APx500

MEASUREMENT SOFTWARE

for all APx series audio analyzers

User's Manual

Audio precision

APx500
User's Manual

for all APx Series audio analyzers
APx500 version 4.5



model APx555

Audio precision

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Section I: Introduction

1

General Information

Welcome to APx500

APx500 version 4.5 software

APx500 is the measurement software used with Audio Precision APx audio analyzers. APx500 version 4.5 is a major release that brings both software innovations and support for the hardware improvements to the APx platform available at the 4.5 release.

See Chapter 3 for a tour of the APx analyzer software.

APx Analyzer Systems

Each member of the APx family of analyzers is a system comprised of a precision measurement instrument (the APx500 analyzer hardware) attached to a personal computer (PC) running APx500 measurement software. The instrument hardware provides analog and digital interface and system clocking. The APx500 software provides configuration, measurement, automation and reporting capabilities.

Hardware

Analyzer Instrument Hardware

APx500 series instruments include:

- the compact, fixed-configuration 2-channel APx515
- the 2-channel APx525
- the 2-channel output / 4-channel input APx526
- the 2-channel APx555. The APx555 analog input and output circuitry outperforms that of any other audio analyzer sold.
- the 2-channel output / 8-channel input APx582
- the 8-channel output / 8-channel input APx585, and
- the 8-channel output / 16-channel input APx586.

See your analyzer's *Installation Instructions and Specifications* booklet for safety, installation, and fusing information and personal computer hardware and operating system requirements.

Hardware Options

For the entire APx525/APx585 family:

- The **APx-ADIO** Advanced Digital I/O option (standard in the APx555).
- The **APx-DSIO** Digital Serial I/O option.
- The **APx-HDMI** HDMI+ARC option
- The **APx-BT-DUO** Bluetooth wireless technology option
- The **APx-PDM** Option
- The **Advanced Master Clock** option (standard in the APx555).

The ADIO and the latest versions of the DSIO and PDM hardware modules support jitter generation and measurement when co-installed with the Advance Master Clock. Contact your Audio Precision representative for information.

For instruments fitted with the 2-channel output module, including the APx582:

- The **APx-AG52** analog generator option adds square wave generation and other improvements (standard equipment in the APx555).

For the instruments fitted with the 2-channel input module:

- The **APx-BW52** high bandwidth option provides up to 1 MHz high-resolution analog input bandwidth (standard equipment in the APx555).

Auxiliary switchers, filters, the APx1701 Transducer Test Interface, and the DCX-127 Multifunction Module are also available for all instruments.

See Chapter 2 for a tour of the APx analyzer hardware.

Analyzer Output signals

Analog signals

Audio output signals are, with some notable exceptions, generated in DSP in APx500 on the PC, and streamed down the USB cable to precision analog output circuits. The exceptions are the low-distortion analog sine generated in the APx555, the analog square wave generated in the APx555 and AG52, and Output

Equalization. These are implemented in instrument hardware.

Digital signals

Digital signals are also generated in DSP in the PC and streamed to a choice of digital interface transmitters in the instrument hardware.

Analyzer Input signals

Analog signals

Audio input signals are acquired using precision interfaces and eventually digitized. The data goes through some initial processing in the hardware, and then is streamed up the USB cable to the PC. The measurements are performed in APx500 DSP, using, for the most part, FFT technology. For THD+N measurements, the APx555 adds an analog notch filter implemented in instrument hardware.

Digital signals

Digital signals are input via a choice of digital interface receivers in the instrument hardware, and streamed to the PC for analysis in DSP.

Software

APx500 software supports all of the instruments in the APx family. APx can be addressed from Sequence Mode or from Bench Mode.

Getting Started

To make a measurement, you must have an APx analyzer attached to the PC. APx500 software will also run without hardware in "demo mode." No measurements can be made in demo mode, but you can explore the user interface and view the APx features and help content.

For complete installation, safety and fusing information and for detailed specifications, refer to the *Installation Instructions and Specifications* booklet provided with your instrument and available online at ap.com.

PC Requirements

APx500 requires a PC with the following characteristics and capabilities:

- Operating system: Microsoft Windows 10, Windows 8, or Windows 7.
- A multi-core processor (at least dual-core) running at a clock speed of at least 2 GHz. Most current processors from Intel and AMD meet these requirements. Note: the Intel Atom processor does not meet our minimum specification.
- At least 2 GB of RAM. More is recommended.
- At least 300 MB of free hard disk space.
- A CD-ROM optical disc drive.
- A USB 2.0 port; two are required for optional switcher use.
- A color monitor and a video card with at least VGA capabilities. Video resolution of 1024 x 768 or greater is recommended.

System performance is sensitive to processor speed; faster processors will yield faster results.

APx500 is data intensive and it is recommended that other data-intensive applications not be run concurrently. This includes Audio Precision's AP2700, APWIN or ATS.

Embedded Help

You can access Help from the Help menu or by right-clicking and selecting Help, or pressing F1 from a context within the application. The Help system includes a Table of Contents, Index and Search capabilities.

Documentation and Support

The *APx500 User's Manual* (this document) is the primary documentation for the APx500 Series, in conjunction with the embedded help system accessible from within the application. The manual is provided as a PDF for the APx515 and as a PDF and a printed book for APx52x/555/58x users. Other documentation includes:

- *APx515 Installation Instructions and Specifications*
- *APx52x/58x Family Installation Instructions and Specifications*
- *APx555 Installation Instructions and Specifications*
- APx-DVD1
  APx-DVD1 is a playable video DVD with menu-driven linear and coded audio test signals for external source use with DVD players.
- APx-CD1
  APx-CD1 is a playable audio CD with linear audio test signals for external source use with CD players.

Audio Precision Web site at ap.com

For more information, go to ap.com. You'll find manuals, audio test signals, Performance Check, utilities, APx sample projects, VB.NET / C#.NET / LabVIEW samples and more.

You can also contact our Technical Support staff at techsupport@ap.com, or by telephoning 503-627-0832 ext. 4, or 800-231-7350 ext. 4 (toll free in the U.S.A.).

Quick Tour: APx Analyzer Hardware



APx555 2-channel advanced analog output, 2-channel advanced analog input, with ADIO, DSIO, HDMI and Bluetooth Duo.

Introduction

The Audio Precision APx Series audio analyzers and the APx500 measurement software bring a fast, intuitive measurement-centric approach that is changing how audio test and measurement is done. This chapter provides a brief overview of the system hardware and will point you to more information.

For safety, fusing or installation information, and for detailed specifications for any of the following instruments, see the corresponding APx Installation and Specifications Booklet.

APx515

The APx515 is a fixed-configuration audio analyzer perfect for both bench and production test.



Analog I/O

The APx515 has two analog output channels, available with both balanced and unbalanced connections; and two analog input channels, each balanced or unbalanced.

Digital I/O

The APx515 is fitted with AES3/SPDIF digital input and output on balanced electrical, unbalanced electrical and optical connectors.

See the *APx515 Installation Instructions and Specifications* booklet for full specifications.

Rear Panel

Power entry module

The power entry module is mounted on APx515 rear panel, and provides the mains power switch, the connector for the mains power cable, the mains voltage selector and the line fuse assembly.



Power and fuse ratings are printed below the power entry module. See the *APx515 Installation Instructions and Specifications* booklet for more information about mains power and fusing.

PC interface

The PC interface connector provides a USB 2.0 interconnection to the controlling personal computer.

Aux control

The two Aux Control connectors (sometimes referred to as GPIO or General Purpose Input/Output) provide the capability to communicate with external devices, transmitting and receiving control commands. See Chapter 102 for more information about Aux Control.

Software options for the APx515

The APx515 enables a basic set of APx500 features and measurements in Sequence Mode. This set can be expanded to the full complement of instrument-appropriate features and measurements (including Bench Mode) by the installation of iButton software option keys. Factory-keyed iButtons are mounted in the APx software option module (pn BSW0.0000). The software option module is connected to the Software Options connector on the APx515 rear panel. Go to page 189 for more information about software options in the APx515.

APx52x/58x series

The APx52x/58x series of audio analyzers provides a modular framework for a wide selection of input/output configurations. Analog I/O is available with as few as 2 input and 2 output channels, up to 8 output and 16 input channels. The standard DIO module provides AES3/SPDIF digital input and output, to which you can add DSIO (chip-level digital serial interface), HDMI+ARC, PDM or Bluetooth wireless technology, provided there is an open module bay in the lower tier.

See the *APx52x/58x Series Installation Instructions and Specifications* booklet for full specifications.

Power switch and indicator light

The master power switch and the power indicator light for the APx52x/58x family are located in the blue module at the right end of the front panel.

APx525



The APx525 is the classic 2-channel configuration, fitted with a Model 103/203 2-channel analog output module, a Model 104/204 2-channel analog input module, and the Model 110/210 DIO module. Read about these I/O modules in detail in the next few pages.

APx526



The APx526 is a 4-channel (analog input) version of the APx525, with 2 analog outputs and 4 analog inputs.

On the analog side, the APx525 has balanced and unbalanced analog inputs and outputs, relay-implemented I/O loopback and common-mode rejection measurements. The digital I/O provides AES3 balanced input and output on XLR connectors, SPDIF/SMPTE unbalanced I/O on BNCs, and optical I/O on Toslink connectors, at sample rates up to 216 kHz. All APx500 generation and measurement features that are applicable to a 2-channel instrument are supported.

APx582



The APx582 is a popular combination of the analog output features of the APx525, and the analog input features of the APx585, described below.

APx585



The APx585 is the world's first true multichannel audio analyzer.

The typical configuration features 8 simultaneous channels of balanced or unbalanced analog inputs and outputs with 216 kHz AES3, SPDIF, and optical digital I/O.

APx586

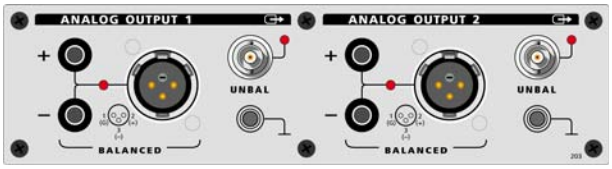


The APx586 is similar to the APx585, but with 8 additional analog input channels on a Model 106/206

module. This provides a total of 16 analog input channels.

Analog I/O for the APx52x/58x series

2-channel analog outputs



The Model 103/203 2-channel analog output module is used in the APx520, 521, 525, 526 and 582 analyzers.

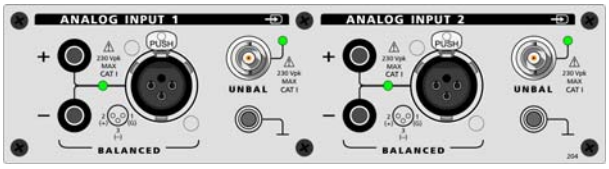
The module has two unbalanced analog output channels available on BNC connectors. Each unbalanced output has selectable source resistances and a maximum output level of 10.61 Vrms.

The module also has two balanced analog output channels available on XLR-M and dual-banana connectors (connected in parallel). Each balanced output has selectable source resistances and a maximum output level of 21.21 Vrms.

AG52 analog generator option

The AG52 analog generator option is available for the APx525 and 526 analyzers, and is provided as standard equipment in the APx555 and APx582 analyzers. AG52 improves THD+N residual performance, increases maximum generator amplitude (26.66 Vrms balanced, 13.33 Vrms unbalanced), and adds a fast rise time square wave generator for general purpose testing and for generating a DIM/TIM distortion stimulus.

2-channel analog inputs



The Model 104/204 2-channel analog input module is used in the APx525 and 526 for analog inputs 1 and 2. The electronically identical Model 107/207 module is used in the APx526 for Analog Inputs 3 and 4.

The module has two unbalanced analog input channels available on BNC connectors. Each unbalanced input can be DC or AC coupled, with selectable termination resistances and a maximum input level of 160 Vrms.

The module also has two balanced analog input channels available on XLR-F and dual-banana connectors (connected in parallel). Each balanced input can be DC or AC coupled, with selectable termination resistances and a maximum input level of 300 Vrms. Two ground lugs are provided.

BW52 high-bandwidth input option

The BW52 high bandwidth analog input option is available to the APx525 and 526, and the performance features of the BW52 are standard in the APx555. The BW52 expands the analyzer bandwidth of analog inputs 1 and 2 to up to 1 MHz, providing unparalleled 1 MHz 24-bit FFTs.

8-channel analog outputs



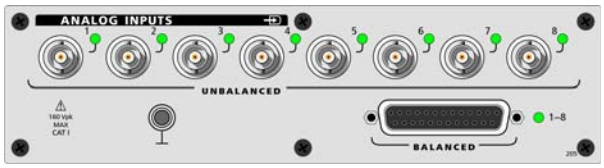
Unbalanced

The Model 101/201 is used in the APx585 and 586 analyzers. The module has eight unbalanced analog output channels available on floating BNC connectors. Each unbalanced output has a source resistance of 50 Ω and a maximum output level of 7.2 Vrms.

Balanced

The module also has eight balanced analog output channels available on a single 25-pin D-Sub connector; see page 11. Each balanced output has a source resistance of 100 Ω and a maximum output level of 14.40 Vrms. A ground lug is provided.

8-channel analog inputs



The Models 105/205 and 106/206 8-channel modules are functionally identical, their only difference being front-panel nomenclature. Model 105/205 is used on the APx582, 585 and 586 for input channels 1–8, while Model 106/206 is used on the APx586 for channels 9–16.

Unbalanced

Each module has 8 unbalanced analog input channels available on floating BNC connectors. Each unbalanced input is DC coupled, with a termination resistance of 100 kΩ and a maximum input level of 115 Vrms.

Balanced

Each module also has 8 balanced analog input channels available on a single 25-pin D-Sub connector; see page 11. Each balanced input is DC coupled, with a termination resistance of 100 kΩ, each leg to ground, and a maximum input level of 115 Vrms.

APx555



The APx555 is the most versatile, highest-performance audio analyzer Audio Precision has ever made. Built on the scaffolding of the APx525, the APx555 includes an analog High Performance Sine Generator and a hybrid High Performance Sine Analyzer, which together drop the residuals in a THD+N measurement to unmatched lows. The High Performance Sine Generator is of extraordinarily low distortion and noise, and has a maximum frequency of 204.75 kHz, well beyond the 80.1 kHz maximum of the DAC generator.

The Advanced Master Clock is standard on the APx555, and provides jitter generation and analysis capability when used with a supporting jitter-enabled digital module (ADIO and current versions of DSIO and PDM). The Advanced Master Clock also provides an AES11 Digital Audio Reference Signal (DARS) reference in and out, a clock sync in and out, and a trigger in and out, all on a connector panel on the rear of the instrument.

The Advanced Digital Input/Output (ADIO) is a two-channel AES3/SPDIF interface with balanced, unbalanced and optical I/O, which is similar to the Digital Interface Module (DIO) module but with additional features. ADIO adds metadata impairment, digital interface signal impairment, digital interface common mode measurement, and the provides digital interface level data to the analyzer. ADIO includes jitter generation and analysis capability when used with the Advanced Master Clock.

All APx500 generation and measurement features that are applicable to a 2-channel instrument are supported.

Analog I/O for the APx555

2-channel analog outputs

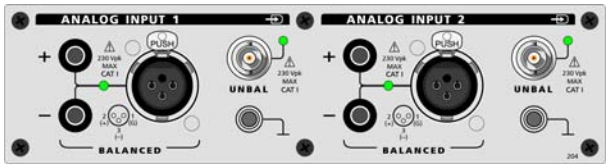


The Model 203 2-channel analog output module is also used in the APx555 analyzers, but the high performance features of the AG52 option are included, with the fast rise time square wave generator for general purpose testing, generating a DIM/TIM distortion stimulus and sine burst signals. Additionally, the High Performance Sine Generator provides an extraordinarily pure sine stimulus at frequencies as high as 204 kHz.

The module has two unbalanced analog output channels available on BNC connectors. Each unbalanced output has selectable source resistances and a maximum output level of 13.33 Vrms.

The module also has two balanced analog output channels available on XLR-M and dual-banana connectors (connected in parallel). Each balanced output has selectable source resistances and a maximum output level of 26.66 Vrms.

2-channel analog inputs



The Model 104/204 2-channel analog input module is also used in the APx555 analog inputs. The features of the BW52 option are included, expanding the analyzer bandwidth of the analog inputs up to 1 MHz, providing unparalleled 1 MHz 24-bit FFTs. Additionally, the High Performance Sine Analyzer uses stereo analog notch filters with a dedicated ADC to provide a system THD+N residuals specification unmatched in the industry.

The module has two unbalanced analog input channels available on BNC connectors. Each unbalanced input can be DC or AC coupled, with selectable termination resistances and a maximum input level of 160 Vrms.

The module also has two balanced analog input channels available on XLR-F and dual-banana connectors (connected in parallel). Each balanced input can be DC or AC coupled, with selectable termination resistances and a maximum input level of 300 Vrms. Two ground lugs are provided.

Digital I/O for APx modular analyzers

Advanced Digital I/O



The Model 219 Advanced Digital I/O module with AES3 interface is the standard digital interface for the APx555 audio analyzer, and is an option for the APx52x/58x instruments.

This module provides both digital input and output on balanced electrical, unbalanced electrical and optical connectors.

Impairments and interface measurements

The Advance Digital I/O differs from the standard Digital I/O in that it offers impairments to the transmitted digital interface signal, and the ability to measure certain characteristics of the received interface signal.

The transmitted interface signal can be impaired with false metadata, validity and parity errors, jitter, noise, common mode and risetime impairments (including cable simulation), to test the tolerance of a downstream device. Measured input interface voltage level can be measured and displayed.

Jitter

The transmitted interface signal can be impaired with jitter, and received jitter can be analyzed in great detail.

Balanced

The balanced interface satisfies AES3, EBU-3250 and IEC60958-4.

The balanced interface has a nominal carrier level of 5 Vpp. Output source resistance is nominally 110 Ω. Input termination resistance can be set to 110 Ω or off (~2 kΩ). When the balanced output connector is selected, the APx500 software embeds Professional status bits in the output signal.

Unbalanced

When the unbalanced output is set to Consumer mode (the default), the unbalanced output is compatible with SPDIF and satisfies IEC60958-3. Nominal carrier level is 0.5 Vpp and output status bits are set to Consumer.

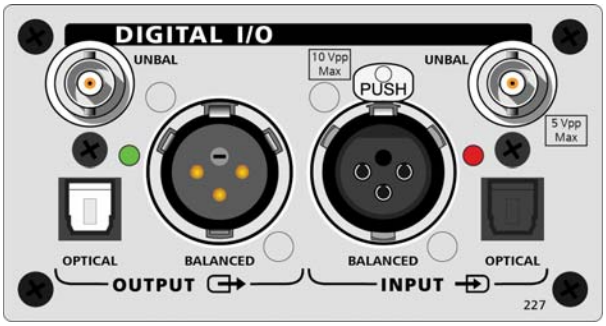
When the unbalanced output is set to Professional mode (see Advanced Settings), the unbalanced output satisfies AES3-id and SMPTE-276M, while remaining largely compatible with IEC-60958-3 and most SPDIF receivers. Nominal carrier level is 1.0 Vpp, and output status bits are set to Professional.

Output source resistance is nominally 75 Ω. Input termination resistance can be set to 75 Ω or OFF (~23 kΩ).

Optical

Compatible with the Toslink interface. When the optical output connector is selected, output status bits are set to Consumer.

Digital input/output



The Digital I/O module (panel nomenclature 227) is the standard DIO for the APx52x and APx58x series of audio analyzers. Earlier analyzers may have been fitted with a DIO module carrying model nomenclature 210 or 110, which have identical electrical features and specifications.

This module provides both digital input and output on balanced electrical, unbalanced electrical and optical connectors.

NOTE: Very early APx585 and 586 instruments were fitted with a Model 109 Digital I/O module, which was not equipped with balanced digital input or output connectors.

Balanced

The balanced interface satisfies AES3, EBU-3250 and IEC60958-4.

The balanced interface has a nominal carrier level of 5 Vpp. Output source resistance is nominally 110 Ω. Input termination resistance can be set to 110 Ω or off (~2 kΩ). When the balanced output connector is selected, the APx500 software embeds Professional status bits in the output signal.

Unbalanced

When the unbalanced output is set to Consumer mode (the default), the unbalanced output is compatible with SPDIF and satisfies IEC60958-3. Nominal carrier level is 0.5 Vpp and output status bits are set to Consumer.

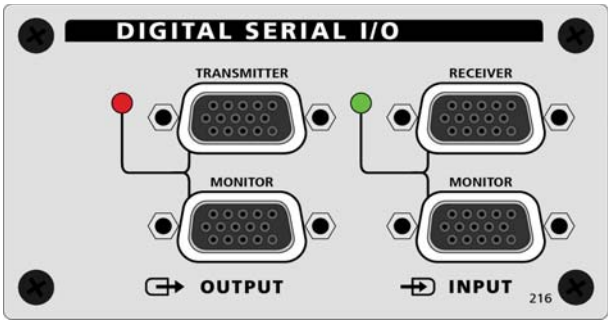
When the unbalanced output is set to Professional mode (see Advanced Settings), the unbalanced output satisfies AES3-id and SMPTE-276M, while remaining largely compatible with IEC-60958-3 and most SPDIF receivers. Nominal carrier level is 1.0 Vpp, and output status bits are set to Professional.

Output source resistance is nominally 75 Ω. Input termination resistance can be set to 75 Ω or OFF (~23 kΩ).

Optical

Compatible with the Toslink interface. When the optical output connector is selected, output status bits are set to Consumer.

Digital Serial I/O option



DSIO

The Digital Serial Input/Output (or DSIO) option provides a flexible chip- or board-level serial input and output interface.

With separate Master Clock, Bit Clock, Frame Clock, Channel Clock and four Data lines, variable signal formats, variable word width, bit depth and synchronization options, the DSIO can address almost any serial interface need.

Formats include TDM, I^2S, DSP (bit-wide pulse) and custom formats. Up to 16 channels can be transmitted and received using the TDM format.

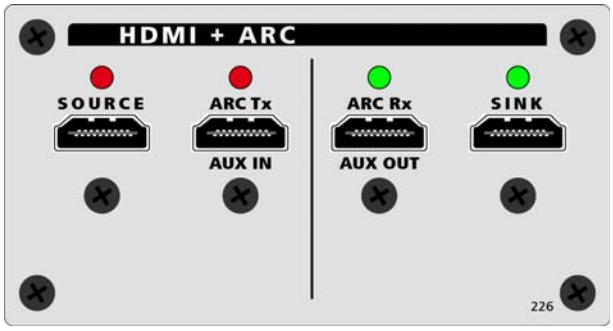
The DSIO option can be ordered with APx525, 526, 582, 585 or 586 instruments, provided there is an open module bay in the lower tier.

Jitter

If the analyzer is fitted with the Advanced Master Clock (standard on the APx555), the current version of the DSIO supports jitter generation and measurement.

Operation of the DSIO is discussed in Chapter 15.

HDMI+ARC option



The APx525, 582 and 585 can be fitted with an optional model 114/214 HDMI+ARC (High Definition Multimedia Interface plus Audio Return Channel) I/O module.

NOTE: Earlier APx585 instruments may be fitted with a Model 112 HDMI module, which does not include ARC support.

About HDMI+ARC

The model 114/214 HDMI+ARC module is fully compatible with HDMI 1.3a; additionally, it supports a subset of HDMI 1.4a, the ARC (Audio Return Channel) feature.

HDMI (High Definition Multimedia Interface) is designed to carry high-bandwidth digital streams providing an audio/video interface that includes content protection and a bi-directional channel for interaction with connected electronic devices.

ARC (Audio Return Channel) provides an additional digital audio channel, which can simplify interface cabling in certain applications, for user convenience.

HDMI+ARC Hardware description (214)

The HDMI option is fitted with four HDMI Type A connectors:

- The **Source** connector outputs APx-generated monocolor video with APx generated audio and HDCP. Alternatively, user video (from Aux In) can be substituted for monocolor video. This connector is inactive when the Signal Path Setup Output Configuration is set to HDMI ARC Tx.
- The **ARC Tx / Aux In** connector provides input for user video (such as a test pattern) to be substituted for APx-generated video available at Source connector.
  When Signal Path Setup Output Configuration is set to **HDMI ARC Tx**, this function is disabled; instead, the connector outputs digital audio and CEC negotiation in support of HDMI ARC.
- The **ARC Rx / Aux Out** connector provides a monitor function, outputting the video and audio present at the Sink connector. EDID metadata from downstream device (typically an HDTV monitor) can be selectively blocked (the default) or passed to the upstream Device Under Test.
  When Signal Path Setup Input Configuration is set to **HDMI ARC Rx**, this function is disabled; instead, the connector inputs digital audio in support of HDMI ARC.
- The **Sink** connector provides measurement input for HDMI from the Device Under Test.
  This connector is inactive when the Signal Path Setup Input Configuration is set to HMI ARC Rx.

CEC

For new instruments and upgrades sold after September, 2013, CEC functionality has been added to Source and Sink connections. ARC Tx and ARC Rx CEC functionality has been expanded. APx500 version 3.4 or later is required. See More About CEC on page 126.

HDMI (112)

Older APx585 instruments may have been fitted with the model 112 HDMI module, which requires APx500 v2.2 or later.

The model 112 HDMI module is fully compatible with HDMI 1.3a.

The front panel is fitted with four Type A HDMI connectors:

- The **Source** connector outputs APx-generated monocolor video with APx generated audio and HDCP. Alternatively, user video (from Aux In) can be substituted for monocolor video.
- The **Aux In** connector provides input for user video (such as a test pattern) to be substituted for APx-generated video available at Source connector.
- The **Aux Out** connector provides a monitor function, outputting the video and audio present at the Sink connector. EDID metadata from downstream device (typically an HDTV monitor) can be selectively blocked (the default) or passed to the upstream Device Under Test.
- The **Sink** connector provides measurement input for HDMI from Device Under Test.

Audio on HDMI

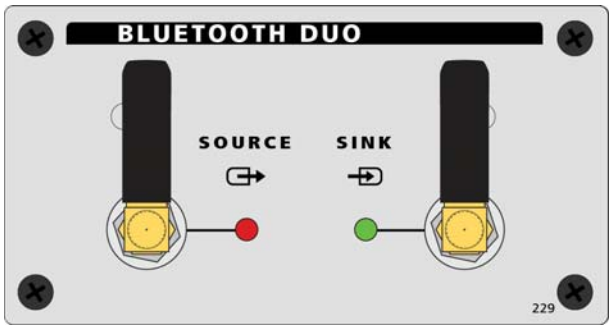
APx500 provides generation and measurement capabilities for audio signals carried on the HDMI interface.

Video is implicit in HDMI, and the AP HDMI interface transmits and receives the required video and metadata signals for proper operation of the HDMI interface. These video and metadata signals can be altered in APx500 in some ways to examine the effect of bitstream changes on audio quality. However, APx500 does NOT provide any measurement or diagnostic capabilities for HDMI video signals, or for all HDMI metadata.

Audio on ARC

ARC (Audio Return Channel) is a feature of HDMI 1.4a that enables digital audio (IEC 60958 / SPDIF) transmission and reception on existing but previously unused conductors in standard HDMI connectors and cables. See More About ARC on page 125.

APx Bluetooth Duo Option



Hardware description

The panel nomenclature for the Bluetooth Duo module (Help > About > Bluetooth Duo Module > ID: BZL2) is 229. The Bluetooth Duo module can be installed in any APx instrument with an E-type chassis (from mid 2012 and later), provided there is an open module bay in the lower tier. Bluetooth cannot be installed in the APx515.

The Bluetooth Duo module is fitted with two Bluetooth radios, one dedicated to source profiles and one to sink. The SMA connectors on the front of the module provide RF (radio frequency) connectivity for each of the radios. Typically, small antennas are connected at this point. One or two 50 Ω cables can also be connected to route the RF signal to a Bluetooth device with an RF cable connection.

When cabling directly to a device (without using antennae), the use of an RF level attenuator of approximately 50 dB is recommended, to be connected between the APx and the device.

Bluetooth codecs

Bluetooth wireless technology uses codecs to compress the audio to usable data rates.

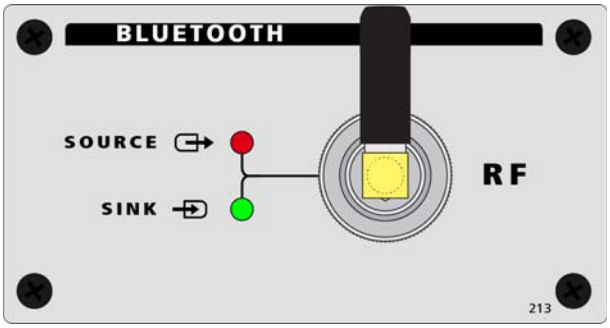
The Audio Precision Bluetooth Duo module supports the CVSD, SBC, mSBC (wideband speech), and AAC codecs, and Qualcomm® aptX™, aptX™ HD, and aptX™ Low Latency audio codecs. Earlier legacy APx Bluetooth modules (APX-BT and APX-BT-WB) are supported by the current APx500 software, but support fewer codecs than Bluetooth Duo.

Bluetooth Duo firmware updates

The Bluetooth Duo option module supports field firmware upgrades.

To check the firmware version and codec support for your Bluetooth Duo module, in APx500 go to Help > About > Bluetooth Duo Module.

APx legacy Bluetooth Option



The legacy Bluetooth Option module is no longer available, but APx500 continues to support this legacy hardware.

Legacy Bluetooth hardware description

The panel nomenclature for the legacy Bluetooth modules (Help > About > Bluetooth Module > ID: BAZL) is 113. The panel nomenclature for the legacy Bluetooth modules (Help > About > Bluetooth Module > ID: BAZW) is 213. The legacy Bluetooth modules can be installed in any APx instrument, provided there is an open module bay in the lower tier. Bluetooth cannot be installed in the APx515.

Typically, a small antenna is connected to the N connector on the Bluetooth front panel. A 50 Ω cable can

also be connected to route the RF signal to a Bluetooth device with an RF cable connection.

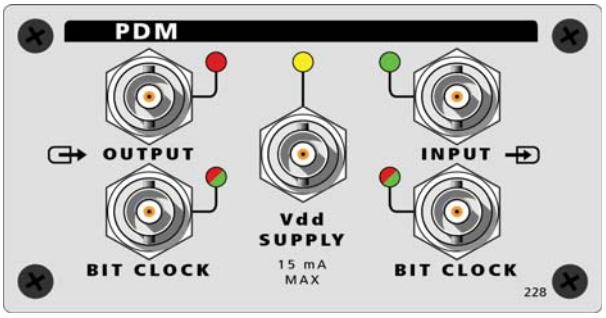
When cabling directly to a device (without using antennae), the use of an RF level attenuator of approximately 50 dB is recommended, to be connected between the APx and the device.

Legacy Bluetooth codecs

Bluetooth wireless technology uses codecs to compress the audio to usable data rates.

Audio Precision BAZL legacy Bluetooth modules support the SBC and Qualcomm® aptX™ audio codecs. BAZW modules also support the mSBC (wideband speech) codec.

APx PDM Option



The PDM option provides a complete solution for addressing circuits or devices with a PDM input or output. The nomenclature for the current PDM option module is 228; the previous module is 217. The PDM module can be installed in any APx instrument except the APx515, provided there is an open module bay in the lower tier.

Output

The PDM Option provides a PDM signal output, which consists of an APx generator audio signal, interpolated by a broad choice of oversampling ratios, and modulated into a 1-bit PDM bitstream. A 4th-order modulator is the default; a 5th-order modulator can be selected. Amplifier Control Codes can be imposed upon the PDM output signal.

Bit Clock

A PDM transmitter (output) typically acts as a slave, with the downstream receiver acting as master. Therefore, in the APx implementation the Bit Clock connection associated with the PDM Output is, by default, configured as a clock input; however, it can also be configured as a clock output by changing the clock direction in software. The LED associated with the Bit Clock connector indicates the clock direction: red when Bit Clock is a clock output, green when it is a clock input.

Bit clock frequencies can range from 128 kHz to 24.576 MHz, depending upon the interpolation rate (baseband audio sample rate) and oversampling ratio. This constrains the Decimated Rates available at higher interpolation ratios.

Input

The PDM Option also provides a signal input with its associated clock connection. The input accepts a 1-bit PDM bitstream, which is then decimated by one of a wide range of oversampling ratios and filtered into baseband audio at the Decimated Rate. The input bitstream can also be analyzed directly (before decimation) in the Signal Analyzer to view out-of-band components. Amplifier Control codes in the PDM input signal can be monitored.

Bit Clock

A PDM receiver (input) typically acts as a master, with the upstream transmitter acting as a slave. Therefore, in the APx implementation, the Bit Clock connection associated with the PDM Input is, by default, configured as a clock output; however, it can also be configured as a clock input by changing the clock direction in software. The LED associated with the Bit Clock connector indicates the clock direction: red when Bit Clock is a clock output, green when it is a clock input.

Bit clock frequencies can range from 128 kHz to 24.576 MHz, depending upon Decimated Rate (baseband audio sample rate) and oversampling ratio.This constrains the Decimated Rates available at higher decimation ratios.

Jitter

A PDM model 228 module installed in an APx with the Advanced Master Clock can apply jitter to the outgoing bit clock, and can analyzer jitter on the incoming bit clock.

Vdd Supply

PDM MEMS microphones devices are typically externally powered. The Vdd Supply connector can provide DC current up to 15 mA, with a voltage range of 0 V to 3.60 V.

PSR testing

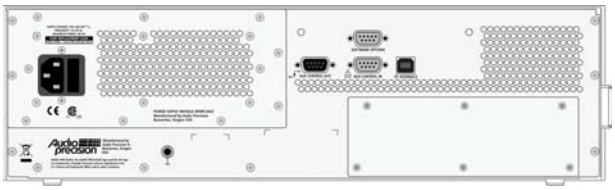
AC signals can be imposed upon the Vdd supply dc voltage to test the PSR (Power Supply Rejection) in the DUT. This connection can also be used for PSRR testing with certain analog DUTs. Use PSR or PSR Sweep to measure the PSR or PSRR.

APx52x/555/58x family rear panel

Power entry module

The power entry module is mounted on the power supply module. The power entry module provides the con-

nector for the mains power cable and the line fuse assembly.



Power and fuse ratings are printed to the right of the power entry module. See the instrument Installation and Specifications booklet for more information about mains power and fusing.

PC interface

The PC interface connector provides a USB 2.0 interconnection to the controlling personal computer.

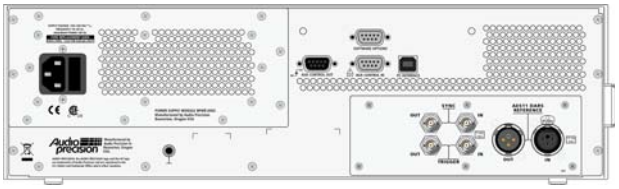
Configuration label

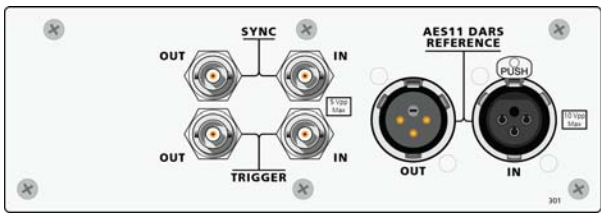
Configuration, serial number and date of manufacture are listed on the configuration label.

Aux control

The two Aux Control (sometimes referred to as GPIO or General Purpose Input/Output) connectors provide the capability to communicate with external devices, transmitting and receiving control commands. See Chapter 102 for more information about Aux Control.

Advanced Master Clock





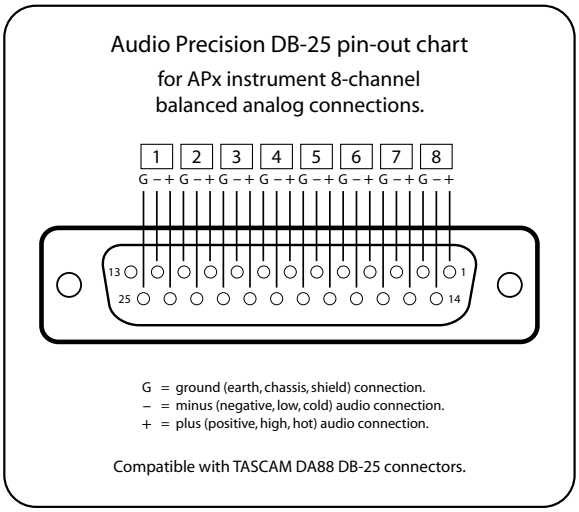
Detail, Sync-Ref panel

Analyzers fitted with the Advanced Master Clock (standard on the APx555 and optional on other models) will have Sync, Trigger and Reference connectors on the rear panel.

The Advance Master Clock also provides jitter generation and measurement capabilities when used with a jitter-compatible digital I/O module.

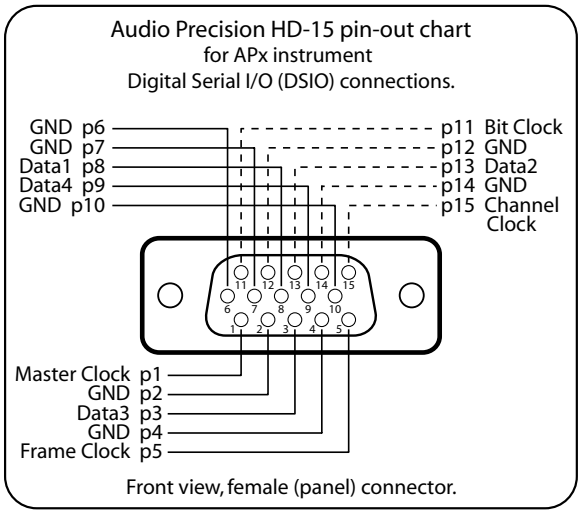
DB-25 Analog Balanced Connector
(used in 8-channel analog I/O modules)

Audio Precision's 8-channel analog input and 8-channel analog output modules use a female DB-25 connector for balanced multichannel inputs and outputs. Pin connections are shown below. Compatible cables and adapters of high performance are available from Audio Precision.



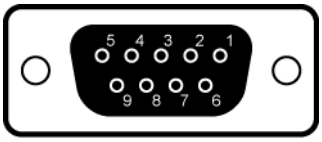
HD-15 Connector
(used in the DSIO module)

The Digital Serial I/O option is fitted with four female HD-15 connectors for multichannel serial interface transmitter, receiver and monitor connections. Pin connections are shown below. Compatible cables are available from Audio Precision.

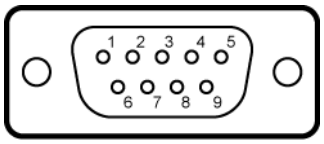


DE-9 connectors
used in all analyzers (rear panel)

Aux Control Out is available on a female DE-9 connector on the rear panel of all APx analyzers. The female DE-9 has the following pin-out:



Aux Control In is available on a male DE-9 connector on the rear panel of all APx analyzers. The male DE-9 has the following pin-out:



Accessories

SWR-2755 switchers



Audio Precision manufactures optional audio switchers that can be used to connect and disconnect many signal channels (up to 192 input and 192 output channels) to the instrument inputs or outputs, under software control. See Chapter 103 for more information about switchers.

APx1701 Transducer Test Interface



The APx1701 Transducer Test Interface is an APx accessory device with both input and output functions. It provides a two-channel power amplifier, microphone powering and impedance characterization circuitry for loudspeaker or microphone acoustic testing.

When connected to an APx system, the APx1701 appears in Signal Path Setup as additional inputs and outputs. You can read more about using an APx1701 in Chapter 20, or in the *APx1701 Installation, Specifications and User's Guide* booklet.

AUX-0025 and AUX-0040 filters



The AUX-0025 and AUX-0040 switching amplifier filters are 2-channel low-pass switching amplifier measurement filters for use with the APx analyzers when testing high slew rate signals, typical of switching amplifier (Class D) outputs. The AUX-0025 passes audio up to 20 kHz; the AUX-0040 extends that bandwidth to 40 kHz.

AUX-0100 filter



The AUX-0100 filter (previously known as the APx581 filter) provides the same filtering as an AUX-0025 in a 8-channel product, and may be a more appropriate choice for use with the APx582, 585 or 586 analyzers.

For more information about the AUX-0025, AUX-0040 or AUX-0100 switching amplifier filters, contact your Audio Precision distributor or representative or visit our Web site at ap.com.

DCX-127 Multifunction Module



The DCX-127 Multifunction Module is an Audio Precision accessory that provides interface and control features not otherwise available in Audio Precision analyzers. Download the *DCX-127 Multifunction Module User's Guide* from the Audio Precision Web site at ap.com for complete operational information for the DCX-127.

The DCX-127 provides a digital multimeter that can be configured to measure DC voltages or resistance; it has two programmable DC voltage sources, and four GPIO Auxiliary Output connections.

3

Introduction to the APx500 Software

Welcome to APx500 v 4.5

APx500 is the measurement software used with Audio Precision APx500 audio analyzers. APx500 analyzers are systems, comprised of a precision measurement instrument (the APx500 analyzer hardware) attached to a personal computer (PC) running APx500 measurement software.

The APx Project

An APx500 project is the set of configurations and instructions for a test and measurement session, with necessary waveform files, data files and images.

A project can be as simple as one brief measurement in the default configuration, or as complex as a series of different signal path configurations encompassing hundreds of measurements directed by an automated sequence and documented in a custom report. See Chapter 4 for more information about APx projects.

Sequence Mode and Bench Mode

Sequence Mode and Bench Mode each have their own workspace. APx normally launches into Sequence Mode. You can switch modes using the selector in the upper right corner of the workspace.

Sequence Mode

This chapter begins with a tour of Sequence Mode on the next page.

Sequence Mode provides easy access to a large number of pre-configured measurements, which can be quickly arranged into an automated sequence that can generate sophisticated reports. For many tests, it's as simple as opening a measurement and clicking Start.

Bench Mode

Bench Mode, on the other hand, is largely unconfigured. Rather than using pre-defined measurements, Bench Mode makes a number of settings and tools available, and allows you to simply adjust and tweak and try different approaches without imposing constraints.

This chapter will look at the key features of Bench Mode beginning on page 20, and will also visit other APx features that are shared by both Sequence Mode and Bench Mode.

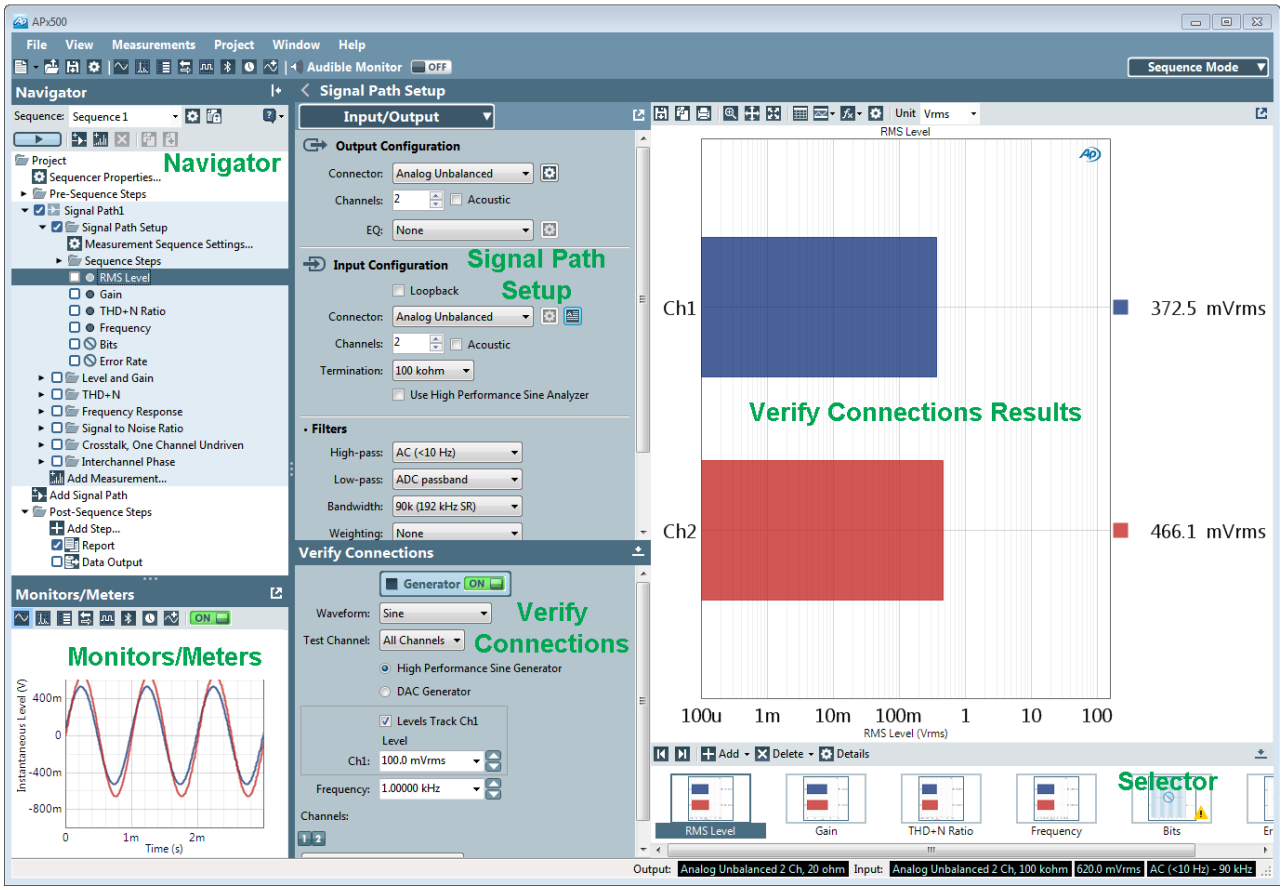
Signal Path Setup

Signal Path Setup includes a menu to select a number of related panels: Input/Output, References, and Switchers. If an Audio Precision DCX-127 is attached, there is a new menu item to control it. If your instrument includes the Advanced Master Clock (AMC), Ref, Sync and Jitter selections are included in the menu.

The References panels, which used to reside in the Navigator, are now consolidated into a menu item in the new Signal Path Setup menu.

The Sequence Mode Workspace: Signal Path Setup

SEQUENCE MODE WORKSPACE (1)



All APx500 workspaces have a Menu bar and a Toolbar at the top, and a Status bar at the bottom.

The Navigator panel is on the left, above the Monitors/Meters panel. The Signal Path Setup menu runs down the center of the workspace, and the measurement Results, Selector and Data Sets grid are on the right.

The Navigator is key to Sequence Mode. Here you add pre-defined measurements to Signal Paths, and select the results you'd like to see. You can create multiple Signal Paths, with different combinations of measurements or even different input/output configurations.

Once you've created a Signal Path or two and populated them with measurements, you can run them all in any order as an automated sequence. You can add stops and prompts and run-time configurations to your sequence, and customize a report to be generated upon completion.

The Signal Path Setup menus

The center panel is Signal Path Setup, and the menu at the top of the panel provides configuration settings for the current signal path. Depending upon the connected APx instrument hardware, the menu items may include

- Input/Output
- References
- Output Switchers
- Input Switchers
- DCX
- Clocks
- Triggers

The Signal Path Setup menu is also available in Bench Mode.

Signal Path Setup is discussed in detail in Chapter 7.

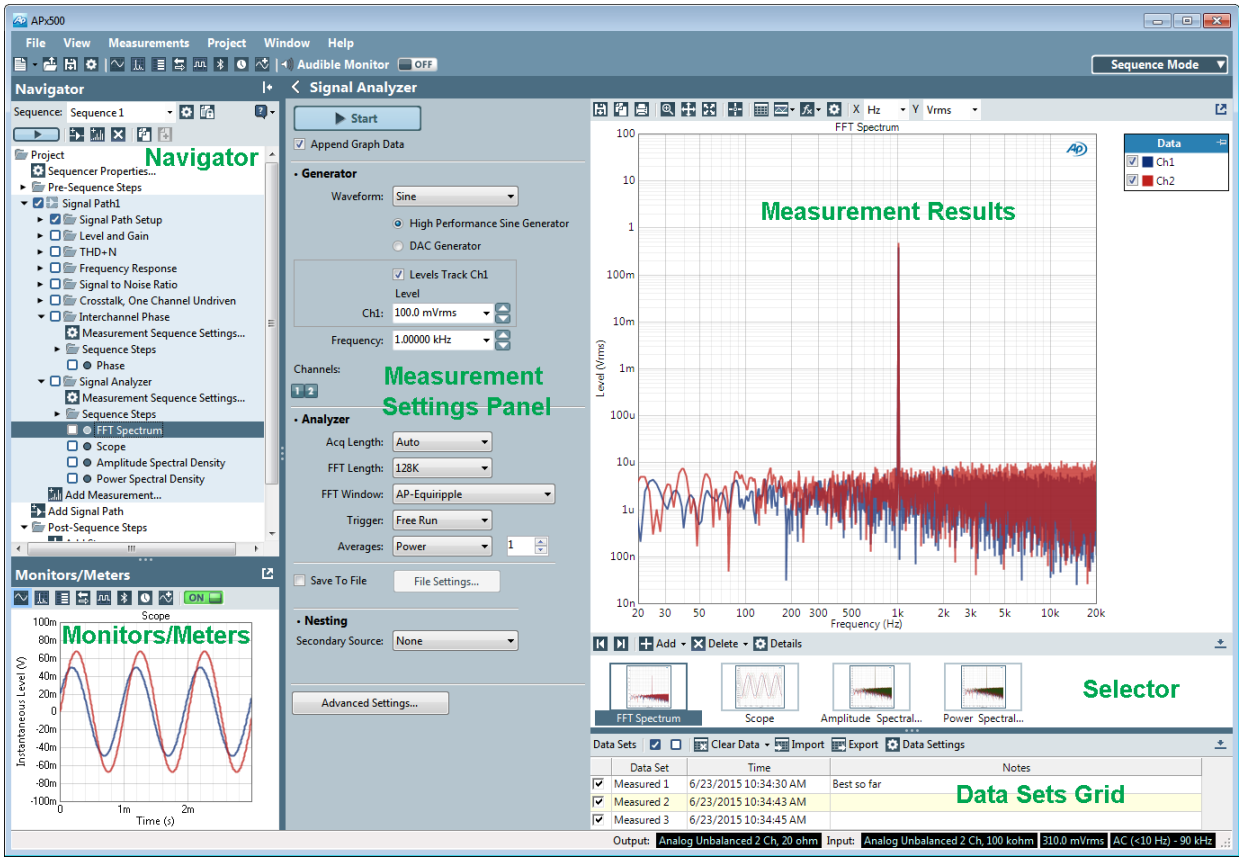
Verify Connections (Sequence Mode only)

Verify Connections provides generator controls and analyzer results that help you to easily check your connections and perform simple diagnostics by sending and receiving signals on the channels you are using.

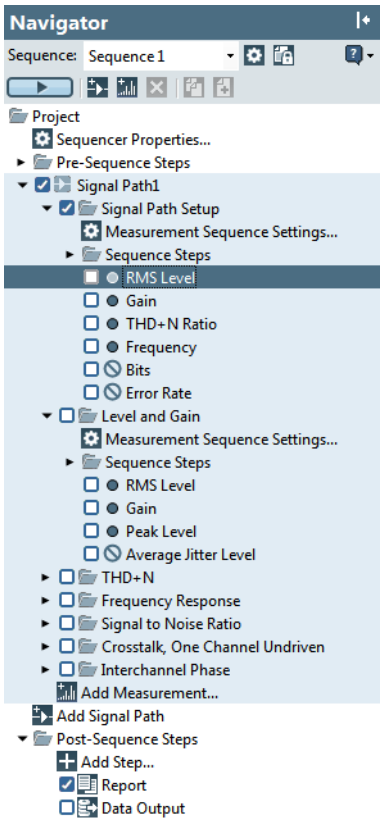
Verify Connections is discussed in detail in Chapter 8.

The Sequence Mode Workspace: Making a Measurement

SEQUENCE MODE WORKSPACE (2)



When you select a Measurement view, Signal Path Setup and Verify Connections disappear, replaced by a Measurement Settings Panel, a Graph Results panel, the Selector; and for some measurements, the Data Sets grid.



The Navigator (Sequence Mode only)

This panel has two key functions:

It serves as a measurement Navigator, where you can add, delete or rename measurements and results, displayed in a tree-and-branch structure.

It also is the interface for the Sequencer, an automation interface where you can select measurements and results, add prompts and build one or more sophisticated automated sequences, all from within the APx500 graphical interface. See Chapter 86 for detailed information about creating and using automated sequences.

Depending upon the context and the task at hand, you may find this panel referred to as the Navigator, or the Sequencer.

The Navigator panel is at the left of the APx500 workspace. Click the **Hide** button to hide the panel; click **Show Navigator** to restore it to view. Note that the show and hide buttons also affect the visibility of the Signal Monitors when they are docked.

What is a Signal Path?

In Sequence Mode, a Signal Path is a combination of input and output format and connector settings, reference levels and accessories settings, and a collection of measurements, measurement settings and measurement results. You can add any number of signal paths to a project. Multiple signal paths are useful to define tests using different paths through a DUT, for example, or different reference levels or measurement settings through the same DUT path. Signal Paths can be renamed to match your testing requirements.

A new project opens with one signal path. You can add, move, rename and delete signal paths from the Navigator panel. Multiple signal paths in a project are useful for:

- testing DUTs that have more than one signal path; for example, analog in / digital in, or 5.1 channels out / stereo out;
- testing the same signal path on a DUT with a different configuration; for example, all levels at 0 dBV versus all levels at –20 dBV;
- cycling through different switcher settings.

A sequence can run through some or all of the signal paths within a project. Use the sequence step checkboxes in the Navigator to add or remove measurements in different signal paths, just as you would add or remove measurements within one signal path.

Remember to change physical connectors as needed when you change signal paths. If a connection change is required in an automated sequence, you should insert a user prompt in the sequence to remind the operator and to provide time to change the connectors.

Measurements (Sequence Mode only)

In Sequence Mode, APx500 provides many different testing tools called *measurements*, such as Signal-to-Noise Ratio or THD+N. Chapter 23, Making Measurements, covers general topics regarding making Sequence Mode measurements. Each Sequence Mode measurement is discussed in detail in Chapters 24 through 85.

You can add any number of measurements to a Signal Path, including more than one instance of the same measurement. You can remove measurements from a Signal Path, and you can rename each measurement to match your testing requirements.

In a new Sequence Mode Project, there are six key measurements initially available:

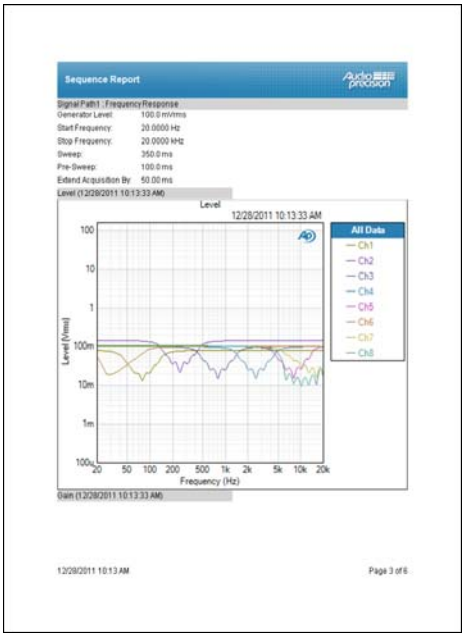
- Level and Gain (see Chapter 56)
- THD+N (see Chapter 84)
- Frequency Response (see Chapter 47)
- Signal to Noise Ratio (see Chapter 79)
- Crosstalk (see Chapter 37)
- Interchannel Phase (see Chapter 53)

Note: some measurements are unavailable in certain configurations, and are marked with a symbol in the Navigator.

Result branches in the Navigator

Each measurement has one or more primary results and optional derived results associated with it. Results are represented in the Navigator tree as branch nodes and are also shown as thumbnails in the Selector. See Chapters 99 and 100 for more information about results.

Reports (Sequence Mode only)



A report is the primary output document of a Sequence, listing setup, measurements and results. A report is created when a Sequence is run, or when a measurement is made using the **Start Measurement** command from the Navigator context menu or the Project menu.

You can create a report showing the results in the current view; you can run a measurement and create a report showing results of all the selected views; or you

can run a sequence and create a report showing the results of all the selected measurements. You can view, edit and format a report, and then export or print it.

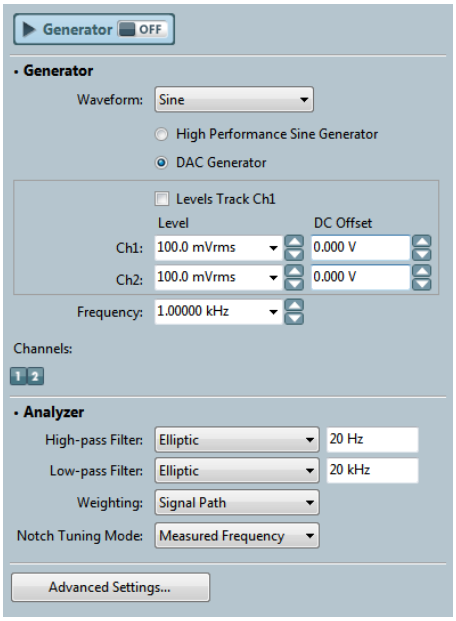
Click on the **Report** node in the Navigator to create a report, or right-click on the node to edit the report's properties. Using APx500's Microsoft Word report feature, you can also create custom reports and report layouts.

See Chapter 87 for more information about reports.

Data Output (Sequence Mode only)

The Data Output feature can save certain sequence results to a CSV file as tabular data. Each time the sequence is run, the target file is opened, and the new data is appended to the table.

The measurement Settings panels (Sequence Mode only)



A typical measurement Settings panel

In a Sequence Mode measurement, the measurement Settings panel runs down the middle of the APx500 workspace. The contents of the panel vary with the measurement or configuration view selected. In general, there are generator settings at the top of the panel (when supported) and analyzer settings (if settings are available) below that.

Start On/Off Button

Most measurements have a **Start** or Generator **On/Off** button, usually at the top of the panel.

Generator settings

Depending upon the measurement selected, the Measurement Settings panel may contain generator waveform source, level, frequency and channel settings. Sweep measurements may include start and stop frequencies or levels, step size, points, logarithmic or linear settings and more. See Chapter 6 for more information about the APx Generator.

Analyzer settings

Depending upon the measurement selected, the measurement Settings panel may contain acquisition and analysis settings including filters, measured or reference channel settings, append settings, regulation controls, FFT settings, reading rate settings and more.

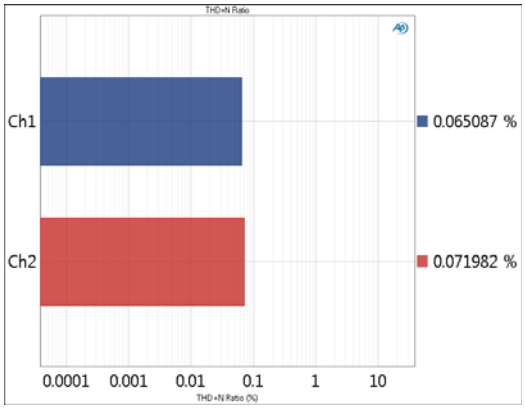
Advanced settings

An Advanced Settings button opens a dialog that allows you to view and set other generator and analyzer settings for the measurement.

Results and the Graph panel

Measurement results are represented as thumbnail views in the measurement's Selector filmstrip display (see page 19) and as branches in the Navigator tree display (page 15).

When a result is selected, the meter bar graph or XY line graph is shown in the main Graph panel (below). The Graph panel can be undocked and repositioned and resized as an independent window. The Selector thumbnails can also be undocked, and become full-featured graph windows.

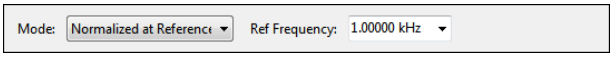


Typical Bar Graph Result

The Result Settings bar

Some measurements present a Result Settings bar between the results Graph panel and the Selector. Different results may offer different controls in the Result Settings bar.

For example, Frequency Response: Relative Level shows the Ref Frequency control, the necessary reference for the relative result view.



Frequency Response: Deviation shows the Min and Max frequency controls.



Adjusting a results setting changes the results view, but this is non-destructive; the original measurement acquisition data are retained.

See Chapter 23 for more information about making measurements.

Working with the Navigator

Moving around in the Navigator

To expand a branch (or *node*) of the Navigator tree, click on a plus-sign expansion box ⊞. To contract an expanded branch, click on a minus-sign contraction box ⊟.

Or, right click on a Navigator node and select **Expand All** or **Collapse**.

Selecting a measurement

To select a measurement, click the measurement branch. The primary result for that measurement will open in the APx500 main Result panel.

Selecting a result

Many measurements have more than one result, represented as sub-branches beneath the Measurement. Click on the sub-branch and the result will open in the APx500 graph panel.

Adding Navigator elements

Add a Signal Path

To add a Signal Path, click on the **Add Signal Path** button at the top of the Navigator or the **Add Signal Path** branch at the bottom of the Project tree. You can also right-click on the Project branch and choose **Add Signal Path** from the context menu.

Add a measurement using a context menu

To add a measurement to a selected Signal Path, right-click in the Navigator area and then choose a measurement from one of the fly-out menus: **Add Measurement, Insert Measurement Before Selection,** or **Insert Measurement After Selection**.

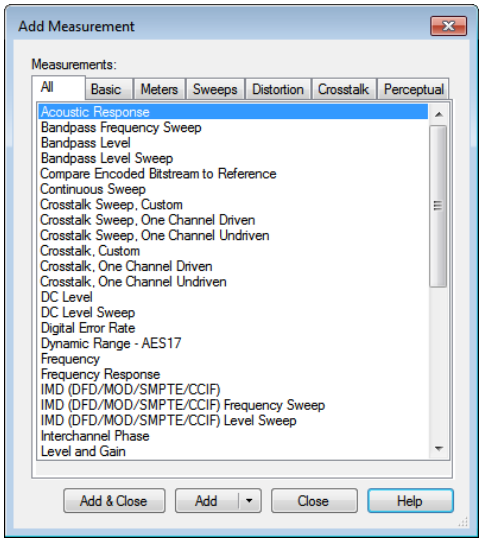
Add a measurement from the Main menu

You can add a measurement from the Main menu. Choose **Project** and then choose a measurement from one of the fly-out menus: **Add Measurement, Insert Measurement Before Selection,** or **Insert Measurement After Selection.**

The Add Measurement dialog

You can open the **Add Measurement** dialog by clicking the **Add Measurement** button at the top of the Navigator or the **Add Measurement** branch at the bottom of the Signal Path tree.

The tabs at the top of the dialog allow you to filter the list. Select one or more measurements and click Add and Close, or choose one of the items on the Add fly-out menu to insert the measurement(s) at a particular place in the signal path.



Typical Add Measurement dialog

Add a result

To add a result to a measurement, right-click on a result and choose **Add Primary Result** or **Add Derived Result**.

Copying and pasting Navigator elements

To copy a Signal Path or a Measurement, right-click on the branch you wish to copy and choose Copy from the context menu. To paste a Signal Path or a Measurement, right-click in the Signal Path you wish to copy into and choose **Paste** from the context menu.

Only one Signal Path Setup branch and Reference Levels branch can exist within each Signal Path. Signal Path Setup and Reference Levels cannot be copied, pasted or deleted.

Moving Navigator elements

You can change the order of Navigator elements by moving the branches within the tree. This can provide a more orderly project, but more importantly it also changes the order of operations within an automated sequence.

To move a Signal Path, select the branch you wish to move and drag it up or down to a new location in the Project.

To move a Measurement, select the branch you wish to move and drag it up or down to a new location within the Signal Path. Measurements cannot be dragged to a different Signal Path. Instead, use the Copy, Paste and Delete functions.

The Signal Path Setup branch and Reference Levels branch are always at the top of

each Signal Path tree. Signal Path Setup and Reference Levels cannot be moved.

Renaming Navigator elements

To rename a Signal Path or a Measurement, click twice on the branch you wish to rename, or right-click on the branch and choose **Rename** from the context menu. The branch name will become available for keyboard editing (indicated by highlighting and a cursor bar).

Deleting Navigator elements

To delete a Signal Path, select the Signal Path branch you wish to remove and click on the **Delete Selected Item** button at the top of the Navigator. You can also right-click on the Signal Path branch you wish to remove and choose **Delete Signal Path** from the context menu.

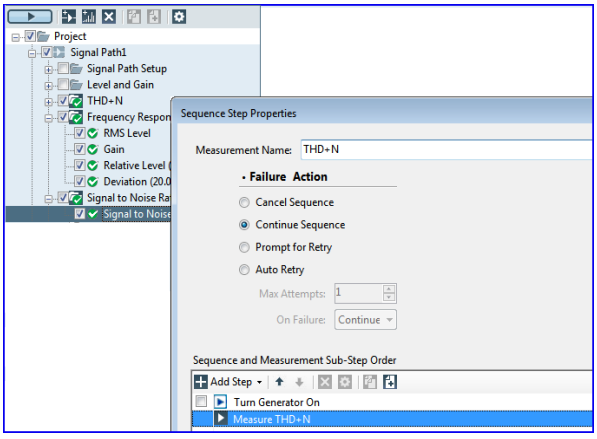
To delete a Measurement, select the Measurement branch you wish to remove and click on the **Delete Selected Item** button at the top of the Navigator. You can also right-click on the Measurement branch you wish to remove and choose **Delete Measurement** from the context menu.

There must be one Signal Path Setup branch in each Signal Path. The Signal Path Setup branch cannot be deleted.

Running a measurement

To run a Measurement, right-click on the branch you wish to run and choose **Run Selected Measurement** from the context menu. This action runs the only the selected measurement and produces a Report for that measurement's results.

Working with the Sequencer (Sequence Mode only)



The Sequencer is an easy-to-use automation feature accessed through the Navigator. Using the mouse, set a green check mark in the measurements you would like to run in a sequence. Click the **Run Sequence** button, and all checked measurements will be performed against limits (if set), and a report will be generated.

Prompts, delays and external routines can be added to the sequence. Pass/Fail results are shown in the Navigator and in the Report.

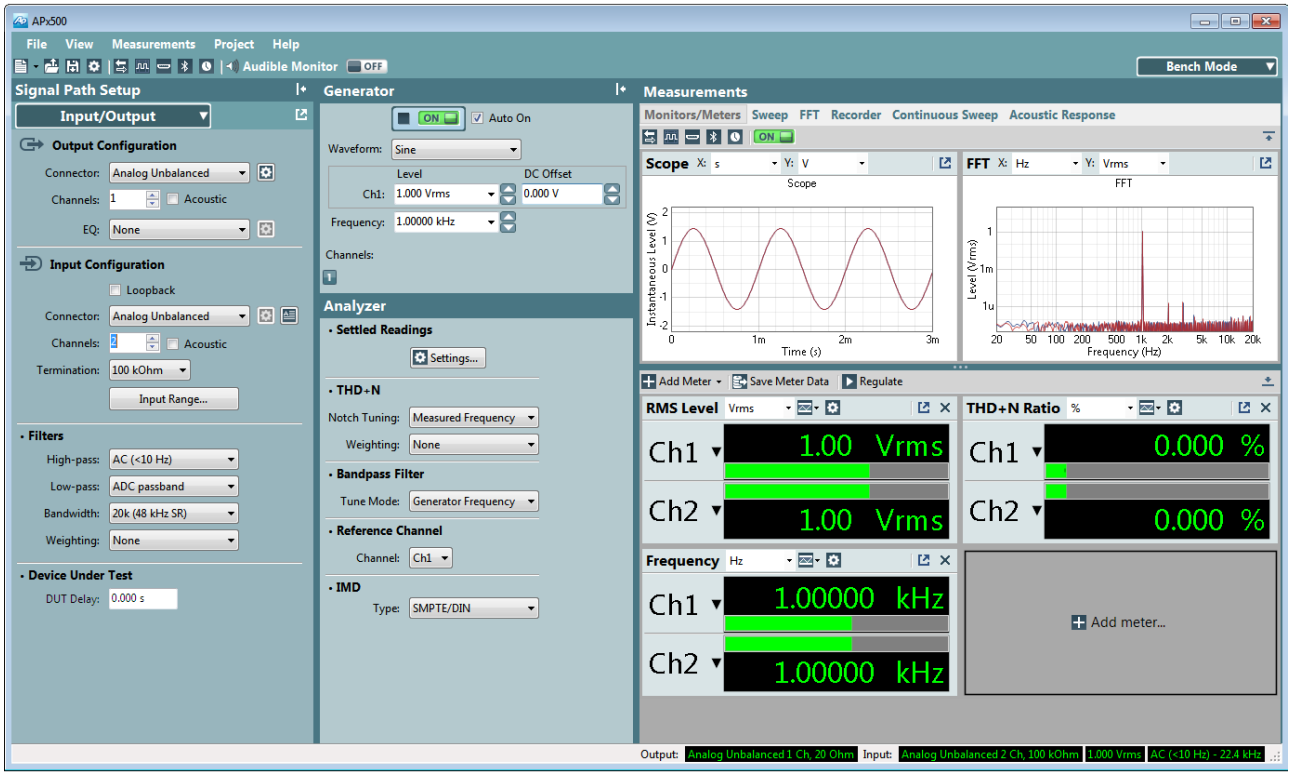
See Chapter 86 for more information about using the Sequencer to automate your measurements.

Running a Sequence

You can select measurements in the Navigator to run in an automated sequence using the APx500 Sequencer, a flexible, powerful and easy-to-use automation tool.

The Bench Mode Workspace

BENCH MODE WORKSPACE



SIGNAL PATH SETUP

GENERATOR AND ANALYZER

MONITORS / METERS AND MEASUREMENT TABS

Bench Mode is an alternative user interface introduced in 2014. Unlike Sequence Mode, Bench Mode does not offer defined measurements, or multiple signals paths or sequences. Instead, Bench Mode provides a set of tools that can be assembled and used in many different ways. AP2700 and APWIN users will recognize similarities between the Bench Mode paradigm and earlier Audio Precision control software.

The Bench Mode Workspace consists of three columns. The column to the left contains a collection of Signal Path Setup panels, very similar to the Signal Path Setup panels in Sequence Mode. The central column contains Generator and Analyzer settings. The right panel provides a selection of Tools displays, with a choice of meter, graph, sweep and other analysis tools.

Bench Mode measurement tools include a number of bar graph/numeric meters that can each be set to one of over 12 signal parameters, real time monitors, and an extremely flexible set of sweep engines for stepped sweep, FFT, time recorder, continuous sweep and acoustic response sweeps. X and Y axes can be set to a broad range of parameters, and nested sweeps are supported.

Bench Mode shares a number of features with Sequence Mode, including the Menu Bar, the Toolbar across the top of the workspace, and the Status Bar across the bottom.

The Signal Path Setup menus are essentially the same as in Sequence Mode, and the Scope and FFT Spectrum Monitors are identical. A toolbar in the Monitors/Meters tabbed view provides access to the same tools that are available in Sequence Mode.

The Signal Path Setup menus

The left panel is Signal Path Setup, and the menu at the top of the panel provides configuration settings for the Bench Mode signal path. Depending upon the connected APx instrument hardware, these menus may include

- Input/Output
- References
- Output Switchers
- Input Switchers
- DCX
- Clocks
- Triggers

The Signal Path Setup menus are also available in Sequence Mode.

Signal Path Setup is discussed in detail in Chapter 7.

The Generator and Analyzer

The Generator and Analyzer in Bench Mode are independent of each other, and to some extent independent of the Bench Mode measurements. **The** Generator, for example, can be turned **On** or **Off**, or have settings made or changed at any time. A measurement sweep can take control of the generator for the sweep duration, and then returns the generator to its previous state. Read more about the Bench Mode Generator in Chapter 88.

The Bench Mode Analyzer provides a number of analysis tools that you can customize for your test. These include settling, filtering and IMD analysis settings. Read more about the Bench Mode Analyzer in Chapter 88.

The Bench Mode Measurements

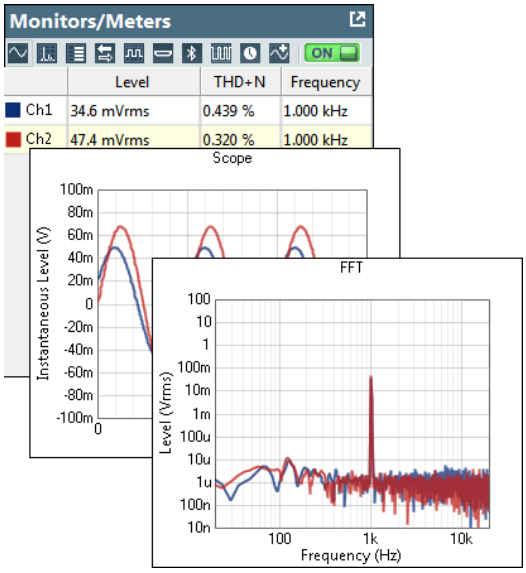
Bench Mode provides tabbed measurement panels, each displaying different measurement tools, including Monitors and Meters, a stepped Sweep, an FFT Scope and Spectrum analyzer, a time Recorder, and Continuous Sweep and Acoustic Response suites. Chapter 88, Bench Mode Overview, covers general topics regarding making Bench Mode measurements. Each Bench Mode measurement tool is discussed in detail in Chapters 89 through 94.

Common Features to Both Sequence Mode and Bench Mode

Features and tools common to both Sequence Mode and Bench Mode are discussed in the next section, including many Menus and Menu items, Project Properties, the Monitors and the Monitors/Meters tool bar, the Status Bar, the Selector, and the Audible Signal Monitor.

Sweep features like importing sweep tables or nesting sweeps are the same, and graph features such as cursors or limits operate in the same way.

The Monitors/Meters



APx500 has a number of Monitors and Meters (depending upon hardware and software options) to provide continuous detailed information about the signals at the analyzer's inputs, and to monitor and set other features, all without changing the primary measurement view. Read more about the Monitors and Meters in Chapter 5.

In Sequence Mode, these are normally docked just below the Navigator, but each signal monitor can be undocked and moved around the screen.

In Bench Mode, the Scope and FFT Spectrum Monitors are at the top of the Monitors/Meters panel, and the other Monitors are available from the Monitors/Meters toolbar.

When turned **ON**, the signal monitors operate continuously. The displays are refreshed many times per second, creating near real-time signal monitors.

The signal monitors can be turned **OFF**, which improves computer performance by reducing the CPU usage.

Use the Monitors/Meters toolbar to select the monitor view.



Choose

-  Scope Monitor
-  Meters Monitor
-  FFT Spectrum (disabled for APx515 unless software option SW-AML is installed.)
-  Aux Control

-  Metadata Monitor: Status Bits/User Bits
-  Metadata Monitor: HDMI (available for instruments fitted with HDMI I/O)
-  Bluetooth Monitor (available for instruments fitted with Bluetooth I/O)
-  Metadata: Status Bits/User Bits (for instruments with a digital interface only).
-  Metadata: HDMI (for instruments fitted with the HDMI Option).
-  PDM Control Codes Monitor (for instruments fitted with the PDM Option.
-  Digital Serial Reference Clock (for instruments fitted with the Digital Serial I/O Option.
-  Analog Sine Generator (for the APx555)

The Menu bar



The **Menu** bar extends across the top of the workspace and provide access to the following menus.

The File menu

New Project (Sequence and Bench Modes)

Opens a new project file. New projects always use a template, either the Standard APx500 Project or another template of your choice. Available templates are shown in the New Project dialog.

Open Project (Sequence Mode)

Opens an existing project file from disk. You will be prompted to save the current project.

Lock Project (Sequence Mode)

Allows you to set a password to lock the current project; also provides the capability to save a project as a Production Test (page 564.)

This function is useful to prevent accidental project changes by a production operator. When locked, all user control of signal path, measurement and sequence settings is disabled. A sequence can still be run, and display and navigation features are available. See Lock/Unlock Project on page 32 for more information.

Unlock Project (Sequence Mode)

Allows you to unlock a project by entering the correct password.

Save Project (Sequence and Bench Modes)

Saves the current project file. If this is the first time the project is being saved, you will be prompted for a project filename.

Save Project as (Sequence and Bench Modes)

Saves the current project under a new filename.

Save Project as Template (Sequence and Bench Modes)

You can make a new project template by saving your current project as a template file. In the dialog, name your template file and write an optional description.

> *You can create project template files to help manage and streamline your measurement work. A project template file contains all project definitions and settings.*

Manage Project Templates (Sequence and Bench Modes)

This opens a dialog in which you can rename or delete a project template. You can also make a template the default project.

You can add, rename or delete project groups from the Edit Groups dialog. You can move a project template from one group to another using the Move to Group dialog.

Templates are saved under My **Documents > APx500 Templates**, where the subfolders are managed as Template Groups. Choose or create a Templates Group from the **Group Name** field.

Manage Attached Project Items (Sequence and Bench Modes)

Image files, generator waveform files, Microsoft Word report layouts, multitone signal definition files and other data can be attached to an APx500 project for use in a measurement or a user prompt. This menu item opens the **Manage Attached Project Items** dialog, page 27.

Import Graph Data (Sequence and Bench Modes)

Imports graph data from a file on disk to be displayed in the current graph.

Export Graph Data (Sequence and Bench Modes)

Exports the graph result data as an Excel spreadsheet (*.xls), a comma separated value text file (*.csv) or a Matlab file (*.mat).

Operating Mode (Sequence and Bench Modes)

Choose Sequence Mode or Bench Mode.

Recent Projects (Sequence and Bench Modes)

Displays a list of recently opened project. Select a project to open.

Exit (Sequence and Bench Modes)

Closes the application.

The View menu

Scope Monitor (Sequence Mode)

Switches the Monitor to the Scope view.

FFT Spectrum Monitor (Sequence Mode)

Switches the Monitor to the FFT Spectrum view.

Meters Monitor (Sequence Mode)

Switches the Monitor to the Meters view.

Aux Control (Sequence and Bench Modes)

Opens the Aux Control Monitor view.

Metadata Monitor: Status Bits/User Bits (Sequence and Bench Modes)

Switches the Monitor to the Status Bits/User Bits view. This selection is not available if the instrument not fitted with DIO or HDMI+ARC.

HDMI: Metadata and CEC (Sequence and Bench Modes)

Switches the Monitor to the HDMI metadata/CEC view. This choice is only available when the instrument is fitted with the HDMI or HDMI+ARC option.

Bluetooth Monitor (Sequence and Bench Modes)

Switches the Monitor to the Bluetooth view, if the Bluetooth option module is installed.

PDM Control Codes/Vdd

Switches the Monitor to the PDM Control Codes/Vdd view. This choice is only available when the instrument is fitted with PDM option.

Serial Reference Clock (Sequence and Bench Modes)

Switches the Monitor to the Digital Serial Reference clock view. This choice is only available when the instrument is fitted with Digital Serial I/O, and Input Configuration is not DSIO.

Analog Sine Generator (Sequence and Bench Modes)

Switches the Monitor to the Analog Sine Generator View (APx555 only). This allows secondary analog sine output while primary generator output is digital.

Hide/Show Navigator (Sequence Mode)

Toggles the Navigator visibility.

Hide/Show Selector (Sequence Mode)

Toggles the Selector visibility.

Turn Monitors On/Off (Sequence and Bench Modes)

Toggles the Signal Monitors On or Off. When Signal Monitors are Off the measurement PC will show better performance.

Audible Signal Monitor (Sequence and Bench Modes)

Allows you to turn the Audible Signal Monitor On or Off, or to adjust settings.

The Measurements menu

Turn Generator On F9 / Turn Generator Off F12 (Sequence and Bench Modes)

If the current measurement is a meter measurement, **Turn Generator On F9** and **Turn Generator Off F12** appear on the Measurements menu. Click these, or press the associated keys (**F9** or **F12**) to turn the Generator **On** or **Off** in the current measurement

Active Measurements List (Sequence and Bench Modes)

Displays a list of Active Measurements in the current Signal Path.

The Project menu

Add Signal Path (Sequence Mode)

Adds a signal path to the project

Add Measurement (Sequence Mode)

Choose a measurement to add from one of the fly-out menus: **Add Measurement, Insert Measurement Before Selection,** or **Insert Measurement After Selection.**

Add Primary Result (Sequence Mode)

Adds a primary result to the selected measurement.

Define New Result (Sequence Mode)

Chose X, Y or Bar from a submenu to add a user-defined results.

Add Derived Result (Sequence Mode)

Adds a Derived Result to the selected result. This is only available when a result is selected in the Navigator.

Delete (Sequence Mode)

Delete items selected on submenu.

Run Sequence (Sequence Mode)

Runs entire sequence defined in the Navigator.

Run Selected Signal Path (Sequence Mode)

Runs a sequence for the selected Signal Path only.

Run Selected Measurement (Sequence Mode)

Runs the selected measurement only.

Run Sequence from Selected Measurement (Sequence Mode)

Runs a sequence beginning at the selected measurement.

Edit Report Properties (Sequence Mode)

Opens the Edit Report Properties dialog.

Edit Data Output Properties (Sequence Mode)

Opens the Edit Data Output Properties dialog.

Sequencer Properties (Sequence Mode)

Displays the Sequencer Properties dialog.

Project Properties (Sequence and Bench Modes)

Displays the Project Properties dialog.

The Window menu

The View menu is not available in Bench Mode.

The Window menu is available from the main Menu bar. Measurements that have been visited are listed in the Window menu, with a check mark indicating the active measurement. Up to ten measurement are listed in the menu; if there are more than ten measurements in the Project, the ten that have been most recently active are listed here. Click the **Windows...** button to open the Windows dialog and view all the measurements.

The Windows dialog (Sequence Mode)

Open the Windows dialog by selecting **Windows...** from the bottom of the Window menu. All measurements in the Project are shown in the Windows dialog. The active measurement is highlighted in the dialog. To activate a different measurement, select the measurement and click **Activate.**

The Help menu

Help on this topic F1 (Sequence and Bench Modes)

Displays context-sensitive Help for the current measurement. Equivalent to pressing function key F1.

Contents (Sequence and Bench Modes)

Displays the Help Contents view.

Index (Sequence and Bench Modes)

Displays the Help Index view.

Register Product (Sequence and Bench Modes)

Register your product with the Audio Precision Web site.

About (Sequence and Bench Modes)

Provides detailed information about APx500 and the current instrument hardware and firmware.

The Toolbar



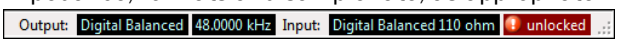
The Toolbar provides a convenient access to often-used commands.

- New Project / New Project from Template
- Open Project
- Save Project
- Project Properties
- Scope Monitor
- FFT Spectrum Monitor
- Meters Monitor
- Aux Control
- Metadata Monitor: Status Bits/User Bits
- Metadata Monitor: HDMI
  (available for instruments fitted with HDMI I/O)
- Bluetooth Monitor
  (available for instruments fitted with Bluetooth I/O)
- PDM Control Codes/Vdd controls
  (available for instruments fitted with PDM)
- Serial Reference Clock
  (available for instruments fitted with DSIO)
- Analog Sine Generator
  (available on the APx555)
- Audible Monitor Configure the Audible Signal Monitor
- OFF Audible Signal Monitor On/Off

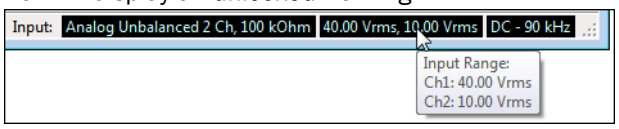
Status bar



The Status Bar (at the bottom of the APx500 window) displays the current output and input connections, impedance, formats and sample rate, as appropriate.



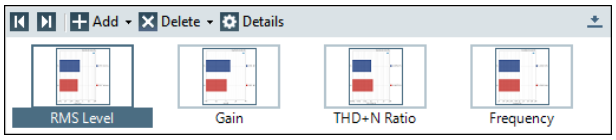
When the Input Configuration is set to a digital input and no signal is present, or if the signal is corrupted or out of range, the sample rate indicator in the Status Bar will display an **unlocked** warning.



When the Input Configuration is set to an analog input, the input range is displayed. If channels are set to different ranges, place the cursor over the range field and the tool tip will display the ranges set for the various channels.

If you are using the APx1701 Transducer Test Interface, clipping, power and thermal overload warnings are displayed in the Status Bar.

The Selector



The Selector runs across the bottom of each measurement view panel, providing a visual means to browse and choose measurement views from thumbnail icons.

Selecting a result

To select a result, click on the result thumbnail.

In addition, you can select an existing derived result by first selecting the source result and then choosing **Go to Derived** from the right-click context menu.

You can also select a source result by first selecting the derived result and then choosing **Go to Source** from right-click context menu.

Moving a result

In the Selector, grab a result thumbnail with the mouse cursor and drag it to a new position in the Selector. The result positions in the Selector determines the relative result locations in a report.

Adding or deleting a result

Select a result and click **Add** or **Delete** in the Selector toolbar.

In addition, you can add a derived result by right-clicking on the result thumbnail and choosing **Add Derived Result**. You can delete a result by right-clicking on the result thumbnail and choosing **Delete**.

Viewing result details

To view result details, select the result and click **Details** in the Selector toolbar.

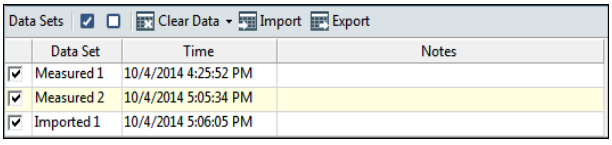
Derived Results

A derived result is an additional result, computed from data in one or more existing results. Examples include smoothing, calculating means, comparisons of data, etc. Read more about Derived Results in Chapter 100.

The Data Sets panel

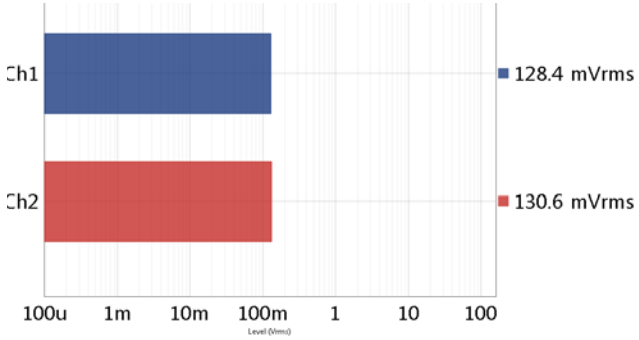
A **Data Set** is created whenever a batch measurement is run. Real-time measurements do not produce a **Data Set.** A **Data Set** is the set of results measured from an acquisition, or imported from a data file.

Data Sets are managed in the **Data Sets** panel, located below the **Selector** in the APx workspace.



See Managing Data Sets on page 649.

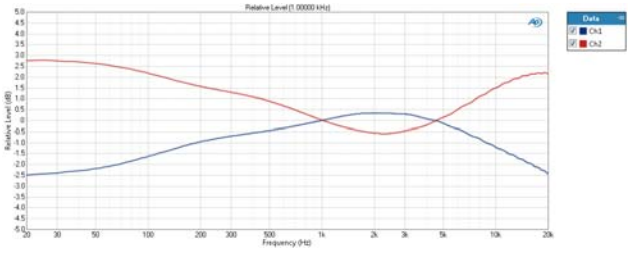
Measurement Result Displays



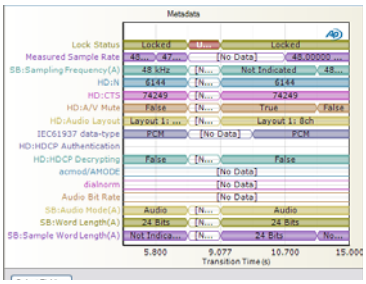
Meter bar single value result

	Ch1		Ch2	
	X	Y	X	Y
1	18.7500	-2.510	18.7500	2.722
2	22.5000	-2.454	22.5000	2.748
3	26.2500	-2.418	26.2500	2.746
4	30.0000	-2.385	30.0000	2.734
5	33.7500	-2.352	33.7500	2.716
6	37.5000	-2.317	37.5000	2.695
7	41.2500	-2.280	41.2500	2.672
8	45.0000	-2.242	45.0000	2.646

Tabular result view



XY graph result



“Logic analyzer” result

APx500 has many measurement result views using different methods and display techniques. Read more about Results in Chapter 99.

Meter Bar Displays

Meter bar displays are used for those measurements that measure only one parameter per channel, providing a clear view of this parameter across many channels. APx500 also uses a meter bar display for distortion product measurements, where several parameters are displayed but the view is limited to one channel.

XY Graphs

APx500 graphs display two parameters of a signal measurement on an XY grid. In a frequency domain display, these parameters are typically level versus frequency, with level on the Y axis (vertical, left), and frequency on the X axis (horizontal, bottom). In a time domain display, these parameters are typically level versus time, with level on the Y axis and time on the X axis. Other relationships can be graphed: phase angle versus frequency, DUT output level versus generator level (linearity) and so on.

A second Y axis can be added to show three parameters simultaneously.

"Logic analyzer" result

The Metadata Recorder logs metadata status changes against time, shown in a logic analyzer type result.

Tabular Displays

Meter and graph measurement values are available as tabular displays in the meter or graph Data Grid.

Undocking Views

One or more measurement views can be undocked to compare readings. Click the **Undock** button on a view, and then select another view to Undock. Undocked views, shown here, can be independently moved and resized, revealing all graph settings for independent adjustment.

The APx Project

The APx Project

An APx500 project is the set of configurations and instructions for a test and measurement session, with necessary waveform files, data files and images.

A project can be as simple as one brief measurement in the default configuration, or as complex as a series of different signal path configurations encompassing hundreds of measurements directed by an automated sequence and documented in a custom report.

Both Sequence Mode and Bench Mode tests are contained within a project.

The Project File

APx500 projects are stored in a project file with the filename extension *.approjx. A co-worker with APx500 and a compatible Audio Precision APx500 series analyzer can open a project file, make the same connections to a DUT and perform the same set of tests.

Project files created in earlier versions of APx500 are opened and converted to the current format.

A project file has one or more signal paths (for Sequence Mode), and contains all the project measurement information, including signal path setup, all settings for each measurement and graph along, with limit information and sequence instructions. Generator waveforms, user prompt images and report layout files are also saved in the project file.

The following data is NOT saved in an APx project file:

- Audio acquisition data.
- External programs called by a sequence step.

Project Templates

A project template file contains all project definitions and settings. You can create project template files to help manage and streamline your measurement work.

Saving a Project Template File

You can make a new project template by saving your current project as a template file. Choose File > Menu > Save Project As Template. In the dialog, name your template file and write an optional description.

Templates are saved under My Documents > APx500 Templates, where the subfolders are managed as Template Groups. Choose or create a Templates Group from the Group Name field.

Check the checkbox to Make this the Default Project Template.

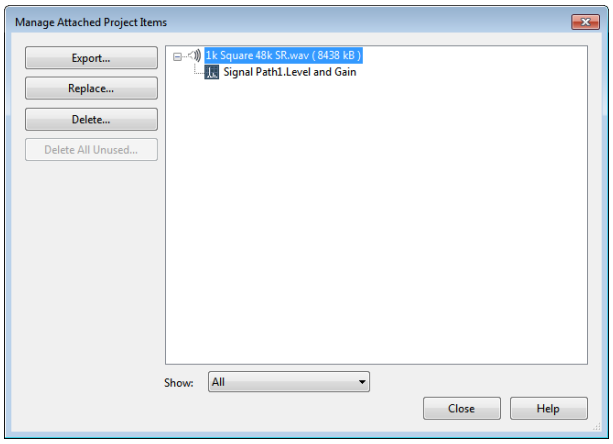
Opening a New Project / Managing Project Templates

Choose File > New Project to open a new project file. New projects always use a use a template, either the Standard APx500 Project or another template of your choice. Available templates are shown in the New Project dialog. You can also rename or delete a template, or change the default template. You can add, rename or delete project groups from the Edit Groups dialog. You can move a project template from one group to another using the Move to Group dialog.

Manage Attached Project Items (Sequence and Bench Modes)

Image files, generator waveform files, Microsoft Word report layouts, multitone signal definition files and other data can be attached to an APx500 project for use in a measurement or a user prompt. Attached items that have been removed from current use remain attached to the project, increasing the project's memory and disk use.

The **Manage Attached Project Items** dialog allows you to view all the current attachments, see where they are used, and manage their usage.



Also see Chapter 14 for more information about generator waveforms, Chapter 64 for multitone signal definition files, Chapter 87 for Microsoft Word report layouts, and page 573 for image files.

Manage Attached Project Items: Export

Export allows you to save an attached file to disk, outside of the APx project file. Custom report layouts, in particular, are created, edited and maintained within the project, and must be exported for use by other projects.

Manage Attached Project Items: Replace

Replace allows you to choose a new file for a particular project item. The new file replaces the current file in all its project uses.

Manage Attached Project Items: Delete

Delete allows you to remove a specific file from the project.

Manage Attached Project Items: Delete all unused

Delete all unused examines the current usage of all the files in the project. The files that are not currently being used are removed from the project.

Project Properties dialog

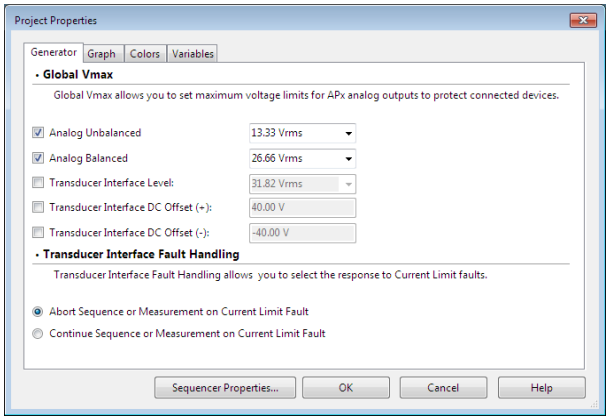
These properties are project-wide (global) settings. They affect all measurements and signal paths within the project, for both Sequence Mode and Bench Mode.

The Project Properties dialog is available by selecting the **Project Properties** icon in the main toolbar, or by selecting **Project Properties** from the Project menu.

There are four tabbed pages in the Project Properties dialog:

- Generator
- Graph
- Colors
- Variables

Generator tab



Global Vmax

Global Vmax is a feature that allows you to set maximum voltage limits for APx analog outputs to protect connected devices. When enabled, **Vmax** is a global setting, effective across all measurements in the project for both Sequence Mode and Bench Mode.

Setting Vmax

Open the Project Properties dialog by clicking **Project Properties** in the Main toolbar, or by selecting **Project Properties** on the Project menu. Choose the **Generator** tab.

Depending upon the analyzer hardware, you will see one or more of the following analog output choices:

> *Note that the Transducer output Vmax settings are available on this dialog whether or not an APx1701 Transducer Test Interface is connected and active.*

For all analyzers except the APx511:

- **Analog Unbalanced**
- **Analog Balanced**
- **Transducer Interface Level**
- **Transducer Interface DC Offset (+):**
- **Transducer Interface DC Offset (–):**

For the APx511:

- **Speaker**

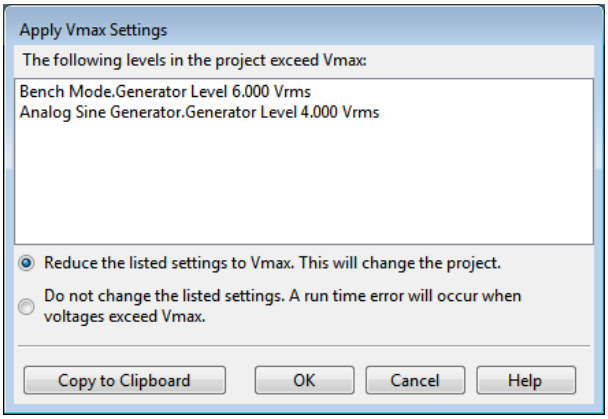
To enable **Global Vmax** for an analog output configuration, check the box for that selection. When checked, each choice provides a field in which you can enter a maximum generator output value, in Vrms, dBV or dBu.

To disable **Global Vmax** for an analog output configuration, uncheck the checkbox for that selection.

If there are no output levels in the project that will exceed Vmax, when **OK** is pressed the dialog closes with no action taken.

If one or more output levels in the project exceeds Vmax, when **OK** is pressed the **Apply Vmax** (see below) dialog opens.

Apply Vmax



When **Vmax** is applied to a project, you can view the project settings that request voltage levels that will exceed Vmax (if any).

This list can be copied as text to the Windows clipboard by pressing the **Copy to Clipboard** button.

Options

There are two options to apply **Vmax** to a project:

- **Reduce the listed settings to Vmax. This will change the project.**
  In this case, **Vmax** will search through the project and change output level values to **Vmax**. Save serve your previous settings. See **Vmax and Stepped Level Sweeps**, below, to read how **Vmax** treats sweep tables.
- **Do not change the listed settings. A run time error will occur when a voltage exceeds Vmax.**
  In this case, none of the settings in the project are changed. Instead, **Vmax** will issue an error at run time. At that time, you must change the output setting or sweep table to continue.

Vmax and Stepped Level Sweeps

When **Vmax** is set to the first option "**Reduce the listed settings to Vmax**," it will examine the levels requested in generator level settings throughout the project. **Vmax** will also examine the levels requested at each step of a stepped level sweep. These actions are taken with stepped level sweep steps:

- For a linear sweep table, the highest step is reduced to **Vmax**, and the rest of the steps are distributed from the lowest value to **Vmax,** linearly. The number of steps in the new table is the same as in the original table.
- For a log sweep table, the highest step is reduced to **Vmax**, and the rest of the steps are distributed from the lowest value to **Vmax,** logarithmically. The number of steps in the new table is the same as in the original table.
- **Vmax** will not edit a custom sweep table. Instead, a run time error will be issued and you must edit the custom sweep table to continue.

Hardware Voltage maximums

Note that APx analyzer models have different maximum voltage output characteristics, and that generator level or **Global Vmax** cannot be set higher than the hardware maximum voltage specification.

	Vrms max	dBV max	dBu max	Vpp max
APx555, 582, 525/6 w/AG52 bal	26.66	28.517	30.736	75.4
as above, unbal	13.33	22.497	24.715	37.7
APx525/6 bal	21.21	26.531	28.749	60
as above, unbal	10.61	20.514	22.733	30
APx585/6 bal	14.4	23.167	25.386	40.72
as above, unbal	7.2	17.147	19.365	20.36
APx515 bal	16	24.082	26.301	45.2
as above, unbal	8	18.062	20.280	22.6
APx511, speaker	5.7	15.117	17.336	16.1

APx hardware maximum voltage output levels

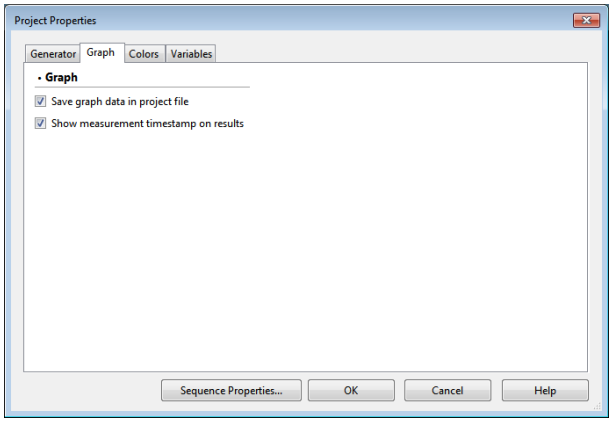
When a project has been saved and then re-opened with hardware that has a lower maximum output voltage, a project conversion warning is issued, and the project level settings are changed as necessary.

Transducer Interface Fault Handling

The power amplifier within the APx1701 Transducer Test Interface has features designed to protect the APx1701 and connected DUTs from damage. The options set here determine how a current limit fault detected in the APx1701 is to be handled. The choices are:

- **Abort Sequence or Measurement on Current Limit Fault (the default)**
- **Continue Sequence or Measurement on Current Limit Fault**

Graph tab



Save graph data in project file

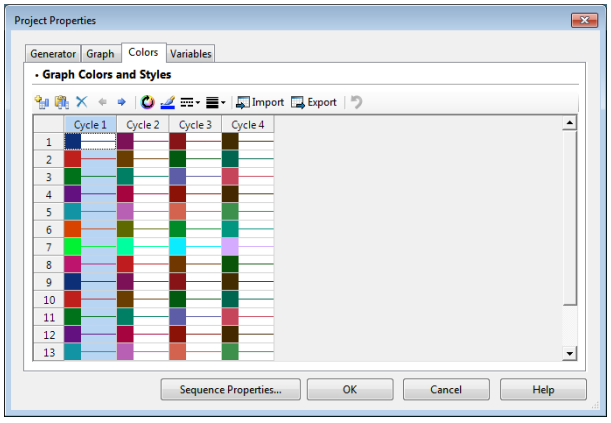
By default, this checkbox is checked, and the graph data for all results in the project is saved in the project file.

Graph data that requires recalculation is not saved. For example, when saving **Relative Level** data, only the data in memory when saving (**Normalized** or **Centered**) will be saved. When a project file that contains result graph data is opened, all graph data that do not require recalculation are displayed in their various measurement results.

Show measurement timestamp on results

Measurement data received into the project, whether as Measured data or Imported data, is stamped with the time and date it entered the project. When this checkbox is checked the measurement timestamp is displayed on the graph panel and in the report.

Colors tab



These properties are project-wide (global) settings, but can be overridden locally for any result. For local bar graph (meter) color overrides, see page 640. For local XY graph color and line style overrides, see page 648.

Colors for bar graph (meter) displays and colors, line style and line width for XY graph traces can be customized from a wide palette, and automatic cycling of colors and styles can be specified for appended and imported measurements.

Graph Colors and Styles

Rows

Each row in this grid applies to an input channel, to a maximum of 16 channels.

Columns

Each column in this grid shows color, line style and line width for one **Color Cycle**. **Color Cycle** columns can be added to the grid, to a maximum of 16.

Color Cycle behavior

For measurements that support **Append Graph Data** and **Import Graph Data**, each **Color Cycle** column applies to a **Data Set**. Within a measurement result, these **Color Cycles** are applied to **Data Sets** in the order the **Data Sets** are added to the result. If the last defined **Color Cycle** has been applied, the next **Data Set** added will have **Cycle 1** applied, and further **Data Set** additions will progress through the **Cycle** columns.

These features allow you to specify colors and styles that can show channels and appended data in a clear and understandable display.

Grid Toolbar

Add Cycle button

Add a **Cycle** column to the grid.

Duplicate Selected Cycle button

Copies the currently selected **Cycle** column and adds it to the grid.

Move Cycle right or left button

Moves a **Cycle** column. This affects the order of the cycles applied to appended and imported data.

Automatically choose new colors button

Changes the color of the selected cells by an arbitrary amount. A quick way of adding different colors to the **Cycle** grid.

Set Color button

Allows selection of a custom color for the selected cells, using the color picker dialog.

Set Line Style button

Allows selection of one of five line styles for the selected cells.

Set Line Width button

Allows selection of one of five line widths for the selected cells.

Import button

Opens an import file browser to select and import a **Trace Style Cycle** file (*.tsc). The styles in the file will overwrite the styles currently in the grid.

Export button

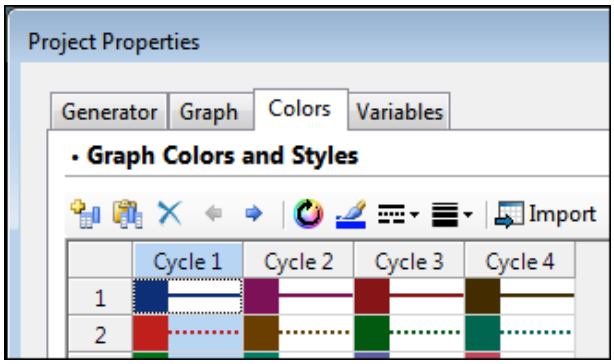
Opens an export file browser to select and export the current **Cycle** grid contents to a **Trace Style Cycle** file (*.tsc).

Reset to Default button

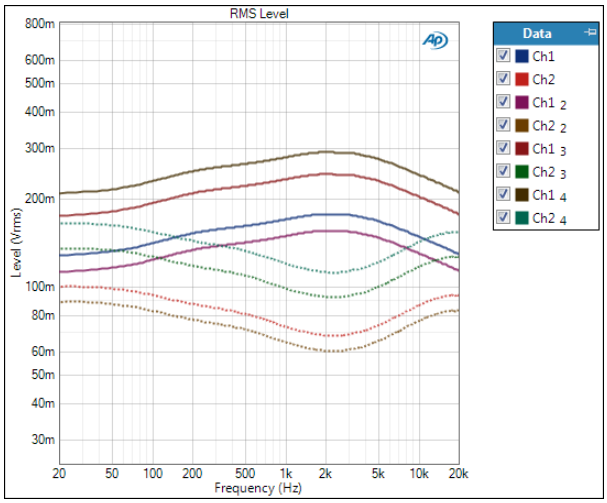
Resets the highlighted cells to the default colors.

Example

For example, a user might want to display noise vs. time for 2 channels, across 4 appended acquisitions. The illustrations show a set of Color Cycles that would provide a bold, solid line for Channel 1 and a bold, dotted line for channel 2. The appended acquisitions would cycle through the four colors.

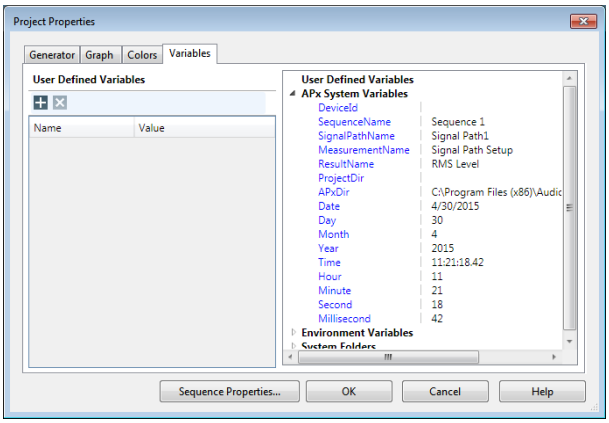


Color Cycles example



The resultant Noise Recorder graph

Variables tab



Overview

APx can use string-based variables to simplify the entry of product model, product serial number, operator name and other information in automated processes; and to create paths and filenames.

APx recognizes these APx Variables:

- **User Defined Variables**
  APx Variables created by the user containing arbitrary text strings.
- **APx System Variables**
  A predefined set of APx Variables useful in an automated sequence.
- **Environment Variables**
  APx Variables corresponding to the Windows Environment variables defined on the host system.
- **System Folders**
  APx Variables corresponding to the list of Windows System folders.

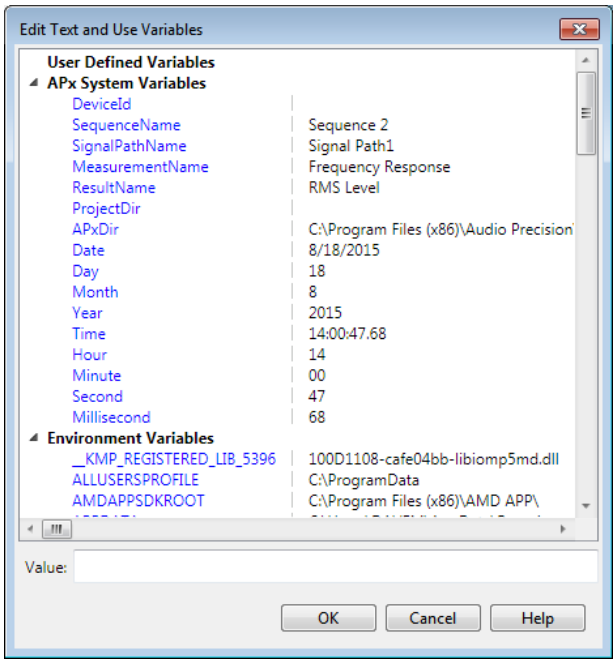
The Variables tab provides a dialog to create User Defined Variables, and visibility of the APx System Variables, Environment Variables, System Folders, and their current values.

Creating a User-defined variable

On the Variables tab, click the **Add Variable** button in the User Defined Variables pane. A **Create Variable** dialog will open. Enter a name for the new variable in the Variable Name field and select **OK**.

In the User Defined Variables grid, enter a value for the new variable. The value may be any text string.

Edit Text and Use Variables dialog



Lock/Unlock Project

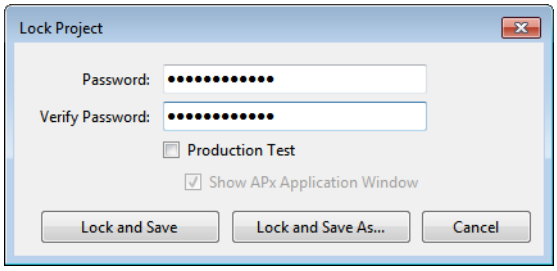
You can Lock an APx project to prevent unauthorized changes. Use a password to lock a project; the password is required to unlock the project.

A locked project can be used to make measurements or run a sequence without unlocking, but changes to the project settings and configuration are not allowed.

To save a project as a **Production Test** (page 564), check the **Production Test** checkbox when locking the file.

Locking a project

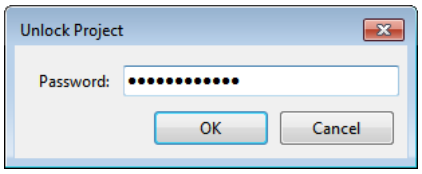
From the Main toolbar, choose the **File menu**, then select **Lock Project...** Or, click the **Lock Project** button in the Navigator's Sequence toolbar.



Enter and verify a password. Click **Lock and Save** to save the file under the current name, or click **Lock and Save as...** top save the locked version under a new name.

Unlocking a project

Open a locked project file. From the Main toolbar, choose the **File menu**, then select **Unlock Project...** Or, click the **Unlock Project** button in the Navigator's Sequence toolbar.

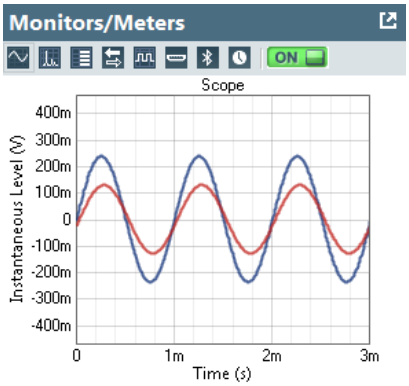


Enter the password and click **OK**.

Monitors/Meters

Introduction

Scope Monitor



The Scope Signal Monitor provides an oscilloscope view on an XY graph.

Settings

The additional settings shown here, and graph display options, are available from a right-click context menu or from the graph panel when **Undocked**.

Interpolation

For time-domain displays, the APx500 graphs are normally plotted using sinc function interpolation. Interpolation adds points to the displayed trace that do not exist in the measured data, to make data trends more easily visualized. The default in APx is to have Interpolation **On**. However, digital domain signals are sometimes best understood when viewing actual samples, with interpolation switched **Off**. Also see “Interpolation and limit failure markers” on page 678.

Residual Display

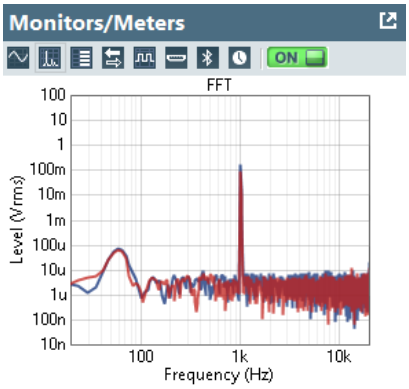
- **Off**
  When Residual Display is **Off** (the default), a time-domain view of the full input signal is displayed.
- **x1 through x100K**
  These selections display a time-domain view of the THD+N residuals after the fundamental is removed, with selectable gain applied. This view requires a periodic signal for meaningful results.

Show Cycles

For periodic waveforms, you can choose to display 1 to 10 cycles of the waveform. 3 cycles is the default.

For non-periodic waveforms, the X-axis is set to 200 ms. When Graph Properties X-Axis is not set to Auto, **Show Cycles** is not available.

FFT Spectrum Monitor



This monitor view is disabled for APx515 unless software option SW-AML is installed.

The FFT Spectrum Signal Monitor displays a frequency-domain spectrum view on an XY graph. Right-click on the monitor to access graph properties.

Settings

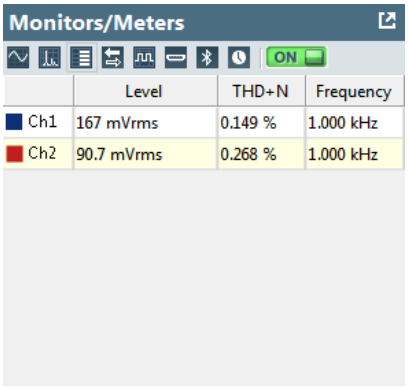
Graph display options and these additional settings are available from a right-click context menu, or from the graph panel when **Undocked**.

- **Show Residual**
  Adds distortion residual FFT display for each channel.
- **Averages**
  For non-periodic waveforms such as noise, averaging multiple acquisitions can provide a more useful view. The FFT monitor will display the average of the last *n* acquisitions. Default is 1 (no averaging); maximum is 1000. **Reset** clears the averaging history and restarts averaging with the next acquisition.

- **Window**
  FFT acquisitions must either be synchronous or have one of a number of amplitude windows applied to provide useful data for interpretation. Each window function brings advantages and disadvantages. The default selection, AP-Equiripple, is a proprietary Audio Precision FFT window that is an excellent choice for most FFT measurements.
- **FFT Length**
  Set the FFT record length here. Options from 256 K samples to 1.2 M samples are available; default is 8 K samples.

Go to page 492 for more information about FFTs and FFT windows.

Meters Monitor



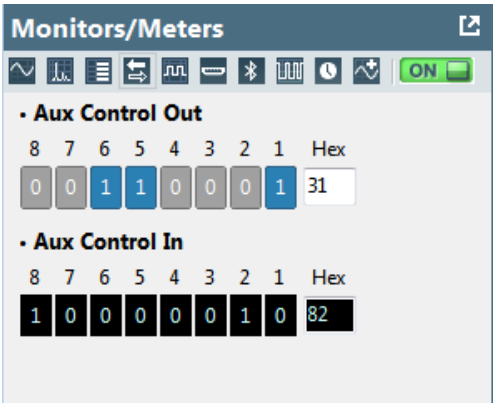
Meters

The Meters Signal Monitor displays a grid with Level, THD+N and Frequency measurements for each channel. The number of channels displayed is set in the Input Configuration settings in the Signal Path Setup view.

Changing Units for the Meters monitor

Right-click on the Meters monitor display to open a context menu, where you can select the units for each of the meter results.

Aux Control Monitor



Reading Aux Control Out and In bit states

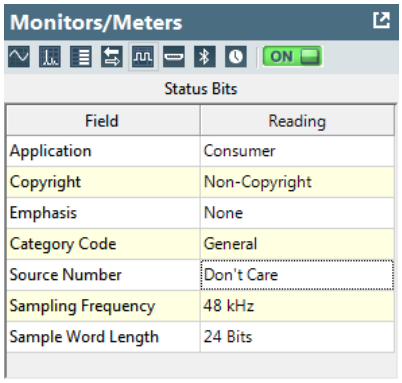
The Aux Control Monitor displays the current state of the Aux Control Out and Aux Control In bits. Bit states are shown both in a binary display and as the hex equivalent.

Setting Aux Control Out bit states

Additionally, Aux Control Out bits can be set from this monitor window, either by clicking on the binary display to toggle the state of a bit, or by entering the hex equivalent in the **Hex:** field.

For more information about Aux Control go to Chapter 102.

Metadata Monitor: Status Bits/User Bits



This monitor is available only for instruments with Digital Input/Output or HDMI.

The Metadata Monitor: Status Bits displays the channel status bits embedded as metadata in the digital input signal. Status bits are supported in the transport streams for the Digital Balanced, Digital Unbalanced, Digital Optical, HDMI ARC Rx and HDMI Sink inputs.

The Digital Serial Input/Output (DSIO) protocol does not support embedded metadata.

Consumer status bits

For consumer application status bits, the docked Monitor view displays this set of status bit fields:

- Application
- Copyright
- Emphasis
- Category Code
- Source Number
- Sampling Frequency
- Sample Word Length

Undock the Monitor to display all available consumer application status bits fields, in both plain text and in hex.

Professional status bits

For professional application status bits, the docked Monitor view displays this set of status bit fields:

- Application
- Emphasis
- Sampling Frequency

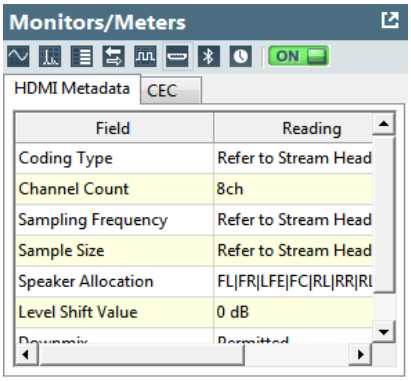
Undock the Monitor to display all available professional application status bits fields, in both plain text and in hex.

User bits

In the undocked Metadata Monitor views, user bits are displayed in hex. User bits are the same for both consumer and professional applications.

Go to page 388 for more information about Status Bits and User Bits.

HD Monitor: Metadata tab



This monitor is available only for instruments fitted with the HDMI Option.

This topic discusses *reading* the Audio InfoFrame data on a received HDMI transport stream in the HDMI Monitor.

For information about *setting* the Audio InfoFrame data to be embedded in the transmitted HDMI transport stream, see page 119.

For HDMI input signals, the docked Monitor view displays these Audio InfoFrame fields:

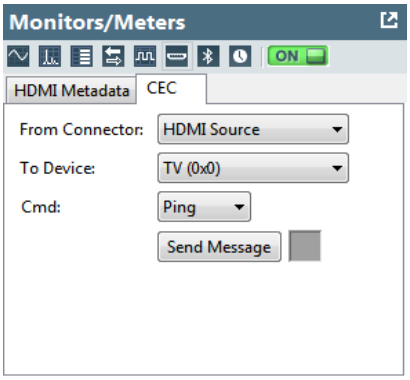
- Coding Type
- Channel Count
- Sampling Frequency
- Sample Size
- Speaker Allocation
- Level Shift Value
- Downmix

Undock the Monitor to display the Audio InfoFrame fields in both plain text and hex, plus these additional HDMI metadata fields:

- N
- CTS
- A/V Mute
- High Bit Rate
- Audio Layout
- HDCP Decrypting

Go to page 390 for more information about HDMI Audio InfoFrame metadata.

HD Monitor: CEC tab



This tab is available if the HDMI+ARC Option (+ CEC enabled) is installed. See More about CEC on page 126.

The CEC tab provides settings and controls to send CEC commands via either the sync or source connectors, and to receive replies acknowledging receipt.

From Connector:

Select the HDMI connector (**Source, Sync, ARC Tx** or **ARC Rx**) that will send the command and receive the acknowledgement.

To Device:

Select the device to be addressed. The menu lists the 16 defined CEC logical addresses.

Cmd:

Select the command to send, **Ping** or **Custom**. A **Ping** is a specific CEC polling message that any addressed device should acknowledge.

When **Custom** is selected, the **Opcode** and **Operands** fields are available to add payload information to the message.

Opcode:

Enter an arbitrary opcode (operational code) here.

Operands (hex):

Enter arbitrary operands here, in hex.

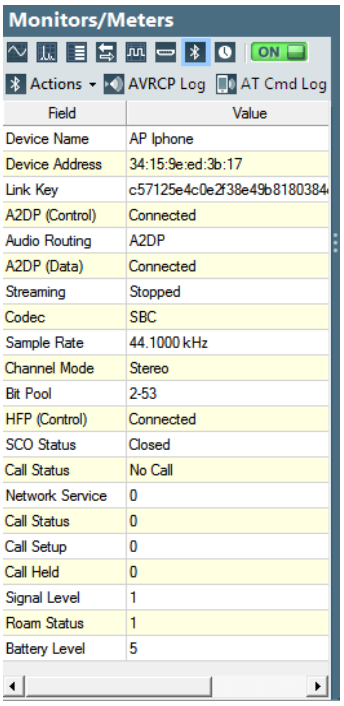
Send Message:

Click this button to sent the message.

The field to the right will remain grayed out until the message is acknowledged (ACK) by the device, then will display a passed graphic . If the message fails, a failed graphic will be displayed.

Note: ARC Tx and ARC Rx can also send certain ARC-specific CEC commands. See CEC for ARC on page 125.

Bluetooth Monitor



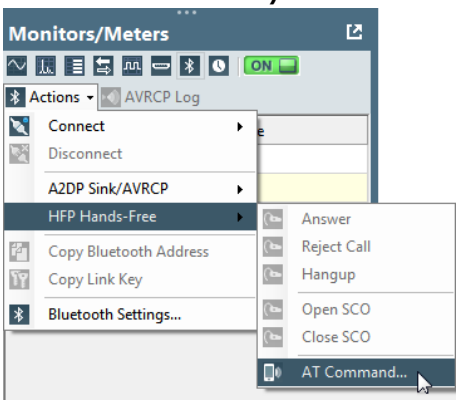
This monitor is available if the Bluetooth Option is installed and active.

The Bluetooth Monitor displays a number of Bluetooth status fields and their current values. Additionally, Bluetooth actions, settings and utility functions are available through a context menu (right-click in the Bluetooth Monitor display.)

Bluetooth Status display

Fields shown are dependent upon profile, connected device and other variables. Here is an example:

Bluetooth actions, settings and utility functions (from monitor context menu)



Right-click on the Bluetooth monitor display to open the context menu. Options include

- Connect.
- Disconnect.
- [One or more Actions, depending upon active profile].
- Copy Bluetooth Address.
- Copy Link Key.
- Open Bluetooth Settings dialog.

AVRCP Command Log

Also see AVRCP in More About Supported Bluetooth Profiles, page 143.

The AVRCP Command Log shows commands sent from APx as AVRCP Controller, and commands received by APx as AVRCP Target. Click on **AVRCP Log** in the Bluetooth Monitor panel to open the AVRCP Command Log panel. You can view and save the logged commands (as a CSV file), and clear the history.

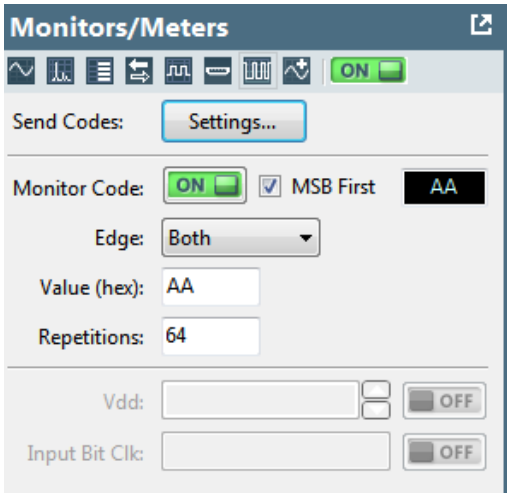
Note that Absolute Volume commands are exchanged in the opposite direction: the AVRCP Target profile sends Absolute Volume commands, and the AVRCP Controller profile receives them.

AT Cmd Log

An AT command is used to control a device. AT commands are a PSTN (Public Switched Telephone Network) legacy, and are widely used for controlling a GSM phone.

If APx is configured with a Bluetooth profile that includes HFP or HSP profiles, the AT Command Log feature will record both transmitted AT commands and received responses. Click on **AT Cmd Log** in the Bluetooth Monitor panel to open the AT Commands panel. You can view and save the logged commands (as a CSV file), and clear the history.

PDM Monitor



See more about PDM on page 160.

This monitor is available if the PDM Option is installed and active.

The PDM Monitor enables monitoring and sending amplifier control codes embedded in a PDM bitstream.

Send Codes:

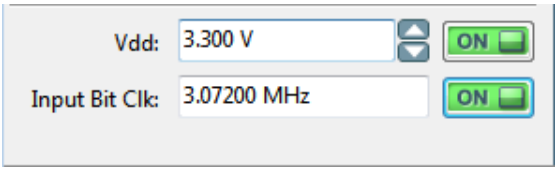
- **Settings...**
  Click this button to open the Send PDM Control Codes dialog. See page 38.

Monitor Code: ON/Off

When **Monitor Code** is **ON**, APx500 monitors the incoming PDM bitstream, looking for a control code that matches the criteria set in the next 4 fields. When a matching code appears, a confirmation field appears in the reading field here and in the Status Bar, displaying the hex value of the desired code. Read more about Control Codes on page 162.

- **MSB First**
  When MSB First is checked, the expected control code must be formatted with MSB first. When **MSB First** is not checked, the desired control code must be formatted with MSB last.
- **Edge**
  Select the data edge to monitor.
- **Value (hex)**
  Enter the expected code value in hex.
- **Repetitions**
  Enter the expected number of repetitions of the control code.

Vdd and Input Bit Clock



The Vdd feature on this panel is only available when the instrument is fitted with a PDM Option module, and when neither Output Configuration nor Input Configuration is set to PDM.

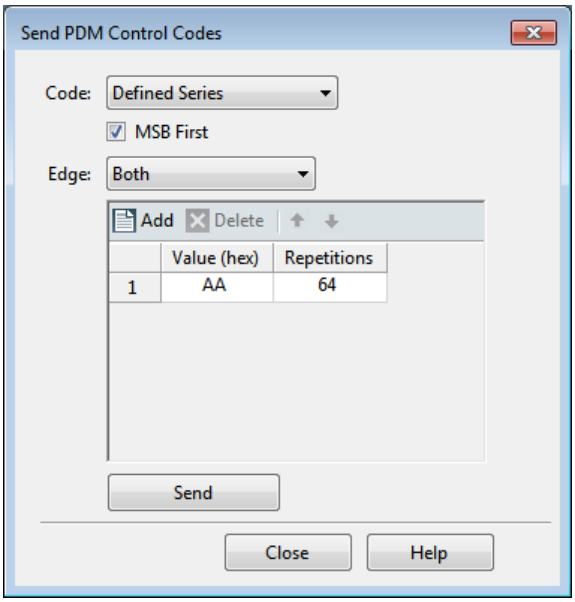
The Input Bit Clock feature is only available when the instrument is fitted with a PDM Option module, and when Input Configuration is not set to PDM.

These features enable power and clock to be supplied to a DUT such as a MEMS microphone or an integrated circuit, while generating and/or analyzing audio in a format other than PDM.

- **Vdd:**
  Set the voltage required for operating power the DUT here. This is available on the PDM module at the Vdd Supply BNC connector, providing DC current up to 15 mA, with a voltage range of +0.8 VDC to +3.60 VDC.
- **ON/OFF**
  This switch turns the Vdd DC power supply **On** or **Off**.
- **Input Bit Clk**
  Set the bit clock rate required by the DUT here. Bit clock is available on the PDM module at the Input Bit Clock BNC connector. The range is 128 kHz to 24.576 MHz.
- **ON/OFF**
  This switch turns the Input Bit Clock **On** or **Off**.

The Input Bit Clock is unavailable in certain configurations due to limited resources. See Clock Availability on page 69.

Send PDM Control Codes dialog



The PDM transmitter can embed amplifier control codes in the PDM bitstream. Click the **Send Codes: Settings** button in the PDM Monitor panel to open the Send PDM Control Codes dialog.

Also see the **Control Codes** discussion in More About PDM on page 162.

This dialog enables configuration for transmitting PDM control codes, and provides a **Send** button to insert the codes into the PDM bit stream in real time.

Code

Choose

- **Defined Series** to send one code value for a defined number of repetitions, or to send a series of different code values. Enter the code values and the number of repetitions in the grid below. Add more rows to the grid to specify other codes.

or

- **Single Code Indefinitely** to repeat one code value indefinitely, until **Stop** is invoked. Enter the code in the **Value** field.

MSB First

Check **MSB First** to send the Control Code MSB (Most Significant Bit) first. When **MSB First** is not checked, the MSB is sent last, and the LSB first.

Edge

Choose **Both**, **Rising** or **Falling** to select the PDM bitstream edge to carry the Control Code data.

Value (hex)

Enter the control code value as a 2 digit hex number.

Repetitions

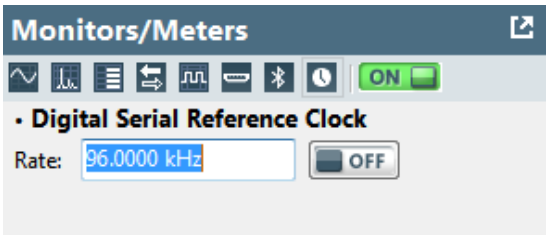
For **Defined Series**, set the number of repetitions here.

Send/Stop

Click the **Send** button to insert the control code(s) into the PDM bit stream. If **Code** is set to **Single Code Indefinitely** when **Send** is invoked, the Send button becomes a Stop button.

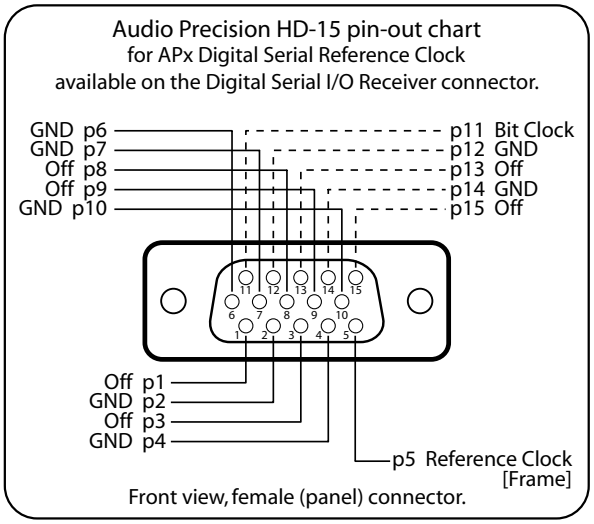
> *Note: PDM Control Codes can be configured and sent during an automated sequence. See Sequencer: Add a Send PDM Control Codes step on page 551 and Sequencer: Add a Send PDM Control Code Indefinitely step on page 551.*

Monitor: Digital Serial Reference Clock



If you have a Digital Serial I/O module installed, you can output a reference clock signal from the module in any configuration where the signal path is not Digital Serial Input. The rate is arbitrary, unrelated to the internal converter rates or other digital I/O rates.

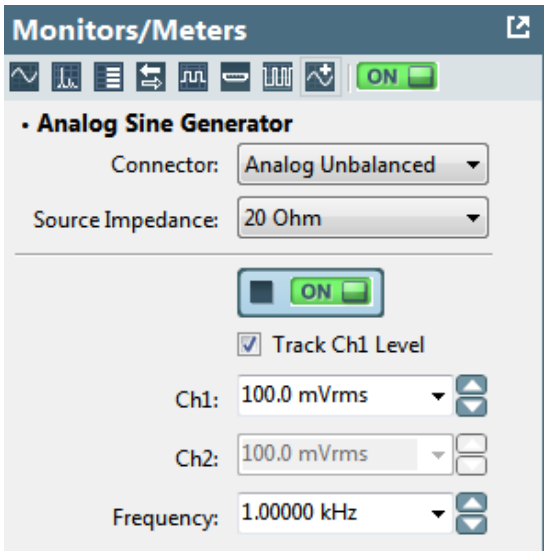
The signal is a 50% duty cycle square wave of 3.3 Vpp with a range of 4 kHz to 216 kHz. Accuracy is ±0.0003% (3 PPM). The output impedance is 50 Ω.



The clock signal appears on pin 5 of the Digital Serial I/O Receiver HD15 jack. Pins 2, 4, 6, 7, 10, 12 and 14 are grounded. Pins 1, 3, 8, 9, 13, 15 are **Off**. Bit Clock (64x) appears on pin 11.

If you have an Audio Precision CAB-DSIO cable kit, use the "FRAME" cable, colored green. This connects to pin 5 and to a grounded pin.

Analog Sine Generator



This feature is only supported by the APx555.

The APx555 has two generators, the standard DSP/DAC generator used across the APx family, and the analog High Performance Sine Generator.

When the APx555 analog outputs are not selected (or, if using the APx1701, Transducer Interface outputs), it is possible to use the High Performance Sine Generator as a second generator. In this case you could use the DAC generator to feed a digital output, while simultaneously using the High Performance Sine Generator to feed a different signal to the analog outputs.

In both Bench Mode and Sequence Mode, you can open a panel to control the Analog Sine Wave Generator feature from the Menu Bar or the Monitors/Meters area.

Choose the desired analog output connector, the source impedance, the channel levels and frequency, and turn the Analog Sine Generator **On** or **Off**.

The Audible Signal Monitor

File input signals cannot be monitored using the Audible Signal Monitor.

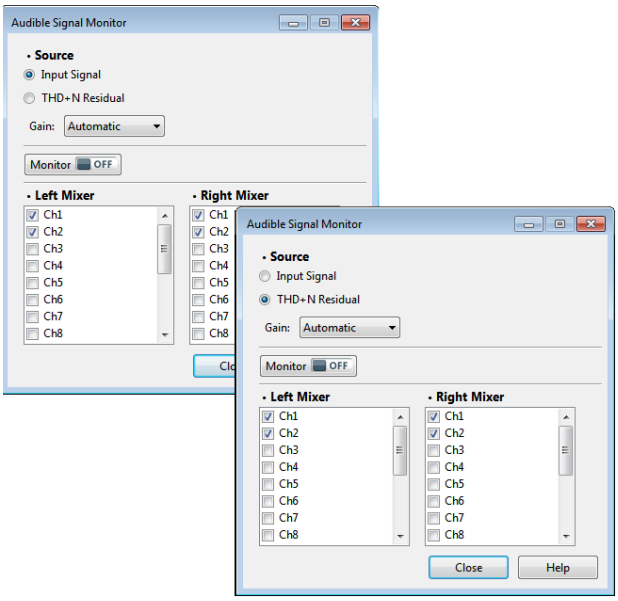
You can listen to the signal applied to the APx500 analyzer inputs by using the Audible Signal Monitor. For some signals and measurements, you can choose to listen to the input signal's THD+N residual.

The Audible Signal Monitor uses the attached PC's software and hardware sound components to play the audio. You must have a PC sound card or an equivalent device, and you must connect headphones, speakers or an audio amplifier to the sound card outputs.

Using the Audible Signal Monitor



Click the Audible Signal Monitor Settings button on the toolbar, or choose **View > Audible Signal Monitor > Settings...** to open the Audible Signal Monitor Settings dialog.



Source

Input Signal

When **Input Signal** is selected, the audio at the system inputs chosen in Signal Path Setup is available for monitoring, subject to the settings made in the Left and Right Mixer panels, below.

THD+N Residual

The **THD+N Residual** setting is only useful when a steady sine signal is present at the system inputs. This setting applies a very deep, narrow notch at the stimulus signal frequency, effectively removing the stimulus. The residual consists of distortion products, hum and noise.

Gain

The APx500 system can measure a very wide range of signal levels. When monitoring the input signals, some signals may be very loud, some very soft. You can choose gain settings here.

- **Automatic** (the default) When **Automatic** is checked, the software automatically adjusts the Audible Signal Monitor volume for a constant level.
- **x1 (0 dB gain)**
- **x10 (20 dB gain)**
- **x100 (40 dB gain)**
- **x1k (60 dB gain)**
- **x10k (80 dB gain)**

- x100k (100 dB gain)

Monitor ON / OFF

The  button toggles the Audible Signal Monitor **On** and **Off**. This button is also available on the toolbar.

Left Mixer / Right Mixer

When **Source** is set to **Input Signal**, the **Left Mixer** and **Right Mixer** panels are available.

All input channels checked in the Left Mixer panel will be summed and routed to the Left PC sound output. Unchecked channels will not be sent to this output.

All input channels checked in the Right Mixer panel will be summed and routed to the Right PC sound output. Unchecked channels will not be sent to this output.

THD+N Monitor Channel

When **Source** is set to **THD+N Residual**, the THD+N Monitor Channel selector is available. Choose the input channel to monitor for THD+N Residual.

Section II: Configuration

6

Generator Waveforms and Controls

Overview

APx analyzers generate a number of audio stimulus signals for both analog and digital outputs.

Signals for digital output are always generated in the digital domain, using digital signal processing (DSP) in the attached computer. Generally, signals for analog output are generated in the digital domain and converted to analog signals using precision digital-to-analog converters (DACs). The APx555 has an additional analog oscillator to generate very low distortion sine signals, and the 555 and analyzers equipped with the AG52 option also have an analog square wave generator.

This chapter looks first at the waveforms generated in DSP and used for digital and analog testing. Next we look at signals used exclusively for digital testing, beginning on page 49. On page 51 we look at the APx555's analog oscillator and its capabilities, and finally on page 51 we examine the AG52 and 555 square wave and DIM waveforms.

DSP signal generation

For the most part, APx stimulus signals are generated in digital signal processing (DSP). Signals are created mathematically or played from digital audio files on disk. These signals can be embedded in digital output streams, or converted to analog output signals using precision digital-to-analog converters (DACs).

Not all selections are available for every hardware or software configuration.

Signals available

For analog (via DAC) or digital output:

- **Sine** (frequency range 0.1 Hz to 80 kHz; optional DC Offset). See page 44.
- **Sine, Burst** (available only for the APx555, analog) See page 52.
- **Sine, Dual** (2 frequencies, on separate channels or summed). See page 45.
- **Sine, Var Phase** (1 frequency. An optional phase offset can be applied to other channels). See page 45.
- **IMD signals** (2 sine frequencies summed according to specific definitions). See page 45.
- **Noise** (various shapes available). See page 47.
- **Generator Waveform** (arbitrary waveform from file on disk). See page 47.

Some measurements use specific, batch-mode stimulus signals:

- **Continuous Sweep** (Farina log-swept sine "chirp," used in Acoustic Response, Continuous Sweep, Frequency Response Impedance Thiele-Small and Loudspeaker Production Test.) See page 49.
- **Multitone** (several tones summed into one complex signal; used in the Multitone Analyzer) See page 49.

For digital output only:

- **Square**. See page 49.
- **Bit test** (digital diagnostic). See page 50.
- **Constant Value** (digital diagnostic)
- **Walking Ones** (digital diagnostic)
- **Walking Zeros** (digital diagnostic)

Common Controls

Most DAC generated signals have these common controls:

- **Levels Track Ch 1**
  If Level Tracks Ch1 is checked, all output channels are set to the level entered for channel 1. If it is unchecked, each channel can be set to a different output level.
- **Level**
  Set the generator level here.

Bit Test, Walking Ones and Walking Zeros have no level controls.

- **DC Offset**
  For a DAC generator continuous sine waves, a DC offset can be added to the signal. For analog output, when sine level is zero the maximum offset is $\sqrt{2}$ times the maximum (±) RMS level for the current instrument and output connector. For digital output, when sine level is zero the maximum offset is 1 D (±).
  If **Level Tracks Ch1** is checked, all output channels have a DC voltage added at the level set in the channel 1 DC Offset field. If it is unchecked, each channel can be offset by a different DC voltage.
  If the generator level is set to zero (0.000 Vrms or 0 FS), the generator will output DC only (pure DC).

Bit Test, Constant Value, Walking Ones and Walking Zeros have no DC offset controls.

Channels

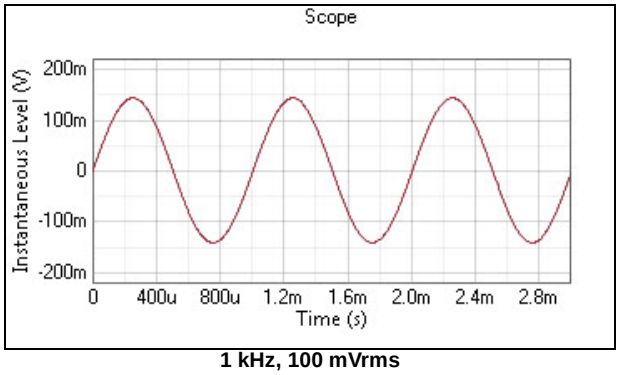
Select the channels to output the generated signal. By default, all available channels are selected.

Waveform Descriptions

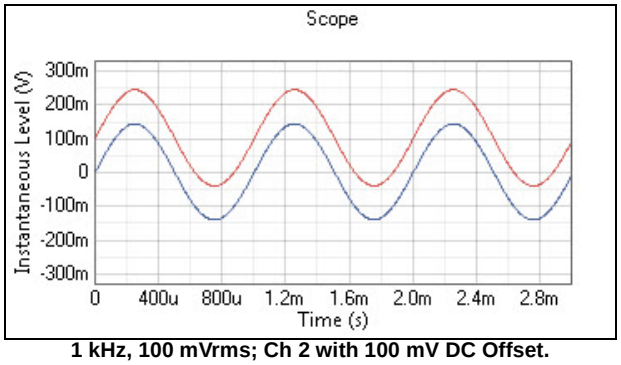
The waveforms cited in the generator descriptions above are examined in more detail here.

DAC Sine Waveform (digitally generated)

The basic sine waveform is a pure tone, a single frequency. Sines are useful in test and measurement because of their purity. When passed through a device under test (DUT), the output signal can be analyzed to see what the DUT has added to the pure sine, whether noise, distortion, non-linear level changes, or other changes.



1 kHz, 100 mVrms



1 kHz, 100 mVrms; Ch 2 with 100 mV DC Offset.

DAC Sine

The digitally-generated DAC sine is discussed here; for the APx555, see High Performance Sine Generator on page 51.

The DAC sine waveform has a frequency range up to 80.1 kHz (see the table below). The purity of the waveform is indicated in the system residual THD+N, to which the generator sine signal is a contributor.

The system residual figure varies with Audio Precision hardware, but is better than:

–102 dB, for the APx515
–103 dB for the APx585/586
–105 dB for the APx525/526 (and APx555, when using the DAC generator).

Please check the APx *Installation Instructions and Specifications* booklet for your instrument for detailed specifications.

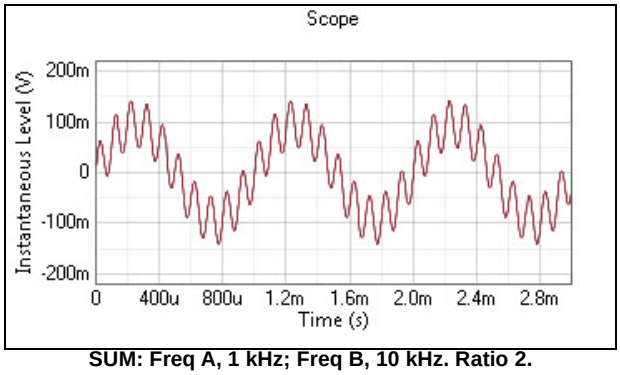
Controls

- **Levels Track Ch 1**
  If Level Tracks Ch1 is checked, all output channels are set to the level entered for channel 1. If it is unchecked, each channel can be set to a different output level.
- **Level**
  Set the generator level here.
- **DC Offset**
  For a DAC generator continuous sine waves, a DC offset can be added to the signal. For analog output, when sine level is zero the maximum offset is $\sqrt{2}$ times the maximum (±) RMS level for the current instrument and output connector. For digital output, when sine level is zero the maximum offset is 1 D (±).
  If **Level Tracks Ch1** is checked, all output channels have a DC voltage added at the level set in the channel 1 **DC Offset** field. If it is unchecked, each channel can be offset by a different DC voltage.
  If the generator level is set to zero (0.000 Vrms or 0 FS), the generator will output DC only (pure DC).

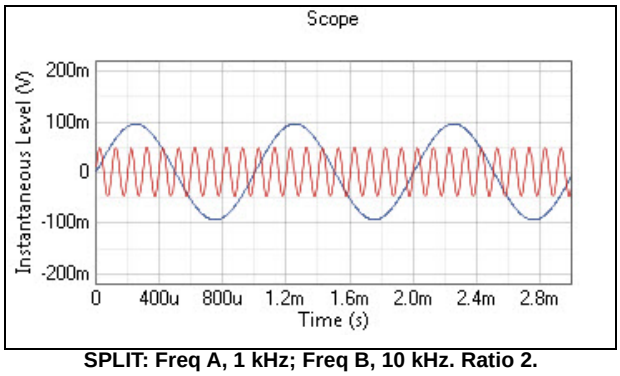
- **Frequency**
  Set the generator frequency here. The DAC generator frequency range is:
  - APx515: 2.00000 Hz to 80.1 kHz
  - APx525/526/582: 0.10000 Hz to 80.1 kHz
  - APx585/586: 5.00000 Hz to 80.1 kHz
  - APx555: 0.00100 Hz to 80.1 kHz
- **Channels**
  Toggle output channels **ON** or **OFF**.

Sine, Dual

Bench Mode and a number of Sequence Mode measurements provide a **Sine, Dual** DSP/DAC generator option, which allows you to modify the sine output in several ways.



SUM: Freq A, 1 kHz; Freq B, 10 kHz. Ratio 2.



SPLIT: Freq A, 1 kHz; Freq B, 10 kHz. Ratio 2.

Controls Specific to Sine, Dual

Frequency A and B

Sine, Dual allows you to set the DSP generator to produce two signals of different frequencies. These can be output independently or summed; see below. Enter the values for **Frequency A** and **Frequency B** here.

Sum

Select **Sum** and enter different frequencies in **Frequency A** and **Frequency B**. The two sine waves are summed to become the output waveform. Use the **FB:FA Level Ratio** control to set the level ratios of the two component sine waves.

Split

Select **Split** and enter different frequencies in **Frequency A** and **Frequency B**. The A and B signals can be routed to different output channels. See Channel Assignments, below.

FB:FA Level Ratio

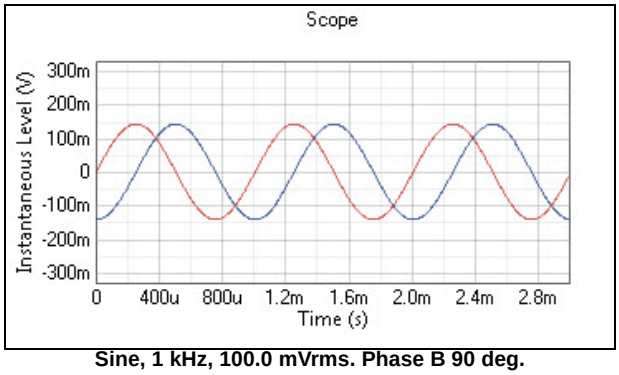
For both Sum and Split, this control sets the ratio of levels between Frequency A and Frequency B signals.

Channel Assignments

For **Split**, use the **Channel Assignments** menu to map frequencies A and B to output channels. By default, frequency A is mapped to odd-numbered channels, and frequency B is mapped to even-numbered channels. For assignments across multiple channels not available in the Menu, select **Custom**.

Sine, Var Phase

Bench Mode and a number of Sequence Mode measurements provide a Sine, Var Phase generator option, which allows you to modify the sine output in several ways.



Sine, 1 kHz, 100.0 mVrms. Phase B 90 deg.

Controls Specific to Sine, Var Phase

Phase B

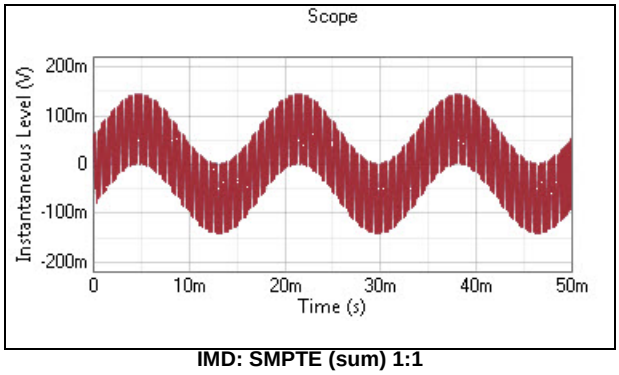
The channel assigned as A is the reference channel. Channels assigned as B can have a phase shift relative to A.

Channel Assignments

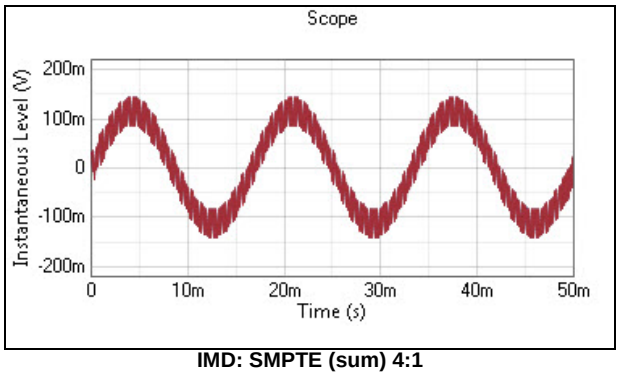
Use the **A/B Channel Assignments** menu to map phase A and phase B to output channels. By default, phase A is mapped to odd-numbered channels, and phase B is mapped to even-numbered channels.

IMD waveforms

Bench Mode and a number of Sequence Mode measurements provide an IMD DSP/DAC generator option, which allows you to modify the sine output to provide IMD (intermodulation distortion) waveforms.



IMD: SMPTE (sum) 1:1



IMD: SMPTE (sum) 4:1

Controls Specific to IMD

DIM signals require the APx analog square wave as a component, restricting DIM to the APx555 and analyzers with the AG52 option installed.

Type

See More about IMD on page 305 for a more detailed description of IMD signals and measurement techniques.

SMPTE 10:1
This is a SMPTE-type stimulus, with a F1:F2 ratio of 10:1. See Frequency 1 and Frequency 2 below.

SMPTE 4:1
This is the standard SMPTE stimulus, with a F1:F2 ratio of 4:1. See Frequency 1 and Frequency 2 below.

SMPTE 1:1
This is a SMPTE-type stimulus, with a F1:F2 ratio of 1:1. See Frequency 1 and Frequency 2 below.

DIN
This is the standard DIN stimulus, with a F1:F2 ratio of 4:1. DIN is quite similar to SMPTE 4:1, using somewhat different frequencies. See Frequency 1 and Frequency 2 below.

DFD/CCIF
DFD and CCIF use the same stimulus tones, with slightly different reporting, discussed in More about IMD on page 305. The tones are of high frequency and close together, placed around a mean frequency and separated by a difference frequency. See Mean Frequency and Diff Frequency, below.

DIM

The DIM waveform is only available for analog outputs in instruments fitted with an analog square wave generator: the APx555, or an APx520, 521, 525, 526 or 582 with option AG52.

DIM 30
DIM 30 mixes a square wave (fixed at 3.15 kHz) with a lower-amplitude sine wave (fixed at 15 kHz). The square wave is filtered by a single-pole low-pass filter at 30 kHz. The ratio of the peak voltage of the square wave to the peak voltage of the sine wave is 4:1.

DIM 100
DIM 100 mixes a square wave (fixed at 3.15 kHz) with a lower-amplitude sine wave (fixed at 15 kHz). The square wave is filtered by a single-pole low-pass filter at 100 kHz. The ratio of the peak voltage of the square wave to the peak voltage of the sine wave is 4:1.

DIM B
DIM B (broadcast) mixes a square wave (fixed at 2.96 kHz) with a lower-amplitude sine wave (fixed at 14 kHz). The square wave is filtered by a single-pole low-pass filter at 30 kHz. The ratio of the peak voltage of the square wave to the peak voltage of the sine wave is 4:1.

DIM B8
DIM B8 mixes a square wave (fixed at 2.96 kHz) with a lower-amplitude sine wave (fixed at 8 kHz). The square wave is filtered by a single-pole low-pass filter at 30 kHz. The ratio of the peak voltage of the square wave to the peak voltage of the sine wave is 4:1.

Frequency 1 and Frequency 2

These controls are for SMPTE and DIN IMD stimulus signals.

For SMPTE, **F1** is typically 60 Hz; **F2** is typically 7 kHz. **F1** range is 40 Hz to 1 kHz; **F2** range is 2 kHz to 60 kHz.

For DIN, **F1** is typically 250 Hz; **F2** is typically 8 kHz. **F1** range is 40 Hz to 1 kHz; **F2** range is 2 kHz to 60 kHz.

Mean Frequency and Diff Frequency

These controls are for DFD/CCIF stimulus signals. **Mean Freq** is typically 12.5 kHz; **Diff Frequency** is typically 80 Hz. **Mean frequency** range is 250 Hz to 60 kHz; **Diff frequency** range is 80 Hz to 2 kHz.

Square Freq and Sine Freq

These are display fields for DIM stimulus signals. **Square Freq** is fixed at 3.15 kHz (2.96 kHz for B signals); **Sine Freq** is fixed at 15 kHz (14 kHz for B, 8 kHz for B8).

Sum

Select **Sum** for typical IMD testing. The two signals are summed to become the output waveform. For SMPTE/DIN, the ratio of F1:F2 is set by the IMD Type selected.

Split

Select **Split** for microphone distortion testing, or other IMD testing that mixed the signals in the DUT or acoustic environment.

Split is useful for microphone distortion testing because a loudspeaker is typically used as a sound source for microphone testing, and loudspeakers have high THD levels compared to the microphone under test, making THD testing impossible. The alternative is to perform IMD testing on the microphone, using two loudspeakers, each outputting a different component of the IMD signal. The mixing of the signals occurs in the microphone.

For SMPTE/DIN, the ratio of F1:F2 is set by the IMD Type selected.

Channel Assignments

For **Split**, use the **Channel Assignments** menu to map frequencies 1 and 2 to output channels. By default, frequency 1 is mapped to odd-numbered channels, and frequency 2 is mapped to even-numbered channels. For assignments across multiple channels not available in the Menu, select **Custom**.

More About Noise Signals

The Noise signals in APx500 are generated in DSP. They have the following characteristics:

White noise

White noise has a power spectrum ideally constant per unit bandwidth from just above DC to half the sampling rate. The crest factor is between 3 and 5, and the probability distribution is close to Gaussian. The generation period (before the waveform repeats) is 2^{32-1} samples long, which is about a day at 48 kHz.

Pink noise

Pink noise is generated from white noise using a filter, and inherits the statistics of the white noise. Pink noise has a power spectrum ideally constant per fractional bandwidth from just above DC to half the sampling rate. The signal power drops off below 10 Hz. The ideal pink power spectrum is maintained ±1.0 dB in the range $2x10^{-4}$ SR to 0.45 SR.

IEC 60268-1 noise

Noise according to the IEC 60268-1 standard is generated from pink noise passed through a weighting filter. IEC 60268-1 has a spectrum that mimics program material, including voice and music.

BS EN 50332-1 noise

BS EN 50332-1 is IEC 60268-1 noise with a reduced crest factor, accomplished by applying soft clipping. The crest factor is between 1.8 and 2.2.
BS EN 50332-1 noise supports headphone testing.

Setting Generator levels when using noise signals

The RMS units assume a sine wave, which has a crest factor of the square root of 2 (approximately 1.414). None of the noise signals have this crest factor, so actual noise values will not be equal to the generator level in RMS-based units.

White noise is generated using a method that guarantees that the peak value is equal to the generator level in peak units; Vp and Vpp will accurately specify the noise level. Note that for digital outputs there is no generator level peak unit.

The filters used in pink noise, IEC 60268-1 noise and BS EN 50332-1 noise alter the signal so that the maximum output cannot be guaranteed in the same way that the white noise can. Signal levels for these noise types will not be equal to the generator level setting for any units, but will be bounded by the Vp setting.

Generator Waveform Files (arbitrary waveforms)

Test signal generation from a waveform file

In addition to the DSP-generated waveforms, APx500 also has the capability to use custom or arbitrary waveforms, loaded into the generator from waveform files available to the computer running APx500. This is particularly useful when coded audio files (dts, Dolby or others) are required to stimulate a decoder in your DUT.

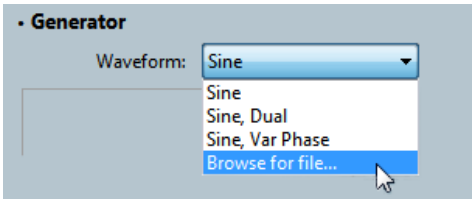
The loaded waveform is “looped” for generator output, so that the end of the waveform is immediately followed by the beginning, providing continuous playback without a jump discontinuity.

Generator waveform files are not restricted to sine waveforms; other waveforms can be used. The Waveform Generator Utility (APxWfmGenerator.exe) allows you to generate .wav files for a wide range of test signals, with selectable sample rates, bit depths and channel counts. It is available from the Audio Precision Web site at “www.ap.com/downloads/file/273”. You are free to copy this utility to other computers or media, where you can generate the audio files you may need as generator waveforms or for external source testing.

You can also create your own generator waveform files. To be useful for measurement, user-created files must be designed with the requirements of the measurement context in mind. Not all audio signals are compatible with certain analyzer measurements nor are useful for measurement. APx500 supports a number of file formats, sample rates and multichannel file formats for Generator Waveform files. See Chapter 96

for a detailed list of supported audio waveform file formats.

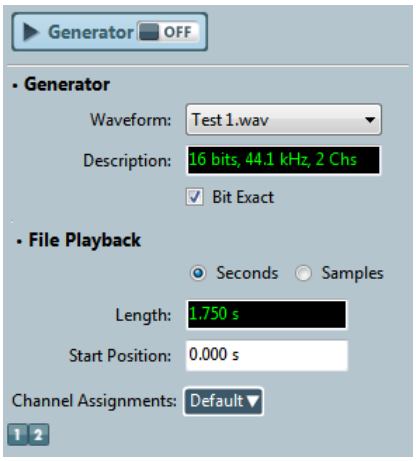
Using a generator waveform file



A measurement that supports generator waveform files displays a Waveform selection list in the Signal Generation panel. Click the Waveform menu in the Generator and choose **Browse for file...** to select an audio file for the generator to use. You can select multiple files.

A selected file becomes the source for that measurement's generator output and is attached to the project file. Once attached to the project, an audio file is available to all other supported measurements as a selection on the Waveform menu. Files attached to the project but not currently used by any measurement can be removed by navigating to the dialog at **File > Manage Attached Project Items...**

Signal Output



Bit Exact

When a digital output is selected, the **Bit Exact** option is available when a linear generator waveform is in use. Checking **Bit Exact** forces the APx generator to output the waveform with no changes whatsoever, with every audio bit exactly as it is in the disk file. Level adjustment is unavailable and dither is off. Coded audio waveforms are always output as **Bit Exact**.

File Playback

When a Generator Waveform file is selected, the File Playback controls appear on the panel.

- **Length** displays the file duration, in seconds or samples.
- **Start Position** allows you to select the point in the file where playback will begin.
- **Channel Assignments** allows you to specify the output channels to which the file channels are assigned.

Analog output: linear audio only

Coded waveforms cannot be decoded to analog audio within the APx500 system.

Linear mono or stereo audio files can be sent to the analog outputs. Mono waveforms are sent to all available output channels. By default, stereo waveforms map channel A audio to odd output channels, channel B audio to even output channels. These assignments can be remapped in the **Advanced Settings** dialog.

Analog output: level adjustment

The level of linear audio files sent to the analog outputs can be adjusted.

Consider that the signal level of the embedded audio in any file is unknown: the file may contain a sine wave at 0 dBFS, or it may contain pink noise at –42 dBFS. This being the case, the signal level at the generator output is also indeterminate. Level adjustments for generator waveform files are calibrated assuming the maximum possible level in the file.

For analog output, a 1 Vrms level setting would output a 0 dBFS sine wave at 1 Vrms; lower signal levels in the file would be output at correspondingly lower voltages.

Digital output: linear or coded audio

Linear mono or stereo audio files and coded audio files can be sent to the digital outputs. Whenever a generator waveform file is selected, the APx500 output sample rate is set to the rate of the file.

Mono waveforms are sent to all available output channels. By default, stereo waveforms map channel A audio to odd output channels, channel B audio to even output channels. These assignments can be remapped in the Advanced Settings dialog. Coded waveforms are not decoded in the APx500 system, but are sent bit exact.

Digital output: level adjustment

When Bit Exact is not set, the level of linear audio files sent to the digital outputs can be adjusted. Level adjustments are not available for coded waveforms, which are send bit exact.

Consider that the signal level of the embedded audio in any file is unknown: the file may contain a sine wave at 0 dBFS, or it may contain pink noise at –42 dBFS. This being the case, the signal level at the generator output is also indeterminate. Level adjust-

ments for generator waveform files are calibrated assuming the maximum possible level in the file.

For digital output, a 0 dBFS level setting would output a 0 dBFS sine wave at 0 dBFS; lower signal levels in the file would be output at correspondingly lower digital levels. A –15 dBFS signal in the file with a –12 dBFS level setting would be output at –27 dBFS.

Dither

For digital output (when **Bit Exact** is not set), a generator waveform file signal is always redithered.

Generator waveform length constraints

File Truncation

Generator waveforms are limited in file length by APx instrument memory resources. A waveform file that exceeds the maximum length will be truncated as it is loaded. A truncated waveform is marked with a warning icon when selected as a generator source.

A generator waveform file is downloaded from the PC into the APx analyzer hardware as needed for a measurement. The hardware memory has a capacity of 128 MB, and audio waveforms that exceed 128 MB will be truncated.

Since APx500 converts 8- and 16-bit PCM waveforms to 32-bit for generator output, and expands coded audio for playback, generator waveform files may use considerably more memory than indicated by their file size on disk.

Truncated waveforms should be avoided, as a jump discontinuity will be introduced on playback at the point of truncation. Best practice is to use the shortest generator waveform file consistent with your measurement requirements.

Maximum File Length table

File formats and sample rates affect the amount of memory used. This table is provided as a guide, indicating the running times that represent the maximum memory available for various formats.

DTS transmit rates

When a legacy dts file (*.dts or *.cpt) with an embedded sample rate of 44.1 kHz is transmitted via HDMI or DSIO, it can sent using an interface transmit rate of 44.1 kHz or 176.4 kHz (4x). If the dts file has an embedded sample rate of 48 kHz, it can be sent using an interface transmit rate of 48 kHz or 192 kHz.

These settings do not convert the sample rate of the embedded audio but only affect how the audio is carried on the interface signal. For DSIO, the **Digital Serial Output Sample Rate** setting must agree with setting made here.

Select

- **Auto** (the default)
  For a 44.1 kHz sample rate file, Auto sets the interface transmit rate to 44.1 kHz.
  For a 48 kHz sample rate file, Auto sets the interface transmit rate to 48 kHz.
- **4x**
  For a 44.1 kHz sample rate file, 4x sets the interface transmit rate to 176.4 kHz.
  For a 48 kHz sample rate file, 4x sets the interface transmit rate to 192 kHz.

Continuous Sweep method

The Continuous Sweep measurement outputs a special broadband stimulus signal to the DUT. The DUT output is acquired by the analyzer and is processed in DSP (digital signal processing) to provide a number of sweep results, all from one acquisition. (The Deviation result is not graphed as a sweep, but is viewed as a single value for each channel).

The Continuous Sweep, Frequency Response and Acoustic Response measurements use the continuous sweep method. Read more about continuous sweep beginning on page 246.

Multitone method

The multitone method uses a stimulus signal made up of many sine waves (a multitone) and analyzes the DUT output with FFT techniques to measure the amplitude and phase of each component sine wave in the result. These values can provide frequency response, phase response, crosstalk response, distortion response, noise response and other results.

Multitones are useful for fast testing applications, and since a multitone stimulus can be brief and unobtrusive, multitone can be considered for in-service testing. Read more about multitone beginning on page 409.

Square wave

The square wave has been used in audio test for many years as a diagnostic tool. A glance at a square wave passed through a DUT and displayed in the time domain (scope) view can reveal a lot about the DUT's coupling, frequency response, slew rate, power supply sag and other characteristics to the experienced eye. In some APx instruments, a square wave is used as a component of the DIM/TIM waveform.

Digitally generated square waves (DSP/DAC) exhibit Gibbs phenomenon ringing artifacts, an unavoidable consequence of mathematics of the DSP generation. In APx instruments, digitally generated square waves are not available for analog output.

The APx555 and an AG52-equipped APx525/526/582 have an analog square wave generator that produces fast rise time square waves with little ringing or overshoot. This waveform is only available to the analog outputs, and also serves as a component of the DIM/TIM waveform.

Controls

Levels Track Ch 1

If Level Tracks Ch1 is checked, all output channels are set to the level entered for channel 1. If it is unchecked, each channel can be set to a different output level.

Level

Set the generator level here.

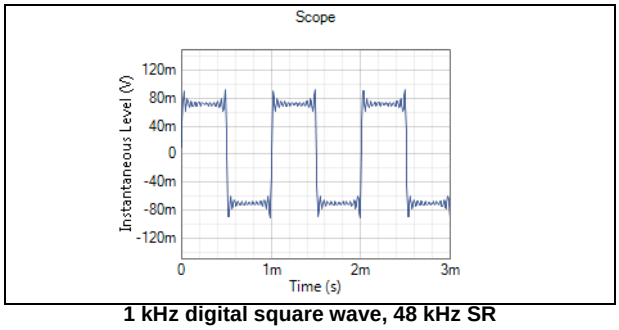
DC Offset

Set the DC Offset here. This control is only available for digitally generated square waves.

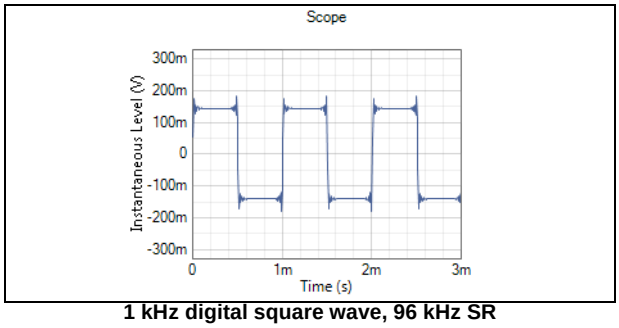
Frequency

Set the generator frequency here. For digital square waves, the range is 100 mHz to SR/6. For analog square waves, the range is 100 mHz to 30 kHz.

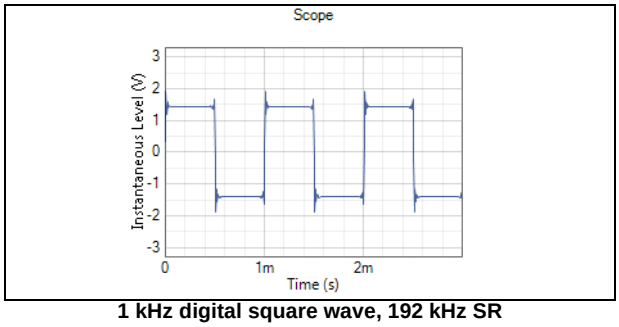
Comparison, digitally generated square waves and analog generated square wave



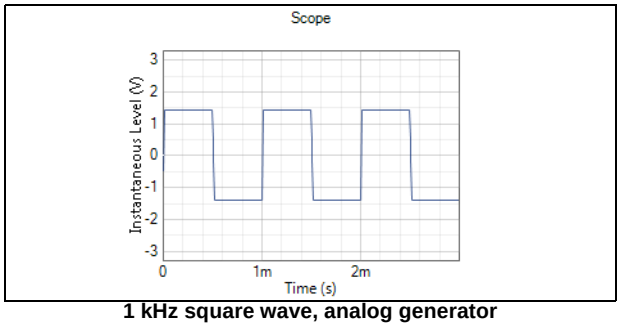
1 kHz digital square wave, 48 kHz SR



1 kHz digital square wave, 96 kHz SR



1 kHz digital square wave, 192 kHz SR



1 kHz square wave, analog generator

Special waveforms for Digital Diagnostics

Walking Bits patterns

The walking bit waveforms are used in diagnosing digital problems in the **Verify Connections** feature of Signal Path Setup. The walking pattern makes it easy to observe “stuck” bits in digital hardware. The walking bits waveforms will also show no errors when passed through a bit-accurate system.

Bit test

Bit Test is a pseudo-random waveform with values uniformly distributed between plus and minus full scale. Algorithms in the generator and analyzer determine sample values.

Bit test exercises a wide range of levels and frequencies and is the most thorough of the bit-accurate waveforms.

Walking Zeros

In the Walking Zeros waveform, all bits are set to 1 except one bit, which is set to 0. This single low bit is continuously incremented from the least significant bit (LSB) to the most significant bit (MSB). When it reaches the MSB it wraps around to the LSB of the selected word length.

Walking Ones

In the Walking Ones waveform, all bits are set to 0 except one bit, which is set to 1. Like Walking Zeros, this single high bit is continuously incremented from LSB to MSB and wrapped back to LSB.

Samples/Step

The time required for one complete cycle of the walking pattern depends upon the output sample rate and the generator Samples/Step value. For example, with Samples/Step set to 48000 and sample rate set to 48 kHz, each step will last for 1 second.

Constant Value waveform

The Constant Value waveform is a continuous stream of data samples at the same fixed value (digital dc). Constant Value mode aids in the investigation of data-dependent errors in digital systems.

Value

Enter the value for the digital data here, in either hexadecimal (base 16) or D units. Specifying the value in hex is often easier, especially if the output bit depth is configured for less than 32 bits. For instance, if the output is configured for 16 bits, the lowest positive constant value is 00010000 hex; this is easier to specify than 30.52 µD.

Note: Constant Value signals of all ones (1s) or all zeros (0s) are treated as invalid Constant Value signals, and will return errors. This implementation serves to flag digital connection errors, which can incorrectly produce a stream of constant ones or zeros. When using Constant Value signals, set the signal to any value except ones or zeros.

Notes on hex values in APx500 Constant Value

1. Truncation of hex numbers at lower bit depths.

The output bit depth selected in Signal Path Setup defines the range of valid hex values for digital levels. For example, a bit depth of 24 requires 6 hex digits (1 hex digit = 4 binary digits). If the bit depth is reduced to 16, only 4 hex digits are required, so the 2 least significant hex digits are not used. If, while bit depth is set to 16, you enter 6 digit hex value (12345600), APx will return the truncated value 12340000.

2. Conversion of hex values to two's complement.

As is almost universal in digital audio systems, APx500 uses two's complement notation for digital audio sample values. Two's complement simplifies digital processing by expressing bipolar signals as positive-only numbers. In APx, hex values are always shown in their two's complement form.

For example, entering '-12345600' with the output configuration in 24-bit mode results in the display 'EDCBAA00 hex', which is the two's complement representation of -12345600 hex.

Units available for Constant Value are

- hex
- D

See Chapter 104 for more information about units of measurement.

Active Bits

When Active Bits is set, any bit that changes state during the diagnostic period (about 1/4 second) is shown as highlighted with color. Inactive bits are shown as white.

Data Bits

When Data Bits is set, the logical state (binary 1 or binary 0) of each bit in the audio word is shown. This is updated at the end of each diagnostic period (about 1/4 second). Binary 1 is shown in highlighted color; binary 0 is shown as white.

Note: The audio word may be any length between 24 bits and 8 bits, depending upon the Bit Depth setting in Signal Path Setup.

Error Rate (digital only)

Read more about bit-accurate (also called bit-exact) measurements beginning on page 277.

The Error Rate display recognizes Audio Precision bit-accurate waveforms, including

- Bit test (usually the best choice for digital error rate measurements.)
- Walking Zeros
- Walking Ones
- Constant Value

When one of these waveforms is received with every sample at precisely the same value as was generated, zero errors are reported and an error rate of 0.000% is displayed. When one of these waveforms is received with sample values different from the values in the original waveform, each differing sample is reported as one error.

Analog square wave generator (APx555 and AG52)

Digitally synthesized square waves (DSP/DAC) exhibit Gibbs phenomenon ringing artifacts. The APx555 and an AG52-equipped APx525/526 have an analog square wave generator that produces fast rise time square waves with little ringing or overshoot.

Signals available

- Square
  A fast rise time analog square wave.
- DIM
  The APx555 and AG52 use the analog square wave generator in combination with the sine generator to produce a DIM IMD waveform.

Analog High Performance Sine Generator (APx555)

The APx555 has a low-distortion RC oscillator which can be selected for sine generation. It is considerably lower in residual harmonic distortion and noise than the DSP/DAC generator, and in combination with the APx555's High Performance Sine Analyzer enables the 555's signature low system residuals.

Signals available

- **Sine** (frequency range 5 Hz to 204 kHz). Very low distortion and extended frequency range.
- **Sine, Burst.**

Sine

The APx555 has a low-distortion RC oscillator which can be selected for sine generation, called the High Performance Sine Generator. It is considerably lower in residual harmonic distortion and noise than the DSP/DAC generator, and in combination with the APx555's High Performance Sine Analyzer enables the 555's signature low system residuals.

The analog sine waveform has a frequency range of 5 Hz to 204.475 kHz. The purity of the waveform is indicated in the system residual THD+N, to which the generator sine signal is a contributor. For the APx555, this figure is better than:

−116.5 dB.

Please check the *APx555 Installation Instructions and Specifications* booklet for detailed specifications.

Controls

Levels Track Ch 1

If Level Tracks Ch1 is checked, all output channels are set to the level entered for channel 1. If it is unchecked, each channel can be set to a different output level.

Level

Set the generator level here.

Frequency

Set the generator frequency here. Range is 5 Hz to 204.475 kHz.

Comparison of the High Performance Sine Generator to the DAC sine generator

Advantages

- Very low THD+N residuals.
- Extended high-frequency range (up to 204 kHz).

Disadvantages

- Split modes, summed modes, offset and other modifications or impairments are not available.
- Frequency accuracy is less precise. (DAC generator 3 ppm; RC oscillator 0.3%).
- RC oscillator cannot be synchronized to a reference.

Sine, Burst

The APx555 provides a general purpose sine burst waveform, which can have an arbitrary sine frequency, an arbitrary **High Time** and **Interval** time, and an arbitrary ratio between **On** and **Off** levels. **Sine, Burst** is only available in the APx555, and only for analog outputs.

Read more about burst signals beginning on page 374.

Controls

Levels Track Ch 1

If Level Tracks Ch1 is checked, all output channels are set to the level entered for channel 1. If it is unchecked, each channel can be set to a different output level.

Level

Set the generator level for the On portion of the burst here.

Frequency

Set the generator frequency here. The **Sine, Burst** generator frequency range is 1 mHz to 80.1 kHz.

Cycles/Seconds

Choose whether to set the burst **High Time** and **Interval** in cycles (at the current sine frequency) or seconds.

High Time

For **Cycles**, set the number of cycles of the sine waveform for the **On** portion of the burst. Minimum is 1; maximum is 16,777,214.

For **Seconds**, set the duration in time for the **On** portion of the burst.

Interval

For **Cycles**, set the number of cycles of the sine waveform for the total interval between burst onsets. Minimum is 2; maximum is 16,777,215.

For **Seconds**, set the duration in time for the total interval between burst onsets.

Low/High Ratio

Set the ratio of the level of the **Off** portion of the burst to the **On** portion of the burst. The level of the **On** portion of the burst is set in the **Level** control.

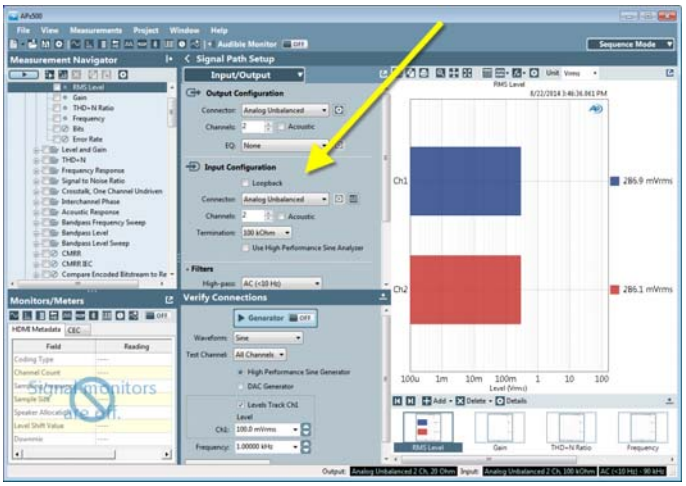
The **Off** portion can be set as high as unity (0 dB), to as low as −80 dB. The low level can also be set to −∞ dB (completely off). To set −∞ dB, enter a value of −10000 dB or smaller.

Channels

Toggle output channels **ON** or **OFF**.

Signal Path Setup

Overview: Signal Path Setup



See Chapter 3, **Introduction to the APx Software** to read more about Signal Paths.

For both Sequence Mode and Bench Mode, **Signal Path Setup** provides a series of menu-selected panels, with access to a collection of tools to select and configure the physical connections between the analyzer and the Device Under Test (DUT) for the current **Signal Path**. This is where you choose your input/output (I/O): digital or analog, balanced or unbalanced, specify impedances or communications protocols, set the number of channels in use, name your inputs and so on. You can also configure references, switchers, DCX, clock sync, triggers and jitter.

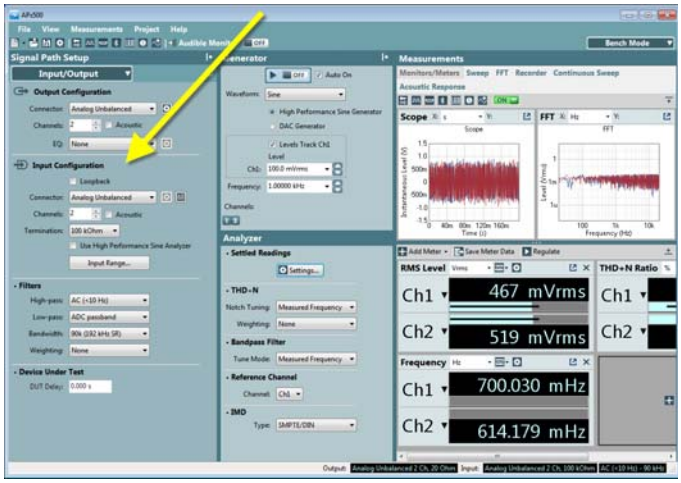
Individual Signal Path Setup panels can be undocked and placed anywhere on the screen.

In **Sequence Mode**, **Signal Path Setup** is in the center of the workspace, along with the control panel for **Verify Connections**. See Chapter 8 for more information about Verify Connections.

In **Bench Mode**, **Signal Path Setup** is on the left side of the workspace.

A drop-down menu provides these selections:

- **Input/Output**, page 54.
  Input and Output configuration, acoustic mode selectors, labels and colors, global input filters, device delay.
- **References**, page 56.
  Output and input level references, mic calibration, frequency reference.
- **Output Switchers**, page 65.
  Output switcher configuration. Available if an output switcher is connected.
- **Input Switchers**, page 65.
  Input switcher configuration. Available if an input switcher is connected.
- **DCX**, page 66.
  Settings for DCX-127 module. Available if a DCX-127 is connected.

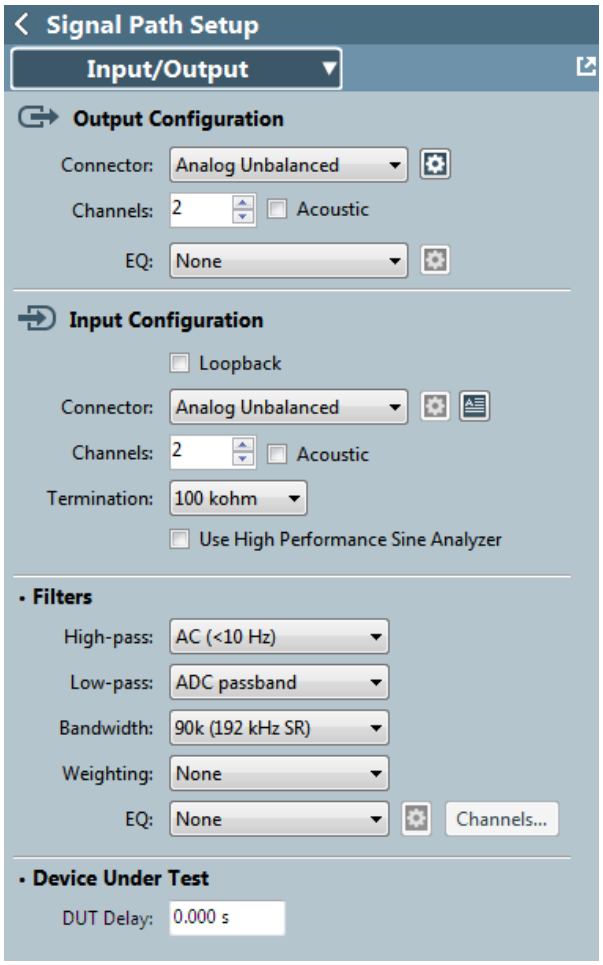


- **Clocks**, page 68.
  Configuration for external reference and synchronization signals, and jitter generation. Available if the Advanced Master Clock is installed.
- **Triggers**, page 70.
  Configuration for trigger output and external trigger input. Available if the Advanced Master Clock is installed.

The Back button

In Sequence Mode, if you have jumped to Signal Path Setup from a measurement, the back button will return you to the measurement.

Signal Path Setup: Input/Output



Dependency upon Analyzer Hardware

APx500 software works with all APx analyzer hardware models and options. The differences in using APx500 with different analyzer hardware models are most apparent in Signal Path Setup, where input and output format, configuration and channel count are dependent upon the analyzer hardware model and options.

APx500, for example, will not show HDMI connection selections when connected to an analyzer that has no HDMI interface installed.

This chapter provides an overview of Signal Path Setup and discusses features common to all analyzer hardware configurations.

I/O Configurations

The Signal Path Setup I/O Configuration panels are where you select the generator output and analyzer input connectors and formats. For each selection, a **Settings** dialog provides complete control over output settings.

Each of these I/O configurations are discussed in detail in the following chapters:

- **Analog I/O** (input/output) configurations for the APx515 analyzer, Chapter 9.
- **Analog I/O** configurations for the APx525 family of analyzers, Chapter 10. APx582 output configuration is also covered in Chapter 10.
- **Analog I/O** configurations for the APx555 analyzer, Chapter 11.
- **Analog I/O** configurations for the APx585 family of analyzers, Chapter 12. APx582 input configuration is also covered in Chapter 10.
- **DIO** (digital I/O) configurations for all analyzers, Chapter 13.
- **ADIO** (advanced digital I/O) configurations, Chapter 14.
- **DSIO** (digital serial I/O) configurations for the APx525, 582 and 585 families, Chapter 15.
- **HDMI** (high definition multimedia interface) and **HDMI ARC** I/O configuration for some models in the APx525, 582 and 585 families, Chapter 16.
- **Bluetooth** wireless technology configurations for APx525, 582 and 585 families, Chapter 17.
- **PDM** (pulse density modulation) configuration for APx525, 582 and 585 families, Chapter 19.
- **Transducer Interface**, which inserts the optional APx1701 Transducer Test Interface accessory between the DUT and the analyzer inputs and outputs, providing power amplification, impedance measurements and microphone interface and powering for electro-acoustic testing. See Chapter 20.
- **ASIO** (audio stream input/output) configuration for all analyzers, Chapter 21. APx515 requires a software option to enable this interface.

For Output Configuration, there is this additional selection:

- **External Source** (open-loop, no generator output) configuration for all analyzers, Chapter 6.

For Input Configuration, there are these additional selections:

- **Loopback**
- **File (Analog Units)** and **File (Digital Units)**, Chapter 22.

Output EQ

For analog outputs, **Output EQ** allows you to insert an equalization filter in the output signal path, after the APx generator. This enables compensation for a non-flat response in the device under test. Read more about **Output EQ** on page 193.

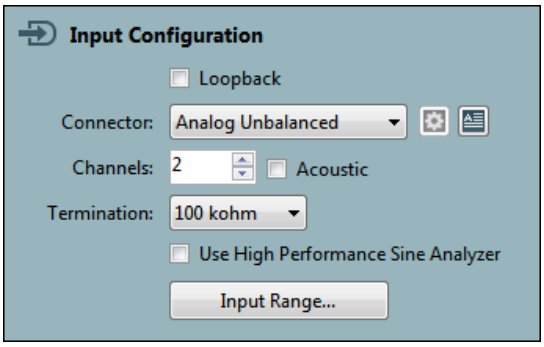
Acoustic Output Configuration

For convenience when performing acoustic tests, APx500 analog outputs can be set to **Acoustic** output configuration. **Acoustic** output configures all generator level setting fields and generator level graph displays to units of pascals or dBSPL. Read more about **Acoustic** output configuration on page 191.

Acoustic Input Configuration

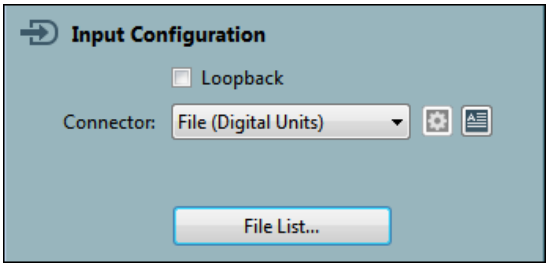
For convenience when performing acoustic tests, APx500 analog inputs can be set to **Acoustic** input configuration. Acoustic input configures all APx500 measurements in the signal path to display levels relative to channel-specific acoustic reference levels. This is useful for making acoustic measurements with multiple microphones. Read more about **Acoustic** input configuration on page 191.

Input Range (Bench Mode only)



For analog inputs in Bench Mode, the **Input Range...** dialog is available to set input ranging. This feature is available in Sequence Mode in the Advanced Settings dialogs for individual measurements. See Chapter 98, Ranging and Settling.

File List (Bench Mode only)



For File Input configuration in Bench Mode, the **File List...** browser is available. This feature is available in Sequence Mode in the Analyzer settings for individual measurements. See Chapter 22, Open Loop measurements.

Filters

Signal Path **Filters** (high-pass, low-pass, weighting and EQ) can be set, which affect the input signal for the entire Signal Path.

> *Note: AC/DC coupling is set by the high-pass filter control.*

Read more about Signal Path Filters and EQ in Chapter 97.

> *Note: for digital inputs when jitter is selected as the measurement source, the Signal Path filters provide different filtering options.*

Device Under Test: DUT Delay

Overview

This feature helps enable problem-free measurements on high-delay devices or systems (such as Bluetooth links or acoustic propagation) by waiting a specified time before analysis begins.

For devices with short delays (<10 ms), APx will perform correctly without any need to add delay in this field.

In a high-delay system, the signal from the APx generator takes some time to propagate through the system to the APx analyzer input.

In some cases, APx may attempt to make a measurement before the generator signal has arrived. In others, the stimulus signal may be truncated.

For the continuous sweep chirp-based measurements, such delay problems may be solved by adjusting the “Extend Acquisition By” control, but this must be adjusted for each measurement. DUT Delay, however, operates on all measurements in the signal path.

Entering a DUT Delay value

There are three ways to enter a value into the DUT Delay field. Note that values entered into DUT Delay, from any source, are overwritten by any subsequent values, from any source.

Manual Entry

If you know the amount of delay the DUT provides, you can type that value directly in the DUT Delay field on the Signal Path Setup panel. You may know the delay value from a device specification, from a previous measurement, or, for acoustic paths, from a measurement of the distance between the transducers.

Using the Measure and Set DUT Delay step

In a sequence, this step is available in Sequence Steps folder under the Signal Path Setup node. This step measures the DUT delay using a cross correlation method, and enters the result in the DUT Delay field. Read more about the **Measure and Set DUT Delay** step beginning on page 541.

Using the A/V Sync Delay Bluetooth feature

If you have the Bluetooth Duo option and are using Bluetooth as an A2DP Source, the AVRCP A/V Sync Delay feature is supported. A/V Sync Delay enables a Bluetooth rendering device (a loudspeaker, for instance) to report its internal delay to the Source device for A/V Sync correction. If the Bluetooth device to which you are connected supports A/V Sync Delay, and you have enable Enter A/V Sync Delay into DUT Delay in your Bluetooth settings, the device delay value will be entered into the DUT Delay field.

The value is reported from the sink device periodically and may vary.

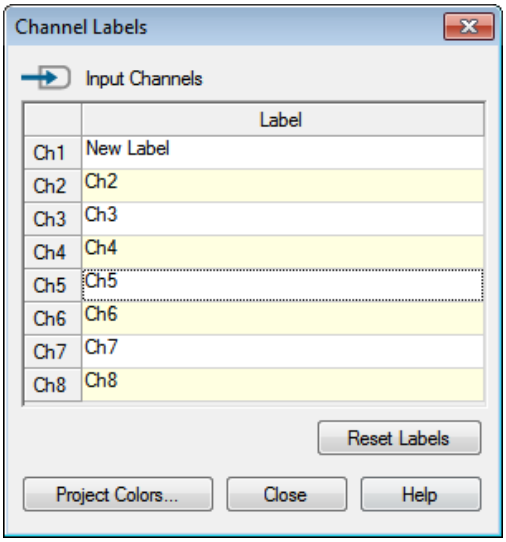
Making other delay measurements with DUT Delay set

The DUT Delay measurement (Chapter 44) temporarily sets the Signal Path Setup DUT Delay value to zero while the measurement is running.

However other methods of measuring delay using APx, such as Impulse Response in Continuous Sweep, do not take the DUT Delay setting into account. For instance, if the DUT Delay field is set to 50 ms, and the measured delay of the device is 22 ms according to the Impulse Response result in Continuous Sweep, then the actual device delay is 72 ms.

Input Labels

In **Signal Path Setup**, click the Labels button to open the Channel Labels dialog.



Here you can rename the input channels with names of your choice. These settings are project-wide, but can be overridden for any measurement result.

For local bar graph (meter) label overrides, see page 640. For local XY graph label overrides, see page 648.

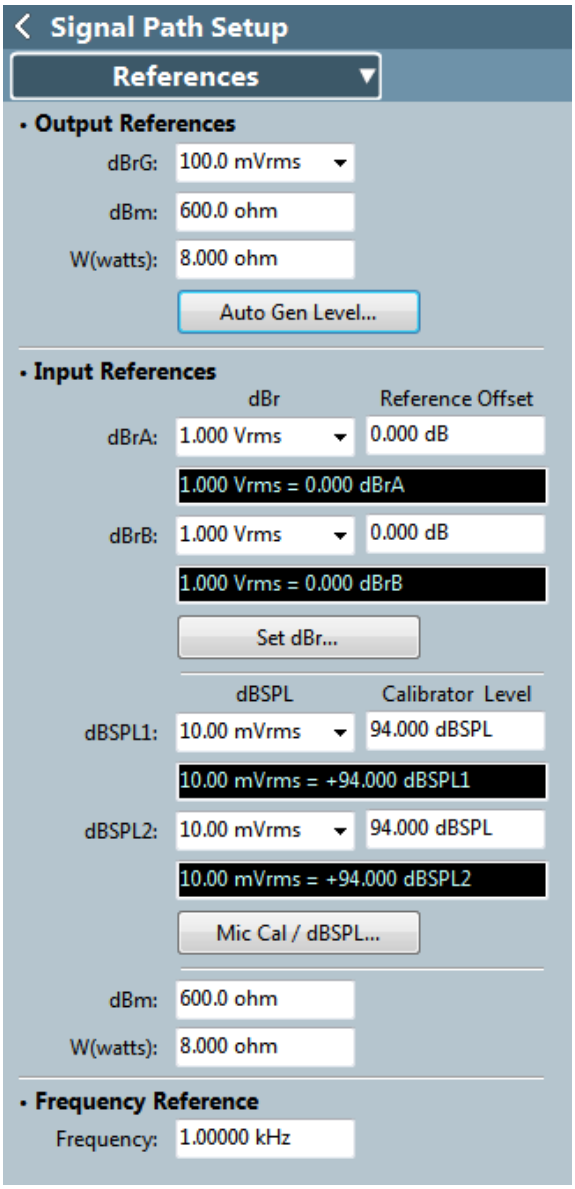
To set project colors and line styles, click the **Project Colors** button in the **Labels** dialog, This opens the Project Properties Colors tab, where you can set graph colors for each channel, or set a color cycle for multiple Data Sets. Read about setting colors on page 30.

Signal Path Setup: References

The References dialogs and options are quite different in a normal I/O mode, when compared to an Acoustic I/O mode. References in normal mode are discussed first.

The References in acoustic mode (when an Acoustic checkbox is checked in **Signal Path Setup > Input/Output**) are discussed beginning on page 62.

References: (Normal Mode)



A signal path through a DUT will have an optimal operating level for each measurement. This level may be specified in a device's specifications, required for a standard measurement method, or may be deter-

mined on the bench by a monitoring a parameter such as distortion.

If, for example, you select a non-referenced unit such as dBV as the unit in a meter display, you will see decibels relative to 1 volt; but if you select the reference unit dBrA, you will see decibels relative to the reference value you have entered.

Output References

If you would like to force all generator settings to acoustic units referenced to an acoustic level, see Acoustic Output configuration on page 62.

dBrG:

You can save a generator output reference level in each domain for a signal path. The saved generator level is called **dBrG**, and can be referenced throughout the signal path when making generator settings by selecting **dBrG** from the units menu.

You can enter a known value as **dBrG** here, or you can use the **Auto Gen Level** feature (see below) to find the optimal generator level setting for your test.

Alternatively, you can determine **dBrG** manually by adjusting the generator level in V**erify Connections** while observing the measured level or measured distortion on the graph, until you achieve your desired results. Once you have found your optimal level, you can copy the value and paste it in the **dBrG** field.

Default values are

- analog generator: 0 dBrG = 100.0 mVrms
- digital generator: 0 dBrG = –20 dBFS

Note: When using dBrG as a unit in a measurement, a setting greater than 0 dBrG is not allowed.

dBm:

Enter a **dBm** output reference impedance here. **dBm** is available as a level unit throughout APx. **dBm** is a power unit, and as such must be referenced to a nominal circuit impedance to be meaningful. **dBm** is typically referenced to 600 Ω. You can enter a different value here.

W (watts):

Enter a **W (watts)** output reference impedance here. **W (watts)** is available as a level unit throughout APx. **W (watts)** is a power unit, and as such must be referenced to a nominal circuit impedance to be meaningful. By default, **W (watts)** is referenced to 8.000 Ω. You can enter a different value here.

Auto Gen Level

The **Auto Gen Level...** button opens a dialog where you can automatically set the **dBrG** by regulating to a target level or distortion. You can read more about the Automatically Set Generator Level beginning on the next page.

Input References

If you would like to force all results to acoustic units referenced to an acoustic level, see Acoustic Input configuration on page 62.

dBrA and dBrB:

You can save two analog input reference levels and two digital input reference levels for a signal path, so that results can be evaluated relative to references. These saved input level references are called **dBrA** and **dBrB**, and are independently set for analog and digital inputs. These can be referenced throughout the signal path when making result settings by selecting **dBrA or dBrB** from the units menu. You can set **dBrA** and **dBrB** arbitrarily, or you can set them interactively with your DUT, while observing measured level or distortion.

Reference Offset

Once a reference has been entered or set, you can offset the level by some arbitrary amount.

Set dBr

This button opens the **Set dBrA, B** dialog, where you can interactively set **dBrA** or **dBrB** from input levels. You can read more about the Set dBrA, B dialog beginning on the next page.

dBSPL1 and dBSPL2

If you are making acoustic measurements, you may want to use **dBSPL** references instead of **dBr** references.

You can save two analog input reference levels and two digital input reference levels for a signal path using sound pressure level (SPL) units, so that results can be evaluated relative to acoustic levels.

These saved input acoustic level references are called **dBSPL1** and **dBSPL2**, and are independently set for analog and digital inputs. These can be referenced throughout the signal path when making result settings by selecting **dBSPL1** or **dBSPL2** from the units menu. You can set **dBSPL1** and **dBSPL2** arbitrarily, or you can set them interactively with your DUT, while observing measured level.

Typically, **dBSPL** is referenced to a measurement microphone and a microphone calibrator, in APx External Source configuration.

Calibrator Level

Microphone calibrators operate at one or more specific acoustic levels, typically 94 dBSPL or 114 dBSPL. Enter the calibrator level here.

Mic Cal / dBSPL

This button opens the **Microphone Calibration /Set dBSPL1, dBSPL2** dialog, where you can interactively set dBSPL1 or 2 from input levels. You can read more about this dialog beginning on page 60.

dBm:

Enter a dBm input reference impedance here. The default is 600.0 Ω.

W (watts):

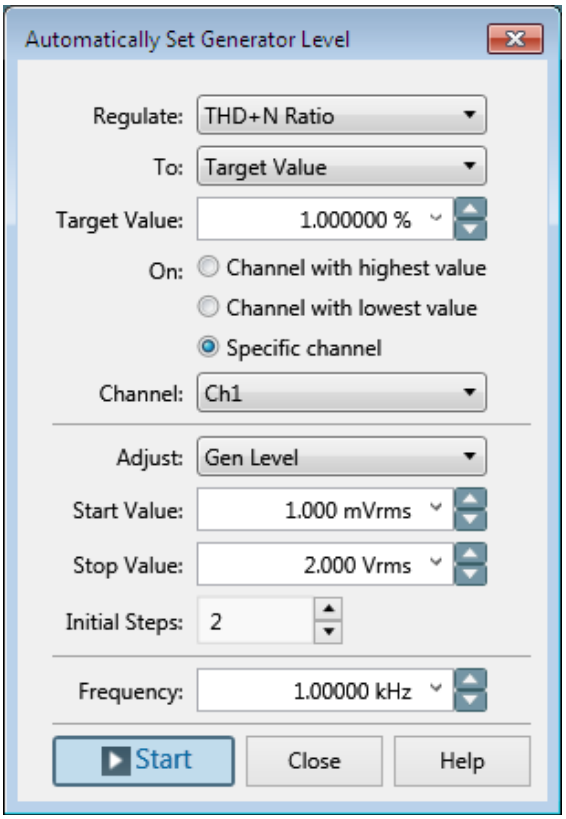
Enter a watts input reference impedance here. The default is 8.000 Ω.

Frequency Reference

Frequency:

Enter a global reference frequency here. The default is 1.0000 kHz.

Automatically Set Generator Level (Auto Gen Level)



Also **see** Reference Levels for general information about references, and Set Acoustic Output Level for a similar feature in Acoustic Output configuration.

Auto Gen Level uses generator regulation to automatically determine the APx generator level that results in a specified RMS Level or THD+N Ratio in the DUT output to satisfy your test requirement. The result is entered in the **References** panel as the **dBrG** value. Read more about regulation on page 481.

Operation

Your DUT should be connected and set up for testing.

From **Signal Path Setup/References**, click **Auto Gen Level...** to open the Automatically **Set Generator Level** dialog. In APx500 4.4, this dialog has been refactored. The regulation functions remain essentially the same, but controls have been clarified and new controls added.

Regulate:

This control selects the parameter in the DUT output you want to regulate. Choose

- **RMS Level** or
- **THD+N Ratio**

To:

This control selects the value you want to attain in the regulated parameter. Choose

- **Target Value**
  Find a specific **Target Value**, set in the following field.
- **Maximum Value**
  Find the **Maximum Value**
- **Minimum Value**
  Find the **Minimum Value**

Target Value:

If **Target Value** is selected in the previous control, enter the value you want to attain here.

On:

This control selects the DUT channel to be regulated. Choose

- **Channel with the highest value**
  All available analyzer input channels are monitored, and the channel with the highest value of the regulation parameter chosen in **Regulate** (see above) is selected for regulation purposes.
- **Channel with the lowest value**
  All available analyzer input channels are monitored, and the channel with the lowest value of the regulation parameter chosen in **Regulate** (see above) is selected for regulation purposes.
- **Specific channel**
  Choose one of the available analyzer input channels.

Channel

If **On** is set to **Specific Channel**, select the input channel here.

Adjust:

The **Adjust** control selects the APx controlled parameter that will be adjusted to find the regulated DUT parameter. For **Auto Gen Level** the only choice is **Generator Level.**

Start Value:

The **Start Value** is the first generator setting where a measurement is made. If it is lower than the **Stop Value,** then it is also the lowest generator setting where a measurement is made. If it is higher than the **Stop Value,** then it is the highest generator setting where a measurement is made.

The regulation search proceeds from Start Value to Stop Value; your choices for these values determine the direction (up or down) of the search.

Stop Value:

If the **Stop Value** is higher than the **Start Value**, then the **Stop Value** is the highest generator setting where a measurement is made. If the **Stop Value** is lower than the **Start Value**, then the **Stop Value** is the lowest generator setting where a measurement is made. The **Stop Value** is also the last generator setting where a measurement is made in the initial search for an interval containing a measured value of the type that is sought.

The higher of Start Value or Stop Value also serves to limit the maximum APx generator output, protecting your DUT.

Initial Steps:

Initial Steps sets the number of intervals dividing the generator level range between **Start Value** and **Stop Value**. The default setting is **2**. Depending upon the parameter to be regulated and characteristics of the DUT, greater values may provide better performance.

Frequency:

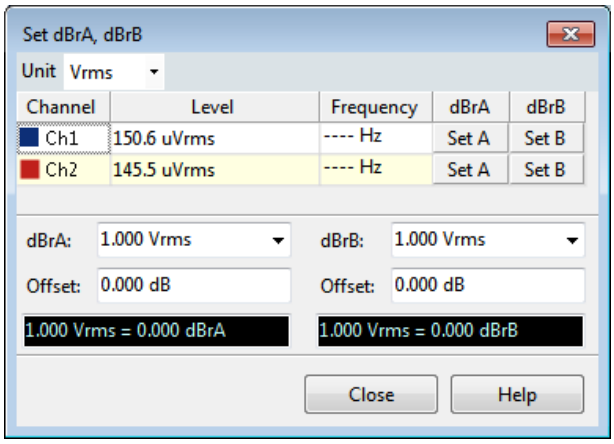
The **Frequency** control sets the APx generator frequency for regulation.

For more about **Start Value**, **Stop Value**, and Initial Steps, read more about regulation on page 481.

Note: Distortion measurements such as THD+N benefit from filtering. When regulating THD+N, consider using the Input Filters in Signal Path Setup > Input/Output to filter the signal used for regulation. Suggested filters are High-pass: 20 Hz, Low-pass: 22 kHz. Remember to reset the filters when you continue to other measurements. See Chapter 97, Bandwidth, Weighting and EQ Filters for more information.

Set dBrA, B dialog

The **Set dBr A, B** dialog allows you to interactively set dBrA or B from input levels.



You can save two analog input reference levels and two digital input reference levels for each signal path. These saved input level references are called **dBrA** and **dBrB**, and are independently set for analog and digital inputs. They can be referenced throughout the signal path so that results can be evaluated relative to references. You can set dBrA and dBrB arbitrarily, or you can set them in interaction with your DUT, while observing measured level or distortion.

If you are making acoustic measurements, you may want to use **dBSPL** references (see page 60). For acoustic measurements using multiple microphones, it may be more useful to set sound pressure level references specific to each microphone and input channel. See Microphone Calibration / Set dBSPL1, dBSPL2 on page 64.

Setting dBrA and dBrB

If you have not yet set up your test, first go to Signal Path Setup to configure your inputs and outputs. See page 54.

Using the generator

In Sequence Mode, use the generator in Verify Connections and the RMS Level or THD+N ratio graphs attached to Signal Path Setup. In Bench Mode, use the Generator and the RMS Level or THD+N Ratio monitor meters.

With your DUT connected and adjusted for testing,

- Click the **Set dBr...** button in the References panel. This opens the **Set dBrA, dBrB** dialog, shown here.
- Turn the Generator **ON**, using a sine waveform.
- Set the Generator and the DUT level controls to produce a DUT output level appropriate to your test. This may involve first using **Auto Gen Level** (see page 58) to set an optimal generator output level, or it may be that you need to make your measurements relative to some standard DUT output level, such as 1 volt rms.

- Use **RMS Level** or **THD+N Ratio** as the graph display (Bench Mode: monitor meter), depending upon which result you want to observe. If you are setting input levels in relation to a specific distortion figure, see the Distortion Target paragraph below.
- Choose a channel that exhibits the level or distortion you want as a reference, and click **Set A** or **Set B** to enter that value as a reference level.
- By default, the level you select will be referenced to the value of 0.000 dBrA (or dBrB). You can redefine how you reference the selected level by entering a value other than 0.000 in the **Offset** field. The display shows the result of your choices.

Using External Source

Setting input reference levels from an external source is slightly different from the operation described above, which uses the internal generator. First, be sure you are configured for External Source. Go to Signal Path Setup and select **None (External)** for the output connector in Output Configuration.

When using an external source, instead of using the generator you will play an audio signal from the DUT.

For setting references, you can use any mid-band audio sine wave signal.

Once the external signal is ON, setting an input reference is the same as using the generator: the RMS Level is shown on the meter bar display, and you can click the Set A or Set B buttons in the Set dBrA, dBrB dialog when your target level or THD+N reading is reached.

Read more about External Source in Chapter 22.

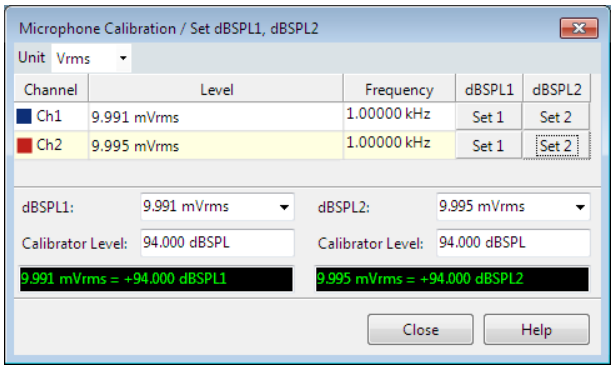
Using a Distortion Target to help set a reference level

If you would like to define your level reference by specifying a maximum acceptable distortion rather than a particular level, select the THD+N Ratio graph (Bench Mode: meter) to read the measured THD+N for the chosen channel.

To help you determine if the distortion exceeds your specification, you can set a limit on the THD+N display. In Bench Mode, set a limit in the usual way. In Sequence Mode, you can enter a value in the **Target THD+N**: field on the graph.

When the measured distortion for a channel exceeds this value the result is flagged in red. The distortion measurements can be filtered using the filters available in the **Signal Path Setup > Input / Output**. See **Signal Path Setup > Filters** on page 617 for more information.

Microphone Calibration /Set dBSPL1, dBSPL2 dialog



Microphone Calibration / Set dBSPL dialog (normal mode).

You can save two analog input reference levels and two digital input reference levels for a signal path using sound pressure level (SPL) units, so that results can be evaluated relative to acoustic levels. These saved input acoustic level references are called **dBSPL1** and **dBSPL2**, and are independently set for analog and digital inputs. These can be referenced throughout the signal path when making result settings by selecting dBSPL1 or dBSPL2 from the units menu. You can set dBSPL1 and dBSPL2 arbitrarily (typically using information from a microphone data sheet), or set them from TEDS data, or you can set them interactively with your DUT, while observing measured level.

Note: TEDS data are only available when using the APx1701 with a TEDS-enabled measurement microphone.

If you would like to force all results to acoustic units referenced to an acoustic level, see Acoustic Input configuration, beginning on the next page.

Typically, dBSPL is referenced to a measurement microphone and a microphone calibrator, in APx External Source configuration, without using the APx generator.

Setting dBSPL with a calibrator

With your measurement microphone (and preamplifier, if required) connected and adjusted for testing,

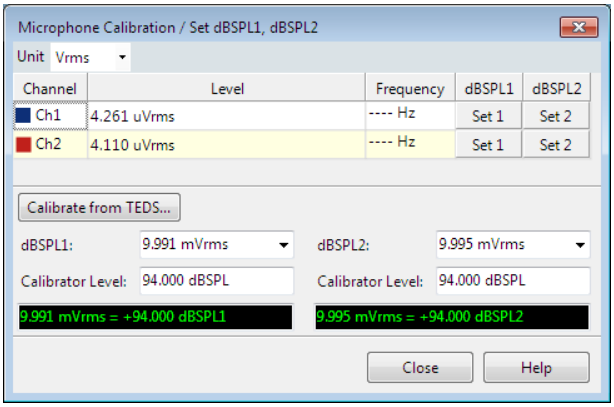
1. Set up the measurement microphone and microphone calibrator following the directions included with the microphone and calibrator. Turn on the microphone and connect it to the analyzer input.
2. Note the calibrator acoustic output level. Depending upon the calibrator, this might be fixed at one level, or switchable. 94 dBSPL and 114 dBSPL are two common calibrator acoustic output levels.
3. Navigate to the References panel.

4. Click the **Set dBSPL...** button on the References panel. This opens the **Set dBSPL1, dBSPL2 / Microphone Calibration** dialog.
5. Enter the nominal calibrator acoustic output level into the Calibrator Level field. By default, this is 94 dBSPL.
6. Mount the calibrator on the microphone connected to Channel 1 and turn it on.
7. The calibrator frequency is measured and displayed in the **Frequency** field. The mic signal is bandpass filtered at the measured calibrator frequency to reduce extraneous low-frequency noise.
8. Click the **Set 1** or **Set 2** button for Channel 1. You can set either reference, dBSPL1 or dBSPL2 from this value.
9. If you have a second measurement microphone you would like to reference, move the calibrator to the second microphone.
10. Click the **Set 1** or **Set 2** button for Channel 1. You can set either reference, dBSPL1 or dBSPL2 from this value.

Setting dBSPL without a calibrator

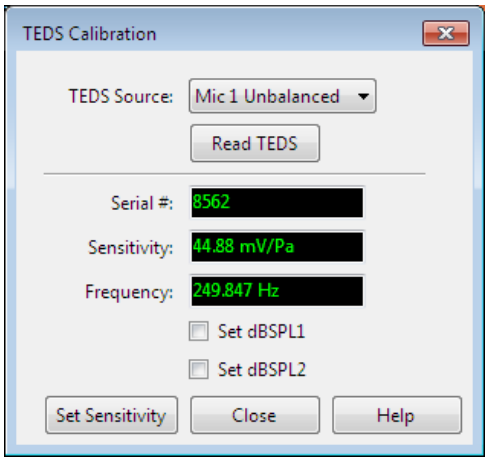
If a calibrator is not available, you can enter nominal references in the dBSPL1 and dBSPL2 fields.

If you are using an APx1701 and a TEDS-enabled measurement microphone, a **Calibrate from TEDS** button will be available on the mic cal dialog.



Microphone Calibration / Set dBSPL dialog (normal mode, with Calibrate from TEDS button).

Open the Calibrate from TEDS dialog to use TEDS data to set dBSPL1 and dBSPL2 references.



Note: TEDS data are only available when using the APx1701 with a TEDS-enabled measurement microphone.

With a TEDS-enabled measurement microphone connected to an APx1701 as part of the analyzer system, you can use the TEDS data to enter calibration information into the APx Mic Cal / dBSPL references.

Select the APx1701 unbalanced microphone input which is connected to the TEDS microphone.

Choices are:

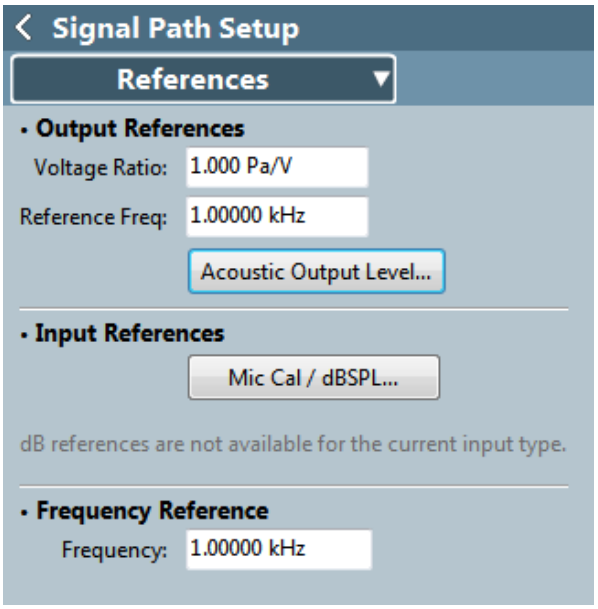
- **Mic1 Unbalanced**
- **Mic2 Unbalanced**

Click **Read TEDS**. The Serial #, Sensitivity and Frequency fields will be populated with the microphone's TEDS data.

When input "normal" mode, you can assign the reference value to either or both dBSPL1 and dBSPL2. Select checkbox(es) and click **Set Sensitivity**.

When in input Acoustic mode, you can assign the reference value to any available analyzer input channel. Choose the channel from the **Apply To** menu. Click **Set Sensitivity**.

Signal Path Setup: References (Acoustic Modes)



Output References (Acoustic)

When the **Acoustic** checkbox in Signal Path Setup: Output Configuration is checked, these Output References selections are available. Checking **Acoustic** expresses the generator levels in acoustic units throughout the signal path.

Voltage Ratio

Voltage Ratio displays the ratio of acoustic level in Pa to electrical level in V, which is established in the Set Acoustic Output Level dialog.

Alternatively, you can enter a known voltage ratio directly in the Voltage Ratio field.

Reference Freq

Reference Freq displays the frequency at which the voltage ratio was established, which is established in the Set Acoustic Output Level dialog.

Alternatively, you can enter a known reference frequency directly in the Reference Freq field.

Acoustic Output Level...

Click this button to open the **Set Acoustic Output Level** dialog. You can read more about this dialog in the next topic.

Input References (Acoustic)

When the **Acoustic** checkbox in Signal Path Setup: Input Configuration is checked, this Input References selection is available. Checking **Acoustic** expresses the analyzer levels in acoustic units throughout the signal path.

Mic Cal / dBSPL...

Click this button to open the **Microphone Calibration / Set dBSPL Per-channel** dialog. You can read more about this dialog beginning on the next page.

Frequency Reference

Frequency:

Enter a global reference frequency here. The default is 1.0000 kHz.

Set Acoustic Output Level

This topic discusses setting acoustic output references when in Acoustic Output configuration. See previous topics in this chapter for more information about Acoustic Output and References.

Auto Gen Level (page 58) is a similar feature used when not in Acoustic Output configuration.

Set Acoustic Output Level uses generator regulation to automatically determine the APx generator level that results in a specified acoustic level (the **Target Value**) in the acoustic test space. Since the analyzer outputs are voltage outputs and acoustic levels are measured in Pa or dBSPL, an acoustic output signal path must have scaling factor associated with it. This is called the **Voltage Ratio**, and it is expressed in Pa/V. The Voltage Ratio is set interactively here in the **Set Acoustic Output Level** dialog by regulating the generator output to the desired acoustic **Target Value**, and is entered in the **References** panel as the **Voltage Ratio** value. Read more about regulation on page 481.

Operation

Your acoustic devices should be placed in the acoustic test space and set up for testing.

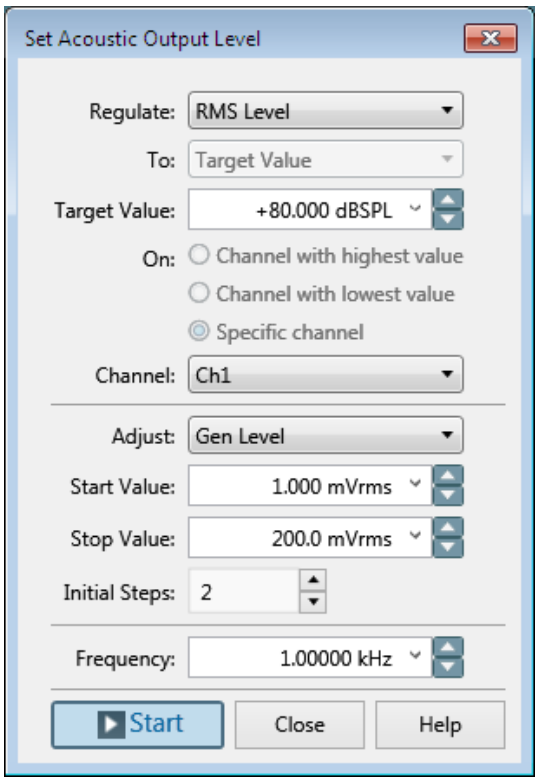
Typically, these include a loudspeaker and a measurement microphone placed in an acoustic testing space, properly oriented and placed at a standard distance from each other. The loudspeaker is connected to the APx generator output via a power amplifier (such as the APx1701), and the microphone is connected to an APx analyzer input.

Before using **Set Acoustic Output Level**, a microphone calibration must be performed to establish the microphone sensitivity.

Once you have set up your acoustic devices, proceed to **Signal Path Setup > Input/Output** and check the **Acoustic Output** and **Acoustic Input** checkboxes in **Output** and **Input Configuration**. The **Acoustic** checkboxes are only available for analog configurations.

In **Signal Path Setup > Input/Output > Output Configuration**, or on the **References** panel, click **Acoustic Output Level...** to open the **Set Acoustic Output Level** dialog. In APx500 4.4, this dialog has been refactored. The regulation functions remain essentially the

same, but controls have been clarified and new controls added.



Regulate:

This control selects the parameter in the DUT output you want to regulate. For Set Acoustic Output Level, there is only this choice:

- **RMS Level**

To:

This control selects the value you want to attain in the regulated parameter. For Set Acoustic Output Level, there is only this choice:

- **Target Value**
  Find a specific **Target Value**, set in the following field.

Target Value:

Enter the **Target Value** you want to attain here. The **Target Value** is the sound pressure level you would like to attain. Be sure that the APx Input Configuration is set to **Acoustic**, and that your microphone is calibrated and set as an APx **Acoustic** input reference.

On:

This control selects the DUT channel to be regulated. For **Set Acoustic Output Level,** there is only this choice:

- **Specific channel**
  Set the channel to which the measurement microphone is connected in the Channel **field**.

Channel

Select the measurement microphone input channel here.

Adjust:

The **Adjust** control selects the APx controlled parameter that will be adjusted to find the regulated DUT parameter. For **Set Acoustic Output Level**, there is only this choice:

- **Gen Level**

Start Value:

The **Start Value** is the first generator setting where a measurement is made. If it is lower than the **Stop Value**, then it is also the lowest generator setting where a measurement is made. If it is higher than the **Stop Value**, then it is the highest generator setting where a measurement is made.

> *The regulation search proceeds from Start Value to Stop Value; your choices for these values determine the direction (up or down) of the search.*

Stop Value:

If the **Stop Value** is higher than the **Start Value**, then the **Stop Value** is the highest generator setting where a measurement is made. If the **Stop Value** is lower than the **Start Value**, then the **Stop Value** is the lowest generator setting where a measurement is made. The **Stop Value** is also the last generator setting where a measurement is made in the initial search for an interval containing a measured value of the type that is sought.

> *The higher of Start Value or Stop Value also serves to limit the maximum APx generator output, protecting your amplifier and loudspeaker.*

Initial Steps:

Initial Steps sets the number of intervals dividing the generator level range between **Start Value** and **Stop Value**. The default setting is **2**. Depending upon the characteristics of the DUT, greater values may provide better performance.

Frequency:

The **Frequency** control sets the frequency of the stimulus signal for regulation. By default, it is 1 kHz. Lower or higher frequencies might be more appropriate for various driver or loudspeaker types, or when Output Equalization is applied.

For more about **Start Value**, **Stop Value**, and **Initial Steps**, read more about regulation on page 481.

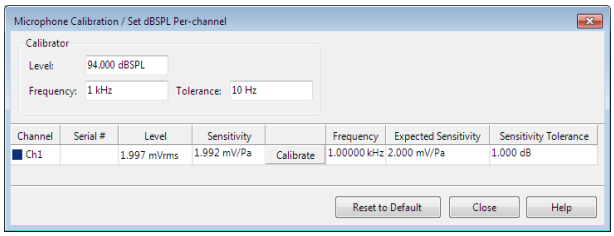
Units

Units available for Acoustic Output and Acoustic Input configuration are

- Pa
- dBSPL

Read more about units in chapter 104.

Microphone Calibration / Set dBSPL Per-channel dialog



This topic discusses microphone calibration and setting dBSPL input references when in Acoustic Input configuration.

Since the analyzer inputs are voltage inputs, an acoustic input signal path must have scaling factor associated with it. This is the **Sensitivity** input reference, expressed in V/Pa. Sensitivity is set interactively (or entered as a known value from the microphone specification) here in the **Mic Cal / dBSPL** dialog, and is displayed in the References (acoustics) dialog.

Microphone Calibration / Set dBSPL Per-channel

For this topic to be relevant, you must go to Signal Path Setup > Input Configuration and set a check in the Acoustic checkbox to enter Acoustic Input configuration.

In Acoustic Input configuration, you can set input reference levels in acoustic units (dBSPL or Pa) independently for each input channel. This allows you to use multiple microphones in an acoustic measurement, where each microphone has been independently referenced to a microphone calibrator.

Setting dBSPL with a microphone calibrator

Calibration of a measurement microphone in an acoustic signal path requires a measurement microphone and a microphone calibrator.

The calibrator frequency is measured and displayed in the Frequency field. The mic signal is tightly bandpass filtered at the measured calibrator frequency to reduce extraneous noise.

1 Set up the measurement microphone(s), preamplifiers and acoustic calibrator following the directions included with the microphones and the calibrator. Turn on the microphones and preamps. Connect the preamp outputs to APx inputs.

2. Note the calibrator acoustic output level. Depending upon the calibrator hardware, this might be fixed at one level, or switchable. 94 dBSPL and 114 dBSPL are two common calibrator acoustic output levels.
3. In APx, go to Signal Path Setup > Input/Output and check the Acoustic checkbox in Input Configuration. You must be set to an analog input to see the Acoustic checkbox.
4. Go to Signal Path Setup > References and click the Mic Cal / dBSPL... button in Input References. This opens the Microphone Calibration / Set dBSPL Per-channel dialog.
5. Enter the nominal calibrator acoustic output level (check the calibrator instructions) into the Calibrator Level field. By default, this is 94 dBSPL.
6. Optional: Enter the nominal calibrator output frequency into the Calibrator Frequency field. Typical calibrator frequencies are 250 Hz or 1 kHz.
7. Optional: Set an acceptable tolerance for frequency in the Tolerance field.
8. Optional: Set the nominal microphone sensitivity in the Expected Sensitivity field.
9. Optional: Set an acceptable tolerance for sensitivity in the Sensitivity Tolerance field.
10. Optional: Enter a serial number or other identifying information in the Serial Number field.
11. Mount the calibrator on the first microphone and turn it on.
12. Click the Calibrate button for Channel 1.
13. Move the calibrator to the second microphone.
14. Click the Calibrate button for Channel 2.
15. Repeat this procedure until a reference level has been set for each microphone.

If you have entered nominal frequency and expected sensitivity values, calibration results that fall within the tolerances you have set will be accepted without comment. Values outside those ranges will be marked in red.

Reset to Default

This button sets the Calibrator Frequency, Calibrator Tolerance, Expected Sensitivity and Sensitivity Tolerance fields to their default settings.

Units

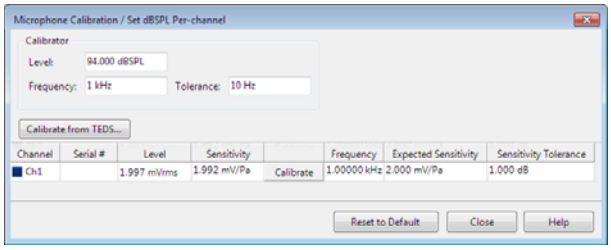
Units available for setting Reference Levels in Acoustic input are

- dBSPL
- Pa

Setting dBSPL without a calibrator (entering a known value)

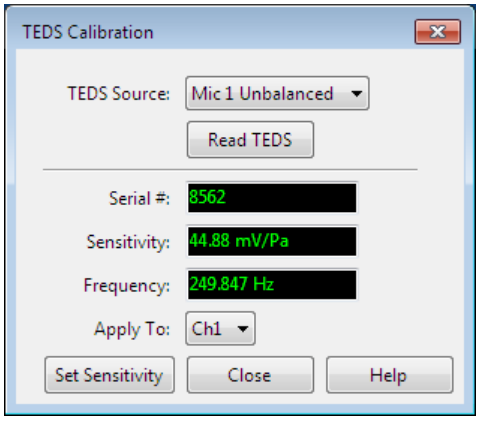
If performing an acoustic calibration of the measurement microphone is not possible, you can enter the microphone's known sensitivity (from a specification or from a previous calibration) in the Sensitivity field for a reference setting.

If you are using an APx1701 and a TEDS-enabled measurement microphone, a **Calibrate from TEDS** button will be available on the mic cal dialog.



Microphone Calibration / Set dBSPL dialog (acoustic mode, with Calibrate from TEDS button).

Open the Calibrate from TEDS dialog to use TEDS data to set dBSPL1 and dBSPL2 references.



Note: TEDS data are only available when using the APx1701 with a TEDS-enabled measurement microphone.

With a TEDS-enabled measurement microphone connected to an APx1701 as part of the analyzer system, you can use the TEDS data to enter calibration information into the APx Mic Cal / dBSPL references.

Select the APx1701 unbalanced microphone input which is connected to the TEDS microphone.

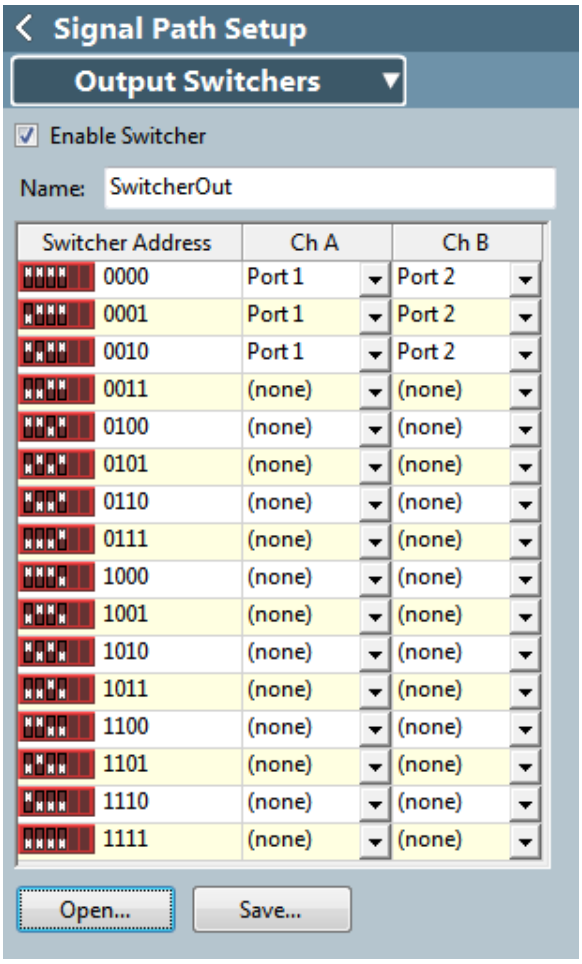
Choices are:

- **Mic1 Unbalanced**
- **Mic2 Unbalanced**

Click **Read TEDS**. The Serial #, Sensitivity and Frequency fields will be populated with the microphone's TEDS data.

When in input Acoustic mode, you can assign the reference value to any available analyzer input channel. Choose the channel from the **Apply To** menu. Click **Set Sensitivity**.

Signal Path Setup: Output and Input Switchers

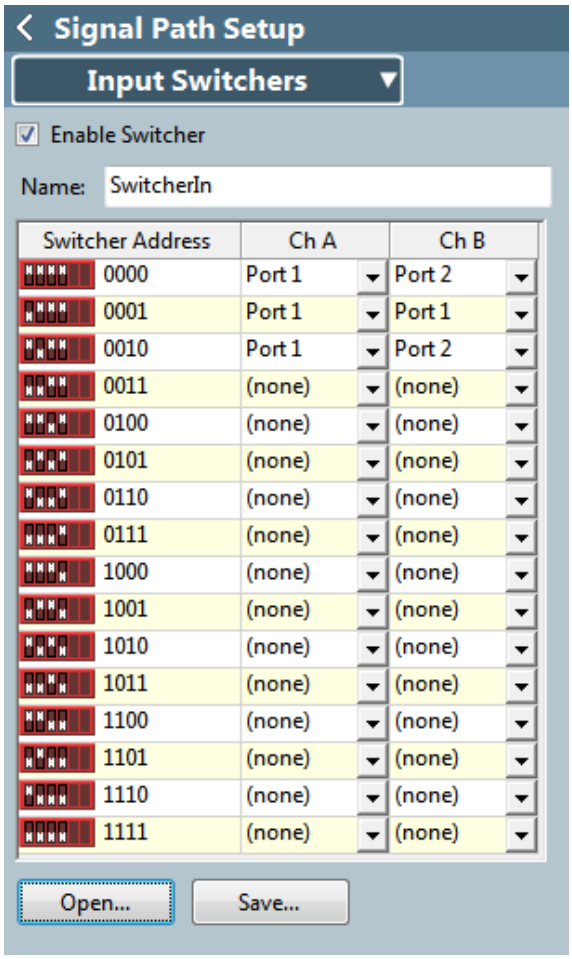


Audio Precision manufactures optional switchers that can be used to connect or disconnect many signal channels (up to 192 input and 192 output channels) to or from the instrument, under software control. See general switcher operating information beginning with the Switchers topic.

To configure a switcher, go to Signal Path Setup and choose Output Switchers or Input Switchers.

Name

You can apply a custom name to a switcher setup in the Name field. This name becomes the default name in the Save dialog. When opening a file, the name of the file opened file appears in the Name field.



Switcher Address

The switcher address is the binary address that enables communication from the APx500 software to a particular switcher hardware unit. On the rear panel of each switcher is a DIP switch with 6 switches; the four switches to the left set the binary address for that unit. To address a switcher unit from the APx500 software, the binary address must match on the switcher hardware and the switcher address controls in the software.

Note that the DIP switch orientation on the hardware is such that switch order is reversed from the digit order in ordinary binary notation, and that 0=UP and 1=DOWN: binary number 0111 corresponds to switch settings of DOWN-DOWN-DOWN-UP. For clarity, both DIP switch position and binary number are shown in the control panel.

Switcher Channels A and B

Each switcher has 2 channels called Channel A and Channel B, that are used to connect to 2 APx outputs (for an output switcher) or inputs (for an input switcher). These are connected to the switcher ports by internal relays, under the APx500 software control.

Switcher Ports

Each switcher has 12 ports that are used to connect to DUT channels according to your test requirements. An input switcher's ports are designed to connect to a DUT's output channels, and an output switcher's ports are designed to connect to a DUT's input channels. For each APx500 signal path, you can make a port selection for Channel A and Channel B for each addressed switcher.

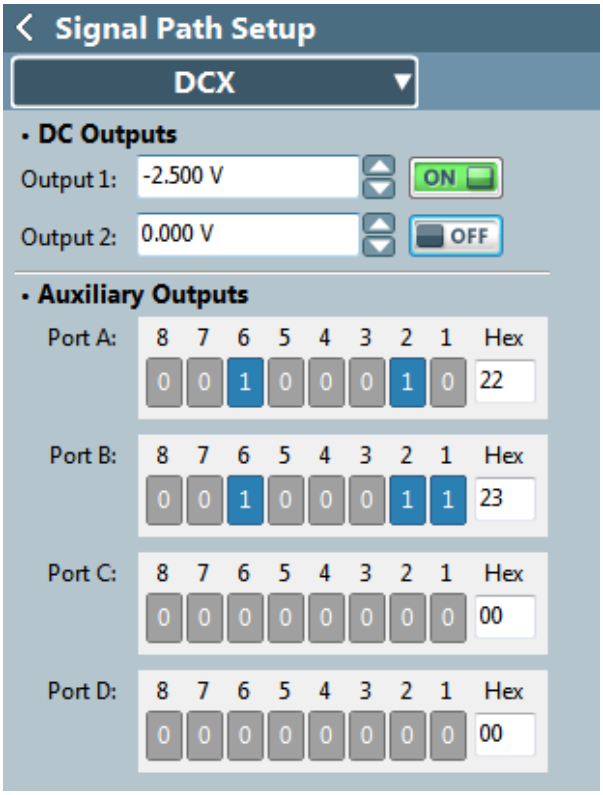
All But ChA

The Output Switcher Channel B port selection list includes All But ChA as a choice. This selection connects switcher Channel B to all ports except the current port connected to Channel A.

Audio Precision manufactures optional switchers that can be used to connect or disconnect many signal channels (up to 192 input and 192 output channels) to the instrument inputs or outputs, under software control.

See Chapter 103 for information about using switchers with APx500 and the APx family of instruments.

Signal Path Setup: DCX



The DCX-127 Multifunction Module is an Audio Precision accessory that provides interface and control features not otherwise available in Audio Precision

analyzers. Download the *DCX-127 Multifunction Module User's Guide* from the Audio Precision Web site at ap.com for complete operational information for the DCX-127.

- In Bench Mode, you can configure the DCX DMM (digital multimeter), the DCX DC Outputs and Auxiliary Outputs from the Signal Path Setup > DCX panel.
- In Sequence Mode, you can only configure the DCX DC Outputs and Auxiliary Outputs from the Signal Path Setup > DCX panel. However, DMM functions are available in DC Level (DCX) and Resistance (DCX) Sequence Mode measurements.

DMM (Bench Mode only)

The DCX has a digital multimeter (DMM) that can be configured as DC voltmeter or an ohmmeter.

Mode

The Mode control allows you to select the DCX meter mode.

- **Off**
  The DCX meter is not enabled.
- **Resistance**
  The DCX meter is configured to measure resistance.
- **DC Level**
  The DCX meter is configured to measure DC Level.

Range

The Range control allows you to select one of 5 fixed maximum meter input ranges, or Auto. The ranges are:

Resistance	*DC Level*
2 MOhm	500 V
200 kOhm	200 V
20 kOhm	20 V
2 kOhm	2 V
200 Ohm	200 mV

Rate

The Rate control allows you to select one of two meter reading rates.

- **6/sec**
  Approximately 6 meter readings are taken per second.
- **25/sec**
  Approximately 25 meter readings are taken per second.

At 6/sec, DCX meter resolution is a full 4 1/2 digits. At 25/sec, the resolution is reduced. The number of digits displayed is the same but the least significant digit is always 5 or 0.

DC Outputs (Sequence Mode and Bench Mode)

The DCX has two DC outputs, which you can configure here.

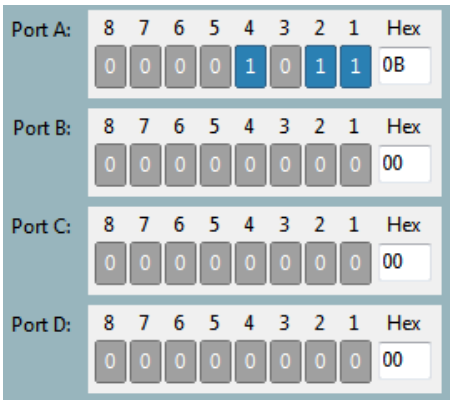
- **Output 1**
  Set the DC Output 1 voltage and ON/OFF status here.
- **Output 2**
  Set the DC Output 2 voltage and ON/OFF status here.

The voltage range for both outputs is ±10.50 V. These outputs have relatively low current capability (20 mA) and should be used as control or signal voltages, not as supply voltages.

Auxiliary Outputs (Sequence Mode and Bench Mode)

The DCX has four auxiliary (GPIO) output ports, similar in function, capability and user interface to the APx Aux Control Out port.

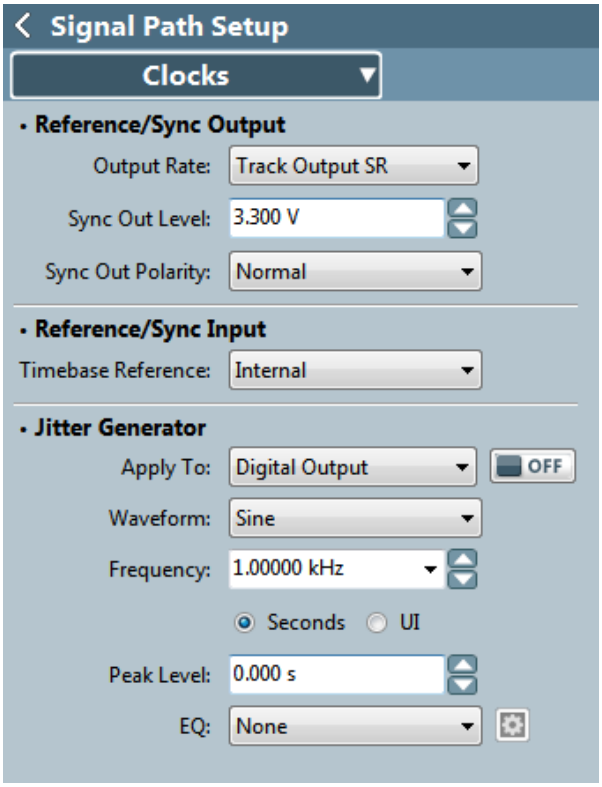
Configure the output ports here.



Ports A, B, and C are general-purpose 8-bit digital ports, available on 9-pin D-Sub connectors on the DCX-127 rear panel. Pin 9 is the common (ground) connection; pins 1–8 are numbered to correspond with bits 1–8.

Port D is a general-purpose 8-bit digital port, available on a 15-pin D-Sub connector labeled J141 on the DCX-127 rear panel. Pins 1, 14, and 15 are the common (ground) connection; pins 10 and 11 are +5 V dc; pin 12 is +15 V dc; pin 13 is –15 V dc, and pins 2–9 correspond with bits 1–8 respectively.

Signal Path Setup: Clocks



The features and controls described here require the Advanced Master Clock, a standard feature on the APx555 and an option on modular APx analyzers. When the Advanced Master Clock is present, these inputs and outputs are available on connectors on the APx rear panel.

Overview

Signal Path Setup: Clocks is the control panel for selecting the reference for the APx master clock time base, for making clock reference input and output settings, and for applying an internally generated jitter signal to APx digital outputs.

Reference/Sync Output

Output Rate:

- **Track Output SR**
  When **Track Output SR** is set, the Reference and Sync outputs track the current APx output sample rate.
  When the APx output is digital, the Reference and Sync output rates are set to the current output sample rate, as set in Signal Path Setup: Output Configuration.
  When the APx generator is playing a waveform, the Reference and Sync output rates are set to the sample rate of the waveform.
  When the APx output is the DAC generator, the Reference and Sync output rates are fixed at 192 kHz, unless an arbitrary Generator Waveform of a different rate is loaded. When the APx output is the High Performance Sine Generator, the Reference and Sync outputs are muted.
- **Custom**
  When **Custom** is set, the **Custom Rate** field is available. The rate set in this field will be the **DARS Reference** and **Sync Output Rates**, which track together from 8 kHz to 216 kHz. If you enter a rate between 216 kHz and 50 MHz, the **Sync Output** will use that rate, but the **DARS Reference Output** will mute.

The Reference/Sync Output is unavailable in certain configurations when Output Rate is set to Custom, due to limited resources. See Clock Availability on page 69.

- **Custom Rate**
  Enter the **Custom Rate** here. The acceptable range is 8 kHz to 50 MHz. The **DARS Reference Output** range is restricted to a maximum frequency of 216 kHz.

Sync Out Level

Set the Sync Out level here. Minimum is 0.8 V; maximum is 3.6 V; default is 3.3 V.

Sync Out Polarity

Select **Normal** or **Inverted**.

Reference/Sync Input

Timebase Reference:

- **Internal**
  The APx timebase is referenced to an internal crystal oscillator. Signals at the Reference In and Sync In connectors are ignored.
- **Ext. AES11 DARS (XLR)**
  The APx timebase is referenced to the digital signal at the **AES11 DARS In** XLR jack on the rear panel.
- **Ext. Sync (BNC)**
  The APx timebase is referenced to the square wave sync signal at the **Sync In** BNC jack on the rear panel.

Ext. Reference Rate:

When using an external Timebase Reference, you must specify the nominal rate of the external signal.

For the DARS input, the range is 8 kHz to 216 kHz. For the Sync input, the range is 4 kHz to 49.1520 MHz.

DARS Termination:

The DARS signal is a special version of an AES3 (IEC60958-4) signal. AES3 signals should be terminated in an impedance of 110 ohms, or bridged by a high impedance.

- **High Impedance**
  This setting is useful when bridging a digital con-

nection (as with a “Y” cable). A balanced digital connection should be properly terminated in 110 ohms at the end of its run.

- **110 ohms**
  With this set, the DARS Reference input is terminated in a resistance of 110 ohms. This is the correct setting for terminating a balanced digital audio connection for AES3 / AES-EBU / IEC60958-4 professional formats.

Jitter Generator

Read more about jitter on page 71.

Jitter is the variation in time of an event, such as a clock signal, from nominal. In APx analyzers, jitter generation is only available for I/O modules that support jitter, and only when the Advanced Master Clock is installed. An APx digital or sync output signal that is impaired by jitter can be used to evaluated jitter tolerance in downstream or synchronized external devices.

A “jitter clock” is provided as a component of the Advanced Master Clock system. This clock can be modulated in time (jittered), and applied as a reference to either a digital output or to the signals provided to the Reference Out and Sync Out connections. The modulating signal can be one of three waveforms, across a range of frequencies and levels.

Apply To

Select the signal to be impaired by jitter.

- **Digital Output**
  This selection will apply jitter to the output of a digital module that supports output jitter, such as the Advanced Digital I/O module.
- **Ref/Sync**
  This selection will apply jitter to the Reference Out and Sync Out signals available on the APx rear panel.

On/Off

Turn the jitter **On** or **Off**.

Waveform

Select a waveform for the jitter generator.

- Sine
- Square
- Noise

Frequency

For the sine and square waveforms, select a frequency. The range is 2 Hz to 200 kHz.

Peak Level

Set the peak level of the jitter here. Since jitter is a time modulation, the level units are seconds or Unit Intervals (UI).

For the sine signal, the maximum peak level varies with frequency, providing 1.591 µs (9.775 UI at 48 kHz) peak level up to 20 kHz. Above 20 kHz, the maximum level derates at 20 dB/decade, falling to 159.1 ns by 200 kHz. When the jittered interface is PDM, sine peak level is limited to 3.5 UI.

For the square wave or noise signals, the maximum peak level is 40 ns.

Units

Since jitter is a time modulation, the level units are seconds.

- **(sec)**
  Choose this to show jitter in seconds.
- **(UI)**
  Choose this to show jitter in Unit Intervals (UI), a unit that scales to the sample rate.

UI definition for Reference/Sync output

As discussed in Unit Intervals (UI) on page 688, the UI is defined differently for AES/SPDIF signals and for clock signals. When **Apply To:** is set to **Reference/ Sync**, and **UI** is selected for **Units**, a **Define UI** control is available for you to choose the UI definition.

Choose

- **Reference Output** to define the UI according to AES31,

or

- **Sync Output** to define the UI by the telecommunications industry practice.

A jitter peak level setting field is available that uses the selected definition, and a reading field displays the jitter peak level in the alternative definition.

EQ

When the jitter waveform is sine, an EQ curve can be applied to the jitter generator. You can use this function to apply an EQ curve that reflects the AES3 jitter tolerance template for the current sample rate, or for any other EQ curve. The EQ curve affects the jitter generator level for both sweep and static measurements.

- None
- **Create New**
  You can create or edit the generator EQ curve in Editing the Generator EQ table.
- **Browse for File**
  Open an existing jitter generator EQ table (*.csv or *.xls file).

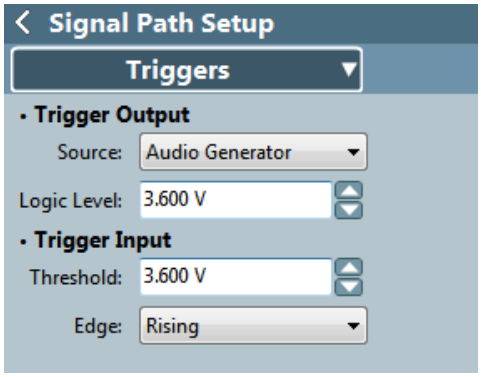
Clock Availability

There are rare cases where APx system resource limitations affect the availability of clock outputs. If the configuration requires too many different clock rates, clock outputs such as the Digital Serial Reference Clock, the PDM Input Bit Clock or Reference/Sync Out may be unavailable and muted.

These cases often arise in configurations where the measurement involved has strict requirements for input/output sample rate relationships, such as Continuous Sweep-based measurements (Continuous Sweep, Frequency Response, Acoustic Response, Loudspeaker Production Test) and the POLQA and PESQ measurements, or when Custom Reference/ Sync Out rates are specified.

For detailed information, please contact Audio Precision Technical Support at techsupport@ap.com.

Signal Path Setup: Triggers



Overview

External trigger inputs and outputs are available when the Advanced Master Clock (AMC) is installed, and are available on connectors on the APx rear panel.

Trigger Output

Trigger Output sends a logic-level pulse when a certain condition is satisfied in APx. You can choose one of two trigger sources.

Source

Choose **Audio Generator**, **Jitter Generator** or **Off**.

Triggering points by trigger source:

Logic Level

When a **Source** is selected, the **Logic Level** field is available. Set the trigger out level here. Minimum is 0.8 V; maximum is 3.6 V; default is 3.3 V.

Output triggering points by trigger source:

Trigger Source	*Trigger goes high at*	*Trigger goes low at*
High Performance Sine Generator	positive sine peak	negative sine peak
Burst sine	At start of high-level burst	At end of high-level burst
Jitter Generator (sine, square)	positive-going zero crossing	negative-going zero crossing
Jitter Generator (noise)	---	---
Sine (DAC)	positive-going zero crossing	negative-going zero crossing
Sine plus offset (DAC)	positive-going zero crossing of sine	negative-going zero crossing of sine
Sine split frequency (DAC)	positive-going zero crossing of freq A	negative-going zero crossing of freq A
Sine split phase (DAC)	positive-going zero crossing of freq A	negative-going zero crossing of freq A
Square	positive-going zero crossing	negative-going zero crossing
IMD (DAC)	positive-going zero crossing of freq 1	negative-going zero crossing of freq 1
DIM	positive-going zero crossing of square	negative-going zero crossing of square
Arbitrary/coded waveform	First sample	Second sample
Noise, all types	---	---
Bit test	First sample of sequence	Second sample of sequence
Constant value	---	---
Walking ones/zeros	At LSB/MSB transition	At next bit transition

Trigger Input

The Sequence Mode **Signal Acquisition** and **Signal Analyzer** measurements and the Bench Mode FFT measurement each have a **Trigger** choice called **External Input**. When this is selected, an external pulse at the rear-panel **Trigger In** connector satisfying the **Threshold** and **Edge** requirements (below) will start the measurement.

Threshold

Set voltage level threshold that will start the measurement. Minimum is 0.8 V; maximum is 3.6 V; default is 3.3 V.

Edge

Choose **Rising** edge or **Falling** edge for trigger input sensitivity.

More About Jitter

An excellent discussion of jitter in digital audio is found in the first chapter of *Measurement Techniques for Digital Audio* by the late Julian Dunn, published by Audio Precision. This book can be downloaded or purchased in hard copy from our Web site at ap.com.

Here is an excerpt:

> *Jitter is the variation in the time of an event*—such as a regular clock signal—*from nominal.*
> For example, the jitter on a regular clock signal is the difference between the actual pulse transition times of the real clock and the transition times that would have occurred had the clock been ideal, that is to say, perfectly regular.
> Against this nominal reference, the zero-crossing transitions of many of the pulses in a jittered data stream are seen to vary in time from the ideal clock timing. Expressed another way, jitter is phase modulation of the digital interface signal.
> The jitter component can be extracted from the clock or digital interface signal to be analyzed as a signal in its own right. Among the more useful ways of characterizing jitter is by examining its frequency spectrum and identifying the significant frequency components of the jitter itself.
>
> **Measuring Jitter**
>
> When very little jitter is present, the pulse transitions are moved back or forth by only small measures of time. When the jitter is increased, the transitions move across a larger range of times.
> Jitter amplitude, then, is a measure of time displacement and is expressed in units of time, either as fractions of a second or unit intervals. For those new to jitter measurement, this can lead to some disconcerting graph labels, with time on the vertical axis versus time on the horizontal axis, for example.
> Jitter frequency is the rate at which this phase-shifting is taking place. Like other noise or interference signals, the jitter modulation signal can be a pure and regular sine wave, a complex waveform or have a completely random character.
>
> **The Unit Interval**
>
> The unit interval (UI) is a measure of time that scales with the interface data rate, and is often a convenient term for interface jitter discussions. The UI is defined as the shortest nominal time interval in the coding scheme. For an AES3 signal at a 48 kHz frame rate, there are 32 bits per subframe and 64 bits per frame, giving a nominal 128 pulses per frame in the channel after bi-phase mark encoding is applied. So, in this case:
>
> 1 UI / (128x48000)=163 ns
>
> The UI is used for several of the jitter specifications in AES31 (the Audio Engineering Society's standard for interfacing two-channel linear digital audio), with the result that the specifications scale appropriately with the data and frame rate. As an example, the dimensions in UI for 96 kHz frame rates are exactly half the size, in seconds, as the dimensions in UI for frame rates of 48 kHz. This scaling matches the scaling of the capabilities and requirements of the receivers and transmitters on the interface.

Read more about time units and the Unit Interval on page 688.

Jitter Support in APx

Jitter generation

Normally, we strive to keep our internal master clocks as stable and accurate as possible. However, the Advanced Master Clock provides an additional clock that can be jittered. This clock can be applied to a jitter-enabled I/O module to impair the output signal to test the jitter tolerance of a device under test. The clock can be jittered with a noise, sine, or square waveform.

Jitter can be applied to a jitter-enabled digital audio output or clock output, or to the **Reference Out** and **Sync Out** signals.

Note that the jitter generator can be equalized in the same way as other APx generators.

Jitter measurement

When a jitter-enabled digital input is selected, the **Measure** selector allows you to chose whether the attached meters read the embedded audio signal or the jitter signal. The jitter signal can be displayed in seconds or in UI.

Note that when jitter is selected as the measurement source, the Signal Path filters provide different filtering options. High-pass provides a 700 Hz filter (the AES3 recommendation for intrinsic jitter measurement) and 50 Hz.

In APx, (as in earlier Audio Precision analyzers) jitter is always expressed in peak units. Three detector types are available for jitter measurements: a peak detector, an average detector, and an rms detector. Jitter results from any of these detectors are scaled in peak units for display. Consequently, a sinusoidal signal will show about the same (peak) value in the display no matter which detector is selected, and signals with higher crest factors will show differing display values.

Hardware support for jitter as of October, 2015

An APx analyzer fitted with the Advanced Master Clock (AMC) and a jitter-enabled input/output module can measure jitter on the incoming digital stream, and can impair the outgoing digital stream by imposing jitter upon it or upon the associated clock output.

The Advanced Master Clock (AMC)

Any jitter generation or measurement capability requires the presence of the Advanced Master Clock (AMC) module. The AMC is standard on all APx555 instruments, and can be ordered as an option for APx52x/58x instruments. The AMC can also be added to existing APx52x/58x instruments in a upgrade package.

The Advanced Digital Input/Output (ADIO) module

The ADIO module supports application of generated jitter to the output data stream. The ADIO also supports measurement of jitter on the input data stream.

The AMC is required for jitter.

The Digital Serial Input/Output (DSIO) module

DSIO modules require a modification to enable jitter. When an instrument is ordered with or upgraded to the AMC, an associated DSIO module will be modified to enable jitter.

For a jitter-enabled DSIO module, the DSIO transmitter or receiver supports application of generated jitter to the master clock, and to the bit and frame clocks, when these are derived from the APx master clock.

A jitter-enabled DSIO module also supports measurement of jitter on the receiver bit clock, when receiver **Bit and Frame Dir** are set to **In**.

The AMC is required for jitter.

The PDM module

PDM modules shipped after early October, 2015 are jitter enabled. These modules carry the model nomenclature “228” on the front panel, and operate under a firmware release of revision 4.00 or greater, indicated in the **Help > About > Product Components** field by **FW: V4.0** in the **PDM Module** section. Contact your Audio Precision representative to upgrade your PDM Option module.

A jitter-enabled PDM module supports application of generated jitter to the clock for PDM output, when **Bit Clk Dir** is set to **Out** in PDM **Output Settings**; or application of generated jitter to the clock for PDM input, when **Bit Clk Dir** is set to **Out** in PDM **Input Settings**. The PDM module also supports measurement of jitter on the clock for PDM input, when the **Bit Clk Dir** is set to In in PDM **Input Settings**.

The AMC is required for jitter.

More About Frequency Scaling in APx

In sampled systems (such as digital audio systems), a signal is sampled at a sampling rate for conversion to the digital domain. When such a digital signal is reconverted, any change in the sampling rate will bring a corresponding change in the frequency of the recovered signal (for very small changes, this effect appears as a change in phase).

Of course, nothing is perfect, and there is always a very small frequency error in sample rate and recovered signal, but this is typically well within the tolerance of both engineered systems and listeners.

In some cases, however, a shift in sampling rate is great enough, or the tolerance of a downstream system is small enough, that a frequency shift in the embedded signal must be avoided. This is the purpose of the **Scale Freq by:** feature in APx analyzer systems.

Normally, the frequency of the embedded signal is related to the output sample rate or the measured input sample rate. **Scale Freq by:** allows you to relate the embedded audio frequency to a fixed rate that you define. This can be the nominal rate (the assumed output or input sample rate, without error), or some other, arbitrary rate.

Application

The sample rate of an APx clock or digital output is quite stable; however, applied jitter or a testing configuration that may allow sample rate stability to be degraded can affect the phase and frequency of the embedded audio signal, even if that sample rate is derived from an APx clock.

In many cases, this is of little consequence, but testing that requires precise frequency accuracy (such as multitone or POLQA testing) may not tolerate much error. Also, APx digital filters frequencies are related to sample rate, and high jitter levels can cause poor filter tracking and instability. This is most apparent when making THD+N measurements. If the input signal has high levels of jitter, measurements of

THD+N levels can be in error, reading too high. Setting Scale Freq by: to Fixed and entering the nominal sample rate will stabilize the notch filter and provide correct THD+N readings.

Frequency Scaling for digital outputs

ADIO outputs (and DSIO and PDM outputs, when configured in a system that supports jitter generation) have a **Scale Freq. by:** control. Normally, this should be set to **Output SR** (the default). If high levels of generator jitter are applied, you may want to set **Scale Freq. by:** to **Fixed** and enter the nominal sample rate.

Frequency Scaling for digital inputs

DIO, ADIO, DSIO, PDM HDMI ARC inputs have a **Scale Freq. by:** control. Normally, this should be set to **Input SR** (the default). If high levels of input jitter are present, you may want to set **Scale Freq. by:** to **Fixed** and enter the nominal input sample rate.

If you are making a sample rate sweep (available in Bench Mode) while measuring a digital input, you should set the input **Scale Freq. by:** to **Output SR** to track the swept sample rate.

Note: clock transport in APx supported digital formats.
The AES3/SPDIF/optical/IEC60958 digital interfaces supported in the APx515 and in the DIO (Digital Input/Output) and ADIO (Advanced Digital Input/Output) modules use one transport stream, in which audio, metadata and sample rate clock are embedded.
The DSIO (Digital Serial Input/Output) and PDM modules use two or more electrical connections, with a clock signal on one line and audio data on one or more separate lines.

Verify Connections
(Sequence Mode)

Verify Connections

It's easy to accidentally cross cables when preparing a test setup. Verify Connections allows you to check your connections by sending and receiving tones on the channels you are using.

The **Verify Connections** panel is at the bottom of the **Signal Path Setup** column, providing diagnosis features for signal interconnection and format in the associated displays at the right side of the Workspace. The **Verify Connections** panel can be hidden.

Operation

If you have not yet set up your test, first go to Signal Path Setup to select and configure inputs and outputs, references and other settings. See Chapter 7 for information about Signal Path Setup.

Generator Off/On

With all your connections made, click Generator On. APx500 will apply the waveform to the selected **Test Channel** (by default, all channels are selected).

Waveform

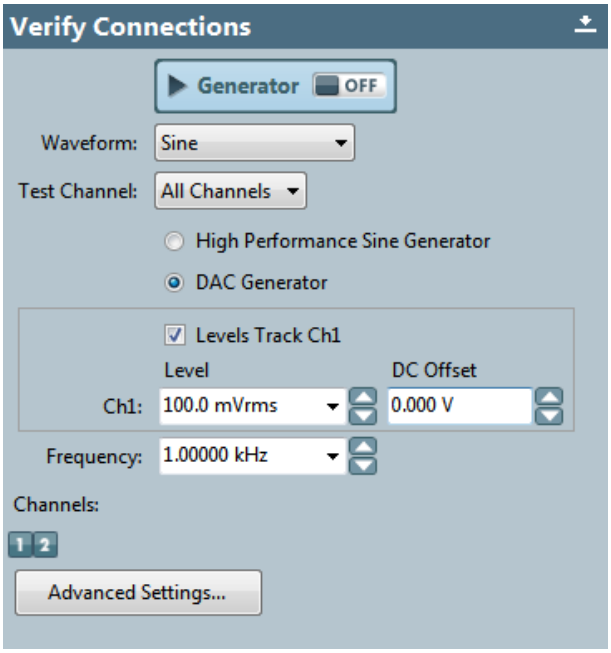
See Chapter 6 for detailed information about the using the APx Generator and making Waveform, Level and Frequency settings, or for information about using External Source.

Test Channel

From the dropdown menu, use **All Channels** or select an individual channel to check for cross-connection.

High Performance Sine Generator / DAC Generator

For an APx555, you have the option of using the High Performance Sine Generator (very low distortion sine waves and wide bandwidth) or the DAC generator, which provides special sine waveforms. See High Performance Sine Generator starting on page 51 for a comparison of features.



Channels

Assign the generator to output channels. By default, all active output channels are selected.

Verifying your Connections

With all your connections made, click **Generator On**. APx500 will apply the waveform to the selected channel (by default, **All Channels** are selected).

Use the Monitors

The Monitors are always available, giving you a simultaneous view of Scope, Spectrum, Meter or Metadata displays of the input signal(s).

Check RMS Level

View the RMS Level results. From the dropdown menu, use **All Channels** or select an individual channel to check for cross-connection.

Check Gain

View the Gain result to evaluate the DUT gain.

Check THD+N Ratio

Click the THD+N Ratio result to view the signal distortion.

Bits and Error Rate: Diagnostics for Digital Signal Paths

For signal paths with both a digital output and input, you can apply special waveforms and measurements as diagnostic tests for digital signal paths. These tools can be helpful in diagnosing digital connection problems such as "stuck bits." When the Output Configuration is digital, the Verify Connections Generator will show:

- Bit test
- Walking Zeros
- Walking Ones

and

- Constant Value

as generator waveform choices. (You could also use a properly formed digital file using these signals as a generator waveform or external source).

Read more about these special waveforms beginning on page 50.

Choose the **Bits** or **Error Rate** results.

Bits Result (digital only)

This view shows a real-time display of the audio bitstream. The bits are numbered from the MSB (most significant bit, bit 24) to the LSB (least significant bit, bit 1). It can be used with any waveform, but is most useful with these special waveforms:

- Walking Zeros
- Walking Ones
- Constant Value

Active Bits

When Active Bits is set, any bit that changes state during the diagnostic period (about 1/4 second) is shown as highlighted with color. Inactive bits are shown as white.

Data Bits

When Data Bits is set, the logical state (binary 1 or binary 0) of each bit in the audio word is shown. This is updated at the end of each diagnostic period (about 1/4 second). Binary 1 is shown in highlighted color; binary 0 is shown as white.

Note: The audio word may be any length between 32 bits and 8 bits, depending upon the Bit Depth setting in Signal Path Setup.

Error Rate Result (digital only)

This view recognizes Audio Precision bit-accurate waveforms, including

- Bit Test
  (This is usually the best choice for digital error rate measurements.)
- Walking Zeros
- Walking Ones
- Constant Value

When one of these bit-accurate waveforms is received with every sample at precisely the same value as was generated, zero errors are reported and an error rate of 0.000% is displayed.

When one of these waveforms is received with sample values different from the values in the original waveform, each differing sample is reported as one error, and the rate is calculated.

Read more about bit-accurate measurements beginning on page 277.

SPS: APx515 analog I/O

Introduction

This chapter discusses the Signal Path Setup analog I/O configurations for the APx515 analyzer. For other APx515 I/O configurations, go to

- **DIO** (digital I/O) configurations, Chapter 13.
- **ASIO** configurations, Chapter 21. (This is a software option for the APx515. See Software Options on page 189.)
- **External Source** configuration (open-loop, using no generator output), Chapter 6.
- **File Input** configurations, Chapter 22.

Common Signal Path Setup I/O Settings

- **Acoustic** (output) mode and output **EQ** are common to all analyzer analog output configurations,
- **Acoustic** (input) mode is common to all analyzer analog input configurations.
- **Loopback**, **Channel Labels**, **Filters** and **Device Under Test: Delay** settings are common to all analyzer input configurations.

Each of these common features is discussed in the **Input/Output** topics in Chapter 7, Signal Path Setup.

Bench Mode

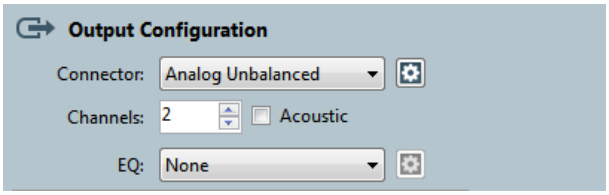
Bench Mode is an alternative operating interface for APx500, and is available as a software option for the APx515. See Software Options on page 189.

Output Configuration

The **Output Configuration** panel allows you to select the number of output channels operating for your test, and to choose the output format and connectors you will be using.

The **Settings** button opens an output **Settings** dialog for each configuration, offering more detailed control.

Output: Analog Unbalanced



This selects the unbalanced analog outputs available on BNC connectors on the instrument front panel.

Channels

The Channels setting allows you to set the number of output channels (1–2) to be tested for this signal path.

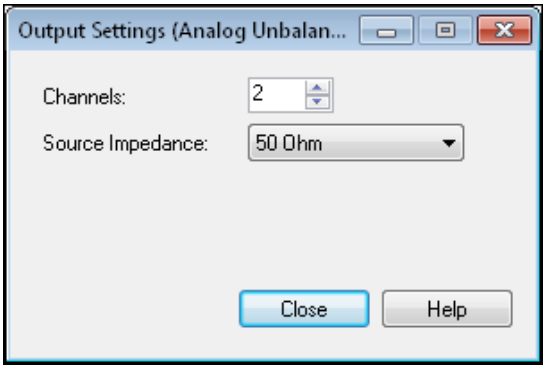
Acoustic

Read more about output **Acoustic** mode in the **Input/Output** topics in Chapter 7, Signal Path Setup.

EQ

Read more about output **EQ** in the **Input/Output** topics in Chapter 7, Signal Path Setup.

Settings for Analog Unbalanced Outputs



Channels

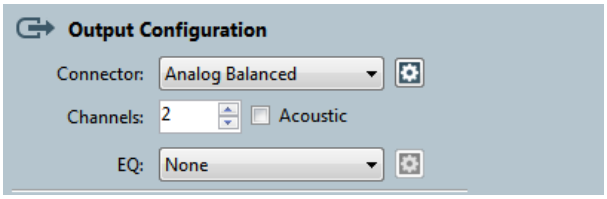
The **Channels** setting appears in both Signal Path Setup and here in the Settings dialog.

Source Impedance

Select the output **Source Impedance** here. For unbalanced outputs, the choices are

- 50 Ω (default)
- 600 Ω

Output: Analog Balanced



This selection enables the balanced analog outputs available on XLR3 male connectors.

Channels

The Channels setting allows you to set the number of output channels (1–2) to be tested for this signal path.

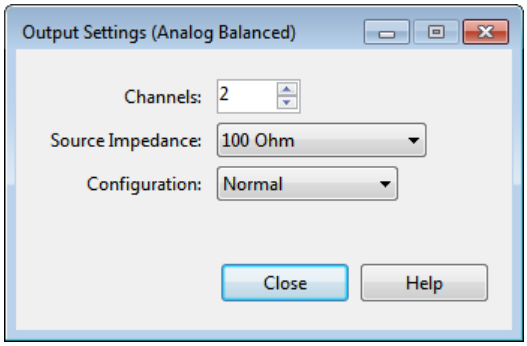
Acoustic

Read more about output **Acoustic** mode in the **Input/Output** topics in Chapter 7, Signal Path Setup.

EQ

Read more about output **EQ** in the **Input/Output** topics in Chapter 7, Signal Path Setup.

Settings for Analog Balanced Outputs



Channels

The Channels setting appears in both Signal Path Setup and here in the Settings dialog.

Source Impedance

Select the output source impedance here. For balanced outputs, the choices are

- 100 Ω
- 600 Ω

Configuration

The Configuration control sets the **Normal Mode / Common Mode Test (CMTST)** configuration for the balanced analog outputs, for all measurements.

Configuration choices are:

- **Normal**
  The default, differentially balanced output configuration. Signal plus (+) is connected to pin 2; signal minus (–) is connected to pin 3. This is the recommended setting for all measurements.
- **CMTST**
  A fixed common-mode output configuration. Signal plus (+) is connected to both pin 2 and pin 3, signal minus (–) is connected to pin 1 (ground). This configuration is recommended only for common-mode testing.

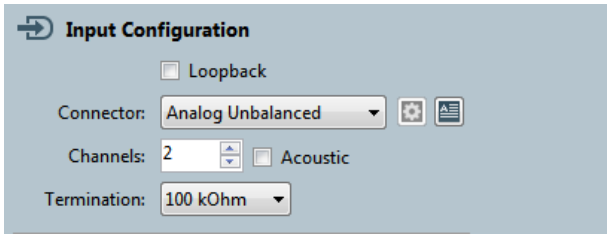
Setting a common mode configuration for all measurements can produce unexpected results. For Sequence Mode common mode rejection ratio tests, use the CMRR measurement instead.

Input Configuration

The Input Configuration panel allows you to select the number of input channels operating for your test, and to choose the input format and connectors you will be using.

The **Settings** button is unavailable for analog inputs.

Input: Analog Unbalanced



Connector

Analog Unbalanced sets both channels to the unbalanced analog inputs, available on BNC connectors on the instrument front panel.

Channels

The Channels setting appears in both Signal Path Setup and here in the Settings dialog.

Acoustic

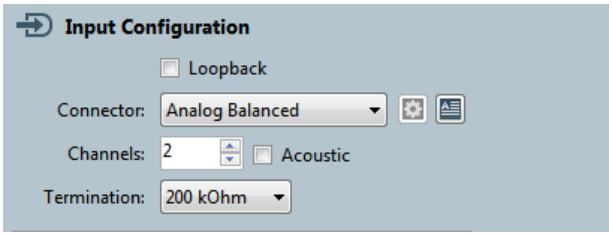
Read more about input **Acoustic** mode in the **Input/Output** topics in Chapter 7, Signal Path Setup.

Termination

This control sets the input termination for both unbalanced input channels. Choices are:

- **100 kΩ** or
- **600 Ω**

Input: Analog Balanced



Connector

Analog Balanced sets both channels to the balanced analog inputs, available on XLR female connectors on the instrument front panel.

Channels

The Channels setting appears in both Signal Path Setup and here in the Settings dialog.

Acoustic

Read more about input **Acoustic** mode in the **Input/Output** topics in Chapter 7, Signal Path Setup.

Termination

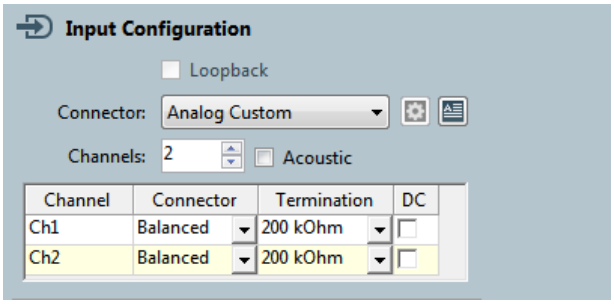
This control sets the input termination for both balanced input channels. Choices are:

- 200 kΩ or
- 600 Ω

Input: Analog Custom

Connector

Analog Custom allows you to set **Connector, Termination** and **Coupling** settings independently for each input channel. A channel set to **Unbalanced** is connected to a front panel BNC connector; a channel set to **Balanced** is connected to a front panel XLR female connector.



Channels

The Channels setting appears in both Signal Path Setup and here in the Settings dialog.

Acoustic

Read more about input **Acoustic** mode in the **Input/Output** topics in Chapter 7, Signal Path Setup.

The Custom configuration grid

Channel

Each channel has a row for custom selections.

Connector

Select the Connector (Balanced or Unbalanced) for a particular channel here.

Termination

Select the Termination for a particular channel here. The Termination options will depend upon the Connector selection.

DC

Select the AC or DC coupling for a particular channel here. Note that when Input Configuration is set to **Custom**, the **High-pass Filter** selections are only AC-coupled. Selecting **DC** in the **Custom** grid overrides the **High-pass Filter** selection for the channel, and sets it to DC-coupled. See page 55 and also Chapter 97 for a detailed discussion of Signal Path Setup Filters.

Filters and Device Under Test: Delay

Filters

See page 55 and also Chapter 97 for a detailed discussion of Signal Path Setup **Filters**. Note that analog system bandwidth (DAC sample rate) is set using the **Low-Pass Filter** menu, and that AC/DC coupling is set using the **High-pass Filter** menu.

Device Under Test

Delay

See the Input/Output topics in Chapter 7, Signal Path Setup, for a detailed discussions of **DUT Delay**.

SPS: APx52x analog I/O and APx582 analog out

Introduction

This chapter discusses the Signal Path Setup analog I/O configurations for the APx520, 521, 525 and 526 analyzers. Additionally, the Output Configuration discussed here applies to the analog outputs of the APx582. For other I/O configurations for these analyzers, go to

- **DIO** (digital I/O) configurations, Chapter 13.
- **ADIO** (advanced digital I/O) configurations, Chapter 14. This an option that can replace the standard DIO.
- **DSIO** (digital serial I/O) configurations, Chapter 15.
- **HDMI+ARC I/O** configuration, Chapter 16.
- **Bluetooth** configurations, Chapter 17.
- **PDM** configuration, Chapter 19.
- **ASIO** configurations, Chapter 21.
- **External Source** configuration, Chapter 6.
- **File Input** configurations, Chapter 22.

Common Signal Path Setup I/O Settings

- **Acoustic** (output) mode and output **EQ** are common to all analyzer analog output configurations,
- **Acoustic** (input) mode is common to all analyzer analog input configurations.
- **Loopback**, **Channel Labels**, **Filters** and **Device Under Test: Delay** settings are common to all analyzer input configurations.

Each of these common features is discussed in the **Input/Output** topics in Chapter 7, Signal Path Setup.

Output Configuration

The **Output Configuration** panel allows you to select the number of output channels operating for your test, and to choose the output format and connectors you will be using.

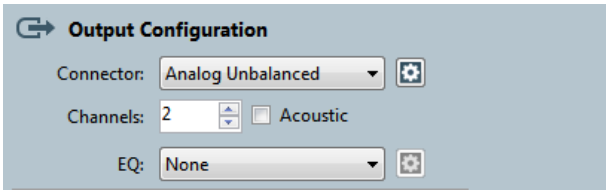
The **Settings** button opens an output **Settings** dialog for each configuration, offering more detailed control.

AG52 Analog Generator option

The AG52 Analog Generator option is available for APx525 family instruments. AG52 provides square wave and DIM waveforms and improved level and noise performance, but has no effect on the available analog output configuration selections.

The AG52 is a standard component for APx582 analyzers.

Output: Analog Unbalanced



This selects the unbalanced analog outputs available on BNC connectors on the instrument front panel.

Channels

The Channels setting allows you to set the number of output channels (1–2) to be tested for this signal path.

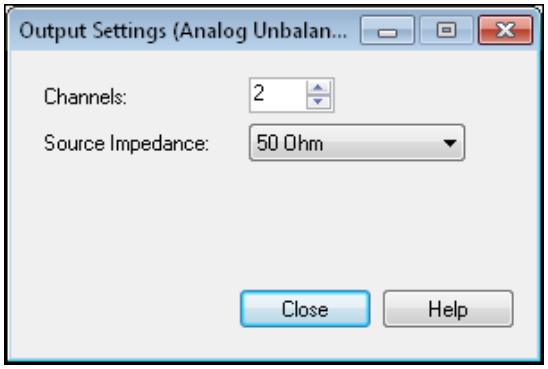
Acoustic

Read more about output **Acoustic** mode in the **Input/Output** topics in Chapter 7, Signal Path Setup.

EQ

Read more about output **EQ** in the **Input/Output** topics in Chapter 7, Signal Path Setup.

Settings for Analog Unbalanced Outputs



Channels

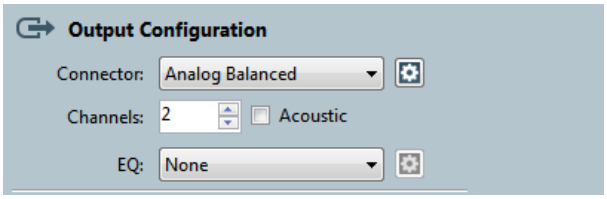
The **Channels** setting appears in both Signal Path Setup and here in the Settings dialog.

Source Impedance

Select the output **Source Impedance** here. For unbalanced outputs, the choices are

- 20 Ω (default)
- 50 Ω
- 75 Ω
- 100 Ω
- 600 Ω

Output: Analog Balanced



This selection enables the balanced analog outputs available on XLR3 male connectors and double banana connectors. The XLRs and the double bananas are wired in parallel.

Channels

The Channels setting allows you to set the number of output channels (1–2) to be tested for this signal path.

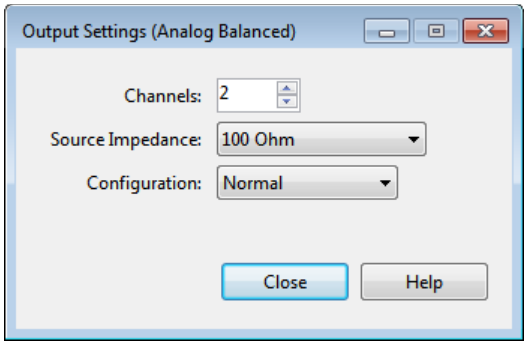
Acoustic

Read more about output **Acoustic** mode in the **Input/Output** topics in Chapter 7, Signal Path Setup.

EQ

Read more about output **EQ** in the **Input/Output** topics in Chapter 7, Signal Path Setup.

Settings for Analog Balanced Outputs



Channels

The Channels setting appears in both Signal Path Setup and here in the Settings dialog.

Source Impedance

Select the output source impedance here. For balanced outputs, the choices are

- 40 Ω
- 100 Ω
- 150 Ω
- 200 Ω
- 600 Ω

Configuration

The Configuration control sets the **Normal Mode / Common Mode Test (CMTST)** configuration for the balanced analog outputs, for all measurements.

Configuration choices are:

- **Normal**
  The default, differentially balanced output configuration. Signal plus (+) is connected to pin 2; signal minus (–) is connected to pin 3. This is the recommended setting for all measurements.
- **CMTST**
  A fixed common-mode output configuration. Signal plus (+) is connected to both pin 2 and pin 3, signal minus (–) is connected to pin 1 (ground). This configuration is recommended only for common-mode testing.\CMTST (IEC pin 2)
  A fixed common-mode output configuration, using the IEC unbalanced leg configuration. Source impedance is fixed at 40 Ω. Signal plus (+) is connect to pin 2 and pin 3, with an additional 10 Ω resistor inserted in the pin 2 leg. Signal minus (–) is connected to pin 1 (ground).
- **CMTST (IEC pin 2)**
  A fixed common-mode output configuration, using the IEC unbalanced leg configuration. Source impedance is fixed at 40 Ω. Signal plus (+) is connect to pin 2 and pin 3, with an additional 10 Ω resistor inserted in the pin 2 leg. Signal minus (–) is connected to pin 1 (ground).

- **CMTST (IEC pin 3)**
  A fixed common-mode output configuration, using the IEC unbalanced leg configuration. Source impedance is fixed at 40 Ω. Signal plus (+) is connect to pin 2 and pin 3, with an additional 10 Ω resistor inserted in the pin 3 leg. Signal minus (–) is connected to pin 1 (ground).

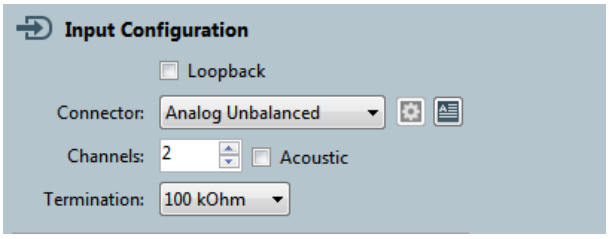
Setting a common mode configuration for all measurements can produce unexpected results. For Sequence Mode common mode rejection ratio tests, use the CMRR measurement instead.

Input Configuration

The Input Configuration panel allows you to select the number of input channels operating for your test, and to choose the input format and connectors you will be using.

The **Settings** button is unavailable for analog inputs.

Input: Analog Unbalanced



Connector

Analog Unbalanced sets both channels to the unbalanced analog inputs, available on BNC connectors on the instrument front panel.

Channels

The Channels setting appears in both Signal Path Setup and here in the Settings dialog.

Acoustic

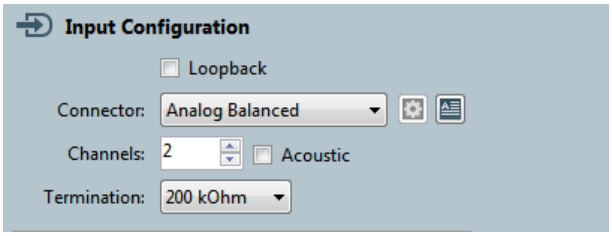
Read more about input **Acoustic** mode in the **Input/Output** topics in Chapter 7, Signal Path Setup.

Termination

This control sets the input termination for both unbalanced input channels. Choices are:

- 100 kΩ
- 300 Ω
- 600 Ω

Input: Analog Balanced



Connector

Analog Balanced sets both channels to the balanced analog inputs, available on XLR female connectors on the instrument front panel.

Channels

The Channels setting appears in both Signal Path Setup and here in the Settings dialog.

Acoustic

Read more about input **Acoustic** mode in the **Input/Output** topics in Chapter 7, Signal Path Setup.

Termination

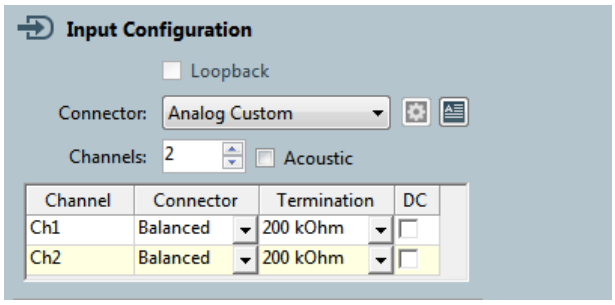
This control sets the input termination for both balanced input channels. Choices are:

- 200 kΩ
- 300 Ω
- 600 Ω

Input: Analog Custom

Connector

Analog Custom allows you to set **Connector, Termination** and **Coupling** settings independently for each input channel. A channel set to **Unbalanced** is connected to a front panel BNC connector; a channel set to **Balanced** is connected to a front panel XLR female connector.



Channels

The Channels setting appears in both Signal Path Setup and here in the Settings dialog.

Acoustic

Read more about input **Acoustic** mode in the **Input/Output** topics in Chapter 7, Signal Path Setup.

The Custom configuration grid

Channel

Each channel has a row for custom selections.

Connector

Select the **Connector** (Balanced or Unbalanced) for a particular channel here.

Termination

Select the **Termination** for a particular channel here. The Termination options will depend upon the Connector selection.

DC

Select the AC or DC coupling for a particular channel here. Note that when Input Configuration is set to **Custom**, the **High-pass Filter** selections are only AC-coupled. Selecting **DC** in the **Custom** grid overrides the **High-pass Filter** selection for the channel, and sets it to DC-coupled. See page 55 and also Chapter 97 for a detailed discussion of Signal Path Setup Filters.

Filters and Device Under Test: Delay

See page 55 and also Chapter 97 for a detailed discussion of Signal Path Setup **Filters**. Note that analog system bandwidth (DAC sample rate) is set using the **Low-Pass Filter** menu, and that AC/DC coupling is set using the **High-pass Filter** menu.

Device Under Test

Delay

See the Input/Output topics in Chapter 7, Signal Path Setup, for a detailed discussions of **DUT Delay**.

11

SPS: APx555 analog I/O

Introduction

For analog signal generation, the APx555 uses either DSP/DAC signal generation, or the High Performance Sine Generator. See Chapter 6 for more information about these generation techniques.

Likewise, the APx555 has dedicated analog notch filters and an additional pair of ADCs for use with the High Performance Sine Analyzer.

However, these high performance features do not affect the configuration controls and options discussed in this chapter, which are much the same as the I/O Configuration controls for other 2-channel APx instruments.

This chapter discusses the Signal Path Setup analog I/O configurations for the APx555 analyzer. For other I/O configurations for the APx555, go to

- **ADIO** (advanced digital I/O) configurations, Chapter 14.
- **DSIO** (digital serial I/O) configurations, Chapter 15.
- **HDMI+ARC I/O** configuration, Chapter 16.
- **Bluetooth** configurations, Chapter 17.
- **PDM** configuration, Chapter 19.
- **ASIO** configurations, Chapter 21.
- **External Source** configuration, Chapter 6.
- **File Input** configurations, Chapter 22.

Common Signal Path Setup I/O Settings

- **Acoustic** (output) mode and output **EQ** are common to all analyzer analog output configurations,
- **Acoustic** (input) mode is common to all analyzer analog input configurations.
- **Loopback**, **Channel Labels**, **Filters** and **Device Under Test: Delay** settings are common to all analyzer input configurations.

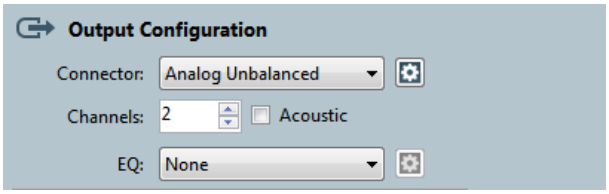
Each of these common features is discussed in the **Input/Output** topics in Chapter 7, Signal Path Setup.

Output Configuration

The **Output Configuration** panel allows you to select the number of output channels operating for your test, and to choose the output format and connectors you will be using.

The **Settings** button opens an output **Settings** dialog for each configuration, offering more detailed control.

Output: Analog Unbalanced



This selects the unbalanced analog outputs available on BNC connectors on the instrument front panel.

Channels

The Channels setting allows you to set the number of output channels (1–2) to be tested for this signal path.

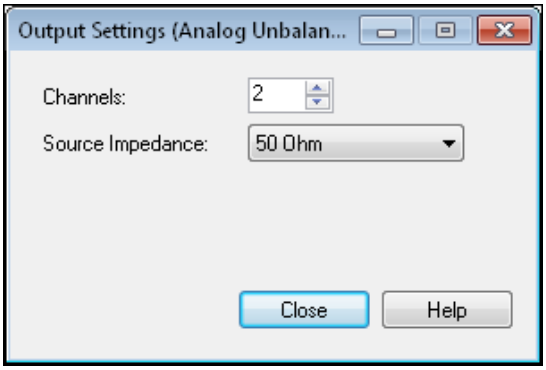
Acoustic

Read more about output **Acoustic** mode in the **Input/Output** topics in Chapter 7, Signal Path Setup.

EQ

Read more about output **EQ** in the **Input/Output** topics in Chapter 7, Signal Path Setup.

Settings for Analog Unbalanced Outputs



Channels

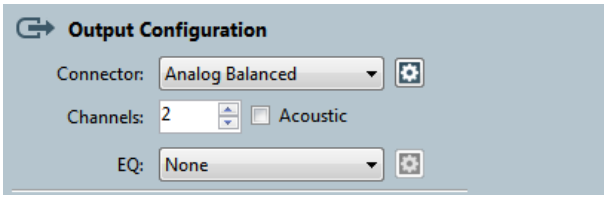
The **Channels** setting appears in both Signal Path Setup and here in the Settings dialog.

Source Impedance

Select the output **Source Impedance** here. For unbalanced outputs, the choices are

- 20 Ω (default)
- 50 Ω
- 75 Ω
- 100 Ω
- 600 Ω

Output: Analog Balanced



This selection enables the balanced analog outputs available on XLR3 male connectors and double banana connectors. The XLRs and the double bananas are wired in parallel.

Channels

The Channels setting allows you to set the number of output channels (1–2) to be tested for this signal path.

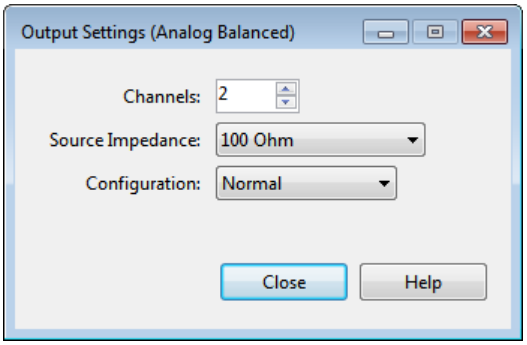
Acoustic

Read more about output **Acoustic** mode in the **Input/Output** topics in Chapter 7, Signal Path Setup.

EQ

Read more about output **EQ** in the **Input/Output** topics in Chapter 7, Signal Path Setup.

Settings for Analog Balanced Outputs



Channels

The Channels setting appears in both Signal Path Setup and here in the Settings dialog.

Source Impedance

Select the output source impedance here. For balanced outputs, the choices are

- 40 Ω
- 100 Ω
- 150 Ω
- 200 Ω
- 600 Ω

Configuration

The Configuration control sets the **Normal Mode / Common Mode Test (CMTST)** configuration for the balanced analog outputs, for all measurements.

Configuration choices are:

- **Normal**
  The default, differentially balanced output configuration. Signal plus (+) is connected to pin 2; signal minus (–) is connected to pin 3. This is the recommended setting for all measurements.
- **CMTST**
  A fixed common-mode output configuration. Signal plus (+) is connected to both pin 2 and pin 3, signal minus (–) is connected to pin 1 (ground). This configuration is recommended only for common-mode testing.\CMTST (IEC pin 2)
  A fixed common-mode output configuration, using the IEC unbalanced leg configuration. Source impedance is fixed at 40 Ω. Signal plus (+) is connect to pin 2 and pin 3, with an additional 10 Ω resistor inserted in the pin 2 leg. Signal minus (–) is connected to pin 1 (ground).
- **CMTST (IEC pin 2)**
  A fixed common-mode output configuration, using the IEC unbalanced leg configuration. Source impedance is fixed at 40 Ω. Signal plus (+) is connect to pin 2 and pin 3, with an additional 10 Ω resistor inserted in the pin 2 leg. Signal minus (–) is connected to pin 1 (ground).

- **CMTST (IEC pin 3)**
  A fixed common-mode output configuration, using the IEC unbalanced leg configuration. Source impedance is fixed at 40 Ω. Signal plus (+) is connect to pin 2 and pin 3, with an additional 10 Ω resistor inserted in the pin 3 leg. Signal minus (–) is connected to pin 1 (ground).

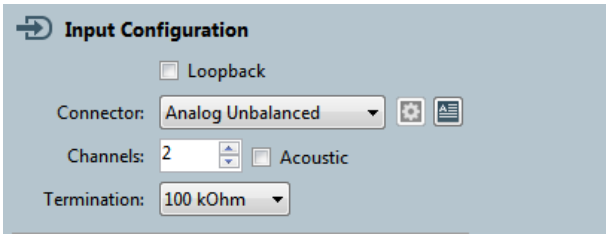
Setting a common mode configuration for all measurements can produce unexpected results. For Sequence Mode common mode rejection ratio tests, use the CMRR measurement instead.

Input Configuration

The Input Configuration panel allows you to select the number of input channels operating for your test, and to choose the input format and connectors you will be using.

The **Settings** button opens an input **Settings** dialog for each configuration, offering more detailed control.

Input: Analog Unbalanced



Connector

Analog Unbalanced sets both channels to the unbalanced analog inputs, available on BNC connectors on the instrument front panel.

Channels

The Channels setting appears in both Signal Path Setup and here in the Settings dialog.

Acoustic

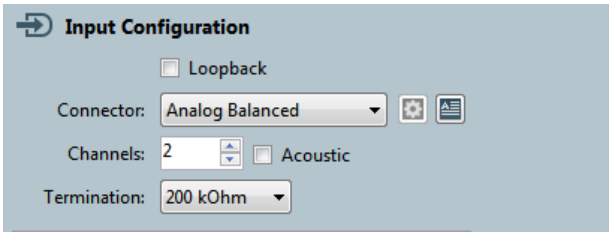
Read more about input **Acoustic** mode in the **Input/Output** topics in Chapter 7, Signal Path Setup.

Termination

This control sets the input termination for both unbalanced input channels. Choices are:

- 100 kΩ
- 300 Ω
- 600 Ω

Input: Analog Balanced



Connector

Analog Balanced sets both channels to the balanced analog inputs, available on XLR female connectors on the instrument front panel.

Channels

The Channels setting appears in both Signal Path Setup and here in the Settings dialog.

Acoustic

Read more about input **Acoustic** mode in the **Input/Output** topics in Chapter 7, Signal Path Setup.

Termination

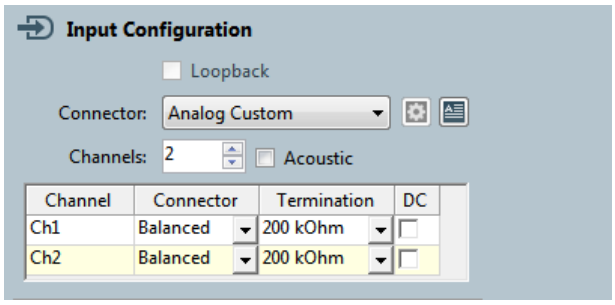
This control sets the input termination for both balanced input channels. Choices are:

- 200 kΩ
- 300 Ω
- 600 Ω

Input: Analog Custom

Connector

Analog Custom allows you to set **Connector, Termination** and **Coupling** settings independently for each input channel. A channel set to **Unbalanced** is connected to a front panel BNC connector; a channel set to **Balanced** is connected to a front panel XLR female connector.



Channels

The Channels setting appears in both Signal Path Setup and here in the Settings dialog.

Acoustic

Read more about input **Acoustic** mode in the **Input/Output** topics in Chapter 7, Signal Path Setup.

The Custom configuration grid

Channel

Each channel has a row for custom selections.

Connector

Select the Connector (Balanced or Unbalanced) for a particular channel here.

Termination

Select the Termination for a particular channel here. The Termination options will depend upon the Connector selection.

DC

Select the AC or DC coupling for a particular channel here. Note that when Input Configuration is set to **Custom**, the **High-pass Filter** selections are only AC-coupled. Selecting **DC** in the **Custom** grid overrides the **High-pass Filter** selection for the channel, and sets it to DC-coupled. See page 55 and also Chapter 97 for a detailed discussion of Signal Path Setup Filters.

Filters and Device Under Test: Delay

Filters

See page 55 and also Chapter 97 for a detailed discussion of Signal Path Setup **Filters**. Note that analog system bandwidth (DAC sample rate) is set using the **Low-Pass Filter** menu, and that AC/DC coupling is set using the **High-pass Filter** menu.

Device Under Test

Delay

See the Input/Output topics in Chapter 7, Signal Path Setup, for a detailed discussions of **DUT Delay**.

SPS: APx58x analog I/O and APx582 analog in

Introduction

This chapter discusses the Signal Path Setup analog I/O configurations for the APx585 and 586 analyzers. Additionally, the Input Configuration discussed here applies to the analog inputs of the APx582. For other APx585 or 586 I/O configurations, go to

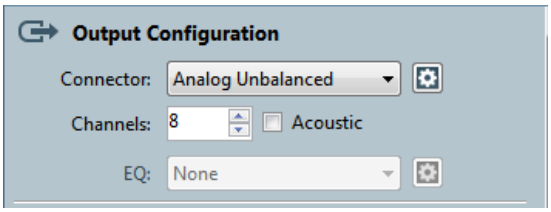
- DIO (digital I/O) configurations, Chapter 13.
- DSIO (digital serial I/O) configurations, Chapter 15.
- HDMI+ARC I/O configuration, Chapter 16.
- Bluetooth configurations, Chapter 17.
- PDM configuration, Chapter 19.
- External Source configuration, Chapter 6.
- File Input configurations, Chapter 22.

Output Configuration

The Output Configuration panel allows you to select the number of output channels operating for your test, and to choose the output format and connectors you will be using.

The **Settings** button opens an output Settings dialog for each configuration, offering more detailed control.

Output: Analog Unbalanced

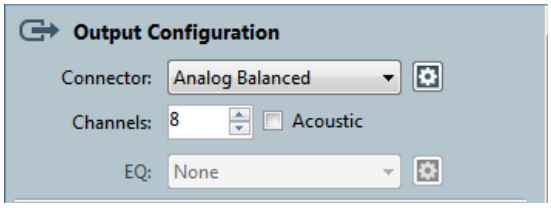


This selects the unbalanced analog outputs available on BNC connectors on the instrument front panel. The source impedance for the unbalanced analog outputs is 50 Ω.

Channels

The Channels setting allows you to set the number of output channels (1–8) to be tested for this signal path.

Output: Analog Balanced



This selection enables the balanced analog outputs available on a 25-pin female D-Sub connector (see page 11) on the instrument front panel. The source impedance for the balanced analog outputs is 100 Ω. When Analog Balanced is selected, the **Channels** selection field (duplicated in the Settings dialog) is available.

Channels

The Channels setting allows you to set the number of output channels (1–8) to be tested for this signal path.

Acoustic

Read more about output **Acoustic** mode in the **Input/Output** topics in Chapter 7, Signal Path Setup.

EQ

Read more about output **EQ** in the **Input/Output** topics in Chapter 7, Signal Path Setup.

For the APx585 and 586, **Output EQ** is unavailable when output Channels is set to more than 2.

Input Configuration

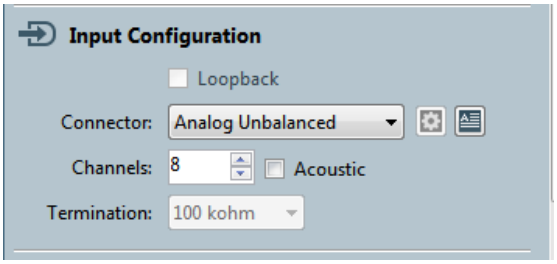
The Input Configuration panel allows you to select the number of input channels operating for your test, and to choose the input format and connectors you will be using. The **Settings...** button opens an input Settings dialog for each configuration, offering more detailed

control. You can also choose identifying names and colors for your test input channels.

Loopback

Loopback is not available for APx582, 585 or 586 analog inputs.

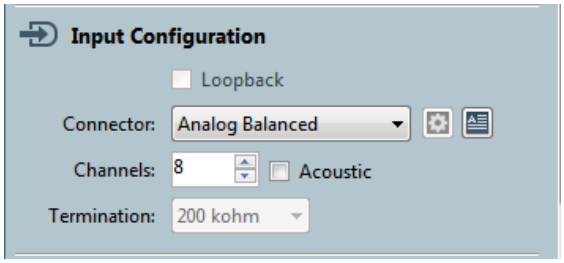
Input: Analog Unbalanced



Connector

Analog Unbalanced sets all channels to the unbalanced analog inputs, available on BNC connectors on the instrument front panel. The APx586 is fitted with 8 additional BNC connectors to provide 16 unbalanced inputs. The input termination for the unbalanced analog inputs is fixed at 100 kΩ.

Input: Analog Balanced



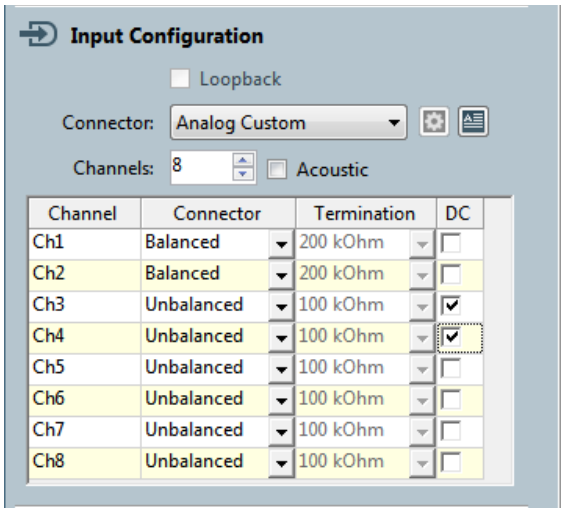
Connector

Analog Balanced sets all channels to the balanced analog inputs, available on 25-pin D-Sub connectors on the instrument front panel. The APx586 is fitted with a second D-Sub connector to provide 16 unbalanced inputs. The input termination for the balanced analog inputs is fixed at 200 kΩ.

Input: Analog Custom

Connector

Analog Custom allows you to choose **Unbalanced** or **Balanced** connections and **Coupling** settings independently for each input channel. Channels set to **Unbalanced** are connected to the front panel BNC connectors, with a fixed input termination of 100 kΩ; channels set to **Balanced** are connected to the front panel D-Sub connector, with a fixed input termination of 200 kΩ;



Channels

The **Channels** setting allows you to choose the number of input channels to be tested for this signal path. For the APx582 or 585, the maximum number of input channels is 8. For the APx586, the maximum number of input channels is 16.

Acoustic

Read more about input **Acoustic** mode in the **Input/Output** topics in Chapter 7, Signal Path Setup.

The Custom configuration grid

Channel

Each channel has a row for custom selections.

Connector

Select the Connector (Balanced or Unbalanced) for a particular channel here.

Termination

Select the Termination for a particular channel here. The Termination options will depend upon the Connector selection.

DC

Select the AC or DC coupling for a particular channel here. Note that when Input Configuration is set to **Custom**, the **High-pass Filter** selections are only AC-coupled. Selecting **DC** in the **Custom** grid overrides the **High-pass Filter** selection for the channel, and sets it to DC-coupled. See page 55 and also Chapter 97 for a detailed discussion of Signal Path Setup Filters.

Filters and Device Under Test: Delay

Filters

See page 55 and also Chapter 97 for a detailed discussion of Signal Path Setup **Filters**. Note that analog system bandwidth (DAC sample rate) is set using the **Low-Pass Filter** menu, and that AC/DC coupling is set using the **High-pass Filter** menu.

Device Under Test

Delay

See the Input/Output topics in Chapter 7, Signal Path Setup, for a detailed discussions of **DUT Delay**.

SPS: Digital I/O

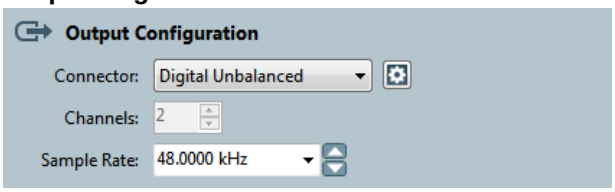
Introduction

Most APx Series audio analyzers have a digital input/output interface (DIO). The software control for the DIO is the same for all APx Series analyzers. For information on the other supported digital interfaces, see Chapter 15 (DSIO), Chapter 16 (HDMI), Chapter 17 (Bluetooth), and Chapter 19, (PDM).

An advanced digital input/output (ADIO) interface is also available, standard on an APx555 and an option for the APx525/585 family. See Chapter 14 for more information about the advanced digital input/output.

Output Configuration

Output: Digital Unbalanced



This selection enables the unbalanced digital electrical output, available on a BNC connector on the instrument front panel.

The **Settings** button opens an output **Settings** dialog for each configuration, offering more detailed control.

Channels

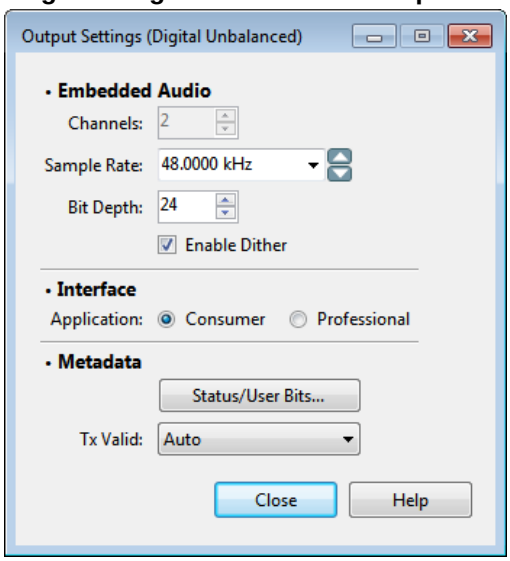
Audio channels are fixed at two.

Sample Rate

To set the output sample rate (Fs), click the up/down arrows to select a standard rate, or enter an arbitrary value in the **Sample Rate** field. Minimum setting is 8 kHz; maximum is 216 kHz. Default is 48 kHz. **Sample Rate** is also available in the Settings dialog.

The APx515 has a minimum output sample rate of 22 kHz.

Settings for Digital Unbalanced Output



Embedded Audio

The **Channels** and **Sample Rate** fields in the Settings dialog duplicate the fields in the Output Configuration panel.

Bit Depth

The Bit Depth control allows you to set the output bit depth (also called word length). The default setting is 24 bits, which is also the maximum setting. Minimum setting is 8 bits.

Enable Dither

When the **Enable Dither** checkbox is checked, dither is **ON** (the default). When **ON**, dither is set according to the bit depth, using TPDF (triangular probability density function) dither at ±1 LSB.

Uncheck the checkbox to turn dither **OFF**.

Interface

Application: Consumer

Consumer application (the default) sets two aspects of the unbalanced digital output signal:

1. It sets the nominal interface signal voltage at the unbalanced output connector to 0.5 Vpp. This is

the defined signal voltage for consumer applications such as SPDIF and IEC60958-3.

2. It sets the status bits for both subframes A and B to the consumer application. The status bits can be customized or even set to professional application in the Set Status Bits / User Bits panel.

Application: Professional

Professional application sets two aspects of the unbalanced digital output signal:

3. 1.It sets the nominal interface signal voltage at the unbalanced output connector to 1.0 Vpp. This is the defined signal voltage for unbalanced professional applications such as AES3id and SMPTE 276M.

4. 2.It sets the status bits for both subframes A and B to the professional application. The status bits can be customized or even set to consumer application in the Set Status Bits / User Bits panel.

Metadata

Status/User Bits

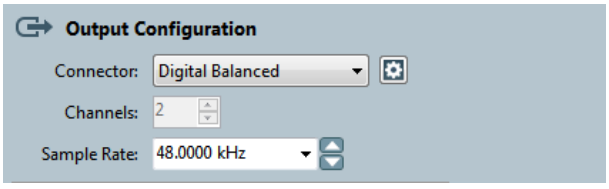
This button opens the Set Status Bits / Set User Bits dialog. See page 611 for more information about setting status bits and user bits.

Tx Valid

The last four time slots of each subframe contain a validity bit, a user data bit, a status bit, and a parity bit. By default, APx automatically transmits the bit as Valid for linear PCM embedded audio, and Invalid for coded audio.

- **Auto**
  APx automatically transmits the validity bit as **Valid** for linear PCM embedded audio, and **Invalid** for coded audio.
- **Valid**
  The validity bit is transmitted as **Valid** without regard to the nature of the embedded audio.
- **Invalid**
  The validity bit is transmitted as **Invalid** without regard to the nature of the embedded audio.

Output: Digital Balanced



This selection enables the balanced digital electrical output, available on a male XLR connector on the instrument front panel.

Note: Earlier APx585/586 Series analyzers may be fitted with a Model 109 Digital I/O module, which does not support balanced DIO.

Channels

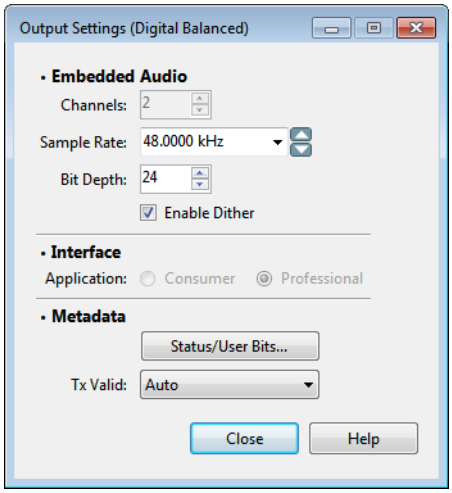
Audio channels are fixed at two.

Sample Rate

To set the output sample rate (Fs), click the up/down arrows to select a standard rate, or enter an arbitrary value in the **Sample Rate** field. Minimum setting is 8 kHz; maximum is 216 kHz. Default is 48 kHz. **Sample Rate** is also available in the Settings dialog.

The APx515 has a minimum output sample rate of 22 kHz.

Settings for Digital Balanced Output



Embedded Audio

The **Channels** and **Sample Rate** fields in the Settings dialog duplicate the fields in the Output Configuration panel.

Bit Depth

The Bit Depth control allows you to set the output bit depth (also called digital word length). The default setting is 24 bits, which is also the maximum setting. Minimum setting is 8 bits.

Enable Dither

When the **Enable Dither** checkbox is checked, dither is **ON** (the default). When **ON**, dither is set according to the bit depth, using TPDF (triangular probability density function) dither at ±1 LSB.

Uncheck the checkbox to turn dither **OFF**.

Interface

Application

The Application buttons are not available in Digital Balanced output; instead, these aspects of the balanced digital output signal are set:

5. 1.The nominal interface signal voltage at the balanced output connector is always set to 5.0 Vpp. This is within the range of defined signal voltages

for balanced professional applications such as AES3, AES/EBU and IEC60958-4.

6. 2.By default, the status bits for both subframes A and B are set to the professional application. The status bits can be customized or even set to consumer application in the Set Status Bits / User Bits panel.

Metadata

Status/User Bits

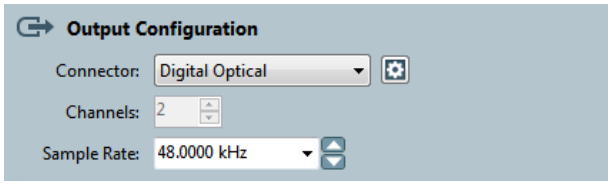
This button opens the Set Status Bits / Set User Bits dialog. See page 611 for more information about setting status bits and user bits.

Tx Valid

The last four time slots of each subframe contain a validity bit, a user data bit, a status bit, and a parity bit. By default, APx automatically transmits the bit as Valid for linear PCM embedded audio, and Invalid for coded audio.

- **Auto**
  APx automatically transmits the validity bit as **Valid** for linear PCM embedded audio, and **Invalid** for coded audio.
- **Valid**
  The validity bit is transmitted as **Valid** without regard to the nature of the embedded audio.
- **Invalid**
  The validity bit is transmitted as **Invalid** without regard to the nature of the embedded audio.

Output: Digital Optical



This selection enables the digital optical output, available on a Toslink connector on the instrument front panel.

Channels

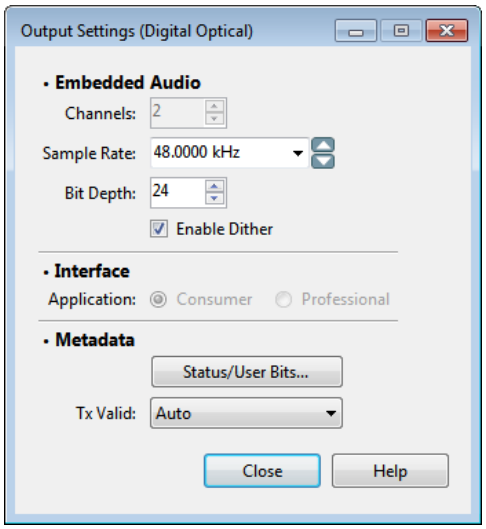
Audio channels are fixed at two.

Sample Rate

To set the output sample rate (Fs), click the up/down arrows to select a standard rate, or enter an arbitrary value in the **Sample Rate** field. Minimum setting is 8 kHz; maximum is 216 kHz. Default is 48 kHz. **Sample Rate** is also available in the Settings dialog.

The APx515 has a minimum output sample rate of 22 kHz.

Settings for Digital Optical Output



Embedded Audio

The **Channels** and **Sample Rate** fields in the Settings dialog duplicate the fields in the Output Configuration panel.

Bit Depth

The Bit Depth control allows you to set the output bit depth (also called digital word length). The default setting is 24 bits, which is also the maximum setting. Minimum setting is 8 bits.

Enable Dither

When the **Enable Dither** checkbox is checked, dither is **ON** (the default). When **ON**, dither is set according to the bit depth, using TPDF (triangular probability density function) dither at ±1 LSB.

Uncheck the checkbox to turn dither **OFF**.

Interface

Application

The Application buttons are not available in Digital Optical output; instead, the status bits for both subframes A and B are set to the consumer application. The status bits can be customized or even set to professional application in the Set Status Bits / User Bits panel.

Metadata

Status/User Bits

This button opens the Set Status Bits / Set User Bits dialog. See page 611 for more information about setting status bits and user bits.

Tx Valid

The last four time slots of each subframe contain a validity bit, a user data bit, a status bit, and a parity bit. By default, APx automatically transmits the bit as Valid for linear PCM embedded audio, and Invalid for coded audio.

- **Auto**
  APx automatically transmits the validity bit as **Valid** for linear PCM embedded audio, and **Invalid** for coded audio.
- **Valid**
  The validity bit is transmitted as **Valid** without regard to the nature of the embedded audio.
- **Invalid**
  The validity bit is transmitted as **Invalid** without regard to the nature of the embedded audio.

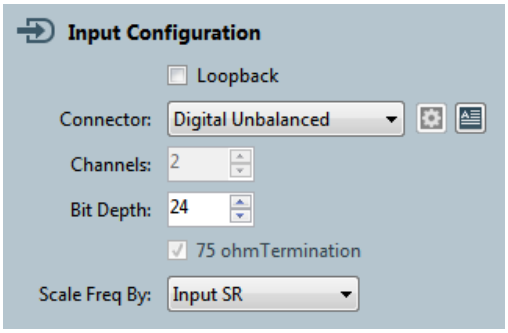
Input Configuration

Operation with absent or corrupt interface signal

When Input Configuration is set to a digital input and no signal is present, or if the signal is corrupt or out of range, the input receiver cannot lock (synchronize) to the interface signal, and no valid audio can be recovered. The sample rate indicator in the Status Bar will display an "unlocked" warning, and any measurement result referencing audio from the unlocked input will display an "----" (invalid) result. See page 636 for more about invalid results.

The **Settings** button is unavailable for DIO inputs.

Input: Digital Unbalanced



This selection enables the unbalanced digital electrical input, available on a BNC connector on the instrument front panel.

Channels

Audio channels are fixed at two.

Bit Depth

The Bit Depth control allows you to set the input bit depth (digital word length). The default setting is 24 bits, which is also the maximum setting. Minimum setting is 8 bits. The input signal is truncated (not dithered down) if Bit Depth is set to a lower value than Bit Depth of the signal.

We recommend that you use the default setting of 24 bits, unless you have a special application that requires that you mask out low-order bits.

75 ohm Termination checkbox

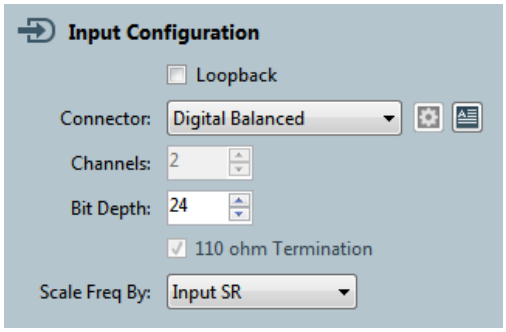
When Digital Unbalanced is selected, the **75 ohm Termination** checkbox is available on the panel. When this checkbox is set (the default), the digital unbalanced input is terminated in a resistance of 75 Ω. This is the correct setting for terminating an unbalanced digital audio connection for both the SPDIF / IEC60958-3 (consumer) and AES-3id / SMPTE276M (professional) formats. If you uncheck the checkbox, the digital unbalanced input is unterminated. An unterminated setting is useful when bridging a digital connection (as with a "T" connector). An unbalanced digital connection should be properly terminated in 75 Ω at the end of its run.

Scale Freq By:

- **Input SR**
  For most testing, use **Input SR** (the default).
- **Output SR**
  If sample rate is being swept, you may want to use the output sampling rate as the scaling reference.
- **Fixed**
  If jitter is present on the input signal, you may want to reference the sampled audio data to a fixed rate, rather than the jittered input rate. This may improve phase and frequency stability of the embedded audio. Set **Scale Freq By:** to **Fixed**, and enter the nominal input sample rate in the **Fixed Rate** field.

See More About Frequency Scaling in APx on page 72 for a detailed explanation.

Input: Digital Balanced



Note: Earlier APx585/586 Series analyzers may be fitted with a Model 109 Digital I/O module, which does not support balanced DIO.

Digital Balanced enables the balanced digital electrical input, available on an XLR female connector on the instrument front panel. This input is optimized to accept signal at the professional signal level for balanced signals compatible with AES3 and IEC60958-4 (5 Vpp).

Channels

Audio channels are fixed at two.

Bit Depth

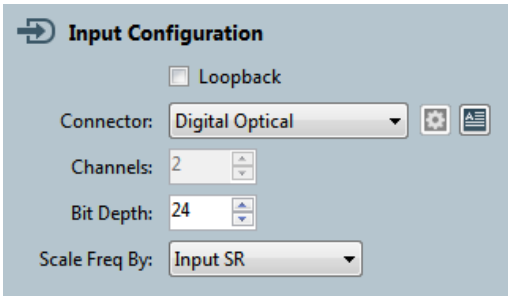
The Bit Depth control allows you to set the input bit depth (digital word length). The default setting is 24 bits, which is also the maximum setting. Minimum setting is 8 bits. The input signal is truncated (not dithered down) if Bit Depth is set to a lower value than Bit Depth of the signal.

We recommend that you use the default setting of 24 bits, unless you have a special application that requires that you mask out low-order bits.

110 Ohm Termination checkbox

When Digital Balanced is selected, the 110 Ohm Termination checkbox is available on the panel. When this checkbox is set (the default), the digital balanced input is terminated in a resistance of 110 Ω. This is the correct setting for terminating a balanced digital audio connection for AES3 / AES-EBU / IEC60958-4 professional formats. If you uncheck the checkbox, the digital balanced input is unterminated. This setting is useful when bridging a digital connection (as with a "Y" cable). A balanced digital connection should be properly terminated in 110 Ω at the end of its run. More settings are available in the Settings dialog.

Input: Digital Optical



Digital Optical enables the digital optical input, available on a Toslink connector on the instrument front panel.

Channels

Audio channels are fixed at two.

Bit Depth

The Bit Depth control allows you to set the input bit depth (digital word length). The default setting is 24 bits, which is also the maximum setting. Minimum setting is 8 bits. The input signal is truncated (not dithered down) if Bit Depth is set to a lower value than Bit Depth of the signal.

We recommend that you use the default setting of 24 bits, unless you have a special application that requires that you mask out low-order bits.

SPS: Advanced Digital I/O

Introduction

The advanced digital input/output (ADIO) interface is the standard digital interface on an APx555, and an option for the APx52x/58x family. The ADIO is similar to the standard DIO described in Chapter 13, but differs in that it offers impairments to the transmitted digital interface signal, and the ability to measure certain characteristics of the received interface signal.

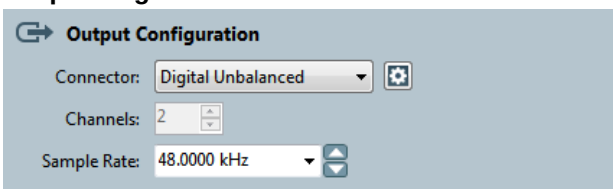
The transmitted interface signal can be impaired with false metadata, validity and parity errors, jitter, noise, common mode and risetime impairments (including cable simulation), to test the tolerance of a downstream device.

The level and sample rate of the received interface signal can be measured, and the jitter can be analyzed in great detail.

For information on the other supported digital interfaces, see Chapter 15 (DSIO), Chapter 16 (HDMI), Chapter 17 (Bluetooth), and Chapter 19, (PDM).

Output Configuration

Output: Digital Unbalanced



This selection enables the unbalanced digital electrical output, available on a BNC connector on the instrument front panel.

The **Edit Connector Settings** button opens an output **Settings** dialog for each configuration, offering more detailed control.

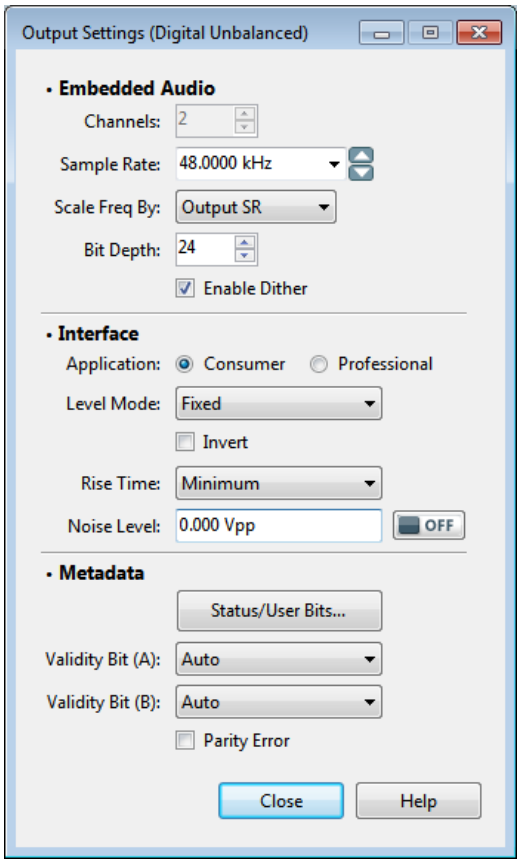
Channels

Audio channels are fixed at two.

Sample Rate

To set the output sample rate (Fs), click the up/down arrows to select a standard rate, or enter an arbitrary value in the **Sample Rate** field. Minimum setting is 8 kHz; maximum is 216 kHz. Default is 48 kHz. **Sample Rate** is also available in the Settings dialog.

Settings for Digital Unbalanced Output



Embedded Audio

Channels

The **Channels** and **Sample Rate** fields in the Settings dialog duplicate the fields in the Output Configuration panel.

Channels are fixed at **2.**

Sample Rate

The **Channels** and **Sample Rate** fields in the Settings dialog duplicate the fields in the Output Configuration panel.

To set the output sample rate (Fs), click the up/down arrows to select a standard rate, or enter an arbitrary value in the **Sample Rate** field. Minimum setting is **8 kHz**; maximum is **216 kHz**. Default is 48 kHz.

Scale Freq By:

- **Output SR**
  For most testing, use **Output SR** (the default).
- **Fixed**
  If you are applying jitter to the output signal at high levels, you may want to reference the embedded audio to a fixed rate, rather than the jittered output rate. This may improve phase and frequency stability of the embedded audio. Set **Scale Freq By:** to **Fixed**, and enter the nominal output sample rate in the **Fixed Rate** field.

Bit Depth

The **Bit Depth** control allows you to set the output bit depth (also called word length). The default setting is 24 bits, which is also the maximum setting. Minimum setting is 8 bits.

Enable Dither

When the **Enable Dither** checkbox is checked, dither is **ON** (the default). When **ON**, dither is set according to the bit depth, using TPDF (triangular probability density function) dither at ±1 LSB.

Uncheck the checkbox to turn dither **OFF**.

Interface

Application: Consumer

Consumer application (the default) sets two aspects of the unbalanced digital output signal:

1. It sets the nominal interface signal voltage at the unbalanced output connector to 0.5 Vpp. This is the defined signal voltage for consumer applications such as SPDIF and IEC60958-3.
2. It sets the status bits for both subframes A and B to the consumer application. The status bits can be customized or even set to professional application in the Set Status Bits / User Bits panel.

Application: Professional

Professional application sets two aspects of the unbalanced digital output signal:

3. 1.It sets the nominal interface signal voltage at the unbalanced output connector to 1.0 Vpp. This is the defined signal voltage for unbalanced professional applications such as AES3id and SMPTE 276M.
4. 2.It sets the status bits for both subframes A and B to the professional application. The status bits can be customized or even set to consumer application in the Set Status Bits / User Bits panel.

Level Mode

- **Fixed**
  The standard voltage for the unbalanced consumer application is 0.5 Vpp; for the unbalanced professional application it is 1.0 Vpp. The Fixed selection sets the interface voltage to the standard, according to the Application settings made above.
- **Custom**
  You can enter a custom value for the interface voltage in the Level field, as an impairment to test a downstream receiver. The acceptable range is 0 Vpp to 2.5 Vpp.

Invert

This checkbox inverts the interface signal waveform to test the tolerance of a downstream receiver.

Rise Time

The rise time of the pulses in the interface waveform should be short for error-free reception. However, cable capacitance and other environmental factors can degrade the signal by lengthening pulse rise time. Rise time impairments can be imposed on the interface signal to test the tolerance of a downstream receiver.

- **Minimum**
  The **Minimum** setting uses no artificial impairments and is the fastest rise time available from the digital transmitter.
- **Cable Simulation**
  The **Cable Simulation** setting lengthens the pulse rise time by simulating the capacitance of a long cable.
- **Custom**
  Enter a custom rise time impairment in the field provided. The acceptable range is 12.00 ns to 100.0 ns.

Noise Level

You can add noise to the interface signal as an impairment to test the tolerance of a downstream receiver. Enter the **Noise Level** in the field provided, and click **ON**. The acceptable range is 0.0 Vpp to 625.0 mVpp.

Metadata

Status/User Bits

This button opens the Set Status Bits / Set User Bits dialog. See page 611 for more information about setting status bits and user bits.

Validity Bit (A) / Validity Bit (B)

The last four time slots of each subframe contain a validity bit, a user data bit, a status bit, and a parity bit. By default, APx automatically transmits the bit as

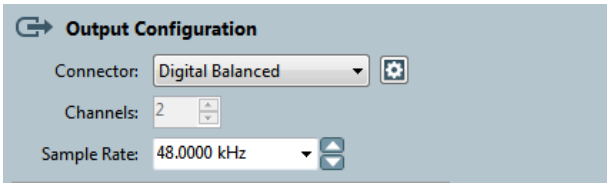
Valid for linear PCM embedded audio, and Invalid for coded audio.

- **Auto**
  APx automatically transmits the validity bit as **Valid** for linear PCM embedded audio, and **Invalid** for coded audio.
- **Valid**
  The validity bit is transmitted as **Valid** without regard to the nature of the embedded audio.
- **Invalid**
  The validity bit is transmitted as **Invalid** without regard to the nature of the embedded audio.

Parity Error

One of the administrative bits in the interface is the parity (**P**) bit. In normal operation, even parity is maintained in each subframe. By setting a checkmark in the **Parity Error** checkbox, the parity bit can be set to produce odd parity, as an impairment to test a downstream receiver.

Output: Digital Balanced



This selection enables the balanced digital electrical output, available on a male XLR connector on the instrument front panel.

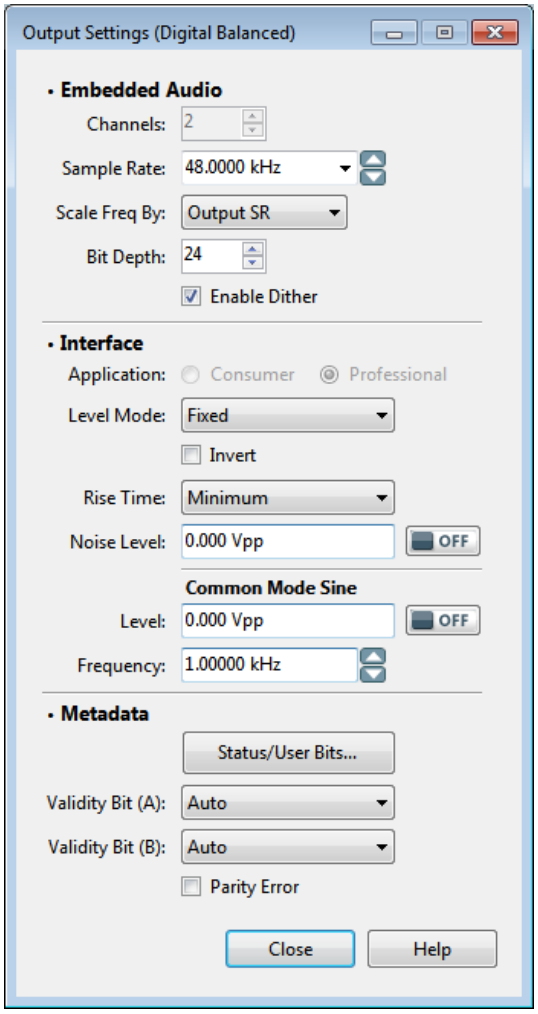
Channels

Audio channels are fixed at two.

Sample Rate

To set the output sample rate (Fs), click the up/down arrows to select a standard rate, or enter an arbitrary value in the **Sample Rate** field. Minimum setting is 8 kHz; maximum is 216 kHz. Default is 48 kHz. **Sample Rate** is also available in the Settings dialog.

Settings for Digital Balanced Output



Embedded Audio

The **Channels** and **Sample Rate** fields in the Settings dialog duplicate the fields in the Output Configuration panel.

Scale Freq By:

- **Output SR**
  For most testing, use **Output SR** (the default).
- **Fixed**
  If you are applying jitter to the output signal at high levels, you may want to reference the embedded audio to a fixed rate, rather than the jittered output rate. This may improve phase and frequency stability of the embedded audio. Set **Scale Freq By:** to **Fixed**, and enter the nominal output sample rate in the **Fixed Rate** field.

Bit Depth

The Bit Depth control allows you to set the output bit depth (also called digital word length). The default setting is 24 bits, which is also the maximum setting. Minimum setting is 8 bits.

Enable Dither

When the **Enable Dither** checkbox is checked, dither is **ON** (the default). When **ON**, dither is set according to the bit depth, using TPDF (triangular probability density function) dither at ±1 LSB.

Uncheck the checkbox to turn dither **OFF**.

Interface

Application

The Application buttons are not available in Digital Balanced output; instead, these aspects of the balanced digital output signal are set:

5. 1.The nominal interface signal voltage at the balanced output connector is always set to 5.0 Vpp. This is within the range of defined signal voltages for balanced professional applications such as AES3, AES/EBU and IEC60958-4.
6. 2.By default, the status bits for both subframes A and B are set to the professional application. The status bits can be customized or even set to consumer application in the Set Status Bits / User Bits panel.

Level Mode

- **Fixed**
  The standard voltage for the unbalanced consumer application is 0.5 Vpp; for the unbalanced professional application it is 1.0 Vpp. The Fixed selection sets the interface voltage to the standard, according to the Application settings made above.
- **Custom**
  You can enter a custom value for the interface voltage in the Level field, as an impairment to test a downstream receiver. The acceptable range is 0 Vpp to 2.5 Vpp.

Invert

This checkbox inverts the interface signal waveform to test the tolerance of a downstream receiver.

Rise Time

The rise time of the pulses in the interface waveform should be short for error-free reception. However, cable capacitance and other environmental factors can degrade the signal by lengthening pulse rise time. Rise time impairments can be imposed on the interface signal to test the tolerance of a downstream receiver.

- **Minimum**
  The **Minimum** setting uses no artificial impairments and is the fastest rise time available from the digital transmitter.
- **Cable Simulation**
  The **Cable Simulation** setting lengthens the pulse rise time by simulating the capacitance of a long cable.
- **Custom**
  Enter a custom rise time impairment in the field provided. The acceptable range is 12.00 ns to 100.0 ns.

Noise Level

You can add noise to the interface signal as an impairment to test the tolerance of a downstream receiver. Enter the **Noise Level** in the field provided, and click **ON**. The acceptable range is 0.0 Vpp to 2.0.

Common Mode Sine

A sine signal can be imposed on the digital interface signal as a common mode impairment, to test the tolerance of a downstream receiver.

Level
Set the common mode sine level here. The acceptable range is 0.0 Vpp to 20.0 Vpp.

Frequency
Set the common mode sine frequency here. The acceptable range is 20 Hz to 100 kHz.

Metadata

Status/User Bits

This button opens the Set Status Bits / Set User Bits dialog. See page 611 for more information about setting status bits and user bits.

Validity Bit (A) / Validity Bit (B)

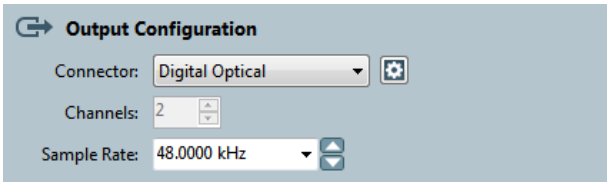
The last four time slots of each subframe contain a validity bit, a user data bit, a status bit, and a parity bit. By default, APx automatically transmits the bit as Valid for linear PCM embedded audio, and Invalid for coded audio.

- **Auto**
  APx automatically transmits the validity bit as **Valid** for linear PCM embedded audio, and **Invalid** for coded audio.
- **Valid**
  The validity bit is transmitted as **Valid** without regard to the nature of the embedded audio.
- **Invalid**
  The validity bit is transmitted as **Invalid** without regard to the nature of the embedded audio.

Parity Error

One of the administrative bits in the interface is the parity (**P**) bit. In normal operation, even parity is maintained in each subframe. By setting a checkmark in the **Parity Error** checkbox, the parity bit can be set to produce odd parity, as an impairment to test a downstream receiver.

Output: Digital Optical



This selection enables the digital optical output, available on a Toslink connector on the instrument front panel.

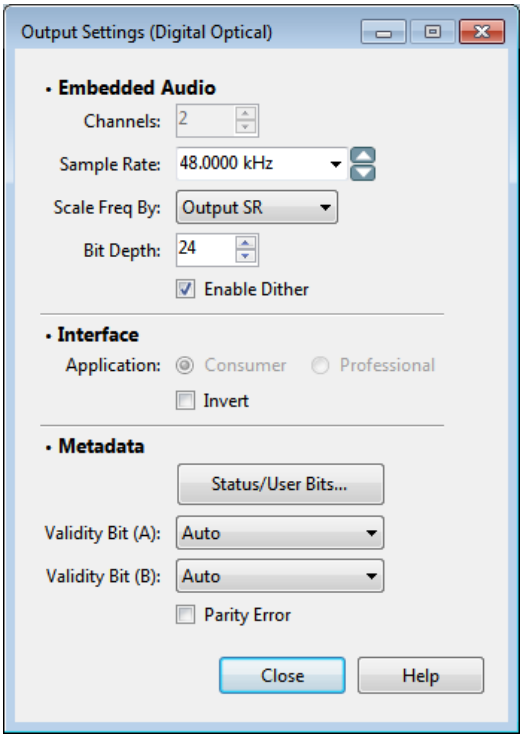
Channels

Audio channels are fixed at two.

Sample Rate

To set the output sample rate (Fs), click the up/down arrows to select a standard rate, or enter an arbitrary value in the **Sample Rate** field. Minimum setting is 8 kHz; maximum is 216 kHz. Default is 48 kHz. **Sample Rate** is also available in the Settings dialog.

Settings for Digital Optical Output



Embedded Audio

The **Channels** and **Sample Rate** fields in the Settings dialog duplicate the fields in the Output Configuration panel.

Scale Freq By:

- **Output SR**
  For most testing, use **Output SR** (the default).
- **Fixed**
  If you are applying jitter to the output signal at high levels, you may want to reference the embedded audio to a fixed rate, rather than the jittered output rate. This may improve phase and frequency stability of the embedded audio. Set **Scale Freq By:** to **Fixed**, and enter the nominal output sample rate in the **Fixed Rate** field.

Bit Depth

The Bit Depth control allows you to set the output bit depth (also called digital word length). The default setting is 24 bits, which is also the maximum setting. Minimum setting is 8 bits.

Enable Dither

When the **Enable Dither** checkbox is checked, dither is **ON** (the default). When **ON**, dither is set according to the bit depth, using TPDF (triangular probability density function) dither at ±1 LSB.

Uncheck the checkbox to turn dither **OFF**.

Interface

Application

The Application buttons are not available in Digital Optical output; instead, the status bits for both subframes A and B are set to the consumer application. The status bits can be customized or even set to professional application in the Set Status Bits / User Bits panel.

Invert

This checkbox inverts the interface signal waveform to test the tolerance of a downstream receiver.

Metadata

Status/User Bits

This button opens the Set Status Bits / Set User Bits dialog. See page 611 for more information about setting status bits and user bits.

Validity Bit (A) / Validity Bit (B)

The last four time slots of each subframe contain a validity bit, a user data bit, a status bit, and a parity bit. By default, APx automatically transmits the bit as Valid for linear PCM embedded audio, and Invalid for coded audio.

- **Auto**
  APx automatically transmits the validity bit as **Valid** for linear PCM embedded audio, and **Invalid** for coded audio.
- **Valid**
  The validity bit is transmitted as **Valid** without regard to the nature of the embedded audio.
- **Invalid**
  The validity bit is transmitted as **Invalid** without regard to the nature of the embedded audio.

Parity Error

One of the administrative bits in the interface is the parity (**P**) bit. In normal operation, even parity is maintained in each subframe. By setting a checkmark in the **Parity Error** checkbox, the parity bit can be set to

produce odd parity, as an impairment to test a downstream receiver.

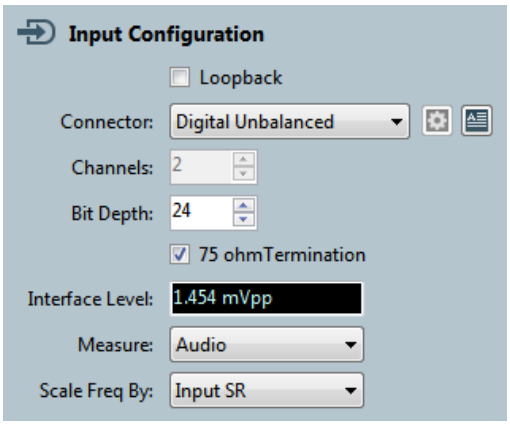
Input Configuration

Operation with absent or corrupt interface signal

When Input Configuration is set to a digital input and no signal is present, or if the signal is corrupt or out of range, the input receiver cannot lock (synchronize) to the interface signal, and no valid audio can be recovered. The sample rate indicator in the Status Bar will display an "unlocked" warning, and any measurement result referencing audio from the unlocked input will display an "----" (invalid) result. See page 636 for more about invalid results.

The **Settings** button is unavailable for ADIO inputs.

Input: Digital Unbalanced



This selection enables the unbalanced digital electrical input, available on a BNC connector on the instrument front panel.

Channels

Audio channels are fixed at two.

Bit Depth

The Bit Depth control allows you to set the input bit depth (digital word length). The default setting is 24 bits, which is also the maximum setting. Minimum setting is 8 bits. The input signal is truncated (not dithered down) if Bit Depth is set to a lower value than Bit Depth of the signal.

We recommend that you use the default setting of 24 bits, unless you have a special application that requires that you mask out low-order bits.

75 ohm Termination checkbox

When Digital Unbalanced is selected, the **75 ohm Termination** checkbox is available on the panel. When this checkbox is set (the default), the digital unbalanced input is terminated in a resistance of 75 Ω. This is the correct setting for terminating an unbalanced digital audio connection for both the SPDIF / IEC60958-3 (consumer) and AES-3id / SMPTE276M (professional) formats. If you uncheck the checkbox, the digital unbalanced input is unterminated. An unterminated setting is useful when bridging a digital connection (as with a "T" connector). An unbalanced digital connection should be properly terminated in 75 Ω at the end of its run.

Interface Level (Sequence Mode)

This field is only available in Sequence Mode when the Advanced Digital Input/Output module is fitted, and Input Configuration is set to Digital Unbalanced or Digital Balanced. It displays the peak-to-peak voltage level of the incoming digital interface waveform.

In Bench Mode, the same information is available from the Digital Interface Level meter.

Measure

The Measure menu is only available for a jitter-enabled Input/Output module when the Advanced Master Clock is installed in the analyzer.

The Measure menu allows you to select whether the audio signal or the jitter is routed to the analyzer input.

- **Audio**
  This is the normal setting. The embedded audio is routed to all measurements and meter readings.
- **Jitter (UI)**
  This is the jitter setting. The jitter signal is extracted from the carrier and routed to all measurements and meter readings. Results are shown in Unit Interval (UI) units.
- **Jitter (sec)**
  This is an alternative Jitter setting. The jitter is measured as above, but the results are shown in seconds (s) units.

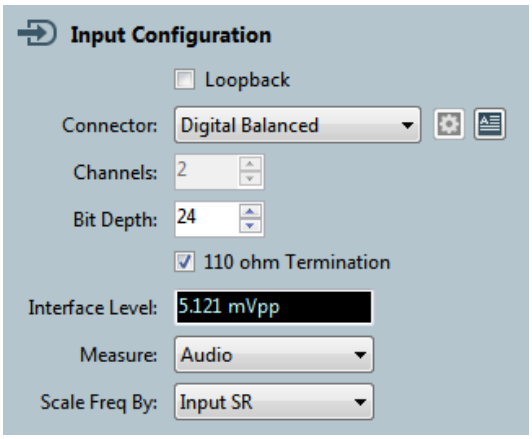
Read more about jitter on page 69.

Scale Freq By:

- **Input SR**
  For most testing, use **Input SR** (the default).
- **Output SR**
  If sample rate is being swept, you may want to use the output sampling rate as the scaling reference.
- **Fixed**
  If jitter is present on the input signal, you may want to reference the sampled audio data to a fixed rate, rather than the jittered input rate. This may improve phase and frequency stability of the embedded audio. Set **Scale Freq By:** to **Fixed**, and enter the nominal input sample rate in the **Fixed Rate** field.

See More About Frequency Scaling in APx on page 72 for a detailed explanation.

Input: Digital Balanced



Digital Balanced enables the balanced digital electrical input, available on an XLR female connector on the instrument front panel. This input is optimized to accept signal at the professional signal level for balanced signals compatible with AES3 and IEC60958-4 (5 Vpp).

Channels

Audio channels are fixed at two.

Bit Depth

The Bit Depth control allows you to set the input bit depth (digital word length). The default setting is 24 bits, which is also the maximum setting. Minimum setting is 8 bits. The input signal is truncated (not dithered down) if Bit Depth is set to a lower value than Bit Depth of the signal.

We recommend that you use the default setting of 24 bits, unless you have a special application that requires that you mask out low-order bits.

110 Ohm Termination checkbox

When Digital Balanced is selected, the 110 Ohm Termination checkbox is available on the panel. When this checkbox is set (the default), the digital balanced input is terminated in a resistance of 110 Ω. This is the correct setting for terminating a balanced digital audio connection for AES3 / AES-EBU / IEC60958-4 professional formats. If you uncheck the checkbox, the digital balanced input is unterminated. This setting is useful when bridging a digital connection (as with a "Y" cable). A balanced digital connection should be properly terminated in 110 Ω at the end of its run. More settings are available in the Settings dialog.

Interface Level (Sequence Mode)

This field is only available in Sequence Mode when the Advanced Digital Input/Output module is fitted, and Input Configuration is set to Digital Unbalanced or Digital Balanced. It displays the peak-to-peak voltage level of the incoming digital interface waveform.

In Bench Mode, the same information is available from the Digital Interface Level meter.

Measure

The Measure menu is only available for a jitter-enabled Input/Output module when the Advanced Master Clock is installed in the analyzer.

- **Audio**
  This is the normal setting. The embedded audio is routed to all measurements and meter readings.
- **Jitter (UI)**
  This is the jitter setting. The jitter signal is extracted from the carrier and routed to all measurements and meter readings. Results are shown in Unit Interval (UI) units.
- **Jitter (sec)**
  This is an alternative Jitter setting. The jitter is measured as above, but the results are shown in seconds (s) units.

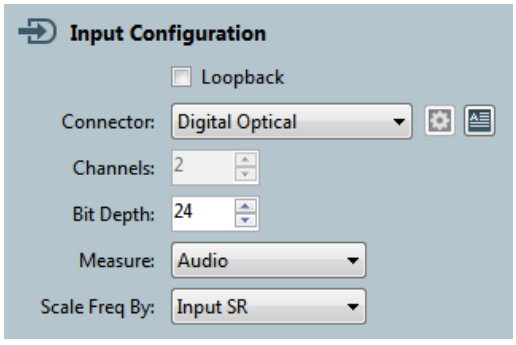
Read more about jitter on page 69.

Scale Freq By:

- **Input SR**
  For most testing, use **Input SR** (the default).
- **Output SR**
  If sample rate is being swept, you may want to use the output sampling rate as the scaling reference.
- **Fixed**
  If jitter is present on the input signal, you may want to reference the sampled audio data to a fixed rate, rather than the jittered input rate. This may improve phase and frequency stability of the embedded audio. Set **Scale Freq By:** to **Fixed**, and enter the nominal input sample rate in the **Fixed Rate** field.

See More About Frequency Scaling in APx on page 72 for a detailed explanation.

Input: Digital Optical



Digital Optical enables the digital optical input, available on a Toslink connector on the instrument front panel.

Channels

Audio channels are fixed at two.

Bit Depth

The Bit Depth control allows you to set the input bit depth (digital word length). The default setting is 24 bits, which is also the maximum setting. Minimum setting is 8 bits. The input signal is truncated (not dithered down) if Bit Depth is set to a lower value than Bit Depth of the signal.

We recommend that you use the default setting of 24 bits, unless you have a special application that requires that you mask out low-order bits.

Measure

The Measure menu is only available for a jitter-enabled Input/Output module when the Advanced Master Clock is installed in the analyzer.

- **Audio**
  This is the normal setting. The embedded audio is routed to all measurements and meter readings.
- **Jitter (UI)**
  This is the jitter setting. The jitter signal is extracted from the carrier and routed to all measurements and meter readings. Results are shown in Unit Interval (UI) units.
- **Jitter (sec)**
  This is an alternative Jitter setting. The jitter is measured as above, but the results are shown in seconds (s) units.

Read more about jitter on page 69.

Scale Freq By:

- **Input SR**
  For most testing, use **Input SR** (the default).
- **Output SR**
  If sample rate is being swept, you may want to use the output sampling rate as the scaling reference.
- **Fixed**
  If jitter is present on the input signal, you may want to reference the sampled audio data to a fixed rate, rather than the jittered input rate. This may improve phase and frequency stability of the embedded audio. Set **Scale Freq By:** to **Fixed**, and enter the nominal input sample rate in the **Fixed Rate** field.

See More About Frequency Scaling in APx on page 72 for a detailed explanation.

DSIO

Introduction

The Digital Serial Interface (or DSIO) option provides a flexible chip- or board-level serial input and output interface. DSIO is a hardware option available for the APx52x, 555 and 58x families of analyzers.

The DSIO allows you to transmit or receive Digital Serial data in a number of different formats. The DSIO physical layer provides connections for Master Clock, Frame Clock, Bit Clock, Channel Clock and 4 Data lines.

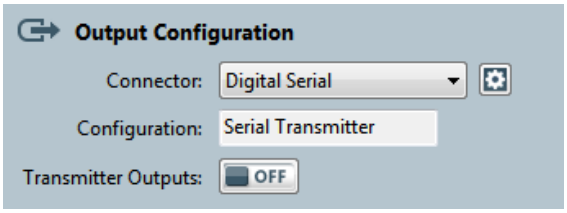
> *The DSIO Receiver Frame Clock is available for arbitrary use as a reference clock, when Input Configuration is not DSIO. See page 38.*

DSIO Jitter Testing

DSIO modules shipped after August 5th, 2015 are jitter-enabled. Earlier DSIO modules can be upgraded to enable jitter. For more information see Jitter Support in APx beginning on page 71.

Output Configuration

Digital Serial (when installed)



This selection enables the Digital Serial output available on the Transmitter HD-15 connector on the instrument front panel; also, the transmitter **Monitor** output is enabled.

Transmitter Outputs

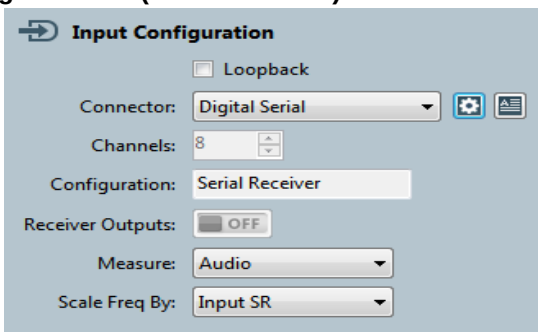
When Output Configuration: Digital Serial is selected, the **Transmitter Outputs ON/OFF** switch (duplicated in the Settings dialog) is available. This is a “kill switch” that, when **OFF**, removes all output voltages from the DSIO transmitter output connector pins.

Settings

The **Settings** button opens the DSIO output **Settings** dialog. Settings for DSIO are quite extensive. See Settings for Digital Serial output on page 108.

Input Configuration

Digital Serial (when installed)



This Connector selection enables the Digital Serial input available on the Receiver HD-15 connector on the instrument front panel; also, the receiver Monitor output is enabled.

Loopback

When the Loopback checkbox is set, the digital input is disconnected from the front panel input connector and is instead routed to the digital output circuits. This function is called GenMon in other Audio Precision products. Loopback allows you to view the signal present at the digital output on the analyzer meters or graphs.

Channels

The Channels setting is a read-only display on the Signal Path Setup panel. Go to the Input Settings (Digital Serial Receiver) dialog to set channels.

Settings

The **Settings** button opens the DSIO input **Settings** dialog. Settings for DSIO are quite extensive. See Settings for Digital Serial input on page 112.

Receiver Outputs

When Input Configuration: Digital Serial is selected, the **Receiver Outputs ON/OFF** switch (duplicated in the Settings dialog) is available. This is a “kill switch” that, when **OFF**, removes all output voltages from the DSIO receiver input connector pins.

Measure

The Measure menu is only available for a jitter-enabled DSIO module when the Advanced Master Clock is installed in the analyzer.

The Measure menu allows you to select whether the receiver audio signal or the bit clock jitter signal is routed to the analyzer.

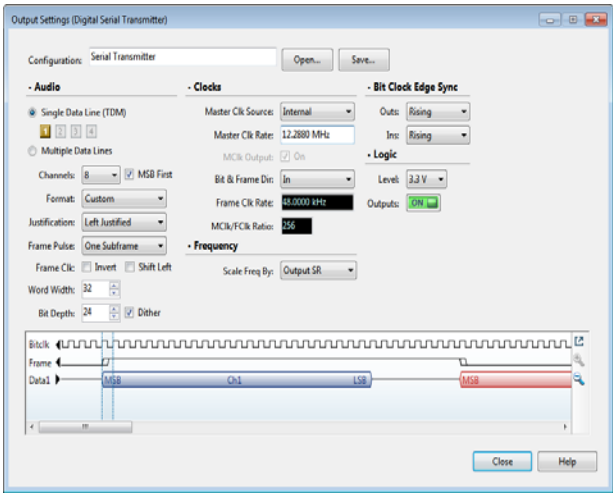
- **Audio**
  This is the normal setting. The audio signal is routed to all measurements and meter readings.
- **Jitter (UI)**
  This is the jitter setting. The jitter signal is extracted from receiver bit clock line and routed to all measurements and meter readings. Results are shown in Unit Interval (UI) units.
- **Jitter (sec)**
  This is an alternative Jitter setting. The jitter is measured as above, but the results are shown in seconds (s) units.

Read more about jitter on page 69.

Operation with absent or corrupt interface signal

When Input Configuration is set to a digital input and no signal is present, or if the signal is corrupt or out of range, the input receiver cannot lock (synchronize) to the interface signal, and no valid audio can be recovered. The sample rate indicator in the Status Bar will display an “unlocked” warning, and any measurement result referencing audio from the unlocked input will display an “----” (invalid) result. See page 636 for more about invalid results.

Output Settings for the Digital Serial Transmitter



This dialog allows you to set, save or load digital serial transmitter configurations. A dynamic timing diagram provides an on-screen display of your configuration settings. Transmitter connections are provided on an HD-15 connector, with four sets of Master Clock, Bit Clock, Channel Clock and Data connections. A monitor connector provides these signals for oscilloscope monitoring.

The channel clock is provided as an aid for users monitoring on an oscilloscope. The channel clock signal is aligned with the data and has a rising edge with each channel change.

See illustrations of typical Digital Serial Input/Output configurations beginning on page 114.

Common Settings for Digital Serial Output

Most of the settings on this panel are common to all Digital Serial Output configurations, and are discussed first.

The settings in the transmitter Clocks area vary with clock configuration, and the four possible cases are discussed following these common settings.

Timing Diagram

The timing diagram at the bottom of the panel is dynamic, changing to illustrate your current settings. Use the zoom controls to contract or expand the horizontal axis of the display, or click the **Undock** button to display a large view.

Configuration

Type a name for your input configuration here.

Open

Click **Open** to view the Open Digital Serial Transmitter Settings dialog, where you can load saved *.stx configuration files.

Save

Click **Save** to view the Save Digital Serial Receiver Settings dialog, where you can save the current configuration as an *.stx file.

Audio

Data Lines

- **Single (TDM)** uses time division multiplexing to embed the audio data in a single data stream. Channels are embedded in the data stream sequentially, beginning with channel 1, followed by a channel 2 followed by channel 3, and so on. Although only a single, multichannel data stream can be configured, this stream can be sent on any combination (or none) of the four data line conductors in the physical layer. Use the **Data Line** buttons to select the output data line(s).
- **Multiple** uses 1, 2 or 4 separate data lines, each outputted on individual data line conductors in the physical layer. Each of the data lines carries 2 channels of audio data.

Channels

- When configured for a single data line (TDM), the Digital Serial Output can transmit 1, 2, 4, 6, 8 or 16 audio data channels. Select the number of channels from the drop-down menu.
- When configured for multiple data lines, the Digital Serial Output can transmit 1, 2, 4, 6, or 8 audio data channels. Select the number of channels from the drop-down menu.

MSB First

The active audio data is a string of bits, whose number is set by Bit Depth. When **MSB First** is set, the MSB (Most Significant Bit) leads the string. When **MSB First** is not set, the LSB (Least Significant Bit) leads the string.

Format

Set the transmitter format here.

- **I2S** (spoken “I-squared-S”) is an acronym for the Philips “inter IC sound” bus. I^2S is a common audio serial interface format. **I2S** sets the Frame Clock low for the first channel (left or “1”) and high for the second channel (right or “2”). If there are more channels, the Frame Clock stays high until the beginning of the next frame.

  I^2S data is offset by one bit clock period from the beginning of each word.
- **DSP**

  In the DSP (digital signal processing) format, the frame clock pulse is the same length as one bit clock period.

  Like I^2S, DSP offsets the data by one bit clock period from the beginning of each word.
- **Custom**

  The Custom selection allows you to set Justification and Frame Pulse in configurations not defined by I^2S or DSP formats.

Justification

- **Left Justified**

  Left Justified sets the active bits fully to the left (leading) edge of the data word. Padding bits, if any, trail the data.
- **Right Justified**

  Right Justified sets the active bits fully to the right (trailing) edge of the data word. Padding bits, if any, lead the data.

Frame Pulse

When **Format** is **Custom**, you can set the width of the Frame Clock Pulse here.

- **One Bit Clock**

  The Frame Clock pulse is one Bit Clock bit wide.
- **One Subframe**

  The Frame Clock pulse is one subframe wide. The subframe is the width of one data word. A subframe is a frame divided by the number of channels on that data line.
- **50% Duty Cycle**

  The Frame Clock pulse is set to a 50% duty cycle, which makes the width of the Frame Clock pulse 1/2 the width of the frame.

Frame Clk: Invert

Check this box to invert the polarity of the Frame Clock.

Frame Clk: Shift Left

Check this box to shift the Frame Clock one Bit Clock pulse to the left, relative to the data.

Word Width

The word width is the width, in bits, of one data word, or subframe. A subframe is a frame divided by the number of channels. Set the word width here. Maximum is 128 bits; minimum is 8 bits. Word width cannot be less than bit depth.

Bit Depth

Set the bit depth (audio data length) here. Maximum is 32 bits; minimum is 8 bits.

When the **Bit Depth** is set to the same value of the **Word Width**, the entire bit allocation for the channel carries audio data. When **Bit Depth** is less than **Word Width**, only some of the bits carry data. The remaining bits are called pads or padding bits. The digital

serial transmitter sets the logic state of padding bits to logical low.

Dither

When the **Dither** checkbox is checked, dither is **ON** (the default). When **ON**, dither is set according to the bit depth, using TPDF (triangular probability density function) dither at ±1 LSB.

Uncheck the checkbox to turn dither **OFF**.

Clocks

The visibility and availability of controls and readings in the transmitter Clocks area depend upon the settings of **Master Clock Source** and **Bit & Frame Clock Direction**. There are four possible configurations; see Cases 1, 2, 3 and 4, following.

A jitter-enabled DSIO module supports application of generated jitter to the transmitter master clock. When the bit clock and frame clock are derived from the APx master clock, these are jittered as well. In **Signal Path Setup > Clocks > Jitter Generator**, select **Apply To: Digital Output** and turn jitter **ON**.

For more information see Jitter Support in APx beginning on page 71.

Frequency

Note: in DSIO, sample rate (SR) is referred to as Frame Clock Rate.

Scale Freq. By:

- **Output SR**
  For most testing, use **Output SR** (the default).
- **Fixed**
  If you are applying jitter to the output clock at high levels, you may want to reference the sampled audio data to a fixed rate, rather than the jittered output rate. This may improve phase and frequency stability of the embedded audio. Set **Scale Freq By:** to **Fixed**, and enter the nominal output rate in the **Fixed Rate** field.

See More About Frequency Scaling in APx on page 72 for a detailed explanation.

Bit Clock Edge Sync

These choices select whether the leading edge of the frame (and data) are synchronized with the rising edge or the falling edge of the bit clock, for DSIO receiver inputs and outputs.

Outs

- **Rising Edge**
  For clocks that are set to OUT, synchronization is at the rising edge of the clock signal.
- **Falling Edge**
  For clocks that are set to **OUT**, synchronization is at the falling edge of the clock signal.

Ins

- **Rising Edge**
  For clocks that are set to **IN**, synchronization is at the rising edge of the clock signal.
- **Falling** Edge
  For clocks that are set to IN, synchronization is at the falling edge of the clock signal.

Logic: Level

This sets the nominal voltage level for the DSIO clock and data signals at the HD-15 receiver connections. Choose the level compatible with your DUT. Choices are 1.8 V, 2.5 V and 3.3 V. The logic level for the monitor connections is fixed at 3.3 V.

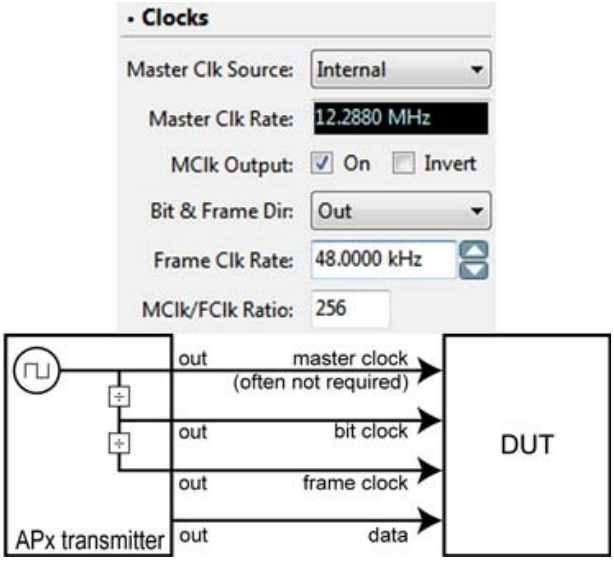
Logic: Outputs

This is a “kill switch” that, when **OFF**, removes all output voltages from the DSIO output connector pins. This switch is duplicated on the Signal Path Setup dialog.

Case 1: Master Clock Internal, Bit & Frame Out

This topic describes settings for Digital Serial Output **Clocks** controls when configured for Case 1.

The APx provides the master clock and the bit and frame clock dividers. The APx sends master clock, bit clock, frame clock and data. In some DUT configurations, a master clock output is not required.



Transmitter Settings: Master Clk Internal, Bit & Frame Out

Master Clock Source: Internal

The bit and frame clocks are generated internally from the APx master clock. In this configuration, the APx DSIO transmitter is the master, and the DUT is slaved to the APx.

Master Clk Rate

This field displays the rate of the Master Clock signal. Master Clock rate is the product of the Sample Rate multiplied by the Ratio value. Minimum is 4 kHz, maximum is 56 MHz.

MClk Output

When this box is checked, the master clock signal appears on the DSIO transmitter output HD-15 connector, pin 1.

Invert

This checkbox is shown if **MClk Output** is checked. Checking the **Invert** checkbox inverts the master clock output signal. When not inverted, the rising edge of the bit clock is synchronized with the rising edge of the master clock. When inverted, the rising edge of the bit clock is synchronized with the falling edge of the master clock

Bit & Frame Clock Dir: Out

The bit and frame clocks are divided internally from the APx master clock. In this configuration, the DUT is slaved to the APx DSIO transmitter.

Frame Clk Rate

Set the frame clock rate (audio data sample rate) here. Minimum is 4 kHz, maximum is 432 kHz. Frame Clock Rate is constrained by channel count and word width.

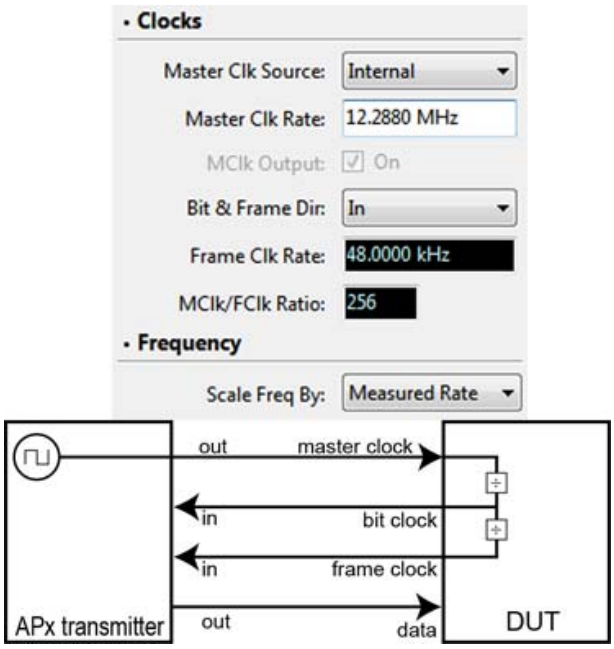
MClk/FClk Ratio

The Ratio is one of two factors that determine the Master Clock Rate, the other being the Frame Clock Rate (the sample rate). Your DUT will accept a specified range of Master Clock frequencies. Once you have chosen your Frame Clock Rate (sample rate), enter a Ratio here to set the Master Clock Rate to an acceptable frequency.

Case 2: Master Clock Internal, Bit & Frame In

This topic describes settings for Digital Serial Output **Clocks** controls when configured for Case 2.

This is a hybrid configuration, where the data flows from the slaved device to the master device (as indicated by bit and frame clock arrows in opposite direction from the data arrow in this diagram). Hybrid configurations are more likely to suffer from errors due to longer propagation delay.



Transmitter Settings: Master Clk Internal, Bit & Frame In

Master Clock Source: Internal

The Master Clock is generated within the APx and is output to the DUT.

Master Clk Rate

Set the Master Clock rate here. Minimum Master Clock Rate is 4 kHz; maximum Master Clock rate is 56 MHz.

MClk Output

MClk Output is forced **On** in this configuration. The master clock signal appears on the DSIO transmitter output HD-15 connector, pin 1.

Bit & Frame Clock Dir: In

The bit and frame clocks are divided from the APx Master Clock externally in the DUT. In this configuration, the APx transmitter is slaved to the DUT.

Frame Clk Rate

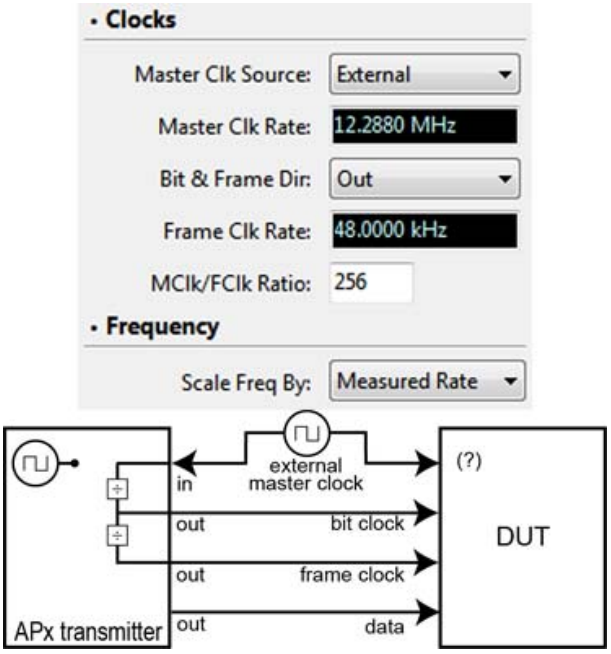
This field displays the rate of the Frame Clock signal, as provided by the DUT.

MClk/FClk Ratio

This field displays the MClk/FClk Ratio.

Case 3: Master Clock External, Bit & Frame Out

This topic describes settings for Digital Serial Output **Clocks** controls when configured for Case 3.



Transmitter Settings: Master Clk External, Bit & Frame Out

Master Clock Source: External

The APx DSIO Master Clock is not used. The DUT provides a Master Clock signal.

Master Clk Rate

This field displays the Master Clock Rate as provided by the DUT

Bit & Frame Clock Dir: In

The bit and frame clocks are divided internally from the external master clock. In this configuration, the APx DSIO transmitter is slaved to the external master clock (typically the DUT).

Frame Clk Rate

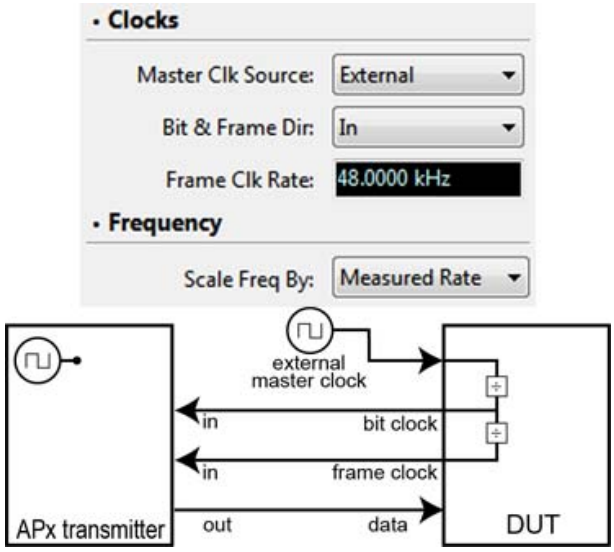
This field displays the rate of the Frame Clock signal.

MClk/FClk Ratio

The Ratio determines the Frame Clock Rate, dividing the Master Clock signal provided by the DUT, by the value entered here.

Case 4: Master Clock External, Bit & Frame In

This topic describes settings for Digital Serial Output **Clocks** controls when configured for Case 4.



Transmitter Settings: Master Clk External, Bit & Frame In

Master Clock Source: External

The APx DSIO Master Clock is not used.

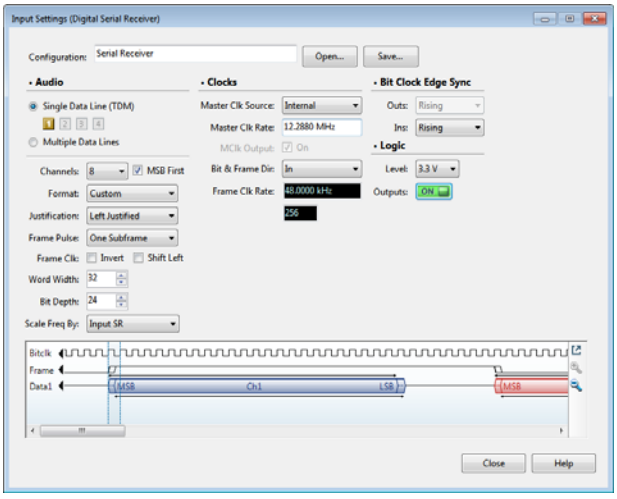
Bit & Frame Clock Dir: In

The bit and frame clocks are provided by the DUT. In this configuration, the APx transmitter is slaved to the DUT.

Frame Clk Rate

This field displays the rate of the Frame Clock signal received from the DUT

Input Settings for the Digital Serial Receiver



This dialog allows you to set, save or load digital serial receiver configurations. A dynamic timing diagram provides an on-screen display of your configuration settings. Receiver connections are provided on an HD-15 connector, with Master Clock, Frame Clock, Bit Clock, Channel Clock and four Data line connections. A moni-

tor connector provides buffered outputs of these signals for oscilloscope monitoring.

The channel clock is provided as an aid for users monitoring on an oscilloscope. The channel clock signal is aligned with the data and has a rising edge with each channel change.

The DSIO Receiver Frame Clock is available for arbitrary use as a reference clock, when Input Configuration is not DSIO. See page 38.

See illustrations of typical Digital Serial Input/Output configurations beginning on page 114.

Common Settings for Digital Serial Input

Most of the settings on this panel are common to all Digital Serial Input configurations, and are discussed first.

The settings in the receiver Clocks area vary with clock configuration, and the four possible cases are discussed following these common settings.

Timing Diagram

The timing diagram at the bottom of the panel is dynamic, changing to illustrate your current settings. Use the zoom controls to contract or expand the horizontal axis of the display, or click the **Undock** button to display a large view.

Configuration

Type a name for your input configuration here.

Open

Click **Open** to view the Open Digital Serial Receiver Settings dialog, where you can load saved *.srx configuration files.

Save

Click **Save** to view the Save Digital Serial Receiver Settings dialog, where you can save the current configuration as an *.srx file.

Audio

Data Lines

- **Single (TDM)** uses time division multiplexing to read the audio data in a single data stream. Channels are read from the data stream sequentially, beginning with channel 1, followed by a channel 2 followed by channel 3, and so on. Use the **Data Line** buttons to select the input data line.
- **Multiple** uses 1, 2 or 4 separate data lines, each received on individual data line conductors in the physical layer. Each of the multiple data lines carries 2 channels of audio data.

Channels

- When configured for a single data line (TDM), the Digital Serial Input can receive 1, 2, 4, 6, 8 or 16 audio data channels. Select the number of channels from the drop-down menu.
- When configured for multiple data lines, the Digital Serial Input can receive 1, 2, 4, 6, or 8 audio data channels. Select the number of channels from the drop-down menu.

MSB First

The active audio data is a string of bits, whose number is the Bit Depth. When **MSB First** is set in the receiver, the MSB (Most Significant Bit) is assumed to lead the string. When **MSB First** is not set, the LSB (Least Significant Bit) is assumed to lead the string.

Format

Set the receiver format here.

- **I2S** (spoken “I-squared-S”) is an acronym for the Philips “inter IC sound” bus. I^2S is a common audio serial interface format. **I2S** sets the input to receive I^2S formatted signals, where the Frame Clock low for the first channel (left or “1”) and high for the second channel (right or “2”). If there are more channels, the Frame Clock stays high until the beginning of the next frame.

  I^2S data is offset by one bit clock period from the beginning of each word.
- **DSP**
  In the DSP (digital signal processing) format, the frame clock pulse is the same length as one bit clock period. Like I^2S, DSP offsets the data by one bit clock period from the beginning of each word. Choose this setting to configure the receiver for the DSP format.
- **Custom**
  The Custom selection allows you to configure the receiver for Justification and Frame Pulse in configurations not defined by I^2S or DSP formats.

Justification

- **Left Justified**
  Left Justified configures the receiver for active bits fully to the left (leading) edge of the data word.
- **Right Justified**
  Right Justified configures the receiver for active bits fully to the right (trailing) edge of the data word.

Frame Pulse

When **Format** is **Custom**, you can set the receiver for the width of the **Frame Clock Pulse** here.

- **One Bit Clock**
  The Frame Clock pulse is one Bit Clock bit wide.

- **One Subframe**
  The Frame Clock pulse is one subframe wide. The subframe is the width of one data word. A subframe is a frame divided by the number of channels on that data line.
- **50% Duty Cycle**
  The Frame Clock pulse is set to a 50% duty cycle, which makes the width of the Frame Clock pulse 1/2 the width of the frame.

Invert Frame Clk

Check this box to set the receiver for inverted polarity of the Frame Clock.

Word Width

The word width is the width, in bits, of one data word, or subframe. A subframe is a frame divided by the number of channels. Set the received word width here. Maximum is 128 bits; minimum is 8 bits. Word width cannot be less than bit depth.

Bit Depth

Set the received bit depth (audio data length) here. Maximum is 32 bits; minimum is 8 bits. When the **Bit Depth** is set to the same value of the **Word Width**, the entire bit allocation for the channel carries audio data. When **Bit Depth** is less than **Word Width**, only some of the bits carry data. The remaining bits are called pads or padding bits.

Scale Freq By:

Note: in DSIO, sample rate (SR) is referred to as Frame Clock Rate.

- **Input SR**
  For most testing, use **Input SR** (the default).
- **Output SR**
  If sample rate is being swept, you may want to use the output sampling rate as the scaling reference.
- **Fixed**
  If jitter is present on the input bit clock, you may want to reference the sampled audio data to a fixed rate, rather than the jittered input rate. This may improve phase and frequency stability of the embedded audio. Set **Scale Freq By:** to **Fixed**, and enter the nominal input sample rate in the **Fixed Rate** field.

See More About Frequency Scaling in APx on page 72 for a detailed explanation.

Bit Clock Edge Sync

These selects whether the leading edge of the frame (and data) are synchronized with the rising edge or the falling edge of the bit clock, for DSIO receiver inputs and outputs.

Outs

- **Rising Edge**
  For clocks that are set to OUT, synchronization is at the rising edge of the clock signal.
- **Falling Edge**
  For clocks that are set to **OUT**, synchronization is at the falling edge of the clock signal.

Ins

- **Rising Edge**
  For clocks that are set to **IN**, synchronization is at the rising edge of the clock signal.
- **Falling** Edge
  For clocks that are set to IN, synchronization is at the falling edge of the clock signal.

Logic: Level

This sets the nominal voltage level for the DSIO clock and data signals at the HD-15 receiver connections. Choose the level compatible with your DUT. Choices are 1.8 V, 2.5 V and 3.3 V. The logic level for the monitor connections is fixed at 3.3 V.

Logic: Outputs

This is a “kill switch” that, when **OFF**, removes all output voltages from the DSIO input connector pins. This switch is duplicated on the Signal Path Setup dialog.

Clocks

The visibility and availability of controls and readings in the receiver **Clocks** area depend upon the settings of **Master Clock Source** and **Bit & Frame Clock Direction**. The explanations here are segmented into the four possible combinations of these settings.

Jitter

A jitter-enabled DSIO module supports application of generated jitter to the receiver master clock. When the bit clock and frame clock are derived from the APx master clock, these are jittered as well. In **Signal Path Setup > Clocks > Jitter Generator**, select **Apply To: Digital Input** and turn jitter **ON**.

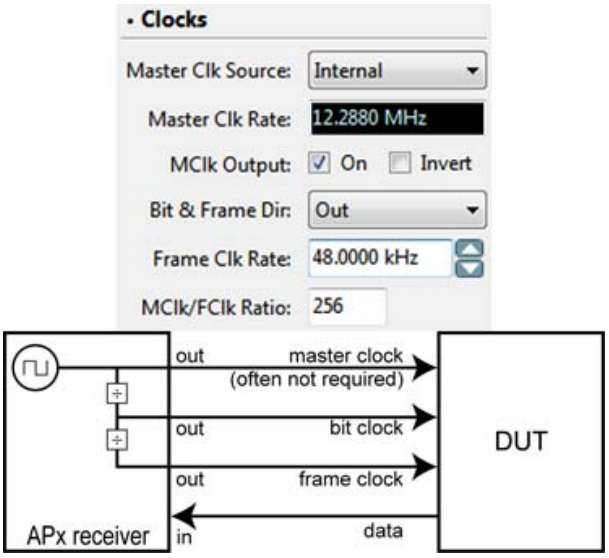
Alternatively, incoming jitter can be measured when the DSIO receiver **Bit and Frame Clk Dir:** is set to **In**. In **Signal Path Setup > Input/Output** and **Input Connector** is set to **Digital Serial**, select a **Jitter** choice from the **Measure** menu.

For more information see Jitter Support in APx beginning on page 71.

Case 1: Master Clock Internal, Bit & Frame Out

The APx provides the master clock and the bit and frame clock dividers. The DUT sends data; the APx send master clock, bit clock and frame clock. In some DUT configurations, only the bit clock and the frame clock are required.

This is a hybrid configuration, where the data flows from the slaved device to the master device (as indicated by bit and frame clock arrows in opposite direction from the data arrow in this diagram). Hybrid configurations are more likely to suffer from errors due to longer propagation delay.



Receiver Settings: Master Clk Internal, Bit & Frame Out

Master Clock Source: Internal

The Bit and Frame Clocks are generated internally from the APx Master Clock. In this configuration, the APx DSIO receiver is the master, and the DUT is slaved to the APx.

Master Clk Rate

This field displays the rate of the Master Clock signal. Master Clock rate is the product of the Sample Rate multiplied by the Ratio value. Minimum is 4 kHz; maximum is 56 MHz.

MClk Output

When this box is checked, the Master Clock signal appears on the DSIO receiver output HD-15 connector, pin 1. Since the DUT is receiving Bit Clock and Frame Clock, the Master Clock signal is often not required by the DUT.

Invert

This checkbox is shown if MClk Output is checked. Checking the Invert checkbox inverts the master clock output signal. When not inverted, the rising edge of the bit clock is synchronized with the rising edge of the master clock. When inverted, the rising edge of the bit clock is synchronized with the falling edge of the master clock.

Bit & Frame Clock Dir: Out

The bit and frame clocks are divided internally from the APx master clock and provided to the DUT.

Frame Clk Rate

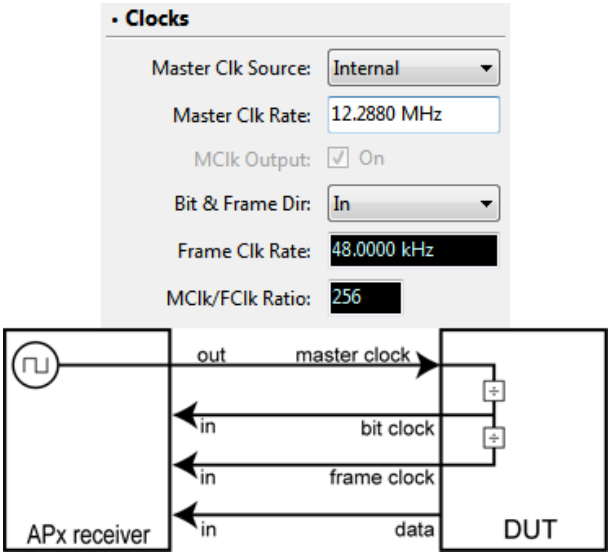
Set the Frame Clock Rate (audio data sample rate) here. Minimum is 4 kHz; maximum is 432 kHz. Frame Clock Rate is constrained by channel count and word width.

MClk/FClk Ratio

The Ratio is one of two factors that determine the Master Clock Rate, the other being the Frame Clock Rate (the sample rate). Your DUT will accept a specified range of Master Clock frequencies. Once you have chosen your Frame Clock Rate (sample rate), enter a Ratio here to set the Master Clock Rate to an acceptable frequency.

Case 2: Master Clock Internal, Bit & Frame In

The DUT provides the bit and frame clock dividers, but has no master clock. APx sends master clock; the DUT sends bit clock, frame clock and data.



Receiver Settings: Master Clk Internal, Bit & Frame In

Master Clock Source: Internal

The Master Clock is generated within the APx and is output to the DUT.

Master Clk Rate

Set the Master Clock Rate here. Minimum Master Clock Rate is 4 kHz; maximum Master Clock Rate is 56 MHz.

MClk Output

MClk Output is forced On in this configuration. The Master Clock signal appears on the DSIO receiver HD-15 connector, pin 1.

Bit & Frame Clock Dir: In

The bit and frame clocks are divided from the APx Master Clock externally in the DUT. In this configuration, the APx receiver is slaved to the DUT.

Frame Clk Rate

This field displays the rate of the Frame Clock signal, as provided from the DUT.

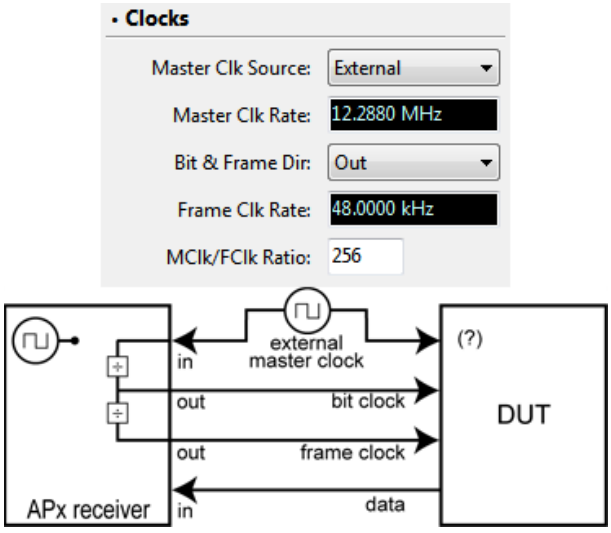
MClk/FClk Ratio

This field displays the MClk/FClk Ratio.

Case 3: Master Clock External, Bit & Frame Out

The DUT provides the master clock, and uses the bit and frame clock dividers in the APx. The DUT may or may not use the master clock (indicated by "?"), which does not affect the APx configuration. The DUT sends master clock and data; the APx sends bit clock and frame clock.

This is a hybrid configuration, where the data flows from the slaved device to the master device (as indicated by bit and frame clock arrows in opposite direction from the data arrow in this diagram). Hybrid configurations are more likely to suffer from errors due to longer propagation delay.



Receiver Settings: Master Clk External, Bit & Frame Out

Master Clock Source: External

The APx DSIO Master Clock is not used.

Master Clk Rate

This field displays the rate of the Master Clock signal provided by the DUT.

Bit & Frame Clock Dir: Out

The bit and frame clocks are divided internally from the external master clock. In this configuration, the DUT is slaved to the APx DSIO receiver.

Frame Clk Rate

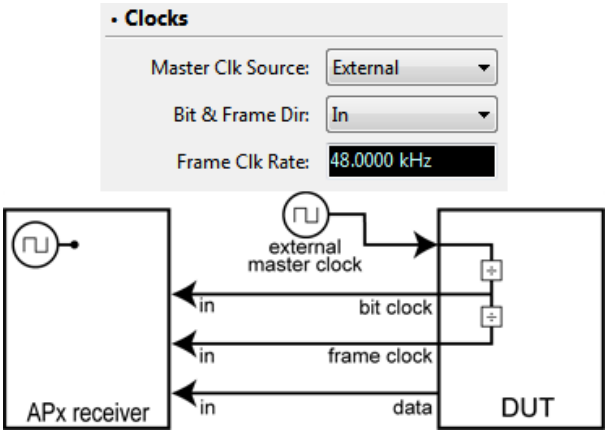
This field displays the rate of the Frame Clock signal.

MClk/FClk Ratio

The ratio is a factor in determining the rate of the Frame Clock signal. Enter a Ratio here to divide the external master clock to an acceptable Frame Clock Rate.

Case 4: Master Clock External, Bit & Frame In

The DUT provides the master clock and the bit and frame clock dividers. The DUT sends bit clock, frame clock and data.



Receiver Settings: Master Clk External, Bit & Frame In

Master Clock Source: External

The APx DSIO Master Clock is not used.

Bit & Frame Clock Dir: In

The bit and frame clocks are provided by the DUT. In this configuration, the APx transmitter is slaved to the DUT.

Frame Clk Rate

This field displays the rate of the Frame Clock signal.

HDMI+ARC

Introduction

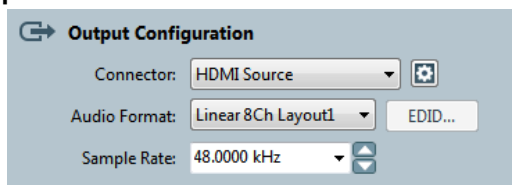
HDMI (High Definition Multimedia Interface) is an interface designed to carry high-bandwidth digital streams providing an audio/video interface that includes content protection and a bi-directional channel for interaction with connected electronic devices.

ARC (Audio Return Channel) provides an additional digital audio channel to the HDMI interconnection that can simplify interface cabling in typical applications, for user convenience. HDMI+ARC is a current hardware option available for APx525, 582 and 585 families of analyzers.

APx Output and Input configurations and settings for HDMI and ARC are independent selections. Configuration and settings documentation for audio over HDMI begins here; documentation for ARC begins on page 123.

HDMI Source Output Configuration

Output: HDMI Source



The HDMI Source selection is only available for an APx analyzer fitted with the HDMI option (model 112) or HDMI+ARC (model 114 or 214) option.

This selection embeds the output audio in the HDMI stream, available on the "SOURCE" type A HDMI connector on the instrument front panel. When HDMI Source is selected, the Audio Format and Sample Rate controls are available. The **EDID** button is available when EDID data from a downstream device is present at the AUX OUT HDMI connector.

Settings

The **Settings** button opens the HDMI output **Settings** dialog. Settings for HDMI are quite extensive. See Output Settings for HDMI Source on page 118.

Audio Format

Choose the HDMI audio format here.

- **Layout 0 LPCM 2ch**
  Layout 0 is only 2 channel, but can output any supported sample rate in any supported video format.
- **Layout 1 LPCM 8ch**
  Layout 1 is 8 channel. Supported sample rates are constrained by selected video format. See Settings to choose video format and for information about supported sample rates.

EDID

This button is only available when there is EDID information at the HDMI SOURCE connector. It opens the EDID Viewer, which displays a property grid that shows the EDID data from the downstream device at the SOURCE HDMI connector. The EDID data can be viewed, saved to a file and reloaded using the EDID Editor (see page 122), for use as the APx instrument EDID sent to an upstream device at the SINK HDMI connector.

Sample Rate (duplicated on the Settings dialog)

To set the output sample rate (Fs) for the embedded digital audio, click the up/down arrows to select a standard rate, or enter an arbitrary value in the **Sample Rate** field. Minimum setting is 30.7 kHz; maximum is 192 kHz. Default is 48 kHz. Supported sample rates are constrained to certain ranges by the selected video format. See **Settings** to choose video format and for information about supported sample rates.

Settings for HDMI are quite extensive. See Output Settings for HDMI Source on page 118.

HDMI Sink Input Configuration

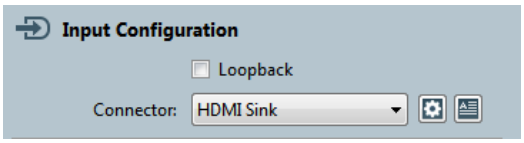
In the Input Configuration panel, you select the number of input channels operating for your test and the connectors you will be using. You can choose references and names for your test input channels.

Loopback

Loopback is only available when Output Configuration is set to a digital output.

When the Loopback checkbox is set, the digital input is disconnected from the front panel input connector and is instead routed to the digital output circuits. This function is called GenMon in other Audio Precision products. Loopback allows you to view the signal present at the digital output on the analyzer meters or graphs.

Input: HDMI Sink



The HDMI Sink selection is only available for an APx analyzer fitted with the HDMI option (model 112) or HDMI+ARC (model 114) option.

HDMI Sink de-embeds the input audio in the HDMI stream presented at the SINK connector.

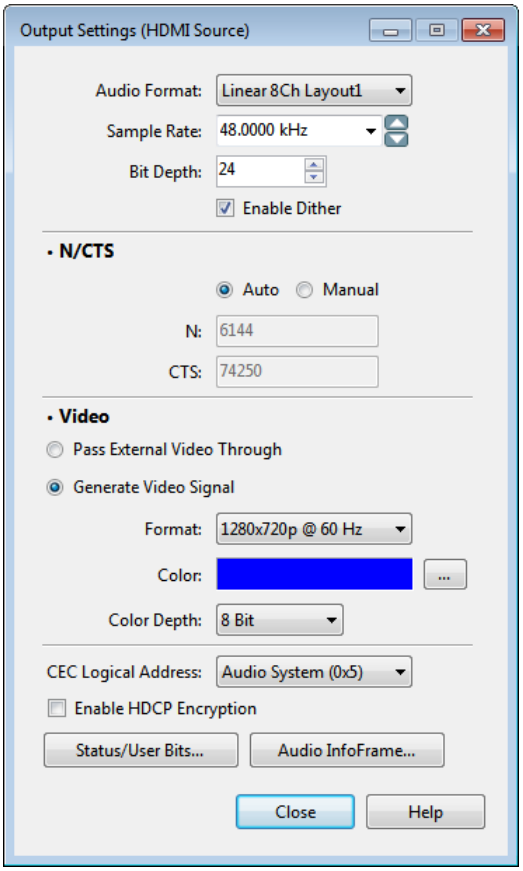
Settings

The **Settings** button opens the HDMI input **Settings** dialog. Settings for HDMI are quite extensive. See Input Settings for HDMI Sink on page 120.

Operation with absent or corrupt interface signal

When Input Configuration is set to a digital input and no signal is present, or if the signal is corrupt or out of range, the input receiver cannot lock (synchronize) to the interface signal, and no valid audio can be recovered. The sample rate indicator in the Status Bar will display an “unlocked” warning, and any measurement result referencing audio from the unlocked input will display an “----” (invalid) result. See page 636 for more about invalid results.

Output Settings for HDMI Source



Audio Format

Audio Format appears on both Signal Path Setup and here in the Settings dialog. Choose the HDMI audio format here.

Layout 0 LPCM 2ch

Layout 0 is only 2 channel, but can output any supported sample rate in any supported video format.

Layout 1 LPCM 8ch

Layout 1 is 8 channel. Supported sample rates are constrained by selected video format. See Video Settings > Format, below, to choose video format and for information about supported sample rates.

Sample Rate

To set the output sample rate (Fs) for the embedded digital audio, click the up/down arrows to select a standard rate, or enter an arbitrary value in the **Sample Rate** field. Minimum setting is 30.7 kHz; maximum is 192 kHz. Default is 48 kHz. Supported sample rates are constrained to certain ranges by the selected video format. See Settings to choose video format and for information about supported sample rates. See **Video Settings: Format** below.

Bit Depth

Bit Depth has the same function for HDMI signal generation as it does in the other digital output modes. Return to Advanced Settings for Signal Path Setup to see this information.

Enable Dither

Dither is enabled by default, and should remain enabled for all ordinary audio signals. Dither is disabled for signals that must be bit-accurate, such as Walking Zeros, Walking Ones, Constant Value and Bit test Random.

N/CTS

At the HDMI transmitter, the audio sample is rate embodied in the relationship between the values of N (the video clock) and CTS (the cycle time stamp, sent as metadata). The downstream HDMI receiver uses N/CTS to re-create the audio clock.

Auto

In **Auto**, N/CTS follows the HDMI standard for all supported video resolutions and audio rates.

Manual

In **Manual**, you are allowed to enter non-standard values as impairments to test the robustness of downstream devices.

Video Settings

HDMI is a video interface with embedded audio. The video content for the APx HDMI generator output can be from either an external video signal brought in at the HDMI AUX IN connector, or from an internal video generator (the default). Select the video source from these choices:

Pass External Video Through

When Pass External Video Through is selected, the video signal at the HDMI AUX IN connector is passed to the HDMI SOURCE connector. Any audio on the external HDMI signal is stripped off and replaced by audio generated in APx500. The APx generator audio format is constrained by the video format of the external signal. The HDCP encryption state is passed through unchanged.

Generate Video Signal (the default)

When Generate Video Signal is selected, the video content for the HDMI SOURCE is generated within the Audio Precision HDMI module. Format, color, color depth and HDCP encryption of this signal are set in the next controls.

- **Format:**
  One of 14 video formats can be set here, ranging from 640x480p @ 60 Hz to 1920x1080p @ 24 Hz. Note that only high-definition video formats support multi-channel high-sample-rate embedded audio. Audio Format and Video Format selections will constrain HDMI channel count and sample rate.

Refer to the chart of HDMI channel count and sample rates by audio and video formats on page 122.

- **Color:**
  Select video content color here. Default color is a blue screen. Click the browser button to select other colors.
- **Color Depth:**
  Select video content color depth here. Choices are 8 bit, or Deep Color modes 10 bit or 12 bit.

HDCP Encryption

For internally-generated video, HDCP encryption may be enabled or disabled by clicking the checkbox.

For externally-provided video, the HDCP encryption state is passed through to the downstream device unchanged. The HDCP Encryption checkbox is unavailable in this mode.

About HDCP encryption

- HDCP (High-bandwidth Digital Content Protection) can be used to protect content transmitted on an HDMI interface.
- When enabled, non-HDCP-compliant downstream devices will not be able to establish a connection. A blank screen or error message may be displayed.

Status/User Bits

This button opens a dialog where you can set the Status Bits and User Bits embedded in the audio carried in the HDMI stream. See More About Metadata on page 388.

Audio InfoFrame

This button opens a dialog where you can set the Audio InfoFrame metadata carried in the HDMI stream. For information about reading Audio InfoFrame data, see page 35.

Audio streams in HDMI contain metadata in the form of status bits, user bits, and for coded audio, burst information. The HDMI Audio InfoFrame contains additional information about the audio stream.

Setting Audio InfoFrame

When Output Configuration is set to HDMI Source, click Settings and then Audio InfoFrame.

Auto

When **Auto** is selected, APx500 sets the HDMI Audio InfoFrame to default values.

When **Auto** is not selected, Audio InfoFrame fields can be set to any available value.

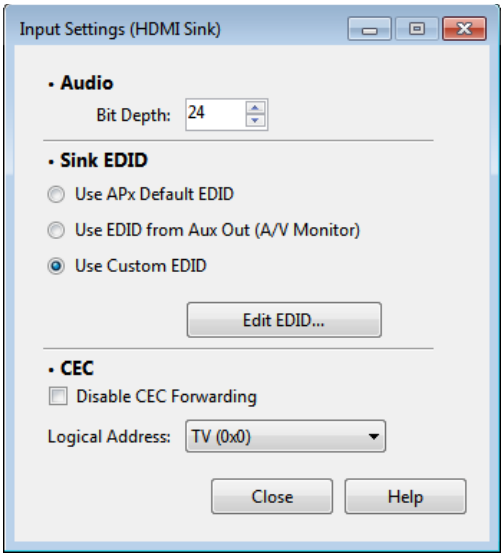
Plain Text display and Hex display

Audio InfoFrame fields can be set from the grid display, reading the plain text interpretations and choosing settings from drop-down menus in the grid. Audio InfoFrame data can also be set by entering hex values for selected bytes in the Audio InfoFrame hex display. These two displays are coupled, and a change in one will be reflected by the corresponding change in the other.

Refer to HDMI.org and the standards document CEA-861-DB for detailed information about Audio InfoFrame and the HDMI transport stream.

Read more about Audio InfoFrame data on page 390.

Input Settings for HDMI Sink



Disable CEC Forwarding

By default, CEC commands from devices at the HDMI Sink or AUX OUT connectors are forwarded between these devices. Select Disable CEC Forwarding to prevent this communication.

About CEC

- CEC (Consumer Electronics Control) is an optional HDMI protocol that uses a dedicated bidirectional bus to communicate consumer control commands (such as volume or channel change) between HDMI-connected devices. Read more about CEC on page 126.

SINK EDID

About EDID and DDC

- APx500 supports the Enhanced Display Data Channel (E-DDC, or simply DDC), which is used by the HDMI source device to read the E-EDID data from the HDMI sink device to learn what audio/video formats it supports. APx supports EDID and E-EDID for HDMI.
- EDID (Extended Display Identification Data) and E-EDID (Enhanced Extended Display Identification Data) are data stored in downstream (sync) devices and communicated to upstream (source) devices.
- EDID contains information about the video and audio capabilities of the downstream device, so that the source device can format the content properly. This means, for example, that a downstream 720x480p video monitor with only two audio channels will cause an upstream surround decoder or Blu-ray player to provide a 720x480p video signal with a stereo downmix, when connected to this device.

EDID 1.4

- With APx500 v 3.1, EDID for HDMI 1.4 is supported, in addition to the previously supported EDID 1.3. Note that in EDID 1.4, some of the EDID bytes are interpreted differently than in EDID 1.3. For more information, see the following standard, available at vesa.org.
- VESA ENHANCED EXTENDED DISPLAY IDENTIFICATION DATA STANDARD (Defines EDID Structure Version 1, Revision 4).

Use APx Default EDID

When this button is set, the APx default EDID is sent to the upstream device connected at the HDMI SINK connector. The APx instrument is configured with default EDID data that will produce expected results for many DUT / instrument / monitor setups. You can view these default settings by using the Edit EDID button, below.

Use EDID from AUX OUT (monitor)

By default, EDID read from the downstream display device on AUX OUT is NOT forwarded to the upstream device connected to SINK. This configuration allows you to analyze the full high-definition audio embedded in the SINK signal without constraining it to the limits of the display device.

To constrain the upstream device on SINK to the capabilities of the monitor connected to AUX OUT, check the Copy EDID box.

Use Custom EDID

In some cases you may want to send custom EDID data to the upstream device. Open the EDID editor by

clicking the Edit EDID button. Read more about the EDID Editor starting at page 122.

Embedded Audio

Bit Depth

The Bit Depth control allows you to set the input bit depth (digital word length). The default setting is 24 bits, which is also the maximum setting. Minimum setting is 8 bits. The input signal is truncated (not dithered down) if Bit Depth is set to a lower value than Bit Depth of the signal.

We recommend that you use the default setting of 24 bits, unless you have a special application that requires that you mask out low-order bits.

Coupling

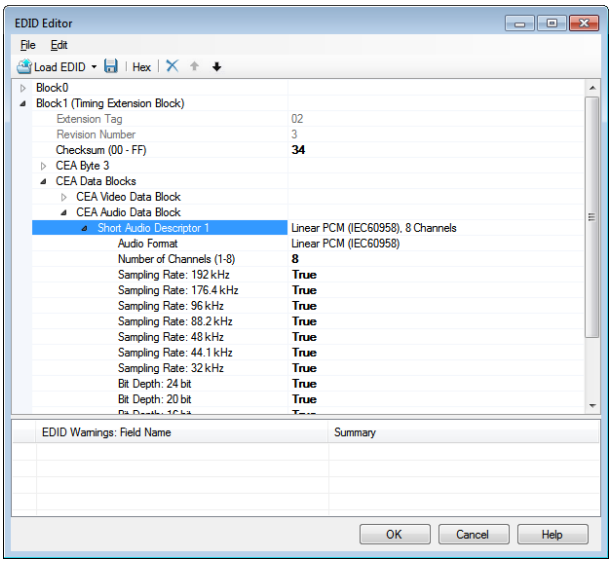
Input signals can be analyzed and displayed in one of two coupling modes:

- **AC**
  AC coupling blocks any dc signal component from analysis, displaying results for only the ac components. AC coupling is the default setting.
- **DC**
  DC coupling includes the dc signal component in analysis and display.

Audio Format	*Video Format*	*Channel Count*	*Sample Rates*
Linear 2Ch Layout 0	any	2	30.7 kHz to 33.3 kHz 42.3 kHz to 45.9 kHz 46.0 kHz to 50.0 kHz 84.6 kHz to 91.8 kHz 92.0 kHz to 100.0 kHz 169.2 kHz to 183.6 kHz 184.0 kHz to 192.0 kHz
Linear 8Ch Layout 1	640x720p @ 60 Hz 720x480p @ 60 Hz 720x576p @ 50 Hz	8	30.7 kHz to 33.3 kHz 42.3 kHz to 45.9 kHz 46.0 kHz to 50.0 kHz
Linear 8Ch Layout 1	720x240p @ 60 Hz 720x288p @ 50 Hz 720x480i @ 60 Hz 720x576i @ 50 Hz	8	30.7 kHz to 33.3 kHz 42.3 kHz to 45.9 kHz 46.0 kHz to 50.0 kHz 84.6 kHz to 91.8 kHz
Linear 8Ch Layout 1	All 1280x720 and 1920x1080 formats	8 (Note: Although the HDMI interface can carry 8 channels at all these sample rates, Blu-ray Discs are limited to 6 channels at 192 kHz sample rate.)	30.7 kHz to 33.3 kHz 42.3 kHz to 45.9 kHz 46.0 kHz to 50.0 kHz 84.6 kHz to 91.8 kHz 92.0 kHz to 100.0 kHz 169.2 kHz to 183.6 kHz 184.0 kHz to 192.0 kHz

HDMI audio constraints by format

The EDID Viewer and the EDID Editor



The EDID Editor.
The EDID Viewer displays information in the same way.

The EDID Viewer

The APx500 EDID Viewer is a property grid that displays the EDID settings received from the downstream device connected to the instrument HDMI SOURCE connector. These properties cannot be edited, since they come from the downstream DUT, but they can be viewed and saved as a *.edid data file. The EDID Viewer is available from the Output Configuration panel when Digital HDMI is selected and a downstream device that transmits EDID is connected. Click **EDID**.

The EDID Editor

The EDID Editor is a property grid that displays the EDID settings to be sent to the upstream device connected to the instrument HDMI SINK connector. These properties can be viewed, edited and saved as a data file. The EDID Editor is available from the Input Settings (Digital HDMI) panel. Click **Edit EDID**.

Using the EDID Editor

Navigate through the grid to the setting of interest, and enter the value or select the option required. Click **OK** to accept the changes you have made.

EDID Audio settings are stored in **Block1 > CEA Data Blocks > CEA Audio Data Block > Short Audio Descriptor n**. By default, APx500 has 2 audio settings, one for 8 channel PCM and one for 2 channel PCM, listed in **Short Audio Descriptor 1** and **Short Audio Descriptor 2**. These settings inform the upstream device that the APx500 SYNC will only accept 8 or 2 channel PCM, at the sample rates and bit depths indicated in the descriptors.

Changing an audio setting

To change an audio setting, you can either add a new audio descriptor or edit one of the existing descriptors.

To add a descriptor, right-click on the **CEA Audio Data Block field. Choose Add CEA Extension Data > Short Audio Descriptor**. Edit the new descriptor to define the new format.

Note: the standard limits the amount of data that can be carried in the EDID. You can add new descriptors, but you will not be allowed to add many without deleting other information.

To edit a descriptor, in the left column select the field to be changed. Select or enter a new value in the right column for that field.

Note: When a defining field such as Audio Format is changed, other fields within the descriptor may be affected. Note that when selecting AC-3 (see below), the Bit Depth fields are replaced with a Maximum Bit Rate field.

Example: changing PCM to Dolby Digital 5.1

Under **Short Audio Descriptor 1**, select the **Audio Format** field in the left column. Click the arrow in the right column, and choose **AC-3** from the drop-down menu. Now change **Number of Channels** to **6**, change all the sample rates except 48kHz to **False**, and set **Maximum Bit Rate** to **2040**.

Hex

The Hex button enables a display of the EDID data in hexadecimal notation. You may have to enlarge the window to view all the data.

Saving an EDID file

Using the File menu commands or the Toolbar buttons, you can save your changes to a proprietary *.edid file, or to a Microsoft Excel file (*.xls) or a text file (*.txt).

Loading EDID

When you first open the EDID Editor in a New Project, the EDID data are set to the APx default.

Using the File menu commands or the Toolbar buttons, you can load EDID settings from an existing *.edid or *.xls file, or from the device at the HDMI **SOURCE** or **AUX OUT** connectors.

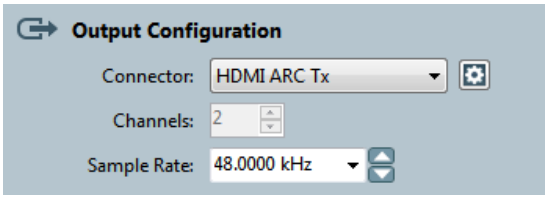
EDID Warnings

Some EDID settings will allow undefined values to be entered and passed with the EDID data. When an undefined value is entered, a warning will appear in the EDID Warning list.

ARC Output Configuration

See page 126 for more about ARC.

Output: HDMI ARC Tx



The ARC Tx selection is only available for an APx analyzer fitted with the HDMI+ARC (model 114) option.

Introduction

HDMI ARC Tx sends digital audio in the IEC 60958 / IEC 61937 (S/PDIF) interface format via the HDMI ARC Tx / AUX IN connector, in accordance with the HEAC ARC (audio return channel) definition in the HDMI 1.4a specification. In this configuration the APx represents a TV, returning audio down an HDMI cable to a home theater receiver Source.

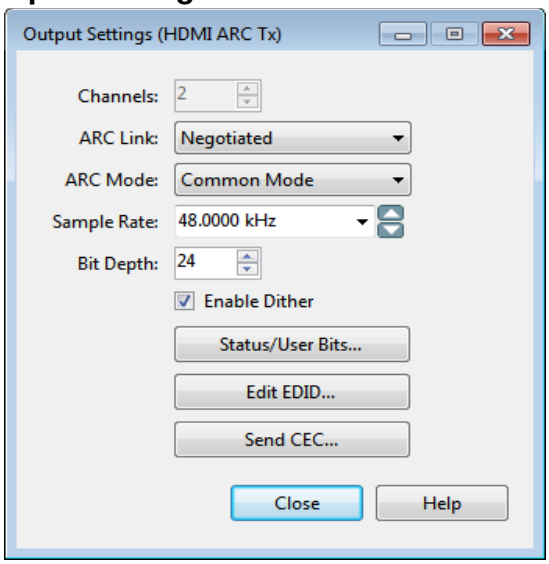
Sample Rate (duplicated on the Settings dialog)

To set the output sample rate (Fs), click the up/down arrows to select a standard rate, or enter an arbitrary value in the **Sample Rate** field. Minimum setting is 8 kHz; maximum is 216 kHz. Default is 48 kHz. **Sample Rate** is also available in the Settings dialog.

Settings

Click the **Settings** button to open the **Output Settings for ARC Tx** dialog.

Output Settings for ARC Tx



ARC Link

- **Negotiated**
  Typically, ARC is negotiated between two devices, using EDID identification and exchange of CEC messages. If this negotiation fails, an ARC link cannot be established. **Negotiated** is the APx default.
- **Audio Forced On**
  This mode ignores all negotiation, and simply streams digital audio data on the ARC connectors, regardless of EDID status or exchange of CEC messages.

ARC Mode

- **Common Mode**
  In Common Mode, the ARC signal is sent commonly on the HEAC+ and HEAC- lines, relative to ground.

- **Single Mode**
  In Single Mode, the ARC signal is sent on the HEAC+ line, relative to ground.

Sample Rate

To set the output sample rate (Fs), click the up/down arrows to select a standard rate, or enter an arbitrary value in the **Sample Rate** field. Minimum setting is 8 kHz; maximum is 216 kHz. Default is 48 kHz. **Sample Rate** is also available in the Settings dialog.

Bit Depth

The **Bit Depth** control allows you to set the output bit depth (also called digital word length). The default setting is 24 bits, which is also the maximum setting. Minimum setting is 8 bits.

Enable Dither

When the **Enable Dither** checkbox is checked, dither is **ON** (the default). When **ON**, dither is set according to the bit depth, using TPDF (triangular probability density function) dither at ±1 LSB.

Uncheck the checkbox to turn dither **OFF**.

Status/User Bits...

The ARC digital audio stream includes IEC 60958 Status Bits and User Bits. This button opens the Set Status Bits / User Bits panel for the ARC audio stream. See page 611.

Edit EDID...

EDID is Extended Display Identification Data, one of several metadata protocols in the HDMI transport stream.

Click the **EDID...** button to open the EDID Editor. See page 122.

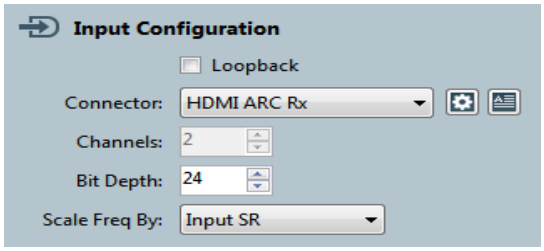
Send CEC...

Establishing, confirming or terminating an ARC link requires exchange of CEC messages. Click **Send CEC...** to open a dialog box. See page 126.

ARC Input Configuration

See page 126 for ore about ARC.

Input: HDMI ARC Rx



The ARC Rx selection is only available for an APx analyzer fitted with the HDMI+ARC (model 114) option.

Introduction

HDMI ARC Rx receives digital audio in the IEC 60958 / IEC 61937 (S/PDIF) interface format via the ARC Rx / HDMI AUX OUT connector, in accordance with the HEAC ARC (audio return channel) definition in the HDMI 1.4a specification. In this configuration the APx represents a home theater receiver, accepting audio sent down an HDMI cable from a TV Sink.

Labels/Colors

Click the **Labels/Colors...** button to open the Labels/Colors dialog, where you can name the input channels with names you choose and change the channel color assignments used in the display and reports. Names and color assignments are maintained independently for the each input configuration selection.

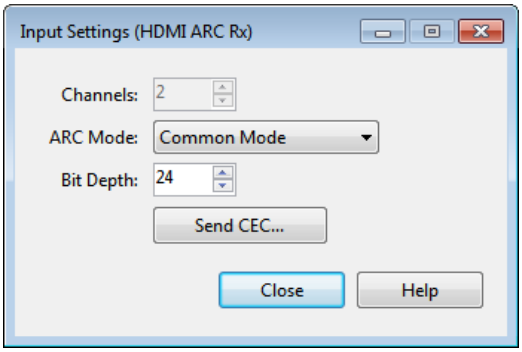
Operation with absent or corrupt interface signal

When Input Configuration is set to a digital input and no signal is present, or if the signal is corrupt or out of range, the input receiver cannot lock (synchronize) to the interface signal, and no valid audio can be recovered. The sample rate indicator in the Status Bar will display an “unlocked” warning, and any measurement result referencing audio from the unlocked input will display an “----” (invalid) result. See page 636 for more about invalid results.

Settings

Click the **Settings** button to open the **Input Settings for ARC Rx** dialog.

Input Settings for ARC Rx



ARC Mode

- **Common Mode**
  In Common Mode, the ARC signal is sent commonly on the HEAC+ and HEAC- lines, relative to ground. This setting configures the receiver for Common Mode.
- **Single Mode**
  In Single Mode, the ARC signal is sent on the HEAC+ line, relative to ground. This setting configures the receiver for Single Mode.

See page 126 for more about the ARC physical layer.

Bit Depth

The Bit Depth control allows you to set the input bit depth (digital word length). The default setting is 24 bits, which is also the maximum setting. Minimum setting is 8 bits. The input signal is truncated (not dithered down) if Bit Depth is set to a lower value than Bit Depth of the signal.

We recommend that you use the default setting of 24 bits, unless you have a special application that requires that you mask out low-order bits.

Scale Freq By:

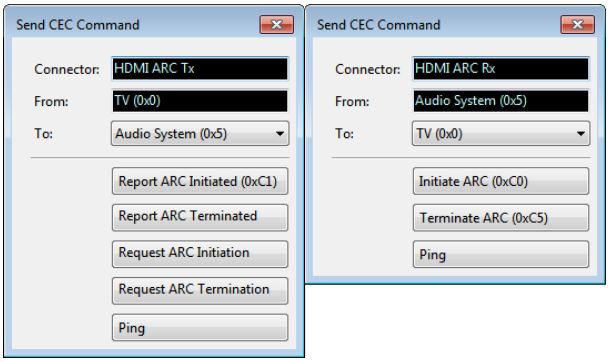
- **Input SR**
  For most testing, use Input SR (the default).
- **Output SR**
  If sample rate is being swept, you may want to use the output sampling rate as the scaling reference.
- **Fixed**
  If jitter is present on the input signal, you may want to reference the embedded audio to a fixed rate, rather than the jittered input rate. This may improve phase and frequency stability of the embedded audio. Set **Scale Freq By:** to **Fixed**, and enter the nominal input sample rate in the **Fixed Rate** field. See More About Frequency Scaling on page 72.

Send CEC...

Establishing, confirming or terminating an ARC link requires exchange of CEC messages.

Click **Send CEC...** to open a dialog box.

Send CEC dialog boxes



When in an HDMI ARC configuration, the Send CEC dialog allows you to send ARC-related CEC commands to the source or sink device connected.

In normal operation, EDID settings and CEC messages are exchanged in a link negotiation process, and use of the commands in this dialog is unnecessary. These are provided for diagnostic purposes.

Tx and Rx dialogs

If the **Send CEC Message** dialog has been opened from Output Settings (HDMI ARC Tx), the role of the APx is "TV," from which CEC commands can be sent to the connected device, defined as "Audio System."

Available CEC messages (shown with their hex equivalents) are:

- Report ARC Initiated (0xC1)
- Report ARC Terminated (0xC2)
- Request ARC Initiation (0xC3)
- Request ARC Termination (0xC4)

If the **Send CEC Message** dialog has been opened from Input Settings (HDMI ARC Rx), the role of the APx is "Audio System," from which CEC commands can be sent to the connected device, defined as "TV."

Available CEC messages (shown with their hex equivalents) are:

- Initiate ARC (0xC0)
- Terminate ARC (0xC5)

More About ARC

APx support of HDMI versions and features

APx HDMI modules fully support HDMI version 1.3a. The APx HDMI+ARC module adds ARC (Audio Return Channel) support, which is a subset of the HDMI 1.4 specification. Other HDMI features added in 1.4a are not supported. In the APx implementation, ARC Tx (transmit) and ARC Rx (receive) input and output selections are entirely independent of the standard HDMI audio configuration and features. The ARC Tx signal appears on the connector also used for HDMI AUX IN, and the ARC Rx uses the connector also used for HDMI AUX OUT.

About ARC

ARC (Audio Return Channel) is a feature of HDMI 1.4a that enables digital audio (IEC 60958 / SPDIF) transmission and reception on existing but previously unused conductors in standard HDMI connectors and cables.

ARC was added to eliminate the need for a coaxial or optical digital cable to return audio from a device designated as an HDMI sink. The classic case is a television (typically connected as a sink to a home theater receiver acting as a video source) that has been configured to receive video directly from an RF antenna, rather than from the receiver. The audio from the on-air signal must be routed to the receiver; in the past this required an additional cable. The HDMI ARC function designates existing conductors in the HDMI cable to act as a digital audio return channel, presenting the audio to the source connector on the receiver.

The ARC carrier and embedded audio formats conform to IEC60958 (AES3-S/PDIF).

Physical Layer

The ARC digital audio signal is carried on a twisted pair of wires in an HDMI cable. These wires are designated HEAC+ and HEAC–. The ARC signal can be carried in two modes. In Single Mode, the digital audio signal is carried on the HEAC+ conductor, relative to the HDMI ground. In Common Mode, the ARC signal is carried commonly on the HEAC+ and HEAC– conductors, relative to ground. Common mode facilitates an HDMI 1.4a feature not supported by APx, carrying an ethernet signal simultaneously in differential mode on the HEAC– and HEAC+ conductors.

CEC

In APx, CEC commands are sent using the HD Monitor panel, CEC tab. See page 35.

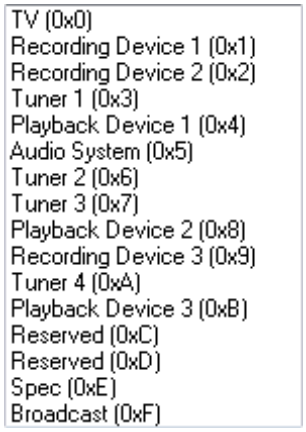
CEC is Consumer Electronics Control, a bus in the HDMI interface. CEC enables HDMI interconnected devices such as a television, a cable set-top box, an audio-video receiver, a disc player, etc. to identify each other and exchange control commands.

These commands are used to simplify operating an audio-video system for the consumer. Selecting the DVD input on a receiver, for example, may also turn the DVD player ON. One IR remote control may be configured to control several different devices.

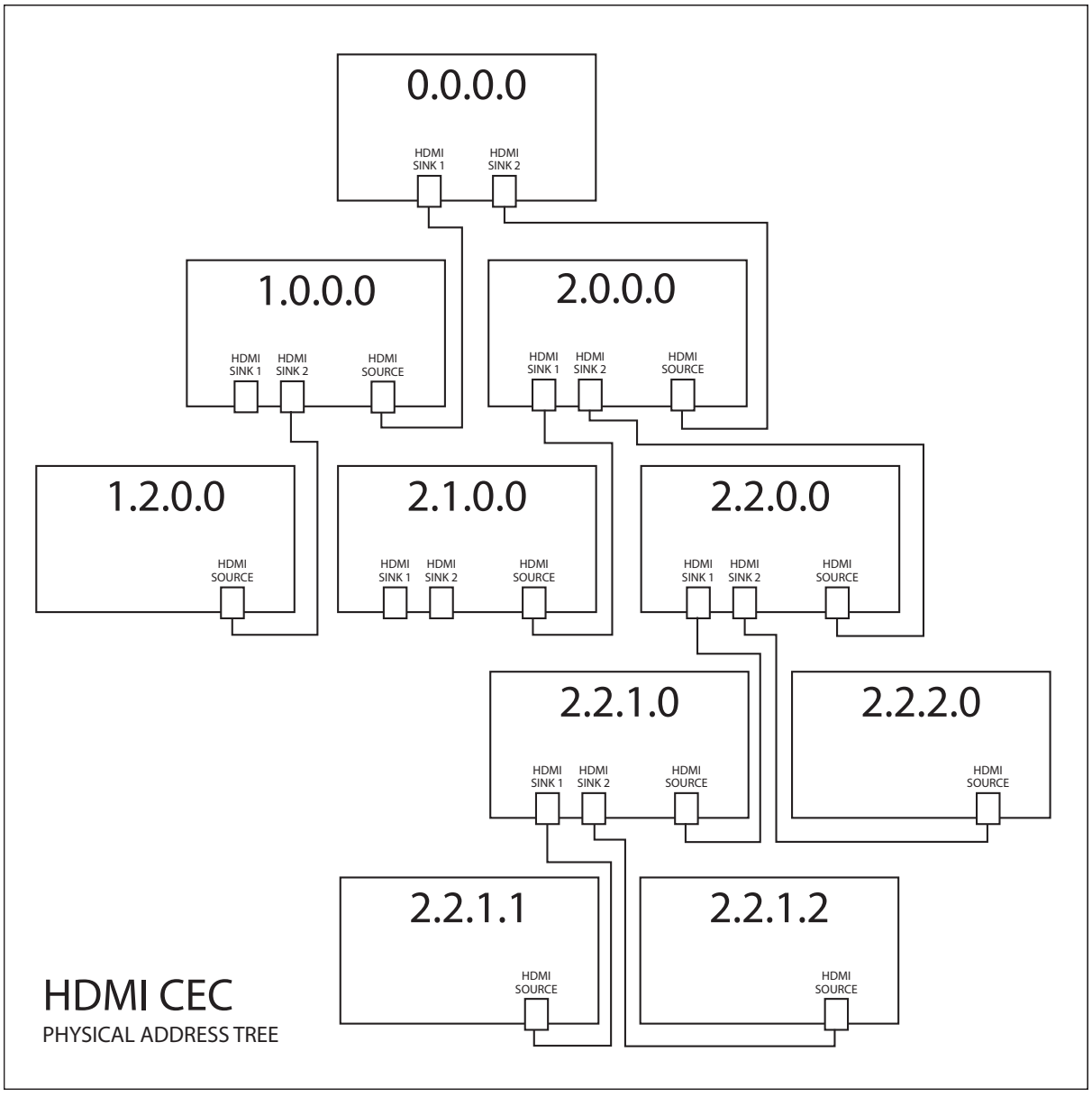
The CEC system assumes the video display or television is the “root” device, and the CEC network expands from this root in a tree pattern, with switchers (e.g. a receiver) as “branches” and source devices as “leaves.” The tree can only be 5 layers deep. Up to 10 devices can be connected to a CEC bus.

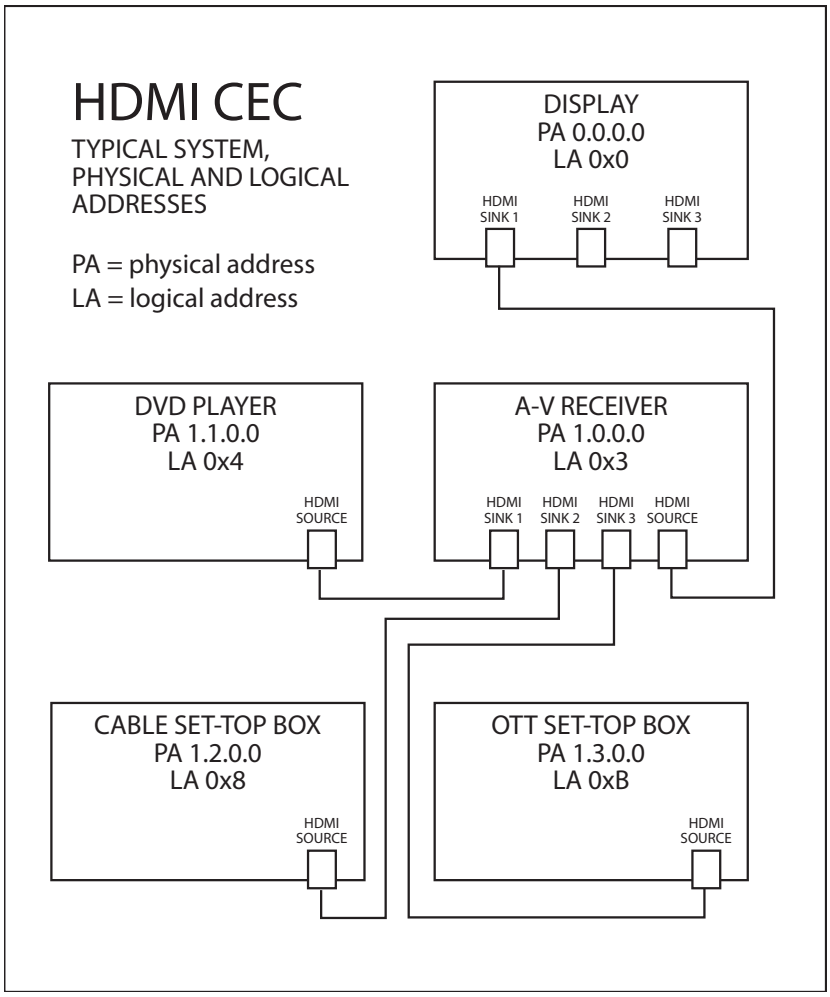
Each device on the tree has a physical address and a logical address. The physical address for the root device is 0.0.0.0. Other devices inherit physical addresses according to the connector and the device each is connected to.

Devices also have logical addresses, corresponding with their roles in the system. There are 16 CEC logical addresses defined. Each device negotiates for a logical address when it is connected to the bus.



CEC Device menu, showing logical addresses.


0.0.0.0
HDMI SINK 1
HDMI SINK 2
1.0.0.0
HDMI SINK 1
HDMI SINK 2
HDMI SOURCE
2.0.0.0
HDMI SINK 1
HDMI SINK 2
HDMI SOURCE
1.2.0.0
HDMI SOURCE
2.1.0.0
HDMI SINK 1
HDMI SINK 2
HDMI SOURCE
2.2.0.0
HDMI SINK 1
HDMI SINK 2
HDMI SOURCE
2.2.1.0
HDMI SINK 1
HDMI SINK 2
HDMI SOURCE
2.2.2.0
HDMI SOURCE
2.2.1.1
HDMI SOURCE
2.2.1.2
HDMI SOURCE
HDMI CEC
PHYSICAL ADDRESS TREE



Detailed information about CEC and HDMI is available at www.hdmi.org.

17

Bluetooth I/O

Introduction

The APx Bluetooth Duo option (page 9) is a hardware module available for the APx modular analyzers (the APx555, and the APx525 and 585 families of analyzers, including the APx582). It provides *Bluetooth*® wireless technology source and sink interfaces. See licensing acknowledgements on the copyright page of this manual (immediately following the title page).

The APx Bluetooth profiles support audio in a lower-quality (voice) mode (the HFP and HSP profiles) and in a higher-quality (music) mode (the A2DP profile).

See more About Bluetooth on page 142 and Using Bluetooth in a Sequence on page 557. See More About Supported Bluetooth Profiles on page 143.

This chapter discusses operation of the Bluetooth Duo option.APx500 version 4.5 supports the Bluetooth Duo option, and also provides support for the legacy Bluetooth modules. Operation of the legacy Bluetooth modules is discussed in Chapter 18.

Identification

The most obvious difference between the legacy Bluetooth modules and the Bluetooth Duo module is the number of external antennas: the legacy modules have one; Bluetooth Duo has two.

The Bluetooth Duo module carries the model number 229, and is identified in the Help > About product components list as Bluetooth Duo Module.

See more About Bluetooth on page 142, and Using Bluetooth in a Sequence on page 557. See More About Supported Bluetooth Profiles on page 143.

The Bluetooth Monitor

The status of a number of current Bluetooth settings is shown in the Bluetooth Monitor. Additionally, Bluetooth actions, settings and utility functions are available through a context menu (right-click in the Bluetooth Monitor display.) See Bluetooth Monitor on page 36.

Bluetooth Discovery, Pairing and Connection

Bluetooth connections require successful radio frequency (RF) communication between the APx analyzer and the DUT, and successful device handshaking through device discoverability, pairing and connection protocols. Data transfer is encrypted and must be authorized by the exchange of link keys, and personal identification number (PIN) codes may be required.

Bluetooth Settings are Global

Bluetooth settings are unique in APx500, in that they are *global*. This means the settings made in the Bluetooth Settings dialog affect the entire project: all measurements, all signal paths.

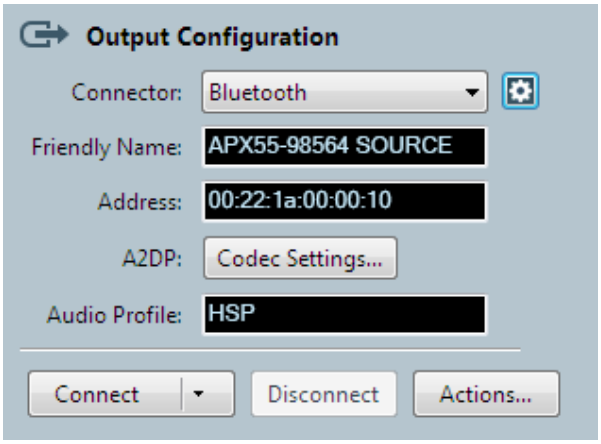
Source, Sink, Output and Input

When the APx **Output Configuration** is set to **Bluetooth**, the instrument can transmit audio using the A2DP Source profile, the HFP Audio Gateway profile, or the HSP Audio Gateway profile. When the APx **Input Configuration** is set to **Bluetooth** input, the instrument can receive audio using the A2DP Sink profile, the HFP Hands Free profile, or the HSP Headset profile.

There is one case where you are can output audio and input audio simultaneously: set the APx Output and APx Input to Bluetooth, and connect to the DUT using the HFP Profile. When the SCO is opened, you can both send and receive audio, since HFP is a bi-directional link.

Output Configuration: Bluetooth

This Signal Path Setup I/O selection enables output of audio from the APx generator to a Bluetooth device under test, using Bluetooth radio frequency (RF) transmission.



Note: The Bluetooth choice is only available for analyzers fitted with the APx Bluetooth Option module.

Connector

Choose **Bluetooth** here. Since the Bluetooth connection isn't valid until you choose, pair and connect with a device, the Bluetooth Settings dialog opens immediately. See page 132.

Settings

Click the **Settings...** button to discover, select, pair, connect and otherwise configure a Bluetooth link. See page 132.

Friendly Name

The friendly name of the APx Bluetooth module's source chip will appear here.

Address

The address of the APx Bluetooth module's source chip will appear here.

A2DP

The codec settings for the A2DP profile can be viewed and edited here.

Audio Profile

This displays the active profile, if any. See More About Supported Bluetooth Profiles on page 143.

Connect

If the APx is already paired with one or more Bluetooth devices, you can open the **Connect** flyout menu to select a device and one or more profiles for connection. See More About Supported Bluetooth Profiles on page 143.

Disconnect

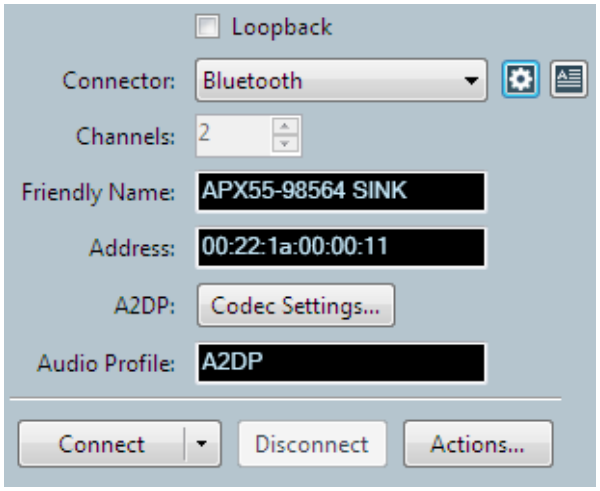
If the APx is already connected to a Bluetooth device, you can click the **Disconnect** button to terminate the connection.

Actions

Click **Actions** to open the **Bluetooth Actions** dialog. If the APx is already connected to a Bluetooth device, you can initiate Bluetooth Actions from this dialog. Only the actions appropriate to the current connection and profile are shown. See Actions on page 140.

Input Configuration: Bluetooth

This selection enables input of audio from a Bluetooth device under test, using Bluetooth radio frequency (RF) transmission.



Note: The Bluetooth choice is only available for analyzers fitted with the APx Bluetooth Option module.

Loopback

Loopback is not available when either the Output or Input Configuration is set to Bluetooth.

Connector

Choose **Bluetooth** here. Since the Bluetooth connection isn't valid until you choose, pair and connect with a device, the Bluetooth Settings dialog opens immediately. See page 132.

Settings

Click the **Settings...** button to discover, select, pair, connect and otherwise configure a link with a Bluetooth device. See page 132.

Edit Channel Labels

In the Channel Labels dialog you can rename the input channels with names of your choice. See page 56. These settings are project-wide, but can be overridden for any measurement result. To set project colors, click the **Project Colors...** button in the Labels dialog, or go to the Project/Sequence Properties dialog, Colors tab. See Setting Graph Colors and Styles on page 30.

Channels

This is a reading field that displays the current input channel count, determined by profile settings.

Friendly Name

The friendly name of the APx Bluetooth module's sink chip will appear here.

Address

The address of the APx Bluetooth module's sink chip will appear here.

A2DP

The codec settings for the A2DP profile can be viewed and edited here.

Audio Profile

This displays the active profile, if any. See More About Supported Bluetooth Profiles on page 143.

Connect

If the APx is already paired with one or more Bluetooth devices, you can open the **Connect** flyout menu to select a device and one or more profiles for connection. See More About Supported Bluetooth Profiles on page 143.

Disconnect

If the APx is already connected to a Bluetooth device, you can click the **Disconnect** button to terminate the connection.

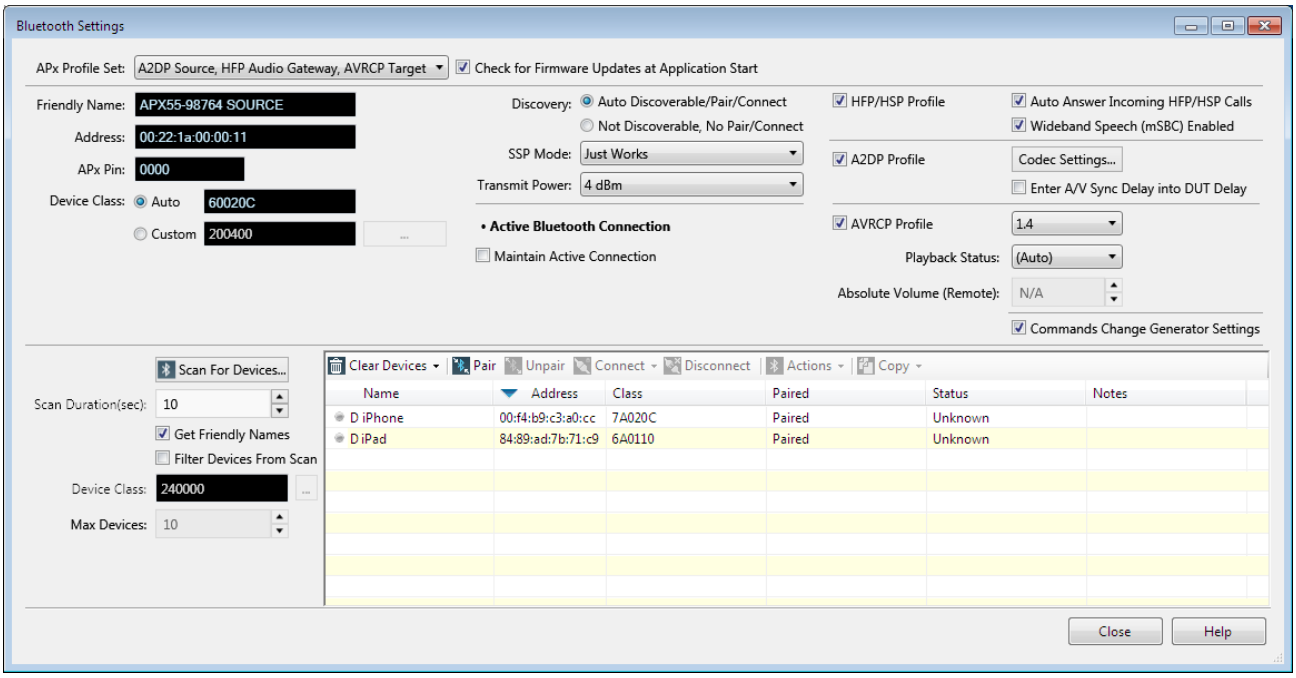
Actions

Click **Actions** to open the **Bluetooth Actions** dialog. If the APx is already connected to a Bluetooth device, you can initiate Bluetooth Actions from this dialog. Only the actions appropriate to the current connection and profile are shown. See Actions on page 140.

Operation with absent or corrupt interface signal

When Input Configuration is set to a digital input and no signal is present, or if the signal is corrupt or out of range, the input receiver cannot lock (synchronize) to the interface signal, and no valid audio can be recovered. The sample rate indicator in the Status Bar will display an "unlocked" warning, and any measurement result referencing audio from the unlocked input will display an "----" (invalid) result. See page 636 for more about invalid results.

Bluetooth settings dialog for Bluetooth Duo



Bluetooth Settings dialog for Bluetooth Duo, shown in the A2DP Source, HFP Audio Gateway, AVRCP Target profile set.

This topic explores Bluetooth Duo settings dialog. The legacy Bluetooth settings dialog is discussed beginning on page 150.

This dialog provides settings for Bluetooth discovery, pairing, connection and other configuration. Unlike most APx settings, Bluetooth profile selection, pairing and connection settings are global to the APx project. Settings made here affect both Bluetooth source and sink configurations across all signal paths in the project.

A flow chart depicting discovery, pairing and connection paths is shown on page 139.

APx Profile Set

APx has four sets of profiles that the Bluetooth Option module can assume. Each set lists the profiles available for connection and/or the profiles available for access by a remote device. Choose an APx Bluetooth Profile Set here.

- A2DP Source, HFP Audio Gateway, AVRCP Target
- A2DP Source, HSP Audio Gateway, AVRCP Target
- A2DP Sink, HFP Hands Free, AVRCP Controller
- A2DP Sink, HSP Headset, AVRCP Controller

In A2DP Source, the APx Bluetooth transmitter mutes the audio when the average signal level falls below –54 dBFS for more than 1 second. We recommend maintaining test levels above –54 dBFS for meaningful results.

Check for Firmware Updates at Application Start

The Bluetooth Duo hardware module can update its firmware in the field, using update files that are included with the APx500 software distribution. See Update Bluetooth Firmware on page 135.

APx Identification

The upper left side of the dialog displays a number of APx Identification settings readable by other Bluetooth devices.

Friendly Name

The friendly name of the APx Bluetooth module's Source or Sink chip is displayed here.

Address

The Address of the APx Bluetooth Source or Sink chip is displayed here.

APx PIN

The APx Bluetooth PIN is displayed here.

Device Class

- **Auto**
  Auto uses the Device Class assigned by default: for the Source chip, 60020C; for the Sink chip, 240408.
- **Custom**
  Custom allows you to select a Device Class for the active APx Bluetooth chip. Use the ... browse button to open the Edit Device Class Dialog (page 137).

Discovery

Auto Discoverable/Pair/Connect

When this is selected (the default), the APx Bluetooth Option is discoverable by remote Bluetooth devices. If the remote device initiates pairing, the APx will pair. If the remote device initiates connection, the APx is available to connect for the profiles checked here.

Not Discoverable, No Pair/Connect

When this is selected, the APx Bluetooth Option is not discoverable by remote devices, and it will not respond to pairing or connection initiated by remote device.

SSP Mode

APx provides four SSP (Secure Simple Pairing) options. The **Just Works** choice is used in Bluetooth device relationships that require little security. The **Numeric Comparison** choices are man-in-the-middle (MITM) interactions for device relationships that require higher security. Choose the SSP Mode that is appropriate for the Bluetooth device you are testing. See More About Secure Simple Pairing on page 144 for detailed information.

- Just Works
- Numeric Comparison, Display Only
- Numeric Comparison, Display+Buttons
- Numeric Comparison, Keyboard

Transmit Power

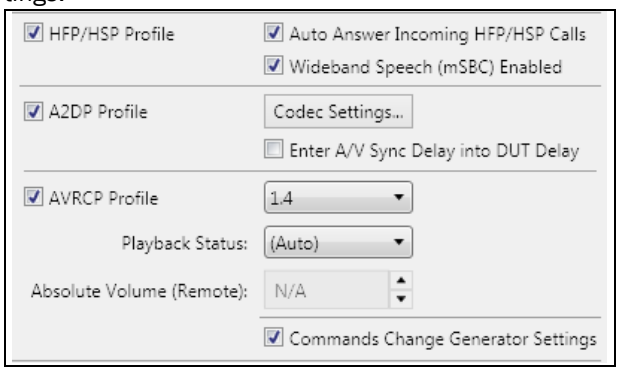
You can choose one of eight RF output transmission power levels between –20 dBm and +8 dBm. The default is +4 dBm. When you change a transmit power setting, the new setting is applied to the NEXT Bluetooth connection.

Active Bluetooth Connection

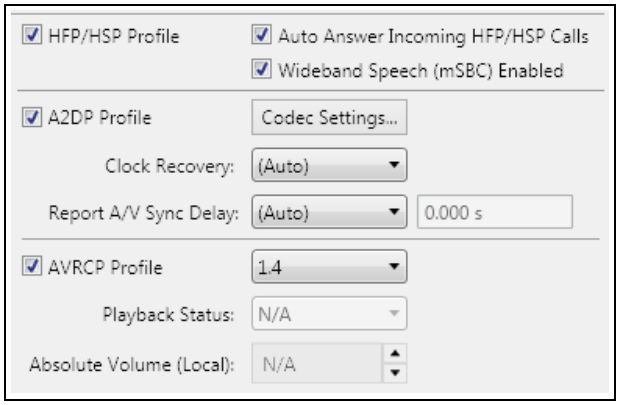
☑ If **Maintain Active Connection** is checked, the APx Bluetooth Option will remain connected to the Bluetooth device under test throughout the project, even across Signal Paths where Bluetooth is not configured as an input or an output. However, if you change to Bench Mode, the Bluetooth device will be disconnected.

Profile Configurations

The upper right side of the dialog displays a number of APx profile configurations. The controls and settings here vary with Source or Sink (Target or Controller) settings.



Profile Configurations for Source/Target settings



Profile Configurations for Sink/Controller settings

HFP/HSP Profile

☑ If this box is checked, the HFP/HSP profile is enabled, and APx support for this profile is disclosed to remote devices. You may elect to use the profile at connection time, or not.

☐ If this box is unchecked, the HFP/HSP profile will not be available on the connection menu, and APx support for this profile is not disclosed to remote devices. These further options are available:

- ☑ Auto Answer Incoming HFP/HSP Calls
- ☑ Wideband Speech (mSBC) Enabled

A2DP Profile (when APx Profile is set to Source)

☑ If this box is checked, the A2DP profile is enabled for Source, and APx support for this profile is disclosed to remote devices. You may elect to use the profile at connection time, or not.

☐ If this box is unchecked, the A2DP profile will not be available on the connection menu, and APx support for this profile is not disclosed to remote devices. These further options are available:

Codec Settings...

This button opens the A2DP Codec Settings dialog. See page 135.

A/V Sync Delay

☐ Enter A/V Sync Delay into DUT Delay. See page 136.

A2DP Profile (when APx Profile is set to Sink)

☑ If this box is checked, the A2DP profile is enabled for Source, and APx support for this profile is disclosed to remote devices. You may elect to use the profile at connection time, or not.

☐ If this box is unchecked, the A2DP profile will not be available on the connection menu, and APx support for this profile is not disclosed to remote devices. These further options are available:

Codec Settings...

This button opens the A2DP Codec Settings dialog. See page 135.

Clock Recovery

Clock Recovery enables or disables the sink clock recovery algorithm. See page 136.

- **Fixed**
  Fixed (the default) sets the A2DP sink sample rate to the negotiated rate.
- **(Auto)**
  Auto enables the A2DP sink clock recovery algorithm.

Report A/V Sync Delay

Report A/V Sync Delay to the connected Source device. See page 136.

- **(Auto)** (the default)
  Auto reports the APx Bluetooth Sink chip A/V Sync Delay.
- **Custom**
  Custom allows you to enter an arbitrary A/V Sync Delay value to be reported.

AVRCP Profile (when APx Profile is set to Target)

☑ If this box is checked, the AVRCP Target profile is enabled, and APx support for this profile is disclosed to remote devices. You may elect to use the profile at connection time, or not.

☐ If this box is unchecked, the AVRCP Target profile will not be available on the connection menu, and APx support for this profile is not disclosed to remote devices.

The latest AVRCP supported by APx is version 1.4. As an alternative (perhaps for diagnostic purposes), you may choose AVRCP version 1.0 here.

These further options are available:

Playback Status

See page 137.

Absolute Volume (Remote)

See page 138.

☑ Commands Change Generator Settings. See page 139.

☑ AVRCP Profile (when APx Profile is set to Controller)

☑ If this box is checked, the AVRCP Target profile is enabled, and APx support for this profile is disclosed to remote devices. You may elect to use the profile at connection time, or not.

☐ If this box is unchecked, the AVRCP Target profile will not be available on the connection menu, and APx support for this profile is not disclosed to remote devices.

The latest AVRCP supported by APx is version 1.4. As an alternative (perhaps for diagnostic purposes), you may choose AVRCP version 1.0 here.

This further option is available:

Absolute Volume (Local)

See page 138.

Scan

The lower left area of the dialog has controls and settings to scan for nearby Bluetooth devices.

Scan for Devices

This button initiates a scan to identify all the discoverable Bluetooth devices within range. Discovered devices are listed in the Current Devices List.

Scan Duration

This setting limits the scan time to the value set. The maximum time is 48 seconds. The default is 10 seconds.

☑ Get Friendly Names

By default, APx asks for a device's friendly name. In automated processes, this process adds extra time. Uncheck this checkbox to disable this feature.

☑ Filter Devices From Scan

You can filter the scan so only a certain Bluetooth Device Class will be shown. Click the browse button to open the Edit Device Class dialog.

Device Class

You can select a Device Class as a filter for the scan. Use the ... browse button to open the Edit Device Class Dialog (page 137).

Max Devices

You can limit the number of devices found in the filtered scan here. The default is 10 devices.

Current Devices List

The Current Devices List occupies most of the lower part of the dialog.

This grid shows the devices discovered in the scan, along with information fields. There are controls to pair, unpair, connect, disconnect, initiate actions and

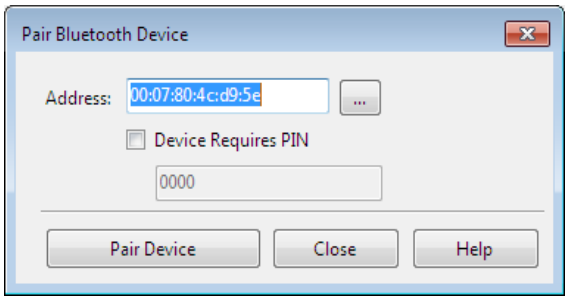
to copy the device Bluetooth address and link key. Select a device to configure the relationship.

Clear Devices

You can choose to remove all devices from the list, or all paired devices, or all unpaired devices. When a device is removed from the list, any pairing information for that device is also removed. Click **Scan** to repopulate the list.

Pair

Pairing provides a way for devices to exchange link keys. To pair with a device, refer to the documentation received with the device. To enable pairing, you may be required to execute a command or series of commands at the device. When pairing is enabled, select the device in the Current Devices List, and click **Pair**. The Pair Bluetooth Device dialog box will open.



The **Browse** ... button in this dialog opens the Known Devices list, from which you may select other devices which have been previously discovered using this computer. See page 142.

Pairing information is saved in the project, until the Current Devices List is cleared, or the until device is Unpaired.

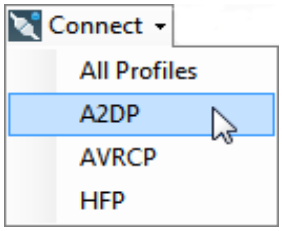
Some devices may require the exchange of a PIN code. The APx Bluetooth PIN code is set to 0000; a field in the pairing dialog allows input of a remote device's PIN code.

Unpair

Unpairs the selected, previously paired device.

Connect

Connects to the selected, previously paired device. A Connect menu opens. Select the profile you would like to offer to the device.



Disconnect

Disconnects the selected, connected device.

Actions

Opens various **Actions** dialogs. This duplicates the function of the Actions button shown on Signal Path Setup, without the necessity of leaving the Bluetooth Settings dialog. See page 140.

Copy

Copies the device address and key link to the Windows Clipboard. This text can then be pasted into another application or tool.

Update Bluetooth Firmware

The Bluetooth Duo hardware module can update its firmware in the field, using update files that are included with the APx500 software distribution.

If **Check for Firmware Updates at Application Start** (located at the top of the Bluetooth Settings panel) is ☑ checked, each time the APx500 application is launched it will check to verify that the Bluetooth firmware is correct for the APx500 version. If the firmware requires updating, the update will begin immediately and a progress dialog will be displayed. The update may take a few minutes.

If **Check for Firmware Updates at Application Start** is ☐ unchecked, the APx500 application will not check for updates at launch. Instead, it will check for updates if and when Bluetooth is selected as an Input or Output. At that time, the update may take a few minutes.

If the Bluetooth firmware is the correct version for the APx500 software, you will not be prompted to update.

Codec Settings

See licensing acknowledgements on the copyright page of this manual (immediately following the title page).

A2DP Codec Settings

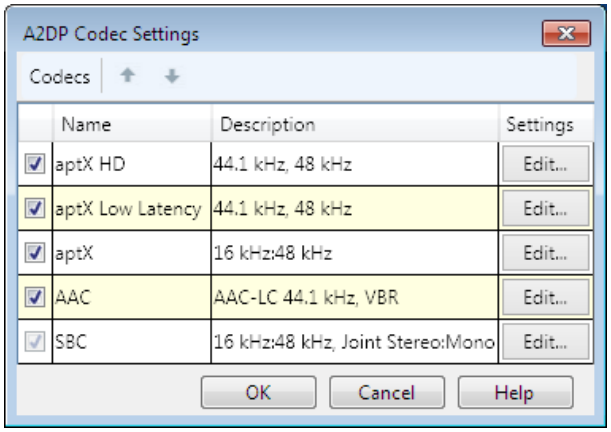
The Bluetooth A2DP profile is required to support the SBC audio codec. Additional codecs are optional.

In APx, you can select which optional codecs you would like to support, and edit features of the codecs you choose.

Using the arrow icons at the top of the dialog, you can set an order of priority

for APx. The DUT will support one or more codes, in its own prioritized list. The device that initiates the A2DP connection will move from the topmost codec downward

until finding the codec that the connected device supports.



Codecs supported, in default order of priority:

aptX HD

You can choose to support either or both of the following sample rates for the aptX HD codec:

- 48 kHz
- 44.1 kHz

aptX Low Latency

You can choose to support either or both of the following sample rates for the aptX Low Latency codec:

- 48 kHz
- 44.1 kHz

aptX

You can choose to support any or all of the following sample rates for the aptX codec:

APx as A2DP source:

- 48 kHz
- 44.1 kHz
- 32 kHz
- 16 kHz

APx as A2DP sink:

- 48 kHz
- 44.1 kHz
- 32 kHz
- 16 kHz

AAC

The AAC codec is fixed at MPEG-2/4, AAC-LC, 44.1 kHz, 2 channels.

By default, variable bit rate (VBR) is set. You can deselect variable bit rate to enforce fixed bit rate.

SBC

You can choose to support any or all of the following sample rates for the SBC codec:

APx as A2DP source:

- 48 kHz
- 44.1 kHz
- 32 kHz
- 16 kHz

APx as A2DP sink:

- 48 kHz
- 44.1 kHz
- 32 kHz
- 16 kHz

Additionally, you can choose to support any or all of the following channel modes:

- Joint Stereo
- Stereo
- Dual Channel
- Mono

Clock Recovery

Bluetooth A2DP Sink Clock Recovery

Clock Recovery: Fixed (the default) does not attempt to algorithmically correct transmission errors, but sets the A2DP sink sample rate at the negotiated rate. This provides stable frequency measurements, but may occasionally be subject to glitches due to packet loss or buffer overruns. Generally, we recommend using the **Fixed** setting, particularly when using the aptX Low Latency codec.

Clock Recovery: (Auto) engages an A2DP sink algorithm that compensates for RF packet loss or sample rate drift between source and sink. Ideally, this compensation is unnoticed by the listener, providing a pleasant listening experience; it will, however, create small but measurable sample rate and audio frequency variations.

A/V Sync Delay

AV Sync Delay features are not supported by the Legacy Bluetooth Option.

This feature of A2DP allows the sink device to notify the source device of any rendering delay, enabling the source device to compensate for the delay.

For example, a source device may provide video to a television monitor via a hard-wired connection, but route the audio through a Bluetooth A2DP connection to speakers or headphones. Any delay in the audio path will cause the audio to lag the video. If the source device is informed of the rendering delay in the sink device, it can delay the video stream by that amount

and maintain proper synchronization between video and audio.

A2DP Source

When A2DP Source is selected, you can choose **Enter A/V Sync Delay into DUT Delay** to apply the delay reported by the remote sink device to the APx DUT Delay feature.

The delay value entered into DUT Delay is updated each time the remote device reports a new delay value.

The Bluetooth Monitor will indicate the delay reported from the remote sink device. If the remote device does not support this feature, the monitor entry will read "N/A".

A2DP Sink

When A2DP Sink is selected, APx provides an A/V Sync Delay value for the remote device to read.

For **Report A/V Sync Delay** you can choose either

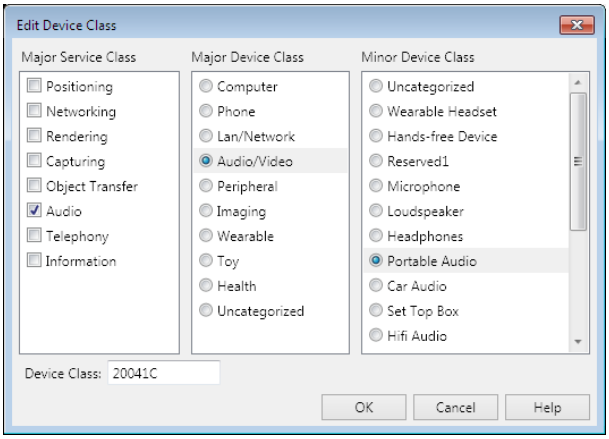
- (**Auto**)
  Auto reports the current delay in the Bluetooth chip internal to the APx, which will vary with configuration and connection.

or

- **Custom**
  Custom allows you to enter an arbitrary delay value for the remote source device to read. The range is 100 ns to 1.6 s.

Edit Device Class

This dialog will generate a Bluetooth Device Class hex code from your selections.



Choose one or more Major Service Class(es), a Major Device Class, and a Minor Device Class (if available.) This dialog will also identify a Device Class if you type a valid hex code in the Device Class field, and press the Enter key on the PC keyboard.

APx Device Class

If you have come to this dialog from the APx Custom Device Class setting, click OK to enter the code generated here as the APx Custom Device Class, which will be asserted in both Source and Sink profiles.

Scan Filter Device Class

If you have come to this dialog from the Filter Devices From Scan setting, click OK to enter the code generated here as the scan filter Device Class.

Note: this feature is not supported by the Legacy Bluetooth Option.

AVRCP Playback Status

AVRCP Playback Status features are not supported by the Legacy Bluetooth Option.

Playback Status is a feature introduced in AVRCP 1.3. This feature is intended to allow devices such as car kits to ask a phone whether a song is playing or not. This prevents certain "out of band" audio noises from the user, such as key presses while dialing a phone number. The car kit will suppress audio from the A2DP source device (also the AVRCP target) when the play status is not PLAYING.

AVRCP Profile selection

The AVRCP Profile checkbox allows you to select AVRCP version 1.4 or version 1.0, or to disable AVRCP altogether. AVRCP version 1.4 is selected as the default. Toggling the AVRCP Profile or version will disconnect any active connection. We recommend that you use AVRCP version 1.4; version 1.0 is available primarily as a diagnostic tool.

AVRCP Controller profile

In the AVRCP Controller profile, APx acts as a A2DP rendering device (a sink such as a speaker or headphone), receiving an audio stream from a remote media player such as a phone. APx uses AVRCP Commands to control the remote media player.

When AVRCP is acting as a Controller, Playback Status is enabled when AVRCP Profile is enabled and set to version 1.4. Playback Status is disabled when AVRCP Profile is unchecked, or when AVRCP Profile is set to version 1.0.

When AVRCP is acting as a Controller, and Playback Status is enabled, and the connected AVRCP Target device supports this notification, the Playback Status is indicated in the Bluetooth Monitor. If the connected AVRCP Target device does not support it,

the Bluetooth Monitor will indicate "N/A".

AVRCP Target profile

In the AVRCP Target profile, APx acts as an A2DP source device, sending an audio stream to a remote rendering device (speaker or headphones). APx responds to AVRCP

Commands sent from the remote rendering device.

When AVRCP is acting as a Target, Playback Status is enabled when AVRCP Profile is enabled and set to version 1.4. Playback Status is disabled when AVRCP Profile is unchecked, or when AVRCP Profile is set to version 1.0.

When AVRCP is acting as a Target, and Playback Status is enabled, you can navigate to **Actions > A2DP Sink/AVRCP > Playback Status** to open a dialog to set a response to the remote AVRCP Controller device. The Playback Status control provides a menu that has the following options: **Auto, Stopped, Playing, Paused, Fwd Seek, Rev Seek, Error.** The default selection is **Auto**.

For any choice except Auto, the selected Playback Status will be passed to the AVRCP Controller device.

If Auto is selected, APx will set the status to **Playing** when the generator is turned on. When A2DP streaming is started, the playback status will be **Paused** when the generator is off. When A2DP streaming is stopped, the playback status will be **Stopped**.

The Playback status value is only updated when the active audio stream is A2DP. Playback status changes should not be made when the audio path is SCO.

Absolute Volume

AVRCP Absolute Volume control features are not supported by the Legacy Bluetooth Option.

AVRCP Absolute Volume Control is a feature introduced in AVRCP 1.3. This feature is intended to allow the source device to specify a percentage value of full volume to the rendering device (speaker or headphones). The rendering device responds with a notification of the current absolute volume level.

APx supports features of Absolute Volume Control in both the AVCRP Controller (A2DP sink) and Target (A2DP source) profiles.

AVCRP Controller profile

In the AVRCP Controller profile, APx acts as a A2DP rendering device (a sink such as a speaker or headphone), receiving an audio stream from a remote media player. APx uses AVRCP Commands to control the remote media player.

When the AVRCP Controller profile is selected, the Absolute Volume checkbox on the Bluetooth Settings panel allows you to enable or disable Absolute Volume. This setting is enabled by default. Toggling this setting will disconnect any active connection.

When Absolute Volume is enabled in the AVRCP Controller profile and the remote Target (source) device supports Absolute Volume, you can navigate to Actions > A2DP Sink/AVRCP > Absolute Volume to open a dialog to set an Absolute Volume level value, which can

be read by the remote AVRCP Target device. Values are integers from 0–127. If the Target device does not support Absolute Volume, this control is unavailable.

...in a sequence

When the AVRCP Controller profile is selected, a sequence step is available to set the Absolute Volume level on the A2DP sink device. When the step is included in a sequence and loaded on an APx analyzer that does not support Absolute Volume, an error occurs when the step is executed. When the step is included in a sequence and loaded on an analyzer that supports Absolute Volume, but connected to a remote device that does not support it, an error occurs when the step is executed.

In a sequence when the AVRCP Controller profile is selected, and the remote AVRCP Target device supports Absolute Volume, the Absolute Volume level set in the local A2DP sink device (the APx analyzer) is displayed on the report for each measurement in the sequence.

AVRCP Target profile

In the AVRCP Target profile, APx acts as an A2DP source device, sending an audio stream to a remote rendering device (speaker or headphones). APx responds to AVRCP Commands sent from the remote rendering device.

When the AVRCP Target profile is selected, the Absolute Volume checkbox on the Bluetooth Settings panel allows you to enable or disable Absolute Volume. This setting is enabled by default. Toggling this setting will disconnect any active connection.

When Absolute Volume is enabled in the AVRCP Target profile and the remote AVRCP Controller (sink) device supports Absolute Volume, you can navigate to Actions > A2DP Sink/AVRCP > Absolute Volume to open a dialog to send an Absolute Volume command to the remote AVRCP Controller device. Values are integers from 0-127. If the remote AVRCP Controller device does not support Absolute Volume, this control is unavailable.

When the AVRCP Target profile is selected, and the remote AVRCP Controller device reports its Absolute Volume level, the level is read and displayed in the APx Bluetooth Monitor. If the remote device does not support Absolute Volume control, the Bluetooth Monitor indicates “N/A” for this value.

When the AVRCP Target profile is selected, a nested sweep parameter called “Absolute Volume” is available. This parameter allows the AVRCP Controller (A2DP sink) device manufacturer to test audio output at various levels, plotted on one graph.

...in a sequence

When the AVRCP Target profile is selected, a sequence step is available to set the Absolute Volume level on the remote device. When the step is included in a sequence, and loaded on an APx analyzer which does not support Absolute Volume, an error occurs when the step is executed. When the step is included in a sequence, and loaded on an analyzer that does support Absolute Volume, but the remote device does not support it, an error occurs when the step is executed.

In a sequence when the AVRCP Target profile is selected and the remote AVRCP Controller device supports Absolute Volume, the Absolute Volume level reported by the remote device is read by APx and displayed on the report for each measurement in the sequence.

Commands Change Generator Settings

When APx is configured for A2DP Source AVRCP Target, the APx generator can now respond to certain AVRCP commands.

AVCRP Target profile

In the AVRCP Target profile, APx acts as an A2DP source device, sending an audio stream to a remote rendering device (speaker or headphones). APx responds to AVRCP Commands sent from the remote rendering device.

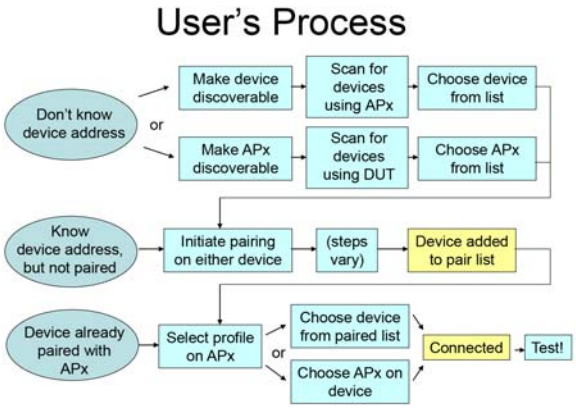
When the AVRCP Target profile is selected, the **Commands Change Generator Settings** ☑ checkbox on the Bluetooth Settings panel allows you to enable or disable this feature. This setting is enabled by default.

AVRCP generator commands are only acted upon when the active measurement has a generator with an ON/OFF button, and the generator is not disabled.

Generator response to AVRCP Commands

- The AVRCP command **Play** turns the generator **On**
- The AVRCP command **Pause** turns the Generator **Off**
- The AVRCP command **Stop** turns the Generator **Off**, and **stops** A2DP streaming.
- The AVRCP command **Next Track** increases the generator **Frequency** equivalent to pressing the **up** arrow on the frequency control in the panel
- The AVRCP command **Prev Track** decreases the generator **Frequency** equivalent to pressing the **down** arrow on the frequency control in the panel.
- The AVRCP command **Volume Up** increases the generator **Level** on **Ch1** equivalent to pressing the **up** arrow on the level control in the panel. **Volume Up** messages are **not** received in legacy Bluetooth modules.
- The AVRCP command **Volume Down** decreases the generator **Level** on **Ch1** equivalent to pressing the **down** arrow on the level control in the panel. **Volume Down** messages are **not** received in legacy Bluetooth modules.

Process for discovery, pairing and connection



Steps in connecting APx to a Bluetooth device

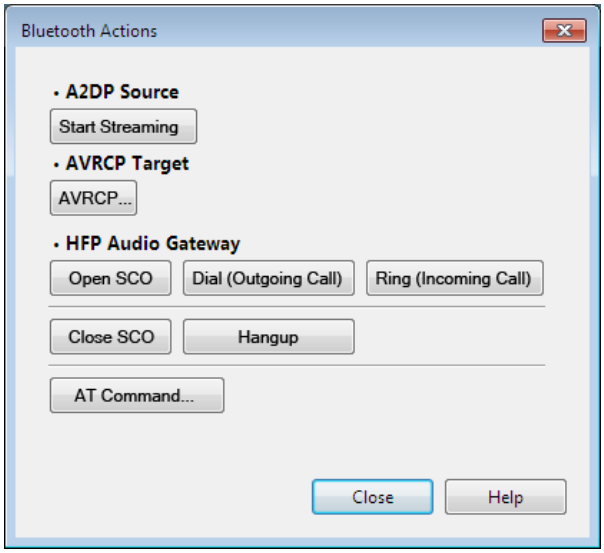
There are 3 steps involved in setting up a connection between any Bluetooth devices. APx500 software provides more visibility and control of options for these communication steps than most DUTs.

- **Discovery**
  Discovery scans the area and lists any Bluetooth devices that are not “undiscoverable” (hidden). Many devices are undiscoverable by default, and must be set to a discoverable mode through a switching sequence.
- **Pairing**
  Pairing establishes a mutual, secure relationship between devices that have at least one compatible profile. Pairing is stored in non-volatile memory in each device. Devices can pair with more than one other device.
- **Connection**
  Paired devices can connect using compatible profiles and roles. Connection enables exchange of audio or control data. Each device can connect with only one device at a time.

Actions Dialogs

Bluetooth Actions are commands that can be sent from the APx analyzer to the Bluetooth device. The current Bluetooth profile(s) and connection determine which commands are available in the Bluetooth Actions dialog box.

Bluetooth Actions for A2DP Source, HFP AG, AVRCP Target



A2DP Source, HFP Audio Gateway, AVRCP Target Actions Dialog

For this profile set, these actions are available:

A2DP Source

For A2DP Source, the actions are

- **Start Streaming / Stop Streaming**

AVRCP Target

- **AVRCP...**
  Opens AVRCP Commands panel, where Absolute Volume commands and Playback Status can be sent.

HFP Audio Gateway

For HFP Audio Gateway, the actions are

- **Open SCO**
- **Dial (Outgoing Call)**
- **Ring (Incoming Call)**
- **Close SCO**
- **Hangup**

A "SCO" is a "Synchronous connection-oriented" link, which is the type of radio link Bluetooth uses for voice data. The SCO link is used in the APx supported protocols HFP and HSP.

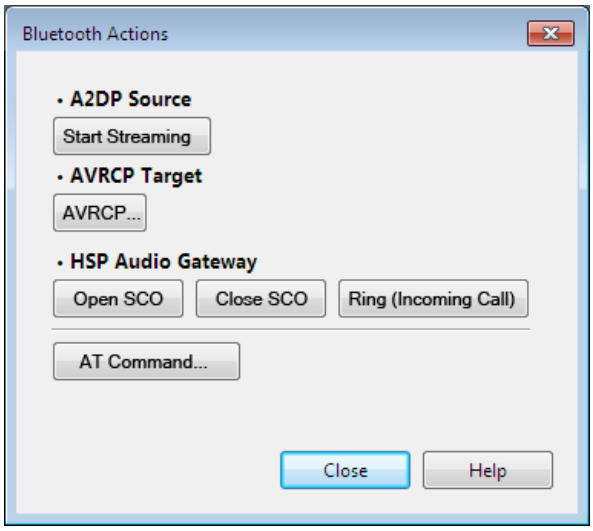
AT Commands

For AT Commands, click the

- **AT Commands...**

button to open the Send AT Commands panel. See page 142.

Bluetooth Actions for A2DP Source, HSP Audio Gateway, AVRCP Target



A2DP Source, HSP Audio Gateway, AVRCP Target Actions Dialog

For this profile set, these actions are available:

A2DP Source

For A2DP Source, the actions are

- **Start Streaming / Stop Streaming**

AVRCP Target

- **AVRCP...**
  Opens AVRCP Commands panel, where Absolute Volume commands and Playback Status can be sent.

HSP Audio Gateway

For HSP Audio Gateway, the actions are

- **Open SCO**
- **Close SCO**
- **Ring (Incoming Call)**

A "SCO" is a "Synchronous connection-oriented" link, which is the type of radio link Bluetooth uses for voice data. The SCO link is used in the APx supported protocols HFP and HSP.

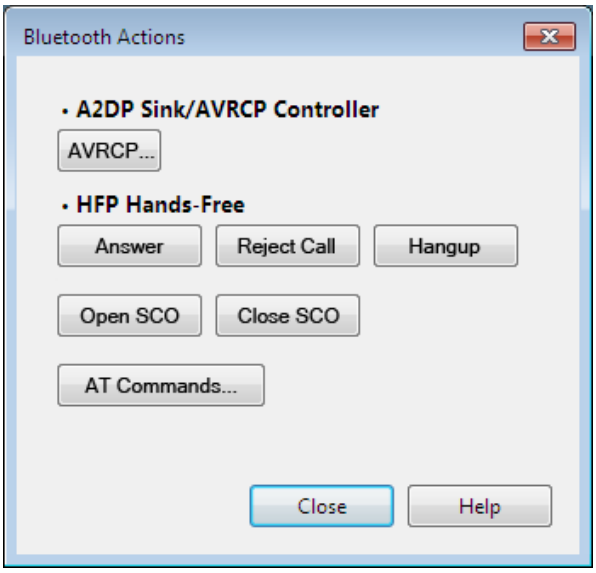
AT Commands

For AT Commands, click the

- **AT Commands...**

button to open the Send AT Commands panel. See page 142.

Bluetooth Actions for A2DP Sink, HFP Hands Free, AVRCP Controller



A2DP Sink, HFP Hands-Free, AVRCP Controller Actions Dialog

For this profile set, these actions are available:

A2DP Sink/AVRCP

For A2DP Sink/AVRCP, click the **AVRCP** button to open the Send AVRCP Command transport controls panel:



PREVIOUS TRACK	REWIND	FAST FORWARD	NEXT TRACK	PLAY	PAUSE	STOP	VOLUME UP	VOLUME DOWN	MUTE

Choose a command to send.

HFP Hands-Free

For HFP Hands Free, the actions are

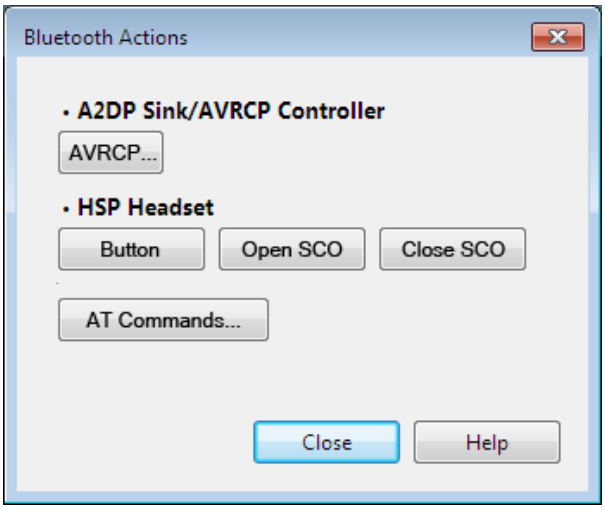
- Answer
- Reject Call
- Hangup
- Open SCO
- Close SCO

AT Commands

- AT Commands...

For AT Commands, click the button to open the Send AT Commands panel. See page 142.

Bluetooth Actions for A2DP Sink, HSP Headset, AVRCP Controller



A2DP Sink, HSP Headset, AVRCP Controller Actions Dialog

For this profile set, these actions are available:

A2DP Sink/AVRCP

For A2DP Sink/AVRCP, click the **AVRCP** button to open the Send AVRCP Command transport controls panel:



PREVIOUS TRACK	REWIND	FAST FORWARD	NEXT TRACK	PLAY	PAUSE	STOP	VOLUME UP	VOLUME DOWN	MUTE

Choose a command to send.

HSP Headset

For HSP Headset, the actions are

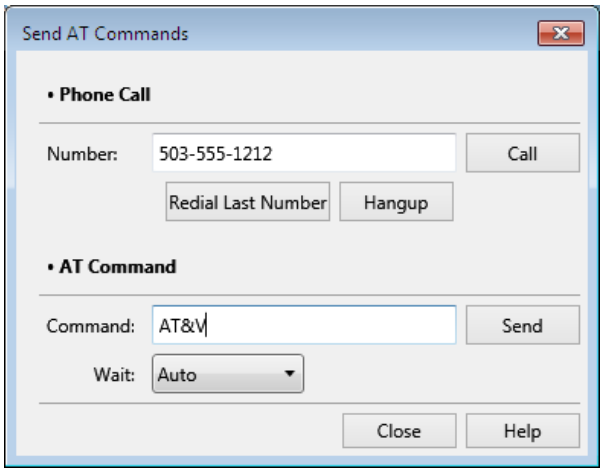
- Button
- Open SCO
- Close SCO

AT Commands

- AT Commands...

For AT Commands, click the button to open the Send AT Commands panel. See page 142.

Bluetooth: Send AT Commands Dialog



You can open the Send AT Commands dialog by clicking **Actions > AT Commands** from Signal Path Setup, or from the context menu opened by right-clicking in the Bluetooth Monitor.

You must have an APx input or output configured for Bluetooth, and be paired and connected to an appropriate Bluetooth HFP profile device.

Phone Call

To **make** a phone call, enter a new phone number in the Number field and click **Call**, or click **Redial Last Number**. Disconnect a phone call by clicking **Hangup**.

AT Command

AT commands are widely used for controlling GSM phones. Lists of AT commands (or the Hayes command set) are easily found online.

Enter any valid AT command as a text string, and click **Send** to transmit the command to the connected device.

Wait

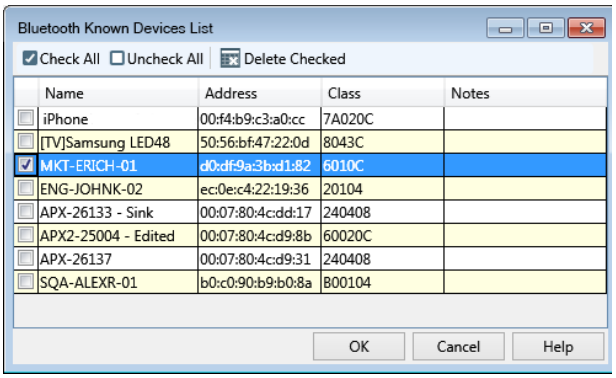
Most AT commands have an expected **OK**, **ERROR**, or other response. Certain AT commands are unsolicited, or are sent with no expectation of a response. APx will indicate an error if an AT command does not provide an expected response.

- **Auto**
  If the sent command begins with "AT", APx waits for an "OK" response, until a timeout of 10 seconds. If the sent command is anything else, APx does not wait for a response.
- **Wait for Response**
  APx waits for an "OK" response to any command, until a timeout of 10 seconds.
- **Do Not Wait**
  APx does not wait for a response

The Known Devices List

You are already familiar with the Bluetooth Current Devices list (page 134, or page 152 for legacy), which is displayed in the large grid central to the Bluetooth Settings panel.

There is also a Known Devices list, which displays every device previously discovered using this computer. This list is stored on the PC, independent of the APx project.



The Known Devices list can be accessed from the Pair Bluetooth Device dialog, or from the Use Specific Device Address **Browse** ... button on the Connect Bluetooth Device Settings in a sequence.

More About Bluetooth...

General

Bluetooth is a short-distance (a few meters) control, data, and audio communications wireless technology. Bluetooth replaces wires in the user's "personal area."

Bluetooth uses low power, frequency-hopping radio in the 2.4 GHz band. Communication is two-way (for handshaking, metadata, etc); some profiles (HFP, for example) support duplex audio (both directions simultaneously); some profiles (A2DP) support only simplex audio (one direction per connection).

Typical uses are mouse, keyboard, cell phone, headphones, hands-free talk and listen.

APx Bluetooth Option

An APx Bluetooth Option module must be fitted in the analyzer instrument to enable Bluetooth transmission and reception. See Bluetooth Option on page 9.

Bluetooth Profiles

Bluetooth has about 30 "profiles" that describe the capabilities and/or current operating modes for Bluetooth devices. For devices to communicate, they must support and share a common profile.

For example, a wireless mouse uses the "HID" profile, which has no audio capabilities. Wireless head-

phones use the “A2DP” profile, which has no cursor control capabilities. The Bluetooth profiles these devices use are not compatible with each other.

APx supported profiles

The APx Bluetooth Option supports four Bluetooth profiles. The supported profiles are

- **A2DP** (Advanced Audio Distribution Profile)
  This is a one-way (source or sink), relatively high-quality stereo audio profile.
- **HFP** (Hands Free Profile)
  This is a bi-directional voice communications audio profile that includes AT-type commands for phone use. Legacy Bluetooth DUTs support only the CSVD codec at sample rate of 8 kHz; HFP 1.6 compatible DUTs support the wideband voice mSBC codec at a sample rate of 16 kHz.
- **HSP** (Head Set Profile)
  This is a simpler version of HFP, using the CSVD codec at 8 kHz and a reduced set of AT commands.
- **AVRCP** (Audio Video Remote Control Profile)
  This provides Play-Pause-Forward-Reverse “remote control” transport-type commands to control an audio source. This profile is typically used in conjunction with A2DP for personal audio player applications.

More about Supported Bluetooth Profiles

HSP

HSP is the “Head Set Profile.” This profile supports voice-quality audio, using the CVSD codec at 8 kHz sample rate. This profile is used for phone-to-headset communication.

Roles

Supported roles are AG “Audio Gateway” (the phone) and HS “Headset”.

Audio

Audio flows in a duplex (bi-directional) mode, connecting the Audio Gateway device to the Headset device.

Actions

A subset of AT commands are also supported for phone operations. Sent AT commands and responses are listed in the AT Command Log. See page 36.

HFP

HFP is the “Hands Free Profile.” This profile supports voice-quality audio, using the CVSD codec at 8 kHz sample rate or the “wideband speech” mSBC codec at 16 kHz. This profile is used for phone-to-headset communication and for phone-to-car kit hands free communication.

Note: HFP wideband speech is only available if the mSBC codec is supported by your Bluetooth module hardware. See Bluetooth Option on page 9 for more information.

Roles

Supported roles are AG “Audio Gateway” (the phone) and HF “Hands Free” (the headset or car kit mic/speaker).

Audio

Audio flows in a duplex (bi-directional) mode, connecting the Audio Gateway device to the Hands Free device.

Actions

A subset of AT commands are also supported for phone operations. Compared to HSP, HFP provides a few more AT commands for hands-free convenience, such as last number redial.

A2DP

A2DP is the “Advanced Audio Distribution Profile.” This profile supports higher bit rate, higher performance stereo audio.

Roles

APx supported profile roles are “source” (transmitting audio) and “sink” (receiving audio). Audio is distributed in one direction only, from the source device to the sink device.

Audio

The SBC codec is mandatory; Audio Precision Bluetooth Options also support the aptX codec. The Bluetooth Duo option also supports aptX HD, aptX Low Latency and AAC codecs. In A2DP Source, the APx Bluetooth transmitter mutes the audio when the average signal level falls below –54 dBFS for more than 1 second. We recommend maintaining test levels above –54 dBFS for meaningful results.

A2DP Start Streaming/Stop Streaming

These controls are used by some devices (such as car kits) to regulate audio traffic.

AVRCP

AVRCP is the “Audio/Video Remote Control Profile.” This profile is used in conjunction with A2DP, and provides “transport” controls such as Play, Pause, Reverse, Forward, and also Playback Status and Absolute Volume Control. AVRCP activity is logged in the AVRCP Command Log. See page 36.

Playback Status and Absolute Volume are not supported by Legacy Bluetooth Option modules.

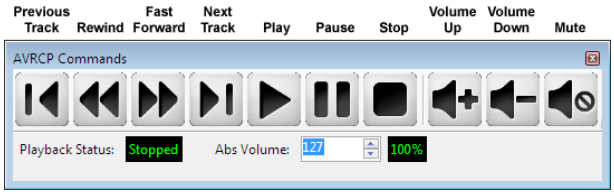
AVRCP Target role

When A2DP is a Source, AVRCP is a Target. In this role, APx supports AVRCP Absolute Volume, which can send an Absolute Volume command to the rendering Sink device, instructing it to raise or lower the Absolute Volume. When APx is the target, AVRCP status, events, and sent and received commands are displayed in the Bluetooth Monitor AVRCP log.

> *Received AVRCP commands are not acted upon in APx, with the exception of Commands Change Generator Settings, if enabled.* *See page 139.*

AVRCP Controller role

When A2DP is a Sink, AVRCP is a Controller. In this role, APx supports AVRCP transport controls, which can be sent to the Source device to control audio playback; additionally, Playback Status events are read, and the sink device's current Absolute Volume is provided for reading by the source device. The supported controls are



Note that the AVRCP **Volume Up, Volume Down** and **Mute** transport controls are not the same as AVRCP Absolute Volume.

Volume Up, Volume Down and Mute scale the audio level at the source in response to commands sent from the sink; Absolute Volume scales the audio level at the sink, in response to commands sent from the source. When Absolute Volume is supported in a sink device, the source device audio level is typically scaled to 100%.

More about Bluetooth SSP (Secure Simple Pairing) Mode

SSP is required for devices using Bluetooth v 2.1 and later. Bluetooth v 2.0 and earlier devices use Legacy Pairing.

APx provides four SSP (Secure Simple Pairing) options. You must choose the option that is appropriate for the Bluetooth device you are testing.

Man-in-the-middle (MITM)

Man-in-the-middle or MITM is the term used in Bluetooth technology to refer to Secure Simple Pairing mechanisms that require human interaction. One device may display a PIN, for example, and the user may be required to enter the same PIN using a keyboard on a second device. MITM methods add security to the pairing process.

APx SSP Modes

Just Works

This mode is used in Bluetooth device relationships that require little security, such as headsets. User interaction is not required.

Numeric Comparison, Display Only

Some Bluetooth devices have only a display screen to interact with a user; an example is a car kit (car stereo head unit). Use this mode when APx is playing the role of such a device.

A use case would be testing a smart phone. APx would be an A2DP Sink, HFP with a display only (the car kit), and the DUT would be the smart phone. The phone would initiate pairing, and APx would display the PIN transmitted from the phone. The user would confirm (using a button on the phone) that the PIN displayed in APx matched the PIN on the phone.

Numeric Comparison, Display+Buttons

Some Bluetooth devices have a display screen and one or more buttons to interact with a user; the button may be used for a binary (Yes/No) response to a query. An example is a smart phone. Use this mode when APx is playing the role of such a device.

A use case would be testing a car kit. APx would be an A2DP Source, HFP Hands-Free with a display and buttons (the smart phone), and the DUT would be the car kit. APx would initiate pairing and transmit a PIN to the DUT. The DUT would display the PIN, and the user would confirm (using a button in the APx prompt dialog) that the PIN in the DUT display and the APx PIN matched.

Numeric Comparison, Keyboard

Some Bluetooth devices have a numeric or alphanumeric keyboard to interact with the user. An example is a computer used as an audio source, distributing iTunes audio to a home entertainment system. Use this mode when APx is playing the role of such a device.

There are not obvious use cases in audio test for this mode, which is included in APx for completeness. A non-audio test use case would be pairing a Bluetooth keyboard to a tablet computer. The computer would display a PIN, and the user would enter a PIN (using the keyboard) that matches the PIN on the tablet screen.

Legacy Pairing

Bluetooth v2.1 and later devices (such as APx) are permitted to use Legacy Pairing modes when pairing with a Bluetooth v2.0 or earlier device.

Glossary of Bluetooth terms

These terms are used in the Audio Precision Bluetooth audio testing implementation.

A2DP
is the Advanced Audio Distribution Profile, with two roles: source and sink. This profile supports higher bit rate, higher performance stereo audio, with sample rates up to 48 kHz. The SBC codec is mandatory; codecs such as mp3, AAC, apt-X and others are optionally supported.

AG
is the duplex Audio Gateway (phone) role, used in HFP and HSP.

apt-X
Qualcomm aptX is an optional high-performance codec used in A2DP.

aptX Low Latency
Qualcomm aptX Low Latency is an optional high-performance low-latency codec used in A2DP.

aptX HD
Qualcomm aptX HD is an optional high-performance high-definition codec used in A2DP.

AT command
is an audible signal used to control a device. AT commands are a PSTN (Public Switched Telephone Network) legacy.

AVDTP
is the Audio/Video Distribution Transport Protocol. This protocol defines A/V stream negotiation, establishment, and transmission procedures. Also specified are the message formats that are exchanged between such devices to transport their A/V streaming in A/V distribution applications.

AVRCP
is the Audio/Video Remote Control Profile. This profile is used in conjunction with A2DP, and provides "transport" controls such as Play, Pause, Reverse, Forward, etc. Events and notifications allow reporting of current status information.

Connect
Paired devices can connect using compatible profiles and roles. Connection enables exchange of audio or control data. Each device can connect with only one device at a time.

CVSD
is Continuously Variable Slope Delta modulation, the codec used in HSP and legacy HFP profiles. With HFP 1.6, higher data rates enable the use of a higher quality codec called mSBC.

Device address
Every Bluetooth device has a unique 48-bit device address, in APx500 displayed in hex format, such as 00:f4:b9:c3:a0:cc.

Device class
A Bluetooth device class is a code that identifies the type of device. Some Bluetooth devices will only recognize devices of a certain class.

Discovery
Bluetooth devices that are not paired can discover each other when they are in range. Some devices allow users to make the device undiscoverable.

duplex
refers to a bi-directional audio channel, across which speaking and listening can occur simultaneously.

eSCO
Extended SCO, available in the Bluetooth 1.2 specification. Adds new packet types (EV1, etc.) and more flexibility in channel parameters, allows retransmission of bad packets. Also see SCO.

EV1
First of a series of packet types available with eSCO, as opposed to the HV1, etc. packet types in SCO.

Friendly name
is an optional name for a Bluetooth device, more easily understood and remembered than the device address. An example is "iPhone".

HF
is Hands Free, the duplex the duplex "speaker/headset" device in the HFP profile.

HFP
is the Hands Free profile, intended to allow hands-free device operation in an automobile. It is similar to HSP, with more phone controls. Roles are AG and HF.

HFP 1.6
is a revision of the HFP specification, which allows new packet types and higher data rates compared to legacy HFP. These data rates and newly supported codecs such as mSBC enable WBS (Wideband Speech) performance.

HS
is Head Set, the duplex "speaker/headset/hearing aid" device in the HSP profile.

HSP
is the Head Set profile, intended to enable Bluetooth mobile phone use. Roles are AG and HS.

HV1
First of a series of packet types available with SCO, as opposed to the EV1, etc. packet types in eSCO.

Link key
is a shared secret exchanged in pairing.

mSBC
is a monaural version of the SBC codec, optimized for use in the HFP 1.6 profile. This codec and the

higher available sample rates enable Wideband Speech operation.

Pair
Bluetooth devices that have discovered each other can be paired by exchanging a link key to form a bond. For some devices, pairing is automatic upon discovery; for others, user interaction is required. Pairing establishes a mutual, secure relationship between devices that have at least one compatible profile. Pairing is stored in non-volatile memory in each device. Devices can pair with more than one other device.

PIN code
A Personal Identification Number embedded in a device or provided by a user that is exchanged in a pairing negotiation.

Profile
One of a number of defined Bluetooth relationships covering a range of devices and use cases. A device can support more than one profile, and devices can be paired acknowledging more than one profile.

SBC
(Sub-band Codec), the mandatory codec for the A2DP Profile. Other codecs are allowed.

SCO
is a synchronous connection oriented channel, a full duplex data channel with 64 kbit/s data rate in each direction. The CVSD codec is used, and the three HV types of data packets are available. Also see eSCO.

sink
is the term used for the device that receives audio in a uni-directional system.

source
is the term used for the device that transmits audio in a uni-directional system.

SSP
is Secure Simple Pairing, the pairing methods required in Bluetooth v2.1. Audio Precision supports the SSP Just Works, Numeric Comparison and Passkey Entry methods.

18

Legacy Bluetooth I/O

Introduction

The APx legacy Bluetooth option is a hardware module previously available for the APx modular analyzers (the APx555, and the APx525 and 585 families of analyzers, including the APx582). It provides *Bluetooth*® wireless technology source and sink interfaces. See licensing acknowledgements on the copyright page of this manual (immediately following the title page).

The APx Bluetooth profiles support audio in a lower-quality (voice) mode (the HFP and HSP profiles) and in a higher-quality (music) mode (the A2DP profile).

The legacy option has been succeeded by the more capable Bluetooth Duo option, but it continues to be supported by the APx500 measurement software.

This chapter discusses operation of the legacy Bluetooth option under the most current version of APx500. Operation of the Bluetooth Duo option is discussed in Chapter 17.

Identification

The most obvious difference between the legacy Bluetooth modules and the Bluetooth Duo module is the number of external antennas: the legacy modules have one; Bluetooth Duo has two.

Legacy Bluetooth modules have the model number 113 or 213 painted on the front panel, and are identified in the Help > About product components list as BAZL or BAZW.

See more About Bluetooth on page 142, and Using Bluetooth in a Sequence on page 557. See More About Supported Bluetooth Profiles on page 143.

The Bluetooth Monitor

The status of a number of current Bluetooth settings is shown in the Bluetooth Monitor. Additionally, Bluetooth actions, settings and utility functions are available through a context menu (right-click in the Bluetooth Monitor display.) See Bluetooth Monitor on page 36.

Bluetooth Discovery, Pairing and Connection

Bluetooth connections require successful radio frequency (RF) communication between the APx analyzer and the DUT, and successful device handshaking through device discoverability, pairing and connection protocols. Data transfer is encrypted and must be authorized by the exchange of link keys, and personal identification number (PIN) codes may be required.

Bluetooth Settings are Global

Bluetooth settings are unique in APx500, in that they are *global*. This means the settings made in the Bluetooth Settings dialog affect the entire project: all measurements, all signal paths.

Source, Sink, Output and Input

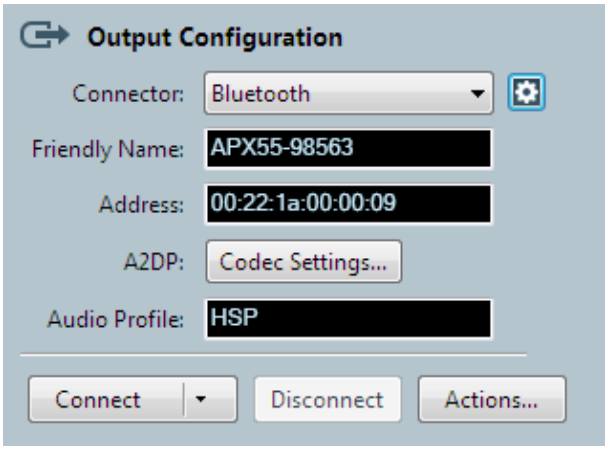
When the APx **Output Configuration** is set to **Bluetooth**, the instrument can transmit audio using the A2DP Source profile, the HFP Audio Gateway profile, or the HSP Audio Gateway profile. When the APx **Input Configuration** is set to **Bluetooth** input, the instrument can receive audio using the A2DP Sink profile, the HFP Hands Free profile, or the HSP Headset profile.

There is one case where you are can output audio and input audio simultaneously: set the APx Output and APx Input to Bluetooth, and connect to the DUT using the HFP Profile. When the SCO is opened, you can both send and receive audio, since HFP is a bi-directional link.

Output Configuration: Bluetooth

This Signal Path Setup I/O selection enables output of audio from the APx generator to a Bluetooth device

under test, using Bluetooth radio frequency (RF) transmission.



Note: The Bluetooth choice is only available for analyzers fitted with the APx Bluetooth Option module.

Connector

Choose **Bluetooth** here. Since the Bluetooth connection isn't valid until you choose, pair and connect with a device, the Bluetooth Settings dialog opens immediately. See page 150.

Settings

Click the **Settings...** button to discover, select, pair, connect and otherwise configure a Bluetooth link. See page 150.

Friendly Name

The friendly name of the APx Bluetooth module will appear here.

Address

The address of the APx Bluetooth module will appear here.

A2DP

The codec settings for the A2DP profile can be viewed and edited here.

Audio Profile

This displays the active profile, if any. See More About Supported Bluetooth Profiles on page 143.

Connect

If the APx is already paired with one or more Bluetooth devices, you can open the **Connect** flyout menu to select a device and one or more profiles for connection. See More About Supported Bluetooth Profiles on page 143.

Disconnect

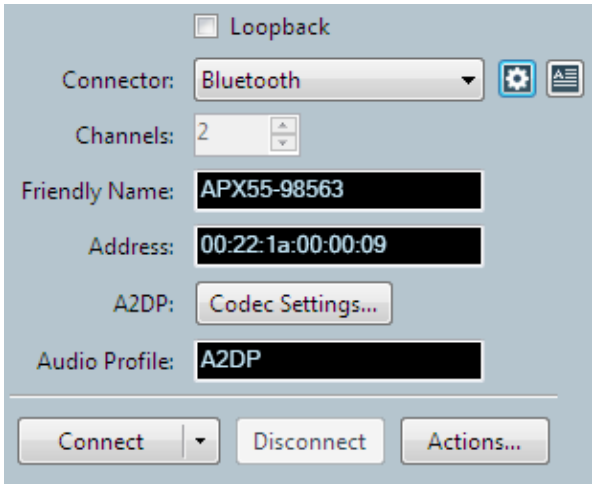
If the APx is already connected to a Bluetooth device, you can click the **Disconnect** button to terminate the connection.

Actions

Click **Actions** to open the **Bluetooth Actions** dialog. If the APx is already connected to a Bluetooth device, you can initiate Bluetooth Actions from this dialog. Only the actions appropriate to the current connection and profile are shown. See Actions on page 140.

Input Configuration: Bluetooth

This selection enables input of audio from a Bluetooth device under test, using Bluetooth radio frequency (RF) transmission.



Note: The Bluetooth choice is only available for analyzers fitted with the APx Bluetooth Option module.

Loopback

Loopback is not available when either the Output or Input Configuration is set to Bluetooth.

Connector

Choose **Bluetooth** here. Since the Bluetooth connection isn't valid until you choose, pair and connect with a device, the Bluetooth Settings dialog opens immediately. See page 150.

Settings

Click the **Settings...** button to discover, select, pair, connect and otherwise configure a link with a Bluetooth device. See page 150.

Edit Channel Labels

In the Channel Labels dialog you can rename the input channels with names of your choice. See page 56. These settings are project-wide, but can be over-

ridden for any measurement result. To set project colors, click the **Project Colors...** button in the Labels dialog, or go to the Project/Sequence Properties dialog, Colors tab. See Setting Graph Colors and Styles on page 30.

Channels

This is a reading field that displays the current input channel count, determined by profile settings.

Friendly Name

The friendly name of the APx Bluetooth module will appear here.

Address

The address of the APx Bluetooth module will appear here.

A2DP

The codec settings for the A2DP profile can be viewed and edited here.

Audio Profile

This displays the active profile, if any. See More About Supported Bluetooth Profiles on page 143.

Connect

If the APx is already paired with one or more Bluetooth devices, you can open the **Connect** flyout menu to select a device and one or more profiles for connection. See More About Supported Bluetooth Profiles on page 143.

Disconnect

If the APx is already connected to a Bluetooth device, you can click the **Disconnect** button to terminate the connection.

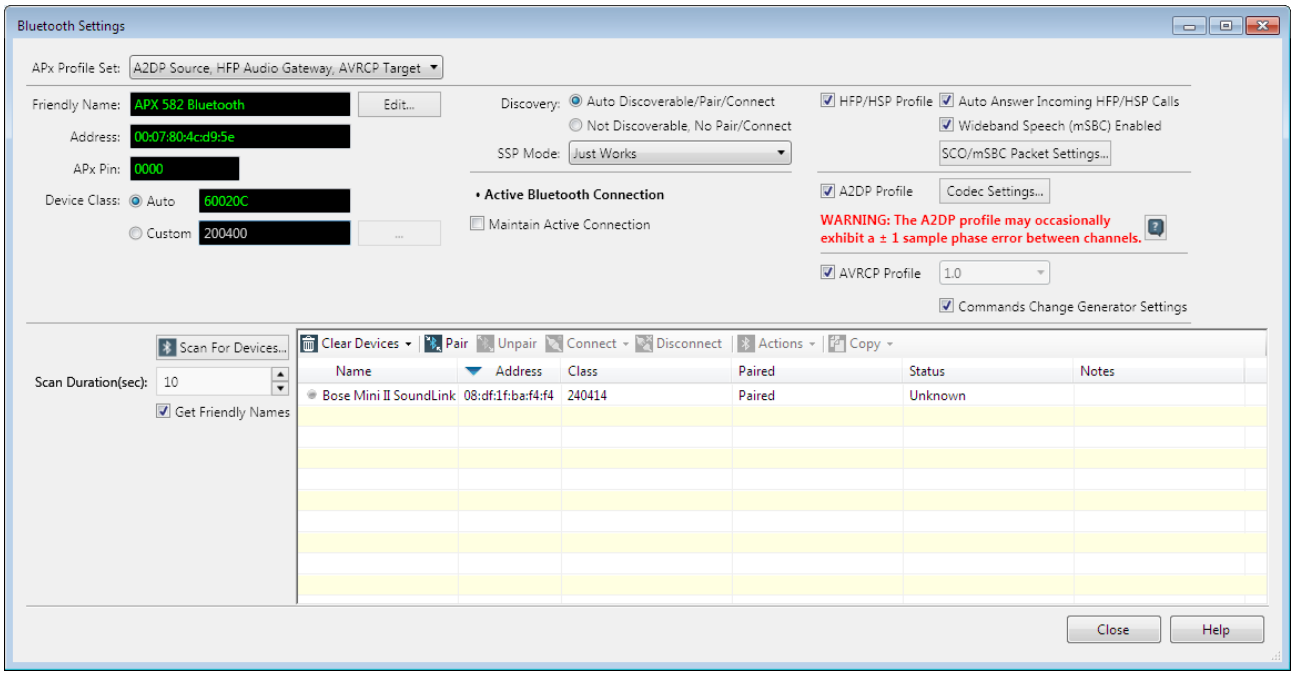
Actions

Click **Actions** to open the **Bluetooth Actions** dialog. If the APx is already connected to a Bluetooth device, you can initiate Bluetooth Actions from this dialog. Only the actions appropriate to the current connection and profile are shown. See Actions on page 140.

Operation with absent or corrupt interface signal

When Input Configuration is set to a digital input and no signal is present, or if the signal is corrupt or out of range, the input receiver cannot lock (synchronize) to the interface signal, and no valid audio can be recovered. The sample rate indicator in the Status Bar will display an “unlocked” warning, and any measurement result referencing audio from the unlocked input will display an “----” (invalid) result. See page 636 for more about invalid results.

Legacy Bluetooth Settings dialog



Bluetooth Settings dialog for legacy Bluetooth, shown in the A2DP Source, HFP Audio Gateway, AVRCP Target profile set.

This topic explores the legacy Bluetooth settings dialog. The Bluetooth Duo settings dialog is discussed beginning on page 132.

This dialog provides settings for Bluetooth discovery, pairing, connection and other configuration. Unlike most APx settings, Bluetooth profile selection, pairing and connection settings are global to the APx project. Settings made here affect both Bluetooth source and sink configurations across all signal paths in the project.

A flow chart depicting discovery, pairing and connection paths is shown on page 139.

APx Profile Set

APx has four sets of profiles that the Bluetooth Option module can assume. Each set lists the profiles available for connection and/or the profiles available for access by a remote device. Choose an APx Bluetooth Profile Set here.

- A2DP Source, HFP Audio Gateway, AVRCP Target
- A2DP Source, HSP Audio Gateway, AVRCP Target
- A2DP Sink, HFP Hands Free, AVRCP Controller
- A2DP Sink, HSP Headset, AVRCP Controller

In A2DP Source, the APx Bluetooth transmitter mutes the audio when the average signal level falls below –54 dBFS for more than 1 second. We recommend maintaining test levels above –54 dBFS for meaningful results.

APx Identification

The upper left side of the dialog displays a number of APx Identification settings readable by other Bluetooth devices.

Friendly Name

The friendly name of the APx Bluetooth module is displayed here.

Edit

You can change the Friendly Name in the Set Friendly Name dialog.

Address

The Address of the APx Bluetooth module is displayed here.

APx PIN

The APx Bluetooth PIN is displayed here.

Device Class

- **Auto**
  Auto uses the Device Class assigned by default: in Source profiles, 60020C; in Sink profiles, 240408.
- **Custom**
  Custom allows you to select a Device Class for the APx Bluetooth module. Use the ... browse button to open the Edit Device Class Dialog (page 137).

Discovery

Auto Discoverable/Pair/Connect

When this is selected (the default), the APx Bluetooth Option is discoverable by remote Bluetooth devices. If

the remote device initiates pairing, the APx will pair. If the remote device initiates connection, the APx is available to connect for the profiles checked here.

Not Discoverable, No Pair/Connect

When this is selected, the APx Bluetooth Option is not discoverable by remote devices, and it will not respond to pairing or connection initiated by remote device.

SSP Mode

APx provides four SSP (Secure Simple Pairing) options. The **Just Works** choice is used in Bluetooth device relationships that require little security. The **Numeric Comparison** choices are man-in-the-middle (MITM) interactions for device relationships that require higher security. Choose the SSP Mode that is appropriate for the Bluetooth device you are testing. See More About Secure Simple Pairing on page 144 for detailed information.

- Just Works
- Numeric Comparison, Display Only
- Numeric Comparison, Display+Buttons
- Numeric Comparison, Keyboard

Active Bluetooth Connection

☑ If **Maintain Active Connection** is checked, the APx Bluetooth Option will remain connected to the Bluetooth device under test throughout the project, even across Signal Paths where Bluetooth is not configured as an input or an output. However, if you change to Bench Mode, the Bluetooth device will be disconnected.

Profile Configurations

The upper right side of the dialog displays a number of APx profile configurations. The controls and settings here vary with Source or Sink (Target or Controller) settings.

HFP/HSP Profile

☑ If this box is checked, the HFP/HSP profile is enabled, and APx support for this profile is disclosed to remote devices. You may elect to use the profile at connection time, or not.

☐ If this box is unchecked, the HFP/HSP profile will not be available on the connection menu, and APx support for this profile is not disclosed to remote devices. These further options are available:

These further options are available:

- ☑ Auto Answer Incoming HFP/HSP Calls
- ☑ Wideband Speech (mSBC) Enabled

Note 1: Wideband Speech is not available in early legacy Bluetooth modules of type BAZL.

Note 2: For Hands Free/Headset profiles, this setting has immediate effect. For Audio Gateway profiles, this setting is applied at the next connection.

SCO/mSBC Packet Settings...

This button opens the SCO/mSBC Settings Dialog. See page 153. These features are not available in early BAZL legacy Bluetooth modules.

A2DP Profile

☑ If this box is checked, the A2DP profile is enabled, and APx support for this profile is disclosed to remote devices. You may elect to use the profile at connection time, or not.

☐ If this box is unchecked, the A2DP profile will not be available on the connection menu, and APx support for this profile is not disclosed to remote devices. This further option is available:

Codec Settings...

This button opens the A2DP Codec Settings dialog. See page 135.

A2DP phase error with APX-BT-WB legacy hardware

A flaw in the firmware supplied by our Bluetooth chipset vendor causes the APx legacy Wideband Speech Bluetooth module (APX-BT-WB) (BAZW) to exhibit a ±1 sample phase error between channels when using the A2DP profile (source and sink), in about 15% of trials. The error, when present, is introduced at the time the Bluetooth settings are committed to the module chipset, and is constant during the operation of the Bluetooth channel. When present, the error affects results sensitive to phase, such as the time domain signal monitor, the group delay result, and the interchannel phase result.

The error only occurs only when using the APX-BT-WB hardware, only in the A2DP profile. HSP and HFP profiles are unaffected and operate correctly.

A warning appears on the Bluetooth Settings panel when APX-BT-WB hardware is in use.

This issue does not exist with early BAZL legacy Bluetooth modules, or with the Bluetooth Duo module.

☑ AVRCP Profile

☑ If this box is checked, the AVRCP profile is enabled, and APx support for this profile is disclosed to remote devices. You may elect to use the profile at connection time, or not.

☐ If this box is unchecked, the AVRCP profile will not be available on the connection menu, and APx support for this profile is not disclosed to remote devices.

The legacy Bluetooth module supports AVRCP version 1.0.

This further option is available when the profile is set to **AVRCP Target:** ☑ **Commands Change Generator Settings**. See page 153.

Scan

The lower left area of the dialog has controls and settings to scan for nearby Bluetooth devices.

Scan for Devices

This button initiates a scan to identify all the discoverable Bluetooth devices within range. Discovered devices are listed in the Current Devices List.

Scan Duration

This setting limits the scan time to the value set. The maximum time is 48 seconds. The default is 10 seconds.

☑ Get Friendly Names

By default, APx asks for a device's friendly name. In automated processes, this process adds extra time. Uncheck this checkbox to disable this feature.

Current Devices List

The Current Devices List occupies most of the lower part of the dialog.

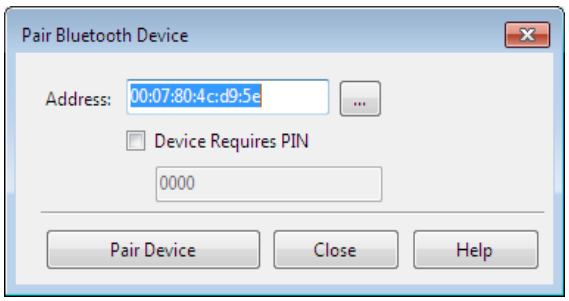
This grid shows the devices discovered in the scan, along with information fields. There are controls to pair, unpair, connect, disconnect, initiate actions and to copy the device Bluetooth address and link key. Select a device to configure the relationship.

Clear Devices

You can choose to remove all devices from the list, or all paired devices, or all unpaired devices. When a device is removed from the list, any pairing information for that device is also removed. Click **Scan** to repopulate the list.

Pair

Pairing provides a way for devices to exchange link keys. To pair with a device, refer to the documentation received with the device. To enable pairing, you may be required to execute a command or series of commands at the device. When pairing is enabled, select the device in the Current Devices List, and click **Pair**. The Pair Bluetooth Device dialog box will open.



The **Browse** ... button in this dialog opens the Known Devices list, from which you may select other devices which have been previously discovered using this computer. See page 142.

Pairing information is saved in the project, until the Current Devices List is cleared, or the until device is Unpaired.

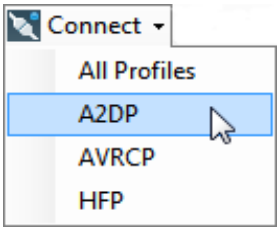
Some devices may require the exchange of a PIN code. The APx Bluetooth PIN code is set to 0000; a field in the pairing dialog allows input of a remote device's PIN code.

Unpair

Unpairs the selected, previously paired device.

Connect

Connects to the selected, previously paired device. A Connect menu opens. Select the profile you would like to offer to the device.



Disconnect

Disconnects the selected, connected device.

Actions

Opens various **Actions** dialogs. This duplicates the function of the Actions button shown on Signal Path Setup, without the necessity of leaving the Bluetooth Settings dialog. See page 140.

Copy

Copies the device address and key link to the Windows Clipboard. This text can then be pasted into another application or tool.

Codec Settings

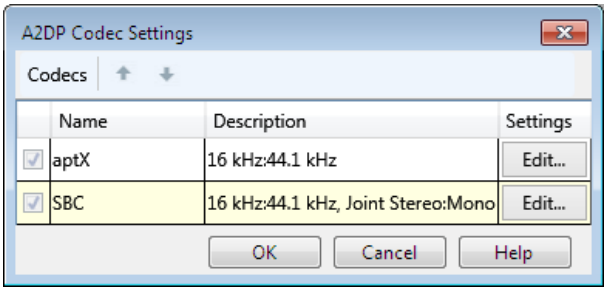
A2DP Codec Settings

The Bluetooth A2DP profile always supports the SBC audio codec. Additional codecs are optional.

In APx, you can select which optional codecs you would like to support, and edit features of the codecs you choose.

Using the arrow icons at the top of the dialog, you can set an order of priority. APx will move from the top-

most codec downward until finding the codec that the DUT supports.



Codecs supported, in default order of priority:

aptX

You can choose to support any or all of the following sample rates for the aptX codec:

APx as A2DP source:

- 44.1 kHz
- 32 kHz
- 16 kHz

APx as A2DP sink:

- 48 kHz
- 44.1 kHz
- 32 kHz
- 16 kHz

SBC

You can choose to support any or all of the following sample rates for the SBC codec:

APx as A2DP source:

- 48 kHz (This sample rate is not supported by early legacy Bluetooth modules of type BAZL.)
- 44.1 kHz
- 32 kHz
- 16 kHz

APx as A2DP sink:

- 48 kHz
- 44.1 kHz
- 32 kHz
- 16 kHz

Additionally, you can choose to support any or all of the following channel modes:

- Joint Stereo
- Stereo
- Dual Channel
- Mono

Commands Change Generator Settings

When APx is configured for A2DP Source AVRCP Target, the APx generator can now respond to certain AVRCP commands.

AVCRP Target profile

In the AVRCP Target profile, APx acts as an A2DP source device, sending an audio stream to a remote rendering device (speaker or headphones). APx responds to AVRCP Commands sent from the remote rendering device.

When the AVRCP Target profile is selected, the **Commands Change Generator Settings** ☑ checkbox on the Bluetooth Settings panel allows you to enable or disable this feature. This setting is enabled by default.

AVRCP generator commands are only acted upon when the active measurement has a generator with an ON/OFF button, and the generator is not disabled.

Commands

- The AVRCP command **Play** turns the generator **On**
- The AVRCP command **Pause** turns the Generator **Off**
- The AVRCP command **Stop** turns the Generator **Off**, and **stops** A2DP streaming.
- The AVRCP command **Next Track** increases the generator **Frequency** equivalent to pressing the **up** arrow on the frequency control in the panel
- The AVRCP command **Prev Track** decreases the generator **Frequency** equivalent to pressing the **down** arrow on the frequency control in the panel.
- AVRCP **Volume Up** messages are **not** received in legacy Bluetooth modules.
- AVRCP **Volume Down** messages are **not** received in legacy Bluetooth modules.

SCO/mSBC Settings

The SCO/mSBC settings are only available if the mSBC codec is supported by your Bluetooth module hardware. Early BAZL legacy Bluetooth modules do not support these features.

SCO/eSCO

This section of the dialog provides management of SCO and eSCO channels and packet types. A SCO is a synchronous connection-oriented link, a radio link used for voice data. eSCO links are enhanced SCO, providing greater flexibility. For additional reading, see the Bluetooth protocols article in Wikipedia.

eSCO Latency (ms):

Set the eSCO latency here.

eSCO Retransmission

- eSCO Retransmission preferences:
  - No Retransmission
  - Power Saving Optimized
  - Link Quality Optimized, or
  - No Preference

For an eSCO connection, you can specify any combination of supported packet types:

- EV1
- EV2
- EV3
- 2-EV3
- 3-EV3
- 2-EV5
- 3-EV5

For a SCO connection, you can specify any combination of supported packet types:

- HV1
- HV2
- HV3

mSBC/Wideband Speech

This tab enables management of mSBC parameters. mSBC requires an eSCO link. mSBC is the modified SBC codec, a component of HFP 1.6.

For an mSBC connection, you can specify

mSBC channel latency in ms

mSBC Retransmission preferences:

- No Retransmission
- Power Saving Optimized
- Link Quality Optimized, or
- No Preference

Packet Types

and any combination of supported packet types:

- HV1
- HV2
- HV3
- EV1
- EV2
- EV3
- 2-EV3
- 3-EV3
- 2-EV5
- 3-EV5

PDM I/O

Introduction

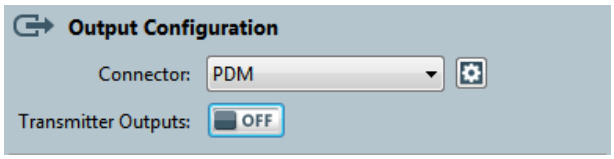
PDM I/O (input/output) requires the APx PDM Option module, described on page 10. This chapter discusses the APx500 Signal Path Setup settings for PDM output and input configurations. Also see More about PDM on page 160.

PDM Jitter Testing

With PDM hardware shipped after August 5th, 2015, jitter testing is available for analyzers fitted with the Advanced Master Clock. For more information see Jitter Support in APx beginning on page 71.

Output Configuration

PDM (when installed)



These controls are shown on **Signal Path Setup** when **PDM** is selected as the **Output Connector**.

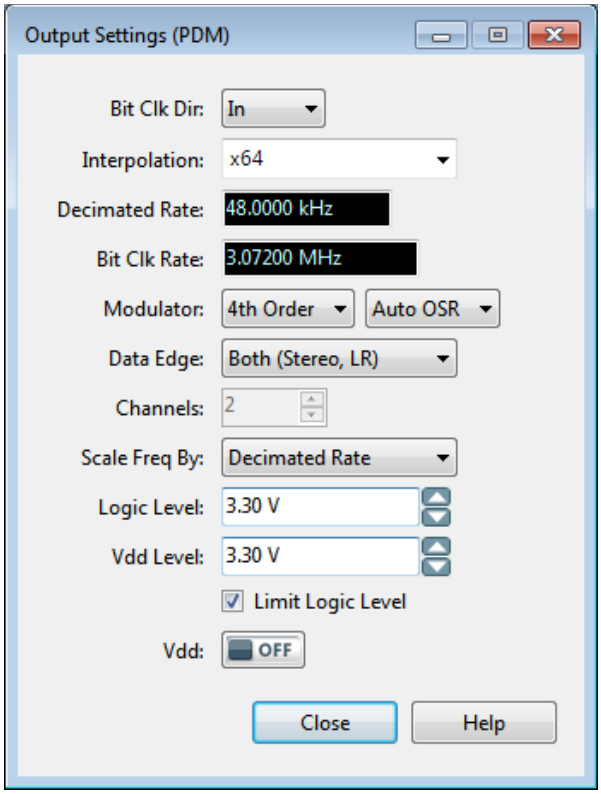
Transmitter Outputs (ON/OFF)

When the PDM Output Bit Clock Direction is set to **Out** (see following), you can switch the transmitter outputs **ON** or **OFF** here.

Settings

The Settings button opens the Output Settings (PDM) dialog, which reveals more PDM settings and readings.

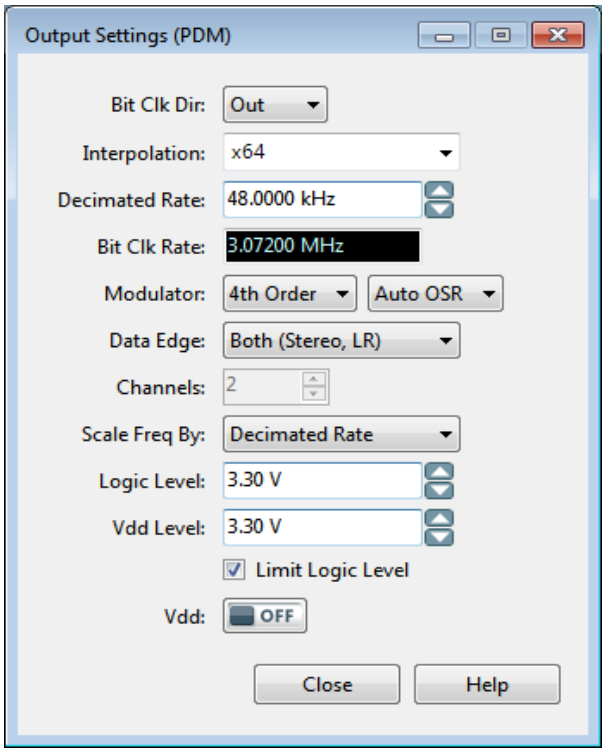
Output Settings (PDM)



Output Settings with Bit Clock set to In (default).

These controls are in the Output Settings (PDM) dialog, available by clicking the **Settings** button from Sig-

nal Path Setup, when Output Configuration is set to PDM.



Output Settings with Bit Clock set to Out.

Bit Clk Dir:

The Bit Clock connector associated with the PDM Output on the PDM module can be configured as an input or an output. Choose a clock direction by selecting Bit Clk Dir: In (the default), or Out.

A typical use case provides that an upstream PDM transmitter (the APx PDM output, representing a MEMS microphone) is connected to a downstream PDM input (the DUT), and that the transmitter receives the clock from the receiver. In this case the APx output-associated clock is set to In, where APx is the slave and the DUT is the master. This is default clock direction setting.

Jitter

A PDM model 228 module installed in an APx with the Advanced Master Clock (AMC) can have jitter applied to the bit clock when PDM is an output and **Bit Clk Dir:** is set to **Out**. In **Signal Path Setup > Clocks > Jitter Generator**, select **Apply To: Digital Output** and turn jitter **ON**.

For more information see Jitter Support in APx beginning on page 71.

Interpolation:

Set the transmitter interpolation factor here.

APx provides a wide range of interpolation settings to accommodate common ratios. The supported Decimated Rates (baseband audio sample rates) are constrained by the interpolation setting, as shown in the table here. Power-of-two rates are shown in bold type.

Interpolation Factor	*Maximum Decimated Rate*	*Auto OSR*
16x	216 kHz	x64
16.67x	216 kHz	x64
21.33x	216 kHz	x64
24x	216 kHz	x64
25x	216 kHz	x64
x32	216 kHz	x64
x33.33	216 kHz	x64
x37.5	216 kHz	x64
x42.67	216 kHz	x64
x48	216 kHz	x64
x50	216 kHz	x64
x62.5	216 kHz	x64
x64	216 kHz	x64
x66.67	216 kHz	x128
x75	216 kHz	x128
x85.33	216 kHz	x128
x96	216 kHz	x128
x100	216 kHz	x128
x125	196.608 kHz	x128
x128	192 kHz	x128
x150	163.84 kHz	x256
x192	128 kHz	x256
x200	122.88 kHz	x256
x250	98.304 kHz	x256
x256	96 kHz	x256
x300	81.92 kHz	x512
x384	64 kHz	x512
x400	61.44 kHz	x512
x500	49.152 kHz	x512
x512	48 kHz	x512
x600	40.96 kHz	x512
x768	32 kHz	x512
x800	30.72 kHz	x512

Decimated Rate:

This control is available when the Bit Clock connector associated with the PDM output is set to Out. The clock direction is set with the Bit Clk Dir control, above.

The Decimated Rate control specifies the decimated sample rate (the baseband audio sample rate) by setting the PDM input Bit Clock output rate. The relationship between the decimated rate and the bit clock rate depends upon the interpolation ratio, set by the Interpolation control, above. For example, a specified Decimated Rate of 48 kHz and an interpolation factor of 64 would set the PDM Bit Clock output to 3.072 MHz (48000 × 64 = 3072000).

To set the decimated rate, click the up/down arrows to select a standard rate, or enter an arbitrary value in the **Decimated Rate** field. Maximum decimated rate is shown in the adjacent table. Minimum supported decimated rate is 4 kHz. Default is 48 kHz.

This control is unavailable when the PDM output Bit Clock is set to **In**. In that case, the field becomes a display that shows the current decimated rate.

Bit Clk Rate:

This is a reading field. The rate of the Bit Clock (which is Decimated Rate × interpolation ratio, the factor set in Interpolation) is displayed here.

Modulator

Select the PDM modulator with this control.

APx can use a 4th-order or 5th-order PDM modulator. A 4th-order modulator is commonly used in MEMS microphone systems, and is the default. The 5th-order modulator provides better noise and distortion performance at most levels and can be selected as an alternative. See More about PDM on page 160.

Auto OSR

APx500 provides four 4th-order and four 5th-order modulators, each optimized for one of four common oversampling ratios (x64, x128, x256 and x512). When **Auto OSR** is selected, these modulators are mapped to other oversampling ratios as shown in the table above. You can force a different mapping by selecting one of the OSRs in the drop-down menu. The current OSR and the current modulator version are shown in the Status Bar.

Data Edge:

One channel of audio data can be carried on each edge (the rising edge or the falling edge) of the clock signal. Select **Both (Stereo, LR)**, **Both (Stereo, RL)**, **Rising** or **Falling** for the PDM transmitter setting.

This selection affects the APx output channel count. For PDM, there are 2 output channels when Data Edge is set to either **Both** setting, and 1 output channel when Data Edge is set to **Rising** or **Falling**. **Both (Stereo, LR)** routes the Channel 1 (left) audio to the rising edge, and the Channel 2 (right) audio to the falling edge; **Both (Stereo, RL)** reverses this.

Channels:

This control is unavailable, and displays the input channel count as set by the Data Edge control, above.

Scale Freq By:

Note: in PDM Output Settings, sample rate (SR) is referred to as Decimated Rate.

- **Decimated Rate**
  For most testing, use **Decimated Rate** (the default).
- **Fixed**
  If you are applying jitter to the output clock at high levels, you may want to reference the sampled audio data to a fixed rate, rather than the jittered output rate. This may improve phase and frequency stability of the embedded audio. Set **Scale Freq By:** to **Fixed**, and enter the nominal decimated rate in the **Fixed Rate** field.

See More About Frequency Scaling in APx on page 72 for a detailed explanation.

Logic Level

This control sets nominal logic level for the PDM bitstream, and the Bit Clock level when Bit Clock is an output. See the table below for voltage ranges.

Vdd Level

This control sets the nominal DC voltage at the **Vdd Supply** BNC connector on the PDM module. See the table below for voltage ranges.

Limit Logic Level

When this box is checked, the maximum **Logic Level** is limited to the current **Vdd Level**, but cannot be less than 0.8 V. When unchecked, Logic Level is independent of Vdd Level.

Additionally, when this box is checked, the minimum Vdd Level is 0.8 V; when unchecked, the minimum Vdd Level is 0 V.

Vdd ON/OFF

This switch turns the Vdd DC power supply **On** or **Off**.

The **Vdd ON/OFF** switch appears here on the PDM Output Settings panel, and also on the PDM **Input Configuration** and **Settings** panels.

Voltage levels by PDM hardware and firmware versions

Module model number	*PDM hardware revision*	*Firmware revision*	*Vdd range*	*Logic level range*	*Jitter support*
215	000	1.00	0.8 V to 3.6 V	1.8 V to 3.3 V	No
215	000	1.10	0.8 V to 3.6 V	1.8 V to 3.3 V	No
215	001, 002	2.00	0.0 V to 3.6 V	1.8 V to 3.3 V	No
228	101	4.00	0.0 V to 3.6 V	0.8 V to 3.3 V	Yes*
228	200	4.00	0.0 V to 3.6 V	0.8 V to 3.3 V	Yes*

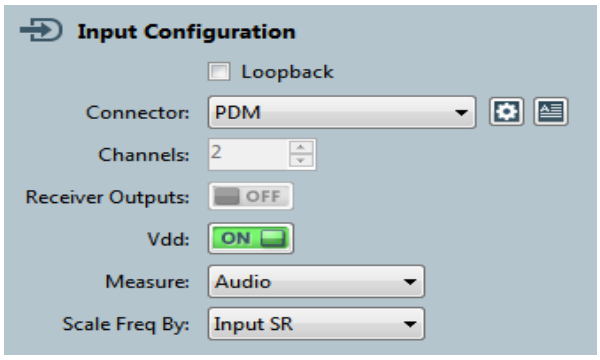
* requires AMC.

PDM Clock Rates by Logic Level

The low voltage logic families slow down dramatically at logic levels below 2 volts, resulting in lower maximum achievable clock rates at the lower logic levels. The table below shows the maximum clock speeds for error-free performance at four logic level voltage ranges.

Voltage Minimum	*Voltage Maximum*	*Maximum Frequency*
0.8 V	<1 V	3.072 MHz
1 V	<1.5 V	6.144 MHz
1.5 V	<2 V	12.288 MHz
2 V	3.3 V	24.576 MHz

PDM Input Configuration



Note: APx measurements are all performed on the baseband audio signal, at the Decimated Rate. However, it is possible to directly view the undecimated PDM bitstream in the Signal Analyzer, when the Signal control is set to PDM Bitstream.

These controls are shown on Signal Path Setup when **PDM** is selected as the Input Connector:

Settings

The Settings button opens the Input Settings (PDM) dialog, which reveals more PDM settings and readings.

Receiver Outputs (ON/OFF)

When the PDM Input Bit Clock Direction is set to **Out** (see following), you can switch the receiver outputs **ON** or **OFF** here.

Vdd (ON/OFF)

APx can supply operating power to a PDM device under test, available on the PCM module at the Vdd Supply BNC connector. See Vdd Level, following.

This switch turns the Vdd DC power supply **ON** or **OFF**.

The **Vdd ON/OFF** switch appears on the PDM Output Settings panel, and also on the PDM Input Configuration and Settings panels.

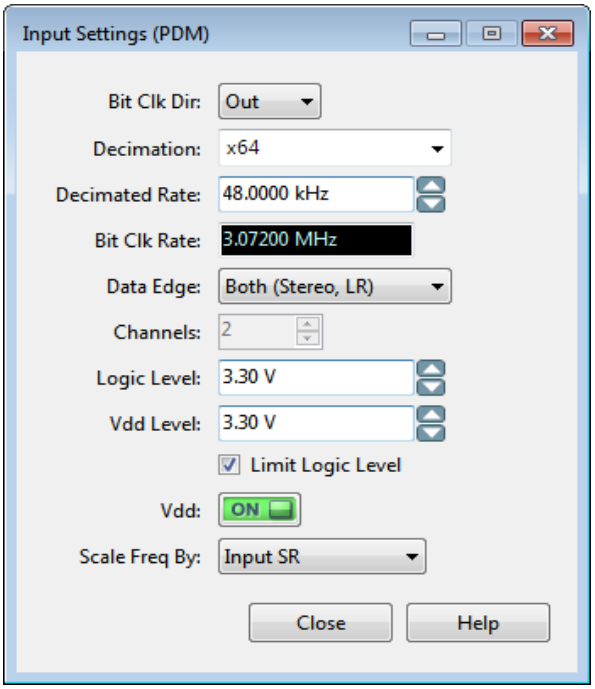
Measure

The Measure menu is only available for a jitter-enabled Input/Output module when the Advanced Master Clock is installed in the analyzer.

- **Audio**
  This is the normal setting. The embedded audio is routed to all measurements and meter readings.
- **Jitter (UI)**
  This is the jitter setting. The jitter signal is extracted from the carrier and routed to all measurements and meter readings. Results are shown in Unit Interval (UI) units.
- **Jitter (sec)**
  This is an alternative Jitter setting. The jitter is measured as above, but the results are shown in seconds (s) units.

Read more about jitter on page 69.

Input Settings (PDM)



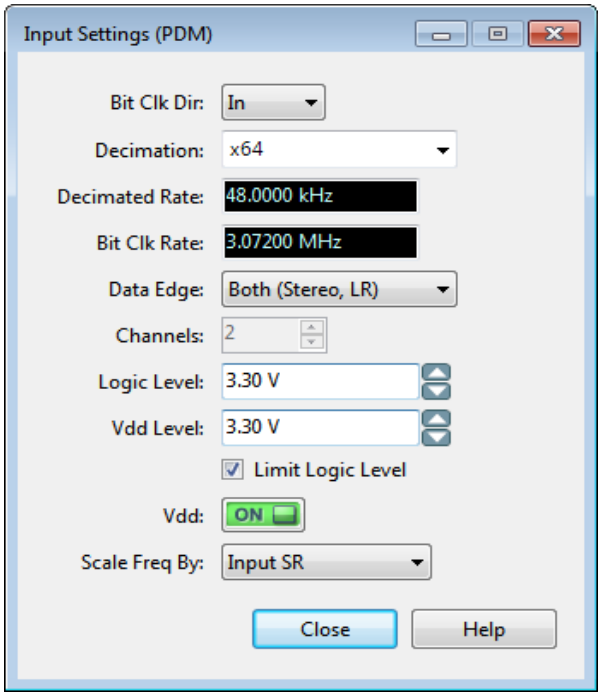
Input Settings with Bit Clock set to Out (default).

These controls are in the Input Settings (PDM) dialog, available by clicking the Settings button from Signal Path Setup, when Input Configuration is set to PDM.

Bit Clk Dir:

The Bit Clock connector associated with the PDM Input on the PDM module can be configured as an input or an output. Choose a clock direction by selecting **Bit Clk Dir: Out** (the default), or **In**.

A typical use case provides that a downstream PDM receiver (the APx PDM input) is connected to a MEMS microphone PDM output (the DUT), and that the receiver also provides the clock to the MEMS module. In this case the APx input-associated clock is set to **Out**, where APx is the master and the DUT is the slave. This is the default clock direction setting.



Input Settings with Bit Clock set to In.

Jitter

A PDM model 228 module installed in an APx with the Advanced Master Clock (AMC) can have jitter applied to the bit clock when PDM is an input and **Bit Clk Dir:** is set to **Out**. In **Signal Path Setup > Clocks > Jitter Generator**, select **Apply To: Digital Input** and turn jitter **ON**.

Alternatively, incoming jitter can be measured when PDM is an input and **Bit Clk Dir:** is set to **In**. In **Signal Path Setup > Input/Output** and **Input Connector** is set to **PDM**, select a **Jitter** choice from the **Measure** menu.

For more information see Jitter Support in APx beginning on page 71.

Decimation:

Set the receiver decimation factor here.

APx provides a wide range of decimation settings to accommodate common ratios. The supported Decimated Rates (baseband audio sample rates) are constrained by the decimation setting, as shown in the table here. Power-of-two rates are shown in bold type.

Note: the x1 setting does not decimate the PDM signal. The undecimated PDM bitstream is presented to the APx measurements.

Decimation Factor	*Maximum Decimated Rate*
x1 (Bitstream)	216 kHz
x3.125	216 kHz
x4	216 kHz
x6.25	216 kHz
x8	216 kHz
x8.33	216 kHz
x10.67	216 kHz
x12.5	216 kHz
x15.625	216 kHz
x16	216 kHz
x16.67	216 kHz
x18.75	216 kHz
x21.33	216 kHz
x24	216 kHz
x25	216 kHz
x31.25	216 kHz
x32	216 kHz
x33.33	216 kHz
x37.5	216 kHz
x42.67	216 kHz
x46.875	216 kHz
x48	216 kHz
x50	216 kHz
x62.5	216 kHz
x64	216 kHz
x66.67	216 kHz
x75	216 kHz
x85.33	216 kHz
x96	216 kHz
x100	216 kHz
x125	196.608 kHz
x128	192 kHz
x150	163.84 kHz
x192	128 kHz
x200	122.88 kHz
x250	98.304 kHz
x256	96 kHz
x300	81.92 kHz
x384	64 kHz
x400	61.44 kHz
x500	49.152 kHz
x512	48 kHz
x600	40.96 kHz
x768	32 kHz
x800	30.72 kHz

Decimated Rate:

This control is available when the Bit Clock connector associated with the PDM input is set to **Out**. The clock direction is set with the Bit Clk Dir control, above.

The Decimated Rate control specifies the decimated sample rate (the baseband audio sample rate) by setting the PDM input Bit Clock output rate. The relationship between the decimated rate and the bit clock rate depends upon the oversampling ratio, set by the Decimation control, above. For example, a specified decimated rate of 48 kHz and an oversampling ratio of 64 would set the PDM Bit Clock out to 3.072 MHz (48000 × 64 = 3072000).

To set the decimated rate, click the up/down arrows to select a standard rate, or enter an arbitrary value in the Decimated Rate field. Maximum decimated rate is shown in the above table. Minimum supported decimated rate is 4 kHz. Default is 48 kHz.

This control is unavailable when the PDM input Bit Clock is set to In. In that case, the field becomes a display that shows the current decimated rate.

Bit Clk Rate:

This is a reading field. The rate of the Bit Clock (which is Decimated Rate × decimation ratio, the factor set in Decimation) is displayed here.

Data Edge:

One channel of audio data can be carried on each edge (the rising edge or the falling edge) of the clock signal. Select Both (Stereo), Rising or Falling for the PDM receiver setting.

This selection affects the APx input channel count. For PDM, there are 2 input channels when Data Edge is set to either **Both** setting, and 1 input channel when Data Edge is set to **Rising** or **Falling**. **Both (Stereo, LR)** routes the data on the rising edge to Channel 1 (left), and the data on the falling edge to channel 2 (right); **Both (Stereo, RL)** reverses this.

Channels:

This control is unavailable, and displays the input channel count as set by the Data Edge control, above.

Logic Level

This control sets nominal logic level for the PDM bitstream, and the Bit Clock level when Bit Clock is an output. See the table on page 157 for voltage ranges.

Vdd Level

This control sets the nominal DC voltage at the **Vdd Supply** BNC connector on the PDM module. See the table on page 157 for voltage ranges.

Limit Logic Level

When this box is checked, the maximum **Logic Level** is limited to the current **Vdd Level**, but cannot be less than 0.8 V. When unchecked, Logic Level is independent of Vdd Level.

Additionally, when this box is checked, the minimum Vdd Level is 0.8 V; when unchecked, the minimum Vdd Level is 0 V.

Vdd (ON/OFF)

This switch turns the Vdd DC power supply **On** or **Off**.

The **Vdd ON/OFF** switch appears on the PDM Output Settings panel, and also on the PDM Input Configuration and Settings panels.

Scale Freq By:

Note: in PDM, sample rate (SR) is referred to as Decimated Rate.

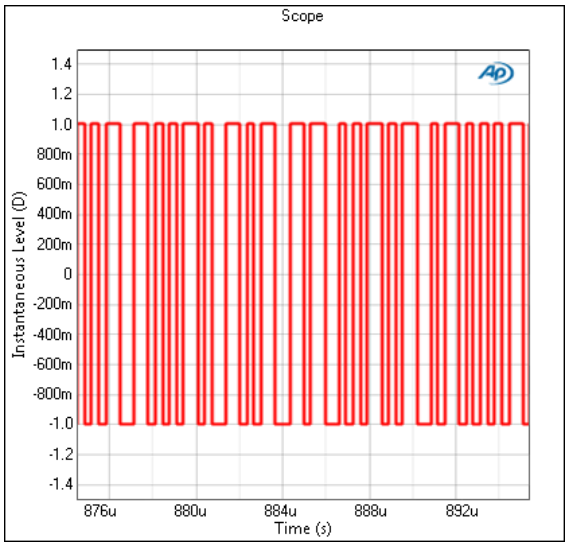
- **Input SR**
  For most testing, use **Input SR** (the default).
- **Output SR**
  If sample rate is being swept, you may want to use the output sampling rate as the scaling reference.
- **Fixed**
  If jitter is present on the input clock, you may want to reference the sampled audio data to a fixed rate, rather than the jittered input rate. This may improve phase and frequency stability of the embedded audio. Set **Scale Freq By:** to **Fixed**, and enter the nominal input sample rate in the **Fixed Rate** field.

See More About Frequency Scaling in APx on page 72 for a detailed explanation.

More about PDM

PDM is Pulse Density Modulation, a system for representing a sampled signal as a stream of single bits using delta sigma conversion (sometimes called sigma delta conversion). PDM is the technology used in many oversampling ADCs and DACs, and is the basis of the

Sony/Philips commercial digital format and disc trade named DSD and SACD, respectively.



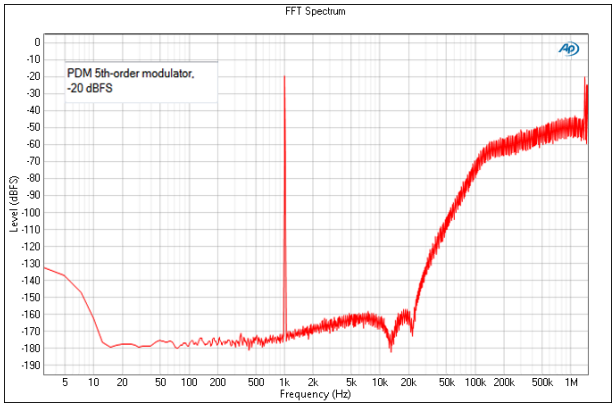
PDM bitstream, time domain view.

To configure the Signal Analyzer to display the PDM bitstream as shown above, see page 489.

A one-bit stream is unacceptably noisy, but very high sampling rates and noise shaping techniques are used to greatly reduce the noise in the audio spectrum. This noise energy is moved from the audio baseband into the area of the spectrum above 20 kHz, where it is inaudible.

Why PDM?

Properly done, PDM can digitally represent high quality audio, and is inexpensive and easy to implement. For these reasons, a PDM stream is now commonly used as the data output of a MEMS (Micro Electro-Mechanical System) microphone. MEMS microphones can be made to be very small and are inexpensive to implement on silicon chips, and are found in many small devices such as cell phones or Bluetooth devices.



Spectrum of PDM bitstream with 5th-order modulator, –20 dBFS

To configure the Signal Analyzer to display the PDM bitstream as shown above, see page 489.

Analog to PDM to Analog

An analog signal can be directly sampled at a high sampling rate (several megahertz or more) and converted to a PDM stream. The PDM signal can be converted back to analog audio by passing it through a low-pass filter.

PCM to PDM

A signal that is coded as PCM (pulse code modulation, the coding widely used in digital audio) can be converted to PDM by interpolating it and reducing the word length to one bit. The ratio of the interpolated PDM bit rate to the PCM sample rate is called the oversampling ratio.

PDM to PCM

A signal that is coded as PDM can be converted to PCM by decimating it and increasing the word length. As mentioned above, the ratio of the PDM bit rate to the decimated PCM sample rate is called the oversampling ratio.

PDM modulator

PDM modulators have a series of integrators, or accumulating nodes. The “order” of a modulator refers to the number of its integrators.

Higher-order modulators provide improved noise and distortion performance over much of the amplitude range, but suffer from increased instability at amplitudes approaching maximum level. To allow full amplitude levels without driving a PDM modulator into instability, limiting techniques are typically employed. In APx, an Overload indicator (shown in the Status Bar) is activated when the modulator exceeds the defined overload point. For the APx 4th-order modulator, the overload point is about –7.5 dBFS. For the 5th-order modulator, the overload point is about –9 dBFS. Limiting is used above this point, enabling both the 4th and 5th order modulators to operate to maximum level without instability. However, noise and distortion rise for signals above the overload point. In practical applications, such performance is deemed acceptable, since voice signals at maximum levels are typically brief transients. In APx, modulator distortion at maximum level is about 1% THD+N.

MEMS microphones

A MEMS microphone system consists of the MEMS sensor, an analog preamplifier, and a PDM modulator, often implemented on a small silicon chip. DC power and a clock signal at the sampling rate are typically provided by downstream devices.

Manufacturers of such devices typically use 4th-order sigma delta modulators at a clock frequency of 3.072 MHz, which in the receiver is typically decimated by a factor of 64 to a baseband sampling rate of 48 kHz.

Control Codes

Some PDM devices can send or receive control codes. A control code is a 2-digit hex number (8 bits) that is inserted into the PDM bit stream, replacing the audio for the duration of the control code sequence. Well-behaved devices will prevent the codes from being played as audio. Control codes are often repeated many times to assure identification and reception.

Control codes are typically used to configure a downstream PDM device. If the device is an amplifier, the code may request a mute or an unmute, or adjust the volume, or request that the device power down.

The codes and their interpretations are not standardized and are specific to manufacturers and devices.

APx implementation

The PDM Monitor reads control codes and provides access to the Send PDM Control Codes dialog. See pages 37 and 38.

Transducer Interface

The APx1701 Transducer Interface

The Signal Path Setup Transducer Interface Outputs and Inputs only function if there is an APx1701 Transducer Test Interface connected to the system. See page 12.

The APx1701 Transducer Test Interface is an APx accessory device with both input and output functions. The APx1701 is primarily designed to drive loudspeakers and headphones in acoustic testing, to measure loudspeaker impedance curves, and to power and accommodate both measurement microphones and microphones under test. More detailed hardware information, interconnection diagrams and use case diagrams are provided in the *APx1701 Transducer Test Interface Installation Instructions, Specifications and User's Guide* booklet, provided with the interface and available for download from the Audio Precision Web site at ap.com.

Output Functions

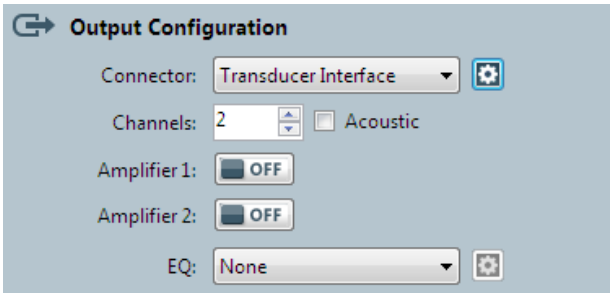
For output measurement functions, the APx1701 provides an audio power amplifier that stands between a driven transducer (typically a loudspeaker system or driver) and the APx analyzer analog outputs. Additionally, a sense resistor and switching are provided for driver impedance measurements.

Input Functions

For input measurement functions, the APx1701 is connected between transducer outputs (typically microphones) and the APx analyzer analog inputs. The APx1701 can provide CCP or Phantom powering to attached microphones. TEDS serial number and sensitivity data can be passed from a TEDS microphone to APx500.

Output Configuration

Output: Transducer Interface



This selection enables the APx1701 Transducer Test Interface output, if available.

The **Settings** button opens an output **Settings** dialog, offering more detailed control. See Settings for Transducer Interface Outputs on the next page.

Channels

The Channels setting allows you to set the number of output channels to be tested for this signal path. Although the APx1701Transducer Test Interface only has two amplifier channels, additional APx analog output channels are available for other use (if your hardware supports more than two channels). Note that the gain scaling (see below) imposed when using the Transducer Interface output affects all analyzer output channels.

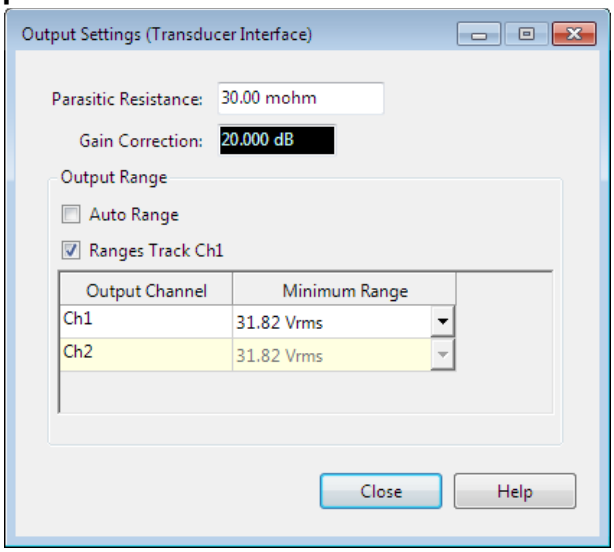
Acoustic

Read more about output **Acoustic** mode in the **Input/Output** topics in Chapter 7, Signal Path Setup.

EQ

Read more about output **EQ** in the **Input/Output** topics in Chapter 7, Signal Path Setup.

Settings for Transducer Interface Outputs



Note: when using the APx1701 Transducer Test Interface, you must interconnect the APx1701 with and your APx analyzer system before proceeding. See Interconnecting APx1701 with an APx Analyzer.

Source Impedance

When the APX1701 is the analyzer output connection, the unbalanced analog output source impedance is fixed at 50 Ω.

Parasitic Resistance

The resistance in a circuit that is present but not desirable for the intended design is called parasitic resistance. In the case of the output circuits of the APx1701, the sense resistor and the DUT load are design elements, but any resistance in circuit connections (including internal conductor resistance, SpeakON connector resistance, and the resistance in the wires connecting the DUT) is parasitic resistance. Parasitic resistance will affect impedance and Thiele-Small results.

For many use cases, the parasitic resistance is so low that its effect on measurements is insignificant, and using the default value of 0.030 ohms (30 mΩ) will not adversely affect results. However, in cases where extreme precision is required, or where the external connection resistance is high (long or small-gauge speaker wires), you must measure and enter the parasitic resistance for best results.

Refer to the *APx1701 Installation, Specifications and User's Guide* booklet for information about determining output circuit parasitic resistance.

Gain Correction

The APx1701 has a fixed gain of 20 dB (a voltage gain of 10). When Output Configuration > Connector is set to Transducer Interface, the generator levels in the APx500 user interface are scaled to accommodate this gain.

For example, if a generator level is set to 10 V rms, the voltage at the APx1701 amplifier output will indeed be 10 V rms, but the voltage at the analyzer analog output will be 20 dB lower, or 1 V rms.

This user interface scaling is in place for all analog outputs on all channels when the Transducer Interface is the Output Connection.

Output Range

Normally, APx generator outputs are automatically switched to an optimal gain range for a given output level. This provides the best generator output performance, in terms of noise and distortion residuals.

However, the automatic range-switching can produce minor transients which can be annoying when amplified and played through a loudspeaker. When Output Configuration > Connector is set to Transducer Interface, you have the option of disabling the auto-ranging, and instead setting a fixed output range to eliminate audible transients. For acoustic testing, where environmental noise and DUT distortion are often orders of magnitude above APx generator performance, setting a fixed range should not affect results.

Auto Range

When this box is unchecked, (the default), the APx generator minimum output range is fixed at the value set in Minimum Range. This can eliminate audible range-switching transients.

When this box is checked, the APx generator output will automatically switch output gain ranges for best performance.

Note: Fixed Transducer Interface output ranging is not available on the APx555 when the generator is set for High Performance Sine or Burst Sine, or when Square Wave or DIM waveforms are selected. It is also not available on the APx525 or 526 with AG52 Option, when Square Wave or DIM waveforms are selected.

Ranges Track Ch1

When this box is checked, the Minimum Range set for channel 1 will be used as the Minimum Range for all channels. Uncheck the box to enter different ranges for different channels.

What is Minimum Range?

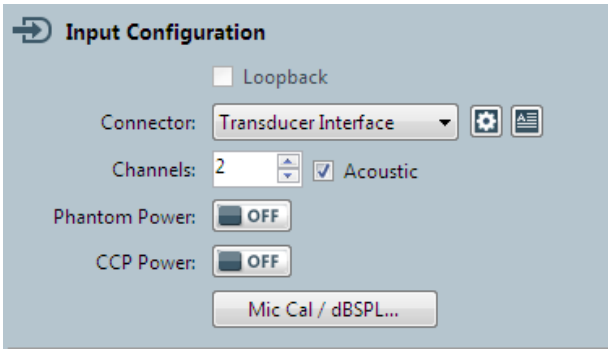
When setting a fixed range, first determine what the maximum level will be for your test. Then set Mini-

mum Range to a value somewhat above your maximum level. The generator range will not be automatically switched as long as project generator levels do not exceed this value. If at run time a generator level exceeds the value in Minimum Range, the generator output range will be switched to the next higher range to accommodate the generator level.

Since, as mentioned above, generator noise and distortion residuals are typically far below real test conditions, in the vast majority of cases we recommend that you set a fixed level Minimum Range to the highest value possible for your APx analyzer.

Input Configuration

Input: Transducer Interface



Note: when using the APx1701 Transducer Test Interface, you must interconnect the APx1701 with and your APx analyzer system before proceeding. See detailed information in the APx1701 Transducer Test Interface Installation, Specifications and User's Guide *booklet.*

The APx1701 is designed to connect directly to balanced or unbalanced microphones, for use of measurement microphones for loudspeaker or headphone testing, or to test a microphone as a DUT. The audio signal from the microphone is not altered, passing through the APx1701 on electrical conductors and relays but not amplified or filtered in any way. The Pass Thru microphone signal(s) from the APx1701 are connected to one or two analyzer balanced analog inputs.

When the APX1701 is the analyzer input connection, the balanced analog input termination is fixed at 200 kΩ.

Loopback

Loopback is not available for the APx1701 Transducer Test Interface.

The **Settings** button opens an output **Settings** dialog, offering more detailed control. See below.

Channels

The **Channels** setting allows you to choose the number of input channels to be tested for this signal path.

Acoustic

Read more about input **Acoustic** mode in the **Input/Output** topics in Chapter 7, Signal Path Setup.

Phantom Power

+48 V phantom power can be provided to the APx1701 balanced microphone inputs.

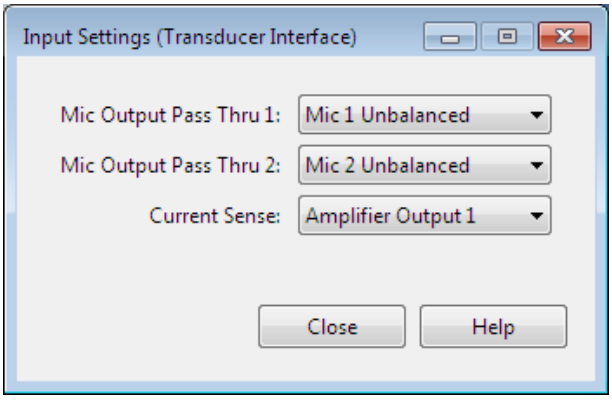
CCP Power

+24 V CCP (constant current power) can be provided to the APx1701 unbalanced microphone inputs.

Mic Cal / dBSPL...

This button opens the Microphone Calibration / Set dBSPL1, dBSPL2 dialog. See pages 60 and 64.

Settings for Transducer Interface Input



Mic Output Pass Thru 1

This menu selects the APx1701 microphone input to appear at the Microphone Output (Pass Thru) connector 1. The choices are:

- **Mic 1 Balanced**
- **Mic 2 Balanced**
- **Mic 1 Unbalanced**
- **Mic 2 Unbalanced**
- **Mic 1 CCP Self-Test**

Mic Output Pass Thru 2

This menu selects the APx1701 microphone input to appear at the Microphone Output (Pass Thru) connector 2. The choices are:

- **Mic 2 Balanced**
- **Mic 2 Unbalanced**
- **Mic 2 CCP Self-Test**

See the *APx1701 Transducer Test Interface Installation, Specifications and User's Guide* booklet for information about performing a CCP Self-Test.

Note: power for these microphones (phantom or CCP) is enabled using the switches on the Input Configuration panel.

Current Sense

There is a 0.1 Ω current sense resistor in the output circuit of each APx1701 amplifier channel. The voltage across this resistor can be measured by the APx analyzer to calculate the current flowing through the amplifier output circuit.

This menu selects which amplifier channel current sense resistor is connected to the Current Sense connector for measurement. The choices are:

- Amplifier Output 1
- Amplifier Output 2

Microphone Powering and the APx1701

In electronics, condenser and capacitor are interchangeable terms. In practice, circuit elements are typically called capacitors, and microphones that have capacitors as their transducer element are typically called condenser microphones.

Condenser microphones use a capacitor as the transducer element. All condenser microphones require powering of some sort. Some designs need a polarizing voltage for the transducer element, and all designs need power for the small amplifier and associated circuitry attached to the transducer.

Measurement microphone CCP powering

Measurement microphones that are pre-polarized (using an electret condenser for the transducer element) are typically powered by a constant current supply (CCP) power system. The APx1701 can provide 4 mA (+24 V CCP) power to such microphones when connected to the unbalanced microphone inputs.

Measurement microphones that require a polarization voltage for the transducer element are typically fitted with a multi-pin connector and are provided with a proprietary power supply from the manufacturer. The APx1701 can use such microphones, but cannot power them. You must provide a power supply.

Studio microphone phantom powering

Condenser microphones designed for recording and sound reinforcement applications are invariably “balanced” devices, and use a powering system called phantom power. A phantom powered microphone may be pre-polarized, or may develop the polarization voltage internally from the phantom power. The APx1701 can provide phantom power to such microphones when connected to the balanced microphone inputs.

In the APx1701 Transducer Test Interface, phantom power is +48 V dc, applied to XLR pins 2 and 3 via 6.8 kΩ resistors, using the grounded shield connection on pin 1 as the return.

CCP Self-Test

You can view the DC voltage applied by the CCP power feature at either of the APx1701 unbalanced microphone inputs, and observe the change in voltage when a load (the CCP microphone) is attached.

In the Settings for Transducer Interface Inputs dialog, choose Mic (1 or 2) CCP Self-Test from the Mic Output Pass Thru (1 or 2) menu. This connects the CCP dc voltage at either of the unbalanced microphone inputs to the associated Mic Output Pass Thru connector. The voltage can be measured and displayed in APx using the DC Level measurement in Sequence Mode, or a DC Level meter in Bench Mode.

When an unbalanced microphone input has CCP applied but is unloaded (no microphone connected), a typical DC Level reading is between +24 Vdc and +25 Vdc. With a CCP microphone connected, the loaded DC Level reading will be lower. We have observed voltages at approximately +13 Vdc and approximately +4 Vdc, when loaded with microphones from different manufacturers.

ASIO I/O

ASIO Output and Input Configurations

For the APx515, this interface requires a software option key. See page 189 for more information about software options.

To connect APx to an ASIO device, first be sure that you have the current manufacturer's ASIO driver for your device properly installed on the PC that is running APx500, and that you have the device connected to the PC and turned on.

An ASIO device that uses USB for its physical interface should not be connected to the PC through the same root USB hub device as the APx instrument, to avoid bandwidth contention between the APx instrument and ASIO device.

Go to Signal Path Setup and select ASIO as an output or input configuration. Then select your device from the ASIO Device list.

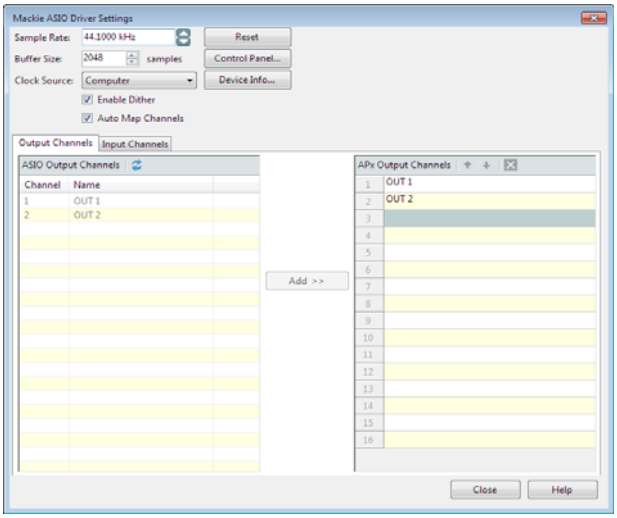
Note: in Demo Mode you can select Demo Device to inspect the ASIO User Interface.

Although more than one ASIO device can be simultaneously connected to the PC and displayed in the ASIO Device list, only one device can be selected as an active, connected device for each APx Signal Path. A different ASIO device can be selected for other Signal Paths within a Project.

Click the **Settings** button to make additional settings for your ASIO device.

ASIO Settings

There is one Settings dialog for both ASIO output and ASIO input. Click the **Output Channels** or **Input Channels** tab on the ASIO channel list table to view the corresponding channel mapping.



The ASIO Settings dialog provides a panel to set buffer size and clock source, and to map the ASIO device output and input channels to the APx analyzer output and input channels. From this dialog, you can also open the device control panel (when supported), and view device information (when supported).

Sample Rate

This field presents you with the supported sample rates in the currently selected ASIO audio device. Changing the sample rate here will issue a command to the ASIO device to change its sample rate.

Reset

Click **Reset** at any time to re-initialize and refresh the connection. The **Refresh Device Channels** button on the ASIO channel list tables has the same function.

Buffer Size

This field presents you with the supported buffer sizes in the currently selected ASIO audio device. Changing the buffer size here will issue a command to the ASIO device to change its buffer size.

Control Panel

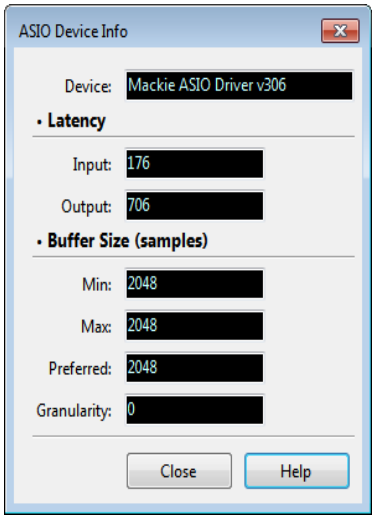
Most ASIO devices provide a control panel where other parameters and readings can be addressed. Some devices provide access to their control panels through an ASIO-specified protocol. In such cases, clicking the Control Panel button will open the device control panel. Other devices may provide control panels that must be accessed in some other way, such as launching a mixing application, etc.

Clock Source

This drop-menu presents you with a list of clock sources supported by the currently selected ASIO audio device. Selecting a clock source here will issue a command to the ASIO device to use that clock source.

Device Info

Most ASIO devices provide device information that can be accessed through an ASIO-specified protocol. In such cases, clicking the Device Info button will populate the APx ASIO Device Info panel with values.



The Device info panel displays this information:

Device

The device manufacturer, model, serial number, etc. can be listed here.

Latency (samples):

Latency is the delay through the ASIO device, which includes both the physical sound device and delay occurring in consequence of PC activity such as disk access, buffering, etc. Latency is reported in samples.

Input

This is the input latency reported by the ASIO driver.

Output

This is the output latency reported by the ASIO driver.

Buffer Size (samples)

A certain amount of PC memory is allotted to the ASIO device driver to ensure that PC interrupts will not adversely affect the ASIO data. Smaller buffer sizes will decrease latency but will present a higher risk of dropped samples. Larger buffer sizes will increase latency but provides a lower risk of dropped samples. Buffer size is reported in samples, and therefore latency will change with a sample rate change.

Min:

This is the minimum buffer size supported by the ASIO driver.

Max:

This is the maximum buffer size supported by the ASIO driver.

Preferred:

This is the default buffer size of the ASIO driver.

Granularity:

This is the degree of resolution supported by the ASIO driver in specifying buffer size. This is typically “powers of 2” (32, 64, 128, etc.), which is indicated by the ASIO signifier “–1”.

Enable Dither

When the **Enable Dither** checkbox is checked ☑, dither is **ON** (the default). When **ON**, dither is set according to the bit depth, using TPDF (triangular probability density function) dither at ±1 LSB.

Uncheck ☐ the checkbox to turn dither **OFF**.

Auto Map Channels and Channel List tables

The APx ASIO interface supports mapping of up to 16 output and 16 input channels.

The channel list table to the left shows the available device output or input channels, and the channel list table to the right shows the corresponding APx output or input channels. These lists work in conjunction with the **Auto Map Channels** checkbox as follows:

When **Auto Map Channels** is checked ☑ (the default), APx will map all device output and input channels (up to 16) to the 16 APx output and input channels, in the order provided by the device driver.

When the **Auto Map Channels** ☐ checkbox is unchecked, APx will map the first device output and input channel to the first APx output and input channel. You can then highlight more device channels and use the **Add >>** button to map them to the APx outputs or inputs.

Moving or deleting mapped channels

Uncheck ☐ **Auto Map Channels** to move or delete mapped channel connections in the APx output or input channel list.

Select a channel and use the **Move Selected Channel Up** or **Move Selected Channel Down** buttons to change the mapping. Use the **Delete Channel** button to remove a channel from the mapping.

More about ASIO

ASIO is a license-free sound device protocol for Microsoft Windows that provides a high-quality low-latency connection to an internal sound card or external audio device (typically via USB or IEEE 1394 Firewire interface). Other common interfaces such as PCI, Ethernet, etc. are supported. The APx500 implementation enables input and output through any ASIO-compatible device. ASIO was developed and specified by Steinberg GmbH.

ASIO avoids the intermediary layers of the Windows operating system, bypassing its relatively poor quality sample rate conversion and mixing capabilities and allowing applications to have bit exact access to the audio streams from a large number of independent channels from a sound card. ASIO supports the use of 24 bit samples and sample rates up to 192K samples per second.

ASIO devices vary in their compatibility with other ASIO systems. Features such as channel count, sample rate, bit depth and overall audio quality also vary widely. Properly implemented, ASIO devices can be bit-accurate while signals remain unmolested in the digital domain; whether or not a particular device achieves this performance depends upon its design.

The APx ASIO I/O capability was added as a means to test ASIO devices, where the ASIO device itself is the DUT. Of course, an ASIO device can serve as an output for the APx generator, or as an input for the APx analyzer. For test and measurement applications, the audio quality of any ASIO device will always be inferior to the APx hardware input and outputs in the analyzer hardware, and will degrade your measurements significantly. We never recommend using an ASIO device as a replacement for the Audio Precision instrument.

Open Loop measurements

Overview

Most APx measurements, whether in Sequence Mode or Bench Mode, are made in the Closed Loop configuration, where the APx generator output is connected to the inputs of the DUT, and the outputs of the DUT are connected back to the analyzer.

However, this is not possible with playback-only devices such as CD or DVD players and personal audio players, or with received broadcast signals or split-site signals carried over a long distance.

APx has solutions for these circumstances, the **External Source** and **File Input** configurations.

External Source configuration

To enter External Source configuration, go to the Signal Path Setup Input/Output menu, and select **None (External)** under Output Configuration. This configures APx to make measurements without using the internal generator.

Compatibility

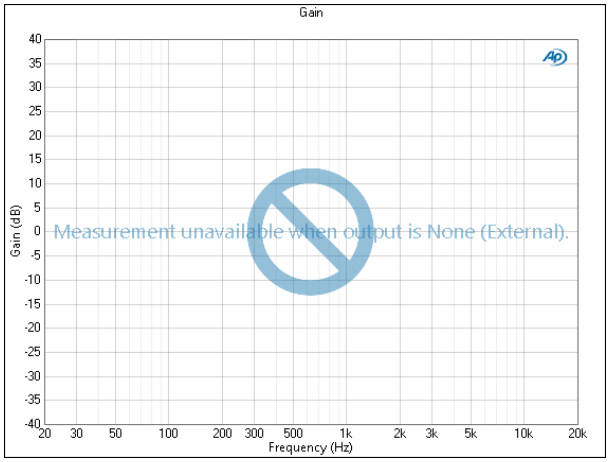
Some measurements are entirely compatible with External Source configuration, and no special considerations are required. For example, many meter measurements such as Level, Phase, Frequency and so on will operate perfectly with or without a signal from the APx generator.

Other measurements, while compatible, require special attention in External Source, such as the application of a sweep table or a multitone signal definition file.

Measurements based on continuous sweep (Continuous Sweep, Frequency Response, Acoustic Response, Impedance/Thiele-Small, Loudspeaker Production Test) require a synchronization between generator and analyzer, and are not compatible with open-loop measurements.

Level sweeps and regulated measurements, some crosstalk measurements, and any gain result depend upon a dynamic relationship between generator and analyzer, and are not compatible with open-loop measurements. Some batch measurements depend upon pre-loaded files (POLQA Averaged, for example), and are not compatible with open-loop.

When Output Configuration is set to **None (External)**, incompatible measurements and results are displayed with text and a mark, like this:



Using the External Source configuration

Setup

In Signal Path Setup Input/Output, go to Output Configuration and select **None (External)** as the Output Connector. This will set the project signal path to the External Source configuration.

No Generator

In Bench Mode, the Generator panel disappears when in **None (External)**. In Sequence Mode, compatible measurement views will no longer show the Generator, but, will display text information or a drop-down list of known source signals that you must use for the measurement. Other measurements do not require known signals, and the External Source panel provides a brief textual explanation of acceptable sources.

File Input Configuration

File Input is another open-loop configuration, where the signal from the APx Generator is not routed through a DUT and then into the Analyzer input.

APx500 software can analyze the data in an audio file accessible to the PC connected to the instrument. This is useful for analyzing DUTs that present their output as an audio file, or for remotely analyzing an audio signal.

Applications

There are a number of applications for File Input.

- The WAV files may be created and saved by the DUT during the test, using the APx500 generator as a signal source. This is typical of DUTs like a digital recorder or a sound card.
- The files may be output from a decoder device testing coded audio streams.
- The files may be from a remote testing session, saved by the Measurement Recorder, for example, to pass on a recording of the analog output of a DUT for further analysis.
- In a sequence, it may be useful to analyze the same audio data more than once using different parameters, such as filtering.

A colleague separated by time or distance could record the output of a DUT to a digital file (using the APx Save to File function or a third-party recorder) for later analysis.

A digital recorder's recording function could be analyzed as the DUT, by using its output file as an analyzer input.

A complex system such as a smart phone can be tested using a combination of External Source and File Input techniques, as described in the Audio Precision Technote 120.

Measurements that support File Input Configuration

All measurements that support the External Source configuration also support File Input configuration, with one exception: the Metadata Recorder. This is because linear WAV files do not contain metadata of interest.

Supported audio file formats

See page 614 for a table of supported audio file formats for file input analysis of audio files.

Setup

In Signal Path Setup, go to **Input Configuration: Connector** and choose **File (Analog Units)** or **File (Digital Units)** from the drop-down list.

Both of these menu choices configure APx500 for a WAV file input, but with different units selected for the results displays.

File (Digital Units)

File (Digital Units) displays the audio data level in the file in digital units.

File (Analog Units)

When analyzing a WAV file that represents an analog measurement acquisition, it is often convenient to view the audio level in the file in the original analog units. This requires scaling the file's digital levels to correspond to the analog levels represented.

Pre-scaled files

Analog measurement acquisitions saved as WAV files by the APx Measurement Recorder contain proprietary metadata that includes this scaling factor, which is used for displaying results when configured for **File (Analog Units)**. You don't need to do any re-scaling.

Unscaled files

Files not recorded by the Measurement Recorder can be scaled by the factor entered in the **Vrms/FS** field. With the default scaling factor of 1.0, for example, a digital level of 1.000 FS (0 dBFS) will be scaled to a display value of 1.000 Vrms.

Operation

Sequence Mode

When **File Input** has been selected in Sequence Mode, each compatible measurement will display a **File List: Choose Files** button to open the **Input File List** dialog. Add the files you want to analyze in the **Input File List**.

When the files are chosen, use the **Analyze** button to make the measurement.

Bench Mode

In Bench Mode, use the **File List** button in Signal Path Setup to open the **Input File List** dialog. The analog scaling field **Vrms/FS** is located just above the **File List** button. Add the files you want to analyze in the **Input File List**.

When the files are chosen, use the **Analyze** in a measurement panel button to make the measurement.

Configuring the DUT

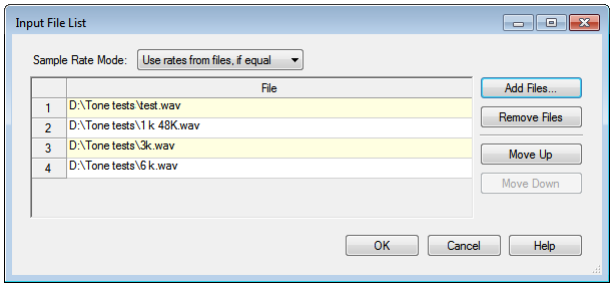
If recording from a DUT, set it up to save WAV files to the location and with the file names specified in the **Input File List**. Turn on the Generator to feed a signal to the DUT, if necessary, and configure the DUT to save the WAV file(s).

If you are measuring an existing file (not recording from the generator but working in External Source), be sure that the file name(s) and location match the name(s) specified in the File List.

Finally, click the **Analyze** button. The first settled reading (typically within the first 0.10 second of the file) is

displayed. To take a reading later in the file, go to the **Settling** tab in **Advanced Settings** or **Settled Readings** and increase the **Delay Time** to the desired point in the file.

Input File List



The **File List...** button opens the **Input File List** dialog.

The Input File List allows you to select WAV files for analysis. Adding a file to this list does NOT load the file into memory, but it inserts a pointer to the file location into the list. You can even add a file name for a WAV file that does not yet exist on the disk (see **Specifying files that do not yet exist**, below). When the **Analyze** button on the measurement panel is clicked, each file is loaded into memory, analyzed, and then cleared from memory.

Sample Rate Mode

You can choose one of three modes to manage the sample rate used to play the input files for analysis.

- **Use rates from files, if equal** (the default)
  In this mode, all files are required to have exactly the same embedded sample rate. Files are played at that rate. An error is flagged during analysis if files of mixed sample rate are selected.
- **Use rate from first file**
  In this mode, all files are played at the sample rate of the first file in the list. If subsequent files have other sample rates, their audio content will be pitch-shifted.
- **Use fixed rate**
  In this mode, all files are played at the sample rate set in the Fixed Rate field. If files have other sample rates, their audio content will be pitch-shifted.

Action Buttons

- The **Add Files...** button opens a file browser for file selection. Select one or more files to add to the list.
- The **Remove Files** button removes highlighted files from the list.
- The **Move Up** and **Move Down** buttons change the list position of a highlighted file.

When all the desired files are in the list, click **OK** to close the dialog and return to the measurement panel, where you can continue and analyze the files.

Specifying files that do not yet exist

It is possible to add to the list WAV files that do not yet exist on the hard drive. This is useful for setting up a test that involves a DUT (such as a digital recorder) that is expected to save a WAV file during the test procedure.

You can specify a file that does not exist by typing in the **File Name:** field in the **Add Files...** browser, or by editing the text in the **Input File List**. Remember to include a valid path to the expected file.

Maximum number of files and channels

The **Input File List** will accept a maximum of 16 files.

Since each file in the **Input File List** could have 1 or 2 channels of audio, it is possible that an acceptable file list (fewer than 16 files) could exceed the analyzer audio channel count maximum, displaying an error message at analysis. If this occurs, remove 1 or more files from the **Input File List**.

Supported file characteristics

- Mono (single channel) or stereo (dual channel) 24-bit WAV files containing linear PCM can be analyzed. Each file, mono or stereo, occupies one line in the list.
- For most measurements, files can have any audio content. For Stepped Frequency Sweep, only compliant sweeps will provide useful results.
- If Sample Rate Mode is **Use rates from files, if equal** (the default), all the files must be of the same sample rate.
- Files can be of any length up to the maximum WAV file length (4 GB). Files can be of mixed length; however, analysis stops at the end of the shortest file.

Errors

Errors may be flagged during analysis:

- if a file on the list does not exist;
- if a file on the list is not a valid linear WAV file;
- if the number of channels in the listed files exceeds 16;
- if files of mixed sample rate are selected, and the Sample Rate Mode is **Use rates from files, if equal** (the default).

External Source Test Signals

audio Precision provides a collection of external stimulus signals in a several different distributions. These include playable media discs and a collection of files

optimized for generator waveform (arbitrary waveform) use with the APx generator. For the stepped frequency sweeps in these collections, Sweep Tables are embedded in the APx software.

The small Audio Precision utility "WfmGeneration.exe" is a convenient way to create any of these files (or a large part of the collection) at a specified resolution and sample rate. WfmGeneration.exe is available as a download from the AP Web site.

Audio Precision Test Signals on Disc

...for External Source use

We provide a full suite of test signals for external source as audio tracks on DVD and audio CD. The audio CD signals are linear audio at 44.1 kHz sample rate and 16-bit resolution; the DVD includes other sample rates and Dolby and dts coded audio as well. The signals are compatible with various measurement views in APx500.

Many of these test signals are also available as linear format audio files at various sample rates and resolutions, using the Waveform Generator utility provided on the Audio Precision Web site at ap.com.

- Playable test signals on video DVD: APx-DVD1
- Playable test signals on audio CD: APx-CD1

...for Generator Waveform use

Versions of these test signals optimized for Generator Waveform use are also available on the Audio Precision Web site at ap.com. Coded files are provided on disc; linear files can be created using the Waveform Generator utility. See the documentation in the Audio Test Signals folder.

...making your own test signals

For many measurements, you can make your own test signals for external source testing. See Creating Your Own Test Signals on page 22 for guidelines.

Audio Precision Test Signals on DVD: APx-DVD1

These test signals are provided as video-menu-driven audio tracks on the APx-DVD1. Video is 720p 30 frame. Formats available are:

- Linear PCM 48 kHz & 96 kHz (2 channels)
- Dolby Digital @ 48 kHz (5.1 channels)
- dts Digital Surround @ 48 kHz (5.1 channels)

All signals provide the same audio on both channels except as noted for Channel ID, Reference Level with LFE, and Crosstalk signals.

Bit test notes

The Bit test signals are prepared differently across the formats.

Bit test

For linear PCM, both channels carry the Bit test random signal. This signal uses the same algorithm used by the APx500 and by other Audio Precision instruments, and it is compatible with all Audio Precision Bittest Random analysis. These tracks provide a stereo Bit test signal that maintains bit accuracy.

Bit test (bitstream)

It is possible to bypass an encoder and insert a Bit test signal directly into a coded audio bitstream. Since lossy codecs cannot maintain bit-accuracy through an encode/decode cycle, this is the only way to test bit-accuracy for Dolby Digital and DTS Digital Surround paths.

A Bit test (bitstream) signal must be played directly into the APx digital input using no decoder to maintain bit accuracy.

APx-DVD1: Audio Test Signals on DVD

Test Signal Name	*Length*	*Description*
Channel_ID_2	0:14	997 Hz @ -20 dBFS. Channel 1 only for 5 seconds, followed by Channel 2 only for 2 seconds. Repeats.
Reference_Level_0dB	1:00	997 Hz @ 0 dBFS
Reference_Level_-20dB	1:00	997 Hz @ -20 dBFS
Reference_Level_-60dB	1:00	997 Hz @ -60 dBFS
Silence_dithered	1:00	Digital zero with triangular dither
Silence_undithered	1:00	Digital zero
MOD-SMPTE_4-to-1	1:00	60 Hz + 7 kHz, 4:1
MOD-SMPTE_1-to-1	1:00	60 Hz + 7 kHz, 1:1
MOD_10-to-1	1:00	60 Hz + 7 kHz, 10:1
DFD	1:00	12.5 kHz mean, 80 Hz difference (12460 Hz + 12540 Hz)
Crosstalk_left_only	1:00	9.997 kHz @ 0 dBFS, L
Crosstalk_right_only	1:00	9.997 kHz @ 0 dBFS, R
Crosstalk_both_channels	1:00	9.997 kHz @ 0 dBFS, L+R

APx-DVD1: Audio Test Signals on DVD		
Test Signal Name	*Length*	*Description*
Freq_sweep_11_0dB	0:18	997 Hz 2 sec, followed by 11 pt sweep 20 Hz-20 kHz @ 0 dBFS
Freq_sweep_11_-1dB	0:18	997 Hz 2 sec, followed by 11 pt sweep 20 Hz-20 kHz @ -1 dBFS
Freq_sweep_11_-20dB	0:18	997 Hz 2 sec, followed by 11 pt sweep 20 Hz-20 kHz @ -20 dBFS
Freq_sweep_31_0dB	0:42	997 Hz 2 sec, followed by 31 pt sweep 20 Hz-20 kHz @ 0 dBFS
Freq_sweep_31_-1dB	0:42	997 Hz 2 sec, followed by 31 pt sweep 20 Hz-20 kHz @ -1 dBFS
Freq_sweep_31_-20dB	0:42	997 Hz 2 sec, followed by 31 pt sweep 20 Hz-20 kHz @ -20 dBFS
Freq_sweep_61_0dB	1:18	997 Hz 2 sec, followed by 61 pt sweep 20 Hz-20 kHz @ 0 dBFS
Freq_sweep_61_-1dB	1:18	997 Hz 2 sec, followed by 61 pt sweep 20 Hz-20 kHz @ -1 dBFS
Freq_sweep_61_-20dB	1:18	997 Hz 2 sec, followed by 61 pt sweep 20 Hz-20 kHz @ -20 dBFS
Slow_freq_sweep_11_0dB	0:36	997 Hz 2 sec, followed by 11 pt slow sweep 20 Hz-20 kHz @ 0 dBFS
Slow_freq_sweep_11_-1dB	0:36	997 Hz 2 sec, followed by 11 pt slow sweep 20 Hz-20 kHz @ -1 dBFS
Slow_freq_sweep_11_-20dB	0:36	997 Hz 2 sec, followed by 11 pt slow sweep 20 Hz-20 kHz @ -20 dBFS
Slow_freq_sweep_31_0dB	1:36	997 Hz 2 sec, followed by 31 pt slow sweep 20 Hz-20 kHz @ 0 dBFS
Slow_freq_sweep_31_-1dB	1:36	997 Hz 2 sec, followed by 31 pt slow sweep 20 Hz-20 kHz @ -1 dBFS
Slow_freq_sweep_31_-20dB	1:36	997 Hz 2 sec, followed by 31 pt slow sweep 20 Hz-20 kHz @ -20 dBFS
Slow_freq_sweep_61_0dB	3:06	997 Hz 2 sec, followed by 61 pt slow sweep 20 Hz-20 kHz @ 0 dBFS
Slow_freq_sweep_61_-1dB	3:06	997 Hz 2 sec, followed by 61 pt slow sweep 20 Hz-20 kHz @ -1 dBFS
Slow_freq_sweep_61_-20dB	3:06	997 Hz 2 sec, followed by 61 pt slow sweep 20 Hz-20 kHz @ -20 dBFS
Level_sweep	0:49	997 Hz @ -20 dBFS for 2 seconds followed by 400 Hz, 0 dBFS to -110 dBFS, 5 dB steps, 2 sec. per point, both channels
Multitone_32	1:00	Multitone with approximate ISO 1/3 octave frequencies from 16 Hz to 20 kHz, with each tone @ -22 dBFS
Bit test Bit test (bitstream)	0:30	Bit test linear form on LPCM tracks; Bit test (bitstream) form on Dolby Digital and DTS Digital Surround tracks.

Audio Precision Test Signals on CD: APx-CD1

These test signals are provided as compact disc audio tracks on APx-CD1. Format available is:

- Linear PCM 44.1 kHz, 16 bit (2 channels)

All signals provide the same audio on all channels except as noted for Channel ID, Reference Level with LFE, and Crosstalk signals.

Bit test notes

Both channels carry the Bit test random signal. This signal uses the same algorithm used by the APx500 and by other Audio Precision instruments, and it is compatible with all Audio Precision Bittest Random analysis. These tracks provide a stereo Bit test signal that maintains bit accuracy.

Note: CD audio has a bit depth of 16 bits. For bit accurate Bit test playback, you must set the analyzer input to 16 bits.

APx-CD1: Audio Test Signals as CD audio tracks

Signal Name	*Track*	*Length*	*Description*
Channel_ID_2	01	0:14	997 Hz @ –20 dBFS. Channel 1 only for 5 seconds, followed by Channel 2 only for 2 seconds. Repeats.
Reference_Level_0dB	02	1:00	997 Hz @ 0 dBFS
Reference_Level_-20dB	03	1:00	997 Hz @ –20 dBFS
Reference_Level_-60dB	04	1:00	997 Hz @ –60 dBFS
Silence_dithered	05	1:00	Digital zeros with triangular dither
Silence_undithered	06	1:00	Digital zeros
MOD-SMPTE_4-to-1	07	1:00	60 Hz + 7 kHz, 4:1
MOD-SMPTE_1-to-1	08	1:00	60 Hz + 7 kHz, 1:1
MOD_10-to-1	09	1:00	60 Hz + 7 kHz, 10:1
DFD	10	1:00	12.5 kHz center, 80 Hz difference (12460 Hz + 12540 Hz)
Crosstalk_left_only	11	1:00	9.997 kHz @ 0 dBFS, L
Crosstalk_right_only	12	1:00	9.997 kHz @ 0 dBFS, R
Crosstalk_both_channels	13	1:00	9.997 kHz @ 0 dBFS, both L and R
Freq_sweep_11_0dB	14	0:18	997 Hz 2 sec, followed by 11 pt sweep 20 Hz–20 kHz @ 0 dBFS
Freq_sweep_11_-1dB	15	0:18	997 Hz 2 sec, followed by 11 pt sweep 20 Hz–20 kHz @ –1 dBFS
Freq_sweep_11_-20dB	16	0:18	997 Hz 2 sec, followed by 11 pt sweep 20 Hz–20 kHz @ –20 dBFS
Freq_sweep_31_0dB	17	0:42	997 Hz 2 sec, followed by 31 pt sweep 20 Hz–20 kHz @ 0 dBFS
Freq_sweep_31_-1dB	18	0:42	997 Hz 2 sec, followed by 31 pt sweep 20 Hz–20 kHz @ –1 dBFS
Freq_sweep_31_-20dB	19	0:42	997 Hz 2 sec, followed by 31 pt sweep 20 Hz–20 kHz @ –20 dBFS
Freq_sweep_61_0dB	20	1:18	997 Hz 2 sec, followed by 61 pt sweep 20 Hz–20 kHz @ 0 dBFS
Freq_sweep_61_-1dB	21	1:18	997 Hz 2 sec, followed by 61 pt sweep 20 Hz–20 kHz @ –1 dBFS
Freq_sweep_61_-20dB	22	1:18	997 Hz 2 sec, followed by 61 pt sweep 20 Hz–20 kHz @ –20 dBFS
Slow_freq_sweep_11_0dB	23	0:36	997 Hz 2 sec, followed by 11 pt slow sweep 20 Hz–20 kHz @ 0 dBFS
Slow_freq_sweep_11_-1dB	24	0:36	997 Hz 2 sec, followed by 11 pt slow sweep 20 Hz–20 kHz @ –1 dBFS
Slow_freq_sweep_11_-20dB	25	0:36	997 Hz 2 sec, followed by 11 pt slow sweep 20 Hz–20 kHz @ –20 dBFS
Slow_freq_sweep_31_0dB	26	1:36	997 Hz 2 sec, followed by 31 pt slow sweep 20 Hz–20 kHz @ 0 dBFS

APx-CD1: Audio Test Signals as CD audio tracks			
Signal Name	*Track*	*Length*	*Description*
Slow_freq_sweep_31_-1dB	27	1:36	997 Hz 2 sec, followed by 31 pt slow sweep 20 Hz–20 kHz @ –1 dBFS
Slow_freq_sweep_31_-20dB	28	1:36	997 Hz 2 sec, followed by 31 pt slow sweep 20 Hz–20 kHz @ –20 dBFS
Slow_freq_sweep_61_0dB	29	3:06	997 Hz 2 sec, followed by 61 pt slow sweep 20 Hz–20 kHz @ 0 dBFS
Slow_freq_sweep_61_-1dB	30	3:06	997 Hz 2 sec, followed by 61 pt slow sweep 20 Hz–20 kHz @ –1 dBFS
Slow_freq_sweep_61_-20dB	31	3:06	997 Hz 2 sec, followed by 61 pt slow sweep 20 Hz–20 kHz @ –20 dBFS
Level_sweep	32	0:49	997 Hz @ –20 dBFS for 2 seconds followed by 400 Hz, 0 dBFS to –110 dBFS, 5 dB steps, 2 sec. per point, both channels
Multitone_32	33	1:00	Multitone with approximate ISO 1/3 octave frequencies from 16 Hz to 20 kHz, with each tone @ –22 dBFS
Bit test	34	0:30	Bittest Random signal on each track. Set analyzer input to 16 bits.

Creating your own test signals

Some of the measurement views in External Source configuration do not require specific known test signals. Suggestions for custom signals for these views follow:

- The Level measurement view will measure the RMS level of any audio signal within the input range of the analyzer.
- The THD+N measurement view will measure the THD+N of any periodic audio signal within the input range of the analyzer. A sound wave is called periodic if the wave pattern repeats continuously. Sine waves, square waves and other repetitive waves are periodic.
- The Crosstalk measurement view will measure the degree to which the signal on one channel is present on another. This requires external source signals that by design have signal present or absent on individual channels in a known way, and that the “mode” set in Crosstalk Signal Acquisition and Analysis corresponds to the signal and to your intent. In External Source configuration, the analyzer software examines the inputs signals to determine driven and undriven channels.
  A crosstalk signal is typically high-frequency (we use approximately 10 kHz), because the most common crosstalk mechanism is capacitive coupling, which increases with frequency. Other frequencies can be used.
- The Interchannel Phase measurement view will measure the phase between channels for any periodic audio signal within the range of the analyzer. A sound wave is called periodic if the wave pattern repeats continuously. Sine waves, square waves and other repetitive waves are periodic.
- The DC Level measurement view requires no audio signal. If a DUT has a dc offset with no audio signal, it will be measured by the analyzer. However, it is possible that a DUT’s dc offset might change in the presence of signal, and any audio signal can be used during the measurement of DC Level. The DC Level meter provides excellent ac rejection.
- The DFD IMD measurement view requires a twin-tone signal, with a pair of tones of equal level set apart by a difference frequency, the pair centered around a mean frequency. The difference frequency must be in the range from 80 Hz to 2 kHz; we use 80 Hz. The mean frequency choice depends to some extent on the difference frequency choice, since the maximum range is the sum and the difference of the two. Mean frequency minimum is 2.5 kHz, maximum is 20 kHz; we use 12.5 kHz. See page 305 for more information about IMD.
- The SMPTE IMD and MOD IMD measurement views require a twin-tone signal. The higher frequency must be in the range from 2 kHz to 20 kHz; we use 7 kHz. The lower frequency choice depends to some extent on the higher frequency choice, with a range from a minimum of 40 Hz to a maximum between 333.33 Hz to 1 kHz; we use 60 Hz. To satisfy industry standards, for MOD the ratio [low:high] of the level of these tones must be 1:1, 4:1 or 10:1; for SMPTE, the ratio must be 1:1 or 4:1. See page 305 for more information about IMD.
- The Signal Analyzer measurement view will provide time domain or frequency domain views of

any audio signal within the input range of the analyzer.

The Measurement Recorder measurement view will provide a record of RMS level versus elapsed time for any audio signal within the input range of the analyzer. The Measurement Recorder also provides THD+N records, which require a periodic waveform.

Creating custom stimulus signals

Stepped Frequency Sweeps

You can create custom Sweep Tables and corresponding WAV files using the dialogs available from the Stepped Frequency Sweep and Bandpass Frequency Sweep measurements.

Multitones

You can create custom multitone signal definitions and corresponding WAV using the dialogs available from the Multitone measurement.

Using third-party tools

For high-performance testing, stimulus signals must be properly constructed and very pure; that is, they must have very low inherent distortion, noise or spurious components. It is possible to create digital files that satisfy these requirements using third-party software such as Matlab, Adobe Audition or Audacity, but care must be taken.

Section III: Measurements

23

Making Measurements

Overview

This chapter contains general information about making measurements in APx500, and about using the common settings and controls in the APx500 generator and analyzer.

For a more focused look, go to chapters 24 through 84, which discuss each measurement in detail. The Navigator and the Selector are covered in Chapter 3. Chapters 99 and 100 look closely at Results. Reports are covered in Chapter 87, as is Automation in Chapter 86.

Sequence Mode or Bench Mode?

Sequence Mode and Bench Mode present two different approaches to audio testing.

If you would prefer pre-defined measurements and results with easy automation and reporting, work in Sequence Mode.

On the other hand, if you are interested in specifying your own generation and analysis tools and working in a more free-form, stimulus and response method, give Bench Mode a try.

Or, mix it up. Run some Sequence Mode measurements, then look at interesting discoveries in Bench Mode, all in the same project.

How to choose measurements

APx500 provides many audio measurements for you to choose from. Which should you use? The answer to that question depends on a number of factors, such as:

The nature of the device under test (DUT)

The type of DUT you are testing will determine which measurements are appropriate. Most audio devices can be evaluated by the "Big Six" set of measurements (see page 181), for example, but if your DUT is a single-channel device, it cannot be tested for crosstalk.

Specified measurement requirements

You may have a requirement to make a specific measurement, such as SMPTE IMD.

The required testing speed

In a production environment, you may want the complete set of measurements to run very quickly. In this case, select only measurements that are essential, and be sure to choose the fastest methods.

The "Big Six"

Audio engineers agree that there is a small group of measurements that will reveal the key characteristics and performance of almost any audio device. At Audio Precision we talk about the "Big Six" measurements as the benchmark set of audio tests.

Level

The basic level measurement provides the output level or amplitude of an audio device. Depending on the device and your test configuration, the result might be displayed in volts, watts, dBV or other level units.

Gain gives the ratio of the DUT output level to its input level.

THD+N

THD+N is an abbreviation for total harmonic distortion plus noise. THD+N measurements have been used for many years as a comprehensive single-value statement of an audio device's overall performance.

Frequency Response

A frequency response test reveals how a device "responds" across the audible spectrum. For most devices, a "flat" response curve, where all frequencies are passed with the same gain, is desirable.

Signal to Noise Ratio

Signal to Noise Ratio or SNR gives a single value that indicates how "noisy" an audio device is. Two measurements are actually required: the first measures the DUT output level (usually at either maximum level or at a nominal operating level); the second measures the residual noise in the device. The signal level is divided by the noise level, and the resulting ratio is usually expressed in decibels.

Crosstalk

Crosstalk is an expression of the degree of signal leakage between DUT channels. Ideally, none of the signal in one channel will appear in a second channel, but in practical devices there is always measurable crosstalk. Crosstalk often varies with frequency.

Phase

Phase is a measure of the lagging or leading of an alternating current waveform with regard to a reference waveform.

Interchannel phase measures the phase relationships between channels in a multichannel device. One channel is chosen as the reference.

Other measurements

Note: the APx515 requires software option keys for some measurements. See page 189.

APx500 performs many measurements beyond the Big Six, including dc level measurements, intermodulation distortion (IMD) measurements, regulated measurements, tone burst measurements and more. There is also a Signal Analyzer tool with Scope and FFT Spectrum displays at very high resolution (FFT length up to 1 million samples).

Measuring using Meter measurements

Many of these methods measure streaming audio, where the generator continuously outputs a stimulus signal, and the analyzer measures the DUT output in real time using a Meter measurement. Meter measurements have single value (meter) results and are displayed as a Bar Graph and a numerical value.

Meter measurements have a **ON / OFF** button on the Generator panel. Meter measurements typically have single value (meter) results.

Stimulus signals for meter measurements include:

- Sine wave;
- Mixed twin tone sine wave;
- Special Sine wave;
- Square wave;
- Mixed square wave and sine wave;
- Generator waveform (an arbitrary waveform played from a file on disk);
- Diagnostic digital signal.

Sine wave

Sine waves are generated in APx DSP (digital signal processing), and are the default waveform for most measurements. For analog configurations, the default frequency is 1 kHz; the default level is 100.0 mVrms. For digital configurations, the default level is –20.000 dBFS. Most APx measurements offer a sine wave.

The High Performance Sine Generator in the APx555 can generate sine waves of lower distortion and wider frequency range than the DSP-generated sine waves used by the DAC generator.

Mixed twin tone sine wave

Mixed twin tone sine waves are generated in APx DSP, and are available for IMD measurements (Chapter 48). The frequencies of the two tones, and for some measurements, the level ratio of the two tones can be set in the corresponding IMD measurement settings panel. The output level is the rms level of the complex waveform that is the sum of the two component tones. For analog configurations, the default level is 100.0 mVrms. For digital configurations, the default level is –20.000 dBFS.

Special Sine

You have the option of choosing **Split Sine, Split Phase or Sine + Offset** generator waveforms for the following measurements:

- Reference Levels
- Dynamic Range AES17
- Frequency
- Interchannel Phase
- Level and Gain
- Maximum Output
- Measurement Recorder
- Signal Analyzer
- Signal to Noise Ratio
- Stepped Level Sweep
- THD+N
- CMRR-IEC
- CMRR-IEC

For these measurements, a single sine wave is the default generator waveform. A Special Sine selection allows you to output sine waves of two different frequencies to different channels, sine waves of the same frequency but different phase to different channels, or sine waves with DC offset.

Split Frequency

Split Frequency allows you to assign a sine wave of **Frequency A** and a sine wave of **Frequency B** to the output channels. By default, **Frequency A** is assigned to odd channel numbers and **Frequency B** to even channel numbers, but these can be reassigned in the **Advanced Settings** dialog for the measurement.

Split Phase

Split Phase allows you to assign a sine wave of **Phase A** (the reference at 0 degrees) and a sine wave of **Phase B** to the output channels. By default, **Phase A** is assigned to odd channel numbers and **Phase B** to even channel numbers, but these can be reassigned in the **Advanced Settings** dialog for the measurement.

Enter a value for **Phase B** in degrees. The range is –180.00 degrees to +179.99 degrees. A negative value sets the B waveform to lag the A waveform; a positive value sets the B waveform to lead the A waveform.

Split (IMD)

For IMD measurements, the Split IMD setting defeats the summing of the two IMD frequencies within the generator. Instead, the signals are "split" and routed independently to channel outputs.

Acoustic testing of microphones sometimes uses two stimulus loudspeakers, each with a single frequency component of an IMD stimulus. The stimulus signals are summed acoustically at the microphone element. This method provides a distortion result for the microphone while minimizing the distortion contributions of the loudspeakers.

Offset

The Offset filed allows you to add a DC offset to the generator output. If the generator level is set to zero (0.000 Vrms or 0 FS), the generator can optionally output DC only (pure DC).

For analog output, when sine level is zero the maximum offset is $\sqrt{2}$ times the maximum RMS level (±) for the current instrument and output connector.

For digital output, when sine level is zero the maximum offset is 1 D (±).

Offsets can be set at different levels across the channels in Advanced Settings. Unset Track First Channel and enter the desired values in the Offset column of the Signal Generation grid.

Square wave

A fast rise time square wave is available in the APx analog generator for the APx555 and analyzers equipped with the AG52 option (page 5) installed. AG52 can be fitted in the APx582 and in the APx52x family of analyzers.

A DSP-generated square wave is available in the APx digital generator for all APx analyzers.

When the hardware or output configuration allows, the square wave is available for the following measurements:

- Frequency
- Level and Gain
- Measurement Recorder
- Signal Analyzer

Mixed square and sine wave (DIM)

A special stimulus signal (the sum of a square wave and a low amplitude sine wave) is required for the DIM IMD measurement (chapters 42 and 43). DIM measurements are only supported in the APX555 and analyzers with the AG52 option (page 5) installed.

Generator waveform

In addition to the DSP-generated waveforms, APx500 has the capability to use custom (or *arbitrary*) waveforms, loaded into the generator from waveform files available to the computer running APx500. This is particularly useful when coded audio files (such as dts, Dolby or others) are required to stimulate a decoder in your DUT. See page 186.

For perceptive audio measurements, speech sample files are used, and can be opened as generator waveform files. See More About POLQA on page 451, and More About PESQ on page 438.

Diagnostic digital signals

Certain digital signals are useful to diagnose or characterize digital transport systems. In APx500, these include

- Walking Zeros
- Walking Ones
- Constant Value
- Bit test

See page 50 for more information about these signals.

Reading Rate

APx meters have a reading rate, the rate at which the meter reads the incoming data and determines and displays the measured value. Faster reading rates put a higher demand upon the PC, and in some circumstances may affect PC performance. The reading rate also has an effect on analyzer performance at low frequencies: each reading rate has a low-frequency point below which analyzer performance falls off. THD+N performance degrades rapidly below this point, with level accuracy degrading a little more slowly.

Faster rates are required to resolve short-lived phenomena; slower rates are required to accurately read values at lower frequencies.

Monitor Meters

Bench Mode meter monitors have a fixed reading rate of 4/s, with accuracy falling off below 12 Hz.

Recorders

A reading is taken at the beginning of the record and at (approximately) the reading rate throughout the record. The final reading is taken at the reading instant that equals or exceeds the sweep end time.

Sequence Mode

Sequence Mode Measurement Recorder can have the reading rate set to a number of fixed choices. Accuracy limits are listed here, by reading rate and frequency:

- at 20/sec > 60 Hz
- at 10/sec > 30 Hz (the default)
- at 5/sec > 15 Hz
- at 3/sec > 9 Hz
- at 2/sec > 6 Hz
- at 1/sec > 3 Hz
- at .5/sec > 1.5 Hz
- at .2/sec > .6 Hz
- at .1/sec > .3 Hz

Bench Mode

Bench Mode Recorder can have the reading rate set to a number of fixed choices. Accuracy limits are listed here, by reading rate and frequency:

- at 256/sec > 1536 Hz
- at 128/sec > 768 Hz
- at 64/sec > 184 Hz
- at 32/sec > 96 Hz (the default)
- at 16/sec > 48 Hz
- at 8/sec > 24 Hz
- at 4/sec > 12 Hz
- at 2/sec > 6 Hz
- at 1/sec > 3 Hz

Bench Mode Sweep

Bench Mode Sweep can have reading rate set to the same fixed choices as Bench Mode Recorder, up to a maximum of 32/sec. An **Auto** selection is added to the Sweep. The default is **Auto**.

Auto reads at a rate of 32/s at higher audio frequencies. At lower sweep frequencies the rate is reduced to maintain accuracy.

Measuring using Batch measurements

Other methods are “batch” measurements, where the stimulus is a signal or series of signals of a certain duration, and the analyzer acquires DUT output that corresponds to the stimulus. Batch measurements have a **Start** button on the Measurement Settings panel. Batch measurements typically have both single value (meter) results and XY graph results.

Signals for batch measurements

Stimulus signals for batch measurements include:

Sweeps:

- Continuous sweep (log-swept sine or *chirp*)
- Stepped sine sweep
- Stepped mixed twin tone sweep
- Regulated sine sweep

Sequential measurements:

- sine on sequential channels, for crosstalk
- sine at two sequential levels, for SNR or Dynamic Range
- regulated sine, for Maximum Output

Digital diagnostic signals:

- Bit test
- Generator waveform (an arbitrary digital diagnostic signal played from a file on disk).

Multitone signals.

Sweeps

Stepped sweeps are a classic method of audio testing, where one parameter is moved (swept) across a range of values while one or more other parameters are measured. The results are usually displayed on an XY graph, with the swept parameter on the X-axis and the measured results on the Y-axis.

Continuous sweep

A continuous sweep is a brief log-swept sine wave (sometimes called a log chirp) that moves continuously across a specified range of frequencies. The DUT output is acquired by the analyzer and is mathematically processed to provide a number of results.

Continuous sweep stimulus signals are used in the Acoustic Response and Continuous Sweep measurements in both Sequence Mode and Bench Mode, and in Frequency Response, Loudspeaker Production Test and Impedance Thiele-Small measurements.

Stepped sweeps

In practical measurements, the sweep is usually stepped across a number of discrete points rather than moved continuously. This provides precision in the determination of point values, and allows the use of settling algorithms to wait for stable data. Bench Mode has a general purpose, configurable Sweep engine.

APx Sequence Mode stepped sweeps include:

Stepped Frequency sweep measurement

The Stepped Frequency Sweep (chapter 81) measurement provides a sine wave stimulus signal that is moved across a range of frequencies in a specified number of points. The DUT output is acquired by the analyzer and processed for display.

Stepped level sweep measurement

The Stepped Level Sweep (chapter 82) measurement provides a sine wave stimulus signal that is moved

across a range of levels in a specified number of points. The DUT output is acquired by the analyzer and processed for display.

Regulated Frequency Sweep

Regulated Frequency Sweeps use a sine wave stimulus signal that is moved across a range of frequencies in a specified number of points. At each point, the generator level is adjusted regulated (adjusted to attain a certain level or distortion figure) at each step. See Regulated Frequency Sweep measurements in Chapter 75. This method is often employed to plot an amplifier frequency sweep at the amplifier's maximum output level.

IMD Level Sweep

A stepped mixed twin tone sweep is a IMD stimulus signal that is moved across a range of levels in a series of points or steps. See IMD Level Sweep in Chapter 50.

Nested Sweeps

Bench Mode measurements and some Sequence Mode sweeps provide a Nested Sweep function.

When Sweeps are nested, one defined sweep (driven by a Primary Source, such as Frequency) is run at steps of second defined sweep (driven by a Secondary Source, such as Level). The effect is to nest frequency sweeps (to follow the example here) in a series of level sweeps, providing a family of responses at different levels.

Ordered measurements

Sine moved sequentially across channels

The Crosstalk measurement uses a sine wave stimulus on each of the active output channels, one at a time. See Chapter [crosstalk chapter].

Sine at two sequential levels

The Signal to Noise measurement (chapter 79) first measures the DUT output with a sine wave stimulus, and then makes a second measurement with the generator output level set to zero.

The Dynamic Range (AES17) (chapter 45) measurement first measures the DUT output with a sine wave stimulus at maximum level, and then makes a second measurement with the generator output level set to a very low value, usually 60 dB below the maximum level.

Other batch measurements

Regulated sine

A regulated sine wave stimulus signal is regulated (adjusted to attain a certain level or distortion figure) before the measurement is made. Sequence Mode and Bench Mode both offer regulation in Generator Level setting, and as an option in stepped sweeps.

Digital error rate (bit test}

Bittest Random (also simply called Bit test) is a pseudo-random waveform with values uniformly distributed between plus and minus full scale. Algorithms in the generator and analyzer determine sample values. Bit test exercises a wide range of levels and frequencies and is the most thorough of the bit-accurate waveforms. See Chapter 41.

Generator waveform

In addition to the DSP-generated waveforms, APx has the capability to use custom (or *arbitrary*) waveforms, loaded into the generator from waveform files available to the computer running APx500. This is particularly useful when coded audio files (such as dts, Dolby or others) are required to stimulate a decoder in your DUT. See page 186.

Multitone Analyzer

The Multitone Analyzer uses a special stimulus signal called a *multitone*. Multitone analysis is very fast, and provides a broad range of results. See Chapter 64.

Diagnostic tools

Signal Analyzer

The Signal Analyzer is a general-purpose diagnostic tool, with a time domain view that emulates a multichannel oscilloscope display, and a frequency domain view that is an FFT Spectrum. The Signal analyzer uses a high-resolution FFT technique for both views. FFT length can be set as high as 1048576 points. Triggering, coupling, FFT window and number of averages are selectable.

A version of the Signal Analyzer is available for both Sequence Mode and Bench Mode.

Signal Analyzer tools in APx500 are:

- Scope
- FFT Spectrum
- Amplitude Spectral Density
- Power Spectral Density

Additionally, a special function in Signal Analyzer allows you to view either the embedded audio or the PDM bitstream, when the PDM option module is installed. See page 489.

Measurement Recorder

The Measurement Recorder is a general-purpose diagnostic tool that provides a record of level or THD+N versus elapsed time. The length of the record can be very long (up to one week), providing a means of monitoring the output of a device under test over an extended period of time. The Measurement Recorder also provides a means of recording the audio acquisition to a file.

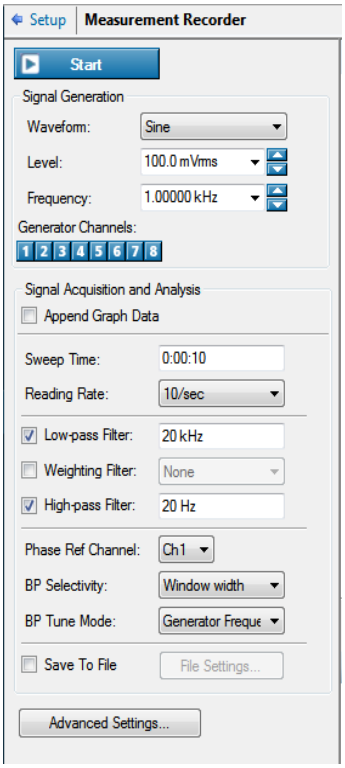
The Measurement Recorder can be used in External Source configuration, but it does not require a specific test signal. It will record any audio signal within the input range of the analyzer.

A version of the Measurement Recorder is available for both Sequence Mode and Bench Mode.

Metadata Recorder

The Metadata Recorder measurement records the state of selected metadata settings in the incoming digital audio signal. If one of the selected settings changes state during the acquisition, the transition instant and the new state are recorded and displayed.

Measurement Settings panel



Each measurement has a measurement settings panel to the right of the Navigator, containing Signal Generation in the upper area, and Signal Analysis controls in the lower area. A range of different controls are available, depending upon system hardware, signal path setup and measurement. This illustration is a typical example.

Signal generation controls are discussed here. Go to page 188 to read about signal acquisition and analysis controls.

Signal Generation controls

Start / Generator ON

Most measurements use a generator stimulus during the measurement process. For the APx500 meter methods, there is a **Generator On** button.

Other methods begin a batch process that returns a result: these measurements use a **Start** button.

DC level, Noise and external source measurements do not use a generator stimulus signal. There is no Generator **On** or **Start** button on these measurement panels.

Signal Path Setup

There is a **Signal Path Setup** button that allows you to switch to the Signal Path Setup panel and make changes. A **Back** button on Signal Path Setup allows a quick return.

Waveform

Select the stimulus waveform here. The waveforms available depend upon the analyzer hardware, configuration and the current measurement. See the relevant measurement chapter for more information.

Depending upon the measurement, you may have a Frequency control(s), Sweep settings or other controls available.

Enter the generator output level in the **Level** field.

EQ

For certain sweep measurements, generator equalization allows you to impose a response curve on the stimulus signal to compensate for a non-flat response in the device under test. See Generator Equalization on page 192.

Channel mute switches

The generator output to any channels or channels can be muted (turned OFF) independent of the generator status. A dark blue channel mute button indicates an unmuted channel (channel ON). Muted (channel OFF) channels are indicated by a pale gray button. By default, all channels are unmuted (turned ON).

Generator Waveforms

Test signal generation from a waveform file

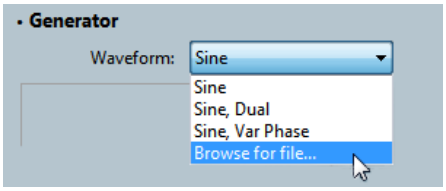
APx500 can use custom generator waveforms (also called arbitrary waveforms) loaded into the generator from waveform files available to the computer running the measurement software. This is particularly useful when coded audio files (dts, Dolby or others) are required to stimulate a decoder in your DUT.

The loaded waveform is “looped” for generator output, so that the end of the waveform is immediately followed by the beginning, providing continuous playback without a jump discontinuity.

Generator waveform files are not restricted to sine waveforms; other waveforms can be used. A number of generator waveform files are available on the Audio Precision Web site at ap.com.

You can also create your own generator waveform files To be useful for measurement, user-created files must be designed with the requirements of the measurement context in mind. Not all audio signals are compatible with certain analyzer views nor are useful for measurement. See page 613 for a table listing the audio file formats supported for generator waveforms.

Using a generator waveform file



A measurement that supports generator waveform files displays a Waveform selection list. The default is **Sine** (generated in DSP). Click the Waveform menu arrow in Signal Generation and choose **Browse for file...** to select an audio file for the generator to use. You can select multiple files.

A selected file becomes the source for that measurement's generator output, and is attached to the project file. Once attached to the project, an audio file is available to all other supported measurements as a selection on the Waveform menu. Files attached to the project but not currently used by any measurement can be removed by navigating to the dialog at **File > Manage Attached Project Items** (see page 27.)

Signal Output

Level adjustment

For analog output, and for digital output of linear waveforms when **Bit Exact** is not set, the level of a generator waveform file can be adjusted in the Generator panel.

> *Level adjustment is not available for coded waveforms. Coded waveforms are always transmitted bit-exact.*

The actual signal level within a generator waveform file is unknown. The file may contain a sine wave at 0 dBFS, or it may contain pink noise at –42 dBFS. This being the case, the generator output is also indeterminate. Level adjustments for generator waveform files are calibrated assuming the maximum possible level in the file.

For analog output, a 1 Vrms level setting would output a 0 dBFS sine wave at 1 Vrms; lower signal levels in the file would be output at correspondingly lower voltages.

For digital output, a 0 dBFS level setting would output a 0 dBFS sine wave at 0 dBFS; lower signal levels in the file would be output at correspondingly lower digital levels.

A –15 dBFS signal in the file with a –12 dBFS level setting would be output at –27 dBFS.

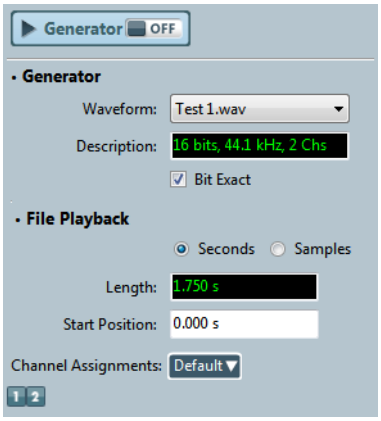
Dither

For digital output (when **Bit Exact** is not set), a generator waveform file signal is always redithered.

Bit Exact

> *Bit Exact is also called bit accurate.*

When a digital output is selected, the **Bit Exact** checkbox is available when a linear generator waveform is in use. Checking **Bit Exact** forces the APx generator to output the waveform with no changes whatsoever, with every audio bit exactly as it is in the disk file. Level adjustment is unavailable and dither is off. Coded waveforms are always transmitted bit-exact.



Generator waveform length constraints

Generator waveforms are limited in file length by APx instrument memory resources (see below). A waveform file that exceeds the maximum length will be truncated as it is loaded. A truncated waveform is marked with a warning icon ⚠ when selected as a generator source.

Truncated waveforms should be avoided, as a jump discontinuity may be introduced on playback at the point of truncation. Best practice is to use the shortest generator waveform file consistent with your measurement requirements.

Memory resources

A generator waveform file is downloaded from the PC into the APx analyzer hardware as needed for a measurement. The hardware memory has a capacity of 128 MB, and audio waveforms that exceed 128 MB will be truncated.

Since APx500 converts 8- and 16-bit PCM waveforms to 32-bit for generator output, and expands coded audio for playback, generator waveform files may use considerably more memory than indicated by their file size on disk.

Maximum File Length table

File formats and sample rates affect the amount of memory used. The table below is provided as a guide, indicating the running times that represent the maximum memory available for various formats.

Generator Waveform Maximum File Length Table

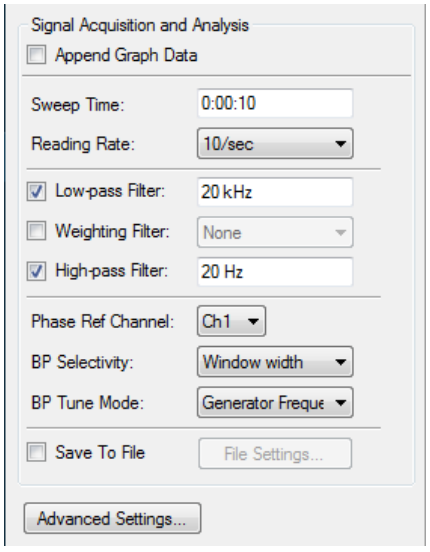
Waveform Type	*Sample Rate*	*Playback Time (sec)*
Mono .WAV	48 kHz	699
Mono .WAV	96 kHz	349
Mono .WAV	192 kHz	174
Stereo .WAV	48 kHz	334
Stereo .WAV	96 kHz	174
Stereo .WAV	192 kHz	87
AC3	48 kHz	699
EC3	192 kHz	174
DTS	48 kHz	699
DTS-HD	192 kHz	43

Supported generator waveform file formats

The table above describes the maximum file length, in seconds, of an audio waveform file that can be transmitted by APx. The playback times are approximate and can vary depending on the audio content, if the file is encoded.

Sample Rate is the rate at which the audio samples are transmitted from APx via the selected output interface. EC3 and DTS-HD are always transmitted over HDMI at 192 kHz regardless of the encoded bit rate of the embedded audio content. AC3 and DTS are always transmitted over the digital interface at 48 kHz, regardless of the choice of digital interface or the encoded bit rate of the embedded audio content.

Analyzer controls



Typical Analyzer controls (Measurement Recorder shown)

Signal acquisition and analysis controls are found in the lower area of the Measurement Settings panel. A range of different controls are available, depending upon system hardware, signal path setup and measurement. This illustration is a typical example.

Signal acquisition and analysis controls are discussed here. Go to page 186 to read about signal generation controls.

Common acquisition and analysis controls

Filtering

Signal Path Setup provides Low-pass filtering (including AC/DC coupling control) High-pass and Weighting filtering for the entire Signal Path. See chapter 97.

Bench Mode Analyzer Controls

Bench Mode provides Settling and some filtering, reference channel and IMD Type controls for the Analyzer.

Sequence Mode Analyzer Controls

Measurements in Sequence Mode often provide additional local filtering, and other controls specific to the measurement.

Append Graph Data

Most batch measurements allow you to append acquisitions. Bench Mode also enables repeating measurements.

Other acquisition and analysis controls

Other controls are available for specific measurements, primarily in Sequence Mode. These include the following:

Ref Channel

For phase measurements, a reference channel can be specified. See chapter 53.

Save to File

The Measurement Recorder and the Multitone Analyzer allow you to save the acquired audio to a file. See chapters 61 and 64.

Sweep Time / Reading Rate

These are adjustments for the Measurement Recorder acquisition. See chapter 61.

Averages

Acoustic Response, Multitone and Signal Analyzer allow you to average a number of acquisitions. See chapters 24, 64 and 78.

Selectivity / Filter Tune Mode

Bandpass Level allows you to adjust and steer the bandpass filter. See chapter 26.

IMD Mode

The IMD measurements allow you to specify which distortion products or combinations of distortion products to view. See chapters 48 and 50.

Trigger

Multitone and Signal Analyzer allow free run or adjustable trigger settings. See chapters 64 and 78.

Measured Channel / Target THD+N

The Maximum Output measurements allow you to specify the regulation target.

Appending and Importing data into Data Sets

Sweep and other batch measurements allow appending and importing of data. Multiple instances of data are managed as **Data Sets**.

For example, with a new sweep acquisition, the graph is populated with a new set of data for display.

This data can be seen as a curve on the graph or as a table of values in the graph data grid for each view. This data is called **Measured Data**, and is shown in the **Data Sets** panel as **Measured 1**.

Append

Set the **Append** checkbox in the Measurement Settings Panel to add more measured data sets to the graph data already acquired. Data added in this way is a shown in the **Data Sets** panel as **Measured 2**, **Measured 3**, etc.

Import

You can also use the **Import** command on the graph data grid panel to bring external data into the graph. Data added in this way is a shown in the **Data Sets** panel as **Imported 1**, **Imported 2**, etc.

Read more about managing Data Sets on page 649.

Advanced Settings

Each Sequence Mode measurement has additional settings available in an Advanced Settings dialog. These vary with the measurement, and are discussed in the individual measurement chapters.

External Source

Some DUTs (disc or other media players, broadcast receivers, transmission links) have no available signal inputs. How is the stimulus signal introduced into the DUT?

The answer is to use the APx External Source configuration, which is an open-loop solution that does not use the instrument generator, but instead provides a test signal that can be transmitted, played or otherwise generated externally.

Not all measurements support the External Source configuration. See Chapter 6 for more information about External Source measurements.

Results

Each measurement has one or more primary results (measured results) associated with it. A primary result is the direct result of a measurement; a primary Level result, for example, is the measured amplitude of the audio signal. See chapter 99.

Result data can be also imported from other measurements, from other applications or from arbitrary lists of values.

Navigating results

Results are represented as branch nodes on the Navigator tree display, and as thumbnails in the Selector filmstrip display. The currently selected result is displayed in the Graph panel.

Primary or derived results can be added to or deleted from a measurement, and can be renamed to suit your needs. Multiple instances of a result can be added to a measurement.

Derived results

A measurement can have one or more user-defined derived results. A derived result is a data point or set of data points derived from an existing result (called the source result) by a mathematical computation such as averaging or normalizing. Derived results can be attached to primary results, imported results or existing derived results. See chapter 100.

Software Options

You can purchase software options to add features to your analyzer. Software options are enabled by one or more iButton keys, which are mounted in the APx Software Option module (pn BSW0.0000). The software option module (when present) is connected to the Software Options connector on the analyzer rear panel.

Adding software options

You can easily add more software options in the field by ordering and installing a factory-keyed iButton.

Viewing installed software options

To see if a software option is installed in your APx analyzer, run APx500 with the analyzer connected. Go to **Help > About** and scroll to the bottom of the Product Components list. If there is an iButton installed, you will see the text "iButton" followed by an identifying code. Click on an iButton line, and the software options enabled by that button will be shown in the Component Details panel.

Software options available for all APx analyzers

SW-POLQA-2

For Sequence Mode POLQA perceptual audio testing and MOS results, software option SW-POLQA-2 provides

- POLQA
- POLQA Averaged

SW-POLQA-2 provides POLQA operation for only 2 audio channels.

SW-PESQ

For Sequence Mode PESQ perceptual audio testing and MOS results, software option SW-PESQ provides

- PESQ
- PESQ Averaged

SW-SPK-PT

For fast Sequence Mode production testing of loudspeakers and drivers, software option SW-SPK-PT provides

- Acoustic Response (with the Rub and Buzz result)
- Loudspeaker Production Test
- Modulated Noise.

SW-SPK-RD

For Sequence Mode loudspeaker testing in R&D, SW-SPK-RD provides

- Acoustic Response (with the Rub and Buzz result)
- Impedance/Thiele-Small.
- Loudspeaker Production Test and
- Modulated Noise

Also included in SW-SPK-RD is authorization for two utilities:

- APx Polar Plot utility and
- APx Waterfall Graph utility.

These utilities are not part of the APx500 measurement software, but can be downloaded from ap.com.

Software options available for the APx515

Without options, the APx515 enables Sequence Mode and a basic set of APx500 features and measurements. This set can be expanded to the full complement of instrument-appropriate features and measurements by the installation of iButton software option keys.

Basic Measurements in the APx515

These measurements are enabled by an APx515 with no software option keys:

- CMRR
- Crosstalk, Custom
- Crosstalk, One Channel Driven
- Crosstalk, One Channel Undriven
- DC Level
- Frequency
- Frequency Response
- Interchannel Phase
- Level and Gain
- Measurement Recorder
- Noise (Q-Peak)
- Noise (RMS)
- Pass/Fail
- Signal to Noise Ratio
- SINAD
- Stepped Frequency Sweep
- Stepped Level Sweep
- THD+N

SW-BEN Option for the APx515

Software Option SW-BEN enables Bench Mode for the 515.

SW-ACR Option for the APx515

For acoustic testing, software option SW-ACR enables the following measurement:

- Acoustic Response (the Rub and Buzz result is not included with this option).

SW-HST Option for the APx515

For high-speed testing, software option SW-HST enables the following measurements:

- Continuous Sweep
- Multitone Analyzer

SW-AML

Software option SW-AML (the Advanced Measurement Library) enables the following measurements:

- Bandpass Level
- Bandpass Frequency Sweep
- Bandpass Level Sweep
- Crosstalk Sweep, Custom
- Crosstalk Sweep, One Channel Driven
- Crosstalk Sweep, One Channel Undriven
- Digital Error Rate
- Dynamic Range - AES17
- IMD Frequency Sweep
- IMD Level Sweep
- Maximum Output
- Maximum Output (CEA-2006)
- Metadata Recorder
- Noise Recorder
- Regulated Frequency Sweep
- Signal Analyzer

- IMD (DFD/MOD/SMPTE/CCIF)
- DC Level Sweep

Acoustic Testing

Acoustic testing brings some additional considerations to test. Preferred units are acoustic, microphone calibration may be necessary, and stimulus EQ may be required to compensate for transducers and provide Leveled Acoustic Output.

Specific acoustic and electro-acoustic measurements such as Acoustic Response, Loudspeaker Production Test and Impedance/Thiele-Small are supported by special Signal Path acoustic features, such as

- **Acoustic** output configuration
- **Acoustic** input configuration and
- **Output EQ** (equalization)

and the generator feature

- **Generator EQ**.

Although these features certainly have uses beyond acoustic testing (such as preemphasis/deemphasis applications or normalizing a response for flat display) their primary application is in acoustic test.

Acoustic Output Configuration

For convenience when performing acoustic tests, APx500 can be set to **Acoustic** output configuration. **Acoustic** output configures all generator level setting fields and generator level graph displays to units of pascals or dBSPL.

Since the analyzer outputs are voltage outputs, an acoustic output signal path must have scaling factor associated with it. This is called the **Voltage Ratio**, and it is expressed in Pa/V. The Voltage Ratio is set interactively in the Set Acoustic Output Level dialog (page 62), and is displayed in the References, Acoustic Mode dialog (also page 62).

Entering Acoustic output configuration

In **Signal Path Setup**, place a check in the **Acoustic** checkbox in **Signal Path Setup: Output Configuration**. **Acoustic** output configuration and the **Acoustic** output checkbox are only available for analog outputs.

You will notice a warning alert ⚠ appear, with a text box reading “Output References Voltage Ratio has not been set. Acoustic output levels may not be accurate.” This is a reminder to Set Acoustic Output Level, as described above.

Acoustic Input Configuration

For convenience when performing acoustic tests, APx500 can be set to **Acoustic** input configuration. Acoustic input configures all APx500 measurements in the signal path to display levels relative to channel-specific acoustic reference levels. This is useful for making acoustic measurements with multiple microphones.

Specifically,

- Level results are reported in dBSPL or Pa (in this case the Pa values are rms).
- Gain results are reported in Pa/Vrms.
- Instantaneous voltages are reported in Pa (in this case the Pa values are instantaneous).
- Impulse response is reported in Pa/V (in this case the Pa values are instantaneous).

The Pa and dBSPL values use the Sensitivity input reference(s), set in Set dBSPL Per-channel / Microphone Calibration dialog, and also displayed in the References dialog. See pages 62 and 64 for information about Acoustic input configuration and setting references.

Entering Acoustic input configuration

In Signal Path Setup, set the **Acoustic** checkbox. Acoustic input configuration and the Acoustic checkbox are only available for analog inputs.

Stimulus Equalization

APx offers two methods to equalize the stimulus signal: **Output Equalization**, which inserts a filter in the output signal path, and **Generator Equalization**, which changes the signal level by frequency as it is being generated. **Output Equalization** is a signal path feature, discussed on page 193. **Generator Equalization** is a feature of some measurements, discussed on page 192. Also see a **Comparison of Generator EQ and Output EQ** on page 193.

To EQ and re-save one or more .wav files, download the APx EQ .wav Files Utility at www.ap.com/display/file/661.

Application

An equalized stimulus signal is useful in a number of cases, including:

Leveled Acoustic Output

In testing microphones or systems with embedded microphones (such as a hearing aid or a mobile device), it is essential that frequency response of the power amplifier, the loudspeaker and the acoustic chamber be characterized and compensated for in the stimulus signal, so that the response of the system under test can be accurately measured. This process is called “leveling the acoustic output.”

In APx, this process requires characterizing the system response using a measurement microphone and a measurement such as Frequency Response or Acoustic Response.

The response curve is exported, and then imported into Output EQ and inverted.

This is best performed with Acoustic output and Acoustic input configurations on, and normalizing the level with Auto Gen Level before the characterization sweep and again after the Output EQ is applied.

Preemphasis and deemphasis

Some devices, such as phonographs, analog tape players or analog radio receivers, are designed to have a flat frequency response across a complete signal path, but non-flat frequency response at points within the signal path, with the goal of reducing noise introduced in the storage media or distribution channel. The response of the recording or transmitting device is modified by a preemphasis equalization curve, and the response of the playback or receiving device is modified by a complementary deemphasis equalization curve.

To test DUTs that incorporate a deemphasis equalization response, it is convenient to use a test sweep that varies in amplitude as if it were passed through the required preemphasis filter, producing a flat measurement response result.

Standard preemphasis and deemphasis curves are provided on the APx Resources disc as EQ tables. These include the RIAA phonograph curves and the 50 µs and 75 µs FM broadcast curves.

Characterization of a loudspeaker “golden unit”

In production loudspeaker testing, a common procedure is to characterize a well-performing loudspeaker as a “golden unit,” and to use its response curve as a basis for pass/fail limits. It is convenient to view the response of the golden unit as relatively flat, even if the true response is irregular.

Generator EQ

Generator EQ (equalization) allows you to vary the APx generator level by frequency in a sweep or multitone signal, in compliance with a curve defined in an EQ table. This enables compensation for a non-flat response in the device under test, as described in Stimulus Equalization.

> *An alternative to Generator EQ that can accomplish a similar goal (for analog outputs) is to insert a filter in the output signal path. See Output EQ, next.*

Measurements that can use Generator EQ

The Generator EQ feature addresses this requirement by allowing you to specify and attach an equalization table to the APx sine sweep or multitone generator for the following measurements:

- Acoustic Response
- Continuous Sweep
- Frequency Response
- Acoustic Response
- Stepped Frequency Sweep
- Multitone (see the next paragraph)

Operational details can be found in the chapters for Acoustic Response, Continuous Sweep, Frequency Response, Acoustic Response and Stepped Frequency Sweep.

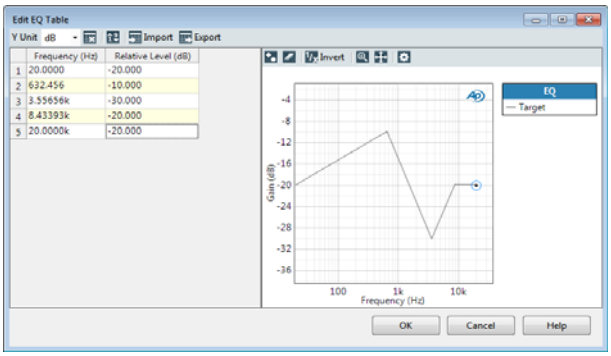
Multitone signal equalization

Since the multitone measurement does not use a swept sine signal, generator equalization cannot be applied in the same way. You can, however, effect the same functionality for multitone measurements. For multitone, the equalization is applied by adjusting the levels of individual tones during the multitone waveform creation process. See page 413.

Applying Generator EQ

For instructions for using a generator equalization table with specific measurements, see the EQ heading in the Common Controls section for Acoustic Response, Continuous Sweep, Frequency Response, Acoustic Response or Stepped Frequency Sweep. For Multitone, Import EQ Table on page 413.

Editing the Generator EQ table



This dialog allows you to import, edit and export generator equalization tables.

Add Point 

Add Point inserts a blank frequency point (row) below the current selection in the table. Enter a level value for each new point. You can edit the frequency or level value arbitrarily.

Delete Selected Points 

Delete Selected Points removes the selected points (rows) from the table.

Invert 

Use this control to invert the EQ curve. This is useful for creating a compensation curve from a measured response curve.

Zoom 

Click on this button and then drag a marquee in the graph to zoom.

Reset Zoom/Pan 

Click on this button to revert to the original zoom and pan views.

Graph Properties 

This button opens a Graph Properties dialog, where you can select ranges, names and log/lin characteristics for the EQ curve graph.

Clear Table 

Clear Table removes all points (rows) from the table.

Sort Ascending 

Sort Ascending sorts the table by frequency, ascending toward the bottom of the table.

Import 

Import opens a file browser window to select and open compatible EQ table files from disk. Excel, CSV and Audio Precision legacy *.adx and *.atsx are supported.

Export 

Export opens a file browser window to save the current EQ table as an Excel or CSV file.

Y Unit

Choose the Y-axis (level) unit here.

Comparison of Output EQ to Generator EQ.

	Output Equalization	*Generator Equalization*
Design	Arbitrary 30-pole filter approximating settings in Output EQ Table, implemented in the digital domain.	Run-time generator amplitude adjustment by frequency, corresponding to settings in Generator EQ Table.
Output Connector	analog outputs only	any analog or digital output
Signal source	any generator or file sourced signal, except AG52 special signals (square and DIM).	Sine signals used in Stepped Frequency Sweep, Acoustic Response, Frequency Response, Continuous Sweep, Loudspeaker Production Test and Multitone measurements only.
Channel count	two channels maximum	all channels
Availability	applied per signal path	applied per measurement
APx555 note	APx555 High-Performance Analog Generator: Output EQ is available as a User Interface choice, but is actually implemented by the Generator Equalization method.	

Output EQ

For analog outputs, **Output EQ** (equalization) allows you to insert an EQ filter in the output signal path, after the APx generator. This enables compensation for a non-flat response in the device under test, as described in Stimulus Equalization on page 191.

An alternative to Output EQ that can accomplish a similar goal for some measurements is to vary the APx generator level by frequency in a sweep or multitone signal. See Generator EQ, previous.

Creating or importing an EQ curve

Open the **EQ** menu in **Signal Path Setup: Output Configuration**. Choose

- **Create New** or
- **Browse for file**.

The EQ response curve is specified in the **Edit Output EQ Table** dialog, and response curves can be imported and saved. The response curve can be easily inverted.

Design

The equalization filter is implemented in the digital domain near the end of the output signal path, before the digital-to-analog converters (DACs). Th e filter has a fixed number of poles (30). The EQ response beyond the range specified in the curve is unspecified.

Application

The equalization filter is only applied to analog outputs.

The filter can be applied to a maximum of 2 channels. If more than two output channels are specified in **Signal Path Setup: Output Configuration**, the filter is made inactive.

The filter is applied to all audio signals generated in DSP, including sine wave, square wave, sweeps, IMD signals, multitone, noise and any non-coded arbitrary generator waveform signal output from files on disc, including POLQA and PESQ stimuli. The filter cannot be applied to the square wave and DIM signals generated in the AG52 option.

Generator output limiting

DAC generator

If the combination of generator level and offset settings and the applied Output EQ curve would require the APx generator to exceed its maximum level, the filter is scaled so that the maximum never exceeds a point 1 dB below the maximum generator output level (which varies with instrument hardware and configuration), while preserving the EQ curve shape. A warning is shown in the Status Bar when such a condition exists.

High Performance Sine Generator

If the combination of generator level setting and the applied Output EQ curve requires the APx generator to exceed its maximum level, the APx generator is held at its maximum level until the required level falls below the maximum. A warning is shown in the Status Bar when such a condition exists.

Generator output limiting when using complex waveforms

If you are testing with complex waveforms (such as speech waveforms for PESQ or POLQA) and find that the generator limiting feature of Output EQ reduces the level so far that you cannot attain the necessary Active Speech Level at the transducer, an alternative method is to use the APx EQ .wav Files Utility. Read about the utility by searching for the file “APx EQ .wav Files Utility Instructions.pdf” at ap.com.

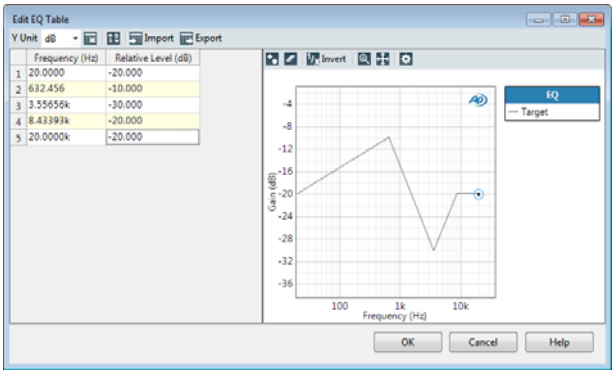
Interaction with Generator EQ

If **Generator EQ** is in effect for a measurement, it overrides **Output EQ**. **Output EQ** is disabled for that measurement.

Automation

You can import (and optionally invert) an Output EQ curve into **Signal Path Setup** during an automated sequence. See **Adding an Import Output EQ Curve Step** on page 551.

Editing the Output EQ table



This dialog allows you to import, edit and export Output EQ tables.

The points in the table are represented graphically by the **Target** trace in the graph. The **Design** trace represents the actual filter curve.

Add Point 

Add Point inserts a blank frequency point (row) below the current selection in the table. Enter a level value for each new point. You can edit the frequency or level value arbitrarily.

Delete Selected Points 

Delete Selected Points removes the selected points (rows) from the table.

Invert 

Use this control to invert the EQ curve. This is useful for creating a compensation curve from a measured response curve.

Zoom 

Click on this button and then drag a marquee in the graph to zoom.

Reset Zoom/Pan 

Click on this button to revert to the original zoom and pan views.

Graph Properties 

This button opens a **Graph Properties** dialog, where you can select ranges, names and log/lin characteristics for the EQ curve graph.

Clear Table

Clear Table removes all points (rows) from the table.

Sort Ascending

Sort Ascending sorts the table by frequency, ascending toward the bottom of the table.

Import

Import opens a file browser window to select and open compatible EQ table files from disk. Excel, CSV and Audio Precision legacy *.adx and *.atsx are supported.

Export

Export opens a file browser window to save the current EQ table as an Excel or CSV file.

Y Unit

Choose the Y-axis (level) unit here.

Input EQ

Input Equalization allows you to insert a filter into the input signal path for all audio inputs. This enables compensation for a non-flat microphone response in the system under test. Input Equalization can also be used to compensate for loudspeaker response in acoustic testing.

Creating or importing an EQ curve

Open the **EQ** menu in **Signal Path Setup: Input Configuration**. Choose

- **Create New** or
- **Browse for file**.

The EQ response curve is specified in the **Edit Input EQ Table** dialog, and response curves can be imported and saved. The response curve can be easily inverted.

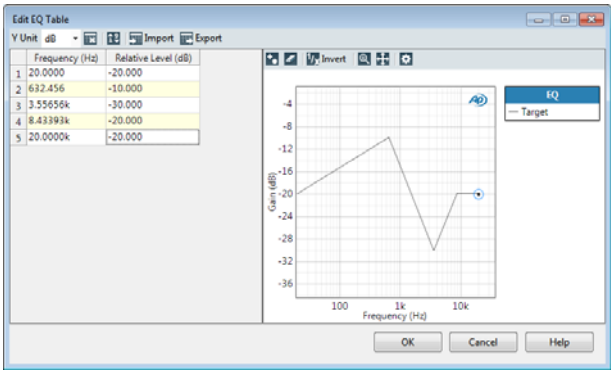
Design

The equalization filter is implemented in the digital domain early in the input signal path. The filter has a fixed number of poles (30). The EQ response beyond the range specified in the curve is unspecified.

Application

The equalization filter can be applied to all audio inputs, and also to demodulated jitter inputs. The filter can be applied to a selection of available channels.

Editing the Input EQ table



This dialog allows you to import, edit and export Output EQ tables. The points in the table are represented graphically by the trace in the graph.

Add Point

Add Point inserts a blank frequency point (row) below the current selection in the table. Enter a level value for each new point. You can edit the frequency or level value arbitrarily.

Delete Selected Points

Delete Selected Points removes the selected points (rows) from the table.

Invert

Use this control to invert the EQ curve. This is useful for creating a compensation curve from a measured response curve.

Zoom

Click on this button and then drag a marquee in the graph to zoom.

Reset Zoom/Pan

Click on this button to revert to the original zoom and pan views.

Graph Properties

This button opens a **Graph Properties** dialog, where you can select ranges, names and log/lin characteristics for the EQ curve graph.

Clear Table

Clear Table removes all points (rows) from the table.

Sort Ascending

Sort Ascending sorts the table by frequency, ascending toward the bottom of the table.

Import 

Import opens a file browser window to select and open compatible EQ table files from disk. Excel, CSV and Audio Precision legacy *.adx and *.atsx are supported.

Export 

Export opens a file browser window to save the current EQ table as an Excel or CSV file.

Y Unit

Choose the Y-axis (level) unit here.

Acoustic Response Measurements

Introduction

For the APx515, this measurement requires a software option key. See page 189 for more information about software options.

The Acoustic Response measurement is designed to facilitate loudspeaker testing, using an APx500 Series audio analyzer with a measurement microphone and an audio power amplifier.

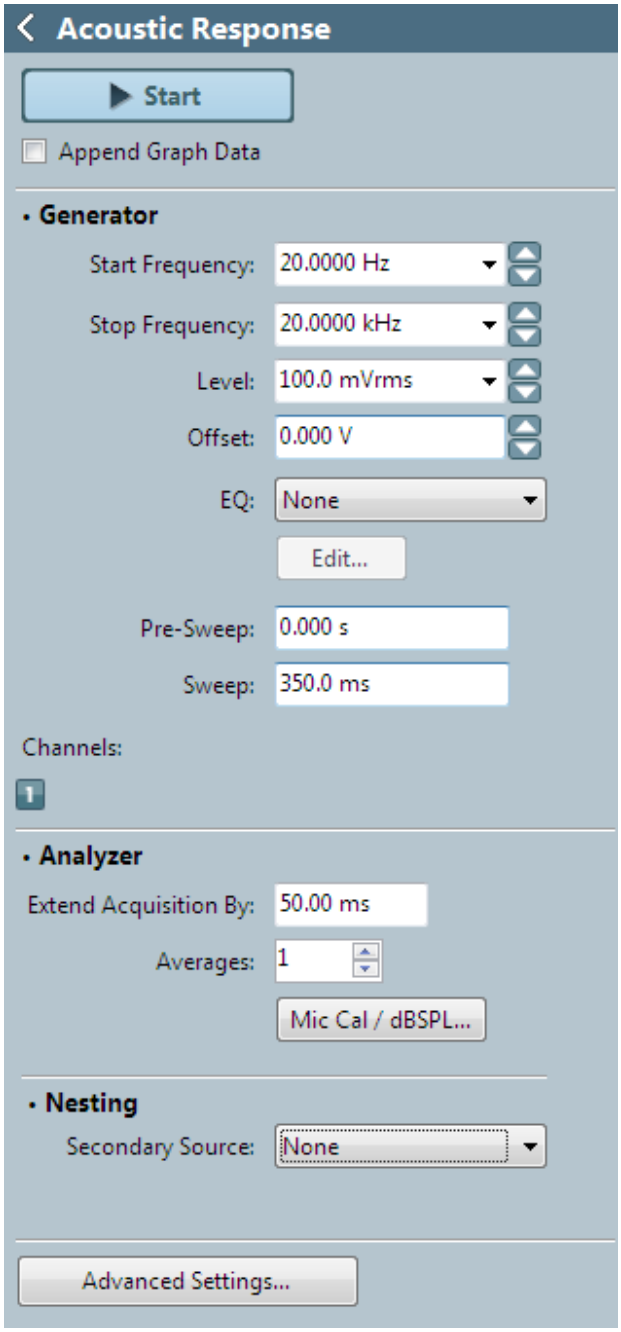
Typically, the APx500 generator output is connected to the power amplifier, which in turn is connected to the loudspeaker under test. A measurement microphone is located in the acoustic field of the loudspeaker, and the output of the microphone preamplifier is connected to an APx500 analyzer input. A specially configured continuous sweep measurement (called the Acoustic Response measurement) is made. See page 246 for more about continuous sweep.

Acoustic Response results available in APx500 are:

- Impulse Response
- Delay
- THD Level
- Energy Time Curve (ETC)
- Phase
- Distortion Product Ratio
- RMS Level (Freq. Response)
- Group Delay
- Distortion Product Level
- Relative Level
- Level and Distortion
- Rub and Buzz *
- Deviation
- THD Ratio
- Acquired Waveform

All of these results are available from a single acquisition.

** The Rub and Buzz results require a software option key. See page 189 for more information about software options.*



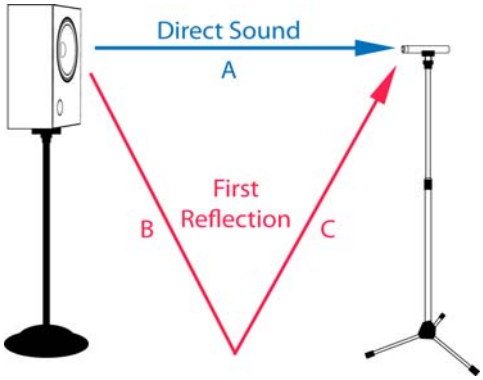
Loudspeaker Testing using Acoustic Response

Acoustic Response needs careful setup and precision third-party instruments to obtain good loudspeaker testing results. You must have a measurement micro-

phone and its associated power supply and preamplifier, a microphone calibrator, a suitable acoustic space, and in most cases a high-quality audio power amplifier.

This first section takes you through the basics of setting up a loudspeaker testing system, using Acoustic Response result tools such as Impulse Response and Energy Time Curve to optimize the APx processing for best measurements.

Positioning the Loudspeaker and Measurement Microphone in the Acoustic Space



Loudspeaker measurements can be made in various acoustic spaces: outdoors, within rooms, or even within special isolation enclosures. The choice of acoustic space will affect the optimal positioning of the loudspeaker and microphone, and will affect the measurement results. This is because it is desirable to measure only the sound traveling directly from the loudspeaker to the microphone, and to avoid measuring sound that has traveled a reflected path, from the loudspeaker to a reflecting surface and then to the microphone.

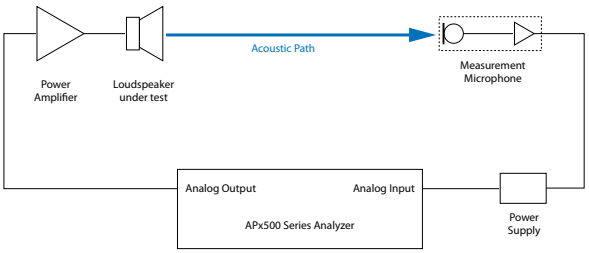
Close reflections are more troublesome than distant reflections, because they typically have greater amplitude and because they may reach the microphone before the impulse response of the system has decayed.

The best measurement results, then, are typically made in acoustic spaces that create minimal close reflections: outdoors, or in large rooms. Floors, walls, ceilings, tables and other objects reflect sound waves, and even in a large room care must be taken to mount the loudspeaker and microphone at a distance from any reflecting surface.

Sound travels at approximately 343 meters per second (1125 feet per second) in air at 20 °C (68 °F); this speed varies primarily with air temperature and humidity, and to a minor extent with pressure and density. This means that in ordinary conditions a sound wave takes approximately 3 ms to travel 1 m, or approximately 0.9 ms to travel 1 ft.

A standard distance between the loudspeaker under test and the measurement microphone is 1 m. Other distances can be used. Ideally, the path of first reflection is longer (in time) than the system impulse response.

Connecting the Loudspeaker and Measurement Microphone



Configuring APx500 for Acoustic Response measurements

After launching APx500, go to Signal Path Setup to configure the analog outputs and inputs for proper interface to the audio power amplifier and the measurement microphone.

Reduce the number of output and input channels to the number actually required for your test. This improves both system performance and clarity of display.

For best performance, reduce the system bandwidth to 40 kHz or even 20 kHz. Few loudspeakers have ultrasonic response, and the APx500 software runs more efficiently at reduced bandwidth.

Verify system electrical and acoustic paths

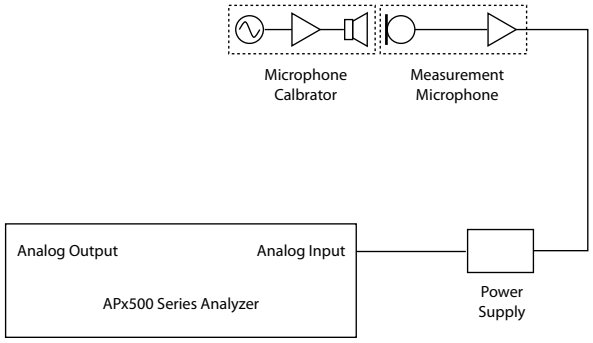
In Verify Current Connections, be sure that the generator level is not set too high, and that the generator frequency is appropriate for your loudspeaker. If the audio power amplifier has high gain or the loudspeaker is very sensitive, it may be possible to damage the loudspeaker, even at the default generator setting of 100 mVrms. You may want to set Global Vmax (page 28).

Click the Generator **ON** and verify that the loudspeaker and microphone are connected and operating properly.

Calibrating the Measurement Microphone and APx500

It is not required that loudspeaker measurements be made in reference to an acoustic calibrator, but it is often done. APx500 can use an acoustic calibration as a dBSPL reference and report measurement results against that reference.

You must have an acoustic calibrator such as the MMC-3 available from Audio Precision to calibrate the system.



Two calibration levels are in standard use for acoustic calibration: 94 dB SPL and 114 dB SPL. If both are available on your calibrator, select the level most suited to your measurement. Mount the calibrator firmly on the microphone capsule and turn it ON.

In APx500, go to Signal Path Setup > References > Input References and click **Mic Cal / Set dBSPL...** In the **Calibrator Level** field, enter the calibrator setting (94 dB SPL or 114 dB SPL, set in the previous step. Then select the input channel (usually Channel 1) and click Set 1 to enter that level as dBSPL1. See Mic Cal / Set dBSPL on page 60.

Operation

If you have not yet set up your test, first go to Signal Path Setup to select and configure inputs and outputs, references and other settings. See Chapter 7 for information about Signal Path Setup.

Acoustic Response has a number of settings that are explained in detail in the following pages, and it can appear to be complicated to operate.

However, most of the settings are similar to Continuous Sweep or Frequency Response and are straightforward. The key difference is working with the Time Window (page 200) to focus the measurement on the desired region of the acquired data.

Append Graph Data

Measured data are grouped in a **Data Set**, which contains all acquisition results. Normally, the current graph data are deleted each time you start a new measurement. If the **Append Graph Data** box is checked, appended data are grouped as additional data sets, as are any imported data. The Data Sets Panel displays the Data Set(s) in the measurement, and provides a visibility checkbox, a Time acquisition field, and a Notes field for each data set.

Read more about Data Sets beginning on page 649.

Generator

The Acoustic Response measurement uses a log-swept sine chirp stimulus signal, swept between the frequencies entered in **Start Frequency** and **Stop Frequency**, at the value set in **Level**.

> *Note: Loudspeakers under test can be damaged by signals that exceed their level or frequency limits. Be sure to set generator and amplifier level and sweep frequencies to values within the limits of the loudspeaker.*

Running the Measurement

To use Acoustic Response, click **Start**. The generator will output the test signal to the DUT on the selected generator channels. Read the results in the selected result view.

Start Frequency and Stop Frequency

These controls set the frequency range of the sweep.

> *Note that the actual sweep stimulus signal begins below the defined Start Frequency and is swept upward, reaching the Start Frequency at the end of the Pre-Sweep time, as described below in Pre-Sweep and Sweep Duration Fields. Further, the stimulus signal is actually swept to 110% of the designated Stop Frequency (or to the Nyquist frequency, whichever is lower).*

See page 246 for more about continuous sweep ranges, which are dependent upon the instrument.

Level

Set the generator level here.

Offset

Set any DC offset to the generator signal here.

EQ

You can equalize the generator signal to make a pre-emphasized stimulus. See Generator Equalization (page 192) for a general discussion of this feature.

Choosing generator equalization

Select **None, Relative** or **Absolute** from the EQ drop-down menu.

- **None** applies no equalization to the generator.
- **Relative** applies the gain or loss specified in the EQ table to the current generator level for each channel.
- **Absolute** sets the generator level for all channels to the level specified in the EQ table. Note: when **Absolute** is selected, current channel level settings are lost.

Edit

If you have chosen to equalize the generator signal, you can edit the EQ table here. See **Edit EQ Table** (page 192).

Pre-Sweep and Sweep duration fields

A continuous sweep stimulus consists of a sine wave swept logarithmically across the sweep range in the period of time set in these fields.

To enable the DUT to stabilize before the beginning of the defined sweep measurement, there is variable **Pre-Sweep** time. The actual sweep stimulus signal begins below the defined **Start Frequency** and is swept upward, reaching the **Start Frequency** at the end of the **Pre-Sweep** time. The sum of the times in **Pre-Sweep** and **Sweep** is the total sweep length.

- The default for **Pre-Sweep** is 0 s. Minimum is 0 s; maximum is 1.0 s.
- The default for **Sweep** is 350.0 ms. Minimum is 50.0 ms; maximum is 5 s.

Longer **Pre-Sweep** settings allow more time for DUT stabilization; longer **Sweep** settings provide greater resolution and signal-to-noise ratio. Of course, increasing either setting will increase measurement time.

Analyzer

Extend Acquisition By

By default, a continuous sweep measurement makes an acquisition slightly longer than the stimulus, to include possible time-delayed artifacts created in the DUT. By default, the acquisition is extended 50 ms longer than the stimulus.

In some cases, you may want to extend the acquisition further. Enter a new value in the **Extend Acquisition By:** field. Minimum extension is 0 s; maximum is 3 s.

Averages

Averaging several sweeps can remove uncorrelated data from the results (for acoustic measurements, this is usually background noise). The sweep will be run the number of times set in the **Averages** field, and the results will be averaged for display.

Mic Cal / dBSPL...

This button opens the **Microphone Calibration / Set dBSPL Per-channel** dialog, also available from the Signal Path Setup > Input Configuration and References panels when in Acoustic Mode.

Nesting

Acoustic Response sweeps can be nested. Read about Nested Sweeps beginning on page 185.

Secondary Source (nested sweeps)

Select the secondary sweep source here. This is the outside, nesting parameter that is varied to change the conditions for the primary sweep through a number of iterations.

Available choices are dependent upon instrument hardware, options, Input/Output settings and Bench Mode Generator settings.

Advanced Settings

Advanced Settings include output level and offset settings and input ranging settings. To view or modify these settings, click **Advanced Settings**. See Advanced Settings for Continuous Sweep based measurements on page 249.

Adjusting the Time Window

The Time Window is a unique feature of the Acoustic Response measurement that is used to select the portion of the acquired data that is used for analysis. The full set of controls for the Time Window are available from the Impulse Response result panel. The Impulse Response and Energy Time Curve (ETC) views are useful in adjusting the Time Window to optimize the measurement results.

For loudspeaker measurement, the Time Window is typically used to select only the direct sound from the loudspeaker, excluding reflections from the acoustic environment.

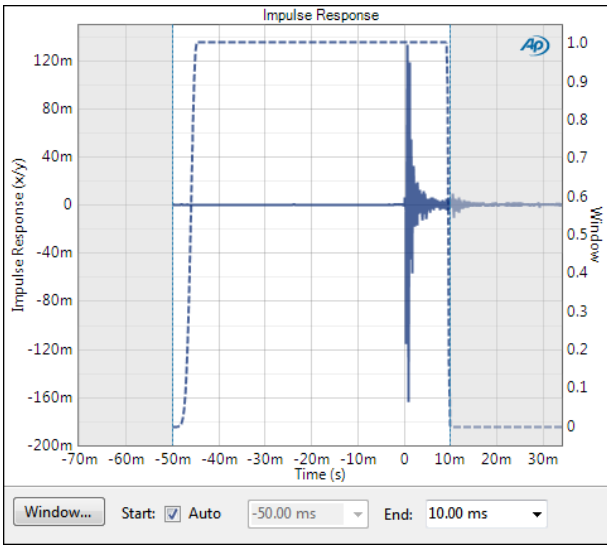
In an open project, the Time Window can be adjusted and readjusted at will, and will affect all the current measured data. When a project is saved, the latest Time Window is saved with the project. When the project is opened, the Time Window associated with the saved data can be viewed on the Impulse Response and ETC results for those measurements, but cannot be adjusted. New appends have an active Time Window that can be fully adjusted.

Time Window display

Gray areas on Impulse Response and ETC result graphs

The regions of the Impulse Response and ETC results that are outside of the Time Window are partially obscured by a gray mask superimposed on the graph.

Additionally, the Impulse Response graph shows the shape of the Time Window as a dashed line.



Gray areas on other result graphs

The adjustment of the Time Window excludes certain time-domain regions from analysis. This affects the reliability of low-frequency measurements in frequency-domain results. In these results, you will notice that the graphs show a gray mask overlaying a portion of the low frequencies. This indicates that the measurement results in the masked area are unreliable. The masked area will change as the Time Window is adjusted.

Time Window Start and End time settings

On the Impulse Response and ETC graphs, 0 on the X-axis corresponds to the start time of the APx generator (the instant that the first audio sample is output when **Start** is pressed). An additional 50 ms is added to the acquisition before 0 to include any activity within the APx system, such as internal ADC/DAC ringing. (This is required for correct processing of the continuous sweep acquisition).

Time Window Controls

These controls duplicate the controls available in the Time Window Settings dialog and on the Impulse Response result panel.

Start and End

By default, **Start: Auto** is checked, causing the beginning of the Time Window to correspond to the beginning of the acquisition, at about –50 ms.

In most cases, the end of the Time Window should be set to a point just before the first reflection by entering a value in the Time Window **End** field, or by dragging the Time Window cursor on the graph using the mouse cursor. This is the default behavior, and is correct for the majority of acoustic tests.

In Loopback, the default Time Window settings place the peak of the impulse response at 0. In real testing in an acoustic environment, device and acoustic delay will delay the acquired signal, placing the peak of the impulse later on the graph. The arrival time of the impulse shown on the Impulse Response graph is the same as the value displayed in the **Delay** result.

In some circumstances, however, acoustic reflections or ambient noise can cause the **Start: Auto** Time Window span to be chosen incorrectly (typically, in situations where the amplitude of the first reflection is larger at the microphone than the amplitude of the initial signal).

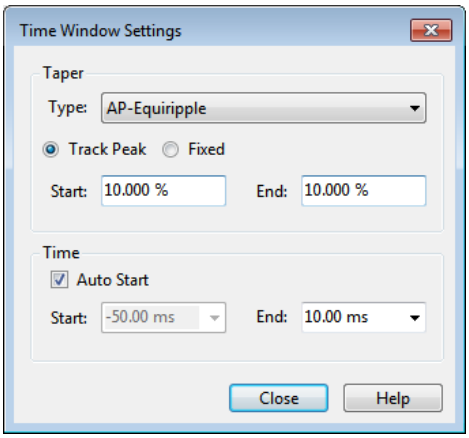
In such a case, deselect **Start: Auto**. The start time will be reset to a default value of 0.00, and the **Start** field will allow user input. You will now be able to adjust both Time Window **Start** and Time Window **End** (by dragging with the mouse cursor or by entering a new value into the **Time Window** fields) to include the desired portion of the signal for analysis.

Time Window Type and Taper

A Time Window selects regions of the time-domain acquisition to include or exclude from analysis. A window provides a central flat, rectangular region with tapers on each side that smoothly transition from 0 to 1 (for the **Start** taper) and 1 to 0 (for the **End** taper).

See Legacy Time Windows on page 211 for Time Window behavior in APx versions prior to version 4.2.

The **Auto: Start** Time Window setting, when combined with a carefully selected **End** time is correct for most applications. However, you can further refine the shape of the Time Window using controls in the Time Window Settings dialog. Click the **Window** Window... button on the Impulse Response result panel to open the Time Window Settings dialog.



Taper

Type

The **Type** menu offers a list of taper shapes, each of which uses the specified type of windowing function (page 493) to window the impulse response before calculating its FFT. The AP-Equiripple window is the default. The **Start** and **End** window tapers follow the shapes available from this menu.

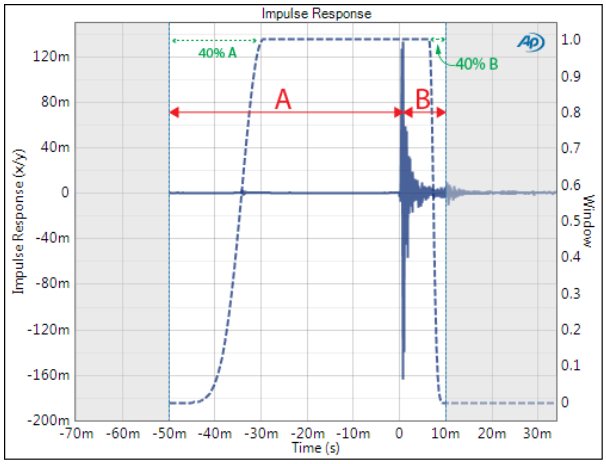
- **Track Peak**
  Track Peak aligns the window to the peak of the impulse in a specific way for each acquisition. This can cause the window to include or exclude slightly different ranges across multiple acquisitions.
  In **Track Peak**, the window is designed with two regions, shown as A and B in the illustration below. The A region extends from **Start** time to the impulse peak, and the B region extends from the impulse peak to **End** time.
- **Fixed**
  Fixed sets the window at the same place in time (defined by **Start** and **End** times) across acquisitions. The **Start** and **End** taper percentages are calculated for the entire window width, without respect to regions.

Start

Start defines the width of the **Start** taper as a percentage of the A region (for **Track Peak**), or as a percentage of the total width of the window (for **Fixed**). By default, the **Start** taper is set to 10%.

End

End defines the width of the **End** taper as a percentage of the B region (for **Track Peak**), or as a percentage of the total width of the window (for **Fixed**). By default, the **End** taper is set to 10%.



Impulse Response graph using AP-Equiripple window, Track Peak, with Start and End tapers set to 40%.

Time

The Time controls on this dialog duplicate the settings on the Impulse Response result panel.

Note: The Auto Start Time checkbox and Time Window Start feature are not available for the acoustic response component of the Loudspeaker Production Test measurement.

Example

Let's run through this procedure once as an example, using the default Time Window settings.

Set up and run an Acoustic Response measurement. In most acoustic environments, a strong second peak can be observed that corresponds to the first acoustic reflection.

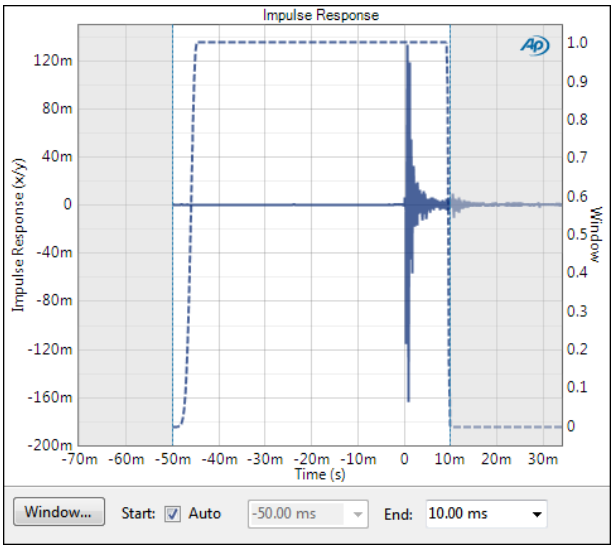
You can make this adjustment in either the Impulse Response view or the ETC view, but the peak is often more easily viewed in the ETC result.

In the ETC view, move the Time Window **End** by dragging it with the mouse cursor to a point just before the first reflection, as shown in the ETC graph on page 203.

Confirm the correct placement of the Time Window **End** this by observing the first reflection peak and the Time Window end placement in the Impulse Response view, which should look like the graph on page 202.

Further confirm by measuring the lengths of the acoustic paths, and obtaining the difference between the direct path and the first reflection path (B+C-A, as shown in the diagram on page 198). That distance should agree closely with the position of the Time Window **End**, when using distance units.

Impulse Response



The Impulse Response provides a graphical display of the IR of the loudspeaker and acoustic space. This view is useful in setting up an acoustic measurement in conjunction with the Energy Time Curve view, using

the Time Window. For more information about impulse response, see page 244.

Time Window controls

These controls duplicate the controls available in the Time Window Settings dialog and on the ETC result panel.

Window...

The **Window** Window... button opens the Time Window Settings dialog, discussed on page 200.

Auto:

When **Auto** is checked, the Time Window **Start** time is automatically set. See page 200.

Start:

When **Auto** is not checked, you can enter a Time Window **Start** time here. Alternatively, you can drag the Time Window **Start** on the graph using the mouse cursor. Use the drop-down menu to select Time Window units.

End

You can enter a Time Window **End** time here. Alternatively, you can drag the Time Window **End** on the graph using the mouse cursor. Use the drop-down menu to select Time Window units.

Display Interpolation

For time-domain displays such as Impulse Response, ETC and Acquired Waveform, the APx500 graphs are normally plotted using sinc function interpolation. Interpolation adds points to the displayed trace that do not exist in the measured data, to make data trends more easily visualized. The default in APx is to have Interpolation **On** for time-domain displays. However, digital domain signals are sometimes best understood when viewing actual samples, with interpolation switched **Off**. Go to Graph Properties (page 646) to set interpolation **On** or **Off**.

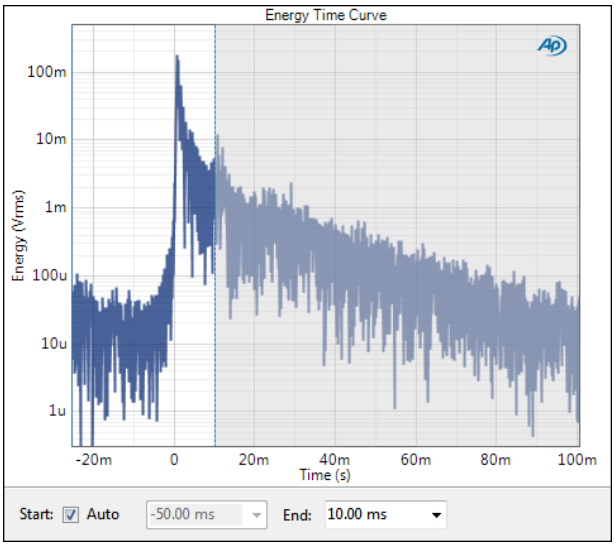
Interpolation can cause limit failure markers to appear to be in an incorrect location. See “Interpolation and limit failure markers” on page 678 for an explanation.

Units

Units available for Impulse Response are:

X-axis	*Y-axis same-domain*	*Y-axis cross-domain*
• s (seconds)	• x/y	• D/V — or— • V/D

Energy Time Curve (ETC)



The Energy Time Curve (ETC) provides a graphical display of the ETC of the loudspeaker and acoustic space. This view is useful in setting up an acoustic measurement. See More About the Energy Time Curve on page 210.

Time Window controls

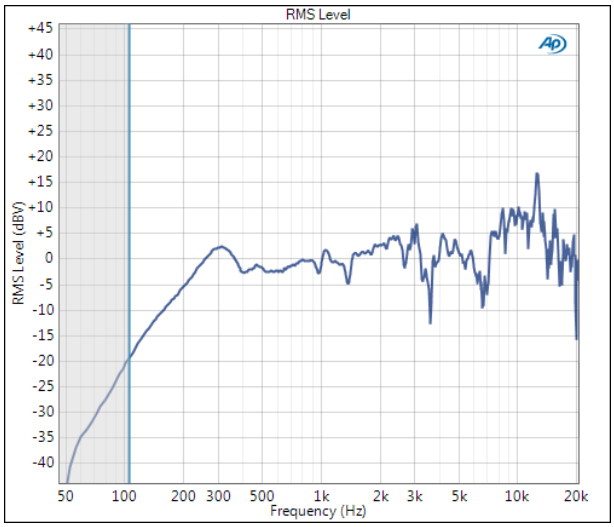
These controls duplicate the controls available in the Time Window Settings dialog and on the Impulse Response result panel.

Units

Units available for Energy Time Curve are:

X-axis	*Y-axis (analog)*	*Y-axis (digital)*
• s	• Vrms • dBV • dBu • dBrA • dBrB • dBSPL1 • dBSPL2 • dBm • W (watts)	• dBFS • FS • %FS • dBrA • dBrB • dBSPL1 • dBSPL2

Level (Frequency Response)



The Level result provides a graphical display of the frequency response of the loudspeaker.

Result Settings

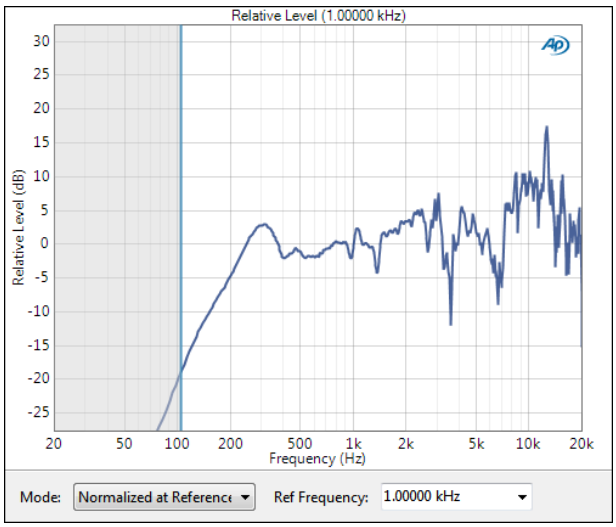
These settings are made in the Result Settings bar, beneath the graph display.

Units

Units available for Acoustic Response Level are:

X-axis	*Y-axis (analog)*	*Y-axis (digital)*
• Hz	• Vrms	• dBFS
• dHz	• dBV	• FS
• F/R	• dBu	• %FS
• %Hz	• dBrA	• dBrA
	• dBrB	• dBrB
	• dBSPL1	• dBSPL1
	• dBSPL2	• dBSPL2
	• dBm	
	• W (watts)	

Relative Level



The Relative Level result is a continuous sweep measurement that provides a graphical display of the frequency response of each channel. In this result the DUT output level is plotted against frequency, relative to the level at a selected frequency set in the **Ref Frequency** field.

This enables you to specify the frequency that will be set as 0 dB and view the response in relation to that frequency. A midband frequency such as 1 kHz is often set as the reference frequency, and is the default here.

Mode

Normalized at Reference

Enter a reference frequency for normalization in the **Ref Frequency** field. The measured level at the frequency you choose is set as 0 dB on the graph, and the DUT response is plotted in relation to this reference. You can change the **Ref Frequency** at any time (except after appending) and the graph will immediately be redrawn to reflect the new setting.

Auto-Centered in Limits

This control is only usable if upper and lower limits are set on this graph. In some applications, the general conformance of a trace to the shape of a set of limits is of importance, with absolute level less important or irrelevant. **Auto-Centered in Limits** centers the data trace relative to the upper and lower limits. To exclude possible exceptional data points at frequency extremes, the frequency range that is considered for the centering operation is constrained by the values in the **Min** and **Max** fields.

Result Settings

These settings are made in the Result Settings bar, beneath the graph display.

Ref Frequency

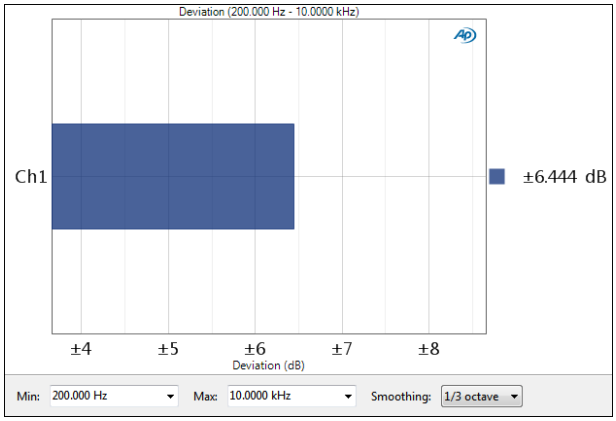
Choose a Reference Frequency here.

Units

Units available for Acoustic Response Relative Level are:

X-axis	*Y-axis*
• Hz	• x/y
• dHz	• %
• F/R	• ppm
• %Hz	• dB

Deviation



The Acoustic Response Deviation result is a single value measurement computed from the continuous sweep acquisition. In this result the frequency deviation (the total range of frequency variation) of each channel is displayed as a meter bar. You can specify a minimum and maximum frequency to define the range be considered in the deviation measurement.

Result Settings

These settings are made in the Result Settings bar, beneath the graph display.

Min

Set the minimum frequency of the range of interest here.

Max

Set the maximum frequency of the range of interest here.

Smoothing

Octave smoothing is a common technique in loudspeaker response measurement, useful in revealing trends by smoothing out anomalies in the response curve. The APx500 implementation uses a hybrid FFT bin averaging and interpolation technique to achieve smooth results even at very low bin densities.

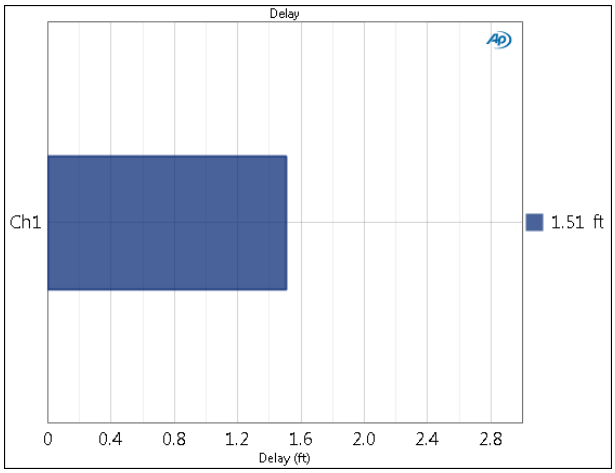
Smoothing effectively passes the raw response data through a constant-Q bandpass filter that is swept with the continuous sweep signal. The bandwidth of this filter is selected in the **Smoothing** field.

Units

Units available for Acoustic Response Deviation are

- dB

Delay



The Acoustic Response Delay result is a single value measurement computed from the impulse response. Delay displays the time delay between the generator output and analyzer input for each channel. For Acoustic Response testing, this is effectively the acoustic delay between the loudspeaker and the measurement microphone.

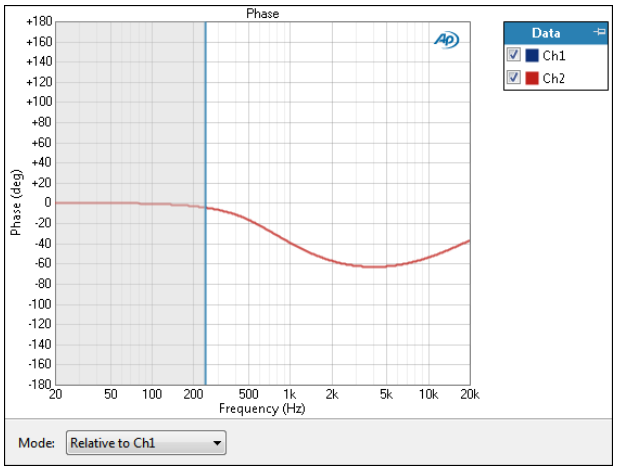
This result is primarily useful for setting up an Acoustic Response test, in conjunction with the ETC view and the Impulse Response view.

Units

Units available for Acoustic Response Delay are

- dB

Phase



The Acoustic Response Phase result is a continuous sweep measurement that displays the DUT output phase for each channel, plotted against frequency.

See page 332 for more information about phase measurements.

Result Settings

These settings are made in the Result Settings bar, beneath the graph display.

Mode

Acoustic Response Phase has four result modes:

Relative to Ch1

In this mode, the absolute phase of channel one is subtracted from the absolute phase of channels numbered greater than 1. The result is plotted against frequency for each channel numbered 2 and above, showing the phase difference (from channel 1) for each channel. Since channel 1 is used as the reference, it is not plotted in this result. This mode shows "unwrapped" phase differences.

Input-to-output

In the input-to-output mode, the absolute phase of each channel, from device input to device output, is plotted against frequency, "unwrapped." Input-to-output mode includes device delay.

Input-to-output, wrapped

This mode shows the same result as input-to-output phase, but "wrapped" within the range of –180° to +180°.

Input-to-output, excess

This mode shows the input-to-output phase (unwrapped), but removes the linear component (the average group delay of the system), leaving the "excess phase."

Some devices have linear phase only over a portion of their passband. For such devices, including frequency ranges where the phase is non-linear affects the excess phase result, which becomes dependent on the stop frequency of the measurement. Although the measurement is technically correct, the variation of the result with stop frequency is disconcerting and not particularly useful.

You may (optionally) use the **Min**: and **Max**: settings to set the expected linear phase range of the device under test. For this result, the device delay is computed as the best linear fit to the phase data between the **Min** frequency and the **Max** frequency. Excess phase is then computed as the difference from the linear fit.

Min and **Max** frequencies cannot extend beyond sweep **Start** and **Stop** frequencies.

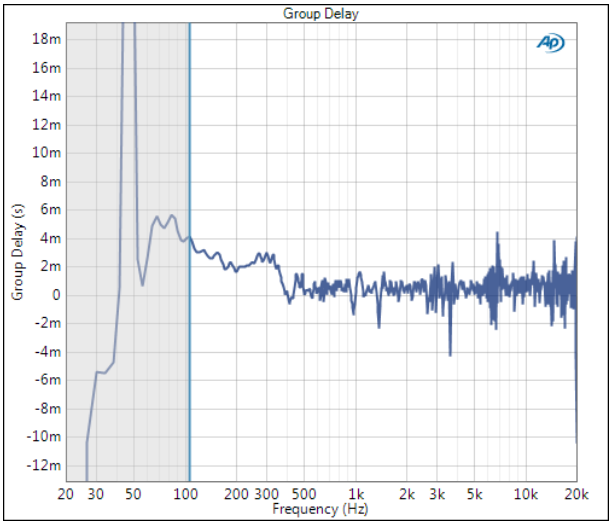
See page 241 for more information about wrapped and unwrapped phase.

Units

Units available for Acoustic Response Phase are:

X-axis	*Y-axis*
• Hz	• deg
• dHz	• rad
• F/R	
• %Hz	

Group Delay



Group delay is a measure of the rate of change of phase shift as a function of frequency. Group delay can also be described as a time delay of a group of frequencies with respect to the generator.

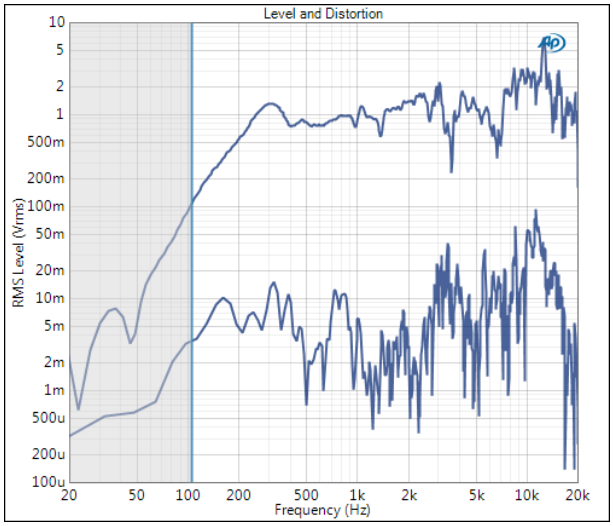
The Group Delay result is a continuous sweep measurement that displays the group delay measured in each channel, plotted against frequency.

Units

Units available for Acoustic Response Group Delay are:

X-axis	*Y-axis*
• Hz	• s (seconds)
• dHz	
• F/R	
• %Hz	

Level and Distortion



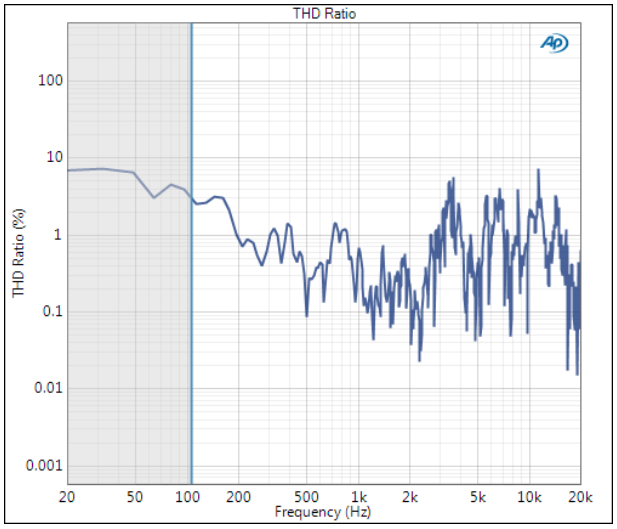
The Acoustic Response: Level and Distortion result is a continuous sweep measurement that provides a graphical display of the fundamental level vs. frequency and the distortion level vs. frequency of the loudspeaker connected to each channel, plotted on the same graph. Distortion curves included total distortion level, H2 distortion level and H3 distortion level.

Units

Units available for Acoustic Response Level and Distortion are:

X-axis	*Y-axis (analog)*	*Y-axis (digital)*
• Hz	• Vrms	• dBFS
• dHz	• dBV	• FS
• F/R	• dBu	• %FS
• %Hz	• dBrA	• dBrA
	• dBrB	• dBrB
	• dBSPL1	• dBSPL1
	• dBSPL2	• dBSPL2
	• dBm	
	• W (watts)	

THD Ratio



The THD Ratio result is a continuous sweep measurement that provides a graphical display of the harmonic distortion response of each channel. In this result the ratio of the level of the THD (total harmonic distortion) to the total signal in the DUT output is plotted against frequency.

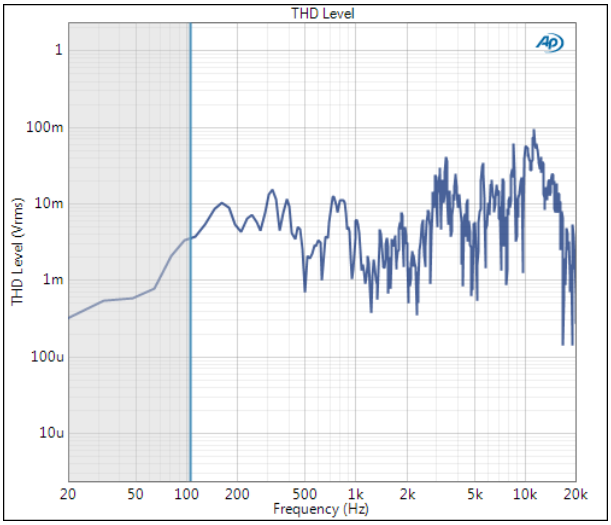
See page 242 for more information about THD.

Units

Units available for Acoustic Response THD Ratio are:

X-axis	*Y-axis*
• Hz	• x/y
• dHz	• %
• F/R	• ppm
• %Hz	• dB

THD Level



The THD Level result is a continuous sweep measurement that provides a graphical display of the har-

monic distortion response of each channel. In this result the level of the THD (total harmonic distortion) in the DUT output is plotted against frequency.

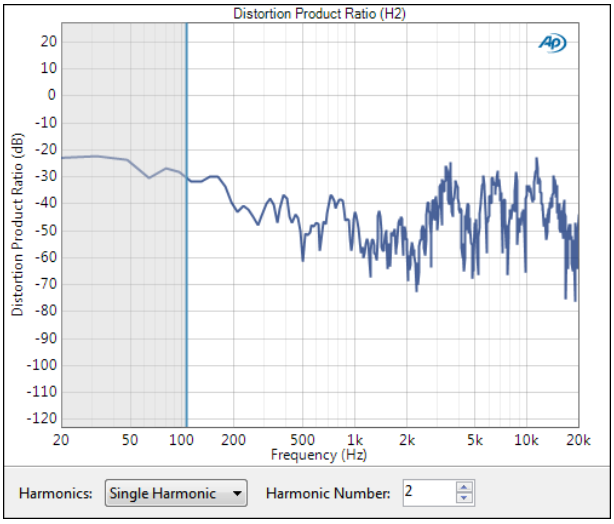
See page 242 for more information about THD.

Units

Units available for Acoustic Response THD Level are:

X-axis	*Y-axis (analog)*	*Y-axis (digital)*
• Hz	• Vrms	• dBFS
• dHz	• dBV	• FS
• F/R	• dBu	• %FS
• %Hz	• dBrA	• dBrA
	• dBrB	• dBrB
	• dBSPL1	• dBSPL1
	• dBSPL2	• dBSPL2
	• dBm	
	• W (watts)	

Distortion Product Ratio



The Acoustic Response Distortion Product Ratio result is a continuous sweep measurement that provides a graphical display of the selected harmonic distortion products present in each channel. In this result the ratio of the level of the selected harmonic distortion product to the fundamental in the DUT output is plotted against frequency.

Additional Controls for Distortion Product Ratio

Harmonics

For a graph of the ratio of one specific harmonic product to the signal level, select **Single Harmonic** and set the **Harmonic Number** to your choice, from 2nd harmonic to 20th harmonic.

For a graph of the sum of the levels of any combination of harmonic products (from **H2** through **H20**), divided by the signal level, select **Sum of Harmonics**.

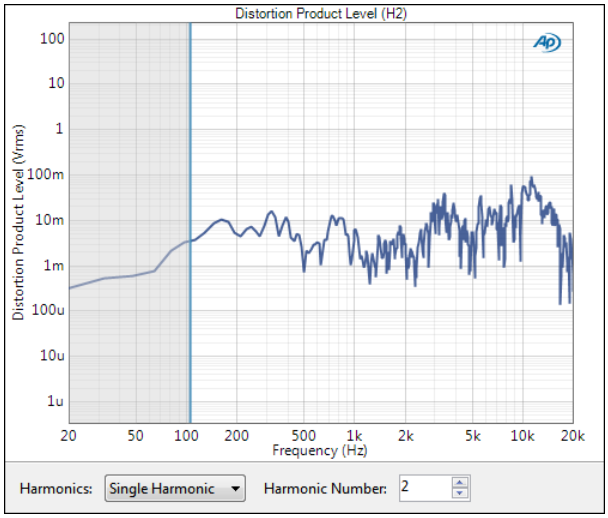
Then click **Select Harmonics...** to open a result definition dialog.

Units

Units available for Acoustic Response Distortion Product Ratio are:

X-axis	*Y-axis*
• Hz	• x/y
• dHz	• %
• F/R	• ppm
• %Hz	• dB

Distortion Product Level



The Acoustic Response Distortion Product Level result is a continuous sweep measurement that provides a graphical display of the selected harmonic distortion products present in each channel. In this result the level of the selected harmonic distortion product in the DUT output is plotted against frequency.

Additional Controls for Distortion Product Level

Harmonics

For a graph of the level of one specific harmonic product, select **Single Harmonic** and set the **Harmonic Number** to your choice, from 2nd harmonic to 20th harmonic.

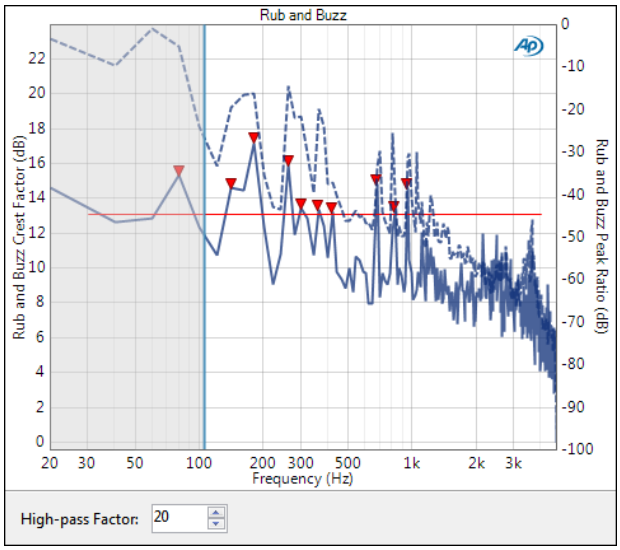
For a graph of the level of the sum of several or all harmonic products (from **H2** through **H20**), select **Sum of Harmonics**, then click **Select Harmonics**... to open a result definition dialog.

Units

Units available for Acoustic Response Distortion Product Level are:

X-axis	*Y-axis (analog)*	*Y-axis (digital)*
• Hz	• Vrms	• dBFS
• dHz	• dBV	• FS
• F/R	• dBu	• %FS
• %Hz	• dBrA	• dBrA
	• dBrB	• dBrB
	• dBSPL1	• dBSPL1
	• dBSPL2	• dBSPL2
	• dBm	
	• W (watts)	

Rub and Buzz



The Rub and Buzz result requires a software option key. See page 189 for more information about software options.

The Acoustic Response: Rub and Buzz result uses a continuous sweep measurement to display the Rub and Buzz Crest Factor and Peak Ratio for each channel. See More about Loudspeaker Testing on page 211 for a more detailed discussion of using rub and buzz and modulated noise in loudspeaker testing.

Rub and Buzz presents two results on one dual axis graph. In the APx implementation, detection of rub and buzz requires a simultaneous interpretation of both of these results; a rub and buzz crest factor result is plotted in reference to the left Y-axis, and a rub and buzz peak ratio result is plotted in reference to the right Y-axis. Limit behavior is different on a graph with dual axes. See Drawing limits for XY Graphs (dual axis) on page 673 and Editing limits for XY Graphs (dual axis) on page 673.

The next two results are simply independent views of the Crest Factor and Peak Ratio views that are combined on this view.

You can split the dual axis graph into two individual graphs. Right-click on the Rub and Buzz result and choose Split from the context menu. Two new results will be added, presenting the data for the Left and Right axes individually.

Additional Controls for Rub and Buzz

High-pass Factor

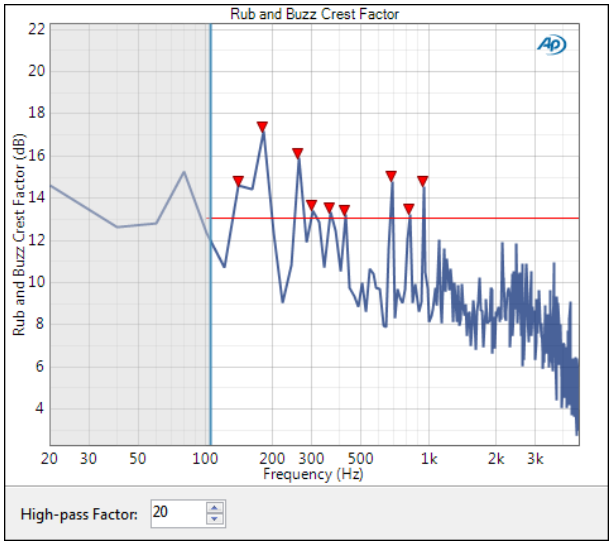
The data plotted on these graphs is high-pass filtered to reveal the rub and buzz signature results. The filter tracks the chirp stimulus frequency, moving upward with the sweep. The corner frequency of the filter is the product of the instantaneous chirp frequency multiplied by the **High-pass Factor**, which can be set in the range from 5 to 30. This results in a graph where the X-axis represents the stimulus frequency, and the plotted data represent the much higher frequency sounds generated by the interaction of the stimulus frequency with loose particles or rubbing components within the loudspeaker system. For interpretation of the results, see More About Loudspeaker Testing on page 211.

Units

Units available for Rub and Buzz are:

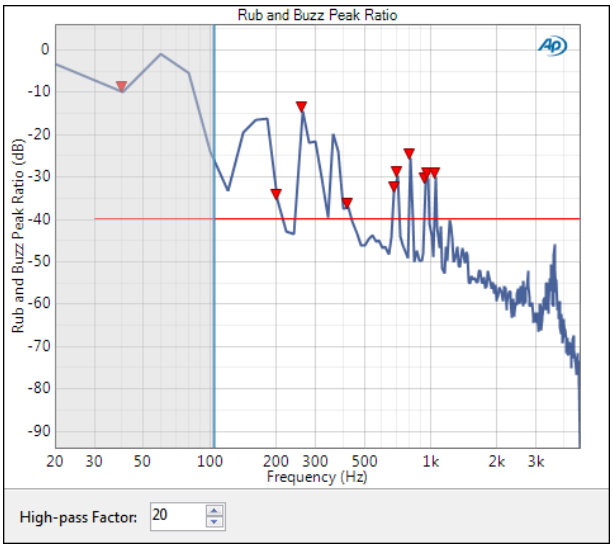
X-axis	*Y-axes*
• Hz	• x/y
• dHz	• %
• F/R	• ppm
• %Hz	• dB

Rub and Buzz Crest Factor



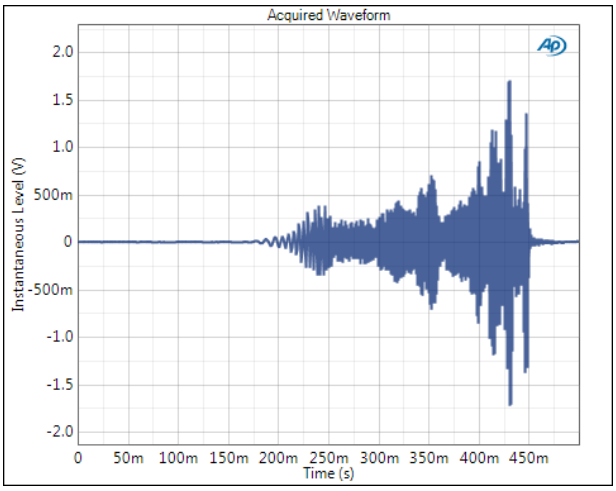
This graph is the Crest Factor trace from the Rub and Buzz result, displayed without the overlay of the Peak Ratio trace.

Rub and Buzz Peak Ratio



This graph is the Peak Ratio trace from the Rub and Buzz result, displayed without the overlay of the Crest Factor trace.

Acquired Waveform



The Acquired Waveform view is a continuous sweep measurement that displays the acquired waveform for each channel. Acquired waveform is plotted against time.

Interpolation

For time-domain displays, the APx500 graphs are normally plotted using sinc function interpolation. Interpolation adds points to the displayed trace that do not exist in the measured data, to make data trends more easily visualized. The default in APx is to have Interpolation **On**. However, digital domain signals are sometimes best understood when viewing actual samples, with interpolation switched **Off**. Also see “Interpolation and limit failure markers” on page 678.

Units

Units available for Acquired Waveform are:

X-axis	*Y-axis (analog)*	*Y-axis (digital)*
• s (seconds)	• V	• D • hex

More about ETC

The acquired Acoustic Response data is transformed into the frequency domain where further processing is performed, including the application of a Hilbert Transform. This result is then inverse transformed back to the time domain for display. This result is commonly called the energy time curve, or ETC.

The ETC is only an approximation of the actual energy arriving at the microphone, since this energy can never be known without the simultaneous measurement of both the velocity and the pressure of the sound; the term energy time curve is actually a misnomer. A more accurate term sometimes found in technical papers is the “analytic signal magnitude,” but as “energy time” has become common usage that term is used here.

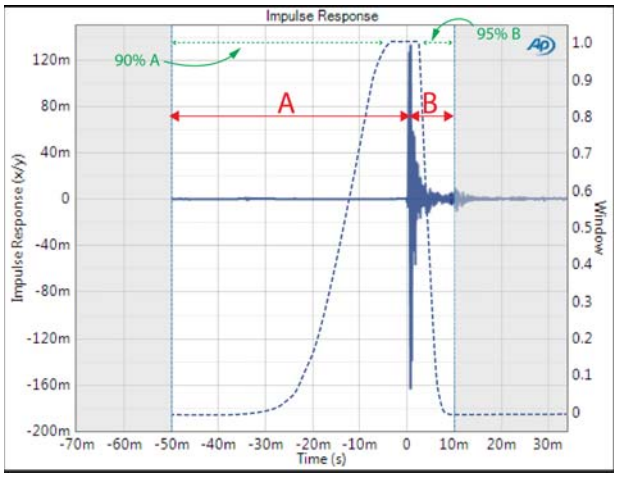
Interpretations of ETC measurements can be misleading. The Energy Time Curve view is included in Acoustic Response as an aid in identifying first reflections for optimal setting of the Time Window.

The energy time curve is useful for determining arrival times and relative energy distribution in time. It is the log magnitude of the impulse response, and may be considered as similar to the envelope of the IR, tracing along the top of the IR curve but not showing any of the negative excursions.

Legacy Time Windows

Acoustic Response time windowing features have been made more powerful and flexible with version 4.2. Some users may find it desirable to maintain the Time Window shapes available in earlier versions. The Time Window in APx version 4.2 can be set to match the windows used in earlier versions, which are discussed next.

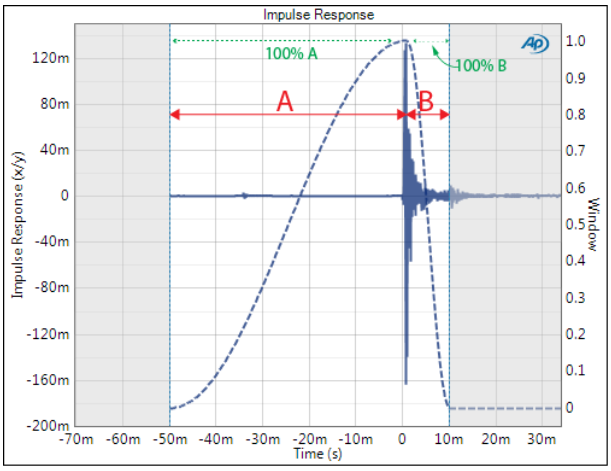
Time Windows in APx500 v2.x through v4.0



The Time Window available in APx v 2.x through v4.0 used AP-Equiripple-shaped tapers and **Track Peak**. The start taper was fixed at 90% of the A region, and the end taper was fixed at 95% of the B region. Read a description of **Track Peak** on page 201.

Time Windows in APx500 v4.1

The Time Window available in APx 4.1 used Hann-shaped tapers and Track Peak. The start taper was fixed at 100% of the A region, and the end taper was fixed at 100% of the B region. Read a description of **Track Peak** on page 201.



More about Loudspeaker Testing

Loudspeakers can exhibit faults in several domains.

Response measurements

Loudspeaker response measurements focus on frequency, phase and distortion response at one or more level settings. Log-swept sine (chirp) measurements such as APx Acoustic Response (see page 197) are particularly effective for loudspeaker response measurements; stepped sine measurement and impulse response measurements can also be useful.

Rub and Buzz detection

Mechanical defects, such as a rubbing coil or particles in the gap, can introduce a number of different spurious signals into the acoustic output of a loudspeaker system. Rub and buzz signals are characteristically of higher frequency than the stimulus tone that excites the defect, and are of high crest factor.

In the Audio Precision implementation (see page 209), rub and buzz is detected by a process that first uses high-pass filtering, providing a residuals-only signal. The filter corner frequency slides upward during the measurement, tracking a multiple of the chirp stimulus frequency. This is followed by a crest factor measurement of the filtered residual, and a peak ratio measurement of the filtered residual vs. the unfiltered rms signal.

- Rub and buzz crest factor = residual signal peak / residual signal RMS.
- Rub and buzz peak ratio = residual signal peak / main (unfiltered) signal RMS.

A combination of high residuals crest factor (above 12 dB) and high peak ratio (above –40 dB) is a reliable indicator of a rub and buzz defect.

In APx500, rub and buzz detection is provided as an additional result to the Acoustic Response measurement. Both crest factor and peak ratio are shown on the same graph, using dual axes. The graph traces can be scaled and panned so that results indicating a defect fall on the same area of the graph, providing a helpful visual correspondence.

Because rub and buzz defects are impulsive and not periodic, averaging of results is not recommended. Averaging can make the defect signature disappear entirely.

Air Leak detection

Loudspeaker designs may specify a completely sealed enclosure, or an enclosure vented with acoustic ports. In either case, air leaks can degrade the performance, and can add spurious signals to the acoustic output of the system. A Modulated Noise ratio measurement is provided in software option SW-SPK-RD and SW-SPK-PT as an air leak detection tool.

Read about Modulated Noise and air leak detection on page 394.

Impedance/Thiele-Small

See Chapter 51.

The Impedance/Thiele-Small measurement measures the complex impedance of a loudspeaker driver, providing impedance response curves and Thiele-Small parameters.

Loudspeaker Production Test

See Chapter 58.

Loudspeaker Production Test combines two types of measurements in one test to facilitate fast loudspeaker testing in a production environment. An external power amplifier, a measurement microphone with power supply and an impedance fixture with a sense resistor are required. An acoustic isolation chamber is optional.

The two types of measurements are

- an acoustic measurement, providing frequency response, phase, distortion and rub and buzz results
  and
- an electromechanical impedance measurement, providing impedance response curves and a subset of Thiele-Small parameter results.

The underlying technology is the continuous sweep, providing both acoustic and impedance results in one fast measurement.

Bandpass Frequency Sweep
(Sequence Mode)

Introduction

For the APx515, this measurement requires a software option key. See page 189 for more information about software options.

The **Bandpass Frequency Sweep** measurement uses a sine wave stimulus signal that is moved across a range of frequencies in a specified number of points. The results are filtered using a bandpass filter that tracks the sweep frequency. Choose the filter width with the **Selectivity** control. X axis is generator frequency; Y axis is DUT output.

Bandpass Frequency Sweep measurement results available in APx500 are:

- RMS Level
- Gain
- Relative Level
- Deviation
- Peak Level
- Average Jitter Level

All of these results are available from a single measurement.

Operation

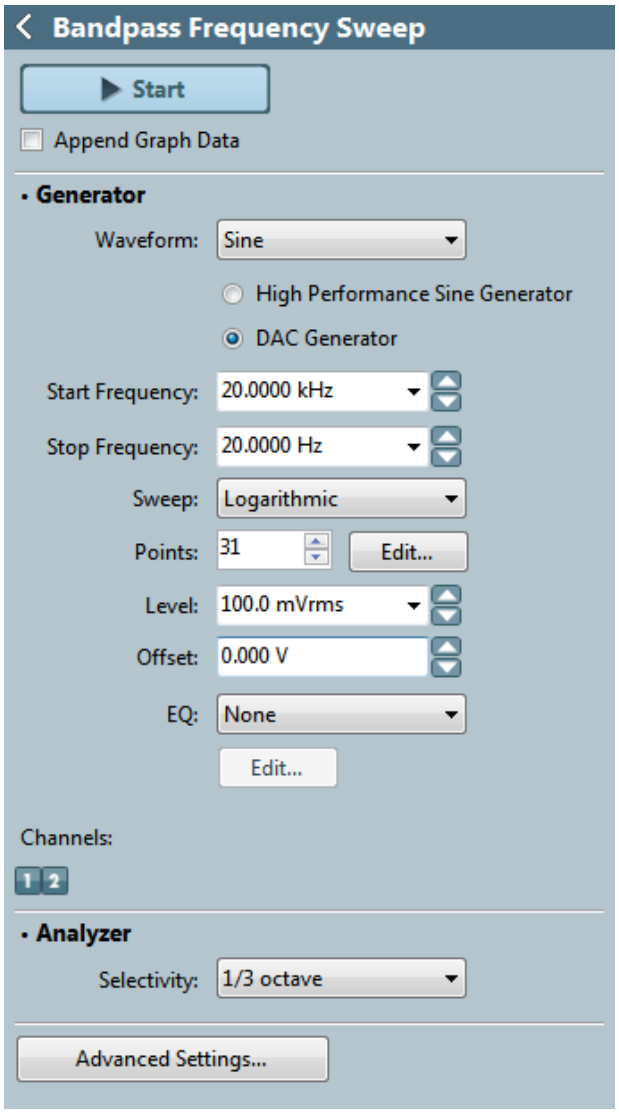
If you have not yet set up your test, first go to Signal Path Setup to select and configure inputs and outputs, references and other settings. See Chapter 7 for information about Signal Path Setup.

Start

For a default sweep of 20 kHz to 20 Hz in 31 logarithmic steps, simply click **Start**. If you want to specify other parameters for your sweep, review **Generator** and **Analyzer** settings, discussed below. Click **Start** when you have approved the settings.

Append Graph Data

Normally, the current graph data for all results is deleted each time you start a new measurement. If the **Append** box is checked, the current data is kept in memory as a **Data Set** (see page 189), and the new measurement data is appended as a new data set. You can append many **Data Sets**.



Generator

The typical configuration for this measurement is the closed-loop configuration, using the APx generator as a stimulus. Alternatively, you can make open-loop

measurements, using either the External Source or File Input configuration.

See Chapter 6 for detailed information about the using the APx Generator and making Waveform, Level and Frequency settings, or for information about using External Source.

Waveform

Select **Sine** for a closed loop measurement using the internal generator. If you select **Browse for file** to choose a generator waveform file, see "Using Sweep Tables with Stepped Frequency Sweep Measurements" on page 507.

High Performance Sine Generator / DAC Generator

For an APx555, you have the option of using the High Performance Sine Generator (very low distortion sine waves and wide bandwidth) or the DAC generator, which provides special sine waveforms. See High Performance Sine Generator starting on page 51 for a comparison of features.

Start Frequency

Set the sweep start frequency here.

Stop Frequency

Set the sweep stop frequency here.

Sweep

Choose

- **Logarithmic** point spacing
- **Linear** point spacing
- **Custom**
  Edit sweep spacing and number of **Points** to create a **Custom** sweep.

Points > Edit

Open the **Sweep Points** dialog to edit import or export the **Sweep Points** table.

Step Size

For linear sweeps, enter the **Step Size** here.

Level

Set the signal level here. This level may be changed if generator equalization is used, and can be adjusted on a per-channel basis in **Advanced Settings** (see page 508).

EQ

You can optionally modify the sweep with by applying a equalization curve to the generator. See Generator Equalization on page 192 for a general discussion of this feature. Choose

- **None**, which applies no equalization to the generator.
- **Relative**, which applies the gain or loss specified in the EQ table to the current generator level for each channel.
- **Absolute**, which sets the generator level for all channels to the levels specified in the EQ table. Note: when **Absolute** is selected, current channel level settings are lost.

Click the **Edit** button to create or change the EQ curve. See the **Edit EQ Table** dialog discussion on page 192.

Channels

Toggle specific output channels on or off by clicking on the Channel number button.

Analyzer

This measurement can be configured to measure jitter in Signal Path Setup > Input > Measure. Read detailed information about jitter generation and measurement beginning on page 69.

Filters

This measurement uses a tunable bandpass filter. No local low pass, high pass or weighting filters are available for this measurement.
However, low pass, high pass and weighting filter settings made in Signal Path Setup > Input/Output will affect this measurement. Read more about APx Filters in Chapter 97.

Selectivity

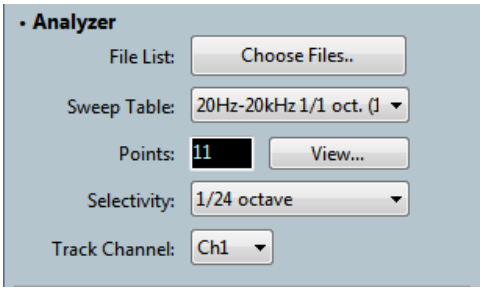
In the **Bandpass Frequency Sweep** measurement you can choose from a number of bandpass filter widths using the **Selectivity** control. The list is ordered from narrowest at the top, to widest at the bottom. The filter is tuned to the APx generator frequency, and follows the sweep.

- **Window width**
  This is the window width of the underlying FFT, typically only a few hertz wide. This selection has very steep skirts and a flat top.
- **1/24 octave**
  This selection has –3 dB points that are 1/24 octave apart.
- **1/12 octave**
  This selection has -–3 dB points that are 1/12 octave apart.
- **1/9 octave**
  This selection has –3 dB points that are 1/9 octave apart.
- **1/6 octave**
  This selection has –3 dB points that are 1/6 octave apart.

- 1/3 octave
  This selection has –3 dB points that are 1/3 octave apart.
- 1 octave
  This selection has –3 dB points that are 1 octave apart.

All of the "x octave" filters are equivalent to second-order analog bandpass filters, with 6 dB/octave skirts. Q is chosen to achieve the specified bandwidth at the –3 dB points.

File List



This control is available if the **Input Configuration** in **Signal Path Setup** is set to one of the two **File** choices. Read more about analyzing audio files in Chapter 22.

Sweep Table

The Sweep Table menu, and the Points, View/Edit and Track Channel controls discussed below, are not shown for a closed loop sweep that uses the internal generator. In that case, the stimulus sweep definition is directly available to the analyzer and these controls are unnecessary.

When the measurement is configured so that the stimulus sweep definition is not available to the analyzer (in **External Source, Generator Waveform** or **File Input** configurations), you must select a **Sweep Table** (see page 507) that matches the definition of the sweep you are using.

The **Sweep Table** menu lists the embedded sweep tables and any new tables that have been added to the project. Select a table from the **Sweep Table** menu, or choose **Create New** or **Browse** for a sweep table file.

Points

This field displays the number of points in the **Sweep Points** table provided to the analyzer.

View / Edit

For embedded sweep tables, click **View** (see page 507) to view or export the table, or to create a WAV file from the table. For other sweep tables, click **Edit** to edit, view or export the table, or to create a WAV file from the table (see page 508).

Selectivity

See the **Selectivity** discussion on the previous page.

Track Channel

When using a sweep table (in **External Source, Generator Waveform** or **File Input** configurations), the analyzer must determine the relation of the incoming signal to the sweep table. When **Auto** is selected, the analyzer continually polls the all channel inputs throughout the sweep, switching between channels as necessary to track the input with the highest signal level.

Alternatively, you can select a specific input channel for the analyzer to use for sweep tracking.

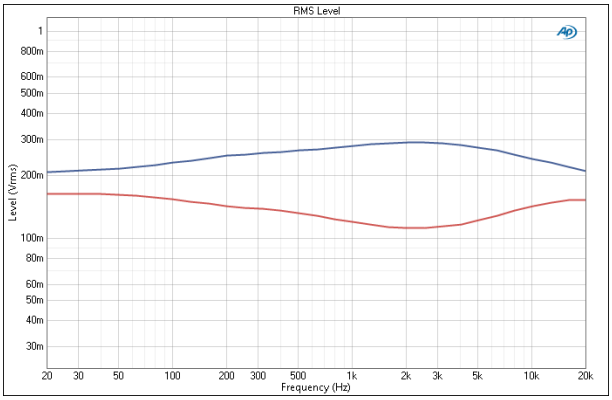
Advanced Settings

If your test requires special adjustments or settings, click **Advanced Settings**. See page 508. See Chapter 104 for more information about units of measurement.

Data Set

If you have appended or imported graph data, the **Data Set** panel in the area beneath the graph display will be populated. Choose the data set(s) to be displayed on the graph. See page 189 for more information about **Data Sets**.

RMS Level



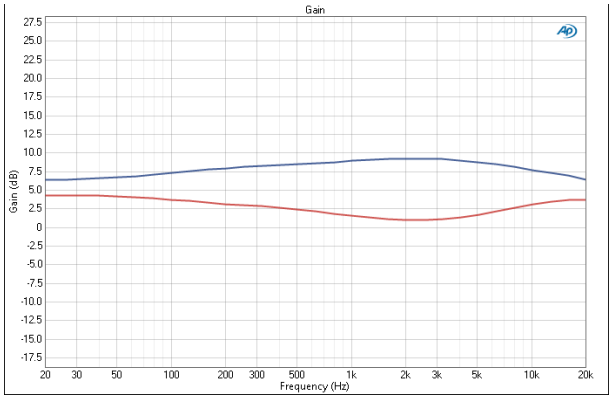
Bandpass Frequency Sweep: RMS Level results uses a sine wave stimulus signal that is moved across a range of frequencies in a specified number of points. This is the conventional frequency response sweep, bandpass filtered.

Units

Units available for Bandpass Frequency Sweep Level results are:

X-axis	*Y-axis (analog)*	*Y-axis (digital)*	*Jitter*
• Hz	• Vrms	• dBFS	• UI
• dHz	• dBV	• FS	• dBUI
• F/R	• dBu	• %FS	• s
• %Hz	• dBrA	• dBrA	
	• dBrB	• dBrB	
	• dBSPL1	• dBSPL1	
	• dBSPL2	• dBSPL2	
	• dBm		
	• W (watts)		

Gain



Gain is a Bandpass Frequency Sweep result that provides a graphical display of the voltage gain in each channel. In this result the DUT gain is plotted against frequency. The **Bandpass Frequency Sweep Gain** measurement is not available in **External Source** configuration.

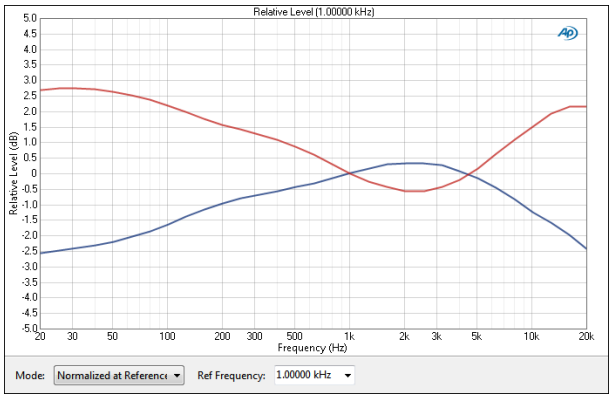
> *Gain results are not available when External Source is selected as the output configuration, when File is selected as the input configuration or when a generator waveform file is selected as the test signal.*

Units

Units available for Bandpass Frequency Sweep: Gain are:

X-axis	*Y-axis same-domain*	*Y-axis cross-domain*
• Hz	• x/y	• FS/Vrms
• dHz	• %	• dB(FS/Vrms)
• F/R	• ppm	—or—
• %Hz	• dB	• Vrms/FS
		• dB(Vrms/FS)

Relative Level



The **Bandpass Frequency Sweep: Relative Level** result uses a sine wave stimulus signal that is moved across a range of frequencies in a specified number of points, providing a conventional frequency response sweep. Additionally, in this result the response curve is plotted in relation to the measured level at a selected frequency. This enables you to specify the frequency where the relative response will be unity and to view the response in relation to that frequency. A midband frequency such as 1 kHz is often set as the reference frequency, and is the default here.

Mode

Normalized at Reference

Enter a reference frequency for normalization in the **Ref Frequency** field. The measured level at the frequency you choose is set as 0 dB on the graph, and the DUT response is plotted in relation to this reference. You can change the **Ref Frequency** at any time (except after appending) and the graph will immediately be redrawn to reflect the new setting.

Auto-Centered in Limits

This control is only usable if upper and lower limits are set on this graph. In some applications, the general conformance of a trace to the shape of a set of limits is of importance, with absolute level less important or irrelevant. **Auto-Centered in Limits** centers the data trace relative to the upper and lower limits. To exclude possible exceptional data points at frequency extremes, the frequency range that is considered for the centering operation is constrained by the values in the **Min** and **Max** fields.

Ref Frequency

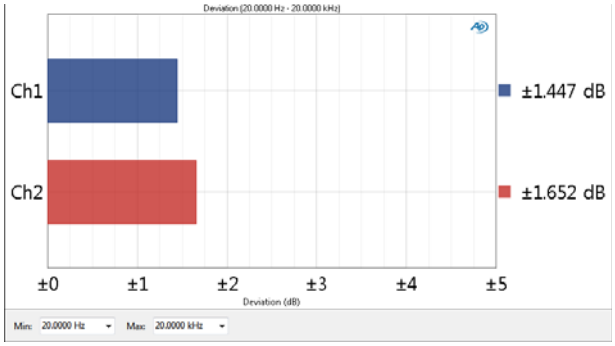
Choose a Reference Frequency here.

Units

Units available for Bandpass Frequency Sweep: Relative Level are:

X-axis	*Y-axis*
• Hz	• x/y
• dHz	• %
• F/R	• ppm
• %Hz	• dB

Deviation



The **Bandpass Frequency Sweep: Deviation** result uses a sine wave stimulus signal that is moved across a range of frequencies in a specified number of points. This result is a single value measurement computed from the stepped sweep that shows the frequency deviation (the total range of frequency variation) of each channel as a meter bar. You can specify a minimum and maximum frequency to define the range of the deviation measurement.

Min

Set the minimum frequency of the range of interest here.

Max

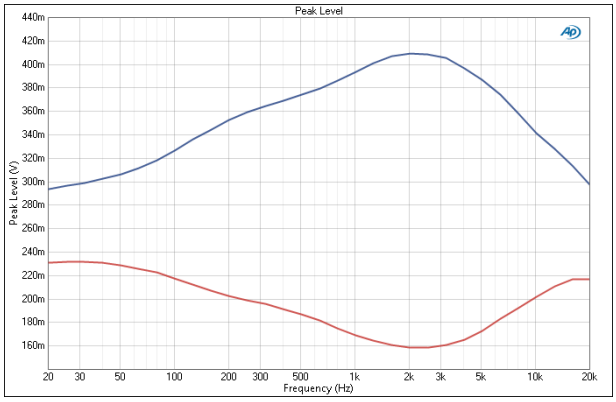
Set the maximum frequency of the range of interest here.

Units

Units available for Bandpass Frequency Sweep: Deviation are

- dB

Peak Level



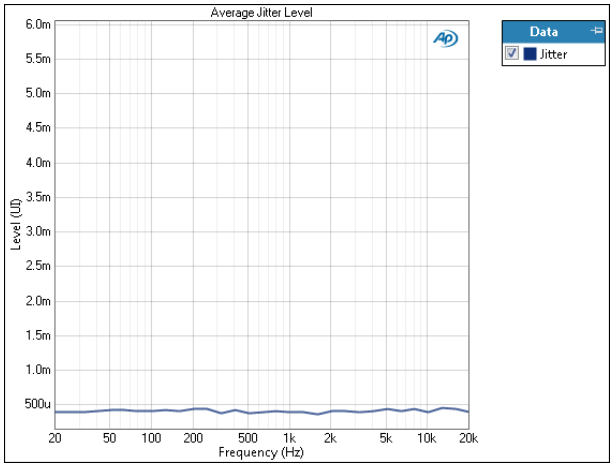
Bandpass Frequency Sweep: Peak Level results uses a sine wave stimulus signal that is moved across a range of frequencies in a specified number of points. This is a frequency response sweep, bandpass filtered, with peak-scaled results.

Units

Units available for Bandpass Frequency Sweep Peak Level results are:

X-axis	*Y-axis (analog)*	*Y-axis (digital)*	*Jitter*
• Hz	• V	• D	• UI
• dHz		• hex	• dBUI
• F/R			• s
• %Hz			

Average Jitter Level



Bandpass Frequency Sweep: Average Jitter Level results uses a sine wave stimulus signal that is moved across a range of frequencies in a specified number of points. The result is the bandpass filtered DUT jitter output, average scaled.

Units

Units available for Bandpass Frequency Sweep Average Jitter Level results are:

X-axis	*Jitter*
• Hz	• UI
• dHz	• dBUI
• F/R	• s
• %Hz	

Bandpass Level
(Sequence Mode)

Introduction

For the APx515, this measurement requires a software option key. See page 189 for more information about software options.

The Bandpass Level measurement provides a bandpass-filtered single-value result displaying the output level from each DUT channel, as measured at each analyzer input.

A bandpass filter is designed to pass a band of frequencies narrower than the system bandwidth. This range is called the *passband*. The *selectivity* of the filter determines the width of the passband; a highly selective filter will have a narrow passband.

Bandpass filters are often used for noise measurements, restricting the measurement to the frequency range of interest.

Bandpass Level results available in APx500 are:

- RMS Level
- Peak Level
- Average Jitter Level

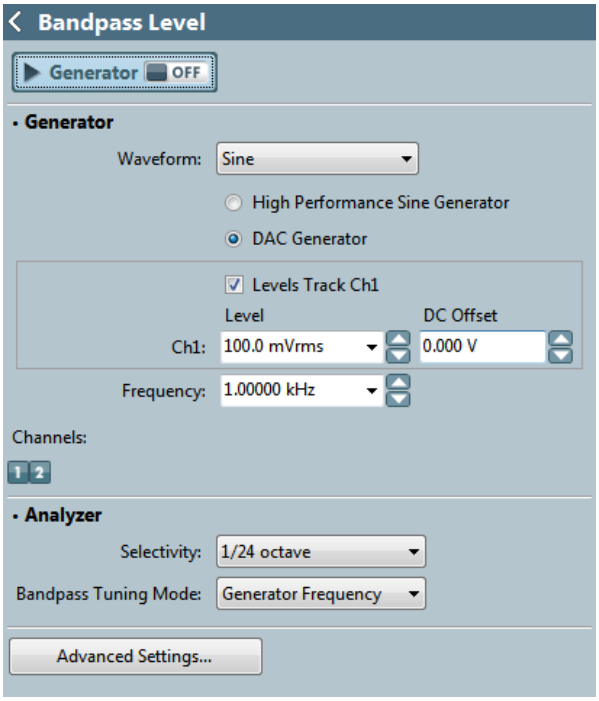
Average Jitter Level results are only available when Jitter is selected in Signal Path Setup > Input/Output > Measure.

Operation

If you have not yet set up your test, first go to Signal Path Setup to select and configure inputs and outputs, references and other settings. See Chapter 7 for information about Signal Path Setup.

Generator

The typical configuration for this measurement is the closed-loop configuration, using the APx generator as a stimulus. Alternatively, you can make open-loop measurements, using either the External Source or File Input configuration.



See Chapter 6 for detailed information about the using the APx Generator and making Waveform, Level and Frequency settings, or for information about using External Source.

High Performance Sine Generator / DAC Generator

For an APx555, you have the option of using the High Performance Sine Generator (very low distortion sine waves and wide bandwidth) or the DAC generator, which provides special sine waveforms. See High Performance Sine Generator starting on page 51 for a comparison of features.

Analyzer

This measurement can be configured to measure jitter in Signal Path Setup > Input > Measure. Read

detailed information about jitter generation and measurement beginning on page 69.

Selectivity

In the Bandpass Level measurement you can choose from a number of bandpass filter widths using the **Selectivity** control. The list is ordered from narrowest at the top, to widest at the bottom.

- **Window width**
  This is the window width of the underlying FFT, typically only a few hertz wide. This selection has very steep skirts and a flat top.
- **1/24 octave**
  This selection has -3 dB points that are 1/24 octave apart.
- **1/12 octave**
  This selection has -3 dB points that are 1/12 octave apart.
- **1/9 octave**
  This selection has -3 dB points that are 1/9 octave apart.
- **1/6 octave**
  This selection has -3 dB points that are 1/6 octave apart.
- **1/3 octave**
  This selection has -3 dB points that are 1/3 octave apart.
- **1 octave**
  This selection has -3 dB points that are 1 octave apart.
- **Rectangular Band**
  This selection has very steep skirts and a flat top. The user defines the lower and upper frequency cutoff points. See **Min Freq: / Max Freq:** below.

All of the “x octave” filters are equivalent to second-order analog bandpass filters, with 6 dB/octave skirts. Q is chosen to achieve the specified bandwidth at the –3 dB points.

Bandpass Tuning Mode

The “Window width” and “x octave” filters are centered at some user-defined frequency for each measurement. This can be

- **Generator Frequency**
  The current APx audio generator frequency.
  When the generator channels are outputting different frequencies (Split Frequency generation), the bandpass filter center is set to the Frequency A. This mode is not available when using a generator waveform file.
- **Jitter Generator Frequency**
  The current APx jitter generator frequency, when jitter generation is available and enabled. Read detailed information about jitter generation and measurement beginning on page 69.
- **Measured Frequency**
  The current measured frequency.
  When the analyzer channels are receiving different frequencies, the bandpass filter for each channel is centered on the frequency in that channel.
- **Fixed Frequency**
  A fixed frequency selected by the user.
  When **Fixed Frequency** is selected, a **Filter Freq:** entry field becomes available beneath the **Bandpass Tuning Mode** control.

Min Freq: / Max Freq:

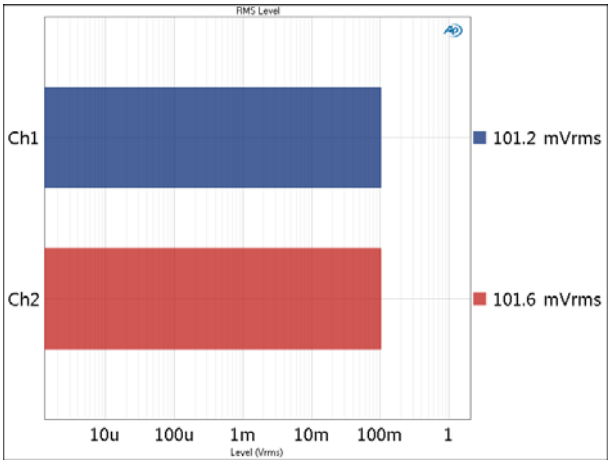
When **Rectangular Band** is the **Selectivity** choice, these fields become available for lower and upper frequency cutoff point entry.

Advanced Settings

If your test requires individual generator settings for each channel or special analyzer ranging or settling, click **Advanced Settings**. See Advanced Settings for single value measurements on page 351.

See Chapter 104 for more information about units of measurement.

RMS Level



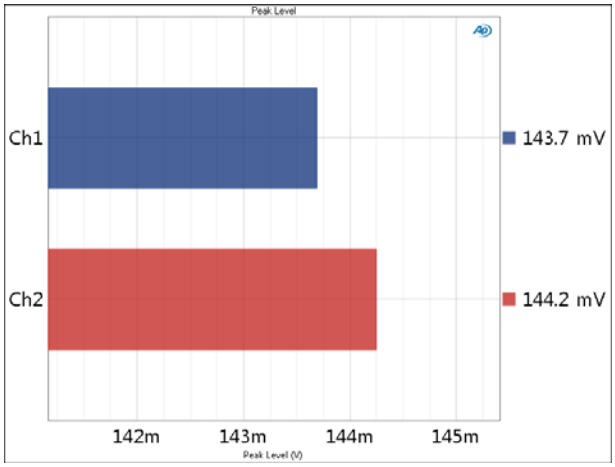
RMS Level provides a single-value meter result, displaying of the rms output level from each DUT channel, as measured at each analyzer input.

Units

Units available for RMS Level are

Analog Signals	*Digital Signals*	*Jitter signals*
• Vrms	• dBFS	• UI
• dBV	• FS	• dBUI
• dBu	• %FS	• s
• dBrA	• dBrA	
• dBrB	• dBrB	
• dBSPL1	• dBSPL1	
• dBSPL2	• dBSPL2	
• dBm		
• W (watts)		

Peak Level



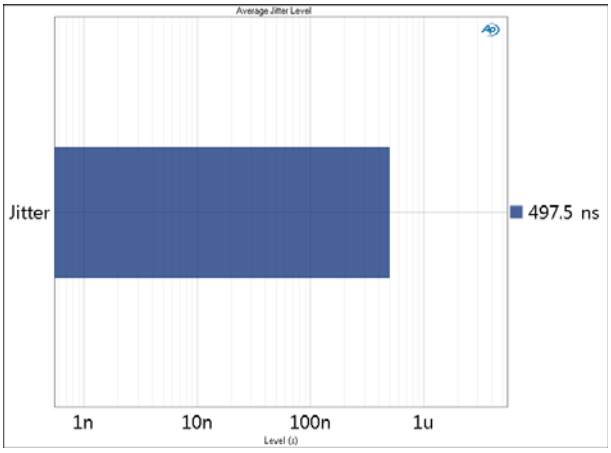
Peak Level provides a single-value meter result, displaying of the peak output level from each DUT channel, as measured at each analyzer input.

Units

Units available for Peak Level are

Analog Signals	*Digital Signals*	*Jitter signals*
• Vrms	• dBFS	• UI
• dBV	• FS	• dBUI
• dBu	• %FS	• s
• dBrA	• dBrA	
• dBrB	• dBrB	
• dBSPL1	• dBSPL1	
• dBSPL2	• dBSPL2	
• dBm		
• W (watts)		

Average Jitter Level



Average Jitter Level provides a single-value meter result, displaying the average jitter level from the digital receiver.

Units

Units available for Average Jitter Level are

Analog Signals	*Digital Signals*	*Jitter signals*
• Vrms	• dBFS	• UI
• dBV	• FS	• dBUI
• dBu	• %FS	• s
• dBrA	• dBrA	
• dBrB	• dBrB	
• dBSPL1	• dBSPL1	
• dBSPL2	• dBSPL2	
• dBm		
• W (watts)		

Bandpass Level Sweep
(Sequence Mode)

Introduction

For the APx515, this measurement requires a software option key. See page 189 for more information about software options.

The **Bandpass Level Sweep** measurement provides a sine wave stimulus signal that is moved across a range of levels in a series of points. The DUT output is acquired by the analyzer, and filtered using a bandpass filter. **Bandpass Level Sweep** measurements are not supported in the **External Source** configuration.

Bandpass Level Sweep results available in APx500 are:

- RMS Level
- Linearity
- Average Jitter Level
- Gain
- Peak Level

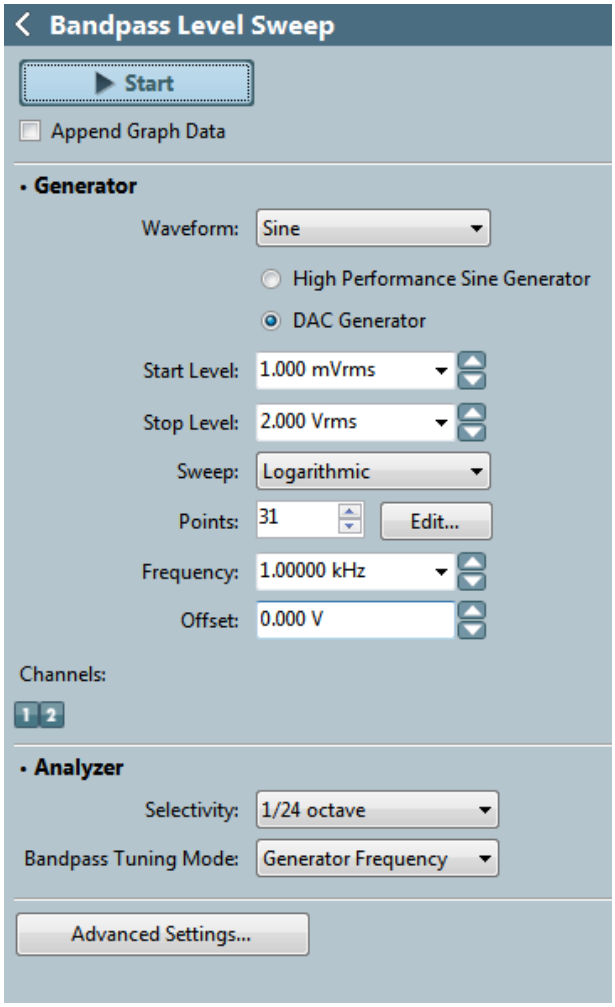
All of these results are available from a single measurement.

Operation

If you have not yet set up your test, first go to Signal Path Setup to select and configure inputs and outputs, references and other settings. See Chapter 7 for information about Signal Path Setup.

The **Bandpass Level Sweep** measurement uses a sine wave at the frequency and range of levels set in the Signal Generation panel as the test signal. This sine wave will be swept from the **Start Level** to the **Stop Level** in the set number of **Points**. The DUT output is acquired by the analyzer, and filtered using a bandpass filter.

Note: if Output Configuration is set to Bluetooth (A2DP Source profile), the Bluetooth hardware imposes a –54 dBFS cutoff. Signals below –54 dBFS are not transmitted. Please consider this when defining the range of levels for your sweep.



Start

For a default sweep of 1 mVrms to 2 Vrms in 31 logarithmic steps, simply click **Start**. If you want to specify other parameters for your sweep, review **Generator** and **Analyzer** settings, discussed below. Click **Start** when you have approved the settings.

Append Graph Data

Normally, the current graph data for all results is deleted each time you start a new measurement. If the **Append** box is checked, the current data is kept in memory as a **Data Set** (see page 189), and the new measurement data is appended as a new data set. You can append many **Data Sets**.

Generator

Waveform

Choose a **Special Sine** option (see page 182) to set a DC offset or to split frequency or phase across channels.

Start Level, Stop Level and Sweep Points

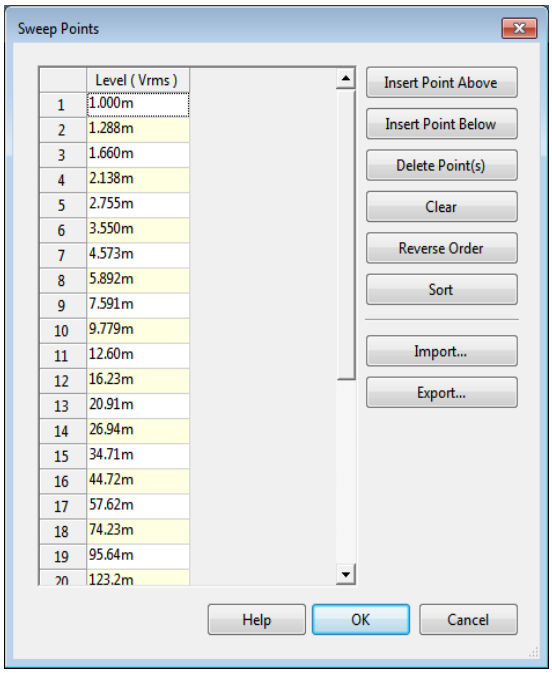
The sweep moves between two levels, set in the **Start Level** and **Stop Level** fields. The default levels are 1.000 mVrms and 2.000 Vrms.

The sweep moves in a specified number of step points, set in the **Points** field. The minimum is 2 points; maximum is 65,535. The default setting is 31.

The sweep point spacing is set by selecting one of the following choices in the Sweep field:

- **Logarithmic** (the default); use the **Points** field to set the number of logarithmically spaced points;
- **Linear**, which provides two methods of adjusting spacing: the **Points** field or the **Step Size** field; or
- **Custom**. Click **Edit** to open the **Sweep Points** dialog, where you can set points arbitrarily, or load or save sweep table files.

Viewing or Editing the Sweep Points table



You can view or edit the current sweep points at any time.

Click **Edit** to open the **Sweep Points** table. The table shows each sweep point and its corresponding level. You can edit this table to add or delete points, or to change the level of a point. Points can be sorted or reversed in order using the controls on the right.

A **Sweep Points** table can be saved as a *.csv file or as a Microsoft Excel *.xls file. A compatible *.csv or *.xls file can be opened and used as a **Sweep Points** table.

Analyzer

This measurement uses a tunable bandpass filter. No low pass, high pass or weighting filters are available for this measurement.
However, low pass, high pass and weighting filter settings made in Signal Path Setup > Input/Output will affect this measurement. Read more about APx Filters in Chapter 97.

Selectivity

In the **Bandpass Level Sweep** measurement you can choose from a number of bandpass filter widths using the **Selectivity** control. The list is ordered from narrowest at the top, to widest at the bottom. The **Window Width** and **x Octave** filters are tuned according to the **Bandpass Tuning Mode** setting.

- **Window width**
  This is the window width of the underlying FFT, typ-

ically only a few hertz wide. This selection has very steep skirts and a flat top.

- **1/24 octave**
  This selection has –3 dB points that are 1/24 octave apart.
- **1/12 octave**
  This selection has –3 dB points that are 1/12 octave apart.
- **1/9 octave**
  This selection has –3 dB points that are 1/9 octave apart.
- **1/6 octave**
  This selection has –3 dB points that are 1/6 octave apart.
- **1/3 octave**
  This selection has –3 dB points that are 1/3 octave apart.
- **1 octave**
  This selection has –3 dB points that are 1 octave apart.
- **Rectangular Band**
  This selection has very steep skirts and a flat top. The user defines the lower and upper frequency cutoff points. See **Min Freq: / Max Freq:** below.

All of the "x octave" filters are equivalent to second-order analog bandpass filters, with 6 dB/octave skirts. Q is chosen to achieve the specified bandwidth at the –3 dB points.

Min Freq: / Max Freq:

When **Rectangular Band** is the **Selectivity** choice, these fields become available for lower and upper frequency cutoff point entry.

Bandpass Tuning Mode

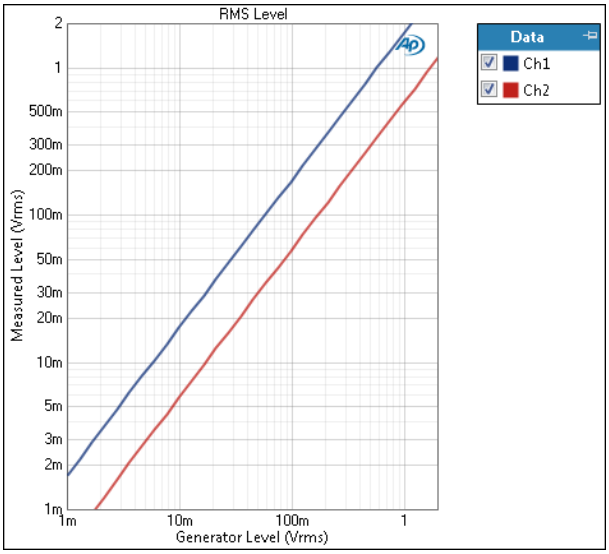
The "Window width" and "x octave" filters are centered at some user-defined frequency for each step of the measurement. This can be

- **Generator Frequency**
  The APx generator frequency for the current step. When the generator channels are outputting different frequencies (Split Frequency generation), the bandpass filter center is set to Frequency A.
- **Jitter Generator Frequency**
  The current APx jitter generator frequency, when jitter generation is available and enabled. Read detailed information about jitter generation and measurement beginning on page 69.
- **Measured Frequency**
  The measured frequency for the current step. When the analyzer channels are receiving different frequencies, the bandpass filter for each channel is centered on the frequency in that channel.
- **Fixed Frequency**
  A fixed frequency selected by the user. When **Fixed Frequency** is selected, a **Filter Freq:** entry field becomes available beneath the **Tune Mode** control.

Advanced Settings

If your test requires individual generator settings for each channel or special analyzer ranging or settling, click **Advanced Settings**. See Advanced Settings for Stepped Level Sweeps on page 519. See Chapter 104 for more information about units of measurement.

RMS Level



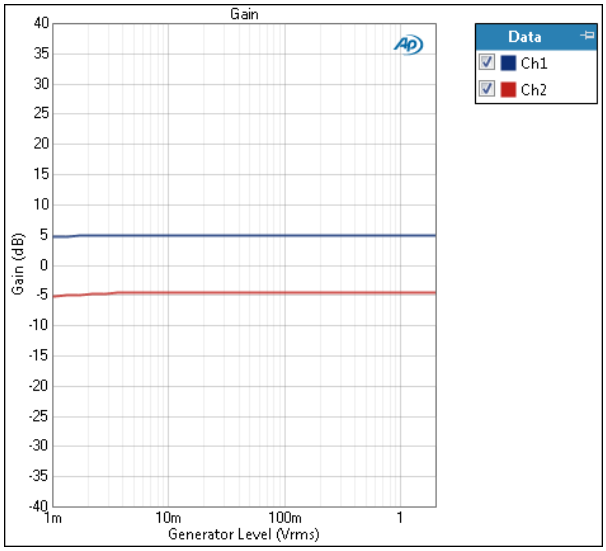
The **Bandpass Level Sweep: RMS Level** result uses a sine wave stimulus signal that is moved across a range of levels in a series of points. X axis is generator level; Y axis is the bandpass filtered DUT output level.

Units

Units available for Level are:

X-axis (analog)	*X-axis (digital)*	*Y-axis (analog)*	*Y-axis (digital)*
• Vrms	• dBFS	• Vrms	• dBFS
• Vp	• FS	• dBV	• FS
• Vpp	• %FS	• dBu	• %FS
• dBV	• dBrG	• dBrA	• dBrA
• dBu		• dBrB	• dBrB
• dBrG		• dBSPL1	• dBSPL1
• dBm		• dBSPL2	• dBSPL2
• W (watts)		• dBm	
		• W (watts)	
		Y-axis (jitter)	
		• UI	
		• dBUI	
		• s	

Gain



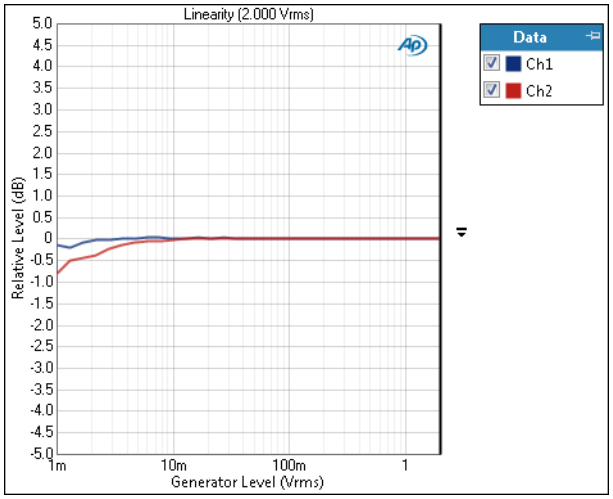
The **Bandpass Level Sweep: Gain** result uses a sine wave stimulus signal that is moved across a range of levels in a series of points. X axis is generator level; Y axis is gain in the DUT. Gain is a ratio, calculated by dividing the bandpass filtered DUT output level by the generator level.

Units

Y-axis units available for Gain are:

X-axis (analog)	*X-axis (digital)*	*Y-axis same-domain*	*Y-axis cross-domain*
• Vrms	• dBFS	• x/y	• FS/Vrms
• Vp	• FS	• %	• dB(FS/Vrms)
• Vpp	• %FS	• ppm	—or—
• dBV	• dBrG	• dB	• Vrms/FS
• dBu			• dB(Vrms/FS)
• dBrG			
• dBm			
• W (watts)			

Linearity



The **Bandpass Level Sweep: Linearity** result uses a sine wave stimulus signal that is moved across a range of levels in a series of points. X axis is generator level; Y axis is linearity in the DUT. Linearity is a ratio, calculated by dividing the bandpass filtered DUT output level by the generator level, normalized for a chosen relative level.

Mode

Normalized at Reference

Enter a reference level for normalization in the **Relative Level** field. To calculate linearity, each channel is normalized so that a linearity of unity (0 dB) is at the same measured DUT output level. Set this reference level in the **Relative Level** field. Default **Relative Level** is 1 Vrms (analog), 1 FS (digital). You can change the **Relative Level** at any time (except after appending) and the graph will immediately be redrawn to reflect the new setting.

Auto-Centered in Limits

This control is only usable if upper and lower limits are set on this graph. In some applications, the general conformance of a trace to the shape of a set of limits

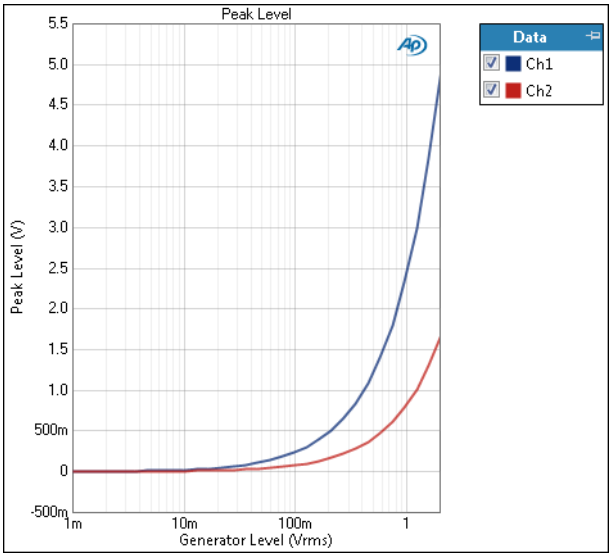
is of importance, with absolute level less important or irrelevant. **Auto-Centered in Limits** centers the data trace relative to the upper and lower limits. To exclude possible exceptional data points at level extremes, the amplitude range that is considered for the centering operation is constrained by the values in the **Min** and **Max** fields.

Units

Units available for Linearity are:

X-axis (analog)	*X-axis (digital)*	*Y-axis*
• Vrms	• dBFS	• x/y
• Vp	• FS	• %
• Vpp	• %FS	• ppm
• dBV	• dBrG	• dB
• dBu		
• dBrG		
• dBm		
• W (watts)		

Peak Level



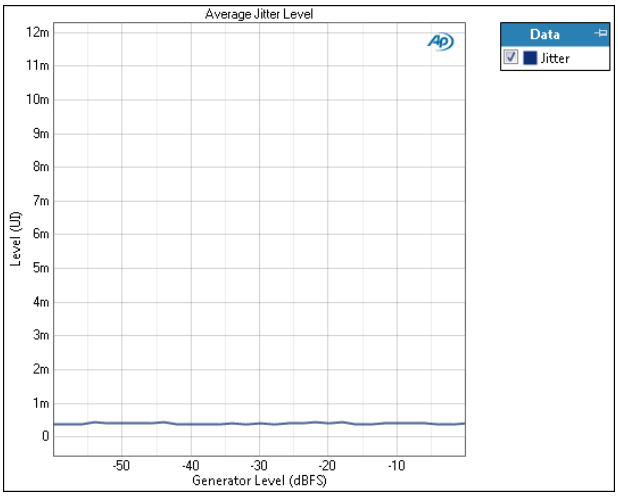
The **Bandpass Level Sweep: Peak Level** result uses a sine wave stimulus signal that is moved across a range of levels in a series of points. X axis is generator level; Y axis is the bandpass filtered DUT output level, scaled in peak units.

Units

Units available for Peak Level are:

X-axis (analog)	*X-axis (digital)*	*Y-axis (analog)*	*Y-axis (digital)*
• Vrms	• dBFS	• V	• D
• Vp	• FS		
• Vpp	• %FS		
• dBV	• dBrG		
• dBu		*Y-axis (jitter)*	
• dBrG		• UI	
• dBm		• dBUI	
• W (watts)		• s	

Average Jitter Level



The **Bandpass Level Sweep: Average Jitter Level** result uses a sine wave stimulus signal that is moved across a range of levels in a series of points. X axis is generator level; Y axis is the bandpass filtered DUT output jitter level, scaled in average units.

Units

Units available for Average Jitter Level are:

X-axis (analog)	*X-axis (digital)*	*Y-axis (jitter)*
• Vrms	• dBFS	• UI
• Vp	• FS	• dBUI
• Vpp	• %FS	• s
• dBV	• dBrG	
• dBu		
• dBrG		
• dBm		
• W (watts)		

CMRR
(Sequence Mode)

CMRR is an analog measurement that uses hardware features found only in the 2-channel analog output circuitry fitted in an APx515, 525, 526, 555 or 582. CMRR measurements are not available in External Source configuration.

CMRR is an abbreviation of Common Mode Rejection Ratio. For a differential input such as a balanced audio input, “common mode” refers to a signal common to both the + and – inputs. Common mode signals are typically spurious signals picked up from the environment common to both leads, or sometimes a dc offset common to both leads. Common mode signals should be rejected to a high degree by differential amplifier inputs.

The CMRR measurement provides a single-value result, displaying the CMRR in each DUT channel.

Also see Chapter 29 for information about CMRR-IEC measurements.

Operation

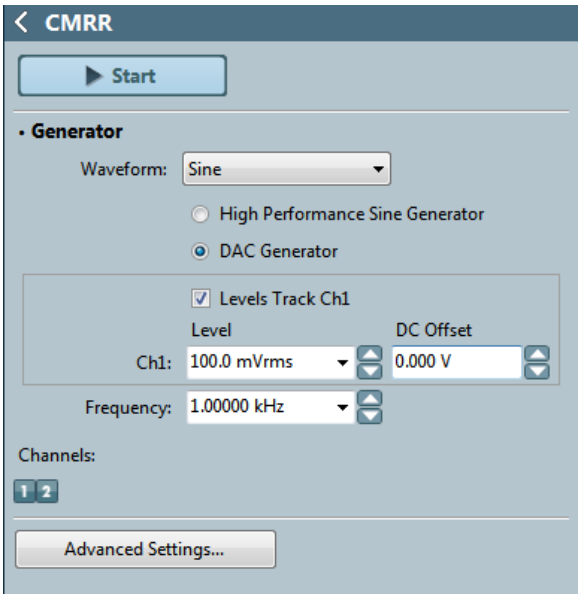
If you have not yet set up your test, first go to Signal Path Setup to select and configure inputs and outputs, references and other settings. See Chapter 7 for information about Signal Path Setup.

Generator

This measurement must be made in the closed-loop configuration with balanced analog outputs, using the APx generator as a stimulus.

To measure CMRR, Click **Start**. The generator will output a sine wave to the DUT on the selected generator channels at the level and frequency set in the Signal Generation panel.

For the first half of the measurement, the generator is connected to the DUT in normal (differential) mode and a measurement of the normal mode rms level is made. The generator outputs are automatically



switched to common mode using a pair of matched source resistors, one in each leg of the output circuit. A second measurement is made. The ratio of the two results is displayed as the Common Mode Rejection Ratio (CMRR).

See Chapter 6 for detailed information about the using the APx Generator and making Waveform, Level and Frequency settings.

High Performance Sine Generator / DAC Generator

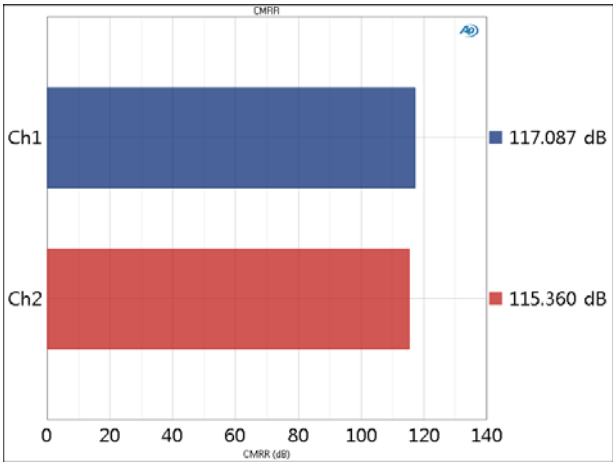
For an APx555, you have the option of using the High Performance Sine Generator (very low distortion sine waves and wide bandwidth) or the DAC generator, which provides special sine waveforms. See High Performance Sine Generator starting on page 51 for a comparison of features.

Advanced Settings

If your test requires individual generator settings for each channel or special analyzer ranging or settling,

click **Advanced Settings**. See Advanced Settings for single value measurements on page 351.

CMRR



Units

Units available for CMRR are

- x/y
- dB

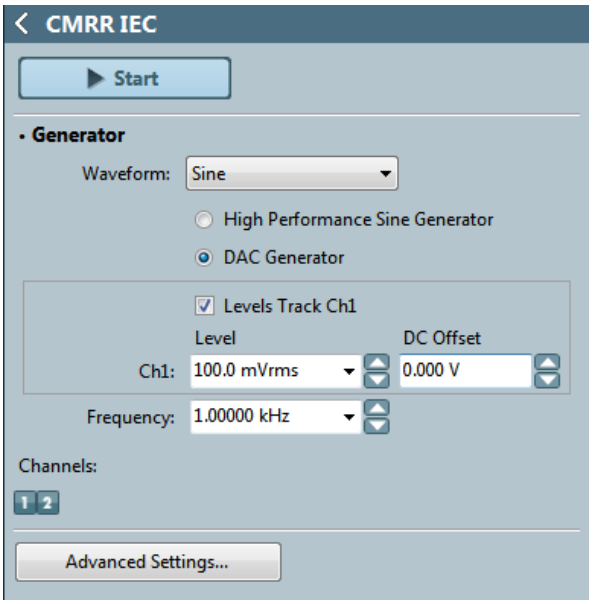
See Chapter 104 for more information about units of measurement.

CMRR IEC
(Sequence Mode)

CMRR-IEC is an analog measurement that uses hardware features found only in the 2-channel analog output module fitted in an APx525, 526, 555 or 582. CMRR measurements are not available in External Source configuration.

CMRR IEC is a technique for measuring CMRR described in the International Electrotechnical Commission document IEC60268-3 that differs from the conventional technique used in the basic CMRR measurement discussed in Chapter 28. Use this method to produce CMRR results that satisfy IEC60268-3.

The CMRR IEC measurement provides single-value results showing the CMRR in each DUT channel by the IEC technique, which a unbalances the generator output circuit by first inserting a resistance in one leg (the Pin 2 measurement), and then the other leg (the Pin 3 measurement). The final CMRR IEC result is the worst-case of the pin 2 or pin 3 measurements.



Operation

If you have not yet set up your test, first go to Signal Path Setup to select and configure inputs and outputs, references and other settings. See Chapter 7 for information about Signal Path Setup.

Generator

This measurement must be made in the closed-loop configuration with balanced analog outputs, using the APx generator as a stimulus.

To measure CMRR IEC, Click **Start**. The generator will output a sine wave to the DUT on the selected generator channels at the level and frequency set in the Signal Generation panel.

For the first one-third of the measurement, the generator is connected to the DUT in normal (differential) mode and a measurement of the normal mode rms level is made. The generator outputs are automatically switched to use an IEC-defined common mode circuit, where one leg of the output circuit is connected directly to the generator output and the other leg is connected through an additional 10 ohm source resistor. A second measurement is made. Then the resistor and the direct connection are exchanged, and a third measurement is made. The worst ratio is reported as the CMRR IEC measurement result.

See Chapter 6 for detailed information about the using the APx Generator and making Waveform, Level and Frequency settings.

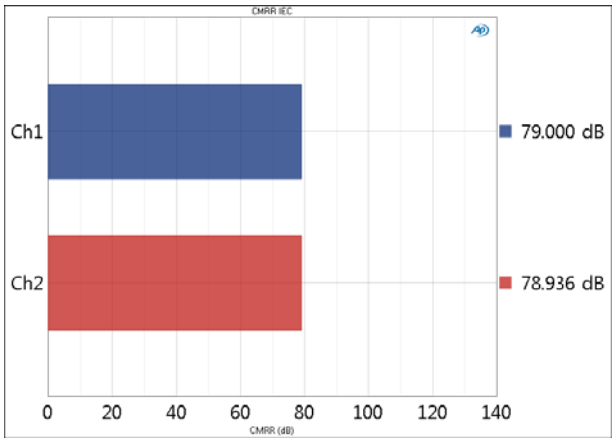
High Performance Sine Generator / DAC Generator

For an APx555, you have the option of using the High Performance Sine Generator (very low distortion sine waves and wide bandwidth) or the DAC generator, which provides special sine waveforms. See High Performance Sine Generator starting on page 51 for a comparison of features.

Advanced Settings

If your test requires individual generator settings for each channel or special analyzer ranging or settling, click **Advanced Settings**. See Advanced Settings for single value measurements on page 351.

CMRR IEC



The CMRR Pin 2 and CMRR Pin 3 results are intermediate results, and are provided for diagnostic purposes. Report the data displayed in the CMRR IEC result.

Units

Units available for CMRR IEC are

- x/y
- dB

See Chapter 104 for more information about units of measurement.

Compare Encoded Bitstream to Reference
(Sequence Mode)

This measurement is not available to the APx515.

This measurement allows you to compare the data in a coded audio bitstream to a reference file. It does not support comparison of linear PCM (IEC60958) bitstreams.

A typical use case is a coded audio (e.g. DTS, Dolby) compliance test, where a test disc (e.g., a Blu-ray Disc or DVD) and matching reference files are provided by a coded audio licensor to a licensee for verification of device compliance. The licensee plays the disc in bitstream mode via an HDMI or optical digital connection, for analysis in an APx analyzer whose HDMI sink EDID is configured to indicate support for such a bitstream. A bit-accurate comparison is made. If even one bit from the DUT does not match the reference file, the measurement indicates a failure.

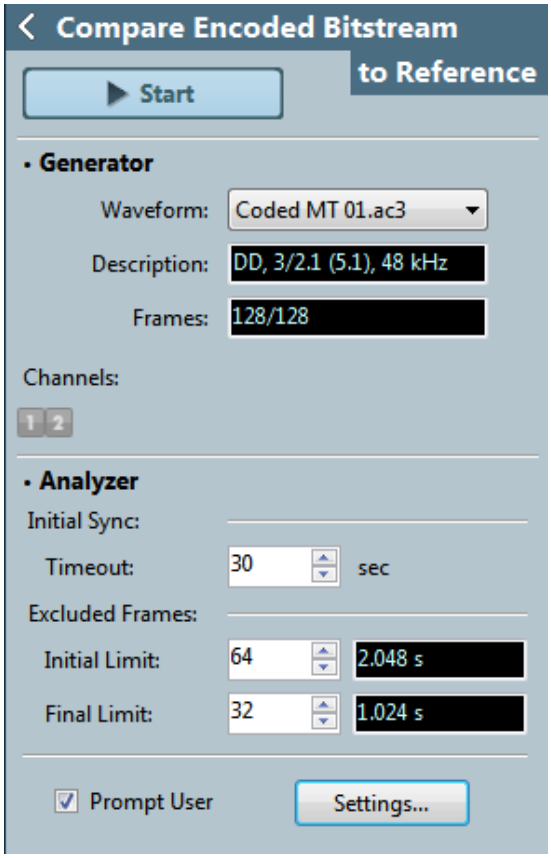
A number of frames at both the beginning and the end of the comparison can be excluded from triggering a measurement failure. These frames are received and examined, but any data mismatches are ignored. Use of these excluded areas prevents errors during disc player “spin up” and “spin down” times from causing the comparison to fail.

Received anomalies and their locations are reported, along with statistical summaries. For error frames, the hex data of both the reference file and the incoming data stream can be viewed.

Operation in Closed Loop mode

This measurement can be used in closed-loop mode to verify the compliance of a device such as an HDMI or IEC60958 repeater. If the APx DSIO module is fitted (for chip-level interface), HDMI or IEC60958 transmitter or receiver integrated circuits can be tested.

For Closed Loop operation, go to Signal Path Setup and select a digital output as your Output Configuration.



Signal Generation

Waveform

Choose a coded generator waveform file here. The same file will be used as the reference file for the comparison. The filename, description and frame count will be displayed in the associated fields.

Operation in Open Loop (External Source) mode

For Open Loop (External Source) operation such as disc player testing, go to Signal Path Setup and select None (External) as your Output Configuration.

Signal Acquisition and Analysis

Reference File:

Choose a reference file here. The file data must be bit-for-bit identical to the embedded coded data in the bit-stream played from the DUT; typically, you will be using a test disc and matching test reference file provided by the coded audio licensor.

The **Description** and **Frame Count** will be displayed in the associated fields.

Initial Sync

Timeout

Many devices require a period of time to start, stabilize and synchronize after a play command has been issued. Set the maximum time allowed to wait for a valid frame here.

Excluded Frames

During startup, many devices issue invalid frames or skip frames altogether. You can exclude a range of frames at the beginning and the end of an acquisition, so that invalid frames in these regions will not cause the measurement to fail.

Initial Limit

Set the number of frames to exclude at the beginning of the acquisition in this field. The adjacent field shows the duration of the excluded range in seconds. This setting is also displayed as a gray band at the left side of the bar graph in the measurement panel.

Final Limit

Set the number of frames to exclude at the end of the acquisition in this field. The adjacent field shows the duration of the excluded range in seconds. This setting is also displayed as a gray band at the right side of the bar graph in the measurement panel.

Prompt User

You can optionally create an on-screen User Prompt for this measurement.

Settings

The optional User Prompt has Settings properties similar to the Pass/Fail Prompts, which are discussed on page 537.

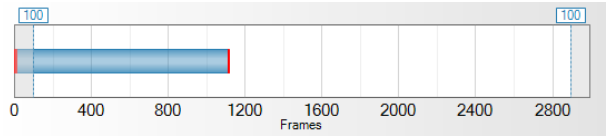
Results for Compare Encoded Bitstream to Reference

Overview

There are three possible results for this measurement.

- Failed due to timeout.
- Failed due to frame error.
- Passed.

Bar Graph Display



The top of the results area has a bar graph display that indicates received frames. Invalid frames are shown in red; valid frames are shown in blue. The excluded areas are represented by gray bands at the left and the right of the display.

Statistical and Descriptive Results

Reference File Frame Count:	2994
Received Frame Count:	1109
Valid Frame Range:	5 - 1111
Skipped Frame Range:	0 - 3
Excluded Error Frames:	4
Error Frame Number:	1112
Error Frame Description:	One or more data values in the fra...

The center of the results area reports statistical and descriptive information as text, for diagnostic purposes.

Reference File Frame Count

The number of frames in the reference file.

Received Frame Count

The number of valid frames received.

Valid Frame Range

The range(s) of valid frames.

Skipped Frame Range

The range(s) of skipped frames.

Excluded Error Frame Range

Error range(s) of error frames in the excluded areas.

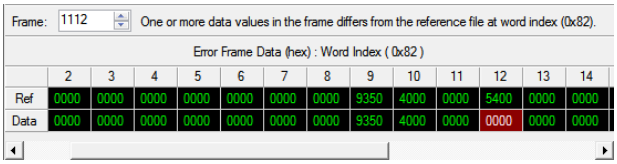
Error Frame Number

The number of the error frame found in the data area (between the excluded areas).

Error Frame Description

A description of the type of fault found in the error frame.

Data Grid viewer



The bottom of the results area displays specific information about invalid and skipped frames.

The Frame window allows you to select among the invalid or skipped frames for more information, which is shown in the data grid below.

The data grid displays hex values for 24 bytes of the selected frame, for both the reference file and the acquired data.

Continuous Sweep Measurements

Introduction

> *For the APx515, this measurement requires a software option key. See page 189 for more information about software options.*

A Continuous Sweep is a brief log-swept sine wave (a Farina log chirp) that moves continuously across a specified range of frequencies. The DUT output is acquired by the analyzer and is mathematically processed to provide a number of results. For more information about continuous sweep measurements, go to page 246.

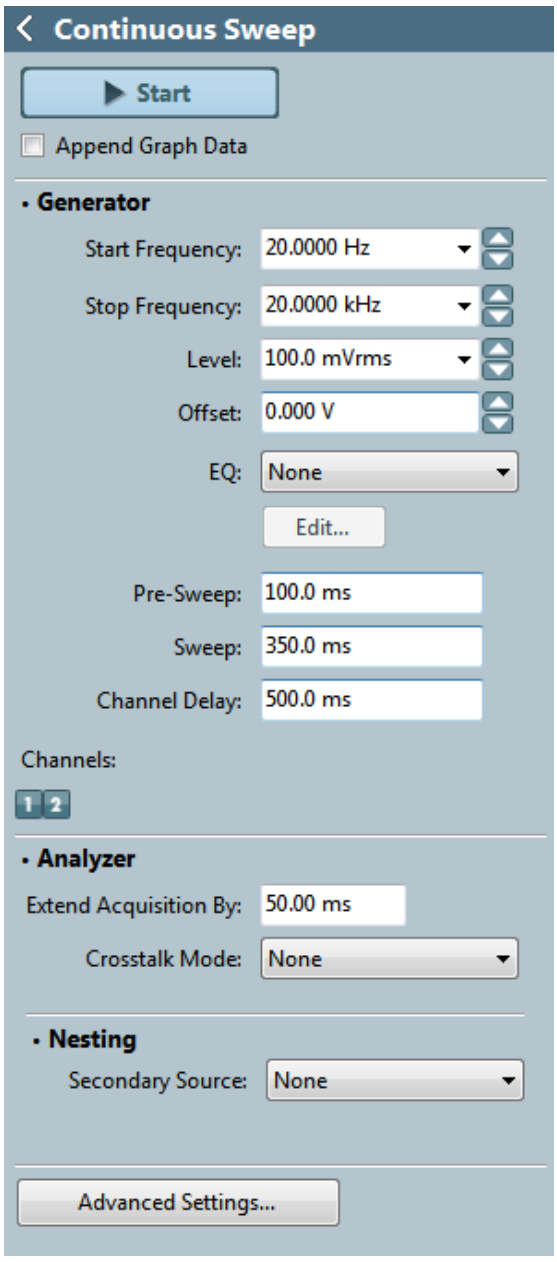
> *Continuous sweep based measurements, ire not supported in External Source or File Input configurations.*

Continuous Sweep results available in APx500 are:

- Level
- Gain
- Relative Level
- Deviation
- Phase
- Group Delay
- THD Ratio
- THD Level
- Distortion Product Ratio
- Distortion Product Level
- Impulse Response
- Acquired Waveform
- Crosstalk One Channel Driven
- Acquired Crosstalk Waveform One Channel Driven
- Crosstalk, One Channel Undriven
- Acquired Crosstalk Waveform One Channel Undriven

Operation

If you have not yet set up your test, first go to Signal Path Setup to select and configure inputs and outputs, references and other settings. See Chapter 7 for information about Signal Path Setup.



Append Graph Data

Measured data are grouped in a **Data Set**, which contains all acquisition results. Normally, the current graph data are deleted each time you start a new measurement. If the **Append Graph Data** box is checked, appended data are grouped as additional data sets, as are any imported data. The Data Sets Panel displays the Data Set(s) in the measurement, and provides a visibility checkbox, a Time acquisition field, and a Notes field for each data set.

Read more about Data Sets beginning on page 649.

Generator

The Continuous Sweep measurement uses a log-swept sine chirp stimulus signal, swept between the frequencies entered in **Start Frequency** and **Stop Frequency**, at the value set in **Level**.

Running the Measurement

To use Continuous Sweep, click **Start**. The generator will output the test signal to the DUT on the selected generator channels. Read the results in the selected result view.

Start Frequency and Stop Frequency

These controls set the frequency range of the sweep. See page 246 for more about continuous sweep ranges, which are dependent upon the instrument.

Level

Set the generator level here.

Offset

Set any DC offset to the generator signal here.

EQ

You can equalize the generator signal to make a pre-emphasized stimulus. See Generator Equalization (page 192) for a general discussion of this feature.

Choosing generator equalization

Select **None, Relative** or **Absolute** from the EQ drop-down menu.

- **None** applies no equalization to the generator.
- **Relative** applies the gain or loss specified in the EQ table to the current generator level for each channel.
- **Absolute** sets the generator level for all channels to the level specified in the EQ table. Note: when **Absolute** is selected, current channel level settings are lost.

Edit

If you have chosen to equalize the generator signal, you can edit the EQ table here. See **Edit EQ Table** (page 192).

Pre-Sweep and Sweep duration fields

A continuous sweep stimulus consists of a sine wave swept logarithmically across the sweep range in the period of time set in these fields.

To enable the DUT to stabilize before the beginning of the defined sweep measurement, there is variable **Pre-Sweep** time. The actual sweep stimulus signal begins below the defined **Start Frequency** and is swept upward, reaching the **Start Frequency** at the end of the **Pre-Sweep** time. The sum of the times in **Pre-Sweep** and **Sweep** is the total sweep length.

- The default for **Pre-Sweep** is 100.0 ms. Minimum is 0 s; maximum is 1.0 s.
- The default for **Sweep** is 350.0 ms. Minimum is 50.0 ms; maximum is 5 s.

Longer **Pre-Sweep** settings allow more time for DUT stabilization, but increase measurement time. Longer **Sweep** settings provide greater resolution and signal-to-noise ratio, but increase measurement time.

Channel Delay

The **Channel Delay** control allows you to specify a delay before the **Start** moments for each channel when in one of the **High Accuracy Crosstalk** modes.

If the **Channel Delay** is zero or very short, the slew rate limiting of a DUT or the time constant in **AC Coupling** can cause a transient when energy from the sweep in a previous channel appears in the analysis of the current channel. The default **Channel Delay** is 500 ms. If DC coupled and the DUT does not have appreciable slew rate limiting, Channel Delay can be set to a very low value, or 0.

Analyzer

The sweep is acquired and processed to provide the various results.

There are no local measurement filters available for continuous sweep measurements. However, low pass, high pass and weighting input filter settings made in Signal Path Setup > Input/Output > Filters will also affect this measurement. See Chapter 97 for more information about filtering.

Extend Acquisition By

By default, a continuous sweep measurement makes an acquisition slightly longer than the stimulus, to include possible time-delayed artifacts created in the DUT. By default, the acquisition is extended 50 ms longer than the stimulus.

In some cases, you may want to extend the acquisition further. Enter a new value in the **Extend Acquisition By:** field. Minimum extension is 0 s; maximum is 3 s.

Crosstalk Mode

Choose the mode of crosstalk measurement (or **None**). This selection changes the sweep alignment in time across channels, and may affect other continuous sweep results. For more information, see Continuous Sweep Crosstalk modes on page 248.

Choices are

- None
- **High speed** (default)
- High accuracy, one channel driven
- High accuracy, one channel undriven

Nesting

Continuous Sweeps can be nested. Read about Nested Sweeps beginning on page 185.

Secondary Source (nested sweeps)

Select the secondary sweep source here. This is the outside, nesting parameter that is varied to change the conditions for the primary sweep through a number of iterations.

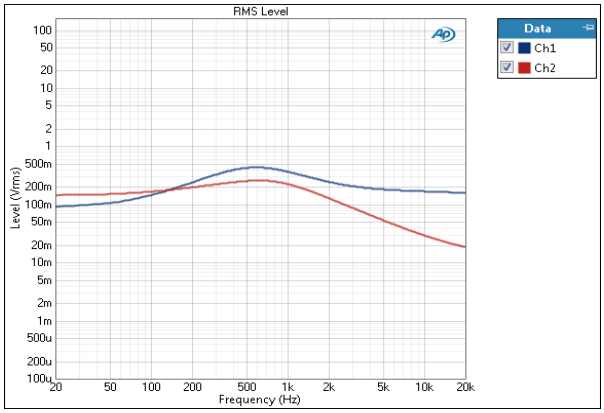
Available choices are dependent upon instrument hardware, options, Input/Output settings and Bench Mode Generator settings.

Advanced Settings

If your test requires individual generator settings for each channel or special analyzer ranging, see Advanced Settings for Continuous Sweep on page 249.

See Chapter 104 for more information about units of measurement.

Continuous Sweep: Level



The Level result is a continuous sweep measurement that provides a graphical display of the frequency response of each channel. In this result the DUT output level is plotted against frequency.

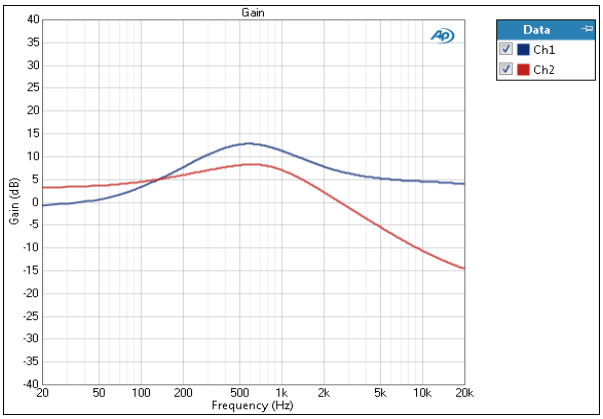
If the settings are identical, this will provide the same results as the **Frequency Response: Level** result.

Units

Units available for Continuous Sweep: Level are:

X-axis	*Y-axis (analog)*	*Y-axis (digital)*
• Hz	• Vrms	• dBFS
• dHz	• dBV	• FS
• F/R	• dBu	• %FS
• %Hz	• dBrA	• dBrA
	• dBrB	• dBrB
	• dBSPL1	• dBSPL1
	• dBSPL2	• dBSPL2
	• dBm	
	• W (watts)	

Continuous Sweep: Gain



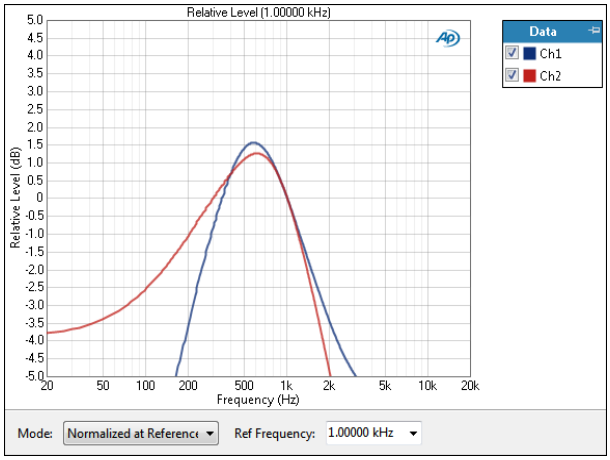
The Gain result is a continuous sweep measurement that provides a graphical display of the frequency response of each channel. In this result the DUT gain is plotted against frequency.

Units

Units available for Continuous Sweep: Gain are:

X-axis	*Y-axis same-domain*	*Y-axis cross-domain*
• Hz	• x/y	• FS/Vrms
• dHz	• %	• dB(FS/Vrms)
• F/R	• ppm	—or—
• %Hz	• dB	• Vrms/FS
		• dB(Vrms/FS)

Continuous Sweep: Relative Level



The Relative Level result is a continuous sweep measurement that provides a graphical display of the frequency response of each channel. In this result the DUT output level is plotted against frequency, relative to the level at a selected reference frequency. This enables you to specify the frequency that will be set as 0 dB and view the response in relation to that frequency. A midband frequency such as 1 kHz is often set as the reference frequency, and is the default here.

If the settings are identical, this will provide the same results as the Frequency Response: Relative Level result.

You can change the reference frequency at any time (except after appending) and the graph will immediately redraw around the new reference.

Result Settings

These settings are made in the Result Settings bar, beneath the graph display.

Mode

Normalized at Reference

Enter a reference frequency for normalization in the **Ref Frequency** field. The measured level at the frequency you choose is set as 0 dB on the graph, and the DUT response is plotted in relation to this reference. You can change the **Ref Frequency** at any time (except after appending) and the graph will immediately be redrawn to reflect the new setting.

Auto-Centered in Limits

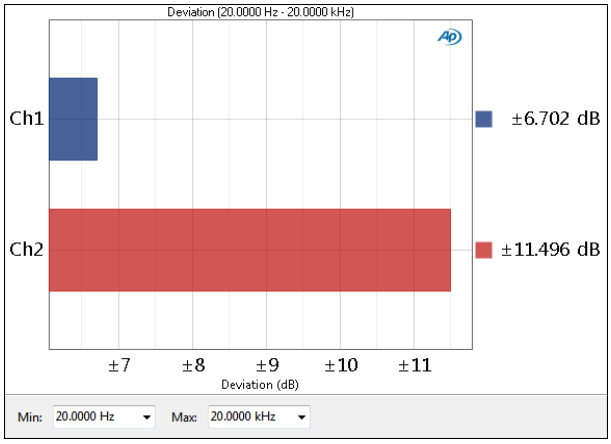
This control is only usable if upper and lower limits are set on this graph. In some applications, the general conformance of a trace to the shape of a set of limits is of importance, with absolute level less important or irrelevant. **Auto-Centered in Limits** centers the data trace relative to the upper and lower limits. To exclude possible exceptional data points at frequency extremes, the frequency range that is considered for the centering operation is constrained by the values in the **Min** and **Max** fields.

Units

Units available for Continuous Sweep: Relative Level are:

X-axis	*Y-axis*
• Hz	• x/y
• dHz	• %
• F/R	• ppm
• %Hz	• dB

Continuous Sweep: Deviation



Continuous Sweep Deviation is a single value result computed from the continuous sweep acquisition. In this result the frequency deviation (the total range of frequency variation) of each channel is displayed as a meter bar. You can specify a minimum and maximum frequency to define the range be considered in the deviation measurement.

To measure Continuous Sweep Deviation, first enter a frequency range in the **Min Frequency** and **Max Frequency** fields below the graph; or, accept the default 20 Hz to 20 kHz range. Click **Start**. you can change the deviation **Min Frequency** or **Max Frequency** settings at any time and the meter bars will immediately redraw to reflect the new settings.

Result Settings

These settings are made in the Result Settings bar, beneath the graph display.

Min

Set the minimum frequency of the range of interest here.

Max

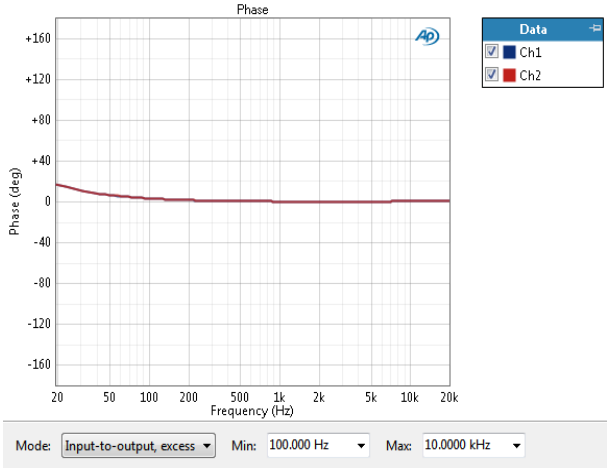
Set the maximum frequency of the range of interest here.

Units

Units available for Continuous Sweep: Deviation are

- dB

Continuous Sweep: Phase



The Continuous Sweep: Phase result displays the DUT output phase for each channel, plotted against frequency. See page 332 for more information about phase measurements.

Result Settings

These settings are made in the Result Settings bar, beneath the graph display.

Mode

Continuous Sweep Phase has four result modes:

Relative to Ch1

In this mode, the absolute phase of channel one is subtracted from the absolute phase of channels numbered greater than 1. The result is plotted against frequency for each channel numbered 2 and above, showing the phase difference (from channel 1) for each channel. Since channel 1 is used as the reference, it is not plotted in this result. This mode shows "unwrapped" phase differences.

Input-to-output

In the input-to-output mode, the absolute phase of each channel, from device input to device output, is plotted against frequency, "unwrapped." Input-to-output mode includes device delay.

Input-to-output, wrapped

This mode shows the same result as input-to-output phase, but "wrapped" within the range of –180° to +180°.

Input-to-output, excess

This mode shows the input-to-output phase (unwrapped), but removes the linear component (the average group delay of the system), leaving the "excess phase."

Some devices have linear phase only over a portion of their passband. For such devices, including frequency ranges where the phase is non-linear affects the excess phase result, which becomes dependent on the stop frequency of the measurement. Although the measurement is technically correct, the variation of the result with stop frequency is disconcerting and not particularly useful.

You may (optionally) use the **Min**: and **Max**: settings to set the expected linear phase range of the device under test. For this result, the device delay is computed as the best linear fit to the phase data between the **Min** frequency and the **Max** frequency. Excess phase is then computed as the difference from the linear fit.

Min and **Max** frequencies cannot extend beyond sweep **Start** and **Stop** frequencies.

See more information about wrapped and unwrapped phase in the next paragraphs.

Units

Units available for Continuous Sweep: Phase are:

X-axis	*Y-axis*
• Hz	• deg
• dHz	• rad
• F/R	
• %Hz	

More about Wrapped and Unwrapped Phase

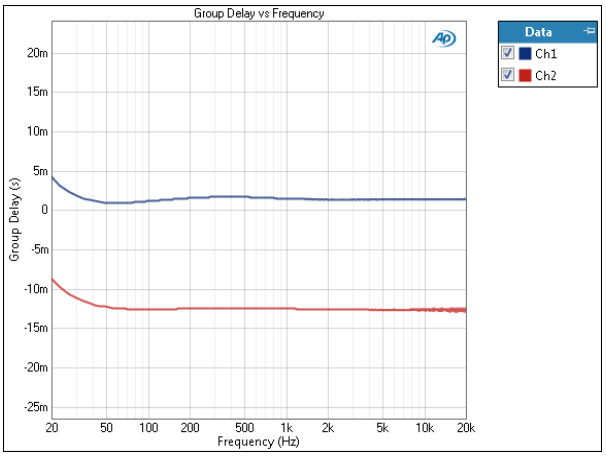
Wrapped phase results

Phase angles within a circle are typically expressed in degrees within the range of 0° to 360°. Using this frame of reference, if the phase of a signal has shifted by exactly 360°, it is considered to have "wrapped" back to 0° and is now "back in phase." Raw, unwrapped phase values are converted to wrapped values by subtracting 360 *n* times, until there is a remainder between 0° and 360°. This remainder is the wrapped value. For example, an unwrapped phase measurement of 742° would be displayed as a wrapped value of 22°. APx500 expresses wrapped phase using the range of –180° to +180°.

Unwrapped phase results

Phase angles of many thousands of degrees can be shown in an unwrapped phase mode.

Continuous Sweep: Group Delay



Group delay is a measure of the rate of change of phase shift as a function of frequency. Group delay can also be described as a time delay of a group of frequencies with respect to the generator.

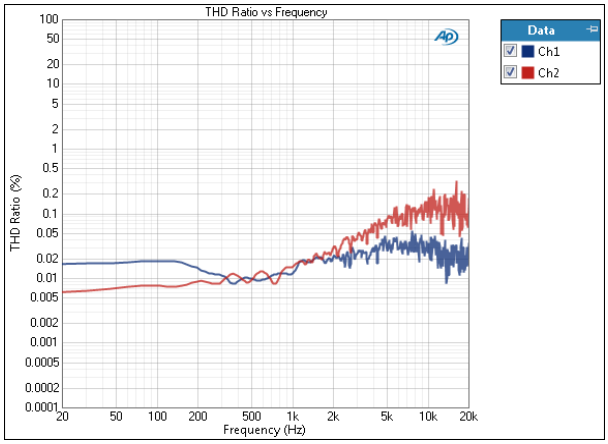
The Group Delay result is a continuous sweep measurement that displays the group delay measured in each channel, plotted against frequency.

Units

Units available for Continuous Sweep: Group Delay are:

X-axis	*Y-axis*
• Hz	• s (seconds)
• dHz	
• F/R	
• %Hz	

Continuous Sweep: THD Ratio



The THD Ratio result is a continuous sweep measurement that provides a graphical display of the harmonic distortion response of each channel. In this result the ratio of the level of the THD (total harmonic distortion) to the fundamental in the DUT output is plotted against frequency.

See more information about THD below.

Units

Units available for Continuous Sweep: THD Ratio are:

X-axis	*Y-axis*
• Hz	• x/y
• dHz	• %
• F/R	• ppm
• %Hz	• dB

More about THD

THD is an abbreviation for Total Harmonic Distortion. This refers to the sum of all the harmonic distortion products in a measurement. Noise is not included in a THD measurement.

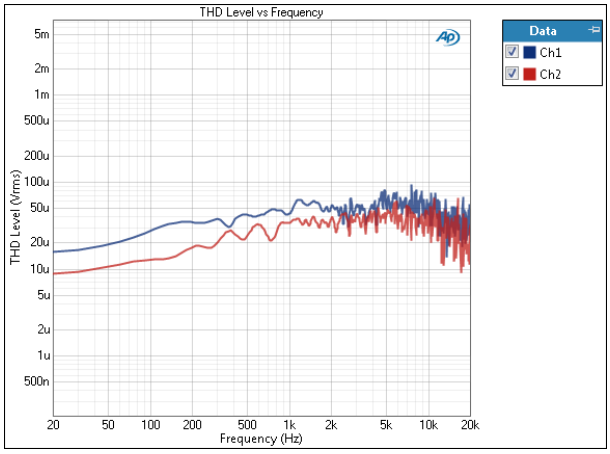
THD without the noise was not easily done until the advent of FFT measurement techniques. The APx500 implementation measures only the bins that contain harmonic distortion products, producing a result that is not influenced by DUT noise or spurious interfering signals. In a very low-noise system, the THD result will be the same as the THD+N result.

The FFT method that provides THD results also brings the capability to selectively measure discrete harmonic distortion products.

In the various THD and Distortion Product ratio measurements in APx, the rms level of the measured distortion is divided by the rms level of the fundamental, unlike the THD+N ratio results discussed above, where the ratio is to the total signal. The difference is negligible for all reasonable distortion levels.

Also see More about THD+N on page 527.

Continuous Sweep: THD Level



The THD Level result is a continuous sweep measurement that provides a graphical display of the harmonic distortion response of each channel. In this result the level of the THD (total harmonic distortion) in the DUT output is plotted against frequency.

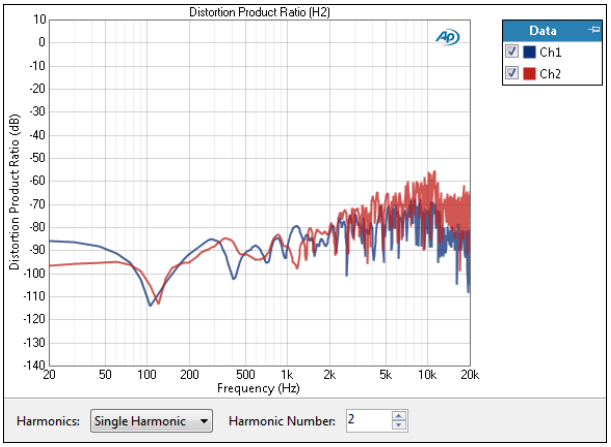
See the previous topic for more information about THD.

Units

Units available for Continuous Sweep: THD Level are:

X-axis	*Y-axis (analog)*	*Y-axis (digital)*
• Hz	• Vrms	• dBFS
• dHz	• dBV	• FS
• F/R	• dBu	• %FS
• %Hz	• dBrA	• dBrA
	• dBrB	• dBrB
	• dBSPL1	• dBSPL1
	• dBSPL2	• dBSPL2
	• dBm	
	• W (watts)	

Continuous Sweep: Distortion Product Ratio



The Continuous Sweep Distortion Product Ratio result is a continuous sweep measurement that provides a graphical display of the selected harmonic distortion products present in each channel. In this result the ratio of the level of the selected harmonic distortion product to the fundamental in the DUT output is plotted against frequency.

Select the harmonic distortion product(s) you wish to view using the **Harmonics** controls, described below.

Result Settings

These settings are made in the Result Settings bar, beneath the graph display.

Harmonics

For a graph of the ratio of one specific harmonic product to the signal level, select **Single Harmonic** and set the **Harmonic Number** to your choice, from 2nd harmonic to 20th harmonic.

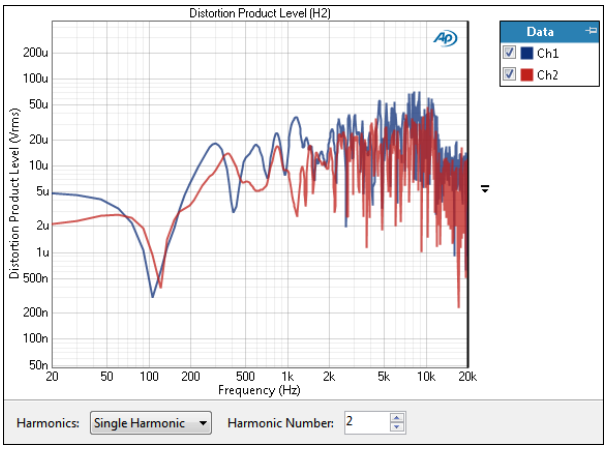
For a graph of the sum of the levels of any combination of harmonic products (from **H2** through **H20**), divided by the signal level, select **Sum of Harmonics**. Then click **Select Harmonics...** to open a result definition dialog.

Units

Units available for Continuous Sweep Distortion Product Ratio are:

X-axis	*Y-axis*
• Hz	• x/y
• dHz	• %
• F/R	• ppm
• %Hz	• dB

Continuous Sweep: Distortion Product Level



The Continuous Sweep Distortion Product Level result is a continuous sweep measurement that provides a graphical display of the selected harmonic distortion products present in each channel. In this result the level of the selected harmonic distortion product in the DUT output is plotted against frequency.

Select the harmonic distortion product(s) you wish to view using the **Harmonics** controls, described below.

Result Settings

These settings are made in the Result Settings bar, beneath the graph display.

Harmonics

For a graph of the level of one specific harmonic product, select **Single Harmonic** and set the **Harmonic Number** to your choice, from 2nd harmonic to 20th harmonic.

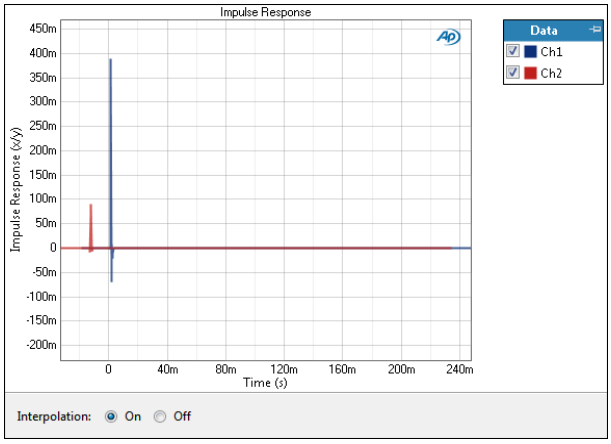
For a graph of the level of the sum of several or all harmonic products (from **H2** through **H20**), select **Sum of Harmonics**, then click **Select Harmonics...** to open a result definition dialog.

Units

Units available for Continuous Sweep Distortion Product Level are:

X-axis	*Y-axis (analog)*	*Y-axis (digital)*
• Hz	• Vrms	• dBFS
• dHz	• dBV	• FS
• F/R	• dBu	• %FS
• %Hz	• dBrA	• dBrA
	• dBrB	• dBrB
	• dBSPL1	• dBSPL1
	• dBSPL2	• dBSPL2
	• dBm	
	• W (watts)	

Continuous Sweep: Impulse Response



The Impulse Response result is a continuous sweep measurement that displays the impulse response for each channel. Impulse response is plotted against time.

See page 244 for more information about impulse response.

Result Settings

These settings are made in the Result Settings bar, beneath the graph display.

Interpolation

For time-domain displays, the APx500 graphs are normally plotted using sinc function interpolation. This is the default setting. However, digital domain signals are sometimes best understood when viewing actual samples, with no interpolation. Turn display interpolation **ON** or **OFF** here.

Units

Units available for Impulse Response are:

X-axis	*Y-axis same-domain*	*Y-axis cross-domain*
• s (seconds)	• x/y	• D/V —or— • V/D

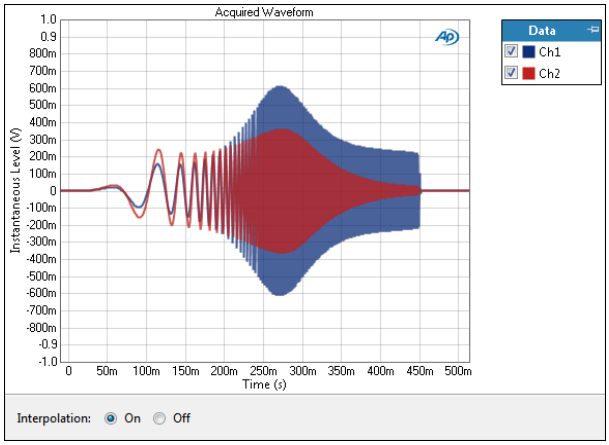
More about Impulse Response

The impulse response of the DUT is derived from the APx500 log-swept continuous sweep measurement. This gives the same result (although with better signal to noise ratio) as would be produced by stimulating the DUT with an impulse and observing the output in the time domain. This result is useful to study artifacts that move or spread the stimulus in the time domain, such as delay, reverberation, echo or reflection, etc.

Impulse response is a property of the device or system under test, and for same-domain measurements (analog to analog, for example) the impulse response is unitless, which we represent in APx500 as “x/y”. For cross-domain measurements, the impulse response is expressed as the relationship between output and input. A DAC, for example, has an impulse response expressed in units of V/D. See Chapter 104 for more information about units of measurement.

An ideal impulse response does not change with stimulus level.

Continuous Sweep: Acquired Waveform



The Acquired Waveform result is a continuous sweep measurement that displays the Acquired Waveform for each channel. See page 246 for more information about acquired waveform.

Result Settings

These settings are made in the Result Settings bar, beneath the graph display.

Interpolation

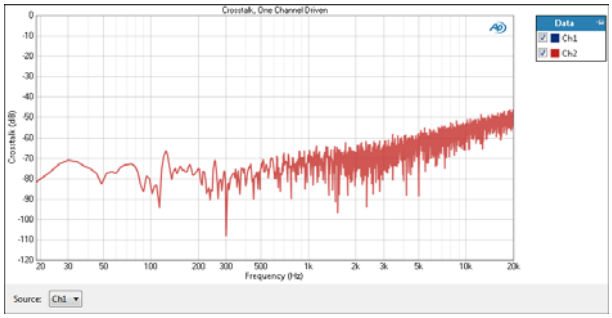
For time-domain displays, the APx500 graphs are normally plotted using sinc function interpolation. Interpolation adds points to the displayed trace that do not exist in the measured data, to make data trends more easily visualized. The default in APx is to have Interpolation **On**. However, digital domain signals are sometimes best understood when viewing actual samples, with interpolation switched **Off**. Also see “Interpolation and limit failure markers” on page 678.

Units

Units available for Acquired Waveform are:

X-axis	*Y-axis (analog)*	*Y-axis (digital)*
• s (seconds)	• V	• D • hex

Continuous Sweep: Crosstalk, One Channel Driven



The Crosstalk, One Channel Driven result provides a graphical display of the crosstalk from a selected source channel into each of the other channels. See page 264 for more information about crosstalk.

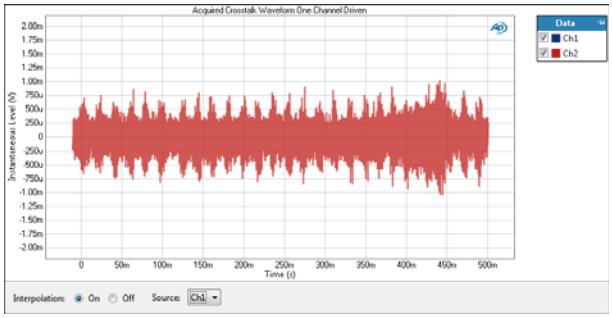
Units

Crosstalk is expressed as a ratio, with the crosstalk in the measured channel divided by the actual signal level in that channel. The result is typically stated in decibels.

Units available for Continuous Sweep: Crosstalk, One Channel Driven are:

X-axis	*Y-axis*
• Hz	• x/y
• dHz	• %
• F/R	• ppm
• %Hz	• dB

Continuous Sweep: Acquired Crosstalk Waveform, One Channel Driven



The Acquired Crosstalk Waveform, One Channel Driven result is a continuous sweep measurement that displays the Acquired Crosstalk Waveform for each undriven channel. This acquisition is only made when Crosstalk Mode is set to **High Accuracy, One Channel Driven**. See page 246 for more information about acquired waveform.

Result Settings

These settings are made in the Result Settings bar, beneath the graph display.

Interpolation

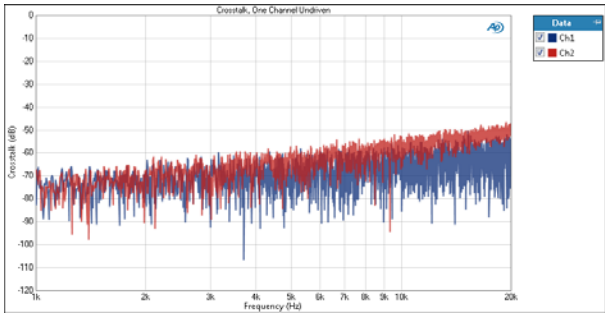
For time-domain displays, the APx500 graphs are normally plotted using sinc function interpolation. Interpolation adds points to the displayed trace that do not exist in the measured data, to make data trends more easily visualized. The default in APx is to have Interpolation **On**. However, digital domain signals are sometimes best understood when viewing actual samples, with interpolation switched **Off**. Also see “Interpolation and limit failure markers” on page 678.

Units

Units available for Acquired Crosstalk Waveform are:

X-axis	*Y-axis (analog)*	*Y-axis (digital)*
• s (seconds)	• V	• D • hex

Continuous Sweep: Crosstalk, One Channel Undriven



The Crosstalk, One Channel Undriven result is a continuous sweep measurement that provides a graphical display of the combined crosstalk into each

channel from all the other channels. See page 264 for more information about crosstalk.

To measure make a continuous sweep Crosstalk, One Channel Undriven measurement, first select a source channel the **Source** list beneath the graph. Click **Start**. You can select another Source channel, and repeat the test by clicking Start again.

You can change the Source channel selection at any time (except after appending) and the graph will immediately redraw to show crosstalk from that new source.

Result Settings

These settings are made in the Result Settings bar, beneath the graph display.

Source

Designate a channel as the Source for crosstalk results.

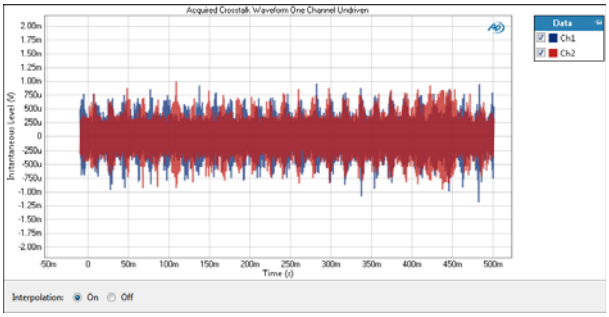
Units

Crosstalk is expressed as a ratio, with the crosstalk in the measured channel divided by the fundamental signal level in that channel. The result is typically stated in decibels.

Units available for Continuous Sweep: Crosstalk, One Channel Undriven are:

X-axis	*Y-axis*
• Hz	• x/y
• dHz	• %
• F/R	• ppm
• %Hz	• dB

Continuous Sweep: Acquired Crosstalk Waveform, One Channel Undriven



The Acquired Crosstalk Waveform, One Channel Undriven result is a continuous sweep measurement that displays the Acquired Crosstalk Waveform for each channel. This acquisition is only made when Crosstalk Mode is set to **High Accuracy, One Channel Undriven**. See page 246 for more information about acquired waveform.

Result Settings

These settings are made in the Result Settings bar, beneath the graph display.

Interpolation

For time-domain displays, the APx500 graphs are normally plotted using sinc function interpolation. Interpolation adds points to the displayed trace that do not exist in the measured data, to make data trends more easily visualized. The default in APx is to have Interpolation **On**. However, digital domain signals are sometimes best understood when viewing actual samples, with interpolation switched **Off**. Also see “Interpolation and limit failure markers” on page 678.

Units

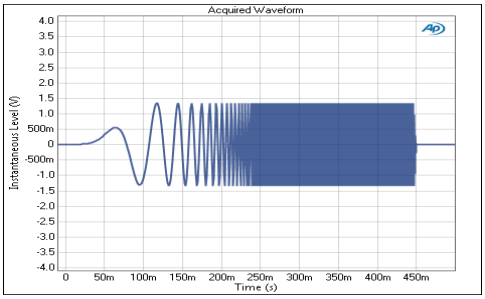
Units available for Acquired Crosstalk Waveform are:

X-axis	*Y-axis (analog)*	*Y-axis (digital)*
• s (seconds)	• V	• D • hex

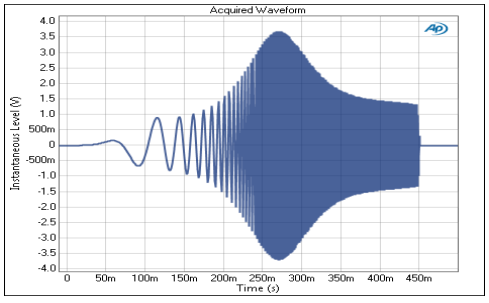
More about Acquired Waveforms

After having passed through the DUT, the continuous sweep waveform is acquired into memory for analysis.

The Acquired Waveform result is a time-domain display of the waveform as acquired.



Acquired Waveform, flat DUT.



Acquired Waveform, EQ’d DUT.

More about Continuous Sweep

The APx Continuous Sweep is an implementation of impulse response measurement methods using log-swept sines. The Continuous Sweep method is used in

the APx500 Continuous Sweep, Frequency Response and Acoustic Response measurements.

A continuous sweep provides a brief broadband stimulus signal to the DUT. This stimulus consists of a log-swept sine wave (often called a *log chirp* or a *chirp*) that moves continuously across a specified range of frequencies. The DUT output is acquired by the analyzer and is mathematically processed to provide a number of results.

Advantages

Continuous sweep is very fast and can provide many high-resolution results from a single acquisition. A typical continuous sweep measurement can return 15 separate results, each made up of many thousands of data points across multiple channels through a range of 20 Hz to 20 kHz, in about 4 seconds.

Limitations

1. When **Crosstalk Type** is set to **High Speed**, the **Continuous Sweep** measurement generates log chirps that are delayed from channel to channel, with channel 2 slightly later than 1, channel 3 slightly later than 2, and so on. These staggered sweeps enable the processing algorithms to determine the interchannel crosstalk for frequencies above 1 kHz, all from the same acquisition. In some circumstances, the staggered sweeps can produce unexpected results. See “Continuous Sweep Crosstalk modes” on page 248 and “Solutions for Unexpected Continuous Sweep Crosstalk Results” on page 249.
   Frequency Response and **Acoustic Response** measurements do not provide crosstalk results, and do not use staggered sweeps.
2. Continuous sweep based measurements must be used in closed loop configuration. Open loop (External Source) configurations are not supported.
3. Total delay through device or system under test must be less than 3 seconds.
4. Sweeps with **Stop** frequencies less than 1 kHz are not supported. See “Sweep range,” below.
5. Harmonic Distortion is measured using the THD method. This method may produce different results from the traditional THD+N method. See “More about THD” on page 242 and “More about THD+N” on page 527.

Sweep range

In the Audio Precision implementation, the sine can be swept from the system minimum (0.1 Hz for the APx525 family, 2.0 Hz for the APx515, and 5.0 Hz for the APx585 family) to a maximum of 80 kHz. The default range is 20 Hz to 20 kHz.

Start Frequency can be set to any frequency within the system’s range, up to a frequency equal to 1/2 the current **Stop Frequency**.

Stop Frequency can be set to any frequency within the range of 1 kHz to 80 kHz (depending upon system bandwidth setting).

Sweep duration

The duration of the sweep can be adjusted from a minimum of 50 ms to a maximum of 5 s, with a default of 350 ms. A shorter duration, of course, provides for a faster test: stimulus, acquisition and processing time are all reduced. Longer sweeps provide greater resolution and a better signal to noise ratio. The 350 ms default length is a good compromise between speed and resolution and provides excellent results.

Pre-sweep duration

The total length of the stimulus includes a pre-sweep time in addition to the sweep time. The pre-sweep is an extension of the sweep that begins slightly below the specified start frequency, reaching the start frequency at the end of the pre-sweep time. The amplitude of the pre-sweep is modulated from zero to the sweep level to avoid an initial transient “pop.” Pre-sweep times can be set from 0 s to 1 s, with a default of 100 ms.

Extend Acquisition

The DUT output is acquired, digitized and processed in DSP. The length of the acquisition is usually somewhat longer than the stimulus sweep to allow for time-delayed artifacts created in the DUT. In APx500, the acquisition can be extended up to 3.00 s after end of stimulus. The default is 50 ms.

Input Range

The analyzer inputs must be ranged to provide an optimal acquisition level for a continuous sweep. The APx500 implementation uses autoranging, with the first range set to the lowest level for the best noise performance. If this range is too low, data from the out-of-range sweep are discarded. The input range is moved up and the sweep is repeated. This process may result in several sweep attempts before the correct range is determined; however, since the sweeps are very fast (typically one second) the total acquisition time is short.

If your sweep is a step in an automated sequence that you would like to run as fast as possible, you can optimize the speed of the ranging process by setting the range floor to the correct range for the measurement.

Ranging is set on the Advanced Settings panel (page 249).

Continuous Sweep Distortion Product results

Similar to the meter bar results, the continuous sweep harmonic distortion product results provide great detail about the makeup of the harmonic distortion. For each channel, a selected harmonic product is plotted against frequency. You can choose to view the plot of any harmonic from 2nd to 20th.

In the Distortion Product Level result, the graph displays rms level. In the Distortion Product Ratio result, the graph displays the ratio of the rms level of the distortion product to the rms level of the fundamental tone.

Troubleshooting Unexpected Continuous Sweep Results

Continuous sweep measurements can quickly gather data for many results across multiple channels. However, adverse conditions in the DUT or system under test can produce unexpected results when using Continuous Sweep. These conditions include high distortion (such as clipping), presence of spurious signals (hum, buzz or other signals), or high crosstalk, including crosstalk of distortion products within the DUT.

The solution generally is to configure or adjust the DUT or system under test to minimize these conditions. In some cases, you may have to choose another APx measurement for your test.

Continuous Sweep Crosstalk modes

Overview

APx500 now offers a selection of Crosstalk modes for Continuous Sweep. These modes affect the time-alignment of the sweep stimulus across channels, and the results available for **Continuous Sweep Crosstalk, One Channel Driven** and **Continuous Sweep Crosstalk, One Channel Undriven**.

These choices offer an alternative to the original Continuous Sweep crosstalk mode (High Speed), which required that the continuous sweep stimulus signal be delayed across output channels.

Here's the explanation for this change:

The High Speed crosstalk method relies on starting the sweep at different times, across channels. For example, the sweep in channel 1 starts at T0, the sweep in channel 2 starts at T1, and so on. In analysis, the signals "crosstalking" from other channels can be identified by their time offsets. This method enables the measurement of multichannel crosstalk very quickly, in one sweep cycle.

The time-offset stimulus and the analysis required to separate the crosstalk signals can create confusing artifacts in the results; it returns no results at all below 1 kHz; in some circumstances, it can affect other continuous sweep results. The Crosstalk Type selection feature allows you to use the High Speed method or choose alternatives that do not use time-offset sweep signals. The High Accuracy crosstalk modes provide excellent results, but require multiple sweep cycles and more test time.

Crosstalk Type

None

Sweeps are aligned in time across channels. There is only one sweep cycle, and no crosstalk results are reported.

High speed (default)

Sweeps are offset in time across channels. There is one sweep cycle, and high-speed crosstalk results are r**eported for both Crosstalk, One Channel Undriven and Crosstalk, One Channel Driven** results.

High accuracy, one channel driven

Sweeps are aligned in time across channels. There are n sweep cycles, where n = channel count. Only one channel is driven per sweep cycle. With each sweep cycle, the driven channel is advanced to the next channel, until all channels have been tested. **Crosstalk, One Channel Driven** are reported.

High accuracy, one channel undriven

Sweeps are aligned in time across channels. There are n sweep cycles, where n = channel count. All channels but one are driven per sweep cycle. With each sweep cycle, the undriven channel is advanced to the next channel, until all channels have been tested. **Crosstalk, One Channel Undriven** results are reported.

Comparison of measurement results

As mentioned above, a key difference in the crosstalk method is the type of stimulus signal used, whether the log chirp is aligned in time across channels, or offset.

High-speed results

No results are returned below 1 kHz. Results between 2 kHz and the maximum sweep frequency agree well with the high-accuracy results. There is a small artifact at the high-frequency end of the sweep.

High-accuracy results

Results are returned across the entire sweep spectrum.

Noise in continuous sweep crosstalk results

Since crosstalk results are often very low in amplitude, the DUT noise or system noise is a measurable component of a crosstalk result when no bandpass filtering is used. Since bandpass filtering is unavailable for continuous sweep measurements, we recommend

using a relatively high stimulus level for continuous sweep crosstalk measurements.

Solutions for Unexpected Continuous Sweep Crosstalk Results

Continuous Sweep High Speed crosstalk method (staggered sweeps across channels)

When Continuous Sweep **Crosstalk Type** is set to **High Speed**, the Continuous Sweep measurement generates log chirps that are delayed from channel to channel, with channel 2 slightly later than 1, channel 3 slightly later than 2, and so on. These "staggered sweeps" enable the analysis software to identify crosstalk signals for all channels within one stimulus/response cycle.

For most testing, this method provides fast, accurate results. However, staggered sweeps and the analysis necessary to interpret the data produce some unexpected conditions and results, which can be complicated by errors in cable connection or DUT characteristics.

- The FFT windowing required to focus on the crosstalk data limits the range of the crosstalk measurements to frequencies above 1 kHz.
- A DUT with different delays across channels will produce erroneous crosstalk results.
- Mis-connection or swapping of channel connections will produce erroneous crosstalk results.
- Similarly, mixing of channels within the DUT will produce erroneous crosstalk results.
- The heuristic required in the crosstalk analysis method creates a measurement floor that is higher than the true instrument floor. This measurement floor is also dependent upon the measurement setup, in particular the relationship between sweep range and sample rate/channel bandwidth. Crosstalk above the measurement floor is correctly measured.

Note: The Frequency Response and Acoustic Response measurements do not provide crosstalk results at all, and thus do not use staggered sweeps.

Alternative crosstalk methods using Continuous Sweep (simultaneous sweeps across channels)

Your measurement requirements or DUT characteristics may make the staggered sweep method undesirable.

The Continuous Sweep measurement provides crosstalk by sequential stimulus/response cycles in these two **Crosstalk Type** selections:

- High accuracy, one channel driven
- High accuracy, one channel undriven

Alternative crosstalk methods using other methods

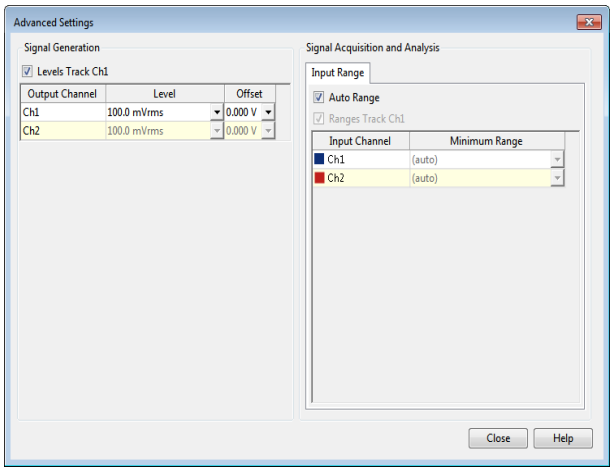
The alternative to using staggered sweeps is to run multiple stimulus/response cycles (one for each channel of interest), with the generator switched to OFF for the channel of interest for each cycle, requiring longer testing times.

For crosstalk meter measurements, you can use the **Crosstalk, One Channel Driven**, **Crosstalk, One Channel Undriven,** or **Crosstalk, Custom** measurements.

For crosstalk versus frequency stepped sweeps, you can use the **Crosstalk Sweep, One Channel Driven**, **Crosstalk Sweep, One Channel Undriven**, **Crosstalk Sweep, Custom** or **Multitone** crosstalk measurements.

Advanced Settings for Continuous Sweep based measurements

The default settings here are appropriate for most measurements. You may want to make adjustments for special situations.



Signal Generation

You can set different output levels or DC offsets for each channel here.

If **Levels Track Ch1** is checked (the default), the generator output level and offset values for channel 1 are copied to the other channels, and output levels for channels greater than 1 cannot be edited. Any changes made to channel 1 are reproduced in the other channels.

To set individual generator output channel levels or offsets, **uncheck** the **Levels Track Ch1** checkbox and

enter values in the output channel Level or Offset fields.

Signal Acquisition and Analysis

Input Range

Digital inputs do not require ranging, and Input Range settings are unavailable when inputs are set to a digital format.

Auto Range

If **Auto Range** is checked (the default), each analog channel determines its input range automatically, based on the level of the input signal.

Continuous Sweep based measurements use a modified autoranging process to find the correct input range.

The sweep acquisition begins in a particular range called the range floor, and moves up from there if necessary. By default, the range floor is the lowest range, but you can specify a higher range by entering the value in the **Minimum Range** field.

If the initial range is correct for the signal, the acquisition is made and the data are processed and displayed. If this range is too low, data from the out-of-range sweep are discarded. The input range is moved up and the sweep is repeated. This process may result in several sweep attempts before the correct range is determined; however, since the sweeps are very fast (typically one second) the total acquisition time is short.

If your sweep is a step in an automated sequence that you would like to run as fast as possible, you can optimize the speed of the ranging process by setting the range floor to the correct range for the measurement.

Fixed Range

Alternatively, you can set a fixed minimum range for the analog inputs. Uncheck **Auto Range** and enter the expected maximum level in the **Minimum Range** filed. The software will choose the lowest range that will accommodate that level; this becomes the minimum range available, and the analyzer will not reset to a lower range. Fixed range is best for signals that vary rapidly, including tone bursts, perceptual audio speech samples, and program material such as voice or music. See **When autoranging isn't suitable** on page 626.

Ranges Track Ch1

If **Ranges Track Ch1** is checked (the default), the fixed input range setting for channel 1 is copied to the other channels, and range settings for channels greater than 1 cannot be edited. Any changes made to channel 1 are reproduced in the other channels. If **Ranges Track Ch1** is not checked, the fixed input range setting for each channel can be set independently.

Read more about ranging in Chapter 98.

Crosstalk Sweep, Custom
(Sequence Mode)

Introduction

For the APx515, this measurement requires a software option. See page 189 for more information about software options.

The Crosstalk Sweep, Custom measurement applies a stepped frequency sweep signal to the DUT. You must take steps to prevent the signal from stimulating the channel of interest, as described later in this chapter. The level measured at each frequency point in the undriven channel is compared to the level measured in a driven channel. The ratio is the crosstalk for that channel. X axis is generator frequency; Y axis is crosstalk ratio.

Read more about crosstalk beginning on page 264.

The Crosstalk Sweep, One Channel Driven measurement is not available in External Source or in File Input configurations.

Operation

If you have not yet set up your test, first go to Signal Path Setup (Chapter 7).

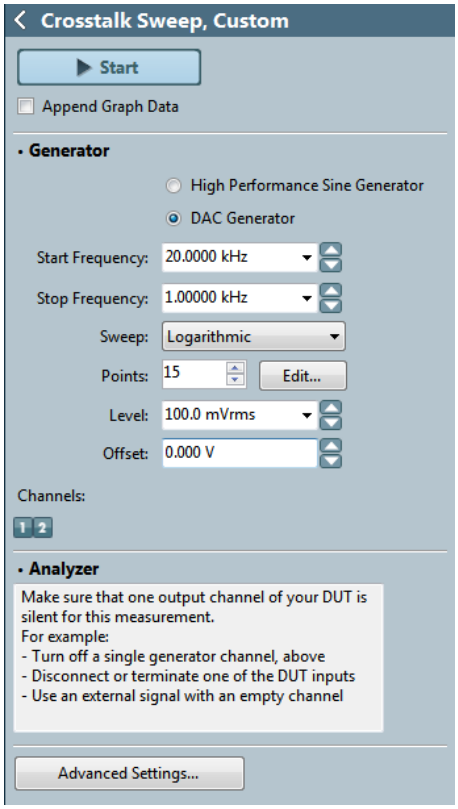
Running the Measurement

The generator outputs the test signal on all channels. You must choose a channel of interest to examine, and remove the generator signal from that channel. The crosstalk (signal leakage) into the channel of interest from all of the remaining channels is measured.

Before you run the sweep, you must prevent the signal from stimulating the channel of interest, as discussed next.

Remove the generator signal from the channel of interest using one of the following methods:

1. **Turn OFF a generator channel.**
   For many DUTs or systems under test, simply turning a channel of interest **OFF** at the generator will provide the most useful **Crosstalk Seep, Custom** results. Use one of the generator channel selection buttons.



2. **Interrupt a channel in the DUT or system under test.**
   For some DUTs or systems, interrupting the signal (by switching or disconnection) for the channel of interest may provide more useful results. In the special case of a DUT that has fewer inputs than outputs and where an input is routed within the DUT to more than one output, interrupting the signal within the DUT is the only way to measure crosstalk across some channel combinations.

You can select another channel of interest without turning off the generator. First, reconnect the current

channel of interest, then disconnect the new channel of interest. Read the updated results immediately.

For a default sweep of 20 kHz to 1 kHz in 15 logarithmic steps, simply click **Start**. The generator will output the sweep signal to the DUT on all available channels. Next, you must remove the generator signal from the channel of interest (as described above). Read the crosstalk measured in each the remaining channel from the graph display.

Append Graph Data

Measured data are grouped in a **Data Set**, which contains all acquisition results. Normally, the current graph data are deleted each time you start a new measurement. If the **Append Graph Data** box is checked, appended data are grouped as additional data sets, as are any imported data. The Data Sets Panel displays the Data Set(s) in the measurement, and provides a visibility checkbox, a Time acquisition field, and a Notes field for each data set.

Read more about Data Sets beginning on page 649.

Generator

The configuration for this measurement must be the closed-loop configuration, using the APx generator as a stimulus.

See Chapter 6 for detailed information about the using the APx Generator and making Waveform, Level and Frequency settings.

High Performance Sine Generator / DAC Generator

For an APx555, you have the option of using the High Performance Sine Generator (very low distortion sine waves and wide bandwidth) or the DAC generator, which provides special sine waveforms. See High Performance Sine Generator starting on page 51 for a comparison of features.

Start

Set the sweep **Start Frequency** here.

Stop

Set the sweep **Stop Frequency** here.

Sweep

Choose

- **Logarithmic** point spacing
- **Linear** point spacing
- **Custom**
  Edit sweep spacing and number of **Points** to create a Custom sweep.

Points

Set the number of sweep points here.

Points / Edit

Open the **Sweep Points** dialog to edit import or export the generator **Sweep Points** table. See page 507.

Step Size

For linear sweeps, enter the **Step Size** here.

Level

Set the signal level here. This level may be changed if generator equalization is used, and can be adjusted on a per-channel basis in **Advanced Settings** (see page 507).

Channels

Toggle specific output channels on or off by clicking on the Channel number button.

Analyzer

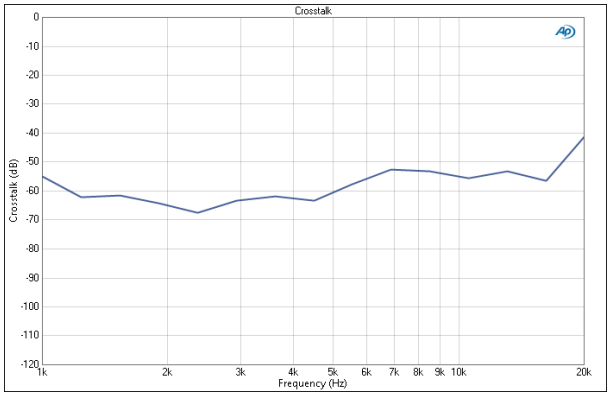
No local low pass, high pass or weighting filters are available for this measurement. However, low pass, high pass and weighting filter settings made in Signal Path Setup > Input/Output will affect this measurement. Read more about APx Filters in Chapter 97.

Advanced Settings

If your test requires special adjustments or settings, click **Advanced Settings**. See page 508.

See Chapter 104 for more information about units of measurement.

Crosstalk result



This result shows the crosstalk introduced into Channel 1 from Channel 2, with both channels driven but the Channel 1 connection between the generator and the DUT disconnected.

Units

Units available for **Crosstalk Sweep, One Channel Undriven: Crosstalk** are:

X-axis	*Y-axis*
• Hz	• x/y
• dHz	• %
• F/R	• ppm
• %Hz	• dB

Crosstalk Sweep, One Channel Driven
(Sequence Mode)

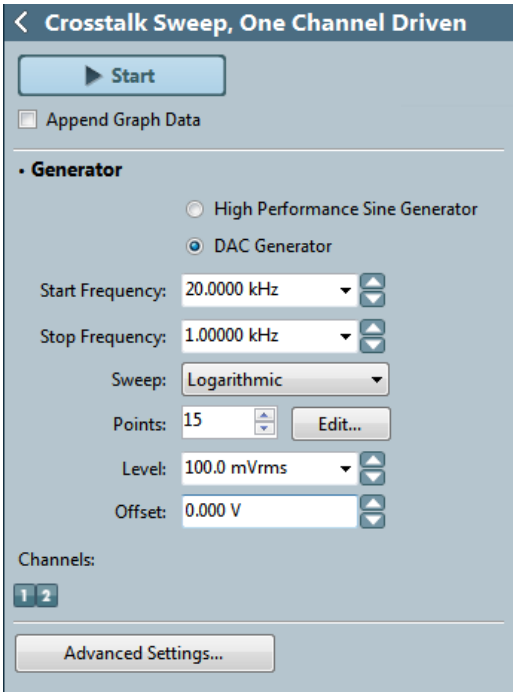
Introduction

For the APx515, this measurement requires a software option. See page 189 for more information about software options.

The **Crosstalk, One Channel Driven** performs a series of stepped frequency sweeps where only one channel is driven at a time. Each sweep applies the stimulus signal to one channel only (initially, Channel 1). Each undriven channel is measured for crosstalk. Then the sweep is run again, with the next channel (Channel 2) driven. This cycle is repeated through the available output channels. The level measured at each frequency point in the undriven channel(s) is compared to the level measured in the driven channel. The ratio of the two is the crosstalk result for that channel. X axis is generator frequency; Y axis is crosstalk ratio.

Read more about crosstalk beginning on page 264.

The Crosstalk Sweep, One Channel Driven measurement is not available in External Source or in File Input configurations.



Operation

If you have not yet set up your test, first go to Signal Path Setup (Chapter 7).

Running the Measurement

For a default sweep of 20 kHz to 1 kHz in 15 logarithmic steps, simply click **Start**. The generator will begin a series of sweeps, driving first Channel 1, then the next channel until the last available channel has been driven. If you want to specify other parameters for your sweep, review **Generator** and **Analyzer** settings, discussed below. Click **Start** when you have approved the settings.

Append Graph Data

Measured data are grouped in a **Data Set**, which contains all acquisition results. Normally, the current graph data are deleted each time you start a new measurement. If the **Append Graph Data** box is checked, appended data are grouped as additional data sets, as are any imported data. The Data Sets Panel displays the Data Set(s) in the measurement, and provides a visibility checkbox, a Time acquisition field, and a Notes field for each data set.

Read more about Data Sets beginning on page 649.

Generator

The configuration for this measurement must be the closed-loop configuration, using the APx generator as a stimulus.

See Chapter 6 for detailed information about the using the APx Generator and making Waveform, Level and Frequency settings.

High Performance Sine Generator / DAC Generator

For an APx555, you have the option of using the High Performance Sine Generator (very low distortion sine waves and wide bandwidth) or the DAC generator, which provides special sine waveforms. See High Performance Sine Generator starting on page 51 for a comparison of features.

Start

Set the sweep **Start Frequency** here.

Stop

Set the sweep **Stop Frequency** here.

Sweep

Choose

- **Logarithmic** point spacing
- **Linear** point spacing
- **Custom**
  Edit sweep spacing and number of **Points** to create a Custom sweep.

Points

Set the number of sweep points here.

Points / Edit

Open the **Sweep Points** dialog to edit import or export the generator **Sweep Points** table. See page 507.

Step Size

For linear sweeps, enter the **Step Size** here.

Level

Set the signal level here. This level may be changed if generator equalization is used, and can be adjusted on a per-channel basis in **Advanced Settings** (see page 507).

Channels

Toggle specific output channels on or off by clicking on the Channel number button.

Analyzer

No local low pass, high pass or weighting filters are available for this measurement. However, low pass, high pass and weighting filter settings made in Signal Path Setup > Input/Output will affect this measurement. Read more about APx Filters in Chapter 97.

Source

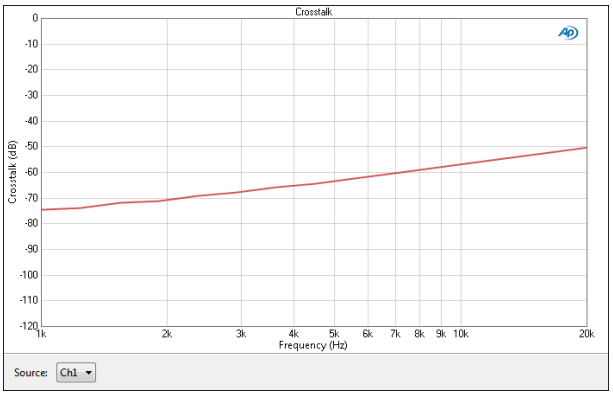
Set the **Source** control beneath the graph in the result panel to view the crosstalk signal(s) introduced into the non-source channel(s).

Advanced Settings

If your test requires special adjustments or settings, click **Advanced Settings**. See page 508.

See Chapter 104 for more information about units of measurement.

Crosstalk result



This result shows the crosstalk introduced into Channel 2, while Channel 1 is set as the Source.

Units

Units available for **Crosstalk Sweep, One Channel Driven: Crosstalk ratio** are:

X-axis	*Y-axis*
• Hz	• x/y
• dHz	• %
• F/R	• ppm
• %Hz	• dB

Crosstalk Sweep, One Channel Undriven
(Sequence Mode)

Introduction

For the APx515, this measurement requires a software option. See page 189 for more information about software options.

The **Crosstalk, One Channel Undriven** measurement performs a series of stepped frequency sweeps where all channels but one are driven. Each sweep applies the stimulus signal to all channels except the undriven channel (initially, Channel 1). The undriven channel is measured for crosstalk. Then the sweep is run again, with the next channel (Channel 2) undriven. This cycle is repeated through the available output channels. The level measured at each frequency point in the undriven channel(s) is compared to the level measured in a driven channel. The ratio of the two is the crosstalk result for that channel. X axis is generator frequency; Y axis is crosstalk ratio.

Read more about crosstalk beginning on page 264.

The Crosstalk Sweep, One Channel Driven measurement is not available in External Source or in File Input configurations.

Operation

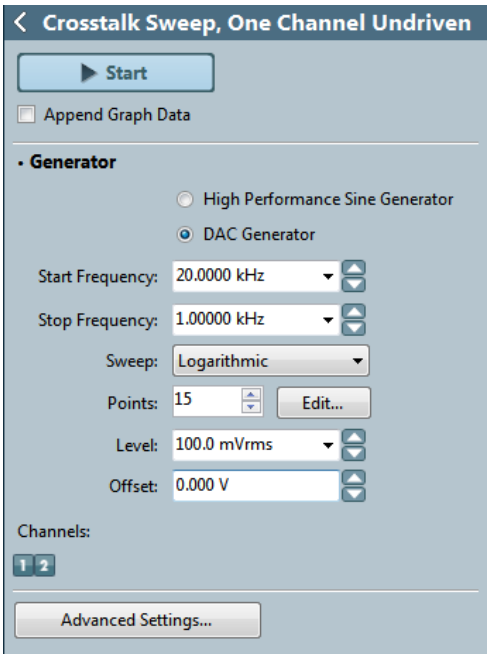
If you have not yet set up your test, first go to Signal Path Setup (Chapter 7).

Running the Measurement

For a default sweep of 20 kHz to 1 kHz in 15 logarithmic steps, simply click **Start**. The generator will begin a series of sweeps, driving first Channel 1, then the next channel until the last available channel has been driven. If you want to specify other parameters for your sweep, review **Generator** and **Analyzer** settings, discussed below. Click **Start** when you have approved the settings.

Append Graph Data

Measured data are grouped in a **Data Set**, which contains all acquisition results. Normally, the current



graph data are deleted each time you start a new measurement. If the **Append Graph Data** box is checked, appended data are grouped as additional data sets, as are any imported data. The Data Sets Panel displays the Data Set(s) in the measurement, and provides a visibility checkbox, a Time acquisition field, and a Notes field for each data set.

Read more about Data Sets beginning on page 649.

Generator

The configuration for this measurement must be the closed-loop configuration, using the APx generator as a stimulus.

See Chapter 6 for detailed information about the using the APx Generator and making Waveform, Level and Frequency settings.

High Performance Sine Generator / DAC Generator

For an APx555, you have the option of using the High Performance Sine Generator (very low distortion sine waves and wide bandwidth) or the DAC generator, which provides special sine waveforms. See High Performance Sine Generator starting on page 51 for a comparison of features.

Start

Set the sweep **Start Frequency** here.

Stop

Set the sweep **Stop Frequency** here.

Sweep

Choose

- **Logarithmic** point spacing
- **Linear** point spacing
- **Custom**
  Edit sweep spacing and number of **Points** to create a Custom sweep.

Points

Set the number of sweep points here.

Points / Edit

Open the **Sweep Points** dialog to edit import or export the generator **Sweep Points** table. See page 507.

Step Size

For linear sweeps, enter the **Step Size** here.

Level

Set the signal level here. This level may be changed if generator equalization is used, and can be adjusted on a per-channel basis in **Advanced Settings** (see page 507).

Channels

Toggle specific output channels on or off by clicking on the Channel number button.

Analyzer

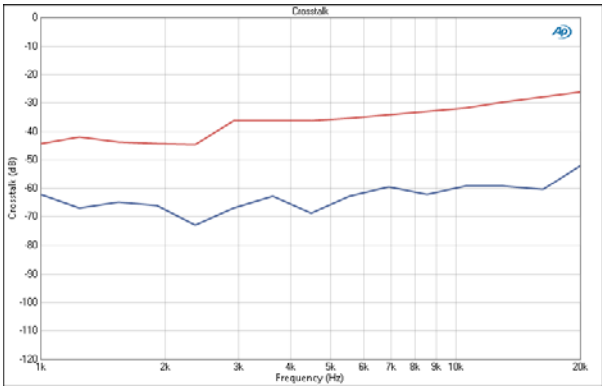
No local low pass, high pass or weighting filters are available for this measurement. However, low pass, high pass and weighting filter settings made in Signal Path Setup > Input/Output will affect this measurement. Read more about APx Filters in Chapter 97.

Advanced Settings

If your test requires special adjustments or settings, click **Advanced Settings**. See page 508.

See Chapter 104 for more information about units of measurement.

Crosstalk result



This result shows the crosstalk introduced into Channel 1 from Channel 2, and the crosstalk introduced into Channel 2 from Channel 1.

Units

Units available for **Crosstalk Sweep, One Channel Undriven: Crosstalk** are:

X-axis	*Y-axis*
• Hz	• x/y
• dHz	• %
• F/R	• ppm
• %Hz	• dB

Crosstalk, Custom
(Sequence Mode)

Crosstalk is unwanted leakage or bleed of a signal from one or more channels to other channels within a device.

The Crosstalk, Custom measurement result provides a measurement of the crosstalk into one undriven DUT channel, while the other channels are driven. For this measurement, the user must manually turn OFF or interrupt the signal to the undriven channel.

More Information

For more information about crosstalk, see page 264. Also see the Crosstalk, One Channel Drive measurement in Chapter 36 and Crosstalk, One Channel Undriven measurement in Chapter 37.

Operation

If you have not yet set up your test, first go to Signal Path Setup to select and configure inputs and outputs, references and other settings. See Chapter 7 for information about Signal Path Setup.

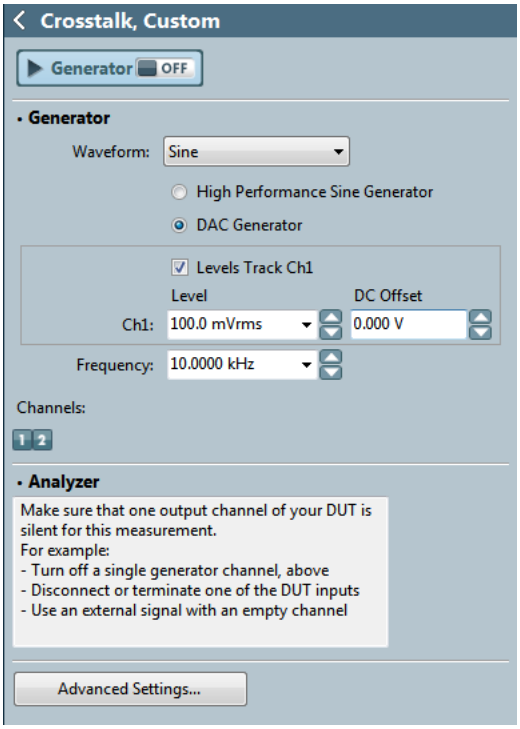
Generator

The typical configuration for this measurement is the closed-loop configuration, using the APx generator as a stimulus. Alternatively, you can make open-loop measurements, using either the External Source or File Input configuration.

See Chapter 6 for detailed information about the using the APx Generator and making Waveform, Level and Frequency settings, or for information about using External Source.

Be sure that external sources or input/generator files correspond with the Crosstalk “Driven Channel” setting.

By default, the Crosstalk, Custom measurement uses a sine wave at the frequency and level set in the Signal Generation panel as the test signal. Since crosstalk typically increases with frequency, 10 kHz has been selected as the default frequency.



Alternatively, you can select a generator waveform file as the test signal, using the file browser in the Waveform drop-down menu. See Chapter 14 for more information about using Generator Waveform files.

The generator outputs the test signal on all channels. You must choose a channel of interest to examine, and remove the generator signal from that channel. The crosstalk (signal leakage) into the channel of interest from all of the remaining channels is measured.

To measure Crosstalk, Custom, turn the Generator **On**. The generator will output the test signal to the DUT on all channels. Next, you must remove the generator signal form the channel of interest (see below). Read the crosstalk measured in each remaining channel from the meter bar display.

Removing the generator signal from the channel of interest

Remove the generator signal from the channel of interest using one of the following methods:

1. **Turn OFF a generator channel.**
   For many DUTs or systems under test, simply turning a channel of interest OFF at the generator will provide the most useful Crosstalk, Custom results. Use one of the generator channel selection buttons. When in External Source mode, playing a waveform with the channel of interest silent will provide the same results.
2. **Interrupt a channel in the DUT or system under test.**
   For some DUTs or systems, interrupting the signal (by switching or disconnection) for the channel of interest may provide more useful results. In the special case of a DUT that has fewer inputs than outputs and where an input is routed within the DUT to more than one output, interrupting the signal within the DUT is the only way to measure crosstalk across some channel combinations.

You can select another channel of interest without turning off the generator. First, reconnect the current channel of interest, then disconnect the new channel of interest. Read the updated results immediately.

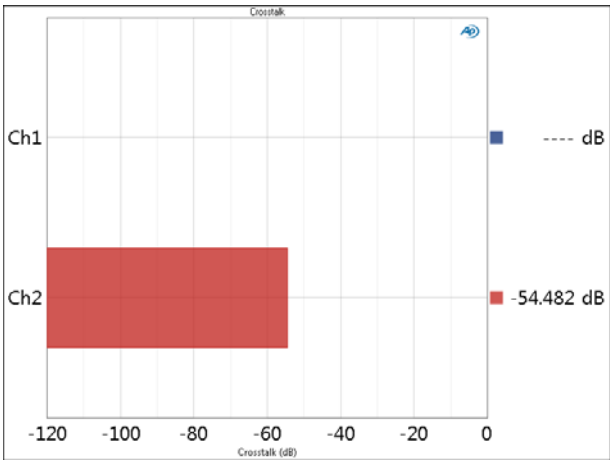
High Performance Sine Generator / DAC Generator

For an APx555, you have the option of using the High Performance Sine Generator (very low distortion sine waves and wide bandwidth) or the DAC generator, which provides special sine waveforms. See High Performance Sine Generator starting on page 51 for a comparison of features.

Advanced Settings

If your test requires individual generator settings for each channel or special analyzer ranging or settling, click Advanced Settings. See Advanced Settings on page 351.

Crosstalk



The Crosstalk measurement result provides a single-value measurement crosstalk in the DUT. In this case, the Channel 2 drive signal has been interrupted in the DUT, so crosstalk results only appear in Channel 2.

Crosstalk is usually expressed as a ratio, with the measured crosstalk in the unstimulated channel divided by the full signal level measured in a stimulated channel. The result is typically stated in decibels or as a percentage.

Units

Crosstalk is usually expressed as a ratio, with the measured crosstalk in the unstimulated channel divided by the full signal level measured in a stimulated channel. The result is typically stated in decibels or as a percentage.

Units available for crosstalk are

- x/y
- %
- ppm
- dB

See Chapter 104 for more information about units of measurement.

Crosstalk, One Channel Driven
(Sequence Mode)

Crosstalk is unwanted leakage or bleed of a signal from one or more channels to other channels within a device.

The Crosstalk, One Channel Driven measurement result provides a measurement of the crosstalk into the unstimulated DUT channel(s), when one channel is stimulated.

More Information

For more information about crosstalk, see page 264. Also see the Crosstalk, Custom measurement in Chapter 35 and Crosstalk, One Channel Undriven measurement in Chapter 37.

Operation

If you have not yet set up your test, first go to Signal Path Setup to select and configure inputs and outputs, references and other settings. See Chapter 7 for information about Signal Path Setup.

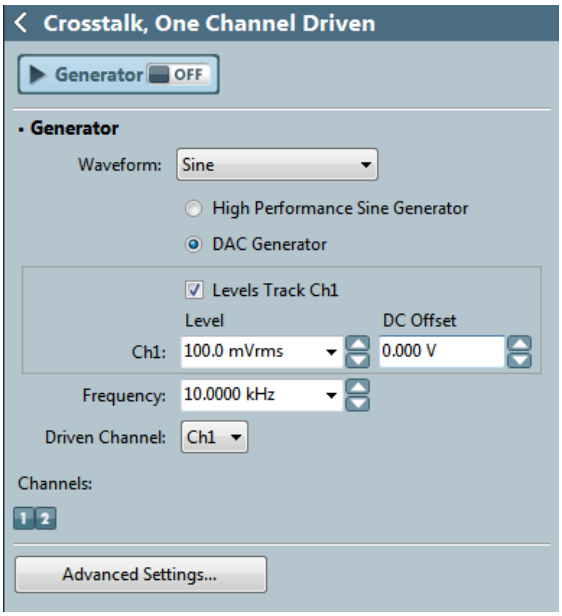
Generator

The typical configuration for this measurement is the closed-loop configuration, using the APx generator as a stimulus. Alternatively, you can make open-loop measurements, using either the External Source or File Input configuration.

See Chapter 6 for detailed information about the using the APx Generator and making Waveform, Level and Frequency settings, or for information about using External Source.

Be sure that external sources or input/generator files correspond with the Crosstalk “Driven Channel” setting.

By default, the Crosstalk, One Channel Driven measurement uses a sine wave at the frequency and level set in the Signal Generation panel as the test signal. Since crosstalk typically increases with frequency, 10 kHz has been selected as the default frequency.



The generator outputs the test signal on one channel only, called the “Driven Channel.” The crosstalk (signal leakage) in each of the remaining channels is measured.

To measure Crosstalk, One Channel Driven, turn the Generator **ON**. The generator will output the test signal to the DUT on the selected Driven Channel.

Since the Crosstalk, One Channel Driven measurement relies on your choice of a Driven Channel, as described above, deselecting other generator channels can produce ambiguous results. We recommend that for Crosstalk measurements all the generator channel selection buttons remain set to **ON**, the default condition.

High Performance Sine Generator / DAC Generator

For an APx555, you have the option of using the High Performance Sine Generator (very low distortion sine waves and wide bandwidth) or the DAC generator,

which provides special sine waveforms. See High Performance Sine Generator starting on page 51 for a comparison of features.

Analyzer

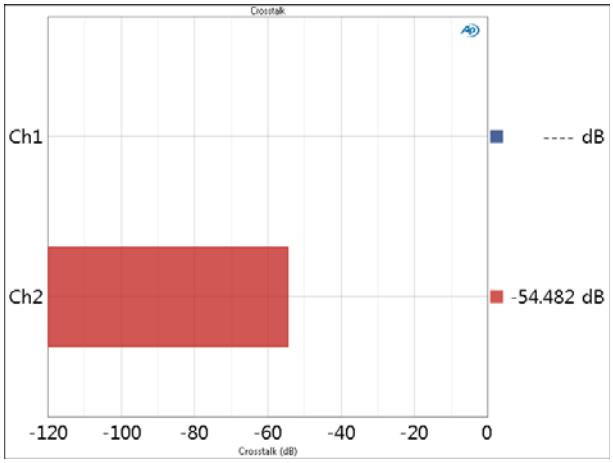
The meter bar display excludes data from the Driven Channel; read the crosstalk measured in each remaining channel from the active meter bars.

You can select another channel as the Driven Channel with the Generator **ON**, and read the updated results immediately.

Advanced Settings

If your test requires individual generator settings for each channel or special analyzer ranging or settling, click Advanced Settings. See Advanced Settings on page 351.

Crosstalk



The Crosstalk measurement result provides a single-value measurement crosstalk in the DUT. In this case, the Driven Channel is Channel 1, so crosstalk results only appear in Channel 2.

Crosstalk is usually expressed as a ratio, with the measured crosstalk in the unstimulated channel divided by the full signal level measured in a stimulated channel. The result is typically stated in decibels or as a percentage.

Units

Units available for crosstalk are

- x/y
- %
- ppm
- dB

See Chapter 104 for more information about units of measurement.

Crosstalk, One Channel Undriven
(Sequence Mode)

Crosstalk is unwanted leakage or bleed of a signal from one or more channels to other channels within a device.

The Crosstalk, One Channel Undriven measurement result provides a measurement of the crosstalk into one unstimulated DUT channel (the *undriven* channel), while the other channels are stimulated. This Crosstalk measurement automatically steps the measurement through the channels.

Note: The Crosstalk, One Channel Undriven measurement is not available in External Source configuration.

More Information

For more information about crosstalk, see page 264. Also see the Crosstalk, Custom measurement in Chapter 35 and Crosstalk, One Channel Driven measurement in Chapter 36.

Operation

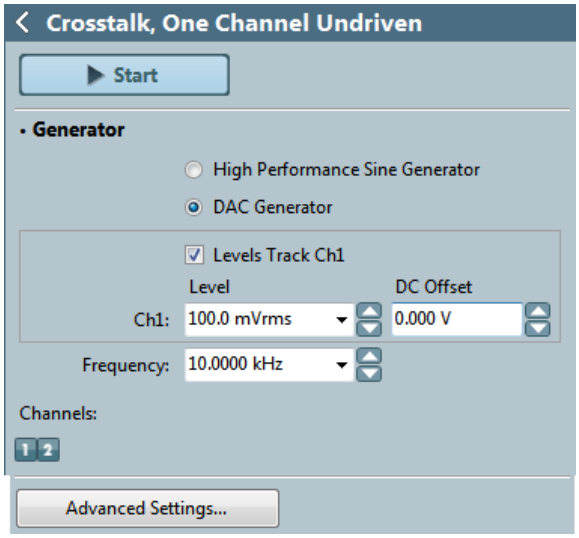
If you have not yet set up your test, first go to Signal Path Setup to select and configure inputs and outputs, references and other settings. See Chapter 7 for information about Signal Path Setup.

Generator

This measurement must be made in the closed-loop configuration with balanced analog outputs, using the APx generator as a stimulus.

See Chapter 6 for detailed information about the using the APx Generator and making Waveform, Level and Frequency settings.

The Crosstalk, One Channel Undriven measurement uses a sine wave at the frequency and level set in the Signal Generation panel as the test signal. Since crosstalk typically increases with frequency, 10 kHz has been selected as the default frequency.



To run Crosstalk, One Channel Undriven, click **Start**. The generator will output the test signal to the DUT on all channels except Channel 1. After the first result is displayed, the measurement runs again with Channel 1 turned **ON**, and Channel 2 **OFF**. This process repeats through all of the available channels. Read the crosstalk measured in each remaining channel from the meter bar display.

Since the Crosstalk, One Channel Undriven measurement relies on stimulating one channel at a time as described above, deselecting generator channel(s) will produce ambiguous results. We recommend that for Crosstalk, One Channel Undriven measurements all generator channel selection buttons remain set to **ON**, the default condition.

High Performance Sine Generator / DAC Generator

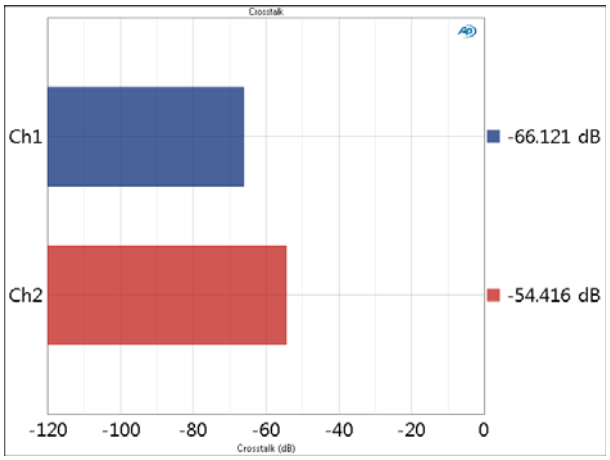
For an APx555, you have the option of using the High Performance Sine Generator (very low distortion sine waves and wide bandwidth) or the DAC generator, which provides special sine waveforms. See High Per-

formance Sine Generator starting on page 51 for a comparison of features.

Advanced Settings

If your test requires individual generator settings for each channel or special analyzer ranging or settling, click Advanced Settings. See Advanced Settings on page 351.

Crosstalk



Units

Crosstalk is usually expressed as a ratio, with the measured crosstalk in the unstimulated channel divided by the full signal level measured in a stimulated channel. The result is typically stated in decibels or as a percentage.

Units available for Crosstalk, One Channel Undriven are

- x/y
- %
- ppm
- dB

See Chapter 104 for more information about units of measurement.

More about Crosstalk

Crosstalk is expressed as a ratio. The measured crosstalk level in a channel is divided by the measured level of the stimulus tone in the source channel. Crosstalk is typically stated in decibels (dB).

Crosstalk should normally be measured as a function of frequency, since the values are likely to vary strongly with frequency. The most common circuit mechanisms causing crosstalk are stray capacitive and inductive coupling. If circuit impedances are approximately constant with frequency, crosstalk caused by a single instance of capacitive coupling will increase with increasing frequency at a 6 dB per octave rate. That is, crosstalk due to single pole capacitive coupling will be 6 dB worse at 4 kHz than at 2 kHz, etc. Crosstalk can also be caused by inductive coupling, shared power supplies, shared ground returns, etc., so the relationship is often not the simple capacitive coupling model.

For traditional techniques, crosstalk is measured selectively. That is, a narrow bandpass filter in the analyzer is tuned to the generator frequency, in order to measure crosstalk at or below the wideband noise level. This is not merely an academic concern; the human ear is able to distinguish coherent signals such as sine waves even when the signal amplitude is 10 dB to 20 dB below the wideband noise level.

Crosstalk from “Channel X” to “Channel Y” is often not identical to crosstalk in the other direction, from “Channel Y” into “Channel X.” Circuit layout and complex stray coupling mechanisms often result in somewhat different values for crosstalk in the two directions. Good values for crosstalk in electronic devices range from –50 dB and below, with crosstalk in better than –100 dB achievable by sufficient isolation. When the application is stereo or surround sound, such values are far higher than really necessary for a full effect. However, if the two devices are carrying independent program material, values of crosstalk of –60 dB to –70 dB and more are desirable.

DC Level
(Sequence Mode)

The DC Level view provides a single-value result, the DC voltage present at the output of each DUT channel.

Input coupling is forced to DC coupling during this measurement.

Operation

If you have not yet set up your test, first go to Signal Path Setup to select and configure inputs and outputs, references and other settings. See Chapter 7 for information about Signal Path Setup.

Running the measurement with no stimulus signal

No stimulus is required to measure DC Level. Simply open the measurement and read the DC Level present at the analyzer inputs.

Generator

Running the measurement with a stimulus signal

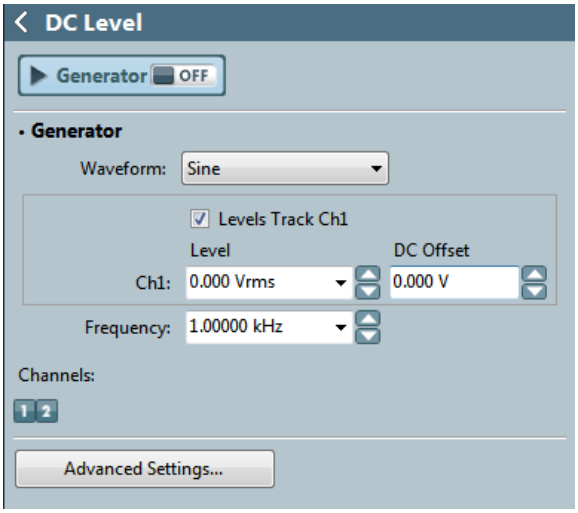
However, you can measure DC Level in the presence of signal, if desired. To measure DC Level in the presence of signal, click the Generator button to **On**. The generator will output the test signal to the DUT on the selected generator channels. Read the DC Level of each channel from the meter bar display.

Selecting the generator waveform

By default, the DC Level measurement uses a sine wave at the frequency and level set in the Signal Generation panel as the test signal. You can add a DC offset to the signal.

Alternatively, you can select a generator waveform file as the test signal, using the file browser in the Waveform drop-down menu.

See Chapter 6 for detailed information about the using the APx Generator and making Waveform, Level and Frequency settings, or for information about using External Source.

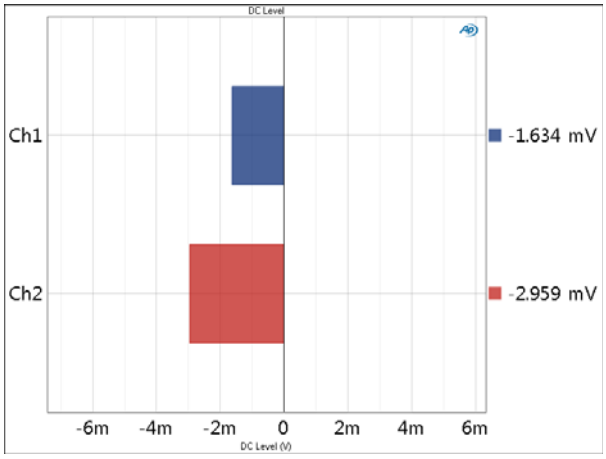


Advanced Settings

Advanced Settings makes available individual channel output level and offset controls, and input ranging and settling controls. See Advanced Settings for DC Level measurements later in this chapter.

See Chapter 104 for more information about units of measurement.

DC Level



The DC Level measurement provides a single-value result showing the DC voltage present at the output of each DUT channel.

The analog measurement range for DC Level depends upon the instrument model; for an APx515 it is from approximately ±10 µV to ±120 V. For an APx525 family analyzer it is from approximately ±10 µV to ±160 V. For an APx582, 585 or 586 it is from approximately ±10 µV to ±115 V.

Units

Analog	*Digital*
• V	• D • hex

See Chapter 104 for more information about units of measurement.

More about DC in APx Analyzers

APx can measure DC Level using these measurements:

- DC Level
- Multitone: DC Level
- Signal Analyzer: FFT Spectrum
- Monitor: FFT Spectrum

DC offset

Unless blocked by a capacitor or a transformer-coupled output, analog audio devices will have some (usually very small) dc component in their output. Typical dc levels (often called dc offset) are in the range of –20 mV to +20 mV.

The embedded audio in a digital signal can have a dc component as well. This is usually very low in level, and may vary with the applied dither.

The Constant Value digital signal is “digital dc” of an arbitrary value.

APx can generate sine signals with an arbitrary dc offset, if desired. This facilitates “dc in the presence of signal” measurements.

DC in the presence of signal measurements

APx can generate stimulus signals with AC and DC components, and can measure the dc component when it is an offset in an ac signal.

Coupling: AC or DC

Input signals can be analyzed and displayed in one of two coupling modes:

- **AC**
  AC coupling blocks any dc signal component from analysis, displaying results for only the ac components. As a consequence, AC coupling introduces a rolloff below 10 Hz. AC coupling is the default setting. For the DC Level measurement only, DC coupling is forced.

Note: Some measurements may require the insertion of a small delay before analysis, to allow for blocking filter settling after a transient (such as generator start).

- **DC**
  DC coupling includes the dc signal component in analysis and display.

For the DC Level measurement, the APx system is DC coupled regardless of user settings.

Advanced Settings for DC Level measurements

The default settings here are appropriate for most measurements. You may want to make minor adjustments for special situations.

Signal Generation

You can set different output levels or DC offsets for each channel here.

If **Levels Track Ch1** is checked (the default), the generator output level and DC offset values for channel 1 are copied to the other channels, and values for channels greater than 1 cannot be edited. Any changes made to channel 1 are reproduced in the other channels.

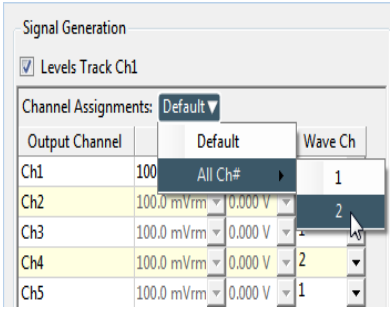
To set individual generator output channel levels or offsets, uncheck the **Levels Track Ch1** checkbox and enter values in the output channel Level or Offset fields.

Set Channel Assignments for special waveforms

For special waveforms like Sine Dual, you can specify the channel assignment.

Set Channel Assignments for generator waveforms

For stereo or multichannel generator waveform files, you can set channel mapping. By default, channel 1 audio is mapped to channel 1 output, channel 2 to channel 2, and so on. If the number of channels in the waveform file is less than the number of output channels, the waveform channels resume numbering at 1 and wrap to the next available output channel. These assignments can be remapped by changing individual settings in the **Wave Ch** column, or by selection one of several presets from the **Set Channel Assignments menu.**



Signal Acquisition and Analysis

Input Range tab

Digital inputs do not require ranging, and the Input Range tab is unavailable when inputs are set to a digital format.

Auto Range

If **Auto Range** is checked (the default), each analog channel determines its input range automatically, based on the level of the input signal. If the input signal level changes beyond a ranging threshold, **Auto Range** will cause the input ranging circuits to move up or down for proper ranging.

Fixed Range

Alternatively, you can set a fixed minimum range for the analog inputs. Uncheck **Auto Range** and enter the expected maximum level in the **Minimum Range** filed. The software will choose the lowest range that will accommodate that level; this becomes the minimum range available, and the analyzer will not reset to a lower range. Fixed range is best for signals that vary rapidly, including tone bursts, perceptual audio speech samples, and program material such as voice or music. See **When autoranging isn't suitable** on page 626.

If a signal greater then the **Minimum Range** is presented to the analog input, the ranging circuits will range up to accommodate the higher level, and then drop again when the signal level falls. The range will not be set below the **Minimum Range**.

Ranges Track Ch1

If **Ranges Track Ch1** is checked (the default), the fixed input range setting for channel 1 is copied to the other channels, and range settings for channels greater than 1 cannot be edited. Any changes made to channel 1 are reproduced in the other channels. If **Ranges Track Ch1** is not checked, the fixed input range setting for each channel can be set independently.

Read more about ranging in Chapter 98.

Delay Time

DC Level results can vary with noise-like characteristics. When a noise measurement (or a DC Level measurement) is run from the navigator or as part of a sequence, the acquisition is delayed for the time set here. This pause can avoid signal disruptions that may occur in the DUT when changing measurements. The **Delay Time** range is 0.0 s to 10.0 s; the default setting is 300.0 ms.

Acquisition Time:

DC Level results can vary with noise-like characteristics. Since noise is not periodic, a very short acquisition may display a result that is well above or below the average noise level. In a noise measurement the signal is acquired over a period of time between 100 ms and 10 s, set in Acquisition Time. Longer acquisitions provide more repeatable results. The default acquisition time is 250 ms.

DC Level (DCX)
(Sequence Mode)

DCX measurements require that a DCX-127 Multifunction Module be connected to the APx analyzer system, typically using an Audio Precision USB/APIB adapter connected to the PC.

The DCX-127 brings additional measurement and control features to an Audio Precision analyzer. The APx DC Level (DCX) measurement uses the DCX DMM (Digital Multimeter) mode to implement a precise DC voltage meter. Download the DCX-127 Multifunction Module User's Guide from the Audio Precision Web site at ap.com for complete operational information for the DCX-127.

Operation

If you have not yet set up your test, first go to Signal Path Setup to select and configure inputs and outputs, references and other settings. See Chapter 7 for information about Signal Path Setup.

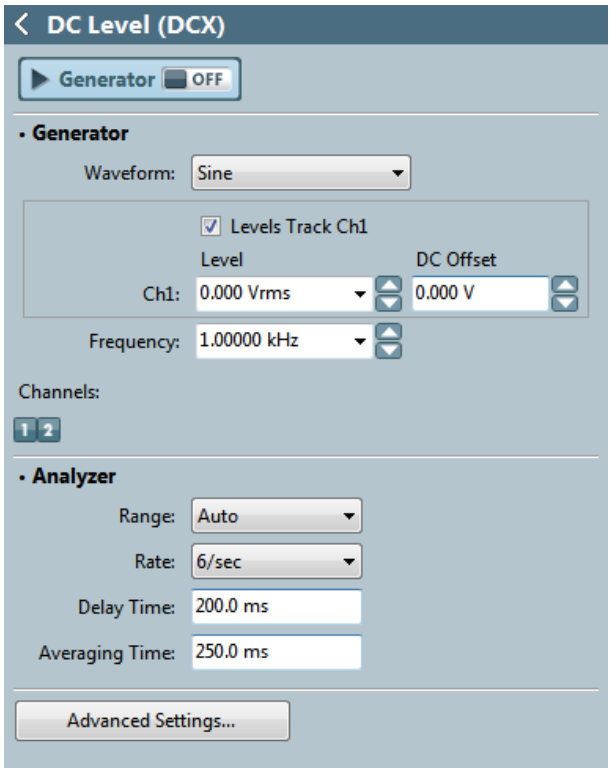
The DCX-127 must be powered on and connected to the measurement PC using an Audio Precision USB/APIB adapter.

Selecting the DC Level (DCX) measurement places the DCX in "DMM" mode. Note that you can configure the DCX DC Outputs or Auxiliary outputs in Signal Path Setup > DCX, independent of any settings or operation in the DC Level (DCX) measurement.

You can make a DC Level measurement with no stimulus signal, or you can use the generator to provide a stimulus to investigate how an audio signal affects the DC Level at the output of a DUT.

Running the measurement with no stimulus signal

No stimulus is required to measure DC Level. Simply open the measurement and read the DC Level present at the analyzer inputs.



Generator

Running the measurement with a stimulus signal

You can also run this measurement in the closed-loop configuration, using the APx generator as a stimulus. See Chapter 6 for detailed information about the using the APx Generator and making Waveform, Level and Frequency settings, or for information about using External Source.

Analyzer

The DCX has a digital multimeter (DMM) that is configured as DC voltmeter for this measurement. The APx analyzer is not used; all the data comes from the DCX.

Range

Select a fixed range for the DC Level measurement, or select **Auto**.

- Auto
- 500 V
- 200 V
- 20 V
- 2 V
- 200 mV

Rate

The Rate control allows you to select one of two meter reading rates.

- **6/sec**
  Approximately 6 meter readings are taken per second.
- **25/sec**
  Approximately 25 meter readings are taken per second.

At 6/sec, DCX meter resolution is a full 4 1/2 digits. At 25/sec, the resolution is reduced. The number of digits displayed is the same but the least significant digit is always 5 or 0.

Delay Time

A Delay Time interval is started when the DC Level (DCX) measurement is started within a sequence. APx ignores DCX data until the time set in Delay Time has passed. Minimum Delay Time is 200 ms, maximum is 10 s, 200 ms is the default.

Averaging Time

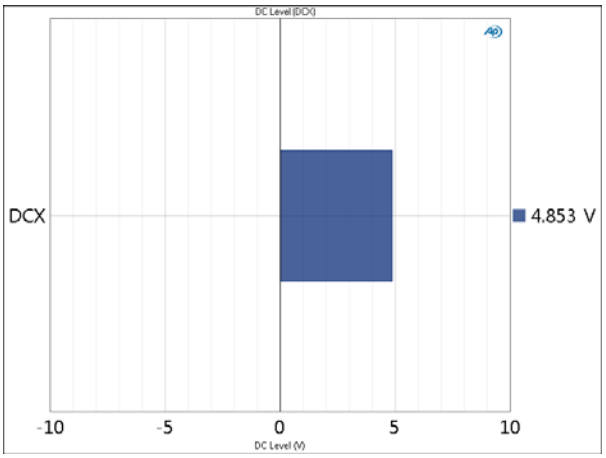
APx performs a rolling average of the DCX readings across a specified time before display. Minimum Averaging Time is 200 ms, maximum is 10 s, 250 ms is the default.

Advanced Settings

Advanced Settings makes available individual channel output level and offset controls, and input ranging and settling controls. See Advanced Settings for single value measurements on page 351.

See Chapter 104 for more information about units of measurement.

DC Level (DCX)



Units

Unit available for DC Level (DCX) is

- V

DC Level Sweep
(Sequence Mode)

For the APx515, this measurement requires a software option key. See page 189 for more information about software options.

The DC Level Sweep measurement uses a DC stimulus that is moved across a range of values in a series of points. The DUT output is acquired by the analyzer. DC Level and Noise results are plotted. The Noise result can be filtered.

The DC Level Sweep measurement is not available in External Source or File Input configurations.

Input coupling is forced to DC Coupling during this measurement.

DC Level Sweep results available in APx500 are:

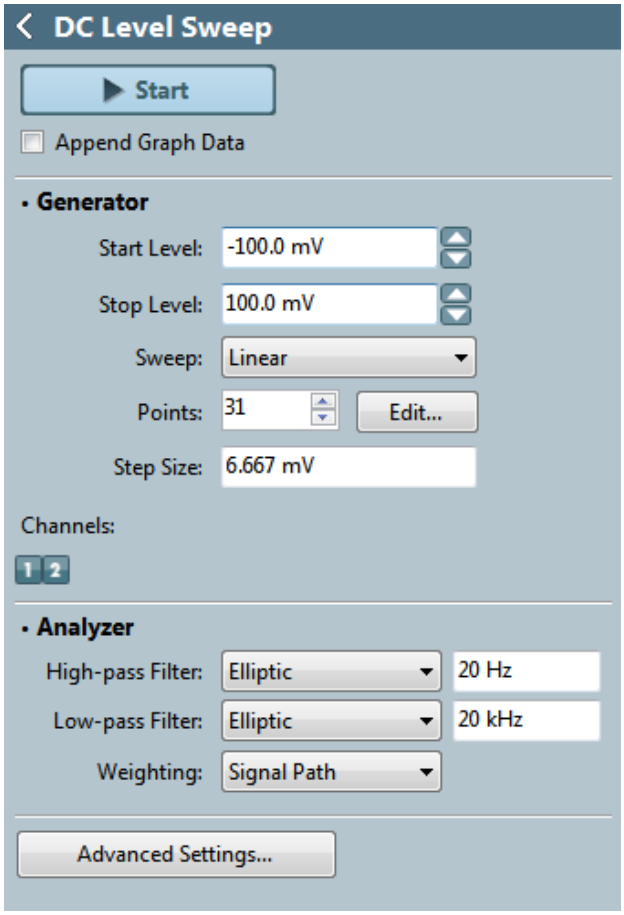
- DC Level
- Noise

Operation

If you have not yet set up your test, first go to Signal Path Setup to select and configure inputs and outputs, references and other settings. See Chapter 7 for information about Signal Path Setup.

The DC Level Sweep measurement does not use an audio signal for a stimulus, but instead sweeps a DC signal across a range. For analog outputs, this is a DC voltage; for digital outputs, it is digital DC (constant value).

To make a DC Level Sweep measurement, click **Start**. The generator will output the DC signal to the DUT on the selected generator channels. The signal will be swept from the **Start Level** to the **Stop Level** in the set number of points. You can select **Logarithmic** or **Linear** point spacing, or choose **Edit...** to customize the **Sweep Points** table.



Append Graph Data

Normally, the current graph data for all results is deleted each time you start a new measurement. If the **Append** box is checked, the current data is kept in memory as a **Data Set** (see page 189), and the new measurement data is appended as a new data set. You can append many **Data Sets**.

Generator

Start Level, Stop Level and Sweep Points

The sweep moves between two levels, set in the **Start Level** and **Stop Level** fields. The default levels are –100.0 mV and +100.0 mV.

The sweep moves in a specified number of step points, set in the **Points** field. The minimum is 2 points; maximum is 65,535. The default setting is 31.

The sweep point spacing is set by selecting one of the following choices in the Sweep field:

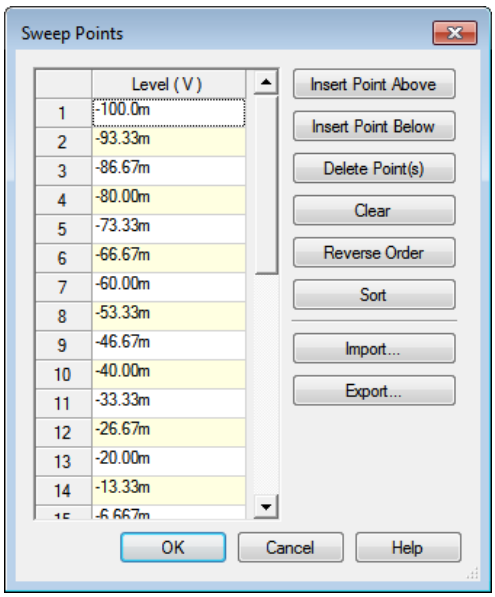
- **Logarithmic**; use the **Points** field to set the number of logarithmically spaced points;
- **Linear** (the default), which provides two methods of adjusting spacing: the **Points** field or the **Step Size** field; or
- **Custom**. Click **Edit** to open the **Sweep Points** dialog, where you can set points arbitrarily, or load or save sweep table files.

Viewing or Editing the Sweep Points table

You can view or edit the current generator sweep points at any time.

Click **Edit** to open the generator **Sweep Points** table. The table shows each sweep point and its corresponding level. You can edit this table to add or delete points, or to change the level of a point. Points can be sorted or reversed in order using the controls on the right.

A **Sweep Points** table can be saved as a *.csv file or as a Microsoft Excel *.xls file. A compatible *.csv or *.xls file can be opened and used as a **Sweep Points** table.



A Linear sweep can include a range that extends into negative values, but a Logarithmic sweep can only include positive values.

Each point is displayed as it is acquired. In a short period of time, the entire sweep is processed, and the small graphs in the Selector will be populated with results. Simply click the result view you want in the Navigator, or choose the graph thumbnail result view in the Selector.

Analyzer

Filters

Local high-pass, low-pass and weighting filters can be applied to the **Noise Level** result. The **DC Level** result is unfiltered.

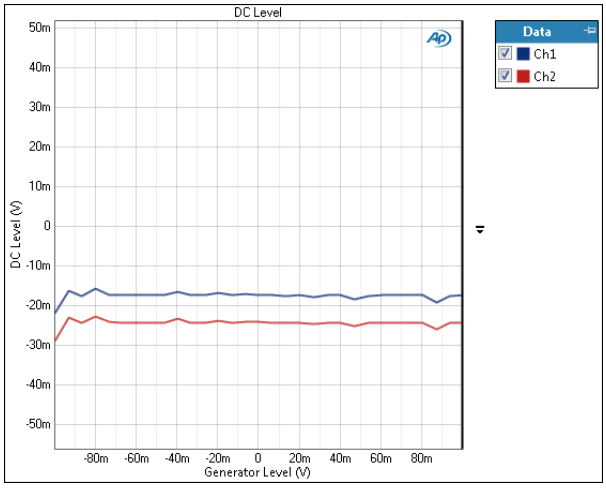
Noise measurements are typically made within a limited passband. The default setting for **Noise Level** measurements uses a 20 kHz low-pass filter and a 20 Hz high-pass filter. See Measurement Filters on page 619 for detailed information about the filters available locally for some measurements.

Low pass, high pass and weighting Signal Path Filters settings (see page 617) made in Signal Path Setup > Input/Output will also affect this measurement. When set, the local low-pass or high-pass filter setting overrides the corresponding Signal Path filter setting while the measurement is active. A local weighting filter setting does not override any Signal Path filter settings; the local weighting filter is applied in addition to any active Signal Path filters.

Advanced Settings

If you'd like to adjust analyzer ranging or settling parameters, click **Advanced Settings**. Read more about Advanced Settings for DC measurements on page 273.

DC Level Result



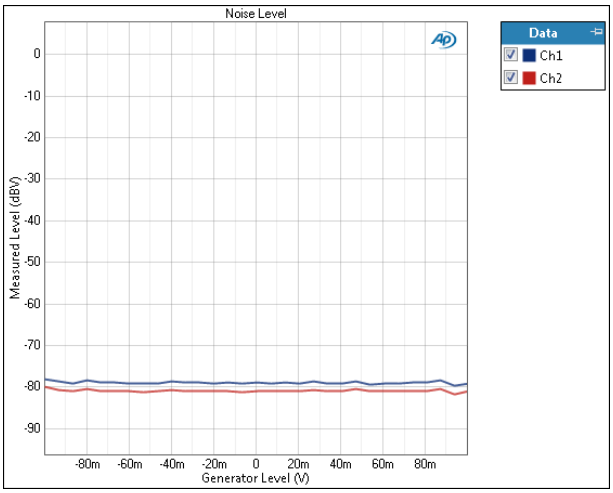
The DC Level Sweep measurement uses a DC stimulus that is moved across a range of values in a series of points. The DUT output is acquired by the analyzer. This view shows DC Level result.

Units

Units available for DC Level Sweep: DC Level are:

X-axis (analog signals)	*Y-axis (analog signals)*
• V	• V
X-axis (digital signals)	*Y-axis (digital signals)*
• D	• D
• hex	• hex

Noise Level Result



The DC Level Sweep measurement uses a DC stimulus that is moved across a range of values in a series of points. The DUT output is acquired by the analyzer. This view shows the Noise result. The Noise result can be filtered.

Units

Units available for DC Level Sweep: Noise Level are:

X-axis (analog signals)	*Y-axis (analog signals)*
• V	• Vrms • dBV • dBu • dBrA • dBrB • dBSPL1 • sBSPL2 • dBm • W (watts)
X-axis (digital signals)	*Y-axis (digital signals)*
• D • hex	• dBFS • FS • %FS • dBrA • dBrB • dBSPL1 • dBSPL2

Advanced Settings for DC Level Sweep measurements

The default settings here are appropriate for most measurements. You may want to make minor adjustments for special situations.

Signal Generation

Generator output levels cannot be modified for Stepped Level Sweeps, and the channel level controls are not available.

Signal Acquisition and Analysis

Input Range tab

Digital inputs do not require ranging, and the Input Range tab is unavailable when inputs are set to a digital format.

Auto Range

If **Auto Range** is checked (the default), each analog channel determines its input range automatically, based on the level of the input signal. If the input signal level changes beyond a ranging threshold, **Auto Range** will cause the input ranging circuits to move up or down for proper ranging.

Fixed Range

Alternatively, you can set a fixed minimum range for the analog inputs. Uncheck **Auto Range** and enter the expected maximum level in the **Minimum Range** filed. The software will choose the lowest range that will accommodate that level; this becomes the minimum range available, and the analyzer will not reset to a

lower range. Fixed range is best for signals that vary rapidly, including tone bursts, perceptual audio speech samples, and program material such as voice or music. See **When autoranging isn't suitable** on page 626.

If a signal greater then the **Minimum Range** is presented to the analog input, the ranging circuits will range up to accommodate the higher level, and then drop again when the signal level falls. The range will not be set below the **Minimum Range**.

Ranges Track Ch1

If **Ranges Track Ch1** is checked (the default), the fixed input range setting for channel 1 is copied to the other channels, and range settings for channels greater than 1 cannot be edited. Any changes made to channel 1 are reproduced in the other channels. If **Ranges Track Ch1** is not checked, the fixed input range setting for each channel can be set independently.

Read more about ranging in Chapter 98.

Delay Time

DC Level results can vary with noise-like characteristics. When a noise measurement (or a DC Level measurement) is run from the navigator or as part of a sequence, the acquisition is delayed for the time set here. This pause can avoid signal disruptions that may occur in the DUT when changing measurements. The **Delay Time** range is 0.0 s to 10.0 s; the default setting is 300.0 ms. This delay is applied at every step of the sweep.

Acquisition Time:

DC Level results can vary with noise-like characteristics. Since noise is not periodic, a very short acquisition may display a result that is well above or below the average noise level. In a noise measurement the signal is acquired over a period of time between 100 ms and 10 s, set in Acquisition Time. Longer acquisitions provide more repeatable results. The default acquisition time is 250 ms. This acquisition time is applied at every step of the sweep.

Digital Error Rate
(Sequence Mode)

Introduction

For the APx515, this measurement requires a software option key. See page 189 for more information about software options.

The Digital Error Rate measurements uses a bit-accurate technique to provide results showing both the number of errors and the error rate in a digital audio system.

For meaningful results, an Audio Precision Bit test waveform should be used. Walking Ones, Walking Zeros and Constant waveforms can also be used.

Go to page 277 for more information about bit-accurate measurements and Bit test, Walking Zeros, Walking Ones and Constant Value waveforms.

Digital Error Rate measurements are not available in analog input configurations.

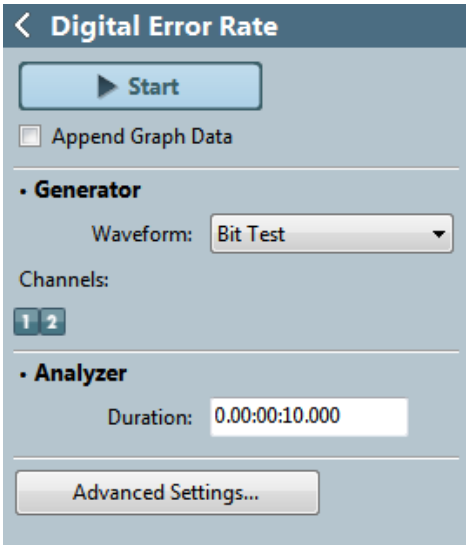
Digital Error Rate measurements available in APx500 are:

- Average Error Rate
- Instantaneous Error Rate
- Total Errors
- Cumulative Errors

Operation

Append Graph Data

Measured data are grouped in a Data Set, which contains all acquisition results. Normally, the current graph data are deleted each time you start a new measurement. If the **Append Graph Data** box is checked, appended data are grouped as additional data sets, as are any imported data. The Data Sets Panel displays the Data Set(s) in the measurement, and provides a visibility checkbox and a Notes field for each data set. Read more about Data Sets beginning on page 649.



Running the measurement

Click **Start**. The generator will output the test signal to the DUT on the selected generator channels. The acquisition will begin and the graph will display the error rate as the signal is acquired. The acquisition will continue until the time set in **Duration**.

Generator

Selecting the generator waveform

By default, the Average Error Rate measurement uses the Bittest Random waveform (see page 50) at a fixed level.

Alternatively, you can select a generator waveform file as the test signal, using the file browser in the Waveform drop-down menu. See Chapter 14 for more information about using Generator Waveform files. For useful results, generator waveforms for Digital Error Rate measurements must be Audio Precision bit-accurate waveforms.

Analyzer

For Digital Error Rate, the analyzer automatically determines whether or not the acquired signal is a compatible bit-accurate signal. If so, the signal is analyzed for error rate, regardless of the current generator setting. Compatible bit-accurate signals are

- Audio Precision Bit test
- Constant Value
- Walking ones
- Walking zeros

Note: Constant Value signals of all ones (1s) or all zeros (0s) are treated as invalid Constant Value signals, and will return errors. This implementation serves to flag digital connection errors, which can incorrectly produce a stream of constant ones or zeros. When using Constant Value signals, set the signal to any value except ones or zeros.

Duration

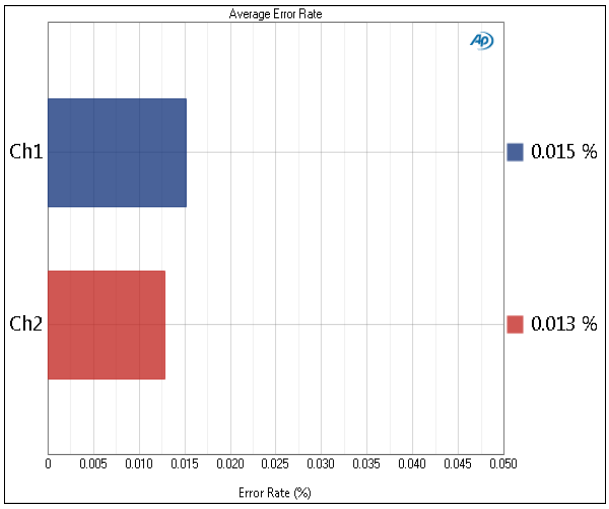
Duration sets the length of the sample acquisition. Minimum time is 0.1 s, maximum time is 7 days.

Enter the Duration following this pattern:

d:hh:mm:ss.s

where d = days, hh = hours, mm = minutes, ss = seconds. Days and fractional seconds are optional.

Average Error Rate measurement



The Digital Error Rate: Average Error Rate measurement result provides a single-value measurement of the average audio bit error rate. The average error rate is calculated by dividing the errors received by the total number of samples received.

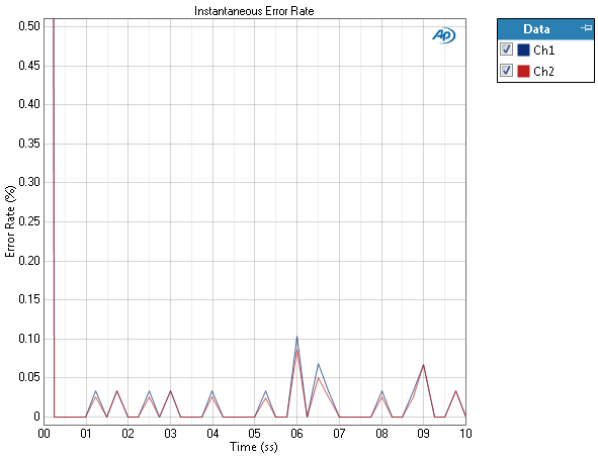
Units

Units available for Average Error Rate are

X-axis

- %
  Error percentage.
- Err/S
  Errors per sample, expressed in exponential notation (also called scientific notation).

Instantaneous Error Rate measurement



The Digital Error Rate: Instantaneous Error Rate measurement result provides a record of the rate of digital errors versus elapsed time.

Units

Units available for Instantaneous Error Rate are

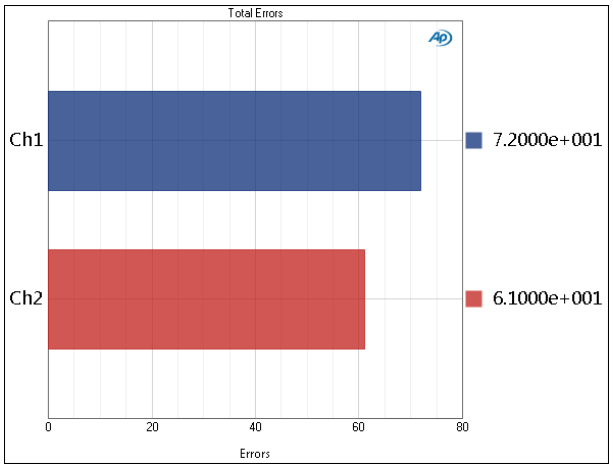
X-axis

- s (seconds)

Y-axis

- %
  Error percentage.
- Err/S
  Errors per sample, expressed in exponential notation (also called scientific notation).

Total Errors measurement



The Digital Error Rate: Total Errors measurement result provides a single-value measurement of the total digital errors acquired.

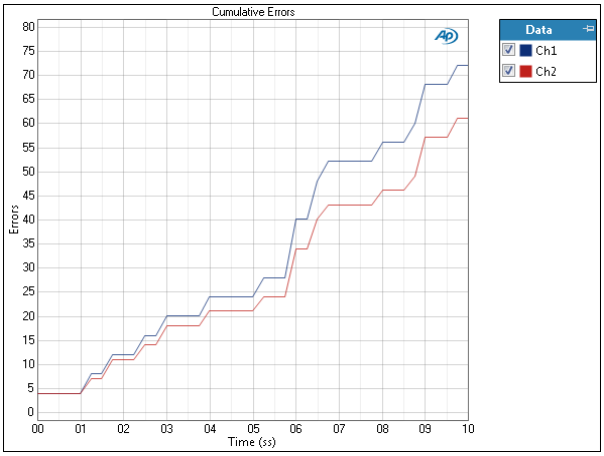
Units

Units available for Total Errors are

X-axis

- The X-axis is unitless. The total number of errors is displayed, using exponential notation (also called scientific notation).

Cumulative Errors measurement



The Digital Error Rate: Cumulative Errors result provides a record of the accumulation of digital errors versus elapsed time.

Units

Units available for Cumulative Errors are

X-axis

- s (seconds)

Y-axis

- %
  Error percentage.
- Err/S
  Errors per sample, expressed in exponential notation.

Generator Units

Generator units available for Constant Value are

- hex
- D

More about Bit-accurate measurements

Bit-accurate measurements are digital domain measurements that use test signals in which the value of every audio sample is known. In analysis, each sample that does not have the correct value is counted as an error. Measurement results may be total errors, error rate, and so on.

A digital device or system that can pass a digital signal no errors is said to be exhibiting bit-accurate performance.

APx500 generates and analyzes four signals used for bit-accurate measurements:

- Bit test
- Walking Zeros
- Walking Ones
- Constant Value

Read more about these waveforms on page 50.

Bit-exact setting

To maintain bit-accuracy when generating a digital waveform for digital output, the signal from the generator must not be dithered, changed in level or modified in any way. The Bit-Exact setting on generator panels enforces this by making level and offset fields unavailable, and by defeating dither (if enabled) while the generator is on.

DIM IMD
(Sequence Mode)

Note: DIM IMD measurements are supported only in the APx555 or in instruments fitted with the AG52 option, when in analog output configurations. Digital output configurations or External Source configuration are not supported. See page 5 for information about the AG52 Option and page 6 for information about the APx555.

DIM stands for Dynamic Intermodulation Distortion. DIM is a technique for measuring IMD under rapidly-changing dynamic conditions, which typically stress analog power amplifiers.

For more information about the DIM method, see page 281. For information about IMD in general, see page 305.

The DIM IMD measurement results available in APx500 are:

- DIM Ratio
- DIM Distortion Product Ratio

Operation

If you have not yet set up your test, first go to Signal Path Setup to select and configure inputs and outputs, references and other settings. See Chapter 7 for information about Signal Path Setup.

Generator

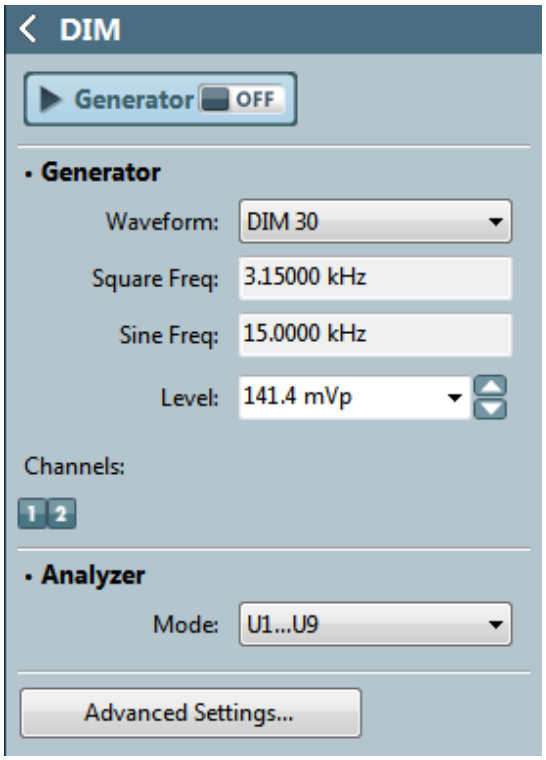
This measurement must be performed in a closed-loop analog-output configuration, using the APx generator as a stimulus. See Chapter 6 for detailed information about the using the APx Generator and making Waveform, Level and Frequency settings.

Selecting the DIM generator waveform

The DIM Ratio measurement uses a DIM test waveform at the level set in the Generator panel.

The DIM test waveforms are as follows:

- **DIM 30**
  The DIM 30 signal is the linear sum of a square wave of 3.15 kHz and a sine wave of 15.00 kHz. The square wave is filtered by a single-pole low-pass filter at 30 kHz.
- **DIM 100**
  The DIM 100 signal is the linear sum of a square wave of 3.15 kHz and a sine wave of 15.00 kHz. The square wave is filtered by a single-pole low-pass filter at 100 kHz.
- **DIM B**
  The DIM B (broadcast) signal is the linear sum of a square wave of 2.96 kHz and a sine wave of 14.00 kHz. The square wave is filtered by a single-pole low-pass filter at 30 kHz.
- **DIM B8**
  The DIM B8 signal is the linear sum of a square wave of 2.96 kHz and a sine wave of 8.00 kHz.



The square wave is filtered by a single-pole low-pass filter at 30 kHz.

In all cases, the ratio of the peak voltage of the square wave to the peak voltage of the sine wave is 4:1.

Analyzer

No local low pass, high pass or weighting filters are available for this measurement. However, low pass, high pass and weighting filter settings made in Signal Path Setup > Input/Output will affect this measurement.

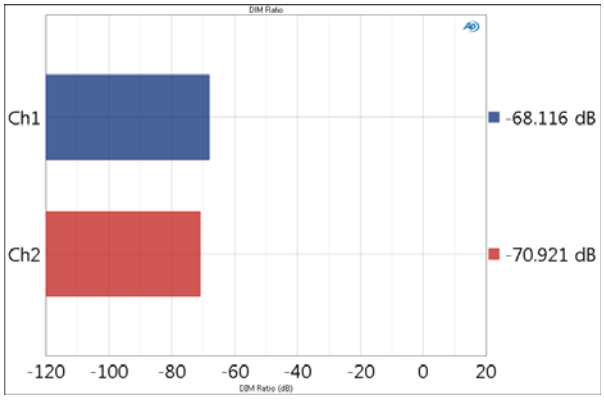
Mode

- **U1...U9**
  The IEC 60268-3 standard defines the DIM measurement, listing nine specific IM components and identifying the corresponding output voltages as U1, U2, U3, U4, U5, U6, U7, U8 and U9. The U1...U9 selection is the APx500 default, and displays the rms summation of all nine voltages expressed as a ratio to the amplitude of the test signal sine wave component.
- **U4+U5**
  This mode includes only the two IM components that fall below the test signal square wave frequency, U4 and U5.
- **Emulation**
  Audio Precision System One, System Two, Cascade and 2700 series instruments measured DIM differently, using a combination of analog filters and rms meters. Choose **Emulation** to view DIM results compatible with results from earlier Audio Precision instruments.

Advanced Settings

If your test requires special settings, click **Advanced Settings**. Advanced Settings for IMD measurements include individual channel generator levels, custom IMD frequencies, individual channel analyzer ranging and custom analyzer settling. See Advanced Settings for IMD on page 306. See Chapter 104 for more information about units of measurement.

DIM Ratio results



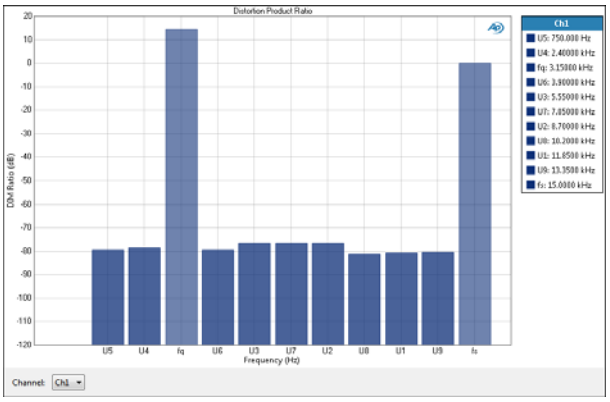
The DIM Ratio result provides single-value measurement results of the IMD (intermodulation distortion) ratio in the output signal from each DUT channel.

Units

Units available for DIM Ratio are

- x/y
- %
- ppm
- dB

DIM Distortion Product Ratio results



The DIM Distortion Product Ratio result provides measurement results of the IMD (intermodulation distortion) ratio in the output signal from each DUT channel.

Read the DIM Distortion Product Ratio for the selected channel from the meter bar display. The fundamental square wave fq, the fundamental sine wave fs and the distortion products U1 through U9 (as chosen by the Mode selector) will be shown in the meter bar display. The graph legend shows the frequency of each distortion product.

Result Settings: Channel

This setting is made in the Result Settings bar, beneath the graph display.

The Distortion Product Ratio result shows only one channel at a time. Select the channel to be viewed from the **Channel** drop-down list.

Units

Units available for DIM Distortion Product Ratio are

- x/y
- %
- ppm
- dB

See Chapter 104 for more information about units of measurement.

More About DIM IMD

DIM stands for Dynamic Intermodulation Distortion. It is a technique to measure the nonlinearity of a device, designed to be particularly sensitive to distortions produced during transient conditions typical of program material. DIM measurements typically use a square wave of about 3 kHz summed with a lower-amplitude sine wave of 14 kHz to 15 kHz. The DIM measurement is defined in the standard IEC 60268-3, sec. 14.12.9.

DIM is also referred to as TIM—Transient Intermodulation Distortion.

For information about IMD in general, see page 305.

DIM Level Sweep
(Sequence Mode)

Note: DIM IMD measurements are supported only in the APx555 or in instruments fitted with the AG52 option, when in analog output configurations. Digital output configurations or External Source configuration are not supported. See page 5 for information about the AG52 Option and page 6 for information about the APx555.

DIM stands for Dynamic Intermodulation Distortion. DIM is a technique for measuring IMD under rapidly-changing dynamic conditions, which typically stress analog power amplifiers.

The DIM Level Sweep measurement provides a DIM stimulus signal (square wave mixed with sine wave, as described on pages 279 and 281) that is moved across a range of levels in a series of points. The DUT output is acquired by the analyzer and processed for display.

For more information about the DIM method, see page 281. For information about IMD in general, see page 305.

DIM Level Sweep results available in APx500 are:

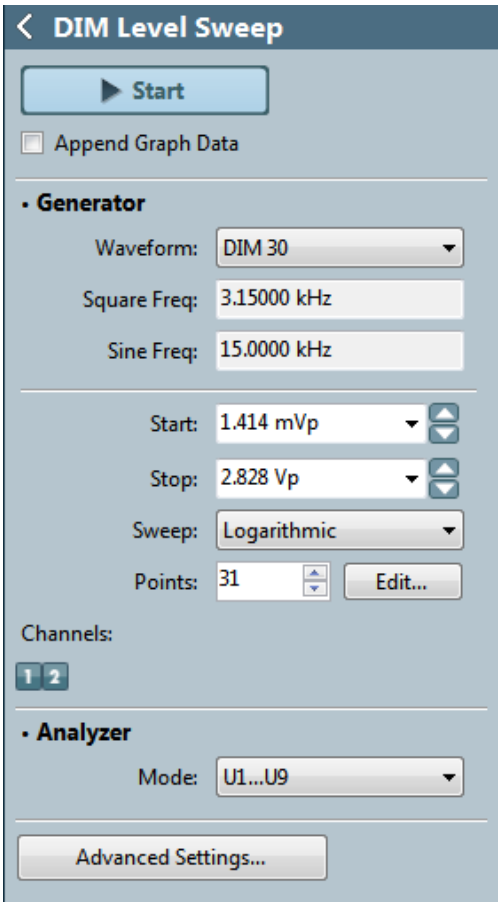
- DIM Ratio
- DIM Ratio vs. Measured Level

Operation

Append Graph Data

Normally, the current graph data for all results is deleted each time you start a new measurement. If the **Append** box is checked, the current data is kept in memory as a **Data Set** (see page 189), and the new measurement data is appended as a new data set. You can append many **Data Sets**.

If you have not yet set up your test, first go to Signal Path Setup (Chapter 7).



Generator

This signal will be swept from the **Start Level** to the **Stop Level** in the set number of **Points**.

Level Settings for DIM IMD

The DIM Level Sweep measurement uses a DIM test waveform at the level set in the Signal Generation panel as the test signal.

The DIM test waveforms are as follows:

- **DIM 30**
  The DIM 30 signal is the linear sum of a square

wave of 3.15 kHz and a sine wave of 15.00 kHz. The square wave is filtered by a single-pole low-pass filter at 30 kHz.

- **DIM 100**
  The DIM 100 signal is the linear sum of a square wave of 3.15 kHz and a sine wave of 15.00 kHz. The square wave is filtered by a single-pole low-pass filter at 100 kHz.
- **DIM B**
  The DIM B (broadcast) signal is the linear sum of a square wave of 2.96 kHz and a sine wave of 14.00 kHz. The square wave is filtered by a single-pole low-pass filter at 30 kHz.
- **DIM B8**
  The DIM B8 signal is the linear sum of a square wave of 2.96 kHz and a sine wave of 8.00 kHz. The square wave is filtered by a single-pole low-pass filter at 30 kHz.

In all cases, the ratio of the peak voltage of the square wave to the peak voltage of the sine wave is 4:1.

Start Level, Stop Level and Sweep Points

The sweep moves between two levels, set in the **Start Level** and **Stop Level** fields. The default levels are 1.000 mVrms and 2.000 Vrms.

The sweep moves in a specified number of step points, set in the **Points** field. The minimum is 2 points; maximum is 65,535. The default setting is 31.

The sweep point spacing is set by selecting one of the following choices in the Sweep field:

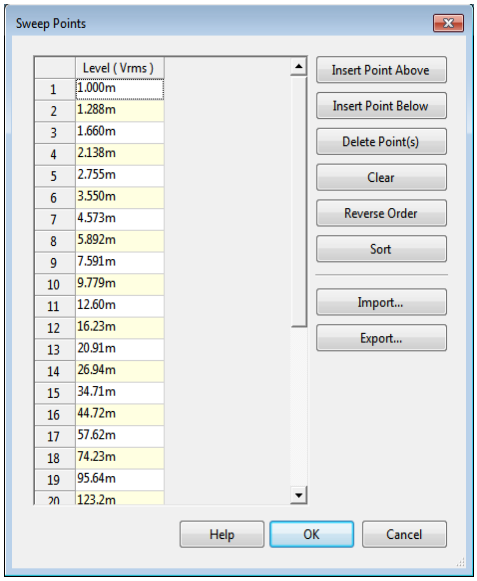
- **Logarithmic** (the default); use the **Points** field to set the number of logarithmically spaced points;
- **Linear**, which provides two methods of adjusting spacing: the **Points** field or the **Step Size** field; or
- **Custom**. Click **Edit** to open the **Sweep Points** dialog, where you can set points arbitrarily, or load or save sweep table files.

Viewing or Editing the Sweep Points table

You can view or edit the current sweep points at any time.

Click **Edit** to open the **Sweep Points** table. The table shows each sweep point and its corresponding level. You can edit this table to add or delete points, or to change the level of a point. Points can be sorted or reversed in order using the controls on the right.

A **Sweep Points** table can be saved as a *.csv file or as a Microsoft Excel *.xls file. A compatible *.csv or *.xls file can be opened and used as a **Sweep Points** table.



Analyzer

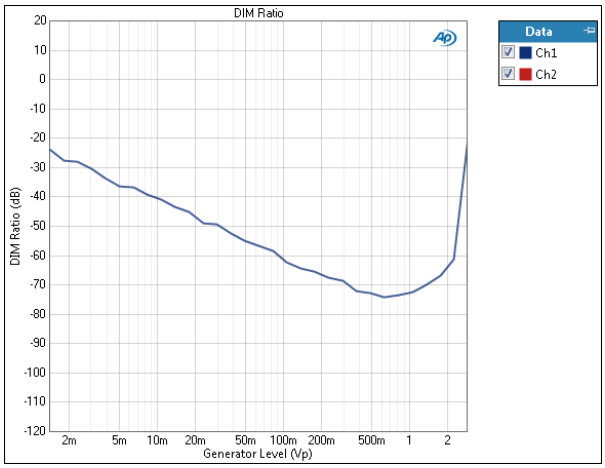
Mode

- **U1...U9**
  The IEC 60268-3 standard defines the DIM measurement, listing nine specific IM components and identifying the corresponding output voltages as U1, U2, U3, U4, U5, U6, U7, U8 and U9. The **U1...U9** selection is the APx500 default, and displays the rms summation of all nine voltages expressed as a ratio to the amplitude of the test signal sine wave component.
- **U4+U5**
  This mode includes only the two IM components that fall below the test signal square wave frequency, U4 and U5.
- **Emulation**
  Audio Precision System One, System Two, Cascade and 2700 series instruments measured DIM differently, using a combination of analog filters and rms meters. Choose Emulation to view DIM results compatible with results from earlier Audio Precision instruments.

Advanced Settings

If your test requires individual generator settings for each channel or special analyzer ranging or settling, click **Advanced Settings**. See Advanced Settings for IMD Level Sweeps on page 315. See Chapter 104 for more information about units of measurement.

DIM Ratio (vs. Generator Level)



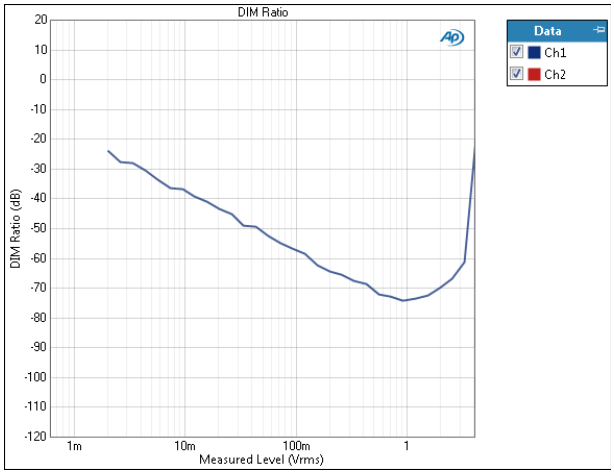
DIM Ratio plots the distortion ratio on the Y axis against the Generator level on the X axis.

Units

Units available for the DIM Ratio vs. Generator Level result are

X-axis	*Y-axis*
• Vrms	• x/y
• Vp	• %
• Vpp	• ppm
• dBV	• dB
• dBu	
• dBrG	
• dBm	
• W (watts)	

DIM Ratio vs. Measured Level



DIM Ratio vs. Measured Level plots the distortion ratio on the Y axis against the DUT output level on the X axis.

Units

Units available for the DIM Ratio vs. Measured Level result are

X-axis (analog)	*X-axis (digital)*	*Y-axis*
• Vrms	• FS	• x/y
• Vp	• %FS	• %
• Vpp	• dBFS	• ppm
• dBV	• dBrA	• dB
• dBu	• dBrB	
• dBrG	• dBSPL1	
• dBm	• dBSPL2	
• W (watts)		

See Chapter 104 for more information about units of measurement.

DUT Delay
(Sequence Mode)

This measurement is not available if Input or Output Connector is ASIO, or Input Connector is File, or Output Connector is None.

DUT Delay measures the length of time an audio signal is delayed through a device under test (DUT) or system under test, such as Bluetooth A2DP, or a speaker or headphone under test in an acoustic space. For chirp-based measurements such as Continuous Sweep, Acoustic Response or Loudspeaker Production Test, the delayed signal may fall out of the analysis window. DUT Delay provides a precise measurement of the delay through the device, which can be used to correct the position of the analysis window by entering the delay value into the Signal Path Setup DUT Delay field (see page 55).

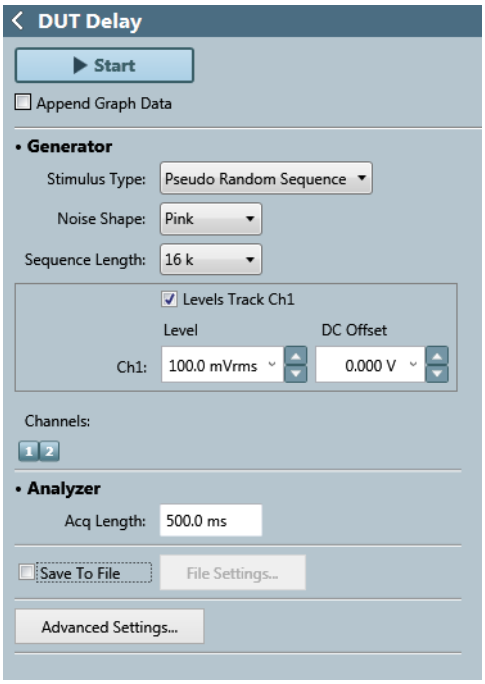
The DUT Delay result may also be useful in itself, in verifying or characterizing a device's inherent delay, or the propagation delay in an acoustic space.

This measurement determines loop delay (the delay between the generator output and the analyzer input) using either a pseudo random sequence (PRS) stimulus signal, as recommended in ITU-T P.501, or a maximum length sequence (MLS) signal. These are deterministic noise-like stimulus signals that can be cross-correlated to a fine granularity, providing a precise delay result.

Note: any delay previously entered in the Signal Path Setup DUT Delay field is set to zero while this measurement is running.

DUT Delay results available in APx500 are:

- **DUT Delay**
  This is a meter that shows the delay through the device under test.
- **Cross Correlation**
  This graph shows the cross-correlation result, which can be useful for diagnostics. There is typically a strong primary peak that shows the initial delay.



- **Acquired Waveform**
  This graph shows the acquired waveform, which can be useful for diagnostics. If the waveform is truncated at the end, increase the acquisition length.

Operation

If you have not yet set up your test, first go to Signal Path Setup to select and configure inputs and outputs, references and other settings. See Chapter 7 for information about Signal Path Setup.

Append Graph Data

Measured data are grouped in a **Data Set**, which contains all acquisition results. Normally, the current graph data are deleted each time you start a new measurement. If the **Append Graph Data** box is checked, appended data are grouped as additional

data sets, as are any imported data. The Data Sets Panel displays the Data Set(s) in the measurement, and provides a visibility checkbox, a Time acquisition field, and a Notes field for each data set.

Read more about Data Sets beginning on page 649.

Generator

The configuration for this measurement must be closed-loop configuration, using the APx generator as a stimulus. Click **Start** to run the measurement and read the results.

Stimulus Type

DUT Delay uses one of two noise-like binary sequences as a stimulus. Although there are many other uses for such stimuli, here they are used exclusively for cross-correlation between the sent and received signals to determine delay.

- **Pseudo Random Sequence**
  (PRS is recommended for this application in ITU-T P.501)
- **Maximum Length Sequence**
  (The MLS sequence cross-correlates better and can provide more accurate results. MLS also has a 0 dBFS crest factor, much higher than the PRS)

Noise Shape

Unfiltered, the noise-like PRS and MLS binary sequences have equal intensity at every frequency (a constant power spectral density; white noise).

These signals can be filtered to become pink noise, where the power spectral density is inversely proportional to the frequency. With pink filtering, each octave carries an equal amount of energy. Pink noise filtering is recommended in acoustic applications, where it offers a reduced risk of overloading high-frequency transducers.

Select the noise shaping here:

- **White**
- **Pink** (the default)

Sequence Length

The sequence length is the number of samples in the stimulus signal. The sequence length divided by the sample rate is the duration of the signal, and therefore longest delay that can be measured. Note that the menu selections use a common shorthand, representing sequence lengths that are actually powers of two, as shown below in parentheses.

- **16k** (16,383) 341 ms at 48 kHz SR
- **32k** (32,767) 683 ms at 48 kHz SR
- **64k** (65535) 1.37 s at 48 kHz SR
- **128k** (131,071) 2.73 s at 48 kHz SR

You must use a sequence length that is longer than the expected delay, at the chosen sample rate.

Levels Track Ch 1

If this box is checked, all channels are set to the same level as channel 1.

Level

Set the output level here.

DC Offset

Set an optional DC offset here.

Analyzer

Acq Length

Set the acquisition length here, in seconds. To capture an entire sequence, set the acquisition length to the sum of the duration of the sequence at the current sample rate, plus any expected delay in the system under test.

The acquisition starts at the moment the generator begins to output the sequence; in a DUT or system with significant delay (like an acoustic path) you may want to extend the acquisition to capture the propagation delay and the full sequence.

Save to File

If this box is checked, the acquisition will be saved to a file.

File Settings

Set file name, format and location in the Save Acquisition to File Settings dialog. See page 382.

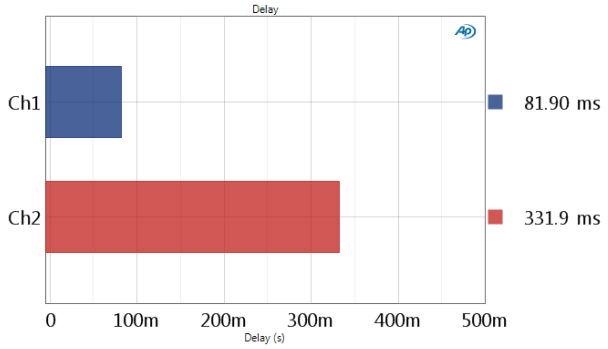
Advanced Settings

Advanced Settings makes available individual channel output level and offset controls, and input ranging controls. See Advanced Settings for single value measurements on page 351. See Chapter 104 for more information about units of measurement.

Diagnosing problems

If the system under test adds a great deal of noise or distortion, or greatly limits the bandwidth of the signal, the acquired signal will not cross-correlate well and the delay result will be spurious. See Diagnosing Problems in DUT Delay cross-correlation on page 289.

Delay



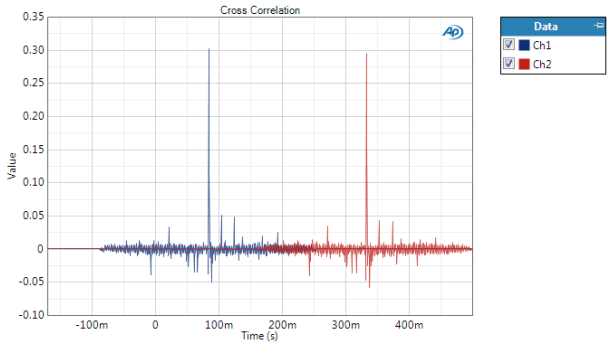
Delay provides a single-value meter result, displaying the Delay for each DUT channel. In this example, channel 1 is delayed by 81.90 ms, and channel 2 is delayed by 331.9 ms

Units

Units available for Delay are:

- seconds

Cross Correlation



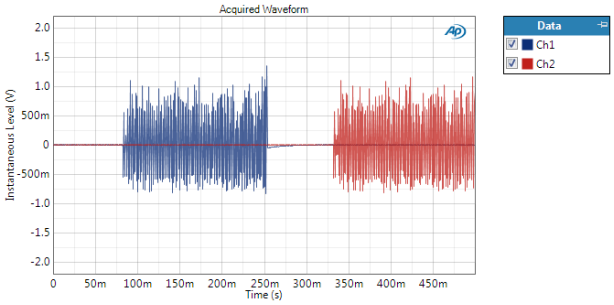
Cross Correlation is a diagnostic result, used to confirm the validity of a result, or provide information about a spurious result. Good cross correlation will show a strong peak at the delay value.

Units

Units available for Cross Correlation are:

- X-axis: unitless Value
- Y-axis: seconds

Acquired Waveform



Acquired Waveform is a diagnostic result, used to confirm the validity of a result, or provide information about a spurious result. A good acquired waveform will show a clear burst of signal surrounded by relative silence.

Interpolation

The interpolation setting is only available when viewing Scope results.

For time-domain displays, the APx500 graphs are normally plotted using sinc function interpolation. Interpolation adds points to the displayed trace that do not exist in the measured data, to make data trends more easily visualized. The default in APx is to have Interpolation **On**. However, digital domain signals are sometimes best understood when viewing actual samples, with interpolation switched **Off**.

Interpolation can cause limit failure markers to appear to be in an incorrect location. See “Interpolation and limit failure markers” on page 678 for an explanation.

Units

Units available for Acquired Waveform are:

- X-axis: Instantaneous Level (V)
- Y-axis: seconds

Diagnosing Problems in DUT Delay cross-correlation

If the system under test adds a great deal of noise or distortion, or greatly limits the bandwidth of the signal, the acquired signal will not cross-correlate well and the delay result will be spurious. We have found that the MLS sequence is much more likely to cross-correlate well in poor conditions.

We recommend that you verify your connections prior to making a DUT Delay measurement, to assure that signal passes well.

Measure and Set DUT Delay sequence step

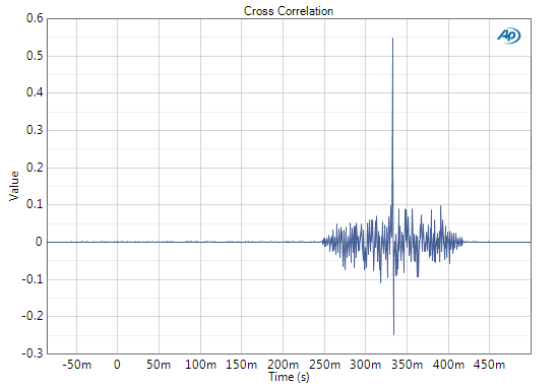
If you are using the Measure and Set DUT Delay sequence step, it is advisable to first manually (outside of a sequence) verify that your DUT or system under test can pass the DUT Delay signals with integ-

rity, and that you get a satisfactory delay result. Use the DUT Delay Measurement to accomplish this. If you are occasionally getting spurious delay results, you can add a DUT Delay Measurement later in the sequence with a limit set, to flag possible DUT Delay errors.

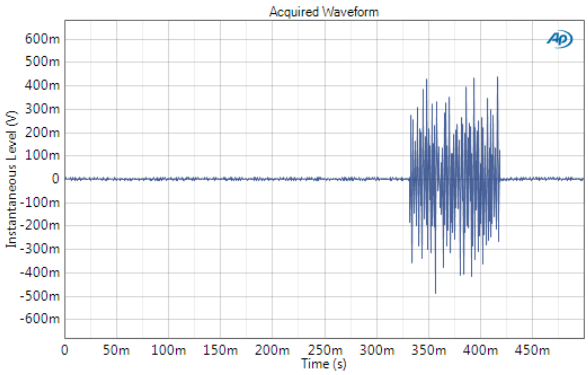
Examples of good and poor cross-correlation

The graphs on the next page were made with the same DUT and connections. The device under test is a digital delay line, set at a delay of 332 ms. The analog output of the delay line was routed through an analog mixer, and the top two graphs were made, with a correct measured delay of 332 ms. A strong white noise signal was added in the mixer to produce the next two graphs, which produced a spurious measured delay of 87.25 ms.

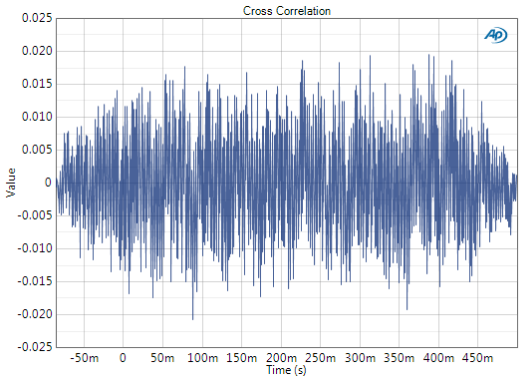
Good cross correlation (top graphs) and poor cross correlation (bottom graphs)



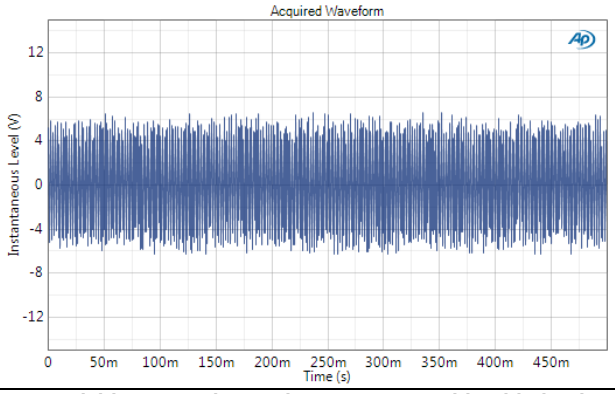
Cross-correlation, Pseudo Random Sequence without added noise. Peak is at 332 ms.



Acquisition, Pseudo Random Sequence without added noise. Peak is at 332 ms.



Cross-correlation, Pseudo Random Sequence with added noise.



Acquisition, Pseudo Random Sequence with added noise.

Dynamic Range (AES17)
(Sequence Mode)

Application

For the APx515, this measurement requires a software option key. See page 189 for more information about software options.

We recommend using this AES17 dynamic range measurement for digital converters. When testing other devices, we recommend using the Signal-to-Noise Measurement described in Chapter 79.

Operation

If you have not yet set up your test, first go to Signal Path Setup to select and configure inputs and outputs, references and other settings. See Chapter 7 for information about Signal Path Setup.

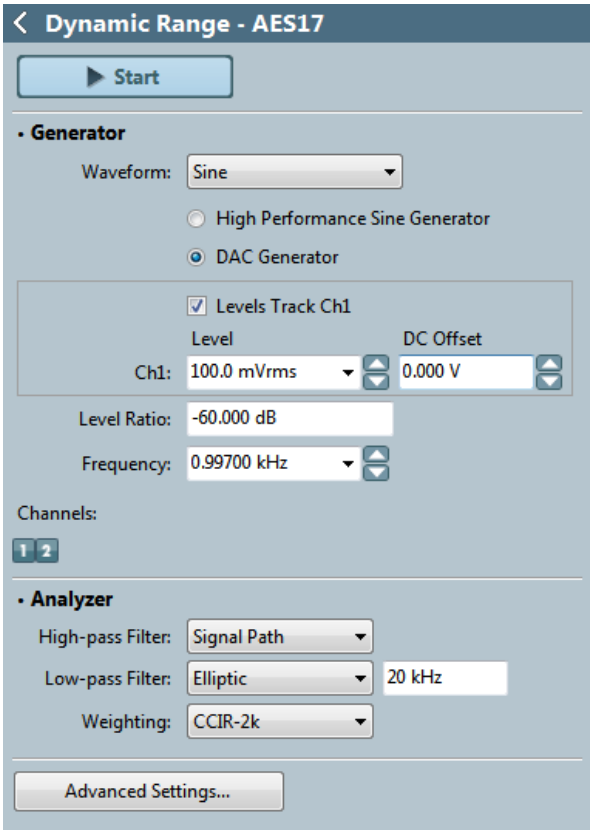
Running the measurement

To measure Dynamic Range, click **Start**. The generator is turned ON briefly at the specified full level on the selected generator channels, and a measurement is made. Then the generator is set to the lower level (set in Level Ratio), the signal is notched out, and a second measurement is made. The ratio between the two results is computed for each channel and displayed as meter bars.

Generator

The typical configuration for this measurement is the closed-loop configuration, using the APx generator as a stimulus. Alternatively, you can make open-loop measurements, using either the External Source or File Input configuration.

See Chapter 6 for detailed information about the using the APx Generator and making Waveform, Level and Frequency settings, or for information about using External Source.



High Performance Sine Generator / DAC Generator

For an APx555, you have the option of using the High Performance Sine Generator (very low distortion sine waves and wide bandwidth) or the DAC generator, which provides special sine waveforms. See High Performance Sine Generator starting on page 51 for a comparison of features.

Level

For dynamic range, you must choose a generator level that represents the maximum level for the device under test (DUT).

DAC testing

For DACs, the maximum level is usually 0 dBFS.

ADC testing

For ADCs, it is usually the analog signal that just produces 0 dBFS at the ADC output. You can set the analog generator to produce a target digital output level in Reference Levels, and then set that generator level as dBrG. Return to the Dynamic Range measurement and choose 0 dBrG as the Generator level.

Level Ratio

AES17 recommends the lower signal be set to 60 dB below the full level signal (the APx500 default). Other ratios can be selected.

Running the measurement in External Source configuration (Open Loop)

When configured for External Source, use a signal source that varies between your desired maximum value, and your desired minimum value (usually 60 dB below the maximum). APx500 uses an algorithm that monitors the incoming signal, waiting for readings that indicate a level change of at least 20 dB. When this can be reliably determined, a tuned notch filter is applied to the lower level acquisition and the ratio between the two levels is displayed as the Dynamic Range result.

The Audio Precision utility APxWfmGenerator.exe (available from www.ap.com) provides a suitable waveform called “Dynamic Range (1 kHz)”. This waveform is a 0.99700 kHz tone at a level of 0 dBFS, alternating with periods of 0.99700 kHz at –60 dBFS.

Analyzer

Filters

Local high-pass, low-pass and weighting filters are available for the lower (typically -60 dB) acquisition of this measurement. A low-pass filter set just above the audio passband (at 20 kHz) is recommended for noise measurements. This is the APx default.

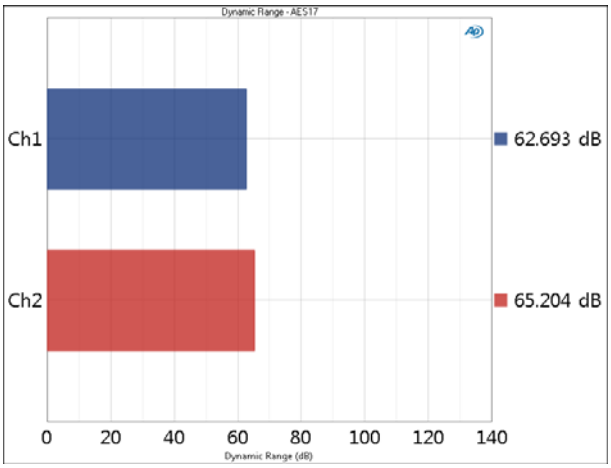
AES17 recommends the CCIR-2k weighting filter for this measurement; this is the APx default. See Measurement Filters on page 619 for detailed information about the filters available locally for some measurements.

Low pass, high pass and weighting Signal Path Filters settings (see page 617) made in Signal Path Setup > Input/Output will also affect this measurement. When set, the local low-pass or high-pass filter setting overrides the corresponding Signal Path filter setting while the measurement is active. A local weighting filter setting does not override any Signal Path filter settings; the local weighting filter is applied in addition to any active Signal Path filters.

Advanced Settings

If your test requires individual generator settings for each channel or special analyzer ranging or settling, click **Advanced Settings**. See Advanced Settings for single value measurement on page 351.

Dynamic Range results



The Dynamic Range (AES17) measurement provides a single-value result showing the AES17 Dynamic Range for each DUT channel.

Units

Units available for Dynamic Range are

- x/y
- dB

See Chapter 104 for more information about units of measurement.

More about Dynamic Range

Overview

Dynamic Range is an expression of the ratio of the largest signal a device can pass to the device’s noise floor. “Largest signal” usually refers to a signal at a specified degree of distortion, often 1%. Signal-to-Noise Ratio and Dynamic Range are essentially the same measurement, except that the signal in SNR is arbitrary (and should be stated in the results), and the signal in Dynamic Range is at the maximum (details of which should also be stated in the results).

For digital converter measurements...

We recommend using this AES17 dynamic range measurement. It is intended specifically for ADC (analog-to-digital converter) and DAC (digital-to-analog converter) dynamic range and “noise in the presence of signal” measurements, as described in Section 9.3 of AES17. A similar method is defined in IEC61606.

This method differs from standard signal-to-noise and dynamic range measurements in that it uses a –60 dBFS stimulus during the noise measurement. This method is used for two reasons:

In both ADCs and DACs, “idle tones” can be produced within the converter in the absence of applied signal. In the method here, a low-level tone is applied to the converter to avoid production of idle channel noise. The low-level tone is removed by a notch filter before measurement.

In some DACs, the output of the device is switched off when there is no signal, providing an unrealistically quiet measurement. The low-level tone (again, notched out before measurement) defeats this muting mechanism.

At –60 dBFS, the tone is so low that any distortion products created are below the noise floor.

For dynamic range measurements of other devices...

We recommend using the Signal-to-Noise measurement configured for dynamic range when testing other devices. It is described in Chapter 79. The low-level signal is not required for non-converter devices.

Frequency
(Sequence Mode)

The Frequency measurement provides a single-value result showing the frequency of the strongest component in the output signal of each DUT channel, as measured at each analyzer input.

Operation

If you have not yet set up your test, first go to Signal Path Setup to select and configure inputs and outputs, references and other settings. See Chapter 7 for information about Signal Path Setup.

Generator

The typical configuration for this measurement is the closed-loop configuration, using the APx generator as a stimulus. Alternatively, you can make open-loop measurements, using either the External Source or File Input configuration.

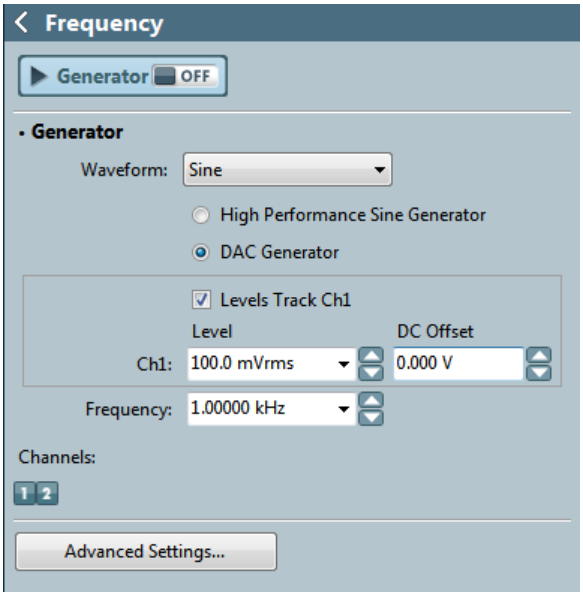
See Chapter 6 for detailed information about the using the APx Generator and making Waveform, Level and Frequency settings, or for information about using External Source.

High Performance Sine Generator / DAC Generator

For an APx555, you have the option of using the High Performance Sine Generator (very low distortion sine waves and wide bandwidth) or the DAC generator, which provides special sine waveforms. See High Performance Sine Generator starting on page 51 for a comparison of features.

Filters

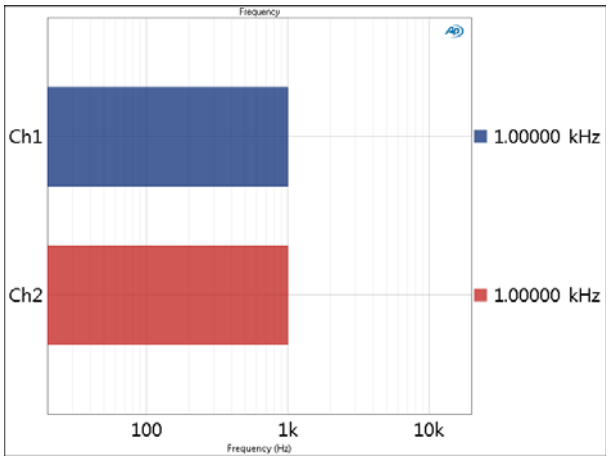
There are no local filters available for this measurement. However, low pass, high pass and weighting filter settings made in Signal Path Setup > Input/Output will affect this measurement.



Advanced Settings

If your test requires individual generator settings for each channel or special analyzer ranging or settling, click **Advanced Settings**. See Advanced Settings for single value measurements on page 56.

Frequency Measurement



Units

Units available for Frequency are

- Hz
- dHz
- F/R
- %Hz

See Chapter 104 for more information about units of measurement.

Frequency Response
(Sequence Mode)

Frequency Response is a continuous sweep measurement that displays the frequency response of each channel, plotted against frequency. A broadband single value deviation result is also computed from the continuous sweep acquisition.

A continuous sweep is a brief log-swept sine wave (a Farina log chirp) that moves continuously across a specified range of frequencies. The DUT output is acquired by the analyzer and is mathematically processed to provide a number of results. For more information about the continuous sweep measurements used in Frequency Response, go to page 246.

Continuous sweep based measurements, including Frequency Response, are not supported in External Source or File Input configurations.

Frequency Response results available in APx500 are:

- Level
- Gain
- Relative Level
- Deviation

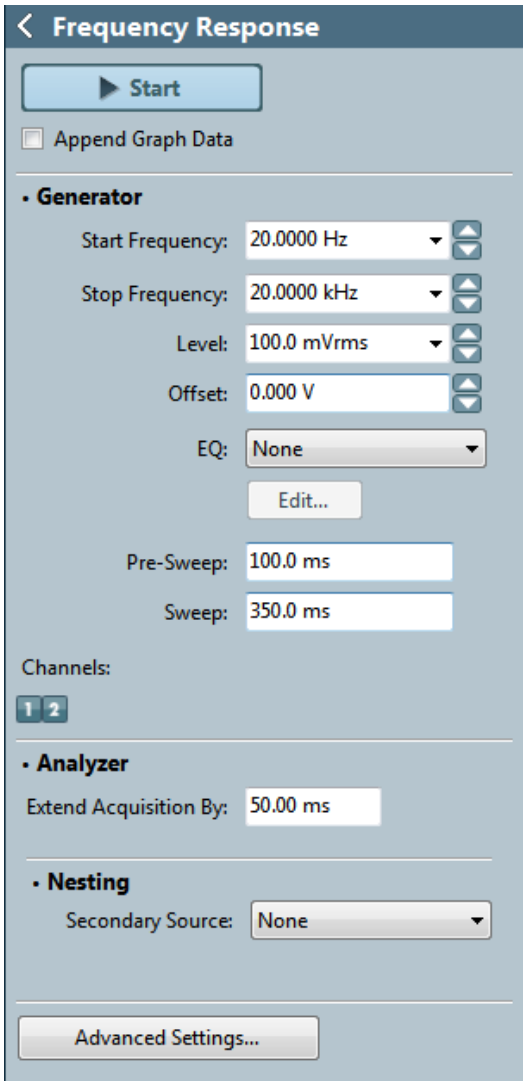
All of these results are available from a single acquisition.

Operation

If you have not yet set up your test, first go to Signal Path Setup to select and configure inputs and outputs, references and other settings. See Chapter 7 for information about Signal Path Setup.

Append Graph Data

Measured data are grouped in a **Data Set**, which contains all acquisition results. Normally, the current graph data are deleted each time you start a new measurement. If the **Append Graph Data** box is checked, appended data are grouped as additional data sets, as are any imported data. The Data Sets Panel displays the Data Set(s) in the measurement, and provides a visibility checkbox, a Time acquisition field, and a Notes field for each data set.



Read more about Data Sets beginning on page 649.

Generator

The Frequency Response measurement uses a continuous sweep (log-swept sine chirp) stimulus signal,

swept between the frequencies entered in **Start Frequency** and **Stop Frequency**, at the value set in **Level**.

Running the Measurement

To use Frequency Response, click **Start**. The generator will output the test signal to the DUT on the selected generator channels. Read the results in the selected result view.

Start Frequency and Stop Frequency

These controls set the frequency range of the sweep. See page 246 for more about continuous sweep ranges, which are dependent upon the instrument.

Level

Set the generator level here.

Offset

Set any DC offset to the generator signal here.

EQ

You can equalize the generator signal to make a pre-emphasized stimulus. See Generator Equalization (page 192) for a general discussion of this feature.

Choosing generator equalization

Select **None, Relative** or **Absolute** from the EQ drop-down menu.

- **None** applies no equalization to the generator.
- **Relative** applies the gain or loss specified in the EQ table to the current generator level for each channel.
- **Absolute** sets the generator level for all channels to the level specified in the EQ table. Note: when **Absolute** is selected, current channel level settings are lost.

Edit

If you have chosen to equalize the generator signal, you can edit the EQ table here. See **Edit EQ Table** (page 192).

Pre-Sweep and Sweep duration fields

A continuous sweep stimulus consists of a sine wave swept logarithmically across the sweep range in the period of time set in these fields.

To enable the DUT to stabilize before the beginning of the defined sweep measurement, there is variable **Pre-Sweep** time. The actual sweep stimulus signal begins below the defined **Start Frequency** and is swept upward, reaching the **Start Frequency** at the end of the **Pre-Sweep** time. The sum of the times in **Pre-Sweep** and **Sweep** is the total sweep length.

- The default for **Pre-Sweep** is 100.0 ms. Minimum is 0 s; maximum is 1.0 s.
- The default for **Sweep** is 350.0 ms. Minimum is 50.0 ms; maximum is 5 s.

Longer **Pre-Sweep** settings allow more time for DUT stabilization, but increase measurement time. Longer **Sweep** settings provide greater resolution and signal-to-noise ratio, but increase measurement time.

Analyzer

The sweep is acquired and processed to provide the various results.

There are no local measurement filters available for continuous sweep measurements. However, low pass, high pass and weighting input filter settings made in Signal Path Setup > Input/Output > Filters will also affect this measurement. See Chapter 97 for more information about filtering.

Extend Acquisition By

By default, a continuous sweep measurement makes an acquisition slightly longer than the stimulus, to include possible time-delayed artifacts created in the DUT. By default, the acquisition is extended 50 ms longer than the stimulus.

In some cases, you may want to extend the acquisition further. Enter a new value in the **Extend Acquisition By:** field. Minimum extension is 0 s; maximum is 3 s.

Nesting

Frequency Response sweeps can be nested. Read about Nested Sweeps beginning on page 185.

Secondary Source

Select the secondary sweep source here. This is the outside, nesting parameter that is varied to change the conditions for the primary sweep through a number of iterations.

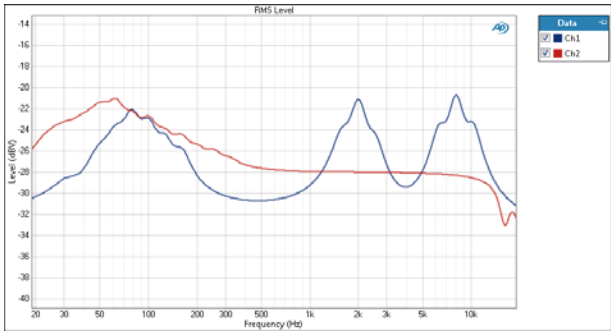
Available choices are dependent upon instrument hardware, options, Input/Output settings and Bench Mode Generator settings. Read about Nested Sweeps beginning on page 185.

Advanced Settings

Advanced Settings include output level and offset settings and input ranging settings. To view or modify these settings, click **Advanced Settings**. See Advanced Settings for Continuous Sweep based measurements on page 249.

See Chapter 104 for more information about units of measurement.

Frequency Response: Level



The Frequency Response: Level result is a continuous sweep measurement that provides a graphical display of the frequency response of each channel. In this result the DUT output level is plotted against frequency.

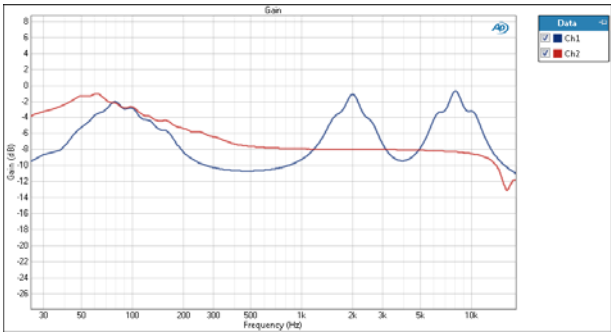
If the settings are identical, this will provide the same results as the Continuous Sweep: Level result.

Units

Units available for Frequency Response: Level are:

X-axis	*Y-axis (analog)*	*Y-axis (digital)*
• Hz	• Vrms	• dBFS
• dHz	• dBV	• FS
• F/R	• dBu	• %FS
• %Hz	• dBrA	• dBrA
	• dBrB	• dBrB
	• dBSPL1	• dBSPL1
	• dBSPL2	• dBSPL2
	• dBm	
	• W (watts)	

Frequency Response: Gain



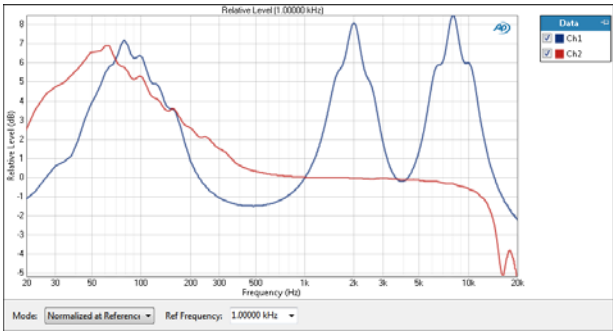
The Frequency Response: Gain result is a continuous sweep measurement that provides a graphical display of the frequency response of each channel. In this result the DUT gain is plotted against frequency.

Units

Units available for Frequency Response: Gain are:

X-axis	*Y-axis same-domain*	*Y-axis cross-domain*
• Hz	• x/y	• FS/Vrms
• dHz	• %	• dB(FS/Vrms)
• F/R	• ppm	—or—
• %Hz	• dB	• Vrms/FS
		• dB(Vrms/FS)

Frequency Response: Relative Level



The Frequency Response: Relative Level result is a continuous sweep measurement that provides a graphical display of the frequency response of each channel. In this result the DUT output level is plotted against frequency, relative to the level at a selected frequency. This enables you to specify the frequency that will be set as 0 dB and view the response in relation to that frequency. A midband frequency such as 1 kHz is often set as the reference frequency, and is the default here.

If the settings are identical, this will provide the same results as the Continuous Sweep: Relative Level result.

Result Settings

These settings are made in the Result Settings bar, beneath the graph display.

Mode

Normalized at Reference

Enter a reference frequency for normalization in the **Ref Frequency** field. The measured level at the frequency you choose is set as 0 dB on the graph, and the DUT response is plotted in relation to this reference. You can change the **Ref Frequency** at any time (except after appending) and the graph will immediately be redrawn to reflect the new setting.

Auto-Centered in Limits

This control is only usable if upper and lower limits are set on this graph. In some applications, the general conformance of a trace to the shape of a set of limits is of importance, with absolute level less important or irrelevant. **Auto-Centered in Limits** centers the data trace relative to the upper and lower limits. To exclude

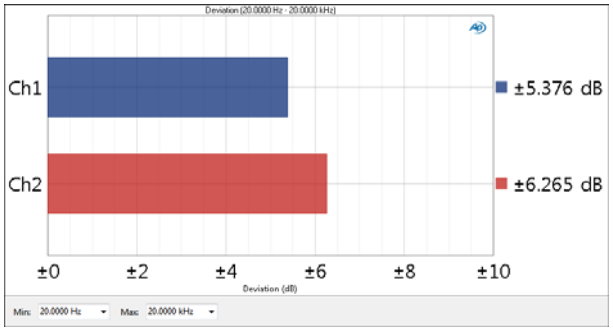
possible exceptional data points at frequency extremes, the frequency range that is considered for the centering operation is constrained by the values in the **Min** and **Max** fields.

Units

Units available for Frequency Response: Relative Level are:

X-axis	*Y-axis*
• Hz	• x/y
• dHz	• %
• F/R	• ppm
• %Hz	• dB

Frequency Response: Deviation



Frequency Response: Deviation is a single value result computed from the continuous sweep acquisition. In this result the frequency deviation (the total range of frequency variation) of each channel is displayed as a meter bar. You can specify a minimum and maximum frequency to define the range to be considered in the deviation measurement.

To measure Frequency Response: Deviation, first enter a frequency range in the **Min Frequency** and **Max Frequency** fields below the graph; or, accept the default 20 Hz to 20 kHz range.

You can change the **Deviation Min Frequency** or **Max Frequency** settings at any time and the meter bars will immediately redraw to reflect the new settings.

Result Settings

These settings are made in the Result Settings bar, beneath the graph display.

Min

Set the minimum frequency of the range of interest here.

Max

Set the maximum frequency of the range of interest here.

Units

Units available for Frequency Response: Deviation are

- dB

IMD
(Sequence Mode)

Introduction

For the APx515, these measurements require a software option key. See page 189 for more information about software options.

IMD (intermodulation distortion) measurements use two tones of different frequencies summed into a stimulus signal. This measurement provides single-value IMD ratio and IMD distortion product results.

The IMD measurement provides stimulus tones and analysis for four important methods of measuring intermodulation distortion: SMPTE, DFD, CCIF and MOD.

Operation

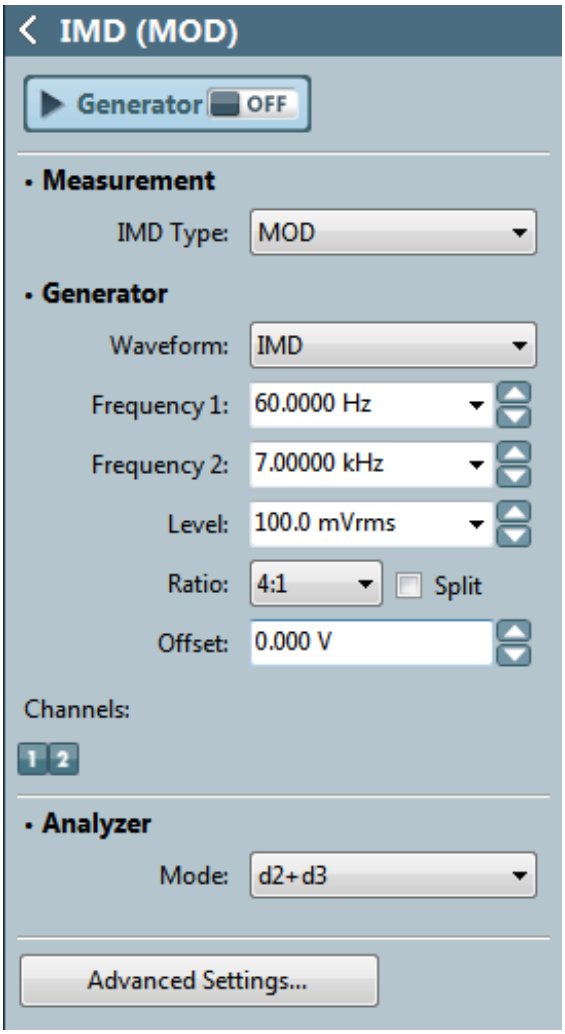
If you have not yet set up your test, first go to Signal Path Setup to select and configure inputs and outputs, references and other settings. See Chapter 7 for information about Signal Path Setup.

Measurement

IMD Type

Select the IMD type here. Read more about IMD beginning on page 305 for descriptions of the IMD types and stimulus frequencies.

- SMPTE
- DFD
- CCIF
- MOD



Generator

Waveform

The typical configuration for this measurement is the closed-loop configuration, using the APx generator as a stimulus. Alternatively, you can make open-loop measurements, using either the External Source or File Input configuration.

See Chapter 6 for detailed information about the using the APx Generator and making Waveform, Level and Frequency settings, or for information about using External Source.

If you have selected None (External Source) in Signal Path Setup Output Configuration, the Signal Generation controls listed below will be not be shown. For generator waveform files or external source use, the selected waveform file must be a correctly formed IMD stimulus corresponding to the selected IMD type. See Chapter 6, External Source, page 173 for information about Stimulus Signal Resources, and page 186 for information about Generator Waveforms.

Frequency

Set the IMD tone to be generated here.

- For **SMPTE** or **MOD**, set **Frequency 1** or **Frequency 2**.
  These measurements use dual-tone stimulus, with a lower, dominant frequency (**Frequency 1**) and a higher frequency of interest (**Frequency 2**). In a nonlinear device, the tones modulate each other. You can choose different frequency values here. The SMPTE and MOD stimulus frequency constraints are listed in the More on IMD discussion beginning on page 305.
- For **DFD** or **CCIF**, set **Mean Frequency** or **Diff Frequency**.
  DFD and **CCIF** use a dual-tone stimulus, with two high frequency tones separated by a **Difference Frequency**, centered around a **Mean Frequency**. You can choose different frequency values here. The DFD and CCIF stimulus frequency constraints are listed in the More on IMD discussion beginning on page 305.

Level

Set the Generator Level here.

Ratio

For **SMPTE**, select a **1:1** or **4:1** ratio (level ratio of **Freq 1** to **Freq 2**). For **MOD**, select **1:1, 4:1** or **10:1**.

Split

The **Split** setting defeats the summing of the two IMD frequencies within the generator. Instead, the signals are “split” and routed independently to channel outputs. **Frequency 1** or the **Mean Frequency** is routed to odd-numbered channels, and **Frequency 2** or the **Diff Frequency** is routed to even-numbered channels.

Acoustic testing of microphones sometimes uses two stimulus loudspeakers, each with a single frequency component of an IMD stimulus. The stimulus signals are summed acoustically at the microphone element. This method provides a distortion result for the microphone while minimizing the distortion contributions of the loudspeakers.

Offset

Set any DC offset here.

Analyzer

Filters

No local low pass, high pass or weighting filters are available for this measurement. However, low pass, high pass and weighting filter settings made in Signal Path Setup > Input/Output will affect this measurement. Read more about APx Filters in Chapter 97.

Mode

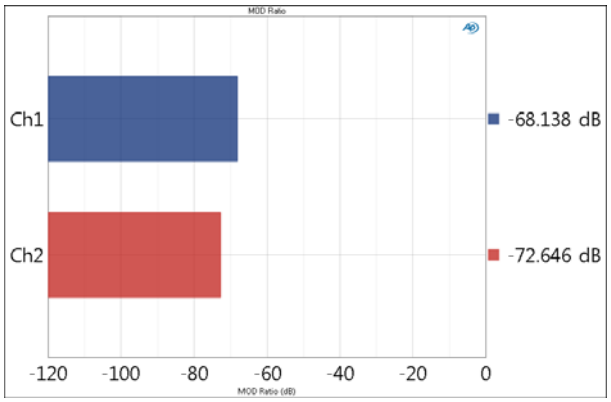
The Mode selector is not available for the SMPTE IMD measurement.

For **DFD**, **CCIF** and **MOD** sweeps, the **Mode** selector allows you to choose the distortion products or combinations of distortion products to be displayed. The default is d2 + d3 (2nd order plus 3rd order products).

Advanced Settings

If your test requires special settings, click **Advanced Settings**. Advanced Settings for IMD measurements include individual channel generator levels, custom IMD frequencies, individual channel analyzer ranging and custom analyzer settling. See Advanced Settings for IMD on page 306.

IMD Ratio



IMD Ratio provides a single-value result showing the IMD (intermodulation distortion) ratio in the output signal from each DUT channel, using the SMPTE, DFD, CCIF or MOD method.

Units

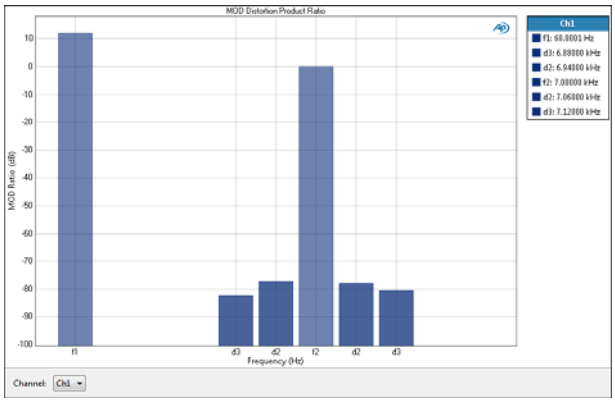
Units available for IMD Ratio are

Y-axis

- x/y
- %
- ppm
- dB

See Chapter 104 for more information about units of measurement.

IMD Distortion Product Ratio



IMD Distortion Product Ratio provides a single-value result that shows the individual second, third, fourth and fifth-order modulation products (d2 through d5) as a ratio to the level of the high frequency stimulus tone (f2), for a selected channel.

Result Settings

These settings are made in the Result Settings bar, beneath the graph display.

Channel

The Distortion Product Ratio result shows only one channel at a time. Select the channel to be viewed from the Channel drop-down list.

Units

Units available for IMD Distortion Product Ratio are

X-axis	*Y-axis*
• Hz	• Stimulus and distortion product tone bars
• dHz	
• F/R	
• %Hz	

See Chapter 104 for more information about units of measurement.

More about Intermodulation Distortion

IMD is an abbreviation for Intermodulation Distortion. IMD is created when two or more audio tones beat with one another (intermodulate) in a non-linear device to produce undesired new tones. The primary mechanism producing IM in most devices is AM (amplitude modulation), which creates sidebands that are at the sum and difference of the frequencies of the original audio tones.

For example, if a 2 kHz audio tone and a 7 kHz audio tone pass through a non-linear device and undergo AM intermodulation distortion, the output signal will contain new tones (modulation products) at 9 kHz (the sum of the tones) and at 5 kHz (the difference of the tones).

The modulation products may also beat with each other and with the original audio signal, creating more modulation products.

Measurements that have two tones in the stimulus are used to measure IMD. IMD measurements in APx500 include DFD, DIM, MOD, SMPTE and CCIF.

SMPTE IMD

SMPTE IMD is a technique for measuring IMD (intermodulation distortion) according to the SMPTE RP120-1983 standard. The DIN intermodulation distortion technique uses a similar method.

The stimulus is a strong low-frequency interfering signal (F1) combined with a weaker high frequency signal of interest (F2). F1 is usually 60 Hz and F2 is usually 7 kHz, at a ratio of F1:F2 = 4:1. The stimulus signal is the sum of the two sine waves. In a distorting DUT, this stimulus results in an AM (amplitude modulated) waveform, with F2 as the “carrier” and F1 as the modulation.

In analysis, F1 is removed, and the residual is bandpass filtered and then demodulated to reveal the AM modulation products. The rms level of the modulation products is measured and expressed as a ratio to the rms level of F2. The SMPTE IMD measurement includes noise within the passband, and is insensitive to FM (frequency modulation) distortion.

The APx500 implementation of SMPTE IMD provides the capability to vary the stimulus frequencies. F1 can be in the range of 40 Hz to 1 kHz; F2 can be in the range of 2 kHz to 60 kHz. However, F2 must always be at least 6 times greater than F1.

The level ratio F1:F2 can be set to the standard 4:1, or to 1:1.

Distortion products out to the 5th order are measured and reported in the distortion product result.

MOD IMD

MOD IMD is similar to SMPTE IMD, and in some cases produces the same results. The MOD stimulus is the same as the SMPTE stimulus, but instead of using AM demodulation for analysis MOD selectively measures the 2nd and 3rd order intermodulation products and combines their values arithmetically. This method

reduces the influence of noise in the result, and is sensitive to any distortion mechanism. When only amplitude modulation IMD exists, and when noise is very low, the MOD results will be identical to the SMPTE results.

The APx500 implementation of MOD IMD provides the capability to vary the stimulus frequencies. F1 can be in the range of 40 Hz to 1 kHz; F2 can be in the range of 240 Hz to 60 kHz. However, F2 must always be at least 6 times greater than F1.

The level ratio F1:F2 can be set to the standard 4:1, or to 10:1 or 1:1.

The 4th and 5th order products are also measured and can be reported in the distortion product result.

DFD IMD

DFD stands for Difference Frequency Distortion. DFD is described in the standards IEC60118 and IEC60268.

The DFD stimulus is two equal-level high-frequency tones f1 and f2, centered around a frequency called the mean frequency, (f1+f2)/2. The tones are separated by a frequency offset called the difference frequency. The two tones intermodulate in a nonlinear DUT to produce sum and difference frequencies.

The APx500 implementation of DFD IMD provides the capability to vary the stimulus frequencies. The mean frequency can be in the range of 250 Hz to 60 kHz; and the difference frequency can be in the range of 80 Hz to 2 kHz.

For analysis DFD selectively measures the 2nd and 3rd order intermodulation products, combines their values arithmetically and provides a result that is the ratio of the sum of the products to a reference voltage defined as 2x the voltage of f2 (effectively, the sum of f1 and f2). In the APx500 implementation, the 4th and 5th order products are also measured and can be reported in the distortion product result.

Because the stimulus tones are high in frequency, DFD is a useful measurement for observing distortion in devices that exhibit distortion that rises with frequency. Since the tones by default are only 80 Hz apart, much of the energy contained in the distortion products will fall near or below the stimulus tones. This makes DFD a good choice for measuring distortion at higher frequencies in band limited devices, where harmonic distortion products from high-frequency stimulus tones would fall out of band.

DFD measurements are made in the same way as CCIF measurements, differing only in amplitude calibration. DFD results are expressed as values 6.02 dB lower than CCIF.

CCIF IMD

The CCIF IMD method is described in document no. 11 of the Commission Mixte, CCIF/UIR, March 1937, issued by the International Telephonic Consultative Committee (CCIF). CCIF no longer exists as an organization, having become the ITU-R division of the International Telecommunications Union (ITU). This method is also referred to as IMD (ITU-R).

The CCIF stimulus is two equal-level high-frequency tones f1 and f2, centered around a frequency called the mean frequency, (f1+f2)/2. The tones are separated by a frequency offset called the difference frequency. The two tones intermodulate in a distorting DUT to produce sum and difference frequencies.

The APx500 implementation of CCIF IMD provides the capability to vary the stimulus frequencies. The mean frequency can be in the range of 250 Hz to 60 kHz; and the difference frequency can be in the range of 80 Hz to 2 kHz.

For analysis CCIF selectively measures the 2nd and 3rd order intermodulation products, combines their values arithmetically and provides a result that is the ratio of the sum of the products to a reference voltage defined as 2x the voltage of f2 (effectively, the sum of f1 and f2). In the APx500 implementation, the 4th and 5th order products are also measured and can be reported in the distortion product view.

Because the stimulus tones are high in frequency, CCIF is a useful measurement for observing distortion in devices that exhibit distortion that rises with frequency.

Since the tones by default are only 80 Hz apart, much of the energy contained in the distortion products will fall near or below the stimulus tones. This makes CCIF a good choice for measuring distortion at higher frequencies in band limited devices, where harmonic distortion products from high-frequency stimulus tones would fall out of band.

CCIF measurements are made in the same way as DFD measurements, differing only in amplitude calibration. CCIF results are expressed as values 6.02 dB higher than DFD.

Filters

Filters are not provided with IMD measurements, but APx Signal Path > Input > Filters can be applied. Any filtering in the range of the fundamentals or the distortion products would give non-standard results of questionable usefulness.

Advanced Settings for IMD

The default settings here are appropriate for most measurements. You may want to make minor adjustments for special situations.

Signal Generation

You can set different output levels or DC offsets for each channel here.

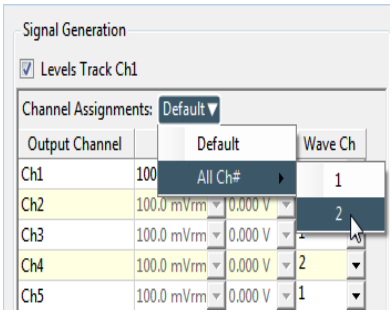
If **Levels Track Ch1** is checked (the default), the generator output level and DC offset values for channel 1 are copied to the other channels, and values for channels greater than 1 cannot be edited. Any changes made to channel 1 are reproduced in the other channels.

To set individual generator output channel levels or offsets, uncheck the **Levels Track Ch1** checkbox and enter values in the output channel Level or Offset fields.

If you are making an IMD measurement using the Split control, note that unchecking Levels Track Ch1 allows you to effectively change the ratio (4:1, for example) set in the Measurement Setup panel. This may create incorrect IMD measurement results.

Set Channel Assignments for generator waveforms

For stereo or multichannel generator waveform files, you can set channel mapping. By default, channel 1 audio is mapped to channel 1 output, channel 2 to channel 2, and so on. If the number of channels in the waveform file is less than the number of output channels, the waveform channels resume numbering at 1 and wrap to the next available output channel. These assignments can be remapped by changing individual settings in the **Wave Ch** column, or by selection one of several presets from the **Set Channel Assignments menu.**



Signal Acquisition and Analysis

Input Range tab

Digital inputs do not require ranging, and the Input Range tab is unavailable when inputs are set to a digital format.

Auto Range

If **Auto Range** is checked (the default), each analog channel determines its input range automatically, based on the level of the input signal. If the input signal level changes beyond a ranging threshold, **Auto Range** will cause the input ranging circuits to move up or down for proper ranging.

Fixed Range

Alternatively, you can set a fixed minimum range for the analog inputs. Uncheck **Auto Range** and enter the expected maximum level in the **Minimum Range** filed. The software will choose the lowest range that will accommodate that level; this becomes the minimum range available, and the analyzer will not reset to a lower range. Fixed range is best for signals that vary rapidly, including tone bursts, perceptual audio speech samples, and program material such as voice or music. See **When autoranging isn't suitable** on page 626.

If a signal greater then the **Minimum Range** is presented to the analog input, the ranging circuits will range up to accommodate the higher level, and then drop again when the signal level falls. The range will not be set below the **Minimum Range**.

Ranges Track Ch1

If **Ranges Track Ch1** is checked (the default), the fixed input range setting for channel 1 is copied to the other channels, and range settings for channels greater than 1 cannot be edited. Any changes made to channel 1 are reproduced in the other channels. If **Ranges Track Ch1** is not checked, the fixed input range setting for each channel can be set independently.

Read more about ranging in Chapter 98.

Settling tab

A Settling tab is available for IMD measurement results. See page 627 for more information about settling.

IMD Frequency Sweeps
(Sequence Mode)

Introduction

For the APx515, these measurements require a software option key. See page 189 for more information about software options.

IMD measurements use two tones of different frequencies summed into a stimulus signal. For IMD Frequency Sweep measurements, one of these tones is held at a fixed frequency while the other is swept through a range of frequencies. The result is displayed as an IMD Ratio result, with the swept frequency on the X-axis and the IMD Ratio on the Y-axis.

The IMD measurements in APx provide stimulus tones and analysis for four important methods of measuring intermodulation distortion: SMPTE, DFD, CCIF and MOD. Additionally, DIM IMD measurements are available with specific analyzer hardware.

Operation

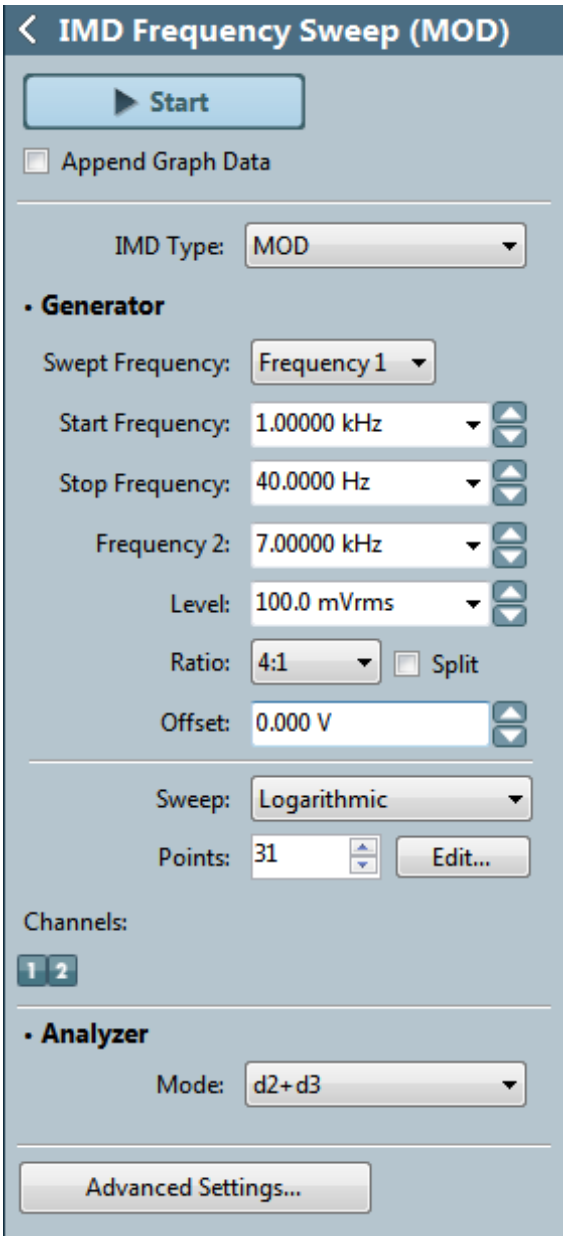
If you have not yet set up your test, first go to Signal Path Setup (Chapter 7).

IMD Frequency Sweep measurements are not available in External Source configuration.

IMD Type

Select the IMD type here. Read more about IMD beginning on page 305 for descriptions of the IMD types and stimulus frequencies.

- SMPTE
- DFD
- CCIF
- MOD



Generator

Swept Frequency

SMPTE or MOD

For SMPTE or MOD, select **Frequency 1** or **Frequency 2** as the Swept Frequency.

DFD or CCIF

For DFD or CCIF, select **Mean Frequency** or **Diff Frequency** as the Swept Frequency.

Start Frequency

Select the **Start** frequency of the swept tone here. This frequency is constrained only by the DAC generator constraints.

Stop Frequency

Select the **Stop** frequency of the swept tone here. This frequency is constrained only by the DAC generator constraints.

(Unswept) Frequency

For SMPTE or MOD, the control located below **Stop Frequency** will be labeled **Frequency 1** or **Frequency 2**.

The SMPTE and MOD stimulus frequency constraints are listed in the More on IMD discussion beginning on page 305.

For DFD or CCIF, the control located below **Stop Frequency** will be labeled **Mean Frequency** or **Diff Frequency**. In any case, enter the desired frequency for the unswept tone here.

The DFD and CCIF stimulus frequency constraints are listed in the More on IMD discussion beginning on page 305.

Level

Set the generator level here.

Ratio

For **SMPTE**, select a **1:1** or **4:1** ratio (level ratio of **Freq 1** to **Freq 2**). For **MOD**, select **1:1, 4:1** or **10:1**.

Split

The **Split** setting defeats the summing of the two IMD frequencies within the generator. Instead, the signals are “split” and routed independently to channel outputs. **Frequency 1** or the **Mean Frequency** is routed to odd-numbered channels, and **Frequency 2** or the **Diff Frequency** is routed to even-numbered channels.

Acoustic testing of microphones sometimes uses two stimulus loudspeakers, each with a single frequency component of an IMD stimulus. The stimulus signals are summed acoustically at the microphone element. This method provides a distortion result for the microphone while minimizing the distortion contributions of the loudspeakers.

Offset

Set any DC offset here.

Sweep

Select **Logarithmic, Linear** or **Custom**.

Points

For **Logarithmic** or **Linear**, set the number of points here.

Edit

Opens the **Sweep Points** editor. See page 507 for information about viewing and editing the **Sweep Points** table.

Step Size

For **Linear**, set the **Step Size** here.

Analyzer

Filters

No local low pass, high pass or weighting filters are available for this measurement. However, low pass, high pass and weighting filter settings made in Signal Path Setup > Input/Output will affect this measurement. Read more about APx Filters in Chapter 97.

Mode

The Mode selector is not available for the SMPTE Frequency Sweep.

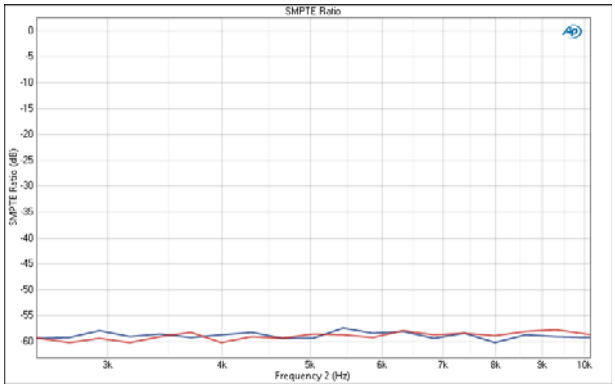
For **CCIF** and **MOD**, the **Mode** selector allows you to choose the distortion products or combinations of distortion products to be displayed. The default is d2 + d3 (2nd order plus 3rd order products).

Advanced Settings

If your test requires special adjustments or settings, click **Advanced Settings**. See page 351.

See Chapter 104 for more information about units of measurement.

IMD Ratio



IMD Ratio plots the distortion ratio on the Y axis against the Generator frequency on the X axis, using the SMPTE, DFD, CCIF or MOD IMD method.

Units

Units available for IMD Ratio are

X-axis (analog)	*Y-axis*
• Hz	• x/y
• dHz	• %
• F/R	• ppm
• %Hz	• dB

IMD Level Sweeps
(Sequence Mode)

Introduction

> *For the APx515, these measurements require a software option key. See page 189 for more information about software options.*

The SMPTE, DFD, CCIF and MOD Level Sweep measurements are combined into one measurement. Use the **IMD Type** menu to select **SMPTE, DFD, CCIF** or **MOD**.

IMD measurements use two tones of different frequencies summed into a stimulus signal. For IMD Level Sweep measurements, this stimulus is swept through a range of levels. The result is displayed as two IMD Ratio results, one with the swept generator level on the X-axis, the other with the measured level on the X-axis. Both plot the IMD Ratio on the Y-axis.

Operation

If you have not yet set up your test, first go to Signal Path Setup (Chapter 7).

> *IMD Level Sweep measurements are not available in External Source configuration.*

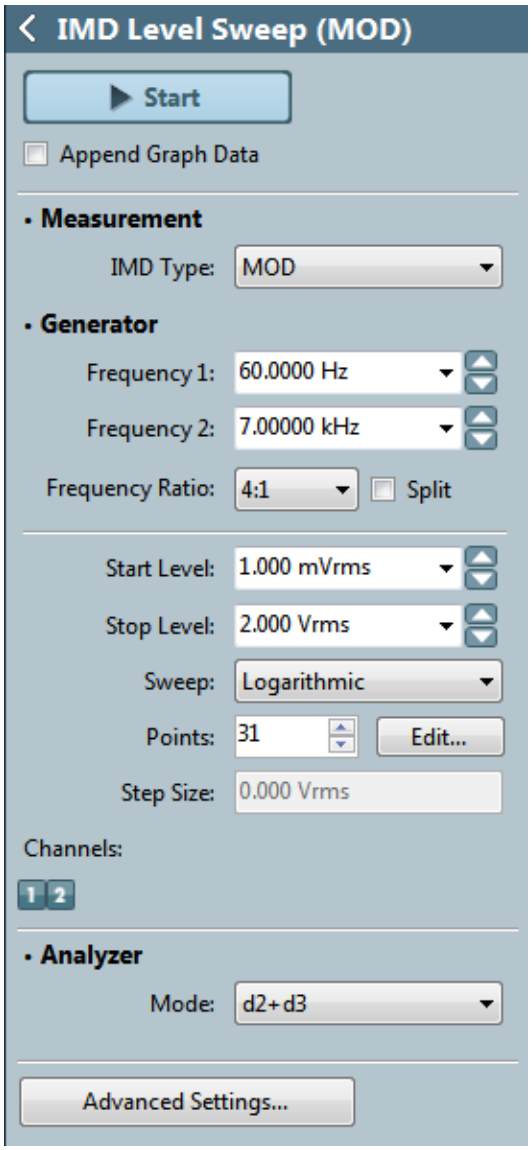
Append Graph Data

Normally, the current graph data for all results is deleted each time you start a new measurement. If the **Append** box is checked, the current data is kept in memory as a **Data Set** (see page 189), and the new measurement data is appended as a new data set. You can append many **Data Sets**.

Measurement

IMD Type

Select the IMD type here. Read more about IMD beginning on page 305.



- SMPTE
- DFD
- CCIF
- MOD

Generator

The configuration for this measurement must be the closed-loop configuration, using the APx sine generator as a stimulus. See Chapter 6 for information about the using the APx Generator and setting Waveform, Level and Frequency.

Frequency

Set the IMD stimulus here.

- For **SMPTE** or **MOD**, set **Frequency 1** or **Frequency 2**. These measurements use dual-tone stimulus, with a lower, dominant frequency (**Frequency 1**) and a higher frequency of interest (**Frequency 2**). In a nonlinear device, the tones modulate each other. You can choose different frequency values here.
  Frequency 1 minimum setting is 40 Hz. Maximum setting varies with the Frequency 2 setting, ranging from 333.33 Hz to 1 kHz. The default is 60 Hz.
  The **Frequency 2** setting range is 2 kHz (240 Hz for MOD) to 20 kHz. The default is 7 kHz.
- For **DFD** or **CCIF**, set **Mean Frequency** or **Diff Frequency**.
  DFD and **CCIF** use a dual-tone stimulus, with two high frequency tones separated by a **Difference Frequency**, centered around a **Mean Frequency**. You can choose different frequency values here.
  Mean Frequency minimum settings vary with the **Difference Frequency** setting, ranging from 2.5 kHz to 12 kHz. Maximum setting is 20 kHz. The default is 12.5 kHz.
  The **Difference Frequency** setting range is 80 Hz to 2 kHz. The default is 80 Hz.

Frequency Ratio

For **SMPTE**, select a **1:1** or **4:1** ratio (level ratio of **Freq 1** to **Freq 2**). For **MOD**, select **1:1, 4:1** or **10:1**.

Split

The **Split** setting defeats the summing of the two IMD frequencies within the generator. Instead, the signals are “split” and routed independently to channel outputs. **Frequency 1** or the **Mean Frequency** is routed to odd-numbered channels, and **Frequency 2** or the **Diff Frequency** is routed to even-numbered channels.

Acoustic testing of microphones sometimes uses two stimulus loudspeakers, each with a single frequency component of an IMD stimulus. The stimulus signals are summed acoustically at the microphone element. This method provides a distortion result for the microphone while minimizing the distortion contributions of the loudspeakers.

Running the measurement

To make an IMD Level Sweep measurement, click **Start**. The generator will output the IMD stimulus to the DUT on the selected generator channels. The level of the signal will be swept from the **Start Level** to the **Stop Level i**n the set number of points. You can select **Logarithmic** or **Linear** point spacing, or choose **Edit...** to customize the **Sweep Points** table.

Start Level

Select the **Start** level of the swept tone here.

Stop Level

Select the **Stop** level of the swept tone here.

Sweep

Select **Logarithmic, Linear** or **Custom**.

Points

For **Logarithmic** or **Linear**, set the number of points here.

Edit

Opens the **Sweep Points** editor. See page 512 for information about viewing and editing the **Sweep Points** table.

Step Size

For **Linear**, set the **Step Size** here.

Analyzer

No local low pass, high pass or weighting filters are available for this measurement. However, low pass, high pass and weighting filter settings made in Signal Path Setup > Input/Output will affect this measurement. Read more about APx Filters in Chapter 97.

Mode

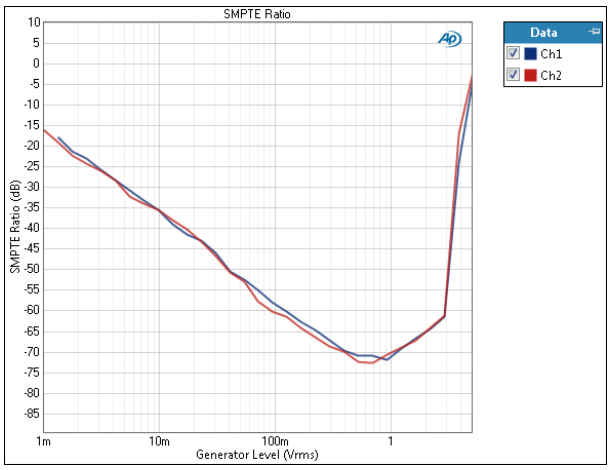
The Mode selector is not available for the SMPTE Frequency Sweep.

For **CCIF** and **MOD**, the **Mode** selector allows you to choose the distortion products or combinations of distortion products to be displayed. The default is d2 + d3 (2nd order plus 3rd order products).

Advanced Settings

If your test requires individual generator settings for each channel or special analyzer ranging or settling, click **Advanced Settings**. See Advanced Settings for Stepped Sweeps at the end of this chapter. See Chapter 104 for more information about units of measurement.

IMD Ratio vs. Generator Level



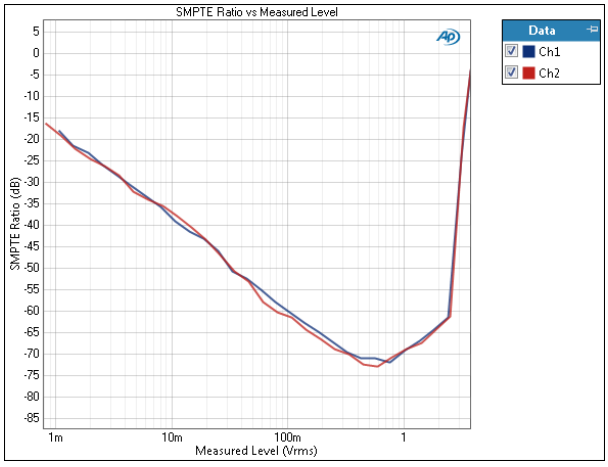
IMD Ratio plots the distortion on the Y axis against the Generator level on the X axis, using the SMPTE, DFD CCIF or MOD IMD method.

Units

Units available for the IMD Ratio result are

X-axis (analog)	*X-axis (digital)*	*Y-axis*
• Vrms	• FS	• x/y
• Vp	• %FS	• %
• Vpp	• dBFS	• ppm
• dBV	• dBrG	• dB
• dBu		
• dBrG		
• dBm		
• W (watts)		

IMD Ratio vs. Measured Level



IMD Ratio vs. Measured Level plots the distortion on the Y axis against the measured DUT output level on the X axis, using the SMPTE, DFD CCIF or MOD IMD method.

Units

Units available for the IMD Ratio vs. Measured Level result are

X-axis (analog)	*X-axis (digital)*	*Y-axis*
• Vrms	• FS	• x/y
• dBV	• %FS	• %
• dBu	• dBFS	• ppm
• dBrA	• dBrA	• dB
• dBrB	• dBrB	
• dBSPL1	• dBSPL1	
• dBSPL2	• dBSPL2	
• dBm)		
• W (watts)		

Advanced Settings for IMD Level Sweep Measurements

The default settings here are appropriate for most measurements. You may want to make minor adjustments for special situations.

Signal Generation

Generator output levels cannot be modified for this measurement.

Signal Acquisition and Analysis

Input Range tab

Digital inputs do not require ranging, and the Input Range tab is unavailable when inputs are set to a digital format.

Auto Range

If **Auto Range** is checked (the default), each analog channel determines its input range automatically, based on the level of the input signal. If the input signal level changes beyond a ranging threshold, **Auto Range** will cause the input ranging circuits to move up or down for proper ranging.

Fixed Range

Alternatively, you can set a fixed minimum range for the analog inputs. Uncheck **Auto Range** and enter the expected maximum level in the **Minimum Range** filed. The software will choose the lowest range that will accommodate that level; this becomes the minimum range available, and the analyzer will not reset to a lower range. Fixed range is best for signals that vary rapidly, including tone bursts, perceptual audio speech samples, and program material such as voice or music. See **When autoranging isn't suitable** on page 626.

If a signal greater then the **Minimum Range** is presented to the analog input, the ranging circuits will range up to accommodate the higher level, and then drop again when the signal level falls. The range will not be set below the **Minimum Range**.

Ranges Track Ch1

If **Ranges Track Ch1** is checked (the default), the fixed input range setting for channel 1 is copied to the other channels, and range settings for channels greater than 1 cannot be edited. Any changes made to channel 1 are reproduced in the other channels. If **Ranges Track Ch1** is not checked, the fixed input range setting for each channel can be set independently.

Settling tabs

Two or three Settling tabs are available for stepped sweep measurement results. These tabs may have different parameters and default values depending upon the result to which they're attached.

Read more about ranging and settling in Chapter 98.

Impedance/Thiele-Small
(Sequence Mode)

This measurement requires a software option key. See page 189 for more information about software options.

Impedance/Thiele-Small is only available when the analyzer input and output are set to an analog, non-acoustic configuration.

Overview

The Impedance/Thiele-Small measurement measures the complex impedance of a loudspeaker driver, providing impedance response curves and Thiele-Small parameters. The voltage across the driver is measured, and the current through the driver is calculated by measuring the voltage across a precision sense resistor in series with the driver. The underlying technology is the continuous sweep.

Impedance/Thiele-Small results available in APx500 are:

- Impedance Magnitude
- Impedance Real
- Thiele-Small parameters
- Impedance Phase
- Impedance Imaginary

These results are available from a single acquisition; however, depending upon the Thiele-Small mode selected, a second measurement may be necessary to obtain the full set of Thiele-Small parameters.

Thiele-Small parameters

The Thiele-Small (T-S) parameters are a set of electromechanical parameters that define the low-frequency performance of a loudspeaker driver.

T-S parameters are often referred to as “small signal” parameters, because the results are most useful when the driver remains in its linear range, not overdriven by a large signal. Driver specifications often include a table of T-S parameters to aid design engineers in driver selection and in enclosure design.

The T-S parameters are derived from measurements of a driver’s impedance curve, with additional information about the mechanical characteristics of the driver.

The additional information can be attained by modifying the mechanical loading of the driver and making a second measurement, or by entering the M_{MD} (moving mass) of the driver, measured outside of APx. These methods are discussed in detail below.

Symbol	*Unit*	*Description*
F_S	Hz	Resonant frequency of the driver
Q_{MS}	----	Mechanical *Q* at F_S
Q_{ES}	----	Electrical *Q* at F_S
Q_{TS}	----	Total *Q* at F_S
S_D	cm^2	Effective surface area of cone
R_E	Ω	Voice coil resistance
L_E	mH	Voice coil inductance
R_2	Ω	LR-2 model: voice coil parallel resistance
L_2	mH	LR-2 model: voice coil parallel inductance
E_{rm}	----	Wright model: motor resistance exponent
K_{rm}	----	Wright model: motor resistance coefficient
E_{xm}	----	Wright model: motor reactance exponent
K_{xm}	----	Wright model: motor reactance coefficient
R_{MS}	Ns/m	Mechanical resistance of suspension
C_{MS}	mm/N	Mechanical compliance of the suspension
M_{MS}	gram	Mechanical mass including air load
V_{AS}	liter	Acoustic volume with same compliance as suspension
Bl	Tm	Magnetic motor strength
η_0	%	Reference efficiency of the driver

R_2 and L_2 are only populated when the LR-2 model is selected. Erm, Krm, Exm and Kxm are only populated when the Wright model is selected.

Impedance Response

The impedance of a loudspeaker driver is given by dividing the voltage across the voice coil by the current passing through it. The reactance of the voice coil and the action of the mechanical system coupled to it do not present a pure resistance. When stimulated by an alternating current, the impedance is expressed as a complex number.

In the APx implementation, the impedance is calculated from a voltage measurement taken across the driver voice coil and the current in the driver circuit. The driver current is calculated by measuring the voltage across a sense resistor of known value, by I = E/R.

The stimulus signal is an audio sweep, resulting in an impedance response curve plotted against frequency. The underlying measurement technology is the APx continuous sweep.

Connections and Test Configurations

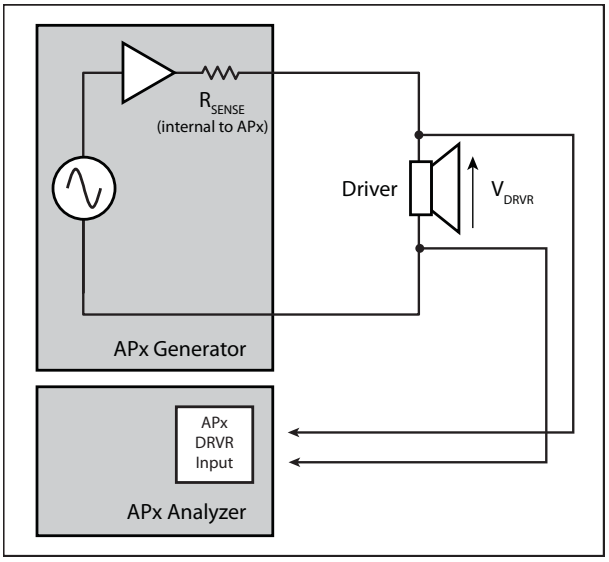
The Impedance/Thiele-Small measurement can use one of several test configurations to derive the driver current from the available voltages. Choose the test configuration suited to your situation.

In these tests, we are not measuring conventional audio parameters; we are measuring voltage or current in circuit components. Consequently, one or two of the analyzer input channels must have specific roles assigned, either to measure the voltage across the driver, the voltage across the sense resistor, or the total voltage across the driver and the sense resistor. Please pay close attention to the roles depicted in the illustrations below and identified on the **Channel** control in the software.

We recommend that you use balanced analyzer inputs for impedance measurements.

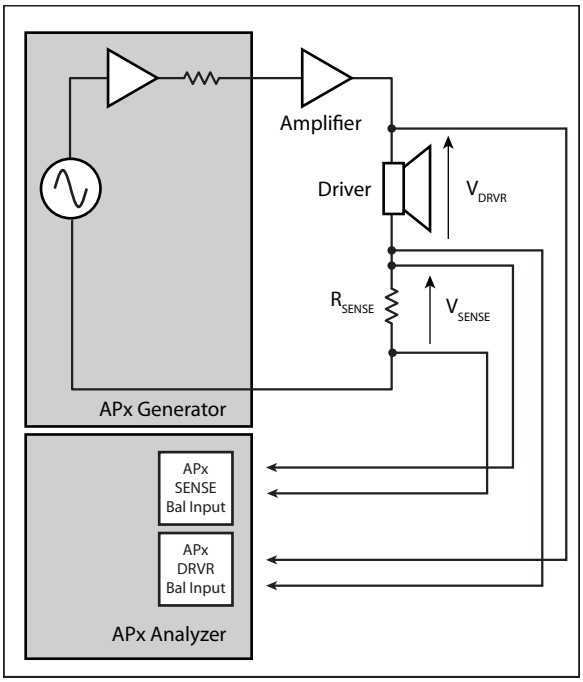
Choose the test configuration suited to your situation.

Internal



This method is convenient but less accurate than the External (2 Ch) configurations discussed below. It is appropriate for low-power demonstration purposes, or when no power amplifier or precision sense resistor is available. The Internal method uses the source resistor (a known value) in the final stage of the APx output circuitry as the current sense resistor. This method assumes that the APx directly drives the loudspeaker driver, and that no external power amplifier or sense resistor is used.

External (2 Ch) (bal)



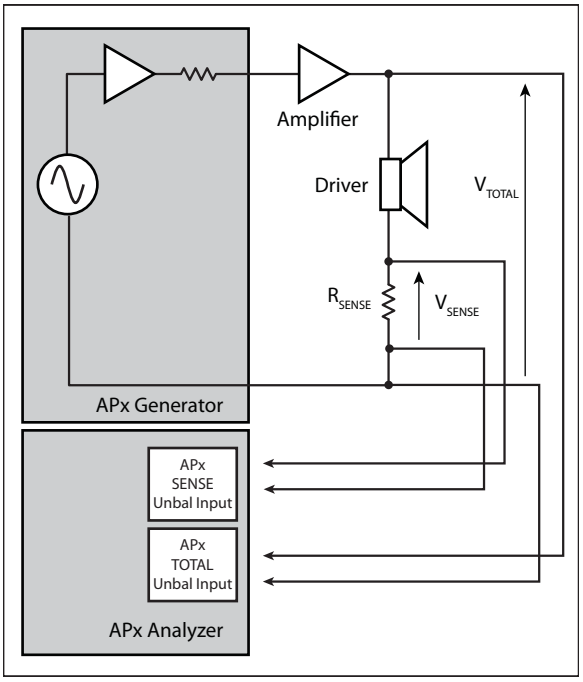
This is our recommended configuration. when making a 2-channel impedance measurement. This configuration is enabled when **Test Configuration** is set to **External (2 Ch)** and the analyzer inputs are set to balanced.

Because V_{DRVR} and V_{SENSE} are measured simultaneously, no assumptions are made regarding the amplifier response. Impedance results are reported correctly across the frequency range.

The Audio Precision IMP1 Impedance Fixture can be used with this configuration.

Avoid the use of a bridged amplifier as the accuracy of results can be significantly degraded due to strong common mode signals at the analyzer input.

External (2 Ch) (unbal)



External (2 Ch) unbal

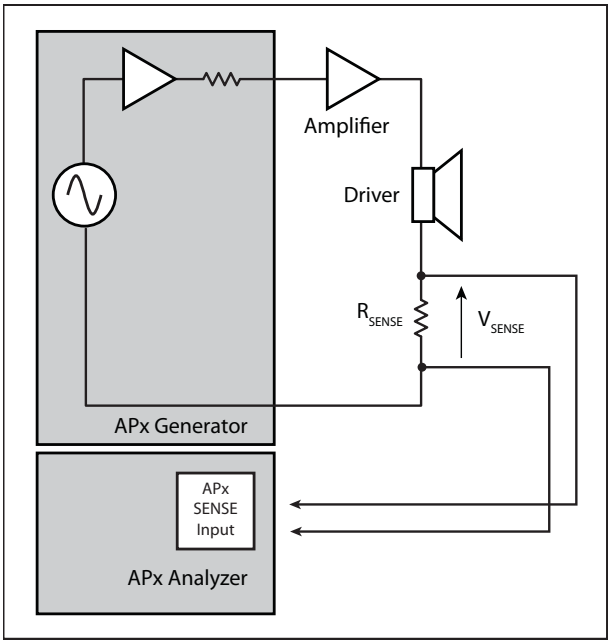
This configuration is enabled when **Test Configuration** is set to **External (2 Ch)** and the analyzer inputs are set to unbalanced.

This configuration is similar to External (2 Ch) (bal), but it allows the use of unbalanced analyzer inputs, which may be more convenient in some situations. V_{DRVR} is calculated by subtracting V_{SENSE} from V_{TOTAL}. When the analyzer is set to unbalanced inputs, this configuration is assumed and voltage measured at channel *n*+1 is subtracted from the voltage at channel *n*, provided the driver voltage.

The Audio Precision IMP1 Impedance Fixture can be used with this configuration.

A bridged amplifier cannot be used with unbalanced analyzer inputs.

External (1 Ch)



External (1 Ch)

This method uses an external power amplifier and sense resistor.

Set **Correction Curve** to select an amplifier correction curve in the project, or choose to **Create New** or **Browse for File**. If the correction curve accurately represents the amplifier in use, this configuration will make accurate impedance response measurements. See Amplifier Correction on page 325.

Enter the value of the sense resistor into the data entry field in APx. This is our recommended configuration when making a 1 channel impedance measurement.

If Correction Curve is set to **None**, the broadband gain of the amplifier must be known from a previous measurement, such as the APx Level and Gain measurement. Enter the gain and the value of the sense resistor into the data entry fields in APx.

When Correction Curve is set to **None**, this configuration cannot correct for the low-frequency rolloff present in an AC-coupled power amplifier. AC coupling will introduce phase shift that clearly affects the driver impedance curve. If your test requires a sweep that extends below 20 Hz, we recommend the use of a DC-coupled power amplifier, or that you use a correction curve as described above.

We recommend that you use balanced analyzer inputs for impedance measurements.

Avoid the use of a bridged amplifier as the accuracy of results can be significantly degraded due to strong common mode signals at the analyzer input.

Mounting the driver for testing

Free air

For the free-air measurement, the driver should be suspended or mounted to a rigid, open frame in free air, as far from reflecting surfaces as possible. The driver should be oriented so that the line of travel of the voice coil is perpendicular to the force of gravity.

Loading the driver for a second measurement

Unless you know the M_{MD} of your driver (see Known M_{MD}, below), the full set of Thiele-Small parameters requires a second measurement, where the driver has been loaded with either added mass or the air compliance of a sealed enclosure.

Added Mass

The driver should be mounted as for a free-air measurement, with mass added to the driver in a balanced pattern that does not interfere with the motion of the cone. This is typically done by placing a uniform toroid of modeling clay (such as Plasticine or Handi-TAK) around the driver dust cap.

Adding mass to the driver will shift its resonant frequency (the peak in the impedance curve) downward. You must add enough mass to shift the resonant frequency by at least 5%. APx accepts a range of 1 g to 1000 g.

Known Volume

The driver should be mounted on a sealed enclosure, in the same orientation to the Earth as indicated in the free air instructions.

If the driver is facing in to the enclosure, calculate the total volume by adding the volume of the enclosure to the volume of the truncated conic section formed by the driver cone.

If the driver is mounted inside a sealed enclosure facing out, you must subtract the volume of the driver cone, frame and magnet assembly from the volume of the enclosure.

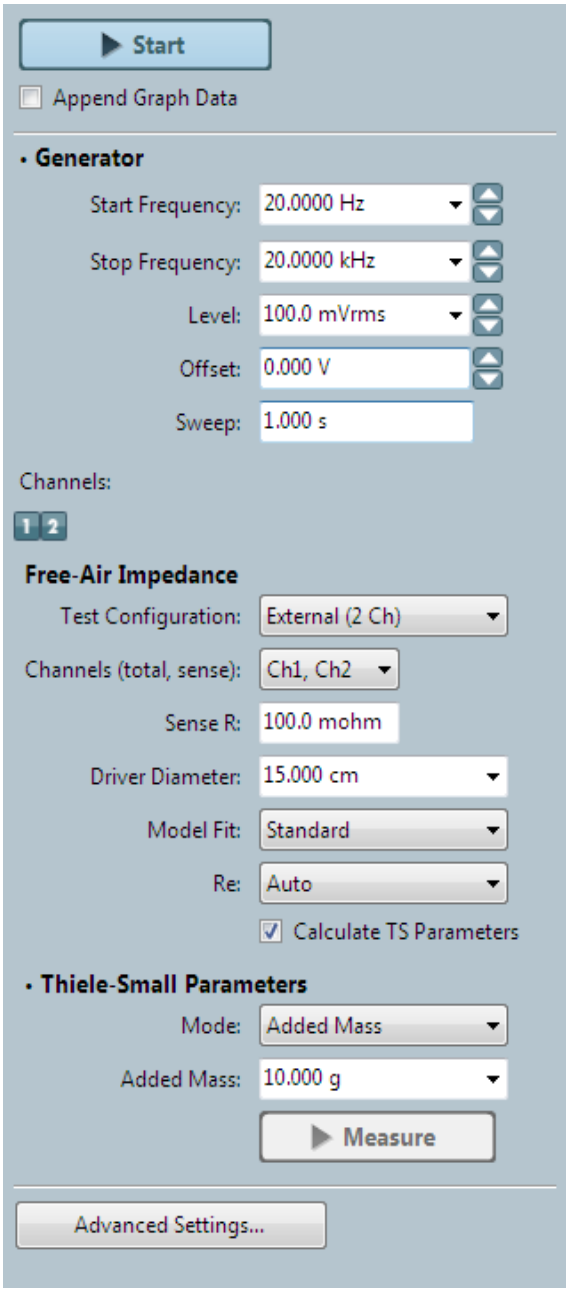
Mounting the driver on a sealed enclosure will shift its resonant frequency (the peak in the impedance curve) upward by reducing the compliance of the system; the smaller the enclosure, the more the resonant frequency is shifted. You must reduce the compliance enough to shift the resonant frequency by at least 5%. APx accepts a range of 0.01 liter to 1000 liters.

Known M_{MD}

Rather than making two measurements for T-S as described above, you can enter the M_{MD} (the moving mass of the driver without air load) and derive the full set of parameters in one measurement pass. The M_{MD} may be available from a manufacturer's specification, or can be obtained by disassembling an identical driver and weighing the moving parts. Enter the M_{MD} into the **Known M_{MD}** field.

The **Known M_{MD}** technique has the advantage of faster and more precise measurements.

Common Controls



Operation

If you have not yet set up your test, first go to Signal Path Setup (Chapter 7).

Note: be certain to disable all input filters for this measurement. Filtering the acquisition will result in incorrect impedance

curves. For low-frequency drivers, be sure that the input high-pass filter is set to DC (DC coupled).

Before you run the measurement

Before you run the measurement, you must choose a Test Configuration and enter a number of values as described in detail below. If you are using the **Known M_{MD}** mode, you must choose that mode in the Thiele-Small Parameters area, and enter the exact value of the M_{MD} in the **Mmd** entry field before running the measurement.

Append Graph Data

Normally, the graph data in memory is deleted each time you start a new measurement. If the **Append Graph Data** box is checked, appended data are grouped as additional data sets, as are any imported data. The Data Sets Panel displays the Data Set(s) in the measurement, and provides a visibility checkbox and a Notes field for each data set.

For the Impedance/Thiele-Small measurement, only the Free-Air Impedance data (the first measurement) can be appended as new Data Sets (Measured 1, Measured 2, etc.). The data acquired in the second Thiele-Small Parameters acquisition (for Added Mass or Known Volume) are applied to the final measured Data Set, and are available for viewing as additional traces on that graph. The second measurement can be run repeatedly, updating the second set of traces but not appending new Data Sets.

Generator

The configuration for this measurement must be the closed-loop configuration, using the APx sine generator as a stimulus. See Chapter 6 for information about the using the APx Generator and setting Waveform, Level and Frequency.

If you have not yet set up your test, first go to Signal Path Setup (Chapter 7).

The Impedance/Thiele-Small measurement uses a continuous sweep (log chirp) stimulus signal, swept between the frequencies entered in **Start Frequency** and **Stop Frequency**, at the value set in **Level**.

Note: Loudspeakers and drivers under test can be damaged by signals that exceed their level or frequency limits. Be sure to set generator and amplifier level and sweep frequencies to values within the limits of the loudspeaker.

Click **Start**. This plots the impedance response curves and populates the Thiele-Small parameter values calculated from the impedance measurement. If you have elected to use the **Known M_{MD}** mode, this will be the full set of T-S parameters. Otherwise, a second measurement pass (see Thiele-Small Parameters, below) is required.

Start Frequency and Stop Frequency

These controls set the frequency range of the sweep. For loudspeaker driver testing, the sweep range is typically set to the nominal range of the driver. See page 246 for more about continuous sweep ranges, which are dependent upon the instrument.

Level

Set the generator level here.

Offset

Set any DC offset to the generator signal here.

Sweep

A continuous sweep stimulus consists of a sine wave swept logarithmically across the sweep range in the period of time set in **Sweep**.

The default for **Sweep** is 1 s. Minimum is 50.0 ms; maximum is 5 s.

Longer **Sweep** settings provide greater resolution and signal-to-noise ratio, but increase measurement time.

Analyzer: Free-Air Impedance

Test Configuration

This measurement supports three test configurations. The selection made here affects the availability of other settings below. The test configurations are:

- **Internal (not available when using the APx1701 Transducer Test Interface)**
- **External (2 Ch)**
- **External (1 Ch)**

See page 318 for a full description of these configurations.

Internal

See a diagram of the Internal test configuration on page 318.

This method is convenient but less accurate than the "External (2 Ch)" configuration discussed below. It is appropriate for low-power demonstration purposes, or when no power amplifier or precision sense resistor is available.

Channel

The Internal Test Configuration requires 1 input channel, which is assigned the role of "drvr". This channel measures the voltage across the loudspeaker driver.

No external power amplifier or external sense resistor is used.

External (2 Ch)

This is our recommended configuration. It requires an external power amplifier and an external sense resistor. It requires 2 analyzer input channels, but because both the voltage across the driver and the current through the driver are measured simultaneously, it produces the best results.

Channel

Balanced inputs

See a diagram of the External (2 Ch) (bal) test configuration on page 318.

When using balanced inputs, the External (2 Ch) Test Configuration requires one input channel to be assigned the role of “drvr”. This channel measures the voltage across the loudspeaker driver.

It requires a second input channel to be assigned the role of “sense”. This channel measures the voltage across the sense resistor.

Channel roles are assigned in adjacent pairs, with “drvr” always the lower numbered channel.

Unbalanced inputs

See a diagram of the External (2 Ch) (unbal) test configuration on page 319.

When using unbalanced inputs, the External (2 Ch) Test Configuration requires one input channel to be assigned the role of “total”. This channel measures the total voltage across the loudspeaker driver and the sense resistor. This is the same as the amplifier output voltage.

It requires a second input channel to be assigned the role of “sense”. This channel measures the voltage across the sense resistor.

Channel roles are assigned in adjacent pairs, with “total” always the lower-numbered channel. The voltage across the driver is calculated by subtracting the sense voltage from the total voltage.

Sense R:

Enter the value of the external sense resistor in this field.

External (1 Ch)

See a diagram of the External (1 Ch) test configuration on page 319.

This configuration allows you to make an impedance measurement using an external power amplifier and an external sense resistor, but requires only 1 analyzer input channel. You must use an amplifier correction curve (see below) or provide a broadband amplifier gain value. We recommend using an amplifier correction curve, as it is more accurate. Characteristics of the amplifier used and the accuracy of the correction curve will affect your results.

Correction Curve:

- **None**
  If you are not using a correction curve, choose **None** and enter a value for the amplifier gain in the field below. The response of your amplifier will affect the accuracy of the impedance curves. See External (1 Ch) Test Configuration on page 319 for more information.
- If there are Amplifier Correction Curves in the project, they will be listed here and available for selection.
- **Create New**
  To create an amplifier correction curve, choose Create New. See **Amplifier Correction** on page 325.
- **Browse for file**
  To use a previously saved amplifier correction curve, choose Browse for file.See **Amplifier Correction** on page 325.

A correction curve cannot be applied when using the APx1701 Transducer Test Interface.

Amplifier Gain

This field is only available when **Test Configuration** is set to **External (1 Ch)** configuration and **Correction Curve** is set to **None**. Determine the gain of your amplifier using a measurement such as the APx Level and Gain, and enter the value here. See External (1 Ch) Test Configuration on page 319 for more information.

Amplifier gain is fixed at 20 dB when using the APx1701 Transducer Test Interface.

Channel

The **External (1 Ch)** Test Configuration requires one input channel to be assigned the role of “sense.” This channel measures the voltage across the sense resistor.

Sense R:

Enter the value of the external sense resistor here. The acceptable range is 0.01 ohm to 1,000 ohm.

Note: The APx1701 Transducer Test Interface has an internal 0.1 ohm sense resistor for each channel. If you are using the APx1701, the value in Sense R: will be fixed at 0.1 ohm.

Driver Diameter

Enter the driver diameter in this field.

Model Fit

After the preliminary set of Thiele-Small parameters are obtained from the free air pass, the result data are processed and a model of the impedance curve is created.

You can choose to view the raw result data, or the processed data using the **Unprocessed / Processed** control on the graph display. The choice of **Unprocessed / Processed** viewing does not affect the operation of the measurement.

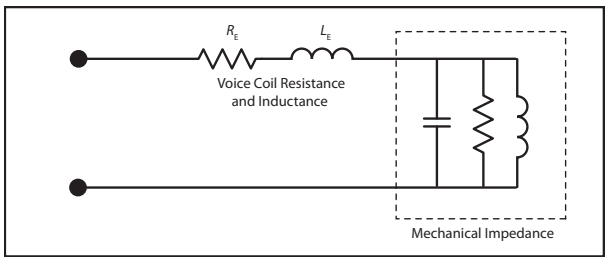
Note: If the ratio between the sense resistor and the driver under test is large (small sense resistor / large driver impedance), very little voltage will be developed across the sense resistor, and the result data may be very noisy. The Unprocessed view may show a great deal of noise, and in severe cases the impedance curve of the driver under test may not fit the available models. See Calculate TS Parameters on the next page.

Choose a Model Fit:

- Standard
- **LR-2** or
- Wright

Standard model

The standard model provides a good fit at lower frequencies but may not represent the physical driver well at higher frequencies.



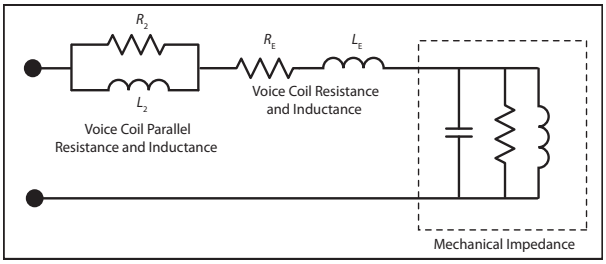
Standard Model

The moving mass of the driver is modeled as a resonant system, shown as the dotted box on the right. All moving coil drivers have a voice coil with resistance and inductance. The Standard Model assumes that the simple series resistance plus inductance of the voice coil accurately models the electrical impedance of the driver.

In a practical system, eddy current losses in the magnet and pole piece cause the real part of the impedance to climb with frequency, which the Standard Model does not predict.

LR-2 model

The LR-2 and Wright models add an additional network of resistance and inductance to the standard model and typically provide a better fit at high frequencies.

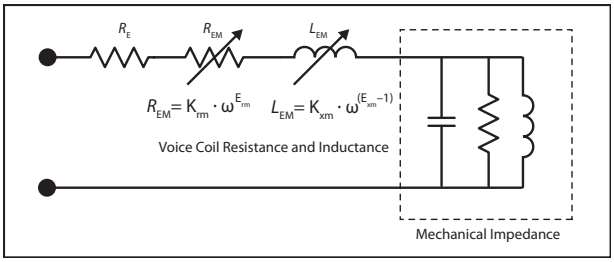


LR-2 Model

The LR-2 Model assumes that the network shown above in series with the mechanical impedance accurately models the electrical impedance of the coil, including eddy current losses. This model provides a more accurate fit to the measured impedance curve than the Standard Model, and can be implemented with physical components or digital filters.

For more information, see Mark Dodd, Wolfgang Klippel, and Jack Oclee-Brown, "Voice Coil Impedance as a Function of Frequency and Displacement", AES 117th Convention, Oct 28-31 2004. This paper is available from the Audio Engineering Society eLibrary at aes.org.

Wright model



Wright Model

The Wright Model assumes that the network shown above in series with the mechanical impedance accurately models the electrical impedance of the coil, including eddy current losses. This model provides a very accurate fit to the measured impedance curve, but cannot be implemented with physical components or digital filters because it uses unrealizable parameters, such as fractional resistance and inductance. With certain software tools, the Wright model is useful in crossover and enclosure design.

For more information, see J. R. Wright, "An Empirical Model for Loudspeaker Motor Impedance", JAES, Vol. 38, No. 10, October 1990. This paper is available from the Audio Engineering Society eLibrary at aes.org.

R_E

R_E is a Thiele-Small parameter representing the DC resistance (DCR) of the voice coil. Choose

- **Auto**
  Auto obtains R_E in finding the best model fit to the impedance curve.
- **Fixed (DCR)**
  Fixed (DCR) allows you to enter the DC resistance of the voice coil, measured outside of APx using a precision ohmmeter.

DCR

Enter the resistance value in the **DCR** field.

Calculate TS Parameters

This checkbox is checked by default, enabling the calculation of Thiele-Small parameters from the impedance measurement.

In some cases (such as high-impedance headphone measurements), the impedance curve of the driver under test may not fit the available models. Derivation of the Thiele-Small parameters will fail, and the measurement will abort. To prevent this, uncheck the **Calculate TS Parameters** checkbox. The device impedance curve will be measured, but Thiele-Small parameters and model fit will not be calculated.

Curve Fit Range

For calculating Thiele-Small parameters, it is best to constrain the high-frequency extent of the range across which the data is fitted to the model curve to approximately 20xFs (Fs is the resonant frequency of the driver).

- **Auto**
  The data that is fitted to the model curve is the full frequency range of the acquisition, which is approximately the same as the range of the generator sweep.
- **Custom**
  The data that is fitted to the model curve is constrained to the range between the **Min Freq** and **Max Freq**, set in the fields below.

Analyzer: Thiele-Small Parameters

Mode

The **Mode** selected in Thiele-Small parameters determines whether or not a second measurement pass is required, and what further information must be entered for the second pass. If you are using the **Known M_{MD}** mode, you must choose that mode and enter the exact value of the M_{MD} in the **Mmd** entry field before you click **Start**. See Known M_{MD} on page 320.

For **Added Mass** or **Known Volume**, you must first run the free-air impedance measurement, as described above in **Running the Measurement**.

When that measurement has been made, choose the driver loading mode for Thiele-Small: **Added Mass** or **Known Volume**.

For **Known M_{MD}**, only the first free-air measurement is necessary.

Added Mass

If you do not know the M_{MD} of the driver, you can choose the **Added Mass** mode to fully populate the table of T-S parameters. You must first run the free-air impedance measurement initiated by the **Start** button, as described above in **Running the Measurement**. For **Added Mass**, you must add mass to the driver as described in **Mounting the driver for testing: Free air** on page 320. This will shift the resonance frequency of the driver downward. Enter the exact value for the added mass in the **Added Mass** field and click **Measure**.

Known Volume

If you do not know the M_{MD} of the driver, you can choose the **Known Volume** mode to fully populate the table of T-S parameters. You must first run the free-air impedance measurement initiated by the **Start** button, as described above in **Running the Measurement**. For **Known Volume**, you must mount the driver to a sealed enclosure of known volume, as described in **Mounting the driver for testing: Free air** on page 320. This will shift the resonance frequency of the driver upward. Enter the exact value of enclosure volume in the field provided and click **Measure**.

Known M_{MD}

If you know the M_{MD} (moving mass) of the driver under test, set **Mode** in Thiele-Small parameters to **Known M_{MD}**, enter the exact value of the M_{MD} in the **Mmd** entry field, and click **Start** as described above in **Running the Measurement**. In this case, only the one free-air measurement is necessary to obtain the full set of Thiele-Small parameters. See **Known M_{MD}** in **Mounting the driver for testing: Free air** on page 320.

Using the **Known M_{MD}** mode is not always possible. If you do not know the M_{MD} (moving mass) from a specification, you must dismantle the driver and weigh its moving components.

Measure

For **Added Mass** or **Known Volume** modes, after the impedance response measurement initiated by the **Start** button is complete, you must make a second measurement to obtain the full set of T-S parameters. Click **Measure** to run the second measurement pass.

Advanced Settings

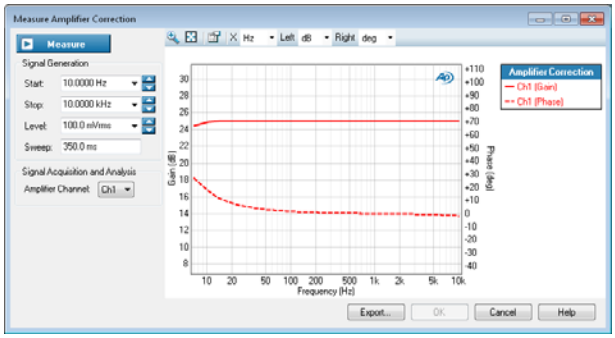
Advanced Settings include output level and offset settings and input ranging settings. To view or modify these settings, click **Advanced Settings**. See Advanced

Settings for Continuous Sweep based measurements on page 249.

Amplifier Correction

For accurate 1-channel impedance response measurements, the amplifier's gain response and phase response must be known.

The **Measure Amplifier Correction** dialog provides a convenient way to make this measurement and attach the correction curve to the project.



Signal Generation

Select a sweep range, level and duration.

Connections and Signal Acquisition and Analysis

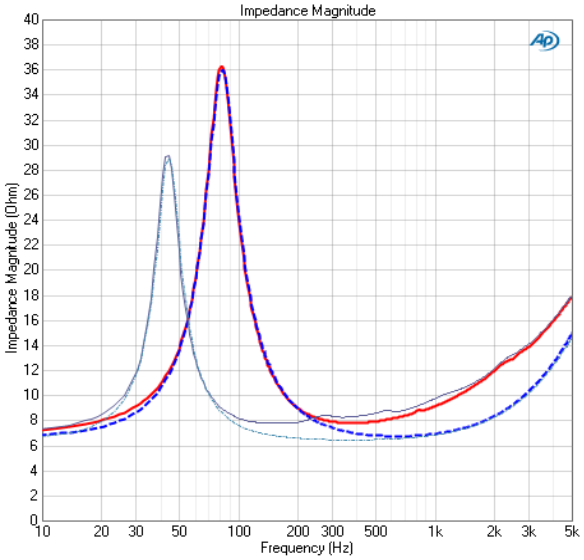
In the **Measure Amplifier Correction** dialog, select the channel to be used for measurement. Connect the amplifier input to the APx generator output for this channel, and the amplifier output to APx analyzer input for this channel.

Click **Measure**. APx will measure and display the amplifier's gain response and phase response.

Click **Export** to save the amplifier correction curve outside the project as an *.xls or *.csv file.

Click **OK** to leave the dialog. You will be prompted to name the amplifier correction curve, which will then be attached to the project.

Impedance Magnitude



This result provides a graphical display of the impedance response of the driver, plotting the complex impedance magnitude versus frequency. This trace is has been **Processed** for viewing using decimation and smoothing. The model fit (Standard model) curve is also plotted on the same graph, shown here as a dashed line.

If a second measurement is made using the **Added Mass** or **Known Volume** driver loads, the loaded trace and its model fit are plotted on the same graph. The example here (and in the other results) was made by adding mass to the driver, producing the traces with the lower magnitude and lower resonance frequency.

Unprocessed/Processed

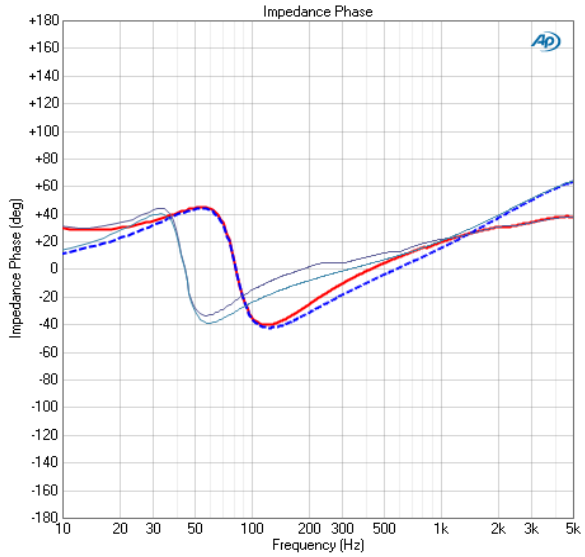
You can choose to view the raw, unprocessed result data, or the processed result data using the **Unprocessed / Processed** control on the graph display. The choice of **Unprocessed / Processed** viewing does not affect the operation of the measurement.

Units

Units available for Impedance/Thiele-Small Impedance Magnitude are:

X-axis	*Y-axis*
• Hz	• ohms
• dHz	
• F/R	
• %Hz	

Impedance Phase



This result provides a graphical display of the impedance phase response of the driver, plotting the impedance phase versus frequency. This trace is has been **Processed** for viewing using decimation and smoothing. The dashed line is the model fit trace, here using the Standard model. Traces for **Added Mass** are also shown.

Unprocessed/Processed

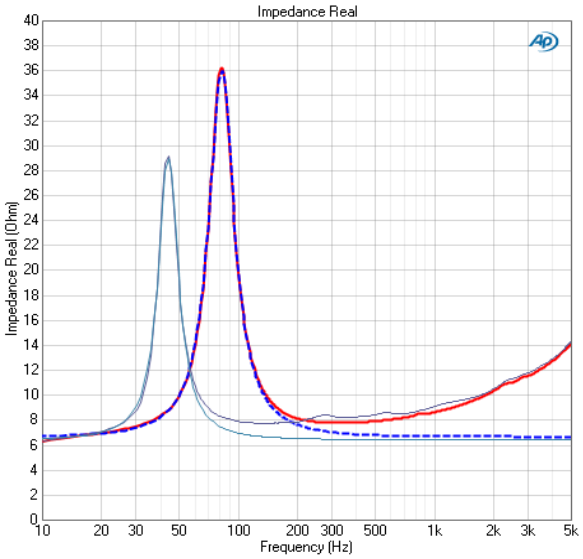
You can choose to view the raw, unprocessed result data, or the processed result data using the **Unprocessed / Processed** control on the graph display. The choice of **Unprocessed / Processed** viewing does not affect the operation of the measurement.

Units

Units available for Impedance/Thiele-Small Impedance Phase are:

X-axis	*Y-axis*
• Hz	• deg
• dHz	• rad
• F/R	
• %Hz	

Impedance Real



Driver impedance is a complex number, with both real and imaginary components. This result provides a graphical display of the real component of the impedance response of the driver, plotting the real impedance value versus frequency. This trace is has been **Processed** for viewing using decimation and smoothing. The dashed line is the model fit trace, here using the Standard model. Traces for **Added Mass** are also shown.

Unprocessed/Processed

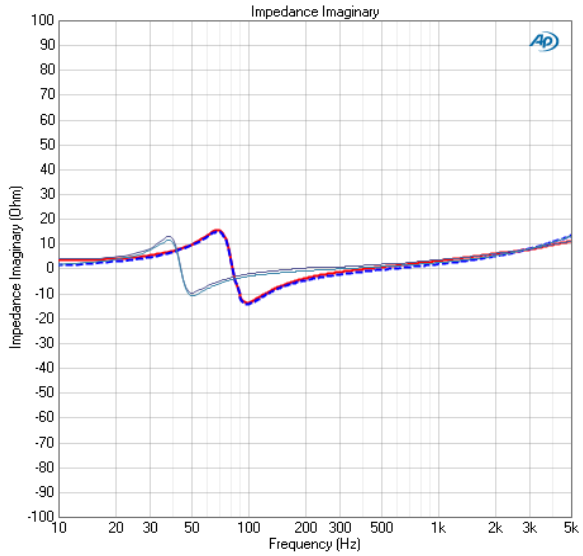
You can choose to view the raw, unprocessed result data, or the processed result data using the **Unprocessed / Processed** control on the graph display. The choice of **Unprocessed / Processed** viewing does not affect the operation of the measurement.

Units

Units available for Impedance/Thiele-Small Impedance Real are:

X-axis	*Y-axis*
• Hz	• ohms
• dHz	
• F/R	
• %Hz	

Impedance Imaginary



Driver impedance is a complex number, with both real and imaginary components. This result provides a graphical display of the imaginary component of the impedance response of the driver, plotting the imaginary impedance value versus frequency. This trace is has been **Processed** for viewing using decimation and smoothing. The dashed line is the model fit trace, here using the Standard model. Traces for **Added Mass** are also shown.

Unprocessed/Processed

You can choose to view the raw, unprocessed result data, or the processed result data using the **Unprocessed / Processed** control on the graph display. The choice of **Unprocessed / Processed** viewing does not affect the operation of the measurement.

Units

Units available for Impedance/Thiele-Small Impedance Magnitude are:

X-axis	*Y-axis*
• Hz	• ohms
• dHz	
• F/R	
• %Hz	

Analysis: Thiele-Small Parameters

This result populates a table with Thiele-Small parameter values calculated from the impedance response measurement. If you have elected to use the **Known M_{MD}** mode, or if you have made the second measurement in **Added Mass** or **Known Volume** mode, this will be the full set of T-S parameters.

If the resonant frequency of the driver has not moved down (for **Added Mass**) or up (for **Known Volume**) at least 5%, an error message will be shown, suggesting that you add more mass, or reduce the enclosure volume.

Otherwise, a single free-air impedance response measurement will only populate a subset of the Thiele-Small parameters.

Parameter	Value	Lower Limit	Upper Limit
F_S	50.50 Hz	----	----
Q_{MS}	2.07	----	----
Q_{ES}	0.56	----	----
Q_{TS}	0.44	----	----
S_D	452.39 cm^2	----	----
R_E	5.51 Ω	----	----
L_E	0.38 mH	----	----
R_2	---- Ω	----	----
L_2	---- mH	----	----
K_{rm}	----	----	----
E_{rm}	----	----	----
K_{xm}	----	----	----
E_{xm}	----	----	----
R_{MS}	14.85 Ns/m	----	----
C_{MS}	---- mm/N	----	----
M_{MS}	---- g	----	----
V_{AS}	---- l	----	----
Bl	---- Tm	----	----
η_o	---- %	----	----

Thiele-Small parameters (partial)

The parameters C_{MS}, M_{MS}, V_{AS}, Bl, and η_0 are not yet populated. Also, R_2 and L_2 are only populated when the LR-2 model is selected, and E_{rm}, K_{rm}, E_{xm} and K_{xm} are only populated when the Wright model is selected.

Parameter	Value	Lower Limit	Upper Limit
F_S	50.50 Hz	----	----
Q_{MS}	2.07	----	----
Q_{ES}	0.56	----	----
Q_{TS}	0.44	----	----
S_D	452.39 cm^2	----	----
R_E	5.51 Ω	----	----
L_E	0.38 mH	----	----
R_2	---- Ω	----	----
L_2	---- mH	----	----
K_{rm}	----	----	----
E_{rm}	----	----	----
K_{xm}	----	----	----
E_{xm}	----	----	----
R_{MS}	14.85 Ns/m	----	----
C_{MS}	0.79 mm/N	----	----
M_{MS}	12.63 g	----	----
V_{AS}	228.30 l	----	----
Bl	6.27 Tm	----	----
η_0	4.99 %	----	----

Thiele-Small parameters (complete)

R_2 and L_2 are only populated when the LR-2 model is selected, and E_{rm}, K_{rm}, E_{xm} and K_{xm} are only populated when the Wright model is selected.

Setting Limits for Thiele-Small parameters

Parameter	Value	Lower Limit	Upper Limit
F_S	55.37 Hz	53.0	56.0
Q_{MS}	2.27	2.00	2.25
Q_{ES}	0.63	----	----
Q_{TS}	0.50	----	----

Thiele-Small parameters with limits applied (detail)

You can set arbitrary limits for any or all of the Thiele-Small parameters by entering limit values into the results grid. Results that fall outside of the limits will be highlighted in the grid, and in a sequence the measurement will be flagged as **Failed**.

Input Sample Rate
(Sequence Mode)

The Input Sample Rate measurement provides a single-value meter result, the measured sample rate of the incoming digital audio signal. Signal Path Setup Input/Output must be set to a digital input.

Generator

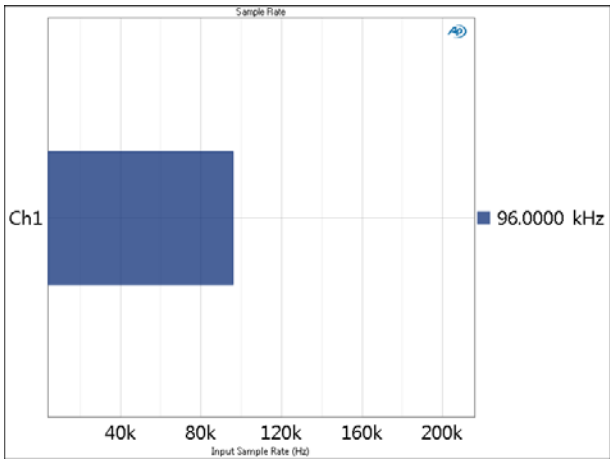
See Chapter 6 for detailed information about the using the APx Generator and making Waveform, Level and Frequency settings, or for information about using External Source.

Advanced Settings

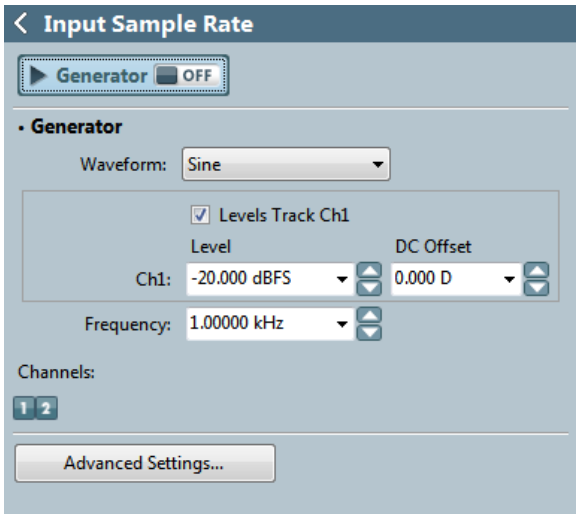
Advanced Settings makes available individual channel output level and offset controls, and input ranging and settling controls. See Advanced Settings for single value measurements on page 351.

See Chapter 104 for more information about units of measurement.

Sample Rate



Input Sample Rate provides a single-value meter result, displaying of the measured sample rate at the digital receiver.



Units

Units available for Input Sample Rate are

- Hz
- dHz
- F/R
- %Hz

Interchannel Phase
(Sequence Mode)

The Interchannel Phase measurement provides a single-value result that shows the relative phase of the DUT channels at a single frequency. One channel is chosen as the phase reference channel, and the remaining channels are measured against it. For more information about phase, see page 332.

Operation

If you have not yet set up your test, first go to Signal Path Setup to select and configure inputs and outputs, references and other settings. See Chapter 7 for information about Signal Path Setup.

Generator

The typical configuration for this measurement is the closed-loop configuration, using the APx generator as a stimulus. Alternatively, you can make open-loop measurements, using either the External Source or File Input configuration.

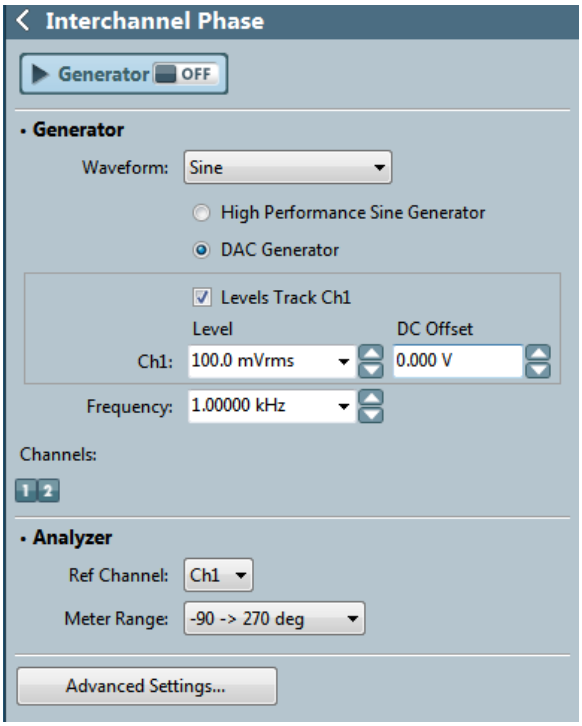
See Chapter 6 for detailed information about the using the APx Generator and making Waveform, Level and Frequency settings, or for information about using External Source.

High Performance Sine Generator / DAC Generator

For an APx555, you have the option of using the High Performance Sine Generator (very low distortion sine waves and wide bandwidth) or the DAC generator, which provides special sine waveforms. See High Performance Sine Generator starting on page 51 for a comparison of features.

Running the measurement

To measure Interchannel Phase, click the Generator button to **On**. The generator will output the test signal to the DUT on the selected generator channels. Read the Interchannel Phase for each channel from the meter bar display. Channel phase is relative to the selected Reference Channel.



Analyzer

No local low pass, high pass or weighting filters are available for this measurement. However, low pass, high pass and weighting filter settings made in Signal Path Setup > Input/Output will affect this measurement.

Ref Channel

The phase of each channel is measured against the selected Reference Channel. There is no phase result for the Reference Channel itself, and no data is displayed for that channel.

Meter Range

You can set the phase meter to examine the phase results in one of three meter ranges:

0 > 360 deg

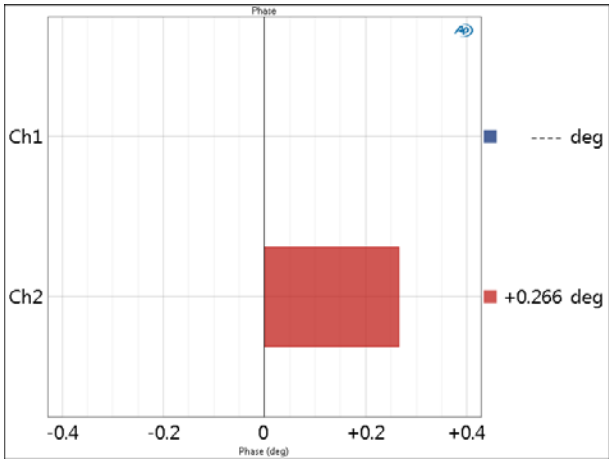
–90 > 270 deg

–180 > 180 deg

Advanced Settings

If your test requires individual generator settings for each channel or special analyzer ranging or settling, click **Advanced Settings**. See Advanced Settings for single value measurements on page 351.

Phase



Phase provides a single-value meter result, displaying the phase between the selected reference channel and the other input channels.

Units

Units available for Interchannel Phase are

- deg
- rad

See Chapter 104 for more information about units of measurement.

Measuring Interchannel Phase when configured for External Source

Operation with an external source is slightly different from the operation described above, which uses the internal generator. First, be sure you are configured for External Source. Go to Signal Path Setup and select **None (External)** for the output connector in Output Configuration.

When measuring from an external source, instead of using the generator you will play an audio signal from the DUT. For Interchannel Phase, the audio signal must be periodic and have two or more channels. See External Source Test Signals on page 173.

Reading the results is the same: the phase difference for each channel is shown on the meter bar display. Channel phase is relative to the selected Reference Channel. The meters will read interchannel phase as long as the external source plays the signal.

See Chapter 6 for more information about External Source measurements.

Running the measurement in File Input configuration

See Analyzing Audio Files in Chapter 22 for more information about File Input configuration.

If you have chosen **File** for an input configuration in Signal Path Setup, first click the **File List...** button to add input file(s) to a list for analysis.

If your DUT is to be recording the WAV file, select a generator waveform and turn the generator **ON**, as described above. Configure your DUT to save a WAV file(s) whose file name(s) and location match the name specified in the File List.

If you are measuring an existing file (not recording from the generator but working in External Source), be sure that the file name(s) and location match the name(s) specified in the File List.

Finally, click the **Analyze** button. The first settled reading (typically within the first 0.10 second of the file) is shown on the meter bar display. To take a reading later in the file, go to the **Settling** tab in **Advanced Settings** and increase the **Delay Time** to the desired point in the file.

More about Phase

The acoustic vibrations that are sound and the electrical vibrations that represent sound as audio signals are waves.

At a given moment in time, two waves of the same frequency can be at the same point in their wave cycle (in phase) or at some different point in the cycle (out of phase). Waves are referenced to circular motion, and a complete cycle of a wave is 360 degrees of angle. 0 degrees, 360 degrees and any integer multiple of 360 all represent the same point on a circle, and all describe signals that are in phase.

Differences in phase may be observed between any two signals. In audio testing, phase is typically measured between channels in multichannel devices, and between the input and output of a device under test.

Jitter Frequency Sweep
(Sequence Mode)

Jitter generation and measurement requires an analyzer fitted with the Advanced Master Clock (AMC), standard in an APx555 and an option for APx525, 526, 582, 585, and 586.

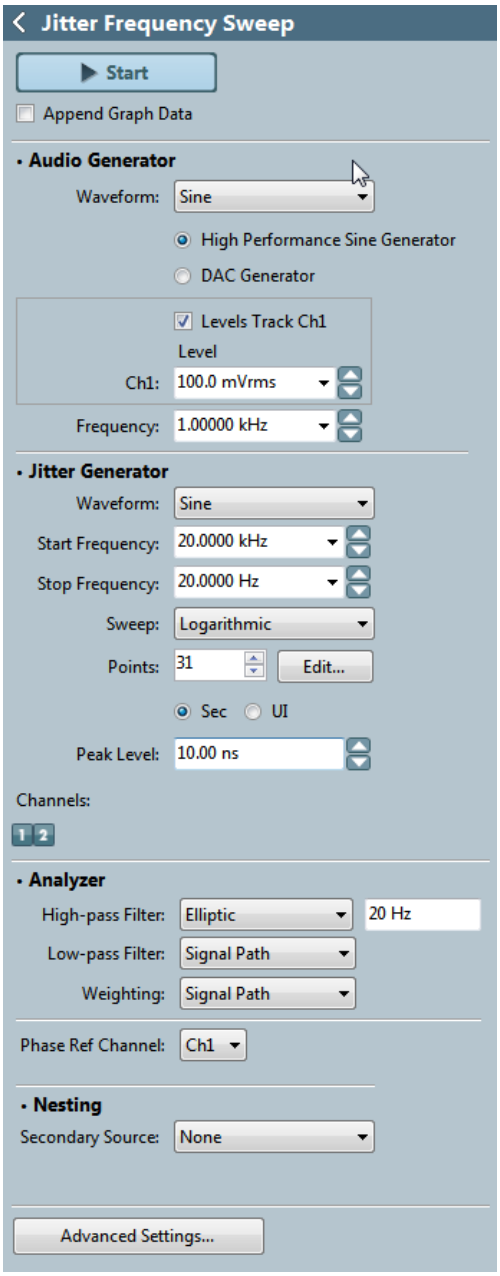
Generated jitter can be applied to an audio stream or clock if the instrument is fitted with a jitter-enabled I/O module, such as ADIO, jitter-enabled DSIO or jitter-enabled PDM. Alternatively, generated jitter can be applied to the AMC Reference Out and Sync Out signals.

Jitter on incoming signals can be analyzed if the instrument is fitted with a jitter-enabled I/O module, such as ADIO, jitter-enabled DSIO or jitter-enabled PDM.

Results for Jitter Frequency Sweep include:

• RMS Level	• Deviation	• THD Level
• Gain*	• Phase	• Distortion Product Ratio
• Relative Level	• THD+N Ratio	• Distortion Product Level
• Peak Level	• THD+N Level	• SINAD
• Average Jitter Level*	• THD Ratio	

* These results will only be populated if the analyzer input is set to measure the jitter signal. See Measuring Audio or Jitter, below.



Overview

The Jitter Frequency Sweep stimulates the DUT with a digital signal. The embedded audio waveform is selected in Audio Generator. The timebase for the digital signal is modulated by the jitter generator, which is swept across a frequency range.

The results are either

- measurements of the audio output of the DUT (analog or digital), which is typically degraded when the timebase is jittered; or

- for DUTs with a digital output, direct measurements of the jitter in the DUT output signal, rather than the embedded audio signal. Select **Audio** or **Jitter** in Signal Path Setup > Input/Output > Input Configuration.

Operation

If you have not yet set up your test, first go to Signal Path Setup to select and configure inputs and outputs, references and other settings. See Chapter 7 for information about Signal Path Setup.

Additionally, you must choose where to apply the jitter, and whether to measure the audio or the jitter on the incoming signal, as described next.

Applying jitter

Go to the **Clocks** panel in Signal Path Setup, and select whether you would like to apply the jitter sweep to a digital output, or to the Reference Out and Sync Out connectors.

Select **Apply To**:

- **Digital Output**
  If you select **Digital Output**, you must also go to Signal Path Setup > Input/Output and select a digital output that supports jitter generation.
- **Reference/Sync**
  If you select **Reference/Sync**, you must connect the Reference Out or the Sync Out to the appropriate sync input on your DUT.

For a jitter sweep, you do not need to **Enable** jitter on the **Clocks** panel. Jitter is automatically enabled for the duration of the sweep.

Measuring Audio or Jitter

For a DUT with a digital output, go to Signal Path Setup > **Input/Output** to choose a digital input for the APx analyzer.

Select **Measure**:

- **Audio**
  Audio de-embeds the audio signal from the digital interface signal and routes it to the analyzer. This is normal APx operation.
- **Jitter (UI)** or **Jitter (sec)**
  Jitter routes the digital interface signal to jitter demodulator and a dedicated ADC (analog to digital converter), and then to the analyzer.

For a default sweep of 20 kHz to 20 Hz in 31 logarithmic steps, simply click **Start**. If you want to specify other parameters for your sweep, review Generator and Analyzer settings, discussed below. Click **Start** when you have approved the settings.

Append Graph Data

Normally, the current graph data for all results is deleted each time you start a new measurement. If the **Append** box is checked, the current data is kept in memory as a **Data Set** (see page 189), and the new measurement data is appended as a new data set. You can append many **Data Sets**.

Audio Generator

The Audio Generator generates the audio signal that is embedded in the digital interface signal, as in most APx measurements.

Waveform, Frequency and Level

The configuration for this measurement must be the closed-loop configuration, using the APx generator as a stimulus.

See Chapter 6 for detailed information about the using the APx Generator and making Waveform, Level and Frequency settings.

Jitter Generator

The Jitter Generator generates the signal that modulates the jitter clock, which is used as the timebase for the digital interface signal used in this measurement.

Waveform

Select **Sine** or **Square**.

Start Frequency

Set the sweep start frequency here. The available range is 2 Hz to 200 kHz.

Stop Frequency

Set the sweep stop frequency here. The available range is 2 Hz to 200 kHz.

Sweep

Choose

- **Logarithmic** (logarithmic point spacing)
- **Linear** (linear point spacing)
- **Custom**
  Edit sweep spacing and number of **Points** to create a **Custom** sweep.

Points

Set the number of sweep points here. The default is 31 points.

Points > Edit

Open the **Sweep Points** dialog to edit, import or export the **Sweep Points** table.

Step Size

For linear sweeps, enter the **Step Size** here.

Peak Level

Set the jitter level here. Note that the level is expressed in peak units of time, seconds or UI, selectable below the **Peak Level** field.

Seconds / UI

Choose the jitter units here.

Channels

Toggle specific output channels on or off by clicking on the **Channel** number button.

EQ

The generated jitter signal can be equalized. Go to Signal Path Setup > **Clocks** > **Jitter Generator** to apply an EQ curve. See the Jitter Generator topic in **Clocks**. beginning on page 69.

Analyzer

Note that for this measurement, the analyzer can measure either the audio signal (de-embedded from the interface signal) or the jitter (demodulated from the interface signal), as selected in Signal Path Setup > **Input/Output**.

Filters

Note that these filters will affect the audio signal measurements or the jitter signal measurements, as selected in Signal Path Setup > Input/Output.

Local high-pass, low-pass and weighting filters are available for this measurement. The low-pass and high-pass filters, if selected, are applied to the entire measured signal and affect all the results. The weighting filter, if selected, affects only the distortion and distortion plus noise results. See Measurement Filters on page 619 for detailed information about the filters available locally for some measurements.

Low pass, high pass and weighting Signal Path Filters settings (see page 617) made in Signal Path Setup > Input/Output will also affect this measurement. When set, the local low-pass or high-pass filter setting overrides the corresponding Signal Path filter setting while the measurement is active. A local weighting filter setting does not override any Signal Path filter settings; the local weighting filter is applied in addition to any active Signal Path filters.

Phase Ref Channel

Interchannel Phase measurements must be referenced to one channel. Set the phase reference channel here.

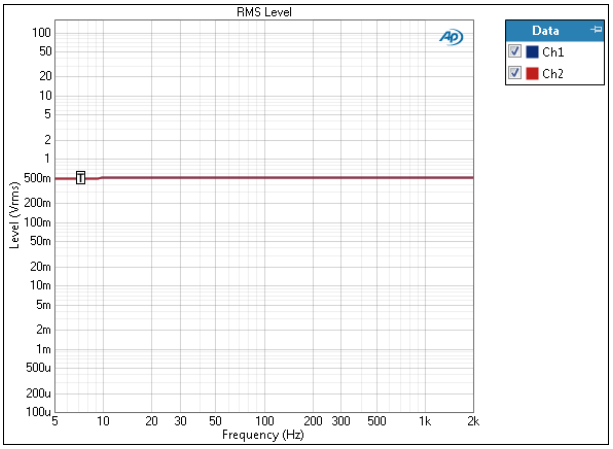
Nesting

Jitter Frequency sweeps can be nested. Read about Nested Sweeps beginning on page 185.

Advanced Settings

If your test requires special adjustments or settings, click Advanced Settings.

RMS Level results



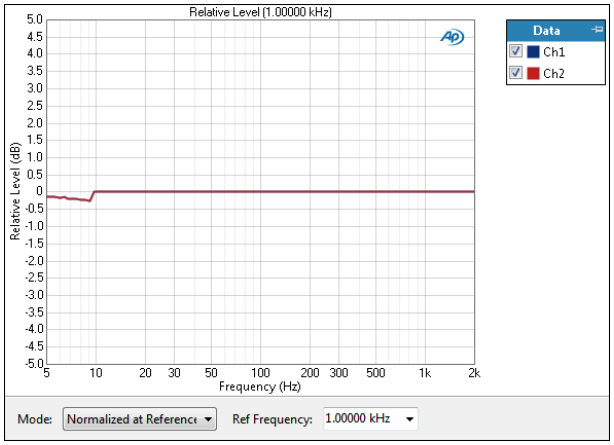
Jitter Frequency Sweep: RMS Level result uses a fixed audio stimulus signal while the jitter generator is moved across a range of frequencies in a specified number of points.

Units

Units available for Jitter Frequency Sweep: RMS Level results are:

X-axis	*Y-axis (analog)*	*Y-axis (digital)*	*Jitter signals*
• Hz	• Vrms	• dBFS	• UI
• dHz	• dBV	• FS	• dBUI
• F/R	• dBu	• %FS	• s
• %Hz	• dBrA	• dBrA	
	• dBrB	• dBrB	
	• dBSPL1	• dBSPL1	
	• dBSPL2	• dBSPL2	
	• dBm		
	• W (watts)		

Jitter Frequency Sweep: Relative Level



The Jitter Frequency Sweep: Relative Level result uses a fixed audio stimulus signal while the jitter generator is moved across a range of frequencies in a specified number of points. Additionally, in this result the response curve is plotted in relation to the measured level at a selected jitter frequency.

Mode

Select one of two result display modes.

Normalized at Reference

Enter a reference frequency for normalization in the **Ref Frequency** field. The measured level at the jitter frequency you choose is set as 0 dB on the graph, and the DUT response is plotted in relation to this reference. You can change the **Ref Frequency** at any time (except after appending) and the graph will immediately be redrawn to reflect the new setting.

Auto-Centered in Limits

This control is only usable if upper and lower limits are set on this graph. In some applications, the general conformance of a trace to the shape of a set of limits is of importance, with absolute level less important or irrelevant. **Auto-Centered in Limits** centers the data trace relative to the upper and lower limits. To exclude possible exceptional data points at frequency extremes, the frequency range that is considered for the centering operation is constrained by the values in the **Min** and **Max** fields.

Result Settings

These settings are made in the Result Settings bar, beneath the graph display.

Ref Frequency

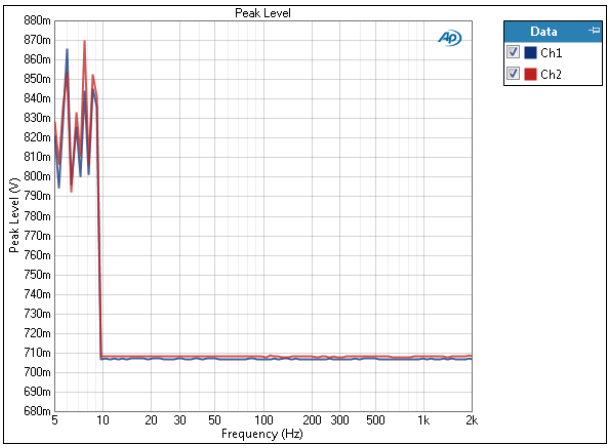
Choose a Reference Frequency here.

Units

Units available for Jitter Frequency Sweep: Relative Level are:

X-axis	*Y-axis*
• Hz	• x/y
• dHz	• %
• F/R	• ppm
• %Hz	• dB

Peak Level



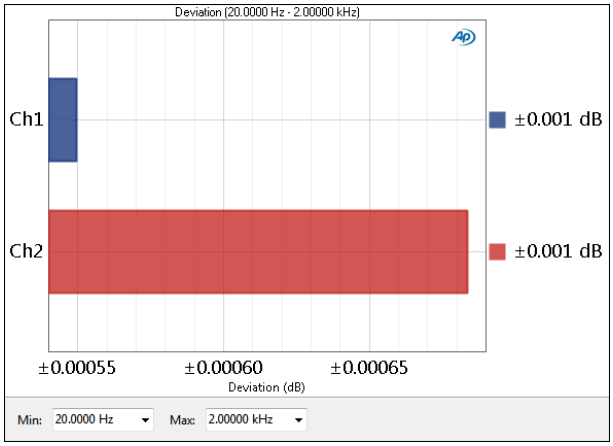
Jitter Frequency Sweep: Peak Level result uses a fixed audio stimulus signal while the jitter generator is moved across a range of frequencies in a specified number of points. This is a frequency response sweep, with peak-scaled results.

Units

Units available for Jitter Frequency Sweep Peak Level results are:

X-axis	*Y-axis (analog)*	*Y-axis (digital)*	*Jitter*
• Hz	• V	• D	• UI
• dHz		• hex	• dBUI
• F/R			• s
• %Hz			

Deviation



The Jitter Frequency Sweep: Deviation result uses a fixed audio stimulus signal while the jitter generator is moved across a range of frequencies in a specified number of points. This result is a single value measurement computed from the stepped sweep that shows the level deviation (the total range of level variation) of each channel as a meter bar. You can specify a minimum and maximum frequency to define the range of the deviation measurement.

Result Settings

These settings are made in the Result Settings bar, beneath the graph display.

Min

Set the minimum frequency of the range of interest here.

Max

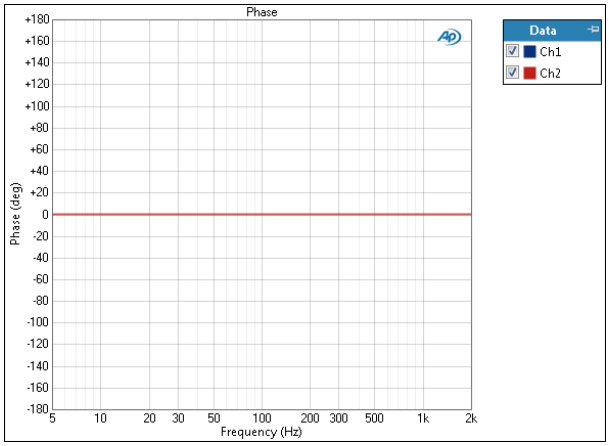
Set the maximum frequency of the range of interest here.

Units

Units available for Jitter Frequency Sweep: Deviation are

- dB

Phase



The Phase result uses a fixed audio stimulus signal while the jitter generator is moved across a range of frequencies in a specified number of points. One channel is chosen as the phase reference channel, and the remaining channels are measured against it.

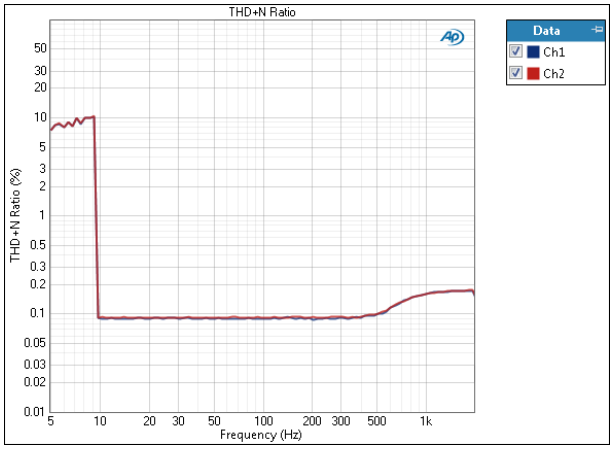
Select a phase reference channel in **Phase: Ref Channel**. The remaining channels are plotted against the selected channel.

Units

Units available for Jitter Frequency Sweep: Phase are:

X-axis	*Y-axis*
• Hz	• deg
• dHz	• rad
• F/R	
• %Hz	

THD+N Ratio



The THD+N Ratio result uses a fixed audio stimulus signal while the jitter generator is moved across a range of frequencies in a specified number of points. In this result the ratio of the level of the THD+N (total harmonic distortion plus noise) to the total signal in the DUT output is plotted against frequency.

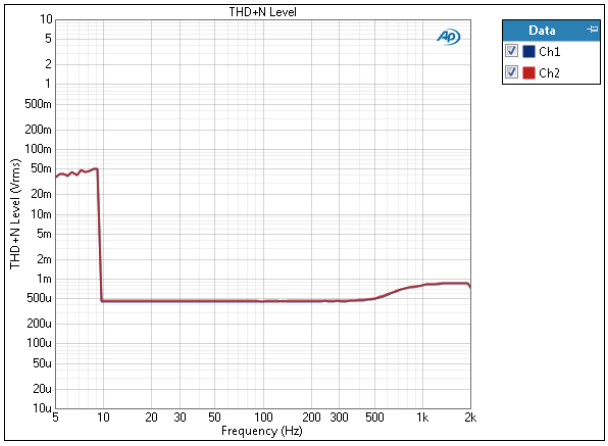
See page 527 for more information about THD+N.

Units

Units available for Jitter Frequency Sweep: THD+N Ratio are:

X-axis	*Y-axis*
• Hz	• x/y
• dHz	• %
• F/R	• ppm
• %Hz	• dB

THD+N Level



The THD+N Level result uses a fixed audio stimulus signal while the jitter generator is moved across a range of frequencies in a specified number of points. In this result the level of the THD+N (total harmonic distortion plus noise) in the DUT output is plotted against frequency.

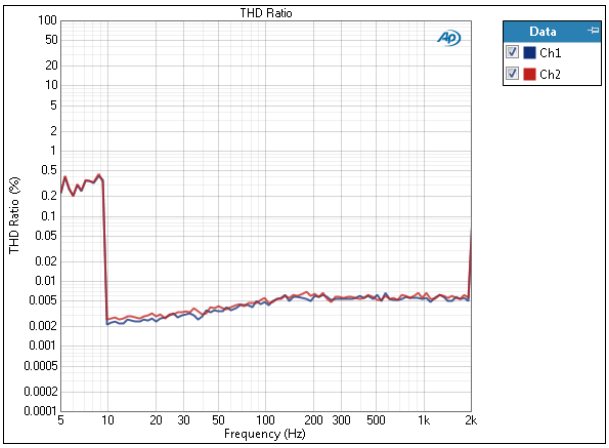
See page 527 for more information about THD+N.

Units

Units available for Jitter Frequency Sweep: THD+N Level are:

X-axis	*Y-axis (analog)*	*Y-axis (digital)*	*Jitter signals*
• Hz	• Vrms	• dBFS	• UI
• dHz	• dBV	• FS	• dBUI
• F/R	• dBu	• %FS	• s
• %Hz	• dBrA	• dBrA	
	• dBrB	• dBrB	
	• dBSPL1	• dBSPL1	
	• dBSPL2	• dBSPL2	
	• dBm		
	• W (watts)		

THD Ratio



The THD Ratio result uses a fixed audio stimulus signal while the jitter generator is moved across a range of frequencies in a specified number of points. In this result the ratio of the level of the THD (total harmonic distortion) to the total signal in the DUT output is plotted against frequency.

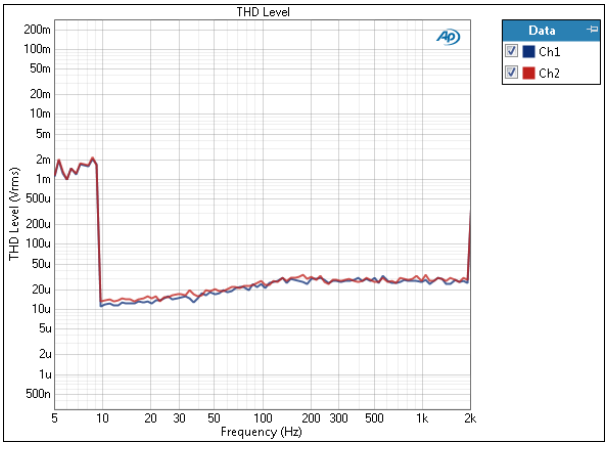
See page 527 for more information about THD.

Units

Units available for Jitter Frequency Sweep: THD Ratio are:

X-axis	*Y-axis*
• Hz	• x/y
• dHz	• %
• F/R	• ppm
• %Hz	• dB

THD Level



The THD Level result uses a fixed audio stimulus signal while the jitter generator is moved across a range of frequencies in a specified number of points. In this result the level of the THD (total harmonic distortion) in the DUT output is plotted against frequency.

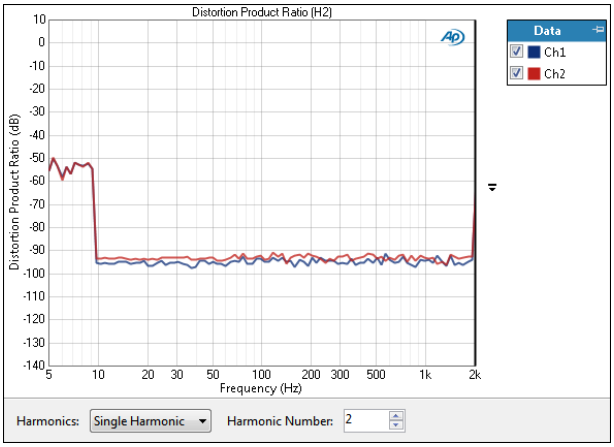
See page 527 for more information about THD.

Units

Units available for Jitter Frequency Sweep: THD Level are:

X-axis	*Y-axis (analog)*	*Y-axis (digital)*	*Jitter signals*
• Hz	• Vrms	• dBFS	• UI
• dHz	• dBV	• FS	• dBUI
• F/R	• dBu	• %FS	• s
• %Hz	• dBrA	• dBrA	
	• dBrB	• dBrB	
	• dBSPL1	• dBSPL1	
	• dBSPL2	• dBSPL2	
	• dBm		
	• W (watts)		

Distortion Product Ratio



The Jitter Frequency Sweep Distortion Product Ratio result uses a fixed audio stimulus signal while the jitter generator is moved across a range of frequencies in a specified number of points. In this result the ratio of the level of the selected harmonic distortion product to the fundamental in the DUT output is plotted against frequency.

Select the harmonic distortion product(s) you wish to view using the **Harmonics** controls, described below.

Additional Controls for Distortion Product Ratio

Harmonics

For a graph of the ratio of one specific harmonic product to the signal level, select **Single Harmonic** and set the **Harmonic Number** to your choice, from 2nd harmonic to 20th harmonic.

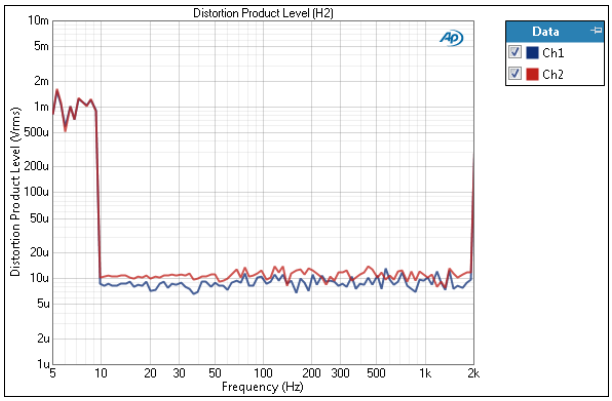
For a graph of the sum of the levels of any combination of harmonic products (from **H2** through **H20**), divided by the signal level, select **Sum of Harmonics**. Then click **Select Harmonics...** to open a result definition dialog.

Units

Units available for Jitter Frequency Sweep Distortion Product Ratio are:

X-axis	*Y-axis*
• Hz	• x/y
• dHz	• %
• F/R	• ppm
• %Hz	• dB

Distortion Product Level



The Jitter Frequency Sweep Distortion Product Level result uses a fixed audio stimulus signal while the jitter generator is moved across a range of frequencies in a specified number of points. In this result the level of the selected harmonic distortion product in the DUT output is plotted against frequency.

Select the harmonic distortion product(s) you wish to view using the **Harmonics** controls, described below.

Additional Controls for Distortion Product Level

Harmonics

For a graph of the level of one specific harmonic product, select **Single Harmonic** and set the **Harmonic Number** to your choice, from 2nd harmonic to 20th harmonic.

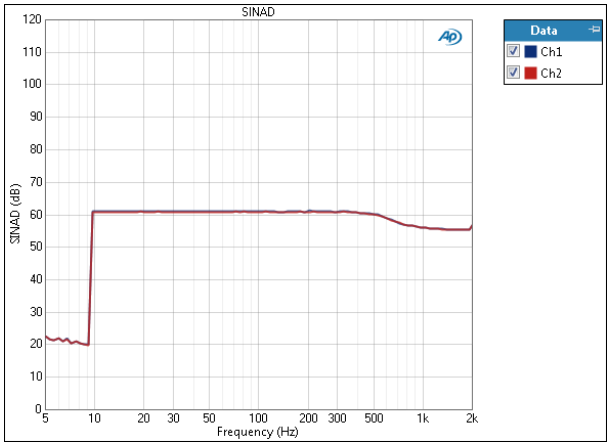
For a graph of the level of the sum of several or all harmonic products (from **H2** through **H20**), select **Sum of Harmonics**, then click **Select Harmonics...** to open a result definition dialog.

Units

Units available for Jitter Frequency Sweep Distortion Product Level are:

X-axis	*Y-axis (analog)*	*Y-axis (digital)*	*Jitter signals*
• Hz	• Vrms	• dBFS	• UI
• dHz	• dBV	• FS	• dBUI
• F/R	• dBu	• %FS	• s
• %Hz	• dBrA	• dBrA	
	• dBrB	• dBrB	
	• dBSPL1	• dBSPL1	
	• dBSPL2	• dBSPL2	
	• dBm		
	• W (watts)		

SINAD



The Stepped Level Sweep: SINAD result uses a fixed audio stimulus signal while the jitter generator is moved across a range of frequencies in a specified number of points. X axis is generator level; Y axis is DUT output SINAD ratio.

Units

Units available for SINAD are:

X-axis (analog)	*X-axis (digital)*	*Y-axis*
• Vrms	• dBFS	• x/y
• Vp	• FS	• dB
• Vpp	• %FS	
• dBV	• dBrG	
• dBu		
• dBrG		
• dBm		
• W (watts)		

Jitter Level Sweep
(Sequence Mode)

Jitter generation and measurement requires an analyzer fitted with the Advanced Master Clock (AMC), standard in an APx555 and an option for APx525, 526, 582, 585, and 586.

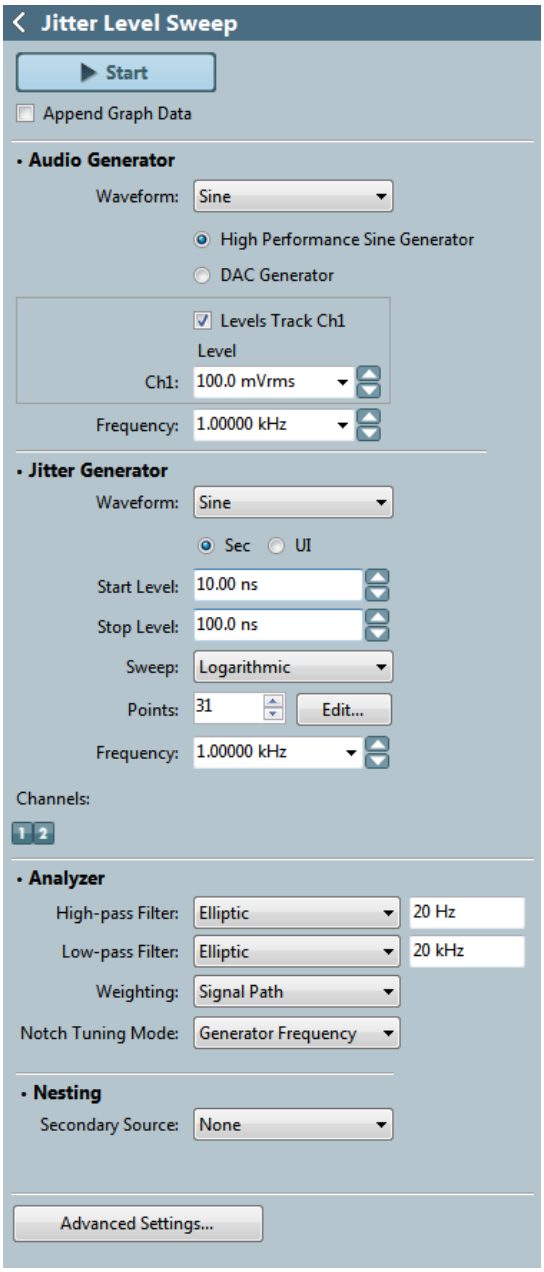
Generated jitter can be applied to an audio stream or clock if the instrument is fitted with a jitter-enabled I/O module, such as ADIO, jitter-enabled DSIO or jitter-enabled PDM. Alternatively, generated jitter can be applied to the AMC Reference Out and Sync Out signals.

Jitter on incoming signals can be analyzed if the instrument is fitted with a jitter-enabled I/O module, such as ADIO, jitter-enabled DSIO or jitter-enabled PDM.

Results for Jitter Level Sweep include:

• Level	• Average Jitter Level*	• THD Level
• Gain*	• THD+N Ratio	• SINAD
• Linearity*	• THD+N Level	• Distortion Product Level
• Peak Level	• THD Ratio	• Distortion Product Ratio

* These results will only be populated if the analyzer input is set to measure the jitter signal. See **Measuring Audio or Jitter**, below.



Overview

The Jitter Level Sweep stimulates the DUT with a digital signal. The embedded audio waveform is selected in Audio Generator. The timebase for the digital signal is modulated by the jitter generator, which is swept across a range of levels.

The results are either

- measurements of the audio output of the DUT (analog or digital), which is typically degraded when the timebase is jittered; or
- for DUTs with a digital output, direct measurements of the jitter in the DUT output signal, rather

than the embedded audio signal. Select **Audio** or **Jitter** in Signal Path Setup > Input/Output > Input Configuration.

Operation

If you have not yet configured your signal path, first go to Signal Path Setup (Chapter 7) to select and configure inputs and outputs, references and other settings.

This measurement must be configured as closed-loop configuration, using the APx generator as a stimulus. Additionally, you must choose where to apply the jitter, and whether to measure the audio or the jitter on the incoming signal, as described next.

Applying jitter

Go to the Clocks panel in Signal Path Setup, and select whether you would like to apply the jitter sweep to a digital output, or to the Reference Out and Sync Out connectors.

Select **Apply To**:

- **Digital Output**
  If you select Digital Output, you must also go to Signal Path Setup > Input/Output and select a digital output that supports jitter generation.
- **Reference/Sync**
  If you select Reference/Sync, you must connect the Reference Out or the Sync Out to the appropriate sync input on your DUT.

For a jitter sweep, you do not need to **Enable** jitter on the Clocks panel. Jitter is automatically enabled for the duration of the sweep.

Measuring Audio or Jitter

For a DUT with a digital output, go to Signal Path Setup > Input/Output to choose a digital input for the APx analyzer.

Select **Measure**:

- **Audio**
  Audio de-embeds the audio signal from the digital interface signal and routes it to the analyzer. This is normal APx operation.
- **Jitter (UI)** or **Jitter (sec)**
  Jitter routes the digital interface signal to jitter demodulator and a dedicated ADC (analog to digital converter) and to the analyzer.

For a default sweep of 20 kHz to 20 Hz in 31 logarithmic steps, simply click **Start**. If you want to specify other parameters for your sweep, review **Generator** and **Analyzer** settings, discussed below. Click **Start** when you have approved the settings.

Append Graph Data

Measured data are grouped in a Data Set, which contains all acquisition results. Normally, the current graph data are deleted each time you start a new measurement.

If the **Append Graph Data** box is checked, appended data are grouped as additional data sets, as are any imported data. The Data Sets Panel displays the Data Set(s) in the measurement, and provides a visibility checkbox, a Time acquisition field, and a Notes field for each data set.

Read more about Data Sets beginning on page 649.

Audio Generator

The Audio Generator generates the audio signal that is embedded in the digital interface signal, as in most APx measurements.

Waveform, Frequency and Level

See Chapter 6 for information about the using the APx Generator and setting Waveform, Level and Frequency.

Jitter Generator

The Jitter Generator generates the signal that modulates the jitter clock, which is used as the timebase for the digital interface signal used in this measurement.

Note: For the case where a jitter level sweep exceeds the derating curve of the jitter generator, the level sweep will skip points that exceed the limit without aborting.

Waveform

Select **Sine, Square** or **Noise**.

Start Level

Set the sweep start level here. The available range is 0 s to 1.592 µs, peak. Default is 10.00 ns.

Stop Level

Set the sweep stop level here. The available range is 0 s to 1.592 µs, peak. Default is 100.00 ns.

Seconds / UI

Choose the jitter units here.

Sweep

Choose

- **Logarithmic** (logarithmic point spacing)
- **Linear** (linear point spacing)
- **Custom**
  Edit sweep spacing and number of **Points** to create a **Custom** sweep.

Points
Set the number of sweep points here. The default is 31 points.

Points > Edit
Open the **Sweep Points** dialog to edit, import or export the **Sweep Points** table.

Step Size
For linear sweeps, enter the **Step Size** here.

Frequency
Set the jitter frequency here.

Seconds / UI
Choose the jitter units here.

Channels
Toggle specific output channels on or off by clicking on the **Channel** number button.

Analyzer
Note that for this measurement, the analyzer can measure either the audio signal (de-embedded from the interface signal) or the jitter (demodulated from the interface signal), as selected in Signal Path Setup > Input/Output.

Filters
> *Note that these filters will affect the audio signal measurements or the jitter signal measurements, as selected in Signal Path Setup > Input/Output.*

Local high-pass, low-pass and weighting filters are available for this measurement. The low-pass and high-pass filters, if selected, are applied to the entire measured signal and affect all the results. The weighting filter, if selected, affects only the distortion and distortion plus noise results. See Measurement Filters on page 619 for detailed information about the filters available locally for some measurements.

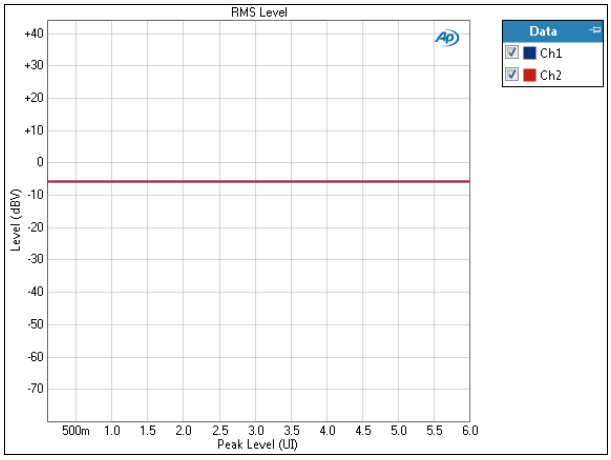
Low pass, high pass and weighting Signal Path Filters settings (see page 617) made in Signal Path Setup > Input/Output will also affect this measurement. When set, the local low-pass or high-pass filter setting overrides the corresponding Signal Path filter setting while the measurement is active. A local weighting filter setting does not override any Signal Path filter settings; the local weighting filter is applied in addition to any active Signal Path filters.

Notch Tuning Mode
For the harmonic distortion results, a tunable notch filter is used to remove the fundamental from the measurement. The tuning can be

- **Generator Frequency**
  The current APx audio generator frequency.
- **Jitter Generator Frequency**
  The APx jitter generator frequency for the current step.
- **Measured Frequency**
  The measured frequency for the current step. When the analyzer channels are receiving different frequencies, the notch filter for each channel is centered on the frequency in that channel.
- **Fixed Frequency**
  A fixed frequency selected by the user. When **Fixed Frequency** is selected, a **Filter Frequency**: entry field becomes available beneath the **Notch Tuning Mode** control.

Nesting
Jitter Frequency sweeps can be nested. Read about Nested Sweeps beginning on page 185.

Advanced Settings
If your test requires special adjustments or settings, click Advanced Settings.

Jitter Level Sweep: RMS Level


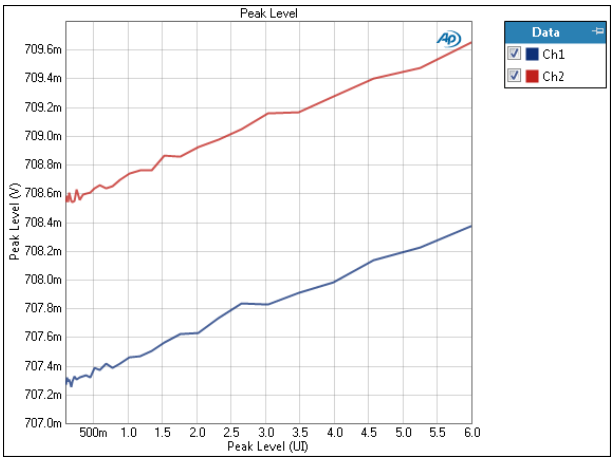
The **Jitter Level Sweep: RMS Level** result uses a fixed audio stimulus signal while the jitter generator is moved across a range of levels in a specified number of points. X axis is generator level; Y axis is DUT output level.

Units

Units available for Level are:

X-axis (analog)	*X-axis (digital)*	*Y-axis (analog)*	*Y-axis (digital)*
• Vrms	• dBFS	• Vrms	• dBFS
• Vp	• FS	• dBV	• FS
• Vpp	• %FS	• dBu	• %FS
• dBV	• dBrG	• dBrA	• dBrA
• dBu		• dBrB	• dBrB
• dBrG		• dBSPL1	• dBSPL1
• dBm		• dBSPL2	• dBSPL2
• W (watts)		• dBm	
		• W (watts)	

Peak Level



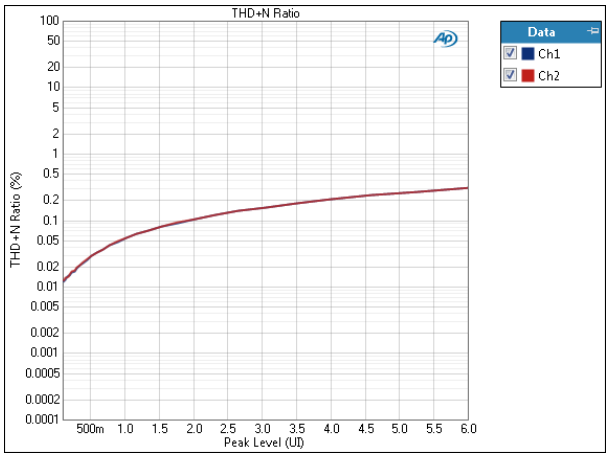
The **Jitter Level Sweep: Peak Level** result uses a fixed audio stimulus signal while the jitter generator is moved across a range of levels in a specified number of points. X axis is generator level; Y axis is the band-pass filtered DUT output level, scaled in peak units.

Units

Units available for Peak Level are:

X-axis (analog)	*X-axis (digital)*	*Y-axis (analog)*	*Y-axis (digital)*
• Vrms	• dBFS	• V	• D
• Vp	• FS		
• Vpp	• %FS		
• dBV	• dBrG		
• dBu		*Y-axis (jitter)*	
• dBrG		• UI	
• dBm		• dBUI	
• W (watts)		• s	

Jitter Level Sweep: THD+N Ratio



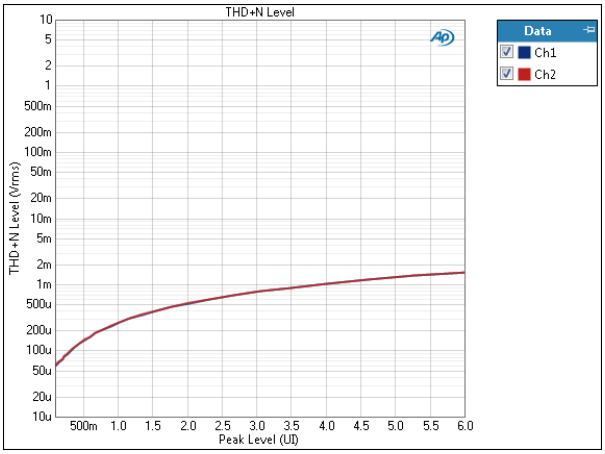
The Jitter Level Sweep: THD+N Ratio result uses a fixed audio stimulus signal while the jitter generator is moved across a range of levels in a specified number of points. X axis is generator level; Y axis is DUT output THD+N ratio.

Units

Units available for THD+N Ratio are:

X-axis (analog)	*X-axis (digital)*	*Y-axis*
• Vrms	• dBFS	• x/y
• Vp	• FS	• %
• Vpp	• %FS	• ppm
• dBV	• dBrG	• dB
• dBu		
• dBrG		
• dBm		
• W (watts)		

Jitter Level Sweep: THD+N Level



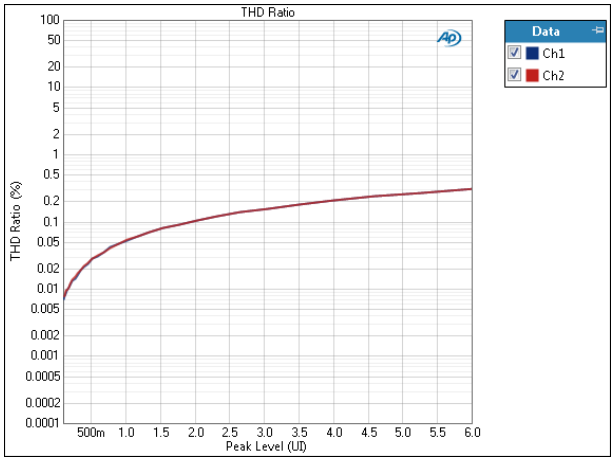
The Jitter Level Sweep: THD+N Level result uses a fixed audio stimulus signal while the jitter generator is moved across a range of levels in a specified number of points. X axis is generator level; Y axis is DUT output THD+N level.

Units

Y-axis units available for THD+N Level are:

X-axis (analog)	*X-axis (digital)*	*Y-axis (analog)*	*Y-axis (digital)*
• Vrms	• dBFS	• Vrms	• dBFS
• Vp	• FS	• dBV	• FS
• Vpp	• %FS	• dBu	• %FS
• dBV	• dBrG	• dBrA	• dBrA
• dBu		• dBrB	• dBrB
• dBrG		• dBSPL1	• dBSPL1
• dBm		• dBSPL2	• dBSPL2
• W (watts)		• dBm	
		• W (watts)	

Jitter Level Sweep: THD Ratio



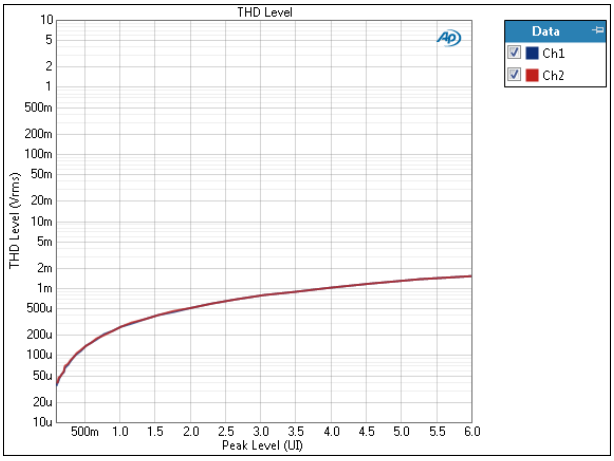
The Jitter Level Sweep: THD Ratio result uses a fixed audio stimulus signal while the jitter generator is moved across a range of levels in a specified number of points. X axis is generator level; Y axis is DUT output THD ratio.

Units

Units available for THD Ratio are:

X-axis (analog)	*X-axis (digital)*	*Y-axis*
• Vrms	• dBFS	• x/y
• Vp	• FS	• %
• Vpp	• %FS	• ppm
• dBV	• dBrG	• dB
• dBu		
• dBrG		
• dBm		
• W (watts)		

Jitter Level Sweep: THD Level



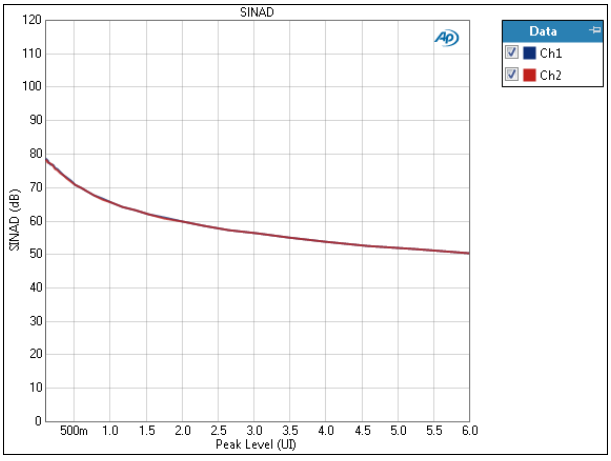
The Jitter Level Sweep: THD Level result uses a fixed audio stimulus signal while the jitter generator is moved across a range of levels in a specified number of points. X axis is generator level; Y axis is DUT output THD level.

Units

Y-axis units available for THD Level are:

X-axis (analog)	*X-axis (digital)*	*Y-axis (analog)*	*Y-axis (digital)*
• Vrms	• dBFS	• Vrms	• dBFS
• Vp	• FS	• dBV	• FS
• Vpp	• %FS	• dBu	• %FS
• dBV	• dBrG	• dBrA	• dBrA
• dBu		• dBrB	• dBrB
• dBrG		• dBSPL1	• dBSPL1
• dBm		• dBSPL2	• dBSPL2
• W (watts)		• dBm	
		• W (watts)	

Jitter Level Sweep: SINAD



The Jitter Level Sweep: SINAD result uses a fixed audio stimulus signal while the jitter generator is moved across a range of levels in a specified number

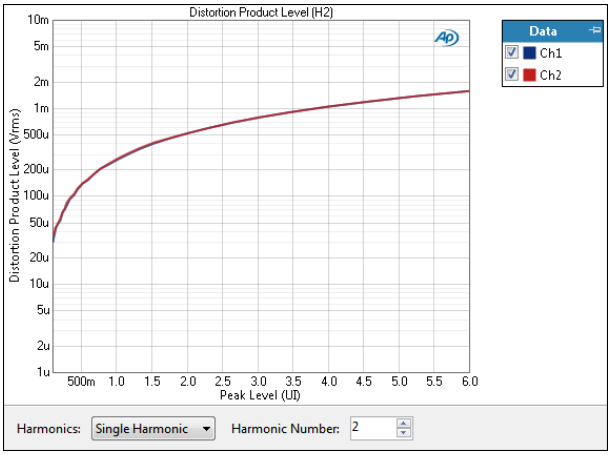
of points. X axis is generator level; Y axis is DUT output SINAD ratio.

Units

Units available for SINAD are:

X-axis (analog)	*X-axis (digital)*	*Y-axis*
• Vrms	• dBFS	• x/y
• Vp	• FS	• dB
• Vpp	• %FS	
• dBV	• dBrG	
• dBu		
• dBrG		
• dBm		
• W (watts)		

Jitter Level Sweep: Distortion Product Level



The Jitter Level Sweep Distortion Product Level result uses a fixed audio stimulus signal while the jitter generator is moved across a range of levels in a specified number of points. In this result the level of the selected harmonic distortion product in the DUT output is plotted against generator level.

Select the harmonic distortion product(s) you wish to view using the **Harmonics** controls, described below.

Harmonics

For a graph of the level of one specific harmonic product, select **Single Harmonic** and set the **Harmonic Number** to your choice, from 2nd harmonic to 20th harmonic.

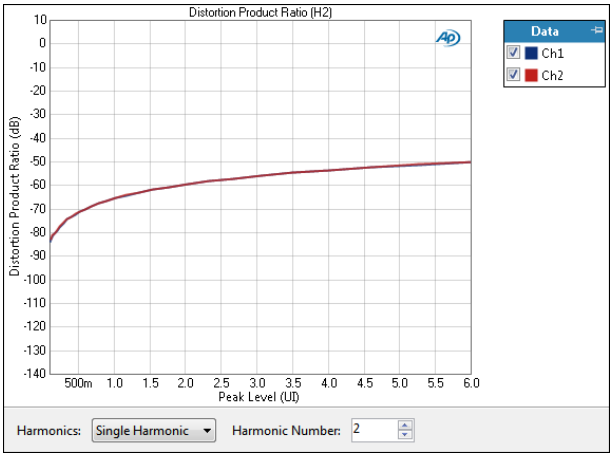
For a graph of the level of the sum of several or all harmonic products (from **H2** through **H20**), select **Sum of Harmonics**, then click **Select Harmonics**... to open a result definition dialog.

Units

Units available for Jitter Level Sweep Distortion Product Level are:

X-axis (analog)	*X-axis (digital)*	*Y-axis (analog)*	*Y-axis (digital)*
• Vrms	• dBFS	• Vrms	• dBFS
• Vp	• FS	• dBV	• FS
• Vpp	• %FS	• dBu	• %FS
• dBV	• dBrG	• dBrA	• dBrA
• dBu		• dBrB	• dBrB
• dBrG		• dBSPL1	• dBSPL1
• dBm		• dBSPL2	• dBSPL2
• W (watts)		• dBm	
		• W (watts)	

Jitter Level Sweep: Distortion Product Ratio



The Jitter Level Sweep Distortion Product Ratio result uses a fixed audio stimulus signal while the jitter generator is moved across a range of levels in a specified number of points. In this result the ratio of the level of the selected harmonic distortion product to the fundamental in the DUT output is plotted against generator level.

Select the harmonic distortion product(s) you wish to view using the **Harmonics** controls, described below.

Harmonics

For a graph of the ratio of one specific harmonic product to the signal level, select **Single Harmonic** and set the **Harmonic Number** to your choice, from 2nd harmonic to 20th harmonic.

For a graph of the sum of the levels of any combination of harmonic products (from **H2** through **H20**), divided by the signal level, select **Sum of Harmonics**. Then click **Select Harmonics...** to open a result definition dialog.

Units

Units available for Jitter Level Sweep Distortion Product Ratio are:

X-axis (analog)	*X-axis (digital)*	*Y-axis*
• Vrms	• dBFS	• x/y
• Vp	• FS	• %
• Vpp	• %FS	• ppm
• dBV	• dBrG	• dB
• dBu		
• dBrG		
• dBm		
• W (watts)		

Level and Gain
(Sequence Mode)

The Level and Gain measurements provide single-value results that show the output level or gain in each DUT channel, as measured at each analyzer input.

Level and Gain results available in APx500 are:

- RMS Level
- Peak Level
- Gain
- Average Jitter Level

Gain is not available in external source configuration, or if a square wave, a generator waveform or File Input has been selected for the measurement.

Average Jitter Level results are only available when Jitter is selected in Signal Path Setup > Input/Output > Measure.

Operation

If you have not yet set up your test, first go to Signal Path Setup to select and configure inputs and outputs, references and other settings. See Chapter 7 for information about Signal Path Setup.

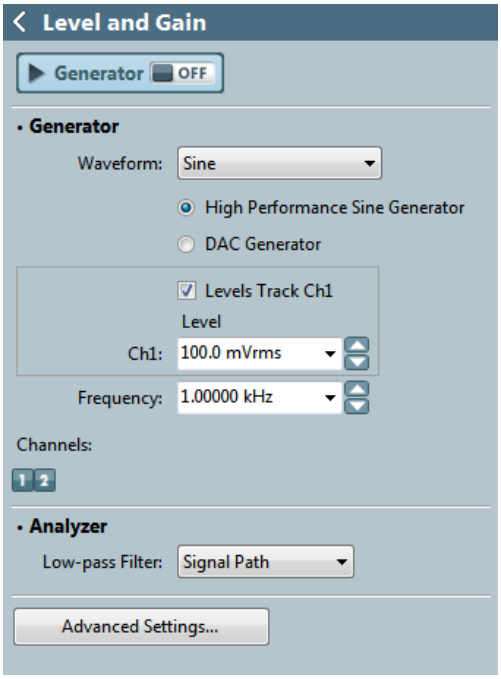
Generator

The typical configuration for this measurement is the closed-loop configuration, using the APx generator as a stimulus. Alternatively, you can make open-loop measurements, using either the External Source or File Input configuration.

See Chapter 6 for detailed information about the using the APx Generator and making Waveform, Level and Frequency settings, or for information about using External Source.

High Performance Sine Generator / DAC Generator

For an APx555, you have the option of using the High Performance Sine Generator (very low distortion sine waves and wide bandwidth) or the DAC generator,



which provides special sine waveforms. See High Performance Sine Generator starting on page 51 for a comparison of features.

Analyzer

This measurement can be configured to measure jitter in Signal Path Setup > Input > Measure. Read detailed information about jitter generation and measurement beginning on page 69.

Low-pass Filter

A local low-pass filter is available for this measurement. The low-pass filter, if selected, is applied to the entire measured signal and affects all the results. See Measurement Filters on page 619 for detailed information about the filters available locally for some measurements.

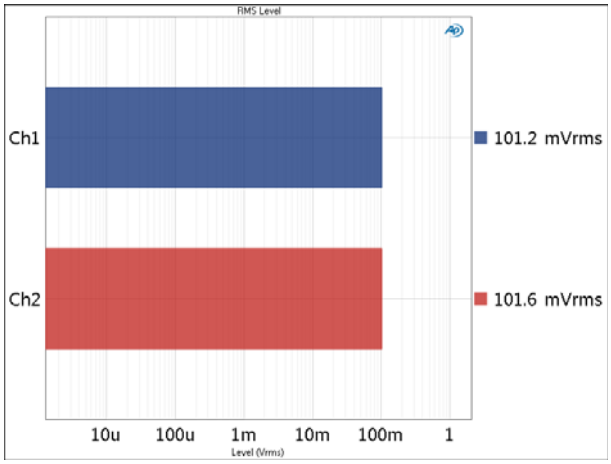
Low pass, high pass and weighting Signal Path Filters settings (see page 617) made in Signal Path Setup > Input/Output will also affect this measurement. When set, the local low-pass filter setting overrides the corresponding Signal Path filter setting while the measurement is active.

Advanced Settings

Advanced Settings makes available individual channel output level and offset controls, and input ranging and settling controls. See Advanced Settings for single value measurements on page 351.

See Chapter 104 for more information about units of measurement.

RMS Level



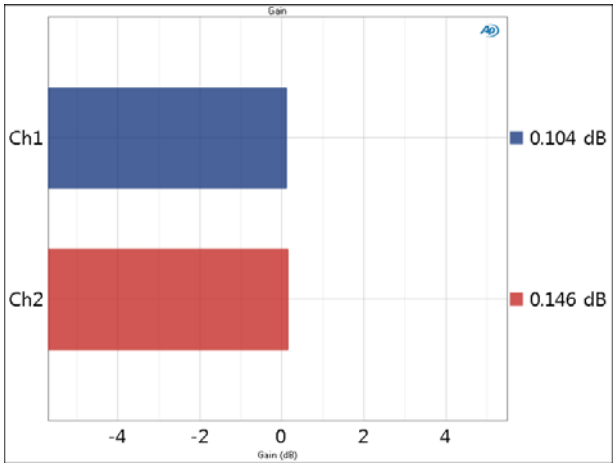
RMS Level provides a single-value meter result, displaying the rms output level from each DUT channel, as measured at each analyzer input.

Units

Units available for RMS Level are

Analog Signals	*Digital Signals*	*Jitter signals*
• Vrms	• dBFS	• UI
• dBV	• FS	• dBUI
• dBu	• %FS	• s
• dBrA	• dBrA	
• dBrB	• dBrB	
• dBSPL1	• dBSPL1	
• dBSPL2	• dBSPL2	
• dBm		
• W (watts)		

Gain



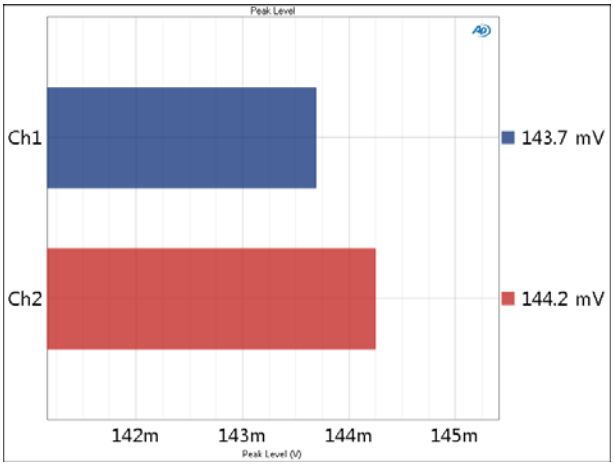
The Gain measurement result provides a single-value measurement of the DUT gain for channel, as measured at each analyzer input.

Units

Units available for Gain are

Same-domain	*Cross-domain*
• x/y	• FS/Vrms
• %	• dB(FS/Vrms)
• ppm	or
• dB	• Vrms/FS
	• dB(Vrms/FS)

Peak Level



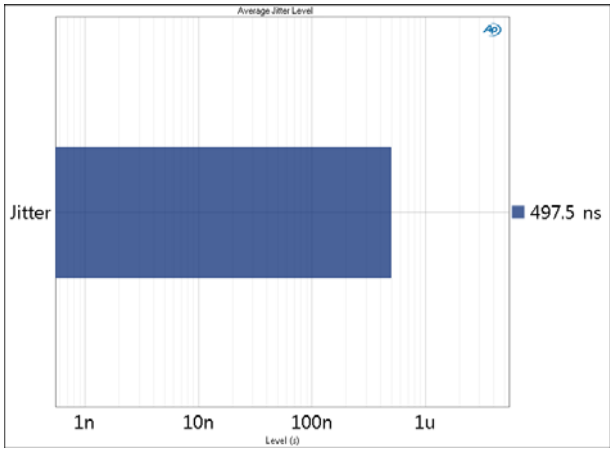
Peak Level provides a single-value meter result, displaying of the peak output level from each DUT channel, as measured at each analyzer input.

Units

Units available for Peak Level are

Analog Signals	*Digital Signals*	*Jitter signals*
• Vrms	• dBFS	• UI
• dBV	• FS	• dBUI
• dBu	• %FS	• s
• dBrA	• dBrA	
• dBrB	• dBrB	
• dBSPL1	• dBSPL1	
• dBSPL2	• dBSPL2	
• dBm		
• W (watts)		

Average Jitter Level



Average Jitter Level provides a single-value meter result, displaying the average jitter level from the digital receiver.

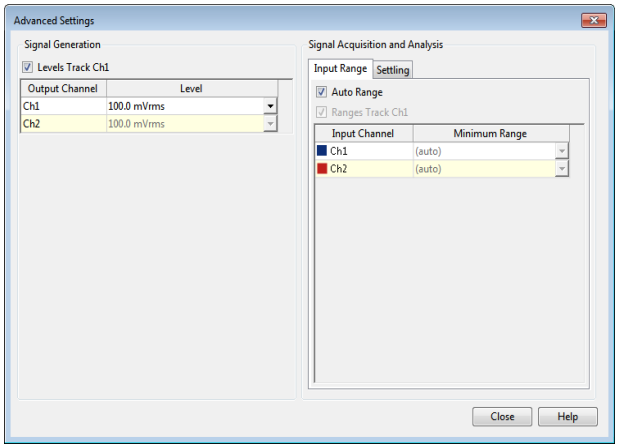
Units

Units available for Average Jitter Level are

Analog Signals	*Digital Signals*	*Jitter signals*
• Vrms	• dBFS	• UI
• dBV	• FS	• dBUI
• dBu	• %FS	• s
• dBrA	• dBrA	
• dBrB	• dBrB	
• dBSPL1	• dBSPL1	
• dBSPL2	• dBSPL2	
• dBm		
• W (watts)		

Advanced Settings for Single Value Measurements

The default settings here are appropriate for most measurements. You may want to make minor adjustments for special situations.



Signal Generation

You can set different output levels or DC offsets for each channel here.

If **Levels Track Ch1** is checked (the default), the generator output level and DC offset values for channel 1 are copied to the other channels, and values for channels greater than 1 cannot be edited. Any changes made to channel 1 are reproduced in the other channels.

To set individual generator output channel levels or offsets, uncheck the **Levels Track Ch1** checkbox and enter values in the output channel Level or Offset fields.

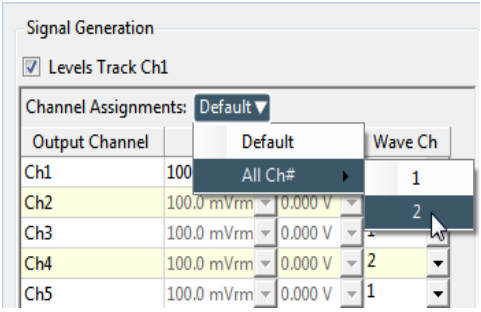
Set Channel Assignments for special waveforms

For special waveforms like Sine Dual, you can specify the channel assignment.

Set Channel Assignments for generator waveforms

For stereo or multichannel generator waveform files, you can set channel mapping. By default, channel 1 audio is mapped to channel 1 output, channel 2 to channel 2, and so on. If the number of channels in the waveform file is less than the number of output channels, the waveform channels resume numbering at 1 and wrap to the next available output channel. These assignments can be remapped by changing individual settings in the **Wave Ch** column, or by selection one of

several presets from the **Set Channel Assignments menu.**



Signal Acquisition and Analysis

Input Range tab

Digital inputs do not require ranging, and the Input Range tab is unavailable when inputs are set to a digital format.

Auto Range

If **Auto Range** is checked (the default), each analog channel determines its input range automatically, based on the level of the input signal. If the input signal level changes beyond a ranging threshold, **Auto Range** will cause the input ranging circuits to move up or down for proper ranging.

Fixed Range

Alternatively, you can set a fixed minimum range for the analog inputs. Uncheck **Auto Range** and enter the expected maximum level in the **Minimum Range** filed. The software will choose the lowest range that will accommodate that level; this becomes the minimum range available, and the analyzer will not reset to a lower range. Fixed range is best for signals that vary rapidly, including tone bursts, perceptual audio speech samples, and program material such as voice or music. See **When autoranging isn't suitable** on page 626.

If a signal greater then the **Minimum Range** is presented to the analog input, the ranging circuits will range up to accommodate the higher level, and then drop again when the signal level falls. The range will not be set below the **Minimum Range**.

Ranges Track Ch1

If **Ranges Track Ch1** is checked (the default), the fixed input range setting for channel 1 is copied to the other channels, and range settings for channels greater than 1 cannot be edited. Any changes made to channel 1 are reproduced in the other channels. If **Ranges Track Ch1** is not checked, the fixed input range setting for each channel can be set independently.

Read more about ranging in Chapter 98.

Settling tab

One or two Settling tabs are available for Single Value measurement results. These tabs may have different parameters and default values depending upon the measurement result to which they're attached. See page 627 for more information about settling.

Level Ratio
(Sequence Mode)

The Level Ratio measurement provides a single-value meter result of the ratio of the level in each channel to the level in the selected Reference channel

If you have not yet set up your test, first go to Signal Path Setup to select and configure inputs and outputs, references and other settings.

Operation

If you have not yet set up your test, first go to Signal Path Setup to select and configure inputs and outputs, references and other settings. See Chapter 7 for information about Signal Path Setup.

Generator

The typical configuration for this measurement is the closed-loop configuration, using the APx generator as a stimulus. Alternatively, you can make open-loop measurements, using either the External Source or File Input configuration.

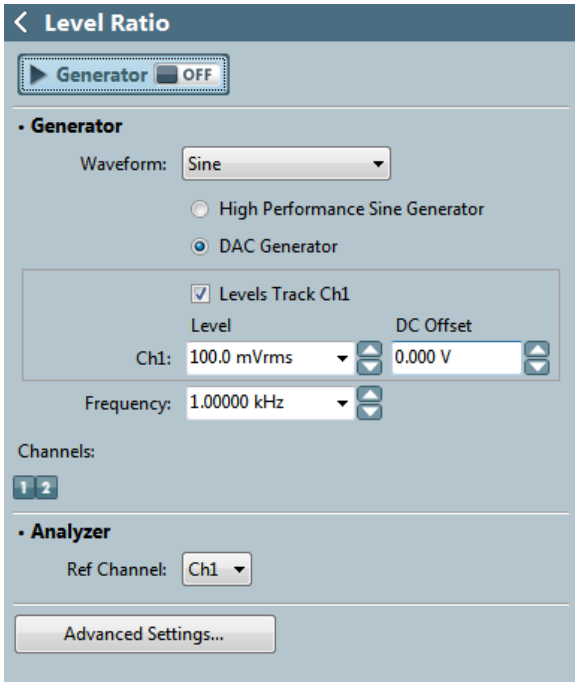
See Chapter 6 for detailed information about the using the APx Generator and making Waveform, Level and Frequency settings, or for information about using External Source.

High Performance Sine Generator / DAC Generator

For an APx555, you have the option of using the High Performance Sine Generator (very low distortion sine waves and wide bandwidth) or the DAC generator, which provides special sine waveforms. See High Performance Sine Generator starting on page 51 for a comparison of features.

Analyzer

No local low pass, high pass or weighting filters are available for this measurement. However, low pass, high pass and weighting filter settings made in Signal Path Setup



> Input/Output will affect this measurement.

Ref Channel

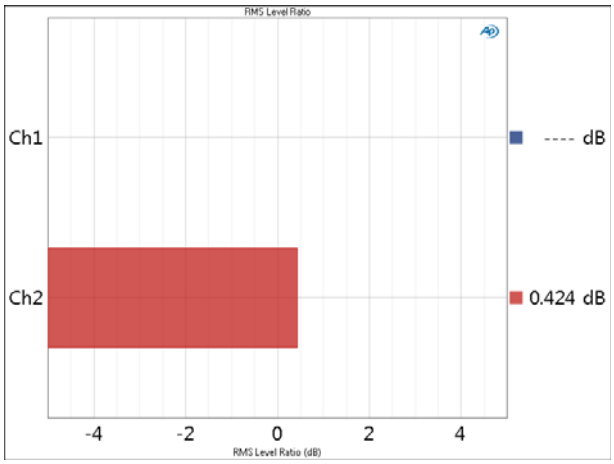
The level ratio of each channel is measured against the selected Reference Channel. There is no level ratio for the Reference Channel itself, and no data is displayed for that channel.

Advanced Settings

Advanced Settings makes available individual channel output level and offset controls, and input ranging and settling controls. See Advanced Settings for single value measurements on page 351.

See Chapter 104 for more information about units of measurement.

Level Ratio



Level Ratio provides a single-value meter result, displaying the ratio of the level in the non-reference input channel(s) to the level in the Reference Channel.

Units

Units available for Level Ratio are

- x/y
- %
- ppm
- dB

Loudspeaker Production Test
(Sequence Mode)

Introduction

This measurement requires a software option key. See page 189 for more information about software options.

Loudspeaker Production Test is only available when the analyzer input and output are set to an analog, non-acoustic configuration.

Loudspeaker Production Test combines two types of measurements in one test to facilitate fast loudspeaker testing in a production environment. An external power amplifier, a measurement microphone with power supply, and an impedance fixture with a sense resistor are required. An acoustic isolation chamber is optional.

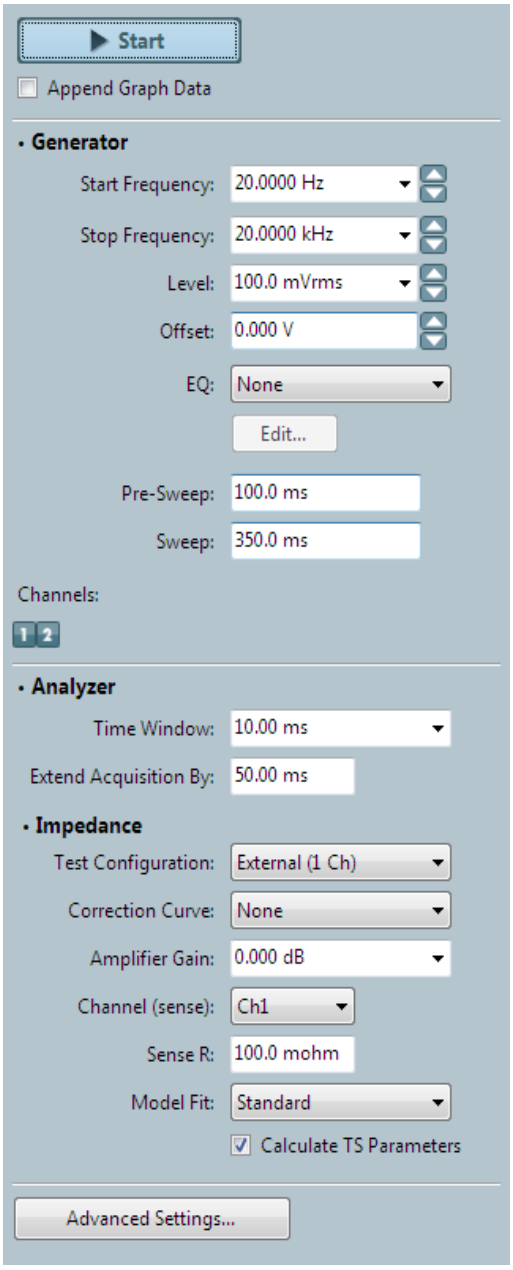
The two types of measurements are

- an acoustic measurement, providing frequency response, phase, distortion, rub and buzz and impulse response results;
- an electromechanical impedance measurement, providing impedance response curves and a subset of Thiele-Small parameter results.

The underlying technology is the continuous sweep, providing both acoustic and impedance results in one fast measurement.

Loudspeaker Production Test results available in APx500 are:

- Level (Freq. Response)
- Relative Level
- Phase
- Distortion Product Ratio
- Distortion Product Level
- Rub and Buzz
- Impulse Response (for polarity
- Impedance Magnitude
- Impedance Phase
- Thiele-Small parameters



All of these results are available from a single acquisition.

The acoustic measurement

Set a measurement microphone in front of the loudspeaker under test. Connect this microphone to its power supply and to an undesignated APx analyzer input. An undesignated input is one that does not have a role assigned to it for the impedance measurement. See **Connections and test configurations** on the next page.

Determining the value for the Time Window

The acoustic measurement is based on the APx Acoustic Response measurement, which makes quasi-anechoic measurements by limiting the duration of the impulse response to the time before the first reflection. This requires setting the measurement **Time Window** to the proper value. In Acoustic Response, this is an interactive process where the user manually sets a cursor on a graph display, or measures the distance for the direct and first reflection paths in the acoustic space. You can read about determining an optimal Time Window value in Chapter 24.

For production testing, the supervising production engineer must determine the optimal value for the **Time Window** (using Acoustic Response or some other method) at the time the production station is being set up, and recording the value for entry into the **Time Window** field in Loudspeaker Production Test. This value can be used until there is a significant change in the acoustic testing space.

One or two of the analyzer input channels must be dedicated to the impedance measurement, measuring voltages and calculating currents. One (or more) of the remaining channels measures the audio acquired by a measurement microphone for an acoustic result.

The impedance measurement

The electromechanical impedance measurement is based on the APx Impedance/Thiele-Small measurement, where the current through the driver and the voltage across the driver must be determined. For fast and reliable production testing, Loudspeaker Production Test only makes only a single-pass, free-air impedance measurement, providing a subset of the Thiele-Small parameters, sufficient to identify manufacturing defects.

An audio power amplifier and a sense resistor are required, and one or two additional analyzer inputs, depending upon the chosen test configuration. The optional Audio Precision IMP1 impedance fixture provides sense resistors, amplifier and analyzer connectors and switching.

You will need to precisely measure the value of the sense resistor, and if you are using the **External (1 Ch)** test configuration, you will need to create an amplifier correction curve. If you do not create an amplifier correction curve, you must precisely measure the broadband gain of the amplifier before making the measurement, and enter the value into the **Amplifier Gain** field.

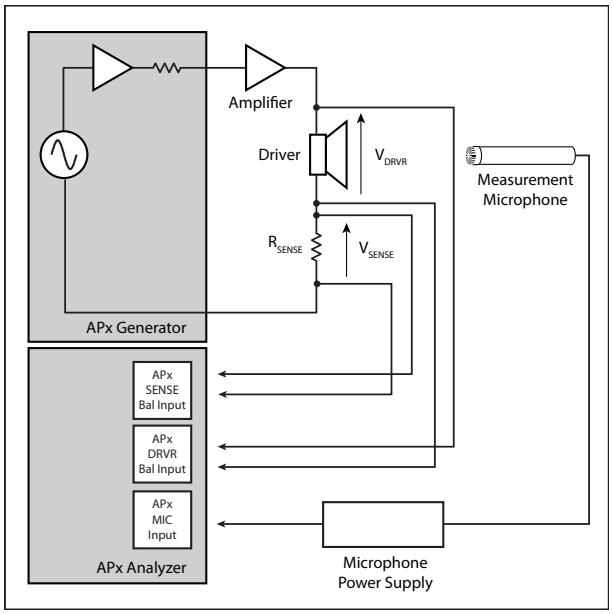
Connections and Test Configurations

Loudspeaker Production Test measurements can use a choice of test configurations to derive the driver current from the available voltages. Select the test configuration suited to your situation.

In these tests, we are not measuring conventional audio parameters; we are measuring voltage or current in circuit components. Consequently, one or two of the analyzer input channels have specific roles assigned, either to measure the voltage across the driver, the voltage across the sense resistor, or the total voltage across the driver and the sense resistor. Please pay close attention to the roles depicted in the illustrations linked here, and identified on the **Channel** control in the software.

The Audio Precision IMP1 Impedance Fixture can be used with either configuration.

External (2 Ch, recommended) (bal)



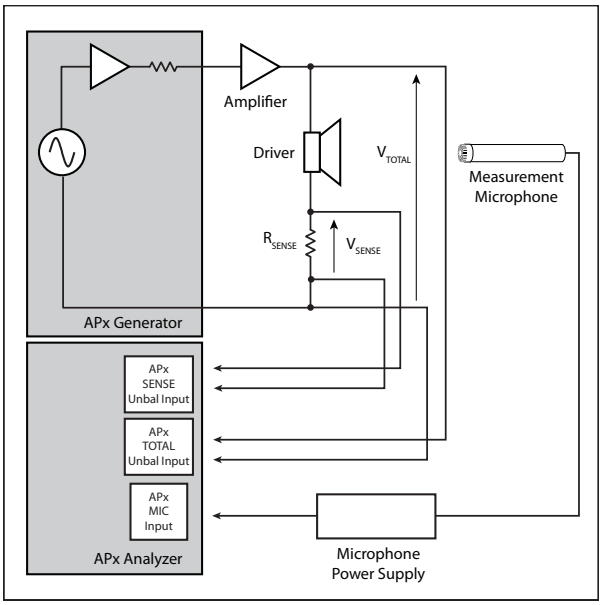
This is our recommended configuration when making a 2-channel impedance measurement. This configuration is enabled when **Test Configuration** is set to **External (2 Ch)** and the analyzer inputs are set to balanced.

The Audio Precision IMP1 Impedance Fixture can be used with this configuration.

We recommend that you use balanced analyzer inputs for impedance measurements.

Avoid the use of a bridged amplifier as the accuracy of results can be significantly degraded due to strong common mode signals at the analyzer input.

External (2 Ch, recommended) (unbal)



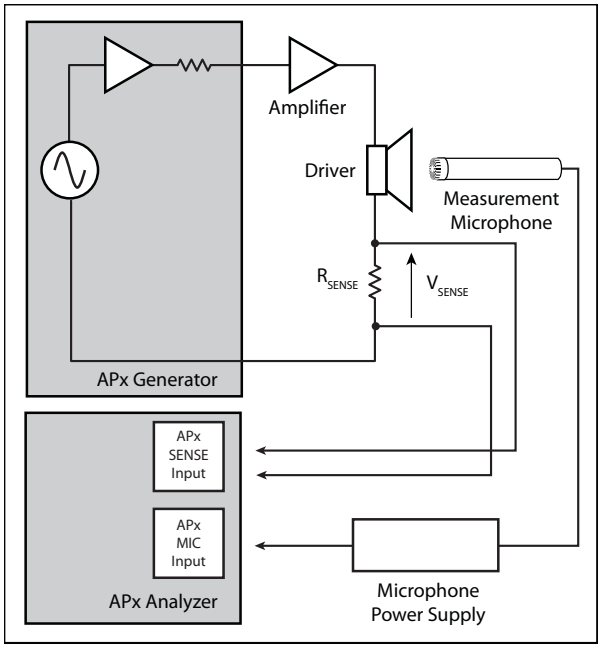
This configuration is enabled when **Test Configuration** is set to **External (2 Ch)** and the analyzer inputs are set to unbalanced.

This configuration is similar to **External (2 Ch)** (bal), but it allows the use of unbalanced analyzer inputs, which may be more convenient in some situations. V_{DRVR} is calculated by subtracting $V_{SENS}E$ from V_{TOTAL}. When the analyzer is set to unbalanced inputs, this configuration is assumed and voltage measured at channel *n*+1 is subtracted from the voltage at channel *n*, provided the driver voltage.

The Audio Precision IMP1 Impedance Fixture can be used with this configuration.

A bridged amplifier cannot be used with unbalanced analyzer inputs.

External (1 Ch)



This method uses an external power amplifier and sense resistor. This is the only configuration that supports Loudspeaker Production Test using an analyzer with only two inputs.

Set **Correction Curve** to select an amplifier correction curve in the project, or choose to **Create New** or **Browse for File**. If the correction curve accurately represents the amplifier in use, this configuration will make accurate impedance response measurements. Enter the value of the sense resistor into the data entry field in APx. This is our recommended configuration when making a 1 channel impedance measurement.

If **Correction Curve** is set to **None**, the broadband gain of the amplifier must be known from a previous measurement, such as the APx Level and Gain measurement. Enter the gain and the value of the sense resistor into the data entry fields in APx.

When **Correction Curve** is set to **None**, this configuration cannot correct for the low-frequency rolloff present in an AC-coupled power amplifier. AC coupling will introduce phase shift that clearly affects the driver impedance curve and resonance frequency. If your test requires a sweep that extends below 20 Hz, we recommend the use of a DC-coupled power amplifier, or that you use a correction curve as described above.

The Audio Precision IMP1 Impedance Fixture can be used with this configuration.

We recommend that you use balanced analyzer inputs for impedance measurements.

Avoid the use of a bridged amplifier as the accuracy of results can be significantly

degraded due to strong common mode signals at the analyzer input.

Operation

If you have not yet set up your test, first go to Signal Path Setup to select and configure inputs and outputs, references and other settings. See Chapter 7 for information about Signal Path Setup.

After making the settings below, click **Start**.

Generator

Append Graph Data

Measured data are grouped in a **Data Set**, which contains all acquisition results. Normally, the current graph data are deleted each time you start a new measurement. If the **Append Graph Data** box is checked, appended data are grouped as additional data sets, as are any imported data. The Data Sets Panel displays the Data Set(s) in the measurement, and provides a visibility checkbox, a Time acquisition field, and a Notes field for each data set.

Read more about Data Sets beginning on page 649.

Generator

The Loudspeaker Test measurement uses a continuous sweep (log-swept sine chirp) stimulus signal, swept between the frequencies entered in **Start Frequency** and **Stop Frequency**, at the value set in **Level**.

Running the Measurement

To use Loudspeaker Production Test, click **Start**. The generator will output the test signal to the DUT on the selected generator channels. Read the results in the selected result view.

Start Frequency and Stop Frequency

These controls set the frequency range of the sweep. For testing a loudspeaker, set the range to the range of your loudspeaker or driver. See page 246 for more about continuous sweep ranges, which are dependent upon the instrument.

Level

Set the generator level here.

Note: Loudspeakers under test can be damaged by signals that exceed their level or frequency limits. Be sure to set generator and amplifier level and sweep frequencies to values within the limits of the loudspeaker.

EQ

You can equalize the generator signal to make a pre-emphasized stimulus. See Generator Equalization (page 192) for a general discussion of this feature.

Choosing generator equalization

Select **None, Relative** or **Absolute** from the EQ drop-down menu.

- **None** applies no equalization to the generator.
- **Relative** applies the gain or loss specified in the EQ table to the current generator level for each channel.
- **Absolute** sets the generator level for all channels to the level specified in the EQ table. Note: when **Absolute** is selected, current channel level settings are lost.

Edit

If you have chosen to equalize the generator signal, you can edit the EQ table here. See **Edit EQ Table** (page 192).

Pre-Sweep and Sweep times

A continuous sweep stimulus consists of a sine wave swept logarithmically across the sweep range in a short period of time.

To enable the DUT to stabilize before the beginning of the defined sweep measurement, there is an adjustable **Pre-Sweep** time. The actual sweep stimulus signal begins below the defined **Start Frequency** and is swept upward, reaching the **Start Frequency** at the end of the **Pre-Sweep** time. The sum of the times in **Pre-Sweep** and **Sweep** is the total duration of the sweep.

- The default for **Pre-Sweep** is 100.0 ms. Minimum is 0 s; maximum is 1.0 s.
- The default for **Sweep** is 350.0 ms. Minimum is 50.0 ms; maximum is 5 s.

Longer **Pre-Sweep** settings allow more time for DUT stabilization, but increase measurement time. Longer **Sweep** settings provide greater resolution and signal-to-noise ratio, but increase measurement time.

Analyzer

Time Window

Enter a value for the **Time Window** here. See page 356 for more information about Time Window values.

Extend Acquisition By

By default, a continuous sweep measurement makes an acquisition 50 ms longer than the stimulus, to include possible time-delayed artifacts created in the DUT.

In some cases, you may want to extend the acquisition further. Enter a new value in the **Extend Acquisition By:** field. Minimum extension is 0 s; maximum is 3 s.

Impedance

Test Configuration

This measurement supports two test configurations. The selection made here affects the availability or options of the **Amplifier Gain, Channel** and **Sense R** settings below. The test configurations are:

- **External (2 Ch)** or
- **External (1 Ch)**

See page 356 for a full description of these configurations.

External (2 Ch, recommended)

For the Loudspeaker Production Test measurement, this configuration is not supported by 2-channel analyzers.

This configuration uses an external power amplifier and an external sense resistor. It requires 2 analyzer input channels for the impedance measurement, but because both the voltage across the driver and the current through the driver are measured simultaneously, it produces the best results.

Since Loudspeaker Production Test also requires at least one channel for the acoustic measurement microphone, the Loudspeaker Production Test measurement only supports the **External (2 Ch)** configuration for analyzers with more than two input channels.

Channel

Balanced inputs

See a diagram of this configuration on page 356.

We recommend the use of balanced inputs for impedance testing. When using balanced inputs, the **External (2 Ch)** Test Configuration requires one input channel to be assigned the role of "**drvr**". This channel measures the voltage across the loudspeaker driver.

It requires a second input channel to be assigned the role of "**sense**". This channel measures the voltage across the sense resistor.

Channel roles are assigned in adjacent pairs, with "**drvr**" always the lower numbered channel.

Unbalanced inputs

See a diagram of this configuration on page 357.

When using unbalanced inputs, the **External (2 Ch)** Test Configuration requires one input channel to be assigned the role of "**total**". This channel measures the total voltage across the loudspeaker driver and the sense resistor. This is the same as the amplifier output voltage.

It requires a second input channel to be assigned the role of "**sense**". This channel measures the voltage across the sense resistor.

Channel roles are assigned in adjacent pairs, with "**total**" always the lower-numbered channel. The voltage across the driver is calculated by subtracting the sense voltage from the total voltage.

External (1 Ch)

See a diagram of this configuration on page 357.

This configuration only requires 1 channel for the impedance measurement, allowing support for 2-channel amplifiers when using Loudspeaker Production Test. A second channel is used for the acoustic measurement. Like the above configurations, it requires an external power amplifier and an external sense resistor.

For the **External (1 Ch)** configuration, you must use an amplifier correction curve (see below) or provide a broadband amplifier gain value. We recommend using an amplifier correction curve, as it is more accurate. Characteristics of the amplifier used and the accuracy of the correction curve will affect your results.

If you are using an APx1701 Transducer Test Interface, Correction Curve is not available.

Correction Curve:

- **None**
  If you are not using a correction curve, choose **None** and enter a value for the broadband amplifier gain in the field below. The response of your amplifier will affect the accuracy of the impedance curves. See External (1 Ch) Test Configuration on page 357 for more information.
- If there are Amplifier Correction Curves in the project, they will be listed here and available for selection.
- **Create New**
  To create an amplifier correction curve, choose Create New. See **Amplifier Correction** on page 325.
- **Browse for file**
  To use a previously saved amplifier correction curve, choose **Browse for file**.

Amplifier Gain

This field is only available when **Test Configuration** is set to the **External (1 Ch)** configuration and **Correction Curve** is set to **None**. Determine the broadband gain of your amplifier using a measurement such as the APx Level and Gain, and enter the value here.

If you are using an APx1701 Transducer Test Interface, amplifier gain is fixed at 20 dB.

Channel

The **External (1 Ch)** Test Configuration requires one input channel to be assigned the role of "**sense**". This channel measures the voltage across the sense resistor.

Sense R:

Enter the value of the external sense resistor here. The acceptable range is 0.01 ohm to 1,000 ohm.

Note: The APx1701 Transducer Test Interface has an internal 0.1 ohm sense resistor for each channel. If you are using the APx1701, the value in Sense R: will be fixed at 0.1 ohm.

Model Fit

After the preliminary set of Thiele-Small parameters are obtained from the free air pass, the result data are processed and a model of the impedance curve is created and displayed on the graph as the **Fit** trace. The result data is also shown. You can choose to view the raw, unprocessed result data, or the processed result data using the **Unprocessed / Processed** control on the graph display. The choice of **Unprocessed / Processed** viewing does not affect the operation of the measurement.

The processing used for the Processed data includes decimation of data points and smoothing.

Note: If the ratio between the sense resistor and the driver under test is large (small sense resistor / large driver impedance), very little voltage will be developed across the sense resistor, and the result data may be very noisy. The Unprocessed view may show a great deal of noise, and in severe cases the impedance curve of the driver under test may not fit the available models. See Calculate TS Parameters, below.

Choose a Model Fit:

- **Standard**
- **LR-2 or**
- **Wright**

See page 323 for detailed information and diagrams of the driver models.

Calculate TS Parameters

This checkbox is checked by default, enabling the calculation of Thiele-Small parameters from the impedance measurement.

In some cases (such as high-impedance headphone measurements), the impedance curve of the driver under test may not fit the available models. Derivation of the Thiele-Small parameters will fail, and the measurement will abort. To prevent this, uncheck the **Calculate TS Parameters** checkbox. The device impedance curve will be measured, but Thiele-Small parameters and model fit will not be calculated.

Curve Fit Range

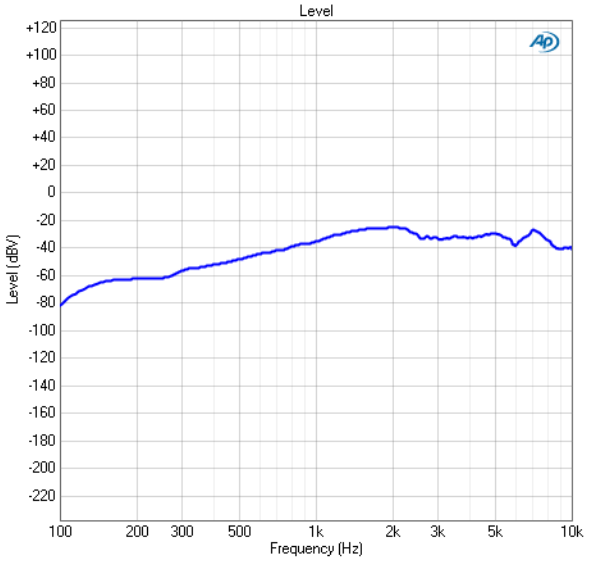
For calculating Thiele-Small parameters, it is best to constrain the high-frequency extent of the range across which the data is fitted to the model curve to approximately 20xFs (Fs is the resonant frequency of the driver).

- **Auto**
  The data that is fitted to the model curve is the full frequency range of the acquisition, which is approximately the same as the range of the generator sweep.
- **Custom**
  The data that is fitted to the model curve is constrained to the range between the **Min Freq** and **Max Freq**, set in the fields below.

Advanced Settings

Advanced Settings include output level and offset settings and input ranging settings. To view or modify these settings, click **Advanced Settings**. See Advanced Settings for Continuous Sweep based measurements on page 249.

Level (Frequency Response)



Loudspeaker Production Test: Level is a continuous sweep result that displays level plotted against frequency. This is the frequency response of the loudspeaker.

A note about the gray area on the graph

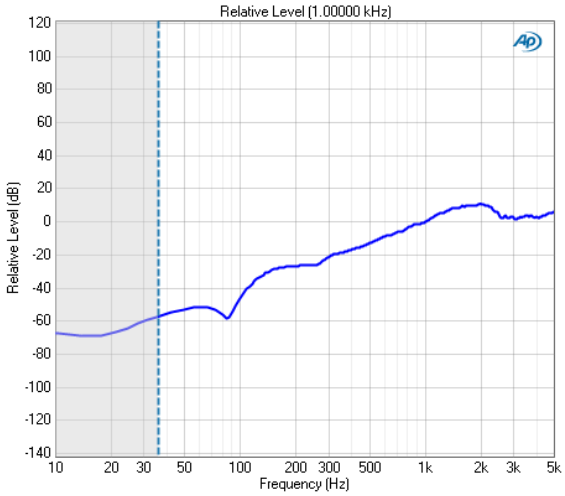
An inevitable consequence of the finite length of an impulse response is that in the frequency domain, the data below some low frequency will be unreliable. In this graph, the area of unreliable data is indicated by gray shading. As the **Time Window** is increased, the frequency of the bound defining this area is decreased.

Units

Units available for Loudspeaker Production Test Level (Frequency Response) are:

X-axis	*Y-axis*
• Hz	• Vrms
• dHz	• dBV
• F/R	• dBu
• %Hz	• dBrA
	• dBrB
	• dBSPL1
	• dBSPL2
	• dBm
	• W (watts)

Relative Level



The Relative Level result is a continuous sweep measurement that provides a graphical display of the frequency response of the loudspeaker. In this result the DUT output level is plotted against frequency, relative to the level at a selected frequency set in the **Ref Frequency** field.

This enables you to specify the frequency that will be set as 0 dB and view the response in relation to that frequency. A midband frequency such as 1 kHz is often set as the reference frequency, and is the default here.

Additional Controls for Relative Level

These settings are made in the Result Settings bar, beneath the graph display.

Mode

Normalized at Reference

Enter a reference frequency for normalization in the **Ref Frequency** field. The measured level at the frequency you choose is set as 0 dB on the graph, and the DUT response is plotted in relation to this reference. You can change the **Ref Frequency** at any time (except after appending) and the graph will immediately be redrawn to reflect the new setting.

Auto-Centered in Limits

This control is only usable if upper and lower limits are set on this graph. In some applications, the general conformance of a trace to the shape of a set of limits is of importance, with absolute level less important or irrelevant. **Auto-Centered in Limits** centers the data trace relative to the upper and lower limits. To exclude possible exceptional data points at frequency extremes, the frequency range that is considered for the centering operation is constrained by the values in the **Min** and **Max** fields.

Ref Frequency

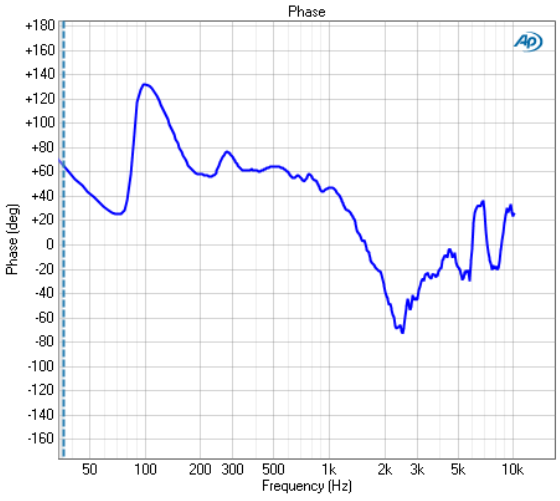
Choose a Reference Frequency here.

Units

Units available for Loudspeaker Production Test: Relative Level are:

X-axis	*Y-axis*
• Hz	• x/y
• dHz	• %
• F/R	• ppm
• %Hz	• dB

Phase



The Loudspeaker Production Test Phase result is a continuous sweep measurement that displays the DUT output phase, plotted against frequency.

See page 332 for more information about phase measurements.

Additional Controls for Phase

These settings are made in the Result Settings bar, beneath the graph display.

Mode

Loudspeaker Production Test Phase has four result modes:

Relative to Ch1

In this mode, the absolute phase of channel one is subtracted from the absolute phase of channels numbered greater than 1. The result is plotted against frequency for each channel numbered 2 and above, showing the phase difference (from channel 1) for each channel. Since channel 1 is used as the reference, it is not plotted in this result. This mode shows "unwrapped" phase differences.

Input-to-output

In the input-to-output mode, the absolute phase of each channel, from device input to device output, is plotted against frequency, "unwrapped." Input-to-output mode includes device delay.

Input-to-output, wrapped

This mode shows the same result as input-to-output phase, but "wrapped" within the range of –180° to +180°.

Input-to-output, excess

This mode shows the input-to-output phase (unwrapped), but removes the linear component (the device delay).

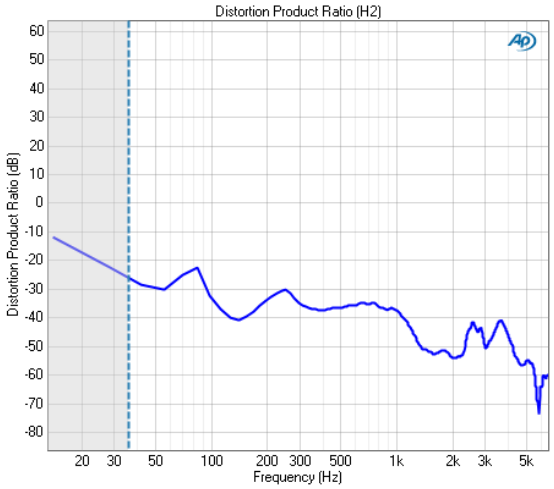
See page 241 for more information about wrapped and unwrapped phase.

Units

Units available for Loudspeaker Production Test Phase are:

X-axis	*Y-axis*
• Hz	• deg
• dHz	• rad
• F/R	
• %Hz	

Distortion Product Ratio



The Loudspeaker Production Test Distortion Product Ratio result is a continuous sweep measurement that provides a graphical display of the selected harmonic distortion products present in each audio input channel. In this result the ratio of the level of the selected harmonic distortion product to the fundamental in the DUT output is plotted against frequency.

Additional Controls for Distortion Product Ratio

Harmonics

For a graph of the ratio of one specific harmonic product to the fundamental level, select **Single Harmonic** and set the **Harmonic Number** to your choice, from 2nd harmonic to 20th harmonic.

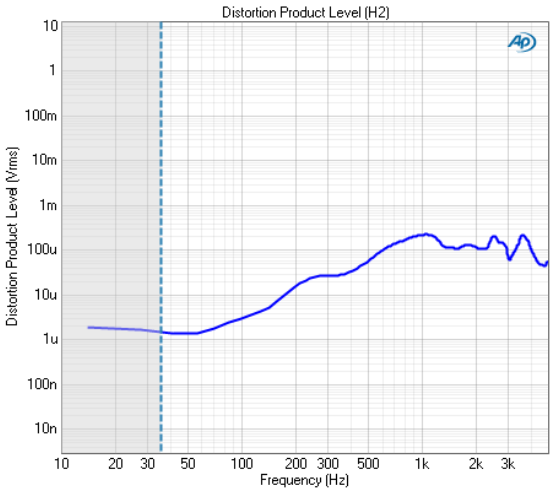
For a graph of the sum of the levels of any combination of harmonic products (from **H2** through **H20**), divided by the fundamental level, select **Sum of Harmonics**. Then click **Select Harmonics...** to open a result definition dialog.

Units

Units available for Loudspeaker Production Test Distortion Product Ratio are:

X-axis	*Y-axis*
• Hz	• x/y
• dHz	• %
• F/R	• ppm
• %Hz	• dB

Distortion Product Level



The Loudspeaker Production Test Distortion Product Level result is a continuous sweep measurement that provides a graphical display of the selected harmonic distortion products. In this result the level of the selected harmonic distortion product in the DUT output is plotted against frequency.

Additional Controls for Distortion Product Level

Harmonics

For a graph of the level of one specific harmonic product, select **Single Harmonic** and set the **Harmonic Number** to your choice, from 2nd harmonic to 20th harmonic.

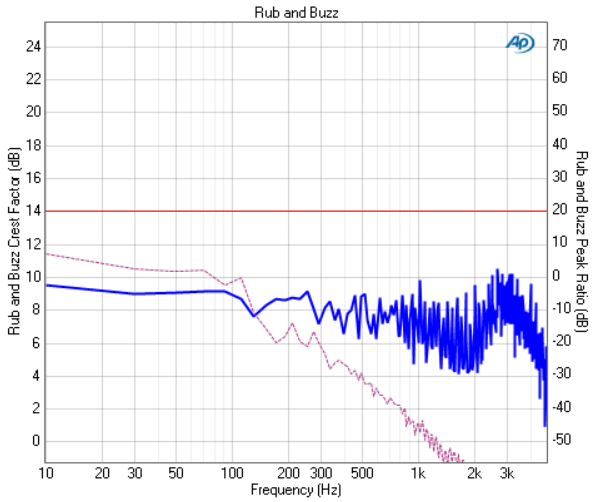
For a graph of the level of the sum of several or all harmonic products (from **H2** through **H20**), select **Sum of Harmonics**, then click **Select Harmonics...** to open a result definition dialog.

Units

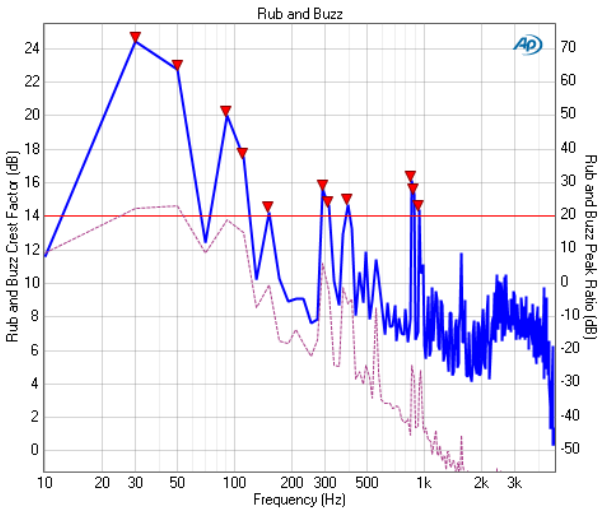
Units available for Loudspeaker Production Test Distortion Product Level are:

X-axis	*Y-axis (analog)*
• Hz	• Vrms
• dHz	• dBV
• F/R	• dBu
• %Hz	• dBrA
	• dBrB
	• dBSPL1
	• dBSPL2
	• dBm
	• W (watts)

Rub and Buzz



Rub and Buzz good result



Rub and Buzz poor result

The Loudspeaker Production Test: Rub and Buzz result uses a continuous sweep measurement to display the Rub and Buzz Crest Factor and Peak Ratio. See More about Loudspeaker Testing on page 211 for a more detailed discussion of using rub and buzz in loudspeaker testing.

Rub and Buzz presents two results on one graph. In the APx implementation, detection of rub and buzz requires a simultaneous interpretation of both of these results; a rub and buzz crest factor result is plotted in reference to the left Y-axis, and a rub and buzz peak ratio result is plotted in reference to the right Y-axis. Limit behavior is different on a graph with 2 Y-axes. See Drawing limits for XY Graphs (two Y-axes) on page 673 and Editing limits for XY Graphs (two Y-axes) on page 673.

You can split the dual axis graph into two individual graphs. Right-click on the Rub

and Buzz result and choose Split from the context menu. Two new results will be added, presenting the data for the Left and Right axes individually.

Additional Control for Rub and Buzz

High-pass Factor

The data plotted on these graphs is high-pass filtered to reveal the rub and buzz signature results. The filter tracks the chirp stimulus frequency, moving upward with the sweep. The corner frequency of the filter is the product of the instantaneous chirp frequency multiplied by the **High-pass Factor**, which can be set in the range from 5 to 30. This results in a graph where the X-axis represents the stimulus frequency, and the plotted data represent the much higher frequency sounds generated by the interaction of the stimulus frequency with loose particles or rubbing components within the loudspeaker system. For interpretation of the results, see More About Loudspeaker Testing on page 211.

Units

Units available for Rub and Buzz are:

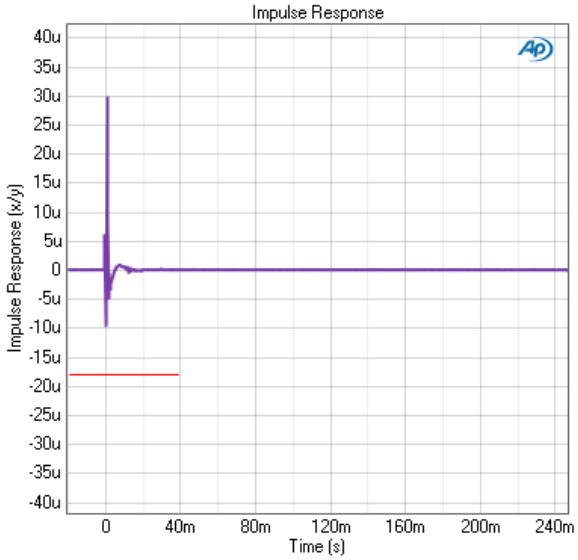
X-axis	*Y-axes*
• Hz	• x/y
• dHz	• %
• F/R	• ppm
• %Hz	• dB

Impulse Response

The Loudspeaker Production Test: Impulse Response view is a continuous sweep result that displays the Impulse Response of the system under test. This result is particularly useful in production test to verify correct system polarity.

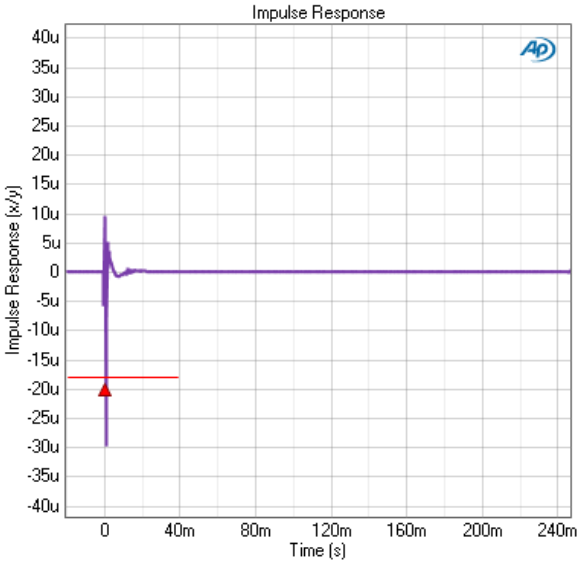
See page 244 for more information about impulse response.

Impulse Response correct polarity



This is a Loudspeaker Production Test Impulse Response result from a system that has correct polarity. The primary impulse is positive. Note the limit set at –18u, to flag reversed polarity.

Impulse Response reversed polarity



This is a Loudspeaker Production Test Impulse Response result from a system that has reversed polarity. The primary impulse is negative. Notice that the limit set at –18u has been crossed and a limit failure marker has been set.

Result Settings bar

Interpolation

For time-domain displays, the APx500 graphs are normally plotted using sinc function interpolation. Interpolation adds points to the displayed trace that do not exist in the measured data, to make data trends more

easily visualized. The default in APx is to have Interpolation **On**. However, digital domain signals are sometimes best understood when viewing actual samples, with interpolation switched **Off**.

Interpolation and limit failure markers.

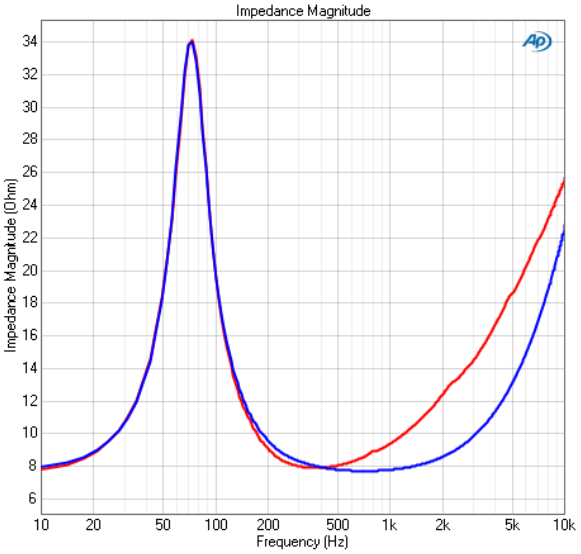
Limit failure markers (small orange triangles) are placed at the first data point that exceeds a limit. When interpolation is on and the graph is zoomed in to high magnification, the limit failure markers may appear to be "floating" in the graph, not at the intersection of the limit line and the interpolated data trace. Although this appears to be an error, it is actually correct behavior. Turn interpolation **Off**, and you will see that the marker is at the first (real) data point that exceeds the limit. There is no data at the intersection, although interpolation suggests there is. See an Illustration of interpolation and limit marker interaction on page 678.

Units

Units available for Impedance/Thiele-Small Impulse Response are:

X-axis	*Y-axis*
• s (seconds)	• x/y

Impedance Magnitude



This result provides a graphical display of the impedance response of the driver, plotting the complex impedance magnitude versus frequency.

Unprocessed/Processed

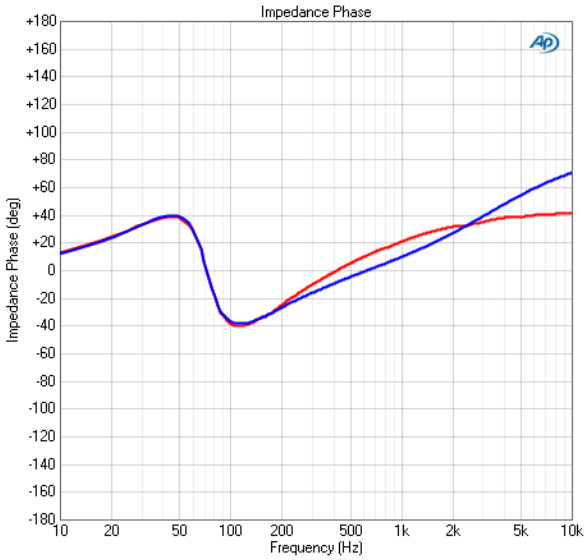
You can choose to view the raw, unprocessed result data, or the processed result data using the **Unprocessed / Processed** control on the graph display. The choice of **Unprocessed / Processed** viewing does not affect the operation of the measurement.

Units

Units available for Impedance/Thiele-Small Impedance Magnitude are:

X-axis	*Y-axis*
• Hz	• ohms
• dHz	
• F/R	
• %Hz	

Impedance Phase



This result provides a graphical display of the impedance phase response of the driver, plotting the impedance phase versus frequency.

Unprocessed/Processed

You can choose to view the raw, unprocessed result data, or the processed result data using the **Unprocessed / Processed** control on the graph display. The choice of **Unprocessed / Processed** viewing does not affect the operation of the measurement.

Units

Units available for Impedance/Thiele-Small Impedance Phase are:

X-axis	*Y-axis*
• Hz	• deg
• dHz	• rad
• F/R	
• %Hz	

Thiele-Small Parameters

Parameter	Value	Lower Limit	Upper Limit
F_S	50.50 Hz	----	----
Q_{MS}	2.07	----	----
Q_{ES}	0.56	----	----
Q_{TS}	0.44	----	----
S_D	452.39 cm^2	----	----
R_E	5.51 Ω	----	----
L_E	0.38 mH	----	----
R_2	---- Ω	----	----
L_2	---- mH	----	----
K_{rm}	----	----	----
E_{rm}	----	----	----
K_{xm}	----	----	----
E_{xm}	----	----	----
R_{MS}	14.85 Ns/m	----	----

Thiele-Small parameters

Thiele-Small populates a table with values calculated from the impedance response measurement data. When compared to a known good set of Thiele-Small parameters for a golden unit, this subset can be useful in identifying manufacturing faults in drivers and loudspeakers. R_2 and L_2 are only populated when the LR-2 model is selected, and E_{rm}, K_{rm}, E_{xm} and K_{xm} are only populated when the Wright model is selected.

Setting Limits for Thiele-Small parameters

Parameter	Value	Lower Limit	Upper Limit
F_S	55.37 Hz	53.0	56.0
Q_{MS}	2.27	2.00	2.25
Q_{ES}	0.63	----	----
Q_{TS}	0.50	----	----

Thiele-Small parameters with limits applied (detail)

You can set arbitrary limits for any or all of the Thiele-Small parameters by entering limit values into the results grid. Results that fall outside of the limits will be highlighted in the grid, and in a sequence the measurement will be flagged as **Failed**.

Maximum Output
(Sequence Mode)

Introduction

For the APx515, this measurement requires a software option key. See page 189 for more information about software options.

Maximum Output measurements use a sine wave stimulus signal that is adjusted to the level that produces the target distortion in a selected channel using regulation. This method is often employed to perform measurements at an amplifier's maximum output level.

Go to page 481 for more information about regulation.

Maximum Output measurements are not available in External Source or File Input configurations.

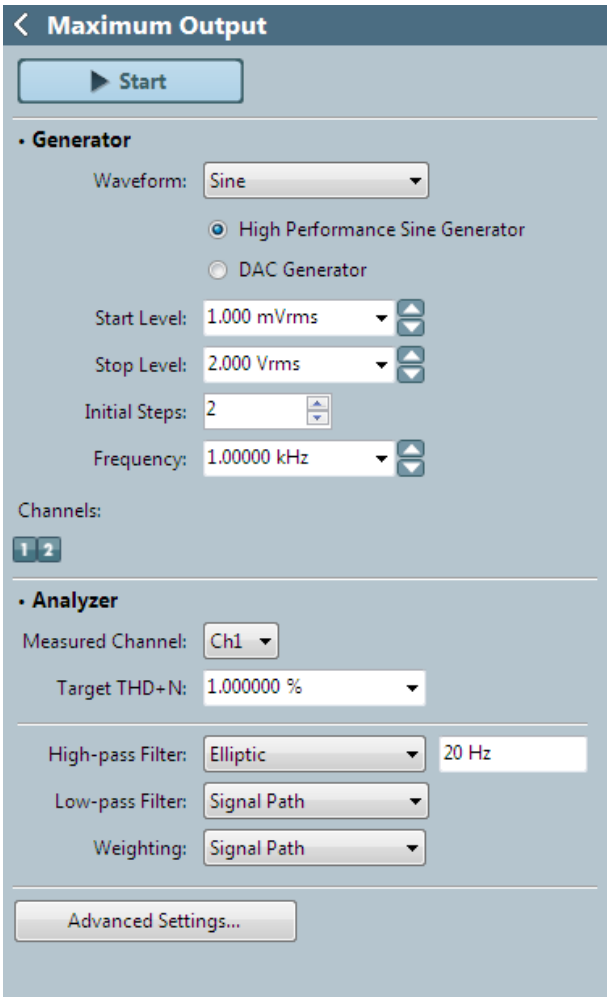
Maximum Output results available in APx500 are:

- RMS Level
- THD+N Level
- THD+N Ratio.

Operation

If you have not yet set up your test, first go to Signal Path Setup to select and configure inputs and outputs, references and other settings. See Chapter 7 for information about Signal Path Setup.

Click **Start**. The generator will output a sine wave to the DUT on the selected generator channels at the frequency set in the Signal Generation panel. The generator level is automatically adjusted while the analyzer observes the **Measured Channel** to achieve the level of distortion you specified. If another channel exhibits more distortion than the channel you initially selected, you can set that channel as the **Measured Channel** and run the measurement again.



Generator

Selecting the generator waveform

By default, the Maximum Output measurement uses a sine wave at the frequency set in the Signal Generation panel as the test signal.

High Performance Sine Generator / DAC Generator

For an APx555, you have the option of using the High Performance Sine Generator (very low distortion sine waves and wide bandwidth) or the DAC generator, which provides special sine waveforms. See High Performance Sine Generator starting on page 51 for a comparison of features.

Start Level

The **Start Level** is the first generator setting where a measurement is made. If it is lower than the **Stop Level**, then it is also the lowest generator setting where the measurement is made. If it is higher than the **Stop Level**, then it is the highest generator setting where the measurement is made.

The regulation search proceeds from Start Level to Stop Level; your choices for these values determine the direction (up or down) of the search.

Stop Level

If the **Stop Level** is higher than the **Start Level**, then the **Stop Level** is the highest generator setting where a measurement is made. If the **Stop Level** is lower than the **Start Level**, then the **Stop Level** is the lowest generator setting where a measurement is made. The **Stop Level** is also the last generator setting where a measurement is made in the initial search for an interval containing a measured value of the type that is sought.

The higher of Start Level or Stop Level also serves to limit the maximum APx generator output, protecting your DUT.

Offset

For the DAC generator, you can enter an optional DC offset here.

Initial Steps

Initial Steps sets the number of intervals dividing the generator level range between **Start Level** and **Stop Level**. The default setting is **2**. Depending upon the parameter to be regulated and characteristics of the DUT, greater values may provide better performance.

Frequency

Choose the generator frequency here.

Analyzer

Measured Channel

Choose a channel to measure for the **Target THD+N**.

Target THD+N

Choose a target distortion and enter the value in the Target **THD+N field.**

Filters

Local high-pass, low-pass and weighting filters are available for this measurement. The low-pass and high-pass filters, if selected, are applied to the entire measured signal and affect all the results. The weighting filter, if selected, affects only the distortion and distortion plus noise results. See Measurement Filters on page 619 for detailed information about the filters available locally for some measurements.

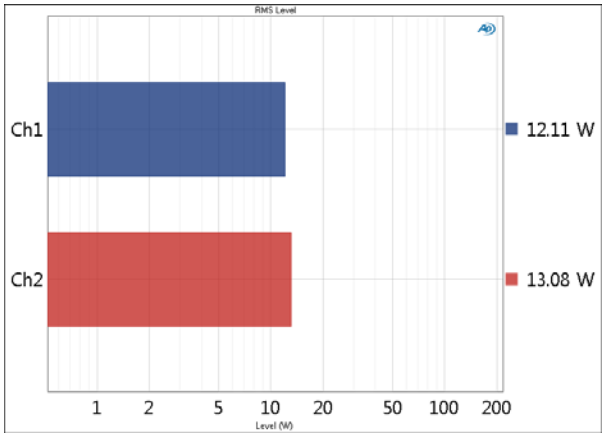
Low pass, high pass and weighting Signal Path Filters settings (see page 617) made in Signal Path Setup > Input/Output will also affect this measurement. When set, the local low-pass or high-pass filter setting overrides the corresponding Signal Path filter setting while the measurement is active. A local weighting filter setting does not override any Signal Path filter settings; the local weighting filter is applied in addition to any active Signal Path filters.

Advanced Settings

If your test requires individual generator settings for each channel or special analyzer ranging or settling, click **Advanced Settings**. See Advanced Settings for Maximum Output on page 369.

See Chapter 104 for more information about units of measurement.

Maximum Output: Level



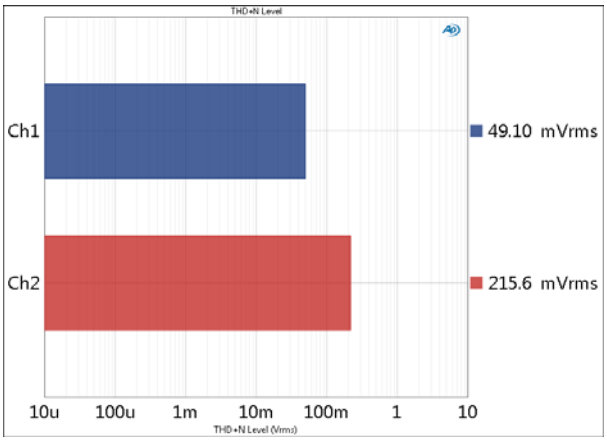
Read the regulated RMS Level for each channel from the meter bar display.

Units

Units available for Maximum Output: RMS Level are

Analog Signals	*Digital Signals*
• Vrms	• dBFS
• dBV	• FS
• dBu	• %FS
• dBrA	• dBrA
• dBrB	• dBrB
• dBSPL1	• dBSPL1
• dBSPL2	• dBSPL2
• dBm	
• W (watts)	

Maximum Output measurements: THD+N Level



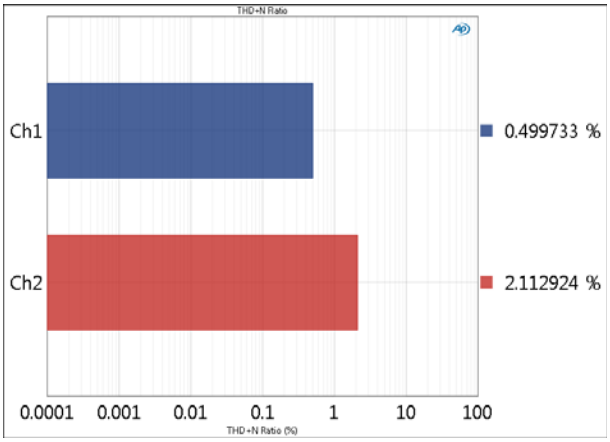
Read the THD+N Level for each channel from the meter bar display.

Units

Units available for Maximum Output: THD+N Level are

Analog Signals	*Digital Signals*
• Vrms	• dBFS
• dBV	• FS
• dBu	• %FS
• dBrA	• dBrA
• dBrB	• dBrB
• dBSPL1	• dBSPL1
• dBSPL2	• dBSPL2
• dBm	
• W (watts)	

Maximum Output measurements: THD+N Ratio



Read the THD+N Ratio for each channel from the meter bar display.

Units

Units available for Maximum Output: THD+N Ratio are

- x/y
- %
- ppm
- dB

Advanced Settings for Maximum Output

The default settings here are appropriate for most measurements. You may want to make minor adjustments for special situations.

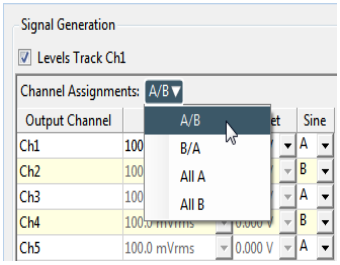
Signal Generation

No advanced settings for signal generation level are available for Maximum Output (CEA-2006) or Regulated Frequency Sweep measurements.

Set Channel Assignments (Maximum Output measurement)

For special waveforms like Sine Dual, you can set channel mapping. By default, channel A audio is mapped to odd numbered output channels, channel B audio to even numbered output channels. These assignments can be remapped by changing individual

settings in the **Sine** column, or by selection one of several presets from the **Set Channel Assignments** menu.



Signal Acquisition and Analysis

Input Range tab

Digital inputs do not require ranging, and the Input Range tab is unavailable when inputs are set to a digital format.

Auto Range

If **Auto Range** is checked (the default), each analog channel determines its input range automatically, based on the level of the input signal. If the input signal level changes beyond a ranging threshold, **Auto Range** will cause the input ranging circuits to move up or down for proper ranging.

Fixed Range

Alternatively, you can set a fixed minimum range for the analog inputs. Uncheck **Auto Range** and enter the expected maximum level in the **Minimum Range** filed. The software will choose the lowest range that will accommodate that level; this becomes the minimum range available, and the analyzer will not reset to a lower range. Fixed range is best for signals that vary rapidly, including tone bursts, perceptual audio speech samples, and program material such as voice or music. See **When autoranging isn't suitable** on page 626.

If a signal greater then the **Minimum Range** is presented to the analog input, the ranging circuits will range up to accommodate the higher level, and then drop again when the signal level falls. The range will not be set below the **Minimum Range**.

Ranges Track Ch1

If **Ranges Track Ch1** is checked (the default), the fixed input range setting for channel 1 is copied to the other channels, and range settings for channels greater than 1 cannot be edited. Any changes made to channel 1 are reproduced in the other channels. If **Ranges Track Ch1** is not checked, the fixed input range setting for each channel can be set independently.

Read more about ranging in Chapter 98.

Settling tab

Two Settling tabs are available for regulated measurements, Settling (Level) and Settling (THD+N).

See Chapter 98 for more information about Settling for regulated measurements.

Maximum Output (CEA-2006)
(Sequence Mode)

Introduction

For the APx515, this measurement requires a software option key. See page 189 for more information about software options.

The Maximum Output (CEA-2006) measurements provide single-value results that show the DUT output for each channel during the burst ON interval, after regulation is performed. Maximum Output (CEA-2006) measurements use a tone-burst stimulus signal that is adjusted to the level that produces a target distortion in a selected channel (regulated). This method is specified in the standard CEA-2006 "Testing & Measurement Methods for Mobile Audio Amplifiers," section 5.2, Dynamic Power testing.

Go to page 481 for more information about regulation, and to page 374 for more information about burst measurements.

Maximum Output (CEA-2006) results are not available in External Source or File Input configurations.

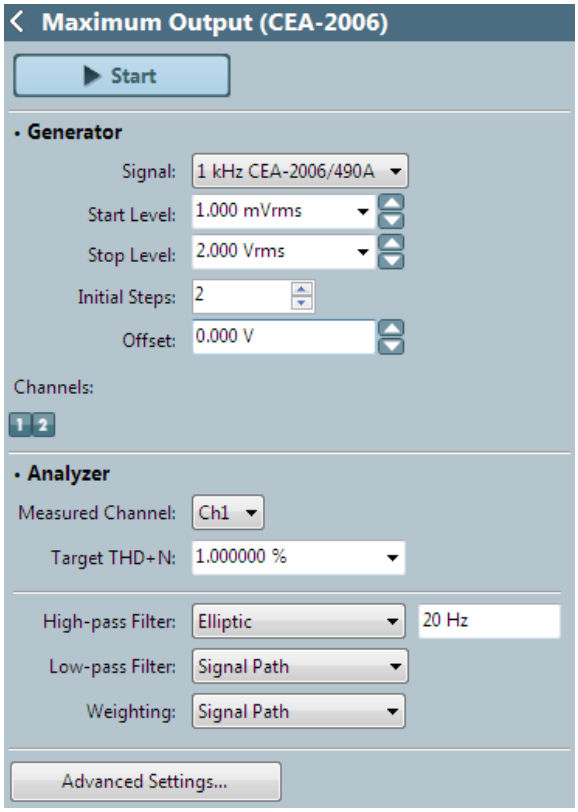
Maximum Output (CEA-2006) results available in APx500 are:

- RMS Level
- THD+N Level
- THD+N Ratio.

Operation

If you have not yet set up your test, first go to Signal Path Setup to select and configure inputs and outputs, references and other settings. See Chapter 7 for information about Signal Path Setup.

Click **Start**. The generator will output a sine burst to the DUT on the selected generator channels at the frequency set in the Signal Generation panel. The generator is automatically adjusted while the analyzer observes the **Measured Channel** during the burst ON interval to achieve the level of distortion you specified. If another channel exhibits more distortion than the channel you initially selected, you can set that channel as the **Measured Channel** and run the measurement again.



Generator

Signal

By default, the Maximum Output (CEA-2006) measurement uses a 1 kHz CEA-2006/490A sine burst as the test signal. You can optionally select a 50 Hz CEA-2006/490A sine burst as the test signal.

Start Level

The **Start Level** is the first generator setting where a measurement is made. If it is lower than the **Stop Level**, then it is also the lowest generator setting where the measurement is made. If it is higher than the **Stop Level**, then it is the highest generator setting where the measurement is made.

The regulation search proceeds from Start Level to Stop Level; your choices for these values determine the direction (up or down) of the search.

Stop Level

If the **Stop Level** is higher than the **Start Level**, then the **Stop Level** is the highest generator setting where a measurement is made. If the **Stop Level** is lower than the **Start Level**, then the **Stop Level** is the lowest generator setting where a measurement is made. The **Stop Level** is also the last generator setting where a measurement is made in the initial search for an interval containing a measured value of the type that is sought.

The higher of Start Level or Stop Level also serves to limit the maximum APx generator output, protecting your DUT.

Initial Steps

Initial Steps sets the number of intervals dividing the generator level range between **Start Level** and **Stop Level**. The default setting is **2**. Depending upon the parameter to be regulated and characteristics of the DUT, greater values may provide better performance.

Offset

Set an optional DC offset voltage here.

Analyzer

Measured Channel

Choose a channel to measure for the **Target THD+N**.

Target THD+N

Choose a target distortion and enter the value in the Target **THD+N field.**

Filters

Local high-pass, low-pass and weighting filters are available for this measurement. The low-pass and high-pass filters, if selected, are applied to the entire measured signal and affect all the results. The weighting filter, if selected, affects only the distortion and distortion plus noise results. See Measurement Filters on page 619 for detailed information about the filters available locally for some measurements.

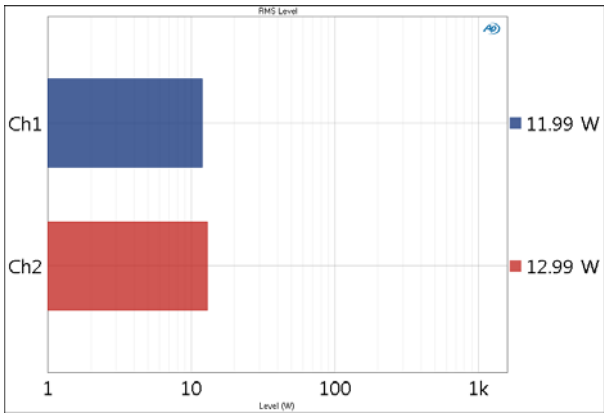
Low pass, high pass and weighting Signal Path Filters settings (see page 617) made in Signal Path Setup > Input/Output will also affect this measurement. When set, the local low-pass or high-pass filter setting overrides the corresponding Signal Path filter setting while the measurement is active. A local weighting filter setting does not override any Signal Path filter settings; the local weighting filter is applied in addition to any active Signal Path filters.

Advanced Settings

If your test requires individual generator settings for each channel or special analyzer ranging or settling, click **Advanced Settings**. See Advanced Settings for Burst Measurements on page 373.

See Chapter 104 for more information about units of measurement.

Maximum Output (CEA-2006): Level



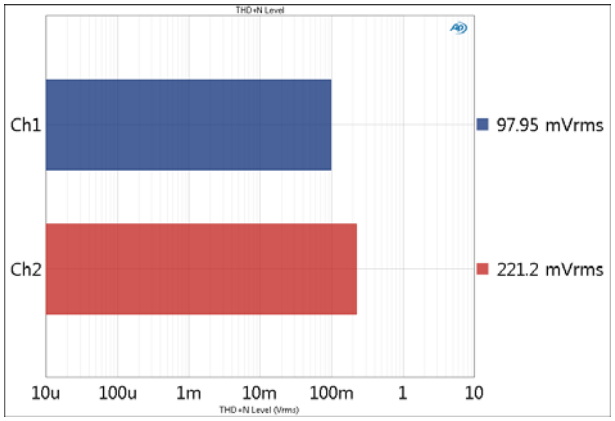
Read the RMS Level for each channel from the meter bar display.

Units

Units available for Maximum Output (CEA-2006): RMS Level are

Analog Signals	*Digital Signals*
• Vrms	• dBFS
• dBV	• FS
• dBu	• %FS
• dBrA	• dBrA
• dBrB	• dBrB
• dBSPL1	• dBSPL1
• dBSPL2	• dBSPL2
• dBm	
• W (watts)	

Maximum Output (CEA-2006): THD+N Level



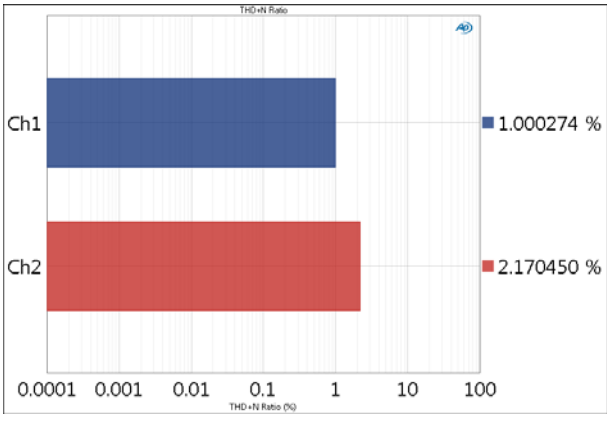
Read the THD+N Level for each channel from the meter bar display.

Units

Units available for Maximum Output (CEA-2006): THD+N Level are

Analog Signals	*Digital Signals*
• Vrms	• dBFS
• dBV	• FS
• dBu	• %FS
• dBrA	• dBrA
• dBrB	• dBrB
• dBSPL1	• dBSPL1
• dBSPL2	• dBSPL2
• dBm	
• W (watts)	

Maximum Output (CEA-2006): THD+N Ratio



Read the THD+N Ratio for each channel from the meter bar display.

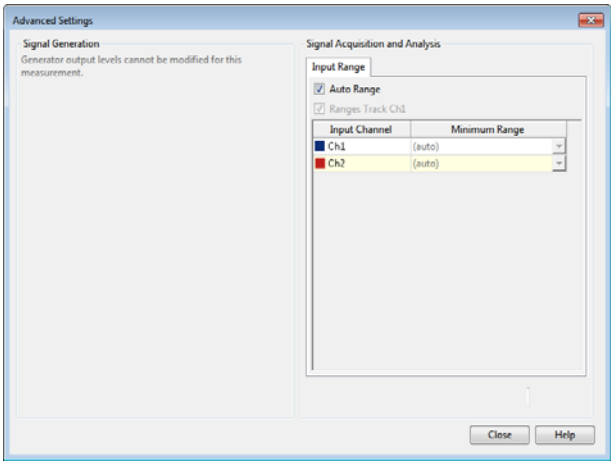
Units

Units available for Maximum Output (CEA-2006): THD+N Ratio are

- x/y
- %
- ppm
- dB

Advanced Settings for Burst Measurements

The default settings here are appropriate for most measurements. You may want to make minor adjustments for special situations.



Signal Generation

There are no advanced settings for signal generation for regulated burst measurements.

Signal Acquisition and Analysis

Input Range tab

Digital inputs do not require ranging, and the Input Range tab is unavailable when inputs are set to a digital format.

Auto Range

Maximum Output (CEA-2006) uses a modified auto-ranging process to find the correct input range.

The burst measurement acquisition begins in the minimum range and moves up from there if necessary. You can specify a higher range by entering the value in the **Minimum Range** field.

The regulation algorithm moves the range upward if necessary while searching for the target distortion.

If your Maximum Output (CEA-2006) measurement is a step in an automated sequence that you would like to run as fast as possible, you can optimize the speed of the ranging process by setting the minimum range to the correct range for the measurement.

Fixed Range

Alternatively, you can set a fixed minimum range for the analog inputs. Uncheck **Auto Range** and enter the expected maximum level in the **Minimum Range** filed.

The software will choose the lowest range that will accommodate that level; this becomes the minimum range available, and the analyzer will not reset to a lower range. Fixed range is best for signals that vary rapidly, including tone bursts, perceptual audio speech samples, and program material such as voice or music. See **When autoranging isn't suitable** on page 626.

If a signal greater then the **Minimum Range** is presented to the analog input, the ranging circuits will range up to accommodate the higher level, and then drop again when the signal level falls. The range will not be set below the **Minimum Range**.

Ranges Track Ch1

If **Ranges Track Ch1** is checked (the default), the fixed input range setting for channel 1 is copied to the other channels, and range settings for channels greater than 1 cannot be edited. Any changes made to channel 1 are reproduced in the other channels. If **Ranges Track Ch1** is not checked, the fixed input range setting for each channel can be set independently.

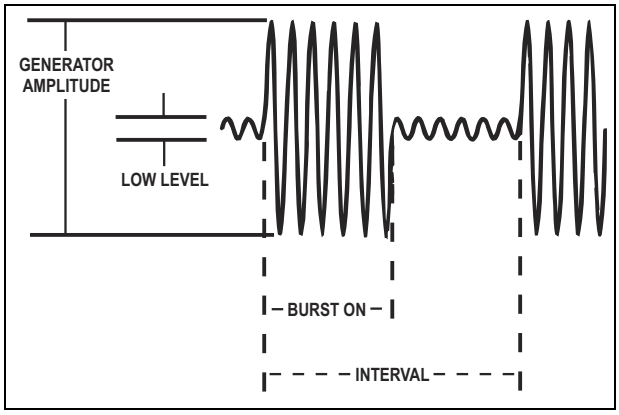
Read more about ranging in Chapter 98.

No settling is available for Maximum Output (CEA-2006) results.

More about burst measurements

"Tone bursts" have been used for many years as stimulus signals for audio testing.

A burst signal is usually a sine wave that is switched between two levels (that is, modulated by a square wave), with the higher level typically 100% of the set generator amplitude and the lower level typically 10% (–20 dB). The "duty cycle" of the burst is usually less than 50% (the "on" time is less than the "off" time). Well-formed tone burst signals will have an integer number of sine cycles in the burst, and the beginning and end of the burst will be zero-crossing.



Tone Burst Diagram

Other common burst signals include "shaped bursts" (sine waves modulated by a wave shape other than a square wave).

Burst signals for CEA-2006 in APx500

The burst signals in APx500 are designed to satisfy the requirements of the standard CEA-2006 "Testing & Measurement Methods for Mobile Audio Amplifiers," section 5.2, Dynamic Power testing.

Two signals are specified in the standard: for full-range amplifiers, the signal shall be a repetition of a burst of 20 cycles of a 1 kHz sine wave at 100%, followed by 480 cycles of 1 kHz at 10% (–20 dB). For limited-range amplifiers (subwoofer amplifiers), the signal shall be a repetition of a burst of 10 cycles of a 50 Hz sine wave at 100%, followed by 20 cycles of 50 Hz at 10% (–20 dB).

APx500 provides both these stimulus signals in the Max Output (Burst) measurement result.

Analyzing burst signals in APx500

The first few cycles of the burst are ignored to minimize spurious results caused by initial transients. For the 1 kHz burst, the first 9 cycles are ignored and the last 11 cycles are analyzed. For the 50 Hz burst, the first 2 cycles are ignored and the last 8 cycles are analyzed. Any delay in the DUT is automatically taken into account.

Measurement Recorder
(Sequence Mode)

The Measurement Recorder is a general-purpose diagnostic tool that provides a record of level, gain, interchannel phase, THD+N ratio or level, DC level, frequency, or SINAD versus elapsed time. The record can be very long (up to one week), providing a means of monitoring the output of a device under test over an extended period of time.

The Measurement Recorder does not require a specific test signal. It can be used with any audio signal within the input range of the analyzer.

Measurement Recorder results available in APx500 are:

- RMS Level
- Gain
- Phase
- THD+N Ratio
- THD+N Level
- THD Ratio
- THD Level
- SINAD
- Bandpass Level
- Frequency
- DC Level
- Peak Level
- Average Jitter Level

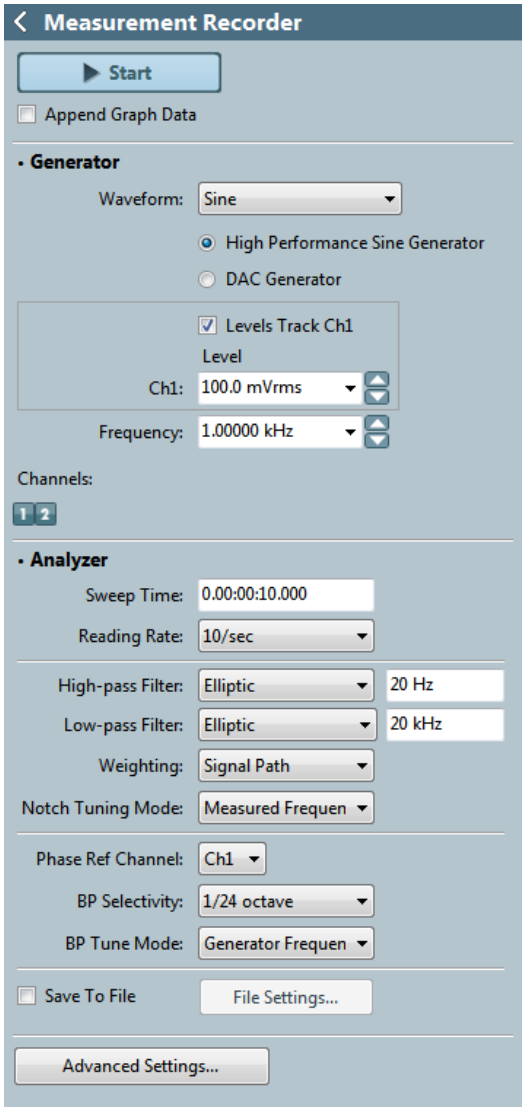
Gain is not available in external source configuration, or if a square wave, a generator waveform or File Input has been selected for the measurement.

Operation

If you have not yet set up your test, first go to Signal Path Setup (Chapter 7) to select and configure inputs and outputs, references and other settings.

Append Graph Data

Measured data are grouped in a Data Set, which contains all acquisition results. Normally, the current graph data are deleted each time you start a new measurement. If the **Append Graph Data** box is checked, appended data are grouped as additional data sets, as are any imported data. The Data Sets Panel displays the Data Set(s) in the measurement, and provides a visibility checkbox and a Notes field for each data set. See page 649 for more information about managing Data Sets.



Running the measurement

To use Measurement Recorder: Level, click **Start**. The generator will output the test signal to the DUT on the selected generator channels. The measurement recording will begin and the graph will display as the

record is acquired. The record will continue until the end time set in **Sweep Time**.

Note: For digital inputs, sample rate must be stable before the Measurement Recorder begins. When playing a digital external source (such as a DVD), we advise that you start the playback device first, then click the measurement Start button.

Generator

The typical configuration for this measurement is the closed-loop configuration, using the APx generator as a stimulus. See Signal Generation (Chapter 6) for detailed information about the using the APx Generator and setting Waveform, Level and Frequency.

Loop Waveform

Most of the operation when using a Generator Waveform is described in Chapter 6. Measurement Recorder has an additional control in this mode. By default, generator waveforms are looped in the generator, providing a continuous generator output. If Loop Waveform is unchecked, the generator waveform is played once, and then ends.

Analyzer

Sweep Time

Sweep Time sets the length of the measurement record. Minimum time is 0.1 s, maximum time is 7 days.

Enter the **Sweep Time** following this pattern:

d:hh:mm:ss.s

where d = days, hh = hours, mm = minutes, ss = seconds. Days and fractional seconds are optional.

Reading Rate

Reading Rate sets a maximum number of times the input signal will be read per second. In most cases the number of readings will be very close to this value. However, the rate is dependent upon PC performance and channel count, and may be lower. See More About Reading Rate on page 383.

Filtering

Local high-pass, low-pass and weighting filters are available for this measurement. The low-pass and high-pass filters, if selected, are applied to the entire measured signal and affect all the results. The weighting filter, if selected, affects only the distortion and distortion plus noise results. See Measurement Filters on page 619 for detailed information about the filters available locally for some measurements.

Low pass, high pass and weighting Signal Path Filters settings (see page 617) made in Signal Path Setup > Input/Output will also affect this measurement. When set, the local low-pass or high-pass filter setting overrides the corresponding Signal Path filter setting while the measurement is active. A local weighting filter setting does not override any Signal Path filter settings; the local weighting filter is applied in addition to any active Signal Path filters.

Notch Tuning Mode

For the harmonic distortion results, a tunable notch filter is used to remove the fundamental from the measurement. The filter can be tuned to:

- **Generator Frequency**
  The current APx audio generator frequency. When the generator channels are outputting different frequencies (Split Frequency generation), the notch filter center is set to Frequency A. This mode is not available when using a generator waveform file.
- **Jitter Generator Frequency**
  The current APx jitter generator frequency, when jitter generation is available and enabled. Read more about jitter beginning on page 69.
- **Measured Frequency**
  The current measured frequency. When the analyzer channels are receiving different frequencies, the notch filter for each channel is centered on the frequency in that channel.
- **Fixed Frequency**
  A fixed frequency selected by the user. When **Fixed Frequency** is selected, a **Filter Frequency** entry field becomes available beneath the **Notch Tuning Mode** control.

Phase Reference Channel

This selection is only relevant to the Measurement Recorder: Phase measurement. See page 378.

BP Selectivity

This control only affects the Bandpass Level result.

For the Bandpass Level result, you can choose from a number of bandpass filter widths using the **BP Selectivity** control. The list is ordered from narrowest at the top, to widest at the bottom. The **Window Width** and **x Octave** filters are tuned according to the **BP Tune Mode** setting.

- **Window width**
  This is the window width of the underlying FFT, typically only a few hertz wide. This selection has very steep skirts and a flat top.
- **1/24 octave**
  This selection has –3 dB points that are 1/24 octave apart.

- **1/12 octave**
  This selection has –3 dB points that are 1/12 octave apart.
- **1/9 octave**
  This selection has –3 dB points that are 1/9 octave apart.
- **1/6 octave**
  This selection has –3 dB points that are 1/6 octave apart.
- **1/3 octave**
  This selection has –3 dB points that are 1/3 octave apart.
- **1 octave**
  This selection has –3 dB points that are 1 octave apart.

All of the "x octave" filters are equivalent to second-order analog bandpass filters, with 6 dB/octave skirts. Q is chosen to achieve the specified bandwidth at the –3 dB points.

BP Min Freq: / BP Max Freq:

When **Rectangular Band** is the **BP Selectivity** choice, these fields become available for lower and upper frequency cutoff point entry.

BP Tune Mode

This control only affects the Bandpass Level result.

The bandpass **Window width** and **x octave** filters are centered at some user-defined frequency for each step of the measurement. This can be:

- **Generator Frequency**
  The current APx audio generator frequency. When the generator channels are outputting different frequencies (Split Frequency generation), the bandpass filter center is set to Frequency A.
- **Jitter Generator Frequency**
  The current APx jitter generator frequency, when jitter generation is available and enabled. Read more about jitter beginning on page 69.
- **Measured Frequency**
  The current measured frequency. When the analyzer channels are receiving different frequencies, the notch filter for each channel is centered on the frequency in that channel.
- **Fixed Frequency**
  A fixed frequency selected by the user. When **Fixed Frequency** is selected, a **Filter Frequency** entry field becomes available beneath the **BP Tune Mode** control.

Save to File

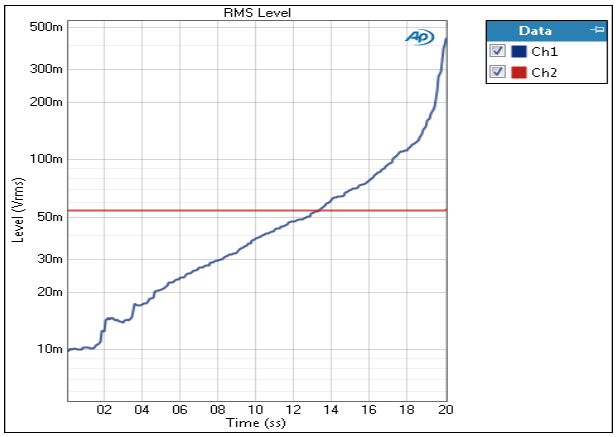
Click this option to save the acquired audio to a file on disk. See **Recording Audio to a File** on page 381.

Advanced Settings

If your test requires individual generator settings for each channel or special analyzer ranging or settling, see Advanced Settings for the Measurement Recorder on page 384. See Chapter 104 for more information about units of measurement.

RMS Level

The Measurement Recorder: RMS Level result will provide a record of RMS level versus elapsed time for any audio signal. The illustration shows a 20-second record of a 10 kHz sine wave on two channels, with a 10 kHz EQ control adjusted from about –20 dB to about +12 dB during the course of the recording.



Units

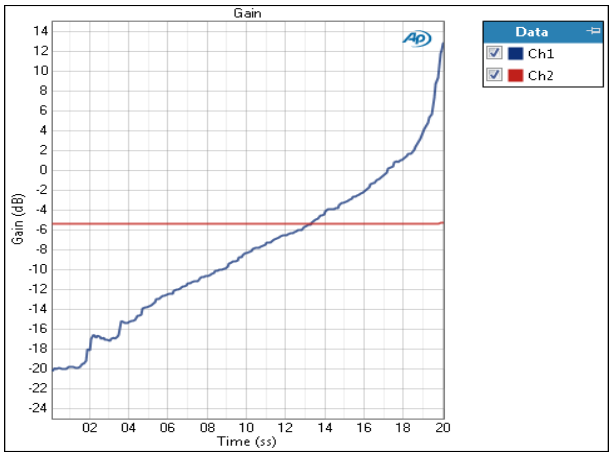
Units available for RMS Level are:

X-axis	*Y-axis (analog)*	*Y-axis (digital)*	*Jitter*
• s	• Vrms	• dBFS	• UI
	• dBV	• FS	• dBUI
	• dBu	• %FS	• s
	• dBrA	• dBrA	
	• dBrB	• dBrB	
	• dBSPL1	• dBSPL1	
	• dBSPL2	• dBSPL2	
	• dBm	• D	
	• W (watts)		
	• V		

Gain

The Measurement Recorder: Gain measurement result will provide a record of gain versus elapsed time for any audio signal. The illustration shows a 20-second record of a 10 kHz sine wave on two channels,

with a 10 kHz EQ control adjusted from about –20 dB to about +12 dB during the course of the recording.



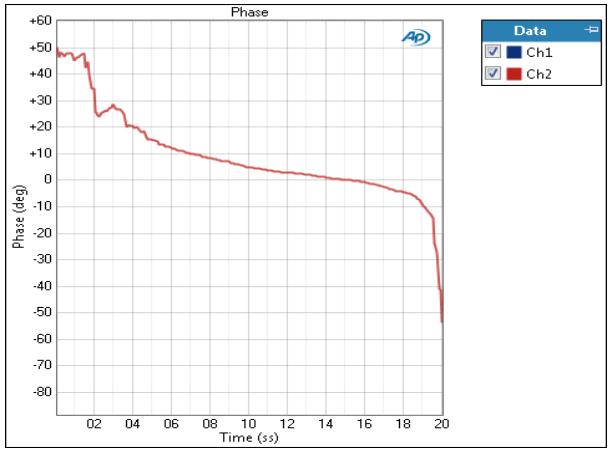
Units

Units available for: Gain are:

X-axis	*Y-axis (same-domain)*	*Y-axis (cross-domain)*
• s	• x/y • % • ppm • dB	• FS/Vrms • dB(FS/Vrms) or • Vrms/FS • dB(Vrms/FS)

Phase

The Measurement Recorder: Phase measurement result will provide a record of relative phase of the DUT channels at a single frequency. One channel is chosen as the phase reference channel, and the remaining channels are measured against it. The illustration shows a 20-second record of a 10 kHz sine wave on two channels, with a 10 kHz EQ control adjusted from about –20 dB to about +12 dB during the course of the recording. For more information about phase, see page 332.

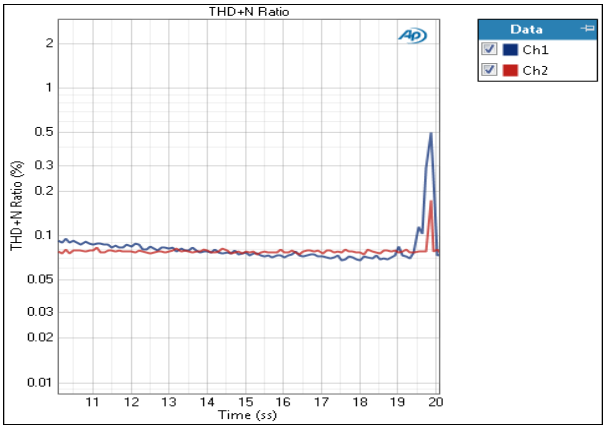


Units

Units available for Phase are:

X-axis	*Y-axis*
• s	• deg • rad

THD+N Ratio



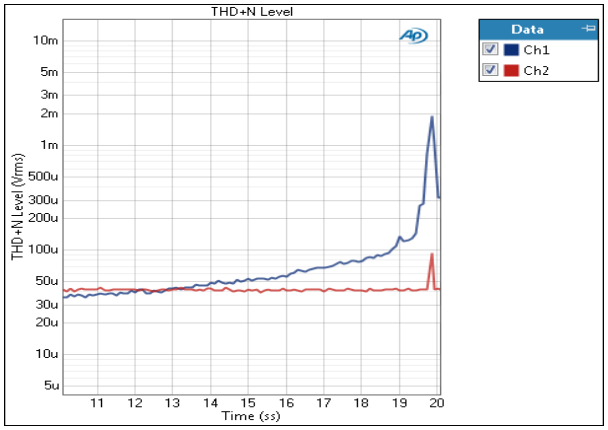
The Measurement Recorder: THD+N Ratio measurement result provides a record of the THD+N ratio versus elapsed time for any audio signal. The illustration shows a 20-second record of a 10 kHz sine wave on two channels, with a 10 kHz EQ control adjusted from about –20 dB to about +12 dB during the course of the recording.

Units

Units available for THD+N Ratio are:

X-axis	*Y-axis*
• s	• x/y • % • ppm • dB

THD+N Level



The Measurement Recorder: THD+N Level measurement result provides a record of THD+N level versus

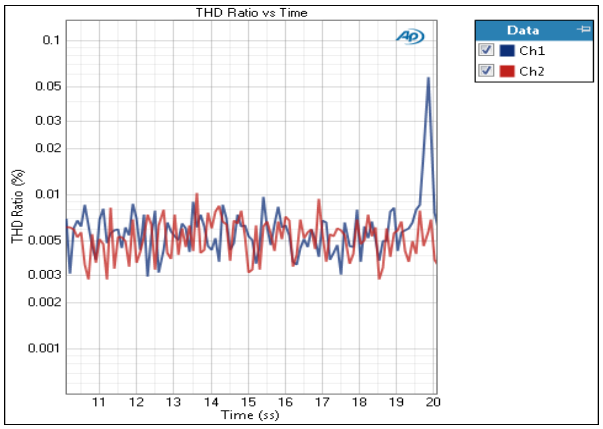
elapsed time for any audio signal. The illustration shows a 20-second record of a 10 kHz sine wave on two channels, with a 10 kHz EQ control adjusted from about –20 dB to about +12 dB during the course of the recording.

Units

Units available for THD+N Level are:

X-axis	*Y-axis (analog)*	*Y-axis (digital)*	*Jitter*
• s	• Vrms • dBV • dBu • dBrA • dBrB • dBSPL1 • dBSPL2 • dBm • W (watts) • V	• dBFS • FS • %FS • dBrA • dBrB • dBSPL1 • dBSPL2 • D	• UI • dBUI • s

THD Ratio



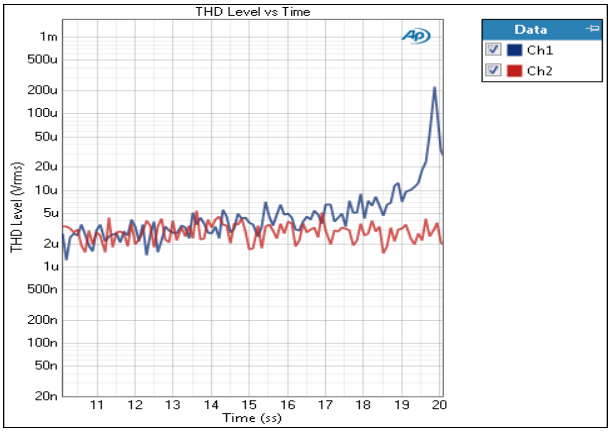
The Measurement Recorder: THD Ratio measurement result provides a record of the THD ratio versus elapsed time for any audio signal. The illustration shows a 20-second record of a 10 kHz sine wave on two channels, with a 10 kHz EQ control adjusted from about –20 dB to about +12 dB during the course of the recording.

Units

Units available for THD Ratio are:

X-axis	*Y-axis*
• s	• x/y • % • ppm • dB

THD Level



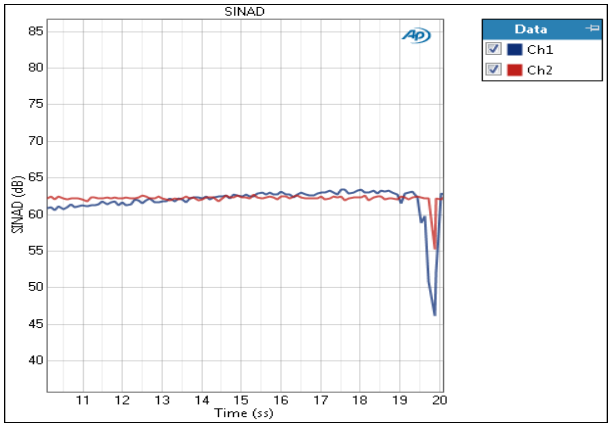
The Measurement Recorder: THD Level measurement result provides a record of THD level versus elapsed time for any audio signal. The illustration shows a 20-second record of a 10 kHz sine wave on two channels, with a 10 kHz EQ control adjusted from about –20 dB to about +12 dB during the course of the recording.

Units

Units available for THD Level are:

X-axis	*Y-axis (analog)*	*Y-axis (digital)*	*Jitter*
• s	• Vrms • dBV • dBu • dBrA • dBrB • dBSPL1 • dBSPL2 • dBm • W (watts) • V	• dBFS • FS • %FS • dBrA • dBrB • dBSPL1 • dBSPL2 • D	• UI • dBUI • s

SINAD



The Measurement Recorder: SINAD measurement result provides a record of the SINAD ratio versus elapsed time for any audio signal. The illustration

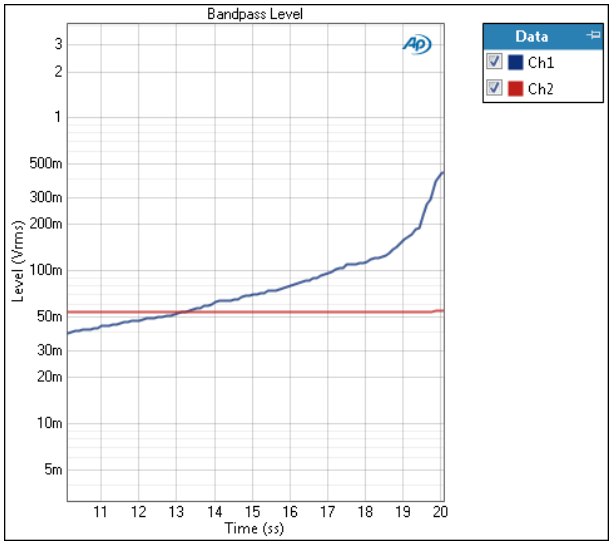
shows a 20-second record of a 10 kHz sine wave on two channels, with a 10 kHz EQ control adjusted from about –20 dB to about +12 dB during the course of the recording.

Units

Units available for SINAD are:

X-axis	*Y-axis*
• s	• x/y • dB

Bandpass Level



The Bandpass Level measurement result provides a record of the Bandpass Level versus elapsed time for any audio signal. The illustration shows a 20-second record of a 10 kHz sine wave on two channels, with a 10 kHz EQ control adjusted from about –20 dB to about +12 dB during the course of the recording.

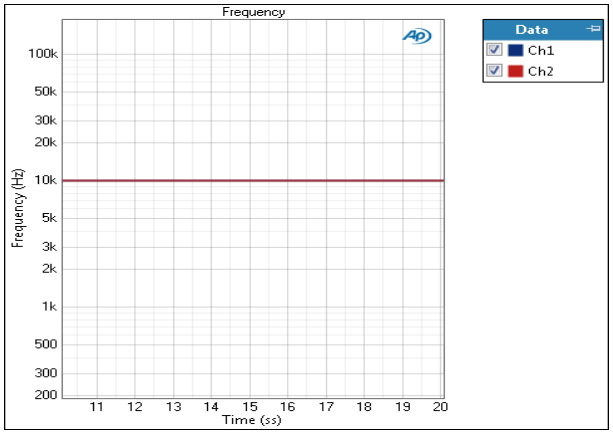
Units

Units available for Bandpass Level are:

X-axis	*Y-axis (analog)*	*Y-axis (digital)*	*Jitter*
• s	• Vrms • dBV • dBu • dBrA • dBrB • dBSPL1 • dBSPL2 • dBm • W (watts) • V	• dBFS • FS • %FS • dBrA • dBrB • dBSPL1 • dBSPL2 • D	• UI • dBUI • s

Frequency

The Measurement Recorder: Frequency measurement result will provide a record of frequency versus elapsed time for any audio signal. The illustration shows a 20-second record of a 10 kHz sine wave on two channels, with a 10 kHz EQ control adjusted from about –20 dB to about +12 dB during the course of the recording.

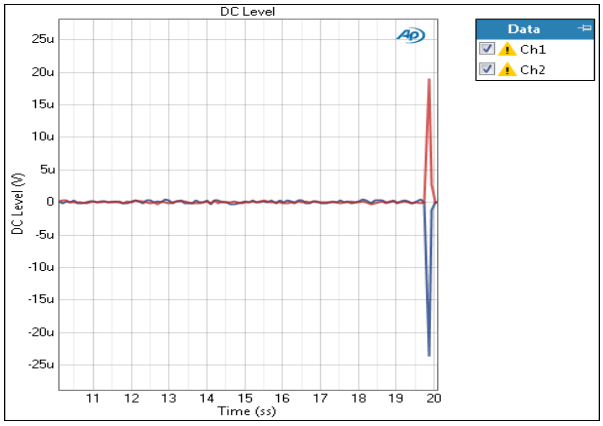


Units

Units available for Frequency are:

X-axis	*Y-axis*
• s	• Hz • dHz • F/R • %Hz

DC Level



The DC Level result is not available when the input is AC coupled.

The Measurement Recorder: DC Level measurement result will provide a record of DC level versus elapsed time for any signal. The Measurement Recorder does not require a specific test signal. It can be used with any audio signal within the input range of the analyzer.

Note: Although the Measurement Recorder result includes Generator and Filter controls, the DC Level measurement incorporates a very tight low-pass filter that rejects any audio signal and makes the analysis fil-

ters irrelevant. Measurement Recorder: DC Level measurements will return the same results whether or not there is an audio signal in the DUT, or whether or not there is filtering applied.

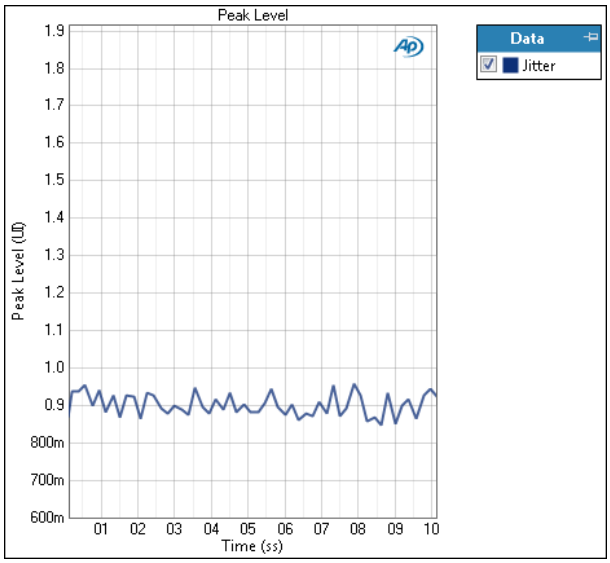
Go to page 266 for more information about DC Level.

Units

Units available for DC Level are:

X-axis	*Y-axis (analog)*	*Y-axis (digital)*
• s (seconds)	• V	• D • hex

Peak Level



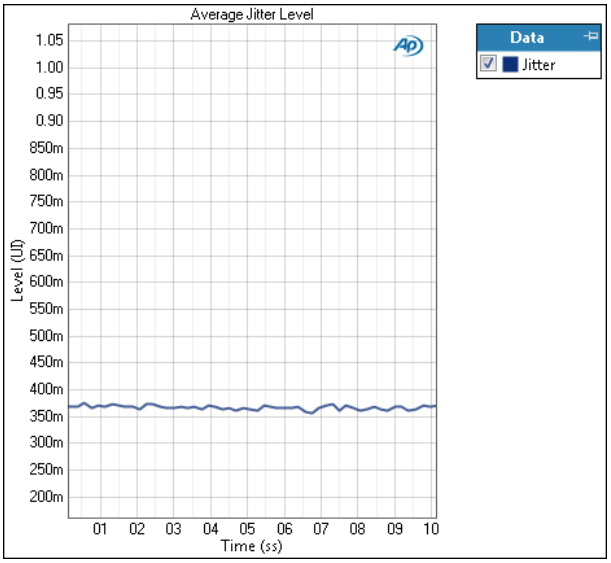
The **Measurement Recorder: Peak Level** result uses a sine wave stimulus signal that is moved across a range of levels in a series of points. X axis is generator level; Y axis is the bandpass filtered DUT output level, scaled in peak units.

Units

Units available for Peak Level are:

X-axis (analog)	*X-axis (digital)*	*Y-axis (analog)*	*Y-axis (digital)*
• Vrms • Vp • Vpp • dBV • dBu • dBrG • dBm • W (watts)	• dBFS • FS • %FS • dBrG	• V *Y-axis (jitter)* • UI • dBUI • s	• D

Average Jitter Level



The **Measurement Recorder: Average Jitter Level** result uses a sine wave stimulus signal that is moved across a range of levels in a series of points. X axis is generator level; Y axis is the bandpass filtered DUT output jitter level, scaled in average units.

Units

Units available for Average Jitter Level are:

X-axis (analog)	*X-axis (digital)*	*Y-axis (jitter)*
• Vrms • Vp • Vpp • dBV • dBu • dBrG • dBm • W (watts)	• dBFS • FS • %FS • dBrG	• UI • dBUI • s

Recording Audio to a File

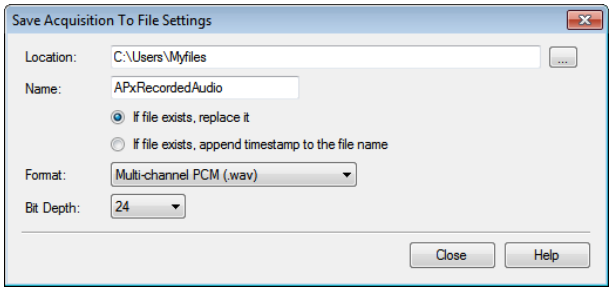
The Measurement Recorder and the Noise Recorder can optionally record acquired audio to a file on disk.

Note: APx timing constraints will sometimes cause a timeout error when saving to a network folder or a removable disk. We recommend saving audio files to a local disk.

On the Measurement or Noise Recorder panel, check the **Save to File** checkbox. Click **File Settings** to specify the file name, location and file format.

Save to File is not available when Input Configuration is set to File (Analog Units) or File (Digital Units).

Save Acquisition to File Settings dialog



Location

Enter or browse to the destination folder here.

Name

Name the output file here. Choose from one of these selections:

- If file exists, replace it
- If file exists, append timestamp to the file name

Generator Trigger

When Signal Generation is set to **Waveform**, there is always some delay between Measurement Recorder **Start** and the instant the waveform playback begins. When **Generator Trigger** is **Off** (unchecked), the file recording begins at the measurement **Start**. When **Generator Trigger** is **On** (checked), the file recording begins at the instant the waveform playback begins.

Format

Choose the file format in the **Format** field.

For both analog inputs and digital acquisitions, there are three linear file formats available:

Multiple Mono PCM (.wav)

If **Multiple Mono PCM (.wav)** (the default) is selected, each audio channel of the acquisition is saved as an individual monaural WAV file, using the Microsoft WAVE_FORMAT_PCM (Type 1, 24 bit) file format. For multichannel audio of channel count n, n mono WAV files are created, with the numeral n appended to the file name.

Multi-channel PCM (.wav)

If **Multi-channel PCM (.wav)** is selected, all the audio channels of the acquisition are saved in one multichannel file, using the WAVE_FORMAT_PCM format. For 1 channel and 2 channel acquisitions, this selection records a standard WAVE_FORMAT_PCM (Type 1, 24 bit). For multichannel acquisitions, this selection records a multichannel WAVE_FORMAT_PCM file in a proprietary 24-bit format, compatible with Sony Sound Forge and other audio editing applications.

Extensible Multi-channel PCM (.wav)

If **Save as Extensible Multi-channel PCM (.wav)** is selected, all the audio channels of the acquisition are saved in one multichannel file, using the Microsoft standard WAVE_FORMAT_EXTENSIBLE format, at 24 bits.

32-bit audio

The DSIO receiver will accept 32-bit audio. In such a case, the audio can be saved in any of the above formats, but the bit depth will be 32 bit fixed point.

Bit Depth

This field is only available for analog acquisitions. The default Bit Depth (word length) is 24 bits. You can also select 16 bits.

Analog scaling and sample rates

When recording an analog acquisition, proprietary scaling metadata is embedded in the WAV file. The scaling factor relates the analog voltage to the digital level (Vrms/FS), and is used for scaling the display of values when using the APx500 file input configuration, or the MATLAB script apx_read_wave.

For analog acquisitions, the sample rate is set to be the same as the sample rate determined by the analyzer input configuration analog bandwidth setting, set in Signal Path Setup, as follows:

Bandwidth	Sample Rate
• 20 kHz	• 44.1 kS/s
• 22 kHz	• 48 kS/s
• 45 kHz	• 96 kS/s
• 90 kHz	• 192 kS/s
• 250 kHz	• 624 kS/s
• 500 kHz	• 1.248 MS/s
• 1 MHz	• 2.496 MS/s

Input bandwidths beyond 90 kHz require the BW52 Option; 1 MHz bandwidth requires input set to 1 channel.

The sample rate can be changed in the Advanced Settings dialog.

Duration for linear recordings

Windows WAV files are limited to 4 GB in size. For 24-bit stereo WAV files, this is approximately 1 hour at 192 kHz. Mono files and files with lower sample rates or bit depths can have longer durations. For longer recordings, APx automatically creates a sequential file (or a set of sequential files) providing a very long recording capability without loss of any data.

Note: The sequential file feature for very long recordings is resource-intensive, and requires a high performance level in terms of PC CPU and hard disk speed, beyond the minimum recommended computer required to run APx500.

Filtering

The Filtering set on the Measurement Recorder/Noise Recorder panel is applied to the measurement, but the audio is recorded without filtering. AC/DC coupling settings apply, however.

Bit depth and sample rate for digital acquisitions

For digital acquisitions, the record sample rate and bit depth are set to match that of the incoming digital signal.

Coded formats available for Measurement Recorder digital acquisitions

Coded audio formats are only available as **Save to File** selections for digital acquisitions in Measurement Recorder.

Coded audio (typically multichannel Dolby or DTS surround sound, coded in an IEC61937 bitstream) is saved in the incoming format. Only coded formats currently supported by the APx waveform generator are supported by Save to File. When connected to the HDMI Sink input, recordings cannot be made if HDCP is enabled at the playback device. See Compare Encoded Bitstream to Reference in Chapter 30 to evaluate bitstreams acquired with HDCP enabled.

First, select the expected coded format from the list:

- Dolby Digital (.ac3)
- Dolby Digital + (.ec3)
- DTS 5.1 Compact (.cpt)
- DTS 5.1 (.dts)
- DTS-HD (.cpt)
- Dolby TrueHD (.mlp)

When you click **Start**, Measurement Recorder waits until the selected coded bitstream is detected, and then begins analysis and saves the coded audio to the named file.

This pause accommodates DUTs such as Blu-ray Disc players, which often output PCM audio while idling between tracks.

If you choose

- Auto Detect Encoded

Measurement Recorder waits until the any of the listed coded bitstreams is detected before analyzing and saving.

Alternatively, you can select

- Raw Encoded Bitstream (.bin)

This saves the raw digital output of the receiver chip as a binary file for analysis using third-party software tools.

Duration for coded recordings

Coded file length is limited only by computer storage capacity, to the maximum duration of the Measurement Recorder acquisition of 7 days. Hundreds of gigabytes of storage are necessary for extended times.

Filtering

Coded waveforms are saved without any processing or modification.

More about Reading Rate

A reading is taken at the beginning of the record and at (approximately) the reading rate throughout the record. The final reading is taken at the reading instant that equals or exceeds the sweep end time.

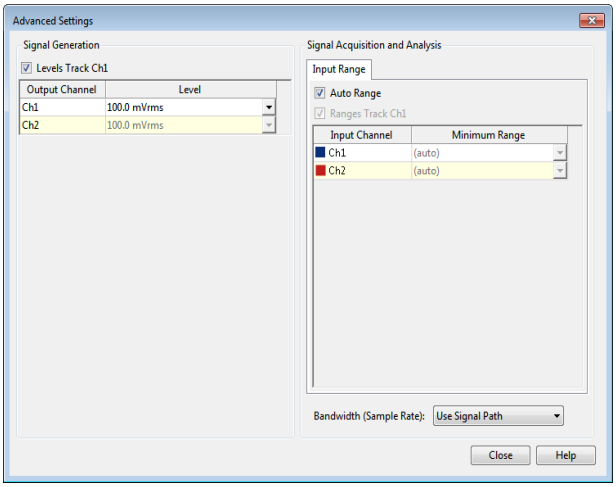
The reading rate also has an effect on analyzer performance at low frequencies. Each reading rate has a low-frequency point below which analyzer performance falls off. THD+N performance degrades rapidly below this point, with level accuracy degrading a little more slowly.

Measurement Recorder and Noise Recorder accuracy limits, by reading rate and frequency:

- at 20/sec > 60 Hz
- at 10/sec > 30 Hz
- at 5/sec > 15 Hz
- at 3/sec > 9 Hz
- at 2/sec > 6 Hz
- at 1/sec > 3 Hz
- at 0.5/sec > 1.5 Hz
- at 0.2/sec > 0.6 Hz
- at 0.1/sec > 0.3 Hz

Advanced Settings for Measurement Recorder

The default settings here are appropriate for most measurements. You may want to make minor adjustments for special situations.



Signal Generation

You can set different output levels or DC offsets for each channel here.

If **Levels Track Ch1** is checked (the default), the generator output level and DC offset values for channel 1 are copied to the other channels, and values for channels greater than 1 cannot be edited. Any changes made to channel 1 are reproduced in the other channels.

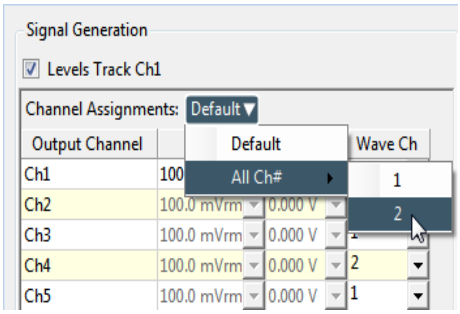
To set individual generator output channel levels or offsets, uncheck the **Levels Track Ch1** checkbox and enter values in the output channel Level or Offset fields.

Set Channel Assignments for special waveforms

For special waveforms like Sine Dual, you can specify the channel assignment.

Set Channel Assignments for generator waveforms

For stereo or multichannel generator waveform files, you can set channel mapping. By default, channel 1 audio is mapped to channel 1 output, channel 2 to channel 2, and so on. If the number of channels in the waveform file is less than the number of output channels, the waveform channels resume numbering at 1 and wrap to the next available output channel. These assignments can be remapped by changing individual settings in the **Wave Ch** column, or by selection one of several presets from the **Set Channel Assignments menu.**



Signal Acquisition and Analysis

Input Range tab

Digital inputs do not require ranging, and the Input Range tab is unavailable when inputs are set to a digital format.

Auto Range

If **Auto Range** is checked (the default), each analog channel determines its input range automatically, based on the level of the input signal. If the input signal level changes beyond a ranging threshold, **Auto Range** will cause the input ranging circuits to move up or down for proper ranging.

Fixed Range

Alternatively, you can set a fixed minimum range for the analog inputs. Uncheck **Auto Range** and enter the expected maximum level in the **Minimum Range** filed. The software will choose the lowest range that will accommodate that level; this becomes the minimum range available, and the analyzer will not reset to a lower range. Fixed range is best for signals that vary rapidly, including tone bursts, perceptual audio speech samples, and program material such as voice or music. See **When autoranging isn't suitable** on page 626.

If a signal greater then the **Minimum Range** is presented to the analog input, the ranging circuits will range up to accommodate the higher level, and then drop again when the signal level falls. The range will not be set below the **Minimum Range**.

Ranges Track Ch1

If **Ranges Track Ch1** is checked (the default), the fixed input range setting for channel 1 is copied to the other channels, and range settings for channels greater than 1 cannot be edited. Any changes made to channel 1 are reproduced in the other channels. If **Ranges Track Ch1** is not checked, the fixed input range setting for each channel can be set independently.

Read more about ranging in Chapter 98.

Bandwidth (Sample Rate)

This control is only available when the analyzer input configuration is set to Analog.

The Bandwidth (Sample Rate) control allows you to override the analog input bandwidth setting in Signal Path Setup. The input bandwidth set here operates only on the Measurement Recorder.

Certain applications that may use the files created here require files of specific sample rate. An example would be the recording of perceptual evaluation signals, which are often required to be at 8 kHz or 16 kHz sample rate. For analog inputs, the default sample rate of a recorded file is determined by the input bandwidth setting, normally set in Signal Path Setup. This control allows selection of other bandwidths/ sample rates. The analog input bandwidth (sample rate) settings, whether made here or in Signal Path Setup, result in channel-count constraints at high sample rates.

The bandwidth choices, followed by their corresponding sample rates, are:

- 2.75 kHz (6 kS/s)
- 3.5 kHz (8 kS/s)
- 5.5 kHz (12 kS/s)
- 7 kHz (16 kS/s)
- 11 kHz (24 kS/s)
- 20 kHz (44.1 kS/s)
- 22 kHz (48 kS/s)
- 40 kHz (88.2 kS/s)
- 45 kHz (96 kS/s)
- 80 kHz (176.4 kS/s)
- 90 kHz (192 kS/s)
- 250 kHz (624 kS/s) (requires the BW52 Option)
- 500 kHz (1.248 MS/s) (requires the BW52 Option)
- 1 MHz (2.496 MS/s) (requires the BW52 Option, set to 1 channel)
- Use Signal Path (the default)

Note that recording files with bandwidths of 250 kHz or greater (with the BW52 Option at sample rates at or above 624 kHz) requires a very fast PC and hard disk, due to the high data rates. Slower systems can suffer recording interruptions and failure.

Metadata Recorder
(Sequence Mode)

Introduction

For the APx515, this measurement requires a software option key. See page 189 for more information about software options.

The Metadata Recorder measurement records the state of selected metadata settings in the incoming digital audio signal. If one of the selected settings changes state during the acquisition, the transition instant and the new state are recorded and displayed.

This measurement is only available when the input is set to digital unbalanced, digital balanced, digital optical, or HDMI.

Operation

If you have not yet set up your test, first go to Signal Path Setup (Chapter 7).

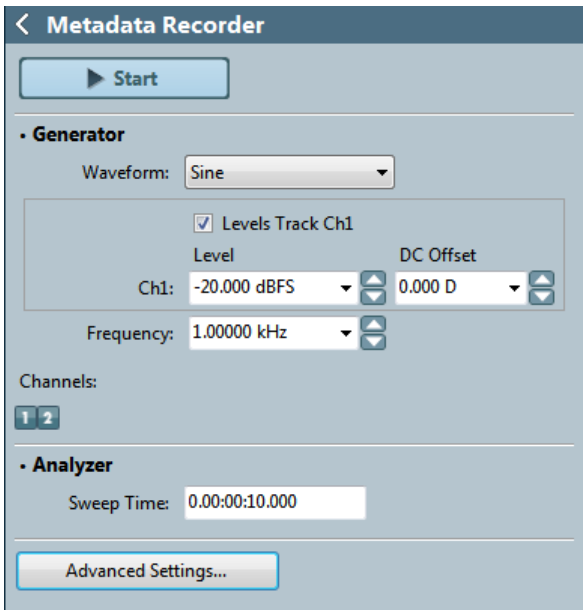
Generator

The typical configuration for this measurement is the closed-loop configuration, using the APx generator as a stimulus. See APx Generator Waveforms and Controls for detailed information about the using the APx Generator and setting Waveform, Level and Frequency.

Alternatively, you can make open-loop measurements, using either the External Source or File Input configuration.

Linear waveforms

Since the Metadata Recorder only reads metadata information, the audio in the transport stream is largely irrelevant. Chose the internal generator or a linear generator waveform. For a closed-loop test, you can set the metadata for the transport stream in **Signal Path Setup > Output Configuration > Digital [Unbalanced] [Balanced] [Optical] [HDMI] > Settings > Status/User Bits or Audio InfoFrame**.



Coded waveforms

For a coded waveform, you must choose a generator waveform of a coded type, such as AC3 or DTS. The coding metadata is embedded in the waveform file, and the transport stream Validity bit is set (indicating not linear audio). You can set other metadata for the transport stream as discussed above.

Analyzer

Acquiring the metadata

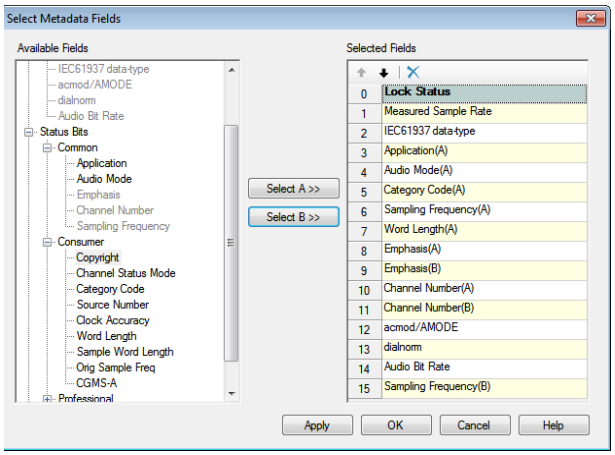
Set **Sweep Time** for the duration you would like to display in the log. The default is 10 s. Click **Start**. The initial state of each metadata field is displayed by a labeled bar. If a field changes state, a transition is indicated, and the label after the transition indicates the new state. The transition is marked with a time stamp referenced to the beginning of the sweep.

The X-axis of the Metadata Recorder is labeled in seconds, but the scale does not represent a linear flow of

time. Instead, (when X-Axis Range is set to **Auto**) the total time of the sweep is shown across the breadth of the graph, and the graph is divided into segments left to right that correspond to the number of transitions. Each segment is marked with the instant of transition.

Select Fields (in the Results display)

You can choose up to sixteen metadata fields for display. Click **Select Fields** to open the selection dialog.



In a New Project, the default fields are

- Lock Status
- Measured Sample Rate
- IEC61937 data-type
- Application(A) [Status Bits]
- Audio Mode(A) [Status Bits]
- Category Code(A) [Status Bits]
- Sampling Frequency(A) [Status Bits]
- Word Length(A) [Status Bits]

Adding fields

Click **Select Fields...** to open the selection dialog.

Open a branch in the tree in the left panel and select a metadata field. In the right panel, select the row in which you would like the data to be displayed. Click the **Select>>** button to assign the field to the display. For Status Bits, choose the **Select A>>** or **Select B>>** button for the subframe you want to display, A or B.

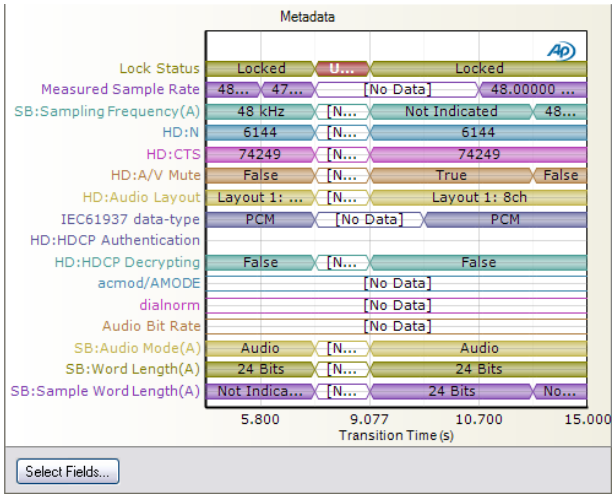
A default color is assigned to each row. Click the browser button at the right of the row to change the display color for that row.

You can remove an entry by clicking the **Delete** button at the top of the right panel. Use the **Up** and **Down** arrows to change the order of the fields.

Abbreviations used in the Metadata Recorder display

"SB" stands for status bits. "(A)" and "(B)" refer to the status bits subframe designation. "HD" stands for HDMI.

Metadata Recorder results



The initial state of each metadata field is displayed by a labeled bar. If a field changes state, a transition is indicated, and the label after the transition indicates the new state. The transition is marked with a time stamp referenced to the beginning of the sweep.

The X-axis of the Metadata Recorder is labeled in seconds, but the scale does not represent a linear flow of time. Instead, (when X-Axis Range is set to Auto) the total time of the sweep is shown across the breadth of the graph, and the graph is divided into segments left to right that correspond to the number of transitions. Each segment is marked with the instant of transition.

Units

Units available for Metadata Recorder are

- s

See Chapter 104 for more information about units of measurement.

More About Audio Metadata

> *Note: for this topic, the designation "IEC60958" represents all the largely compatible audio transport streams, including AES/EBU, AES3, S/PDIF, SMPTE 276M and so on.*

Metadata has been defined as "data about other data." Some digital audio files and digital transport streams carry metadata that describe aspects of the stream and the audio content.

Transport streams and metadata

IEC60958

This industry standard transport stream carries audio samples, embedded clock and metadata. For each subframe (or channel) of audio, there is a synchronizing/identification Preamble, followed by audio data, followed by a Validity Bit, a User Bit, a Channel Status Bit (normally just called a Status Bit) and a Parity Bit. The User Bits and Status Bits are collected over 192 frames, providing metadata in the form of a block of 384 user bits and Subframe A and Subframe B blocks of 192 channel status bits.

Coded audio streams

Low-bit rate coded audio such as Dolby Digital (AC3) and DTS Digital Surround can be embedded in the IEC60958 transport stream as defined in the standard IEC61937. In such cases, the Validity Bit is set, indicating a non-PCM signal. Additionally, the embedded IEC61937 audio carries its own metadata, specific to the coding format.

HDMI: Stream metadata

The HDMI transport stream carries a great deal of data and metadata. Although in audio test and measurement we are not concerned with much of the HDMI signal, the video formats and high-level stream metadata do affect a receiver's ability to correctly accept embedded audio, and these metadata settings and readings are exposed in APx500.

HDMI: Audio InfoFrame

HDMI carries metadata in InfoFrames, and APx500 can set and read Audio InfoFrame data.

Embedded audio stream metadata

Although audio on HDMI is packetized and multiplexed into a high-bit rate video stream, the format of the audio mimics IEC60958 audio and includes user bits, status bits and metadata related to coded audio.

EDID

HDMI also carries EDID, which is not directly related to the audio but affects how a downstream device may react to the transport stream.

HDCP

HDMI streams are often encrypted using HDCP encryption, indicated by a metadata flag.

Audio streams that do not include metadata

Analog audio does not carry any digital metadata. Also, pure PCM sampled audio data is simply a pulse train of samples, and does not carry any metadata. I2S, TDM and other chip-level protocols are of this type; thus, the signals passing in or out of the DSIO ports have no metadata.

Audio files and metadata

Linear files

Generally speaking, linear audio files such as WAV files do not contain metadata. There are housekeeping data that all files contain so that the computer can work with the files, but not metadata about the audio content.

That being said, there are variants on WAV files that contain channel allocation information or specialized broadcast production information. It is even possible to package coded audio with its attendant metadata in a WAV file. But common WAV files, and the WAV linear test signals we provide, do not carry metadata.

When such a file is routed out through a transport stream transmitter (e.g., IEC60958 or HDMI) metadata is added by the transmitter.

Coded audio files

Coded audio files such as AC3, DTS, MLP and DTSHD contain the proprietary metadata for the coding format. When such a file is routed out through a transport stream transmitter (e.g. IEC60958 or HDMI), the coding metadata is maintained as embedded IEC61937 metadata, and transport stream metadata is added by the transmitter.

IEC60958 Status Bits and User Bits

Overview

The family of digital interface formats variously known as IEC60958, AES3, AES/EBU, S/PDIF, and SMPTE 276M share a common history of development by standards committees made up of audio experts from around the world. The formats are, for the most part, compatible. The primary differences are in the physical interface and in the use and interpretation of channel status bits.

In practice, if the physical interfaces are compatible and the sample rate and bit depth are supported, a signal transmitted conforming to any of these standards will almost always be converted to audio in a receiving device.

Status Bits

The channel status information is carried in the Status Bits. Apart from the first two bits, the meaning of the bits is defined differently for the consumer and professional formats. See the tables at the end of this chapter.

User Bits

The user bit can be used to carry user-specific information.

Status bits and user bits over HDMI

Although the physical layer of the high-speed, packeted transport signal of HDMI is completely incompatible with any of the above interfaces, nonetheless the supported audio protocols at the input or output of HDMI transmitters or receivers are compliant with IEC60958 and the coded audio layer IEC61937. Channel status bits and user bits are supported within the audio streams carried by HDMI. The default for HDMI is the consumer application.

HDMI and HDMI Audio InfoFrame

Overview

The HDMI transport stream carries a great deal of information, including video data, audio data and extensive metadata, including several types of InfoFrames that describe various aspects of the data in the transport stream.

Of interest in audio test and supported in APx500 is the Audio InfoFrame, which contains information about the audio format, with additional channel allocation and downmix information for linear PCM audio.

"Refer to Stream Header"

Note that the CEA-861-D standard that defines Audio InfoFrame supports HDMI, but also other transport streams. Some fields of the Audio InfoFrame are considered redundant in the HDMI application and are required to be set to 0, "Refer to Stream Header". "Stream Header" indicates the IEC60958 channel status bits and, for coded streams, the IEC61937 burst info data.

Audio InfoFrame fields

Coding Type (CT)

In HDMI use, audio Coding Type is considered redundant and is required to be set to 0 ("Refer to Stream Header"). For testing purposes, APx500 can ignore this requirement and transmit audio Coding Type indicators at any supported setting. Coding Type and Channel Count share Data Byte 1.

Channel Count (CC)

Audio channel count can be set to 0 ("Refer to Stream Header"), or to 2–8 channels for multichannel linear PCM use. For testing purposes, APx500 can transmit channel count indicators at any supported setting. Coding Type and Channel Count share Data Byte 1.

Sampling Frequency (SF)

In HDMI use, audio sampling frequency is considered redundant and is required to be set to 0 ("Refer to Stream Header"). For testing purposes, APx500 can ignore this requirement and transmit sampling frequency indicators at any supported setting. Sampling Frequency and Sample Size share Data Byte 2.

Sample Size (SS)

In HDMI use, audio sample size is considered redundant and is required to be set to 0 ("Refer to Stream Header"). For testing purposes, APx500 can ignore this requirement and transmit sample size indicators at any supported setting. Sampling Frequency and Sample Size share Data Byte 2.

Speaker Allocation (CA)

The speaker allocation field is provided for Layout 1 multichannel LPCM streams, which do not carry speaker data. Metadata is embedded in the HDMI signal indicating the desired mapping of channels 1–8 to one of 32 loudspeaker arrangements. For testing purposes, APx500 can transmit speaker allocation data at any supported setting. Speaker Allocation is carried in Data Byte 4.

Level Shift Value (LSV)

The level shift field is provided for multichannel linear PCM streams, which do not carry downmix data. For testing purposes, APx500 can transmit level shift data at any supported value. Level Shift Value and Downmix share Data Byte 5.

Downmix (DM)

The downmix inhibit field is provided for multichannel linear PCM streams, which do not carry downmix data. For testing purposes, APx500 can transmit downmix permitted (0) or prohibited (1). Level Shift Value and Downmix share Data Byte 5.

In APx500, outgoing Audio InfoFrame metadata can be set in the HDMI Output Settings dialog; see page 119. Incoming Audio InfoFrame metadata can be read in the Metadata Monitor: HDMI; see page 35.

Metadata tables

Status Bits consumer interpretation

See the latest edition of the standard IEC60958 Part 3 for more information.

BITS	*LABEL*	*INTERPRETATION*		
0	application: consumer or professional	0: consumer; 1: professional		
1	non-audio	0: audio data is linear PCM samples 1: other than linear PCM samples		
2	copyright	0: asserted; 1: not asserted		
3-5	emphasis	000: Emphasis not indicated 100: CD-type emphasis		
6-7	channel status mode	00: mode zero; other values reserved		
8-15	category code	(bit 8 is LSB)		
16-19	source number	(bit 16 is LSB)		
20-23	channel number	(bit 20 is LSB)		
24-27	sampling frequency	0000: 44.1 kHz 0100: 48 kHz 1100: 32 kHz		
28-29	clock accuracy	10: Level I, ±50 ppm 00: Level II, ±1000 ppm 01: Level III, variable pitch shifted		
30-31	reserved			
32	word length (field size)	0: Maximum length 20 bits 1: Maximum length 24 bits		
33-35	word length		if bit 32 = 1	if bit 32 = 0
		000:	not indicated	not indicated
		101:	24 bits	20 bits
		001:	23 bits	19 bits
		010:	22 bits	18 bits
		011:	21 bits	17 bits
		100:	20 bits	16 bits
36-39	reserved			
40-191	reserved			

Status Bits professional interpretation

See the latest edition of the standard IEC60958 Part 4 for more information.

BITS	*LABEL*	*INTERPRETATION*
0	application: consumer or professional	0: consumer; 1: professional format
1	non-audio	0: audio data is linear PCM samples 1: other than linear PCM samples
2-4	emphasis	000: Emphasis not indicated 100: No emphasis 110: CD-type emphasis 111: J-17 emphasis
5	lock	0: not indicated 1: unlocked
6-7	sampling frequency	00: not indicated (or see byte 4) 10: 48 kHz 01: 44.1 kHz 11: 32 kHz

8-11	Channel mode (SCDSR represents “single channel double sample rate”)	0000: not indicated (default to 2 ch) 0001: 2 channel 0010: 1 channel (monophonic) 0011: primary / secondary 0100: stereo 0101: reserved for user applications 0110: reserved for user applications 0111: SCDSR (see byte 3 for ID) 1000: SCDSR (stereo left) 1001: SCDSR (stereo right) 1111: Multichannel (see byte 3 for ID)		
12-15	user bit management	0000: no indication 0001: 192-bit block as in channel status 0010: As defined in AES18 0011: user-defined 0100: as in IEC60958-3 (consumer)		
16-18	use of aux sample word	0000: not defined, audio max 20 bits 0001: used for main audio, max 24 bits 0010: used for coordination signal, audio max 20 bits 0011: user-defined		
19-21	source word length	000: 001: 010: 011: 100: 101:	if max = 24 bits not indicated 23 bits 22 bits 21 bits 20 bits 24 bits	if max = 20 bits not indicated 19 bits 18 bits 17 bits 16 bits 20 bits
22-23	alignment level	00: not indicated 01: –20 dB FS 10: –18.06 dB FS		
24-31	channel identification	if bit 31 = 0 then channel number is 1 plus the numeric value of bits 24-30. if bit 31 = 1 then bits 4–6 define a multichannel mode and bits 0–3 give the channel number within that mode.		
32-33	digital audio reference signal (DARS)	00: not a DARS 10: DARS grade 2 (+ / –10 ppm) 01: DARS grade 1 (+ / –1 ppm)		
35-38	sampling frequency	0000: not indicated 1000: 24 kHz 0100: 96 kHz 1001: 22.05 kHz 0101: 88.2 kHz 1101: 176.4 kHz 1111: User defined		
39	sampling frequency scaling	0: no scaling 1: apply factor of 1 / 1.001 to value		
48-79	alphanumerical channel origin	four-character label using 7-bit ASCII with no parity. Bits 55, 63, 71, 79 = 0.		
80-111	alphanumerical channel destination	four-character label using 7-bit ASCII with no parity. Bits 87, 95, 103, 111 = 0.		
112-143	local sample address code	32-bit binary number representing the sample count of the first sample of the channel status block.		
144-175	time of day code	32-bit binary number representing time of source encoding in samples since midnight.		
176-183	reliability flags	0: data in byte range is reliable 1: data in byte range is unreliable		
184-191	CRCC	00000000: not implemented nnnnnnnn: error check code for bits 0–183		

Modulated Noise
(Sequence Mode)

This measurement requires a software option key. See page 189 for more information about software options.

The Modulated Noise measurement is specifically designed to detect air leaks in a sealed loudspeaker/enclosure system. Modeling an air leak as a burst of amplitude-modulated noise, this measurement uses a low-frequency stimulus to excite the system, and applies filtering and AM detection to analyze the speaker output and provide a modulated noise ratio result.

Read about Modulated Noise and air leak detection at the end of this chapter.

Operation

If you have not yet set up your test, first go to Signal Path Setup to select and configure inputs and outputs, references and other settings. See Chapter 7 for information about Signal Path Setup.

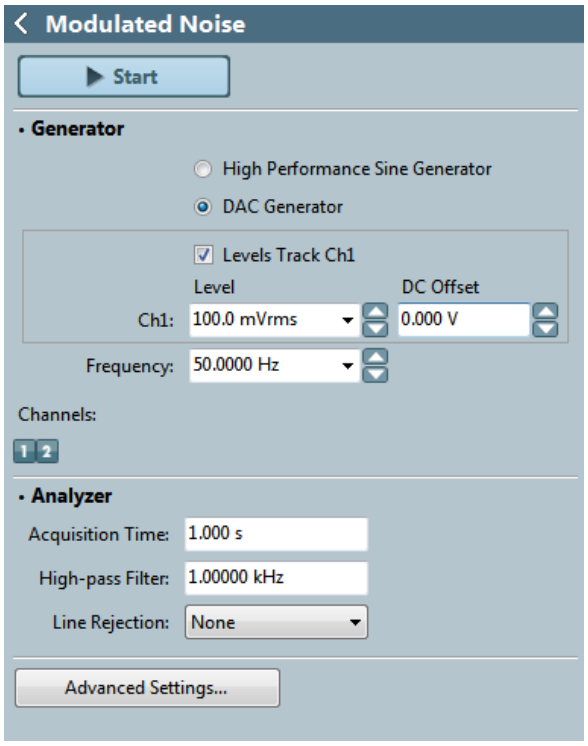
Generator

This measurement must be performed in the closed-loop configuration, using the APx generator as a stimulus.

Selecting the generator waveform

By default, the Modulated Noise measurement uses a 50 Hz sine wave at the frequency and level set in the Generator panel as the test signal. You can set other frequencies, but low frequencies are the most effective in detecting air leaks. It is important to drive the loudspeaker with enough level to get the air moving in the enclosure.

See Chapter 6 for detailed information about the using the APx Generator and making Waveform, Level and Frequency settings.



High Performance Sine Generator / DAC Generator

For an APx555, you have the option of using the High Performance Sine Generator (very low distortion sine waves and wide bandwidth) or the DAC generator, which provides special sine waveforms. See High Performance Sine Generator starting on page 51 for a comparison of features.

Analyzer

The acquired signal is first comb-filtered, with a comb set at the generator frequency and its harmonics.

Acquisition Time

Set the acquisition time here.

High-pass filter

Use this filter to reduce environmental noise. The default corner frequency is 1 kHz, with a range of 10 Hz to 3 kHz. Adjust this filter to find the highest modulated noise ratio. When set, the local high-pass filter acts on the acquired audio, after comb filtering. Read about Modulated Noise and air leak detection at the end of this chapter.

Low pass, high pass and weighting Signal Path Filters settings (see page 617) made in Signal Path Setup > Input/Output will also affect this measurement. Unlike other measurements, setting the local high-pass filter in Modulated Noise does not override the high-pass filter in Signal Path Setup.

If a filter in Signal Path Setup is set to reduce or eliminate a frequency range that includes the Modulated Noise stimulus (set in Generator > Frequency), the Modulated Noise measurement may fail to make an acquisition.

Line Rejection

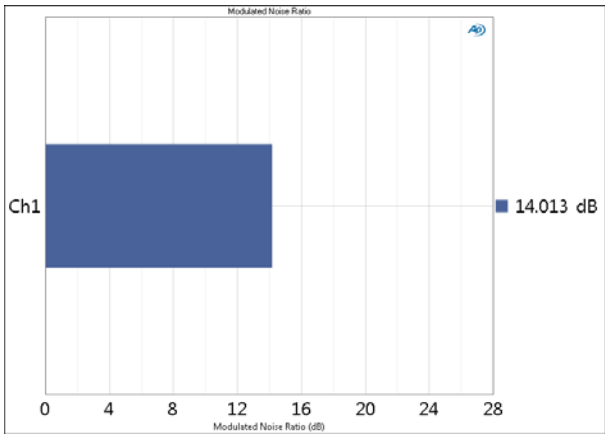
This optional filter will notch out the mains power line frequency and its harmonics, if necessary. Set the filter to the nominal mains frequency in your region, 50 Hz or 60 Hz.

Advanced Settings

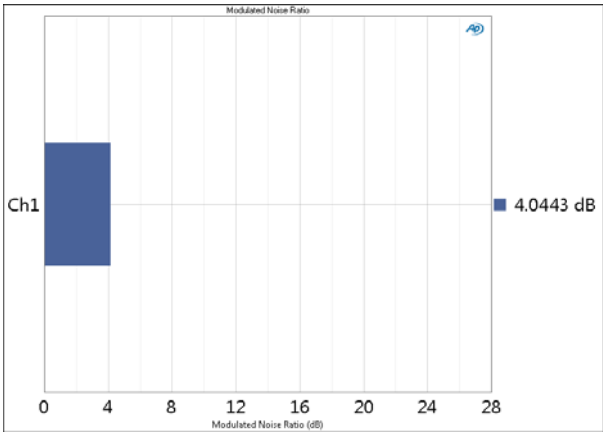
Advanced Settings makes available individual channel output level and offset controls, and input ranging and settling controls. See Advanced Settings for single value measurements on page 351.

See Chapter 104 for more information about units of measurement.

Modulated Noise Ratio



Loudspeaker enclosure with an air leak.



Same loudspeaker enclosure with air leak plugged.

Units

Units available for Modulated Noise Ratio are

- x/y
- dB

Modulated Noise and air leak detection

Loudspeaker designs may specify a completely sealed enclosure, or an enclosure vented with acoustic ports. In either case, air leaks can degrade the performance, and can add spurious signals to the acoustic output of the system. A loudspeaker with air leaks presents a higher Modulated Noise Ratio result (in dB) compared to the same loudspeaker without air leaks measured under identical conditions.

The Modulated Noise Ratio result values are relative, and will depend on a number of factors such as the environmental conditions, the physical test setup, the choice of filters and associated parameters selected, the amount of air leakage, the relative position of the microphone with respect to the position of the air leak—to name a few.

Modulated Noise

A Modulated Noise ratio measurement is provided in software option SW-SPK-RD and SW-SPK-PT as an air leak detection tool. A leak will create noise that is modulated by movements of the speaker cone. A sine wave is used to drive the speaker. Under these circumstances, the signature of a leak will be short bursts of noise occurring in rhythm with excursions of the driver at the stimulus fundamental frequency and its harmonics.

The Modulated Noise ratio measurement first removes the sine wave signal and its harmonics but leaves the noise portion of the signal intact. The user may suppress the impact of environmental noise by adjusting the corner frequency of a high pass filter. Additionally, the user may opt to remove any line frequency and its harmonics from the noise. The remaining noise signal is then analyzed to provide the average power of the residual stimulus at the fundamental frequency of the original sine wave and its harmonics. This is compared to the average power of the noise at other frequencies. The result of this comparison is finally converted to decibels to provide the reported value.

Suppressing environmental noise

Successful detection of air leaks requires analysis uncontaminated by environmental noise, to the degree possible. Ideally, the loudspeaker and measurement microphone are placed in a quiet acoustic chamber. However, in production and in many R&D bench situations, environmental noise such as HVAC rumble, fan noise, and mains hum are unavoidable.

If the loudspeaker is in a noisy environment, the corner frequency of the measurement's high pass filter may be adjusted with the goal of obtaining the highest ratio. An optional filter in the Modulated Noise measurement may also be selected to suppress mains hum.

Using multiple microphones

An air leak may be located on the side or rear surface of an enclosure. If the measurement microphone is only directed to the front surface, the air leak may escape detection. Using an analyzer with two or more inputs enables the simultaneous use of multiple measurement microphones for more complete evaluation and detection.

Multitone Analyzer
(Sequence Mode)

Introduction

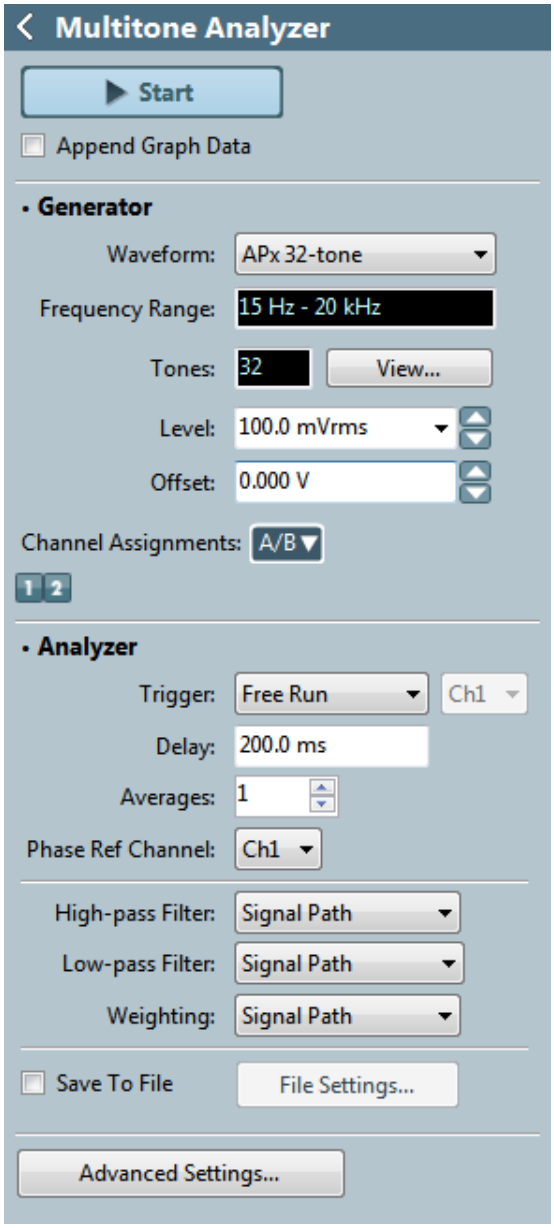
For the APx515, this measurement requires a software option key. See page 189 for more information about software options.

The Multitone Analyzer uses a special stimulus signal called a *multitone*. Multitone analysis is very fast, and provides a broad range of results.

The Multitone Analyzer results available in APx500 are:

- Level
- Gain
- Relative Level
- Deviation
- Phase
- Phase (at Tone)
- TD+N Ratio
- TD+N Level
- TD+N Spectrum
- Crosstalk
- Crosstalk (at Tone Pair)
- Total Level
- Tone Level
- Maximum Level (at Tone)
- Noise Density Spectrum
- Noise RMS Level
- Signal to Noise Ratio
- DC Level
- Frequency Shift
- FFT Spectrum
- Acquired Waveform

For more information about multitone, go to page 409.



Operation

If you have not yet configured the instrument I/O for your DUT and test, first go to Signal Path Setup (Chap-

ter 7). When Generator and Analyzer (below) are set for your test, click **Start**.

Append Graph Data

Measured data are grouped in a **Data Set**, which contains all acquisition results. Normally, the current graph data are deleted each time you start a new measurement. If the **Append Graph Data** box is checked, appended data are grouped as additional data sets, as are any imported data. The Data Sets Panel displays the Data Set(s) in the measurement, and provides a visibility checkbox, a Time acquisition field, and a Notes field for each data set.

Read more about Data Sets beginning on page 649.

Generator

Waveform

The APx generator can create a multitone directly from a multitone signal definition, or it can play an audio (WAV) file that contains a multitone.

Using a pre-defined multitone signal definition

A number of pre-defined multitone signal definitions are available for selection from the **Waveform** drop-down menu. The illustration at left shows the pre-defined definition “APx 32-tone” as the selection. Pre-defined multitone signal definitions cannot be edited.

When a pre-defined multitone signal definition is currently selected, you can read the multitone signal’s **Min Frequency, Max Frequency** and the number of **Tones** in the display fields. Click the **View** button to open the **View Multitone Signal Definition** dialog, where you can see the definition in detail, export the definition as a data file, or use it to create and save an audio (WAV) multitone file.

Creating and using user-defined multitone signal definitions

You can create custom multitone signal definitions. From the **Waveform** drop-down menu, choose **Create New...** to open the **Create Multitone Signal Definition** dialog (page 411). Choose a signal definition as a starting point, then click **OK** to continue to the **Edit Multitone Signal Definition** dialog (page 412), where you can edit the definition in detail, export the definition as a CSV or XLS data file, or use it to create and save an audio (WAV) multitone file. Click **OK** to save your definition in the project and add it to the **Waveform** menu.

Loading a multitone signal definition from a file on disk

You can import a multitone signal definition from a file on disk. From the **Waveform** drop-down menu, choose **Browse for file...** and select a CSV or XLS multitone signal definition file using the file browser.

Using a linear audio file on disk as a generator waveform

You can use the APx generator waveform feature with the multitone analyzer, much as you can with other measurements. From the **Waveform** drop-down menu, choose **Browse for file...** and select a linear audio (WAV) file using the file browser. For proper operation and analysis, the audio contained in the file must be a valid, well-formed multitone. For a linear (WAV) file, APx500 extracts a multitone signal definition from the audio in the file. This definition is required to provide the frequency, level and phase information necessary for analysis. The extracted definition can be examined by clicking the **View** button to open the **View Multitone Signal Definition** dialog (page 411). When the multitone is played by the APx generator, the original audio file is used for signal generation, not the extracted definition. See Generator Waveform files on page 186.

Using a coded audio file on disk as a generator waveform

Coded audio files (ac3, dts, etc.) can be used as generator waveform files. The file must contain coded audio derived from a valid and well-formed multitone signal. Additionally, the multitone signal definition data used to create the original multitone must be available to APx500. When a coded multitone file is selected, a new field called **Signal Definition** appears in the settings panel. Use this selector to choose a multitone signal definition that corresponds to the multitone in the coded file. This definition is required to provide the frequency, level and phase information necessary for analysis. See Generator Waveform files on page 186.

Description

This field is only displayed when a generator waveform is currently loaded in the generator. It displays information about the waveform file.

Signal Definition

This field is displayed in **External Source** configuration, or when a coded generator waveform is currently loaded in the generator. It provides a selector to choose a multitone signal definition that corresponds to the multitone in use (either the externally sourced multitone, or the multitone in the coded file). See **Edit Multitone Signal Definition** on page 412. This definition is required to provide the frequency information necessary for analysis.

Min Frequency / Max Frequency / Tones

These fields display information about the multitone definition currently loaded in the generator. See **View Multitone Signal Definition** (page 411) and **Edit Multitone Signal Definition** (page 412).

Level

Set the generator output level in this field.

The **Level** control in an APx measurement sets the level of the generator output to the rms value entered in the **Level** field. With a sine wave, the result is what you might expect: the generator outputs a single tone at 1 Vrms.

However, a multitone waveform is the sum of two or more sine waves of different frequency and phase, and possibly different level. These tones are summed, resulting in a complex waveform whose crest factor (page 416) will be higher than that of a sine wave. The **Level** control sets the amplitude of the complex waveform, but the individual component stimulus tone levels in a multitone signal are typically well below the rms level of the complex waveform.

Channel

The generator will output a multitone to the DUT on the selected generator channels (page 186).

Start

The **Start** button runs the measurement. For closed loop operation, **Start** generates the multitone signal and prepares the analyzer for acquisition. For **External Source** (open loop) measurements, **Start** prepares the analyzer for acquisition, awaiting **Trigger**.

Analyzer

Trigger

By default, the Multitone Analyzer **Trigger** is set to **Free Run**, which is the untriggered condition. In many cases, especially closed-loop configurations, this setting will provide fast, reliable and highly accurate multitone measurements. However, some use cases or testing environments will require other trigger configurations.

- **Signal**
  This trigger mode is the most discriminating. The incoming audio on a specified channel is compared to the multitone signal definition in memory, with selectable degrees of discrimination (see **Channel, Delay, Match %** and **SNR**, below). The analysis is only triggered when a match between the incoming audio and the multitone signal definition is accomplished, after any trigger **Delay**. Tones that are very close together (separated by only a few bins) are more difficult to trigger on.
- **Generator**
  This trigger mode begins analysis at the moment the internal generator outputs a signal, after any trigger **Delay**.
- **Level**
  This trigger mode begins analysis when the incoming audio on a specified channel exceeds the specified level, after any trigger **Delay**.
- **Free Run** (the default)
  This trigger has no conditions. Analysis begins when the Start button is clicked, after any trigger **Delay**.

Channel

For **Signal** and **Level** triggering, select the triggering channel here.

Delay

The trigger **Delay** inserts time between a successful trigger event and the actual start time of the acquisition. The range is 0.000 s to 1.000 s; default is 200.0 ms.

Level

For **Level** triggering, set a threshold **Level** here.

Match %

For **Signal** triggering, set the desired degree of matching here. The range is from 25 % (rough match) to 100 % (exact match).

SNR

For **Signal** triggering, set the required signal-to-noise ratio here. The range is from 20 dB to 100 dB.

Averages

For non-periodic signals such as noise, averaging multiple acquisitions can provide a more useful view. This control allows you to specify how many acquisitions to average. The default is 1; maximum is 8.

Phase Ref Channel

Select the reference channel for the phase measurement here. The phase of the other channels is calculated relative to the reference channel.

Filters

Local high-pass, low-pass and weighting filters are available for this measurement. The low-pass and high-pass filters, if selected, are applied to the entire measured signal and affect all the results. The weighting filter, if selected, affects only the distortion and distortion plus noise results. See Measurement Filters on page 619 for detailed information about the filters available locally for some measurements.

Low pass, high pass and weighting Signal Path Filters settings (see page 617) made in Signal Path Setup > Input/Output will also affect this measurement. When set, the local low-pass or high-pass filter setting overrides the corresponding Signal Path filter setting while the measurement is active. A local weighting filter setting does not override any Signal Path filter settings;

the local weighting filter is applied in addition to any active Signal Path filters.

Save to File

You can record the multitone audio signal to a file as the signal is being acquired. Click **Save to File** before you run the multitone measurement. For multitone, only the frequency-shifted FFT record is saved, not the entire acquisition. See Recording Multitone Audio to a File on page 416 for detailed information.

File Settings

When saving a multitone acquisition to a file, click **File Settings** to specify the file name and location. For subsequent acquisitions, data are overwritten to the same file name, with no warning. To keep multiple acquisitions, rename the file for each acquisition.

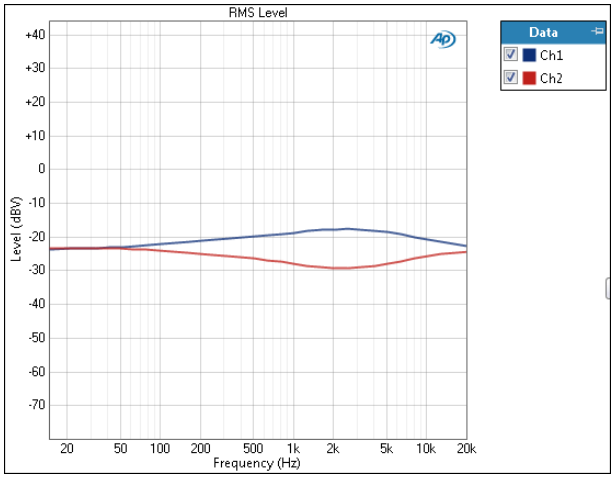
Advanced Settings

If you'd like to adjust analyzer ranging or settling parameters, click **Advanced Settings**, where you will also find multitone-specific settings including A-B signal mapping, trigger timeout, maximum frequency shift and skirt width. See page 415 for information about Advanced Settings for Multitone.

See Chapter 104 for more information about units of measurement.

A multitone result provides a limited number of measured data points (one for each tone in the stimulus, as you can see in the graph data grid (page 649). Values between these data points implied by the plotted curve are interpolated for display.

Multitone Analyzer: Level



The **Multitone Analyzer: Level** result uses the multitone method to provide a graphical display of the level vs. frequency (frequency response) for each channel of the DUT. The rms value of each tone is measured, and a response curve is plotted through these values.

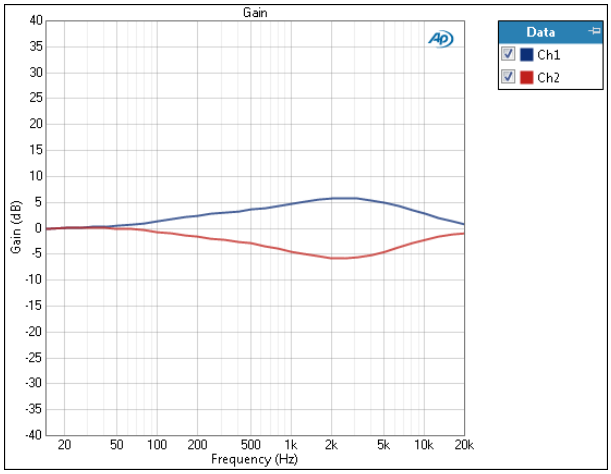
The **Skirt Width** control (page 416) in **Advanced Settings** can affect the values reported in this result.

Units

Units available for Level results are

X-axis	*Y-axis (analog)*	*Y-axis (digital)*
• Hz	• Vrms	• dBFS
• dHz	• dBV	• FS
• F/R	• dBu	• %FS
• %Hz	• dBrA	• dBrA
	• dBrB	• dBrB
	• dBSPL1	• dBSPL1
	• dBSPL2	• dBSPL2
	• dBm	
	• W (watts)	

Multitone Analyzer: Gain



Gain measurements are not available when External Source is selected as the output configuration, when File is selected as the input configuration, or when a square wave or generator waveform file is selected as the test signal.

The **Multitone Analyzer: Gain** result uses the multitone method to provide a graphical display of the gain vs. frequency (gain response) for each channel of the DUT. The rms value of each tone is measured, and a response curve is plotted through these values.

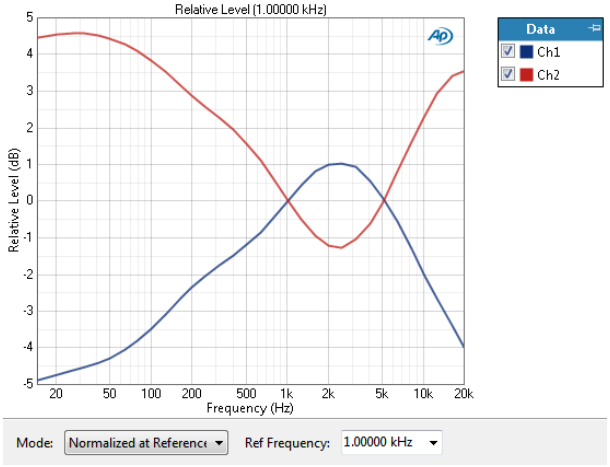
The **Skirt Width** control (page 416) in **Advanced Settings** can affect the values reported in this result.

Units

Units available for Multitone: Gain are

X-axis	*Y-axis (same-domain)*	*Y-axis (cross-domain)*
• Hz	• x/y	• FS/Vrms
• dHz	• %	• dB(FS/Vrms)
• F/R	• ppm	or
• %Hz	• dB	• Vrms/FS
		• dB(Vrms/FS)

Multitone Analyzer: Relative Level



The **Multitone Analyzer: Relative Level** result uses the multitone method to provide a graphical display of the level vs. frequency (frequency response) for each channel of the DUT, relative to the measured level at a reference frequency. This enables you to specify a frequency that will be set as 0 dB, and to view the response in relation to that frequency. The rms value of each tone is measured, and a response curve is plotted through these values.

The **Skirt Width** control (page 416) in **Advanced Settings** can affect the values reported in this result.

Additional Controls for Relative Level

Ref Frequency

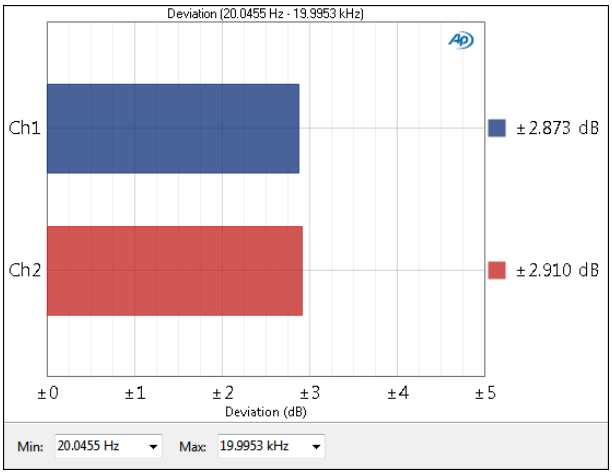
Enter a reference frequency here. The measured level at the frequency you choose is set as 0 dB on the graph, and the DUT response is plotted in relation to this reference. You can change the **Ref Frequency** at any time and the graph will immediately be redrawn to reflect the new setting. A midband frequency such as 1 kHz is often set as the reference frequency, and is the default here.

Units

Units available for **Multitone: Relative Level** are

X-axis	*Y-axis*
• Hz	• x/y
• dHz	• %
• F/R	• ppm
• %Hz	• dB

Multitone Analyzer: Deviation



The **Multitone Analyzer: Deviation** result uses the multitone method to provide the frequency deviation (the total range of frequency variation) of each DUT channel, displayed as a meter (single value) result. You can specify a minimum and maximum frequency to define the range to be considered in the deviation measurement.

The **Skirt Width** control (page 416) in **Advanced Settings** can affect the values reported in this result.

Additional Controls for Deviation

Min Frequency / Max Frequency

Deviation measurements are often limited to a certain frequency range; the default here is 20 Hz to 20 kHz. You can change set the range across which deviation is calculated by entering new values into the **Min Frequency** and **Max Frequency** fields. You can change this range at any time and the meter bars will immediately be redrawn to reflect the new settings.

A note about data sets and the Deviation result

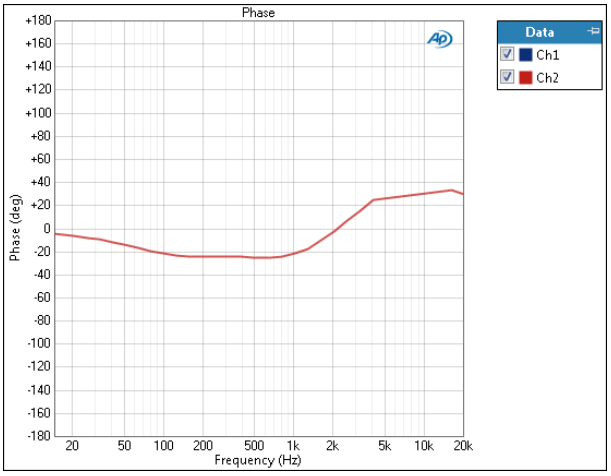
You can append or import graph data to Multitone measurements, but the Deviation result only shows the deviation in the latest acquired data. Other data sets cannot be selected for Deviation.

Units

Units available for **Multitone: Deviation** results are

- dB

Multitone Analyzer: Phase



The **Multitone Analyzer: Phase** result uses the multitone method to provide a graphical display of the interchannel phase vs. frequency (interchannel phase response) for each channel of the DUT. The phase value of each tone is measured and compared to the phase value of the reference channel at the same tone, and a response curve is plotted through these values.

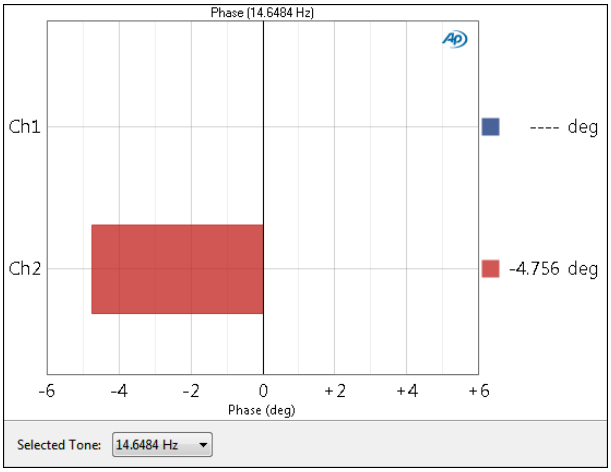
Read more about phase on page 332.

Units

Units available for **Multitone: Phase** are

X-axis	*Y-axis*
• Hz	• deg
• dHz	• rad
• F/R	
• %Hz	

Multitone Analyzer: Phase (at Tone)



The **Multitone Analyzer: Phase at (Tone)** result uses the multitone method to provide a graphical display of the interchannel phase at any tone in the multitone, for each channel of the DUT. The phase value of the selected tone is measured and compared to the phase value of that tone in the reference channel.

Read more about phase on page 332.

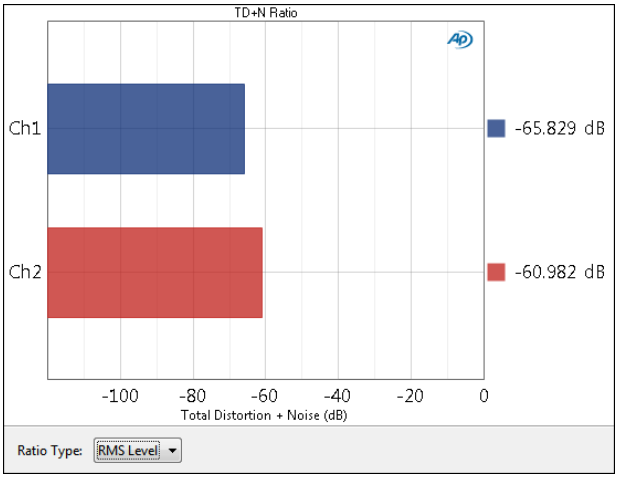
Units

Units available for **Multitone: Phase (at Tone)** are

- deg
- rad

See Chapter 104 for more information about units of measurement.

Multitone Analyzer: TD+N Ratio



Operation

See the Multitone Analyzer main topic on page 397 for controls and settings common to all multitone measurements.

The **Multitone Analyzer: TD+N Ratio** result uses the multitone method to provide total distortion + noise ratio results. How this is measured depends upon the setting of **Ratio Type.**

The **Skirt Width** control (page 416) in **Advanced Settings** can affect the values reported in this result.

TD+N Ratio (Ratio Type set to RMS Level)

This result is the ratio of two intermediate results.

First, the rms distortion + noise level (the sum of the rms distortion + noise across the multitone bandwidth) is determined for each DUT channel. Tone bins and bins in the current **Skirt width** are excluded from this result. All other bins, odd and even are summed. This provides a total distortion + noise (TD+N) level result. In a second operation, all the bins are summed to provide a total rms level result. The total rms level is divided by the TD+N level, to produce a TD+N ratio result.

TD+N Ratio (Ratio Type set to Max Level)

This result is the ratio of two intermediate results.

First, the rms distortion + noise level is determined in the same way as described above, providing a TD+N level result. In a second operation, the selected tone bin and the bins in that tone's current **Skirt width** are summed to provide an rms level result. This result is then scaled (the attenuation applied in the multitone signal creation process is factored out), providing a maximum level value. This maximum level value is divided by the TD+N level, producing an alternative TD+N ratio result.

Ratio Type:

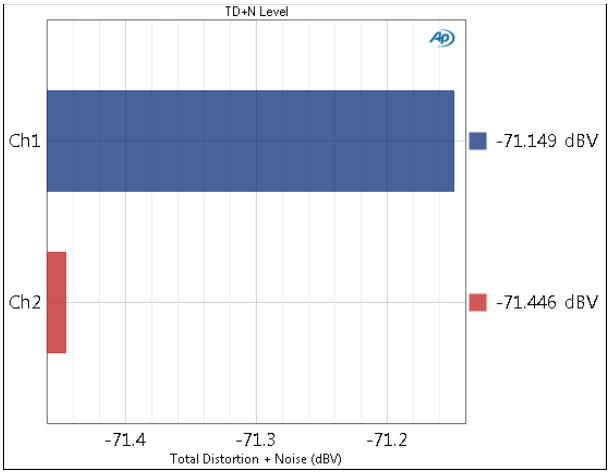
Choose the **Ratio Type** (described above) in this field.

Units

Units available for **Multitone: TD+N Ratio** results are:

- x/y
- %
- ppm
- dB

Multitone Analyzer: TD+N Level



The **Multitone Analyzer: TD+N Level** result uses the multitone method to provide the rms total distortion + noise level (the sum of the rms distortion + noise across the multitone bandwidth) for each DUT channel, displayed as a meter (single value) result. To produce a total distortion + noise result, tone bins and bins in the current **Skirt width** are excluded from measurement. All other bins, odd and even, are summed.

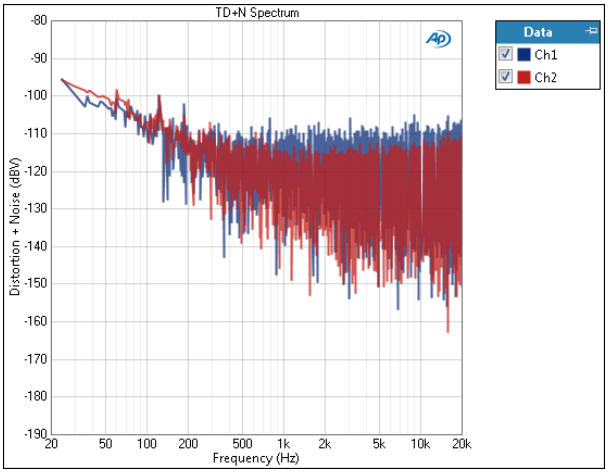
The **Skirt Width** control (page 416) in **Advanced Settings** can affect the values reported in this result.

Units

Units available for **Multitone: TD+N Level** are

Analog	*Digital*
• Vrms	• dBFS
• dBV	• FS
• dBu	• %FS
• dBrA	• dBrA
• dBrB	• dBrB
• dBSPL1	• dBSPL1
• dBSPL2	• dBSPL2
• dBm	
• W (watts)	

Multitone Analyzer: TD+N Spectrum



The Multitone Analyzer: TD+N Spectrum result provides an FFT spectrum display of the acquired signal. All bins, odd and even, that are not tone bins nor within the current **Skirt width** are plotted in this display. This removes the stimulus tones and reports a total distortion + noise result.

> *This result provides approximately the full number of measured data points (one for each non-tone bin in the FFT analysis). The graph data grid (page 649) initially shows the number of data points on the graph display, but can be set to show All Points.*

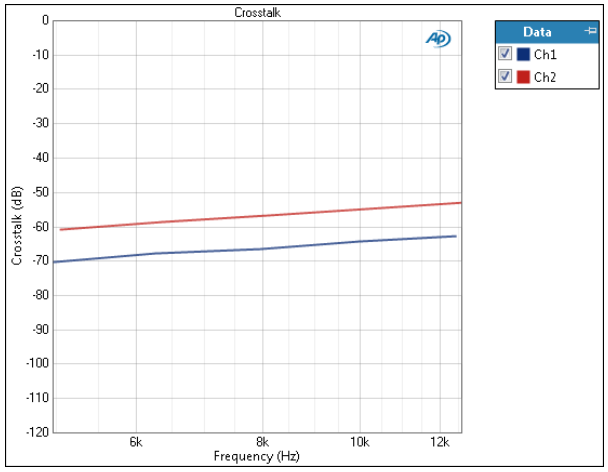
The **Skirt Width** control (page 416) in **Advanced Settings** can affect the values reported in this result.

Units

Units available for Multitone: TD+N Spectrum are

X-axis	*Y-axis (analog)*	*Y-axis (digital)*
• Hz	• Vrms	• dBFS
• dHz	• dBV	• FS
• F/R	• dBu	• %FS
• %Hz	• dBrA	• dBrA
	• dBrB	• dBrB
	• dBSPL1	• dBSPL1
	• dBSPL2	• dBSPL2
	• dBm	
	• W (watts)	

Multitone Analyzer: Crosstalk



The **Multitone Analyzer: Crosstalk** result uses the multitone method to provide a graphical display of the crosstalk vs. frequency (crosstalk response) for each channel of the DUT. The crosstalk of each tone pair is measured, and a response curve is plotted through these values.

The **Crosstalk** results require that specific crosstalk tone exist in the multitone signal. If there are no such tones in the stimulus signal, Crosstalk results will not be available. See **Multitone Crosstalk Measurements** on page 411 for more information.

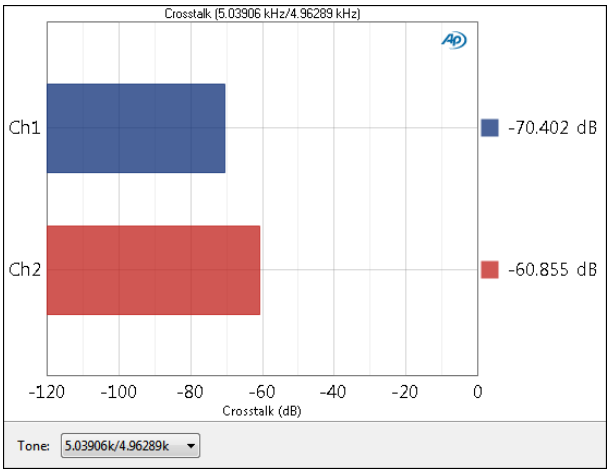
The **Skirt Width** control (page 416) in **Advanced Settings** can affect the values reported in this result.

Units

Units available for **Multitone: Crosstalk** are

X-axis	*Y-axis*
• Hz	• x/y
• dHz	• %
• F/R	• ppm
• %Hz	• dB

Multitone Analyzer: Crosstalk (at Tone Pair)



The **Multitone Analyzer: Crosstalk (at Tone Pair)** result uses the multitone method to provide the crosstalk level for a selected tone pair for each DUT channel, displayed as a meter (single value) result.

The **Crosstalk** results require that specific crosstalk tone exist in the multitone signal. If there are no such tones in the stimulus signal, Crosstalk results will not be available. See **Multitone Crosstalk Measurements** on page 411 for more information.

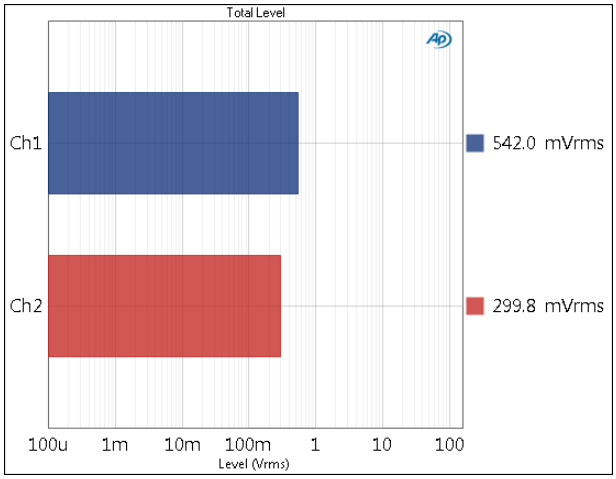
The **Skirt Width** control (page 416) in **Advanced Settings** can affect the values reported in this result.

Units

Units available for **Multitone: Crosstalk (at Tone Pair)** are:

- x/y
- %
- ppm
- dB

Multitone Analyzer: Total Level



The **Multitone Analyzer: Total Level** result uses the multitone method to provide the total rms level (the sum of the rms levels of all tones and their skirts) for each DUT channel, displayed as a meter (single value) result.

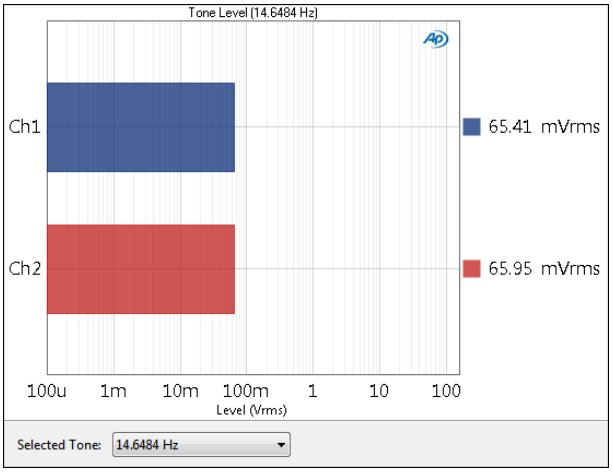
The **Skirt Width** control (page 416) in **Advanced Settings** can affect the values reported in this result.

Units

Units available for **Multitone: Total Level** are

Analog	*Digital*
• Vrms	• dBFS
• dBV	• FS
• dBu	• %FS
• dBrA	• dBrA
• dBrB	• dBrB
• dBSPL1	• dBSPL1
• dBSPL2	• dBSPL2
• dBm	
• W (watts)	

Multitone Analyzer: Tone Level



The **Multitone Analyzer: Tone Level** result uses the multitone method to provide the rms level for a selected tone for each DUT channel, displayed as a meter (single value) result.

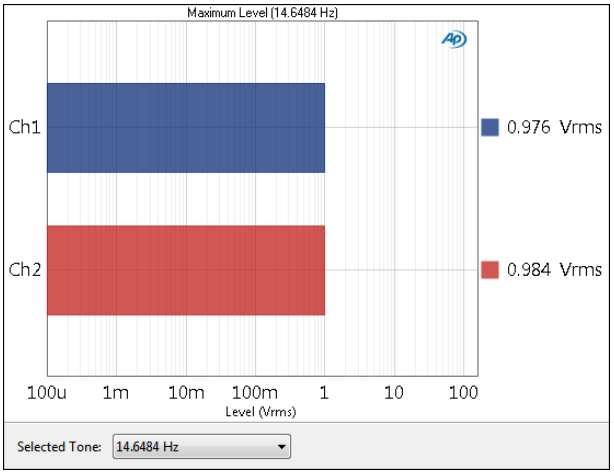
The **Skirt Width** control (page 416) in **Advanced Settings** can affect the values reported in this result.

Units

Units available for **Multitone: Tone Level** are

Analog	*Digital*
• Vrms	• dBFS
• dBV	• FS
• dBu	• %FS
• dBrA	• dBrA
• dBrB	• dBrB
• dBSPL1	• dBSPL1
• dBSPL2	• dBSPL2
• dBm	
• W (watts)	

Multitone Analyzer: Maximum Level (at Tone)



The **Multitone Analyzer: Maximum Level (at Tone)** result uses the multitone method to provide the calculated maximum level for a selected tone for each DUT channel, displayed as a meter (single value) result.

The **Skirt Width** control (page 416) in **Advanced Settings** can affect the values reported in this result.

Explanation of Max Level

Since a multitone waveform is a complex waveform, the sum of several or many individual tones, none of the individual tones can be at the maximum level of the waveform. In other words, the crest factor of the waveform will always be greater than the crest factor for a sine wave (1.414).

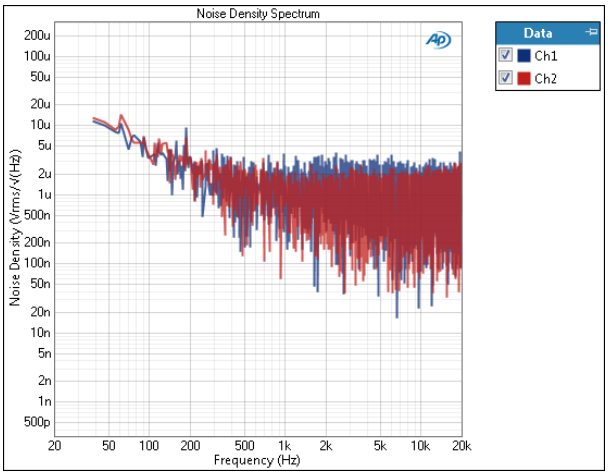
To provide a close estimation of a signal to noise ratio measurement made in the conventional method, the selected tone bin and the bins in that tone's current **Skirt width** are summed to provide an rms level result. This result is then scaled (the attenuation applied in the multitone signal creation process is factored out), providing a maximum level value.

Units

Units available for **Multitone: Maximum Level (at Tone)** are

Analog	*Digital*
• Vrms	• dBFS
• dBV	• FS
• dBu	• %FS
• dBrA	• dBrA
• dBrB	• dBrB
• dBSPL1	• dBSPL1
• dBSPL2	• dBSPL2
• dBm	
• W (watts)	

Multitone Analyzer: Noise Density Spectrum



The **Multitone Analyzer: Noise Density Spectrum** result provides an FFT spectrum display of the acquired signal. All odd-numbered bins that are not within the current Skirt width are plotted in this display; even numbered bins are not reported, ignoring stimulus tones and distortion products. This produces a noise-only result. This result is scaled for display in the accepted unit for noise density results: V/root(Hz) for analog inputs, or FS/root(Hz) for digital inputs.

> *This result provides approximately 1/2 the full number of measured data points (one for each odd-numbered bin in the FFT analysis). The graph data grid (page 649) initially shows the number of data points on the graph display, but can be set to show All Points.*

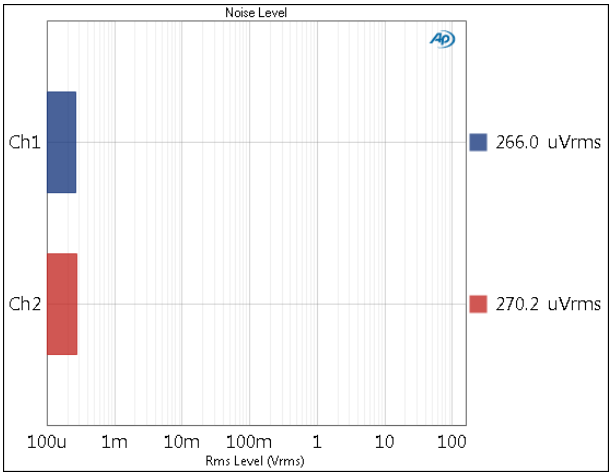
The **Skirt Width** control (page 416) in **Advanced Settings** can affect the values reported in this result.

Units

Units available for **Multitone: Noise Density Spectrum** are

X-axis	*Y-axis (analog)*	*Y-axis (digital)*
• Hz	• V/root(Hz)	• FS/root(Hz)
• dHz		
• F/R		
• %Hz		

Multitone Analyzer: Noise Level



The **Multitone Analyzer: Noise Level** result uses the multitone method to provide the rms noise level (the sum of the rms noise across the multitone bandwidth) for each DUT channel, displayed as a meter (single value) result. To produce a noise-only result, even-numbered bins and bins in the current **Skirt width** are excluded from measurement. Instead, such bins are assigned the values in the adjacent odd-numbered bins. All bins are summed.

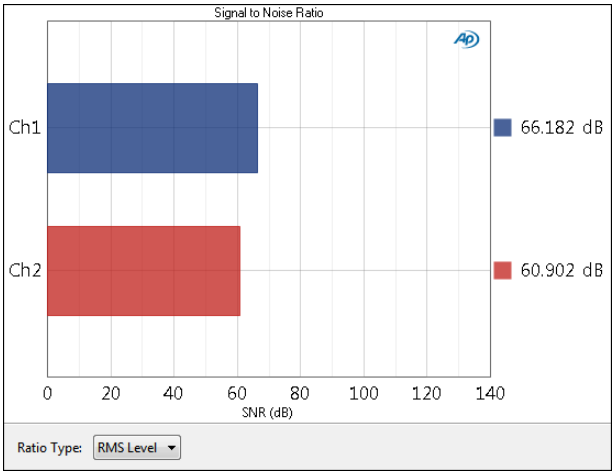
The **Skirt Width** control (page 416) in **Advanced Settings** can affect the values reported in this result.

Units

Units available for **Multitone: Noise Level** are

Analog	*Digital*
• Vrms	• dBFS
• dBV	• FS
• dBu	• %FS
• dBrA	• dBrA
• dBrB	• dBrB
• dBSPL1	• dBSPL1
• dBSPL2	• dBSPL2
• dBm	
• W (watts)	

Multitone Analyzer: Signal to Noise Ratio



Operation

See the Multitone Analyzer main topic on page 397 for controls and settings common to all multitone measurements.

The **Multitone Analyzer: Signal to Noise Ratio** result uses the multitone method to provide signal to noise ratio results. How this is measured depends upon the setting of **Ratio Type**.

The **Skirt Width** control (page 416) in **Advanced Settings** can affect the values reported in this result.

Signal to Noise Ratio (Ratio Type set to RMS Level)

This result is the ratio of two intermediate results.

First, the rms noise level (the sum of the rms noise across the multitone bandwidth) is measured for each DUT channel. To produce a noise-only result, even-numbered bins and bins in the current **Skirt Width** are excluded from measurement. Instead, the excluded bins are assigned the values in the adjacent odd-numbered bins. All bins are summed. This provides a total rms noise level result.

In a second operation, all the bins are summed to provide a total rms level result.

The total rms level is divided by the total noise level, to produce a signal to noise ratio result.

Signal to Noise Ratio (Ratio Type set to Max Level)

This result is the ratio of two intermediate results.

First, the rms noise level (the sum of the rms noise across the multitone bandwidth) is measured for each DUT channel. in the same way as described above, providing a total rms noise level result.

In a second operation, the selected tone bin and the bins in that tone's current **Skirt width** are summed to provide an rms level result. This result is then scaled the attenuation applied in the multitone signal creation process is factored out), providing a maximum level value.

This maximum level value is divided by the total noise, producing an alternative signal to noise ratio result.

See **Maximum Tone Level** on page 405 for an explanation of the calculations that provide maximum level.

Ratio Type

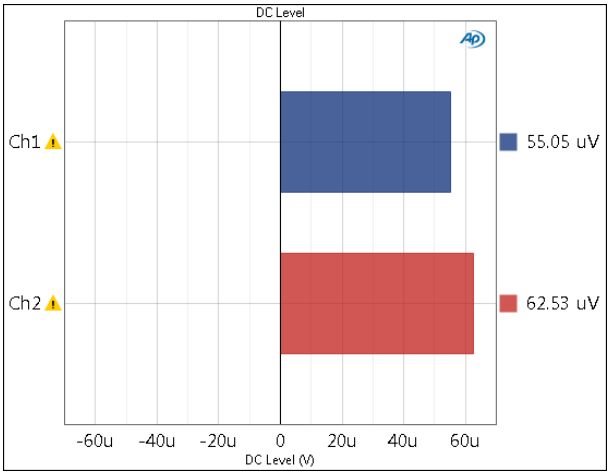
Choose the **Ratio Type** (described above) in this field.

Units

Units available for **Multitone: Signal to Noise Ratio** are:

- x/y
- dB

Multitone Analyzer: DC Level



The **Multitone Analyzer: DC Level** result uses the multitone method to measure dc level in the presence of signal for each DUT channel, displayed as a meter (single value) result. If **AC Coupled** is selected in the analog input configuration settings in Signal Path Setup, the Multitone Analyzer: DC Level result is not available. Change the setting to **DC Coupled** and start the measurement again.

The **Skirt Width** control (page 416) in **Advanced Settings** can affect the values reported in this result.

The analog measurement range for DC Level depends upon the instrument model:

- for an APx515, it is from approximately ±10 µV to ±120 V.
- for an APx52x or 555 family analyzer, it is from approximately ±10 µV to ±160 V.
- for an APx58x, it is from approximately ±10 µV to ±115 V.

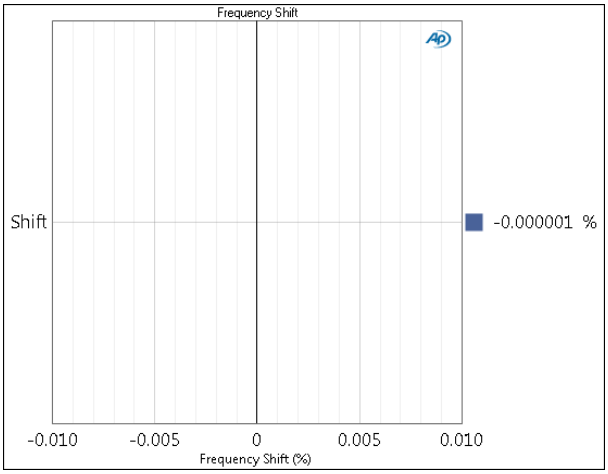
See More about DC Level on page 266.

Units

Units available for **Multitone: DC Level** are

Analog signals	*Digital signals*
• V	• D
	• hex

Multitone Analyzer: Frequency Shift



Operation

See the Multitone Analyzer main topic on page 397 for controls and settings common to all multitone measurements.

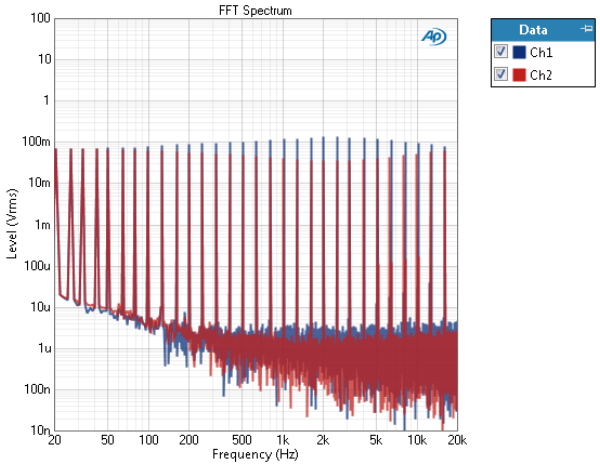
Multitone Analyzer: Frequency Shift is a diagnostic result that reports the degree of frequency shifting necessary to bring the signal acquired from the DUT into the precise match required for synchronous (windowless) FFT analysis.

Units

Units available for **Multitone: Frequency Shift** are:

- %
- ppm

Multitone Analyzer: FFT Spectrum



The **Multitone Analyzer: FFT Spectrum** result provides an FFT spectrum display (a frequency domain display) of the acquired signal. Every bin is plotted in this display. For a multitone stimulus, this will show the individual tones rising from a floor of distortion + noise.

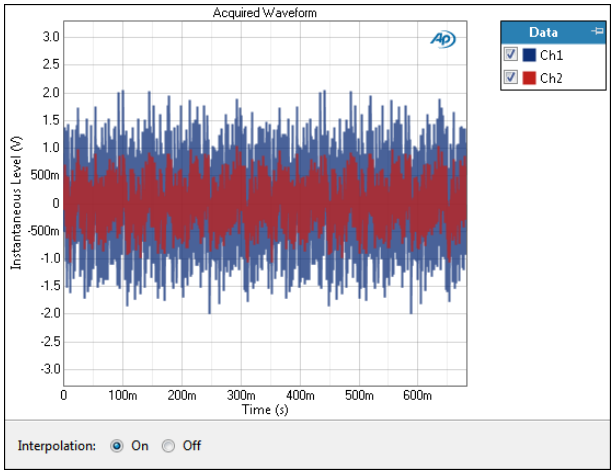
> *This result provides the full number of measured data points (one for each bin in the FFT analysis). The graph data grid (page 649) initially shows the number of data points on the graph display, but can be set to show All Points.*

Units

Units available for **Multitone: FFT Spectrum** are

X-axis	*Y-axis (analog)*	*Y-axis (digital)*
• Hz	• Vrms	• dBFS
• dHz	• dBV	• FS
• F/R	• dBu	• %FS
• %Hz	• dBrA	• dBrA
	• dBrB	• dBrB
	• dBSPL1	• dBSPL1
	• dBSPL2	• dBSPL2
	• dBm	
	• W (watts)	

Multitone Analyzer: Acquired Waveform



The **Multitone Analyzer: Acquired Waveform** result provides a time domain display of the acquired signal.

Units

Units available for **Multitone: Acquired Waveform** are

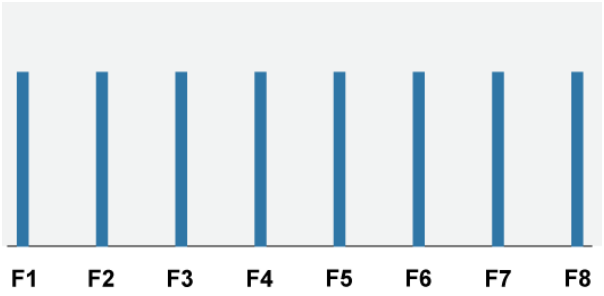
Analog signals	*Digital signals*
• V	• D
	• hex

More about multitone

Multitone testing uses FFT analysis with a special stimulus waveform. A multitone stimulus signal is a complex waveform, the combination of many sine waves. There are typically from 3 to 30 or more tones in a multitone signal; to the ear, it sounds like a dissonant organ chord. The analysis can provide frequency response, phase response, crosstalk response, distortion response, noise response and other results.

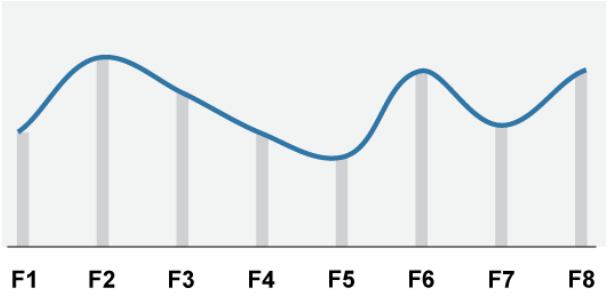
Multitones are useful for fast testing applications, and since a multitone stimulus can be brief and unobtrusive, multitone can be considered for in-service testing.

The example below shows a diagram representing a frequency domain view of an 8-tone multitone signal.



The multitone signal is applied to the DUT and an acquisition of the device's output is made for analysis.

The FFT data can be interpreted in a number of ways, extracting amplitude, phase, and other results. Multitone can provide detailed, accurate and fast measurements using a short multitone "burst."



Multitone: Level displays a frequency response curve that plots the level across all of the tones at analysis. The amplitudes in the FFT bins that correspond to the multitone stimulus tones are shown as a light gray lines. The dark line represents the frequency response curve.

APx500 Multitone implementation

To produce useful results, the APx multitone analyzer must perform a windowless FFT on the acquired multitone audio signal. This requires that the analyzer be provided with the precise frequency values for each of the tones in the multitone stimulus signal. For multitones generated in the APx generator from a multitone signal definition, this information is available within the system. For playback of generator waveforms or for external source, multitone signal definition information must be made available to the analyzer.

FFT analysis

The multitone is analyzed in the frequency domain. The frequency and phase results of both tone bins and non-tone bins are used to derive the various measurement results.

Windowless FFT

Windowed FFTs filter the FFT record in the time domain using amplitude window functions to eliminate the effect of any discontinuity at the record ends. However, windowed FFTs provide limited frequency resolution.

Current technology has made it trivial to create stimulus signals with exactly the same sample rate and length as the FFT, permitting FFT analysis without using windows in most use cases. This method produces FFT results of very high frequency resolution and none of the distortions created by window leakage.

Bins, bandwidth and bin width

A frequency-domain FFT result contains amplitude and phase values in frequency bins. The number of bins is equal to one-half of the FFT record length: an FFT length of 8192 samples would produce a result with 4096 bins.

The bandwidth of the result is equal to one-half of the sample rate: an FFT record of a sample rate of 48 kHz would produce a result with a 24 kHz bandwidth.

The bin width is equal to the bandwidth divided by the number of bins: the examples above would produce a result with bin widths of 5.86 Hz. The amplitude and phase value of each bin would be the sum of the amplitudes and phases all frequencies that fall into that bin.

As another example, an FFT record of a length of 48000 samples and a sample rate of 48 kHz would have 24000 bins across a bandwidth of 24 kHz. Each bin would be 1 Hz wide.

Multitone and even and odd bins

Tones in an Audio Precision multitone stimulus signal are always set on even-numbered bins. This has the desirable effect of causing any distortion products to fall into even bins as well. At analysis time, all bins can be measured, or only tone bins, or only non-tone bins, or only odd bins. This enables the analysis software to report level (tone bins), distortion plus noise (all non-tone bins), noise (all odd bins) and so on. When reporting only the odd bins, you would correctly expect that the noise would be about half of the value of the

true noise, as only half the bins are used. APx500 solves this by reporting values for all bins, using the value of the adjacent odd bin for each ignored even bin.

Skirts

A DSP generated and analyzer multitone is very precise, and a view of the results in loopback reveals zero stimulus energy outside of any FFT bin. But real world DUTs can exhibit a variable frequency shift, commonly jitter or flutter. Even when the multitone signal is properly frequency-shifted to match the bin centers, jitter or flutter will spread stimulus tone energy around the tone frequency, creating skirts surrounding the FFT bin(s) in analysis.

For multitone results based on rms level results, APx500 includes the energy in the skirts by including adjacent bins. For multitone results based on noise results, APx500 rejects the energy in the skirts by excluding adjacent bins. By default, the skirt width is 3.00 % of the tone frequency. For a 1 kHz tone, this would be a skirt width of 30 Hz.

Signal Definition

Multitone signal definition creation

A well-formed multitone waveform requires tones that are on specific frequency centers (for windowless FFTs, below) and with optimized phase relationships (for a low crest factor). These will vary with the chosen Sample Rate and Length. APx500 provides a comprehensive multitone creation capability within the Multitone Measurement environment. A multitone signal definition can be created, imported or extracted from an audio file. This definition can be loaded directly into the APx generator, or it can be exported as definition data or used to create a multitone WAV file.

Signal Generation

Output of an internally generated multitone (closed loop)

The APx generator can output a multitone from a multitone signal definition, or by playing a multitone audio file using APx500's generator waveform capability.

Output of a generator waveform multitone (from a WAV file)

When the APx generator is outputting a multitone by playing a generator waveform WAV file, a multitone signal definition is extracted from the audio in the WAV file to provide the analyzer with the necessary frequency center information.

Output of a generator waveform multitone (from a coded audio file)

APx500 does not incorporate decoders to open or analyze coded audio files. Therefore, when the APx generator is outputting a multitone by playing a generator waveform coded audio file, a multitone signal definition cannot be extracted from the audio in the file. A multitone signal definition containing frequency center information that corresponds to the audio in the coded file must be provided.

Signal Analysis

Acquisition

The multitone is acquired and the frequencies of the tones are identified. These are compared to the frequencies in the current multitone signal definition. The acquired audio is sample rate converted (SRC) to shift the pitch of the acquired multitone until the frequencies exactly match the frequencies in the multitone signal definition. This is required for windowless FFT analysis.

External Source configuration (open loop)

When APx500 is making a multitone measurement in external source configuration, a multitone signal definition containing frequency center information that corresponds to the audio being played in the DUT must be provided.

Triggering

The analysis of the multitone does not begin until the trigger function is satisfied. Four trigger modes and a trigger delay allow you to optimize the acquisition and analysis.

Results

Level

FFT analysis provides level and phase results for each frequency bin. In APx500 multitone measurements, these data are used in a number of ways.

- The rms level of every bin is reported. This provides the FFT spectrum result.
- The rms level of every non-tone bin is reported, with skirt exclusion. This provides the TD+N (total distortion + noise) result, which is also used as a source to derive the TD+N Ratio and TD+N Level results.
- The rms level of each stimulus tone bin is reported, with skirt inclusion. From these are derived the Level, Gain, Relative Level, Deviation, Total RMS Level, RMS Level (at Tone), Maximum Level (at Tone) results, and the signal component of Signal to Noise Ratio.
- The rms level of each odd bin is reported, with skirt exclusion. From these are derived Noise RMS Level, Noise Density Spectrum and the noise component of Signal to Noise Ratio.
- The instantaneous voltage level in the zero (0) Hz bin is reported as DC Level.

Multitone Crosstalk Measurements

APx multitone signals that have not been specifically configured for crosstalk measurements are mono; that is, the same multitone signal is applied to all APx output channels. Crosstalk measurements, however, require that one or more tones exist in each channel that do not exist in another, so that the degree of crosstalk can be observed and measured.

APx allows you to modify a mono multitone signal definition to create two related multitones that contain crosstalk tones; that is, that have one or more tones unique to each signal. The resultant multitone signal definitions are called “A” and “B”.

In APx, you have the choice of measuring crosstalk at one specified frequency or tone, or at all tones. APx accomplishes this by adding a tone 0.5 % higher than the specified tone(s) and then deleting the original specified tone(s), for signal A; and by adding a tone 0.5 % lower than the specified tone(s) and then deleting the original specified tone(s), for signal B.

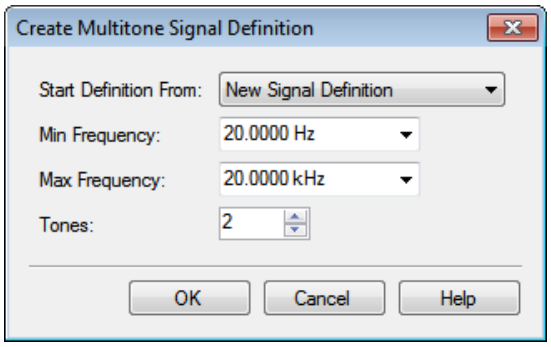
View Multitone Signal Definition

See **Edit Multitone Signal Definition** dialog on page 412 for detailed discussions of the fields in both the **View Multitone Signal Definition** and the **Edit Multitone Signal Definition** dialogs. The fields in these two dialogs and the descriptions of their content are the same.

The **View Multitone Signal Definition** dialog allows you to view a multitone signal definition that cannot be edited. You can optionally export the signal definition or create an audio file (WAV file) from the definition. Multitone signal definitions that cannot be edited include preset multitone signal definitions, signal definitions extracted from audio files, and signal definitions associated with External Source stimulus signals in open loop testing.

Create a Multitone Signal Definition

Also see **Edit Multitone Signal Definition** on page 412.



This dialog opens from **Signal Generation > Waveform > Create New...** on the Multitone Analyzer settings panel. The selections made here can be edited in more detail or changed entirely in the following dialog, **Edit Multitone Signal Definition** (page 412).

Start Definition From

Select an existing multitone definition from this list to use as a starting point for your new multitone.

New Signal Definition is the default, which provides a simple, 2-tone definition as a starting point. **Sample Rate** is set to 48 kHz for an analog signal path, or to the current sample rate for a digital signal path. **Length** is set to 8192 samples, **Min Frequency** to approximately 20 Hz and **Max Frequency** to approximately 20 kHz. The actual frequencies are constrained by the **Sample Rate** and **Length** defined in **New Signal Definition**. You can edit the Min **Frequency**, **Max Frequency** and **Tones** fields as described below.

Min Frequency

The minimum frequency in the definition selected in **Start Definition From** is shown here.

If **Start Definition From** is set to **New Signal Definition**, you can edit the frequency in this field to establish a new **Min Frequency** for the definition you are creating. Note that the available frequencies are constrained by the **Sample Rate** and **Length** defined in **New Signal Definition**.

Max Frequency

The maximum frequency in the definition selected in **Start Definition From** is shown here. If **Start Definition From** is set to **New Signal Definition**, you can edit the frequency in this field to establish a new **Max Frequency** for the definition you are creating. Note that available frequencies depend upon the **Sample Rate** and **Length** settings in the definition selected in **Start Definition From**.

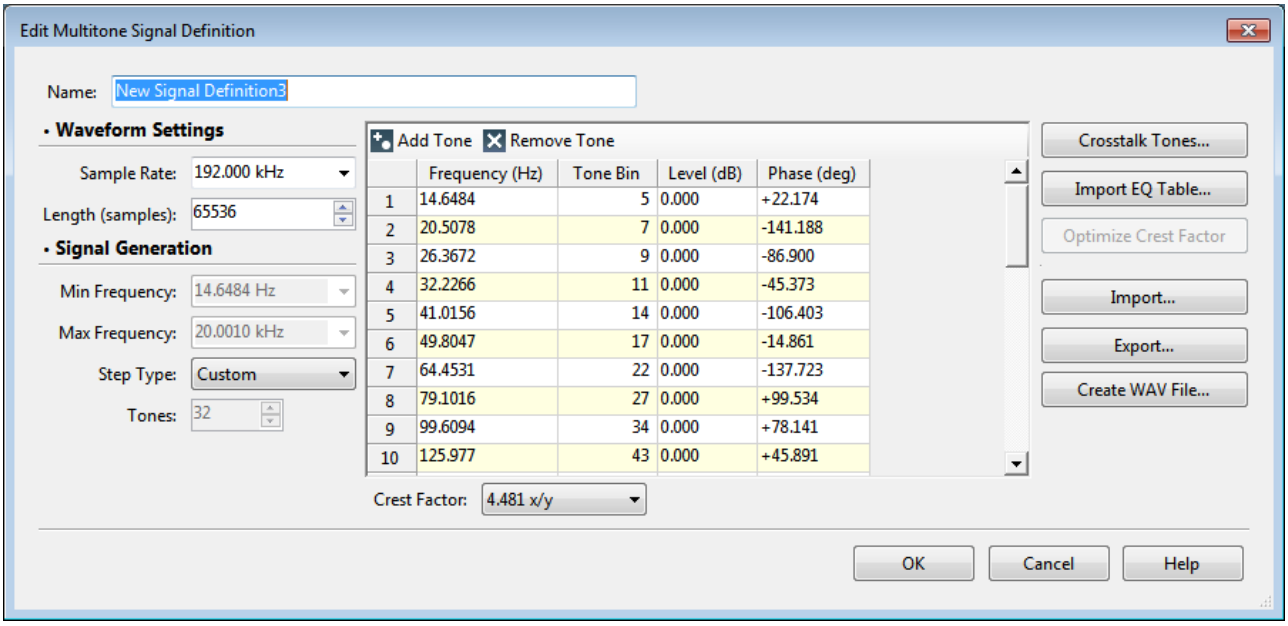
Tones

The number of tones in the definition selected in **Start Definition From** is shown here. If **Start Definition From** is set to **New Signal Definition**, you can edit this field to change the number of tones for the definition you are creating.

OK

When you have made your selections, click **OK** to open the next dialog, **Edit Multitone Signal Definition**.

Edit Multitone Signal Definition



This dialog allows you to edit a multitone signal definition. You can optionally import a signal definition, export the signal definition or create an audio file (WAV file) from the definition.

APx500 does not allow all multitone signal definitions to be edited.

Name

Enter a unique name for your multitone signal definition here.

Waveform Settings

Sample Rate

Enter the sample rate for your multitone here.

Length (samples)

Enter the length of the multitone signal here.

Signal Generation

Values entered in these fields will immediately be propagated into the tone description grid, to the right.

Min Frequency

Enter the minimum desired frequency for your multitone signal here.

Max Frequency

Enter the maximum desired frequency for your multitone signal here.

Step Type

The **Step Type** determines the spacing between the tones in the multitone signal. Choices are

- **Logarithmic**
  Logarithmic will logarithmically space the number of tones set in Tones across the frequency range set in Min and Max Frequency.
- **Linear**
  Linear will linearly space the number of tones set in Tones across the frequency range set in Min and Max Frequency. When the Step Type is Linear, the Step Size field (below) becomes available.
- **Custom**
  When any tone has been manually edited in the grid (added, deleted, or the frequency is changed), the Step Type becomes Custom, and the Min Frequency, Max Frequency and Tones fields become unavailable.

Tones

Enter the number of tones here.

Step Size

When the **Step Type** is **Linear**, the **Step Size** field becomes available. **Step Size** is equal to the range between **Min** and **Max Frequency** divided by the number of **Tones**. Editing the value of any of these three variables will change the values of the other two.

Tone Description Grid

The grid in the center of the dialog lists each tone in the multitone definition, sorted by frequency. When the information is available, the Tone Bin, Level and Phase are also listed.

Crest Factor

This field shows the crest factor (page 416) of the current multitone definition.

Crosstalk Tones...

When creating or editing a multitone definition, you can create tones on alternate channels that are slightly offset in frequency, for the purpose of measuring crosstalk. This button opens the **Create Crosstalk Tones** dialog. See Creating Multitone Crosstalk Tones on page 416 for a detailed discussion.

Import EQ Table...

See Generator Equalization on page 192 for general information on this subject.

You can equalize a multitone by entering various levels in the **Level** column in the **Edit Multitone Signal Definition** dialog. You can also import an existing generator equalization table file and apply it to the current multitone signal definition. Click **Import EQ Table...** and use the file browser to select a compatible generator EQ table in *.xls or *.csv format.

The level applied to tones that are between EQ points in the file will be set to an interpolated value.

Import EQ Table operations are additive. Reset all tone level values to 0 dB before re-importing to prevent addition.

Optimize Crest Factor

When you have chosen all the tones and optionally edited levels, added crosstalk tones, and applied an EQ curve, click **Optimize Multitone**. This control applies a set of algorithms to find phase settings for each tone that result in a multitone waveform with a lower crest factor (page 416). Optimization proceeds for 1 second, and uses the optimized waveform if it shows an improved crest factor. Subsequent applications of **Optimize Multitone** will re-optimize the waveform, but there is rarely any improvement over the optimization achieved in the first pass. Multitone waveforms with a great number of tones may show little or no improvement in crest factor after optimization.

Note: if a low crest factor (page 416) is important to your application, consider designing a multitone with fewer tones.

Import...

Click **Import...** to open a multitone signal definition from a file on disk. Use the file browser to select a CSV or XLS multitone signal definition file.

Export...

Your multitone signal definition is added to the current project as an attached project item. You can also export the signal definition as a CSV or XLS file. Click **Export...** to open a Save File browser.

Create WAV File....

You can use the current multitone signal definition to create a multitone audio file, in the mono or stereo linear PCM WAV format. Click **Create WAV File...** and enter the **Duration** (up to 5 minutes) and **Bit Depth** (either 16-bit or a 24-bit) in the dialog box. Click **OK** to open a file browser.

Multitone Crosstalk Measurements

APx multitone signals that have not been specifically configured for crosstalk measurements are mono; that is, the same multitone signal is applied to all APx output channels. Crosstalk measurements, however, require that one or more tones exist in each channel that do not exist in another, so that the degree of crosstalk can be observed and measured.

APx allows you to modify a mono multitone signal definition to create two related multitones that contain crosstalk tones; that is, that have one or more tones unique to each signal. The resultant multitone signal definitions are called “A” and “B”.

In APx, you have the choice of measuring crosstalk at one specified frequency or tone, or at all tones. APx accomplishes this by adding a tone 0.5 % higher than the specified tone(s) and then deleting the original specified tone(s), for signal A; and by adding a tone 0.5 % lower than the specified tone(s) and then deleting the original specified tone(s), for signal B.

This results in one or more crosstalk “tone pairs” between the two signals, offset slightly around the original tone centers.

Operation

The first task is to create a multitone with crosstalk tones, as described above. This is done in the Edit Multitone Signal Definition dialog. First, design the multitone definition with tones, sample rate, etc. according to your testing requirements.

Then click **Crosstalk Tones...** to open the **Create Crosstalk Tones** dialog and specify the tones you want. Close the dialog.

Observe that there are now an **“A”** and a **“B”** tab for the signal definition grid, each listing the tones in the two multitones. Optimize the crest factor and click **OK** to save the definition in the project.

Results

Two multitone crosstalk results are available.

Crosstalk result

The **Crosstalk** result requires at least two crosstalk tone pairs in the stimulus multitone, and provides a crosstalk ratio vs. frequency result. See Multitone Ana-

lyzer: Crosstalk (page 404), and the Multitone Analyzer main topic (page 397) for more detailed operational information.

Crosstalk (at Tone Pair) result

The Crosstalk (at Tone Pair) result provides a crosstalk ratio value for the selected tone pair. See Multitone Analyzer: Crosstalk (at Tone Pair) (page 404) and the Multitone Analyzer main topic (page 397) for more detailed operational information.

Removing or Changing Crosstalk Tones

If you open a multitone signal definition that already has A/B crosstalk tone pairs, you will not be able to add more tone pairs. If you click **Crosstalk Tones...** from the **Edit Multitone Signal Definition** dialog (page 412), you will see that the only available option is **Remove Crosstalk Tones**. Accepting this option will delete the "B" signal definition, leaving a mono multitone definition that is not configured for crosstalk measurement.

Note that accepting this option does not return the specified crosstalk tones to their original frequencies, but instead keeps the "A" definition, in which the specified crosstalk tones were shifted up by 0.5 %.

Mapping the A and B multitones to APx output channels

The APx generator can output a maximum of two different signals simultaneously. When the output format supports more than two signals, these two signals can be arbitrarily mapped to the output channels using controls in Advanced Settings. The default mapping is the "A" multitone signal to odd channels, and the "B" multitone signal to even channels.

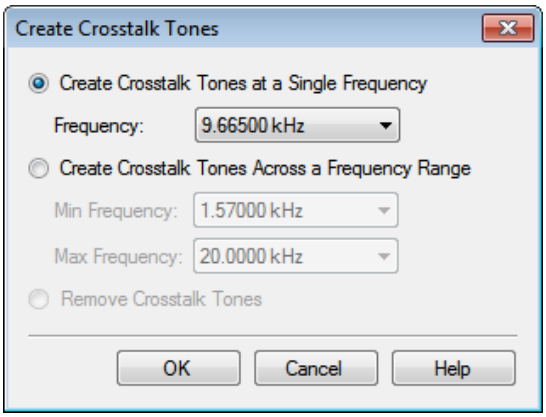
Creating Multitone Crosstalk Tones

In APx500, multitone definitions and waveforms that do NOT contain crosstalk tones are monaural, single-channel definitions or waveforms.

For crosstalk measurements, the APx500 multitone implementation requires a stereo, two-channel definition or waveform. For each frequency of interest, a pair of tones is created (one tone slightly above the frequency of interest, and one tone slightly below). The upper tone of the tone pair is placed in the "A" channel, and the lower in the "B" channel.

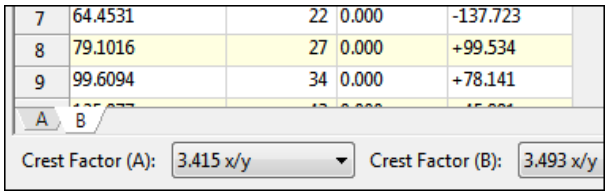
Create Crosstalk Tones dialog

In the **Edit Multitone Signal Definition** dialog (page 412), click **Crosstalk Tones...** The **Create Crosstalk Tones** dialog will open.



Create Crosstalk Tones at a Single Frequency

Use the first selection to create a pair of crosstalk tones at a single frequency of interest. Choose the frequency from the list of available tones in the current multitone signal definition. The single-channel multitone signal definition will be converted to a two-channel signal definition. (When you return to the previous dialog, you will notice that an "A" and a "B" tab have been added to bottom of the tone definition grid.) The tone you select will be removed from the definition, and a crosstalk tone pair will be created. The upper tone will be placed in the A channel, and the lower tone will be placed in the B channel.



Create Crosstalk Tones Across a Frequency Range

Use this selection to create a pair of crosstalk tones at every frequency within a range of interest. Set the range minimum and maximum frequencies using the **Min** and **Max** lists, which show the available tones in the current multitone signal definition. The single-channel multitone signal definition will be converted to a two-channel signal definition. (When you return to the previous dialog, you will notice that an "A" and a "B" tab have been added to bottom of the tone definition grid.} The tones you select will be removed from the definition, and a crosstalk tone pair will be created for each tone. The upper tones of each pair will be placed in the A channel, and the lower tones will be placed in the B channel.

"Not enough FFT bins" error

At low frequencies for short multitone FFT lengths, there will not enough FFT bins available to create a crosstalk tone pair. To create a successful multitone with crosstalk tones, either increase the **Length**, or specify crosstalk tones of higher frequencies.

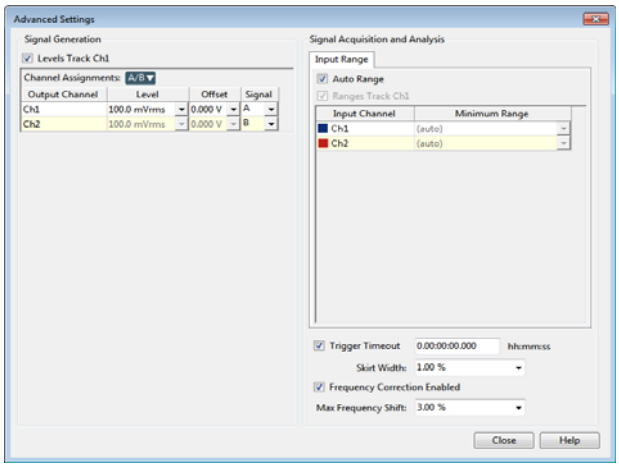
Remove Crosstalk Tones

If a multitone signal definition already has crosstalk tones on the A and B channels, new crosstalk tones cannot be added without first removing the existing crosstalk tones. Selecting **Remove Crosstalk Tones** deletes the B channel, resulting in a monaural, single-channel multitone signal definition.

Note that removing the existing crosstalk tones does NOT return the tones to their original frequencies, as were specified in the original multitone signal definition. That information is not carried forward in the definition. Instead, the tone(s) that had been shifted up slightly (the A channel crosstalk tones) become the tones in the new definition.

Advanced Settings for the Multitone Analyzer

The default settings here are appropriate for most measurements. You may want to make minor adjustments for special situations.



Also see More About Multitone and APx500 Multitone implementation on page 409.

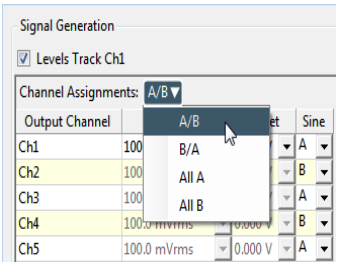
Signal Generation

If **Levels Track Ch1** is checked (the default), the generator output level and DC offset values for channel 1 are copied to the other channels, and values for channels greater than 1 cannot be edited. Any changes made to channel 1 are reproduced in the other channels.

To set individual generator output channel levels or offsets, uncheck the **Levels Track Ch1** checkbox and enter values in the output channel Level or Offset fields.

Set Channel Assignments for crosstalk multitones

For multitones that include crosstalk tones, you can also set channel mapping. By default, crosstalk multitones map channel A audio to odd output channels, and channel B audio to even output channels. These assignments can be remapped by changing individual settings in the **Signal** column, or by selection one of several presets from the **Set Channel Assignments** menu.



Set Channel Assignments for generator waveforms

For stereo generator waveform multitone files, you can set channel mapping. By default, channel 1 audio is mapped to channel 1 output, and channel 2 to channel 2. These assignments can be remapped by changing individual settings in the **Wave Ch** column.

Signal Acquisition and Analysis

Input Range tab

> *Digital inputs do not require ranging, and the Input Range tab is unavailable when inputs are set to a digital format.*

Auto Range

If **Auto Range** is checked (the default), each analog channel determines its input range automatically, based on the level of the input signal. If the input signal level changes beyond a ranging threshold, **Auto Range** will cause the input ranging circuits to move up or down for proper ranging.

Fixed Range

Alternatively, you can set a fixed minimum range for the analog inputs. Uncheck **Auto Range** and enter the expected maximum level in the **Minimum Range** filed. The software will choose the lowest range that will accommodate that level; this becomes the minimum range available, and the analyzer will not reset to a lower range. Fixed range is best for signals that vary rapidly, including tone bursts, perceptual audio speech samples, and program material such as voice or music. See **When autoranging isn't suitable** on page 626.

If a signal greater then the **Minimum Range** is presented to the analog input, the ranging circuits will range up to accommodate the higher level, and then drop again when the signal level falls. The range will not be set below the **Minimum Range**.

Ranges Track Ch1

If **Ranges Track Ch1** is checked (the default), the fixed input range setting for channel 1 is copied to the other channels, and range settings for channels greater than 1 cannot be edited. Any changes made to channel 1 are reproduced in the other channels. If **Ranges Track Ch1** is not checked, the fixed input range setting for each channel can be set independently.

Read more about ranging in Chapter 98.

Advanced Settings for Multitone does not have a Settling tab.

Trigger Timeout

When a **Trigger** selection other than **Free Run** (untriggered) is set and **Trigger Timeout** is not set, the APx500 software will wait indefinitely for a valid trigger condition. When **Trigger Timeout** is set, the operation will wait for the duration set in the **Trigger Timeout** field, and then will cancel the measurement with a notice. Minimum time is 0.1 s, maximum time is 7 days.

Enter the **Trigger Timeout** following this pattern:

d:hh:mm:ss.s

where d = days, hh = hours, mm = minutes, ss = seconds. Days and fractional seconds are optional.

Skirt Width

Flutter, jitter or other variable frequency shift in the device or transmission channel will create sidebands of energy around each of the tones, called skirts. Some small amount of energy is also spread into the skirts as an artifact of the multitone frequency correction. For correct rms level measurements, energy in the skirts must be added to the energy in the tone. For correct noise measurements, energy in the skirts must be excluded from the measurement.

The **Skirt Width** control specifies the width around the tone that is to be considered as skirt energy. The default (and maximum) is 3 %.

Frequency Correction Enabled

In practical devices and transmission channels, audio signals can undergo frequency changes (sample rate changes). APx multitone measurements, which use unwindowed FFTs for analysis, must frequency shift the acquired waveform so that it matches the internal reference exactly. This checkbox enables frequency correction.

By default, **Frequency Correction** is checked (on) for digital inputs. The multitone measurement will run faster with **Frequency Correction** disabled.

Max Frequency Shift

When **Frequency Correction** is enabled, the **Max Frequency Shift** field is available.

The value in **Max Frequency Shift** represents the maximum percentage of shift allowed. If the input signal requires a shift beyond this value, the measurement will fail. The default value is 3 %; the range is 0 % to 5 %.

Crest Factor

The crest factor of a waveform is the ratio of its peak value to its rms value. Crest factor can be expressed as a ratio or fraction (4:1 or 4/1, for example) or as a decibel (12.041 dB, in this case). When expressed as a ratio or fraction, crest factor is often reduced to lowest terms (4, in this case).

In APx500, the amplitude of a multitone waveform is set to the value in the generator **Level** setting field. The higher the waveform's crest factor, the lower the level of the component tones within the waveform.

Typical crest factors:

WAVEFORM	RATIO	DECIBEL
square wave	1	0 dB
sine wave	1.414	3.01 dB
triangle wave	1.732	4.77 dB
ambient noise (typical)	3	10 dB
speech (typical)	4	12 dB
APx 32-tone multitone	4.481	13.028 dB
some musical instruments (typical)	8–10	18–20 dB
fast transient (such as a gunshot)	30 or more	30 dB or more

Recording Multitone Audio to a File

This topic discusses the **Save to File** feature when invoked from the Multitone Analyzer. See Recording Audio to a File on page 381 for a discussion of the **Save to File** feature when invoked from the Measurement Recorder.

Overview

The Measurement Recorder and the Noise Recorder can optionally record acquired audio to a file on disk.

Transform record

As explained in APx500 Multitone Implementation on page 409, at analysis time a multitone signal must match the multitone signal definition in sample length and in sample rate. This often requires frequency (sample rate) shifting and truncation of the raw input acquisition. It is this truncated and frequency-shifted

audio, the audio actually used for analysis, that is recorded to file; not the raw audio acquisition. This audio is also be referred to by the terms *transform record* or *FFT record*.

Note: APx timing constraints will sometimes cause a timeout error when saving to a network folder or a removable disk. We recommend saving audio files to a local disk.

On the Measurement or Noise Recorder panel, check the **Save to File** checkbox. Click **File Settings** to specify the file name, location and file format.

Save to File is not available when Input Configuration is set to File (Analog Units) or File (Digital Units).

Save Acquisition to File Settings dialog

Location

Enter or browse to the destination folder here.

Name

Name the output file here. Choose from one of these selections:

- If file exists, replace it
- If file exists, append timestamp to the file name

Format

Choose the file format in the Format field.

For both analog inputs and digital acquisitions, there are three linear file formats available:

Multiple Mono PCM (.wav)

If **Multiple Mono PCM (.wav)** (the default) is selected, each audio channel of the acquisition is saved as an individual monaural WAV file, using the Microsoft WAVE_FORMAT_PCM (Type 1, 24 bit) file format. For multichannel audio of channel count n, n mono WAV files are created, with the numeral n appended to the file name.

Multichannel PCM (.wav)

If **Multichannel PCM (.wav)** is selected, all the audio channels of the acquisition are saved in one multichannel file, using the WAVE_FORMAT_PCM format. For 1 channel and 2 channel acquisitions, this selection records a standard WAVE_FORMAT_PCM (Type 1, 24 bit). For multichannel acquisitions, this selection records a multichannel WAVE_FORMAT_PCM file in a proprietary 24-bit format, compatible with Sony Sound Forge and other audio editing applications.

Extensible Multichannel PCM (.wav)

If **Save as WAVE_FORMAT_EXTENSIBLE** is selected, all the audio channels of the acquisition are saved in one multichannel file, using the Microsoft standard WAVE_FORMAT_EXTENSIBLE format, at 24 bits.

Bit Depth

This field is only available for analog acquisitions. The default Bit Depth (word length) is 24 bits. You can also select 16 bits.

Analog scaling and sample rates

When recording an analog acquisition, proprietary scaling metadata is embedded in the WAV file. The scaling factor relates the analog voltage to the digital level (Vrms/FS), and is used for scaling the display of values when using the APx500 file input configuration.

For analog acquisitions, the sample rate is set to be the same as the sample rate determined by the analyzer input configuration analog bandwidth setting, set in Signal Path Setup, as follows:

Bandwidth	Sample Rate
22 kHz	48 kHz
45 kHz	96 kHz
90 kHz	192 kHz
250 kHz	624 kHz
500 kHz	1.248 MHz
1 MHz	2.496 MHz

Input bandwidths beyond 90 kHz require the BW52 Option; 1 MHz bandwidth requires input set to 1 channel.

The sample rate can be changed in the Advanced Settings dialog.

Duration for linear recordings

Windows WAV files are limited to 4 GB. For 24-bit stereo WAV files, this is approximately 1 hour at 192 kHz. Mono files and files with lower sample rates or bit depths can have longer durations. For longer recordings, APx automatically creates a sequential file (or a set of sequential files) providing a very long recording capability without loss of any data.

Note: The sequential file feature for very long recordings is resource-intensive, and requires a high performance level in terms of PC CPU and hard disk speed, beyond the minimum recommended computer required to run APx500.

Noise (Q-peak)
(Sequence Mode)

Noise (Q-Peak) Level view provides a single-value measurement of the noise level present at the output of each DUT channel, using a Q-peak detector and the CCIR/ITU-R BS 468-4 weighting curve. Use this measurement if you are required to provide CCIR/ITU-R BS 468-4 results.

Preparing for a Noise Measurement

If you have not yet set up your test, first go to Signal Path Setup to select and configure inputs and outputs, references and other settings. See Chapter 7 for information about Signal Path Setup.

To measure the residual noise in a DUT, select **None** as the waveform, which will run the measurement with no stimulus signal. We recommend disconnecting any devices from the DUT inputs (including the APx generator, if connected), and terminating the DUT inputs with a low resistance, typically 0 ohms to a few hundred ohms.

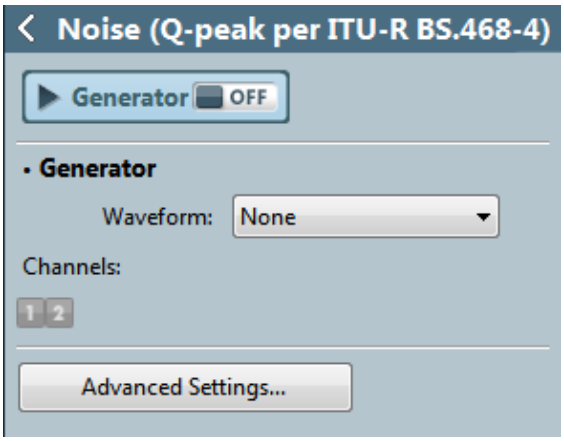
To test using a noise stimulus, select **Noise** or **Browse for File**, and connect the generator outputs to the DUT inputs.

Generator

Typically, the Noise (Q-Peak) measurement is used without a stimulus signal (**Waveform: None**), but a Noise signal or Generator Waveform files are also available.

If you choose to make a noise measurement with a stimulus signal, these signals are available. When **Waveform** is set to **Noise**, these choices will be available from the **Noise Shape** selector:

- White
- Pink
- IEC 60268-1
- BS EN 50332-1



See **More About Noise Signals** on page 423 for descriptions of these signals.

Alternatively, you can select a generator waveform file as the test signal, using the file browser in the Waveform drop-down menu.

See Chapter 6 for detailed information about the using the APx Generator and making Waveform, Level and Frequency settings, or for information about using External Source.

Level

Set the generator level here. Note that with Noise signals the generator level values are scaled differently; see **More About Noise Signals** on page 423.

Analyzer

The average Q-peak level across the default **Acquisition Time** of 250 ms is shown on the meter bar display. To adjust the **Acquisition Time** or the **Delay Time** (which sets the time of the beginning of the acquisition), go to **Advanced Settings**.

This measurement can be configured to measure jitter in Signal Path Setup > Input > Measure. Read

detailed information about jitter generation and measurement beginning on page 69.

Filtering

A fixed CCIR/ITU-R BS 468-4 weighting curve filter is used for Noise (Q-peak) measurements.

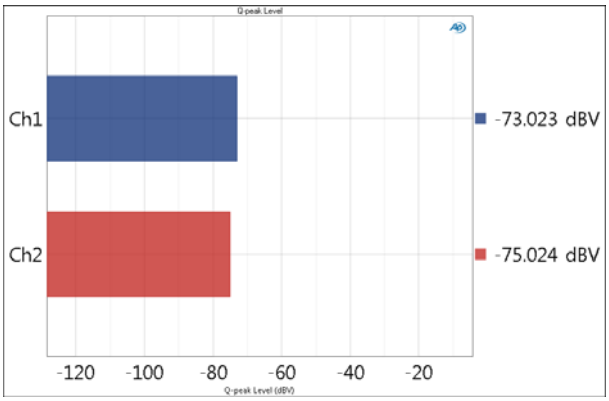
Low pass, high pass and weighting input filter settings made in Signal Path Setup > Input/Output > Filters will also affect this measurement. See Chapter 97 for more information about filtering.

Advanced Settings

Advanced Settings makes available individual channel output level and offset controls, and input ranging and settling controls. See Advanced Settings on page 422.

See Chapter 104 for more information about units of measurement.

Q-Peak Level



Noise (Q-Peak) Level provides a single-value meter result, displaying the Q-Peak noise level from each DUT channel, as measured at each analyzer input.

Units

Units available for Noise (Q-peak) are

Analog Signals	*Digital Signals*
• Vrms	• dBFS
• dBV	• FS
• dBu	• %FS
• dBrA	• dBrA
• dBrB	• dBrB
• dBSPL1	• dBSPL1
• dBSPL2	• dBSPL2
• dBm	
• W (watts)	

Noise (RMS)
(Sequence Mode)

The Noise (RMS) measurement provides a single-value result that shows the noise present at the output of each DUT channel, using an rms detector. Typically, the Noise measurement is used without a stimulus signal (**Waveform: None**), but a Noise signal or Generator Waveform files are also available.

The Noise (RMS) measurement available in APx500 is:

- Noise: RMS Level

Preparing for a Noise Measurement

If you have not yet set up your test, first go to Signal Path Setup to select and configure inputs and outputs, references and other settings. See Chapter 7 for information about Signal Path Setup.

To measure the residual noise in a DUT, select **None** as the waveform, which will run the measurement with no stimulus signal. We recommend disconnecting any devices from the DUT inputs (including the APx generator, if connected), and terminating the DUT inputs with a low resistance, typically 0 Ω to a few hundred ohms.

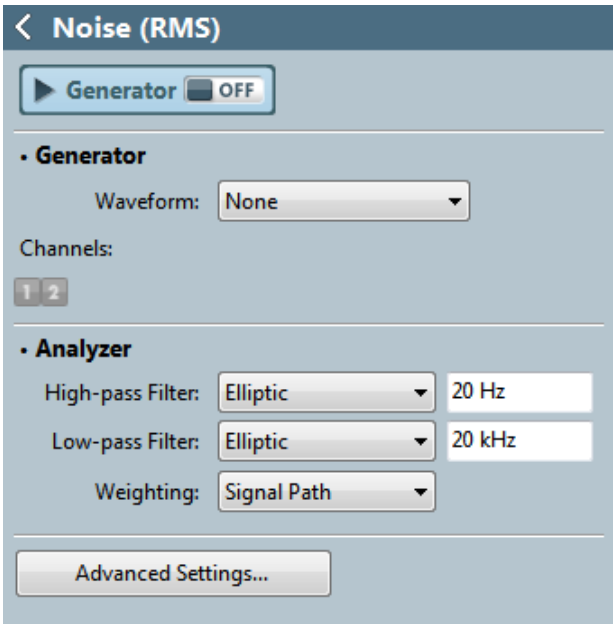
To test using a noise stimulus, select **Noise** or **Browse for File**, and connect the generator outputs to the DUT inputs.

Generator

If you choose to make a noise measurement with a stimulus signal, these signals are available. When **Waveform** is set to **Noise**, these choices will be available from the **Noise Shape** selector:

- White
- Pink
- IEC 60268-1
- BS EN 50332-1

See **More About Noise Signals** on page 423 for descriptions of these signals.



Alternatively, you can select a generator waveform file as the test signal, using the file browser in the Waveform drop-down menu.

See Chapter 6 for detailed information about the using the APx Generator and making Waveform, Level and Frequency settings, or for information about using External Source.

Level

Set the generator level here. Note that with Noise signals the generator level values are scaled differently; see **More About Noise Signals** on page 423.

Analyzer

This measurement can be configured to measure jitter in Signal Path Setup > Input > Measure. Read detailed information about jitter generation and measurement beginning on page 69.

Filters

Noise measurements are typically made within a limited passband. The default setting for noise level measurements uses a 20 kHz low-pass filter and a 20 Hz high-pass filter.

Local high-pass, low-pass and weighting filters are available for this measurement. The low-pass and high-pass filters, if selected, are applied to the entire measured signal and affect all the results. The weighting filter, if selected, affects only the distortion and distortion plus noise results. See Measurement Filters on page 619 for detailed information about the filters available locally for some measurements.

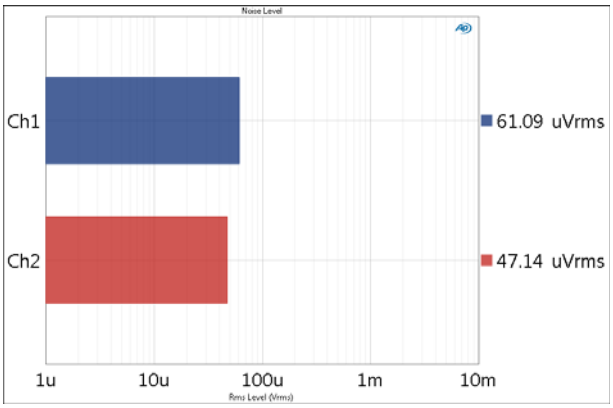
Low pass, high pass and weighting Signal Path Filters settings (see page 617) made in Signal Path Setup > Input/Output will also affect this measurement. When set, the local low-pass or high-pass filter setting overrides the corresponding Signal Path filter setting while the measurement is active. A local weighting filter setting does not override any Signal Path filter settings; the local weighting filter is applied in addition to any active Signal Path filters.

Advanced Settings

Advanced Settings makes available individual channel output level and offset controls, and input ranging and settling controls. See Advanced Settings, next.

See Chapter 104 for more information about units of measurement.

Noise (RMS) Level



Noise (RMS) Level provides a single-value meter result, displaying the rms noise level from each DUT channel, as measured at each analyzer input.

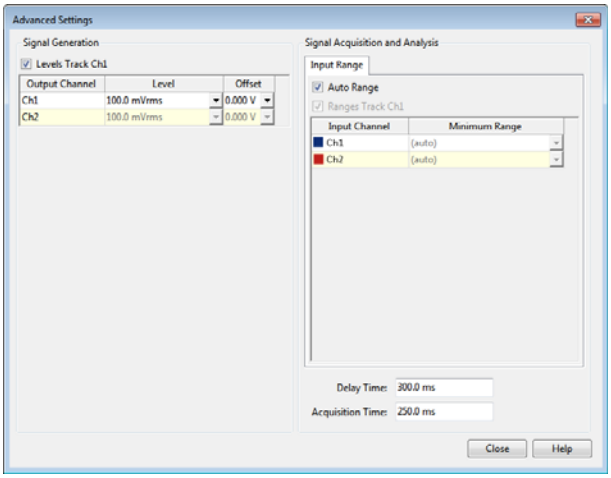
Units

Units available for Noise (RMS) Level are

Analog Signals	*Digital Signals*	*Jitter signals*
• Vrms	• dBFS	• UI
• dBV	• FS	• dBUI
• dBu	• %FS	• s
• dBrA	• dBrA	
• dBrB	• dBrB	
• dBSPL1	• dBSPL1	
• dBSPL2	• dBSPL2	
• dBm		
• W (watts)		

Advanced Settings for Noise Measurement

The default settings here are appropriate for most measurements. You may want to make minor adjustments for special situations.



Signal Generation

If Waveform is set to None, no signal generation settings will be available.

You can set different output levels or DC offsets for each channel here.

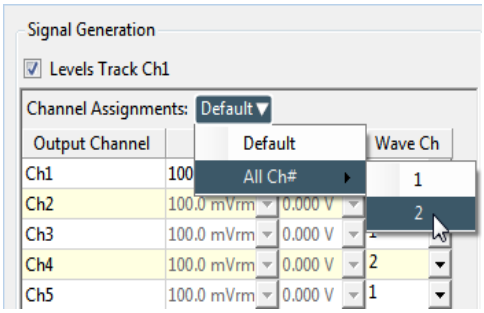
If **Levels Track Ch1** is checked (the default), the generator output level and DC offset values for channel 1 are copied to the other channels, and values for channels greater than 1 cannot be edited. Any changes made to channel 1 are reproduced in the other channels.

To set individual generator output channel levels or offsets, uncheck the **Levels Track Ch1** checkbox and enter values in the output channel Level or Offset fields.

Set Channel Assignments for generator waveforms

For stereo or multichannel generator waveform files, you can set channel mapping. By default, channel 1

audio is mapped to channel 1 output, channel 2 to channel 2, and so on. If the number of channels in the waveform file is less than the number of output channels, the waveform channels resume numbering at 1 and wrap to the next available output channel. These assignments can be remapped by changing individual settings in the **Wave Ch** column, or by selection one of several presets from the **Set Channel Assignments menu.**



Signal Acquisition and Analysis

Input Range tab

Digital inputs do not require ranging, and the Input Range tab is unavailable when inputs are set to a digital format.

Auto Range

If **Auto Range** is checked (the default), each analog channel determines its input range automatically, based on the level of the input signal. If the input signal level changes beyond a ranging threshold, **Auto Range** will cause the input ranging circuits to move up or down for proper ranging.

Fixed Range

Alternatively, you can set a fixed minimum range for the analog inputs. Uncheck **Auto Range** and enter the expected maximum level in the **Minimum Range** filed. The software will choose the lowest range that will accommodate that level; this becomes the minimum range available, and the analyzer will not reset to a lower range. Fixed range is best for signals that vary rapidly, including tone bursts, perceptual audio speech samples, and program material such as voice or music. See **When autoranging isn't suitable** on page 626.

If a signal greater then the **Minimum Range** is presented to the analog input, the ranging circuits will range up to accommodate the higher level, and then drop again when the signal level falls. The range will not be set below the **Minimum Range**.

Ranges Track Ch1

If **Ranges Track Ch1** is checked (the default), the fixed input range setting for channel 1 is copied to the other channels, and range settings for channels greater than 1 cannot be edited. Any changes made to channel 1 are reproduced in the other channels. If **Ranges Track Ch1** is not checked, the fixed input range setting for each channel can be set independently.

Read more about ranging in Chapter 98.

Delay Time

When the noise measurement is run from the Navigator or as part of a sequence, the acquisition is delayed for the time set here. This pause can avoid signal disruptions that may occur in the DUT when changing measurements. The Delay Time range is 0.0 s to 10.0 s; the default setting is 300.0 ms.

Acquisition Time

Since noise is aperiodic, a very short acquisition may display a result that is well above or below the average noise level. In the Noise: RMS Level measurement the signal is acquired over a period of time between 100 ms and 10 s, set in Acquisition Time. Longer acquisitions provide more repeatable results, but update more slowly. The default acquisition time is 250 ms.

More About Noise Signals

The Noise signals in APx500 are generated in DSP. They have the following characteristics:

White noise

White noise has a power spectrum ideally constant per unit bandwidth from just above DC to half the sampling rate. The crest factor is between 3 and 5, and the probability distribution is close to Gaussian. The generation period is 2^{32-1} samples long, which is about a day at 48 kHz.

Pink noise

Pink noise is generated from white noise using a filter, and inherits the statistics of the white noise. Pink noise has a power spectrum ideally constant per fractional bandwidth from just above DC to half the sampling rate. The signal power drops off below 10 Hz. The ideal pink power spectrum is maintained ±1.0 dB in the range $2x10^{-4}$ SR to 0.45 SR.

IEC 60268-1 noise

Noise according to the IEC 60268-1 standard is generated from pink noise passed through a weighting filter. IEC 60268-1 has a spectrum that mimics program material, including voice and music.

BS EN 50332-1 noise

BS EN 50332-1 is IEC 60268-1 noise with a reduced crest factor, accomplished by applying soft clipping. The crest factor is between 1.8 and 2.2. BS EN 50332-1 noise supports headphone testing.

Setting Generator levels when using noise signals

The RMS units assume a sine wave, which has a crest factor of the square root of 2 (approximately 1.414). None of the noise signals have this crest factor, so actual noise values will not be equal to the generator level in RMS-based units.

White noise is generated using a method that guarantees that the peak value is equal to the generator level in peak units; Vp and Vpp will accurately specify the noise level. Note that for digital outputs there is no generator level peak unit.

The filters used in pink noise, IEC 60268-1 noise and BS EN 50332-1 noise alter the signal so that the maximum output cannot be guaranteed in the same way that the white noise can. Signal levels for these noise types will not be equal to the generator level setting for any units, but will be bounded by the Vp setting.

Noise Recorder (RMS)
(Sequence Mode)

Introduction

For the APx515, this measurement requires a software option key. See page 189 for more information about software options.

Noise Recorder (RMS) is a diagnostic tool that provides a record of an RMS noise measurement versus elapsed time. The record can be very long (up to one week), providing a means of monitoring the output of a device under test over an extended period of time. Typically, the Noise Recorder is used without a stimulus signal (Waveform: None), but a Noise signal or Generator Waveform files are also available.

The Noise Recorder result available in APx500 is:

- RMS Level

Recording an audio file to disk

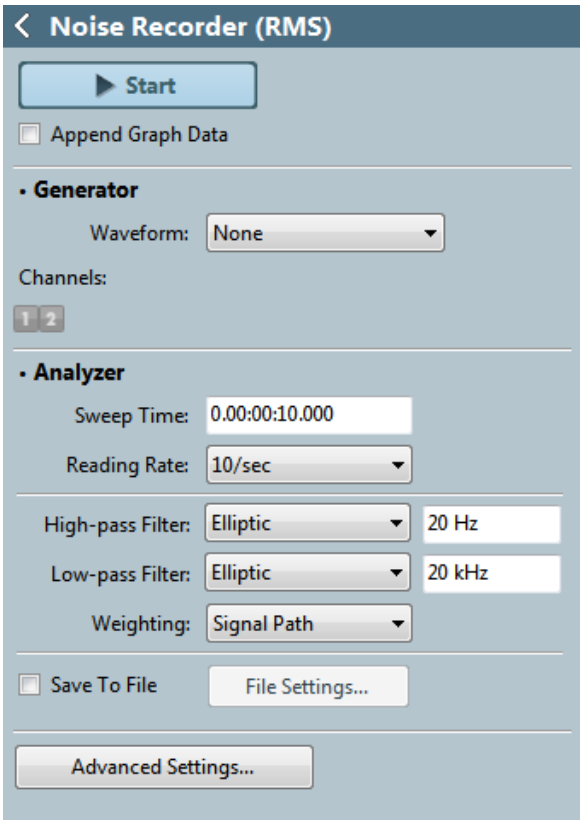
The Noise Recorder also provides a means of recording the audio acquisition to a file. See **Recording Audio to a File** on page 381 for more information.

Preparing for a Noise Recorder measurement

If you have not yet set up your test, first go to Signal Path Setup (Chapter 7).

To measure the residual noise in a DUT, select **None** as the waveform, which will run the measurement with no stimulus signal. We recommend disconnecting any devices from the DUT inputs (including the APx generator, if connected), and terminating the DUT inputs with a low resistance, typically 0 ohms to a few hundred ohms.

To test using a noise stimulus, select **Noise** or **Browse for File**, and connect the generator outputs to the DUT inputs.



Operation

When Waveform is set to **None**, no stimulus is generated. Click **Start** and the RMS level of the noise in the DUT output is recorded.

When Waveform (below) is set to **Noise** or **File**, click **Start**. The generator will output the stimulus signal to the DUT on the selected generator channels at the level set in the Signal Generation panel. The RMS level of the noise in the DUT output is recorded.

Append Graph Data

Normally, the current graph data for all results is deleted each time you start a new measurement. If the **Append** box is checked, the current data is kept in

memory as a **Data Set** (see page 189), and the new measurement data is appended as a new data set. You can append many **Data Sets**.

Generator

Typically, noise measurements are made with no stimulus signal. Once connections to the DUT outputs are made, the noise level is measured and displayed.

However, signal generation is available.

When **Waveform** is set to **Noise**, these choices will be available from the **Noise Shape** selector:

- White
- Pink
- IEC 60268-1
- BS EN 50332-1

See **More About Noise Signals** on page 423 for descriptions of these signals.

Alternatively, you can select a generator waveform file as the test signal, using the file browser in the Waveform drop-down menu. See Generator Waveforms on page 186.

Level

Set the generator level here. Note that with Noise signals the generator level value are scaled differently; see **More About Noise Signals** on page 423.

Analyzer

Sweep Time

Sweep Time sets the length of the measurement record. Minimum time is 0.1 s, maximum time is 7 days.

Enter the **Sweep Time** following this pattern:

d:hh:mm:ss.s

where d = days, hh = hours, mm = minutes, ss = seconds. Days and fractional seconds are optional.

Reading Rate

Reading Rate sets a maximum number of times the input signal will be read per second. In most cases the number of readings will be very close to this value. However, the rate is dependent upon PC performance and channel count, and may be lower. See More About Reading Rate on page 383.

Filters

Low-pass, weighting and high-pass filters are available for this measurement. In the case of the Noise Recorder, all three filters (if applied) act on the RMS Level result.

Noise measurements are typically made within a limited passband. The default setting for noise level measurements uses a 20 kHz low-pass filter and a 20 Hz high-pass filter.

Low pass, high pass and weighting Signal Path Filters settings (see page 617) made in Signal Path Setup > Input/Output will also affect this measurement. When set, the local low-pass or high-pass filter setting overrides the corresponding Signal Path filter setting while the measurement is active. A local weighting filter setting does not override any Signal Path filter settings; the local weighting filter is applied in addition to any active Signal Path filters.

Save to File

Click this option to save the acquired audio to a file on disk. See Recording Audio to a File on page 381.

Running the measurement in File Input configuration

See Analyzing Audio Files in Chapter 22 for more information about File Input configuration.

If you have chosen **File** for an input configuration in Signal Path Setup, first click the **File List...** button to add input file(s) to a list for analysis.

If your DUT is to be recording the WAV file, select a generator waveform and turn the generator **ON**, as described above. Configure your DUT to save a WAV file(s) whose file name(s) and location match the name specified in the File List.

If you are measuring an existing file (not recording from the generator but working in External Source), be sure that the file name(s) and location match the name(s) specified in the File List.

Finally, click the **Analyze** button. The first settled reading (typically within the first 0.10 second of the file) is shown on the meter bar display. To take a reading later in the file, go to the **Settling** tab in **Advanced Settings** and increase the **Delay Time** to the desired point in the file.

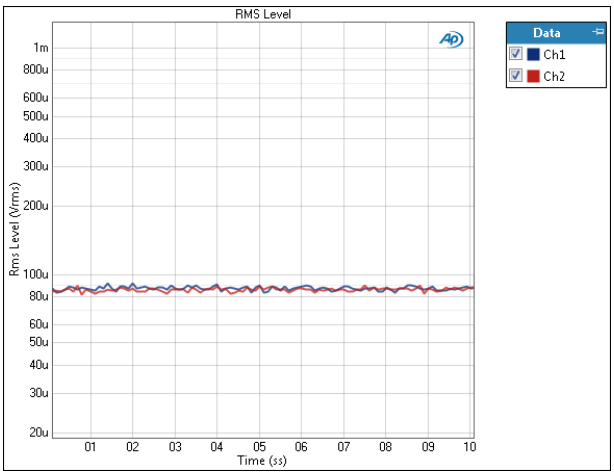
Advanced Settings

If your test requires individual generator settings for each channel or special analyzer ranging or settling, see Advanced Settings for the Noise Recorder on page 427.

Noise Recorder: RMS Level

The Measurement Recorder: Level measurement result will provide a record of RMS level versus elapsed time for any audio signal. The illustration

shows a 10-second record of a noise signal on two channels.



Units

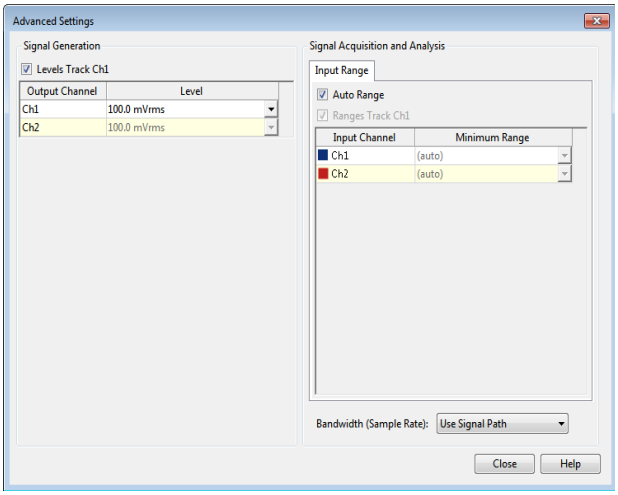
Units available for Measurement Recorder: Level are:

X-axis	*Y-axis (analog)*	*Y-axis (digital)*
• s	• Vrms	• dBFS
	• dBV	• FS
	• dBu	• %FS
	• dBrA	• dBrA
	• dBrB	• dBrB
	• dBSPL1	• dBSPL1
	• dBSPL2	• dBSPL2
	• dBm	
	• W (watts)	

See Chapter 104 for more information about units of measurement.

Advanced Settings for Noise Recorder

The default settings here are appropriate for most measurements. You may want to make minor adjustments for special situations.



Signal Generation

If Waveform is set to None, no signal generation settings will be available.

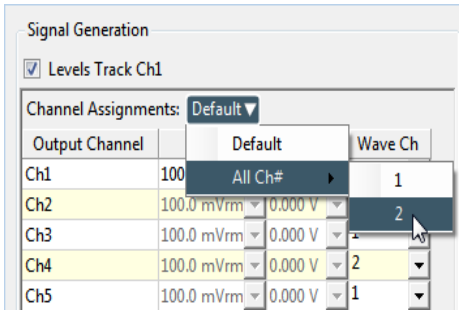
You can set different output levels or DC offsets for each channel here.

If **Levels Track Ch1** is checked (the default), the generator output level and DC offset values for channel 1 are copied to the other channels, and values for channels greater than 1 cannot be edited. Any changes made to channel 1 are reproduced in the other channels.

To set individual generator output channel levels or offsets, uncheck the **Levels Track Ch1** checkbox and enter values in the output channel Level or Offset fields.

Set Channel Assignments for generator waveforms

For stereo or multichannel generator waveform files, you can set channel mapping. By default, channel 1 audio is mapped to channel 1 output, channel 2 to channel 2, and so on. If the number of channels in the waveform file is less than the number of output channels, the waveform channels resume numbering at 1 and wrap to the next available output channel. These assignments can be remapped by changing individual settings in the **Wave Ch** column, or by selection one of several presets from the **Set Channel Assignments menu.**



Signal Acquisition and Analysis

Input Range tab

Digital inputs do not require ranging, and the Input Range tab is unavailable when inputs are set to a digital format.

Auto Range

If **Auto Range** is checked (the default), each analog channel determines its input range automatically, based on the level of the input signal. If the input signal level changes beyond a ranging threshold, **Auto Range** will cause the input ranging circuits to move up or down for proper ranging.

Fixed Range

Alternatively, you can set a fixed minimum range for the analog inputs. Uncheck **Auto Range** and enter the expected maximum level in the **Minimum Range** filed. The software will choose the lowest range that will accommodate that level; this becomes the minimum range available, and the analyzer will not reset to a lower range. Fixed range is best for signals that vary rapidly, including tone bursts, perceptual audio speech samples, and program material such as voice or music. See **When autoranging isn't suitable** on page 626.

If a signal greater then the **Minimum Range** is presented to the analog input, the ranging circuits will range up to accommodate the higher level, and then drop again when the signal level falls. The range will not be set below the **Minimum Range**.

Ranges Track Ch1

If **Ranges Track Ch1** is checked (the default), the fixed input range setting for channel 1 is copied to the other channels, and range settings for channels greater than 1 cannot be edited. Any changes made to channel 1 are reproduced in the other channels. If **Ranges Track Ch1** is not checked, the fixed input range setting for each channel can be set independently.

Read more about ranging in Chapter 98.

Bandwidth (Sample Rate)

This control is only available when the analyzer input configuration is set to Analog.

The Bandwidth (Sample Rate) control allows you to override the analog input bandwidth setting in Signal Path Setup. The input bandwidth set here operates only on the Noise Recorder.

Certain applications that may use the files created here require files of specific sample rate. For analog inputs, the default sample rate of a recorded file is determined by the input bandwidth setting, normally set in Signal Path Setup. This control allows selection of other bandwidths/sample rates. The analog input bandwidth (sample rate) settings, whether made here or in Signal Path Setup, result in channel-count constraints at high sample rates.

The bandwidth choices, followed by their corresponding sample rates, are:

- 2.75 kHz (6 kS/s)
- 3.5 kHz (8 kS/s)
- 5.5 kHz (12 kS/s)
- 7 kHz (16 kS/s)
- 11 kHz (24 kS/s)
- 20 kHz (44.1 kS/s)
- 22 kHz (48 kS/s)
- 40 kHz (88.2 kS/s)
- 45 kHz (96 kS/s)
- 80 kHz (176.4 kS/s)
- 90 kHz (192 kS/s)
- 250 kHz (624 kS/s) (requires the APx555 or the BW52 Option)
- 500 kHz (1.248 MS/s) (requires the APx555 or the BW52 Option)
- 1 MHz (2.496 MS/s) (requires the APx555 or the BW52 Option, set to 1 channel)
- Use Signal Path (the default)

Note that recording files at bandwidths above 90 kHz (sample rates above 192 kHz) requires a very fast PC and hard disk, due to the high data rates. Slower systems can suffer recording interruptions and failure.

Pass/Fail
(Sequence Mode)

Pass/Fail is not a measurement, but a Navigator node designed to provide an opportunity for the operator or some external device to insert a **Pass** or **Fail** result in a sequence, unrelated to APx audio measurement results. The **Pass/Fail** result is included in the Report.

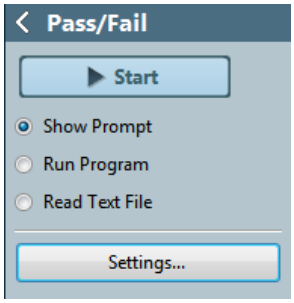
Whether or not this step presents a **Pass, Fail** or **Cancel** result, the Sequence continues.

Pass/Fail is only useful in an automated sequence.

Methods of Response

There are 4 methods of response to the Pass/Fail step in a sequence.

1. The Operator responds to a condition (an indicator light in the DUT, for example) by clicking the **Pass**, **Fail** or **Cancel** button on the prompt displayed on the computer screen.
2. An **Aux Control In** condition is met (a specific pin goes high, for example) for **Pass**, **Fail** or **Cancel**. This could be in response to the operator pushing a physical button, or could be a signal from another device involved in the test. A temperature-sensing device, for example, could send an **Aux Control In Fail** signal if the DUT temperature were out of range. **Aux Control In** conditions for **Pass**, **Fail** or **Cancel** can be set in **Project/Sequence Properties > Dialog Button Aux Control** (see page 537).
3. The step calls a program, which runs to produce a text string. If the resulting text string matches the expected result, the step reports **Pass**; in any other case (wrong result, illegal path, etc.) the step reports **Fail**.
4. The step opens a text file that contains a text string. If the text string matches the expected result, the step reports **Pass**; in any other case (wrong text, illegal path, etc.) the step reports **Fail**.



Operation

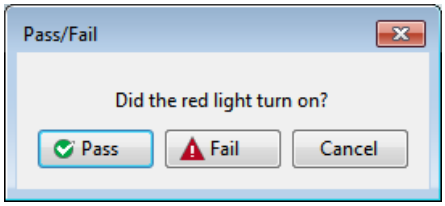
Open **Pass/Fail** and place a check mark on its node to enter it as a sequence step. Move the node to the desired position in the sequence.

Choose the method of response

- Show Prompt
- **Run Program**, or
- Read Text File

Click **Settings** to configure the response.

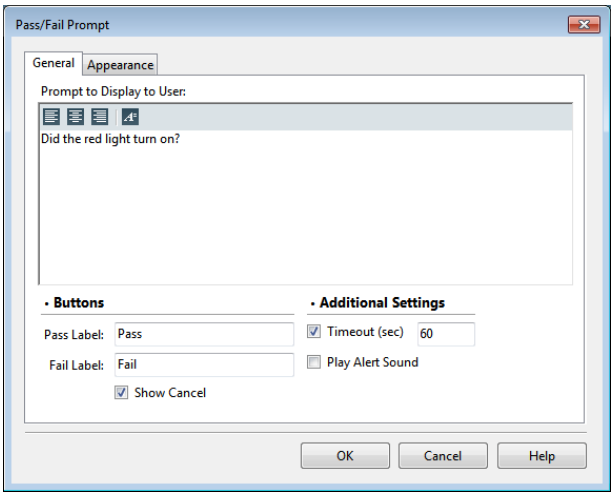
Show Prompt settings



Example prompt

General Tab

Prompt to Display to User



Add any text you want to appear in the prompt here. You can also paste formatted text from the Windows clipboard into the text entry field. Long text can be scrolled in the prompt.

The toolbar at the top of the text entry area provides left, center and right alignment buttons. The font style button opens a dialog where you can select a new font, change point size, color, weight, and so on.

Pass Button

By default, the **Pass** button is labeled “Pass”. You can change the button label by setting the label checkbox and entering new text in the label field.

Fail Button

By default, the **Fail** button is labeled “Fail”. You can change the button label by setting the label checkbox and entering new text in the label field.

Show Cancel

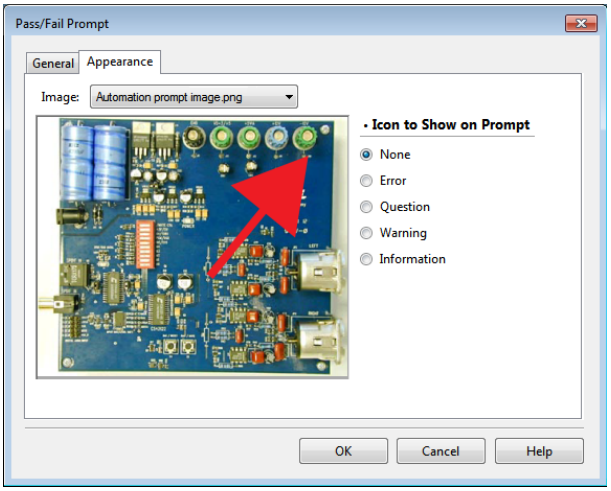
By default, there in no **Cancel** button in the prompt. If you would like to add a **Cancel** button, set the checkbox here.

Close Dialogs using Aux Control

The operator can use **Aux Control In** commands to close the prompts dialogs during production test, enabling the use of physical devices rather than clicking on the computer screen. To configure **Aux Control In** to respond, right-click on the Project node to open the **Project/Sequence Properties** dialog (page 537).

Appearance tab

Image



You can place an image at the top of the user prompt. This could be a photograph or diagram with additional visual information to help the user.

Beneath the Image area, click the **Browse...** button and navigate to an image file (page 573). Select the image and click **Open** to place the image in the prompt. When an image is used, the image file is also attached to the project file.

Larger images will cause the prompt window size to increase to fill the available screen space, and will be scaled to fit in the prompt window.

Click **Delete** to remove the image from the prompt. Image files attached to the project but not currently in use can be removed by navigating to the dialog at **File > Manage Attached Project Items...** (see page 27).

Icon to Show on Prompt

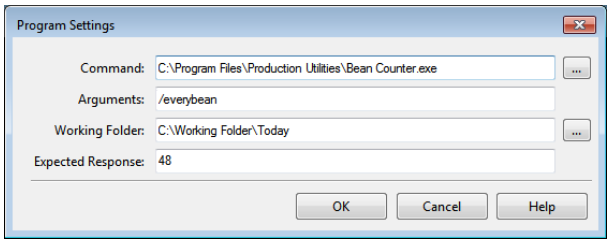
Check one of the following (or **None**) to place a standard Windows attention icon on the prompt.

- None
- **Error**
- **Question**
- **Warning**
- **Information**

Run Program

When configured to **Run Program**, the **Pass/Fail** step calls a program, which runs to produce a text string. If the resulting text string matches the expected result,

the step reports **Pass**; in any other case (wrong result, illegal path, etc.) the step reports **Fail**.



Command:

Enter in the path and command for the external program here, or browse to the program using the browse button.

Arguments:

Enter in the command arguments here, if any.

Working Folder:

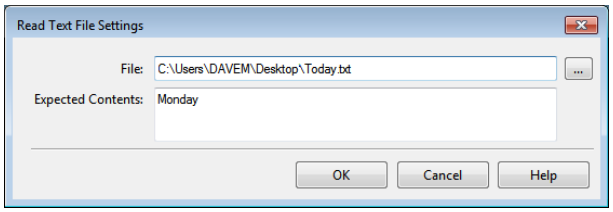
Enter in the working folder here, if necessary, or browse to the folder using the browse button.

Expected Response

Enter the exact text you expect as a response.

Read Text File

When configured to **Read Text File**, the **Pass/Fail** step opens a text file. If the resulting text string in the file matches the expected result, the step reports **Pass**; in any other case (wrong text string, illegal path, etc.) the step reports **Fail**.



File:

Enter in the path and filename for the text file here, or browse to the file using the browse button.

Expected Contents

Enter the exact text you expect to be in the file.

PESQ
(Sequence Mode)

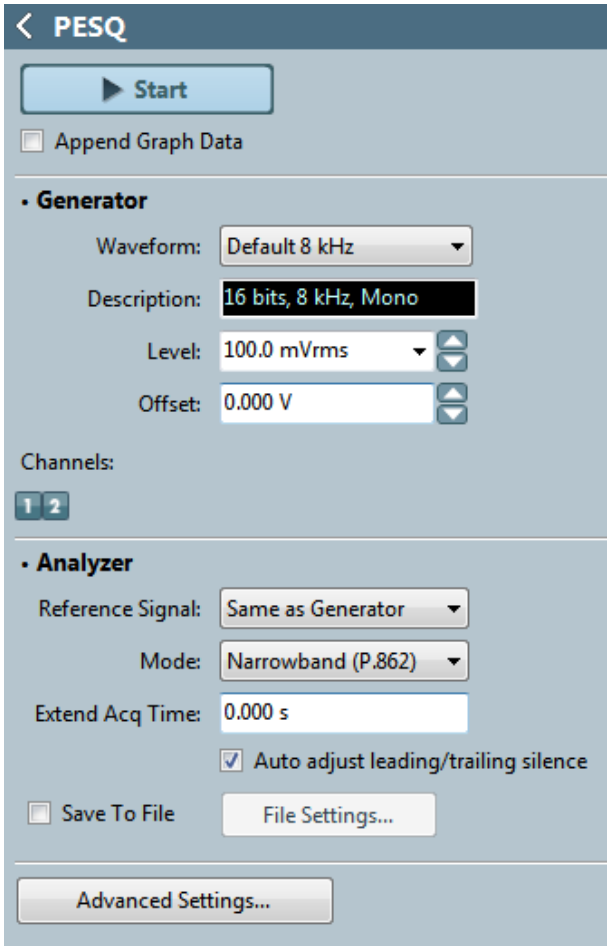
This measurement requires a software option key. See page 189 for more information about software options.

PESQ (Perceptual Evaluation of Speech Quality) is a method of grading the quality of an audio communications channel, resulting in a MOS (Mean Opinion Score). See page 438 for more information about PESQ.

The PESQ measurement uses a speech sample stimulus signal that is passed through the DUT and compared with the original speech signal serving as a reference. A MOS result and a number of diagnostic cause analysis results are provided.

The PESQ measurement results available in APx500 are:

- MOS (P.862.1)
- PESQ
- MOS vs Time
- Average Delay
- Delay vs Time
- Acquired Waveform
- Reference Waveform



Operation

If you have not yet set up your test, first go to Chapter 7 “Signal Path Setup.” When Generator and Analyzer settings (below) are set for your test, click **Start**.

Append Graph Data

Normally, the current graph data for all results is deleted each time you start a new measurement. If the **Append** box is checked, the current data is kept in memory as a **Data Set** (see page 189), and the new measurement data is appended as a new data set. You can append many **Data Sets**.

Generator

Waveform

For the PESQ measurement, the APx generator can play a speech sample that conforms to PESQ requirements. APx provides one speech sample at a sample rate of 8 kHz and one at 16 kHz embedded in the PESQ measurement. Select one of these, or browse for a waveform file. An additional 32 speech samples are installed as .WAV waveforms during the installation of APx500.

Conforming waveforms are mono WAV files at a sample rate of 8 kHz or 16 kHz. Content should be speech

samples. Length is typically 5 to 10 seconds; maximum length is 35 seconds.

Description

This field displays information about the current waveform.

Level

Set the generator output level in this field.

The Level control sets the level of the generator output to the value entered in the Level field, for a maximum level in the waveform file. See Generator Waveform Files on page 47 for a general discussion of generator level when playing waveform files.

Channel

The generator will output the speech samples to the DUT on the selected generator channels.

Start

The Start button runs the measurement. For closed loop operation, Start outputs the speech sample signal and prepares the analyzer for acquisition. For External Source (open loop) measurements, Start prepares the analyzer for acquisition, awaiting a speech sample from the DUT.

Using Output EQ with PESQ waveforms

See Output EQ on page 193.

If you are testing with complex waveforms (such as speech waveforms for PESQ) and find that the generator limiting feature of Output EQ reduces the level so far that you cannot attain the necessary Active Speech Level at the transducer, an alternative method is to use the APx EQ .wav Files Utility. Read about the utility by searching for the file “APx EQ .wav Files Utility Instructions.pdf” at ap.com.

Analyzer

Filters

There are no local filters available for this measurement; the global filters in Signal Path Setup > Input/Output will affect this measurement, if set. Read more about APx Filters in Chapter 97.

Reference Signal

in Closed Loop configuration...

In closed loop configuration, the **Reference Signal** is usually the **Generator Waveform** that is currently selected.

In situations where you have externally modified or EQ’d a Reference Waveform (equalized to compensate for loudspeaker response, for example), you have the option to select an alternative **Reference Signal** (the unmodified reference waveform, for example) here.

in External Source configuration...

When in External Source (open loop) configuration, you must choose a reference speech sample file here. The speech sample file (or compensated speech sample file) being played from the DUT must correspond to the **Reference Signal** chosen here.

Mode

Conforming PESQ speech sample files have an 8 kHz sample rate for testing narrowband channels, and a 16 kHz sample rate for testing wideband channels. The **Mode** setting is forced to **Wideband** for 16 kHz sample rate reference files. For 8 kHz sample rate reference files, the **Mode** should be set to **Narrowband**. In mixed narrowband/wideband environments, set the **Mode** selector to **Wideband**, even if the reference file has a sampling rate of 8 kHz.

Extend Acq. Time

The acquisition time is set to the length of the speech sample (selected either in Waveform, for closed loop, or **Reference Signal** for External Source (open loop), plus the time set in this field. If you suspect you are getting poor results because the acquisition length is truncating the input signal, add time here.

This field is not available if **Input Configuration** is set to **File**. In **File Input**, the acquisition time is set to the length of the input file. In either case, the maximum acquisition length is 35 seconds.

Auto adjust leading/trailing silence

The PESQ implementation will automatically adjust the length of the leading and trailing silence of a speech sample to ensure that the length conforms to the ITU recommendations. When this checkbox is checked (the default), the automatic length adjustment is ON.

For PESQ conformance testing, switch this feature OFF to remain backward compatible.

Save to File

You can record the PESQ audio signal to a file as the signal is being acquired. Click **Save to File** before you run the PESQ measurement. See Recording Audio to a File on page 381 for more information.

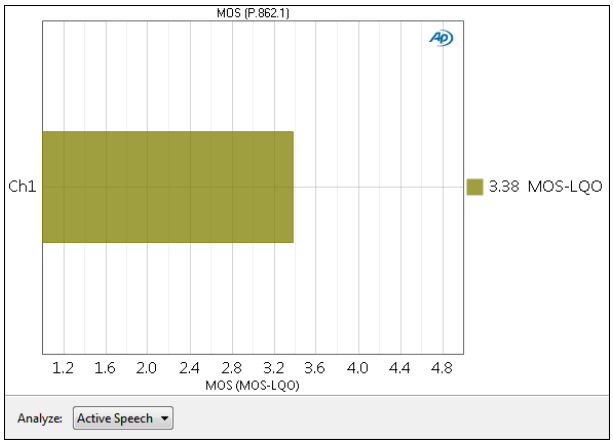
File Settings

When saving a PESQ acquisition to a file, click **File Settings** to specify the file name and location.

Advanced Settings

If you’d like to adjust analyzer ranging parameters, click **Advanced Settings**. See Advanced Settings for PESQ on page 437. Read more about Units in Chapter 104.

PESQ: MOS (P.862.1) result



The **PESQ: MOS** result is the primary result for the PESQ measurement. The PESQ score is mapped to MOS-LQO using P.862.1 for narrowband analysis, or P.862.2 for wideband analysis.

Additional Control for MOS result

Analyze

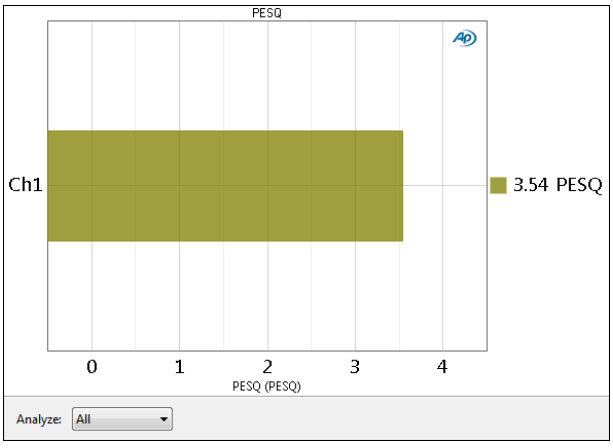
- **All**
  Analyzes the entire speech sample. This mode is supported by the ITU-T P.862 standard for MOS results.
- **Active Speech**
  Analyzes only the speech portions of the sample.
- **Silence**
  Analyzes only the relative silence between the speech portions.

Units

Units for PESQ: MOS are:

- MOS-LQO

PESQ: PESQ result



The **PESQ: PESQ** result shows the PESQ score for narrowband analysis. The PESQ score should be treated as an intermediate result, and the MOS scores reported instead.

Additional Control for PESQ result

Analyze

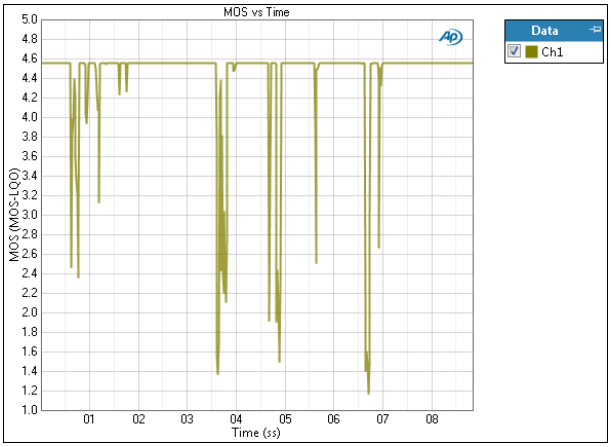
- **All**
  Analyzes the entire speech sample. This mode is supported by the ITU-T P.862 standard for MOS results.
- **Active Speech**
  Analyzes only the speech portions of the sample.
- **Silence**
  Analyzes only the relative silence between the speech portions.

Units

Units for PESQ: PESQ results are:

- PESQ

PESQ: MOS vs Time result



The PESQ: MOS vs Time is useful for cause analysis. The result shows a graph of the MOS score versus elapsed time.

Units

Units for PESQ: Delay vs Time are:

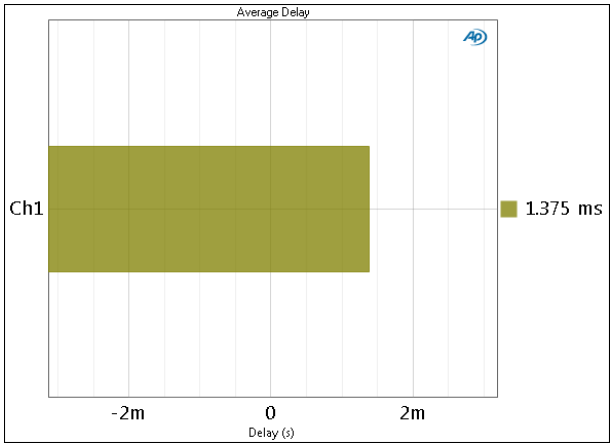
X-axis

- s (seconds)

Y-axis

- MOS-LQO

PESQ: Average Delay result



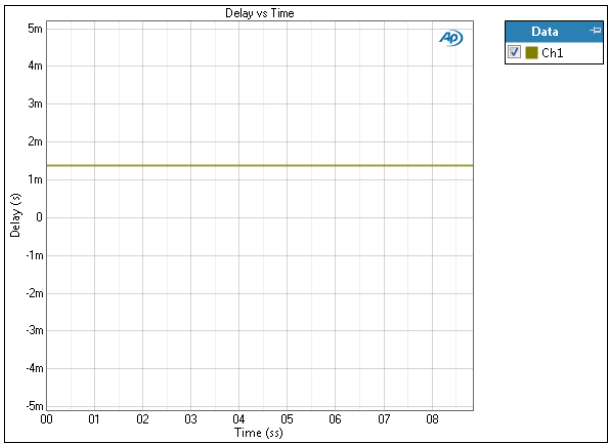
The PESQ: Average Delay result is useful for cause analysis. The result shows the average signal delay through the DUT or system.

Units

Units for PESQ: Average Delay are:

- s (seconds)

PESQ: Delay vs Time result



The PESQ: Delay vs Time result is useful for cause analysis. The result shows signal delay time versus elapsed time.

Units

Units for PESQ: Delay vs Time are:

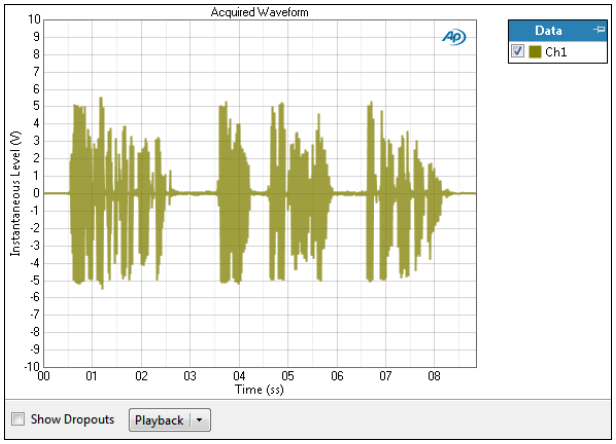
X-axis

- s (seconds)

Y-axis

- s (seconds)

PESQ: Acquired Waveform result



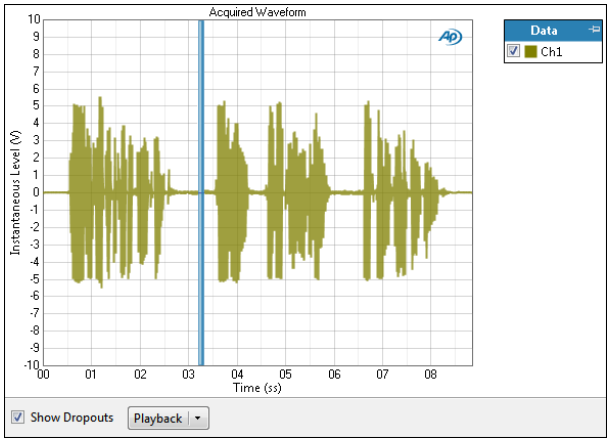
The PESQ: Acquired Waveform result is useful for cause analysis. The result shows a time-domain view of the acquired speech sample waveform.

Additional Control for Acquired Waveform result

Show Dropouts

APx compares the acquired waveform to the reference waveform. A signal dropout is defined as a location where there is signal in the reference waveform but no signal in the acquired waveform.

When the Show Dropouts checkbox is checked (the default), APx highlights dropout locations on the graph.



Playback

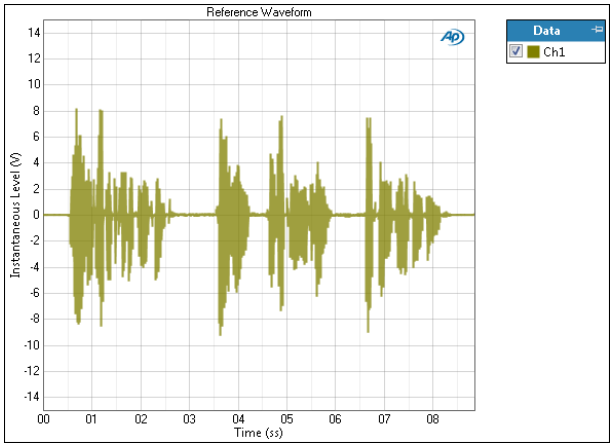
You can play the PESQ measurement Acquired Waveform through the PC Windows audio. Click the Playback menu to list the available acquired channels, and play the desired channel by selecting it.

Units

Units available for PESQ: Acquired Waveform are:

X-axis	*Y-axis (analog)*	*Y-axis (digital)*
• s (seconds)	• V	• D • hex

PESQ: Reference Waveform result



The PESQ: Reference Waveform result is result is useful for cause analysis. The result shows a time-domain view of the reference speech sample waveform.

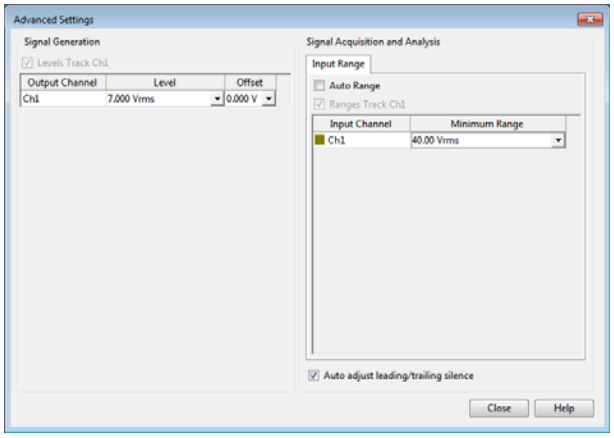
Units

Units for PESQ: Reference Waveform are:

X-axis	*Y-axis (analog)*	*Y-axis (digital)*
• s (seconds)	• V	• D • hex

Advanced Settings for PESQ

The default settings here are appropriate for meter (single value) measurements under most conditions. You may want to make minor adjustments for special situations.



Signal Generation

If **Levels Track Ch1** is checked (the default), the generator output level value for channel 1 is copied to the other channels, and output levels for channels greater than 1 cannot be edited. Any changes made to channel 1 are reproduced in the other channels.

To set individual generator output channel levels, uncheck the **Levels Track Ch1** checkbox and enter values in the output channel Level fields.

Signal Acquisition and Analysis

Input Range tab

> *Digital inputs do not require ranging, and the Input Range tab is unavailable when inputs are set to a digital format.*

Auto Range

If **Auto Range** is checked (the default), each analog channel determines its input range automatically, based on the level of the input signal. If the input signal level changes beyond a ranging threshold, **Auto Range** will cause the input ranging circuits to move up or down for proper ranging. You will hear relays clicking when the ranging changes.

Fixed Range

Alternatively, you can set a fixed minimum range for the analog inputs. Uncheck **Auto Range** and enter the expected maximum level in the **Minimum Range** filed. The software will choose the lowest range that will accommodate that level; this becomes the minimum range available, and the analyzer will not reset to a lower range. Fixed range is best for signals that vary rapidly, including tone bursts, perceptual audio speech samples, and program material such as voice or music.

If a signal greater then the **Minimum Range** is presented to the analog input, the ranging circuits will range up to accommodate the higher level, and then drop again when the signal level falls. The range will not be set below the **Minimum Range**.

Ranges Track Ch1

If **Ranges Track Ch1** is checked (the default), the fixed input range setting for channel 1 is copied to the other channels, and range settings for channels greater than 1 cannot be edited. Any changes made to channel 1 are reproduced in the other channels. If **Ranges Track Ch1** is not checked, the fixed input range setting for each channel can be set independently.

Read more about ranging in Chapter 98.

Auto Range and PESQ

PESQ and other perceptual audio measurements use voice samples as test signals, and operate best when ranging is fixed. See **When autoranging isn't suitable** on page 626.

More about PESQ and perceptual audio testing

Overview

PESQ (Perceptual Evaluation of Speech Quality), like POLQA, is a method of grading the quality of an audio communications channel. Perceptual audio evaluation methods were developed to enable evaluation of audio quality in channels of compromised performance, where conventional measurements such as THD+N may provide results that are not useful.

Although still in wide use, PESQ has been succeeded by POLQA, which avoids certain weaknesses in PESQ, and is extended towards higher bandwidth audio signals.

Perceptual audio evaluation techniques are based on subjective experiments with listeners, and do not use simple periodic waveforms or sweeps as most APx measurements do; consequently, the perceptual evaluation results cannot be stated as physical values such as volts or hertz. Instead, listeners' opinions of quality (or algorithms designed to respond similarly) are averaged and scored.

PESQ and POLQA are both methods of producing MOS results without actually using a room full of listeners. Instead, pre-recorded speech samples are passed through a DUT or system and are compared with the original speech sample. A sophisticated algorithm evaluates the difference between the reference audio and the degraded audio, and produces a result.

Channels of compromised audio quality

Many speech communications channels have limited audio capabilities, when compared to full fidelity channels optimized for music transmission. The limited capabilities of speech channels may provide other benefits: lower cost, lower bandwidth, lower power requirements, and so on.

Most of the I/O interfaces offered in an APx analyzer have very high fidelity capabilities, but in some cases the configuration for a low data rate DUT may make PESQ or POLQA-type evaluations valuable. Passing a stimulus signal through a telephone network, for example, will likely degrade the signal substantially, as will low data rate codecs. The Bluetooth HSP and HFP profiles, in particular, are of moderate fidelity. Bluetooth manufacturers often use PESQ or POLQA to characterize Bluetooth devices operating in these profiles.

MOS (Mean Opinion Score)

Perceptual audio evaluations are typically made with a number of listeners, who provide opinions about the quality of audio samples they have listened to. Typically, many audio samples, many listeners and many iterations are used, and the results are averaged into a MOS (Mean Opinion Score). MOS is expressed in a unitless scale from 1 to 5, representing the average opinion of quality, as shown in this table:

MOS	*Quality*	*Impairment*
5	Excellent	Imperceptible
4	Good	Perceptible but not annoying
3	Fair	Slightly annoying
2	Poor	Annoying
1	Bad	Very annoying

MOS results in APx are reported as MOS-LQO (LQO stands for Listening-only Quality, Objective). For PESQ results, these reference the PESQ-to-MOS mapping: P.862.1 for narrowband, and P.862.2 for wideband.

PESQ ITU-T P.862 (2001)

The PESQ algorithm is defined in ITU-T Recommendation P.862 (2001).

Narrowband / Wideband testing

Narrowband has a bandwidth of 300–3400 Hz and uses a filter in analysis to emulate the characteristics of a telephone handset. Digital files for PESQ narrowband have a sampling rate of 8 kHz. Narrowband PESQ results are mapped to MOS-LQON according to P.862.1.

Wideband has a bandwidth of 50–7000 Hz and uses flat filtering. Digital files for PESQ wideband have a sampling rate of 16 kHz. Wideband PESQ results are mapped to MOS-LQOW according to P.862.2. The mapping algorithm in P.862.2 is slightly different from the algorithm in P.862.1.

In either case, the PESQ result itself should be considered an intermediate result, and the MOS score the primary result.

The standard forbids mixing of wideband and narrowband results. If wideband networks have to be compared to narrowband networks, then the wideband version of PESQ has to be used in both cases.

See http://www.itu.int/rec/T-REC-P.862/ for the latest ITU standard.

The PESQ score is available for the entire signal, or for the active speech parts of the signal only, or for the silent periods only.

Speech samples

PESQ evaluates speech samples. You can record your own samples, or obtain speech sample files from other sources. The ITU has created a number of samples designed to cover the range of speech sounds uttered by men and women in a range of languages. These are available at http://www.itu.int/net/itu-t/sigdb/genaudio/Pseries.htm and http://www.itu.int/rec/T-REC-P.501-200912-S/en. See the ITU P501E documentation at the second link

for detailed information about creating and using speech samples.

For PESQ, APx500 provides one speech sample at a sample rate of 8 kHz and one at 16 kHz embedded in the measurement. An additional 32 speech samples are installed as .WAV files during the installation of APx500, at C:\Program Files (x86)\Audio Precision\APx500 [version]\Waveforms\PESQ\...

The PESQ algorithm has been extensively tested against a number of European and Asian languages and performs well. For detailed information, refer to the ITU publication ITU-T-REC-P862 3-200711.pdf.

Averaging

Averaging multiple speech samples across multiple iterations provides the most reliable MOS results. APx500 provides a PESQ (Averaged) measurement that enables you to average up to 64 speech samples or iterations.

Other results

The primary PESQ results in APx500 are the MOS (P.862.1) result and the MOS Average (P.862.1) result.

Other results are provided that are unsupported by ITU-T P.862. These results may be helpful in diagnostics and cause analysis in the DUT or system.

PESQ (Averaged)
(Sequence Mode)

This measurement requires a software option key. See page 189 for more information about software options.

PESQ (Perceptual Evaluation of Speech Quality) is a method of grading the quality of an audio communications channel, resulting in a MOS (Mean Opinion Score). See page 438 for more information about PESQ.

This measurement provides a means to average a number of PESQ measurements, producing a single MOS Average result per channel. Best practice recommends at least two male and two female voice samples, each averaged several times. Some testers will use many different voice samples, repeated and averaged many times.

The PESQ (Averaged) measurement uses a speech sample stimulus signal that is passed through the DUT and compared with the original speech signal serving as a reference. A MOS result and a number of diagnostic cause analysis results are provided.

The PESQ (Averaged) measurement is not available in External Source (open loop) configuration, or when Input Configuration is set to File.

The PESQ (Averaged) measurement results available in APx500 are:

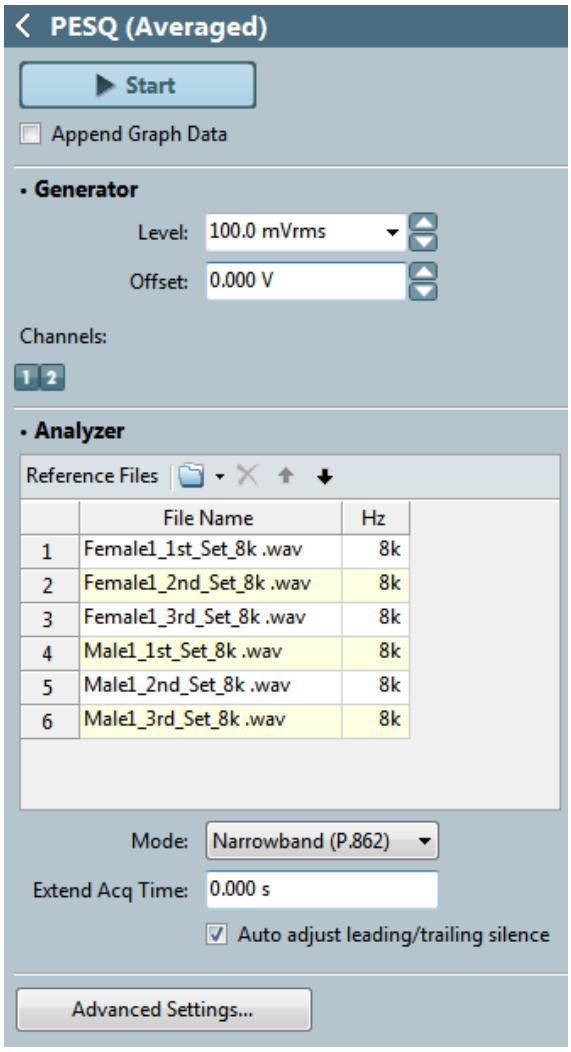
- MOS Average
- File MOS

Operation

If you have not yet set up your test, first go to Chapter 7 “Signal Path Setup.”

Start

When Signal Generation and Signal Analysis (below) are set and you have imported all your reference files, click Start. The files will be played through the DUT, one at a time.



Append Graph Data

Normally, the current graph data for all results is deleted each time you start a new measurement. If the **Append** box is checked, the current data is kept in memory as a **Data Set** (see page 189), and the new

measurement data is appended as a new data set. You can append many **Data Sets**.

Generator

Level

Set the generator output level in this field.

The Level control sets the level of the generator output to the value entered in the Level field, for a maximum level in the waveform file. See Chapter 14 for a general discussion of generator level when playing waveform files.

Generator Channels

The generator will output the speech samples to the DUT on the selected generator channels.

Reference Files

For a PESQ measurement, a reference file in the APx is played through the generator, and the output of the DUT or communications channels under test is compared to the reference file. For multiple averaging iterations with the same file or with different files, browse to enter the files in the grid here. These reference files will be played and analyzed in the order shown in the grid. The minimum number of files is 2; the maximum number of files is 64.

Audio Precision provides 32 speech samples, installed as .WAV waveforms during the installation of APx500.

Conforming waveforms are mono WAV files at a sample rate of 8 kHz or 16 kHz. Content should be speech samples. Length is typically 5 to 10 seconds; maximum length is 35 seconds.

Use the folder icon to open a file browser. You can add files already in the project, or you can browse for files on disk. If you choose files of mixed sampling rates, the analysis Mode will be constrained to Wideband.

Select a file to delete it, or to move it up or down in the list using the arrow buttons.

Analyzer

Filters

> *There are no local filters available for this measurement; the global filters in Signal Path Setup > Input/Output will affect this measurement, if set. Read more about APx Filters in Chapter 97.*

As the generator moves through the Reference Files list, the analyzer acquires the signal and matches the audio received to the reference files, one at a time. The PESQ algorithm assigns a PESQ score, which is then mapped to a MOS score. A running average of the MOS scores is displayed, with the final value being the average of the scores of all the reference files.

Mode

Conforming PESQ speech sample files have an 8 kHz sample rate for testing narrowband channels, and a 16 kHz sample rate for testing wideband channels. The Mode setting is forced to Wideband for 16 kHz sample rate reference files. For 8 kHz sample rate reference files, the Mode should be set to Narrowband. In mixed narrowband/wideband environments, set the Mode selector to Wideband, even if the reference file has a sampling rate of 8 kHz. For files of mixed sampling rates, the Mode will be constrained to Wideband.

Extend Acq. Time:

The acquisition time is set to the length of the speech sample (selected either in Waveform, for closed loop, or Reference Signal for External Source (open loop), plus the time set in this field. If you suspect you are getting poor results because the acquisition length is truncating the input signal, add time here.

Auto adjust leading/trailing silence

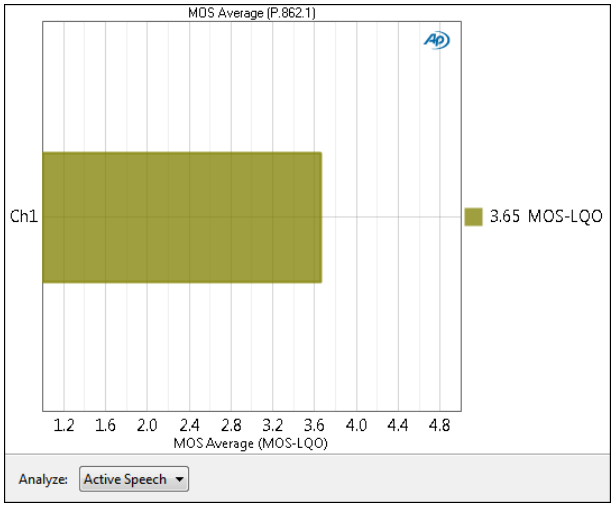
The PESQ implementation will automatically adjust the length of the leading and trailing silence of a speech sample to ensure that the length conforms to the ITU recommendations. When this checkbox is checked (the default), the automatic length adjustment is ON.

For PESQ conformance testing, switch this feature OFF to remain backward compatible.

Advanced Settings

If you'd like to adjust analyzer ranging parameters, click **Advanced Settings**. See Advanced Settings for PESQ on page 437. Read more about Units in Chapter 104.

PESQ (Averaged): MOS Average result



The PESQ: MOS Average (P.862.1) result is the primary result for the PESQ (Averaged) measurement. The PESQ score is mapped to MOS-LQO using P.862.1 for narrowband analysis, or P.862.2 for wideband analysis. This result provides an average of all the MOS scores of the speech samples in the Reference Files list.

Additional Control for MOS result

Analyze

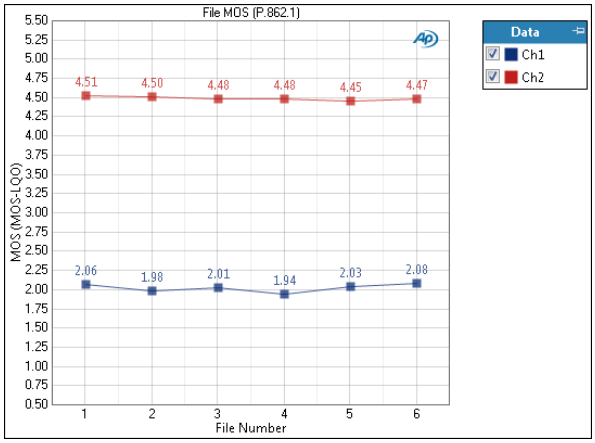
- **All**
  Analyzes the entire speech sample. This mode is supported by the ITU-T P.862 standard for MOS results.
- **Active Speech**
  Analyzes only the speech portions of the sample.
- **Silence**
  Analyzes only the relative silence between the speech portions.

Units

Units for PESQ (Averaged): MOS Average are:

- MOS-LQO

PESQ (Averaged): File MOS result



The PESQ (Averaged): File MOS (P.862.1) result is useful for cause analysis. The result shows the MOS score for each of the files in the Reference Files list.

Additional Control for File MOS result

Analyze

- **All**
  Analyzes the entire speech sample. This mode is supported by the ITU-T P.862 standard for MOS results.
- **Active Speech**
  Analyzes only the speech portions of the sample.
- **Silence**
  Analyzes only the relative silence between the speech portions.

Units

Units for PESQ (Averaged): File MOS are:

X-Axis

- File number

Y-Axis

- MOS-LQO

POLQA
(Sequence Mode)

This measurement requires a software option key. See page 189 for more information about software options.

Note: POLQA measurements depend upon correct level settings throughout the signal path, and will provide inaccurate results when levels are not properly set. See Level alignment for POLQA on page 453 for detailed information.

POLQA (Perceptual Objective Listening Quality Assessment) is a method of grading the quality of an audio communications channel, resulting in a MOS (Mean Opinion Score). See page 451 for more information about POLQA.

The POLQA measurement uses a speech sample stimulus signal that is passed through the DUT and compared with the original speech signal serving as a reference. A MOS result and a number of diagnostic cause analysis results are provided.

The POLQA measurement results available in APx500 are:

- MOS (LC IRS P.863)
- POLQA G.107
- MOS vs Time
- Average Delay
- Delay vs Time
- Delay vs Time (Resampled)
- POLQA Result Data
- Acquired Waveform
- Reference Waveform

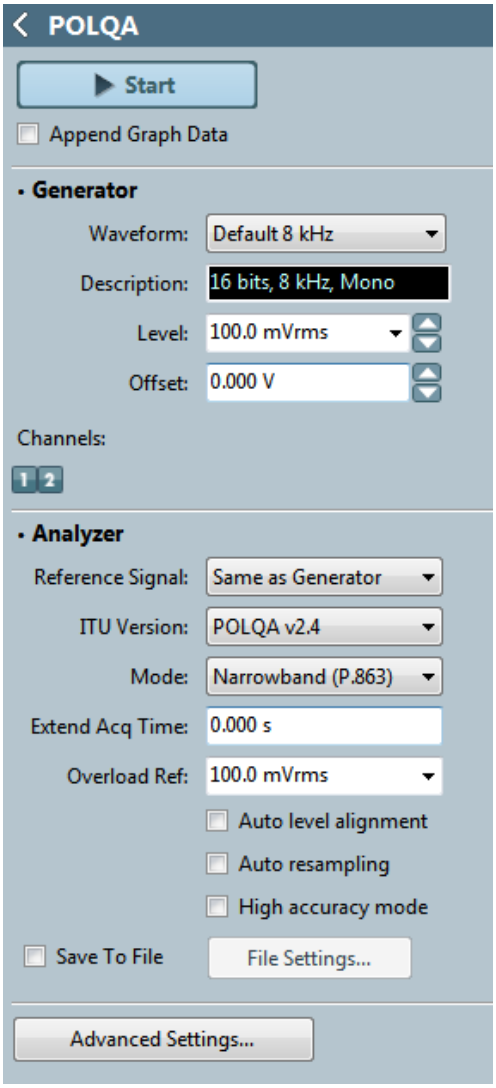
Operation

If you have not yet set up your test, first go to Chapter 7, Signal Path Setup.

When Signal Generation and Signal Analysis (below) are set for your test, click **Start**.

Append Graph Data

Measured data are grouped in a Data Set (see page 189), which contains all acquisition results. Normally, the current graph data are deleted each time you start a new measurement. If the **Append Graph Data** box is checked, appended data are grouped as additional data sets. The Data Sets Panel (see page 649) displays the Data Set(s) in the measurement, and provides a visibility checkbox and a Notes field for each data set.



Generator (closed loop configuration)

Waveform

For POLQA, a speech sample used as a test stimulus is called the *reference signal*.

For the POLQA measurement, the APx generator can play a speech sample that conforms to POLQA requirements; this is the reference signal. APx provides one speech sample at each of three sample rates: 8 kHz, 16 kHz and 48 kHz, embedded in the POLQA measurement. Select one of these, or browse for a waveform file. More POLQA speech samples are installed at C:\Program Files (x86)\Audio Precision\APx500 [version]\Waveforms\POLQA\...

Conforming waveforms are mono WAV files at a sample rate of 8 kHz, 16 kHz or 48 kHz, at a 16-bit word length (bit depth). Content should be speech samples. Length is typically 5 to 10 seconds; maximum length is 12 seconds. Files must be properly filtered and optimized for an Active Speech Level (ASL) of –26 dBov, using the ITU-T P.56 meter. See Level alignment for POLQA on page 453 for more information.

Description

When using a POLQA speech signal from a file on disk, this field displays information about the current waveform.

Level

Set the generator output level in this field.

For analog output, the file overload level (0 dBFS) is scaled to the value set in Level. A properly made POLQA speech signal will have an ASL of –26 dBov (equivalent to –26 dBFS), so the ASL in the analog output will be 26 dB below the value set in Level.

For digital output, the Level should be set to –0.000 dBFS to maintain correct ASL in the digital output stream.

See Level alignment for POLQA on page 453 for detailed information. See Generator Waveform Files on page 47 for a general discussion of generator level when playing waveform files.

Offset

Set an optional DC offset to the signal here.

Channel

The generator will output the speech samples to the DUT on the selected generator channels. Note that the Audio Precision POLQA-2 software option only enables POLQA analysis on 2 channels.

Using Output EQ with POLQA waveforms

See Output EQ on page 193.

If you are testing with complex waveforms (such as speech waveforms for POLQA) and find that the generator limiting feature of Output EQ reduces the level so far that you cannot attain the necessary Active Speech Level at the transducer, an alternative method is to use the APx EQ .wav Files Utility. Read about the utility by searching for the file “APx EQ .wav Files Utility Instructions.pdf” at ap.com.

External Source (open loop) configuration

Go to Signal Path Setup to choose the None (External) Output Configuration.

For External Source, the APx generator is not used. The DUT is a player or receiver that provides the stimulus signal. To enable the POLQA measurement to operate, the signal must be a conforming speech sample and must match the speech sample selected in Signal Acquisition and Analysis: Reference Signal. Read more about External Source in Chapter 6.

File Input configuration

Read more about File Input Configuration on page 172.

If you have chosen **File** for an input configuration in Signal Path Setup, the **Analyze** button will be added to the top of the measurement settings panel. Click the **File List...** button to add input file(s) to a list for analysis. To enable the POLQA measurement to operate, the signal must be a conforming speech sample and must match the speech sample selected in **Signal Generation: Waveform** (for closed loop) or **Signal Acquisition and Analysis: Reference Signal** (for open loop / external source).

If your DUT is to be recording the WAV file, select a generator waveform speech sample and click **Start**, as described above. Configure your DUT to save a WAV file(s) whose file name(s) and location match the name specified in the **File List**.

If you are measuring an existing file (not recording from the generator but working in External Source), be sure that the file name(s) and location match the name(s) specified in the **File List.** For POLQA measurements, the matching speech sample file must be available for selection as the **Reference Signal.**

Finally, click the **Analyze** button. The results will be shown on the graph display.

Start

The **Start** button runs the measurement. For closed loop operation, **Start** outputs the speech sample signal and prepares the analyzer for acquisition. For

External Source (open loop) measurements, **Start** prepares the analyzer for acquisition, awaiting a speech sample from the DUT.

Analyzer

For POLQA, the signal to be analyzed is called the *degraded signal*.

The SW-POLQA-2 option only provides POLQA analysis on input channels 1 and 2. To authorize more channels, contact your Audio Precision representative.

Filters

There are no local filters available for this measurement; the global filters in Signal Path Setup > Input/Output will affect this measurement, if set. Filter use in POLQA measurements may degrade MOS scores. Read more about APx Filters in Chapter 97.

Reference Signal

in Closed Loop configuration...

In closed loop configuration, the **Reference Signal** is usually the **Generator Waveform** that is currently selected.

In situations where you have externally modified or equalized a Reference Waveform (to compensate for loudspeaker response, for example), you have the option to select an alternative **Reference Signal** (the unmodified reference waveform, for example) here.

in External Source configuration...

When in External Source (open loop) configuration, you must choose a reference speech sample file here. The speech sample file (or compensated speech sample file) being played from the DUT must correspond to the **Reference Signal** chosen here.

Note: the use a reference file whose sample rate differs from the target file is not standards compliant, and an error will be thrown. To suppress this error, set the Auto Resampling checkbox in Advanced Settings.

ITU Version

The original POLQA V1.1 description in ITU-T P.863 (2011) has been superseded by POLQA V2.4, described in ITU-T P.863 (2014). Choose the POLQA version you would like to test against here. See More About POLQA on page 451.

Mode

Conforming POLQA speech sample files have an 8 kHz sample rate for testing Narrowband (P.863) channels, a 16 kHz sample rate for testing Wideband (P.863) channels, and a 48 kHz sample rate for testing Super-Wideband (P.863) channels. The mode is selected automatically by the POLQA measurement based on the sample rate contained in the reference file.

A conforming POLQA speech file is also optimized for an Active Speech Level (ASL) of –26 dBov ASL, using the ITU-T P.56 meter. See Level alignment for POLQA on page 453.

Extend Acq. Time

The acquisition time is set to the length of the speech sample (selected either in **Waveform**, for closed loop, or **Reference Signal** for External Source (open loop), plus the time set in this field. If you suspect you are getting poor results because the acquisition length is truncating the input signal, add time here; 10 s is the maximum.

This field is not available if **Input Configuration** is set to **File**. In **File Input**, the acquisition time is set to the length of the input file. In either case, the maximum acquisition length is 35 seconds.

Save to File

You can record the POLQA audio signal to a file as the signal is being acquired. Click **Save to File** before you run the POLQA measurement. See Recording Audio to a File on page 381 for more information.

File Settings

When saving a POLQA acquisition to a file, click **File Settings** to specify the file name and location.

Overload Ref:

A key difference between POLQA and PESQ is in sensitivity to signal level. PESQ ignores level, but POLQA will produce poor MOS scores if the active speech level (ASL) presented to the algorithm is not matched (within a certain range) to the ASL of the reference file, which itself must be ASL-optimized for POLQA.

The **Overload Ref:** field is available when **Auto level alignment** is OFF (the default; see below). Enter the overload value of the DUT or system under test here to correctly align the ASL. For analog signals, the value set here is typically the clipping point of the analog DUT; for digital signals, the value is typically 0 dBFS. See Level alignment for POLQA on page 453.

Auto Level Alignment

APx provides an **Auto level alignment** checkbox to automatically align the input stream ASL to the level required, which makes good MOS scores easier to obtain. This is useful if your test requirements do not consider signal level, or if temporarily defeating level sensitivity is helpful in diagnosis. However, when **Auto level alignment** is ON, the APx POLQA measurement is not compliant with the ITU-T P.863 standard.

See Level alignment for POLQA on page 453 for information on adjusting levels to optimize MOS scores while remaining standards-compliant.

Auto resampling

If this checkbox is checked, the POLQA algorithm defaults to the lowest sampling rate when it detects a difference between the reference and target signals, and suppresses the error that is thrown. This may result in an improved score, but the result will not be standards compliant.

High accuracy mode

When checked, High-accuracy mode engages a mechanism in the POLQA algorithm that shifts the time alignment of the degraded signal in sub-sample steps to find the optimal (highest) MOS score. Use this feature if it provides better results in your test. See Sub-sample time shift on page 453.

Warnings

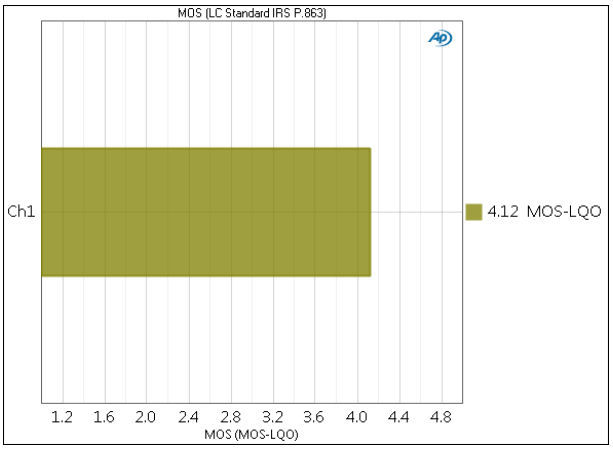
If the level of either the reference signal or the degraded signal is too high or too low, a warning symbol is displayed. View POLQA Result Data result (page 449), or the POLQA log file (page 454) for level data.

If either signal is too long, a warning symbol is displayed. POLQA signals must be less than 12 seconds long.

Advanced Settings

If you'd like to adjust analyzer ranging parameters, click **Advanced Settings**. See Advanced Settings on page 450. Read more about Units in Chapter 104.

POLQA: MOS (LC IRS P.863) result



The **POLQA: MOS** result is the primary result for the POLQA measurement. The POLQA score is mapped to MOS-LQO LC STANDARD IRS P.863.

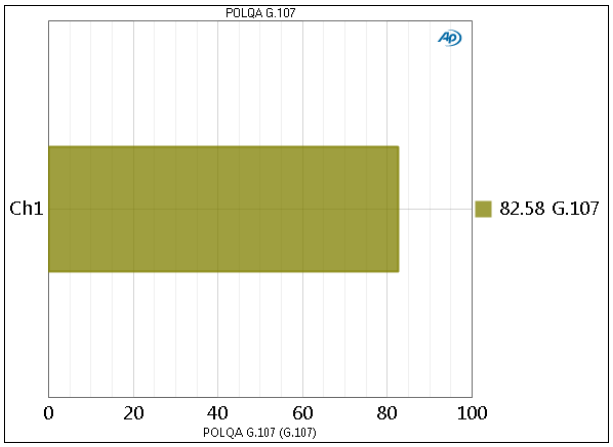
Units

Units for POLQA: MOS (LC STANDARD IRS P.863).

are:

- MOS-LQO

POLQA: POLQA G.107 result



The **POLQA: POLQA G.107** result shows the POLQA score for Wideband and SuperWideband analysis. This result is invalid and not populated with data in Narrowband analysis.

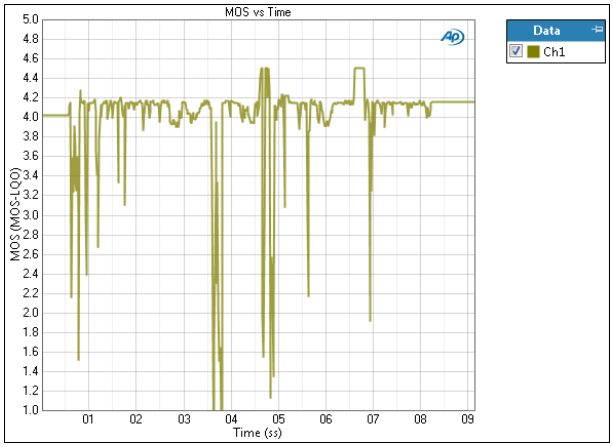
The POLQA G.107 score should be treated as an intermediate result, and the MOS scores reported instead

Units

Units for POLQA: POLQA G.107 results are:

- G.107

POLQA: MOS vs Time result



The POLQA: MOS vs Time is useful for cause analysis. The result shows a graph of the MOS score versus elapsed time.

Units

Units for POLQA: Delay vs Time are:

X-axis

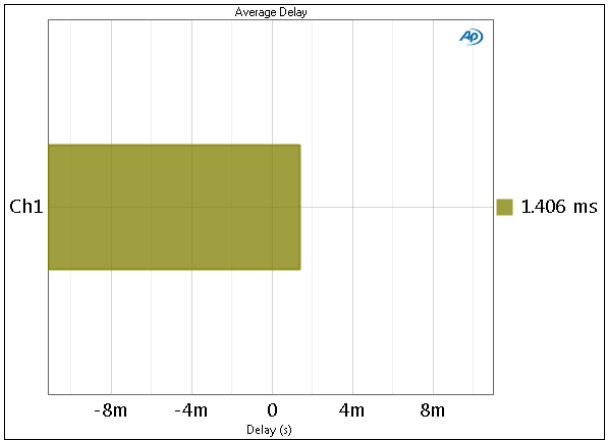
- s (seconds)

Y-axis

- MOS-LQO

Read more about Units in Chapter 104.

POLQA: Average Delay result



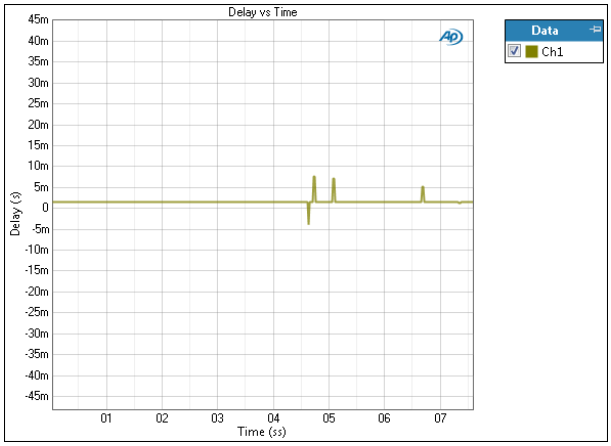
The POLQA: Average Delay result is useful for cause analysis. The result shows the average signal delay through the DUT or system.

Units

Units for POLQA: Average Delay are:

- s (seconds)

POLQA: Delay vs Time result



The POLQA: Delay vs Time result is useful for cause analysis. The result charts the delay between reference and acquisition over the span of the POLQA measurement.

Units

Units for POLQA: Delay vs Time are:

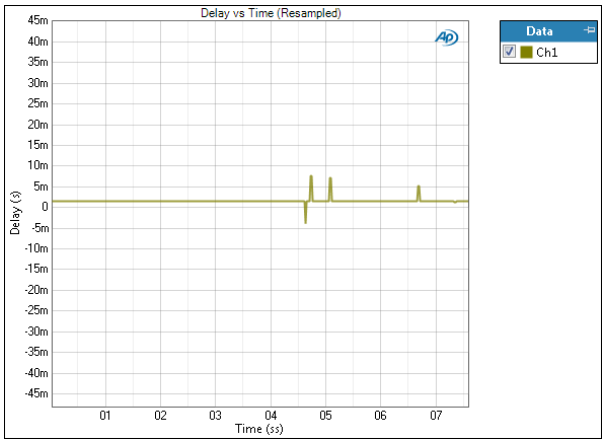
X-axis

- s (seconds)

Y-axis

- s (seconds)

POLQA: Delay vs Time (Resampled) result



The POLQA: Delay vs Time (Resampled) result is useful for cause analysis. The result charts the delay between reference and acquisition over the span of the POLQA measurement, after **Auto resampling** has been applied.

Units

Units for POLQA: Delay vs Time are:

X-axis

- s (seconds)

Y-axis

- s (seconds)

POLQA: POLQA Result Data

POLQA Result Data - Default 8 kHz

	Data	Ch1	Ch2
1	MOS-LQO (P.863)	4.06	3.87
2	Active Speech Level (ASL) - Reference	-25.946 dBov	-25.946 dBov
3	Active Speech Level (ASL) - Degraded	-25.920 dBov	-25.842 dBov
4	Pause Level - Reference	-69.754 dBov	-69.754 dBov
5	Pause Level - Degraded	-68.349 dBov	-68.079 dBov
6	Active Speech Ratio (ASR) - Reference	0.587	0.587
7	Active Speech Ratio (ASR) - Degraded	0.585	0.585
8	Signal to Noise Ratio (SNR) - Reference	39.464 dB	39.464 dB
9	Signal to Noise Ratio (SNR) - Degraded	39.535 dB	39.487 dB
10	Level of Signal - Reference	-27.724 dB	-27.724 dB
11	Level of Signal - Degraded	-27.721 dB	-27.623 dB
12	Pitch - Reference	164.968 Hz	164.968 Hz
13	Pitch - Degraded	164.588 Hz	164.958 Hz
14	Attenuation	-0.025 dB	-0.104 dB
15	Minimum Transmission Delay	-3.625 s	-2.125 s
16	Maximum Transmission Delay	6.125 s	3.000 s
17	Processing Sample Rate	8.00000 kHz	8.00000 kHz
18	Estimated Sample Rate	8.00000 kHz	8.00000 kHz
19	Resampling Applied	Disabled	Disabled

POLQA Result data displays 19 results calculated by the POLQA algorithm. Each time a POLQA measurement is made, the 19 results are updated. The same information is written to the POLQA log file.

Reviewing these data while setting levels for a POLQA measurement can be helpful, as the reference ASL

and the degraded ASL are both shown, and the Overload Ref value can be adjusted to properly align these values.

Units

Units for POLQA: Delay vs Time are:

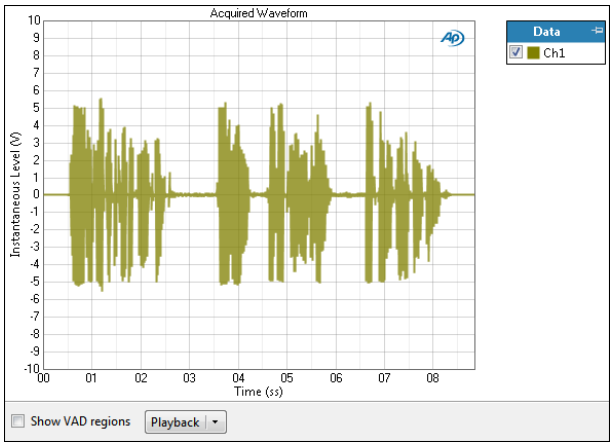
X-axis

- s (seconds)

Y-axis

- s (seconds)

POLQA: Acquired Waveform result



The POLQA: Acquired Waveform result is useful for cause analysis. The result shows a time-domain view of the acquired speech sample waveform.

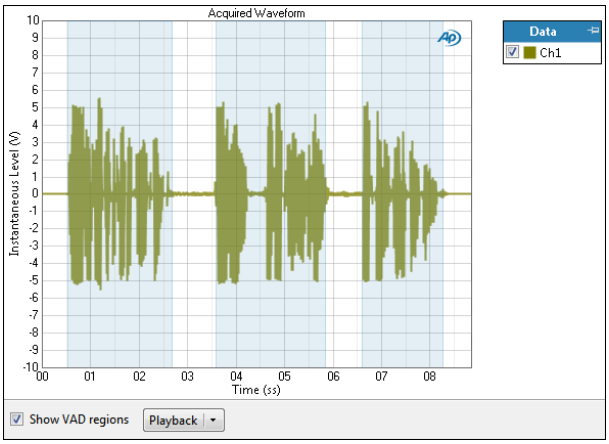
Additional Controls for Acquired Waveform result

Show VAD

If **Show VAD regions** is checked the result will indicate graphically on the acquired waveform where the POLQA algorithm's Voice Activity Detection has found voice sounds and performed its analysis. VAD data is not exported nor retained in the project.

Playback

You can play the POLQA measurement Acquired Waveform through the PC Windows audio. Click the Playback menu to list the available acquired channels, and play the desired channel by selecting it.

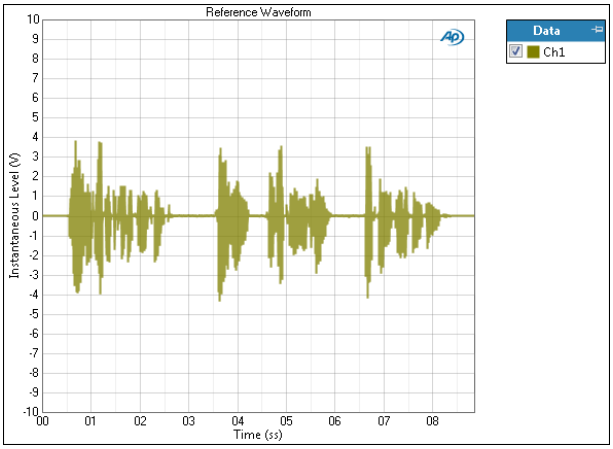


Units

Units available for POLQA: Acquired Waveform are:

X-axis	*Y-axis (analog)*	*Y-axis (digital)*
• s (seconds)	• V	• D • hex

POLQA: Reference Waveform result



The POLQA: Reference Waveform result is result is useful for cause analysis. The result shows a time-domain view of the reference speech sample waveform.

Units

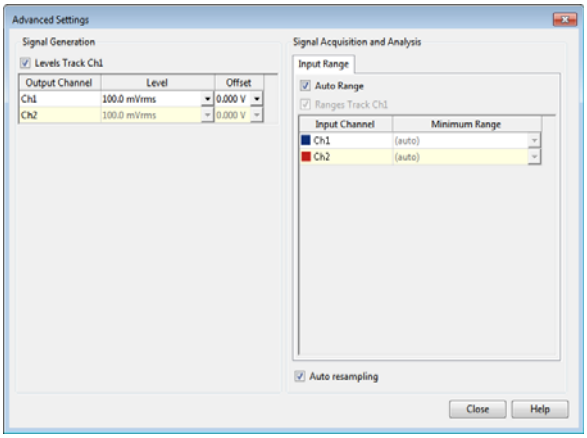
Units for POLQA: Reference Waveform are:

X-axis	*Y-axis (analog)*	*Y-axis (digital)*
• s (seconds)	• V	• D • hex

Advanced Settings for POLQA

The default settings here are appropriate for meter (single value) measurements under most conditions.

You may want to make minor adjustments for special situations.



Signal Generation

If **Levels Track Ch1** is checked (the default), the generator output level value for channel 1 is copied to the other channels, and output levels for channels greater than 1 cannot be edited. Any changes made to channel 1 are reproduced in the other channels.

To set individual generator output channel levels, uncheck the **Levels Track Ch1** checkbox and enter values in the output channel Level fields.

Signal Acquisition and Analysis

Input Range tab

> *Digital inputs do not require ranging, and the Input Range tab is unavailable when inputs are set to a digital format.*

Auto Range

If **Auto Range** is checked (the default), each analog channel determines its input range automatically, based on the level of the input signal. If the input signal level changes beyond a ranging threshold, **Auto Range** will cause the input ranging circuits to move up or down for proper ranging. You will hear relays clicking when the ranging changes.

Fixed Range

Alternatively, you can set a fixed minimum range for the analog inputs. Uncheck **Auto Range** and enter the expected maximum level in the **Minimum Range** filed. The software will choose the lowest range that will accommodate that level; this becomes the minimum range available, and the analyzer will not reset to a lower range. Fixed range is best for signals that vary rapidly, including tone bursts, perceptual audio speech samples, and program material such as voice or music.

If a signal greater then the **Minimum Range** is presented to the analog input, the ranging circuits will range up to accommodate the higher level, and then drop again when the signal level falls. The range will not be set below the **Minimum Range**.

Ranges Track Ch1

If **Ranges Track Ch1** is checked (the default), the fixed input range setting for channel 1 is copied to the other channels, and range settings for channels greater than 1 cannot be edited. Any changes made to channel 1 are reproduced in the other channels. If **Ranges Track Ch1** is not checked, the fixed input range setting for each channel can be set independently.

Read more about ranging in Chapter 98.

Auto Range and POLQA

POLQA and other perceptual audio measurements use voice samples as test signals, and operate best when ranging is fixed. See **When autoranging isn't suitable** on page 626.

More about POLQA and perceptual audio testing

Overview

The POLQA perceptual measurement algorithm is a joint development of OPTICOM, SwissQual and TNO, protected by copyrights and patents and available under license as software for various platforms.

POLQA (Perceptual Objective Listening Quality Assessment), like PESQ, is a method of grading the quality of an audio communications channel. Perceptual audio evaluation methods were developed to enable evaluation of audio quality in channels of compromised performance, where conventional measurements such as THD+N may provide results that are not useful.

POLQA is the successor of PESQ (ITU-T Rec. P.862). POLQA avoids weaknesses of the current P.862 model and is extended towards handling of higher bandwidth audio signals. Further improvements target the handling of time scaled signals and signals with many delay variations. Similarly to P.862,[2] (PESQ), POLQA supports measurements in the common telephony band (300–3400 Hz), but in addition it has a second operational mode for assessing HD-Voice in wideband and super-wideband speech signals (50–14000 Hz). POLQA also targets the assessment of speech signals recorded acoustically by an artificial head with mouth and ear simulators.

Perceptual audio evaluation techniques are based on subjective experiments with listeners, and do not use simple periodic waveforms or sweeps as most APx measurements do; consequently, the perceptual evaluation results cannot be stated as physical values such as volts or hertz. Instead, listeners' opinions of quality (or algorithms designed to respond similarly) are averaged and scored.

PESQ and POLQA are both methods of generating MOS results without actually using a room full of listeners. Instead, pre-recorded speech samples are passed through a DUT or system and are compared with the original speech sample. An algorithm evaluates the difference between the reference audio and the degraded audio, and produces a result.

ITU-T P.863 (2011) (POLQA V1.1)

POLQA V1.1 is defined in ITU-T Recommendation P.863, introduced in 2011. See http://www.itu.int/rec/T-REC-P.863 “Superceded and Withdrawn components.”

ITU-T P.863 (2014) (POLQA V2.4)

POLQA Version 1.1 has been superseded by Version 2.4, described in ITU-T P.863 (2014). Version 2.4 is also referred to as “POLQA 2015” and POLQA Edition 2. See http://www.itu.int/rec/T-REC-P.863 “In force components.”

Channels of compromised audio quality

Many speech communications channels have limited audio capabilities, when compared to full fidelity channels optimized for music transmission. The limited capabilities of speech channels may provide other benefits: lower cost, lower bandwidth, lower power requirements, and so on.

Most of the I/O interfaces offered in an APx analyzer have very high fidelity capabilities, but in some cases the configuration for a low data rate DUT may make POLQA evaluations valuable. Passing a stimulus signal through a telephone network, for example, will likely degrade the signal substantially, as will low data rate codecs. The Bluetooth HSP and HFP profiles, in particular, are of moderate fidelity. Bluetooth manufacturers often use POLQA to characterize Bluetooth devices operating in these profiles.

MOS (Mean Opinion Score)

Perceptual audio evaluations are typically made with a number of listeners, who provide opinions about the quality of audio samples they have listened to. Typically, many audio samples, many listeners and many iterations are used, and the results are averaged into a MOS (Mean Opinion Score). MOS is expressed in a unitless scale from 1 to 5, representing the average opinion of quality, as shown in this table:

MOS	*Quality*	*Impairment*
5	Excellent	Imperceptible
4	Good	Perceptible but not annoying
3	Fair	Slightly annoying
2	Poor	Annoying
1	Bad	Very annoying

MOS results in APx are reported as MOS-LQO (LQO stands for Listening-only Quality, Objective). For POLQA, the MOS-LQO results reference LC STANDARD IRS P.863.

Active Speech Level (ASL) in POLQA

Active Speech Level is a level definition in POLQA that is the result of a measurement of speech samples made by a specific method, compliant with a meter defined in ITU-T P.56. The P.56 meter has optimized measurement detection and ballistics for this purpose, and largely ignores the silences between utterances, focusing instead on the active portions of the speech waveform.

A key difference between POLQA and PESQ is in sensitivity to signal level. PESQ ignores level, but POLQA will produce poor MOS scores if the active speech level (ASL) presented to the algorithm is not matched (within a certain range) to the ASL of the reference file, which itself must be ASL-optimized for POLQA.

An **Overload Ref:** field is available. Enter the overload value of the DUT or system under test here to correctly match the ASL. For analog signals, the value set here is typically the clipping point of the analog DUT; for digital signals, the value is typically 0 dBFS.

APx provides an **Auto level alignment** checkbox to automatically align the input stream ASL to the level required, which makes good MOS scores easier to obtain. This is useful if your test requirements do not consider signal level, or if temporarily defeating level sensitivity is helpful in diagnosis. However, when **Auto level alignment** is **ON**, the APx POLQA measurement is not compliant with the ITU-T P.863 standard.

See **Level alignment for POLQA** (beginning on page 453) for information on adjusting levels to optimize MOS scores while remaining standards-compliant.

Speech samples

POLQA evaluates speech samples. You can record your own samples, or obtain speech sample files from other sources. The ITU has created a number of samples designed to cover the range of speech sounds uttered by men and women in a range of languages. These are available at

- http://www.itu.int/net/itu-t/sigdb/genaudio/Pseries.htm

and

- http://www.itu.int/rec/T-REC-P.501/en.

See the ITU P501E documentation at the second link for detailed information about creating and using speech samples.

For POLQA, APx500 provides one speech sample at a sample rate of 8 kHz, one at 16 kHz, and one at 48 kHz embedded in the measurement. More speech sample files are provided as part of the APx distribution, and are saved to the PC hard disk. These speech

samples have been optimized by filtering and having ASL set to –26 dB dBov using the ITU-T P.56 meter.

Sub-sample time shift

The algorithm moves the degraded signal in time to align with the reference signal for comparison. This time alignment is accurate to a resolution of 1 sample.

Sub-sample time shift is defined as an offset in the time alignment of the two signals greater than zero but less than one sample, which can occur in systems under test. Sub-sample time shift can adversely affect MOS scores.

High-Accuracy Mode engages a mechanism in the POLQA algorithm that shifts the time alignment of the degraded signal in sub-sample steps to find the optimal (highest) MOS score. Use this feature if it provides better results in your test.

Averaging

Averaging multiple speech samples across multiple iterations provides the most reliable MOS results. APx500 provides POLQA (Averaged) measurements that enable you to average up to 64 speech samples or iterations.

Other results

The primary POLQA results in APx500 are the MOS (LC Standard IRS P.863) result and the MOS Average (LC Standard IRS P.863) result.

Other results are provided that are unsupported by ITU-T P.863. These results may be helpful in diagnostics and cause analysis in the DUT or system.

A POLQA log file saves 19 results for each POLQA iteration. See page 454.

Level Alignment for POLQA

ASL and dBov

Active Speech Level (ASL) is a level definition in POLQA that is the result of a measurement of speech samples made by a specific method, compliant with a meter defined in ITU-T P.56. The P.56 meter has optimized measurement detection and ballistics for this purpose, and largely ignores the silences between utterances, focusing instead on the active portions of the speech waveform.

The POLQA algorithm is more critical than the PESQ algorithm in a number of ways. In particular, POLQA scores are sensitive to signal level where PESQ scores were not. For a good score in POLQA, the ASL of the degraded signal must closely match that of the reference signal, which itself must be ASL-optimized for POLQA.

dBov units are referenced to the level which causes overload in a system. POLQA requires that speech test signals are maintained at –26 dBov throughout the tested system for correct MOS scores (see the graph). In the acoustic domain, the speech signals should be presented at 73 dBA for monotic narrow bandwidth measurements, and at 79 dBA for diotic wideband measurements. See below. A gain adjustment is available in APx to properly scale input signals. This adjustment is in the form of an entry field called Overload Ref, where the known overload level of the device connected to the analyzer input is entered. See below.

POLQA Use cases

You may find it helpful to refer to the illustrated use cases listed in POLQA Use Cases, beginning on page 455.

APx analyzer analog input ranging

Because the level of a speech signal varies, the analog input range may change during a PESQ or POLQA measurement. We recommend that you determine the maximum level that your DUT will present to the analyzer input, and set fixed ranging to accommodate that level. Read more about ranging in Chapter 98.

ASL in all-digital signal paths

In the digital domain, an ASL of –26 dBov is –26 dBFS. Since 0 dBFS is the overload point in a digital system, in most cases the –26 dBov ASL is correctly maintained throughout the path, and eventually input to the APx analyzer and passed to the POLQA algorithm at the correct level.

ASL in analog and cross-domain paths

The analog component of a signal path will have an arbitrary signal level scaling, and an analog overload point unrelated to 0 dBFS. In this case the input and/or output gain of the DUT must be adjusted to apply and maintain an ASL within the DUT of –26 dB, relative to the system overload point. Set the DUT overload point in the **Overload Ref:** field in APx to scale the signal properly.

You can find the overload point of a DUT using standard distortion measurement techniques. Measure the rms output level of a DUT that is on the threshold of overload, and use that value in the Overload Ref: field.

ASL in acoustic paths

For a system that includes an acoustic path, the electrical parts of the path should be set so that the test signal speech is at an ASL of –26 dBov (relative to the DUT's overload point), and the acoustic part of the path should be set so that the test signal speech is at an ASL of 73 dBA (dBSPL with A-weighting) for monotic narrow bandwidth measurements, and at

79 dBA for diotic wideband measurements. The final part of the path is typically a microphone and preamplifier, and the microphone output must be scaled properly. Enter a value in the **Overload Ref:** field for scaling, and observe the "Active Speech Level (ASL) – Degraded" result in the POLQA log file. If the level in the log file is not close to –26 dBov, enter greater or lesser scaling factors until –26 dBov is attained.

It can be difficult to achieve an ideal ASL through an acoustic path, and you may find that setting Auto Level Alignment to ON is useful in producing satisfactory MOS-LQO results.

Auto level alignment

Auto level alignment enables two simultaneous stages of compensation:

- APx automatically scales the signal so that it is applied to the algorithm at the optimal level.
- The POLQA algorithm implements its internal auto level mode, where both the reference signal and the degraded signal are forced to an ASL of –26 dBov.

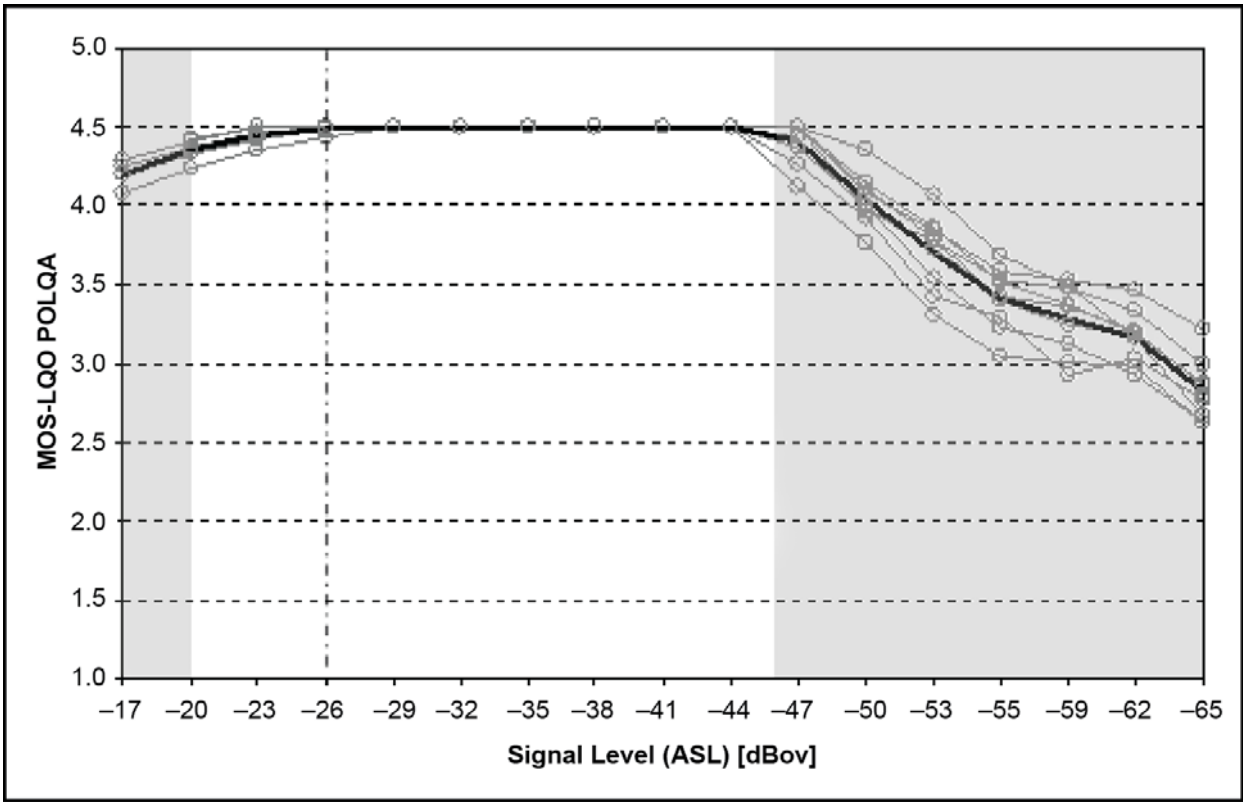
Generally, for standards-compliant POLQA MOS results in APx the **Auto level alignment** checkbox must be OFF.

You should optimize levels in your signal path so that the speech samples are at –26 dBov within the DUT or system under test, and so that the signal presented to the analyzer inputs is at the correct level to present an ASL of –26 dBov to the POLQA algorithm, after scaling in the analyzer input and converter stages.

APx speech files

More POLQA speech samples are installed at C:\Users\{user}\AppData\Local\Audio Precision\APx500\POLQA\PolqaResultData.csv.

The speech sample files provided by Audio Precision for POLQA have been optimized, with ASLs set to –26 dBov using the ITU-T P.56 meter. In digital loopback, a POLQA measurement using the Default 48 kHz file will score at MOS 4.75 (the maximum for 48 kS/s). In other I/O configurations, generator level settings and DUT or system gain will typically change the level and the MOS score. You can find information about the P.56 meter at http://www.itu.int/rec/T-REC-P.56/.



Level differences between reference and degraded signals are allowed, but must be within the range of +5 dB above to –20 dB below the –26 dBov level, as shown here.

The POLQA Log File

The POLQA log file records 18 results calculated by the POLQA algorithm. Each time a POLQA measurement is made, the 18 results and a time stamp are appended to the log file.

Reviewing this file while setting levels for a POLQA measurement can be helpful, as the reference ASL and the degraded ASL are both shown, and the Overload Ref value can be adjusted to properly align these values.

The file is saved at

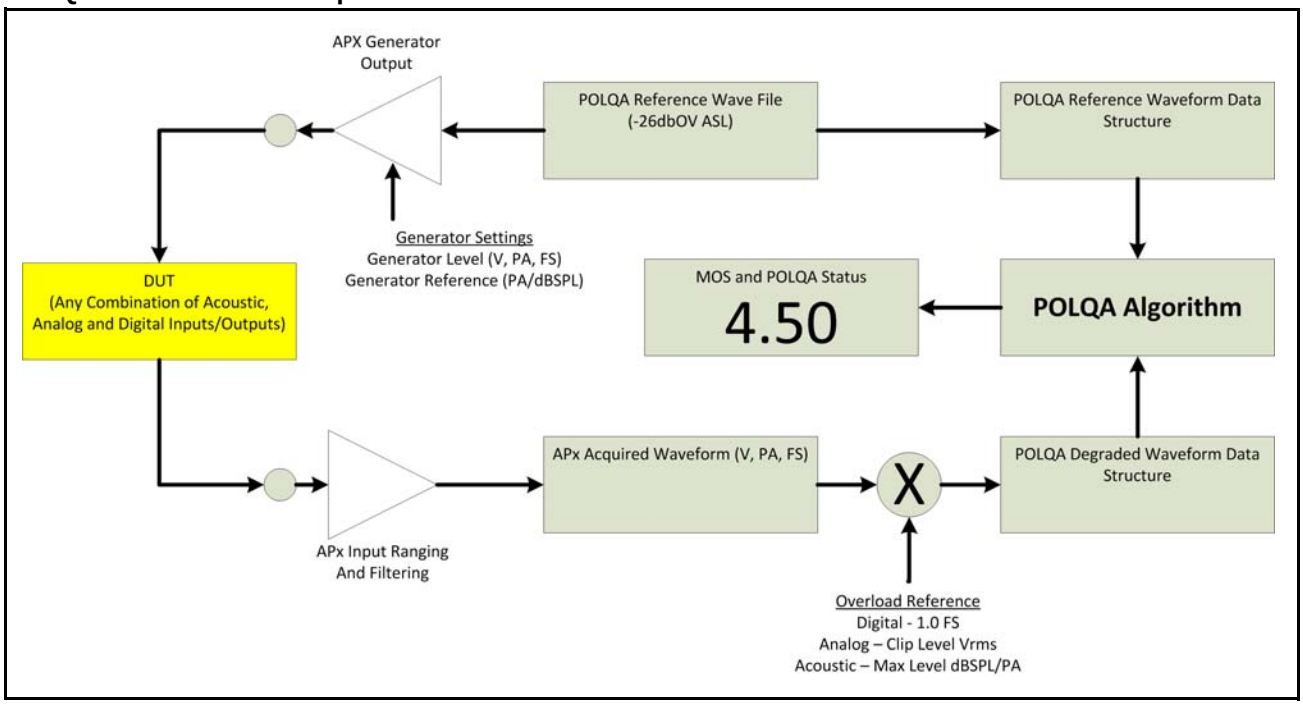
- C:\Users\{user}\AppData\Local\Audio Precision\APx500\POLQA\PolqaResultData.csv.

The first row of each appended measurement contains the date stamp. The first column shows the result name, the second column channel 1 data, and the third column channel 2 data. See the example in the APx500 embedded help.

POLQA Use Case Diagrams

A functional diagram of the POLQA architecture is provided, along with conceptual block diagrams for 12 POLQA use cases.

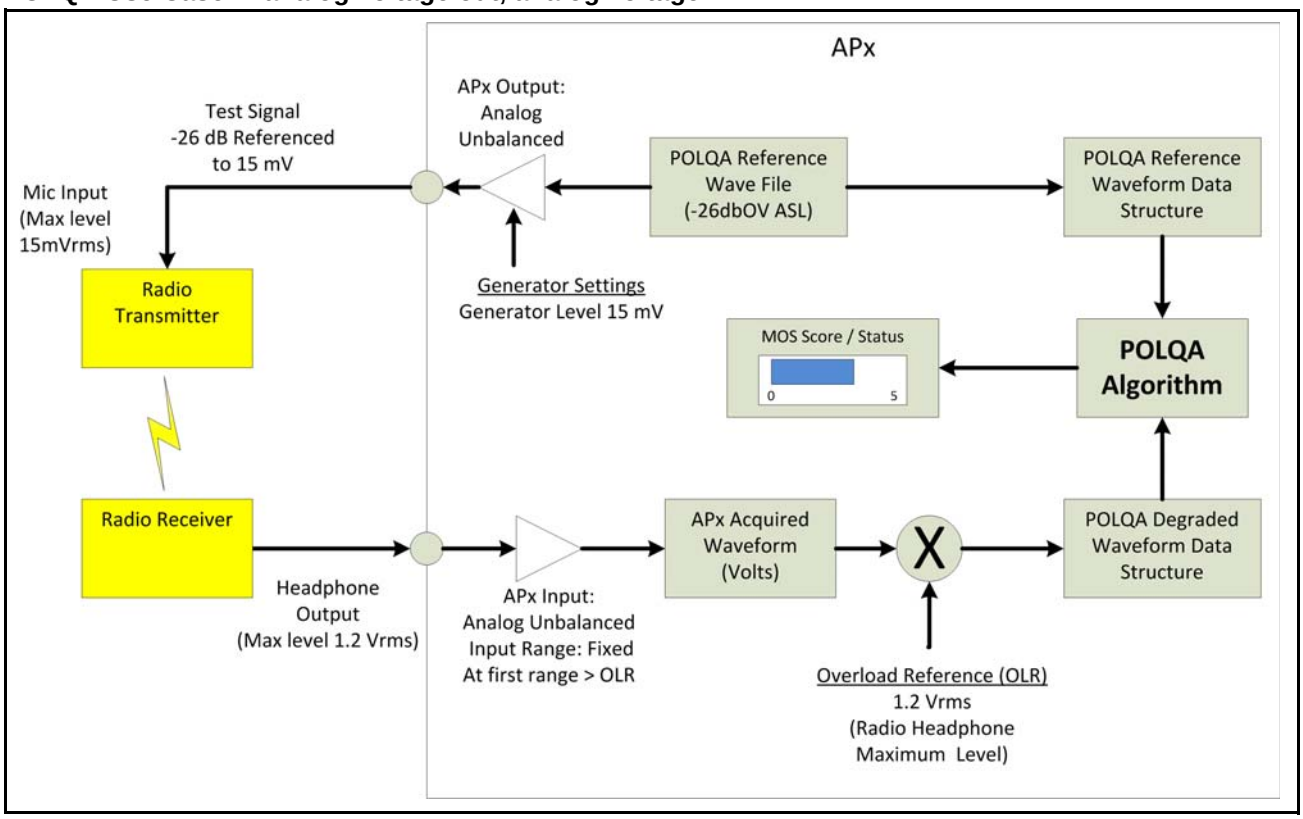
POLQA Architecture as implemented in APx500



APx uses an algorithm licensed from OPTICOM GmbH to perform POLQA evaluations. The algorithm requires properly formed reference signals and properly level-aligned degraded signals for correct MOS results.

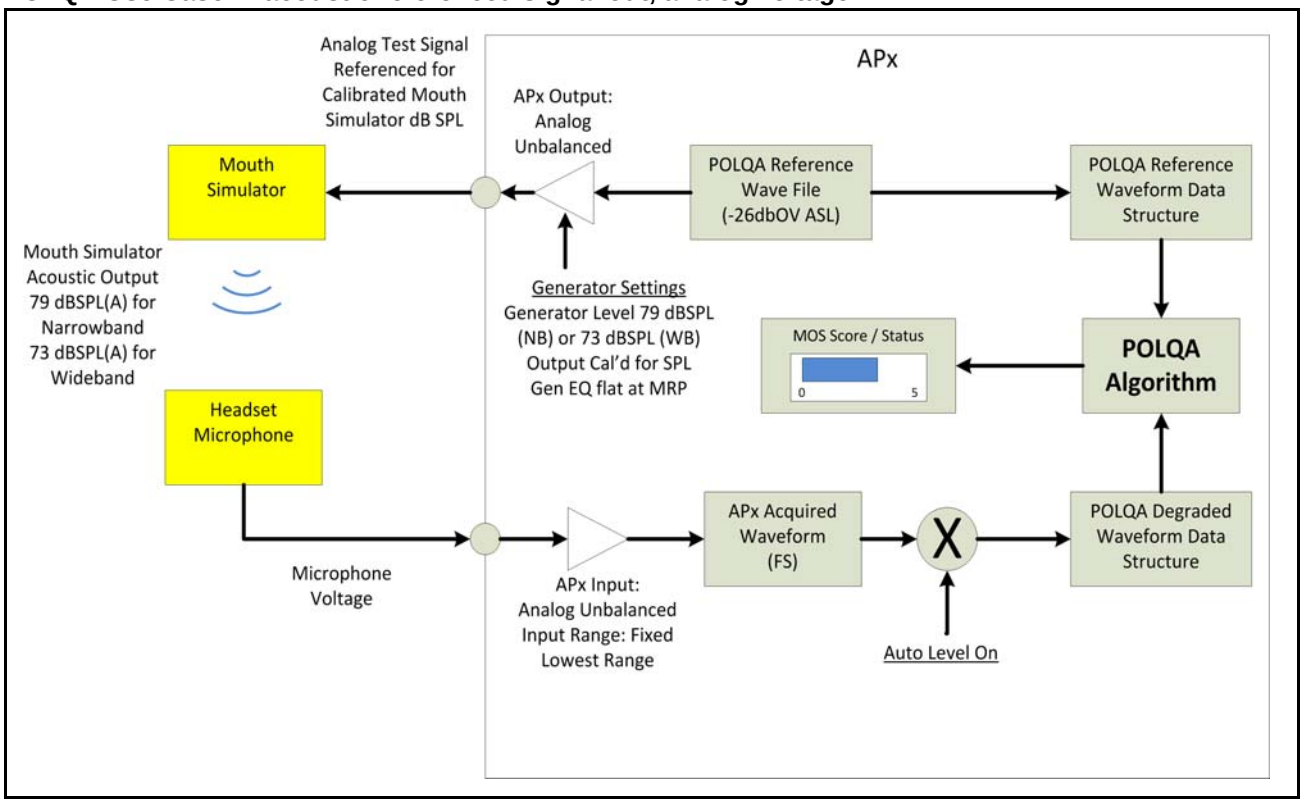
APx displays a number of results and warning flags from the POLQA algorithm in the POLQA and POLQA Averaged measurements User Interface (UI). Some data from the algorithm are not displayed in the UI, but are available in the POLQA log file. See page 454 for more information about the POLQA log file.

POLQA Use Case 1: analog voltage out, analog voltage in.

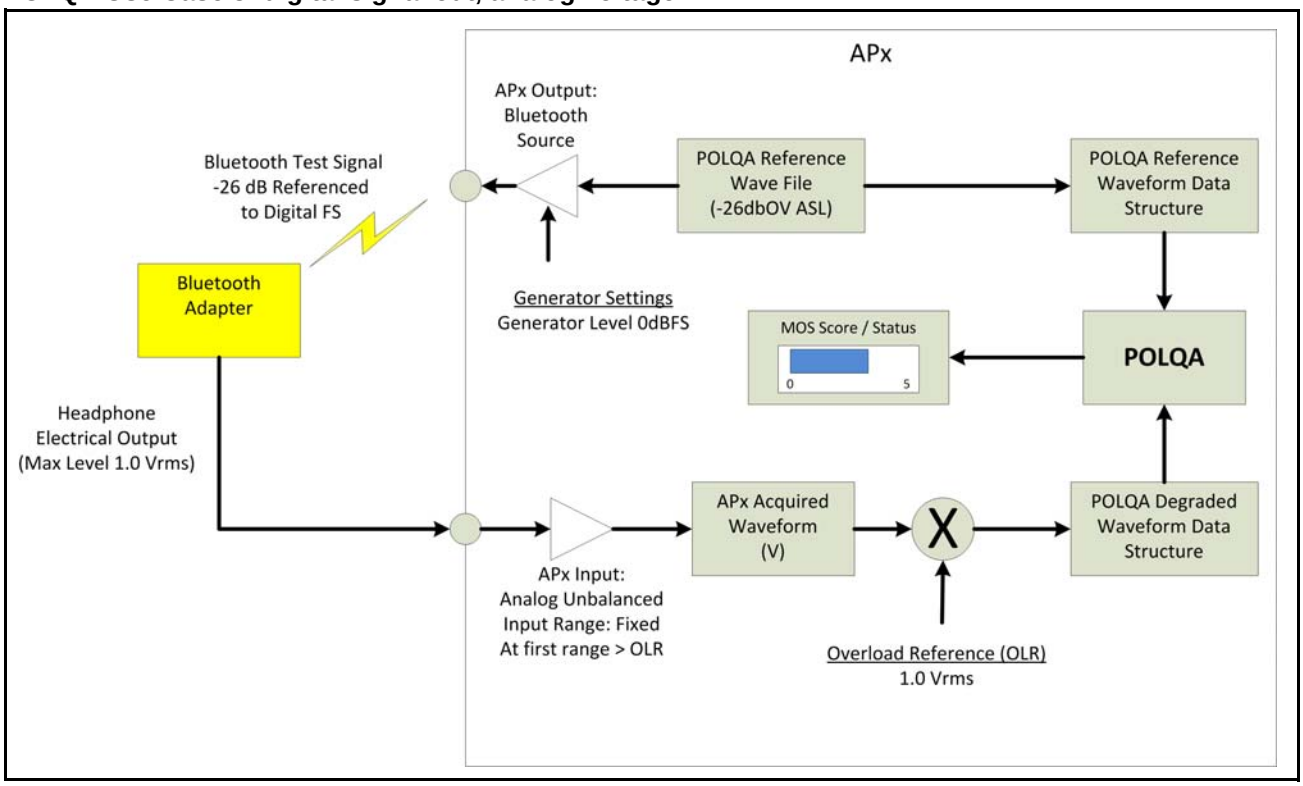


This illustration shows a radio link in the DUT system, but any analog-in, analog-out DUT system belongs to this use case.

POLQA Use Case 2: acoustic referenced signal out, analog voltage in

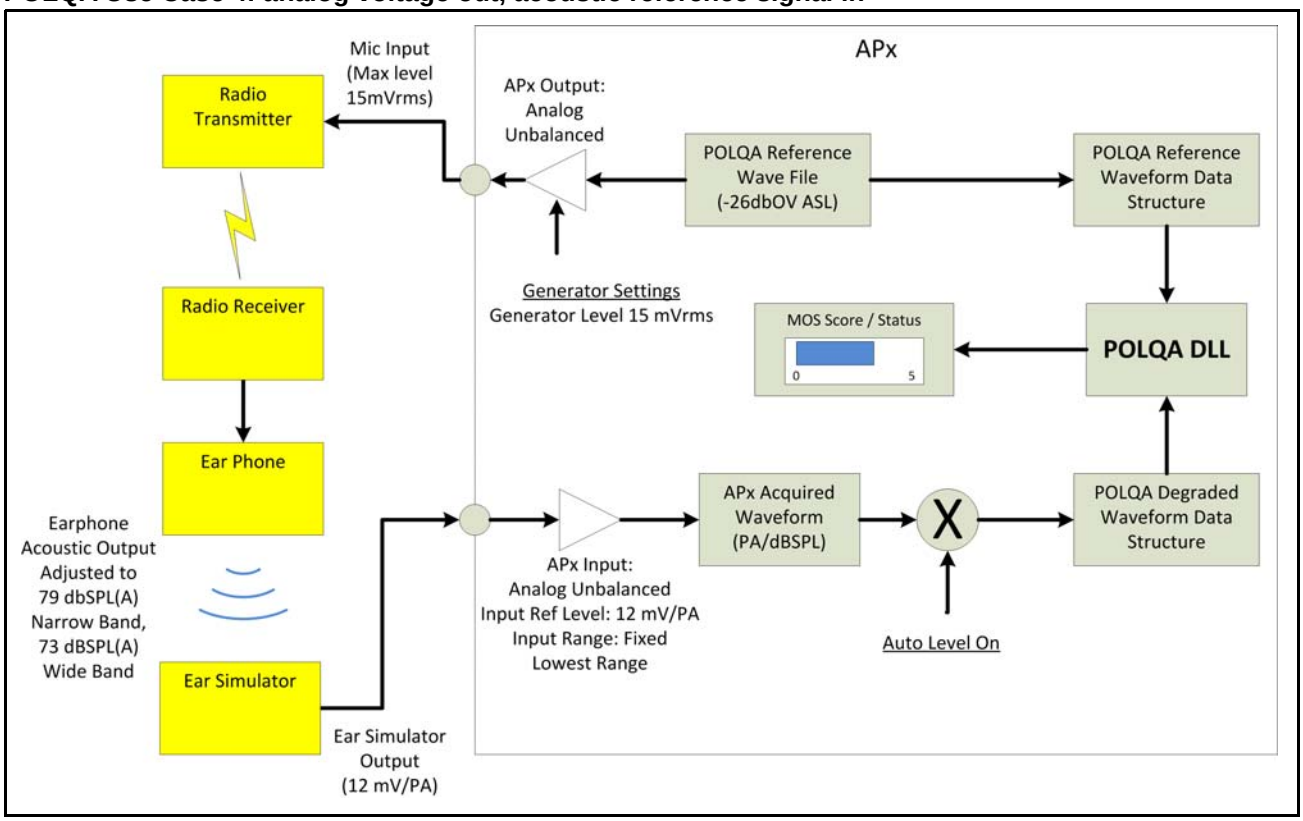


POLQA Use Case 3: digital signal out, analog voltage in

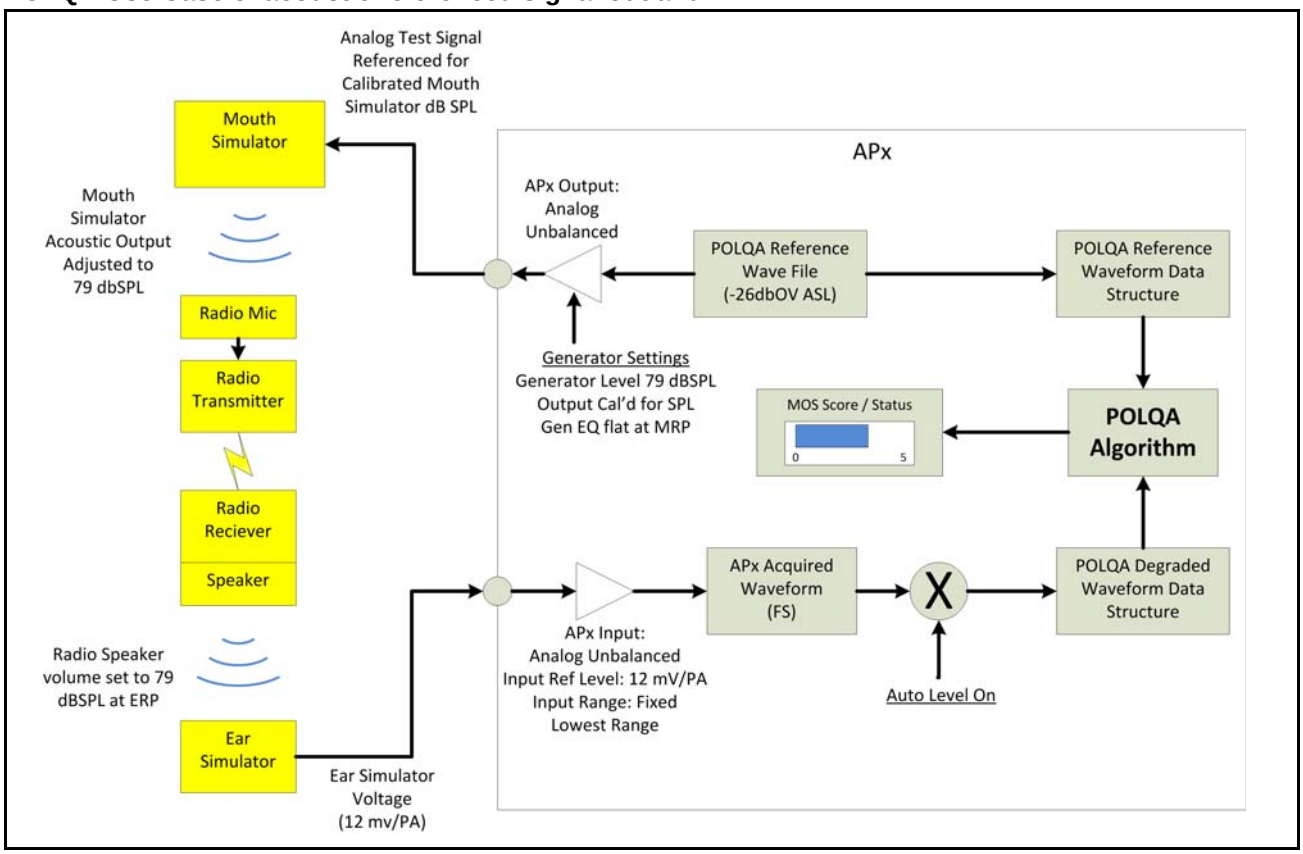


This use case shows a digital signal from the generator (in this case, using Bluetooth) to a device with an analog output.

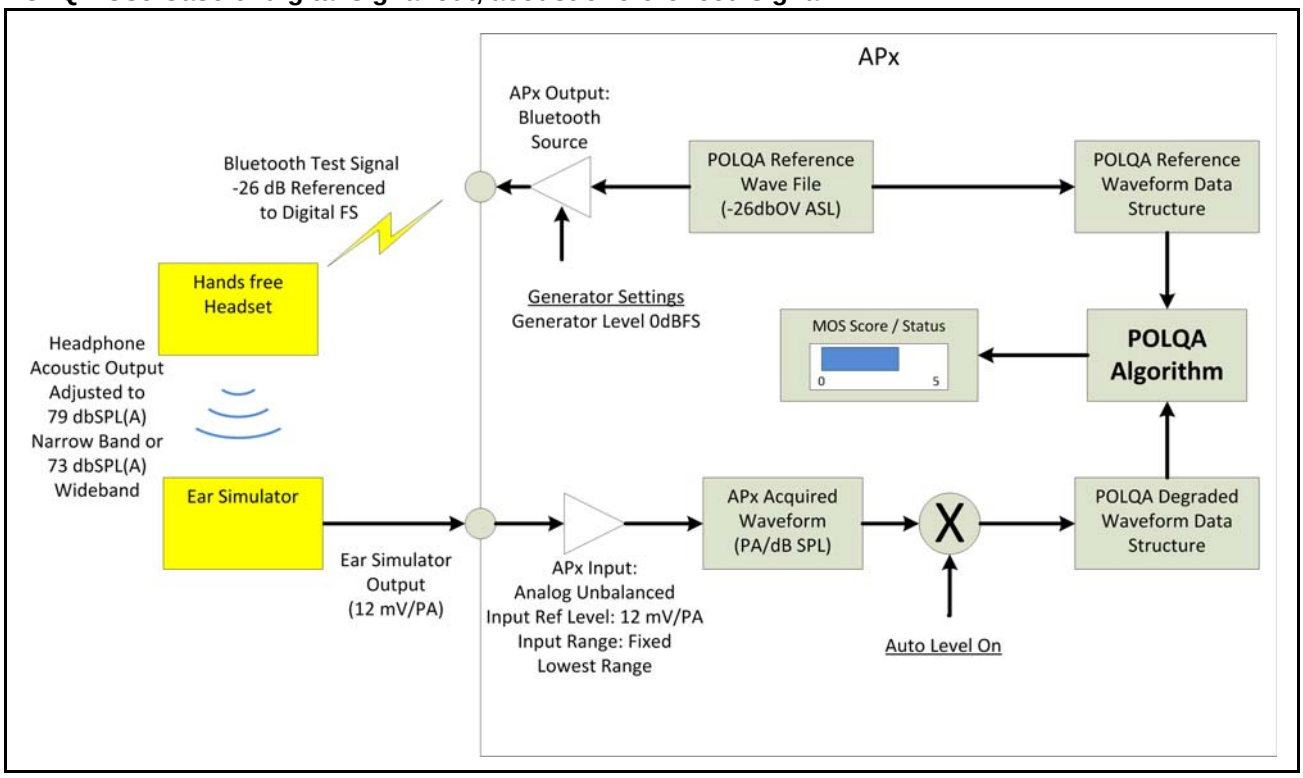
POLQA Use Case 4: analog voltage out, acoustic reference signal in



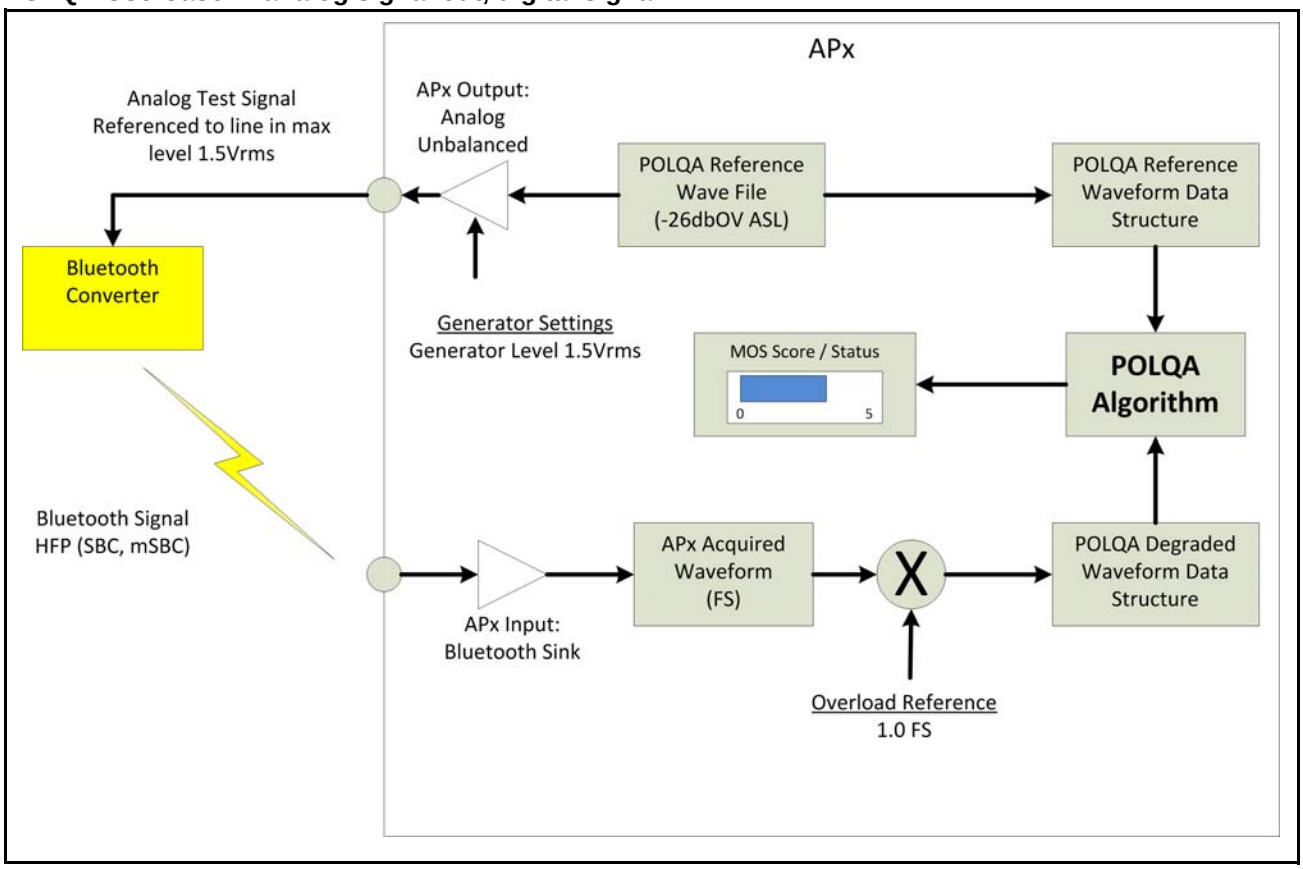
POLQA Use Case 5: acoustic referenced signal out and in



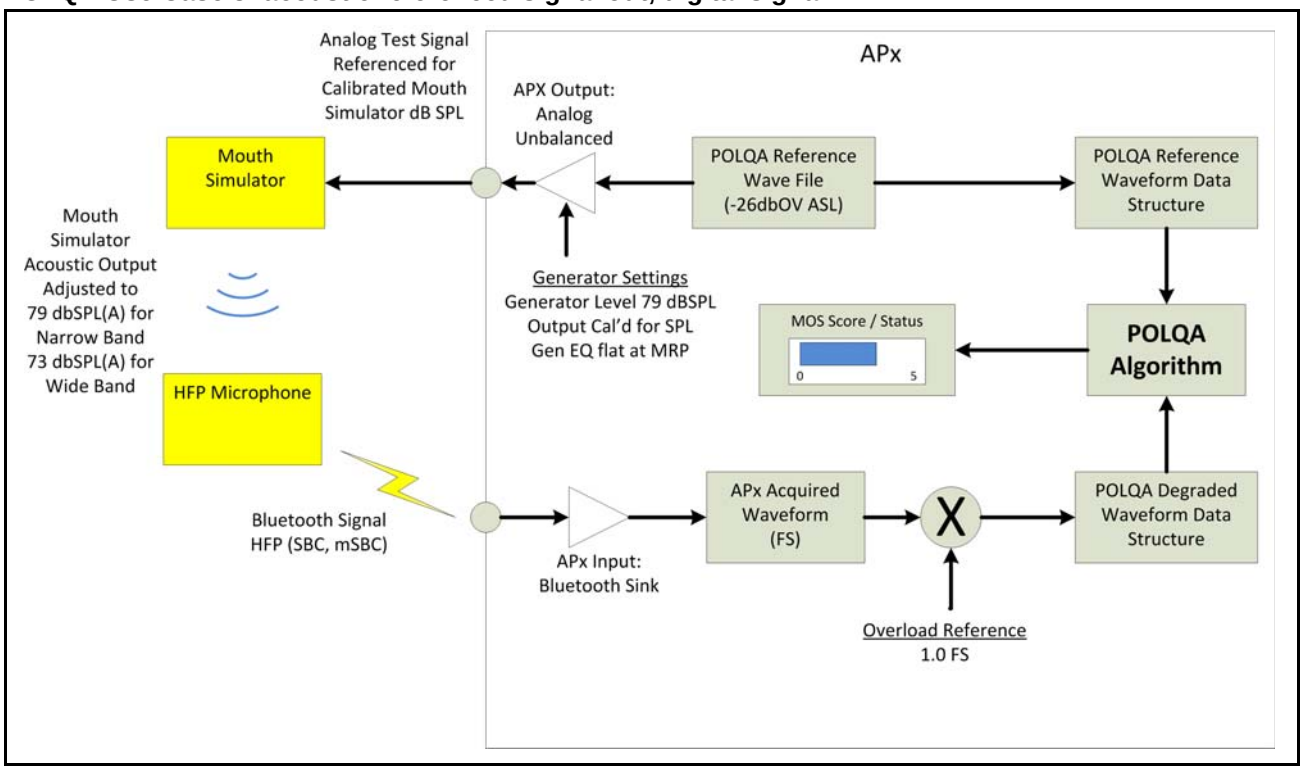
POLQA Use Case 6: digital signal out, acoustic referenced signal in



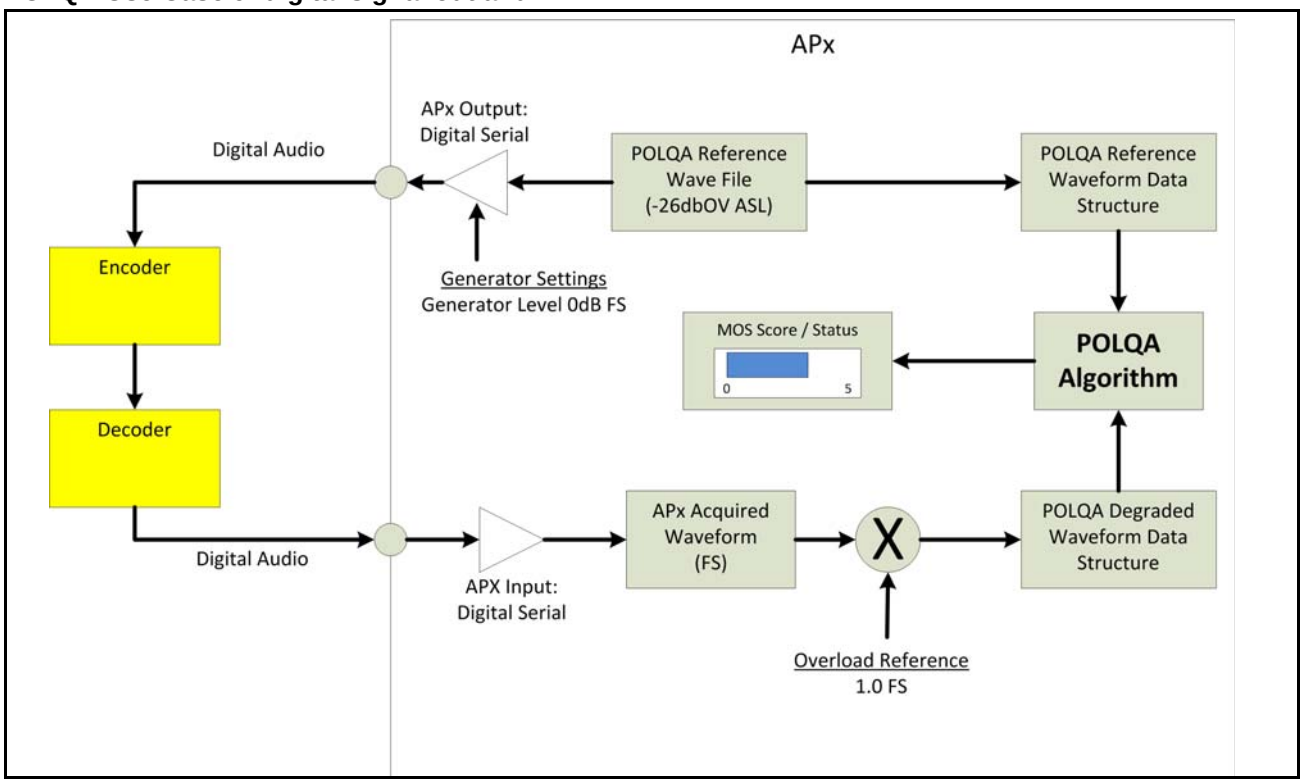
POLQA Use Case 7: analog signal out, digital signal in



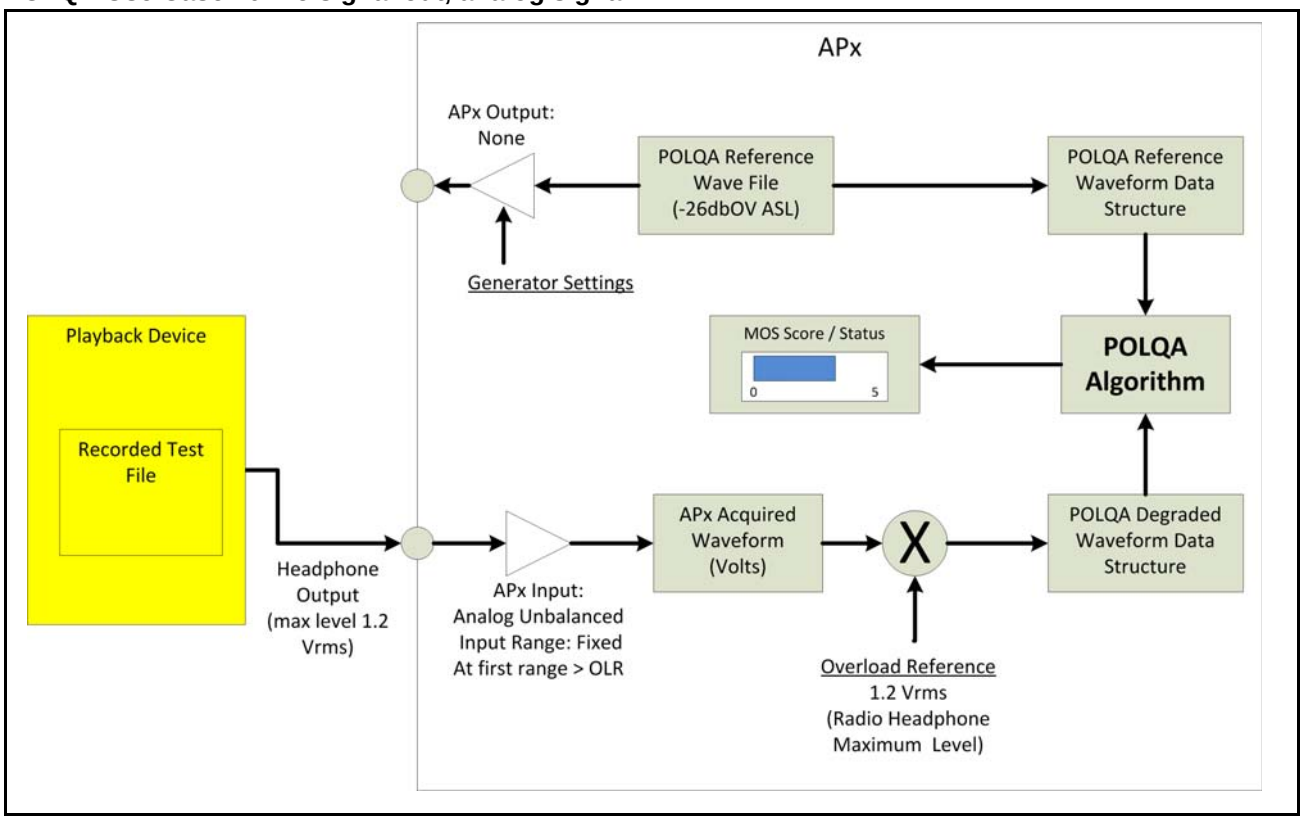
POLQA Use Case 8: acoustic referenced signal out, digital signal in



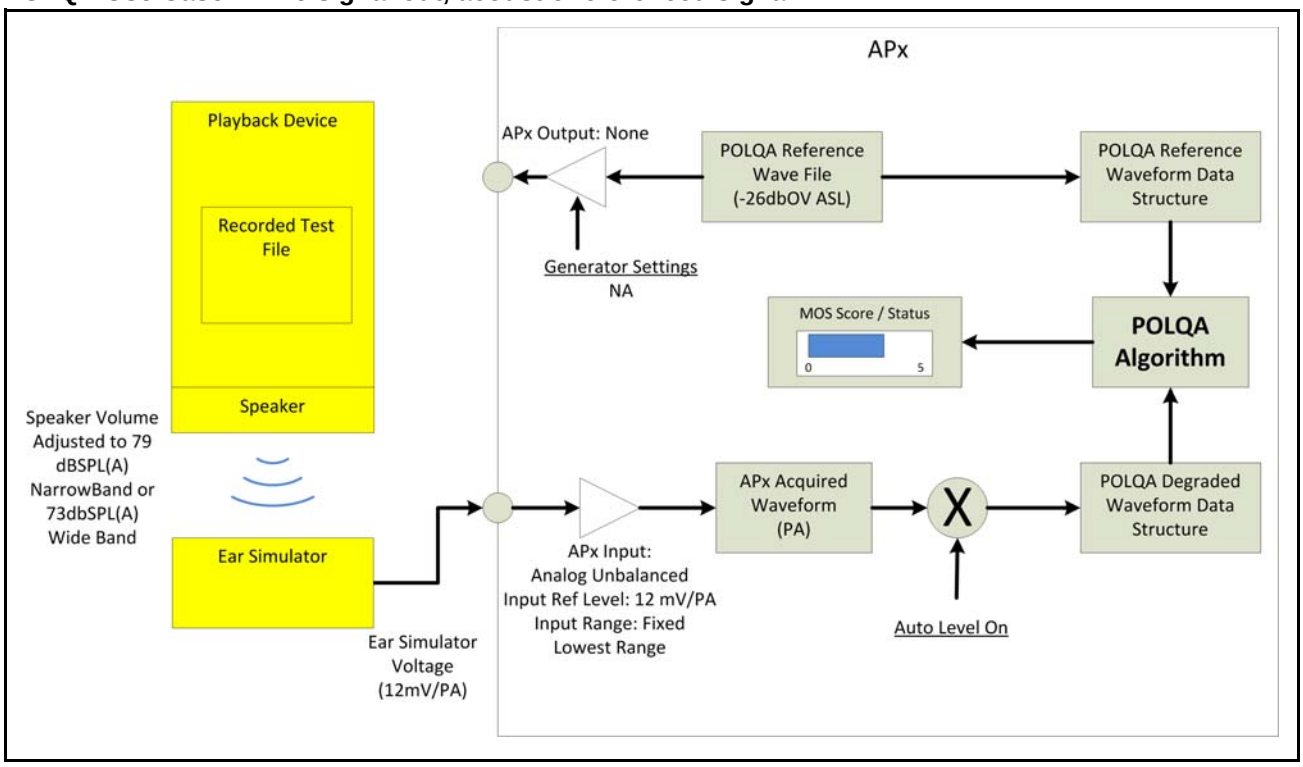
POLQA Use Case 9: digital signal out and in



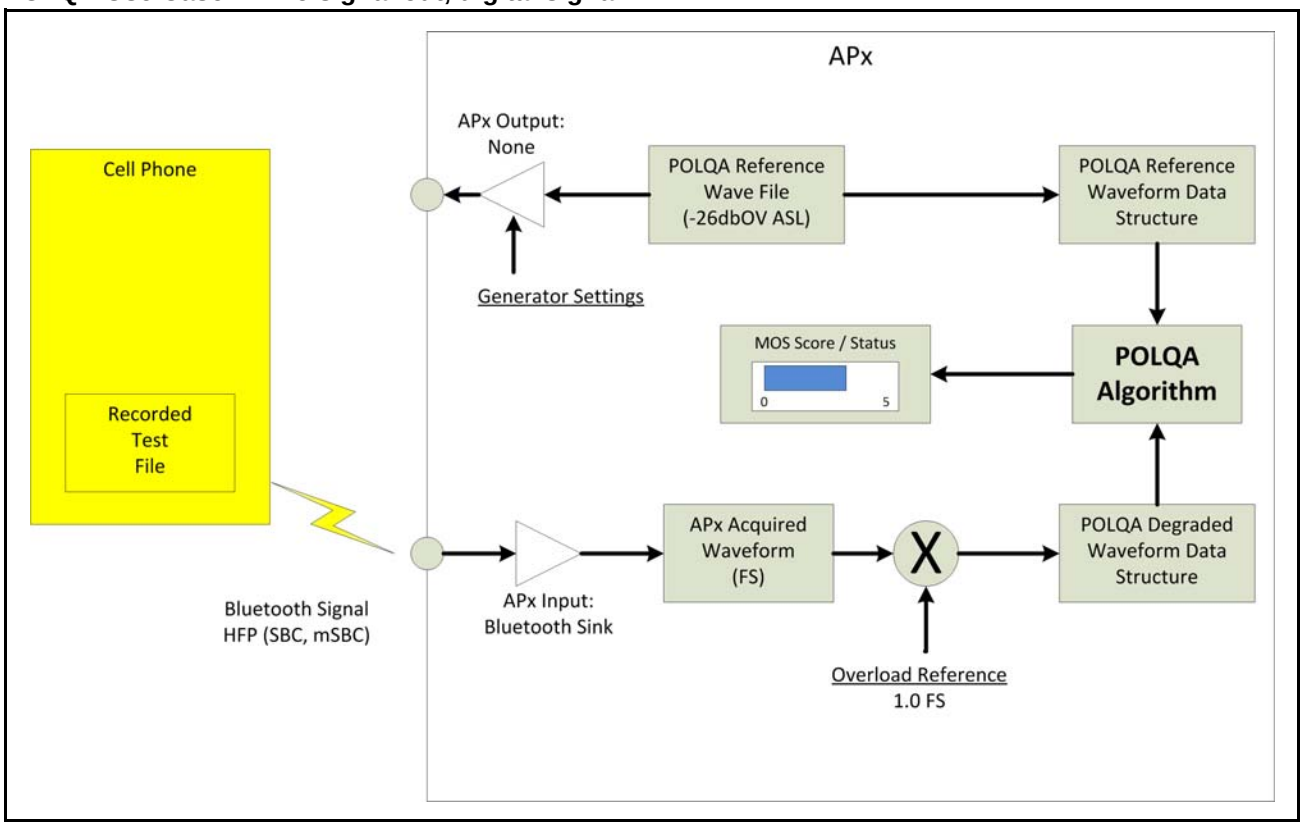
POLQA Use Case 10: no signal out, analog signal in



POLQA Use Case 11: no signal out, acoustic referenced signal in



POLQA Use Case 12: no signal out, digital signal in



POLQA (Averaged)
(Sequence Mode)

This measurement requires a software option key. See page 189 for more information about software options.

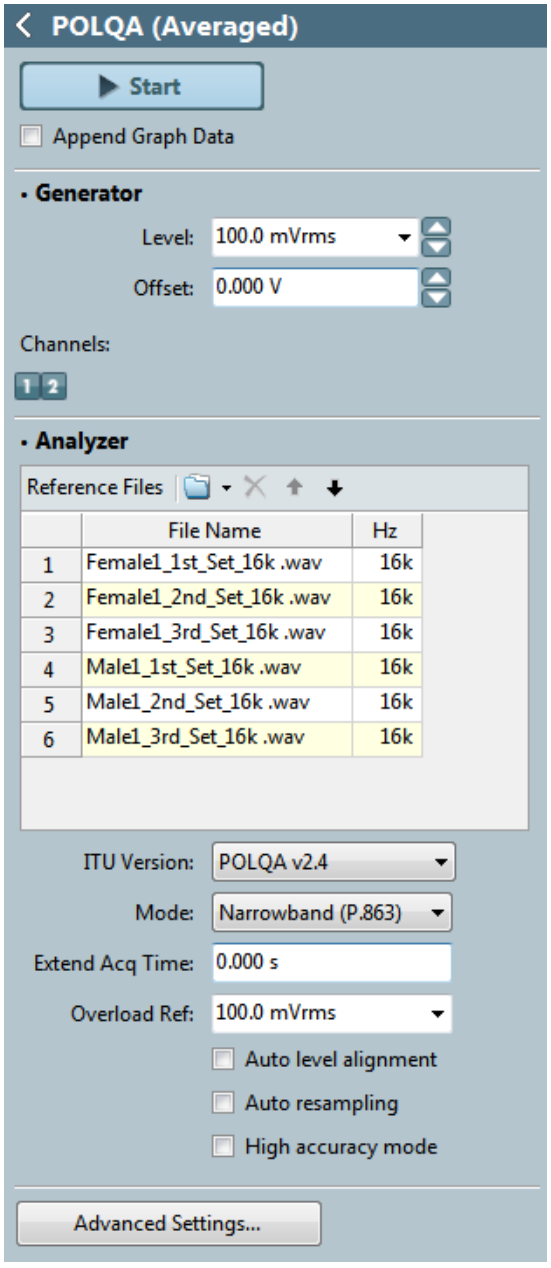
Note: POLQA measurements depend upon correct level settings throughout the signal path, and will provide inaccurate results when levels are not properly set. See Level alignment for POLQA on page 453 for detailed information.

The POLQA (Averaged) measurement is not available in **External Source** (open loop) configuration, or when Input Configuration is set to **File**.

See page 451 for more information about POLQA.

The POLQA (Averaged) measurement results available in APx500 are:

- MOS Average (LC IRS P.863)
- File MOS (LC IRS P.863)



Overview

POLQA (Perceptual Objective Listening Quality Assessment), like PESQ, is a method of grading the quality of an audio communications channel. Perceptual audio evaluation methods were developed to enable evaluation of audio quality in channels of compromised performance, where conventional measurements such as THD+N may provide results that are not useful.

This measurement provides a means to average a number of POLQA measurements, producing a single MOS Average result per channel. Best practice recommends at least two male and two female voice samples, each averaged several times. Some testers will use many different voice samples, repeated and averaged many times.

Operation

If you have not yet set up your test, first go to Chapter 7, Signal Path Setup.

Append Graph Data

Measured data are grouped in a Data Set (see page 189), which contains all acquisition results. Normally, the current graph data are deleted each time you start a new measurement. If the **Append Graph Data** box is checked, appended data are grouped as additional data sets. The Data Sets Panel (see page 649) displays the Data Set(s) in the measurement, and provides a visibility checkbox and a Notes field for each data set.

Generator

For POLQA, a speech sample used as a test stimulus is called the *reference signal*.

Level

Set the generator output level in this field.

For analog output, the file overload level (0 dBFS) is scaled to the value set in Level. A properly made POLQA speech signal will have an ASL of –26 dBov (equivalent to –26 dBFS), so the ASL in the analog output will be 26 dB below the value set in Level.

For digital output, the Level should be set to –0.000 dBFS to maintain correct ASL in the digital output stream.

See Level alignment for POLQA on page 453 for detailed information. See Generator Waveform Files on page 47 for a general discussion of generator level when playing waveform files.

Offset

Set an optional DC offset to the signal here.

Channels

The generator will output the speech samples to the DUT on the selected generator channels. Note that the Audio Precision POLQA-2 software option only enables POLQA analysis on 2 channels.

Start

When Signal Generation and Signal Analysis (below) are set and you have imported all your reference files, click Start. The files will be played through the DUT, one at a time.

Analyzer

For POLQA, the signal to be analyzed is called the *degraded signal*.

The SW-POLQA-2 option only provides POLQA analysis on input channels 1 and 2. To authorize more channels, contact your Audio Precision representative.

As the generator moves through the **Reference Files** list, the analyzer acquires the signal and matches the audio received to the reference files, one at a time.

The POLQA algorithm assigns a POLQA score, which is then mapped to a MOS score. A running average of the MOS scores is displayed, with the final value being the average of the scores of all the reference files.

Reference Files

For a POLQA measurement, a reference file in the APx is played through the generator, and the output of the DUT or communications channels under test is compared to the reference file. For multiple averaging iterations with the same file or with different files, browse to enter the files in the grid here. These reference files will be played and analyzed in the order shown in the grid. The minimum number of files is 2; the maximum number of files is 64.

Audio Precision provides 48 speech samples for POLQA usage, installed as .WAV waveforms during the installation of APx500. POLQA speech samples are installed at C:\Program Files (x86)\Audio Precision\APx500 [version]\Waveforms\POLQA\... Conforming waveforms are mono WAV files at a sample rate of 8 kHz, 16 kHz or 48 kHz. Content should be speech samples. Length is typically 5 to 10 seconds; maximum length is 12 seconds.

A conforming POLQA speech file is also filtered and optimized for an Active Speech Level (ASL) of -26 dBov ASL, using the ITU-T P.56 meter. See Level alignment for POLQA on page 453 for detailed information.

Use the folder icon to open a file browser. You can add files that are already in the project, or you can browse for files on disk. If you choose files of mixed sampling rates, the analysis **Mode** will automatically be set to the highest resolution mode found in the files. You can force the analysis mode to a lower resolution by making a selection in the **Mode** menu.

> *Note: the use a reference file whose sample rate differs from the target file is not standards compliant, and an error will be thrown. To suppress this error, set the Auto Resampling checkbox in Advanced Settings.*

Select a file to delete it, or to move it up or down in the list using the arrow buttons.

Filters

> *There are no local filters available for this measurement; the global filters in Signal Path Setup > Input/Output will affect this measurement, if set. Filter use in POLQA measurements may degrade MOS scores. Read more about APx Filters in Chapter 97.*

Mode

Conforming POLQA speech sample files have an 8 kHz sample rate for testing Narrowband (P.863) channels,

a 16 kHz sample rate for testing Wideband (P.863) channels, and a 48 kHz sample rate for testing Super-Wideband (P.863) channels. The mode is selected automatically by the POLQA measurement based on the sample rate contained in the reference file.

Extend Acq. Time:

The acquisition time is set to the length of the speech sample, selected in **Waveform**, plus the time set in this field. If you suspect you are getting poor results because the acquisition length is truncating the input signal, add time here.

Auto level alignment

A key difference between POLQA and PESQ is sensitivity to signal level. PESQ ignores level, but POLQA will produce poor MOS scores if the active speech level (ASL) presented to the algorithm is not matched (within a certain range) to the ASL of the reference file, which itself must be ASL-optimized for POLQA.

When **Auto level alignment** is **OFF** (the default), the **Overload Ref:** field is available. Enter the overload value of the DUT or system under test here to correctly align the ASL. For analog signals, the value set here is typically the clipping point of the analog DUT; for digital signals, the value is typically 0 dBFS. See Level alignment for POLQA on page 453.

APx provides an **Auto level alignment** checkbox to automatically align the input stream ASL to the level required, which makes good MOS scores easier to obtain. This is useful if your test requirements do not consider signal level, or if temporarily defeating level sensitivity is helpful in diagnosis. However, when **Auto level alignment** is **ON**, the APx POLQA measurement is not compliant with the ITU-T P.863 standard.

See Level alignment for POLQA on page 453 for information on adjusting levels to optimize MOS scores while remaining standards-compliant.

Warnings

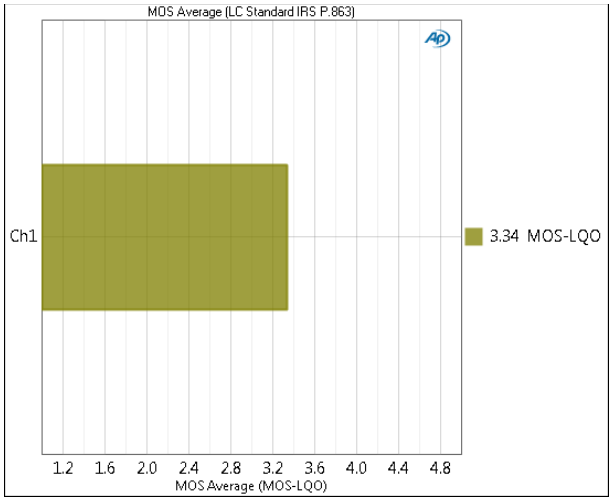
If the level of either the reference signal or the degraded signal is too high or too low, a warning symbol is displayed. Check the POLQA log file (see page 454) for level data.

If either signal is too long, a warning symbol is displayed. POLQA signals must be less than 12 seconds long.

Advanced Settings

If you'd like to adjust analyzer ranging parameters, click **Advanced Settings**. See Advanced Settings for POLQA on page 450. Read more about Units in Chapter 104.

POLQA (Averaged): MOS Average result



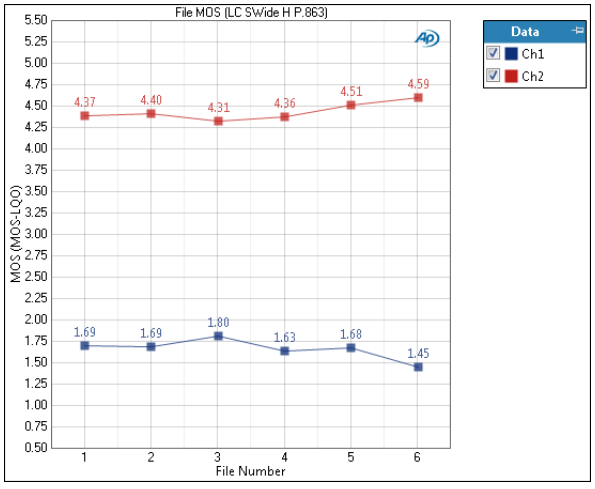
The **POLQA: MOS Average** (LC Standard IRS P.863) result is the primary result for the POLQA (Averaged) measurement. The POLQA score is mapped to MOS-LQO using LC Standard IRS P.863. This result provides an average of all the MOS scores of the speech samples in the Reference Files list.

Units

Units for POLQA (Averaged): MOS Average are:

- MOS-LQO

POLQA (Averaged): File MOS result



The POLQA (Averaged): File MOS (LC Standard IRS P.863) result is useful for cause analysis. The result shows the MOS score for each of the files in the Reference Files list.

By default, the graph display for this result has Show Markers: Y Labels enabled on the Graph Properties dialog. See page 646.

Units

Units for POLQA (Averaged): File MOS are:

X-Axis

- File number

Y-Axis

- MOS-LQO

PSR
(Sequence Mode)

Introduction

This measurement requires the PDM Option. See page 10 and Chapter 19 for more information.

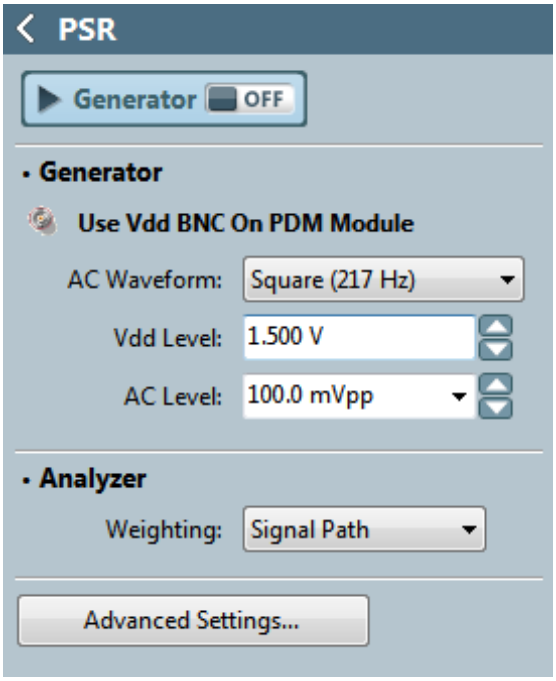
Overview

PSR is a power supply rejection measurement typically used in DUTs that are PDM transmitters. A small AC signal is imposed on the PDM Vdd power supply, and the residual of that signal is measured in the DUT output. PSR (or PSRR) can also be measured in other powered DUTs, such as converter or opamp devices, as long as the Vdd voltage and current limits of the PDM Vdd output are sufficient for the DUT. For DUTs that require more current, a unity gain DC coupled amplifier can be inserted between the APx generator and the DUT.

Unique Characteristics

The PSR and PSR Frequency Sweep measurements have some characteristics unique in APx500:

- **Availability**
  The PSR measurements are only available if the PDM Option is installed.
- **Vdd Supply**
  When the **Generator** in a PSR measurement is turned **On**, DC power is applied to the PDM **Vdd Supply** connector, regardless of the current Signal Path settings. When the **Generator** is turned **Off**, DC power at the PDM **Vdd Supply** connector reverts to the Signal Path setting.
- **APx Generator**
  For a PSR measurement, the output of the APx **Generator** is added to the DC on the **Vdd Supply** BNC power connector, and does not appear at any other output. The analyzer output connector as specified in Signal Path Setup is irrelevant. The generator signal is available to the PSR measurement, even if the **Output Connector** is set to **None**.



There are four results for PSR:

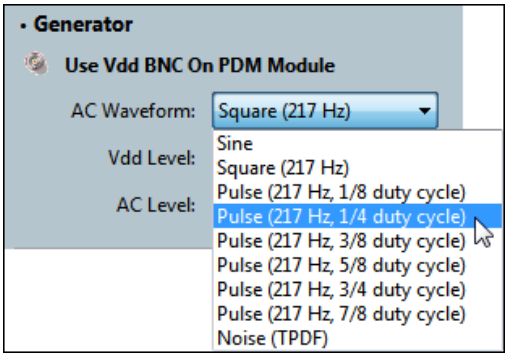
- PSR
- PSR+N
- PSRR (analog inputs only)
- PSRR+N (analog inputs only)

Generator

When this measurement is selected, the PDM **Vdd Supply** is switched to **Off**, if currently **On**.

Click the **Generator** button to toggle the **Generator** to **On**. The signal selected in **AC Waveform** is added to the DC available at the PDM **Vdd Supply** connector, and the **Vdd Supply** is switched to **On**.

AC Waveform



Choose the waveform to be added to the **Vdd Supply** voltage.

- Sine
- Square (217 Hz)
- Pulse (217 Hz, 1/8 duty cycle)
- Pulse (217 Hz, 1/4 duty cycle)
- Pulse (217 Hz, 3/8 duty cycle)
- Pulse (217 Hz, 5/8 duty cycle)
- Pulse (217 Hz, 3/4 duty cycle)
- Pulse (217 Hz, 7/8 duty cycle)
- Noise (TPDF)

The square wave and pulse choices simulate the digitally noisy environment of a GSM mobile phone. 216.667 Hz is approximately the frequency of the GSM frame rate, and the pulses simulate the noise with a variable number of time slots used, from 1 to 8.

The noise signal is a pseudo-random noise waveform having a triangular probability density function and a peak level set by the user.

Vdd Level

Set the (unmodulated) Vdd DC level here. This is the level before the AC signal is added. The range is 0 V to 3.6 V.

AC Level

Set the level of the AC level here. Maximum AC Level varies with Vdd Level. The range is constrained by the value set in Vdd Level, with the extremes being 10 mVpp to 720 mVpp

Frequency

This field is available if **AC Waveform** is set to **Sine**. Set the frequency of the sine wave here. The range is 10 Hz to 22 kHz.

Analyzer

Weighting Filter

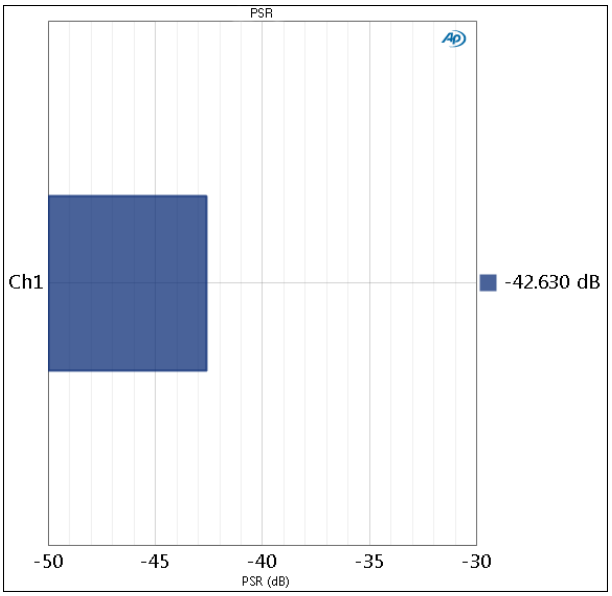
A local weighting filter is available for this measurement. See Measurement Filters on page 619 for detailed information about the filters available locally for some measurements.

Low pass, high pass and weighting Signal Path Filters settings (see page 617) made in Signal Path Setup > Input/Output will also affect this measurement. When set, the local low-pass or high-pass filter setting overrides the corresponding Signal Path filter setting while the measurement is active. A local weighting filter setting does not override any Signal Path filter settings; the local weighting filter is applied in addition to any active Signal Path filters.

Advanced Settings

If your test requires special adjustments or settings, click **Advanced Settings**. See Advanced Settings for PSR on page 470. See Chapter 104 for more information about units of measurement.

PSR: PSR Result



When **AC Waveform** is set to **Sine**, the PSR acquisition is filtered first by a weighting filter, if applied, then by a window width bandpass filter at the sine frequency. The PSR is the RMS level inside the bandpass filter.

When **AC Waveform** is set to **Square (217 Hz)**, the PSR acquisition is filtered first by a weighting filter, if applied, then by window width bandpass filters at the fundamental (217 Hz) and at the odd harmonics of 217 Hz, to the 101st harmonic. The PSR is the RMS sum of the levels inside the bandpass filters.

When **AC Waveform** is set to one of the **Pulse** choices, the PSR acquisition is filtered first by a weighting filter, if applied, then by window width bandpass filters at the fundamental (217 Hz) and at the even and odd harmonics of 217 Hz, to the 101st harmonic.

When **AC Waveform** is set to **Noise**, the PSR acquisition is filtered by a weighting filter, if applied. The PSR is the RMS level of the broadband result. For Noise, the PSR and PSR+N results are the same.

Note: the actual square wave frequency is 216.667 Hz. It is displayed nominally as 217 Hz.

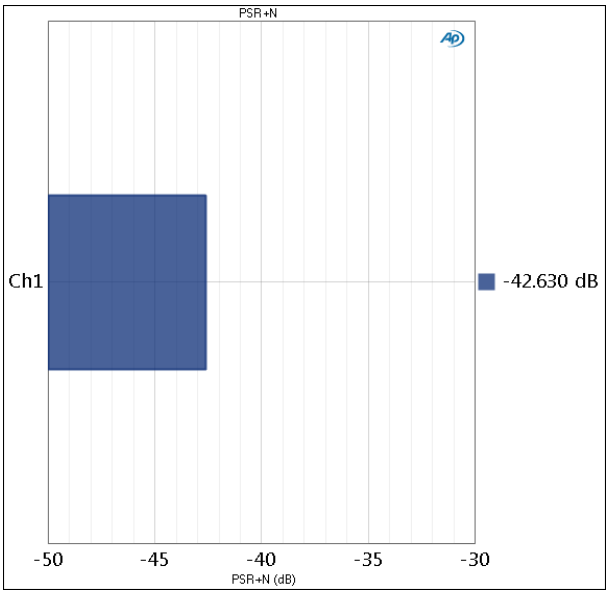
The use of window-width filters provides a low-noise estimate of the PSR, since the power outside the filters is rejected. This enables PSR measurements to be made on microphones which are not in an acoustically quiet environment.

Units

Units available for PSR results are

Analog Signals	*Digital Signals*
• Vrms	• dBFS
• dBV	• FS
• dBu	• %FS
• dBrA	• dBrA
• dBrB	• dBrB
• dBSPL1	• dBSPL1
• dBSPL2	• dBSPL2
• dBm	
• W (watts)	

PSR: PSR+N Result



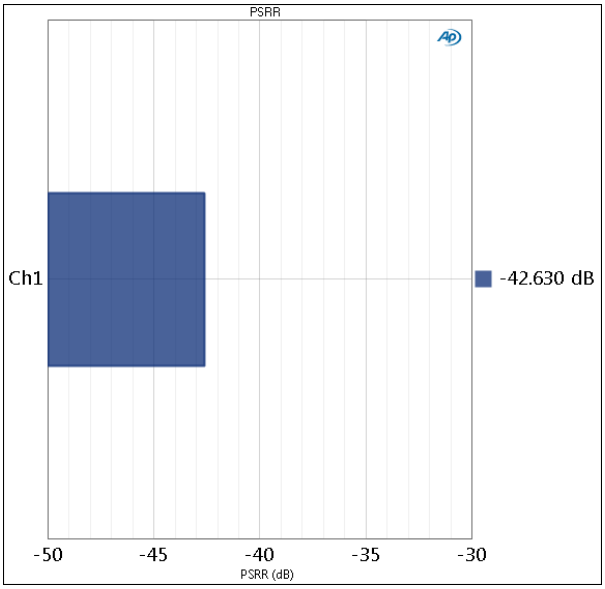
The PSR+N (Power Supply Rejection plus Noise) acquisition is filtered by a weighting filter, if applied. The PSR+N is the RMS level of the broadband result. For the Noise waveform, the PSR and PSR+N results are the same.

Units

Units available for PSR+N results are

Analog Signals	*Digital Signals*
• Vrms	• dBFS
• dBV	• FS
• dBu	• %FS
• dBrA	• dBrA
• dBrB	• dBrB
• dBSPL1	• dBSPL1
• dBSPL2	• dBSPL2
• dBm	
• W (watts)	

PSR: PSRR Result



PSRR is Power Supply Rejection Ratio. This nomenclature is used to describe the ratio of the level of a residual power supply perturbation found in the output signal, to the perturbation in the power supply. In APx, this result is only available when the analyzer input is analog.

When **AC Waveform** is set to **Sine**, the PSRR acquisition is filtered first by a weighting filter, if applied, then by a window width bandpass filter at the sine frequency. The PSRR is the ratio of the RMS level inside the bandpass filter to the RMS level of the AC Waveform.

When **AC Waveform** is set to **Square (217 Hz)**, the PSRR acquisition is filtered first by a weighting filter, if applied, then by window width bandpass filters at the fundamental (217 Hz) and at the odd harmonics of 217 Hz, to the 101st harmonic. The PSRR is the ratio of the RMS sum of the levels inside the bandpass filters to the RMS level of the AC Waveform.

When **AC Waveform** is set to one of the **Pulse** choices, the PSRR acquisition is filtered first by a weighting filter, if applied, then by window width bandpass filters at the fundamental (217 Hz) and at the even and odd harmonics of 217 Hz, to the 101st harmonic. The PSRR is the ratio of the RMS sum of the levels inside the bandpass filters to the RMS level of the AC Waveform.

When **AC Waveform** is set to **Noise**, the PSRR acquisition is filtered by a weighting filter, if applied. The PSRR is the ratio of the RMS level of the broadband result to the RMS level of the AC Waveform. For Noise, the PSRR and PSRR+N results are the same.

Note: the actual square wave frequency is 216.667 Hz. It is displayed nominally as 217 Hz.

The use of window-width filters provides a low-noise estimate of the PSRR, since the power outside the filters is rejected. This enables PSRR measurements to be made on microphones which are not in an acoustically quiet environment.

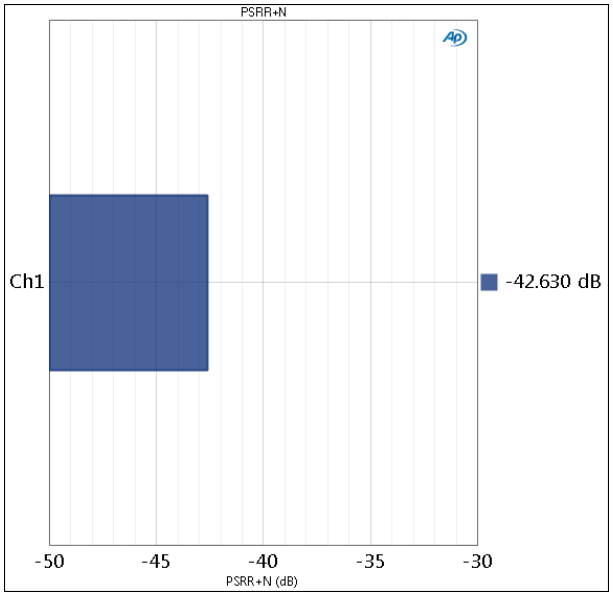
Units

Units available for PSRR results are

- x/y
- %
- ppm
- dB

Read more about Units in Chapter 104.

PSR: PSRR+N Result



PSRR is Power Supply Rejection Ratio plus Noise. This nomenclature is used to describe the ratio of the level of a residual power supply perturbation plus noise found in the output signal, to the AC signal that is the source of perturbation in the power supply. In APx, this result is only available when the analyzer input is analog.

The PSRR+N (Power Supply Rejection Ratio plus Noise) acquisition is filtered by a weighting filter, if applied. The PSRR+N is the ratio of the RMS level of the broadband result to the RMS level of the AC Waveform. For Noise, the PSRR and PSRR+N results are the same.

Units

Units available for PSRR+N results are

- x/y
- %
- ppm
- dB

Advanced Settings for PSR

This measurement requires the PDM option.

The default settings here are appropriate for measurements under most conditions. You may want to make minor adjustments for special situations.

Signal Generation

There are no advanced settings for signal generation for this measurement.

Signal Acquisition and Analysis

Input Range tab

Digital inputs do not require ranging, and the Input Range tab is unavailable when inputs are set to a digital format.

Auto Range

If **Auto Range** is checked (the default), each analog channel determines its input range automatically, based on the level of the input signal. If the input signal level changes beyond a ranging threshold, **Auto Range** will cause the input ranging circuits to move up or down for proper ranging.

Fixed Range

Alternatively, you can set a fixed minimum range for the analog inputs. Uncheck **Auto Range** and enter the expected maximum level in the **Minimum Range** filed. The software will choose the lowest range that will accommodate that level; this becomes the minimum range available, and the analyzer will not reset to a lower range. Fixed range is best for signals that vary rapidly, including tone bursts, perceptual audio speech samples, and program material such as voice or music. See **When autoranging isn't suitable** on page 626.

If a signal greater then the **Minimum Range** is presented to the analog input, the ranging circuits will range up to accommodate the higher level, and then drop again when the signal level falls. The range will not be set below the **Minimum Range**.

Ranges Track Ch1

If **Ranges Track Ch1** is checked (the default), the fixed input range setting for channel 1 is copied to the other channels, and range settings for channels greater than 1 cannot be edited. Any changes made to channel 1 are reproduced in the other channels. If **Ranges Track Ch1** is not checked, the fixed input range setting for each channel can be set independently.

Settling Tab

A Settling tab is available for these measurement results.

Read more about ranging and settling in Chapter 98.

PSR Frequency Sweep
(Sequence Mode)

Introduction

This measurement requires the PDM Option. See page 10 and Chapter 19 for more information.

Overview

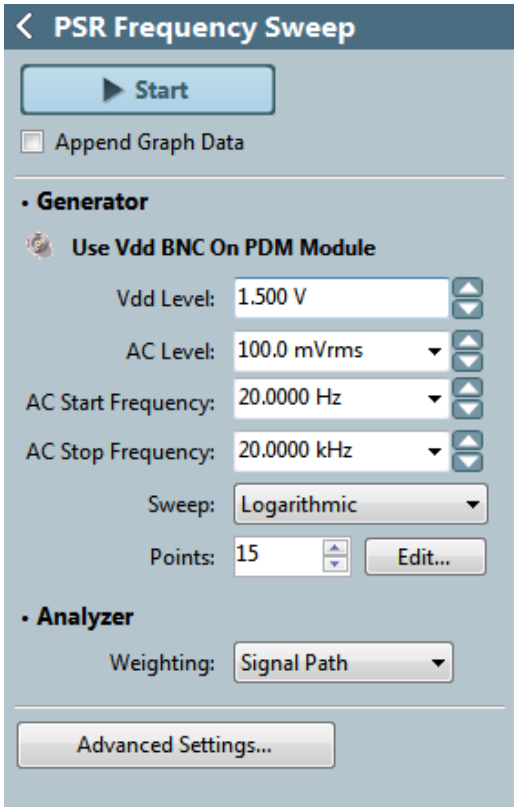
PSR is a power supply rejection measurement typically used in DUTs that are PDM transmitters. A small AC signal is imposed on the PDM Vdd power supply, and the residual of that signal is measured in the DUT output. PSR (or PSRR) can also be measured in other powered DUTs, such as converter or opamp devices, as long as the Vdd voltage and current limits of the PDM Vdd output are sufficient for the DUT.

For PSR Frequency Sweep, the AC signal is a sine wave, swept across a range of frequencies.

Unique Characteristics

The PSR and PSR Frequency Sweep measurements have some characteristics unique in APx500:

- **Availability**
  The PSR measurements are only available if the PDM Option is installed.
- **Vdd Supply**
  When the **Generator** in a PSR measurement is turned **On**, DC power is applied to the PDM **Vdd Supply** connector, regardless of the current Signal Path settings. When the **Generator** is turned **Off**, DC power at the PDM **Vdd Supply** connector reverts to the Signal Path setting.
- **APx Generator**
  For a PSR measurement, the output of the APx **Generator** is added to the DC on the **Vdd Supply** BNC power connector, and does not appear at any other output. The analyzer output connector as specified in Signal Path Setup is irrelevant. The generator signal is available to the PSR measurement, even if the **Output Connector** is set to **None**.



There are four results for PSR Frequency Sweep:

- PSR
- PSR+N
- PSRR (analog inputs only)
- PSRR+N (analog inputs only)

When this measurement is selected, the PDM **Vdd Supply** is switched to **Off**, if currently **On**.

Start

For a default sweep of 20 kHz to 20 Hz in 31 logarithmic steps, simply click **Start**. If you want to specify other parameters for your sweep, review **Signal Gener-**

ation and **Signal Acquisition and Analysis** settings, discussed below. Click **Start** when you have approved the settings.

Append Graph Data

Normally, the current graph data for all results is deleted each time you start a new measurement. If the **Append** box is checked, the current data is kept in memory as a **Data Set** (see page 189), and the new measurement data is appended as a new data set. You can append many **Data Sets**.

Generator

When this measurement is selected, the PDM **Vdd Supply** is switched to **Off**, if currently **On**.

Vdd Level

Set the (unmodulated) Vdd DC level here. This is the level before the AC signal is added. The range is 0 V to 3.6 V.

AC Level

Set the level of the AC level here. The range is constrained by the value set in Vdd Level, with the extremes being 1 mVpp to 254.6 mVpp.

AC Start Frequency

Set the sweep start frequency here. The range is 10 Hz to 22 kHz.

AC Stop Frequency

Set the sweep stop frequency here. The range is 10 Hz to 22 kHz.

Sweep

Choose

- **Logarithmic** point spacing
- **Linear** point spacing
- **Custom**
  Edit sweep spacing and number of **Points** to create a **Custom** sweep.

Points

Set the number of sweep points here.

Points > Edit

Open the **Sweep Points** dialog to edit, import or export the **Sweep Points** table.

Step Size

For linear sweeps, enter the **Step Size** here.

Level

Set the signal level here.

Analyzer

Filters

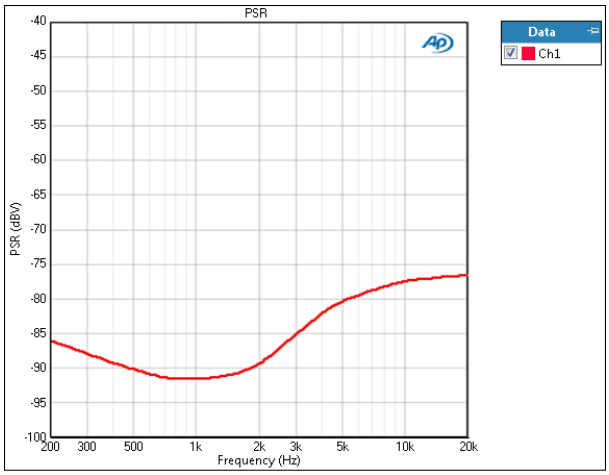
A local weighting filter is available for this measurement. See Measurement Filters on page 619 for detailed information about the filters available locally for some measurements.

Low pass, high pass and weighting Signal Path Filters settings (see page 617) made in Signal Path Setup > Input/Output will also affect this measurement. When set, the local low-pass or high-pass filter setting overrides the corresponding Signal Path filter setting while the measurement is active. A local weighting filter setting does not override any Signal Path filter settings; the local weighting filter is applied in addition to any active Signal Path filters.

Advanced Settings

If your test requires special adjustments or settings, click **Advanced Settings**. See Advanced Settings for PSR on page 470. See Chapter 104 for more information about units of measurement.

PSR Frequency Sweep: PSR Result



The PSR acquisition is filtered first by a weighting filter, if applied, then by a window-width bandpass filter that is swept at the sine frequency. The PSR is the RMS level inside the bandpass filter, plotted against frequency.

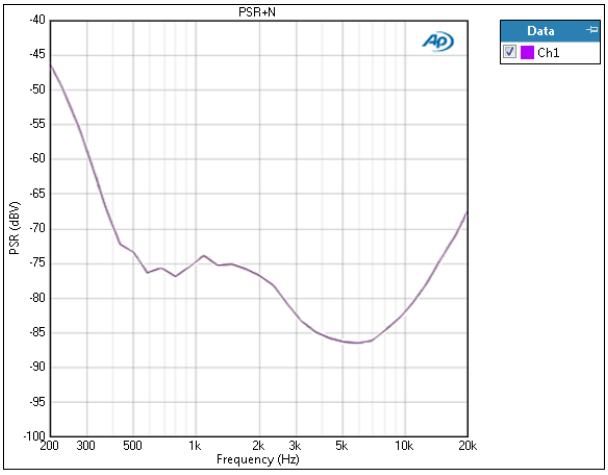
The use of window-width filters provides a low-noise estimate of the PSR, since the power outside the filters is rejected. This enables PSR measurements to be made on microphones which are not in an acoustically quiet environment.

Units

Units available for PSR results are

X-axis	*Y-axis (analog)*	*Y-axis (digital)*
• Hz	• Vrms	• dBFS
• dHz	• dBV	• FS
• F/R	• dBu	• %FS
• %Hz	• dBrA	• dBrA
	• dBrB	• dBrB
	• dBSPL1	• dBSPL1
	• dBSPL2	• dBSPL2
	• dBm	
	• W (watts)	

PSR Frequency Sweep: PSR+N Result



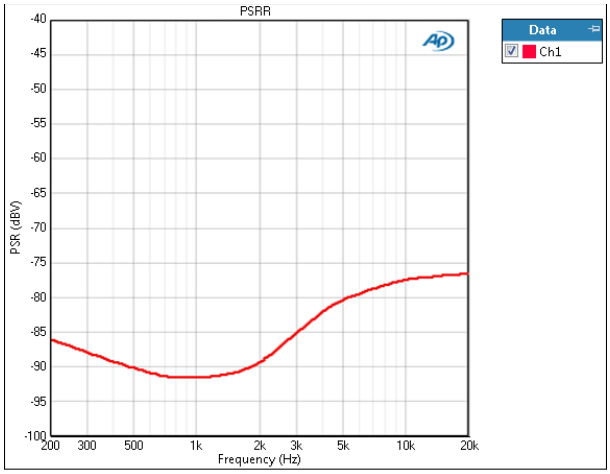
The PSR+N (Power Supply Rejection plus Noise) acquisition is filtered by a weighting filter, if applied. The PSR+N is the RMS level of the broadband result, plotted against frequency.

Units

Units available for PSR results are

X-axis	*Y-axis (analog)*	*Y-axis (digital)*
• Hz	• Vrms	• dBFS
• dHz	• dBV	• FS
• F/R	• dBu	• %FS
• %Hz	• dBrA	• dBrA
	• dBrB	• dBrB
	• dBSPL1	• dBSPL1
	• dBSPL2	• dBSPL2
	• dBm	
	• W (watts)	

PSR Frequency Sweep: PSRR Result



PSRR is Power Supply Rejection Ratio. This nomenclature is used to describe the ratio of the level of a residual power supply perturbation found in the output signal, to the perturbation in the power supply. In APx, this result is only available when the analyzer input is analog.

The PSRR acquisition is filtered first by a weighting filter, if applied, then by a window-width bandpass filter that is swept at the sine frequency. The PSRR is the ratio of the RMS level inside the bandpass filter to the RMS level of the perturbation in the power supply, plotted against frequency.

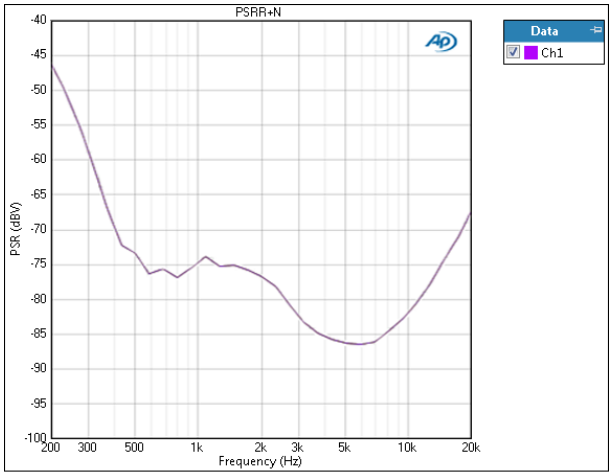
The use of window-width filters provides a low-noise estimate of the PSRR, since the power outside the filters is rejected. This enables PSRR measurements to be made on microphones which are not in an acoustically quiet environment.

Units

Units available for PSRR results are

X-axis	*Y-axis*
• Hz	• x/y
• dHz	• %
• F/R	• ppm
• %Hz	• dB

PSR Frequency Sweep: PSRR+N Result



PSRR+N is Power Supply Rejection Ratio plus Noise. This nomenclature is used to describe the ratio of the level of a residual power supply perturbation plus noise found in the output signal, to the AC signal that is the source of perturbation in the power supply. In APx, this result is only available when the analyzer input is analog.

The PSRR+N (Power Supply Rejection Ratio plus Noise) acquisition is filtered by a weighting filter, if applied. The PSRR+N is ratio of the RMS level of the broadband result to the AC signal that is the source of the perturbation in the power supply, plotted against frequency.

Units

Units available for PSRR+N results are

X-axis	*Y-axis*
• Hz	• x/y
• dHz	• %
• F/R	• ppm
• %Hz	• dB

Regulated Frequency Sweep
(Sequence Mode)

Introduction

For the APx515, this measurement requires a software option key. See page 189 for more information about software options.

Regulated Frequency Sweeps use a sine wave stimulus signal that is moved across a range of frequencies in a specified number of points. At each point, the generator level is adjusted to the level that produces the target value in the selected channel. Go to page 481 for more information about regulation.

Regulated Frequency Sweeps are not available in External Source configuration.

Regulating THD+N is often employed to plot an amplifier frequency sweep at the amplifier's maximum output level.

Regulated Frequency Sweep results available in APx500 are:

- Level (frequency response)
- THD+N Level
- THD+N Ratio
- Generator Level

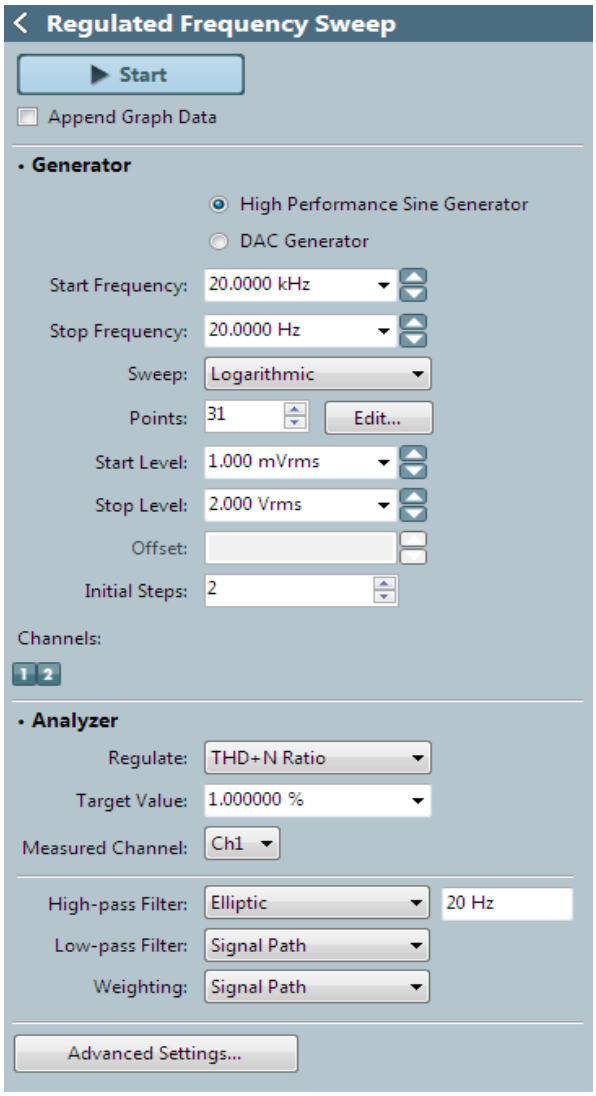
All of these results are available from a single measurement.

Operation

If you have not yet set up your test, first go to Signal Path Setup (Chapter 7).

Append Graph Data

Measured data are grouped in a **Data Set**, which contains all acquisition results. Normally, the current graph data are deleted each time you start a new measurement. If the **Append Graph Data** box is checked, appended data are grouped as additional data sets, as are any imported data. The Data Sets Panel displays the Data Set(s) in the measurement, and provides a visibility checkbox, a Time acquisition field, and a Notes field for each data set.



Read more about Data Sets beginning on page 649.

Generator

Regulated Frequency Sweep only supports a sine waveform in a closed-loop configuration.

To make a regulated frequency sweep measurement, first define the sweep by selecting a sweep **Start Frequency**, **Stop Frequency**, the number of sweep points and **Sweep** type.

High Performance Sine Generator / DAC Generator

For an APx555, you have the option of using the High Performance Sine Generator (very low distortion sine waves and wide bandwidth) or the DAC generator, which provides special sine waveforms. See High Performance Sine Generator starting on page 51 for a comparison of features.

Start Frequency

Set the sweep start frequency here.

Stop Frequency

Set the sweep stop frequency here.

Sweep

Choose

- **Logarithmic** (logarithmic point spacing)
- **Linear** (linear point spacing)
- **Custom**
  Edit sweep spacing and number of **Points** to create a **Custom** sweep.

Points

Set the number of sweep points here. The default is 31 points.

Points > Edit

Open the **Sweep Points** dialog to edit, import or export the **Sweep Points** table.

Step Size

For linear sweeps, enter the **Step Size** here.

Start, Stop and Points for a stepped frequency sweep

The sweep begins at the **Start Frequency** and ends at the **Stop Frequency**. By default, these are set at 20 kHz and 20 Hz, and the sweep moves in logarithmic step points from high frequency to low.

You can enter **Start** and **Stop** frequencies of your choice, and sweep from low to high if you want. The sweep moves in a specified number of points, set in the Points field. The minimum is 2 points; maximum is 65,535. The default setting is 31, which corresponds to 1/3 octave per point in a logarithmic sweep across 20 Hz to 20 kHz. By default, the sweep is logarithmic. You can also select a linear sweep, in which case the **Step Size** filed becomes available.

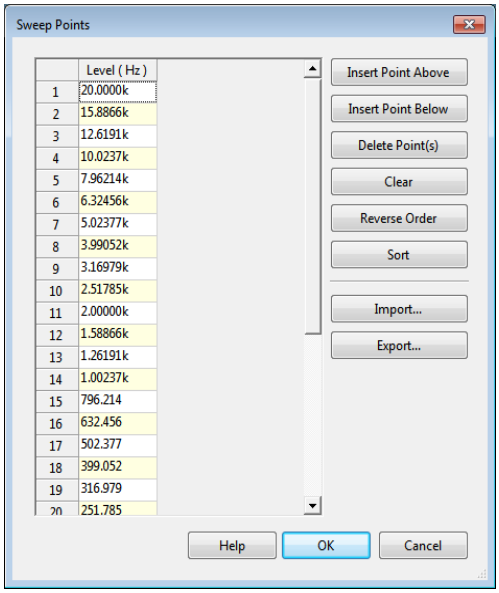
The sweep point spacing is set by selecting one of the following choices in the Sweep field:

- **Logarithmic** (the default); use the **Points** field to set the number of logarithmically spaced points;
- **Linear**, which provides two methods of adjusting spacing: the **Points** field or the **Step Size** field; or
- **Custom**. Click **Edit** to open the **Sweep Points** dialog, where you can set points arbitrarily, or load or save sweep table files.

Viewing or Editing the Sweep Points table

Click Edit to open the Sweep Points table. This table shows each point and its corresponding frequency. You can edit this table to add or delete points or to change the frequency of a point. Points can be sorted or reversed in order using the controls on the right.

A Sweep Points table can be saved as a *.csv file or as a Microsoft Excel *.xls file. A compatible *.csv or *.xls file can be opened and used as a Sweep Points table.



Start Level

The **Start Level** is the first generator setting where a measurement is made for each frequency point. If it is lower than the **Stop Level**, then it is also the lowest generator setting where the measurement is made. If it is higher than the **Stop Level**, then it is the highest generator setting where the measurement is made.

The regulation search proceeds from Start Level to Stop Level; your choices for these values determine the direction (up or down) of the search.

Stop Level

If the **Stop Level** is higher than the **Start Level**, then the **Stop Level** is the highest generator setting where a measurement is made for each frequency point. If the **Stop Level** is lower than the **Start Level**, then the **Stop Level** is the lowest generator setting where a measurement is made. The **Stop Level** is also the last generator setting where a measurement is made in the initial

search for an interval containing a measured value of the type that is sought.

The higher of Start Level or Stop Level also serves to limit the maximum APx generator output, protecting your DUT.

Offset

For the DAC generator, you can enter an optional DC offset here.

Initial Steps

Initial Steps sets the number of intervals dividing the generator level range between **Start Level** and **Stop Level**. The default setting is **2**. Depending upon the parameter to be regulated and characteristics of the DUT, greater values may provide better performance.

Click **Start**. The generator will output a sine wave to the DUT on the selected generator channels at the frequency set in the Signal Generation panel. At the first sweep point the software regulates the generator level to the **Target** value. When the result for the first point is acquired, the sweep proceeds to the next point, regulating the generator at each point until the sweep is completed.

Since each step is regulated with settling, this sweep will proceed more slowly than an unregulated frequency sweep.

Analyzer

Regulate

This control selects the parameter in the DUT output you want to regulate. Choose

- **RMS Level** or
- **THD+N Ratio**

Target Value

Enter the value you want to attain here.

Measured Channel

Choose the input channel to measure for regulation here.

Filters

Local high-pass, low-pass and weighting filters are available for this measurement. The low-pass and high-pass filters, if selected, are applied to the entire measured signal and affect all the results. The weighting filter, if selected, affects only the distortion and distortion plus noise results. See Measurement Filters on page 619 for detailed information about the filters available locally for some measurements.

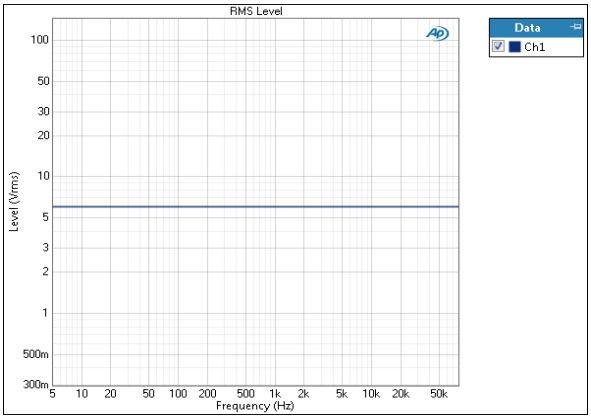
Low pass, high pass and weighting Signal Path Filters settings (see page 617) made in Signal Path Setup > Input/Output will also affect this measurement. When set, the local low-pass or high-pass filter setting overrides the corresponding Signal Path filter setting while the measurement is active. A local weighting filter setting does not override any Signal Path filter settings; the local weighting filter is applied in addition to any active Signal Path filters.

Read more about regulation on page 481 for a detailed discussion of these methods.

Advanced Settings

If your test requires individual generator settings for each channel or special analyzer ranging or settling, see Advanced Settings for Stepped Sweeps on page 508. See Chapter 104 for more information about units of measurement.

Regulated Frequency Sweep: Level



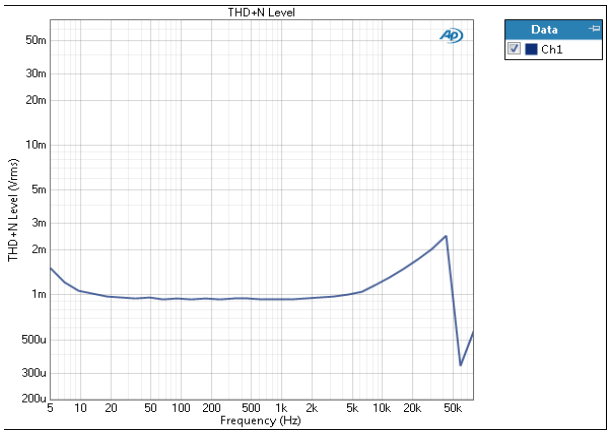
The Level result plots the DUT measured output level on the Y axis, against the generator frequency on the X axis.

Units

Units available for Level are:

X-axis	*Y-axis (analog)*	*Y-axis (digital)*
• Hz	• Vrms	• dBFS
• dHz	• dBV	• FS
• F/R	• dBu	• %FS
• %Hz	• dBrA	• dBrA
	• dBrB	• dBrB
	• dBSPL1	• dBSPL1
	• dBSPL2	• dBSPL2
	• dBm	
	• W (watts)	

Regulated Frequency Sweep: THD+N Level



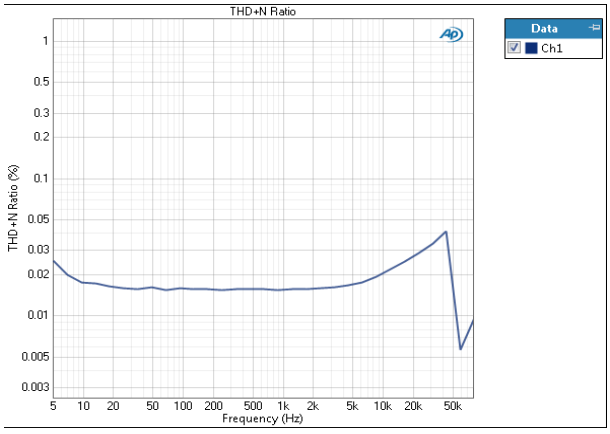
The THD+N Level result plots the DUT measured THD+N output level on the Y axis, against the generator frequency on the X axis. This method is often employed to plot an amplifier frequency sweep at the amplifier's maximum output level.

Units

Units available for THD+N Level are:

X-axis	*Y-axis (analog)*	*Y-axis (digital)*
• Hz	• Vrms	• dBFS
• dHz	• dBV	• FS
• F/R	• dBu	• %FS
• %Hz	• dBrA	• dBrA
	• dBrB	• dBrB
	• dBSPL1	• dBSPL1
	• dBSPL2	• dBSPL2
	• dBm	
	• W (watts)	

Regulated Frequency Sweep: THD+N Ratio



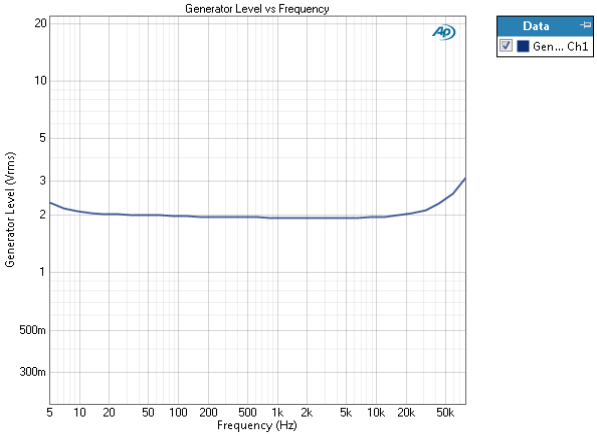
The THD+N Ratio result plots the DUT measured THD+N output Ratio on the Y axis, against the generator frequency on the X axis. This method is often employed to plot an amplifier frequency sweep at the amplifier's maximum output level.

Units

Units available for THD+N Ratio are:

X-axis	*Y-axis*
• Hz	• x/y
• dHz	• %
• F/R	• ppm
• %Hz	• dB

Regulated Frequency Sweep: Generator Level



The Level result plots the generator output level on the Y axis, against the generator frequency on the X axis.

Units

Units available for Generator Level results are:

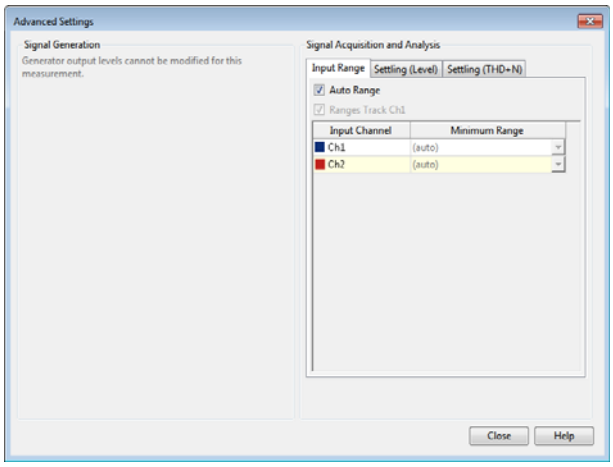
X-axis	*Y-axis (analog)*	*Y-axis (digital)*
• Hz	• Vrms	• FS
• dHz	• Vp	• %FS
• F/R	• Vpp	• dBFS
• %Hz	• dBV	• dBrG
	• dBu	
	• dBrG	
	• dBm	
	• W (watts)	

Y-axis (acoustic output configuration)

- Pa
- dBSPL

Advanced Settings for Regulated Measurements

The default settings here are appropriate for most measurements. You may want to make minor adjustments for special situations.



Signal Generation

There are no advanced settings for signal generation for regulated sweep measurements.

Signal Acquisition and Analysis

Input Range tab

Digital inputs do not require ranging, and the Input Range tab is unavailable when inputs are set to a digital format.

Auto Range

If **Auto Range** is checked (the default), each analog channel determines its input range automatically, based on the level of the input signal. If the input signal level changes beyond a ranging threshold, **Auto Range** will cause the input ranging circuits to move up or down for proper ranging.

Fixed Range

Alternatively, you can set a fixed minimum range for the analog inputs. Uncheck **Auto Range** and enter the expected maximum level in the **Minimum Range** filed. The software will choose the lowest range that will accommodate that level; this becomes the minimum range available, and the analyzer will not reset to a lower range. Fixed range is best for signals that vary rapidly, including tone bursts, perceptual audio speech samples, and program material such as voice or music. See **When autoranging isn't suitable** on page 626.

If a signal greater then the **Minimum Range** is presented to the analog input, the ranging circuits will range up to accommodate the higher level, and then drop again when the signal level falls. The range will not be set below the **Minimum Range**.

Ranges Track Ch1

If **Ranges Track Ch1** is checked (the default), the fixed input range setting for channel 1 is copied to the other channels, and range settings for channels greater than 1 cannot be edited. Any changes made to channel 1 are reproduced in the other channels. If **Ranges Track Ch1** is not checked, the fixed input range setting for each channel can be set independently.

Read more about ranging in Chapter 98.

Settling tab

No settling is available for Maximum Output (CEA-2006) measurements.

Two Settling tabs are available for regulated measurements, Settling (Level) and Settling (THD+N).

See Chapter 98 for more information about Settling for regulated measurements.

More about regulation

Overview

In APx, regulation is the mechanism of varying one parameter (usually generator output level) to produce a specific value in the DUT response. Most often, this is a minimum level or THD+N ratio, a maximum level or THD+N ratio, or a selected target level or THD+N ratio. In some cases, other parameters can be regulated.

Regulation is useful to automatically search out desired test settings. For example, perhaps a test requirement is that an amplifier's distortion is to be measured at a 2 volt rms output level. The regulated **Auto Gen Level** feature can automatically determine the AP generator level to produce this output level.

In another example, it may be a test goal to perform a frequency sweep on an amplifier, where each step of the sweep is at the amplifier's nominal maximum output. If the maximum output is defined at 1% THD+N, then, at each step of the frequency sweep, the stimulus level is adjusted up or down until the target distortion is reached.

Regulation is used in several places in the APx500 software:

- **Auto Gen Level** (both Sequence Mode and Bench Mode)
- **Set Acoustic Level** (both Sequence Mode and Bench Mode)
- **Regulated Frequency Sweep** measurement (Sequence Mode)
- **Maximum Output** measurement (Sequence Mode)
- **Maximum Output CEA-2006** measurement (Sequence Mode)

- The **Regulate** button available on the Bench Mode Monitors/Meters panel
- Bench Mode Sweep when **Regulate each point** is selected.

APx regulation method

APx regulates by sequentially applying two algorithms:

1. The first algorithm locates an interval between the **Stop** and **Start** values where the stimulus signal results in a measured value of the desired type (maximum, minimum or a target value).

   a. The user defines a range for the adjusted parameter, entering a **Start Value** and a **Stop Value**. These values should encompass the range of stimulus values that the user expects will result in a measured value of the desired type in the DUT. The start value may be higher or lower than the **Stop Value**; the regulation search proceeds from **Start** to **Stop**, so the choice of these values determines the search direction.

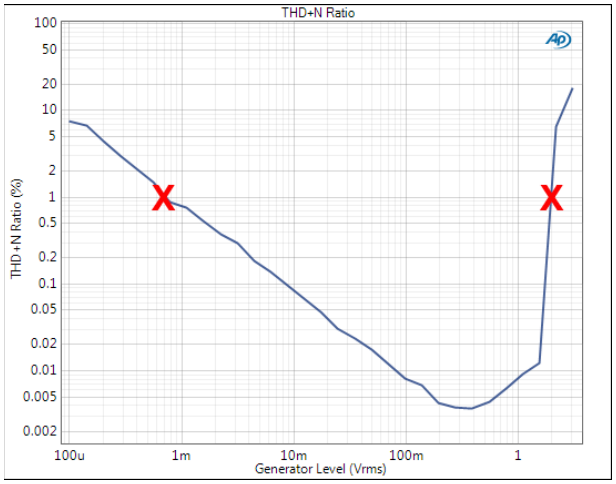
   b. The **Start-Stop** range is divided into interval steps, with the default number of initial steps set at 2. Beginning at **Start**, the regulation search moves toward the **Stop** value, examining each interval until it finds an interval containing the type of value sought. If regulation is seeking a specific value of the measured quantity, this means finding an interval with the measured value above the target value at one endpoint, and the measured value below the target value at the other endpoint.
2. The interval identified above is passed to the second stage of regulation, where a second algorithm quickly determines a stimulus value that results in the a measured value of the type that is sought.

Recommendations for successful regulation

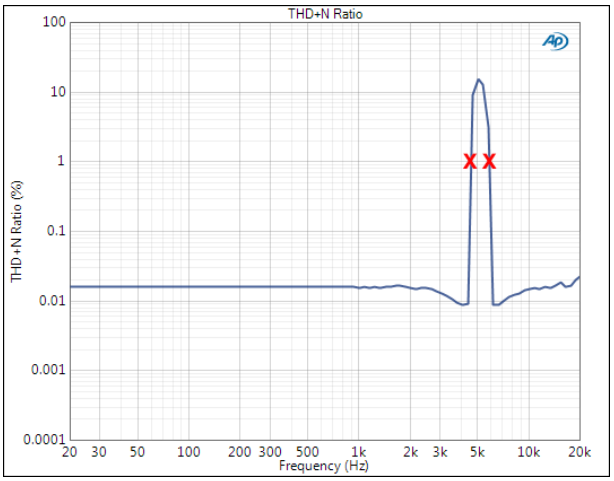
You should have some knowledge of the characteristics of your DUT before you attempt regulation, especially an estimate of what range of stimulus signals might produce the target response you are looking for. A THD+N Ratio result may have two points, for example, that will produce a target THD+N Ratio result, as shown in this illustration. It may be useful to make a Stepped Level Sweep to see where these points fall, and set your **Start** and **Stop Values** to exclude the unwanted **Target** point, and to choose the direction of the search.

Don't set **Start** at zero (0), which will cause a regulation failure.

If regulation fails before a complete search is finished in the first stage of regulation (above), or if it finishes successfully but the final measured value is far from the desired value, increasing the amount of settling applied to the measurements could improve the measurements and might allow regulation to terminate successfully close to the desired value.



THD+N Ratio vs. Stimulus Level



THD+N Ratio vs. Stimulus Frequency

Why use more steps?

For **Target Values** that may only exist for a very narrow range of stimulus values, using two regulation steps may entirely miss the points where the target level is attained, as shown in the first illustration. Two steps would not find this distortion peak, but 10 steps would.

Alternatively, you may want to step upward towards the expected target value in small steps, to avoid accidentally driving the DUT to high range of output.

Why use fewer steps?

More steps make for slower regulation, which is multiplied in a Regulated Frequency Sweep, where each frequency step is regulated. For production test, we recommend that the DUTs in production be well characterized in development, so the that any regulation procedures can be optimized for fast testing.

Resistance (DCX)
(Sequence Mode)

DCX measurements require that a DCX-127 Multifunction Module be connected to the APx analyzer system, typically using an Audio Precision USB/APIB adapter connected to the PC.

The DCX-127 brings additional measurement and control features to an Audio Precision analyzer. The APx Resistance (DCX) measurement uses the DCX DMM (Digital Multimeter) mode to implement a precise ohmmeter. Download the DCX-127 Multifunction Module User's Guide from the Audio Precision Web site at ap.com for complete operational information for the DCX-127.

Operation

If you have not yet set up your test, first go to Signal Path Setup to select and configure inputs and outputs, references and other settings. See Chapter 7 for information about Signal Path Setup.

The DCX-127 must be powered on and connected to the measurement PC using an Audio Precision USB/APIB adapter.

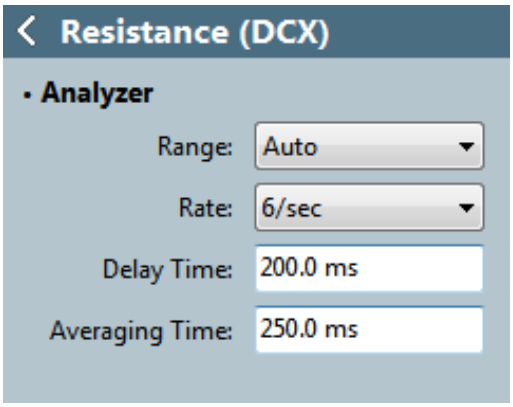
Selecting the Resistance (DCX) measurement places the DCX in "DMM" mode. Note that you can configure the DCX DC Outputs or Auxiliary outputs in Signal Path Setup > DCX, independent of any settings or operation in the Resistance (DCX) measurement.

Generator

No Generator functions are provided for the Resistance (DCX) measurement.

Analyzer

The DCX has a digital multimeter (DMM) that is configured as an ohmmeter for this measurement. The APx analyzer is not used; all the data comes from the DCX.



Range

Select a fixed range for the Resistance measurement, or select **Auto**.

- Auto
- 2 MOhm
- 200 kOhm
- 20 kOhm
- 2 kOhm
- 200 Ohm

Rate

The Rate control allows you to select one of two meter reading rates.

- **6/sec**
  Approximately 6 meter readings are taken per second.
- **25/sec**
  Approximately 25 meter readings are taken per second.

At 6/sec, DCX meter resolution is a full 4 1/2 digits. At 25/sec, the resolution is reduced. The number of digits displayed is the same but the least significant digit is always 5 or 0.

Delay Time

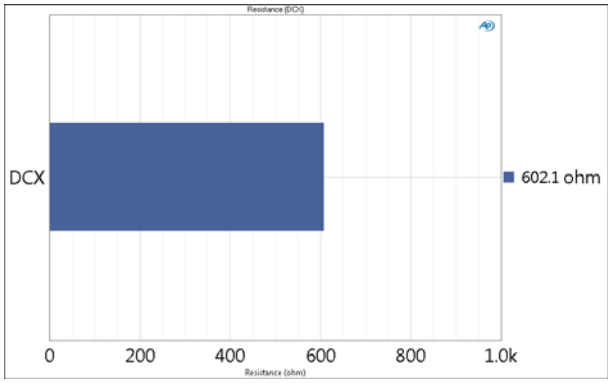
A Delay Time interval is started when the Resistance (DCX) measurement is started within a sequence. APx ignores DCX data until the time set in Delay Time has passed. Minimum Delay Time is 200 ms, maximum is 10 s, 200 ms is the default.

Averaging Time

APx performs a rolling average of the DCX readings across a specified time before display. Minimum Averaging Time is 200 ms, maximum is 10 s, 250 ms is the default.

See Chapter 104 for more information about units of measurement.

Resistance (DCX)



Units

Units available for Resistance (DCX) are

- ohm

Signal Acquisition
(Sequence Mode)

Introduction

Signal Acquisition is a general purpose tool that makes an audio acquisition (or a series of appended or nested audio acquisitions) that can be saved with the project. The only result is the signal acquisition. No analysis is done.

Operation

If you have not yet set up your test, first go to Signal Path Setup (Chapter 7). If you are configured for External Source, see the topic below.

Append Graph Data

Measured data are grouped in a **Data Set**, which contains all acquisition results. Normally, the current graph data are deleted each time you start a new measurement. If the **Append Graph Data** box is checked, appended data are grouped as additional data sets, as are any imported data. The Data Sets Panel displays the Data Set(s) in the measurement, and provides a visibility checkbox, a Time acquisition field, and a Notes field for each data set.

Read more about Data Sets beginning on page 649.

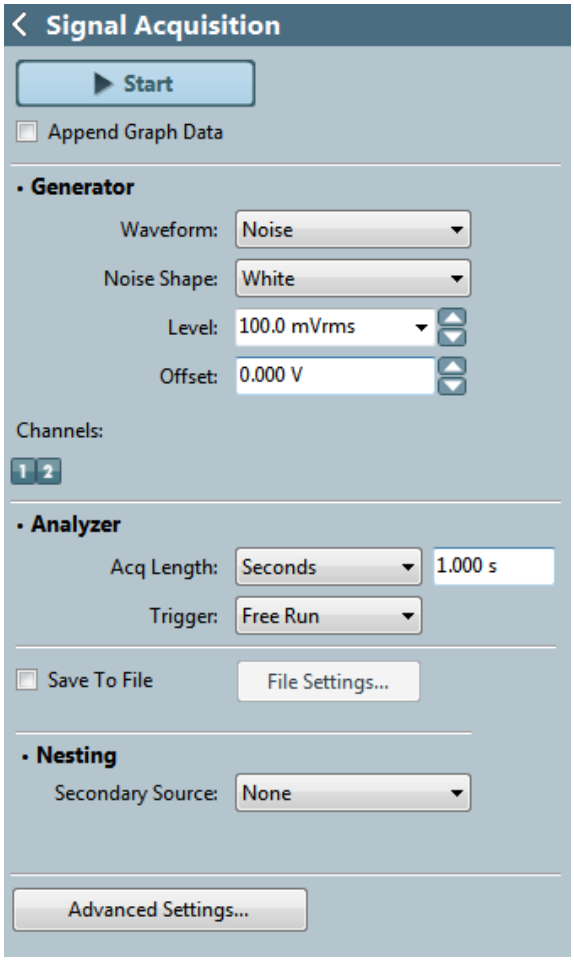
Generator

The typical configuration for this measurement is the closed-loop configuration, using the APx generator as a stimulus. Alternatively, you can make open-loop measurements, using either the External Source or File Input configuration.

See Chapter 6 for detailed information about the using the APx Generator and making Waveform, Level and Frequency settings, or for information about using External Source.

High Performance Sine Generator / DAC Generator

For an APx555, you have the option of using the High Performance Sine Generator (very low distortion sine waves and wide bandwidth) or the DAC generator, which provides special sine waveforms. See High Performance Sine Generator starting on page 51 for a comparison of features.



Running the measurement

Click **Start**. The generator will output the test signal to the DUT on the selected generator channels. In a short period of time, the signal will be acquired and the waveform displayed.

If you would like to view the DUT output using no stimulus signal (for a view of DUT noise, for example), set the Level in Signal Generation to zero. Alternatively, you can use the Signal Analyzer in External Source configuration.

Analyzer

Filters

No local low pass, high pass or weighting filters are available for this measurement. However, low pass, high pass and weighting filter settings made in Signal Path Setup > Input/Output will affect this measurement. Read more about APx Filters in Chapter 97.

Acq Length

Set the acquisition length here. Use the drop-down menu to choose seconds or samples.

Trigger

- **Free Run**
  When trigger is set to **Free Run**, the acquisition begins when the **Start** button is clicked, plus any time or sample count set in **Delay**.
- **+0 Crossing**
  When trigger is set to **+0 Crossing**, the acquisition begins at the input waveform's first positive-going zero crossing after any time or sample count set in **Delay**, for the input channel selected.
- **Generator**
  - When **Trigger** is set to **Generator** and **Generator Waveform** is set to **Sine (High Performance Sine Generator)**, the acquisition begins at the waveform's first positive peak plus any time or sample count set in **Delay**. For the High Performance Sine Generator, if the **Delay** time is set in samples it is scaled to the input sample rate.
  - When **Trigger** is set to **Generator** and **Generator Waveform** is set to **Sine (DAC)**, the acquisition begins at the waveform's first positive-going zero crossing plus any time or sample count set in **Delay**.
  - When **Trigger** is set to **Generator** and **Generator Waveform** is set to **Square**, the acquisition begins at the waveform's first rising edge plus any time or sample count set in **Delay**.
  - When **Trigger** is set to **Generator** and **Generator Waveform** is set to **Noise**, the acquisition will not trigger.
  - When **Trigger** is set to **Generator** and **Generator Waveform** is set to **File**, the acquisition begins at the file's first sample, plus any time or sample count set in **Delay**. A trigger is generated each time the file loops back to its first sample.
- **External**
  External is only available for analyzers equipped with the Advanced Master Clock. The acquisition begins at the first trigger pulse received at the rear-panel **Trigger In** BNC connector plus any time or sample count set in **Delay**. Set the **Trigger Input** in **Signal Path Setup > Triggers** to the desired voltage threshold and pulse edge.

Signal

The **Signal** field is only available when **Input Connector** in **Signal Path Setup** is set to **PDM**.

- **Decimated Audio**
  The decimated PDM audio (at the baseband sample rate and bandwidth) is the input signal.
- **PDM Bitstream**
  The PDM bitstream (at the PDM bit clock rate and bandwidth) is the input signal. This is useful to view the out-of-band noise and other components of the PDM bitstream.

Note: When the PDM input decimation is set to x1, (see "Settings for PDM Input" on page 159) the PDM signal is not decimated. In that case, either of the Signal selections in the Signal Analyzer will present the undecimated PDM bitstream for the analyzer acquisition.

Save To File

You can record the audio signal to a computer file as the signal is being acquired. Click **Save to File** before you run Signal Acquisition. See **Recording Audio to a File** on page 381.

File Settings

When saving an audio signal to a file, click **File Settings** to specify the file name and location.

Nesting

Signal Acquisition records can be nested. See Nested Sweeps on page 185.

Secondary Source

Select the secondary sweep source here. This is the outside, nesting parameter that is varied to change the conditions for the recording through a number of iterations. Available choices are dependent upon instrument hardware, options and Input/Output settings.

Advanced Settings

If your test requires individual generator settings for each channel or special analyzer ranging, delay time or bandwidth settings, click **Advanced Settings**.

Signal Analyzer
(Sequence Mode)

Introduction

For the APx515, this measurement requires a software option key. See page 189 for more information about software options.

The Signal Analyzer is a general-purpose diagnostic tool that provides a time-domain Scope result view, an FFT Spectrum result view and noise density views.

Signal Analyzer results available in APx500 are:

- Scope
- FFT Spectrum
- Amplitude Spectral Density
- Power Spectral Density

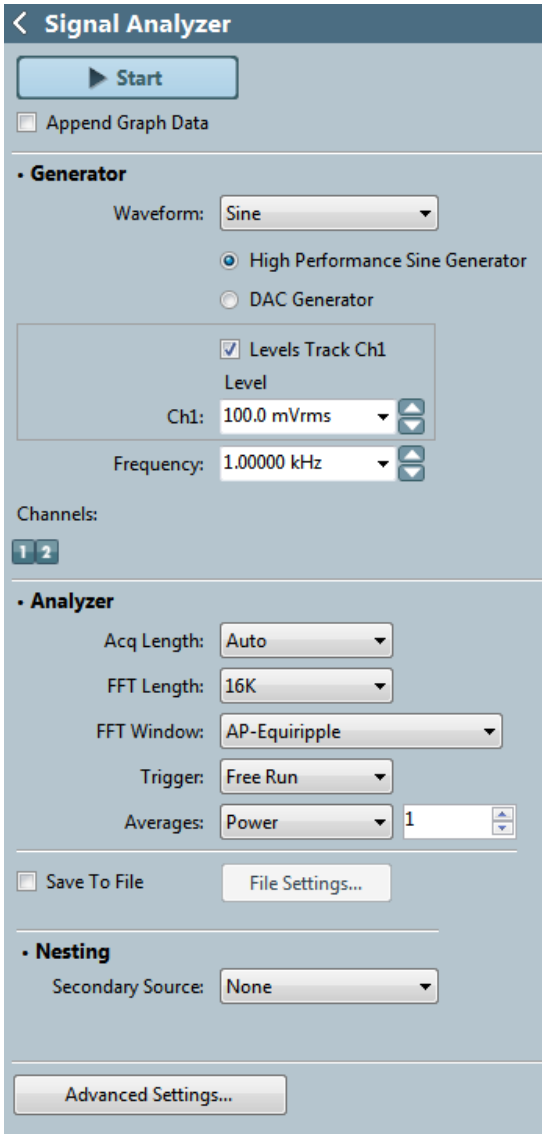
Operation

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Append Graph Data

Measured data are grouped in a **Data Set**, which contains all acquisition results. Normally, the current graph data are deleted each time you start a new measurement. If the **Append Graph Data** box is checked, appended data are grouped as additional data sets, as are any imported data. The Data Sets Panel displays the Data Set(s) in the measurement, and provides a visibility checkbox, a Time acquisition field, and a Notes field for each data set.

Read more about Data Sets beginning on page 649.



Generator

The typical configuration for this measurement is the closed-loop configuration, using the APx generator as a stimulus. Alternatively, you can make open-loop measurements, using either the External Source or File Input configuration.

See Chapter 6 for detailed information about the using the APx Generator and making Waveform, Level and Frequency settings, or for information about using External Source.

High Performance Sine Generator / DAC Generator

For an APx555, you have the option of using the High Performance Sine Generator (very low distortion sine waves and wide bandwidth) or the DAC generator, which provides special sine waveforms. See High Performance Sine Generator starting on page 51 for a comparison of features.

Running the measurement

Click **Start**. The generator will output the test signal to the DUT on the selected generator channels. In a short period of time, the signal will be acquired and processed, and the two small graphs in the Selector will be populated with results.

If you would like to view the DUT output using no stimulus signal (for a view of DUT noise, for example), set the Level in Signal Generation to zero. Alternatively, you can use the Signal Analyzer in External Source configuration.

Analyzer

No local low pass, high pass or weighting filters are available for this measurement. However, low pass, high pass and weighting filter settings made in Signal Path Setup > Input/Output will affect this measurement. Read more about APx Filters in Chapter 97.

Acq Length

Set the acquisition length here, in seconds or in samples. **Auto**, the default, sets the acquisition length to the FFT length.

FFT Length

Set the FFT record length here. Options from 256 points to 1.2M points are available, depending upon the number of input channels in use; default is 16K points.

See “More About FFTs” on page 492.

FFT Window

FFT acquisitions must either be synchronous or have one of a number of amplitude windows applied to provide useful data for interpretation. Each window function brings advantages and disadvantages. The default selection, AP-Equiripple, is a proprietary Audio Precision FFT window that is an excellent choice for most FFT results.

The APx500 Signal Analyzer FFT window choices are:

- None
- Hann
- Blackman-Harris 3-term
- Blackman-Harris 4-term
- Flat Top
- AP-Equiripple

Detailed descriptions of the FFT window functions begin on page 493.

Trigger

- **Free Run**
  When trigger is set to **Free Run**, the acquisition begins when the **Start** button is clicked, plus any time or sample count set in **Delay**.
- **+0 Crossing**
  When trigger is set to **+0 Crossing**, the acquisition begins at the input waveform’s first positive-going zero crossing after any time or sample count set in **Delay**, for the input channel selected.
- **Generator**
  - When **Trigger** is set to **Generator** and **Generator Waveform** is set to **Sine (High Performance Sine Generator)**, the acquisition begins at the waveform’s first positive peak plus any time or sample count set in **Delay**. For the High Performance Sine Generator, if the **Delay** time is set in samples it is scaled to the input sample rate.
  - When **Trigger** is set to **Generator** and **Generator Waveform** is set to **Sine (DAC)**, the acquisition begins at the waveform’s first positive-going zero crossing plus any time or sample count set in **Delay**.
  - When **Trigger** is set to **Generator** and **Generator Waveform** is set to **Square**, the acquisition begins at the waveform’s first rising edge plus any time or sample count set in **Delay**.
  - When **Trigger** is set to **Generator** and **Generator Waveform** is set to **Noise**, the acquisition will not trigger.
  - When **Trigger** is set to **Generator** and **Generator Waveform** is set to **File**, the acquisition begins at the file’s first sample, plus any time or sample count set in **Delay**. A trigger is generated each time the file loops back to its first sample.
- **External**
  External is only available for analyzers equipped with the Advanced Master Clock. The acquisition begins at the first trigger pulse received at the rear-panel **Trigger In** BNC connector plus any time or sample count set in **Delay**. Set the **Trigger Input** in **Signal Path Setup > Triggers** to the desired voltage threshold and pulse edge.

Averages

For non-periodic waveforms such as noise, averaging multiple acquisitions can provide a more useful view. By setting **Averages** to more than 1, you can specify

how many acquisitions to average. Default is 1; maximum is 1000.

- **Power**
  The scope result does not use power averaging. Power averaging affects only the frequency domain results. See “More About Averaging” on page 494 for a description of power averaging.
- **Synchronous**
  When **Trigger** is set to **Generator**, **Synchronous Averaging** is an option. Synchronous averaging affects both the time domain and the frequency domain results. See “More About Averaging” on page 494 for a description of synchronous averaging.

Max Hold and Min Hold are not averaging functions, but use the multiple acquisitions available to the Averages feature to find maximum or minimum values for each FFT bin.

- **Max Hold**
  Max Hold displays the highest value for each bin found in all the acquisitions set in **Averages**. As the acquisitions progress, the value in each bin can only go up. The scope result does not use **Max Hold**. This selection only affects the frequency domain results.
- **Min Hold**
  Min Hold displays the lowest value for each bin found in all the acquisitions set in **Averages**. As the acquisitions progress, the value in each bin can only go down. The scope result does not use **Min Hold**. This selection only affects the frequency domain results.

Signal

The Signal field is only available when Input Connector in Signal Path Setup is set to PDM.

- **Decimated Audio**
  The decimated PDM audio (at the baseband sample rate and bandwidth) is the input signal.
- **PDM Bitstream**
  The PDM bitstream (at the PDM bit clock rate and bandwidth) is the input signal. This is useful to view the out-of-band noise and other components of the PDM bitstream.

Note: When the PDM input decimation is set to x1, (see “Settings for PDM Input” on page 159) the PDM signal is not decimated. In that case, either of the Signal selections in the Signal Analyzer will present the undecimated PDM bitstream for the analyzer acquisition.

Save To File

You can record the audio signal to a computer file as the signal is being acquired. Click **Save to File** before you run the measurement. See **Recording Audio to a File** on page 381.

File Settings

When saving an audio signal to a file, click **File Settings** to specify the file name and location.

Nesting

Signal Analyzer records can be nested. See Nested Sweeps on page 185.

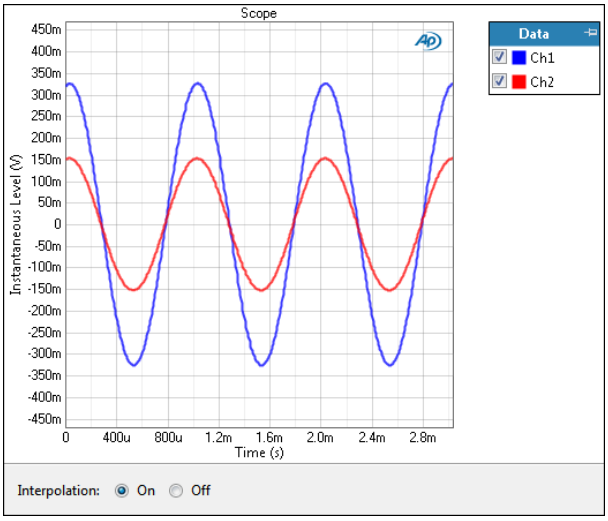
Secondary Source

Select the secondary sweep source here. This is the outside, nesting parameter that is varied to change the conditions for the recording through a number of iterations. Available choices are dependent upon instrument hardware, options and Input/Output settings.

Advanced Settings

If your test requires individual generator settings for each channel or special analyzer ranging or FFT windowing, click **Advanced Settings**. See Advanced Settings for the Signal Analyzer on page 491.

Signal Analyzer: Scope



The Signal Analyzer is a general purpose diagnostic tool that uses a high-resolution FFT technique to provide several different display results. Scope is a time domain result, displaying the signal amplitude versus time, analogous to an oscilloscope view.

Interpolation

The interpolation setting is only available when viewing Scope results.

For time-domain displays, the APx500 graphs are normally plotted using sinc function interpolation. Interpolation adds points to the displayed trace that do not exist in the measured data, to make data trends more easily visualized. The default in APx is to have Interpolation **On**. However, digital domain signals are sometimes best understood when viewing actual samples, with interpolation switched **Off**.

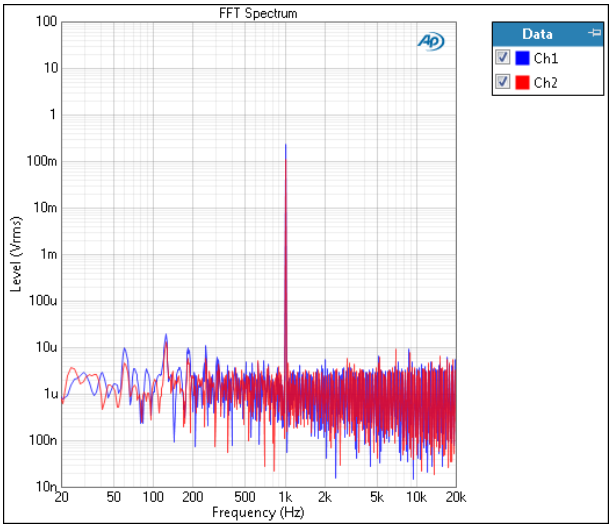
Interpolation can cause limit failure markers to appear to be in an incorrect location. See “Interpolation and limit failure markers” on page 678 for an explanation.

Units

Units available for Signal Analyzer: Scope are

X-axis	*Y-axis (analog)*	*Y-axis (digital)*
• s	• V	• D • hex

Signal Analyzer: FFT Spectrum



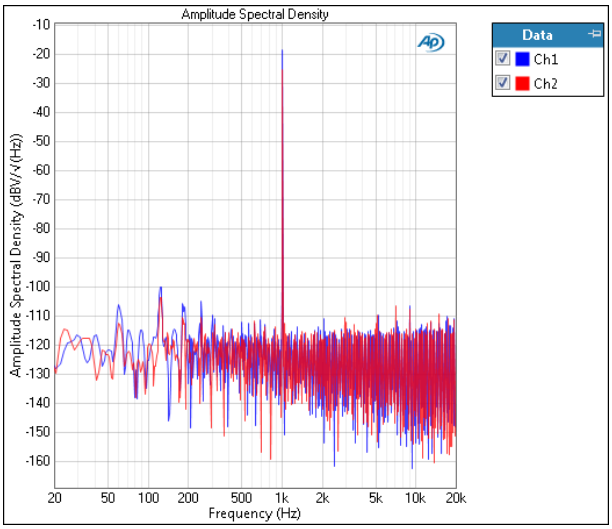
The Signal Analyzer is a general purpose diagnostic tool that uses a high-resolution FFT technique to provide several different display results. FFT Spectrum is a frequency domain result, displaying the signal amplitude versus frequency.

Units

Units available for Signal Analyzer: FFT Spectrum are

X-axis	*Y-axis (analog)*	*Y-axis (digital)*
• Hz • dHz • F/R • %Hz	• Vrms • dBV • dBu • dBrA • dBrB • dBSPL1 • dBSPL2 • dBm • W (watts)	• dBFS • FS • %FS • dBrA • dBrB • dBSPL1 • dBSPL2

Signal Analyzer: Amplitude Spectral Density



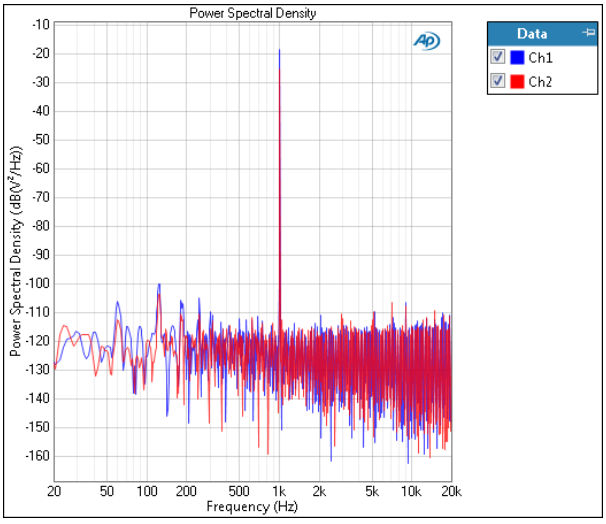
The Signal Analyzer is a general purpose diagnostic tool that uses a high-resolution FFT technique to provide several different display results. Amplitude Spectral Density is a frequency domain result, displaying the signal amplitude per root Hertz, versus frequency.

Units

Units available for Signal Analyzer: Amplitude Spectral Density are

X-axis	*Y-axis (analog)*	*Y-axis (digital)*
• Hz • dHz • F/R • %Hz	• Vrms/√(Hz) • dBV/√(Hz) • dBu/√(Hz)	• FS/√(Hz) • dBFS/√(Hz)

Signal Analyzer: Power Spectral Density



The Signal Analyzer is a general purpose diagnostic tool that uses a high-resolution FFT technique to pro-

vide several different display results. Power Spectral Density is a frequency domain result, displaying the signal amplitude squared per Hertz, versus frequency.

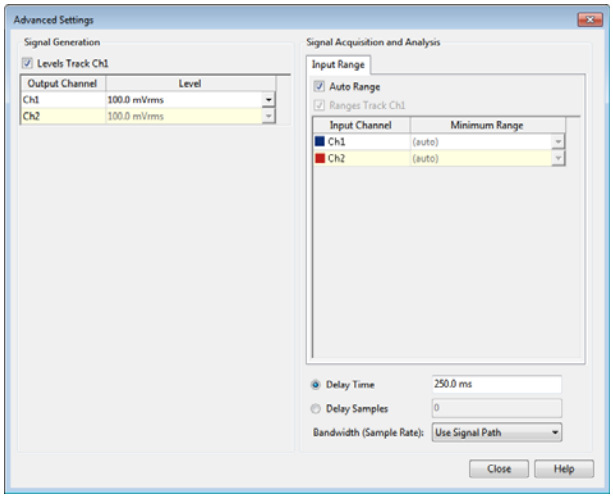
Units

Units available for Signal Analyzer: Power Spectral Density are

X-axis	*Y-axis (analog)*	*Y-axis (digital)*
• Hz	• dB(V^2/Hz)	• dB(FS^2/Hz)
• dHz	• V^2/Hz	• FS^2/Hz
• F/R		
• %Hz		

Advanced Settings for Signal Analyzer

The default settings here are appropriate for most measurements. You may want to make minor adjustments for special situations.



Signal Generation

You can set different output levels or DC offsets for each channel here.

If **Levels Track Ch1** is checked (the default), the generator output level and DC offset values for channel 1 are copied to the other channels, and values for channels greater than 1 cannot be edited. Any changes made to channel 1 are reproduced in the other channels.

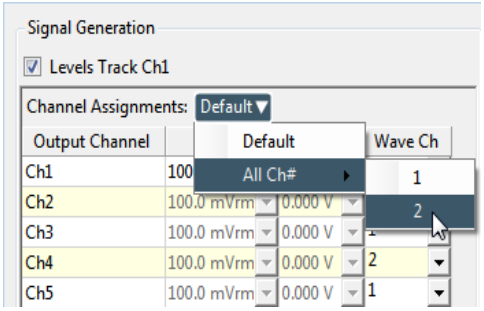
To set individual generator output channel levels or offsets, uncheck the **Levels Track Ch1** checkbox and enter values in the output channel Level or Offset fields.

Set Channel Assignments for special waveforms

For special waveforms like Sine Dual, you can specify the channel assignment.

Set Channel Assignments for generator waveforms

For stereo or multichannel generator waveform files, you can set channel mapping. By default, channel 1 audio is mapped to channel 1 output, channel 2 to channel 2, and so on. If the number of channels in the waveform file is less than the number of output channels, the waveform channels resume numbering at 1 and wrap to the next available output channel. These assignments can be remapped by changing individual settings in the **Wave Ch** column, or by selection one of several presets from the **Set Channel Assignments menu.**



Signal Acquisition and Analysis

Input Range

Digital inputs do not require ranging, and Input Range settings are unavailable when inputs are set to a digital format.

Auto Range

If **Auto Range** is checked (the default), each analog channel determines its input range automatically, based on the level of the input signal. If the input signal level changes beyond a ranging threshold, **Auto Range** will cause the input ranging circuits to move up or down for proper ranging.

Fixed Range

Alternatively, you can set a fixed minimum range for the analog inputs. Uncheck **Auto Range** and enter the expected maximum level in the **Minimum Range** filed. The software will choose the lowest range that will accommodate that level; this becomes the minimum range available, and the analyzer will not reset to a lower range. Fixed range is best for signals that vary rapidly, including tone bursts, perceptual audio speech samples, and program material such as voice or music. See **When autoranging isn't suitable** on page 626.

If a signal greater then the **Minimum Range** is presented to the analog input, the ranging circuits will range up to accommodate the higher level, and then drop again when the signal level falls. The range will not be set below the **Minimum Range**.

Ranges Track Ch1

If **Ranges Track Ch1** is checked (the default), the fixed input range setting for channel 1 is copied to the other channels, and range settings for channels greater than 1 cannot be edited. Any changes made to channel 1 are reproduced in the other channels. If **Ranges Track Ch1** is not checked, the fixed input range setting for each channel can be set independently.

Read more about ranging in Chapter 98.

Delay Time/Delay Samples

The **Delay (s)/Delay Samples** setting allows you to delay the acquisition by a specific period of time or a specific number of samples relative to the measurement **Start** or a trigger event, as described in **Trigger**.

This is sometimes necessary to avoid analysis of spurious signals created when using a coded generator file with a DUT that requires some time to match the coded format, or to allow the filter to settle when AC coupled.

For **Delay (s)**, enter the delay in seconds. Minimum time is 0.0 s; maximum is 60.0 s; the default is 250 ms.

If **Trigger** is set to **Generator**, you can choose to enter the delay in **Samples** rather than seconds. Minimum sample count is 0; maximum is 12,960,000.

Bandwidth (Sample Rate)

This control is only available when the analyzer input configuration is set to Analog.

The Bandwidth (Sample Rate) control allows you to override the analog input bandwidth setting in Signal Path Setup. The input bandwidth set here operates only on the Signal Analyzer.

If your system analog bandwidth is high, you can reduce it here for faster measurements. If your system bandwidth is low, you can expand it here for more detailed Signal Analyzer results.

The analog input bandwidth (sample rate) settings, whether made here or in Signal Path Setup, result in channel-count constraints at high sample rates.

The bandwidth choices, followed by their corresponding sample rates, are:

- 2.75 kHz (6 kS/s)
- 3.5 kHz (8 kS/s)
- 5.5 kHz (12 kS/s)
- 7 kHz (16 kS/s)
- 11 kHz (24 kS/s)
- 20 kHz (44.1 kS/s)
- 22 kHz (48 kS/s)
- 40 kHz (88.2 kS/s)
- 45 kHz (96 kS/s)
- 80 kHz (176.4 kS/s)
- 90 kHz (192 kS/s)
- 250 kHz (624 kS/s) (requires the BW52 Option)
- 500 kHz (1.248 MS/s) (requires the BW52 Option)
- 1 MHz (2.496 MS/s) (requires the BW52 Option, set to 1 channel)
- Use Signal Path (the default)

Note that recording files at bandwidths above 90 kHz (sample rates above 192 kS/s) requires a very fast PC and hard disk, due to the high data rates. Slower systems can suffer recording interruptions and failure.

More about FFTs

The Fast Fourier Transform, or FFT, is a mathematical technique that deconstructs a complex waveform into its component sine waves. Presented with a segment of data which represents the amplitude vs. time relationships of a complex waveform, an FFT will derive two sets of results: the amplitude spectrum (amplitude vs. frequency) and the phase spectrum (phase vs. frequency) of the signal. This information is sufficient to describe the relationships of the sine waves which make up the waveform, and with further computation the FFT results can be interpreted for several different types of display.

Much of the signal analysis in APx500 is performed using FFT techniques. These results are typically displayed in the frequency domain, as level vs. frequency or phase vs. frequency spectrum graphs. Some results (such as Impulse Response and Scope) are displayed in the time domain, as level vs. time.

FFT length

The length (in samples) of the audio data being transformed is called the FFT length. Since this length consists of samples, its length in time varies with the sample rate. Many of the nominal FFT lengths shown in the APx menu are convenient approximations. The actual lengths are shown here:

Nominal (menu)	Actual Length in samples
256	256
512	512
1K	1,024
2K	2,048
4K	4,096
8K	8,192
12K	12,000
16K	16,384
24K	24,000
32K	32,768
48K	48,000

64K	65,536
96K	96,000
128K	131,072
192K	192,000
256K	262,144
300K	312,000
512K	524,288
600K	624,000
1M	1,048,576
1.2M	1,248,000

FFT bandwidth

The bandwidth of the FFT is from DC to 1/2 the sample rate.

FFT bins and bin width

The bandwidth of the FFT is divided into bins, the number of which is 1/2 the FFT length. The bin width (also called line spacing) defines the frequency resolution of the FFT. The FFT provides amplitude and phase values for each bin.

The bin width is stated in hertz. The bin width can be calculated by dividing the sample rate by the FFT length; or by dividing the bandwidth by the number of bins (which is equal to 1/2 the FFT length).

Examples

As an example, at a sample rate of 48 kHz and an FFT length of 16,384 (the 16K setting), the bandwidth would be DC to 24 kHz, the number of bins would be 8,192, and the bin width would be 24 kHz / 8,192 = 2.93 Hz.

With care, you can select sample rates and FFT lengths that will result in a specific bin width; for example, an FFT length of 65,536 samples (APx menu setting of 64K) and a sample rate of 65,536 Hz (we're assuming a digital input, with the signal at that sample rate) provides a bin width of exactly 1 Hz.

The FFT lengths of 312000, 624000 and 1248000 are included for use with the BW52, to provide integer bin widths at the sample rates used for BW52 bandwidths.

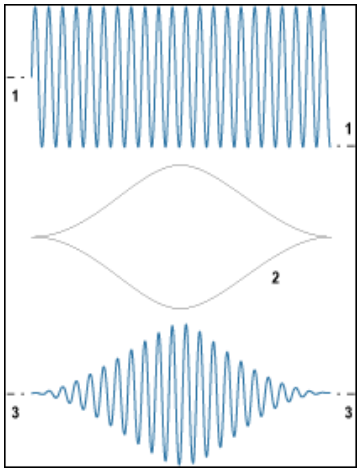
FFT windows

The Fourier transform has the effect of connecting the end of the FFT record to its beginning, forming an endless loop of data in the calculation.

In almost all cases, there is a discontinuity at the "splice," and just like a bad audio edit, the discontinuity is an impulse that spills energy across the spectrum. Unmodified, this energy swamps the useful data. In APx500 we use FFT windowing to eliminate the discontinuity.

An FFT window is an amplitude envelope that is applied to the FFT record before the transform. FFT windows change the data, and therefore affect the results. However, a number of windows have been developed that provide useful results while eliminating the looping discontinuity.

The example below shows the Hann window in the time domain, with the FFT record at the top, the window function in the center and the modified FFT record at the bottom. Other window functions have different shapes to optimize the resultant FFT for different purposes. What they all have in common is a tapering to zero at both ends, to eliminate the discontinuity.



The windows provided with APx500 are detailed here:

	Minimum sidelobe atten. (dB)	*Mainlobe width (bins)*	*Maximum level error (dB)*
Hann	----	4	–1.43
Blackman-Harris 3-term	58	6	–1.10
Blackman-Harris 4-term	92	8	–0.83
Flat Top	94	10	–0.02
AP-Equiripple	179	12	–0.50
Dolph-Chebyshev 150	147	12	–0.60
Dolph-Chebyshev 200	195	16	–0.46
Dolph-Chebyshev 250	240	20	–0.37

Hann

The Hann window is a raised cosine window that provides good selectivity near the top of the main lobe (about –6 dB at one bin away from center and about –30 dB at two bins away), with no side lobes. Its skirts more than 3 bins off center are not as steep as the Blackman-Harris window. The Hann window causes approximately –1.5 dB maximum amplitude error due to window attenuation, if the signal is at the extreme edge of the bin.

Blackman-Harris 3-term

The Blackman-Harris 3-term window is about –5 dB in the bins adjacent to the center, about –20 dB two bins off, and about –160 dB three bins off. Response fall-off is monotonic. Scallop loss is 1.1 dB maximum.

Blackman-Harris 4-term

The Blackman-Harris window is a 4-term minimum side lobe window, as used in AP2700 software. When compared to the Hann window, it is not quite as selective across the central several bins (about –3 dB in the adjacent bins and about –14 dB at two bins off), but has steeper skirts beyond that point. The Blackman-Harris window has side lobes below –92 dB (response fall-off is not monotonic). It has a reasonably flat top with a maximum amplitude error of about –0.8 dB if the signal is at the extreme edge of the bin.

Flat Top

The Flat-Top window is designed for the greatest amplitude measurement accuracy. It provides a maximum amplitude error due to window attenuation of less than –0.02 dB. However, its selectivity is poorer than the other windows. The Flat-Top window is the appropriate window for accurate amplitude measurements (such as when measuring individual harmonics) except when signals are so closely spaced that its selectivity becomes a problem.

AP-Equiripple

The Equiripple window, developed at Audio Precision, is an approximation to the Dolph-Chebyshev window that has the narrowest main lobe width for a given maximum side lobe depth. The main lobe is approximately 12 bins wide; that is, the first null is about six bins from the main lobe center. The first side lobe, which is also the highest, is –147 dB from the main lobe. The maximum amplitude error with a signal at the bin boundary is about 0.5 dB.

Dolph-Chebyshev

The Dolph-Chebyshev windows are true equiripple windows. For a given minimum sidelobe attenuation, they provide the minimum possible mainlobe width. The Dolph-Chebyshev 250 window is capable of revealing details in very low noise floors that other windows obscure.

More About Averaging

The Acoustic Response and Signal Analyzer measurements use averaging to reduce noise in measurement results. This technique averages the signal over a specified number of acquisitions by summing the signals from all acquisitions and dividing the result by the number of acquisitions.

Synchronous averaging

Synchronous averaging operates on a time domain acquisition. Synchronous averaging is useful to examine coherent time domain waveforms by reducing the average level (but not the variance) of noise and other non-coherent signals. Since the frequency domain results in a measurement are derived from the time domain acquisition, synchronous averaging affects spectrum results as well.

Power averaging

Power averaging (spectrum averaging) operates only on a frequency domain result. The power spectra for all averaged acquisitions are summed and then divided by the number of acquisitions. Power averaging helps reveal coherent frequency components by reducing the variance (but not the average level) of noise and other non-coherent signals.

Application in APx500

Synchronous averaging is available to all the results in the **Signal Analyzer** measurements, when using a **Generator Waveform** file as the stimulus signal, and triggering from the generator. The generator waveform is looped end-to-end for playback in the generator, and each repetition is synchronously acquired and added to the averaging process.

Synchronous averaging is also available to all the results in the **Acoustic Response** measurement.

Power averaging is available to the **Signal Analyzer** FFT Spectrum, Amplitude Spectral Density and Power Spectral Density results, and to the **FFT Monitor**.

Averaging acquisitions from 2 to 1000 are available. A selection of 1 disables averaging.

Signal to Noise Ratio
(Sequence Mode)

The Signal to Noise Ratio measurement provides a single-value result that shows the signal to noise ratio (SNR) of the output signal from each DUT channel. To use the SNR measurement to produce a dynamic range result, set the initial generator level to produce the maximum level in the DUT.

For the AES17 Dynamic Range measurement for digital converters, go to Chapter 45.

Operation

If you have not yet set up your test, first go to Signal Path Setup to select and configure inputs and outputs, references and other settings. See Chapter 7 for information about Signal Path Setup.

Running the measurement

To measure SNR, click **Start**. The generator is turned on briefly, then off. The results are compared and displayed as meter bars.

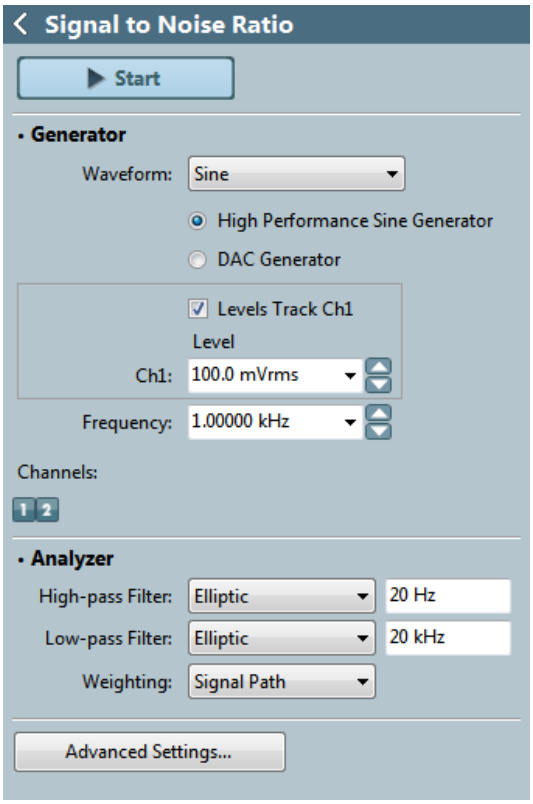
Generator

The typical configuration for this measurement is the closed-loop configuration, using the APx generator as a stimulus. Alternatively, you can make open-loop measurements, using either the External Source or File Input configuration.

See Chapter 6 for detailed information about the using the APx Generator and making Waveform, Level and Frequency settings, or for information about using External Source.

High Performance Sine Generator / DAC Generator

For an APx555, you have the option of using the High Performance Sine Generator (very low distortion sine waves and wide bandwidth) or the DAC generator, which provides special sine waveforms. See High Performance Sine Generator starting on page 51 for a comparison of features.



Level

For signal to noise ratio, you must choose a generator level that represents the maximum level for the device under test (DUT).

Analyzer

Filters

Local high-pass, low-pass and weighting filters are available for the “noise” (generator **OFF** or low level) portion of this measurement. See Measurement Filters on page 619 for detailed information about the filters available locally for some measurements.

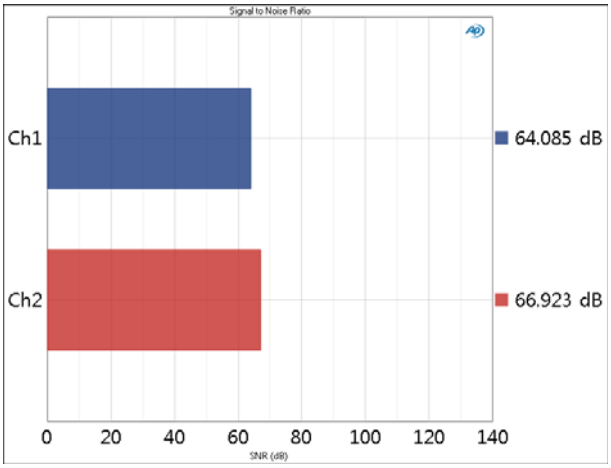
Low pass, high pass and weighting Signal Path Filters settings (see page 617) made in Signal Path Setup > Input/Output will also affect this measurement. When set, the local low-pass or high-pass filter setting overrides the corresponding Signal Path filter setting while the measurement is active. A local weighting filter setting does not override any Signal Path filter settings; the local weighting filter is applied in addition to any active Signal Path filters.

Advanced Settings

If your test requires individual generator settings for each channel or special analyzer ranging or settling, click **Advanced Settings**. See Advanced Settings for single value measurement on page 351.

See Chapter 104 for more information about units of measurement.

Signal to Noise Ratio



The Signal to Noise Ratio measurement provides a single-value result showing the SNR for each DUT channel.

Units

Units available for SNR are

- x/y
- dB

More about SNR

Overview: SNR

Signal-to-noise ratio (SNR) was first used to evaluate the intelligibility of analog radio voice communications, and expressed the difference between the nominal signal level and the noise at the radio receiver. In excellent conditions, the voice might be 40 dB above the noise; in the worst conditions, the signal would be “lost in the noise” and unintelligible.

Strictly speaking, then, the “signal” level is arbitrary, although it is usually taken to be the “nominal” program level. SNR is actually two measurements: first, of the signal level; and second, of the noise level with the signal turned off. These two measurements are expressed as a ratio, almost always in decibels.

Like most noise measurements, the SNR results are typically bandpass limited using high and low pass filters, or a weighting filter. Filter use should be stated in the results.

The APx500 Signal-to-Noise Ratio measurement makes the two measurements and computes the ratio in one operation.

Overview: Dynamic Range

Dynamic range is an expression of the ratio of the largest signal a device can pass to the device's noise floor. “Largest signal” usually refers to a signal at a specified degree of distortion, often 1%. Signal-to-noise ratio and dynamic range are essentially the same measurement, except that the signal in SNR is arbitrary (and should be stated in the results), and the signal in dynamic range is at the maximum (details of which should also be stated in the results).

For the AES17 Dynamic Range measurement for digital converters, see Chapter 45.

SINAD
(Sequence Mode)

SINAD provides a single-value measurement of SINAD (Signal to Noise and Distortion) in the output signal from each DUT channel. ENOB is the mathematical equivalent of SINAD, and ENOB results are provided as well.

More Information

See More about SINAD and More about ENOB on page 498.

Operation

If you have not yet set up your test, first go to Signal Path Setup to select and configure inputs and outputs, references and other settings. See Chapter 7 for information about Signal Path Setup.

Running the measurement

To measure SINAD, click the Generator button to **On**. The generator will output the test signal to the DUT on the selected generator channels. Read the SINAD ratio for each channel from the meter bar display. Click the Generator button to **Off**.

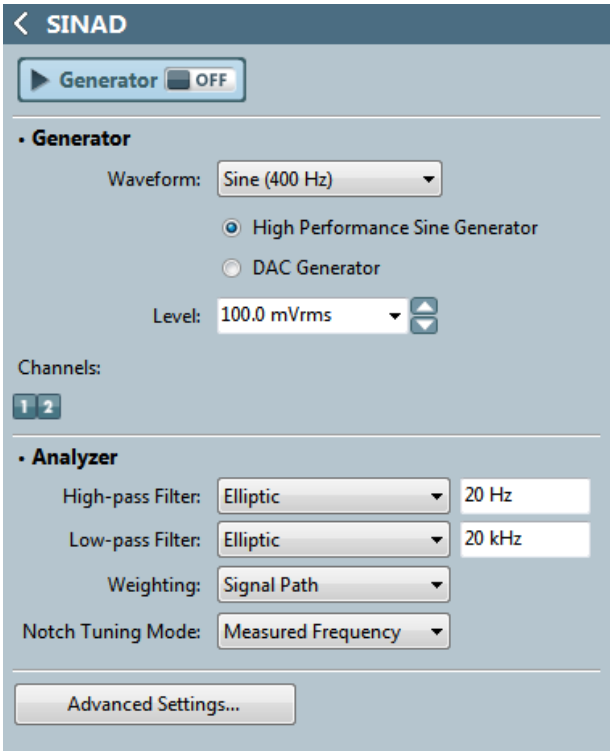
Generator

The typical configuration for this measurement is the closed-loop configuration, using the APx generator as a stimulus. Alternatively, you can make open-loop measurements, using either the External Source or File Input configuration.

See Chapter 6 for detailed information about the using the APx Generator and making Waveform, Level and Frequency settings, or for information about using External Source.

Selecting the generator waveform

By default, the SINAD measurement uses a sine wave at 400 Hz at the level set in the Signal Generation panel as the test signal. Alternatively, you can select a 1 kHz sine wave.



High Performance Sine Generator / DAC Generator

For an APx555, you have the option of using the High Performance Sine Generator (very low distortion sine waves and wide bandwidth) or the DAC generator, which provides special sine waveforms. See High Performance Sine Generator starting on page 51 for a comparison of features.

Analyzer

Filters

Local high-pass, low-pass and weighting filters are available for this measurement. See Measurement Filters on page 619 for detailed information about the filters available locally for some measurements.

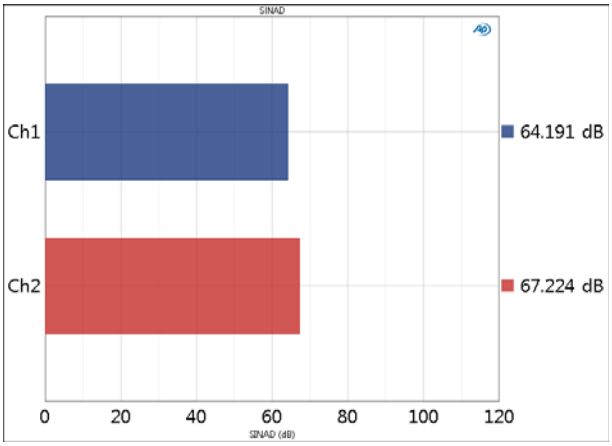
Low pass, high pass and weighting Signal Path Filters settings (see page 617) made in Signal Path Setup > Input/Output will also affect this measurement. When set, the local low-pass or high-pass filter setting overrides the corresponding Signal Path filter setting while the measurement is active. A local weighting filter setting does not override any Signal Path filter settings; the local weighting filter is applied in addition to any active Signal Path filters.

Advanced Settings

If your test requires individual generator settings for each channel or special analyzer ranging or settling, click **Advanced Settings**. See Advanced Settings for single value measurement on page 351.

See Chapter 104 for more information about units of measurement.

SINAD



The SINAD measurement provides a single-value result showing the SINAD for each DUT channel.

Units

Units available for SINAD are

- x/y
- dB

More about SINAD

SINAD is an abbreviation for Signal-to-Noise and Distortion. It is a measure of audio quality used in low signal-to-noise ratio situations such as radio communications systems. SINAD is usually performed with a stimulus signal of 400 Hz or 1 kHz.

SINAD is defined as the ratio of the total signal (desired signal plus the sum of all distortion and noise components) to the sum of all distortion and noise components. SINAD can never be less than 1, and is always positive when expressed in dB. SINAD is the reciprocal of THD+N.

More about ENOB

ENOB (Effective Number Of Bits) is a measure of quality of a signal, expressed in terms that relate to digital acquisition processes. ENOB is the mathematical equivalent of SINAD (and, by extension, THD+N under similar conditions), by the conversion shown here:

$$\text{ENOB} = \frac{\text{SINAD} - 1.76}{6.02}$$

Stepped Frequency Sweep
(Sequence Mode)

The **Stepped Frequency Sweep** measurement uses a sine wave stimulus signal that is moved across a range of frequencies in a series of steps or points. The DUT output is acquired by the analyzer and processed for display. X axis is the sweep frequency; Y axis is DUT output.

Stepped Frequency Sweep measurement results available in APx500 are:

- RMS Level (freq. response)
- Gain
- Relative Level
- Deviation
- Phase
- THD Ratio
- THD Level
- THD+N Ratio
- THD+N Level
- Distortion Product Ratio
- Distortion Product Level
- SINAD
- Peak Level
- Average Jitter Level

All of these results are available from a single measurement.

Operation

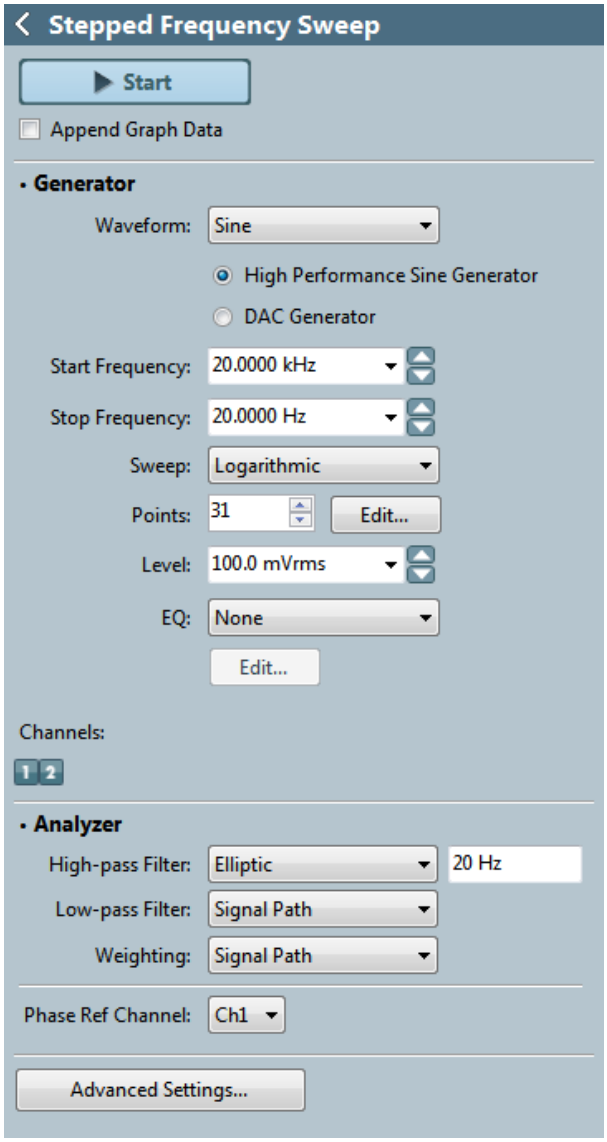
If you have not yet set up your test, first go to Signal Path Setup to select and configure inputs and outputs, references and other settings. See Chapter 7 for information about Signal Path Setup.

Running the Measurement

For a default sweep of 20 kHz to 20 Hz in 31 logarithmic steps, simply click **Start**. If you want to specify other parameters for your sweep, review **Generator** and **Analyzer** settings, discussed below. Click **Start** when you have approved the settings.

Append Graph Data

Measured data are grouped in a **Data Set**, which contains all acquisition results. Normally, the current graph data are deleted each time you start a new measurement. If the **Append Graph Data** box is checked, appended data are grouped as additional data sets, as are any imported data. The Data Sets Panel displays the Data Set(s) in the measurement,



and provides a visibility checkbox, a Time acquisition field, and a Notes field for each data set.

Read more about Data Sets beginning on page 649.

Generator

The typical configuration for this measurement is the closed-loop configuration, using the APx generator as a stimulus. Alternatively, you can make open-loop measurements, using either the External Source or File Input configuration.

See Chapter 6 for detailed information about the using the APx Generator and making Waveform, Level and Frequency settings, or for information about using External Source.

High Performance Sine Generator / DAC Generator

For an APx555, you have the option of using the High Performance Sine Generator (very low distortion sine waves and wide bandwidth) or the DAC generator, which provides special sine waveforms. See High Performance Sine Generator starting on page 51 for a comparison of features.

Start Frequency

Set the sweep start frequency here.

Stop Frequency

Set the sweep stop frequency here.

Sweep

Choose

- **Logarithmic** point spacing
- **Linear** point spacing
- **Custom**
  Edit sweep spacing and number of **Points** to create a Custom sweep.

Points

Set the number of sweep points here.

Points / Edit

Open the **Sweep Points** dialog to edit import or export the generator **Sweep Points** table. See page 507.

Step Size

For linear sweeps, enter the **Step Size** here.

Level

Set the signal level here. This level may be changed if generator equalization is used, and can be adjusted on a per-channel basis in **Advanced Settings** (see page 508).

Offset

Set any DC offset to the generator signal here.

EQ

You can optionally modify the sweep with by applying a equalization curve to the generator. See “Generator Equalization” on page 192 for a general discussion of this feature. Choose

- **None**, which applies no equalization to the generator.
- **Relative**, which applies the gain or loss specified in the EQ table to the current generator level for each channel.
- **Absolute**, which sets the generator level for all channels to the levels specified in the EQ table. Note: when **Absolute** is selected, current channel level settings are lost.

Click the **Edit** button to create or change the EQ curve. See the **Edit EQ Table** dialog discussion on page 192.

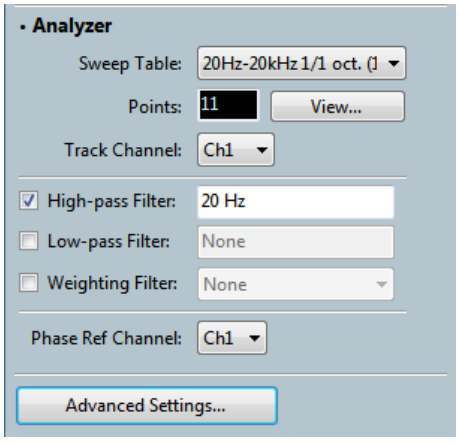
Ch

Toggle specific output channels on or off by clicking on the Channel number button.

Analyzer

This measurement can be configured to measure jitter in Signal Path Setup > Input > Measure. Read detailed information about jitter generation and measurement beginning on page 69.

Sweep Table



The analyzer Sweep Table menu, and the Points, View/Edit and Track Channel controls discussed below, are not shown for a closed loop sweep that uses the internal generator. In that case, the stimulus sweep definition is directly available to the analyzer and these controls are unnecessary.

When the measurement is configured so that the stimulus sweep definition is not available to the analyzer (in **External Source, Generator Waveform** or **File Input** configurations), you must select an analyzer

Sweep Table (see page 507) that matches the definition of the sweep you are using.

The analyzer **Sweep Table** menu lists the embedded sweep tables and any new tables that have been added to the project. Select a table from the analyzer **Sweep Table** menu, or choose **Create New** or **Browse** for a sweep table file.

Points

This field displays the number of points in the analyzer **Sweep Points** table.

View / Edit

For embedded analyzer sweep tables, click **View** (see page 507) to view or export the table, or to create a WAV file from the table. For other analyzer sweep tables, click **Edit** to edit, view or export the table, or to create a WAV file from the table (see page 508).

Filters

Local high-pass, low-pass and weighting filters are available for this measurement. The low-pass and high-pass filters, if selected, are applied to the entire measured signal and affect all the results. The weighting filter, if selected, affects only the distortion and distortion plus noise results. See Measurement Filters on page 619 for detailed information about the filters available locally for some measurements.

Low pass, high pass and weighting Signal Path Filters settings (see page 617) made in Signal Path Setup > Input/Output will also affect this measurement. When set, the local low-pass or high-pass filter setting overrides the corresponding Signal Path filter setting while the measurement is active. A local weighting filter setting does not override any Signal Path filter settings; the local weighting filter is applied in addition to any active Signal Path filters.

Phase Ref Channel

Interchannel Phase measurements must be referenced to one channel. Set the phase reference channel here.

Track Channel

When using a sweep table (in **External Source, Generator Waveform** or **File Input** configurations), the analyzer must determine the relation of the incoming signal to the sweep table. When **Auto** is selected, the analyzer continually polls all the channel inputs throughout the sweep, switching between channels as necessary to track the input with the highest signal level.

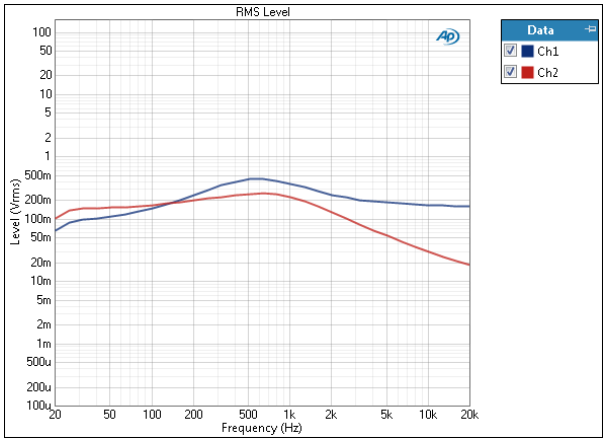
Alternatively, you can select a specific input channel for the analyzer to use for sweep tracking.

Advanced Settings

If your test requires special adjustments or settings, click **Advanced Settings**. See page 508.

See Chapter 104 for more information about units of measurement.

RMS Level results



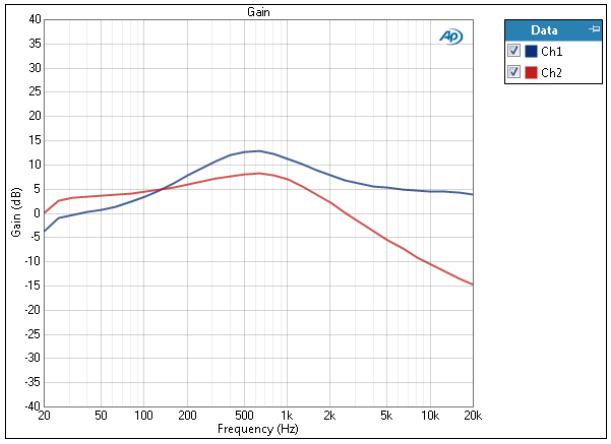
Stepped Frequency Sweep: RMS Level results uses a sine wave stimulus signal that is moved across a range of frequencies in a specified number of points. This is the conventional frequency response sweep.

Units

Units available for Stepped Frequency Sweep: RMS Level results are:

X-axis	*Y-axis (analog)*	*Y-axis (digital)*	*Jitter signals*
• Hz	• Vrms	• dBFS	• UI
• dHz	• dBV	• FS	• dBUI
• F/R	• dBu	• %FS	• s
• %Hz	• dBrA	• dBrA	
	• dBrB	• dBrB	
	• dBSPL1	• dBSPL1	
	• dBSPL2	• dBSPL2	
	• dBm		
	• W (watts)		

Gain



Gain is a stepped frequency sweep result that provides a graphical display of the voltage gain in each channel. In this result the DUT gain is plotted against frequency. The stepped frequency sweep Gain measurement is not available in External Source configuration.

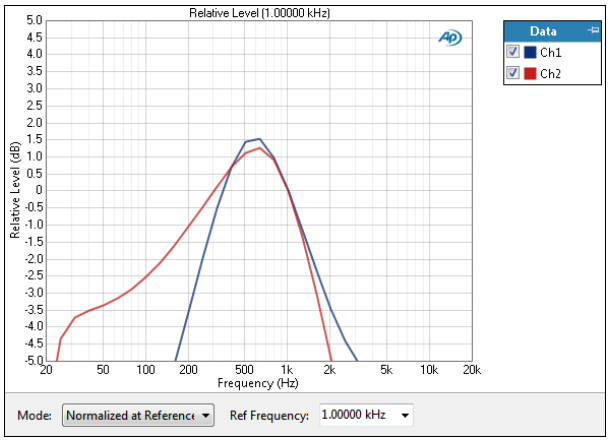
Gain results are not available when External Source is selected as the output configuration, when File is selected as the input configuration or when a generator waveform file is selected as the test signal.

Units

Units available for Stepped Frequency Sweep: Gain are:

X-axis	*Y-axis same-domain*	*Y-axis cross-domain*
• Hz	• x/y	• FS/Vrms
• dHz	• %	• dB(FS/Vrms)
• F/R	• ppm	—or—
• %Hz	• dB	• Vrms/FS
		• dB(Vrms/FS)

Stepped Frequency Sweep: Relative Level



The Stepped Frequency Sweep: Relative Level result uses a sine wave stimulus signal that is moved across a range of frequencies in a specified number of points, providing a conventional frequency response sweep. Additionally, in this result the response curve is plotted in relation to the measured level at a selected frequency. This enables you to specify the frequency where the relative response will be unity and to view the response in relation to that frequency. A midband frequency such as 1 kHz is often set as the reference frequency, and is the default here.

Mode

Select one of two result display modes.

Normalized at Reference

Enter a reference frequency for normalization in the **Ref Frequency** field. The measured level at the frequency you choose is set as 0 dB on the graph, and the DUT response is plotted in relation to this reference. You can change the **Ref Frequency** at any time (except after appending) and the graph will immediately be redrawn to reflect the new setting.

Auto-Centered in Limits

This control is only usable if upper and lower limits are set on this graph. In some applications, the general conformance of a trace to the shape of a set of limits is of importance, with absolute level less important or irrelevant. **Auto-Centered in Limits** centers the data trace relative to the upper and lower limits. To exclude possible exceptional data points at frequency extremes, the frequency range that is considered for the centering operation is constrained by the values in the **Min** and **Max** fields.

Result Settings

These settings are made in the Result Settings bar, beneath the graph display.

Ref Frequency

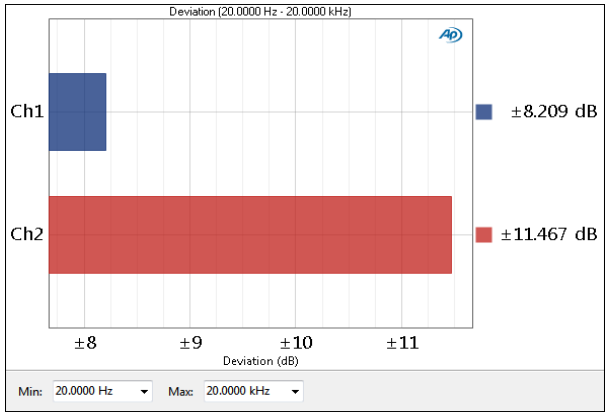
Choose a Reference Frequency here.

Units

Units available for Stepped Frequency Sweep: Relative Level are:

X-axis	*Y-axis*
• Hz	• x/y
• dHz	• %
• F/R	• ppm
• %Hz	• dB

Deviation



The Stepped Frequency Sweep: Deviation result uses a sine wave stimulus signal that is moved across a range of frequencies in a specified number of points. This result is a single value measurement computed from the stepped sweep that shows the frequency deviation (the total range of frequency variation) of each channel as a meter bar. You can specify a minimum and maximum frequency to define the range of the deviation measurement.

Result Settings

These settings are made in the Result Settings bar, beneath the graph display.

Min

Set the minimum frequency of the range of interest here.

Max

Set the maximum frequency of the range of interest here.

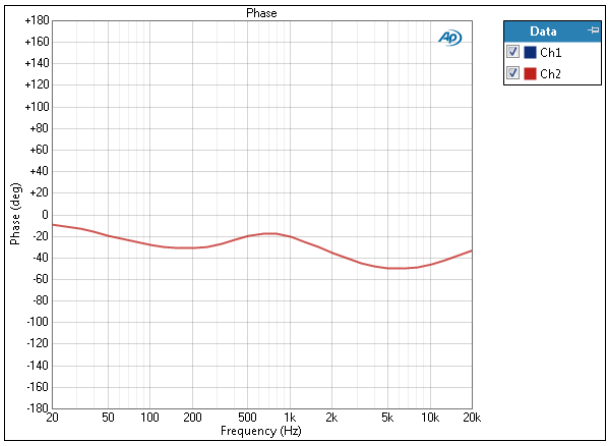
Units

Units available for Stepped Frequency Sweep: Deviation are

- dB

See Chapter 104 for more information about units of measurement.

Phase



The Phase result is a stepped frequency sweep measurement that provides a graphical display of the Phase response of each channel. One channel is chosen as the phase reference channel, and the remaining channels are measured against it.

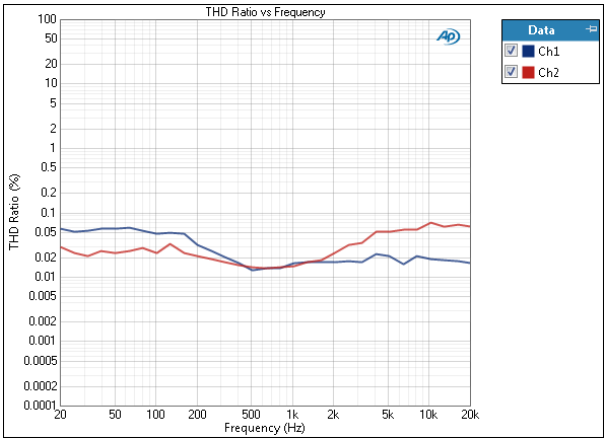
Select a phase reference channel in **Phase: Ref Channel**. The remaining channels are plotted against the selected channel.

Units

Units available for Stepped Frequency Sweep: Phase are:

X-axis	*Y-axis*
• Hz	• deg
• dHz	• rad
• F/R	
• %Hz	

THD Ratio



The THD Ratio result is a stepped frequency sweep measurement that provides a graphical display of the THD response of each channel. In this result the ratio of the level of the THD (total harmonic distortion) to the total signal in the DUT output is plotted against frequency.

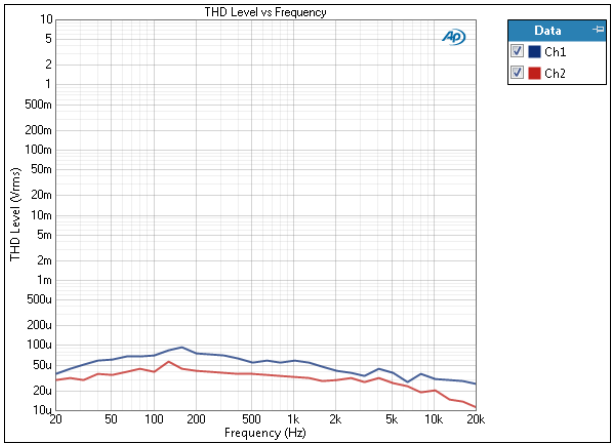
See page 527 for more information about THD.

Units

Units available for Stepped Frequency Sweep: THD Ratio are:

X-axis	*Y-axis*
• Hz	• x/y
• dHz	• %
• F/R	• ppm
• %Hz	• dB

THD Level



The THD Level result is a stepped frequency sweep measurement that provides a graphical display of the THD response of each channel. In this result the level of the THD (total harmonic distortion) in the DUT output is plotted against frequency.

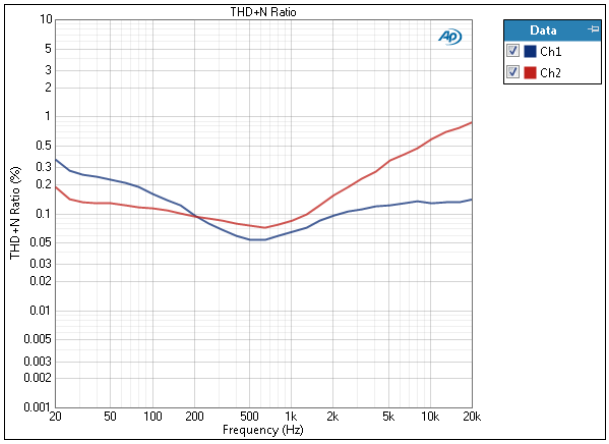
See page 527 for more information about THD.

Units

Units available for Stepped Frequency Sweep: THD Level are:

X-axis	*Y-axis (analog)*	*Y-axis (digital)*	*Jitter signals*
• Hz	• Vrms	• dBFS	• UI
• dHz	• dBV	• FS	• dBUI
• F/R	• dBu	• %FS	• s
• %Hz	• dBrA	• dBrA	
	• dBrB	• dBrB	
	• dBSPL1	• dBSPL1	
	• dBSPL2	• dBSPL2	
	• dBm		
	• W (watts)		

THD+N Ratio



The THD+N Ratio result is a stepped frequency sweep measurement that provides a graphical display of the THD+N response of each channel. In this result the ratio of the level of the THD+N (total harmonic distortion plus noise) to the total signal in the DUT output is plotted against frequency.

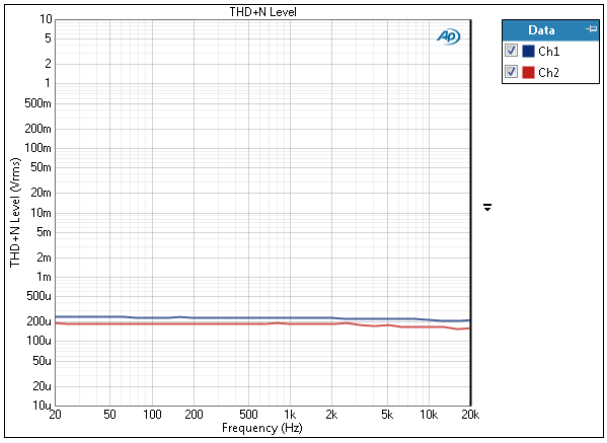
See page 527 for more information about THD+N.

Units

Units available for Stepped Frequency Sweep: THD+N Ratio are:

X-axis	*Y-axis*
• Hz	• x/y
• dHz	• %
• F/R	• ppm
• %Hz	• dB

THD+N Level



The THD+N Level result is a stepped frequency sweep measurement that provides a graphical display of the THD+N response of each channel. In this result the level of the THD+N (total harmonic distortion plus noise) in the DUT output is plotted against frequency.

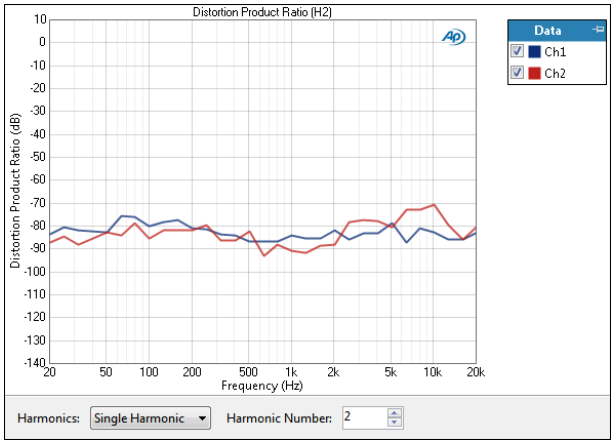
See page 527 for more information about THD+N.

Units

Units available for Stepped Frequency Sweep: THD+N Level are:

X-axis	*Y-axis (analog)*	*Y-axis (digital)*	*Jitter signals*
• Hz	• Vrms	• dBFS	• UI
• dHz	• dBV	• FS	• dBUI
• F/R	• dBu	• %FS	• s
• %Hz	• dBrA	• dBrA	
	• dBrB	• dBrB	
	• dBSPL1	• dBSPL1	
	• dBSPL2	• dBSPL2	
	• dBm		
	• W (watts)		

Distortion Product Ratio



The Stepped Frequency Sweep Distortion Product Ratio result provides a graphical display of the selected harmonic distortion products present in each channel. In this result the ratio of the level of the selected harmonic distortion product to the fundamental in the DUT output is plotted against frequency.

Select the harmonic distortion product(s) you wish to view using the **Harmonics** controls, described below.

Additional Controls for Distortion Product Ratio

Harmonics

For a graph of the ratio of one specific harmonic product to the signal level, select **Single Harmonic** and set the **Harmonic Number** to your choice, from 2nd harmonic to 20th harmonic.

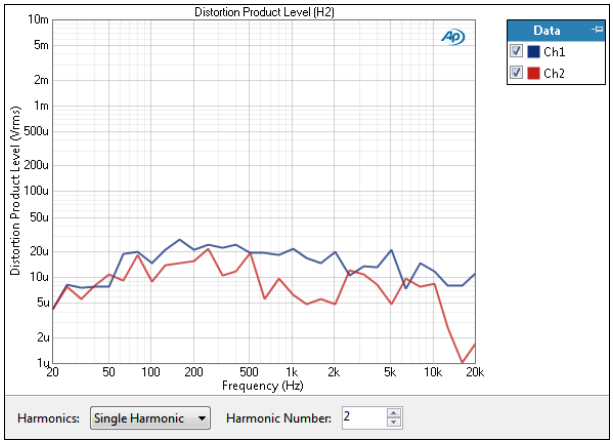
For a graph of the sum of the levels of any combination of harmonic products (from **H2** through **H20**), divided by the signal level, select **Sum of Harmonics**. Then click **Select Harmonics...** to open a result definition dialog.

Units

Units available for Stepped Frequency Sweep Distortion Product Ratio are:

X-axis	*Y-axis*
• Hz	• x/y
• dHz	• %
• F/R	• ppm
• %Hz	• dB

Distortion Product Level



The Stepped Frequency Sweep Distortion Product Level result provides a graphical display of the selected harmonic distortion products present in each channel. In this result the level of the selected harmonic distortion product in the DUT output is plotted against frequency.

Select the harmonic distortion product(s) you wish to view using the **Harmonics** controls, described below.

Additional Controls for Distortion Product Level

Harmonics

For a graph of the level of one specific harmonic product, select **Single Harmonic** and set the **Harmonic Number** to your choice, from 2nd harmonic to 20th harmonic.

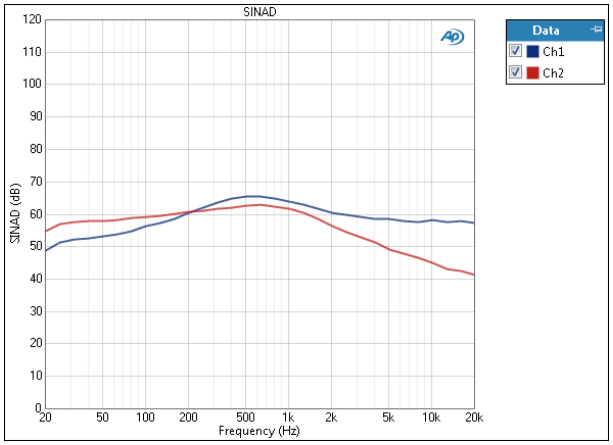
For a graph of the level of the sum of several or all harmonic products (from **H2** through **H20**), select **Sum of Harmonics**, then click **Select Harmonics...** to open a result definition dialog.

Units

Units available for Stepped Frequency Sweep Distortion Product Level are:

X-axis	*Y-axis (analog)*	*Y-axis (digital)*	*Jitter signals*
• Hz	• Vrms	• dBFS	• UI
• dHz	• dBV	• FS	• dBUI
• F/R	• dBu	• %FS	• s
• %Hz	• dBrA	• dBrA	
	• dBrB	• dBrB	
	• dBSPL1	• dBSPL1	
	• dBSPL2	• dBSPL2	
	• dBm		
	• W (watts)		

SINAD



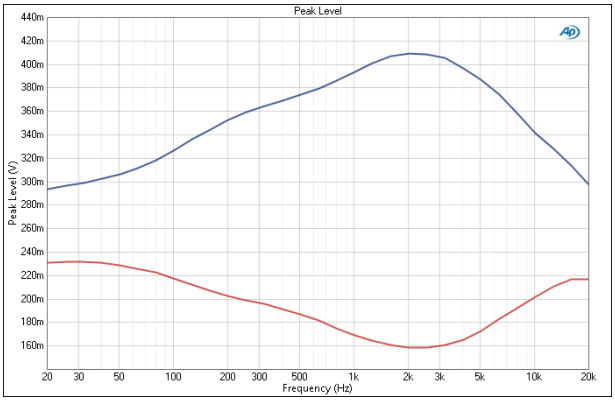
The Stepped Level Sweep: SINAD result uses a sine wave stimulus signal that is moved across a range of levels in a series of points. X axis is generator level; Y axis is DUT output SINAD ratio.

Units

Units available for SINAD are:

X-axis (analog)	*X-axis (digital)*	*Y-axis*
• Vrms	• dBFS	• x/y
• Vp	• FS	• dB
• Vpp	• %FS	
• dBV	• dBrG	
• dBu		
• dBrG		
• dBm		
• W (watts)		

Peak Level



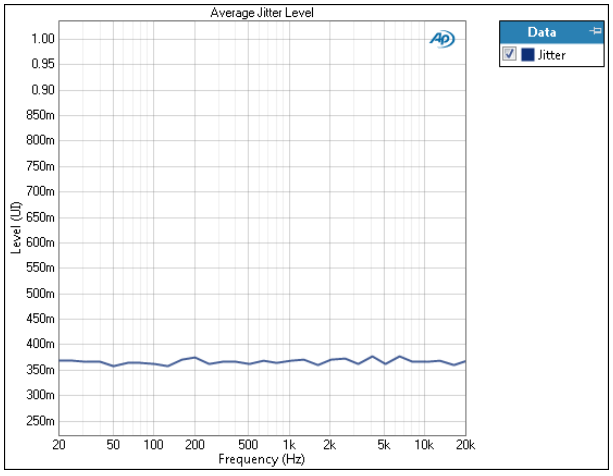
Stepped Frequency Sweep: Peak Level results uses a sine wave stimulus signal that is moved across a range of frequencies in a specified number of points. This is a frequency response sweep, with peak-scaled results.

Units

Units available for Stepped Frequency Sweep Peak Level results are:

X-axis	*Y-axis (analog)*	*Y-axis (digital)*	*Jitter*
• Hz	• V	• D	• UI
• dHz		• hex	• dBUI
• F/R			• s
• %Hz			

Average Jitter Level



Stepped Frequency Sweep: Average Jitter Level results uses a sine wave stimulus signal that is moved across a range of frequencies in a specified number of points.

Units

Units available for Stepped Frequency Sweep Level results are:

X-axis	*Y-axis (analog)*	*Y-axis (digital)*	*Jitter*
• Hz	• Vrms	• dBFS	• UI
• dHz	• dBV	• FS	• dBUI
• F/R	• dBu	• %FS	• s
• %Hz	• dBrA	• dBrA	
	• dBrB	• dBrB	
	• dBSPL1	• dBSPL1	
	• dBSPL2	• dBSPL2	
	• dBm		
	• W (watts)		

Using Sweep Tables with Stepped Frequency Sweep Measurements

A stepped frequency sweep depends upon the analyzer correctly tracking the stimulus sweep as it moves across its range.

When the source of the signal is the APx generator in an internal (closed loop) sweep, the tracking information is shared from generator to analyzer in real time. Additionally, the analyzer informs the generator when a tone has been successfully acquired, so that the generator can move to the next tone.

When a generator waveform or external source is used as the stimulus, or when a file is used as the analyzer input, a table that defines the sweep points must be provided to the analyzer. The analyzer then tracks the sweep (as defined in the **Sweep Table**) by monitoring one or more input channels. This method works well, unless the input signal is severely degraded.

When measuring from an external source, instead of using the generator, you will play an audio signal from the DUT.

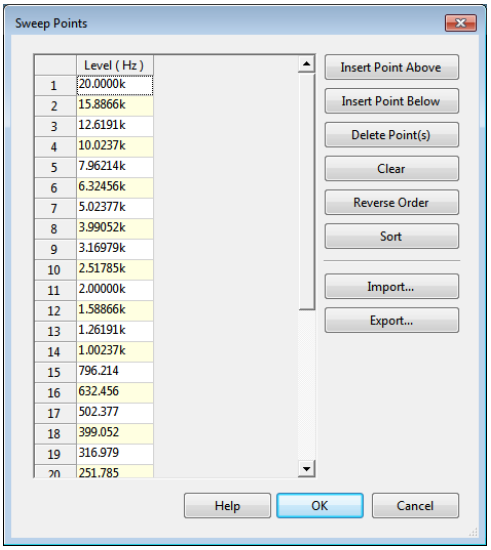
For a stepped frequency sweep, you must select one of the AP stepped frequency sweep test signals listed on the External Source panel, which are provided on-disc in several different formats. See “AP Test Signals on Disc” on page 174 and “Stepped Frequency Sweep Signals” on page 509.

When you have the test signal ready on the DUT and have selected it in APx500, click **Start** and play the test signal from the DUT. The analyzer will wait for the pilot tone, and then wait for each frequency point in the sweep. The signal is acquired and analyzed at each point, and the results are graphed.

If the wait for the pilot tone or any sweep point exceeds 60 seconds, the sweep will time out and end.

Viewing or Editing the Sweep Points tables

Stepped sweeps are defined by the points in a sweep table.

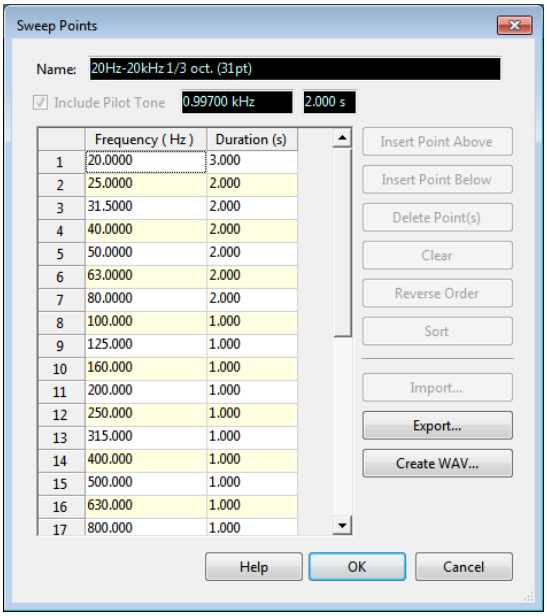


Generator Sweep Points Table (internal sweep).

For internal sweeps (generated in real time), the dialog above allows you to specify generator frequencies and to add, delete and order the sweep points. The generator **Sweep Points** table can be exported to an XLS or CSV file; compatible files can be imported into the current sweep.

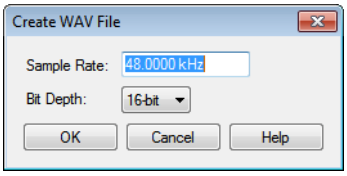
For external sweeps (from generator waveforms, file input or external source), an analyzer **Sweep Table** is necessary for the analyzer to track the sweep. The dialog below has the features of the internal generator **Sweep Points** dialog, with the addition of pilot tone settings and tone duration settings. You also have the

option of creating a WAV file from the **Sweep Table**, as discussed next.



Analyzer Sweep Points Table (external sweep)

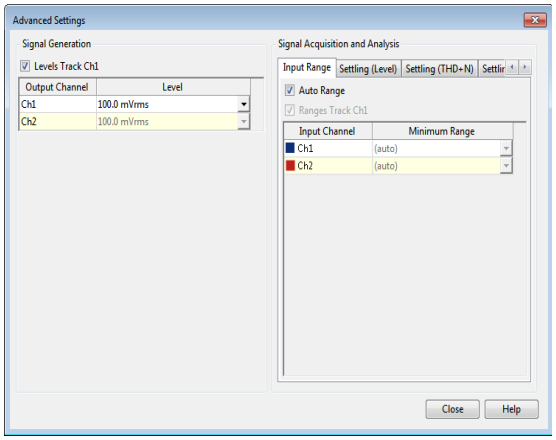
Create a WAV file from a Sweep Table



When you click **Create WAV File...** in the Sweep Points dialog, this dialog opens. Enter the **Sample Rate** (any sample rate from 22 kS/s to 192 kS/s) and **Bit Depth** (either 16-bit or a 24-bit) you require for your sweep WAV file. Click **OK** to open a file browser to save the file.

Advanced Settings for Stepped Frequency Sweep Measurements

The default settings here are appropriate for most measurements. You may want to make minor adjustments for special situations.



Signal Generation

You can set different output levels or DC offsets for each channel here.

If **Levels Track Ch1** is checked (the default), the generator output level and DC offset values for channel 1 are copied to the other channels, and values for channels greater than 1 cannot be edited. Any changes made to channel 1 are reproduced in the other channels.

To set individual generator output channel levels or offsets, uncheck the **Levels Track Ch1** checkbox and enter values in the output channel Level or Offset fields.

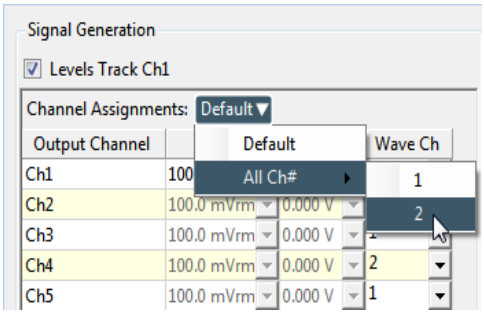
Set Channel Assignments for special waveforms

For special waveforms like Sine Dual, you can specify the channel assignment.

Set Channel Assignments for generator waveforms

For stereo or multichannel generator waveform files, you can set channel mapping. By default, channel 1 audio is mapped to channel 1 output, channel 2 to channel 2, and so on. If the number of channels in the waveform file is less than the number of output channels, the waveform channels resume numbering at 1 and wrap to the next available output channel. These assignments can be remapped by changing individual settings in the **Wave Ch** column, or by selection one of

several presets from the **Set Channel Assignments menu.**



Signal Acquisition and Analysis

Input Range tab

Digital inputs do not require ranging, and the Input Range tab is unavailable when inputs are set to a digital format.

Auto Range

If **Auto Range** is checked (the default), each analog channel determines its input range automatically, based on the level of the input signal. If the input signal level changes beyond a ranging threshold, **Auto Range** will cause the input ranging circuits to move up or down for proper ranging.

Fixed Range

Alternatively, you can set a fixed minimum range for the analog inputs. Uncheck **Auto Range** and enter the expected maximum level in the **Minimum Range** filed. The software will choose the lowest range that will accommodate that level; this becomes the minimum range available, and the analyzer will not reset to a lower range. Fixed range is best for signals that vary rapidly, including tone bursts, perceptual audio speech samples, and program material such as voice or music. See **When autoranging isn't suitable** on page 626.

If a signal greater then the **Minimum Range** is presented to the analog input, the ranging circuits will range up to accommodate the higher level, and then drop again when the signal level falls. The range will not be set below the **Minimum Range**.

Ranges Track Ch1

If **Ranges Track Ch1** is checked (the default), the fixed input range setting for channel 1 is copied to the other channels, and range settings for channels greater than 1 cannot be edited. Any changes made to channel 1 are reproduced in the other channels. If **Ranges Track Ch1** is not checked, the fixed input range setting for each channel can be set independently.

Set Channel Assignments for special waveforms

For special waveforms like Sine Dual, you can specify the channel assignment.

Read more about ranging in Chapter 98.

Stepped Frequency Sweep Signals

Most of these sweeps are provided on playable optical disc media available from Audio Precision. These sweeps are available at several sample rates, several signal levels, several channels counts, and as linear and a choice of coded formats.

- APx-DVD1 has a range of sample rates and formats.
- APx-CD1 has only 16-bit linear files at 44.1 kHz sample rate.

A subset of somewhat shorter versions of the linear files are also available for generator waveform use. These are available on the AP Web site at ap.com.

Coded waveforms (such as Dolby or DTS) must be decoded in the DUT before analysis.

The six sweeps at the end of the list are Dolby or DTS proprietary stimulus signals, available on discs provided to licensees of Dolby or DTS technologies.

The Audio Precision sweeps begins with a short (2 second to 5 second) 997 Hz pilot tone at –20 dBFS. This sets the analyzer to its ready condition. The rest of the sweep is at 0 dBFS, –1 dBFS or –20 dBFS, as selected.

Sweep definitions for stimulus signals on Audio Precision media.

20 Hz–20 kHz 1 octave (11 point)

Media: APx-CD1, APx-DVD1

Description: Pilot tone, followed by 11 pt sweep (approximate 1-octave centers) 20 Hz–20 kHz @ 0 dBFS.

Sweep duration: 00:15.

20 Hz–20 kHz 1 octave (11 point) (slow)

Media: APx–CD1, APx-DVD1

Description: Pilot tone, followed by 11 pt sweep (approximate 1-octave centers) 20 Hz–20 kHz @ 0 dBFS. Low-frequency tones are long to allow stabilization of DUT and analyzer.

Sweep duration: 00:33.

20 Hz–20 kHz 1/3 octave (31 point)

Media: APx-CD1, APx-DVD1

Description: Pilot tone, followed by 31 pt sweep (approximate 1/3-octave centers) 20 Hz–20 kHz @ 0 dBFS.

Sweep duration: 00:39.

20 Hz–20 kHz 1/3 octave (31 point) (slow)

Media: APx-CD1, APx-DVD1

Description: Pilot tone, followed by 31 pt sweep (approximate 1/3-octave centers) 20 Hz–20 kHz @ 0 dBFS. Low-frequency tones are long to allow stabilization of DUT and analyzer.

Sweep duration: 01:33.

20 Hz–20 kHz 1/6 octave (61 point)

Media: APx-CD1, APx-DVD1

Description: Pilot tone, followed by 61 pt sweep (approximate 1/6-octave centers) 20 Hz–20 kHz @ 0 dBFS.

Sweep duration: 1:15.

20 Hz–20 kHz 1/6 octave (61 point) (slow)

Media: APx-CD1, APx-DVD1

Description: Pilot tone, followed by 61 pt sweep (approximate 1/6-octave centers) 20 Hz–20 kHz @ 0 dBFS. Low-frequency tones are long to allow stabilization of DUT and analyzer.

Sweep duration: 03:03.

20 Hz–22 kHz 1/6 octave (62 point)

Description: Pilot tone, followed by 62 pt sweep (approximate 1/6-octave centers) 20 Hz–22.4 kHz @ 0 dBFS.

Proprietary sweep definitions from Dolby.

You must be a licensee and obtain stimulus files or media from Dolby Labs.

Dolby 1/6 Oct. sweep (AutoTest 2.0) 20 Hz–20 kHz

Proprietary sweep available from Dolby for licensees of Dolby technology.

Dolby 1/6 Oct. sweep 20 kHz–20 Hz (60pt)

Proprietary sweep available from Dolby for licensees of Dolby technology.

Dolby TrueHD 20 Hz–24 kHz (60pt)

Proprietary sweep available from Dolby for licensees of Dolby technology.

Dolby TrueHD 24 kHz–20 Hz (60pt)

Proprietary sweep available from Dolby for licensees of Dolby technology.

Dolby TrueHD 20 Hz–48 kHz (60pt)

Proprietary sweep available from Dolby for licensees of Dolby technology.

Dolby TrueHD 48 kHz–20 Hz (60pt)

Proprietary sweep available from Dolby for licensees of Dolby technology.

Dolby TrueHD 20 Hz–96 kHz (60pt)

Proprietary sweep available from Dolby for licensees of Dolby technology.

Dolby TrueHD 96 kHz–20 Hz (60pt)

Proprietary sweep available from Dolby for licensees of Dolby technology.

Dolby 1/6 Oct. sweep (Test DVD 1.5)

Proprietary sweep available from Dolby for licensees of Dolby technology.

Dolby 1/6 Oct. LFE (20–120 Hz) sweep (Test DVD 1.5)

Proprietary sweep available from Dolby for licensees of Dolby technology.

Proprietary sweep definitions from DTS.

You must be a licensee and obtain stimulus files or media from DTS.

DTS 20 Hz–20 kHz @ –20 dBFS (30 point)

Proprietary sweep available from DTS for licensees of DTS technology.

DTS 20 Hz–300 Hz @ –20 dBFS (15 point)

Proprietary sweep available from DTS for licensees of DTS technology.

DTS 20 Hz–40 kHz (31 point)

Proprietary sweep available from DTS for licensees of DTS technology.

Stepped Level Sweep
(Sequence Mode)

The **Stepped Level Sweep** measurement provides a sine wave stimulus signal that is moved across a range of levels in a series of points. The DUT output is acquired by the analyzer and processed for display. **Stepped Level Sweep** measurements are not supported in the **External Source** configuration.

Stepped Level Sweep results available in APx500 are:

- Level
- Gain
- Linearity
- THD+N Ratio
- THD+N Level
- THD Ratio
- THD Level
- THD+N Ratio vs. Measured Level
- THD+N Level vs. Measured Level
- THD Ratio vs. Measured Level
- THD Level vs. Measured Level
- SINAD
- Peak Level
- Average Jitter Level

All of these results are available from a single measurement.

Operation

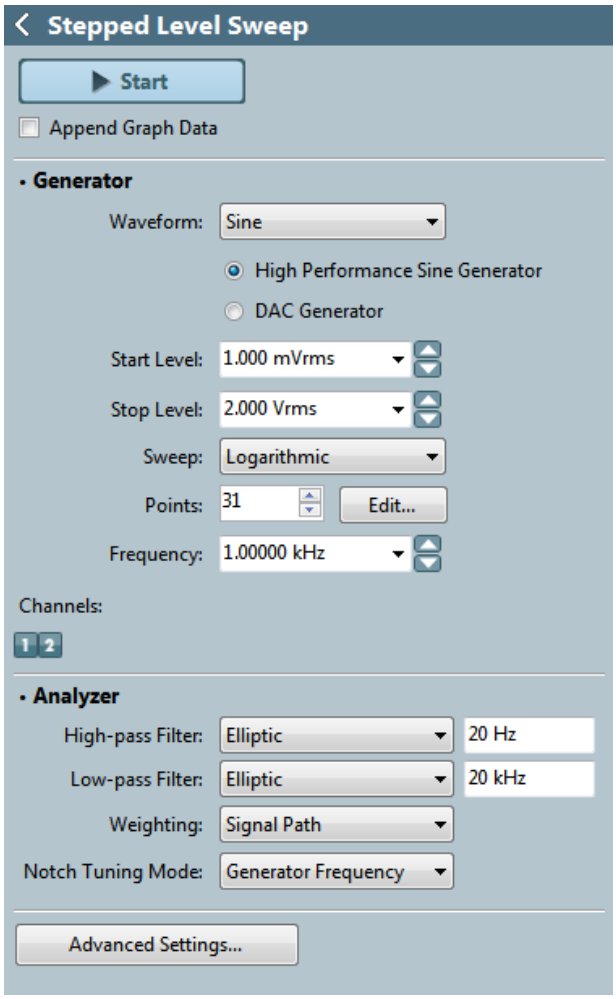
If you have not yet set up your test, first go to Signal Path Setup to select and configure inputs and outputs, references and other settings. See Chapter 7 for information about Signal Path Setup.

Running the Measurement

For a default sweep of 1 mVrms to 2 Vrms in 31 logarithmic steps, simply click **Start**. If you want to specify other parameters for your sweep, review **Generator** and **Analyzer** settings, discussed below. Click **Start** when you have approved the settings.

Append Graph Data

Measured data are grouped in a **Data Set**, which contains all acquisition results. Normally, the current graph data are deleted each time you start a new measurement. If the **Append Graph Data** box is checked, appended data are grouped as additional data sets, as are any imported data. The Data Sets Panel displays the Data Set(s) in the measurement,



and provides a visibility checkbox, a Time acquisition field, and a Notes field for each data set.

Read more about Data Sets beginning on page 649.

Generator

The configuration for this measurement must be the closed-loop configuration, using the APx generator as a stimulus.

See Chapter 6 for detailed information about the using the APx Generator and making Waveform, Level and Frequency settings.

High Performance Sine Generator / DAC Generator

For an APx555, you have the option of using the High Performance Sine Generator (very low distortion sine waves and wide bandwidth) or the DAC generator, which provides special sine waveforms. See High Performance Sine Generator starting on page 51 for a comparison of features.

Note: if Output Configuration is set to Bluetooth (A2DP Source profile), the Bluetooth hardware imposes a –54 dBFS cutoff. Signals below –54 dBFS are not transmitted. Please consider this when defining the range of levels for your Stepped Level sweep.

Start Level, Stop Level and Sweep Points

The sweep moves between two levels, set in the **Start Level** and **Stop Level** fields. The default levels are 1.000 mVrms and 2.000 Vrms.

The sweep moves in a specified number of step points, set in the **Points** field. The minimum is 2 points; maximum is 65,535. The default setting is 31.

The sweep point spacing is set by selecting one of the following choices in the Sweep field:

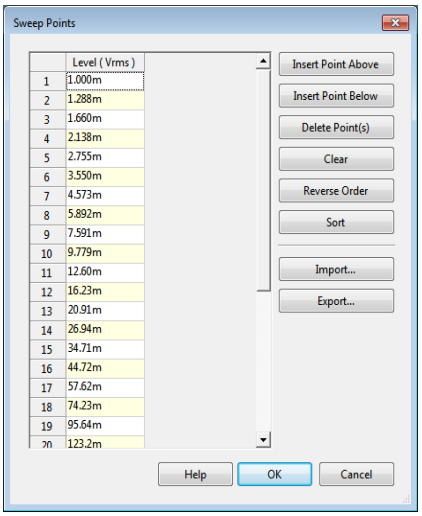
- **Logarithmic** (the default); use the **Points** field to set the number of logarithmically spaced points;
- **Linear**, which provides two methods of adjusting spacing: the **Points** field or the **Step Size** field; or
- **Custom**. Click **Edit** to open the **Sweep Points** dialog, where you can set points arbitrarily, or load or save sweep table files.

Viewing or Editing the Sweep Points table

You can view or edit the current generator sweep points at any time.

Click **Edit** to open the generator **Sweep Points** table. The table shows each sweep point and its corresponding level. You can edit this table to add or delete points, or to change the level of a point. Points can be sorted or reversed in order using the controls on the right.

A **Sweep Points** table can be saved as a *.csv file or as a Microsoft Excel *.xls file. A compatible *.csv or *.xls file can be opened and used as a **Sweep Points** table.



Analyzer

Filters

Local high-pass, low-pass and weighting filters are available for this measurement. The low-pass and high-pass filters, if selected, are applied to the entire measured signal and affect all the results. The weighting filter, if selected, affects only the distortion and distortion plus noise results. See Measurement Filters on page 619 for detailed information about the filters available locally for some measurements.

Low pass, high pass and weighting Signal Path Filters settings (see page 617) made in Signal Path Setup > Input/Output will also affect this measurement. When set, the local low-pass or high-pass filter setting overrides the corresponding Signal Path filter setting while the measurement is active. A local weighting filter setting does not override any Signal Path filter settings; the local weighting filter is applied in addition to any active Signal Path filters.

Notch Tuning Mode

For the harmonic distortion results, a tunable notch filter is used to remove the fundamental from the measurement. The filter can be tuned to:

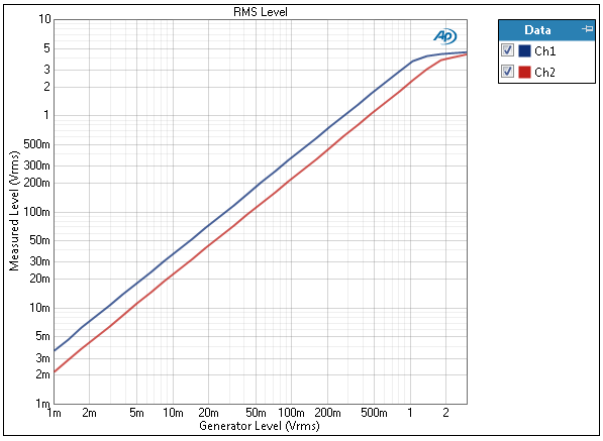
- **Generator Frequency**
  The current APx audio generator frequency. When the generator channels are outputting different frequencies (Split Frequency generation), the notch filter center is set to Frequency A. This mode is not available when using a generator waveform file.
- **Jitter Generator Frequency**
  The current APx jitter generator frequency, when jitter generation is available and enabled. Read more about jitter beginning on page 69.

- **Measured Frequency**
  The current measured frequency. When the analyzer channels are receiving different frequencies, the notch filter for each channel is centered on the frequency in that channel.
- **Fixed Frequency**
  A fixed frequency selected by the user. When **Fixed Frequency** is selected, a **Filter Frequency** entry field becomes available beneath the **Notch Tuning Mode** control.

Advanced Settings

If your test requires individual generator settings for each channel or special analyzer ranging or settling, click **Advanced Settings**. See Advanced Settings for Stepped Level Sweeps on page 519. See Chapter 104 for more information about units of measurement.

Stepped Level Sweep: Level



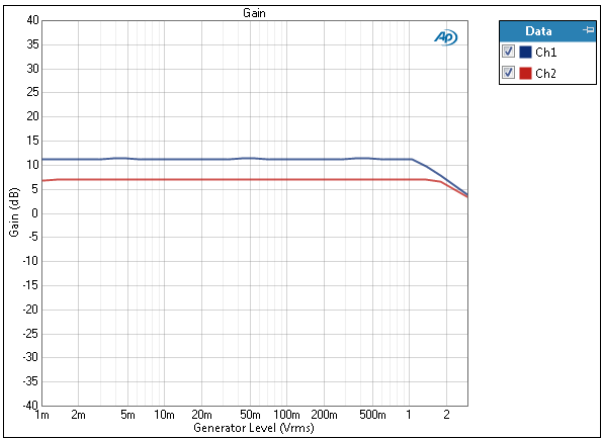
The Stepped Level Sweep: Level result uses a sine wave stimulus signal that is moved across a range of levels in a series of points. X axis is generator level; Y axis is DUT output level.

Units

Units available for Level are:

X-axis (analog)	*X-axis (digital)*	*Y-axis (analog)*	*Y-axis (digital)*
• Vrms	• dBFS	• Vrms	• dBFS
• Vp	• FS	• dBV	• FS
• Vpp	• %FS	• dBu	• %FS
• dBV	• dBrG	• dBrA	• dBrA
• dBu		• dBrB	• dBrB
• dBrG		• dBSPL1	• dBSPL1
• dBm		• dBSPL2	• dBSPL2
• W (watts)		• dBm	
		• W (watts)	

Stepped Level Sweep: Gain



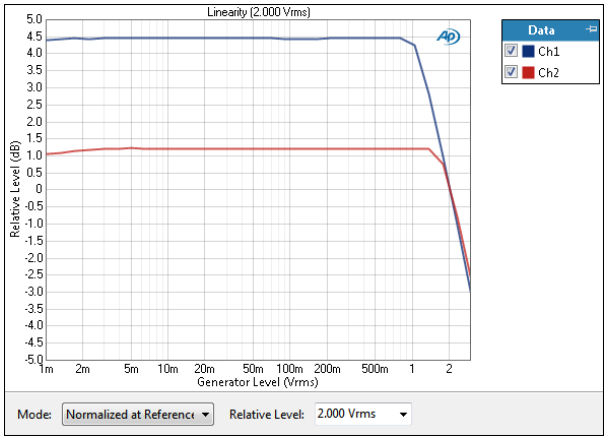
The Stepped Level Sweep: Gain result uses a sine wave stimulus signal that is moved across a range of levels in a series of points. X axis is generator level; Y axis is gain in the DUT. Gain is a ratio, calculated by dividing the DUT output level by the generator level.

Units

Y-axis units available for Gain are:

X-axis (analog)	*X-axis (digital)*	*Y-axis same-domain*	*Y-axis cross-domain*
• Vrms	• dBFS	• x/y	• FS/Vrms
• Vp	• FS	• %	• dB(FS/Vrms)
• Vpp	• %FS	• ppm	—or—
• dBV	• dBrG	• dB	• Vrms/FS
• dBu			• dB(Vrms/FS)
• dBrG			
• dBm			
• W (watts)			

Stepped Level Sweep: Linearity



The Stepped Level Sweep: Linearity result uses a sine wave stimulus signal that is moved across a range of levels in a series of points. X axis is generator level; Y axis is linearity in the DUT. Linearity is a ratio, calcu-

lated by dividing the DUT output level by the generator level, normalized for a chosen relative level.

Mode

Normalized at Reference

Enter a reference level for normalization in the **Relative Level** field. To calculate linearity, each channel is normalized so that a linearity of unity (0 dB) is at the same measured DUT output level. Set this reference level in the **Relative Level** field. Default **Relative Level** is 1 Vrms (analog), 1 FS (digital). You can change the **Relative Level** at any time (except after appending) and the graph will immediately be redrawn to reflect the new setting.

Auto-Centered in Limits

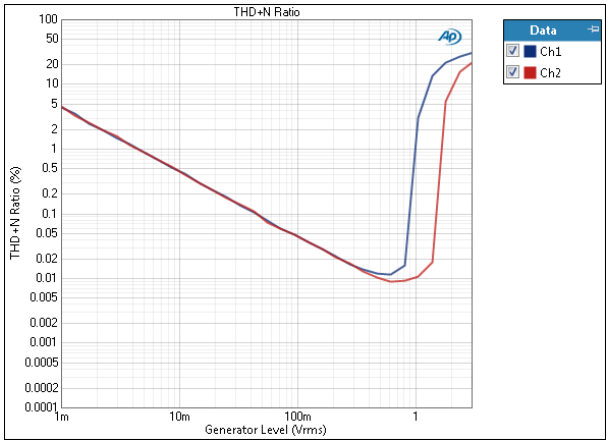
This control is only usable if upper and lower limits are set on this graph. In some applications, the general conformance of a trace to the shape of a set of limits is of importance, with absolute level less important or irrelevant. **Auto-Centered in Limits** centers the data trace relative to the upper and lower limits. To exclude possible exceptional data points at level extremes, the amplitude range that is considered for the centering operation is constrained by the values in the **Min** and **Max** fields.

Units

Units available for Linearity are:

X-axis (analog)	*X-axis (digital)*	*Y-axis*
• Vrms	• dBFS	• x/y
• Vp	• FS	• %
• Vpp	• %FS	• ppm
• dBV	• dBrG	• dB
• dBu		
• dBrG		
• dBm		
• W (watts)		

Stepped Level Sweep: THD+N Ratio



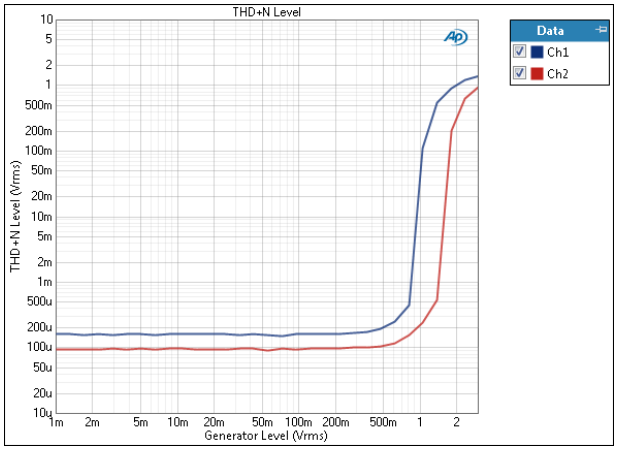
The Stepped Level Sweep: THD+N Ratio result uses a sine wave stimulus signal that is moved across a range of levels in a series of points. X axis is generator level; Y axis is DUT output THD+N ratio.

Units

Units available for THD+N Ratio are:

X-axis (analog)	*X-axis (digital)*	*Y-axis*
• Vrms	• dBFS	• x/y
• Vp	• FS	• %
• Vpp	• %FS	• ppm
• dBV	• dBrG	• dB
• dBu		
• dBrG		
• dBm		
• W (watts)		

Stepped Level Sweep: THD+N Level



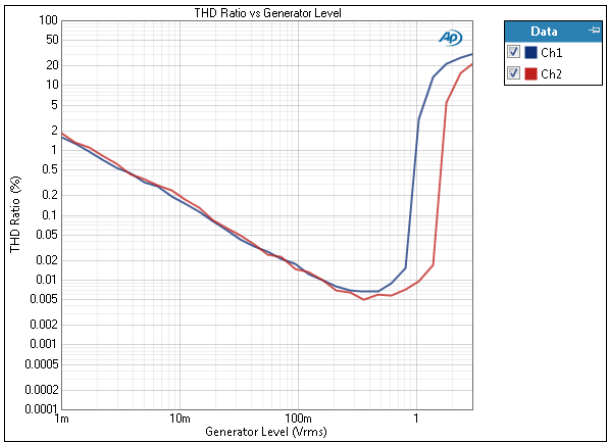
The Stepped Level Sweep: THD+N Level result uses a sine wave stimulus signal that is moved across a range of levels in a series of points. X axis is generator level; Y axis is DUT output THD+N level.

Units

Y-axis units available for THD+N Level are:

X-axis (analog)	*X-axis (digital)*	*Y-axis (analog)*	*Y-axis (digital)*
• Vrms	• dBFS	• Vrms	• dBFS
• Vp	• FS	• dBV	• FS
• Vpp	• %FS	• dBu	• %FS
• dBV	• dBrG	• dBrA	• dBrA
• dBu		• dBrB	• dBrB
• dBrG		• dBSPL1	• dBSPL1
• dBm		• dBSPL2	• dBSPL2
• W (watts)		• dBm	
		• W (watts)	

Stepped Level Sweep: THD Ratio



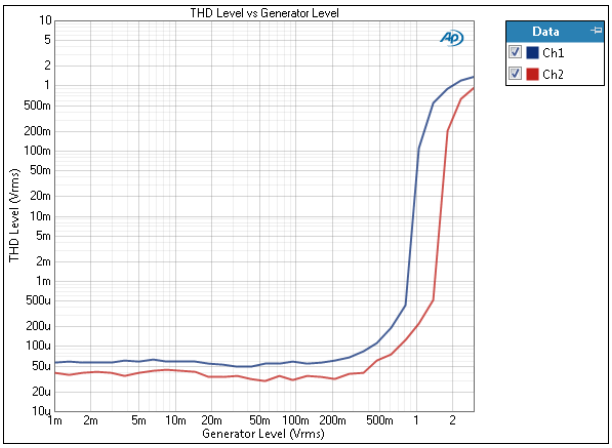
The Stepped Level Sweep: THD Ratio result uses a sine wave stimulus signal that is moved across a range of levels in a series of points. X axis is generator level; Y axis is DUT output THD ratio.

Units

Units available for THD Ratio are:

X-axis (analog)	*X-axis (digital)*	*Y-axis*
• Vrms	• dBFS	• x/y
• Vp	• FS	• %
• Vpp	• %FS	• ppm
• dBV	• dBrG	• dB
• dBu		
• dBrG		
• dBm		
• W (watts)		

Stepped Level Sweep: THD Level



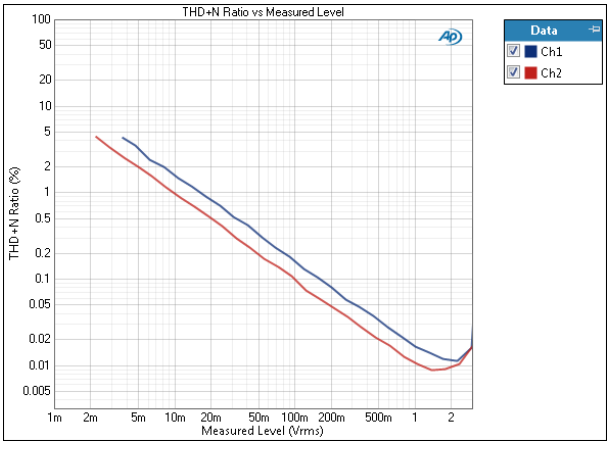
The Stepped Level Sweep: THD Level result uses a sine wave stimulus signal that is moved across a range of levels in a series of points. X axis is generator level; Y axis is DUT output THD level.

Units

Y-axis units available for THD Level are:

X-axis (analog)	*X-axis (digital)*	*Y-axis (analog)*	*Y-axis (digital)*
• Vrms	• dBFS	• Vrms	• dBFS
• Vp	• FS	• dBV	• FS
• Vpp	• %FS	• dBu	• %FS
• dBV	• dBrG	• dBrA	• dBrA
• dBu		• dBrB	• dBrB
• dBrG		• dBSPL1	• dBSPL1
• dBm		• dBSPL2	• dBSPL2
• W (watts)		• dBm	
		• W (watts)	

Stepped Level Sweep: THD+N Ratio vs. Measured Level



The Stepped Level Sweep: THD+N Ratio vs. Measured Level result uses a sine wave stimulus signal that is moved across a range of levels in a series of points. X

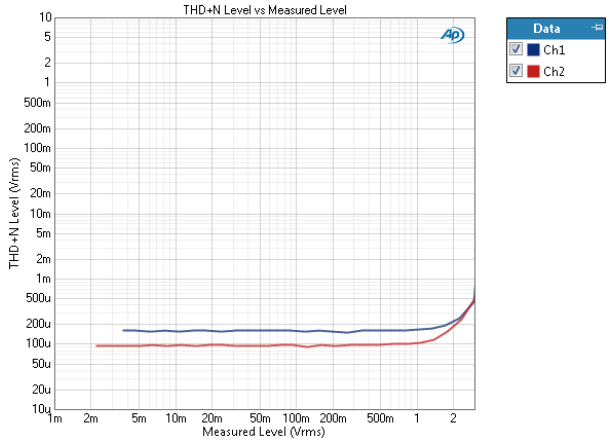
axis is measured DUT output level; Y axis is DUT output THD+N ratio.

Units

Units available for THD+N Ratio vs. Measured Level are:

	X-axis (digital)	*Y-axis*
• Vrms	• dBFS	• x/y
• dBV	• FS	• %
• dBu	• %FS	• ppm
• dBrA	• dBrA	• dB
• dBrB	• dBrB	
• dBSPL1	• dBSPL1	
• dBSPL2	• dBSPL2	
• dBm		
• W (watts)		

Stepped Level Sweep: THD+N Level vs. Measured Level



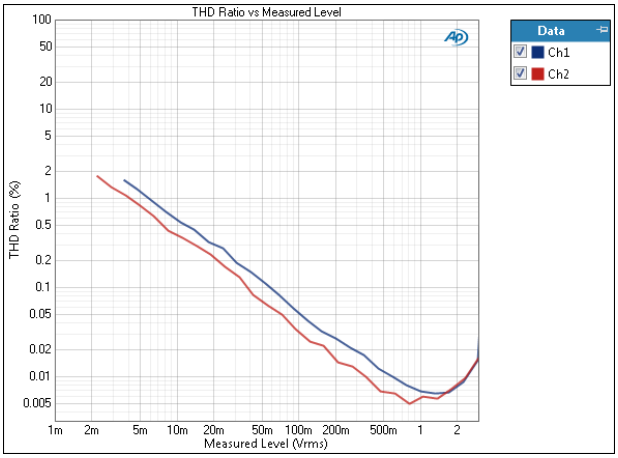
The Stepped Level Sweep: THD+N Level vs. Measured Level result uses a sine wave stimulus signal that is moved across a range of levels in a series of points. X axis is measured DUT output level; Y axis is DUT output THD+N level.

Units

Y-axis units available for THD+N Level vs. Measured Level are:

X-axis (analog)	*X-axis (digital)*	*Y-axis (analog)*	*Y-axis (digital)*
• Vrms	• dBFS	• Vrms	• dBFS
• dBV	• FS	• dBV	• FS
• dBu	• %FS	• dBu	• %FS
• dBrA	• dBrA	• dBrA	• dBrA
• dBrB	• dBrB	• dBrB	• dBrB
• dBSPL1	• dBSPL1	• dBSPL1	• dBSPL1
• dBSPL2	• dBSPL2	• dBSPL2	• dBSPL2
• dBm		• dBm	
• W (watts)		• W (watts)	

Stepped Level Sweep: THD Ratio vs. Measured Level



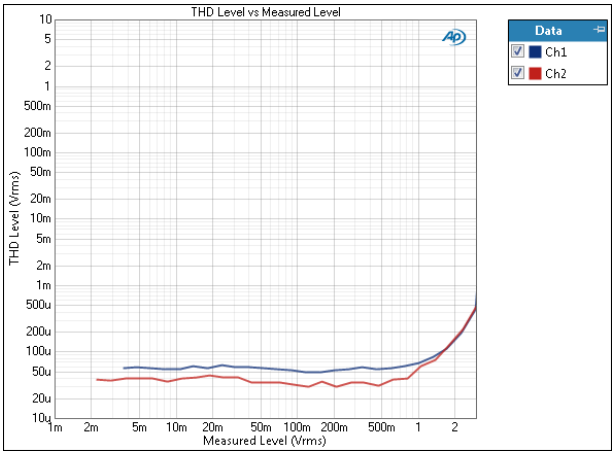
The Stepped Level Sweep: THD Ratio vs. Measured Level result uses a sine wave stimulus signal that is moved across a range of levels in a series of points. X axis is measured DUT output level; Y axis is DUT output THD ratio.

Units

Units available for THD Ratio vs. Measured Level are:

X-axis (analog)	*X-axis (digital)*	*Y-axis*
• Vrms	• dBFS	• x/y
• dBV	• FS	• %
• dBu	• %FS	• ppm
• dBrA	• dBrA	• dB
• dBrB	• dBrB	
• dBSPL1	• dBSPL1	
• dBSPL2	• dBSPL2	
• dBm		
• W (watts)		

Stepped Level Sweep: THD Level vs. Measured Level



The Stepped Level Sweep: THD Level vs. Measured Level result uses a sine wave stimulus signal that is

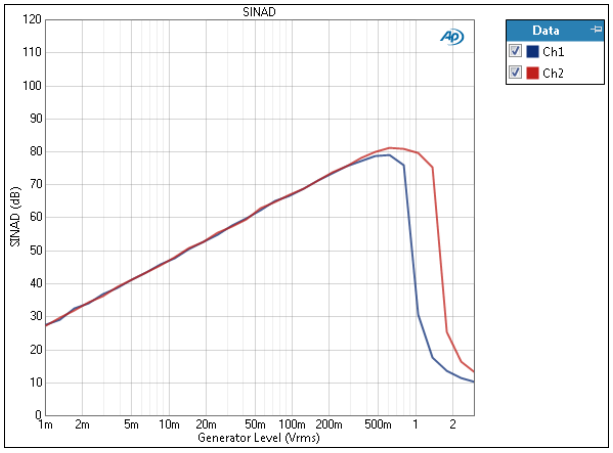
moved across a range of levels in a series of points. X axis is measured DUT output level; Y axis is DUT output THD level.

Units

Y-axis units available for THD Level vs. Measured Level are:

X-axis (analog)	*X-axis (digital)*	*Y-axis (analog)*	*Y-axis (digital)*
• Vrms	• dBFS	• Vrms	• dBFS
• dBV	• FS	• dBV	• FS
• dBu	• %FS	• dBu	• %FS
• dBrA	• dBrA	• dBrA	• dBrA
• dBrB	• dBrB	• dBrB	• dBrB
• dBSPL1	• dBSPL1	• dBSPL1	• dBSPL1
• dBSPL2	• dBSPL2	• dBSPL2	• dBSPL2
• dBm		• dBm	
• W (watts)		• W (watts)	

Stepped Level Sweep: SINAD



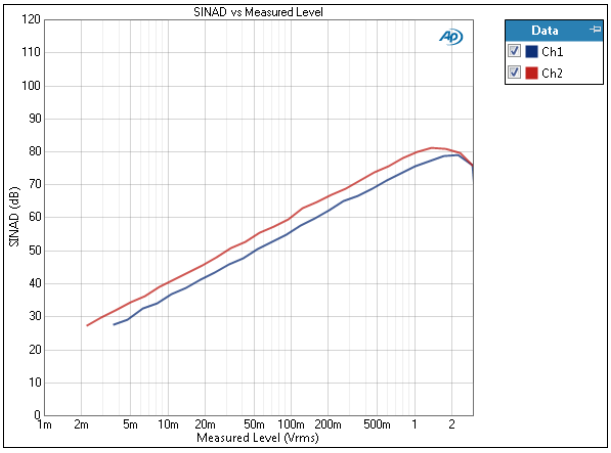
The Stepped Level Sweep: SINAD result uses a sine wave stimulus signal that is moved across a range of levels in a series of points. X axis is generator level; Y axis is DUT output SINAD ratio.

Units

Units available for SINAD are:

X-axis (analog)	*X-axis (digital)*	*Y-axis*
• Vrms	• dBFS	• x/y
• Vp	• FS	• dB
• Vpp	• %FS	
• dBV	• dBrG	
• dBu		
• dBrG		
• dBm		
• W (watts)		

Stepped Level Sweep: SINAD vs. Measured Level



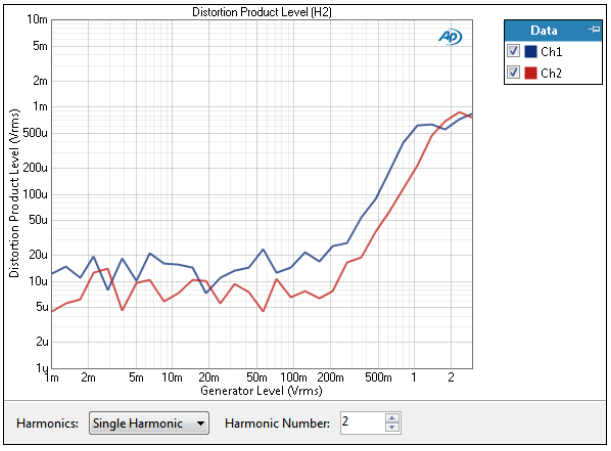
The Stepped Level Sweep: SINAD vs. Measured Level result uses a sine wave stimulus signal that is moved across a range of levels in a series of points. X axis is measured level; Y axis is DUT output SINAD ratio.

Units

Units available for SINAD vs. Measured Level are

X-axis (analog)	*X-axis (digital)*	*Y-axis*
• Vrms	• dBFS	• x/y
• dBV	• FS	• dB
• dBu	• %FS	
• dBrA	• dBrA	
• dBrB	• dBrB	
• dBSPL1	• dBSPL1	
• dBSPL2	• dBSPL2	
• dBm		
• W (watts)		

Stepped Level Sweep: Distortion Product Level



The Stepped Level Sweep Distortion Product Level result provides a graphical display of the selected harmonic distortion products present in each channel. In this result the level of the selected harmonic distor-

tion product in the DUT output is plotted against generator level.

Select the harmonic distortion product(s) you wish to view using the **Harmonics** controls, described below.

Harmonics

For a graph of the level of one specific harmonic product, select **Single Harmonic** and set the **Harmonic Number** to your choice, from 2nd harmonic to 20th harmonic.

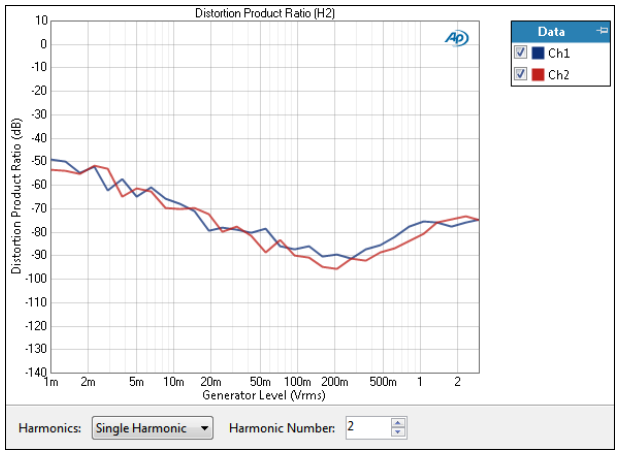
For a graph of the level of the sum of several or all harmonic products (from **H2** through **H20**), select **Sum of Harmonics**, then click **Select Harmonics...** to open a result definition dialog.

Units

Units available for Stepped Level Sweep Distortion Product Level are:

X-axis (analog)	*X-axis (digital)*	*Y-axis (analog)*	*Y-axis (digital)*
• Vrms	• dBFS	• Vrms	• dBFS
• Vp	• FS	• dBV	• FS
• Vpp	• %FS	• dBu	• %FS
• dBV	• dBrG	• dBrA	• dBrA
• dBu		• dBrB	• dBrB
• dBrG		• dBSPL1	• dBSPL1
• dBm		• dBSPL2	• dBSPL2
• W (watts)		• dBm	
		• W (watts)	

Stepped Level Sweep: Distortion Product Ratio



The Stepped Level Sweep Distortion Product Ratio result provides a graphical display of the selected harmonic distortion products present in each channel. In this result the ratio of the level of the selected harmonic distortion product to the fundamental in the DUT output is plotted against generator level.

Select the harmonic distortion product(s) you wish to view using the **Harmonics** controls, described below.

Harmonics

For a graph of the ratio of one specific harmonic product to the signal level, select **Single Harmonic** and set the **Harmonic Number** to your choice, from 2nd harmonic to 20th harmonic.

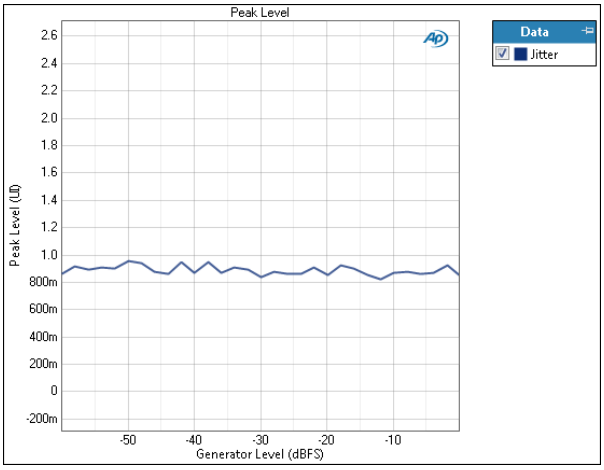
For a graph of the sum of the levels of any combination of harmonic products (from **H2** through **H20**), divided by the signal level, select **Sum of Harmonics**. Then click **Select Harmonics...** to open a result definition dialog.

Units

Units available for Stepped Level Sweep Distortion Product Ratio are:

X-axis (analog)	*X-axis (digital)*	*Y-axis*
• Vrms	• dBFS	• x/y
• Vp	• FS	• %
• Vpp	• %FS	• ppm
• dBV	• dBrG	• dB
• dBu		
• dBrG		
• dBm		
• W (watts)		

Peak Level



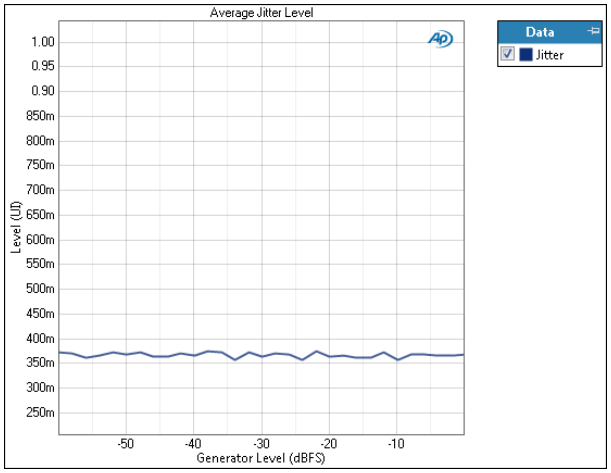
The **Stepped Level Sweep: Peak Level** result uses a sine wave stimulus signal that is moved across a range of levels in a series of points. X axis is generator level; Y axis is the bandpass filtered DUT output level, scaled in peak units.

Units

Units available for Peak Level are:

X-axis (analog)	*X-axis (digital)*	*Y-axis (analog)*	*Y-axis (digital)*
• Vrms	• dBFS	• V	• D
• Vp	• FS		
• Vpp	• %FS		
• dBV	• dBrG		
• dBu		*Y-axis (jitter)*	
• dBrG		• UI	
• dBm		• dBUI	
• W (watts)		• s	

Average Jitter Level



The **Stepped Level Sweep: Average Jitter Level** result uses a sine wave stimulus signal that is moved across a range of levels in a series of points. X axis is generator level; Y axis is the bandpass filtered DUT output jitter level, scaled in average units.

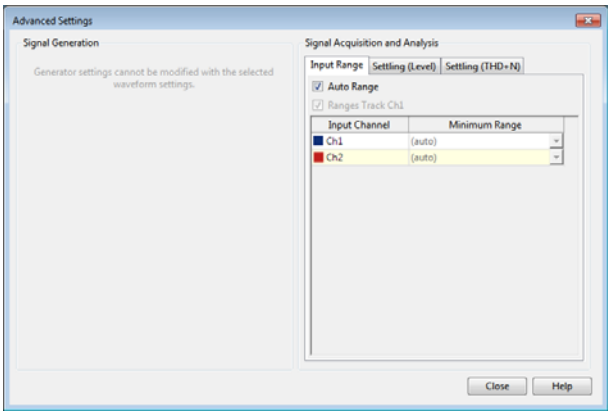
Units

Units available for Average Jitter Level are:

X-axis (analog)	*X-axis (digital)*	*Y-axis (jitter)*
• Vrms	• dBFS	• UI
• Vp	• FS	• dBUI
• Vpp	• %FS	• s
• dBV	• dBrG	
• dBu		
• dBrG		
• dBm		
• W (watts)		

Advanced Settings for Stepped Level Sweep Measurements

The default settings here are appropriate for most measurements. You may want to make minor adjustments for special situations.



Signal Generation

Generator output levels cannot be modified for Stepped Level Sweeps, and the channel level controls are not available.

Set Channel Assignments for special waveforms

For special waveforms like Sine Dual, you can specify the channel assignment.

Signal Acquisition and Analysis

Input Range tab

Digital inputs do not require ranging, and the Input Range tab is unavailable when inputs are set to a digital format.

Auto Range

If **Auto Range** is checked (the default), each analog channel determines its input range automatically, based on the level of the input signal. If the input signal level changes beyond a ranging threshold, **Auto Range** will cause the input ranging circuits to move up or down for proper ranging.

Fixed Range

Alternatively, you can set a fixed minimum range for the analog inputs. Uncheck **Auto Range** and enter the expected maximum level in the **Minimum Range** filed. The software will choose the lowest range that will accommodate that level; this becomes the minimum range available, and the analyzer will not reset to a lower range. Fixed range is best for signals that vary rapidly, including tone bursts, perceptual audio speech samples, and program material such as voice or music. See **When autoranging isn't suitable** on page 626.

If a signal greater then the **Minimum Range** is presented to the analog input, the ranging circuits will range up to accommodate the higher level, and then drop again when the signal level falls. The range will not be set below the **Minimum Range**.

Ranges Track Ch1

If **Ranges Track Ch1** is checked (the default), the fixed input range setting for channel 1 is copied to the other channels, and range settings for channels greater than 1 cannot be edited. Any changes made to channel 1 are reproduced in the other channels. If **Ranges Track Ch1** is not checked, the fixed input range setting for each channel can be set independently.

Settling tabs

Two or three Settling tabs are available for stepped sweep measurement results. These tabs may have different parameters and default values depending upon the result to which they're attached.

Read more about ranging and settling in Chapter 98.

TEDS Data

Reading of TEDS data requires the APx1701 Transducer Test Interface accessory.

Also see More About TEDS, below; and using TEDS for microphone calibration on pages 60 and 64.

The TEDS Data measurement provides a tabular display of the TEDS data read from a connected TEDS-enabled measurement microphone.

Preliminary conditions

An APx1701 must be connected to your PC and APx analyzer. A TEDS-enabled microphone must be connected to an APx1701 unbalanced microphone input. Read about the APx1701 on page 12.

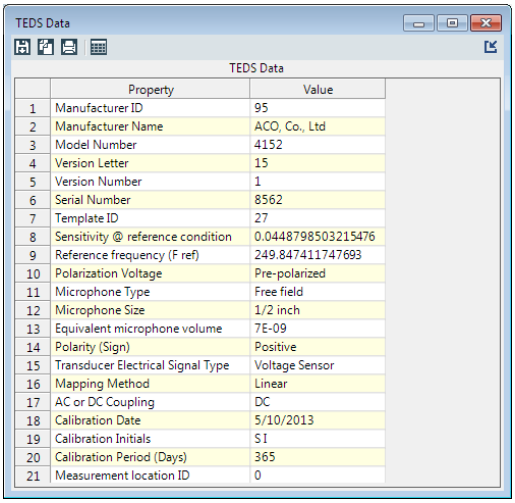
TEDS Source

Select the microphone to read TEDS data from. Choices are:

- Mic 1 Unbalanced
- Mic 2 Unbalanced

Making the Measurement

Click **Start**. The results panel will be populated with TEDS data.



	Property	Value
1	Manufacturer ID	95
2	Manufacturer Name	ACO, Co., Ltd
3	Model Number	4152
4	Version Letter	15
5	Version Number	1
6	Serial Number	8562
7	Template ID	27
8	Sensitivity @ reference condition	0.0448798503215476
9	Reference frequency (F ref)	249.847411747693
10	Polarization Voltage	Pre-polarized
11	Microphone Type	Free field
12	Microphone Size	1/2 inch
13	Equivalent microphone volume	7E-09
14	Polarity (Sign)	Positive
15	Transducer Electrical Signal Type	Voltage Sensor
16	Mapping Method	Linear
17	AC or DC Coupling	DC
18	Calibration Date	5/10/2013
19	Calibration Initials	SI
20	Calibration Period (Days)	365
21	Measurement location ID	0

Export

The TEDS data can be exported in Excel or CSV format.

More About TEDS

Some measurement microphones and other transducers store Transducer Electronic Data Sheet (TEDS) information that can be read by connected devices. The Audio Precision APx1701 Transducer Test Interface can read data from TEDS 0.9 and TEDS 1.0 devices connected to either of the unbalanced microphone inputs. This data is passed to the APx500 software via the USB connection for display, reporting, and entry into calibration dialogs.

The APx References microphone calibration dialogs can read TEDS data when an APx1701 is selected as the connector in Signal Path Setup > Input Configuration. The Sequence Mode TEDS Data measurement can read a broad range of microphone TEDS data.

For R&D testing, these data are used as data-sheet values in calibration reports. For production testing in controlled production setups and environments, the TEDS data can be used in lieu of actual calibration values.

TEDS is described in detail in the IEEE 1451 family of standards.

APx stores TEDS FDL, TDL and manid.txt files in the \AudioPrecision\APx500 4.4\TEDS\ directory. If the APx does not have an FDL or TDL file required by a TEDS device, the software shall raise an error indicating that it is not currently supported by the APx. Additional TEDS FDL and TDL files can be added as necessary.

THD+N
(Sequence Mode)

The THD+N measurement provides single-value results that show the THD+N (total distortion plus noise) in the output signal from each DUT channel, as measured at each analyzer input. The THD and Noise values are also reported in separate results. For more information about THD+N go to page 527.

THD+N results available in APx500 are:

- THD+N Ratio
- THD+N Level
- THD Ratio
- THD Level
- Noise Ratio
- Noise Level
- Distortion Product Level
- Distortion Product Ratio

Operation

If you have not yet set up your test, first go to Signal Path Setup to select and configure inputs and outputs, references and other settings. See Chapter 7 for information about Signal Path Setup.

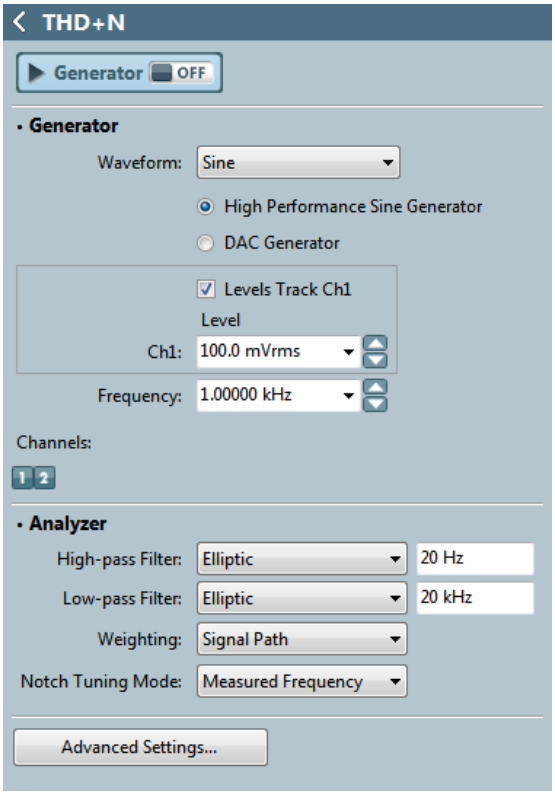
Running the measurement

To measure THD+N Ratio, click the Generator button to **On**. The generator will output the test signal to the DUT on the selected generator channels. Read the THD+N Ratio of each channel from the meter bar display.

Generator

The typical configuration for this measurement is the closed-loop configuration, using the APx generator as a stimulus. Alternatively, you can make open-loop measurements, using either the External Source or File Input configuration.

See Chapter 6 for detailed information about the using the APx Generator and making Waveform, Level and Frequency settings, or for information about using External Source.



High Performance Sine Generator / DAC Generator

For an APx555, you have the option of using the High Performance Sine Generator (very low distortion sine waves and wide bandwidth) or the DAC generator, which provides special sine waveforms. See High Performance Sine Generator starting on page 51 for a comparison of features.

Analyzer

Filters

Local high-pass, low-pass and weighting filters are available for this measurement. See Measurement Fil-

ters on page 619 for detailed information about the filters available locally for some measurements.

Low pass, high pass and weighting Signal Path Filters settings (see page 617) made in Signal Path Setup > Input/Output will also affect this measurement. When set, the local low-pass or high-pass filter setting overrides the corresponding Signal Path filter setting while the measurement is active. A local weighting filter setting does not override any Signal Path filter settings; the local weighting filter is applied in addition to any active Signal Path filters.

Notch Tuning Mode

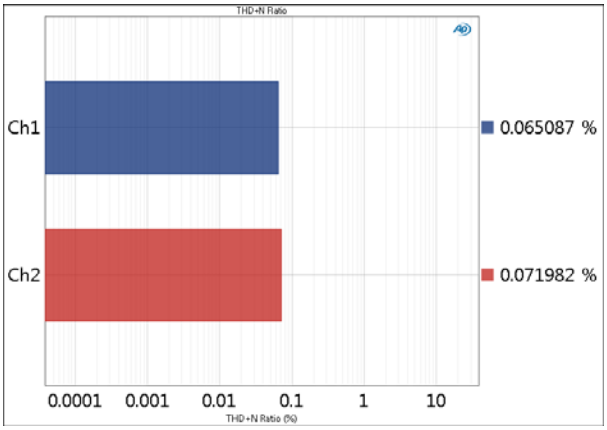
For the harmonic distortion results, a tunable notch filter is used to remove the fundamental from the measurement. The filter can be tuned to:

- **Generator Frequency**
  The current APx audio generator frequency. When the generator channels are outputting different frequencies (Split Frequency generation), the notch filter center is set to Frequency A. This mode is not available when using a generator waveform file.
- **Jitter Generator Frequency**
  The current APx jitter generator frequency, when jitter generation is available and enabled. Read more about jitter beginning on page 69.
- **Measured Frequency**
  The current measured frequency. When the analyzer channels are receiving different frequencies, the notch filter for each channel is centered on the frequency in that channel.
- **Fixed Frequency**
  A fixed frequency selected by the user. When **Fixed Frequency** is selected, a **Filter Frequency** entry field becomes available beneath the **Notch Tuning Mode** control.

Advanced Settings

If your test requires individual generator settings for each channel or special analyzer ranging or settling, click **Advanced Settings**. See Advanced Settings for single value measurement on page 351.

THD+N Ratio



The THD+N Ratio result provides a single-value measurement of the THD+N (total harmonic distortion plus noise) ratio in the output signal from each DUT channel.

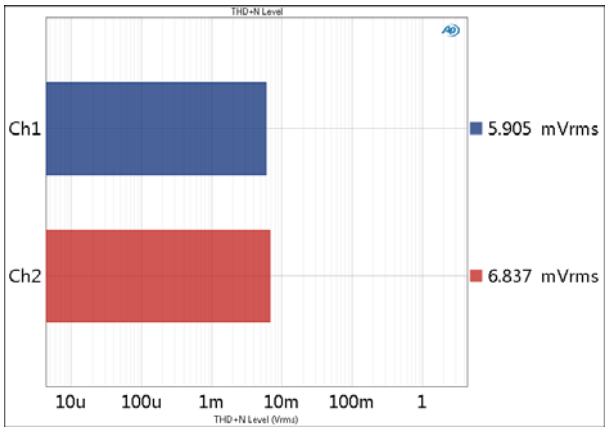
Units

Units available for THD+N Ratio are

- x/y
- %
- ppm
- dB

See Chapter 104 for more information about units of measurement.

THD+N Level



The THD+N Level result provides a single-value measurement of the THD+N (total harmonic distortion plus noise) level in the output signal from each DUT channel.

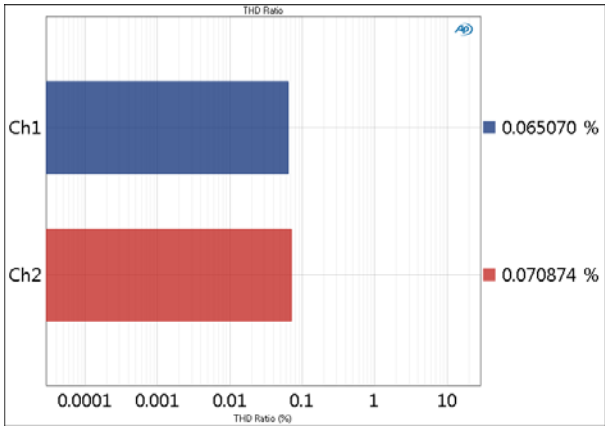
Units

Units available for THD+N Level are

Analog Signals	*Digital Signals*
• Vrms	• dBFS
• dBV	• FS
• dBu	• %FS
• dBrA	• dBrA
• dBrB	• dBrB
• dBSPL1	• dBSPL1
• dBSPL2	• dBSPL2
• dBm	
• W (watts)	

See Chapter 104 for more information about units of measurement.

THD Ratio



The THD Ratio result provides a single-value measurement of the THD (total harmonic distortion) ratio in the output signal from each DUT channel.

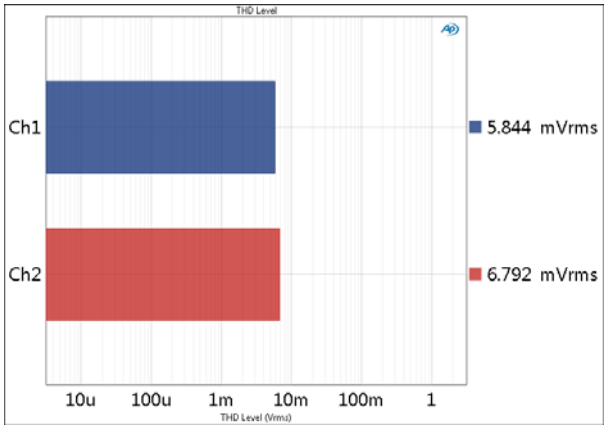
Units

Units available for THD Ratio are

- x/y
- %
- ppm
- dB

See Chapter 104 for more information about units of measurement.

THD Level



The THD Level result provides a single-value measurement of the THD (total harmonic distortion) level in the output signal from each DUT channel.

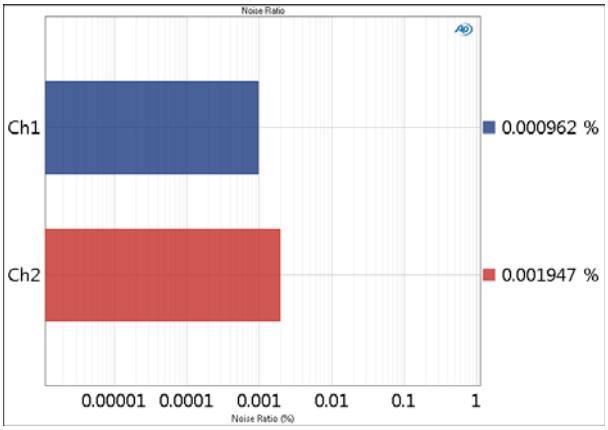
Units

Units available for THD Level are

Analog Signals	*Digital Signals*
• Vrms	• dBFS
• dBV	• FS
• dBu	• %FS
• dBrA	• dBrA
• dBrB	• dBrB
• dBSPL1	• dBSPL1
• dBSPL2	• dBSPL2
• dBm	
• W (watts)	

See Chapter 104 for more information about units of measurement.

Noise Ratio



The Noise Ratio result provides a single-value measurement of the Noise ratio in the output signal from each DUT channel.

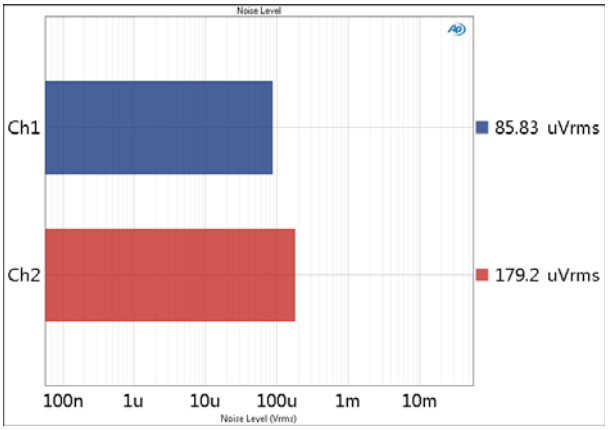
Units

Units available for Noise Ratio are

- x/y
- %
- ppm
- dB

See Chapter 104 for more information about units of measurement.

Noise Level



The Noise Level result provides a single-value measurement of the Noise level in the output signal from each DUT channel.

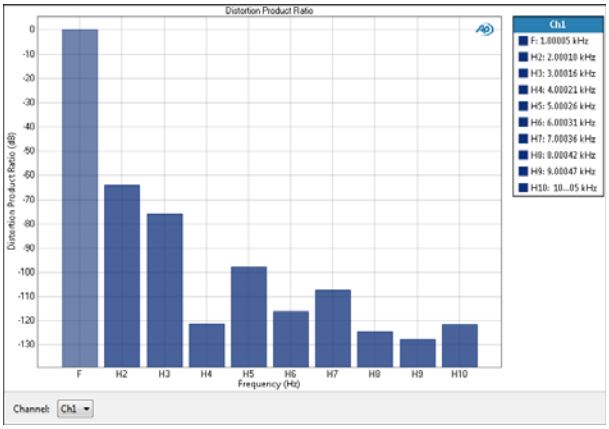
Units

Units available for Noise Level are

Analog Signals	*Digital Signals*
• Vrms	• dBFS
• dBV	• FS
• dBu	• %FS
• dBrA	• dBrA
• dBrB	• dBrB
• dBSPL1	• dBSPL1
• dBSPL2	• dBSPL2
• dBm	
• W (watts)	

See Chapter 104 for more information about units of measurement.

THD+N Distortion Product Ratio



The THD+N Distortion Product Ratio result displays a single-value measurement of the ratio of a distortion product (single harmonic) to the total signal for a selected channel, for the harmonics 2–10. The level displayed in each bar is the rms level of that harmonic tone divided by the rms level of the total signal.

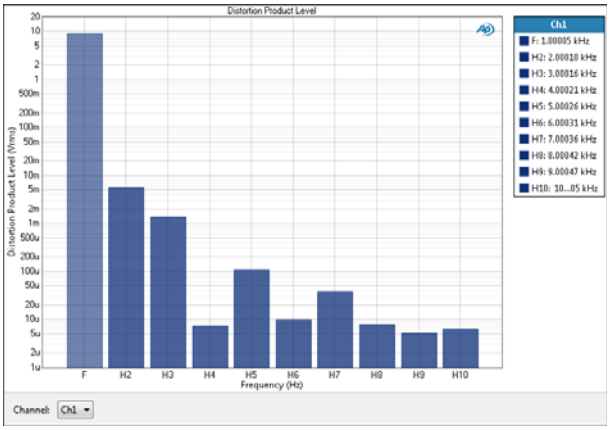
Units

Units available for THD+N Distortion Product Ratio are

- x/y
- %
- ppm
- dB

See Chapter 104 for more information about units of measurement.

THD+N Distortion Product Level



The THD+N Distortion Product Level result displays a single-value measurement of the rms level of distortion products (harmonics) for the harmonics 2–10. The level displayed in each bar is the rms level of that harmonic tone.

Units

Units available for THD+N Distortion Product Level are

Analog Signals	*Digital Signals*
• Vrms	• dBFS
• dBV	• FS
• dBu	• %FS
• dBrA	• dBrA
• dBrB	• dBrB
• dBSPL1	• dBSPL1
• dBSPL2	• dBSPL2
• dBm	
• W (watts)	

See Chapter 104 for more information about units of measurement.

More about THD+N

THD+N

THD+N is an abbreviation for Total Harmonic Distortion plus Noise. Total Harmonic Distortion is the sum of all the harmonic distortion products in a measurement, as opposed to selective harmonic distortion, which measures discrete harmonic products separately. THD+N adds noise to this figure.

THD+N is the most common distortion measurement for audio signals. In the APx500 meter (single value) THD+N and Stepped Sweep results, the measurement is performed in the conventional manner: a single sine wave stimulus tone is applied to the DUT input(s). In analysis, the bandwidth is usually controlled using filters; the stimulus tone is removed, and the signals that remain are reported as one measurement. These remainders are the total harmonic distortion (all harmonics within the bandwidth), noise from the DUT, and any other signals such as hum, spurious signals and other distortion products.

THD+N is an excellent and trusted quick check on several aspects of a system's overall performance, a quick, easy and robust mark of quality. However, because of its lack of discrimination, THD+N is not always the most useful measurement for troubleshooting.

THD+N is the reciprocal of SINAD (see page 498).

For THD+N Level, the measured distortion plus noise (the signal with the stimulus tone removed) is expressed as an rms level.

For THD+N Ratio, the rms level of the measured distortion plus noise (the signal with the stimulus tone removed) is divided by the rms level of the total signal. The result displayed on a bar meter or as a point on an XY sweep graph. THD+N ratio is most often stated in as a percentage or as a decibel value, where 0 dB represents the total signal.

Note: The total signal level is the rms level of the signal filtered by the filter selections defined in Signal Path Setup. The RMS Level result uses the measurement High-pass and Low-pass filter settings which override the filtering defined in Signal Path Setup for all of the measurement results. The total signal result is not presented along with the results of this measurement.

Also see More about THD (Total Harmonic Distortion, without noise results) on page 242.

THD+N Distortion Product Meter bar results

The harmonic distortion product results offer much more detail into the makeup of the harmonic distortion. For each channel, the rms level of stimulus tone is displayed as f, the fundamental, followed by the rms level of the harmonics 2f through 10f.

In the Distortion Product Level result, the bars show rms level. In the Distortion Product Ratio result, the bars represent the ratio of the rms level of the distortion product to the rms level of the entire signal.

Vdd Ramp
(Sequence Mode)

This measurement requires the PDM Option (see page 10).

Note: The Vdd Ramp measurement can sweep the Vdd voltage to 0 VDC to 3.6 VDC. However, early PDM Option modules do not support the Vdd Ramp feature. PDM modules that support the Vdd Ramp measurement operate under a firmware release of revision 2.00 or greater, indicated in the Help > About > Product Components field by FW: V2.0 in the PDM Module section. Contact your Audio Precision representative to upgrade your PDM Option module.

Overview

Vdd Ramp is measurement that characterizes the audio output of a device while its operating power is swept from one value to another. The range of voltage and the current capability are both modest, designed to provide swept operating voltage to a small integrated circuit. It is typically used with DUTs that are PDM transmitters.

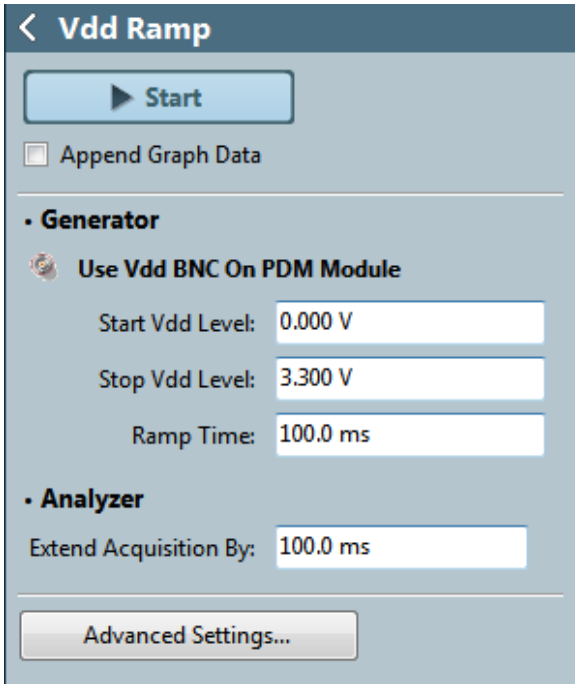
Other powered DUTs, such as converter or opamp devices, can be tested with Vdd Ramp, as long as the Vdd voltage and current limits of the PDM Vdd output are sufficient for the DUT. For DUTs that require more current, a unity gain DC coupled amplifier can be inserted between the APx generator and the DUT.

The Vdd Ramp result can reveal thumps or other transients in the DUT output related to Vdd changes.

Unique Characteristics

The Vdd Ramp measurement has some features notable in APx500:

- **Availability**
  The Vdd Ramp measurement is only available if the PDM Option is installed. See the Note above regarding early PDM Option modules.
- **Vdd Supply**
  When the Vdd measurement is started, DC power is applied to the PDM Vdd Supply connector, regardless of the current Signal Path settings. When the measurement is completed, DC power at the PDM Vdd Supply connector reverts to the Signal Path setting.
- **APx Generator**
  For a Vdd Ramp measurement, the output of the APx Generator does not appear at any analog or digital audio output. The analyzer output connector as specified in Signal Path Setup is irrelevant.



Append Graph Data

Measured data are grouped in a **Data Set**, which contains all acquisition results. Normally, the current

graph data are deleted each time you start a new measurement. If the **Append Graph Data** box is checked, appended data are grouped as additional data sets, as are any imported data. The Data Sets Panel displays the Data Set(s) in the measurement, and provides a visibility checkbox, a Time acquisition field, and a Notes field for each data set.

Read more about Data Sets beginning on page 649.

Generator

When this measurement is selected, the PDM Vdd Supply is switched to **Off**, if currently **On**. Click the **Start** button to begin the measurement. The Vdd Supply is switched to **On**, and the dc voltage on the Vdd connector is swept from the **Start Value** to the **Stop** Value, in a linear sweep which has the duration set in Ramp Time.

Start Vdd Level

Set the Vdd start voltage here. Start Value can be lower or higher than Stop Value.

Stop Vdd Level

Set the Vdd stop voltage here. The Vdd voltage remains at this level until the end of the time set in Extend Acquisition By. Stop Value can be lower or higher than Start Value.

Ramp Time

Set the duration of the linear sweep here.

Analyzer

Extend Acquisition By

Extend the acquisition by time set in this field. The enables viewing the stability of the DUT output for a period after the Vdd operating voltage has reached its maximum (Stop Vdd Level). The measurement ends at after the time set here.

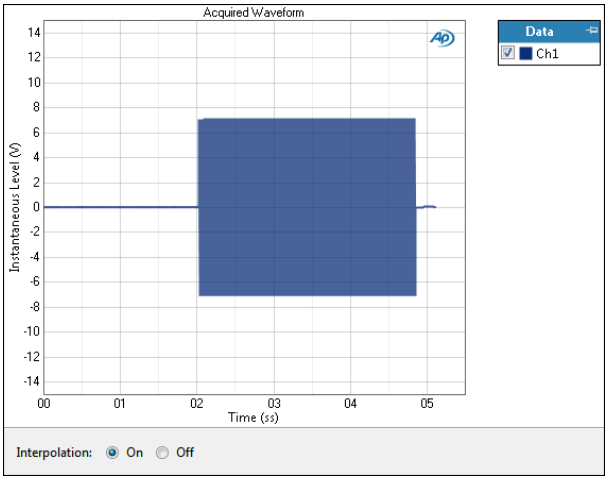
Result

There is one result for Vdd Ramp, the time-domain acquisition from the input selected in Signal Path Setup.

Advanced Settings

If your test requires special analyzer ranging, see Advanced Settings, following. See Chapter 104 for more information about units of measurement.

Acquired Waveform Result



Units

Units available for Vdd Ramp are

X-axis	*Y-axis (analog)*	*Y-axis (digital)*
• s	• V	• D • hex

Advanced Settings for Vdd Ramp

Signal Acquisition and Analysis

Input Range tab

Digital inputs do not require ranging, and the Input Range tab is unavailable when inputs are set to a digital format.

Auto Range

If **Auto Range** is checked (the default), each analog channel determines its input range automatically, based on the level of the input signal. If the input signal level changes beyond a ranging threshold, **Auto Range** will cause the input ranging circuits to move up or down for proper ranging.

Fixed Range

Alternatively, you can set a fixed minimum range for the analog inputs. Uncheck **Auto Range** and enter the expected maximum level in the **Minimum Range** filed. The software will choose the lowest range that will accommodate that level; this becomes the minimum range available, and the analyzer will not reset to a lower range. Fixed range is best for signals that vary rapidly, including tone bursts, perceptual audio speech samples, and program material such as voice or music. See **When autoranging isn't suitable** on page 626.

If a signal greater then the **Minimum Range** is presented to the analog input, the ranging circuits will range up to accommodate the higher level, and then

drop again when the signal level falls. The range will not be set below the **Minimum Range**.

Ranges Track Ch1

If **Ranges Track Ch1** is checked (the default), the fixed input range setting for channel 1 is copied to the other channels, and range settings for channels greater than 1 cannot be edited. Any changes made to channel 1 are reproduced in the other channels. If **Ranges Track Ch1** is not checked, the fixed input range setting for each channel can be set independently.

Read more about ranging in Chapter 98.

Automation

Introduction

Using the built-in **Sequencer** in Sequence Mode, APx500 measurements can be run in a programmed sequence to automate repeated tasks, to ensure consistent methods in R&D or to evaluate a series of devices against pass / fail limits in production. Sequences are saved with the project.

You can also control APx500 using the application programming interface (API). The APx500 API is discussed briefly beginning on page 567. Detailed API documentation and API examples are provided on the Audio Precision Web site at ap.com.

The Sequencer

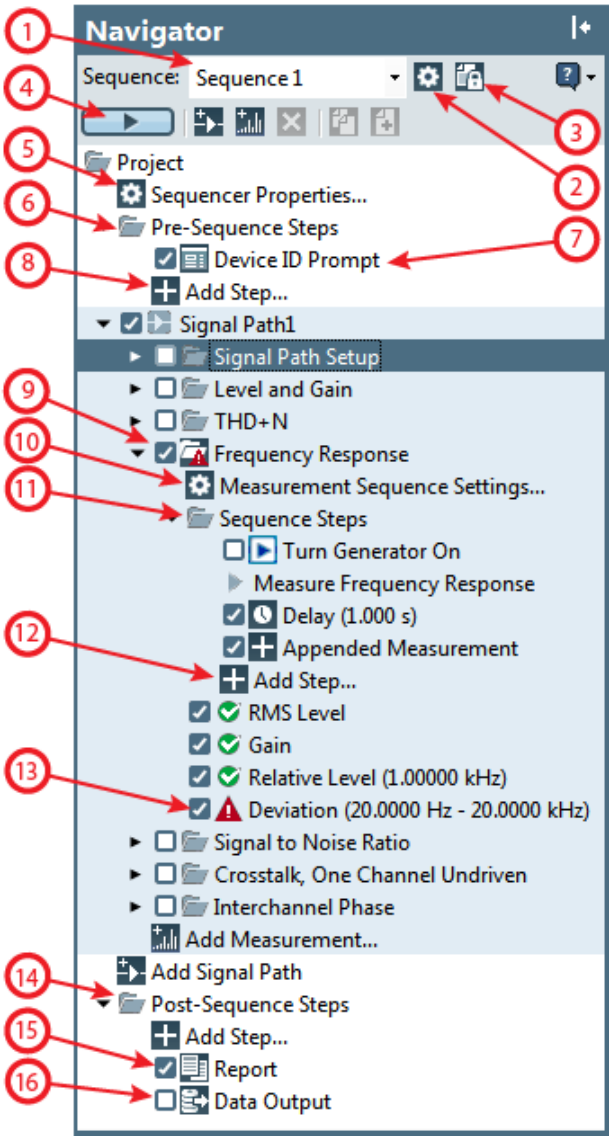
The **Sequencer** is a flexible and powerful automation interface that allows you to choose any combination of signal paths and measurements to be included in one or more automated sequences.

The **Sequencer** is controlled from the **Navigator** panel, which uses a tree structure interface to add and organize signal paths, measurements and measurement results, and to select some of these measurements and results to be included in a sequence. You can read more about the Navigator beginning on page 15.

In addition to its general-use automation capabilities, the Sequencer also provides a Production Test mode (see page 564) that allows you to create an entire suite of manufacturing-floor production tests, complete with simplified user prompts, interactive operator control, password-controlled supervisor access, pass/fail limits, detailed reports, production run statistics and raw data output.

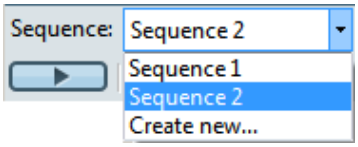
Sequencer features on the Navigator

The call-outs in this illustration are explained on the next page.



The Navigator panel, displaying sequence functionality. Callouts detailed on the following page.

1. The **Sequence** menu.
   By default, there is one sequence (Sequence 1) in a new project. You can add new sequences and select the active sequence from this menu.



2. **Add/Edit Sequences**
   Click **Add/Edit Sequences** to open the **Edit Sequences** dialog (page 564), where you can create and define a new sequence and set sequence properties (including Production Test mode).
3. **Lock Project**
   Production Test supervisors use the **Lock Project** feature (page 32) to switch to the restricted-access Production Test menu mode.
4. **Run Sequence**
   Click the **Run Sequence** button to run the current sequence.
5. **Sequencer Properties**
   Click **Sequencer Properties** (page 536) to set global, project-wide properties for the Sequencer.
6. **Pre-Sequence Steps**
   You can attach **Pre-Sequence Steps** to any sequence. These will run before the selected measurements.
7. The default Pre-Sequence Step is **Device ID Prompt**, which has been checked and will be included in th**e sequence.**
8. Use **Add Step** to attach additional Pre-Sequence steps such as Prompt, Delay, Set Aux Control, etc.
9. Notice that only the **Frequency Response** measurement has been selected (using the checkbox) to be included in Sequence 1.
10. **Measurement Sequence Settings**
    The Signal Path Setup branch and each measurement branch in the Navigator have a **Measurement Sequence Settings** node (page 538). Here you can set measurement-specific sequence behaviors, such as what the failure action should be.
11. **Sequence Steps**
    Signal Path Setup and each measurement also have a **Sequence Steps** folder. When this is expanded you will see the default steps (typically **Turn Generator On** and **Measure**). You will also see any steps that have been added.
    If you have more than one sequence step attached to this measurement, the right-click context menu available on the **Sequence Steps** node has a **Copy Multiple Steps...** selection.
    This opens a dialog where you can specify which steps you would like to copy.
12. You can attach sequence steps to any measurement in a Signal Path. Default steps include **Run Measurement** and **Turn Generator On**, and you can attach additional steps such as **Prompt**, **Delay**, **Set Aux Control**, etc., from the **Add Step...** menu. See Managing Sequence Steps on page 540.
13. Notice that the **Deviation** result is marked with a **Failed** flag because of a limit failure. The other results of this measurement have **Passed**.
14. **Post-Sequence Steps**
    Similar to the Pre-Sequence Steps discussed above, you can attach **Post-Sequence Steps** that run after the selected measurements. The default steps are **Report** and **Data Output**, and you can attach additional steps such as **Prompt**, **Delay**, **Set Aux Control**, etc., from the **Add Step...** menu.
15. **Report**
    The **Report** is a default **Post-Sequence** step. The **Report** is the primary output document of results from APx500 sequences and individual measurements. Reports are typically generated at the end of any sequence, or when a single measurement is run from the Navigator. See chapter 87 for more information about reports.
16. **Data Output**
    Data Output is a default **Post-Sequence Step** that sends sequence meter results to a CSV file as tabular data. Read more about Data Output beginning on page 535.

Running a sequence

Select the measurements and results to be included by placing checks in the Navigator checkboxes, and click **Run Sequence** or from the menu toolbar go to **Project > Run Sequence.** A **Running Sequence** window will open to show the progress of the sequence. Click **Cancel** to quit the sequence.

> *At least one measurement must be selected to run a sequence.*

Settling in a sequence

For those measurements that can be settled, the results are settled for reporting as the sequence is run. When the sequence is completed, a report is generated. Go to Chapter 98 for detailed information about settling.

Running a sequence at a selected measurement

Instead of running an entire sequence, you can select a measurement in the sequence and run the sequence from that point.

Right-click the measurement you would like to select as a starting point, and select **Run Sequence From Here**; or, highlight the measurement, and from the Main menu select **Project > Run Sequence from Selected Measurement**.

Running only a selected measurement

You can select one measurement, and run it as if it were in a sequence. Results will be settled and a report will be created.

Right-click the measurement you would like to run, and select **Run Selected Measurement**; or, highlight the measurement and from the Main menu select **Project > Run Selected Measurement**.

Report Data

All the settings and results of a sequence, called the report data, are stored in temporary memory. If an entire sequence is run again, the memory is flushed and all the report data are replaced by data from the new sequence. If a partial sequence is run (see above), only the data for the selected measurements are replaced. If a new measurement is added to the sequence, its results will be added to the report.

You can clear the report data manually. Right-click on the Report node at the bottom of the Navigator tree, and select **Clear Report Data**.

Information shown in a report is organized in the same order as measurements and results in the current Navigator arrangement.

Data Output

The Data Output feature sends certain sequence results to a CSV file as tabular data. Each time the sequence is run, the target file is opened, and the new data is appended to the table.

This allows you to view and organize sequence results in a spreadsheet or database program.

Enable **Data Output** by setting a checkmark in the **Data Output** node at the bottom of the Navigator tree. You can specify the CSV file name and location by right-clicking on the **Data Output** node to open the **Data Output Properties** dialog.

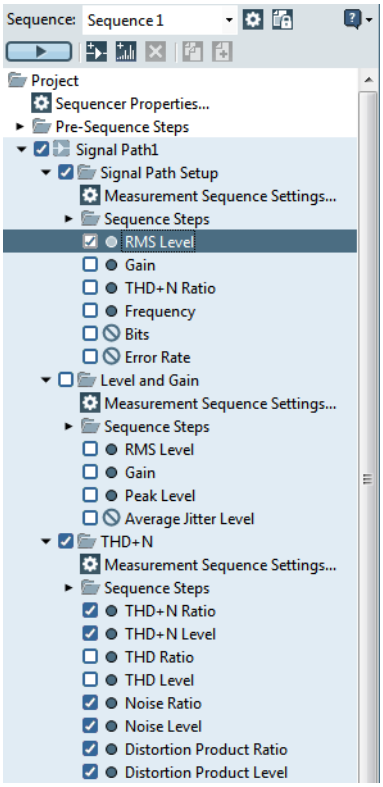
Data Output reports date, start time, device ID, limits, sequence pass or fail and meter values. XY graph values are not reported.

Opened in a spreadsheet, a typical series of sequences might follow this pattern.

	A	B	C	D	E	F	G	H	I
1	Data	Date	Start Time	Device ID	Sequence Result	M1 Name	M2 Name	M3 Name	M4 Name
2	Upper Limit					x	x	x	x
3	Lower Limit					x	x	x	x
4	Sequence Run	11/14/2011	13:50:01		PASS	y	y	y	y
5	Sequence Run	11/14/2011	13:51:01		PASS	y	y	y	y
6	Sequence Run	11/14/2011	13:52:01		PASS	y	y	y	y
7	Sequence Run	11/14/2011	13:53:01		PASS	y	y	y	y
8	Sequence Run	11/14/2011	13:54:01		PASS	y	y	y	y

Selecting and deselecting measurements

In the Navigator panel, set a check in the checkbox box next to any measurement to add the measurement to a sequence. Click again to remove the check mark and remove the measurement from the sequence.



If no measurements or results are selected, selecting a higher branch on the tree selects all the measurements that are within it. If a measurement or result is already selected, selecting or deselecting a higher branch includes or excludes that branch from the sequence without changing the selection status of the measurements or results that are within it. This allows you to choose a group of measurements, and then to include or exclude that group from a sequence with just one click.

To deselect all measurements in a branch and its lower branches, right-click on the branch and choose **Uncheck All** from the context menu.

Sequencer Properties settings

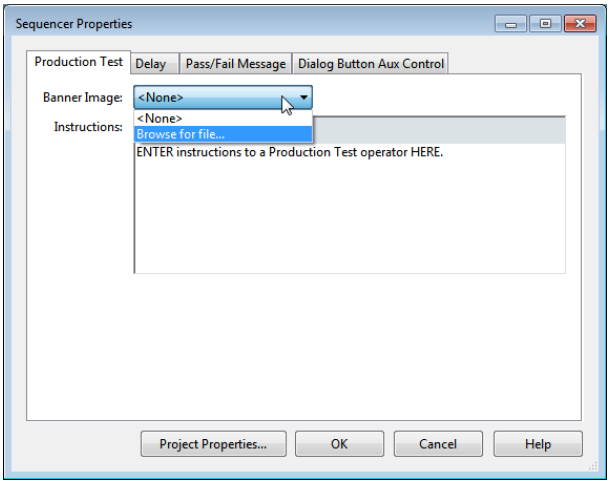
These properties are project-wide settings, and affect Sequencer behavior across all Signal Paths. Other project-wide settings are available in the Project Properties dialog (page 28).

The Sequencer Properties dialog is available by selecting the **Sequencer Properties** icon in the Navigator tree, by selecting **Sequencer Properties** from the Project menu, or by selecting **Sequencer Properties** from the right-click context menu at the Navigator **Project** node.

There are four tabbed pages in the Sequencer Properties dialog:

- **Production Test**
- **Delay**
- **Pass/Fail Message**
- **Dialog Button Aux Control**

Sequencer Properties: Production Test tab



When a sequence has been designated as a Production Test sequence (see page 564), an optional graphical banner and optional text can be shown to the operator at the beginning of the each Production Test sequence in the project. You can specify the banner and text here.

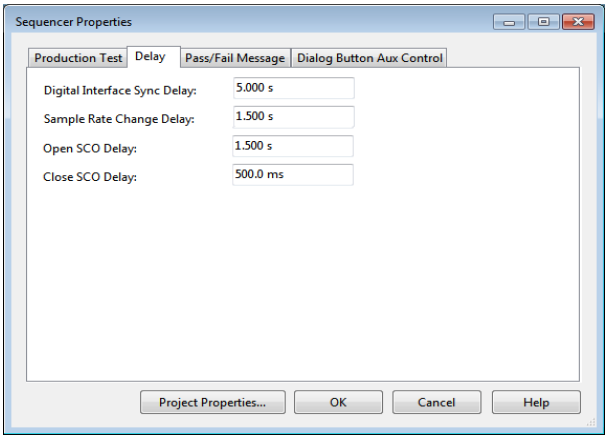
Banner Image:

Use the file browser to insert an optional image file here.

Instructions:

Enter optional instructions to the production test operator here.

Sequencer Properties: Delay tab



Digital devices may require some time to establish or re-establish synchronization. When a digital configuration is changed during a sequence, delay time must be added to the sequence execution to allow the devices to synchronize. Depending upon configuration, one or more of these delay settings will be available.

Digital Interface Sync Delay (digital input or output)

This delay is inserted into a sequence when a signal path that is configured with a digital input or output is activated. Default delay is 5 seconds.

Sample Rate Change Delay (digital output active)

This delay is inserted into a sequence when the output configuration is digital and a measurement with a different sample rate is activated. Default delay is 1.5 seconds.

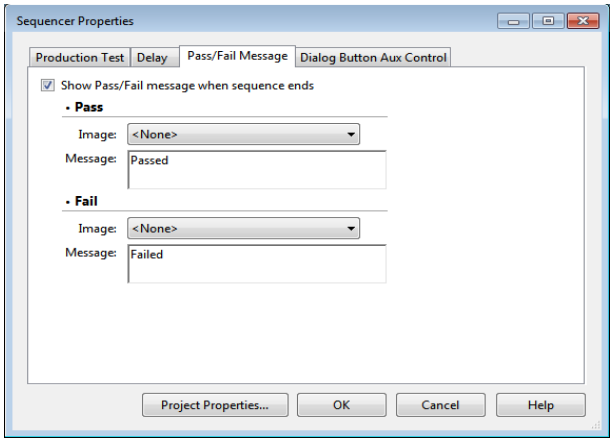
Open SCO Delay (Bluetooth Option only)

This delay is inserted after a Bluetooth **Open SCO** action is sent.

Close SCO Delay (Bluetooth Option only)

This delay is inserted after a Bluetooth **Close SCO** action is sent.

Sequencer Properties: Pass/Fail Message tab



An on-screen message can be shown to the operator at the end of a sequence.

To show a Pass/Fail message, click the **Show Pass/Fail Message** checkbox.

Pass Message

If all the measurements in the sequence pass, the **Pass** message is shown. The default message text is "Passed". Click in the **Message** box to edit the message. You can add an image using the **Image** file browser button.

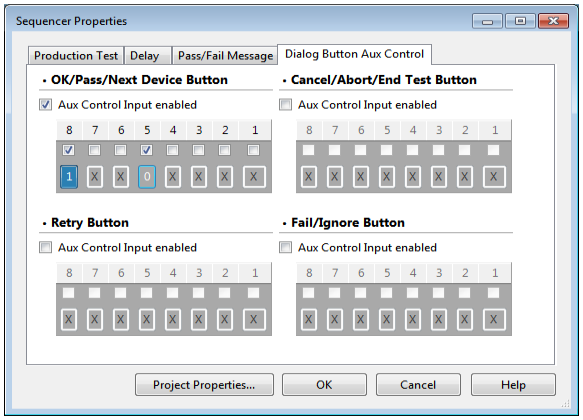
Fail Message

If one or more measurements in the sequence fails, the **Fail** message is shown. The default message text is "Failed". Click in the **Message** box to edit the message. You can add an image using the **Image** file browser button.

Closing the Pass/Fail Message dialog

The **Pass/Fail Message** dialog that is displayed to the user contains an **OK** button. The user clicks this button to close the dialog box. Alternatively, the user can close the dialog using an **Aux Control In** command. See **Dialog Button Aux Control,** next.

Sequencer Properties: Dialog Button Aux Control tab



Dialog buttons with four designations may appear in user prompts and dialogs generated during a sequence. These buttons require a response from the user, which can be accomplished by a mouse click or keyboard stroke on the automation PC, or by sending a specific signal to the Aux Control Input connector.

The four designations are:

- **OK/Pass/Next Device Button**
- **Retry Button**
- **Cancel/Abort /End Test Button**
- **Fail/Ignore Button**

To activate any of these on-screen buttons using Aux Control, check the **Aux Control Input Enabled** checkbox for the button you wish to activate.

Then, set the desired **Aux Control In** pattern in the control display. To do this, activate each bit you are interested in by setting a checkmark under the bit. Then set the desired state (0 or 1) for each activated bit. The state of bits that are not activated (unchecked, state shown as "X") will be ignored for this control.

In the illustration above, the **OK/Pass/Next Device Button** will be activated when **Aux Control In** bit 8 is high (1) and bit 5 is low (0). The states of bits 7, 6, 4, 3, 2, and 1 are ignored.

Sequence-specific Aux Control Run and End settings are available in the Aux Control Sequence Settings dialog, page 565. See Chapter 102 for more information about Aux Control.

Building a Sequence

A sequence is built in the Navigator panel. A sequence has at least one Signal Path selected.

For example, in a new project, click Run Sequence and you will get a Report of the Signal Path Setup configuration.

Select the Level and Gain measurement in the Navigator and click Run. This time you'll get a report that also has Level and Gain results. You will notice that Level and Gain is marked "Passed," which means it did not fail any limits. In this case, it's because no limits have been set.

You can add limits to your measurements, you can add more measurements, you can add more signal paths. You can also change the order of the measurements by dragging them up or down in the Navigator.

Each measurement node can be expanded to show a Measurement Sequence Settings icon, and a Sequence Steps folder. Measurement Sequence Settings determine the behavior if the measurement should fail limits, and are discussed in detail beginning on page 538.

The Sequence Steps folder can be expanded to show what optional steps are associated with this measurement. Many steps are available, adding delay, or run-

ning a program, or prompting the operator, for example. You can add steps, delete them, modify them and re-order them as you wish. The Sequence Step folder and the steps that can be added are discussed in detail beginning on page 539.

Sequences and Settling

For single value measurements, settling is applied only as a measurement or sequence is **Run**.

Stepped Sweep measurements are always settled, and Continuous Sweep and Signal Analyzer measurements are always unsettled.

See Chapter 98 for more information about settling.

Reports

The Report is the output document of a sequence. A report is generated when a sequence is run.

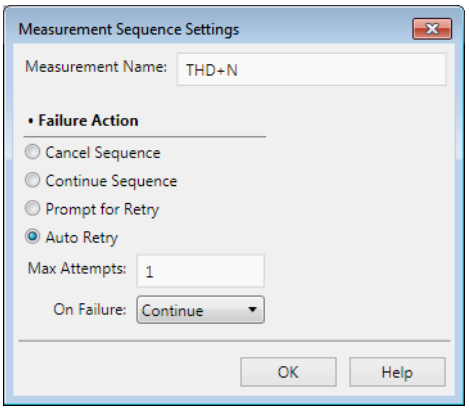
See Chapter 87 for more information about reports.

Measurement Sequence Settings

Each measurement has sequence properties associated with it that determine the behavior of the sequence step(s) if that measurement should fail.

Click the **Measurement Sequence Settings** node below each measurement in the Navigator to access the settings dialog, or by right-clicking on a measurement node and choosing **Measurement Sequence Settings** in the context menu.

The **Measurement Sequence Settings** dialog will open.



Measurement Sequence Settings, most measurements

Measurement Name

The parent measurement name from the Navigator is shown here. You can edit the name here, and your changes will be made on the Navigator branch as well.

Failure Action

When a limit (or **Auto Generator Set** action, see below) is exceeded in the selected measurement, you can direct the Sequencer to:

Cancel Sequence

Stop the sequence on a limit failure.

Continue Sequence (the default)

Flag the measurement as **Failed** and continue.

Prompt for Retry

Display a prompt indicating a limit failure, with **Cancel, Retry** and **Ignore** commands available to the operator.

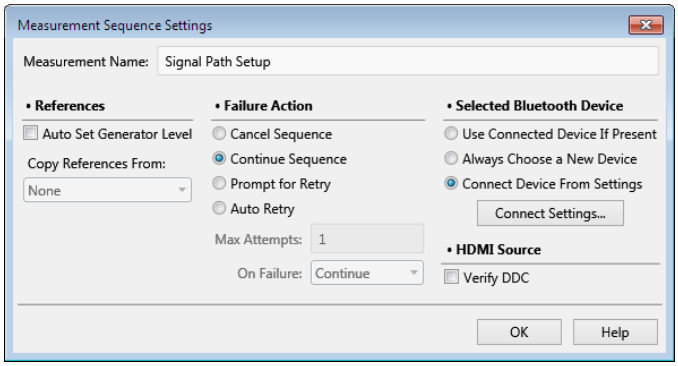
Auto Retry

Retry the step up to the number of times set in **Max Attempts**. If the step still fails after number of attempts set in **Max Attempts**, the Sequence will either **Continue** or **Cancel,** as selected in the **On Failure** menu.

Note: If the sequence is not configured properly (calling for a file or program step that does not exist, for example), the sequence will always abort.

Signal Path Setup: additional settings

If **Measurement Sequence Settings** has been accessed from the **Signal Path Setup** node, you will see more settings for **References**, and, depending up configuration, **HDMI Source** or **Selected Bluetooth Device.**



Measurement Sequence Settings, Signal Path Setup

Common Settings

Measurement Name and **Failure Action** are common to Measurement Sequence Settings for all measurements, and are discussed on the previous page.

References

Auto Set Generator Level

For **Signal Path Setup** only, Measurement Sequence Settings includes an **Auto Set Generator Level** checkbox. When checked, this will invoke the **Auto Gen Level** feature of **Signal Path Setup > References.** If **Auto Gen Level** fails, the selected **Failure Action** will be invoked.

Copy References From

For **Signal Path Setup** only. If there is more than one **Signal Path** in the sequence, you can copy the references from another **Signal Path** into the current **Signal Path**. This allows you, for example, to copy microphone calibration settings from one **Signal Path** to another.

Selected Bluetooth Device

See **Using Bluetooth in a Sequence** on beginning on page 557.

HDMI Source

Verify DDC

For **Signal Path Setup** in **HDMI Source Output Configuration** only, Measurement Sequence Settings includes an **HDMI Source: Verify DDC** checkbox. When checked, this will verify DDC (Display Data Channel) connection with the downstream device. If DDC verification is enabled and the DDC read fails, the sequence shall fail with either a **DDC error** or a **Receiver Not Connected** error. See More About DDC and EDID on page 120.

Sequence Steps

The Sequence Steps folder

The Signal Path Setup node and each measurement node in the Navigator have a **Sequence Steps** folder.

For most measurements, the **Sequence Steps** folder contains three default steps, a **Turn Generator On** step, a **Measure [measurement name]** step (such as Measure Frequency Response), and an **Add Step** button. These are all discussed next.

Any steps that are added to the measurement are also contained in this folder. The various steps in the folder can be rearranged to change the order of operation.

The Sequence Steps folder for Signal Path Setup

The Sequence Steps folder for a Signal Path Setup node contains an additional default step, **Measure and Set DUT Delay**. See page 541.

Default steps

Turn Generator On step

Most measurements have a **Turn Generator On** step that precedes the **Measure [measurement name]** step. It is not necessary to have this step checked to run the measurement; typically, a measurement will turn the generator **ON** and **OFF** as part of its normal operation.

However, there are situations that require the generator to be **ON** before the measurement begins. Here are two use cases:

Use case 1

The most common use case is to allow the operator to make adjustments to the DUT or the APx generator while observing a meter or other result, until an optimal setting for measurement is achieved.

Use case 2

Another case would be a measurement configured for **File Input** that requires a DUT (such as a digital recorder) to receive a generated signal and record it to a WAV file. In the APx interface, such measurements have a **Generator On** button, and an **Analyze** button to provide this functionality. When in a sequence, the **Turn Generator On** step accomplishes the former, and the **Measure [measurement name]** step accomplishes the latter.

Measure [measurement name] step

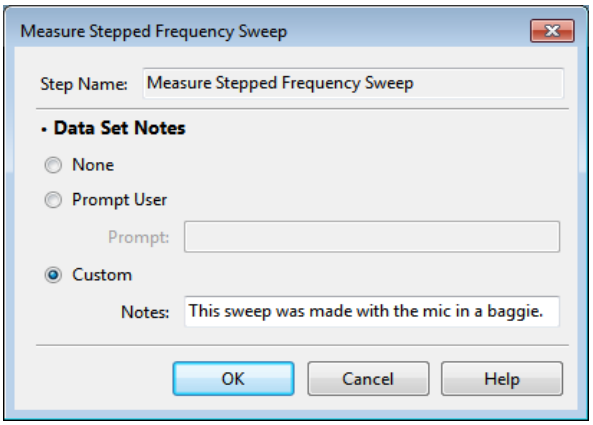
All measurement nodes have a **Measure [measurement name]** step that initializes the measurement.

If the measurement node is checked, the measurement will run. The usefulness of the **Measure [measurement name]** step is to determine in which order the various steps run. Move the **Measure [measurement name]** step up or down to change the order.

Adding Data Set Notes to a measurement

For measurements that support Append Graph Data, you can open a dialog to enter a prompt or note for each data set, if you are running multiple iterations of the measurement in the signal path.

For each iteration of a repeated measurement in the Signal Path, double-click on a **Measure [measurement name]** step (such as Measure Frequency Response), or right-click on the step and select **Properties** from the context menu. This dialog will open:



None

None, the default, will add no prompt or note to this iteration of the measurement.

Prompt User

You can enter text here that will prompt the operator to enter text in the **Notes** field while the Sequence is running. Perhaps the operator would enter an identifying number or scan a bar code in response to the prompt. Text entered by the operator will appear in the Report as a Data Sets Note for this iteration.

Custom

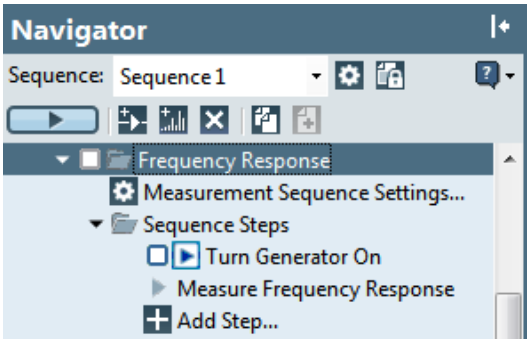
Text entered here will appear in the Report as a Data Sets Note for this iteration.

Sequence and Measurement Step Order

You can add any number of steps to display user prompts, require user input, run external programs, add delay, set aux control out or turn the generator on. Once added, these steps can be enabled, disabled or deleted. Each step's properties can be edited, and the steps can be moved into any order.

Each measurement has a Measurement Sequence Setting node, and a Sequence Steps folder. The default and any added steps are listed in the Sequence Steps folder.

Managing Sequence Steps



Sequence Steps

Expand the **Sequence Steps** folder below each measurement. You can view the sequence steps attached to the measurement, add and delete steps, change their order, review their settings, and enable or disable any step in the current sequence. You can attach any number of sequence steps to a measurement.

Copying Multiple Sequence Steps

If you have more than one sequence step attached to this measurement, the right-click context menu available on the **Sequence Steps** node has a **Copy Multiple Steps...** selection. This opens a dialog where you can specify which steps you would like to copy.

Default Step: Turn Generator On

A **Turn Generator On** step (page 539) is attached to every measurement where it is appropriate. This step is provided to turn the generator **ON** independent of the running of measurement, to allow the operator to make adjustments to the DUT or the APx generator. Place a checkmark in the box to enable the step.

It is not necessary to have this step checked to run the measurement or sequence; the measurement will turn the generator **ON** and **OFF** as part of its normal "Measure" operation.

Default Step: Measure [measurement name]

A Measure [measurement name] step is attached to every measurement where it is appropriate. This step invokes the parent measurement when the sequence is run.

Managing Steps

Click the **Add Step** button, then select a step type from the menu. Available steps will vary with hardware, measurement and configuration.

Once added, the order in which the steps run in the sequence can be changed by moving the steps (by dragging with the mouse) within the Sequence Steps folder.

The following steps are available. Each step is discussed in detail at the page reference listed.

- Measure and Set DUT Delay step (Signal Path Setup only, page 541).
- Device ID Prompt step (Pre-Sequence only, page 542)
- Prompt step (page 543)
- Input step (page 545)
- Program step (page 546)
- Delay step (page 546)
- Set Digital Sync Delay step (page 547)
- Appended Measurement step (page 547)
- Set Aux Control Out step (page 548)
- Export Result Data step (page 548)
- Import Result Data step (page 549)
- Import Limits Data step (page 549)
- Import Output EQ Curve step (page 551)
- PDM Control Codes step (page 551)
- Set Vdd Output step (page 552)
- Set Switcher Configuration step (page 552)
- Send CEC Command step (page 553)
- Set DCX Aux Output step (page 553)
- Set DCX DC Output (step page 554)
- Make Result Visible step (page 554)
- User Evaluation of Result step (page 554)
- Start Analog Generator step (page 556)
- Stop Analog Generator step (page 557)
- Clear Variable Value step (page 557)

Bluetooth steps

Additionally, there are a number of steps specific to using Bluetooth in a sequence:

- Bluetooth steps (beginning page 557)
-
-
-
-
-

Enabling or Disabling steps for the current sequence

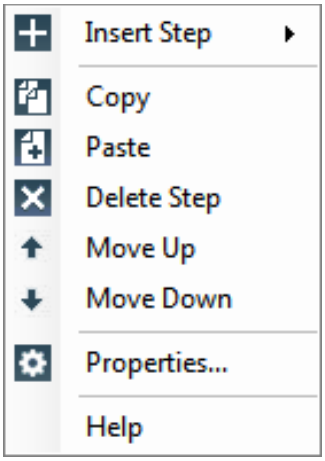
The checkbox at the left of each step row indicates whether or not that step is enabled or disabled in the sequence; steps are enabled ☑ (checked) by default. Uncheck the box ☐ to disable the step.

Selecting steps

Select a step by highlighting the step's row using the cursor. Select multiple steps by clicking while pressing the **Shift** or **Ctrl** keys.

Moving steps

Steps are processed in the sequence according to their position in the tree. You can move selected steps up or down in the list by selecting the step(s) and dragging with the cursor, or by right-clicking on the step and selecting **Move Up** or **Move Down**



Deleting steps

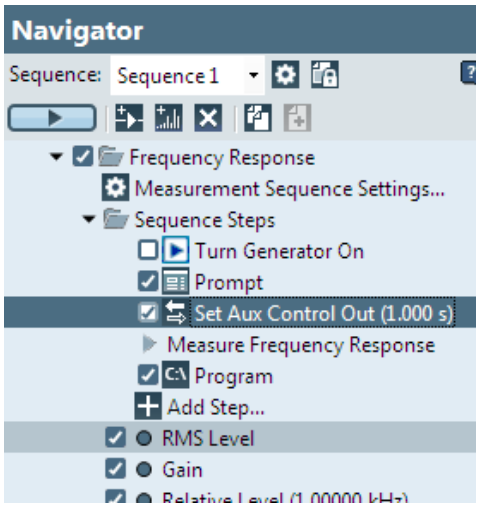
Delete a select step by right-clicking on the step and selecting **Delete Step**.

Editing step properties

To edit existing step properties, right-click on the step and choose **Properties...** in the context menu.

Copying and Pasting steps

Copy a selected step by right-clicking on the step and selecting **Copy**. Paste a copied step by right-clicking on the step and choosing **Paste**. Steps are pasted above the currently highlighted step.



In the example above, we added a **Prompt** step and a **Set Aux Control Out** step, and then moved them above the default **Measure Frequency Response** step. Then we added the final **Program** step at the end.

Measure and Set DUT Delay step

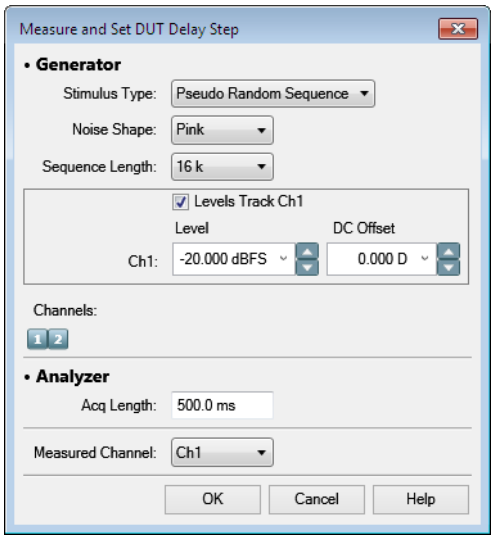
The **Measure and Set DUT Delay** step is only available as a Signal Path Setup sequence step. This step uses a PRS or MLS sequence stimulus with cross-correlation, in the same manner as the DUT Delay measurement, discussed in detail in Chapter 44.

> *This step is not available if Input or Output Connector is ASIO, Input Connector is File, or Output Connector is None.*

DUT Delay measures the length of time an audio signal is delayed through a device under test (DUT) or system under test, such as Bluetooth A2DP, or a speaker or headphone under test in an acoustic space. For chirp-based measurements such as Continuous Sweep, Acoustic Response or Loudspeaker Production Test, the delayed signal may fall out of the analysis window. Measure and Set DUT Delay provides a precise measurement of the delay through the device or system under test, and then enters the delay value into the Signal Path Setup DUT Delay field (see page 55).

This measurement determines loop delay (the delay between the generator output and the analyzer input) using either a pseudo random sequence (PRS) stimulus signal, as recommended in ITU-T P.501, or a maximum length sequence (MLS) signal. These are deterministic noise-like stimulus signals that can be cross-correlated to a fine granularity, providing a precise delay result.

Note: any delay previously entered in the SPS DUT Delay is set to zero while this measurement is running.



Generator

Stimulus Type

DUT Delay uses one of two noise-like binary sequences as a stimulus. Although there are many other uses for such stimuli, here they are used exclusively for cross-correlation between the sent and received signals to determine delay.

- **Pseudo Random Sequence**
  (PRS is recommended for this application in ITU-T P.501)
- **Maximum Length Sequence**
  (The MLS sequence cross-correlates better and can provide more accurate results. MLS also has a 0 dBFS crest factor, much higher than the PRS)

Noise Shape

Unfiltered, the noise-like PRS and MLS binary sequences have equal intensity at every frequency (a constant power spectral density; white noise).

These signals can be filtered to become pink noise, where the power spectral density is inversely proportional to the frequency. With pink filtering, each octave carries an equal amount of energy. Pink noise filtering is recommended in acoustic applications, where it offers a reduced risk of overloading high-frequency transducers.

Select the noise shaping here:

- **White**
- **Pink** (the default)

Sequence Length

The sequence length is the number of samples in the stimulus signal. The sequence length divided by the sample rate is the duration of the signal, and therefore longest delay that can be measured. Note that the menu selections use a common shorthand, representing sequence lengths that are actually powers of two, as shown below in parentheses.

- **16k** (16,383) 341 ms at 48 kHz SR
- **32k** (32,767) 683 ms at 48 kHz SR
- **64k** (65535) 1.37 s at 48 kHz SR
- **128k** (131,071) 2.73 s at 48 kHz SR

You must use a sequence length that is longer than the expected delay, at the chosen sample rate.

Levels Track Ch 1

If this box is checked, all channels are set to the same level as channel 1.

Level

Set the output level here.

DC Offset

Set an optional DC offset here.

Analyzer

Acq Length

Set the acquisition length here, in seconds. To capture an entire sequence, set the acquisition length to the sum of the duration of the sequence at the current sample rate, plus any expected delay in the system under test.

The acquisition starts at the moment the generator begins to output the sequence; in a DUT or system with significant delay (like an acoustic path) you may want to extend the acquisition to capture the propagation delay and the full sequence.

Diagnosing problems

If the system under test adds a great deal of noise or distortion, or greatly limits the bandwidth of the signal, the acquired signal will not cross-correlate well and the delay result will be spurious. See Diagnosing Problems in DUT Delay cross-correlation beginning on page 289.

Device ID Prompt step

To have the operator enter a device ID (typically, the device serial number) for a sequence, enable the **Device ID Prompt** step at the beginning of the sequence. The **Device ID Prompt** step is in the **Pre-Sequence Steps** folder. If the **Device ID Prompt** is enabled and **Automatically Generate File Name** is set in the **Report Properties Auto-Save** tab, the device ID will be used in the report filename.

Note: in this case, avoid the characters \ / : * ? " < > | in the **Device ID**. These are illegal characters in a Win-

dows file name, and APx will replace these with underscores "_" before saving.

Prompt step (General tab)

You can add a step that inserts an operator prompt, with optional operator response input accepted. Prompts allow you to provide the operator with information and the opportunity to interact with the sequence.

The **Input step** is similar, with fewer features. It does provide the opportunity to rename the Pass/Fail response buttons with any text.

The **Prompt** step has two tabbed pages: **General** and **Appearance**.

Inserting a Prompt step

...at the beginning of a sequence

Expand the **Pre-Sequence Steps** folder in the Navigator, click **Add Step**, and choose **Prompt...**

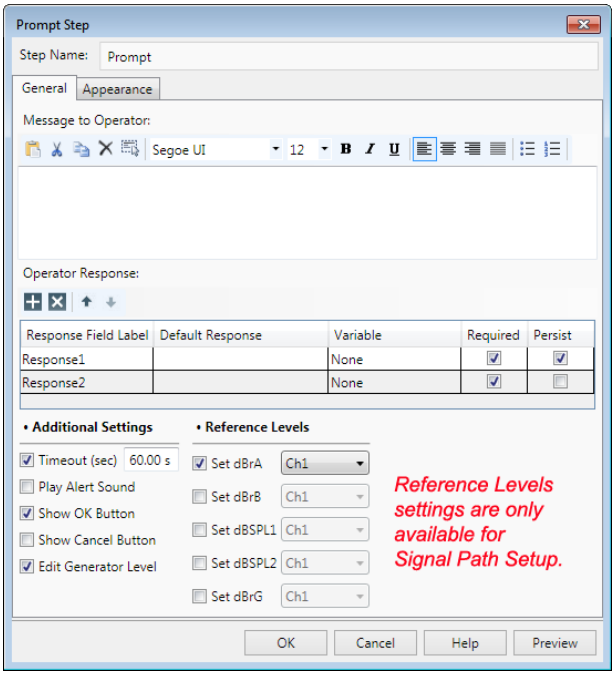
...within a sequence

Expand the **Sequence Steps** folder in a measurement node in the Navigator, click **Add Step**, and choose **Prompt...**

...at the end of a sequence

Expand the **Post-Sequence Steps** folder in the Navigator, click **Add Step**, and choose **Prompt...**

In any of these cases, the following dialog will open to the General tab:



Step Name:

You can name a Prompt step to help organize your sequence.

General tab: Message to Operator:

Add any text message that you want to appear in the prompt here. You can also paste formatted text from the Windows clipboard into the text entry field. The message window can be scrolled to allow viewing of long messages.

The toolbar at the top of the text entry area provides editing controls and font and paragraph settings.

General tab: Operator Response:

This feature optionally allows the operator to respond to the prompt by entering information such as operator ID, device ID or other information.

Click **Add Operator Response** to add a response field to the prompt.

Click **Delete Operator Response** to delete a response field from the prompt.

Click **Move Up** or **Move Down** to change the order of operator response fields on the prompt.

Enter text for the **Response Field Label** to be shown on the prompt.

Enter optional text as an editable **Default Response** to be shown on the prompt.

Select or create an optional **Variable** (page 31) to contain the operator response text.

If **Required** is checked, the prompt will not close without an operator response, unless canceled.

If **Persist** is checked and a variable has been selected to contain the response, the response entered by the operator will persist throughout the sequence.

General tab: Additional Settings

Timeout

When the **Timeout** box is checked, the value entered in the field to the right sets the maximum prompt window display time, in seconds. Minimum timeout is 1 second, maximum is 3600 seconds (one hour).

If **Timeout** is checked, but neither **Show OK Button** nor **Show Cancel Button** are checked, the measurement proceeds. This is useful to display an informational prompt that requires no user action.

If **Timeout** is checked and either or both **Show OK Button** and **Show Cancel Button** are checked, when the **Timeout** time is reached the subsequent steps in the current measurement are skipped, and the measurement fails. The sequence continues to the next measurement.

When the **Timeout** box is unchecked, the prompt window will remain open until the user closes it by the

OK button or the **Cancel** button, or the equivalent **Dialog Button Aux Control** (page 537) command.

Play Alert Sound

If the **Play Alert Sound** is ☑ checked, a Windows alert sound will be played through the PC when the prompt is displayed. The specific sound chosen depends on the prompt icon chosen on the Appearance tab, and your Windows settings. As with any sound played by Microsoft Windows, you must have a sound card installed in your PC and have the volume control and other Windows settings properly adjusted to hear the sound.

Show OK Button

Check ☑ this box to place an **OK** button OK on the prompt. Clicking the **OK** button closes the prompt window and continues the sequence, if any required operator response is entered. Alternatively, the operator or an external device can continue the sequence with an Aux Control command (if an operator response is required, it must also be entered for the sequence to continue). See Dialog Button Aux Control on page 537.

Show Cancel button

Check ☑ this box to add a **Cancel** button Cancel to the prompt. Clicking the **Cancel** button closes the prompt window and ends the sequence. Alternatively, the operator or an external device can cancel the sequence with an Aux Control command. See Dialog Button Aux Control on page 537.

Edit Generator Level

Not all measurements require or enable this prompt setting.

Certain measurements may require operator adjustment of the DUT or APx generator during the sequence. The **Turn Generator On** step (page 539) must be previously set to turn the generator on. **Edit Generator Level** opens a prompt during the sequence that displays generator level controls to enable the operator to observe DUT output while making APx generator adjustments. When ready, the operator clicks **OK** to continue the sequence. See the Preview below for an example.

Edit Generator Level in Acoustic output mode

When the Signal Path is set for output Acoustic mode, the prompt shows the Edit Generator Level settings in acoustic units (Pa or dBSPL).

General tab: Reference Levels

These controls appear only for the **Signal Path Setup** node.

The purpose of these controls is to allow the operator to play an audio signal or make an adjustment in response to a prompt, and to save the resultant input level as a reference.

When prompted, the operator performs the required action to present a signal to the analyzer (such as playing a test signal from the DUT or using a mic calibrator), selects dBrA, dBrB, dBSPL1 or dBSPL2 and the input channel of interest, and clicks **OK** to save the reference(s).

If the operator has been prompted to edit the generator level (as discussed above), that level can be saved as dBrG.

Reference Levels in Acoustic input mode

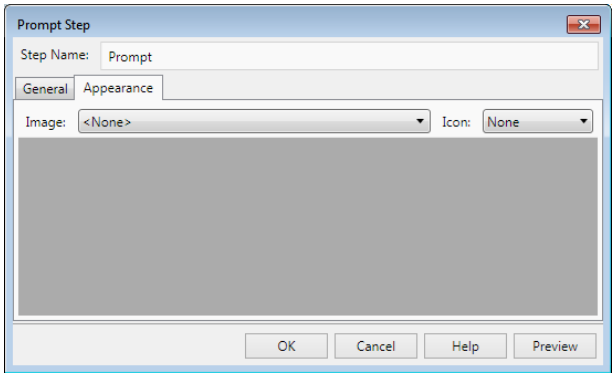
When the Signal Path is set to Acoustic input mode (page 191), the available input reference is dBSPL.

Preview

Click **Preview** to view the prompt as currently designed.

Prompt (Appearance tab)

Once you've navigated to the **Prompt** step dialog (page 543) choose the **Appearance** tab to add an image or icons to the prompt. The following dialog will open.



Appearance tab: Image

You can place an image at the top of the user prompt. This could be a company logo or a photograph or diagram with additional visual information to help the user.

Click the **Image...** button to navigate to an image file. Select the image and click **Open** to place the image in the prompt. When an image is used, the image file is also attached to the project file.

Select **<None>** to remove the image from the prompt. Image files attached to the project but not currently in use can be removed by navigating to the dialog at **File > Manage Attached Project Items** (see page 27).

Supported image file formats

- .bmp
- .jpg
- .jpeg
- .gif
- .tif

- .tiff
- .png
- .ico

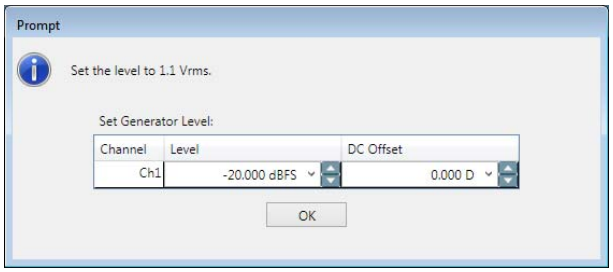
Appearance tab: Icon

Check one of the following (or None) to place a standard Windows attention icon on the prompt.

- None
-  Error
-  Question
-  Warning
-  Information

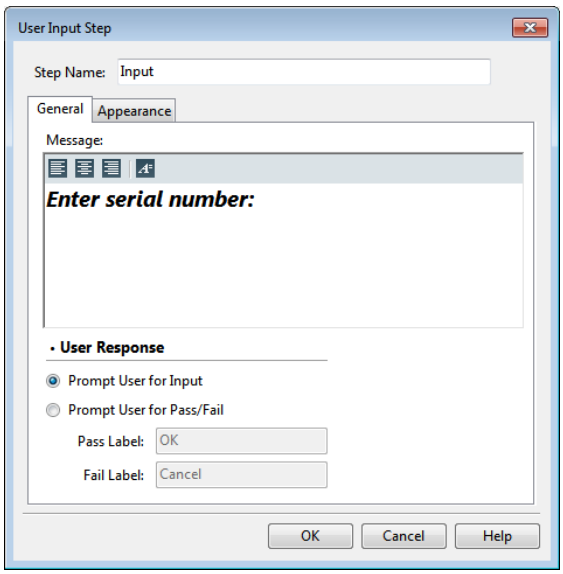
Preview

Press the **Preview** button to show how the finished prompt will look when displayed in the sequence.



Input (General tab)

You can add a step to any measurement in a sequence that pauses the sequence, and prompts the operator for a response.



This response can be in the form of entering a text string, or by clicking an on-screen button notifying the operator that the measurement has passed or failed limits.

> *The Prompt step offers similar operator input features along with other useful features.*

Step Name:

You can name an Input step to help organize your sequence.

Message:

Add any text message to the operator that you want to appear in the prompt. You can also paste formatted text from the Windows clipboard into the text entry field. The message window can be scrolled to allow viewing of long messages.

The toolbar at the top of the text entry area provides editing controls and font and paragraph settings.

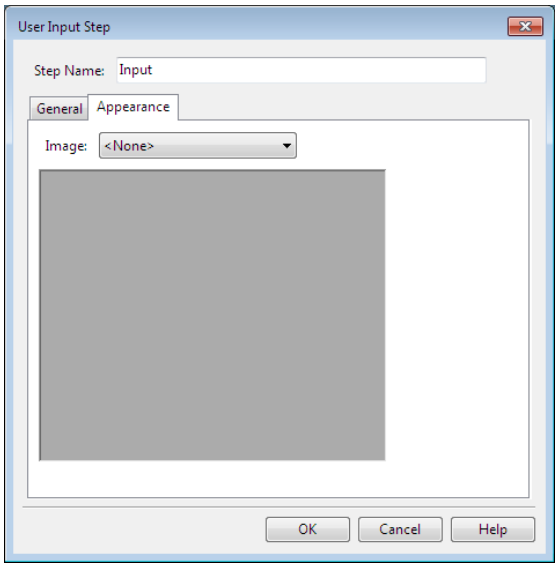
User Response:

- **Prompt User for Input**
  If Prompt User for Input is selected, the operator will see the Message and a text entry field for response.
- **Prompt User for Pass/Fail**
  If Prompt User for Pass/Fail, the operator will see either the pass or fail button, labeled with the text you have chosen below. The Pass/Fail buttons can be labeled with any text, such as OK / CANCEL, or MOVE CONNECTOR / RETRY. The operator responds by clicking the button.

Input (Appearance tab)

You can place an image at the top of the Input prompt. This could be a company logo or a photograph or dia-

gram with additional visual information to help the user.



Click the **Image...** button to navigate to an image file. Select the image and click **Open** to place the image in the prompt. When an image is used, the image file is also attached to the project file.

Select **<None>** to remove the image from the prompt. Image files attached to the project but not currently in use can be removed by navigating to the dialog at **File > Manage Attached Project Items** (see page 27).

Supported image file formats

- .bmp
- .jpg
- .jpeg
- .gif
- .tif
- .tiff
- .png
- .ico

Program step

You can add a step to a sequence that allows you to run an external program at one or more chosen points in the sequence.

Inserting a Program step

...at the beginning of a sequence

Expand the **Pre-Sequence Steps** folder in the Navigator, click **Add Step**, and choose **Program...**

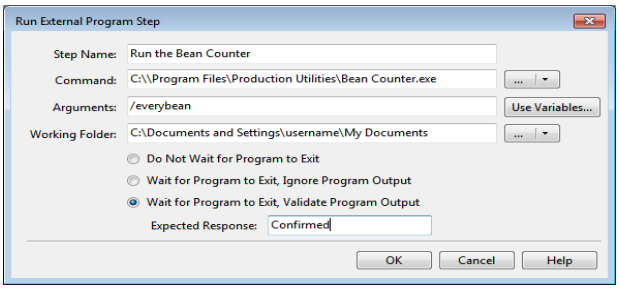
...within a sequence

Expand the **Sequence Steps** folder in a measurement node in the Navigator, click **Add Step**, and choose **Program...**

...at the end of a sequence

Expand the **Post-Sequence Steps** folder in the Navigator, click **Add Step**, and choose **Program...**

In any of these cases, the following dialog will open:



Step Name:

You can name a Run External Program step to help organize your sequence.

Command:

Type in the command for the external program here. You can use Variables (page 31), or browse to the program using the **Browse** ... button.

Arguments:

Type in the command arguments here, if any.

Use Variables...

Use Variables opens the **Edit Text and Use Variables** dialog, which allows you to build a string of arguments using variables and text entry. Select and double-click a variable to enter it into the variable string; type or delete text in the string as required.

Working Folder:

Type in the working folder here, if required. You can use Variables, or browse to the working folder using the **Browse** ... button.

Do Not Wait for Program to Exit

If this option is selected, the sequence will issue the command and then proceed (or, end) without waiting.

Wait for Program to Exit, Ignore Program Output

If this option is selected (the default), the sequence will pause until the program is finished.

Wait for Program to Exit, Validate Program Output

If this option is selected, the sequence will pause until the program is finished and returns a text string that matches the string entered into the **Expected Response** field.

Delay step

You can add a step to any measurement in a sequence that pauses the sequence for a specified time.

Inserting a Delay step

...at the beginning of a sequence

Expand the **Pre-Sequence Steps** folder in the Navigator, click **Add Step**, and choose **Delay...**

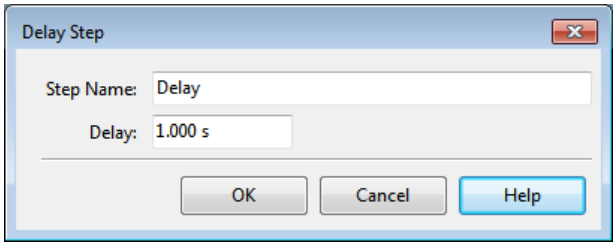
...within a sequence

Expand the **Sequence Steps** folder in a measurement node in the Navigator, click **Add Step**, and choose **Delay...**

...at the end of a sequence

Expand the **Post-Sequence Steps** folder in the Navigator, click **Add Step**, and choose **Delay...**

In any of these cases, the following dialog will open:



Step Name:

You can create a distinct name for this step to help organize your sequence.

Delay:

In this field, type in the delay time in s (seconds). Minimum is 0.000 s; maximum is 300.0 s.

Digital Sync Delay step

Digital devices may require some time to establish or re-establish synchronization. When a digital connection or sample rate is changed during a sequence, delay time must be added to the sequence execution to allow the devices to synchronize.

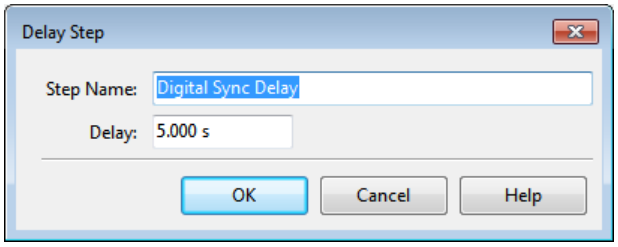
In most cases, the Delay tab features in Sequencer Properties (page 536) will accommodate these sync issues. But when the connection change or sample rate change happens external to APx (disconnecting and reconnecting digital cable, instructing a DUT to change sample rate), Sequencer Properties is not aware of these changes.

We recommend that when a sequence is expected to include external digital switching or sample rate changes that you add a Digital Sync Delay step to pause long enough for synchronization to occur. If synchronization lock occurs before the delay time has expired, the sequence will continue immediately.

Inserting a Digital Sync Delay step

Expand the **Sequence Steps** folder in a measurement node in the Navigator, click **Add Step**, and choose **Digital Sync Delay...**

The following dialog will open:



Step Name:

You can create a distinct name for this step to help organize your sequence.

Delay:

In this field, type in the delay time in s (seconds). Minimum is 0.000 s; maximum is 300.0 s.

Appended Measurement step

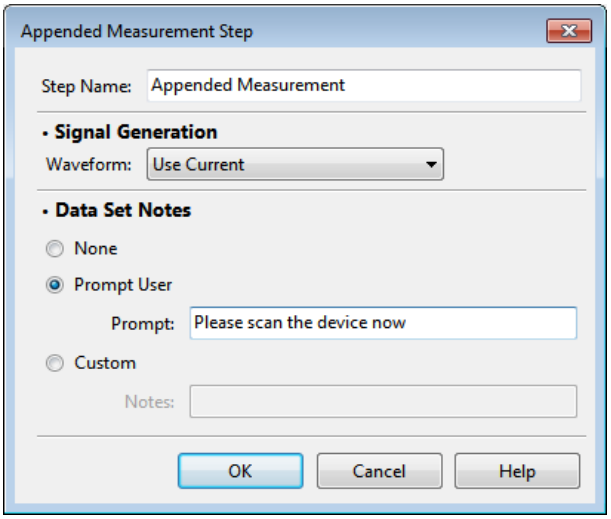
For measurements that support **Append**, you can one or more **Appended Measurement** steps to a measurement, each invoking an additional appended measurement. This allows you to make variations on a measurement within one sequence step, rather than sequencing multiple repeated measurements.

For measurements that support more than one generator waveform choice, you can select the waveform to use for each appended measurement.

Inserting an Appended Measurement step

Expand the **Sequence Steps** folder in a measurement node in the Navigator, click **Add Step**, and choose **Appended Measurement...**

The following dialog will open:



For example, in the Measurement Recorder, you may have **Split Frequency** selected as the waveform on the measurement panel, along with two generator waveform files called **Test1** and **Test2**. In the example here,

- **Use Current** would select the current waveform, **Split Frequency;**
- **Test1.wav** or **Test2.wav** would select one of the generator waveform files.

Data Sets Notes

You can cause text to be added to the **Data Set Notes** field. This text will appear in the report.

None

No **Data Set Note** will appear in the report for this measurement.

Prompt User

You can enter text here that will prompt the operator to cause text to be entered in the **Notes** field while the Sequence is running. Perhaps the operator would enter an identifying number or scan a bar code in response to the prompt here.

Custom

Text you enter here will appear as a **Data Set Note** in the report.

Set Aux Control Out step

You can add a step to any measurement in a sequence to issue an Aux Control Out command, and to pause for a specified delay time after the command is issued. Read about Aux Control in Chapter 102.

Inserting a Set Aux Control Out step

...at the beginning of a sequence

Expand the **Pre-Sequence Steps** (page 538) folder in the Navigator, click **Add Step**, and choose **Set Aux Control Out...**

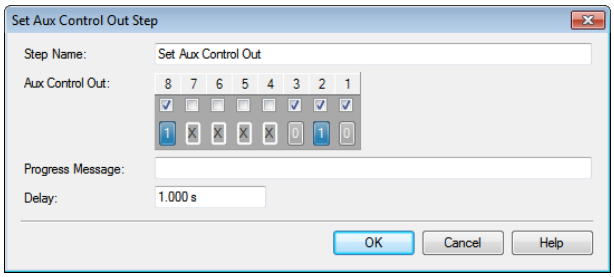
...within a sequence

Expand the **Sequence Steps** folder in a measurement node in the Navigator, click **Add Step**, and choose **Set Aux Control Out...**

...at the end of a sequence

Expand the **Post-Sequence Steps** folder in the Navigator, click **Add Step**, and choose **Set Aux Control Out...**

In any of these cases, the following dialog will open:



Step Name

Enter an optional name for this step.

Aux Control Out

Set the desired **Aux Control Out** pattern in the control display. To do this, activate each bit you are interested in by setting a checkmark under the bit. Then set the desired state (0 or 1) for each activated bit. The state of bits that are not activated (unchecked, state shown as "X") will be ignored for this control.

In this illustration, **Aux Control Out** will be set as follows: bits 8 and 2 are high (1) and bits 3 and 1 are low (0). The states of bits 7, 6, 5 and 4 are ignored; their previous states are maintained.

Progress Message

You can enter an optional text message in the **Progress Message** field. This message will be displayed to the operator for the length of time set in **Delay**, below.

Delay

An **Aux Control Out** command may start an external event that takes some time to complete. To wait for completion, you can enter an optional delay time here. The next step in the sequence is not executed until the delay time has passed.

The delay time begins at the instant that the Aux Control Out command is sent.

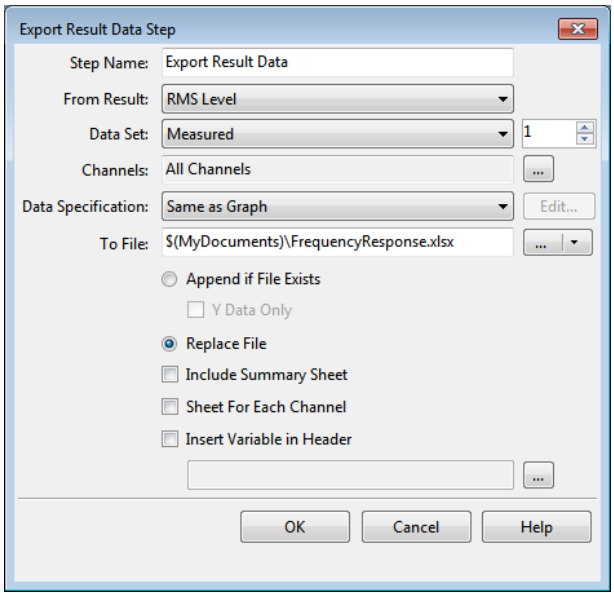
Export Result Data step

You can add a step to any measurement in a sequence to export APx result data from that measurement.

Inserting an Export Result Data step

Expand the **Sequence Steps** folder in a measurement node in the Navigator, click **Add Step**, and choose **Export Result Data...**

The following dialog will open:

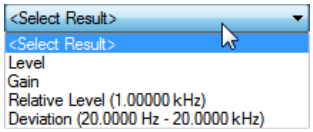


Step Name

Enter an optional name for this step.

From Result

Many measurements have more than one result. Choose the result from which you want to export data.



Data Set

For results that support data sets, you will be able to select the data set from this menu. Choose

- **Measured** or
- **Imported**

Select the number of the measured or imported data set from the numerical selector at right.

Meter results do not have these options.

Channels

Channels displays the measurement channels for which data will be exported. Click the ellipses ... button to open the **Choose Channels** dialog.

Data Specification

For results that have exportable data, this menu allows you to specify the range, distribution and number of data points to export. Choose

- **Same as Graph** (the default)
  The number of data points used to create the current graph; this is always fewer than **All Points**.
- **All Points**
  Every measured point in the result.

You can also specify a custom set of data points to be exported. Choose

- **Create New...**
  Create New opens the **Edit Data Specification** (page 651) dialog. You can define and save a Data Specification as an *.xyspec file.
- **Browse to File...**
  Browse to File allows you to find and import an existing *.xyspec Data Specification file.

To File

Enter the path and file name for the exported data file here. You can use Variables (page 31), or browse to the program using the **Browse** ... button. You can save in Excel spreadsheet formats, CSV or MATLAB formats.

Append if File Exists

This option opens the file on disk, appends the current data to the end of the data in the file, and re-saves the file.

Y Data only

For XY results, you can specify that only the Y-axis data be exported. This feature is only available when appending to .xls or .xlsx files, and then only when the data specification is not set to “Same as Graph.”

Replace File

This option erases the file on disk and replaces it with a new file containing the current data.

Include Summary Sheet

For Excel spreadsheets, you can optionally specify that a sheet be created to summarize the result data.

Sheet for Each Channel

For Excel spreadsheets, you can optionally specify that each channel be exported to numbered sheets within the file. This option made available when the target file extension in the **To File** field is *.xls or *.xlsx.

Insert Variable in Header

You can enter text, or enter a variable by clicking the ellipses button ... to open a **Edit Text and Use Variable** dialog (page 32). This information will appear in a header row preceding the exported data.

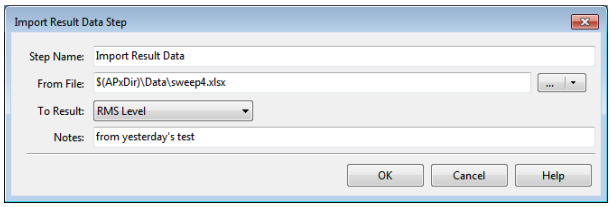
Import Result Data step

For measurements that support imported data sets, you can add a step in a sequence to import APx result data into that measurement.

Inserting an Import Result Data step

Expand the **Sequence Steps** folder in a measurement node in the Navigator, click **Add Step**, and choose **Import Result Data...**

The following dialog will open:



Step Name

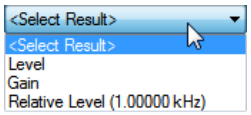
Enter an optional name for this step.

From File

Enter the path and file name here. You can use path variables (page 31), or browse to the file using the **Browse** ... button. You can open Excel spreadsheet formats, CSV or legacy Audio Precision data file formats.

To Result

Many measurements have more than one result. Choose the result into which you want to import data.



Be sure that the imported data and the selected result are compatible.

Notes

You can add a note to the imported data set.

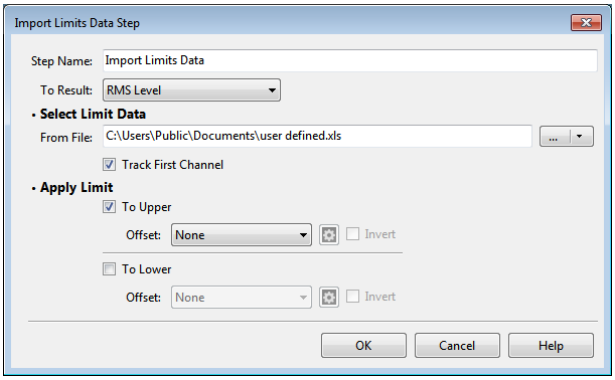
Import Limits Data step

For measurements that support imported limits, you can add a step in a sequence to import, apply and offset APx limit data for that measurement.

Inserting an Import Limits Data step

Expand the **Sequence Steps** folder in a measurement node in the Navigator, click **Add Step**, and choose **Import Limits Data...**

The following dialog will open:



Step Name

Enter an optional name for this step.

To Result

Many measurements have more than one result. Choose the result for which you want to import limits.

Select Limit Data

From File

Select **From File** to import limits from a data file. Enter the path and file name here. You can use path variables (page 31), or browse to the file using the **Browse** ... button. Supported formats are Excel spreadsheet formats, CSV, or legacy Audio Precision data file formats.

Be sure that the imported data and the selected result are compatible.

Track First Channel

Select **Track First Channel** to cause the limit for all channels to track the imported data in the first channel.

Apply Limit

To Upper / To Lower

Choose whether to apply the imported data to an upper limit, to a lower limit, or both.

Invert Data

If the limit is being applied to a relative result (such as Relative Level), an Invert Data checkbox will be available. Check the Invert Data checkbox to invert the limit data.

Offset:

When imported from a file, the data have arbitrary Y values, according to the values in the data file. When imported from the graph, the data have the same Y values as the current data in the selected data set. Depending upon data set choice, these values may be identical to the current result points and the limit curve may be drawn directly on the current result trace.

In either case, you will typically want to offset the limit curve above or below the result curve by scaling the limit Y values, using a constant offset value, or by using an offset curve.

You can select a file already attached to the project (if any), or

Create New...

The **Offset > Create New...** menu item opens the **Edit Limit Offset Curve** dialog (page 676). In this dialog, you can define an offset value that is constant across the X-range, or an offset curve whose Y-values vary across the X-range.

Click **OK** to save the offset curve in the project.

You can **Export** the curve as an Excel or CSV data file.

You can also **Import** an existing offset curve from an Excel or CSV data file.

Browse for file...

The **Offset > Browse for File** menu item opens a file browser.

Properties

For an existing offset curve, the **Properties** button opens the **Edit Limit Offset Curve** dialog (page 676).

Invert Offset Curve

Check the **Invert** checkbox to invert the limit offset curve.

Import Output EQ Curve step

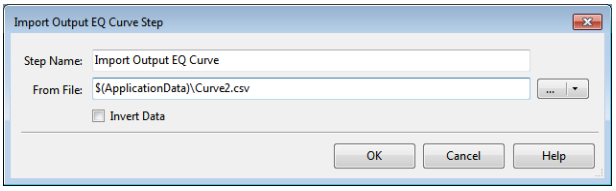
Also see Output Equalization on page 193.

You can add a sequence step to **Signal Path Setup** to import an **Output EQ** curve. The current **Output Configuration** must be **Analog**.

Inserting an Output EQ Curve step

Expand the **Sequence Steps** folder in a measurement node in the Navigator, click **Add Step**, and choose **Import Output EQ Curve...**

The following dialog will open:



Step Name:

Enter an optional name for this step.

From File:

Select **From File** to import limits from a data file. Enter the path and file name here. You can use path variables (page 31), or browse to the file using the ... **Browse** button. Supported formats are Excel spreadsheet formats, CSV, or legacy Audio Precision data file formats.

Invert

If this checkbox is set, the EQ curve will be inverted.

PDM Control Code(s) steps

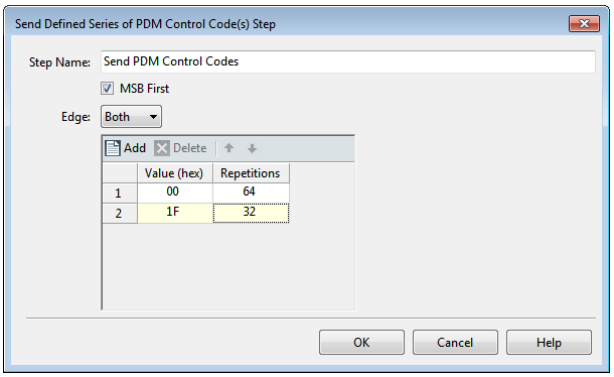
Inserting a Send Defined Series PDM Control Code step

This step is only available if Output Configuration for this Signal Path is set to PDM.

You can add a step to any measurement in a sequence that sends a defined series of PDM control codes.

Expand the **Sequence Steps** folder in a measurement node in the Navigator, click **Add Step**, and choose **PDM Control Code(s)...**

The following dialog will open:



Step Name

You can create a name for this step to help organize your sequence.

MSB First

Check **MSB First** to send the Control Code MSB (Most Significant Bit) first. When **MSB First** is not checked, the MSB is sent last.

Edge

Choose **Both**, **Rising** or **Falling** to select the PDM bitstream edge to carry the Control Code data.

Grid

Add or delete PDM control codes here, and define the **Value (hex)** and **Repetitions** for each code.

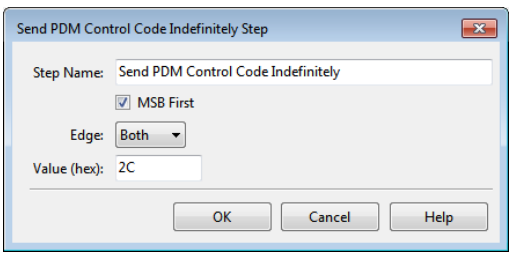
Inserting a Send PDM Control Code Indefinitely step

This step is only available if Output Configuration for this Signal Path is set to PDM.

You can add a sequence step that sends a PDM control code indefinitely. The PDM module will continue sending the specified code until a **Stop Sending PDM Control Code** step is encountered in the sequence, or until a signal path that does not use the PDM output is encountered. See the **Stop Sending PDM Control Code** step below.

Expand the **Sequence Steps** folder in a measurement node in the Navigator, click **Add Step**, and choose **Input...**

The following dialog will open:



Step Name

You can create a name for this step to help organize your sequence.

MSB First

Check **MSB First** to send the Control Code MSB (Most Significant Bit) first. When **MSB First** is not checked, the MSB is sent last.

Edge

Choose **Both**, **Rising** or **Falling** to select the PDM bit-stream edge to carry the Control Code data.

Value (hex)

Define the **Value (hex)** for the code.

Inserting a Stop PDM Control Code step

After inserting a **Send PDM Control Code Indefinitely** step, you can insert a **Stop Sending PDM Control Code** step.

Expand the **Sequence Steps** folder in a measurement node in the Navigator, click **Add Step**, and choose **PDM Control Codes... > Stop Sending Code...**

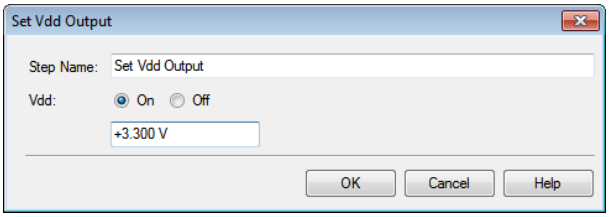
Set Vdd Output step

You can add a step to any measurement in a sequence that switches the Vdd dc power supply **ON** or **OFF**.

Inserting a Set Vdd Output step

Expand the **Sequence Steps** folder in a measurement node in the Navigator, click **Add Step**, and choose **Set Vdd Output...**

The following dialog will open:



Step Name

You can create a name for this step to help organize your sequence.

Vdd

The **Set Vdd Output** step will set the PDM Vdd voltage when executed as part of any signal path (including those containing a PDM connector) and will change the Vdd output accordingly.

If the active signal path contains a PDM connector, the PDM Vdd will be set to the value configured by the user for the PDM signal path when the sequence ends. If the active signal path does not contain a PDM connector, the PDM Vdd voltage will be returned the value set by user in the PDM Monitor.

This feature enables power to be supplied to a DUT such as a MEMS microphone or an integrated circuit, while generating and/or analyzing audio in a format other than PDM.

On/Off

Choose whether this step turns the Vdd output **ON** or **OFF**.

Enter the voltage required for operating power the DUT in entry field. The Vdd power is available on the PDM module at the **Vdd Supply** BNC connector, providing DC current up to 15 mA, with a voltage range of +0 VDC to +3.60 VDC.

Set Switcher Configuration step

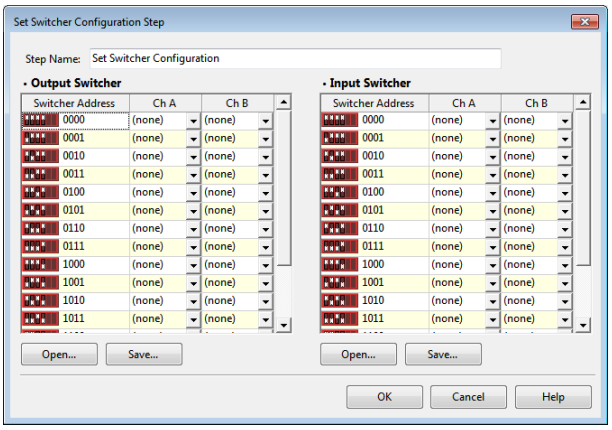
In a Sequence, the initial switcher settings for each Signal Path are set in **Signal Path Setup > Switcher Settings...** for that Signal Path.

To set or change switcher settings while a Sequence is in progress, add a **Set Switcher Configuration** step.

Insert a Set Switcher Configuration step

Expand the **Sequence Steps** folder in a measurement node in the Navigator, click **Add Step**, and choose **Set Switcher Configuration...**

The following dialog will open:



Step Name

You can create a name for this step to help organize your sequence.

Input and Output Switcher Settings dialog

These **Input** and **Output Switcher Settings** dialog presents the same settings as the **Signal Path Setup > Switcher Settings** dialog, discussed beginning on page 681.

Behavior

At the beginning of each active measurement in a sequence, the switcher configuration is set to the configuration in **Signal Path Setup** for the active Signal Path.

When a sequence is running, if the active measurement contains a **Set Switcher Configuration** step, the switcher configuration is set according to the step. The

new configuration persists until the next **Set Switcher Configuration** step, or until the next measurement begins, when the configuration reverts to the Signal Path Setup switcher configuration as described above.

At the end of the Signal Path in a Sequence, the switcher configuration is reset to Signal Path Setup switcher configuration.

If no switchers are connected, the sequence continues normally, with no error or warning.

Adding an Import Amplifier Correction Curve step

Read more about Amplifier Correction on page 325.

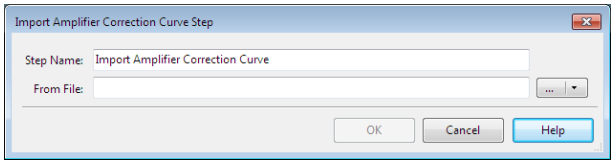
You can add a sequence step to an **Impedance/Thiele-Small** or **Loudspeaker Production Test** measurement to import an **Amplifier Correction Curve**. The current **Output Configuration** must be **Analog**, and the measurement **Test Configuration** must be **External (1 Ch)**.

> *Note: importing an Amplifier Correction Curve using a sequence step is not supported when the output connector is configured for Transducer Interface.*

Inserting an Amplifier Correction Curve step

Expand the **Sequence Steps** folder in an **Impedance/Thiele-Small** or **Loudspeaker Production Test** node in the Navigator, click **Add Step**, and choose **Import Amplifier Correction Curve...**

The following dialog will open.



Step Name:
Enter an optional name for this step.

From File:
Enter the path and file name here. You can use path variables (page 31), or browse to the file using the ... **Browse** button. Supported formats are Excel spreadsheet formats, CSV, or legacy Audio Precision data file formats.

Send CEC Command step

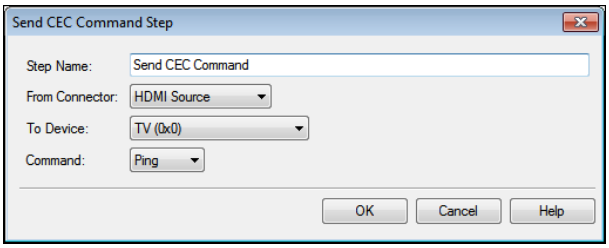
Read more about CEC on page 126.

You can add a sequence step to **Signal Path Setup** to send a CEC command from the HDMI Source or Sink connector. Either (or both) Output Configuration and Input Configuration must be currently set to HDMI to access this step.

Inserting a Send CEC Command Step

Expand the **Sequence Steps** folder in a measurement node in the Navigator, click **Add Step**, and choose **Send CEC Command...**

The following dialog will open:



Step Name:
Enter an optional name for this step.

From Connector:
Select the HDMI connector (**Source, Sync, ARC Tx** or **ARC Rx**) that will send the command and receive the acknowledgement.

To Device:
Select the device to be addressed. The menu lists the 16 defined CEC logical addresses.

Command:
Select the command to send, **Ping** or **Custom**. A **Ping** is a specific CEC polling message that any addressed device should acknowledge.

When **Custom** is selected, the **Opcode** and **Operands** fields are available to add payload information to the message.

Opcode:
Enter an arbitrary opcode (operational code) here.

Operands (hex):
Enter arbitrary operands here, in hex.

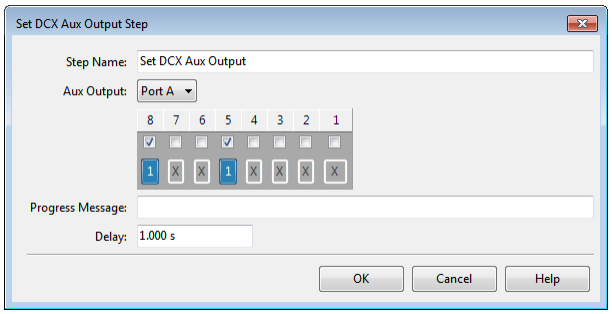
Set DCX Aux Output step

You can add a sequence step to a measurement to make an Aux Output setting on an attached DCX-127.

Inserting a Set DCX Aux Output step

Expand the **Sequence Steps** folder in a measurement node in the Navigator, click **Add Step**, and choose **Set DCX Aux Output...**

The following dialog will open:



Step Name:
You can create a distinct name for this step to help organize your sequence.

Aux Output:
Choose the **DCX Port** from the menu, and make **Aux Output** settings in the display. Use a checkmark to enable a line, and set the line high or low by choosing a **1** or a **0**.

Progress Message:
Enter an optional progress message here.

Delay:
Add any delay needed for the **Aux Output** command and connected device to attain the required state.

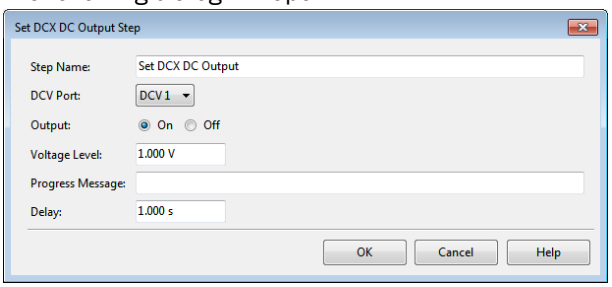
Set DCX DC Output step

You can add a sequence step to a measurement to set and adjust a DC Output on an attached DCX-127.

Inserting a Set DCX DC Output step

Expand the **Sequence Steps** folder in a measurement node in the Navigator, click **Add Step**, and choose **Set DCX DC Output...**

The following dialog will open:



Step Name:
You can create a distinct name for this step to help organize your sequence.

DCV Port:
Choose the DCX **DCV Port** from the menu.

Output:
Switch the DC voltage **On** or **Off**.

Voltage Level:
Set the DC **Voltage Level** here.

Progress Message:
Enter an optional progress message here.

Delay:
Add any delay needed for the DC **Voltage Output** and connected device to attain the required state.

Make Result Visible step

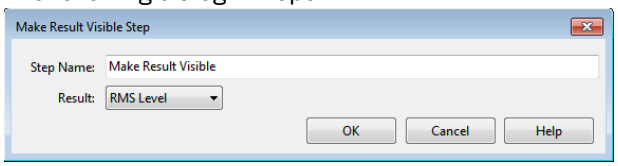
When you run a sequence, you may have noticed that for each measurement, the most recently selected result is displayed as the measurement runs.

You can change this behavior by adding the **Make Result Visible** step, which instead causes a specific chosen result to be displayed. This can be useful when a user must visually evaluate one or more results as a sequence runs.

It may be helpful to insert a **Delay** step (page 546) after the **Make Result Visible** step to allow more time for the user to make an evaluation.

Expand the **Sequence Steps** folder in a measurement node in the Navigator, click **Add Step**, and choose **Make Result Visible...**

The following dialog will open:



Step Name:
You can create a distinct name for this step to help organize your sequence.

Result:
Select the result to display from this menu.

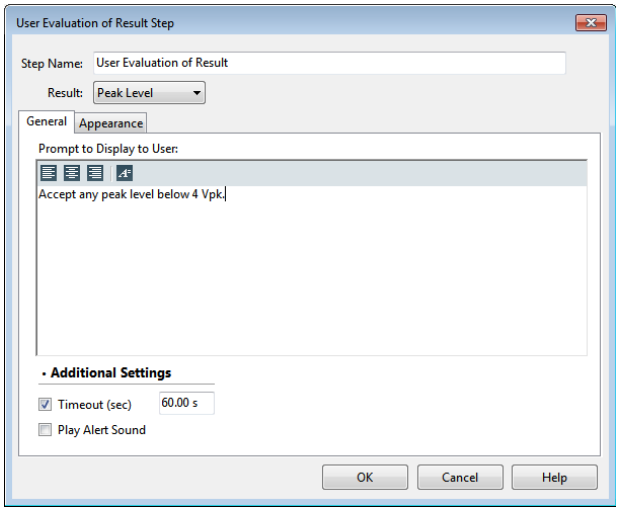
User Evaluation of Result step

Some sequences require that a user evaluate a result that is not easily evaluated by an analyzer. An example might be a stair-step pattern of automated level changes that a user can visually judge, but a machine cannot.

Inserting a User Evaluation of Result step

Expand the **Sequence Steps** folder in a measurement node in the Navigator, click **Add Step**, and choose **User Evaluation of Result...**

The following dialog will open:



Step Name:

Enter an optional name for this step.

Result:

Most measurements have more than one result. Select the result you would like to display to the user or operator from this list. “Peak Level” is shown in the example here.

Prompt to Display to User:

Enter an optional prompt for the user here. “Accept any peak level below 4 Vpk.” is shown in the example here.

Timeout

Set maximum time that the sequence will wait for a response before continuing.

Play Alert Sound

Play a sound to alert the user.

Appearance tab

There is an associated **Appearance** tab in this dialog, where you can specify an image and an icon to display.

Set Transducer Interface Input step

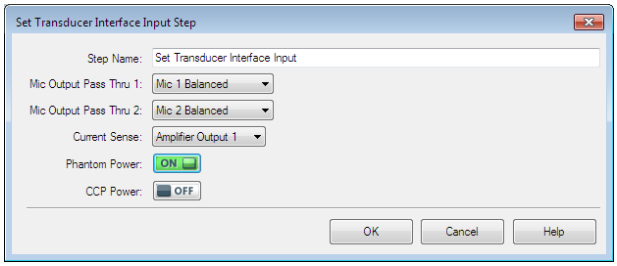
Read about the APx1701 Transducer Test Interface on page 12, and Settings for Transducer Interface Inputs on page 165.

You can add a sequence step to a measurement to make input settings for an attached APx1701.

Inserting a Set Transducer Interface Input step

Expand the **Sequence Steps** folder in a **Signal Path** node in the Navigator, click **Add Step**, and choose **Set Transducer Interface Input...**

The following dialog will open.



Step Name:

You can create a distinct name for this step to help organize your sequence.

Mic Output Pass Thru 1

This menu selects the APx1701 microphone input to appear at the Microphone Output (Pass Thru) connector 1. The choices are:

- **Mic 1 Balanced**
- **Mic 2 Balanced**
- **Mic 1 Unbalanced**
- **Mic 2 Unbalanced**
- **Mic 1 CCP Self-Test**

Mic Output Pass Thru 2

This menu selects the APx1701 microphone input to appear at the Microphone Output (Pass Thru) connector 2. The choices are:

- **Mic 2 Balanced**
- **Mic 2 Unbalanced**
- **Mic 2 CCP Self-Test**

See the *APx1701 Transducer Test Interface Installation, Specifications and User’s Guide* booklet for information about performing a CCP Self-Test.

Current Sense

This menu selects which amplifier channel current sense resistor is connected to the Current Sense connector for measurement. The choices are:

- **Amplifier Output 1**
- **Amplifier Output 2**

Phantom Power

Set this control to **ON** to apply +48 V phantom power to the two APx1701 balanced microphone inputs.

CCP Power

Set this control to **ON** to apply +24 V CCP (constant current power) to the two APx1701 unbalanced microphone inputs.

Set Transducer Interface Output step

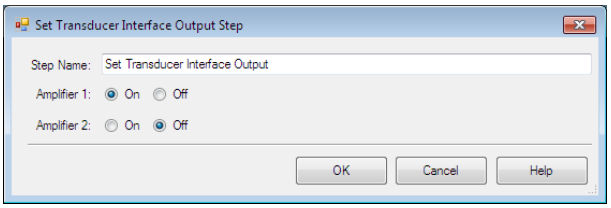
Read about the APx1701 Transducer Test Interface on page 12, and Settings for Transducer Interface Outputs on page 164.

You can add a sequence step to a measurement to make output settings for an attached APx1701.

Inserting a Set Transducer Interface Output step

Expand the **Sequence Steps** folder in a **Signal Path** node in the Navigator, click **Add Step**, and choose **Set Transducer Interface Output...**

The following dialog will open.



Step Name:

You can create a distinct name for this step to help organize your sequence.

Amplifier 1, Amplifier 2

You can switch the APx1701 amplifier channels 1 or 2 **OFF** or **ON** with this step.

> *Note: Configurations that are set by a sequence step remain as set by the last applicable sequence step executed in the sequence when the sequence ends. If the you want to restore the settings at the end of a sequence, then you must use a sequence step at the end of the sequence, or manually restore through the user interface after the sequence ends.*

Calibrate from TEDS step

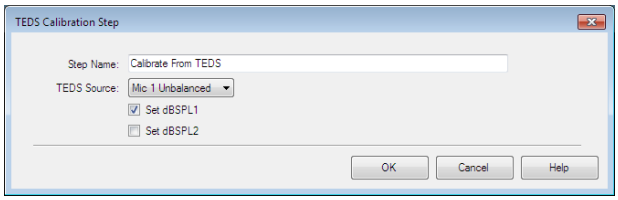
Also see APx1701 Transducer Test Interface on page 12, Microphone Calibration on page 60, Microphone Calibration (acoustic) on page 64, and More About TEDS on page 521.

You can add a sequence step to a measurement to calibrate a TEDS-enabled measurement microphone from its TEDS data, when using an attached APx1701.

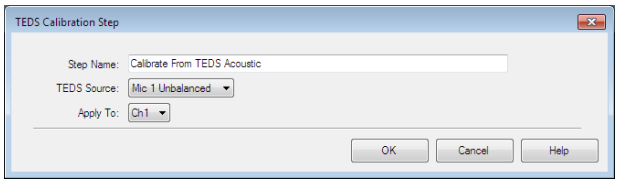
Inserting a Calibrate from TEDS step

Expand the **Sequence Steps** folder in a **Signal Path** node in the Navigator, click **Add Step**, and choose **Calibrate from TEDS...**

The following dialog will open.



Calibrate from TEDS sequence step in "normal" mode.



Calibrate from TEDS sequence step in acoustic mode.

Step Name:

You can create a distinct name for this step to help organize your sequence.

TEDS Source

Choose the APx1701 input connected to the microphone you would like to read TEDS data from. Choices are:

- **Mic 1 Unbalanced**
- **Mic 2 Unbalanced**

Signal Path Input Configuration in "normal" mode

Using the checkboxes, choose the APx reference you would like to set from the TEDS data. Choices are:

- **Set dBSPL1**
- **Set dBSPL2**

Signal Path Input Configuration in acoustic mode

Using the menu, choose the APx channel you would like to which you'd like to apply the TEDS data. You can choose any one of the available analog input channels.

Start Analog Sine Generator step

> *This step is only available for the APx555, when Output Configuration is set to a non-analog output connector.*

The APx555 enables the use of the analog generator as an additional signal source, when **Output Configuration** is set to a non-analog output connection.

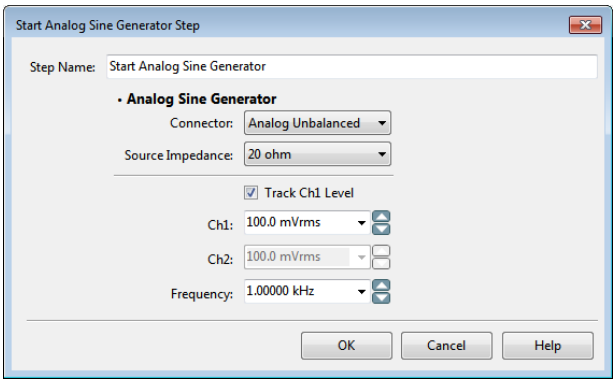
When configured this way, you can add a sequence step to turn on the analog generator (as distinct from the **Turn Generator On** step available in all sequences).

This step will start the analog sine generator, which, in this configuration, can be controlled independently from the DSP generator connected to a digital output.

Inserting a Start Analog Generator step

Expand the **Sequence Steps** folder in a measurement node in the Navigator, click **Add Step**, and choose **Start Analog Sine Generator...**

The following dialog will open:



Step Name:

You can create a distinct name for this step to help organize your sequence.

Analog Sine Generator

The controls in the Analog Sine Generator area set the configuration for the APx555 High Performance Sine Generator. These are the same controls and settings that are available in the Signal Path Setup Input/Output analog unbalanced and balanced panels (Chapter 11).

Stop Analog Generator step

This step is only available for the APx555, when Output Configuration is not set to an analog output connector.

The APx555 enables the use of the analog generator as an additional signal source, when Output configuration is set to a non-analog output connection.

When this generator has been turned on in a sequence stop, you can add this sequence step to turn off the analog generator.

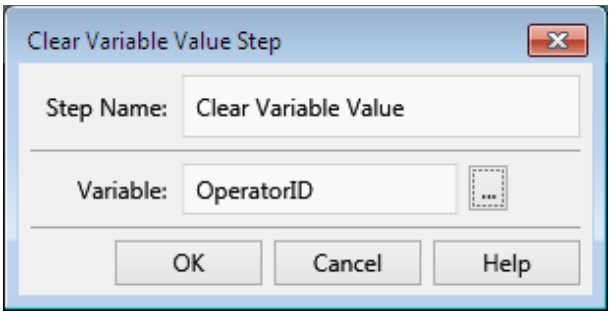
Inserting a Stop Analog Generator step

Expand the **Sequence Steps** folder in a measurement node in the Navigator, click **Add Step**, and choose **Stop Analog Sine Generator...**

No settings dialog is displayed; a **Stop Analog Generator** command is simply inserted into the sequence.

Clear Variable Value step

With this step, you can clear a current variable value during a sequence.



You can either enter the variable name here, or search the list of existing variables by clicking the ... **Browse** button. Read more about Variables on page 31.

Using Bluetooth in a Sequence

These steps are only available if Output Configuration for this Signal Path is set to Bluetooth.

Unlike other APx Output and Input Configurations, which are simply established by software selection and a cable connection, Bluetooth configurations require discovery, pairing, profile selection and so on, in an extended negotiation between the APx analyzer and the Bluetooth device under test.

Project-wide settings

Only one APx Bluetooth Profile Set can be specified in a project. Go to Signal Path Setup and choose Bluetooth for either your Output Configuration or your Input Configuration. Open Bluetooth Settings and choose the APx Profile Set appropriate for your testing in this project.

In the Project Menu, choose Project Properties and add any User Variables you would like.

Sequence-wide settings

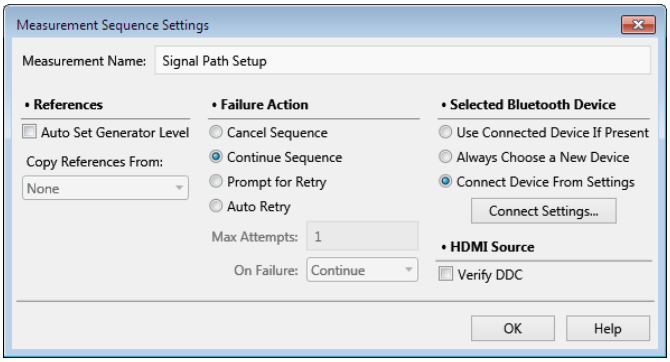
In the Project Menu, choose Sequencer Properties (or click Sequencer Properties in the Navigator and enter your preferences for Production Test, Delay, Pass/Fail Message and Dialog Button Aux Control settings. In particular, you may need to add delays may to allow the DUT sufficient time to execute SCO commands and sample rate changes. See page 536.

Signal Path settings

When automating Bluetooth testing, settings and steps that provide a means of specifying and negotiating the connection, connecting and disconnecting, and initiating actions must be included.

Although it is possible to connect to a Bluetooth device automatically in a sequence, in many cases steps requiring run-time interaction with the operator are necessary.

Connecting a device in a Sequence

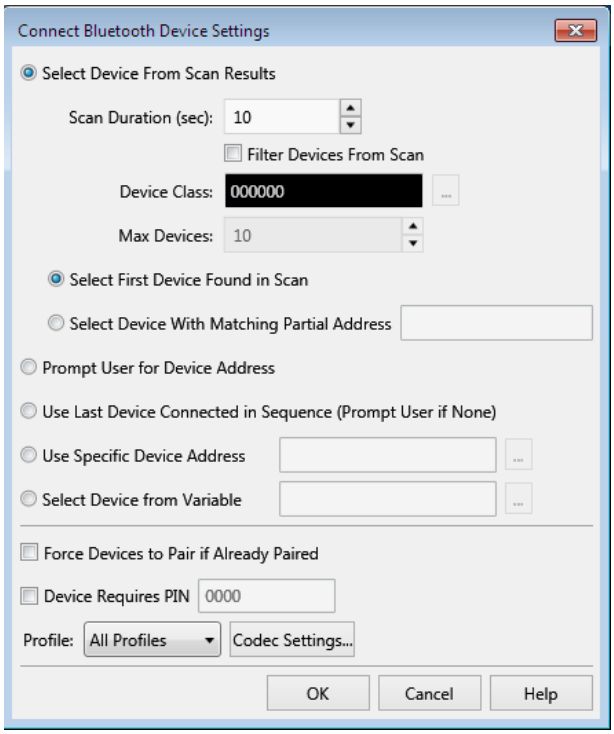


A Bluetooth device used in a sequence must be connected for testing. If there are multiple signal paths in the sequence, each signal path can be connected to a different device. However, global settings such as sink/source profile sets are in force across the entire project.

For each Signal Path, specify how to connect to a Bluetooth device in the Measurement Sequence Settings dialog. In the Selected Bluetooth Device area, you have three choices:

- **Use Connected Device if Present**
  This selection assumes the APx is connected to a device, either by connecting in the Signal Path Setup Bluetooth Settings dialog before running the sequence, or in a previous Signal Path in the sequence. If no device is connected, the operator will be prompted to select one at run time. See Connect Bluetooth Device run time prompt, page 559.
- **Always Choose a New Device**
  The operator is prompted to select a device at run time. See Connect Bluetooth Device run time prompt, page 559.
- **Connect Device from Settings**
  A device is selected automatically, using the criteria specified in the Connect Settings dialog. Click the Connect Settings button to open the settings dialog. See the next topic.

Connect Bluetooth Device settings



Connect Bluetooth Device Settings enables APx to automatically establish a Bluetooth connection during a sequence, using a number of criteria specified here.

Select Device From Scan Results

This choice will perform a Bluetooth scan at run time. If the specified criteria are met, a device will be selected, paired and connected. Many of the selection filters are not available with Legacy Bluetooth Option.

Scan Duration (sec):

This setting limits the scan time to the value set. The maximum time is 48 seconds. The default is 10 seconds.

- **Filter Devices From Scan** (not available with Legacy Bluetooth Option)
- **Device Class** (not available with Legacy Bluetooth Option)
- **Max Devices** (not available with Legacy Bluetooth Option)

Select First Device Found in Scan

This setting selects the first device found in the scan.

Select Device with Matching Partial Address

You can enter the first part (starting at the left) of a Bluetooth address here. The first device found that matches will be selected.

You can enter as few as the first two hex digits, or as much as the entire address. The address can be entered with or without separating colons (:).

Prompt User for Device Address

A prompt will appear at sequence run time. The operator must enter the device address.

Use Last Device Connected in Sequence (Prompt User if None)

Use the address of the most recent device connected earlier in the sequence. If no device has been connected, a prompt will appear. The operator must enter the device address.

Use Specific Device Address

Enter a specific device address here, or click the **Browse** ... button to select a device from the Bluetooth Known Devices list (page 142).

Select Device from Variable

Enter the name of an existing User Variable whose value is a valid device address, or click the **Browse** ... button to select a variable. See page 31.

Force Devices to Pair if Already Paired

If the selected device has already been paired with APx, force the device to pair.

Device Requires PIN

If the device requires a PIN, check this box and enter the PIN in the field to the right.

Profile

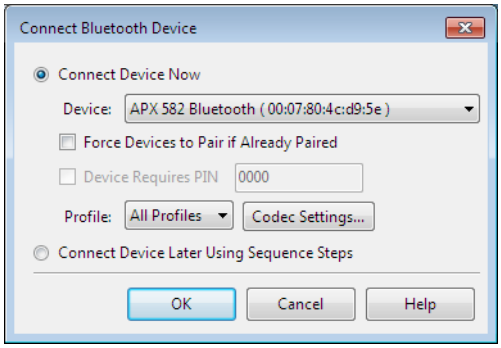
Connect to a select profile, or to select all profiles.

Codec Settings

This button opens the appropriate Codec Settings dialog for your configuration. See page 135. For legacy Bluetooth codecs, see page 152.

Note that later in the sequence, you can disconnect a device with a sequence step, or connect to a different device with a sequence step.

Connect Bluetooth Device run-time prompt



This is a sequence run time prompt. If, when a sequence is started, no Bluetooth device is connected, or Connect Bluetooth Device Settings asserts “Always Choose New Device”, the Connect Bluetooth Device prompt opens.

Connect Device Now

Select this choice to connect to the selected device immediately, when OK is clicked.

Device

Use this menu to select a Bluetooth device.

- **None** (if the Current Devices list is empty, this is the current condition).
- if the Current Devices list is populated, the discovered devices are shown. Highlight a device to select it.
- **Select New Device** opens the Scan for Bluetooth Devices dialog to repopulate the Current Devices list. Click Scan for Devices, then highlight a device and click Select Device.

Force Devices to Pair if Already Paired

If this device has been previously paired using this project, force it to pair now.

Device Requires PIN

If the device requires a PIN, enter it here.

Profile

Choose the profile to be used for the immediate connection.

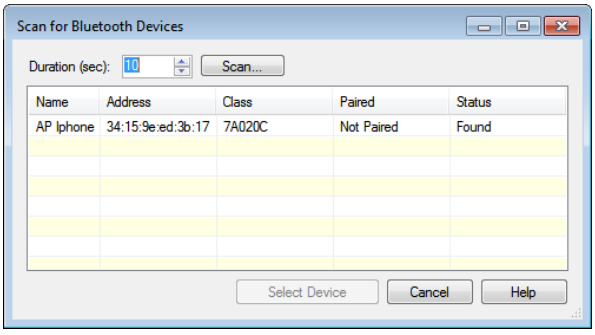
Codec Settings

Edit codec settings.

Connect Device Later Using Sequence Steps

Select this choice to connect later in the sequence. Be sure to insert a Connect Bluetooth Device (page 560) step in a subsequent measurement, and configure the step for the Bluetooth device.

Scan for Bluetooth Devices run-time prompt



This is a sequence run time dialog, called by the **Select New Device** choice from the Device menu on the Connect Bluetooth Device prompt.

Scan for Devices

This button initiates a scan to identify all the discoverable Bluetooth devices within range. Discovered devices are shown and added to the Current Devices list.

Scan Duration

This setting limits the scan time to the value set. The maximum time is 48 seconds. The default is 10 seconds.

Get Friendly Names

By default, APx asks for a device's friendly name. In automated processes, this process adds extra time. Uncheck this checkbox to disable this feature.

Filter Devices From Scan

You can filter the scan so only a certain Bluetooth Device Class will be shown. Click the **Browse** ... button to open the Edit Device Class dialog. This feature is not supported by the Legacy Bluetooth Option.

Device Class

If a Device Class has been selected, it is displayed here. This feature is not supported by the Legacy Bluetooth Option.

Max Devices

You can limit the number of devices found in the scan here. The default is 10 devices. This feature is not supported by the Legacy Bluetooth Option.

Current Devices List

This list shows the devices discovered in the scan, along with information fields. Highlight a device row to configure the relationship. There are controls to pair or unpair the device.

Clear Devices

This opens a menu to removes all or a selection of devices from the list.

Pair

Pairing provides a way for devices to exchange link keys. To pair with a device, refer to the documentation received with the device. To enable pairing, you will typically be required to execute a command or series of commands at the device. When pairing is enabled, select the device in the Current Devices List, and click Pair. A Pairing dialog box will open.

Some devices may require the exchange of a PIN code. The APx Bluetooth PIN code is set to 0000; a field in the pairing dialog allows input of a remote device's PIN code.

Pairing information is saved in the project, until the Current Devices List is cleared, or the until device is Unpaired.

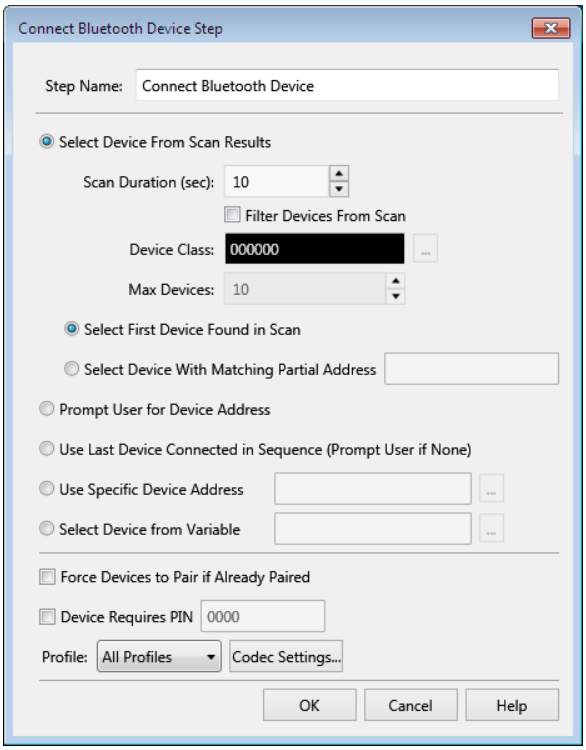
Unpair

Unpairs the selected, previously paired device.

Select Device

In the Current Devices List, highlight the device and click Select Device.

Connect Bluetooth Device Step



This dialog is called by the Bluetooth sequence step **Connect Bluetooth Device**.

Other then the Step Name field, all the settings here are the same as those in the Connect Bluetooth Device Settings dialog, page 558.

Step Name

Enter a Step Name here.

See page 558 for complete descriptions for the rest of the settings choices in Connect Bluetooth Device step.

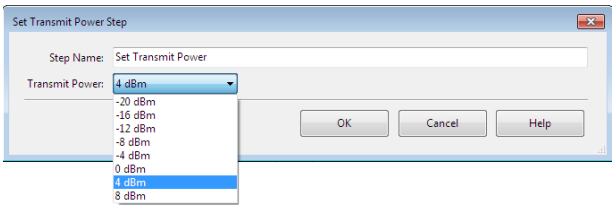
Disconnect Bluetooth Device Step

This step disconnects the connected Bluetooth device.

Set Transmit Power step

Set Transmit Power opens the Set Transmit Power step dialog, where you can set one of eight RF output trans-

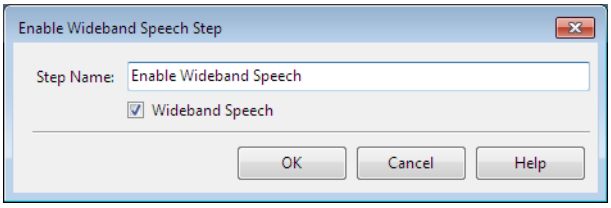
mission power levels between –20 dBm and +8 dBm. The default is +4 dBm.



Click OK to place the step in the sequence.

The Set Transmit Power feature is not supported by the Legacy Bluetooth Option.

Enable Wideband Speech step



This step allows you to enable or disable Bluetooth mSBC Wideband Speech.

Note: HFP 1.7 wideband speech is only available if the mSBC codec is supported by your Bluetooth module hardware. See Legacy Bluetooth Option for more information.

This step requires that a Bluetooth device is connected.

Inserting an Enable Wideband Speech step

Expand the **Sequence Steps** folder in a measurement node in the Navigator, click **Add Step**, and choose **Bluetooth > Enable Wideband Speech...**

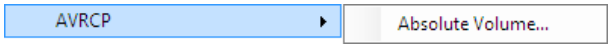
Step Name

Enter an optional **Step Name** here.

Wideband Speech

- If this checkbox is checked, **Wideband Speech** is enabled.
- If it is not checked, **Wideband Speech** is disabled.

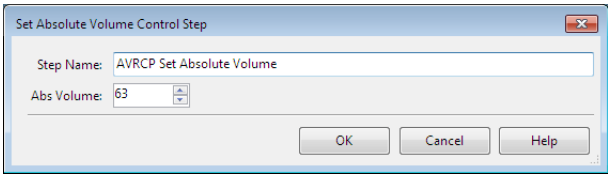
Bluetooth Set Absolute Volume step



AVRCP Absolute Volume control features are not supported by the Legacy Bluetooth Option (page 9).

Inserting a Set Absolute Volume step.

Expand the **Sequence Steps** folder in a measurement node in the Navigator, click **Add Step**, and choose **Bluetooth > AVRCP > Absolute Volume.**



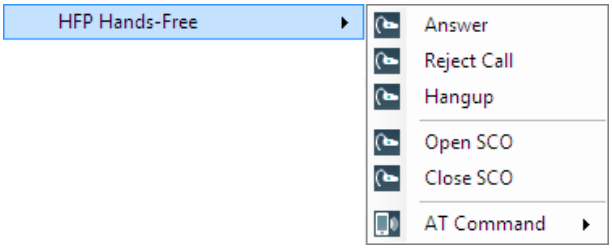
A2DP Source / AVRCP Target profile

When APx is configured as a source, you can add a step to a sequence to send an AVRCP absolute volume control command to the remote sink device. The range is 0 to 127.

A2DP Sink / AVRCP Controller profile

When APx is configured as a sink, you can add a step to a sequence to send an absolute volume status that the remote source device can read. The range is 0 to 127.

HFP Hands Free action steps



Inserting an HFP Hands-Free Action step

The current Bluetooth profile must include HFP Hands-Free.

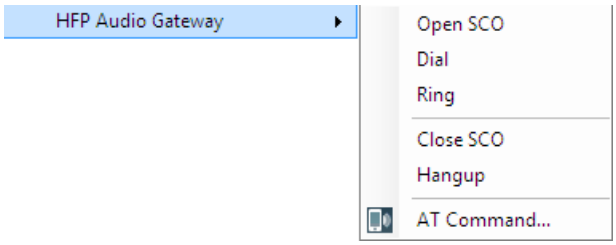
Expand the **Sequence Steps** folder in a measurement node in the Navigator, click **Add Step**, and choose **Bluetooth > HFP Hands-Free >.**

When using the HFP Hands Free profile (a sink profile), the following actions are available:

- Answer
- Reject Call
- Hangup
- Open SCO
- Close SCO
- AT Command opens the AT Command Step dialog (see page 563)

Read more about Bluetooth Actions beginning on page 140.

HPF Audio Gateway action steps



Inserting an HFP Audio Gateway step

The current Bluetooth profile must include HFP Audio Gateway.

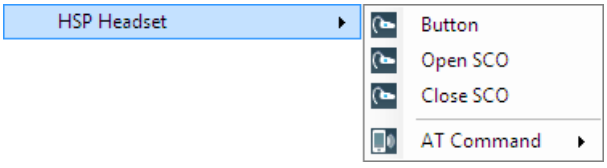
Expand the **Sequence Steps** folder in a measurement node in the Navigator, click **Add Step**, and choose **Bluetooth > HFP Audio Gateway >.**

When using the HFP Audio Gateway profile (a source profile), the following actions are available:

- Open SCO
- Dial
- Ring
- Close SCO
- Hangup
- AT Command opens the AT Command Step dialog (see page 563)

Read more about Bluetooth Actions beginning on page 140.

HSP Headset action steps



Inserting an HSP Headset step

The current Bluetooth profile must include HSP Headset.

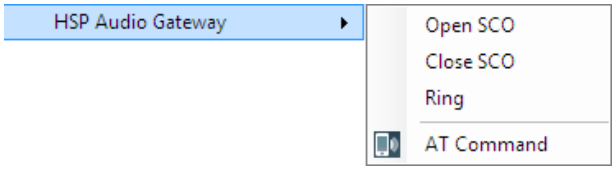
Expand the **Sequence Steps** folder in a measurement node in the Navigator, click **Add Step**, and choose **Bluetooth > HSP Headset >.**

When using the HSP Headset profile (a sink profile), the following actions are available:

- Custom
- Open SCO
- Close SCO
- AT Command opens the AT Command Step dialog (see page 563)

Read more about Bluetooth Actions beginning on page 140.

HSP Audio Gateway action steps



Inserting an HSP Audio Gateway step

The current Bluetooth profile must include HSP Audio Gateway.

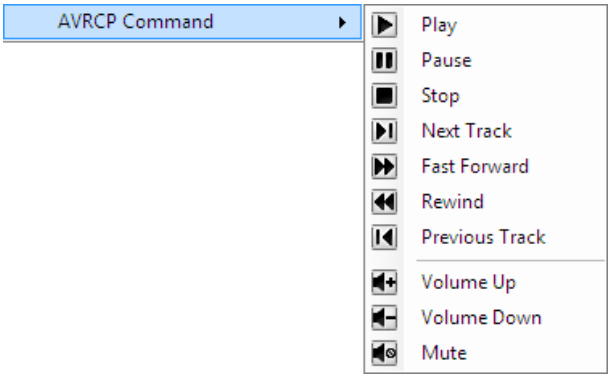
Expand the **Sequence Steps** folder in a measurement node in the Navigator, click **Add Step**, and choose **Bluetooth > HSP Audio Gateway >.**

When using the HSP Audio Gateway profile (a source profile), the following actions are available:

- Open SCO
- Close SCO
- Ring
- AT Command opens the AT Command Step dialog (see page 563)

Read more about Bluetooth Actions beginning on page 140.

AVRCP Command action steps



Inserting an AVRCP Command step

The current Bluetooth profile must include AVRCP Controller.

Expand the **Sequence Steps** folder in a measurement node in the Navigator, click **Add Step**, and choose **Bluetooth > AVRCP Command >.**

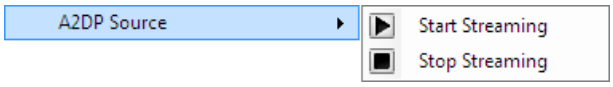
When using the AVRCP Controller profile (a sink profile), the following commands are available to control the source device:

- Play
- Pause
- Stop
- Next Track
- Fast Forward

- Rewind
- Previous Track
- Volume Up
- Volume Down
- Mute

Read more about AVRCP Controller commands on page 143.

A2DP Source action steps



The current Bluetooth profile must include A2DP Source.

When using the A2DP Source profile, the following A2DP commands are available:

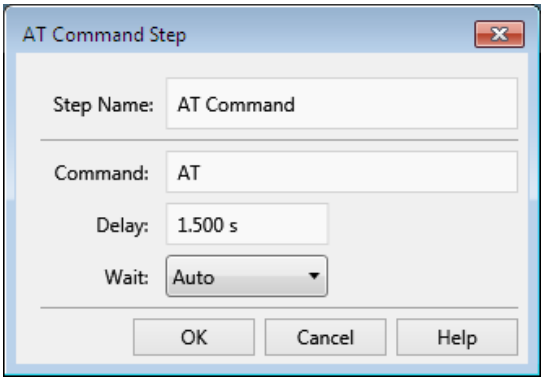
- Start Streaming
- Stop Streaming

Read more about A2DP on page 143.

AT Command step

When APx is set to any HFP or HSP Bluetooth profile, you can insert add an AT Command step to any measurement in a sequence. To insert a step: expand the **Sequence Steps** folder in a measurement node in the Navigator, click **Add Step**, and choose **Bluetooth > [HSP] [HFP] > AT Command...**

The following dialog will open:



See page 142 for more information about AT commands.

Step Name:

Enter an optional name for this step.

Command:

AT commands are widely used for controlling a GSM phone. Enter any valid AT command as a text string.

Delay:

A device may receive an AT command and respond "OK" in a timely fashion, but may not be able to immediately complete the requested action. APx delays the completion of the AT command step for the time entered in the Delay field. Valid entries are in the range of 0 seconds to 300 seconds; 1.5 seconds is the default.

Wait

Most AT commands have an expected OK, ERROR, or other response. Certain AT commands are unsolicited, or are sent with no expectation of a response. APx will indicate an error if an AT command does not provide an expected response.

- **Auto**
  If the sent command begins with "AT", APx waits for an "OK" response, until a timeout of 10 seconds. If the sent command is anything else, APx does not wait for a response.
- **Wait for Response**
  APx waits for an "OK" response to any command, until a timeout of 10 seconds.
- **Do Not Wait**
  APx does not wait for a response.

Using Pass / Fail Limits in an automated sequence

Automated testing is usually done in conjunction with Pass / Fail limits. A limit is a value you choose as a maximum or minimum acceptable value for a measurement parameter.

You may, for example, require that no device channel have a signal-to-noise ratio less than 40 dB, or that the device's frequency response never vary more than 3 dB across a certain bandwidth. You can set limits in the signal-to-noise ratio and frequency response measurements to reflect these requirements. When the sequence is run, tests that satisfy the limits are marked as **Passed**; those that exceed the limits are marked as **Failed**.

Displaying a message on Pass/Fail

You can display a message to the operator at the end of a sequence in response to Pass/Fail. See page 537.

Sending an Aux Control Out command on Pass/Fail

You can set the Aux Control Out state at the end of a sequence in response to Pass/Fail. See page 566.

See page 671 for more information about creating limits.

Sequences and Settling

For single value measurements, settling is applied only as a measurement or sequence is **Run**.

Stepped Sweep measurements are always settled, and Continuous Sweep and Signal Analyzer measurements are always unsettled.

See Chapter 98 for more information about settling.

Reports

The Report is the output document of a sequence. A report is generated when a sequence is run.

See Chapter 87 for more information about reports.

Production Test

A Production Test project is designed for product testing in a manufacturing production facility.

A project saved as a Production Test is always a locked project; the control to designate a project as a Production Test is found on the Lock Project dialog.

Use case

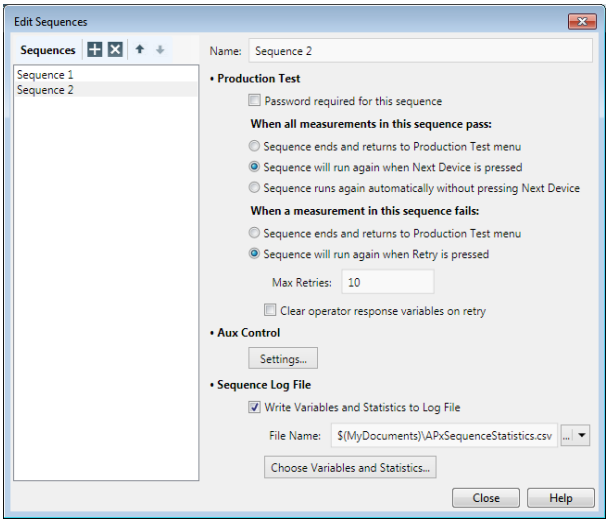
For a Production Test, the assumption is made that two types of users will be interacting with the project: **supervisors** and **operators**.

Supervisors are typically managers or engineers who are authorized to design, verify and maintain the APx500 production test project. Supervisors have password access to unlock and modify the project.

Operators are individuals who run the production test project on the manufacturing floor. Operators do not have password access to the APx project, and are presented with a simplified interface of menus and prompts to test the manufactured products quickly, accurately and easily.

Edit Sequences dialog

You can add one or more sequences (up to 32) to a project from this dialog. You can set or edit a number of settings for each sequence here, including designation as a Production Test sequence.



Edit Sequence: Sequences

All the sequences in the project are shown in the panel on the left.

Click + to add a sequence.

Click ✕ to delete a highlighted sequence, click ↑ to move a highlighted sequence up in the list, click ↓ to move a highlighted sequence down the list. List order determines the order of the sequence buttons on the Production Test menu.

Edit Sequences: Name

You can rename the selected sequence here.

Edit Sequences: Production Test

The settings here determine the behavior of the selected sequence when run in Production Test mode.

Password required for this sequence

When the Production Test menu is displayed, only sequences that are not password protected are shown. Sequences that require a password are saved in the project, but they are not visible in the Production Test menu, nor are they available to the operator. A supervisor must unlock the project to access a password-required sequence.

When all measurements in this sequence pass:

- **Sequence ends and returns to the Production Test menu**
  The operator is presented with the full menu of choices.
- **Sequence will run again when Next Device is pressed**
  The sequence that has just completed is the only available sequence. It will run again when the operator presses **Next Device**.

- **Sequence runs again automatically without pressing Next Device**
  The sequence that has just completed will run again without pressing **Next Device**.

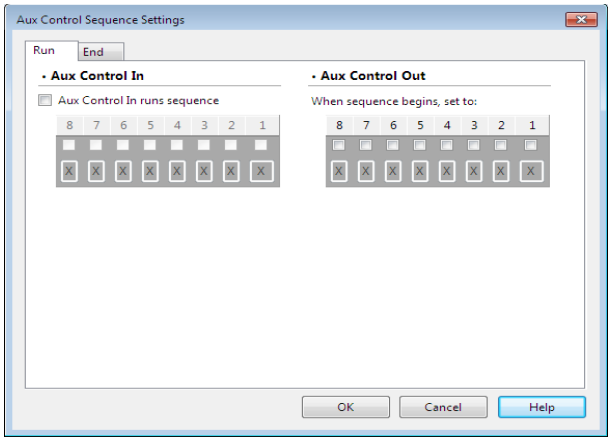
When a measurement in this sequence fails:

- **Sequence ends and returns to Production Test menu**
  The operator is presented with the full menu of choices.
- **Sequence will run again when Retry is pressed**
  The sequence that has just completed will run again when the operator presses Retry.
  - **Max Retries**
    This is the maximum number of retries allowed, set by the supervisor in the field here.
  - **Clear operator response variables for each retry**
    If this checkbox is unchecked, the current variables entered as operator responses are maintained. If the checkbox is checked, the variables entered as operator responses that are not marked **Persist** are cleared. The **Persist** setting for operator response variables is set in the sequencer **Prompt** step (page 543). Operator response variables marked **Persist** and all other variables (including user variables that are not operator responses) are maintained and not cleared.

Edit Sequences: Aux Control Sequence Settings

You can set a number of **Aux Control** settings related to this sequence. Click the **Aux Control: Settings** button to open the **Aux Control Sequence Settings: Run** (page 565) and **Aux Control Sequence Settings: End** (page 566) dialogs.

Aux Control Sequence Settings: Run tab



Running a Sequence with an Aux Control IN command

When this control is enabled, an **Aux Control In** command that satisfies the pattern set here will run the sequence.

Check the **Aux Control In Runs Sequence** checkbox to enable this control.

Then, set the desired **Aux Control In** pattern in the control display. To do this, activate each bit you are interested in by setting a checkmark under the bit. Then let the desired state (0 or 1) for each activated bit. The state of bits that are not activated (unchecked, state shown as “X”) will be ignored for this control.

In the illustration above, the sequence will run when **Aux Control In** bit 8 is high (1) and bit 5 is low (0). The states of bits 7, 6, 4, 3, 2, and 1 are ignored.

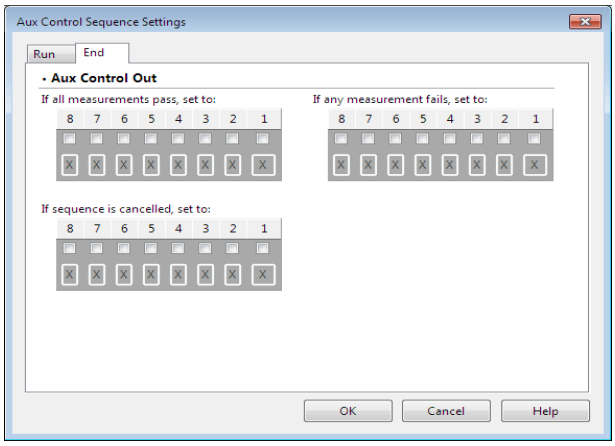
Setting the initial state of Aux Control Out when a sequence begins

If you are using **Aux Control Out** in a sequence, the initial state of the **Aux Control Out** bits must be set to the desired pattern. If a sequence is repeated, the **Aux Control Out** bits must be reset to this pattern at the beginning of the repeating sequence. This control allows you to specify the initial state of **Aux Control Out** for the beginning of the sequence.

Set the desired **Aux Control Out** pattern in the control display **When Sequence Begins, Set To:**. To do this, activate each bit you are interested in by setting a checkmark under the bit. Then set the desired state (0 or 1) for each activated bit. The state of bits that are not activated (unchecked, state shown as “X”) will be ignored for this control.

In the illustration above, at the beginning of the sequence **Aux Control Out** will be set as follows: bits 8 and 7 are low (0), bits 4, 3, 2 and 1 are high (1). The states of bits 6 and 5 are ignored; their previous state is maintained. See Chapter 102 for more information about Aux Control.

Aux Control Sequence Settings: End tab



Setting the state of Aux Control Out when a sequence ends

A sequence can end three ways: it is canceled, or it ends successfully with all measurements passed, or it ends successfully with at least one measurement failed. **Sequence Properties: End** allows you to send an **Aux Control Out** command for any of these sequence results.

Set the desired **Aux Control Out** pattern in the control display for any of the three end states (**All Measurements Passed, Any Measurement Failed, Sequence Canceled**).

To do this, activate each bit you are interested in by setting a checkmark under the bit. Then set the desired state (0 or 1) for each activated bit. The state of bits that are not activated (unchecked, state shown as "X") will be ignored for this control.

In the illustration above, when the sequence ends with all measurements passed, **Aux Control Out** will be set as follows: bits 8 and 7 are low (0), bits 4, 3, 2 and 1 are high (1). The states of bits 6 and 5 are ignored; their previous state is maintained. See Chapter 102 for more information about Aux Control.

Edit Sequences: Sequence Log file

You can save variables (page 31) and production test statistics (page 566) for a sequence or a series of sequences in a log file.

Write Variables and Production Test Statistics to a Log File

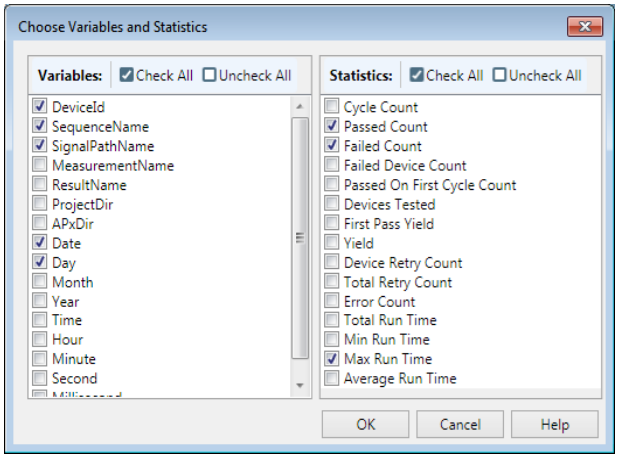
Check this box to save variables and statistics for a sequence or a series of sequences in a log file.

File Name

When **Write Variables and Statistics to a Log File** is checked, you can enter a file name and path here. You can use variables in specifying the path by accessing the menu to the right of the entry field.

Choose Variables and Statistics

Click here to open the **Choose Variables and Statistics** dialog.



Production Test Statistics

Production test statistics provide an organization with data to evaluate product production runs and the performance of the production testing regimen.

The statistical data can be written to the Sequence Log File (see page 566) for evaluation. You can choose to report any combination of these data types:

- **Cycle Count**
  The number of times the sequence has been run since the project was loaded.
- **Passed Count**
  The number of times the sequence was executed and passed since the project was loaded. This would also equal the number of devices passed assuming the user doesn't run the same device again after it passes.
- **Failed Count**
  The number of times the sequence was executed and failed since the project was loaded.
- **Failed Device Count**
  The number of devices that failed to pass after all allowed retries are exhausted.

Retry failures are not counted in this statistic. Only failure of the final retry is counted.

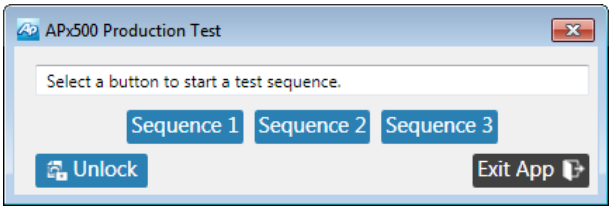
- **Passed on First Cycle Count**
  The number of devices that passed without a retry.
- **Devices Tested**
  Total Number of devices tested = (Passed Count) + (Failed Device Count).
- **First Pass Yield**
  First Pass Yield = (Passed on First Cycle Count) / (Devices Tested).
- **Yield**
  Yield = (Passed Count) / Devices Tested

- **Device Retry Count**
  Total number of sequence retries that were executed for a particular device. If the device passed on the first try the Device Retry Count will be 0.
- **Total Retry Count**
  The summation of Device Retry Counts since the project was loaded.
- **Error Count**
  The total number of APx Errors encountered during the sequence run since the project was loaded.
- **Total Run Time**
  The total time spent running the sequence from beginning to end for all runs since the project was loaded.
- **Min Run Time**
  The minimum time required for the sequence to run from beginning to end since the project was loaded.
- **Max Run Time**
  The maximum time required for the sequence to run from beginning to end since the project was loaded.
- **Average Run Time**
  The average time required for sequence run from beginning to end since the project was loaded.

Opening a project saved as a production test

When a project has been saved as a Production Test, it opens showing the Production Test menu. A supervisor with access to the project password can unlock the project to make changes; other users are confined to running the sequences listed on the Production Test menu.

The Production Test menu



The Production Test menu is shown when the project has been locked with a password and saved as a Production Test. The menu has these features:

- A text prompt or greeting
- A list of available operator sequences in the project. A sequence must have at least one measurement and not be password protected to be shown in this list. This list may be empty.
- An **Unlock** button, for supervisor access to the project. The project password is required to unlock a project.
- An **Exit App** button, which closes the project and exits APx500.

The Production Test menu may be displayed with the APx500 user interface, or the APx500 user interface may be optionally suppressed by the supervisor so that only the Production Test menu appears.

The Production Test menu is designed by the supervisor using the APx500 Navigator and Sequencer, and especially the Edit Sequences dialog, page 564.

The APx500 API

Automating APx500 with the APx500 API

You can control APx500 from external programs by calling functions using the APx500 Application Programming Interface (API). The APx500 User's Manual does not cover use of the APx500 API; instead, refer to the documentation cited below.

We are currently supporting API automation for programmers using Visual Basic .NET, Visual C# .NET, and National Instruments' LabVIEW programming environment. You will find tools and information on the Audio Precision Web site at ap.com. You can download the APx API Developer Tools at http://ap.com/display/file/138. For more information about APx automation and the APx API, go to http://ap.com/products/apx/automation.

Tools from Microsoft

Visual Basic.NET

To create or edit VB.NET programs, you will need a Visual Basic editor for the .NET framework, such as Microsoft Visual Studio or Microsoft Visual Basic Express Edition. See information below regarding a free download of Visual Studio Express.

Visual C#.NET

To create or edit Visual C# .NET programs, you will need a current version Visual C# editor for the .NET framework, such as Microsoft Visual Studio or Microsoft Visual C# Express Edition. See information below regarding a free download of Visual Studio Express.

Visual Studio Express Free Download

You can download the current version of Visual Studio Express at http://www.microsoft.com/express/Windows.

Note: you must run Visual Studio or Visual Studio Express as an Administrator to open APx500 API templates and projects.

On the Audio Precision Web site at ap.com:

Refer to the file Getting_Started_with_the_APx-_API.pdf.

API / Documentation Tools

- APx500 Programmer's Reference Guide
  The APx500 Programmer's Reference Guide (PRG) documents the published API library for APx500, with support for VB.NET. The filename is "APx-500_API_PRG.chm". This is a stand-alone Microsoft HTML Help file (H1). This file is installed with APx500 and is available via the Windows Start > All Programs > Audio Precision > APx500 n.n > API menu.
- API browser
  "APx_API_Browser.exe" is a utility that generates a complete hierarchical tree for the APx500 API. This utility is available on the Audio Precision Web site at ap.com. The program will generate documentation directly from the API files installed as components of APx500. This file is installed with APx500 and is available via the Windows Start > All Programs > Audio Precision > APx500 n.n > API menu.

API / Examples

A number of sample files have been developed to help you get started with using the APx500 API. The examples include:

- APx Information Sample
- Configure Switchers
- Control Generator Settings
- Display Meter Results
- Display Sequence Results
- Enable Signal Monitors
- Export Sample
- Manufacturing Audio Test
- Measurement Control Sample
- Read Sweep Values
- Reference Levels
- Run Measurement
- Run Sequence

API / LabVIEW Driver

National Instruments' LabVIEW is a graphical programming language that uses icons instead of text to create applications. LabVIEW is in wide use in automated test applications. For more information about LabVIEW, visit NI's Web site at http://www.ni.com/labview/.

Audio Precision provides a LabVIEW driver for APx500 on the Audio Precision Web site at ap.com. This driver is version specific; check the Audio Precision Web site for version information and other documentation.

The LabVIEW driver is quite comprehensive, with LabVIEW VIs for almost all of the current APx500 measurements. Visit our Web site at ap.com for more information about the LabVIEW driver.

API / Project Templates and Help

These resources are setup files that install APx API Visual Basic and Visual C# project templates. These projects include completely integrated Microsoft Help documentation for the APx API. Installers are provided for the current versions of:

- Microsoft Visual Basic Express Edition
- Microsoft Visual C# Express Edition
- Microsoft Visual Studio

Getting Started

A basic overview of the APx API is provided in

- API\Getting_Started_with_the_APx_API.pdf.

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Reports

A report is the primary output of results from APx500 measurements. A report is created when a Sequence is run, or when a measurement is made using the **Start Measurement** command (found in the Navigator context menu and the Project menu).

Report types

There are two types of report:

- the APx Default Layout report, a simple, largely fixed approach. See page 569.
- the Microsoft Word report, which can be customized. See page 570.

The report type can be selected from the **Type** list in the **Edit Properties > General**, described on page 571.

For a quick output document from a single measurement view, see Printing and Exporting Results on page 573.

Your choice of report type and options will greatly affect what results are reported and how the results are presented. The Microsoft Word report, though more complicated to use than APx Default Layout report, allows you to completely control the content and presentation of the results using embedded tags.

Report Properties

Right-click on the Report node in the Navigator to open the Edit Properties dialog box.

- You can select the report **Type**, and add a title or notes in the **Edit Properties > General** tab. See page 571.
- Reports can automatically be saved at the completion of a sequence. You can select **Auto-Save** options in the **Edit Properties > Auto-Save** tab. See page 571.
- You can select and format graphics to insert in your report in the **Edit Properties > Logo** tab. See page 572.

What's included in a report?

A report can include information about the APx instrument used, the configuration of each signal path, and measurement results in detail (including graphic display of graphs and meters) or in a summary. Reports can include a title, a time stamp, notes, and selected images. Pass/Fail information is reported and can be used to filter the report content.

Viewing a report

If the **Report** node in the **Navigator** is checked, the report is shown at the completion of the sequence or measurement. If the node is not checked, the report is created but not shown. Click the **Report** node to show the report.

The APx Default Layout report is viewed in an APx500 window. Navigation, printing, save to file and exporting controls are provided in a Toolbar in the viewing window.

The Microsoft Word report is viewed in an instance of Microsoft Word invoked by APx500. Navigation, formatting, save to file, printing, etc. are controlled from within Microsoft Word.

The APx Default Layout report type

The APx Default Layout report is the default report type. You can also select it by choosing APx Default Layout as the Type in the Edit Properties > General dialog (right-click on the Report node in the Navigator). See Edit Report Properties.

An APx Default Layout report takes the form of one or more 8.5" by 11"-formatted pages created within APx500 and initially displayed in an undocked APx500 window.

Once displayed, an APx Default Layout report can be printed to any printer installed for Microsoft Windows. Reports can also be exported to a *.PDF (Adobe Acrobat), *.html (HTML, or hypertext markup lan-

guage),*.rtf (rich text format), *.xls (MS Excel spreadsheet) or *.txt (text) file. Reports can be formatted with custom margins and have a custom title and company logo or other image applied.

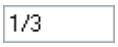
An APx Default Layout report can include measurement setup and summary information, test results, pass / fail results (or only failed results), user prompts and meter bars or XY graphs.

APx Default Layout Report toolbar



-  **Edit Properties** opens the Edit Properties dialog box.
  You can select the report Type, specify included information and set margins, show date, time or page numbers and add a title or notes in the **Edit Properties > General** tab. See page 571.

  Reports can automatically be saved at the completion of a sequence. You can select Auto-Save options in the **Edit Properties > Auto-Save** tab. See page 571.

  You can select and format graphics to insert in your report in the **Edit Properties > Logo** tab. See page 572.
-  **Export** opens the Export browser. Export your report to PDF, HTML, RTF, XLS or CSV.
-  **Print** opens the Windows Print dialog.
-  **Copy** copies the current page to the Windows clipboard.
-  **Find** opens the Find dialog.
-  **Single Page View** sets the view to one page.
-  **Multiple Page View** sets the view to multiple pages.
-  **Continuous Scroll** enables vertical scrolling of pages.
-  **Zoom Out**
-  **Zoom In**
-  **Zoom menu**
-  **Previous Page**
-  **Next Page**
-  **Current Page Number**

Microsoft Word Report

The APx Microsoft Word report type is an option that provides much more flexibility in the selection and presentation of results, compared to the APx Default Layout report type. This is a more complicated approach, however, and requires some effort to properly format a custom layout to suit your needs.

Microsoft Word report layouts

This method applies a Contents Control tag to each report element (such as a result, result name, image, setting, etc.), and places the results and tags into an Microsoft Word layout file. A layout file acts as a template, containing the Contents Control tags and the formatting for a custom report. APx provides a default layout for Microsoft Word; you can use this file as a starting point for your custom layouts. Once you have created and saved a custom layout file by editing the default layout, you can select it as your report layout in the future.

Select Microsoft Word as the **Type** in the **Edit Properties > General** dialog. See **Edit Properties** on page 571.

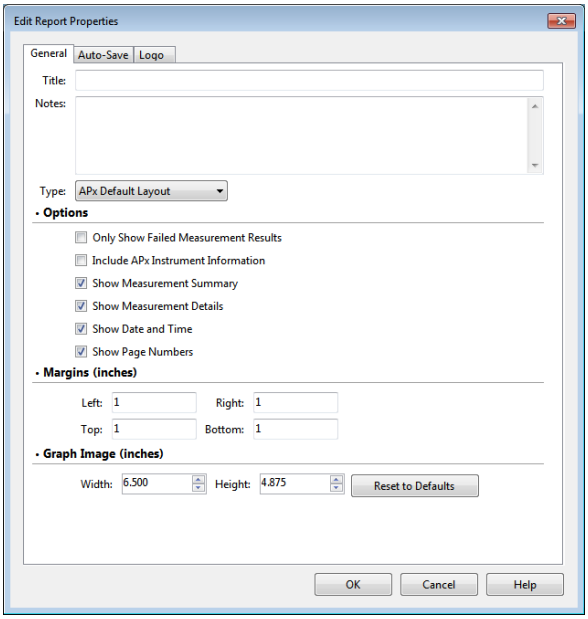
Required software

To create or edit Microsoft Word reports, you must have Microsoft Word 2013 or Microsoft Word 2010 installed on the PC. Microsoft Word 2003 and 2007 are no longer supported. Specifics follow:

Tutorials for Microsoft Word reports

There are three tutorials in the APx500 online Help to instruct you in working with the Microsoft Word report feature. We recommend that you use the tutorials from within the Help file.

Edit Properties: General tab



Choosing the Report Type

APx Default Layout

The APx Default Layout is the default report type selection. This choice creates a multipage report with graphics in the standard APx style that can be viewed in an APx window. You can export the report in a number of different formats. See APx Default Layout for detailed information about using this report type.

Microsoft Word

The Microsoft Word report type requires more effort in formatting, but can be completely customized. See Microsoft Word report for detailed information about using this report type.

APx Default Layout: Report Options

Only Show Failed Measurement Results

Only failed measurement results are reported.

Include APx Instrument Information

Adds the APx Instrument ID, its adjustment date and the APx500 software version number.

Show Measurement Summary / Show Measurement Details

These controls combine to set the degree of detail needed for your report.

APx Default Layout: Margins

Margins for the Microsoft Word style are set from within Microsoft Word.

Margins (inches)

Set page margins here. Units are inches. Defaults are 1" on each side.

APx Default Layout: Other Settings

Show Date and Time

If this box is checked, the date and time that the report was generated is printed at the bottom of each page.

Show Page Numbers

If this box is checked, the page number is printed at the bottom of each page.

Microsoft Word report layout: Report Options

You can select a default or custom layout here and can open Word to edit the layout.

Both Layout types

Title

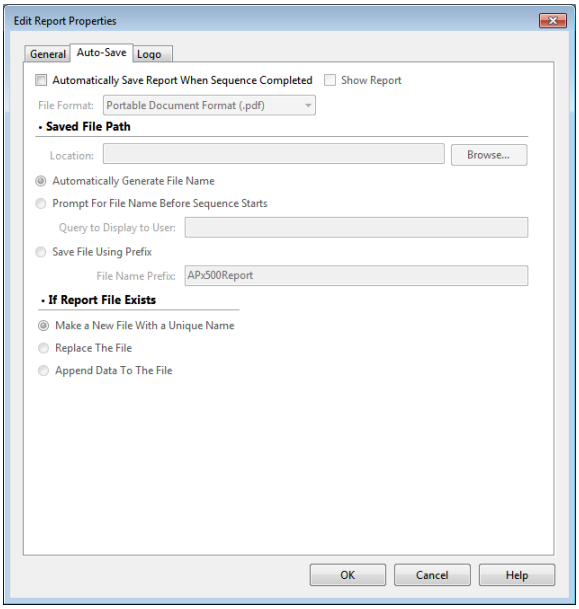
You can enter a report title with text of your choice. For the APx Default Layout, the title appears in the report title bar at the top of each page. For the Microsoft Word report type, the title appears in the “Title” tag.

Notes

You can enter notes for the report with text of your choice. The notes appear below the report title bar at the top of the first page.

You can replace the default title with text of your choice. The title appears in the report title bar at the top of each page.

Auto-Save Tab



Automatically Save Report When Sequence Completed

If this box is checked, the report is automatically saved to disk at the end of the sequence.

Show Report

If you have checked **Automatically Save Report When Sequence Completed**, you can choose to also show the report on-screen when it is generated, or to save it without showing.

File Format

For APx Default Layout reports, choose the format for automatic report save here. Options are

- PDF (Adobe Portable Document Format)
- HTML (Hypertext Markup Language)
- RTF (Rich Text Format)
- XLS (Microsoft Excel spreadsheet)
- CSV (Comma Separated Value text; data only)

For Microsoft Word reports, the format is

- .docx

Saved File Path

Location

Choose the folder for the automatically saved reports here.

Automatically Generate File Name

If this box is checked, the APx500 project name is used as the file name for automatic report saving. A time/date stamp is added to the file name. For example, a project named “My Project” and save on the afternoon of January 20, 2010 might have the name “My_Project_01_20_2010_15_14_04.docx”.

*Note: in this case, avoid the characters \ / : * ? " < > | in the Device ID. These are illegal characters in a Windows file name, and APx will replace these with underscores "_" before saving.*

Prompt for File Name Before Sequence Starts

If this box is checked, a prompt at the end of the sequence will query the user for a file name for the report (such as DUT serial number), for automatic report saving.

Query to Display to User:

Enter the text for the user filename prompt query here.

Save File Using Prefix

This option will save the report with a prefix to the file name.

File Name Prefix

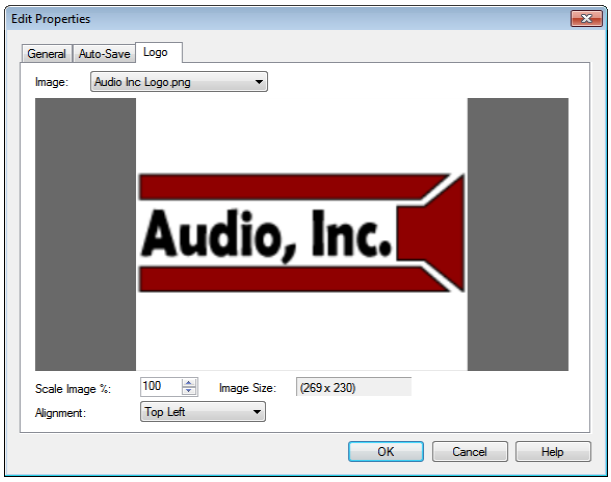
Enter the file name prefix here.

If Report File Exists

If the file already exists, you can select how Auto-Save behaves.

- **Make a new file with a unique name**
  APx will add a number (1, 2, 3, etc.) to the end of the file name, incrementing the number by 1 with each save.
- **Replace the file**
  APx will delete the previous file and save the new file with the same name.
- **Append data to the file**
  Each save operation will append the new data to the end of the file.
- **Update the data in the file** (Microsoft Word reports only)
  The file on disk will be opened, and new data will replace old data on a measurement-by-measurement basis.
  We recommend that you finalize your report layout before you use this feature. After the report is created the first time, subsequent reports use the layout in the file on disk. If you make layout changes (such as adding new tags), these changes will be ignored when the next report is generated. To save new layout changes, delete the report file on disk before creating a new report. Note that all the data in the old report file on disk will be lost.

Logo tab



You can place a logo or other image in your report. For the APx Default report style, the artwork is placed at the top of each report page, just below the report title bar. For the Microsoft Word report style, the artwork is placed at the “Logo” tag.

Logo

If an image file is loaded for this report, it is shown in the **Logo** field.

Image Size

If an image file is loaded for this report, its size in pixels is shown in the **Image Size** field.

Delete

Click the **Delete** button to remove an image file from the report.

Browse

Click the **Browse** button to open a file browser. Navigate to the image file you wish to select, and click **Open**. Supported image file types are *.bmp, *.jpg, *.gif, *.tif, *.png and *.ico.

Scale Image

You can reduce the size of the image in the report by setting **Scale Image** to less than 100%. Image aspect ratio is maintained when scaled. For the Microsoft Word report style, the artwork can be re-scaled in Word.

Alignment

You can choose whether the image is aligned to the left, the center, or the right of the report. For the Microsoft Word report style, the artwork can be re-aligned in Word.

About Image files

Image files are used by APx500 to display graphical images in Sequencer user prompts, or to display images such as company logos in reports. Image files are inserted for current use and are also attached to the project file.

Image files attached to the project but not currently in use can be removed by navigating to the dialog at **File > Manage Attached Project Items**. See page 27 for more about managing attached project items.

Supported image file formats...

- .bmp
- .jpg
- .jpeg
- .gif
- .tif
- .tiff
- .png
- .ico

Printing or Exporting Results

This feature only prints or exports results from the current view. For more powerful reporting capabilities for measurements and entire sequences, see the APx Default Layout report on page 569, or the Microsoft Word report on page 570.

To print or export results of a single measurement view, navigate to the view, run the measurement and click **Print/Export Results for this view** from the toolbar menu at the top of the meter bar or graph display. The results are settled for the report and are compared to limits, if set.

*The **Copy graph image to clipboard** feature does not generate a set of printable measurement results. It simply copies the graph image to the Windows clipboard. This image can then be pasted into another application. If the data on screen is unsettled, the image that is copied is also unsettled.*

The results are shown in an undocked APx500 window. Use the window's Toolbar commands to navigate, format or to print or export the results.

Toolbar



- Edit Properties **Edit Properties** opens the Edit Properties dialog box.

  You can set margins, show date, time or page numbers and add a title and notes in the **Edit Properties > General** tab.

  You can select and format a graphic to insert in your results printout in the **Edit Properties > Logo** tab.

- **Export** opens the Export browser. Export your report to PDF, HTML, RTF, XLS or CSV.
- Print... **Print** opens the Windows Print dialog.
- **Copy** copies the current page to the Windows clipboard.
- **Find** opens the Find dialog.
- **Single Page View** sets the view to one page.
- **Multiple Page View** sets the view to multiple pages.
- **Continuous Scroll** enables vertical scrolling of pages.
- **Zoom Out**
- **Zoom In**
- 100 % **Zoom menu**
- **Previous Page**
- **Next Page**
- 1/3 **Current Page Number**

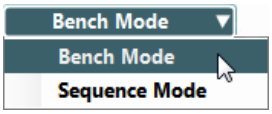
Section IV: Bench Mode

Bench Mode

For the APx515, Bench Mode requires a software option key. See page 189 for more information about software options.

Overview

APx500 has two operating modes: Sequence Mode and Bench Mode, each with its own workspace. APx normally launches into Sequence Mode. You can switch to Bench Mode using the selector in the upper right corner of the workspace.



Sequence Mode and Bench mode use the same underlying measurement engines, and when configured in the same way will provide identical results.

Bench Mode

Bench Mode is an alternative user interface introduced in 2014. Unlike Sequence Mode, Bench Mode does not offer defined measurements. Instead, Bench Mode provides a set of tools that can be assembled and used in many different ways. AP2700 and APWIN users will recognize similarities between the Bench Mode paradigm and earlier Audio Precision control software.

Bench Mode tools include many number bar graph meters that can each be set to one of over 12 signal parameters, real time monitors, and an extremely flexible set of sweep engines for stepped sweep, FFT, time recorder, continuous sweep and acoustic response sweeps. X and Y axes can be set to a broad range of parameters, and nested sweeps are supported.

The Bench Mode Workspace consists of three columns. The column to the left contains a collection of Signal Path Setup panels, very similar to the Signal Path Setup panels in Sequence Mode. The central column contains Generator and Analyzer settings. The right panel provides a selection of Tools displays, with a choice of meter, graph, sweep and other analysis tools.

Bench Mode Signal Path

Bench Mode supports only one signal path, and no Sequencer automation. There is no Navigator panel, since there are no Signal Paths nor sequenced measurements to navigate, select, deselect or run.

The Signal Path Setup menus are in Bench Mode, so you can configure the Bench Mode signal path inputs, outputs, filtering, switchers and so on, just as you can for a Sequence Mode signal path.

Bench Mode Generator and Analyzer

In Sequence Mode, the generator and analyzer are always associated with a particular measurement, and are often constrained to waveforms and operations suited to that measurement. In Bench Mode, the Generator and Analyzer settings are largely independent from the Bench Mode measurements you may be using.

Bench Mode Measurements and tools

Bench Mode provides a collection of tools and measurements. There are Signal Monitors, Meters, tools for initiation actions in a Bluetooth or HDMI or PDM configuration, and stepped Sweep, FFT, Recorder, Continuous Sweep and Acoustic Response measurements, each with a large suite of selectable results and displays.

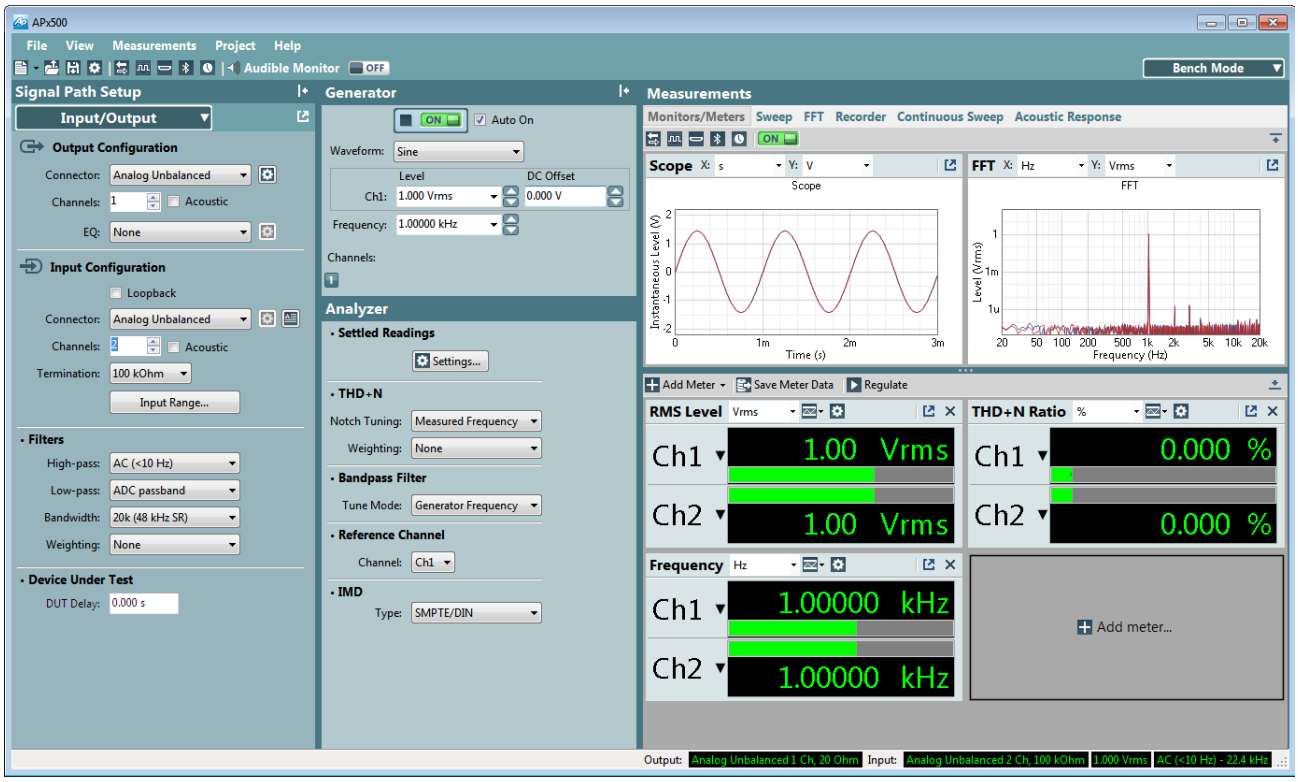
These tools and measurements are each very similar to tools and measurements in Sequence Mode, with a broader usability and application.

Hardware Support

Bench Mode and Sequence Mode support the same hardware, hardware options and accessories. Bench Mode offers slightly more capability in DCX-127 use, and in digital interface waveform analysis.

The Bench Mode Workspace

BENCH MODE WORKSPACE



SIGNAL PATH SETUP

GENERATOR AND ANALYZER

MONITORS / METERS AND MEASUREMENT TABS

Signal Path Setup

The Signal Path Setup panel and menus are located on the left of the Bench Mode workspace. The Signal Path Setup panel can be hidden.

With one exception, Signal Path Setup is identical to Signal Path Setup in Sequence Mode, with the same menus, controls and features. You can read about Signal Path Setup in Chapter 7.

Bench Mode Signal Path Setup exceptions

When Input Configuration is set to a digital input that supports interface waveform measurements, an Interface Level reading field is available on the Sequence Mode Signal Path Setup I/O menu.

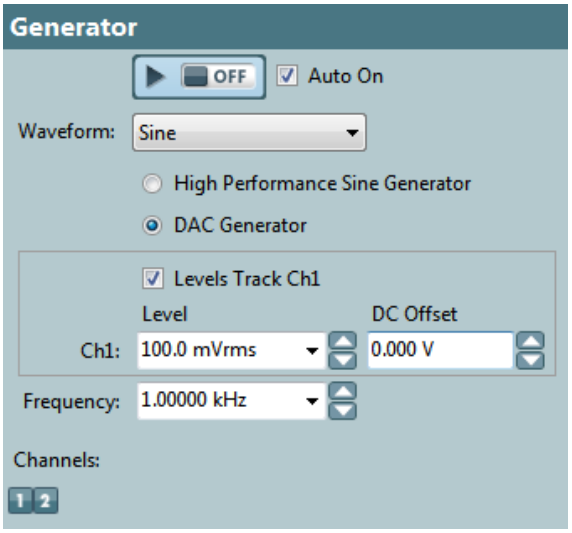
Digital Interface Level

Since there is a Digital Interface Meter available in Bench Mode, the Signal Path Setup reading field is unnecessary.

DCX measurement settings

Sequence Mode provides these settings in the context of the DCX measurements. In Bench Mode, the settings are on the Signal Path Setup DCX panel.

Generator



The Bench Mode Generator panel is located in the central set of panels in Bench Mode, above the Bench Mode analyzer panel. These panels can be hidden.

The generator is available throughout a Bench Mode session and can be adjusted at will. Three of the Bench Mode measurements (the Sweep, Continuous Sweep and Acoustic Response) temporarily take con-

trol of the generator to perform sweeps. When such sweeps are finished, the generator returns to its previous state.

On/Off

Turn the generator **ON** or **OFF** with this switch.

When **OFF**, the generator remains **OFF** unless an **Auto On** condition turns it on. When the **Auto On** condition ends, the generator reverts to **OFF**.

When **ON**, the generator remains on at its current settings unless a sweep is in progress. The sweep temporarily takes control of the generator. When the sweep is finished, the generator returns to its previous state.

Auto On

When **Auto On** is selected and the generator is off, initializing a sweep will temporarily turn the generator on. When the sweep is finished, generator is turned off.

Waveform

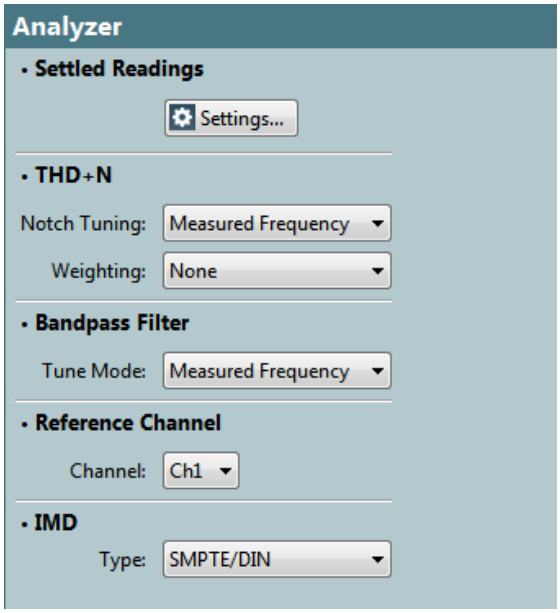
Select the generator waveform here. See Chapter 6, Signal Generation, for a full description of available generator waveforms and controls, such as Level and Frequency. The available Waveform choices change by instrument hardware and Output Configuration.

Channels

Assign the generator to output channels. By default, all active output channels are selected.

Note that **Output EQ** and other settings in **Signal Path Setup > Input/Output > Output Configuration** will affect the generator signal output. See page 55 for more information about Output EQ.

Analyzer



The Bench Mode Analyzer panel is located in the central set of panels in Bench Mode, below the Bench Mode generator panel. These panels can be hidden.

The analyzer settings are available throughout a Bench Mode session and can be adjusted at will. These settings are global within Bench Mode and affect all signal analysis.

Settled Readings

The meter readings that provide data to the Bench Mode Sweep and Recording measurements are settled. Settling is a method to identify a meter reading that represents a measured parameter accurately, excluding spurious variations caused by stimulus changes or other transient phenomena. In Bench Mode, Settling is a mechanism that provides stable, consistent results for the Sweep and Recorder tools. Read more about settling in Chapter 98.

Settings

Click the **Settings** button to open the Bench Mode **Settling** dialog, described in detail beginning on page 630.

THD+N

For meters that measure THD+N (THD+N Ratio and THD+N Level), the notch tuning and weighting filters must be specified.

See More about THD+N for a discussion of the notch filter used in THD+N measurements.

Notch Tuning

- **Generator Frequency**
  The notch is tuned to the APx generator frequency for the current step. When the generator channels are outputting different frequencies (Split Fre-

quency generation), the bandpass filter center is set to Frequency A.

- **Measured Frequency**
  The notch is tuned to the measured frequency for the current step. When the analyzer channels are receiving different frequencies, the bandpass filter for each channel is centered on the frequency in that channel.
- **Fixed Frequency**
  The notch is tuned to a fixed frequency selected by the user. When Fixed Frequency is selected, a Filter Freq: entry field becomes available beneath the Tune Mode control.

Frequency

When Notch Tuning is set to Fixed Frequency, enter the desired frequency here.

Weighting

A choice of weighting filters can be applied to the THD+N meter:

- A-wt.
- B-wt.
- C-wt.
- CCIR-1k
- CCIR-2k
- CCITT
- C-message

Deemphasis filters are also included, for use with pre-emphasized DUTs or stimulus signals.

- 50us de-emph.
- 75us de-emph.
- 50us de-emph. + A-wt
- 75us de-emph. + A-wt.

Notice that this is the same set of filters that can be applied to the overall input signal in Signal Path Setup > Input/Output. These filters are effectively in series, and are additive in their effect, which is rarely a desired result.

Also, consider that the Signal Path Setup > Input/Output filters are applied to the entire signal, while the filters in this area are only applied to the THD+N meter, which displays the distortion residuals.

Read more about filters in APx in Chapter 97.

Bandpass Filter

The bandpass meter differs from the RMS Level meter only in that it follows a narrow bandpass filter. The center of the filter is set by Tune Mode.

Tune Mode

- **Generator Frequency**
  The notch is tuned to the APx generator frequency for the current step. When the generator channels are outputting different frequencies (Split Frequency generation), the bandpass filter center is set to Frequency A.
- **Measured Frequency**
  The notch is tuned to the measured frequency for the current step.
  When the analyzer channels are receiving different frequencies, the bandpass filter for each channel is centered on the frequency in that channel.
- **Fixed Frequency**
  The notch is tuned to a fixed frequency selected by the user. When **Fixed Frequency** is selected, a **Filter Freq:** entry field becomes available beneath the **Tune Mode** control.

Frequency

When **Tune Mode** is set to **Fixed Frequency**, enter the desired frequency here.

Reference Channel

For Phase or Level Ratio measurements, you must specify a reference channel.

Channel

Select the reference channel here.

IMD

IMD measurements require special analysis methods.

Type

Select the IMD analysis method here. You should match the analysis mode to the generator IMD waveform used for stimulus. MOD IMD analysis is not available in Bench Mode.

- **SMPTE**
  This setting will analyze SMPTE and DIN IMD when used with an appropriate stimulus waveform.
- **DFD**
  This setting will analyze DFD IMD when used with an appropriate stimulus waveform In Bench Mode, only the d2 result is reported.
- **CCIF**
  This setting will analyze CCIF (ITU-R) IMD when used with an appropriate stimulus waveform. In Bench Mode, only the d2 result is reported.
- **DIM**
  This setting will analyze DIM IMD when used with an appropriate stimulus waveform. Audio Precision System One, System Two, Cascade and 2700 series instruments measured DIM differently than the **U1...U9** or **U4+U5** modes found in the APx Sequence Mode DIM measurements, using a combination of analog filters and rms meters. In Bench Mode, DIM analysis uses **Emulation** mode analy-

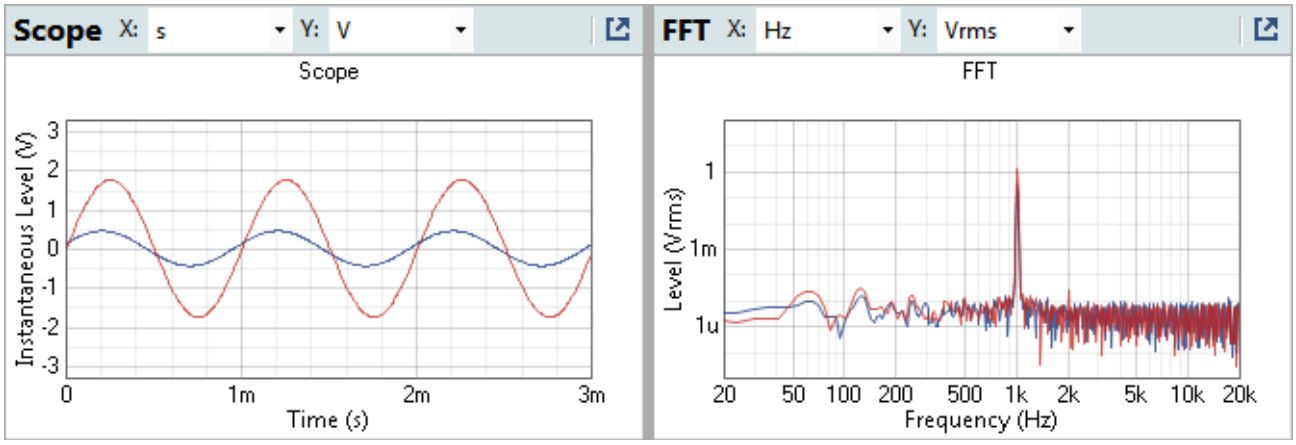
sis, providing DIM results compatible with results from earlier Audio Precision instruments.

Measurements

There are six tabbed panels in the Measurements panel, each discussed in detail in the following chapters.

- **Monitors/Meters**, Chapter 89.
- **Sweep**, Chapter 90.
- **FFT**, Chapter 91.
- **Recorder**, Chapter 92.
- **Continuous Sweep**, Chapter 93.
- **Acoustic Response**, Chapter 94.

Monitors / Meters
(Bench Mode)



Overview

The initial Bench Mode view displays the Monitors and Meters, offering multiple real-time views of the analyzed input signals.

Scope and FFT monitors

The Scope and FFT monitors are normally docked beneath the Monitors toolbar.

These the same monitors that are used in Sequence Mode.They monitor the current analyzer inputs, displaying up to 16 channels, depending upon the input configuration. When undocked, additional monitor display controls are available. Read more about the Scope Monitor on page 33, and more about the FFT Monitor on page 33.

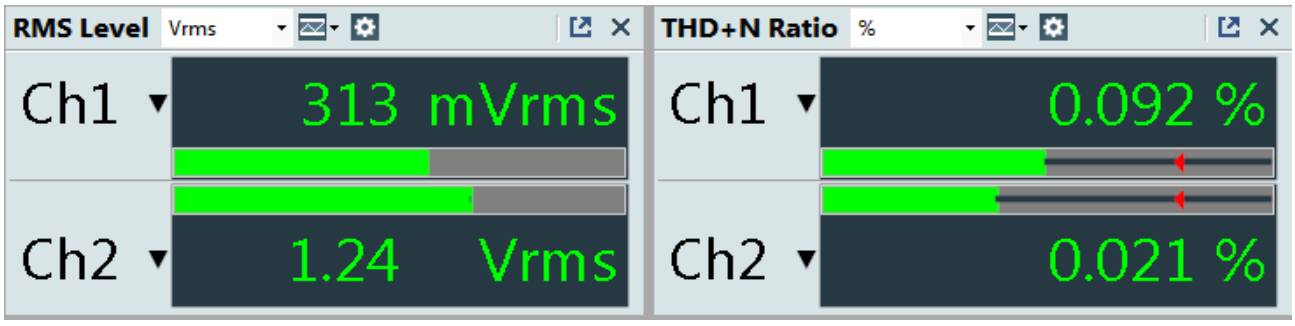
Undocking, Docking and Resizing

Click the **Undock** button to move a monitor out of the Bench Mode workspace. An undocked monitor can be resized by grabbing any of the meter edges or corners with the mouse cursor and dragging. Click the **Dock** button to return a monitor to the workspace. When undocked or opened these panels remain visible and available, even when other Bench Mode Tools, such as Sweep or Recorder are selected.

Meters

The meters are unique to Bench Mode. Each meter displays current readings as a numerical value and as a bar graph.

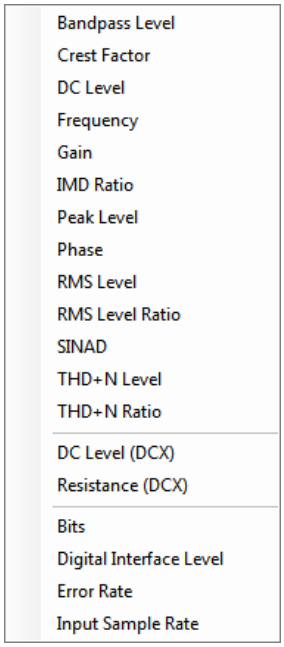
In a New Project, there are three meters shown:



- RMS Level
- Frequency
- THD+N Ratio

Adding a Meter

Click the Add Meter button and select from the menu. You can add as many meters as you want, and you can have more than one meter that displays the same parameter. Too many active meters, however, will slow system performance.



Available choices will vary with analyzer model, hardware and Signal Path Setup configuration. See detailed descriptions of Bench Mode meters on page 585.

Save Meter Data

Click Save Meter Data to open a dialog to save the current data to a csv file. Enter (or browse to) the path and filename for the csv file. Click **OK**.

If the **Append** checkbox is checked, the latest data will be added to the end of the data in the csv file. If **Append** is unchecked, previously saved data (if any) is deleted and replaced with the data in the current save operation.

The data values are unsettled, and are taken from the readings at the moment the **OK** button in the **Save Meter Data** dialog is clicked.

Regulate

The **Regulate** feature allows you to perform a regulation operation to find the parameter setting value that results in a specific reading value in a second parameter. You may, for example, want to find the generator level that produces a certain amount of distortion in the DUT. Click the **Regulate** button Regulate to open the **Regulation Settings** dialog. Read more about Bench Mode Regulation on page 586.

Arranging meters in the workspace

You can re-arrange docked meters by grabbing the meter title bar with the mouse cursor and dragging the meter to a new position.

Channels

Each meter displays 2 channels. For configurations with more than 2 input channels, you can add meters and specify additional channels.

Limits

Limits can be set for either or both channels of any meter.

Editing Limits

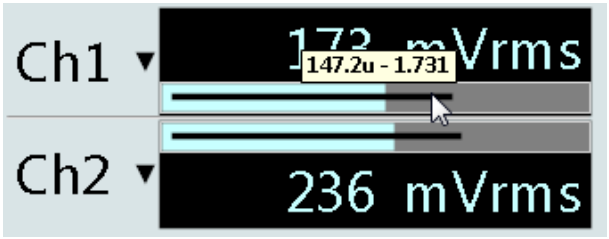
In a Bench Mode meter, select **Edit Limits** from the **Limits** menu, or right-click on the meter and select the same from the context menu.

Clear Limits

To remove all limit points, select **Clear Limits** from the **Limits** menu or from the right-click context menu.

Min/Max indicator

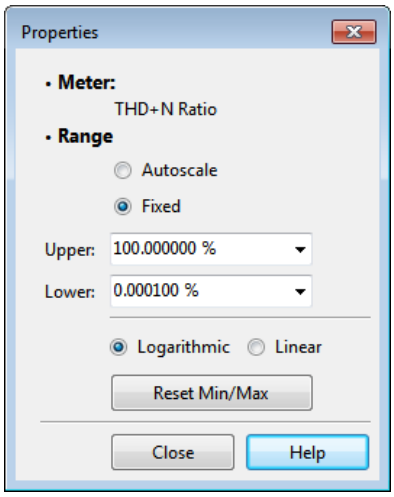
The bar graph display shows the current reading as a green bar. The excursion range of the bar graph is shown as a narrow black bar overlaying the green reading bar.



When the cursor is moved over the Min/Max indicator, a Tool Tip shows the val0ue of the minimum and maximum excursions.

The Min/Max indicator can be clear by clicking **Reset MIn/Max** in the Properties dialog.

Properties



Click the Edit Properties button to open the Properties dialog.

Meter

The meter type is displayed here

Range

Autoscale

Autoscale sets the meter range around the current signal range.

Fixed

Fixed allows you to specify a fixed graph range by **Upper** and **Lower** range values.

Upper

Set the meter range upper value here.

Lower

Set the meter range lower value here.

Logarithmic / Linear

When **Logarithmic** is selected, the meter scale follows a logarithmic rule. When **Linear** is selected, the meter scale follows a linear rule. The logarithmic scale is not available for all units.

Reset Min/Max

Click this button to reset the Meter Min/Max display.

Bench Mode Meter Descriptions

Bandpass Level

The Bandpass Level meter differs from the RMS Level meter only in that it follows a narrow bandpass filter. See page 580 for information about setting the Bench Mode Analyzer Bandpass Filter.

Crest Factor

This meter calculates and displays the Crest Factor for the selected channels. Crest factor is the peak amplitude of the waveform divided by the RMS value of the waveform, and is an indicator of the peakiness of a waveform.

DC Level

This meter reads the DC Level measured at the selected channels.

Frequency

This meter reads the frequency of the dominant tone in the signal at the selected channels.

Gain

This meter calculates and displays the gain in the selected channels. Gain is the ratio of the APx generator level to the level at the analyzer inputs.

IMD Ratio

This meter reads intermodulation distortion ratio for the selected channels. The intermodulation type (SMPTE/DIN, DFD, CCIF or DIM) as selected by the IMD Type selector in the Bench Mode Analyzer panel. See page 580 for information about setting the Bench Mode Analyzer IMD Type filter.

Note: The Bench Mode IMD meter is disabled when the system bandwidth is set higher than 90 kHz. The IMD stimulus signals and distortion products all fall well within the 90 kHz band, and overall system performance is improved with this meter disabled.

Peak Level

This meter reads the AC peak level for the selected channels. For a sine wave, the peak level is one-half the peak-to-peak level, and 1.41 times the RMS level.

Phase

This meter reads the interchannel phase between the selected channels and the Reference Channel. See page 580 for information about setting the Bench Mode Analyzer Reference Channel.

RMS Level

This meter displays the RMS Level read on the selected input channels. Read more about RMS Level beginning on page 689.

RMS Level Ratio

This meter displays the ratio of the RMS Level read on the selected input channels to the RMS Level on the selected Reference Channel. See page 580 for infor-

mation about setting the Bench Mode Analyzer Reference Channel.

SINAD

SINAD is a distortion ratio value, reciprocal of THD+N. Read more about SINAD beginning on page 498.

THD+N Level

THD+N Level displays the absolute RMS level of the THD+N residuals (the total signal with the fundamental removed) read on the selected input channels. Read more about THD+N beginning on page 527.

THD+N Ratio

THD+N Ratio displays the ratio of the THD+N residuals (the RMS value of the total signal with the fundamental removed) read on the selected input channels, to the RMS value of the total signal. Read more about THD+N beginning on page 527.

DC Level (DCX)

DC Level (DCX) displays the DC level measured by an optional Audio Precision DCX-127 Multifunction Module, if connected. See page 12 for information about the DCX-127.

Resistance (DCX)

DC Resistance (DCX) displays the ohmic resistance measured by an optional Audio Precision DCX-127 Multifunction Module, if connected. See page 12 for information about the DCX-127.

Bits

The Bits Meter is only available when the input is digital. The Bits Meter shows a bit-by-bit view of the embedded data in the digital signal audio word for each subframe. The two subframes correspond to Channel 1 and Channel 2.

In the bar display, the bits are numbered from the most significant bit (MSB, or bit 31 of the word) on the left, to the least significant bit (LSB, or bit 0 of the word) on the right. The Bits meter examines the signal in repeated periods of approximately 1/4 second.

When Active Bits is selected, the bar display indicates green for a bit that has changed state during the measurement period, and black for a bit that has not. The hex display shows the hexadecimal value of the difference between the changed samples.

When Data Bits is selected, the indicators display green for a bit that is at data 1 at the moment of measurement, and black for a bit that is at data 0. The hex display shows the hexadecimal value of the measured sample.

Digital Interface Level

The Digital Interface Level meter is only available when the Advanced Digital I/O module is installed, and the input is Digital Balanced or Digital Unbalanced. This meter reads the peak-to-peak level of the electrical digital interface signal (carrier).

This signal is nominally between 2 Vpp and 7 Vpp for the balanced connection when terminated in 110 ohms; it is nominally 1 Vpp for the professional unbalanced connection when terminated in 75 ohms; it is nominally 0.5 Vpp for the consumer unbalanced connection when terminated in 75 ohms.

Error Rate

This meter reads the rate of bit errors in the embedded audio of a digital signal. This meter requires a stimulus signal that can be evaluated for bit-accuracy, which includes Bit Test, Constant Value, Walking Ones and Walking Zeros.

Input Sample Rate

This meter reads the input sample rate for digital inputs.

Regulation in Bench Mode

Bench Mode offers a regulation feature in the Monitors/Meters panel, and as an option in the Sweep measurement. With the exception of the **Start** button, the dialogs are the same; both will be discussed in this topic. The dialogs are also similar to the Auto Gen Level regulation dialog (page 58), and to the Acoustic Output Level regulation dialog (page 62) in Signal Path Setup.

Monitors/Meters > Regulate

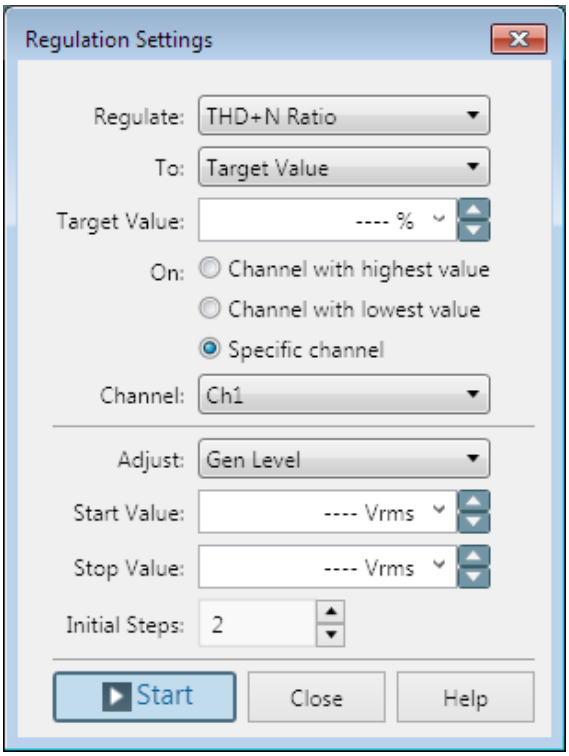
Regulation in the Bench Mode Monitors/Meters panel allows you to perform a regulation operation to find a parameter setting value, that results in a specific reading value in a second parameter. You may, for example, want to find the generator level that produces a certain amount of distortion in the DUT.

Click **Regulate** to open the editing dialog.

Sweep Point Regulation

A regulation feature is also available for the Bench Mode Sweep. When the Sweep panel **Regulate each point** checkbox is checked, regulation is performed on every sweep point before it is displayed. Click the **Edit Regulation Settings** button to open the editing dialog.

Edit Regulation Settings Dialog



When accessed from Monitors/Meters, the Regulation Settings dialog includes a **Start** button. When accessed from the Sweep panel, regulation is automatically started at each sweep point.

In APx500 4.4, these dialogs have been refactored. The regulation functions remain essentially the same, but controls have been clarified and new controls added.

Regulate:

This control selects the parameter in the DUT output you want to regulate. For example, if you are searching for the generator level that results in a specific THD+N Ratio value (the Target, Min or Max Value), select THD+N Ratio here. Choose from

- **Bandpass Level**
- **Crest Factor**
- **DC Level**
- **DC Level (DCX)**
- **Frequency**
- **Gain**
- **IMD Ratio**
- **Input Sample Rate**
- **Peak Level**
- **Phase**
- **Resistance (DCX)**
- **RMS Level**
- **RMS Level Ratio**
- **SINAD**
- **THD+N Level**
- **THD+N Ratio**

To:

This control selects the value you want to attain in the regulated parameter. Choose

Target Value

Find a specific Target Value, set in the following field.

- **Maximum Value**
  Find the **Maximum Value**
- **Minimum Value**
  Find the **Minimum Value**
- **Target Value:**
  If **Target Value** is selected in the previous control, enter the value you want to attain here.

On:

This control selects the DUT channel to be regulated. Choose

- **Channel with the highest value**
  All available analyzer input channels are monitored, and the channel with the highest value of the regulation parameter chosen in **Regulate** (see above) is selected for regulation purposes.
- **Channel with the lowest value**
  All available analyzer input channels are monitored, and the channel with the lowest value of the regulation parameter chosen in **Regulate** (see above) is selected for regulation purposes.
- **Specific channel**
  Choose one of the available analyzer input channels.

Channel

If **On** is set to Specific Channel, select the input channel here.

Adjust:

The **Adjust** control selects the APx controlled parameter that will be adjusted to find the regulated DUT parameter. Choose from:

- **Gen Level**
- **Gen Offset**
- **Jitter Frequency**
- **Jitter Level (UI)**
- **Jitter Level (sec)**
- **DCX DCV1 Level**
- **DCX DCV2 Level**

Start Value:

The **Start Value** is the first parameter setting where a measurement is made. If it is lower than the **Stop Value**, then it is also the lowest parameter setting where a measurement is made. If it is higher than the **Stop Value**, then it is the highest parameter setting where a measurement is made.

The regulation search proceeds from Start Value to Stop Value; your choices for these values determine the direction (up or down) of the search.

Stop Value:

If the **Stop Value** is higher than the **Start Value**, then the **Stop Value** is the highest parameter setting where a measurement is made. If the **Stop Value** is lower than the **Start Value**, then the **Stop Value** is the lowest parameter setting where a measurement is made. The **Stop Value** is also the last parameter setting where a measurement is made in the initial search for an interval containing a measured value of the type that is sought.

For voltages, the higher of Start Value or Stop Value also serves to limit the maximum APx voltage, protecting your DUT.

Initial Steps:

Initial Steps sets the number of intervals dividing the parameter level range between **Start Value** and **Stop Value**. The default setting is **2**. Depending upon the parameter to be regulated and characteristics of the DUT, greater values may provide better performance.

Frequency:

The **Frequency** control sets the APx generator frequency for regulation.

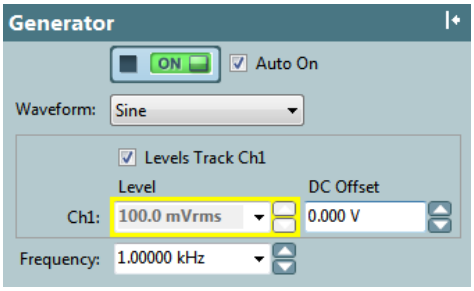
Read more about regulation on page 481 for a detailed discussion of these methods.

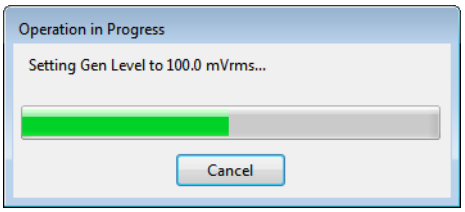
Note: Distortion measurements such as THD+N benefit from filtering. When regulating THD+N, consider using the Input Filters in Signal Path Setup > Input/Output to filter the signal used for regulation. Suggested filters are High-pass: 20 Hz, Low-pass: 22 kHz. Remember to reset the filters when you continue to other measurements. See Bandwidth Limiting and Weighting Filters for more information.

See Measurement Filters on page 619 for detailed information about the filters available locally for some measurements.

Progress view

During regulation, the adjusted parameter setting is highlighted on-screen, and a progress dialog or progress bar comments on the settings as they are varied.





When regulation is successful in a Monitors/Meters operation, the desired value for the regulated parameter is attained, and the adjusted parameter setting retains the adjusted value. When regulation is successful in a sweep point operation, the point is plotted and the sweep moves to the next point.

When regulation is unsuccessful, an error message is displayed.

Sweep
(Bench Mode)

For the APx515, Bench Mode requires a software option key. See page 189 for more information about software options.

Read about the Bench Mode workspace, Generator and Analyzer panels in Chapter 88.

Overview

Bench Mode Sweep is a general-purpose stepped-sweep engine that sweeps a primary source parameter (on the X axis) in a number of steps, providing a graph of the response of one or two result parameters (on the Y axes). Typical sweeps are a frequency response sweep, where the generator frequency is swept across the audio band while the DUT's output level is graphed, or a level response sweep, where the generator level is swept across the DUT's range, while the DUT THD+N is graphed.

Sweep in Bench Mode can be considered a combination of the Stepped Frequency Sweep and Stepped Level Sweep as presented in Sequence Mode, with some Bench Mode changes and additions. You can read more about these measurements in Chapters 81 and 82.

Generator and Analyzer settings

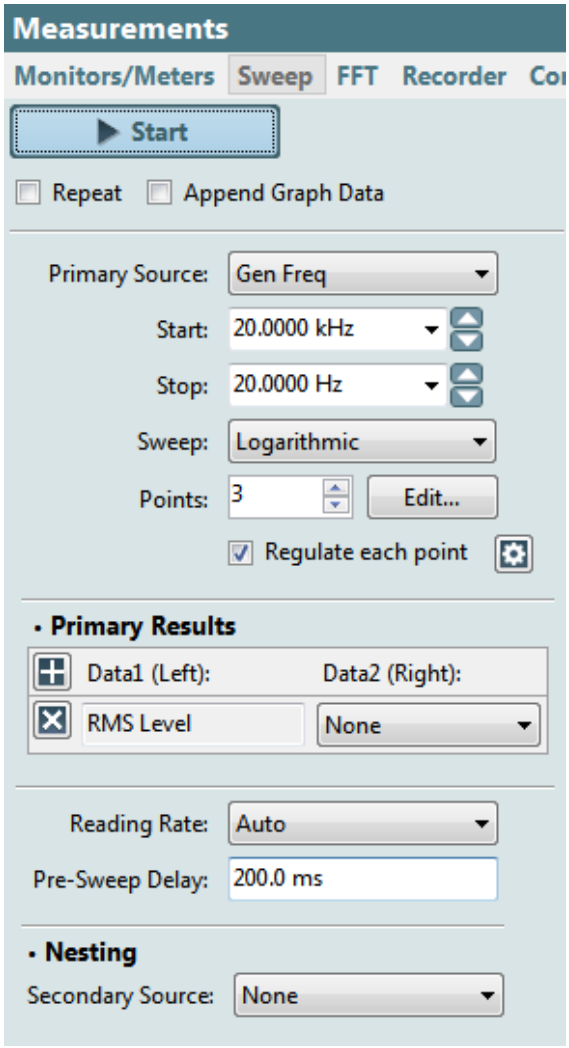
In Bench Mode, these settings are made in the Generator and Analyzer panels. See pages 578 and 579.

Nested Sweeps

A Bench Mode Sweep can be run iteratively at different steps of an additional parameter (the secondary source) in what is referred to as a nested sweep. See Nesting, below on page 591.

Operation

If you have not yet set up your test, first go to Signal Path Setup to select and configure inputs and outputs, references and other settings. Read about the Signal Path Setup menu in Chapter 7.



Start

To make a sweep, click **Start**. The Sweep will start at the first point and continue until the results for the last point are plotted.

For File Input configuration, the Start button becomes the Analyze button.

Repeat

When the **Repeat** checkbox is checked, the sweep (or nested sweep) repeats continually. Unchecking the **Repeat** checkbox or clicking **Stop** will cancel **Repeat**.

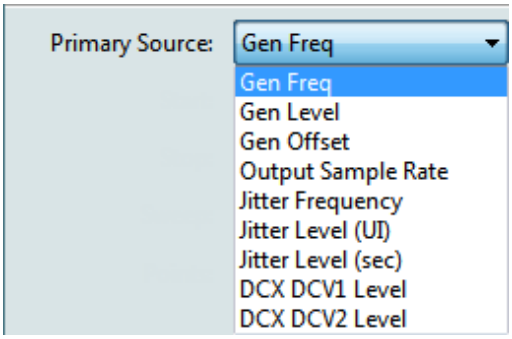
Append Graph Data

Measured data are grouped in a **Data Set**, which contains all acquisition results. Normally, the current graph data are deleted each time you start a new measurement. If the **Append Graph Data** box is checked, appended data are grouped as additional data sets, as are any imported data. The Data Sets Panel displays the Data Set(s) in the measurement, and provides a visibility checkbox, a Time acquisition field, and a Notes field for each data set.

Read more about Data Sets beginning on page 649.

Primary Source

Select the primary sweep source here. This is the parameter that will be swept and plotted on the X-axis. Available choices are dependent upon instrument hardware, options, Input/Output settings and Bench Mode Generator settings. Choices may include:



Output Sample Rate is only available when Output Configuration > Connector is set to a digital output.

If Output Configuration > Connector is set to None (External), or Input Configuration > Connector is set to a File input, you must use a Sweep Table. Read about using Sweep Tables beginning on page 178.

Start

Set the sweep start value here.

Stop

Set the sweep stop value here.

Sweep

Choose

- **Logarithmic** point spacing
- **Linear** point spacing
- **Custom**
  Edit the **Sweep Points** to create a **Custom** sweep.

Some Primary Sources allow only Linear spacing.

Points

Set the number of sweep points here.

Edit

Open the **Sweep Points** dialog to edit, import or export the **Sweep Points** table.

Step Size

For **Linear** sweeps, enter the **Step Size** here.

Regulate each point

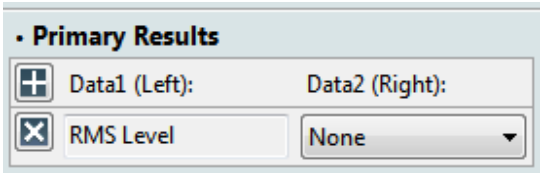
Check this box to regulate each sweep point according to the settings in the Regulation Settings dialog, discussed below on page 592. Click the **Edit...** button to open the dialog.

Primary Results

In Bench Mode Sweep, Primary results consist of a Bench Mode meter reading (the Data) on the left or right Y-axis, and a Bench Mode sweep Primary Source on the X-Axis. See Bench Mode Meter descriptions in Chapter 89.

There is an exception to this rule: when the Primary Source is set to Generator Level, four additional results are available, all providing the option of plotting the measured level (rather than the generator level) on the X-axis.

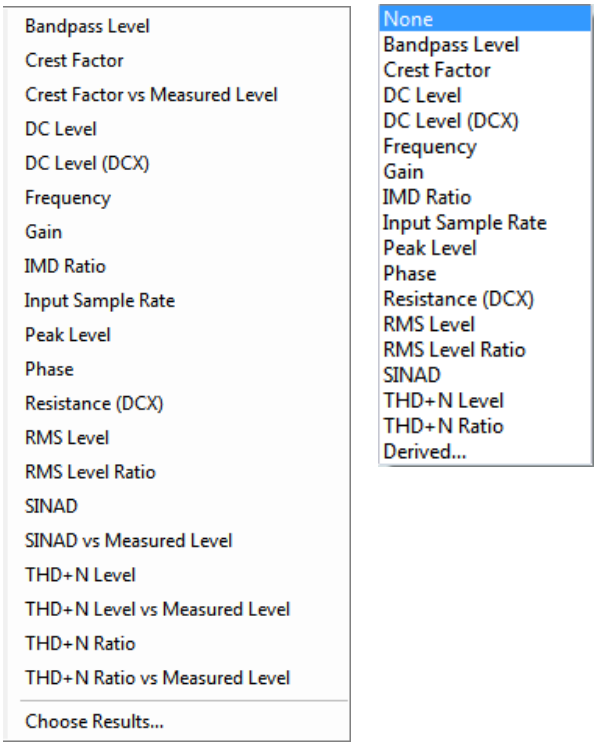
Initially, only one result (RMS Level) is set, plotted as Data1 on the left Y axis of the graph.



To add more primary results to the left axis, click the Add Data1 (Left) button and select one or more results.

To add a primary or derived result to the right axis, open the Data2 (Right) menu for a given existing Data1 result, and select a result.

Available choices will vary with analyzer model, hardware and Signal Path Setup configuration. Choices may include:



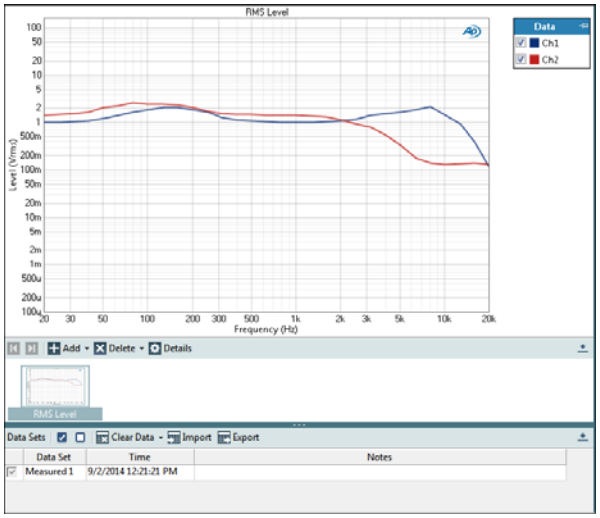
Data 1 Menu Data 2 Menu

The "vs Measured Level" results are only available when the Primary Source is Generator Level.

Remove a row by clicking the **Delete** button. Both left and right axis results are removed. Results can also be added or deleted from the Selector filmstrip.

Note that in the Data 1 menu, you can select **Choose Results...** to open the **Add Primary Results** dialog, where you can add multiple results. Also, in the **Data 2** menu, you can select **Derived...** to add **Derived Results**. See Chapter 100 for more information about Derived Results.

Typical Result Display



Settling

If Settling is enabled in the **Settling** dialog (**Analyzer > Settled Readings > Settings...**), the data for each point in the sweep is settled before plotting. Read more about Settling in Bench Mode beginning on page 579.

Reading Rate

Sweep uses a software meter to (RMS Level, for example) to read the value of the input signal for each selected result at each sweep step. **Reading Rate** sets a maximum number of times the input signal will be read per second. The default for Sweep is **Auto**.

In most cases the number of readings will be very close to the specified value. However, the rate is dependent upon PC performance and channel count, and may be lower. Read more about Reading Rate beginning on page 183.

Pre-Sweep Delay

If the generator is off when the Sweep started, you may want to delay the beginning of the sweep to allow any transients to dissipate, or to allow the DUT to warm to some nominal operating temperature. Enter a delay time between 0 seconds and 5 seconds in the **Pre-Sweep Delay** field.

Nesting

Bench Mode sweeps can be nested. See Nested Sweeps on page 185.

Secondary Source

Select the secondary sweep source here. This is the outside, nesting parameter that is varied to change the conditions for the primary sweep through a number of iterations. Available choices are dependent upon instrument hardware, options, Input/Output settings and Bench Mode Generator and other settings.

Sweep Point Regulation

A regulation feature is available for the Bench Mode Sweep, and also for the Bench Mode Meters/Monitors panel. With the exception of the **Start** button, the dialogs are the same; both are discussed in the topic on page 586.

FFT
(Bench Mode)

For the APx515, Bench Mode requires a software option key. See page 189 for more information about software options.

Read about the Bench Mode workspace, Generator and Analyzer panels in Chapter 88.

Overview

Bench Mode FFT is a general purpose diagnostic tool that uses a high-resolution FFT technique to provide several different display results. Read More About FFTs beginning on page 492.

FFT in Bench Mode is essentially the same as the Signal Analyzer measurement that is presented in Sequence Mode, with some Bench Mode additions. Read more about the Signal Analyzer in Chapter 78.

Generator and Analyzer settings

In Bench Mode, these settings are made in the Generator and Analyzer panels. See pages 578 and 579.

Nested Sweeps

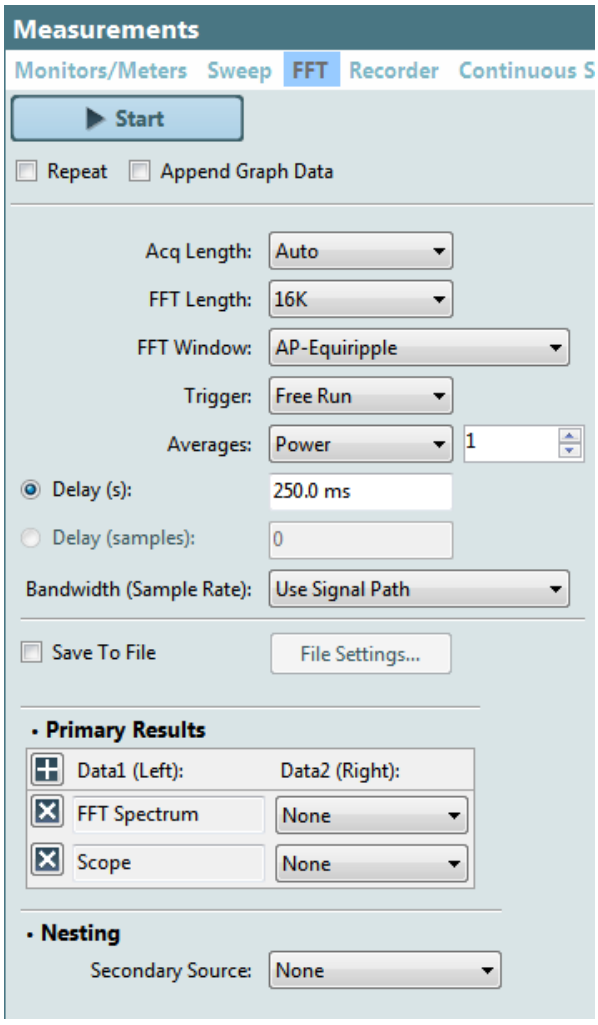
A Bench Mode FFT can be run iteratively at different steps of a fourth parameter (the secondary source) in what is referred to as a nested sweep. See Nesting, below on page 596.

Operation

If you have not yet set up your test, first go to Signal Path Setup to select and configure inputs and outputs, references and other settings. Read about the Signal Path Setup menu in Chapter 7.

Start

To run FFT, click **Start**. The signal selected in the Generator (if any) will be applied to the DUT, and an FFT of the DUT's output will be graphed here.



Repeat

When the **Repeat** checkbox is checked, the FFT acquisition (or nested acquisitions) repeats continually. Unchecking the **Repeat** checkbox or clicking **Stop** will cancel **Repeat**.

Append Graph Data

Measured data are grouped in a **Data Set**, which contains all acquisition results. Normally, the current graph data are deleted each time you start a new measurement. If the **Append Graph Data** box is checked, appended data are grouped as additional data sets, as are any imported data. The Data Sets Panel displays the Data Set(s) in the measurement, and provides a visibility checkbox, a Time acquisition field, and a Notes field for each data set.

Read more about Data Sets beginning on page 649.

Acq. Length

Set the acquisition length here, in seconds or in samples. Auto, the default, sets the acquisition length to match the FFT length.

FFT Length

Set the FFT record length here. Options from 256 points to 1.2M points are available, depending upon the number of channels in use; default is 16K points.

FFT Window

FFT acquisitions must either be synchronous or have one of a number of amplitude windows applied to provide useful data for interpretation. Each window function brings advantages and disadvantages. The default selection, AP-Equiripple, is a proprietary Audio Precision FFT window that is an excellent choice for most FFT measurements.

Read about FFT windows beginning on page 493.

Trigger

- **Free Run**
  When trigger is set to **Free Run**, the acquisition begins when the **Start** button is clicked, plus any time or sample count set in **Delay**.
- **+0 Crossing**
  When trigger is set to **+0 Crossing**, the acquisition begins at the input waveform's first positive-going zero crossing after any time or sample count set in **Delay**, for the input channel selected.
- **Generator**
  - When **Trigger** is set to **Generator** and **Generator Waveform** is set to **Sine (High Performance Sine Generator)**, the acquisition begins at the waveform's first positive peak plus any time or sample count set in **Delay**. For the High Performance Sine Generator, if the **Delay** time is set in samples it is scaled to the input sample rate.
  - When **Trigger** is set to **Generator** and **Generator Waveform** is set to **Sine (DAC)**, the acquisition begins at the waveform's first positive-going zero crossing plus any time or sample count set in **Delay**.
  - When **Trigger** is set to **Generator** and **Generator Waveform** is set to **Square**, the acquisition begins at the waveform's first rising edge plus any time or sample count set in **Delay**.
  - When **Trigger** is set to **Generator** and **Generator Waveform** is set to **Noise**, the acquisition will not trigger.
  - When **Trigger** is set to **Generator** and **Generator Waveform** is set to **File**, the acquisition begins at the file's first sample, plus any time or sample count set in **Delay**. A trigger is generated each time the file loops back to its first sample.
- **External**
  External is only available for analyzers equipped with the Advanced Master Clock. The acquisition begins at the first trigger pulse received at the rear-panel **Trigger In** BNC connector plus any time or sample count set in **Delay**. Set the **Trigger Input** in **Signal Path Setup > Triggers** to the desired voltage threshold and pulse edge.

Averages

For non-periodic waveforms such as noise, averaging multiple acquisitions can provide a more useful view. By setting **Averages** to more than 1, you can specify how many acquisitions to average. Default is 1; maximum is 1000.

- **Power**
  The scope result does not use power averaging. Power averaging affects only the frequency domain results.
- **Synchronous**
  When **Trigger** is set to **Generator**, **Synchronous Averaging** is an option. Synchronous averaging affects both the time domain and the frequency domain results.

Read **More About Averaging** beginning on page 494.

Max Hold and Min Hold are not averaging functions, but use the multiple acquisitions available to the Averages feature to find maximum or minimum values for each FFT bin.

- **Max Hold**
  Max Hold displays the highest value for each bin found in all the acquisitions set in **Averages**. As the acquisitions progress, the value in each bin can only go up.
  The scope result does not use **Max Hold**. This selection only affects the frequency domain results.

- **Min Hold**
  Min Hold displays the lowest value for each bin found in all the acquisitions set in **Averages**. As the acquisitions progress, the value in each bin can only go down.
  The scope result does not use **Min Hold**. This selection only affects the frequency domain results.

Delay (s) / Delay Samples

The **Delay (s)/Delay Samples** setting allows you to delay the acquisition by a specific period of time or a specific number of samples relative to the measurement **Start** or a trigger event, as described in **Trigger**.

This is sometimes necessary to avoid analysis of spurious signals created when using a coded generator file with a DUT that requires some time to match the coded format, or to allow the filter to settle when AC coupled.

For **Delay (s)**, enter the delay in seconds. Minimum time is 0.0 s; maximum is 60.0 s; the default is 250 ms.

If **Trigger** is set to **Generator**, you can choose to enter the delay in **Samples** rather than seconds. Minimum sample count is 0; maximum is 12,960,000.

Signal

The APx515 does not support the PDM I/O module.

The **Signal** field is only available when **Input Connector** in **Signal Path Setup** is set to **PDM**.

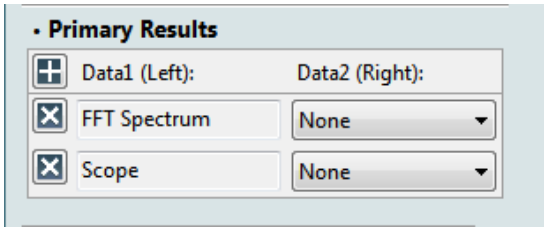
- **Decimated Audio**
  The decimated PDM audio (at the baseband sample rate and bandwidth) is the input signal.
- **PDM Bitstream**
  The PDM bitstream (at the PDM bit clock rate and bandwidth) is the input signal. This is useful to view the out-of-band noise and other components of the PDM bitstream.

Note: When the PDM input decimation is set to x1, (see Settings for PDM Input on page 159) the PDM signal is not decimated. In that case, either of the Signal selections in the Signal Analyzer will present the undecimated PDM bitstream for the analyzer acquisition.

Primary Results

In Bench Mode FFT, Primary results consist one of four FFT results on the left or right Y-axis. The X-axis is scaled in the result domain (either frequency domain or time domain).

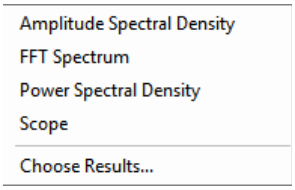
Initially, only two results are set, plotted as Data1 on the left Y axis of the graph.



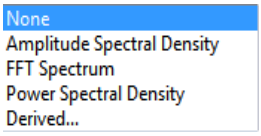
To add more primary results to the left axis, click the Add Data1 (Left) button and select one or more results.

To add a primary or derived result to the right axis, open the Data2 (Right) menu for a given existing Data1 result, and select a result.

Choices include:



Data 1 Menu

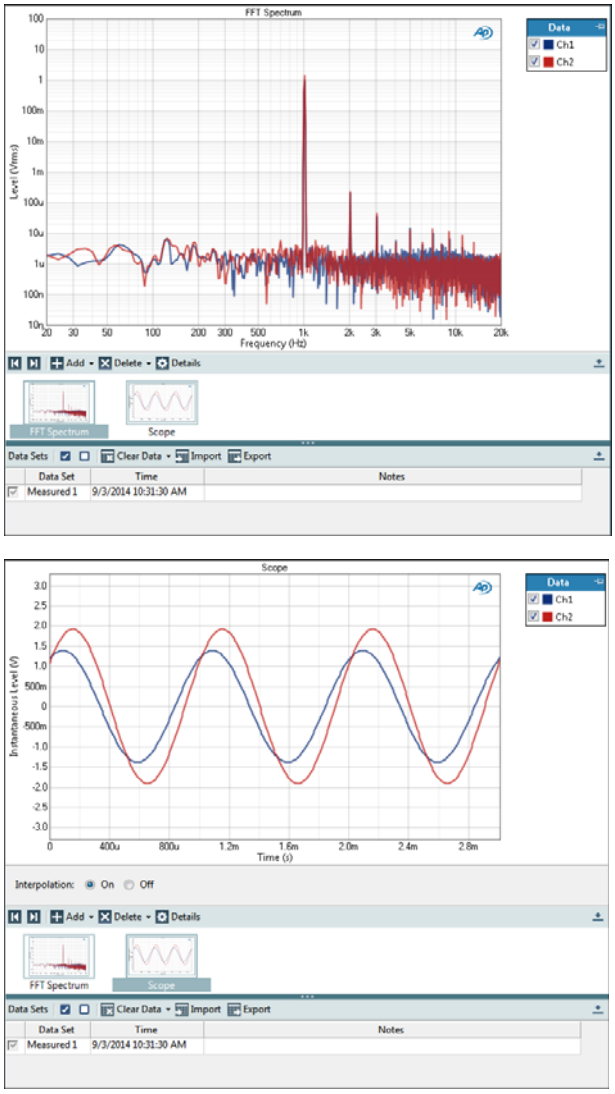


Data 2 Menu

Remove a row by clicking the **Delete** button. Both left and right axis results are removed. Results can also be added or deleted from the Selector filmstrip.

Note that in the Data 1 menu, you can select **Choose Results...** to open the **Add Primary Results** dialog, where you can add multiple results. Also, in the **Data 2** menu, you can select **Derived...** to add **Derived Results**. See Chapter 100 for more information about Derived Results.

Typical Result Displays



Spectrum and scope result displays.

Settling

FFT results are always unsettled.

Save To File

You can record the audio signal to a computer file as the signal is being acquired. Click **Save to File** before you run the measurement recorder. See **Recording Audio to a File** on page 381 for detailed information.

File Settings

When saving a audio signal to a file, click **File Settings** to specify the file name and location.

Nesting

Bench Mode FFTs can be nested. See Nested Sweeps on page 185.

Secondary Source

Select the secondary sweep source here. This is the outside, nesting parameter that is varied to change the conditions for the recording through a number of iterations. Available choices are dependent upon instrument hardware, options, Input/Output settings and Bench Mode Generator and other settings.

Recorder
(Bench Mode)

For the APx515, Bench Mode requires a software option key. See page 189 for more information about software options.

Read about the Bench Mode workspace, Generator and Analyzer panels in Chapter 88.

Overview

Bench Mode Recorder is a general-purpose diagnostic tool that provides a record of a number of measurements versus elapsed time. The record can be very long (up to one week), providing a means of monitoring the output of a device under test over an extended period of time. The Recorder does not require a specific test signal. It can be used with any audio signal within the input range of the analyzer, or with no signal.

The Recorder also provides a means of recording the audio acquisition to a file. See **Save to File**, below on page 599.

Recorder in Bench Mode is similar to Measurement Recorder presented in Sequence Mode, with some Bench Mode changes and additions. You can read about the Measurement Recorder in Chapter 61.

Generator and Analyzer settings

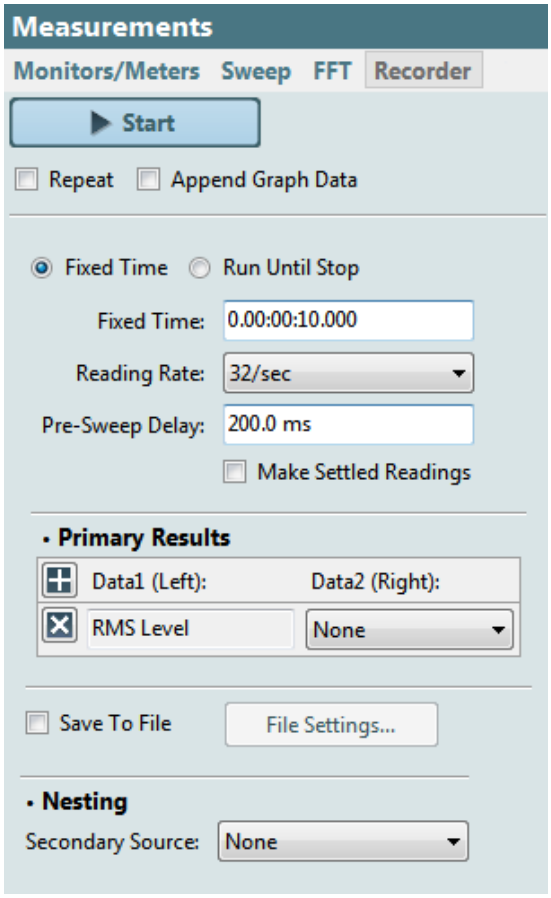
In Bench Mode, these settings are made in the Generator and Analyzer panels. See pages 578 and 579.

Nested Sweeps

A Bench Mode Sweep can be run iteratively at different steps of an additional parameter (the secondary source) in what is referred to as a nested sweep. See Nesting, below on page 599.

Operation

If you have not yet set up your test, first go to Signal Path Setup to select and configure inputs and outputs, references and other settings. Read about the Signal Path Setup menu in Chapter 7.



Start

To make a measurement recording, click **Start**. The measurement recording will begin and the graph will display as the record is acquired. The record will continue until the end time set in **Fixed Time**, or until **Stop** is pushed.

Note: For digital inputs, sample rate must be stable before the Recorder begins. When

playing a digital external source (such as a DVD), we advise that you start the playback device first, then click the Recorder Start button.

Repeat

When the **Repeat** checkbox is checked, the recording (or nested sweep) repeats continually. Unchecking the **Repeat** checkbox or clicking **Stop** will cancel **Repeat**.

Note: if Append Graph Data is checked while Repeat is set, a great deal of data will be quickly amassed and the computer memory or storage may be overwhelmed.

Append Graph Data

Measured data are grouped in a **Data Set**, which contains all acquisition results. Normally, the current graph data are deleted each time you start a new measurement. If the **Append Graph Data** box is checked, appended data are grouped as additional data sets, as are any imported data. The Data Sets Panel displays the Data Set(s) in the measurement, and provides a visibility checkbox, a Time acquisition field, and a Notes field for each data set.

Read more about Data Sets beginning on page 649.

Primary Source

For the Bench Mode Recorder, the Primary Source plotted on the graph X-axis is always **Time**, in seconds (s).

Fixed Time

Fixed Time sets the length of the measurement record. Minimum time is 0.1 s, maximum time is 7 days, or 12 million points at the selected reading rate, whichever is less.

Enter the desired time in the **Fixed Time** field following this pattern:

d:hh:mm:ss.s

where d = days, hh = hours, mm = minutes, ss = seconds. Days and fractional seconds are optional.

Run Until Stop

This selection will record until 12 million points are acquired, unless **Stop** is clicked.

Reading Rate

Recorder uses a software meter to (RMS Level, for example) to read the value of the input signal for each selected result at each sweep step. **Reading Rate** sets a maximum number of times the input signal will be read per second. The default for Recorder is **32/sec**.

In most cases the number of readings will be very close to the specified value. However, the rate is dependent upon PC performance and channel count, and may be lower. Read more about Reading Rate beginning on page 183.

Pre-Sweep Delay

If the generator is off when the Recorder is started, you may want to delay the beginning of the recording to allow any transients to dissipate, or to allow the DUT to warm to some nominal operating temperature. Enter a delay time between 0 seconds and 5 seconds in the Pre-Sweep Delay field.

Make Settled Readings

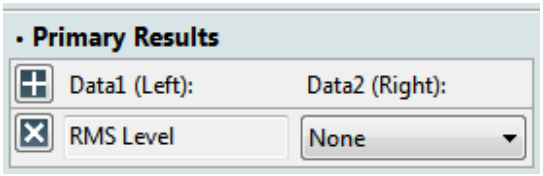
Settling parameters are set in the **Settling** dialog available in the Bench Mode **Analyzer** panel. You can engage or disengage settling for the Recorder here. When the **Make Settled Readings** checkbox is checked, every Recorder reading is settled, which can slow the result display, depending upon Settling settings and the nature of the incoming signal. Read about Settling in Bench Mode beginning on page 579.

When the **Make Settled Readings** checkbox is unchecked, the raw, unsettled data is plotted.

Primary Results

In Bench Mode Recorder, Primary results consist of a Bench Mode meter reading (the Data) on the left or right Y-axis, and Time on the X-Axis. See Bench Mode Meter descriptions in Chapter 89.

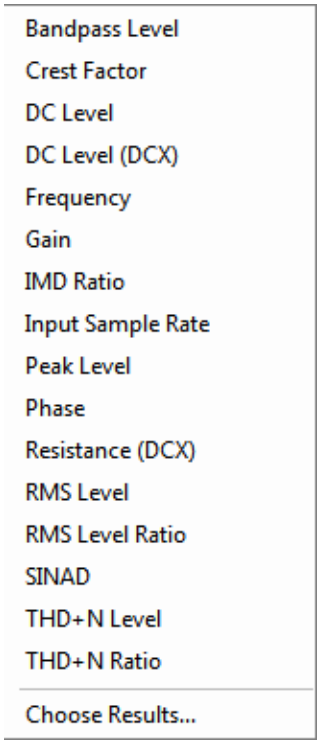
Initially, only one result (RMS Level) is set, plotted as Data1 on the left Y axis of the graph.



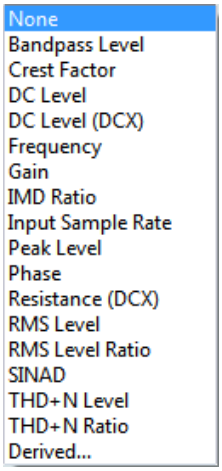
To add more primary results to the left axis, click the Add Data1 (Left) button and select one or more results.

To add a primary or derived result to the right axis, open the Data2 (Right) menu for a given existing Data1 result, and select a result.

Available choices will vary with analyzer model, hardware and Signal Path Setup configuration. Choices may include:



Data 1 Menu

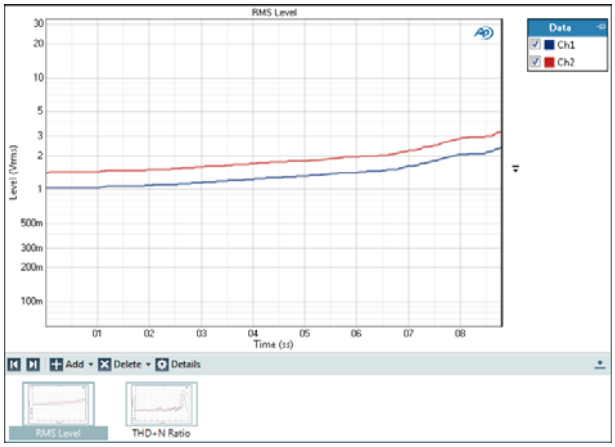


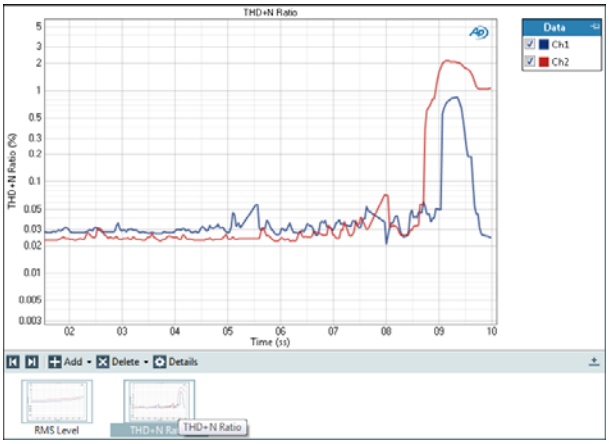
Data 2 Menu

Remove a row by clicking the **Delete** button. Both left and right axis results are removed. Results can also be added or deleted from the Selector filmstrip.

Note that in the Data 1 menu, you can select **Choose Results...** to open the **Add Primary Results** dialog, where you can add multiple results. Also, in the **Data 2** menu, you can select **Derived...** to add **Derived Results**. See Chapter 100 for more information about Derived Results.

Typical Result Displays





These results show a 10 second recording of Level and THD+N results, while the DUT gain is increased.

Save To File

You can record the audio signal to a computer file as the signal is being acquired. Click **Save to File** before you run the measurement recorder. See **Recording Audio to a File** on page 381 for detailed information.

File Settings

When saving a audio signal to a file, click **File Settings** to specify the file name and location.

Nesting

Bench Mode Recorder **Fixed Time** records can be nested. **Run Until Stop** records cannot be nested. See Nested Sweeps on page 185.

Secondary Source

Select the secondary sweep source here. This is the outside, nesting parameter that is varied to change the conditions for the recording through a number of iterations. Available choices are dependent upon instrument hardware, options, Input/Output settings and Bench Mode Generator and other settings.

Continuous Sweep
(Bench Mode)

For the APx515, Bench Mode requires a software option key. See page 189 for more information about software options.

Read about the Bench Mode workspace, Generator and Analyzer panels in Chapter 88.

Overview

Bench Mode Continuous Sweep is a brief log-swept sine wave (a Farina log chirp) that moves continuously across a specified range of frequencies. The DUT output is acquired by the analyzer and is mathematically processed to provide a number of results. For more information about continuous sweep measurements, go to page 246.

Continuous sweep based measurements are not supported in External Source or File Input configurations.

Continuous Sweep in Bench Mode is essentially the same measurement that is presented in Sequence Mode, with some Bench Mode additions. You can read about the Continuous Sweep measurement in Chapter 31.

Generator and Analyzer settings

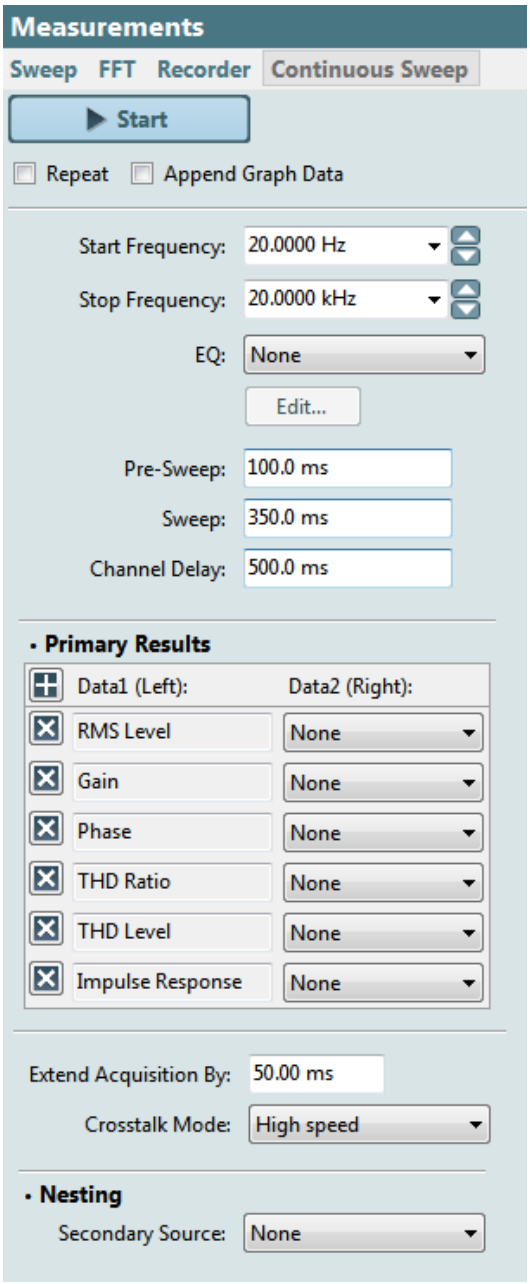
In Bench Mode, these settings are made in the Generator and Analyzer panels. See pages 578 and 579.

Nested Sweeps

A Bench Mode Continuous Sweep can be run iteratively at different steps of an additional parameter (the secondary source) in what is referred to as a nested sweep. See Nesting, below on page 603.

Operation

If you have not yet set up your test, first go to Signal Path Setup to select and configure inputs and outputs, references and other settings. Read about the Signal Path Setup menu in Chapter 7.



Start

To make a Continuous Sweep, click **Start**. The generator will output a continuous sweep to the DUT on the selected generator channels at the level set in the Generator panel. In a short period of time, the signal will be acquired and processed, and each of the small graphs in the Selector will be populated with results. Simply click the result view you want.

Repeat

When the **Repeat** checkbox is checked, the sweep (or nested sweep) repeats continually. Unchecking the **Repeat** checkbox or clicking **Stop** will cancel **Repeat**.

Note: if Append Graph Data is checked while Repeat is set, a great deal of data will be quickly amassed and the computer memory or storage may be overwhelmed.

Append Graph Data

Measured data are grouped in a **Data Set**, which contains all acquisition results. Normally, the current graph data are deleted each time you start a new measurement. If the **Append Graph Data** box is checked, appended data are grouped as additional data sets, as are any imported data. The Data Sets Panel displays the Data Set(s) in the measurement, and provides a visibility checkbox, a Time acquisition field, and a Notes field for each data set.

Read more about Data Sets beginning on page 649.

Primary Source

The primary source for a continuous sweep measurement is the log-swept sine signal. The range between the **Start** and **Stop** frequencies provides the X-axis for most of the result graphs.

Start Frequency and Stop Frequency

These controls set the frequency range of the sweep.

EQ

You can equalize the generator signal to make a pre-emphasized stimulus. See Generator Equalization (page 192) for a general discussion of this feature.

Choosing generator equalization

Select **None, Relative** or **Absolute** from the EQ drop-down menu.

- **None** applies no equalization to the generator.
- **Relative** applies the gain or loss specified in the EQ table to the current generator level for each channel.
- **Absolute** sets the generator level for all channels to the level specified in the EQ table. Note: when **Absolute** is selected, current channel level settings are lost.

Edit

If you have chosen to equalize the generator signal, you can edit the EQ table here. See **Edit EQ Table** (page 192).

Pre-Sweep and Sweep duration fields

A continuous sweep stimulus consists of a sine wave swept logarithmically across the sweep range in the period of time set in these fields.

To enable the DUT to stabilize before the beginning of the defined sweep measurement, there is variable **Pre-Sweep** time. The actual sweep stimulus signal begins below the defined **Start Frequency** and is swept upward, reaching the **Start Frequency** at the end of the **Pre-Sweep** time. The sum of the times in **Pre-Sweep** and **Sweep** is the total sweep length.

- The default for **Pre-Sweep** is 100.0 ms. Minimum is 0 s; maximum is 1.0 s.
- The default for **Sweep** is 350.0 ms. Minimum is 50.0 ms; maximum is 5 s.

Longer **Pre-Sweep** settings allow more time for DUT stabilization, but increase measurement time. Longer **Sweep** settings provide greater resolution and signal-to-noise ratio, but increase measurement time.

Channel Delay

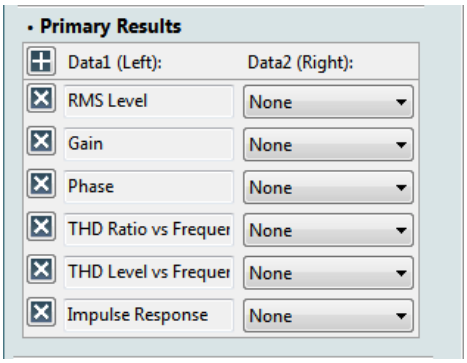
The **Channel Delay** control allows you to specify a delay before the **Start** moments for each channel when in one of the **High Accuracy Crosstalk** modes.

If the **Channel Delay** is zero or very short, the slew rate limiting of a DUT or the time constant in **AC Coupling** can cause a transient when energy from the sweep in a previous channel appears in the analysis of the current channel. The default **Channel Delay** is 500 ms. If DC coupled and the DUT does not have appreciable slew rate limiting, Channel Delay can be set to a very low value, or 0.

Primary Results

In Bench Mode Continuous Sweep, Primary results include 16 sets of data derived from the sweep.

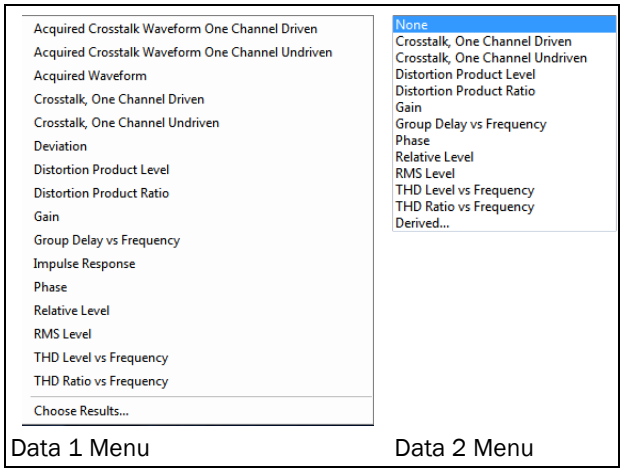
Initially, only six results are set, plotted as Data1 on the left Y axis of the graph.



To add more primary results to the left axis, click the Add Data1 (Left) button and select one or more results.

To add a primary or derived result to the right axis, open the Data2 (Right) menu for a given existing Data1 result, and select a result.

Available choices will vary with analyzer model, hardware and Signal Path Setup configuration. Choices may include:

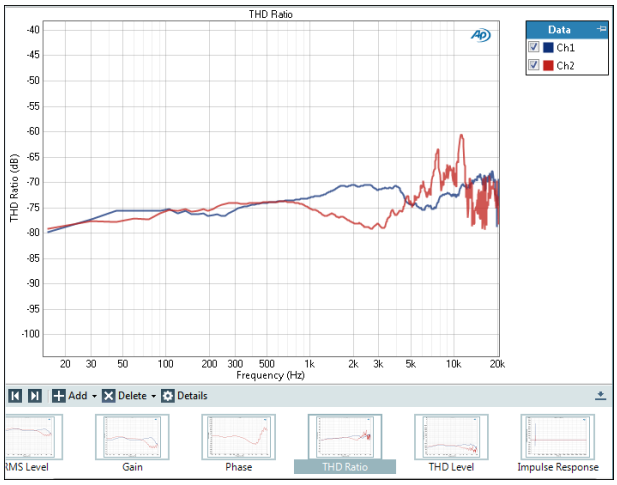


Data 1 Menu Data 2 Menu

Remove a row by clicking the **Delete** button. Both left and right axis results are removed. Results can also be added or deleted from the Selector filmstrip.

Note that in the Data 1 menu, you can select **Choose Results...** to open the **Add Primary Results** dialog, where you can add multiple results. Also, in the **Data 2** menu, you can select **Derived...** to add **Derived Results**. See Chapter 100 for more information about Derived Results.

Typical Result Display



Continuous Sweep result displays are shown in detail beginning on page 239.

Settling

Continuous sweep results are always unsettled.

Extend Acquisition By

By default, a continuous sweep measurement makes an acquisition slightly longer than the stimulus, to include possible time-delayed artifacts created in the DUT. By default, the acquisition is extended 50 ms longer than the stimulus.

In some cases, you may want to extend the acquisition further. Enter a new value in the **Extend Acquisition By:** field. Minimum extension is 0 s; maximum is 3 s.

Crosstalk Mode

Choose the mode of crosstalk measurement (or **None**). This selection changes the sweep alignment in time across channels, and may affect other continuous sweep results. For more information, see Continuous Sweep Crosstalk modes on page 248.

Choices are

- None
- **High speed** (default)
- High accuracy, one channel driven
- High accuracy, one channel undriven

Nesting

Bench Mode Continuous Sweeps can be nested. See Nested Sweeps on page 185.

Secondary Source

Select the secondary sweep source here. This is the outside, nesting parameter that is varied to change the conditions for the primary sweep through a number of iterations. Available choices are dependent upon instrument hardware, options, Input/Output settings and Bench Mode Generator and other settings.

Acoustic Response
(Bench Mode)

For the APx515, Bench Mode requires a software option key. See page 189 for more information about software options.

Read about the Bench Mode workspace, Generator and Analyzer panels in Chapter 88.

Overview

Bench Mode Acoustic Response uses a continuous sweep, with the addition of a time window feature to enable quasi-anechoic acoustic measurements.

The Acoustic Response measurements are designed to facilitate loudspeaker testing using an APx500 Series analyzer with a measurement microphone and an audio power amplifier.

Typically, the APx500 generator output is connected to the power amplifier, which in turn is connected to the loudspeaker under test. A measurement microphone is located in the acoustic field of the loudspeaker, and the output of the microphone preamplifier is connected to an APx500 analyzer input. A specially configured continuous response measurement (called the Acoustic Response measurement) is made.

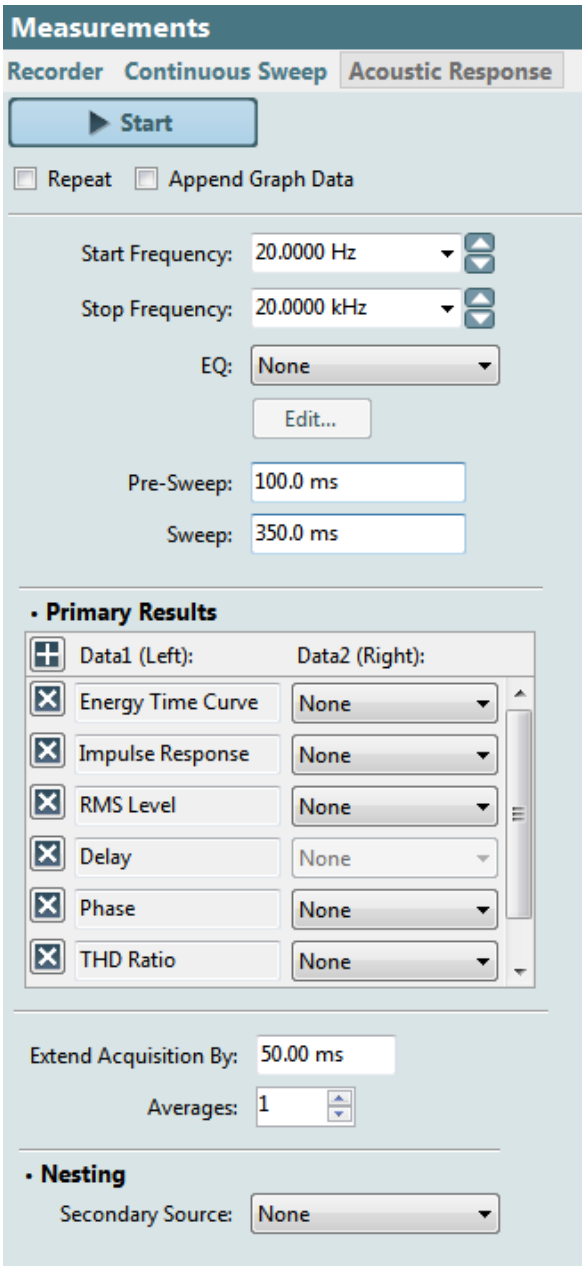
For more information about continuous sweep measurements, go to page 246.

Continuous sweep based measurements are not supported in External Source or File Input configurations.

Acoustic Response in Bench Mode is essentially the same measurement that is presented in Sequence Mode, with some Bench Mode additions. Read more about Acoustic Response and in particular, tips for setting up you measurements, in Chapter 24.

Generator and Analyzer settings

In Bench Mode, these settings are made in the Generator and Analyzer panels. See pages 578 and 579.



Nested Sweeps

A Bench Mode Acoustic Response sweep can be run iteratively at different steps of an additional parameter (the secondary source) in what is referred to as a nested sweep. See Nesting, below on page 607.

Operation

If you have not yet set up your test, first go to Signal Path Setup to select and configure inputs and outputs, references and other settings. Read about the Signal Path Setup menu in Chapter 7.

Start

To make an Acoustic Response measurement, click **Start**. The generator will output a continuous sweep to the DUT on the selected generator channels at the level set in the Generator panel. In a short period of time, the signal will be acquired and processed, and each of the small graphs in the Selector will be populated with results. Simply click the result view you want.

Be sure that you have set the generator Level low enough that the loudspeaker will not be damaged. Also, be sure you have set the generator Start and Stop frequencies so that the loudspeaker will not be damaged by out-of-band signals. See Loudspeaker Testing using Acoustic Response beginning on page 197, for more information.

Repeat

When the **Repeat** checkbox is checked, the sweep (or nested sweep) repeats continually. Unchecking the **Repeat** checkbox or clicking **Stop** will cancel **Repeat**.

Note: if Append Graph Data is checked while Repeat is set, a great deal of data will be quickly amassed and the computer memory or storage may be overwhelmed.

Append Graph Data

Measured data are grouped in a **Data Set**, which contains all acquisition results. Normally, the current graph data are deleted each time you start a new measurement. If the **Append Graph Data** box is checked, appended data are grouped as additional data sets, as are any imported data. The Data Sets Panel displays the Data Set(s) in the measurement, and provides a visibility checkbox, a Time acquisition field, and a Notes field for each data set.

Read more about Data Sets beginning on page 649.

Primary Source

The primary source for a continuous sweep measurement is the log-swept sine signal. The range between the **Start** and **Stop** frequencies provides the X-axis for most of the result graphs.

Start Frequency and Stop Frequency

These controls set the frequency range of the sweep.

EQ

You can equalize the generator signal to make a pre-emphasized stimulus. See Generator Equalization (page 192) for a general discussion of this feature.

Choosing generator equalization

Select **None, Relative** or **Absolute** from the EQ drop-down menu.

- **None** applies no equalization to the generator.
- **Relative** applies the gain or loss specified in the EQ table to the current generator level for each channel.
- **Absolute** sets the generator level for all channels to the level specified in the EQ table. Note: when **Absolute** is selected, current channel level settings are lost.

Edit

If you have chosen to equalize the generator signal, you can edit the EQ table here. See **Edit EQ Table** (page 192).

Pre-Sweep and Sweep duration fields

A continuous sweep stimulus consists of a sine wave swept logarithmically across the sweep range in the period of time set in these fields.

To enable the DUT to stabilize before the beginning of the defined sweep measurement, there is variable **Pre-Sweep** time. The actual sweep stimulus signal begins below the defined **Start Frequency** and is swept upward, reaching the **Start Frequency** at the end of the **Pre-Sweep** time. The sum of the times in **Pre-Sweep** and **Sweep** is the total sweep length.

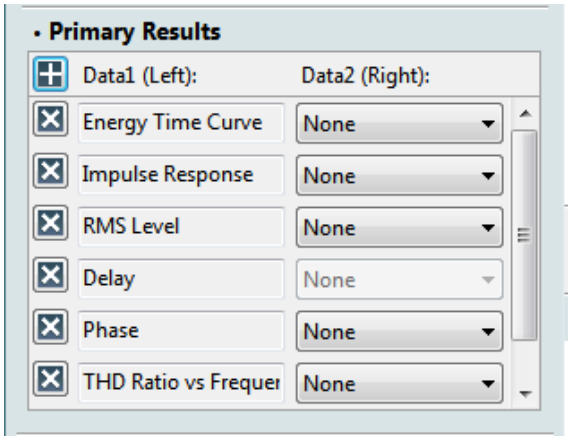
- The default for **Pre-Sweep** is 100.0 ms. Minimum is 0 s; maximum is 1.0 s.
- The default for **Sweep** is 350.0 ms. Minimum is 50.0 ms; maximum is 5 s.

Longer **Pre-Sweep** settings allow more time for DUT stabilization, but increase measurement time. Longer **Sweep** settings provide greater resolution and signal-to-noise ratio, but increase measurement time.

Primary Results

In Bench Mode Acoustic Response, Primary results include 16 sets of data derived from the sweep.

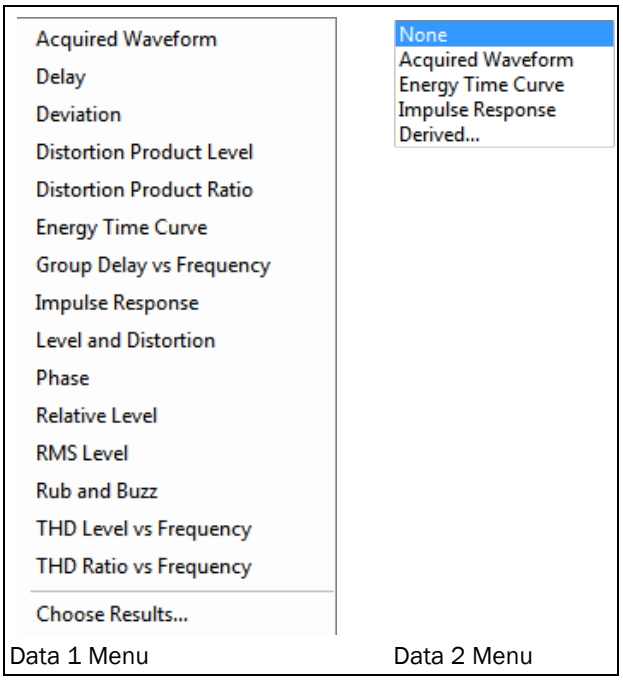
Initially, only six results are set, plotted as Data1 on the left Y axis of the graph.



To add more primary results to the left axis, click the Add Data1 (Left) button and select one or more results.

To add a primary or derived result to the right axis, open the Data2 (Right) menu for a given existing Data1 result, and select a result.

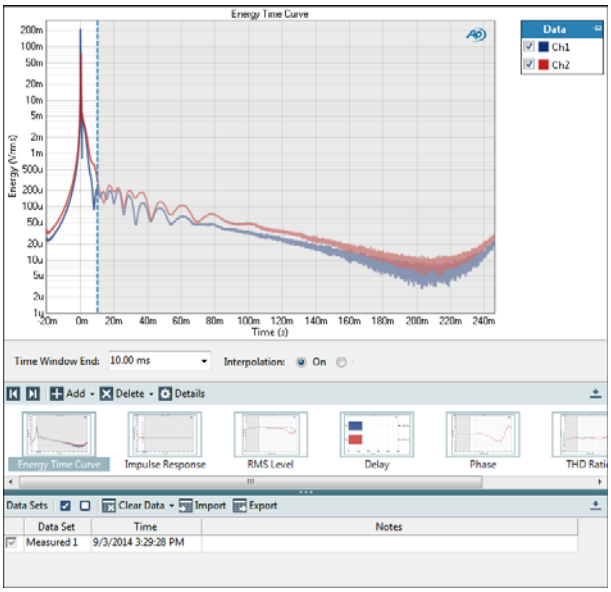
Available choices will vary with analyzer model, hardware and Signal Path Setup configuration. Choices may include:



Remove a row by clicking the **Delete** button. Both left and right axis results are removed. Results can also be added or deleted from the Selector filmstrip.

Note that in the Data 1 menu, you can select **Choose Results...** to open the **Add Primary Results** dialog, where you can add multiple results. Also, in the **Data 2** menu, you can select **Derived...** to add **Derived Results**. See Chapter 100 for more information about Derived Results.

Typical Result Display



Acoustic Response result displays are shown in detail beginning on page 203.

Settling

Continuous sweep results are always unsettled.

Extend Acquisition By

By default, a continuous sweep measurement makes an acquisition slightly longer than the stimulus, to include possible time-delayed artifacts created in the DUT. By default, the acquisition is extended 50 ms longer than the stimulus.

In some cases, you may want to extend the acquisition further. Enter a new value in the **Extend Acquisition By:** field. Minimum extension is 0 s; maximum is 3 s.

Averages

Averaging several sweeps can remove uncorrelated data from the results (for acoustic measurements, this is usually background noise). The sweep will be run the number of times set in the **Averages** field, and the results will be averaged for display.

Nesting

Bench Mode Acoustic Responses can be nested. See Nested Sweeps on page 185.

Secondary Source

Select the secondary sweep source here. This is the outside, nesting parameter that is varied to change the conditions for the primary sweep through a number of iterations. Available choices are dependent

upon instrument hardware, options, Input/Output settings and Bench Mode Generator and other settings.

Section V: Reference

IEC60958 Metadata

Set Status Bits / Set User Bits

For more information about status bits and user bits, go to page 389.

This topic discusses **setting** the status bits and user bits to be embedded in the digital output signal. For information about **reading** the status bits on a digital input, go to the Metadata Monitor: Status Bits on page 34. For information about recording changes in status bits or users bits, see Chapter 62, the Metadata Recorder.

Status bits are supported in the transport streams for the Digital Balanced, Digital Unbalanced, Digital Optical and Digital HDMI inputs.

The Digital Serial Input/Output (DSIO) protocol does not support embedded metadata.

Setting Status Bits

In one of the supported digital modes, from Signal Path Setup click **Settings > Status/User Bits**. The **Set Status Bits / User Bits** dialog will open.

Auto

When **Auto** is selected, APx500 chooses the consumer application or the professional application depending upon your digital output choices. The selected application is shown in the checkbox name: **Auto (Consumer)**, for example. In **Auto**, all status bits are set to default values. User bits are set to 00.

When **Auto** is not selected, the status bits and user bits can be set to any available value.

Subframes A and B are the same (not available in HDMI)

Each frame of audio in the transport stream has 2 subframes, A and B. These are sometimes called channels A and B. For the AES3/SPDIF transport stream, the status bits for A and B are typically set to the same values, but can be set independently.

When **Subframes A and B are the same** is checked, any setting made in subframe A status bits is copied to subframe B. When **Subframes A and B are the same** is not checked, the entry fields for subframe B become available, and status bits for the subframes can be set independently.

Note: some early versions of APx instrument hardware do not support independent setting of subframe A and subframe B status bits.

For HDMI, the Status Bits for subframes A and B are always set to the same values, and this control is not available.

Auto Increment Local Address

This setting is only available when the instrument is fitted with an Advanced Digital I/O module. When checked, the Local Address increments in integers at the sample rate.

Auto Increment Time of Day

This setting is only available when the instrument is fitted with an Advanced Digital I/O module. When checked, the Time of Day increments in integers at the sample rate.

User Defined CRC ChA

This setting is only available when the instrument is fitted with an Advanced Digital I/O module. Enter any hex number in the adjacent field.

User Defined CRC ChB

This setting is only available when the instrument is fitted with an Advanced Digital I/O module. Enter any hex number in the adjacent field.

Display Grids

Plain Text display and Hex display

Status bits can be set from the grid displays, reading the plain text interpretations and choosing settings from drop-down menus in the grid. Status bits can also be set by entering hex values for selected bytes in the Status Bits hex display. These two displays are coupled, and a change in one will be reflected by the corresponding change in the other.

User bits

You can arbitrarily set user bits to any available value by entering hex values for selected bytes in the User Bits hex display. User bits are associated with frames, not subframes; they are not affected by status bit subframe A or B settings.

Note: some early versions of APx instrument hardware do not support setting of user bits.

How Digital Output Configuration Choices Affect Status Bits

In APx500, making a digital output format selection also affects digital metadata by setting the application for the Auto mode on the Set Status Bits / User Bits panel. The application for Auto mode is set as follows:

Digital Unbalanced Output

You can set Digital Unbalanced to either **Application: Consumer** or **Application: Professional** on the Settings panel for this output. The default is **Application: Consumer**. Note that these choices also set the output voltage for the unbalanced output. See Settings for Digital Unbalanced Output for more information.

The **Application** settings can be overridden by deselecting **Auto** on the **Set Status Bits / User Bits** panel.

The electrical settings persist, regardless of metadata settings.

Digital Balanced Output

The Digital Balanced transport stream is output with professional application status bits, when **Auto** is set on the **Set Status Bits / User Bits** panel. This application setting can be overridden by deselecting **Auto**.

The Digital Balanced transport stream is always output with a peak-to-peak level of 5.0 V, within the recommendations of AES3 and IEC60958-4.

Digital Optical Output

The Digital Optical transport stream is output with consumer application status bits, when **Auto** is set on the Set **Status Bits / User Bits** panel. This application setting can be overridden by deselecting **Auto**.

HDMI Source

The HDMI transport stream carries an interpretation of the AES3/IEC60958 protocol, including status bits. These status bits are set to consumer application, when **Auto** is set on the **Set Status Bits** panel. This application setting can be overridden by deselecting **Auto**.

HDMI ARC Tx

The ARC transport stream carries IEC60958 audio, although with different physical layer characteristics. Status bits are set to consumer application, when **Auto** is set on the **Set Status Bits** panel. This application setting can be overridden by deselecting **Auto**.

Digital Serial Output

The Digital Serial Input/Output (DSIO) protocol does not support embedded metadata.

PDM

The PDM Input/Output does not support embedded metadata.

Supported Audio File Formats

...for import as Generator Waveforms

The following audio file formats can be imported as Generator Waveforms (Arbitrary Waveforms).

Format	*Detail*	*Sample Rate*	*Filename Extension*
Linear PCM			
WAVE_FORMAT_PCM (Type 1)	8 to 32 bit mono or stereo†	Any rate in the range of 4 kHz—216 kHz	*.wav
WAVE_FORMAT_EXTENSIBLE	8 to 32 bit multichannel†	Any rate in the range of 4 kHz—216 kHz	*.wav
WAVE_FORMAT_PCM (proprietary mapping)*	8 to 32 bit multichannel†	Any rate in the range of 4 kHz—216 kHz	*.wav
AP legacy waveform	24 bit mono or stereo	Any rate in the range of 4 kHz—216 kHz	*.agm, *.ags
Coded			
Dolby Digital	5.1 surround	32 kHz or 48 kHz	*.ac3
Dolby Digital Plus	7.1 surround	48 kHz	*.ec3
Dolby TrueHD	8 channel	48 kHz, 96 kHz and 192 kHz	*.mlp
MAT (Dolby HD container)	stream	48 kHz, 96 kHz and 192 kHz	*.mat
dts Digital Surround	5.1 surround	48 kHz and 96 kHz	*.dts
dts Digital Surround compact	5.1 surround	48 kHz and 96 kHz	*.cpt
dts-HD High Resolution Audio	6 channel	48 kHz and 96 kHz	*.dtshd
dts-HD Master Audio	8 channel	48 kHz, 96 kHz and 192 kHz	*.dtshd

* Ambiguities within WAVE_FORMAT_PCM allow undocumented multiple channel configurations. This particular proprietary format is in common use, and can be created in Sony Sound Forge and other waveform editors.

† For 32-bit audio, proprietary 32-bit fixed point .wav formats are supported.

...for import as File Input

The following audio file formats can be imported for analysis using the File Input feature. Note that only files from acquisitions saved by APx500 will contain cross-domain scaling data.

Format	*Detail*	*Sample Rate*	*Filename Extension*
Linear PCM			
WAVE_FORMAT_PCM (Type 1)	8 to 32 bit mono or stereo†	Any rate in the range of 4 kHz—216 kHz	*.wav
WAVE_FORMAT_EXTENSIBLE	8 to 32 bit multichannel†	Any rate in the range of 4 kHz—216 kHz	*.wav
WAVE_FORMAT_PCM (proprietary mapping)*	8 to 32 bit multichannel†	Any rate in the range of 4 kHz—216 kHz	*.wav

* Ambiguities within WAVE_FORMAT_PCM allow undocumented multiple channel configurations. This particular proprietary format is in common use, and can be created in Sony Sound Forge and other waveform editors.

† For 32-bit audio, proprietary 32-bit fixed point .wav formats are supported.

...for export as Record to File (linear audio)

These audio file formats can be exported and saved to disc from Measurement Recorder, Noise Recorder and Multitone Recorder.

Format	*Detail*	*Sample Rate*	*Filename Extension*
Linear PCM			
WAVE_FORMAT_PCM (Type 1)	8 to 32 bit mono or stereo†	For analog acquisitions, 6 kHz, 8 kHz, 12 kHz, 16 kHz, 24 kHz, 44.1 kHz, 48 kHz, 96 kHz or 192 kHz. If fitted with BW52, 624 kHz, 1248 kHz and 2496 kHz are also available. For digital acquisitions, files are saved at the same sample rate as the acquisition.	*.wav
WAVE_FORMAT_EXTENSIBLE	8 to 32 bit multichannel†	as above.	*.wav
WAVE_FORMAT_PCM (proprietary mapping)*	8 to 32 bit multichannel†	as above.	*.wav

* Ambiguities within WAVE_FORMAT_PCM allow undocumented multiple channel configurations. This particular proprietary format is in common use, and can be created in Sony Sound Forge and other waveform editors.

† For 32-bit audio, proprietary 32-bit fixed point .wav formats are supported.

...for export as Record to File (coded audio)

These audio file formats can be exported and saved to disc from Measurement Recorder, digital acquisitions only. Coded audio (typically multichannel Dolby or DTS surround sound, coded in an IEC61937 bitstream) is saved in the incoming format.

Format	*Sample Rate*	*Filename Extension*
Coded Audio		
Dolby Digital	File is saved at the same sample rate as the acquisition.	*.ac3
Dolby Digital Plus	File is saved at the same sample rate as the acquisition.	*.ec3
Dolby TrueHD	File is saved at the same sample rate as the acquisition.	*.mlp
dts Digital Surround	File is saved at the same sample rate as the acquisition.	*.dts
dts Digital Surround compact	File is saved at the same sample rate as the acquisition.	*.cpt
dts-HD	File is saved at the same sample rate as the acquisition.	*.dtshd
raw encoded bitstream	File is saved at the same sample rate as the acquisition.	*.bin

Bandwidth Limiting, Weighting and EQ Filters

Introduction

Also see More About Filters on page 622.

Bandwidth limiting or weighting filters are commonly used in distortion and noise measurements. Bandwidth limiting filters cut off the extremes of the frequency spectrum to eliminate low or high frequency signals that might confuse results. Weighting filters approximate the characteristics of a transmission system or of human hearing with a more complex filter curve.

Always state the filtering used when reporting distortion or noise results.

Signal Path Setup: Filters

In Sequence Mode, each signal path has a set of filters that, when applied, affect the entire signal path. Bench Mode Signal Path Setup has the same set of filters available.

Local Measurement Filters

In addition to the Signal Path Filters, some Sequence Mode measurements and some Bench Mode Analyzer features have local filters available.

Signal Path Filters

The **Filters** controls on the **Signal Path Setup > Input/Output** panel enable high-pass, low-pass, bandwidth, weighting and EQ filters. In Sequence Mode, these filters affect all the incoming audio throughout the current signal path; in Bench Mode, the filters affect all the incoming audio for the entire Bench Mode project. The Signal Path Setup filters are implemented in the time domain.

Note: We do not recommend using input filtering when using the Impedance/Thiele-Small measurement or the Loudspeaker Production Test measurement. Impedance curves derived with input filtering in effect might be in error. For low-frequency drivers, be sure that the input signal path is DC-coupled.

High-pass (analog or digital audio signals)

- **DC**
  This selection provides no high-pass filtering. The signal is DC coupled.

Read more about DC in APx analyzers beginning on page 266.

- **AC (<10 Hz)**
  This selection provides AC coupling flat to 20 Hz, and about 0.1 dB down at 10 Hz.
- **Butterworth**
  You can specify a custom 4-pole Butterworth high-pass filter here, by entering a value for the corner frequency in the field; the acceptable range is 10 Hz to 90 kHz (standard analog input), 10 Hz to 1 MHz (APx555 or BW52 analog input), 10 Hz to 100 kHz (digital input). The signal is AC coupled.
- **Elliptic**
  You can specify a custom 5-pole elliptic high-pass filter here, by entering a value for the corner frequency in the field; the acceptable range is 10 Hz to 90 kHz (standard analog input), 10 Hz to 1 MHz (APx555 or BW52 analog input), 10 Hz to 100 kHz (digital input). The filter has a passband ripple of 0.01 dB and a sharp roll-off to –60 dB. It closely matches the response of the frequency-domain filters in earlier versions of APx500. The signal is AC coupled.

High-pass (jitter signals)

- **700 Hz (AES3)**
  This filter conforms to the AES3 recommendation for intrinsic jitter measurement.
- **Butterworth**
  You can specify a custom 4-pole Butterworth high-pass filter here, by entering a value for the corner frequency in the field; the acceptable range is 50 Hz to 150 kHz. The signal is AC coupled.

- **Elliptic**
  You can specify a custom 5-pole elliptic high-pass filter here, by entering a value for the corner frequency in the field; the acceptable range is 50 Hz to 150 kHz. The filter has a passband ripple of 0.01 dB and a sharp roll-off. The signal is AC coupled.

Note that the corner frequency for the high-pass filter cannot be higher than the corner frequency for the low-pass filter.

Low-pass (analog audio signals)

ADC Passband

All analysis in APx analyzers is performed in digital signal processing (DSP), so all analog input signals are digitized using high-performance analog-to-digital converters (ADCs). To avoid aliasing, ADC designs effectively filter the audio to just below 1/2 the sample rate (Fs/2, the Nyquist frequency). Therefore, selecting an ADC sample rate sets a de facto low-pass filter. In APx500 this filtering is referred to as *ADC Passband*.

Bandwidth

For ADC Passband, these bandwidth choices are available:

- 20 kHz (SR 48 kHz)
- 45 kHz (SR 96 kHz)
- 90 kHz (SR 192 kHz)
- 250 kHz (SR 624 kHz) (APx555 or BW52 option)
- 500 kHz (SR 1.248 MHz) (APx555 or BW52 option)
- 1 MHz (SR 2.496 MHz) (APx555 or BW52 option)

AES17 (20 kHz)

This filter is an 8-pole elliptic filter with a corner frequency at 20 kHz, satisfying the AES17 recommendation for converter measurements and other measurements in the presence of high out-of-band noise. When this filter is selected, the ADC sample rate is set to 48 kHz.

AES17 (40 kHz)

This filter is an 8-pole elliptic filter with a corner frequency at 40 kHz, satisfying the AES17 recommendation for converter measurements and other measurements in the presence of high out-of-band noise. When this filter is selected, the ADC sample rate is set to 96 kHz.

Butterworth

You can specify a custom 8-pole low-pass Butterworth filter here, by entering a value for the corner frequency in the field. The acceptable range is 10 Hz to 1 MHz (when supported by hardware). An 8-pole Butterworth filter is not as sharp as an elliptic filter. The Butterworth filter is –3 dB at the corner frequency and falls off at a rate of 48 dB per octave.

When this filter is selected, the ADC sample rate is set to the next range above the requested corner frequency. The final filter curve will reflect the combination of the Butterworth filter and the ADC Passband filter.

Elliptic

You can create a custom 8-pole low-pass elliptic filter here, by entering a value for the corner frequency in the field. The acceptable range is 1 kHz to 1 MHz. When this filter is selected, the ADC sample rate is set to the next range above the requested corner frequency. The elliptic filter has a passband ripple of 0.01 dB and a very sharp roll-off to –60 dB.

When this filter is selected, the ADC sample rate is set to the next range above the requested corner frequency. The final filter curve will reflect the combination of the elliptic filter and the ADC Passband filter. Since the elliptic filter has such a sharp roll-off, the combined response will be dominated by the elliptic filter.

Low-pass (digital signals)

Fs/2

The bandwidth of a digital audio signal cannot exceed one-half the sample rate (Fs/2). The **Fs/2** setting passes a digital signal at its maximum bandwidth, without any additional low-pass filtering.

Butterworth

You can specify a custom 8-pole low-pass Butterworth filter here, by entering a value for the corner frequency in the field. The acceptable range is 10 Hz to 100 kHz.

Elliptic

You can specify a custom 8-pole low-pass elliptic filter here, by entering a value for the corner frequency in the field. The acceptable range is 10 Hz to 100 kHz.

Low-pass (jitter signals)

- **700 Hz (AES3)**
  This filter conforms to the AES3 recommendation for intrinsic jitter measurement.
- **Butterworth**
  You can specify a custom 8-pole Butterworth high-pass filter here, by entering a value for the corner frequency in the field; the acceptable range is 50 Hz to 150 kHz. The signal is AC coupled.
- **Elliptic**
  You can specify a custom 8-pole low-pass elliptic filter here, by entering a value for the corner frequency in the field; the acceptable range is 50 Hz to 150 kHz. The filter is implemented in the time

domain, and has a passband ripple of 0.01 dB and a sharp roll-off. The signal is AC coupled.

Note that the corner frequency for the low-pass filter cannot be lower than the corner frequency for the high-pass filter.

Weighting

APx500 provides weighting and deemphasis filters for noise and distortion measurements. When applied from Signal Path Setup, weighting filters affect the full signal. This may be appropriate for a noise measurement, but for a THD+N measurement it is recommended that you use a filter dedicated to a THD+N measurement or meter, which will only filter the residual signal.

See Local Measurement Filters, below.

APx500 provides these weighting filters:

- A-wt.
- B-wt.
- C-wt.
- CCIR-1k
- CCIR-2k
- CCITT
- C-message

Deemphasis filters are also included, for use with pre-emphasized DUTs or stimulus signals.

- 50us de-emph.
- 75us de-emph.
- 50us de-emph. + A-wt
- 75us de-emph. + A-wt.

EQ

See Input Equalization on page 621.

APx500 provides EQ (equalization) filtering, enabling compensation for a non-flat microphone response in the system under test. Input Equalization can also be used to compensate for loudspeaker response in acoustic testing. The filter is a 30 pole IIR filter, and can be applied to any or all available input channels. Select **Create New** to open the **Edit Input EQ Table** dialog (see page 621); or choose **Browse for File** from the menu. The **Channels** button allows you to choose the channels to be equalized.

Local Measurement Filters

Measurement filters are “local”, affecting only the measurement (Sequence Mode) or meters (Bench Mode) to which the filter is attached.

Sequence Mode

Many Sequence Mode measurements have filter controls that affect that measurement only. These filters use the same DSP algorithms and have the same characteristics as an equivalent Signal Path Filter.

High-pass (analog or digital audio signals)

- **Signal Path**
  This selection applies no local high-pass filter at the measurement level. Only the filtering set in Signal Path Setup > Filters (which could range from none to highly filtered) will be in effect.
- **Butterworth**
  You can specify a custom 4-pole Butterworth high-pass filter here, by entering a value for the corner frequency in the field; the acceptable range is 10 Hz to 90 kHz (standard analog input), 10 Hz to 1 MHz (APx555 or BW52 analog input), 10 Hz to 100 kHz (digital input). The signal is AC coupled. Selecting a filter here overrides any high-pass filter set in Signal Path Setup > Filters.
- **Elliptic**
  You can specify a custom 5-pole elliptic high-pass filter here, by entering a value for the corner frequency in the field; the acceptable range is 10 Hz to 90 kHz. The filter has a passband ripple of 0.01 dB and a sharp roll-off to -60 dB. It closely matches the response of the frequency-domain filters in earlier versions of APx500. The signal is AC coupled. Selecting a filter here overrides any high-pass filter set in Signal Path Setup > Filters.

High-pass (jitter signals)

- **Signal Path**
  This selection applies no local high-pass filter at the measurement level. Only the filtering set in Signal Path Setup > Filters (which could be none, or substantial) will be in effect.
- **Butterworth**
  You can specify a custom 4-pole Butterworth high-pass filter here, by entering a value for the corner frequency in the field; the acceptable range is 50 Hz to 150 kHz. The signal is AC coupled. Selecting a filter here overrides any high-pass filter set in Signal Path Setup > Filters.
- **Elliptic**
  You can specify a custom 5-pole elliptic high-pass filter here, by entering a value for the corner frequency in the field; the acceptable range is 50 Hz to 150 kHz. The filter has a passband ripple of 0.01 dB and a sharp roll-off to –60 dB. The signal is AC coupled. Selecting a filter here overrides any high-pass filter set in Signal Path Setup > Filters.

Note that the corner frequency for the high-pass filter cannot be higher than the corner frequency for the low-pass filter.

Low-pass (analog audio signals)

- **Signal Path**
  This selection applies no local low-pass filter at the measurement level. Only the filtering set in Signal Path Setup > Filters (which could range from none to highly filtered) will be in effect.
- **Butterworth**
  You can specify a custom 8-pole low-pass Butterworth filter here, by entering a value for the corner frequency in the field. The acceptable range is 10 Hz to 1 MHz (when supported by hardware). An 8-pole Butterworth filter is not as sharp as an elliptic filter. The Butterworth filter is –3 dB at the corner frequency and falls off at a rate of 48 dB per octave. Selecting a filter here overrides any low-pass filter set in Signal Path Setup > Filters.

Note that you cannot specify a corner frequency higher than the current ADC passband upper frequency limit.

- **Elliptic**
  You can create a custom 8-pole low-pass elliptic filter here, by entering a value for the corner frequency in the field. The acceptable range is 1 kHz to 1 MHz. When this filter is selected, the ADC sample rate is set to the next range above the requested corner frequency. The elliptic filter has a passband ripple of 0.01 dB and a very sharp roll-off. Selecting a filter here overrides any low-pass filter set in Signal Path Setup > Filters.

Note that you cannot specify a corner frequency higher than the current ADC passband upper frequency limit.

Low-pass (digital audio signals)

- **Fs/2**
  The bandwidth of a digital audio signal cannot exceed 1/2 the sample rate (Fs/2). The Fs/2 setting passes a digital signal at its maximum bandwidth, without any additional low-pass filtering.
- **Butterworth**
  You can specify a custom 8-pole low-pass Butterworth filter here, by entering a value for the corner frequency in the field. The acceptable range is 10 Hz to 100 kHz.
- **Elliptic**
  You can specify a custom 8-pole low-pass elliptic filter here, by entering a value for the corner frequency in the field. The acceptable range is 10 Hz to 100 kHz.

Low-pass (jitter signals)

- **700 Hz (AES3)**
  This filter conforms to the AES3 recommendation for intrinsic jitter measurement.
- **Butterworth**
  You can specify a custom 8-pole low-pass Butterworth filter here, by entering a value for the corner frequency in the field; the acceptable range is 50 Hz to 150 kHz. The filter is implemented in the time domain. The signal is AC coupled.
- **Elliptic**
  You can specify a custom 8-pole low-pass elliptic filter here, by entering a value for the corner frequency in the field; the acceptable range is 50 Hz to 150 kHz. The filter is implemented in the time domain, and has a passband ripple of 0.01 dB and a sharp roll-off. The signal is AC coupled.

Note that the corner frequency for the low-pass filter cannot be lower than the corner frequency for the high-pass filter.

Weighting

APx500 provides weighting and deemphasis filters for noise and distortion measurements. A weighting filter, if selected, is applied to the noise or distortion results only, but not to the entire acquisition or measurement. The filters have the same characteristics as the Signal Path weighting filters, listed previously in this chapter.

Relationship between Sequence Mode Signal Path and Measurement Filters

Keep these rules in mind:

1. Setting a measurement (local) **low-pass** or **high-pass** filter overrides the corresponding Signal Path filter, while the measurement is active. Measurement (local) low-pass and high-pass filters affect the full signal for that measurement.
2. Setting a measurement (local) **weighting** filter DOES NOT override the Signal Path weighting filter; also, the measurement (local) weighting filter affects only the THD+N residuals. If you set both the Signal Path and measurement weighting filters, you will be filtering the signal twice: once the full signal, once the residuals only.
3. When a custom low-pass filter is requested as a Signal Path filter, the ADC sample rate may be shifted to accommodate the request, as explained in the previous section.
   However, when a custom low-pass filter is requested as a measurement (local) filter, the ADC sample rate is never changed; instead, the corner frequency entry field is constrained to be less than the bandwidth allowed by the current ADC sample rate.

Bench Mode

The Bench Mode analyzer provides a “local” THD+N Weighting filter that is only active for THD+N and Noise meters, with the same selections and character-

istics as the Sequence Mode Weighting measurement filters described above.

Input Equalization

Input Equalization allows you to insert a filter into the input signal path for all audio inputs. This enables compensation for a non-flat microphone response in the system under test. Input Equalization can also be used to compensate for loudspeaker response in acoustic testing.

Creating or importing an EQ curve

Open the **EQ** menu in **Signal Path Setup: Input Configuration**. Choose

- **Create New** or
- **Browse for file**.

The EQ response curve is specified in the Edit Input EQ Table dialog (see next), and response curves can be saved and imported. The response curve can be easily inverted.

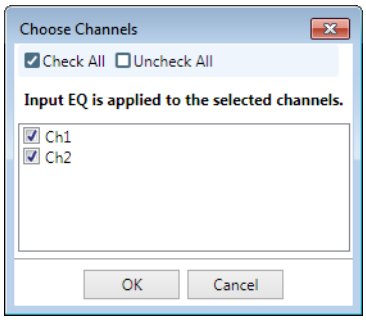
Design

The equalization filter is implemented in the digital domain early in the input signal path. The filter has a fixed number of poles (30). The EQ response beyond the range specified in the curve is unspecified.

Application

The equalization filter can be applied to all audio inputs, and also demodulated jitter inputs. The filter can be applied to a selection of available channels.

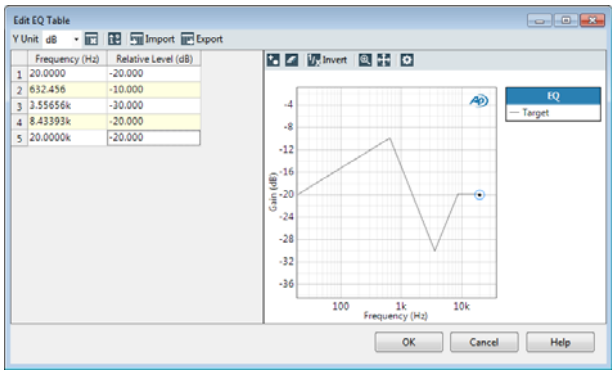
Click the **Channels** button to open the **Choose Channels** dialog.



Edit Input EQ Table dialog

This dialog allows you to import, edit and export Input Equalization tables.

The points in the table are represented graphically by the trace in the graph.



Y Unit

Choose the Y-axis (level) unit here.

Clear Table

Clear Table removes all points (rows) from the table.

Order Frequencies

Order Frequencies sorts the table by frequency, ascending toward the bottom of the table.

Import

Import opens a file browser window to select and open compatible EQ table files from disk. Excel, CSV and Audio Precision legacy *.adx and *.atsx are supported.

Export

Export opens a file browser window to save the current EQ table as an Excel or CSV file.

Add Point

Add Point inserts a blank frequency point (row) below the current selection in the table. Enter a level value for each new point. You can edit the frequency or level value arbitrarily.

- Minimum Frequency: 1 Hz
- Maximum Frequency: 1.04858 MHz

Frequencies must be unique, and ordered from low to high. See **Order Frequencies**, above.

- Minimum Level: -100 dB
- Maximum Level: 100 dB

Delete Selected Points

Delete Selected Points removes the selected points (rows) from the table.

Invert

Use this control to invert the EQ curve. This is useful for creating a compensation curve from a measured response curve.

Zoom

Click on this button and then drag a marquee in the graph to zoom.

Revert Zoom/Pan

Click on this button to revert to the original zoom and pan views.

Graph Properties

This button opens a Graph Properties dialog, where you can select ranges, names and log/lin characteristics for the EQ curve graph.

More About Filters in APx

ADC Passband

All analysis in APx analyzers is performed in digital signal processing (DSP), so all analog input signals are digitized using high-performance analog-to-digital converters (ADCs). To avoid aliasing, ADC designs effectively filter the audio to just below 1/2 the sample rate (Fs/2, the Nyquist frequency). Therefore, selecting an ADC sample rate sets a de facto low-pass filter. In APx500 this filtering is referred to as ADC Passband.

DSP filtering

In APx500 version 4.0 and later, most DSP filters are real time filters, implemented in the time domain. Time-domain filters can accurately model classic filter designs, including Butterworth and elliptic.

Butterworth

APx provides a user-definable 4-pole Butterworth high-pass filter, which falls off at a rate of 24 dB per octave, and an 8-pole Butterworth low-pass filter, at 48 dB per octave. Butterworth filters provide a very flat, ripple-free passband response. Enter a value for the corner frequency in the field.

Elliptic

Alternatively, APx provides a user-definable 5-pole elliptic high-pass filter, and an 8-pole elliptic low-pass filter. The elliptic filters fall off much more quickly than the Butterworth filter, at the price of some ripple in the passband and the stopband.

Bandpass

In the Sequence Mode, filtering for the bandpass measurements is accomplished in the frequency domain using FFT techniques. In the Bench Mode Bandpass measurements, the bandpass filtering is accomplished in DSP the time domain.

Notch

A narrow notch filter is used to remove the fundamental frequency from distortion measurements. With one exception, notch filters are implemented in DSP in the time domain (see below).

Analog filtering

Capacitive coupling

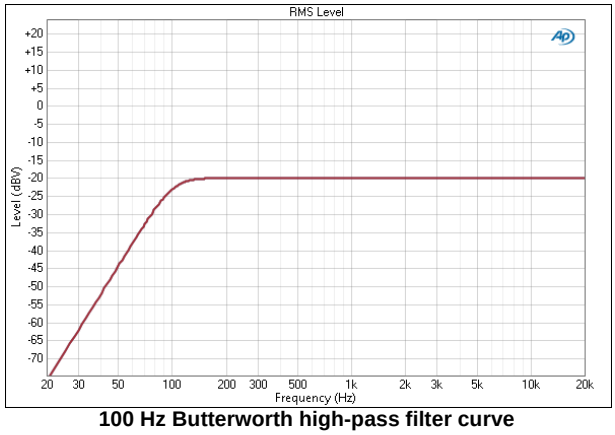
APx analyzers can be DC coupled throughout the signal path. When AC coupled, a coupling capacitor is inserted early in the signal path, with the effect of rolling off the low-frequency response below 10 Hz.

Notch

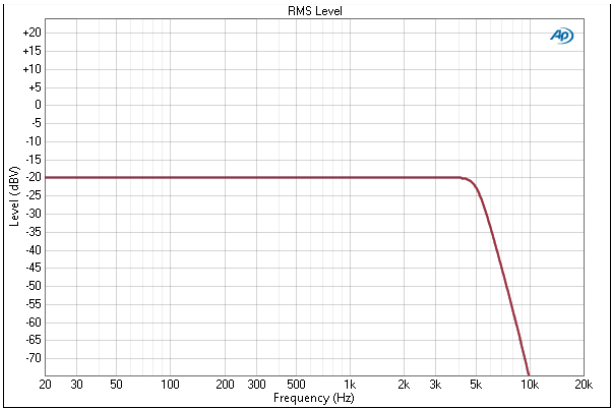
In the APx555, an additional ADC is provided to digitize the notched analog signal. A stereo pair of analog notch filters are used in this path, previous to the additional ADC.

Typical Filter Curves

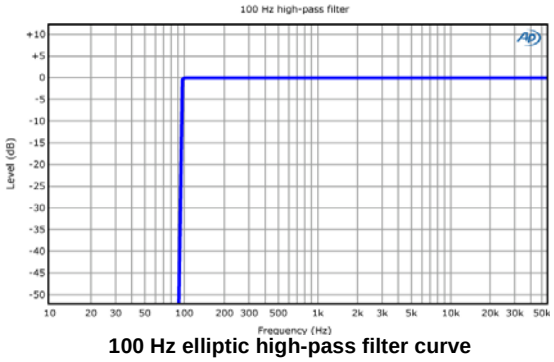
High-pass and low-pass curves



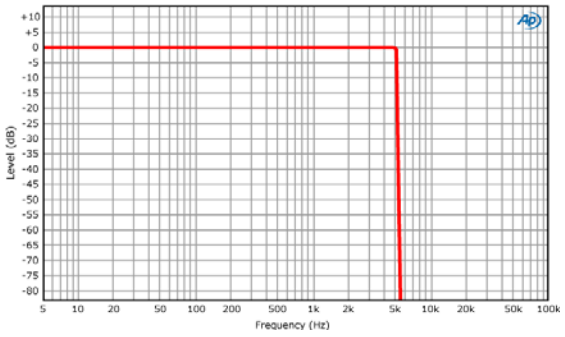
100 Hz Butterworth high-pass filter curve



5 kHz Butterworth low-pass filter curve

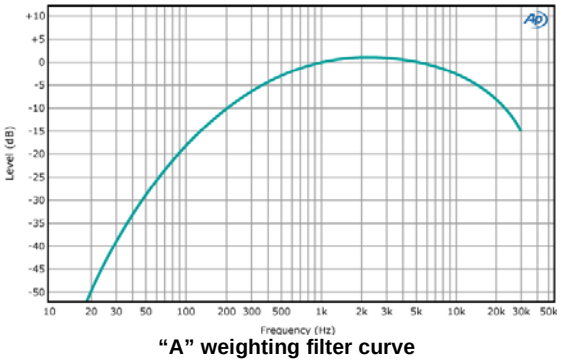


100 Hz elliptic high-pass filter curve

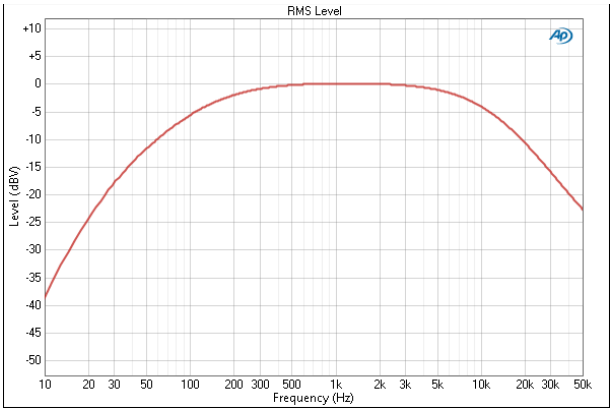


5 kHz elliptic low-pass filter curve

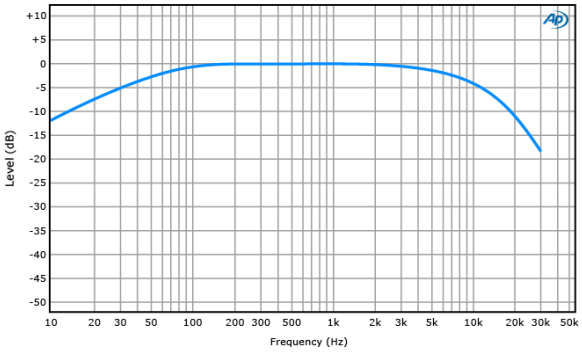
Weighting and deemphasis filter curves



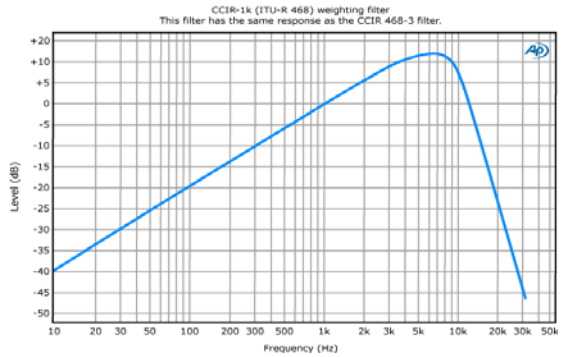
“A” weighting filter curve



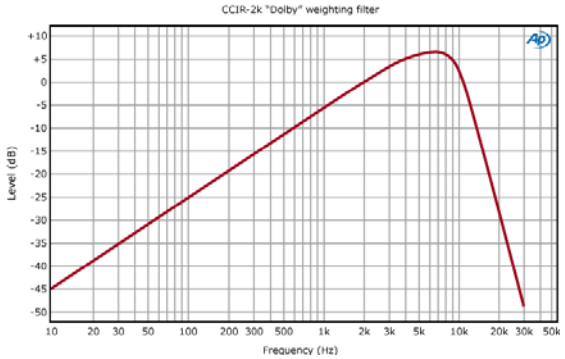
“B” weighting filter curve



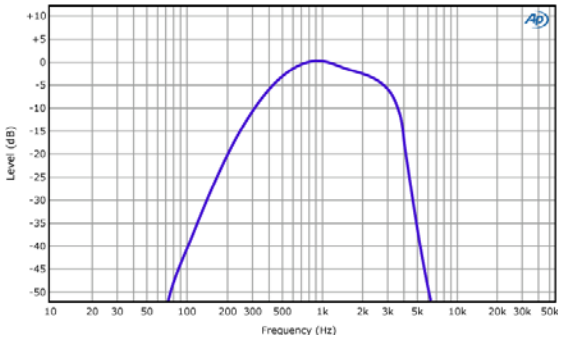
“C” weighting filter curve



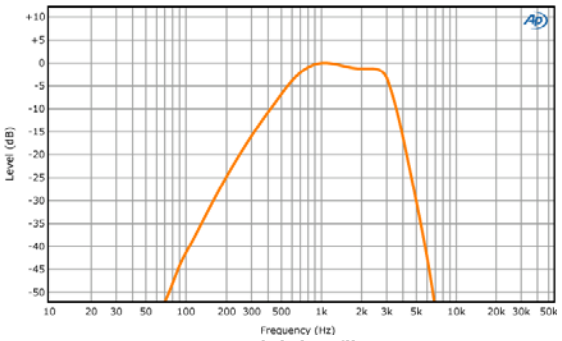
CCIR-1k (ITU-R 468) weighting filter curve



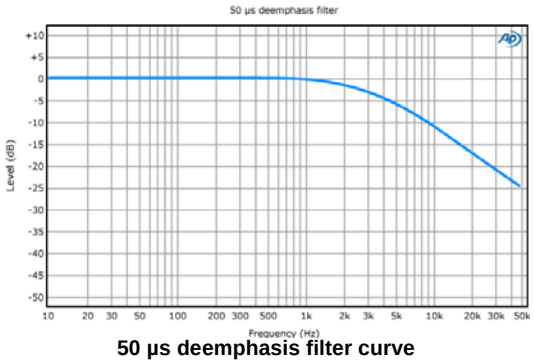
CCIR-2k "Dolby" weighting filter curve



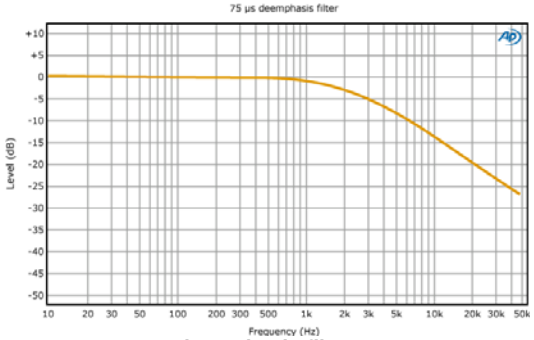
CCITT weighting filter curve



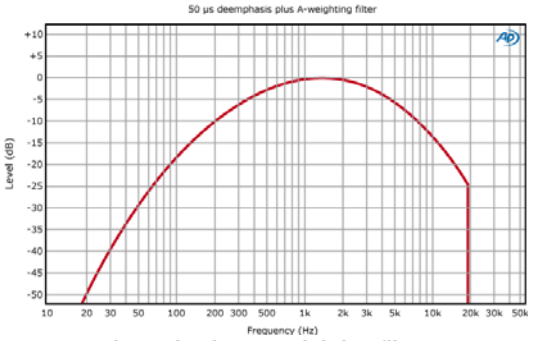
C-Message weighting filter curve



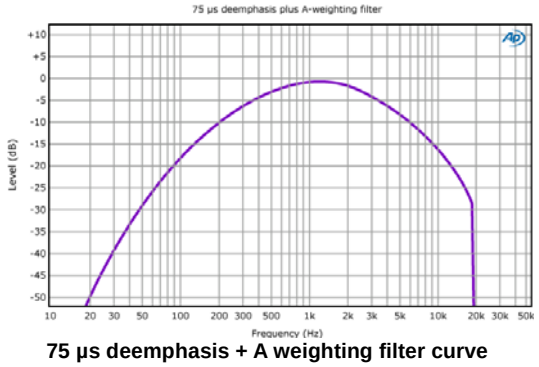
50 µs deemphasis filter curve



75 µs deemphasis filter curve

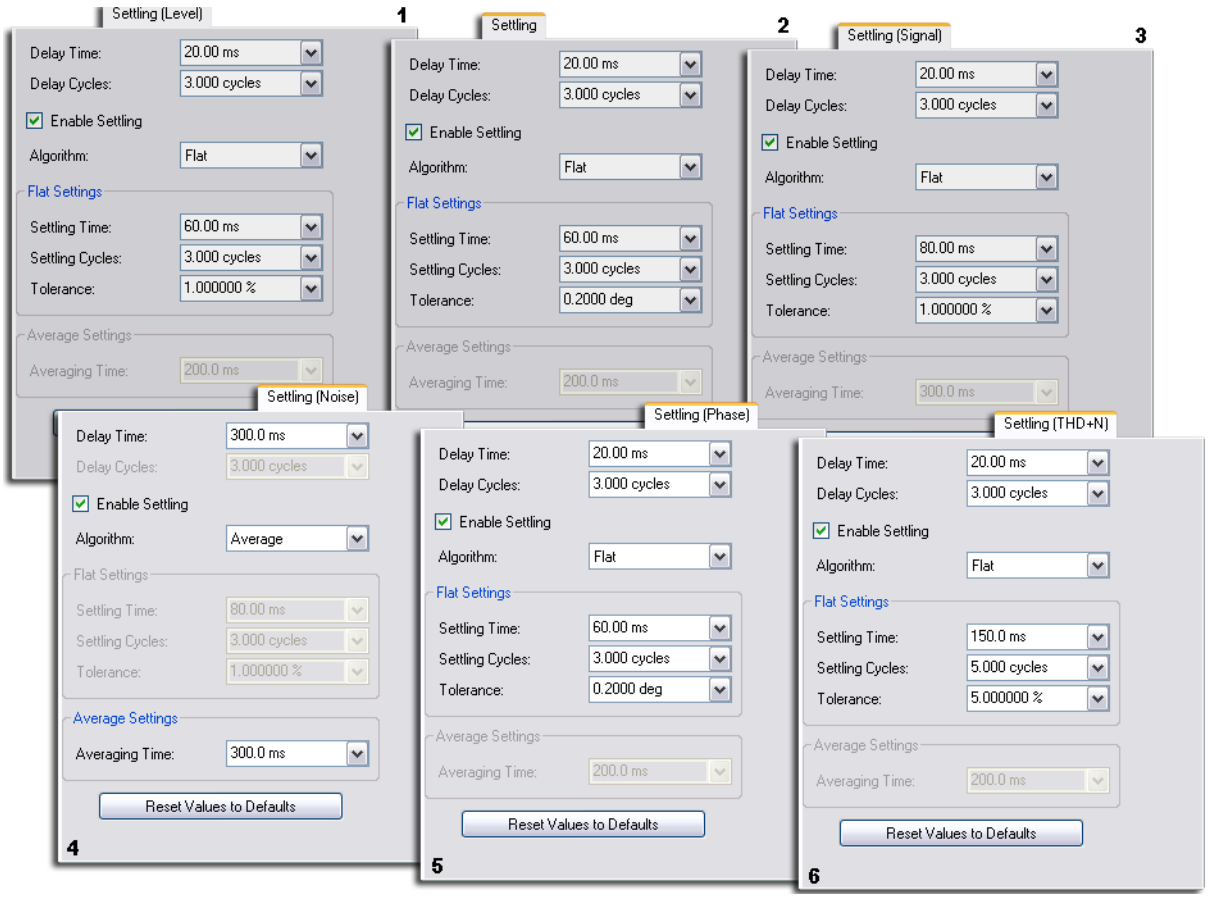


50 µs deemphasis + A weighting filter curve



75 µs deemphasis + A weighting filter curve

Ranging and Settling



Ranging

Audio amplifier circuits are required to operate over vast dynamic ranges: 80, 100 or even 120 dB or more. Distortion and noise at the extremes are to be expected.

The need for ranging

The input circuits of an audio meter or analyzer employ switchable attenuation or gain to accommodate the extremes of dynamic range for accurate, high-resolution measurement.

Fixed ranging

When an input signal is low, the analyzer gain should be set to amplify that signal to an optimum level for measurement. When an input signal is high, the analyzer gain should be reduced to prevent overload and to bring the signal down into the optimum measurement range.

Most meters and analyzers provide a way for you to manually set ranging to accommodate the input signal level. In Audio Precision instruments, the ranging choices are shown in a dialog that allows you to select the maximum level you expect.

For the APx515, six ranges are available for both unbalanced and balanced analog inputs:

- 0 Vrms to 250 mVrms
- 250 mVrms to 800 mVrms
- 800 mVrms to 2.5 Vrms
- 2.5 Vrms to 8 Vrms
- 8 Vrms to 25 Vrms
- 25 Vrms to 80 Vrms

For the APx582, 585 and 586 (8-channel input module), six ranges are available for both unbalanced and balanced inputs:

- 0 Vrms to 320 mVrms
- 320 mVrms to 1 Vrms
- 1 Vrms to 3.2 Vrms
- 3.2 Vrms to 10 Vrms
- 10 Vrms to 32 Vrms
- 32 Vrms to 100 Vrms

For the APx520, 521, 525, and 526 (2-channel input module), seven ranges are available. For the unbalanced analog inputs they are:

- 0 Vrms to 320 mVrms
- 320 mVrms to 1 Vrms
- 1 Vrms to 3.2 Vrms
- 3.2 Vrms to 10 Vrms
- 10 Vrms to 32 Vrms
- 32 Vrms to 100 Vrms
- 100 Vrms to 160 Vrms

For the balanced analog inputs, the seventh range is extended to a higher voltage:

- 100 Vrms to 300 Vrms

For the APx555, eleven ranges are available. These are the same for both the unbalanced and balanced analog inputs:

- 0 Vrms to 310 mVrms
- 310 mVrms to 620 mVrms
- 620 mVrms to 1.25 Vrms
- 1.25 Vrms to 2.5 Vrms
- 2.5 Vrms to 5 Vrms
- 5 Vrms to 10 Vrms
- 10 Vrms to 20 Vrms
- 20 Vrms to 40 Vrms
- 40 Vrms to 80 Vrms
- 80 Vrms to 160 Vrms
- 160 Vrms to 320 Vrms

Choose the lowest range whose upper limit is greater than the expected maximum input signal. The easiest way to do this is to enter the expected maximum input level into the **Minimum Range** field in the **Advanced Settings > Input Range** dialog. The software will choose the lowest range that will accommodate that level.

If a signal greater then the **Minimum Range** is presented to the analog input, the ranging circuits will range up to accommodate the higher level, and then drop again when the signal level falls. The range will not be set below the **Minimum Range**.

Autoranging

Autoranging employs a circuit that measures the input voltage and automatically selects the correct range for the signal. As the signal level varies, the range will be switched up or down to keep the signal in the best range for measurement.

Autoranging works well for steady signals or signals whose levels change slowly. Most test stimulus signals are consistent, making autoranging a good choice for most testing.

When autoranging isn't suitable

However, signals that are very dynamic (such as tone bursts, music, voice or noise signals) may cause the autoranging circuits to switch too often for your test purposes. Signals with very fast pulses or spikes change too quickly for the autoranging circuit to react and may cause incorrect measurements due to ranging errors. Batch measurements that experience a range change during an acquisition will aborts, as will acquisitions being saved to file. POLQA, PESQ and other perceptual audio measurements use voice samples as test signals, and require the use of fixed ranging.

In these cases, evaluate the signal range carefully, or estimate the range if you must, and manually set the analyzer ranging to accommodate the input signal. See **Fixed Ranging**, beginning on the previous page.

Also, autoranging takes time. When the signal levels are known and a sequence would require many range changes, using fixed ranging will provide faster testing.

Ranges Track Ch1

If **Ranges Track Ch1** is checked (the default), the fixed input range setting for channel 1 is copied to the other channels, and range settings for channels greater than 1 cannot be edited. Any changes made to channel 1 are reproduced in the other channels. If **Ranges Track Ch1** is not checked, the fixed input range setting for each channel can be set independently.

Autoranging for Continuous Sweep and Frequency Response measurements

Measurements based on Continuous Sweep (Acoustic Response, Continuous Sweep, Frequency Response, Impedance/Theile-Small and Loudspeaker Production Test) use a modified autoranging process to find the correct input range.

For these measurements, the sweep acquisition begins in a particular range called the *range floor*, and moves up from there if necessary. By default, the range floor is the instrument's lowest range, but you can specify a higher range by entering the value in the **Min Range** field in the Advanced Settings dialog.

If the initial range is correct for the signal, the acquisition is made and the data are processed and displayed. If this range is too low, data from the out-of-range sweep are discarded. The input range is moved up and the sweep is repeated. This process may result in several sweep attempts before the correct range is determined; however, since the sweeps are very fast (typically one second) the total acquisition time is short.

If your sweep is a step in an automated sequence that you would like to run as fast as possible, you can optimize the speed of the ranging process by setting the range floor to the correct range for the measurement.

Settling Overview

APx500 uses settling mechanisms to obtain representative and consistent readings during sweeps and other measurements.

Practical devices will take a finite time to stabilize when an input signal changes, and at each sweep point a DUT will have an initial moment of instability. Also, practical audio analyzers require a minimum time to acquire sufficient data for proper analysis. These instabilities can vary with frequency, level, periodicity of signal and other factors. To provide consistent and accurate measurements, APx500 uses settling mechanisms to obtain stable, settled readings.

When settling is enabled, the analyzer pauses before reporting the result for each measurement or sweep point, providing time for DUT stabilization. The settling mechanisms attempt to make the pause as short as possible consistent with obtaining a settled reading. Since DUTs often take longer to stabilize when stimulated by a low frequency and acquisitions must be longer for sufficient low frequency resolution, you will notice that the sweep progresses more slowly through the low frequencies. This is due to settling.

APx500's settling mechanisms are optimized for fast, stable readings, but some of the settling parameters are available for adjustment. Depending upon the DUT, the nature of your measurement and your required results, you may find that adjusting the default settling parameters will give you faster or more consistent sweeps.

Settling in Sequence Mode

Measurements that can be settled

Signal to Noise Ratio and Crosstalk measurement are always settled, both for on-screen display and for reports. Stepped Sweeps are settled at each point, both for on-screen display and for reports.

Single value "meter" measurements in APx500 (Level and Gain, THD+N, Interchannel Phase, DC Level and the IMD measurements) appear on-screen as real-time measurements, unsettled.

When a meter measurement is started from the Navigator, (whether singly or as part of a sequence) these measurements are settled before the results are reported.

These defaults can be overridden. Settling can be defeated by unchecking the Enable checkbox and setting delays to zero in the Advanced Settings dialog.

Measurements that cannot be settled

Continuous Sweep measurements (including Frequency Response measurements) do not have settling available. Instead, the stimulus includes a pre-sweep signal to avoid transient-induced "pops" in the DUT.

The Signal Analyzer views are not settled. The Signal Analyzer FFT Spectrum view employs selectable averaging to reduce transient and non-periodic components in the results.

Adjusting Settling Parameters

Settling is normally enabled with default settings. These settings are correct for most measurements and do not require adjustment. Settling is applied when a report is generated, either singly or as part of an automated sequence.

If you would like to optimize the settling for speed or to accommodate a challenging signal, the **Settling** tab (or tabs) in **Advanced Settings** provide options.

Settling tabs

Some measurements in APx500 measure only one type of signal and require only one set of settling parameters. Others measure two or even signal types and have independent settling parameters for each. SNR is an example, where the signal and the noise are settled with different settings. Stepped Frequency sweeps have three settling tabs.

In the various Advanced Settings dialogs, one or more Settling tabs allow you to view and adjust the settling parameters. The illustration on page 626 shows the settling tabs and default settings for

1. Settling for Level measurements.
2. Settling for Interchannel Phase measurements.
3. Settling for SNR Signal measurements.
4. Settling for SNR Noise measurements.
5. Settling for Stepped Sweep Phase measurements
6. Settling for Stepped Sweep THD+N measurements.

In the explanations below, the default settings for each Settling tab are described.

Initial delay

To avoid spurious transient signals caused by DUT instability at the onset or change of the stimulus tone, the first moment of the acquisition is not measured. Measurement begins after the initial delay. The initial delay is calculated using the values entered in the **Delay Time** and **Delay Cycles** fields. The actual delay time is the greater of:

- the value set in **Delay Time**, or

  Delay time defines an initial measurement delay in seconds. The effect of this delay is the same at all frequencies.

  Minimum delay time is 0 s.

  Maximum delay time is 10 s.

  Defaults:
  20.00 ms for all measurements except SNR (Noise), which is 300.00 ms.

- the value set in **Delay Cycles** multiplied by the period of the generated stimulus signal.

  Delay cycles defines an initial measurement delay in number of cycles of the stimulus waveform. This setting produces a greater time delay at low frequencies.

  Minimum delay cycles is 0.

  Maximum delay cycles is 100.

  Defaults:
  3.000 cycles for all measurements except SNR (Noise), for which the Delay Cycles control is not available.

These settings interact to produce a frequency-dependent delay curve with an inflection at the point where **Delay Time** and the time calculated using **Delay Cycles** are the same. For SNR (Noise) the delay time is independent of frequency.

Initial delay settings are observed even if the **Enable Settling** checkbox is unchecked. Initial delay can be set to none by entering 0 in both fields.

Time and Cycles in Settling dialogs

For both the pre-settling Delay and the Flat algorithm Settling window, the calculation of total time from the values entered in the Time and Cycles fields follows the same rules:

The time is the greater of:

- the value set in Delay (or Settling) Time, or
- the value set in Delay (or Settling) Cycles value multiplied by the period of the generated signal.

In both cases, these settings produce intersecting curves, as shown in the diagram. With default values, this has the effect of lengthening the time at lower frequencies.

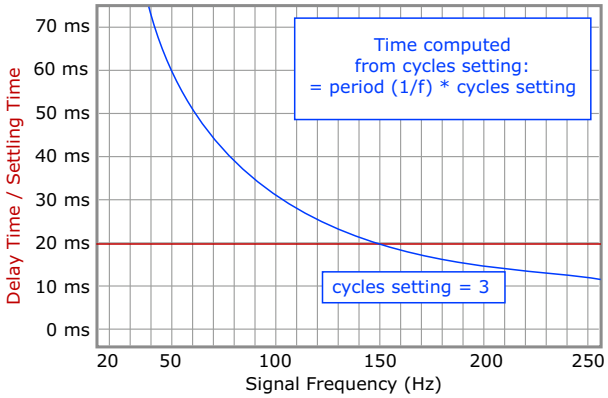


Illustration of both Initial delay and Settling delay calculation.

Example 1

The period of a 1 kHz sine wave is 1 ms. For a measurement at 1 kHz with typical Delay settings, the value of Time = 20 ms, and the calculated value of Cycles is 3 × 1 ms = 3 ms. Actual setting is the greater value, or 20 ms. The 3 cycles setting exceeds 20 ms at 150 Hz, which has a period of 6.667 ms.

Example 2

The period of a 100 Hz sine wave is 10 ms. For a measurement at 100 Hz with the same settings, the value of Time = 20 ms, and the calculated value of Cycles is 3 × 10 ms = 30 ms. Actual setting is the greater value, 30 ms.

Settling algorithms

After the initial delay, APx500 uses one of two settling algorithms. **Flat** is usually more suitable for periodic signals and is the default for most measurements. **Average** is sometimes more suitable for aperiodic (noise-dominated) signals. You can select either algorithm when available. **Enable Settling** must be checked to use a settling algorithm; if the box is unchecked, the signal will be measured directly after the initial **Delay Time / Delay Cycles** time has passed.

Defaults:
Flat is the default algorithm for all measurements except SNR (Noise), for which Average is the default algorithm.

The Flat Algorithm

The flat algorithm settings describe a time/level window within which a series of measurements must fall for the last point to be considered settled. The width of the window is the time calculated using the values entered in the **Settling Time** and **Settling Cycles** fields. The actual settling time is the greater of:

- the value set in Settling Time {default: 60.00 ms}, or

  In conjunction with Settling Cycles, Settling Time defines the width of the Flat algorithm time window in seconds. The effect of this setting is the same at all frequencies.

  Minimum settling time 0 s.

  Maximum settling time is 10 s.

  Defaults:
  60.00 ms for all measurements except the two Stepped Sweep (THD+N) tabs, which are 150.00 ms, and SNR (Noise) which is 80.00 ms, if Flat is selected.

- the value set in Settling Cycles multiplied by the period of the generated stimulus signal.

  In conjunction with Settling Time, Settling Cycles defines the width of the Flat algorithm time window in seconds. This setting produces a wider time window at low frequencies.

  Minimum settling cycles is 0.

  Maximum settling cycles is 100.

  Defaults:
  3.000 cycles for all measurements except the two Stepped Sweep (THD+N) tabs, which are 5.000 cycles, and SNR (Noise), for which the Settling Cycles control is not available.

These settings interact to produce a frequency-dependent delay curve with an inflection at the point where **Settling Time** and the time calculated using **Delay Cycles** are the same.

The height of the window is the instantaneous measured level multiplied by:

- twice the value set in Tolerance.

  As each acquired point is measured, the Flat algorithm window is redefined around its level. The height of the window is ±{Tolerance percentage}, and is always centered vertically above and below the measured value.

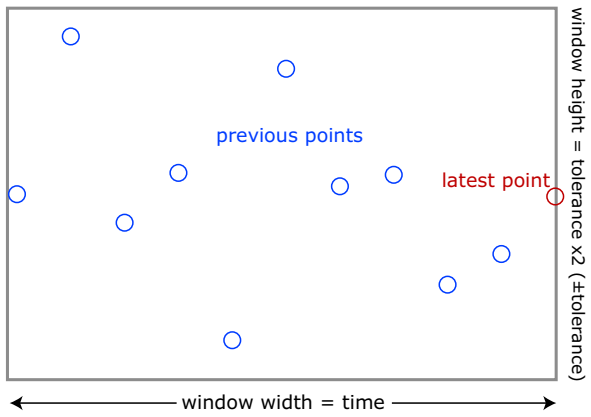
  Minimum tolerance percentage is ±0%.

  Maximum tolerance percentage is ±1000%.

  Defaults:
  1 % for all measurements except the two Stepped Sweep (THD+N) tabs, which are 5 %, and the SNR (Noise), for which the Flat Algorithm selection is not available.

Flat algorithm window

When using the Flat settling algorithm, it is useful to imagine a “window” that qualifies the latest point in a series of measurements as a “settled” measurement.



Flat algorithm settling window.

The Settling Time setting defines the width of the window in time. The current or latest point is shown at the right of the window. Previous points are shown to the left of the latest point. The number of points in the window depends on the measurement rate and the width of the window (Settling Time).

The Tolerance setting defines the height of the window. Since Tolerance is figured as a plus/minus percentage, a 1% Tolerance setting (±1%) defines a window height of 2%. The latest point is centered at 0%.

If any of the points in the window width (Settling Time) fall outside the window height (±Tolerance), the algorithm moves to the next measurement point for a new evaluation. When all the points in the window width are also within the window height, the measurement is considered settled and the latest point is reported as the measurement value.

If the flat algorithm cannot produce a settled result in a time of ten windows, the settling is considered failed and the data in the most recent time of five windows are averaged to produce a result.

The Settling Average Algorithm

This algorithm averages the data to be settled over a period of time to produce a result. The period is set by:

- the value set in Averaging Time {default: 200.0 ms}.

  Minimum averaging time is 0 s.
  Maximum averaging time is 10 s.

  Defaults:
  200.00 ms for all measurements except the SNR (Noise) tab, which is 300.00 ms.

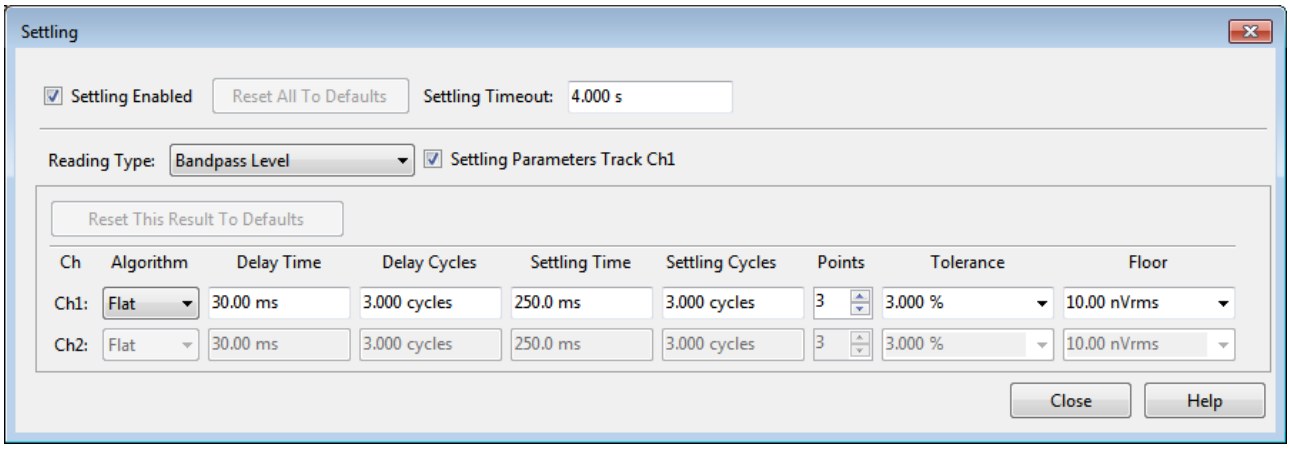
Reset Values to Defaults

Click **Reset Values to Defaults** to restore default settings.

Settling Dialog, Bench Mode

See Settling Overview on page 627, and More About Settling in Bench mode on page 631.

APx500 uses settling mechanisms to obtain representative and consistent readings during sweeps. In Bench Mode, Settling is applied at each step of a Sweep, and can optionally be applied at each reading in the Recorder. The Monitors/Meters, FFT, Continuous Sweep and Acoustic Response are always unsettled.



Settling Enabled

This checkbox enables or disables **Settling**.

Reset All to Defaults

This switch resets all settling parameters for all results to the default values.

Settling Timeout

This value sets the maximum time that APx will wait for a settled reading. If the reading is not settled within this time, the reading is marked with a “T” mark T (timeout) on the graph. The readings within the latest **Settling** window (see page 629) are averaged, and this value is used for the failed reading.

Reading Type

Select the **Reading** type that you'd like to examine from the menu. Each **Reading** type has been populated with default settling parameters appropriate for that **Reading**. You can edit the parameters for each **Reading** type (and for each channel, if **Settling Parameters Track Ch1** is not set). Parameters are remembered for each **Reading** type and channel.

Settling Parameters Track Ch1

When the **Settling Parameters Track Ch1** checkbox is set, all input channels are set to the same settling parameters for each result type. When **Levels Track Ch1** is unchecked, each channel's settling parameters can be set individually.

Reset to This Reading to Defaults

This button resets the parameters for the currently selected **Reading** type only.

Algorithm

Settling uses one of two algorithms. **Flat** is the primary method, and **Average** is an alternative for noisy signals.

Choose **None**, **Flat** or **Average**.

Flat

The Flat algorithm reports a data point when the signal is stable enough that the difference between any two consecutive readings is:

- less than plus or minus the percentage value specified in the **Tolerance** field (or, for very low level signals, less than plus or minus the absolute level value set in the **Floor** field), and
- that this degree of stability is held through the longer of
  - the length of time set in the **Settling Time** field, or
  - the length of time that is the sum of the periods of the number of cycles (at the current measured frequency) set in the **Settling Cycles** field, or
  - the number of readings specified in the **Points** field.

Average

The **Average** settling algorithm takes the mathematical average of the signal data points across the longer of

- the length of time set in the **Settling Time** field, or
- the length of time that is the sum of the periods of the number of cycles (at the current measured frequency) set in the **Settling Cycles** field, or
- the number of readings specified in the **Points** field.

Delay Time

Enter a **Delay Time** value here. **Delay Time** defines an initial measurement delay in seconds, and is one of the two parameters that determines the initial delay for each reading type. The initial delay is the sum of **Delay Time** and the time defined by **Delay Cycles**.

The length of **Delay Time** is the same at all frequencies.

- Minimum **Delay Time** is 0 s.
- Maximum **Delay Time** is 10 s.

Delay Cycles

Enter a **Delay Cycles** value here. **Delay Cycles** defines a length of time that is the sum of the periods of the number of cycles (at the current measured frequency) set in the **Delay Cycles** field. The initial delay is the sum of **Delay Time** and the time defined by **Delay Cycles**.

The length of **Delay Cycles** varies with frequency, with longer delays at lower frequencies.

- Minimum **Delay Cycles** is 0.
- Maximum **Delay Cycles** is 500.

Settling Time

Along with **Settling Cycles** and **Points**, **Settling Time** defines the width of the algorithm time window. The window width is the longer of **Settling Time** or **Settling Cycles**.

The width defined by **Settling Time** is the same for all frequencies.

- Minimum **Settling Time** is 0 s.
- Maximum **Settling Time** is 5 s.

Settling Cycles

Along with **Settling Time** and **Points**, **Settling Cycles** defines the width of the algorithm time window. The window width is the longer of **Settling Time** or **Settling Cycles**.

The width defined by **Settling Cycles** is the length of time that is the sum of the periods of the number of cycles set, at the current measured frequency. This causes settling to be slower, but to have fewer failures at low frequencies. If **Setting Cycles** is greater than zero, the **Settling** window is wider (settling for a longer time) at low frequencies.

- Minimum **Setting Cycles** is 0.
- Maximum **Setting Cycles** is 500.

Points

Enter the **Points** here. This defines the minimum number of data points to be evaluated for settling.

Tolerance

This field is available when **Algorithm** is **Flat**. This specifies the amount of variability you are willing to accept from test to test, as a percentage of signal level.

Floor

This field is available when **Algorithm** is **Flat**.

When working at the bottom of a measurement range, it can be difficult to get readings that satisfy the **Tolerance** setting, because **Tolerance** is a percentage and therefore relative to the signal level. When signal level is very small, level variations (largely due to noise) can consistently exceed the **Tolerance** and cause **Settling** to time out.

Floor helps avoid **Settling** timeout at low levels by establishing a threshold where the relative **Tolerance** evaluation is replaced by the absolute **Floor** evaluation, using ± the value set in **Floor** to set the settling range (the height of the **Settling** window.)

For **Reading Types** that are ratios (such as THD+N Ratio), the **Floor** values are also in ratio units. Many users find the per cent (%) ratio unit more intuitive to use in this application than the other available **Floor** ratio units, such as dB.

More about Settling in Bench Mode

Overview

Practical devices will take a finite time to stabilize when an input signal changes, and at each sweep point a DUT will have an initial moment of instability. Also, practical audio analyzers require a minimum time to acquire sufficient data for proper analysis. These instabilities can vary with frequency, level, periodicity of signal and other factors. To provide consistent and accurate measurements, APx500 uses settling mechanisms to obtain a stable, settled reading.

When settling is enabled, the analyzer pauses at each sweep point to allow for stabilization. The settling mechanisms attempt to make the pause as short as possible consistent with obtaining a settled reading. Since DUTs often take longer to stabilize when stimulated by a low frequency and acquisitions most be longer for sufficient low frequency resolution, you will

notice that the sweep progresses more slowly through the low frequencies. This is due to settling.

APx500's settling mechanisms are optimized for fast, stable readings, but the settling parameters are available for adjustment. Depending upon the DUT, the nature of your measurement and your required results, you may find that adjusting the default settling parameters will give you faster or more consistent sweeps.

Settling Concepts

Since much of the signal disruption occurs at the beginning of a sweep step because of instrument and DUT stabilization, the settling function is designed to:

- Ignore early readings, and
- Determine when a reading should be considered stable enough for data to be acquired.

Initial Delay

Settling in APx ignores early readings by providing initial delay settings. You can set initial delay by direct time entry and by cycles entry, which lengthens the delay at lower frequencies. Any measurements that arrive within the period between the beginning of a sweep step and the time set in the **Delay Time** field are discarded. Typical default **Delay Time** settings are in the 20 ms to 100 ms range, depending on the result selected.

Determination of stable data

APx uses one of two settling algorithms to determine when data is stable enough to measure.

Flat

Flat provides highly repeatable results and is the default algorithm selection. The Settling window is a useful aid to understanding the **Flat** algorithm.

The **Flat Algorithm** reports a data point when the signal is stable enough that the difference between any two consecutive readings is:

- less than plus or minus the percentage value specified in the **Tolerance** field (or, for very low level signals, less than plus or minus the absolute level value set in the **Floor** field), and
- that this degree of stability is held through the longer of
  - the length of time set in the **Settling Time** field, or
  - the length of time that is the sum of the periods of the number of cycles (at the current measured frequency) set in the **Settling Cycles** field, or
  - the number of readings specified in the **Points** field.

The **Flat** algorithm ensures that the signal variations have been settled to the specified **Tolerance** (or **Floor**) for some time.

Average

Average is useful for noise measurements and for other random or aperiodic signals. The **Average** algorithm is particularly useful when the signal is fundamentally noisy and might never settle within a practical tolerance.

The **Average** settling algorithm takes the mathematical average of the signal data points across the longer of

- the length of time set in the **Settling Time** field, or
- the length of time that is the sum of the periods of the number of cycles (at the current measured frequency) set in the **Settling Cycles** field, or
- the number of readings specified in the **Points** field.

The **Average** algorithm is particularly useful when the signal is fundamentally noisy and might never settle within a practical tolerance.

Different measurements and readings may require different settling settings for fast measurement and consistent results. APx provides default settings for a number of reading types, each of which can be customized.

Settling Tolerance

Settling Tolerance represents the amount of variability you are willing to accept from test to test. A tolerance of 0.1% (about 0.01 dB) may be appropriate for frequency response tests on a DUT with excellent characteristics; at the other extreme, tolerances of 10% to 25% may be necessary to obtain data under noisy conditions. The default value of 1% is a good starting place.

Settling Floor

When working near the very bottom of the dynamic range for a particular APx measurement, the difference between two adjacent values or quantization levels can be large compared to the criteria in the settling algorithm. At levels greater than a few percent of full scale, the value set in the Tolerance field is normally the determining parameter, but close to the bottom of a measurable range another technique is needed to keep the settling from slowing down the sweep or even stopping it entirely.

The **Floor** setting provides this protection. Whenever the value set in the **Floor** field is larger than the product of the measured signal multiplied by the **Tolerance** percentage, the algorithm uses the Floor value instead of waiting for the signal to satisfy the **Tolerance** criteria.

For example, with a **Tolerance** of 1% and a **Floor** setting of 1 µV, the **Floor** takes over when the measured signal drops below 100 µV. When **Floor** is in effect, settling is satisfied when three adjacent points are within the absolute value of the level set in the **Floor** field.

The default values of **Floor** for each meter are chosen to be approximately the resolution of that meter on its most sensitive range. Since resolution increases as the reading rate slows, it may be appropriate to change the default **Floor** values when the reading rate is fixed at a specific value.

Settling Timeout

It is possible with extremely noisy signals that the instability of the measured parameter is consistently greater than the **Tolerance** value entered on the **Settling** panel, and that it is impossible to satisfy the **Settling** algorithm and get a settled reading.

The **Timeout** control exists to keep your sweep from running slowly or even locking up under these conditions. If a settled reading cannot be made within the period of time entered in the **Timeout** field, the reading is marked with a "T" mark T (timeout) on the graph. APx will average the readings within the latest Settling window, enter that value as a data point and move on to the next sweep step.

The default **Timeout** value (per step) is 4.000 sec.

See Time and Cycles in Settling dialogs on page 628, and Flat Algorithm window on page 629.

99

Results: Meters, Graphs, and Tables

Overview

APx500 measurements have one or more primary results. You can also specify Derived results, User Defined results, and Imported results. Results are viewed in Result Displays.

Primary results

Each measurement has one or more *primary* results associated with it. A primary result is the direct result of a measurement; a primary Level result, for example, is the measured amplitude of the audio signal. Most measurements have several primary results associated with them. Frequency Response, for example, has four: Level, Gain, Relative Level and Deviation.

A new measurement initially presents its complete set of primary results in the Navigator tree and the Selector filmstrip.

A batch measurement (page 184) primary result initially has one *Data Set*, called **Measured1**. This is the data gathered from measuring the acquisition. For measurements and results that support appending, multiple Data Sets can be appended to primary results and are called **Measured2**, **Measured3** and so on. Meter measurements (page 182) do not support appending measurement data.

Derived results

A measurement can also have one or more user-defined Derived results. A Derived result is a data point or set of data points derived from an existing result (called the source result) by a mathematical computation such as averaging or normalizing. Derived results can be attached to primary results, imported results or existing Derived results.

See Chapter 100 for more information about Derived Results.

User Defined results

You can define a new result and add it to a measurement. The data for a Defined result is read from a file on disk at measurement run time, or when the (Refresh) button is clicked. See page 652 for more information about Defined results.

Imported data

Data can be imported into an XY Graph result from a file on disk. The data may have been exported from an APx measurement, or saved from another application, or created as an arbitrary list of values. Any compatible data can be imported into a result to form a new Data Set for that result. Imported Data Sets are called **Imported1**, **Imported2** and so on. Bar Graph results do not support importing data from a file on disk.

See page 650 for more information about importing result data.

Result Displays

Bar Graphs/Meters

Meter measurements (page 182) have only one result parameter per channel. Meters typically show results as Bar Graph displays (also called Meters). Meters also display numerical results on the Bar Graph and in the Graph Data grid. As real-time measurements, meter measurement Bar Graphs and numerical displays are constantly updating.

Some batch measurements have one or more single-parameter results, which are displayed as Bar Graphs. See more about Bar Graphs starting on page 637.

XY Graphs

Most results of batch measurements (page 184) have two parameters per channel (level vs. frequency, for example) and are displayed as XY Graphs. The data being graphed can be viewed as numerical results in the Graph Data grid. See more about XY Graphs starting on page 641.

Other result displays

Some results of a few APx measurements (Metadata Recorder, POLQA, PESQ and Impedance/Theile-Small are examples) do not use either Bar Graphs or XY Graphs. Displays for such measurements are discussed in the individual measurement chapters.

Lists and tables

The graphic displays for both Bar Graphs and XY Graphs are generated from underlying data points. You can view and export a list (for Meter results) or a table (for XY or other results) containing the data values. See Graph Data on page 649.

Working with results

Navigating results

Results are represented as branch nodes on the Navigator tree display, and as thumbnails in the Selector filmstrip display. The currently selected result is displayed in the Graph panel.

Primary or derived results can be added to or deleted from a measurement, and can be renamed to suit your needs. Multiple instances of a result can be added to a measurement.

Adding, deleting or renaming results

Add, delete or rename a primary or a derived result to a measurement using any of these methods:

- **Navigator**
  Right-click on a result and choose **Add Primary Result**, **Add Derived Result, Delete Result** or **Rename**.
- **Selector toolbar**
  Select a result and click **Add** or **Delete** in the Selector toolbar.
- **Selector filmstrip**
  Right-click on a result and choose **Add Derived Result, Delete** or **Rename**.
- **Selector filmstrip**
  Right-click between results and choose **Add Primary Result**.
- **Project menu**
  Select a result in the Navigator or the Selector, and from the main Menu bar choose **Project > Add Primary Result**, **Project > Add Derived Result** or **Project > Delete Result**.
- **Graph display**
  Right click in the large Graph result display and choose **Add Derived Result**, or click the **Add Derived Result** button in the Graph toolbar.

Selecting a result

To select a result, click on the result branch in the Navigator, or on the result thumbnail in the Selector.

In addition, you can select an existing derived result by first selecting the source result and then choosing Go to Derived from the Project menu or from the context menus available in the Navigator, the Selector or the Graph.

You can also select a source result by first selecting the derived result and then choosing Go to Source from the Project menu or from the context menus available in the Navigator, the Selector or the Graph.

Viewing a result display

View a result display in the Graph panel by selecting the result.

Viewing result details

To view result details, select the result and click Details in the Selector toolbar. This is particularly useful when working with Derived Results, as it reveals the relationships between source and derived results. See page 648.

Moving a result

In the Selector, grab a result thumbnail with the mouse cursor and drag it to a new position in the Selector. The result positions in the Selector determines the relative result locations in a report.

Printing or exporting a result

You can print or export a single result by selecting the result, then clicking Print/Export this Result. Such results are always unsettled. See Settling on page 627.

Including a result in a sequence

Any result that is checked in the Navigator, and whose source measurement is checked, will be included in a sequence.

Including a result in a Sequence report

If Report is checked in the Navigator, any result that is included in a sequence will be included in the sequence report.

You can also run an individual measurement by right-clicking on the measurement in the Navigator and choosing Start Selected Measurement from the context menu. If Report is checked in the Navigator, any result that is checked for that measurement will be included in the measurement report. If Settling is available for the result, it is applied.

"----" (invalid) results

Some configurations or results are invalid, and tables or graphs that display such a result show four dashes "----" followed by the current unit symbol.

Such a result is not a number. It represents an invalid condition for measurement, or no data.

For example, in phase measurements one channel is selected as a reference channel. As such, it has no numerical value, and its result is displayed as "---- deg". A reference channel is ignored for pass/fail evaluation and for derived results computations.

In another example, a THD+N Ratio measurement with no input provides nothing for the distortion ratio algorithm to work with. The THD+N Ratio result for a channel with no input is "----%". Such a result will cause a failure when compared against limits.

Similarly, an unlocked digital input has no valid audio result. The result is displayed as "----dBFS".

Derived results attached to "----" results

When a derived result is attached to a result that includes a reference channel (such as a phase result), the reference channel is ignored in the computations.

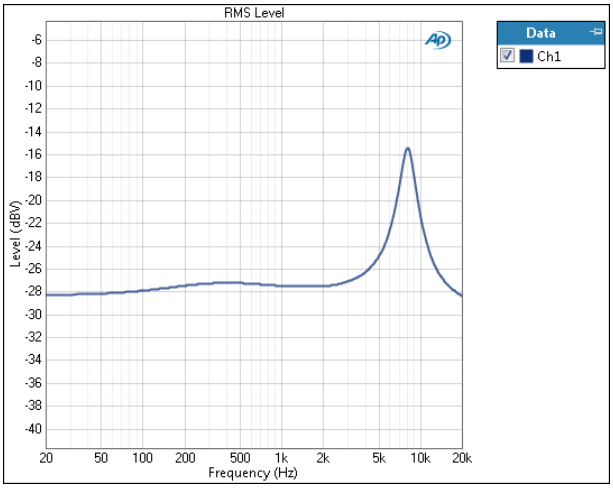
When a derived result is attached to a result that includes one or more channels with "----" results, and one or more of such channels is included in the result specification, the derived result is "----".

Introduction to Result displays

In APx500, measurement results are shown as meter Bar Graphs, as XY graphs or as values in a table or grid. These results can be viewed on-screen, printed or output as data. Results can be also compared to *limits,* so that you can evaluate the performance of your DUT against a performance requirement. This chapter looks at the features of each type of display, the properties you can set for each display, and how to apply limits to your results.

Meter displays show a one-parameter Bar Graph result for each channel. An example is a simple level measurement, as shown here. Channel 1 is measured at 0 dBV.

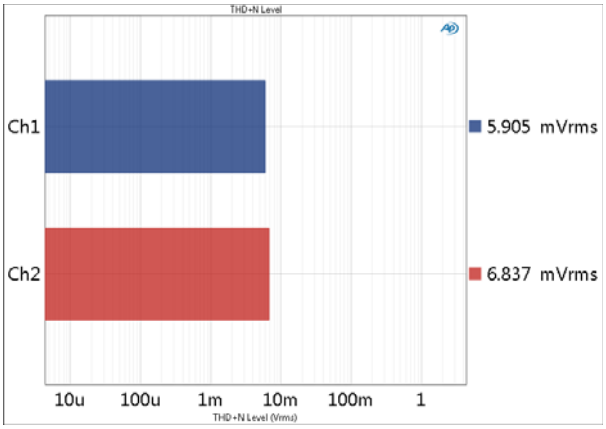
Graphs show the relationship between two parameters for each channel. The example below shows the level of Channel 1 as it varies by frequency. The DUT has a 12 dB peak in the response, centered at about 6.5 kHz.



Tables and grids can show one, two or more parameters per channel, with values for each parameter arranged in a column like a spreadsheet. You'll find grid displays in the Signal Monitor view and in the Reference Levels dBr A and B settings. The data behind each meter bar and Graph display can also be viewed as a table. The table to the right shows part of the data for the frequency response Graph above.

Bar Graph (Meter) Displays

Bar Graph (Meter) displays are used for those measurements that measure only one parameter per channel, providing a clear view of this parameter across many channels. APx500 also uses a vertical Bar Graph display for distortion product measurements, where several parameters are displayed but the view is limited to one channel.



The Bar Graph Panel

The Bar Graph Panel contains the Bar Graph and the Bar Graph toolbar. Below the Bar Graph Panel is the Selector filmstrip, where you can choose which result to view, add results, etc.

Result Name

The name of the result is at the top of the Graph. You can change the result name in the Graph Properties dialog, in the Selector or in the Navigator/Sequencer.

Measurement Timestamp

The current date and time are shown at the top right of the Graph. You can enable or disable the timestamp in the Project/Sequence Properties dialog.

Zoom

You can zoom in and out using the scroll wheel on your mouse; if you do not have a scroll control, you can hold the **Ctrl** button while dragging the mouse pointer left or right in the Graph area or on the X-axis scale. Click **Set Zoom/Pan to Original** in the toolbar or context menu to return the Graph to its original size.

Pan

Click anywhere in the Graph or in the X-Axis scale and drag to move the Graph image.

X-Axis

Click on the chevron icon (next to the X-axis title) to open a menu that provides access to **Units** and **Log/Linear** settings, **Autoscale On/Off** and the **Graph Properties** dialog.

Bar Graph toolbar



The Bar Graph (Meter) toolbar extends across the top of the Graph Panel. The commands in the toolbar are also available in the context menu invoked by right-clicking in the Graph area.

Save Graph image to disk



Click this button to save the meter Bar Graph to disk as an image file. A file browser window opens to enable you to name the file and browse to a folder. Supported file formats are

- .bmp
- .jpg
- .emf
- .png
- .pdf

This image can then be imported into another application. If the data on screen is unsettled, the image that is saved is also unsettled.

Copy Graph image to clipboard



Click this button to copy the meter Bar Graph to the Windows clipboard as a bitmap image. This image can then be pasted into another application. If the data on screen is unsettled, the image that is copied is also unsettled.

Print/Export Results for this view



Click this button to print or export the results of the current measurement. If limits are set, the results are compared to the limits.

Zoom in



Click this button and then drag the mouse pointer across the Graph to define an area. When you release the mouse button, the display will zoom to that area.

Zoom out to original



Click this button to return a zoomed or panned display to its original magnification and location.

Fit view to data



This control autoscales the graph to the current data, at the moment the **Fit view to data** button is clicked.

Show Graph data



Click this button to open a data view window that displays the meter data. From this window, you can copy the data to the Windows clipboard as text, or export the data as a Microsoft Excel spreadsheet file or CSV (comma separated value) text file.

Draw/Edit Limits



This button opens a menu where you can select one of the two following choices:



Draw Limits

Click this button to open the Draw Limits dialog.



Edit Limits

Click this button to open the Edit Limits dialog. See page 671 for information about limits.

Edit Graph Properties

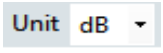


Click this button to open the Edit Graph Properties dialog. See page 639 for information about editing meter Bar Graph properties.

(Pan)

To pan (slide the X axis of the Graph without changing the current magnification), click and drag anywhere in the Graph, or click and drag in the X-axis scale.

Units



A measurement produces a result whose value can be expressed within a certain family of measurement units appropriate to that parameter, so the list of unit choices available on the meter bar display is determined by the type of measurement.

Choose a measurement unit for your display from the Units drop-down list. Some units are absolute; others calculate values in reference to values set in reference fields in APx500.

See Chapter 104 for more information about units of measurement.

Undock and Dock



At the right of the Bar Graph toolbar is the Undock button. This places the meters in a new window (undocks them) that can be moved and resized. When a meter display is undocked, clicking the Dock button will return it to its fixed location in the workspace.

Additional features on the context menu

Right-click in the Graph area to open a context menu. The commands on the toolbar are also available here, with these additional commands:

Edit annotations

You can add text annotations to the Graph display. These annotations are included with the Graph in the Report and when copying or saving the Graph image.

Right-click on the Graph, and select choose **Edit Annotations** from the context menu. This opens the Annotation Toolbox.



To add an annotation, click the button in the Annotation Toolbox, and drag the mouse cursor to create a box on the Graph. Using the Toolbar, choose font size, style and color. Use the **Fill Color** tool to color the box background. Type in the box to enter text for the annotation.

To edit an existing annotation, open the Annotation Toolbox and select the annotation box. Change the font style, size or color in the toolbar. Select and edit the type in the box.

To move the text annotation box, click on the box frame and drag the box to a new position.

To resize the text annotation box, click on a box corner and drag to resize the box.

To delete a text annotation, open the Annotation Toolbox, select the annotation box to be deleted, and click the **Delete Annotation** button.

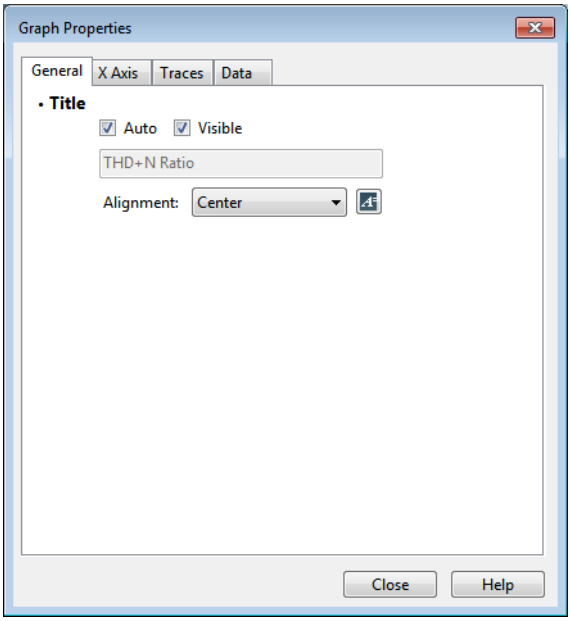
Autoscale

Autoscale is a shortcut that sets the Graph Properties **Range** checkbox to **Auto**. Right-click on the Graph, then select **Autoscale > X Axis** from the context menu.

Meter Bars: Edit Graph Properties

The Graph Properties dialog allows you to edit the meter Bar Graph title, X-Axis title and range.

General tab



Title

For most results, when **Auto** is checked the default name for the measurement result (RMS Level, for example) is entered into the Title field. For a Defined Result, when **Auto** is checked the Title string from the referenced file is entered.

When **Auto** is unchecked, you can enter any title into the Title field.

Visible

When **Visible** is checked the title is applied to the Graph.

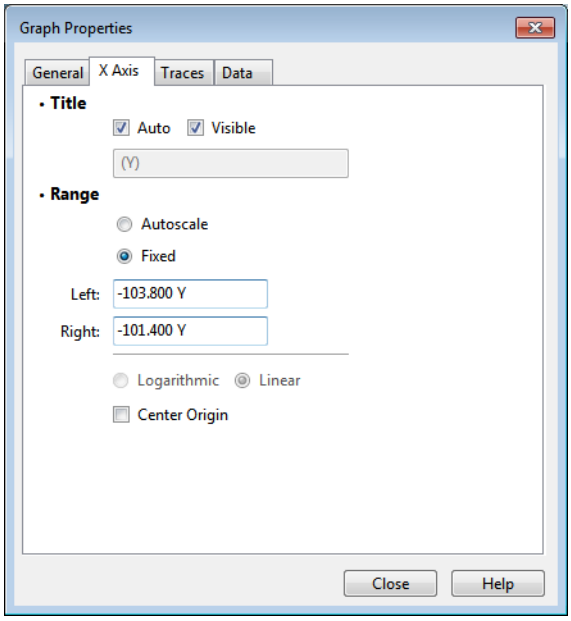
Alignment

Select Left, Center or Right title alignment.

Font...

The **Font...** button opens the Windows font browser. You can modify the title font family, style, size, color and so on from this dialog. Default font is MS Sans Serif 8 point.

X-Axis tab



Title

When Auto is not checked, you can enter a title for the X-Axis in the Title field.

Auto

When Auto is checked, the X-Axis label shows the parameter and the units (Level [dBV], for example).

Visible

When Visible is checked the X-Axis title is applied to the Graph.

Range

Range sets the values at the left and right edges of the Graph. Enter values in the Left and Right range fields. You can also select units from the range field drop-down menus.

Autoscale

Autoscale sets the range around the current signal characteristics.

Logarithmic

When Logarithmic is checked, the X-Axis scale follows a logarithmic rule.

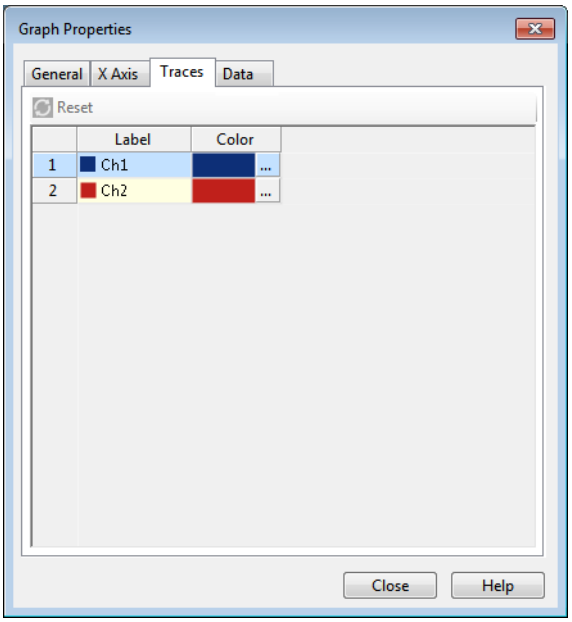
Linear

When Linear is checked, the X-Axis scale follows a linear rule.

Center Origin

For a Bar Graph Defined Result, you can specify whether or not the Graph origin (0 point) is centered.

Traces tab



Open the **Graph Properties** dialog and go to the **Traces** tab. By default, the settings are derived from the global setting made in the **Labels** dialog, accessed from **Signal Path Setup**, and from the global settings made in the **Colors** tab, accessed from the **Project/ Sequence Properties** dialog. In the **Traces** tab you can edit the local settings for each Bar Graph channel, overriding the global settings.

Label

Enter a new **Label** for the channel in this field. This setting is local and will only affect this result.

Color

Click on the button to open a color picker. Choose a new **Color** for this channel. This setting is local and will only affect this result.

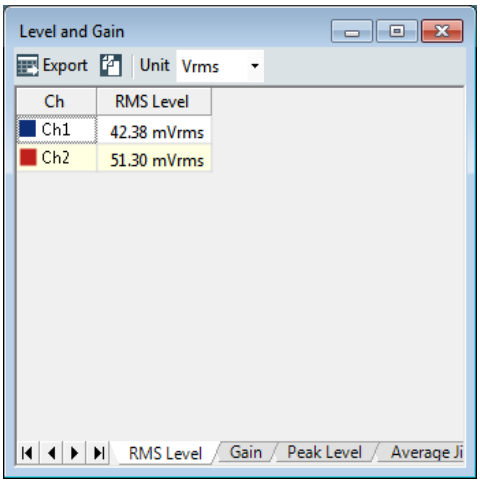
Alternatively, you can right click in a displayed value in a meter result (Bar Graph display) to open a context menu and edit the **Label** and **Color** for that channel.

For more information about setting global labels, see page 56. For information about global colors and line styles, see page 32.

Data Tab

The Data Tab is used for User Defined Results. See page 652.

Graph data for meters



The data window for meters shows the values for each meter bar.

Tabbed Pages

For measurements that have more than one result view, tabs at the bottom of the data grid give you access to the Graph data for the other results.

Units

For each page, you can select units that differ from the initial Graph units.

Export

Click the Export button to save the data for that page as a Microsoft Excel spreadsheet file (*.xls) or as a comma-separated-value (*.csv) text file.

Copy to clipboard

Click the Copy button to copy the data to the Windows clipboard as tab-delimited text. This data can be pasted into other applications such as Microsoft Word or MS Excel.

XY Graphs

APx500 graphs display two (and occasionally three) parameters of a signal measurement on an XY grid.

In a frequency domain display, these parameters are typically level versus frequency, with level on the Y axis (vertical, left), and frequency on the X axis (horizontal, bottom). In a time domain display, these parameters are typically level versus time, with level on the Y axis and time on the X axis. Other relationships can be graphed: phase angle versus frequency, DUT output level versus generator level (linearity) and so on.

Dual axis graphs

An example of a three parameter Graph is the Rub and Buzz result in the Acoustic Response measurement. Crest Factor is on the Y axis (vertical, left), Peak Ratio is on the second Y axis (vertical, right) and frequency is on the X axis (horizontal, bottom). Read more about Dual Axis graphs starting on page 645.

The XY Graph Panel

The XY Graph Panel contains the XY Graph and the XY Graph toolbar. Below the XY Graph Panel is the Selector filmstrip, where you can choose which result to view, add results, etc.

Result Name

The name of the result is at the top of the Graph. You can change the result name in the Graph Properties dialog, in the Selector or in the Navigator/Sequencer.

Measurement Timestamp

The current date and time are shown at the top right of the Graph. You can enable or disable the timestamp in the Project/Sequence Properties dialog.

Zoom

You can zoom in and out by hovering the mouse pointer in the Graph area and using the scroll wheel on your mouse; if you do not have a scroll wheel, you can hold the **Ctrl** button while dragging the mouse pointer in the Graph area or on an X- or Y-axis scale, creating a marquee area. When you release the mouse, the Graph will zoom to the marquee area. Click **Set Zoom/Pan to Original** in the toolbar or context menu to return the Graph to its original size.

Pan

Click anywhere in the Graph or in an X- or Y-axis scale. The mouse pointer will take the shape of a hand; drag to move the Graph image. Click **Set Zoom/Pan to Original** in the toolbar or context menu to return the Graph to its original position.

X-axis and Y-axis

By default, there is only one active Y-axis (the left axis) on an XY Graph. You can add a second set of data that share the same X-axis but are plotted to a second Y-axis, on the right. Click the chevron icon on the right side of the Graph to define the right Y-axis. Read more about Graphs with two Y-axes starting on page 645.

For an active axis, click the chevron icon (next to an axis title) to open a menu that provides access to **Units** and **Log/Linear** settings, **Autoscale On/Off** and the **Graph Properties** dialog.

Graph toolbar



Note: The following commands are also available by right-clicking in the Graph display and choosing the command from the right-click menu.

Save Graph image to disk



Click this button to save the Graph to disk as an image file. A file browser window opens to enable you to name the file and brows to a folder. Supported file formats are

- .bmp
- .jpg
- .emf
- .png
- .pdf

This image can then be imported into another application.

Copy Graph image to clipboard



Click this button to copy the Graph to the Windows clipboard as a bitmap image. This image can then be pasted into another application.

Print/Export Results for this View



Click this button to print or export the results of the current measurement. If limits are set, the results are compared to the limits.

Zoom in



The Zoom control gives you the ability to enlarge an area of the Graph. Click the Zoom button, and the mouse pointer becomes crosshairs (a cross).

Horizontal and vertical zoom

Drag the crosshairs across the Graph to define a marquee area for enlargement. Release the mouse button and the Graph will zoom to the area defined.

You can also zoom by holding the **Ctrl** button down and dragging a marquee in the Graph.

Horizontal zoom only

Hold the **Ctrl** button while you drag the crosshairs, and the marquee will define an area for horizontal enlargement.

You can also zoom horizontally by hovering the mouse pointer over the X-axis scale and using the scroll control on the mouse; if you do not have a scroll control, you can hold the **Ctrl** button while dragging the mouse pointer left or right on the X-axis scale.

Vertical zoom only

Hold the **Shift** button while you drag the crosshairs, and the marquee will define an area for vertical enlargement.

You can also zoom vertically by hovering the mouse pointer over a Y-axis scale and using the scroll control on the mouse; if you do not have a scroll control, you can hold the **Ctrl** button while dragging the mouse pointer up or down a Y-axis scale.

Zoom out to original



Click this button to return a zoomed display to the original magnification and panning position.

You can also return to the original by double-clicking in the Graph area.

Fit view to data



This control autoscales the graph to the current data, at the moment the **Fit view to data** button is clicked.

(Pan)

To pan (slide the X and Y axes of the Graph without changing the current magnification), click and drag anywhere in the Graph. To pan horizontally, click and drag in the X-axis scale. To pan vertically, click and drag in a Y-axis scale.

Show/Hide Cursors



Click this button to show or hide the Graph cursors and the associated cursor legend panel. See Cursors on page 644 for detailed information.

Show Graph data



Click this button to open a data view window that displays the Graph data. From this window, you can copy the data to the Windows clipboard as text. You can also export the underlying Graph result data as an Excel spreadsheet (*.xls), a comma separated value text file (*.csv) or a Matlab file (*.mat).

Draw/Edit Limits



This button opens a menu where you can select one of the two following choices:



Draw Limits

Click this button to open the Draw Limits dialog.



Edit Limits

Click this button to open the Edit Limits dialog. See page 671 for information about limits.

Edit Graph Properties



Click this button to open the Edit Graph Properties dialog.

Units



A measurement produces a result whose value can be expressed within a certain family of measurement units appropriate to that parameter, so the list of unit choices available on the meter bar display is determined by the type of measurement.

Choose a measurement unit for your display from the Units drop-down list. Some units are absolute; others calculate values in reference to values set in reference fields in APx500.

Undock / Dock



At the right end of the Graph toolbar is the **Undock** button. This sets the current Graph in a new window that can be moved and resized. In the new window, click the **Dock** button to return the Graph to its docked location.

Additional features

Edit Annotations

You can add text annotations to the Graph display. These annotations are included with the Graph in the Report and when copying or saving the Graph image.

Right-click on the Graph, and select choose **Edit Annotations** from the context menu. This opens the Annotation Toolbox.



To add an annotation, click the button in the Annotation Toolbox, and drag the mouse cursor to create a box on the Graph. Using the Toolbar, choose font size, style and color. Use the **Fill Color** tool to color the box background. Type in the box to enter text for the annotation.

To edit an existing annotation, open the Annotation Toolbox and select the annotation box. Change the font style, size or color in the toolbar. Select and edit the type in the box.

To move the text annotation box, click on the box frame and drag the box to a new position.

To resize the text annotation box, click on a box corner and drag to resize the box.

To delete a text annotation, open the Annotation Toolbox, select the annotation box to be deleted, and click the **Delete Annotation** button.

Autoscale

For the X-axis or the Y-axis, or both, autoscale causes the Graph scale to automatically expand or contract to fit the range of the measurement. These selections have the same functions as the **Auto** checkboxes in the Graph Properties dialog.

Graph Legend

The Graph Legend controls which data are visible on the Graph, and how the data are displayed.

Channel identification conventions

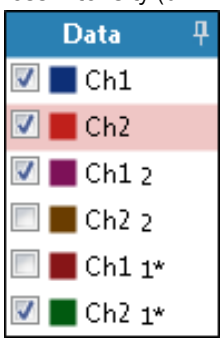
For a single acquisition, the Legend will show an entry for each channel. By default, the channel names take the form of Ch1, Ch2 and so on. If channels have been renamed, the new name is displayed.

If data is appended or imported, the Legend will show entries for each channel of each data set. Appended data are identified by a number in subscript, corresponding to the “Measured n” number automatically assigned to the appended data; imported data are similarly identified by a number corresponding to the “Imported n” number. Imported data channels are also marked with an asterisk (*) to differentiate them from appended measurements.

If the Data Set selector is set to **All Data**, the Legend will show all channels of all data sets.

Using the Legend to set trace visibility

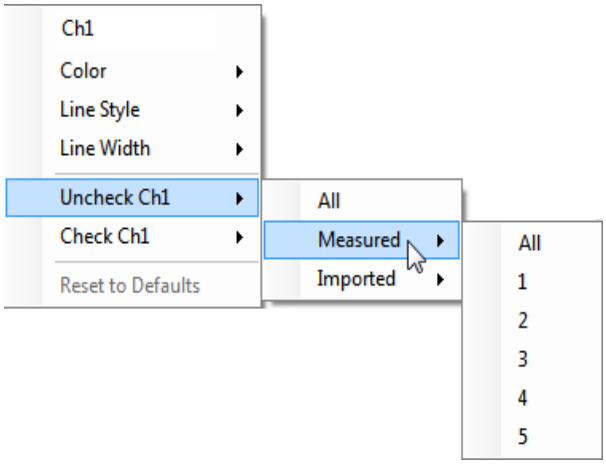
If you click on a Legend entry, the entry will be highlighted in the Legend. Using **Ctrl+Click**, you can highlight more than one entry. On the Graph, the traces that correspond to highlighted Legend entries remain full visible, while any other traces are displayed with less intensity (dimmed).



Each Legend entry has a checkbox. If the checkbox is checked, the trace will be visible on the Graph. If the checkbox is not checked, the trace will not be visible on the Graph.

To quickly set visibility on or off for a channel across all measured or imported data sets, right-click on a

Legend entry and choose **Check Ch n** or **Uncheck Ch n**, and the choose **All**, or **Measured**, or **Imported**.



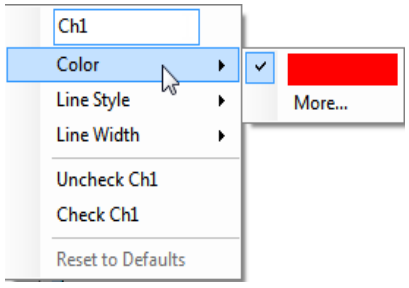
Managing the Legend

With large channel counts and many data sets, the Legend list can grow quite long. When the Legend list grows too long for the display area, scroll arrows will appear at the top and bottom of the Legend display to allow you to move through the Legend entries.

If a channel name is too long for the space allocated to the Legend display, you can see the full channel name in the tool tip that appears when you hover the mouse cursor over the Legend entry. Alternatively, you can drag the bar separating the Graph from the Legend to the left. This will reduce the Graph display area and enlarge the Legend display area.

Using the pin control at the top of the Legend display, you can set the display to show both checked and unchecked entries, or to only show checked entries. When the pin is set to show only checked (visible) entries, hovering over the Legend display with the mouse cursor will temporarily show the entire list.

Using the Legend to set local trace colors and styles



You can set local (measurement result) trace names, colors and styles using the Legend display. To set global (project-wide) names, colors and styles, see “Setting Global Graph Colors and Styles” on page 32.

To set local (measurement result) trace names, colors and styles, right click on a Legend entry to open this context menu.

You can change the channel name by typing in the name field. Click on **Color**, **Line Style** or **Line Width** to set the trace style.

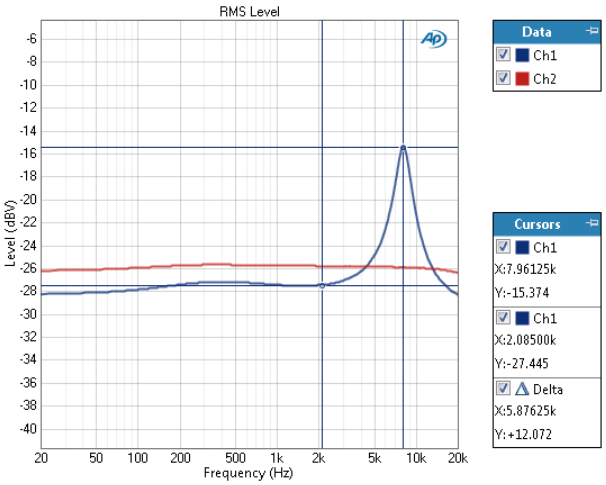
Cursors



You can add one or two cursors to an XY Graph display to point to and evaluate points of interest on the traces. An optional Delta result will display the difference between the two cursors scaled in the units chosen for the Graph.

Click the **Show/Hide Cursors** button on the Graph toolbar or on the context menu that appears when you right-click on the Graph. This is a “toggle” switch that will show or hide the cursors and the cursor legend.

When the cursors are displayed, the cursor legend also appears to the right of the Graph, under the Graph legend.



Cursors: attached and unattached

Cursors can be independent of the Graph traces (unattached), or can be attached to any available trace. By default, both cursors are attached to the Channel 1 trace, as shown. When a cursor is unattached, you can move it to any point on the Graph by grabbing either of the cursor graticules or their point of intersection and dragging the cursor. When a cursor is attached to a trace, you can only grab the vertical graticule and drag the cursor in a horizontal direction.

Cursor Legend

Managing the Legend

If a channel name is too long for the space allocated to the Legend display, you can see the full channel name in the tool tip that appears when you hover the mouse pointer over the Legend entry. Alternatively, you can drag the bar separating the Graph from the Legend to the left. This will reduce the Graph display area and enlarge the Legend display area.

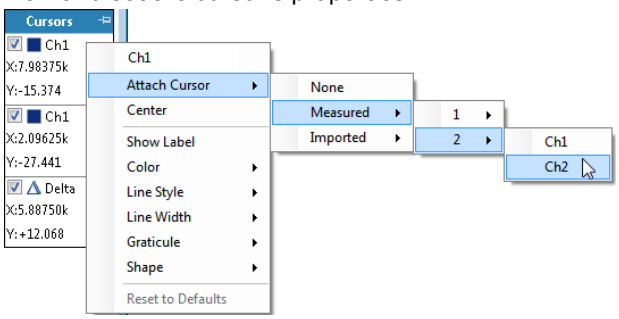
Using the pin control at the top of the Legend display, you can set the display to show both checked and unchecked entries, or to only show checked entries. When the pin is set to show only checked (visible) entries, hovering over the Legend display with the mouse pointer will temporarily show the entire list.

Setting cursor visibility

Each Legend entry has a checkbox. If the checkbox is checked, the cursor (or Delta result) will be visible on the Graph and included in the report. If the checkbox is not checked, the cursor (or Delta result) will not be visible on the Graph nor be included in the report.

Opening the cursor properties context menu

In the cursor legend, right-click in one of the cursor areas. A flyout context menu will open, allowing you to view and set the cursor's properties.



Labeling a cursor

From the cursor's properties context menu, click on the cursor's label (the first item in the menu). The label will become an editable field where you can modify the cursor name. By default, the cursor labels take the form of Cursor1(Ch1) and so on.

Attaching a cursor to a trace

From the cursor's properties context menu, go to **Attach Cursor >** and select the Graph trace you would like to attach the cursor to, or **None**. If data is appended or imported, the menu will show entries for each channel of each data set. Acquired and appended data are grouped in "Measured" submenu, and imported data are similarly grouped in the "Imported" submenu.

Centering a cursor

Sometimes you may zoom or pan to a portion of the Graph that does not include the cursor. From the cursor's properties context menu, choose **Center** to bring the cursor to the center of the current Graph view.

Setting cursor colors and styles

Show label

From the cursor's properties context menu, choose **Show Label**. This will toggle the cursor label visibility on the Graph display.

Color, Line Style, Line Width

From the cursor's properties context menu, you can set cursor names, colors and styles.

Graticule

The lines which define the cursor's position are the graticules. From the cursor's properties context menu, you can set a cursor's horizontal or vertical graticules to **None**, **Short** or **Long**.

Shape

By default, the intersection of a cursor's graticules is marked with a small circle.

From the cursor's properties context menu, you can choose other shapes to indicate the intersection of the graticules.

Dual Axis Graphs

You can add a second Y-axis to APx XY Graphs, enabling simultaneous viewing of two sets of data that share a common X-axis.

- View the same data with two different units: Vrms and dBV, for example.
- View two different primary results: Level and THD+N, for example.
- View a primary result and a Derived Result on the same graph.

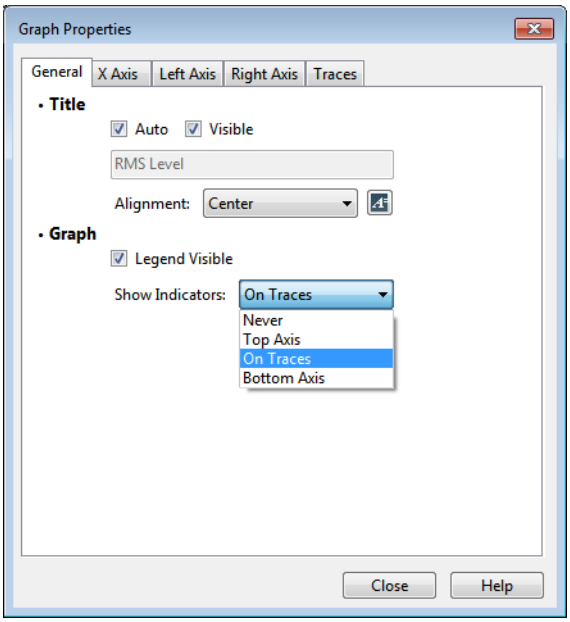
The Acoustic Response / Loudspeaker Test Rub and Buzz result uses this feature to plot the crest factor against the left axis and the peak ratio against the right axis, providing a graphically compelling Rub and Buzz indicator.

To add a second axis to an XY Graph, click the chevron icon (hover the mouse pointer on the right side of the Graph to see the icon) and select **Define Axis**. Choose the result you'd like to plot against the right axis.

XY Graphs: Edit Graph Properties

The Graph Properties dialog allows you to edit the Graph title and X- and Y-Axis titles and ranges.

General tab



Title

The Graph title always carries the name of the measurement (RMS Level, for example). To this you may add more descriptive text in the Title field.

Alignment

Select **Left**, **Center** or **Right** title alignment.

Font...

The **Font...** button opens the Windows font browser. You can modify the title font family, style, size, color and so on from this dialog. Default font is MS Sans Serif 8 point.

Visible

When **Title Visible** is checked the title is applied to the Graph.

Legend Visible

The legend is the boxed display containing channel number checkboxes on the right side of the Graph. When Legend Visible is checked the legend is applied to the Graph.

Interpolation (time-domain graphs)

For time-domain displays such as Scope or Acquired Waveform, the APx500 graphs are normally plotted using sinc function interpolation. Interpolation adds points to the displayed trace that do not exist in the measured data, to make data trends more easily visualized. The default in APx is to have Interpolation **On** for time-domain displays. However, digital domain signals are sometimes best understood when viewing actual samples, with interpolation switched **Off**. Set display interpolation **On** or **Off** here.

Interpolation can cause limit failure markers to appear to be in an incorrect location. See “Interpolation and limit failure markers” on page 678 for an explanation.

Show Indicators (some XY Graphs)

Some XY Graphs (see Stepped Frequency Sweep, for example) can show indicators at data points on the trace. This feature is most useful with graphs with relatively few data points.

In graphs with many data points, the markers and labels will overlap and obscure each other. Check **Show Indicators** to add markers.

Select

- **Never**
- **Top Axis**
- **On Traces** (the default)
- **Bottom Axis**

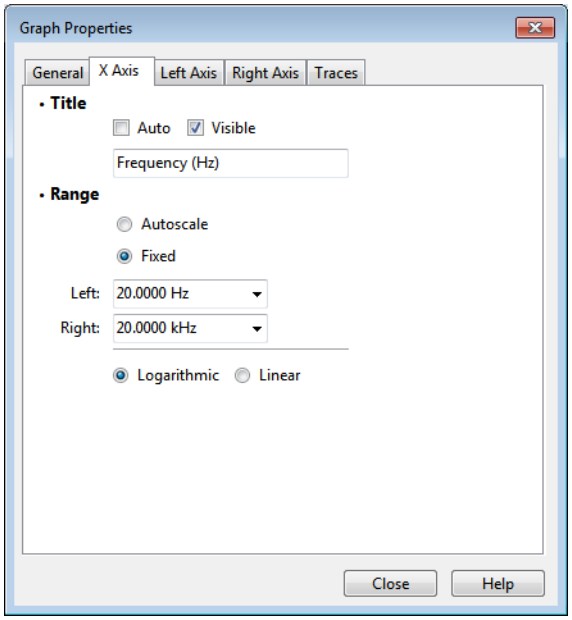
Show Markers (PESQ Averaged and POLQA Averaged only)

PESQ Averaged and POLQA Averaged graphs can show markers with X and Y values for each point plotted.

Select

- **X Labels**
  File number is shown on marker
- **Y Labels**
  MOS for file is shown on marker

X-Axis tab



Title

When **Auto** is not checked, you can enter a title for the **X-Axis** in the **Title** field.

Auto

When **Auto** is checked, the X-Axis label shows the parameter and the units, such as Generator Level (dBV).

Visible

When **Visible** is checked the X-Axis title is applied to the Graph.

Range

Range sets the values at the left and right edge of the Graph. Enter values in the **Left** and **Right** range fields. You can also select units from the range field drop-down menus.

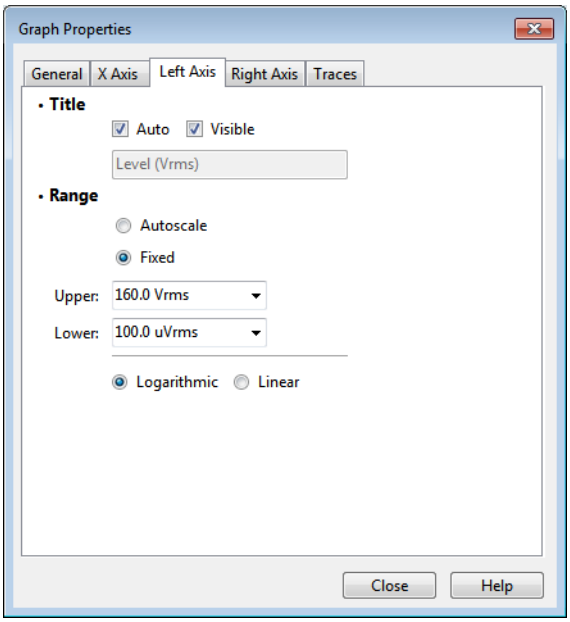
Auto

Auto sets the range around the current signal characteristics.

Logarithmic

When **Logarithmic** is checked, the X-Axis scale follows a logarithmic rule. When unchecked, the X-Axis scale follows a linear rule. The logarithmic scale is not available for certain units.

Left Axis tab



Dual axis graphs will have a Left Axis and a Right Axis tab.

Title

When **Auto** is not checked, you can enter a title for the Y-Axis in the Title field.

Auto

When **Auto** is checked, the Y-Axis label shows the parameter and the units (Level [dBV], for example).

Visible

When **Visible** is checked the Y-Axis title is applied to the Graph.

Range

Range sets the values at the top and bottom edges of the Graph. Enter values in the Upper and Lower range fields. You can also select units from the range field drop-down menus.

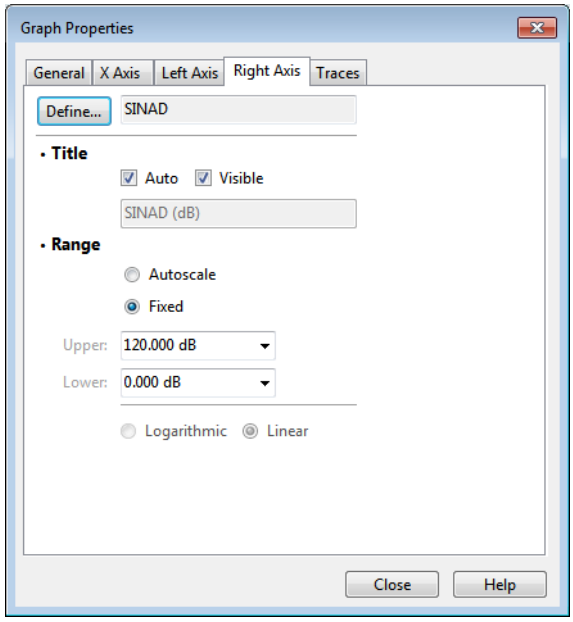
Auto

Auto sets the Y-Axis range around the current signal characteristics.

Logarithmic

When **Logarithmic** is checked, the Y-Axis scale follows a logarithmic rule. When unchecked, the Y-Axis scale follows a linear rule.

Right Axis tab



Dual axis graphs will have a Left Axis and a Right Axis tab.

Title

When **Auto** is not checked, you can enter a title for the Y-Axis in the Title field.

Auto

When **Auto** is checked, the Y-Axis label shows the parameter and the units (Level [dBV], for example).

Visible

When **Visible** is checked the Y-Axis title is applied to the Graph.

Range

Range sets the values at the top and bottom edges of the Graph. Enter values in the Upper and Lower range fields. You can also select units from the range field drop-down menus.

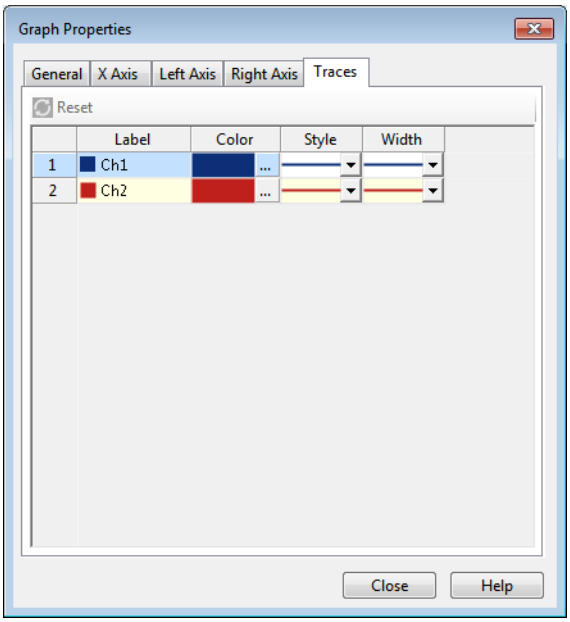
Auto

Auto sets the Y-Axis range around the current signal characteristics.

Logarithmic

When **Logarithmic** is checked, the Y-Axis scale follows a logarithmic rule. When unchecked, the Y-Axis scale follows a linear rule.

Traces tab



Open the **Graph Properties** dialog and go to the **Traces** tab. By default, the settings are derived from the global setting made in the **Labels** dialog, accessed from **Signal Path Setup**, or from the global settings made in the **Colors** tab, accessed from the **Project/ Sequence Properties** dialog. In the **Traces** tab you can edit the local settings for each trace, overriding the global settings.

Label

Enter a new **Label** for the channel in this field. This setting is local and will only affect this result.

Color

Click on the button to open a color picker. Choose a new **Color** for this channel. This setting is local and will only affect this result.

Style

Select a trace **Style** from the drop down menu. This setting is local and will only affect this result. This setting is local and will only affect this result.

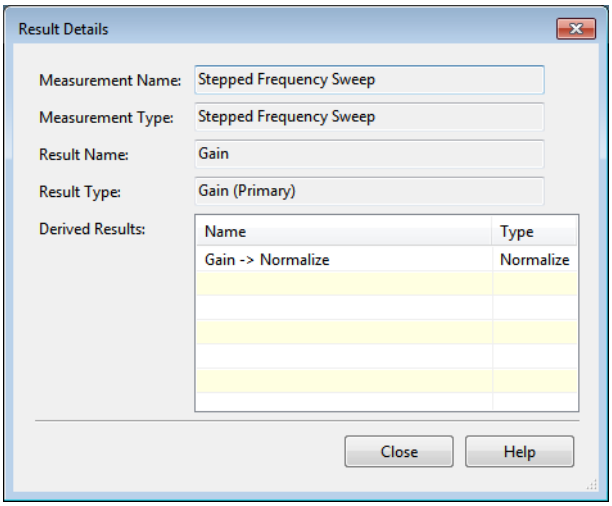
Width

Select a trace **Width** from the drop down menu.

Alternatively, you can right click in a Legend entry to open a context menu to edit the **Label**, **Color, Line Style** or **Line Width** for that channel.

For more information about setting global labels, see page 56. For information about global colors and line styles, see page 32.

The Result Details dialog

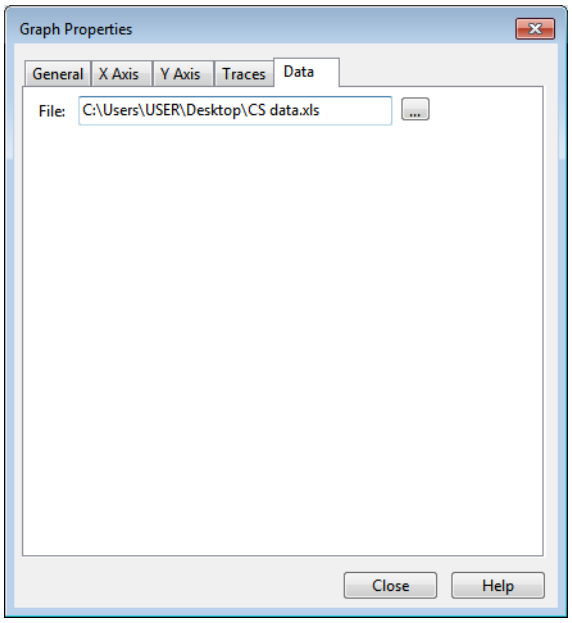


To view result details, select the result and click **Details** in the **Selector** toolbar. This is particularly useful when working with **Derived Results**, as it reveals the relationships between source and derived results. Read more about **Derived Results** in Chapter 100.

The **Result Details** dialog is informational only. It provides a display of a result's key characteristics, including:

- **Measurement Name**
  This field displays the name of the measurement providing the data for the result(s). The measurement name can be in the Navigator or Selector panes.
- **Measurement Type**
  This field displays the type of measurement, which cannot be changed.
- **Result Name**
  This field displays the name of the current result. The result name can be edited in the Navigator or Selector panes.
- **Result Type**
  This field displays the type of result, which cannot be changed.
- **Source**
  This field only appears if the current result is a derived result. It shows the source result for the current result.
- **Derived Results**
  If there are other results derived from the current result, they are shown here.

Data tab (Defined Results only)



This tab only appears for a Defined result.

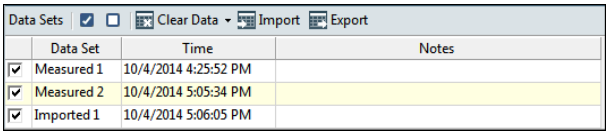
File

Enter the path and filename for the data file to be referenced by the Defined result, or use the Browser button to browse to the file. Read more about Defined Results on page 652.

Managing Data Sets

A **Data Set** is the set of results measured from an acquisition, or imported from a data file.

Data Sets are managed in the **Data Sets** panel, located below the **Selector** in the APx workspace.

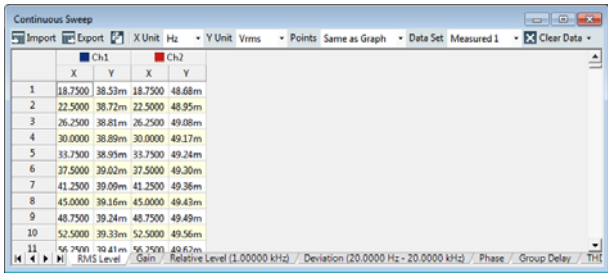


Data Sets panel

The **Data Sets** panel displays all the **Measured** and **Imported** data sets in a grid format.The first column provides visibility checkboxes; the second column displays the name of the data set; the third column displays the date and time of the acquisition or importation; the fourth column is available for notes. The notes are saved with the project.

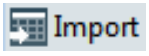
Click **Hide** to hide the **Data Sets** panel. When hidden, click **Show Data Sets** to restore the panel.

XY Graph data



The Graph data window shows the values for each data point in the Graph. There are X and Y values for each channel.

For measurements that have more than one result, tabs at the bottom of the data grid give you access to Graph data pages for the other results.

Import 

You can import Graph data from a file on disk to be displayed in the current Graph. The imported data is managed as a new data set, and is designated by the next available Imported Data Set number (n+1).

Data can be imported from a Microsoft Excel spreadsheet, a CSV (comma separated value) text file, or from Audio Precision *.adx or *.atsx legacy data files. If your data comes from some other source, data within the file must be arranged in the same format as a data file exported from APx500. Go to page 650 for detailed information about importing data.

Export 

Click the **Export** button to save the data for that page as a Microsoft Excel spreadsheet file (*.xls) or as a comma-separated-value (*.csv) text file. For Excel sheets, each tabbed page is saved as a new worksheet page within the file. For CSV files, tabbed pages are listed in sequential order within the file. For CSV files, tabbed pages are listed in sequential order within the file. Go to page 650 for detailed information about importing data.

Copy to clipboard 

Click the Copy button to copy the data to the Windows clipboard as tab-delimited text. This data can be pasted into other applications such as Microsoft Word or MS Excel.

X Units / Y Units

For each page, you can select units that differ from the initial Graph units.

Points

The default number of data points is **Same as Graph**, the same as the number of Graph data points. This number can be large for continuous sweeps. You can

reduce the number of points shown in the data grid by choosing a smaller number from the Points dropdown list, but you will lose resolution in the data shown or exported. The Points control is not available in the stepped sweep data grid dialogs, because Stepped Sweeps normally have a manageable number of points.

Note that Same as Graph reports only the points shown in the current Graph view. Zooming or panning the graph will affect the number of points reported.

Data

For appended or imported data, you can select the data set you would like to view or export.

Clear Data

Clear Data allows you to remove the data from one Data Set, or to remove all data from the graph.

Page Tabs

For measurements that have more than one result, tabs at the bottom of the Data Grid give you access to Graph data pages for the other results.

Import Graph (result) Data

Result data can be imported from other measurements, from other applications or from arbitrary lists of values. An imported result is not a primary result; it is not associated with a current measurement acquisition, and therefore does not produce the set of related primary results typically associated with a measurement. Importing Level data, for example, does not automatically provide related Gain data. Imported data results from each importation are gathered into a new data set, assigned default names such as **Imported1, Imported2,** and so on.

Data can be imported from a Microsoft Excel spreadsheet, a CSV (comma separated value) text file, or from Audio Precision *.adx or *.atsx legacy data files.

Result data cannot be imported into a derived result.

Data compatibility

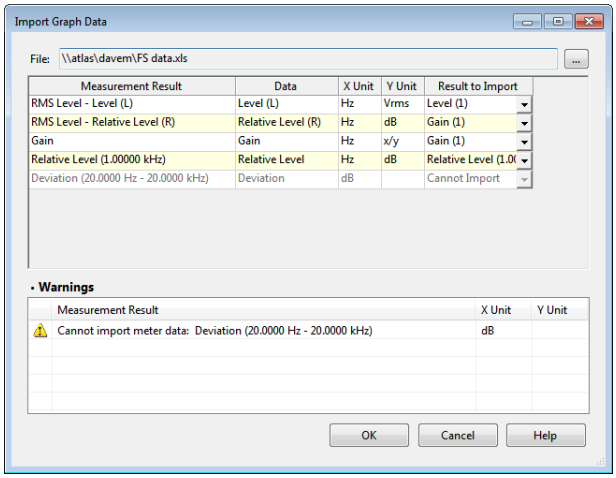
Imported result data must meet certain criteria to be compatible for importation and display.

1. If the data comes from a source other than an APx500 export, data within the file must be arranged in the same format as a data file exported from APx500 (see Export Graph Data, next).
2. Measurement results represented in the data file must be compatible with the current measurement. Compatibility is determined by unit family.

   For example, imported result data in Vrms units are compatible with measurement results in Vrms.

Import Graph Data dialog

The Import Graph Data dialog is available from the Graph Data panel, the Data Sets panel, or the graph right-click context menu.



Compatible Results

Measurement results represented in the data file that are compatible with the current measurement are shown in the upper grid in the dialog. If more than one result is available for a measurement, you can select the result(s) to import.

Incompatible Results

Measurement results in the data file that are not compatible with any result in the current measurement are shown in the **Warnings** grid, and will not be imported.

Export Graph (result) Data

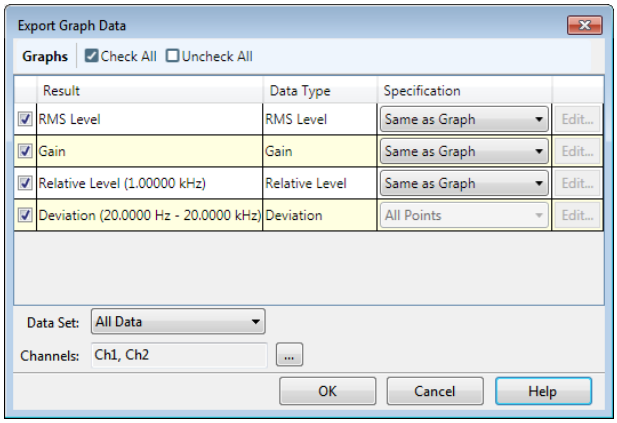
You can export the underlying graph result data as an Excel spreadsheet (*.xls or *.xlsx), a comma separated value text file (*.csv) or a Matlab file (*.mat).

The default settings are

- **Result:** all checked
- **Specification: Same as Graph**

Export Graph Data dialog

The Export Graph Data dialog is available from the Graph Data panel, the Data Sets panel, or the graph right-click context menu.



For measurements with more than one result, you can select which result(s) you would like to export. For graphs with more than one data set, you can select which data set to export. You can also specify the number of points to be exported, which can be greater or fewer than the points currently displayed.

The data specifications here are shared with the data specifications for Graph Data Settings.

For Excel sheets, each result (shown as a tabbed page in the Graph Data Grid) is saved as a new worksheet page within the file. For CSV files, measurement results are listed in sequential order within the file.

When you have made your export choices, click **OK**. A file browser will open where you can choose a name and location for the exported file.

Check All / Uncheck All

Select or deselect all result checkboxes for export.

Result

All results from the current measurement are listed in the grid. To include result data in the exported data file, check the checkbox for desired results.

Data Type

This column shows the data type for each result. Since the results can be renamed, the Data field can be helpful by revealing the underlying data type.

Specification

For continuous-sweep based measurements, there may be thousands of data points. You can reduce the number of data points exported with the Specification control. For results that have exportable data, this column provides menus to specify the range, distribution and number of data points to export. Choose

- **Same as Graph** (the default)
  The number of data points used to create the current graph; this is always fewer than All Points.
- **All Points**
  Every measured point in the result.

You can also specify a custom set of data points to be exported. Choose

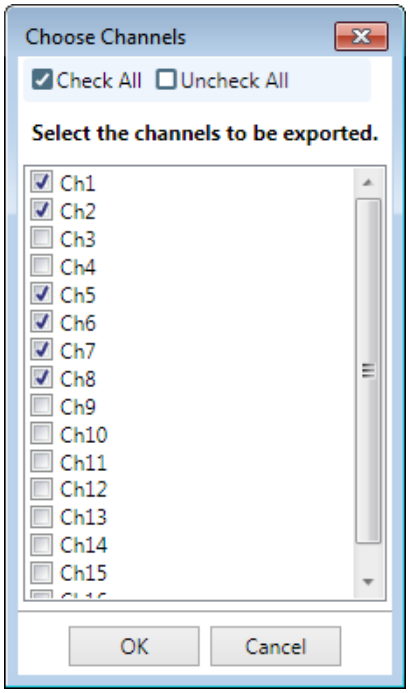
- **Create New...**
  Create New opens the Edit Data Specification dialog (see below). You can define and save a Data Specification as an *.xyspec file.
- **Browse to File...**
  Browse to File allows you to find and import an existing *.xyspec Data Specification file.

Data Set

For measurements that have more than one data set, you can select All Data Sets or a specific data set from this menu.

Channels

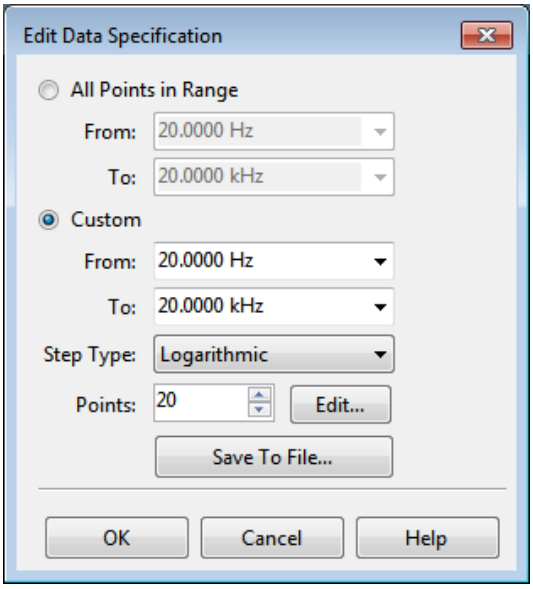
Channels displays the measurement channels for which data will be exported. Click the ellipses ... button to open the **Choose Channels** dialog.



Data Specification dialog

You can specify in detail the data you would like to export from an XY graph result.

A Data Specification is saved with the project as an Attached Project Item, and is available to other measurements in the project. You can also save the data specification in an independent file.



All Points in Range

This choice will select all points within a reduced X-axis range. Set the extent of the X-axis range using the **From** and **To** entry fields.

From/To

Set the X-axis range here.

Custom

Custom allows you to specify the X-axis range, the number of points and the distribution of points across the range.

From/To

Set the X-axis range here.

Step Type

Choose the distribution (step size) of the X-axis points in the **Points** table.

- **Logarithmic**
- **Linear**
- **Custom**

If you change a point's X-value in the data **Points** table, the **Step Type** will change type from **Logarithmic** or **Linear** to **Custom**. If you change the **Step Type** from **Custom** to either **Logarithmic** or **Linear**, the points will be redistributed and any custom point settings will be lost.

Points

Set the number of **Points** desired here.

Edit

This button opens a data **Points** table, where you can enter and reorganize points, and import or export data point table files.

Save to File

A Data Specification is saved with the project as an Attached Project Item (see page 27), and is available to other measurements in the project. You can also save the data specification in an independent file, with the filetype ".xyspec". Click **Save to File** to open a **Save Data Specification** dialog.

User Defined Results

Overview

You can define a new result and add it to a measurement. The data for a Defined Result is read from a file on disk at measurement run time, or when the (Refresh) button is clicked).

A Defined Result can be of the XY Graph type, or of the Bar Graph (Meter) type. It can contain any data, related or unrelated to the measurement to which it is attached.

Any units for the data can be arbitrarily assigned, and do not need to correspond to units used in APx500.

Creating and Using a Defined Result

You can attach a Defined Result to most measurements. Use one of these three methods to create a Defined Result.

3. Right-click on a measurement node in the Navigator, and select **Define New Result**. Choose **(X, Y)** or **Bar**.
4. Or, select a measurement and from the Project menu, choose **Define New Result**. Choose **(X, Y)** or **Bar**.
5. Or, click **Add** in the Selector menu bar. Choose **Defined Result**, then choose **(X, Y)** or **Bar**.

The Defined Result will be added to the measurement, and the Graph Properties dialog will open to the Data tab. In the File field, enter the path and filename for the data file to be referenced by the Defined Result, or us the Browser button to browse to the file.

The Graph Title, X and Y legend and units used in the Defined Result are imported from the file. Graph Properties (scale, alignment, title, trace color, etc.) are set in APx500 in the Graph Properties dialog.

See Data File Formats for Defined Results on the next page.

Application

You may define any result you find useful and add it to a measurement. You may assign any unit names to the data, including unit names not used in APx500.

Use Case 1

Goal: You wish to make an acquisition in Measurement Recorder, and to measure the audio with a fast

RMS measurement, evaluating the waveform at a reading rate of 1000 times per second.

1. Save the acquisition from Measurement Recorder

2. Make a fast RMS measurement with an external utility program that outputs its results in a CSV file.

3. Define a New Result attached to Measurement Recorder.

4. Reference that Defined Result to the CSV file.

Use Case 2

Goal: You wish to aggregate results from 3 different APx500 2-channel measurements into one Graph.

1. Run each measurement.

2. Export the data from the desired result from each measurement to CSV files.

3. Aggregate the data in an external program, such as Microsoft Excel.

4. Format the 3 sets of stereo data into a 6-channel format.

5. Define a New Result attached to a measurement of your choice.

6. Reference that Defined Result to the CSV file.

Data File Formats for Defined Results

File types

- Microsoft Excel spreadsheet (*.xls). Only the first sheet in a workbook will be read.

  or

- Comma-separated value (CSV) text file (*.txt).

	A	B
1	Level	
2	Trace 1	
3	X	Y
4	Hz	Vrms
5	20	0.1002349
6	20.447743	0.1002349
7	20.90551	0.1001984
8	21.373525	0.1001984
9	21.852017	0.1001984

```
Level,,,
Trace 1,,,
X,Y,,
Hz,Vrms,,
20,0.100234876,,
20.44774311,0.100234876,,
20.90550991,0.100198422,,
21.3735248,0.100198422,,
21.85201722,0.100198422,,
```

Excel Spreadsheet **CSV**

See a guide for proper arrangement of data in the file below.

Data arrangement in the file

The data arrangement is discussed in terms of a spreadsheet grid. A CSV file, if used, should correspond to this arrangement. See the illustration on the previous page.

	A	B	[Note]
1	Title	{empty}	[This can be any text string, or empty]
2	Trace Name	{empty}	[This can be any text string, or empty. It can be centered in merged cells A-B]
3	X-axis legend	Y-axis legend	[This can be any text string, or empty]
4	X-axis unit name	Y-axis unit name	[This can be any text string, or empty]
5	{X-data1}	{Y-data1}	[This can be any value, or NaN]
6	{X-data2}	{Y-data2	[This can be any value, or NaN]
7	{X-data3}	{Y-data3}	[This can be any value, or NaN]
8	{X-data4}	{Y-data4}	[This can be any value, or NaN]
9	{X-data5}	{Y-data5}	[This can be any value, or NaN]

- In APx500, NaN (Not a Number) represents an undefined value resulting from an invalid operation.
- As it is imported, the data is checked to verify that the values in column A monotonically increase. The data are and sorted if necessary.
- Cells in rows 1-4 can be empty
- When an empty row ≥ 5 occurs, it is interpreted as the end of the data.
- When an XY result has one empty X or Y cell in a row ≥ 5, it is interpreted as an error.
- For more Traces 2-16, if required, repeat this pattern in columns C-D, E-F, and so on.

Derived Results

Introduction

Derived results accomplish many of the same functions as Computes in other Audio Precision software.

Most measurement results, whether primary, imported, or derived, can have a derived result attached. A result that has a derived result attached is called a *source* result. The derived result is computed from the source result, using one of a set of mathematical functions such as arithmetic mean (average), normalize, invert, etc. See page 635 and Chapter 99 for more information about primary or imported results.

The availability of derived result functions depends upon the type of source result the derived result is attached to. A meter result such as **Level and Gain: RMS Level**, for example, does not support **Normalize** or **Invert**, but an XY result such as **Frequency Response: Level** does.

Not all results support a derived result. Additionally, derived results do not support importation of graph (result) data (see page 650).

Working with Derived Results

Navigating, adding, deleting and renaming derived results is the same as working with primary results. See more information about working with results on page 636.

Of particular interest when working with derived results is the Result Details dialog, which reveals the relationships between the primary (source) result and the derived result. See page 648.

Go to page 666 for several illustrated examples that show the software user interface for attaching and configuring derived results.

Adding a Derived Result

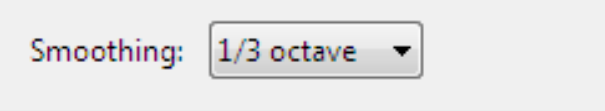
A derived result can be added using any of the following methods. Some methods allow deleting or renaming the derived result as well.

- **Navigator**
  Right-click on a result and choose **Add Primary Result**, **Add Derived Result, Delete Result** or **Rename**.
- **Selector toolbar**
  Select a result and click **Add** or **Delete** in the Selector toolbar.
- **Selector filmstrip**
  Right-click on a result and choose **Add Derived Result, Delete** or **Rename**.
- **Project menu**
  Select a result in the Navigator or the Selector, and from the main Menu bar choose **Project > Add Primary Result**, **Project > Add Derived Result** or **Project > Delete Result**.
- **Graph display**
  Right click in the large Graph result display and choose **Add Derived Result**, or click the **Add Derived Result** button in the Graph toolbar.

Derived Result settings

Each derived result has settings to define the computations that are applied to the source result. These can be very simple and straightforward, requiring the entry of just one value in a setting field. Or, the settings can involve multiple sets of specifications, each of which define different computations across a range of source data.

Single parameter settings



Derived results such as **Offset**, **Smooth**, **Normalize**, **Invert** and **Compare** (when attached to a meter result)

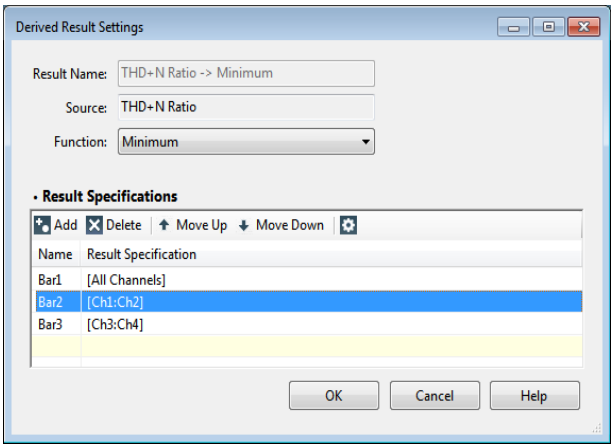
have only a single parameter to set, which is available as an entry field or a menu in the **Result Settings** bar. Smooth is shown above.

Other derived results require Result Specifications, accessed through a Result Settings dialog.

The Result Settings dialog

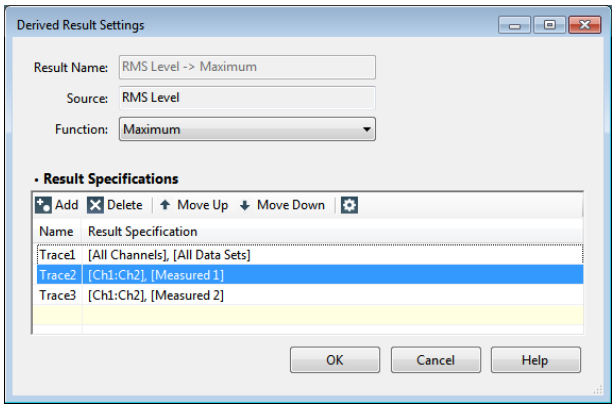
Compare (when attached to an XY result) and **Statistics** group derived results require more a detailed result specification, defining the channels and/or data sets to be considered.The Result Settings dialog for such derived results opens when you first add the result; you can return to the settings dialog using one of the following methods:

- **Selector toolbar**
  With a derived result selected, click the **Derived Result Settings...** button.
- **Selector filmstrip**
  Right-click on a derived result and choose **Derived Result Settings...**
- **Graph display**
  Right click in the large Graph result display and choose **Derived Result Settings...**



Single value (meter) derived result settings dialog.

Single value (meter) source results only support single value derived results. XY source results can have derived results attached that may have an XY output, or a single value output, depending upon the derived result function and the intention of the user.



XY graph derived results settings dialog.

Source

This field displays is the source result (or results) upon which the current derived result is acting.

Function

For Statistics group derived results, this field displays the currently applied Statistics group function. This function can be changed at any time, without redefining result specifications.

Result Specifications

Each Derived Result begins with one Result Specification. A Result Specification

Add

Initially, each derived result has one result specification. Double-click or select **Properties** to open the Result Specification dialog for editing.

The default specification is named **Bar1** (for meter results) or **Trace1** (for XY results), and occupies the first line of the grid. Click **Add** to add more result specifications, which will take the default names **Bar2, Bar3** or **Trace2**, **Trace3** and so on. You can rename each specification.

You can also add a result specification from the context menu available by right-clicking in the grid area and selecting **Add Result Specification...**

Delete

When there is more than one result specification in the grid, you can remove a result specification by highlighting it in the grid and clicking **Delete**, or by right-clicking and selecting **Delete Result Specification** from the context menu.

Move Up / Move Down

When there is more than one result specification in the grid, you can move a result specification by highlighting it in the grid and clicking **Move Up** or **Move Down**, or by right-clicking and selecting **Move Up** or **Move Down** from the context menu.

Properties

You can view and edit the result specification properties by double-clicking the specification line in the grid, or by highlighting the line and clicking the **Properties** button, or by right-clicking and selecting **Properties** from the context menu.

Functions available for derived results

- Offset (page 657)
- Compare (page 657)
- Harmonic Sum (page 658)
- Data Distribution (page 659).
- Normalize (page 659)
- Invert (page 659)
- Smooth (page 660)
- Specify Data Points (page 661)

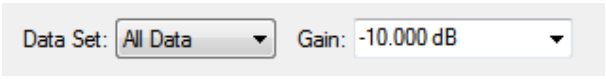
Min/Max Statistics group:

- Minimum (page 663)
- Maximum (page 663)
- Geometric Mean (page 664)
- Arithmetic Mean (page 664)
- RMS (root mean square) (page 664)
- Standard Deviation (page 664)
- Max Difference from Geometric Mean (page 664)
- Max Difference from Arithmetic Mean (page 664)
- Power Average (page 664)

Offset

Offset is a derived results function that shifts the value(s) of the result data by a specified amount. Offset can be attached to either single value (meter) results or XY source results.

One of two algorithms is used, depending upon the unit domain.



Offset applied to audio level unit domains

For audio levels and ratios of audio levels, Offset is accomplished by application of a **Gain Value**, multiplying or dividing the source result value. The **Gain Value** is specified in the Result Settings bar.

For example, specify a +10 dB offset for an existing result of 1 Vrms by entering 10 dB in the **Gain Value** field. 1 Vrms = 0 dBV, and derived result will be +10 dBV, which is equal to 3.162 Vrms.

Offset applied to other unit domains

For any DC level, Frequency or Phase result, for Acquired Waveform results, or for the Signal Analyzer Scope result, offset is accomplished by adding or subtracting an offset value to the source result. The **Offset Value** is specified in the Result Settings bar.

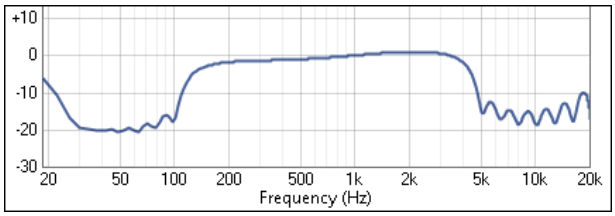
...when attached to meter results

When attached to a meter source result, Offset returns the current value for each channel, shifted by the Offset value and displayed as single value (meter) bar graphs.

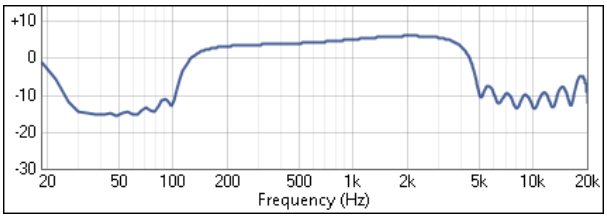
...when attached to XY results

When attached to an XY source result, Offset returns the value for each data point, shifted by the Offset value and displayed as XY graph traces.

The first trace shown below has been normalized at 1 kHz.



The second trace below is the normalized result, offset by +5 dB.

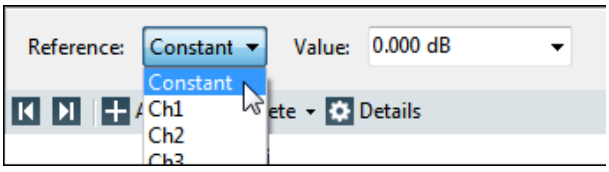


Compare (Ratio or Level)

Compare is a derived results function that compares data to a constant or to the data in another channel or data set. Compare can be attached to either single value (meter) results or XY source results.

For source results in absolute units, such as volts or watts, Compare computes the difference between the values. For source results in relative units, such a decibels (dB), Compare computes the ratio between the values.

attached to a meter result



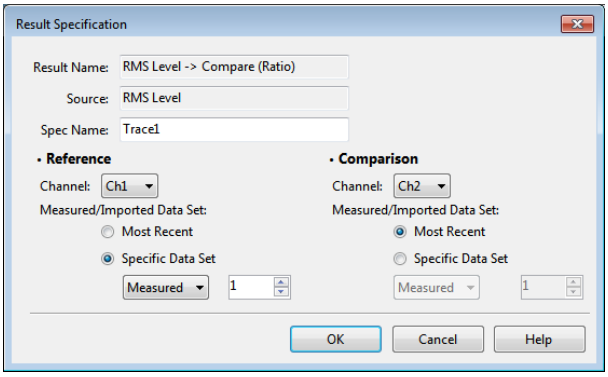
When attached to a meter source result, Compare finds the current value for each channel, compared to

the value of the reference. The reference can be set to a constant value, or to the current value in any channel. The comparison values are single value (meter) results, displayed as bar graphs.

attached to an XY result

When attached to an XY source result, Compare finds the current Y value for each data point, compared to the Y value of the reference data point. The comparison data points can be selected from any channel and from any data set; similarly, the reference data points can be selected from any channel and from any data set. The comparisons are displayed as XY graph traces.

These selections are made in the Result Specification.



Typical Result Specification dialog for Compare when attached to an XY source.

In this application, Compare supports up to 16 specifications, managed in a Derived Result Settings dialog (page 655).

A note about channel count in derived results...

In a derived result setting you can select channel numbers beyond the channel count available via the physical interface connectors on your APx analyzer.

This is because all APx500 systems support up to 16 input channels of audio data, when the input configuration is set to File.

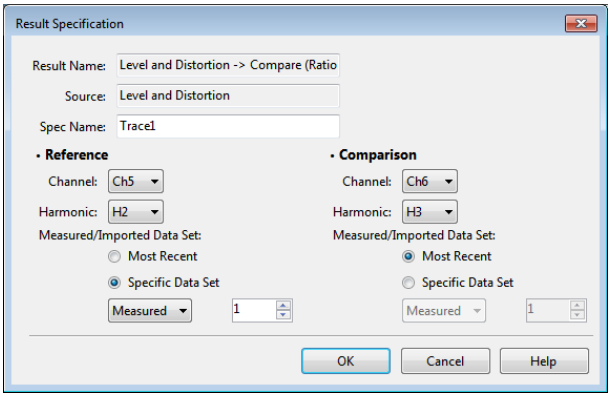
Be sure to select channel numbers that are relevant to your test setup.

A note about specifying non-existent data sets in derived results...

You can specify data sets that have not yet been acquired or imported. This supports use in automated sequences, where the derived result can be defined before the sequence has completed all of its append activities.

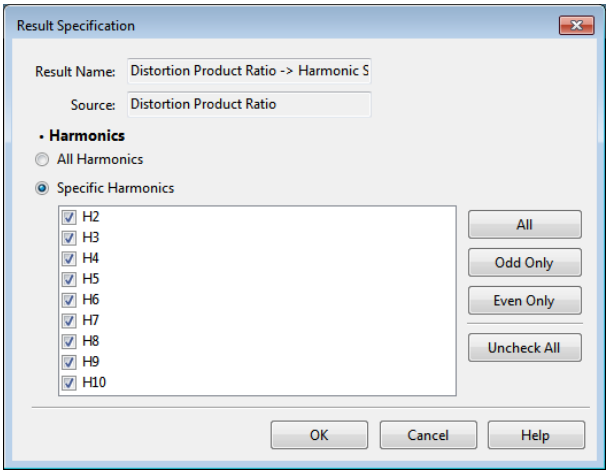
Special Compare Specifications for Acoustic Response

When Compare is attached to an Acoustic Response Level and Distortion result, both Channels and Harmonics can be selected.



Harmonic Sum (Ratio or Level)

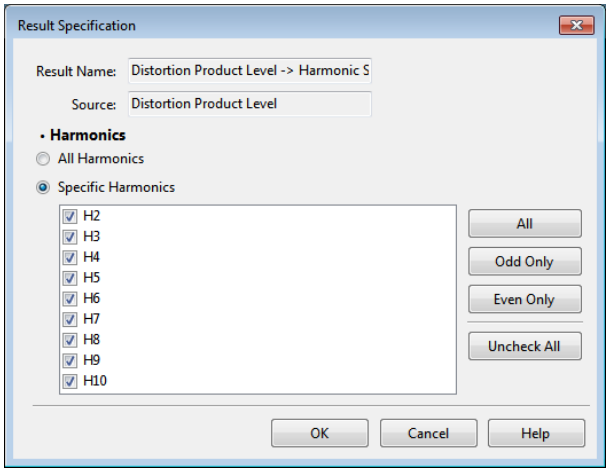
Harmonic Sum Ratio



The Harmonic Sum Ratio derived result can only be attached to the THD+N Distortion Product Ratio result.

The derived result provides the sum of the levels of any combination of harmonic products H2 through H10, divided by the signal level. The result is expressed as a ratio.

Harmonic Sum Level

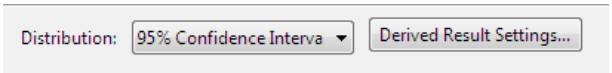


The Harmonic Sum Level derived result can be attached to the THD+N Distortion Product Level result.

The derived result provides the sum of the levels of any combination of harmonic products H2 through H10.

> *These derived results are available for THD+N Distortion Product Ratio or Distortion Product Level results only.*

Data Distribution



This result displays the confidence interval, a measure of the reliability of an estimate, such as a mean. When computing a MOS average, the ITU P.862.1 requires, among other results, a 95% confidence interval result.

Assuming the data distribution is Gaussian, the 95% point corresponds to 1.96 standard deviations from the mean, and the result displayed is 1.96 times the standard deviation of the input data.

The result is expressed as a mean value followed by a +/- value for the deviation.

Normalize

> *Normalize can only be attached to an XY result.*

Normalize is a derived results function that shifts an XY trace so that the data point at the X-axis reference is normalized (set to the unit value of the base unit of the unit domain). The X-axis for XY graphs is usually *frequency*, but for some results is *time*.

Normalization is accomplished by first converting the result units to the base unit for the unit domain, then dividing the value of each data point by the value of the data point at the selected X-axis reference.

For Vrms, the unit value (normalization value) is

- 1 Vrms.

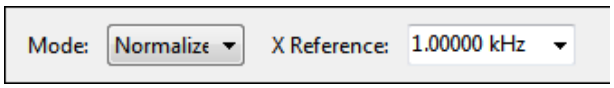
For Vrms, the unit value (normalization value) is

- 1 W.

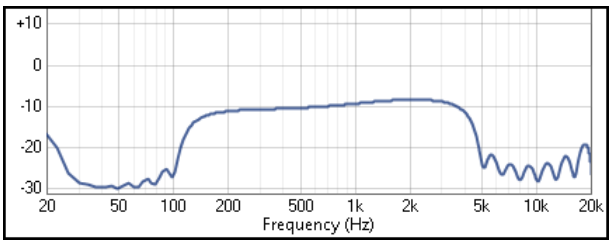
For decibel unit, the unit value (normalization value) is

- 0 dB.

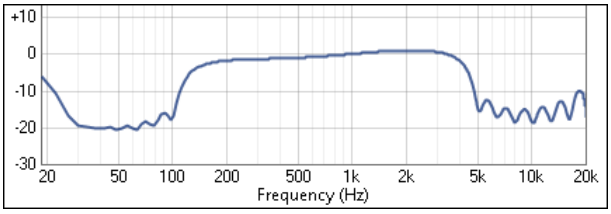
Enter the X-axis reference point in the **X Reference** field.



The first trace shown below is the Level result, before normalization.



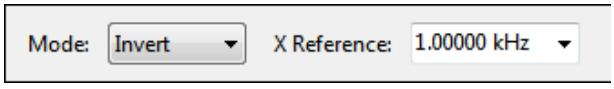
The second trace below has been normalized at 1 kHz.



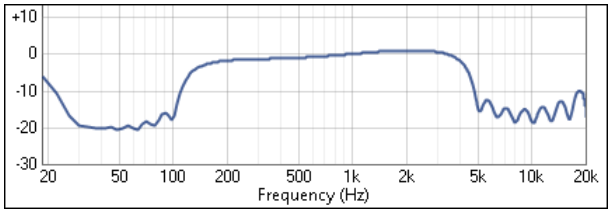
Invert

> *Invert can only be attached to an XY result.*

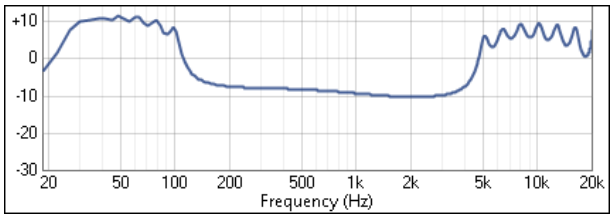
Invert is a derived results function that “flips” an XY trace around a user-defined pivot point. Set the pivot point by choosing an X-axis reference, entered in the **X Reference** field on the Result Settings bar.



The first trace shown below has been normalized at 1 kHz.



The second trace below is the normalized result above, inverted with an X Reference (pivot point) of 1 kHz.



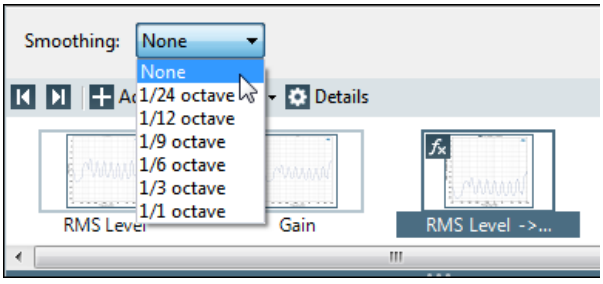
Smooth (frequency domain)

Smooth is a derived results function that is primarily used to smooth frequency-domain XY results. Frequency-domain smoothing is a common technique in loudspeaker response measurement, useful in revealing trends by reducing anomalies in the response curve for viewing. We recommend using smoothing in the Acoustic Response measurement.

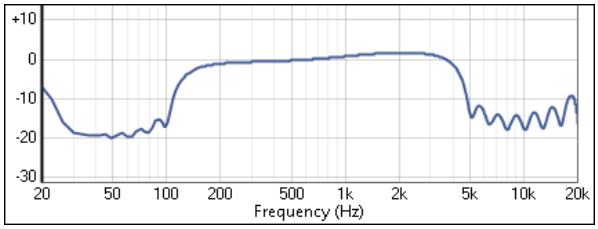
The smoothing algorithm for frequency-domain XY results effectively passes the data through a constant-Q bandpass filter. The bandwidth of this filter is selected in the Smoothing field. The APx500 implementation uses a hybrid FFT bin averaging and interpolation technique to achieve smooth results even at very low bin densities.

When a Smooth derived result is selected, the Smoothing field will be available in the Results Setting Bar. Note that the Deviation result, which does not support derived results, has a dedicated smoothing function in the Acoustic Response measurement.

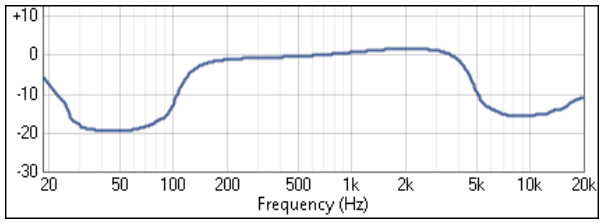
Select the desired value from the dropdown list.



The first trace displays the unsmoothed level results.



The trace below has had 1/3 octave smoothing applied.



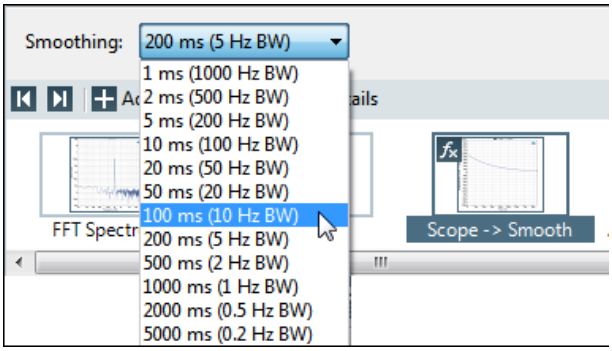
Smooth (time domain)

As mentioned above, Smooth is a derived results function that is primarily used to smooth frequency-domain XY results. However, it can also be attached to time-domain XY results.

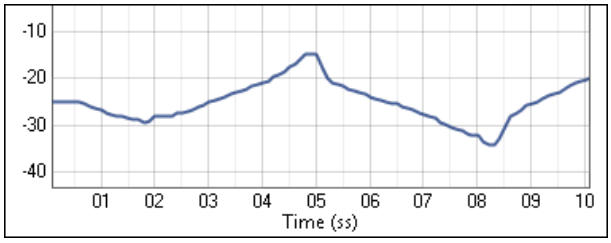
The smoothing algorithm for time-domain XY results effectively passes the data through a single-pole low-pass filter. The time constant and bandwidth of this filter is selected in the Smoothing field.

When a Smooth derived result attached to a time-domain result is selected, the Smoothing field will be available in the Results Setting Bar.

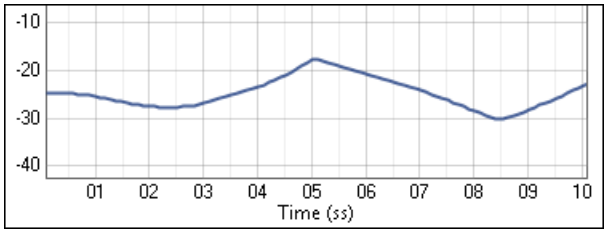
Select the desired value from the dropdown list.



The first trace displays an unsmoothed level result from the Measurement Recorder, showing a DUT whose gain varies over time.

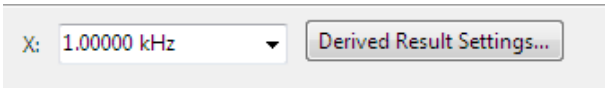


The trace below has had smoothing applied.



Note that the Measurement Recorder has a maximum reading rate of 20 points per second, which provides only 200 data points in a 10 second acquisition. With so few points, the traces do not appear as smooth as a typical frequency domain trace, which may have thousands of data points in an acquisition.

Specify Data Points Single Value



Specify Data Points Single Value can only be attached to results whose source is a continuous-sweep technology measurement: Acoustic Response, Continuous Sweep, and Frequency Response.

Specify Data Points Single Value is a derived results function that allows you to select a specific frequency for reporting results in a continuous-sweep based measurement. Specify Data Points can be applied to measured results or imported results.

After attaching Specify Data Points Single Value to a result, enter the frequency of choice to select the data point to report.

Specify Data Points Single Values finds the desired data point for each channel for the data sets selected in the Result Specification, and displays the value of that data point on a bar graph.

Explanation

Measurements based on continuous sweep technology (Acoustic Response, Continuous Sweep, and Frequency Response) gather data points at thousands of frequencies. The exact frequencies used are determined at the time of measurement, and depend upon the sweep settings. The exact frequencies of sweeps appended with identical settings will also vary slightly from one another.

If you are seeking measurements at one specific frequency (at 1.00000 kHz, for example), you will find that continuous-sweep based measurements will provide values at frequencies very close to, but not exactly at, your specified frequency. A typical 20 Hz to 20 kHz logarithmic sweep, for example, might return data near 1 kHz at exactly 0.99750 kHz and 1.00125 kHz.

Specify Data Points Single Value allows you to select a specific frequency for reporting results. The value of the nearest neighbor in the source data is reported at the chosen frequency.

Use cases

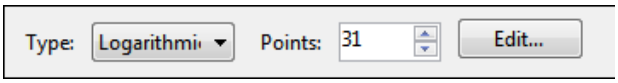
If you are required to report measurements that specify data at an exact frequency, use Specify Data Points Single Value to select the required frequency.

If you are programmatically gathering results across appended sweeps, for example, slight variances in reported frequencies can complicate programming. Use Specify Data Points Single Value to ensure that every sweep has a valid result at the selected frequency.

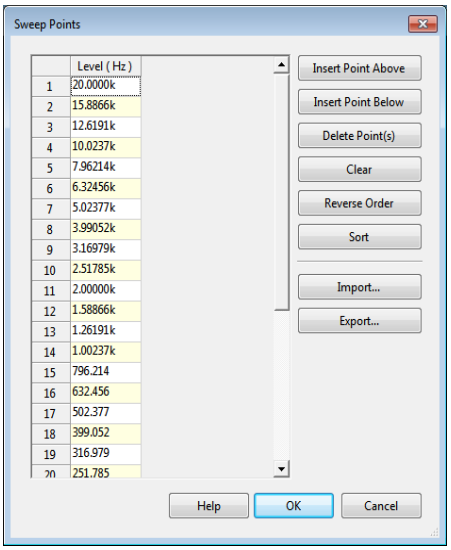
Specify Data Points (X, Y)

Specify Data Points can only be attached to results whose source is a continuous-sweep technology measurement: Acoustic Response, Continuous Sweep, and Frequency Response.

Specify Data Points is a derived results function that allows you to select specific frequencies for reporting results for a continuous-sweep based measurement.



After attaching Specify Data Points to a result, use the controls in the Result Settings bar to select the data points to report. Choose Logarithmic or Linear for regularly-spaced frequencies, and set the number of desired points. For exact control, click Edit to open the Sweep Points dialog and create, import or export a custom Sweep Points table.



Explanation

Measurements based on continuous sweep technology (Acoustic Response, Continuous Sweep, and Frequency Response) gather data points at thousands of frequencies. The exact frequencies used are determined at the time of measurement, and depend upon the sweep settings. The exact frequencies of sweeps appended with identical settings will also vary slightly from one another.

If you are seeking measurements at one or more specific frequencies (at 1.00000 kHz, for example, or at the series of conventional frequencies used in a 1/3-octave sweep), you will find that continuous-sweep based measurements will provide values at frequencies very close to, but not exactly at, your specified frequencies. A typical 20 Hz to 20 kHz logarithmic sweep, for example, might return data near 1 kHz at exactly 0.99750 kHz and 1.00125 kHz.

Specify Data Points allows you to select specific frequencies for reporting results. The value of the nearest neighbor in the source data is reported at the chosen frequency.

Use cases

If you are required to report measurements requirements specify data at exact frequencies, use Specify Data Points to select the required frequencies.

If you are programmatically gathering results across appended sweeps, for example, slight variances in reported frequencies can complicate programming. Use Specify Data Points to ensure that every sweep has valid results at the selected frequencies.

Min/Max Statistics group:

The Statistics group is a set of derived results that apply statistical computations to APx500 source results. For a given source result type (single value or XY) and a given derived result output type (single value or XY), functions in the Statistics group share common settings and specifications.

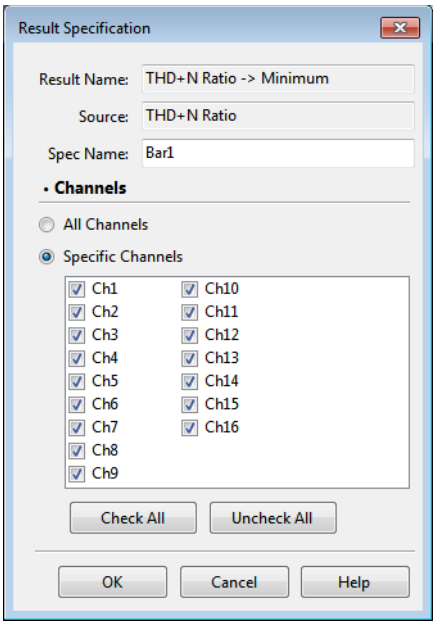
Once you have attached a Statistics group derived result, you can select and view other statistical functions without needing to define a new result. Simply select a different function from the drop-down menu on the Results Settings bar or in the Result Specification dialog.

Statistics group functions

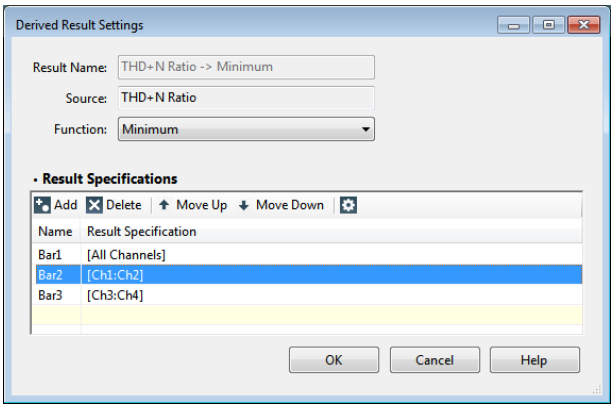
- Minimum (page 663)
- Maximum (page 663)
- Geometric Mean (page 664)
- Arithmetic Mean (page 664)
- RMS (root mean square) (page 664)
- Standard Deviation (page 664)
- Max Difference from Geometric Mean (page 664)
- Max Difference from Arithmetic Mean (page 664)

Statistics Group (single value output) attached to single value (meter) results

When attached to a meter result, a statistics (single value output) derived result gathers information across specified channels, and displays a single value result as a meter bar. Channels are selected in the Result Specification dialog, shown here.



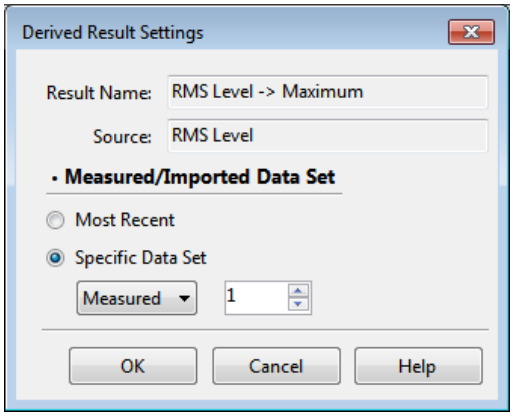
Up to 16 result specifications are supported, managed in the Derived Result Settings dialog shown here.



Statistics Group (single value output) attached to XY results

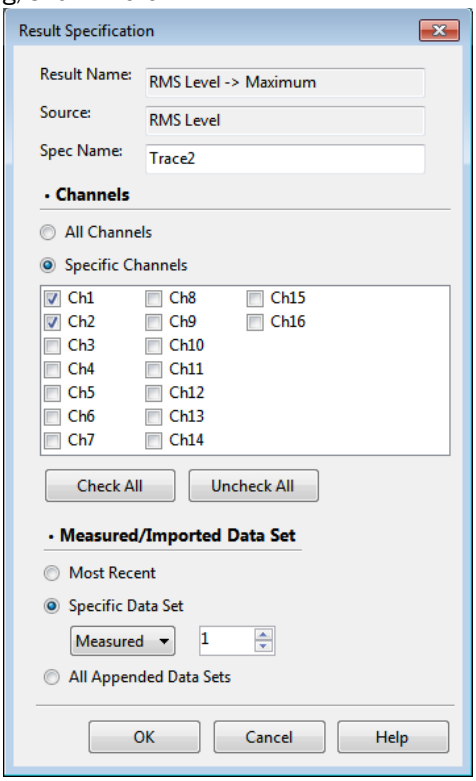
When attached to an XY result, a statistics (single value output) derived result gathers information for each channel for a specified data set, and displays a single value result as a meter bar for each channel.

Data sets are selected in the Result Specification dialog.

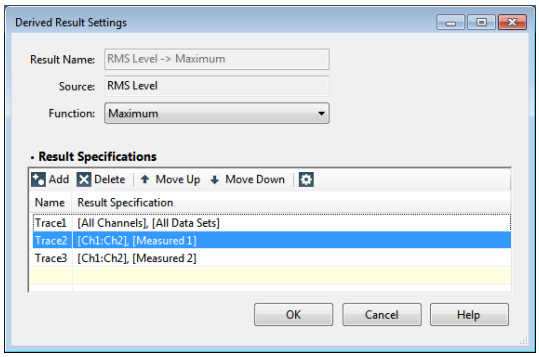


Statistics Group (XY output)

A Statistics group (XY output) derived result can only be attached to an XY source result. Information is gathered across selected channels for selected data sets. The derived result is displayed as an XY graph, with a trace for each result specification. Channels and data sets are selected in the Result Specification dialog, shown here.

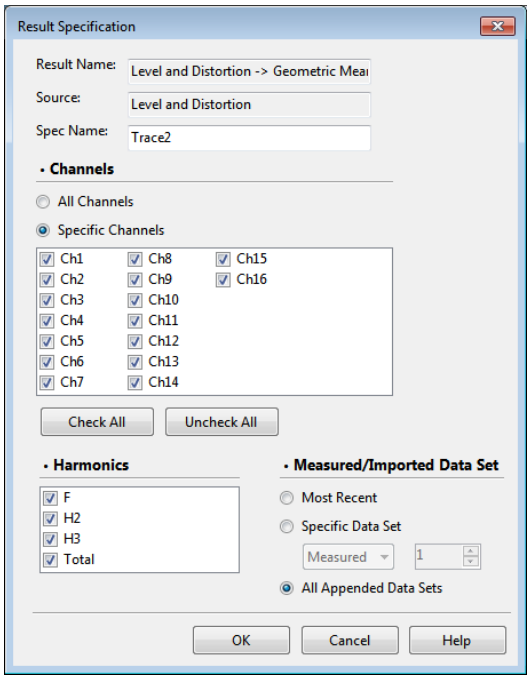


Up to 16 result specifications are supported, managed in the Derived Result Settings dialog, shown here.



Special Statistics Specifications for Acoustic Response

When a Statistics Group derived result is attached to an Acoustic Response Level and Distortion result, both Channels and Harmonics can be selected.



Statistics Group functions descriptions

Minimum

Minimum is a Statistics group function that finds the lowest value in the result data it acts upon.

Maximum

Maximum is a Statistics group function that finds the highest value in the result data it acts upon.

Geometric Mean

Geometric Mean is a Statistics group function that finds the value of the geometric mean of the result data it acts upon. The geometric mean is defined as nth root of the sum of the squares of the values of n data points.

Formula:

$$\left(\prod_{i=1}^{n} a_i\right)^{1/n} = \sqrt[n]{a_1 a_2 \cdots a_n}.$$

Arithmetic Mean

Arithmetic Mean is a Statistics group function that finds the arithmetic mean (average) of the result data it acts upon. The arithmetic mean is defined as the sum the values of all data points divided by the number of all data points.

Formulae:

(x1+x2+x3...xn)/n.

or, in summation notation,

$$\bar{x} = \frac{1}{n} \cdot \sum_{i=1}^{n} x_i$$

RMS (root mean square)

RMS is a Statistics group function that finds the root mean square of the result data it acts upon. The root mean square (also called the quadratic mean) is defined as the square root of the sum of the squares of the values of all data points.

Standard Deviation

Standard deviation is a Statistics group function that finds the standard deviation (the degree of variation from the mean) of the result data it acts upon.

Definition of standard deviation

The standard deviation function shows the degree of variation from the arithmetic mean (average).

Standard deviation is the square root of the arithmetic mean of the variance of each point. The variance is the squared deviation (the difference between the point value and the arithmetic mean of all point values, squared). To compute this, first find the arithmetic mean of all points. Then find the difference of each point from the mean, and square each of these results. This is the variance for each point. Then find the arithmetic mean of all the variances. Lastly, find the square root of this mean.

For example, for 9 points (2, 4, 6, 4, 3, 5, 7, 3, 2), the average (arithmetic mean) is the sum of all the values divided by the number of values, or 36/9 = 4.

The deviation of each point from the average (4) is (–2, 0, 2, 0, –1, 1, 3, –1, –2).

The squared deviation, or variance, of each point is (4, 0, 4, 0, 1, 1, 9, 1, 4).

The average of these variances is 24/9 = 2.67.

The square root of this average is the standard deviation = 1.633

Formulae:

Where the standard deviation is represented by σ (sigma), the arithmetic mean is represented by μ (mu), x is the value of each point, and N is the number of points:

$$\sigma = \sqrt{\frac{(x_1 - \mu)^2 + (x_2 - \mu)^2 + \cdots + (x_N - \mu)^2}{N}}$$

or, in summation notation,

$$\sigma = \sqrt{\frac{1}{N}\sum_{i=1}^{N}(x_i - \mu)^2}$$

Max Difference from Geometric Mean

Max Difference from Geometric Mean is a Statistics group function that finds the difference between the value of the data point(s) with the maximum difference from the geometric mean and the value of the geometric mean, in the result data it acts upon.

Max Difference from Arithmetic Mean

Max Difference from Arithmetic Mean is a Statistics group function that finds the difference between the value of the data point(s) with the maximum difference from the arithmetic mean and the value of that mean, in the result data it acts upon.

Power Average

Power Average (spectrum averaging) is a Statistics group function, with primary applications in acoustics, as listed below.

Power Averaging (single value output)

When attached to an XY frequency domain level result in Vrms, Power Averaging (single value output) squares each data point in a selected acquisition, divides the result by the number of data points and takes the square root of the result. The power average for each channel is displayed as a meter bar. To constrain the result to a frequency range, first apply the Specify Data Points derived result to the data, and apply Power Averaging (single value result to the Specify Data Points result.

Power Averaging (XY output)

When attached to an XY frequency domain level result in Vrms, for each X value Power Averaging (XY output) squares the Y value for all selected channels and acquisitions, divides the result by the product of the

number of channels and acquisitions, and displays the square root as a resulting Y value for that X. This provides a single frequency domain level trace. Select channels and acquisitions in the Result Specification. Power Averaging (XY output) supports up to 16 specifications, managed in the Derived Result Settings dialog.

Applications

Applications in acoustics include:

- Averaging acoustic response curves over an area to produce a “listening window” averaged response curve. This is also known as a spatial average.
- In tuning automotive sound systems, a few microphones are positioned at distributed locations near each seat location and the responses of sets pertaining to each seat are averaged to find the response representative of each seat.
- Averaging the frequency response curves of multiple microphones in an array.
- Averaging the response of multiple headphone measurements after refitting, to account for variations in fit. (IEC 60268-7)

Channel count in a derived result

In a derived result setting you can select channel numbers beyond the channel count available via the physical interface connectors on your APx analyzer.

This is because all APx500 systems support up to 16 input channels of audio data, when the input configuration is set to File.

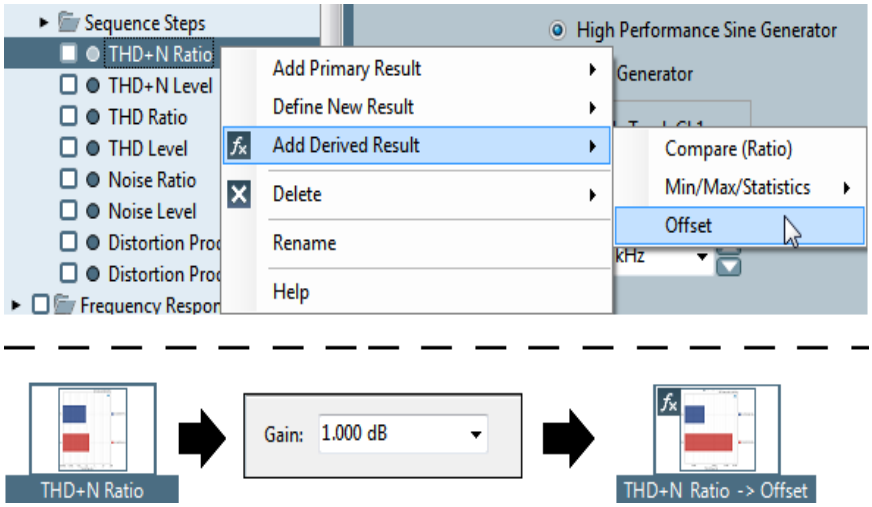
Be sure to select channel numbers that are relevant to your test setup.

Non-existent data sets in a derived result

You can specify data sets that have not yet been acquired or imported. This supports use in automated sequences, where the derived result can be defined before the sequence has completed all of its append activities.

Example 1. Attach Offset to THD+N Ratio

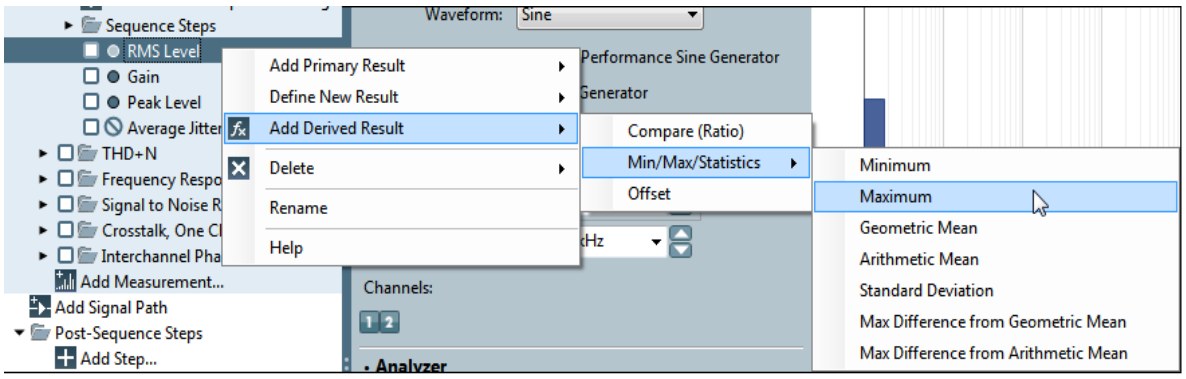
Attaching a Derived Result Offset Function to a THD+N Ratio result.

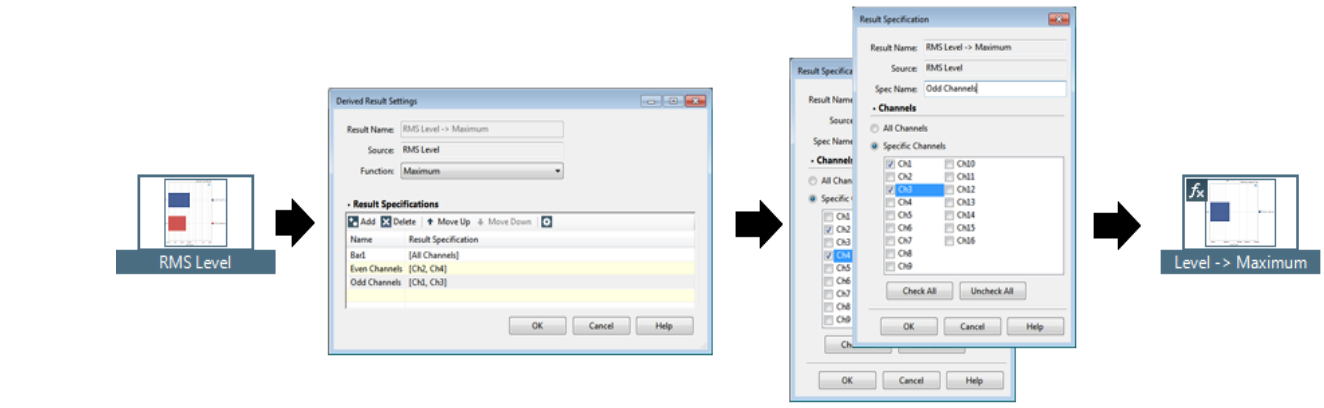


Attaching the Derived Result to the source result places a Gain field on the Result Settings bar, where you can specify the offset gain. We have entered 10.00 %. The Derived Result is of the single value type, with meter bars showing the offset for each channel.

Example 2. Attach Maximum (single value) to RMS Level

Attaching a Derived Result Maximum Function to an RMS Level result.

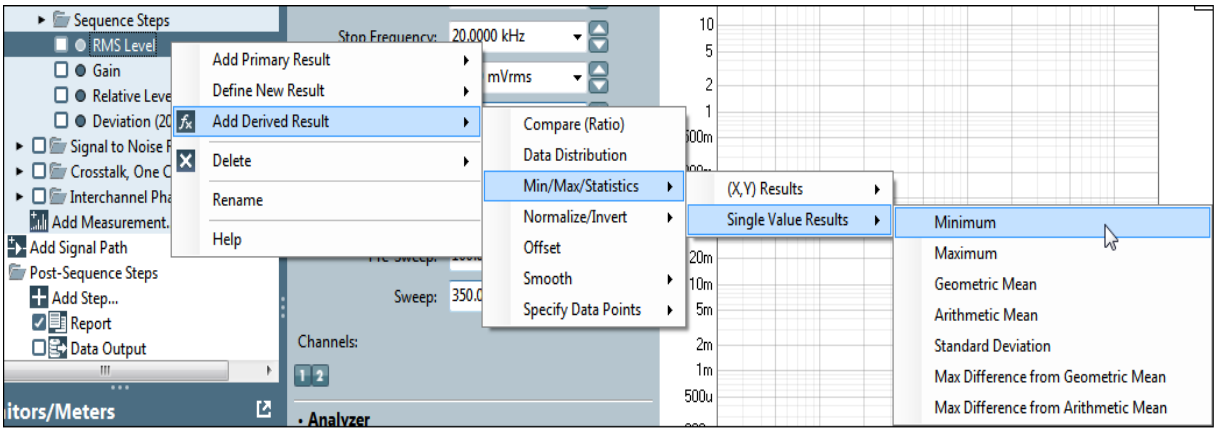




Attaching the Derived Result to the source result opens a Result Settings dialog, where you can specify one or more Result Specifications. We have made two specifications: one that returns the maximum across the even channels, and one that returns the maximum across the odd channels. The Derived Result is of the single value type, with one meter bar for each specification.

Example 3. Attach Minimum (single value) to Frequency Response Level

Attaching a Derived Result Minimum Function to a Frequency Response Level result.

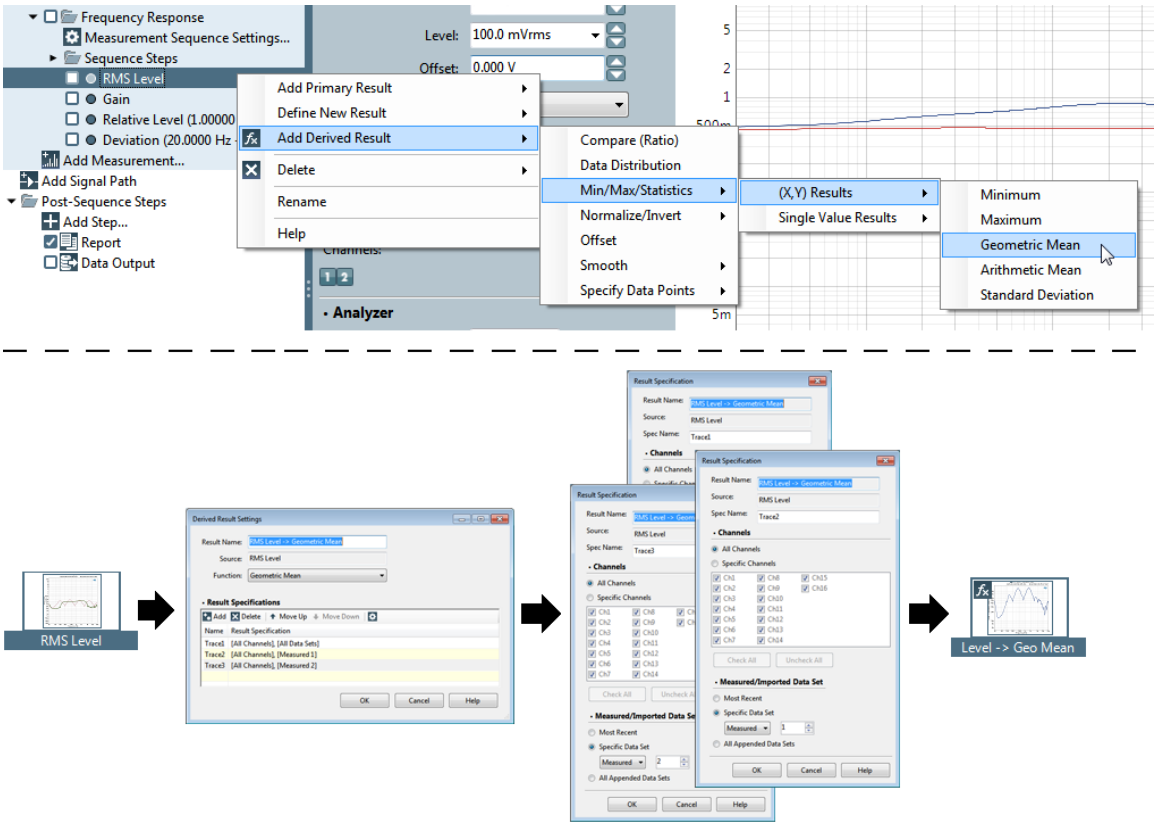




Attaching the Derived Result to the source result opens a Result Settings dialog, where you can specify the Data Set to be considered. We have kept the default, “Most Recent”. The Derived Result is of the single value type, with one meter bar for each channel.

Example 4. Attach Geometric Mean (XY) to Frequency Response Level

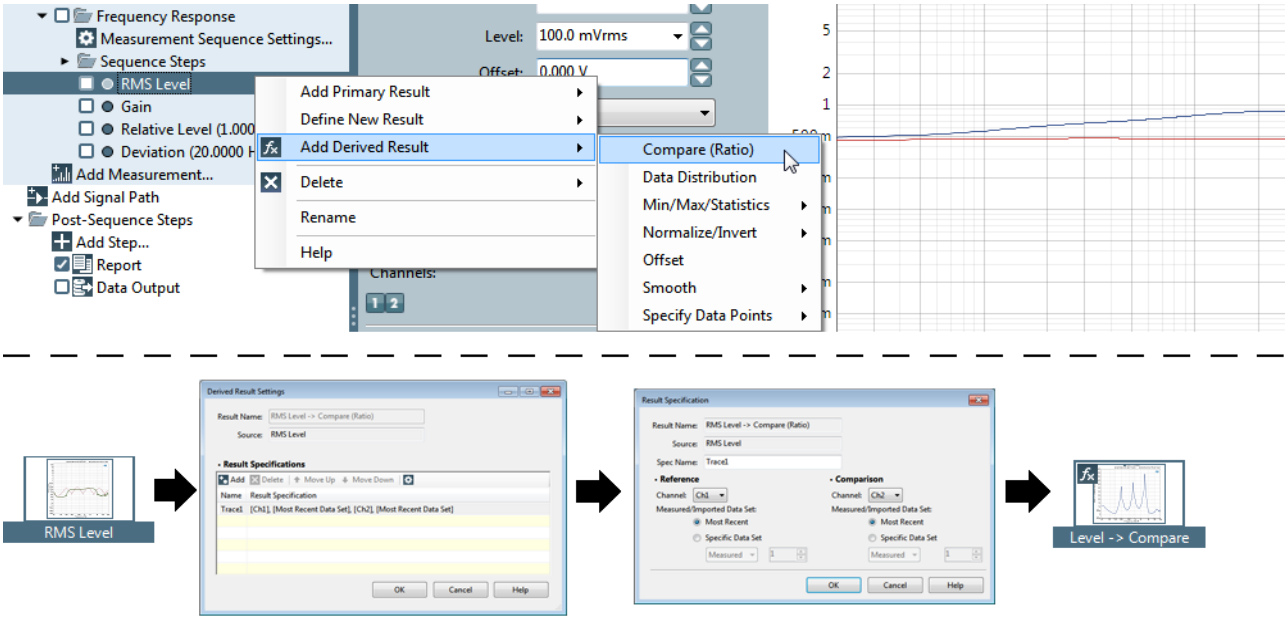
Attaching a Derived Result Geometric Mean (XY) Function to a Frequency Response Level result.



Attaching the Derived Result to the source result opens a Result Settings dialog, where you can specify one or more Result Specifications. We have made three specifications: one that returns the geometric mean across channels 1 and 2, for Data Set 1; one that returns the geometric mean across channels 1 and 2, for Data Set 2, and that returns the geometric mean across all channels for all Data Sets. The Derived Result is of the XY type, with one graph trace for each specification.

Example 5. Attach Compare (XY) to Frequency Response Level

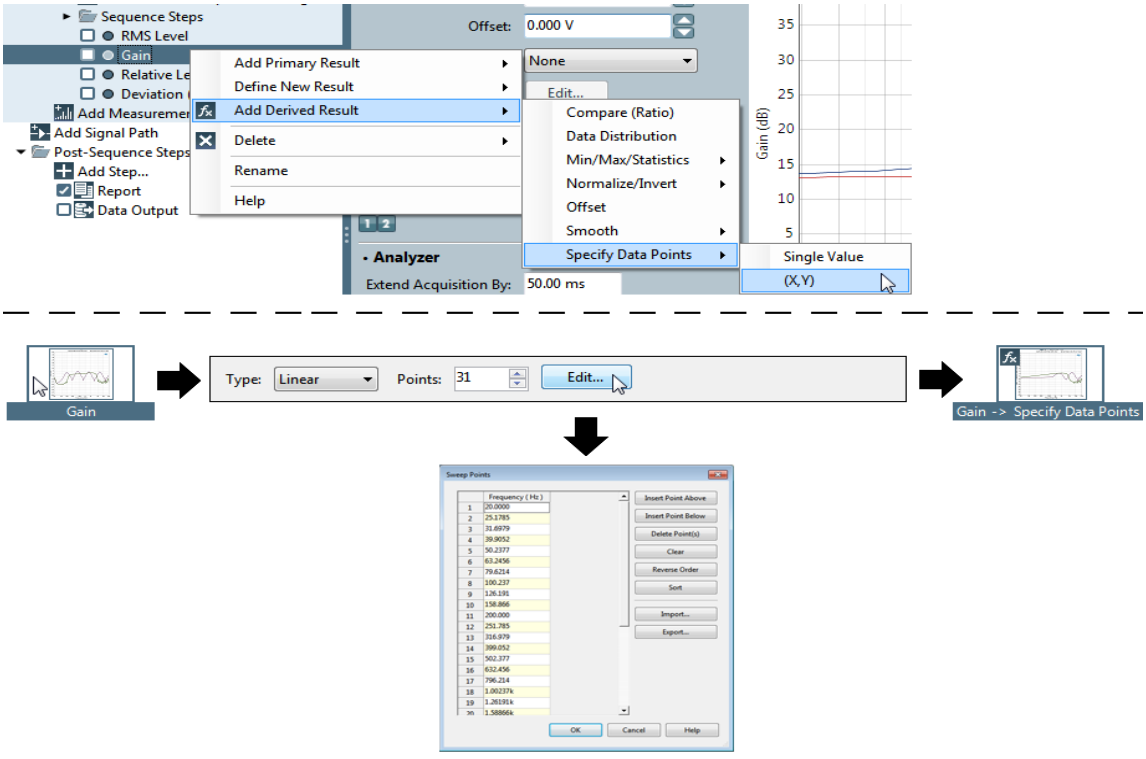
Attaching a Derived Result Compare (XY) Function to a Frequency Response Level result.



Attaching the Derived Result to the source result opens a Result Settings dialog, where you can specify one or more Result Specifications. We have only one specification, the default, which compares the data in channel 1 with the data in channel 2, for the most recent data set. The Derived Result is of the XY type, with one graph trace for each specification.

Example 6. Attach Specify Data Points (XY) to Frequency Response Gain

Attaching a Derived Result Specify Data Points Function to a Frequency Response Gain result.



Attaching the Derived Result to the source result places Type and Points fields on the Result Settings bar, where you can specify Linear or Logarithmic and the number of points, or you can open a Sweep Points dialog to add or delete specific points. We have kept the defaults. The Derived Result is of the XY type, with all traces redrawn with the new data points.

Limits

Overview:

Limits are preset values to which measurement values are compared during testing. In APx500, limit values or limit curves can be attached to a measurement. As you make the measurement, you can see whether or not the measurement values exceed your limits.

You can apply both upper and lower limits to an APx measurement. These are used as Pass / Fail criteria when comparing a device to specifications or to a "golden unit."

When a measurement or sequence is run, measurements that fall entirely between limits are marked as **Passed**. Measurements that exceed limits are marked as **Failed**.

Depending upon settings in a measurement's **Measurement Sequence Settings** (see page 538), the sequence behavior may be affected by result pass or fail status. The sequence may cancel, continue, prompt for a retry, or automatically retry.

Go to page 563 for more information about using pass/fail limits in a sequence.

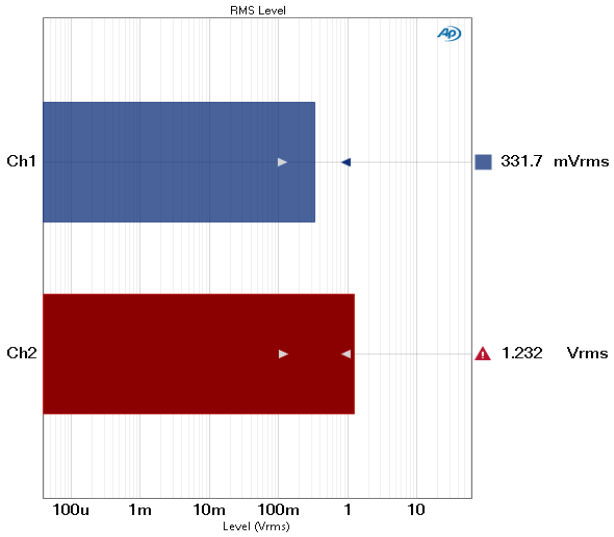
Limits for meter graph results

You create limits for meter (single value) results by specifying a single point on the result scale as an upper or lower limit. Meter limits can be marked on a meter graph with the **Draw Limits** tool, or entered in a table in the **Edit Limits** dialog.

Limit Markers

In a meter measurement, limits are displayed as arrows overlaid on the meter bar graph. A measurement that exceeds a limit changes the color of the meter bar to dark red.

Limits can be moved by grabbing a limit arrow using the mouse pointer and dragging it to a new position.

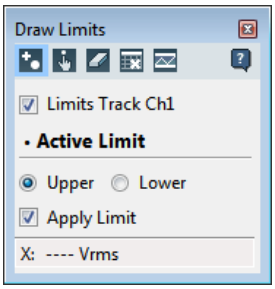


Limit arrow markers on a meter graph.

Drawing limits for meter results

Select **Draw Limits** from the **Draw/Edit Limits** menu on the meter graph toolbar, or right-click on the graph and select the **Draw Limits** item from the menu.

Drawing a limit point



By default, **Draw Limits** is set to **Limits Track Ch1** and **Active Limit** to **Upper**. When no limit has yet been created, only the **Add Points** tool is available.

- With the **Add Points** tool selected, click in the graph to create a limit point.

- Use the **Move Points** tool to drag an existing point to a new location.
- Use the **Erase Points** tool to remove a point.
- Click **Delete Limit** to entirely remove a limit.
- Click the **Edit Limits** button to go to the **Edit Limits** dialog box, where you can enter and edit limits by keyboard entry.

Limits Track Ch1

With **Limits Track Ch1** set, all channels will have these limit points attached. To set different limits (or have no limits) for other channels, deselect **Limits Track Ch1** and move the points on the graph independently.

Active Limit

Select the **Upper** or **Lower Active Limit** (the limit that the tools in this dialog are currently affecting) using the selections here.

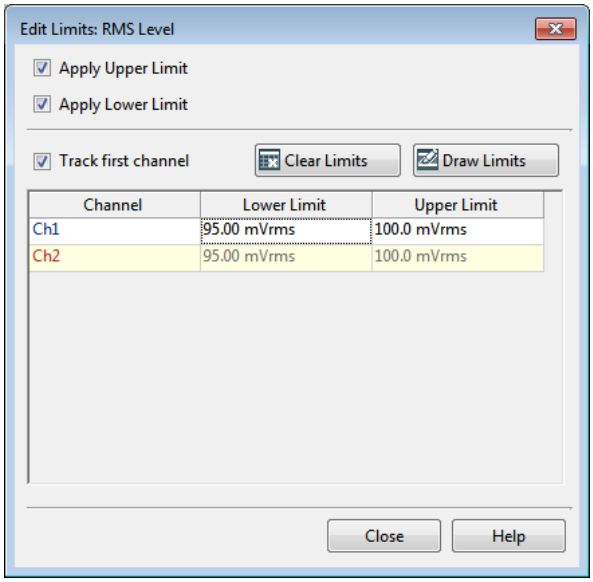
Apply Limit

Limit data that is not applied remains attached to a result but is not visible on the graph. In Sequence Mode, a limit must be applied to be active in a sequence and to display Pass/Fail markers in the Navigator.

To apply a limit to a graph and sequence, check the **Apply Limit** checkbox.

Edit limits for meter results

Select **Edit Limits** from the **Draw/Edit Limits** menu on the graph toolbar, or right-click on the graph and select the **Edit Limits** item from the menu.



The **Edit Limits** for meters (single-value measurements) dialog allows you to enter or edit limits by keyboard entry into a limit grid. If you have used the **Draw Limits** tool before opening **Edit Limits**, the values for the limit points you have drawn will be shown in the limits grid.

Apply Upper/Apply Lower Limit

Limit data that is not applied remains attached to a result but is not visible on the graph. In Sequence Mode, a limit must be applied to be active in a sequence and to display Pass/Fail markers in the Navigator.

To apply limits to a graph and sequence, check either or both **Apply Upper Limit** or **Apply Lower Limit** checkboxes.

Track First Channel

If **Track First Channel** is checked, the limit values for channel 1 are copied to the other channels, and the limit values for channels greater than 1 cannot be edited. Any changes made to channel 1 are reproduced in the other channels. Uncheck **Track First Channel** to enable different limit settings for each channel.

Clear Limits

To remove all limit points, click **Clear Limits**.

Draw Limits

Click **Draw Limits** to open the **Draw Limits** dialog.

Creating New Limits

New limits can be created in the **Edit Limit** dialog by entering limit values directly into the grid.

Limits for XY graph results

You can apply upper and lower limits to APx500 XY graphs. These can be used as Pass / Fail criteria when comparing a device to specifications or to a "golden unit."

Limits for XY graph results are themselves XY curves, although a limit might be *scalar*, a flat curve with only one Y value across all X values.

Single- and dual-axis XY graphs

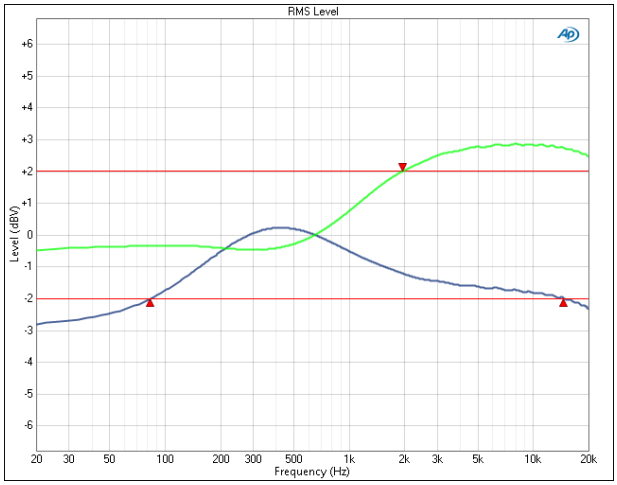
APx graphs can plot two results on the same display, using a left and right Y-axis. The following topics point out the differences in drawing, editing and applying limits when working with one and two Y-axis displays.

> *Right Axis Limit settings are only available for graph results that use two Y-axes.*

Limit Display

In an XY measurement, limits are displayed as red traces overlaid on the XY graph. A measurement that

crosses a limit is marked with a red triangle at the intersection.



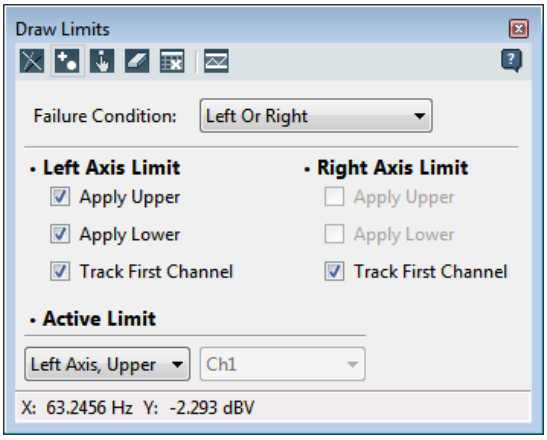
XY limit curves can be drawn directly on a graph display with the **Draw Limits** tool, entered from the keyboard in the **Edit Limits** dialog, copied from the current graph data, or imported from a data file.

Offsetting limits

Once created, an XY limit can be offset (its Y values can be shifted up or down in a defined way).

Drawing limits for XY graphs

Select **Draw Limits** from the **Edit Limits** menu on the graph toolbar, or right-click on the graph and select the **Draw Limits** item from the menu.



Draw Limits toolbar

- When no limit has yet been created, the **Draw Limit** tool is selected. Drag the mouse pointer across the graph to create an upper limit curve. After this first entry, the **Add Points** tool is selected.
- Use the **Add Points** tool to extend the curve. Click the mouse pointer on the graph to add a point.
- Use the **Move Points** tool to drag an existing point to a new location.
- Use the **Erase Points** tool to remove a point.
- Click **Delete Limit** to entirely remove a limit curve.
- Click the **Edit Limits** button to go to the **Edit Limits** dialog box, where you can import, export, offset and edit limits by keyboard entry.

Failure Condition (for 2-axis graphs only)

You can specify whether a limit failure is flagged when a point on either the left or right axis exceeds a limit, or when points on both left and right axes exceed a limit, for the same X value.

Left Axis Limit / Right Axis Limit

Right Axis Limit settings are only available for graph results that use two Y-axes.

Apply Upper/Apply Lower

Limits remain attached to a result until deleted. However, for a limit to be visible and active, it must be applied. Check the appropriate checkboxes. In Sequence mode, a limit must be applied to be active in a sequence and to display Pass/Fail markers in the Navigator.

By default, **Draw Limits** is set to **Track First Channel** and **Active Limit: Upper** or **Left Axis, Upper**.

Track First Channel

With **Track First Channel** set, all channels will have these limit curves attached. To draw different limit curves (or have no limit curves) for other channels, deselect **Track First Channel**.

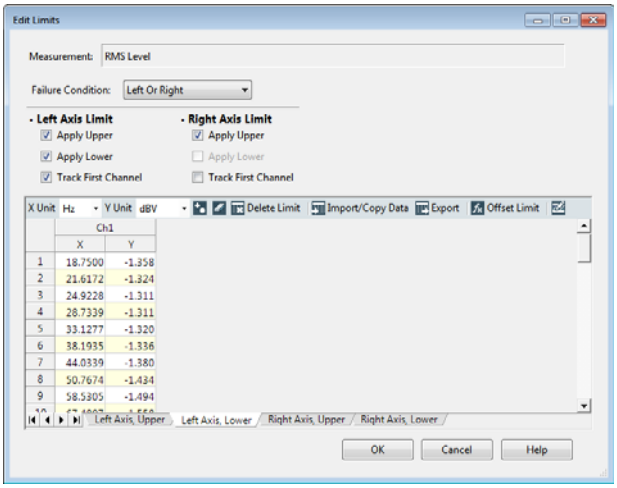
Active Limit

Select the **Active Limit** (the limit that the tools in this dialog are currently affecting) using the menu selections here.

Editing Limits for XY graphs

Select **Edit Limits** from the Edit Limits menu on the graph toolbar, or right-click on the graph and select the **Edit Limits** item from the menu.

The **Edit Limits** dialog provides tools to create, edit, import, export and offset limits. You can also copy current graph data to create new limits. If you have used the **Draw Limits** tool before opening **Edit Limits**, the values for the limit points you have drawn will be shown in the limits grid.



Failure Condition (for 2-Y axis graphs only)

You can specify whether a limit failure is flagged when a point on either the left or right axis exceeds a limit, or when points on both left and right axes exceed a limit, for the same X value.

Left Axis Limit / Right Axis Limit

Right Axis Limit settings are only available for graph results that use two Y-axes.

Apply Upper / Apply Lower

Limits remain attached to a result until deleted. However, for a limit to be visible and active, it must be applied. Check the appropriate checkboxes. In Sequence mode, a limit must be applied to be active in a sequence and to display Pass/Fail markers in the Navigator.

To apply limits to a graph and sequence, in **Left Axis Limit** (or **Right Axis Limit**, when available) check either **Apply Upper** or **Apply Lower** (or both) checkboxes.

Track First Channel

If **Track First Channel** is checked, the X and Y limit values for channel 1 are copied to the other channels, and the limit values for channels greater than 1 cannot be edited. Any changes made to channel 1 are reproduced in the other channels. Uncheck **Track First Channel** to enable different limit settings for each channel.

The Limits Grid; Selecting Upper or Lower Limits

Most of the **Edit Limits** dialog is taken up by the limits grid. The limits grid displays the X and Y values for each current limit point for all channels.

Tabs

Upper and lower limits for left and right axes are available on tabbed grid pages, with the tabs located at the bottom of the grid.

Creating New Limits

New limits for XY graphs can be created in four ways:

- Drawing limits from the **Draw Limits** dialog.
- Adding a limit point into the limit grid and entering a value into the grid. See **Add Point**, below.
- Importing a limit curve from an external limits file. See **Import/Copy Data**, below.
- Importing a limit curve by copying the current graph data into the limits grid. See **Import/Copy Data**, below.

You can add or delete limit points using the **Add Point** or **Delete Points** tools. You can edit the existing X or Y values of any point by entering a new value in the corresponding cell.

Add Point

Each horizontal line in the Limit Grid represents one limit point. Select a point by placing the cursor on a line of data on the grid. Click **Add Point** to add a new point below the selected point.

Delete Points

To remove selected limit points from the grid, click **Delete Points.**

Delete Limit

To remove all limit points from this tab of the grid, click **Delete Limit.** This deletes this limit, but other limits (Upper or Lower, for example) are retained.

Import/Copy Data

This button opens the **Import Limit Data** dialog, discussed on page 675.

Export

Limit data can be exported to external files such as an Excel file or CSV file. Each set of data (Upper Left, Lower Left, Upper Right, Lower Right) is exported saved to a separate file; select the appropriate limit source page for export, using the tabs at the bottom of the grid.

These files can be edited in Microsoft Excel or a text editor and imported back into APx500 if desired.

Offset Limits

An XY limit can be offset, having its Y values shifted up or down in a defined way to create related limit curves.

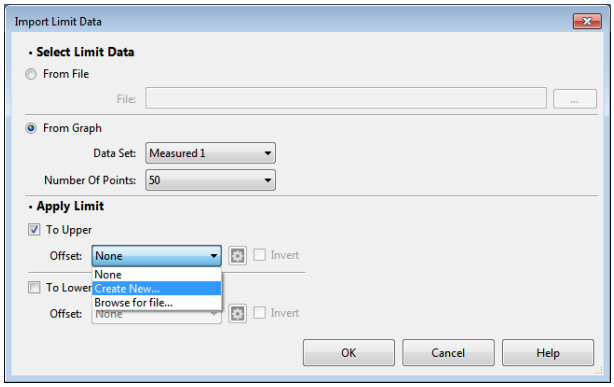
Offset Limits opens the Offset Limit Data dialog (see below), which allows you to offset measured or imported data.

Want more Points?

Here's a tip if you want very high-resolution limits. Export the Graph Data to an Excel or CSV text file, then import that file into the Limit Editor, using All Points in the data specification. Now use Offset Limits to create upper and lower limits

Import Limit Data dialog

From the **Edit Limits** dialog, click **Import/Copy Data** to open the **Import Limit Data** dialog.



Rather than drawing limit data or creating data in the limit grid, you can import limits from external data files, or you can copy convert data from the current measurement and convert it into a limit curve.

Supported file formats include Microsoft Excel spreadsheets (*.xlsx or *.xls) and comma-delimited text files (*.csv), as well as the AP2700 and ATS-2 formats (*.adx) and (*.atsx). Each set of data (Upper Left, Lower Left, Upper Right, Lower Right) is imported from separate files; use the tabs at the bottom of the Limit grid to select the corresponding target page. When limit data are imported, all current limit data are first cleared from memory.

When importing data from a file exported from an APx graph Data Grid, only data corresponding to the current measurement (THD Level, for example) will be imported. If there are no such data in the file, no limit data will be imported.

Select Limit Data

From File

Select **From File** to import limits from a data file. Enter the path and file name here, or browse to the file using the **Browse** button. Supported formats are Excel spreadsheets, CSV, or legacy Audio Precision data files.

From Graph

Select **From Graph** to create limits from data in the current measurement.

Data Set:

If the measurement has more than one data set, select the desired data set here.

Number of Points:

For continuous-sweep based measurements, there may be thousands of data points. A reduced set of data points can be imported from the graph data to define a limit. Choose

- **Same as Graph** (the default)
  This is the number of data points used to create the current graph. For continuous-sweep based measurements, Same as Graph is always fewer points than the number of points acquired.
- Or, choose a specific number of points (from **10** points to **500** points).

Apply Limit

To Upper / To Lower

Choose whether to apply the imported data to an upper limit, to a lower limit, or both.

Invert Data

If the limit is being applied to a relative result (such as Relative Level), an Invert Data checkbox will be available. Check the Invert Data checkbox to invert the limit data.

Offset:

When imported from a file, the data have arbitrary Y values, according to the values in the data file. When imported from the graph, the data have the same Y values as the current data in the selected data set. Depending upon data set choice, these values may be identical to the current result points and the limit curve may be drawn directly on the current result trace.

In either case, you will typically want to offset the limit curve above or below the result curve by scaling the limit Y values, using a constant offset value, or by using an offset curve.

- **"Filename.csv"**
  You can select a file already attached to the project (if any).
- **Create New...**
  This item opens the **Edit Limit Offset Curve** dialog (page 676).
- **Browse for file...**
  This opens a file browser where you can select an offset curve file to attach to the project.

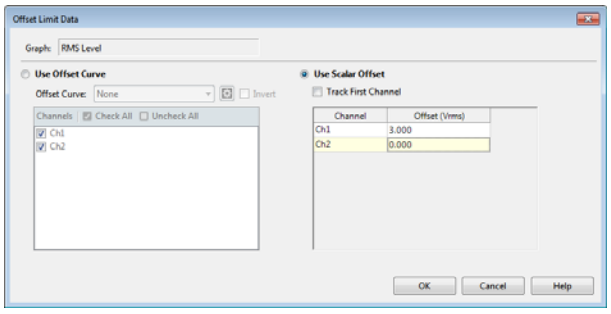
Properties

For an existing offset curve, the **Properties** button opens the **Edit Limit Offset Curve** dialog.

Invert Offset Curve

Check the **Invert** checkbox to invert the limit offset curve.

Offset Limits dialog



The Offset Limits dialog provides an easy way to raise or lower an entire limit curve (changing all the Y values) by a specific amount. This is useful if your current limit curve is of the correct shape, but requires a shift to either tighten or loosen your pass / fail criteria.

Scalar offsets

In most cases XY limits are offset up or down by a fixed amount; the Y values of an entire limit curve may be raised by 1 dB, for example. In APx, this is called a *scalar offset*. The Offset Limit dialog (see page 676) provides a table in which scalar offsets can be applied to existing limits on a per-channel basis.

Offset curves

You can also offset XY limits by a Y-value amount that varies with the X-value by applying an offset curve.

Using Offset Limits with copied graph data

When used with Copy Graph Data, Offset Limits is also a quick way to create limits that mimic measured data, such as that from a “golden unit.”

- In the Edit Limits dialog, select the upper limit grid page with the Selected Limit **Upper** button. Click **Copy Graph Data**. From the drop-down list, choose the number of points of data to be copied.
- Choose **Offset Limit**.
- Enter a value (perhaps +3 Vrms) in the Offset field for Channel 1. Click **OK**.
- Select the lower limit page. Repeat the previous steps for the lower limit, but enter –3 Vrms. Click OK.

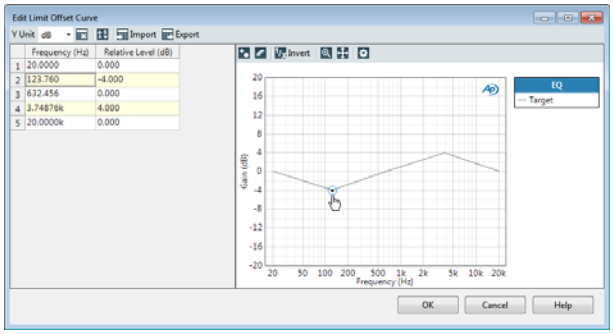
Upper limits are typically offset by positive voltage values; lower limits by negative voltage values. By following the steps in this example, you will have created a pair of limits about 6 Vrms apart surrounding the existing Channel 1 data.

Track First Channel

If the Track First Channel checkbox in the Offset Limits dialog is checked, all channels will be offset by the value entered in the Channel 1 offset field.

Note: When a limit offset is applied, the values in the Limits Grid are permanently changed to incorporate the offset. You may continue to edit the Limit values by changing values in the Limits Grid, by using the Draw Limits tool or by applying new limit offsets.

Edit Limit Offset Curve dialog



This dialog allows you to import, edit and export a limit offset curve/table. The points in the table are represented graphically by the trace in the graph.

You can make a scalar offset by setting the entire X-axis range to the same value, or design an offset curve by setting different Y values across the X range.

Clear Table

Clear Table removes all points (rows) from the table.

Order Values

Order Values sorts the table by X-axis value, ascending toward the bottom of the table.

Import

Import opens a file browser window to select and open compatible limit offset table files from disk. Excel, CSV and Audio Precision legacy *.adx and *.atsx are supported.

Export

Export opens a file browser window to save the current limit offset table as an Excel or CSV file.

Add Point

Add Point inserts a blank X-axis point (row) below the current selection in the table. Enter a level value for each new point. You can edit the X-axis frequency or level value arbitrarily.

For frequency X values:

- Minimum Frequency: 1 Hz
- Maximum Frequency: 1.04858 MHz

For level X values:

- Minimum Level: –100 dB
- Maximum Level: 100 dB

X-axis values must be unique, and ordered from low to high. See **Order Values**, above.

Delete Selected Points

Delete Selected Points removes the selected points (rows) from the table.

Invert

Use this control to invert the offset curve.

Zoom

Click on this button and then drag a marquee in the graph to zoom.

Revert Zoom/Pan

Click on this button to revert to the original zoom and pan views.

Graph Properties

This button opens a Graph Properties dialog, where you can select ranges, names and log/lin characteristics for the offset curve graph.

Examples

Centering new data within limits

In some applications, the general conformance of a trace to the shape of a set of limits is of importance, with absolute level less important or irrelevant.

You can automatically center new data within existing limits for the **Relative Level** results of these measurements:

- Acoustic Response (Chapter 24)
- Bandpass Frequency Sweep (Chapter 25)
- Continuous Sweep (Chapter 31)
- Frequency Response (Chapter 47)
- Multitone Analyzer (Chapter 64)
- Stepped Frequency Sweep (Chapter 81)

You can automatically center new data within existing limits for the **Linearity** results of these measurements:

- Bandpass Level Sweep (Chapter 27)
- Stepped Level Sweep (Chapter 82)

Refer to the measurement chapters for operational details.

Setting XY limits directly

You can draw limits or enter limit values directly in a limit table. For example, perhaps it is important that your DUT perform to an industry standard that requires that frequency response be ±3 dB below 100 Hz and above 15 kHz, and ±1 dB at points between.

You can simply draw or enter these limits in an APx500 measurement that provides frequency response results, and run the test or sequence. For the example here, values were entered in the Edit Limits dialog for the Frequency Response measurement. The limits grid showed these entries:

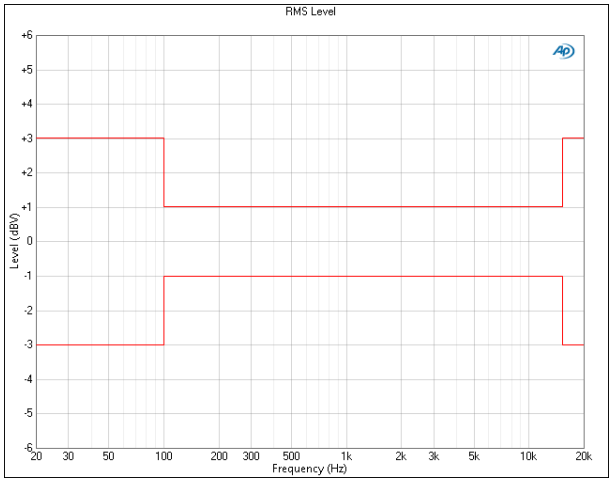
	Ch1	
	X	Y
1	20.0000	+3.000
2	100.000	+3.000
3	100.000	+1.000
4	15.0000k	+1.000
5	15.0000k	+3.000
6	20.0000k	+3.000

Upper

	Ch1	
	X	Y
1	20.0000	-3.000
2	100.000	-3.000
3	100.000	-1.000
4	15.0000k	-1.000
5	15.0000k	-3.000
6	20.0000k	-3.000

Lower

The limits might appear on your graph like this:



Creating limits from measurement data

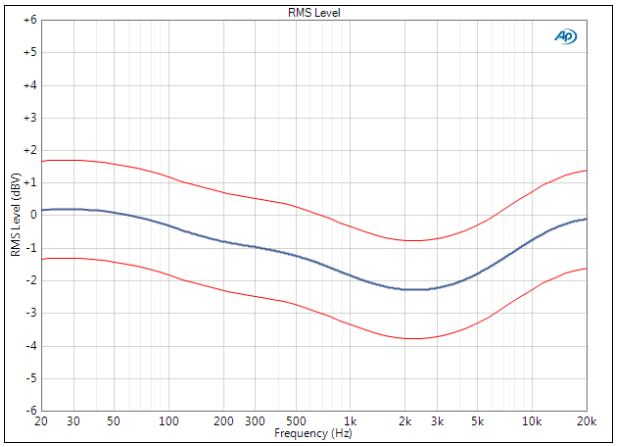
You may have a “golden unit,” a DUT that performs very well, that you would like to use as a standard for your Pass / Fail testing. For an X/Y graph, you can create limits from the measured response of this unit.

The example given below is for a curve derived from a measurement, with scalar limits applied. See Edit Limits > Import Limits on page 675 and Offset Limit on page 676 for other options.

For example, let’s use the frequency response of a golden unit.

- Measure the response.
- In the **Edit Limits** dialog, select **Import/Copy Data**.
- In the **Import Limit Data** dialog, choose **From Graph**. Set **Number of Points** to **50**.
- In the Apply Limit section, choose **To Upper**, and choose **Offset >**
- With the **Upper** tab selected, choose **Offset Limits**, select **Use Scalar Offset**, and enter **1.5**.
- With the **Lower** tab selected, choose **Offset Limits**, select **Use Scalar Offset**, and enter **–1.5**.

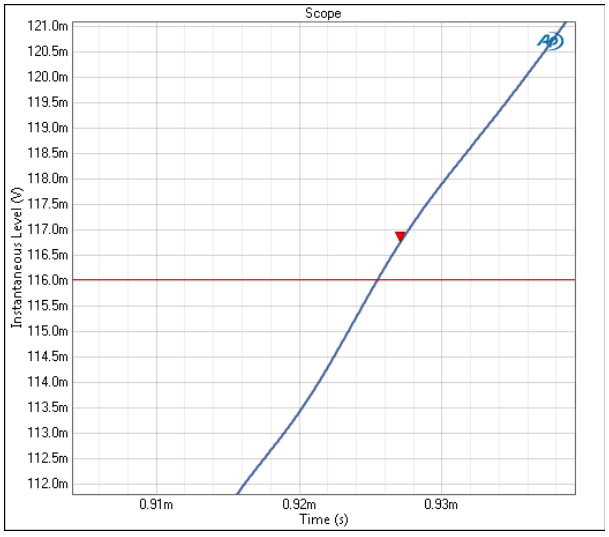
These limits appear on your graph as shown (the red lines are the upper and lower limits; the blue line in the middle is the golden unit data):



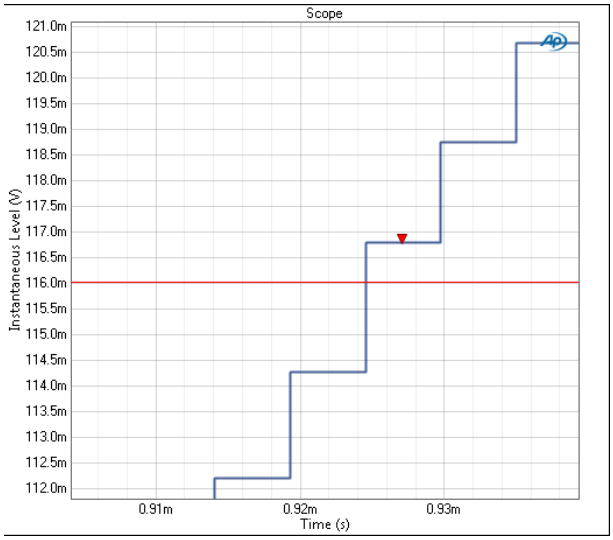
Creating offset limits from graph data is discussed in detail on page 676.

Interpolation and limit failure markers interaction

Limit failure markers (small red triangles) are placed at the first data point that exceeds a limit. When interpolation is on and the graph is zoomed in to high magnification, the limit failure markers may appear to be “floating” in the graph, not at the intersection of the limit line and the interpolated data trace. Although this appears to be an error, it is actually correct behavior. Turn interpolation **OFF**, and you will see that the marker is at the center of the first (real) data point that exceeds the limit. There is no data at the intersection, although interpolation suggests that there is.



Limit failure marker appearing to “float” above limit line when interpolation is ON.



Limit failure marker shown to be centered on data point when interpolation is OFF.

Aux Control

Overview

Aux Control (sometimes referred to as GPIO or General Purpose Input/Output) provides the capability to communicate with external devices, transmitting and receiving control commands via general-purpose 8-bit digital ports, available on 9-pin D-Sub connectors on the APx500 series instrument rear panel.

Aux Control Out

Aux Control Out commands can be set manually in the APx500 software, or can be set automatically in response to a specified condition in an APx500 sequence. Applications include control of mechanical devices such as turntables, mains power, indicator lights, annunciators, and so on.

Aux Control In

Aux Control In commands can be read visually in the APx500 software, or can be used to trigger actions in an APx500 sequence. Applications include input of operator controls (such as a foot switch) and reading of device states.

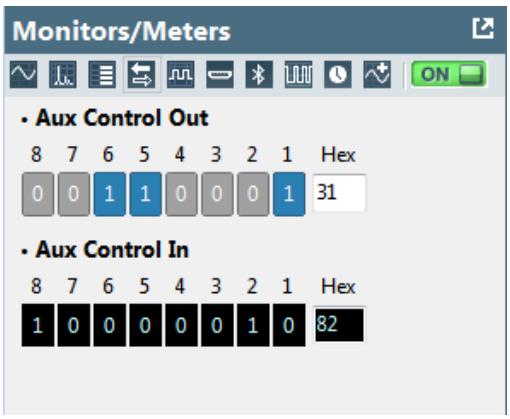
Reading and Setting Aux Control

Viewing and manually setting Aux Control has been placed in a Monitor view, which allows you to work with Aux Control without leaving the current Measurement view.

Go to the monitors and click the **Aux Control** button.

The Aux Control Monitor displays the current state of the Aux Control Out and Aux Control In bits. Bit states are shown both in a binary display and as the hex equivalent.

Additionally, Aux Control Out bits can be set from this monitor window, either by clicking on the binary display to toggle the state of a bit, or by entering the hex equivalent in the Hex: field.



Using Aux Control with Sequences

You can read or set Aux Control states from within a sequence step. See pages 543 and 548.

You can start a sequence with an Aux Control In command, and set the Aux Control out state at the beginning of the sequence. See page 565. You can also set the Aux Control out state at the end of a sequence, in response to pass, fail or sequence cancellation conditions. See page 566.

You can also cause sequence dialogs, such as Pass or Fail, to be closed with an Aux Control In command. See page Dialog Button Aux Control on page 537.



The Aux Control interface used in the Sequence has three states for each bit: 1, 0, and “X”. X is an “ignore” or “don’t care” setting.

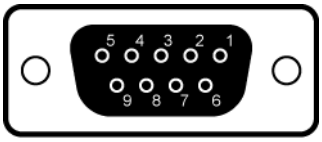
For an Aux Control In interface, X means that for that bit, either a 1 or a 0 will satisfy the pattern.

For an Aux Control Out interface, X means that the current state of the bit will be left unchanged.

Aux Control Physical Layer

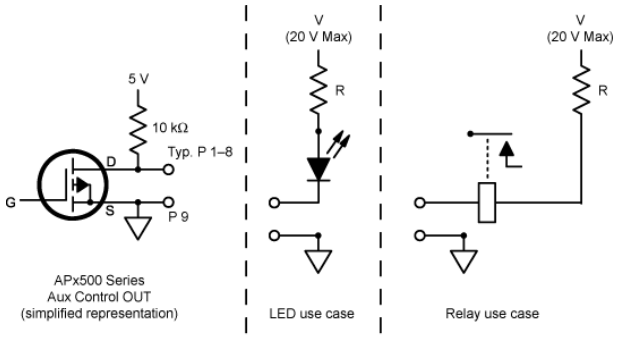
Aux Control Out

Aux Control Out uses open-drain relay drivers, capable of sinking 250 mA per pin. The On resistance is nominally 1.3 Ω. The open drain is pulled high to 5 V with a 10 kΩ resistor.



Aux Control Out DE-9F connector

Pin 9 is the common (ground) connection; pins 1–8 are numbered to correspond with the Aux Control bits 1–8.

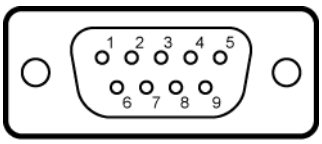


LEDs are typically connected using TTL driver circuits, but can be connected directly as shown above. V and R depend upon the LED selected and other circuit considerations.

Relays are typically connected using TTL driver circuits, but can be connected directly as shown above. If the relay package does not provide an internal suppression diode, be sure to protect the circuit by connecting one across the relay coil. V and R depend upon the relay selected and other circuit considerations. Maximum current per Aux Control Out pin is 250 mA.

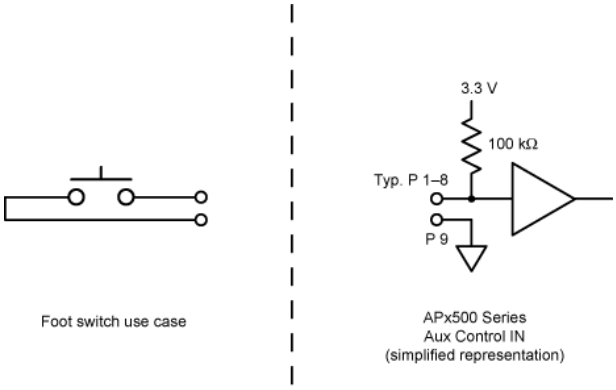
Aux Control In

The input port is a TTL-compatible interface with 5 V tolerant inputs. Vinh (high) is 2.0 V and Vinl (low) is 0.8 V. The inputs are pulled high with a 100 kΩ resistor to 3.3 V. Maximum continuous input voltage is 5.2 V.



Aux Control In DE-9M connector

Pin 9 is the common (ground) connection; pins 1–8 are numbered to correspond with the Aux Control bits 1–8.



Input switches and sensors are often connected using TTL driver circuits, but can be connected directly as shown above.

Switchers

Introduction

Audio Precision manufactures optional switchers that can be used to connect or disconnect many signal channels (up to 192 input and 192 output channels) to the instrument inputs or outputs, under software control.

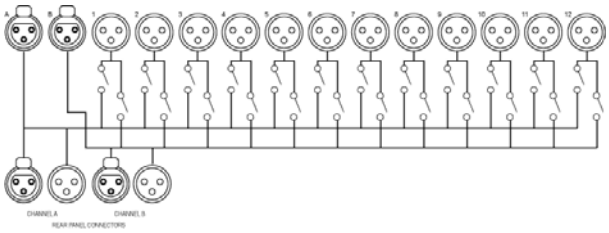
This chapter discusses the APx500 switcher control interface and provide an overview of switcher features and applications.

Switchers can also be configured from within a sequence by adding a Switcher Configuration step. See page 552.

See the SWR-2755 User's Manual (provided with the switcher and available on the Audio Precision Web site) for more information.



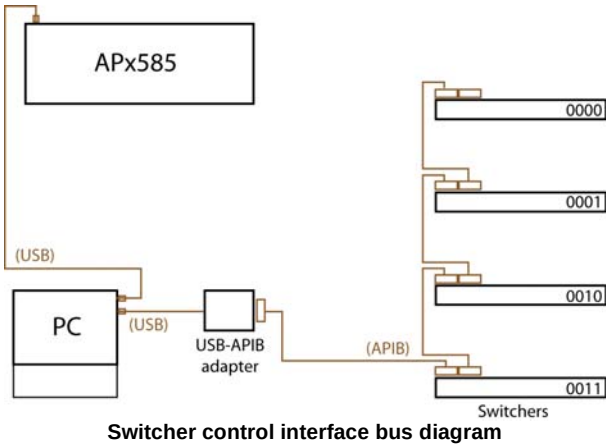
Balanced output switcher SWR-2755M



Block diagram of SWR-2755M switcher

Connecting the switcher to the control computer

Audio Precision switchers use the Audio Precision Interface Bus (APIB) for switcher control. To use a switcher with APx500, you must have at least 2 USB 2.0 ports on the control computer. One of the ports must be connected to the APx500 instrument; the other port must be connected to an Audio Precision USB-APIB adapter, which can then be connected to one or more switchers using APIB. The USB-APIB is an accessory interface available from Audio Precision.



Switcher control interface bus diagram



Audio Precision USB-APIB adapter

Setting switcher addresses

APx500 can control up to 16 input and 16 output switchers, for a total of 192 switched input channels and 192 switched output channels. Each switcher is identified by setting a DIP switch on the SWR-2755 hardware to a binary address, and entering the matching binary address in the software.

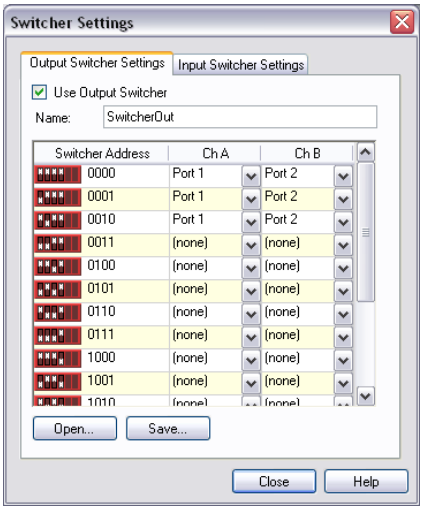
Switcher address settings

Each switcher must be assigned a unique binary address. On the switcher rear panel is a DIP switch with 6 switches; the four switches to the left set the binary address for that unit. To address a switcher from the APx500 software, the binary address must

match on the switcher hardware and the switcher address controls in the software.

Note that the DIP switch orientation on the hardware is reversed from the digit order in ordinary binary notation, and that 0=UP and 1=DOWN: binary number 0111 corresponds to switch settings of DOWN-DOWN-DOWN-UP. For clarity, both DIP switch position and binary number are shown in the control panel.

From Signal Path Setup, click **Switcher Settings**. Choose the **Output Switchers Settings** tab or the **Input Switchers Settings** tab to view a Switcher panel.



Switcher Address

This is the binary address for each switcher hardware unit. On the switcher rear panel is a DIP switch with 6 switches; the four switches to the left set the binary address for that unit. To address a switcher from the APx500 software, the binary address must match on the switcher hardware and the switcher address controls in the software.

Switcher Channels A and B

Each switcher has 2 channels called Channel A and Channel B, that are used to connect to 2 APx 500 series outputs (for an output switcher) or inputs (for an input switcher). These are connected to the switcher ports by internal relays, under the APx500 software control.

Switcher Ports

Each switcher has 12 ports that are used to connect to DUT channels according to your test requirements. An input switcher's ports are designed to connect to a DUT's output channels, and an output switcher's ports are designed to connect to a DUT's input channels. For each APx500 signal path, you can make a port selection for Channel A and Channel B for each addressed switcher.

All But ChA

The Channel B port selection list includes All But ChA as a choice. This selection connects switcher Channel B to all ports except the current port connected to Channel A.

Saving switcher settings

APx500 switcher settings are always saved with the project file, and are included in the project when the file is re-opened.

Additionally, output and input switcher settings can be saved as individual files from the Signal Path Switchers panels. Output switcher settings files have the filetype *.swo, and input switcher settings files have the filetype *.swi.

Click the **Save** button to save the current switcher settings as a separate file. Click the **Open** button to open previously saved switcher files.

Creating a signal path for each step

Each Navigator signal path sets one state for each of the switchers connected, set in the Advanced Settings panel for that signal path. For example, the input switcher at address 0000 could have its ChA connector switched to Port 1, and its ChB connector switched to Port 2.

To switch to two different DUT channels, create a second Navigator signal path using the same 0000 switcher address, but set ChA to Port 3 and ChB to Port 4. Make the necessary audio connections to match. When you move from Signal Path 1 to Signal Path 2, the switcher will change to Ports 3 and 4.

Switcher example

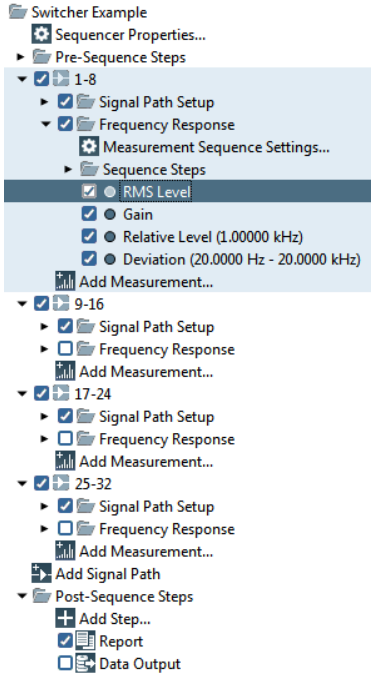
The block diagram on the next page shows an APx585 testing a 32x8 mixer using four output switchers. Generator pairs 1–2 are connected to Channels A and B on switcher 0000, 3–4 to A and B on switcher 0001 and so on.

The switcher ports are connected to various mixer inputs. Ports 1–2, 3–4, 5–6 and 7–8 from 0000, for example, carry Generator signal from 1–2, and are wired to mixer inputs 1–2, 9–10, 17–18, and 25–26.

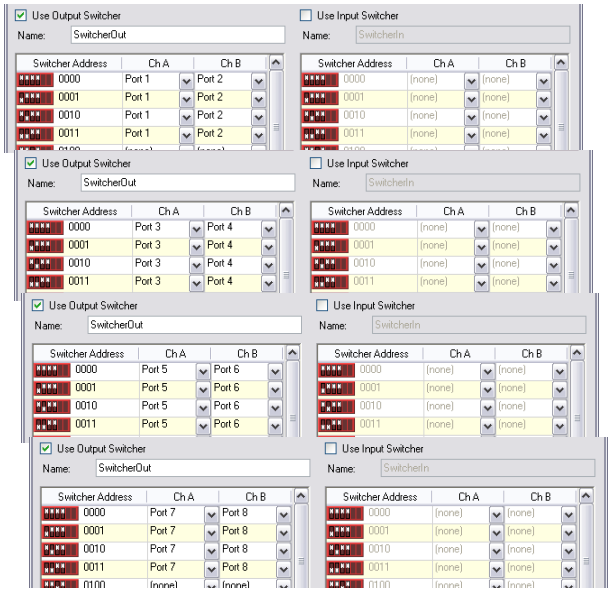
The eight outputs from the mixer are connected directly to the analyzer inputs. If there were more outputs to be tested, input switchers could be added to the setup.

The four Navigator signal paths shown here shift the switchers through the four states necessary to test all the mixer channels. The four switcher address config-

urations show the switcher states specified in each of the four signal paths.



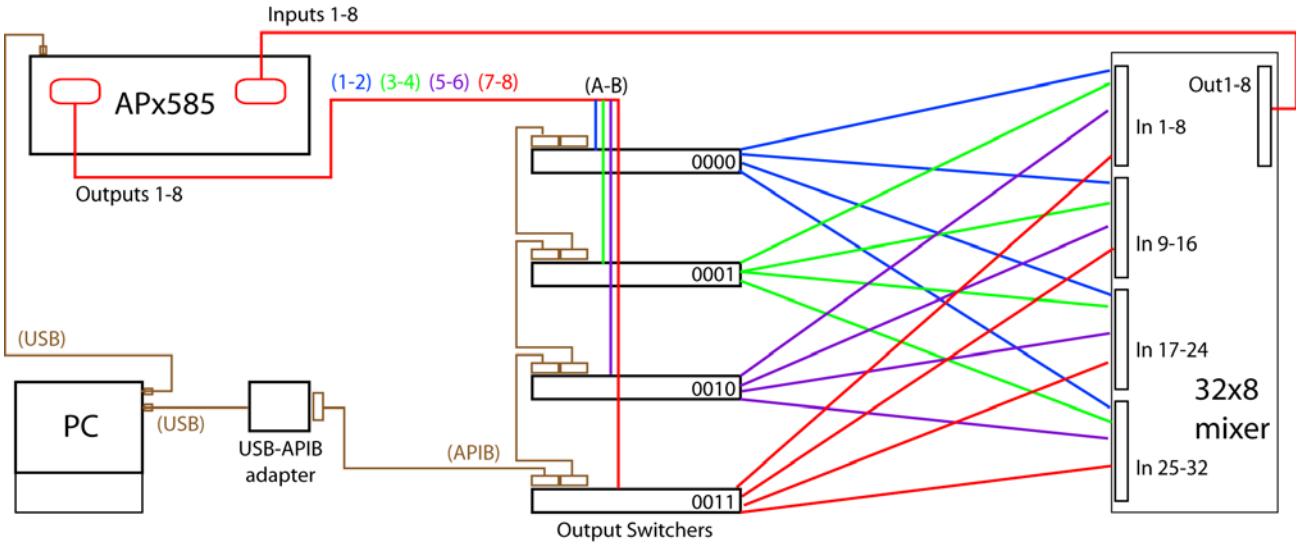
Four Signal Paths in the Navigator



Four sets of switcher settings

When this sequence is run, each signal path performs its measurements on eight channels at a time, and then switches to the next signal path and the next eight channels.

All results are shown in the sequence report.



Block Diagram

Units of Measurement

Unit Domains

Many different units of measurement are used to describe audio signals and audio system components and characteristics. Values can be expressed in absolute terms such as the volt, the watt, the ohm, the hertz and the second. Values can also be displayed as the ratio of one value to another using units like the decibel or percent. In addition, APx500 provides relative units which express the relationship of a value to a user-supplied reference, or to another APx500 setting.

Measurements using Audio Precision analyzers fall into one of eight *unit domains*. Each unit domain has a *base unit*, from which the equivalent units are calculated.

Depending on the current APx500 view and the parameter you are setting or measuring, different units choices in different unit domains will be available. When you request a value in different units, APx500 recalculates the value.

Unit domains used in APx500:

- Analog Level
- Digital Level
- Ratio
- Ratio, cross-domain
- Relative
- Frequency
- Phase
- Time
- Acoustic
- No Unit (Impulse Response only)

Units: Analog Level

V—volts (instantaneous voltage)

This unit is used in the APx500 DC Level measurement, and in the time domain views, including Signal Monitors: Scope, Signal Analyzer: Scope; in Acoustic Response: Impulse Response and Acquired Waveform, Continuous Sweep: Impulse Response and Acquired Waveform, Multitone: Acquired Waveform, Multitone: Noise Density and Multitone: DC Level measurements.

Vrms—volts (rms)

This unit is used in APx500 generator settings and analyzer measurements. This setting accurately calibrates the generator output in Vrms when a sine wave is being generated. When using other waveforms (such IMD twin-tone stimulus signals) with Vrms, the signal has the same peak-to-peak voltage as a sine wave at that setting. For the analog outputs, Vrms is an open-circuit value and the actual output voltage will be less with a load. In analyzer measurements, Vrms readings are always made using an rms detector and are correct for any waveform.

Vp—volts (peak)

This unit is used in APx500 generator settings. This calibration is arithmetically scaled to the Vrms value by the square root of 2, approximately 1.414. Since non-sinusoidal waveforms are calibrated to have the same peak-to-peak value as a sine wave, the Vp unit should be correct for most non-sinusoidal waveforms. For the analog outputs, Vp is an open-circuit value and the actual output voltage will be less with a load.

Vpp—volts (peak-to-peak)

This unit is used in APx500 generator settings. This calibration is arithmetically scaled to the Vrms value by 2 times the square root of 2, approximately 2.828. Since non-sinusoidal waveforms are calibrated to have the same peak-to-peak value as a sine wave, the Vpp unit should be correct for most non-sinusoidal waveforms. For the analog outputs, Vpp is an open-circuit value and the actual output voltage will be less with a load.

dBV—decibels relative to 1.000 Vrms

This is usually the correct choice of units for audio levels in consumer audio equipment and systems. The dBV is a voltage unit and, like the dBu, is independent of circuit impedance. For the analog outputs, dBV is

an open-circuit value and the actual output voltage will be less with a load. See **More about decibels** on page 689.

dBu—decibels relative to 0.7746 Vrms

This is usually the correct choice of units for audio levels in professional audio equipment and systems. The dBu is a voltage unit and, unlike the dBm, is independent of circuit impedance. For the analog outputs, dBu is an open-circuit value and the actual output voltage will be less with a load. See **More about decibels** on page 689 and **More about dBm vs. dBu** on page 689.

dBm—decibels relative to 1 mW

dBm is rarely the correct choice of units for audio measurements. dBm units are often Improperly chosen when dBu units should be used. See dBm versus dBu on page 689.

Like watts, the dBm is a unit of power used in APx500 generator settings and analyzer measurements.

dBm as generator output units

The APx generator outputs have a source impedance of 100 Ω for the balanced connections and 50 Ω for the unbalanced connections. When using dBm as a generator unit, the internal generator level is set to the value that produces the desired power in the load, calculated for the balanced (100 Ω source impedance) outputs. For a correct calculation, you must enter the value of the load resistance in the Reference Levels > Advanced Settings dialog. 600 Ω is the most common circuit impedance for dBm measurements.

As mentioned, this calculation is made correctly for the balanced outputs. The unbalanced outputs present a source impedance of 50 Ω, which is not considered in the dBm units calculation. When using the unbalanced outputs with dBm generator units, the power in the load will not match the value specified in the generator Level field.

Note: units of power (dBm and watts) in generator output settings are features included in Audio Precision instruments as a nod to tradition, but are rarely useful. We do not recommend the use of dBm or watts as generator units.

dBm as analyzer input units

For the analyzer inputs, dBm is accurately measured and calculated for both the balanced and unbalanced inputs. For a correct calculation, you must enter the load impedance at the output of your DUT in the Reference Levels > Advanced Settings dialog. 600 Ω is the most common circuit impedance value for dBm measurements.

W—watts

Like dBm, the watt is a unit of power used in APx500 generator settings and analyzer measurements.

watts as generator output units

The APx generator outputs have a source impedance of 100 Ω for the balanced connections and 50 Ω for the unbalanced connections. When using watts as a generator unit, the internal generator level is set to the value that produces the desired power in the load, calculated for the balanced (100 Ω source impedance) outputs. For a correct calculation, you must enter the value of the load resistance in the Reference Levels > Advanced Settings dialog. 8 Ω is the default load for watts measurements.

As mentioned, this calculation is made correctly for the balanced outputs. The unbalanced outputs present a source impedance of 50 Ω, which is not considered in the watts units calculation. When using the unbalanced outputs with watts generator units, the power in the load will not match the value specified in the generator Level field.

Note: units of power (dBm and watts) in generator output settings are included features in Audio Precision instruments as a nod to tradition, but are rarely useful. We do not recommend the use of dBm or watts as generator units.

watts as analyzer input units

For the analyzer inputs, watts are accurately measured and calculated for both the balanced and unbalanced inputs. For a correct calculation, you must enter the load impedance at the output of your DUT in the Reference Levels > Advanced Settings dialog. 8 Ω is the default load for watts measurements.

ohm (Ω)

The ohm is the unit of electrical resistance (R). Its relationship to voltage (V) and current (I) in a circuit is stated by ohm's law, V=IR.

Units: Digital Level

FS—digital full scale

Digital full scale is defined in terms of a sine wave whose peak just reaches the maximum positive code. The rms value of such a sine wave is 1 FS. See **More about Digital Level units** on page 689.

Other Audio Precision analyzers have used the term FFS (Fraction of Full Scale) for this unit of digital level. 1 FFS = 1 FS.

%FS—per cent of full scale

%FS is the ratio of the rms value of the signal to 1 FS, expressed as a decimal fraction and multiplied by 100. A sine wave half the value of full scale would be indicated as 50 %FS.

dBFS—decibels relative to full scale

This is a logarithmic measurement defined as 20 log10(A/B), where A is the rms level of the signal to

be expressed and B is level of a full scale sine wave (1 FS). For example, a signal at 0.1 FS equals –20 dBFS. dBFS is probably the most common unit of measurement used for digital audio amplitudes. 0 dBFS is the full-scale reference.

D—instantaneous digital level

D, the instantaneous digital level, is defined in terms of an instantaneous signal whose peaks just reach the maximum digital codes. The peak value of such a signal that reaches the maximum positive code is +1 D; the peak value of such a signal that reaches the maximum negative code is –1 D.

This unit is used in the APx500 DC Level measurement, and in the time domain views, including Signal Monitors: Scope, Signal Analyzer: Scope; in Acoustic Response: Impulse Response and Acquired Waveform, Continuous Sweep: Impulse Response and Acquired Waveform, Multitone: Acquired Waveform, Multitone: Noise Density and Multitone: DC Level measurements.

hex—hexadecimal

The hex unit express the instantaneous numerical value of the embedded audio sample in base 16 (hexadecimal) notation, and is used for Constant Value waveform levels, and in the APx500 DC Level measurement, and in the time domain views, including Signal Monitors: Scope, Signal Analyzer: Scope; in Acoustic Response: Impulse Response and Acquired Waveform, Continuous Sweep: Impulse Response and Acquired Waveform, Multitone: Acquired Waveform, Multitone: Noise Density and Multitone: DC Level measurements.

Units: Ratio

X/Y

The X/Y unit is the simple ratio between two numbers.

%—per cent

The per cent unit expresses the ratio of two numbers multiplied by 100. For example, a ratio of 0.7 is expressed as 70 %.

ppm—parts per million

Parts per million expresses the ratio of two numbers, multiplied by 1,000,000. For example, a ratio of 0.00005 equals 50 ppm.

dB—decibel

The decibel is always a ratio of two amplitudes, expressed logarithmically. For voltage amplitudes, decibels are computed by dB=20 log10(V1/V2). See **More about decibels** on page 689.

Specific ratio units, such as dBu or %Hz, are discussed in more detail in the relevant sections on amplitude or frequency units.

Units: Ratio, cross-domain

FS/Vrms

For cross-domain measurements, the FS/Vrms unit is the ratio between output and input levels, for devices (such as an ADC) that have an analog input and a digital output.

Vrms/FS

For cross-domain measurements, the Vrms/FS unit is the ratio between output and input levels, for devices (such as a DAC) that have a digital input and an analog output.

D/V

For cross-domain impulse response measurements, the D/V unit is the relationship between output and input for devices (such as an ADC) that have an analog input and a digital output.

V/D

For cross-domain impulse response measurements, the V/D unit is the relationship between output and input for devices (such as a DAC) that have a digital input and an analog output.

See Analog Level units earlier in this chapter for more information about V and Vrms. See Digital Level units for more information about FS and D.

Units: Relative

dBrG

dBrG is a user memory, storing decibels relative to a user-defined output (generator) reference level setting. There are two independent dBrG memories: one for an analog output level and one for a digital output level. See **More about decibels** on page 689.

dBrA and dBrB

dBrA and dBrB are user memories, storing decibels relative to user-defined input (analyzer) references. See **More about decibels** on page 689.

You can manually enter dBrA and dBrB values in the fields in the Reference Levels view, or you can set them from measured input levels using the Set buttons in the same view. There are two independent pairs of dBr memories: one pair for analog input levels and one pair for digital input levels.

dBSPL1 and dBSPL2

dBSPL1 and dBSPL2 are user memory references just like dBrA and dBrB, with nomenclature to support Sound Pressure Level work.

Units: Frequency

Hz—hertz

This is the basic frequency unit, expressing the number of cycles of a signal in one second.

The other (following) frequency units used in APx500 measurements are relative units, expressing the relationship of the measured frequency to a reference frequency you have entered into the References Levels > Advanced Settings dialog.

dHz—delta hertz

dHz is the difference in frequency between the measured frequency and the reference frequency; dHz=F-R.

F/R

This is a frequency ratio unit, obtained by dividing the measured frequency F by the reference frequency R.

%Hz—per cent hertz

%Hz is obtained by dividing the measured frequency by the reference frequency and multiplying the result by 100; %Hz=100(F/R). For example, a measured frequency of 950 Hz would be expressed as 95.0 %Hz of a reference frequency of 1 kHz.

Units: Phase

deg—degree

The degree is the basic unit of phase measurement, representing 1/360 of a complete revolution of phase.

rad—radian

One radian (rad) is the angle subtended at the center of a circle by an arc of circumference equal in length to the radius of the circle.

2π rad = 360°

1 rad = 360° / 2π = 57.2958° (approx).

Units: Time

s—seconds

Seconds are the basic time unit, abbreviated s. At values common in audio, the display may be in ms (milliseconds), µs (microseconds) or ns (nanoseconds).

For Acoustic Response measurements, the measured time can be divided by the speed of sound in air, approximately 340 m/s or 1116 ft/s), resulting in distance units:

ft—feet

and

m—meters

UI—unit interval

The *unit interval* (UI) is a measure of time that scales with the interface data rate, and is often a convenient term for interface jitter discussions. The UI is defined as the shortest nominal time interval in the coding scheme.

AES3, IEC60958, S/PDIF interface signals

For an AES3 or S/PDIF signal at a 48 kHz frame rate, there are 32 bits per subframe and 64 bits per frame, giving a nominal 128 pulses per frame in the channel after bi-phase mark encoding is applied. So, if the sample rate is 48 kHz:

1 UI / (128x48000)=163 ns.

The UI is used for several of the jitter specifications in AES31 (the Audio Engineering Society's standard for interfacing two-channel linear digital audio), with the result that the specifications scale appropriately with the data and frame rate. As an example, the dimensions in UI for 96 kHz frame rates are exactly half the size, in seconds, as the dimensions in UI for frame rates of 48 kHz. This scaling matches the scaling of the capabilities and requirements of the receivers and transmitters on the interface.

Sync, Digital Serial (DSIO) and PDM clock signals

These signals are simple square wave pulse trains, and do not have standards-defined frame lengths or bi-phase mark encoding like AES/SPDIF signals. The definition of UI above does not apply to these signals.

For these signals, APx uses the telecommunications industry definition of UI, in which 1 UI is the time to transmit 1 symbol. For Sync, DSIO and PDM clock signals, this is the bit clock period, so

1 UI = 1/fb, where fb is the bit clock rate in hertz.

Although monaural PDM carries one symbol per bit clock period, and stereo PDM carries two symbols per bit clock period, the definition of UI for PDM signals in APx remains the same: 1 UI = 1/fb.

Units: Acoustic

Pa—pascal

The pascal (Pa) is a measure of force, defined as 1 newton per square meter. Atmospheric pressure is measured in pascals.

dBSPL—decibels sound pressure level

Sound pressure level is often expressed in decibels relative to a pressure measured in pascals. 0 dBSPL is defined as 20 µPa, considered the threshold of human hearing.
1 Pa = 94 dBSPL.

Units: Unitless x/y (Impulse Response only)

When the input and output domains of a DUT are the same (analog to analog, for example), the impulse response is a unitless property of the device. We express this in APx500

as "x/y".

See More about Impulse Response on page 244.

More about decibels

The decibel (dB), in one form or another, is arguably the most widely-used unit of measurement in the field of audio. The ear's response to both sound amplitude and frequency is usually best examined in logarithmic terms, and the dB, being a logarithmic unit of measurement, is often the right choice.

It is important to remember that the dB is always a ratio of two values, and for a meaningful expression both of the values must be known. While "8 dB" means nothing, "8 dB below the input signal" gives us a reference and is meaningful. Whenever using the simple dB unit, the expression must identify the reference value.

A large collection of units of measurement (we've seen it referred to as "the dB zoo") uses the decibel ratio with the reference stated as part of the definition of the unit. Examples include dBV (decibels in relation to 1 Vrms); dBu (decibels in relation to 0.7746 Vrms); dBFS (decibels in relation to digital full scale); dBSPL (decibels of sound pressure in relation to 20 micropascals of pressure) and so on.

The formula for decibels in voltage levels is

$dB=20 \log_{10}(V1/V2)$

where V1 and V2 are the two related voltages.

The formula for decibels in power levels is

$dB=10 \log_{10}(P1/P2)$

where P1 and P2 are the two related powers.

More about RMS Level

Rms (root mean square), is the preferred form of ac signal detection which measures amplitude in terms of its equivalent power content, regardless of signal wave shape.

The rms voltage level (Vrms) is the most common level measurement for audio signals, although this level is often stated in a variety of related units such as dBV or dBu. Rms values are typically lower than peak values for the same signal.

More about dBm versus dBu

Although still in use, the dBm unit of measurement is a legacy from earlier days of audio technology when measuring the audio power transferred in impedance-matched circuits was important. The dBm is a unit of power (referenced to 1 mW), and the audio circuits of the day often had an impedance of 600 Ω. The voltage drop across a 600 Ω resistor dissipating 1 mW is 0.7746 V, and this voltage (which reads 0 dBm on a dBm meter across 600 Ω came to be regarded as "zero level" in professional audio circles.

However, most audio circuits do not have a 600 Ω impedance. Microphone circuits may be 150 Ω, and headphone circuits may be 8 Ω. This is even more the case in modern design, where circuit outputs often have very low impedances (from perhaps 10 Ω to around 100 Ω) and input circuits often are "bridging," with impedances of 10,000 Ω or even 100,000 Ω. A milliwatt dissipated in any of these circuits will not read 0.7746 V, and the dBm meter (which is simply a voltmeter calibrated for a 600 Ω circuit) will be in error. The circuit impedance must always be taken into account in power measurements.

Because of this, the dBm can be misleading. It has fallen out of use (except for specific applications where power transfer is an important consideration and the circuit impedances are known) and its place in professional audio has largely been taken by the dBu, where the "u" can be taken to mean "unloaded." The dBu is a valid level measurement in audio circuits of any impedance.

The value of 0 dBu, 0.7746 Vrms, is the same voltage as 0 dBm in a 600 Ω circuit; in 600 Ω, 0 dBu = 0 dBm. However, at other levels the voltage divider formed by the source and termination impedances will cause the values of dBu and dBm to diverge. In other impedances, the values will not be equivalent even at the zero level.

Unless you know the circuit impedance and have a clear reason to use the dBm reference, use dBu.

More about digital level units

All of the units for digital domain audio amplitudes refer to digital full scale. In pulse-code-modulation (PCM) audio, the amplitude of each audio sample is represented as a number. The maximum and minimum values of these numbers vary with the digital word length, but they are precisely defined. A signal which attempts to exceed these values can drive a digital system into overload; in a popular expression, the signal "runs out of numbers."

Word Length	*Positive Full-Scale Value*	*Negative Full-Scale Value*
16 bit	7FFF Hex	8000 Hex

24 bit	7FFFFF Hex	800000 Hex
32 bit	7FFFFFFF Hex	80000000 Hex

In AES17 the Audio Engineering Society defines digital full scale in terms of a sine wave whose positive peak just reaches the maximum positive digital code. The rms value of this sine wave is the full-scale reference, defined as 1 FS or 0 dBFS. Note that other waveforms (noise or music, for example) have different crest factors than a sine wave (crest factor is the ratio of a signal's peak amplitude to its rms amplitude) and will have different rms values at the amplitude at which their peaks just reach the maxima.

A consequence of defining full scale as an rms value is that high-level signals that have a lower crest factor than a sine wave can display rms values greater than 1 FS or 0 dBFS, even though the digital maximum and minimum codes are not exceeded. For example, a square wave whose top is at the maximum positive digital code will have an rms value of 1.414 FS or +3.01 dBFS.

Glossary

A-weighting filter

a specific noise-weighting filter (ANSI S1.4, IEC Recommendation 179) used to produce noise measurements that correlate well with human observations.

ac mains

the utility-provided ac power source from which most electronics equipment is operated. Also power line, ac line and line voltage.

acceptance limit

In pass/fail testing, the poorest measured performance specification which a manufacturer is willing to accept and ship in a product.

ADC

Analog to digital converter, also A/D converter and A-to-D converter. A device for converting an analog input signal into a series of digital values representing the instantaneous amplitude of the signal at regular sampling intervals.

AES

Audio Engineering Society, with headquarters in New York City.

AES3, AES/EBU

See **digital audio signal, bi-phase coded.**

AES3, AES/EBU

a digital interface standard for professional audio equipment interconnection. Also AES/EBU, IEC60958. See **digital audio signal, bi-phase coded**.

alias

a nonlinear signal product falling within the audio band, caused by the presence of an out-of-band (above 1/2 the sampling rate) signal during the original A/D conversion. Alias signals fall at sum and difference frequencies of the original signal frequency (or its harmonics) and the sampling rate (or its harmonics).

analog recording, processing:

Analog techniques in which a signal is represented, recorded, processed, or transmitted as a continuously variable quantity.

ANSI

American National Standards Institute, a U.S.-based standards organization.

anti-alias filter:

a low-pass filter preceding an A/D converter to prevent signals at frequencies greater than one-half the sampling rate from reaching the converter. Signals above one-half the sampling rate cannot be unambiguously converted and would appear in the digital output as signals at incorrect frequencies.

arbitrary waveform

see generator waveform file.

average responding

a form of ac signal detection indicative of the average absolute value of a waveform, typical of analog meter movements.

balanced line

an audio transmission line where the signal is applied differentially between two conductors, each of which has equal impedances to ground or common.

band-pass filter

a filter which passes a specific frequency band essentially without attenuation while attenuating frequencies both below and above the specified band.

band-reject filter

a filter which attenuates the frequencies within a specified frequency range while passing essentially without attenuation the frequencies below and above that range. If the rejection range is narrow, a band reject filter is also often called a notch filter.

bin, FFT

the basic frequency resolution-determining division of an FFT-computed spectrum. The FFT process cannot resolve signals of different frequency falling into the same bin. The bin width of an FFT can be computed from (sample rate)/(number of amplitude samples) in the digitized waveform used as input to the FFT process. Bin and line are used interchangeably.

bit

binary digit, which may have only two possible states (ON/OFF, HIGH/LOW, 1/0, etc.)

bit depth

The number of bits in an audio coding system. In high-performance linear PCM, bit depth is typically 16, 20 or 24. Telecom and compressed coded audio may use fewer bits, with depths of 14, 13, 12 or 8. Also called “word length” or “word width.”

bits of resolution

the number of bits of the binary word by which signals are represented in a digital recording or transmission system. Each bit adds approximately 6 dB to the theoretical dynamic range available. Thus, a 16-bit digital system is capable of approximately 96 dB dynamic range, etc.

bridging

a relatively high impedance input (usually balanced) which may be connected across a lower impedance circuit without significantly affecting the levels in the lower impedance circuit.

byte

an 8-bit digital word.

CCIR

Comité Consultatif International des Radiocommunications (International Radio Consultative Committee), an international standards-setting organization with headquarters in Geneva, Switzerland.

CCIR 468

a CCIR specification which includes, among other things, standards for weighted and unweighted noise measurements. The weighted standard specifies the CCIR weighting filter and a quasi-peak detector (see weighting filter). The unweighted standard specifies a 22 Hz to 22 kHz bandwidth limiting filter and an rms detector.

CCIR-2k

Also CCIR-ARM. A noise measurement technique developed by Dolby Laboratories, which uses the weighting filter shape specified in CCIR Recommendation 468 but with the unity-gain frequency at 2 kHz rather than 1 kHz, and an average-responding detector.

clipping

the action of a system in flattening and squaring off signal peaks when driven with a signal whose peak amplitude is beyond its linear signal-handling capability.

CMRR

Common mode rejection ratio.

CMT

Common mode testing.

CMTST

common mode test; also, the name of the APx output mode for fixed common mode testing.

consumer

In audio, often used to designate digital audio format. See **digital audio signal, bi-phase coded**.

continuous sweep

APx500 term for log chirp measurement. In the continuous sweep measurement view a broadband stimulus is generated, acquired and processed providing a family of measurement results.

crest factor

the ratio of a signal's peak amplitude to its rms amplitude.

crosstalk

unwanted signal coupling from one channel of a multi-channel transmission or recording system to another.

DAC

Digital to analog converter, also D/A converter and D-to-A converter. A device which converts a stream of digital numbers, each representing the amplitude of a signal at a particular sampling time, into a corresponding analog signal.

dB

decibel.

dB/octave

dB per octave. A standard means of referring to the ultimate rejection slope (attenuation versus frequency) of a band-limiting filter. Each pole of a band-limiting filter produces an ultimate rejection slope of 6 dB/octave (20 dB/decade). Thus, a 3-pole filter will have a rejection slope of 18 dB/octave (60 dB/decade).

dBFS

decibels with respect to digital full scale. The full scale amplitude (0 dBFS value) is the rms value of a sine wave whose positive peak just reaches positive full scale.

dBm

dB relative to a reference value of 1.000 milliwatts. dBm is a power unit and requires knowl-

edge of power levels (voltage and current, or voltage and impedance, or current and impedance) rather than merely voltage.

dBr

relative dB; dB relative to an arbitrary reference value. The reference value must be stated for this to be a meaningful unit.

dBu

dB relative to a reference of 0.7746 V.

dBV

dB relative to a reference value of 1.000 V.

decade

the interval between two frequencies with a ratio of exactly 10:1, such as the range from 20 Hz to 200 Hz, or from 1 kHz to 10 kHz.

decibel

decibel, a ratio unit for expressing signal amplitudes. If the amplitudes are expressed in voltage, dB = 20 log10 (V1/V2). If the amplitudes are expressed in power, dB = 10 log10 (P1/P2).

detector

the precision ac to dc conversion section of a measurement instrument, located following all ac signal processing and prior to the indicating portion. Detectors are classified according to which parameter of the input ac signal the output dc value linearly follows-true rms, average, peak, etc.

difference product

an intermodulation distortion signal at the frequency which is the difference between two applied signal frequencies. Sometimes referred to as difference tone. For example, test signals of 13 kHz and 14 kHz will produce a 1 kHz difference product when applied to a device which has asymmetrical nonlinearity.

digital audio signal, bi-phase coded

In the consumer and professional audio field, digital audio is typically carried from point to point as bi-phase coded signal, commonly referred to as AES3, AES/EBU or S/PDIF. There are electrical and bitstream protocol differences among the variations of bi-phase coded digital audio, but the various signals are largely compatible.

a. protocols

The “professional” and “consumer” protocols are indicated in the digital signal’s status bits. With some exceptions, the protocols are compatible; even if status bit information is lost, the audio is usually passed. Although either protocol can be carried on any of the electrical or optical formats, systems using the professional formats typically use the professional protocol, and those using consumer formats use the consumer protocol. In APx500 series instruments, by default the digital audio protocol is set to “consumer.” When you are using the electrical digital output, you can optionally set the Professional checkbox in the Signal Path Setup view. This sets the electrical signal to the professional level, and also sets the protocol and status bits to “professional.”

b. electrical formats

AES3, AES/EBU, IEC60958-4 all refer to an electrically balanced professional format, with a 2–7 volt pp signal in a 110 ohm impedance.

AES-3id and SMPTE276M refer to an electrically unbalanced professional format, with a 1 volt pp signal in a 75 ohm impedance.

S/PDIF and IEC60958-3 refer to an electrically unbalanced consumer format, with a 0.5 volt pp signal in high impedance.

In APx500 series instruments, the digital audio output is unbalanced and is by default set to 0.5 volts pp, a “consumer” setting compatible with S/PDIF and IEC60958-3. If the Professional checkbox is set in the Signal Path Setup view, the level is set to 1 volt, compatible with AES-3id and SMPTE276M. The digital input is unbalanced and by default set to high impedance, compatible with S/PDIF and IEC60958-3. If the 75 Ohm Termination checkbox is set in the Signal Path Setup view, the impedance is set to 75 ohms, compatible with AES-3id and SMPTE276M.

c. optical formats

Digital optical formats have been left to manufacturers to define. Ours is compatible with Toshiba’s Toslink format, as are most digital audio optical inputs and outputs.

digital recording or processing

a technique in which the original signal is periodically sampled and the amplitude value at each sampling instant is converted into a number represented by a binary word.

DIN

Deutsches Institute für Normung, a German standards organization.

dither

dither is low amplitude noise added prior to a signal quantization in order to reduce distortion, improve linearity, and extend the available dynamic range downwards below that of an undithered system of the same number of bits. Level is typically one-half to one LSB in amplitude.

DSP

digital signal processor. A specialized microprocessor designed for highly efficient processing (filter-

ing, FFT, etc.) of digitized analog waveforms. Also, digital signal processing, however implemented.

DUT

a common abbreviation in the test and measurement field for “device under test.”

dynamic range

the difference, usually expressed in dB, between the highest and lowest amplitude portions of a signal, or between the highest amplitude signal which a device can linearly handle and the noise level of the device.

EDID

EDID (Extended Display Identification Data). Used in the HDMI interface, EDID is information stored in downstream display devices and communicated upstream to source devices.

EQ

equalization, or an equalizer.

equalizer

circuitry or equipment which produces a varying (usually adjustable) amplitude as a function of frequency.

EUT

a abbreviation in the test and measurement field for “equipment under test.” Audio Precision uses the term DUT for “device under test.”

even order distortion

Even order distortion products are produced by nonlinearities mathematically described by even value exponents. These nonlinearities produce an asymmetrical shape in the output versus input transfer characteristic of the device. Examples include 2nd harmonic distortion and difference frequency intermodulation distortion.

FFT

Fast Fourier Transform, a technique to compute the amplitude versus frequency and phase versus frequency information from a set of amplitude versus time samples of a signal.

flat

constant gain or attenuation across a frequency band, unfiltered.

frequency domain

a means of representing a signal as a plot of amplitude (normally on the vertical axis) versus frequency (normally on the horizontal axis). Spectrum analyzers represent signals in the frequency domain.

fundamental

the lowest frequency component (normally also the highest amplitude) of a periodic signal.

fundamental rejection

the amount, usually expressed in dB, by which a THD+N analyzer rejects the fundamental component of the input signal. The lowest measurable distortion of a THD+N analyzer is limited by fundamental rejection, along with several other attributes.

gain

gain is the ratio of output signal level to input signal level.

generator waveform file

an audio file on disk that is loaded into the APx generator as the test signal; an arbitrary waveform.

graphic equalizer

an equalizer in which the gain at each portion of the spectrum is controlled by a separate fader. The faders are typically vertically-mounted slide controls, so that the positions of the control knobs forms an approximation to the frequency response of the equalizer.

ground loop

an inadvertent signal path formed when interconnecting the chassis of two or more pieces of equipment, each possessing a safety ground. Ground loops can cause hum-related interference.

group delay

the relative time delay between different spectral portions of a signal.

harmonic

a spectral component at an exact integer multiple of a fundamental frequency.

HDCP

HDCP (High-bandwidth Digital Content Protection) can be used to protect content transmitted on an HDMI interface.

HDMI

High Definition Multimedia Interface, designed to carry high-bandwidth digital streams providing an audio/video interface that includes content protection and a bi-directional channel for interaction with connected electronic devices.

high-pass filter

a filter which passes all frequencies above a specified value essentially without attenuation, while attenuating frequencies below that value.

hum

interference at power mains-related frequencies. Hum directly at the mains frequency and odd harmonics is characteristic of magnetic coupling into the affected system. Hum at the second harmonic of the mains frequency is typically caused by inad-

equate filtering of power supplies with full-wave rectifiers.

IEC

International Electrotechnical Commission, a body responsible for preparing and publishing international standards for the electrical and electronics fields. The IEC is based in Geneva, Switzerland.

IEC60958

See digital audio signal, bi-phase coded.

IMD

intermodulation distortion.

jitter

the undesirable cycle-to-cycle variation in the period of a reference clock, such as are used in digital audio converters. Jitter can cause modulation sidebands and noise if converters operate from a jittered clock. Excessive jitter in an interface can cause digitally-interfaced equipment to malfunction.

jump discontinuity

for audio waveforms, a jump discontinuity is a near-instantaneous change in level, usually created by a waveform edit.

level

the magnitude or amplitude of a signal, which may be expressed in a wide variety of units such as volts, dBm, dBu, watts, etc.

limiting

the action of a limiter in preventing output levels above a specified value. Also, the action of a conventional amplifier when driven beyond its linear range. In this latter context, “limiting” and “clipping” are sometimes used interchangeably except that clipping may imply a more abrupt characteristic than limiting.

limits testing

testing in which measurements are compared to acceptable values so that a pass/fail decision may be made. Also called pass / fail testing.

line level

a relatively high amplitude range suitable for transmission of audio signals. Line level is typically in the 0 dBu to +8 dBu range.

low-pass filter

a filter which passes all frequencies below a specified frequency essentially without attenuation, while attenuating frequencies above that value.

LSB

least significant bit. The bit in a binary word representing the smallest possible value change.

matching impedances

Matching assuring that the impedances of sources and loads connected together in a system are equal, so as to assure maximum power transfer.

notch filter

a band reject filter with a narrow rejection band, often used to eliminate the fundamental frequency for THD+N measurements or to reject a specific spectral component such as power mains hum or a feedback frequency in a public address system.

Nyquist rate (Nyquist frequency)

1/2 the sample rate in a digital system, the frequency above which signals cannot be unambiguously coded.

octave

the interval between a 2:1 range of frequencies, such as 400 Hz to 800 Hz or 5 kHz to 10 kHz.

odd order distortion

distortion products produced by nonlinearities mathematically described with odd order exponents. These nonlinearities cause a symmetrical shape of the output versus input transfer characteristic of a device. An example would be third harmonic distortion, or the 13 kHz product produced by 14 kHz and 15 kHz signals (2*F1 - F2).

pass/fail

testing in which measurements are compared to acceptable values so that a pass/fail decision may be made. Also called limits testing.

passband

the frequency band of a filter in which signals are essentially unattenuated.

PCM

see **Pulse code modulation.**

PDM

see **Pulse density modulation.**

peak

the maximum instantaneous excursion of a signal.

peak level

Peak voltage (Vp) is the maximum instantaneous excursion of a signal. Peak-to-peak voltage (Vpp) is the maximum amplitude difference between positive-going and negative-going peaks of a signal. Peak voltage level is another common level measurement for audio signals. Peak values are typically higher readings than rms values for the same signal.

peak to peak

the maximum amplitude difference between positive-going and negative-going peaks of a signal.

perceptual coding

low-bit-rate coding of a digital signal according to an understanding of human perception of sound, so that the most (perceptually) important data in the signal are coded with the greatest accuracy.

pink noise

noise whose spectral power distribution is such that the power per octave, per decade, or in any other equal-percentage section is the same anywhere across the spectrum. For example, pink noise has the same power in the octave between 50 Hz and 100 Hz as in the octave between 10 kHz and 20 kHz.

power gain

the ratio of output signal power to input signal power of an amplifier.

PPM

(1) in test and measurement, parts per million (usually in lowercase, ppm). (2) in professional audio and, broadcasting, peak program level meter.

preamplifier

amplifier stage or stages used to bring low-level signals, such as those from microphones and phonograph pickup cartridges, up to a higher, standard level suitable for signal routing, mixing, monitoring, etc.

processor

an amplifier designed to modify characteristics of a signal such as frequency response or dynamic range. Processors in common use in broadcasting and professional audio include equalizers, compressors, limiters, reverberation units, modulation processors, noise reduction units, noise gates, etc.

professional

In audio, often used to designate digital audio format. See **digital audio signal, bi-phase coded.**

psophometric filter

a filter whose response is based on the frequency response of the human hearing system. Most noise weighting filters are psophometric filters.

pulse code modulation

PCM. A form of data transmission in which amplitude samples of an analog signal are represented by digital numbers.

pulse density modulation

PDM. A system for representing a sampled signal as a stream of single bits using delta sigma conversion (sometimes called sigma delta conversion). PDM is the technology used in many oversampling ADCs and DACs, and is the basis of the Sony/Philips commercial digital format and disc trade-named DSD and SACD, respectively.

pulse width modulation

PWM. A form of data transmission in which amplitude samples of an analog signal are represented by the duty factor of a pulse train. Often used in switching (Class D) amplifiers.

PWM

see **Pulse width modulation**.

Q factor

a measurement of the selectivity or sharpness of a bandpass or bandreject filter. For a bandpass filter, the Q is equal to the ratio of the center frequency to the bandwidth at the –3dB points. For example, a bandpass filter with a Q of 5, tuned to a center frequency of 1,000 Hz, will have a 3 dB bandwidth of 200 Hz.

random noise

noise whose amplitude-vs.-time distribution is mathematically random and unpredictable, never repeating. The spectrum of a random noise signal is continuous with power at all frequencies, rather than power only at certain points as with pseudo-random noise.

rectangular probability density function dither

RPDF or rectangular dither. Dither which has equal probability of occurrence at any amplitude value between plus one-half LSB and minus one-half LSB deviation from the nominal value. Thus, a graph of probability versus digital value is a rectangle.

residual distortion

the irreducible minimum distortion of an audio generator and analyzer. Residual distortion is thus the “floor” below which the instrument is not useful; measurements above but approaching within 6 to 10 dB of the residual distortion value are less accurate.

residual noise

the irreducible noise in a measurement instrument which sets a floor for amplitude measurements.

resolution

the smallest change in a measured parameter to which a measurement instrument can respond.

ripple

(1) undesired ac variations on a dc power supply output. (2) variations in frequency response in the passband of a filter.

RMS

root mean square, the preferred form of ac signal detection which measures amplitude in terms of its equivalent power content, regardless of signal waveshape.

rms level

Rms (root mean square), is the preferred form of ac signal detection which measures amplitude in terms of its equivalent power content, regardless of signal waveshape.The rms voltage level (Vrms) is the most common level measurement for audio signals, although this level is often stated in a variety of relative units such as dBV or dBu. Rms values are typically lower than peak values for the same signal.

rolloff frequency

the frequency considered the transition between the attenuated and non-attenuated portions of the frequency response of a high-pass or low-pass filter. The filter has a specified attenuation, usually either 3.01 dB or the magnitude of the ripple, at the rolloff frequency.

RPDF

See rectangular probability density function dither.

RSS

root sum square; a method for combining the power of a number of signals by squaring each, summing (adding) these squared values, and finally extracting the square root of the sum.

S/N ratio, Signal-to-noise ratio

the difference in level between a reference output signal (typically at the normal or maximum operating level of the device) and the device output with no signal applied. Signal-to-noise ratio is normally stated in dB. The device input conditions for the noise measurement must be specified, such as "input short-circuited" or with a specific value of resistance connected at the device input instead of a signal.

S/PDIF

See **digital audio signal, bi-phase coded.**

sample rate, sample frequency

the frequency at which the signal is sampled in a digital system. The sample rate must exceed twice the highest analog frequency to be converted. Commonly-used sample rates are 48 kHz, 44.1 kHz, and 32 kHz.

second order product

distortion products produced by a term with the exponent "2" (i.e., a squaring function) in the device transfer function.

separation

the isolation (usually stated in dB) between channels of a stereo or multichannel device. Crosstalk.

shaped dither

dither in which the power-versus-frequency distribution has been modified to place most of the power at frequencies where the human hearing system is not as sensitive.

shielded cable

a construction technique for cables in which a metallic outer conductor completely surrounds one or more signal-carrying conductors. This outer shield is normally connected to ground (earth) at one or both ends.

short circuit

zero-Ohm connection.

sigma-delta

an A/D converter technique characterized by a relatively low-resolution quantizer operated at very high speeds, digitally low-pass filtered to obtain the final digital signal.

single value

APx500 single value measurement views return one value per channel, e.g. "1.1 Vrms."

single-ended

unbalanced.

skirt, filter

the portion of response curve of a bandpass, high-pass, or low-pass filter where the attenuation increases at a rapid rate.

slew rate

the rate of change of a signal amplitude, typically expressed in Volts per microsecond.

slew rate limiting

the condition when an amplifier output is unable to follow the rate of change of the input signal.

SMPTE

Society of Motion Picture and Television Engineers. A professional and standards-setting organization.

SMPTE276M

See **digital audio signal, bi-phase coded**.

S/PDIF

See **digital audio signal, bi-phase coded**.

SPDIF

Sony Philips Digital Interface; a digital interface for consumer audio equipment. Sometimes also referred to as the EIAJ interface. The SPDIF is similar to the professional AES/EBU interface, but is normally an unbalanced coaxial signal of lower amplitude. Most of the status byte definitions are different between SPDIF and AES/EBU.

spectrum analyzer

a measurement instrument which measures and displays a signal as a frequency domain representation of amplitude (usually vertical scale) versus frequency (horizontal scale). Spectrum analyzers

may be designed with a variety of technologies including real-time analyzers (bandpass filter bank), heterodyne or scanning analyzers with a local oscillator, mixer, and selectivity at a fixed intermediate frequency, scanning tunable filters, and FFT analyzers.

status bits

Metadata carried in professional and consumer digital audio bitstreams. See **digital audio signal, bi-phase coded**.

stepped frequency sweep

In APx500 a stepped frequency sweep generates a sine wave at a series of frequencies, and plots the results against the frequency steps.

stepped level sweep

In APx500 a stepped level sweep generates a sine wave at a series of levels, and plots the results against the level steps.

stopband

the frequency band across which a filter has at least some minimum specified value of attenuation.

sum product

an intermodulation distortion product at a frequency equal to the sum of two input frequencies.

sweep

A sweep shows one parameter as a function of a second parameter. The second parameter is moved (swept) across a range of values. Results are displayed as XY graphs or as a table of values. In audio measurement, many sweeps display level or distortion as a function of swept frequency. Amplifier testing often requires level sweeps, displaying output level or distortion as a function of swept input level.

symmetrical

balanced

terminating

connecting the specified load resistance or impedance to a device.

termination

a specific resistance or impedance value which must be connected to the output or input of a device under test for certain parameters to be measured.

THD

THD stands for Total Harmonic Distortion, a measure of all the harmonic distortion products in the DUT's output, with the fundamental stimulus tone removed, and without consideration of noise. THD measurements are usually made with band-limiting or weighting filters. True THD measurements can be valuable but are rare (although easily done in the Audio Precision APx analyzers), since with most techniques it is difficult to measure the distortion products without also measuring the noise. See THD+N.

THD+N

THD+N stands for Total Harmonic Distortion plus Noise, a measure of everything in the DUT's output, with the fundamental stimulus tone removed. Noise and other interfering signal such as hum, buzz and aliased high-frequencies are reported in the result. THD+N measurements are usually made with band-limiting or weighting filters. THD+N measurements are regarded as a key benchmark of a system's performance; the measurement is easy to perform and widely understood, and it shows at a glance not only distortion performance but also indicates a system's immunity (or lack of it) to noise and other interfering signals.

third octave, 1/3 octave

a bandwidth of 1/3 octave, or a frequency ratio of 1.2599:1. Three successive frequency changes by this ratio result in a total frequency change of 2:1 (one octave).

third order

distortion products produced by a cube (exponent of 3) term in a devices's nonlinear transfer function.

time domain

a means of representing a signal as a graph of amplitude (usually on the vertical axis) versus time (on the horizontal axis). An oscilloscope produces a time domain representation of a signal.

Toslink

See **digital audio signal, bi-phase coded**.

TPDF

See triangular probability density function dither.

transducer gain

the ratio of output power level that an amplifier will deliver to a load of specified resistance, to the power level that the amplifier's driving source will deliver to a specified resistance equal to the nominal input resistance of the amplifier.

triangular probability density function dither

TPDF dither or triangular dither. Dither in which a graph of the probability of occurrence of an amplitude rises linearly from zero at values plus-or-minus one least significant bit above or below the nominal value, to unity at the nominal value. The graph thus forms a triangle of unity value at the horizontal center (nominal digital value on the hori-

zontal digital value scale), falling to zero at one LSB to the left or right on the horizontal scale.

trigger

an event which causes another event or action, often initiating a signal generation or acquisition.

unbalanced

an audio connection in which the desired signal is present as a voltage with respect to ground or common, rather than as a differential signal across a pair of balanced conductors.

voltage gain

the ratio of output signal voltage to input signal voltage of an amplifier.

VU meter

a volume unit meter, used to indicate program levels in broadcasting, recording, and similar applications.

weighting filter

a filter with varying attenuation as a function of frequency so as to produce a measurement where the various spectral components affect the measurement in a specified fashion. Most commonly-used weighting filters are attempts to correspond to the varying response of the human hearing system in order to produce measurements (usually of noise) which correlate well with human observations.

white noise

noise whose spectral power distribution is such that there is equal power per Hz, anywhere in the spectrum. For example, white noise will have the same power in the 30 Hz bandwidth between 70 Hz and 100 Hz as in the 30 Hz bandwidth between 10,000 Hz and 10,030 Hz.

window

an amplitude-vs.-time function used to multiply the corresponding samples of a digitized waveform before computing an FFT. Window functions go to zero at the two ends of the record. This provides better selectivity, reducing the signal spreading when non-windowed FFTs are computed on signals which are not exactly synchronous in the signal buffer length being transformed.

word length, word width

See **bit depth.**

XLR connector

a high-quality balanced connector designed for audio applications. Sometimes also called a Cannon connector.

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