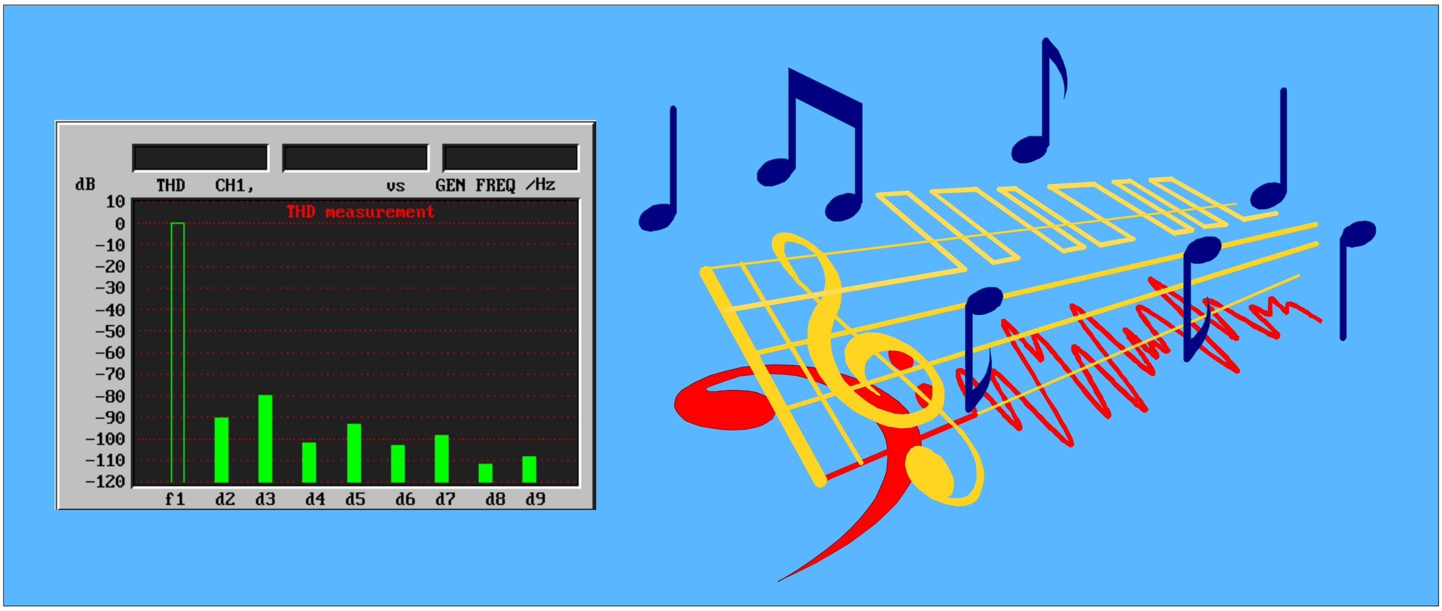


Audio Analyzer UPL

Basics and Operation

UPL Firmware Version 3.01

April 2002

Klaus Schiffner

ROHDE&SCHWARZ

Measurements of Analog Audio Signals

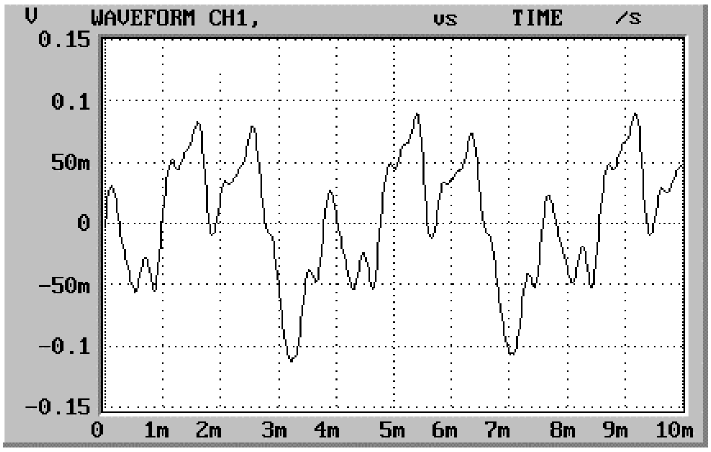


Fig.: 1-1

Analysis of analog parameters:

- frequency and spectrum
- level, signal to noise
- distortion
- intermodulation
- etc.

Generation of the necessary test signals:

- sine waves
- multitone signals
- noise
- etc.

Measurements of Digital Audio Signals

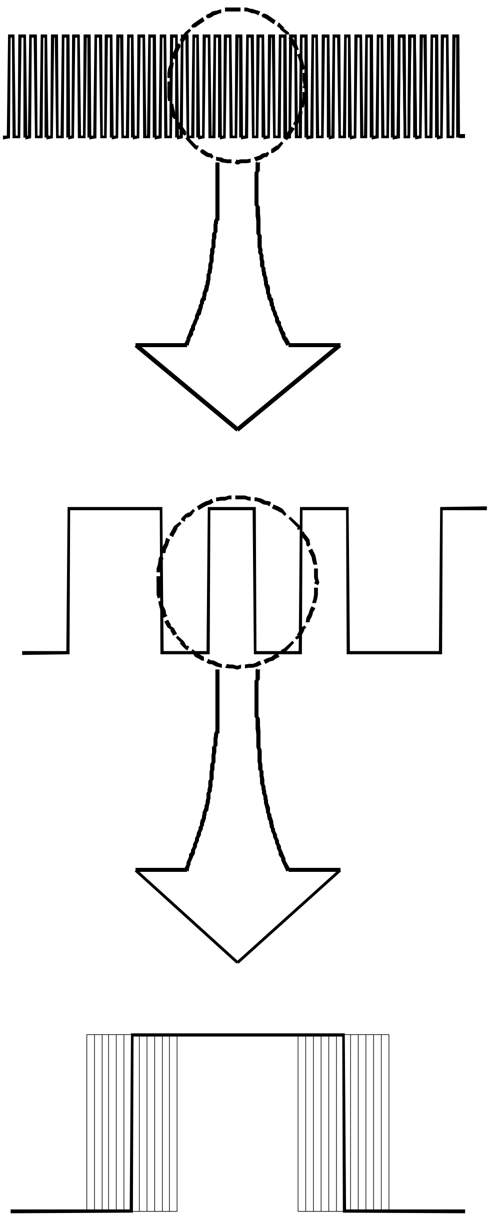


Fig.: 1-2

Analysis of the Audio Data:

- Frequency, Spectrum
- Level, Signal to Noise
- Distortion
- Intermodulation
- etc.

Analysis of the Digital Protocol Data:

- Channel Status Bits
- User Bits
- Validity Bit
- Parity Bit

Interface Tests:

- Puls Amplitude
- Sample Frequency
- Jitter Amplitude
- Jitter Spectrum
- Common Mode Noise
- etc.

Generation of the necessary Test Signals

Instrument Architecture

- **Generators**
  - wide range of test signals thanks to digital signal generation
  - analog generator for extrem pure sine waves
  - variety of sweep functions
  - analog and digital outputs for all signals

- **Analyzers**
  - analog and digital interfaces
  - digital signal processing
  - analog preprocessing
  - variety of data presentation
  - FFT analysis
  - display in time domain (WAVEFORM function)
  - protocol analysis of digital audio data
  - digital interface tests

- **Processor section**
  - standard PC
  - MS-DOS
  - colour TFT display
  - hard disk
  - floppy drive
  - hardcopy to printer or plotter
  - interfaces for keyboard, mouse, ext. monitor
  - test data can be processed further with standard software
  - application programs and measurement functions can be easily loaded

Block Diagram

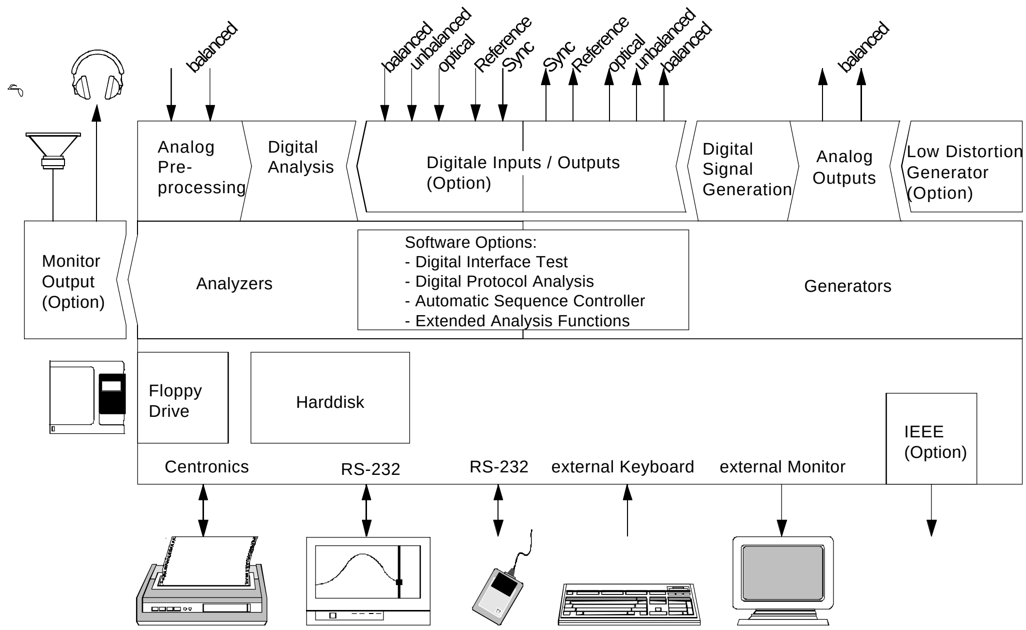


Fig.: 1-3e

Models of the UPL

- **UPL** stock no. 1078.2008.06
  - standard model for universal applications

- **UPL16** stock no. 1078.2008.16
  - special version for type approval measurements on GSM mobiles
  - includes Digital Audio Interface (DAI) for GSM applications
  - UPL-B4, -B6, and -B10 built-in as standard

- **UPL66** stock no. 1078.2008.66
  - no display, no control elements; designed for system applications
  - UPL-B4 built-in as standard

Options of the UPL

- **Low Distortion Generator for the analog outputs**

  UPL-B1 stock no. 1078.4400.02
  - extreme pure sine waves and extended frequency range to 110 kHz

- **Digital Interfaces**

  UPL-B2 stock no. 1078.4000.02
  UPL-B29 stock no. 1078.5107.02
  - AES/EBU, S/P DIF and optical interfaces for 48 / 96 kHz
  - either UPL-B2 or UPL-B29 can be fitted

- **Software for the Analysis of the Digital Audio Protocol**

  UPL-B21 stock no. 1078.3856.02
  - generation and analysis of Channel Status, User Bits, Validity Bit, Parity

- **Software for Jitter and Interface Tests**

  UPL-B22 stock no. 1078.3956.02
  - for measurements of the physical parameters of digital audio interfaces

- **Software to generate Coded Audio Signals**

  UPL-B23 stock no. 1078.5188.02
  - to measure audio decoders

- **IEEE-488 Interface Software**

  UPL-B4 stock no. 1078.3804.02
  - to control the UPL via IEEE bus or RS-232 interface

- **Audio Monitor (Headphone Connector, Speaker built-in)**

  UPL-B5 stock no. 1078.4600.03
  - listen to analog or digital inputs, also after filtering of the input signal

- **Extended Analysis Functions**

  UPL-B6 stock no. 1078.4500.02
  - transfer and coherence function, rub & buzz, third-octave analysis, etc.

- **Hearing Aids Acoustic Chamber and Test Accessories**

  UPL-B7 stock no. 1090.2704.02
  - complete accessories and software for measurements acc. to diff. standards

- **Mobile Phone Test Sets**

  UPL-B8 stock no. 1117.3505.02
  - for acoustical measurements on GSM phones

  UPL-B9 stock no. 1154.7500.02
  - for acoustical measurements on 3. generation mobile phone

- **Universal Sequence Controller**

  UPL-B10 stock no. 1078.3904.02
  - to execute complete measurement sequences or macros
  - comfortable "learn"-mode and program generator

- **LAN Interface**

  UPL-B11 stock no. 1154.7600.02
  - to connect UPL to Novell Networks

- **Automatic Line Measurements acc. to ITU-T O.33**

  UPL-B33 stock no. 1078.4852.02
  - standardized test routine to check the audio lines of broadcasters

- **150** W **Modification**

  UPL-U3 stock no. 1078.4900.02
  - changes the source impedance of the analog generator from 200 Ω to 150 Ω

- **Upgrade for UPL 16**

  UPL-U81 stock no. 1154.7900.02
  – upgrade to validated test cases according to GSM release 1999

- **Adapter Set (XLR/BNC)**

  UPL-Z1 stock no. 1078.3704.02
  - 2 adapters male /female each

- **Audio Switcher**

  UPZ stock no. 1120.8004.02 (Input Version: 8 femal inputs, 2 male outputs)
  UPZ stock no. 1120.8004.03 (Output Version: 8 male outputs, 2 femal inputs)
  - 8 channel audio switcher directly controlled from the UPL
  - see chapter 10 for details

Operation

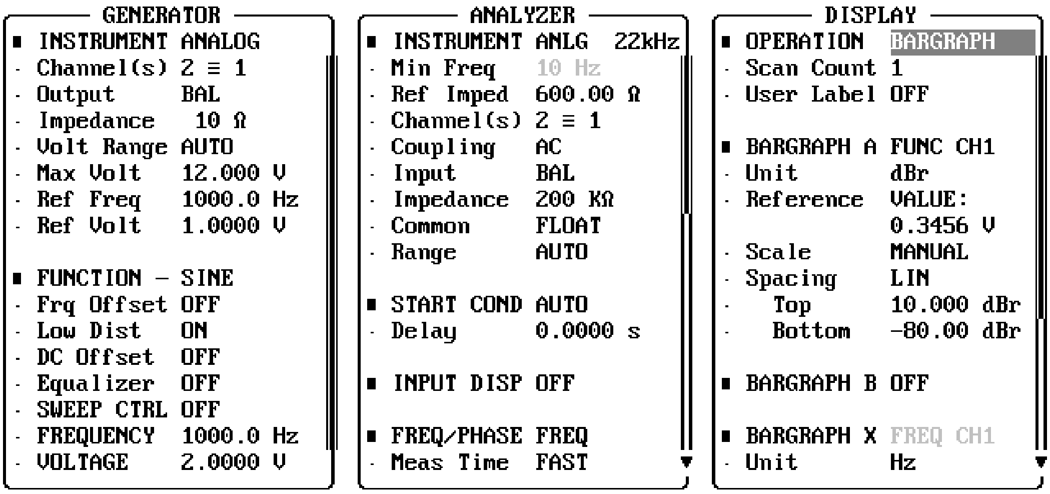


Fig.: 1-4

- **Front panel**
  - hardkeys
  - softkeys
  - rotary knob

- **External keyboard**
  - AT-compatible keyboards may be used
  - all functions available, partly by key combinations using Alt key or Ctrl key
  - country coded keyboards can be selected

- **Mouse**
  - for installation the mouse driver has to be stored under the name mouse.com in the path C:\MOUSE

Panel Structure

related functions and settings are combined to form panels:

- **Generator Panel**
  - selection of analog / digital instrument
  - output configuration
  - level and signal function
  - sweep functions
- **Analyzer Panel**
  - selection of analog / digital instrument and frequency range
  - input configuration
  - measurement functions
  - trigger conditions
  - selection of filters
- **Filter Panel**
  - definition of filter parameters
- **Status Panel**
  - user definable panel, only in combination with graphics
- **File Panel**
  - interface to internal PC
  - to store and to recall setups or series of measured values
- **Display Panel**
  - to define the graphical presentation
  - scaling of x- and y-axis
  - to define tolerance lines
- **Graph Panel**
  - selection of cursor- and marker-functions
  - graphical analysis of measured values
- **Options Panel**
  - settings for printer/plotter, IEEE bus and RS232
external keyboard and external monitor
  - information about built-in options, hard- and software versions
  - selection how the internal calibration routines should be carried out
  - definition of colours and line styles, etc.

General Instructions for Use

1. First select the instrument

- when selecting an instrument also its settings are loaded (see "Parameter-Link-Function" also)
- when changing an instrument, e.g. the choice of functions and thus the possible display modes may be changed

2. Always proceed from "top to bottom" in the panels

- variations in parameters of individual menu items may affect the selection or the range of values of menu items further down, however not the menu items above
- example: the lines to define the sweep parameters will appear after switching on the sweep function
- advantage: panels always are as short as possible, and so they are clearly arranged!

3. Open the window before entering data or selecting parameters

- the keys on the front panel have two functions: first is to select the panels, second function is the input of numbers
- arrow keys are used for scrolling in the panel first, after the window has been opened they are used to select functions

4. Edit the Display Panel only after generator and analyzer have been set

- everything which can be displayed graphically also depends on the selected measurement function
- many setting parameters of the Display panel are automatically adopted from other panels, if desired, speeding up the operation

Selecting Parameters and Entry of Numerical Data

Selecting a Parameter:

- first move to appropriate line using arrow keys or spin wheel
- second open the window (SELECT key or space bar on external keyboard),
- then

  – either: select a function by using the arrow keys or spin wheel
  – or: key in the first letter of the desired function using the external keyboard

- at last press the "Enter" key

Entry of Numerical Data:

- either: open the window and use the keys to enter numeric values,
then close using the "Enter" key
the unit as used before remains valid

- or: open the window and use the keys to enter numeric values,
then close by selecting a unit using the softkeys or the corresponding function keys on the external keyboard

- or: change the numeric values by using the rotary knob; therefore the postion of the number can be selected

The valid range of values for the desired function is always displayed on the screen at the left bottom side.

Use of an external Keyboard

Relation between UPL front-panel keys and key combinations on an external keyboard:

DATA/PANEL keypad:

Front-panel key	Ext. keyboard
GEN	Alt G
ANLR	Alt A
FILTER	Alt T
STATUS	Alt S
FILE	Alt F
DISPLAY	Alt D
GRAPH	Alt R
OPTIONS	Alt O
SHOW I/O	Alt I
?↔?	Alt Z

EDIT keypad:

Front-panel key	Ext. keyboard
SELECT	Space
BACKSP	←
CANCEL	Esc
ENTER	Enter

CURSOR/VARIATION keypad:

Front-panel key	Ext. keyboard
HELP	F1
PgUp	Page Up
PgDn	Page Down
|←, →|	|←, →|
? , ? , ? , ?	←, ↑, →, ↓
ENTER	Enter
to start of panel	Home
to end of panel	End

CONTROL keypad:

Front-panel key	Ext. keyboard
START	Ctrl F5
SINGLE	Ctrl F6
STOP CONT	Ctrl F7
H COPY	Ctrl F8
SYSTEM	Ctrl F9
OUTPUT OFF	Ctrl F11
REM LOCAL	Ctrl F12

Hints and Help Function

The Audio Analyzer UPL has lots of Help functions to support the user:

1. HELP Function

- A Help information can be called for any input field in the panels, selectable in German or English language
- the Help information always corresponds to the firmware version installed

2. SHOW I/O

If it is impossible to display a result, maybe because of inappropriate input signals, the window will display the message "-Input?- Press SHOW I/O". Fulfilling this request by pressing this key sets a graphic to show the currently active inputs/outputs and a text giving information about why the display of measured values is not possible.

3. Error Messages

Red coloured windows will help the user to find out incorrect setting.

4. Entries outside the specified range of values

Entries outside the specified range are not accepted, a warning is audible and the entry is changed to the appropriate minimum or maximum.

5. OUTPUT OFF

All outputs of the UPL can be switched off by pressing this key, e.g. not to endanger the device under test.

Status Display

The status information is always displayed in the top right section of the screen and contains information on the current status of the generator, analyzer and the sweep system.

Generator Status:

GEN OFF:	generator switched off
GEN RUNNING:	generator outputs signal
GEN BUSY:	DSP is processing the waveform
GEN HALTED:	no output signal, the setting is not yet concluded or invalid
GEN OVERRUN:	external sample rate too high for the digital generator
OUTPUT OFF:	all outputs switched off

Analyzer status:

ANL WAIT FOR TRIGGER:	analyzer waits for trigger condition set
ANL 1: ___ 2: ___	separate status information for channels 1 and 2
OFF:	channel switched off
OVER:	overrange
UNDR:	underrange
RANG:	analyzer ranging
SNGL:	single measurement running
CONT:	continuous measurement running
TERM:	single measuremnt finished
STOP:	measurement stopped
CAL:	internal DCoffset calibration running
ORUN:	external sample rate too high for the digital analyzer

Sweep status:

SWP OFF:	no sweep
SWP INVALID:	sweep invalid or not yet started
SWP TERMINATED:	single sweep finished
SWP STOPPED:	sweep was stopped and can be continued
SWP CONT RUNNING:	continuous sweep running
SWP SNGL RUNNING:	single sweep running
SWP MANU RUNNING:	manual sweep running
SWP UNDERRANGE:	valid, but inaccurately measured values occured during a sweep (underrange)

Other status displays:

PRINTER NOT READY:	no printer connected
CONVERTING SETUP:	setup of a previous firmware version is being converted
WAIT FOR CAL ANA OFFSET:	analyzer requires an internal offset calibration, but it is disabled at the moment

Application Setups

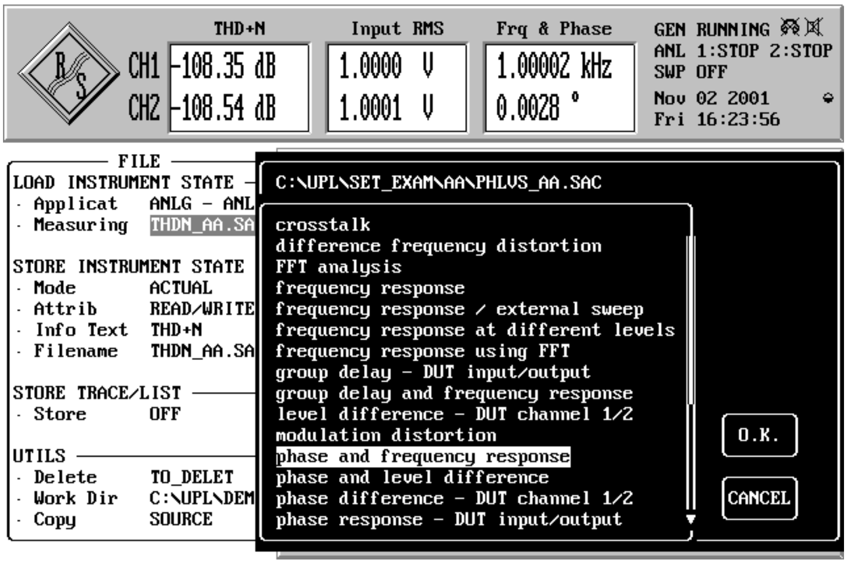


Fig.: 1-5

Loading of Pre-defined Setups using the FILE panel:

Each Audio Analyzer UPL contains a collection of setups, covering the most common applications. Setup files are available for the different domains (analog or digital) of generator and analyzer. They are stored in the UPL directories SETEXAM\AA to SETEXAM\DD, where the first character denotes the generator domain, the second the analyzer domain. Each setup comes with an info text describing the application, the setups are sorted in alphabetical order according to this info text.

All setups use identical presettings, for example the generator output level is set to 1 V, etc. The experienced user is free to adapt the files and/or the info text to specific requirements or to extend the directories by adding own SAC setups. Upon new installation of UPL firmware, all application setups are updated. For this reason modified setups should be stored under another file name.

Applicat:

- USER DEF: Instrument setup according to user-defined measurement or instrument presetting
- ANGL - ANGL: generator- / analyzer domain
  ANGL - DIG
  DIG - ANGL
  DIG - DIG

Measuring: to select the application setups

Storing and Loading of Setups

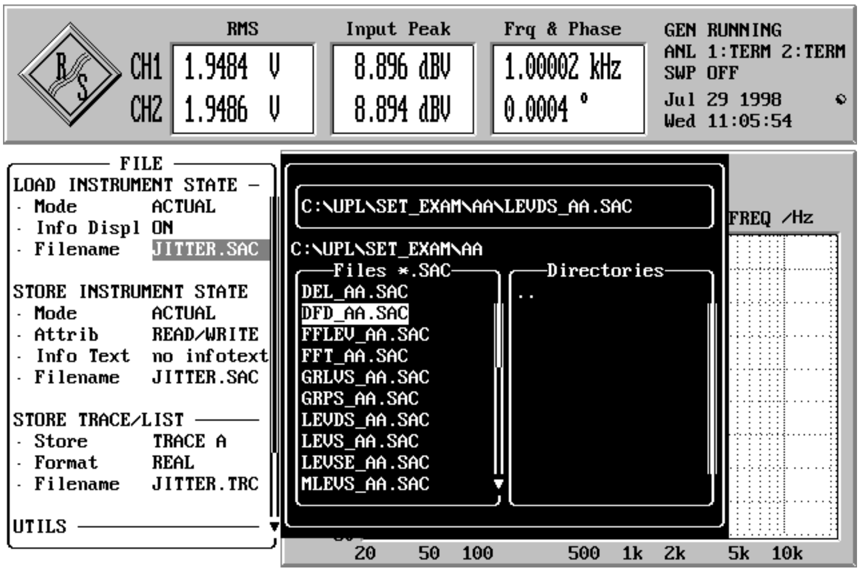


Fig.: 1-5

Storing and Loading of Instrument Setups by using the FILE Panel:

Mode:

- DEFAULT: loads the Rohde & Schwarz default setup
- COMPLETE: complete instrument setup includes all latent, also currently not active functions (load and store)
- ACTUAL: actual setup includes only the currently active instruments and functions (load and store)
- SETUP: actual and complete setups can be loaded
- VIEW PCX: displays files in PCX format
- ACTUAL+DATA: the actual setup together with measured values and traces will be stored
- ASCII: creates a text file to document all active settings of the instrument

Filename: according to MS-DOS regulations

Attrib:

- READ ONLY: the file is write-protected
- READ/WRITE: the file can be deleted or overwritten

Info Text:

- a comment with up to 39 characters can be keyed in to describe the measurement; setting "Info Displ ON" will show this text on the bottom of the screen when loading setups.

Storing Series of Measured Values

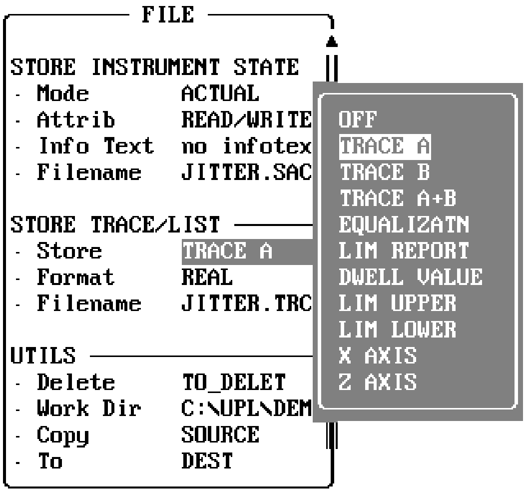


Fig.: 1-6

Loading and Storing of Measured Values and Block/List Data

to be used later

- for the purpose of comparison
- as reference traces
- for generation of lists used in the list sweep
  (possibly after modifying using the text editor EDIT from DOS)
- to be called from other programs and their data further processed

Format:

REAL:	data stored in binary format; rapid access when loaded in UPL
ASCII:	ASCII format, can be further processed using any text editor or other programs; measured values stored in basic units
EXPORT:	contains the trace data without header information to export the data directly to other software programs; measured values stored in display units

FILE Organisation

Working Directory:

- files can be summed up for certain projects or instrument users
- is stored together with the other settings in the setup

File Extensions:

The individual types of files are characterized by reserved file extensions which are listed in the table below:

Extension	Meaning
.AES	Report information AES/EBU + S/P DIF
.AWD	Format for arbitrary waveform files produced by UPD-K2 software
.BAS	BASIC program to be used with UPL-B10
.BAT	Batch file for automatic execution of several programs; reserved (DOS)
.BPZ	Binary file with poles / zeros
.CAL	Calibration file; reserved for calibration factors
.CFG	File with control instructions for working directories
.COE	Coefficient file for filters
.COM	Executable programs; (eg BIOSW.COM); reserved (DOS)
.CPR	compressed arbitrary waveform file
.DWL	Dwell time for automatic generator sweeps: loaded in the GENERATOR panel, menu item "Dwell List"
.ERR	Error file for exceedings of the limits loaded in the DISPLAY panel by selecting Trace A/B →FILE; OPERATION →LIM REPORT;
.EXE	Executable programs; reserved (DOS)
.FTF	Amplitude/Frequency table for generation of noise in the frequency domain
.GL	Screen Copy (HPGL format) is generated in the UPL for subsequent output to a HPGL printer
.HLP	Help File
.LOG	Prolog and epilog for HPGL
.LLW	Limit curve (LOWER) loaded in the DISPLAY panel under the heading LIMIT CHECK, "Lim, Lower" menu item "File name"
.LUP	Limit curve (UPPER) loaded in the DISPLAY panel under the heading LIMIT CHECK, "Lim, Upper" menu item "File name"

Extension	Meaning
.NPZ	Pole/zero files for filters
.NRM	Normalization file reserved for filters
.OUT	DSP files reserved for programs to be down-loaded to the DSPs
.PAC	Report analysis (AES/EBU, S/P DIF), screen control file for channel status data
.PAU	Report analysis (AES/EBU, S/P DIF), screen control file for user data
.PCX	Screen Copy (pixel format) generated in the UPL for later editing and output to a printer
.PGC	Report generation (AES/EBU, S/P DIF), channel status data, file for channel status data
.PGU	Report generation (AES/EBU, S/P DIF), user data, file for user data
.PLT	Color palette information for PCX
.PPC	Report generation (AES/EBU, S/P DIF), channel status data, file for definable report panel
.RLP	Reference lowpass
.SAC	Partial setup; loaded in the file panel under the heading LOAD INSTRUMENT, menu item Mode →ACT SETUP
.SCO	Setup; loaded in the FlLE panel under the heading LOAD INSTRUMENT, menu item Mode →COMPL SETUP
.SPF	Sweep list for frequency of the generator or selective rms measurement; loaded in the GENERATOR panel, menu item "FREQUENCY", "MEAN FREQ" or in the ANALYZER panel, menu item "File name" for X - or Z -axis sweep
.SPI	Sweep list for burst interval, loaded in the GENERATOR panel, menu item "ON-TIME" for X - or Z -axis sweep
.SPO	Sweep list for burst duration, loaded in the GENERATOR panel, menu item "ON-TIME" for X - or Z -axis sweep
.SPV	Sweep list for generator voltage loaded in the GENERATOR panel, menu item "VOLTAGE" or "TOTAL VOLT" (depending on the function) for X - or Z -axis sweep
.TRC	Trace lists for recording of measured values, loaded in the DISPLAY panel by selecting Trace A/B →FILE; OPERATION →CURVE PLOT;
.TTF	Time table for generation of arbitrary signals
.VEQ	Equalizer file, loaded in the GENERATOR panel, menu item " Equal File"
.TXT	Text file to document instrument settings
.WAV	Standard format used for audio signals on PC sound cards
.ZPZ	Pole-zero file reserved for filters

Transfer of Parameters

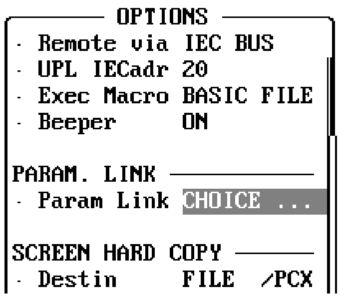


Fig.: 1-7

The Parameter Link Function

- offers the possibility to transfer generator and analyzer settings to another signal or test function or to another instrument.
- for the transfer a corresponding command line must exist in the "new" instrument/function
- if transfer is not possible a warning will be given

Explanation of Menu Items:

Changing Gen Function keeps:

• FUNCTION Parameters	setting of generator function

Changing Gen Instrument keeps

• Output Config.	configuration settings of generator, no transfer A↔D
• FUNCTION including Parameters	signal function and associated settings

Changing Anl Function keeps

• FUNCTION Parameters	settings of measurement functions

Changing Anl Instrument keeps

• Input Config.	configuration settings of analyzer inputs, no transfer A↔D
• START COND	start condition settings
• INPUT DISP	settings of Input Display
• FREQ/PHASE	settings of frequency counter and phase measurement
• FUNCTION including Parameters	functions and associated settings

Function tracking Gen ® Anl

• MDIST, DFD, POL	when the signal function of generator is changed, the appropriate measurement function for the analyzer is set

Hidden Commands

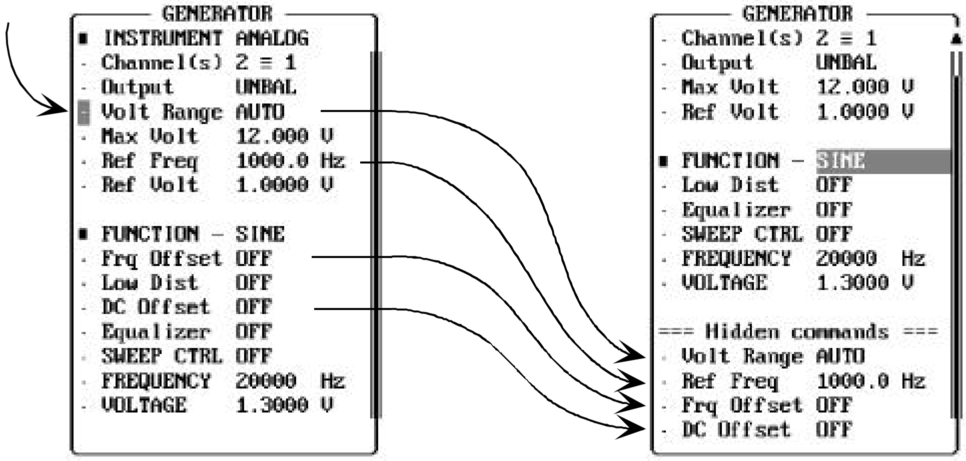


Fig.: 1-9

The function “Hidden Commands”

- the panels for particular UPL applications can be simplified by clearing the visible range from menu lines that are momentarily not required.
- when a menu line is eliminated, a new section „Hidden commands“ appears at the end of the panel, which now contains the eliminated menu lines.
- to „hide“ and to „restore“ a line is done by selecting it in the check box (see fig. 1-9), and then using the BACKSPACE key
- the hidden lines can still be used
- if an external keyboard is connected, the „Hidden commands“ sections can be completely hidden by pressing Ctrl H

Analog Generator

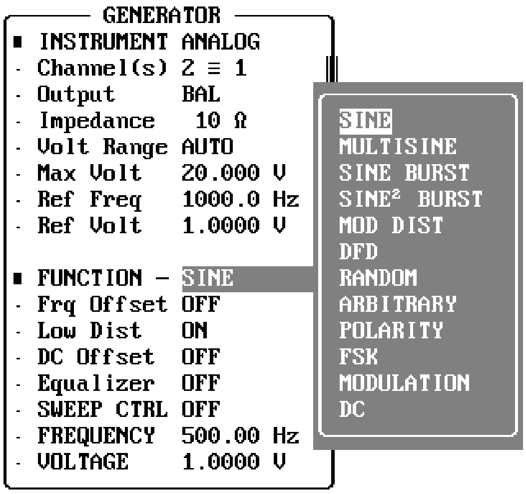


Fig.: 2-1

Universal DSP Generator:

- built-in as a standard
- large variety of signal functions
- frequency range 0.1 Hz to 21.75 kHz

Low Distortion Generator (Option UPL-B1):

- analog circuitry
- sinewaves 10 Hz to 110 kHz
- lower THD
- low distortion generator can be used as an additional source to generate analog audio signals or to superimpose jitter or common mode signals onto the digital audio data stream

Level Reference and Output Power:

- always the no-load voltage is set!
- the outputs are short-circuit-proof
- peak value of the current is limited to about 200 mA
- the maximum settable voltage can be limited not to damage sensitive devices under test

Output Configuration and Impedances:

- the complete generator is designed to be floating to frame potential
- can be configured in unbalanced / balanced mode
- balanced outputs: 10 Ω, 200 Ω (150 Ω as an option), 600 Ω
- unbalanced mode: 5 Ω

Balanced Outputs of the UPL:

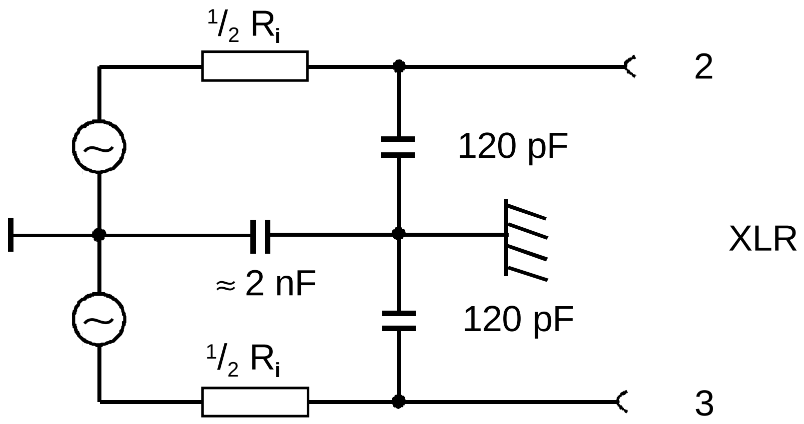


Fig.: 2-2

Unbalanced Mode of the Outputs of UPL:

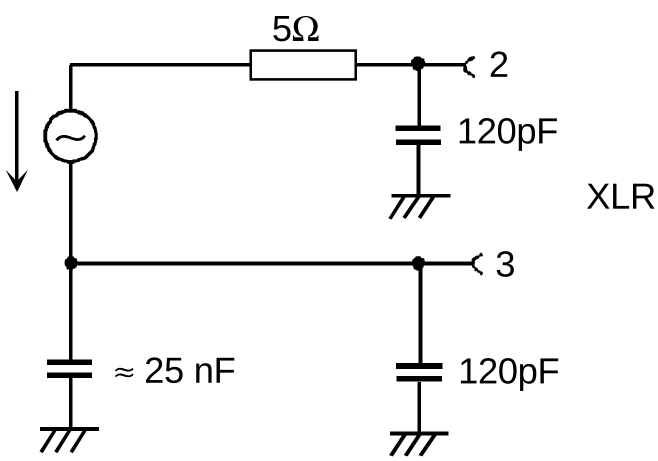


Fig.: 2-3

Digital Interfaces (Option UPL-B2 / UPL-B29)

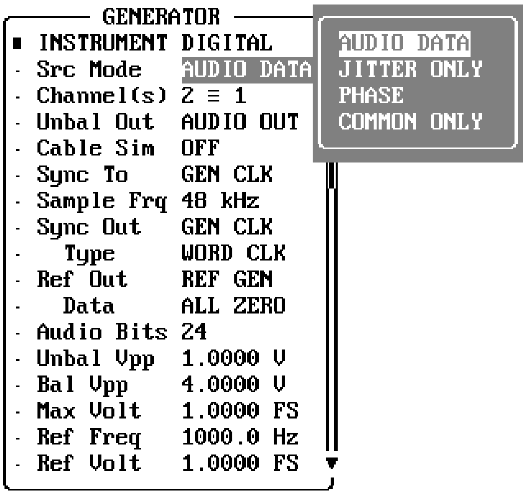


Fig.: 2-4

Source Mode:

- AUDIO DATA — the function generator generates digital audio data; a jitter signal produced by the auxiliary generator (UPL-B1) or a common mode signal can be added
- JITTER ONLY — the digital output provide a jittered signal, no audio data are generated
- PHASE — like AUDIO DATA, the frame phase of the audio data stream to the REF output can be set
- COMMON ONLY — a common mode signal is superimposed onto the audio data stream, no audio data are generated

Frequency Range:

- DIGITAL — 2-channel digital generator
  - up to 55 / 106 kHz sample frequency (UPL-B2 / UPL-B29),
  - 0.1 Hz to 21.9 kHz signal frequency (UPL-B2)
  - 0.1 Hz to 43.8 kHz signal frequency (UPL-B29)

Outputs:

- balanced — XLR connectors
- unbalanced — BNC connectors
- optical — TOSLINK connectors
- signals available on all interfaces simultaneously

Unbalanced Output:

- AUDIO OUT — the generated audio data are present at the BNC connector
- AUDIO IN — the digital audio data received at the UNBAL or BAL input are present at the BNC connector, e.g. to connect an oscilloscop to examine the data stream

Cable Simulation:

- OFF — switched off
- LONG CABLE — cable simulation of an about 100 m long audio cable (lowpass behaviour)

Pulse Amplitude of the Digital Data Stream:

- Unbal Vpp — sets the output voltage of the digital data stream (pulse amplitude, peak-to-peak voltage) at the BNC interface upon termination with nominal impedance of 75 Ω
- Bal Vpp — sets the output voltage of the digital data stream (pulse amplitude, peak-to-peak voltage) at the XLR interface upon termination with nominal impedance of 110 Ω

Generation of Protocol Data (PROTOCOL):

The protocol data, generated together with the digital audio data, can be taken from a file, set in the panel according to AES 3 standard or set in binary format.
CRC as well as time code can be calculated and output too.
See also chapter 6 "Digital Data Transmission and Digital Interfaces".

- ENHANCED — only available with option UPL-B21 installed;
  all channel status data can be set, non-static data (time code, CRC) will be changed dynamically with analyzer function PROTOCOL switched on;
  different data available on both channels
- STATIC — only static data are generated;
  same data on both channels;
  entry from panel only with option UPL-B21 installed
- PANEL OFF — panel switched off; the last-defined state is permanently retained; short panel because of deleted lines

Sample Frequencies and Synchronization

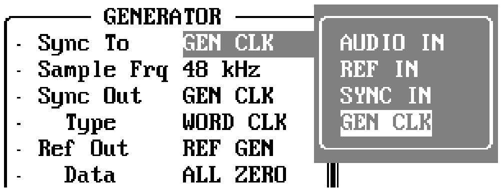


Fig.: 2-5

Setable Sample Frequencies:

- with the fixed frequencies like 44.1 kHz, 48 kHz and 96 kHz the clock is generated internally or synchronized via the SYNC IN input
- using VALUE the frequency of the internal clock can be entered in the range of 27 to 55 kHz with UPL-B2, 35 to 106 kHz with UPL-B29
- when selecting EXTERN the clock is used, which is applied to the SYNC IN input;
  IMPORTANT: The applied frequency has to be entered.
  If the frequency entered does not match the frequency applied, all signals generated are correspondingly shifted in frequency!
- selecting SYNC TO ANL causes the clock to be adopted from the AUDIO IN or REF IN input signal of the analyzer

Synchronization of the Generator:

The digital audio generator can be synchronized to the following signals:

- GEN CLK — synchronization to the internal clock generator
- AUDIO IN — audio signal at the analyzer inputs
- REF IN — DARS signal (Digital Audio Reference Signal), XLR connector at the rear panel
- SYNC IN — synchronization to the following signals, applied to the BNC connector at the rear panel:

- VIDEO 50	video frequency of 50 Hz (Europe)
- VIDEO 60	video frequency of 60 Hz (USA)
- 1024 kHz	reference signal of 1024 kHz
- WORD CLK	synchronisation to the wordclock of the signal applied
- WORD CLK INV	synchronisation to the inverted wordclock of the signal applied

Outputs to synchronize External Instruments:

The digital audio generator produces the following synchronization signals at its SYNC OUT output (BNC connector at the rear panel):

- AUDIO IN — the digital audio input signal (front panel)
- REF IN — the DARS signal from the rear XLR connector
- SYNC PLL — signal from the internal synchronization PLL (e.g. input signal with eliminated jitter)
- GEN CLK — internal generator clock

Either the wordclock signal (sample frequency) or the biphase clock signal (128 times the sample frequency) can be selected.

Outputs for Reference Signals:

The digital audio generator can generate the following synchronization signals at the REF OUT connector (XLR connector at the rear panel):

- AUDIO IN — the buffered input signal
- AUD IN RCLK — audio-input signal which is reclocked by the internal synchronization PLL
- AUDIO OUT — generated audio signal (same as on the front panel)
- REF GEN — generated reference signal, which can be defined constant low (ALL ZERO) or constant high (ALL ONE)

Signal Generation:

Besides of the different audio signals, the digital generator can also generate data streams with superimposed jitter signals; it is also possible, to superimpose common mode signals at the balanced data lines.
For more details see chapter 9 "Digital Signal Processing and Interface Tests".

Audio Bits:

- signals might be quantizised using less bits than limited by the word length e.g. to simulate CD player quality
- output signal is calculated with full resolution and rounded to the specific number of "Audio Bits"

Generator Functions

Available Signals:

• SINE	single sine, dither may be included
• MULTISINE	up to 17 sines simultaneously may be set in level, frequency and (with UPL-B6) phase
• SINE BURST	sine burst signal high-level, low-level, on-time and burst interval length selectable
• SINE² BURST	asymmetrical sine burst signal positive or negative pulses can be generated
• MOD DIST	test signal to measure modulation distortion frequencies and level ratio adjustable
• DFD	test signal to measure difference frequency distortion acc. to IEC 268 resp. IEC 118 frequency difference and mean frequency, resp. the two single frequencies can be set; sweep for mean frequency / upper frequency available
• RANDOM	random noise specify the amplitude density distribution or specify the frequency distribution; for fast frequency response measurements with FFT
• ARBITRARY	waveforms to be specified in time domain up to 16384 points; output of stored audio signals using different file formats
• POLARITY	half-wave signal to check polarity
• FSK	frequency shift keying, required for CCITT O.33
• STEREO SINE	two-channel signals for both digital output channels (UPL-B6)
• MODULATION	AM or FM modulated sinewave signals
• DC	output of DC voltages
• CODED AUDIO	generation of coded audio signals (using UPL-B23)

Generator Sweep Function

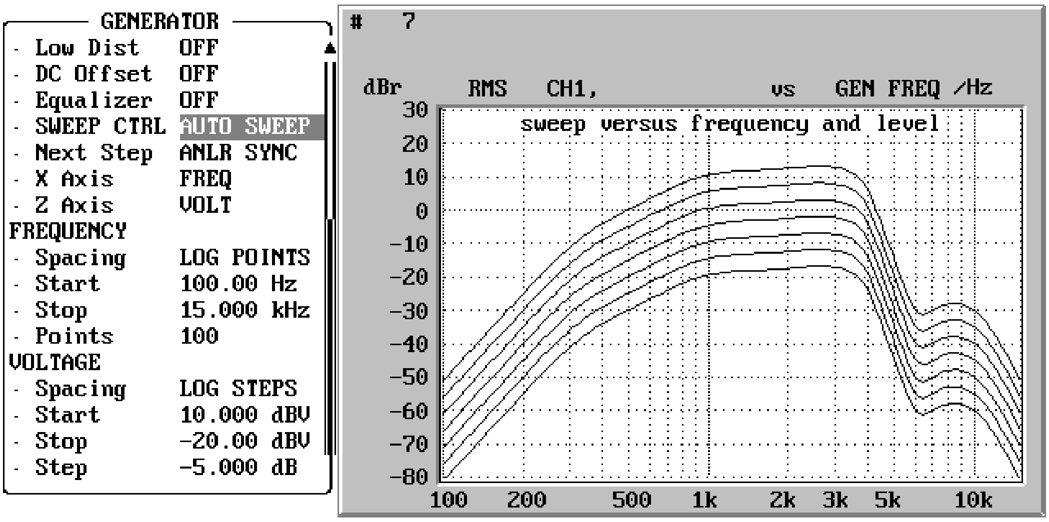


Fig.: 2-6

Sweep Parameters:

- 1-dimensional sweep: X-parameters are changed
- 2-dimensional sweep: X- and Z-parameters are changed
- each list of sweep parameters provides 1024 points as a maximum

Different Sweep Modes:

- AUTO SWEEP: (= "standard" sweep); data of the sweep parameters are obtained from user specifications; sweep runs automatically
- AUTO LIST: data of the sweep parameters are read from a file; sweep runs automatically
- MANU SWEEP: data of the sweep parameters are obtained from user specifications; sweep is controlled by means of the rotary knob and/or the cursor keys
- MANU LIST: data of the sweep parameters are read from a file; sweep is controlled by means of the rotary knob and/or the cursor keys

Analog Analyzers

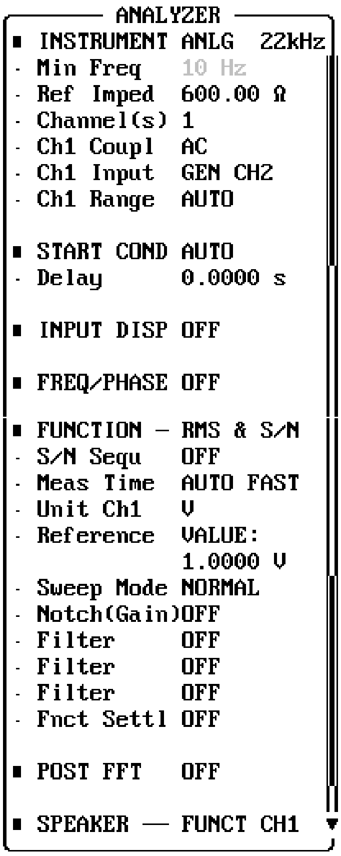


Fig.: 3-1

Organization of Analyzer Panel:

Configurations:

- selection of instruments and frequency range
- reference impedance for power units
- setting the test inputs
- start conditions

Higher-level functions:

- input display
- frequency measurement, phase measurement, group delay

Measurement functions:

- measurement function
- post-FFT
- audio monitoring

Input Circuitries

Balanced inputs of the UPL:

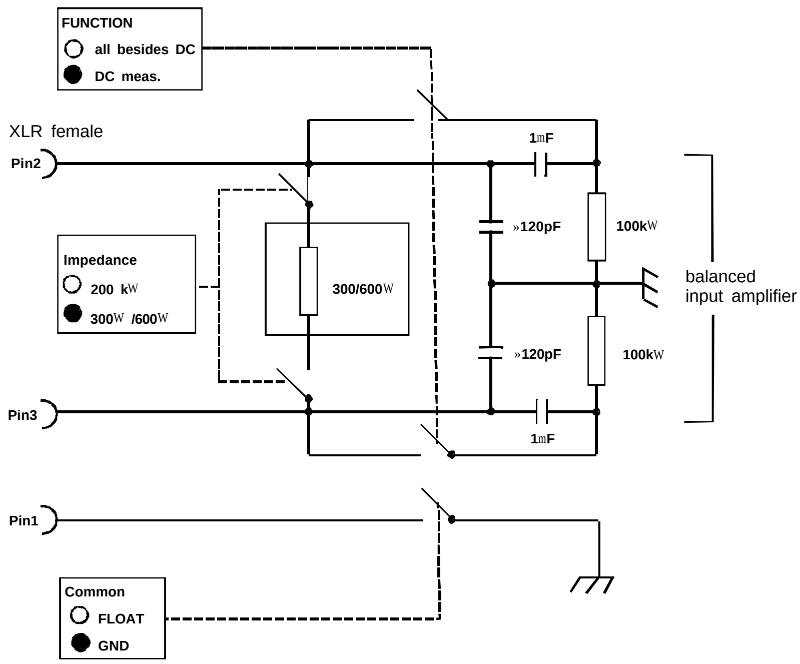


Fig.: 3-2e

Unbalanced mode of the UPL input using the adapter:

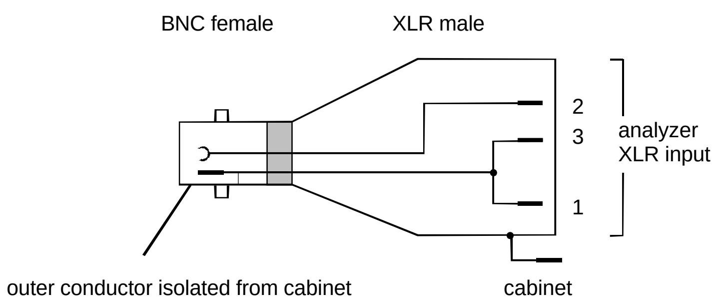


Fig.: 3-3e

Configuration of the Analog Analyzers

Frequency ranges:

• ANLG 22kHz	DC/10 Hz to 21.9 kHz
• ANLG 110kHz	DC/20 Hz to 110 kHz

Measurement Channels:

• 1	channel 1 is active
• 2	channel 2 is active
• 1 & 2	both channels are active and can be configured individually
• 2 ≡ 1	both channels are active and equally configured, 2 follows 1
• 1 ≡ 2	as above, 1 follows 2

Coupling:

• AC	AC coupling; a DC offset of the DUT will not be transmitted
• DC	DC coupling; test signals down to 0 Hz are picked up and considered in the results of RMS, RMS selective, Peak, Quasi Peak, DC, FFT and Waveform measurements

Inputs:

• BAL	balanced to ground measurement inputs
• GEN CH1 CH2 GEN CROSSED	internal connection of one or both analyzer channel(s) to GEN the other generator channel; input connectors of the analyzer inactive

Impedances:

300 Ω / 600 Ω / 200 kΩ

Reference of Potential:

- UPL uses amplifiers with differential inputs, referred to frame ground;
  Pin 1 of the XLR connector can be switched (important when using the adapter):
  - FLOAT: pin 1 (and so also the outer conductor of the adapter) floating against frame ground
  - GROUND: pin 1 connected to frame ground (PE conductor)

Selection of Range:

• AUTO	automatic selection of the range
• FIX	range is retained in any case
• LOWER	no switch to lower ranges, but overloads cause higher ranges

Digital Analyzer (Option UPL-B2 / UPL-B29)

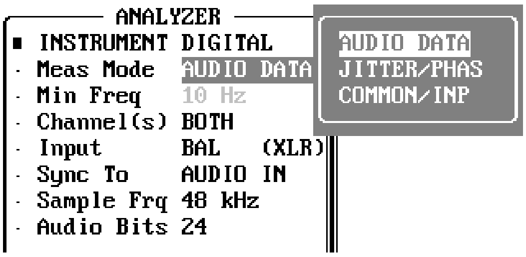


Fig.: 3-4

Meas Mode:

- AUDIO DATA — measurement of digital audio data
- JITTER/PHAS — measurement of jitter at the selected digital input and of the phase offset to the reference signal
- COMMON/INP — measurement of common mode signals and digital pulse amplitude

Frequency Range:

- DIG 48kHz — dual-channel digital analyzer
  - up to 55 / 106 kHz sample frequency (UPL-B2 / UPL-B29),
  - up to 21.9 / 43.8 kHz signal frequency (UPL-B2 / UPL-B29)

Inputs:

- BAL XLR — balanced digital input
- UNBAL BNC — unbalanced digital input
- OPTICAL — optical digital interface (TOSLINK)
- INTERN — internal connection from generator

Analyzer Sample Rates and Synchronization

Fig.: 3-5

Synchronization of the Analyzer:

measuring audio data, the digital analyzer can be synchronized to the following input signals:

- AUDIO IN — to the digital audio input signal
- REF IN — to the reference input signal (XLR connector on the rear panel)

for the measuring mode JITTER/PHASE, jitter reference can be selected:

- GEN CLK — synchronization to the internal clock generator
- VARI (PLL) 44.1 (PLL) 48.0 (PLL) 96.0 (PLL) — to the input signal whose jitter has been eliminated via the internal synchronization PLL; VARI gives the maximum range of 35 to 55 kHz sampling rate (UPL-B2), 35 to 106 kHz (UPL-B29)

Settable Sample Frequencies:

- with the setting **VALUE** or with the fixed sample rates, the analyzer physically works using the applied clock rate, independent from the setting. The setting of the sample rate is necessary for interpretation, because for the internal filters and for the frequency calculations the set sample rate is taken into account;
- setting **AUTO** takes the measured sample rate for the calculation of the results;
- setting **CHAN STATUS** takes the sample rate information from the corresponding channel status bits;
- the frequency applied to the UPL must not exceed 55 kHz (106 kHz using UPL-B29), since, otherwise, incorrect measurements or abortion of the measurement may occur.

Audio Bits:

- signals can be analysed using less bits than given by word length e.g. to simulate the quality of CD input circuitries
- LSBs not used for the analysis are cut off

Ways of Starting the Analyzer (external Sweep)

• AUTO	continuous measurement without trigger conditions
• TIME TICK	single measurements are carried out at regular time intervals
• TIME CHART	measured valued are entered in a time chart in the time pattern that can be entered; e.g. useful to display intermediate results
• FREQ CH1/2	**External Frequency Sweep:** tracing of measured values because a change in frequency was noted at analyzer input; frequency measured by FFT analysis, also works with noisy signals
• FREQ FST CH1/2	**External Frequency Sweep:** as above, but faster frequency measurement; works only with clean signals
• VOLT CH1/2	**External Voltage Sweep:** tracing of measured values because a change in voltage was noted at the analyzer input
• LEV TRG CH1/2	triggering because of a detected voltage at input 1 or 2
• EDG TRIG CH1/2	triggering because of a detected voltage edge at input 1 or 2

Higher-level Functions of Analyzer

INPUT DISP:

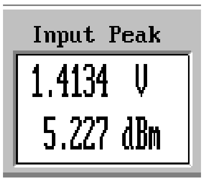


Fig.: 3-6

- Off — input display switched off
- PEAK — display of input peak value
- RMS — display of input RMS value
  available with THD measurements
- DIG INP AMP — display of digital pulse amplitude (in measurement mode COMMON/INP only)

Frequency, Phase and Group Delay Measurements:

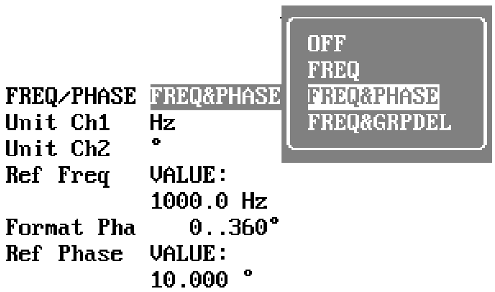


Fig.: 3-7

With UPL, frequency measurements are calculated from the zero crossings when measuring RMS; with all the other measurement functions they are calculated by FFT analysis.

Combined frequency and phase or group delay measurements available in dual-channel mode only.

Settling Function

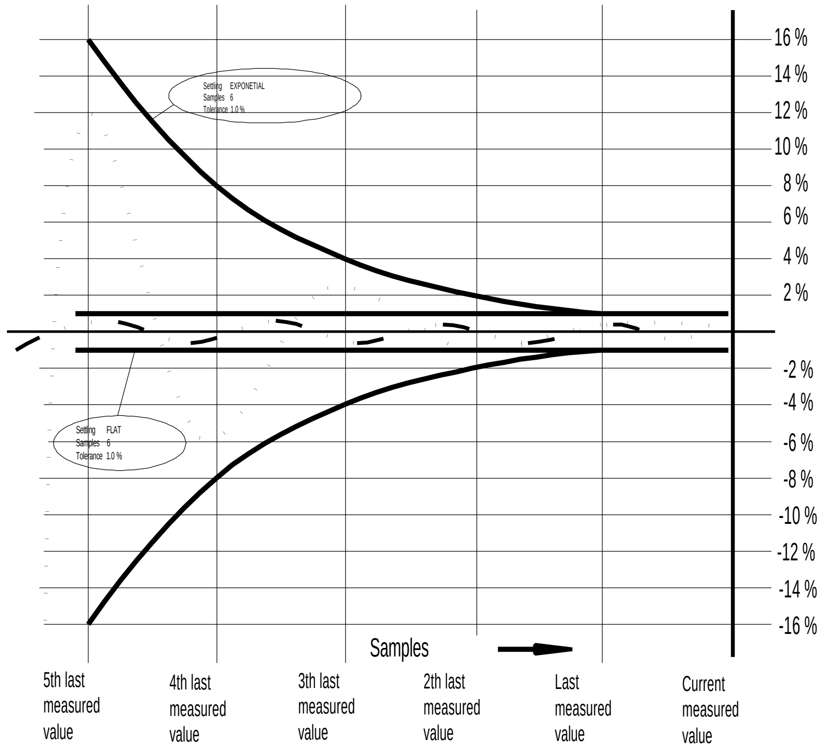


Fig.: 3-8

Why is Settling required?

Generator setting → settling of DUT → display of valid result

- settling known → delay function
- settling unknown → settling function
- external generator → settling function

Settling means:

- measured value only displayed if it is within a certain tolerance
- can also be combined with delay function

Settling is used for:

- external sweep
- frequency measurement
- phase measurement
- function measurements
- can be combined with settling for external sweep mode

Tolerance + Resolution

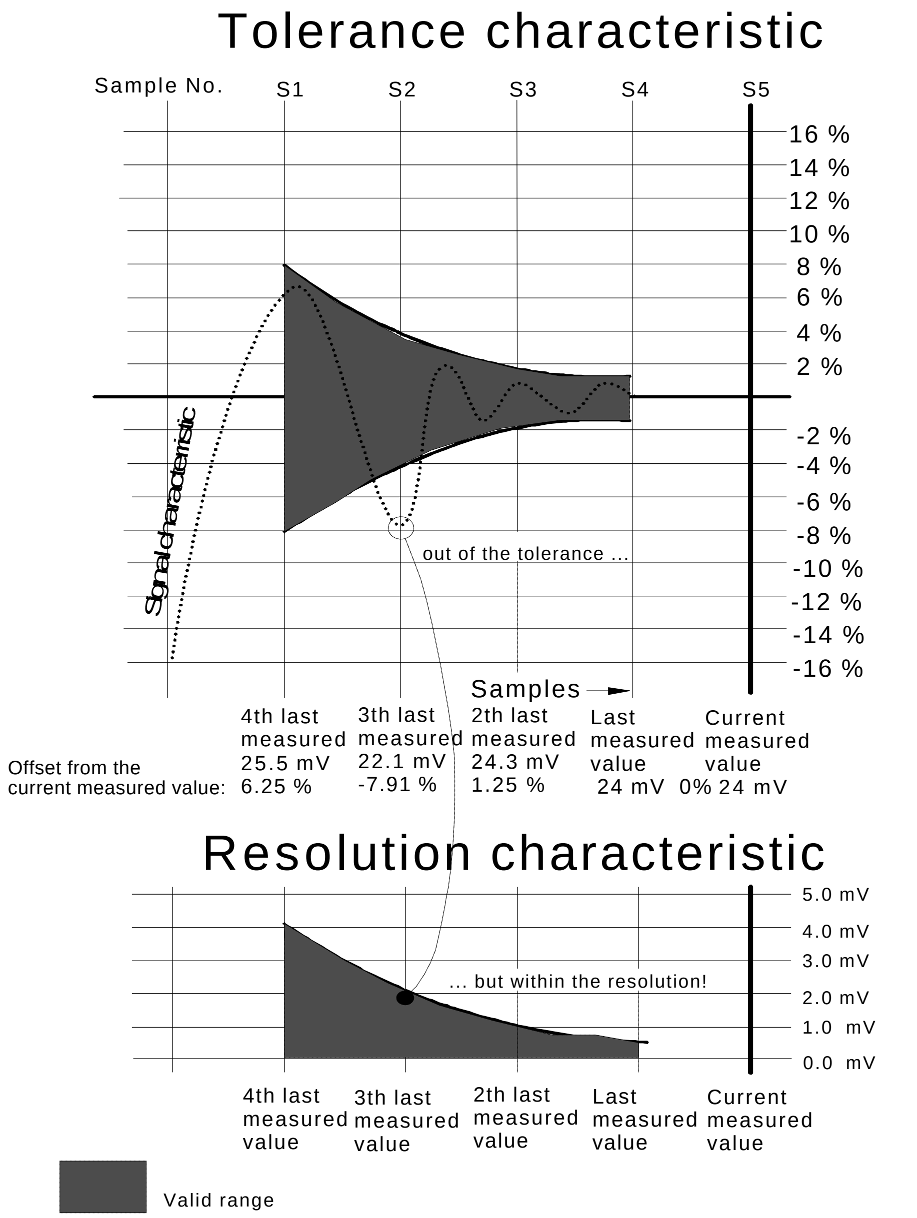


Fig.: 3-9

How does Settling work?

- value is measured
- compared with n measured values stored before
- checking the tolerance limits
- measured value will be displayed

Settling Definitions

Settling EXPONENTIAL

- exponential characteristic (tolerance funnel)
- calculated to the power of 2
- for test items with normal exponential transient response

Settling FLAT

- tolerance band
- stricter settling condition

Settling AVERAGE

- arithmetic averaging
- after restart output only when the number of samples are collected
- after this working like a "shift register"

Tolerance

- maximal permissible deviation from the steady-state final value
- entry in % or dB

Resolution

- for measurements at the lower measurement limit
- in case of superimposed noise
- values away from the exponential tolerance characteristic will be checked a second time by the user-defined resolution

Timeout

- abort of the measurement loop

External Sweep with Settling Function

Example:

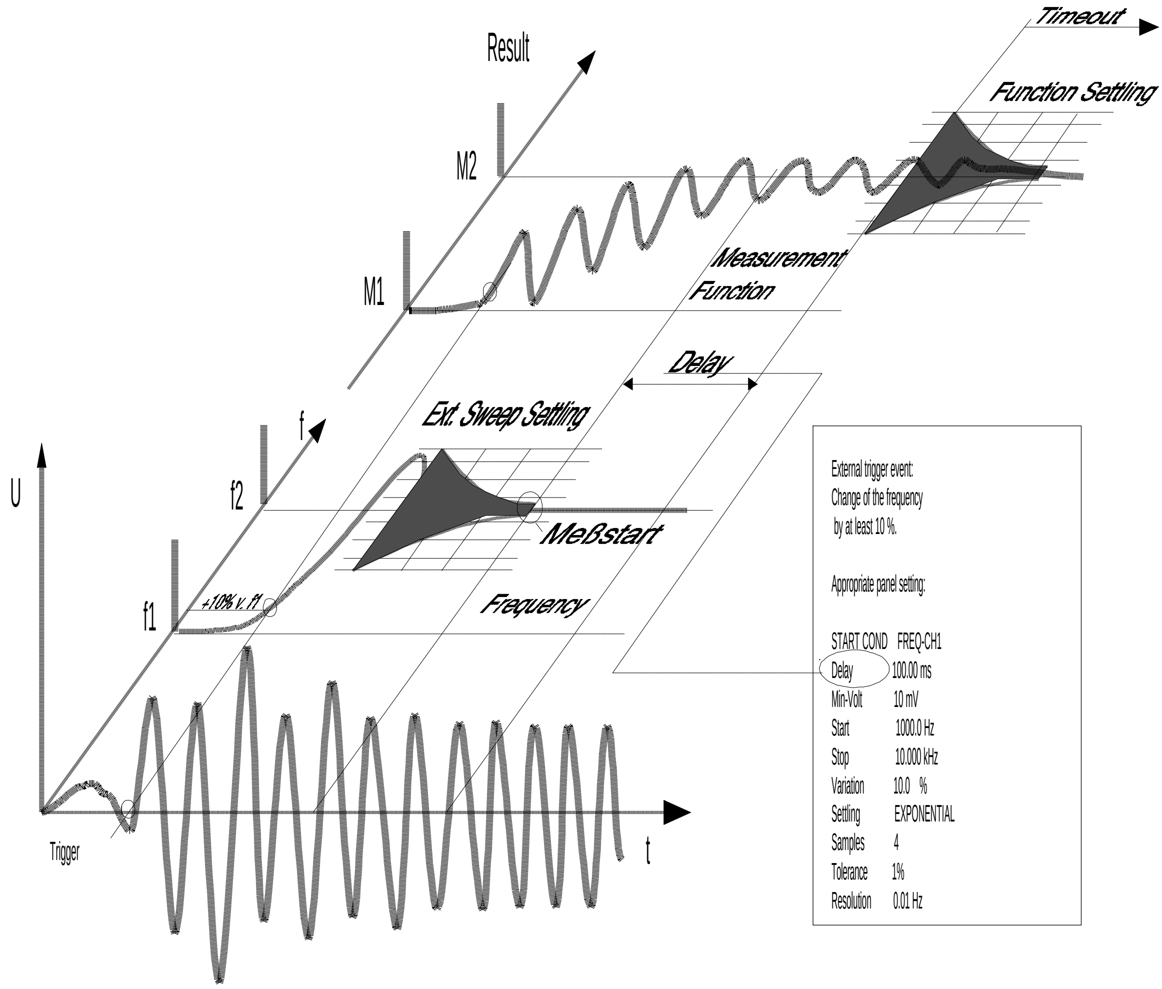


Fig.: 3-10

1. check whether the input level reaches the "Min Volt"
2. wait for frequency stabilization
3. frequency within the range defined by "Start" and "Stop"?
4. wait the delay time
5. perform function measurement
6. check function settling if required
7. display the results of function measurement
8. start a new frequency measurement if frequency has varied by at least the "Variation"

Common Parameters for all Measurement Functions

DC Suppress:

with the digital measurement functions, the DC component of the signal to be measured is suppressed

S/N Sequence:

- generator signal alternately switched on and off
- measurement of signal and noise value
- output impedance remains constant when generator is switched off

Reference:

• VALUE	selectable from 100 pV to 1000 V
• STORE CH1/2	measurement result of channel 1 or 2 is stored on pressing the Enter key
• GEN TRACK	present generator voltage is used as reference value
• MEAS CH1/2	each level measurement result of channel 1 or 2 is used as reference for both channels
• DIG OUT AMP	the measured value of the digital pulse amplitude is referred to the value set in generator panel

Reference Frequency / Reference Phase:

- similar to reference function

FILTER:

- depending from measurement function, up to 3 filters can be switched on.
  See also chapter 4, "Filter and Equalizer"

POST FFT:

- Post FFT is a FFT analysis subsequent to the functions RMS, THD+N and WOW&FL.
  See also chapter 5, "FFT Analysis"

RMS Measurement

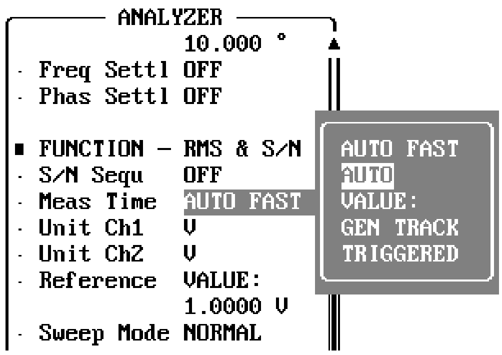


Fig.: 3-11

Selection of Measuring Time:

• AUTO FAST and AUTO	automatic matching to the signal period, differences in the number of samples taken into account and therefore the measurement inaccuracy (AUTO : 0.1 %, AUTO FAST: 1 %); advantage: measuring time always adapted optimally e.g. when using frequency sweep function
• VALUE	numeric entry of measuring time; range: 100 µs to 10 s, depending on type of analyzer
• TRIGGERED	mode, where a single measurement is started as soon as the input level reaches the trigger level; for example to measure short pulses
• GENTRACK	measuring time for one period of the generator signal, at high frequencies for a few periods; measuring time is calculated from the generator signal

Remarks on the fixed integration times:

- Meas Time is integer multiple of signal period:
  → optimum integration effect, steady display
- Meas Time larger than, yet no integer multiple of signal period:
  → beats occur in display
- Meas Time smaller than signal period:
  → no integration effect, AC measurement follows the signal waveform

Selective RMS Measurement

Selective RMS Measurement with narrow Bandpass or Bandstop Filter

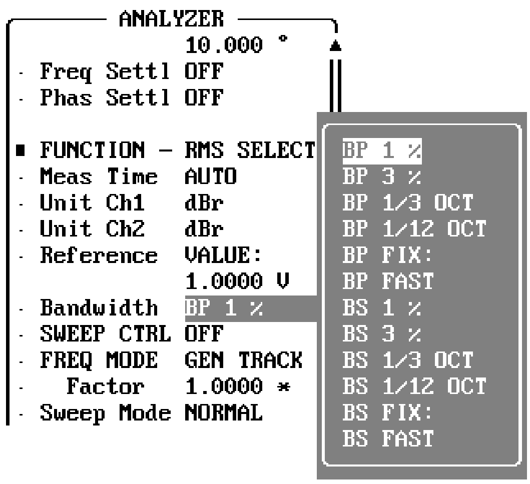


Fig.: 3-12

Selectable Bandwidth:

• BP 1% / BS 1%	geometrically symmetrical bandwidth 1 %
• BP 3% / BS 3%	geometrically symmetrical bandwidth 3 %
• BP 1/3 OCT BS 1/3 OCT	geometrically symmetrical bandwidth of a third octave (≈ 26 %)
• BP 1/12 OCT BS 1/12 OCT	geometrically symmetrical bandwidth of one pitch (≈ 6 %)
• BP FAST BS FAST	same as BP/BS 1/3 OCT, for 40 dB attenuation only, but with considerably shorter settling time
• BP FIX BS FIX	arithmetically symmetrical bandwidth using numerical entry

Frequency Sweep of Selective RMS Measurement

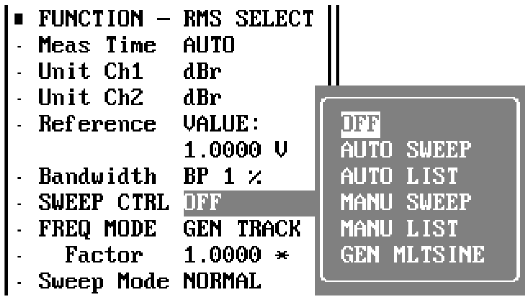


Fig.: 3-13

Sweep Control:

- AUTO SWEEP — sweep runs automatically, sweep parameter data are calculated
- MANU SWEEP — sweep is controlled by the rotary knob or cursor keys, sweep parameter data are calculated
- AUTO LIST — sweep runs automatically, sweep parameter data are read from a file
- MANU LIST — sweep is controlled by the rotary knob or cursor keys, sweep parameter data are read from a file
- GEN MLTSINE — the bandpass center frequency is successively set to the multisine frequencies in the generator panel, it operates in the same way as the AUTO LIST function

Remarks on MANU LIST and MANU SWEEP:

- movement of rotary knob in both directions possible
- fast rotation allows sweep points to be left out which is indicated by gaps in the graphical presentation
- reverse rotation allows to the interrupted curves to be completed

Additional Parameters to be selected:

- similar to generator sweep function

Peak and Quasi-Peak Weighting

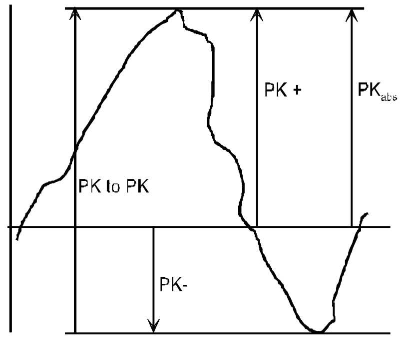


Fig.: 3-14

Peak Measurement:

- peak detector follows the waveform without delay

Quasi-Peak Measurement:

- peak detection with defined rising and falling times according to ITU-T J.16 (formerly CCIR 468-4)

Definition of Peak Values:

• PK +	positive peak value
• PK -	negative peak value
• PK to PK	peak-to-peak value
• PK abs	the higher of Pk+ and Pk- value

Interval Time:

In the Peak and Quasi-Peak measurement, the maximum peak value of the input signal is determined and displayed within the monitoring interval selected under "Intv Time". Then the memory is cleared and the measurement starts again.

• FIX 50 MS	50 ms	(only with PEAK)
• FIX 200 MS	200 ms	(only with PEAK)
• FIX 1000 MS	1000 ms	(only with PEAK)
• FIX 3 S	3000 ms	(only with Quasi PEAK)
• VALUE	numerical entry 20 ms to 100 s	

THD-Measurement

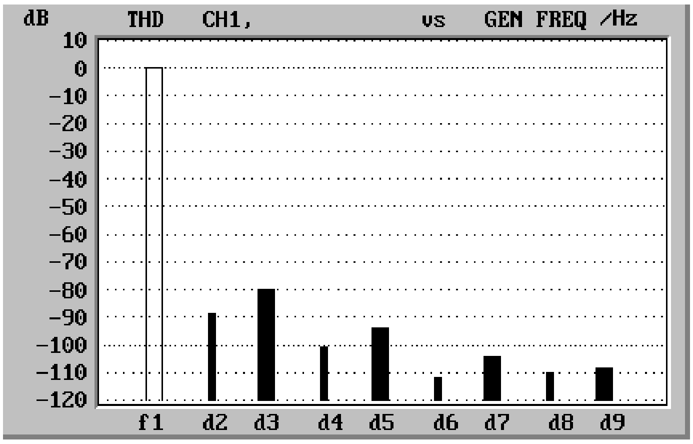


Fig.: 3-15

Principle of Measurement:

- THD = RMS value of harmonics in relation to the total RMS value
- amplitudes of the harmonics are measured selectively (DFT)
- d_2 to d_9
- if necessary only single harmonics are measured
- care about measurement bandwidth!

Measurement Mode:

- SELECT di — any combination of harmonics d_2 to d_9
- All even di — all even harmonics d_2, d_4, d_6, d_8
- All odd di — all odd harmonics d_3, d_5, d_7, d_9
- All di — d_2 to d_9
- LEV SEL di, LEV even di, LEV odd di, LEV All di
  as above, but display as level values

Dynamic Mode:

- FAST — fast measurement, for THD worse than -90 dB
- PRECISION — analog notch filter switched on, highest dynamic

Fundamental:

- AUTO — highest amplitude in spectrum is automatically determined as fundamental
- VALUE — numerical entry of fundamental frequency
- GEN TRACK — generator frequency ("Frequency") is used as fundamental frequency, preferably used for signals with high harmonic content

THD+N-Measurement

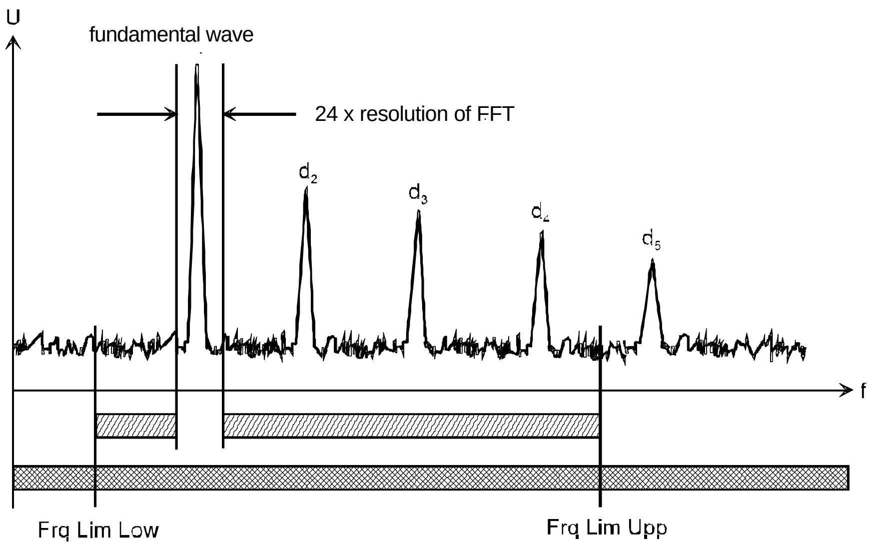


Fig.: 3-16e

Principle of Measurement:

- THD+N = RMS value of harmonics plus broadband noise in relation to the total RMS value
- fundamental wave is filtered out (24 x resolution of FFT)
- total remaining energy is measured within the specified band limits
- care about measurement bandwidth!

Measurement Mode:

- THD+N — negative dB-values
- SINAD — positive dB-values
- NOISE — as THD+N, however the harmonic lines are not weighted
- LEVEL THDN — level measurement of harmonics plus noise
- LEVEL NOISE — level measurement of noise

Dyn Mode

- FAST — fast measurement with lower dynamic
- PRECISION — measurement performed with highest dynamic and with the analog notch filter cut in

Rejection: (with Dyn Mode FAST only)

- NARROW — fundamental blanked with extremely narrow filtering
- WIDE — fundamental rejected with notch filter characteristic

Modulation Distortion

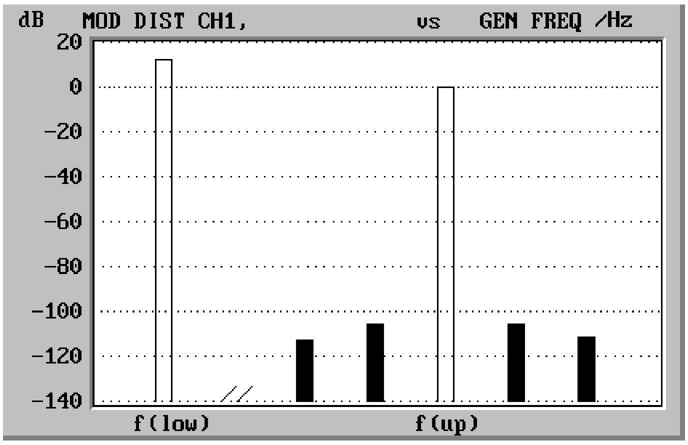


Fig.: 3-17

Principle of Measurement:

- test signal consists of useful sinewave signal (e.g. 8 kHz) and interfering sinewave signal (e.g. 60 Hz), level 4 times as high
- selective measurement of intermodulation products of 2nd and 3rd order
- in relation to RMS value of useful signal (f_{up})
- total modulation distortion = square sum of dm_2 plus dm_3
- unaffected by noise due to selective measurement!

Dynamic Mode:

- FAST — fast measurement, for IMD worse than -80 dB
- PRECISION — analog notch filter switched on, highest dynamic

Difference Frequency Distortion

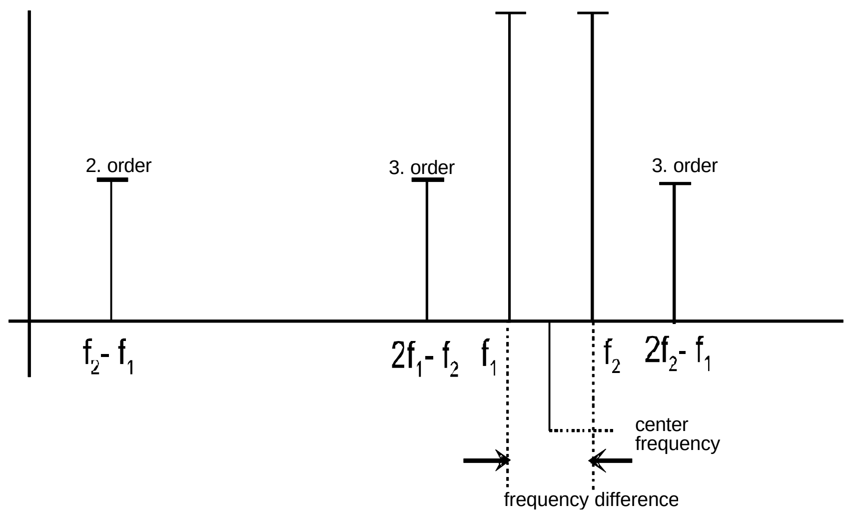


Fig.: 3-18e

Principle of Measurement:

- test signal consists of 2 sinewave signals (e.g. 10 and 11 kHz) at the same level
- DIN IEC 268 part 3 defines DFD of 2nd and 3rd order
- selective Measurement of different frequency distortion d_2 or d_3
- DFD measurement refers to RMS value of useful signals
- unaffected by noise due to selective measurement!

Measurement Mode:

- IEC 268 — standard for measurements on amplifiers: the intermodulation products are referred to the RMS value of both input components
- IEC 118 — standard for measurements on hearing aids: the intermodulation products are referred to the RMS value of the single, higher frequency component, i.e. both modes for DFD of 2nd order differ in the results by 6 dB

Applications of THD and Intermodulation Distortion Measurements

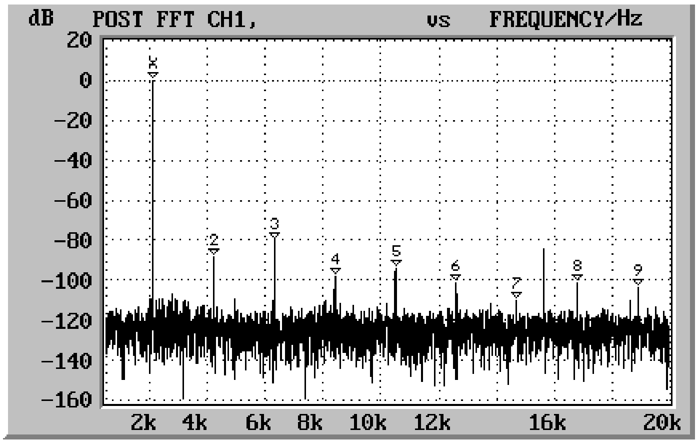


Fig.: 3-19

THD+N Measurement:

- useful for test signals in the lower frequency range (< 5 kHz), otherwise the distortion products are out of the audio bandwidth
- FFT analysis helps to find out if the measurement is effected by nonharmonic components or by noise
- effects typically occurring with digital signal processing (e.g. mixing products from sample frequency) also influence THD+N measurement

THD Measurement:

- analog applications where single harmonics are predominant, e.g. measurement of 3rd harmonic distortion on tape machines
- different reasons in order if even or odd harmonic components are predominant: Examples: overdriven amplifiers show high odd components (spectrum of squarewave!), unbalanced push-pull amplifier stages generate high even components

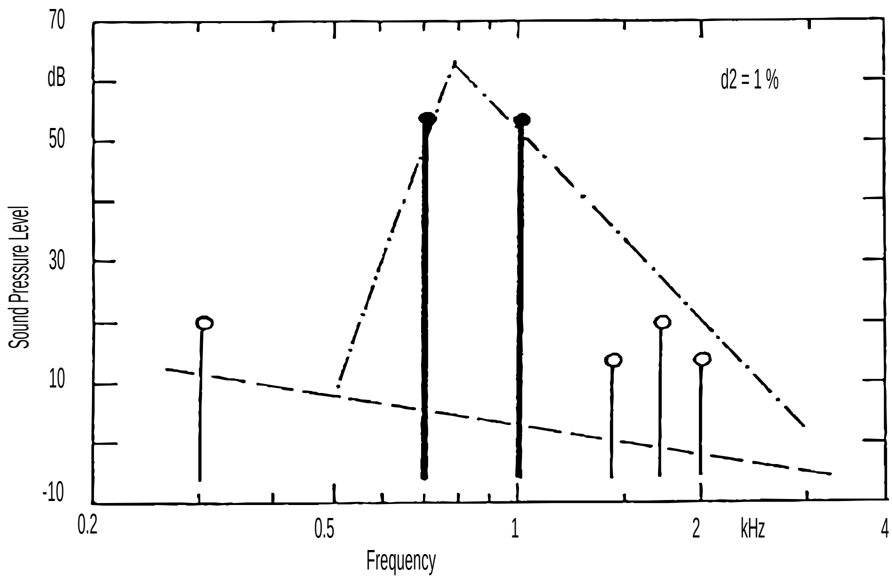


Fig.: 3-20

Modulation Distortion:

- shows the behaviour of test items if more than one test signal is applied
- in analog measuring technique quite difficult, because interfering products are very close to the useful sinewave

Difference Frequency Distortion:

- used especially with test signals in the higher frequency range, because d_2-component always occurs in the lower frequency band
- easy to be measured by using a fixed bandpass filter at for example 1 kHz, independent if the test signals are swept
- different frequency components are more disturbing and more easily audible than harmonic distortion products, because they are nonharmonic and more often above the masking threshold of the human ear (see figure 3-20, the difference frequency is audible, the sum frequency and the second harmonic distortion products are masked by the human ear)

Waveform

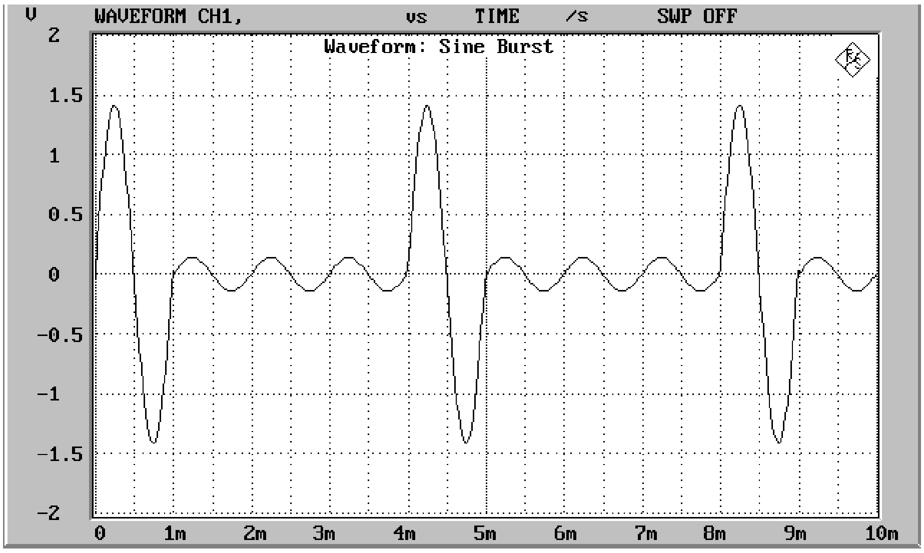


Fig.: 3-21

- displays the input signal in time domain
- trigger level and trigger slope can be set
- also the start of a SINE BURST signal from the generator can be used to trigger; application: measurement of signal delay times through the DUT
- interpolation steps can be selected for the display of the traced waveform; in case of few samples per period of the input signal a smoothed display can be obtained.
- in **COMPRESSED** mode, the input signal is first fed to a peak-value detector. Following this, the number of samples set under Comp Factor are combined. This peak value is then used as the input signal of the waveform function. Thus the x-axis is so to speak compressed, which allows long times to be acquired. If compression factor is set in the way, that at least one period fits into the time window used for one measurement value, the **positive envelope curve** of the test signal is displayed.
  Application: measurements on automatic gain control circuitries
- in the **UNDERSAMPLE** mode x-axis is compressed similar to the compressed mode, but the samples are arithmetically averaged, therefore the **time template** of the test signal is displayed.

Polarity Test

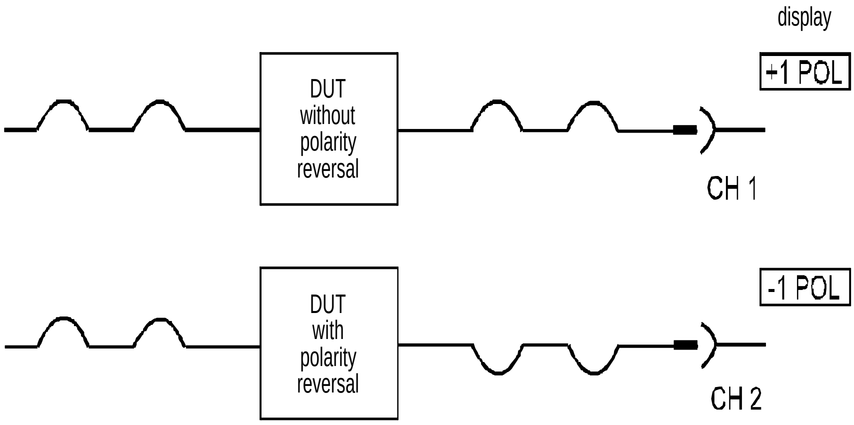


Fig.: 3-22e

- checks whether a DUT passes on an applied signal with the same or with reversed polarity

Wow & Flutter-Measurement

Principle of Measurement:

- a sine tone with fixed frequency is used for the test
- variations in the speed of mechanically moved part (e.g. pitch rolls) cause frequency deviation of the test signal during its reproduction
- these deviations are determined and specified in percent of the carrier frequency
- to simulate the characteristic of the human ear a weighting filter (4 Hz-bandpass) is used

Measurement Standards:

• DIN/IEC	3.15 kHz, Quasi-Peak detector
• NAB	3.00 kHz, average rectifier
• JIS	3.00 kHz, RMS weighting
• 2 Sigma 5 s 2 Sigma 10 s	2-Sigma weighting according to IEC 386/1988 with integration time selectable 5 or 10 s

DC Measurement

- DC voltages can be measured at the balanced inputs.
- reference point is the outer conductor of the BNC adapter and connecting point 3 of the XLR connector.

FFT-Analysis, Transfer- and Coherence Function

- are described in chapter 5, "FFT-Analysis"
- option UPL-B6 necessary for Transfer- and Coherence Function

3^{rd} Octave Analysis

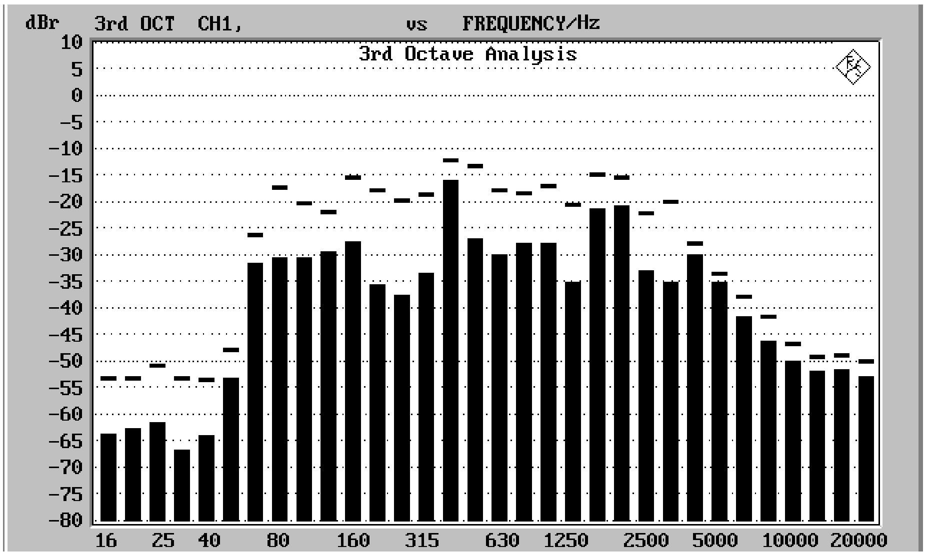


Fig.: 3-23

- option UPL-B6 necessary for 3^{rd} Octave Analysis
- levels in up to 32 third-octave bands are measured simultaneously
- real-time measurements of frequency bands using narrow-band bandpass filters
- analysis performed according to standard IEC 1260 of 1995 with level accuracy of class 0 (± 1,0 dB)
- setting lower and upper frequency limit, the third-octave bands can be selected
- max hold function with different decay characteristic available

12^{th} Octave Analysis

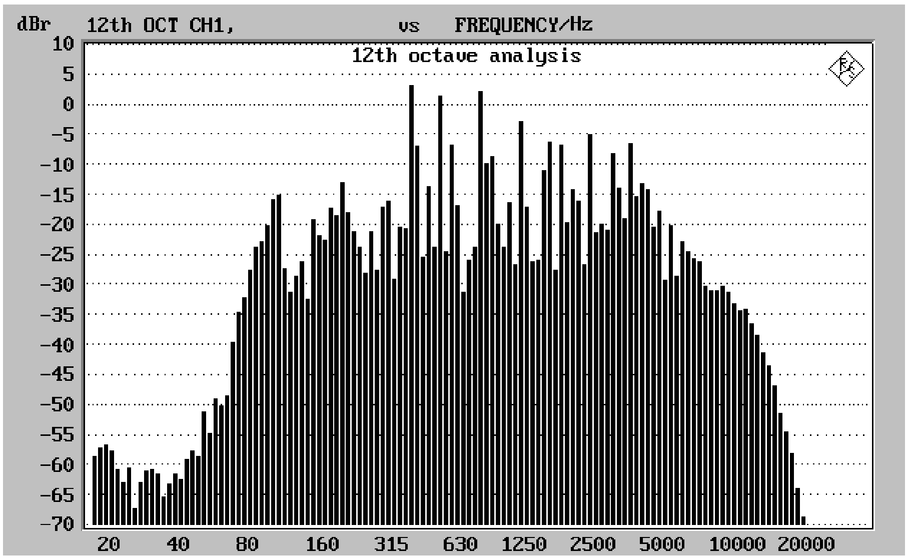


Abb.: 3-24

- option UPL-B6 necessary for 12^{th} Octave analysis
- levels in up to 125 frequency bands can be measured simultaneously
- measured by means of a special zoom FFT, measurement time is a multiple of FFT time
- main application: acoustical measurements on mobile phones

Rub & Buzz

- option UPL-B6 necessary for Rub & Buzz measurement
- special measurement used to detect manufacturing faults in loudspeakers; for example, defective membranes produce more distortion products, quite often in limited frequency bands
- the measurement is based on a THD+N measurement where only components of higher order are measured, a swept sine wave is used as the stimulus
- in addition, simultaneous frequency response measurement is carried out, measurement of polarity can be switched on
- the whole measurement is optimized in speed for production applications

Option UPL-B5: Audio Monitor

built-in loudspeaker and headphones output

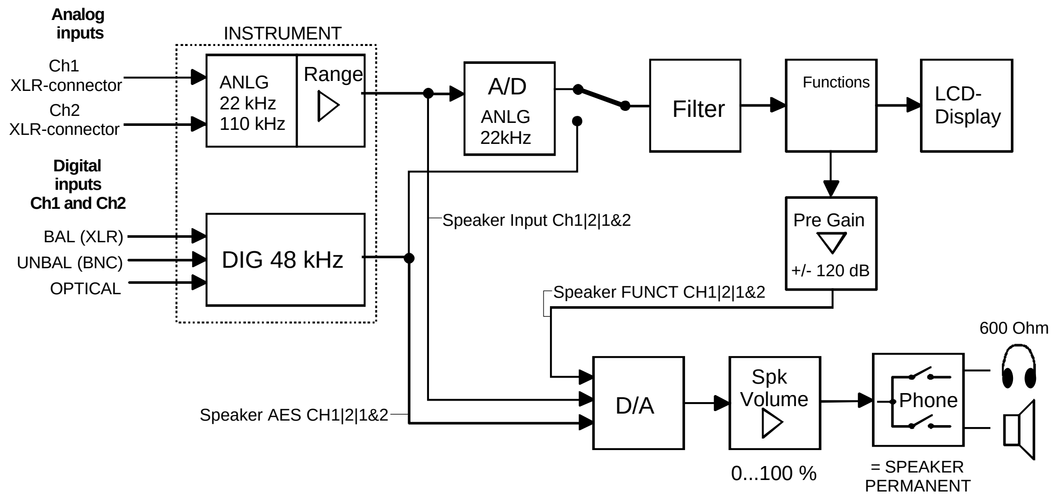


Fig.: 3-25

Monitoring of

Measurement Functions	Monitoring possible	Output Signal =
RMS, RMS Sel, PEAK, Quasi PEAK	yes	filtered or unfiltered input signal
THD+N, RUB&BUZZ	yes	residual signal or unfiltered input signal
DC, THD, MOD DIST, DFD, WOW&FL, FILTER SIM, COHER, FFT, POLARITY, WAVEFORM, 3rd Octave, 12th Octave	no	input signal

Technical Data of the Headphones Output:

recommended impedance of headphones:	600 Ω
connector:	6,3 mm phone jack
max. output voltage:	8 V
max. output current:	50 mA

Notes:

- Headphones output also can be used to connect an oscilloscope, but the excellent data usually supplied by the measuring path of the UPL are not provided due to noise sidebands of the internal PLL.
- In a special mode, UPL-B5 can also be used to generate DC voltages, e.g. to work as a power supply for hearing aids

Settings:

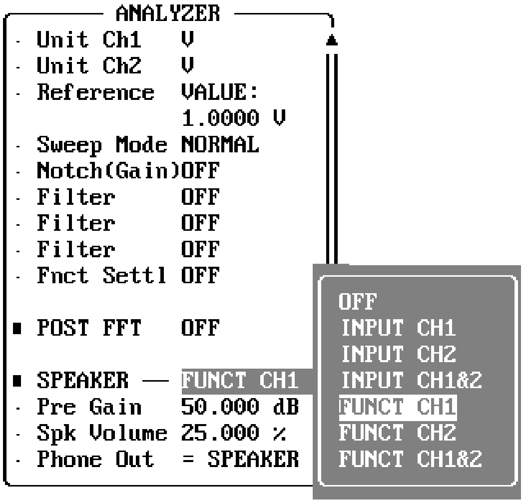


Fig.: 3-26

SPEAKER

- OFF: monitor output switched off
- INPUT: monitoring of the analog analyzer input signal on channel 1, 2 or on both (stereo operation)
- AES: monitoring of the left, the right or of both channels of the interfaces of the UPL-B2 option
- FNCT: monitoring of the analyzer output with the level functions, Rub & Buzz and THD+N measurement

Pre Gain:

digital gain or attenuation of the function output
for example necessary to make filtered signal
components audible

Spk Volume:

volume for the monitor output
also adjustable by the key "+/-" and using the rotary knob

Phone Out

- = SPEAKER: speaker-off key acts on loudspeaker and phones output; when connecting headphones, loudspeaker is switched off
- PERMANENT: phones output permanently switched on; speaker-off key acts on loudspeaker only.

Filter and Equalizer

The functionality of filtering is available with Audio Analyzer UPL on the analyzer part (function "filter") as well as on the generator part (function "equalizer"). In signal analysis, filters are mainly used to suppress certain frequency bands, to do measurements selectively or for weighting measurements according to the human ears' characteristic. Filtering of the generator output signal can be used to compensate the frequency response of the measurement setup or to simulate the effect of preemphasis.

"Filter"-Function of the Analyzer

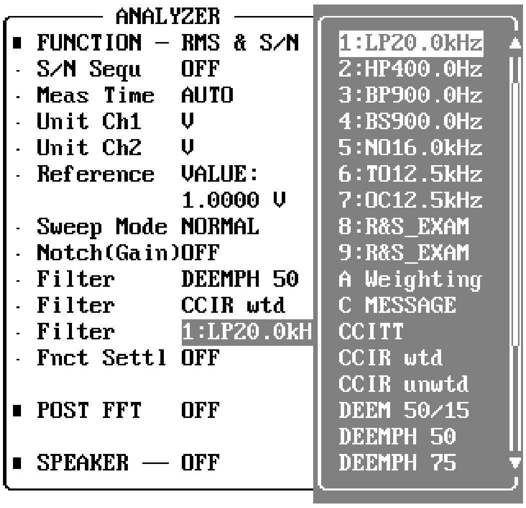


Fig.: 4-1

- different types of filters selectable
- all common weighting filters available
- with UPL: up to 3 filters can be set in the analyzer panel at the same time

Filter Characteristics

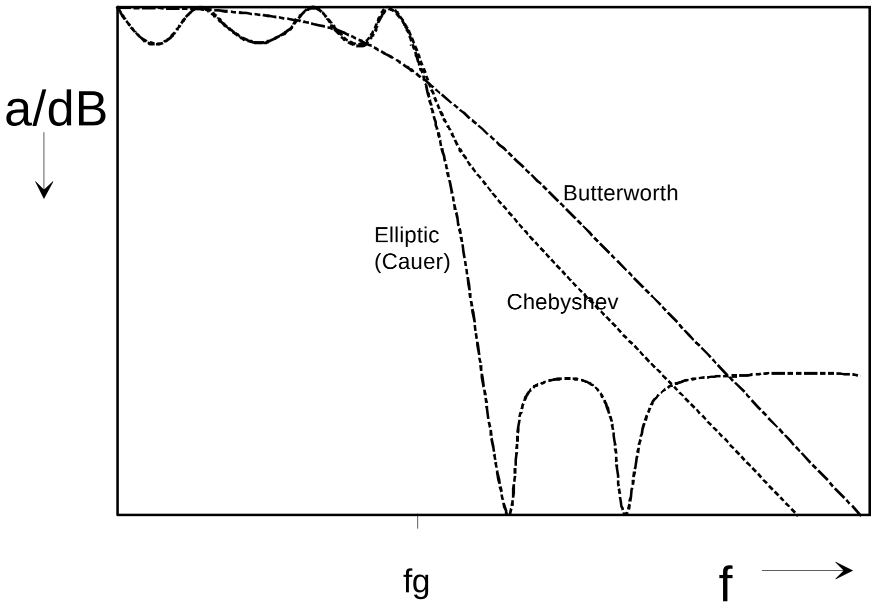


Fig.: 4-2

Features:

- Butterworth — no ripple; monotone frequency response
- Chebyshev — passband ripple; monotone frequency response in stopband
- Cauer — passband and stopband ripple; steep edges

Filter Settings

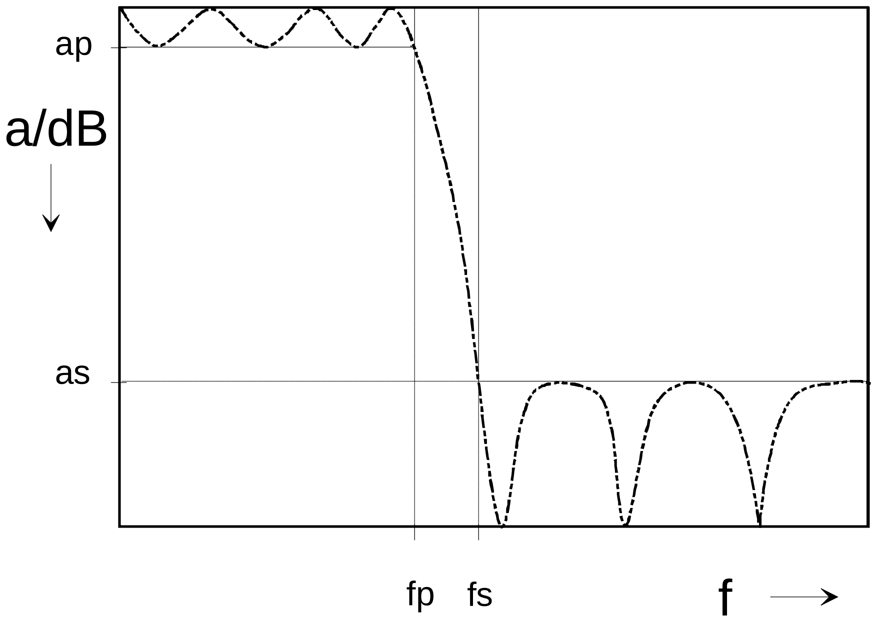


Fig.: 4-3

Passband (fp):

- lower resp. upper cutoff frequency of passband
- -0.1 dB point

Stopband (fs):

- read only display
- frequency where the set attenuation is reached the first time

Attenuation (as):

- to specify the filter attenuation, UPL sets to the next possible value
- minimum attenuation in stopband range

Ripple:

- defined by UPL
- + 0 / -0.1 dB

Steepness and Attenuation

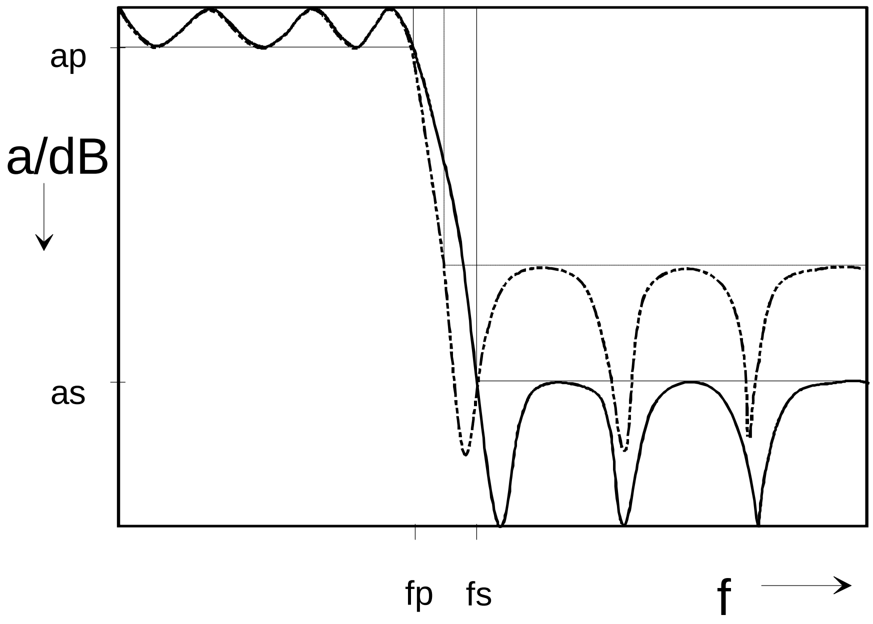


Fig.: 4-4

Cauer Filter get steeper with decreasing attenuation. If an extreme steep filter is needed, it might be better to cascade this by selecting two or three filters with the same characteristic, but less attenuation.

Examples for Filter Design Software:

- FDAS — Momentum Data Systems
files can be directly downloaded by UPL
- ASPI
- FILSYN
- FCAD — Linear Technology
- Monarch — Althena Group

User-definable Filters

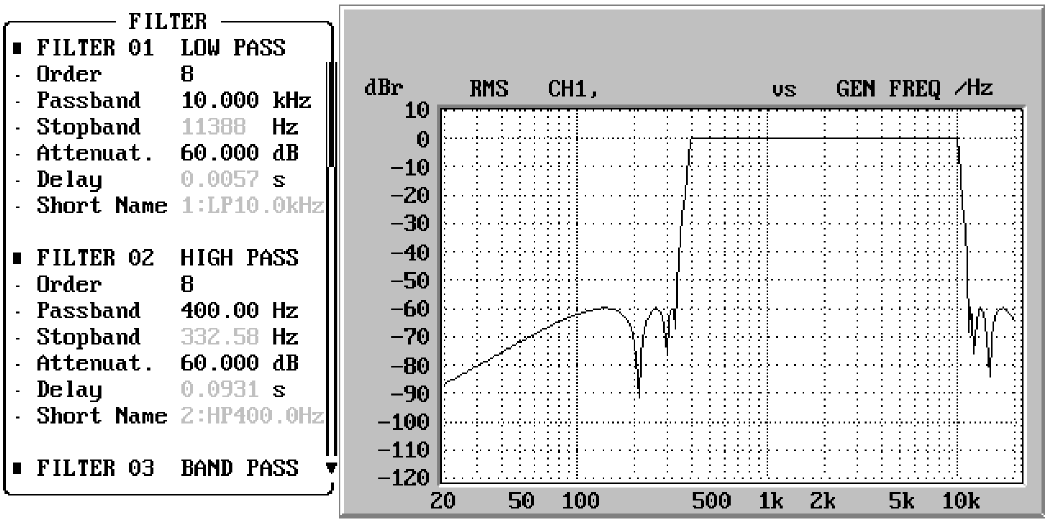


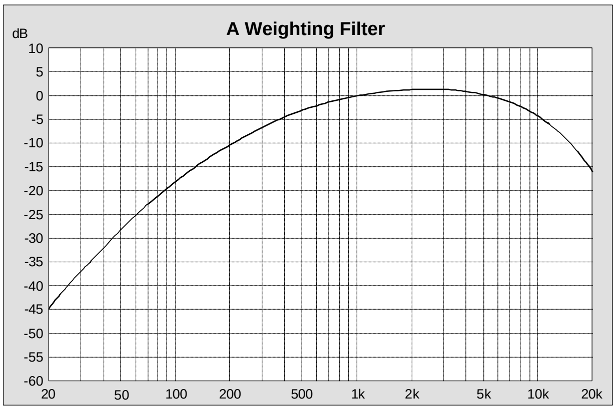
Fig.: 4-5

- 9 filter items can be stored with any setup
- type of filter selectable
- recursive filters (= IIR filters) with 4 or 8 poles;
  by reducing the filter order from 8 to 4, less steep but faster filters can be created
- Cauer characteristic derived from filter catalogue of SAAL 1988
- passband ripple < 0.1 dB absolute, that means frequency response less than ± 0.05 dB
- filters can be superposed, e.g. 40 dB lowpass filter two times assigned gives a very steep 80 dB lowpass (16 poles)
- short name is provided to the filters automatically (→Analyzer-panel)
- filters also can be read in from a file (calculated by filter design software)

Weighting Filters

- automatically matched to the current sample rate
- frequency response absolutely complies to the standards
- cannot be changed by the user

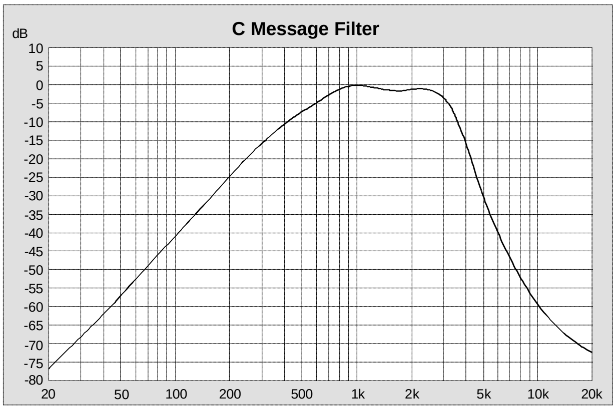
Available Weighting Filters:



Filter: A Weighting

Standard(s): DIN 45412

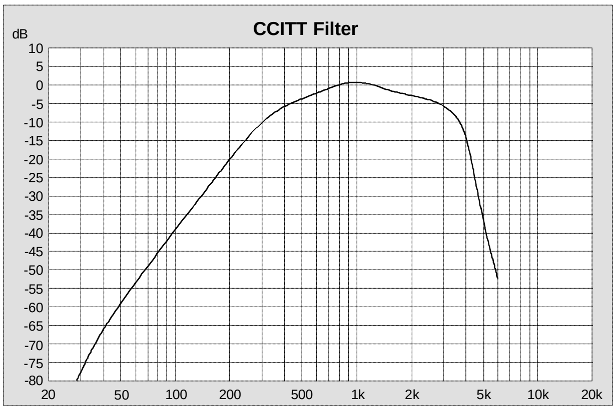
Application: weighting for noise measurements



Filter: C Message

Standard(s): IEEE Rec. 743-84

Application: transmission measurements



Filter: CCITT 0.41

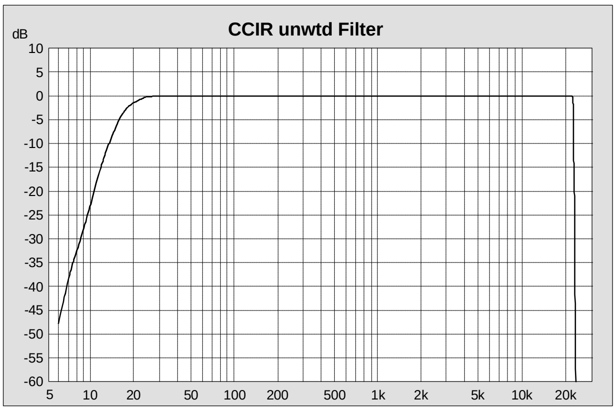
Standard(s): CCITT Rec. 0.41
IEEE Rec. 743-84
CISPR 6-76
CCITT Rec. P.53

Application: psophometric measurements

Filter: CCIR unwtg

Standard(s): CCIR Rec. 468-4

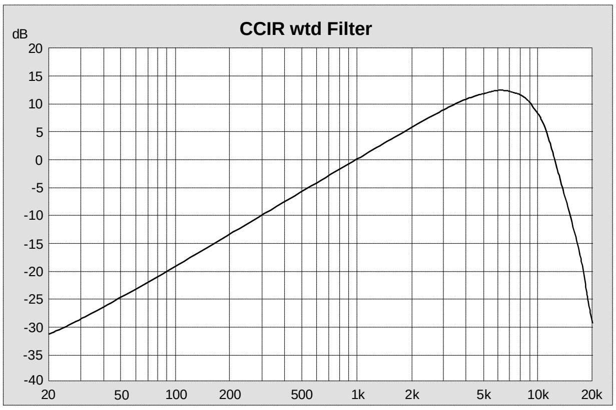
Application: bandpass from 22.4 Hz to 22.4 kHz for band-limited, unweighted measurements



Filter: CCIR 468-4

Standard(s): CCIR Rec. 468-4
DIN 45405
CCITT Rec. N.21
CISPR 6-76

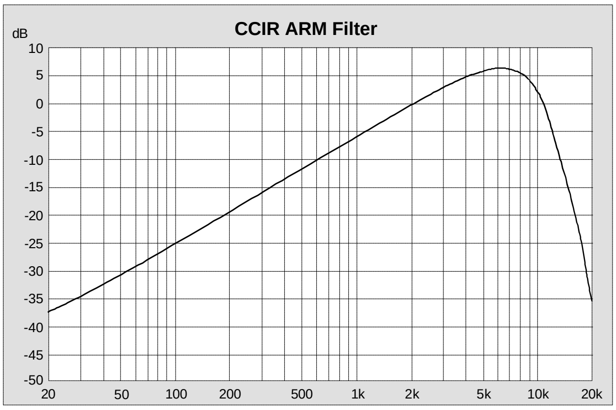
Application: weighting for noise measurements



Filter: CCIR ARM

Standard(s): CCIR
DOLBY
NAB Standard

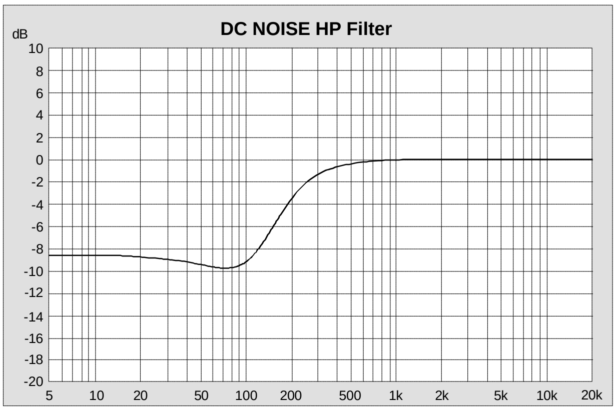
Application: weighted noise measurements



Filter: DC noise HP

Standard(s): ARD Spec. 3 / 4
ARD Spec. 12/2

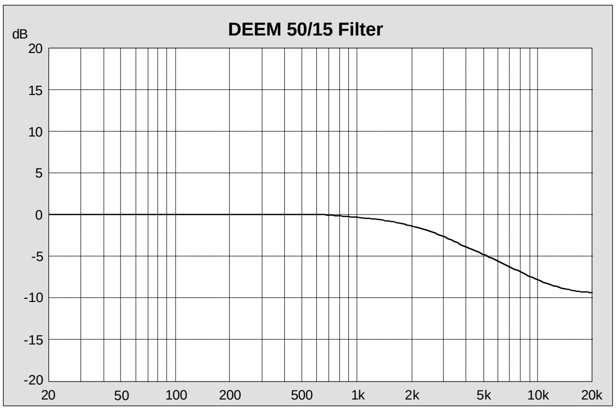
Application: highpass for DC noise measurements



Filter: DEEM 50/15

Standard(s): CCIR Rec. 651

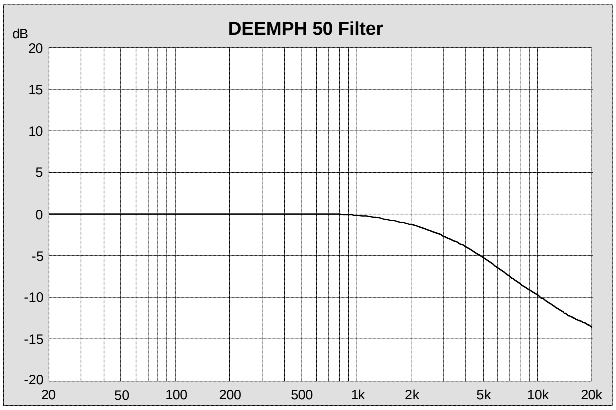
Application: CD measurements



Filter: Deemph 50

Standard(s): ARD Spec. 5/3.1

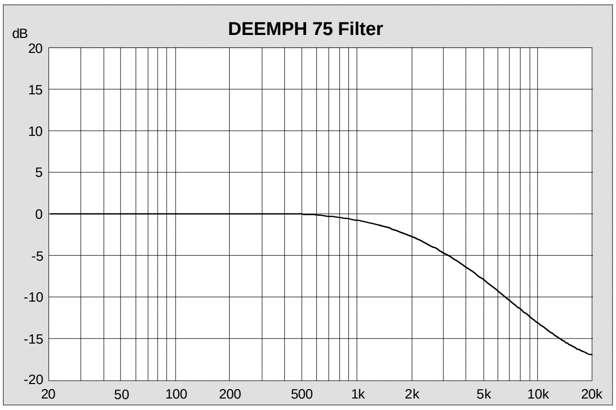
Application: interference and noise measurements to DIN 45405



Filter: Deemph 75

Standard(s): ARD Spec. 5/3.1

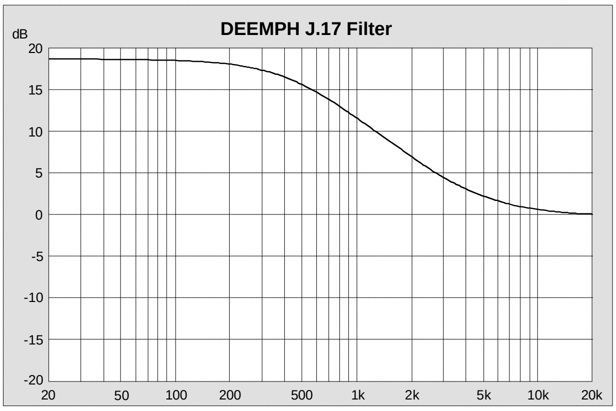
Application: interference and noise measurements to DIN 45405



Filter: Deemph J.17

Standard(s): CCITT J.17

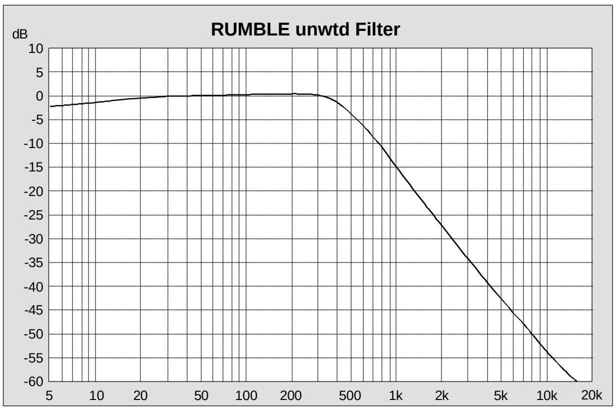
Application: interference and noise measurements to DIN 45405



Filter: Rumble unwtg

Standard(s): DIN 368.3
DIN 45539

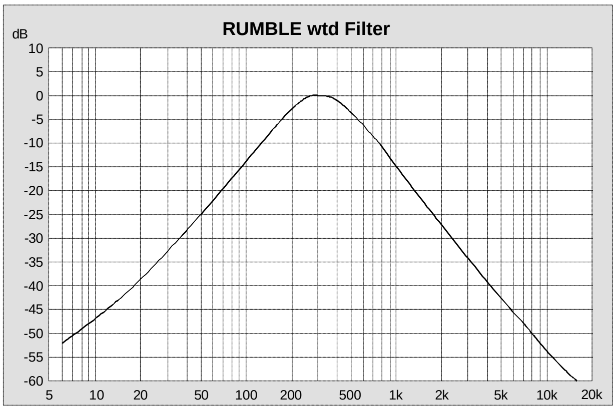
Application: testing of record players,
noise measurements



Filter: Rumble wtg

Standard(s): DIN 45539

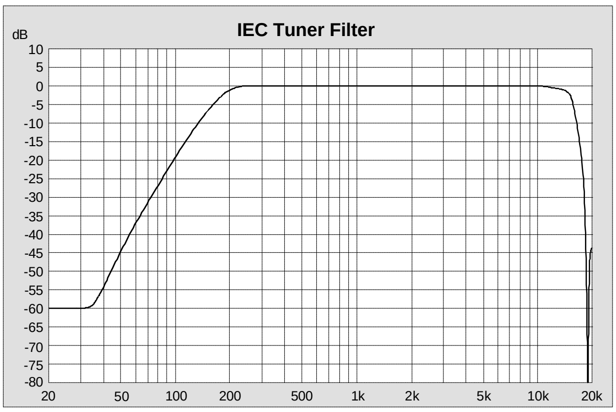
Application: testing of record players,
noise measurements



Filter: IEC TUNER

Standard(s): DIN/IEC 315

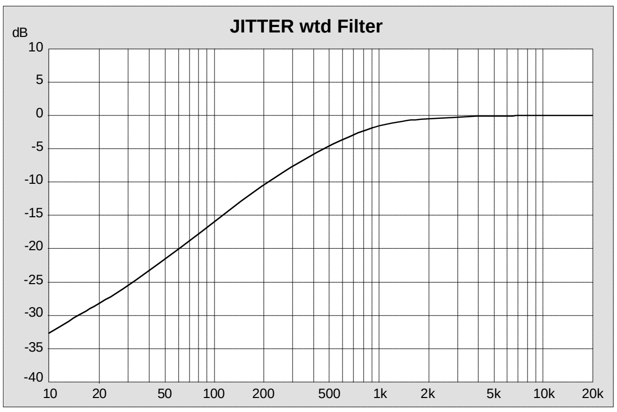
Application: measurements on
HiFi tuners



Filter: Jitter wtd

Standard(s): AES 3

Application: weighted measurement of
jitter transfer function



Filter Simulation

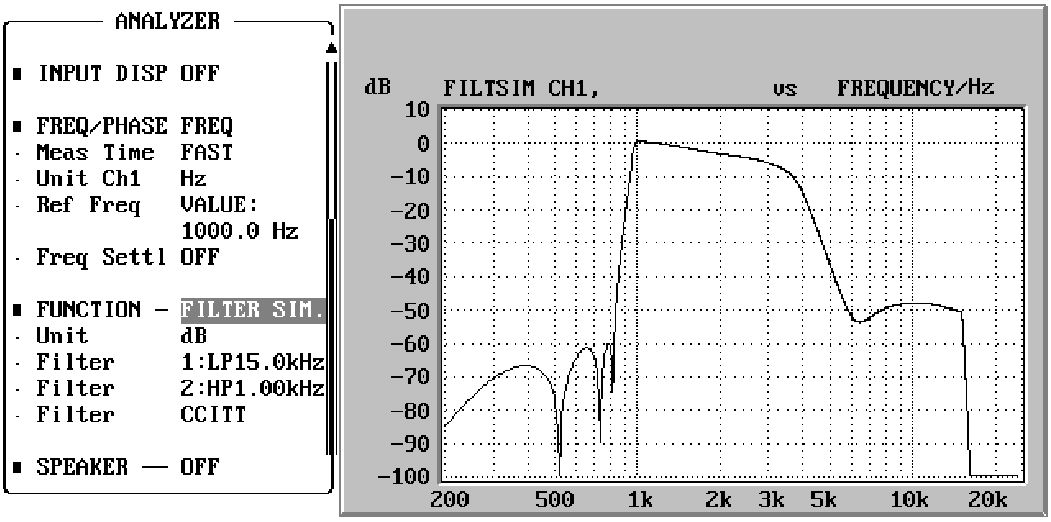


Fig.: 4-21

Displays the sum frequency response of a selectable combination of pre-defined filters as well as of user-defined filters.

Analog Notch Filter

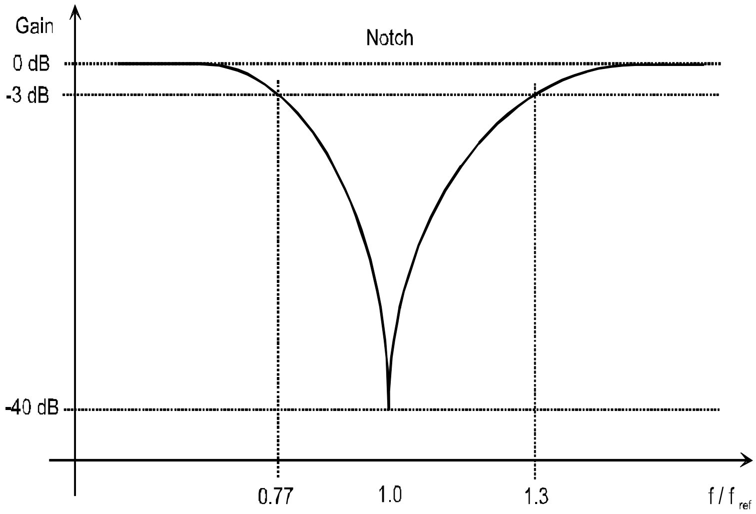


Fig.: 4-22

For the functions RMS & S/N, Q PK & S/N and FFT, the analog analyzers offer an analog notch filter of 2nd order to be activated for narrow-band suppression of interfering frequency lines.

Notch (Gain):

- OFF — filter off
- 0 dB — filter on, no gain effective
- 12 dB AUTO — filter on, gain 12 dB
- 30 dB AUTO — filter on, gain 30 dB
  if analyzer is overdriven, the notch gain will be reduced step by step, which is indicated by adding "AUTO".

Notch Freq:

- VALUE — numerical entry of the center frequency
- GEN TRACK — center frequency refers to the frequency specified under generator panel

"Equalizer" Function of the Generator

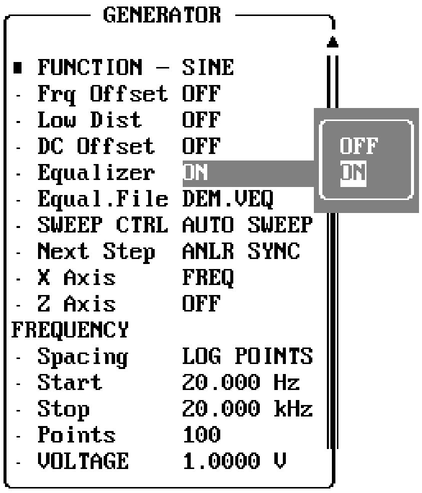


Fig.: 4-23

- generation of signals with defined frequency response
- available with sine, sine burst, multisine, DFD and random signals
- realized by activating a equalizer table which includes frequency specifications and the corresponding voltage amplification factors

Applications:

- simulating the effect of preemphasis
- frequency response compensation of the measurement setup
- measurement of the real frequency response of the device under test in comparison to the one of an ideal device

Example:
Setting to compensate the frequency response of the measurement setup:

1 measure the frequency response of the setup without DUT connected
2. store the measured trace as inverted equalizer file
3. connect the DUT
4. activate the equalizer function in generator panel and load the equalizer file
5. start the measurement, frequency response of measurement setup (i.e. amplifier, cables, etc.) will now be compensated

Basics of FFT Analysis

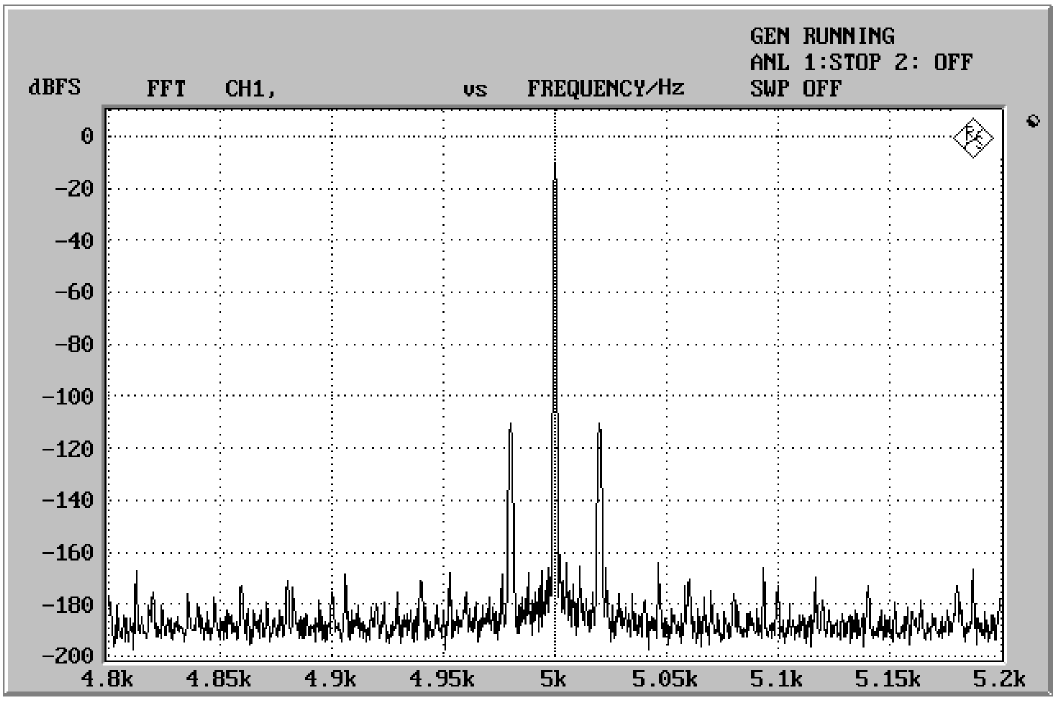


Fig.: 5-0

What is the purpose of the Fourier-Transformation?

- the Fourier-Transformation enables the mapping of any energy-limited signal by sin- and cos-functions

- the Fourier-Transformation (the frequency spectrum) of a signal contains frequency, amplitude and phase of the spectral components

What kind of signals are analyzed?

Every time-variable signal has a characteristic spectrum, i.e. every signal x(t) corresponds to a Fourier-Transform X(f).

Summary of different classes of signals and their Fourier-Transform:

the figure shows an overview of time signals and spectra with respect to:

- periodic / non-periodic
- discrete / continuous

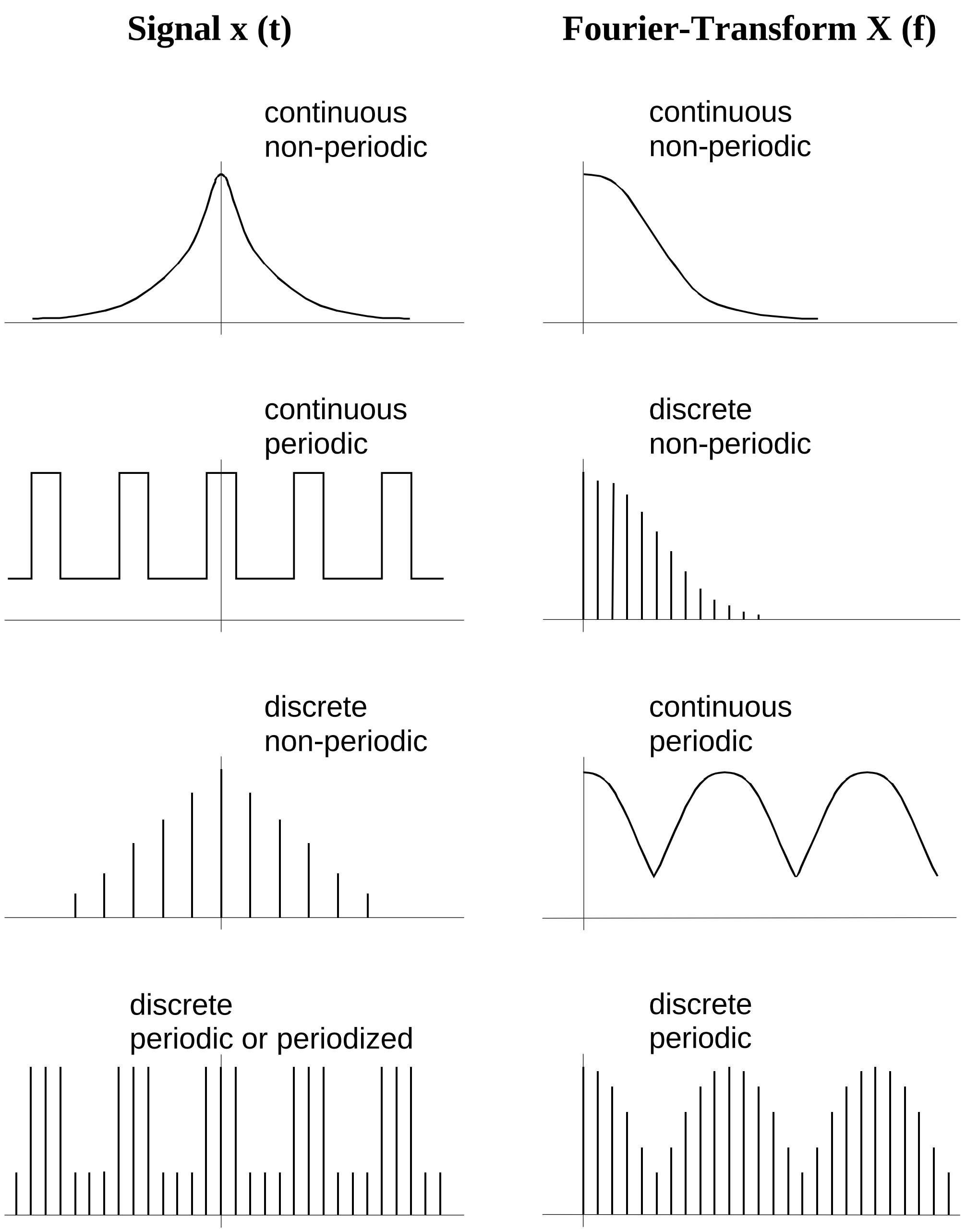


Fig.: 5-1

How is analysis done?

General Fourier-Transformation

$$F(jw) = \int_{-\infty}^{+\infty} f(t) \cdot e^{-jwt} dt$$

frequency domain $\leftrightarrow$ time domain

The **Discrete Fourier-Transformation (DFT)** is calculated from the sampled values of a time-limited signal f (t)

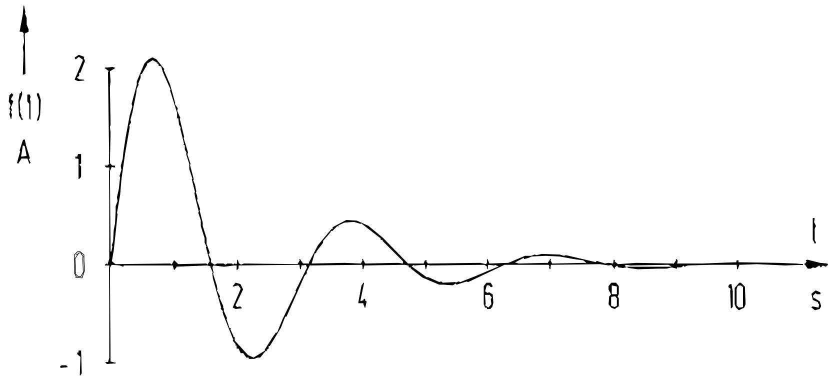


Fig.: 5-2 the transient time function $f(t) = 3\,e^{-t/T} \sin 2\pi ft$ with the damping factor T = 2s and a frequency of $f = 1/\pi s^{-1}$

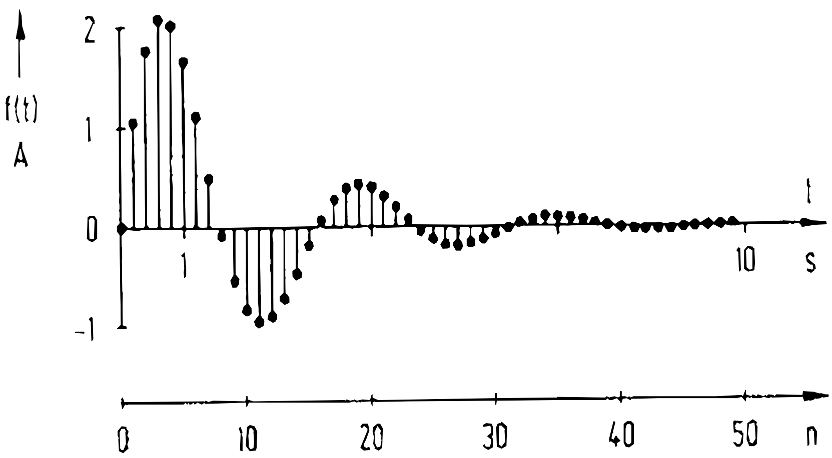


Fig.: 5-3 sampled time function: sample interval $T_S = 0.2$ s, number of samples N = 50, measuring time $NT_S = 10$ s

The samples of the signal f (t) exist only at discrete times nT_S. To get the calculation algorithm for the Discrete Fourier Transform (DFT) the following replacements are necessary:

- t is replaced by nT_S
- f(t) is replaced by f(nTs)
- $e^{-j\omega t}$ is replaced by $e^{-j\omega t\, nT_S}$

The Fourier Integral is then approximated by a sum of rectangles with a height f(nt) and a width of T_S

$$\int f(t)dt \approx T_s \sum f(nT_s)$$

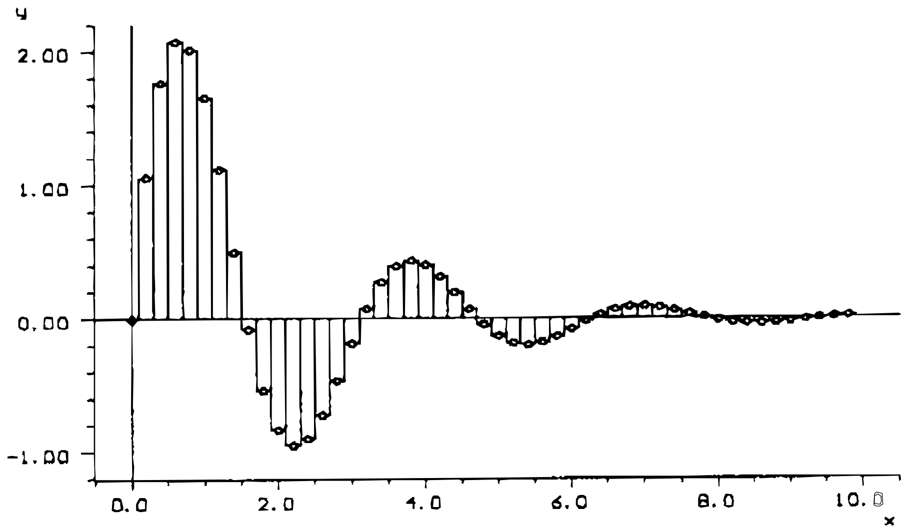


Fig.: 5-4

The factor T_S is omitted in the definition of the DFT resulting in the following algorithm:

$$\begin{aligned} F_d(jw) &= \sum f(nT_s) \cdot e^{-jwT_s} \\ &= \sum f(nT_s) \cdot \cos wnT_s - j \sum f(nT_s) \cdot \sin wnT_s \end{aligned}$$

From this equations the following properties are derived:

- for f the DFT is periodic with $1/T_S$

  Conclusion for sampling frequency f_S:
  the repetitive spectra of the DFT must not overlap; therefore the highest signal frequency f_{max} should be : $f_{max} < ½ f_S$ (sampling theorem)

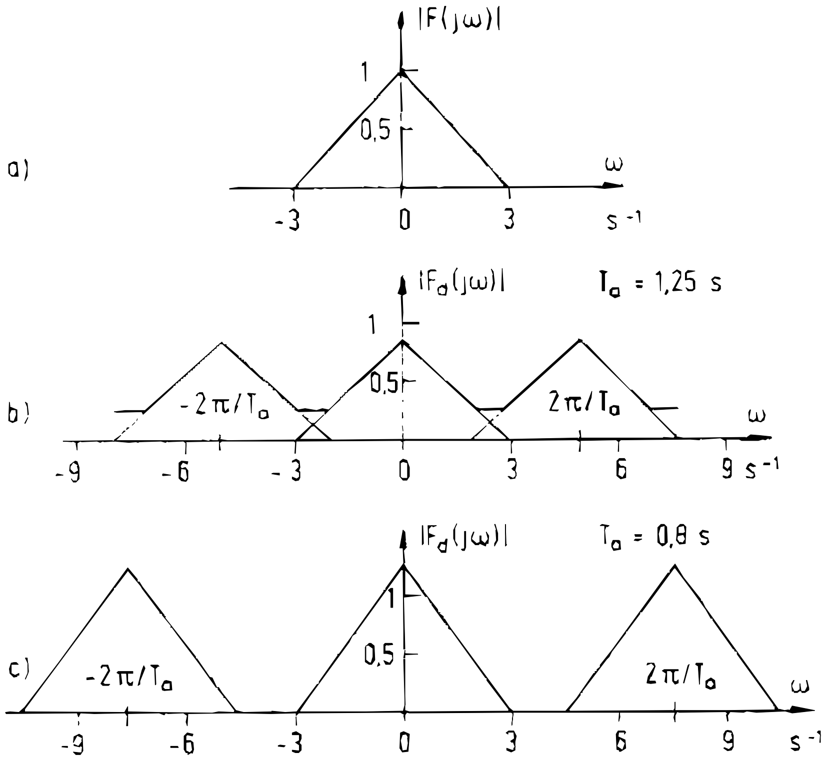


Fig.: 5-5

a) Fourier-transform of a bandlimited signal with $\omega_{max} < 3\ s^{-1}$

b) Envelope of the DFT of signal a) with a sample interval T_S = 1.25 s. Spectra are repetitive with a spacing of $\omega = 2\pi/T_S = 5\ s^{-1}$. Spectra overlap, the sampling frequency is too low. The maximum amplitude decreases from 1 to $1/T_S$ = 0.8.

c) Envelope of the DFT of signal a) with T_S = 0.8 s. The repetitive spectra do not overlap, sampling frequency f_S is sufficient. The maximum amplitude rises from 1 to $1/T_S$ =1.25.

- The DFT has to be calculated for discrete values of ω only, because there is only a limited number of discrete data f (nT_S) available. That means, only for a limited number of discrete frequencies amplitudes can be calculated.
  These discrete frequencies have a spacing of $\Delta f = 1/NT_S$.

  This is the spectral resolution of the measurement, it is reciprocal to the measurement time NT_S.

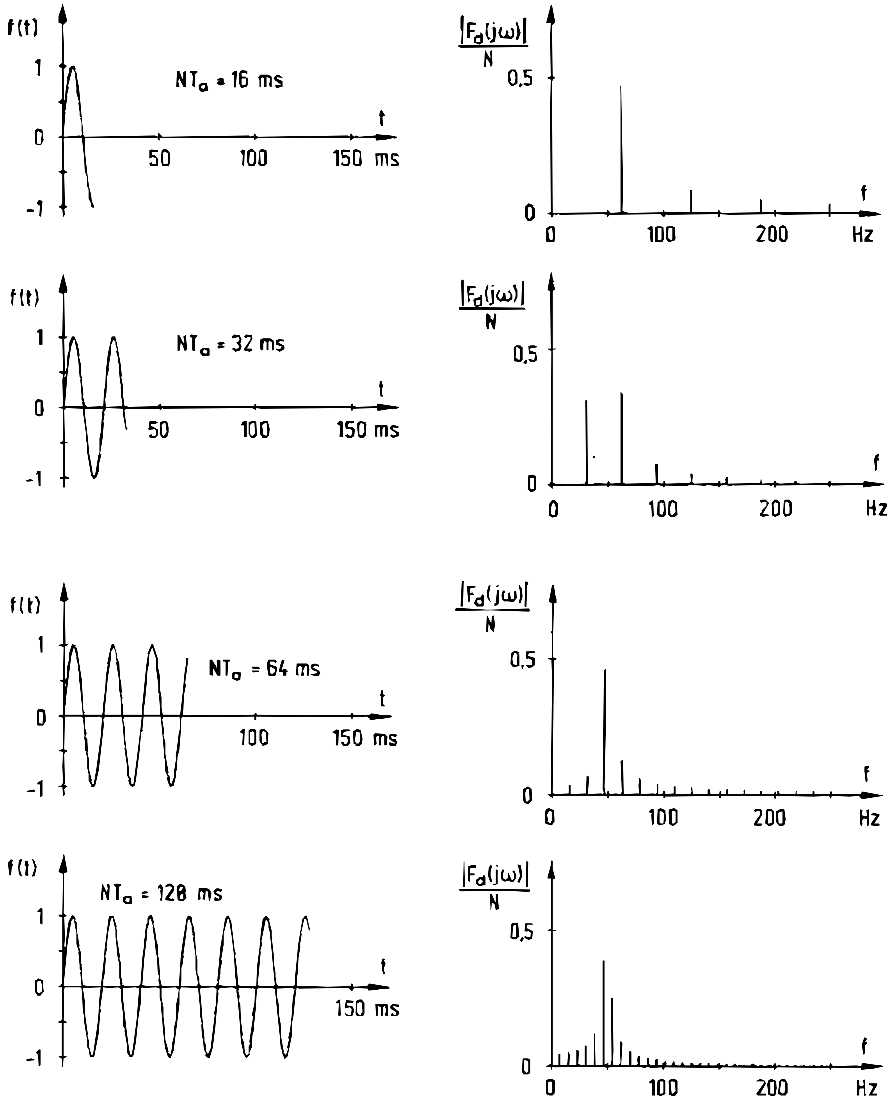


Fig.: 5-6 The spectral resolution is proportional to the measurement time NT_S. A 50 Hz sinusoidal is sampled with f_S = 1000 Hz, the DFT is calculated and shown. The discrete lines are spaced by $\Delta f = 1/NT_S$. The spectra repeat at f_S = 1000 Hz.

The above formula for the DFT needs N² complex multiplications to calculate the spectrum of the complete set of samples. The **Fast Fourier Transformation (FFT)** is a clever algorithm to calculate the DFT that reduces the necessary multiplications to $n \cdot \log_2 N$ by using certain symmetrics of the sampled data. The FFT algorithm was first described by Cooly and Tukey (IBM) in 1965.

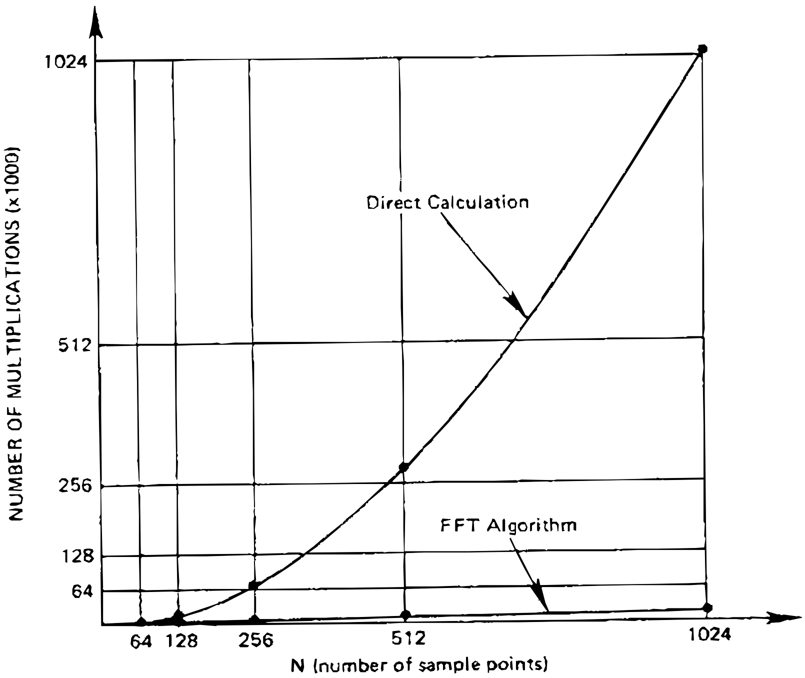


Fig.: 5-7 Comparison of multiplications required by direct calculation and FFT algorithm

Application of the DFT is done in four steps:

1st step: waveform sampling

2nd step: truncation (time-limitation because of the limited number of samples)

3rd step: "periodization" of the time signal

4th step: application of the DFT equation (generally in form of the FFT algorithm)

Two problems are evident:

→ truncation

→ periodization

Problem of truncation (= time limitation)

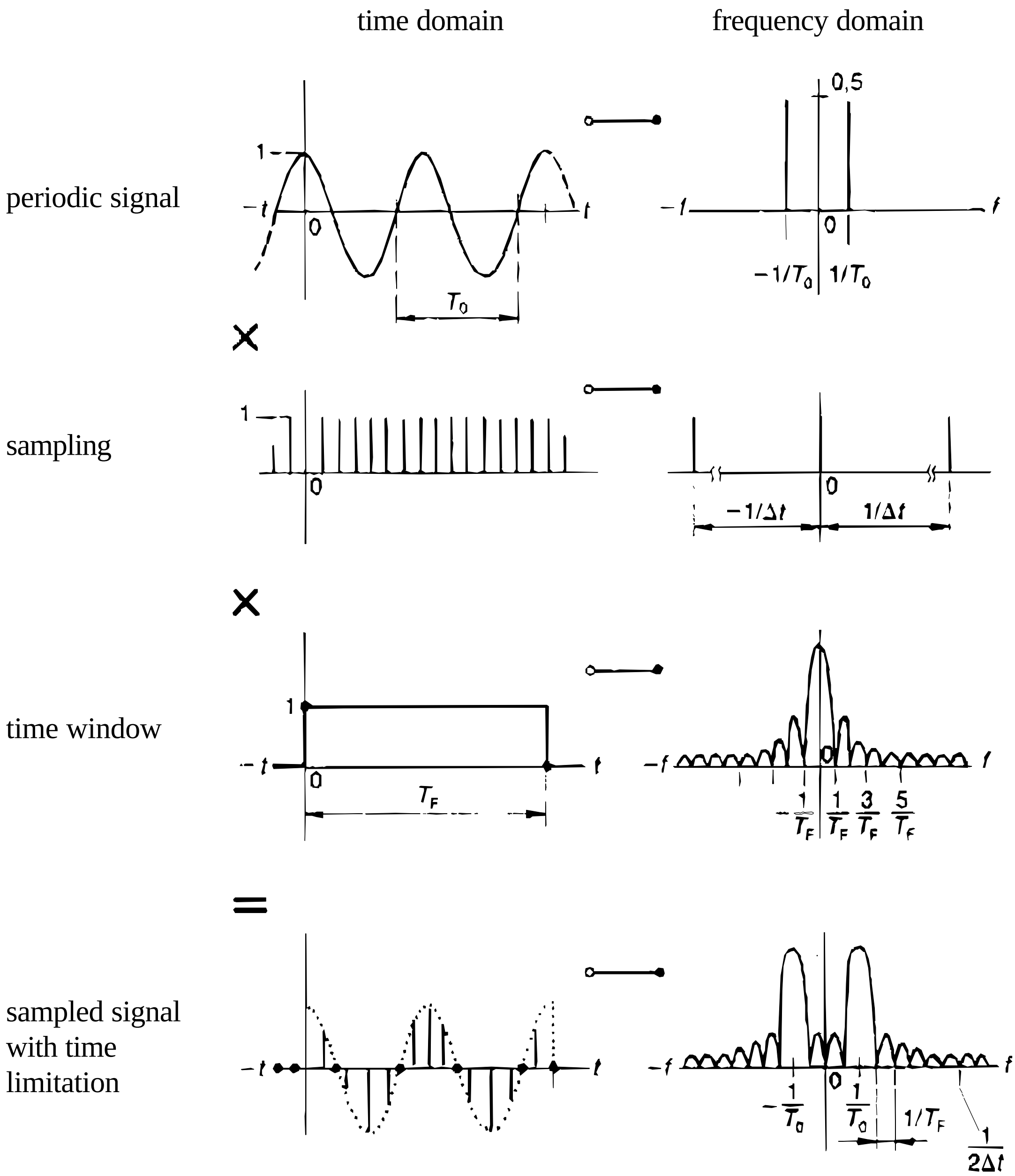


Fig.: 5-8 due to time limitation the spectral resolution is reciprocal to the measuring time

Problem of "periodization"

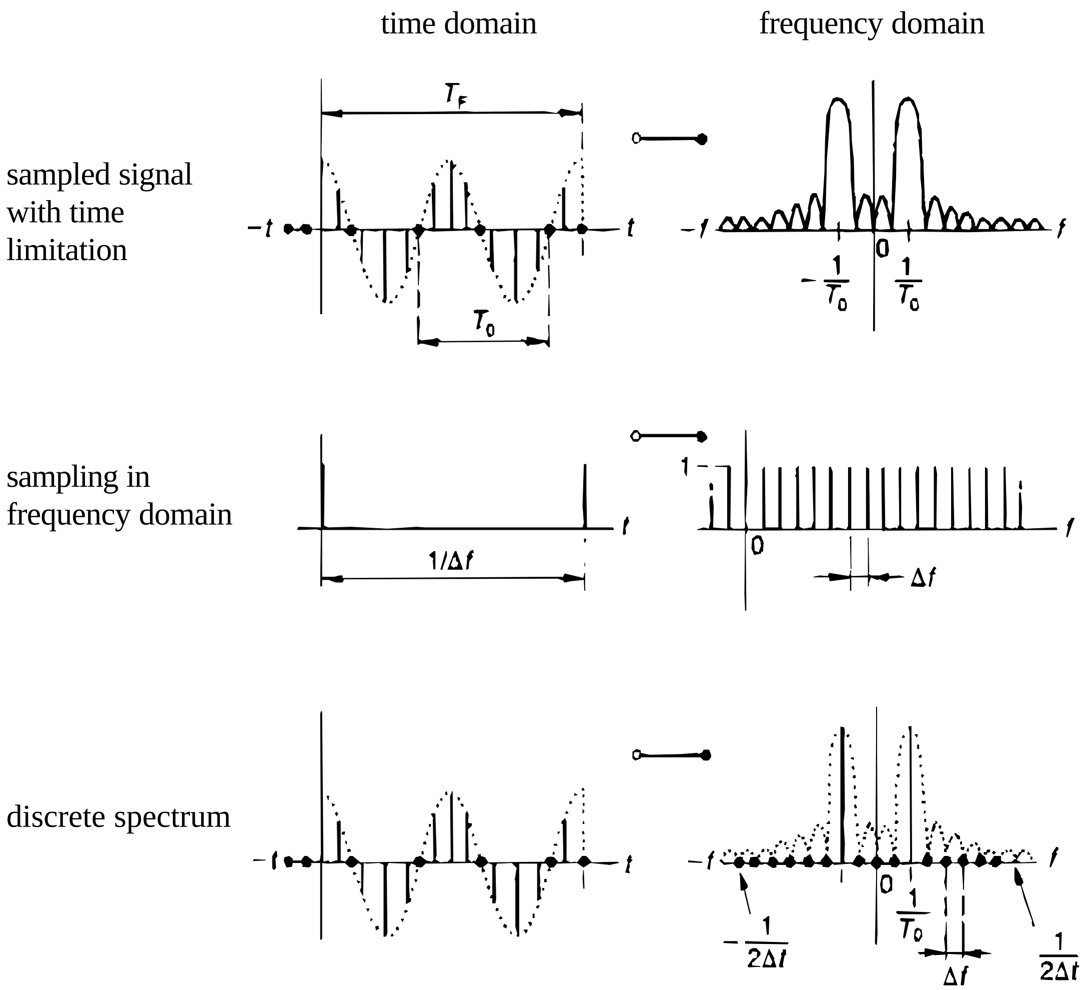


Fig.: 5-9 DFT of a band-limited periodic waveform with a truncation interval equal to period

This example represents the only case for which the discrete and the continuous fourier transform are exactly the same within a scaling constant. Equivalence of the two transforms requires:

→ the time function must be periodic

→ the time function must be band-limited

→ the sampling frequency must be at least two times the largest frequency component of the signal

→ the truncation (time window, measuring window) must be exactly an integer multiple of the signal period

Comparison between band-limited periodic waveforms
with a truncation interval **equal to period** and **not equal to period**

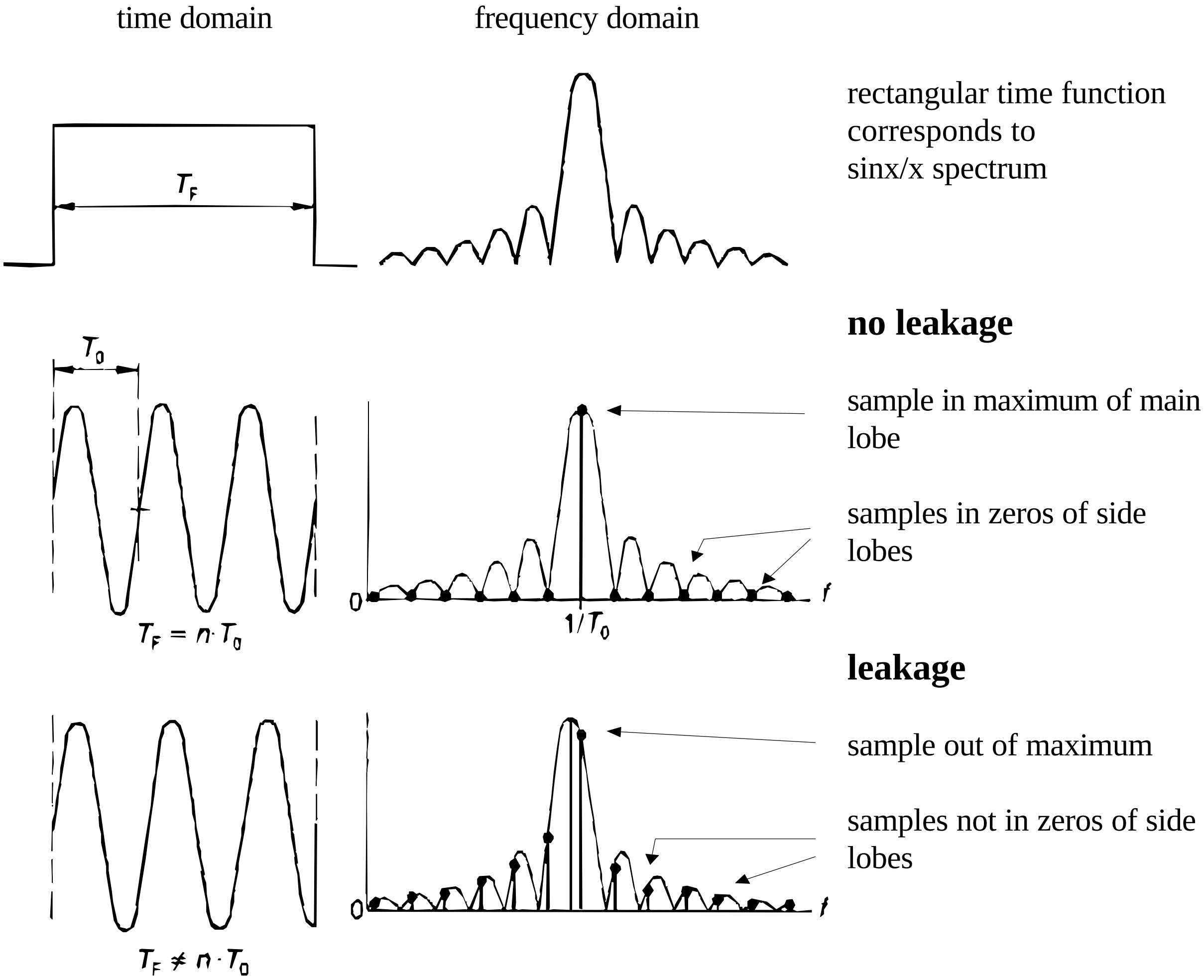


Fig.: 5-10

For signal (3) the zeros of the *sin x/x* - function are not coincident with each sample value as in the previous example (2), resulting in additional impulses in the frequency domain.

The effect of truncation at other than a multiple of the period is to create a periodic function with sharp discontinuities. Intuitively we expect that these sharp changes in the time domain result in additional components in the frequency domain.

Viewed in the frequency domain a time domain truncation is equivalent to the **convolution** of a *sin x/x* - function with the discrete spectral line representing the original frequency function (remember, a periodic function has a discrete spectrum i.e. spectral lines of zero width).

Consequently, the spectrum is not composed of discrete lines but rather a continuous frequency function with a maximum centred at the original line and a series of so-called sidelobes. This effect is termed **leakage** and is inherent in the DFT because of the unavoidable time domain truncation.

To minimize truncation errors so called **window functions** are used. The objective is to attenuate the sampled signal at its "edges", so that no steps occur.

Compared to the rectangular window (= no window at all) all other windows show a broadening of the main lobe but a better damping of the side lobes.

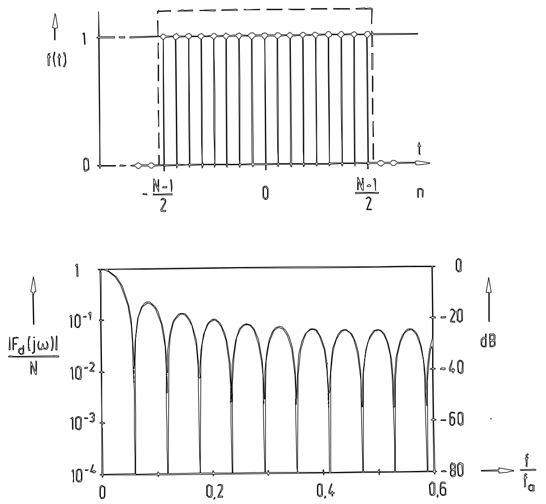


Fig.: 5-11

Rectangular Window

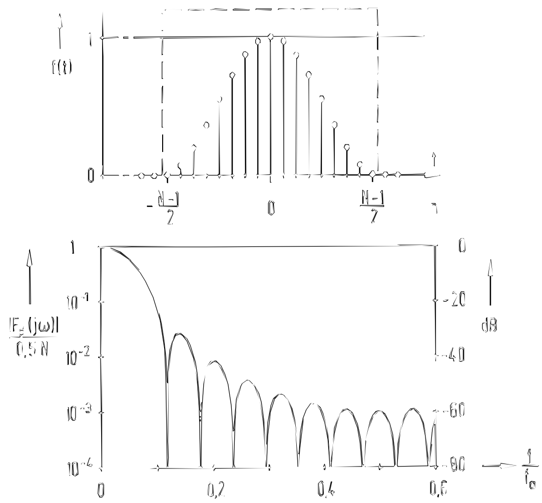


Fig.: 5-12

Hann Window

All window functions are a compromise between

- → frequency resolution of the main lobe (maximal for the rectangular window)
- → side lobe attenuation

Literature:
Schnorrenberg, Werner: "Spektrumanalyse", Vogel Verlag Würzburg 1990
Schrüfer, Elmar: "Signalverarbeitung", Hauser Verlag München-Wien 1992
Eckenberger, Ulrich: "FFT-Analyse", Rohde & Schwarz München

FFT Analysis with UPL

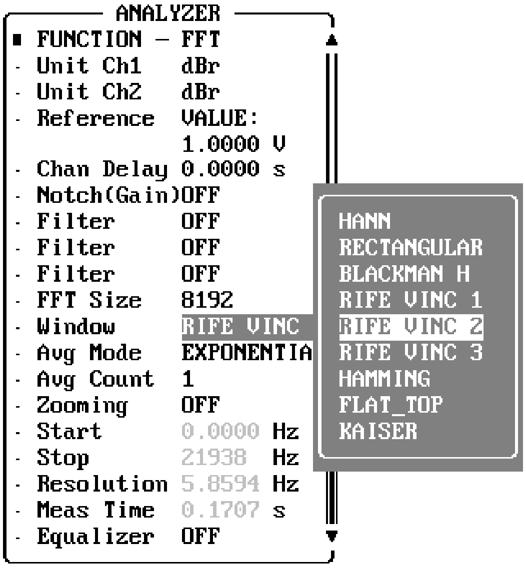


Fig.: 5-13

Channel Delay

- using dual-channel measurements, time delays of the device under test can be compensated for example to compare input and output spectra

FFT Size:

- FFT size, settable from 256 to 8k (16k using zoom FFT) in binary steps; the larger the FFT size, the better the frequency resolution, however the longer the measuring time

Window:

• HANN	for standard applications
• RECTANGULAR	no window function
• BLACKMAN-H	quite steep, however considerable leakage
• RIFE-VINC 1/2/3	excellent suppression of distant interferences, with increasing order the width of the mainlobe decreases
• HAMMING	no significant advantage
• FLAT TOP	top of the mainlobe is as wide as to include always two adjacent lines with approximately the same height
• KAISER	adjustable by setting the parameter β

Average Mode:

• NORMAL	averaging using the set number of FFTs
• EXPONENTIAL	continuous averaging with exponential characteristic

Zoom FFT

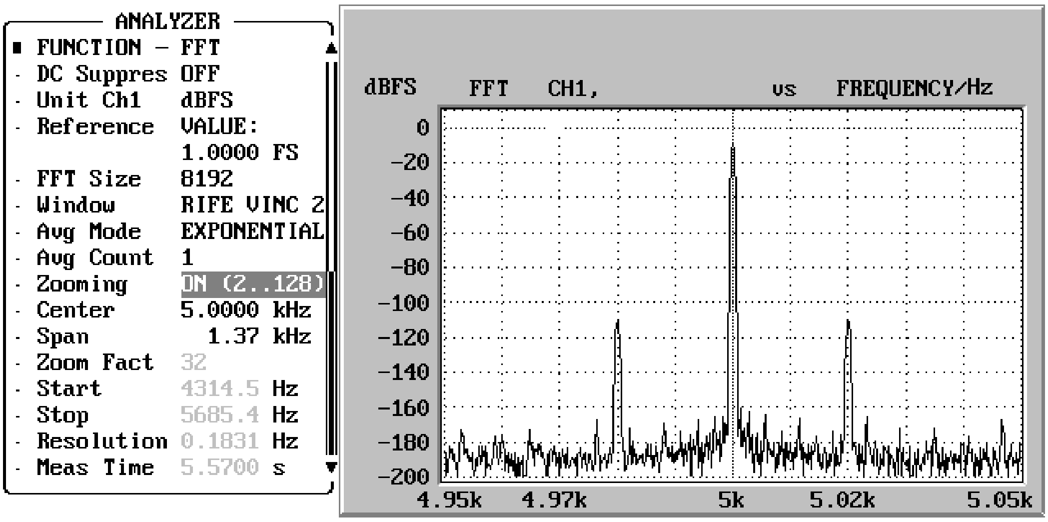


Fig.: 5-14

Zooming:

- OFF — "normal" FFT analysis
- ON(2...128) — FFT analysis using 16k points and zoom of the frequency range by selecting center frequency and span
  - selectable factors 2, 4, 8, ... up to 128
  - measuring time doubled with each zoom step

Post FFT

Post FFT is a FFT analysis which is calculated after the main measurement function. It gives additional information about the test signal, for example besides the THD+N result, the spectrum is displayed to give more details about the different spectral components.

- available with the functions
  - RMS & S/N
  - THD+N / SINAD
  - WOW & FLUTTER

Transfer and Coherence Function

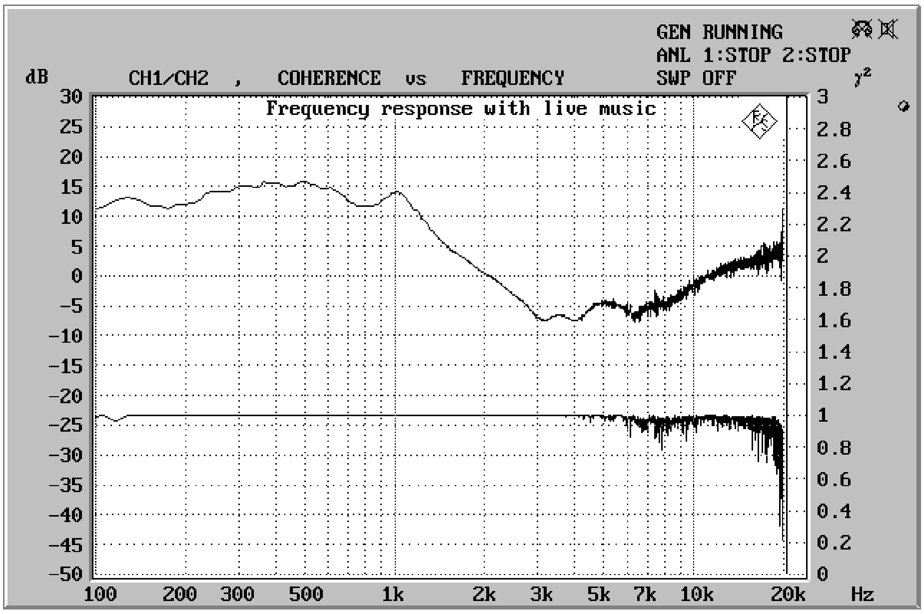


Fig.: 5-15

Application:

Modern devices like coders or intelligent hearing aids use dynamic signal processing where the transfer characteristic depends on the input signals. Stationary test signals as used with conventional measurements do not activate these dynamic processes. Transfer and coherence function use test signals which cover the entire audio spectrum simultaneously, like noise or even music or human speech. Therefore they stimulate also the dynamic processing of the device-under-test.

Measurement Method:

A complex FFT is calculated for each channel (input and output of the device-under-test), the cross-correlation between the two FFTs is formed and averaged.
The two channels are measured at exactly the same time, for delays through the device-under-test, an interchannel delay of up to ± 10 seconds can be keyed in.
Transfer function gives a sort of "frequency response" information, while coherence shows how much the output of the device depends on its input signal. Coherence = 1 means that at this frequency channel 2 is linear dependent on channel 1.

Non-standardized Digital Interfaces

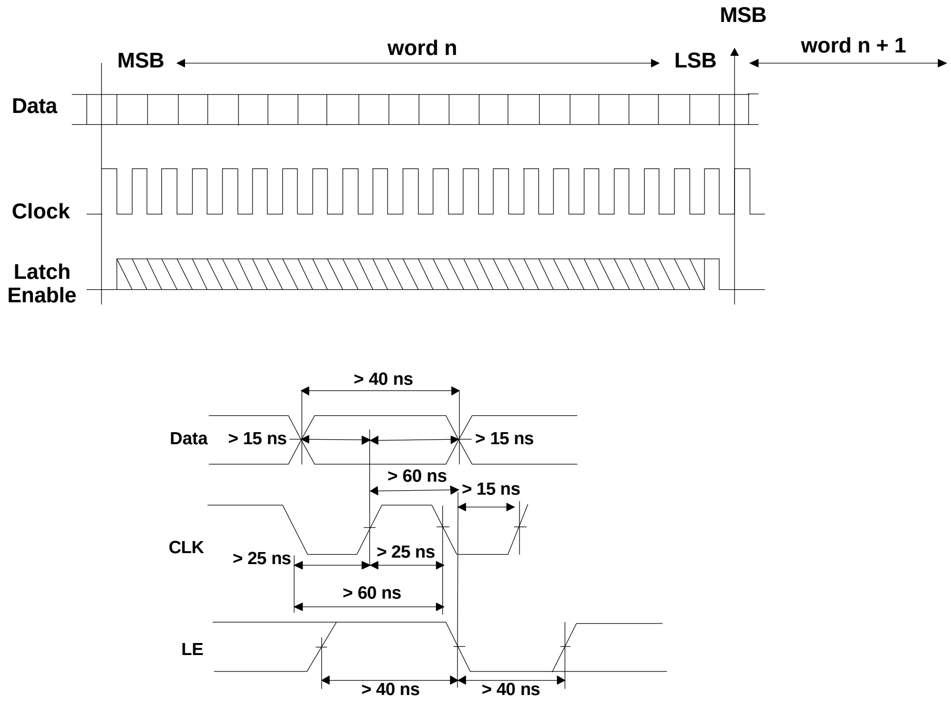


Fig.: 6-1: Example from the data sheet of a 20 bit D/A converter

Very often converter chips and other integrated circuits for audio applications use three serial lines at their inputs / outputs:

- data signal
- bit clock
- word clock (latch enable)

The timing of these different signals can be taken from the data sheets of the manufacturers.

AES / EBU-Interface (according to AES 3)

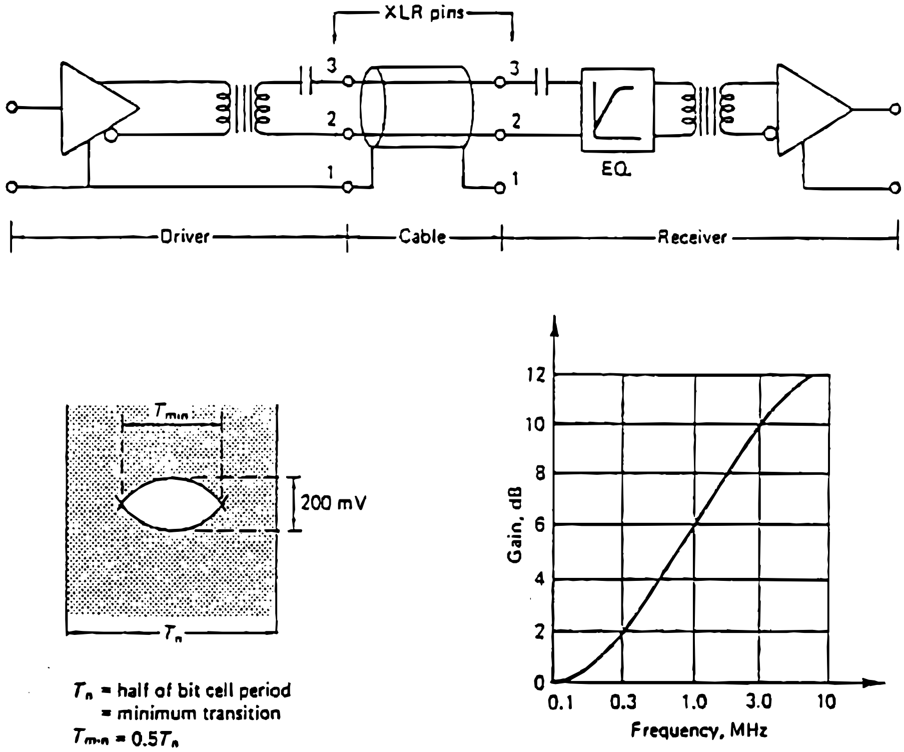


Fig.: 6-2

- shielded twisted wire pairs like used with balanced analog transmission
- impedance 110 Ω, XLR connector
- driver: 2 to 7 V $_{\text{peak to peak}}$
- sensitivity of receiver: see eye pattern
- suggested equalization for cable length of more than 100 m

Bi-Phase Mark Coding

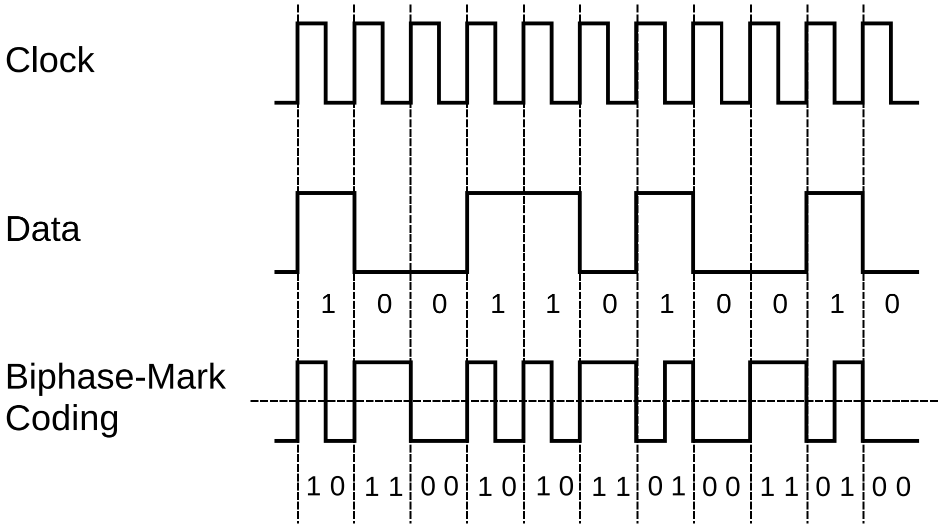


Fig.: 6-3

Principle:

- clockrate = 2 x bitrate
- with "Low"-signal change with every bit
- with "High"-signal additional change with every bit

Advantages:

- self-clocking
- DC-free
- polarity independent

Frame Structure

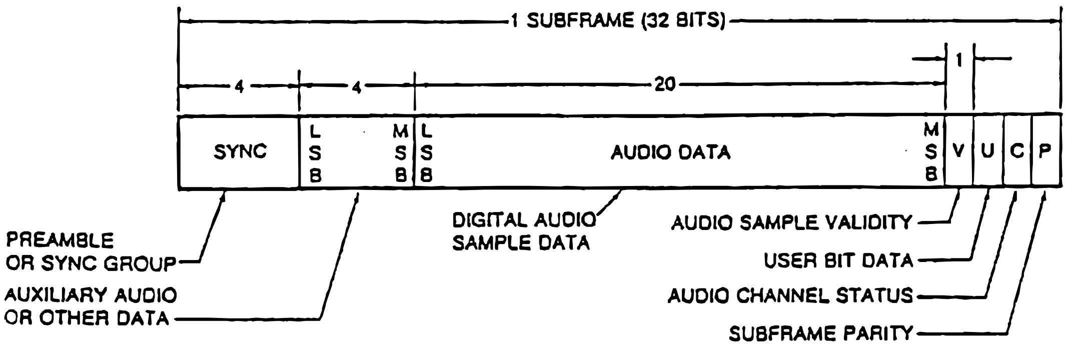


Fig.: 6-4

Frame Structure:

- sample rates 22.05 kHz, 24 kHz, 32 kHz, 44.1 kHz, 48 kHz, 88.2 kHz
  96 kHz, 176.4 kHz and 192 kHz
- 32 bit subframes
- 2 subframe = 1 frame (channel A and B)
- 20 bit audio data
- 4 bit additional data
- validity bit: 0 = audio sample is errorfree
  1 = sample is defective
- user bit
- channel status bit
- parity bit

Synchronisation of the Frames

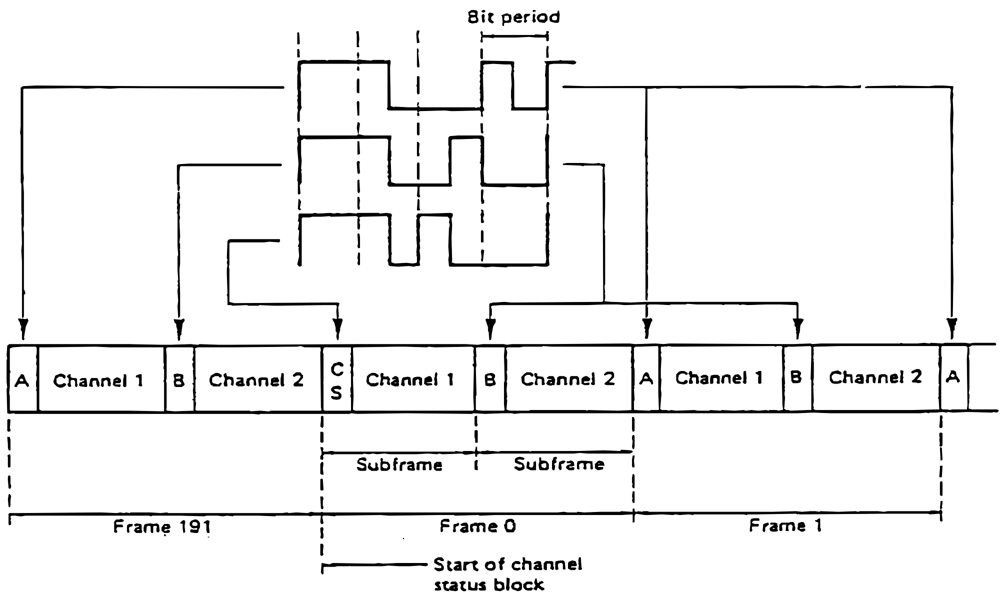


Fig.: 6-5

- 1. subframe = channel A (left)
  2. subframe = channel B (right)
- each subframe starts with a preamble (4 bit)
  to distinguish channel A or B
- 192 frames build up a block, which is marked by a special preamble
  ("Channel Status Block Sync", always indicates channel A!)

Coding of Channel Status Bits

<table>
<tr><th>Byte</th><th>Bit 0</th><th>1</th><th>2</th><th>3</th><th>4</th><th>5</th><th>6</th><th>7</th></tr>
<tr><td>0</td><td>Format:
consumer /
professional</td><td>Mode:
linear PCM /
non-PCM</td><td colspan="3">Audio signal emphasis:
not ind – no emph – 50/15 – J.17</td><td>Locking of source sample frequency</td><td colspan="2">Sampling rate:
not ind – 32 kHz–
44.1 kHz – 48 kHz</td></tr>
<tr><td>1</td><td colspan="4">Channel mode:
not ind – double left – stereo – mono – user(6) –
two chan – double right – user(10) – prim/sec –
double vect – multi chan</td><td colspan="4">User bit management:
not ind – AES18 – block – user def</td></tr>
<tr><td>2</td><td colspan="3">Use of auxiliary sample bits:
no – 20+coord – 24 – user def</td><td colspan="3">Source word length:
not ind – 20/16 – 22/18 – 23/19 –
24/20 – 21/17</td><td colspan="2">Alignment:
not ind – EBU R68 –
SMPTE RP155 - reserved</td></tr>
<tr><td>3</td><td colspan="8">Future multichannel function description</td></tr>
<tr><td>4</td><td colspan="2">Grade:
n.d. – 2 – 1 – reserved</td><td></td><td colspan="4">Enhanced rate:
not ind. – 22.05 kHz – 24 kHz – 88.2 kHz – 96 kHz – 176.4 kHz – 192 kHz – user def.</td><td>Scaling:
off – 1/1.001</td></tr>
<tr><td>5</td><td colspan="8">Reserved</td></tr>
</table>

Byte	
6 7 8 9	**Alphanumeric channel origin data**
10 11 12 13	**Alphanumeric channel destination data**
14 15 16 17	**Local sample address code (32-bit binary)**
18 19 20 21	**Time-of-day sample address code (32-bit binary)**
22	**Reliability flags**
23	**Cyclic redundancy check character**

Fig.: 6-6

- coding separately for both channels
- 192 bits build up 24 bytes (each of 8 bits)

Consumer Interface S/P DIF (acc. to IEC 958)

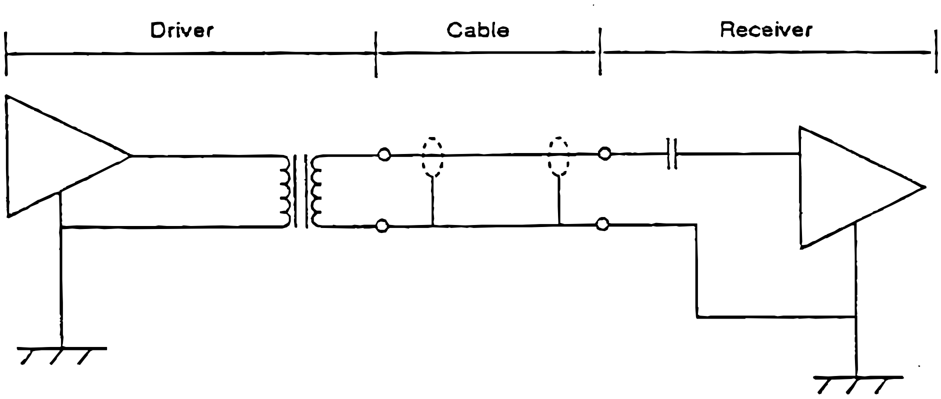


Fig.: 6-7

- unbalanced 75 Ω-line
- BNC- or Cinch-connector
- amplitude 0,5 V_{peak}

Note:

According to AES 3, in the meantime the AES / EBU-interface has also be defined as unbalanced 75 Ω-lines, amplitude 1 V !
Audio Analyzer UPL already fulfils this recommendation.

Coding of Channel Status Bits

Byte	Bit 0	1	2	3	4	5	6	7
0	**Format:** consumer professional	**Mode:** linear PCM non-PCM	**Copy:** protect free	**Emphasis:** no emph 50/15		**Channels:** 2 chans 4 chans	**Mode:**	
1	**Category:** general – CD-IEC – PCM – DAT – BC-Japan – Synth – ADC n – CD – VTR – BC-Europe – micro – mixer – ADC – Fs-conv – sampler – DCC – soft – BC-US							**L-bit:** ni/1st Gen. ori/pre
2	**Source:**				**Channel:** D.C. – A – B – C – D – E – F – G – H – I – J – K – L – M – N – O			
3	**Rate:** not ind. – 22.05 kHz 24 kHz – 32 kHz – 44.1 kHz – 48 kHz – 88.2 kHz – 96 kHz – 176.4 kHz – 192 kHz				**Precision:** 1000 ppm – 50 ppm 12.5% – reserved			
4-23								

Fig.: 6-8

MADI Interface (Multi Channel Audio Digital Interface)

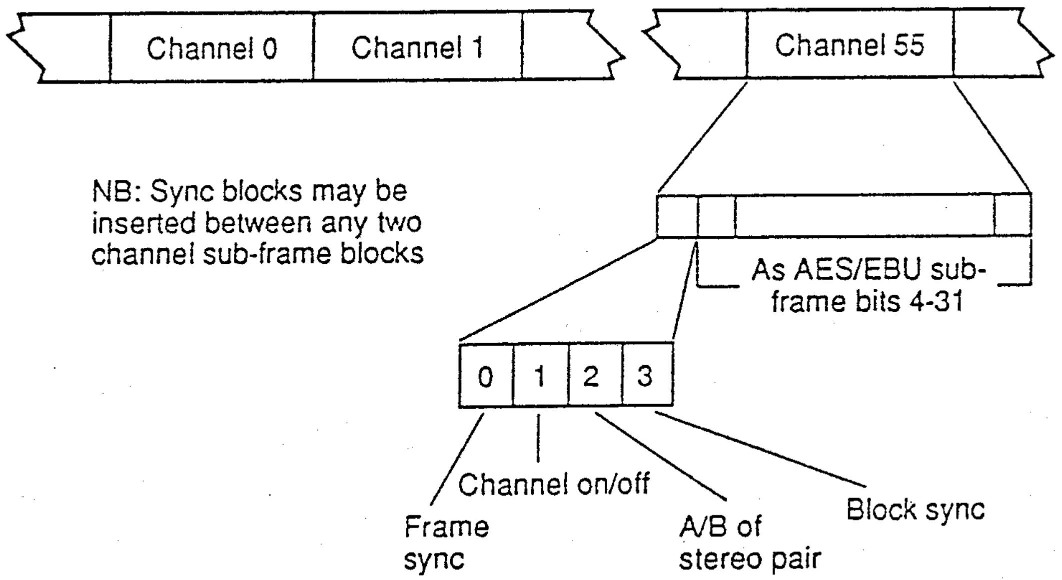


Fig.: 6-9

While AES/EBU and S/P DIF interface are designed for stereo applications, MADI is used for digital connections between multichannel recorders or mixing consoles. MADI has been designed by the companies SONY, MITSUBISHI, NEVE and SSL.

MADI offers 56 digital audio channels, which are transmitted by a single BNC cable. A second cable is used for the sync signal. MADI mainly uses the subframe structure of the AES/EBU protocol; but the first four bits are different. All 56 subframes are multiplexed within one sample period.

Generating of Protocol Bits

many possibilities to generate:

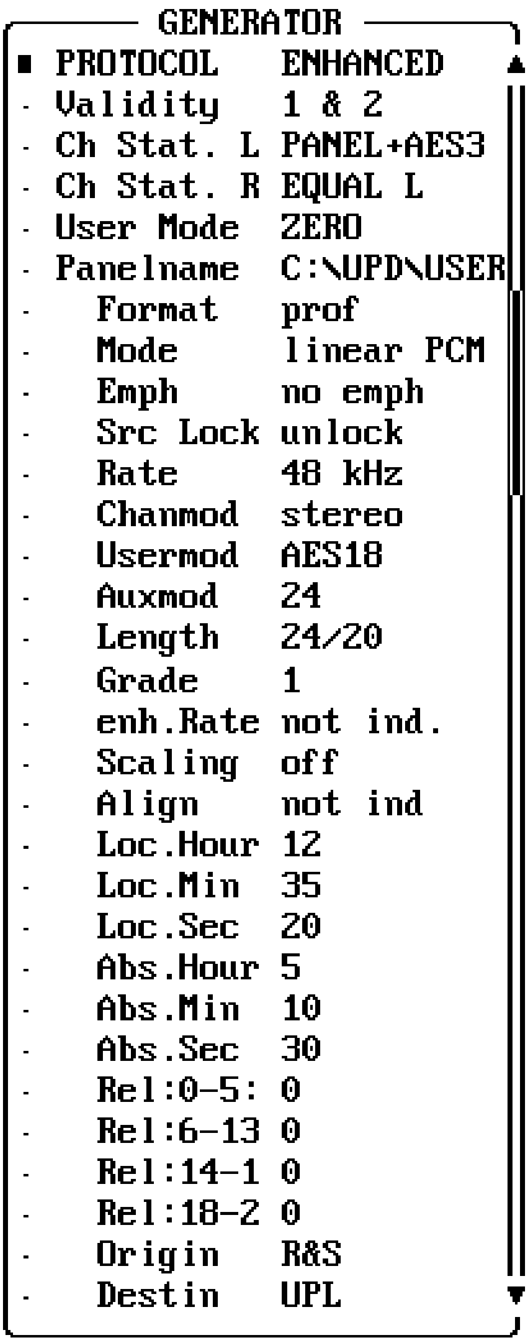


Fig.: 6-10

Valid Chan:	sets the validity identification in channel 1, 2 or 1&2
Ch Stat.L/R	defines the mode of generating the channel status data: data set to zero, load from a file or defined by panel entry
User Mode:	sets the user data to zero or generates user data by reading from a file
Panel Name:	specification of a file to define the programmable panel for setting the channel status bits, see an example to set the bits according to AES 3

Binary Input of Channel Status Bits

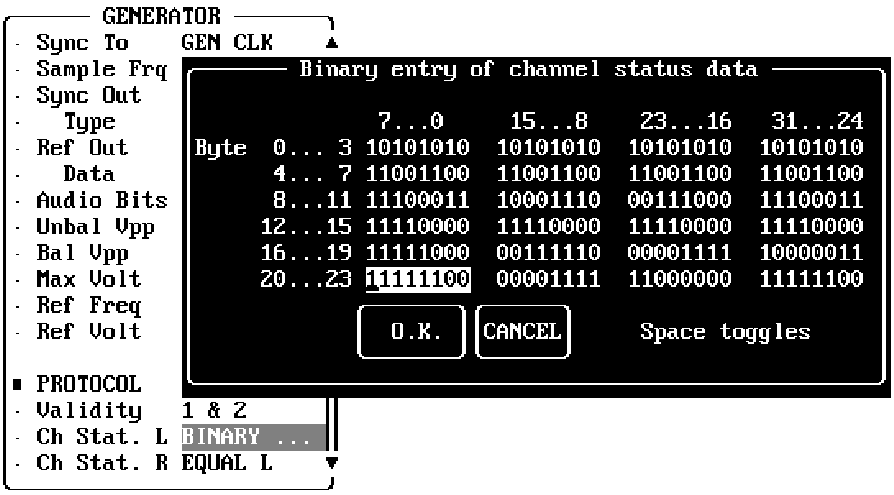


Fig.: 6-11

Using the menu as shown above, the channel status data can be entered in binary format. The key SELECT (respectively the space bar on the external keyboard) switches the individual bit from 0 to 1 or vice versa.
Application: Generation of bit patterns to be compared with the pattern at the output of the device under test, as they are displayed by the protocol analysis in binary mode.

Display of Channel Status Data

Bit sequences are displayed
- in binary format,
- in Hex format or
- as ASCII characters

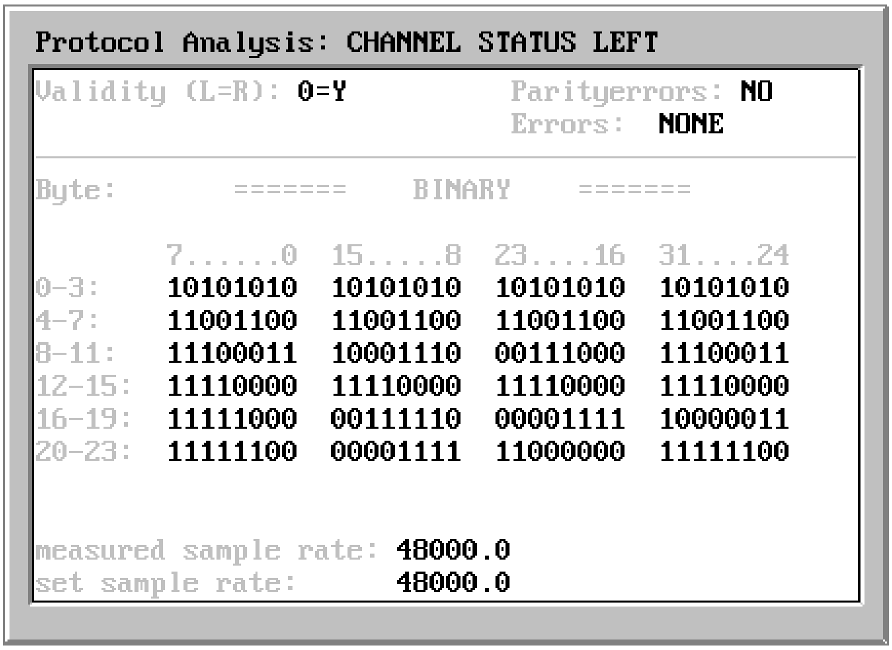


Fig.: 6-12

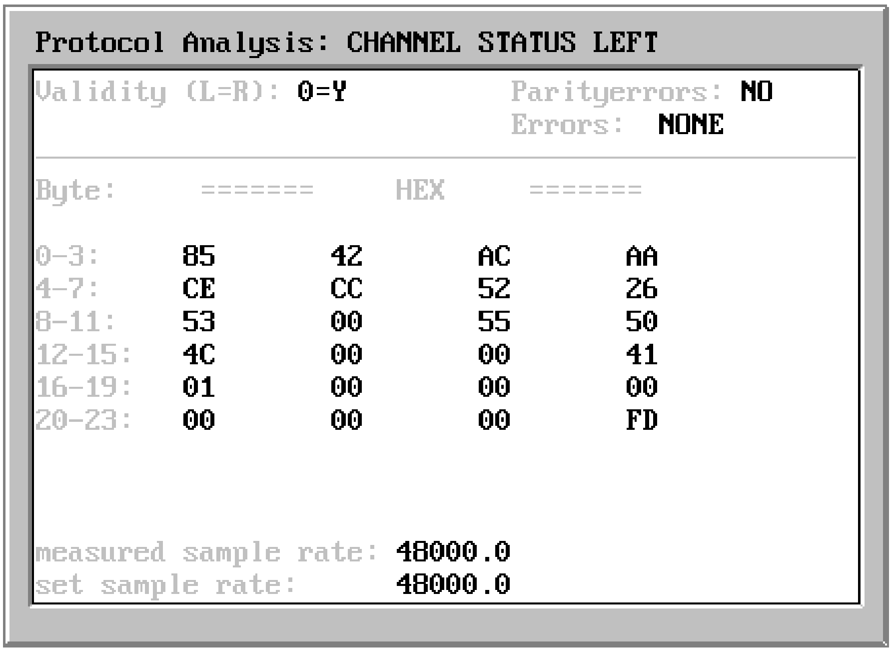


Fig.: 6-13

Display of Information

Displaying the interpreted information
- according to AES 3,
- according to IEC 958
- or to be edited by the user

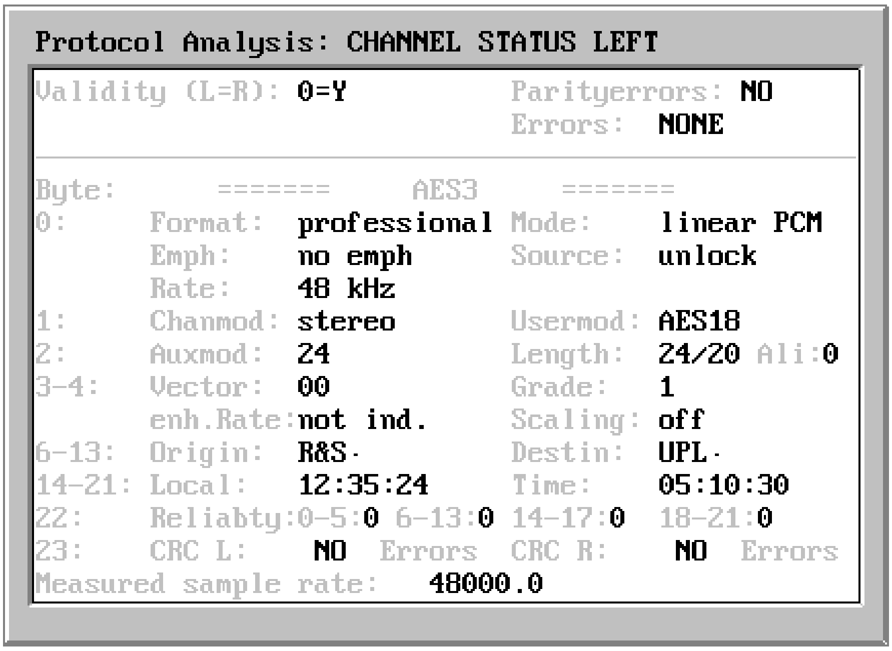


Fig.: 6-14

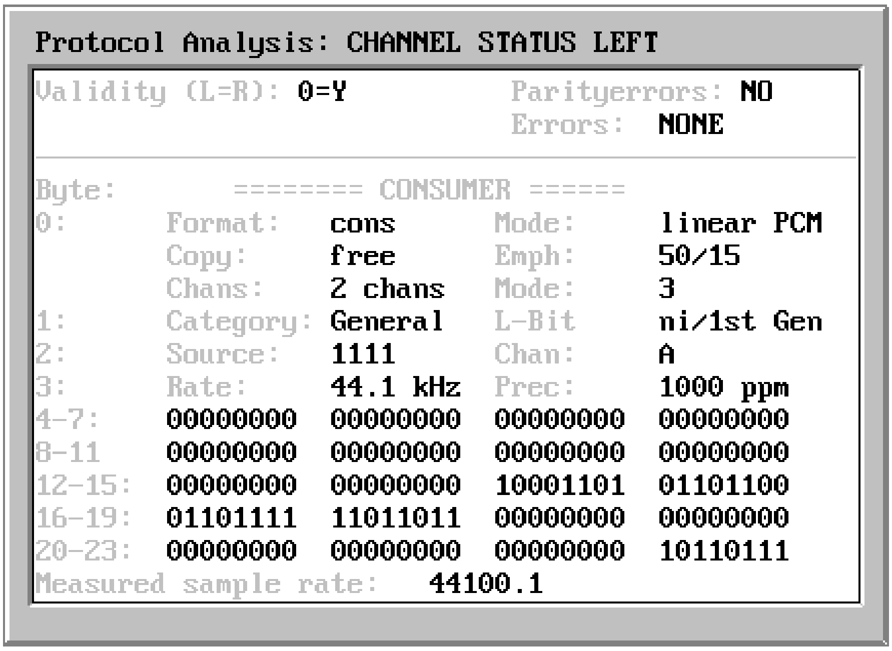


Fig.: 6-15

Digital Amplitude Units

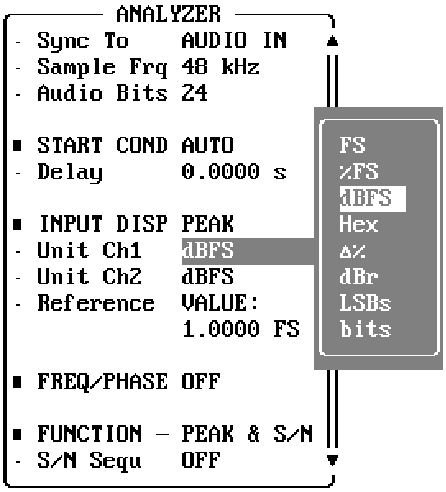


Fig.: 7-1

FS:	full scale sinusoidal signal = 1 FS at RMS measurement full scale sinusoidal signal = 1 FS at Peak measurement
%FS:	conversion formula: $V_{[FS]} * 100$
dBFS:	conversion formula: $20 * \lg (V_{[FS]})$
Hex:	conversion formula: $V_{[FS]} * 65535$ the measured full-scale value is displayed as a 6-digit hexadecimal number (6 digits = 24 bits = 23 bit mantissa + 1 sign bit)
Δ%:	conversion formula: $(V_{[FS]}/V_{Ref}-1) * 100$
dBr:	conversion formula: $20 * \lg (V_{[FS]}/V_{Ref})$
LSB:	conversion formula: $V_{[FS]} * 2^{(\text{audio bits} - 1)}$ depending on number of audio bits
bits:	conversion formula: $\mathrm{ld} (V_{[FS]} * 2^{(\text{audiobits} - 1)} + 1)$
UI:	one UI is specified to be the smallest distance between two slopes of the digital signal and corresponds to half of the clock period of the digital bi-phase clock; with audio signals one UI is 1/128 of the sample period.

Digital Interface Parameters

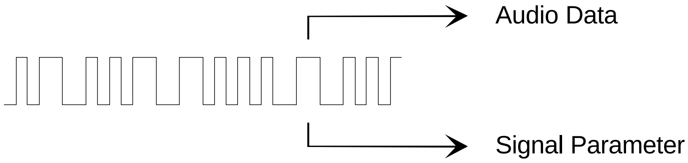


Fig.: 7-2

In digital audio interfaces there are two types of signal:

- the digitally coded audio signal
- the physical digital signal

The physical digital signal can be described with the same analog parameters as an analog signal. Normally, the peak-to-peak voltage and the sample frequency.

Digital audio signals are transmitted via balanced or unbalanced lines and are in principle subject to the same interference effects as an analog signal. Thus noise or other ac voltages can be coupled in, which influence the transmission quality. These super-imposed interferences are far less effective with balanced lines than with unbalanced lines, but there they can also be effective in case of insufficient common mode rejection.

The following interference effects have to be considered:

- cable attenuation — reduces the amplitude of the digital signal
- low-pass filter behaviour — of the transmission lines modifies the signal, the signal slopes are rounded off
- coupled-in interference voltage — e.g. hum or interference from switched power supplies modulate the signal and make decoding impossible at a certain level
- common mode voltages — on balanced lines
- shifting of signal slopes (jitter) — e.g. via line reflections, lead to distortions of the source signal during A/D or D/A conversion or make the correct regeneration of the bit pattern in the next input stage impossible
- inadequate synchronization — during connection of different digitally coded signals can lead to doubled or omitted frames

Pulse Amplitude, Sampling Rate, Common Mode Interference

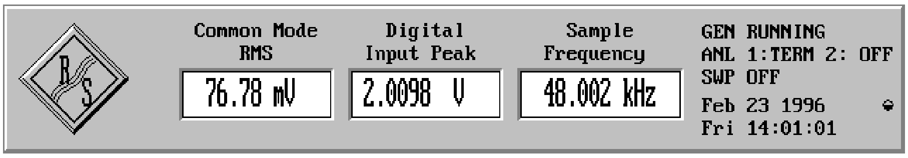


Fig.: 7-3

With the Audio Analyzer UPL and the options UPL-B2/B29 plus UPL-B22 all important measurements for the analysis of digital interface parameters can be performed and the necessary test signals can be generated for this purpose.

Analyzer:

- measurement of pulse amplitude: level range 0 to 10 V; also related to the generator output signal to measure the attenuation of the cable
- measurement of sample frequency: 35 to 55 kHz (UPL-B2), up to 106 kHz (-B29)
- measurement of time difference: between output and input signal to determine time delay of digital audio components
- measurement of phase: between audio input and reference input
- common mode test: amplitude, frequency, phase, spectrum, waveform
- jitter amplitude
- jitter spectrum

Generator:

- pulse amplitude: 0 to 8 V (balanced, matched) 0 to 2 V (unbalanced, matched)
- sample frequency: 35 to 55/106 kHz, adjustable
- common mode signals: frequency 20 Hz to 21.75 kHz; up to 110 kHz with option UPL-B1; amplitude 0 to 20 V
- cable simulation: simulates a cable length of about 100 m
- jitter signals: sine, noise, frequency 10 Hz to 21.75 kHz (to 110 kHz with UPL-B1); amplitude 0 to 5 UI
- input signals with jitter: can be output jitter-free (reclock)

Jitter

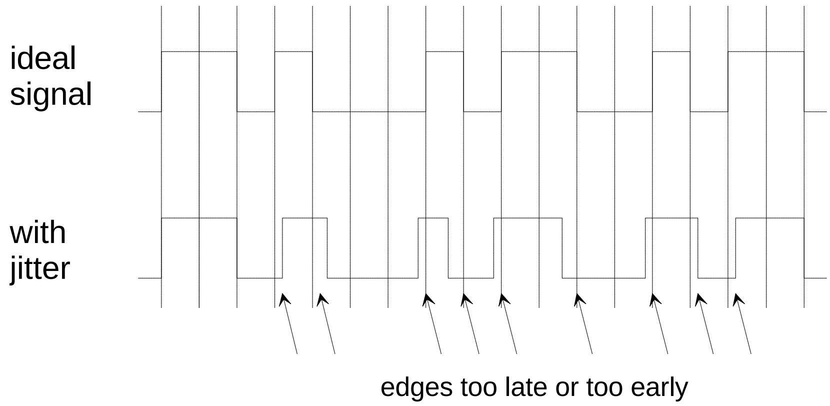


Fig.: 7-4

Jitter is the short-term variation of the impulse slopes of a digital signal around the optimum equidistant points of time. The temporary deviation of the real zero crossings referring to their set-point time is considered.

Causes for Jitter

Jitter can be divided up into the following criteria:

- **Random Jitter** is generated by:
  - thermal noise and shot noise of the semiconductor circuits
  - interference from control signals
  - impulse noise
  - noise on balanced cables
  - phase noise caused by frequency limiting components

- **Pattern-Dependent Jitter** is generated by:
  - spectral density of data pattern
  - bandwidth of transmission path
  - interference of data pattern
  - crosstalk and reflections
  - unstable duty cycle of data pattern

The term pattern-dependent jitter is a special kind of jitter which is generated as a reaction to the distortion of certain patterns in the transmitted sequence of signals, see fig. 7-5. By means of low-pass filtering of the signal, e.g. with a cable, the signal slopes are rounded off. Thus the zero crossings are displaced. At the comparator of the following clock regeneration circuitry the slopes are recovered with a time delay dependent on the bit sequence, so jitter is generated.

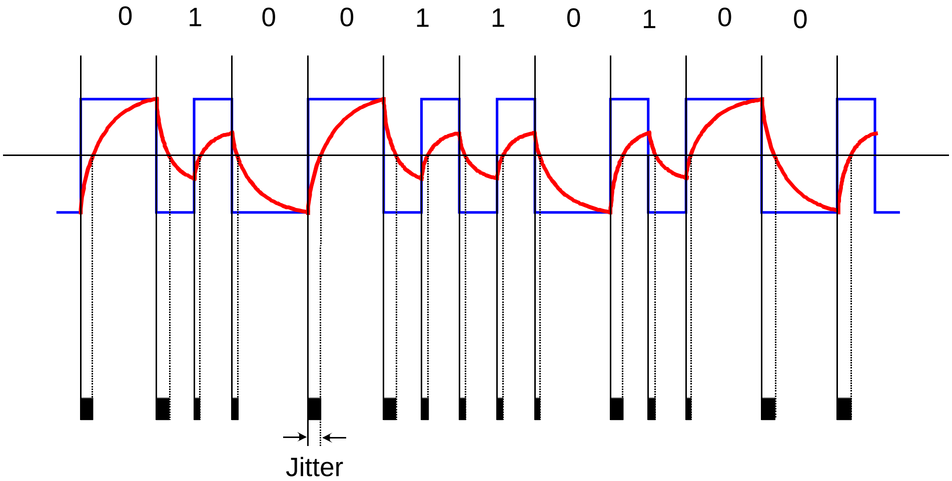


Fig.: 7-5

Reference of Jitter Measurements

If results of different jitter values are compared, it is very important which reference the jitter measurement is referred to.
We distinguish between three methods of measurement:

- **Timing Jitter:**
  The jitter measurement is performed at an ideal, non-jittered reference clock.

- **Alignment Jitter:**
  The ideal reference clock is not available and the measurement signal to be jittered is referred to a clock, which may jitter itself.

- **Sample-to-Sample Jitter:**
  In this method the time distance between two signal slopes is measured continuously and the distance of the previous measurement is subtracted. Thus the modification of the jitter is measured via time referred to the signal itself.

Sampling Jitter - Data Jitter

Each instrument of a transmitting chain generates a jitter itself. This **intrinsic jitter** is determined by the phase noise of the clock generation/clock synchronization.
If this intrinsic jitter is to be measured, please consider that a jitter is also generated via bit patterns (pattern-dependent jitter). For this reason the audio signal has to be "tuned out" during measurement of the **"Sampling Jitter."** This is done by synchronizing the measurement instrument to the wordclock signal.
When measuring **"Data Jitter"** we synchronize to biphase clock, i.e. the time deviation of each data bit is determined.

Jitter Amplitude

The **jitter amplitude** indicates the deviation of the signal slope concerning the ideal point of time.

This indication can be done as follows:

- **Phase Difference** D j
  If jitter is considered to be a phase modulation, the jitter amplitude can be indicated as a peak-to-peak value of the phase deviation $\Delta\varphi$ (in degree or rad).

- **Time Difference** D**t**
  If jitter is regarded as time deviation from the ideal zero crossing, the jitter amplitude is indicated as maximum single deviation Δt_{pos} respectively Δt_{neg} or as peak-to-peak value of the jitter in nanoseconds.

- **Unit Interval UI**
  Meanwhile the jitter amplitude is mostly indicated at the unit UI (Unit Interval). A Unit Interval is the smallest temporary set-point distance between two signal slopes and corresponds to half a clock period by means of which the digital signal is clocked. Thus the indication in UI is independent from the sample rate and allows to compare jitter values from audio signals of different sample rates, or determination of jitter random values without indicating the sample frequency each time.
  For 48 kHz-AES/EBU signals, 1 UI corresponds to 163 ns

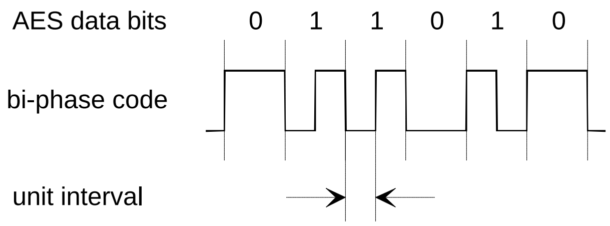


Fig.: 7-6

Eye Pattern

A simple method for jitter presentation is the **eye pattern**. On an oscilloscope the time sequences of the digital data stream are superimposed. A phase-locked, stable reference clock with the bit frequency provides for triggering of the oscilloscope when displaying the absolute jitter. The eye opening is made by the different sequence of high/low states and the rounding off of signal slopes due to low-pass filtering of the transmission line.

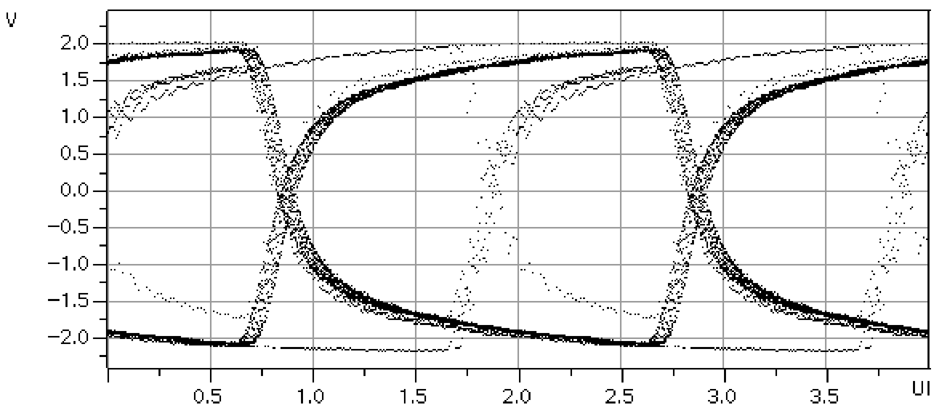


Fig.: 7-7

The vertical eye opening shows the level difference between High and Low level. If the vertical eye opening is small, even little amplitude interferences cause errors when evaluating bit states.

The horizontal eye opening is limited by the jitter, the jitter amplitude (peak-to-peak value) can be derived from the moving of the edges. This horizontal eye opening is responsible for the correct synchronization of the following input stage.

The eye pattern is mostly used for fast overview measurements. Therefore, more than basic information about jitter characteristics is not possible, because the unclear lines can hardly be used for precise evaluation, especially with noisy jitters.

Jitter Frequency and Jitter Spectrum

The **jitter frequency** is the frequency by means of which the data signal is modulated in phase. According to the origin of the jitter it can be a sine-wave, a non-systematic jitter is stochastic, it has continuous spectrum characteristics.

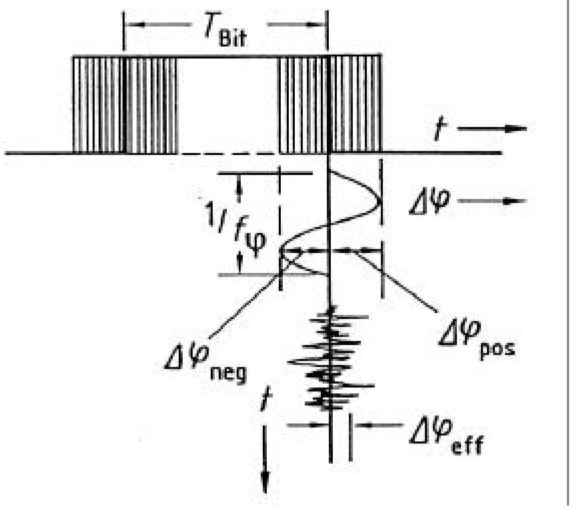


Fig.: 7-8

Checking the jitter spectrum helps to find the causes of jitters and to eliminate them if necessary. Thus complex jitter signals can be determined in quality and quantity. If there are several jitter sources, e.g. due to jitter accumulation along a digital transmission line, they might possibly be differentiated by means of the spectrum.

Jitter Transfer Function

In a transmission line, jitter increases with each audio component. Else every cable causes further jitter due to bandwidth limitation of the digital signal. Reclocking circuits help to resynchronize the digital signal by means of PLL and thus reduce the jitter. But ideal reclocking circuits cannot be realised, so a certain amount of jitter still exists at the output of the PLL circuits.

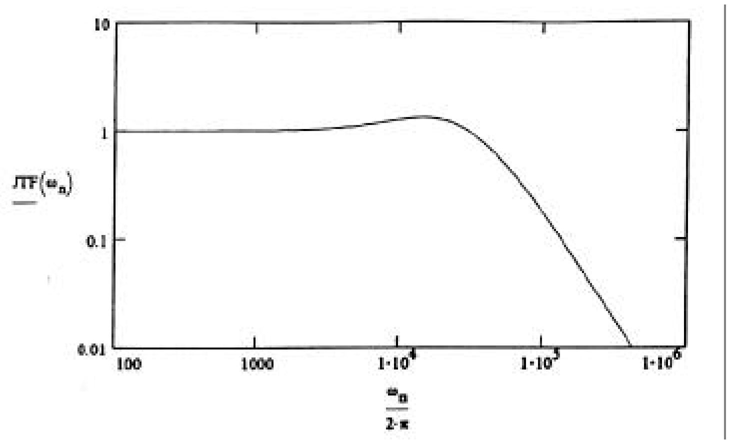


Fig.: 7-9

The **jitter transfer function** is defined as relation of the output jitter spectrum to the input jitter spectrum. It shows the jitter attenuation or the jitter amplifying effect of each part in the transmission line.
Fig. 7-9 shows a typical example. At frequencies below the PLL cut-off frequency the output signal follows the input signal, jitter remains unchanged. At higher frequencies an increasing attenuation starts with increasing jitter frequency. The output jitter is basically determined by the jitter of the reclocking circuit. But in this example even an amplified jitter might also occur. Components of similar behaviour in series might therefore clearly increase the problem of jitter accumulation.

Jitter Effects and Jitter Susceptibility

Jitter might have two different effects in the different parts of a transmission:

- synchronisation interferences
- increase of distortion

As it happens in other applications too, a decrease in audio quality will occur if certain limits are overdriven. The jitter susceptibility as well as the jitter effects will be described in more details in the following.

Synchronisation Interferences

When switching together digital audio instruments the system clock has to be regenerated in each component. If the data jitter is so strong, that the single data bits cannot be identified anymore, this leads to a drastical increase of the bit error rate. The extreme case would be a failure of synchronization, the connection is interrupted.

To avoid synchronisation errors - especially important in cascaded transmission lines - jitter must not exceed certain limits. For sound studio equipment the limit ranges between 50 to 150 ns (0,3 to 0,9 UI) depending from the spectral composition of the jitter.

The AES3-1992 defines the jitter limit values for the digital signal transmission as follows: The maximum one-sided deviation of the signal slopes of the ideal 48 kHz clock must not exceed ± 20 ns. This corresponds to a peak-to-peak value of 40 ns or 0.25 UI. The deviation is measured at the 50% points of the signal level. Limit values for shape or frequency of the jitter are not defined for this purpose.

Jitter on transmission lines can be measured by modern audio analyzers. Nevertheless, the importance of these measurements has decreased, because more and more reclocking circuitries are used on the inputs of each single unit within a transmission line.

Increase of Distortions

During each sampling (A/D conversion) and reconstruction (D/A conversion) of the analog audio signal the deviation of the temporary position of the samples because of the jittered clock leads to distortions of the original signal. This is shown in fig. 7-10, using the sampling process as an example it will be explained in more details.

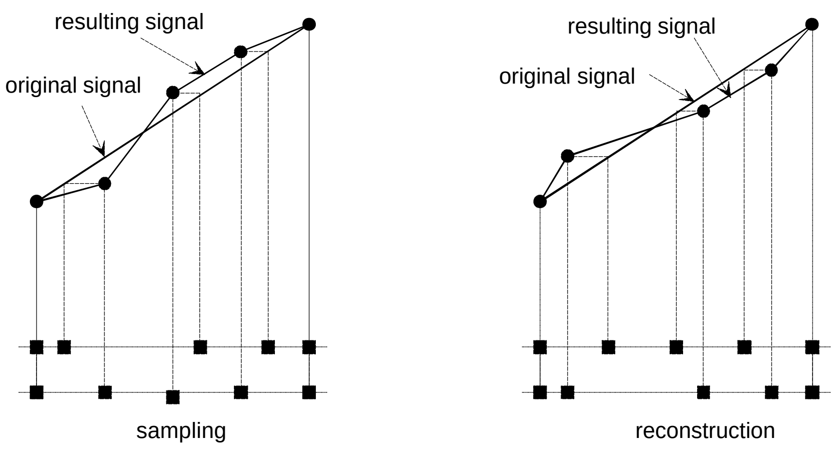


Fig.: 7-10

We will start with a jittered sample clock of the ADC. Therefore the single samples will not be taken on equidistant time points, sometimes they are taken too early (sample no. 2) or too late (samples no. 3 and 4) from the original signal. In a later part of the audio transmission, these samples will be used to reconstruct the audio signal, but now with a different time frame of the clock (in our example at ideal equidistant time points). The resulting signal is different from the original one.
In the other case of D/A conversion similar effects happen as shown in the right hand part of figure 7-10.

This decrease in the audio quality can be detected by the means of total harmonic distortion measurements. AES 17 therefore specifies the so called "Jitter Susceptibility Measurement". In case of a D/A conversion, the clock of the audio signal will be jittered, the resulting distortions will be measured using THD+N measurement (figure 7-11).
Using A/D conversion, an analog test signal is needed while the clock of the ADC will be jittered. The distortions have to be measured on the digital output.

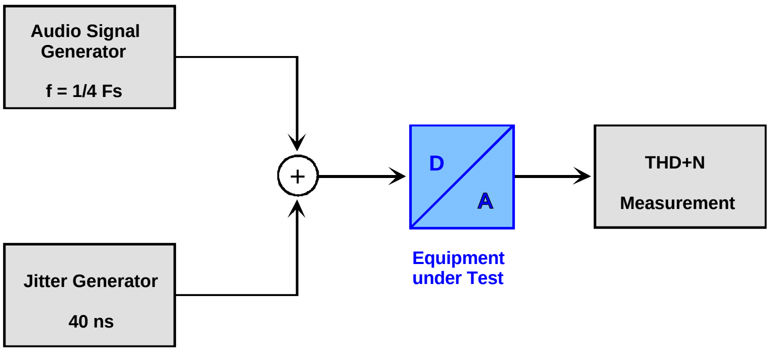


Fig.: 7-11

For estimating the jitter limit value the following questions arise:
To which extent does the sampling time have to be shifted to generate an error of one quantization step?

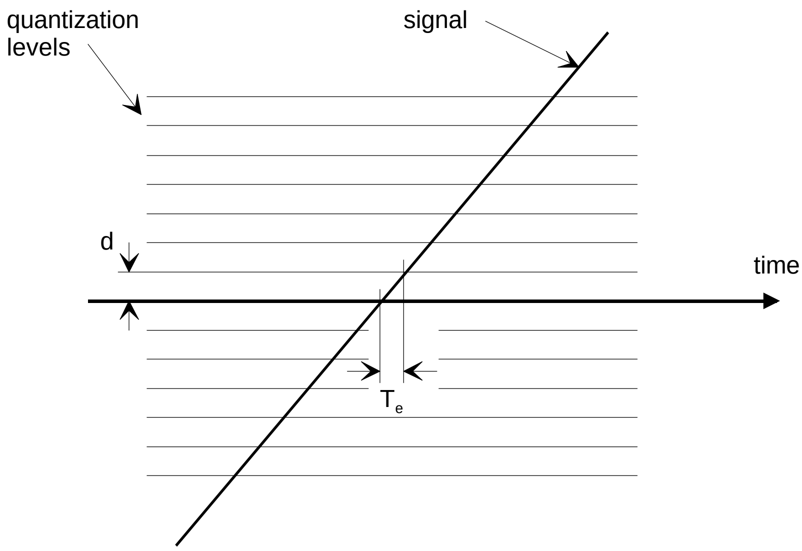


Fig.: 7-12

Fig. 7-12 shows the zero crossing and the corresponding quantization steps. If the sine signal has a frequency of 20 kHz and a maximum level (worst case), the time error T_e for a 16 bit system can be calculated as shown below:

$$T_e = 2^{1-N} / w = 243ps$$

If the digital signal is resynchronized, e.g. during A/D conversion the time error causes an amplitude error which superimpose the signal.
If this error can be neglected it has to be smaller than half a quantization interval. For the above example of a 16 bit quantization, i.e. < 121 ps. For a 20 bit system the value must be smaller by the factor of 24, i.e. < 7.6 ps!

Taking a simple example of a sine-wave jitter modulation, distortions occur in the analog domain in form of side bands whose amplitude is proportional to the jitter amplitude and the frequency distance of the tone signal corresponds to the jitter frequency. An example is shown in fig. 7-13.

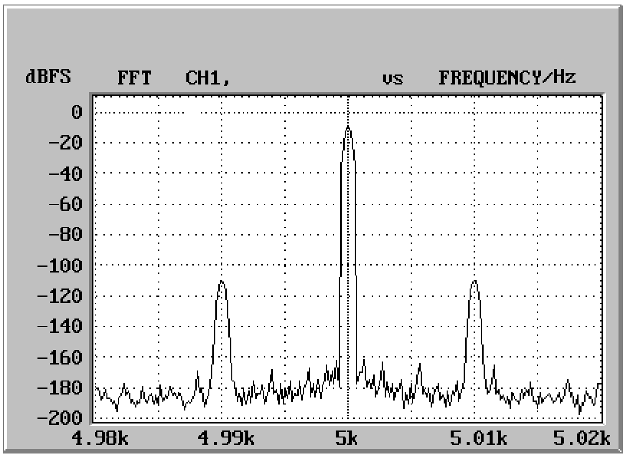


Fig.: 7-13

Jitter frequencies of the word clock in the range of 1 kHz to 10 kHz can be clearly heard in the analog signal since they mostly represent disharmonic components to the music signal. Low frequented jitter components generate signal parts very close to the stimulating frequency and cannot be noticed by the human ear due to masking effects.

Literature:
- Mäusl, Rudolf / Schlagheck, Erhard: "Meßverfahren der Nachrichtenübertragungstechnik", Hüthig Buch Verlag1991
- Dittel, Volker / Ladwig, Peter: "Wie analog ist die digitale Welt?", Bericht zur 18. Tonmeistertagung, Verlag K.G. Saur, 1995
- Dunn, Julian / Dennis, Ian: "The Diagnosis and Solution of Jitter-Related Problems in Digital Audio Systems", AES Preprint No. 3868, Audio Engineering Society
- Sieber, Hartmut: "Ausgewählte Verfahren und deren Anwendung zur Ermittlung von Jitterkenngrößen", FHS Düsseldorf

Jitter Measurements using Audio Analyzer UPL

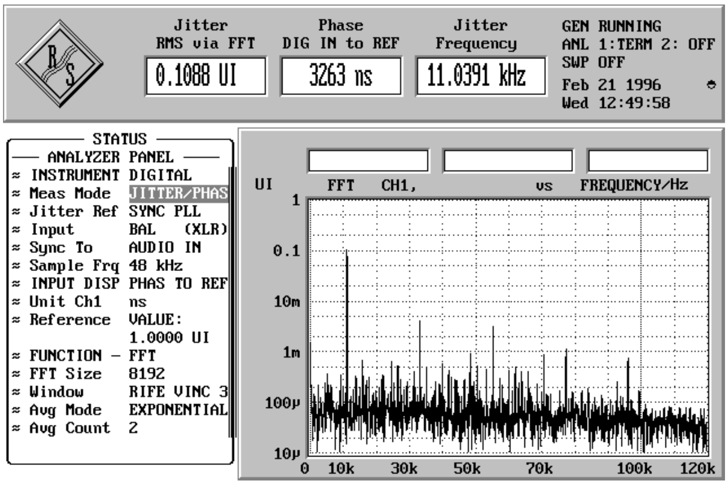


Fig.: 7-14

Using the Audio Analyzer UPL and the options UPL-B2/B29 plus UPL-B22, jitter can be measured with amplitude and frequency, spectrum and waveform can be displayed.

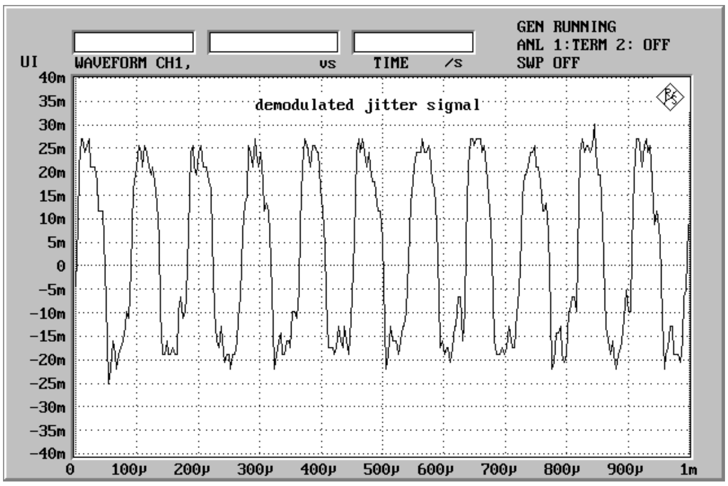


Fig.: 7-15

For the test of input circuitries, the clock of the generator signal can be superimposed by jitter of variable amplitude. Sine and noise function of jitter are particularly suited for the practise. An input signal with jitter can be output jitter-free.

Synchronization of Digital Audio Components

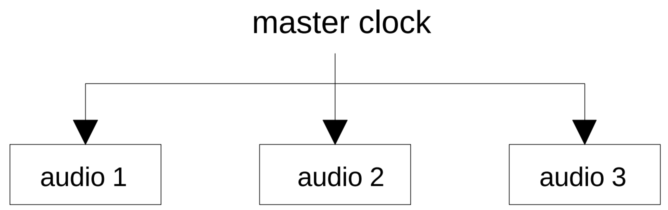


Fig.: 7-16

If several, digitally-coded signals have to be interconnected, i.e. at a mixing console or a multi-channel tape machine, the synchronization is of special importance. Though audio signals using AES/EBU format or S/P DIF format are self-synchronizing, the clock frequencies of the several sources are about to float because of the tolerances of the crystals. Associated frames, which contain the respective instantaneous values (samples) of the left and right channels, do not arrive at the same time. This can be noticed as omission or doubling of individual samples. This effect can be avoided by distributing a master clock to all audio components for synchronization, whenever different digital audio signals have to be interconnected.

Another problem occurs by the different delay times of the individual components because of different processing times. Or, in other words, the time delay between the synchronization clock and the audio output signal. This shift is specified for each individual component as phase between the synchronization input and the audio signal output. According to AES3, it must not exceed one forth of the frame length (= 32 UI).

To synchronize the individual components in a sound studio, different signals are used:

- DARS = "Digital Audio Reference Signal" according to AES11; it is an AES/EBU signal, where all data bits are set to zero
- word clock rectangular signal of sample frequency
- bi-phase clock rectangular signal of 128 times the sample frequency
- video sync signals using video frequency of 50 Hz in Europe (PAL, SECAM) using video frequency of 60 Hz in USA (NTSC)
- 1024 kHz reference signal used in studios for synchronization to video signals

In professional sound studios, the DARS signal is used more and more (balanced lines, XLR connectors);
in video applications video sync signals and reference signals of 1024 kHz can be found very often (unbalanced lines, BNC connectors).

Synchronization of Audio Analyzer UPL

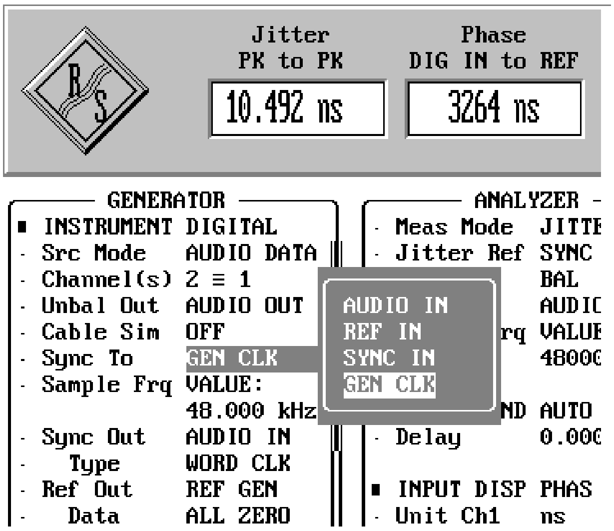


Fig.: 7-17

For measurement purposes, generator and analyzer have also to be synchronized to external master clocks; on the other hand it is also possible, that the measuring instrument distributes the synchronization signal, and thus works as the "master".

The Audio Analyzer UPL can do both. The necessary interfaces and all the signals are described in details in chapter 2 "Generator" and chapter 3 "Analyzer".

In addition, the Audio Analyzer UPL can measure phase between its reference input and its audio signal input, this gives the time delay between the synchronization input and the audio signal output of the device under test.
Measurement of delay time through the device under test is possible too.

Effects of Digital Signal Processing

Digital Data Formats:

Currently two number schemes can be found: two's complement and binary offset. The difference between both is only the polarity of the MSB.
UPL supports only the two's complement, which is the mostly used format in the audio range.

Subharmonics:

Signals, which have an integer frequency ratio to the sampling rate, are sampled and quantizised always at the same points of the curve. Therefore the quantization error correlates with the signal and gets "harmonic".
The noise floor mutates to discrete harmonic lines.

Help:

- Function "Freq Offset" in the Generator panel changes the generator frequency by 0.1%.
- to use 997 Hz signals instead of 1 kHz (as recommended in AES17)

Peak Rectification:

In analog signal processing the peak value can be easily detected by using diodes. Therefore in digital systems the largest sample is taken, but this might not correspond to the true peak value.

Example:

- The use of the INPUT PEAK function, the peak detection of a signal with 16 kHz at sampling rate of 48 kHz shows an instable Input-Peak display.
- The use of the measurement function PEAK & S/N gives a much more stable result, because the instrument interpolates to the true peak value.

Quantization Noise

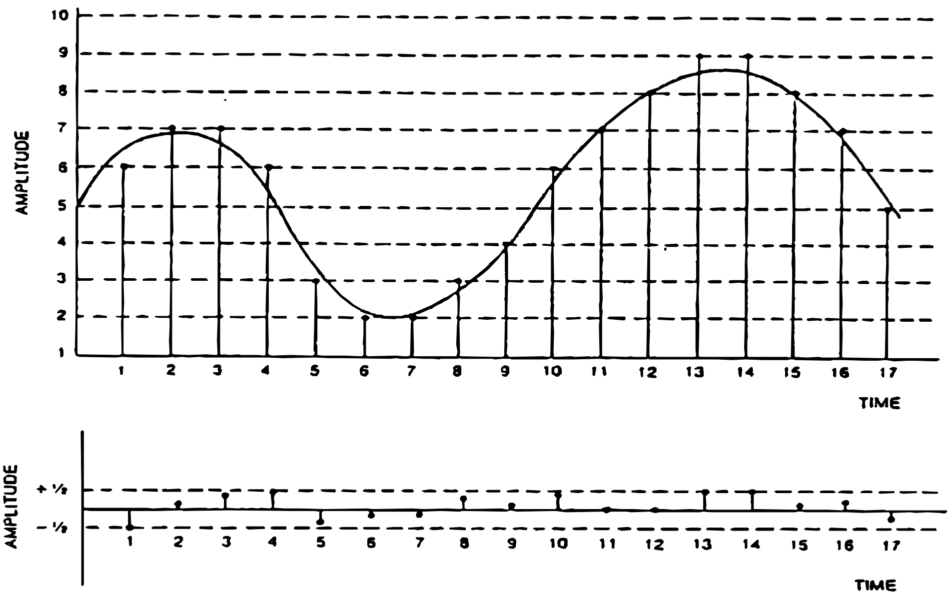
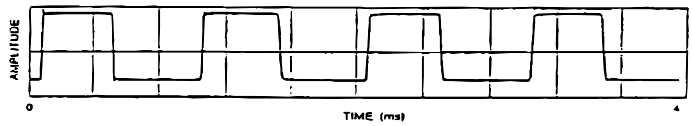


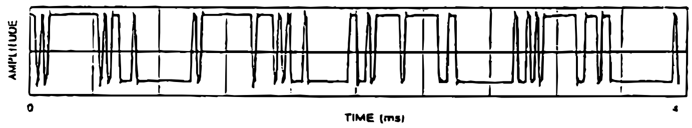
Fig.: 7-18

- when converting from analog to digital, the continuous analog signal is mapped to a discrete digital signal, using a finite number of quantization steps
- level errors are caused by the amplitude quantization, which appear as additional quantization noise
- in an ideal system the quantization error is between 0.5 and -0.5 quantization intervals
- each amplitude value occurs with the same probability = rectangular probability density function (PDF)
- quantization noise is white noise

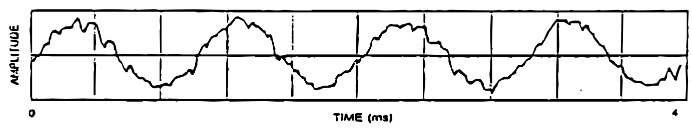
Dither



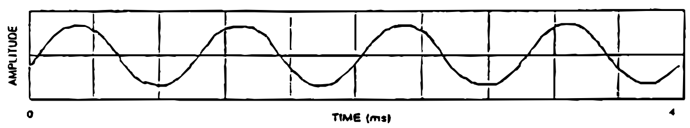
(A) A 1 kHz sine wave with amplitude of one-half LSB is quantized to produce a square wave.



(B) Dither of one-third LSB rms amplitude has been added to the sine wave before quantization.



(C) Modulation carries the encoded sine wave information, as can be seen after 32 averagings.



(D) Modulation carries the encoded sine wave information, as can be seen after 960 averagings

Fig.: 7-19

- dither is a noise-like signal of low level (usually at about 1 LSB), that is added to the analog signal prior to quantization

- example: a very low sinusoidal signal (amplitude 1 LSB) is applied to the converter, at the output it is converted to a rectangular signal; by adding dither, the rectangular signal looks like a pulse-width modulated signal, which contains the sinusoidal signal. Because the noise components are not correlated to the signal, the original signal can be reconstructed by means of averaging

- dither is used with different PDF,
  in the case of rectangular PDF optimum dither amplitude is 1 LSB

- dither generation with Audio Analyzer UPL:
  - dither amplitude
  - probability density function (PDF): Gaussian, Triangle, Equivalent distribution

DISPLAY Panel

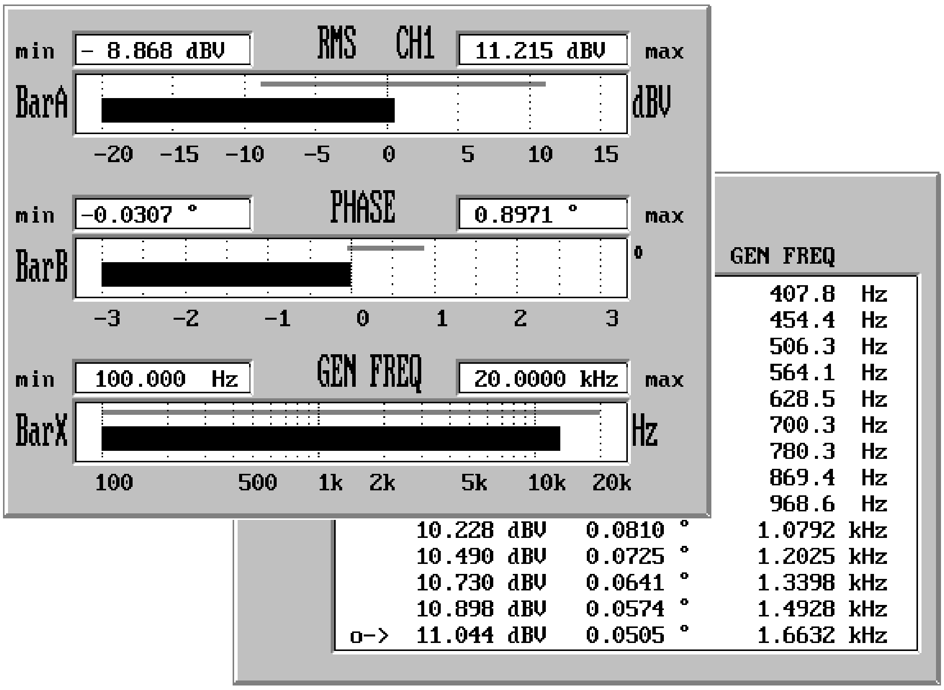


Fig.: 8-1

Selection of Graphical Data Presentation (OPERATION):

• CURVE PLOT:	line diagram, 2 measured functions can be displayed
• BARGRAPH	analog display in form of bars; 3 bars at the same time; extreme values marked by trailing pointers
• SPECTRUM	display of FFT frequency spectrum / transfer and coherence function; display of histogram with intermodulation measurements, THD measurements and third-octave analysis
• SWEEP LIST SPECTR LIST	results of a sweep respectively spectrum are listed
• SWP LIM REP SPC LIM REP	displays the values exceeding the tolerances
• PROTOCOL	protocol data of the digital interface are displayed

Line Diagram and Frequency Spectrum

- two dependent variables (TRACE A and TRACE B) can be displayed above an independent axis
- scaling of TRACE A and B can be different
- the scaling can be set manually or can be adapted automatically:
  - therefore the sweep or FFT start/stop values are adopted for the x axis
  - the y axis will be rescaled once by using the minimum and maximum values of the measured results
- overrange values are not displayed (the curve is interrupted)
- underrange values are indicated in the status line in the top right corner of the display
- measured values not fitting into the selected coordinate system are displayed at top or bottom, after rescaling they are displayed correctly
- with intermodulation and THD measurements histograms are displayed, the frequency axis being not true to scale and invariable
- after selecting the GRAPH Panel, softkeys are available to activate cursors and markers and to change the scaling
- to read the measured values of the curves there are two cursors which can be moved across the display using the rotary knob or the direction keys; they jump from value to value
- also, the cursors can be moved outside the coordinate system and indicate the appertaining values

Basic Settings of DISPLAY Panel

Mode: (with FFT spectrum only)

- DEL BEF WR — new trace will overwrite the previous trace
- MAX HOLD — keeps the maxima of the spectrum lines, average function has to be switched off
- WATERFALL — „spatial“ impression with more than one curve, can only be selected with 1-channel analyzer

Scan Count: (not available with spectrum display)

defines the number of traces to be measured or displayed:

- 1 — one trace respectively pair of traces, new trace will overwrite the previous one(s)
- >1 — several sweeps are recorded; the last 17 are stored internally and can be processed

User Label:

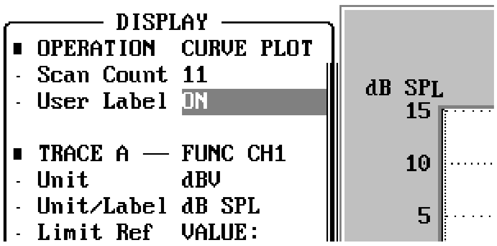


Fig.: 8-2

- the user can assign units labelling of his own,
  example: display of sound pressure level dBSPL for acoustical measurements

Parameters for Display of Traces

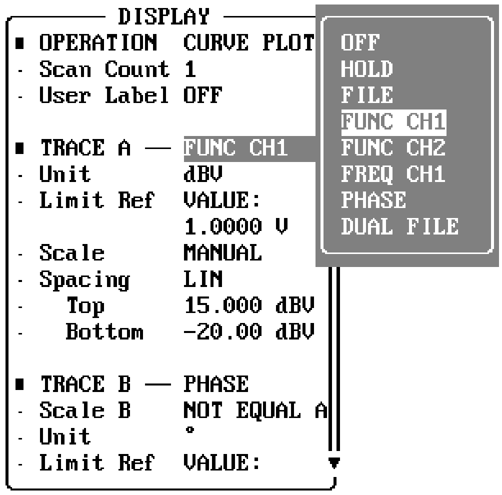


Fig.: 8-3

Possibilities to display Curves (TRACE A/B):

• OFF	no display
• FUNC CH1/2	results of the measurement of the function selected in ANALYZER panel are displayed
• FREQ CH1/2	results of frequency measurement are displayed
• PHASE	results of phase measurement are displayed
• INP RMS CH1/2	RMS measurement results of input function
• HOLD	continues to display the previous trace
• FILE	measurement results stored in a file can be recalled and displayed
• DUAL FILE	a pair of traces stored in a file can be recalled and displayed
• GROUP DELAY	group delay calculated from the phase measurement will be displayed

Reference of Traces

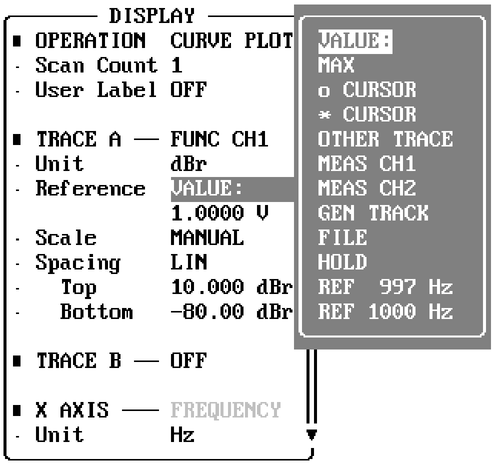


Fig.: 8-4

Reference used for Relative Units

can either be a single numerical value or a data record

• MAX	maximum value of the sweep is adopted once
• * CURSOR o CURSOR	the value on which the cursor is placed is adopted once
• VALUE	numeric entry of reference value
• FILE	reference trace is loaded from a file
• OTHER TRACE	trace data of the other curve serves as reference values; every new measured value is used immediately
• MEAS CH1/2	reference to the measured values of the indicated channel
• GEN TRACK	generator setting is used as reference
• HOLD	reference trace is not changed any more
• FILE INTERN	can be selected if a reference trace was stored in the file
• REF 1000Hz/997Hz	the value at the standard frequency is adopted

NORMALIZE Command

• VALUE	when using a reference unit (i.e. dBr), and also using a reference value, which is not fixed (i.e. OTHER TRACE), then an individual point of the curve can be defined as 0 dBr by moving the whole trace.
• * CURSOR o CURSOR	the value to normalize the trace is taken from the value on which the cursor is placed

Limit Check

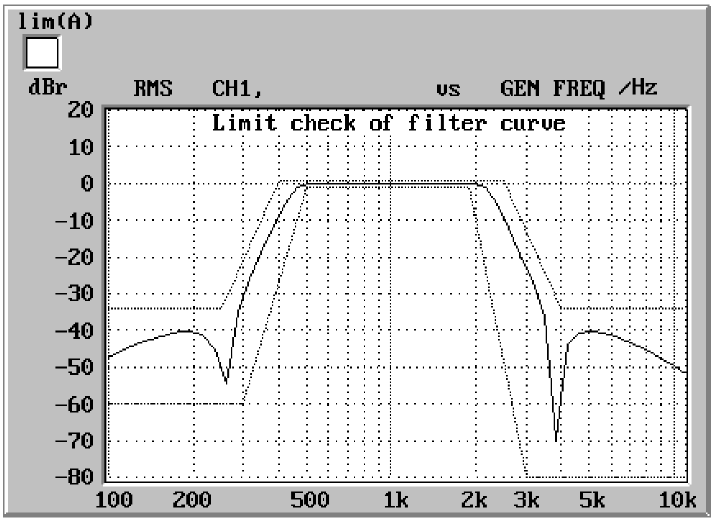


Fig.: 8-5

- a lower and an upper limit can be defined for the measurement (LIMIT CHECK)
- one or both curves commonly can be checked, but there is only one tolerance band
- instead of a fixed limit values also tolerance curves can be loaded from a file.
  These files contain x-y pairs, with the y-value being a factor which is multiplied with the set reference value from the display panel.
  application: fast go/nogo tests
- for frequency response measurements a macro (LIMIT.BAS) can be used to generate these limit curves as it is described in an application note;

Settings in the OPTIONS Panel

Switching the Measurement Display Off (Meas Disp):

- ON — measured value and status displays are switched on
- OFF — measured value and status displays are switched off, this increases speed for measurement routines; sweeps run about 15 % faster
- this function can be switched via an external keyboard using the key combination Ctrl D

Reading Rate of Measurement Results (Read Rate):

For better reading, the output speed can be reduced.

- MAX SPEED — maximum output speed
- 6/s — 6 measurement results / second
- 3/s — 3 measurement results / second
- 1/s — 1 measurement result / second

Resolution of Measurement Results (Read Resol):

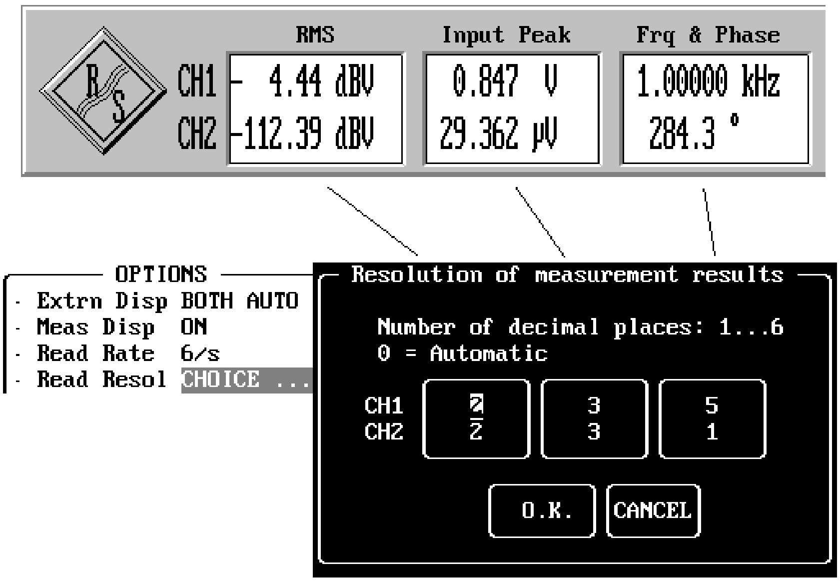


Fig.: 8-6

- A selection box appears in which the resolution of the measured value in trailing decimal places can be input for each measurement result window.

Graphics Display with Selectable Colours (TRACES COLOR/LINE):

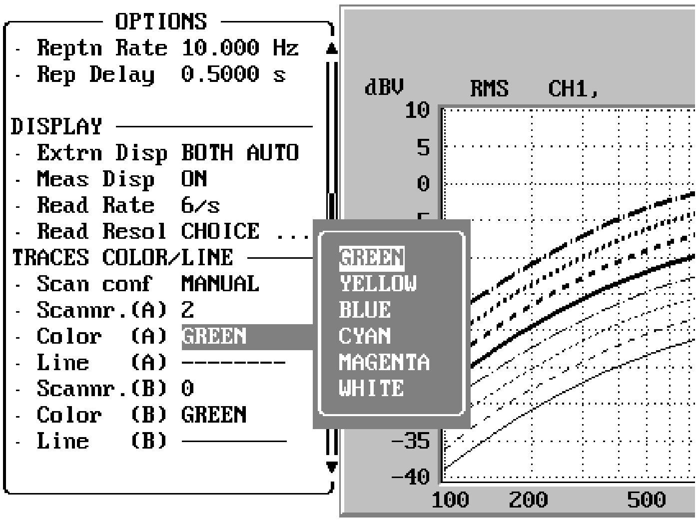


Fig.: 8-7

When individual traces or bars are displayed, a specific colour / shade of gray and line patterns can be assigned to each trace or bar for easy distinction.

- MANUAL — colour/shade of grey and line pattern can be assigned for each single trace
- DEFAULT — automatic assignment, trace A = green, trace B = yellow (corresponds to earlier firmware versions)
- AUTO COLOR — automatic assignment of colours in the sequence green, yellow, blue, cyan, magenta and white (trace A is shown in solid lines, trace B in dotted lines)
- AUTO LINE — automatic assignment of line patterns in the sequence ——— - - - - ········ and ·-·-·-·- (trace A is shown in green, trace B in yellow lines)

STATUS Panel

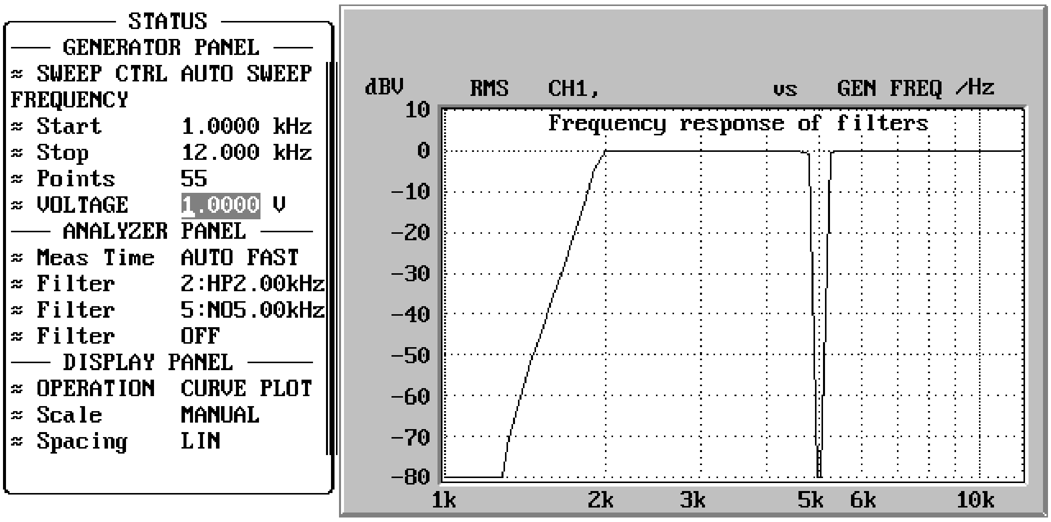


Fig.: 8-8

- instrument settings can be displayed together with graphical and numerical results
- any command line of all panels can be selected and combined in the status panel
- UPL can also be operated from this panel
- all important information can be printed on a single hardcopy

Data Documentation

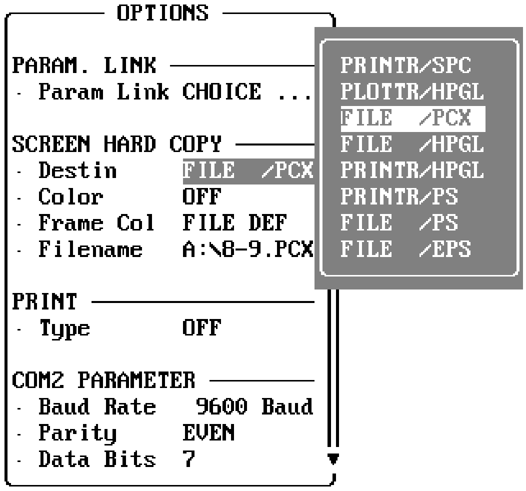


Fig.: 8-9

Basically:

- use the "H COPY" key to initiate
- output of the screen content, but without softkey line
- screen content displayed at the instant the key is pressed
  → if necessary stop measurement first!
- if only graphics is desired, use ▫↔☐ key

Different Types of Output:

1. data on printer using printer-specific format
2. data on printer using HPGL format
3. data on printer using PostScript format
4. outputs on plotter (HPGL format)
5. PCX format stored to file
6. HPGL data stored to file
7. stored to file in PostScript format
8. stored to file in encapsulated PostScript format

Printer Output is performed in the Background:

- printer output while UPL is performing further measurements
- the output can be accelerated by stopping the measured value output

Data on Printer (Printer-specific Format)

PRINTER

Selection of Printer (Printname):

- UPL offers 139 printer drivers
- all printer drivers are shown in the selection box of the OPTIONS panel

Frame Col:

Selecting the background colour of a GRAPH panel frame and the measurement result display

- WHITE — white
- FILE DEF — colour displayed as defined in the file C:\UPL\REF\PRN_BW.PLT resp. C:\UPL\REF\PRN_CL.PLT for black-and-white or colour printers

Resolution (Prn Resol):

- low / medium / high — to select different "quality" of print results; whether a resolution can be set and which one depends on the printer used

Comment:

- additional comment of max. 1500 characters
- comment is also stored when storing a COMPL SETUP, extension ".CTX"
- ON: — opens the comment window: to create comments, to edit comments and to execute screen hardcopy
- OFF: — H COPY key starts printing without comment

Left Mrgn:

- margin of a hard copy
- value range 0 to 80 characters
- not all printers support the positioning of the graphics printout!

Orientatn:

• LANDSCAPE:	horizontal format
• PORTRAIT:	vertical format

Scaling:

- for X and Y axes
- to adapt the printer resolution to the print format
- Note: when using non-integer multiples, pixels are suppressed or printed several times

Print Width / Print Height

- read only display to inform about the actual print format as it results from the scaling of the printer used

PRINT:

measured values or other block data are output to printer as numbers (in ASCII Code).

• TRACE A/B	prints the measured values selected for trace A or B
• TRACE A + B	prints both traces
• EQUALIZATN	prints the values of the equalization table
• DWELL VALUE	prints the values for the sweep dwell time
• LIM REPORT	prints only the values exceeding the limits
• LIM UPPER LIM LOWER	prints the upper/lower limit curve
• X AXIS Y AXIS	prints the values of the X/Y-axis

Output in PostScript Format on a Printer or to a File

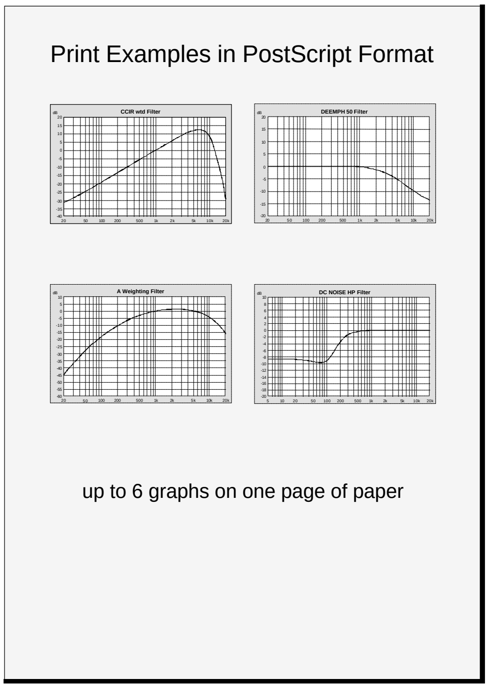


Fig.: 8-10

Color:

- ON — colours and line patterns will be printed as defined in the file C:\UPL\REF\PS.CFG; information is stored in colours in the file
- OFF — print of shades of grey as defined in the file C:\UPL\REF\PS.CFG; only black-and-white information is stored in file

Copy:

- SCREEN — the complete screen content is output
- CURVE/GRID — curves including scale and scale labels are output
- CURVE — only output of curves

Frame Col:

Selecting the background colour of a GRAPH panel frame and the measurement result display

- WHITE — white
- FILE DEF — colour displayed as defined in the file C:\UPL\REF\PRS.CFG

Comment:

- additional comment of up to 1500 characters possible (as described for standard printers)

Paper Size:

- DIN A4 or LETTER format selectable

Orientatn:

- LANDSCAPE: — horizontal format
- PORTRAIT: — vertical format

Selection of Interfaces (Plot on):

- COM 1/2 — connection to one of the two serial interfaces (RS 232); parameters of data transmission for transmitter and receiver must match each other, as a rule settable on plotter by wiping switches, on the UPL in the OPTIONS panel
- LPT1 — some plotters can also be operated using the parallel printer interface
- IEC BUS — output using IEEE-488 bus

Plots/Page:

- number of UPL plots to be positioned on a PostScript page

Outputs on Plotter

PLOTTER

- control characters in HPGL format, specifically defined for plotters
- also supported by some laser printers
- sequence for initialisation necessary (file UPL\REF)
- resolution being defined by the output unit instead of by the UPL
- size of the output defined by the configuration of the plotter

Color:

- ON relation of colours to colour pens
- OFF only one colour pen is used

Copy:

- SCREEN the complete screen content is output
- CURVE/GRID curves including scale and scale labels are output
- CURVE only output of curves

Selection of Interfaces (Plot on):

- COM 1/2 connection to one of the two serial interfaces (RS 232) parameters of data transmission for transmitter and receiver must match each other, as a rule settable on plotter by wiping switches, on the UPL in the OPTIONS panel
- LPT1 some plotters can also be operated using the parallel printer interface
- IEC BUS output using IEEE-488 bus

Storing to PCX Files

- PCX format was defined by the ZSoft company for PC Paintbrush
- accepted by most MS Windows programs

Filename:

- filename under which the PCX information is stored (for floppy disk use A:\)

Color:

- ON/OFF — PCX information is stored in black-and-white mode or in colours
- PCX file includes palette information, by the UPL it consists of 16 entries
  - using colours, the proportion of red, green and blue for the colour mode is specified in the file C:\UPL\REF\PCX_CO.PLT
  - for black-and-white printouts, black/grey/white shades are assigned in the file C:\UPL\REF\PCX_BW.PLT

Frame Col:

Selecting the background colour of a GRAPH panel frame and the measurement result display

- WHITE — white
- FILE DEF — colour displayed as defined in the file C:\UPL\REF\PRN_BW.PLT resp. C:\UPL\REF\PRN_CL.PLT for black-and-white or colour displays

Storing to HPGL File

- for processing with programs with graphic import capabilities
- result is not always optimal. A few programs ignore e.g. the text format instructions or do not properly present the colours. Therefore, programs are available for matching and printing purposes.

Universal Sequence Controller Software UPL-B10

Extends the measuring instrument to a automatic measuring system:

- realisation of additional measuring functions
- to control external instruments using any interface
- display of measured results in non-electric units, e.g. for acoustical applications
- complete application software to measure e.g. tape machines, CD players, etc.
- controlled by the sequence controller, external instruments can be remote-controlled via IEEE488 bus or RS-232 interface
- automatic tests in production lines

The Software Language:

A modified Rohde & Schwarz BASIC Language is included as a part of the UPL software and runs simultaneously. Other software languages would be to large to run under MS-DOS simultaneously with UPL software.

- automatic program generation
- up to 64 kByte BASIC program and 64 kByte BASIC data
- macros can be used to control the UPL and to shift data
- Rohde & Schwarz BASIC is a software tool to control external instruments very easily, no other complex software tools are necessary
- easy to edit, no compiler necessary
- commands are easy to understand, no expert knowledge necessary
- using the SHELL command a direct access to DOS is possible, e.g. to change the directory; the command EXIT leads back to BASIC again

Installation and Update:

- The Universal Sequence Controller software is activated by keying in an installation key number.
- The installation of the Universal Sequence Controller software has to be done only once, whenever installing a new firmware version of the UPL, also the Universal Sequence Controller software will be updated automatically.

Commands and Logging Mode

BASIC Extensions:

Rohde & Schwarz BASIC for the UPL is extended by the following commands:

- UPL OUT <string>
- UPL IN <string variable>
- UPL GTL [B | G | U]
- UPL BLOCKIN <array variable>
- UPL BLOCKOUT <array variable>

Syntax of the Commands:

The syntax of the Universal Sequence Controller software is basically the same as the one used for IEEE bus control (short form of the SCPI commands), except for a few differences:

- the suffix 1 must be omitted with the headers, but not with the parameters.
- optional key words (in square brackets in the manual) must not be used.

Logging Mode:

The function key F2 switches the logging mode on and off.
In the case of "on", all entries used for setting the UPL are appended to the BASIC program as command lines.
The generated command lines are always appended at the end of the BASIC program. They can be moved to any position in the program by changing the line number.
All settings of the UPL can be logged, except for the control keys "Start" to "REM/LOCAL" and the direction keys.

CONTROL Key Commands:

- START — UPL OUT"init:cont on" — or: UPL OUT"force start"
- SNGL — UPL OUT"init:cont off" — or: UPL OUT"force single"
- STOP/CONT — UPL OUT"abort" — or: UPL OUT"force stop"
or: UPL OUT"force cont"
- HCOPY — UPL OUT"hcop:wait"
- LCD ON/OFF — UPL OUT"disp:enab on" resp. off
- LOCAL — UPL GTL

Program Structure for UPL Generator

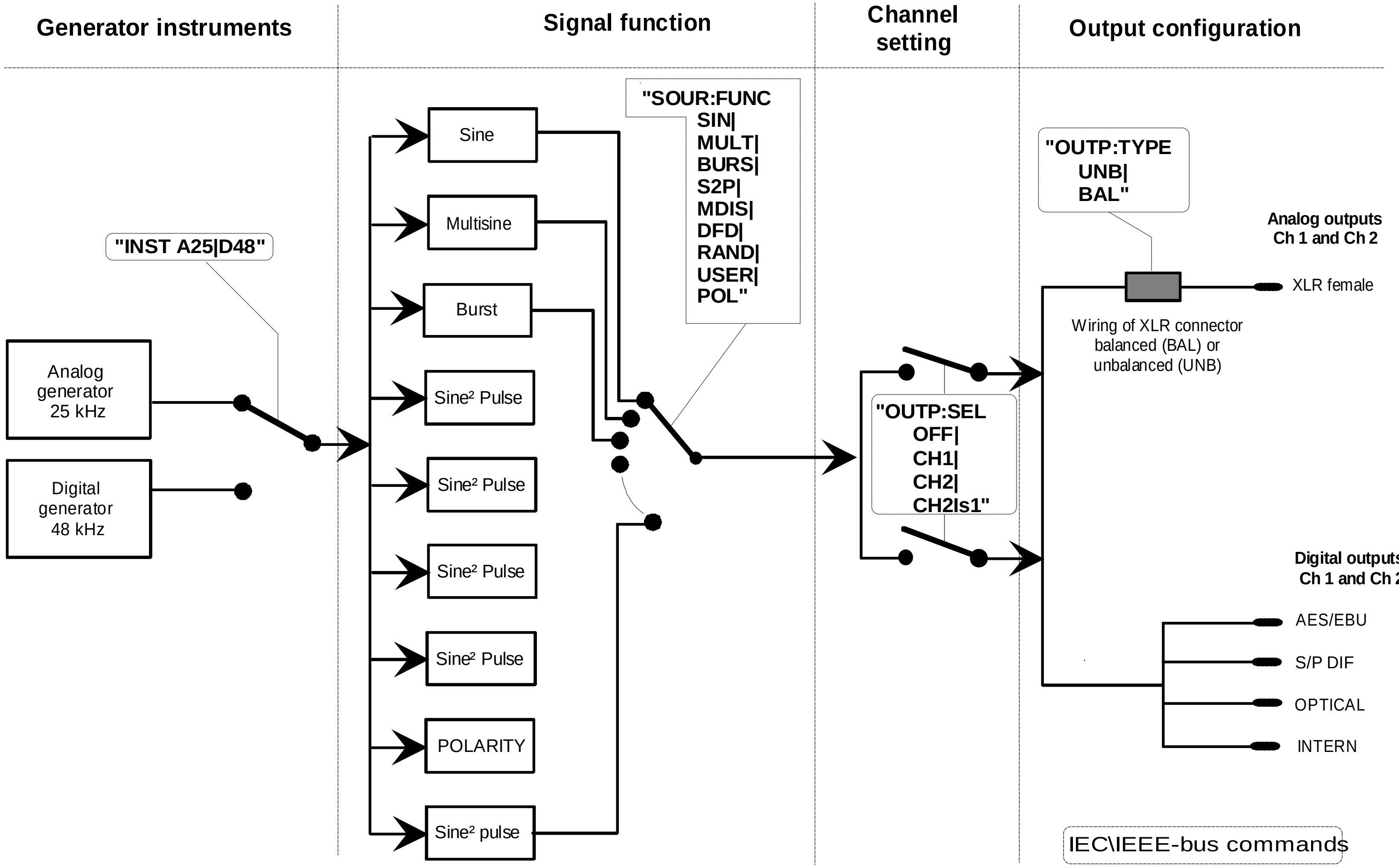


Fig.: 9-3

Program Structure for UPL Analyzer

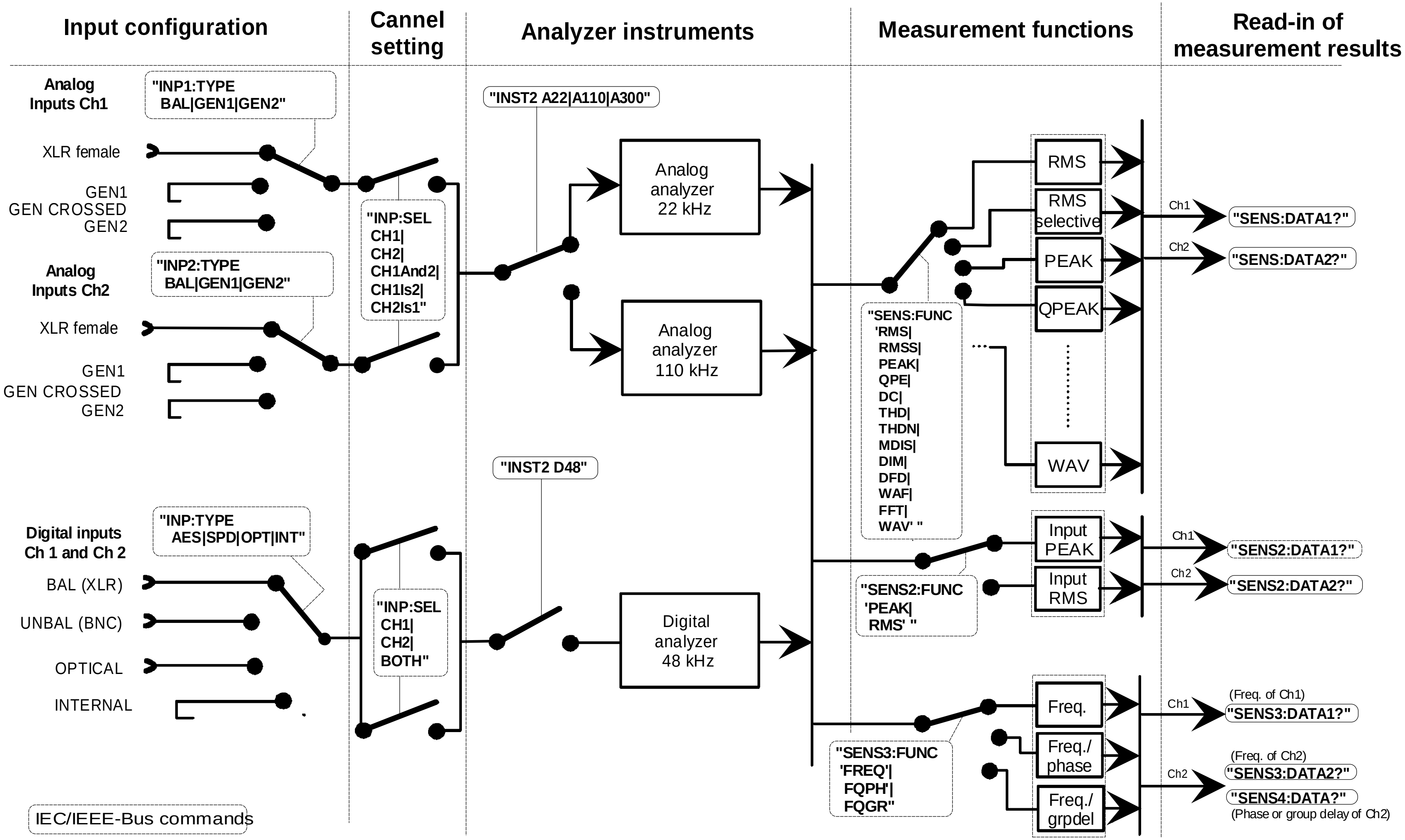


Fig.: 9-4

Selected Control Commands

Commands for reading in Single Measured Values:

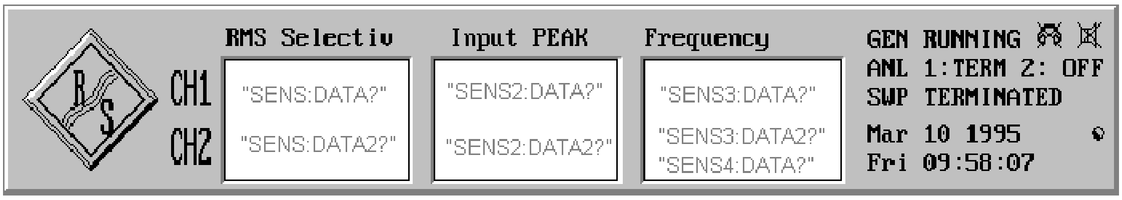


Fig.: 9-5

- Function CH1: UPL OUT "sens:data?":UPL IN A$
- Function CH2: UPL OUT "sens:data2?":UPL IN A$
- Input PEAK CH1: UPL OUT "sens2:data?":UPL IN A$
- Input PEAK CH2: UPL OUT "sens2:data2?":UPL IN A$
- Freq CH1: UPL OUT "sens3:data?":UPL IN A$
- Freq CH2: UPL OUT "sens3:data2?":UPL IN A$
- Phase: UPL OUT "sens4:data?":UPL IN A$

Commands for reading in the Block Data:

- Trace A: UPL OUT "trac? trac1":UPL BLOCKIN A(0)
- Trace B: UPL OUT "trac? trac2":UPL BLOCKIN B(0)
- X-values: UPL OUT "trac? list1":UPL BLOCKIN X(0)
- Z-values: UPL OUT "trac? list2":UPL BLOCKIN Z(0)

Commands for Synchronisation:

- UPL OUT "*WAI"

Command to trigger the Measurement and wait for the Measuring Time:

- UPL OUT "init;*WAI"

Transfer of Block Data

To transfer Lists of Measurements as Block Data:

As a matter of principle block data are stored in their basic units and also have to be transferred in their basic units!

- level values in Volt
- frequency values in Hertz
- distortion values in Percent

Thus a freely scaling is possible even if measured curves have been stored.

Measured curves using other units have to be calculated by the BASIC program.

Block data (traces) are transferred using the commands BLOCKOUT respectively BLOCKIN. They are always handled by the program as data fields, which have to be defined large enough before (max. length 1024 values).

Block data are always transferred via a buffer, what is to be read in must be indicated before by means of a query and then stored to the buffer.

Example:

```
UPL OUT"trac? trac1"
UPL BLOCKIN W(0)
```

To read back the data by using the command BLOCKOUT, they have to be stored to a buffer first, then they can be transferred from the buffer to the display.

Example:

```
UPL BLOCKOUT W(0),1024
UPL OUT"trac trac1"
```

Block data are always read in completely, that means all measured values are transferred when using the command BLOCKIN. Whenever data have to be read out - by using the command BLOCKOUT - the number of transferred values have to be defined,
e.g. UPL BLOCKOUT W(0),100.
This means, that a array W is transferred, starting from index 0 to index 99 (= 100 values).

The BASIC Screen

Under software control different modes are available to use the screen:

- BASIC uses the main part of the screen, UPL builds up a field for display of measured values at the upper part of the screen.
  BASIC can be used with all features of graphics and labelling, including softkeys used by BASIC.

- BASIC uses the whole screen, no display of results. Therefore the measurement display has to be switched off (by the command UPL OUT"disp:ann off" or by pressing <CTRL D> from the UPL operation).
  BASIC can be used with all features of graphics and labelling, including softkeys used by BASIC.

- While a BASIC program is running, the command UPL GTL U switches the screen to be used by the UPL. In part-screen mode the left part of the screen can be used for labelling instead of the panel display.
  The softkeys are used by the UPL. After the program has stopped, the screen switches back to BASIC.

- When the screen is used by BASIC, the UPL graphics can be copied to the BASIC screen by the command UPL GTL G.
  BASIC can be used with all features of graphics and labelling, combined with the graphics of the UPL

Switching between Program Control and UPL Operation:

The command UPL GTL interrupts the program and switches to the UPL screen. The analyzer can be operated manually now.
Pressing the F3 key switches to the BASIC screen and the program will continue where it has been interrupted before.

Control of External Instruments

The Audio Analyzer UPL already includes the IEEE-488 interface in its basic configuration, so there is no additional option necessary to control external instruments.

Universal Commands

IEC ATN	line ATN active
IEC NATN	line ATN not active
IEC DCL	device clear
IEC EOI	line EOI active
IEC NEOI	line EOI not active
IEC IFC	send IFC
IEC LLO	local lockout
IEC PPD	parallel poll disable
IEC PPE k1 k2	parallel poll enable
IEC PPL v%	trigger of parallel poll
IEC PPU	parallel poll unconfigure
IEC RLC	release of control
IEC RQS	send service request
IEC REN	line REN active
IEC NREN	line REN not active
IEC SPD	serial poll disable
IEC SPE	serial poll enable
IEC TERM a	definition of input terminator
IEC TIME a	time-out monitor
IEC UNL	unaddressing of devices as listener
ON SRQ GOSUB m	
ON SRQ GOTO n	event-controlled branching on SRQ
ON SRQ RETURN	inhibit of branching on SRQ

Addressable Commands

IEC ADR a	assignment of IEC address
IEC GET	group execute trigger
IEC GTL	go to local
IEC IN a1,[;a2],v$	read in data
IEC$IN v$	input of strings without addressing
IEC%IN v%	input of character without addressing
IEC LAD a	send listener address
IEC MTA (IECUNT)	deaddressing of devices
IEC OUT a1 [;a2], s$	send string
IEC$OUT s$	send string without address
IEC%OUT a%	send single character without address
IEC PCON b,k1;k2	parallel poll configure
IEC PPC	parallel poll configure
IEC SAD a	secondary addressing of devices
IEC SDC	selected device clear
IEC SPL b1[;b2],v%	serial poll
IEC TAD a	send talker address
IEC TCT	take control
IEC UNL	send unlisten
IEC WMLA	wait for listener address
IEC WMTA	wait for talker address
IEC WTC	wait for take control

Example how to control an instrument via IEEE-488 bus:

IEC OUT address,"command"

That's it!

Example for a Program

Using a small example, a measurement sequence will be explained. The following program loads a sequence of setups and executes the measurements.

```
10 REM ***********************
20 REM * example for a sequence
30 REM * to measure amplifiers
40 REM ***********************
50 T=2: REM waiting time in seconds
60 UPL OUT "MMEM:LOAD:STAT 0, 'LEVS_PA.SAC' "
70 UPL OUT "disp:conf p"
80 UPL GTL U
90 UPL OUT "init;*wai"
100 GOSUB 1000: REM go to waiting loop
110 UPL OUT "MMEM:LOAD:STAT 0, 'SNRA_PA.SAC' "
120 UPL OUT "init;*wai"
130 GOSUB 1000: REM go to waiting loop
140 UPL OUT "MMEM:LOAD:STAT 0, 'THD_PA.SAC' "
150 UPL OUT "init;*wai"
160 GOSUB 1000: REM go to waiting loop
170 UPL OUT "MMEM:LOAD:STAT 0, 'DFD_PA.SAC' "
180 UPL OUT "init;*wai"
190 GOSUB 1000: REM go to waiting loop
200 UPL OUT "MMEM:LOAD:STAT 0, 'CRSS_PA.SAC' "
210 UPL OUT "disp:conf p"
220 UPL OUT "init;*wai"
230 UPL OUT "OUTP:SEL CH2"
240 UPL OUT "DISP:TRAC:FEED 'HOLD' "
250 UPL OUT."DISP:TRAC2:FEED 'SENS:DATA' "
260 UPL OUT DISP:TRAC2:Y:EQU OFF"
270 UPL OUT "DISP:TRAC2:Y:RLEV:MODE CH2M"
280 UPL OUT "init;*wai"
290 GOSUB 1000: REM go to waiting loop
300 STOP
1000 A=TIME: REM waiting loop
1010 IF (TIME-A)<100*T THEN 1010
1020 RETURN
```

- in the lines 60, 110, etc. setups are loaded, after this the measurements are started by the command "init;*wai"

- to display the measurements on the screen, the graphic mode is switched on in line 80 ("GTL U"), a full-screen graph is selected (command "disp:conf p" in line 70)

- to have some time to look to the measured results, the program brunches to a waiting loop (lines 100, 130, etc.), the waiting time can be set by the variable T in line 50

- from line 200 on, a crosstalk measurement is done in both directions; beginning with crosstalk from channel 1 to 2, the measurement runs as defined by the setup

- for the other direction, a few changings have to be made. The easiest way is to switch to manual operation of the UPL and to key-in the different settings in the panels while using the logging mode to add the program lines to the example program. After renumbering we got the lines 230 to 270.

Macro Operation

Settings and measurement sequences written as BASIC programs can be called and used on the Audio Analyzer UPL in different ways:

Call from BASIC User Interface:

Once the BASIC user interface has been activated by pressing F3 (on the external keyboard) or BACKSP (on the UPL keyboard), the program can be loaded with LOAD"program name" (softkey or F11) and then started with RUN (softkey or F6). The program name needs to be typed during loading. After the program has been quit, the UPL interface is reactivated with F3 or LOCAL key.

Automatic Start of a BASIC Program on switch-on:

The UPL can be configured to load and execute a particular program at switch-on by modifying the file USERKEYB.BAT. After the program has been quit, the UPL interface is reactivated with F3 or LOCAL key.

Call from UPL User Interface (Exec Macro):

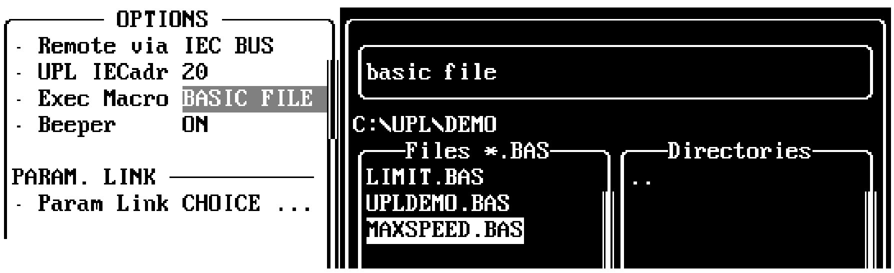


Fig.: 9-4

Via the menu item "Exec Macro" on the OPTIONS panel, the BASIC program names can be selected with the aid of the standard file box. The selected program is loaded and started automatically. After the program has been quit, the UPL interface is reactivated automatically. As no file name needs to be typed, a macro of this type can also be started without an external keyboard.

Call from an External Program via IEEE-Bus Interface:

Any BASIC program can be loaded and started with the IEEE bus command "SYST:PROG:EXEC'file name'". This results in very modular measurement tasks; the controller is not directly involved with how the measurement is executed by the UPL and thus can handle other tasks in the meantime.

Audio Measurements on Surround Sound Decoders (using UPL-B23)

Application

Up to now, measuring surround decoders necessarily involved defining and storing coded test sequences on a DVD or the PC hard disk. The DVD player/PC was connected to the DUT, where the test signals were decoded and finally measured at the analog outputs. Since the test files and the measurements ran on different instruments, synchronization was difficult, leading to extended measurement times.

Solution with Audio Analyzer UPL plus Option UPL-B23:

This option enables the Audio Analyzer UPL to generate AC-3-coded test signals directly with the built-in generator. The measurements are synchronized automatically between the generator and the analyzer.

This has the following advantages:

- The internal synchronization considerably speeds up measurements
- Test sequences can be combined much more flexibly, since the number of channels, frequency or level sweep, start and stop frequency/level as well as the number of sweep points can be set directly. Settings are made in a similar way to those for a standard analog sweep
- The test signals are no longer recorded on DVD/PC, thus saving time previously spent on combining and coding the test signals
- Additional hardware, such as a PC or DVD player, is not required

Functional description:

Several thousand AC-3-coded test files are stored on the UPL audio analyzer hard disk, with each individual file representing a defined frequency/level combination. The files required for a sweep are loaded into the DSP and replayed until the analyzer yields a settled measurement result. The analyzer then switches automatically to the next file (= next frequency/level point), and the next measurement is triggered until the complete sweep has been processed.
Each of the WAV files used contains one or more complete sine periods. The files can thus be combined to form a test sequence without interruption and artefacts. The DUT remains synchronized to the AC-3 data stream.

Operation:

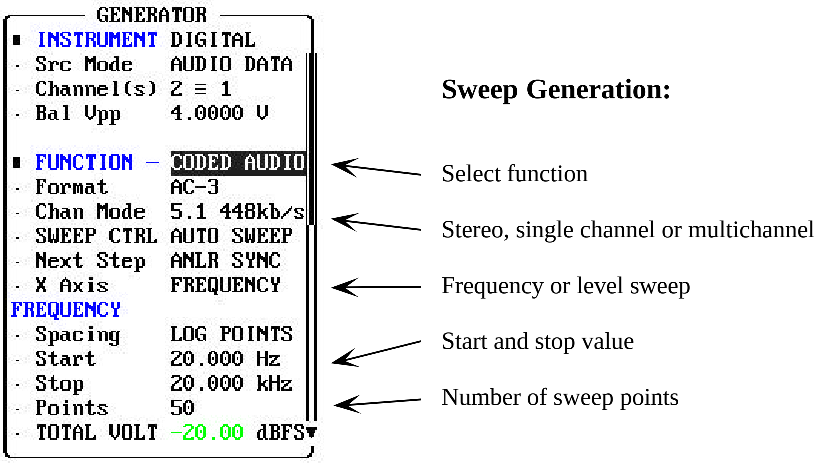


fig.: 10-1

Format:

• AC-3	currently only AC-3 format (Dolby Digital) is supported, other data formats in preparation

Channel Mode:

• 2/0 192kb/s	stereo signals, coded with 192 kb/s
• 5.1 448kb/s	5.1-surround signals, coded with 448 kb/s
• L 448kb/s, C 448kb/s, R 448kb/s, LS 448kb/s, RS 448kb/s, LFE 448kb/s	single-channel signals, coded with 448 kb/s for crosstalk measurements

Variation Mode:

• Voltage	the amplitude can be varied, (frequency fixed)
• Frequency	the frequency can be varied, (amplitude fixed)

Sweep Control:

settings made in a similar way to those for a standard analog sweep

Audio Switcher UPZ

fig.: 10-2

Applications:

- In order to test **5.1 decoders**, the six channels are connected to the Audio Analyzer UPL via the Audio Switcher UPZ. The UPZ is controlled directly from the UPL panel via an RS-232-C interface.
- For **professional surround applications**, the Audio Switcher UPZ comprises 8 channels, with two output channels to allow the two UPL measurement channels to be used simultaneously.
- Through its **RS-232-C interface**, Audio Switcher UPZ can also be controlled directly from a PC. This opens up new possibilities; for instance, in broadcasting stations, where studio operations require the switching of several audio channels.
- The UPZ may also be used in **production**; for instance, when car radios are tested, measurements can be performed at all four loudspeaker outputs.
- It is possible to cascade up to 16 input switchers plus 16 output switchers, allowing up to **128 input and output channels** to be switched.

Input /Output Version

Like the Audio Analyzer UPL, the Audio Switcher UPZ has XLR connectors. Since there is a difference between male and female connectors in the XLR system, the UPZ is available both as an input and an output model.

UPZ stock no. 1120.8004.02 (input version: 8 female inputs, 2 male outputs)
UPZ stock no. 1120.8004.03 (output version: 8 male outputs, 2 female inputs)

Measurements on FM Tuners using Signal Generator SML/SMV

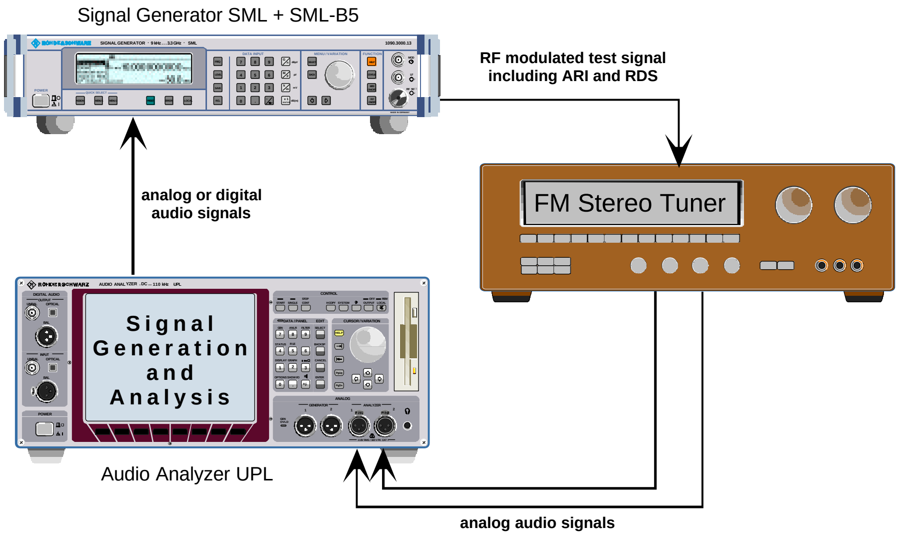


fig.: 10-3

To test FM stereo tuners RF modulated audio test signals have to be used. After the demodulation in the device under test these signals will be measured. For car radio applications also traffic announcements (ARI = Automotive Radio Information) as well as RDS data (Radio Data System) have to be generated.
Using Rohde & Schwarz Signal Generators SML or SMV together with the option Stereo/RDS Coder SML-B5 all this is possible.

Measurements using Audio Analyzer UPL:

- The Stereo/RDS Coder SML-B5 provides analog and digital modulation inputs. Test signals will be generated by the Audio Analyzers UPL, after passing the modulator and the device under test they will be measured by the analyzer part of the UPL. The advantage is that generation and analysis is automatically synchronised, resulting in short measuring sequences.
- The measurements can be automated. Using the Automatic Sequence Controller UPL-B10 complete test programs can run on the UPL, while Signal Generator SML/SMV are controlled via the IEEE or the RS-232 interface.
- In production lines the whole test setup can be controlled by an external computer.
- To measure car radios or surround receivers providing more than two audio outputs, the Audio Switcher UPZ can be used, as described on the page before.

Automatic Line Measurements acc. to ITU-T O.33

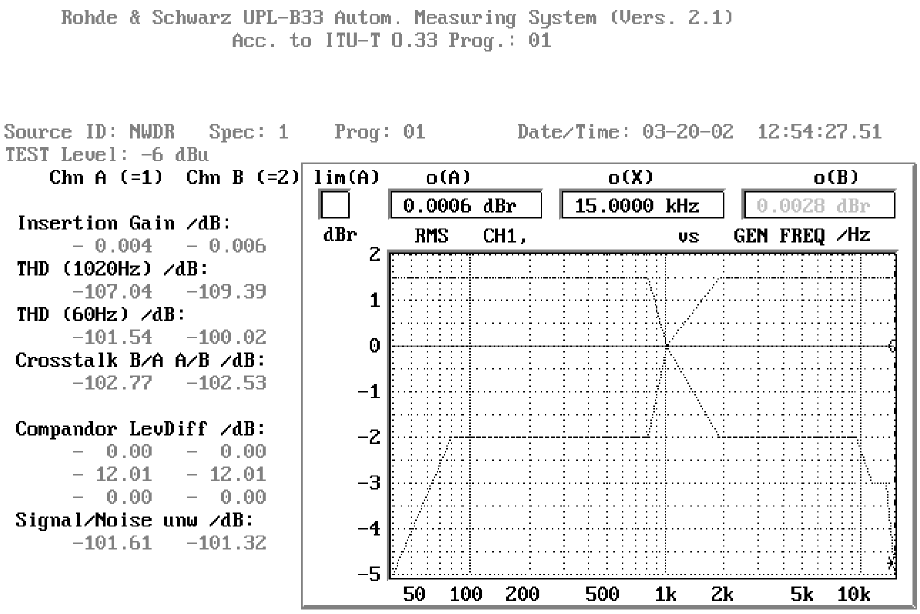


fig.: 10-4

To measure audio links in broadcast and TV applications even between different countries, the ITU recommendation O.33 describes an international procedure. This allows to use test instruments of different manufactures simultaneously, where normally generator and analyzer are located at different places. Synchronisation between both instruments is done by using special FSK signals. The operator can use the standard test sequences as defined by the ITU-T O.33, or he can define his own sequences.

The software package UPL-B33 realises comfortable analysis and display of the results. The operator can adapt the sequences to his special needs or he can add new sequences quite easily.

Option UPL-B10 is needed for the use of UPL-B33.

Measurements on Hearing Aids

fig.: 10-5

Audio Analyzer UPL in conjunction with option UPL-B7 is a complete test system for all standard measurements on hearing aids. To carry out such measurements, UPL only requires the options Audio Monitoring (UPL- B5) and Universal Sequence Controller (UPL-B10). The test system meets all the requirements relevant in the production, quality management and service of hearing aids.
The HEARPRO software supplied with the system allows the user to generate test routines tailored to the specific characteristics of the device under test. The type and sequence of measurements are freely selectable. All test parameters can be accurately defined.

Option UPL-B7 includes:

- a compact acoustic test chamber
- a complete set of cables
- a 2 cm^3 coupler with built-in microphone and calibration adapter
- a set of battery adapters for all commercial battery sizes for DUT power supply

Calibration of the complete test setup requires a sound level calibrator and a test microphone which are not part of the equipment supplied.

Acoustical Measurements on Mobil Phones using UPL16 or UPL-B8 or UPL-B9

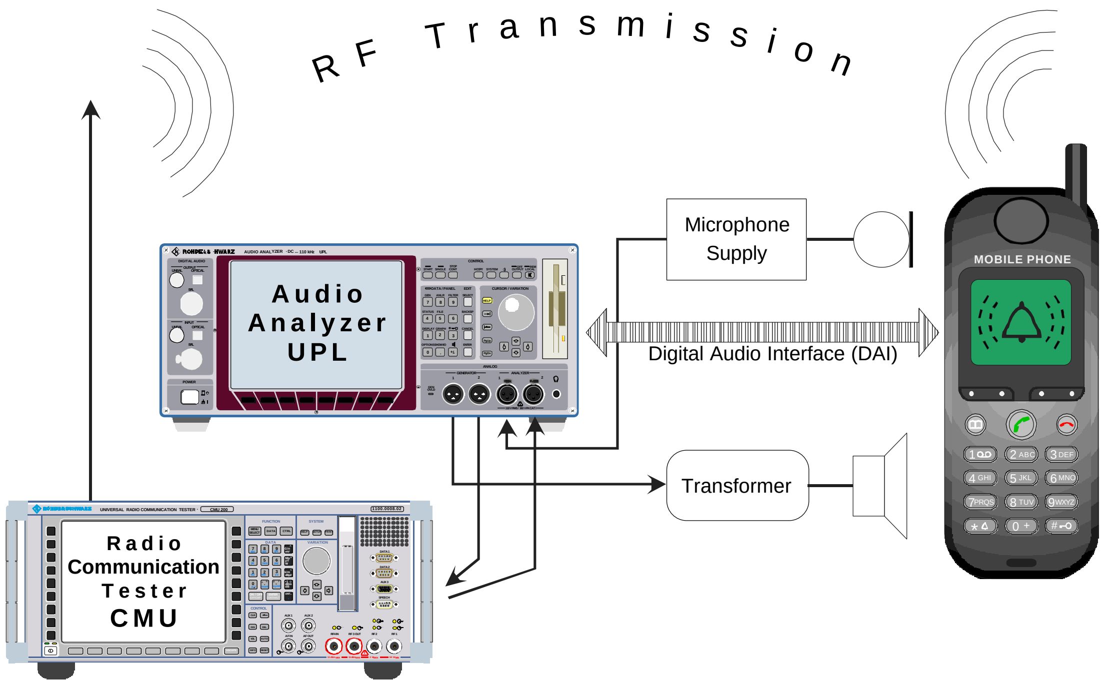


fig.: 10-6

Various test methods have been standardised for determining acoustic characteristics. **Audio Analyzer UPL16** was developed for conformance tests on GSM mobiles. It performs all audio measurements in line with chapter 30 of GSM 11.10, phase 2 (which is now 3GPP TS 51.010-1 chapter 30). Access to the internal digital signals of special test mobile phones is via the standard digital audio interface (DAI).

To measure commercial GSM phones, the **option Mobile Phone Test Set UPL-B8** is available. It enables GSM mobile telephones to also be measured via normal air interface (RF link). Because speech coder/decoder is included in the measurement path too, special multitone signals with sophisticated measurement technology are used. However, the method is not standardised world-wide, and therefore cannot be used for type approval tests.

During the development of third-generation mobile radio a goal was to define test methods which enable measurements of the acoustic characteristics via the normal air interface. The task was to measure all the characteristics in normal operating mode, which means that all the signal processing algorithms in the mobile telephone should remain active during the measurement. For those tests now defined in 3GPP TS 26.132, artificial speech or speech like signals are used.

Up to 1999 there were different acoustic requirements for GSM and 3GPP mobile phones. A harmonisation of the tests and requirements for GSM and 3GPP mobile phones was done early 2001 now specifying the same tests for 3GPP and GSM release 4 mobiles.

The **option Mobile Phone Test Set UPL-B9** for the Audio Analyzer UPL provides all acoustic tests defined in 3GPP TS 26.132 resp. 3GPP TS 51.010 using Audio Analyzer UPL06, UPL66 or UPL16. Besides the acoustical interface (artificial mouth/ear), a Radio Communication Tester CMU200 is needed to establish the RF connection. All tests are validated by an independent test house.

Acoustical Tests of GSM and 3GPP Mobile Phones with UPL16 (DAI) or UPL plus UPL-B9 (Air Interface)	**3GPP TS 51.010 GSM Release 99** (DAI Interface) Chapter	**3GPP TS 51.010 GSM Release 4** (Air Interface) Chapter	**3GPP TS 26.132 3GPP Release 4** (Air Interface) Chapter
Sending sensitivity/frequency response	**30.1***	**30.12***	**7.4***
Sending loudness rating	**30.2***	**30.13***	**7.2***
Receiving sensitivity/frequency response	**30.3***	**30.14***	**7.4***
Receiving loudness rating	**30.4***	**30.15***	**7.2***
Side tone masking rating	**30.5.1***	**30.16***	**7.5***
Listener side tone rating	30.5.2		
Echo loss	30.6.1 (release 4)	**30.17.1***	**7.7***
Stability margin	**30.6.2***	**30.17.2***	**7.6***
Sending distortion	**30.7.1***	**30.18***	**7.8***
Receiving distortion	30.7.2		**7.8***
Sidetone distortion	30.8		
Out-of-band signals (sending / receiving)	30.9		
Idle channel noise (sending / receiving)	30.10		**7.3***
Ambient noise rejection	30.11 (release 4)	**30.19***	**7.9***

*** validated test cases**

fig.: 10-7

Contents