



ROHDE & SCHWARZ

Test and Measurement
Division

Operating Manual

UMTS MOBILE PHONE TESTS for GSM and WCDMA

R&S[®] UPL-B9

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1 Overview

The acoustic transmission and reproduction quality of a mobile phone is its most important characteristic in everyday use. Even the most visually appealing design and wonderfully sophisticated operating concept are not much use, when the user cannot or can hardly understand what is being said at the other end.

Instruments and methods for measuring acoustic characteristics are therefore essential tools for assessing the quality and suitability of a mobile phone.

Special test mobile phones have been required for type-approval testing up to now. The special Audio Analyzer UPL16 performs all audio measurements in line with chapter 30 of GSM 11.10 on special mobile phones under test with a digital audio interface (DAI).

Since a digital audio interface is no longer required even for type-approval tests on 3rd generation mobile phones, it was necessary to develop new test methods that allowed measurements to be performed via the air interface and the normal speech coder and decoder. The test signals have to simulate human voice in the frequency and time domain to be able to test the mobile phone in the normal operating mode with DSP assessment such as voice activity detector (VAD), noise suppression, echo cancellation, etc switched on.

These tests are based on new standards for 3GPP mobile phones. The test methods are stipulated in 3GPP TS 26.132 and the values to be attained in 3GPP TS 26.131.

As of release 4 of the GSM 51.010 standard (successor to GSM 11.10), even GSM mobile phones may be tested to 3GPP TS 26.132.

The UPL-B9 option (UMTS Mobile Phone Tests) of the Audio Analyzer UPL is now available for measuring the acoustic characteristics of 3GPP and GSM mobile phones. The measurements are in line with 3GPP TS 26.131, TS 26.132 and TS 51.010 and have been validated by an independent test house for type-approval testing on GSM and WCDMA mobile phones.

2 Preparation and Start of Application Software

Required Measuring Instruments and Accessories

The Audio Analyzer UPL with the following options is required for the measurements:

- Extended Analysis Functions UPL-B6
- Universal Sequence Controller UPL-B10
- UMTS Mobile Phone Tests UPL-B9 (version 2.0)

The GSM mobile phone under test is driven by the Universal Digital Radio Communication Tester CMU200 via the RF interface. This tester simulates a base station for the mobile phone so that a call can be set up. For GSM, the CMU200 must be equipped with the options CMU-B21 (signalling unit), CMU-B52 (speech coder/decoder) and the appropriate software options for the GSM band used. For WCDMA, the option CMU-B69 is required. The firmware level in the CMU200 must be 3.50 or higher.

Acoustic devices such as an artificial mouth, artificial ear and other accessories are required for the measurements. The following equipment from Brüel & Kjær or G.R.A.S. is normally used:

Device	Description	Type
Telephone test head	Device for fixing the DUT in the prescribed position	B&K 4602B
Ear simulator	Measuring microphone with adapters for connection to the ear piece of the DUT	B&K 4185 (type 1)
Wideband ear simulator	Measuring microphone with adapters for connection to the ear piece of the DUT	B&K 4195 (type 3.2)
Artificial mouth	Special loudspeaker for simulation of the mouth	B&K 4227
Head and torso simulator	Torso simulator with artificial ear (type 3.3) and artificial mouth	B&K 4128 D
Acoustic calibrator	Sound level calibrator for calibrating the measuring microphone	B&K 4231
Microphone power supply	Power supply and preamplifier for the measuring microphone	B&K 2690A0S2 or G.R.A.S.

Note: *With the amplifier set to 0 dB, the microphone power supply B&K 2690A0S2 produces too much noise for measuring idle noise and distortion. It is therefore advisable to set a gain of 20 dB.*

A cable with a BNC connector and special small or angled banana plugs is required for connecting the artificial mouth, as the space between the mouth connectors and the test rack (B&K 4602B) is too small for common banana plugs.

The transformer supplied with option UPL-B9 must be connected between generator output 1 of Audio Analyzer UPL and the connector of the artificial mouth. The transformer matches the impedance of the loudspeaker in the artificial mouth to that of the generator output of the UPL. Without this transformer, the available power is too low for driving the artificial mouth.

Alternatively, a power amplifier, preferably with a gain of approx. 10 dB to 20 dB, can be connected between generator output and mouth instead of the transformer. In this case, the gain set must be kept absolutely stable after calibration.

A cable with male (analyzer) and female (generator) XLR connector is supplied for connection to the "Speech" connector of the Digital Radio Communication Tester CMU.

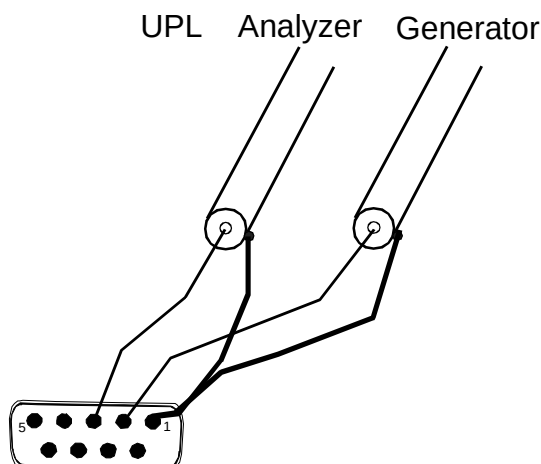


Fig. 1 Assignment of 9-contact speech connector on CMU front panel

This cable is configured for connection to link handler #1 in the CMU. Depending on the CMU hardware, link handler #2 in the CMU can also be used for GSM; in this case, the supplied adapter must be inserted between the cable and the speech connector on the CMU.

Caution: *If normal measuring results are not obtained when measuring GSM mobile phones, please repeat the measurement with the adapter inserted!*

An external PC keyboard must also be connected to the UPL (large DIN connector). The desired driver for country-specific keyboards may be defined in the C:\UPL\USERKEYB.BAT file (see section 2.15.4 of UPL manual).

The BASIC program required for automatic sequence control and the files for generating the artificial voice are on the three floppies supplied with the UPL-B9 option. The audio analyzer must meet the following firmware requirements:

- UPL firmware version 3.01 or higher
- Option Extended Analysis Functions UPL-B6 installed
- Option Universal Sequence Controller UPL-B10 installed
- Option Mobile Phone Tests UPL-B9 installed,
(will be done automatically during software installation)
- UPL configured with 64 kbyte program memory and 32 kbyte data memory for automatic sequence control (using configuration tool UPLSET setting 3)

Installing the Software

The application software is installed with the aid of the UMTSINST.BAT installation program on program floppy 1. The installation number of the option UMTS Mobile Phone Tests UPL-B9 must be known.

Caution: *The software can only be installed on the specified Audio Analyzer UPL with matching serial number.*

- **Quit the measurement software by pressing the SYSTEM key on the instrument or Ctrl F9 on the keyboard.**
- **Insert floppy No. 1.**
- **Select floppy disk drive (enter A:).**
- **Call the installation program (enter UMTSINST).
You are requested to enter the installation number of the UPL-B9.**
- **Enter the installation number supplied with the UPL-B9. If the number does not match the serial number of UPL, the installation is aborted.**
- **Insert floppy No. 2 when asked and press any key.**
- **Insert floppy No. 3 when asked and press any key.**
- **Return to UPL program (enter C:\UPL)
The UMTSINST program creates the C:\3GPP directory in the Audio Analyzer UPL (if it is not already available) and copies the BASIC program, the artificial voice and all setups and files required for the application into this directory.**

Test Setup

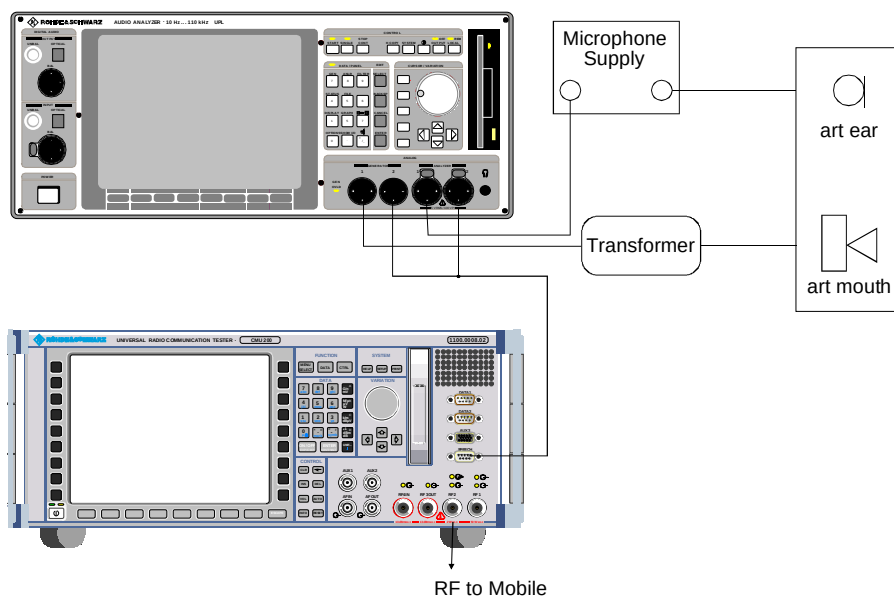


Fig. 2 Test setup and connection of external components

Starting the Application Software

The application program is executed by way of the universal sequence controller. The Audio Analyzer UPL is switched to automatic sequence control using the F3 key (on the external keyboard).

The logging function must be off; this is the case when "logging off" is displayed at the bottom right of the screen. With the logging function on, all commands entered in the manual mode would be appended to the program and use up memory. The logging mode is switched on and off using the F2 key on the external keyboard.

The application programs must be called from path C:\3GPP in order to find all the required program routines and setups. The path can be changed in any of the following ways:

- in the manual mode with the "Working Dir" command in the FILE panel
- by calling one of the setups required for measurements on mobile phone
- in the automatic sequence control mode with the BASIC command
UPL OUT "MMEM:CDIR 'C:\3GPP'"
- under BASIC with the SHELL command by entering CD\3GPP and pressing EXIT
- at DOS level by entering CD\3GPP

Program floppy 1 contains the BASIC program 3GPP_TST.BAS for measurements on GSM and WCDMA mobile phones. It is loaded and started by entering the following:

- LOAD"3GPP_TST"
- RUN

The softkeys displayed at the bottom of the screen in the automatic sequence control mode can be used instead.

Configuring the Application

"Default-Printer" is factory-set in the OPTION panel. This means that the printer configuration does not depend on the setup and the printer used last by the Audio Analyzer UPL remains configured. Consequently, new settings need not be made by the user. It is useful to select the desired printout with type, format and scaling in the manual mode before the program is started. All subsequent printouts triggered with the hardcopy key will then be printed with these settings.

IMPORTANT: *Correct execution of the software cannot be ensured if settings in the setup are changed.*

Setup Conversion for Firmware Updates

For an update of the UPL firmware, the setups may have to be converted. This is done automatically when the setup is loaded, but the conversion delays the loading. To avoid this, the setups can be converted before the application software is started:

- at DOS level by calling the UPL conversion program:

DO_CONV \3GPP

This converts all setups in the 3GPP directory.

IMPORTANT: *Please note that a previous firmware version can no longer be used in the UPL after the conversion of setups:*

3 Operating Concept

Softkeys are displayed at the bottom of the screen for operation and test program selection. The softkey functions are also assigned to hardkeys on the external keyboard so that the keyboard can also be used for selecting program routines.

After the program has been started, the title page

**Measurement of
GSM and WCDMA Mobile Phones
according to 3GPP TS 26.132
with Audio Analyzer UPL**

and the following softkey line are displayed:

F5	F6	F7	F8	F9	F10	F11	F12
	CONT						

After F6 CONT has been pressed, the following request is displayed:

**Selection of
Ear Type used**

F5	F6	F7	F8	F9	F10	F11	F12
	TYPE 1	TYPE 3.2L	TYPE 3.2H	TYPE 3.3	TYPE 3.4		

After a type has been selected, the following request is displayed on the screen:

**Please establish
call to Mobile
and set CMU
to Bit Stream Handset Low (GSM)
or Voice Speechcodec Low (WCDMA)**

To do so, switch on the mobile phone. After successful registration, press the CALL TO MOBILE key on the CMU or dial a number on the mobile phone and press the transmit key. The selection of the speech coder in the CMU depends on the standard used. If GSM is used, HR (half rate version 1 traffic mode), FR (full rate version 1 traffic mode), EFR (full rate version 2 traffic mode) and AMR codecs can be selected; with WCDMA, only the AMR codec is used. For standard-compliant measurements, the use of AMR at 12.2 kbit/s is prescribed so as to minimize the corruption of the measurement results by the coding. The EFR codec for GSM is identical to AMR 12.2 and can therefore also be used. The speech codec must be set to "Low" for all measurements. In the case of GSM, the setting is made in the Network menu, under Bit Stream, Speechcod./Handset Low; in the case of WCDMA, under BS Signal, Dedicated Channel, Voice settings, Speechcodec Low. If WCDMA is used, the setting can only be changed after the call has been terminated.

The following softkey line is displayed:

F5	F6	F7	F8	F9	F10	F11	F12
	CONT						

After F6 CONT has been pressed, the following message is displayed:

Measurement of
GSM and WCDMA Mobile Phones
according to 3GPP TS 26.132
with Audio Analyzer UPL
select Test to be performed

The measurements on the mobile phone under test can now be started, since all required calibration values are stored in the UPL.

IMPORTANT:

When the test setup is installed for the first time, the microphone in the artificial ear, the artificial mouth and the speech coder in the CMU have to be calibrated (see "Calibration Routines"). In this case, the message requesting a call setup to the mobile phone under test can be skipped with CONT.

To select the individual measurements, the softkeys F5 to F12 with abbreviations for the measurement names are displayed.

F5	F6	F7	F8	F9	F10	F11	F12
END	SEND	REC_NOM	REC_MAX	STMR	ECHO_TCL	STB_LOSS	→

Clicking a key starts the associated test routine. Since there are more selection items than softkeys, the next set of softkey definitions is called with the F12 key.

F5	F6	F7	F8	F9	F10	F11	F12
←	DIST_SND	DIST_REC	IDLE_SND	IDLE_REC	AMB_NOI		→

----CALIBRATION----

EXP-FILES

F5	F6	F7	F8	F9	F10	F11	F12
←	EAR	MOUTH	CMU-COD				DELETE

If F12 shows an arrow pointing to the right, press F12 to see the next set of softkey definitions. Press F5 showing an arrow towards the left to go back to the previous set. At the lowest level, F5 shows END. After pressing F5 END, the query "Do you really want to quit?" is displayed and the application can be terminated (see page 38).

Measurements

General

Special problems caused by the coding and decoding algorithms of mobile phones are encountered when measuring acoustic characteristics. Vocoder are used to attain the lowest possible data rate. In this case, not the actual voice signals but only the filter and fundamental parameters required for signal reconstruction are transmitted.

Purely sinusoidal tones normally used for the measurements cannot be transmitted with such a system. In type-approval tests, the coder and decoder have therefore so far been excluded from the measurement. A specially prepared test mobile phone required a digital audio interface (DAI) for the transmission of audio signals with linear PCM coding. The Audio Analyzer UPL16 is equipped with this DAI interface, allowing direct transmission of the test signals to and from a mobile phone under test with a DAI interface.

In commercially available mobile phones, this interface is not accessible; measurements during normal operation can therefore only be performed via the air interface with the speech coder and decoder included. A digital audio interface is no longer provided even for type-approval testing of 3rd generation mobile phones. As mentioned above, measurements using sinusoidal tones cannot be performed because the static sinusoidal input signal becomes a more or less stochastic output signal as a result of coding, particularly in the medium and high audio frequency ranges.

Signals similar to voice therefore have to be used for the measurement, i.e. either artificial voice according to ITU-T P.50 or a multitone signal according to ITU-T P.501 is possible. At the same time, modulation of the signal in time must largely correspond to voice, since many modern mobile phones use algorithms for interference suppression which use the modulation to distinguish the useful from the interfering signal.

The test routines in the UPL use an amplitude-modulated multitone signal according to ITU-T P.501 as described in 3GPP TS 26.132 for echo loss measurement.

Notes on Individual Measurements

The measurements to be performed are described below in the sequence in which they are carried out.

Perform all measurements in an anechoic chamber with sufficient absorption against interfering sound. Since special distortion measurements and the measurement of idle noise place high demands on measurement conditions, the A-weighted noise in the test chamber must be below 30 dB(A).

Measurements are started by pressing the corresponding softkey or function key on the external keyboard. When the measurement is completed, the results are shown and the following softkey line is displayed:

F5	F6	F7	F8	F9	F10	F11	F12
	CONT		ABS_SENS	EXP_FILE	TRC_FILE	PCX_FILE	HARDCOPY

A return to the selection level is possible with CONT or the results can be printed or saved (see section 6, Processing of Measurement Results).

Sending Frequency Response and Loudness Rating

Sending Frequency Response

The sending frequency response is specified as the transmission ratio in dB of the voltage at the decoder output to the input noise pressure at the artificial mouth.

The mobile phone under test is installed in the LRGP (loudness rating guarding position) according to ITU-T P.76, and the receiver is sealed to the artificial ear.

Tones with a sound pressure of -4.7 dBPa are created with the artificial mouth at the MRP (mouth reference point), and the corresponding output voltage is measured at the CMU speech decoder output and evaluated.

The sending frequency response must be within the limit lines specified according to table 1 of 3GPP TS 26.131. The absolute sensitivity is not yet taken into account.

Table 1 Limit lines according to table 1 of 3GPP TS 26.131

Frequency (Hz)	Upper limit (dB)	Lower limit (dB)
100	-12	
200	0	
300	0	-12
1000	0	-6
2000	4	-6
3000	4	-6
3400	4	-9
4000	0	

The offset of the measured frequency response to the upper or lower limit line is calculated and then the whole trace is shifted by the mean value of the maximum and minimum offset. Then another limit check is performed. If the shifted curve is now within the limit lines, PASS is output, otherwise FAIL is displayed. The limit check is performed at each measured frequency. If the measured value and the end point of a limit line are not at the same frequency, it may happen that the trace slightly crosses a corner of the limit line although there are no limit violations.

Sending Loudness Rating

The sending loudness rating (SLR) takes into account the absolute loudness in the transmit direction and weights the tones in compliance with the normal sensitivity of the average human ear.

To this end, the frequencies (Hz) of bands 4 to 17 are evaluated according to table 1 of ITU-T P.79.

Table 2 Frequencies (Hz) of bands 4 to 17 according to table 1 of ITU-T P.79

200	1000
250	1250
315	1600
400	2000
500	2500
630	3150
800	4000

The sensitivity at each frequency is defined as the ratio dBV/Pa referenced to the rated internal level in dBm0, and the sending loudness rating is calculated according to formula 5-1 of ITU-T P.79.

Due to the input sensitivity tolerance of the CMU speech coder, the individual sensitivity of the CMU used has to be taken into account in order to calculate the sending loudness rating (see calibration routines). According to 3GPP TS 26.131, the sending loudness rating should be between 5 dB and 11 dB, with lower dB values corresponding to greater loudness (5 dB = maximum loudness, 11 dB = minimum loudness). The measured SLR is indicated in a window in the frequency response display and checked for compliance with these limits. In addition to the numeric value, either PASS or FAIL is displayed.

The general PASS or FAIL information is obtained from the limit check of the frequency response curve and the loudness rating. PASS is output only if both the curve and the loudness value are within tolerances.

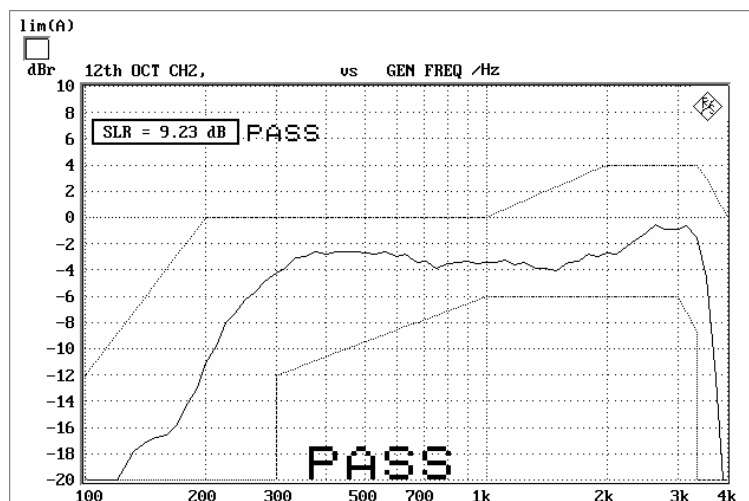


Fig. 3 Sending frequency response with SLR value displayed

Receiving Frequency Response and Loudness Rating

The following two routines are available, as the permissible limit values of the loudness rating depend on the loudness set in the mobile phone under test: REC_NOM checks for compliance with rated loudness setting and REC_MAX checks whether maximum loudness is set.

Receiving Frequency Response

The receiving frequency response is specified as the transmission ratio in dB of the sound pressure in the artificial ear to the input voltage at the speech coder input of the CMU. The measured sound pressure is referenced to the ear reference point (ERP). For ear type 1, the measuring microphone is directly applied to the ERP and no further correction is required. For ear types 3.x, the measuring microphone is applied to the drum reference point (DRP), which is why any measured value has to be converted to the ERP by means of calibration factors. According to 3GPP TS 26.132, the measurement may be referenced to the drum reference point (DRP) even in special cases. For this reason, after the measurement routine has been called, an additional query is displayed asking whether the measurement is to reference to the ERP or the DRP.

The mobile phone under test is installed in the LRGP (ITU-T P.76), and the receiver is sealed to the artificial ear.

The speech coder is driven such that tones with a system reference level of -16 dBm0 are obtained. The sound pressure in the artificial ear is measured and evaluated.

Ear type 1 will no longer be used for measurements on 3GPP mobile phones. Therefore, 3GPP TS 26.131 defines limit values only for ear types 3.x, whereas the limit values specified in 3GPP TS 51.010 (previously GSM 11.10) are still valid for ear type 1.

Table 3 Limit lines according to table 30.2 of 3GPP TS 51.010 (ear type 1)

Frequency (Hz)	Upper limit (dB)	Lower limit (dB)
100	-12	
200	0	
300	2	-7
500	*	-5
1000	0	-5
3000	2	-5
3400	2	-10
4000	2	

* Intermediate values are obtained by drawing a straight line between the specified values and using a logarithmic frequency scale and a linear dB scale.

When ear type 1 is used, the receiving frequency response must be within the limit lines specified in table 30.2 of 3GPP TS 51.010. When ear type 3.x is used, it must be within the limit lines specified in table 2 of 3GPP TS 26.131. The absolute sensitivity is not yet taken into account.

Table 4 Limit lines according to table 2 of 3GPP TS 26.131 (ear type 3.x)

Frequency (Hz)	Upper limit (dB)	Lower limit (dB)
70	-12	
200	2	
300	*	-9
500	*	*
1000	*	-7
3000	*	*
3400	*	-12
4000	2	

* Intermediate values are obtained by drawing a straight line between the specified values and using a logarithmic frequency scale and a linear dB scale.

The offset of the measured frequency response to the upper or lower limit line is calculated and then the whole trace is shifted by the mean value of the maximum and minimum offset. Then another limit check is performed. If the shifted curve is now within the limit lines, PASS is output, otherwise FAIL is displayed. The limit check is performed at each measured frequency. If the measured value and the end point of a limit line are not at the same frequency, it may happen that the trace slightly crosses a corner of the limit line although there are no limit violations.

Receiving Loudness Rating

The receiving loudness rating (RLR) takes into account the absolute loudness in the receive direction and weights the tones in compliance with the normal sensitivity of the average human ear.

To this end, the frequencies (Hz) of bands 4 to 17 are evaluated according to table 1 of ITU-T P.79.

Table 5 Frequencies (Hz) of bands 4 to 17 according to table 1 of ITU-T P.79

200	1000
250	1250
315	1600
400	2000
500	2500
630	3150
800	4000

The sensitivity at each frequency is specified as the ratio dBPa/V referenced to the rated internal level, and the receiving loudness rating is calculated according to formula 5-1 of ITU-T P.79.

Due to the input sensitivity tolerance of the CMU speech decoder, the individual sensitivity of the CMU used has to be taken into account in order to calculate the receiving loudness rating (see calibration routines).

The receiving loudness rating depends on the receiving loudness set on the mobile phone under test and, according to 3GPP TS 26.131, should be between -1 dB and +5 dB at a rated loudness setting, with lower dB values corresponding to a higher loudness.

The RLR must not fall below -13 dB when maximum loudness is set on the mobile phone. To prevent damage to the human ear, the maximum receiving loudness must not exceed a certain value.

The measured RLR is indicated in a window in the frequency response display and checked for compliance with these limits. In addition to the numeric value, either PASS or FAIL is displayed.

The general PASS or FAIL information is obtained from the limit check of the frequency response curve and the loudness rating. PASS is output only if both the curve and the loudness value are within tolerances.

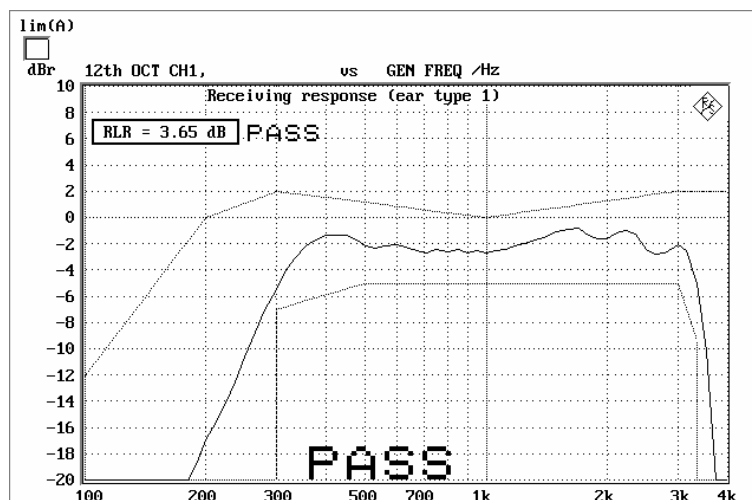


Fig. 4 Receiving frequency response with RLR value displayed

Sidetone Masking Rating (STMR)

The sidetone path is the deliberate output of a part of the signal picked up by the microphone to the phone's receiver. This is to create a natural hearing impression for the person speaking on the phone as is encountered under normal conditions involving an acoustic path between mouth and ear.

The mobile phone under test is installed in the LRGP (ITU-T P.76), and the receiver is sealed to the artificial ear.

The STMR can only be measured according to standard using ear type 1 or ear type 3.2 (low leakage). Measurements with other ear types can be performed for comparison.

The artificial mouth generates tones with a sound pressure of -4.7 dBPa at the MRP (mouth reference point), and the sound pressure is measured in the artificial ear.

The attenuation of the sidetone path is determined at each frequency according to table 1 of ITU-T P.79, and the sidetone masking rating (STMR) is calculated according to formula 5-1 of ITU-T P.79 with the weighting factors of table 3 of ITU-T P.79 taken into account. In addition, the attenuation of the sidetone path is displayed as a curve.

When the phone is set to rated receiving loudness, the STMR should be within 13 dB and 23 dB according to 3GPP TS 26.131.

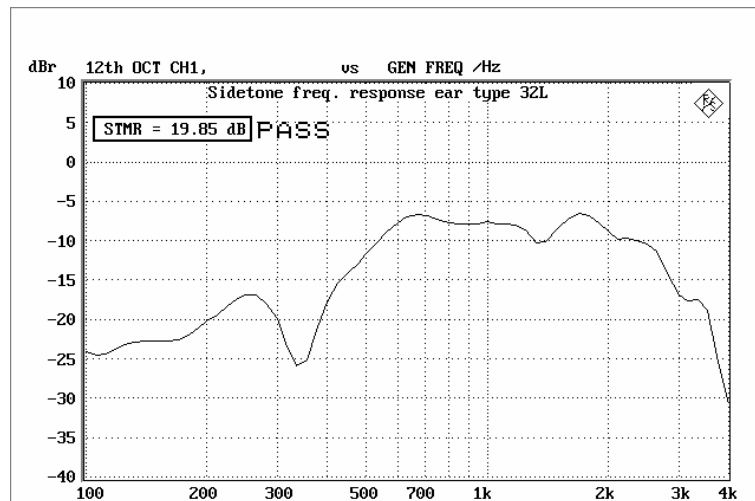


Fig. 5 Typical measurement of sidetone masking rating

Echo Loss (TCLw)

The echo loss is the attenuation between the speech coder input and the speech decoder output (gain of speech coder + decoder = 1). Normally the echo loss is caused by internal acoustic coupling between the telephone receiver and the microphone. Since the echo considerably reduces the sound transmission quality, it must not exceed a certain value.

The mobile phone under test is suspended in free air in the anechoic test chamber (previous specifications used the LRGP).

A modulated multitone signal according to ITU-T P.501 is generated as a test signal and applied to the speech coder. First, the spectral energy distribution of the generated signal is measured in the third-octave bands from 200 Hz to 4 kHz. Then, the spectral distribution in the output signal of the speech decoder is measured. The echo loss is calculated from the differences of the bands according to ITU-T G.122. As an option, the mobile phone under test can be fed for approx. 10 s with the male and female version of artificial voice according to ITU-T P.50 prior to this measurement. This training sequence is to facilitate optimization for potential echo cancellers.

The actual gain of the speech coder and decoder must also be considered in the result. This value is available in the CMU after calibration of the coder. In addition, the attenuation of the echo path is displayed as a curve.

3GPP TS 26.131 specifies an echo loss of at least 46 dB; mobile phones with good echo cancellers can meet this requirement. Since the microphone of the mobile phone under test also picks up any side noise and treats it like an echo, it is essential that the test chamber is well shielded against external noise.

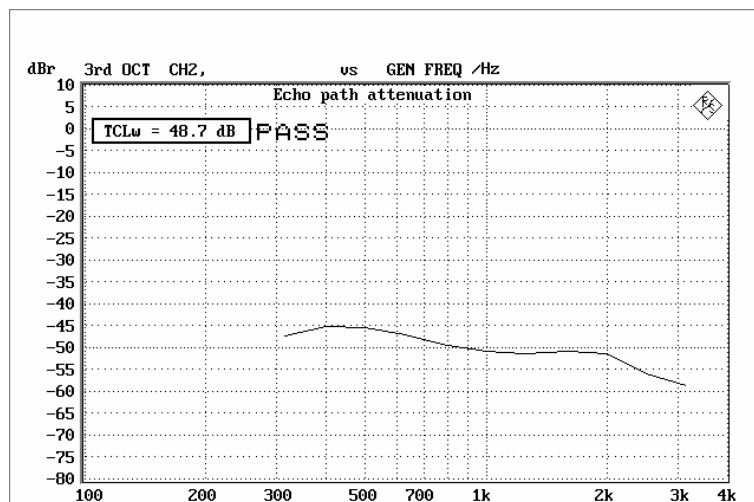


Fig. 6 Typical result of echo loss measurement

Stability Margin

The stability margin is measured to test the susceptibility of the phone to acoustic feedback and instability.

For the test, the telephone is placed on an even, hard board with the receiver and microphone pointing downwards.

A loop is closed in the UPL between the receiving and the voice channel, and an overall gain of 6 dB is set. The gain of the coder is automatically taken into account (see also Echo Loss).

To activate the loop, a noise signal of -10 dBm0 in line with ITU-T O.131 is applied for 1 s and then switched off, with the loop remaining closed.

The test person has to listen to determine whether any resonances or oscillations are produced. If there are no oscillations, the minimum requirements according to 3GPP TS 26.131 for a stability margin of 6 dB are met.

Sending Distortion

The S/N ratio in the transmit path is measured as a function of the sound level.

A pulsed sinusoidal tone of 1015 Hz with a pulse length of approx. 340 ms is used for the measurement. At this frequency, coding yields a sufficiently stable output signal. Voice recognition continues to be active in the mobile phone under test due to this pulsating signal.

The mobile phone under test is installed in the LRGP (ITU-T P.76), and the receiver is sealed to the artificial ear.

The test signal is generated with the artificial mouth at the MRP (mouth reference point) and the SINAD value of the received signal is measured at the CMD decoder output.

The acoustic reference level (ARL) is defined as the sound pressure which creates a signal level of -10 dBm0 in the transmit channel. An automatic routine varies the sound pressure at the artificial mouth until the desired level is attained. This value is then used as a reference for determining the SINAD value versus level.

The SINAD value is measured at sound pressures between -35 dB and +10 dB relative to the acoustic reference level (ARL) and compared with the limits specified in table 7 of 26.131 of 3GPP.

In modern mobile phones, the measured trace may be below the limit line at low levels, causing FAIL to be displayed. The reason for this is that at low levels, the modulation recognition does not function adequately as a voice signal. This behaviour is indicated by a typical steep rise of the signal level above the limit value of -20 dB. Standardization committees are therefore currently working on a modified measurement. In the meantime, the behaviour must be individually evaluated from the trace. Consequently, measurement of the sending distortion has been suspended as a required type approval measurement, until a new test specification is defined.

Table 6 Limit lines specified in table 7 of 3GPP TS 26.131

dB relative to ARL	Level ratio
-35 dB	17.5 dB
-30 dB	22.5 dB
-20 dB	30.7 dB
-10 dB	33.3 dB
0 dB	33.7 dB
7 dB	31.7 dB
10 dB	25.5 dB

The measurement is performed up to a maximum sound pressure of 10 dBPa at the artificial mouth if the value 10 dB relative to ARL with 10 dBPa cannot be attained. The actual trace may therefore end at a lower pressure. This occurs for mobile phones under test which have a low sensitivity in the transmit direction.

If the measured trace is above the limit line, PASS is output, otherwise FAIL is displayed.

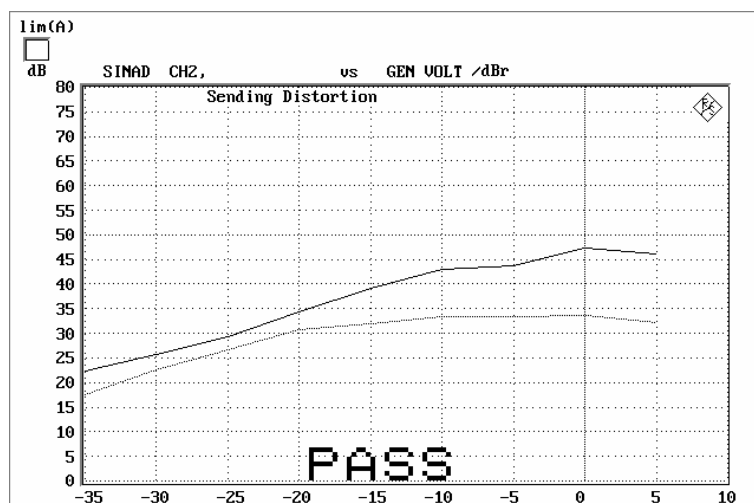


Fig. 7 Sending distortion measurement

Receiving Distortion

The S/N ratio in the receiving path is measured as a function of the acoustic signal level.

A pulsed sinusoidal tone of 1015 Hz is used for the measurement. At this frequency, coding yields a sufficiently stable output signal. Voice recognition continues to be active in the mobile phone under test due to this pulsating signal.

The mobile phone under test is installed in the LRGP (ITU-T P.76), and the receiver is sealed to the artificial ear.

The test signal is applied to the input of the CMU speech coder, and the SINAD value of the sound pressure in the artificial ear is measured with psophometric weighting according to ITU-T G.714.

The SINAD value of the sound pressure is measured at levels between -45 dBm0 and 0 dBm0 and compared with the limits given in table 8 of 3GPP TS 26.131.

Table 7 Limit lines specified in table 8 of 3GPP TS 26.131

Level	Level ratio
-45 dBm0	17.5 dB
-40 dBm0	22.5 dB
-30 dBm0	30.5 dB
-20 dBm0	33.0 dB
-10 dBm0	33.5 dB
-3 dBm0	31.2 dB
0 dBm0	25.5 dB

The measurement is performed up to a maximum sound pressure of 10 dBP a in the artificial ear; the actual trace may end at an earlier point.

If the measured trace is above the limit line, PASS is output, otherwise FAIL is displayed.

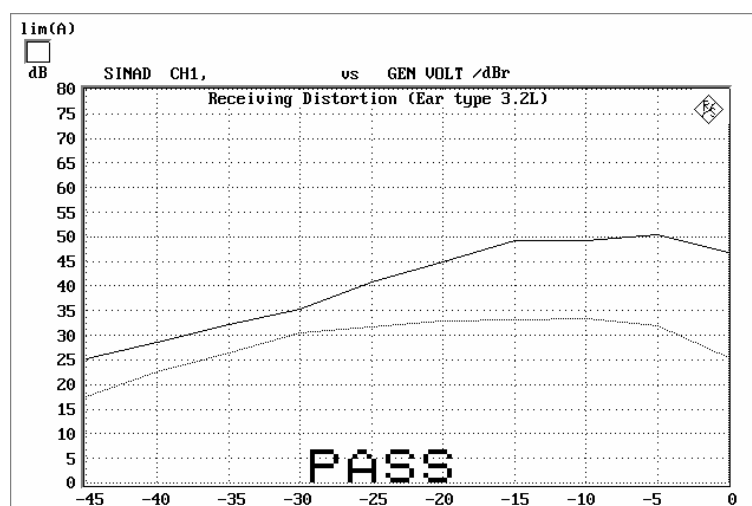


Fig. 8 Typical result of receiving distortion messung

Idle Channel Noise Sending

The noise voltage at the speech decoder output is measured with the phone set up in a quiet environment (<30 dB(A)).

The mobile phone under test is installed in the LRGP (ITU-T P.76), and the receiver is sealed to the artificial ear.

The decoder output voltage is measured, weighted psophometrically according to ITU-T G.223 and recalculated for the internal level in dBm0p.

To keep the mobile phone under test in the normal operating mode, a pulsed signal is applied. The noise level is measured during the signal pauses. The voice activity decoder (VAD) is activated and the mobile phone remains in the active normal sending mode.

The idle noise level should not exceed -64 dBm0p. The measured noise voltage is also displayed as a spectrum, making it easier to find causes if the limit value is exceeded.

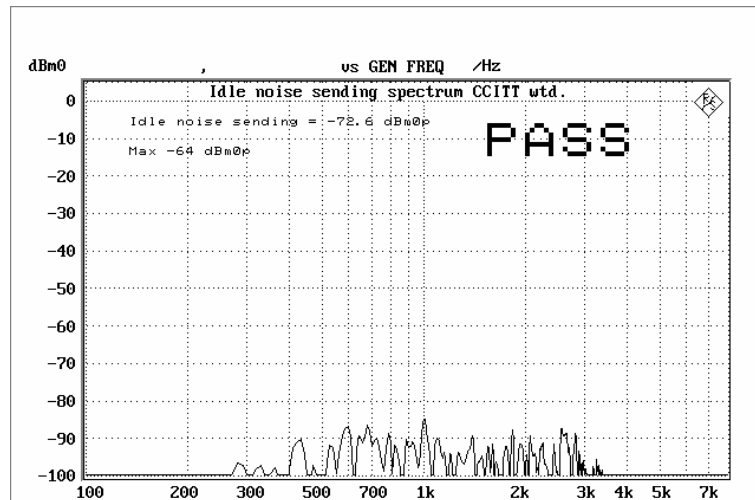


Fig. 9 Idle noise sending

Idle Channel Noise Receiving

The sound pressure in the artificial ear is measured with the phone set up in a quiet environment.

The mobile phone under test is installed in the LRGP (ITU-T P.76), and the receiver is sealed to the artificial ear.

The sound pressure in the artificial ear is measured with A-weighting on.

To keep the mobile phone under test in the normal operating mode, a pulsed signal is applied to the speech coder input. The noise level is measured during the signal pauses. The voice activity decoder is activated and the mobile phone remains in the active normal receiving mode.

With rated loudness set on the mobile phone, the sound pressure should not exceed -57 dBPa(A).

At maximum receiving loudness, the sound pressure should not exceed -54 dBPa(A). The measured noise voltage is also displayed as a spectrum, making it easier to find causes if the limit value is exceeded.

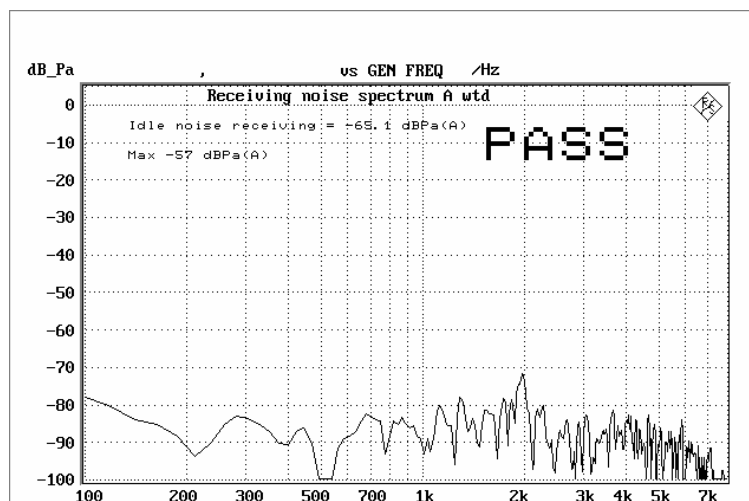


Fig. 10 Idle noise receiving

This measurement makes high demands on the sound insulation of the test chamber and the S/N ratio of the measuring microphone including preamplifier in the artificial ear. A comparison measurement with the test mobile phone switched off or without a DUT shows the measurement reserves of the test equipment. Due to the inherent noise of the Audio Analyzer UPL, measurements can be made to about -80 dBPa(A) at 0 dB microphone gain, and even to lower values when a higher microphone gain is set.

Ambient Noise Rejection

Ambient noise rejection describes the weighted ratio of voice signal transmission to noise in the environment. An ANR value >0 dB means that voice as the useful signal is transmitted more loudly than any ambient noise. The minimum requirement according to 3GPP TS 26.131 is $\text{ANR} > -3$ dB. A value ≥ 3 dB should at least be attained.

To perform this measurement, a homogeneous noise field for simulating the noise in the environment has to be generated. This sound field must be generated by additional loudspeakers and noise generators. To obtain a sufficiently homogeneous sound field, several uncorrelated generators and loudspeakers are required. The use of 2 to 8 generators and loudspeakers is common practice. The noise sources have to generate pink noise ($1/f$). The permissible error in the relevant third-octave bands must be smaller than ± 3 dB with the frequency response of the loudspeakers used also being taken into account.

The measurement of ambient noise rejection is divided into several single measurements.

F5	F6	F7	F8	F9	F10	F11	F12
←	DIST_SND	DIST_REC	IDLE_SND	IDLE_REC	AMB_NOI		→

After selection of AMB_NOI, the following softkey line is displayed:

Ambient noise field				Measurement			
F5	F6	F7	F8	F9	F10	F11	F12
←		ADJUST		START			→

The sound field must be set to the required sound pressure level of 70 dB(A) using ADJUST.

To measure the sound field, the reference microphone is used (as for mouth calibration). Fix the reference microphone to the mouth reference point (MRP) using a suitable support. For the sound field measurement, all other components such as artificial mouth or artificial ear must be removed.

After the ADJUST key has been pressed, a bargraph of the measured A-weighted sound pressure is displayed. It has to be set to a value of 70 ± 1 dB. The tolerance limits are indicated by markers. Moreover, the bargraph colour changes from green to yellow if the value is outside the tolerance limits.

After the CONT key has been pressed, the spectral distribution is measured automatically with an automatic check for compliance with the stipulated absolute level and the pink distribution. If a $1/f$ -weighted noise density (pink noise) is precisely complied with, a third-octave analysis yields identical amplitudes in each third-octave band. The tolerance of ± 3 dB is complied with if the difference between the largest and smallest band measured is less than 6 dB. If this difference is larger, a warning will be output on the screen. A warning will also be output on the screen if the absolute sound pressure is outside the permissible tolerance.

This adjustment and testing of the sound field can practically be regarded as a calibration and has to be repeated only if something changes in the sound field generation. The long-term stability of the noise generators and loudspeakers must of course be sufficient. Otherwise, this measurement routine must be repeated as often as required.

For the actual measurement of ANR, the artificial mouth and the artificial ear must be installed again. The MRP must be installed at the same position as the reference microphone before.

The mobile phone under test is installed in the LRGP (ITU-T P.76), and the receiver is sealed to the artificial ear.

Set up a call to the CMU and set the bit stream to "Handset Low" or the dedicated channel voice to "Speechcodec Low".

Press the START key to start the measurement of the room noise sensitivity.

After completion of the measurement, the request to switch off the noise field will be displayed on the screen. If this is confirmed, the speech sending sensitivity will be measured automatically and the ANR value calculated afterwards.

The ANR value must exceed -3 dB.

Ambient Noise Rejection

ANR = 3.24 dB

PASS

Fig. 11 Typical result of ambient noise rejection measurement

4 Calibration Routines

All calibration data is stored on the UPL hard disk and is therefore automatically available again after every restart. The calibration values for the UPL-B9 option are stored independently of other options.

Calibration of Artificial Ear

Prior to the measurements, the absolute sensitivity of the microphone in the artificial ear must be determined using a sound level calibrator such as the Brüel & Kjær 4231 with a sound pressure level of 94 dB SPL or a sound pressure of 1 Pa at 1 kHz.

Note:

The calibration values of the different ear types are stored separately. Consequently, a calibration need not be performed after a change of the ear type if the physically identical ear has been calibrated before.

Calibration of Ear Type 1

- **Switch off the microphone power supply.**

Note: *The 200 V polarization voltage of the microphone may cause a slight electric shock. The current is harmless, but the microphone preamplifier may be damaged.*

- **Remove the microphone from the artificial ear.**
- **Screw back the microphone capsule and switch on the operating voltage.**
- **Insert the microphone fully into the adapter of the sound level calibrator and switch on the calibrator.**

Note: *After inserting the microphone wait about 10 s to allow for static pressure compensation.*

- **Select the CALIBRATION level with F12 key.**

----CALIBRATION----						EXP-FILES	
F5	F6	F7	F8	F9	F10	F11	F12
←	EAR	MOUTH	CMU-COD				DELETE

- **Call the calibration routines with the EAR key.**

EAR TYPE1		EAR TYPE3.2L		EAR TYPE3.2H		TYPE 3.3		TYPE 3.4	
F5	F6	F7	F8	F9	F10	F11	F12		
BACK	EAR_T1	EAR_T32L	T32L_DAT	EAR_T32H	T32H_DAT	EAR_T33	EAR_T34		

- **Call the test routine using the EAR_T1 key.**

The output voltage of the microphone is measured and the sensitivity displayed with reference to 1 Pa. With 20 dB preamplification of the microphone (recommended value), the displayed sensitivity must be about 10 times the value in the calibration certificate of the microphone

capsule (typical value for microphone capsule 4134 of artificial ear 4185 is approx. 12 mV/Pa, display = 120 mV/Pa). If the voltage measured is below 3 mV, an error message is displayed. Possible error sources are, for example, a switched-off microphone power supply or a disabled calibrator. In this case, the program requests that the test be repeated. After switching on the microphone power supply, wait approx. 20 s before restarting the measurement with RUN.

The reference value measured is stored in a nonvolatile memory and used for all subsequent measurements with the artificial ear type 1.

Calibration of Ear Type 3.2 Low Leakage

- **Connect the sound level calibrator tightly to the artificial ear using the adapter DP0939 and switch on the calibrator.**
- **Select the CALIBRATION level using the F12 key.**

----CALIBRATION-----						EXP-FILES	
F5	F6	F7	F8	F9	F10	F11	F12
←	EAR	MOUTH	CMU-COD				DELETE

- **Call the calibration routines using the EAR key.**

EAR TYPE1		EAR TYPE3.2L		EAR TYPE3.2H		TYPE 3.3	TYPE 3.4
F5	F6	F7	F8	F9	F10	F11	F12
BACK	EAR_T1	EAR_T32L	T32L_DAT	EAR_T32H	T32H_DAT	EAR_T33	EAR_T34

- **Call the test routine using the EAR_T32L key.**

The output voltage of the microphone in the ear is measured and the sensitivity displayed with reference to 1 Pa. If the voltage measured is below 3 mV, an error message is displayed. Possible error sources are, for example, a switched-off microphone power supply or a disabled calibrator. In this case, the program requests that the test be repeated. After switching on the microphone power supply, wait approx. 20 s before restarting the measurement with RUN.

The measured reference value is stored in a nonvolatile memory and used for all subsequent measurements with ear type 3.2L.

Reading the Calibration Data of the Artificial Ear of Type 3.2L:

The frequency response of the artificial ear of type 3.2L is supplied on a floppy together with the artificial ear. The data is used for transforming the measurement values from the drum reference point to the ear reference point.

- **Insert the floppy supplied with the ear into the UPL drive.**
- **Call the routine with the T32L_DAT key.**

The OES_LL.ADA calibration file is automatically searched for and read. The modified data is stored on the UPL hard disk. This procedure need only be repeated after a change of the calibration data, e.g. after a recalibration of the ear by the manufacturer or when a physically different ear of the same type is used.

If the file required is not found on the floppy, the routine requests the user to insert the calibration floppy.

Calibration of Ear Type 3.2 High Leakage

- **Connect the sound level calibrator tightly to the artificial ear using the adapter DP0939 and switch on the calibrator.**
- **Select the CALIBRATION level using the F12 key.**

----CALIBRATION---- ...						EXP-FILES	
F5	F6	F7	F8	F9	F10	F11	F12
←	EAR	MOUTH	CMU-COD				DELETE

- **Call the calibration routines using the EAR key.**

EAR TYPE1		EAR TYPE3.2L		EAR TYPE3.2H		TYPE 3.3		TYPE 3.4	
F5	F6	F7	F8	F9	F10	F11	F12		
BACK	EAR_T1	EAR_T32L	T32L_DAT	EAR_T32H	T32H_DAT	EAR_T33	EAR_T34		

- **Call the test routine using the EAR_T32H key.**

The output voltage of the microphone in the ear is measured and the sensitivity displayed with reference to 1 Pa. If the voltage measured is below 3 mV, an error message is displayed. Possible error sources are, for example, a switched-off microphone power supply or a disabled calibrator. In this case, the program requests that the test be repeated. After switching on the microphone power supply, wait approx. 20 s before restarting the measurement with RUN.

The measured reference value is stored in a nonvolatile memory and used for all subsequent measurements with ear type 3.2H.

Reading the Calibration Data of the Artificial Ear of Type 3.2H:

The frequency response of the artificial ear of type 3.2H is supplied on a floppy together with the artificial ear. The data is used for transforming

the measurement values from the drum reference point to the ear reference point.

- **Insert the floppy supplied with the ear into the UPL drive.**

- **Call the routine with the T32H_DAT key.**

The OES_HL.ADA calibration file is automatically searched for and read. The modified data is stored on the UPL hard disk. This procedure need only be repeated after a change of the calibration data, e.g. after a recalibration of the ear by the manufacturer or when a physically different ear of the same type is used.

If the file required is not found on the floppy, the routine requests the user to insert the calibration floppy.

Calibration of Ear Type 3.3

- **Connect the sound level calibrator tightly to the artificial ear using the adapter UA-1546 and switch the calibrator on.**

- **Select the CALIBRATION level using the F12 key.**

----CALIBRATION---- ...						EXP-FILES	
F5	F6	F7	F8	F9	F10	F11	F12
←	EAR	MOUTH	CMU-COD				DELETE

- **Call the calibration routines using the EAR key.**

EAR TYPE1		EAR TYPE3.2L		EAR TYPE3.2H		TYPE 3.3		TYPE 3.4
F5	F6	F7	F8	F9	F10	F11	F12	
BACK	EAR_T1	EAR_T32L	T32L_DAT	EAR_T32H	T32H_DAT	EAR_T33	EAR_T34	

- **Call the test routine using the EAR_T33 key.**

The output voltage of the microphone in the ear is measured and the sensitivity displayed with reference to 1 Pa. If the voltage measured is below 3 mV, an error message is displayed. Possible error sources are, for example, a switched-off microphone power supply or a disabled calibrator. In this case, the program requests that the test be repeated. After switching on the microphone power supply, wait approx. 20 s before restarting the measurement with RUN.

The measured reference value is stored in a nonvolatile memory and used for all subsequent measurements with ear type 3.3.

The standard calibration data according to ITU-T P57 are used automatically for ear type 3.3.

Calibration of Ear Type 3.4

- Remove the pinna and the ear canal simulator, connect the sound level calibrator tightly to the artificial ear using the short steel adapter and switch the calibrator on.
- Select the CALIBRATION level using the F12 key.

----CALIBRATION---- ...						EXP-FILES	
F5	F6	F7	F8	F9	F10	F11	F12
←	EAR	MOUTH	CMU-COD				DELETE

- Call the calibration routines using the EAR key.

EAR TYPE1		EAR TYPE3.2L		EAR TYPE3.2H		TYPE 3.3		TYPE 3.4	
F5	F6	F7	F8	F9	F10	F11	F12		
BACK	EAR_T1	EAR_T32L	T32L_DAT	EAR_T32H	T32H_DAT	EAR_T33	EAR_T34		

- Call the test routine using the EAR_T34 key.

The output voltage of the microphone in the ear is measured and the sensitivity displayed with reference to 1 Pa. If the voltage measured is below 3 mV, an error message is displayed. Possible error sources are, for example, a switched-off microphone power supply or a disabled calibrator. In this case, the program requests that the test be repeated. After switching on the microphone power supply, wait approx. 20 s before restarting the measurement with RUN.

The measured reference value is stored in a nonvolatile memory and used for all subsequent measurements with ear type 3.4.

The standard calibration data according to ITU-T P57 is used automatically for ear type 3.4.

Calibration of Artificial Mouth

Calibration of the artificial mouth does not depend on the ear type used. A recalibration is therefore not required when the ear type is changed.

Before a mobile phone can be tested, the absolute sensitivity and frequency response of the artificial mouth have to be measured and corrected with the aid of a previously calibrated pressure-field microphone. The measuring microphone removed from artificial ear type 1 can be used for this purpose or an additional microphone capsule is screwed to the microphone preamplifier. The standard microphone is used as a reference for determining the frequency response of the mouth. The frequency response of the microphone can be ignored in the test frequency range (100 Hz to 8 kHz) (see also calibration certificate of microphone capsule).

Since interfering sound falsifies the corrections, the artificial mouth must be calibrated in the sound-proof test chamber.

First of all, the measuring microphone has to be calibrated.

- **Insert the measuring microphone fully into the adapter of the sound level calibrator and switch on the calibrator.**

Note: After inserting the microphone, wait about 10 s to allow for static pressure equalization.

- **Select the CALIBRATION level using the F12 key.**

----CALIBRATION----- ..						EXP-FILES	
F5	F6	F7	F8	F9	F10	F11	F12
←	EAR	MOUTH	CMU-COD				DELETE

- **Call the calibration routines using the MOUTH key.**

MOUTH CALIBRATION							
F5	F6	F7	F8	F9	F10	F11	F12
BACK	REF_MIC	CAL_MOU					

- **Call the test routine using the REF_MIC key.**

The output voltage of the microphone is measured and the sensitivity displayed with reference to 1 Pa. With 20 dB preamplification of the microphone (recommended value), the displayed sensitivity must be about 10 times the value in the calibration certificate of the microphone capsule (typical value for microphone capsule 4134 of artificial ear 4185 is approx. 12 mV/Pa, display = 120 mV/Pa). If the voltage measured is below 3 mV, an error message is displayed. Possible error sources are, for example, a switched-off microphone power supply or a disabled calibrator. In this case, the program requests that the test be repeated.

After switching on the microphone power supply, wait approx. 20 s before restarting the measurement with RUN.

Fit the microphone at right angles to the mouth at the mouth reference point (MRP) using the gauge supplied with the mouth (positioning at right angles is necessary because microphone capsule 4134, e.g. of ear 4185, is pressure-calibrated).

- **Select the CALIBRATION level using the F12 key.**

----CALIBRATION----- ..						EXP-FILES	
F5	F6	F7	F8	F9	F10	F11	F12
←	EAR	MOUTH	CMU-COD				DELETE

- **Call the calibration routines using the MOUTH key.**

MOUTH CALIBRATION							
F5	F6	F7	F8	F9	F10	F11	F12
BACK	REF_MIC	CAL_MOU					

- **Call the test routine using the CAL_MOU key.**

The sound pressure generated at the MRP is set to exactly -4.7 dBPa in an automatic measurement routine at 1 kHz. The generator voltage required is stored in a nonvolatile memory and used as a reference for all subsequent settings with the artificial mouth. If the sound pressure cannot be adjusted to -4.7 dBPa, an error message is displayed with a request to check the connection of the artificial mouth and to repeat the measurement. A possible error source could be that the transformer supplied is not connected between the generator and the artificial mouth.

The uncorrected frequency response of the artificial mouth is measured and displayed. Next, the frequency response is measured with the inverse frequency response correction automatically selected in the generator (equalization). Residual errors caused by nonlinearities of the speaker in the mouth are measured and taken into account in the final equalization file as fine correction.

To verify the results, the absolute sound pressure versus frequency is measured at a sound pressure of 4.7 dBPa (reference value for most of the measurements). The absolute sound pressure at each frequency must be within a tolerance band of -4.7 dBPa ± 0.2 dB. Correct calibration without interfering sound yields an almost straight line in the middle between the two limit lines.

Calibration of CMU Speech Coder

The calibration of the speech coder and decoder is necessary to be able to calculate absolute loudness. Calibration has to be performed only once and must be repeated only if the CMU used is replaced. If the CMU is equipped with model 14 of the Link Handler CMU-B21, GSM as well as WCDMA use the same paths. Calibration of the coder is therefore identical for both operating modes and need not be repeated when switching from GSM to WCDMA or vice versa.

Auxiliary settings required for calibration can be found in the CMU under Bit Stream (for GSM) and under BS Signal, Dedicated Channel, Voice settings (for WCDMA) (firmware version 3.50 or higher). In the case of GSM, this menu is only accessible for an active call. In contrast, with WCDMA the call must first be terminated and then set up again.

- **Select the CALIBRATION level using the F12 key.**

----CALIBRATION---- ..						EXP-FILES	
F5	F6	F7	F8	F9	F10	F11	F12
←	EAR	MOUTH	CMU-COD				DELETE

- **Call the calibration routines with the CMU-COD key.**

The following information is displayed:

Calibration of Coder – Decoder Path in Radiocomm Tester CMU

Please establish call to mobile
and set Bit Stream to 'Decoder Cal'

The following softkey line is displayed:

F5	F6	F7	F8	F9	F10	F11	F12
	CONT						

Set up a call to the mobile phone. Set bit stream or voice setting on the CMU to "Decoder Cal" and then press the CONT key.

The actual voltage at the decoder output of the CMU is now measured for a digital full-scale signal and the required correction value is calculated and saved in the UPL. The following request is then displayed:

Calibration of

Coder – Decoder Path

in Radiocomm Tester CMU

now set Bit Stream to 'Encoder Cal'

The following softkey line is displayed:

F5	F6	F7	F8	F9	F10	F11	F12
	CONT						

After CONT has been pressed, the input sensitivity of the speech coder is measured and the input voltage required for digital full scale is measured at the speech coder and saved in the UPL. At the same time, the loop gain from the speech coder input to the speech coder output is calculated and saved.

5 Processing of Measurement Results

Printing, Storing and Displaying of Measurement Results

All results, the log file as well as all traces and pictures saved by keystroke are stored in a specific directory. The default directory is C:\3GPP\RESULTS. This directory is displayed when the program is started. An additional subdirectory for saving the test results can be generated using the keyboard to sort the results better.

If you key in "a:" then all results are stored directly on a floppy disc.

IMPORTANT: *The program does not check the available space on the floppy disk. An error can occur if the floppy is removed or full.*

The result of each measurement is graphically or numerically displayed on the screen and, if applicable, a PASS or FAIL verdict is output. All individual numeric results such as loudness rating are automatically appended to a RES_3GPP.LOG result file.

The following softkeys are displayed:

F5	F6	F7	F8	F9	F10	F11	F12
	CONT		ABS-SENS	EXP-FILE	TRC-FILE	PCX-FILE	HARDCOPY

The items ABS-SENS, EXP-File and TRC-FILE are displayed only if associated values are available for storage. The key name will be deleted after storage, allowing the user to see at any time whether the file has already been saved.

Pressing the CONT key brings back the selection menu for the various measurements.

When the ABS_FILE key is pressed, the absolute measured values of the displayed trace are saved in export format in a file. This file has the name ABSxx.EXP, where xx represents a consecutive number (of max. 5 digits). This file can be directly processed by spreadsheet programs such as EXCEL. This item is displayed only after a frequency response measurement.

When the EXP_FILE key is pressed, the displayed trace is saved in export format in a file. This file has the name EXPxx.EXP, where xx represents a consecutive number (of max. 5 digits). This file can be directly processed by spreadsheet programs such as EXCEL.

When the TRC_FILE key is pressed, the displayed trace is saved in ASCII format in a file. This file has the name TRCxx.TRC, where xx represents a consecutive number (of max. 5 digits). This allows processing of measurement results with other programs.

When the PCX_FILE key is pressed, the screen content is copied into a PCX file. This file has the name PICxx.PCX, where xx represents a consecutive number (of max. 5 digits). Thus the measurement results can also be used in word processing programs, for example. To allow also numeric values to be stored as a PCX file, the whole screen content without the softkey line is copied.

Since the TRC, EXP, ABS and PCX files are consecutively numbered, it is useful to copy the files of a measurement sequence, for example, and to save them under a new name. In this case, the original files can be deleted. Thus results can be identified more easily and a mix-up between them avoided.

To this end, a DOS shell can be called after termination of the test program (e.g. with key F5) by entering the command SHELL <RETURN>. The files can be copied or renamed (as standard in the C:\3GPP\RESULTS directory) using DOS commands. Entering EXIT <RETURN> brings back BASIC without the program being deleted. The program can be immediately restarted by entering RUN.

The screen content can be output to a printer by pressing the HARDCOPY key.

The desired printer settings are not selected by the program. The printer configuration selected last and set in the UPL manual mode will be chosen. For this reason, the printer, scaling and format must be manually set once in the OPTION panel of UPL prior to the measurement. It is recommended to select a LOW or MEDIUM resolution and integer scale factors for the printer output. If fractional scale factors (especially values <1) are used, the pixel values are interpolated and the print quality is reduced.

It may be useful to first print a test copy to check the print quality. Contrary to manual operation, no COMMENT line is printed in this case and the program automatically sends a FORM FEED after each print to throw out the hardcopy.

All the purely numerical values are automatically added to the RES_3GPP.LOG result file after each measurement. Thus all numeric measurement results can be called again and evaluated after a measurement sequence has been performed.

As with TRC and PCX files, it may be useful to copy the RES_3GPP.LOG file after a measurement sequence and to rename it. Afterwards the RES_3GPP.LOG file can be deleted. Thus results can be identified more easily and a mix-up between them avoided. To this end, a DOS shell can be called after termination of the test program (e.g. with key F5) by entering the command SHELL <RETURN>. The RES_3GPP.LOG file can then be copied and renamed using common DOS commands.

Entering EXIT <RETURN> brings back BASIC without the program being deleted. The program can be immediately restarted by entering RUN.

Old result files of a measurement sequence can be deleted by means of the DELETE EXP-FILES menu item in the 3GPP_TST program.

- **Select the CALIBRATION level using the F12 key.**

----CALIBRATION----- ..				EXP-FILES			
F5	F6	F7	F8	F9	F10	F11	F12
←	EAR	MOUTH	CMU-COD				DELETE

- **Press the DELETE key.**

A query is displayed asking whether all result files (also RES_3GPP.LOG) are really to be deleted. If the user confirms, all result files are stored under *.OLD, i.e. they are not deleted right away. In a second clear procedure, these backup copies are overwritten.

6 Terminating the Application

As long as the arrow → is displayed below the F12 key, another set of softkeys can be called using F12. With F5, the user can return to the previous set of softkeys, as long as the arrow ← is displayed below the key. If F5 displays END, there is no previous set.

F5	F6	F7	F8	F9	F10	F11	F12
END	SEND	REC_NOM	REC_MAX	LSTR	ECHO	STAB-MRG	→

If END is selected by pressing the F5 key, the following query is displayed:

- **"Do you really want to quit?"**
<Y><N>

Upon confirmation with Y, the program is aborted but not deleted. The softkey line for BASIC is automatically restored.

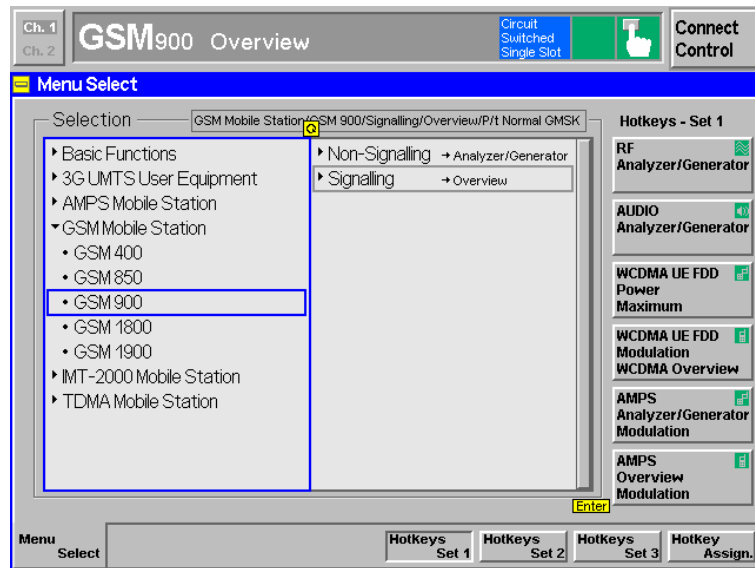
The software can be terminated any time under BASIC with the key combination CTRL BREAK. The program can be continued with CONT and restarted with RUN.

7 Appendix A Settings on the Radio Communication Tester CMU200

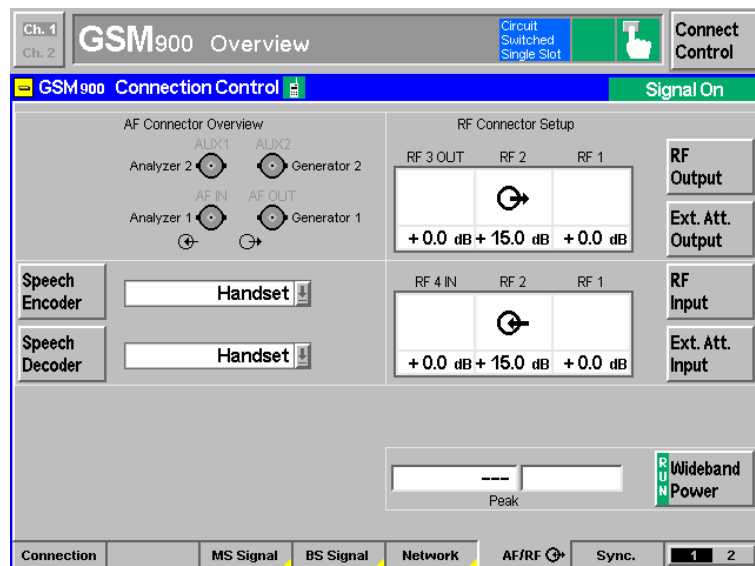
Firmware 3.50

Settings for GSM:

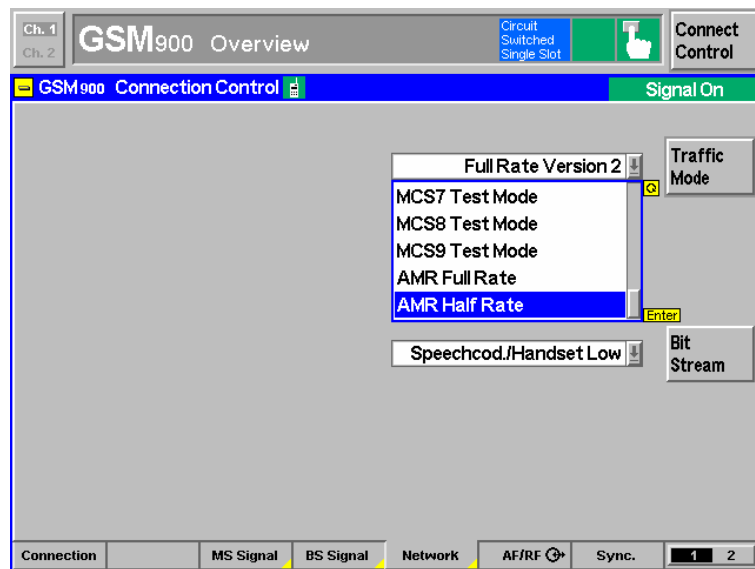
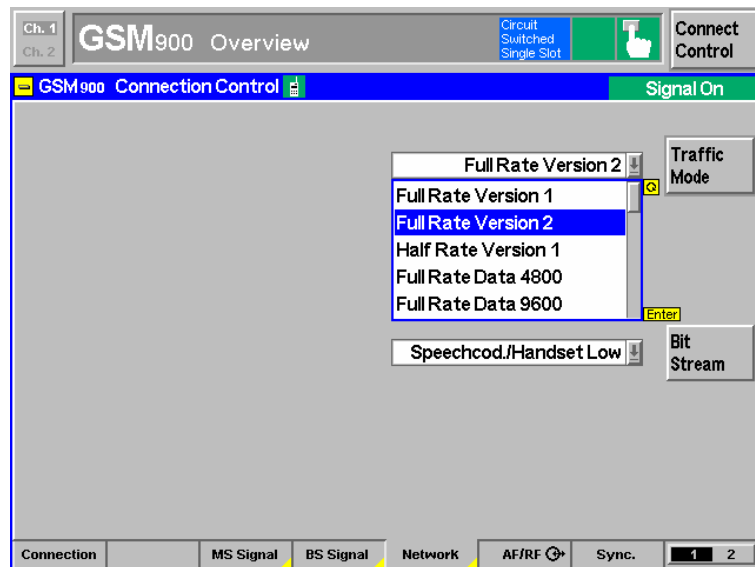
Selection of the GSM band:



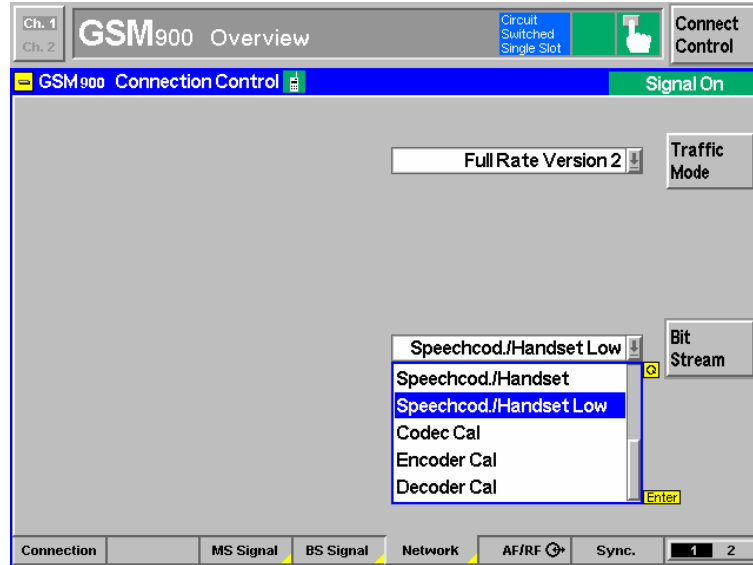
Taking a coupling loss of the antenna coupler into account:



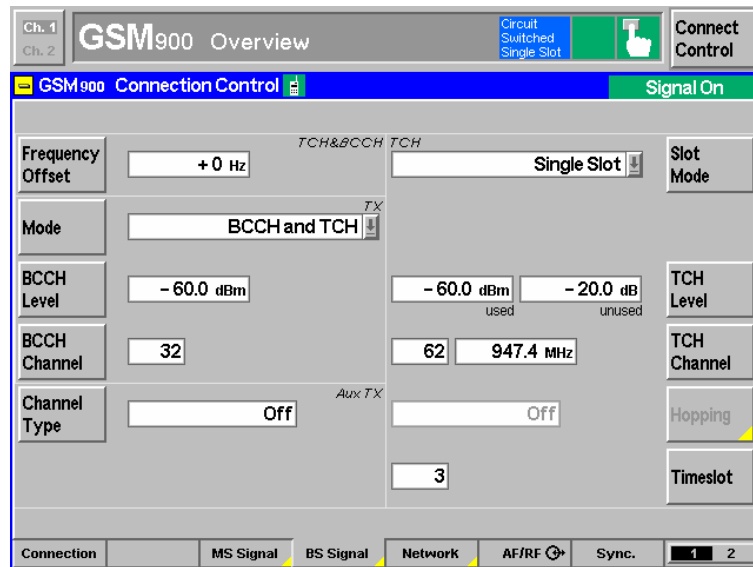
Selection of the FR (full rate version 1), EFR (full rate version 2), HR (half rate version 1), AMR full rate or AMR half rate speech coder:



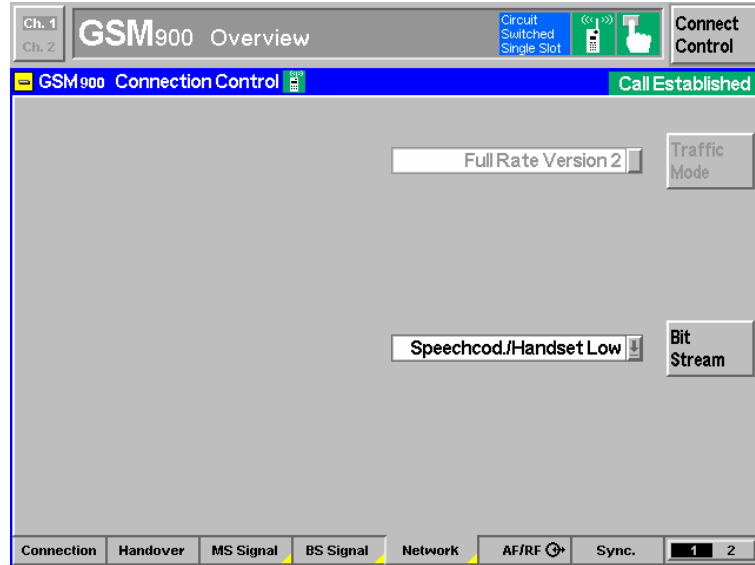
Selection of the Bit Stream Handset Low setting for the measurements or Decoder Cal and Encoder Cal for the calibration:



Setting of the desired TCH and BCCH levels:

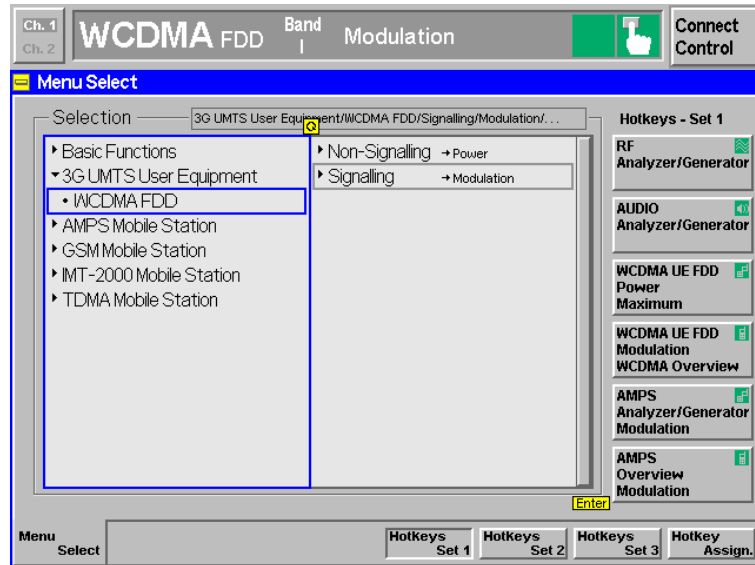


Typical setting with EFR speech coder (corresponds to AMR 12.2) for measurements in the call established status:

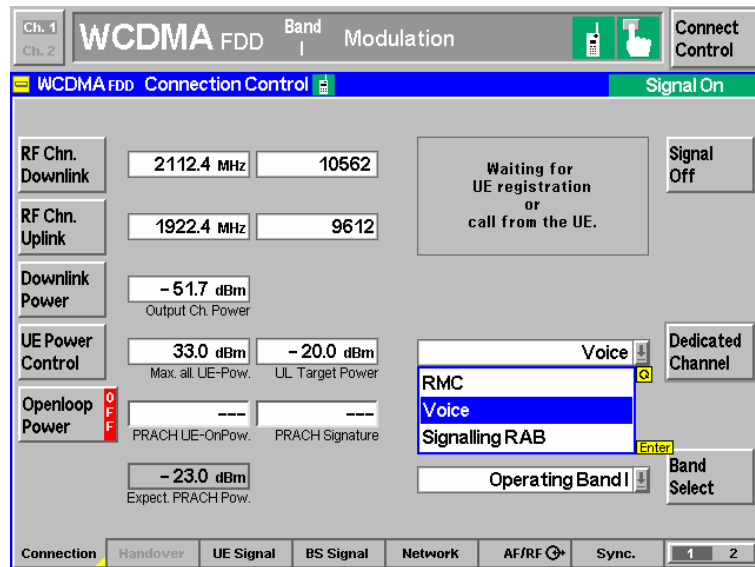


Settings for UMTS WCDMA FDD:

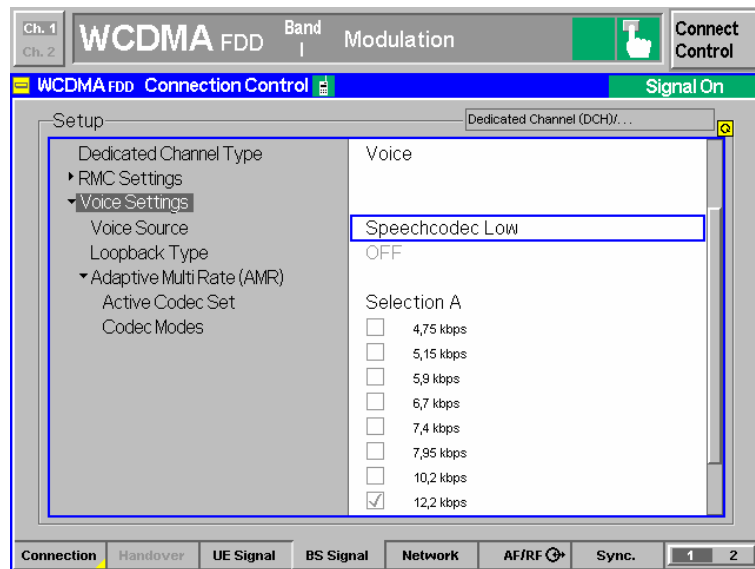
Selection of mode:



Setting dedicated channel to voice:



Selection of the Speechcodec Low setting and AMR Bit Rate for the measurements or Decoder Cal and Encoder Cal for the calibration (selection is possible only if call is deactivated):



Mobile phone registered status, codec settings possible:

The screenshot shows the 'WCDMA FDD Connection Control' window in 'Registered' status. The interface includes fields for RF Chn. Downlink (2112.4 MHz, 10562), RF Chn. Uplink (1922.4 MHz, 9612), Downlink Power (-51.7 dBm), UE Power Control (33.0 dBm, -20.0 dBm), and Openloop Power (-23.0 dBm). A central instruction box says 'Make a call from the UE or press the "Connect UE" key.' The right side has buttons for 'Signal Off', 'Unregister', 'Connect UE', and 'Dedicated Channel'. The bottom status bar shows 'Connection' selected, with other options like 'Handover', 'UE Signal', 'BS Signal', 'Network', 'AF/RF', and 'Sync.'.

Parameter	Value
RF Chn. Downlink	2112.4 MHz, 10562
RF Chn. Uplink	1922.4 MHz, 9612
Downlink Power	-51.7 dBm
UE Power Control	33.0 dBm, -20.0 dBm
Openloop Power	-23.0 dBm

Call active (connected) status for the measurements:

The screenshot shows the 'WCDMA FDD Connection Control' window in 'Connected' status. The interface is similar to the 'Registered' status, but the central instruction box says 'Disconnect the UE by pressing the "Disconnect UE" key.' The right side has buttons for 'Signal Off', 'Unregister', and 'Disconnect UE'. The bottom status bar shows 'Connection' selected, with other options like 'Handover', 'UE Signal', 'BS Signal', 'Network', 'AF/RF', and 'Sync.'.

Parameter	Value
RF Chn. Downlink	2112.4 MHz, 10562
RF Chn. Uplink	1922.4 MHz, 9612
Downlink Power	-51.7 dBm
UE Power Control	33.0 dBm, -20.0 dBm
Openloop Power	-23.0 dBm