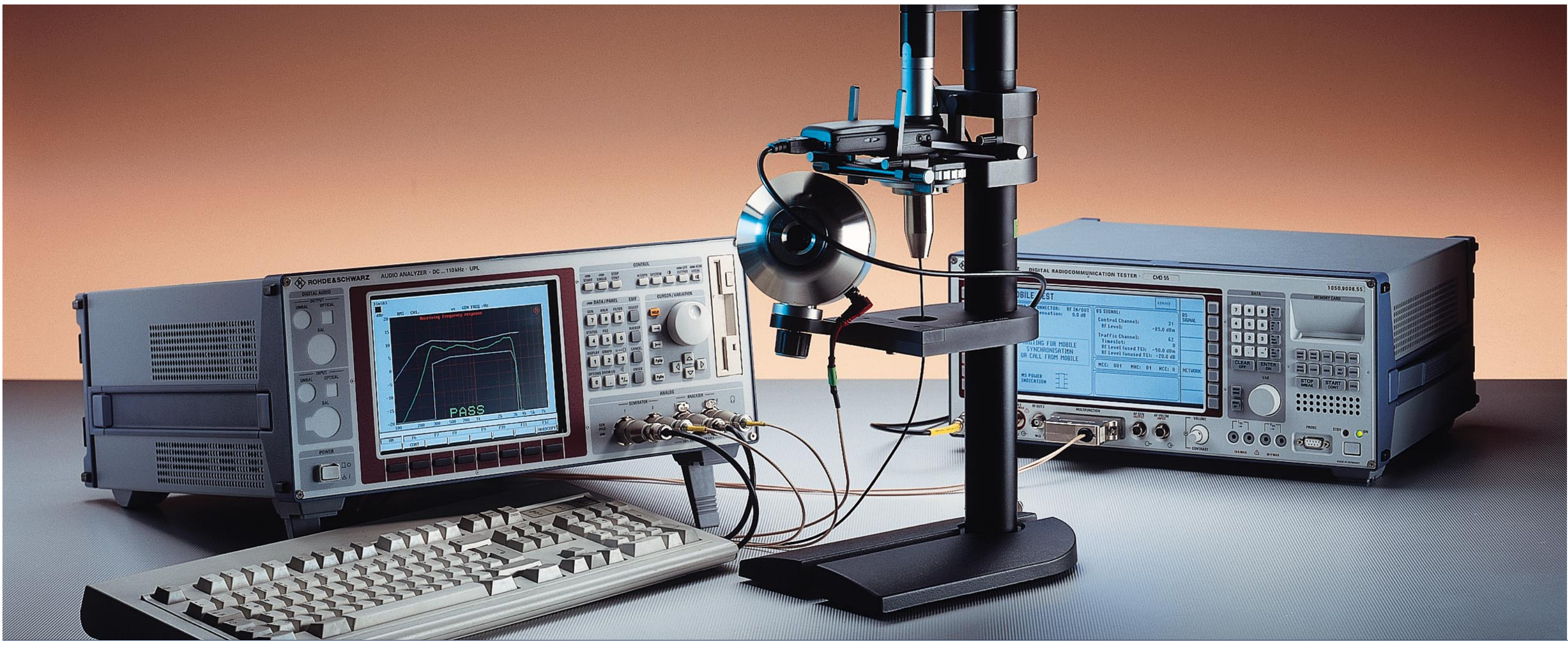

Products: Audio Analyzer UPL Digital Radiocommunication Tester CMD

Acoustic Measurements on GSM Mobile Phones with Audio Analyzer UPL and Digital Radiocommunication Tester CMD

Application Note

This application note describes test methods for commercial GSM mobiles which yield results comparable with those of a type-approval test.

ROHDE&SCHWARZ

Änderungen vorbehalten – Tilman Betz 02.99 – Application Note1GA39_0D

Contents

1 Introduction

The acoustic transmission and reproduction quality of a mobile phone is its most important characteristic in every-day use. The most visually appealing design or a wonderfully sophisticated means of operation are not much use, when the operator user hardly understand what is being said at the other end.

Instruments and procedures for measuring acoustic characteristics are therefore essential tools for determining the quality and suitability of a mobile.

The special Audio Analyzer UPL16 was developed for acoustic measurements for the type approval of GSM mobiles. It performs all audio measurements in line with chapter 30 of GSM 11.10 on special test mobiles which are provided with a digital audio interface (DAI).

There is however great interest in testing mobiles without a DAI. Trade journals, consumer test institutes, or GSM network operators are particularly interested in measuring and comparing acoustic characteristics of commercial mobiles. Network operators, for instance, must be able to check customer complaints or test the quality of supplied phones. A highly accurate test method is also required in the quality assurance of commercial mobiles and for sampling inspection in production.

Mobile Phone Test UPL-B8 of Audio Analyzer UPL is now available for these applications. With the aid of this option all the necessary audio measurements can be performed on conventional GSM mobiles without the DAI interface.

2 Preparation

Required Measuring Instruments and Accessories

The Audio Analyzer UPL with the following options is required :

- Extended Analysis Functions UPL-B6
- Universal Sequence Controller UPL-B10
- Mobile Phone Test UPL-B8

The GSM test mobile is driven by CMD Digital Radiocommunication Tester via the RF interface. CMD simulates a base station for the mobile so that a call can be set up. Depending on the required GSM band, CMD 52, CMD55 or CMD65 is used. The selected CMD must be equipped with the Real-Time Speech Coder/Decoder option CMD-B5.

Acoustic devices such as an artificial mouth, artificial ear and other accessories, are also required for the measurements. The following equipment from Brüel & Kjaer 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)
Artificial Mouth	Special loudspeaker for simulation of the mouth	B&K 4227
Acoustic Calibrator	Sound level calibrator for measuring microphone	B&K 4231
Microphone Power Supply	Power supply and preamplifier for the measuring microphone	B&K 2690A0S2 or G.R.A.S. XYZ

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 at least 20 dB. A low-noise power supply such as XYZ from G.R.A.S is preferable.*

Adapter Set UPL-Z1 using commercial BNC cables is recommended for connecting the CMD accessories to Audio Analyzer UPL.

A cable a with BNC connector and a special angled banana plug is required for connecting the artificial mouth, as the space between the mouth connector and the test rack is too small for common banana plugs.

The transformer supplied with option UPL-B8 is 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 UPL’s generator . Without this transformer the available power is too low for driving the artificial mouth.

Alternatively an amplifier could be connected between generator output and mouth instead of the transformer.

When Adapter UPL-Z1 is used, a cable with male (analyzer) and female (generator) XLR connectors or BNC connectors is required for to connect to the multi-function generator of the Digital Radiocommunication Tester CMD. The cables should be wired as follows:

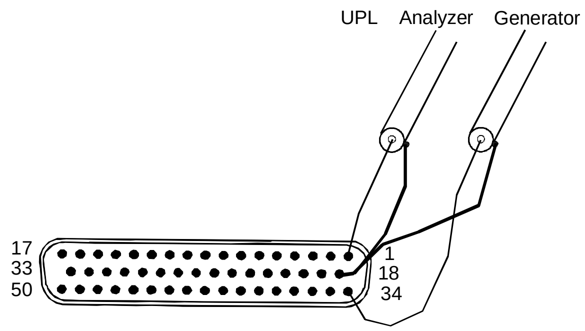

Fig. 1 Assignment of 50-contact multipoint connector at CMD front panel

An external PC keyboard must also be connected to the UPL (large DIN connector). A driver for country-specific keyboards can be defined in the C:\UPL\USERKEYB.BAT file, see the 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 option UPL-B8. The audio analyzer should meet the following firmware requirements:

- UPL firmware version 1.40 or higher
- Extended Analysis Functions option UPL-B6 installed
- Universal Sequence Controller option UPL-B10 installed
- Mobile Phone Test option UPL-B8 installed
  (will be done automatically during the installation of the software)
- 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 be installed with the aid of the PHONINST.BAT installation program on program floppy 1. The installation number of the optional Mobile Phone Test UPL-B8 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 PHONINST)
You are requested to enter the installation number of option UPL-B8

➢ Enter the installation number supplied with option UPL-B8. If the number does not match the serial number of the 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 PHONINST program creates the C:\PHONETST directory in the audio analyzer (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

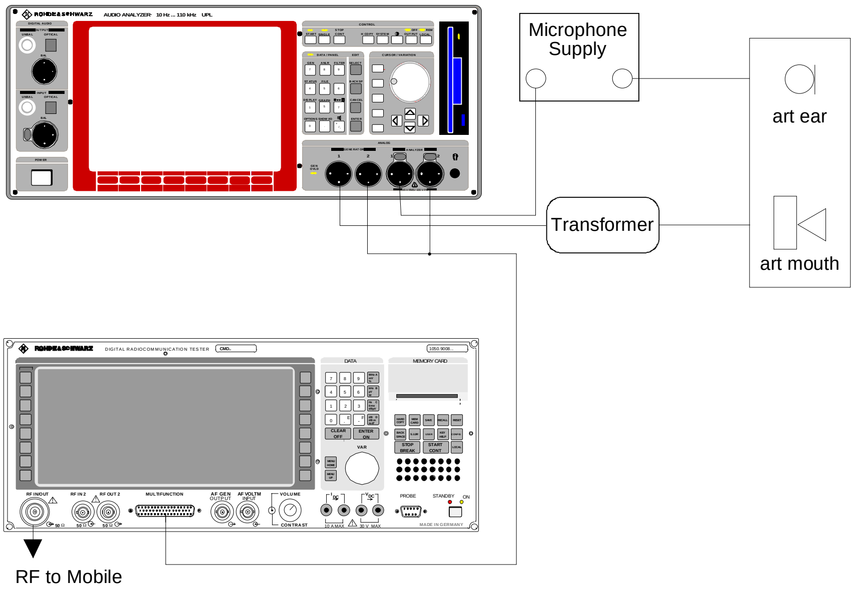

Fig. 2 Test setup and connection of external components

Starting the Application Software

The application program is executed by the automatic sequence control. The audio analyzer is switched to automatic sequence control using the F3 key (on the external keyboard).

The logging function is switched off; check set "logging off" is displayed at the bottom right of the screen; toggle logging on and off with F2. With the logging function on, all commands entered in the manual mode would be appended to the program and so use up memory.

The application programs are called from path C:\PHONETST 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 the mobile
- in the automatic sequence control mode with the BASIC command line UPL OUT "MMEM:CDIR 'C:\PHONETST"
- under BASIC with the SHELL command by entering CD\PHONETST and pressing ENTER
- at DOS level by entering CD\PHONETST.

Program floppy 1 contains the BASIC program GSM_TST.BAS for measurements on GSM mobiles. It is loaded and started by entering:

1. LOAD"GSM_TST"
2. 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, the printer last used by the audio analyzer remains configured. New settings need not therefore be made by the user. It is useful to select the desired printout with type, format, and scaling in 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 guaranteed 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 the delay, the setups can be converted before the application software is started; at DOS level call the UPL conversion program:

DO_CONV \PHONETST

This converts all setups in the PHONETST directory.

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 be used for selecting program routines.

After the program has been started, the title page:

"Measurement of
GSM Mobiles

via Speech Codec
with Audio Analyzer UPL"

and the following softkey line are displayed:

F5	F6	F7	F8	F9	F10	F11	F12
	CONT						

Press F6 CONT, the following message is displayed on the screen:

“*Please establish call to Mobile and set CMD to Speech Mode Handset*“

To do so press the MANUAL TEST key on the CMD and then switch on the mobile. After successful registration, press the CALL TO MOBILE key on the CMD or dial a number on the mobile and press the transmit key.

The following softkey line are displayed:

F5	F6	F7	F8	F9	F10	F11	F12
	CONT						

Press F6 CONT, the following message is displayed:

"*Measurement of GSM Mobiles via Speech Codec with Audio Analyzer UPL*"

"*Select Test to be performed*"

The measurements on the test mobile can now be started as all required calibration values are stored in the UPL.

During the initial installation of the test setup, the microphone in the artificial ear, the artificial mouth and the CMD voice codec have to be calibrated (see “Calibration Routines” on page). In this case the message requesting a call setup to the test mobile can be skipped with CONT.

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

F5	F6	F7	F8	F9	F10	F11	F12
END	SEND	RECEIVE	STMR	LSTR	ECHO	STAB-MRG	->

A click on the respective key starts the test routine. Since there are more selection items than softkeys, the next set of softkey definitions are called with F12.

F5	F6	F7	F8	F9	F10	F11	F12
<-	DIST_SND	DIST_REC	IDLE_SND	IDLE_REC			->

--- CALIBRATION ---

F5	F6	F7	F8	F9	F10	F11	F12
<-	CAL_MIC	CAL_MOU	CODEC	ADJ_SLR	ADJ_RLR		

If F12 shows an arrow, press F12 to see the next set of softkey definitions. Press F5 to go back to the previous set. If F5 shows END, pressing F5 ends the program.

4 Measurements

General

Special problems are encountered when measuring acoustic characteristics, caused by the GSM coder and decoder algorithms.

In type-approval tests, where highly accurate measurements are required, the coder and decoder are excluded from the measurement, as test mobiles are equipped with a digital audio interface DAI for the transmission of audio signals with linear PCM coding. Audio Analyzer UPL16 is also equipped with a DAI interface, so that direct transmission of the test signal to and from the mobile is possible. In commercial mobiles measurements during normal operation can only be performed via the air interface with the voice coder and decoder included. A so-called vocoder is used to attain the lowest possible data rate, only the filter and fundamental parameters required for signal reconstruction are transmitted, not the actual voice.

Standard 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. If, for instance, a tone of approx. 2.5 kHz is applied to the telephone with a constant sound pressure, the amplitude of the signal obtained at the decoder output varies by approx. 20 dB, which makes the signal unsuitable for measurements.

With frequencies up to slightly above 1 kHz, the sinusoidal tone is transmitted with sufficient stability to allow common distortion measurements to be performed at 1 kHz using a sinewave signal.

Sufficient stability throughout the transmission range can only be achieved with test signals simulating the characteristics of the human voice with tones that are harmonic multiples of the fundamental. Whether the results obtained for the fundamental are favourable depends on how far the values coincide with the clock of the coding algorithm. Through a skilful choice of fundamental frequencies, test signals with an overlapping spectral distribution can be generated giving a sufficient number of test points in subsequent measurements at different fundamental frequencies so that a practically continuous frequency response curve is obtained. Evaluation is by means of FFT analysis with a special window function and selection of result bins.

After sorting and smoothing, the result is displayed as a frequency response curve and, depending on the measurement, the sending and receiving loudness rating is calculated in line with CCITT PI79 and indicated in the graphics display. As with type-approval measurements via the DAI interface, the measured frequency response in the transmit and receive direction is checked for compliance with the limits specified by GSM 11.10, and a PASS or FAIL verdict is issued as appropriate.

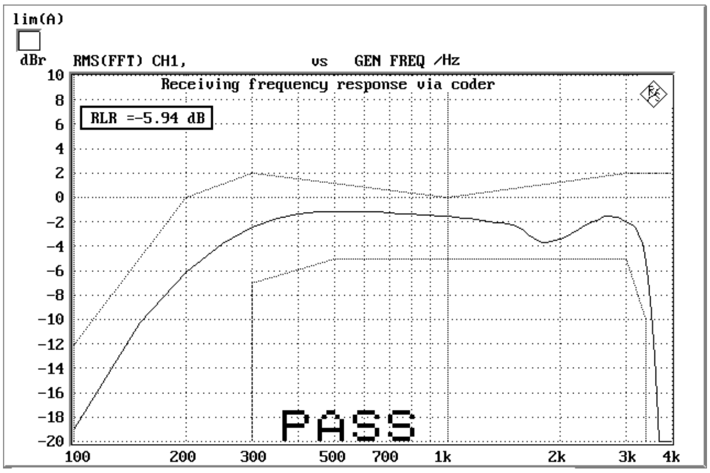

Fig. 3 Example results display with RLR values and PASS verdict

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 isolation against interfering sound. Since special distortion measurements and particularly the measurement of idle noise set high demands on measurement conditions, the A-weighted noise in the test chamber should 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				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 transmit 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 under test is installed in the LRGP position (loudness rating guard ring position to CCITT P.76) and the speaker 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 CMD’s voice decoder output and evaluated.

The transmit frequency response must be within the tolerances specified by GSM 11.10, table 30.1. The absolute sensitivity is not yet taken into account.

Table 1 Tolerances specified by GSM 11.10, table 30.1

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 curve 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, a PASS is output, if not, FAIL is displayed. The limit check is performed at each measured frequency. If the measured value and the end point of the limit curve are not at the same frequency, it may happen that the trace slightly crosses a corner of the limit curve 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 of bands 4 to 17 are evaluated according to table 2 of CCITT P.79.

Table 2 Frequencies of bands 4 to 17 according to table 2 of CCITT P.79.

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

Due to multitone analysis, the above frequencies may shift slightly. The maximum deviation of the individual frequencies from the rated values is 5 %, the resulting errors are negligible.

The sensitivity at each frequency is defined as the ratio dBV/Pa referring to the rated internal level in dBm0, and the sending loudness rating is calculated according to formula 4.19b of CCITT P.79. The result is corrected by a total of -0.3 dB according to table 3 of CCITT P.79.

Due to the inevitable input sensitivity spread of the CMDs coder, there is a degree of uncertainty in the calculation of the sending loudness rating. The sensitivity of the CMD can be taken into using of a special tuning routine, is a test mobile with known SLR is used for adjustments (see “calibration routines” on page).

According to GSM 11.10 the sending loudness rating should be between 5 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 but not checked for compliance with limits.

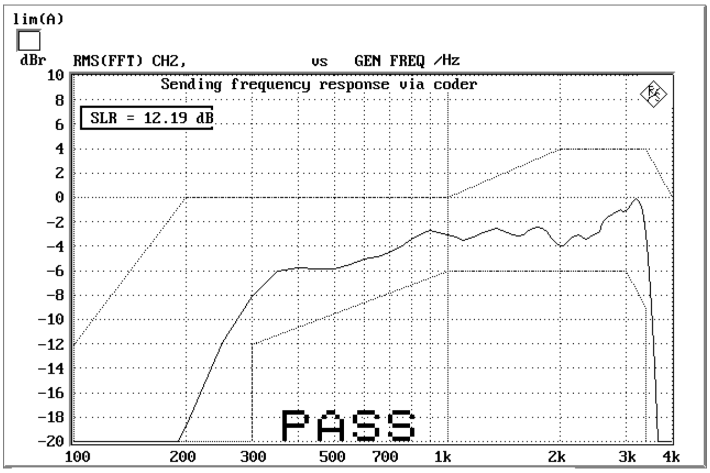

Fig. 4 Sending frequency response with SLR values

Receiving Frequency Response and Loudness Rating

Receiving Frequency Response

The input frequency response is specified as the transmission ratio in dB of the sound pressure in the artificial ear to the input voltage at the voice coder input of CMD.

The mobile under test is installed in the LRGP position (CCITT P.76) and the speaker is sealed to the artificial ear.

The voice coder is driven so that tones with an internal reference level of -16 dBm0 are obtained. The noise pressure in the artificial ear is measured and evaluated.

The receiving frequency response must be within the limit lines specified in table 30.2 of GSM 11.10. The absolute sensitivity is not yet taken into account.

Table 3 Limit lines according to GSM 11.10 table 30.2

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 when a straight line is drawn between the specified values and a logarithmic frequency scale and a linear dB scale are used.

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

Receiving Loudness Rating

The receiving loudness rating (SLR) 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 2 of CCITT P.79.

Table 4 Frequencies (Hz) of bands 4 to 17 according to table 2 of CCITT P.79

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

Due to multitone analysis, the above frequencies may shift slightly. The maximum deviation of the individual frequencies from the rated values is 5 %, the resulting errors are negligible.

The sensitivity at each frequency is specified as a ratio in dBPA/V referring to the rated internal signal level, and the receiving loudness rating is calculated according to formula 4.19c of CCITT P.79 with correction of the ear sensitivity according to table 4 of CCITT P.79.

Due to the inevitable input sensitivity spread of the CMD's voice coder, there is degree of uncertainty in the calculation of the sending loudness rating. The sensitivity of the CMD can be taken into account using a special tuning routine if a test mobile with known RLR is used for adjustments (see “calibration routines” on page).

The receiving loudness rating depends on the volume set on the test mobile and, according to GSM 10.11, should be between -1 V and +5 V at a rated volume setting, with lower dB values corresponding to a higher volume.

The RLR should not fall below -13 dB when maximum volume is set on the phone (i.e. the maximum receiving loudness should not exceed a certain value to avoid damage to the human ear). The measured SLR is indicated in a window in the frequency response display but not checked for compliance with limits.

Sidetone Masking Rating STMR

The so-called sidetone path is the desired output from the part of the signal picked up by the microphone from the phone's speaker. This should create a natural hearing impression for the person speaking on the phone as is encountered under normal call conditions, i.e. via the acoustic path between his mouth and ear.

The mobile under test is installed in the LRGP position (CCITT P.76) and the speaker is sealed to the artificial ear.

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 suppression of the sidetone path is determined at each frequency according to table 2, CCITT P.79, and the side tone masking rating STMR calculated according to formula 8.4 of CCITT P.79 with the weighting factors of tables 6 and 4 of CCITT P.79 taken into account.

When the phone is set to the rated volume, the STMR should be within 8 and 18 dB.

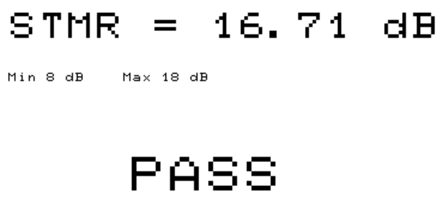

Fig. 5 Display of numeric values on the screen, e.g. for sidetone masking rating

Listener Sidetone Rating LSTR

The listener sidetone rating defines the effect of interference sound on the voice quality. The telephone microphone not only picks up the caller's voice but also any noise in the environment. The listener sidetone rating is the ratio of the wanted to the unwanted sound. For measuring the LSTR, a standard sound field is required, which is created with the aid of eight noise generators producing pink noise with a sound pressure of 70 dB(A). Provided the eight sources are adequately arranged in the test chamber, a homogeneous sound field is obtained in the center. Refer to GSM 11.10, section 30.5.2.4.2, for information on setup and levels.

Since a standard sound field is required for the LSTR measurement, the measurement is far more involved than most of the other acoustic measurements, but the complexity is necessary to test for the effects of interfering sound on the transmission quality. An automatic evaluation is included in the test program.

The sound field has to be created by means of external generators. The test program determines the listener sidetone rating on the assumption that the sound field is in compliance with the standard.

The mobile under test is installed in the LRGP position (CCITT P.76) and the speaker sealed to the artificial ear.

The setup is installed in a chamber with a standard sound field. The energy distribution in this field is defined by GSM 11.10 and therefore known.

The energy of the sound pressure in the artificial ear is measured by means of third-octave analysis in the 14 bands with center frequencies from 200 to 4000 Hz, and the suppression of the listener sidetone path is determined for each band from the known rated values of the sound field. The listener sidetone rating LSTR is then calculated with formula 8.4 of CCITT P.79 by taking into account the weighting factors of tables 6 and 4 of CCITT P.79.

The LSTR should not be less than 15 dB.

Echo Loss

The echo loss is the attenuation between the voice coder input and the voice decoder output (gain of voice 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 should not exceed a certain value.

To obtain realistic results, an artificial voice is used for the echo loss test. The currently applicable GSM 11.10 standard does not take into account that the RMS value (referred to the peak level) of the stochastic signal of the artificial voice is considerably lower than that of previously used test signals with sinusoidal tones (crest factor of voice signals approx. 20 dB compared with 3 dB of sinewave signal). As a result the system can only be driven at low level and the demanded echo loss value of 46 dB corresponds to about the theoretical quantization noise of the GSM system. For this reason a PASS verdict will normally not be issued from this test. Although these measurements are of great importance from an acoustic point of view, the echo loss measurement in the type approval test of GSM mobiles is currently suspended until a new definition is available.

The numeric values obtained in the echo loss measurement may nevertheless be used for a quality assessment of the telephone.

The mobile under test is installed in the LRGP position (CCITT P.76) and the speaker is sealed to the artificial ear.

The artificial voice defined by CCITT P.50 is generated as a test signal and applied to the voice coder. The voice is sent for 10 seconds. During this period the spectral energy distribution of this signal is measured in the third-octave bands from 200 Hz to 4 kHz. The same part of the voice is sent again for 10 seconds and the spectral distribution in the output signal of the voice decoder is measured. The echo loss is then calculated from the differences of the bands according to CCITT G.122. This measurement is performed for a male and a female voice and the final result is the mean value of the two measurements.

The real gain of the voice coder and decoder must also be considered in the result. The gain of the coder path is be determined using of an auxiliary routine (see “calibration routines” on page).

GSM 11.10 specifies an echo loss of at least 46 dB. Normally this cannot be achieved for the above reasons, but values of up to approximately 40 dB can be expected. Since the microphone also picks up any side noise and treats it like an echo, it is essential that the test chamber is shielded against external noise.

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 receive and the voice channel and an overall gain of 6 dB 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 CCITT O.131 is applied for 1 seconds and then switched off with the loop remaining closed.

Listening for whether resonances or oscillations are produced. If there are no oscillations, the minimum requirements to GSM 11.10 for a stability margin of 6 dB are complied with.

Sending Distortion

The S/N ratio in the transmit path is measured as a function of the sound level. As specified by GSM 11.10, the voice coder is excluded from the measurement, but when a standard GSM mobile is used, this measurement can only be performed with the voice coder and decoder included. For this reason the limit values to GSM 11.10 may be taken as a reference but they need not necessarily be adhered to.

A sinusoidal tone of 1015 Hz is used for the measurement. At this frequency, coding yields a sufficiently stable output signal.

The mobile under test is installed in the LRGP position (CCITT P.76) and the speaker 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's 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 limit lines specified in table 30.3 of GSM 11.10.

Table 5 Limit lines specified in table 30.3 of GSM 11.10

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 so the actual trace may end at an lower pressure

If the measured trace is above the limit line, a PASS is output otherwise a FAIL is displayed. Since this measurement includes voice coding in contrast to type approval tests, a PASS verdict cannot always be expected. In this case the offset of the trace to the limit line has to be visually checked.

Receiving Distortion

The S/N ratio in the receive path is measured as a function of the acoustic signal level. As specified by GSM 11.10 the voice coder is excluded from the measurement, but when a standard GSM mobile is used, the measurement can only be performed with the voice coder and decoder included. For this reason the limit values to GSM 11.10 can be taken as a reference but they need not necessarily be adhered to.

A sinusoidal tone of 1015 Hz is used for the measurement. At this frequency, coding yields a sufficiently stable output signal.

The mobile under test is installed in the LRGP position (CCITT P.76) and the speaker is sealed to the artificial ear.

The test signal is applied to the input of the CMD voice coder and the SINAD of the sound pressure in the artificial ear is measured with psophometric weighting to CCITT G.714.

The SINAD of the sound pressure is measured at levels between -45 dBm0 and 0 dBm0 and compared with the limit lines given in table 30.4 of GSM 11.10.

Table 6 Limit lines given in table 30.4 of GSM 11.10

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 dBPa in the artificial ear, so that the actual trace may end at a lower pressure.

If the measured trace is above the limit line, a PASS is issued otherwise a FAIL. Since this measurement includes voice coding in contrast to the type approval test, a PASS verdict cannot always be expected. In this case the offset of the trace to the limit line has to be visually checked.

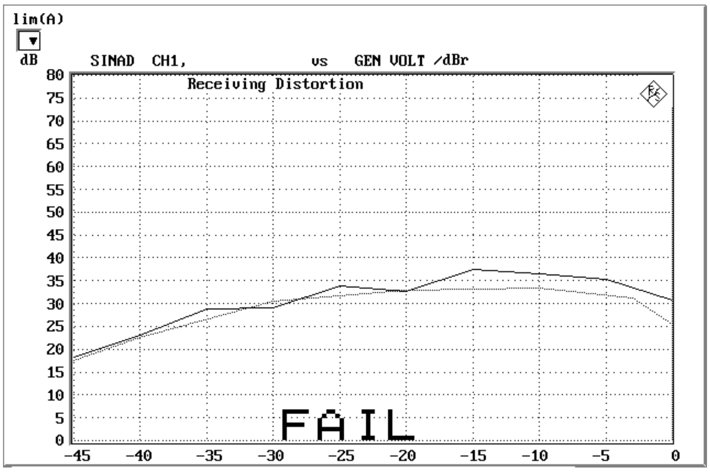

Fig. 6 Typical result of receiving distortion measurement

Idle channel noise sending

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

The mobile under test is installed in the LRGP position (CCITT P.76) and the speaker is sealed to the artificial ear.

The decoder output voltage is measured, psophometrically weighted according to CCITT G.223 and calculated at the internal level in dBm0p.

The idle noise level should not exceed -64 dBm0p.

Idle channel noise receiving

The sound pressure in the artificial ear is measured with the phone set up in a quiet environment (<30 dB(A)).

The mobile under test is installed in the LRGP position (CCITT P.76) and the speaker is sealed to the artificial ear.

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

With optimum volume set on the mobile, the sound pressure should not exceed -57 dBPa(A).

At maximum volume, the sound pressure should not exceed -54 dBPa(A).

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 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.

Thus the compliance with the interfering noise level of below 30 dB(A) as specified by GSM 11.10 for the test chamber can be checked. 30 dB(A) corresponds to -64 dBPa(A) (1 Pa corresponds to a sound pressure level of 94 dB).

5 Calibration Routines

Microphone Calibration

Before a mobiles can be rested, the absolute sensitivity of the microphone in the artificial ear is determined using a sound level calibrator such as 4231 from Brüel & Kjaer with a sound pressure level of 94 dBSPL or a sound pressure of 1 Pa at 1 kHz.

1. After switching 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 amplifier may be damaged*

2. Remove the microphone from the artificial ear.

3. Screw back the microphone capsule and switch on the operating voltage.

4. 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.*

1. Select the CALIBRATION level with function key F12.

   --- CALIBRATION ---

F5	F6	F7	F8	F9	F10	F11	F12
<-	CAL_MIC	CAL_MOU	CODEC	ADJ_SLR	ADJ_RLR		

2. Call the test routine with function key F6 CAL_MIC.

The output voltage of the microphone is measured and the sensitivity displayed with reference to 1 Pa. With 0 dB preamplification of the microphone, the displayed sensitivity should be about the value in the calibration certificate of the microphone capsule (typical value for microphone capsule 4134 and artificial ear 4185 is approx. 12 mV/Pa). If the measured voltage is below 3 mV, an error message is displayed. Possible error sources may be a switched off microphone power supply or a disabled calibrator. In this case the program requests the test to be repeated. After switching on the microphone power supply wait for 20 seconds before restarting the measurement with RUN.

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

Artificial Mouth Calibration

Before a mobiles can be rested, the absolute sensitivity and frequency response of the artificial mouth have to be measured and corrected with the aid of the removed and previously calibrated measuring microphone of the artificial ear. The measuring 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 should be calibrated in a sound-proof test chamber.

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

- Press F12 to display the calibration keys:

--- CALIBRATION ---

F5	F6	F7	F8	F9	F10	F11	F12
<-	CAL_MIC	CAL_MOU	CODEC	ADJ_SLR	ADJ_RLR		

- Call the test routine by pressing function key F7 CAL_MOU.

  First the sound pressure generated at the MRP is set to exactly -4.7 dBPa in an automatic measurement routine at 1 kHz. The required generator voltage 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 supplied transformer is not connected between the generator and the artificial mouth.

  The uncorrected frequency response of the artificial mouth is now 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 non-linearities of the speaker in the mouth are measured and considered as fine correction in the final equalization file.

  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 Voice Coder Loop

For measuring the echo loss or stability margin, the test signal is routed via the voice coder to the test mobile and back again via the voice decoder. To obtain correct measurement results the total gain of this loop must be known. This loop cannot be closed in the CMD.

A mobile has to be connected to the CMD’s RF interface to measure the loop gain. Using a standard Layer 3 command, the mobile can be set so that the audio data sent via the RF interface is returned unchanged, i.e. a total echo is obtained for the audio data. The overall loop gain is then the measured echo loss. The measured gain is saved and used for corrections in subsequent echo loss measurements. The typical gain of the CMD loop is approx. 19 dB.

➢ Press F12 to display the calibration keys:

--- CALIBRATION ---

F5	F6	F7	F8	F9	F10	F11	F12
<-	CAL_MIC	CAL_MOU	CODEC	ADJ_SLR	ADJ_RLR		

➢ Call the test routine with the F8 CODEC function key

➢ Select the CODEC CAL speech mode when requested to do so.
This selection is only possible if the LOOP COMMAND is activated with ENABLE. Select LOOP COMMAND on the CMD by pressing NETWORK and then NEXT PAGE and then ENABLE.

An echo loss measurement is now performed with the aid of the artificial voice but, in contrast to a normal echo loss measurement, the connected mobile is set to total echo. This echo is measured according to CCITT P.50 for a male and a female voice. The mean value of the two voices gives the final result which should be approx. 19 dB.

Adjustment Routine for Sending Loudness Rating (SLR) Using a Reference Mobile

The sensitivity of the CMD voice decoder must be considered when measuring the sending loudness rating. With the previously described calibration of the voice coder loop, only the sum gain but not the separate gains of coder and decoder can be measured.

For this reason the nominal gain of the voice decoder is used for calculating the SLR, but the nominal gain may differ by up to approx. 2 dB. The accuracy of the SLR calculation can be considerably increased when a test mobile with a known SLR is used for calibration, e.g. a mobile from a type approval mobile with measurement via the DAI interface.

The adjustment routine asks for the rated SLR of the connected mobile, performs three SLR measurements and then averages the results. A correction value is calculated from the difference between the nominal SLR and the measured SLR, stored and used for all subsequent SLR measurements. This adjustment takes into account the characteristics of the CMD and need only be repeated when the CMD is replaced.

- The mobile under test is installed in the LRGP position (CCITT P.76) and the speaker is sealed to the artificial ear.

- Set up a CALL to the CMD and select the HANDSET mode, if necessary.

- Press F12 display the calibration keys:

--- CALIBRATION ---

F5	F6	F7	F8	F9	F10	F11	F12
<-	CAL_MIC	CAL_MOU	CODEC	ADJ_SLR	ADJ_RLR		

- Call the test routine by pressing function key F9 ADJ_SLR.
- Enter the known SLR value and confirm with RETURN.

The program performs the measurement and saves the correction value in a nonvolatile memory.

Adjustment Routine for Receiving Loudness Rating (CLR) Using a Reference Mobile

The sensitivity of the CMD's voice coder must be considered when measuring the receiving loudness rating. With the previously described calibration of the voice coder loop, the sum gain but not the separate gains of coder and decoder can be measured.

For this reason the nominal gain of the voice decoder is used for calculating the RLR, but the nominal gain may differ by up to approx. 2 dB. The accuracy of the RLR calculation can be considerably increased when a test mobile with known RLR is used for calibration, e.g. a mobile from a type approval measurement via the DAI interface.

The adjustment routine asks for the rated RLR of the connected mobile, performs three RLR measurements and then averages the results. A correction value is calculated from the difference between the nominal RLR and the measured RLR, saved and used for all subsequent RLR measurements. This adjustment takes into account the characteristics of the CMD and need only be repeated when the CMD is replaced.

- The mobile under test with known SLR is installed in the LRGP position (CCITT P.76) and the speaker sealed to the artificial ear.

- Set up a CALL to the CMD and select the HANDSET mode, if necessary.

- Press to the display F12 calibration keys:.

--- CALIBRATION ---

F5	F6	F7	F8	F9	F10	F11	F12
<-	CAL_MIC	CAL_MOU	CODEC	ADJ_SLR	ADJ_RLR		

- Call the test routine by pressing function key F10 ADJ_RLR.

- Enter the known RLR value and confirm with RETURN.
  The program performs the measurement and saves the correction value in nonvolatile memory.

6 Processing of Measurement Results

Printing, Storing and Displaying of Measurement Results

The result of each measurement is graphically or numerically displayed on the screen and, if applicable, a PASS or FAIL verdict is output.

The following softkeys are displayed.

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

Pressing the CONT key brings back the selection menu for the measurement.

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, with xx representing a consecutive number (of max. 5 digits). This allows processing of measurement results with other programs. The TRC_FILE key has to function when the results are numerically displayed.

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

Since both the TRC and the PCX files are consecutively numbered, it is useful to copy the files of a measurement sequence, for instance, and to save them under a new name. In this case the original TRCxx.TRC and PICxx.PCX files can be cleared. Thus results can be identified more easily and a mix up between them avoided. (The files can be copied and renamed using common DOS commands.)

Call end a DOS shell called after terminating of the test program (e.g. with key F5) by entering the command SHELL <RETURN>. Entering EXIT <RETURN> brings back BASIC without the program being cleared. The program can be restarted immediately by entering RUN.

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

Printer type and desired settings are not selected by the program but the printer selected last and set in UPL manual mode will be chosen. For this reason the desired printer, scaling and format should 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 as far as possible integer scale factors for the printer output. If fractional scale factors (especially values <

1) are used, the pixels values are interpolated and the print quality could be 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.

Numerical values are automatically added to a result file after each measurement. This file has the name RES_GSM.LOG.

Each result is written in a separate line with the measurement in plain text, showing the value, date and time. Thus all numeric measurement results can be called again after a test sequence has been performed and evaluated.

As with TRC and PCX files, it may be useful to copy the RES_GSM.LOG file after a measurement sequence and to save it under a new name. After this the RES_GSM.LOG file can be cleared. 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 and saved under another name using common DOS commands.

Entering EXIT <RETURN> brings back BASIC without the program being cleared. The program can be immediately restarted with RUN.

7 Terminating the Application

As long as the arrow → is displayed below the F12 key, another set of softkeys can be called with this key. With F5 the user can return to the previous set of softkeys, so 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	RECEIVE	STMR	LSTR	ECHO	STAB-MRG	->

After selecting END by pressing the F5 key, the following query is displayed:

- "Do you want to terminate the program <Y><N>?"

  Upon confirmation with Y, the program is aborted but not cleared. The softkey line for BASIC is then 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.

8 Ordering information

Audio Analyzer		
UPL	DC to 110 kHz	1078.2008.06
Option required for UPL		
UPL-B6	Extended Analysis	1078.4500.02
UPL-B8	Mobile Phone Test Set	1117.3505.02
UPL-B10	Universal Sequence Controller	1078.3904.02
Radiocommunicattion Tester		
CMD52	GSM 900 /GSM 1800	1050.9008.52
CMD53	GSM 900 /GSM 1800	1050.9008.53
CMD55	GSM 900 /GSM 1800	1050.9008.55
Option required for CMD		
CMD-B5	Real Time Speech Coder/Decoder for GSM	1051.8657.02

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