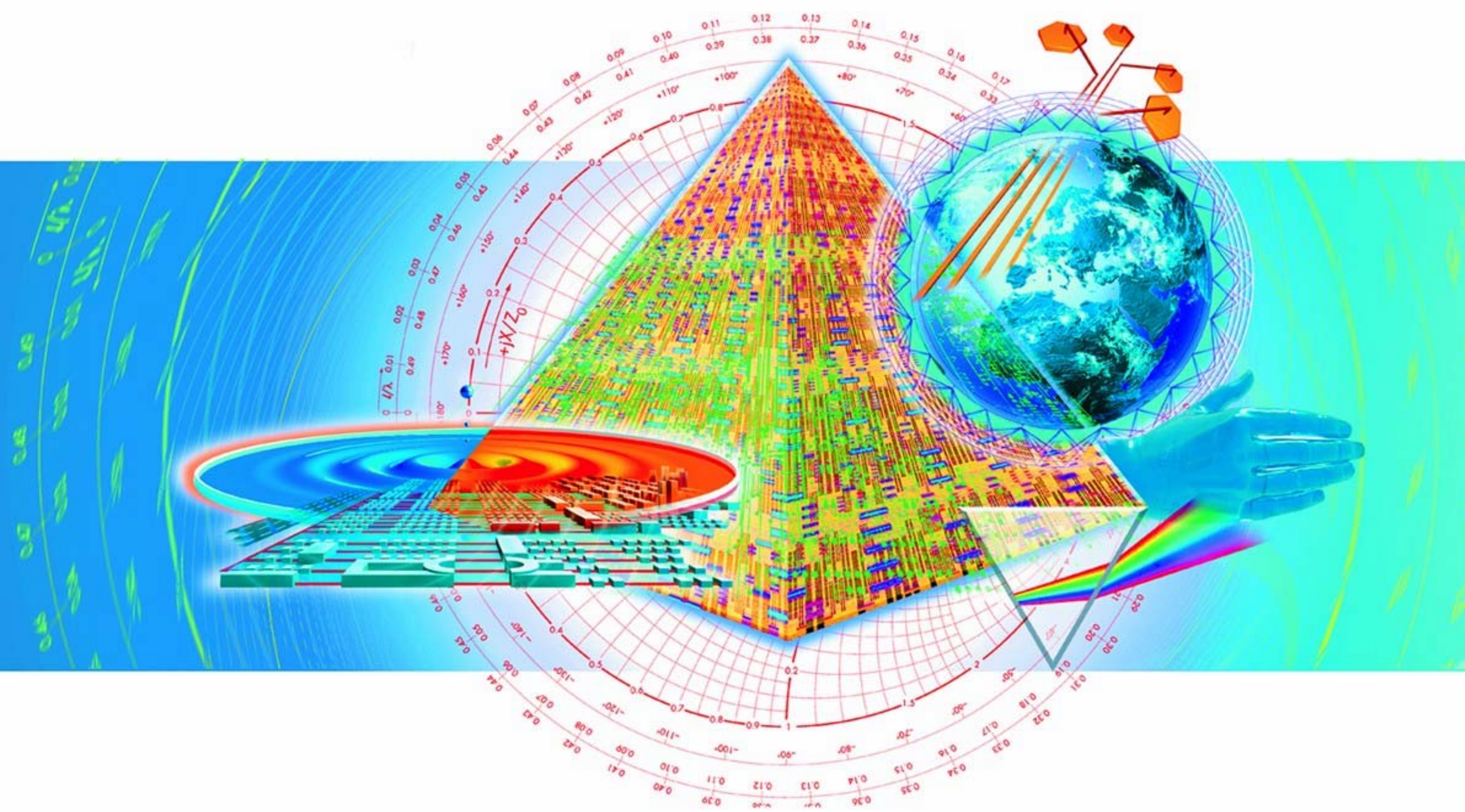
Division 1
Sales Engineer University
11th – 15th October 2004

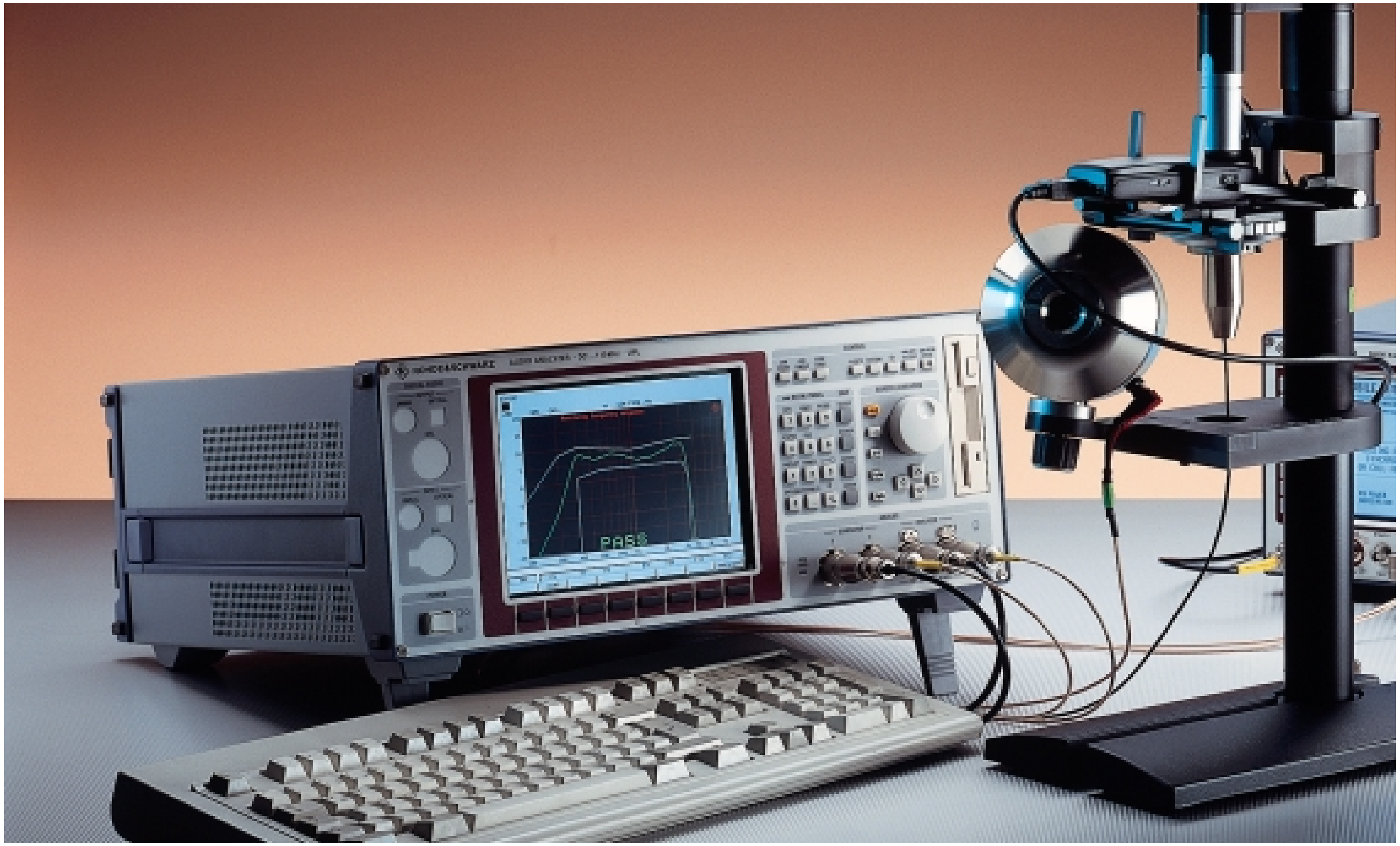


Module: **Fr_11**
Acoustic measurements for mobile phones and hearing aids

Date: **15th October 2004**

TZ-GB1-1MAT-040308-P-00.00-TS21-04-Fr_11

Acoustical Measurements with UPL-B9



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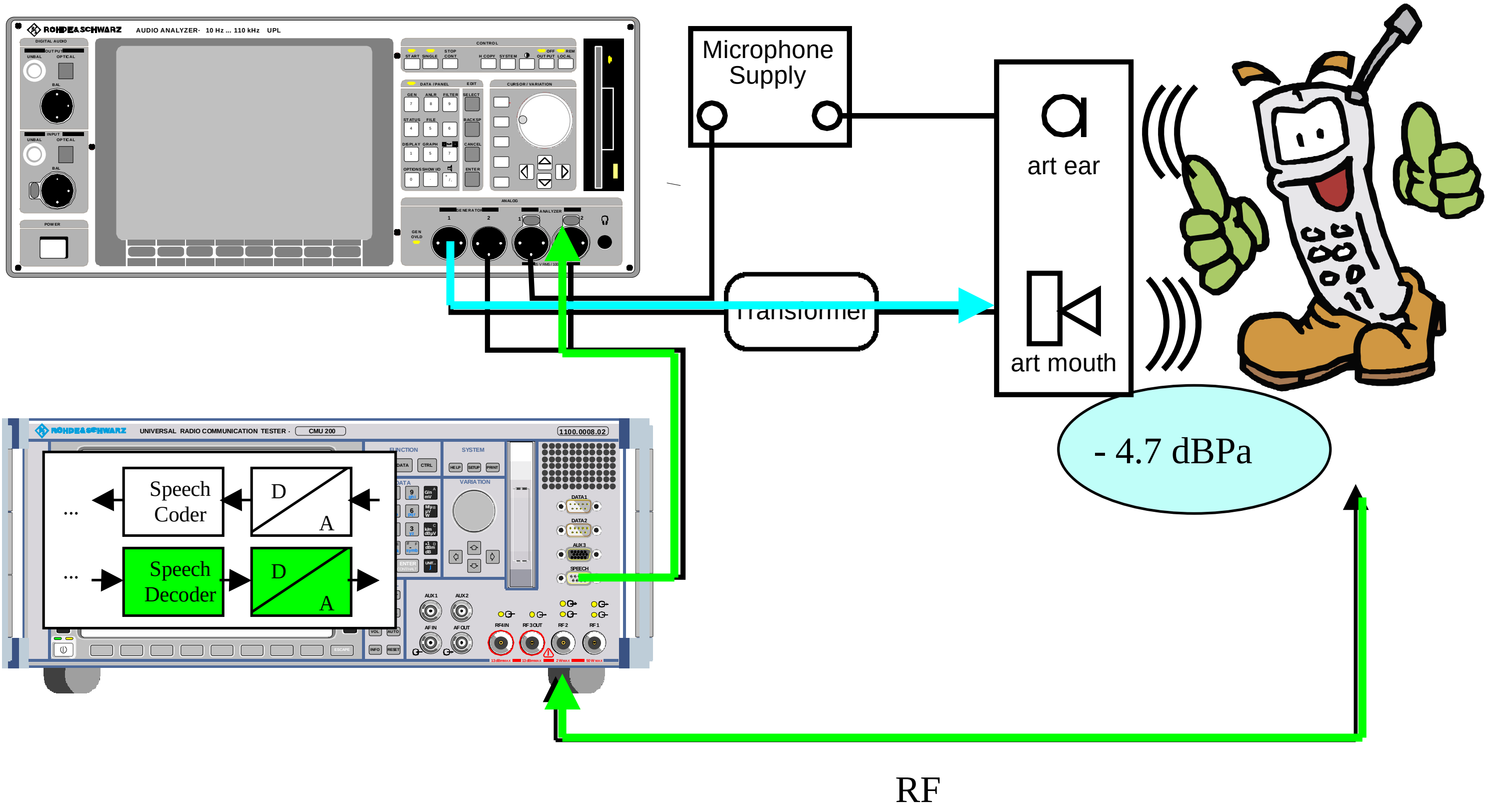
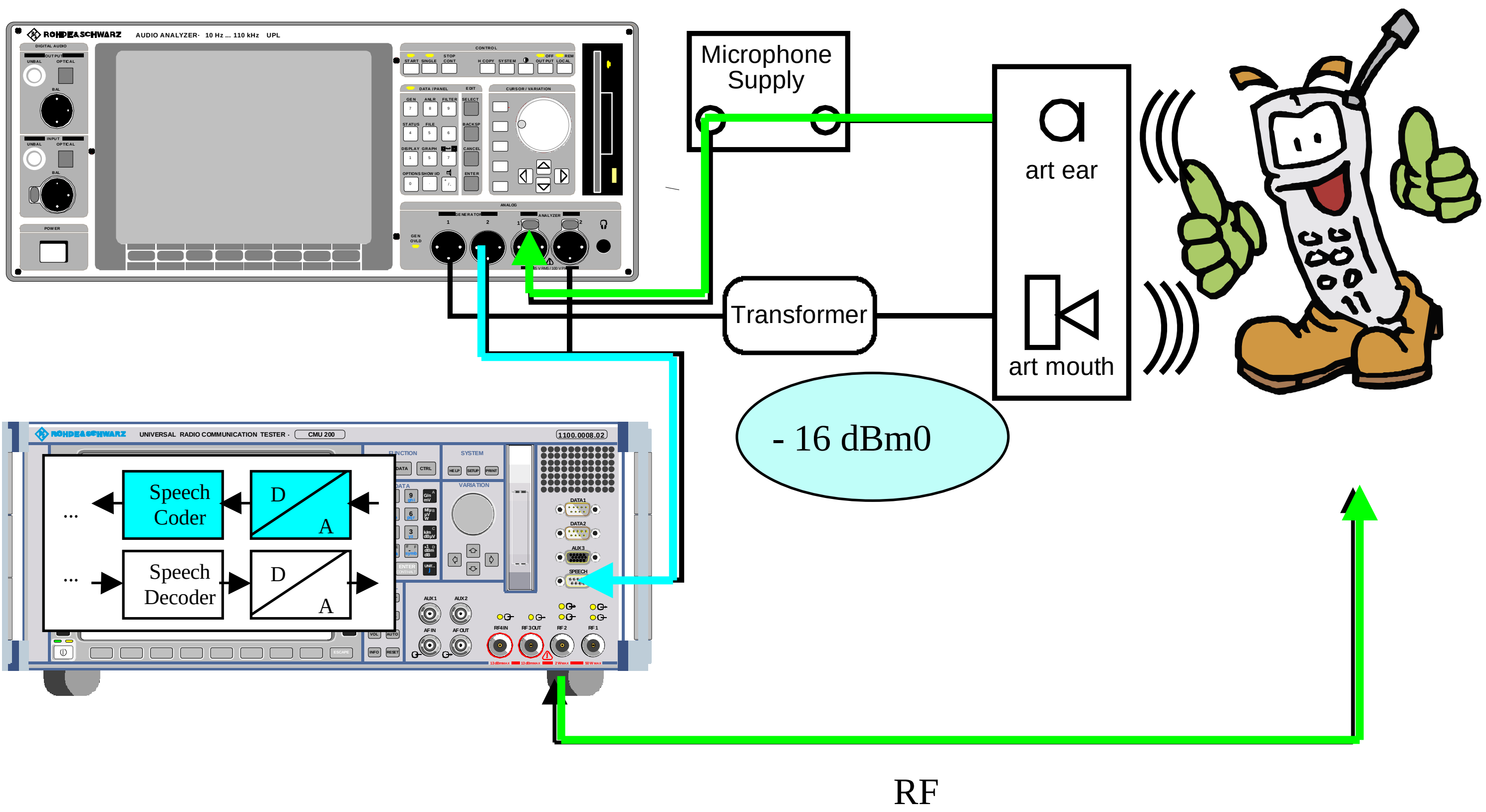
Setup:



Signals: • Amplitude modulated Multi-Tone Signal according to ITU-T P.501

Tests:
- Sending Frequency Response (decoder output voltage / sound pressure level)
- Sending Loudness Rating (weighted absolute loudness over all frequencies)

Setup:

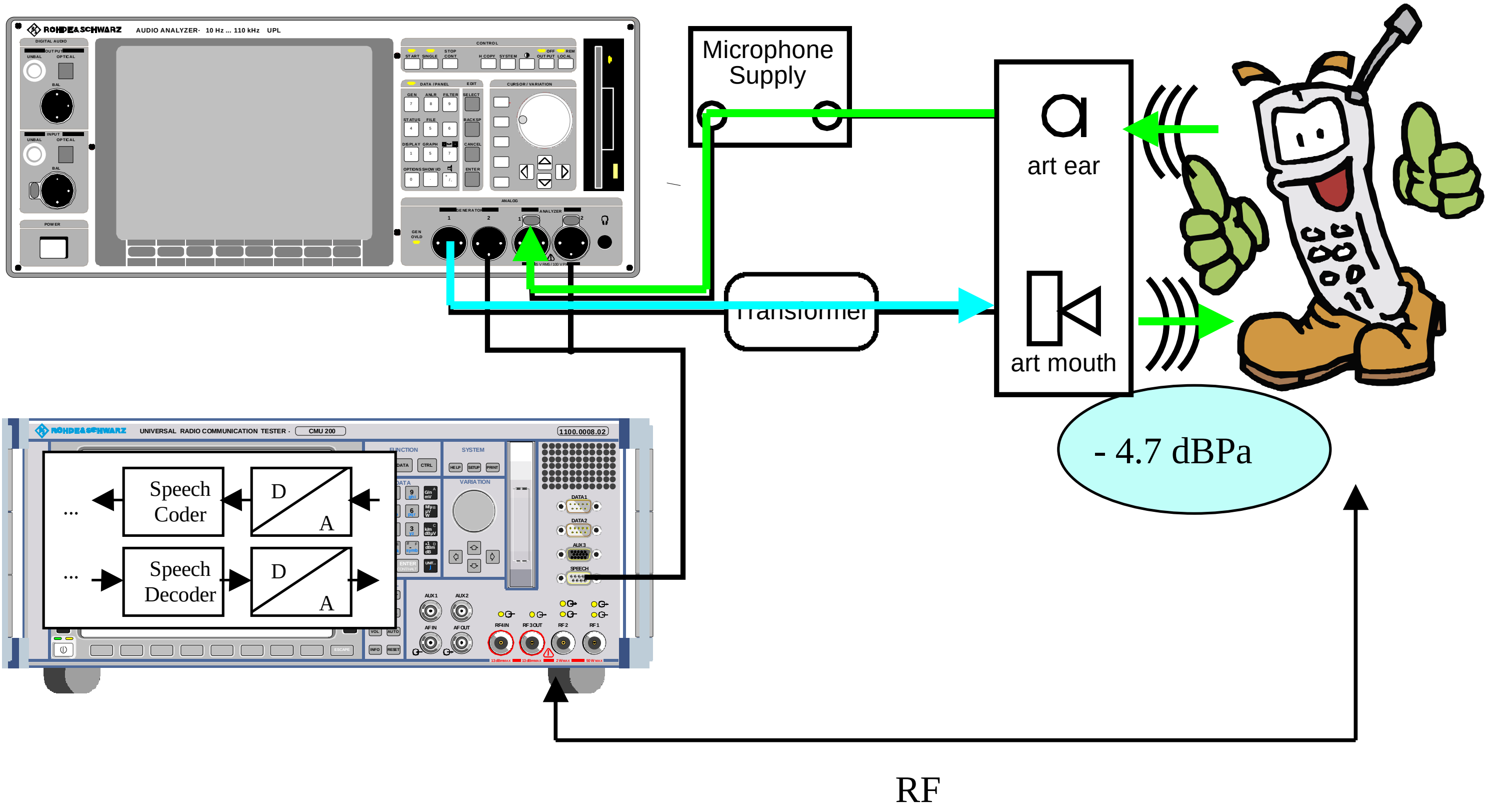


Signals: • Amplitude modulated Multi-Tone Signal according to ITU-T P.501

Tests:
- Receiving Frequency Response (sound pressure level / coder input voltage)
- Receiving Loudness Rating (weighted absolute loudness over all frequencies)

Side Tone Masking Rating (STMR)

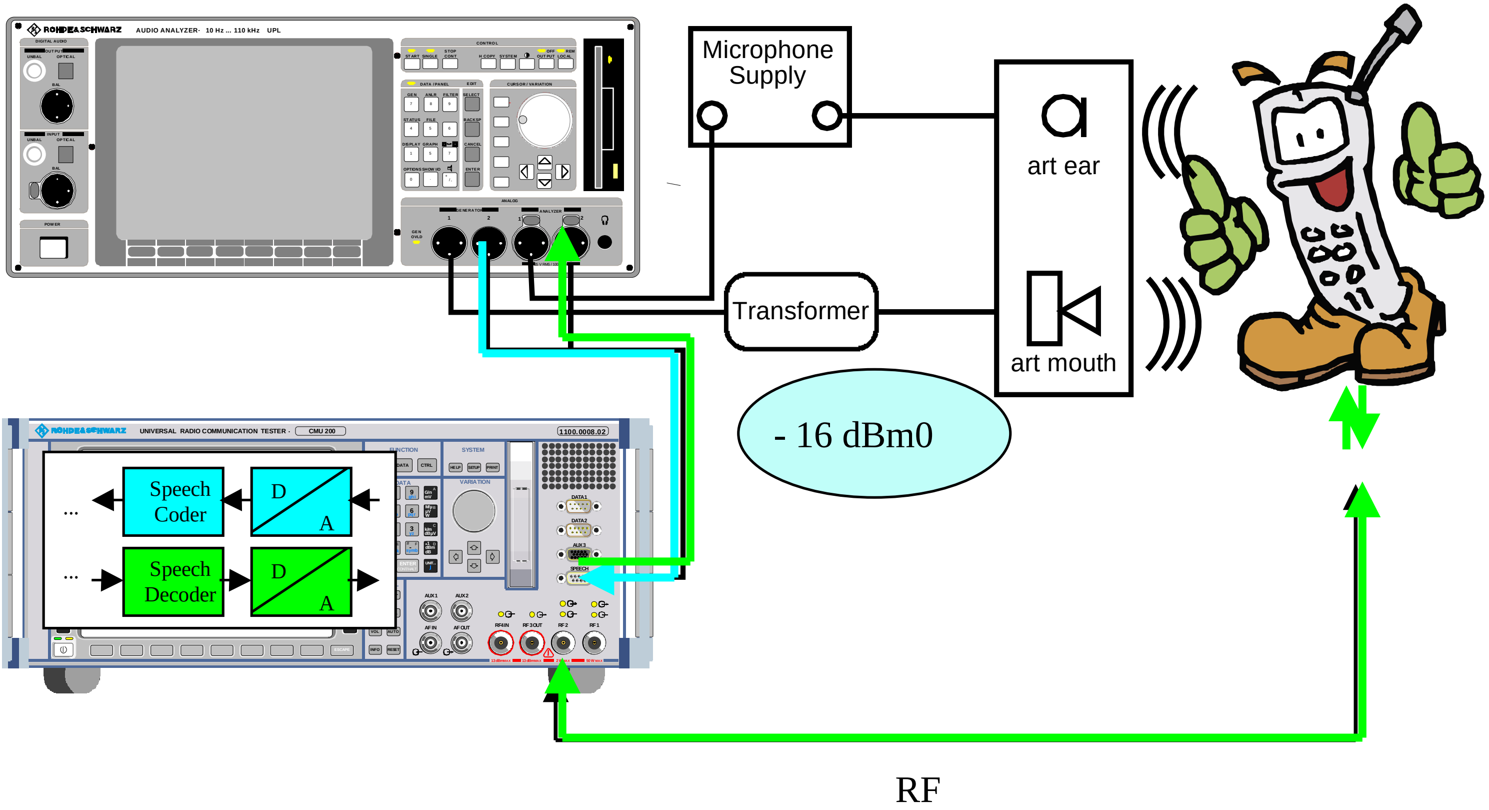
Setup:



Signals: • Measurements at several individual frequencies

Tests: • Side Tone Masking Rating (weighted path attenuation between microphone and loudspeaker, required for natural hearing impression)

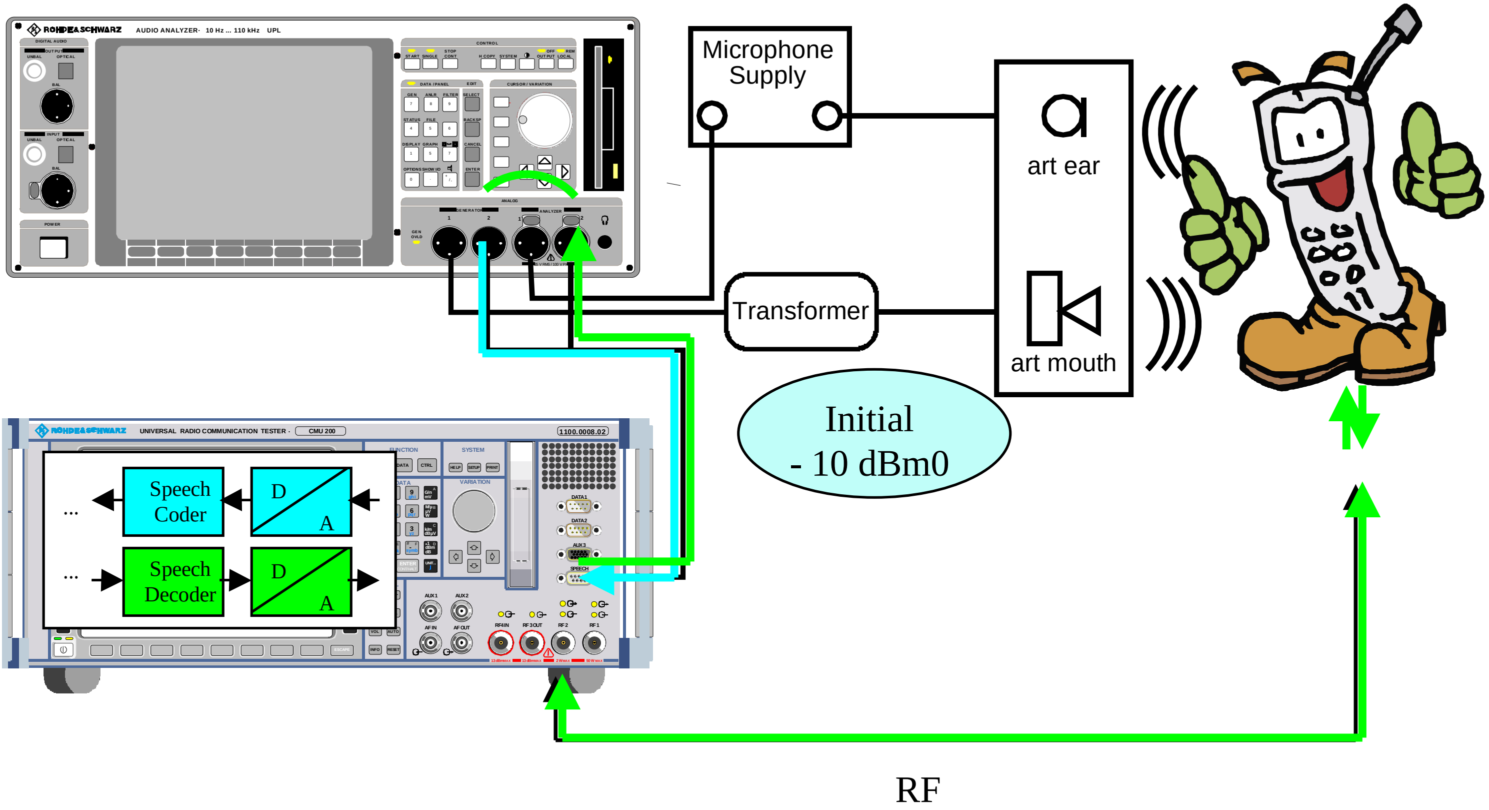
Setup:



Signals: • Amplitude modulated Multi-Tone Signal according to ITU-T P.501

Tests: • Echo Loss (weighted ratio of decoder output and encoder input voltage, caused by unwanted acoustic coupling inside the phone)

Setup:

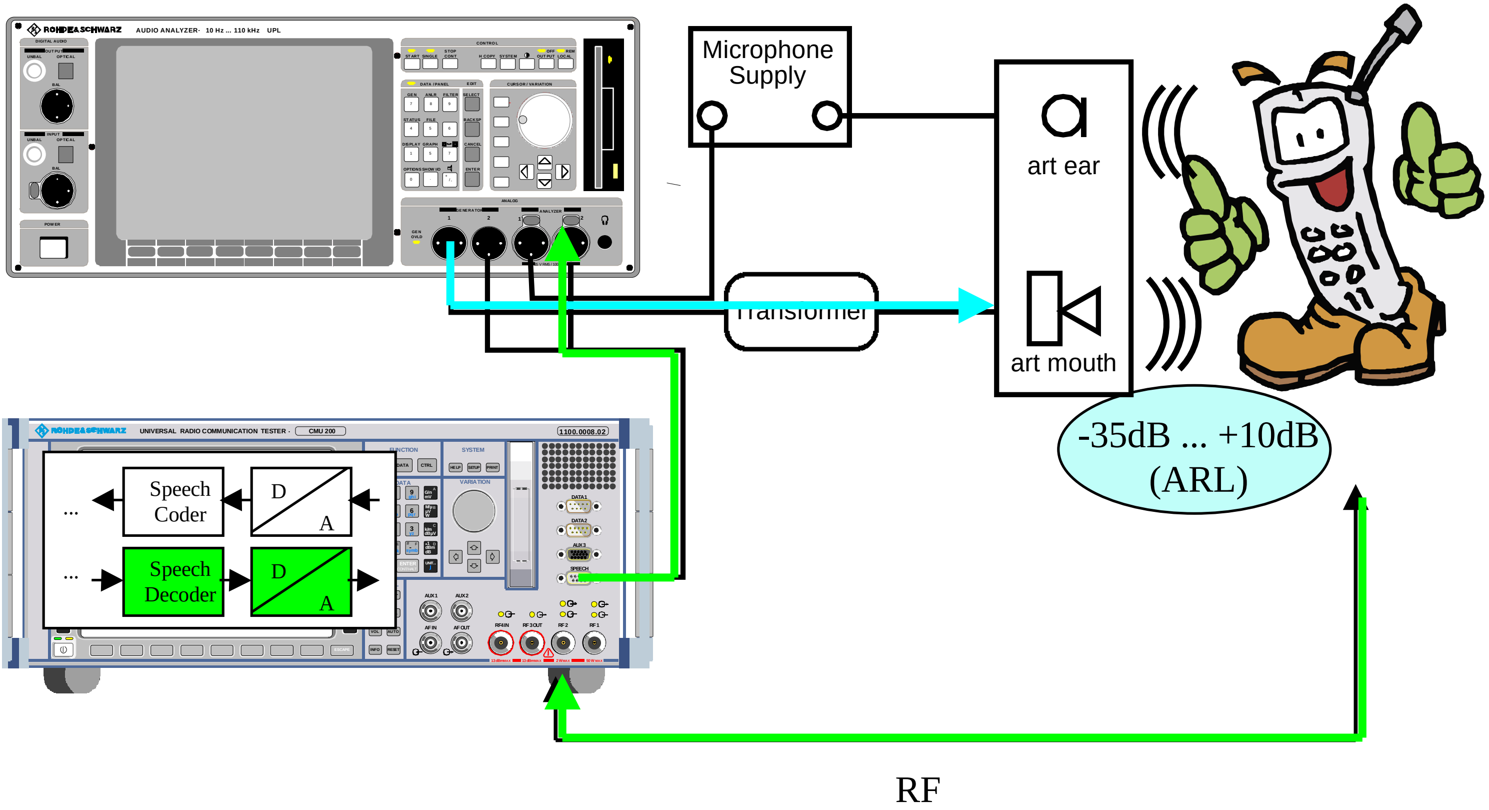


Signals: • Initial noise signal to acticate loop, switched off during test
• Loop inside UPL

Tests: • Stability Margin (check on oscillations at a total loop gain of 6dB; phone placed with speaker and microphone on a hard board)

Sending Distortion

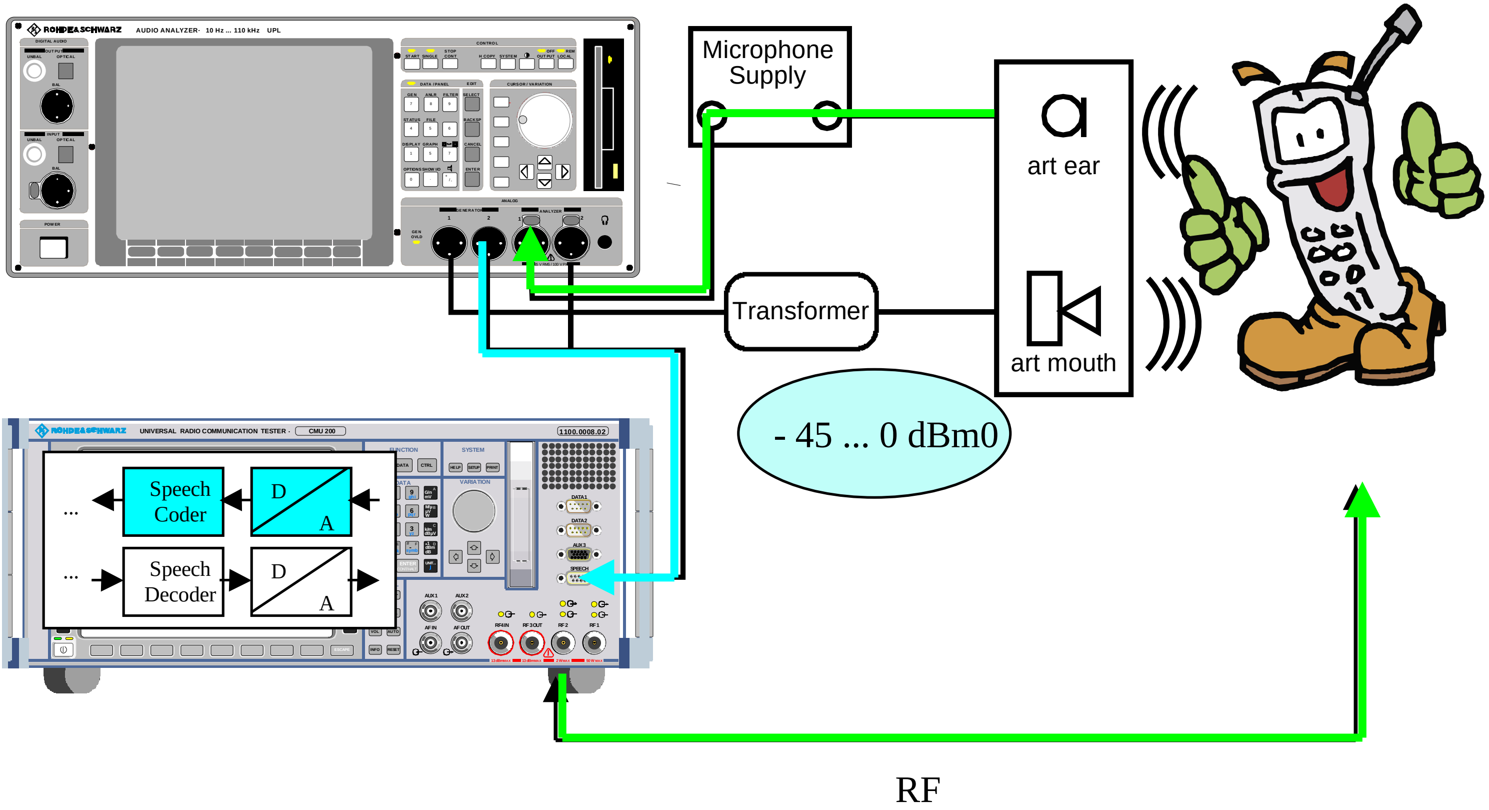
Setup:



Signals: • Pulsed 1015 Hz Tone (required for voice recognition in phone)

Tests: • Sending Distortion (SINAD value of decoder output voltage at different sound pressure levels)

Setup:



Signals: • Pulsed 1015 Hz Tone (required for voice recognition in phone)

Tests: • Receiving Distortion (SINAD value of sound pressure level at different coder input voltages)

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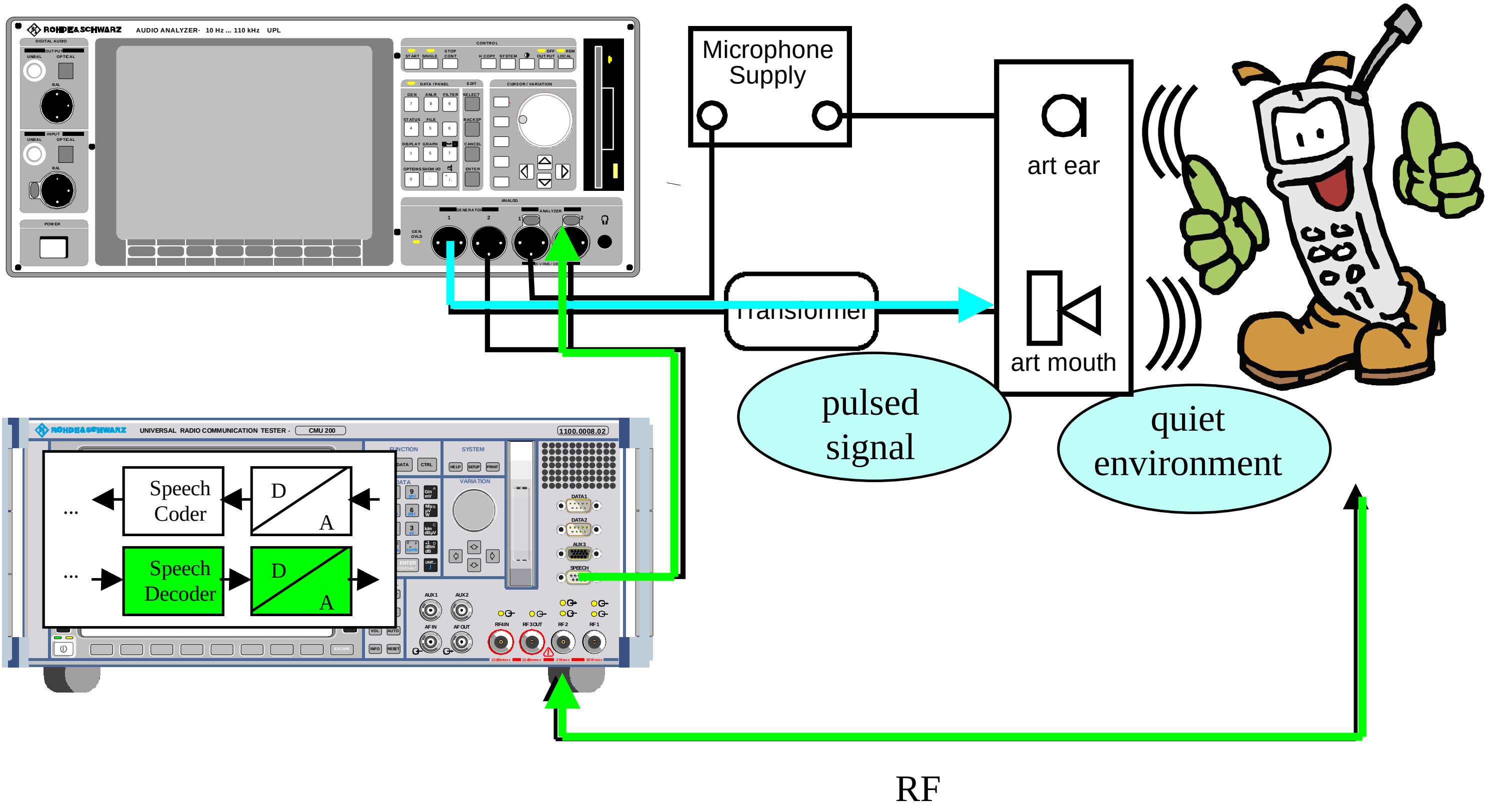
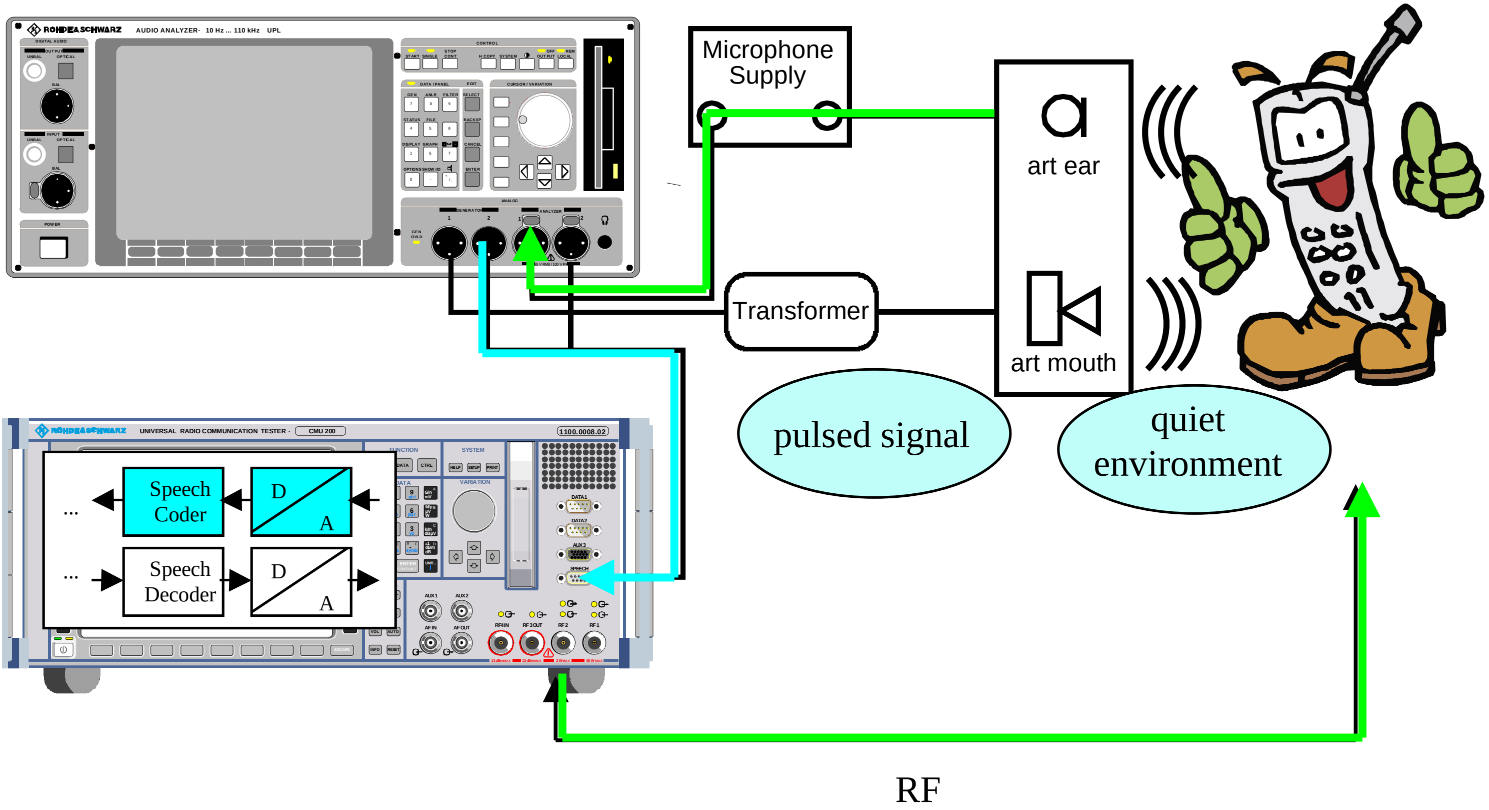
Setup:



Signals: • Pulsed Signal, required for voice recognition in phone (measurements during off-time of this signal)

Tests: • Idle Channel Noise Sending (psophometrically weighted decoder output voltage during off-time of stimulus)

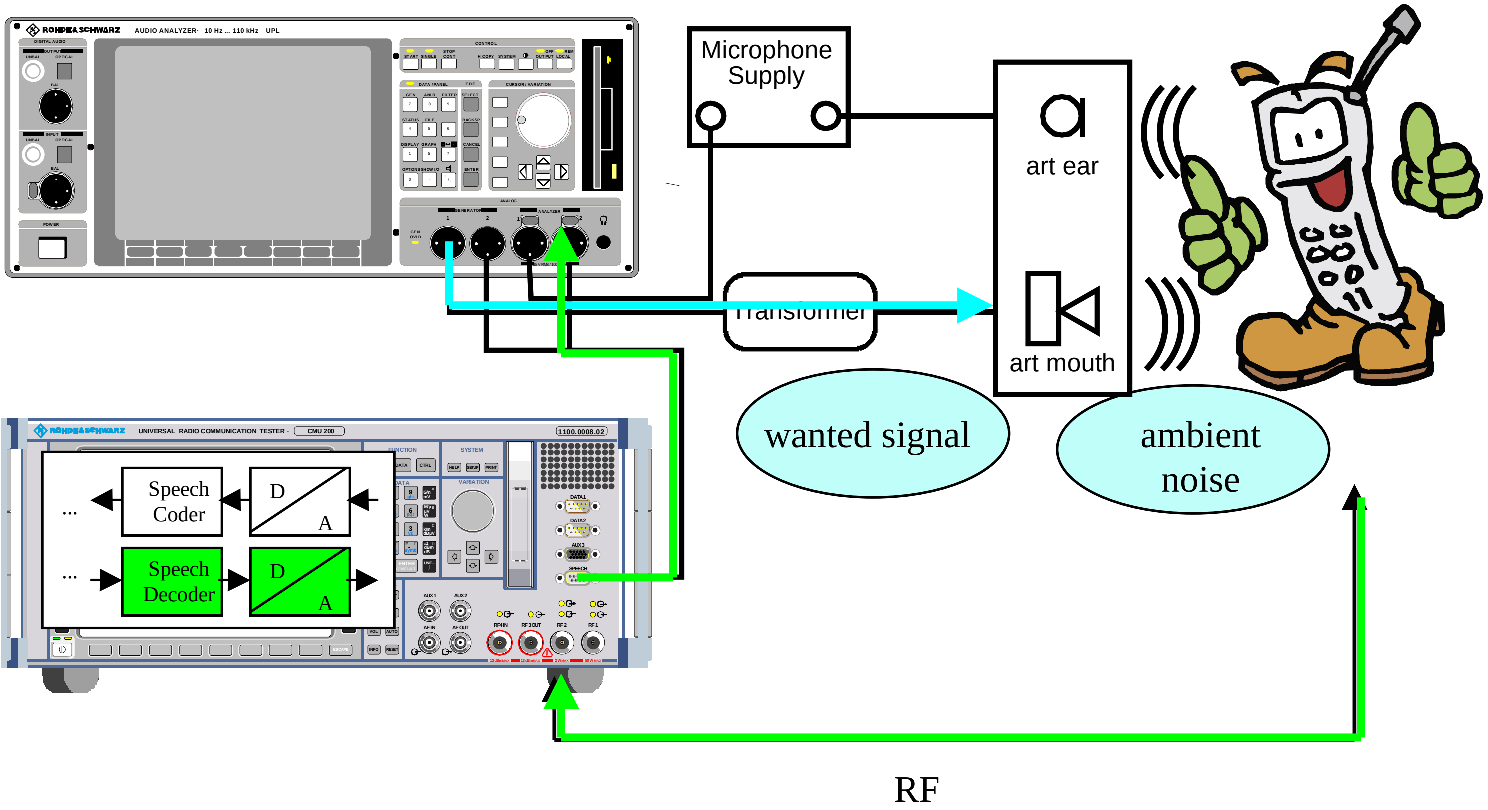
Setup:



Signals: • Pulsed Signal, required for voice recognition in phone (measurements during off-time of this signal)

Tests: • Idle Channel Noise Receiving (A-weighted sound pressure level during off-time of stimulus)

Setup:



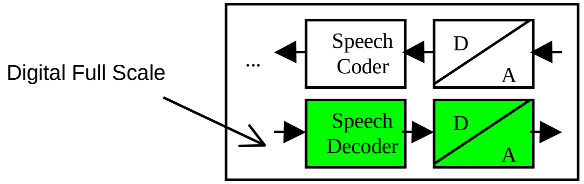
Signals: • Pink Ambient Noise
• Wanted Signal modulated Multi-Tone Signal

Tests: • Ambient Noise Rejection (ratio of decoder output signal for wanted signal and ambient noise)

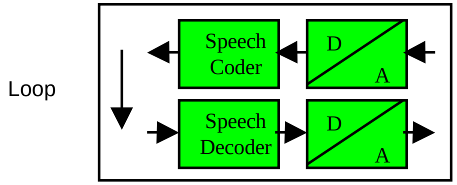
Artificial Ear: 1.) calibration with sound pressure calibrator (1kHz, 0 dBPa)

Artificial Mouth: 1.) calibration of reference microphone with sound pressure calibrator
2.) calibration of artificial mouth with reference microphone (- 4.7 dBPa)

CMU: 1.) decoder calibration:
measurement of decoder output for a digital full-scale signal applied to speech decoder



2.) encoder calibration:
measurement of encoder input voltage, required for digital full-scale signal at speech codec



Acoustical Tests of GSM and Third-Generation Mobile Telephones

The acoustic reproduction quality of a mobile phone is its most important characteristic in everyday use. In view of this, the trend towards ever smaller and lighter telephones does not exactly make life easier for developers. Just as there is a limit to how small the keys of the telephone can be, so it is impossible to alter the given distance between the human ear and mouth. As a result, the smaller the telephone becomes, the less favourable are the acoustic conditions. If a mobile telephone can no longer provide adequate communication, it is nothing more than a toy; the prettiest design and most impressive gimmicks are not of much use if callers cannot or can hardly understand each other.
Measuring the acoustic characteristics of a mobile telephone is all the more important, since users do not themselves immediately notice deficiencies. Considerably deviating transmit frequency response, distortion and noise only become noticeable when callers complain of difficulty understanding each other – and even then, they tend to suspect a poor connection via the base station before exposing the acoustical quality of the own mobile as the culprit.

Consequently, ensuring acoustic characteristics has been a key issue since the beginnings of GSM mobile radio communication. Different regulations specify those measurements for GSM and Third-Generation mobile telephones:

GSM standardised measurements via DAI interface:
GSM mobile phones have to be measured according to GSM 11.10-1 chapter 30 which is now 3GPP TS 51.010-1 chapter 30. In the past those measurements were only defined using a digital audio interface (DAI) at the mobile phone thus avoiding the measuring problems caused by the speech coding algorithms. The digital signals are tapped before the coder input and after the coder output for these measurements. The DAI has since been used for the development and type-approval testing of the acoustic characteristics. For this task the special model Audio Analyzer UPL16 was developed. It is equipped with the Digital Audio Interface DAI as defined in GSM Phase 2, 11.10, section 36.4 which is now 3GPP TS 44.014 section 10. Together with Radiocommunication Tester CRTC all tests defined in GSM 11.10-1 chapter 30 are available, all mandatory tests are validated for type approval measurements for GSM phase 2.

Since the DAI interface is not accessible in a standard mobile telephone, acoustic measurements were always limited to specially prepared test telephones equipped with this extra interface.

GSM measurements via air interface:
Besides the type-approval measurements via the DAI interface, Rohde & Schwarz had already developed methods several years ago that enabled GSM mobile telephones to also be measured via normal air interface (RF link). This was made possible by using multitone signals with sophisticated measurement technology. With only minor deviations, these methods yielded the same results as measurement via the DAI interface. Together with the Radio Communication Tester CMD or CMU200, the Audio Analyzer UPL with option UPL-B8 created a test set which was soon adopted by manufacturers, test houses, and service providers throughout the world. The great advantage of this measurement method is that any commercially available telephone can be measured without special modifications. However, the method is not standardised worldwide, and therefore cannot be used for type approval tests.

3GPP standardised measurements via air interface:
During the development of third-generation mobile radio a goal was to define test methods which enable measurements of the acoustic characteristics via the normal air interface. The task was to measure all the characteristics in normal operating mode, which means that all the signal processing algorithms in the mobile telephone should remain active during the measurement. Those tests using artificial speech or speech like signals are now defined in 3GPP TS 26.132 and the according requirements in 3GPP TS 26.131. The option UPL-B9 for the Audio Analyzer UPL provides all acoustic tests defined in 3GPP TS 26.132 using Audio Analyzer UPL06, UPL66 or UPL16.

Harmonisation of test specifications:
Up to and including release 99 there are different acoustic requirements for GSM and 3GPP mobile phones. GSM mobile phones have to be tested as specified in GSM 03.50 and former GSM 11.10-1 which is now 3GPP TS 51.010-1 and 3GPP mobile phones or dual mode mobile phones as specified in 3GPP TS 26.131 and TS 26.132. A harmonisation of the tests and requirements for GSM and 3GPP mobile phones was done early 2001 now specifying the same tests for 3GPP and GSM release 4 mobiles.

Validation:
In order to use those tests for type approval, the tests and procedures must be proved and validated by an independent test house. This was done for the combination of Audio Analyzer UPL16 with Radiocommunication Tester CRTC for GSM phase 2 and was now done as well as for the combination of Audio Analyzer UPL with Radio Communication Tester CMU200 for release 99 via DAI and release 4 mobiles via air interface.

All those tests are now available using Audio Analyzer UPL and Radio Communication Tester CMU200. For 3GPP mobile phones all testcases are validated, for GSM release 99 mobiles all mandatory testcases are validated using Audio Analyzer UPL16 and DAI interface.

Available Standardised Acoustical Tests of Mobile Phones with Audio Analyzer UPL and Radio Communication Tester CMU200:

Acoustical Tests of GSM and 3GPP Mobile Phones with Option UPL-U81 (DAI) or UPL-B9 (Air interface)	**3GPP TS 51.010 GSM Release 99** (DAI interface) Chapter	**3GPP TS 51.010 GSM Release 4** (Air interface) Chapter	**3GPP TS 26.132 3GPP Release 4** (Air interface) Chapter
Sending sensitivity/frequency response	**30.1***	**30.12***	**7.4***
Sending loudness rating	**30.2***	**30.13***	**7.2***
Receiving sensitivity/frequency response	**30.3***	**30.14***	**7.4***
Receiving loudness rating	**30.4***	**30.15***	**7.2***
Side tone masking rating	**30.5.1***	**30.16***	**7.5***
Listener side tone rating	30.5.2		
Echo loss	30.6.1 (release 4)	**30.17.1***	**7.7***
Stability margin	**30.6.2***	**30.17.2***	**7.6***
Sending distortion	**30.7.1***	**30.18***	**7.8***
Receiving distortion	30.7.2		**7.8***
Sidetone distortion	30.8		
Out-of-band signals (sending / receiving)	30.9		
Idle channel noise (sending / receiving)	30.10		**7.3***
Ambient noise rejection	30.11 (release 4)	**30.19***	**7.9***

*** validated testcases**

Instruments and options needed:

	3GPP TS 51.010 **GSM Release 99** (DAI interface)	**3GPP TS 51.010** **GSM Release 4** (Air interface)	**3GPP TS 26.132** **3GPP Release 4** (Air interface)
Audio Analyzer	**UPL16** + UPL-U81	**UPL16** + UPL-B9 or **UPL06 / 66** + UPL-B6 + UPL-B9 + UPL-B10	**UPL16** + UPL-B9 or **UPL06 / 66** + UPL-B6 + UPL-B9 + UPL-B10
Radio Communication Tester	**CMU200** + CMU-B21 + CMU-B52 + CMU-Kxx depending on band used	**CMU200** + CMU-B21 + CMU-B52 + CMU-Kxx depending on band used	**CMU200** + CMU-B21 + CMU-B52 + CMU-Kxx depending on band used Options for other standards under development

Typical acoustical equipment needed for measurements on mobile phones:

Telephone test head	B&K 4602B
Ear simulator	B&K 4185 Type 1 B&K 4195 Type 3.2 low leakage and high leakage
Artificial mouth	B&K 4227
Head and torso simulator	B&K 4128C (ear type 3.3)
Acoustic calibrator	B&K 4231
Microphone power supply	B&K 2690A0S2
Acoustic test chamber	e.g. Studio Box Type S

Acoustic Measurements of CDMA2000 Mobiles

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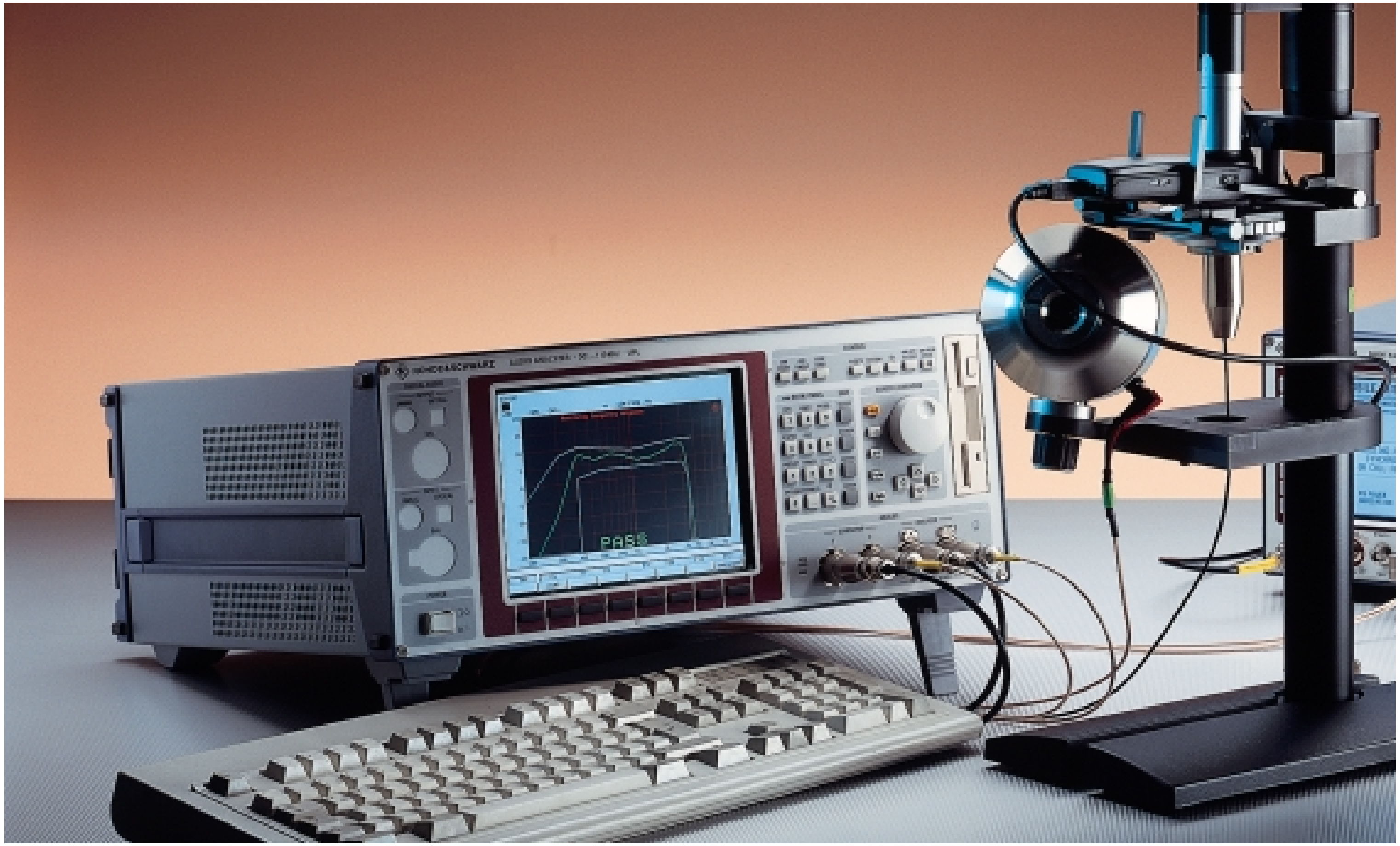
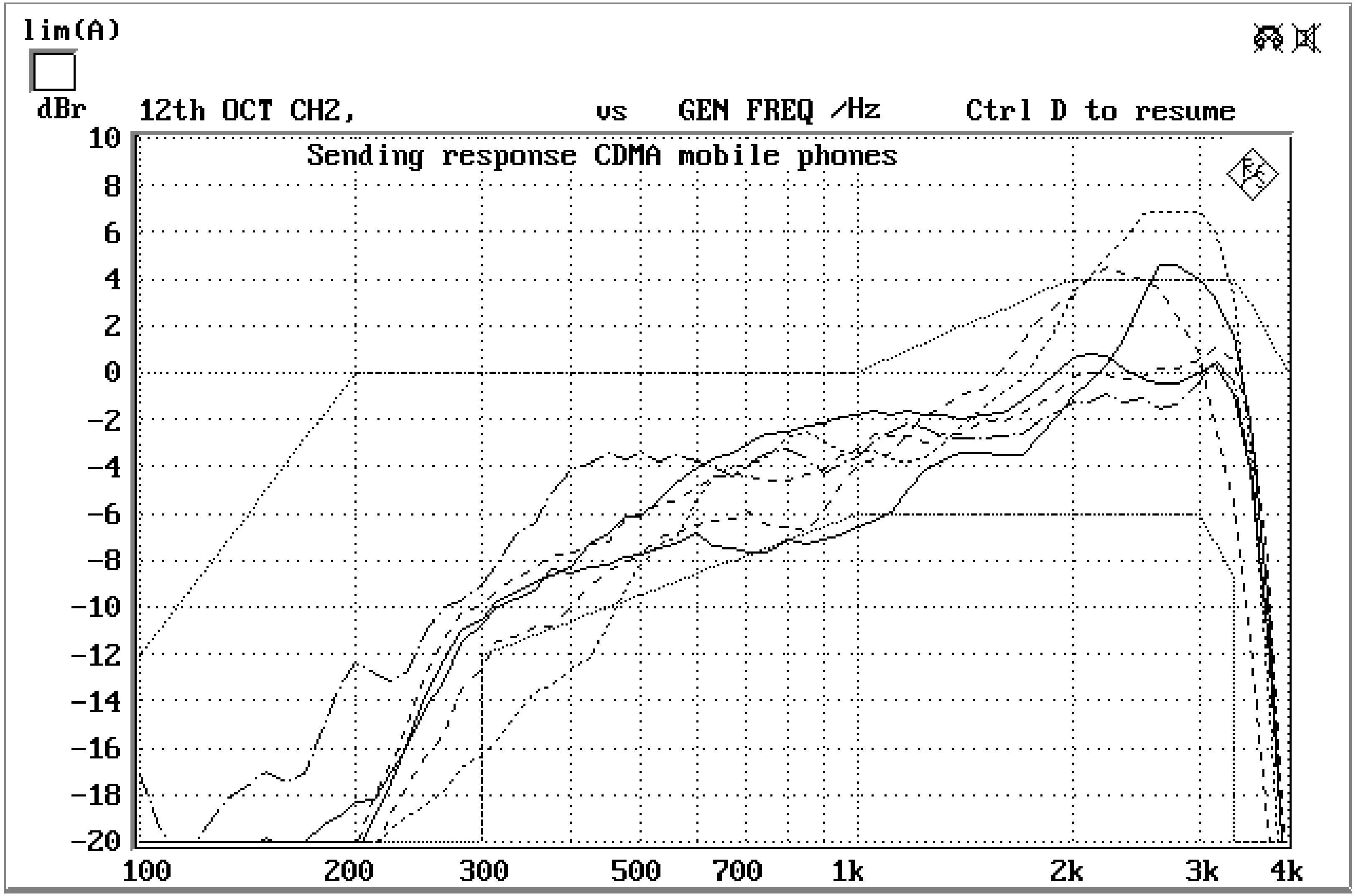


The test of CDMA2000 mobile phones is covered by 3GPP2 standard gremium.

Standards for acoustic measurement of mobile phones are only defined for AMPS, not yet for CDMA1 or CDMA2000!

The actual measurement results of different CDMA mobile phones show clearly that no standard exists!



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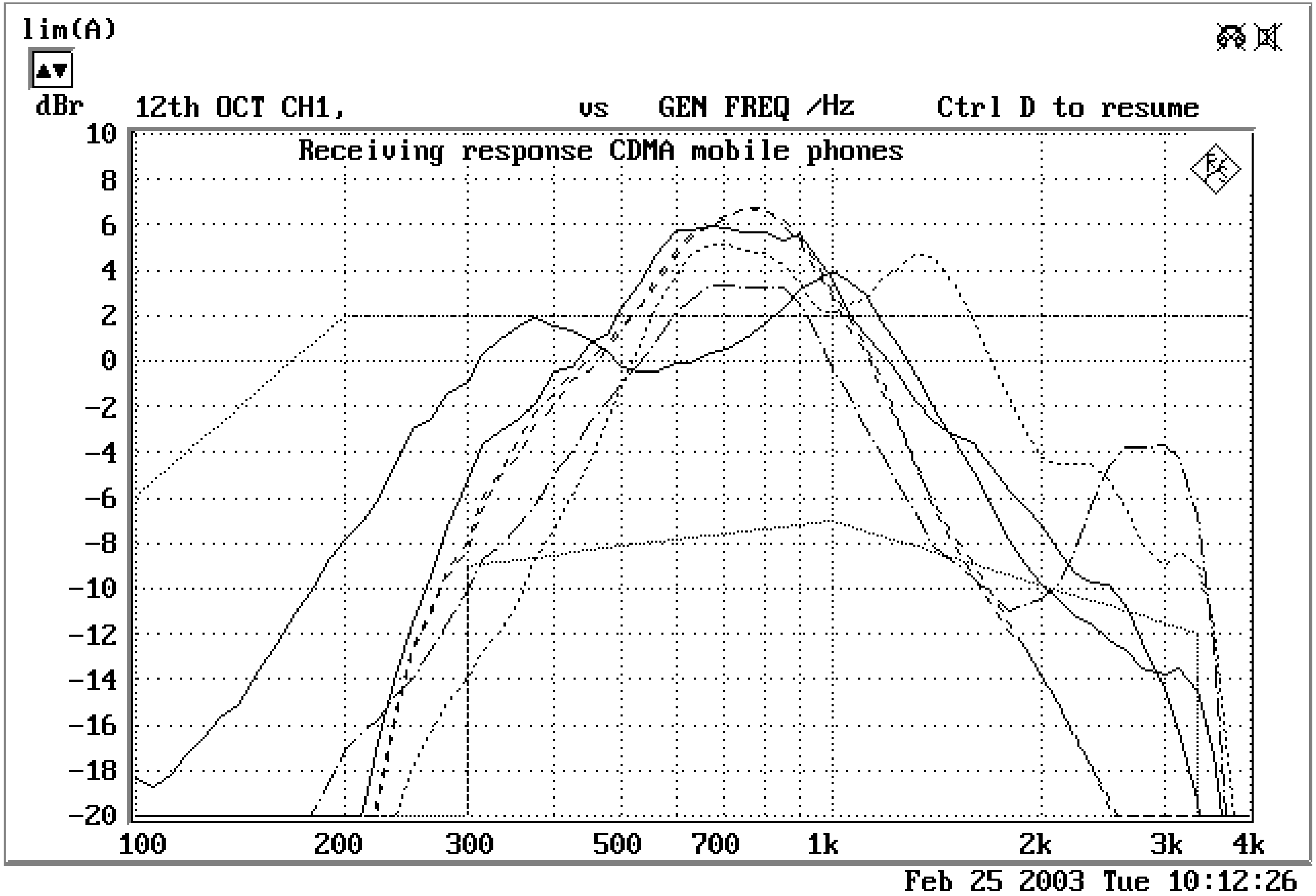

lim(A)
dBr
12th OCT CH1, vs GEN FREQ /Hz Ctrl D to resume
Receiving response CDMA mobile phones
10
8
6
4
2
0
-2
-4
-6
-8
-10
-12
-14
-16
-18
-20
100
200
300
500
700
1k
2k
3k
4k
Feb 25 2003 Tue 10:12:26

Besides the different frequency responses also the measured absolute loudness ratings differ strongly from the nominal values in 3GPP standard.

Measured sending loudness ratings vary from 10.17 dB to 26.81 dB, the nominal value should be 8 dB ± 3dB so most CDMA mobile phones have not enough sending sensitivity.

Measured receiving loudness ratings vary from –5.62 dB to –17.81 dB in maximum volume setting, verdict value is > -13 dB so some mobile phones compensate lacking sending sensitivity by enhanced receiving volume but some of them are thus exceeding the maximum allowed sound pressure level in the ear!

A new work proposal is on the way to create a new standard for acoustic measurements of CDMA2000 mobile phones - but this takes time!

Why should CDMA2000 and 3GPP mobile phones have different acoustic requirements? Everey mobile phone must comply to the same acoustic specifications. This is the only way to ensure interoperability independent from transmission standard GSM, UMTS, CDMA ....

The proposal of R&S is to use the already defined 3GPP acoustic tests also for CDMA mobile phones, this is widely accepted by many mobile phone manufactureres at least as an intermediate solution.

In order to bring this item forward and to enable the manufactureres to measure and compare acoustic behaviour of CDMA2000 mobile phones R&S developped and distributed an update of option UPL-B9 in February 2003 to all users of UPL-B9 for the measurement of CDMA2000 mobile phones. The idea is to establish an industry standard!

ROHDE&SCHWARZ

Test and Measurement
Division

Addendum to Option UPL-B9

CDMA2000 MOBILE PHONE TESTS
Update of UPL-B9 for CDMA

Version 1.0

Printed in the Federal
Republic of German

1 Overview

The acoustic measurement of 3GPP and GSM mobile phones using Audio Analyzer UPL with option UPL-B9 together with Radio Communication Tester CMU200 is a standard test solution used worldwide according to the standards 3GPP TS26.131 and 3GPP TS26.132. Those tests are validated for type approval.

The problem doing the same for CDMA or CDMA2000 mobile phones is the lack of a standard describing the acoustic requirements and tests for those phones – those standards are missing!

An action was started to generate according standards similar to the already existing 3GPP standards. As this process will take time Rohde & Schwarz developped provisional tests for CDMA mobile phones based on the existing standards for 3GPP and GSM mobile phones. Those tests can be used to develop or improve phones but they are not validated for type approval due to the lacking standards!
As there is no reason why a CDMA mobile phone should have a different sound than other phones those tests may help to harmonise the acoustical behaviour between standards.

Intensive tests of different existing mobile phones for IS-95 or CDMA2000 networks show the importance of those tests. Frequency responses show severe deviations violating the existing limits for 3GPP mobile phones and the measured loudness ratings can exceed limits by more than 15 dB and can even produce harmful sound pressure levels in the earpiece!

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
- 3G Mobile Phone Tests UPL-B9 (version 1.0)

This is the normal requirement for measurements of 3GPP phones and the same acoustical interfaces are used as for option UPL-B9.

The cable with male (analyzer) and female (generator) XLR connector which is supplied for connection to the "Speech" connector of the Digital Radio Communication Tester CMU is wired for the input and output of link handler 1 in the CMU and can thus not be be used for CDMA tests because input and output of link handler 2 are used in this case. So it is necessary to make a new cable or to change the wiring in the existing cable.

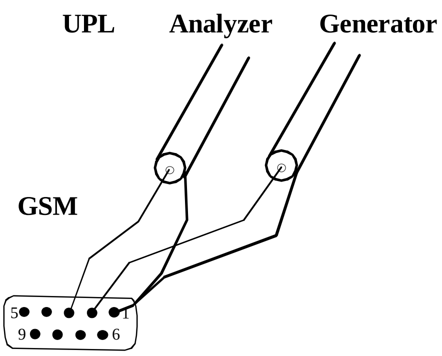

Fig. 1 Assignment of 9-contact speech connector on CMU front panel of existing cable for GSM connection.

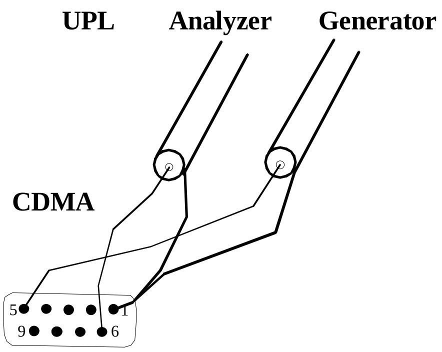

Fig. 2 Assignment of 9-contact speech connector on CMU front panel for CDMA connection.

The Radio Communication Tester CMU200 must be equipped with the following options:

Signaling Unit	R&S CMU-B83	1150.0301.12
Speech Codec	R&S CMU-B85	1100.7002.12

Depending on the band used:

Software CDMA 450 MHz band	R&S CMU-K83
Software CDMA Cellular Band	R&S CMU-K84
Software CDMA PCS Band	R&S CMU-K85
Software CDMA IMT2000 Band	R&S CMU-K86

3 Installing the Software

The application software consists of the files CDMAINST.BAT and CDMA.LZH, both files must be copied to an installation floppy. The program is installed with the aid of the CDMAINST.BAT installation program.

Caution: *The program runs only on an Audio Analyzer UPL with option UPL-B9 already installed.*

- **Quit the measurement software by pressing the SYSTEM key on the instrument or Ctrl + F9 on the keyboard.**
- **Insert the installation floppy.**
- **Select floppy disk drive (enter A:).**
- **Call the installation program CDMAINST.**
  Return to the UPL program (enter C:\UPL).

The CDMAINST program unpacks the installation files and copies all BASIC macros into the directory C:\3GPP.

4 Measuring CDMA mobile phones

The measurement of CDMA mobile phones is similar to the procedure described for GSM mobile phones in the manual for UPL-B9, please refer to this manual for general handling.

Load and Start the program CDMA_TST.BAS

Depending on the desired network and Service options set all necessary parameters in the Radio Communication Tester CMU200 and make a call setup.

Caution: *Only the speech coder options marked with LOW can be used, only those coder versions give correct calibrated levels!*

The following speech coders are available at the moment:

- 8k QCELP (Low)
- 8k enhanced (Low)
- 13k QCELP (Low)

All acoustical calibration values are taken from the program 3GPP_TST.BAS so no new calibration is necessary as long as mouth and ear were already calibrated before.
The coder and decoder sensitivity can be different for the CDMA speech coder so it is necessary to do the speech coder calibration in CDMA mode. Those calibration values are stored independently from the speech coder values measured with program 3GPP_TST.BAS so it is possible to change between GSM and CDMA measurements without further recalibration of the coder.

Difference to standard 3GPP TS 26.132:

According to 3GPP TS 26.132 the sending and receiving distortion measurements are normally done with pulsed sinusoidal signals at 1015 Hz. At this special frequency some codecs used for CDMA coding show a very bad SINAD value which would generally lead to a FAIL for the distortion measurement. It was found that at lower frequencies as well as at higher frequencies the codec has less problems with the coding of the signal. The intention of distortion measurement is to verify the microphone or the speaker with its circuitry and not the coder, therefore the test frequency was changed to 500 Hz for the distortion in sending direction and 800 Hz for receiving direction of CDMA phones. The higher frequency in receiving direction was chosen because at lower frequencies some speakers may run into problems with higher sound pressure levels.

Test and Measurement
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3GPP Hand-Held Handsfree
MOBILE PHONE TESTS

Addendum to Option R&S UPL-B9

Version 1.0

Printed in the Federal
Republic of German

1 Overview

The acoustic measurement of 3GPP and GSM mobile phones using Audio Analyzer R&S UPL with option R&S UPL-B9 together with Radio Communication Tester R&S CMU200 is a standard test solution used worldwide according to the standards 3GPP TS26.131 and 3GPP TS26.132. Those tests are validated for type approval.

More and more mobile phones now have the possibility to be operated in handsfree mode. The mobile can lie on the desk or can be held in the hand at some distance during a telephone call. This feature enables that several persons can listen to the conversation or that the user can make notes or uses his laptop during the call without handling problems of the mobile phone.

The standards 3GPP TS 26.131 and TS 26.132 define requirements and test specifications for the tests of the acoustic behaviour of the mobile phone in this mode but not all details are specified yet.

With this update of the option R&S UPL-B9 all already defined acoustic tests for hand-held handsfree mobile phones can be done.

Those tests are additional programs installed in option R&S UPL-B9 and do not affect the validated testcases for the normal handset tests.

2 Required Measuring Instruments and Accessories

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

- Extended Analysis Functions R&S UPL-B6
- Universal Sequence Controller R&S UPL-B10
- 3G Mobile Phone Tests R&S UPL-B9 (version 1.0)

The Radio Communication Tester R&S CMU200 must be equipped with the following options:

- Signalling Unit R&S CMU-B21
- Speech Codec R&S CMU-B52

Depending on the band used:

- Software R&S CMU-K20 – K24

For the tests of the hand-held handset mode the artificial mouth according to ITU-T P51 (same as for handset measurements) and a free field microphone is needed. The microphone power supply and preamplifier used for the normal handset tests can be used as well but an additional free field calibrated microphone capsule like G.R.A.S. 40AC or B&K 4191 is needed.

All test must be done in an anechoic room (ambient noise less than 24 dBspl(A)), the room should be big enough to give nearly free field conditions at the hands free reference point. A normal testbox can usually not be used for handsfree measurements!

3 Installing and starting the Software

The application software consists of the files HHHFINST.BAT and HHHF.LZH, both files must be copied to an installation floppy. The program is installed with the aid of the HHHFINST.BAT installation program.

Caution: *The program runs only on an Audio Analyzer R&S UPL with option R&S UPL-B9 already installed.*

- **Quit the measurement software by pressing the SYSTEM key on the instrument or Ctrl + F9 on the keyboard.**
- **Insert the installation floppy.**
- **Select floppy disk drive (enter A:).**
- **Call the installation program HHHFINST.**
  Return to the R&S UPL program (enter C:\UPL).

The HHHFINST program unpacks the installation files and copies all BASIC macros into the directory C:\3GPP.

Load and Start the program 3G_HHHF.BAS

Set all necessary parameters in the Radio Communication Tester CMU200 and make a call setup.

Caution: *Only the bit stream setting HANDSET LOW can be used!*

4 Definitions according to 3GPP standards

The following text parts referring to handsfree testing are excerps from the standards 3GPP TS26.131 and 3GPP TS26.132

Positioning and definition of Handsfree reference point according to 3GPP TS 26.132:

5.1.3.3 Handheld hands-free

Either HATS or a free-field microphone with a discrete P. 51 artificial mouth may be used to measure Hand-Held Hands-free type UE.

The current version for hand-held handsfree testing only supports the use of discrete P.51 mouth and free field microphone.

If a free-field microphone with a discrete P. 51 mouth are used, they should be configured to the Hand-Held Hands-free UE as per Figure 5 for receiving measurements and Figure 6 for sending measurements. The measurement instrument should be located at a distance d_{HF} from the centre of the visual display of the Mobile Station. The distance d_{HF} is specified by the manufacturer.

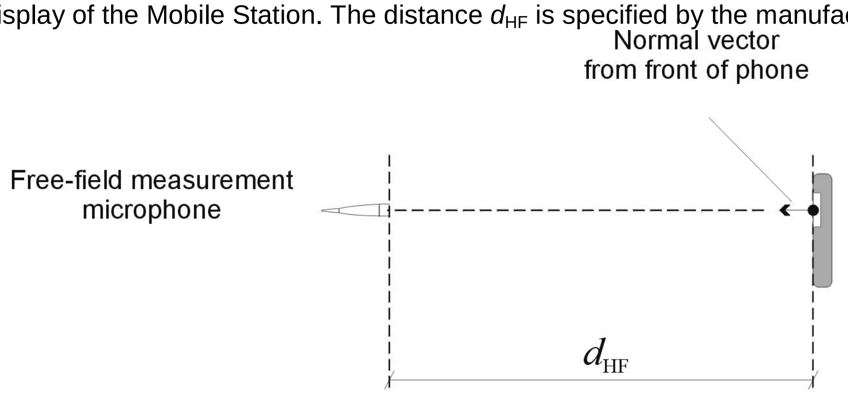

Figure 5: Configuration of Hand-Held Hands-free UE, free-field microphone for receiving measurements.

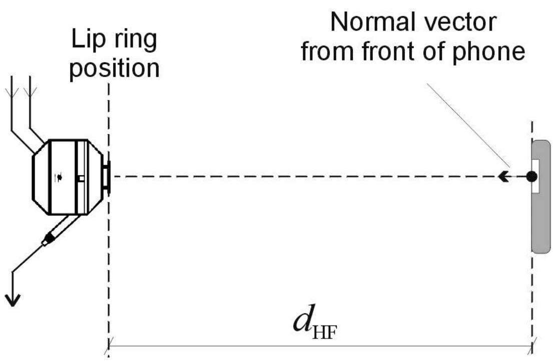

Figure 6: Configuration of Hand-Held Hands-free UE, discrete P. 51 artificial mouth for sending measurements.

6.1.2 Hands-free terminals

Hands-free terminals generally should be tested in their typical environment of application. Care must be taken, that e.g. noise levels are sufficiently low in order not to interfere with the measurements.
For Desk-Top hands-free terminals the appropriate requirements shall be taken from ITU-Recommendation P.340.
The broadband noise level shall not exceed –70 dBPa(A). The octave band noise level shall not exceed the values specified in Table 2.

TABLE 2/P.340 Noise level

Center frequency (Hz)	Octave band pressure level (dBPa)
63	–45
125	–60
250	–65
500	–65
1 k	–65
2 k	–65
4 k	–65
8 k	–65

Echo measurements shall be conducted in realistic rooms with an ambient noise level less then -70 dBPa(A).

Testcase Sending and Receiving Loudness Rating:

- **Requirement according to 3GPP TS 26.131:**

5.2.4 Connections with handheld hands-free UE

The nominal values of SLR/RLR to/from the POI shall be:

SLR = 13 +/- 4 dB;

RLR = 6 +12 / - 4 dB.

Compliance shall be checked by the relevant tests described in TS 26.132.
Where a user controlled volume control is provided, the RLR shall meet the nominal value at one setting of the control.

- **Tests according to 3GPP TS 26.132:**

7.2.4 Connections with Handheld hands-free UE

7.2.4.1 Sending Loudness Rating (SLR)

a) The test signal to be used for the measurements shall be the artificial voice according to ITU-Recommendation P. 50 or a speech like test signal as described in ITU-T Recommendation P.501. The type of test signal used shall be stated in the test report. The spectrum of acoustic signal produced by the artificial mouth is calibrated under free field conditions at the MRP.

The test signal level shall be –4,7 dBPa, measured at the MRP. The test signal level is averaged over the complete test signal sequence. The broadband signal level then is adjusted to –28,7 dBPa at the HFRP or the HATSHFRP (as defined in P. 581) and the spectrum is not altered. The spectrum at the MRP and the actual level at the MRP (measured in third octaves) is used as reference to determine the sending sensitivity S_{mJ}.

b) The hands-free terminal is setup as described in subclause 5.1.3.3. The sending sensitivity shall be calculated from each band of the 14 frequencies given in table 1 of ITU-T Recommendation P.79, bands 4 to 17. For the calculation the averaged measured level at the electrical reference point for each frequency band is referred to the averaged test signal level measured in each frequency band at the MRP.

c) The sensitivity is expressed in terms of dBV/Pa and the SLR shall be calculated according to ITU-T Recommendation P.79, formula (A-23b), over bands 4 to 17, using m = 0,175 and the sending weighting factors from ITU-T Recommendation P.79, table 1.

7.2.4.2 Receiving Loudness Rating (RLR)

a) The test signal to be used for the measurements shall be the artificial voice according to ITU-Recommendation P. 50 or a speech like test signal as described in ITU-T Recommendation P.501. The type of test signal used shall be stated in the test report. The test signal level shall be –16 dBm0, measured at the digital reference point or the equivalent analogue point. The test signal level is averaged over the complete test signal sequence.

b) The hands-free terminal is setup as described in subclause 5.1.3.3. If HATS is used then it is freefield equalized as described in ITU-T Recommendation P.581. The equalized output signal of each artificial ear is power-averaged on the total time of analysis; the “right ” and “left” signals are voltage-summed for each 1/3 octave band frequency band; these 1/3 octave band data are considered as the input signal to be used for calculations or measurements. The receiving sensitivity shall be calculated from each band of the 14 frequencies given in table 1 of ITU-T Recommendation P.79 , bands 4 to 17.

For the calculation the averaged measured level at each frequency band is referred to the averaged test signal level measured in each frequency band.

c) The sensitivity is expressed in terms of dBPa/V and the RLR shall be calculated according to ITU-T Recommendation P.79 [16], formula (A-23c), over bands 4 to 17, using m = 0,175 and the receiving weighting factors from table 1 of ITU-T Recommendation P.79.

d) No leakage correction shall be applied. The hands-free correction as described in P.340 shall be applied. To compute the Receiving loudness rating (RLR) for hands-free terminals (see also ITU-T Recommendation P.340) when using the combination of left and right ear signals from HATS the HFL_E has to be 8 dB, instead of 14 dB. For further information see ITU-T Recommendation P.581.

Testcase Sending and Receiving Frequency Response:

- **Requirement according to 3GPP TS 26.131:**

5.4.5 Handheld hands-free UE sending

The sending sensitivity frequency response from the MRP to the SS audio output (digital output of the reference speech decoder of the SS) shall be as follows:
The sending sensitivity frequency response shall be within the mask which can be drawn with straight lines between the breaking points in table 5 on a logarithmic (frequency) - linear (dB sensitivity) scale.

Table 5: Hands-free sending sensitivity/frequency response

Frequency (Hz)	Upper limit	Lower limit
200	0	
250	0	
315	0	-14
400	0	-13
500	0	-12
630	0	-11
800	0	-10
1 000	0	-8
1 300	2	-8
1 600	3	-8
2 000	4	-8
2 500	4	-8
3 100	4	-8
4 000	0	

NOTE: All sensitivity values are expressed in dB on an arbitrary scale.

Compliance shall be checked by the relevant test described in TS 26.132.

5.4.6 Handheld hands-free UE receiving

The receiving sensitivity frequency response from the SS audio input (analogue or digital input of the reference speech encoder of the SS) to the ERP shall be as follows:
The receiving sensitivity frequency response shall be within the mask which can be drawn with straight lines between the breaking points in table 6 on a logarithmic (frequency) - linear (dB sensitivity) scale.

Table 6: Hands-free receiving sensitivity/frequency response

Frequency (Hz)	Upper limit	Lower limit
200	0	
250	0	
315	0	
400	0	
500	0	
630	0	
800	0	-12
1 000	0	-12
1 300	0	-12
1 600	0	-12
2 000	0	-12
2 500	0	-12
3 100	0	-12
4 000	0	

NOTE: All sensitivity values are expressed in dB on an arbitrary scale.

Compliance shall be checked by the relevant test described in TS 26.132.

- **Tests according to 3GPP TS 26.132:**

7.4.5 Hand-Held hands-free UE sending

a) The test signal to be used for the measurements shall be the artificial voice according to ITU-Recommendation P. 50 or a speech like test signal as described in ITU-T Recommendation P.501. The type of test signal used shall be stated in the test report. The spectrum of acoustic signal produced by the artificial mouth is calibrated under free field conditions at the MRP. The test signal level shall be –4,7 dBPa, measured at the MRP. The test signal level is averaged over the complete test signal sequence. The broadband signal level then is adjusted to –28,7 dBPa at the HFRP or the HATSHFRP (as defined in P. 581) and the spectrum is not altered.
The spectrum at the MRP and the actual level at the MRP (measured in third octaves) is used as reference to determine the sending sensitivity S_{mJ}.

b) The hands-free terminal is setup as described in subclause 5.1.3.3. Measurements shall be made at one third-octave intervals as given by the R.40 series of preferred numbers in ISO 3 [17] for frequencies from 100 Hz to 4 kHz inclusive. For the calculation the averaged measured level at each frequency band is referred to the averaged test signal level measured in each frequency band.

c) The sensitivity is expressed in terms of dBV/Pa.

7.4.6 Hand-Held hands-free UE receiving

a) The test signal to be used for the measurements shall be the artificial voice according to ITU-Recommendation P. 50 or a speech like test signal as described in ITU-T Recommendation P.501. The type of test signal used shall be stated in the test report. The test signal level shall be –16 dBm0, measured at the digital reference point or the equivalent analogue point. The test signal level is averaged over the complete test signal sequence.

b) The hands-free terminal is setup as described in subclause 5.1.3.3. If the HATS is used then it is freefield equalized as described in ITU-T Recommendation P.581 . The equalized output signal of each artificial ear is power-averaged on the total time of analysis; the “right ” and “left” signals are voltage-summed for each 1/3 octave band frequency band; these 1/3 octave band data are considered as the input signal to be used for calculations or measurements. Measurements shall be made at one third-octave intervals as given by the R.40 series of preferred numbers in ISO 3 [17] for frequencies from 100 Hz to 4 kHz inclusive. For the calculation the averaged measured level at each frequency band is referred to the averaged test signal level measured in each frequency band.

c) The sensitivity is expressed in terms of dBPa/V.

Tests and definitions according to 3GPP TS 26.132:

7.5 Sidetone characteristics

7.5.3 Hands-free UE (all categories)

No requirement for other than echo control.

7.6 Stability loss

Hands-free UE (all categories): no requirement other than echo loss.

Testcase Echo Loss:

- **Requirements according to 3GPP TS 26.131:**

5.7.3 Acoustic echo control in an handheld hands-free UE

The TCLw for the hands-free UE shall be 40 dB at the nominal setting of the volume control in quiet background conditions and 33 dB at the maximum user selectable volume control setting. If acoustic echo control is provided using some form of echo cancellation technique, the cancellation algorithm should be designed to cope with the expected reverberation and dispersion. In the case of the hands-free UE, this reverberation and dispersion may be time variant. Compliance with this requirement shall be checked by the relevant test described in TS 26.132.

- **Tests according to 3GPP TS 26.132:**

7.7.2 Acoustic echo control in a Hands-free UE

TCLw:
The hands-free is setup in a room where it is intended to be used, eg. for an office type hands-free UE a typical “office-type” room should be used; a vehicle-mounted hands-free UE should be tested in a vehicle or vehicle simulator, as specified by the UE manufacturer. [For reference on a suitable vehicle simulator see ETSI 0358 601 (TR101110) Digital Cellular Telecommunications System (Phase 2+) .]
The ambient noise level shall be less than -70 dBPa(A). The attenuation from reference point input to reference point output shall be measured using a speech like test signal .
Before the actual test a training sequence consisting of 10 s artificial voice male and 10 s artificial voice female according to ITU-T Recommendation P.50 is altered.
Either a logarithmically spaced multi-sine or PN-sequence test signal shall be used.
When using a logarithmically spaced multi-sine test signal, it is defined as:

$$s(t) = \sum_i \left[\left[A + \mu_{AM} \cos(2\pi t * f_{AM}) \right] * \cos(2\pi t * f_{0i}) \right]$$

with A = 0,5
f_{AM} = 4 Hz, μ_{AM} = 0,5
$f_{0\,i}$= 250Hz * $2^{(i/3)}$;i=1..11
CF= 14dB ±1 dB (10 dB + 4,26 dB due to 100% AM modulation)

CF = Crest Factor = Peak to RMS ratio
The training sequence level shall be –16 dBm0 in order not to overload the codec. The test signal level shall be -10 dBm0. The TCLw is calculated according to ITU-T Recommendation G.122 [8], annex B, clause B.4 (trapezoidal rule). For the calculation the averaged measured echo level at each frequency band is referred to the averaged test signal level measured in each frequency band. The length of the test signal shall be at least one second (1,0 s).
Note:
Full scale of coder input signal corresponds to +3,14 dBm0 with sinusoidal signal, CF= 3dB. A test signal with a CF of maximum 15 dB can thus have a level of up to –8,86 dBm0 without overloading the codec. In order to get best dynamic range the signal amplitude should be as high as possible.
When using a PN-sequence, it should comply with ITU-T Recommendation P.501 with a length of 4096 points (for the 48 kHz sampling rate) and a crest factor of 6 dB. The duration of the test signal is 250ms. The test signal level is –3 dB_{m0}. The low-crest factor is achieved by random-alternation of the phase between -180° and 180°.
The training sequence level shall be –16 dBm0 in order not to overload the codec. The TCLw is calculated according to ITU-T Recommendation G.122 [8], annex B, clause B.4 (trapezoidal rule). For the calculation the averaged measured echo level at each frequency band is referred to the averaged test signal level measured in each frequency band.
Care should be taken that the terminal under test considers the test signal as a speech-like signal.

Testcase Sending and Receiving Distortion:

- **Requirements according to 3GPP TS 26.131:**

No definition for handheld hands-free

- **Tests according to 3GPP TS 26.132:**

7.8.1 Sending Distortion

The handset, headset, or hands-free UE is setup as described in clause 5.
A sine-wave signal with a frequency in the range 1 004 Hz to 1 025 Hz is applied at the MRP.
The level of this signal is adjusted until the output of the terminal is -10 dBm0. The level of the signal at the MRP is then the ARL.
The test signal shall be applied at the following levels: -35, -30, -25, -20, -15, -10, -5, 0, 5, 10 dB relative to ARL.
The ratio of the signal to total distortion power of the signal output of the SS shall be measured with the psophometric noise weighting (see ITU-T Recommendations G.712 and 0.132).
NOTE: It should be ensured that the test signal is treated by speech processing algorithms as a speech-like signal, and not a noise-like signal. Test signals with a time-stationary envelope may be treated by certain algorithms, e.g. noise suppression algorithms defined in 3GPP TS 06.77 R99[17], as a noise-like signal.

7.8.2 Receiving

The handset, headset, or hands-free UE is setup as described in clause 5.
A sine-wave signal with a frequency in the range 1 004 Hz to 1 025 Hz shall be applied at the signal input of the SS at the following levels: -45, -40, -35, -30, -25, -20, -15, -10, -5, 0 dBm0.
The ratio of the signal-to-total distortion power shall be measured at the ERP with the psophometric noise weighting (see ITU-T Recommendations G.712 and 0.132).
NOTE: It should be ensured that the test signal is treated by speech processing algorithms as a speech-like signal, and not a noise-like signal. Test signals with a time-stationary envelope may be treated by certain algorithms, e.g. noise suppression algorithms defined in 3GPP TS 06.77 R99[17], as a noise-like signal.

Tests and definitions according to 3GPP TS 26.132:

7.9 Ambient Noise Rejection

Hands-free UE (all categories):

For further study

5 Measurements

Before the first measurement can be started all acoustic equipment must be calibrated once. The calibration must be done in the real test environment and at the used test position in the chamber in order to calibrate the influence of the chamber as well. All calibration values are stored and automatically used next time. So if the arrangement is unchanged the calibration needs only to be repeated from time to time to check the equipment.

Calibration:

Calibration of Free Field Microphone

The absolute sensitivity of the microphone must determined using a sound level calibrator such as the Brüel & Kjaer 4231 with a sound pressure level of 94 dBSPL or a sound pressure of 1 Pa at 1 kHz.

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

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

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

---CALIBRATION--- EXP-FILES

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

➢ **Call the calibration routines using the MICRO 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 sensitivity displayed must be about 10 times the value in the calibration certificate of the microphone capsule (typical value for microphone capsule 40 AC or B&K 4191 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 the test to be repeated. After switching on the microphone power supply wait approx. 20 seconds before restarting the measurement with RUN.
The reference value measured is stored in a nonvolatile memory and used for all subsequent measurements.

Calibration of Artificial Mouth

The absolute sensitivity and frequency response of the artificial mouth have to be measured and corrected with the aid of the previously calibrated free-field measuring microphone. 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 4 kHz) (see also calibration certificate of microphone capsule).

Attention: If you use a pressure calibrated microphone for the mouth calibration the microphone must be mounted in a right angle position to the mouth reference point (MRP) but if a free field microphone is used it must be fitted in line with the mouth axis!

Fit the microphone in line to the mouth axis at the mouth reference point (MRP) using the gauge supplied with the mouth

Select the CALIBRATION level using the F12 key.

---CALIBRATION--- EXP-FILES

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

- **Call the calibration routines using the MOUTH 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 is 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 non-linearities of the speaker in the mouth are measured and taken into account in the final equalization file as fine correction.

The program then asks for the next calibration point:
Calibration of Sound Pressure Level at Hands Free Reference Point HFRP

The absolute sound pressure level at the HFRP must be set to –28.7 dBPa. This is done in an automatic adjustment routine after placing the measuring microphone at the HFRP.

Attention: The Hands Free Reference point is in axis of the artificial mouth but the actual distance has to be defined by the manufacturer of the mobile phone under test. The typical distance is 50 cm.

Fit the microphone in line to the mouth axis at the hands free reference point (HFRP) using a fixture for the microphone.

Be careful that no obstacles are disturbing the free field conditions and that the chamber is closed as in normal measuring conditions!

- **Continue the calibration using the CONT key.**

The sound pressure generated at the HFRP is set to exactly –28.7dBPa in an automatic measurement routine using the measuring signal according to ITU-TP.501. The generator voltage required is stored in a nonvolatile memory and used as a reference for all subsequent settings

Then for the last calibration point the measuring microphone must be fitted again at the MRP using the test fixture parts. Again the microphone must be placed in line to the mouth axis.

- **Continue the calibration using the CONT key.**

In this step the actual spectrum of the measuring signal at the MRP is measured and stored as reference spectrum for all frequency response measurements.

Calibration of Coder in CMU

The coder calibration is identical to the standard coder calibration for handheld mobile phones so if the coder was already calibrated no additional calibration is necessary.

Measurements: (tests in bold are available)

The selection of the test and the result handling is identical to that for handheld mobile phones, please refer to the manual for option UPL-B9.

SLR: Acoustical signal at HFRP of –28.7 dBPa, calculation of SLR in $1/3^{rd}$ octave bands, test included in sending frequency response.

RLR: Signal with level –16 dBm0, measure with free field microphone at HFRP, calculation in $1/3^{rd}$ octave bands, correction to P.340 with HFLe = -14 dB. Test included in receiving frequency response.

Idle channel noise: Not yet defined for HHHF but tests are important. Tests are identical to handset tests but verdict values must be changed. It was decided to reduce the idle noise verdict values provisionally by 10 dB for handsfree tests (-54 dBm0 for sending and –44 dBPa(A) for receiving).

Frequency response Sending: Acoustical signal at HFRP of –28.7 dBPa, frequency response in $1/12^{th}$ octave bands.

Frequency response Receiving: Signal with –16 dBm0, measure at free field microphone, frequency response in $1/12^{th}$ octave bands.

3GPP Hand-Held Handsfree Mobile Phone Tests

Sidetone: No requirements for HHHF

Stability loss: No requirements for HHHF

Echo Loss: Measurement and calculation as for Handset, other verdict value (33 dB for max volume).

Distortion sending: The test is identical to the handset test but the acoustical reference level is adjusted at –20 dBm0 instead of –10 dBm0 because this point cannot always be reached in handsfree mode. Presumably the verdict values must be changed.

Distortion receiving: The same test as for handset can be used but also the verdict values must be changed in future.

Ambient Noise Rejection: For further study, for tests the existing test for handheld can be used.

ROHDE&SCHWARZ

Test and Measurement
Division

Operating Manual

3G MOBILE PHONE TESTS

UPL-B9

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Version 1.0

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Contents

1 Overview

The acoustic transmission and reproduction quality of a mobile phone is its most important characteristic in everyday use. The most visually appealing design or a 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 so far been required for type-approval testing. 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, new test methods had to be developed allowing to perform measurements via the air interface and the normal voice 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 like VAD (voice activity detector), noise suppression, echo canceller and so on 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 (3G 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.

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
- 3G Mobile Phone Tests UPL-B9 (version 1.0)

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. The radiocommunication tester CMU200 must be equipped with option CMU-B21, versatile signalling unit, option CMU-B52, speech codec and the firmware for the GSM band used. The CMU must be equipped with firmware 3.0 or higher.

Acoustic devices such as an artificial mouth, artificial ear and other accessories are required for the measurements. The following equipment from Brüel & Kjaer or G.R.A.S. is normally used (also HATS can be 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
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. AA12

Note: *With the preamplifier 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 low-noise power supply such as AA12 from G.R.A.S is preferable.*

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

The transformer supplied with the option UPL-B9 is connected between the generator output 1 of the 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 generator output. Without this transformer, the available power is too low for driving the artificial mouth.

Alternatively, a power amplifier preferably with a gain of approx. 0 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.

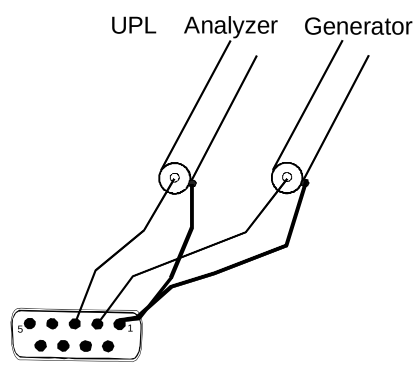

Fig. 1 Assignment of 9-contact speech connector on CMU front panel

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, section 2.15.4).

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 3.00 or higher
- Option Extended Analysis Functions UPL-B6 installed
- Option Universal Sequence Controller UPL-B10 installed
- Option 3G Mobile Phone Tests UPL-B9 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 is installed with the aid of the 3GPPINST.BAT installation program on program floppy 1. The installation number of the 3G Mobile Phone Tests Option 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 3GPPINST).**
  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 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 the UPL program (enter C:\UPL).**
  The 3GPPINST program creates the C:\3GPP 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

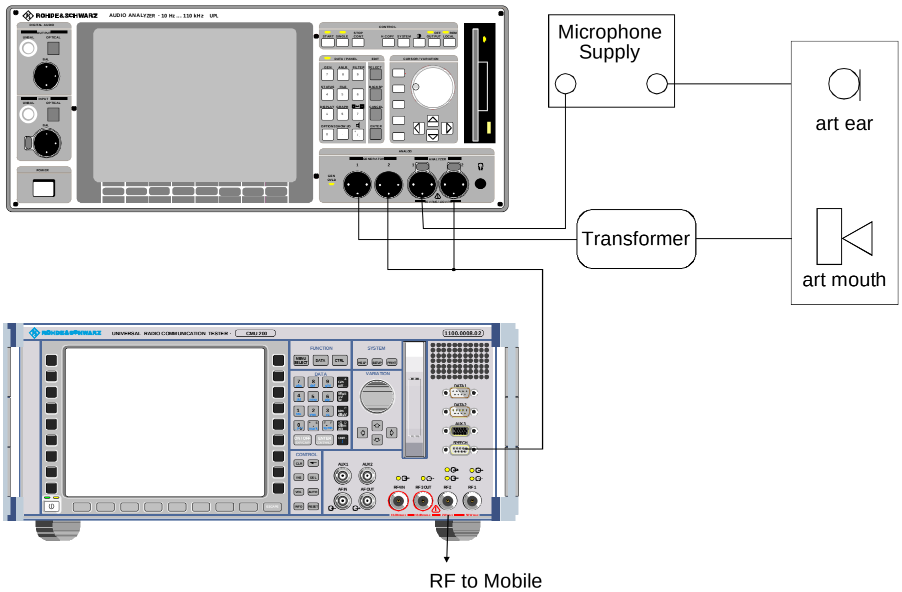

Fig. 2 Test setup and connection of external components

Starting the Application Software

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

The logging function is switched off; check that "logging off" is displayed at the bottom right of the screen; use the F2 key to enable/disable the logging function. 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 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 phones
- 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 mobile phones. It is loaded and started by entering:

- 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, but that the printer used last by the Audio Analyzer UPL 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 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 be used for selecting program routines.

After the program has been started, the title page

Measurement of
GSM 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 CMD

to Bit Stream Handset Low

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 bit stream setting is possible only in the "Call established" state.

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 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 voice 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	STAB-MRG	→

A click on a key starts the test routine associated. 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 towards 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 program can be quit.

4 Measurements

General

Special problems are encountered when measuring acoustic characteristics caused by the coder and decoder algorithms of mobile phones. Vocoders 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 so far been excluded from the measurement. A 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, so that direct transmission of the test signals to and from a mobile phone under test with DAI interface is possible.

In commercially available mobile phones, this interface is not accessible so that measurements during normal operation can only be performed via the air interface with the voice 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. artificial voice to ITU-T P.50 or a multitone signal to ITU-T P.501. 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 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 test chamber with sufficient absorption against interfering sound. Since special distortion measurements and the measurement of idle noise set 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 position (loudness rating guard ring position to ITU-T 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 CMU voice decoder output and evaluated.

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

Table 1 Tolerances specified 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 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, 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 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 1 of ITU-T P.79.

Table 2 Frequencies of bands 4 to 17 according to table 2 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 referred to the rated internal level in dBm0, and the sending loudness rating is calculated according to formula 2.1 of ITU-T P.79.

Due to the input sensitivity tolerance of the CMU voice 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 will be output only if both the curve and the loudness value are within tolerances.

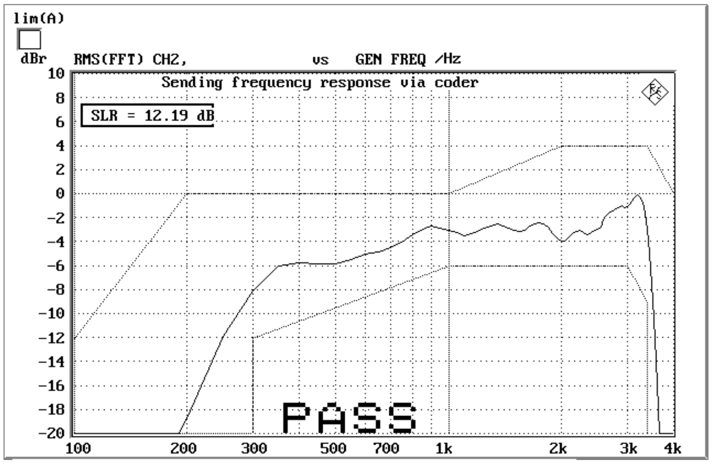

Fig. 3 Sending frequency response with SLR value displayed

Receiving Frequency Response and Loudness Rating

The following two routines are available since 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 voice coder input of the CMU. The measured sound pressure is referred to the ear reference point (ERP). For ear type 1, the measuring microphone is directly applied to the ERP and a further correction is not 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 into the ERP by means of calibration factors.

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

The voice coder is driven so 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 when a straight line is drawn between the specified values, and a logarithmic frequency scale and a linear dB scale are used.

When ear type 1 is used, the receiving frequency response must be within the tolerances specified to table 30.2 of 3GPP TS 51.010. When ear type 3.x is used, it must be within the tolerances specified to 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 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 (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 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 a ratio in dBPA/V referred to the rated internal signal level, and the receiving loudness rating is calculated according to formula 2.1 of ITU-T P.79.

Due to the input sensitivity tolerance of the CMU voice coder, 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 may not fall below -13 dB when maximum loudness is set on the phone, i.e. the maximum receiving loudness may not exceed a certain value to avoid damage to the human ear.

The RLR measured 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 will be output only if both the curve and the loudness value are within the tolerances.

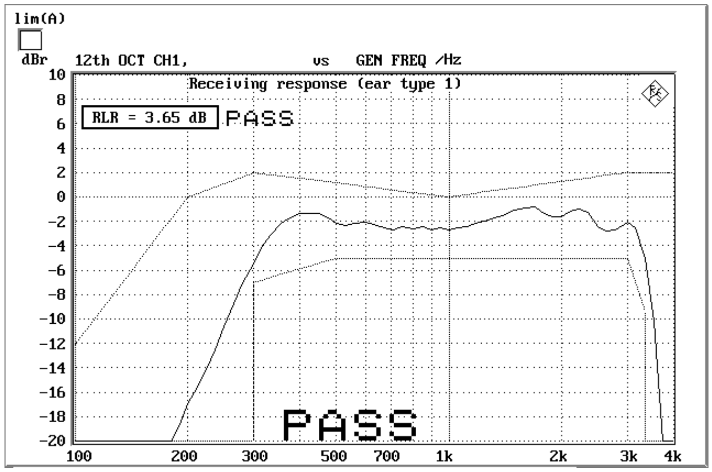

Fig. 4 Receiving frequency response with RLR value displayed

Sidetone Masking Rating (STMR)

The sidetone path is the deliberate output of 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 that involve an acoustic path between mouth and ear.

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

The STMR can only be measured with ear type 1 or ear type 3.2 Low Leakage.

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 1 of ITU-T P.79, and the sidetone masking rating (STMR) is calculated according to formula 2.1 of ITU-T P.79 with the weighting factors of table 3 of ITU-T P.79 taken into account.

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

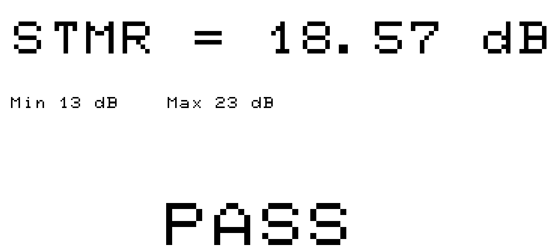

Fig. 5 Typical measurement of sidetone masking rating

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 may not exceed a certain value.

The mobile phone under test is suspended in free air in the anechoic chamber (former releases used the LRGP position!).

A modulated multitone signal to ITU-T P.501 is generated as a test signal and applied to the voice 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 voice decoder is measured. The echo loss is calculated from the differences of the individual 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 voice coder and decoder must also be considered in the result. This value is available in the CMU after calibration of the coder.

3GPP TS 26.131 specifies an echo loss of at least 46 dB which can be achieved by mobile phones using good echo cancellers. 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.

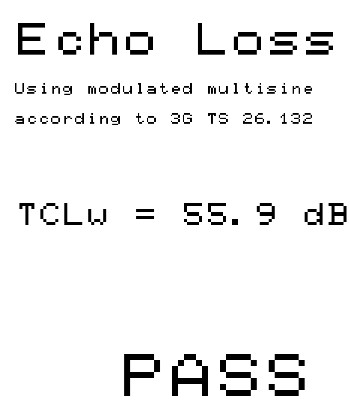

Fig. 7 Typical result of echo loss measurement

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 whether resonances or oscillations are produced. If there are no oscillations, the minimum requirements to 3GPP TS 26.131 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.

A pulsed sinusoidal tone of 1015 Hz with a pulse lenght of approx. 250 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 position (ITU-T 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 CMU’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 7 of 3GPP TS 26.131.

Table 5 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.

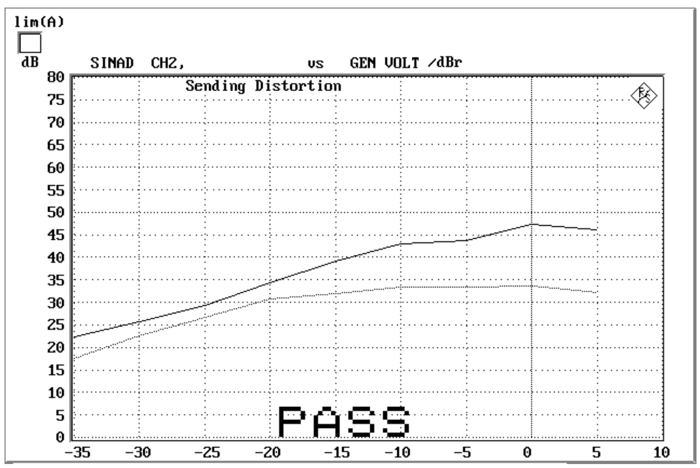

Fig. 7 Sending distortion measurement

Receiving Distortion

The S/N ratio in the receiving path is measured as a function of the sound 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 position (ITU-T P.76), and the speaker is sealed to the artificial ear.

The test signal is applied to the input of the CMU voice coder, and the SINAD value of the sound pressure in the artificial ear is measured with psophometric weighting 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 limit lines given in table 8 of 3GPP TS 26.131.

Table 6 Limit lines given 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 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, PASS is output, otherwise FAIL.

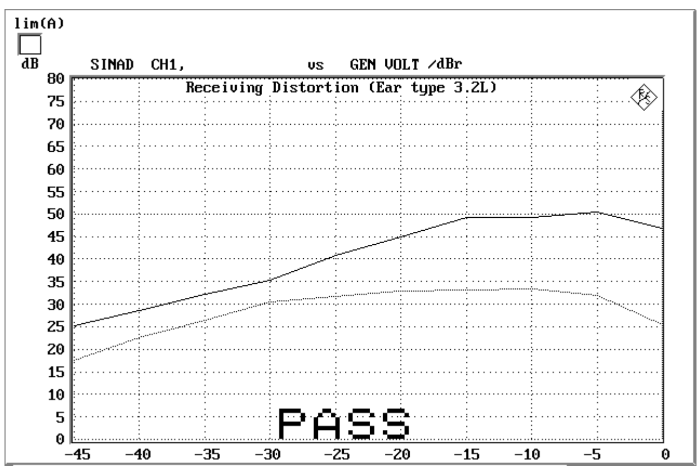

Fig. 8 Typical result of receiving distortion measurement

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

The decoder output voltage is measured, psophometrically weighted according to ITU-T G.223 and calculated at 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 is activated and the mobile phone remains in the active normal sending mode.

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.

The mobile phone under test is installed in the LRGP position (ITU-T P.76), and the speaker 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 voice coder input. The noise level in the artificial ear 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).

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 mobile phone under test 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 (ANR) 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 to 3GPP TS 26.131 is 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. In order to obtain a sufficiently homogeneous sound field, several uncorrelated generators and loudspeakers are required. The use of 4 or 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 position (ITU-T P.76), and the speaker is sealed to the artificial ear.

Set up a call to the CMU and set the bit stream to "Handset 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 values must exceed -3 dB.

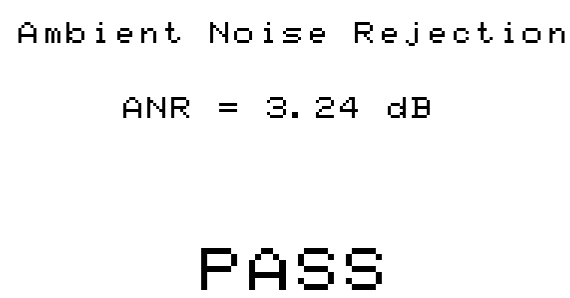

Fig. 9 Typical result of ambient noise rejection measurement

5 Calibration Routines

All calibration values are stored on the harddisk of the UPL and are thus nonvolatile. The storage of the calibration values is independent of other UPL options.

Calibration of Artificial Ear

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

Note:

The calibration values of the different ear types are stored separately. So, 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 at least 10 s to allow for static pressure compensation.*

- **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_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 sensitivity displayed must be about 10 times 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, 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 the test to be repeated. After switching on the microphone power supply wait approx. 20 seconds 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 noise 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 the test to 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 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 using 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 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_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 the test to 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 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 using 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 the test to 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 ear type 3.3. The standard calibration data 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 the test to 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 ear type 3.4.

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

Calibration of Artificial Mouth

The 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 measuring 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 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 must be calibrated in a 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 into the calibrator wait at least 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 sensitivity displayed must be about 10 times 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, 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 the test to 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 is 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 non-linearities 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 Voice Coder

The calibration of the voice coder and decoder is required 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.

Auxiliary settings required for calibration can be found in the CMU under bit stream (firmware version 3.0 or higher). Since this menu is accessible only for active call, a call to a mobile phone has to be established first.

➢ **Select the CALIBRATION level with 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 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						

Establish call to mobile phone, set bit stream on 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 voice coder is measured and the input voltage required for digital full scale is measured at the voice coder and saved in the UPL. At the same time, the loop gain from the voice coder input to the voice coder output is calculated and saved.

6 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 the C:\3GPP\RESULTS directory. This path is defined in the 3GPP_TST.BAS program in the Path$ variable and can be modified, if required.

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. The user can thus see any time whether the file has already been saved.

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

When the ABS_FILE key is pressed, the absolute measured values are saved in export format in a file. This file has the name ABSxx.EXP, xx representing a consecutive number (of max. 5 digits). This file can directly be 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, xx representing a consecutive number (of max. 5 digits). This file can directly be processed by spreadsheet programs such as EXCEL.

When the TRC_FILE key is pressed, the trace displayed is saved in ASCII format in a file. This file has the name TRCxx.TRC, xx representing 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, xx representing 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 and saved under another name (as standard in the C:\3GPP\RESULTS directory) using common DOS commands. Entering EXIT <RETURN> brings back BASIC without the program being deleted. 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 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 the 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 save it under a new name. 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 file RES_3GPP.LOG can be copied and saved under another name using common DOS commands.

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

Old result files of a measurement sequence may 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 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 delete procedure these backup copies are overwritten.

7 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.

Produkte: Audio Analyzer R&S UPL Signal Generator R&S SML

Messungen an FM-Radiotunern mit dem Audio Analyzer R&S UPL und dem Signalgenerator R&S SML mit Option SML-B5

Application Note 1GA43_0D

Um die Audioqualität von FM-Rundfunk-Tunern zu ermitteln ist eine ganze Reihe von Messungen notwendig. In der vorliegenden Applikationsschrift wird ein Programm vorgestellt, das diese Messungen gemäß der Norm DIN EN 60315-4 ermöglicht.

ROHDE & SCHWARZ

Änderungen vorbehalten – 1GPP Betz 01,2003– 1GA43_0D

Inhalt

1 Überblick

Um die Audioqualität von FM-Rundfunk-Tunern zu ermitteln ist eine ganze Reihe von Messungen notwendig. In der vorliegenden Applikationsschrift wird ein Programm vorgestellt, das diese Messungen gemäß der Norm DIN EN 60315-4 ermöglicht.

2 Funktionsprinzip

Der Audio Analysator R&S UPL kann alle notwendigen Testsignale erzeugen. Mit diesen Signalen wird der Stereocoder (Option R&S SML-B5) im Signalgenerator gespeist und damit das Sendesignal moduliert. Dieses Signal wird dem Antenneneingang des Prüflings zugeführt und die demodulierten Audiosignale am Ausgang des Prüflings werden dem Audio Analysator zur Messung zugeführt.

Mit der Option Selbststeuerung R&S UPL-B10 kann der Audio Analysator R&S UPL komplette Messabläufe automatisch ablaufen lassen und dabei noch über die IEC-Bus- oder RS232-Schnittstelle den Signalgenerator mitsteuern. Damit sind auch Messungen wie z.B. über HF-Signalpegel automatisch durchführbar.

3 Anforderungen an Hardware und Software

Erforderliche Geräte und Hilfsmittel

Für die Erzeugung und Messung der Audiosignale wird ein Audio Analysator R&S UPL mit Option R&S UPL-B10 benötigt. Zur Erzeugung der Hochfrequenzsignale dient der Signalgenerator R&S SML (alternativ SMV). Er muß mit der Option R&S SML-B5, Stereo/RDS-Coder ausgerüstet sein.

Außerdem wird eine externe Tastatur und ggf. ein Drucker benötigt.

Die für den automatischen Ablauf erforderlichen BASIC-Programme befinden sich auf einer Diskette, die Sie bei Ihrer örtlichen Rohde & Schwarz-Niederlassung erhalten oder die Sie als File von der Download-Area der Rohde & Schwarz-Web-Site heruntergeladen, entpackt und auf eine Diskette kopiert haben. Folgende Software-Voraussetzungen müssen vom R&S UPL erfüllt sein:

- R&S UPL-Firmware Version 3.01 oder höher,
- Selbststeueroption R&S UPL-B10 installiert,
- Der R&S UPL ist mit 64 kBytes Programm- und 32 kBytes Datenspeicher konfiguriert (mittels Konfigurationstool R&S UPLSET Einstellung 3 wählen).

4 Messaufbau

Benötigt werden der Audio Analyzer R&S UPL, der Signalgenerator R&S SML, sowie evtl. ein HP-Deskjet kompatibler Drucker zur Dokumentation der Ergebnisse.

Audio Analyzer und Signalgenerator sind über den IEC-Bus zu verbinden. Für spezielle Messmodule stehen auch Routinen zur Verfügung, die den Signalgenerator über die RS232-Schnittstelle steuern. Messaufgaben können so über den IEC-Bus als Makro im R&S UPL gestartet werden und die notwendigen Einstellungen werden im R&S SML über die RS232-Schnittstelle vom R&S UPL aus vorgenommen. Da IEC-Bus- und RS232-Steuerung im R&S SML wechselweise erfolgen kann, steht der R&S SML und der R&S UPL nach Abarbeitung eines Messmakros über IEC-Bus auch für weitere Aufgaben im System zur Verfügung, Details siehe Anhang A.

Der Drucker wird an die Centronics-Schnittstelle des R&S UPL angeschlossen.

Der HF-Ausgang des R&S SML wird an den unsymmetrischen Antennen-Eingang des Tuners angeschlossen. Da die Ausgangsimpedanz des R&S SML 50 Ω beträgt, muß der Generator an den Tuner angepasst werden. Üblicherweise wird der koaxiale 75 Ω-Antenneneingang verwendet werden, wir empfehlen hier die Verwendung des Rohde & Schwarz Anpassgliedes vom Typ RAM, das unter der Bestellbezeichnung 358.5414.02 bezogen werden kann.

Sollte der zu messende Empfänger nur über einen symmetrischen Eingang verfügen, so muß ein entsprechender Symmetrieübertrager mit 240 Ω bzw. 300 Ω Impedanz dazwischengeschaltet werden. Auf alle Fälle muß die Einfügungsdämpfung dieser Anpasselemente berücksichtigt werden, da sich die Messungen auf den HF-Pegel am Antenneneingang des Tuners beziehen und nicht auf den Ausgangspegel des Generators. Der tatsächliche Dämpfungswert muß unter SETUP eingegeben werden und wird dann vom Programm entsprechend berücksichtigt. Die Dämpfungswerte können aus den Unterlagen zu den Anpassgliedern entnommen werden, bzw. sind auf diesen aufgedruckt.

Der Generatorausgang 1 des Audio Analyzers R&S UPL wird an den externen Modulationseingang links und der Generatorausgang 2 an den Modulationseingang rechts an der Rückseite des Signalgenerators R&S SML angeschlossen. Der Ausgang des Tuners für den linken Kanal wird mit dem Analysatoreingang 1 und der rechte Kanal mit Eingang 2 des R&S UPL verbunden.

Bei allen Messungen ist auf richtige Erdung zu achten, um z.B. Brummschleifen zu vermeiden. Da Tuner üblicherweise nicht mit Schutzerde angeschlossen werden und daher floatende Ausgänge haben, sollten die Eingänge des R&S UPL geerdet werden wenn nicht durch den Antennenanschluß an den R&S SML eine Erdung herbeigeführt wird. Mit der Auswahl "R&S UPL Input Selection" kann dies im Menüpunkt SETUP gewählt werden.

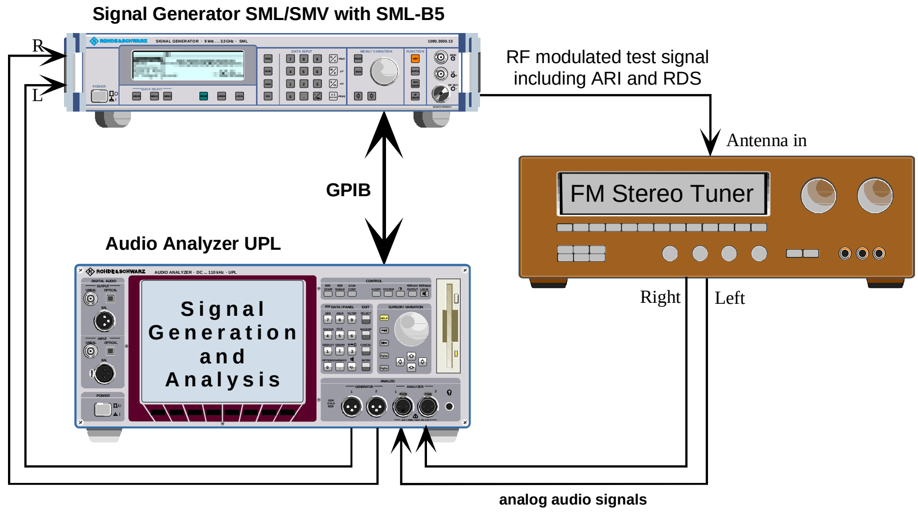

Bild 1: Messaufbau für die Tuner-Messung

5 Installieren der Software

Die Installation der Applikations-Software erfolgt mit Hilfe des Installationsprogramms INSTALL.BAT, das ebenfalls auf der gelieferten Floppy enthalten ist:

- Verlassen der R&S UPL -Software mit der Taste "SYSTEM" oder Ctrl F9 auf der Tastatur
- Diskette einlegen
- Umschalten auf Diskettenlaufwerk (Eingabe A:)
- Aufrufen des Installationsprogramms (Eingabe INSTALL)
- Rückkehr zum R&S UPL -Programm (Eingabe C:\UPL)

Das Programm INSTALL erzeugt auf dem R&S UPL die Directory C:\TUNER (falls diese noch nicht vorhanden ist), und kopiert die BASIC-Programme sowie die für die Applikation notwendigen Setups in dieses Verzeichnis.

Starten der Applikations-Software

Das Applikationsprogramm wird unter der R&S UPL -Selbststeuerung ausgeführt. Nach Starten des R&S UPL-Programms wird mittels der Taste F3 (auf dem externen Keyboard) auf die Selbststeuerung umgeschaltet.

Hierbei ist zu beachten, daß die Logging-Funktion abgeschaltet ist, wie man am Schriftzug "logging off" erkennen kann, der am rechten unteren Rand des Bildschirms eingeblendet wird. Bei eingeschalteter Logging-Funktion würden im Handbetrieb eingegebene Befehle an das Programm angehängt und so unnötig Speicherplatz beanspruchen. Das Ein- bzw. Ausschalten des Logging-Modes geschieht mit der Taste F2 auf der externen Tastatur.

Die Applikationsprogramme müssen aus dem Pfad C:\TUNER aufgerufen werden, da alle Programmodule und Setups in diesem Pfad gesucht werden. Der Pfad kann auf eine der folgenden Arten umgestellt werden:

- auf R&S UPL-Ebene mit dem Befehl "Working Dir" im FILE-Panel
- durch Aufruf eines der für die Tuner-Messungen benötigten Setups
- auf R&S UPL-B10 -Ebene durch die BASIC-Befehlszeile UPD OUT "MMEM:CDIR '\TUNER' "
- über die SHELL der Selbststeuerung mittels Eingabe von CD TUNER und anschließendem EXIT
- auf DOS-Ebene durch Eingabe von CD TUNER

Die Programmdiskette enthält das BASIC-Programm TUNER.BAS für den automatischen Meßablauf. Es wird geladen und gestartet durch Eingabe von:

- LOAD"TUNER"
- RUN

Hierzu können natürlich auch die entsprechenden Softkeys verwendet werden, die beim Aufruf der Selbststeuerung am unteren Bildschirmrand eingeblendet werden.

- Bei Auslieferung sind die Setups so konfiguriert, daß die Meßergebnisse auf einem "Default"-Drucker ausgedruckt werden können, das bedeutet, daß die Einstellungen des zuletzt vom R&S UPL benutzten Druckers verwendet werden. Der Bildschirm des R&S UPL ist auf Farbwiedergabe eingestellt, ein externer Monitor wird angesteuert.

WICHTIG: bei Änderung an den Setups ist der einwandfreie Ablauf der Software nicht sichergestellt!

Setup Konvertierung bei Firmware-Updates

Bei einem Firmware-Update des R&S UPL müssen die Setups möglicherweise konvertiert werden. Beim Laden der Setups geschieht dies automatisch, hierdurch ergeben sich jedoch unnötige Wartezeiten bei jedem Ladevorgang. Wenn dies stört, können die Setups vor dem Start der Applikationssoftware konvertiert und neu abgespeichert werden. Hierfür gibt es zwei Möglichkeiten:

- auf DOS-Ebene durch Aufruf des Konverterprogramms: DO_CONV \TUNER hierbei werden alle Setups in der Directory TUNER konvertiert
- auf R&S UPL -Ebene durch Laden jedes Setups und anschließendem Speichern

WICHTIG: bei auf READ ONLY gesetzten Setups muß zuerst das "r"-Attribut gelöscht werden (auf der DOS-Ebene mit dem Befehl ATTRIB -r).

6 Normgemäße Messungen

Normmeßbedingungen

Alle Messungen sind unter den sogenannten Normmeßbedingungen durchzuführen. Neben der Einhaltung der richtigen Versorgungsspannung, der vorgesehenen Umgebungstemperatur, etc. bedeutet dies auch, daß für die Messungen eine eventuell vorhandene Rauschsperre abgeschaltet werden muß, um die Messungen nicht zu verfälschen.

Die für die Messungen verwendeten Antennensignale müssen bestimmten Bedingungen genügen (Norm-Radiofrequenz-Eingangssignal). Um diese nicht bei jeder Einzelmessung erläutern zu müssen werden sie hier zusammenfassend angegeben:

Normmeßfrequenzen

Frequenzbereich in MHz	Normmeßfrequenz in MHz
65,8 bis 73,0	69
76,0 bis 90,0	83
87,5 bis 104,0	94
87,5 bis 108,0	98

Die Normmeßfrequenz hängt vom Frequenzbereich des Empfängers ab. Unter dem Menüpunkt Setup kann die gewünschte Meßfrequenz im Bereich von 65 ... 108 MHz eingegeben werden.

Normfrequenzhübe

Betriebsart/Signal	RMSD ± 50 kHz	RMSD ± 75 kHz
Mono	± 50 kHz	± 75 kHz
Stereo	± 45 kHz	± 67,5 kHz
Pilotton	± 4,5 kHz	± 6,75 kHz

Der Normfrequenzhub ist gleich dem maximal zulässigen Hub RMSD bei Mono und 90 % von RMSD bei Stereo. Unter dem Menüpunkt Setup kann der maximale Systemhub von ± 50 kHz oder ± 75 kHz gewählt werden, der zulässige Nutzhub (90 % von RMSD) und der Pilottonhub (9 % von RMSD) werden daraus automatisch berechnet.

Preemphase

Zur Verminderung des Rauschens wird normalerweise im Ton-Rundfunk die Eigenschaft üblicher Sprach- und Musiksignale ausgenützt, daß die Signalamplituden zu höheren Frequenzen hin abnehmen. Das ermöglicht, bei der Modulation im Sender die Verwendung einer Preemphase, die die hohen Signalfrequenzen anhebt. Der inverse Frequenzgang im Empfänger senkt diese Signalanteile wieder ab so daß über alles wieder ein ebener Frequenzgang entsteht, gleichzeitig wird aber durch die Absenkung der hohen Frequenzen im Empfänger das Rauschen deutlich reduziert. Meßtechnisch muß aber Sorge dafür getragen werden, daß durch die Preemphase auch bei höheren Meßfrequenzen nicht der Maximalhub des

Systems überschritten wird, daher ist bei Messungen über die Modulationsfrequenz und eingeschalteter Preemphase der Hub so einzustellen, daß der maximal zulässige Hub erst bei der Modulationsfrequenz 15 kHz erreicht wird.

Der Frequenzgang der Preemphase entspricht einem Hochpaß erster Ordnung mit der vorgegebenen Zeitkonstante, länderspezifisch wird hier 50 µs oder 75 µs verwendet, es können auch Messungen ohne Preemphase durchgeführt werden.

Eine Preemphase mit einer Zeitkonstante von 50 µs führt zu einer Signalanhebung um ca. den Faktor 4.8 bei 15 kHz bezogen auf tiefe Frequenzen, daraus ergibt sich der maximal einstellbare Hub für Frequenzgangmessungen zu ca. 20 % des Maximalhubs bei 50 µs Preemphase und ca. 14 % bei 75 µs Preemphase. Auch bei der Messung mit der Normmeßfrequenz 1 kHz ist der Einfluß noch zu berücksichtigen, die Einstellung beträgt hier ca. 95 % des Maximalhubs bei 50 µs Preemphase und ca. 90 % bei 75 µs Preemphase.

Unter dem Menüpunkt Setup kann die verwendete Preemphase 0 µs (keine Preemphase), 50 µs oder 75 µs gewählt werden.

Normmodulationsfrequenz

Hier ist die Normbezugsfrequenz 1 kHz zu verwenden.

Normeingangspegel

Der Normeingangspegel definiert das am Tunereingang verfügbare Antennensignal, hierfür sind 70 dB(fW) festgelegt, auch ausdrückbar als 40 dB(pW). In der Praxis wird weitaus häufiger die Antennenspannung angegeben. Umgerechnet entsprechen 70 dB(fW) gleich 866 µV an einer Eingangsimpedanz von 75 Ω.
Unter dem Menüpunkt Setup kann die zu verwendende Meß-Antennenspannung im Bereich von 0.1 ... 10 mV eingegeben werden, dieser Wert wird bei allen Messungen, die nicht in Abhängigkeit des HF-Signalpegels durchgeführt werden, verwendet.

Filter

Bei einigen Messungen an den Tonfrequenzausgängen ist die Verwendung eines Bandpaßfilters vorgesehen. Der Durchlaßbereich ist 200 Hz bis 15 kHz, zur Unterdrückung von Pilottonresten muß die Dämpfung bei 19 kHz größer 50 dB sein. Dieses Filter ist im R&S UPL unter der Bezeichnung IEC TUNER direkt wählbar, es entspricht der Norm bzw. übertrifft diese.

Zur Bewertung von Störsignalen wird das A-Bewertungsfilter nach IEC 60651-1 verwendet.

Konfiguration (Menüpunkt SETUP)

F5	F6	F7	F8	F9	F10	F11	F12
END	FRQ_RESP	THDN_FRQ	THDN_DEV	CRSS_FRQ	CRSS_LEV	CRSS_DEV	→

Mit der Taste F12 wird die nächste Ebene der Softkeyauswahl eingeblendet:

F5	F6	F7	F8	F9	F10	F11	F12
←	S/N_LEV	IN/OUT	PIL_SUPP		ALL	RECALL	SETUP

Mit der Taste SETUP das Konfigurationsmenü auswählen, es erscheint die folgende Anzeige:

```
Tuner Program Setup

                                        Range            Value

SML GPIB Address                         0...31           28
Rated Max System Deviation              50/75 kHz         75
Preemphase                               0/50/75 us       50
Attenuation antenna matching             0...20 dB        4
Measuring RF Level in mV                 0.1...10 mV      0.87
Measuring Frequency                     65...108 MHz      98
UPL Input selection               Ground=0 Float=1        1
THD+N Display selection               dB=0     %=1        0

!! Enter values with RETURN, do not use arrow keys !!
```

Die gewünschten Werte eingeben und jeweils mit RETURN abschließen, nach Eingabe des letzten Wertes werden alle Werte in ein File geschrieben und automatisch beim Neustart des Programmes wieder verwendet.

Bedienung

F5	F6	F7	F8	F9	F10	F11	F12
END	FRQ_RESP	THDN_FRQ	THDN_DEV	CRSS_FRQ	CRSS_LEV	CRSS_DEV	→

Ein Druck auf die entsprechende Taste startet sofort die Meßroutine. Da mehr Auswahlpunkte als Softkeys zur Verfügung stehen, werden mit der Taste F12 nacheinander weitere Ebenen der Softkeyauswahl eingeblendet:

F5	F6	F7	F8	F9	F10	F11	F12
←	S/N_LEV	IN/OUT	PIL_SUPP		ALL	RECALL	SETUP

Mit der Taste F12 kommt man auf die nächste Ebene, solange dort ein Rechtspfeil eingeblendet wird. Mit der Taste F5 kommt man jeweils eine Ebene zurück solange dort ein Linkspfeil eingeblendet wird. In der untersten Ebene wird dort END eingeblendet, ein Druck auf die Taste F5 bewirkt die Abfrage “Programm beenden?“ und dient zum Beenden des Meßprogrammes.

Mit dem Tastendruck auf die entsprechende Taste wird die zugehörige Meßfunktion gestartet, vorher muß aber der zu messende Tuner auf die Meßfrequenz abgestimmt werden. Nach dem Start des Programmes ist der R&S SML bereits auf die gewünschte Meßfrequenz und den gewünschten Antennenpegel eingestellt.

Nach jeder einzelnen Messung wird das Ergebnis am Bildschirm dargestellt und es erscheint die Softkeyauswahl:

F5	F6	F7	F8	F9	F10	F11	F12
	CONT			EXP-FILE	TRC-FILE	PCX-FILE	PRINTER

Damit kann eine Messkurve als Export-, Trace-File oder als PCX-Bild abgespeichert werden bzw. eine Hardcopy auf einen Drucker ausgegeben werden. Die Files werden in der Directory C:\TUNER\RESULTS gespeichert. Nach Abspeicherung wird die jeweilge Tastenbezeichnung gelöscht, so wird eine versehentliche doppelte Abspeicherung vermieden.

Ein Druck auf die Taste CONT führt wieder zum Auswahlmenü der einzelnen Messungen zurück.

Ein Druck auf die Taste EXP_FILE bewirkt bei einer Kurvendarstellung die Abspeicherung der Meßkurven im ASCII-Export-Format in einer Datei. Diese Datei hat den festen Namen EXPxx.EXP, wobei xx für eine fortlaufende Nummer steht (maximal 5 Stellen). Damit ist eine direkter Import und Weiterverarbeitung der Meßergebnisse mit anderen Programmen wie z.B. Excel möglich.

Ein Druck auf die Taste TRC_FILE bewirkt bei einer Kurvendarstellung die Abspeicherung der Meßkurven im ASCII-Format in einer Datei. Diese Datei hat den festen Namen TRCxx.TRC, wobei xx für eine fortlaufende Nummer steht (maximal 5 Stellen). Diese TRC-Files können im R&S UPL wieder geladen und dargestellt werden.

Ein Druck auf die Taste PCX_FILE bewirkt eine Kopie des Bildschirmes in eine PCX-Datei. Diese Datei hat den festen Namen PICxx.PCX, wobei xx für eine fortlaufende Nummer steht (maximal 5 Stellen). Damit ist eine Einbindung der Ergebnisanzeige z.B. in Textverarbeitungsprogramme möglich. Es wird immer der ganze Bildschirm, jedoch ohne Softkeyzeile, kopiert.

Da bei den EXP-, TRC- und den PCX-Dateien nur eine fortlaufende Numerierung erfolgt, ist es sinnvoll z.B. nach einer Meßreihe die entstandenen Dateien zu kopieren und ggf. umzubenennen. Es können dann die ursprünglichen Dateien wieder gelöscht werden. Dieses Verfahren erlaubt eine bessere Zuordnung der Meßergebnisse und vermeidet Verwechslungen.

Dazu kann nach Beendigung des Meßprogrammes (z. B. mit der Taste F5) durch Eingabe des Kommandos SHELL <RETURN> eine DOS-Shell aufgerufen werden von der aus die Dateien mit DOS-Befehlen kopiert oder umbenannt werden können (standardmäßig in der Directory C:\TUNER\RESULTS). Mit der Eingabe von EXIT <RETURN> kommt man wieder zu Basic zurück, ohne daß das Programm gelöscht wurde. Durch Eingabe von RUN kann das Programm sofort wieder gestartet werden.

Ein Druck auf die Taste PRINTER bewirkt eine Kopie des Bildschirminhaltes auf einen angeschlossenen Drucker. Die gewünschten Druckereinstellungen werden in diesem Fall nicht vom Programm vorgenommen. Der Drucker bleibt so eingestellt, wie er zuletzt manuell am R&S UPL gewählt und benutzt wurde. Es muß daher einmalig vorher in der manuellen Bedienung des R&S UPL im Option-Panel der gewünschte Drucker, sowie die Skalierung und Druckrichtung gewählt werden. Es empfiehlt sich, vorzugsweise Resolution LOW oder MEDIUM und möglichst ganzzahlige Skalierungsfaktoren für die Druckausgabe zu verwenden. Gebrochene Skalierungsfaktoren (insbesondere Werte kleiner 1) führen zur Interpolation der Pixelpunkte und können daher die Druckqualität verschlechtern.

Mit der Softkeyauswahl „RECALL“ werden alle gespeicherten Datensätze angezeigt. Nach Eingabe des gewünschten Satzes werden die Daten geladen und es erscheinen wie nach einer Messung die numerischen Meßergebisse.

Mit der Softkeyauswahl „ALL“ wird ein automatischer Meßablauf aller Meßfunktionen gestartet, dabei werden automatisch alle Grafiken zwischengespeichert und vom Programm ggf. ausgewertet. Nach Beendigung aller Messungen erscheinen dann die numerischen Meßergebisse und die Auswahl:

```
MEASUREMENT OF FM RADIO TUNER WITH AUDIO ANALYZER UPL


Measurement results:
  Audio level Left @ 1kHz @ 90%RMSD:          1.422V
  Audio level Right @ 1kHz @ 90%RMSD:         1.433V

  Maximum Signal/Noise ratio A wtd.:          75.9dB
  RF level for 50 dB S/N A wtd.:              59.9µV
  RF level for 40 dB S/N A wtd.:               3.3µV
  RF level for 30 dB S/N A wtd.:               2.6µV

  Sensitivity for Stereo switching:           23.6µV

  Pilot suppression:                          75.0dB
  Minimum pilot or spurious suppression:      64.6dB
```

F5	F6	F7	F8	F9	F10	F11	F12
BACK			VIEW		SAVE		REPORT

Mit der Taste SAVE wird zunächst eine Eingabe der zusätzlichen Reportangaben aufgerufen und dann werden nach Angabe eines Speichernamens (max 8 Zeichen) die Messergebnisse abgespeichert. Folgende Files werden in der Directory C:\TUNER\RESULTS erzeugt:

- Name.REP   Die Report-Angaben
- Name.RES   Die numerischen Messergebnisse
- Name.FRQ   Das PCX-File der Frequenzgangmessung
- Name.TNF   Das PCX-File der THD+N über Frequenz Messung
- Name.TND   Das PCX-File der THD+N über Deviation Messung
- Name.CRF   Das PCX-File der Crosstalk über Frequenz Messung
- Name.CRL   Das PCX-File der Crosstalk über HF-Pegel Messung
- Name.CRD   Das PCX-File der Crosstalk über Deviation Messung
- Name.SNL   Das PCX-File der S/N über HF-Pegel Messung
- Name.IOL   Das PCX-File der IN/OUT überHF-Pegel Messung
- Name.PIS   Das PCX-File der Pilotton Unterdrückung

Für den Report werden die gewünschten Druckereinstellungen temporär vom Programm vorgenommen. Es wird davon ausgegangen, daß der Drucker kompatibel zu einem Deskjet ist, dieser Druckertyp wird von vielen anderen Druckern wie auch von Laserjet-Druckern emuliert. Nach erfolgtem Ausdruck werden die ursprünglichen Druckereinstellungen wieder hergestellt.

Mit der Taste VIEW können die Grafiken nacheinander aufgerufen werden.

7 Messfunktionen

Tonfrequenzgang

Der Tonfrequenzgang eines UKW-Empfängers wird beeinflußt durch die Qualität des Zwischenfrequenzteils, des Detektors, des Stereo-Decoders und der Demphasis-Schaltung.

Die Messung erfolgt unter Normbedingungen, aber ohne Bandpaßfilter.

Die nach der Norm für UKW-FM-Übertragung vorgesehene Emphase von 50 oder 75 µs wird durch den Signalgenerator R&S SML nachgebildet, das bedeutet, daß niederfrequente Tonsignale mit geringerem Hub moduliert werden, durch die Emphase erhöht sich der Hub dann zur oberen Frequenzgrenze auf den maximal zulässigen Frequenzhub.

Durch die Deemphasis-Schaltung im Tuner wird dieser Effekt kompensiert, so daß sich ein möglichst linearer Frequenzgang des Audio-Signals ergeben sollte. Moderne Geräte weisen Abweichungen von maximal 1 dB an der unteren Frequenzgrenze und maximal 3 dB am oberen Ende des Übertragungsbereiches auf (Bezugswert 1 kHz).

Die Pegelabweichung zwischen den beiden Stereo-Kanälen ist ebenfalls ein Qualitätskriterium, da Pegeldifferenzen die Mitte-Ortung im Stereo-Klangbild verschieben.

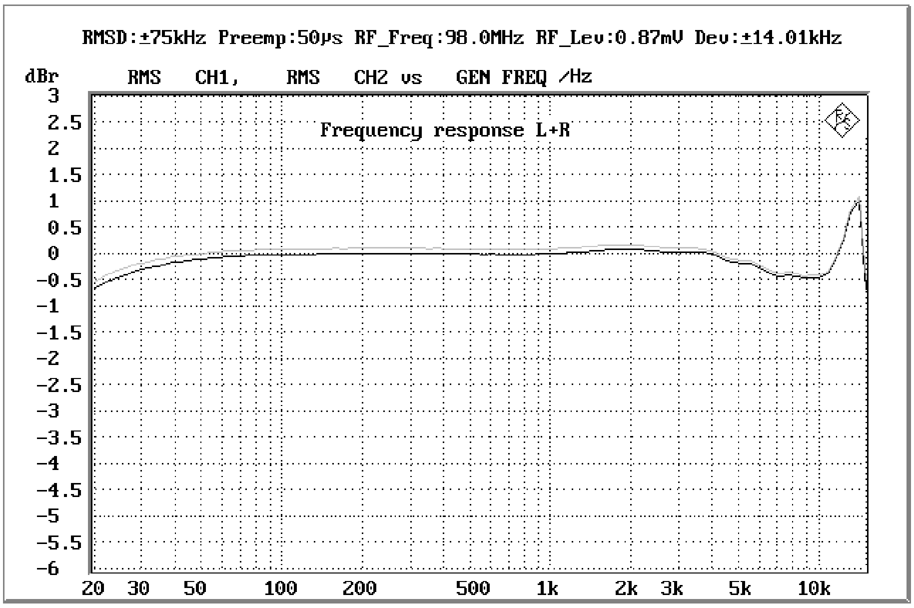

Bild 1: Tonfrequenzgang eines Stereo-Empfängers, bezogen auf den 1 kHz-Wert des linken Kanals

Harmonische Verzerrungen und Rauschen als Funktion der Modulationsfrequenz

Verzerrungen entstehen zum einen in den Hochfrequenz- und Zwischenfrequenzstufen sowie im Detektor des Empfängers, aber auch in den Schaltkreisen der NF-Verstärkung. Die IEC 60315 beschreibt auch Messungen, die die Einflüsse des Verstärkerteils charakterisieren. Der Hauptanteil der Störungen entsteht aber im allgemeinen im Tuner-Teil.

Zur Messung der harmonischen Gesamtverzerrungen wird der Empfänger unter Normmeßbedingungen betrieben. Es werden gleichzeitig beide Stereo-Kanäle moduliert, wobei die Modulationsfrequenz von 20 Hz bis 5 kHz gesweept wird. Gemessen werden die harmonischen Verzerrungen und das Rauschen, bezogen auf das Gesamtausgangssignal. Dieser Wert wird - wahlweise in % oder dB - über der Modulationsfrequenz aufgetragen und grafisch dargestellt (THD+N-Messung). Der in den Normmeßbedingungen vorgesehene Bandpaß kann hier nicht verwendet werden, da auch Frequenzen unter 200 Hz gemessen werden sollen. Um die Messungen allerdings nicht durch Pilottonreste zu verfälschen wird durch ein 15 kHz Tiefpaßfilter die Meßbandbreite eingeengt.

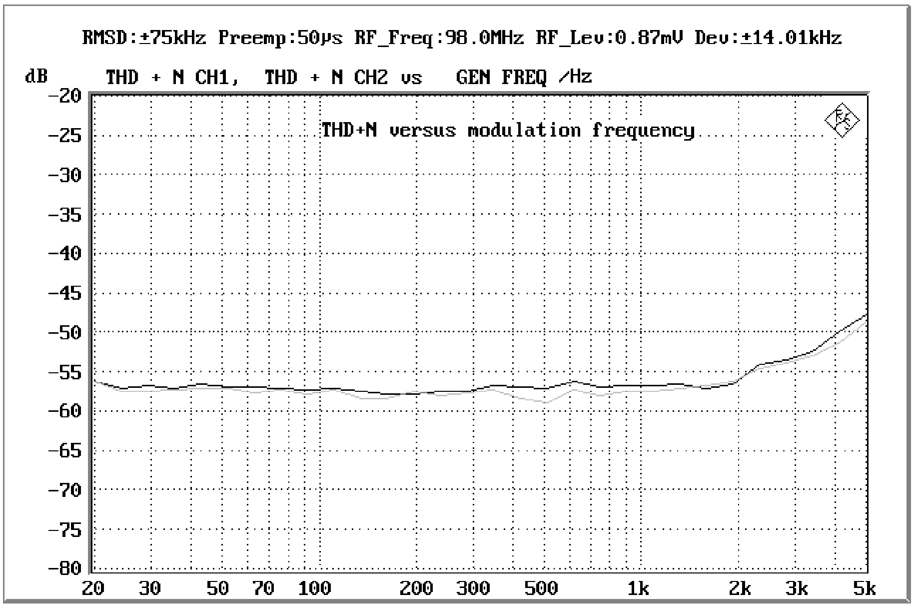

Bild 2: THD+N-Messung dargestellt über der Modulationsfrequenz

Harmonische Verzerrungen und Rauschen als Funktion des Modulationshubs

In Bild 2 wurde der Gesamtklirrfaktor über die Frequenz bei einem durch die Preemphase vorgegeben reduzierten Hub gemessen. Speziell die Detektorstufe des Tuners kann aber eine starke Abhängigkeit über den tatsächlichen Hub ergeben, für diesen Einfluß gibt es die zusätzliche Messung des Klirrfaktors bei 1 kHz über den Signalhub.

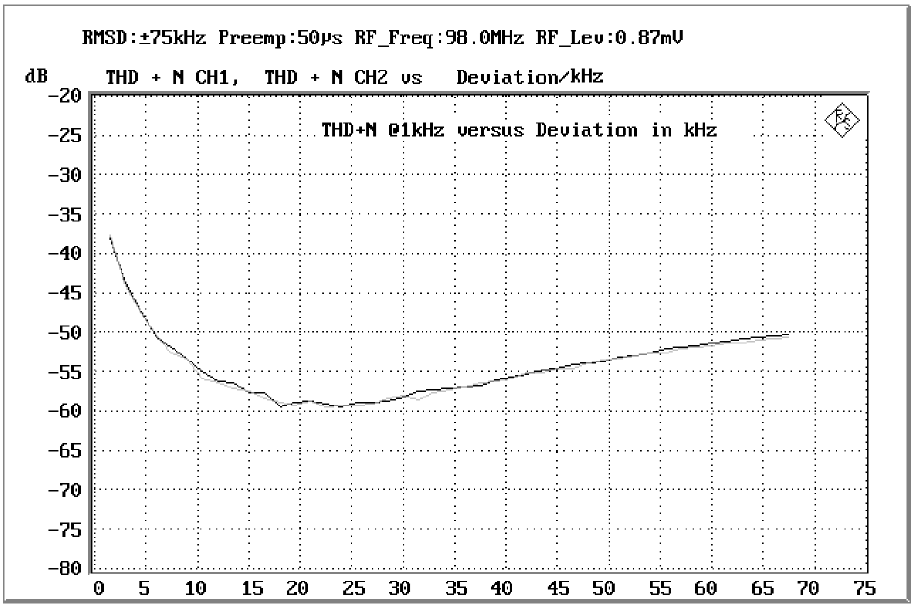

Bild 3: THD+N-Messung dargestellt über den Modulationshub

Bei geringen Hüben überwiegt der Rauschanteil während mit zunehmendem Hub die harmonischen Verzerrungen zunehmen. Bild 3 zeigt ein Ergebnis eines Oberklasse Tuners mit geringem Anstieg bei größerem Hub.

Übersprechdämpfung in Abhängigkeit der Modulationsfrequenz

Übersprechen entsteht, wenn Signalanteile eines Kanals in den anderen Tonkanal eingekoppelt werden. Es vermindert die Kanaltrennung und somit den Stereoeffekt. Die Übersprechdämpfung ist das Pegelverhältnis des gewünschten Signals im einen Kanal zum unerwünschten Signal im anderen Kanal, das von ersterem eingekoppelt wird. Die Angabe erfolgt als Dämpfungswert in dB, das Übersprechen wird in beiden Richtungen gemessen.

Zur Messung wird der Empfänger unter Normbedingungen betrieben. Wie bei der Messung des Tonfrequenzganges wird jedoch die Emphase eingeschaltet und daher bei tiefen Frequenzen ein geringerer Hub eingestellt. Zuerst wird nur der linke Kanal moduliert, wobei die Modulationsfrquenz von 200 Hz bis 15 kHz variiert wird. Die Pegel in beiden Kanälen werden gemessen und ins Verhältnis gesetzt. Um Rauschanteile zu unterdrücken wird selektiv gemessen. Die Messung wird in gleicher Weise mit dem modulierten rechten Kanal wiederholt. Die Ergebnise werden grafisch dargestellt wie in Bild 4 zu sehen.

Übliche Werte für die Übersprechdämpfung bei 1 kHz liegen im Bereich 30...40 dB.

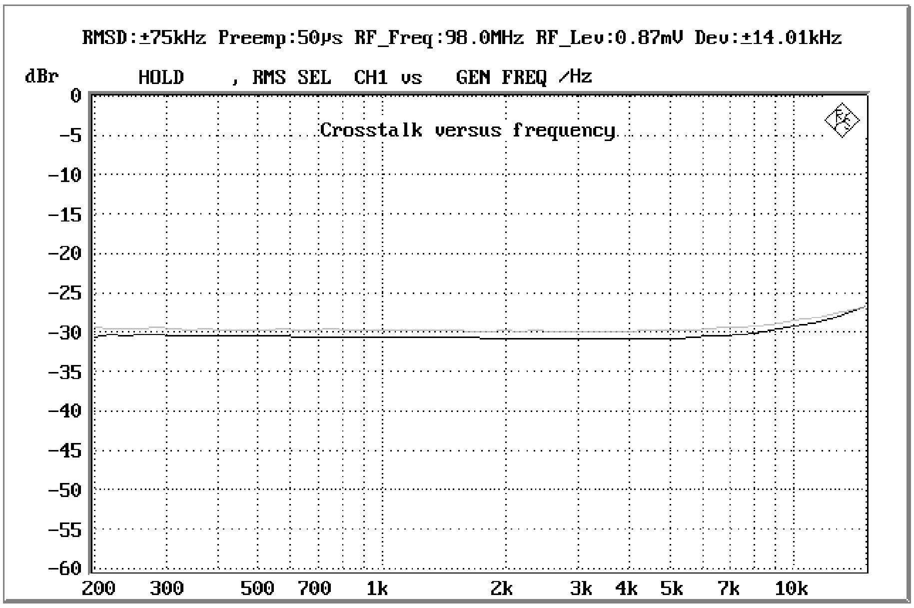

Bild 4: Übersprechdämpfung dargestellt in Abhängigkeit der Modulationsfrequenz.

Übersprechdämpfung in Abhängigkeit des Modulationshubs

Auch die Übersprechdämpfung kann vom Modulationshub abhängen. Um diesen Effekt zu untersuchen gibt es noch die Möglichkeit, die Übersprechdämpfung in Abhängigkeit des Hubes zu messen, die Messung erfolgt analog wie bei der Messung über die Modulationsfrequenz.

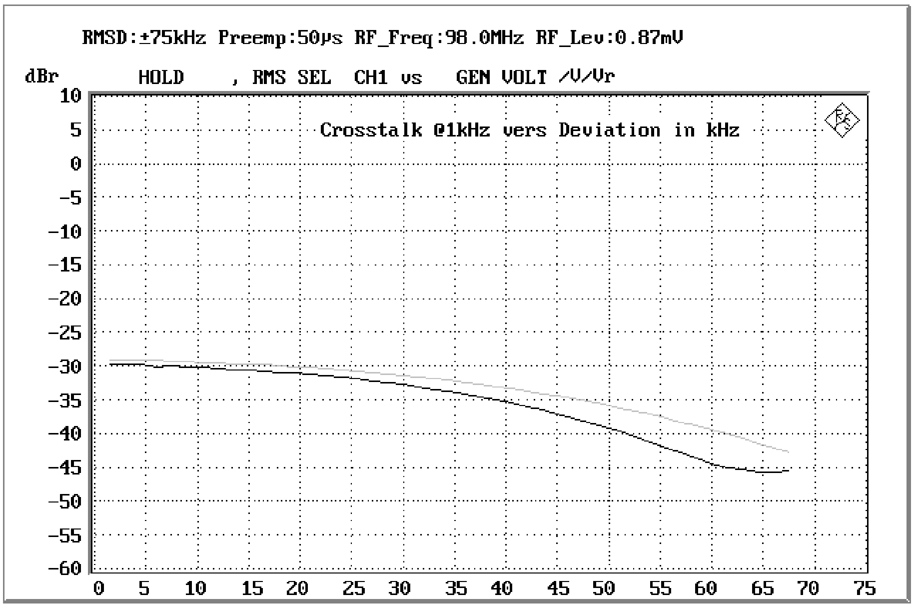

Bild 5: Übersprechdämpfung dargestellt in Abhängigkeit des Modulationshubs

Übersprechdämpfung als Funktion des HF-Pegels / Stereo-Umschaltschwelle

Eine weitere Prüfung ist das Übersprechen in Abhängigkeit vom Antennen-Eingangspegel. Der Empfänger wird unter Normbedingungen betrieben, allerdings mit Maximalhub. Der Eingangspegel wird beginnend bei 100 nV bis 10 mV erhöht. Die Ergebnisse werden grafisch aufgetragen, ein Beispiel zeigt Bild 6.

Mit Hilfe dieser Messung zeigt sich das Verhalten des Tuners beim Empfang schwacher Stereo-Sender. Bei sehr schwachen Antennen-Signalen ist nur Mono-Empfang möglich, d.h. beide Kanäle übertragen das gleiche Signal. In der Meßgrafik ist dieser Bereich an der fehlenden Übersprechdämpfung (Wert 0 dB) zu erkennen. Wird die Antennenspannung erhöht, so beginnt der Stereodecoder ab einem bestimmten Pegel zu arbeiten. Diese Stereo-Schwelle ist in Bild 6 an der plötzlichen Zunahme der Übersprechdämpfung gut zu erkennen. Im Prüfprotokoll wird der Pegel der Stereo-Schwelle angegeben, dazu wird der Punkt ausgewertet, der mindestens 10 dB Übersprechdämpfung in beide Richtungen ergibt.

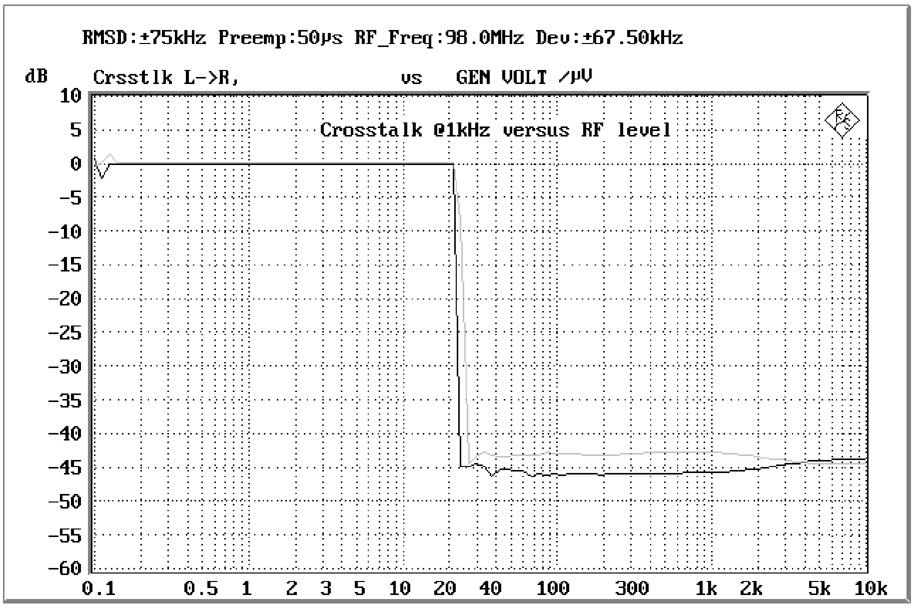

Bild 6: Übersprechdämpfung dargestellt in Abhängigkeit des Antennenpegels

Bedingt durch das Pilottonverfahren erhöht sich aber mit dem Einschalten des Stereodecoders das Rauschen, wie auch in der Eingangssignal / Ausgangssignal-Kennlinie (Bild 8) zu sehen ist. Diese plötzliche Zunahme des Rauschens wird häufig als störend empfunden, vor allem wenn dieser Bereich immer wieder durchlaufen wird wenn das Eingangssignal des Senders schwankt. Im fahrenden Kraftfahrzeug ist dies häufig der Fall. Aus diesem Grund wurde die gleitende Stereo-Umschaltung entwickelt, bei der bei zunehmenden Antennenpegeln das Übersprechen zwischen den Stereo-Kanälen gleitend erhöht wird. Der Einsatz einer derartigen Schaltung ist in der Grafik an der allmählichen Zunahme der Übersprechdämpfung ab der Stereo-Schwelle zu erkennen.

Viele Rundfunkempfänger verfügen über eine Anzeige für den Stereo-Betrieb. Die Anprechschwelle dieser Stereoanzeige kann, muß aber nicht mit dem Pegel identisch sein, an dem der Stereodecoder zu arbeiten beginnt. Vor allem bei gleitender Stereo-Umschaltung wird der Stereo-Empfang häufig erst dann signalisiert, wenn die Empfangsfeldstärke für eine entsprechende Kanaltrennung ausreicht.

Signal/Rauschverhältnis in Abhängigkeit vom Eingangspegel

Das Signal/Rauschverhältnis ist das Verhältnis der vom Signal herrührenden Tonfrequenzspannung zur Rauschspannung. Die IEC 60315-4 erlaubt die Verwendung verschiedener Bewertungsfilter, allerdings hat sich das im HiFi-Sektor verwendete Meßverfahren mit A-Bewertung als Standard durchgesetzt, es bildet die Grundlage für die hier beschriebene Messung.

Prinzipiell kann bei Empfängern das Signal/Rauschverhältnis auf mehrere Möglichkeiten ermittelt werden:

- Beim sequentiellen Verfahren wird bei moduliertem Empfangssignal der Wert der Tonausgangsspannung gemessen, dann die Modulation abgeschaltet und der Wert für das Rauschen ermittelt, dies entspricht der Messung der In/Out-Kennlinie, siehe Bild 9.
- Beim Simultanverfahren wird bei moduliertem Empfangssignal der Pegel des 1 kHz-Tonsignals gemessen, durch Einsatz entsprechender Bandsperren wird sodann die Rauschspannung ermittelt, wobei das HF-Signal weiterhin moduliert ist. Da unter gewissen Umständen die Rauschausgangsspannung eines FM-Empfängers bei Anwesenheit des modulierten Signals erhöht ist, trifft dieses Meßverfahren die Gegebenheiten des praktischen Betriebs besser.

Im Rahmen dieser Applikation wird das Signal/Rauschverhältnis mit Hilfe des Simultanverfahrens unter Verwendung der Effektivwertmessung und mit einem A-Bewertungsfilter ermittelt. Dies entspricht den in der HiFi-Welt üblichen Gegebenheiten.

Der Empfänger wird unter Normbedingungen mit Maximalhub betrieben. Die Modulation erfolgt in Stereo mit einem 1 kHz Signal. Durch den Einsatz der Bandpaßfilter wird der Meßbereich auf 200 Hz bis 15 kHz begrenzt, Einflüsse durch Netzbrumm oder nicht ausreichende Pilottonunterdrückung werden dadurch nicht berücksichtigt. Nach Ermittlung des Wertes für die Tonausgangsspannung wird die 1 kHz Komponente durch ein Notchfilter für die Messung ausgeblendet, und die Rauschspannung gemessen. Um den Wert für das Rauschen nicht durch Klirrkomponenten des 1 kHz Signals zu beeinflussen, wird im Audio Analyzer R&S UPL eine Meßfunktion verwendet, bei der auch eventuelle Harmonische unberücksichtigt bleiben. Aus Signal- und Rauschspannung wird der S/N-Wert berechnet und grafisch in Abhängigkeit vom HF-Eingangspegel dargestellt (Bild 7).

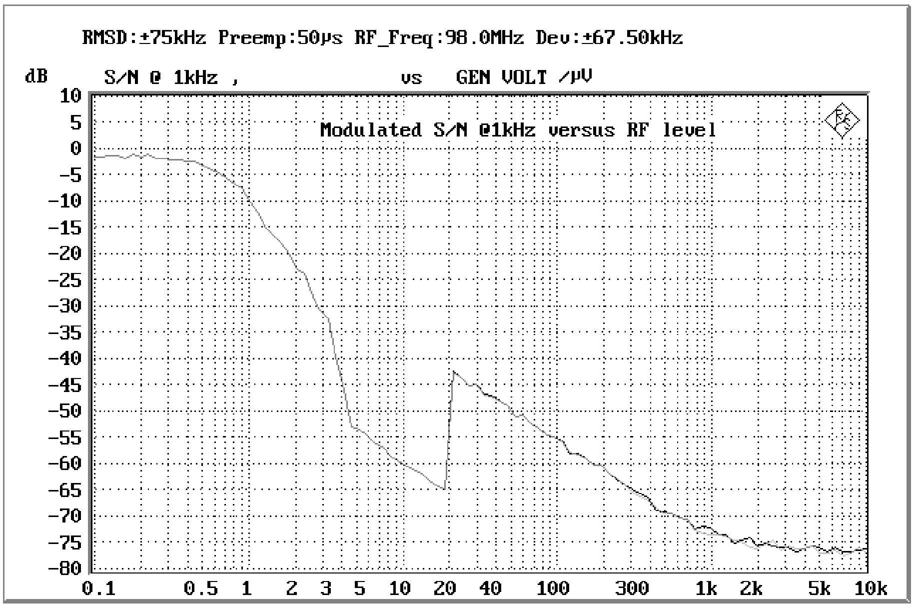

Bild 7: Signal/Rauschverhältnis in Abhängigkeit von der Antennenspannung.

Eingangssignal / Ausgangssignal-Kennlinie

Die Eingangssignal / Ausgangssignal-Kennlinie zeigt die Beziehung zwischen der Antenneneingangsspannung und dem vom Tuner erzeugten Tonsignal auf. Sie ist eine der wichtigsten Messungen, zumal eine Vielzahl von Informationen aus dieser Grafik entnommen werden können, vor allem wenn auch die Rauschausgangsspannung in die Messung einbezogen wird.

Der Empfänger wird unter Normmeßbedingungen betrieben. Für die Messung der Tonausgangsspannung wird mit Maximalub moduliert und für die Rauschmessung mit Hub 0. Der Antennenpegel wird von 100 nV bis 10 mV logarithmisch gesweept, das Tonausgangssignal wird gemessen und grafisch dargestellt, wobei die maximale Ausgangsspannung zu 0 dB gesetzt wird. Der Pegelsweep wird nun wiederholt, wobei die Rauschausgangsspannung aufgezeichnet wird. Es ergibt sich eine Darstellung wie in Bild 8 gezeigt.

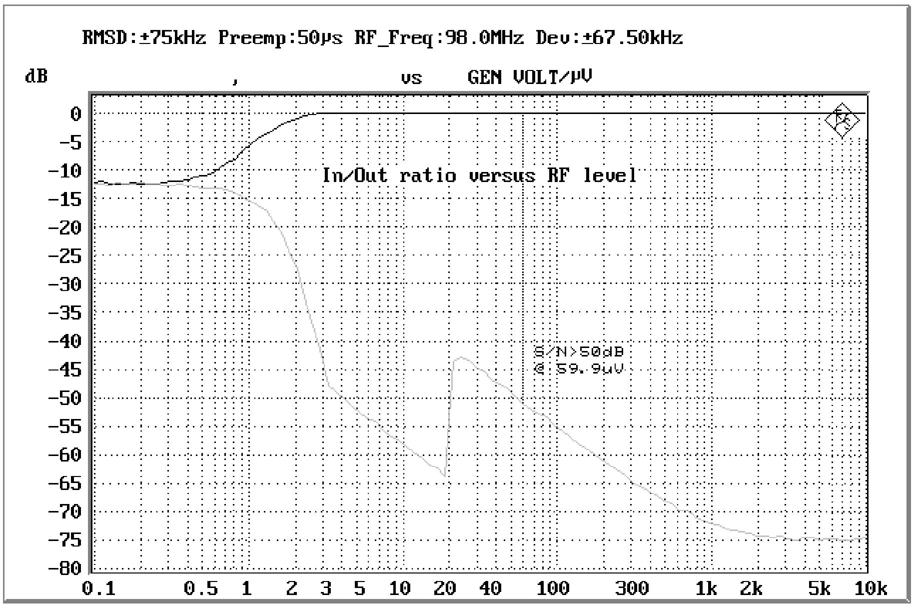

Bild 8: Ausgangssignal/Eingangssignal-Kennlinie eines Tuners mit Einblendung des 50 dB S/N-Wertes

Aus der grafischen Darstellung können nun folgende Informationen entnommem werden:

- Verlauf des Tonausgangssignals

Erst ab einer bestimmten Antenneneingangsspannung ist der Tuner in der Lage, ein Tonsignal aus dem HF-Signal zu detektieren. In der Abbildung ist dieser Punkt daran zu erkennen, daß sich die Signalkurve von der Rauschkurve trennt. Der zugehörige Pegel kann als die absolute Empfindlichkeit des Tuners bezeichnet werden, spielt jedoch zur Charakterisierung des Empfängers nur eine untergeordnete Rolle.

Prinzipiell zeigt die Ausgangssignal-Kurve einen schnellen Anstieg mit zunehmendem Antennensignal, der dann in ein konstantes Signal übergeht. Je nach Empfänger wird dieser Maximalpegel (der gleichzeitig der Bezugspegel für die Messung ist) bei unterschiedlichem HF-Eingangssignal erreicht.

- Rauschsignal

Mit zunehmendem HF-Signal wird das Rauschen immer geringer, bis es einen Minimalwert erreicht. Dieser Minimalwert ist im Monobetrieb kleiner als bei stereophonem Empfang.

Im Stereobetrieb zeigt der Pegelsweep des Eingangssignals zunächst dieselbe Charakteristik wie bei einem monophonen Signale. Ab einem bestimmten Eingangssignal beginnt der Stereodecoder zu arbeiten (Stereoschwelle), was sich zunächst in einer deutlichen Zunahme des Rauschens bemerkbar macht. Mit zunehmendem Eingangssignal wird das Rauschen wiederum reduziert, erreicht jedoch im allgemeinen nicht den Minimalwert des monophonen Empfangs.

- Maximales Signal/Rauschverhältnis

Aus dem Maximalwert des Tonausgangssignals und dem Minimalwert des Rauschens kann der maximale Signal/Rausch-Abstand ermittelt werden. Dieser Wert wird auch im Prüfprotokoll mit ausgedruckt.

- Rauschbegrenzte Empfindlichkeit

Mit rauschbegrenzter Empfindlichkeit wird der Antennenpegel bezeichnet, der ein Tonsignal mit definiertem Signal/Rauschverhältnis erzeugt. Es ist damit ein Empfindlichkeitswert, der gleichzeitig eine Aussage über die Wiedergabequalität des Tonsignals beinhaltet.

Dieser Empfindlichkeitswert soll für HiFi-Stereoempfänger bei 50 dB Signal/Rauschabstand geprüft werden.

Als Beispiele für heutige qualitativ hochwertige Tuner können Empfindlichkeiten von etwa 3 µV für Mono-Empfang und 30 bis 40 µV für Stereo-Empfang angenommen werden. Gleichzeitig wird noch die Empfindlichkeit für 40 dB und 30 dB Signal/Rauschabstand festgestellt, oftmals werden diese Werte erst unterhalb der Stereoumschaltschwelle erreicht. Diese Werte werden zur Information ebenfalls im Protokoll mit ausgedruckt.

Unterdrückung von Pilotton und Hilfsträger

Zur Kennung einer Stereo-Rundfunksendung wird der sogenannte Pilotton, ein Signal bei 19 kHz, mitübertragen. Um an den Tuner angeschlossene Geräte, wie z.B. Verstärker oder Kasettenrekorder nicht zu stören, müssen der Pilotton sowie seine Hilfsträger im Tuner ausreichend unterdrückt werden. Dies geschieht durch eine entsprechende Beschaltung des Stereodecoders oder durch den Einsatz von Filtern am Tuner-Ausgang.

Der Grad der Unterdrückung von Pilotton, Hilfsträger, sowie sonstiger Störprodukte ist ebenfalls ein Qualitätskriterium für einen Tuner. Die Tonfrequenzspannung wird unter Normmeßbedingungen bei Maximalhub gemessen und als Bezug für 0 dB der Darstellung verwendet. Danach wird die Nutzmodulation auf 0 eingestellt und die verbleibenden Frequenzanteile werden dargestellt. Die Unterdrückung des Pilotsignales wird gemessen und im Protokoll ausgedruckt, gleichzeitig wird nach der höchsten Signalkomponente gesucht und ebenfalls als Störunterdrückungswert im Protokoll ausgedruckt.

Die Darstellung zeigt das Spektrum der Signale (Bild 9). Man kann hier die einzelnen Frequenzanteile gut erkennen.

Tuner höherer Qualität sollen alle Frequenzanteile oberhalb des Übertragungsbreiches um mindestens 50 dB unterdrücken.

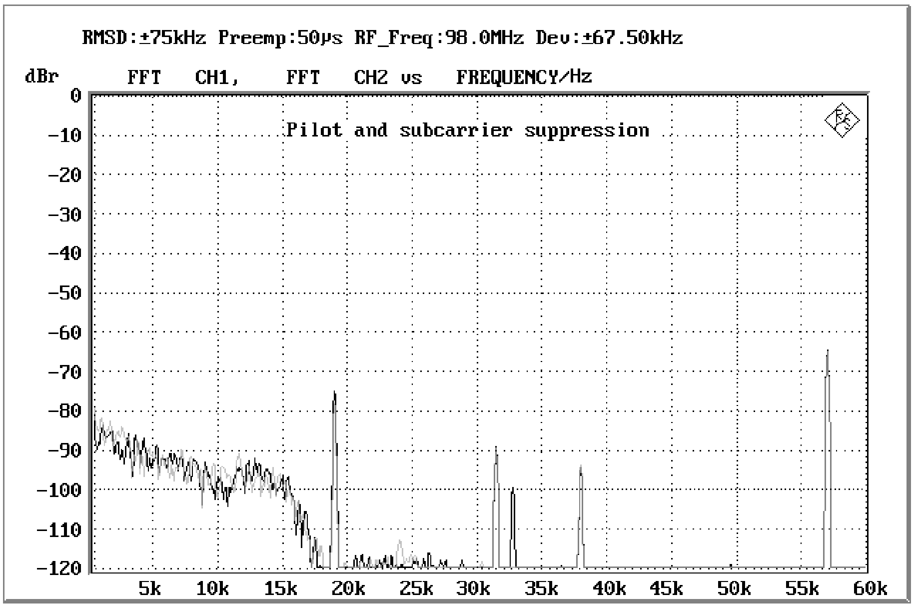

Bild 9: Ausgangs-Spektrum eines Tuner ohne Modulation mit Rest-Pilotton, Hilfsträger und Störkomponenten

8 Demoprogramme für Produktionstests

Mit der Installation werden auch automatisch die Programme MULTFREQ.BAS und FASTDIST.BAS erzeugt. Diese Programme dienen als Beispiele für schnelle Frequenzgang- oder Klirrfaktormessungen für die Produktion. Nach Aufruf und Start von Multfreq.bas erscheint zunächst die Abfrage ob ca. 20, 50 oder 100 Frequenzgangpunkte gemessen werden sollen. Danach erfolgt nach jedem Tastendruck von SPACE eine Messung und Darstellung des gemessenen Frequenzganges einschliesslich einer PASS/FAIL-Prüfung mit vorgegebenen Frequenzgangtoleranzen. Mit der ESC-Taste kann das Programm abgebrochen werden.

Das Programm FASTDIST wird ähnlich bedient, nach dem Start erfolgt die Abfrage nach dem Grenzwert des Klirrfaktors in Prozent. Die Messung wird ebenfalls mit der SPACE-Taste gestartet, der Klirrfaktorwert (THD+N) wird gemessen und auf Einhaltung des Grenzwertes geprüft. Mit der ESC-Taste kann das Programm abgebrochen werden.

9 Literatur

Meßverfahren für Funkempfänger für verschiedene Sendearten
Teil 4: Empfänger für frequenzmodulierte Ton-Rundfunksendungen
DIN EN 60315-4 (deutsche Übersetzung von IEC 60315-4) 12/1998

10 Bestellinformation

Audio Analyzer	**R&S UPL**	**1078.2008.06**
Option Universelle Ablaufsteuerung	**R&S UPL-B10**	**1078.3904.02**
Signalgenerator	**R&S SML01**	**1090.3000.11**
Signalgenerator	**R&S SML02**	**1090.3000.12**
Signalgenerator	**R&S SML03**	**1090.3000.13**
Signalgenerator	**R&S SMV03**	**1147.7509.13**
Option Stereo/RDS-Coder	**R&S SML-B5**	**1147.8805.02**
Matching Pad 50/75 Ohm	**R&S RAM**	**0358.5414.02**

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Products: Audio Analyzer R&S UPL Signal Generator R&S SML

Measurements on RF radio tuners with Audio Analyzer R&S UPL and Signal Generator R&S SML with option -B5

Application Note 1GA43_0E

A great number of measurements have to be performed to determine the quality of FM tuners. This Application Note presents a program that permits these measurements to be carried out in line with DIN EN60315-4.

ROHDE&SCHWARZ

Subject to change – 1GPP Betz 01.2003 – 1GA43_0E

Contents

1 Overview

A great number of measurements have to be performed to determine the audio quality of FM tuners. With the aid of the program described in this Application Note, these measurements can be performed in line with DIN EN60315-4.

2 Operating Principle

The Audio Analyzer R&S UPL generates all required test signals. The signals are applied to the stereo coder (option R&S SML-B5) in the signal generator and modulated onto the transmitter signal. The modulated signal is forwarded to the antenna input of the DUT. The demodulated audio signals at the DUT output are transferred to the audio analyzer for measurements.

When used together with the optional Universal Sequence Controller R&S UPL-B10, the Audio Analyzer R&S UPL automatically executes complete measurement sequences and controls the signal generator via the IEC/IEEE-bus or RS-232-C interface. Thus, even measurements versus the RF signal level, for instance, can be automatically performed .

3 Hardware and Software Requirements

Required Measuring Instruments and Accessories

An Audio Analyzer R&S UPL with option R&S UPL-B10 is required for generating and measuring audio signals. RF signals are generated by the Signal Generator R&S SML (alternatively R&S SMV) which must be equipped with the Stereo/RDS Coder Option R&S SML-B5.

An external keyboard and, if necessary, a printer are needed in addition.

The BASIC programs required for the automatic measurements are stored on a diskette available from your local Rohde & Schwarz representative or can be downloaded in the form of a file from the Rohde & Schwarz website, unpacked and then stored on a diskette. The R&S UPL should meet the following software requirements:

- UPL firmware version 3.01 or higher
- Universal Sequence Controller Option UPL-B10 installed
- R&S UPL configured with 64 Kbyte program memory and 32 Kbyte data memory (using configuration tool UPLSET setting 3)

4 Test Assembly

Required are the Audio Analyzer R&S UPL, Signal Generator R&S SML and probably an HP-deskjet-compatible printer for result documentation.

Link the audio analyzer and the signal generator via the IEC/IEEE bus. Routines are available for specific measurement modules, which control the signal generator via the RS-232-C interface. Measurements can thus be started in the R&S UPL by means of a macro via the IEC/IEEE bus, and necessary settings in the R&S SML can be made from the R&S UPL via the RS-232-C interface. Since control via IEC/IEEE-bus and RS-232-C interface can be used alternately in the R&S SML, the R&S SML and the R&S UPL can also be used for other tasks in the system after a measurement macro has been processed via the IEC/IEEE bus. For details refer to Appendix A.

The printer is connected to the Centronics interface of the R&S UPL.

Connect the RF output of the R&S SML to the unbalanced antenna input of the tuner. Since the R&S SML has an output impedance of 50 Ω, the generator has to be matched to the tuner. Usually, the coaxial 75 Ω antenna input is used. We recommend to use the Matching Pad R&S RAM which can be ordered under 358.5414.02.

If the receiver to be measured is only equipped with a balanced input, an adequate balun with an impedance of 240 Ω or 300 Ω must be connected in between. Since the measurements are referenced to the RF level at the antenna input of the tuner and not to the generator output level, the insertion loss of these matching elements must always be taken into account. The actual loss must be entered under SETUP and will then be considered by the program. For loss values, refer to the matching pad documentation or to the label on the pads.

Connect generator output 1 of the Audio Analyzer R&S UPL to the external modulation input (left) and generator output 2 to the modulation input (right) at the rear of Signal Generator R&S SML. Connect the tuner output for the left channel to analyzer input 1 and the right channel to input 2 of the R&S UPL.

Make sure that measurements are performed with adequate grounding, e.g. to eliminate hum pick-up. Since tuners are normally not grounded and their outputs are floating, the inputs of the R&S UPL should be grounded provided a ground connection is not established via the antenna connected to the R&S SML. A selection can be made under SETUP / R&S UPL Input Selection.

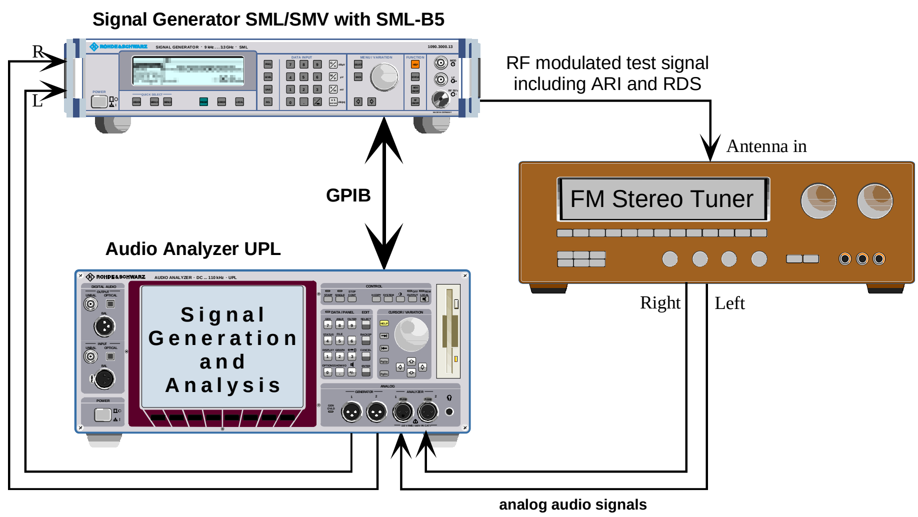


Fig. 1: Test assembly for tuner measurement

5 Installing the Software

The application software is installed with the aid of the INSTALL.BAT installation program which is also stored on the supplied floppy disk.

- Quit the R&S UPL software by pressing the SYSTEM key on the instrument or Ctrl F9 on the keyboard.
- Insert floppy disk.
- Select floppy disk drive (enter A:).
- Call the installation program (enter INSTALL).
- Return to UPL program (enter C:\UPL).

The INSTALL program creates the C:\TUNER directory in the R&S UPL (if it is not already available) and copies the BASIC programs and all setups required for the application into this directory.

Starting the Application Software

The application program is executed by way of the automatic sequence control of R&S UPL. After starting the UPL program, press the F3 key (on the external keyboard) to select automatic sequence control.

The logging function should be switched 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 so use up memory. The logging mode can be switched on and off with the F2 key of the external keyboard.

The application programs have to be called from path C:\TUNER where all program modules and setups are searched for. The path can be changed in the following ways:

- In the R&S UPL with the "Working Dir" command in the FILE panel
- By calling one of the setups required for tuner measurements
- In the R&S UPL-B10 with the BASIC command line UPD OUT "MMEM:CDIR '\TUNER' "
- Via the SHELL of the automatic sequence control by entering CD TUNER and then EXIT
- At DOS level by entering CD TUNER

The program floppy contains the BASIC program TUNER.BAS for automatic sequence control. The program is loaded and started by entering

- LOAD"TUNER"
- RUN

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

- Upon delivery, the setups are configured so that measurement results are output to a "default" printer. This means that the printer settings used last by R&S UPL will be used again. The screen of the R&S UPL is set to colour display and an external monitor is driven (may be used).

IMPORTANT: Correct execution of the software cannot be guaranteed if changes are made in the setups.

Setup Conversion for Firmware Updates

For updating the UPL firmware, the setups may have to be converted. This is done automatically when the setup is loaded, but the conversion may delay the loading. To avoid this, the setups can be converted and stored before the application software is started. This can be done in two ways:

- At the DOS level by calling the conversion program
  DO_CONV \TUNER, which converts all setups in the TUNER directory
- In the R&S UPL by loading and storing each setup

IMPORTANT: In the case of READ ONLY setups, the "r" attribute has to be deleted first (at the DOS level with command ATTRIB -r).

6 Measurements to Standard

Standard Test Conditions

All measurements are to be performed under test conditions specified by the standard. In addition to correct supply voltage, specified ambient temperature, etc, this also means that any squelch that may be used has to be switched off in order not to impair the measurements.

The antenna signals used for the measurements must meet certain requirements (standard radio frequency input signal). To avoid description for each single measurement, all the conditions are listed and described below:

Standard Test Frequency

Frequency range in MHz	Standard frequency in MHz
65.8 to 73.0	69
76.0 to 90.0	83
87.5 to 104.0	94
87.5 to 108.0	98

The standard test frequency is a function of the frequency range of the receiver. The desired test frequency in the range from 65 MHz to 108 MHz can be entered under SETUP.

Standard Frequency Deviation

Operating mode/signal	RMSD ± 50 kHz	RMSD ± 75 kHz
Mono	± 50 kHz	± 75 kHz
Stereo	± 45 kHz	± 67.5 kHz
Pilot tone	± 4.5 kHz	± 6.75 kHz

The standard frequency deviation corresponds to the permissible rms deviation (RMSD) for mono and to 90% of RMSD for stereo. The maximum system deviation of ±50 kHz or ±75 kHz can be selected under SETUP. The permissible useful deviation (90% of RMSD) and the pilot tone deviation (9% of RMSD) are automatically calculated from this value.

Preemphasis

The fact that the amplitude of voice and music signals goes down when the frequency increases is normally utilized in sound broadcasting to reduce the noise. Preemphasis can be used in this case to increase the high signal frequencies during modulation in the transmitter. The inverse frequency response in the receiver again reduces these signal components so that a flat overall frequency response is obtained, but the reduction of the high frequencies in the receiver significantly reduces the noise. When measurements are performed, care must however be taken that the maximum deviation of the system is not exceeded even at high frequencies while preemphasis is on. Therefore, if measurements are performed at the modulation frequency with preemphasis on, the deviation should be adjusted so that the maximum permissible deviation is only attained at a modulation frequency of 15 kHz.

The frequency response of preemphasis has the effect of a 1st order highpass filter with predefined time constant. A country-specific value of 50 µs or 75 µs is used. Measurements without preemphasis can also be performed.

Preemphasis with a time constant of 50 µs increases the signal by a factor of about 4.8 at 15 kHz relative to low frequencies. This yields the maximum deviation that can be adjusted for frequency response measurements (approx. 20% of the maximum deviation with 50 µs preemphasis and approx. 14% with 75 µs preemphasis). This influence must also be taken into account for measurements with the 1 kHz standard test frequency. In this case the setting is approx. 95% at 50 µs preemphasis and approx. 90% at 75 µs preemphasis.

A preemphasis of 0 µs (no preemphasis), 50 µs or 75 µs can be selected under SETUP.

Standard Modulation Frequency

The 1 kHz standard reference frequency should be used in this case.

Standard Input Level

The standard input level determines the antenna signal at the tuner input. It is specified with 70 dB(fW) which corresponds to 40 dB(pW). In practice, the antenna voltage is specified in most cases. A value of 70 dB(fW) corresponds to 866 µV at an input impedance of 75 Ω.
The antenna test voltage to be used can be entered under SETUP in the range 0.1 mV to 10 mV. This value is used for all measurements that are performed without regard to the RF signal level.

Filters

A bandpass filter is required for some measurements at the audio frequency outputs. The filter's passband range is 200 Hz to 15 kHz. To suppress residual pilot tones, the attenuation at 19 kHz must be higher than 50 dB. This filter can be directly selected in the R&S UPL under IEC TUNER. It either meets or exceeds the standard.

The A weighting filter to IEC 60651-1 is used for weighting noise signals.

Configuration (SETUP menu)

F5	F6	F7	F8	F9	F10	F11	F12
END	FRQ_RESP	THDN_FRQ	THDN_DEV	CRSS_FRQ	CRSS_LEV	CRSS_DEV	→

Press key F12 to display the next level of the softkey labels.

F5	F6	F7	F8	F9	F10	F11	F12
←	S/N_LEV	IN/OUT	PIL_SUPP		ALL	RECALL	SETUP

Press the SETUP key to select the configuration menu. The following is displayed:

Tuner Program Setup

	Range	Value
SML GPIB Address	0...31	28
Rated Max System Deviation	50/75 kHz	75
Preemphase	0/50/75 us	50
Attenuation antenna matching	0...20 dB	4
Measuring RF Level in mV	0.1...10 mV	0.87
Measuring Frequency	65...108 MHz	98
UPL Input selection	Ground=0 Float=1	1
THD+N Display selection	dB=0 %=1	0

!! Enter values with RETURN, do not use arrow keys !!

Enter the desired values and confirm with RETURN. After the last value has been entered, all values are stored in a file and automatically used each time the program is restarted.

Operation

F5	F6	F7	F8	F9	F10	F11	F12
END	FRQ_RESP	THDN_FRQ	THDN_DEV	CRSS_FRQ	CRSS_LEV	CRSS_DEV	→

Clicking the respective key starts the test routine. Since there are more selection items than softkeys, the next softkey levels are called with F12.

F5	F6	F7	F8	F9	F10	F11	F12
←	S/N_LEV	IN/OUT	PIL_SUPP		ALL	RECALL	SETUP

The next higher level can be selected with the F12 key as long as the → arrow is displayed below the key. With F5, the user can return to the next lower level as long as the ← arrow is displayed below the key. At the lowest level, END is displayed instead of the arrow. After pressing F5, the query "Do you really want to quit?" is displayed allowing the test program to be terminated.

A test routine is started by clicking the respective key, but the tuner to be measured has to be aligned to the test frequency before the routine is started. When the program is started, the R&S SML is automatically set to the desired test frequency and the desired antenna level.

After each measurement, the result is displayed and the softkeys are labelled as follows:

F5	F6	F7	F8	F9	F10	F11	F12
	CONT			EXP-FILE	TRC-FILE	PCX-FILE	PRINTER

A measured trace can now be stored in the form of an export file, trace file or PCX picture, or a hardcopy can be printed. The files are stored in the C:\TUNER\RESULTS directory. After storage, the respective key label is blanked to prevent the trace being stored twice.

Clicking the CONT key restores the selection menu for the various measurements.

When the EXP_FILE key is pressed, the displayed traces can be saved in a file in ASCII export format. This file has the name EXPxx.EXP, xx representing a consecutive number (of max. 5 digits). This allows direct import and processing of measurement results by means of other programs such as Excel.

When the TRC_FILE key is pressed, the displayed trace is saved in a file in ASCII format. This file has the name TRCxx.TRC, xx representing a consecutive number (of max. 5 digits). The TRC files can be reloaded in the R&S UPL and displayed.

The screen content can be copied into a PCX file with the aid of the PCX_FILE key. This file has the name PICxx.PCX, xx representing a consecutive number (of max. 5 digits). Thus the measurement results can also be used in word processing programs, for instance. The entire screen except for the softkey line is always copied.

Since both the EXP, TRC and 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. The original files can then be deleted. Thus, results can be identified more easily and a mixup 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 then be copied or renamed with the aid of DOS commands (standard procedure in the C:\TUNER\RESULTS directory). Entering EXIT <RETURN> restores BASIC without the program being cleared. The program can be immediately restarted with RUN.

The screen content can be output to a printer by pressing the PRINTER key. In this case the desired printer settings are not selected by the program. The printer remains set as selected last in the manual mode of the R&S UPL. The desired printer, scaling and format should therefore be manually set once in the OPTION panel of the R&S UPL prior to the measurement. It is recommended to select 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 pixel values are interpolated and the print quality might be reduced.

When RECALL is selected, all saved data records are displayed. After a data record has been selected, the data is loaded and results are displayed in numeric form like after a measurement.

An automatic sequence of all measurement functions is started with ALL. All measured traces are temporarily stored and evaluated by the program, if required. After all measurements have been terminated, the results are displayed numerically and softkeys are labelled as shown below:

```
MEASUREMENT OF FM RADIO TUNER WITH AUDIO ANALYZER UPL


Measurement results:
  Audio level Left @ 1kHz @ 90%RMSD:          1.422V
  Audio level Right @ 1kHz @ 90%RMSD:         1.433V

  Maximum Signal/Noise ratio A wtd.:          75.9dB
  RF level for 50 dB S/N A wtd.:              59.9µV
  RF level for 40 dB S/N A wtd.:               3.3µV
  RF level for 30 dB S/N A wtd.:               2.6µV

  Sensitivity for Stereo switching:           23.6µV

  Pilot suppression:                          75.0dB
  Minimum pilot or spurious suppression:      64.6dB
```

F5	F6	F7	F8	F9	F10	F11	F12
BACK			VIEW		SAVE		REPORT

When the SAVE key is pressed, the user is first asked to enter additional information for the report, and then the results are stored after a file name has been entered (max. 8 characters). The following files are created in the C:\TUNER\RESULTS directory.

- Name.REP Report information
- Name.RES Numeric measurement results
- Name.FRQ PCX file of frequency response measurement
- Name.TNF PCX file for THD+N versus frequency measurement
- Name.TND PCX file for THD+N versus deviation measurement
- Name.CRF PCX file for crosstalk versus frequency measurement
- Name.CRL PCX file for crosstalk versus RF level measurement
- Name.CRD PCX file for crosstalk versus deviation measurement
- Name.SNL PCX file for S/N versus RF level measurement
- Name.IOL PCX file for IN/OUT versus level measurement
- Name.PIS PCX file for pilot tone suppression

The program temporarily selects the printer settings desired for report printing. It is assumed that the printer is deskjet-compatible. This printer type is emulated by laserjet and many other printers. After printout, the original settings are restored.

Using the VIEW key the grahpics can be reviewed in a sequence.

7 Measurement Functions

Audio Frequency Response

The audio frequency response of a UHF receiver is influenced by the quality of the IF section, detector, stereo coder and deemphasis circuit.

The measurement is performed under the conditions specified by the standard but without a bandpass filter.

The emphasis of 50 µs or 75 µs specified by the standard for VHF FM transmissions is simulated in the Signal Generator R&S SML. This means that low-frequency audio signals are modulated with a low deviation. The frequency deviation is then increased by emphasis to the maximum permissible deviation at the upper frequency limit.

This effect is compensated for by the deemphasis circuit in the tuner so that the frequency response of the audio signal becomes as linear as possible. Modern instruments have a deviation of max. 1 dB at the lower frequency limit and of max. 3 dB at the upper end of the transmission range (referenced to 1 kHz).

The level deviation between the two stereo channels is also a quality criterion because level differences shift the center for stereo sound impression.

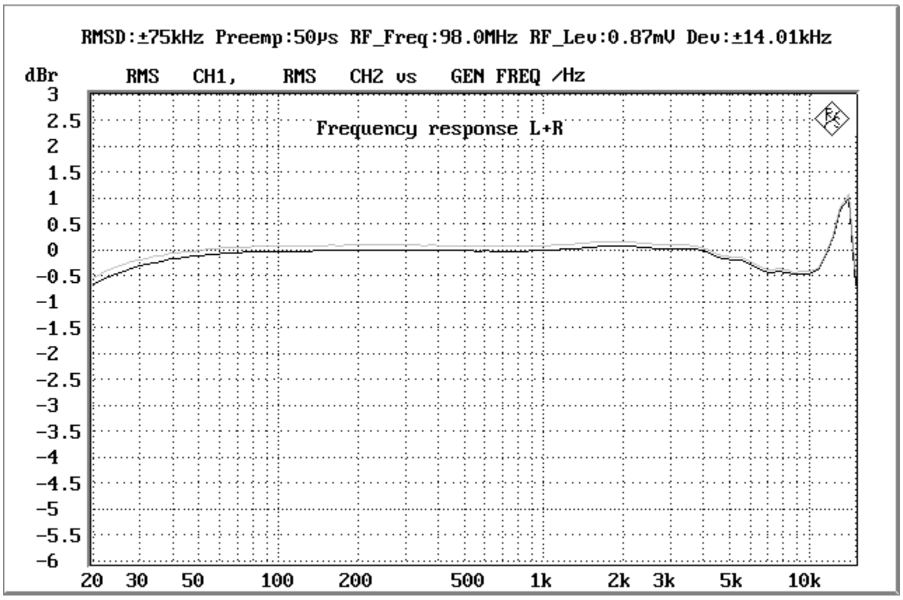


Fig. 1: Audio frequency response of a stereo receiver referenced to 1 kHz in the left channel

THD and Noise as a Function of Modulation Frequency

Distortions are caused by RF and IF sections and by the detector in the receiver but also by AF amplification circuits. IEC 60315 specifies also measurements that characterize the effects caused by the amplifier section. However, most of the distortions are normally produced by the tuner.

For THD measurements, the receiver is operated under standard conditions. The two stereo channels are modulated simultaneously; the modulation frequency is swept between 20 Hz and 5 kHz. THD and noise are measured with reference to the total output signal. The result - in % or dB - is graphically displayed versus the modulation frequency (THD+N measurement). The bandpass filter specified by the standard cannot be used here because frequencies below 200 Hz are also to be measured. In order not to corrupt measurement results by residual pilot tones, the measurement bandwidth is limited by a 15 kHz lowpass filter.

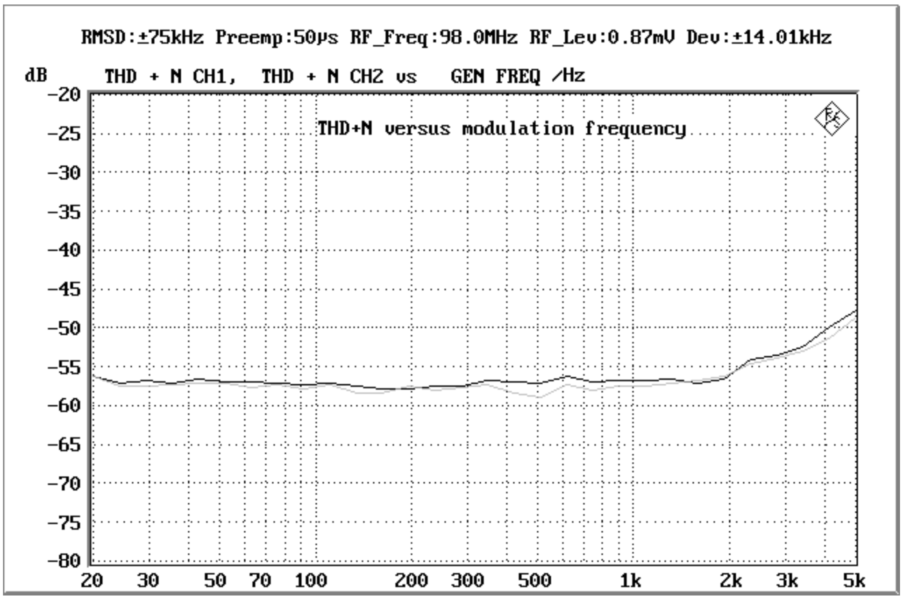


Fig. 2: THD+N measurement versus modulation frequency

THD and Noise as a Function of Modulation Deviation

In Fig. 2, THD was measured versus frequency at a deviation reduced by preemphasis. Particularly in the detector stage of the tuner, the THD may considerably vary depending on the actual frequency deviation. This effect can be determined by a THD measurement at 1 kHz versus the signal deviation.

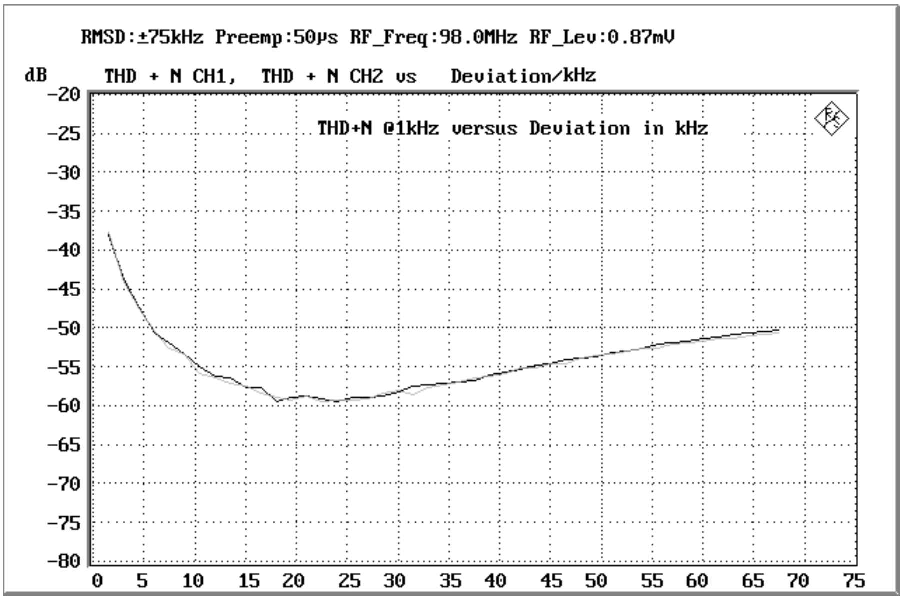


Fig. 3: THD+N measurement versus modulation deviation

When the deviation is small, the noise component dominates while the THD increases with increasing deviation. Fig. 3 shows the result of a high-class tuner where THD increases only slightly at larger deviations.

Crosstalk as a Function of Modulation Frequency

Crosstalk occurs when signal components of a channel are coupled into another audio channel. This reduces channel separation and thus impairs the stereo effect. Crosstalk attenuation is the level ratio of the wanted signal in a channel to the unwanted signal coupled into the other channel. It is specified as attenuation in dB. Crosstalk is measured in both directions.

The measurement is performed under the conditions specified by the standard. As with measurements of the audio frequency response, emphasis is switched on so that a smaller deviation is set at low frequencies. Only the left channel is modulated at first with a modulation frequency that is varied between 200 Hz and 15 kHz. The level is measured in both channels and the ratio is formed. To suppress the noise components, a selective measurement is carried out. The measurement is repeated in the modulated right channel. Results are graphically displayed as shown in Fig. 4.

Common crosstalk values at 1 kHz are within 30 to 40 dB.

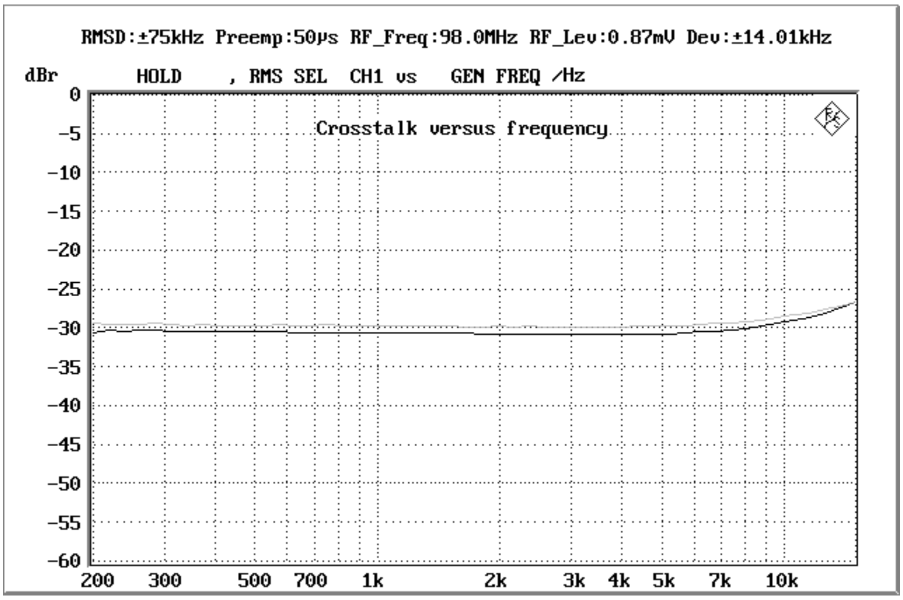


Fig. 4: Crosstalk attenuation as a function of modulation frequency

Crosstalk Attenuation as a Function of Modulation Deviation

The crosstalk attenuation may also be a function of modulation deviation. To investigate this effect, the crosstalk attenuation can be measured as a function of the deviation. The measurement is performed analogously to that versus the modulation frequency.

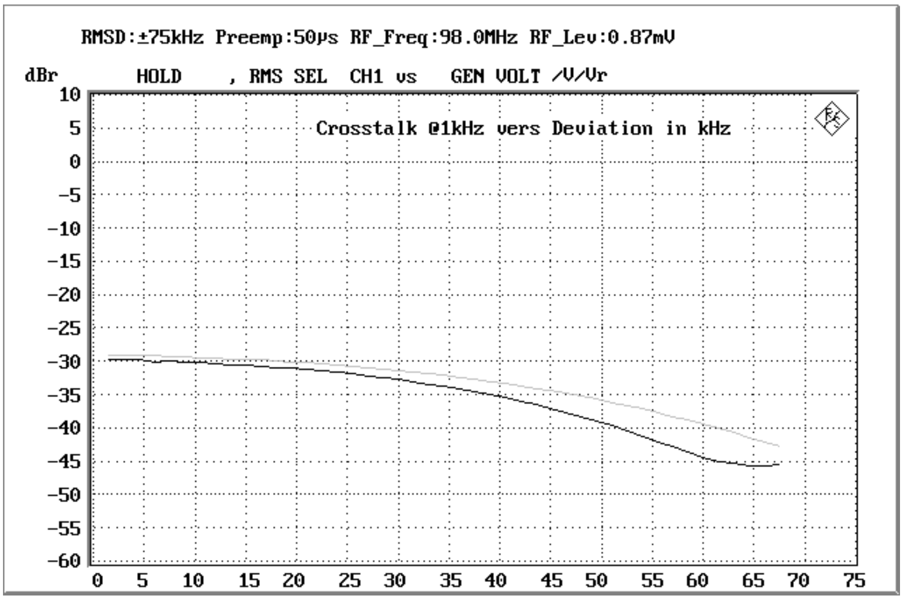


Fig. 5: Crosstalk attenuation as a function of modulation deviation

Crosstalk as a Function of RF Level / Stereo Switching Threshold

Another test is crosstalk measurement as a function of antenna input level. The receiver is operated under conditions specified by the standard but is set to maximum deviation. Starting at 100 nV, the input level is increased to 10 mV. Results are graphically displayed as is shown in Fig. 6.

This measurement shows the behaviour of the tuner when weak stereo signals are received. In the case of very weak antenna signals, reception is in the mono mode, i.e. the same signal is transmitted in both channels. In the diagram, this can be identified by the absence of crosstalk attenuation (0 dB). When the antenna voltage is increased, the stereo decoder starts operating at a certain level. This stereo threshold is clearly visible in Fig. 6 because of the sudden increase in crosstalk attenuation. The level of the stereo threshold is specified in the test report. For this purpose, a point is evaluated that has a crosstalk attenuation of 10 dB in both directions.

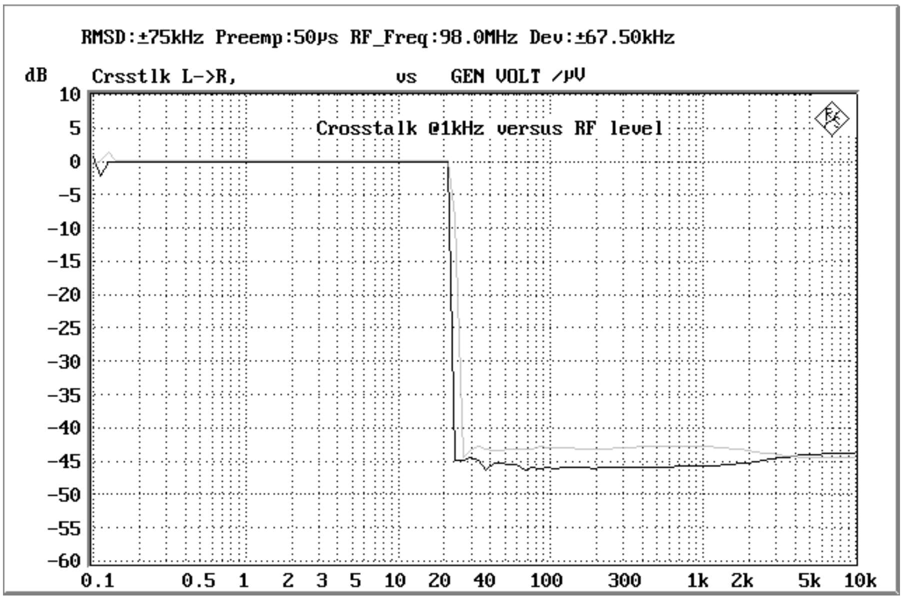


Fig. 6: Crosstalk attenuation as a function of the antenna level

Since the pilot-tone method is used, the noise increases when the stereo decoder is switched on. This is shown by the input signal / output signal characteristic in Fig. 8. The sudden noise increase is often disturbing particularly if this area is repeatedly traversed as the transmitter input signal varies. This is often the case in moving vehicles. For this reason smooth stereo switchover was developed where the crosstalk between the stereo channels is gradually increased with increasing antenna level. The use of such a circuit can be seen in the diagram where the crosstalk increases gradually above the stereo switching threshold.

Stereo operation is signalled on many FM broadcast receivers. However, the response threshold for this stereo indication need not be identical with the level at which the stereo coder starts to operate. Particularly when smooth stereo switchover is used, stereo reception is often signalled only when the received field strength is sufficiently high for adequate channel separation.

S/N Ratio as a Function of Input Level

The S/N ratio is the ratio of the audio frequency voltage of the signal to the noise voltage. According to IEC 60315-4, different weighting filters may be used with these measurements, but the test method with A-weighting in the HiFi sector has been adopted as standard and is the basis for the measurements described below.

The S/N ratio of receivers can be determined in different ways:

- When the sequential method is used and a modulated signal is received, the audio output voltage is measured, then modulation is switched off and the noise is measured. This corresponds to the in/out measurement shown in Fig. 9.
- When the simultaneous method is used and a modulated signal is received, the level of the 1 kHz audio signal is measured. The noise voltage is then determined with the aid of appropriate bandstop filters while a modulated RF signal is present. Since the noise output voltage of an FM receiver increases under certain circumstances when a modulated signal is present, this method more accurately represents the conditions encountered in practical applications.

In this application, the S/N ratio is measured with the simultaneous method. The rms value is measured with an A-weighting filter. This corresponds to the conditions used in the HiFi world.

The receiver is operated under the conditions specified by the standard and set to maximum deviation. The signal is stereo-modulated with a 1 kHz signal. The use of the bandpass filter reduces the measurement range to between 200 Hz and 15 kHz; effects of hum or insufficient pilot tone suppression are not taken into account. After the audio output voltage has been determined, the 1 kHz component is separated by a notch filter and the noise voltage is measured. In order not to influence the noise by the THD of the 1 kHz signal, a measurement is chosen in the Audio Amplifier R&S UPL, where any harmonics are also ignored. The S/N ratio is calculated from the signal voltage and the noise voltage and graphically displayed as a function of the RF input level (Fig. 7).

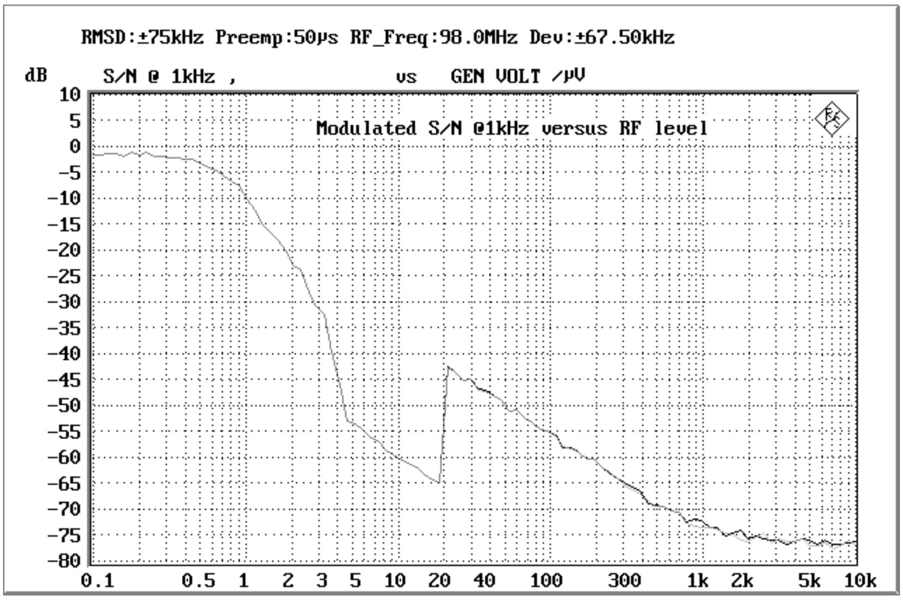


Fig. 7: S/N ratio as a function of the antenna voltage

Input Signal / Output Signal Characteristic

The in/out characteristic illustrates the relation between the antenna input voltage and the audio signal generated by the tuner. This is one of the most important measurements because the diagram provides a great deal of information particularly when the noise output voltage is considered in the measurement.

The receiver is operated under the conditions specified by the standard. For measuring the audio output voltage, the signal is modulated with maximum deviation; for the noise measurement with 0 deviation. The antenna level is logarithmically swept from 100 nV to 10 mV. The audio output signal is measured and graphically displayed with the maximum output voltage set to 0 dB. The level sweep is repeated and the noise output voltage is recorded. A diagram as shown in Fig. 8 is obtained.

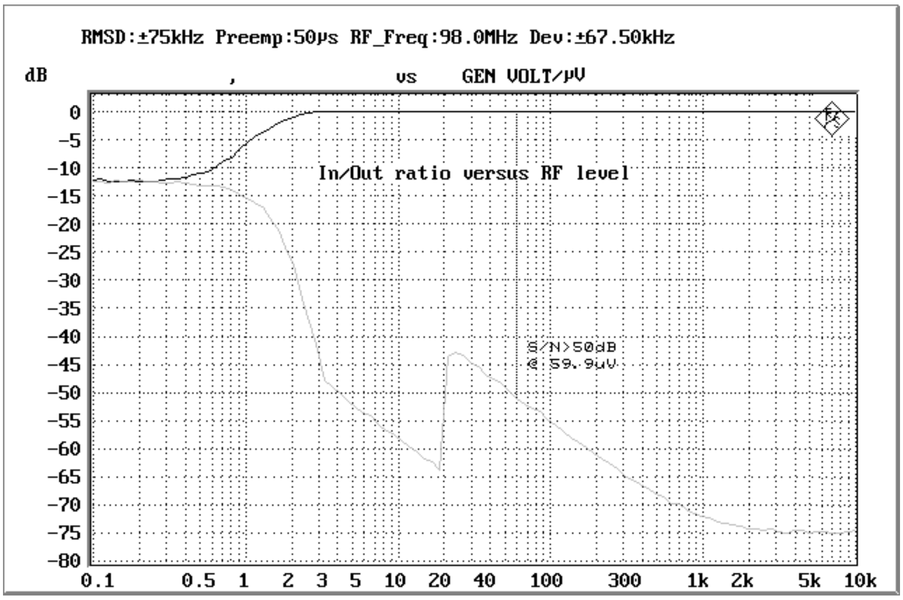


Fig. 8: Output / input signal characteristic of tuner with 50 dB S/N ratio

The following information can be obtained from the diagram:

• Characteristic of audio output signal

Only above a certain antenna input voltage will the tuner be able to detect an audio signal in the RF signal. This is at the point in the diagram where the signal characteristic and the noise characteristic separate. The associated level can be referred to as the absolute sensitivity of the tuner but plays only a minor role in amplifier characteristics.

The output signal characteristic always shows a steep rise when the antenna signal increases and then continues at a constant level. Depending on the receiver, this maximum level (which is also the reference level for the measurement) is attained with different RF levels.

• Noise signal

As the RF signal increases, the noise goes down until it attains its minimum. This minimum value is lower in the mono mode than in the stereo mode.

In the stereo mode, the level sweep of the input signal first shows the same characteristic as for a mono signal. When a certain signal level is attained, the stereo decoder starts operating (stereo threshold). This is first noticed by a clear increase of noise. As the input signal level rises, the noise is reduced again but normally does not attain the minimum value attained in mono reception.

- Max. S/N ratio

The maximum S/N ratio can be determined from the maximum audio output signal level and the minimum noise level. This value is also documented in the test report.

- Noise-limited sensitivity

The noise-limited sensitivity is the antenna level at which an audio signal with defined S/N ratio is obtained. The sensitivity value is at the same time a measure of the replay quality of the audio signal.

If HiFi stereo receivers are used, this value should be tested at an S/N ratio of 50 dB.

For instance, sensitivities of approximately 3 μV for mono reception and of 30 to 40 μV for stereo can be assumed for modern, high-quality tuners. The sensitivity for an S/N ratio of 40 dB and 30 dB is also determined. These values are often attained only below the stereo switchover threshold. They are documented in the report for information only.

Suppression of Pilot Tone and Subcarrier

A pilot tone is transmitted at 19 kHz to identify stereo broadcast transmissions. In order not to disturb instruments such as amplifiers and recorders connected to the tuner, the pilot tone and its subcarriers must be sufficiently suppressed in the tuner. This is done by circuits in the stereo coder or by means of filters at the tuner output.

Another quality criterion of a tuner is the suppression of pilot tone, auxiliary carrier or other interfering products. The audio frequency voltage normally measured at maximum deviation and at conditions specified by the standard is then used as a 0 dB reference in the display. Subsequently, the useful signal modulation is set to 0 and the remaining frequency components are displayed. The pilot signal suppression is measured and documented in the report. At the same time, the highest signal component is searched for and also documented in the report as interference suppression.

Fig. 9 shows the signal spectrum. The individual frequencies can be clearly identified.

High-quality tuners should suppress all frequency components above the transmission range by at least 50 dB.

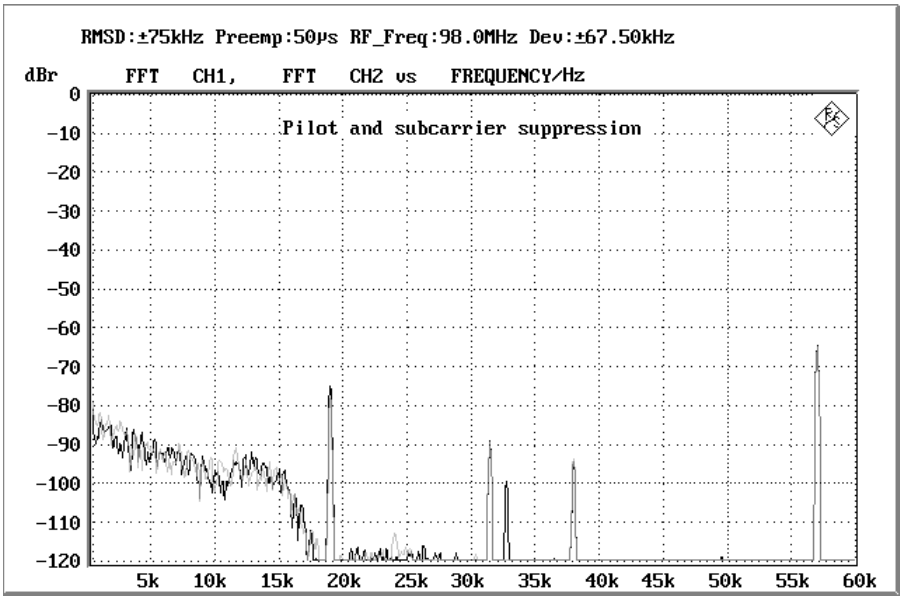


Fig. 9: Unmodulated output spectrum of a tuner with residual pilot tone, subcarrier and interfering components

8 Demo Programs for Production Tests

With the installation of the tuner program automatically the programs MULTFREQ.BAS and FASTDIST.BAS are generated. Those programs are intended as examples for fast frequency response and distortion measurements in production line use.

After loading and starting the program MULTFREQ it will be asked if the measurement should be done with approximately 20, 50 or 100 frequency points. Then after each hit of the SPACE key a measurement is done and displayed and the measured frequency response is PASS/FAIL checked versus the given frequency tolerance limits. The program can be stopped using the ESC key.

The program FASTDIST works in a similar way but after start it asks for the limit value of the measured distortion. The measurement is started with the SPACE key, the measured distortion value (THD+N) will be displayed and checked against the verdict value. The program can be stopped using the ESC key.

9 References

Methods of measurement on radio receivers for various classes of emission - Part 4: Receivers for frequency-modulated sound broadcasting emissions. IEC 60315-4: 1997

10 Ordering Information

Audio Analyzer	**R&S UPL**	**1078.2008.06**
Universal Sequence Controller (option)	**R&S UPL-B10**	**1078.3904.02**
Signal Generator	**R&S SML01**	**1090.3000.11**
Signal Generator	**R&S SML02**	**1090.3000.12**
Signal Generator	**R&S SML03**	**1090.3000.13**
Signal Generator	**R&S SMV03**	**1147.7509.13**
Stereo/RDS Coder (option)	**R&S SML-B5**	**1147.8805.02**
Matching Pad 50/75 Ohm	**R&S RAM**	**0358.5414.02**

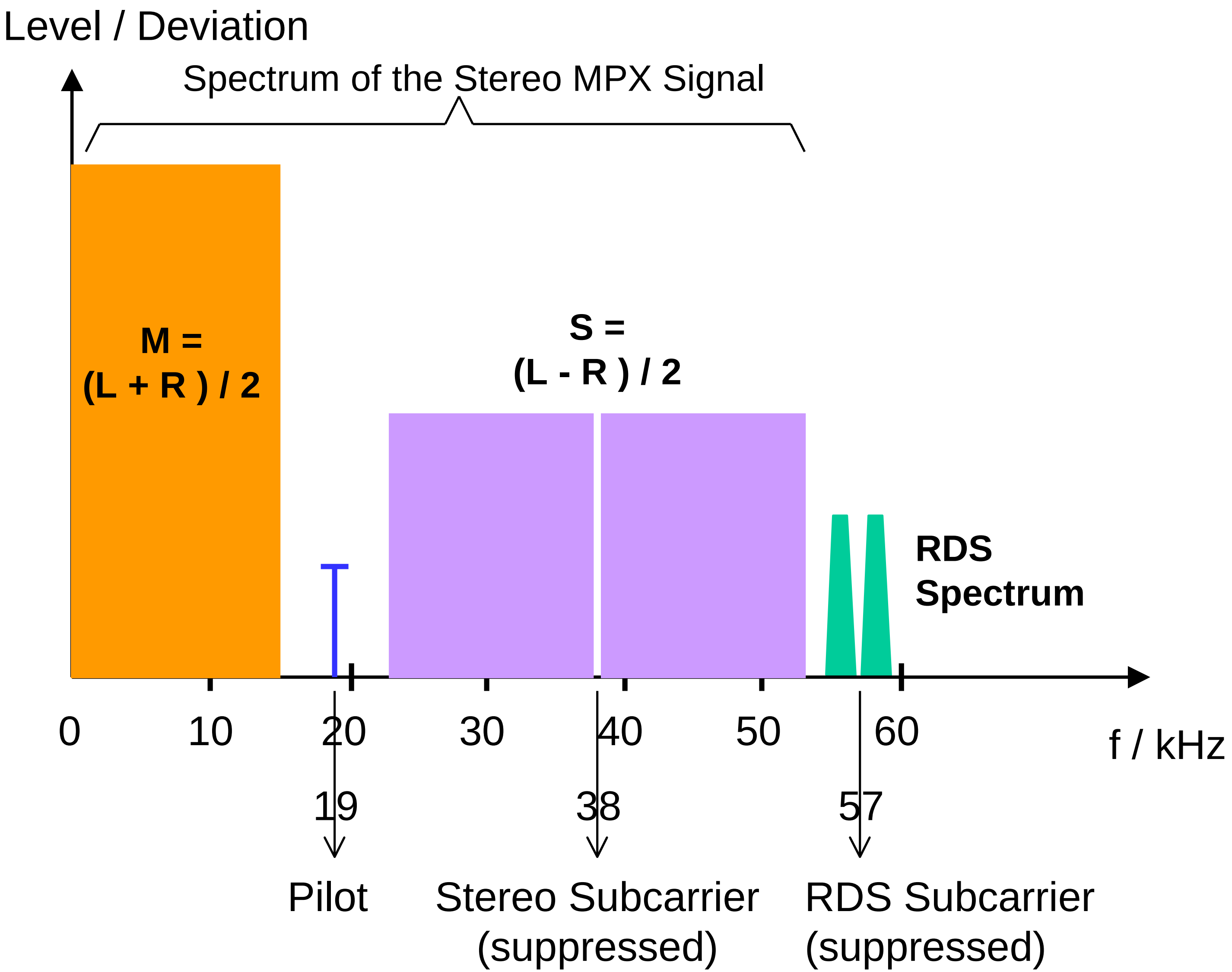
ROHDE & SCHWARZ GmbH & Co. KG · Mühldorfstraße 15 · D-81671 München · Postfach 80 14 69 · D-81614 München · Tel (089) 4129 -0 · Fax (089) 4129 - 13777 · Internet: http://www.rohde-schwarz.com

FM Stereo Tests with SML/SMV and UPL

Overview

- FM Basics
- Standards for FM Radio
- World-Wide Standards
- Requirements
- Setup
- Measurements
- Further Information
- Typical Measurement Results

FM Basics



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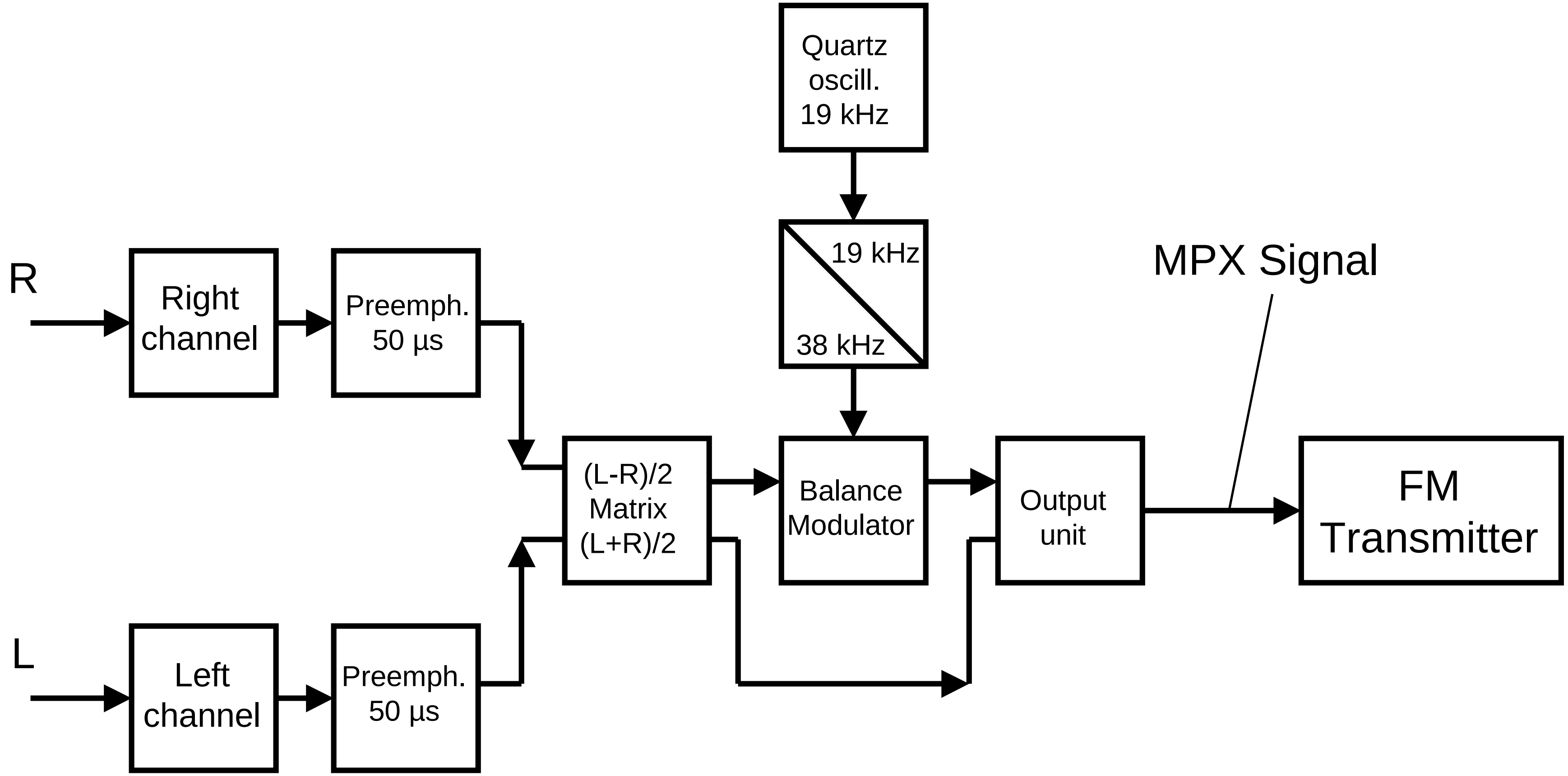
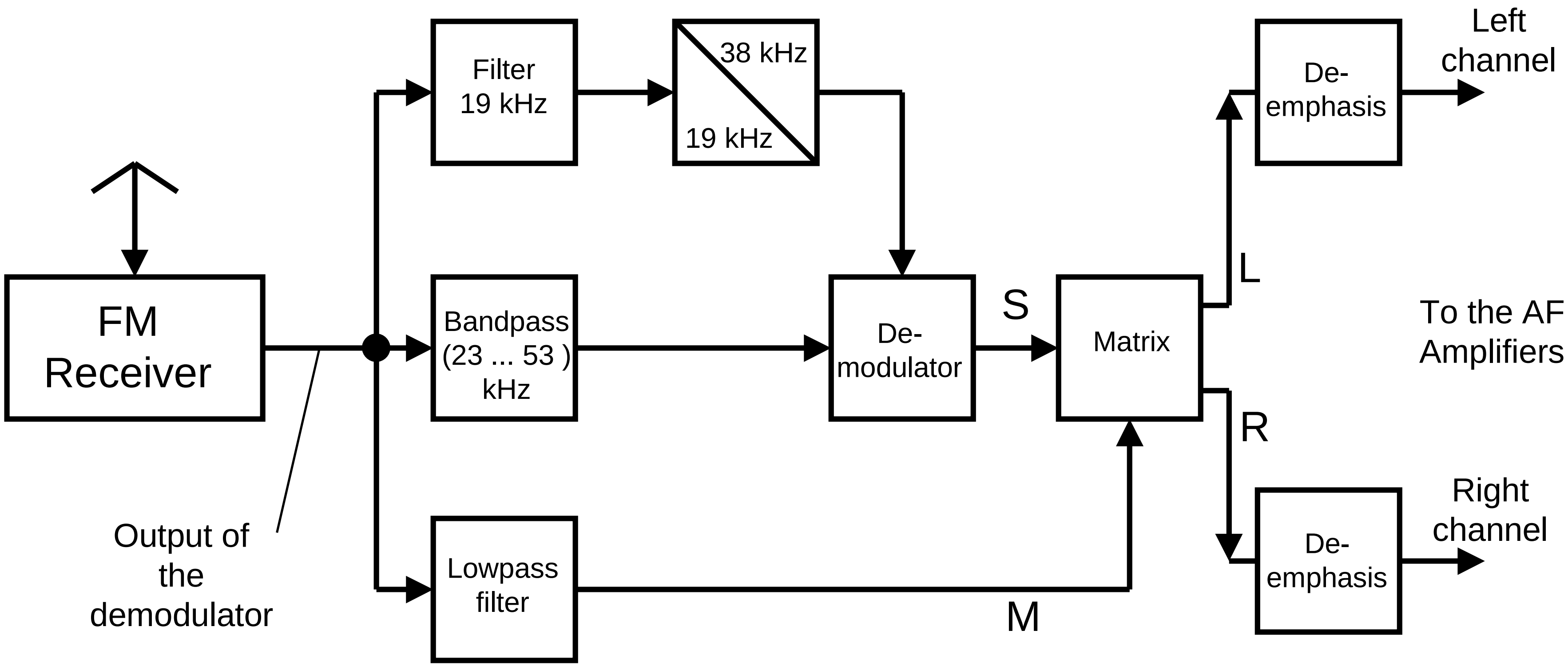
FM Basics



Block diagram of the stereo coder (TX)

FM Basics



Block diagram of the stereo decoder (RX)

Standards for FM Radio

RECOMMENDATION ITU-R BS.450-2
TRANSMISSION STANDARDS FOR FM SOUND BROADCASTING AT VHF

Worldwide standards differ in following points:

Rated Maximum System Deviation RMSD	50 kHz or 75 kHz
Preemphasis	50 µs or 75 µs
Frequency Range	65.8 MHz to 108 MHz

Measurement definitions:

IEC 60315-4 Methods of Measurement on radio receivers for various classes of emission - Part 4: Receivers for frequency-modulated sound braodcasting emissions (1997)

Standards for FM Radio

General definitions for system deviation:

Type / Signal	RMSD ± 50 kHz	RMSD ± 75 kHz
Mono	± 50 kHz	± 75 kHz
Stereo	90 % of RMSD -> ± 45 kHz	90 % of RMSD -> ± 67.5 kHz
Pilot	9 % of RMSD -> ± 4.5 kHz	9 % of RMSD -> ± 6.75 kHz

World-Wide Standards

TABLE 1a

Terrestrial FM sound broadcasting (above 30 MHz)

Country/Geographical area	International agreements			Information related to current emission applications: Frequency bands used (MHz)								Modulation characteristics							Polarization			Transmitter frequency tolerances (RR Article 1)	
	☐ Geneva 60	☐ Stockholm 61	☐ Geneva 84	☐ Others	☐ 66-68	☐ 68-73	☐ 73-74	☐ 76-87.5	☐ 87.5-108	☐ 88.0-108	☐ Others	☐ Monophonic	☐ Stereophonic	☐ Polar-modulation system	☐ Pilot-tone system	☐ Channel spacing (1) (kHz)	☐ Pre-emphasis/ de-emphasis (µs)	☐ Maximum frequency deviation (kHz)	☐ Horizontal	☐ Vertical	☐ Mixed	Current requirement	Long-term design objective
Germany (Federal Republic of)		+	+						+			+	+		+	100	50	±75	+	Except [illegible]			
Aruba										+			+			200	75	±75		+	+		
Australia									+				+		+	200	50	±75	+	+	+		
Bahamas										+			+		+	200	75	±75	+				
Bangladesh (People's Republic of)			+						+			+				200	50	±75	+				
Cyprus (Republic of)			+						+				+		+	100	50	±75			+		
Vatican City State		+	+						+				+		+	100	75	±75			+		
Colombia (Republic of)			+							+			+			200	75	±75			+		
Korea (Republic of)			+							+		+	+		+	200	75	±75			+		
Denmark			+						+				+		+	100	50	±75	+				
Ecuador										+			+		+	200		±75		+			
Spain			+						+				+		+	100	50	±75	+	+	+		
United State of America											87.8-108	+	+		+	200	75	±75	+	+	+		
Finland			+						+				+		+	100	50	±75	+	+	+		
France			+						+			+	+		+	100	50	±75	+	+		20×10^{-6}	
Gambia (Republic of the)			+							+		+	+		+		75	±75		+			
Hungary (Republic of)	+	+	+		+	+			+			+	+		+	30 100	50	±50 ±75	+				
India (Republic of)											100-108	+	+		+	100	50	±75			+		
Iran (Islam Republic of)			+						+			+	+		+		50	±75	+		+		
Italy			+						+				+		+	100	50	±75			+		
Japan				X							76-90				+	100	50	±75	+	+			

01-T01-1

World-Wide Standards

TABLE 1a *(continued)*

Country/Geographical area	International agreements: Geneva 60	Stockholm 61	Geneva 84	Others	Frequency bands used (MHz): 66-68	68-73	73-74	76-87.5	87.5-108	88.0-108	Others	Modulation characteristics: Monophonic	Stereophonic	Polar-modulation system	Pilot-tone system	Channel spacing (1) (kHz)	Pre-emphasis/de-emphasis (µs)	Maximum frequency deviation (kHz)	Polarization: Horizontal	Vertical	Mixed	Transmitter frequency tolerances (RR Article 1): Current requirement	Long-term design objective
Kuwait (State of)			+						+				+			100	50	±75			+		
Lithuania		+	+		+	+	+		+			+	+	+	+	30 100	50 75	±50 ±75	+	+			
Mali (Republic of)			+						+				+		+	100	50	±75	+	+			
Morocco (Kingdom of)		+	+						+			+	+		+		75	±75	+		+		
Norway		+	+						+				+		+	100	50	±75	+	few	+		
New Zealand				ITU-R as Rec 412							88-100		+		+	50	50	±75		+	+		
Oman (Sultanate of)			+						+				+		+	100	50	±75	+	+			
Papua New Guinea			+						+			+	+		+	100	50	±75	+				
Netherlands (Kingdom of)			+						+			+	+		+	100	50	±75	+	+			
Qatar (State of)										+			+			200	50	±75			+		
Czech Republic									+			+	+		+	100	50	±75	+	+	+		
United Kingdom of Great Britain and Northern Ireland			+						+				+		+	100	50	±75			+		
Rwanda (Republic of)	+		+						+			+				100	50	±75	+				
Senegal (Republic of)			+						+			+	+		+	100	50	±75	+				
Singapore (Republic of)										+			+		+	300	50	±75			+		
Slovenija (Republic of)		+	+						+				+			100	50	±75	+	+	+		
South Africa (Republic of)			+						+				+		+	100	50	±75		+			
Sweden			+						+				+		+	100	50	±75	+				
Switzerland (Confederation of)			+						+				+		+	100	50	±75	+	few	few		
Turkey			+						+				+		+	100	75	±50	+				

(1) For definition see Recommendation ITU-R BS.412. It is not meant the frequency spacing in overlapping service areas or tuning steps of the receiver.

01-T01-7

World-Wide Standards

Country/Geographical area	Information related to current receiving applications				Additional information		Remarks
	Recommended or used IF (MHz)	Oscillator position		Electromagnetic immunity requirements for receivers	Compressor or compander systems	Supplementary information	
		High	Low				
Germany (Federal Republic of)	10.7	+		EN 55 020	Yes	ARI, RDS	Variable pre-emphasis at transmitter site in order to avoid excess of $\pm$75 kHz frequency deviation
Aruba	10.7	+					
Australia	10.7	+				ACS on 57 kHz (RDS) 67 kHz and below 95 kHz	
Bahamas							
Bangladesh (People's Republic of)	10.7	+					
Cyprus (Republic of)							
Vatican City State					Compression +10 dB		
Colombia (Republic of)	10.7				No	SCA (67 kHz)	
Korea (Republic of)	10.7	+			Optimod FM 8200	No	
Denmark	10.7	+		EMC	Yes	RDS	
Ecuador	10.7						
Spain	10.7	+				RDS, SCA (67 kHz)	

World-Wide Standards

Country/Geographical area	Information related to current receiving applications				Additional information		Remarks
	Recommended or used IF (MHz)	Oscillator position		Electromagnetic immunity requirements for receivers	Compressor or compander systems	Supplementary information	
		High	Low				
United States of America	10.7	Not defined		FCC 47 CFR 15	Optional	RBDS (RDS), SCA	
Finland	10.7	+			ORBAN compressor	RDS	
France	10.7	+			Yes, mainly for local radio	RDS	Synchronous frequency VHF-FM service for motorists in stereophonic mode along motorways. Frequency tolerance among all synchronous transmitters: 10^{-9}
Gambia (Republic of)	10.8	+					
Hungary (Republic of)	10.7	Not defined		EN 55020, draft Hungarian standard		ARI, RDS, SCA pilot, MBS	
India (Republic of)	10.7		+			RDS, SCA (experimental transmissions)	
Iran (Islam Republic of)	10.7	+		No	No	RDS	
Italy	10.7	+			Compressor of deviation control		“ISORADIO” – ISO frequency VHF-FM service for motorists in monophonic mode is introduced along the motorways
Japan	10.7	+				DARC	
Kuwait (State of)	10.7	+					
Lithuania (Republic of)	10.7	+					
Mali (Republic of)	10.7						

World-Wide Standards

Country/Geographical area	Information related to current receiving applications				Additional information		Remarks
	Recommended or used IF (MHz)	Oscillator position		Electromagnetic immunity requirements for receivers	Compressor or compander systems	Supplementary information	
		High	Low				
Morocco (Kingdom of)							
Norway	10.7	+			Yes	RDS	
New Zealand	10.7	+				SCA use being developed	100-108 MHz presently used for domestic services
Oman (Sultanate of)					None	None	
Papua New Guinea					None	None	
Netherlands (Kingdom of the)	10.7	Left to manufacturer		Comply with EEC standards	Yes	RDS, CSI	
Qatar (State of)						No	
Czech Republic	10.7	+			Compression	RDS	
United Kingdom of Great Britain and Northern Ireland	10.7	+		REC, EEC EMC Directive; Radiation EN 55013; Immunity 55020	Yes	RDS	
Rwanda (Republic of)	10.7	+					
Senegal (Republic of)	10.7						
Singapore (Republic of)	10.7	+			Optimod	SCA	
Slovenija (Republic of)	10.7	+			Yes	RDS	
South Africa (Republic of)	10.7	+	+	No	Optimod	RDS, SST	SST still on trial
Sweden	10.7	+		No	Yes, audioprocessing (compression, limiter)	RDS	
Switzerland (Confederation of)	10.7	+				ARI, RDS	
Turkey	10.7		+	No	No	No	

Requirements for FM Radio Measurements

Simple receiver tests:

- SML01/02/03 or SMV03 with option SML-B5 (Stereo/RDS coder)

General-purpose test system for FM tuners:

- UPL (with option UPL-B10: Universal Sequence Controller)
- SML01/02/03 or SMV03 with option SML-B5 (Stereo/RDS coder)

Setup (1)

Simple receiver tests

Signal Generator SML with SML-B5

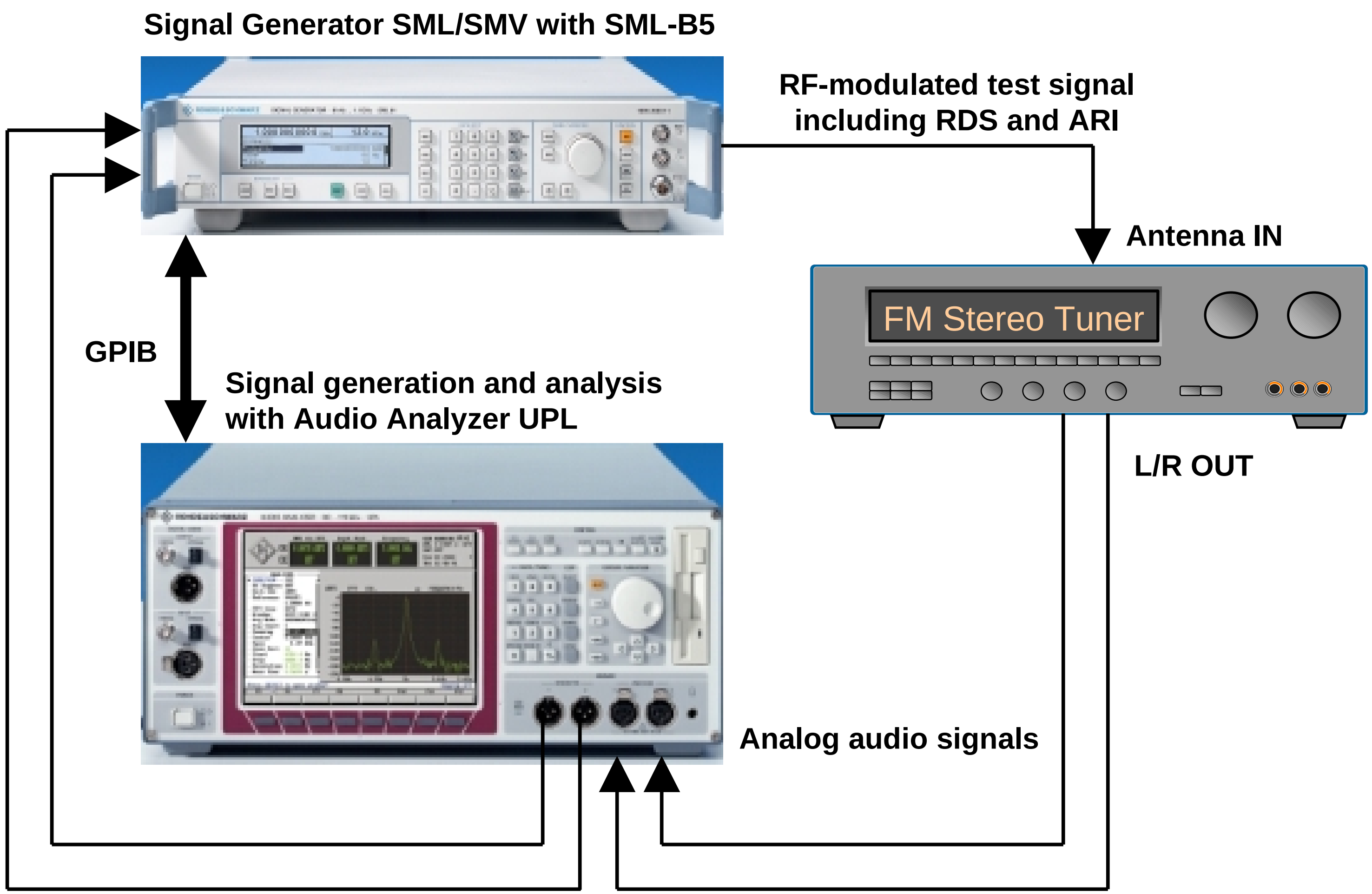
RF modulated test signal
including RDS and ARI

Principle tests, e.g.:

- Pilot on/off ⇒ Stereo on/off
- Variation of pilot deviation ⇒ Stereo on/off
- RDS/ARI on/off ⇒ Test of RDS/ARI display

Setup (2)

General-purpose test system for FM tuners



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Measurements

The following measuerements according to IEC 60315 - 4 can be automatically performed with the measurement software which is part of the Application Note 1GA43_0E „Measurement on FM Tuners ...“:

- Frequency Response
- Total Harmonic Distortion + Noise (THD+N) vs. Frequency
- Total Harmonic Distortion + Noise (THD+N) vs. Deviation
- Crosstalk vs. Frequency
- Crosstalk vs. RF level
- Crosstalk vs. Deviation
- S/N vs. Level
- In/Out Ratio vs. RF Level
- Pilot Suppression

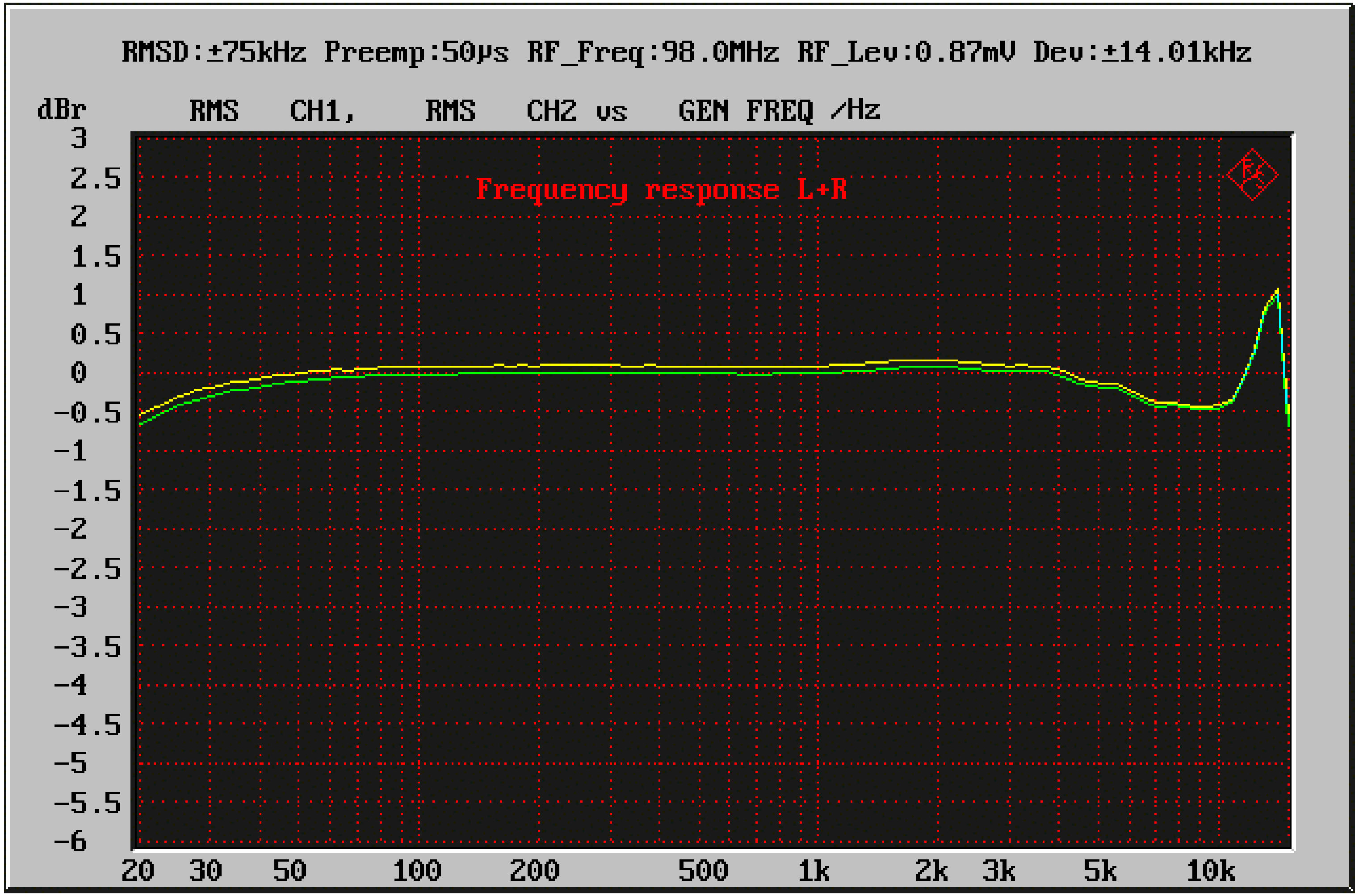
Further Information

- Application Note 1GA43_0e „Measurement on FM Tuners ... " including test software „TUNER.BAS" for UPL
- Sales Circular M02/1821 "Stereo/RDS-Coder R&S SML-B5"
- SML-B5 Data Sheet (PD 0757.7152.21
- UPL Data Sheet (PD 0757.2238.25)
- SML Data Sheet (PD 0757.0935.23)
- SMV Data Sheet (PD 0757.7175.21)

All information available from Gloris / R&S Website

Typical Measurement Results

- Frequency Response 20 Hz ... 15 kHz



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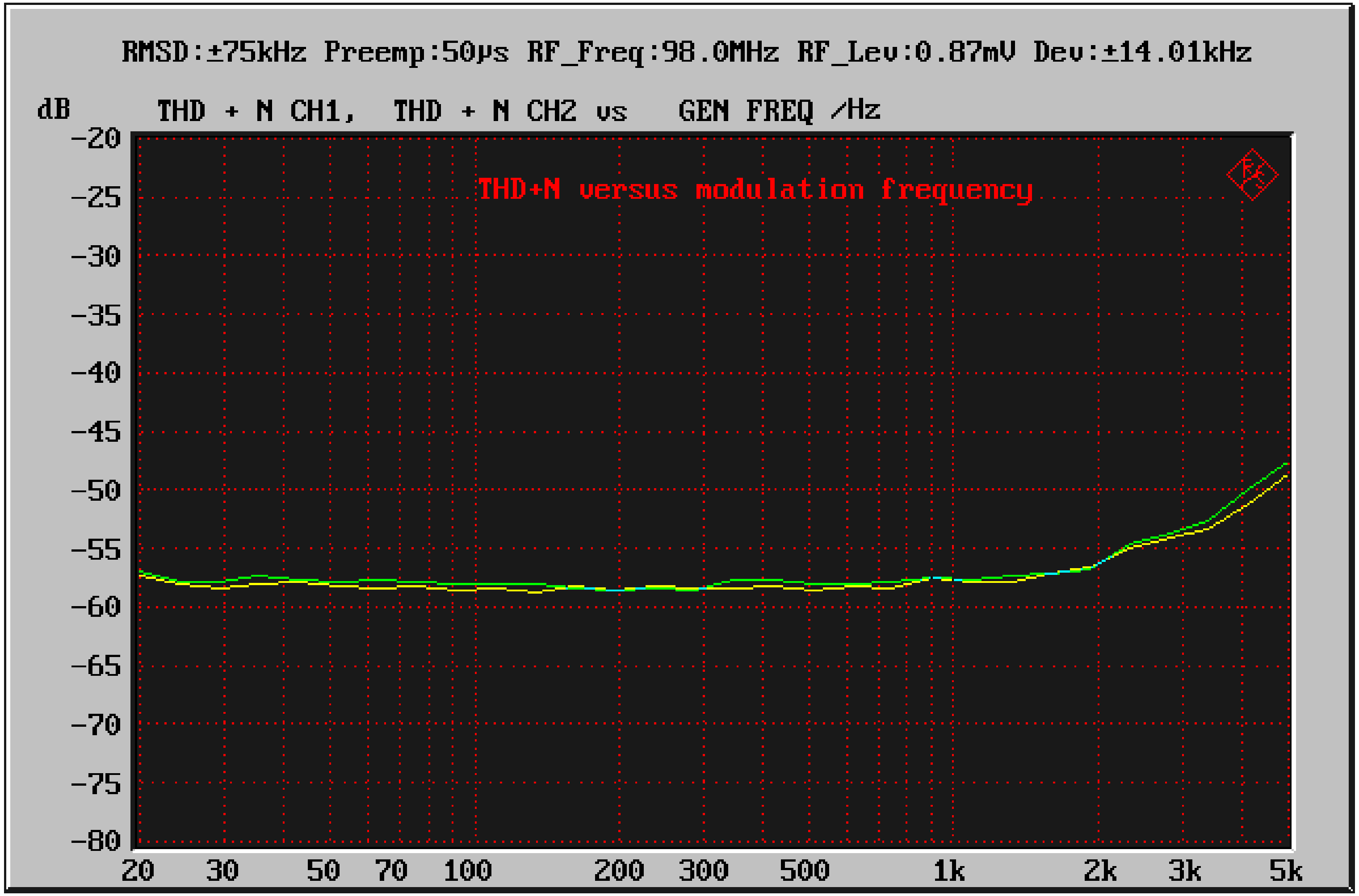
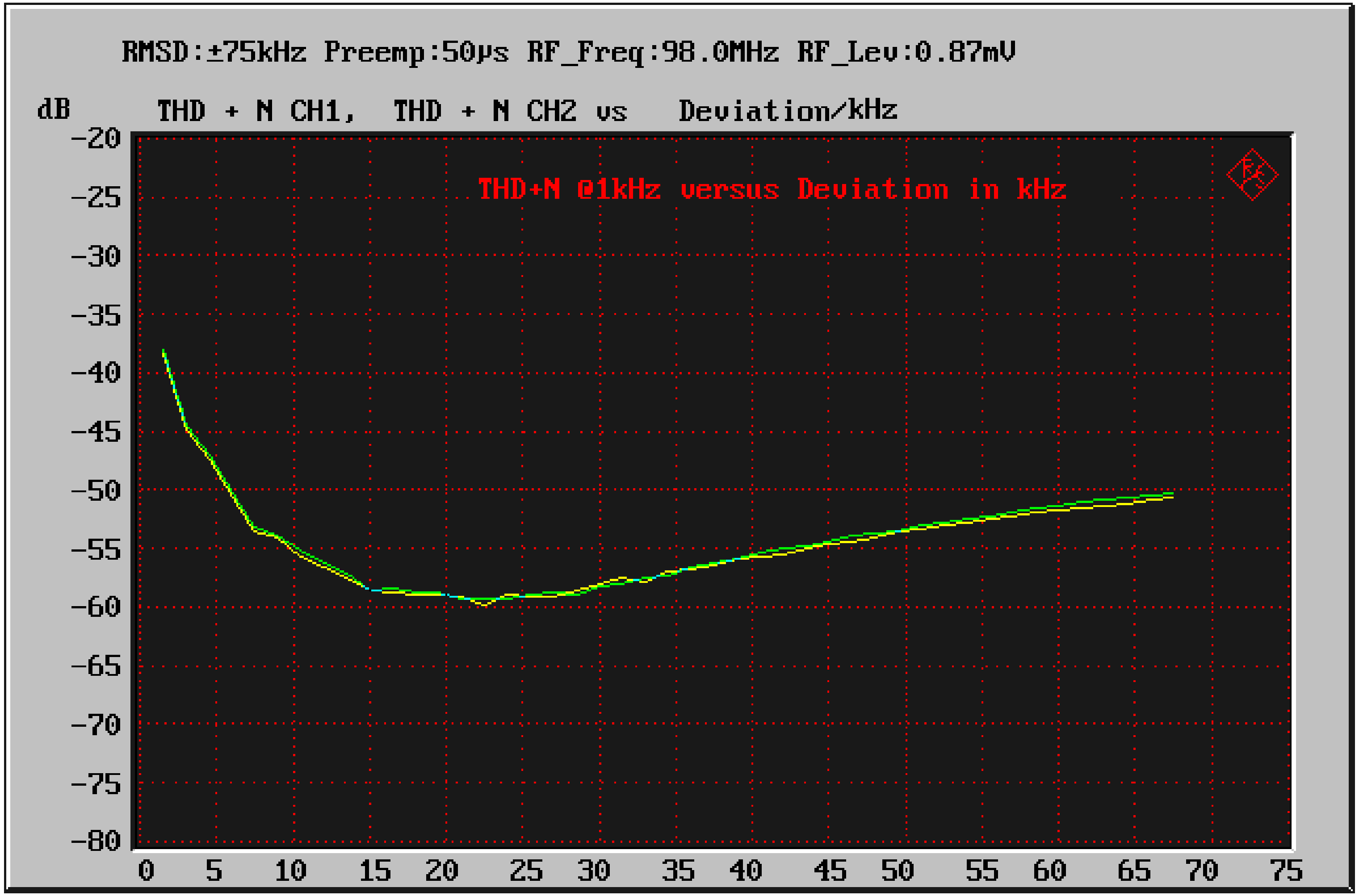
Typical Measurement Results

- Total Harmonic Distortion + Noise versus Frequency



Typical Measurement Results

- Total Harmonic Distortion + Noise versus Deviation



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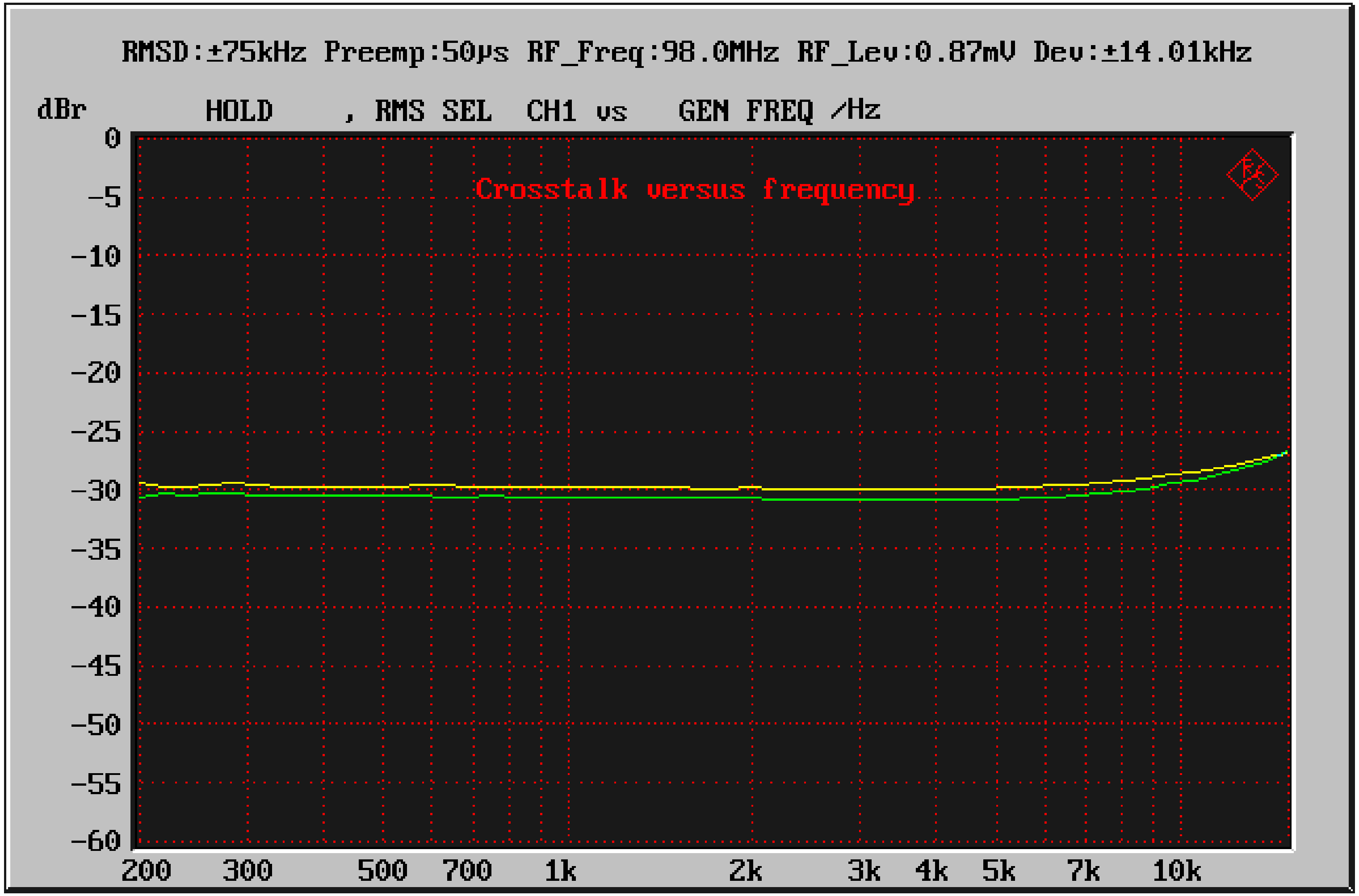
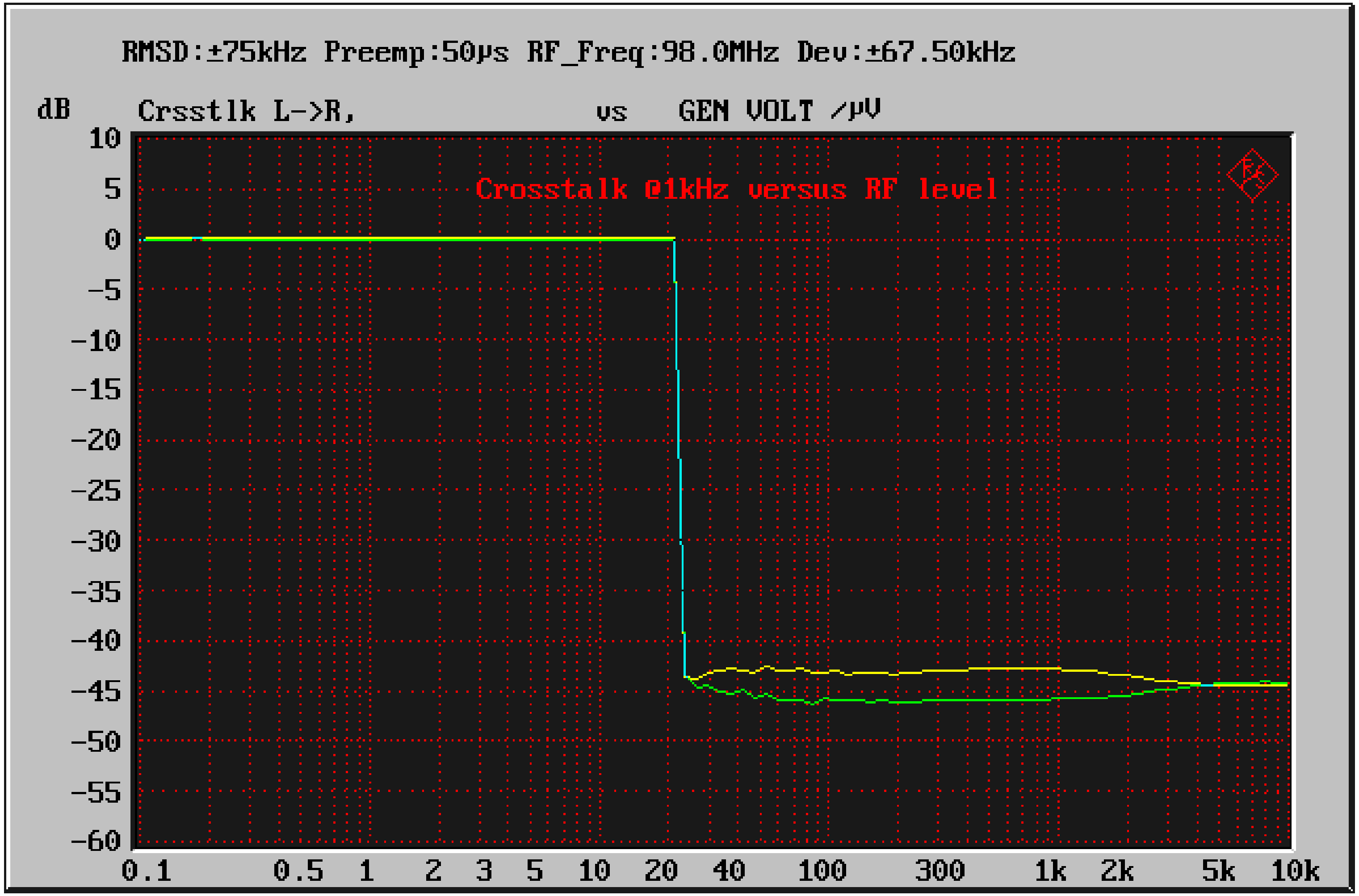
Typical Measurement Results

- Crosstalk versus Frequency



Typical Measurement Results

- Crosstalk versus RF Level



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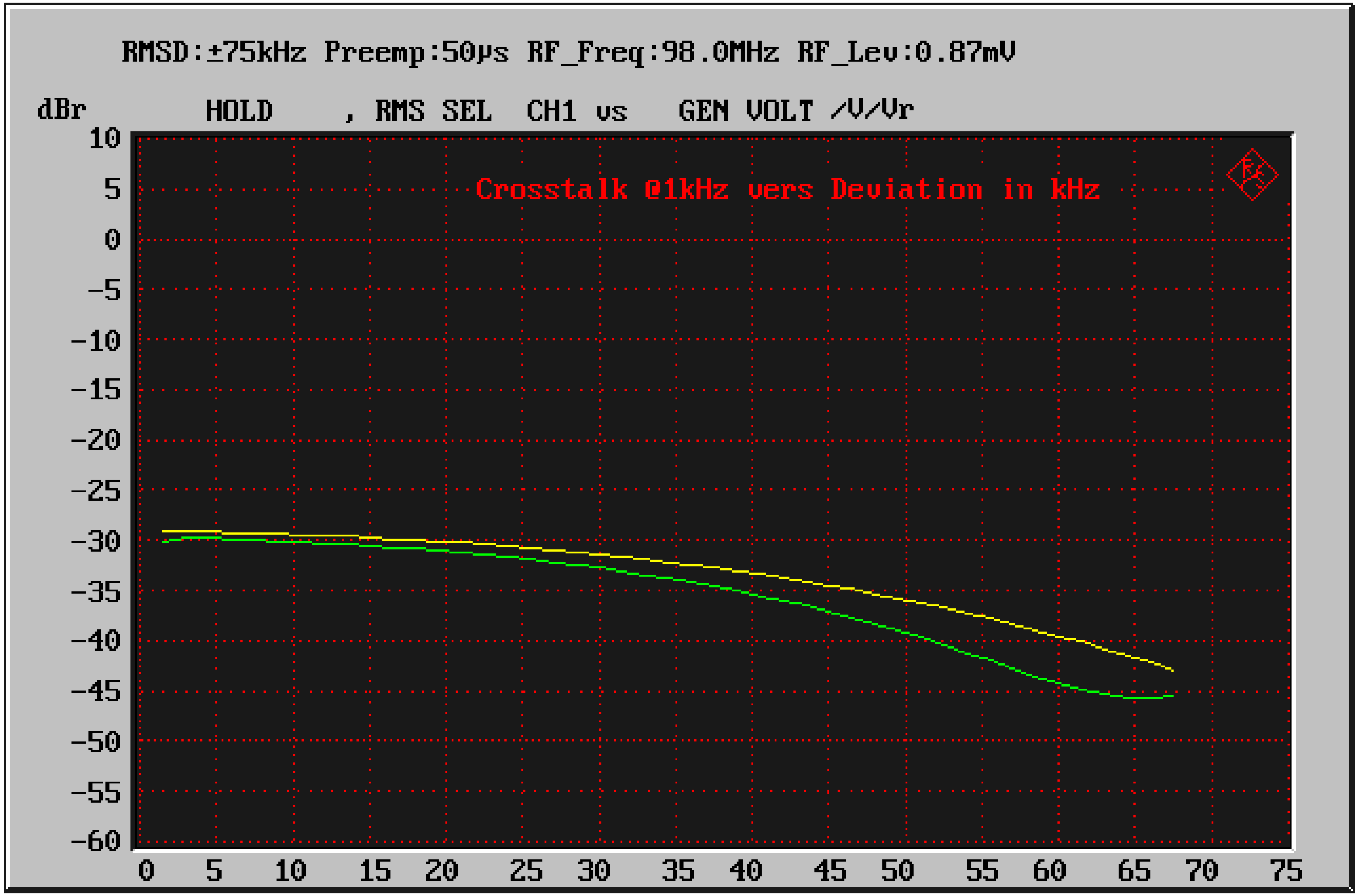
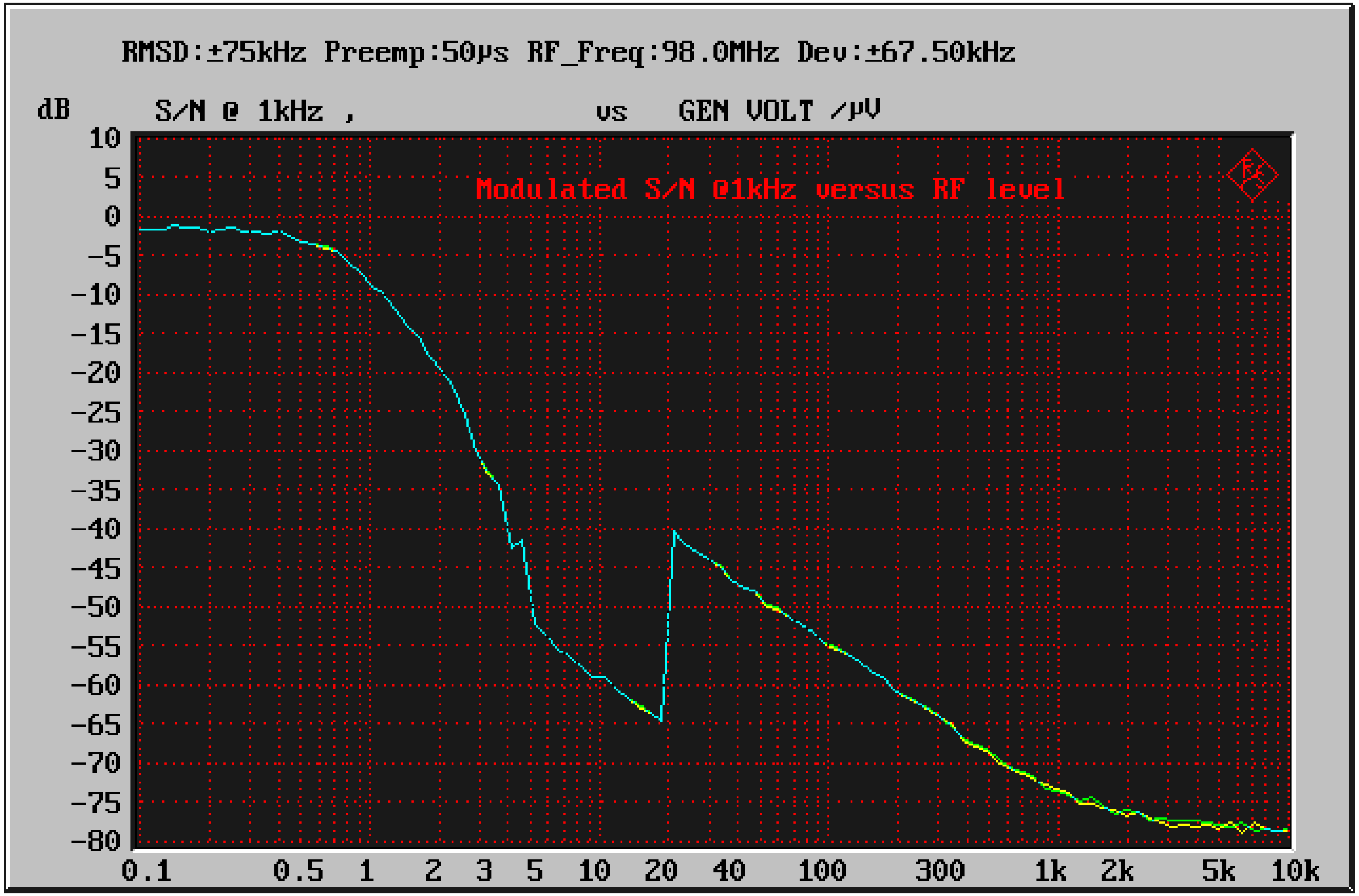
Typical Measurement Results

- Crosstalk at 1 kHz versus Deviation



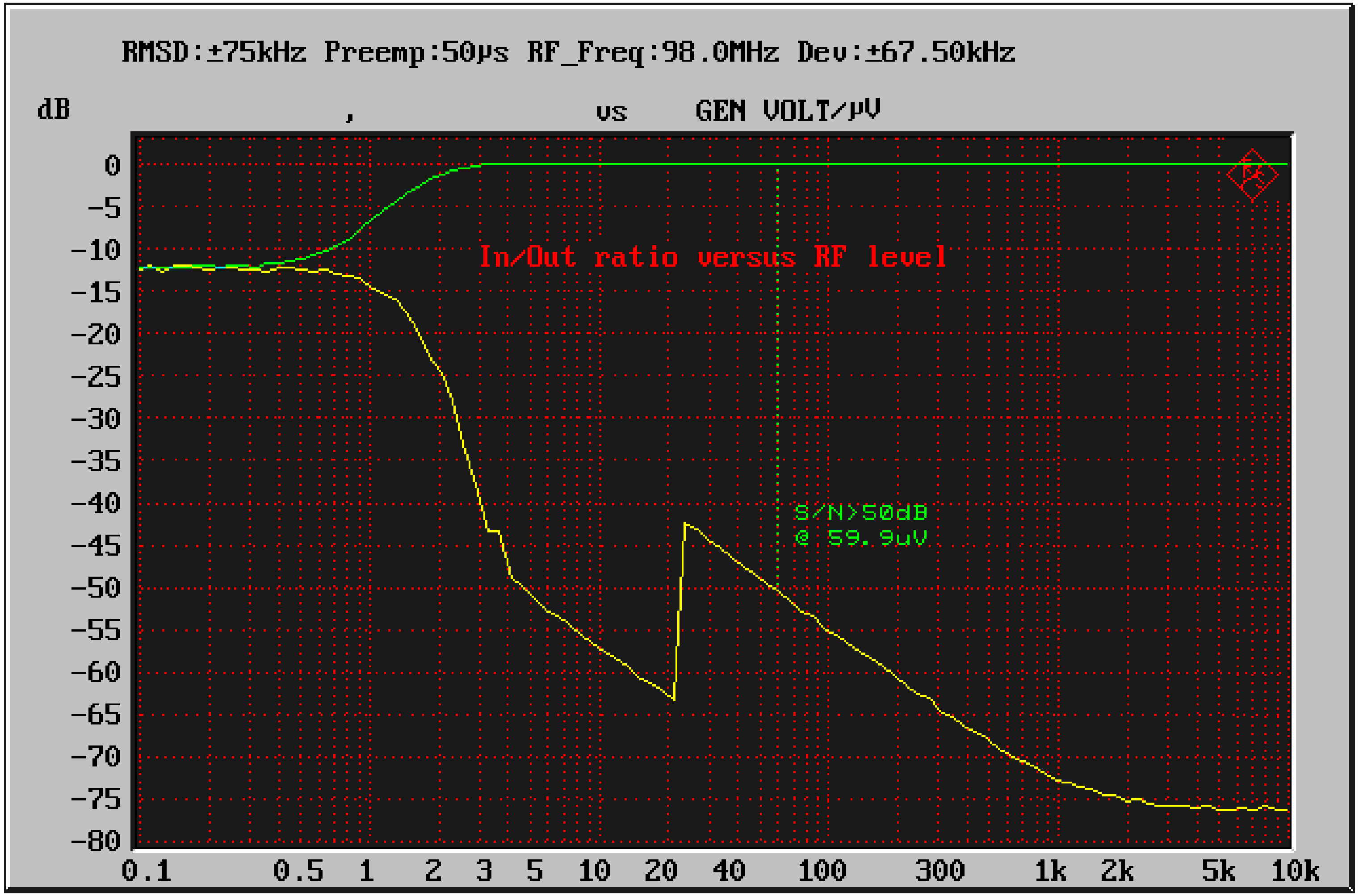
Typical Measurement Results

◆Signal / Noise Ratio at 1 kHz without Harmonics



Typical Measurement Results

◆IN / OUT Characteristics



Typical Measurement Results

◆Pilot and Spurious Suppression