Acoustic Measurements

on

GSM Mobile Phones

with

CMU200

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

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

Products involved

Hardware

Audio Generator and Analyzer	CMU-B41
Speech Codec	CMU-B52
Signaling Unit	CMU-B21

Software (at least one of the following ones)

Software GSM400	CMU-K20
Software GSM850	CMU-K24
Software GSM900	CMU-K21
Software GSM1800	CMU-K22
Software GSM1900	CMU-K23

based on sw V3.00 and higher

The option *Audio Generator and Analyzer CMU-B41* also includes the Multitone measurement feature.
For a description of the features please refer to chapter 4 of CMU Operating Manual, Id.No. 1100.4903.12 .

The option *Speech Codec CMU-B52* contains Fullrate, Enhanced Fullrate and Halfrate codec.

ETSI documents, eg. GSM 04.08 or TS 44.008, define „Fullrate Version 1“ as Fullrate speech codec and „Fullrate Version 2“ as Enhanced Fullrate speech codec.

CMU200 - ... Audio Option B41 installed?

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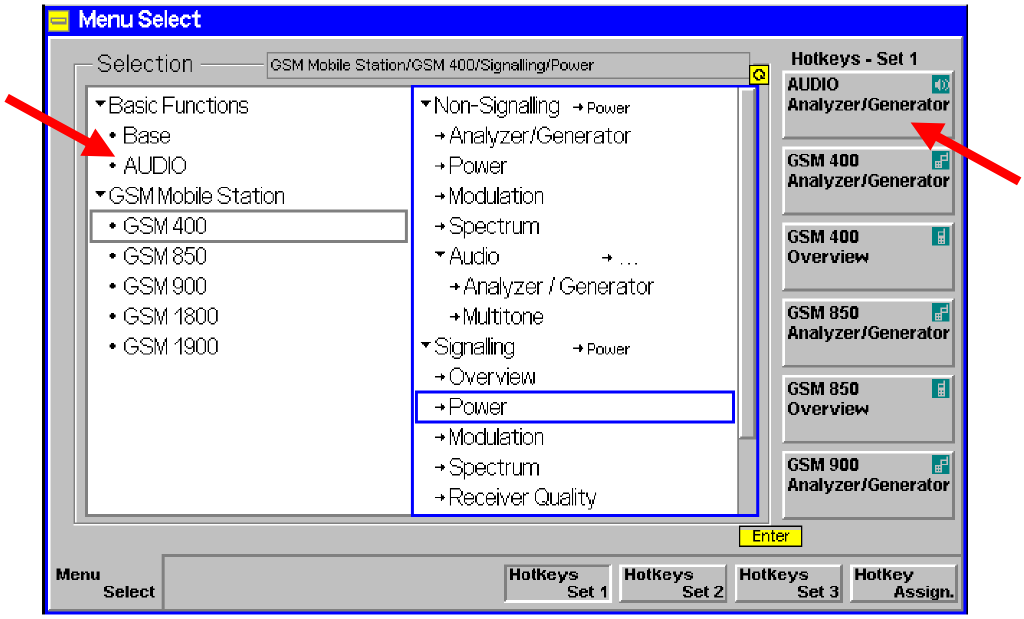

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The option *Audio Generator and Analyzer CMU-B41* also includes the Multitone measurement feature.

The Audio option *CMU-B41* is a *Basic Functions* feature.
This means it can be used network **independent**.

CMU200 - Audio function group

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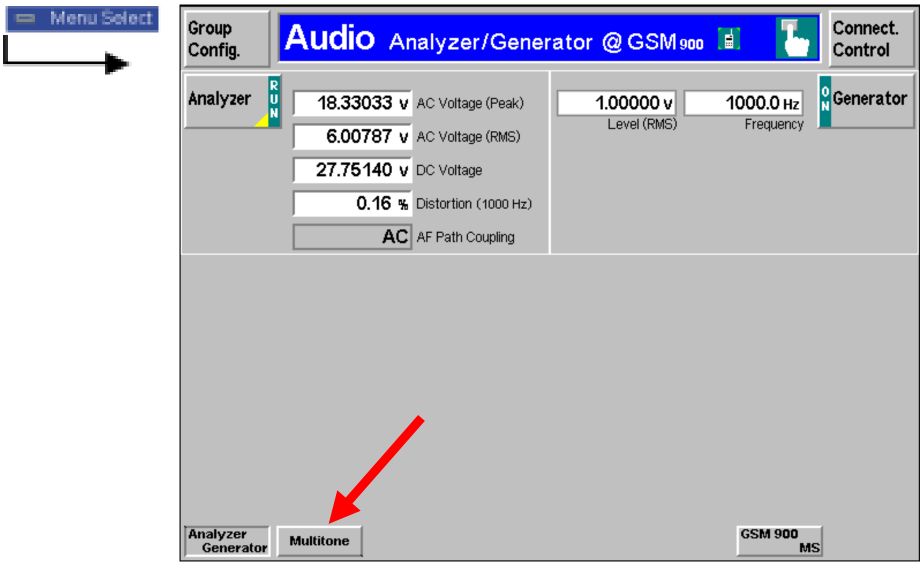
Audio Generator and Analyzer Option CMU-B41



The main menu *Analyzer/Generator* defines the sinusoidal signal generated by the audio generator and displays the voltage of the measured audio signal.
The *Analyzer/Generator* menu is opened from the main menu *Menu Select* (with associated key at front of instrument) or via the *Audio* hotkey which is available in all *GSM400/850/900/1800/1900-MS* measurement menus. Compared to the standalone case, the latter option offers an extended functionality.

Standalone *audio* measurements are performed with default connector settings, the audio signals being applied to the connectors *AF IN* (input) and *AF OUT* (output) on the front panel of CMU.

If *Audio* is used in the context of *GSM400/850/900/1800/1900-MS* measurements, the *AF/RF* tab of the associated *Connect. Control* menu allows to select the input source of the CMU speech encoder and the output destination of its speech decoder.

Note:

In addition to the features reported in this section, option CMU-B41 offers some extended functionality that is accessible via remote control only:

- Secondary audio circuit; analogous to the primary circuit
- Variable band pass filter
- Frequency counter 10Hz ... 204.8kHz

For a description of the additional features refer to chapter 6 of CMU Operating Manual, Id.No. 1100.4903.12 .

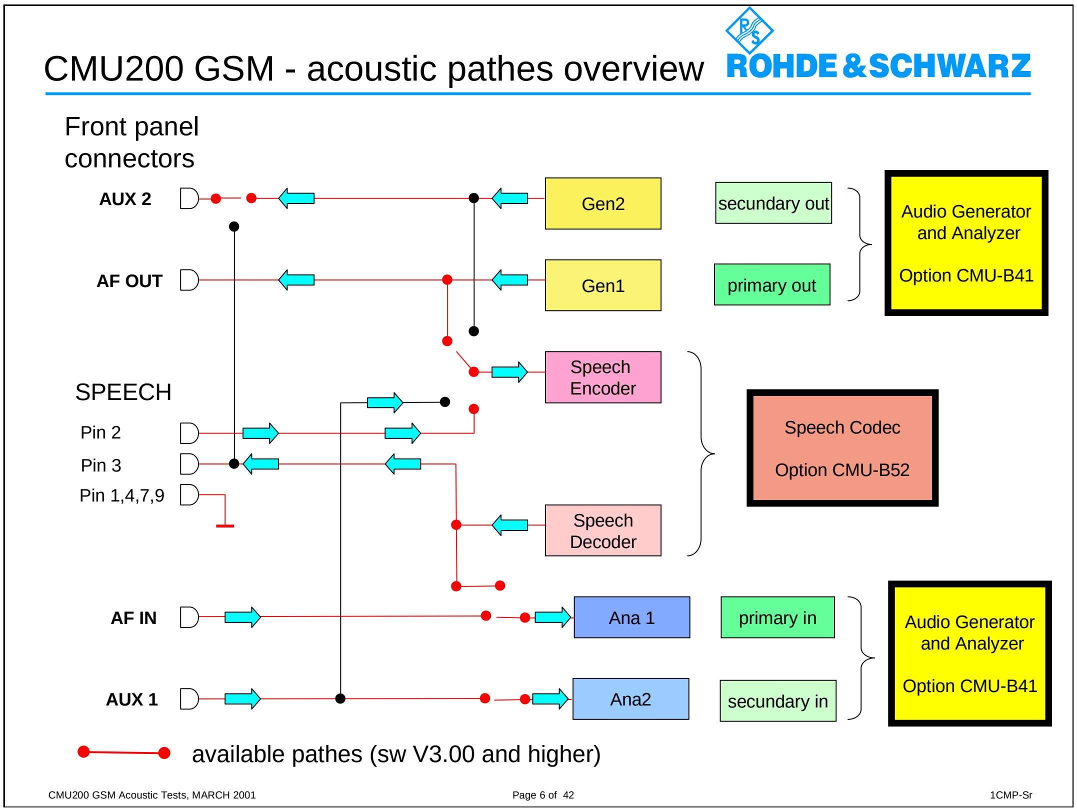
How to test the Speech Codec of a mobile phone

with

Speech Codec Option CMU-B52 and

with or without

Audio Generator and Analyzer Option CMU-B41.



The am. schematic shows(red marked) the currently available acoustic pathes if the options Audio Generator and Analyzer CMU-B41 and Speech Codec CMU-B52 are installed.

The *Audio* measurement is divided into the two subsystems for AF Generator and AF Analyzer control.

In **remote control**, *two* independent circuits are provided:-

- In the primary audio circuit, the audio signals are applied to the connectors AF OUT (output, AF generator signal) and AF IN (input) on the CMU front panel. The primary audio circuit corresponds to the *Audio Analyzer/Generator* menu and the associated configuration menu.
- In the secondary audio circuit, the audio signals are applied to the connectors AUX 2 (output, AF generator signal) and AUX 1 (input) on the CMU front panel.

The secondary audio circuit can not be controlled manually.

With the exception of the input and output connectors, the two audio circuits are identical. All remote control commands are analogous.

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(1) Acoustic measurements on GSM mobiles

with

Speech Codec Option CMU-B52

and with

Audio Generator and Analyzer Option CMU-B41

This application is used to perform a subjective quality check of the mice and earphone of a mobile phone.

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

In commercial mobiles measurements during normal operation can only be performed via the air-interface with the voice encoder and decoder included.

A so-called vocoder is used to attain the lowest possible data rate, only the filter and fundamental parameters required for signal reconstruction are transmitted, not the actual voice.

The audio generator of option CMU-B41 uses sinwave tones that cover some restrictions on the results measured:

Measurements using sin tones cannot be performed because the static sinwave input signal becomes a more or less stochastic output signal as a result of coding, particularly in the medium and high audio frequency ranges. If, for instance, a tone of approx 2.5 .... 2.7kHz is applied to the mobile phone with a constant sound pressure, the amplitude of the signal obtained at the decoder output varies by approx 20dB which makes the signal unsiutable for measurements.

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

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Acoustic tests with Speech Codec option CMU-B52

Where do I find the parameter settings?	Menu Location ?
What do these parameters mean?	Functions ?
When are the parameters available?	Signalling Status ?

Various so-called *Bit Stream* parameters are involving the CMU speech codec, option *CMU-B52*.

The various parameter settings are supported in the corresponding network, eg. *GSM400 Software CMU-K20*.

A good guidline is the Operating Manual for *GSM400/850/900/1800/1900 Software* (Id.No. 1115.6088.12), chapter 4, part *Signalling: Connection Control*.

CMU GSM - Acoustic Tests

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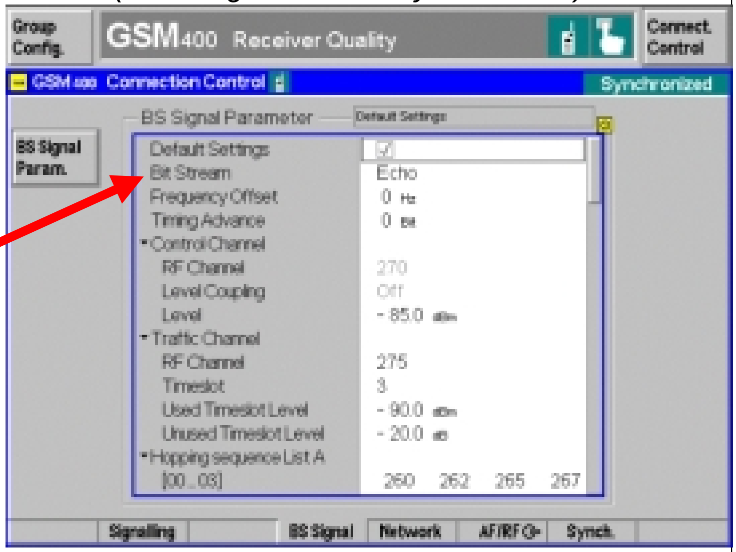
Acoustic tests with Speech Codec option CMU-B52

Where do I find the parameter settings?

see (1) *BS Signal Parameters (State Signal Off/On, Synchronized)*



The *BS Signal* tab configures the signals of the CMU. This includes the selection of the transmit data, so-called *Bit Stream.*

The *Bit Stream* determines the data transmitted on the traffic channel and the signal path.

CMU GSM - Acoustic Tests

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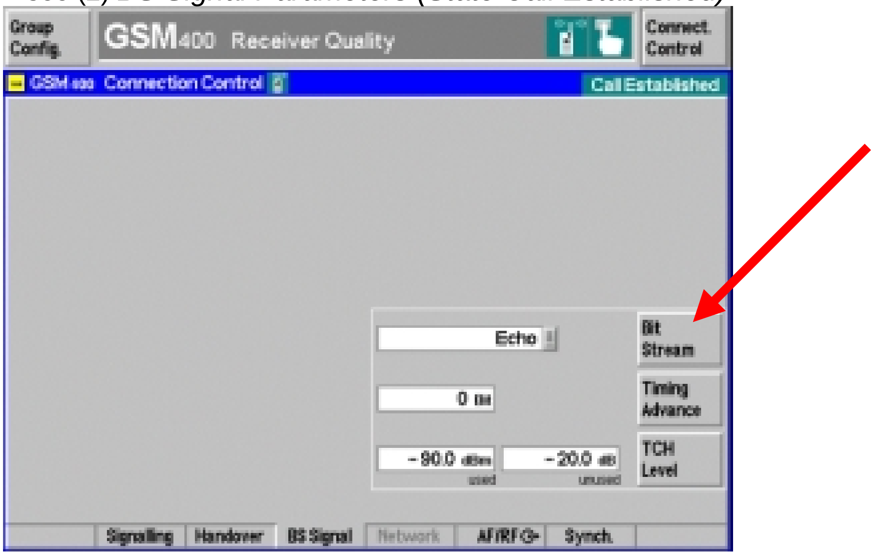
Acoustic tests with option CMU-B52

Where do I find the parameter settings?

see (2) *BS Signal Parameters (State Call Established)*



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The *BS Signal* tab configures the signals of the CMU. This includes the selection of the transmit data, so-called *Bit Stream.*

The *Bit Stream* determines the data transmitted on the traffic channel and the signal path.

For acoustic tests you can choose between the following mode settings:

- *Handset*
- *Handset Low*
- *Decoder Cal*
- *Encoder Cal*
- *Codec Cal*

All these tests are performed with an Audio Analyzer.

CMU GSM - Acoustic Tests

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Acoustic tests with option CMU-B52

What **do these parameters mean?**

Handset

The CMU sends and receives speech frames

that are routed to the internal speech codec on option board CMU-B52.

The *Bit Stream* determines the data transmitted on the traffic channel and the signal path.

Analog signals are provided via the front panel connector *SPEECH*.

The analog input signal at connector *SPEECH* is **amplified by 22.5 dB**.

The pin assignment of SPEECH connector is shown on a later page.

CMU GSM - Acoustic Tests

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Acoustic tests with option CMU-B52

What **do these parameters mean?**

Handset Low

The CMU sends and receives speech frames

that are routed to the internal speech codec on option board CMU-B52.

The *Bit Stream* determines the data transmitted on the traffic channel and the signal path.

Analog signals are provided via the front panel connector *SPEECH*.

The analog input signal at connector *SPEECH* is **not amplified**.

The pin assignment of SPEECH connector is shown on a later page.

CMU GSM - Acoustic Tests

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Acoustic tests with option CMU-B52

What **do these parameters mean?**

Decoder Cal

The speech codec option CMU-B52 provides a
digital 1 kHz fullscale sinewave signal
via the D/A converter to its analog output (*SPEECH* connector pin 3).

The *Bit Stream* determines the data transmitted on the traffic channel and the signal path.

This signal is used for external calibration of the analog output path.

For calibrating the codec of option board CMU-B52 the mobile station under test is in the signaling mode call established.

With the decoder calibration you can determine the full scale output level ref to 3.14dBm0.

In the next calibration step the encoder of the option board CMU-B52 has to be determined.

CMU GSM - Acoustic Tests

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Acoustic tests with option CMU-B52

What do these parameters mean?

Encoder Cal

The speech codec option CMU-B52 loops the input signal after A/D and D/A conversion directly to the analog output (*SPEECH* connector pin 3).

The *Bit Stream* determines the data transmitted on the traffic channel and the signal path.

This signal is used for external calibration of the analog input path.

For calibrating the codec of option board CMU-B52 the mobile station under test is in the signaling mode call established.

With a defined input signal (to be inserted on pin 2 of SPEECH connector) the sensitivity of the internal A/D converter can be measured and referred to the measured output level in setting *Decoder Cal*.

CMU GSM - Acoustic Tests

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Acoustic tests with option CMU-B52

What do these parameters mean?

Codec Cal

The CMU sends and receives speech frames that are routed to the internal speech codec option CMU-B52.

The CMU sends a close loop message to the mobile station to activate an internal test loop.

The *Bit Stream* determines the data transmitted on the traffic channel and the signal path.

Analog signals are provided via the *SPEECH* connector at the front panel.

The analog input signal at connector *SPEECH* pin 2 is **not** amplified.

With the *Codec Cal* test path setting you now can perform the acoustic tests of the mobile station under test.

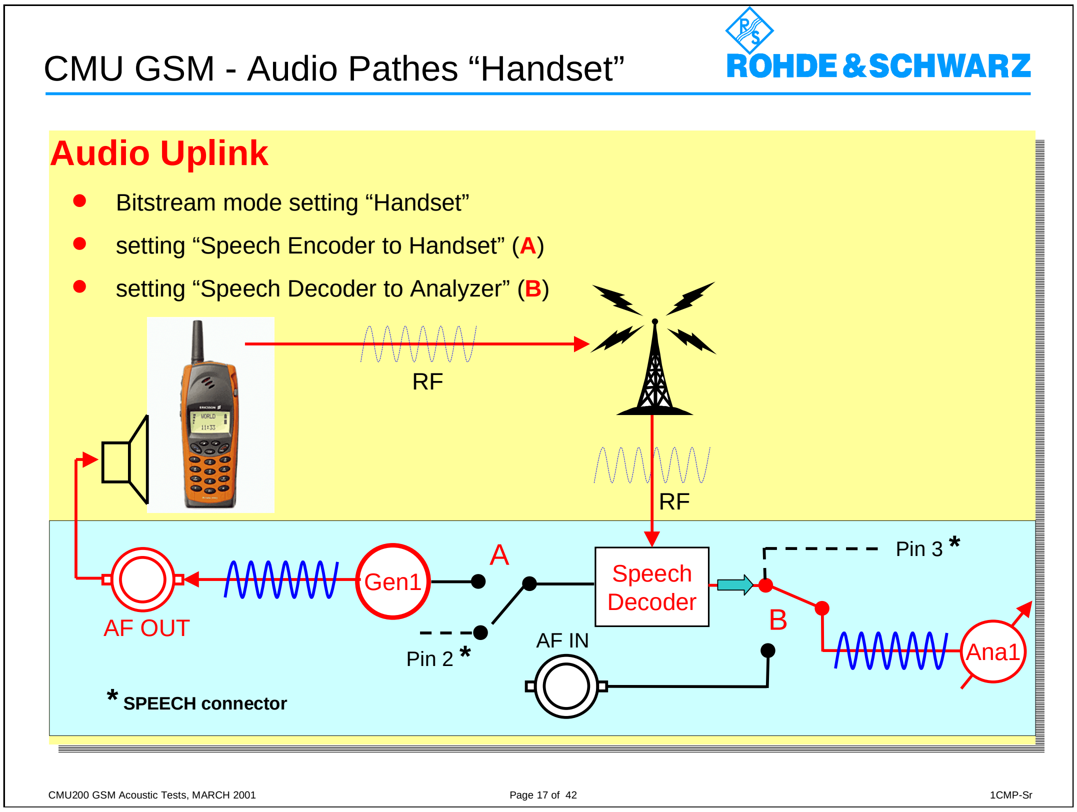
The test command is transmitted to the mobile station, then the mobile loops back the received speech frames.
This test is performed according to the so-called CLOSE_TCH_LOOP_CMD of GSM spec. 04.14 chapter 8, resp. 3GPP spec. TS 44.014 chapter 8.

With a defined input signal (to be inserted on pin 2 of SPEECH connector) the overall loop gain of the Coder - Decoder - Loop (for setting *Handset Low*) can be determined.

The mostly used setting

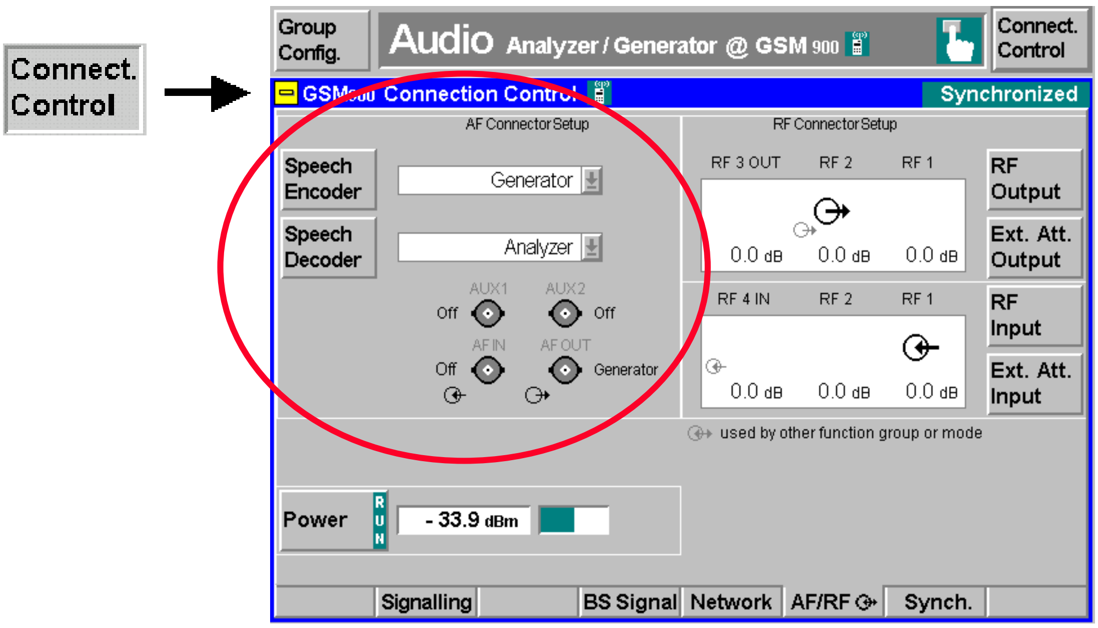
Handset mode

in detail



The *AF/RF* tab (function group *GSM400/850/900/1800/1900-MS*, *Non Signalling* or *Signalling* mode) configures for instance the connectors for AF signals.

This includes the setting of the input source of the CMU speech encoder and the output destination of its speech decoder.



If the *Audio Generator and Analyzer* (option CMU-B41) is not fitted, the speech codec (option CMU-B52) is connected to the 9-pole *SPEECH* (handset) connector on the CMU front panel, see chapter 8 of the CMU200 Operating Manual (Id.No. 1100.4903.12) .

CMU GSM - Audio Pathes “Handset”

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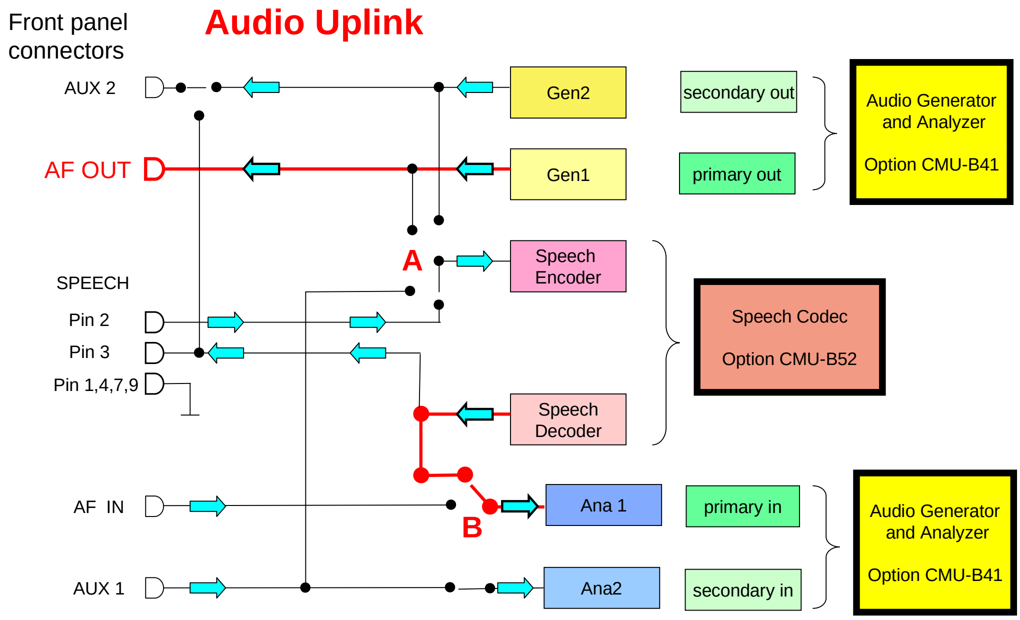
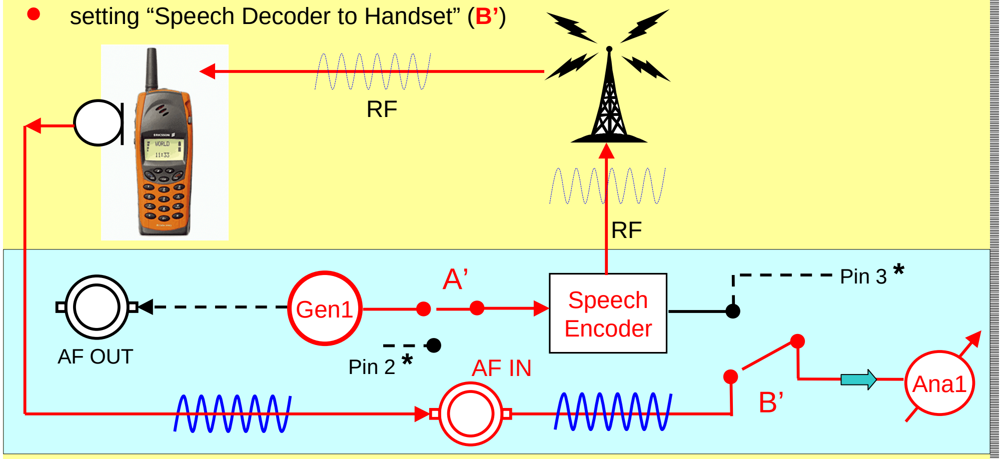


Audio Downlink

- Bitstream mode setting "Handset"
- setting "Speech Encoder to Generator" (**A'**)
- setting "Speech Decoder to Handset" (**B'**)



* **SPEECH connector**

CMU GSM - Audio Pathes “Handset”

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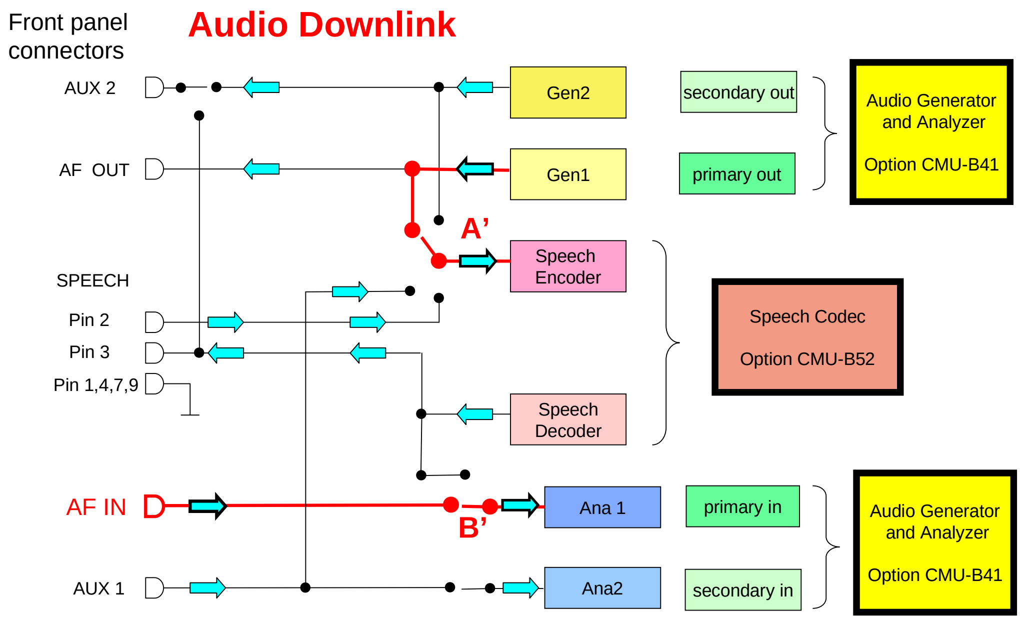

AF generator

Output impedance	<4 Ω
Maximum output current	20 mA
AF sine generator	
Frequency range	20 Hz to 20 kHz
Frequency uncertainty	same as time base + half resolution
Frequency resolution	0.1 Hz
Output level range	10 µV to 5 V
Output level resolution	
at level <10 mV	10 µV
at level ≥10 mV	0.1%
Output level uncertainty	
at level ≥1 mV and frequency ≤10 kHz	≤1.5% + resolution
THD+N[7]	
at level ≥100 mV into load ≥600 Ω	≤0.05%
THD[7]	
at level ≥100 mV into load ≥600 Ω	≤0.025%

[7] Measurement bandwidth: 21.9 kHz

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AF analyzer

Input impedance	1MΩ || 100 pF
AF voltmeter	
Frequency range	50 Hz to 20 kHz
Level Range	50 µV to 30 V
Level Resolution	
at level <1 mV	1 µV
at level ≥1 mV	0.1%
Level uncertainty	
at 1 mV ≤level≤ 2 V	<1.0% + resolution
at 2 V <level≤20 V	<2.0% + resolution
THD+N meter	
Measurement bandwidth	21 kHz
Frequency range	100 Hz to 10 kHz
Level Range	10 mV to 30 V
Resolution	0.01% THD+N
Inherent distortion	
at 100 mV ≤ level≤ 20 V	<0.05% THD+N
Uncertainty	
at 100 mV ≤ level≤ 2 V	<1% + inherent distortion
at 2 V <level≤20 V	<2% + inherent distortion

Subsystem AFANalyzer...:FILTer (Filter)

The subsystem `AFANalyzer:...FILTer` contains the commands for the configuration of the audio analysis filter. The input path of the AF analyzer is as shown below:

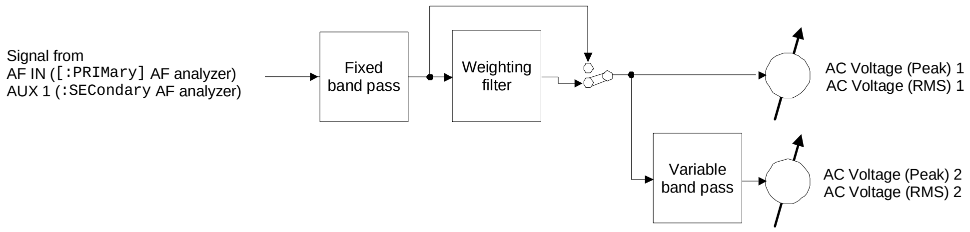

Fig. 6-1 AF analyzer input path configuration

Note: This subsystem has no equivalent in manual control.

Weighting Filter commands:

CONFigure:AFANalyzer[:PRIMary]:FILTer:WEIGhting <*Weighting*>
CONFigure:AFANalyzer:SECondary:FILTer:WEIGhting <*Weighting*>>

<*Weighting*>	Description of parameters
CME |	Switch on C-message weighted filter
CCI |	Switch on CCITT weighting filter
OFF	No weighting filter

These commands select the weighting filter after the fixed band pass.

AF Analyzer filter config - IEEE bus

CONFigure:AFANalyzer[:PRIMary]:FILTer:VBPass:CFRequency <Center>
CONFigure:AFANalyzer:SECondary:FILTer:VBPass:CFRequency <Center> — Frequency

<Frequency>	Description of parameters	Def. value	Def. unit	Unit ring
20 Hz to 20000 Hz	Center frequency of band pass	1000	Hz	

Description of command	FW vers.
This command determines the center frequency of the variable band pass.	≥ V2.12

CONFigure:AFANalyzer[:PRIMary]:FILTer:VBPass:BWIDth <Bandwidth>
CONFigure:AFANalyzer:SECondary:FILTer:VBPass:BWIDth <Bandwidth> — Bandwidth

<Frequency>	Description of parameters	Def. value	Def. unit	Unit ring
10 Hz to 1000 Hz	Bandwidth of band pass	200	Hz	

Description of command	FW vers.
This command determines the 3 dB bandwidth of the variable band pass.	≥ V2.12

CONFigure:AFANalyzer[:PRIMary]:FILTer:WEIGhting <Weighting>
CONFigure:AFANalyzer:SECondary:FILTer:WEIGhting <Weighting>> — Weighting Filter

<Weighting>	Description of parameters	Def. value	Def. unit	Unit ring
CME | **CCI |** **OFF**	Switch on C-message weighted filter Switch on CCITT weighting filter No weighting filter	OFF	-	Freq.

Description of command	FW vers.
This command selects the weighting filter after the fixed band pass (see Fig. 6-1).	≥ V2.12

AF Analyzer filter config - IEEE bus

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CONFigure:AFANalyzer[:PRIMary]:FILTer:BPASs <*Band pass*> **CONFigure:AFANalyzer:SECondary:FILTer:BPASs <*Band pass*> >**				Fixed Band Pass Selection
<Band pass>	Description of parameters	Def. value	Def. unit	Unit ring
	CMU band pass filter with a 3 dB bandwidth of	BP16	–	
BP01 |	0 Hz to 250 Hz			
BP02 |	6 Hz to 250 Hz			
BP03 |	50 Hz to 250 Hz			
BP04 |	0 Hz to 3000 Hz			
BP05 |	6 Hz to 3000 Hz			
BP06 |	50 Hz to 3000 Hz			
BP07 |	300 Hz to 3000 Hz			
BP08 |	0 Hz to 4000 Hz			
BP09 |	6 Hz to 4000 Hz			
BP10 |	50 Hz to 4000 Hz			
BP11 |	300 Hz to 4000 Hz			
BP12 |	0 Hz to 15000 Hz			
BP13 |	6 Hz to 15000 Hz			
BP14 |	50 Hz to 15000 Hz			
BP15 |	300 Hz to 15000 Hz			
BP16 |	0 Hz to 21000 Hz			
BP17 |	6 Hz to 21000 Hz			
BP18	50 Hz to 21000 Hz			
Description of command				FW vers.
This command selects the first band pass in the AF Analyzer.				≥ V2.12

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(2) Acoustic measurements on GSM mobiles

with

Speech Codec Option CMU-B52

but without

Audio Generator and Analyzer Option CMU-B41

This application is used in type approval tests where highly accurate measurements are required.

Audio measurements are performed in line with GSM 11.10, resp. TS 51.010-1 3GPP Release 4, on special test mobiles which are provided with Digital Audio Interface (DAI).

There is however great interest in testing mobiles without DAI.

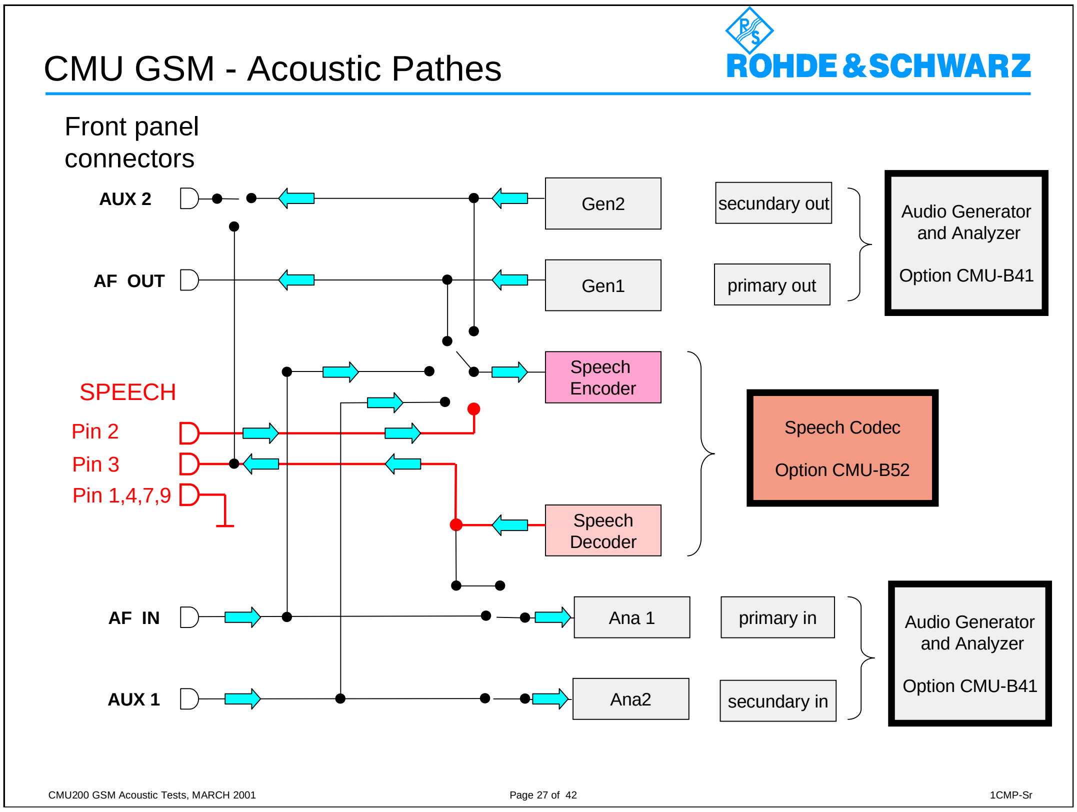
Trade journals, consumer test institutes or network operators are particularly interested in measuring and comparing acoustic characteristics of mobile phones. Network operators for instance must be able to check customer complaints or test the quality of supplied phones. A highly accurate test method is also required in the quality assurance of mobiles and for sampling inspection in production facilities.

The typical test setup is peformed in combination with an audio test system, like Audio Analyzer UPL06 and UPL16 with corresponding options and accessories.

These highly accurate acoustic measurements do **not** require the option

Audio Generator and Analyzer CMU-B41.

If the *Audio Generator and Analyzer* option CMU-B41 is not fitted, the speech codec option CMU-B52 is connected to the 9-pole *SPEECH* (handset) connector on the CMU front panel.



A typical test setup is peformed in combination with an audio test system, like Audio Analyzer UPL16 with corresponding options and accessories.

Such an Audio Analyzer is connected to the CMU200 at the frontpanel connector SPEECH at its corresponding contacts.

The GSM test mobile is driven by CMU200 via the air-interface, connected through the relevant RF frontpanel connector. CMU200 simulates a base station so that a call can be setup.

Two test pathes have to be considered.

In Uplink direction (sending direction):

acoustic input of mobile under test (mice) to decoder output (pin 3)

In Downlink direction (receiving direction):

encoder input (pin 2) to acoustic output of mobile under test (speaker)

Acoustic devices such as an artificial mouth, artificial ear and other accessories are required for these measurements.

CMU200 - SPEECH connector values

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Pin Assignment (see CMU Operating Manual chapter 8)

Pin 1,4,7,9	Ground	
Pin2	Handset In	for Signaling Unit CMU-B21
Pin 3	Handset Out	for Signaling Unit CMU-B21
Pin 5	Handset In	for 2nd Signaling Unit CMU-B21 or CDMA Signaling Unit for TDMA
Pin 6	Handset Out	for 2nd Signaling Unit CMU-B21 or CDMA Signaling Unit for TDMA
Pin 8	Power Supply	+5VDC, max 100mA

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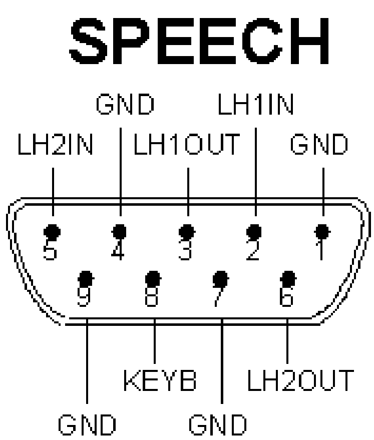
see chapter 8 of the CMU200 Operating Manual (Id.No. 1100.4903.12) :



Levels in mode setting „Handset“

The following RMS levels correspond to the 0dBm0 level.

Input level (Pin 2):

using an external generator	0.05 Vrms
using the internal generator (option CMU-B41)	0.5 Vrms

Output level (Pin 3):	0.5 Vrms

The given levels may vary approx 10% in both directions depending on the AD/DA converter used by the speech coder.

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Levels in mode setting „Handset“

Full scale levels

Full scale is defined as 3.14dBm0.
Therefore the maximum input and output levels are:-

Full scale input level (Pin 2):

using an external generator	0.072 Vrms
using the internal generator (option CMU-B41)	0.72 Vrms

Full scale output level (Pin 3):	0.72 Vrms

The given levels may vary approx 10% in both directions depending on the AD/DA converter used by the speech coder.

Impedances in mode setting „Handset“

Input Impedance (Pin 2)	100 kOhm
Output Impedance (Pin 3)	10 Ohm

Audio Delays

In Uplink direction (sending direction):

acoustic input of mobile under test to decoder output (pin 3) approx 125ms

In Downlink direction (receiving direction):

encoder input (pin 2) to acoustic output of mobile under test approx 145ms

each incl mobile under test connected

CMU200 - DAI Configuration

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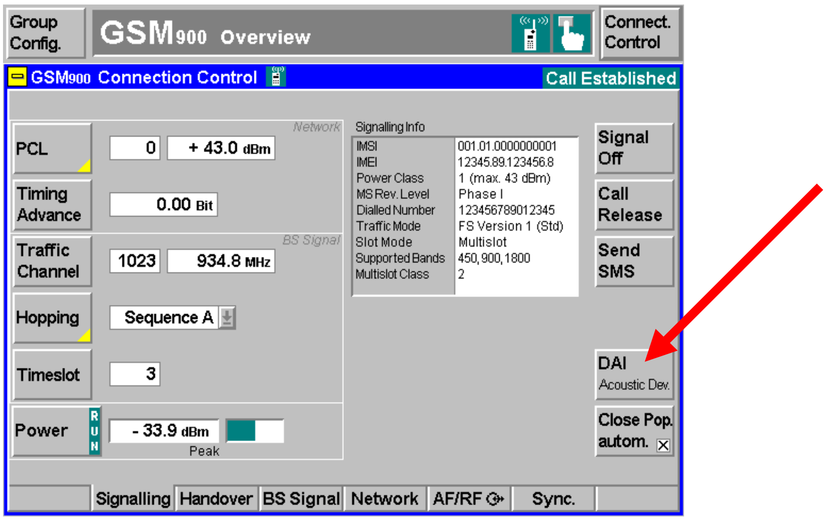
DAI = Digital Audio Interface

see Signaling Control with Call Established (state Call Established)



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In signaling mode Call Established you can configure the Digital Audio Interface (DAI) of the mobile station provided.

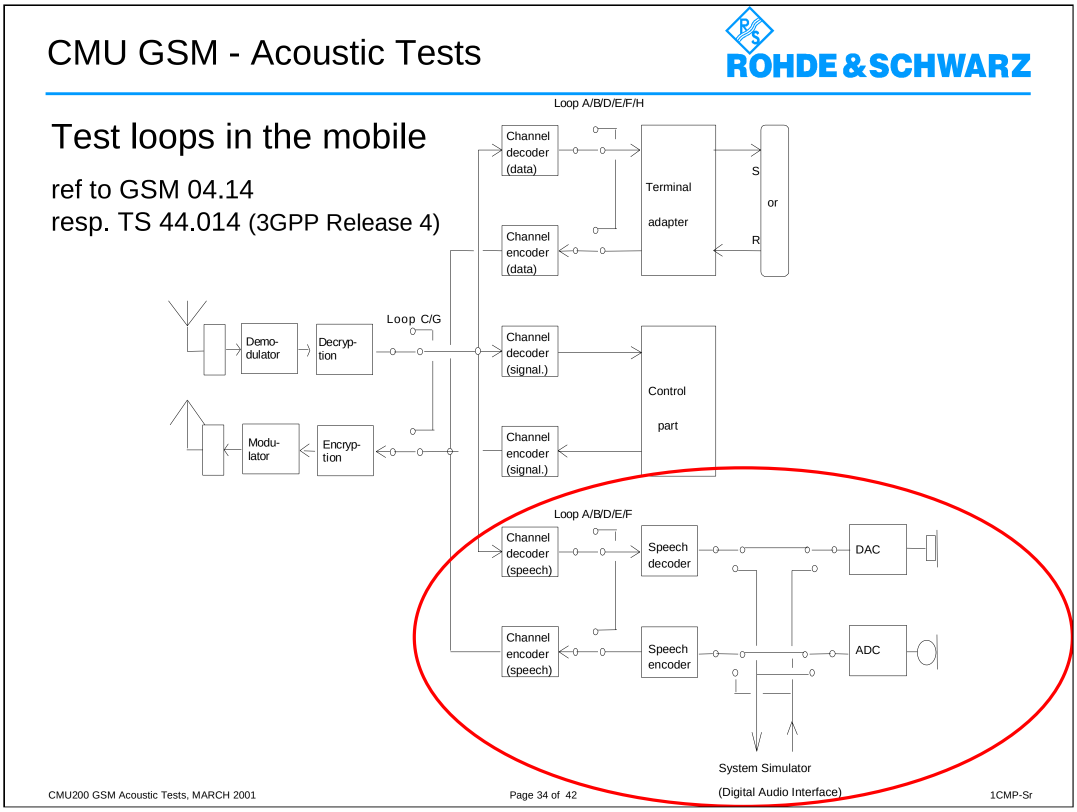
The *DAI Acoustic Dev.* determines the routing of the speech data (DAI of the mobile or internal, ie. normal mode) and
which device is being tested (speech codec/DTX functions or A/D and D/A) as follows:

The DAI can be set to one of the following modes:

Normal	Normal operation of the mobile(default setting during a call setup)
Decoder	Test of speech decoder / DTX functions (downlink)
Encoder	Test of speech encoder / DTX functions (uplink)
Acoustic Devices	Test of acoustic devices and A/D and D/A

When entering the *Call Established* state, the DAI setting is always *Normal*.

The other settings must be choosen explicitly after each call setup.



A number of internal test loops are required providing access to isolated functions of the MS without introducing new physical interfaces just for the reason of type approval testing.

The above figure shows a functional block diagram of a reference MS containing the different test loops.

NOTE:

It should be emphasized that these test loops only describe the functional behaviour of the MS with respect to its external interfaces; physical implementation of the loops is completely left open to the manufacturer.

A particular loop is activated in an MS by transmitting the appropriate command message to the MS.

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(3) Acoustic measurements at mobile phones

with

Audio Generator and Analyzer Option CMU-B41

and its

Multitone Functionality

This application is used to perform a subjective quality check of the mice and earphone of a mobile phone.

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

In commercial mobiles measurements during normal operation can only be performed via the air-interface with the voice encoder and decoder included.

A so-called vocoder is used to attain the lowest possible data rate, only the filter and fundamental parameters required for signal reconstruction are transmitted, not the actual voice.

The audio generator of option CMU-B41 uses sinwave tones that cover some restrictions on the results measured:

Measurements using sin tones cannot be performed because the static sinwave input signal becomes a more or less stochastic output signal as a result of coding, particularly in the medium and high audio frequency ranges. If, for instance, a tone of approx 2.5 .... 2.7kHz is applied to the mobile phone with a constant sound pressure, the amplitude of the signal obtained at the decoder output varies by approx 20dB which makes the signal unsiutable for measurements.

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

CMU200 - Multitone (option CMU-B41)

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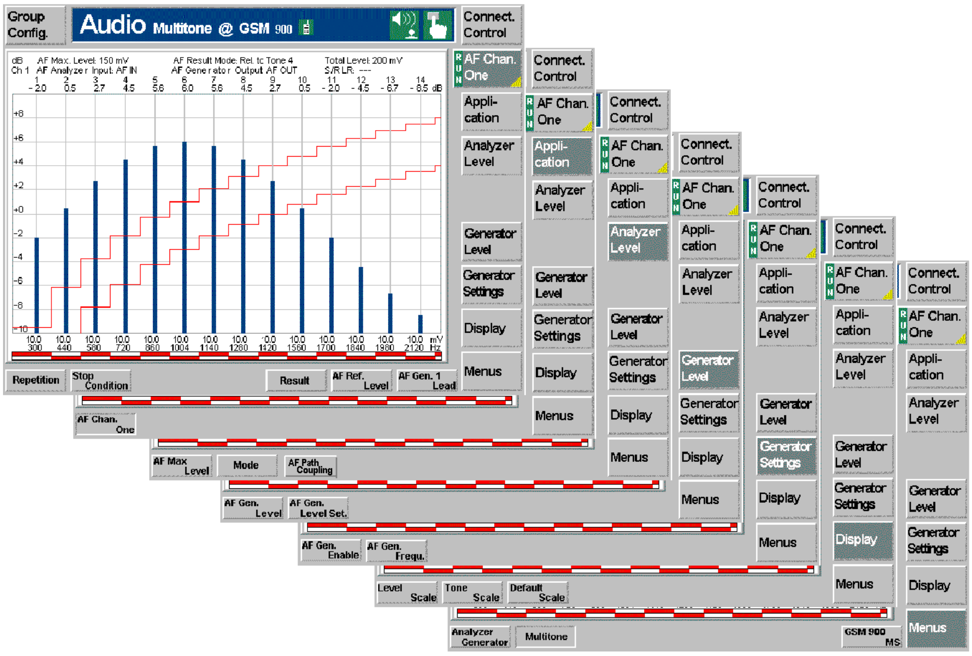

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The graphical measurement menu *Multitone* shows the results of the multitone audio measurement.

To perform an *Multitone* measurement, the CMU generates a composite audio signal that represents the superposition of up to 20 individual fixed-frequency tones with configurable frequency and level. An audio signal containing the same tones can be analyzed in a single measurement and displayed in a bar chart.

The *Multitone* measurement is thus a fast method to determine the level of up to 20 different tones at known frequencies and to perform a limit check for all results.

Possible applications are also frequency response and intermodulation measurements.

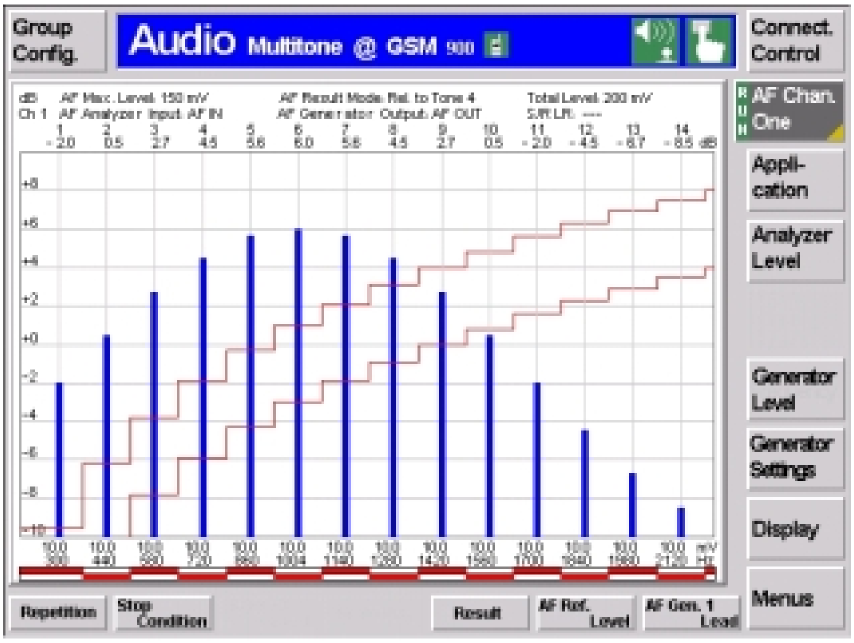
How wide are the setting ranges?

AF Frequency of test tone 10Hz to 15999Hz
AF Level of test tone 0.0V to 5.0V

Each test tone can be enabled(switched on) or disabled(switched off) individually.
Further setting possibilities are described in the manual.

The measurement menu *Multitone* is opened via the main menu *Menu Select* (with the associated key at the front of the instrument) or from the *Analyzer/Generator* menu using the *Multitone* hotkey.

CMU200 - Multitone (option CMU-B41)



The subsystem *Multitone* measures the level of an audio test signal comprising up to 20 test tones.
The subsystem corresponds to the measurement menu *Multitone* and the associated popup menu *Multitone Configuration.*

In analogy to the AF Generator and AF Analyzer subsystems explained before, the *Multitone* measurement provides two independent circuits:

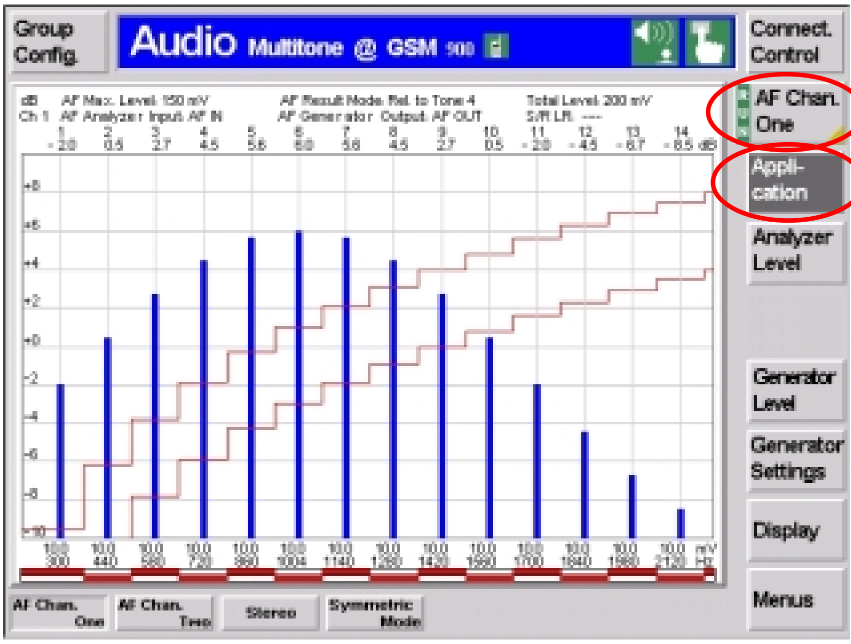
- In the first audio channel the audio signals are applied to the connectors AF OUT (output, AF generator signal) and AF IN (input) on the CMU front panel.

  The first audio channel corresponds to the *Multitone* menu and the associated configuration menu.
- In the second audio channel the audio signals are applied to the connectors AUX 2 (output, AF generator signal) and AUX 1 (input) on the CMU front panel.

  The secondary audio circuit can **not** be controlled manually.

With the exception of the input and output connectors, the two audio circuits are identical. All remote control commands are analogous.

CMU200 - Multitone (option CMU-B41)



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Application

The *Application* softkey activates one of the applications of the *Multitone* measurement. At present, only one application – *AF Chan. One* – is available.

AF Chan. One

The *AF Chan. One* hotkey selects the *Multitone* measurement on channel one. This means that the audio signals are applied to the connectors AF IN (CMU input) and AF OUT (CMU output) on the front panel. A second audio channel is available in remote control.

Remote control
Audio channel no. one is identified by the third-level keyword AF1Channel.
The second audio channel is identified by AF2Channel.

CMU200 - Multitone (option CMU-B41)

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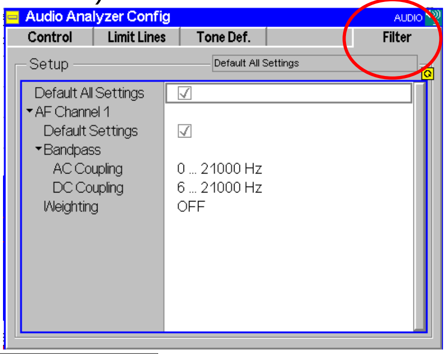
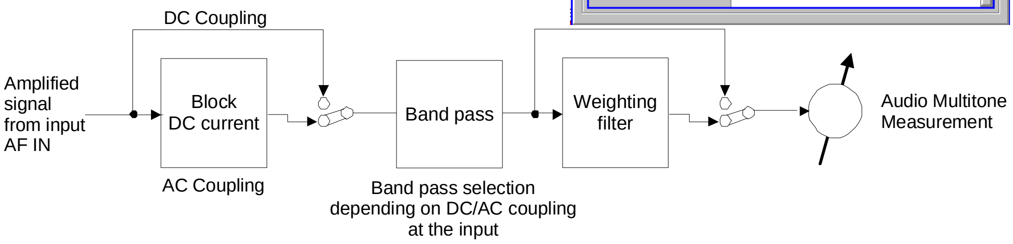
Path Configuration (TX Tests Configuration – Filter)

The *Filter* tab configures the receive path of the CMU for the *Multitone* measurement.





Signal path for Multitone measurements

The *Filter* tab configures the receive path of the CMU for the *Multitone* measurement.

The audio receive path of the CMU may contain the following filter stages:

AF Path Coupling Capacitor stage to block the DC component of the AF input signal including a possible DC offset of the input amplifier. With DC coupling, the complete AF input signal is measured.

Weighting Weighting filter according to CCITT or C-message weighted filter.

Band Pass Audio band pass filter with selectable bandwidth to limit the input frequencies to a definite audio band and eliminate unwanted signal components. The allowed bandwidth depends on the *AF Path Coupling*.

The audio results are generated at the end of the audio receive path, after the audio signal has passed all filter stages that are switched on.

(4) Summary

This application is used to perform a subjective quality check of the mice and earphone of a mobile phone.

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

In commercial mobiles measurements during normal operation can only be performed via the air-interface with the voice encoder and decoder included.

A so-called vocoder is used to attain the lowest possible data rate, only the filter and fundamental parameters required for signal reconstruction are transmitted, not the actual voice.

The audio generator of option CMU-B41 uses sinwave tones that cover some restrictions on the results measured:

Measurements using sin tones cannot be performed because the static sinwave input signal becomes a more or less stochastic output signal as a result of coding, particularly in the medium and high audio frequency ranges. If, for instance, a tone of approx 2.5 .... 2.7kHz is applied to the mobile phone with a constant sound pressure, the amplitude of the signal obtained at the decoder output varies by approx 20dB which makes the signal unsiutable for measurements.

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

CMU GSM - Acoustic Tests

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Products involved

Hardware

Audio Generator and Analyzer	CMU-B41
Speech Codec	CMU-B52
Signaling Unit	CMU-B21

Software (at least one of the following ones)

Software GSM400	CMU-K20
Software GSM850	CMU-K24
Software GSM900	CMU-K21
Software GSM1800	CMU-K22
Software GSM1900	CMU-K23

based on sw V3.00 and higher

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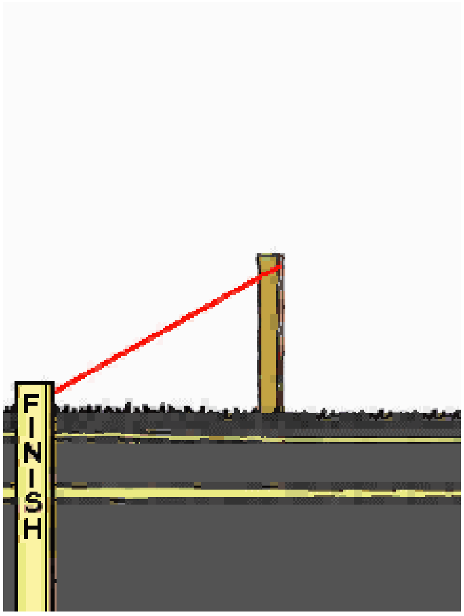
The option *Audio Generator and Analyzer CMU-B41* also includes the Multitone measurement feature.

The option *Speech Codec CMU-B52* contains Fullrate, Enhanced Fullrate and Halfrate codec.

ETSI documents, eg. GSM 04.08 resp. 3GPP TS 24.008, define „Fullrate Version 1“ as Fullrate speech codec and „Fullrate Version 2“ as Enhanced Fullrate speech codec.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