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Philips SySol_ME Training Session

Audio Calibration

Agenda

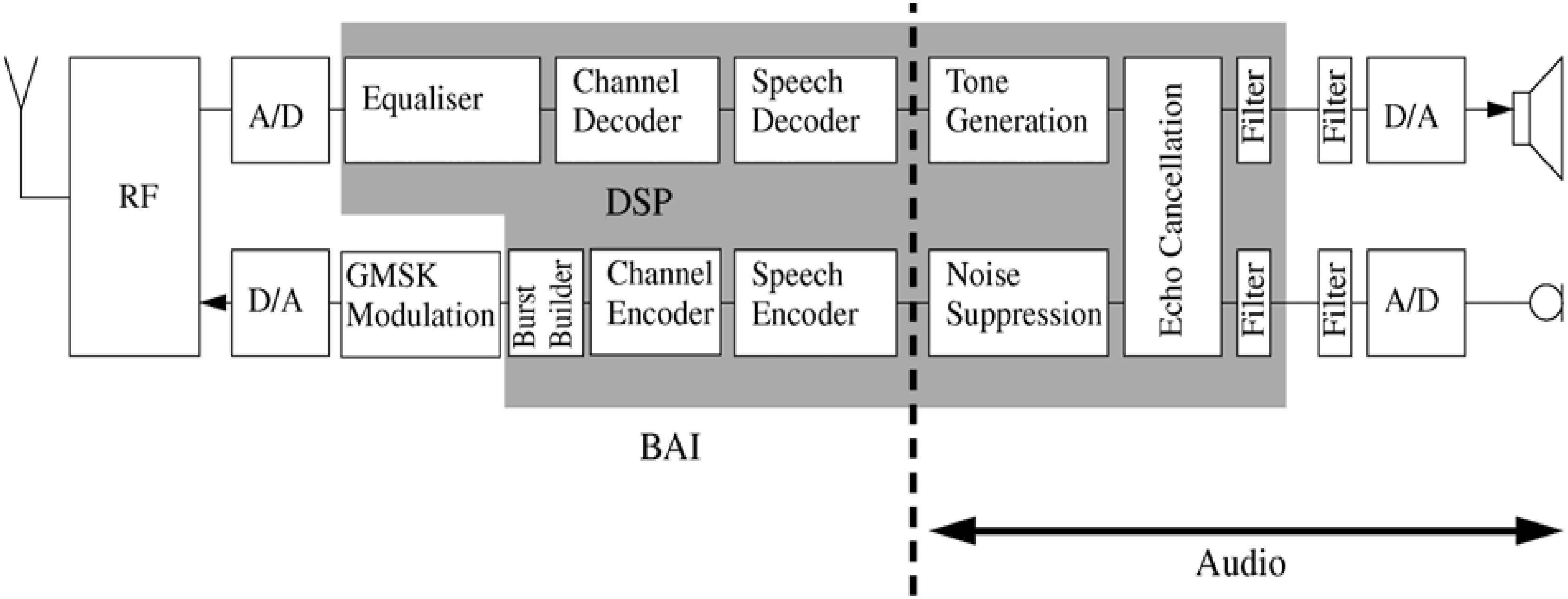
- Audio in GSM Mobile Phone
- Hardware Components in Audio Path
- HW Structure of PCF50732
- VSP in PCF50732
- Audio Firmware in R.E.A.L DSP
- Acoustic Test Bench
- Test Cases in FTA for Audio
- How to Tune the Audio
- TDMA Noise

Audio in GSM Mobile Phone

The importance of the Audio Quality in mobile phones is often underestimated…

…but the audio quality of the phone is the first impression which the final customer has!

Audio in GSM Mobile Phone



- only audio path is covered from this presentation
- sending direction from MIC to speech encoder
- receiving direction from speech decoder to speaker

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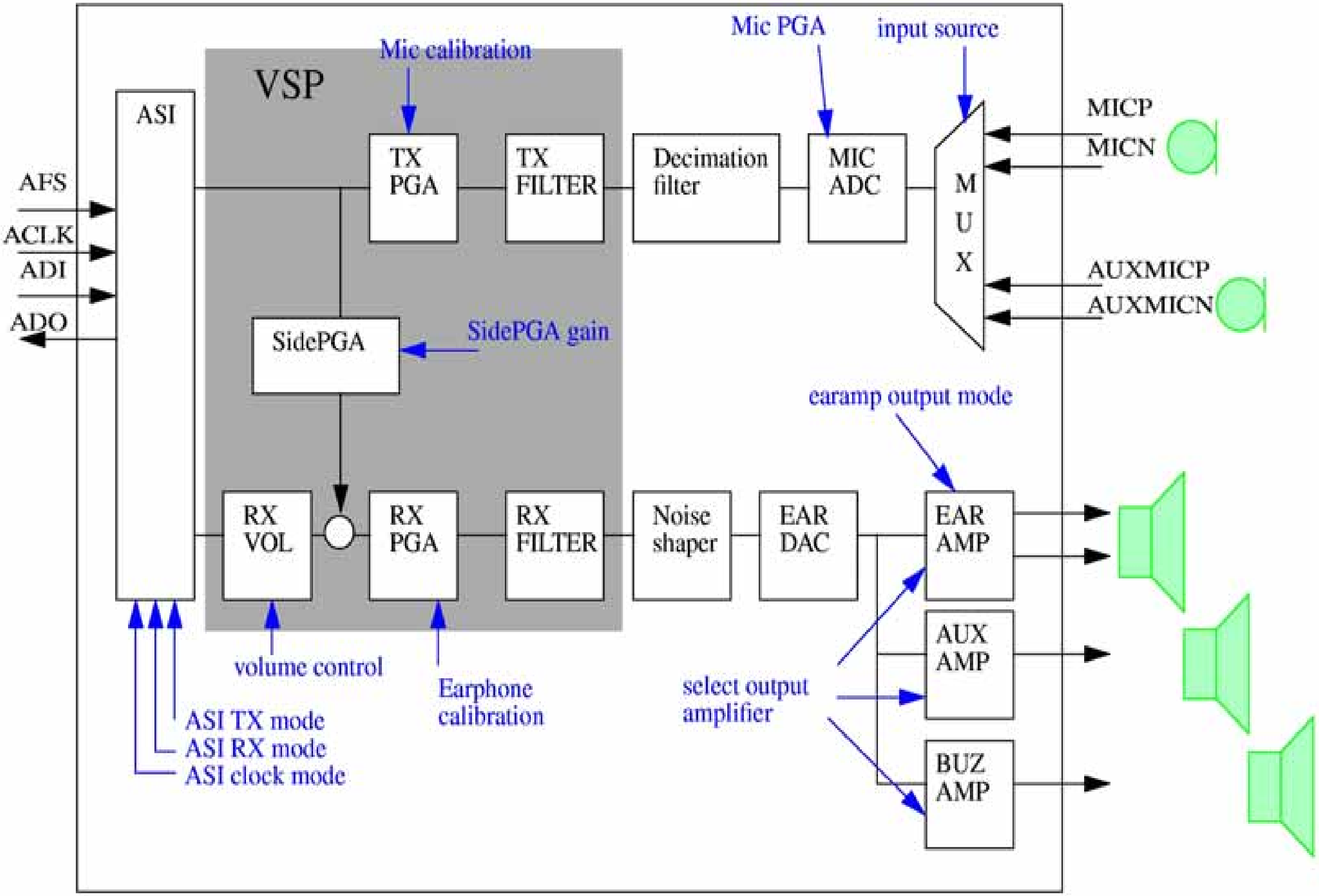
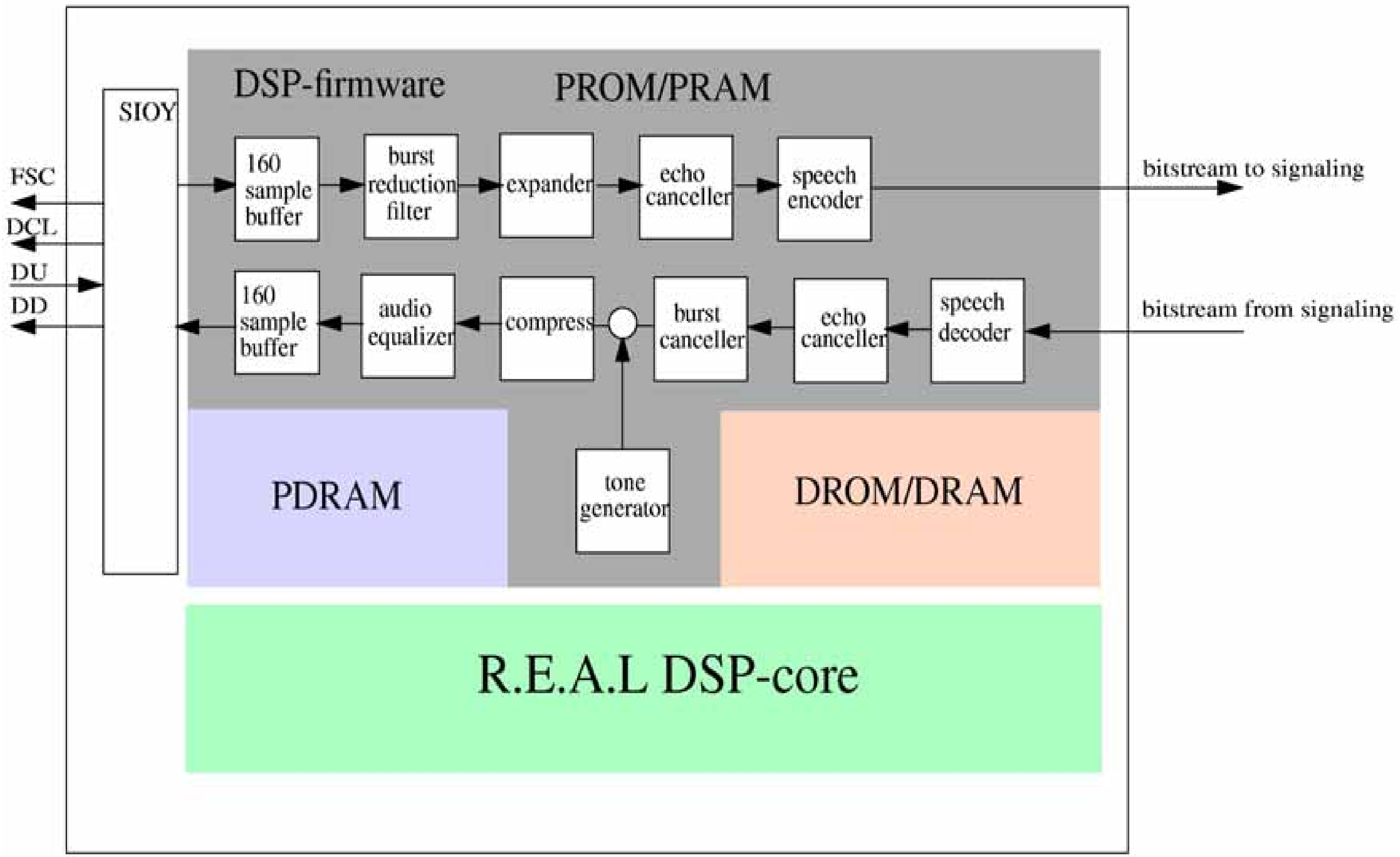
Hardware Components in Audio Path

Baseband Audio Interface : PCF50732



Hardware Components in Audio Path

Baseband Engine R.E.A.L DSP : PCF5087x



Hardware Components in Audio Path

System Solutions with their Baseband Engines&DSP Firmware masks

System Solution	Baseband Engine	DSP Firmware Mask
Sysol1	PCF50872/877	7V1
Sysol2GSM	PCF50877/874	AV3
Sysol2GPRS	PCF50874	CV1/CV3
Sysol3	PCF50874/874-5	CV3
SysolME	PCF50874-6	CV5

Hardware Components in Audio Path

Availability of the different blocks in different DSP Firmware versions

Feature	CV5-mask Multitask	CV3-mask Multitask	CV1-mask Multitask	AV3-mask Multitask	7V1-mask Singletask
Sample buffers	√	√	√	√	√
Burst reduction filter	√	√	√	√	√
Expander (dynamic noise suppressor)	√	√	X	X	X
Echo canceller	√	√	√	√	√
Speech encoder/decoder	FR,EFR,AMR	FR,EFR,AMR	FR,HR,EFR	FR,HR,EFR	FR,HR,EFR
Burst canceller(bad frame)	√	√	√	√	√
Compressor	√	√	√	√	√
Audio equalizer	√	√	√	√	√

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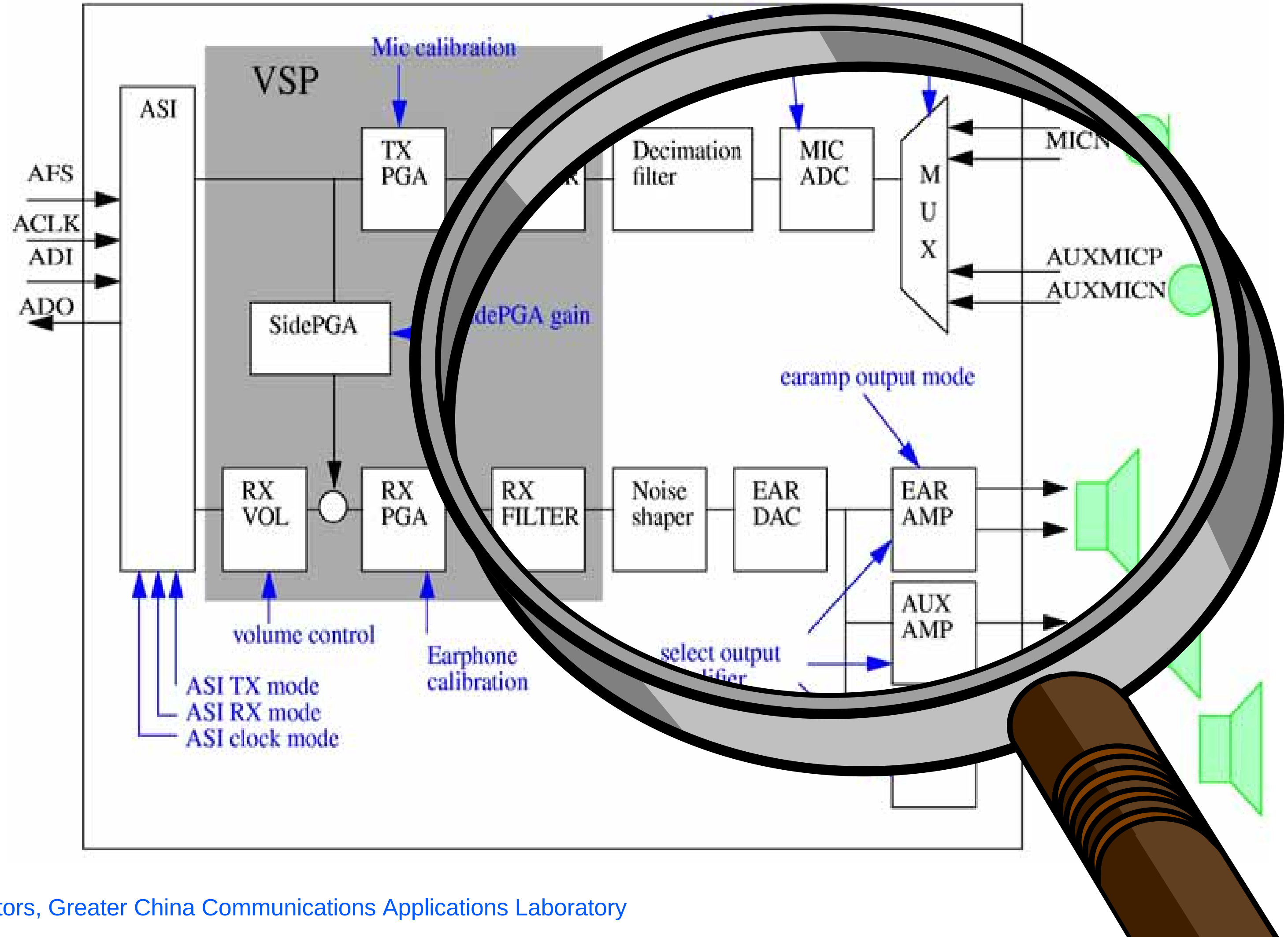
HW Structure of PCF50732

The Internal Circuits

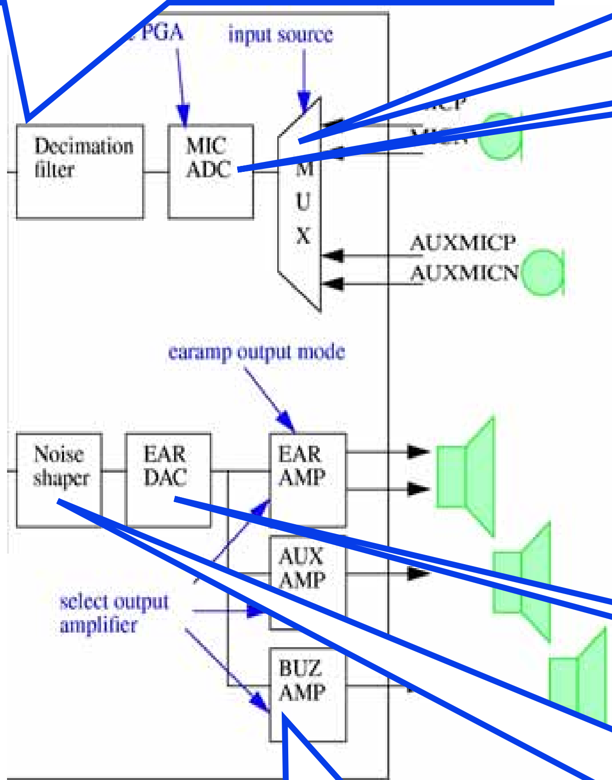


HW Structure of PCF50732

Decimates incoming bit stream from 1MHz to 40kHz/16bit

Multiplexer to select either the MIC- or the AUXMIC-input

Sigma-Delta-ADC with 1MHz bit stream output



Main features of the analog-IO-circuits :

- 2 selectable microphone inputs
- MICADC with 8kHz sampling rate and 16bit data width
- EARDAC with 8kHz sampling rate and 16bit data width
- 3 selectable output amplifiers with

  -1 W at differential EARAMP-ouputs

  -0.25 W at single-ended other outputs

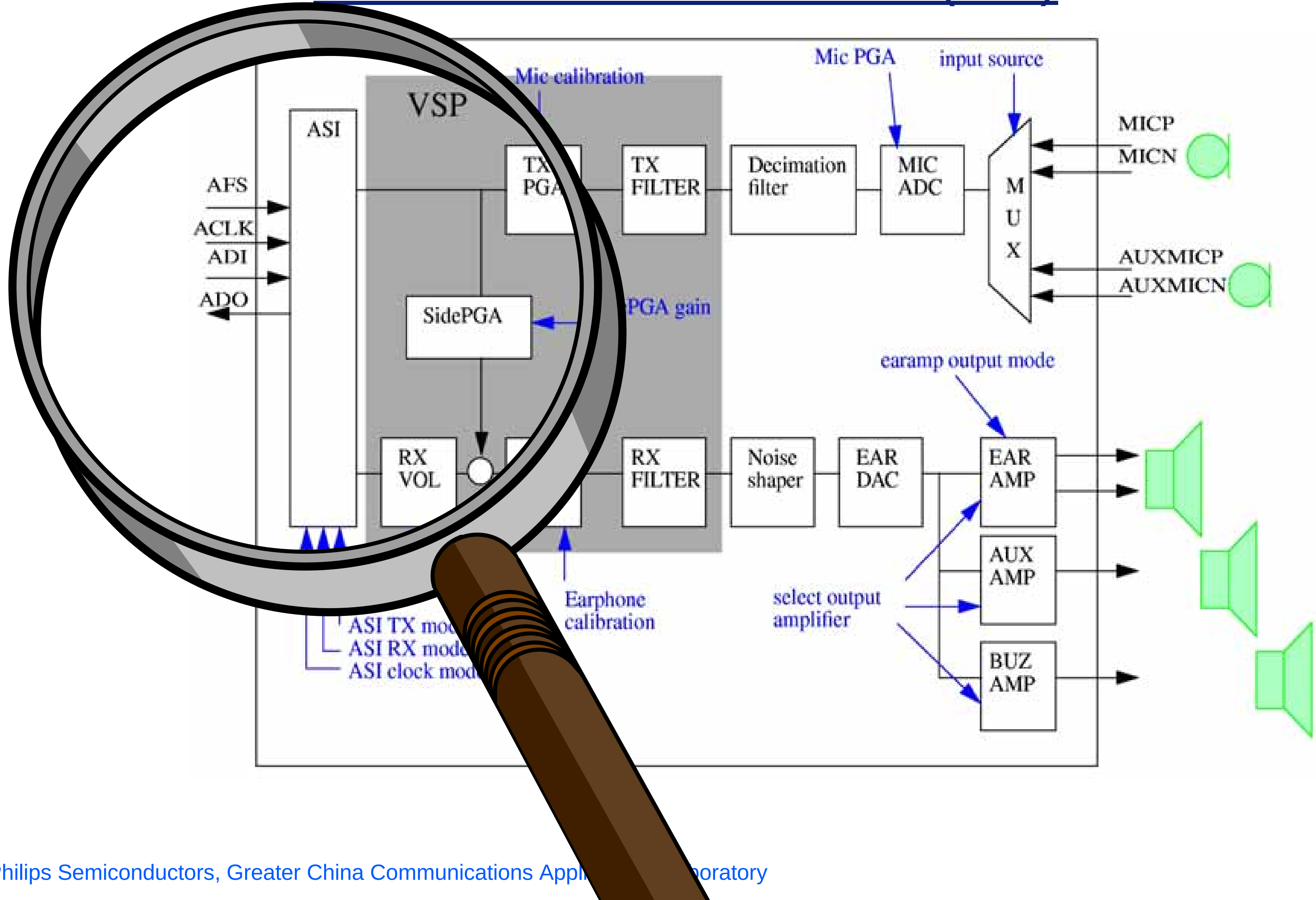
DAC including lowpass filter

1 of 3 16Ω driving amplifiers

2nd order digital Sigma-Delta-Modulator, generates 1MHz bit stream

HW Structure of PCF50732

The Audio Serial Interface (ASI)



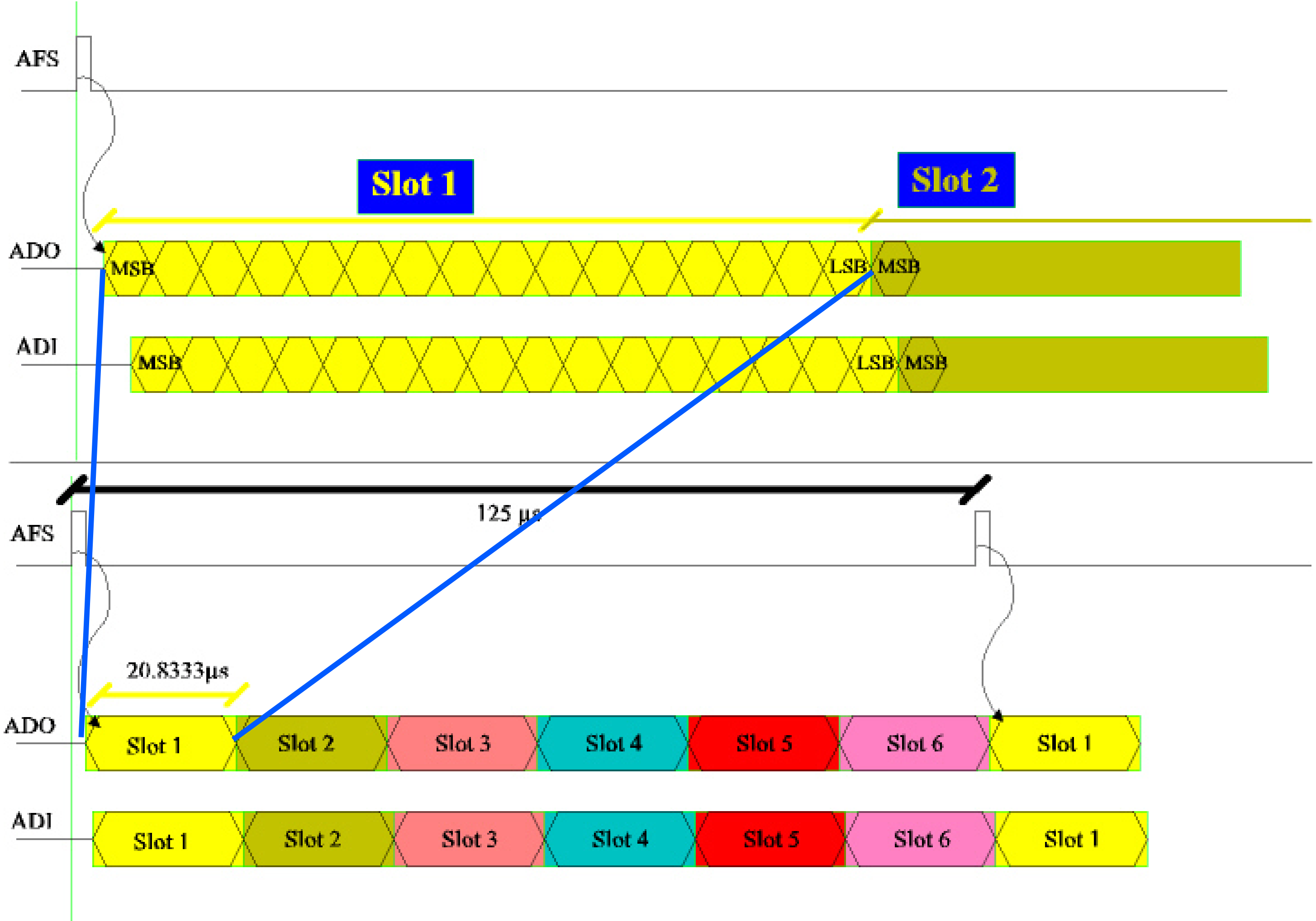
HW Structure of PCF50732

What is the ASI?

- The ASI is a digital, serial interface to transmit and receive digital audio data
- In Sysol it is parametrized to exchange data conform to IOM-2-standard
- The data were exchanged slot-by-slot. 6 slots are 1 frame. In a slot the data are 16bit wide
- In Sysol only slot 3 (1..6) is used in uplink- and downlink direction
- Multi device: 1 master (DSP), max. 16 slaves
- Adjustable 4 to 16 slots
- Single and double clock mode

HW Structure of PCF50732

ASI Frame Structure



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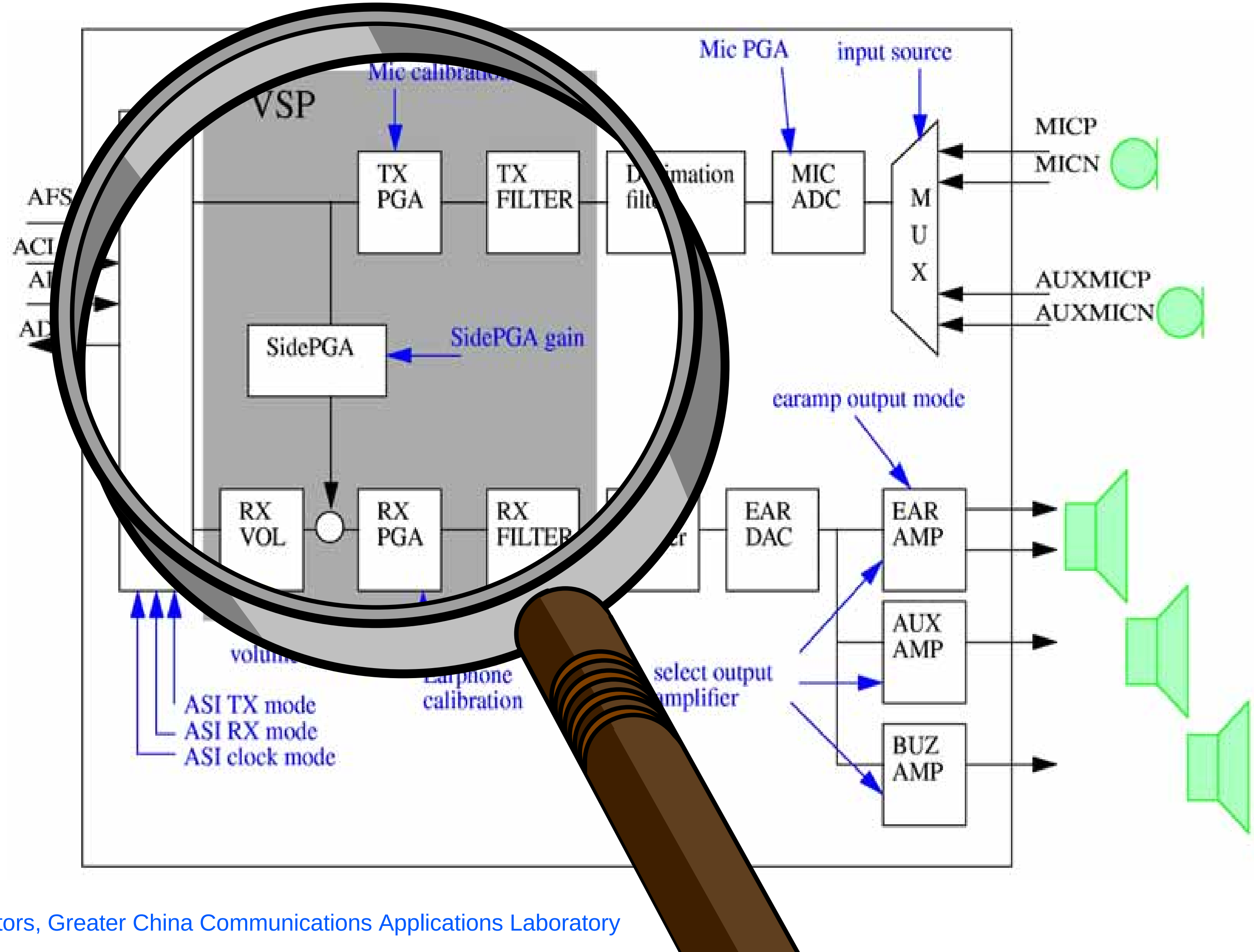
VSP in PCF50732

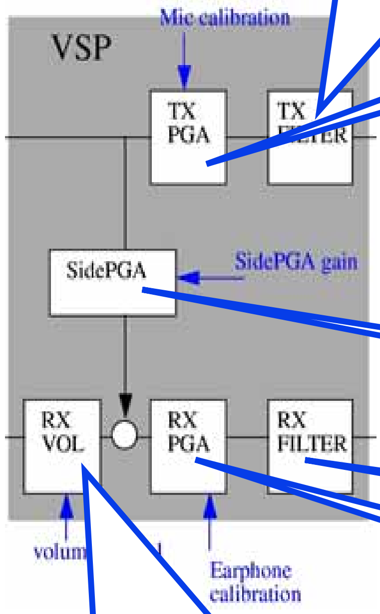
Voice Signal Processor(VSP)



VSP in PCF50732

Digital bandpass filter with passband from 300-3400 Hz

TxPGA controls the optimal matching of the external connected microphone to the signal processing units



VSP is an additional DSP only for processing audio-data

The VSP realizes :

- the shown audio data flow
- 4 digital gains (TxPGA, SidePGA, RxPGA, RxVol)
- digital filtering in Tx- and Rx-direction

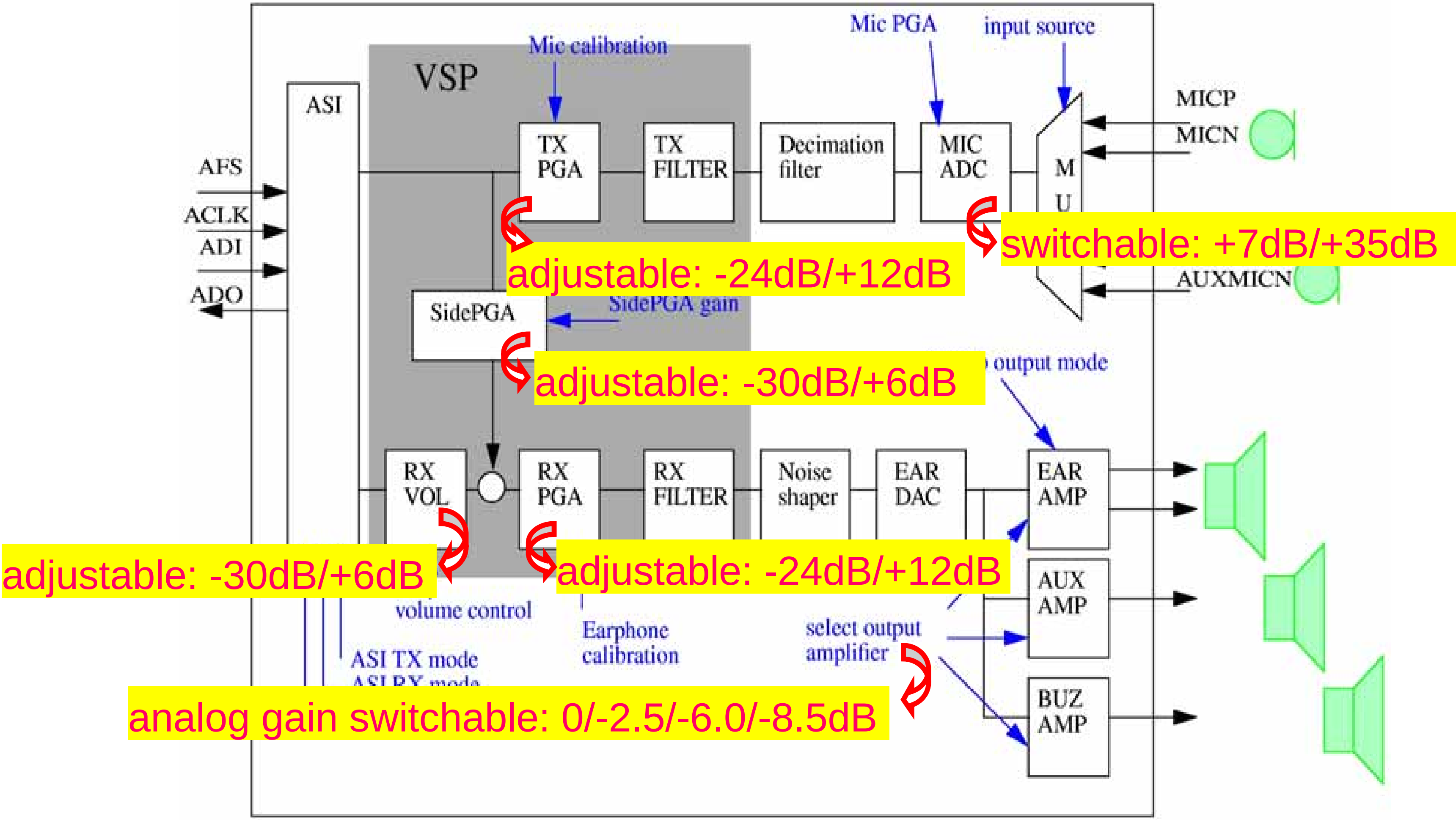
SidePGA loops a part of the microphone signal back to the speaker

Digital bandpass filter with passband from 300-3400 Hz

RxVol sets the volume of the speaker. One of up to 10 stages could be selected by the user via the MMI.

RxPGA controls the optimal matching of the external connected speaker to the signal processing units

VSP in PCF50732



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VSP in PCF50732

There are only 2 limited adjustable analog gains:

- MICHI +7dB/+35dB in TX direction
- Analog (Output) Gain 0dB/-2.5dB/-6.0dB/-8.5dB in RX direction

But there are 4 fine adjustable digital gains:

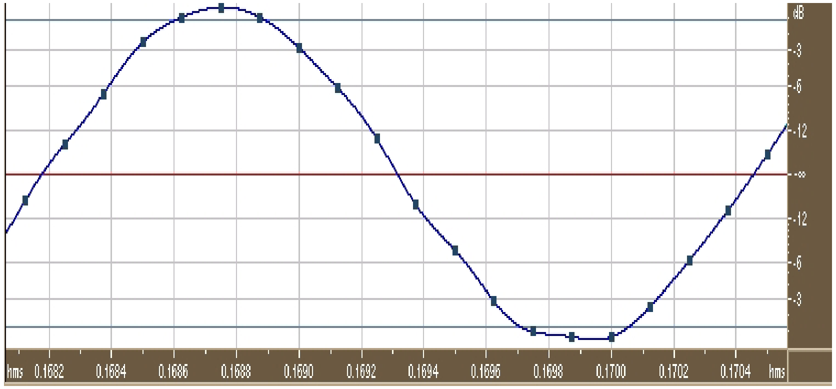
- TxPGA
- RxPGA
- SidePGA
- RxVol

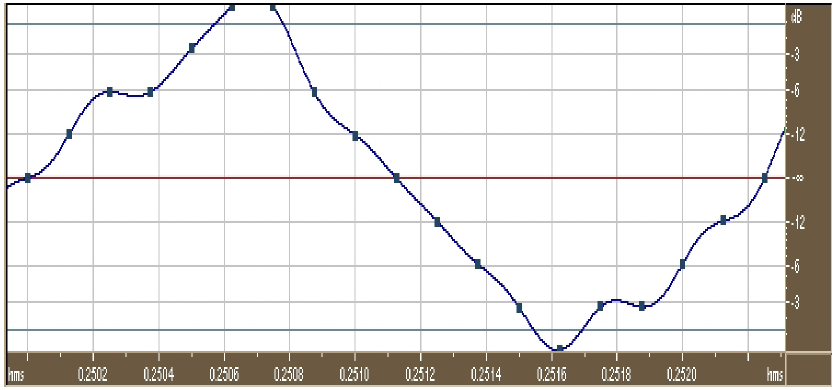
VSP in PCF50732

Please remark that digital gains could lead to resolution or distortion problems !

A sine signal with a level of -60dB was digitally amplified to max

A sine signal with a level of -80dB was digitally amplified to max





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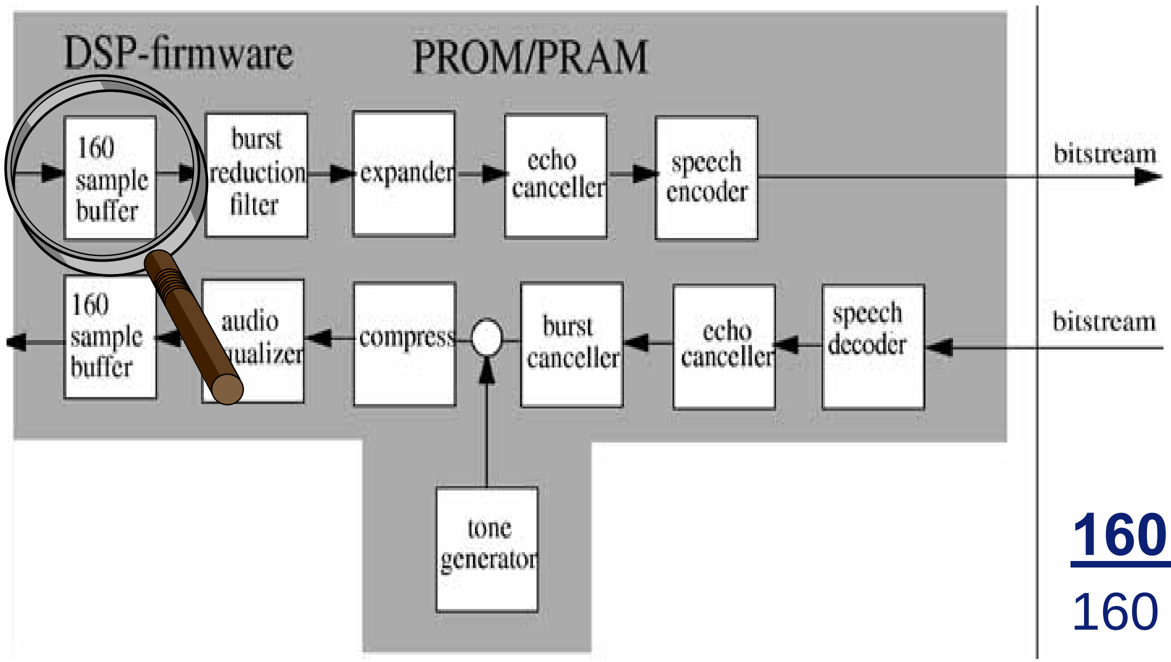
Audio Firmware in R.E.A.L DSP



160 sample buffer:

160 received audio samples (16bit, 8kHz) from BAI are stored into

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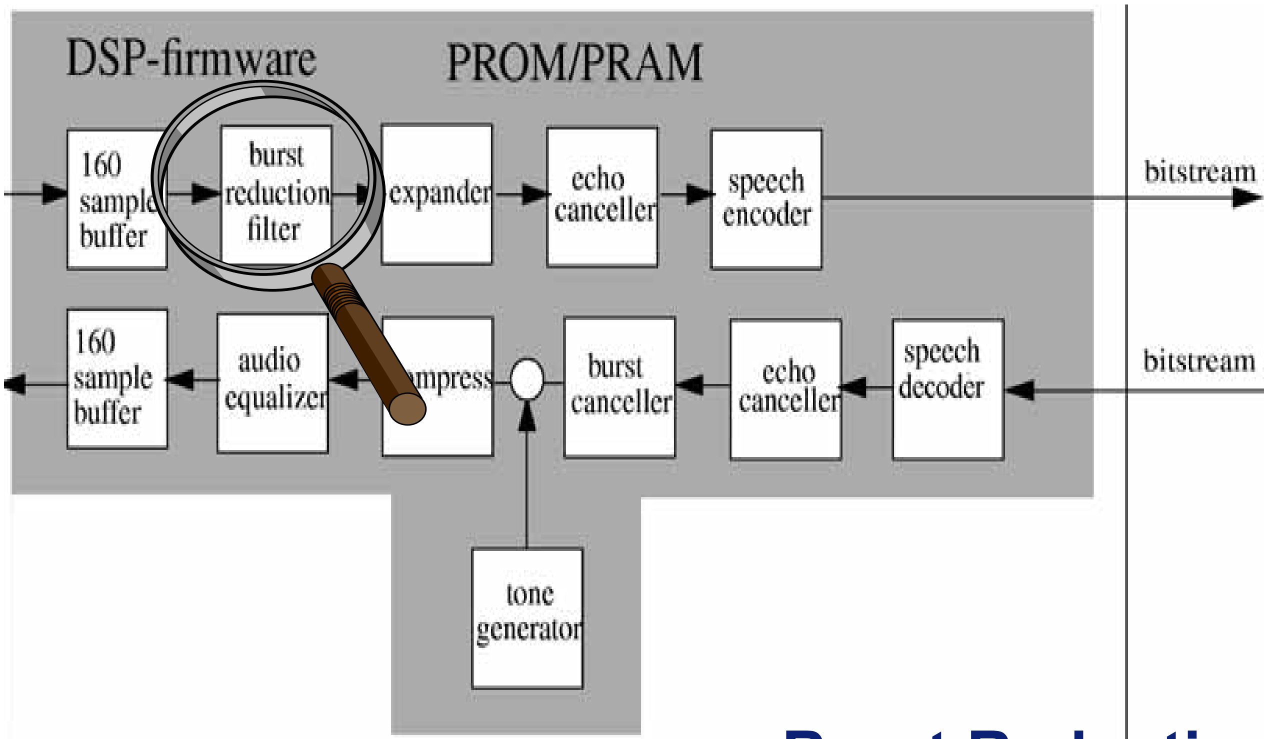
Audio Firmware in R.E.A.L DSP



Burst Reduction filter:

Bandstop filter which attenuates the GSM-burst-frequency of 216.7Hz(TDMA noise) and all harmonics.

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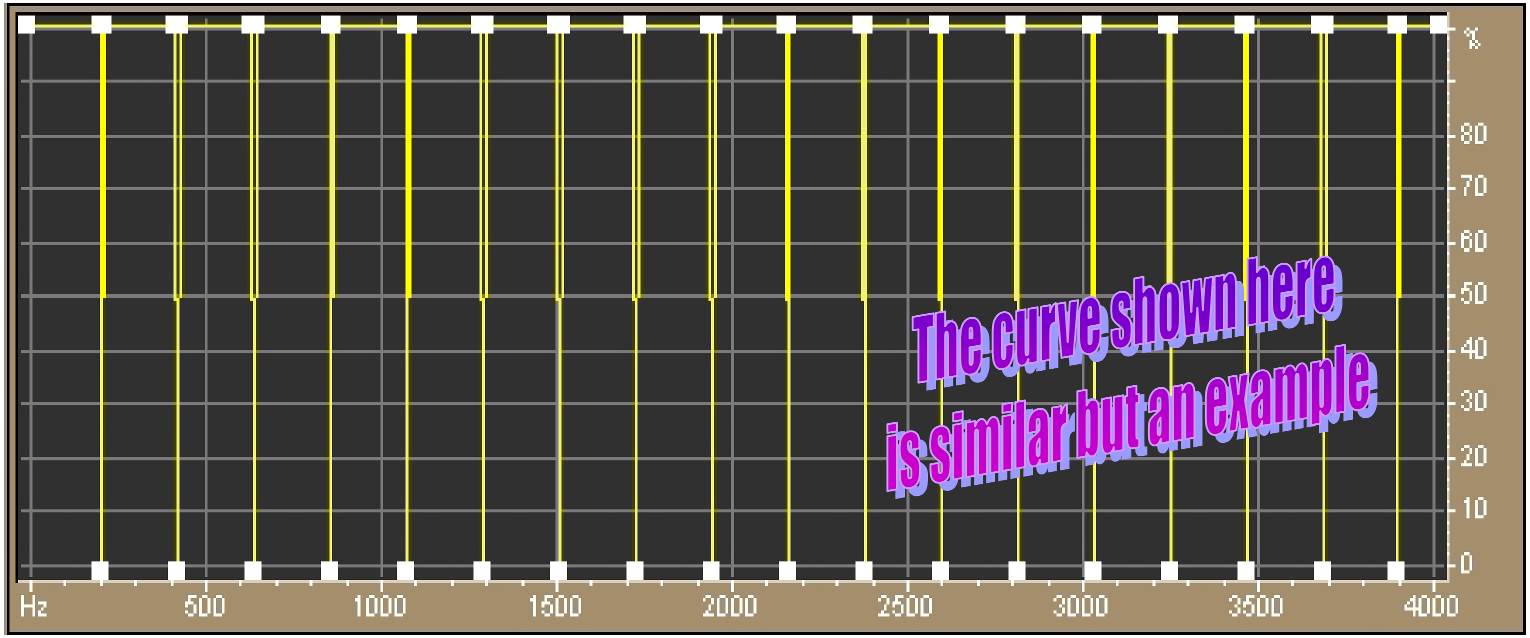
Audio Firmware in R.E.A.L DSP

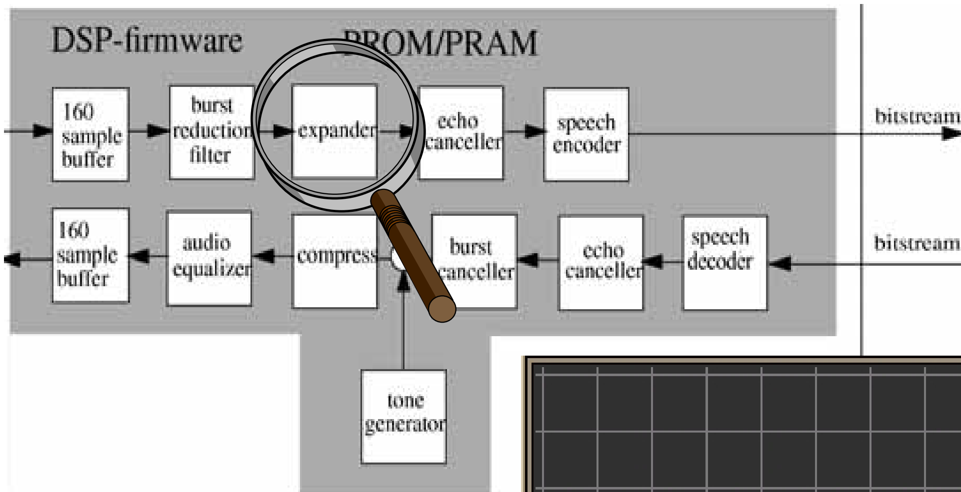
Burst Reduction Filter :



A sine signal with a underlayed GSM-burst-signal (216.7Hz, peaks all 4.6ms, 0.577ms width, -20dB)

The same but filtered @ 216.7 Hz and all harmonics

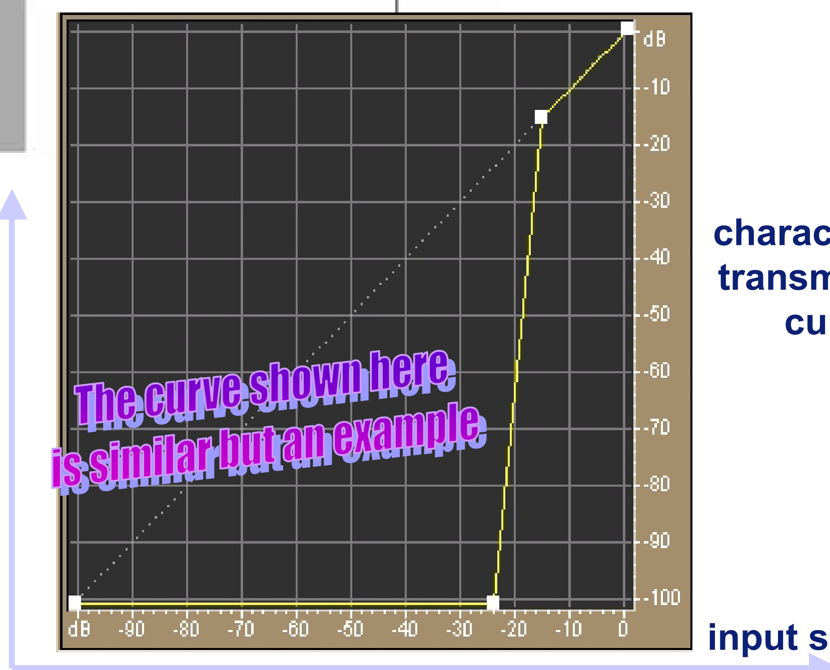
Audio Firmware in R.E.A.L DSP



expander:

Noise suppression by reducing all level under an defined threshold

output signal level /[dB]

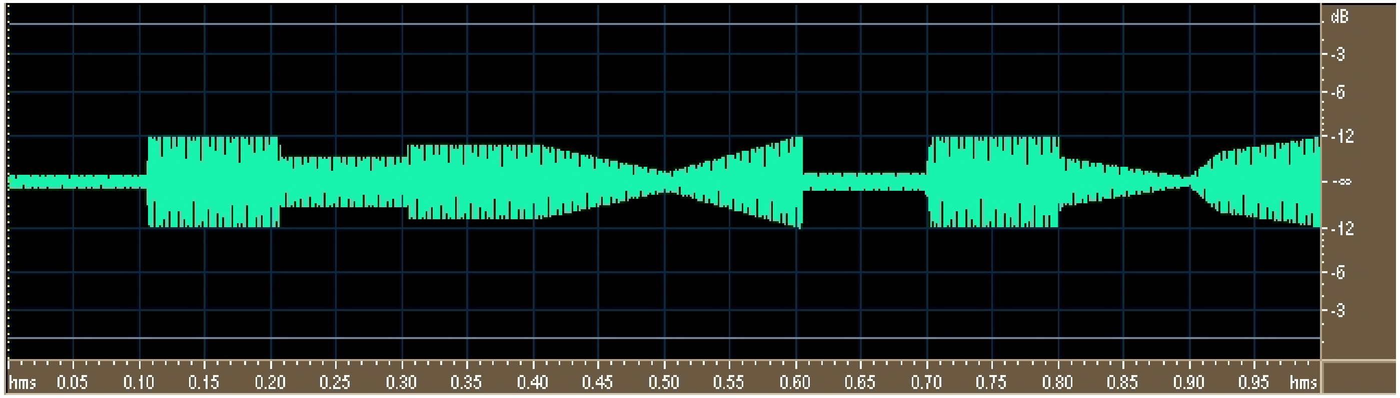


characteristic transmission curve

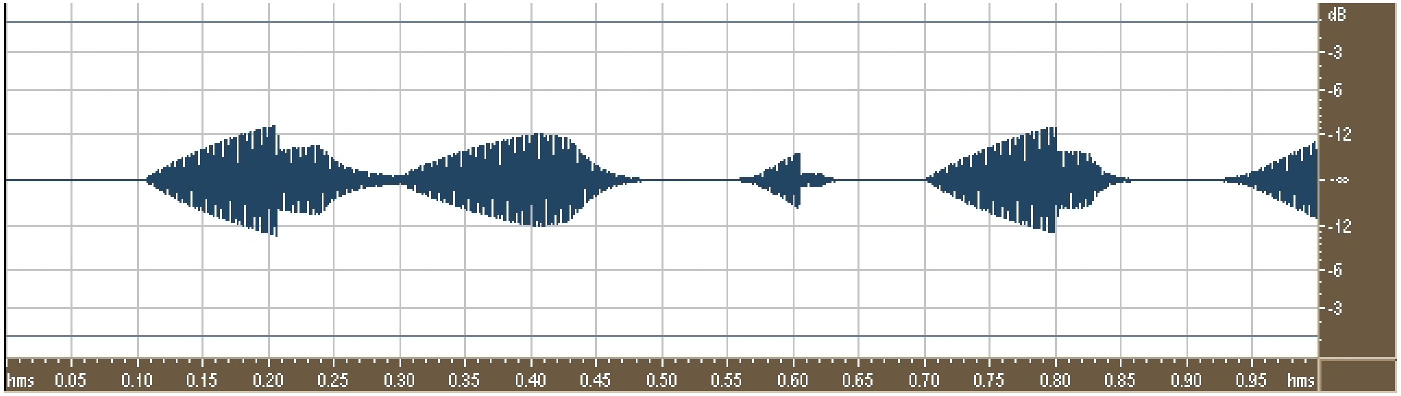
input signal level /[dB]

Audio Firmware in R.E.A.L DSP

Expander :

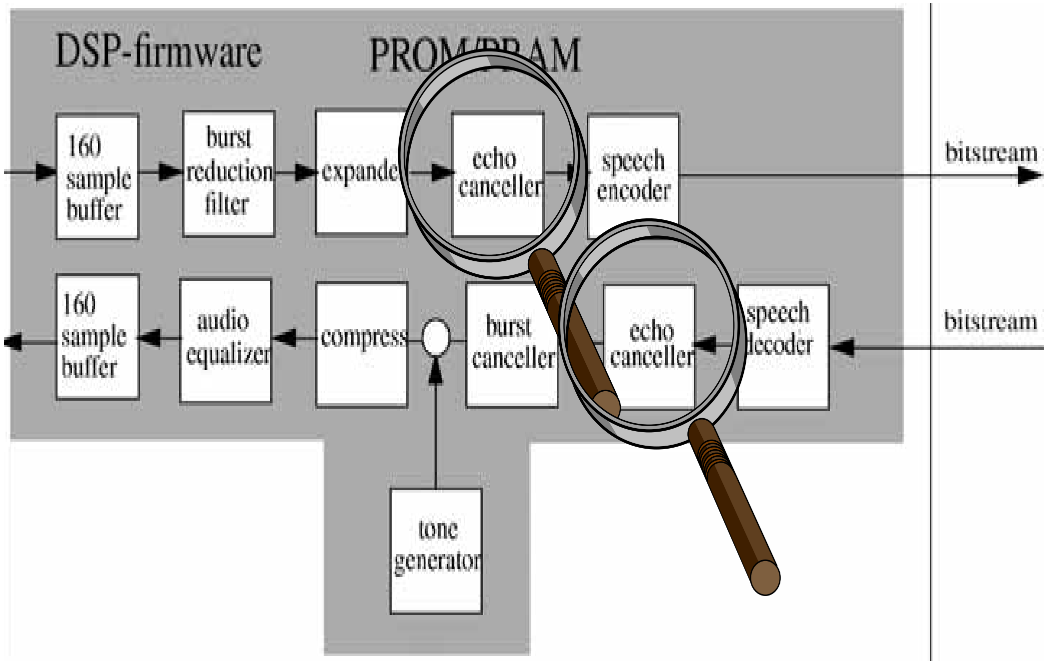


signal origin:



signal expanded:

Audio Firmware in R.E.A.L DSP



Echo Canceller:

Due to the small dimensions of the mobile phone it could be possible that the signal from the loudspeaker is feed back to the microphone.

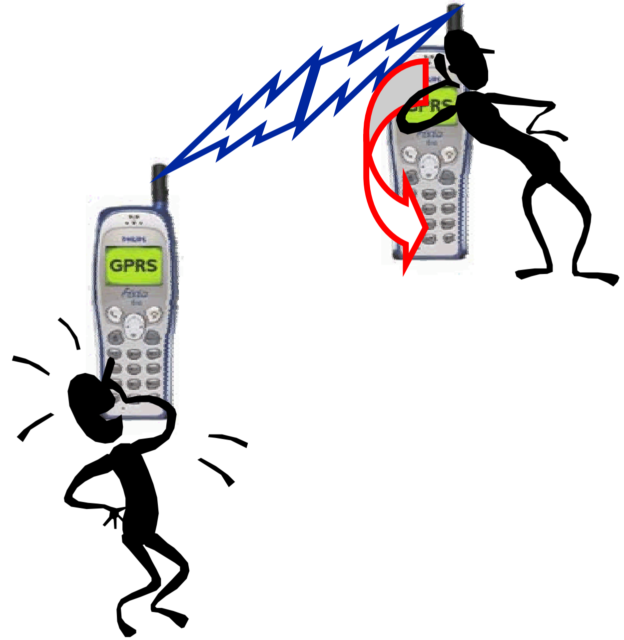
To prevent this an echo canceling algorithm is implemented.

Audio Firmware in R.E.A.L DSP

There are different ways of echo origin possible :

Near End

Far End



Audio Firmware in R.E.A.L DSP

The used algorithm depends on the set mode:

<u>Handset Mode</u>

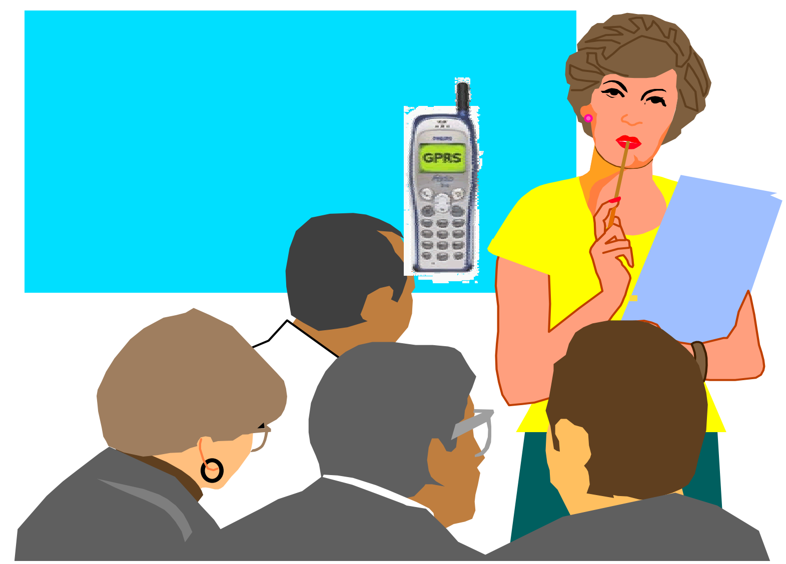
1 speaker/listener only,

phone closed to speaker/listeners mouth/ear.

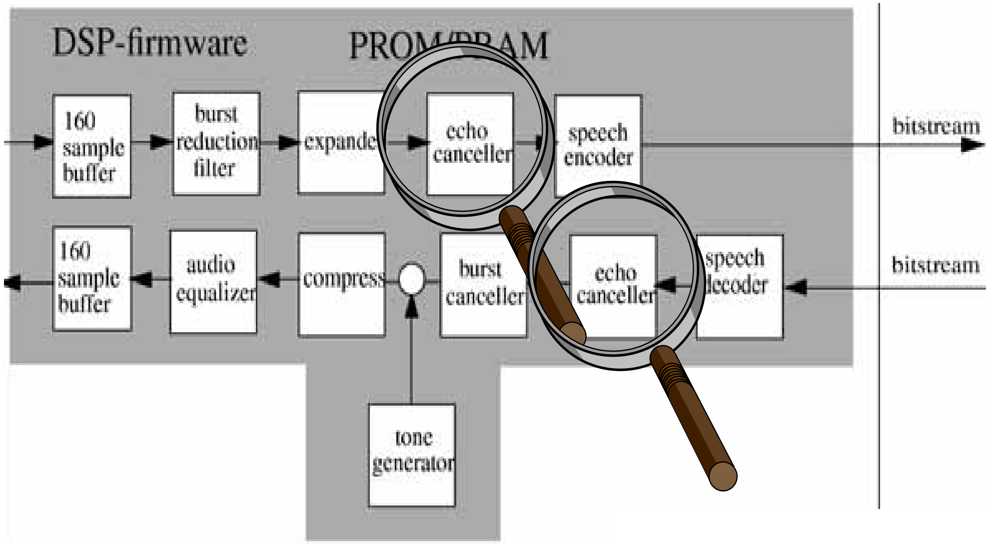
<u>Handsfree Mode</u>

multiple speakers/listeners

phone (microphone/loudspeaker) far away from speakers/listeners

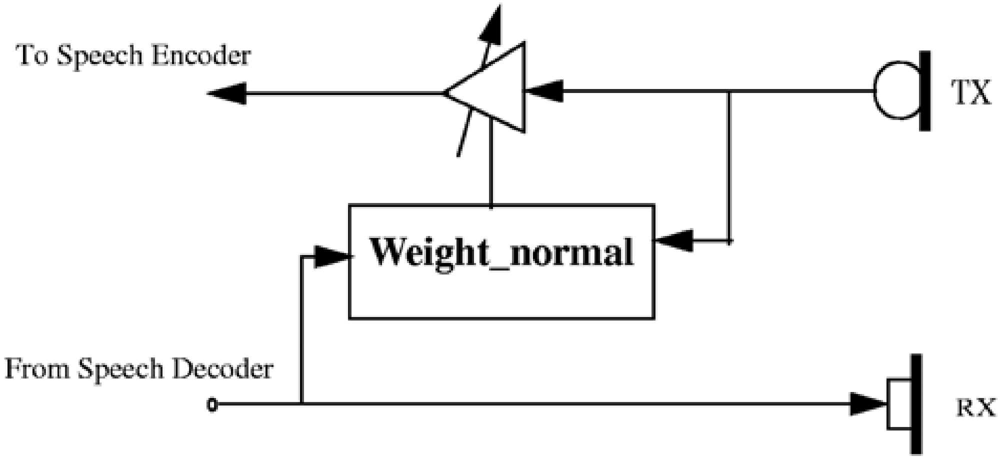


Audio Firmware in R.E.A.L DSP

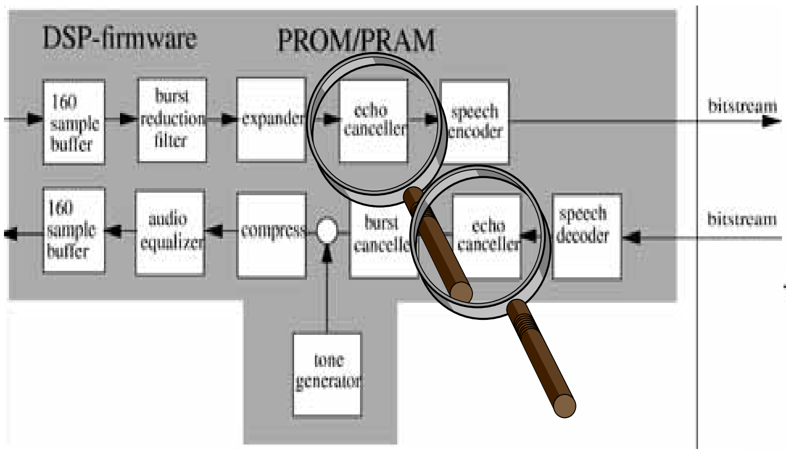


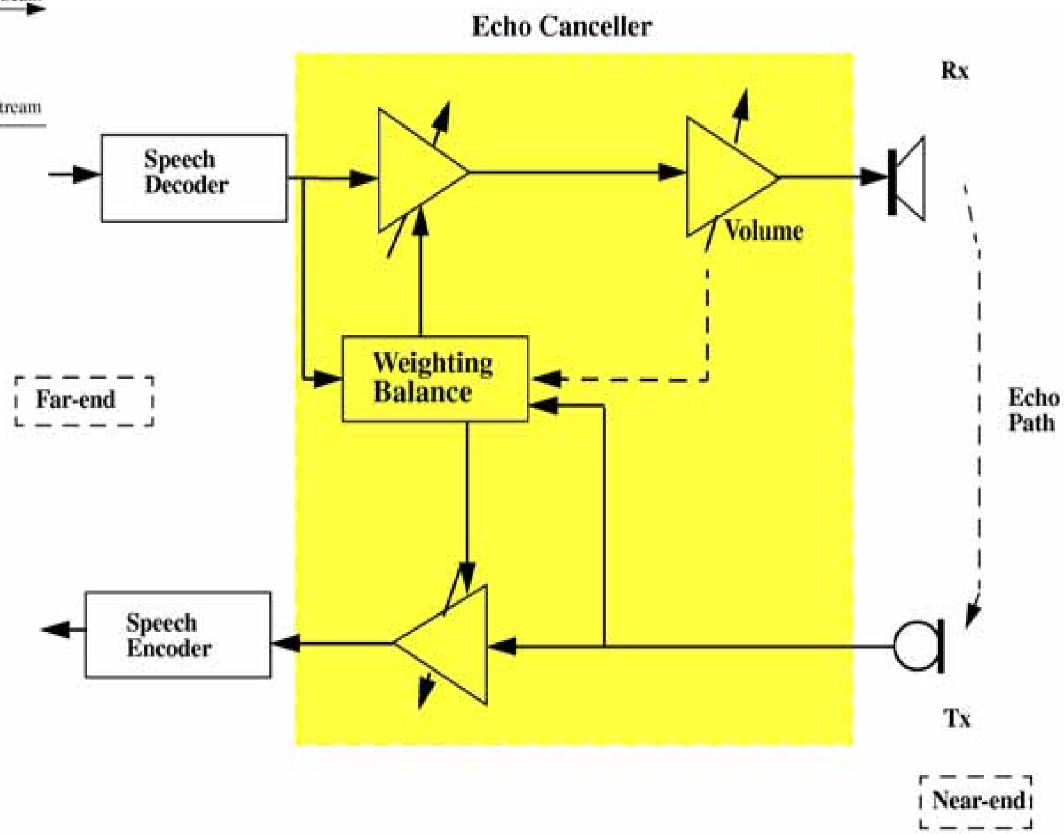
Algorithm in handset mode:

Only mic-signal is attenuated in relation to Tx- and Rx-signal strength



Audio Firmware in R.E.A.L DSP

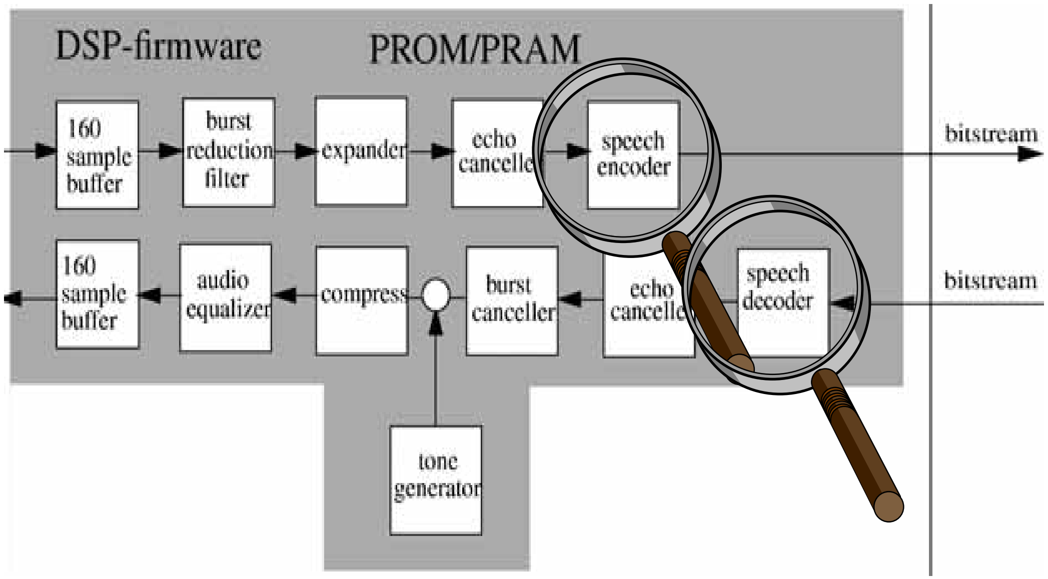




Algorithm in handsfree mode

Mic- and speaker signal are attenuated in relation to Tx- and Rx-signal strength and set speaker volume.

Audio Firmware in R.E.A.L DSP



Speech Encoder/Decoder

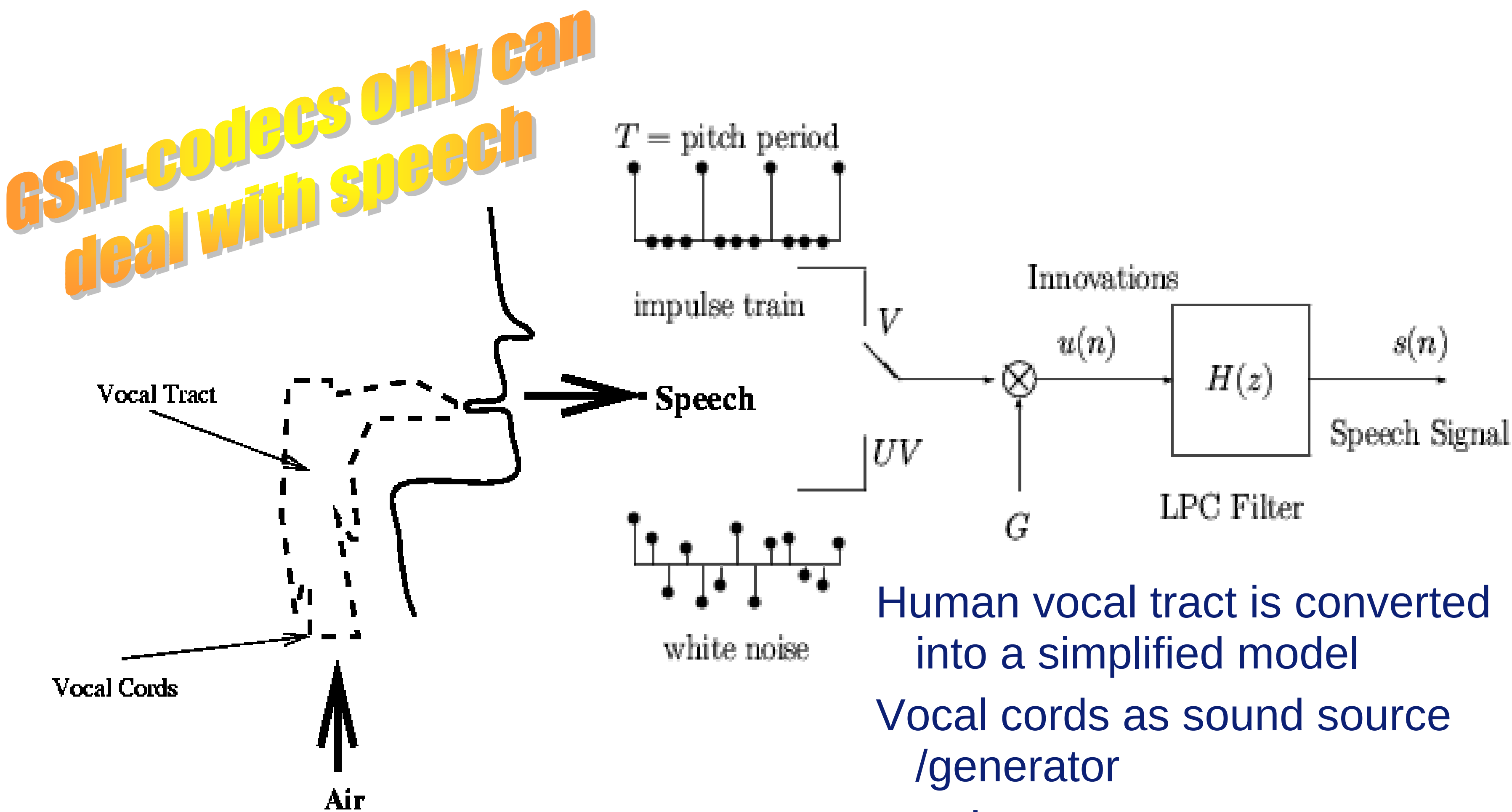
Compresses/decompresses the audio data from/to 104kBit/s to 13kBit/s (FR).

The function of compressing is equivalent to MP3 but the idea behind is different.

There are 4 different codecs specified : FR, EFR, HR, AMR.

GSM-codecs only can deal with speech

Audio Firmware in R.E.A.L DSP



Human vocal tract is converted into a simplified model

Vocal cords as sound source /generator

vocal tract as resonator

Audio Firmware in R.E.A.L DSP

GSM-codecs only can deal with speech

<u>Advantages of the Modeling</u>

The model is known on the transmitter- and receiver-side.

Only the parameters of the model had to be transmitted.

<u>Disadvantages of the Modeling</u>

Only speech could be transmitted.

Music Sample(8kHz, 16bit)

Coded/Encoded with HR codec

Audio Firmware in R.E.A.L DSP

GSM-codecs only can deal with speech

4 Different codecs :

- **FR** = full rate codec defined in GSM 06.10, “the classic”, outgoing bitstream 13kBit/s

- **HR** = half rate codec, defined in GSM 06.20
With better coding algorithms the HR-codec reaches the performance of the FR but the bitstream has only 6.5kBit/s

- **EFR** = enhanced full rate, defined in GSM 06.60
With better coding algorithms the audio performance is better than FR but the bitstream remains on 13kBit/s

- **AMR-NB** = adaptive multirate narrow band, defined in GSM 06.90
The data rate varies between 4.75 and 12.2 kBit/s depending on the channel quality

Audio Firmware in R.E.A.L DSP

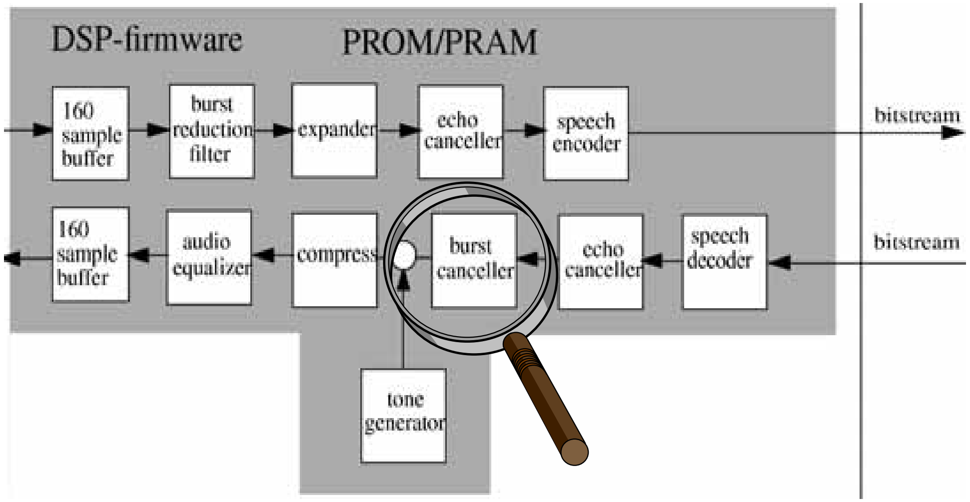
Audio Samples

- FR (full rate)
- HR (half rate) (8kHz, 16bit)
- EFR (enhanced full rate)
- AMR-WB (adaptive multi rate wide band) max. bit rate (16kHz, 16bit)
- AMR-WB (adaptive multi rate wide band) min. bit rate

original

encoded/decoded signal

Audio Firmware in R.E.A.L DSP



Burst Canceller

Suppresses received bad audio frames in FR additional to the codec

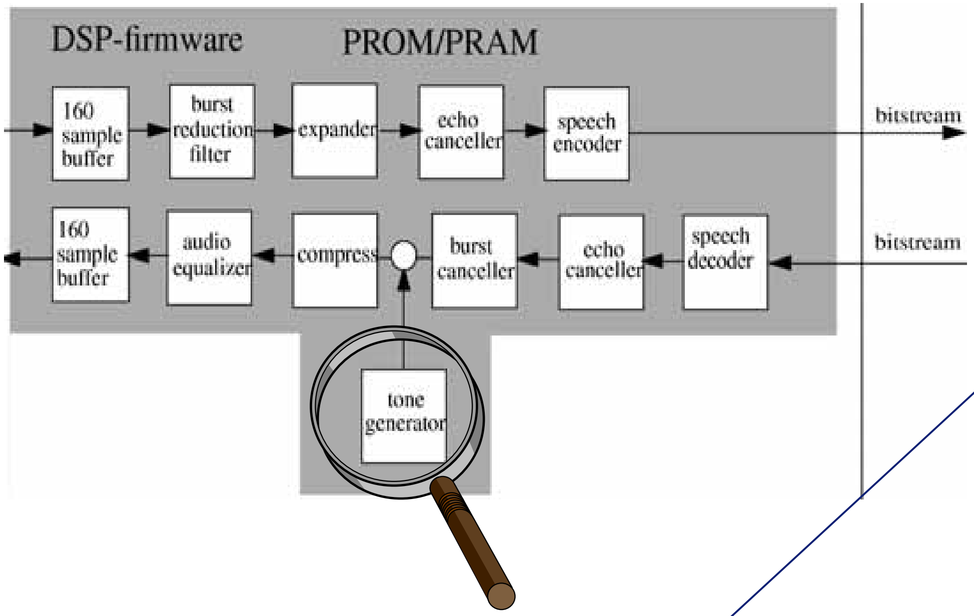
In FR there are only 3 check bits present which can decide if a received audio frame is good or bad.

These are not enough to detect reliable bad frames. So it could happen that distorted signals could reach the loudspeaker.

To prevent this the audio signal is checked for great surges. If an abnormal behaviors is detected the audio signal is set to 0.

Additional a GSM-message (RxQual = BER estimation) is interpreted which is not used in the speech codec but could be used for a better bad frame indication.

Audio Firmware in R.E.A.L DSP

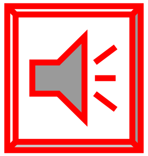


Tone Generator

- generates dial-tones
- generates comfort noise

Comfort Noise

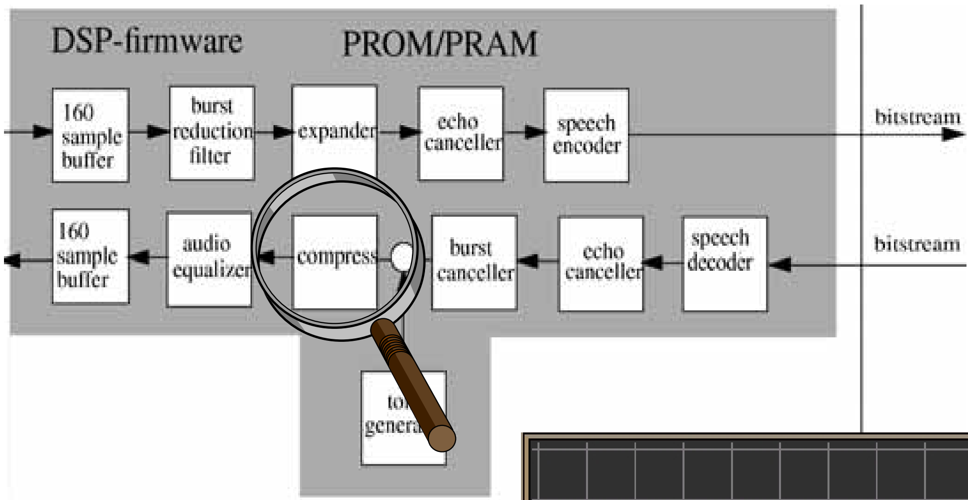
In a speech connection very often a period of silence occurs.
If the mobile performs DTX (dis-continous transmission) then the physical data transmission of the speech signal is interrupted.
To prevent silence in the speaker artificial noise is generated in the MS itself.



Dial Tones

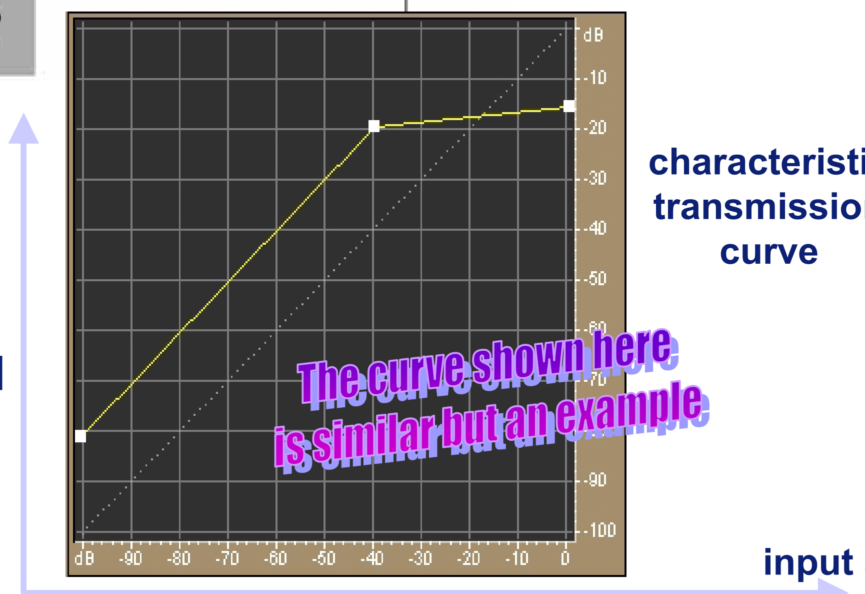
Every number is defined as 2 mixed sine signals.
DTMF = Dual Tone Multiple Frequency

Audio Firmware in R.E.A.L DSP



Compressor

Dynamic reduction by reducing all level above a defined threshold



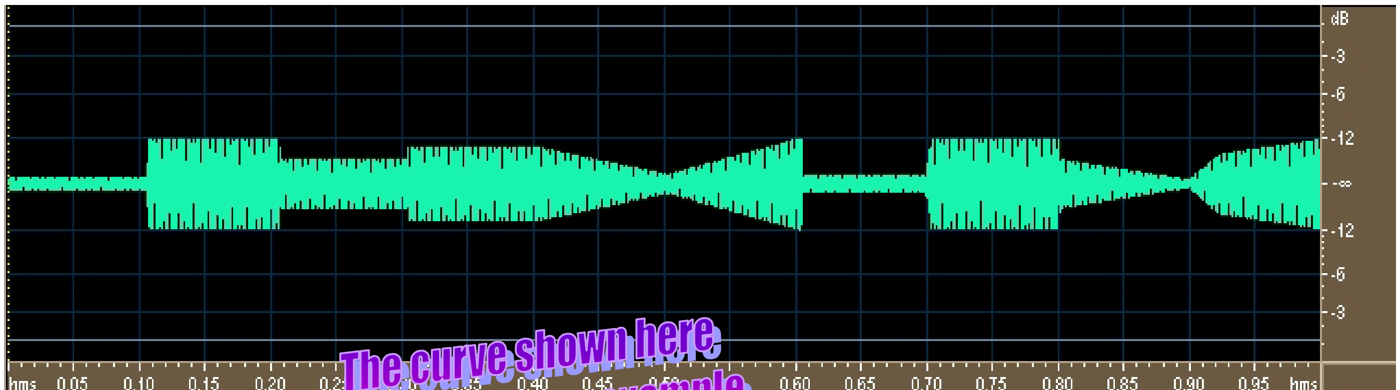
characteristic transmission curve

output signal level /[dB]

input signal level /[dB]

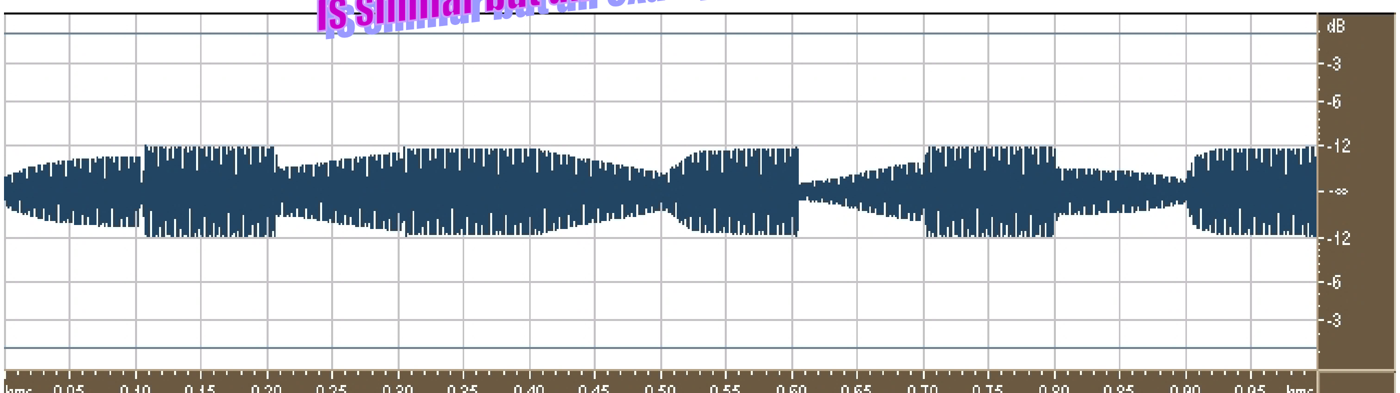
Audio Firmware in R.E.A.L DSP

Compressor :



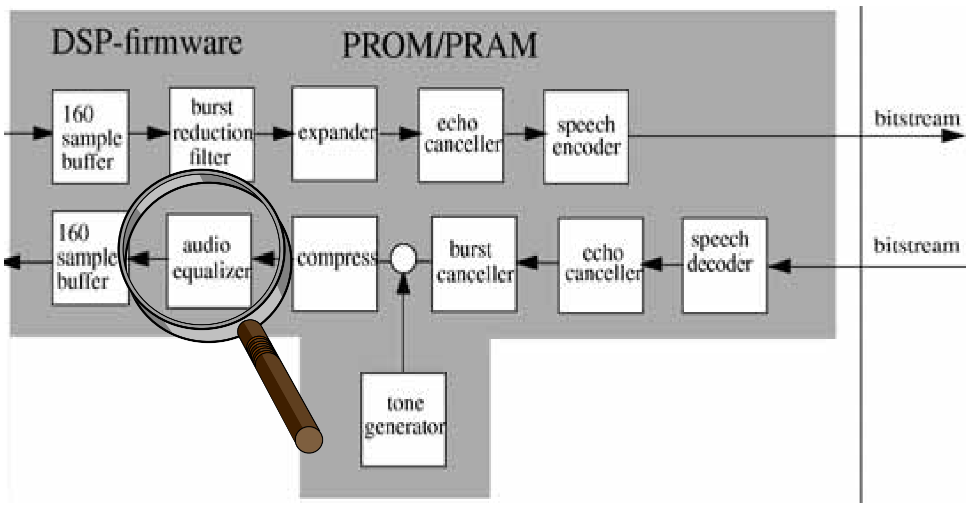
signal origin:

The curve shown here is similar but an example



signal compressed:

Audio Firmware in R.E.A.L DSP



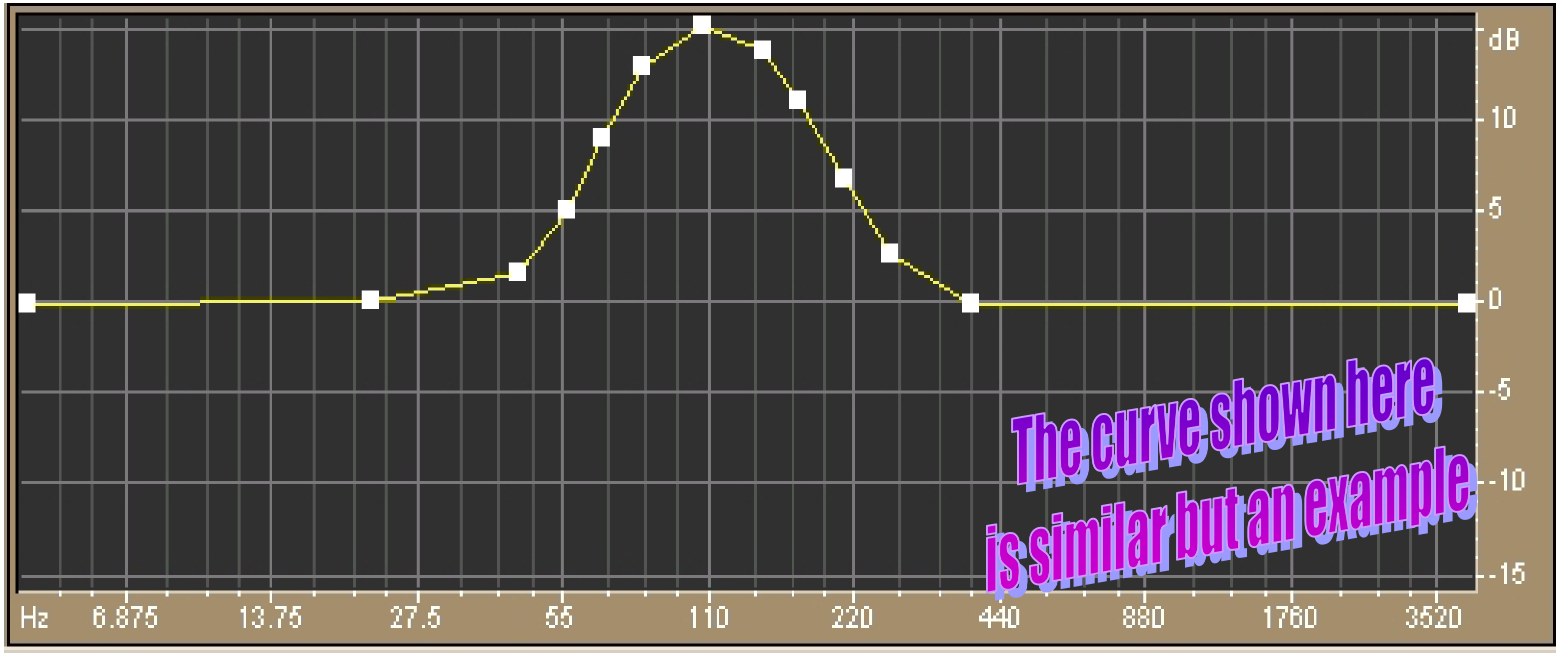
Audio Equalizer

Adapts the acoustic behaviour to the subjective desires of the final customer

We use for instance a bass boost function which amplifies the range below 300Hz

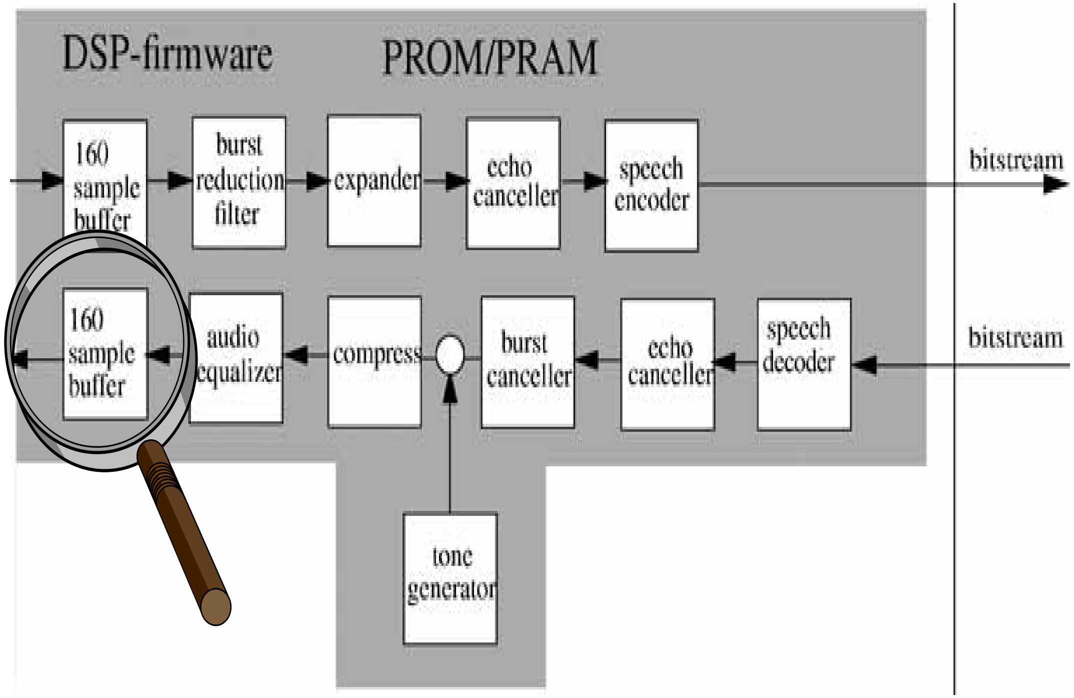
Audio Firmware in R.E.A.L DSP

Audio Equalizer Sample



speech origin : speech equalized :

Audio Firmware in R.E.A.L DSP



160 sample buffer :

160 audio samples (16bit, 8kHz) to transmit to BAI are stored into

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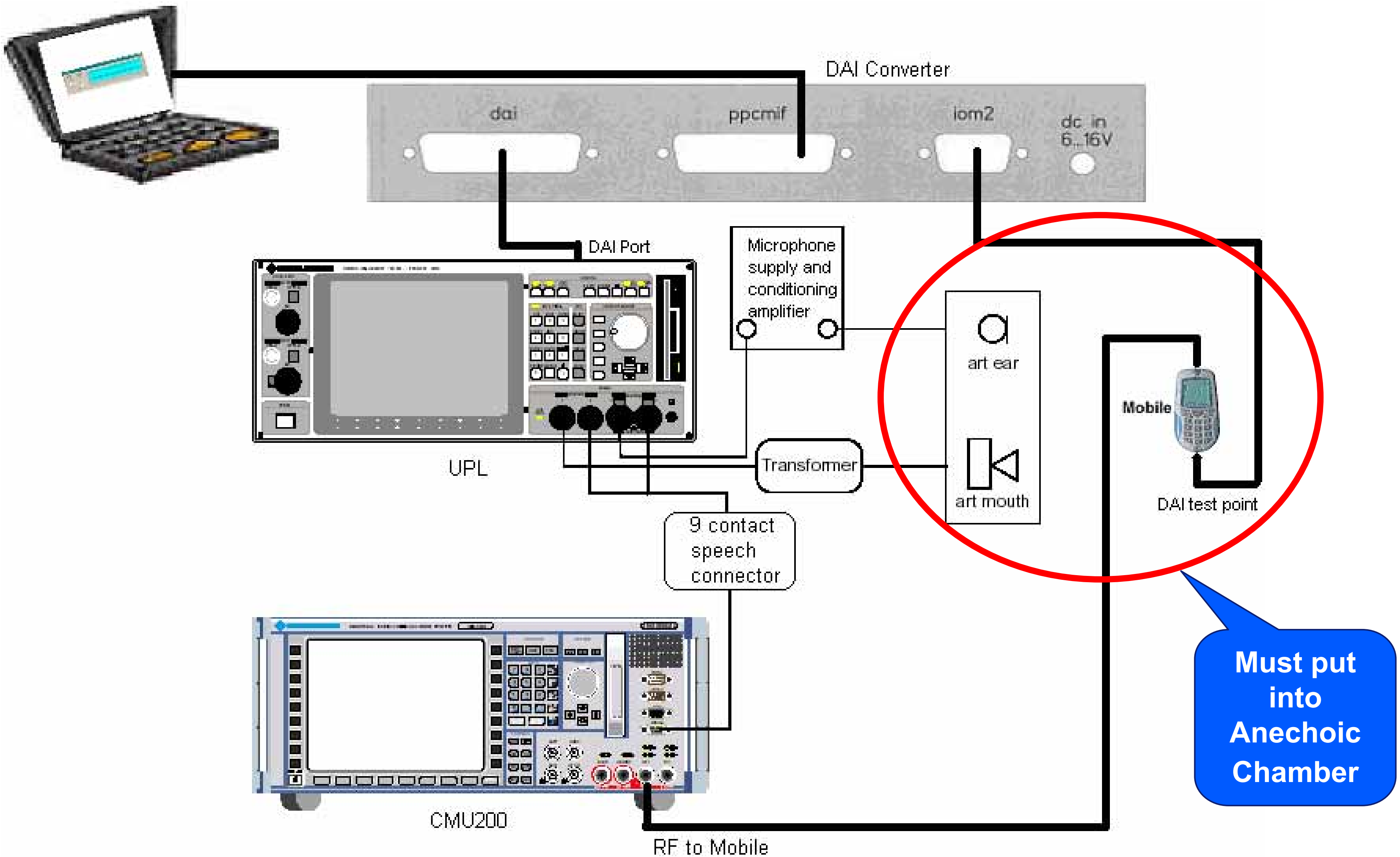
Acoustic Test Bench

Requirements

- Phone with running phone software
- PC with running TAT-tool which complies to the phone software
- Serial connection between PC and phone
- Phone with accessible IOM-2-interface
- DAI-converter
- Audio Analyzer UPL16 with GSM-test-macro
- CMU200 with B52 option
- Anechoic chamber
- Telephone test head
- Ear Simulator
- Mouth Simulator
- Two channel Microphone Power Supply and Conditioning Amplifier
- Transformer
- Acoustic calibrator
- 9-contact speech connector

Acoustic Test Bench



Acoustic Test Bench

Account for devices (UPL)

- Roles description

  UPL16 is the audio generator and analyzer
- Configure

  1. Load configure file from system (DAI test)

     \..F3➔C:\GSM➔shell➔Exit➔F11➔ Load”r99_tst

     ➔Ready➔F6_run➔select_type➔test…
  2. Load configure file from system (OA test)

     \..F3➔C:\ phonetst➔shell➔Exit➔F11➔ Load”gsm_tst

     ➔Ready➔F6_run➔select_type➔test…
  3. Load configure file for TDMA noise test

     \..F3➔File➔A:\RX_NOISE.SAC➔Enter➔GRAPH test…

Acoustic Test Bench

Account for devices (CMU200)

- Roles description

  CMU200 is a full function BS for building a call with MS

- Configure

  1. DAI Test(with DAI)

     \..signal_off➔network➔Bit_stream➔handset_low

     \..call setup➔MS signal➔DAI➔acoustic_device

  2. OA Test(without DAI)

     \..signal_off➔network➔Bit_stream➔handset_low

     \..call setup➔MS signal➔DAI➔normal

Acoustic Test Bench

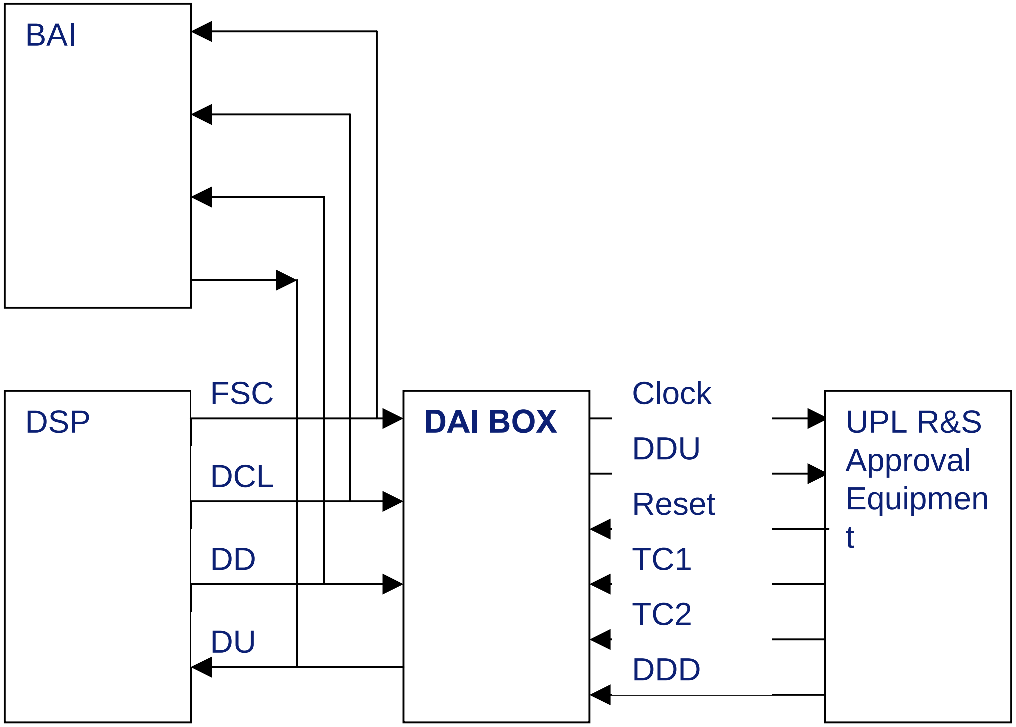
Account for devices (DAI converter)

- Roles description
  - ➔ Dedicated to Audio tests for Final Type Approval
  - ➔ Used to interface between the IOM2 bus from DSP&BAI and the audio analyzer test equipment of Approval (UPL16-B8/B9)
  - ➔ Convert the 16 bit data (slot configuration) on continous 104 kHz data flow (13 bits)
  - ➔ Management of the reset signal provided by the test bench
- Configure

1. DC in: 6~16V (typical 7.5V)
2.

Acoustic Test Bench

- Connections DSP-DAI-UPL



Acoustic Test Bench

- Connector of IOM-2 interface

DAI Converter		Mobile Phone IOM-2	
Pin of connector	Signal of DAI	Pin of OM6357	Signal of IOM-2
1	DD	A9	ADO(DU)
3	DU	A8	ADI (DD)
6	DCL	B8	ACLK (DCL)
7	FSC	B9	AFS (FSC)
9	VDDD	C9	VDD1
others	GND	GND	GND

Acoustic Test Bench

Account for devices (Ear Simulator)

- Roles description

  Measuring microphone with adapters for connection to the ear piece of the DUT

- Configure

  1. DC Supply: 24V
  2. Configure for its Amplifier

CH.	Start Hz	Stop kHz	OUT	UNIT
1	20	10	100m	V/Pa
2	0.1	100	1	V/Pa

Acoustic Test Bench

Account for devices (Mouth Simulator)

- Roles description

  Special loudspeaker for simulation of the mouth

- Configure

  1. Impendence converter, scale4:1
  2.

Transformer Unit: 100Hz~100kHz
Input: 10V RMS Max.
Output: Load 4ohms Min.

- Audio in GSM Mobile Phone
- Hardware Components in Audio Path
- HW Structure of PCF50732
- VSP in PCF50732
- Audio Firmware in R.E.A.L DSP
- Acoustic Test Bench
- Test Cases in FTA for Audio
- How to Tune the Audio
- TDMA Noise

Test Cases in FTA for Audio

There are two kinds test method

- DAI test (use DAI converter)
- OA (Over Air) test(without DAI converter)

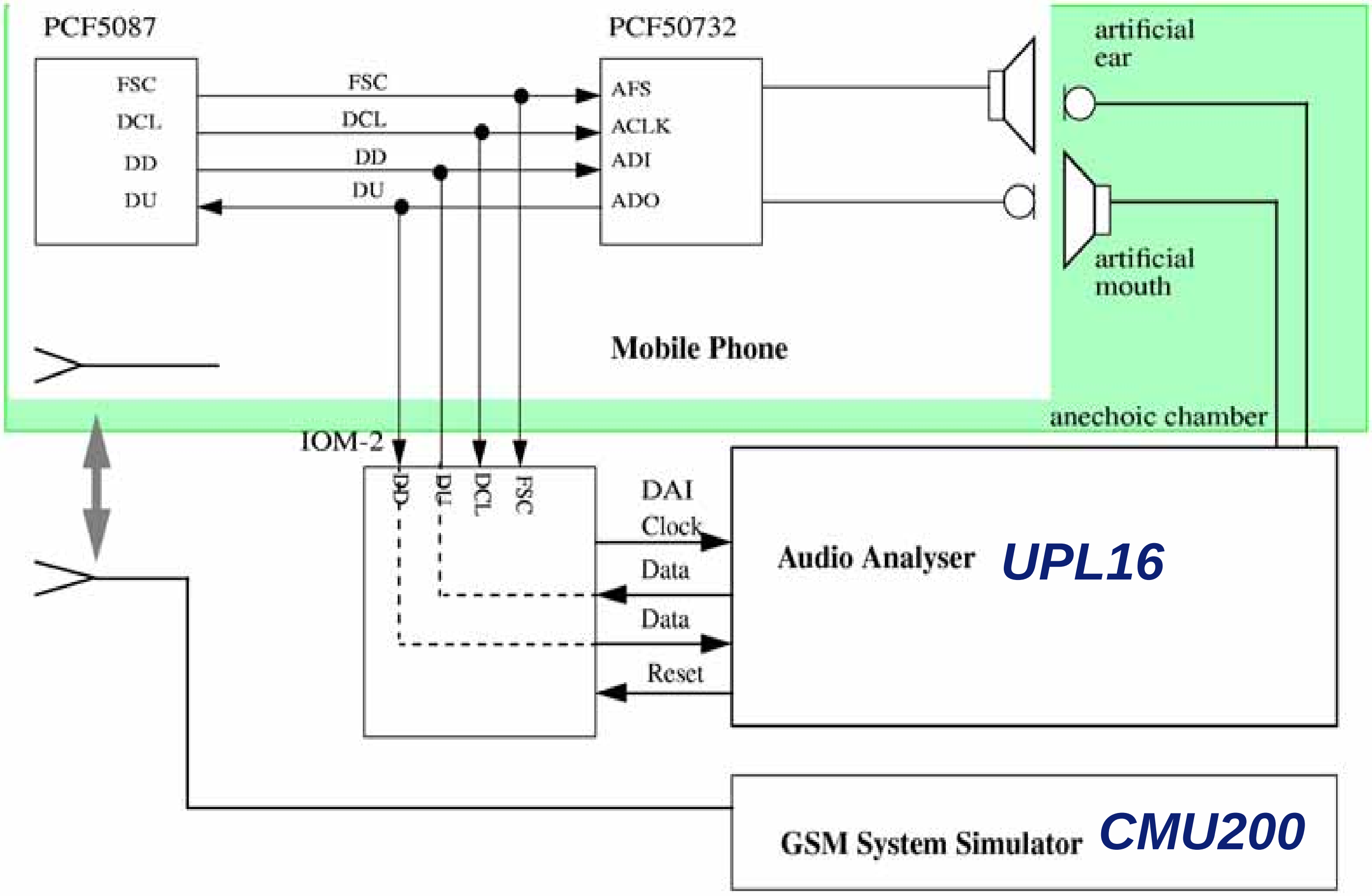
The requirements for audio are defined in GSM 11.10, chapter 30 “Speech teleservices”

There are 15 test cases for testing the audio compliance of the MS defined and 7 of 15 must be fullfilled

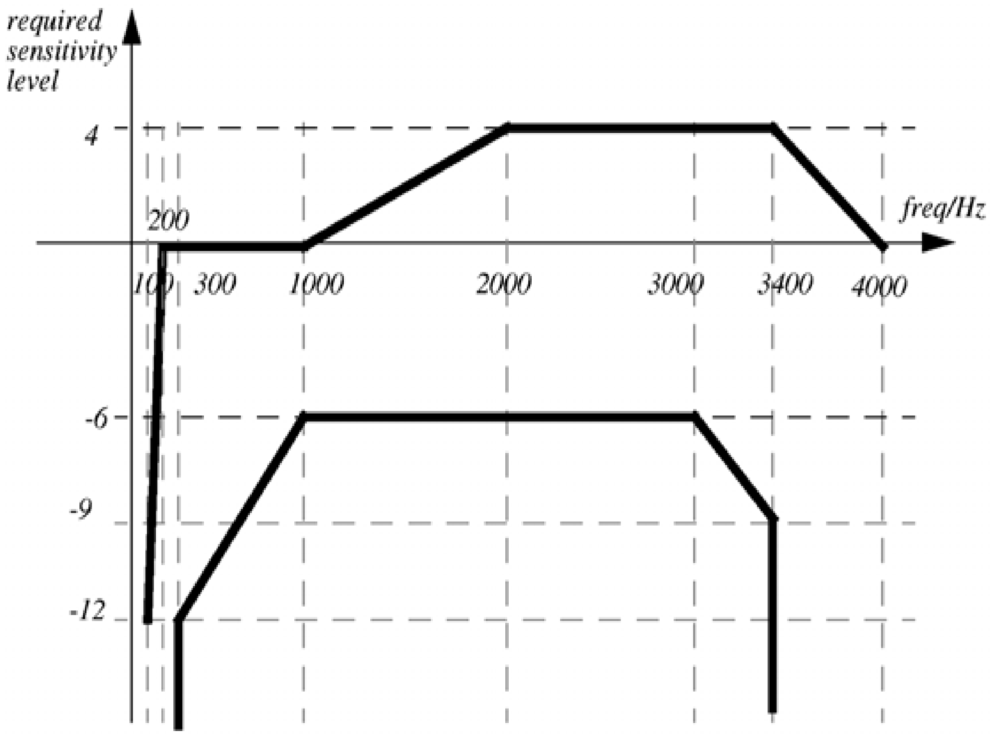
GSM 11.10, 30.1: Sending Sensitivity/Frequency Response
GSM 11.10, 30.2: Sending Loudness Rating(SLR)
GSM 11.10, 30.3: Receiving Sensitivity/Frequency Response
GSM 11.10, 30.4: Receiving Loudness Rating(RLR)
GSM 11.10, 30.5.1: Side Tone Masking Rating (STMR)
GSM 11.10, 30.6.2: Stability Margin
GSM 11.10, 30.7.1: Sending Distortion

Test Cases in FTA for Audio

Test setup which is used in customer support

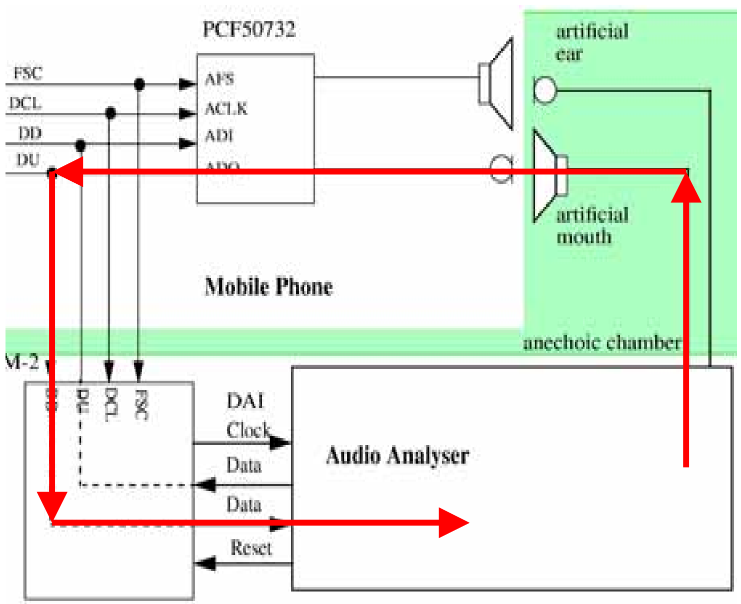


Test Cases in FTA for Audio



<u>*Sending Sensitivity/Frequency Response GSM11.10, 30.1</u>

The frequency of the audio path from MIC to the digital output must fit to pre-defined masks



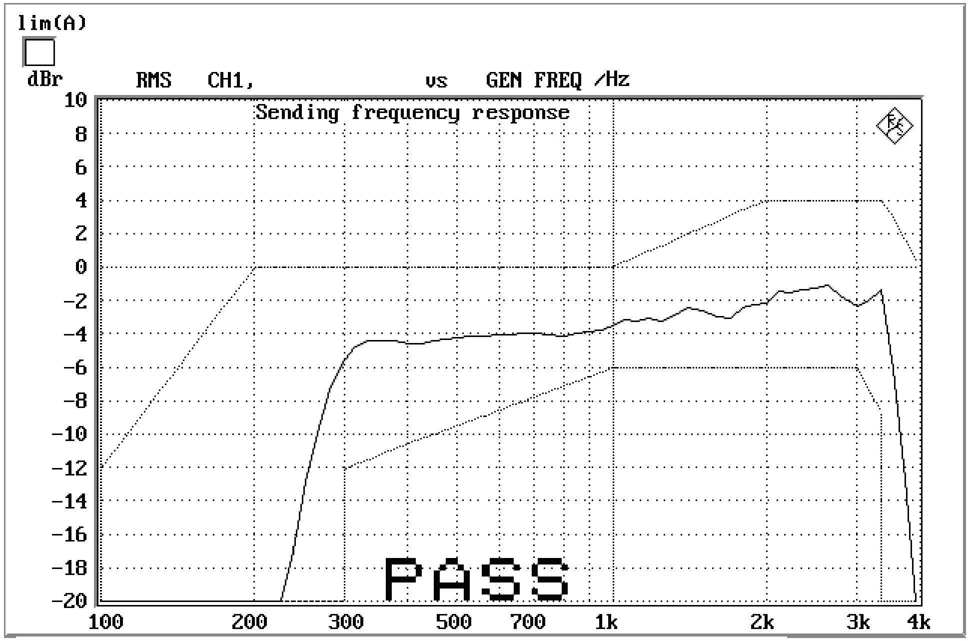
The absolute sensitivity keeps unconsidered.
The sound pressure on the artificial mouth is 4.7dBPa
voltages are measured in the digital BAI-output signal

Test Cases in FTA for Audio

Limit lines according to GSM 11.10 table 30.1

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

Test example:



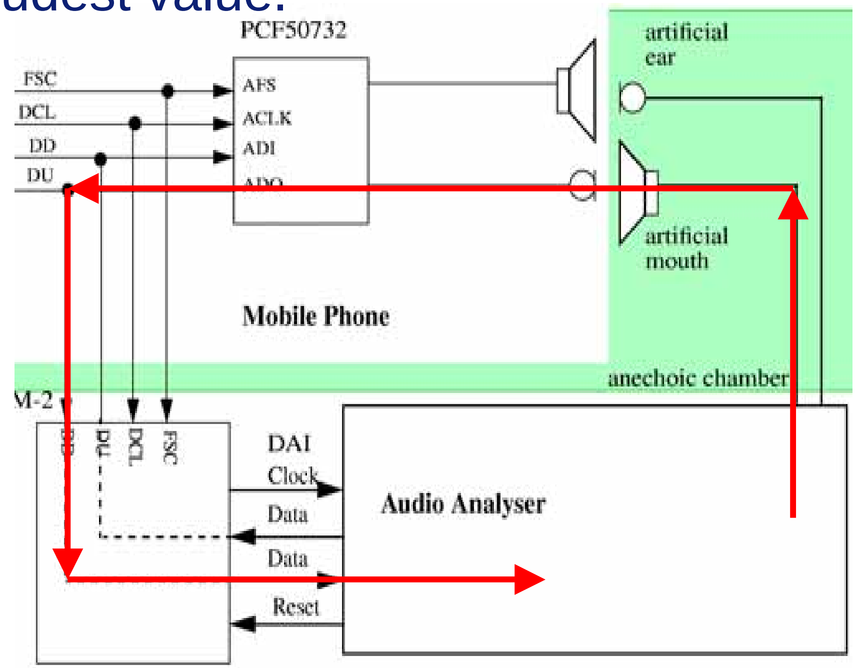
Test Cases in FTA for Audio

*Sending Loudness Rating GSM11.10, 30.2

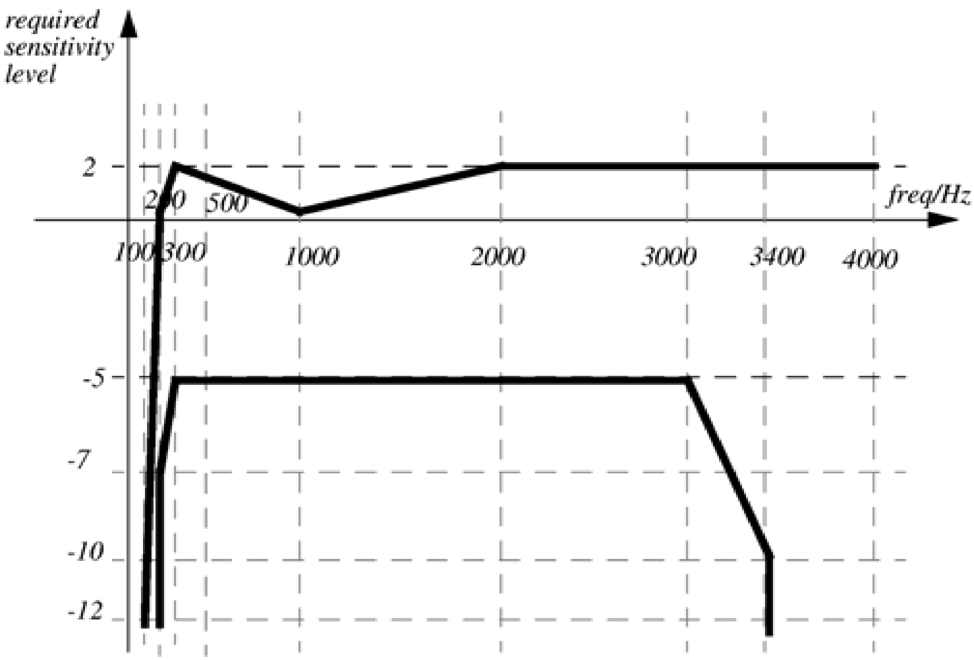
The loudness of the audio path from MIC to the digital output must be 5dB< SLR <11dB

The loudness is measured at 14 frequencies, 11dB is the smallest allowed loudness 5dB is the loudest value.

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

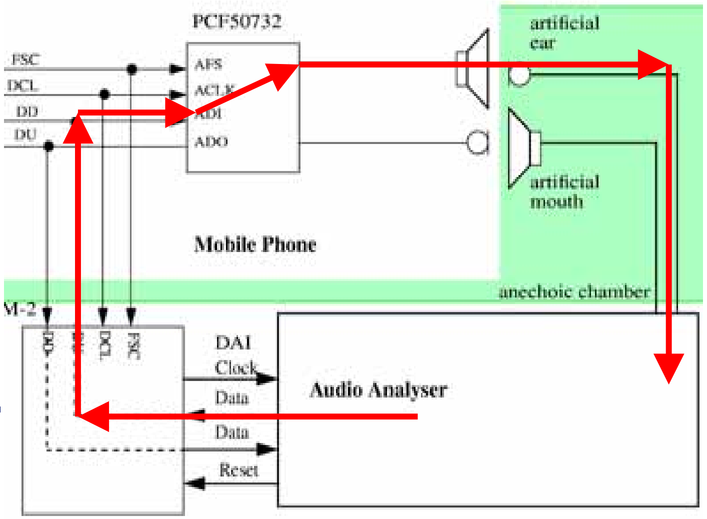


Test Cases in FTA for Audio



***Receiving Sensitivity/Frequency Response GSM11.10, 30.3**

The loudness of the audio path from the digital input to the speaker must fit to pre-defined masks.



The absolute sensitivity keeps unconsidered.
The sound pressure on the digital input is equivalent to -16dBm0
Voltages are measured with the artificial ear which must be calibrated

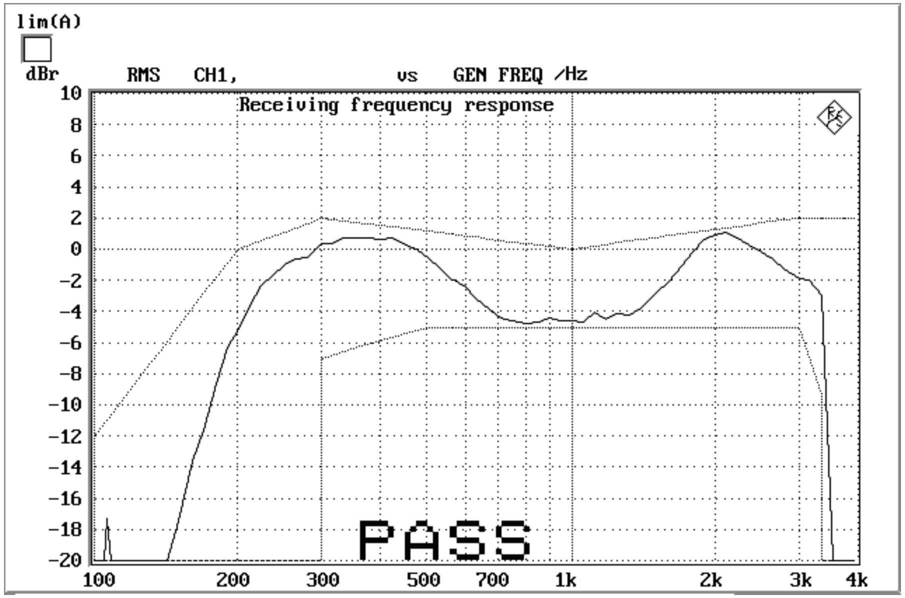
Test Cases in FTA for Audio

Limit lines according to GSM 11.10 table 30.2(ear type1)

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.

Test example:



Test Cases in FTA for Audio

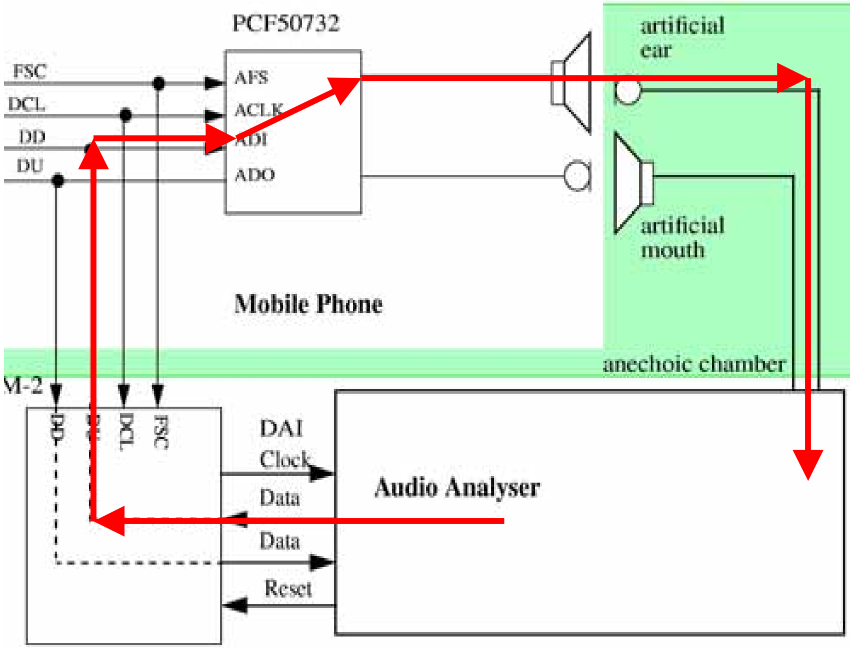
*Receiving Loudness Rating GSM11.10, 30.4

The frequency of the audio path from the digital input to the speaker must be -1dB < RLR <5dB

The loudness is measured at 14 frequencies, 5dB is the smallest allowed loudness and -1dB is the loudest value

For the maximal volume setting RLR shall not be less than –13dB. i.e. not louder than –13dB.

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



Test Cases in FTA for Audio

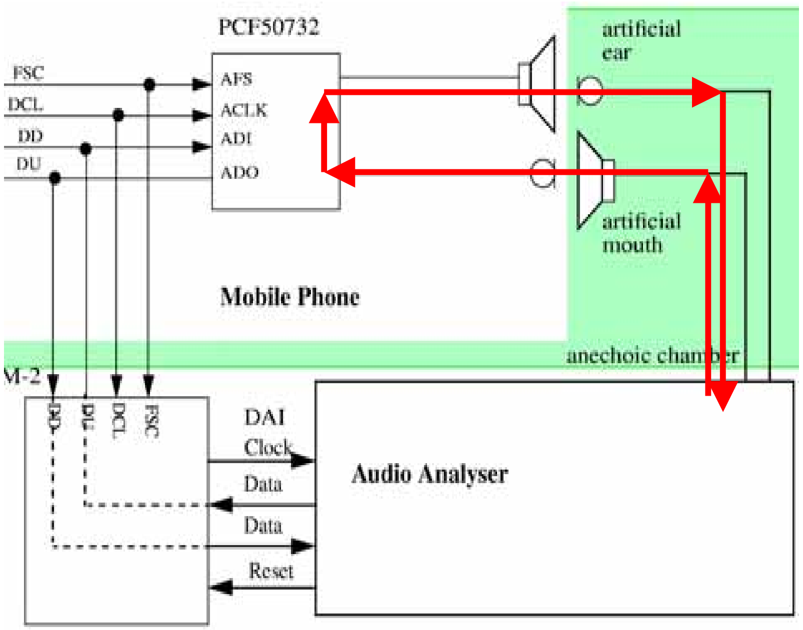
<u>*Sidetone Masking Rating (STMR) GSM11.10, 30.5.1</u>

The attenuation in the sidetone path must be 8dB< STMR < 18dB, STMR is too high and will lead to an uncomfortable feeling for the user in noisy conditions-> We advice to target 16 or 17 dB

Inside the phone the mic-signal is feed back to the speaker-signal. This path is called sidetone path.

The sound pressure on the artificial mouth is -4.7dBPa.

The loudness is measured at 14 frequencies.



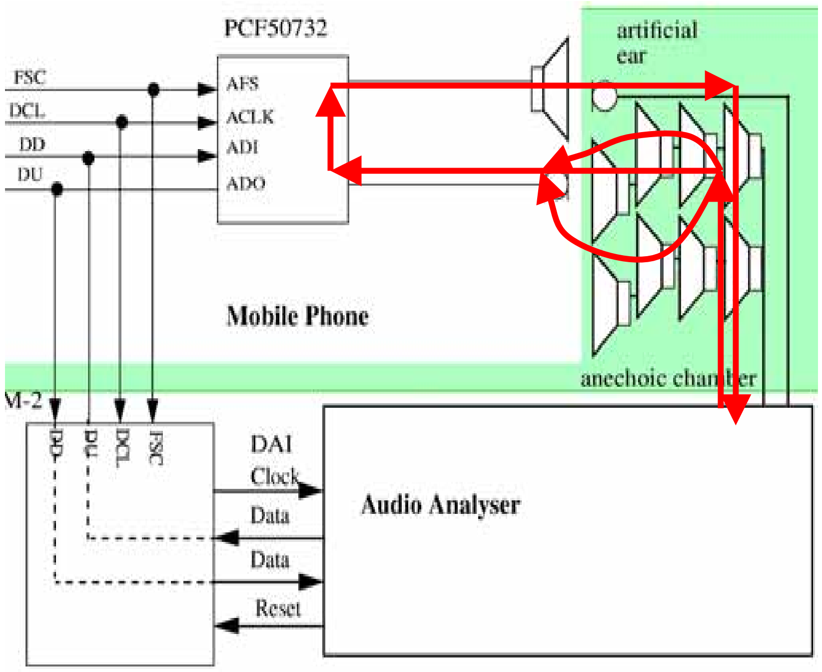
Test Cases in FTA for Audio

Listener Sidetone Rating (STMR) GSM11.10, 30.5.2

The attenuation for interfering signals (white noise sound field) in the sidetone path must be >15dB

Inside the phone the mic-signal is feed back to the speaker-signal. This path is called sidetone path.

8 sound sources produce a homogenic interferer field external sources are needed



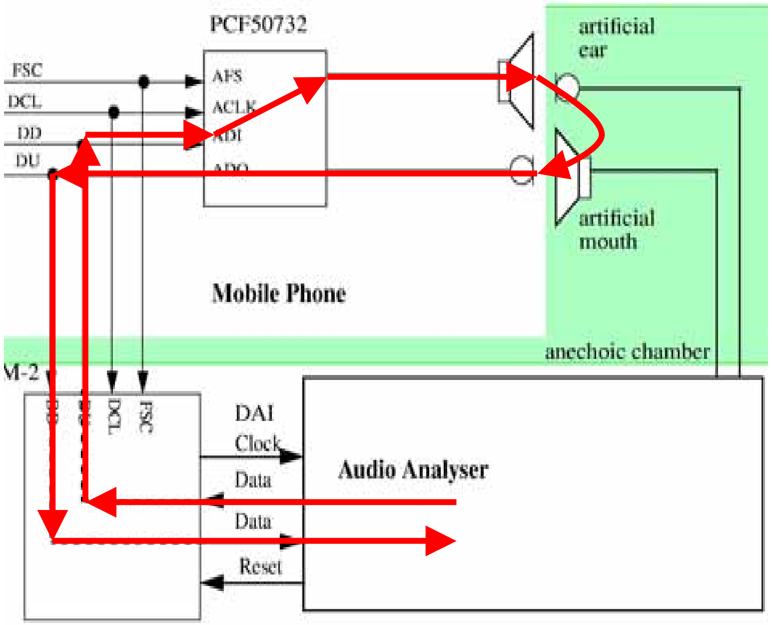
Test Cases in FTA for Audio

Echo Loss (EL) GSM11.10, 30.6.1

The attenuation for speech signals from the speaker to the microphone of the MS >46dB

Due to the small dimensions of the MS it is possible that speech generated from the speaker is feed back to the microphone of the MS

EL is measured with artificial male and female speech signals



Test Cases in FTA for Audio

*Stability Margin GSM11.10, 30.6.2

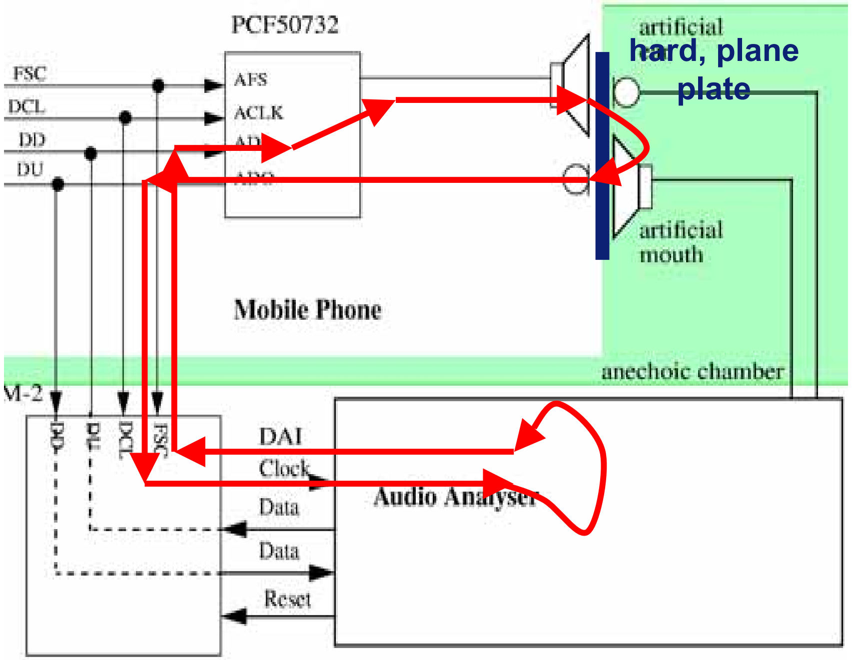
The loop attenuation for feedback loops >6dB

The MS is laying on a hard, plane plate

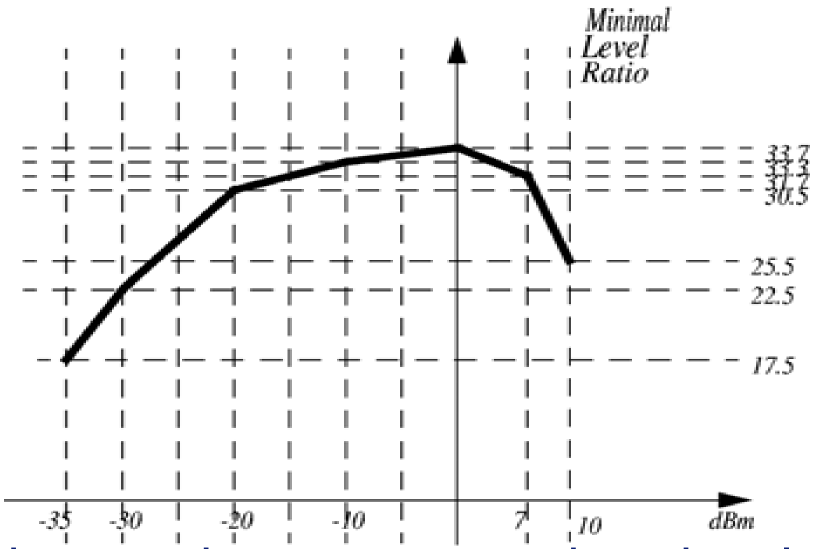
The overall feedback loop gain is adjusted to 6dB

To stimulate the loop a noise signal with a level of -10dBm0 is inserted for 1s

There must be no oscillation after this stimulation



Test Cases in FTA for Audio

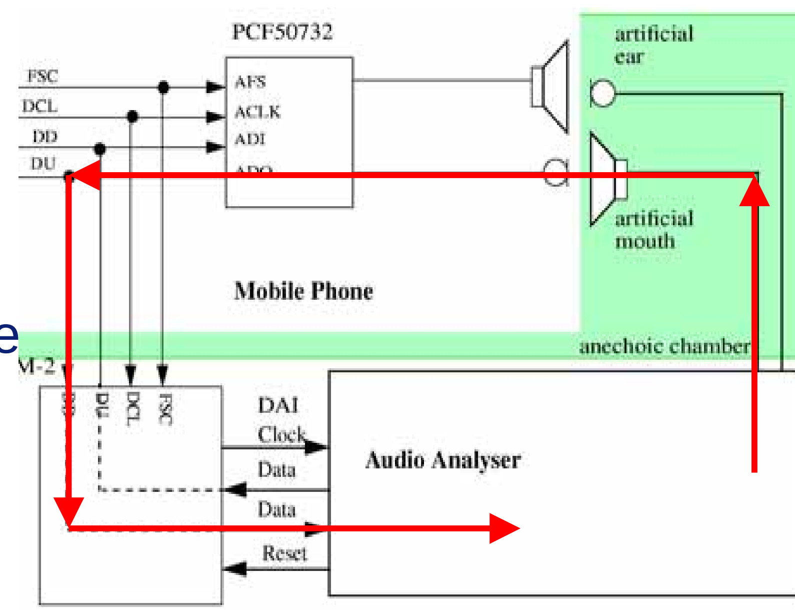


***Sending Distortion GSM11.10, 30.7.1**

The distortion must lower than limits

(but the measured curve must be above the limit curve)

Without DAI, the results will lower 3-5dB



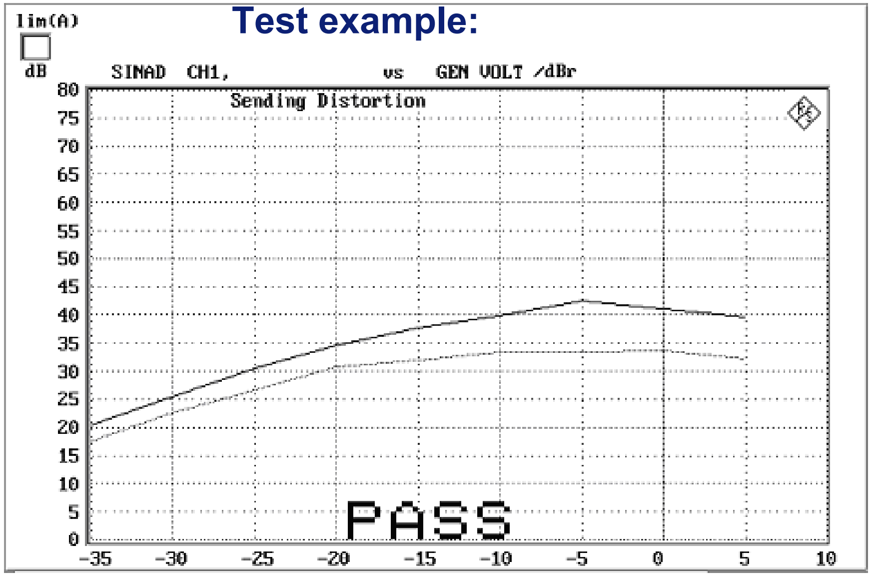
- The mouth generates a sine signal with a frequency of 1015Hz
- The level of this signal is measured on the digital output of the BAI
- The level on the mouth is varied until the measured signal has -10dBm0 (ARL)
- Then the measurement is done with 7 levels relative to ARL

Test Cases in FTA for Audio

Limit lines specified in table 30.3 of GSM 11.10

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

Test example:



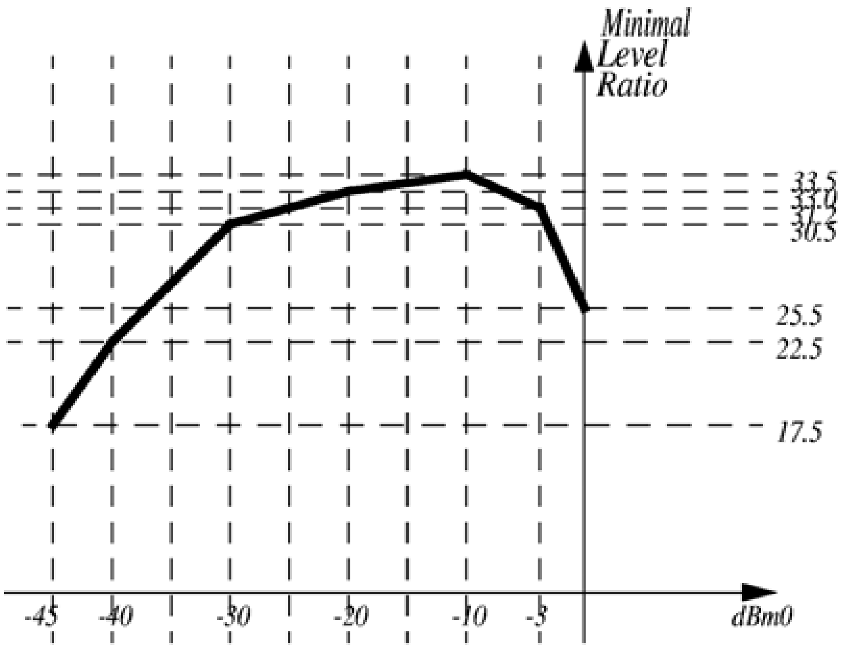
What can improve the results when the test is Failed:

- The measurement have to be done with DAI and not through the codecs (SINAD 3 to 5 dB better by DAI->R&S information)

-The 216 Hz immunity have to be good enough. As the distortion measurement is performed when radio is in conducted way the 216 Hz level is usually low.

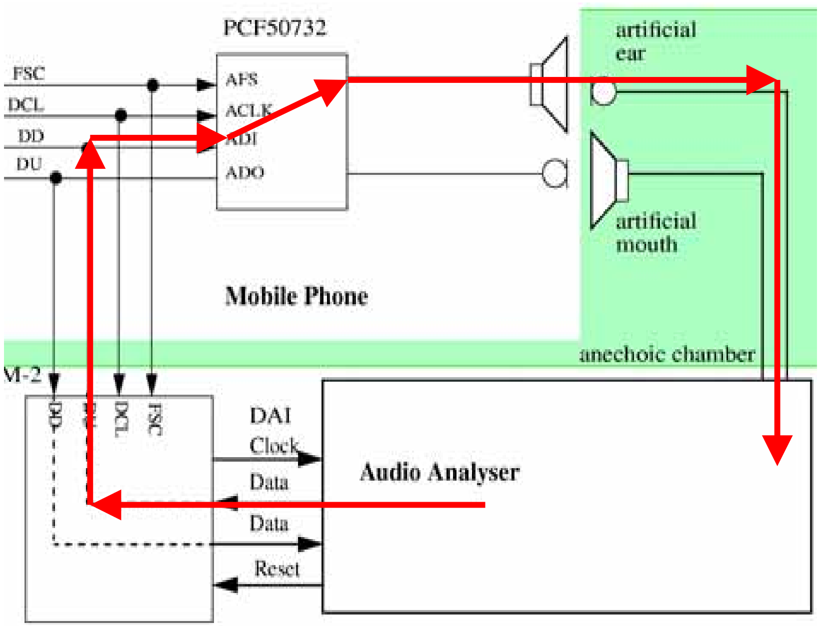
-The most important is to use a very quiet acoustic room.

Test Cases in FTA for Audio



Receiving Distortion GSM11.10, 30.7.2

The distortion must lower than limits (but the measured curve must be above the limit curve)



The analyzer generates on its digital output a sine signal with a frequency of 1015Hz at a ARL-level

The measurement is done with 7 levels relative to ARL

PHILIPS

Test Cases in FTA for Audio

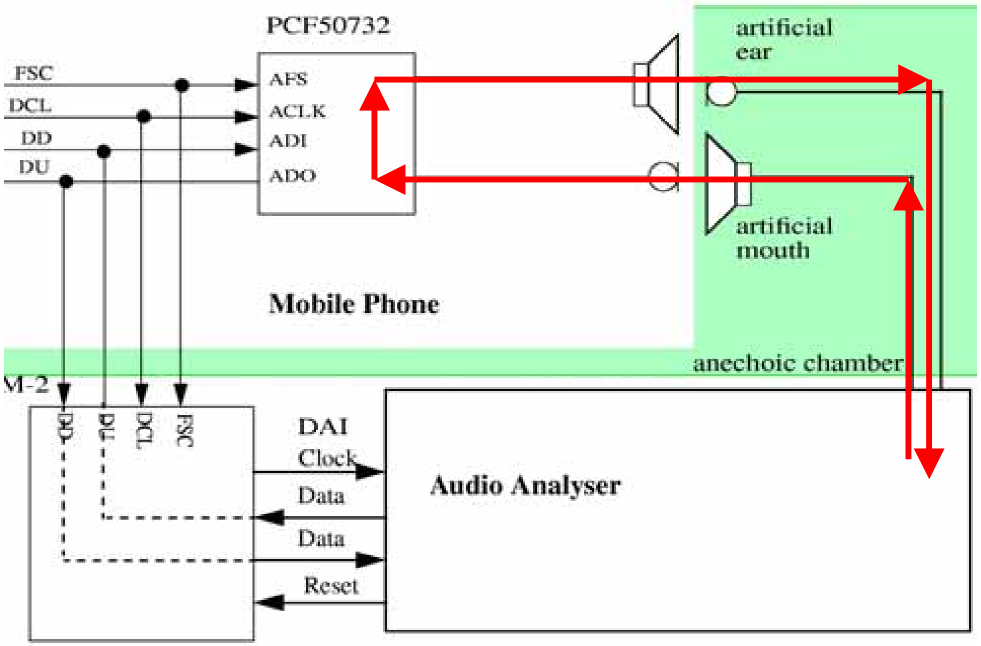
Sidetone Distortion GSM11.10, 30.8

The 3rd harmonic distortion must be smaller than 10% @ 3 frequencies

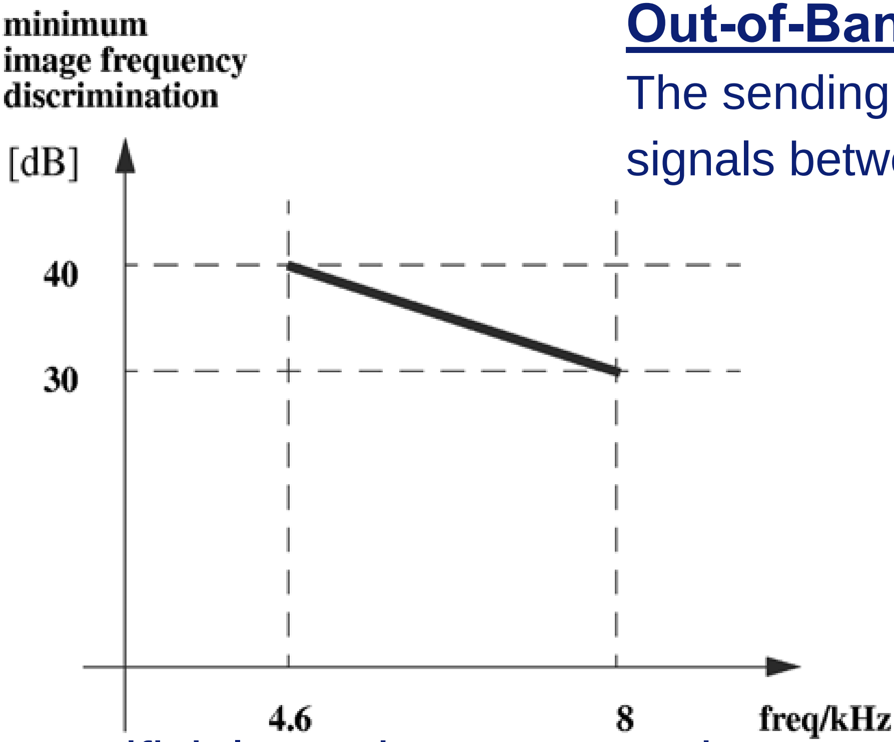
Inside the phone the mic-signal is feed back to the speaker-signal.

This path is called sidetone path.

The distortion is measured @ 315 Hz, 500 Hz, 1kHz

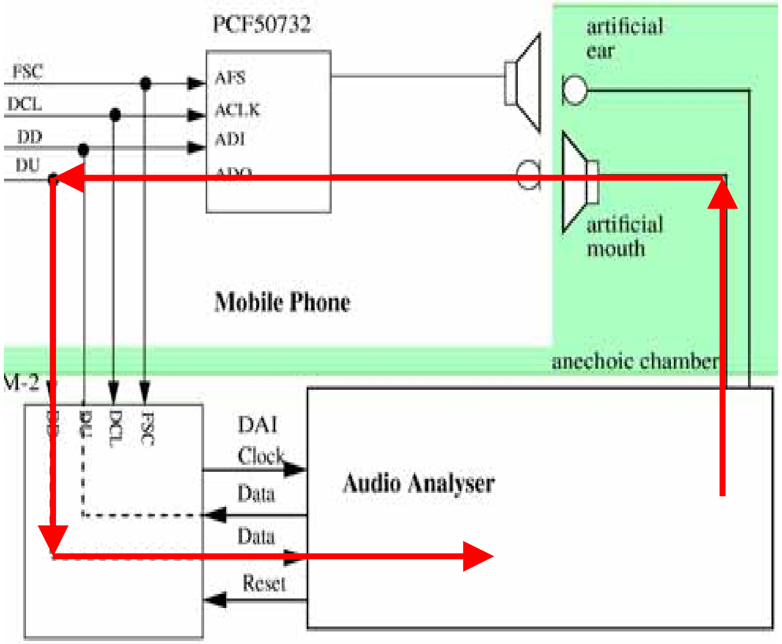


Test Cases in FTA for Audio



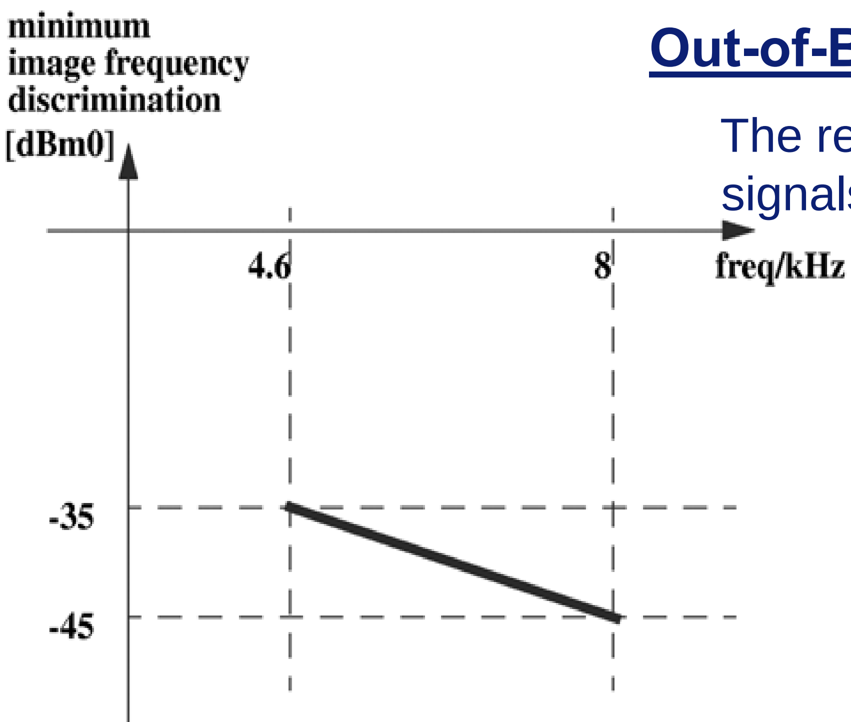
Out-of-Band-Signals Sending GSM11.10, 30.9.1

The sending path has to be insensitive for spurious signals between 1/2 sample rate and sample rate



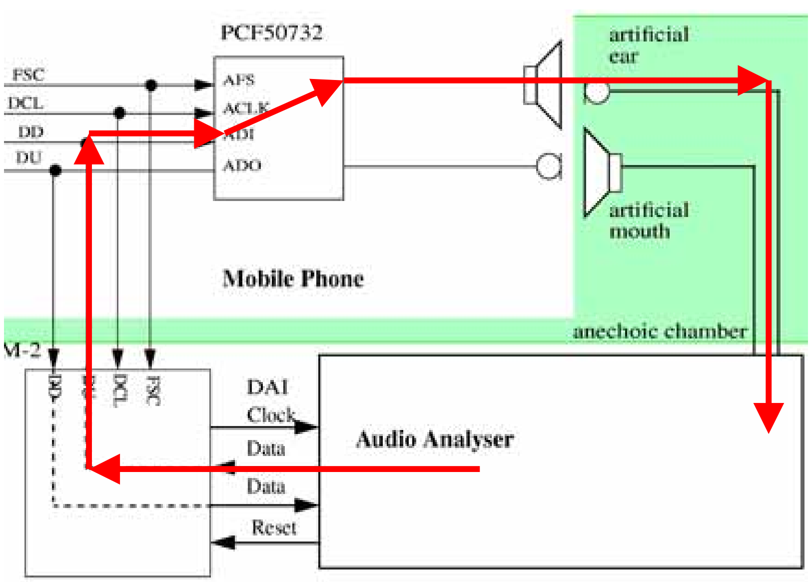
The artificial mouth generates sine signal with 1kHz and -4.7dBPa. The level of spurious "out-of-band signals must be lower than the upstanding curve.

Test Cases in FTA for Audio



Out-of-Band-Signals Receiving GSM11.10,30.9.2

The receiving path has to be insensitive for spurious signals between 1/2 sample rate and sample rate



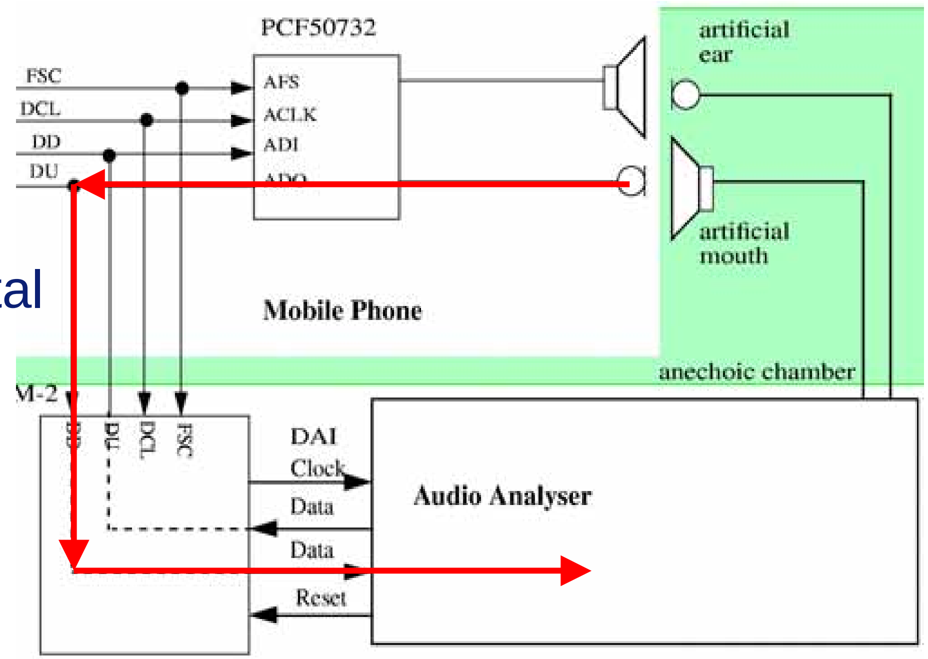
The analyzer generates on its digital output a sine signal of 1kHz.
The level of spurious “out-of-band” signals must be lower than the upstanding curve.

Test Cases in FTA for Audio

Idle Channel Noise Sending GSM11.10, 30.10.1

The idle noise must be smaller than -64dBm0p

The sending signal is measured in a quiet environment.
The anechoic chamber must be isolated very good from environmental Noise.
The analyzer does not produce any Signal.

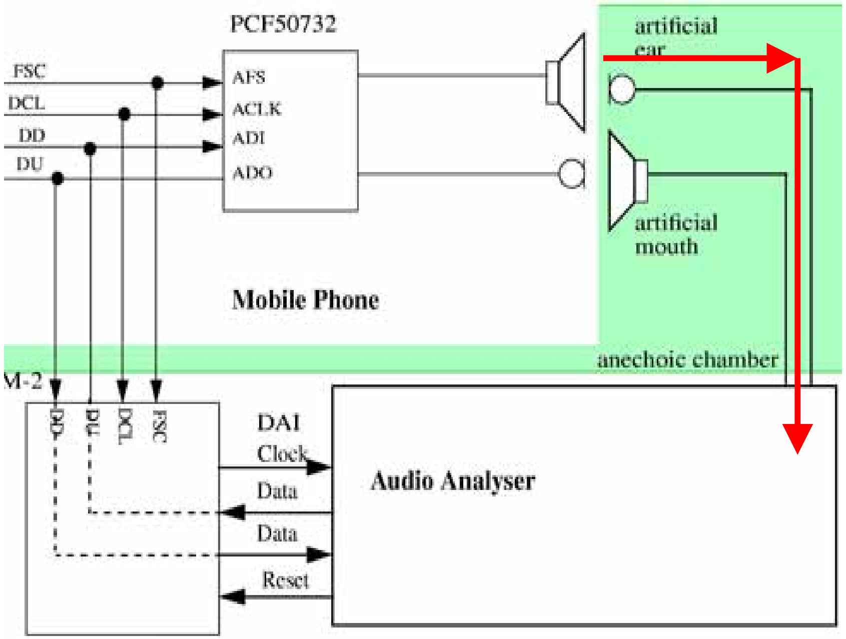


Test Cases in FTA for Audio

<u>Idle Channel Noise Receiving GSM11.10, 30.10.2</u>

The idle noise must be smaller than -54dBPa @ maximum volume setting, -57dBPa @ nominal volume setting

- The receiving signal is measured in a quiet environment.
- The anechoic chamber must be isolated very good from environmental noise.
- The analyzer does not produce any signal.
- The volume of the phone is set to a nominal volume setting which is normally an adjustment in the middle and to maximum.



PHILIPS

- Audio in GSM Mobile Phone
- Hardware Components in Audio Path
- HW Structure of PCF50732
- VSP in PCF50732
- Audio Firmware in R.E.A.L DSP
- Acoustic Test Bench
- Test Cases in FTA for Audio
- How to Tune the Audio
- TDMA Noise

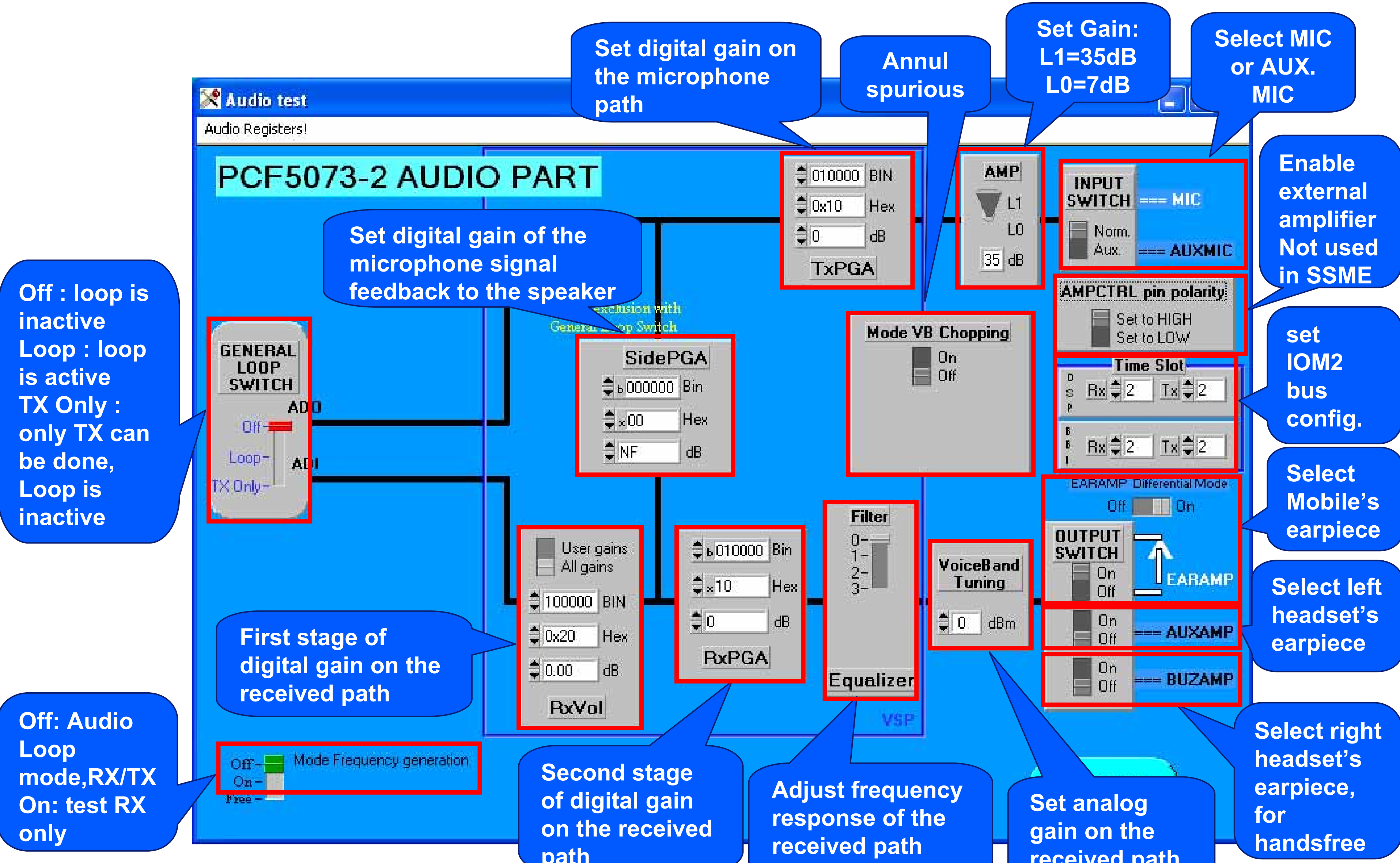
How to Tune the Audio

Main Purposes

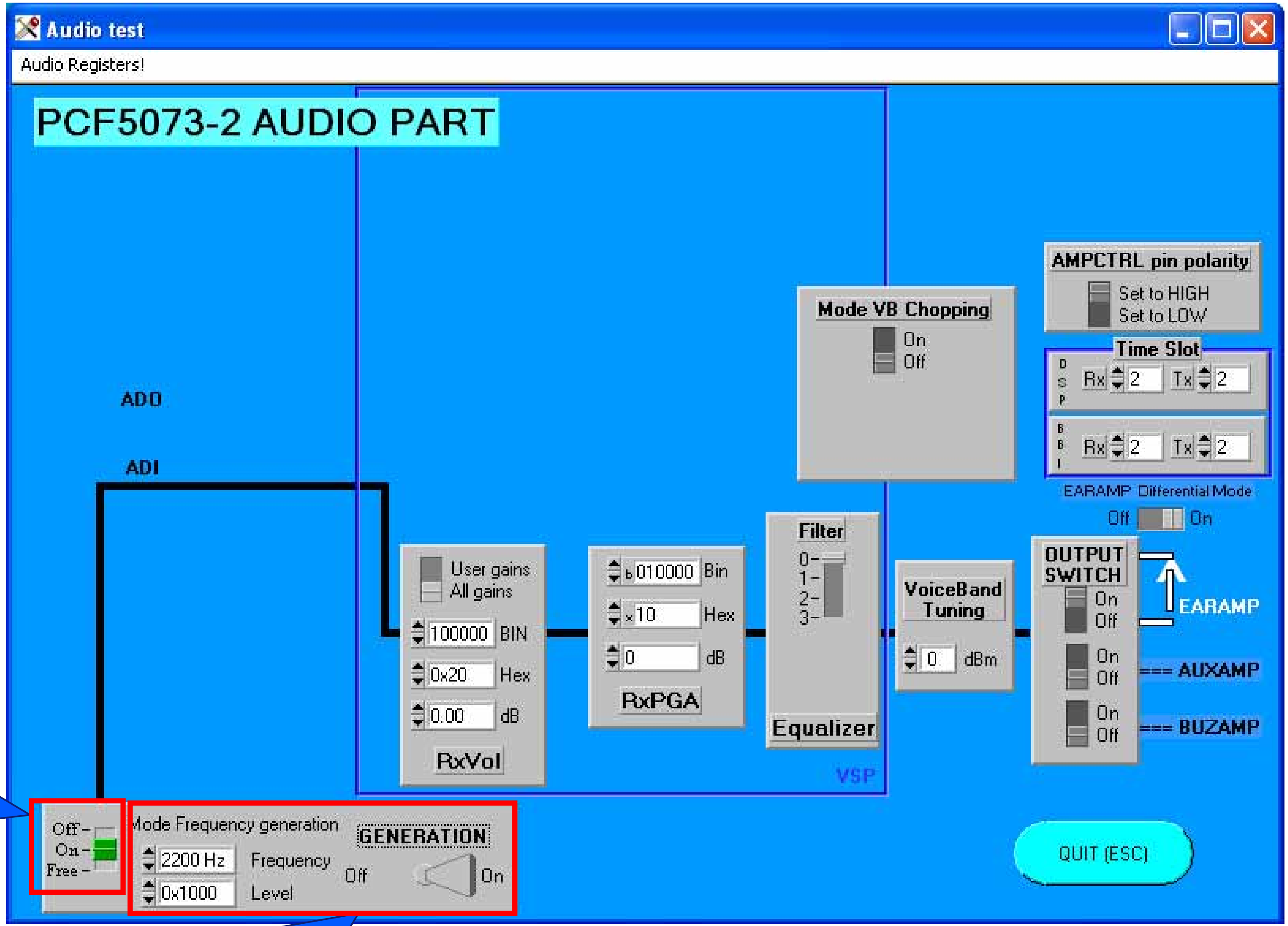
- the phone has to fullfill the GSM-requirements
- the phone has to fullfill subjective criterias which are defined by the final users

Sometimes it is not easy to fullfill both requirements !!

How to Tune the Audio TAT Audio Loop: Main Window



How to Tune the Audio Mode Frequency generation

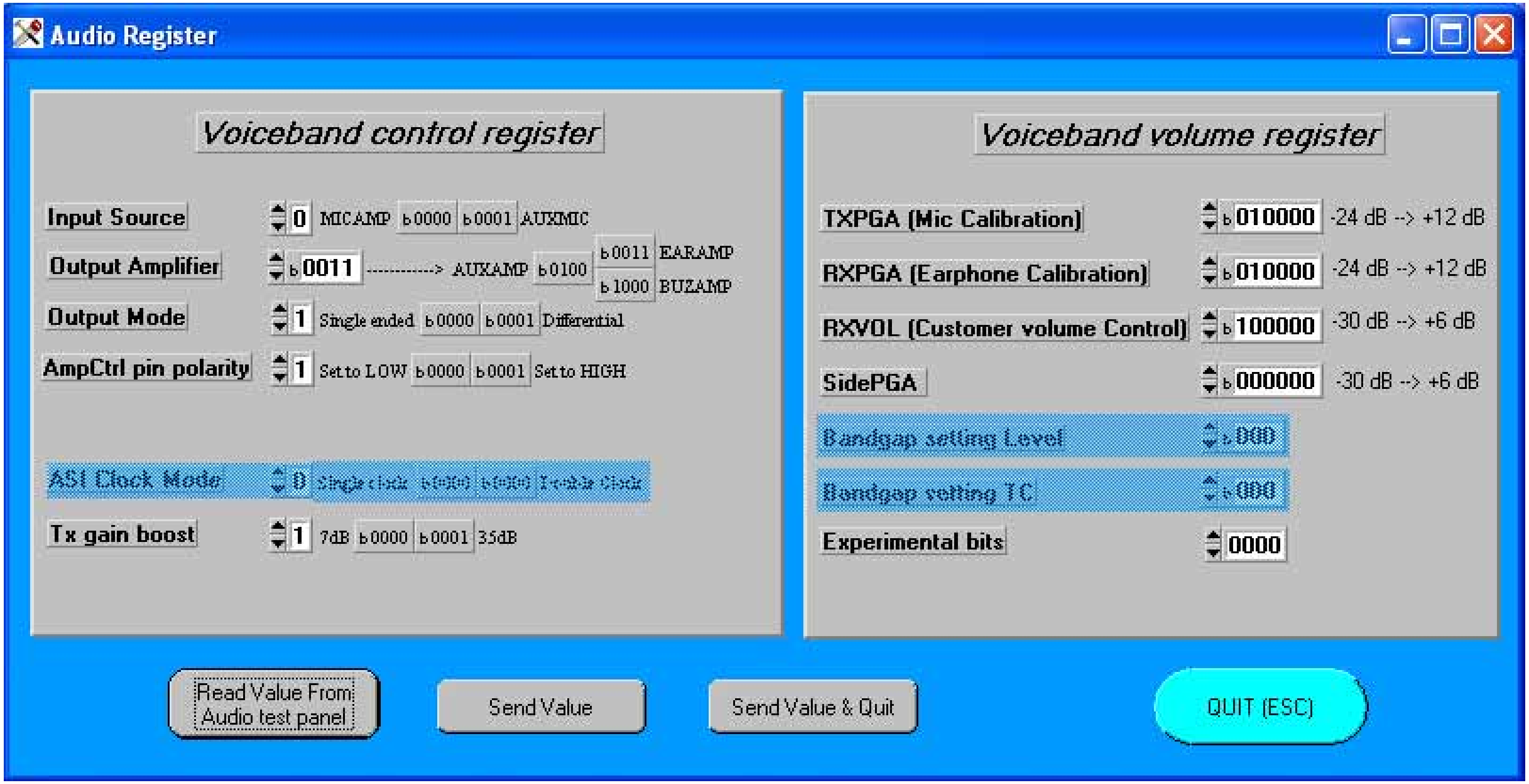


Test the received audio path

Generate a sound: Select Frequency, Level, Place GENERATION “On”

How to Tune the Audio

TAT Audio Loop: Audio Registers



How to Tune the Audio Audio Data Section

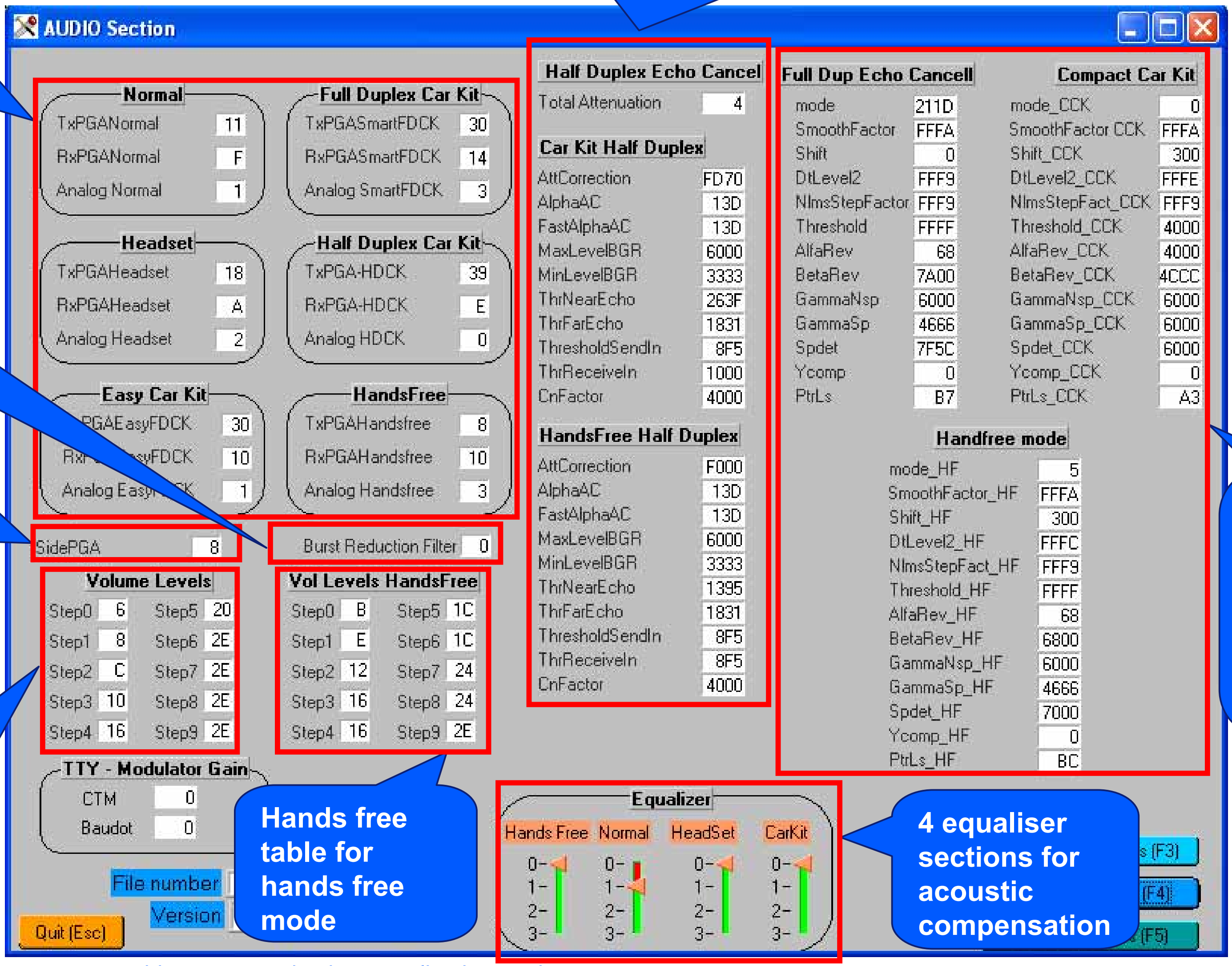
Three sections for Half duplex echo suppresser

set audio gains specific to each modes

reduce the 217 Hz harmonics on mic. Path

Side PGA: only for normal and headset mode

Volume level table for modes: normal, headset, different Car Kit



Three sections for Acoustic Echo canceller algorithm (full duplex)

Hands free table for hands free mode

4 equaliser sections for acoustic compensation

How to Tune the Audio

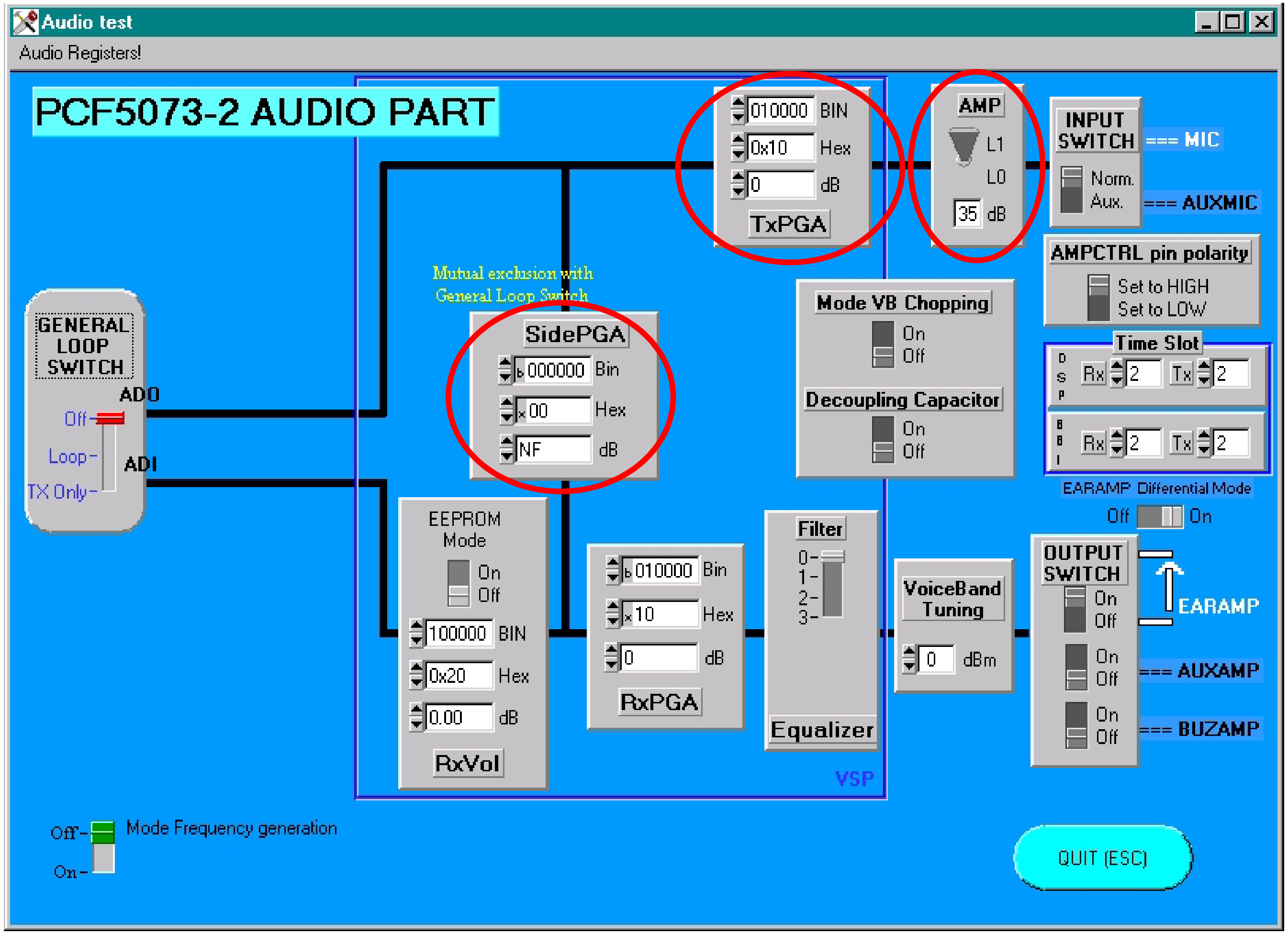
1. Sending Loudness Alignment

Pre-conditions

SidePGA set to 0

Do test 30.2
until loudness
reaches the
lowest allowed limit

Vary TxPGA
and switch AMP



How to Tune the Audio

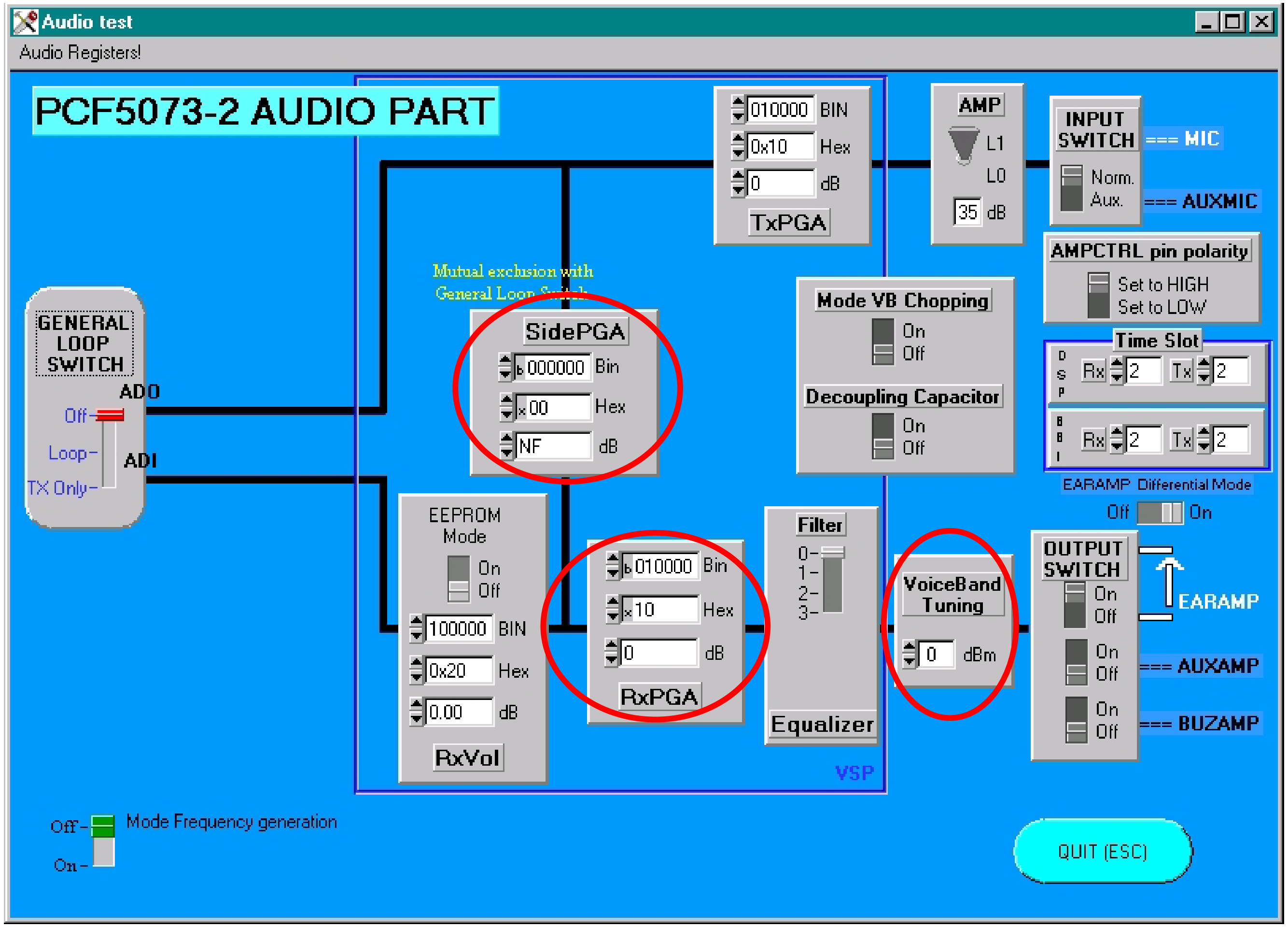
2. Receiving Loudness Alignment

Pre-conditions

SidePGA set to 0

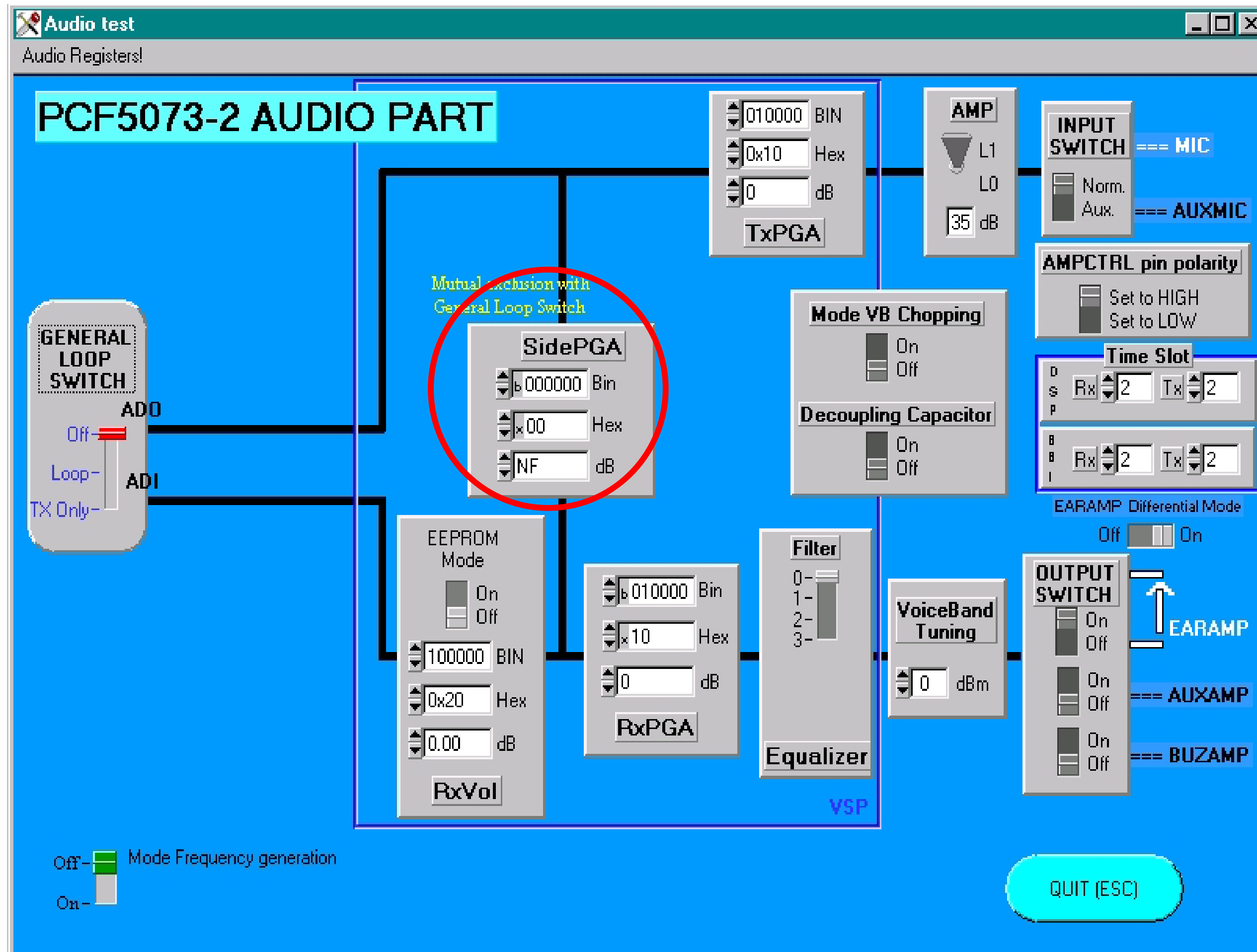
Do test 30.4 until loudness reaches the lowest allowed limit

Vary RxPGA and switch VoiceBand-Tuning



How to Tune the Audio

3. Receiving Frequency alignment



Pre-conditions

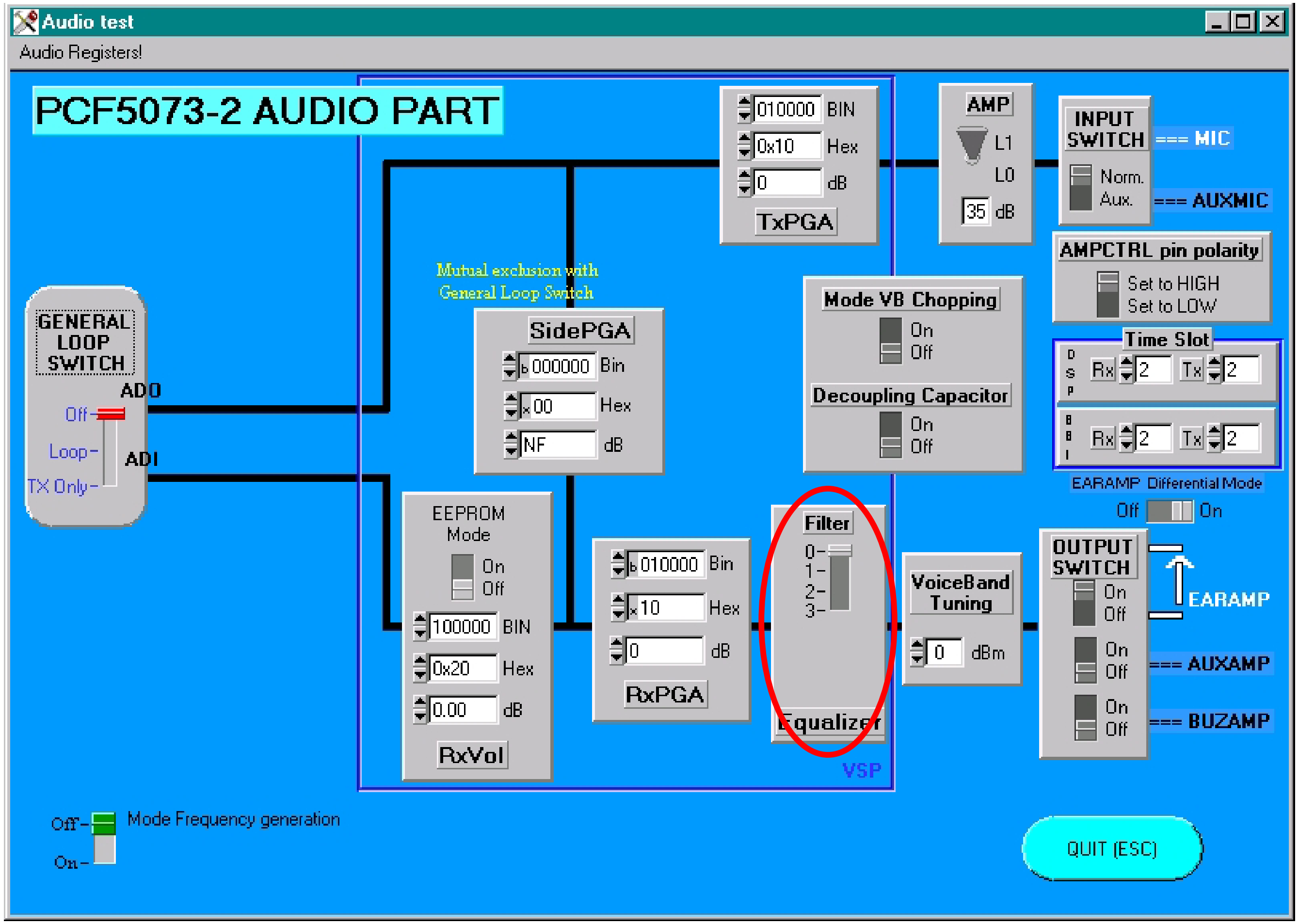
SidePGA set to 0

Do test 30.3
until loudness
reaches the
lowest allowed limit

Vary SidePGA

How to Tune the Audio

4. Sidetone Loudness Alignment



Pre-conditions

Do test 30.5.1 until frequency repsonse fits to the masks

Vary Filter

- Audio in GSM Mobile Phone
- Hardware Components in Audio Path
- HW Structure of PCF50732
- VSP in PCF50732
- Audio Firmware in R.E.A.L DSP
- Acoustic Test Bench
- Test Cases in FTA for Audio
- How to Tune the Audio
- TDMA Noise

TDMA Noise

The main work for a transceiver audio part is to assure a good 217 Hz and harmonics (TDMA noise) immunity; this is not controlled by above audio tests.

TDMA Noise: The 217 Hz harmonics components come from the radio part. When the radio Power Amplifier is activated, it generates 900 MHz or 1800 MHz signal during 1/8 of time each 4.615 ms (1 time slot, without considering GPRS feature). The 4.615 ms correspond to 216.684 Hz (Called 217 Hz, also called TDMA noise)

TDMA Noise can transmit to the audio part by three ways.

- Through the supplies
- Demodulation effect of component through different paths on the board(track coupling)
- Demodulation effect of component through the radio air interface (from the antenna to microphone for example)

Following audio test is able to check the immunity to the first two phenomenon mentioned above

TDMA Noise

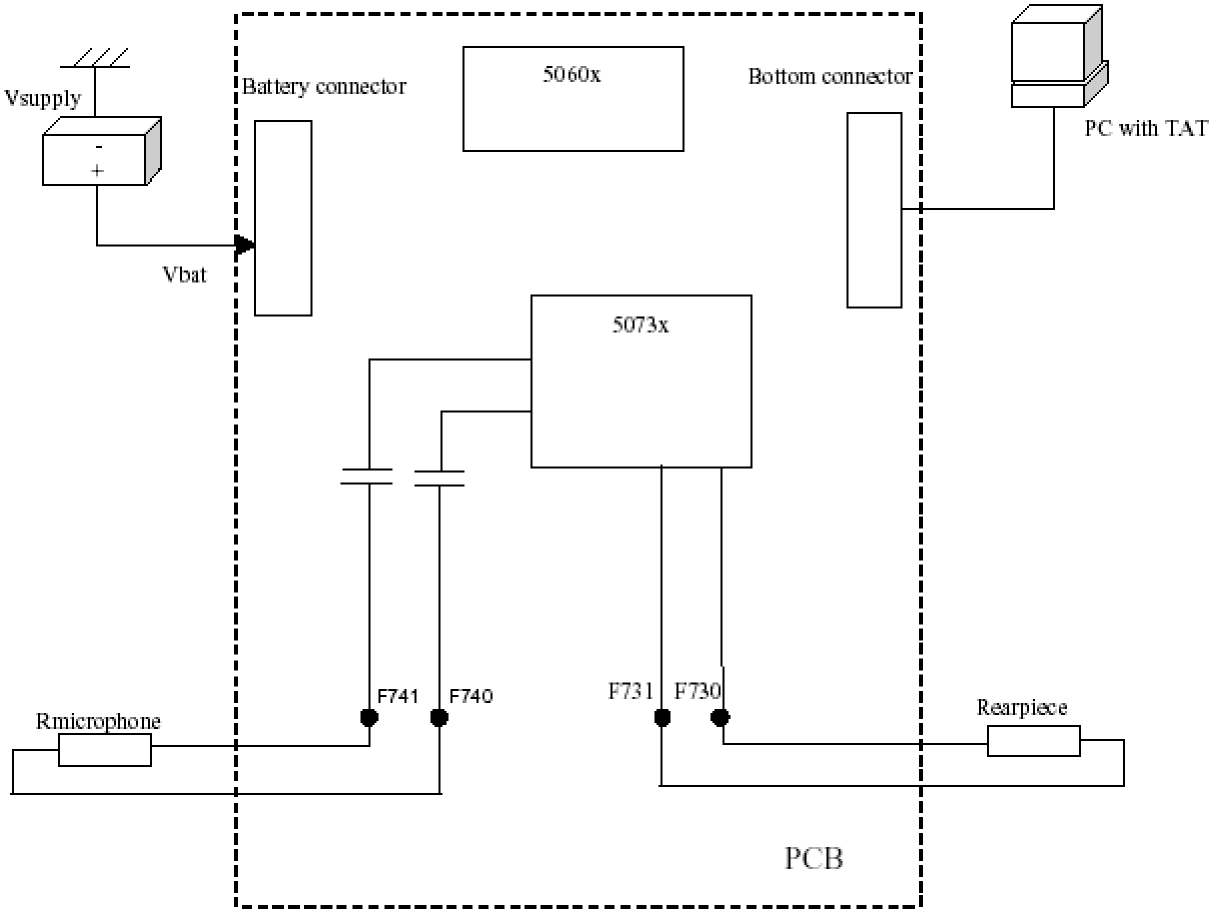
Test equipments

To perform this test the following equipment or equivalent is required

- TAT software
- Dynamic signal analyzer HP 35670A (audio spectrum analyzer able to scan a frequency band width from 0 to 100KHz min)
- Power supply (Fluke PM2813) or 66319B (Hewlett Packard)
- Audio analyser 10 Hz-100kHz UPA3 (Rodhe & Schwartz)

TDMA Noise

Test Setup



PHILIPS

TDMA Noise

Audio loop mode

- Set a 2.2Kohms instead microphone and a 16 Ohms resistor instead the speaker
- Activate HFAVDD to 3.05 Volts (regulator panel)
- Set the GPIO2 to 0 (Sys. Ctrl→GPIO/GPON/PWM panel)
- Set the following Gain Values (Audio panel):
  Gtx=35dB (MICHI=35dB,TXPGA=0 dB); Grx=0dB (RXVOL=RXPGA=RXANA=0dB)
- Switch on the general switch (audio loop mode)
- Activate the radio part (max power: Level5 for GSM and level0 for DCS)

A spectrum analysis has to be performed and the 217Hz fundamental and harmonics 2,3 and 4 levels but also the noise floor level (at 800Hz between two harmonics) have to be measured twice (once in GSM mode and once in DCS mode) on :

-VBAT

-Between MICN and MICP (F740-F741)

-Between EARN and EARP (F730-F731)

TDMA Noise

Measurement result

- Measurements in GSM mode:	217Hz	H2	H3	H4	Noise Level 800 Hz	Limit (max) dBV
VBAT						Around -20
Between MICAMP N & P (F740-F741)						-102
Between EARP1 and EARP2 (F730-F731)						-85

- Measurements in DCS mode:	217Hz	H2	H3	H4	Noise Level 800 Hz	Limit (max) dBV
VBAT						Around -20
Between MICAMP N & P (F740_F741)						-102
Between EARP1 and EARP2 (F730_F731)						-85

TDMA Noise

RX path only

Only the RX part is tested

- Set a 16 Ohms resistor instead the speaker
- Set the following Gain Values (audio panel) :

  Grx=0dB (RXVOL=RXPGA=RXANA=0dB)
- Activate HFAVDD to 3.05 Volts (regulator panel)
- Set GPIO2 to 0 and select normal output for Normal mode measurement

  or set GPIO2 to 1 and select Buz output for LM 4877 outputs measurement for hands free mode
- Select the mode frequency generation (f =none, level=1000) and activate the switch.
- Activate the radio part (max power)

A spectrum analysis has to be performed and the 217 Hz fundamental and harmonics 2 3 and 4 levels but also the noise floor level (at 800Hz between two harmonics) have to be measured twice (once in GSM mode and once in DCS mode) on:

-Between EARN and EARP (F730-F731)

TDMA Noise

Measurement result

-Measurements in GSM mode:	217Hz	H2	H3	H4	Noise Level 800 Hz	Limit (max) dBV
EARP1 and EARP2 Normal mode(F730-F731)						-105
EARP1 and EARP2 Hands free mode(F730-F731)						-65

-Measurements in DCS mode:	217Hz	H2	H3	H4	Noise Level 800 Hz	Limit (max) dBV
EARP1 and EARP2 Normal mode(F730-F731)						-105
EARP1 and EARP2 Hands free mode(F730-F731)						-65

If the results are above the indicated limits the issue can come from:

- Power supply rejection

- Demodulation phenomenon

To suppress these noise need fine tuning and also depends on different layout and shielding etc., maybe require a mechanic and is the customer's liability

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