Responsible Author: Rainer Dietsch

Issue: 1.0

DAI Acceptance Test Specification

G99 WP13 DAI Prototype

Issue: 1.0

List of Updates

Issue	Date	Updated Paragraphs	Description
0.1 Draft	20-Sep-1999	all	document creation
1.0 p	19-Oct-1999	1,2,3,8	document prepared for review
1.0	4-Nov-1999	all	inputs from cross check by Robert Weidenhöfer

1. Introduction

1.1. Purpose

A coarse introduction into audio tests is given in this document. The acceptance tests for the DAI are derived from the tests required by GSM recommendation.

1.2. Scope

This document specifies tests to prove the functionality of the digital audio interface (DAI) hardware and software.

1.3. Organization of the Document

After giving an overview of the DAI in chapter 5 , the audio tests required by GSM recommendations are described in 6 . Chapter 7 summarizes for which of these tests the DAI is needed. Finally chapter 8 specifies a few tests which prove the functionality of the DAI.

2. General

2.1. Distribution

version	Distribution
1.0p	Robert Weidenhöfer, Thomas Richter
1.0	

2.2. Applicability

This paper is meant for internal usage in TCMC only.

3. Abbreviations

ARL	acoustic reference level
DAI	digital audio interface
DD	data down
DSP	digital signal processor
DU	data up
EL	echo loss
EVITA	Evaluation Verification Integration Test and Application Platform
FTA	final type approval
GSM	global system for mobile communication
ITU	International Telecommunication Union
LSTR	listener side tone rating
R.E.A.L.	Reconfigurable Embedded Architecture (name of a DSP architecture)
RLR	receiving loudness rating
RX	receive
SLR	sending loudness rating
STMR	side tone masking rating
TAL	telephone acoustic coupling loss
TX	transmit

4. References

[1] GSM Recommendation 11.10-1, version 4.24.0, December 1998

[2] ITU-T Recommendation P.79, Calculation of loudness ratings for telephone sets

[3] ITU-T Recommendation P.64, Determination of sensitivity/frequency characteristics of local telephone systems

[4] ITU-T Recommendation G.223, Assumptions for the calculation of noise on hypothetical reference circuits for telephony

5. DAI Overview

Figure 1 shows how the DAI is connected to a mobile phone and to the GSM system simulator.

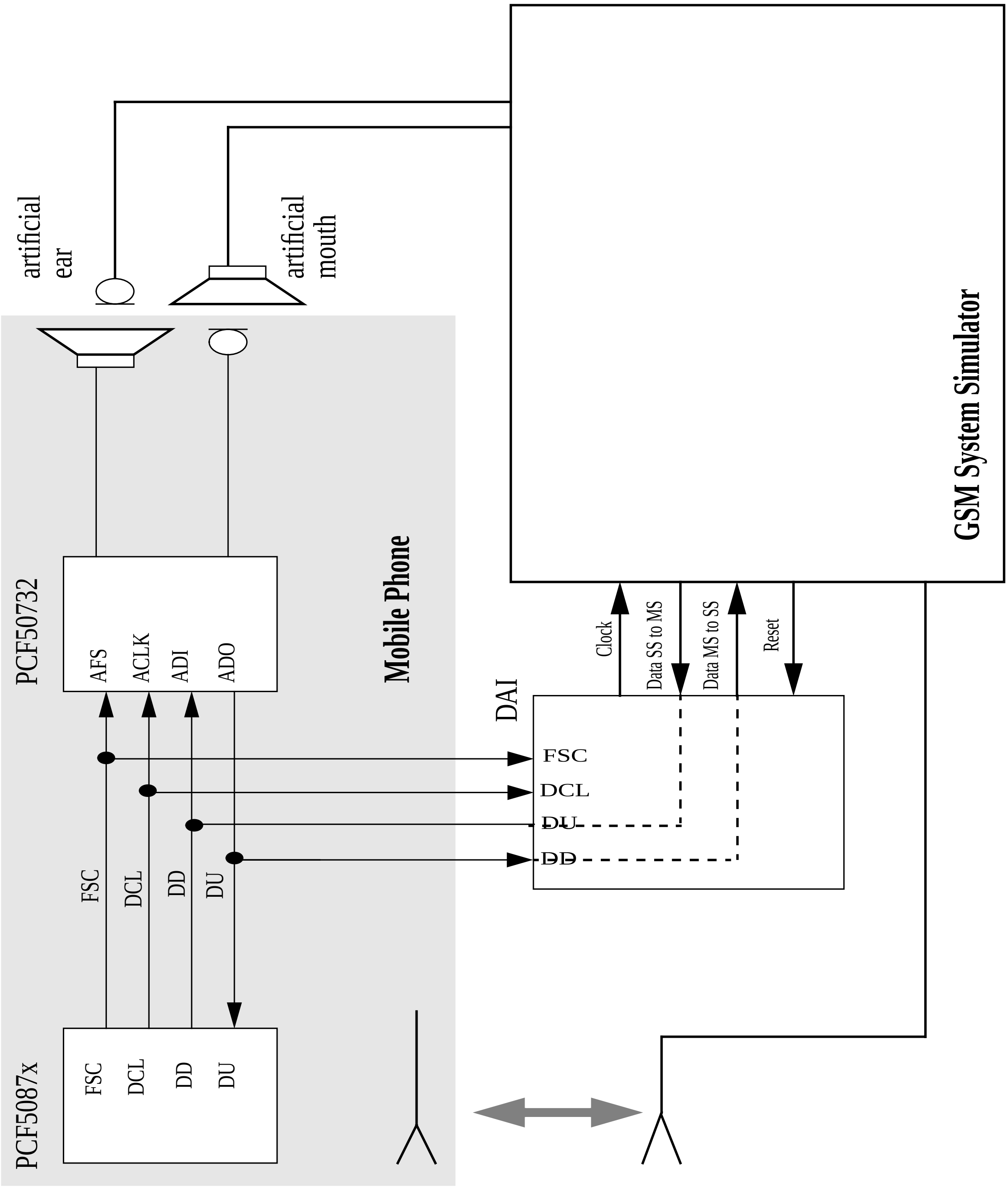

Figure 1: DAI application diagram

Figure 2 gives a top level diagram of the DAI itself.

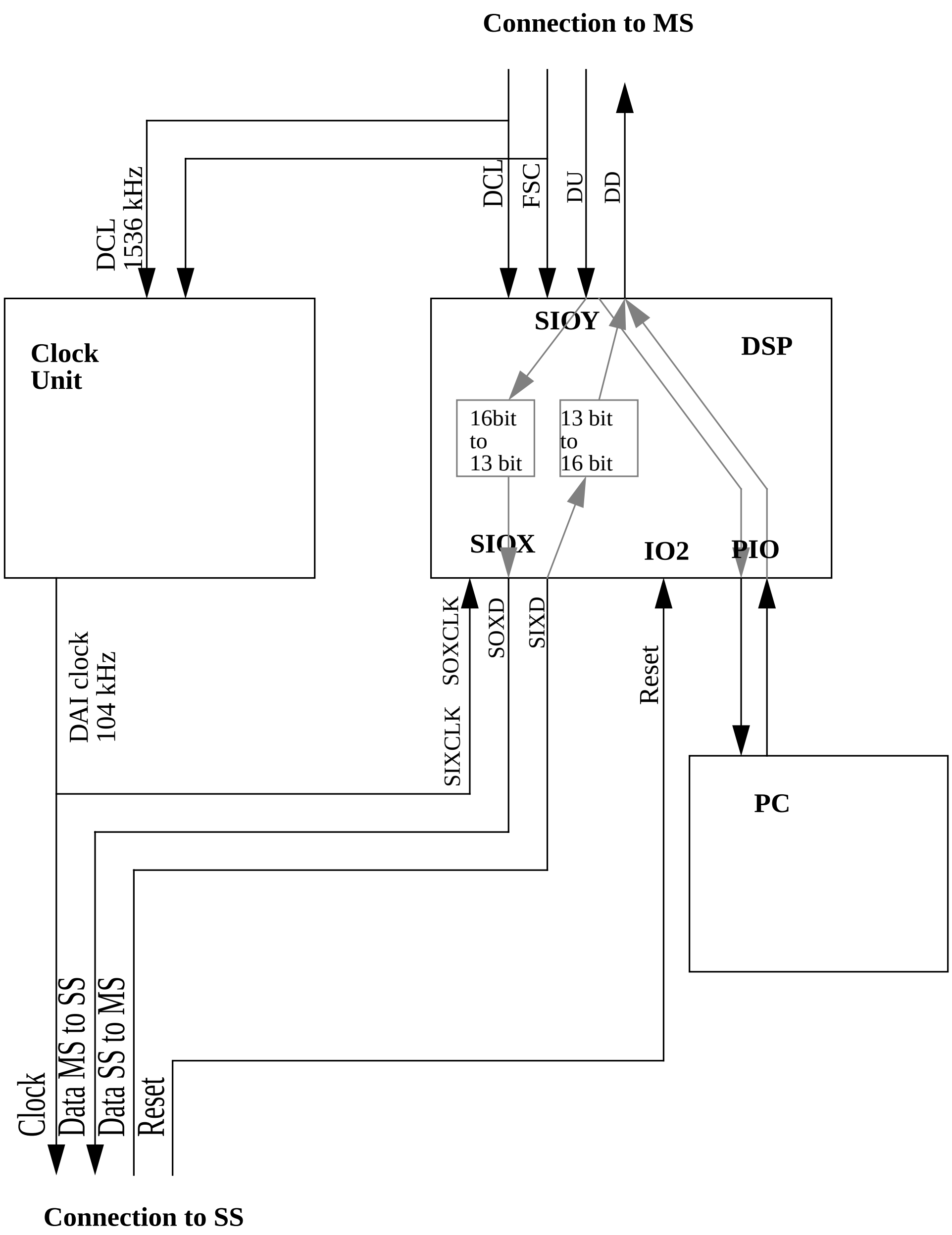

Figure 2: DAI top level diagram

6. Audio tests required by GSM Recommendation 11.10

This chapter describes the tests specified in GSM recommendation 11.10, chapter 30 Speech teleservices. These are the audio tests performed on a mobile at type approval.

6.1. Sending Sensitivity/Frequency response

The sending sensitivity / frequency response is to guarantee that the microphone path frequency response is within an acceptable range.

Test setup

- The handset is mounted to an artificial head.
- A pure sine tone (-4.7 dBPa sound pressure) is applied at the artificial mouth.
- The digital output power is measured.

Figure 3 shows the signal flow for this test.

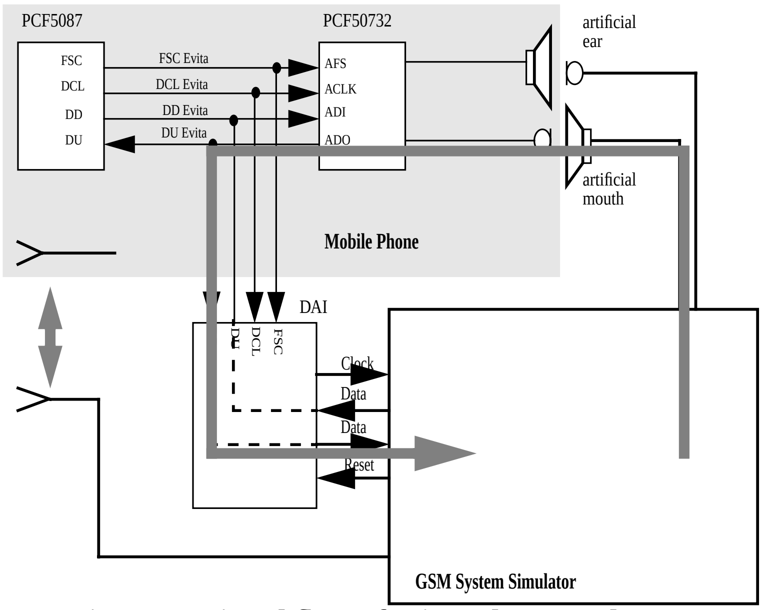

Figure 3: Signal flow of microphone path tests

Test procedure

The test is performed for tones in the range of 100 Hz to 4000 Hz.

Test requirements

Figure 4 shows the required output levels.

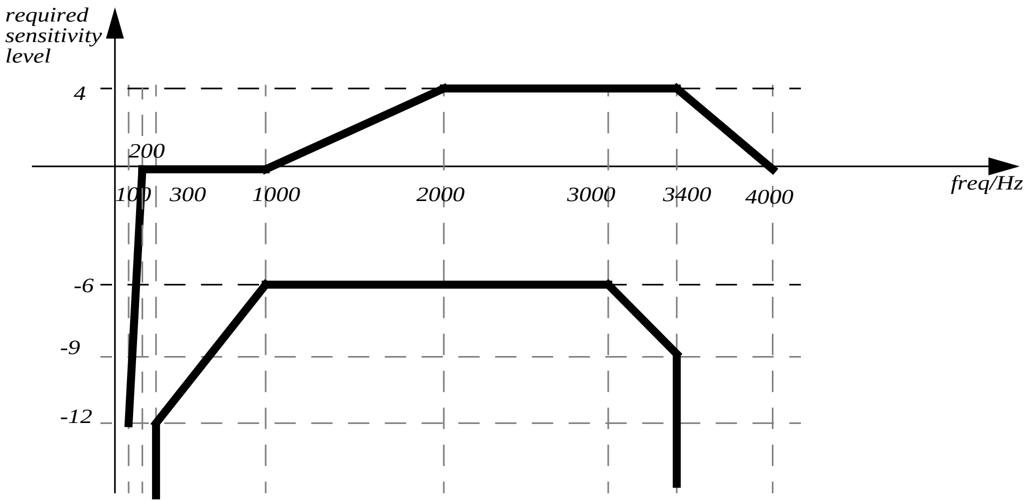

Figure 4: Sending frequency response requirements

6.2. Sending Loudness Rating

The sending loudness rating is to guarantee that the gain of the microphone path is within the defined range.

Test setup
- The handset is mounted to an artificial head.
- A pure sine tone (-4.7 dPa sound pressure) is applied at the artificial mouth.
- The digital output power is measured.

The signal flow is shown in figure 3 .

Test procedure
The test is performed for tones in the range of 100 Hz to 4000 Hz. One SLR value is calculated as described in[2].

Test requirements
The SLR value must be 8+/-3dB, i.e. 5dB < SLR < 11dB.

6.3. Receiving sensitivity / frequency response

The receiving loudness rating is to guarantee that the frequency response of the loudspeaker path is within the defined limits.

Test setup
- The handset is mounted to an artificial head.
- A pure digital sine tone (-16 dBm0) is applied to the digital interface.
- The analog output power is measured.

Figure 5 shows the signal flow for this test.

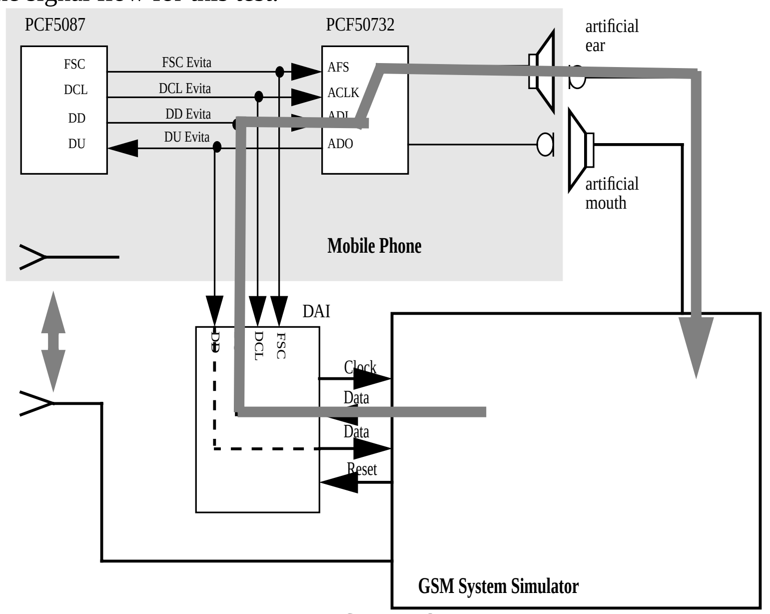

Figure 5: Signal flow of loudspeaker path tests

Test procedure
The test is performed for tones in the range of 100 Hz to 4000 Hz.

Test requirements
Figure 6 shows the required output levels.

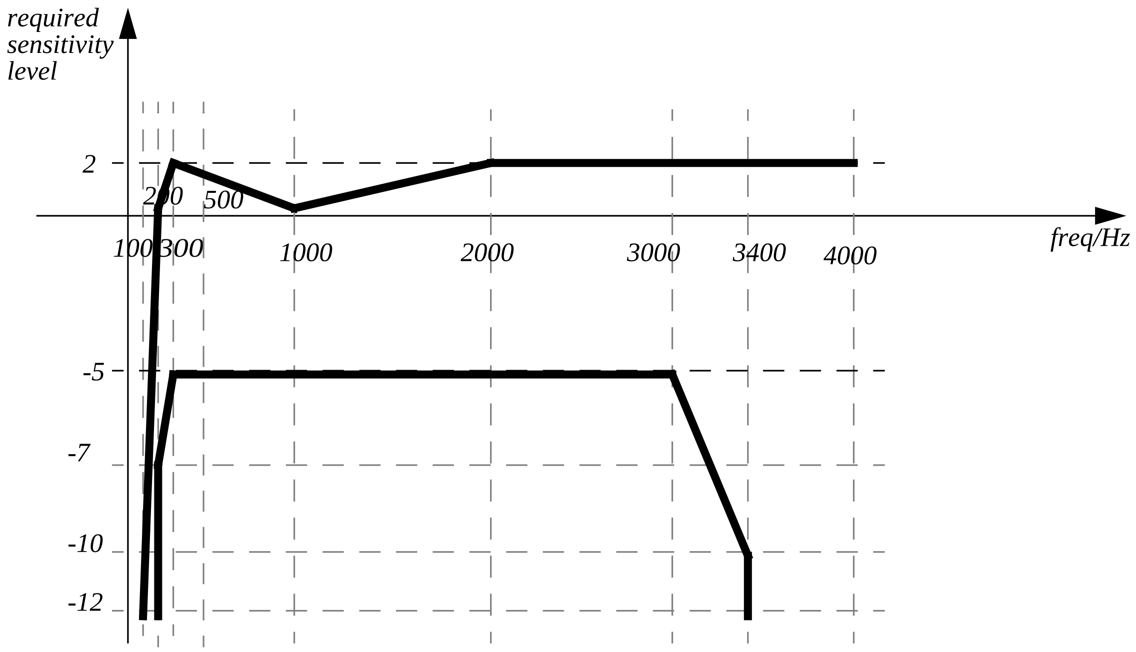

Figure 6: Sending frequency response requirements

6.4. Receiving loudness rating

The sending loudness rating is to guarantee that the gain of the microphone path is within the defined range.

Test setup
- The handset is mounted to an artificial head.
- A pure digital sine tone (-16 dBm0) is applied at the digital interface.
- The acoustic output power is measured.

The signal flow of this test is like shown in figure 5 .

Test procedure
The test is performed for tones in the range of 100 Hz to 4000 Hz.

Test requirements
For at least one volume setting the RLR value must be 2 +/-3dB, i.e. -1dB < RLR < 5dB. For the maximal volume setting RLR shall not be less than -13dB, i.e. not louder than -13dB.

6.5. Side Tone Masking Rating (STMR)

The STMR defines the talker sidetone.

Test setup
- The handset is mounted to an artificial head.
- A pure sine tone (-4.7dBPa) is applied at the artificial mouth.
- The analog output power is measured at the artificial ear while no receive signal is applied.

Figure 7 shows the signal flow of this test.

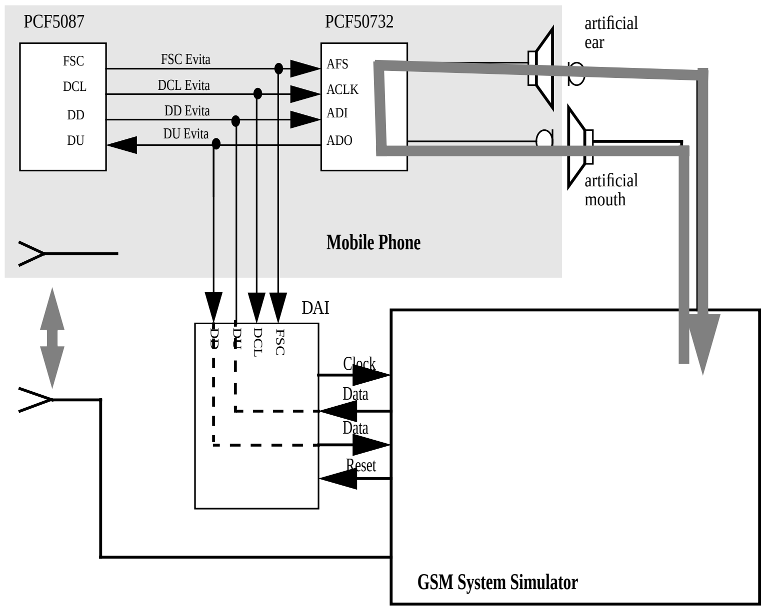

Figure 7: Signal flow of sidetone tests

Test procedure
The test is performed for tones in the range of 100 Hz to 4000 Hz.

Test requirements
The STMR value must be 13+/- 5dB for the reference volume value, i.e. for that volume setting which is used for the RLR measurement.

6.6. Listener Side Tone Rating (LSTR)

The LSTR defines how the background noise is perceived.

Test setup
- The handset is mounted to an artificial head.
- A pink noise signal with 70 dBA (-24dBPa(A)) is applied at the artificial mouth.
- The analog output power is measured at the artificial ear while no receive signal is applied.

The signal flow diagram for this test is shown in figure 7 .

Test procedure
The test is performed for one-third octave bands in the range of 200 Hz to 4000 Hz.

Test requirements
The LSTR value must not be less than 15dB.

6.7. Echo loss (EL)

Echo loss is one of two measurements concerning the ‘Telephone acoustic coupling loss’ (TAL). It checks the acoustic echo from loudspeaker to microphone.

Test setup
- The handset is mounted to an artificial head.
- The volume control of the mobile is set to the maximum value.
- A male / female artificial speech signal is applied as receive signal.
- The analog signal power of the decoded transmit signal is measured at the system simulator while no additional transmit signal is applied at the artificial head.

The signal flow of this test is shown in figure 8 .

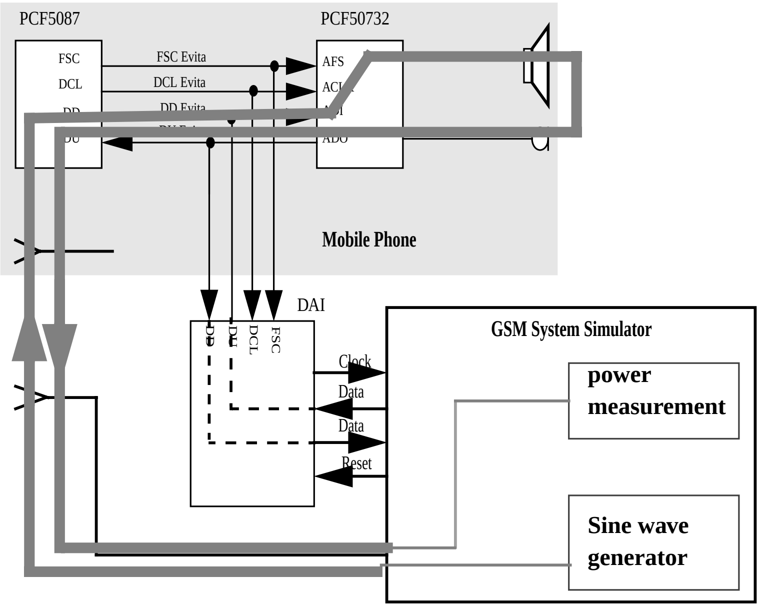

Figure 8: Signal flow of echo loss test

Test procedure
The test is performed for male and for female artificial speech. The results are averaged.

Test requirements
The EL value must be at least 46dB.

6.8. Stability margin

The stability margin measurement is the second measurement concerning the ‘Telephone acoustic coupling loss’ (TAL).

Test setup

- The handset is placed on a hard plain surface, loudspeaker and microphone phasing the surface.
- The volume control is set to maximum.

The signal flow of this test is shown in figure 9 .

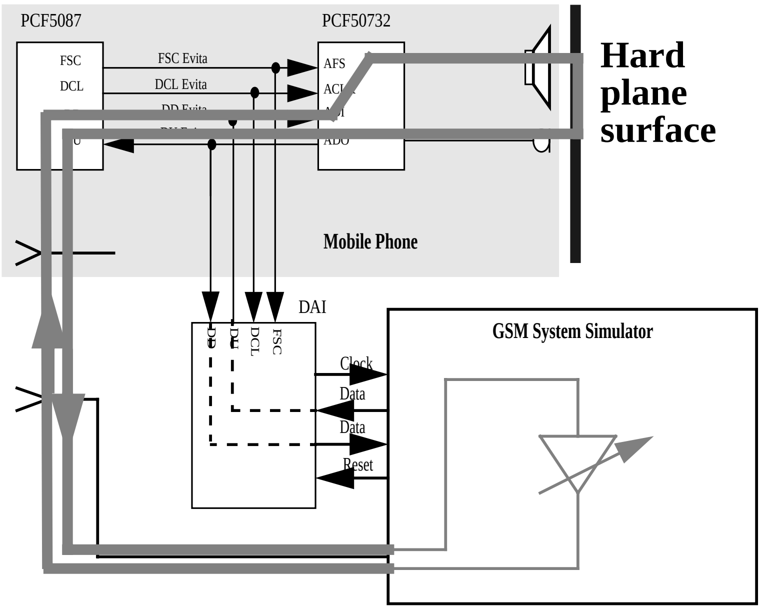

Figure 9: Signal flow of stability margin test

Test procedure
A gain is inserted into the loop. The gain is increased until instability occurs.

Test requirements
The stability margin shall be at least +6dB.

6.9. Sending distortion

Test setup

- The handset is mounted to an artificial head.
- A sine wave signal in the range 1004 Hz to 1025 Hz is applied at the artificial mouth.
- The signal to total distortion ratio is measured at the DAI while no receive signal is applied.
- Psophometric noise weighting is used

The signal flow of this test is shown in figure 3 .

Test procedure
Test signals at levels -35 dB to + 10 dB relative to the ARL (acoustic reference level) are applied

Test requirements
The level ratio shall always be higher than the curve shown in the following figure.

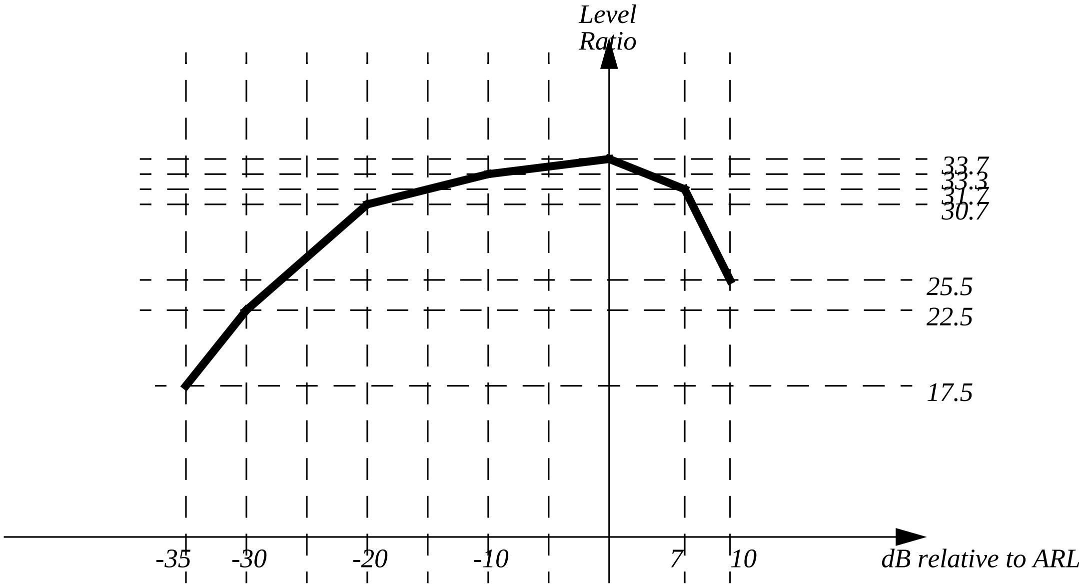

Figure 10: Sending distortion requirement

6.10.Receiving distortion

Test setup
- The handset is mounted to an artificial head.
- A sine wave signal in the range 1004 Hz to 1025 Hz is applied at DAI.
- The signal to total distortion ratio is measured at the artificial ear while no receive signal is applied.
- Psophometric noise weighting is used

The signal flow of this test is shown in figure 5 .

Test procedure
The test is carried out at sound pressures between -50dBPa and +10dBPa.

Test requirements
The receiving Signal to total distortion ratio shall be higher than the curve shown in figure 11

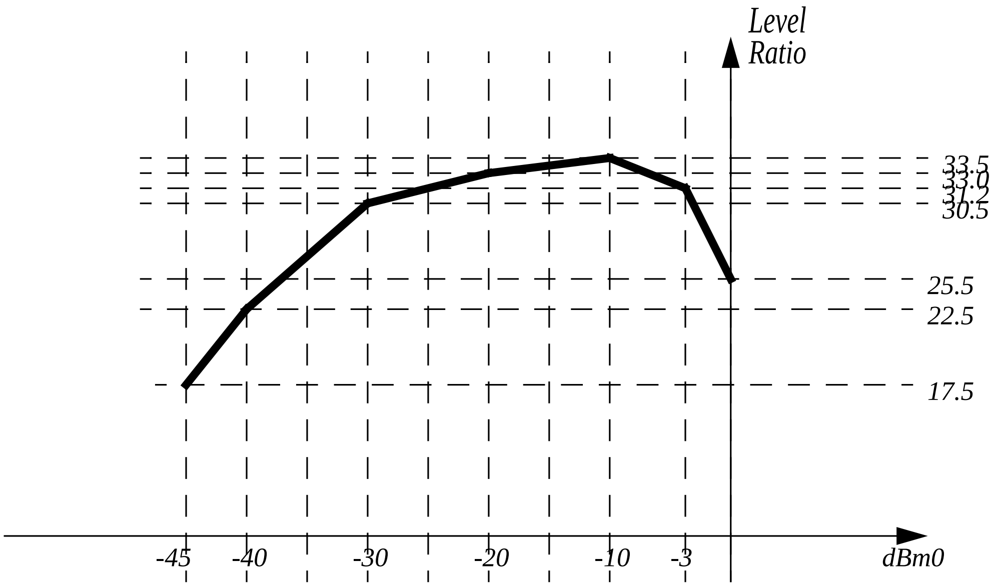

Figure 11: Receiving distortion requirement

6.11.Sidetone distortion

Test setup

- The handset is mounted to an artificial head
- A pure sine tone of -4.7 dBPa is applied at the artificial mouth.
- The third harmonic distortion is measured at the artificial ear.

The signal flow of this test is shown in figure 7 .

Test procedure
This test is performed for frequencies 315 Hz, 500 Hz and 1000 Hz.

Test requirements
The third harmonic distortion may not be greater than 10%.

6.12.Sending Out-of-Band Signals

Test setup

- The handset is mounted to an artificial head
- A pure sine tone of -4.7 dBPa is applied at the artificial mouth.
- The level of inband image frequencies is measured.

The signal flow of this test is shown in figure 3 .

Test procedure
This test is performed for input signals in the range of 4.6 kHz to 8 kHz.

Test requirements

The level of any image frequency must be below a reference obtained at 1 kHz by at least the amount shown in figure 12 .

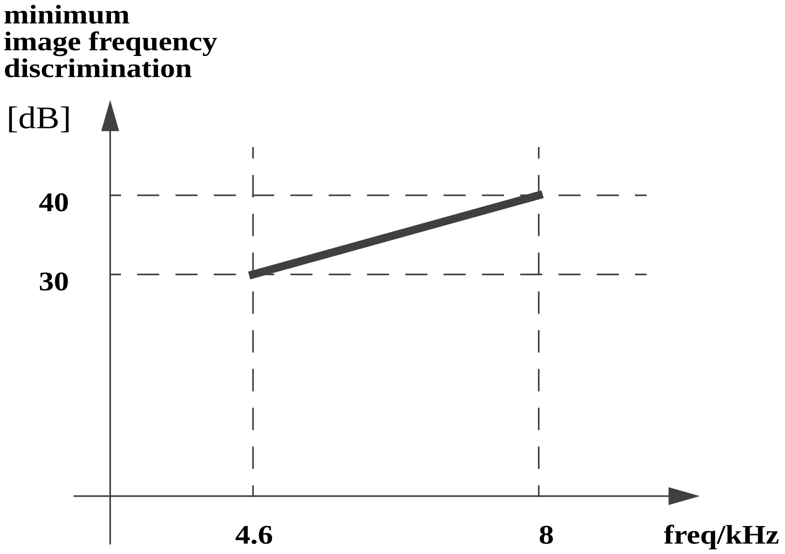

Figure 12: Sending out of band frequency rejection requirement

6.13.Receiving Out-of-Band Signals

Test setup

- The handset is mounted to an artificial head
- A pure sine tone of 0dBm0 is applied at the digital interface.
- The level of spurious out of band signals is measured at the artificial ear.

The signal flow of this test is shown in figure 5 .

Test procedure

This test is performed for input signals in the range 300 Hz to 3400 Hz.

Test requirements

The level of out of band signals shall be lower than the in band acoustic level obtained by a digital signal at 1 kHz at the level shown in figure 13 .

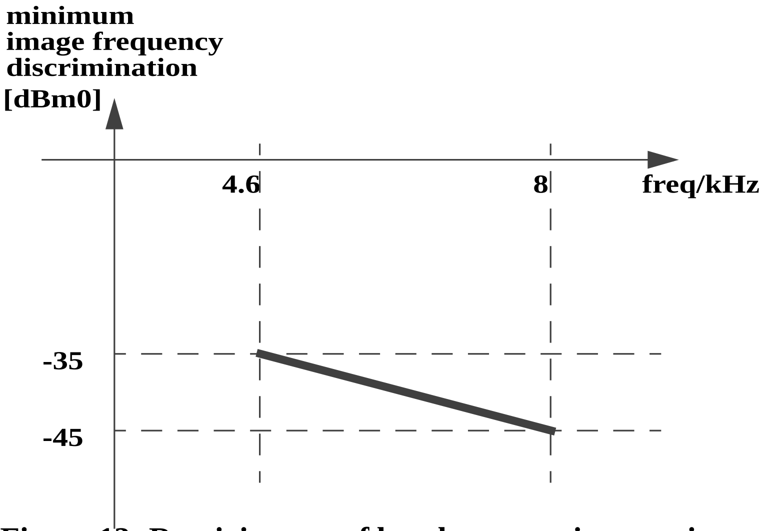

Figure 13: Receiving out of band suppression requirement

6.14.Sending Idle channel noise

Test setup

- The handset is mounted to an artificial head
- The level of the digitized microphone signal is measured at the DAI while no signal is applied at the microphone.

The signal flow of this test is shown in figure 3 .

Test procedure

The measurement is done under silent condition.

Test requirements

The idle noise shall not exceed -64dBm0p.

6.15.Receiving Idle channel noise

Test setup

- The handset is mounted to an artificial head
- The level at the loudspeaker signal is measured at the artificial ear while no signal is applied by the DAI.

The signal flow of this test is shown in figure 5 .

Test procedure

The test is performed for the nominal and for the maximum volume setting.

Test requirements

The measured noise shall not exceed -57dBPa(A) at the nominal volume setting and -54dBPa(A) at the maximum volume setting.

7. DAI usage for audio tests

The following table summarises the audio tests and shows, for which tests the digital audio interface is needed.

Table 1: Dai usage for audio tests

Test	GSM recommendation	DAI usage
Sending Frequency response	11.10, 30.1	read DU
Sending loudness rating	11.10, 30.2	read DU
Receiving Frequency response	11.10, 30.3	write DD
Receiving loudness rating	11.10, 30.4	write DD
Side tone masking rating	11.10, 30.5.1	no
Listener side tone rating	11.10, 30.5.2	no
Echo loss	11.10, 30.6.1	no
Stability margin	11.10, 30.6.2	no
Sending distortion	11.10, 30.7.1	read DU
Receiving distortion	11.10, 30.7.2	write DD
Sidetone distortion	11.10, 30.8	no
Sending out of band signals	11.10, 30.9.1	read DU
Receiving out of band signals	11.10, 30.9.2	write DD
Sending idle channel noise	11.10, 30.10.1	read DU
Receiving idle channel noise	11.10, 30.10.2	write DD

It can be seen, that the DAI has to perform only two different tasks. In some tests it has to write a signal on DD, in other tests it has to read a signal from DU.

8. Acceptance Test

The DAI acceptance test should be performed with a mobile which already has passed type approval. The DAI program has to be called with the parameter ‘AUDIO’ and the respective tests have to be called at the FTA tester. It is not necessary that the tests are fulfilled in all points, since tuning of the audio is not part of this project. Only the functionality of the DAI has to be proven.

It is sufficient to show this functionality for the RX and for the TX path separately. Therefore the tests described in 6.1 (GSM recommendation 11.10, 30.1) and 6.3 (GSM recommendation 11.10, 30.3) are required.

8.1. Play File

This test is done without GSM system tester. A file stored on the DAI-PC is played. Call

DAI audio <play_file.bin> 1

and listen to the output. The loudspeaker signal must sound reasonable.

8.2. Record File

This test is done without GSM system tester. The microphone signal is recorded. Call DAI by

DAI audio 40000 <rec_file.bin>

and talk to the microphone until the DAI program has finished. Control whether the recorded file is ok by playing it.

8.3. Bit Exact Audio Loop

An audio loop is programmed on the DSP of the mobile. A file stored on the DAI-PC is played on the DU line of the mobile and the DD signal is recorded. Call the DAI software by

DAI audio <play_file.txt> <rec_file.txt>

and compare the two files afterwards. It is expected that the recorded file is a delayed version of the played file. Also the three LSBs of the 16 bit words may be modified.

8.4. RX-Test

Call test 30.3 at the GSM system simulator and listen to the loudspeaker signal of the mobile. A sequence of sine tones (increasing frequency) is written to the digital audio interface. These tones must be heared at the loudspeaker.

The DAI software is called by

DAI audio

8.5. TX-Test

Call test 30.1 at the GSM system simulator and connect the mobile to the acoustic head. The frequency response measured by the system tester must look reasonable. Note that it is not required to fulfil the test for all frequencies.

The DAI software is called by

DAI audio