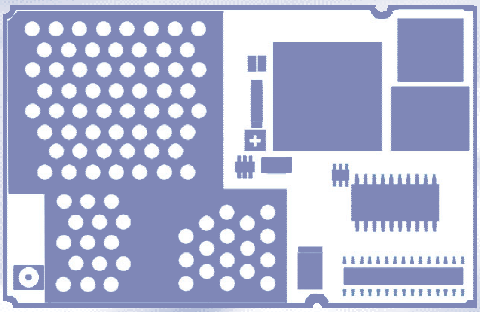
SIEMENS
mobile



# Audio Interface Design for GSM Applications

**Siemens Cellular Engines**

**Version:** v04
**DocID:** mc4x_5x_AN_02_Audio_v04

Application Note 02

| | |
|---|---|
| Application Note 02: | **Audio Interface Design for GSM Applications** |
| Supported products: | **MC4x, MC5x, MC38x, TC45** |
| Version: | **v04** |
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| DocId: | **mc4x_5x_AN_02_Audio_v04** |
| Status: | **Confidential / Released** |

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# 0 Document history

This chapter reports modifications and improvements over previous versions of the document.

Preceding Application Note 02: "Audio Interface Design", Version 03
New Application Note 02: "Audio Interface Design for GSM Applications", Version **v04**

| Chapter | What is new |
|---|---|
| 1.1 | Added further supported products. |
| 2 | New overview of audio interfaces. |
| 4 | Added information on typical audio problems in GSM applications |
| 5 | Recommendations for RF decoupling. |
| 6.2.1 | New Chapter: “Calculating dB” |
| 6.3 | Settings of AT^SAIC storable to user profile (not applicable to Siemens GSM modules released before August 2003) |
| 7 | Added information on Siemens Car Kit Portable |

# 1 Introduction

This application note provides technical recommendations for integrating audio accessories into cellular applications based on Siemens GSM modules. It discusses various solutions for typical design approaches, evaluates strategies of overvoltage protection, explains the concept of handsfree operation, and then focuses on audio specific AT commands. A list of sales contacts and a summary of the discussed audio accessories is included.

## 1.1 Supported products

- MC45
- MC46
- MC55
- MC56
- MC58
- MC388
- MC389
- TC45

## 1.2 Related documents

[1] Hardware Interface Description of your Siemens wireless module
[2] AT Command Set for your Siemens wireless module
[3] Release Notes related to the firmware of your Siemens wireless module
[4] Application Note 14: Audio and Battery Parameter Download

The latest product information and technical documents are ready for download on the Siemens Website or may be obtained from your local dealer or the Siemens Sales department.
To visit the Siemens Website you can use the following link:
http://www.siemens.com/wm

## 1.3 Approval Considerations

The Siemens GSM modules listed above have been type approved for use with the Siemens reference setup presented in Chapter 8. Regarding audio performance, compliance with the International standard TS 51010-1 (successor to ETS 300 607-1 and GSM 11.10) and GCF recommendations has been certified.

When designing a GSM application you are advised to make sure whether the final product is standard compliant. This is particularly important for mobile phones, PDAs or other handheld transmitters and receivers incorporating a GSM module. Depending on the individual design, such devices may require additional Type Approval, for example, if the device includes a handset held close to the ear.

Outside Europe, there may be further international, national or government standards and regulations to be observed for Type Approval.

## 1.4 Terms and Abbreviations

| Abbreviation | Meaning |
|---|---|
| AF | Audio Frequency |
| DAI | Digital Audio Interface |
| dBm0 | Digital level, 3.14dBm0 corresponds to full scale, see ITU G.711, A-law |
| DSB45 | Development Support Box 45 |
| EMC | Electro Magnetic Compatibility |
| EMI | Electromagnetic Interference |
| EPP | Earpiece Positive |
| EPN | Earpiece Negative |
| ESD | Electrostatic discharge |
| FFC | Flat Flexible Cable |
| GSM | Global System for Mobile Communication |
| MICP | Microphone Positive |
| MICN | Microphone Negative |
| n.c. | Not connected |
| NLMS | Normalized Least-Mean-Square |
| opamp | Operational amplifier |
| PCB | Printed Circuit Board |
| SIM | Subscriber Identifier Module |
| SNR | Signal-to-noise ratio |
| RF | Radio Frequency |
| VANA | Voltage Analog |
| VREF | Voltage Reference |

# 2 Overview of audio interfaces

Using the AT^SAIC command you can switch back and forth between each of the analog interfaces and the digital interface. See also Chapter 6.3 for details on how to use AT^SAIC. The two analog outputs are identically designed. The two analog inputs are slightly different and intended for use with different devices.

The DAI replaces the AD/DA converter unit of the module. Thus, all digital filters, gains and DSP functions are usable via DAI. If a flat frequency response without influence of DSP is requested, audio modes 5 or 6 are recommended.

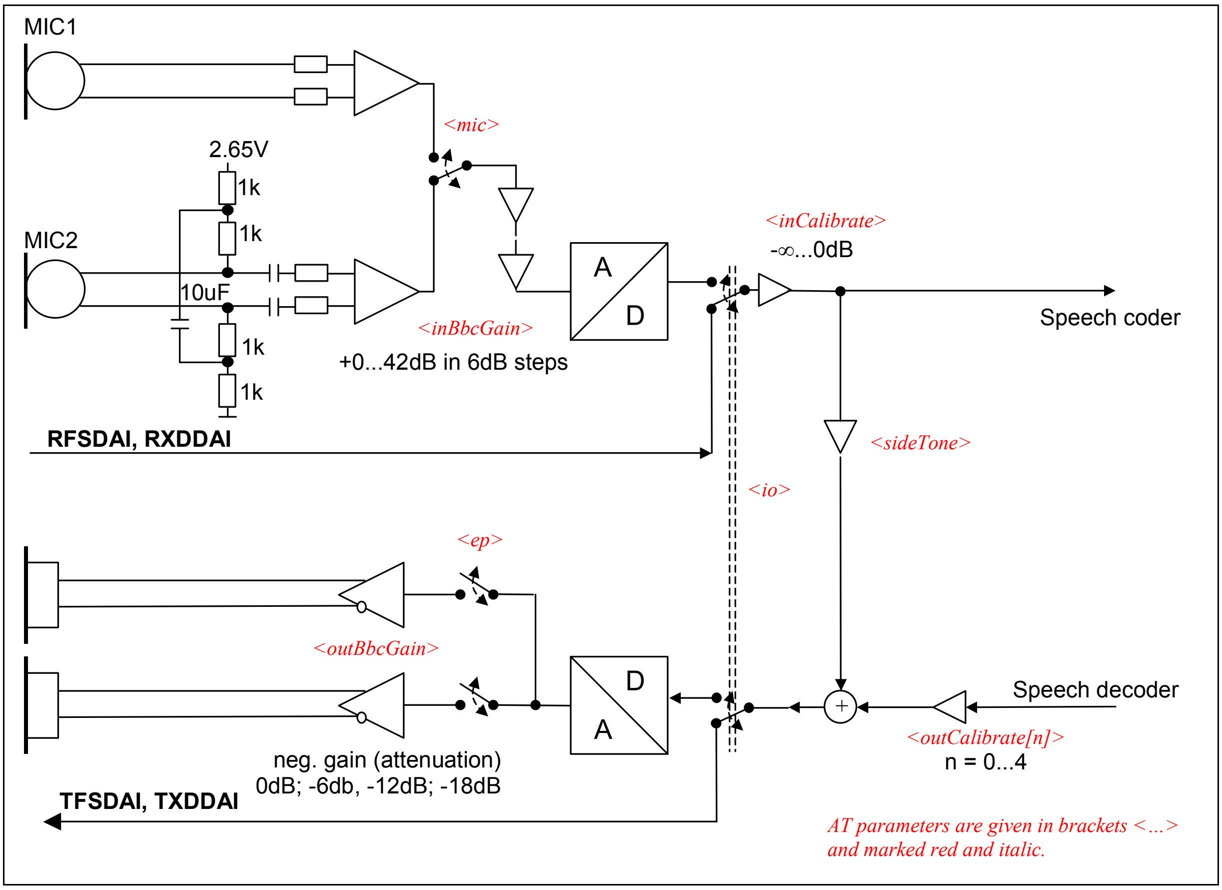


Figure 1: Block diagram of digital and analog interfaces selectable with AT^SAIC

## 2.1 General usage of microphone interface 1 and 2

Microphone interface 1 is high impedance (~50kOhm) and shall be used preferably if an opamp or a CODEC is connected or additional microphone feeding is needed.

Microphone interface 2 is high impedance (~1kOhm) and shall be used preferably if internal microphone feeding can be used, e.g. for an internal microphone.

To reduce or increase the gain of the module you may use the AT^SNFI command. Compare chapter 6.2.3.

# 3 Solutions for the digital audio interface (DAI)

Figure 2 and Figure 3 show an example of using the digital audio interface of the module. The Motorola codec MC145483 can be replaced with a DSP. For a GSM module the frequency of the clock generator is not fix, but the MC145483 accepts only discrete frequencies. Framesync master is the module (TFSDAI line) and thus the GSM network.

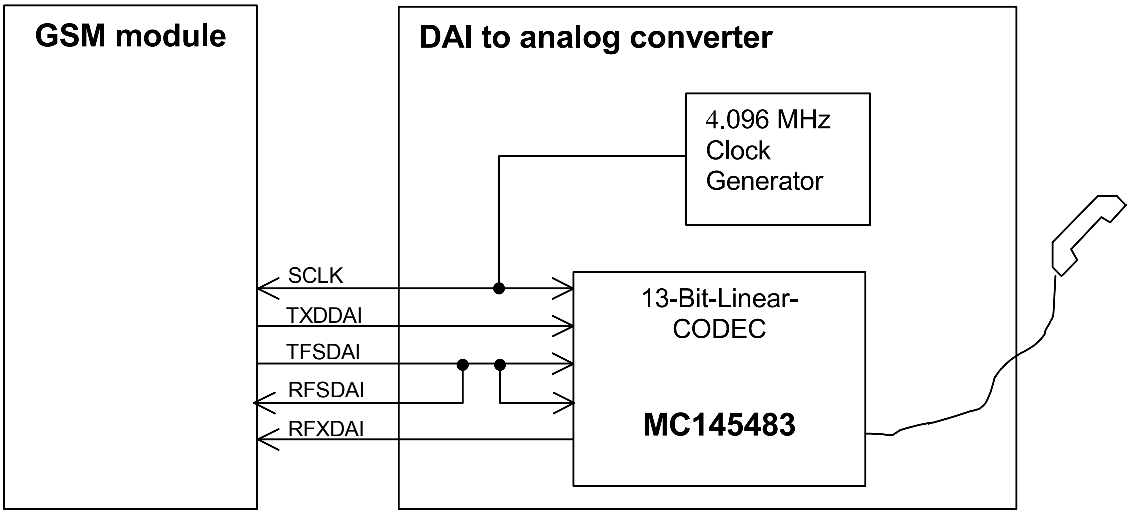


Figure 2: Block circuit for DAI to analog converter

This DAI analog converter is well suited for evaluating and testing a telephone handset and can be used instead of the headset interface of DSB45.

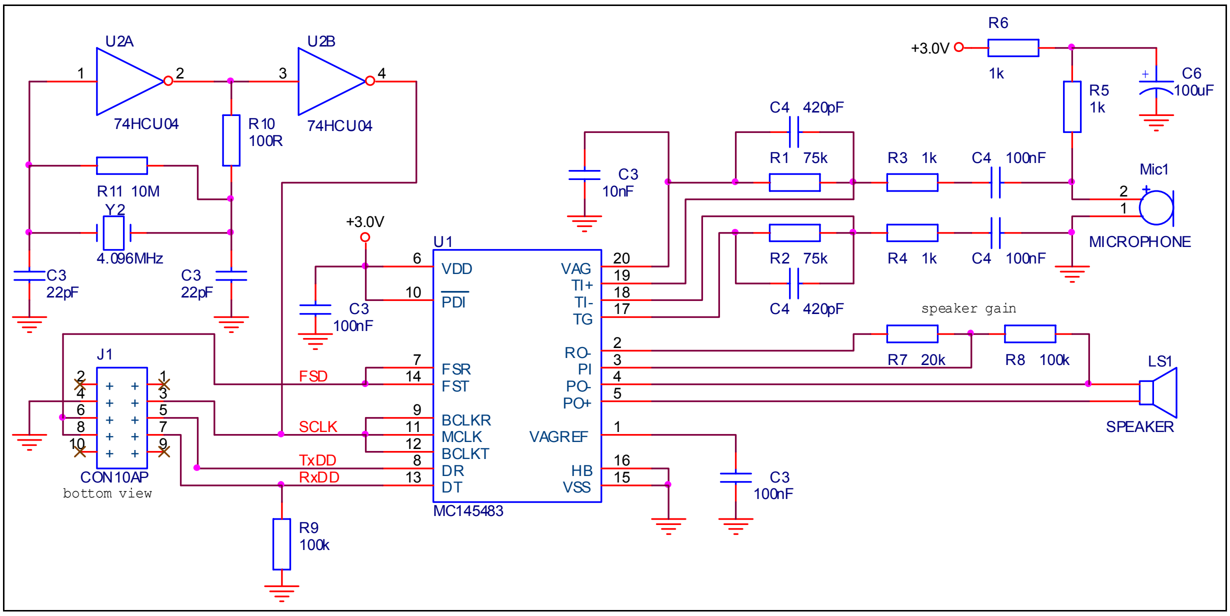


Figure 3: Sample circuit for analog to DAI box

The logical levels and the interface at connector J1 are compatible to the DSB45 DAI interface.

# 4 Solutions for analog audio interfaces

A major audio problem arises with the demodulation of 900/1800/1900 MHz RF frequency from the own antenna. Semiconductors always are a source of RF demodulation. Each semiconductor needs to be RF-decoupled using a capacitor from 10 to 33pF.

It is recommended to use a microphone with a sensitivity of at least –44 ±3 dB/Pa at 2V and 2kΩ (0dB=1V/Pa, 1kHz). It should be equipped with two internal EMI capacitors for GSM 900/1800 MHz bands. External ESD protection is required to protect the microphone from damage. Even a high-quality microphone should be placed at least 5 cm away from the antenna.

Other audio problems may be due to insufficient filtering of power supply and different GND levels between PCBs.

## 4.1 Internal microphone feeding (audio input 2 only)

Your GSM module comes with an internal microphone feeding at the module's audio input 2 as shown in Figure 4. The microphone signal is very sensitive to any disturbances from the power supply, ground bouncing or direct RF intrusion and demodulation. Therefore, a balanced microphone line will be the best choice.

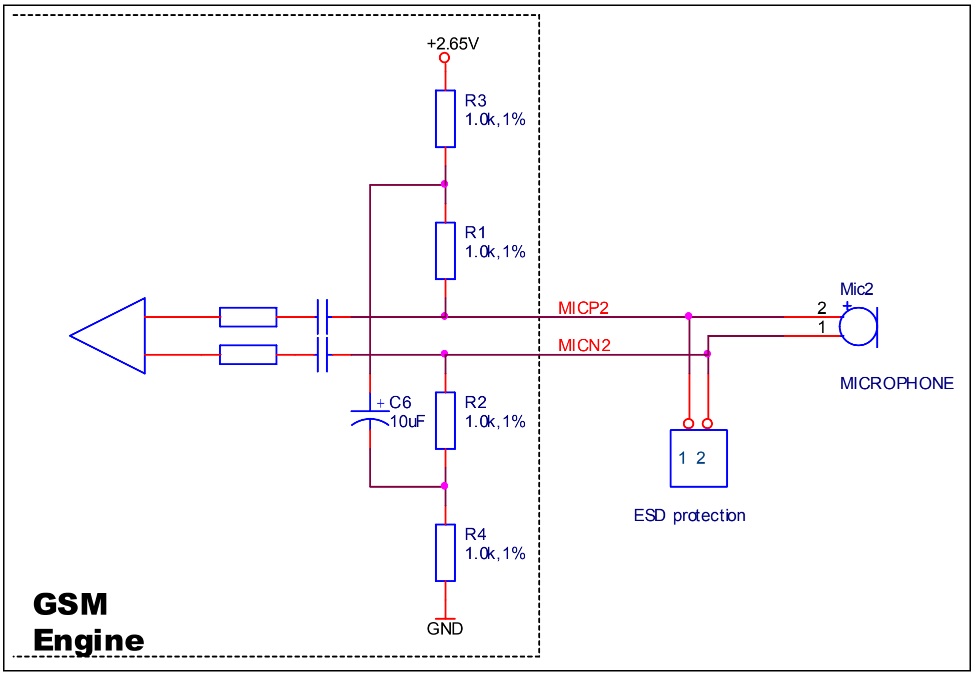


Figure 4: Idealized diagram of the fed audio input 2 terminal

## 4.2 External microphone feeding

For an external feeding solution audio input 1 of the module is the best choice (shown in Figure 5).

You may also use audio input 2 with external microphone feeding, but due to its lower input resistance you will need to invest some more parts to obtain the same result as with interface 1. For details see Figure 6 and Figure 7.

ESD devices used in the following two solutions are explained in Chapter 5.

### 4.2.1 Simple microphone feeding

This chapter discusses simple feeding circuits for an electret microphone (R1, R2, C6).

Figure 5 presents a solution for audio input 1.

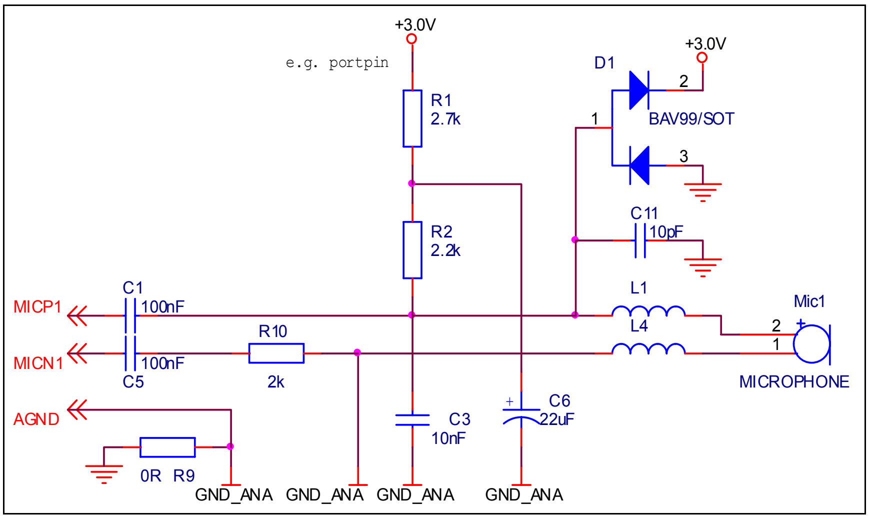


Figure 5: Circuit of microphone feeding at the module's non-fed audio interface 1

If the module input with the lower impedance (audio input 2) is used, or if there is a large distance between the microphone and the module, then an additional opamp is recommended. If the output resistance of the operational amplifier at Figure 6 is below 3 Ohms, the circuit will be sufficiently balanced, also at the fed lower impedance input 2.

The opamp shown in Figure 6 should be placed as far as possible from the antenna.

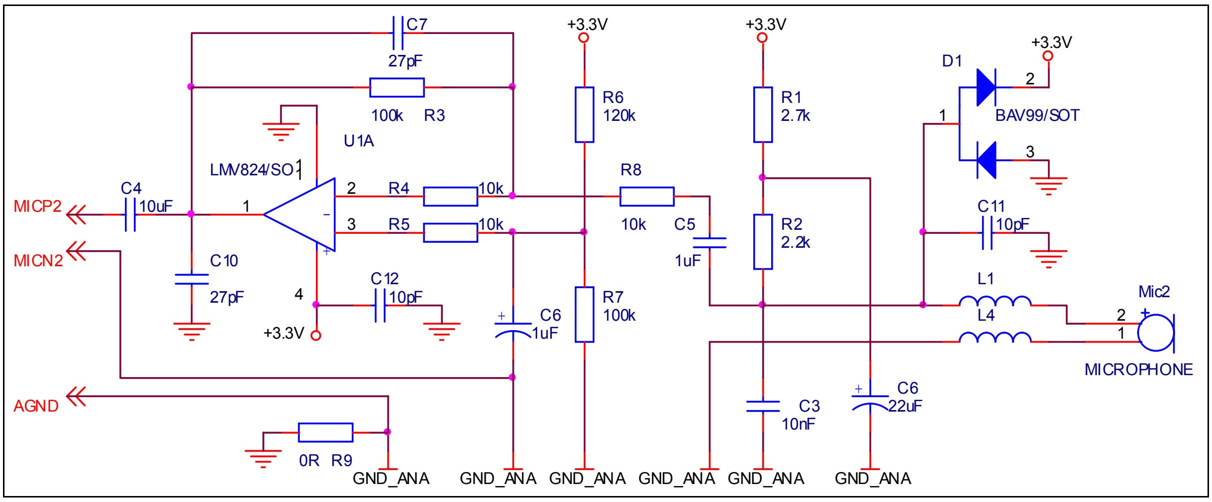


Figure 6: Circuit of microphone feeding at fed audio input 2 with opamp near microphone

For the example shown in Figure 6, C10, R4 and R5 have been chosen to suppress the demodulation near the operational amplifier. C11 has been added to suppress the demodulation at diode D1. As the cable length of acoustic devices usually is some cm, ferrite beads L1 and L4 (e.g. Murata BLM18HG601) are recommended to protect the circuit from RF intrusion. Avoid placing a GND next to these parts.

The GND_ANA net should be a separate small net connected to GND at a single point. This can be done, for example, by using a resistor as shown in the example above (R9) or connecting to the module's AGND pin.

## 4.2.2 Balanced microphone feeding

Figure 7 shows a balanced feeding circuit for a microphone. The distance from the GSM module should be kept as short as possible. This is essential to take into account the small level of the microphone signal and to avoid ground bouncing between GSM module and application.
If you decide to apply this external feeding circuit, we recommend to use a microphone with a sensitivity of at least –38 ±3 dB/Pa at 2V and 2kΩ (0dB=1V/Pa, 1kHz).

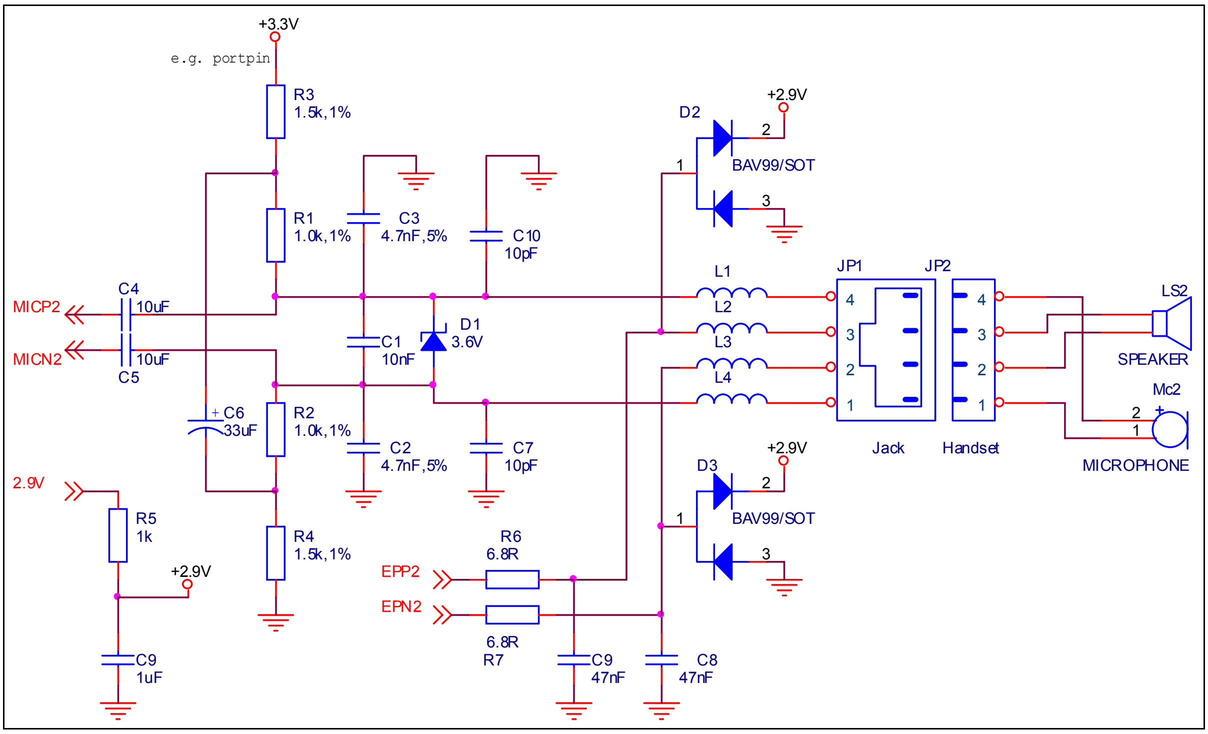


Figure 7: Circuit of a balanced microphone feeding

R3, R4, C6 have been chosen to smoothen the feeding voltage. C7 and C10 have been added to suppress the demodulation at the D1 diode. At the non-fed input MIC1, the capacitors C4 and C5 may be 100nF.

## 4.3 Galvanically coupling the speaker signal

The GSM module is able to drive speakers held close to the ear (Figure 8). C3 forms a 2.5kHz, 1st order noise deemphasis filter. A corresponding preemphasis is preconfigured in audio mode AT^SNFS=2 and 3. This reduces noise floor of the power amplifier.

The frequency response and loudness largely depends on the measurement method, casing, fitting and impedance. An impedance of 16Ω (louder) to 32Ω is recommended. The outputs of the modules listed in Chapter 1.1 are not short circuit protected.

Figure 7 also shows a sample circuit of speakers directly connected to the GSM module. R6 and R7 are necessary to prevent module amplifiers from oscillating while C8 and C9 protect the module from ESD pulses.

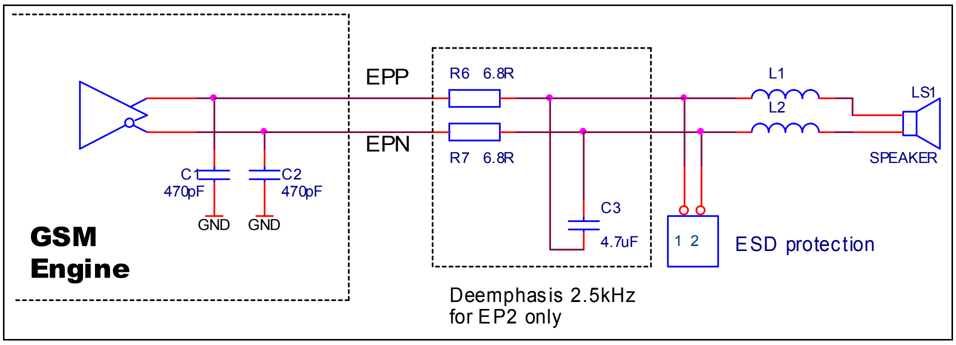


Figure 8: Idealized diagram of the audio output terminal

## 4.4 Decoupling the speaker signal

The speaker signal is balanced. Due to ground bouncing this signal cannot be directly used. Therefore it is recommended to apply an opamp which transforms the reference point. C10, R1, R2 avoid RF-demodulation at the output of U1.

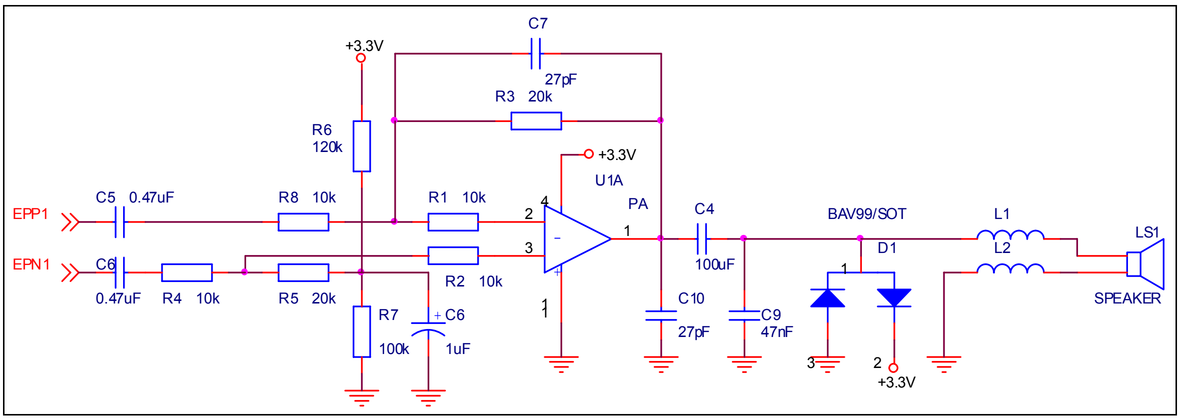


Figure 9: Speaker decoupling

# 5 Overvoltage protection

This chapter describes solutions for overvoltage protection recommended for ESD protection. You can use one of the described solutions or a combination of several methods.

## 5.1 Spark gaps

The most common way to provide ESD protection is to use spark gaps, as for example applied to the SIM interfaces of Siemens GSM modules.

Spark gaps should be located close to the possible place of flashover. One tip must be connected to the ground plane, the other one to the point to be protected.

Advantages: Low cost, if included in the layout.
Disadvantage: Value of the ignition voltage is fuzzy.

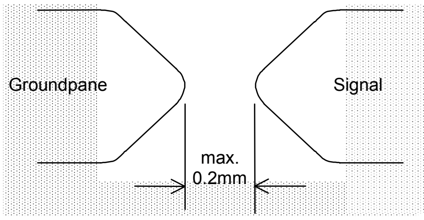


Figure 10: Spark gap

## 5.2 Clamp diodes

A pair of diodes (e.g. BAV99) is required; the anode must be connected to GND and the cathode to the positive supply voltage (see the schematics shown in Figure 6, Figure 7 and Figure 9).
Advantages: Low cost, requires minimum space.
Disadvantages: The overvoltage is fuzzy as it largely depends on the internal resistor of the diode. The effect can be improved with Schottky diodes or high current diodes. Semiconductors always are a source of RF demodulation. Each semiconductor needs to be RF decoupled using a capacitor from 10 to 33pF.

## 5.3 Clamp diodes with external serial resistor

A resistor connected in series to the clamp diodes will reduce the current flowing through the diodes. This way, voltage drops over the diodes can be minimized. The higher the value of the resistor the lower the voltage drop of the diodes, but the higher the voltage drop of the resistor.
Semiconductors always are a source of RF demodulation. Each semiconductor needs to be RF decoupled using a 10..33pF capacitor.

## 5.4 Z-Diodes

Especially supplied electret microphones should be protected with a Z-diode connected in parallel (see Figure 7).
Semiconductors always are a source of RF demodulation. Each semiconductor needs to be RF decoupled using a capacitor from 10 to 33pF.

## 5.5 Capacitors

In addition to the solutions described above, it is recommended to use capacitors connected to GND, especially in the audio lines (see Figure 6, Figure 7 and Figure 9).

# 6 Using AT commands to control the audio interface

The audio parameters of all GSM engines can easily be configured with AT commands. Below you can find a number of examples showing how to use audio specific AT commands. The full set of AT commands is specified in [2].

## 6.1 Supported audio modes

Table 1 contains a summary of the audio modes supported by Siemens GSM modules. Further details are explained in [1] and [2]. The audio mode can be selected with the AT^SNFS command.

Table 1: Selectable audio modes

| No. | Name | AT command | Purpose | MIC2 feeding | Description |
|---|---|---|---|---|---|
| 1 | Default Handset | AT^SNFS=1 | Approval configuration | 2.65V | MIC1, EP1, adapted to Votronic Handset HH-SI-30.3 with DSB45 Support Box<br>Not adjustable |
| 2 | Basic Handsfree | AT^SNFS=2 | | 2.65V | MIC2, EP2, adapted to Siemens CarKit Portable<br>5 volume steps selectable |
| 3 | Headset | AT^SNFS=3 | | 2.65V | MIC2, EP2, adapted to Mono-Headset S45<br>5 volume steps selectable |
| 4 | User Handset | AT^SNFS=4 | | 2.65V | MIC1, EP1, Votronic Handset HH-SI-30.3 with DSB45 Support Box<br>5 volume steps selectable |
| 5 | Plain Codec | AT^SNFS=5 | | 0V | MIC1, EP1, no filtering<br>5 volume steps selectable |
| 6 | Plain Codec | AT^SNFS=6 | | 0V | MIC2, EP2, no filtering<br>5 volume steps selectable |

When shipped from factory, Siemens GSM modules are set to interface 1 and audio mode 1. This configuration is optimized for the Votronic HH-SI-30.3/V1.1/0 handset and used for type approving the Siemens reference equipment. Audio mode 1 has fix parameters which cannot be altered. To adjust the settings of the Votronic handset simply select another audio mode.

## 6.2 Adjusting the volume

There are several ways to adjust the volume of the connected audio devices. Each audio mode uses 5 volume steps, which can be selected with the parameters <outStep> or <level>. The steps can be set with the following commands (where <outStep> or <level> are identical).

AT^SNFV=<outStep>
AT+CLVL=<level>
AT^SNFO=<outBbcGain>,<outCalibrate[0]>,...<outCalibrate[4]>,<outStep>,<sideTone>

The *values* of the 5 volume steps <outStep> and <level> can be specified with the parameters <outCalibrate[0]> …<outCalibrate[4]> of the AT^SNFO command. Table 2 contains the module's factory settings.

No matter which command you use to set the volume, the selected step (<outStep> or <level>) will be stored non-volatile when the GSM module is powered down with AT^SMSO or reset with AT+CFUN=1,1.

Users should be aware that the selected volume step is a global setting, i.e. when selecting another audio mode with AT^SNFS the value of <outStep> or <level> does not change. This is also true for mute operation, which will be retained when you switch back and forth between different audio modes. To mute the microphone you have two commands: AT^SNFM and AT+CMUT.

All the parameters configurable with AT^SNFO need to be saved with AT^SNFW for use after restart, except for <outStep> or <level>. Please take into account that AT^SNFW will save all values currently selected in audio modes 2 to 6.

Table 2: Default values of audio modes (subject to change)

| Mode | Default settings (to be queried with AT^SNFO?) | |
|---|---|---|
| ^SNFS=1 | ^SNFO=1,16384,16384,16384,16384,16384,4,8192 | (no steps) |
| ^SNFS=2 | ^SNFO=1,4685,6301,8500,11205,15115,4,0 | (2.5dB steps) |
| ^SNFS=3 | ^SNFO=2,1253,2452,4891,9759,16383,4,682 | (6dB steps) |
| ^SNFS=4 | ^SNFO=1,10337,11598,13014,14602,16384,4, 8192 | (1dB steps) |
| ^SNFS=5 | ^SNFO=0,16384,16384,16384,16384,16384,4,0 | (no steps) |
| ^SNFS=6 | ^SNFO=0,16384,16384,16384,16384,16384,4,0 | (no steps) |

## 6.2.1 Calculating dB

dBm0 is a measure of digital telecommunication signals. It relates to the digital coded signal on the digital side of the network. It should not be confused with any electrical unit. For example, 3,15dBm0 corresponds to a fully gained digital coded sine wave. For Siemens GSM modules, the correlation between the digital value and the amplitude of the analog signal at 1kHz is givenss in [1].

dBm0p is the psophometric weighted dBm0 value.

dBV relates to 1Vrms and dBm relates to 1mWrms while you can convert each other assuming a telecom typical nominal resistance of 600Ohms as

1Veff = 0dBV ~ + 2.2 dBm.

dB is just a gain (G) or attenuation (negative gain). It can be used for both RF and AF, while for RF you usually compare power (P) at a constant impedance and for AF you compare voltages (U).

G = 10*log(P1/P2) = 20*log(U1/U2)

Some common values / samples:

| | |
|---|---|
| U1/U2=10 → G=20dB | P1/P2=10 → G=10dB |
| U1/U2=2 → G~6dB | P1/P2=2 → G~3dB |
| U1/U2=20 → G~26dB | P1/P2=20 → G~13dB |

## 6.2.2 Specifying the value of the volume steps

In audio modes 2 – 6, the value of the volume steps can be specified with the parameters <outCalibrate[0]>,...<outCalibrate[4]> of AT^SNFO.

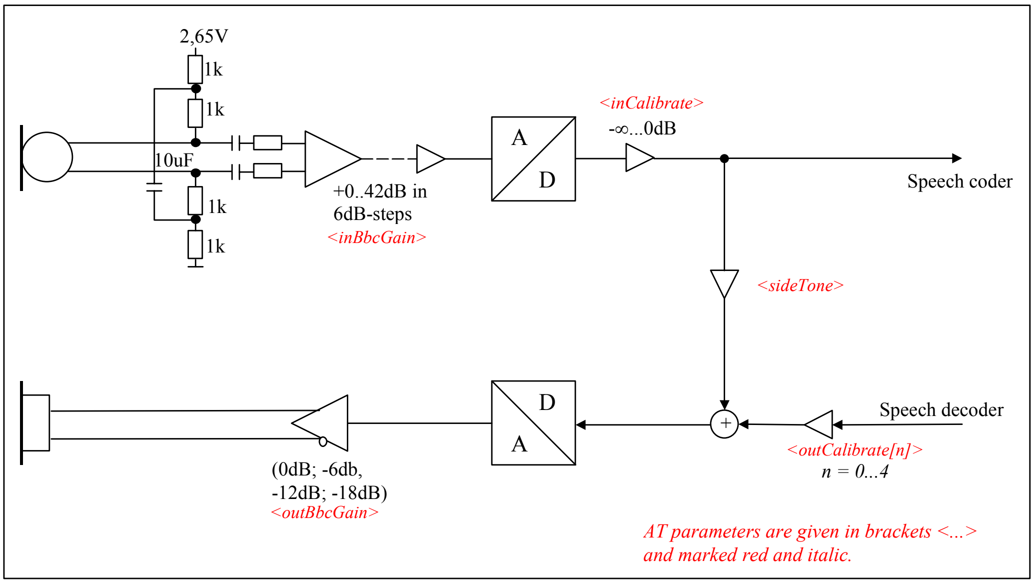


Figure 11: Audio parameters selectable with AT commands

Examples:

| | |
|---|---|
| AT^SNFS=4 | (select User Handset) |
| AT^SNFO? | |
| ^SNFO=1,10337,11598,13014,14602,16384,4,4096 | (default 1dB steps) |
| steps | |
| AT^SNFS=4 | (select User Handset) |
| AT^SNFO=1,**6528,8192,10368,13056,16384**,4,4096 | (2dB steps) |
| AT^SNFW | (write to non-volatile memory) |

or

| | |
|---|---|
| AT^SNFS=4 | (select User Handset) |
| AT^SNFO=1,**4096,5824,8192,11616,16384**,4,4096 | (3dB steps) |
| AT^SNFW | (write to non-volatile memory) |

Now, AT+CLVL=<0...4> or AT^SNFV=<0...4> will adjust the volume according to the steps thus defined.

All permanent settings selected with AT^SNFO and saved with AT^SNFW can be reset to their default values:

| | |
|---|---|
| AT^SNFD | (recall manufacturer default) |

The value of the side tone is adapted automatically depending on the volume. It is sufficient to set the side tone only once, according to the requirements of the used equipment. This eliminates the need to make changes, whenever you reconfigure the remaining audio parameters with AT^SNFO.

The following example shows an alternative approach to use 6dB step analog attenuators. This way, better noise characteristic can be achieved at smaller loudness rates. The AT+CLVL command does not work for this method of loudness control because it is kept at a fix value. Therefore, type the full command line if a new volume step is needed.

| | |
|---|---|
| AT^SNFS=4 | (select User Handset) |
| AT+CLVL=4 | (default) |
| AT^SNFO=1,0,0,0,0,16384,4,4096 | (default) |
| AT^SNFO=**0**,0,0,0,0,**16384**,4,4096 | (default + 6 dB) |
| AT^SNFO=**2**,0,0,0,0,**16384**,4,4096 | (default – 6 dB) |
| AT^SNFO=**2**,0,0,0,0,**12288**,4,4096 | (default – 9 dB) |
| AT^SNFO=**3**,0,0,0,0,**16384**,4,4096 | (default – 12 dB) |
| | |
| AT^SNFD | (recall manufacturer default) |

### 6.2.3 Changing microphone sensitivity

The microphone path contains 6dB step analog amplifiers and a digital multiply value. As described in the previous chapter the setting can be made permanent.

| | |
|---|---|
| AT^SNFS=4 | (select User Handset) |
| AT^SNFI? | |
| ^SNFI=5, 32767 | (default) |
| | |
| AT^SNFI=**2**, 32767 | (default – 18 dB) |
| AT^SNFW | (write to non-volatile memory) |

These permanent settings can be reversed by

| | |
|---|---|
| AT^SNFD | (recall manufacturer default) |

### 6.2.4 Working with external audio processing codec

If an external audio processing codec is used, you should adjust the levels between codec and module as high as possible in order to improve SNR on this path. e.g.:

| | |
|---|---|
| AT^SNFS=4 | (select User Handset) |
| AT^SNFO=**0**,10337,11598,13014,14602,16384,4,8192 | (no analog attenuation for speaker) |
| AT^SNFI=**0**,32767 | (no analog gain for microphone) |

## 6.3 Changing physical audio interface

To switch back and forth between all three audio interfaces you can use the command
AT^SAIC=<io>[,<mic>[,<ep>]]

The AT^SAIC Write command is usable only in audio modes 2 – 6. If AT^SNFS=1, any attempt to use the AT^SAIC Write command returns "+CME ERROR: operation not allowed". This is because all default parameters in audio mode 1 are determined for type approval and are not adjustable.

The factory defaults of AT^SAIC vary with the selected audio mode.
If AT^SNFS=1 or 4 or 5, then AT^SAIC=2,1,1
If AT^SNFS=2 or 3 or 6, then AT^SAIC=2,2,2

Examples:

AT^SAIC=1 selects the digital interface only.
AT^SAIC=2,1,2 selects the analog interfaces MIC1 and EP2.
AT^SAIC=2,2,3 selects the analog interfaces MIC2, EP1 and EP2 while both speakers always get the same output power.

The settings made with AT^SAIC of AT^SNFS can be stored to the audio profile set with AT^SNFW. AT^SNFD can be used to reset the factory defaults.

## 6.4 Extended usage of the MICP2 pin

The power supply of the MICP2 pin (2nd analog audio interface) is programmable via the command

AT^SNFM=[<MicSwitch>],MicVccCtl>]

This gives you greater flexibility in connecting audio accessories or using the MICP2 pin for a variety of functions other than audio.

The parameter <MicSwitch> mutes or activates the microphone at the currently selected audio interface. The parameter can only be set when there is an active call.

In addition, the parameter <MicVccCtrl> controls the power supply of MICP2. Note that the first parameter <MicSwitch> must be omitted when setting <MicVccCtrl>.
AT^SNFM=,0: Permanently switches off the power supply at MICP2.
AT^SNFM=,1: Permanently switches on the power supply at MICP2.
AT^SNFM=,2: Default setting. Power at MICP2 is applied only during a call.

This means that with AT^SNFM=,0 or AT^SNFM=,1 the power supply can be controlled independently of GSM activity. The permanent power supply can be used to feed an audio application even when the GSM part is inactive, for example if the host device integrates a dictaphone or voice recorder connected in parallel to MICP2 and MICN2.

The setting AT^SNFM=,1 makes MICP2 a permanent 2.65V general purpose output. However, the 2kOhms inner resistance of the MICP2 circuit allows to drive a low-current device only or requires to include an additional transistor or gate to increase output power. Also, consider the time constant resulting from the 2kOhms inner resistance and 10µF capacitance of the MICP2 circuit.

# 7 Handsfree concept

The Handsfree mode (AT^SNFS=2) has been optimized for the “Siemens Car Kit Portable HKP-500” and for a special arrangement of microphone, speaker and user (see Figure 13). Physically, audio interface 2 (default) or 1 can be used.

EP2 output is followed by a power amplifier with 20..40dB gain. The external microphone amplifier with 26...40dB gain needs to have good noise characteristic. Final adjustments can be done easily with AT^SNFO, AT^SNFI and AT^SNFW.

For product information on the Siemens Car Kit Portable HKP-500 see Chapter 9.

Figure 12: Siemens Car Kit Portable HKP-500

## 7.1 Mechanical and quality issues

High sensitivity of microphone and small speaker distortion increase the efficiency of DSP echo cancellation and noise reduction routine.

If the microphone is sealed with rubber or glue on its backside, this reduces the backward sensitivity of the microphone.

The speaker shall combine high modulation of membrane with low distortion. A lot of distortion is produced in the plastic housing of the speaker or, for example, in the glass display of PDA or phone. A high speaker volume will be achieved if forward and backward volume of the speaker are well separated by sealing the speaker at the housing.

## 7.2 Short introduction to DSP algorithms

The Handsfree mode involves several DSP algorithms which are influencing each other more or less.

`echo cancelling algorithm:`
Based on the NLMS algorithm, an adaptive digital filter searches for parts of the known speaker signal within the microphone signal in order to substract them. It‘s filter length is 22msec.

`automatic gain adaptions`
This is used as echo suppression if the echo cancelling cannot work properly or the result of the cancelling algorithm is not sufficient. It attenuates the party that is currently not speaking and it amplifies the currently speaking party.

`dynamic compression limitation:`
For car applications it is nice to raise the far end talker loudness automatically above the car environmental noise.

`noise reduction algorithm`
This is a collection of algorithms using the various voice characteristics or the characteristic of car noise in order to keep the voice understandable while the noise is dropped.

## 7.3 Handsfree reference application

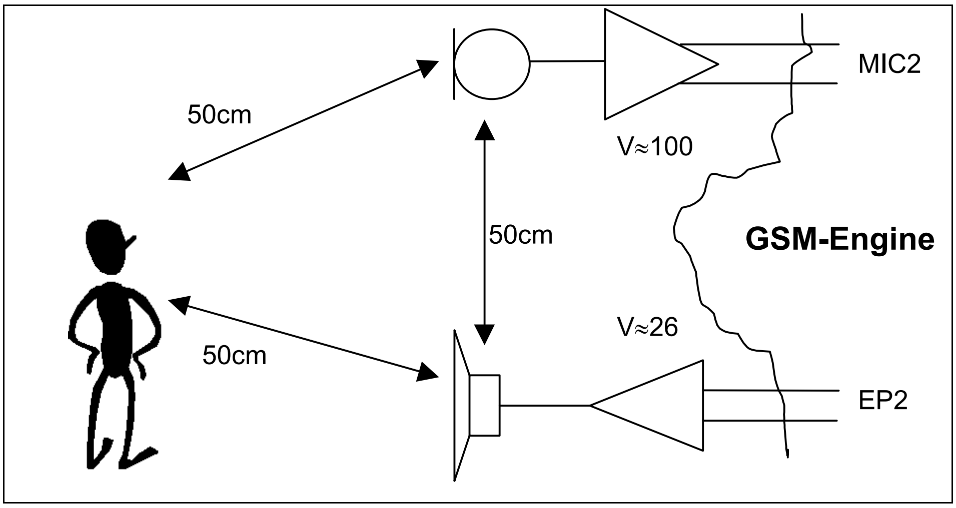


Figure 13: Handsfree arrangement with Car Kit Portable

In handsfree mode, there is no filter implemented. Therefore, the frequency response of the connected acoustic devices should be flat.

There is nearly no automatic gain control in the microphone path, due to the better voice quality. Users can only adjust the loudness. A dynamic compression of approx. 10dB, depending on environmental noise, is activated in the receive path, which is pleasant, particularly in noisy environment (car).

## 7.4 Handsfree reference environment

For the purpose of testing and evaluating the performance of a handsfree application, a Siemens Car Kit Portable HKP-500 can easily be connected to the DSB45 Support Box. The Car Kit Portable needs to be fed by an external 12V power source, while the DSB45 must be supplied from an additional 9V power supply unit. The quality achieved with this simple environment is already usable for vehicle mounted hands free products.

# 8 Siemens Reference Setup

To give one example, the following description proceeds from the audio design approach used for the Siemens reference GSM setup.

The reference equipment includes the following components:

- GSM module
- DSB45 Support Box (evaluation kit designed to test and type approve Siemens cellular engines and to provide a sample configuration for application engineering)
- SIM card reader integrated on the PCB of the DSB45 Support Box
- Handset type Votronic HH-SI-30.3/V1.1/0
- Antenna cable that connects the antenna connectors of the GSM module and of the DSB45 Support Box
- PC as MMI

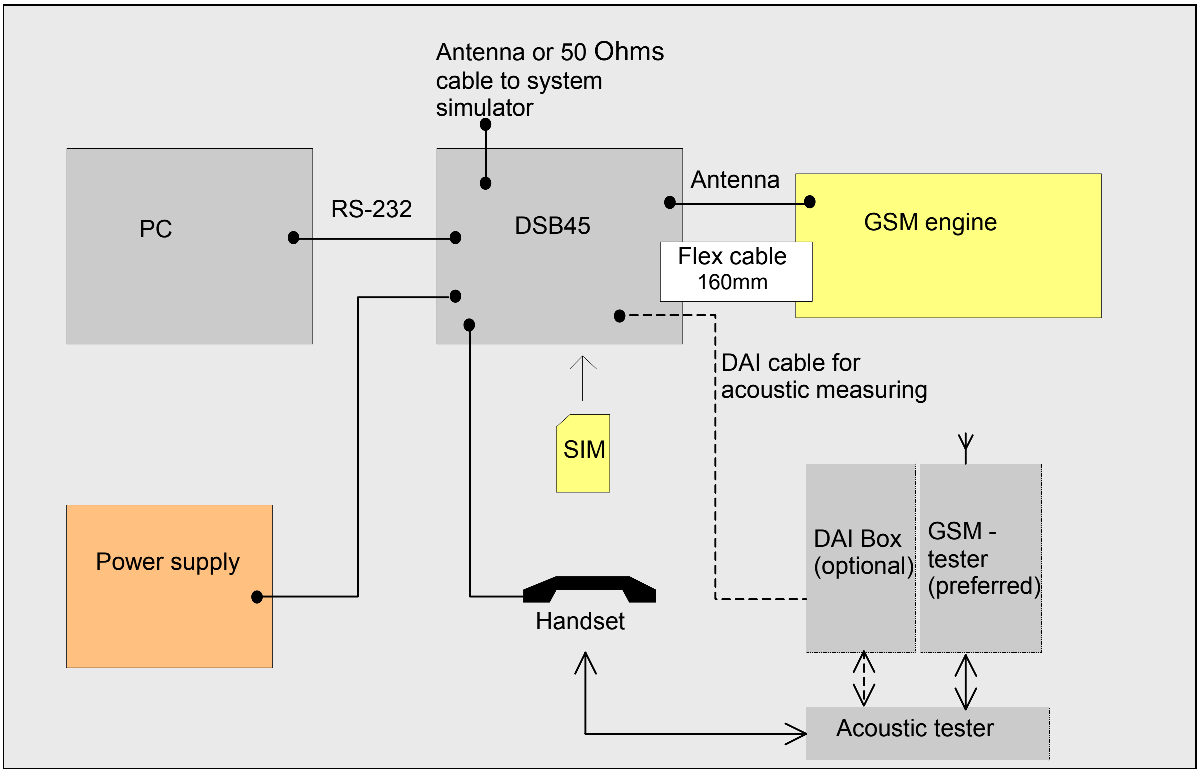


Figure 14: Siemens reference equipment

Audio approval measurements can be done using the following equipment:

1. DAI-Box while the acoustic audio transmission path is evaluated excluding the GSM connection

and

2. GSM tester, while the acoustic path including the GSM codec is evaluated by the acoustic tester.

# 9 List of parts and accessories

Table 3: Summary of products listed in this Application Note

| Product | Company | Ordering information |
|---|---|---|
| Siemens Car Kit Portable | Siemens | Siemens ordering number: L36880-N3015-A117 |
| DSB45 Support Box | Siemens | Siemens ordering number: L36880-N8101-A100-3 |
| Votronic Handset | VOTRONIC | Votronic HH-SI-30.3/V1.1<br>VOTRONIC<br>Entwicklungs- und Produktionsgesellschaft für elektronische Geräte mbH<br>Saarbrücker Str. 8<br>D-66386 St. Ingbert<br>Phone: 06 89 4 / 92 55-0<br>Fax: 06 89 4 / 92 55-88<br>e-mail: contact@votronic.com |
| Infineon DAI-Box | Infineon | Q67250-H0241 |

## 9.1 Suppliers of acoustic devices

The following list does not represent any kind of quality evidence for specific products. Its just a starting point for further investigations:

Table 4: Suppliers of acoustic devices

| Company | URL | Country | Product types |
|---|---|---|---|
| Panasonic | www.panasonic.com | USA | mic, rcv, spk |
| Hosiden | www.hosiden.co.jp | Japan | mic, rcv, spk |
| Bujeon | www.bujeon.com | Korea | mic, rcv, spk |
| Keyrin | www.keyrin.com | Korea | rcv, spk |
| YiLi, IEA | www.yili-e.com,<br>www.ieahk.com.hk | China,<br>Hong Kong | mic |

Abbreviations:

mic: microphones
rcv: receiver
spk: speaker