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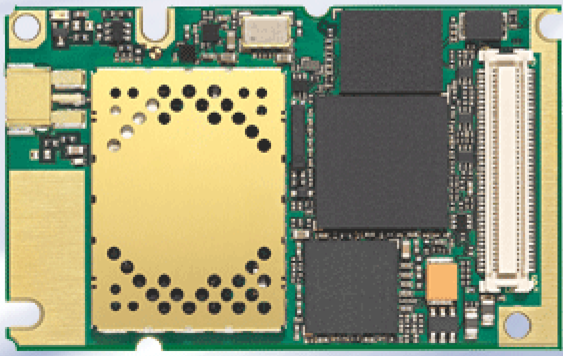

Application Note 02

Audio Interface Design for GSM Applications

Siemens Cellular Engine

Version:	03
DocId:	AC65_AC75_AN02_Audio_v03
Supported Products:	AC65, AC75

Application Note 02:	**Audio Interface Design for GSM Applications**
Version:	**03**
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Status	**Confidential / Released**
Supported Products:	**AC65, AC75**

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0 Document History

New document: AN02: Audio Interface Design for GSM Applications, Version **03**

Preceding document: AN02: Audio Interface Design for GSM Applications, Version 02

Chapter	What is new
4.8.1	Added sample circuit to connect via digital interface.

New document: AN02: Audio Interface Design for GSM Applications, Version 02

Preceding document: AN02: Audio Interface Design for GSM Applications, Version 01

Chapter	What is new
Throughout document	Added AC65 as supported product.
2, 6.3.2	Figure 1 and Figure 30: Modified the position of inCalibrate and outCalibrate in both block diagrams.
3.1	Specified jitter from an ideal 512kHz clock. Corrected description of master PCM timing with long or short frame selected.
3.2	Changed description of slave mode.
4.1	Replaced previous Figure 10 with new Figure 10 and Figure 11. Added recommended values.
4.8	Modified Figure 23.
6.1	Audio mode 6 is now named VDA Hands-free mode (see Table 4).
6.2	Added new parameters for command AT^SAIC.
6.3	Modified default setting for audio mode 6 in Table 6.
8	Added new chapter: VDA Hands-free Mode Details.

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 speakerphone 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

This Application Note applies to the following Siemens products:

- AC65 Module
- AC75 Module

1.2 Related Documents

[1] Hardware Interface Description related to your Siemens product
[2] AT Command Set related to your Siemens product
[3] Release Notes related to the firmware of your Siemens GSM module
[4] VDA Specification for Car Hands-free Terminals, December 2004, Version 1.5

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 in Section 1.1 have been type approved for use with the Siemens reference equipment described in chapter 8. Regarding audio performance, compliance with the TS 51010-1 specification and GCF recommendations has been certified for the parameters provided by audio mode 1 and audio interface 1. The settings are optimized for the reference handset (type Votronic) connected to the evaluation kit DSB75. To ensure that the reference parameters are always within the limits demanded by the standards they cannot be changed by AT command. Furthermore, the reference parameters are set as factory default.

If the customer's application is a mobile device with inbuilt microphone and speaker it needs to be compliant with the TS 51010-1 requirements specified for frequency response, loudness, listener side tone and noise suppression. All measurements shall be done with the mobile device connected to the artificial mouth and the artificial ear test set. Siemens offers assistance in adjusting the internal DSP to ensure that mobile devices with internal microphone and loudspeaker are fully standard compliant. This is achieved by creating a customized audio parameter file which can be downloaded onto each mobile device. The service is available on request, for further information please contact Siemens.

If the customer's application connects to an external headset or speakerphone it is not necessary to fulfill the above GSM specifications. Yet, to achieve a high standard of audio quality in the final product and to shorten the time of development Siemens is offering the same audio parameter customization service. For this kind of applications, the focus is on customizing frequency response, echo cancellation and noise reduction parameters according to the properties of the connected peripherals.

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

1.4 Abbreviations

Table 1: 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
DSB75	Development Support Board 75
EMC	Electro Magnetic Compatibility
EMI	Electromagnetic Interference
EPP	Earpiece Positive
EPN	Earpiece Negative
ESD	Electrostatic discharge
FFC	Flat Flexible Cable
GCF	Global Certification Forum
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
RF	Radio Frequency
SIM	Subscriber Identifier Module
SNR	Signal-to-noise ratio
TDD	Time Division Duplex
VANA	Voltage Analog
VREF	Voltage Reference

A complete list of abbreviations is provided in [1].

2 Overview of Audio Interfaces

The GSM module comprises three audio interfaces:

- A serial digital audio interface (DAI) designed for PCM (Pulse Code Modulation).
- Two analog audio interfaces, each with one analog microphone input and one analog speaker output.

This means you can connect up to three different audio devices, although only one interface can be operated at a time. Using the AT^SAIC command you can switch back and forth between each of the analog interfaces and the digital interface. See also Section 6.2 for details on how to use AT^SAIC.

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 required, we recommend audio modes 5 or 6.

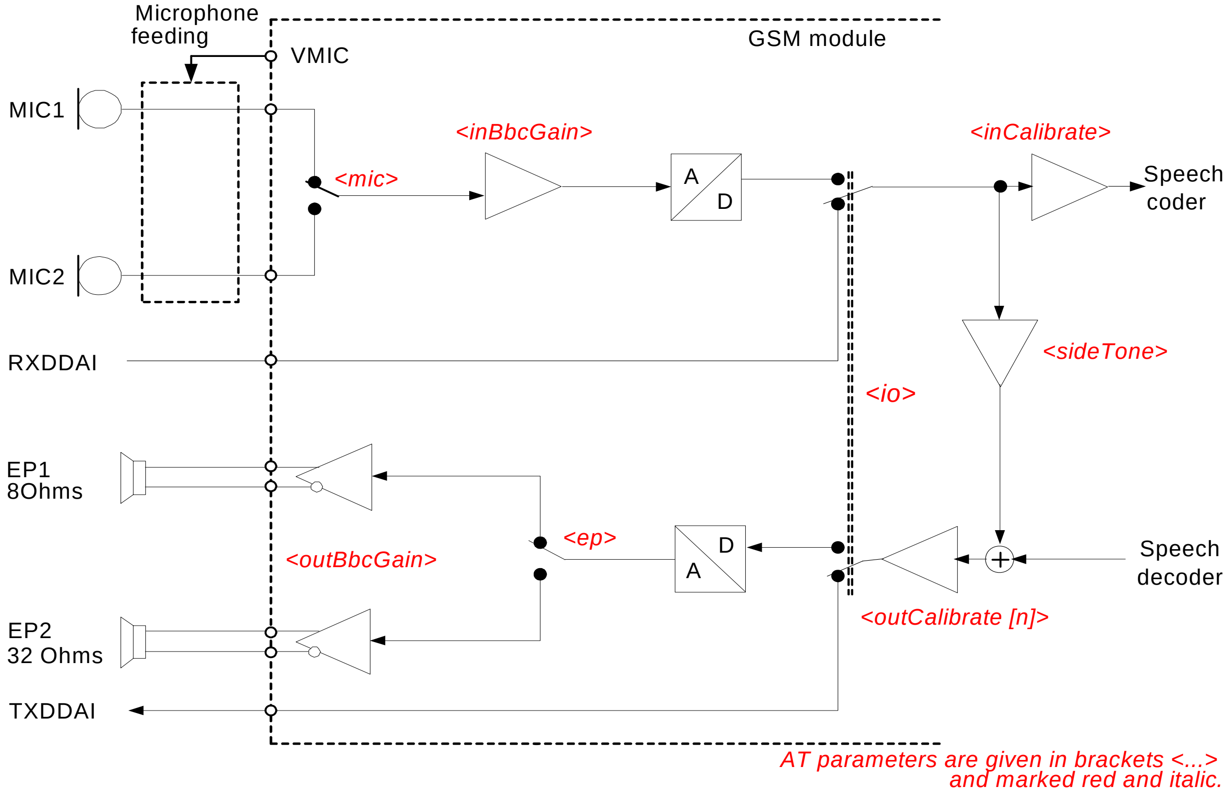


Figure 1: Block diagram of digital and analog interfaces selectable with AT^SAIC

3 Digital Audio Interface (DAI)

The DAI can be used to connect audio devices capable of PCM (Pulse Code Modulation) or for type approval. The following sections describe the PCM interface functionality.

The PCM functionality allows the use of a codec like for example the MC145483. This codec replaces the analog audio inputs and outputs during a call, if digital audio is selected by AT^SAIC.

The PCM interface is configurable with the AT^SAIC command (see [2]) and supports the following features:

- Master and slave mode
- Short frame and long frame synchronization
- 256 kHz or 512 kHz bit clock frequency

For the PCM interface configuration the parameters <clock>, <mode> and <framemode> of the AT^SAIC command are used. The following table lists possible combinations:

Table 2: Configuration combinations for the PCM interface

Configuration	<clock>	<mode>	<framemode>
Master, 256kHz, short frame	0	0	0
Master, 256kHz, long frame	0	0	1
Master, 512kHz, short frame	1	0	0
Master, 512kHz, long frame	1	0	1
Slave, 256kHz, short frame	0 or 1[1]	1	0
Slave, 256kHz, long frame	0 or 1	1	1
Slave, 512kHz, short frame	0 or 1	1	0
Slave, 512kHz, long frame	0 or 1	1	1

1. In slave mode the BCLKIN signal is directly used for data shifting. Therefore, the clock frequency setting is not evaluated and may be either 0 or 1.

In all configurations the PCM interface has the following common features:

- 16 Bit linear
- 8 kHz sample rate
- the most significant bit MSB is transferred first
- 125 µs frame duration
- common frame sync signal for transmit and receive

Table 3 shows the assignment of the DAI0…6 pins to the PCM interface signals. To avoid hardware conflicts different pins are used as inputs and outputs for frame sync and clock signals in master or slave operation. The table shows also which pin is used for master or slave. The data pins (TXDAI and RXDAI) however are used in both modes. Unused inputs have to be tied to GND, unused outputs must be left open.

Table 3: Overview of DAI pin functions

Signal name on B2B connector	Function for PCM Interface		Input/Output
DAI0	TXDAI	Master/Slave	O
DAI1	RXDAI	Master/Slave	I
DAI2	FS (Frame sync)	Master	O
DAI3	BITCLK	Master	O
DAI4	FSIN	Slave	I
DAI5	BCLKIN	Slave	I
DAI6	nc		I

3.1 Master Mode

To clock input and output PCM samples the PCM interface delivers a bit clock (BITCLK) which is synchronous to the GSM system clock. The frequency of the bit clock is 256kHz or 512kHz. Any edge of this clock deviates less than ±100ns (Jitter) from an ideal 256-kHz clock respectively deviates less than ±320ns from an ideal 512kHz clock.

The frame sync signal (FS) has a frequency of 8 kHz and is high for one BITCLK period before the data transmission starts if short frame is configured. If long frame is selected the frame sync signal (FS) is high during the whole transfer of the 16 data bits. Each frame has a duration of 125µs and contains 32 respective 64 clock cycles.

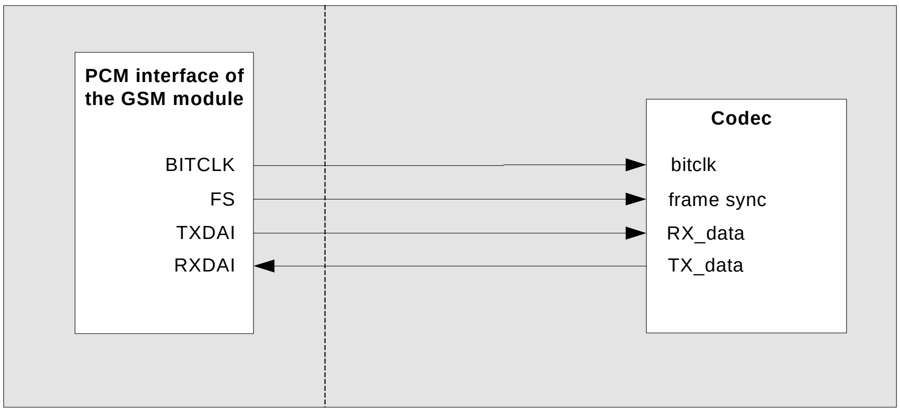


Figure 2: Master PCM interface Application

The timing of a PCM short frame is shown in Figure 3. The 16-bit TXDAI and RXDAI data are transferred simultaneously in both directions during the first 16 clock cycles after the frame sync pulse. The duration of a frame sync pulse is one BITCLK period, starting at the rising edge of BITCLK. TXDAI data is shifted out at the next rising edge of BITCLK. RXDAI data (i.e. data transmitted from the host application to the module's RXDAI line) is sampled at the falling edge of BITCLK.

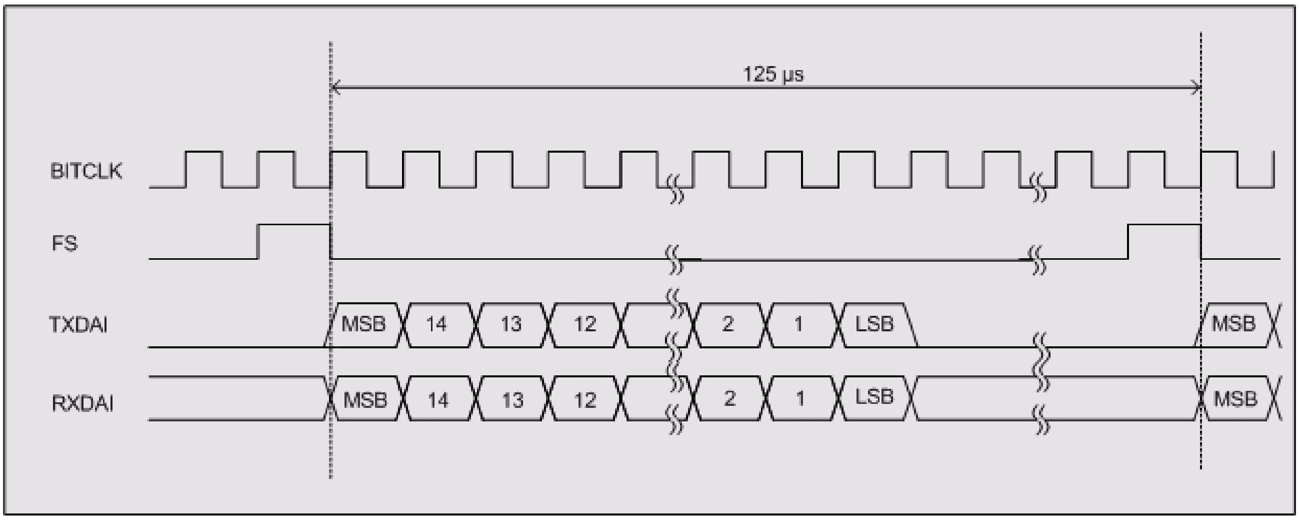


Figure 3: Short Frame PCM timing

The timing of a PCM long frame is shown in Figure 4. The 16-bit TXDAI and RXDAI data are transferred simultaneously in both directions while the frame sync pulse FS is high. For this reason the duration of a frame sync pulse is 16 BITCLK periods, starting at the rising edge of BITCLK. TXDAI data is shifted out at the same rising edge of BITCLK. RXDAI data (i.e. data transmitted from the host application to the module's RXDAI line) is sampled at the falling edge of BITCLK.

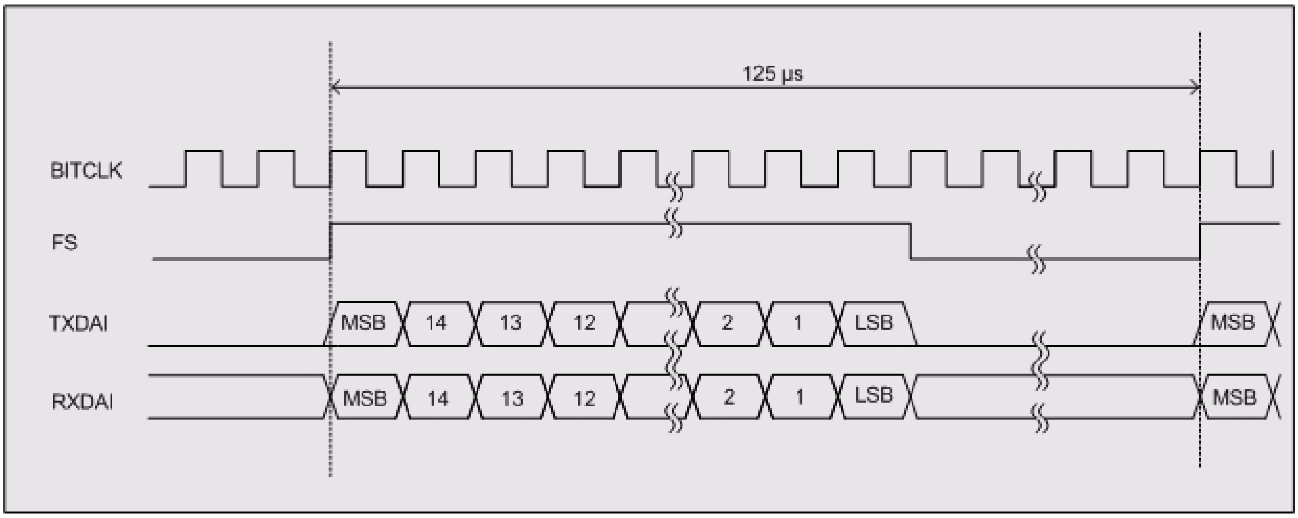


Figure 4: Long Frame PCM timing

3.2 Slave Mode

In slave mode the PCM interface is controlled by the external bit clock and the external frame sync signal applied to the pins BCLKIN and FSIN and delivered either by the connected codes or another source. The bit clock frequency can be in the range of 256kHz -125ppm to 512kHz +125ppm.

Data transfer starts at the falling edge of FSIN if the short frame format is selected, and at the rising edge of FSIN if long frame format is selected. With this edge control the frame sync signal is independent of the frame sync pulse length.

TXDAI data is shifted out at the rising edge of BCLKIN. RXDAI data (i.e. data transmitted from the host application to the module's RXDAI line) is sampled at the falling edge of BCLKIN.

The deviation of the external frame rate from the internal frame rate must not exceed ±125ppm. The internal frame rate of nominal 8kHz is synchronized to the GSM network.

The difference between the internal and the external frame rate is equalized by doubling or skipping samples. This happens for example every second, if the difference is 125ppm.

The resulting distortion can be neglected in speech signals.

The pins BITCLK and FS remain low in slave mode.

Figure 5 shows the typical slave configuration. The external codec delivers the bit clock and the frame sync signal. If the codec itself is not able to run in master mode as for example the MC145483, a third party has to generate the clock and the frame sync signal.

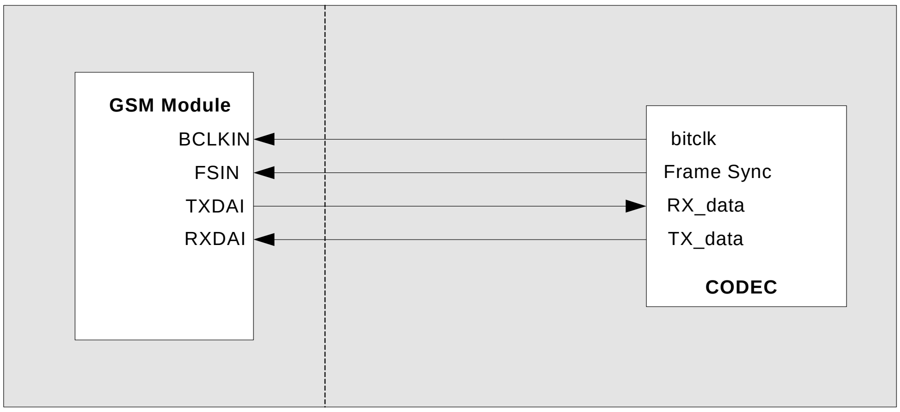


Figure 5: Slave PCM interface application

The following figures show the slave short and long frame timings. Because these are edge controlled, frame sync signals may deviate from the ideally form as shown with the dotted lines.

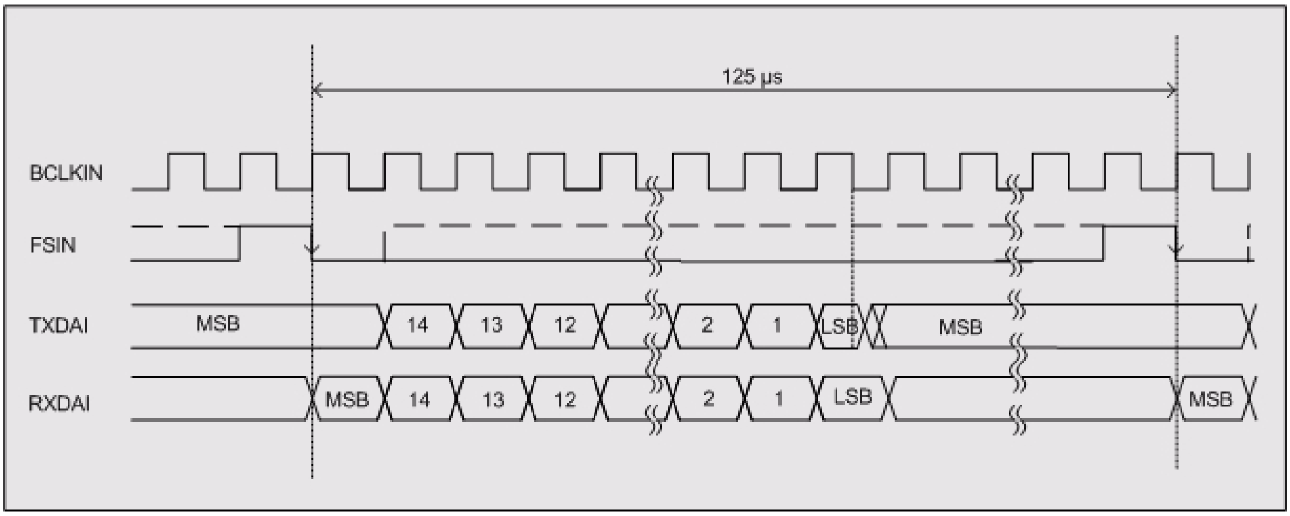


Figure 6: Slave PCM Timing, Short Frame selected

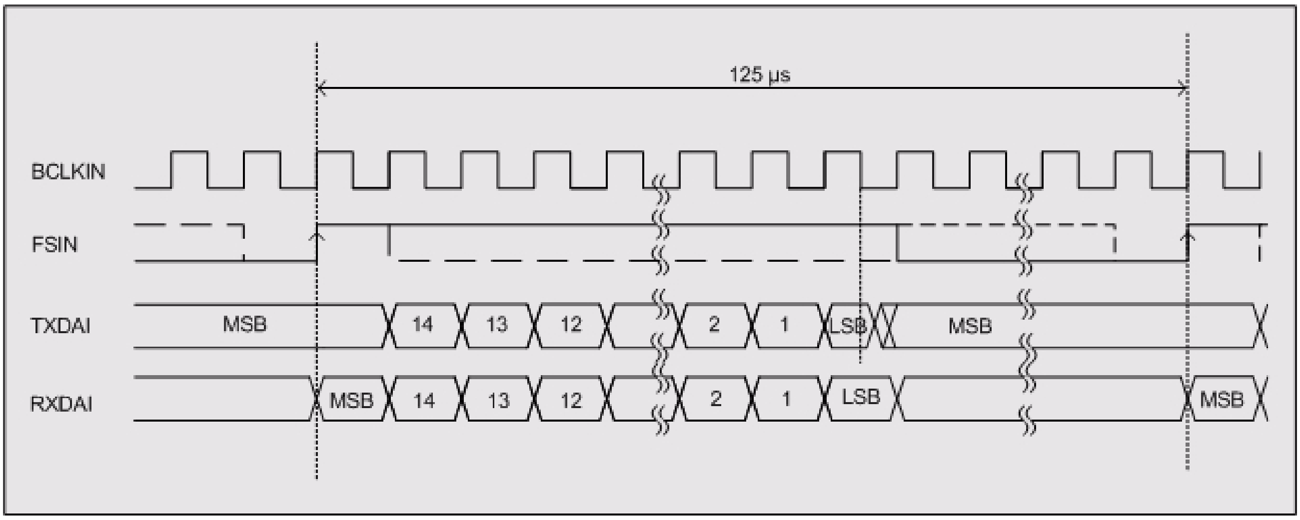


Figure 7: Slave PCM Timing, Long Frame selected

3.3 DAI Example

Figure 8 and Figure 9 show an example of using the digital audio interface of the module. The Motorola codec MC145483 can be replaced by a DSP or a codec with compatible interface. Framesync master is the module (FS line) and thus the GSM network.

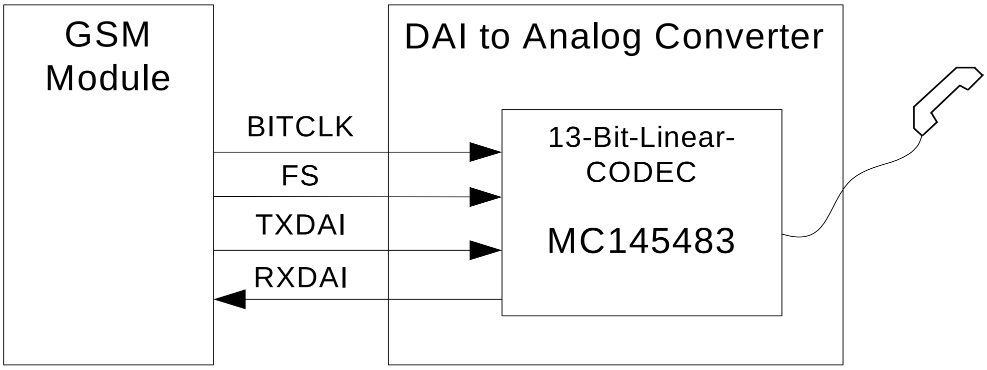


Figure 8: Block circuit for DAI to analog converter

This DAI analog converter is well suited for evaluating and testing a telephone handset and can be used instead of the headset interface of the DSB75.

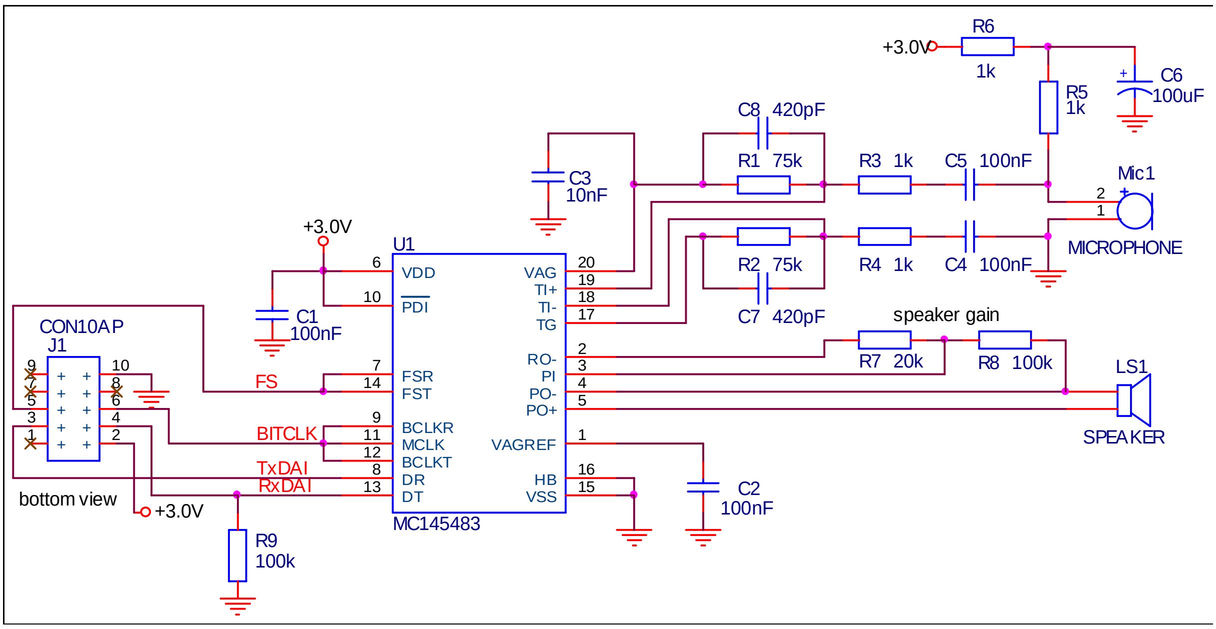


Figure 9: Sample circuit for analog to DAI box

The logical levels and the interface at connector J1 are compatible to the DSB75 DAI interface.

4 Analog Audio Interfaces

To employ audio interfaces 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 900/1800Hz bands. External ESD protection is required to protect the microphone from damage. Even a high-quality microphone should be placed at least 5cm away from the antenna.

Keep in mind that big level differences between the two audio inputs may cause cross talk from the higher level input to the lower level input.

4.1 General Usage of Microphone Interface 1 and 2

The microphone interfaces MIC1 and MIC2 are identical. MIC1 corresponds to EP1 and MIC2 corresponds to EP2, respectively (see also Figure 1).

To reduce or increase the gain of the module you may use the AT^SNFI command (compare Section 6.3.3).

Biasing is needed for the input voltage MICPx.

Recommended bias voltage: 1.0V … 1.6V

Figure 10 and Figure 11 show two examples in which the bias is derived from the internal voltage supply VMIC via a simple resistor network. Additional capacitors can be applied to block the DC voltages.

Note: The voltage VMIC needs filtering if gains larger than 20dB are used. The filter can be realized as a simple first order *RC*-network (Figure 10, Figure 11: R_1 and C_1).

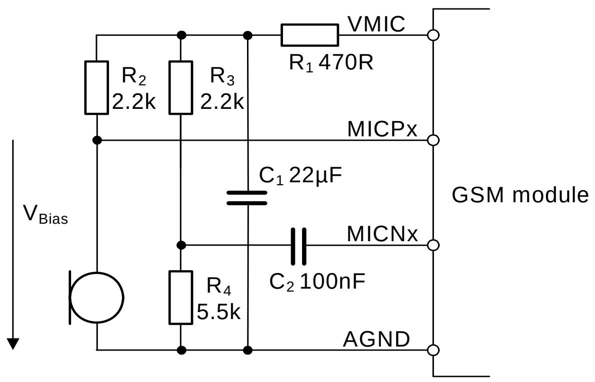


Figure 10: Recommended values for a typical electret microphone

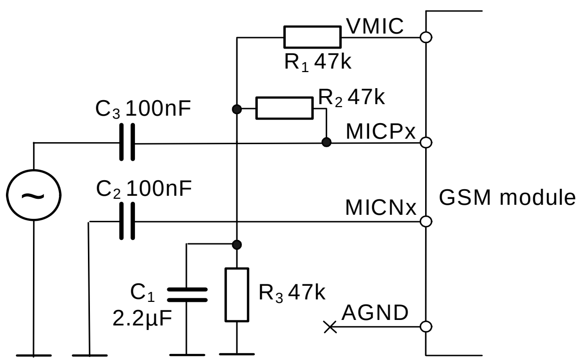
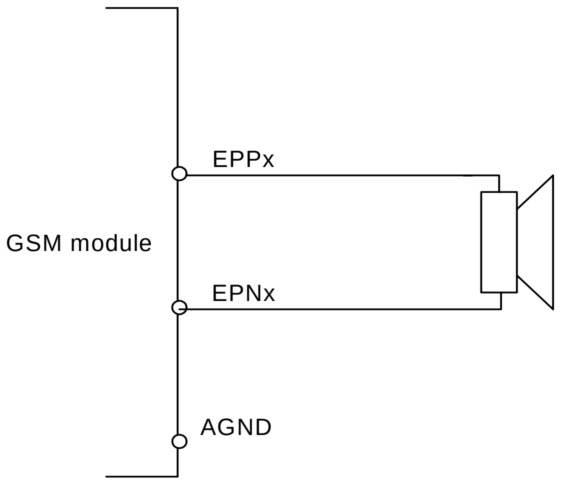


Figure 11: Recommended values for line input configuration with OpAmp

4.2 General Usage of the Loudspeaker Interface

The GSM module comprises two analog speaker outputs: EP1 and EP2. Output EP1 is able to drive a load of 8Ohms while the output EP2 is able to drive a load of 32Ohms. Each interface can be connected in balanced and in single ended configuration (see figures below).

If single ended configuration is used, AGND must be separated from any other GND. In general a balanced configuration is recommended. For line output configuration the use of EP2 is recommended, because of better power supply rejection compared to EP1.



Loudspeaker impedance

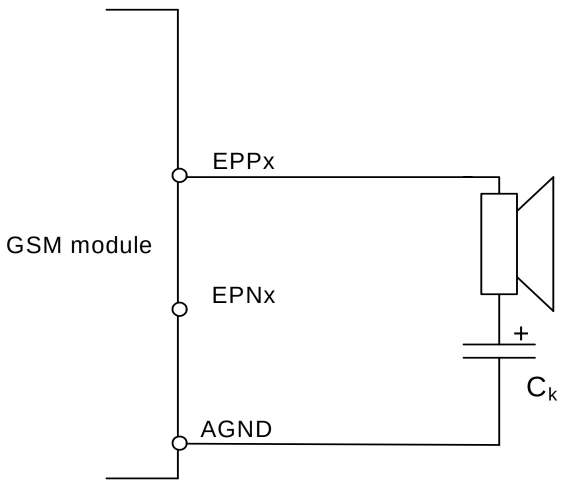
EPP1/EPN1

Z_L = typ. 8Ohm

EPP2/EPN2

Z_L = typ. 32Ohm

Figure 12: Balanced loudspeaker configuration



Loudspeaker impedance

EPP1/EPN1

Z_L = typ. 8Ohm

C_k = 220µF

EPP2/EPN2

Z_L = typ. 32Ohm

C_k = 47µF

Figure 13: Single ended loudspeaker configuration

4.3 Simple Microphone Feeding

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

Currently available microphones need a simple feeding circuit to operate. Due to the high sensitivity of microphones, these circuits shall be designed with special care. For details see Figure 15 and Figure 16. Figure 14 presents a solution for audio input 1.

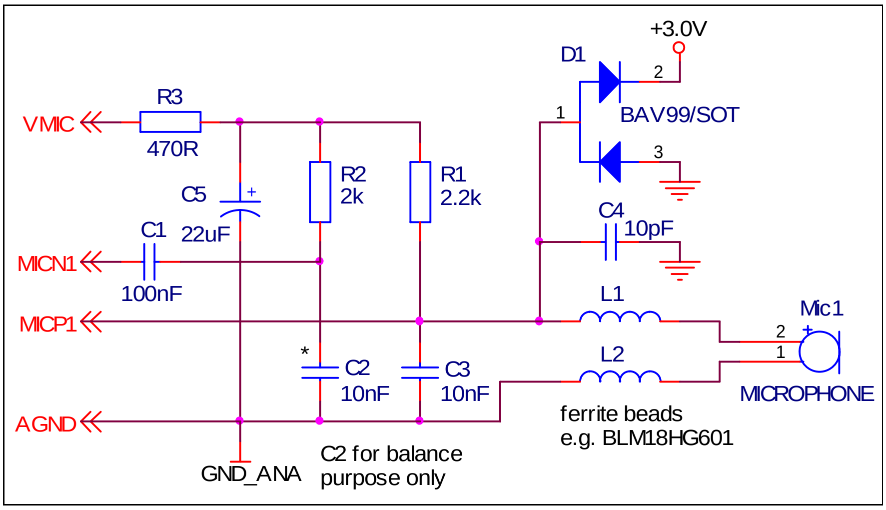


Figure 14: Circuit of microphone feeding at the module's audio interface 1 or 2

D1 and/or C3 supply ESD protection especially for the sensitive microphone. C4 has been added to suppress the demodulation at diode D1. R2 and C2 form a balance bridge to reduce Time Division Duplex (TDD, 217Hz) noise influence from VMIC feeding supply. C1 together with R1 and the microphone supply the bias voltage for GSM module at MICP1. Bias voltage 1.0V…1.6V for GSM module must always be supplied externally. Ferrite beads L1 and L4 (e.g. Murata BLM18HG601) are recommended to protect the circuit from RF intrusion.

If there is a large distance between the microphone and the module, then an additional opamp is recommended. The opamp shown in Figure 15 should be placed as far as possible from the antenna.

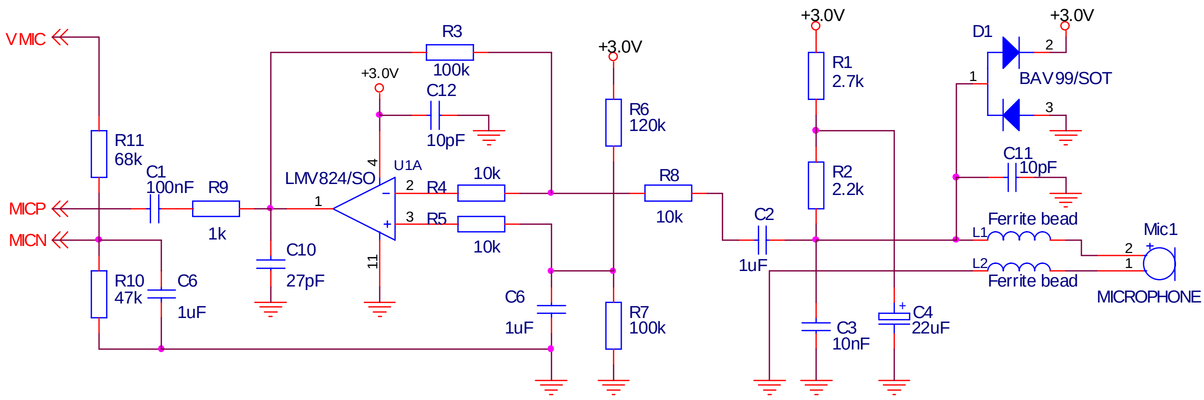


Figure 15: Circuit of microphone feeding followed by a pre-amplifier near microphone

For the example shown in Figure 15, 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 GND plane next to these ferrite beads and to the microphone pads.

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4.4 Balanced Microphone Feeding

Figure 16 shows a balanced feeding circuit for a microphone. The solution of Figure 16 is realized on the DSB75.

Balanced microphone feeding is recommended if the cable of the final product (for example a GSM desktop telephone) is not shielded and exceeds 10cm.

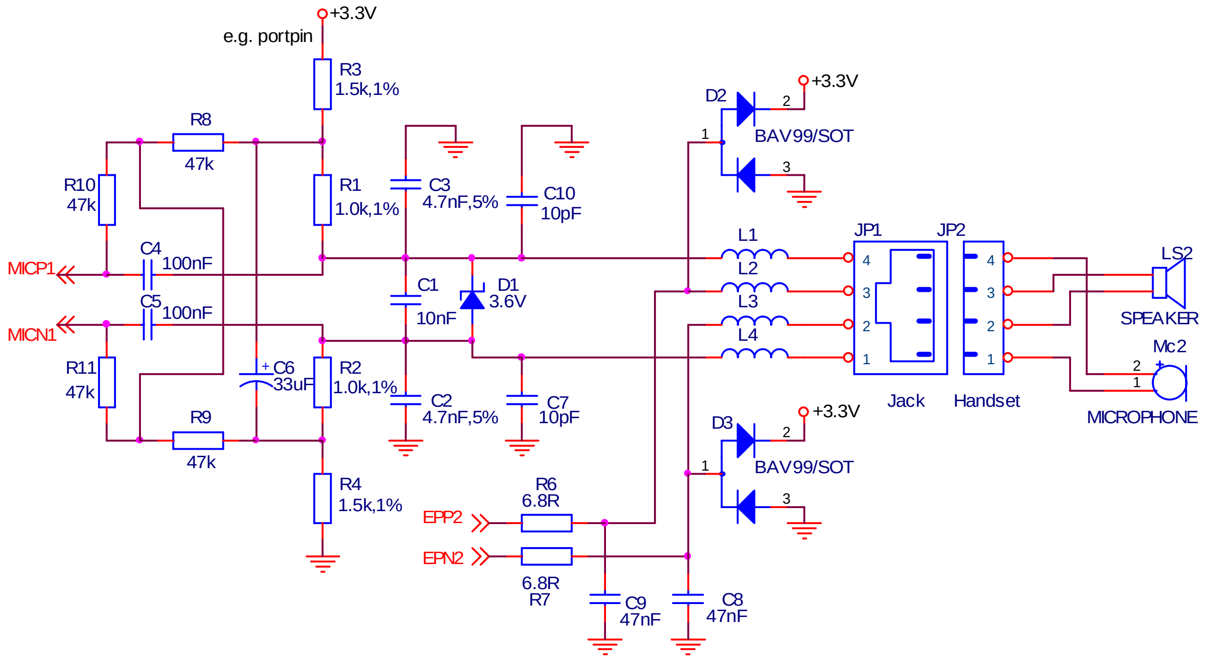


Figure 16: Circuit of a balanced microphone and earpiece feeding

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. The coupling capacitors C4 and C5 should be at least 100nF. R8…R11 supply the bias voltage for the inputs.

4.5 Galvanic Coupling of the Speaker Signal

The GSM module is able to directly drive speakers. The frequency response and loudness largely depends on the measurement method, casing, fitting and impedance. An impedance of 8Ω at interface 1 and 32Ω at interface 2 is recommended. The outputs of the modules listed in Section 1.1 are not short circuit protected.

Figure 16 shows a sample circuit of speakers directly connected to the GSM module. C8 and C9 as well as D2 and D3 protect the module from ESD pulses. Ferrite beads L2 and L3 are recommended to protect the EP lines from the RF signal being caught by the speaker.

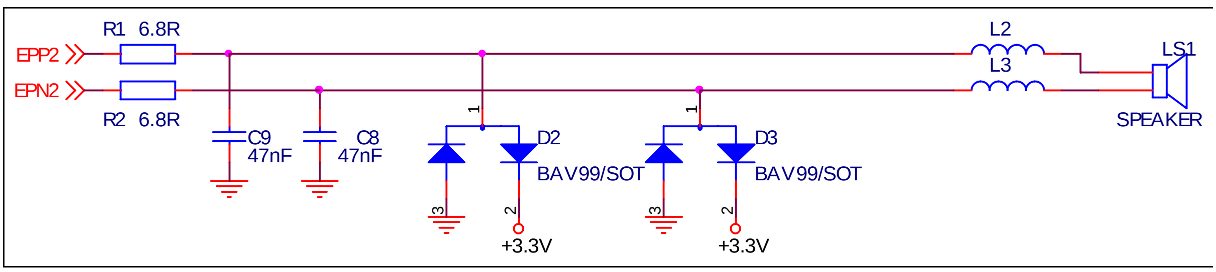


Figure 17: Sample of a module output driving a speaker

4.6 Decoupling the Speaker Signal

The speaker signal is balanced. Due to ground bouncing between module and other GND this signal cannot be simply used as unbalanced signal (see also Section 4.9.1). Therefore it is recommended to apply an opamp which transforms the GND reference point.

For line output configuration the use of EP2 is preferred, because of better power supply rejection compared to EP1.

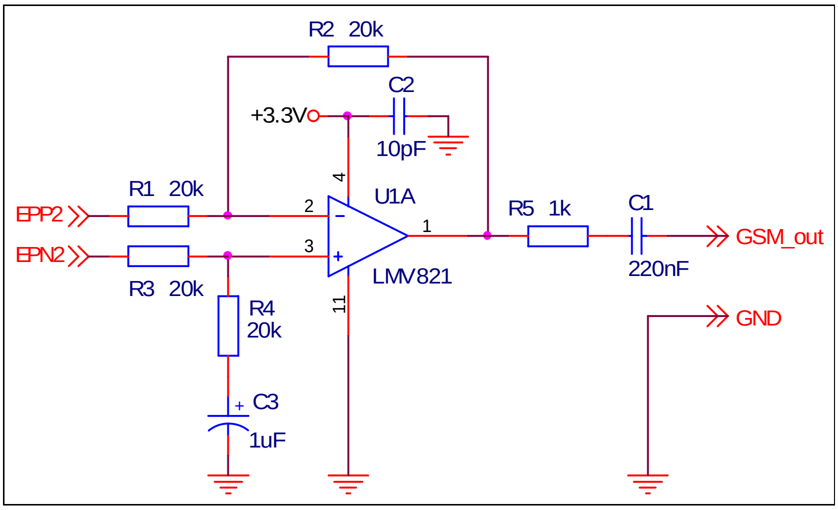


Figure 18: Circuit for transforming balanced into unbalanced – sample 1

R5 in this picture protects the opamp against RF coming from other circuits.

Often a simpler circuit for transforming balanced into unbalanced signals is sufficient. The circuit according to Figure 19 is suitable e.g. for connecting the balanced outputs to an audio codec, like AC97, if no digital audio interface can be used.

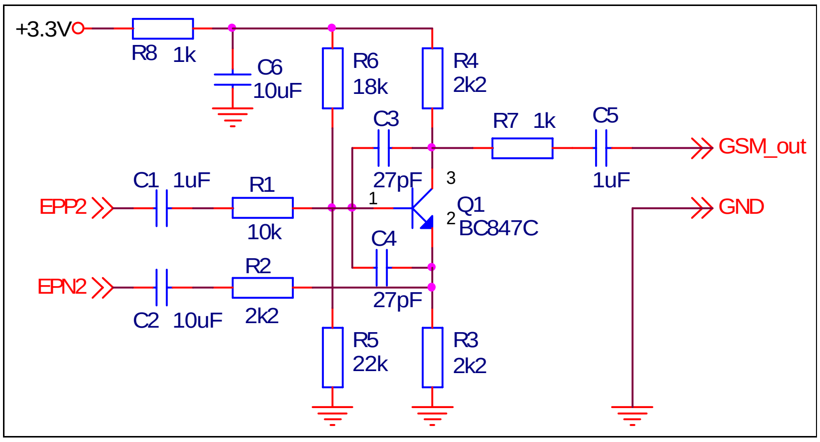


Figure 19: Circuit for transforming balanced into unbalanced – sample 2

Figure 20 shows a sample circuit of headsets connected to the GSM module. C10, R1, R2 avoid RF-demodulation at the output of U1 (see also Section 4.9.2).

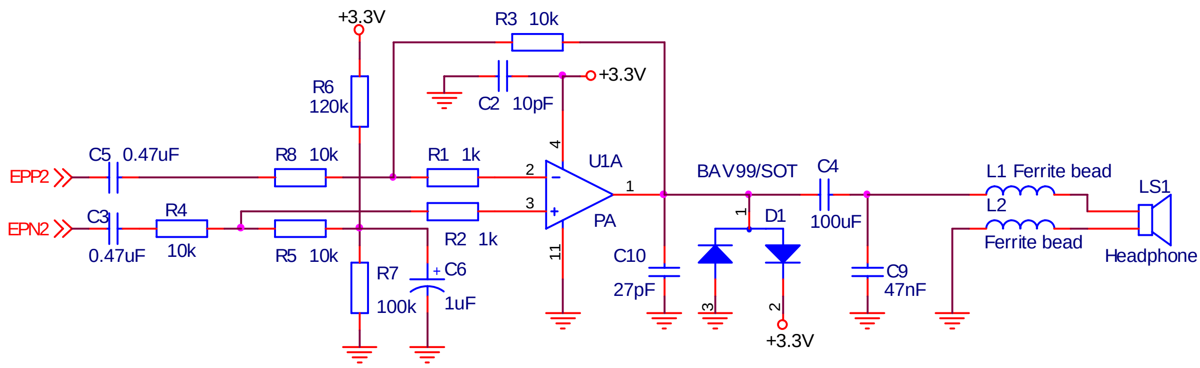


Figure 20: Circuit for connecting a headset

4.7 Stereo Headphone Concept

To use the modules' balanced speaker output together with different other signal sources (e.g. MP3 stereo source) the following concepts are suggested:

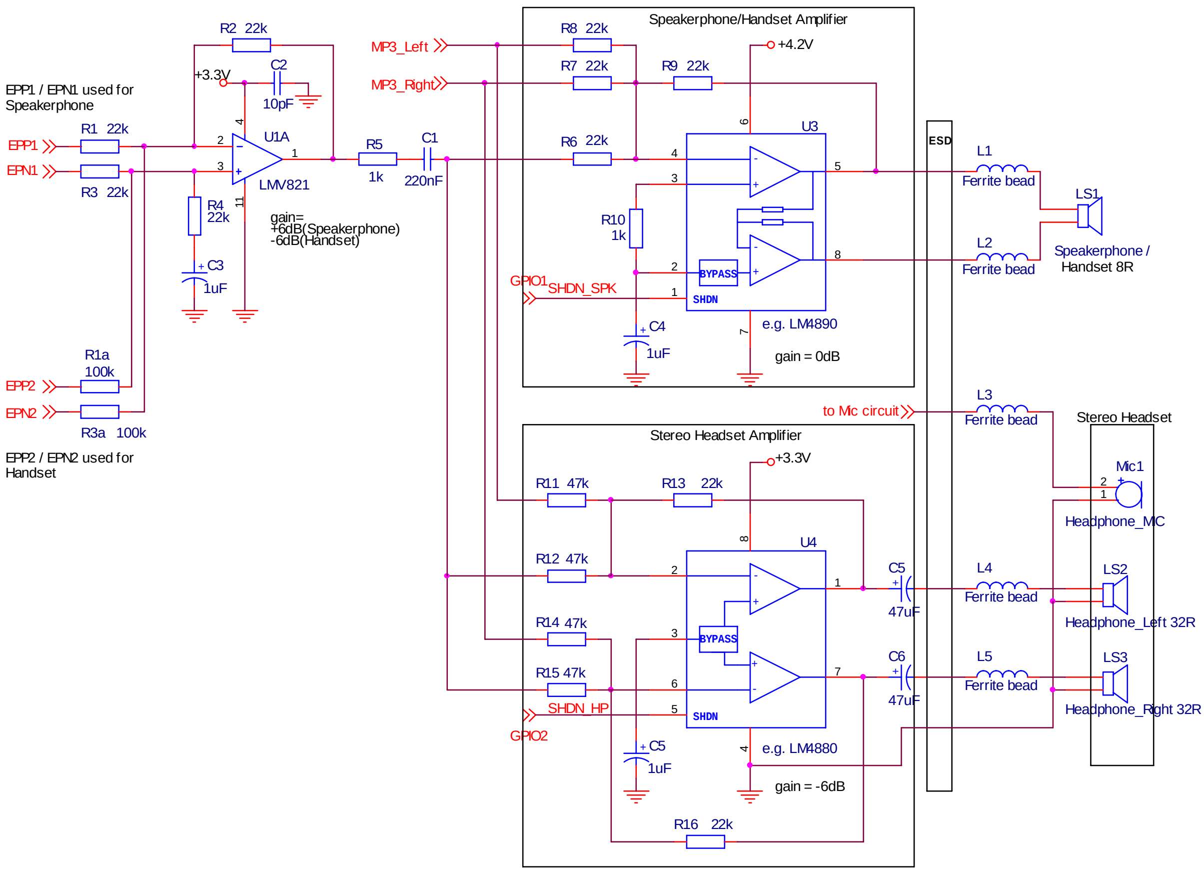


Figure 21: Circuit concept for stereo headphone - sample 1

EPP1/EPN1 and EPP2/EPN2 are used for a simple gain switch mechanism that can be used for switching between speakerphone and handset mode. U1 converts the balanced signal into an unbalanced one. This unbalanced audio signal is split into speaker and headphone amplifier (U3 and U4). U3 can play back GSM sound as well as mono sound from any other source.

Figure 22 offers a similar flexibility as sample 1 but is less complex. The disadvantage is the need for bigger electrolytic capacitors.

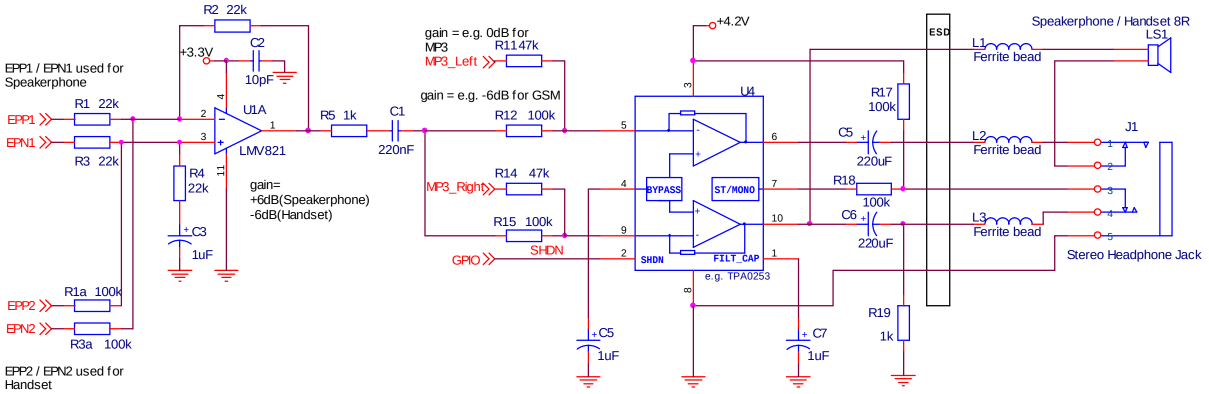


Figure 22: Circuit concept for stereo headphone - sample 2

4.8 Sample Circuit for a PDA Application Using an AC97 Codec

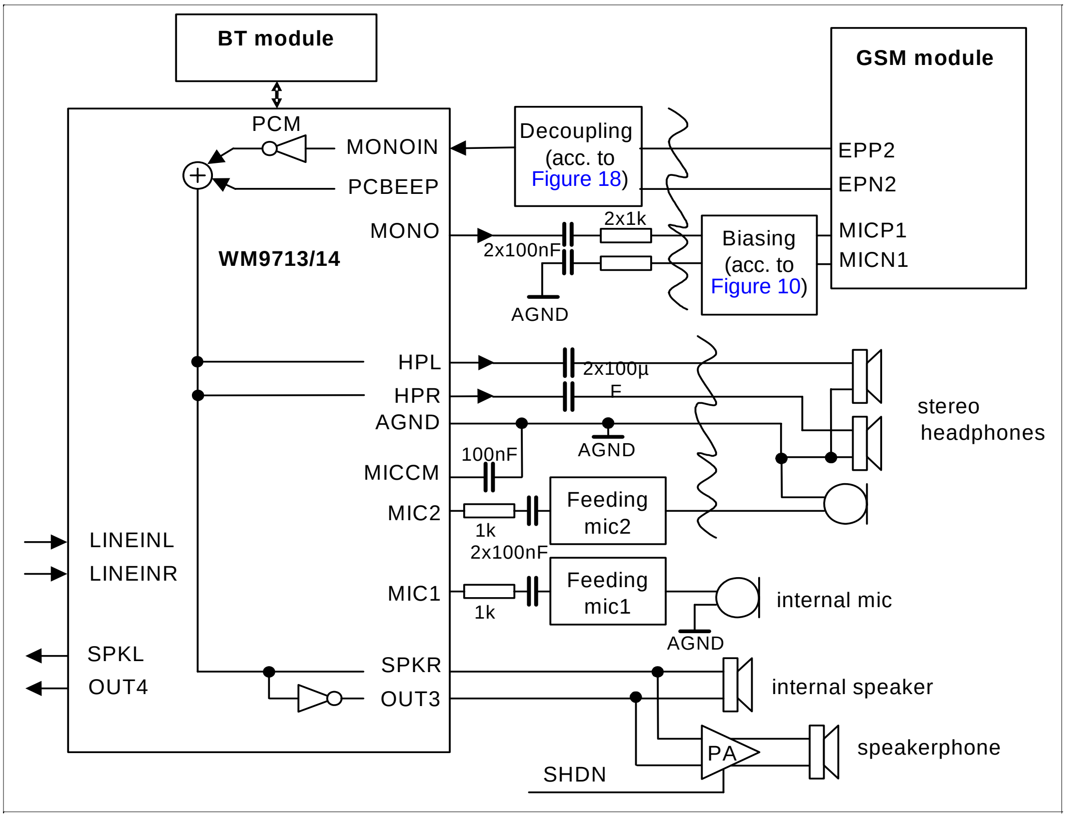


Figure 23: Block circuit of a complex application using WM9713 or WM9714 codec

The complex codec from Wolfson incorporates already many of the commonly needed audio functions. If a complex functionality is requested, it is recommended to use the capabilities of these especially designed chips.

One of the benefits of the Wolfson solution is the possibility of substracting signal EPP1 and EPP2 before it is fed into the internal switch matrix. This saves the conversion circuit from the balanced signal into unbalanced.

4.8.1 Connecting via Digital Interface of WM9713

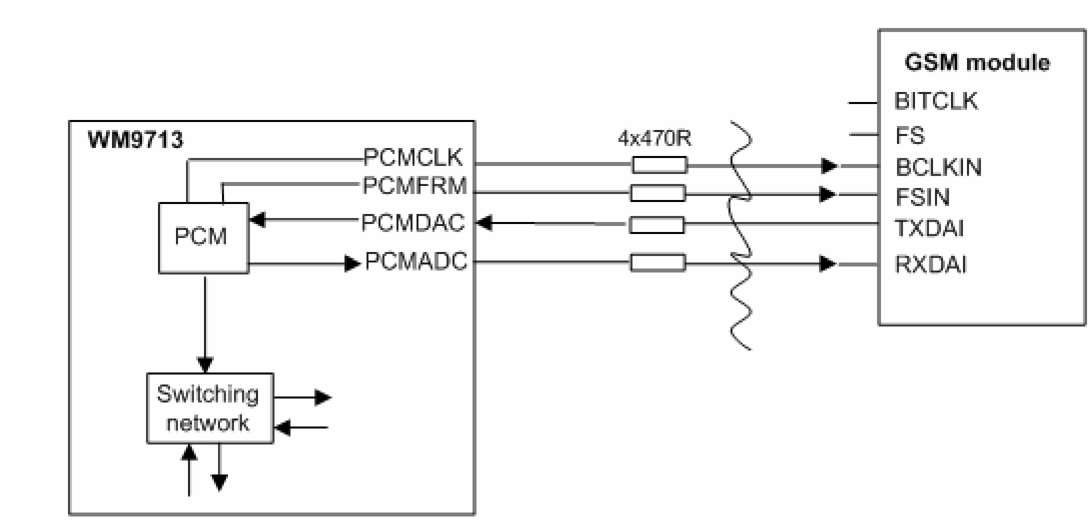


Figure 24: Block circuit of an application connecting via digital interface of the WM9713 codec

To enable the possibility of connecting via digital PCM interface the GSM module **must** be in slave mode and the codec must be in master mode.

The below sample settings connect both interfaces:

GSM module	**WM9713**
AT^SNFS=2	Reg. 36h=0xC603
AT^SAIC=1,1,1,0,1,0	

4.9 Typical Audio Issues

4.9.1 Power Supply Related Audio Problems

Audio problems may occur due to insufficient power supply filtering and different GND levels between PCBs.

The example in Figure 25 shows the problem of ground bouncing which, in principle, can affect speaker and microphone path:

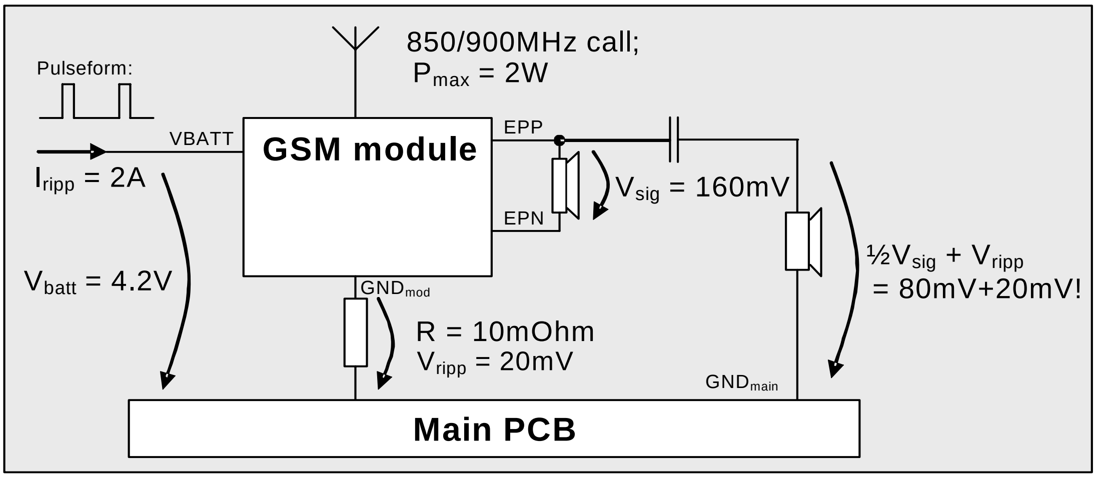


Figure 25: Ground bouncing problem (example)

The example assumes the following scenario:

- Typical peak power consumption is 2A.
- Typical connector resistance is 5…20mOhms.

In the example the speaker connected from EP to GND would have an SNR of only 12dB! This problem can be resolved by

- a balanced usage of outputs or
- by connecting the unbalanced speaker to AGND of the GSM module (see [1]).

For more information refer to Section 4.6.

4.9.2 RF Related Audio Problems

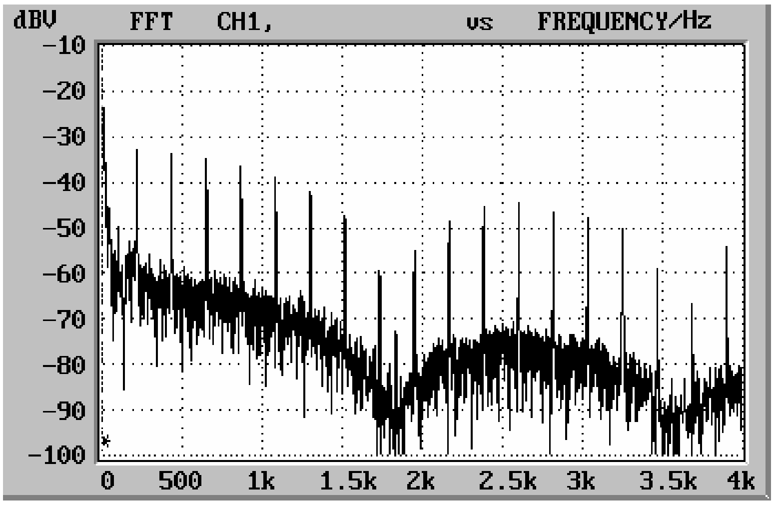


Figure 26: TDD noise/humming in the output spectrum

The microphone should be equipped with two internal EMI capacitors for GSM 900/1800 MHz bands (12…18pF + 33pF).

Semiconductors are always a source of RF demodulation. This behavior can only be cancelled by hardware and layout changes!

- Each semiconductor (e.g. ESD diodes etc.) needs to be RF-decoupled using a capacitor from 10 to 33pF.
- Serial resistors >50Ohms including the internal capacitance of the semiconductor (e.g. opamp or analog switches etc.) act as low pass for GSM frequencies (see Figure 27).
- On the PCB, these capacitors and resistors shall be placed close to the semiconductor and/or input pins.

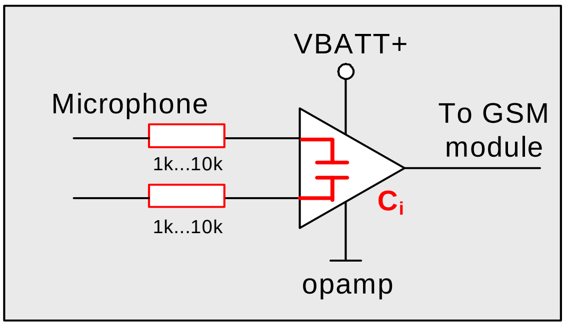


Figure 27: Recommended serial resistors for semiconductor inputs

4.9.3 How to Avoid Echo Problems at the Headset Interface

Commonly, a mono-headset uses 4 wires and a stereo-headset uses 5 wires to connect to a GSM device. Two or three wires are needed for the headphone and two for a balanced or unbalanced microphone and for the push to talk button.

Using of a 3 wire mono-headset interface or a 4 wire stereo-headset interface is possible, but not recommended. As shown in Figure 28 a 3 wire mono-headset interface may generate an electrical echo which can be heard by the far end listener.

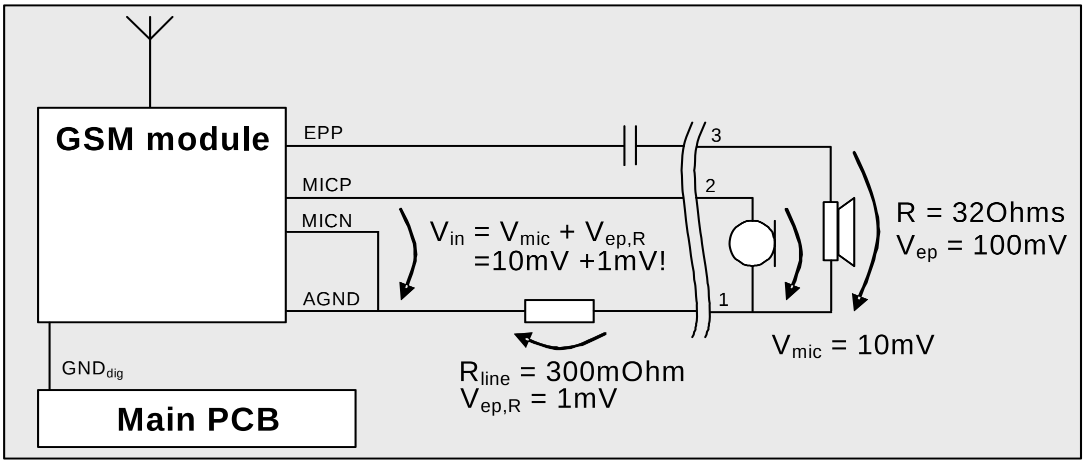


Figure 28: Principle circuit explaining the echo at 3 wire mono-headset interfaces

The example in Figure 28 assumes the following scenario:

- Typical earpiece resistance: 32Ohms
- Typical cable, connector and EMI part resistances together: 0.1…2Ohms

Recommended solution: To avoid the effect of reduced voice quality while using a 3 wire mono-headset or a 4 wire stereo-headset, Siemens recommends a 4 wire mono-headset interface or a 5 wire stereo-headset interface (either balanced headset-speaker or separated GND wires for microphone and headset-speaker)!

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 antenna 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.

Disadvantages: Value of the ignition voltage is fuzzy.

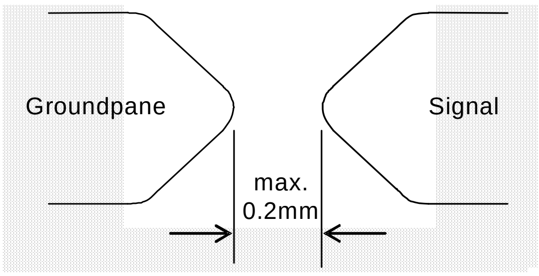


Figure 29: Spark gap

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 15, Figure 16and Figure 20.

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 in parallel.

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 to 33pF capacitor in parallel.

5.4 Z-Diodes

Especially supplied electret microphones should be protected with a Z-diode connected in parallel (see Figure 16).

Semiconductors always are a source of RF demodulation. Each semiconductor needs to be RF decoupled using a capacitor from 10 to 33pF in parallel.

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 15, Figure 16 and Figure 20).

6 Using AT Commands to Control Audio Interfaces

The audio parameters of all GSM modules 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 4 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. You have the choice of six audio modes selectable with AT^SNFS. By default, each audio mode is assigned to one of the two interfaces: audio modes 1, 4 and 5 are allocated to the 1st interface, and audio modes 2, 3 and 6 to the 2nd interface.

If this default assignment fits the design of the host application, the AT^SNFS command is sufficient to select an audio mode and to activate the interface automatically assigned to this mode.

Table 4: Selectable audio modes

No.	Name	AT command	Purpose	Description of mode and interface assigned by default
1	Default Handset	AT^SNFS=1	Approval configuration of module	MIC1, EP1, adapted to Votronic Handset HH-SI-30.3 with DSB75 Support Board, not adjustable. Default: Audio interface 1 (MIC1, EP1).
2	Basic Speakerphone	AT^SNFS=2		MIC2, EP2, adapted to Siemens CarKit Portable, 5 volume steps selectable. Default: Audio interface 2 (MIC2, EP2).
3	Headset	AT^SNFS=3		MIC2, EP2, adapted to standard Mono-Headsets, 5 volume steps selectable. Default: Audio interface 2 (MIC2, EP2).
4	User Handset	AT^SNFS=4	Approval configuration of customer application	MIC1, EP1, Votronic Handset HH-SI-30.3 with DSB75 Support Board, 5 volume steps selectable. Default: Audio interface 1 (MIC1, EP1).
5	Plain Codec	AT^SNFS=5		MIC1, EP1, no filtering, 5 volume steps selectable. Default: Audio interface 1 (MIC1, EP1).
6	VDA Hands-free	AT^SNFS=6		MIC2, EP2, no filtering, 5 volume steps selectable. Default: Audio interface 2 (MIC2, EP2)

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.

If an audio interface is operated with audio modes other than those assumed by default, the AT^SAIC command can be used in addition to AT^SNFS. AT^SAIC (see Section 6.2) selects the analog interface and changes the microphone (MIC1 or MIC2) and speaker (EP1 or EP2) activated by default. So, using both AT commands increases the combinations of audio interfaces and audio modes, giving greater flexibility to connect different devices.

6.2 Changing Physical Audio Interface

To switch back and forth between all three audio interfaces you can use the command:

AT^SAIC=<io>[, <mic>[, <ep>][, <clock>, <mode>, <frame_mode>]]

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 (see also Section 6.1 and [2]).

The factory defaults of AT^SAIC vary with the selected audio mode.

Table 5: Default values of AT^SAIC

Mode	Default settings
^SNFS=1 or 4 or 5	AT^SAIC=2,1,1,0,0,0
^SNFS=2 or 3 or 6	AT^SAIC=2,2,2,0,0,0

Examples:

AT^SAIC=1,,,0,0,1 selects the digital PCM interface in Master mode with 256kHz clock and long frame.
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.

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

6.3 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 6 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 6: Default values of audio modes (subject to change)

Mode	Default settings (to be queried with AT^SNFO?)
^SNFS=1	^SNFO=1,16384,16384,16384,16384,16384,4,6878 (no steps)
^SNFS=2	^SNFO=2,4096,5792,8192,11584,16384,4,0 (3dB steps)
^SNFS=3	^SNFO=1,4096,5792,8192,11584,16384,4,8192 (3dB steps)
^SNFS=4	^SNFO=1,4096,5792,8192,11584,16384,4,6878 (3dB steps)
^SNFS=5	^SNFO=0,4096,5792,8192,11584,16384,4,0 (3dB steps)
^SNFS=6	^SNFO=1,4096,5792,8192,11584,16384,4,0 (3dB steps)

6.3.1 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 scaled digital coded sine wave. For Siemens GSM modules, the correlation between the digital value and the amplitude of the analog signal at 1kHz is given 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 ~ + 2.2dBm.
- 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 (V).

  G = 10*log(P1/P2) = 20*log(V1/V2)

  Some common values / samples:

  V1/V2 = 10 → G = 20dB   P1/P2 = 10 → G = 10dB

  V1/V2 = 2 → G ≈ 6dB   P1/P2 = 2 → G ≈ 3dB

  V1/V2 = 20 → G ≈ 26dB   P1/P2 = 20 → G ≈ 13dB

6.3.2 Specifying the Value of the Volume Steps

The figure below shows the possibilities to influence the volume of the output via the command AT^SNFO.

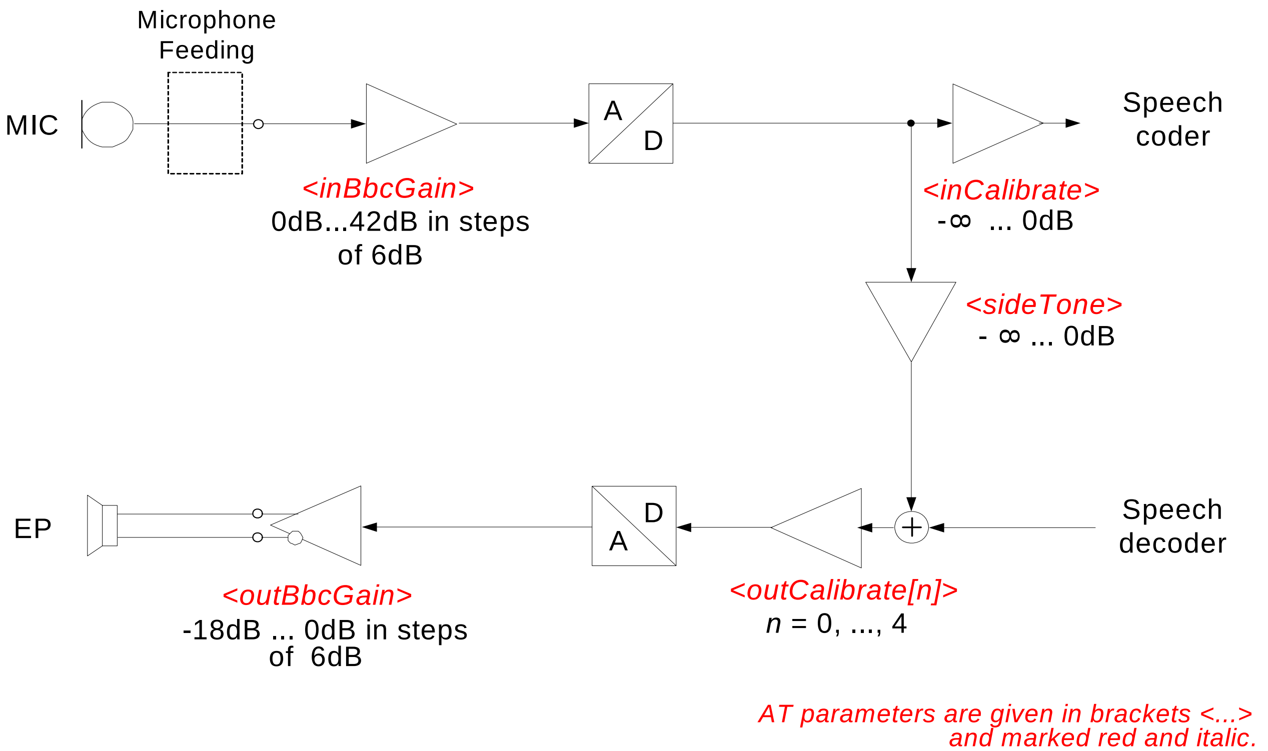


Figure 30: Audio parameters selectable with AT commands

Examples:

AT^SNFS=4	(select User Handset)
AT^SNFO?	
^SNFO=1,4096,5824,8192,11616,16384,4,4096	(default 3dB steps)
5 defined <outCalibrate> volume steps	
AT^SNFS=4	(select User Handset)
AT^SNFO=1,6528,8192,10368,13056,16384,4,6878	(2dB steps)
AT^SNFW	(write to non-volatile memory)
or	
AT^SNFS=4	(select User Handset)
AT^SNFO=1,10337,11598,13014,14602,16384,4,6878	(1dB 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 the manufacturer's default values using the command:

AT^SNFD

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^SNFO=0,16384,16384,16384,16384,16384,4,4096	(default + 6 dB)
AT^SNFO=0,12288,12288,12288,12288,12288,4,4096	(default + 3 dB)
AT^SNFO=1,16384,16384,16384,16384,16384,4,4096	(default volume)
AT^SNFO=1,12288,12288,12288,12288,12288,4,4096	(default - 3 dB)
AT^SNFO=2,16384,16384,16384,16384,16384,4,4096	(default - 6 dB)
AT^SNFO=2,12288,12288,12288,12288,12288,4,4096	(default - 9 dB)
AT^SNFO=3,16384,16384,16384,16384,16384,4,4096	(default - 12 dB)
AT^SNFD	(recall manufacturer default)

6.3.3 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)

6.4 Usage of the Internal Microphone Supply

6.4.1 Configuration of the Internal Microphone Supply

The microphone power supply voltage (VMIC) is programmable via the command:

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

This gives you greater flexibility in connecting audio accessories or using the microphone interfaces 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.

The parameter <MicVccCtl> controls the power supply voltage VMIC. Note that the first parameter <MicSwitch> must be omitted when setting <MicVccCtl>.

AT^SNFM=,0: Permanently switches off the power supply at VMIC.
AT^SNFM=,1: Permanently switches on the power supply at VMIC.
AT^SNFM=,2: Default setting. Power at VMIC 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 MICP1 and MICN1 respective MICP2 and MICN2.

6.4.2 Call Handling and Internal Microphone Supply

Because of voltage protection, the absolute maximum voltage on any MIC pin is limited. While the module is in an operating state with VMIC=off and inactive audio, the voltage at any MIC pin must not exceed ±0.3V relative to AGND, i.e., audio level 0.6Vpp (see [1]) otherwise undervoltage shutdown may be caused.

Only if VMIC=on and active audio, the voltage applied to any MIC pin can be in the range of +2.7V to -0.3V with audio level of 1.6Vpp (full scale).

To meet these requirements please ensure the appropriate control of voltage levels as well as of external microphone amplifiers or codec devices connected to MIC inputs.

The following examples suggest a possible usage of VMIC control (AT^SNFM=,1):

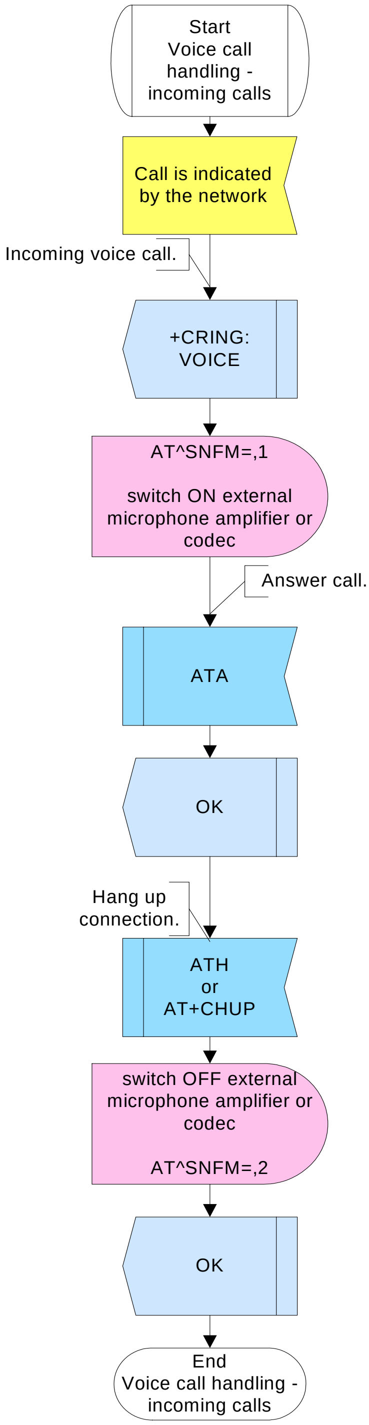


Figure 31: Example VMIC control for incoming call

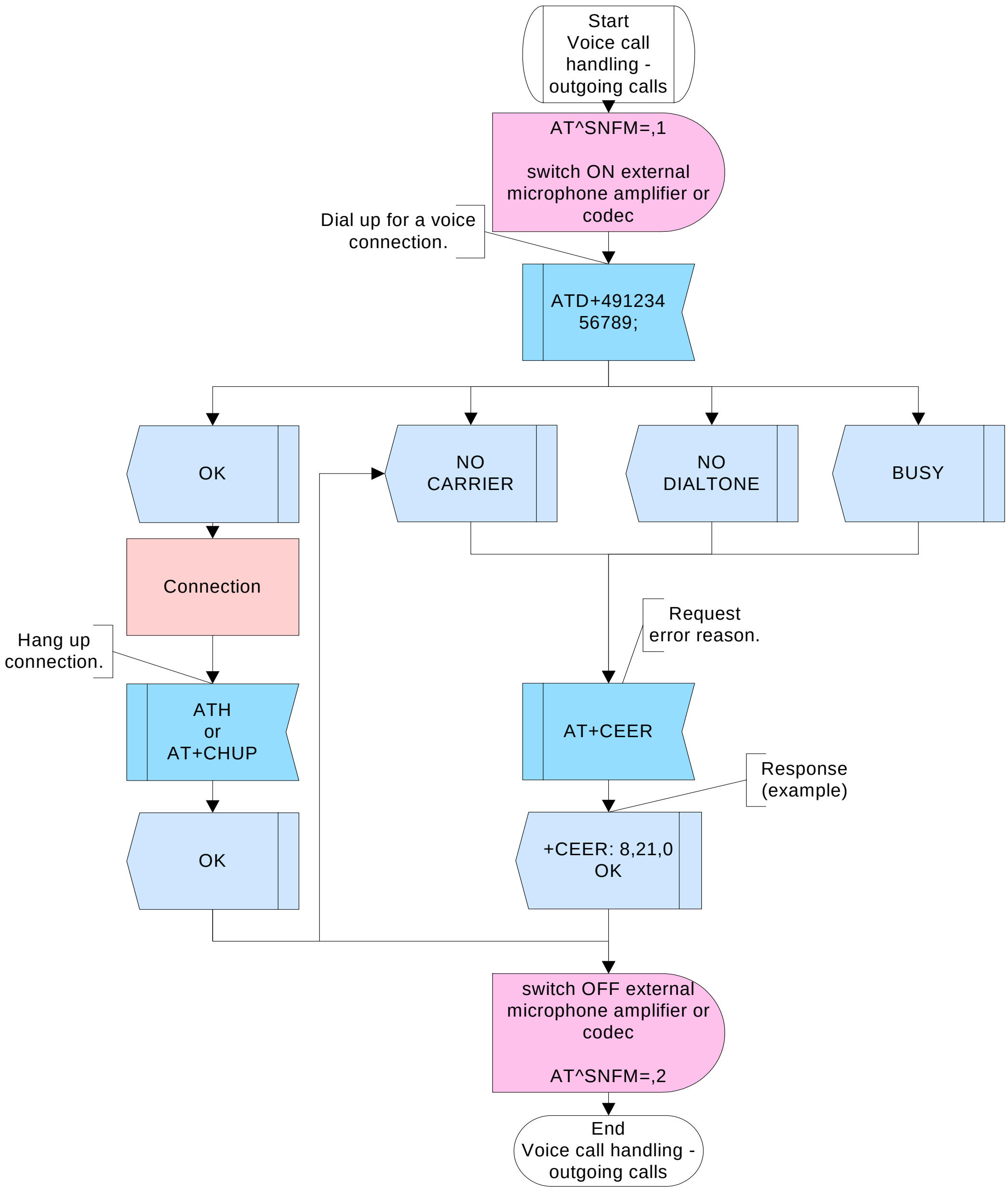


Figure 32: Example VMIC control for outgoing calls

7 Speakerphone Mode Details

The speakerphone 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. Physically, audio interface 2 (default) or 1 can be used.

EP2 output is followed by a power amplifier with 8...40dB gain. The external microphone amplifier with 26...40dB gain needs to have good noise characteristic. Final adjustments can be done easily with AT commands.

For product information on the Siemens Car Kit Portable HKP-500 see Chapter 10.

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 space of the speaker are well separated by sealing the speaker at the housing.

8 VDA Hands-free Mode Details

The VDA hands-free mode (AT^SNFS=6) is based on the VDA Specification for Car Hands-free Terminals, Version 1.5. The VDA specification defines requirements and measurement techniques for car hands-free terminals For details see [4].

The below Figure 33 shows a typical vehicle mounted hands-free application. The user is replaced by a Head and Torso Simulator (HATS) with an artificial ear and mouth.

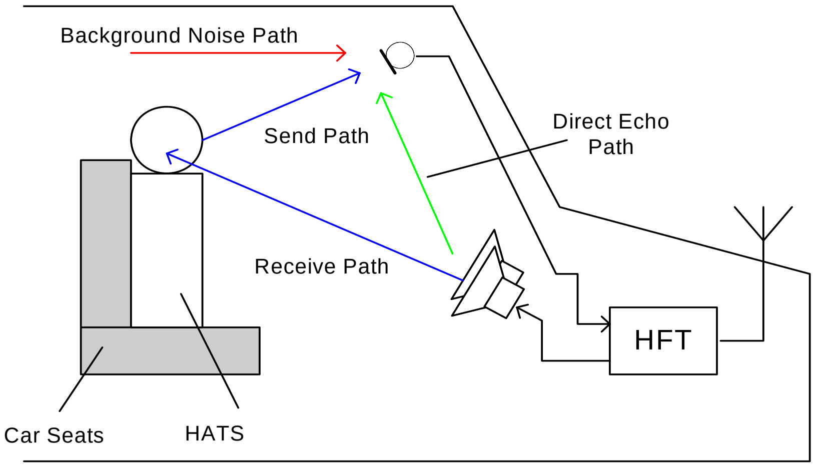


Figure 33: Vehicle mounted hands-free application

Besides the vehicle itself a hands-free application comprises several further components, namely a microphone, the microphone circuit, the vehicle's sound system with one or more loudspeakers and a hands-free terminal (HFT). The HFT contains the module, control processor and a user interface. It provides the circuit switched speech connection to the wireless network (e.g. GSM). The necessary analogue to digital converters and digital to analogue converters are also included as well as appropriate amplifiers, which adjust the gains, and a digital signal processing unit for filters, acoustic echo cancellation (AEC) and noise reduction (NR).

The pre-defined VDA hands-free audio mode (AT^SNFS=6) configures a set of audio parameters for the module that provide the required hands-free quality for a reference setup with a mid-sized station wagon. This VDA mode is therefore a good starting point for implementing customer-specific hands-free applications. However, the VDA mode's parameter settings may have to be adapted to suit individual customer environments.

9 Siemens Reference Setup

The Siemens reference GSM equipment used for type approving includes the following components (for details see Chapter 10):

- GSM module
- DSB75 Support Board (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 DSB75
- Handset type Votronic HH-SI-30.3/V1.1/0
- U.FL-R-SMT antenna connector and U.FL-LP antenna cable
- Li-Ion battery
- PC as MMI

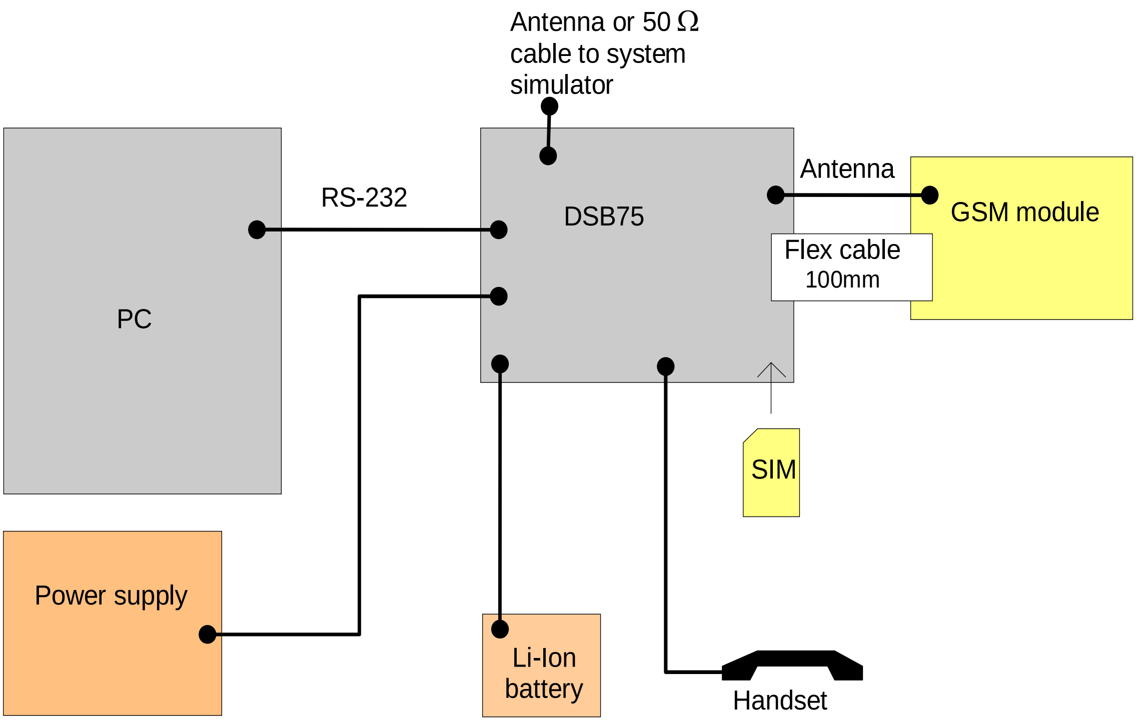
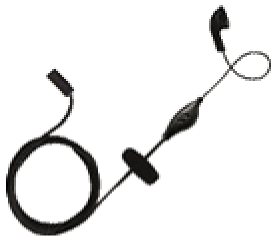


Figure 34: Siemens reference equipment

10 Parts and Accessories

The following table lists parts and accessories mentioned in this Application Note.

Table 7: Parts and Accessories

Product	Ordering Information
Siemens DSB75 Support Board 	Siemens ordering number: L36880-N8811-A100
Votronic Handset	Votronic HH-SI-30.3/V1.1 VOTRONIC Entwicklungs- und Produktionsgesellschaft für elektronische Geräte mbH Saarbrücker Str. 8 D-66386 St. Ingbert Phone: 06 89 4 / 92 55-0 Fax: 06 89 4 / 92 55-88 e-mail: contact@votronic.com
Siemens Car Kit Portable HKP-500 	Siemens ordering number: L36880-N5601-A109
Siemens Mono Headset 	Siemens ordering number, e.g., for HHS 500: L36880-N5601-A107

10.1 Suppliers of Acoustic Devices

The following table lists suppliers of acoustic devices. The list is not representative nor does it evaluate the quality of specific products. It is simply meant as a starting point for further investigation:

Table 8: Suppliers of acoustic devices

Company	URL	Country	Products
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, www.ieahk.com.hk	China, Hong Kong	mic

Abbreviations:

mic: microphones

rcv: receiver

spk: speaker