

The **INTERNET** of THINGS
made **Plug&Play**

































Telit[®] wireless
solutions



AUDIO SETTINGS APPLICATION NOTE

APPLICABILITY TABLE

PRODUCTS

		GC864 SERIES
		GE864 SERIES
		GE865 SERIES
		GL865 SERIES
		GL868 SERIES
		GE910 SERIES
		GE910-QUAD V3
		GL865-DUAL V3
		GL865-QUAD V3
		GL868-DUAL V3
		GE866-QUAD
		UL865 SERIES
		UL865-N3G
		UE910 SERIES
		HE910 SERIES
		HE910 MINI PCIE

SPECIFICATIONS SUBJECT TO CHANGE WITHOUT NOTICE

LEGAL NOTICE

These Specifications are general guidelines pertaining to product selection and application and may not be appropriate for your particular project. Telit (which hereinafter shall include, its agents, licensors and affiliated companies) makes no representation as to the particular products identified in this document and makes no endorsement of any product. Telit disclaims any warranties, expressed or implied, relating to these specifications, including without limitation, warranties or merchantability, fitness for a particular purpose or satisfactory quality. Without limitation, Telit reserves the right to make changes to any products described herein and to remove any product, without notice.

It is possible that this document may contain references to, or information about Telit products, services and programs, that are not available in your region. Such references or information must not be construed to mean that Telit intends to make available such products, services and programs in your area.

USE AND INTELLECTUAL PROPERTY RIGHTS

These Specifications (and the products and services contained herein) are proprietary to Telit and its licensors and constitute the intellectual property of Telit (and its licensors). All title and intellectual property rights in and to the Specifications (and the products and services contained herein) is owned exclusively by Telit and its licensors. Other than as expressly set forth herein, no license or other rights in or to the Specifications and intellectual property rights related thereto are granted to you. Nothing in these Specifications shall, or shall be deemed to, convey license or any other right under Telit's patents, copyright, mask work or other intellectual property rights or the rights of others.

You may not, without the express written permission of Telit: (i) copy, reproduce, create derivative works of, reverse engineer, disassemble, decompile, distribute, merge or modify in any manner these Specifications or the products and components described herein; (ii) separate any component part of the products described herein, or separately use any component part thereof on any equipment, machinery, hardware or system; (iii) remove or destroy any proprietary marking or legends placed upon or contained within the products or their components or these Specifications; (iv) develop methods to enable unauthorized parties to use the products or their components; and (v) attempt to reconstruct or discover any source code, underlying ideas, algorithms, file formats or programming or interoperability interfaces of the products or their components by any means whatsoever. No part of these Specifications or any products or components described herein may be reproduced, transmitted, transcribed, stored in a retrieval system, or translated into any language or computer language, in any form or by any means, without the prior express written permission of Telit.

HIGH RISK MATERIALS

Components, units, or third-party products contained or used with the products described herein are NOT fault-tolerant and are NOT designed, manufactured, or intended for use as on-line control equipment in the following hazardous environments requiring fail-safe controls: the operation of Nuclear Facilities, Aircraft Navigation or Aircraft Communication Systems, Air Traffic Control, Life Support, or Weapons Systems ("High Risk Activities"). Telit, its licensors and its supplier(s) specifically disclaim any expressed or implied warranty of fitness for such High Risk Activities.

TRADEMARKS

You may not and may not allow others to use Telit or its third party licensors' trademarks. To the extent that any portion of the products, components and any accompanying documents contain proprietary and confidential notices or legends, you will not remove such notices or legends.

THIRD PARTY RIGHTS

The software may include Third Party Right software. In this case you agree to comply with all terms and conditions imposed on you in respect of such separate software. In addition to Third Party Terms, the disclaimer of warranty and limitation of liability provisions in this License shall apply to the Third Party Right software.

TELIT HEREBY DISCLAIMS ANY AND ALL WARRANTIES EXPRESS OR IMPLIED FROM ANY THIRD PARTIES REGARDING ANY SEPARATE FILES, ANY THIRD PARTY MATERIALS INCLUDED IN THE SOFTWARE, ANY THIRD PARTY MATERIALS FROM WHICH THE SOFTWARE IS DERIVED (COLLECTIVELY "OTHER CODE"), AND THE USE OF ANY OR ALL THE OTHER CODE IN CONNECTION WITH THE SOFTWARE, INCLUDING (WITHOUT LIMITATION) ANY WARRANTIES OF SATISFACTORY QUALITY OR FITNESS FOR A PARTICULAR PURPOSE.

NO THIRD PARTY LICENSORS OF OTHER CODE SHALL HAVE ANY LIABILITY FOR ANY DIRECT, INDIRECT, INCIDENTAL, SPECIAL, EXEMPLARY, OR CONSEQUENTIAL DAMAGES (INCLUDING WITHOUT LIMITATION LOST PROFITS), HOWEVER CAUSED AND WHETHER MADE UNDER CONTRACT, TORT OR OTHER LEGAL THEORY, ARISING IN ANY WAY OUT OF THE USE OR DISTRIBUTION OF THE OTHER CODE OR THE EXERCISE OF ANY RIGHTS GRANTED UNDER EITHER OR BOTH THIS LICENSE AND THE LEGAL TERMS APPLICABLE TO ANY SEPARATE FILES, EVEN IF ADVISED OF THE POSSIBILITY OF SUCH DAMAGES.

Copyright © Telit Communications PLC.

CONTENTS

1	Introduction	9
1.1	Scope	9
1.2	Audience	9
1.3	Contact Information, Support	9
1.4	Text Conventions	10
1.5	Related Documents	10
2	Audio Section Overview	11
3	Audio Front End	13
3.1	HS/HF Physical Audio Paths	14
4	I2C Bus	16
5	Digital Voice Interface	17
6	Audio Equalizer	18
6.1	Digital Gain Control	18
6.2	Biquad Audio Filter	18
7	AFE/DVI Digital Gains & Sidetone	19
7.1	AFE/DVI Digital Gains	19
7.2	Sidetone	19
8	Text Telephone Typewriter (TTY)	21
9	DTMF Tones	23
9.1	DTMF Local Generation	23
9.2	DTMF Transmission on Uplink	23
9.2.1	DTMF Generated by AT Command	23
9.2.2	DTMF Generated by an External Source	24
9.3	Embedded DTMF Decoder	25
9.3.1	Decoder Enhancement	26
10	Acoustic Echo Canceller	27
10.1	AEC of First Generation	28
10.1.1	Echo Canceller	29
10.1.2	Automatic Gain Control	29
10.1.3	Noise Reduction	30
10.1.4	Comfort Noise Injector	30
10.2	AEC of Second Generation	31

11	eCall	33
12	Audio Streaming	34
12.1	AT#SPCM Command	34
12.1.1	Play Local PCM Stream on Local APK	35
12.1.2	Play a PCM Stream on a Far-End Device	37
12.1.3	Play a PCM Stream on a Far-End Device and Local SPK	38
12.1.4	Convert a Local Audio Signal from MIC into a Local PCM Stream	39
12.1.5	PCM Stream on Output Serial Line	40
12.1.6	From MIC to Output Serial Line	41
12.1.7	RX/TX PCM Streams	42
12.2	AT#APLAY Command	43
12.2.1	PCM Audio File Played on Speaker	43
12.2.2	PCM Audio File Sent on Uplink	45
12.3	AT#SPCM Command & DTMF	46
12.4	AT#SAMR Command	47
12.4.1	Play Local AMR Stream on Local SPK	48
12.4.2	Play an AMR Stream on a Far-End Device	49
12.5	AT#SAMR Command & DTMF	49
13	Acronyms & Abbreviations	50
14	Modules & SW Ver. Tables	51
15	Appendixes	52
15.1	Microphone Line Guidelines	52
15.1.1	Single Ended Input Amplifier	52
15.1.2	Differential Input Amplifier	52
15.1.2.1	<i>Practical Suggestions</i>	53
15.1.3	HandsFree	55
15.1.4	Definitions	56
15.1.4.1	<i>Normal Spoken Condition</i>	56
15.1.4.2	<i>Nominal Sensitivity</i>	56
15.1.4.3	<i>Microphone Electrical Level</i>	56
15.2	Electret Microphone	57
15.2.1	Principle of Operation	57
15.2.2	Types	58
15.2.3	Practical Suggestions	60
15.3	Speaker Line Guidelines	61
15.3.1	Single Ended (SE)	61
15.3.2	Differential (BTL)	61
15.4	Audio Profile Parameters for AEC of the First Generation	62
15.4.1	Configuration of an AEC of the First Generation	63
15.5	Audio Codec Information and Setting	69
16	Document History	72

FIGURES

Fig. 1: Main Functional Blocks of the Audio Section	12
Fig. 2: AFE Block Diagram	13
Fig. 3: Master and Slave Configuration	17
Fig. 4: TTY Device/Module Connection	21
Fig. 5: Acoustic Echo Canceller of First Generation	28
Fig. 6: Acoustic Echo Canceller of Second Generation	31
Fig. 7: Local SPK Plays a Local External PCM Stream	35
Fig. 8: Play a PCM Stream on Far-End Device.....	37
Fig. 9: Play a PCM Stream on Far-End Device and Local SPK.....	38
Fig. 10: PCM Stream Coming from Local MIC	39
Fig. 11: Remote Audio Signal from Downlink to Output Serial Line.....	40
Fig. 12: From MIC to Output Serial Line	41
Fig. 13: RX and TX PCM Streams	42
Fig. 14: PCM Audio File on SPK	43
Fig. 15: AT#SPCM & DTMF	46
Fig. 16: Local SPK Plays a Local External Stream	48
Fig. 17: Play a Stream on Far-End Device.....	49
Fig. 18: Single Ended Input Amplifier	52
Fig. 19: Differential Input Amplifier	52
Fig. 20: AC Coupling Capacitors	53
Fig. 21: Unbalanced Bias Circuitry	54
Fig. 22: Balanced Bias Circuitry	54
Fig. 23: ECM.....	57
Fig. 24: ECM-FET Schematic Block.....	58
Fig. 25: Electret Microphones.....	59
Fig. 26: Unbalanced Microphone Biasing Circuitry	60
Fig. 27: Single Ended Output Amplifier	61
Fig. 28: Differential Output Amplifier	61
Fig. 29: AEC of First Generation: Echo Chain	65
Fig. 30: Position of Microphone-Speaker System.....	66
Fig. 31: Threshold Echo Loss Behavior	67

TABLES

Tab. 1: Modules SW Versions and AT#PSET Command	18
Tab. 2: DTMF Tones	23
Tab. 3: Modules SW Versions and AT#DTMF Command	25
Tab. 4: Modules SW Versions and AT#CPUMODE Command	26
Tab. 5: Modules SW Versions and AT#DTMF=3 Command	26
Tab. 6: AEC of First Generation & Modules SW Versions	28
Tab. 7: AEC of Second Generation & Modules SW Versions	31
Tab. 8: AT#SPCM Parameters.....	34
Tab. 9: Local SPK Plays a Local External PCM Stream	35
Tab. 10: Play a PCM Stream on Far-End Device	37
Tab. 11: Play a PCM Stream on Local SPK and Far-End Device	38
Tab. 12: PCM Stream Coming from Local MIC.....	39
Tab. 13: Remote Audio Signal from Downlink to Output Serial Line	40
Tab. 14: From MIC to Output Serial Line	41
Tab. 15: RX and TX PCM Streams	42
Tab. 16: AT#SAMR Parameters.....	47
Tab. 17: Local SPK Plays a Local External Stream	48
Tab. 18: Play a Stream on Far-End Device	49
Tab. 19: ECM Electrical Characteristics.....	58
Tab. 20: Audio Profile Parameters Ranges.....	62
Tab. 21: Basic AEC Audio Profiles.....	62
Tab. 22: Audio Profile: Office, Open Air, Small/Medium Car	63

1 INTRODUCTION

1.1 Scope

Scope of this document is to give an overview of the audio section of the Telit's modules, and provides examples about the use of the AT commands related to the audio. The following topics are covered: HS/FS physical audio paths, I2C bus, digital voice interface, audio equalizer, sidetone, TTY, DTMF, acoustic echo canceller, audio streaming. The Appendix collects short descriptions about microphone, speaker, and audio profile parameters for AEC.

1.2 Audience

The present application note is intended for those users who want to know and properly configure the audio section of the Telit's modules.

1.3 Contact Information, Support

For general contact, technical support services, technical questions and report documentation errors contact Telit Technical Support at:

TS-EMEA@telit.com

TS-AMERICAS@telit.com

TS-APAC@telit.com

Alternatively, use:

<http://www.telit.com/support>

For detailed information about where you can buy the Telit modules or for recommendations on accessories and components visit:

<http://www.telit.com>

Our aim is to make this guide as helpful as possible. Keep us informed of your comments and suggestions for improvements.

Telit appreciates feedback from the users of our information.

1.4 Text Conventions



Danger – This information **MUST** be followed or catastrophic equipment failure or bodily injury may occur.



Caution or Warning – Alerts the user to important points about integrating the module, if these points are not followed, the module and end user equipment may fail or malfunction.



Tip or Information – Provides advice and suggestions that may be useful when integrating the module.

All dates are in ISO 8601 format, i.e. YYYY-MM-DD.

1.5 Related Documents

- [1] AT Commands Reference Guide, 80000ST10025a
- [2] Telit 3G Modules AT Commands Reference Guide, 80378ST10091A
- [3] Digital Voice Interface Application Note, 80000NT10004A
- [4] HE/UE910, UL865 Digital Voice Interface Application Note, 80000NT10050A
- [5] GE910 Family Digital Voice Interface Application Note, 80000NT10099A
- [6] GL865/GL868 V3, GE866 Digital Voice Interface, 80000NT10104A
- [7] Telit eCall Solution Application Note, 8000NT10046A
- [8] Telit CMUX Implementation User Guide, 1vv0300994
- [9] Audio Tuning for HandsFree Systems Application Note, 80464NT11417A



In the present guide is used the notation [x]/[y] to refer to documents of different series of modules. Refer to the document in accordance with the module you are using.

2 AUDIO SECTION OVERVIEW

The module receives the voice from the downlink in digital format, converts it in analog signal, and sends the converted voice to the speaker (SPK). The microphone (MIC) captures the voice in analog format; the module converts the voice in digital format, and sends it to the uplink. The Fig. 1 shows the main blocks of the audio section. The gray blocks are the items external to the module. The Customer Audio Interface block includes the customer devices used to interface the module with the transducers (SPK, MIC).

This document dedicates a chapter to each functional block of the Fig. 1. The chapter describes the AT commands used to set the block configuration, or gives you information on more detailed documents concerning the block.



Before using the AT commands described in these pages, it is required to enter the AT#SELINT=2 command to set the suitable AT command interface style. Refer to the AT Command Reference Guide of the module you are using to have detailed information on AT commands syntax and their parameters, see chapter 14.

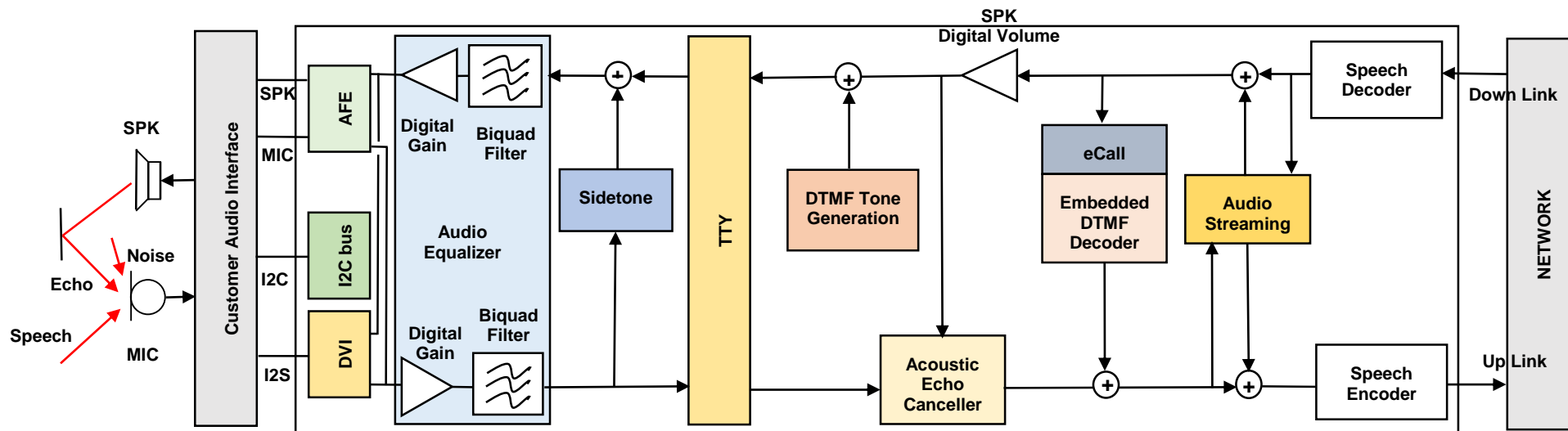


Fig. 1: Main Functional Blocks of the Audio Section

3 AUDIO FRONT END

The module receives the voice, coming from the downlink, in digital format, and routes it to the Audio Front End. The AFE converts the digital voice in analog signal, and sends it to the speaker.

The microphone captures the voice, and routes it to the Audio Front End. The AFE transforms the analog voice in digital format, and sends it on the uplink. Refer to Fig. 1.

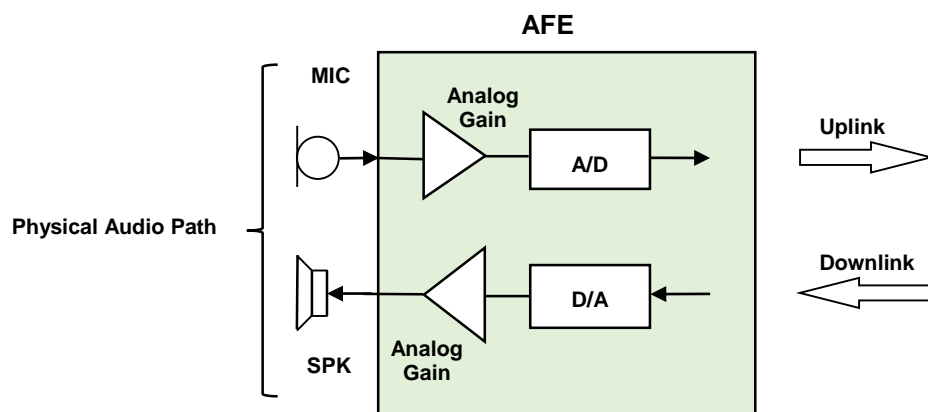


Fig. 2: AFE Block Diagram

The AFE provides two operation modes, featuring a high quality DAC/ADC conversion with buffering stages, allowing the connection of external acoustic transducers.

- In power-down mode:
all analog parts are powered down and all digital part clocks are switched off.
- In audio mode:
all digital filters are connected to the interface buffers and the analog part is enabled.

3.1 HS/HF Physical Audio Paths

Refer to chapter 14 to know which modules provide an Audio Front End with two physical audio paths. To distinguish the two physical audio paths the following historical acronyms are used:

- HS (HandSet) is the combined unit of microphone and earpiece used in connection with a standard telephone, internal audio transducers.
- HF (HandsFree) is the name of equipment that can be used without hands during a call, external audio transducers.



You can use only one physical path at a time.

The two physical audio paths have different electrical characteristics. Refer to the Hardware User Guide of the module you are using to have detailed information.



For modules having only one physical audio path, AT#CAP=<n> is a dummy command, returns OK response and no actions are performed by the module. This solution was adopted for backward compatibility reasons.

Use the AT#CAP command to switch between the two audio paths. Two modes are available:

- Software
- Hardware

Software mode

AT#CAP=2 ← Use software mode to select HS audio path
OK

AT#CAP=1 ← Use software mode to select HF audio path
OK

Hardware mode

AT#CAP=0 ← Use hardware mode to select audio path
OK

After entering the previous command, select the audio path by means of the AXE pin:

- AXE pin = HIGH to select HS audio path.
- AXE pin = LOW to select HF audio path.



- 1) You can switch between the audio paths also during a call. When hardware mode is not used, AXE pin can be unconnected.
- 2) Several AT commands described in this document contain the couple of characters “HS” or “HF” to distinguish the two physical audio paths on which the commands work. In the next pages, to avoid the repetition of the same command working on different audio path, will be used the following rule: the “x” character is “S” for HS path or “F” for HF path.
- 3) For the modules without HF audio path, the Hands Free Commands (x = F) are dummy commands. The commands return OK response and no actions are performed by the modules. This solution was adopted for backward compatibility reasons.

Use the following AT command to set the SPK analog gain:

AT#HxRECG=<level>

OK

Use the following AT command to set the MIC analog gain:

AT#HxMICG=<level>

OK



- 1) The commands above work only on AFE block (refer to Fig. 2). DVI block is not affected.
- 2) Refer to the Hardware User Guide of the module that you are using to have information on the electrical characteristics of the two couples of input/output (MIC/SPK) lines.

The following Open Audio Loop command is used only for testing.

AT#OAP=[<mode>]

4 I2C BUS

Use the I2C bus to configure the Audio Codec internal registers. The Audio Codec is included in the Customer Audio Interface block. Refer to Fig. 1.

The documents [3], [4], [5], and [6] give information on the physical I2C bus, and examples of the AT commands used to configure the Audio Codec internal registers. Refer to the document in accordance with the module that you are using. See chapter 14.

5 DIGITAL VOICE INTERFACE

Referring to Fig. 1. The module receives the voice coming from the downlink, in digital format, and routes it to an external Codec through the Digital Voice Interface (DVI); the Audio Codec decodes the voice and sends it to the speaker.

The microphone captures the voice; the Audio Codec transforms the analog voice in digital format and directs it to the DVI. The module receives the voice from the DVI, in digital format, and sends it on the uplink.

Telit's modules provide a physical DVI interface based on the standard I2S bus. Before establishing a voice call, you must select one of the two configurations shown in the figure below by means of the suitable AT commands

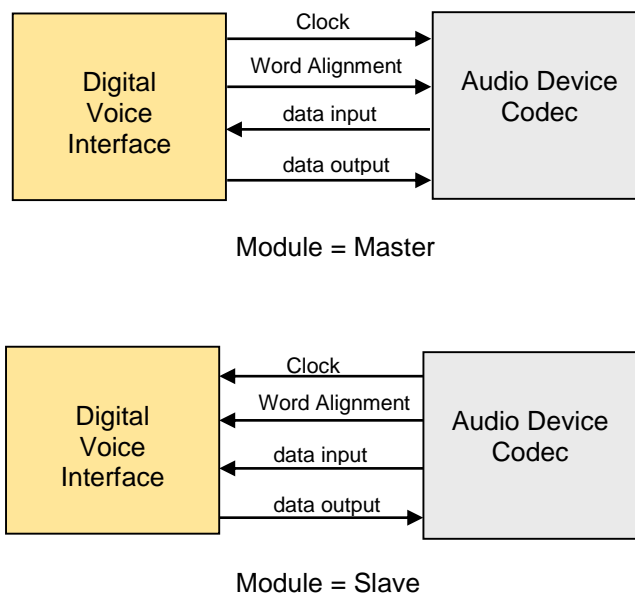


Fig. 3: Master and Slave Configuration

The documents [3], [4], [5], and [6] give information on the physical DVI bus and examples of the AT commands used to set the required configuration; refer to the document in accordance with the module that you are using, see chapter 14.

6 AUDIO EQUALIZER

The Audio Equalizer shown in Fig. 1 includes two sections, one dedicated to the SPK transducer and the other to the MIC transducer. Each section includes a Digital Gain Buffer and a Biquad Filter. The following sub-chapters describe the AT commands used to manage digital gain and filtering.



These AT commands affect AFE and DVI blocks, refer to Fig. 1.

6.1 Digital Gain Control

The first two parameters of the AT#PSET command set the digital gains of the SPK and MIC transducers. See also chapter 15.4.

AT#PSET=<scal_in>,<scal_out>,...

The table below shows the software versions of the modules series supporting the AT#PSET command, see also chapter 14.

SW Versions
10.00.xxy
13.00.xxy
16.00.xxy

Tab. 1: Modules SW Versions and AT#PSET Command

6.2 Biquad Audio Filter

The AT commands below configure the Biquad filters connected to SPK and MIC transducers. Refer to documents [1]/[2] to have detailed information on their parameters.

Here are the AT commands to configure respectively the MIC and SPK transducer filters:

AT#BIQUADIN

AT#BIQUADOUT

Here are the AT commands to configure respectively the MIC and SPK transducer extended filters:

AT#BIQUADINEX

AT#BIQUADOUTEX

7 AFE/DVI DIGITAL GAINS & SIDETONE

7.1 AFE/DVI Digital Gains

The command below sets up the SPK digital volume level, refer to Fig. 1:

AT+CLVL=<vol>



If the used module provides two audio paths (HS and HF), the volume level does not change even if the audio path is changed.

Enable/disable the global muting of the speaker audio line.

AT#SPKMUT=<n>

The microphone of the active path can be muted with the following AT command.

AT+CMUT=<n>

7.2 Sidetone

The sidetone is the effect of sound picked up by the microphone and instantly introduced at a low signal level into the receiver (speaker). Refer to Fig. 1.



The AT commands below use the following rule: the “x” character is “F” to configure the headset sidetone, “S” to configure the handset sidetone.

The following AT commands enable/disable the handset or headset sidetone.

AT#SHxSD=<mode>

Check the current <mode> value.

AT#SHxSD?

Use the third parameter of the following AT command to set the digital tone attenuation.

AT#PSET=<scal_in>,<scal_out>,<side_tone_atten>,...

8 TEXT TELEPHONE TYPEWRITER (TTY)

Fig. 4 shows a TTY device connected to the module; the TTY characters are coded using two tones: 1400 Hz and 1800 Hz. The GSM speech coder does not support these tones; the CTM modem, embedded in the module, makes them compatible with the speech coder. The CTM modem transforms the two unsupported TTY tones into four tones: 400Hz, 600Hz, 800Hz, and 1000Hz. The speech coder is able to manage the four tones. The TTY device is connected to the module using one of the two physical audio paths: HS or HF, refer to chapter 3.1 . This feature enables the transmission of the TTY data and voice jointly on the speech channel.

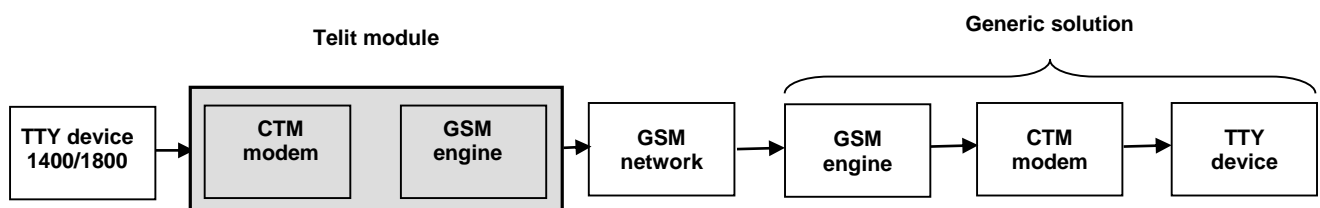


Fig. 4: TTY Device/Module Connection

Example:

Connect the TTY device to the module. In accordance with the used physical audio connection, configure the audio path as indicated below. Assume to use software mode to select HF audio path, refer to chapter 3.1.

AT#CAP=1 ← Use software mode to select HF audio path
OK

AT+CLVL=5 ←Set the volume of the active audio path
OK

AT#HFMICG=4 ←Set the HF microphone input gain
OK

AT#SHFSD=0 ←Disable the side tone on HF audio output
OK

AT#SHFEC=0 ←Disable echo canceller function on HF audio path
OK

AT#SHFAGC=0 ←Disable automatic gain control on HF audio path
OK

AT#SHFNR=0 ←Disable noise reduction on HF audio path
OK

AT# TTY =? ←Check the available values
#TTY: (0-1)
OK

AT# TTY? ←Check the current value
TTY: 0
OK

AT# TTY =1 ←Enable TTY functionality
OK

Now, the voice call is able to support jointly voice and TTY data.

9 DTMF TONES

The DTMF tones are used to control telephone systems. Using DTMF tones, the following characters can be sent: 0,1,2,3,4,5,6,7,8,9,A,B,C,D,* and #. Each character is converted in a couple of frequencies as shown in the table below. The rows represent low frequencies, and the columns represent the high frequencies. For example, the character '1' is sent as two sinusoidal tones of 697 and 1209 Hz.

	1209Hz	1336Hz	1477Hz	1633Hz
697Hz	1	2	3	A
770Hz	4	5	6	B
852Hz	7	8	9	C
941Hz	*	0	#	D

Tab. 2: DTMF Tones

9.1 DTMF Local Generation

The following AT commands reproduce locally DTMF tones:

AT#TONE=<tone>,<duration>

AT#TONEEXT=<toneld>,<act>

9.2 DTMF Transmission on Uplink

In general, DTMF tones sent on the uplink are generated by the module using the suitable AT command or by an external device connected to the module through the MIC line. The following sub-chapters describe these two scenarios.

9.2.1 DTMF Generated by AT Command

Using the following AT command, the module sends the suitable command to the network infrastructure to generate on the other audio party the correspondent DTMF tone. The module embeds the DTMF command in a network message, and sends it during the voice call.

AT+VTS=<dtmfstring>[,duration]

Example:

Check the range of supported values:

AT+VTS=?

(0-9,#,*,A-D,P),(0-255)

OK

Check the tone duration of the single character:

AT+VTD?

1

OK

Dialing the number in voice mode:

ATD04x419x40y;

OK

Send the following sequence of tones:

AT+VTS=123456789

OK

Hang up the voice call:

ATH

OK

9.2.2 DTMF Generated by an External Source

An external source, typically a corded phone, generates DTMF tones, and sends them to the module using the input line. In turn, the module digitizes, encodes, and sends on the uplink the received couples of frequencies.

In the receiving device the signal would be reconstructed, but since the digital transmission of the voice channel is compressed and optimized for voice, this reconstruction depends on the kind of voice compression used for the transmission, and generally will not perfectly match the original signal.

There are four main types of compression for the voice channel: Full Rate, Half Rate, Enhanced Full Rate, and Adaptive Multi Rate. Only the first one has no distortion, while the other three offer a different trouble level.

Suggestion:

- do not generate DTMF signal by means of an external device
- use AT+VTS command to send DTMF signal
- set the sidetone off during DTMF signal sending

If an external DTMF source is used, the distortion due to voice channel compression should be accepted. The maximum input line level is:

$$1,03 \text{ Vpp} / 365 \text{ mVrms} @ \text{ Mic Gain} = 0\text{dB}$$

If the level of DTMF signal is high, it causes saturation of the AFE amplifier and unwanted harmonic components are generated.

9.3 Embedded DTMF Decoder

Telit’s modules provide an embedded DTMF decoder that recognizes automatically the DTMF tones coming from the downlink speech channel. Refer to Fig. 1. The embedded decoder sends, through the current AT interface, an URC reporting the last decoded digit with the following format:

#DTMFEV: x where x = 0,1, 2, 3, 4, 5, 6, 7, 8, 9, A, B, C, D, *, and #

The AT command below enables/disables the DTMF decoder for modules equipped with SW versions shown in Tab. 3, see also chapter 14:

AT#DTMF=1

SW Versions
10.00.xxy
12.00.xxy
13.00.xxy
16.00.xxy

Tab. 3: Modules SW Versions and AT#DTMF Command

If the module is equipped with a SW Version shown in Tab. 4 (see also chapter 14), you should issue the AT command below to improve the CPU clock and ensure the proper CPU performances for real time decoding.

AT#CPUMODE=1

SW Versions
10.00.xxy
16.00.xxy

Tab. 4: Modules SW Versions and AT#CPUMODE Command

See chapters 12.3, and 12.5.

9.3.1 Decoder Enhancement

You can improve the performance of the embedded decoder by means of the following AT commands:

- **AT#DTMF=3** command for modules equipped with SW Versions shown in table below (see also chapter 14).

SW Versions
10.00.xxy
13.00.xxy
16.00.xxy

Tab. 5: Modules SW Versions and AT#DTMF=3 Command

- **AT#DTMFCFG** command for tuning internal parameters.

Use both AT#DTMF and AT#DTMFCFG commands, where it is permitted, to reach a high overall decoding performance. To have more information refer to documents [1]/[2].



You should use the default values for standard applications. Use carefully the tuning.

10 ACOUSTIC ECHO CANCELLER

Acoustic echo is formed when the sound emitted by the speaker is reflected from the walls, ceilings, floor, furniture, people, and so on back to the microphone. To overcome this phenomenon, the modules provide an Acoustic Echo Canceller (AEC). Refer to Fig. 1.

Telit's modules provide two types of AEC, in accordance with the processor installed on them. The following terms distinguish the two types:

- AEC of first generation
- AEC of second generation

The three AT commands below are common to both types of AECs and affect AFE and DVI blocks.

AT#PRST	Audio Profile Factory Configuration
AT#PSAV	Audio Profile Configuration Save
AT#PSEL	Audio Profile Selection

Both AECs include the following three blocks (algorithms), see Fig. 5, and Fig. 6:

- Echo Canceller
- Automatic Gain Control
- Noise Reduction

The following AT commands enable/disable respectively the Echo Canceller, Automatic Gain Control, and Noise Reduction functions on the audio path currently used. If the module provides only one path, does not matter if $x = S$ or $x = F$, the command works on the single physical path. Refer to the Applicability Tables (chapter 14) to see the number of physical path provided by your module.

AT#SHxEC=<mode>

AT#SHxAGC=<mode>

AT#SHxNR=<mode>

10.1 AEC of First Generation

The Fig. 5 shows the basic blocks of an AEC of first generation, and Tab. 6 shows the series of modules equipped with such type of AEC, see also chapter 14.

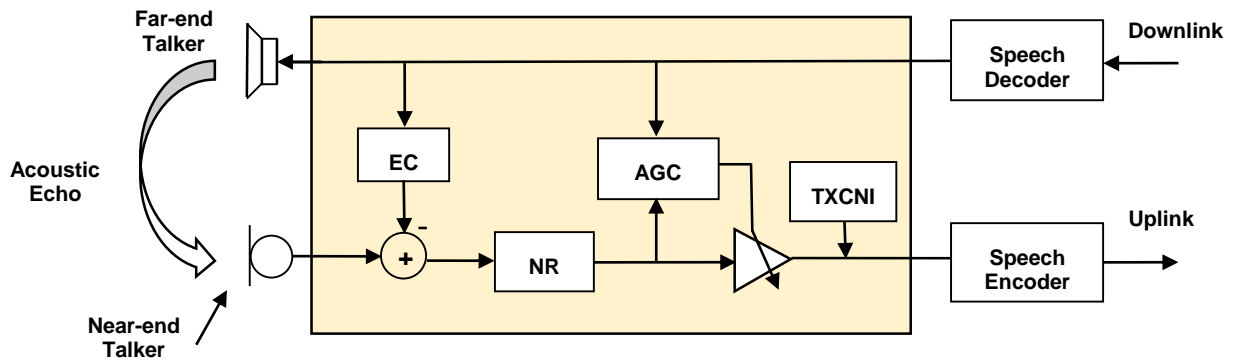


Fig. 5: Acoustic Echo Canceller of First Generation

SW Versions
10.00.xxy

Tab. 6: AEC of First Generation & Modules SW Versions

Here are the blocks (algorithms) building the Acoustic Echo Canceller:

- Echo Canceller (EC)
- Automatic Gain Control (AGC)
- Noise Reduction (NR)
- Comfort Noise Injection (TXCNI)

The first three algorithms are configured by means of the following AT command.

AT#PSET

This command has several parameters to configure the algorithms; the next sub-chapters describe which parameters are dedicated to each one of the algorithm. Chapter 15.4 gives some examples of parameters values.



EC, AGC, and NR configurations are not allowed if Active Audio Profile "0" is active.

10.1.1 Echo Cancellor

Echo Cancellor uses a FIR filter. It reflects the inverse of the acoustic coupling between speaker and microphone. Echo Cancellor is controlled through the AT#PSET command, the used parameters are:

- <adaption_speed> Adaptation Speed.

It represents the capacity of the algorithm to adapt to the variation of echo characteristics; higher value means faster adaptation but less accurate convergence and conversely.

- <filter_length> FIR Filter Length.

It represents the impulse response length to build the estimated filter, thus it is the maximum cancelable echo delay.

- <rxtxrelation> Power Relation RX →TX.

This value is used to tell the algorithm about the real world signal power relation between speaker and microphone and their acoustic coupling.

10.1.2 Automatic Gain Control

Automatic Gain Control is based on the measurement of the power relation between RX and TX signals. The measurement result will indicate to attenuate or not the TX signal, this block can be called "Automatic Reduction of Gain".

The amount of the attenuation is controlled by means of the AT#PSET command. The parameters dedicated to the AGC are:

- <add_atten> Additional Attenuation.

It represents the additional attenuation added to the attenuation calculated by the AGC.

- <min_atten> Minimum Attenuation.

It represents the minimal attenuation in spite of lower calculated attenuation.

- <max_atten> Maximum Attenuation.

It represents the maximum attenuation in spite of higher calculated attenuation.

AGC threshold for Double Talk detection and digital gain in Uplink.

AT#HFCFG

10.1.3 Noise Reduction

NR filter is based on the spectral weighting algorithm implemented with sub-bands from 300Hz to 4000Hz. It filters background noise introduced by the environment (e.g., noise of car engine) and the residual noise from the impulse response not removed by the Echo Canceller.

Noise Reduction filter is controlled by means of the AT#PSET command, the parameters dedicated to the NR are:

- <nr_atten> Noise Max Attenuation.

It is the maximum attenuation that can be introduced by the NR algorithm.

- <nr_w_0> Noise Weighting Factor Band 300-500Hz.

A higher value causes better noise reduction but also a higher distortion of the speech signal.

- <nr_w_1> Noise Weighting Factor Band 500-4000Hz.

A higher value causes better noise reduction but higher distortion of the speech signal.

10.1.4 Comfort Noise Injector

Here is the command to enable, disable, and configure the Comfort Noise Injection (TXCNI). Refer to document [1].

AT#TXCNI=<support>,<gain>,<floor>

10.2 AEC of Second Generation

The Fig. 6 shows the basic blocks of an AEC of second generation, and Tab. 7 lists the software versions of the modules equipped with such type of AEC, see also chapter 14. To have more detailed information on the AEC tuning refer to document [9].

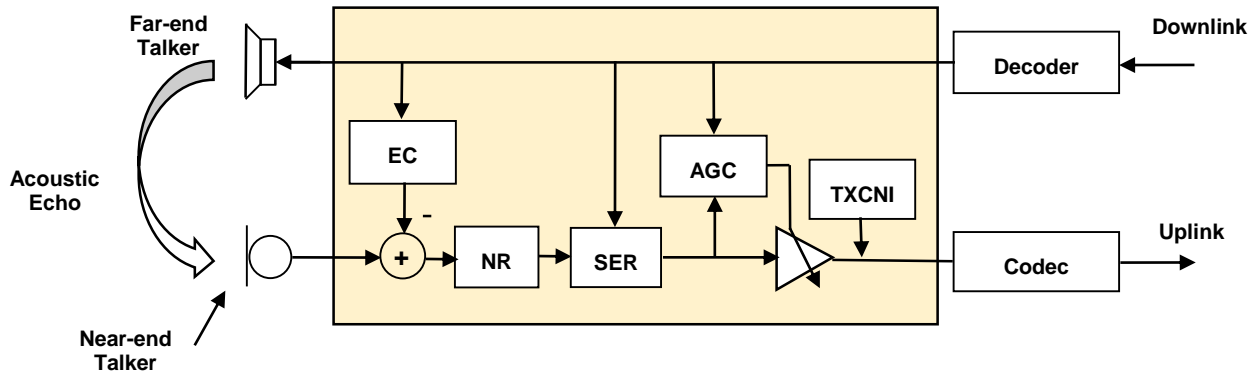


Fig. 6: Acoustic Echo Cancellation of Second Generation

SW Versions
12.00.xxy
13.00.xxy
16.00.xxy

Tab. 7: AEC of Second Generation & Modules SW Versions

Here are the blocks (algorithms) building the Acoustic Echo Canceller:

- Echo Canceller (EC)
- Automatic Gain Control (AGC)
- Noise Reduction (NR)
- Comfort Noise Injection (TXCNI)
- Spectral Echo Canceller (SER)

The AEC of second generation is configured by means of the following Echo Reducer Configuration AT command: AT#ECHOCFG.

After you have issued the command, the ">" prompt is displayed on the terminal. The command is ready to accept your parameters values. For more information on AT command syntax refer to document [2].



Echo Reducer Configuration is not allowed if Active Audio Profile '0' is active. AT#ECHOCFG command returns ERROR message, see AT#PSEL command.

AT#ECHOCFG=0

>the command expects 39 user parameters values

AT#ECHOCFG=1

>the command expects user parameter value corresponding to index 1

AT#ECHOCFG=39

>the command expects user parameter value corresponding to index 39

AT#ECHOCFG=14,15,16,34

>the command expects user parameters values corresponding to indexes 14, 15, 16, 34



As stated at the beginning of this chapter, refer to document [9] to have detailed information on the AEC tuning.

11 ECALL

To have information on the eCall topic refer to document [7].

12 AUDIO STREAMING

Telit's modules provide specific AT commands to send and receive PCM stream through:

- GSM voice channels (Uplink, Downlink)
- Input/Output Serial lines

Refer to the AT Command Reference Guide [1]/[2] of the module you are using to have detailed information on AT commands syntax and their parameters. The following sub-chapters describe the available scenarios in accordance with the used AT commands and their parameters configurations.

12.1 AT#SPCM Command

AT#SPCM command works with the following voice channels:

- Full Rate
- Half Rate
- Enhanced Full Rate

The command is independent from the codec of the used voice channel. Here is its syntax:

AT#SPCM=<mode>,<dir>

The table below summarizes the couples of parameters supported by the command. Each couple defines a scenario.

<dir>	<mode>		
	1	2	3
0	Chapter 12.1.1	Chapter 12.1.4	Not supported
1	Chapter 12.1.2	Chapter 12.1.5	Chapter 12.1.7
2	Chapter 12.1.3	Chapter 12.1.6	Not supported

Tab. 8: AT#SPCM Parameters



The modules equipped with a software version 12.00.xxx provide the AT#SPCM command with three parameters: <mode>,<dir>,<format>. The last one configures the number of bit of the PCM format: 0 → 8 bit, 1 → 16 bit.

12.1.1 Play Local PCM Stream on Local APK

Here is the AT#SPCM command parameters configuration to play a PCM stream on local SPK. The PCM stream comes from a local external source connected to the module through an input serial line, refer to Fig. 7.

AT#SPCM=1,0

Tab. 9 summarizes the scenarios in accordance with the software version. You must establish a voice call.

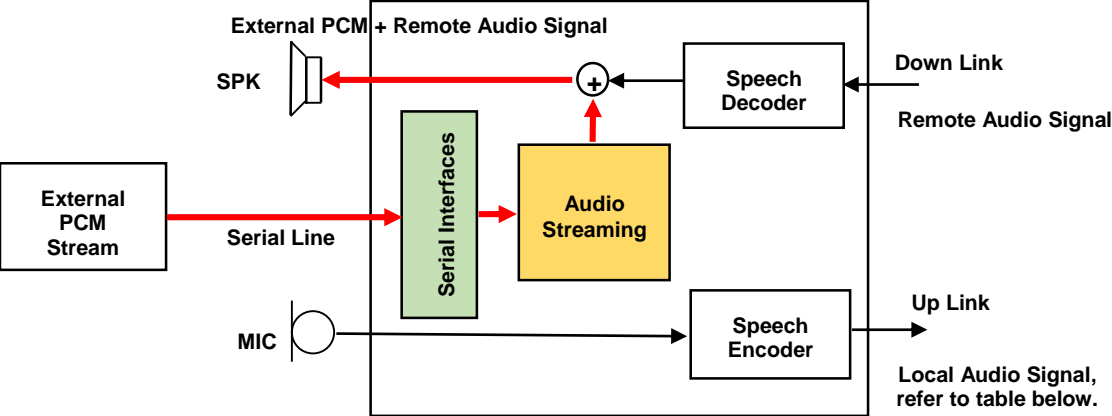


Fig. 7: Local SPK Plays a Local External PCM Stream

		Local SPK plays:	
Software Versions	Uplink channel:	Downlink channel:	Local External PCM Stream + Remote Audio Signal
10.00.xxx	OFF	ON	
12.00.xxx	ON	ON	
13.00.xxx	OFF	ON	
16.00.xxx	OFF	ON	

Tab. 9: Local SPK Plays a Local External PCM Stream

See the examples below.

Example 1: the voice call is not established. Only the PCM stream, coming from a local external source, is played by the local SPK.

AT#SPCM=1,0 ← Play on local SPK the PCM stream coming from the serial line.

CONNECT

Now, the user can send the PCM stream on the serial line.

+++ ← Issue the escape sequence to enter the command mode.

NO CARRIER

Example 2: the voice call is established. The PCM stream, coming from a local external source, is played by the local SPK. Refer to Tab. 10 to see if the downlink channel of the module you are using is enabled. If the downlink channel is ON, also the remote voice is played by the SPK.

ATD<phone number>; ← Establish a voice call
OK

AT#SPCM=1,0 ← Play on local SPK the PCM stream coming from the serial line
CONNECT

Now, the user can send the PCM stream on the serial line.

+++ ← Issue the escape sequence to enter the command mode.
NO CARRIER

12.1.2 Play a PCM Stream on a Far-End Device

Here is the AT#SPCM command parameters configuration to play a PCM stream on a far-end device. The PCM stream comes from a local external source connected to the module through an input serial line, refer to Fig. 8.

AT#SPCM=1,1

Tab. 10 summarizes the scenarios in accordance with the software version. You must establish a voice call.

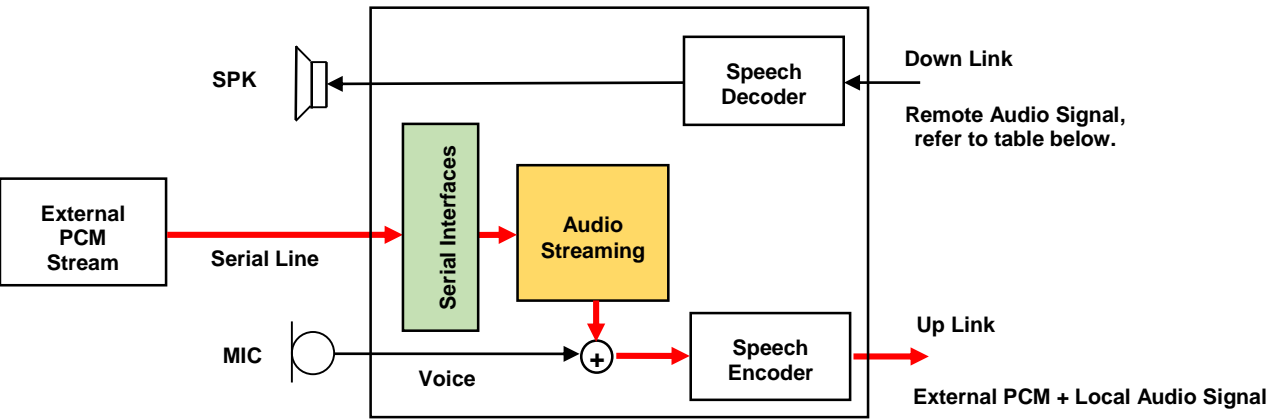


Fig. 8: Play a PCM Stream on Far-End Device

Software Versions	Downlink channel:	Far-End Device plays:	
		Uplink channel:	
10.00.xxx	OFF	ON	Local External PCM Stream + Local Audio Signal
12.00.xxx	ON	ON	
13.00.xxx	OFF	ON	
16.00.xxx	OFF	ON	

Tab. 10: Play a PCM Stream on Far-End Device

12.1.3 Play a PCM Stream on a Far-End Device and Local SPK

Here is the AT#SPCM command parameters configuration to play a PCM stream on local SPK, and send it to uplink audio channel. The PCM stream comes from a local external source connected to the module through an input serial line, refer to Fig. 9.

AT#SPCM=1,2

Tab. 11 summarizes the scenarios in accordance with the software versions. You must establish a voice call.

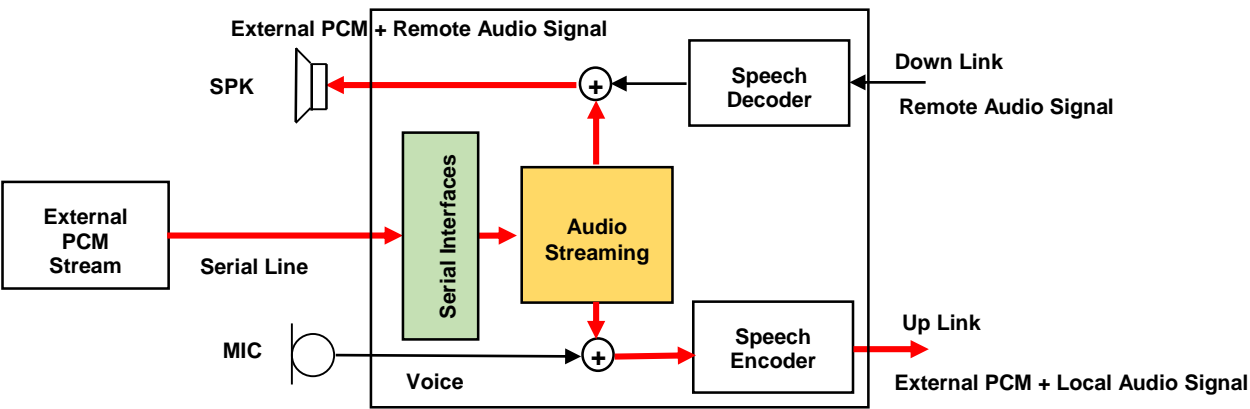


Fig. 9: Play a PCM Stream on Far-End Device and Local SPK

Software Versions	Far-End Device plays:	Local SPK plays:	
	Uplink channel:	Downlink channel:	
10.00.xxx	Local Ext. PCM Stream + Local Audio Stream	Remote Audio Signal	Local External PCM Stream
16.00.xxx			

Tab. 11: Play a PCM Stream on Local SPK and Far-End Device

12.1.4 Convert a Local Audio Signal from MIC into a Local PCM Stream

Here is the AT#SPCM command parameters configuration to receive a voice signal from local MIC. Then, the module converts the voice signal into a PCM stream and routes it to an output serial line. Refer to Fig. 10.

AT#SPCM=2,0

Tab. 12 summarizes the scenarios in accordance with the software versions. You must establish a voice call.

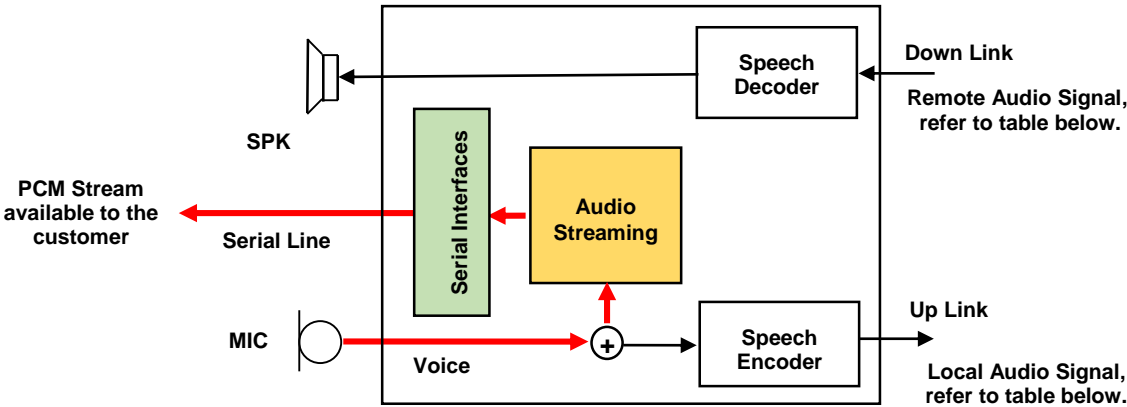


Fig. 10: PCM Stream Coming from Local MIC

Software Versions	Uplink channel:	Downlink channel:	On output serial line is available:
10.00.xxx	OFF	OFF	PCM Stream from local MIC
12.00.xxx	ON	ON	
13.00.xxx	OFF	OFF	
16.00.xxx	OFF	OFF	

Tab. 12: PCM Stream Coming from Local MIC

12.1.5 PCM Stream on Output Serial Line

Here is the AT#SPCM command parameters configuration to receive a remote audio signal from the downlink speech channel. The module converts the remote audio signal into PCM stream, and then routes it to an output serial line. Refer to Fig. 11

AT#SPCM=2,1

Tab. 13 summarizes the different scenarios in accordance with the software version. You must establish a voice call.

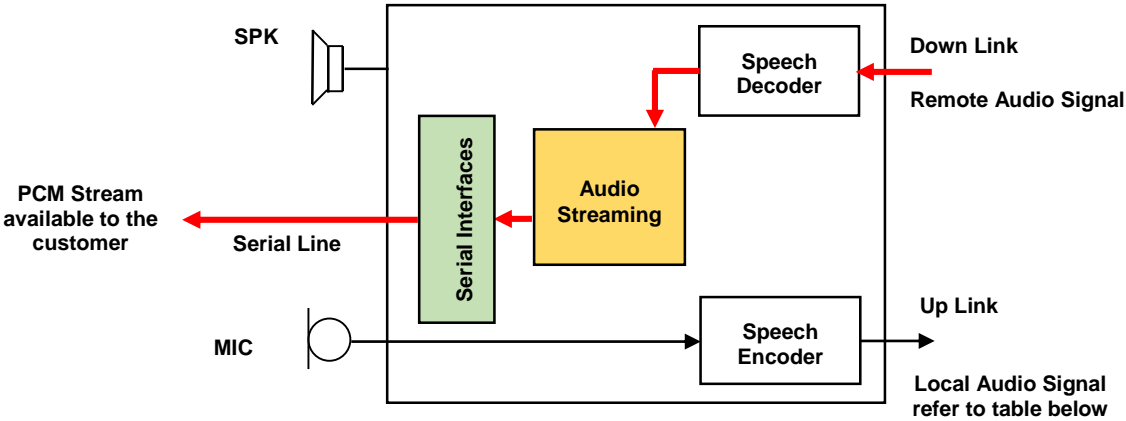


Fig. 11: Remote Audio Signal from Downlink to Output Serial Line

Software Versions	Uplink channel:	On output serial line is available:	
		Downlink channel:	
10.00.xxx	OFF	Remote Audio Signal	
12.00.xxx	ON		
13.00.xxx	OFF		
16.00.xxx	OFF		

Tab. 13: Remote Audio Signal from Downlink to Output Serial Line

12.1.6 From MIC to Output Serial Line

Here is the AT#SPCM command parameters configuration to receive a voice signal from local MIC. Then, the module converts the voice signal into PCM stream, and then routes it to an output serial line. Refer to Fig. 12.

AT#SPCM=2,2



This configuration is obsolete, use AT#SPCM=2,0.

Tab. 14 summarizes the scenarios in accordance with the software versions. You must establish a voice call.

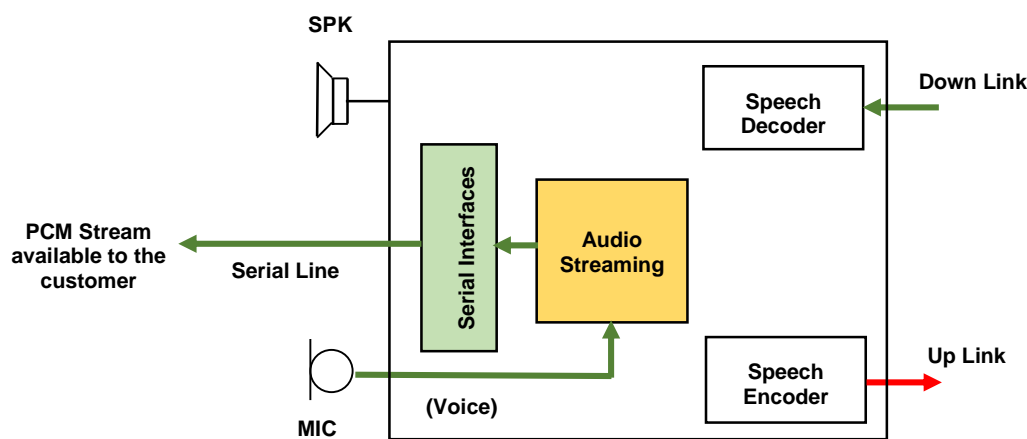


Fig. 12: From MIC to Output Serial Line

Software Versions	Far-End Device plays:	On output serial line is available:
	Uplink channel:	Downlink channel:
10.00.xxx	OFF	OFF
16.00.xxx		

Tab. 14: From MIC to Output Serial Line

12.1.7 RX/TX PCM Streams

Here is the AT#SPCM command parameters configuration to send a TX PCM stream on the uplink speech channel, and receive a remote audio signal on the downlink speech channel. Refer to Fig. 13.

AT#SPCM=3,1

Tab. 15 summarizes the scenarios in accordance with the software versions. You must establish a voice call.

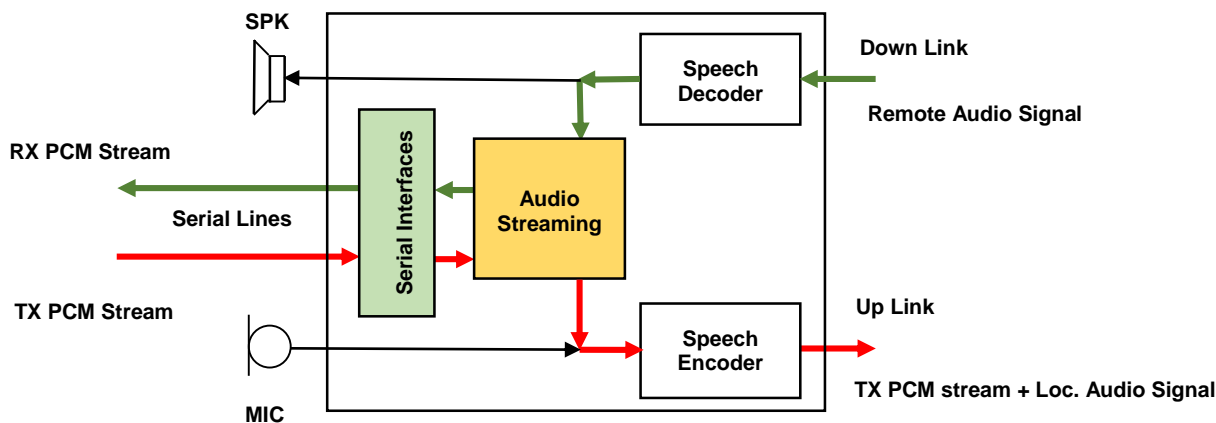


Fig. 13: RX and TX PCM Streams

Software Versions	Far-End Device plays:	Local SPK plays:
	Uplink channel:	Downlink channel:
12.00.xxx	TX PCM Stream + Loc. Audio Signal	Remote PCM Audio Signal
16.01.xx0		

Tab. 15: RX and TX PCM Streams

12.2 AT#APLAY Command

The AT#APLAY command allows the user to send a PCM audio file to the speaker or uplink speech channel. The PCM audio file is stored in the module memory. The modules, shown in the table below, provide several AT commands managing PCM audio files; refer to documents [1]/[2].

SW Versions
/
13.00.xxy
/

12.2.1 PCM Audio File Played on Speaker

Here is the AT#APLAY command parameters configuration to send a PCM audio file to the speaker. The extension of the file is "pcm".

AT#APLAY=1,0,"file_name.pcm"

The establishment of a voice call is not required; the module is in idle mode.

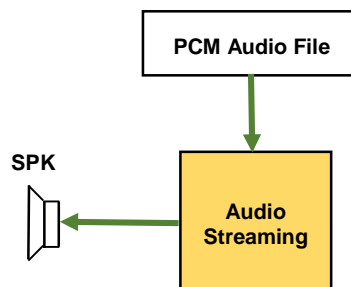


Fig. 14: PCM Audio File on SPK

Here is an example.

Enter the command below. After the command responded with the "CONNECT" prompt, you are allowed to send the PCM audio file using the Send File option provided by the Terminal tool (use RAW ASCII protocol). If the transfer has been successfully ended, the module returns the "OK" message.

If the transferred PCM audio file is stored successfully, the module sends the following Unsolicited Result Code: #ASENDEV: 0.

- The first parameter is the name of the destination file that will be stored in the memory of the module. Write "pcm" file extension in lowercase characters and enclose the file name in straight quotes (string type).

- The second parameter is the size of the file expressed in bytes. Use the "Properties" dialog box of Windows to know the file size in bytes.

AT#ASEND="GOOFY.pcm",320000

CONNECT

OK

#ASENDEV: 0

← URC notifying that the file storing is done successfully.

AT#ALIST

← list the PCM files stored in the module.

#ALIST: "GOOFY.pcm",320000,F238

AT#APLAY=1,0,"GOOFY.pcm"

← play the PCM file on speaker.

OK

#APLAYEV: 0

← URC notifying that the playing is ended.

AT#ADELF="GOOFY.pcm"

← delete the selected PCM file stored in the module.

OK

12.2.2 PCM Audio File Sent on Uplink

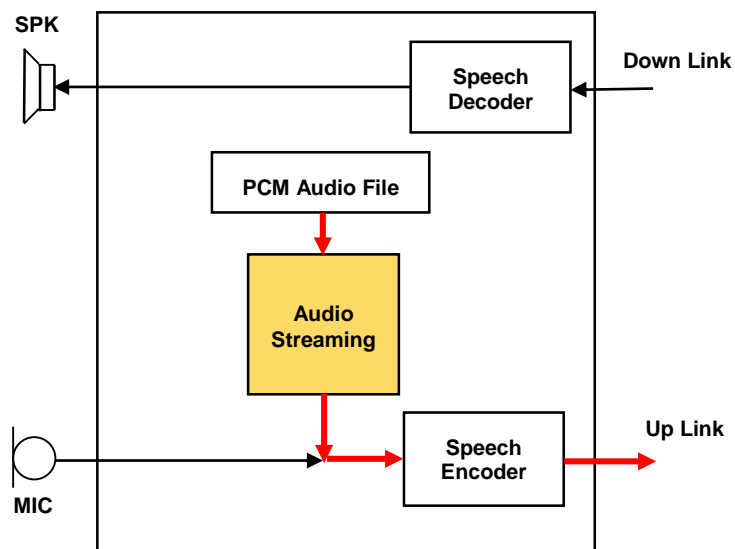
Here is the AT#APLAY command parameters configuration to send a PCM Audio file on the uplink speech channel. The extension of the file is "pcm".



This feature is in progress.

AT#APLAY=1,1,"file_name.pcm"

You must establish a voice call.



12.3 AT#SPCM Command & DTMF

Telit's modules allow the user to start the execution of the AT#SPCM command while receiving and decoding DTMF tones from the downlink speech channel. The table below lists the software version of the modules series that provide this feature, see also chapter 14.

SW Versions
/
16.00.xxy
/

Fig. 15 shows the Application Processor connected to the module by means two input/output serial lines. The serial lines are connected to two AT commands parser instances which decode and executes the entered AT commands.

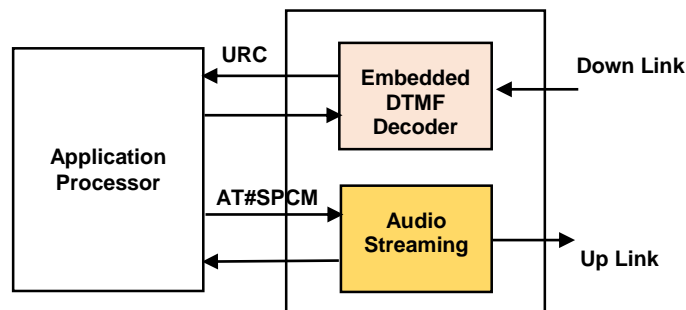


Fig. 15: AT#SPCM & DTMF

1. On the upper serial line, the Application Processor sends the AT commands to initialize the DTMF decoder, and receives the related URC when the remote side sends a DTMF.
2. On the lower serial line, the Application Processor sends the AT#SPCM command after receiving the expected DTMF URC.

To implement this configuration can be used the CMUX Protocol, refer to document [8].

12.4 AT#SAMR Command

AT#SAMR command works with the following voice channel:

- Adaptive Multi Rate

Here are the supported codec:

- AMR 4.75 (default)
- AMR 5.15
- AMR 5.95
- AMR 6.70
- AMR 7.40
- AMR 7.95
- AMR 10.2
- AMR 12.2

Here is the syntax of the command:

AT#SAMR=<mode>,<dir>

The table below summarizes the couples of parameters supported by the command. Each couple defines a scenario.

<mode>	
1	
<dir>	
0	Chapter 12.4.1
1	Chapter 12.4.2

Tab. 16: AT#SAMR Parameters

12.4.1 Play Local AMR Stream on Local SPK

Here is the AT#SAMR command parameters configuration to play a stream on local SPK. The source is an AMR stream coming from a local external device connected to the module through an input serial line. Refer to Fig. 16.

AT#SAMR=1,0

Tab. 17 shows the data running on the uplink and downlink channels in accordance with the AT#SAMRCFG command, when a voice call is established.

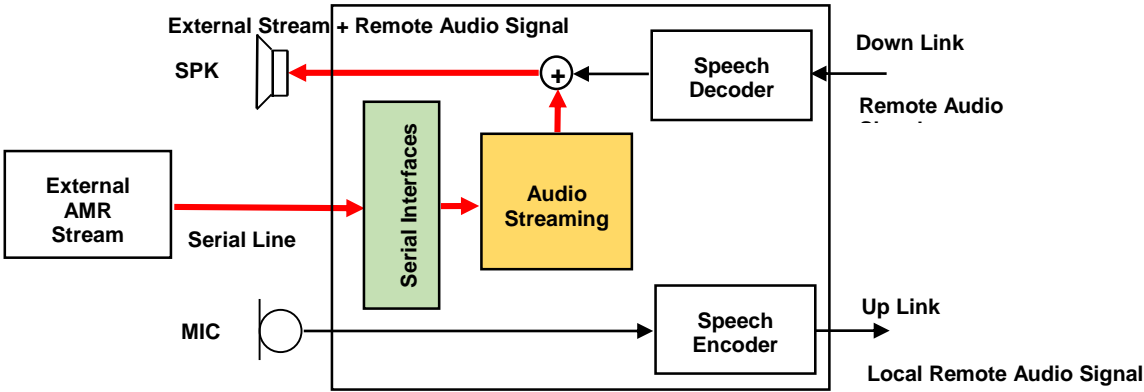


Fig. 16: Local SPK Plays a Local External Stream

Software Versions	AT#SAMRCFG	Local SPK plays:	
		Uplink	Downlink
16.00.xx0	AT#SAMRCFG=0,0,1,1	OFF	OFF
	AT#SAMRCFG=0,0,1,0	OFF	ON
	AT#SAMRCFG=0,0,0,1	ON	OFF
	AT#SAMRCFG=0,0,0,0 (default)	ON	ON

Tab. 17: Local SPK Plays a Local External Stream

12.4.2 Play an AMR Stream on a Far-End Device

Here is the AT#SAMR command parameters configurations to play a stream on a far-end device. The source is an AMR stream coming from a local external device connected to the module through an input serial line. Refer to Fig. 17.

AT#SAMR=1,1

Tab. 18 shows the data running on the uplink and downlink channels in accordance with the AT#SAMRCFG command, when a voice call is established.

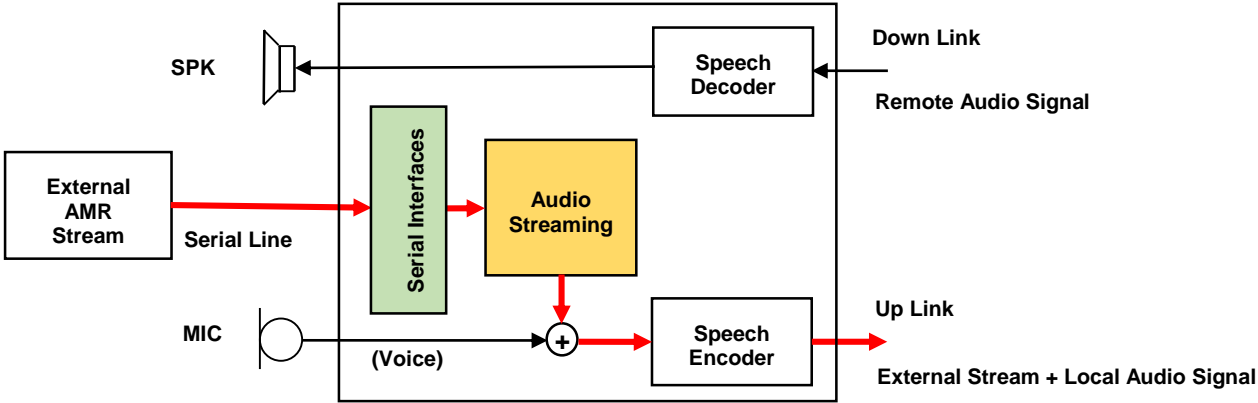


Fig. 17: Play a Stream on Far-End Device

Software Versions	AT#SAMRCFG	Far-End Device plays:		Downlink
		Uplink		
10.01.xx0 13.00.xx6 16.00.xx0	AT#SAMRCFG=0,0,1,1	OFF	Local External Stream	OFF
	AT#SAMRCFG=0,0,1,0	OFF		ON
	AT#SAMRCFG=0,0,0,1	ON		OFF
	AT#SAMRCFG=0,0,0,0 (default)	ON		ON

Tab. 18: Play a Stream on Far-End Device

12.5 AT#SAMR Command & DTMF

You can use the AT#SAMR commands in the same way is used AT#SPCM command, see chapter 12.3.

13 ACRONYMS & ABBREVIATIONS

@	At the condition of...
AC	Alternate Current
AEC	Acoustic Echo Cancellor
AFE	Audio Front End
AGC	Automatic Gain Control
AMR	Adaptive Multi-Rate
BTL	Bridge-Tied Load
CMRR	Common Mode Rejection Ratio
CTM	Cellular Text Telephone Modems
DTMF	Dual Tone Multi Frequency
DVI	Digital Voice Interface
EC	Echo Cancellor
ECM	EleCtret Microphone
EMI	Electro Magnetic Interferences
FIR	Finite Impulse Response
I2C	Inter-Integrated Circuit
I2S	Inter-IC Sound
MIC	Microphone
NR	Noise Reduction
NVM	Non-Volatile Memory
PCB	Printed Circuitry Board
PCM	Pulse Code Modulation
RF	Radio Frequency
SEC	Spectral Echo Cancellor
SNR	Signal Noise Ratio
SPK	Speaker
TTY	Text Telephone Typewriter
URC	Unsolicited Result Code

14 MODULES & SW VER. TABLES

	SW Versions	Audio paths	DVI document	AT Ref. Guide
GC Series			See chapter 1.5	
GC864-QUAD	10.00.xx2	HS, HF	[3]	[1]
GC864-QUAD V2	10.00.xx2			
GC864-DUAL V2	10.00.xx2			
GE/GL Series				
GE864-QUAD	10.00.xx2			
GE864-QUAD V2	10.00.xx2			
GE864-QUAD AUTO V2	10.00.xx2			
GE864-QUAD ATEX	10.00.xx2			
GE864-DUAL V2	10.00.xx2			
GE864-GPS	10.00.xx4			
GE865-QUAD	10.00.xx2			
GL865-DUAL	10.00.xx4			
GL865-QUAD	10.00.xx4			
GL868-DUAL	10.00.xx4			

	SW Versions	Audio paths	DVI document	AT Ref. Guide
HE910 Series			See chapter 1.5	
HE910 ¹	12.00.xx3	HS	[4]	[2]
HE910-GA	12.00.xx3			
HE910-EUR	12.00.xx3			
HE910-NAR	12.00.xx3			
HE910 MINI PCIE	12.00.xxx			
UE/UL Series				
UE910-EUR	12.00.xx4			
UE910-NAR	12.00.xx4			
UL865-EUR	12.00.xx4			
UL865-NAR	12.00.xx4			
UL865-N3G	12.00.xxx			

	SW Versions	Audio paths	DVI document	AT Ref. Guide
GE Series			See chapter 1.5	
GE910-QUAD	13.00.xx4	HF	[5]	[1]
GE910-GNSS		HF		
GE910-QUAD AUTO	13.00.xx5	HF		
GE910-QUAD V3	16.00.xx3	HS, HF		

	SW Versions	Audio paths	DVI document	AT Ref. Guide
GL Series			See chapter 1.5	
GL865-DUAL V3	16.00.xx2	HS, HF	[6]	[1]
GL865-QUAD V3				
GL868-DUAL V3				
GE866-QUAD	16.00.xx3			

The present document describes the features of the products equipped with the software versions equal or higher than the versions shown in the tables.

¹ HE910 is the “type name” of the products marketed as HE910-G & HE910-DG

15 APPENDIX

In this Annexes section are collected detailed descriptions of some topics introduced in the previous chapters.

15.1 Microphone Line Guidelines

15.1.1 Single Ended Input Amplifier

The figure below shows a basic example of single ended input microphone amplifier (unbalanced MIC buffer).

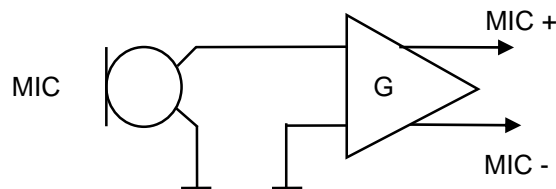


Fig. 18: Single Ended Input Amplifier

This configuration has this drawback: any noise signal appearing on input terminal is amplified as the desired signal coming from the MIC.

The advantage is that it has only one input signal therefore it requires less space on a PCB, and solves critical PCB space problems but, in most cases, introduces the need of strong shielding.

15.1.2 Differential Input Amplifier

The figure below shows a basic example of a differential input microphone amplifier (balanced MIC buffer).

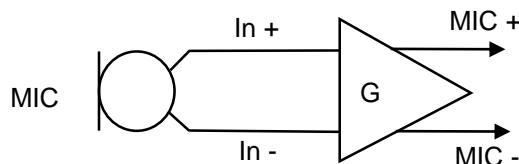


Fig. 19: Differential Input Amplifier

A differential input amplifier provides a Common Mode Rejection Ratio (CMRR): only the difference between the input terminals is amplified.

15.1.2.1 Practical Suggestions

Here is a list of rules concerning the PCB design for differential input amplifier (balanced MIC buffer) and mechanical rules for the MIC housing.

- Lay the differential PCB tracks close to each other from microphone to amplifier to improve EMI rejection.
- Lay the differential PCB tracks over a ground plane to shield them from EMI.
- Use equal AC coupling capacitors on both In+ and In- differential tracks. Ground the capacitor near the source to improve the common-mode noise cancellation, refer to figure below.

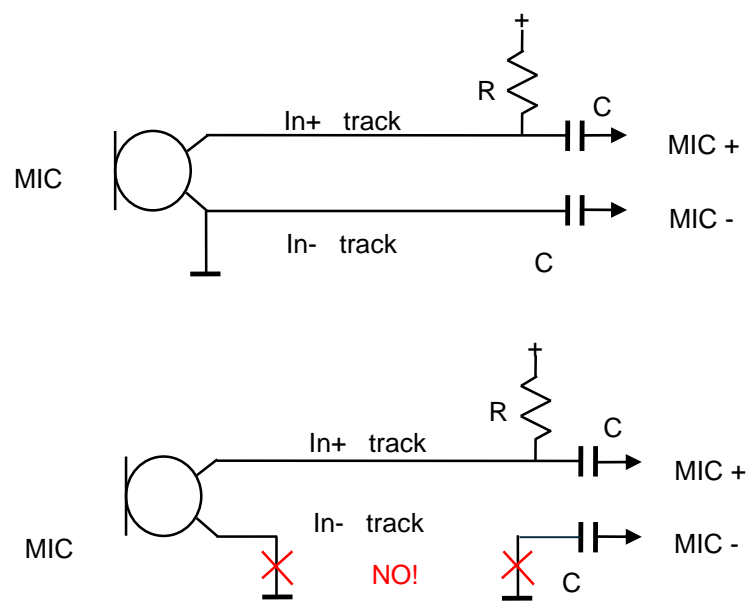


Fig. 20: AC Coupling Capacitors

- Use, as RF rejection, equal shunt capacitors or series ferrite beads on both In+ and In- differential tracks.

- Use, if balanced circuitry, equal load resistors for differential source on both differential tracks, refer to Fig. 22. For ECM microphone refer to chapter 15.2

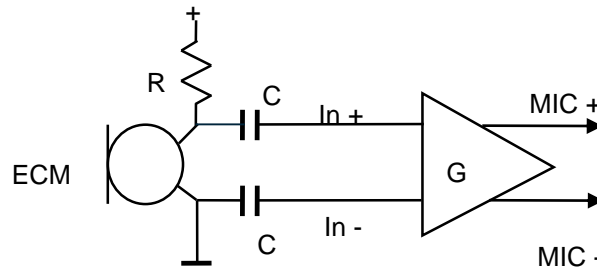


Fig. 21: Unbalanced Bias Circuitry

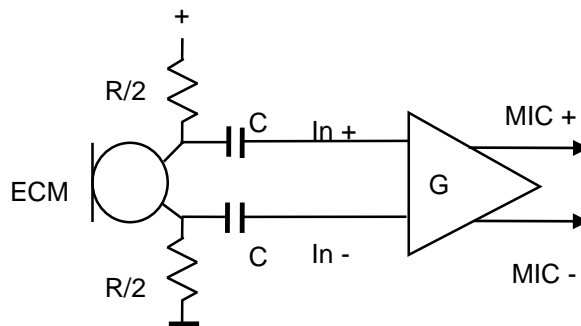


Fig. 22: Balanced Bias Circuitry

- Place a ground plane close (ideally beneath) the differential tracks. This provides a direct return path for RF signals to decrease coupling between audio and other signals.
- Use for the microphone biasing voltage a dedicated voltage regulator and a capacitor multiply circuitry (if needed), refer to chapter 15.2.3.
- Make sure that the microphone tracks in the PCB do not cross or run parallel to noisy tracks (especially the power line).

15.1.3 HandsFree

Here is a list of rules concerning the HandsFree system design with microphone fitted in the same case of the speaker.

- Use the maximum distance between microphone and speaker, at least 7 cm.
- Seal the rear side (no back cavity) of an omni-directional microphone type (typical application) to avoid unwanted signals; make divergent the microphone and speaker axes.
- Set the minimum internal gain of the module if you use an external microphone amplifier.

In a HandsFree system, the microphone and the speaker are far from module, driven by external amplifiers. When designing the external microphone amplifier you must take into account the voice attenuation, due to the distance between the talker and the microphone itself. In addition, you must consider that the microphone will pick up ambient noise.

To overcome these problems, it is preferable to set the gain of the microphone 10 dB lower than the calculated value for a "Nominal Sensitivity" see chapter 15.1.4.2. An increased voice volume of the talker, which will speak louder because of the ambient noise, will compensate the corresponding reduction in signal level.

Usually, the distance between the microphone and the talker in a car cabin is 40/50 cm. In these conditions, the attenuation is considered around 20 dB.

15.1.4 Definitions

15.1.4.1 *Normal Spoken Condition*

For a cellular phone, the “Normal Spoken” condition takes place when the talker mouth is about 7 cm far from the microphone. Under these condition the voice produces an acoustic pressure of -4,7 dBPa @ 1 kHz on the microphone membrane. During a call, this level varies according to the volume of the talker voice. Here is the typical dynamic range:

- The strongest voice level condition is when the talker is screaming: the signal increases by +20dB.
- The lowest voice level condition is when the talker is whispering: the voice level decreases by -50dB.

These limits must be considered for designing the external microphone amplifier.

15.1.4.2 *Nominal Sensitivity*

The “Nominal Sensitivity” is defined as follow:

- The output voltage level for a specified acoustic stimulus under specific load condition.

In the Microphone Manufacturer Data Sheet, the “Nominal Sensitivity” is expressed in dBV/Pa or in dBV/μbar. Because 0.1 Pa = 1 μbar, the difference between the two measuring units is:

$$\Delta = 20 \log_{10} 0.1 = -20\text{dB} \quad \text{as example: } -40 \text{ dBV/Pa} = -60 \text{ dBV/}\mu\text{bar}$$

15.1.4.3 *Microphone Electrical Level*

If the “Nominal Sensitivity” of a microphone is known, it is possible to calculate the voltage level on its pins under “Normal Spoken” condition. A microphone having the “Nominal Sensitivity” of -45 dBVrms/Pa will produce an electrical equivalent signal at its pins:

$$\text{Mic Level} = (-45) + (-4.7) = -49.7 \text{ dBV}_{\text{rms}}$$

$$\text{Mic Voltage} = 10^{(-49.7 / 20)} = 3.3 \cdot 10^{-3} \text{ V}_{\text{rms}}$$

15.2 Electret Microphone

15.2.1 Principle of Operation

The electret microphone is based on the electret material. It is a stable dielectric with a permanently embedded static electric charge, which will not decay for hundreds of years, due to the high resistance of the material. Its name comes from electrostatic and magnet.

The EleCtret Microphone (ECM) is a type of condenser microphone whose audio pickup section has a structure of a condenser consisting of a diaphragm and an opposite back plate. Refer to Fig. 23. Usually a very high voltage (tens or hundreds volts) should be applied externally to polarize such a condenser. However, because the electric charge can be maintained in a polymer film by the electret effect, thereby the polarizing direct-current high voltage is eliminated. Therefore, it could offer the desired long-term stability and ultra-flat frequency response.

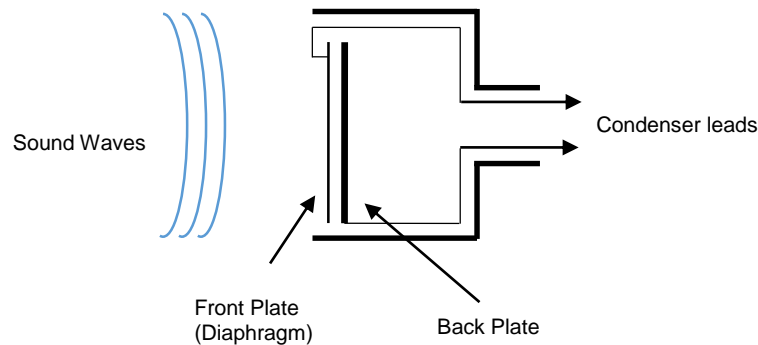


Fig. 23: ECM

The sound waves impinging on the diaphragm cause the capacitance between it and the back plate to change in sympathy.

15.2.2 Types

There are three ECM technologies, according to where the electret film is used:

- Foil-type. The diaphragm itself is made of an electret polymer film. This is the most common type, but offering also the lowest quality, since the electret material doesn't make a very good diaphragm.
- Back-type. An electret film is applied to the back plate of the microphone capsule and the diaphragm is made of a superior, uncharged material.
- Front-type. In this newer type, the back plate is eliminated, and the condenser is formed by the diaphragm and the inside surface of the capsule: the diaphragm is metalized and the electret film is adhered to the inside front cover.

Because the capacitance is relatively small, the electrical impedance is high and unmanageable; to overcome this, the metalized diaphragm is connected to the input of a FET to transform the impedance to a lower level suitable for feeding an amplifier via screened lead. The ECM can be thought as a voltage generator; by means of the FET it is converted in a current generator, see the figure and table below. The table lists some typical ECM electrical characteristics; verify on microphone manufacturer Data Sheet the characteristics of your ECM.

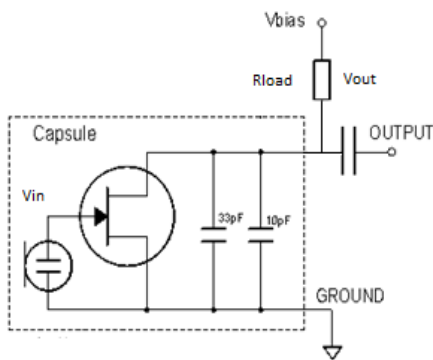


Fig. 24: ECM-FET Schematic Block

Nominal sensitivity	-45dBV _{rms} /1Pa (+/- 3dB)
Line coupling	AC
Nominal Voltage	2V
Range of Using Voltage	[1÷10] V
Consumption Current	[150 ÷ 500] µA
Impedance	2,2KΩ
Signal to Noise Ratio	56dB @ 1KHz/1Pa (A curve)
EMI capacitor between terminals	10pF, 33pF

Tab. 19: ECM Electrical Characteristics

An ECM usually needs a biasing voltage to work properly. Referring to Fig. 24: the R_{load} resistor polarizes the FET and converts its current signal in a V_{out} voltage signal:

$$V_{out} = G_{forward} \times R_{load} \times V_{in}$$

Increment the R_{load} resistor value to increment the V_{out} voltage. The saturation should be avoided.

ECMs have a hot lead where the positive biasing must be connected; the cold lead is connected to the capsule. If the polarity of the bias is reversed, then the microphone will not work properly, its sensibility decreases about of 30 dB.

The microphone leads and the PCB tracks work as antenna and pick up the module RF. The FET rectifies the RF and amplifies the rectified signal that cannot be more isolated from the right audio signal. To avoid this phenomena, the best microphones contain two small value capacitors inside the capsule (one for 900MHz and the second for 1800-1900MHz band), which act as RF bypass to short-circuit the RF frequency components to ground, avoiding rectification phenomena, see Fig. 24.

To improve the shielding action, the cold lead is directly connected to its shield, and a lot of ground vias improve the effect of the capacitors included in the capsule, see Fig. 24. The Fig. 25 shows, on the left side, the best shielded Electret Microphone.

When the shielding action is not enough, you hear a sort of noise (called buzzing), whose level depends from TX power.

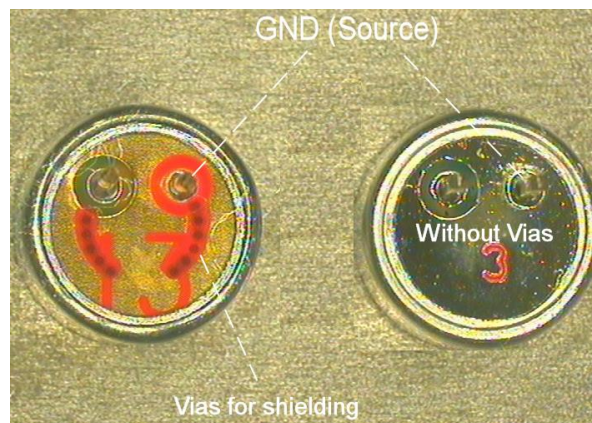


Fig. 25: Electret Microphones

15.2.3 Practical Suggestions

The figure below shows an example of the unbalanced microphone biasing circuitry. Here is a list of suggestions:

- Get the supply biasing voltage from a dedicated voltage regulator (U1) to eliminate the noise present on the power lines; this regulator can be the same for all the audio sections.
- Drive the microphone by a capacitor multiply circuit (R1-Q1-C2).
- Use a shielded cable if the microphone is wire connected.
- Use C4 and C3 capacitors equal or greater than 200nF to obtain a cut frequency lower than 300Hz. In addition, C3 should be connected to GND1, which should be equipotential with GND2.

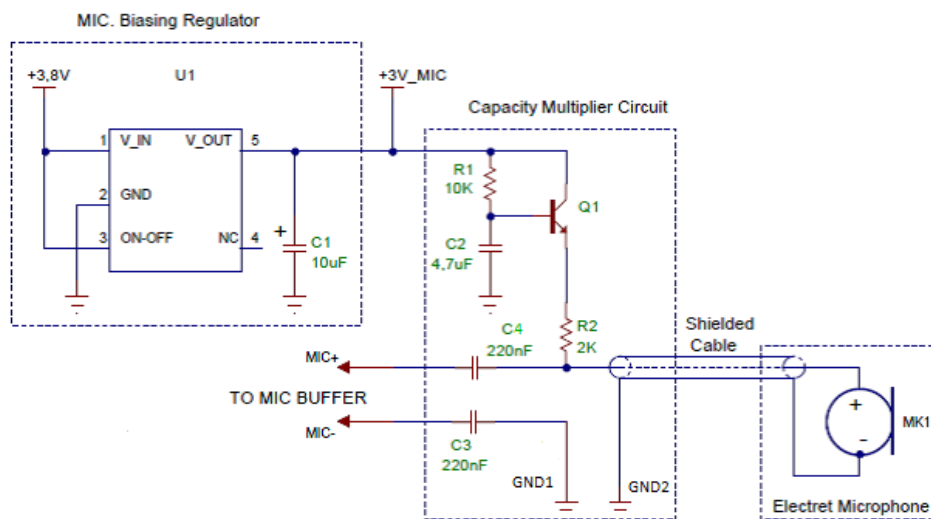


Fig. 26: Unbalanced Microphone Biasing Circuitry

15.3 Speaker Line Guidelines

15.3.1 Single Ended (SE)

A Single Ended Output Amplifier should be AC coupled to the load, as shown in Fig. 27. Without the output coupling capacitor, the half-supply bias across the load would result in both increased internal IC power dissipation and permanent speaker damage. It is an obsolete configuration.

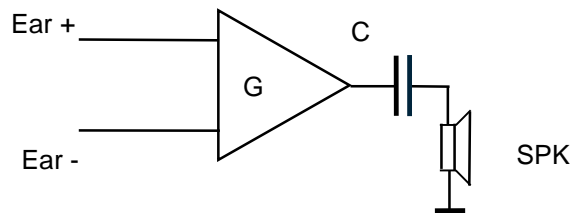


Fig. 27: Single Ended Output Amplifier

This configuration needs only one active track, the routing of signal line requires less PCB space.

15.3.2 Differential (BTL)

A Differential Output Amplifier provides a differential signal output, where the Vout- side is the mirror image of the Vout+ side, without any AC coupling output capacitor, as shown in Fig. 28. In the differential drive, while one side of the amplifier is slewing up the other side is slewing down, and vice versa: the voltage swing on the load is doubled.

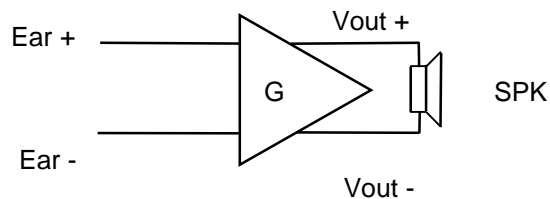


Fig. 28: Differential Output Amplifier

15.4 Audio Profile Parameters for AEC of the First Generation

The table below shows the values ranges of the AT#PSET command parameters, highlighting that the parameters affects three main audio features: Audio Equalizer, Sidetone, and AEC of the first generation. Refer to Fig. 1 to relate the command parameters with the affected audio section blocks, and Tab. 6 to see which modules families support the AT#PSET command.

		Parameter	Description	Range	Step
Audio Equalizer		<scal_in>	Digital Input Scaling	[-6÷ +6] dB	1dB
		<scal_out>	Digital Output Scaling	[-6÷ +6] dB	1dB
Sidetone		<side_tone_atten>	Sidetone Attenuation	[0÷ +24] Db; -1 = OFF	1dB
AEC	EC	<adaption_speed>	Adaption Speed	0 = slow 1 = medium 2 = fast	
		<filter_length>	FIR Filter Length	{10÷50} ms	1ms
		<rxtxrelation>	Power Relation RX→TX	[-90÷ +90] dB	1dB
	NR	<nr_atten >	Noise Max Attenuation	[6÷42] dB	6dB
		<nr_w_0>	Noise Weighting Factor Band 300-500Hz	0=lower weight ... 9=higher weight	
		<nr_w_1>	Noise Weighting Factor Band 500-4000Hz	0=lower weight ... 9=higher weight	
	AGC	<add_atten>	Gain Additional Attenuation	[0÷ +90] dB	1dB
		<min_atten>	Gain Minimum Attenuation	[0÷ +90] dB	1dB
		<max_atten>	Gain Maximum Attenuation	[0÷ +90] dB	1dB

Tab. 20: Audio Profile Parameters Ranges

Modules provide one standard and three extended basic AEC profiles, corresponding to four basic environments. Refer to table below.

Audio Profile	
Audio Pro. Identifier	Description
0	standard
1	office
2	open air
3	small /medium car

Tab. 21: Basic AEC Audio Profiles

All parameters are stored in NVM as default factory profiles, usable as they are or as a starting setup. Every profile can be used on both audio physical paths: HS, and HF.

		Audio Parameters	Audio Profile		
			Office	Open Air	Small/Medium Car
Audio Equalizer		Digital Input Scaling	0	0	0
		Digital Output Scaling	0	0	0
Sidetone		Sidetone Attenuation	-1	-1	-1
AEC	EC	Adaption Speed	2	2	2
		FIR Filter Length	50	50	50
		Power Relation RX→TX	-5	-10	-20
	NR	Noise Max Attenuation	6	12	12
		Noise Weighting Factor Band 300-500Hz	2	4	6
		Noise Weighting Factor Band 500-4000Hz	2	4	6
	AGC	Gain Additional Attenuation	6	9	12
		Gain Minimum Attenuation	0	0	0
		Gain Maximum Attenuation	12	18	24

Tab. 22: Audio Profile: Office, Open Air, Small/Medium Car

15.4.1 Configuration of an AEC of the First Generation

This chapter provides basic suggestions to configure an AEC of first generation, refer to Fig. 5.

The geometry of the used speaker-microphone system is a very important aspect of the AEC parameters tuning, therefore as first step:

- decide the spatial setup of the speaker-microphone system

To obtain the right speech levels:

- Set the internal microphone gain as low as possible by means of the audio AT commands;
- Use an external amplifier if the speaker requires additional gain; see Customer Audio Interface in Fig. 1;
- Set only internal gain for fine-tuning of the overall downlink gain. Use the audio AT commands.

Here is a list of tuning actions according to the used audio interface (AFE or DVI). Perform these actions only if needed:

AFE interface, tune:

- internal analog gains provided by the Audio Front End (AFE), see chapter 3.1;
- internal digital gains provided by the Audio Equalizer, see chapter 6.1;
- Biquad audio filters, provided by the Audio Equalizer, to equalize the transducers, see chapter 6.2.

DVI interface, tune:

- internal digital gains provided by the Audio Equalizer, see chapter 6.1;

- Biquad audio filters, provided by the Audio Equalizer, to equalize the transducers, see chapter 6.2.



Sidetone is always a harmful effect. In HandsFree application, it must be disabled to help Echo Canceller action.

EC: the max echo path

EC algorithm accepts an echo delay value equal or less than 50 ms, see chapter 10.1.1, and <filter_length> parameter in Tab. 20. For a good cancellation it is enough to take into account only the main echo coming from the near reflections.

The following expression provides the max echo path:

$$L = t \times v_s = 0,05[s] \times 340 \left[\frac{m}{s} \right] = 17m$$

Where:

L = max echo path

t = max echo delay

vs = sound velocity

EC: estimation of the RXTX Threshold

EC algorithm estimates the RX (Pdownlink) and TX (Puplink) signals to accomplish the Echo Canceller activity. Refer to Fig. 29, and Tab. 21. Because the algorithm does not take automatically in account the speaker amplifier gain as well as losses in the air path, this information has to be passed to EC.

The <rxtxrelation> parameter (Power Relation RX□TX) of the AT#PSET command is used to configure the EC algorithm in accordance with the current scenario.

In the following example is used the AFE interface; the gains values of the figure below are reported only as examples.

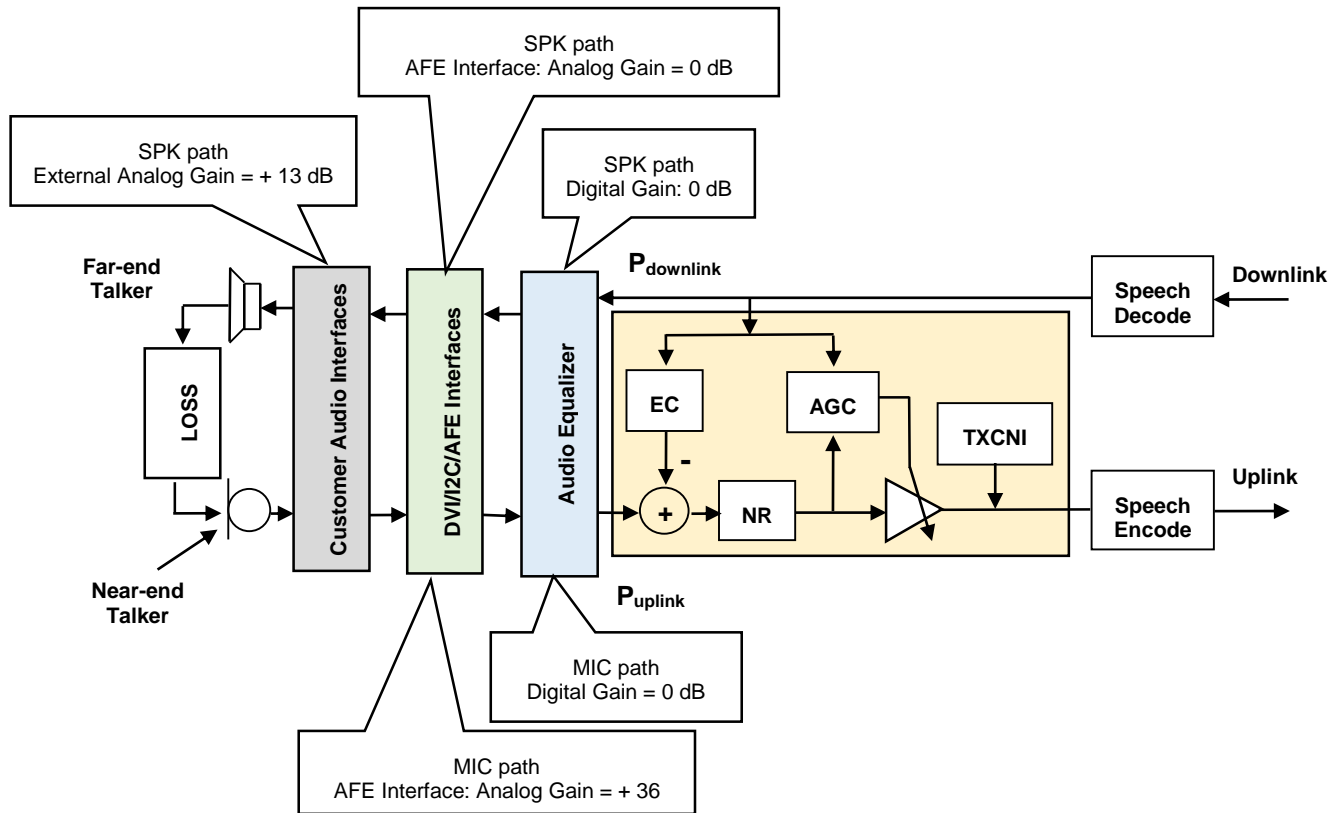


Fig. 29: AEC of First Generation: Echo Chain

The EC algorithm can compare the two power values P_{downlink} and P_{uplink} only if the following relation is respected:

$$P_{\text{uplink}} < P_{\text{downlink}} + (13 - \text{LOSS} + 36)$$

That means:

$$P_{\text{uplink}} - (49 - \text{LOSS}) < P_{\text{downlink}}$$

In Telit Lab has been arranged a measurement bench to evaluate the Echo LOSS in a silent room for a HandsFree application. The used position of the microphone and speaker was that suggested by ITU-T P.340 Recommendation. See Fig. 30.

In-air attenuation, speaker, microphone, and their coupling affect the estimation of the echo LOSS.

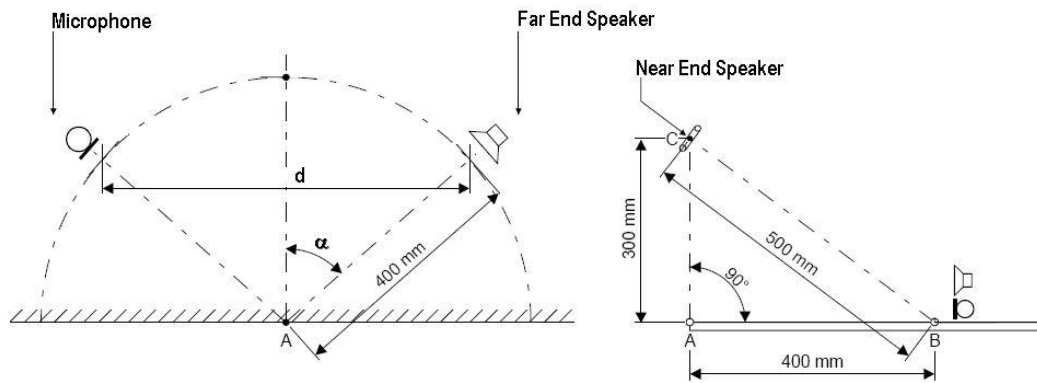


Fig. 30: Position of Microphone-Speaker System

Fig. 31 is the result of the evaluation of the ECHO LOSS:

- X axis indicates the <rxtxrelation> parameter (Power Relation RX→TX). It is the input variable of the measure, its value has been changed from – 40 dB to 50 dB to accomplish the measure itself
- Y axis indicates the result of the measure: the Echo Loss in dB

The figure shows that for <rxtxrelation> values less than about 5 dB, echo is reduced by 50 dB. For the spatial condition used during the measurement, 5 dB is the max limit to have 50 dB of echo attenuation.

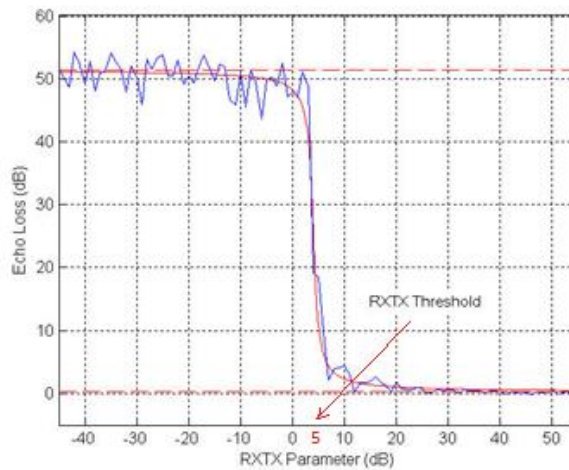


Fig. 31: Threshold Echo Loss Behavior



The RXTX Threshold value of 5 dB can guarantee a good behavior of AEC for the used spatial configuration; it is good practice to keep some dB below this value. Use 0 dB to have more margin.

Audio Configurations with different:

- amplifications
- spatial position
- transducers

need new parameters, here are some rules to obtain the new ones:

- Higher gains has to be compensated lowering the Power Relation RX→TX parameter.
- Lower gains has to be compensated increasing the Power Relation RX→TX parameter.
- Lower LOSS (because of reduction of the distance between speaker and microphone, or modifying the relative orientation of the two elements) has to be compensated like the case of higher gain, conversely if the LOSS increases.

When the Power Relation RX→TX has been adjusted working on the <adaption_speed> and <filter_length> parameters of the EC algorithm, the AGC algorithm can be tuned following the next steps:

- <min_atten> and <max_atten>: minimum and maximum attenuation define the range in which AGC works. It is dependent on application and increases with echo level (due to spatial position or high gains).
- <add_atten>: additional attenuation permits to the user to add a fixed contribution to the final value of gain calculated by AGC.

Suggested AGC parameters:

Typical values for Full Duplex:

- 6 dB for additional attenuation <add_atten>
- 0 dB for minimum attenuation <min_atten>
- 12 dB for maximum attenuation <max_atten>

These values are a good compromise between reducing echo and maintaining double-talk quality.

If you need a high echo cancellation and the full duplex feature is not required, it is possible to reproduce a half-duplex behavior setting:

- <add_atten> = 90 dB for additional attenuation
- <max_atten> = 90 dB for maximum attenuation

When Power Relation RX→TX gets over the RX→TX Threshold value, the AGC starts to work using the maximum attenuation (90 dB).

Intermediate solutions between Full and Half duplex are very closely related to particular application.

Suggested NR parameters:

The tuning of the NR algorithm is based on subjective tests.

For maximum attenuation, typical value of <nr_atten>, parameter is in the range: 6 ÷ 18 dB, bigger values determine very poor speech quality, see Tab. 20.

When tuning the weights parameters <nr_w_0>, <nr_w_1> it is important to start with the lower ones and then increase values, keeping a small difference between the two (at least making them equal), see Tab. 20.

15.5 Audio Codec Information and Setting

These AT commands could be useful to improve the performances of the DTMF and eCall features.

Use the following AT command to get codec information about a voice call. It is valid for both GSM/GPRS and HSPA-GSM/GPRS Standards. Even if the mobile is registered on UTRAN network, usually the Operator assigns to the mobile a GSM channel to carry on a voice call.

AT#CODECINFO = <format>,<mode>

Example:

AT#CODECINFO=1,1 ← enable codec information
OK

ATD<phone number>;
#CODECINFO: "HAMR","FR","EFR","HR","FAMR","HAMR"
OK

NO CARRIER ← remote hang up
#CODECINFO: "None","FR","EFR","HR","FAMR","HAMR"

Use the following AT command to select a codec for a voice call. It is valid for both GSM/GPRS and HSPA-GSM/GPRS Standards. Even if the mobile is registered on UTRAN network, usually the Operator assigns to the mobile a GSM channel to carry on a voice call.

AT#CODEC = <codec>

Example

AT#CODEC?

#CODEC: 0 ← all the codec are enabled
OK

AT#CODECINFO=1,1 ← enable codec information
OK

ATD<phone number>; ← establish the call
#CODECINFO: "HAMR","FR","EFR","HR","FAMR","HAMR"
OK

NO CARRIER ← remote hang up
#CODECINFO: "None","FR","EFR","HR","FAMR","HAMR"

AT#CODEC=1 ← select FR mode
OK

ATD<phone number>; ← establish the call
#CODECINFO: "FR","FR"
OK

NO CARRIER ← remote hang up
#CODECINFO: "None","FR"

16 DOCUMENT HISTORY

Revision	Date	Changes
0	2007-04-13	First issue
1	2008-01-18	<ul style="list-style-type: none">• Added GE863-SIM to the Application Table• Correct AT#VTS
2	2008-04-17	<ul style="list-style-type: none">• Editing changes and integration fix of Rev. 0 and Rev. 1
3	2009-05-14	<ul style="list-style-type: none">• Deleted UC864-E from Applicability table• Introduced family models contents' discrimination.• Applied the new documents' layout• Fixed minor references errors (figures' and tables' numbers)• Deleted the Safety Recommendations paragraph• Edited the EVK2 paragraph• Edited Figures 17,19• Added figure 20bis
4	2010-10-04	<ul style="list-style-type: none">• Added GE/GC864-QUAD V, GE864-QUAD Automotive V2 and 2GL865-DUAL to the applicability table
5	2016-02-22	This revision uses a new format. New topics and Modules & SW Versions Tables have been added. The chapters structure has been changed.

