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LEON-G100 / LEON -G200

LEON Audio Application Note

Application Note

Abstract

This document provides information and procedures for the resolution of audio related problems on LEON-G100 / LEON-200 GSM/GPRS modules.
Moreover a procedure for tuning of Hands-Free algorithm (Echo canceller, Automatic Gain Control, Noise Reduction) is described.

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Contents

1 Introduction

This document provides information and procedures for the resolution of potential audio related problems on the LEON-G100/LEON-G200 GSM/GPRS module as well as the DTMF signaling decoder functionality available through +UDTMFD AT command, implemented following the multi-part ETSI Standard ES 201 235 [6]. As an example, the document describes a procedure for tuning the hands-free algorithm (Echo canceller, Automatic Gain Control, Noise Reduction).

For a detailed description of audio parameters and AT commands, refer to the u-blox AT Commands Manual [1].

For a detailed description of LEON audio features refer to the LEON-G100/LEON-G200 System Integration Manual [2].

The following symbols are used to highlight important information within the document:

☞ An index finger points out key information pertaining to integration and performance.

⚠ **A warning symbol indicates actions that could negatively impact or damage the module.**

☞ This document applies to the following products:

- LEON-G100 series
- LEON-G200 series

2 Introduction to HF algorithm tuning

After the connection of external audio devices (i.e. microphone and loudspeaker) to the wireless module, an acoustic echo might be heard by the far-end user. This problem typically occurs when the device gain is set high to work at a distance (i.e. in hands-free application). LEON provides a hands-free algorithm to remove echo. This algorithm is controlled by HF parameters within the uplink audio path in use (refer to u-blox AT Commands Manual [1], +USPM and +UHFP command), that can be stored in EEP dynamic parameters profile (refer to u-blox AT Commands Manual [1], chapter 3.1 and &W command).

Chapter 3 presents a step-by-step procedure for choosing parameters removing echo heard on the far end side. In case of HF systems with high echo coupling and high non-linearity on the loudspeaker (making the EC cancellation ineffective), a No duplex set-up is recommended, as presented in chapter 4.

The section 2.1 reports a description of HF algorithm and parameters meaning.

2.1 HF algorithm description

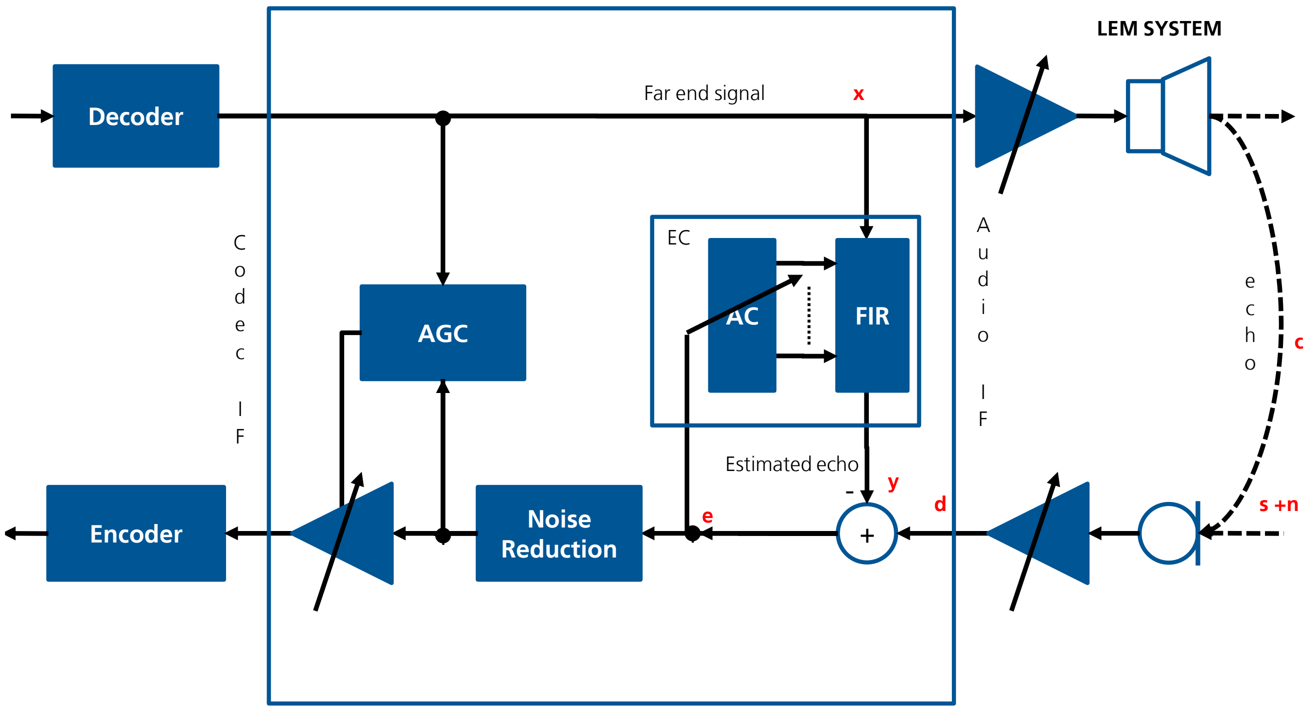

Figure 1: Block diagram for hands-free

The LEM system (Loudspeaker-Enclosure-Microphone) is a non-linear, time varying system.

The FIR filter emulates the LEM system behavior generating an estimate (y) of the acoustic echo (c) produced by signal (x) on LEM. This estimated echo is subtracted from the microphone signal (d =s+c+n) where near-end speech (s) and noise (n) are present.

Since the LEM is time varying, the coefficients of the FIR filter must be adapted by the Adaptation Control block (AC); AC is a Block NLMS adaptive algorithm based on the residual echo (e) heard when the near-end speaker is silent (s=0). When the near-end speaker is not silent (double talk condition) the filter adaptation is suspended. The detection of double talk condition is based on the known TX and RX power relation in single talk condition (s=0). RxTx relation is a fixed parameter of the LEM.

Residual echo and noise (e), are then lowered by Noise Reduction (NR) and Automatic Gain Control (AGC). AGC is disabled when in double talk.

2.2 HF algorithm parameters

HF algorithm parameters can be set using the corresponding parameters of the AT+UHFP (more details in u-blox AT Commands Manual [1]).

2.2.1 Parameters for block activation and initialization

HF_ALGORITHM_INIT

This parameter is used by the audio driver when a call is started to initialize the algorithms.

HF_ALGORITHM_RESTART

This parameters is used by the audio driver during a call to restart adaptation without reinitializing (for example after a handover).

These parameters are sets of flags that control the activity and initialization of the EC, AGC and NR blocks.

Flag	Meaning
Bit #0 set	Echo Cancelation (EC) initialization
Bit #1 set	EC restart (without coefficient initialization)
Bit #2 set	EC on
Bit #3 set	EC adaptation on
Bit #4 set	Noise Reduction initialization
Bit #5 set	Noise Reduction on
Bit #6 set	Noise reduction works with additional AGC
Bit #7 set	Automatic Gain Control (AGC) initialization
Bit #8 set	AGC on

Table 1: HF_ALGORITHM_RESTART flags explanation

The bits setting is not mutually exclusive; more than one bit can be set at the same time.

Examples:

Configuration	Remarks
SWITCH =0x018D =bin 000110001101	EC initialized and on EC adaptation on Automatic Gain Control initialized and on
SWITCH =0x010E =bin 000100001110	EC on EC adaptation on EC restart Automatic Gain Control on

Table 2: HF_ALGORITHM_RESTART examples

2.2.2 Parameters for FIR filter adaptation

Parameter	Range	Default value	Remarks
STEP_WIDTH	0 to 32767	13107	This parameter impacts how fast the FIR coefficients change. The higher this value, the echo characteristic adaptation is faster, but the echo cancellation accuracy reduces. Lower values assure more accurate (but slower) convergence. Limit: STEP_WIDTH *BLOCK_LENGTH<=2*32767
LMS_LENGTH	2 to 400	300	This is the maximum impulsive response of the FIR filter considered by the adaptive LMS algorithm, in samples. Max time length: 400*T = 50 ms Length of the filter depends on the delay of echo and reverberation respect to the downlink signal (echo path length). For example in a car, the typical echo path is around 30-32 ms. So 30 ms / 125 µs = 240 samples. LMS_LENGTH = 240 Limitation: 2<= LMS_LENGTH+ LMS_OFFSET<=400 (DSP memory limit)
LMS_OFFSET	0 to 400	8	This parameter is used by the LMS adaptation algorithm and indicates the expected delay of the echo after the RX signal, in samples. Example of calculation: Sample period T= 1/8000 s= 125 µs Loudspeaker to microphone distance on a phone: L= 10 cm Sound velocity V= 340 m/s Delay of echo D =L/V = 0.1/ 340 = 294 µs Number of samples=D/T= 2.35 → LMS_OFFSET =2
BLOCK_LENGTH	2, 4, 5 and 8	4	LMS coefficient adaptation block length. The higher this number, the slower but more accurate the adaptation converge
RXTX_RELATION	-960 to 960	-300	This parameter checks the power relation between Rx (loudspeaker) and Tx (microphone) signals to recognize the double talk condition from the echo condition. The system is considered to be in double talk condition when the TX power (microphone signal) is higher than the maximum expected echo power: Tx(dB) > Rx (dB) - RxTx(dB) with RxTx(dB)=RXTX_RELATION*3/32

☞ This is the most critical parameter in hands-free operation. Values typical for handset are in the range of 50 to 150 while for back speaker the range goes from -100 to -400.

When in double talk, adaptation of FIR and AGC is suspended.

2.2.3 Parameters for AGC

Parameter	Range	Default value	Remarks
ADD_ATTEN	0 to 960	0	This value is added to the calculated attenuation as bias
MIN_ATTEN	0 to 960	0	Minimum attenuation of the microphone signal by the AGC. If calculated attenuation is lower than MIN_ATTEN, then attenuation is increased to MIN_ATTEN.
MAX_ATTEN	0 to 960	491	Maximum attenuation of the microphone signal by the AGC. If the calculated attenuation is higher than MAX_ATTEN, then the attenuation is decreased to MAX_ATTEN.

Table 3: AGC parameters description

The AGC parameters update the following attenuations

- Additional Attenuation Level (dB) = 3/32* ADD_ATTEN
- Minimum Attenuation Level (dB) = 3/32* HF_MIN_ATTEN
- Maximum Attenuation Level (dB) = 3/32* HF_MAX_ATTEN

2.2.4 Parameters for Noise Reduction

The Noise Reduction operates on 8 frequency bands (band 0: 0-250 Hz; band 1: 250-750 Hz...band 7: 3250-3750 Hz). In band 0 the ear is less sensitive. For each band the NR computes a gain to be applied (attenuation).

Parameter	Range	Default value	Remarks
NR_SW_2	0 to 32767	4096	This is the maximum attenuation that can be introduced by NR. It is linear; where 32767 means 1 (0 dB; in this case no attenuation allowed, so there is no noise reduction). Very low values allow a strong attenuation but voice can result distorted (metallic). A good compromise is that the value is included in the range that goes from 4096 to 16384 (-18 to -6 dB)
NR_U_FAK_0	0 to 16384	4096	This is the weighting factor for frequency band 0 (0 Hz - 250 Hz). Increasing this factor causes a better noise reduction in this band but also higher distortion of speech. Linear; weighting factor =NR_U_FAK_0 / 32768
NR_U_FAK	0 to 16384	4096	Factor of NR in the bands 1 to 7 (250 Hz - 3750 Hz). This is the weighting factor for frequency band 0 (1 to 7 (250 Hz - 3750 Hz). Increasing this factor causes a better noise reduction in this band but also higher speech distortion. Linear; weighting factor =NR_U_FAK_0 / 32768

Table 4: Noise reduction parameters description

Examples:

Configuration	Value	Remarks
NR_SW_2	4096	0.125 = -18 dB gain (18 dB is the maximum attenuation)
NR_U_FAK_0	16384	Weighting factor = 0.5
	8192	Weighting factor = 0.25
NR_U_FAK	16384	Weighting factor = 0.5
	4096	Weighting factor = 0.125

Table 5: Noise reduction parameters examples

3 Procedure for Echo Canceller tuning

This is a step by step procedure to tune parameters on the audio path in use (Refer to u-blox AT Commands Manual [1] for the AT+USPM command description) for the removal of the echo heard on the far-end side.

☞ Refer to u-blox AT Commands Manual [1] for more details on parameters in all AT commands to be used in the tuning procedure (AT+USGC, AT+UMGC, AT+UHFP, AT+UUBF, AT+UDBF, AT&W, AT&F, AT&Y). Check the path index for uplink and downlink to be used in these commands.

1. Regulate the gain on speaker and microphone used so that speech is not distorted on both uplink and downlink.

☞ This is very important because the Echo Canceller algorithm works efficiently only in linear mode. It should be checked (if possible also by oscilloscope) that the speech signal is not clipped.

2. Tune the gain on downlink path using AT+USGC command if the speech signal on speaker is distorted.

3. Tune the gain on uplink path using AT+UMGC command, if the speech signal from microphone is distorted.

4. Begin tuning the HF parameters, starting with the AT command:

   AT+UHFP=<uplink_path_num>,0x000d,0x000e,30000,250,0,2,-960,0,0,500,4096,16384,16384

   where the syntax command is

   AT+UHFP=<uplink_path_num>,<hf_algorithm_init>,<hf_algorithm_restart>,<step_width>,<lms_length>,<lms_offset>,<block_length>,<rxtx_relation>,<add_atten>,<min_atten>,<max_atten>,<nr_sw_2>,<nr_u_fak_0>,<nr_u_fak>

   and the parameters meaning is:

<hf_algorithm_init>=0x000D	(EC started and initialized)
<hf_algorithm_restart>=0x000E	(EC restarted without reinitializing)
<step_width>=30000	
<lms_length>=250	(250*0.125 =31.25 msec)
<lms_offset>=0	(0 msec)
<block_length>=2	
<rxtx_relation>=-960	(-960*3/32 = -90 dB)
<add_atten>=0	(0 dB minimum AGC attenuation)
<min_atten>=0	(0 dB additional AGC attenuation)
<max_atten>=500	(500*3/32 = 47 dB maximum AGC attenuation)
<nr_sw_2>=4096	(20 log(4096/32767) = -18 dB minimum NR attenuation)
<nr_u_fak_0>=16384	(16384/32768 = 0.5 weighting factor for frequency band 0)
<nr_u_fak>=16384	(16384/32768 = 0.5 weighting factor for frequency band 1-7)

   Since the algorithm is adaptive, some seconds are needed to converge after that AT+UHFP command is sent. Reconsider points 2 to 4, if this command has no effect on Echo.

   If EC correctly works a difference should be heard turning off the EC with the following AT command.

   AT+UHFP=<uplink_path_num>,0x0000,0x0000,30000,250,0,2,-960,0,0,500,4096, 16384, 16384

(echo returns)

☞ Particular cases:

- LMS_LENGTH can be raised in case of loud reverberation
- LMS_OFFSET can be raised in case of long delay in echo
- In very critical case, if echo never disappears, try to find a minimum residual echo configuration

5. Starting from the parameter configuration of none or minimum echo, raise RXTX_RELATION from -960 to xxx where the echo suppression starts to fail. Then set RXTX_RELATION to xxx - 50

   Expected final value should be around -300 for a loudspeaker system, about 100 for a handset.

6. Some improvements can be achieved if the values STEP_WIDTH, BLOCK_LENGTH are changed.

☞ This limit must always be respected: STEP_WIDTH *BLOCK_LENGTH<=2*32767

Lower values of STEP_WIDTH assure more accurate (but slower) convergence.

Higher values of BLOCK_LENGTH assure more accurate (but slower) convergence.

☞ Searching for the best accuracy / speed balance for EC convergence, start to attempt within these limits:

BLOCK_LENGTH=2 STEP_WIDTH=15000 to 32000

BLOCK_LENGTH=4 STEP_WIDTH= 7500 to 15000

BLOCK_LENGTH=5 STEP_WIDTH= 6000 to 12000

7. Lower the AGC and NR algorithms setting the following parameters to remove a minimal residual echo is present (if present):

   HF_ALGORITHM_INIT=0x01FD

   HF_ALGORITHM_RESTART=0x016E

☞ Raise also ADD_ATTEN with steps of 30 (3 dB).

8. The final setting should be like this example:

   HF_ALGORITHM_INIT=0x01FD

   HF_ALGORITHM_RESTART=0x016E

   STEP_WIDTH=30000

   LMS_LENGTH=250

   LMS_OFFSET=3

   BLOCK_LENGTH=2

   RXTX_RELATION=-300

   ADD_ATTEN=0

   MIN_ATTEN=0

   MAX_ATTEN=200

   NR_SW_2=4096

   NR_U_FAK_0=16384

   NR_U_FAK=16384

9. The parameters can be stored in EEP dynamic parameters profile using the AT&W command (refer to the chapter 3.1).

3.1 Storing the parameters in the profile

This procedure must be followed to save the parameters in the EEP dynamic parameters:

a. Write the run-time configuration to NVRAM with the AT&W command (e.g. AT&W0; more details in the u-blox AT Commands Manual [1])
b. Assure the boot loading is performed with the desired parameter profile (e.g. profile 0 if the parameter saving was performed with AT&W0; use AT&Y0 to select this)
c. Save the run-time configuration to Flash memory with the AT+CPWROFF command
d. Reboot/PWR_ON reset of the device

4 No-duplex configuration for highly distorting HF device with high echo couplings

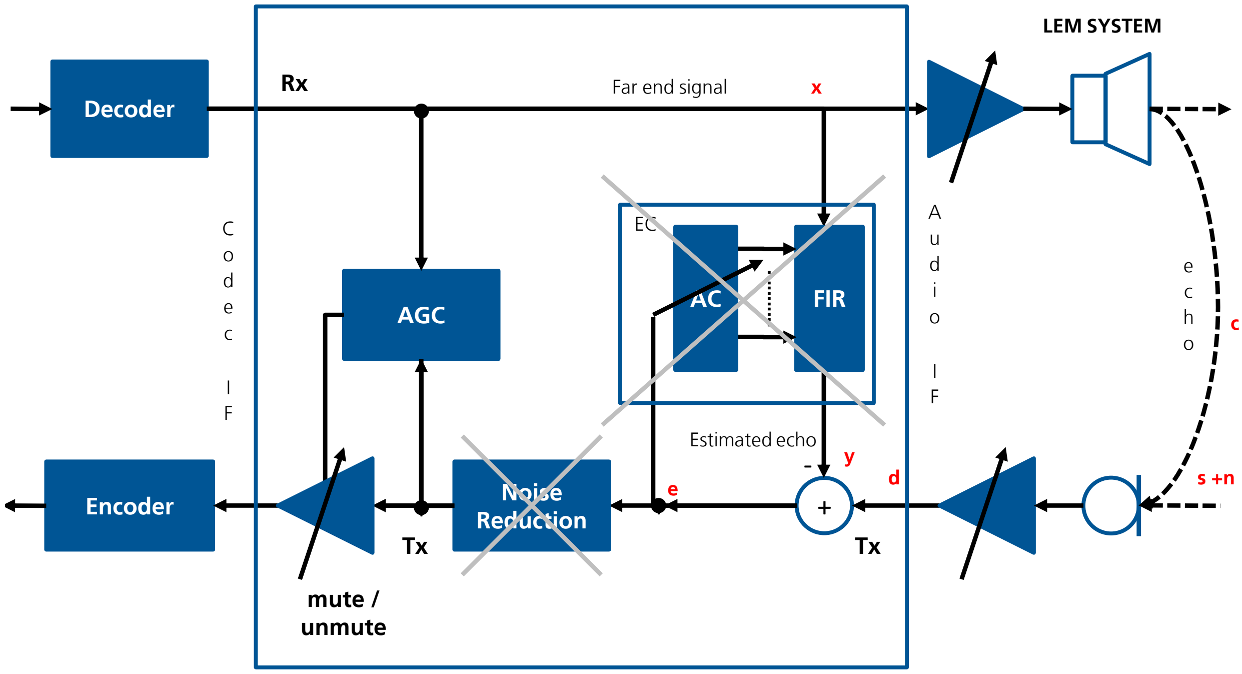

Figure 2: LEM system for highly distorting

The no-duplex set-up is recommended for HF systems with high echo coupling (e.g.: RXTX_RELATION < -300) and high non-linearity on loudspeaker which make ineffective the EC cancellation.

☞ In particular, the no-duplex configuration makes use of the AGC only as muting/un-muting device of the TX path (see Figure 2).

The operating conditions of this no-duplex configuration are:

1. Far-end speech (receiver) active: Tx muted by AGC, high and constant attenuation
2. Far-end speech (receiver) inactive: Tx un-muted (AGC off), both when near-end speech is active or inactive

The muting/un-muting is governed by the following equation:

If (Rx(dB) > Tx(dB) + RxTx(dB))

Tx is muted

Else

Tx is un-muted

where Rx(dB) and Tx(dB) are the power estimates of the Rx and Tx signals, expressed in dB (see 2.2.2 for RxTx(dB) and further details).

Best no-duplex behavior can be reached when:

1. Best no-duplex behavior on Tx muting is achieved with high echo levels (which shall lead to low RXTX_RELATION values, e.g. RXTX_RELATION < -300).

   In general, the echo power level present on Tx path can be expressed as:

   Echo(dB)= Ec + Rx(dB)

   where Ec is the echo coupling level, expressed in dB. RXTX_RELATION tuning actually represents the estimation procedure of Ec.

   Ec can be non-linear, i.e. could depend on Rx power: Ec = function of Rx(dB). A non linearity of coupling levels due to loudspeaker saturation (i.e. Ec (high RX(dB)) < Ec (low Rx(dB))) can improve the switching capabilities.

2. Best no duplex behavior on Tx un-muting: noise is present on Tx path when both near-end and far-end are inactive. The noise can be due to:

   a. Noisy near-end environment
   b. Noisy HF microphone

4.1 Echo coupling estimation / RXTX_RELATION setting procedure

The following convention are assumed in this chapter:

- RxTxRel: RXTX_RELATION, parameter to be tuned
- RxTxRel*: tuned value of RxTxRel
- Near-end speech: talker's speech at the HF device
- Far-end speech: talker's speech at the receiver's end, where the echo is heard
- Receiver: far-end

Step 1: testing environment preparation

1a. Put the hands-free device in a typical environment (e.g. with some background noise). The hands-free device will be the near-end side.

1b. Use a PSTN line telephone located in another room as far-end (receiver, who is hearing echo). Optionally, use a mobile with headset. Avoid couplings between near-end and far-end.

Step 2: HF device configuration

2a. Configure AGC to high attenuation performance, e.g.:

MIN_ATTEN =500

MAX_ATTEN =500

ADD_ATTEN =500

2b. Disable EC/NR and leave AGC only:

Init Command: 0x180

2c. Set microphone/loudspeaker gains to their final values.

☞ Afterwards, the gain must always be set to the values set in this step.

It is recommended that the microphone gain is high enough to have a clear speech transmitted on TX even with the near end talker far away from the HF device.

High microphone gain can also be needed to increase the background noise level when near-end and far-end are inactive.

There is no limit in increasing the loudspeaker gain except for the intelligibility of speech at maximum volume. Moderate saturation of the loudspeaker power level can be advantageous for switching performance.

Step 3: make a call and set the HF loudspeaker volume high

3. Set the loudspeaker volume to high: e.g. AT+CLVL=80

Step 4: muting/unmuting of the Tx path and the echo coupling level of the device

4a. Echo coupling level: set RxTxRel to 960. Talk intensively at the receiver's side. The full echo should be heard. Tx path is un-muted.

4b. Tx muting: Set RxTxRel -960 (preferably within the same call). Talk intensively at the receiver's side. No echo, no noise should be heard: Tx path is muted.

Step 5: RxTxRel tuning

☞ A negative value is expected, < -300

5a. During the call, increase RxTxRel (starting from -960) until an echo on far-end receiver's side is heard. Ensure that the near-end activity is minimal to none (i.e. do not shout into the hands-free device).

The echo appears with RxTxRel = -300:

RxTxRel_EL=-300

5b. Decrease RxTxRel for e.g. 100 (~10 dB) and verify that no echo is heard.

After this RxTxRel*=-400

Step 6: Switching performance verification with estimated RxTxRel*

The step is needed to verify that the HF device has the following behavior:

a. far-end speech (receiver) active: Tx muted

b. far-end speech (receiver) inactive: Tx unmuted, both when near-end speech is active or inactive

The behavior is verified at the far-end (receiver's) side.

6a. Verify, with inactive near-end, that there is no echo during far-end speech activity (Tx muted). If there is some echo, check again 5b) and in case restart tuning from step 4.

6b. Verify that, with inactive near-end, the near-end background noise (Tx unmuted) is heard. If Tx is muted, increase RxTxRel* until Tx is unmuted.

Case 1): RxTxRel*> RxTxRel_EL.

- Verify that EC and NR are off
- Verify that the HF device is not in noise-free conditions (anechoic room, etc)
- Verify that the microphone gain is high enough

☞ If it still persists that RxTxRel*> RxTxRel_EL, the device is very likely not suitable for operation in no-duplex HF configuration.

6c. Verify the switching threshold in presence of double talk, i.e. the minimum distance of the near-end speech source to the microphone, or the maximum admissible level of near-end background noise.

With both near-end and far-end active, verify if and when the switching from muted Tx to unmuted Tx occurs, increasing the near-end speech power. Increase the near-end speech by reducing the distance between the speaker and the HF microphone and/or speaking loudly. Verify the effect on the receiver's side (both echo and near-end speech pass through the unmuted Tx). If the switching is likely to occur during normal operating conditions, and the results are very annoying, try decreasing the microphone settings: restart from 2c).

4.2 No duplex configuration example

HF_ALGORITHM_INIT=0x0180
HF_ALGORITHM_RESTART=0x0100
STEP_WIDTH= don't care
LMS_LENGTH= don't care
LMS_OFFSET= don't care
BLOCK_LENGTH= don't care
RXTX_RELATION= has to be estimated
ADD_ATTEN=500
MIN_ATTEN=500
MAX_ATTEN=500
NR_SW_2= don't care
NR_U_FAK_0= don't care
NR_U_FAK= don't care

5 Hands-free device examples

Some examples for the hands-free device setting are reported as follows. The examples change the HF algorithm setting and impact the uplink speech path only.

It is supposed the gains on speaker and microphone used were already tuned so that speech is not distorted on both uplink and downlink (See point 1-3 of the chapter 3).

In all the examples the <uplink_path_num> parameter is speech uplink path currently used. This value must be the same as <main_uplink> parameters in +USPM command and it can be checked with the following command:

Command	Response
AT+USPM?	+USPM: <main_uplink>,<main_downlink>,<alert_sound>,<headset_indication>,<vmic_ctrl> OK

5.1 Weak AGC examples

Scenario	Command
RXTX_RELATION=-100	AT+UHFP=<uplink_path_num>,0x01fd,0x016e,30000,250,3,2,-100,0,0,200,4096,16384,16384
RXTX_RELATION=-200	AT+UHFP=<uplink_path_num>,0x01fd,0x016e,30000,250,3,2,-200,0,0,200,4096,16384,16384
RXTX_RELATION=-300	AT+UHFP=<uplink_path_num>,0x01fd,0x016e,30000,250,3,2,-300,0,0,200,4096,16384,16384

5.2 Moderate AGC examples

Scenario	Command
RXTX_RELATION=-200	AT+UHFP=<uplink_path_num>,0x01fd,0x016e,30000,250,3,2,-200,100,100,500,4096,16384,16384
RXTX_RELATION=-300	AT+UHFP=<uplink_path_num>,0x01fd,0x016e,30000,250,3,2,-300,100,100,500,4096,16384,16384
RXTX_RELATION=-400	AT+UHFP=<uplink_path_num>,0x01fd,0x016e,30000,250,3,2,-400,100,100,500,4096,16384,16384

5.3 Strong AGC examples

Scenario	Command
RXTX_RELATION=-200	AT+UHFP=<uplink_path_num>,0x01fd,0x016e,30000,250,3,2,-200,200,200,500,4096,16384,16384
RXTX_RELATION=-300	AT+UHFP=<uplink_path_num>,0x01fd,0x016e,30000,250,3,2,-300,200,200,500,4096,16384,16384
RXTX_RELATION=-400	AT+UHFP=<uplink_path_num>,0x01fd,0x016e,30000,250,3,2,-400,200,200,500,4096,16384,16384

5.4 No duplex configurations examples

Scenario	Command
RXTX_RELATION=-300	AT+UHFP=<uplink_path_num>,0x0180,0x0100,30000,250,3,2,-300,500,500,500,4096,16384,16384
RXTX_RELATION=-400	AT+UHFP=<uplink_path_num>,0x0180,0x0100,30000,250,3,2,-400,500,500,500,4096,16384,16384
RXTX_RELATION=-500	AT+UHFP=<uplink_path_num>,0x0180,0x0100,30000,250,3,2,-500,500,500,500,4096,16384,16384

6 DTMF decoder

6.1 About ETSI DTMF

Dual-tone multi-frequency (DTMF), also known as Touch Tone, is used for telephone signaling over the line in the voice frequency band to the local exchange. Prior to DTMF the phone systems had used a system known as pulse dialing to dial numbers, which works by rapidly disconnecting and connecting the calling party's phone line, like turning a light switch on and off.

The multi-part ETSI Standard ES 201 235 [6] specifies how to apply the Dual Tone Multi-Frequency (DTMF) signaling system to transmitters and receivers. It conforms to the International Telecommunication Union (ITU-T) Recommendation Q.23 [3] and it provides a complete set of requirements for all applications intending to use DTMF signaling.

The level of detail enables manufacturers of telecommunications equipment incorporating DTMF signaling to design the equipment such that it facilitates highly reliable signaling. It applies to DTMF signaling in the local access network, in which the transmission path between transmitter and receiver corresponds to a 2-wire analogue subscriber line, as well as to DTMF signaling over an end-to-end transmission path in the telecommunication network.

6.2 About DTMF

Dual-tone multi-frequency signaling is a standard in telecommunication systems. It has been gaining popularity for some years now because of its numerous advantages over the traditional telephone signaling scheme. In the DTMF scheme, a telephone is equipped with a keypad as shown in Figure 3. The A, B, C, and D keys are usually not present on a regular telephone keypad. Each key represents the sum of a pair of tones. One tone is from the high-frequency group between 1 kHz and 2 kHz, and the other tone is from the low-frequency group below 1 kHz. These frequencies are selected carefully so that the DTMF signal, which is the sum of the two tones, can be clearly distinguished as the signaling tone even in the presence of speech waveforms that might occur on the line.

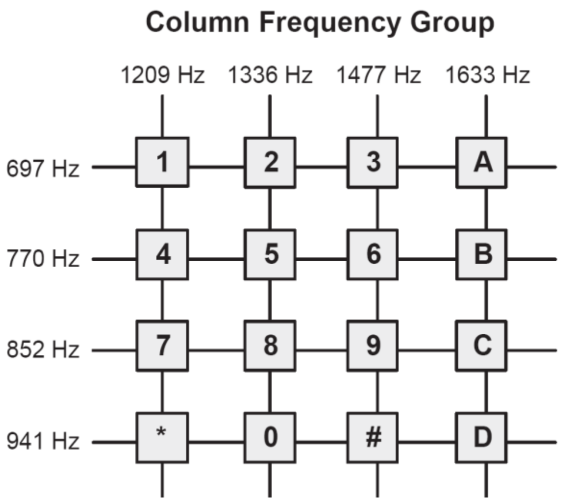

Figure 3: Touch-Tone telephone keypad: a row and a column tone is associated with each digit

6.2.1 The DTMF signal definitions

The tone frequencies, as defined by the Precise Tone Plan, are selected such that harmonics and inter-modulation products do not cause an unreliable signal. The frequency is not a multiple of another, the difference between any two frequencies does not equal any of the frequencies, and the sum of any two frequencies does not equal any of the frequencies. The frequencies were initially designed with a ratio of 21/19, which is slightly less than a whole tone.

7 DTMF signaling decoder on wireless modules

☞ The feature is not supported by LEON-G200 series or by LEON-G100-07x and previous versions.

The u-blox wireless modules can be configured to perform DTMF detection on the RX speech channel. The DTMF decoder is part of the In-Band modem feature and it can be configured through the +UDTMFD AT command (for more details on the command description, refer to u-blox commands manual [1]).

7.1 Implementation

The DTMF decoder must be enabled through AT commands once per module power cycle and before any call set up, e.g.:

```
AT+UDTMFD=1,2
OK
```

At each call set up, the DTMF decoder is automatically enabled. During the call, the DTMF decoder provides URCs for each detected digit, e.g.:

```
+UUDTMFD: 4
```

the digit ‘4’ has been detected.

7.2 Performance criteria

Various standards bodies (ITU-T, ETSI, EIA/TIA), mobile network operators (NTT, AT&T) and other players in the communication industry (MITEL, Bellcore) have established different performance tests and criteria for DTMF decoders.

The u-blox decoder implementation has followed the ETSI specifications as described in the multi-part ETSI Standard ES 201 235 [6]. However, the AT interface allows the decoder configuration for the performance criteria customization on need.

There are two main performance indicators for DTMF detectors:

- **Detection performance** – is the ability to correctly decode the DTMF tones in various network conditions. Modern networks use compression which introduces distortions that may invalidate at detector input a correctly generated DTMF tone
- **Speech immunity** – is the DTMF talk-off abatement performance. Talk-off is the term that describes when a human voice is able to trigger DTMF tones during a telephone call. Talk-off occurs when the DTMF detector tries to translate sounds into DTMF tones causing false detections

The decoder performance is also characterized by the robustness towards digit repetitions (special case of false detection), caused by e.g. interruptions in the DTMF tones. The ETSI standard specifies that a detected digit shall be unaffected by disturbances having a duration of less than 20 ms. Nevertheless, such condition can be not sufficient to avoid false digit repetitions in case of networks characterized by high distortions or speech frame losses.

In some conditions, the overall performance may be improved by increasing the tone duration and the pause between tones (inter-digit interval); in other words, the performance is higher if there is the possibility to decrease the digit transmission rate and tune the detector accordingly.

In general, the higher the speech immunity, the higher the risk of missed detections. The right trade-off between detection performance and talk-off abatement performance depends on application.

7.2.1 Decoder configuration

At each module power cycle, the decoder is configured with factory-programmed values.

It can be reconfigured through AT command at any time, even run-time during a call, e.g.:

```
AT+UDTMFD=1,2,4,400,10,3
```

The decoder has six configuration parameters:

Parameter	Range	Default value	Description
<urc_en>	0: disable 1: enable	N.A.	Enables the URCs on a specific AT terminal. Mandatory parameter
<mode>	0: disabled 1: normal 2: robust	N.A.	DTMF feature enabling/disabling and activation mode definition. Mandatory parameter
<att_cfg>	0-15	4	Controls the accepted signal levels. The signal is scaled down by 24 dB at the detector input.
<threshold>	100-10000	400	Controls the accepted signal levels. The digit recognition starts when the output of the analysis filter bank reaches the value of 400
<immunity>	0-20	14	Calibrates the speech immunity strength.
<max_int>	1-255	2	Controls the false digit repetitions. The expected minimum pause between the digits is 40 ms. Maximum signal interruption is 20 ms.

Table 1: +UDTMD parameter description and factory-programmed values

The factory-programmed values could vary along different products or product versions.

☞ By default, it is suggested to activate the DTMF in robust mode.

☞ <att_cfg> and <threshold> default values are optimized for best performance in terms of signal level operating range, complying with the ETSI requirements.

7.2.1.1 Activation mode (<mode> parameter)

The detector can be activated in the normal and the so-called robust mode. The robust mode is characterized by a reduced risk of any kind of false detections. The robustness is achieved by analyzing the input signal in the time domain. In fact human voice, melodies and other signals, as well as speech codecs like AMR or other disturbances from the network that potentially cause false detections, are generally “touch tone – like generators” for very short time.

The DTMF detector in robust mode meets the ETSI expectations in terms of detection performance ETSI TR 126 975 [7].

Robust mode advantages: case study

Figure 4 shows a normalized signal as presented at the DTMF decoder input, corresponding to the digit sequence "1,2,3,4,5". The signal has been generated by key presses (approx. 500 ms) on a VoIP telephone connected with a u-blox wireless module. The short burst before every tone is actually the start of the tone itself that is interrupted by the network after ~ 25/30 ms and restored after ~ 100 ms.

The normal mode is affected by false digit repetitions. The short bursts are detected as an independent DTMF tone, thus the detector output is "1,1,2,2,3,3,4,4,5,5". In robust mode, the burst are rejected and digits correctly outputted.

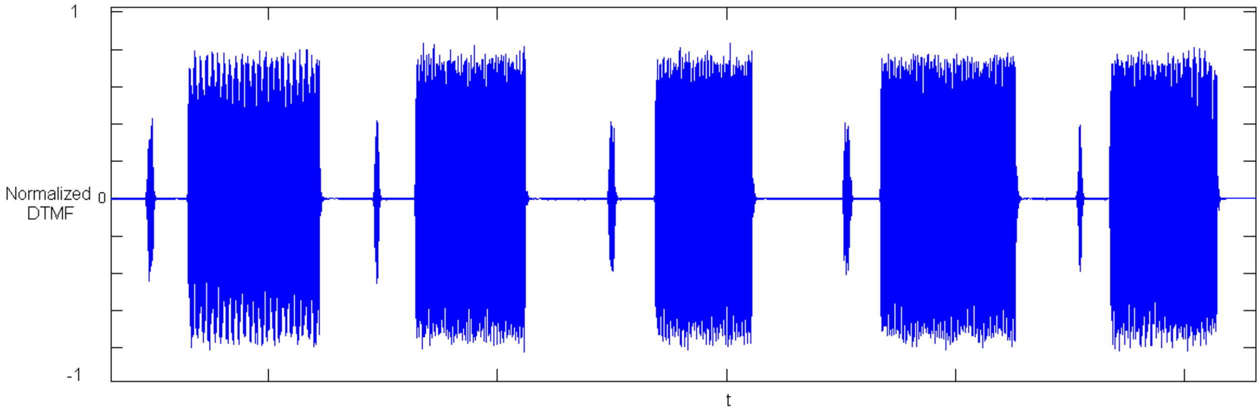

Figure 4: Normalized tones corresponding to digit presses '1','2','3','4','5', with interruptions caused by VoIP-based connection

7.2.1.2 Accepted signal levels (<att_cfg> and <threshold> parameter)

<att_cfg> applies an attenuation on the input signal in steps of 6 dB: 0 for 0 dB attenuation, 15 for 90 dB attenuation, which corresponds to mute signal at decoder input. In general, the parameter can be used to adapt the decoder to special network conditions (e.g. extremely high or extremely low tone levels). <att_cfg> should be configured as low as possible but avoiding overflows. There is an overflow protection mechanism that automatically scales down signals that lead the detector into algorithmic overflow. The automatic scaling is acknowledged through the following URC:

```
+UUDTMFDE: 1
```

and the new attenuation can be retrieved getting the last parameter of the read command, e.g.

```
AT+UDTMFD?
+UDTMFD: 1,2,4,100,14,2,5
OK
```

the attenuation has been increased from 4 to 5. The overflow protection mechanism only increases the attenuation. If overflow is notified, it is not guaranteed that the decoder performs in best condition and additional attenuation might be required. A detection result prior to an overflow notification shall be considered unreliable.

<threshold> is the current threshold applied on the signal level to be considered valid (i.e. to enter the operating condition). Higher thresholds give better performance especially in terms of speech immunity and false detections, with the cost of the increase of the lower boundary of the operating range. This parameter is not expressed in dB.

There is a relationship between <att_cfg> and <threshold>: the decoder performance at a specific signal level does not change if for each 6 dB attenuation increase the threshold is doubled (which corresponds to 6 dB increase, too).

In general, it should not be necessary to change the <att_cfg> and <threshold> parameters. Nevertheless, standards and operators may require slightly different operating conditions for DTMF. Generally, ranges of 20-30 dB are required, while the not-operating condition may vary from -29 to -40 dBm.

The default values are selected to operate according to the (in terms of operating conditions, most exacting) requirements from ETSI ES 201 235-4 [5], for more details refer to the Table 6.

Conditions with default values (in robust mode)	Valid	Not Valid
Signal level x (dBm0)	-36 <= x <= -3	x < -40

Table 6: Default values handling

The levels are expressed in decibels with respect to 0x7FFF clipping value.

7.2.1.3 Immunity/talk-off abatement (<immunity> parameter)

<immunity> calibrates the decoder respect to speech immunity performance: 0 for no immunity, 20 for maximum immunity performance. Unlike the robust mode, the talk-off abatement algorithm is based on the spectral analysis of the signal.

For certain end-to-end applications in which the talk-off abatement is not relevant (since voice or other disturbing signals are not injected in the voice channel), the speech immunity can be lowered or even completely disabled, having the advantage of an improved detection performance with e.g. low bit-rate codecs.

The default vale (14), combined with the robust mode, complies with the ETSI requirements for speech immunity ETSI ES 201 235-4 [5]. According to ETSI ES 201 235-3 [4], ETSI ES 201 235-4 [5], 'Table 2: Signal condition requirements, NOTE2', the talk-off performance is not directly specified as set of requirements for the existence or non-existence of signal conditions. The performance is indirectly specified through the speech immunity requirements of clause 4.2 in ETSI ES 201 235-3 [4] and ETSI ES 201 235-4 [5], 'Speech immunity performance'.

Table 7 provides the tests results with four different detector configurations respect to speech immunity (default values for the other parameters used). Talk-offs represent the number of false detections during the testing.

ETSI Speech Immunity Test 20 minutes test signal	<immunity>=0 normal mode	<immunity>=0 robust mode	<immunity>=14 normal mode	<immunity>=14 robust mode	ETSI reference
Talk-offs	9900	4800	100	5	5

Table 7: Tests results with four different detector configurations

7.2.1.4 False digit repetitions (<max_int> parameter)

Network conditions can generate more or less short interruptions of tones that may cause false detections – digit repetitions. ETSI requires that a decoder is unaffected by disturbances a duration of less than 20 ms, which may be not sufficient for network conditions.

The **<max_int>** parameter allows the tuning of the maximum interruption that a detected tone may have, such that is still interpreted as a single digit and thus avoiding false digit repetitions.

The <max_int> parameter also represents the minimum expected pause between two DTMF tones. Therefore if a decoder is configured to compensate interruptions up to e.g. 80 ms (<max_int>=4), the DTMF transmitter shall be configured to generate tones with a pause between them larger than 80 ms, otherwise the decoder recognizes two subsequent tones associated to the same digit as a single digit.

By default <max_int> is set to 40 ms, which is ETSI compliant according to receiver's digit recognition condition requirement in ETSI ES 201 235-4 [5], cit. "any tone shall be preceded by the continuous absence of a valid signal condition for more than 40 ms".

7.2.1.5 Not configurable signal condition and tolerances / default values

Table 8 reports the not configurable signal conditions. Signal conditions and tolerances comply with ETSI ES 201 235-3 [4] and ETSI ES 201 235-4 [5].

Signal conditions an tolerances	Valid	Not Valid
Frequency Deviation	<= ± (1,5% + 2) Hz	
Twist (signal level difference)	< 12 dB	
Reverse Twist (signal level difference)	< 12 dB	

Table 8: Not configurable factory-programmed signal conditions and tolerances on LEON modules

- Twist: the lower tones are higher in amplitude than the higher tones
- Reverse twist: the lower tones are lower in amplitude than the higher tones

7.3 DTMF performance measurements

DTMF performance is measured with respect to speech immunity and detection performance.

7.3.1 Speech immunity

The tests about speech immunity tests have been performed according to ETSI ES 201 235-4 [5], Paragraph 4.3 and Annex A and Annex B connecting the u-blox module with a network simulator using a full-rate speech codec. The test results presented in the Table 9 have been obtained with factory-programmed configuration values, only varying the <mode> and <immunity> parameters. The full speech immunity is reached if the DTMF detector has maximum 5 talk-offs (i.e. false detections caused by 20 minutes of speech-like test signal injected into the detector).

ETSI Speech Immunity Test 20 minutes test signal	<immunity>=0 normal mode	<immunity>=0 robust mode	<immunity>=14 normal mode	<immunity>=14 robust mode	ETSI reference [5]
Talk-offs	9900	4800	100	5	5

Table 9: Speech immunity tests

The factory-configured DTMF decoder activated in robust mode passes the speech immunity test.

7.3.2 Detection performance

Detection performance measurement and benchmarking has been performed with respect to performance results published by ETSI in ETSI TR 126 975 [7], Chapter 10 "Performances with DTMF tones", implementing the described test procedure on a sub-set of experiments. This ETSI document is not intended to be a DTMF decoder specification. Rather, it evaluates the transparency of the FR and AMR speech codecs to DTMF tones.

The benchmarking with the ETSI reference DTMF decoder is considered a valid performance measurement. It points out the problems that introduce the widely used speech codecs adopted by 2G and 3G wireless networks: it results that **the AMR low bit rate modes are not transparent to DTMF tones** (refer to ETSI TR 126 975 [7] and Table 11).

The tests have been performed with factory-programmed configuration values, only varying the <mode> and <immunity> parameter. Five different experiments from ETSI TR 126 975 [7] at various signals levels and with or without frequency deviation and reverse twist have been considered. Each experiment is made up of 20 repetitions of a sequence of 16 DTMF digits with tone 80 ms duration and 80 ms pause duration.

Table 10 and Table 11 illustrate the DTMF decoder performance with respect to two different speech codecs:

- FR GSM 13 kb/s codec
- AMR 4.75 kb/s codec

7.3.2.1 Full rate GSM 13 kb/s codec

Each element in the Table 1 reports the percentage of undetected digits and the percentage of false detections. For each x/y table element, x represents the percentage of undetected DTMF digits and y represents the percentage of out-of-sequence digits (false detections).

FR GSM 13 kb/s	<immunity>=0 normal mode	<immunity>=0 robust mode	<immunity>=14 normal mode	<immunity>=14 robust mode	ETSI reference
exp7: -6 dBm	0/30	0/0	0/0	0/0	**0/0**
exp8: -16 dBm	0/22	0/0	0/0	0/0	**0/0**
exp9: -26 dBm	0/8	0/0	0/0	0/0	**0/0**
exp10: -16 dBm+frequency deviation	0/8	0/0	0/0	0/0	**0/0**
exp11: -13 dBm with -6 dB (reverse) twist	0/16	0/0	0/0	0/0	**0/0**

Table 10: Results for each experiment (rows) for each decoder configuration (columns)

☞ 100% of detections is achieved with the factory-programmed detector in both normal and robust mode, without false detections.

False detections are present only in normal mode with completely disabled immunity (<immunity>=0). This configuration, which represents a configuration at boundary conditions, is not recommended.

7.3.2.2 AMR 4.75 kb/s codec

Each element in the Table 11 reports the percentage of undetected digits and the percentage of false detections. For each x/y table element, x represents the percentage of undetected DTMF digits and y represents the percentage of out-of-sequence digits (false detections)

AMR 4.75 kb/s codec	<immunity>=0 normal mode	<immunity>=0 robust mode	<immunity>=14 normal mode	<immunity>=14 robust mode	ETSI reference
exp7: -6 dBm	0/24.0	0/0	4.7/0	20.9/0	**21.3/0**
exp8: -16 dBm	0/7.8	0/0	1.6/0	22.1/0	**24.8/0**
exp9: -26 dBm	0/0.6	0/0	1.8/0	19.7/0	**27.5/0**
exp10: -16 dBm+frequency deviation	0/12.5	0.3/0	1.6/0	19.0/0	**26.9/0**

AMR 4.75 kb/s codec	<immunity>=0 normal mode	<immunity>=0 robust mode	<immunity>=14 normal mode	<immunity>=14 robust mode	ETSI reference
exp11: -13 dBm with -6 dB (reverse) twist	0/8.1	16.6/0	11.8/0	34.7/0	**35.9/0**

Table 11: Results for each experiment (rows) for each decoder configuration (columns)

☞ Better detection performance than the ETSI reference are achieved with the factory-programmed detector in robust mode (23.28% vs 27.28% in average).

☞ Almost 100% of detections is achieved if the immunity in robust mode is disabled.

☞ exp11 with artificially added negative twist represents unreal/rare network situations, refer to chapter 7.3.2.3.

7.3.2.3 Discussion

ETSI-compliant detector

The DTMF decoder in robust mode and default setting (<immunity> = 14) performs as expected by ETSI requirements, both with respect to speech immunity and detection performance.

AMR transparency

ETSI verified that low-bit rate codecs, in particular the AMR 4.75 kb/s codec, are not transparent to DTMF tones, especially the shorter ones, if an ETSI-compliant decoder is used. For instance, the AMR codecs have a tendency to add negative twist to DTMF signals. This is revealed by results of experiment exp11, in which an additional negative twist of 6 dB has been artificially added to DTMF tones prior AMR encoding. The DTMF factory setting for twist valid condition has been relaxed from the minimum recommended of 6 dB in ETSI ES 201 235-4 [5] to 12 dB: for more details refer to chapter 7.2.1.5. Nevertheless, the exp11 signal conditions can be considered really boundary conditions, which are rare in real network situations.

Immunity configuration

With a cost of a reduced speech immunity performance, the u-blox DTMF decoder can be tuned to be more or less transparent to speech codec modes, acting on the <immunity> parameter. In particular, with the disabled immunity (<immunity>=0), it can cope with distortions introduced by the AMR 4.75 kb/s codec maintaining a detection performance close to 100%. The reduced immunity performance can be acceptable in controlled conditions of talk-off sources. A typical application which does not need speech immunity performance is the terminal end-to-end signaling, in which the microphone at DTMF generator side is disabled.

Normal mode

The normal mode combined with a proper level of immunity can give the right balance between the detection performance and speech immunity performance. See for example detection results with AMR codec, <immunity>=14 in normal mode detection, close to 98% hits without false detections, 100 talk-offs.

Tone duration

It is a recommendation for the transmitter. For end-to-end signaling, especially with low-bit-rate codecs, a minimum of 80 ms for tone duration is recommended. There are generally no benefits in having tones lasting more than 120 ms (on the contrary, the risk of false digit repetitions is increased).

Pause duration and <max_int>

It is a recommendation for the transmitter. ETSI recommends that if the DTMF signaling pause duration is controlled automatically by the transmitter, the duration of the pause between any individual DTMF tone combination shall not be less than 65 ms. On need, the <max_int> can be configured accordingly to the transmitter's configuration, as proposed in the chapter 7.2.1.4.

Half-Rate (HR) codecs

The half-rate codecs may dramatically worsen the decoder performance. As stated by ETR 229 [8], "a serious commercial application using DTMF in the speech channel should not be supported with the GSM half rate codec.". This statement is valid for any codec working on half-rate channels, like for instance the HR-AMR (Half-Rate AMR). The half-rate speech channels are not only characterized by the distortions of low-bit rate codecs, but also by a higher error rate since the actual payload data rate is halved with respect to the full rate channel (for example, 6.5 kb/s vs 13kb/s).

☞ The module can be configured to not perform calls on half-rate channels through AT+UDCONF command (refer to chapter 7.4).

7.4 Configuration examples

Performance estimates of the following configuration examples are given for error free conditions (no speech frame drops). Frame drops may cause false digit repetitions that can be coped with <max_int> parameter configuration as discussed in the chapter 7.2.1.4.

7.4.1 ETSI-compliant decoder

It is achieved by the decoder enabled in robust mode with factory-programmed parameters:

```
AT+UDTMFD=1,2
OK
AT+UDTMFD?
+UDTMFD: 1,2,4,400,14,2,4
OK
```

Characterized by full speech immunity, the detection rate of this configuration can be less than 100% with low bit-rate codecs, as presented in the chapter 7.3.

7.4.2 ETSI-compliant decoder with guaranteed speech channel QoS

To get rid of low-bit rate codecs distortions, the module can be configured to support and make calls only with a reduced speech codec set. The speech codec is configured through +UDCONF AT command; for more details refer to u-blox AT commands manual [1].

7.4.2.1 EFR, FR codec set restriction

For example, the ETSI-compliant decoder working with Enhanced Full Rate (EFR) and Full Rate (FR) codecs only guarantees a 100% detection performance with full speech immunity:

```
AT+UDCONF=30,6
OK
AT+UDTMFD=1,2
```

```
OK
```

7.4.2.2 Full-rate channel restriction

It is possible to restrict the channels only excluding half-rate channels.

```
AT+UDCONF=30,7
OK
```

configures the module to use FR, EFR and FR-AMR codecs.

☞ For the usage of +UDCONF command for speech codec configuration, refer to the u-blox AT commands manual [1].

7.4.3 Custom DTMF detectors for low quality speech channels

A DTMF decoder can be configured to provide good performance also with low bit-rate codecs, at a cost of lower speech immunity or restricted operating range. In both cases the transmitter shall work in controlled condition.

The following configurations are guideline and need actual on-field tuning and validation.

7.4.3.1 Reduced speech immunity

A good detection performance with low bit-rate codecs can be reached just turning on the decoder in normal mode, keeping the factory-programmed parameters:

```
AT+UDTMFD=1,1
OK
AT+UDTMFD?
+UDTMFD: 1,1,4,400,14,2,4
OK
```

According to performance measurements, talk-offs increase statistically from 5 to 100, while the detection rate increases from approx 75% to 98% with the worst AMR codec case (4.75 kb/s). Varying the immunity parameter, the balance between talk-offs and detection rate can be differently distributed.

If the talk-off performance is not an issue, the immunity can be completely disabled and robust mode turned on (to avoid false detections).

```
AT+UDTMFD=1,2,,,0,2
OK
AT+UDTMFD?
+UDTMFD: 1,1,4,400,0,2,4
OK
```

The detection rate on worst AMR case should now be close to 100%.

7.4.3.2 Reduced operating range

The reducing of the operating range improve the speech immunity and in general performance vs false detection, e.g.:

```
AT+UDTMFD=1,1,,1200
OK
AT+UDTMFD?
+UDTMFD: 1,1,4,3200,14,2,4
OK
```

is a configuration that should have an improved talk-off with respect to chapter 7.4.3.1 at the cost of reduction of a minimum operating level shift from approx -36 dBm to -18 dBm.

Appendix

A List of Acronyms

Abbreviation / Term	Explanation / Definition
AC	Adaptation Control block
AGC	Automatic Gain Control
AMR	Adaptive Multi Rate
AT	AT Command Interpreter Software Subsystem, or attention
DBF	Downlink Biquad Filters command
DRC	Digit Recognition Condition
DSP	Digital Signal Processing
DTMF	Dual-Tone Multi-Frequency
EC	Echo Canceller algorithm
EEP	Electrically Erasable and Programmable ROM
EFR	Enhanced Full Rate
FIR	Finite Impulse Response filter
FR	Full Rate
HF	Hands-free Algorithm
HR	Half Rate
LEM	Loudspeaker-Enclosure-Microphone
LMS	Least Mean Square
MGC	Microphone Gain control command
NLMS	Normalized Least Mean Square
NR	Noise Reduction algorithm
PSTN	Public Switched Telephone Network
RX	Receiver
SGC	Speaker Gain control command
TX	Transmitter
UBF	Uplink Biquad Filters command
URC	Unsolicited Response Code

Related documents

[1] u-blox AT Commands Manual, Docu No WLS-SW-11000
[2] LEON-G100 / LEON-G200 System Integration Manual, Docu No GSM.G1-HW-09002
[3] ITU-T Recommendation Q.23: Technical features of push-button telephone sets
[4] ETSI ES 201 235-3 V1.3.1, Specification of Dual Tone Multi-Frequency (DTMF) Transmitters and Receivers; Part 3: Receivers
[5] ETSI ES 201 235-4 V1.3.1, Specification of Dual Tone Multi-Frequency (DTMF) Transmitters and Receivers; Part 4: Receivers for use in Terminal Equipment for end-to-end signaling
[6] Work Items with ETSI Document Number of '201 235', see Work Programme search database, http://www.etsi.org/
[7] ETSI TR 126 975 V10.0.0 (2011-04), Performance characterization of the Adaptive Multi-Rate (AMR) speech codec (also 3GPP TR 26.975 version 10.0.0 Release 10)
[8] ETR 229: October 1995 (GSM 06.08 version 4.0.0), Performance characterization of the GSM half rate speech codec

[1] and [2] documents are available on our homepage (http://www.u-blox.com).

☞ For regular updates to u-blox documentation and to receive product change notifications, register on our homepage.

Revision history

Revision	Date	Name	Status / Comments
-	Mar. 12, 2010	ague	Initial release
1	May. 11, 2012	ague	Added chapter 5 - Hands-free device examples Updated document number for AT commands manual
2	Jan. 04, 2013	vpah	Added DTMF decoder description

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