WIRELESS

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LISA-U1 / LISA-U2 series

Audio

Application Note

Abstract

This document provides information and procedures to resolve audio related problems with LISA-U1 / LISA-U2 series series modules.
In particular, it describes the procedures for tuning the hands-free algorithm (echo cancellation, automatic gain control, noise reduction) and the external audio management.

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Contents

1 Introduction

This document provides information and procedures for the resolution of the potential audio related problems on LISA-U1 / LISA-U2 series modules. It also addresses the DTMF signaling decoder functionality available via the +UDTMFD AT command, implemented following the multi-part ETSI Standard ES 201 235 [5].

As an example, the document describes a procedure for tuning the hands-free algorithm (Echo cancellation, Automatic Gain Control, Noise Reduction) and the management of an external codec.

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

For a detailed description of LISA-U1 / LISA-U2 series module audio interface, refer to the LISA-U series 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.**

1.1 Scope

☞ This document applies to the following products:

- LISA-U1 series
- LISA-U2 series

2 Introduction to HF algorithm tuning

After connecting external audio devices (i.e. microphone and loudspeaker) to the wireless module, the far-end user might hear an acoustic echo. This problem typically occurs when the device gain is set high to work at a distance (i.e. in hands-free application). LISA-U1 / LISA-U2 modules provide a hands-free algorithm to remove the echo. The HF parameters control the algorithm within the uplink audio path in use (refer to AT+USPM and AT+UHFP command in u-blox AT Commands Manual [1]), stored in the NVM dynamic parameters profile (refer to section 3.1 and AT&W command description in u-blox AT Commands Manual [1]).

Section 3 presents a step-by-step procedure for choosing parameters to remove the 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 described in the section 4.

Section 2.1 describes the HF algorithm and parameters meaning.

2.1 HF algorithm description

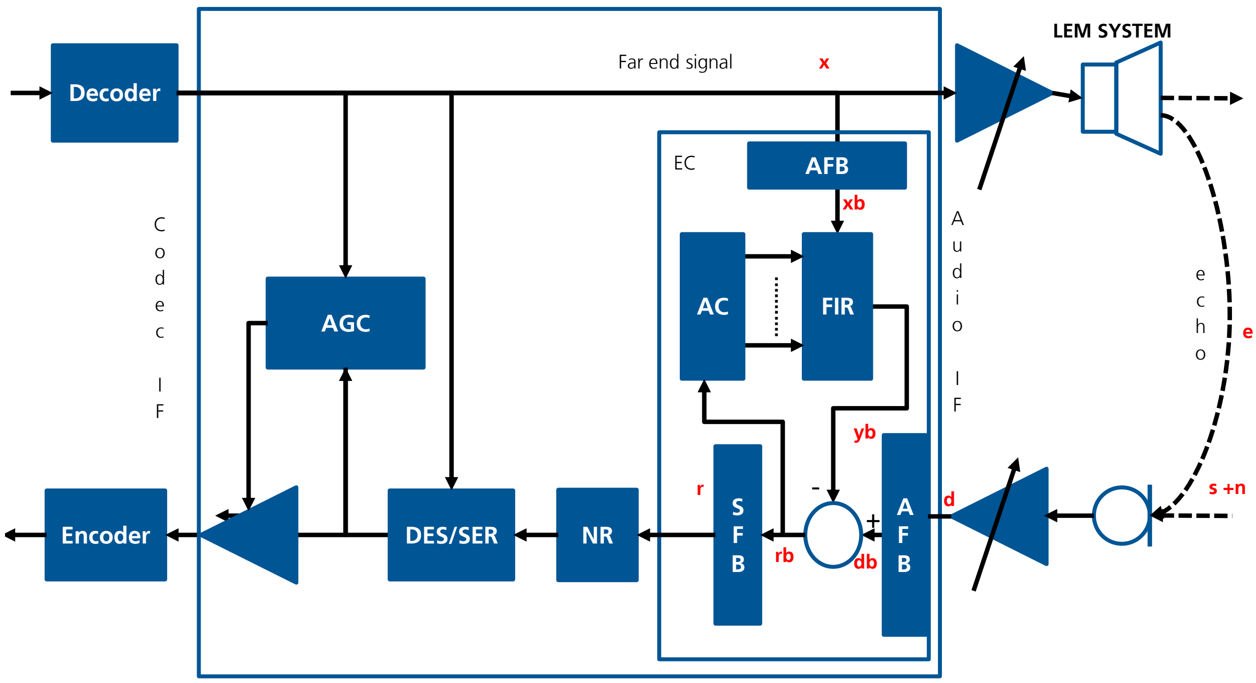

Figure 1: Block diagram for hands-free

The LEM system (Loudspeaker-Enclosure-Microphone) is a non-linear, time varying system. The microphone signal (d) is composed of near-end speech (s), echo (e) of the far end signal(x) and noise (n).

The echo canceller (EC) splits the downlink (x) and microphone (d) signal into sub-bands signals (xb, db) by 2 Analysis Filter Bank (AFB). For each sub-band a FIR filter emulates the LEM system behavior for the corresponding sub-band and generates an estimate (yb) of the acoustic echo produced by the signal (xb) on the LEM in that sub-band. For each sub-band the estimated echo (yb) is subtracted from the microphone sub-band signal (db).The residual echo (rb) of each sub-band is re-combined by a Synthesis Filter Bank (SFB) to a single full spectrum residual echo signal (r).

Since the LEM is time varying, the Adaptation Control block (AC) adapts the coefficients of each FIR filters; AC is a block NLMS adaptive algorithm based on the residual echo (rb) 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. Full spectrum residual echo and noise (r), are then lowered by Noise Reduction (NR), Dynamic Echo Suppressor (DES, LISA-U1 series only) or Spectral Echo Reduction (SER, LISA-U2 series only) and Automatic Gain Control (AGC). AGC is disabled when in double talk.

2.2 HF algorithm parameters

The HF parameters of the +UHFP AT command sets the corresponding algorithm parameters: for the command description refer to the u-blox AT Commands Manual [1].

2.2.1 Parameters for block activation and initialization

HF_ALGORITHM_INIT Range 0x0000, 0x07FF

The audio driver uses this parameter to initialize the algorithm when a call starts. This parameter is a set of flags that control the activity and initialization of the EC, AGC and NR blocks.

Flag	LISA-U1 series	LISA-U2 series
Bit #0 set	Echo Cancellation (EC) initialization	Unused
Bit #1 set	EC restart (without coefficient initialization)	Unused
Bit #2 set	EC on	Echo Cancellation (EC) initialization and on
Bit #3 set	Unused	Unused
Bit #4 set	Noise Reduction initialization	Unused
Bit #5 set	Noise Reduction on	Noise Reduction initialization and on
Bit #6 set	Unused	Unused
Bit #7 set	Automatic Gain Control (AGC) initialization	Unused
Bit #8 set	AGC on	Automatic Gain Control (AGC) initialization and on
Bit #9 set	Dynamic Echo Suppression (DES) INIT	Unused
Bit #10 set	Dynamic Echo Suppression ACTIVE	Spectral Echo Reduction (SER) initialization and on

Table 1: HF_ALGORITHM_INIT flags explanation

Examples:

Configuration	Command	Remarks
EC only	`AT+UHFP=0,0x0004,,,,,,,0,0,500,8192,7500,7500,2,100,100,100,60,60,60`	
NR only	`AT+UHFP=0,0x0020,,,,,,,0,0,500,8192,7500,7500,2,100,100,100,60,60,60`	
AGC only	`AT+UHFP=0,0x0100,,,,,,,0,0,500,8192,7500,7500,2,100,100,100,60,60,60`	
EC+AGC+NR	`AT+UHFP=0,0x0124,,,,,,,0,0,500,8192,7500,7500,2,100,100,100,60,60,60`	
All off	`AT+UHFP=0,0x0000,,,,,,,0,0,500,8192,7500,7500`	Optional parameters can be omitted

Table 2: HF_ALGORITHM_INIT examples

2.2.2 Unavailable parameters

The parameters presented in this section are not used. They are solely maintained in the command for backwards compatibility with LEON-G100 / LEON-G200 series modules.

In the response to the test command (AT+UHFP?) their value is shown as 'NA' (Not Available).

In the set command, they can be omitted, e.g.:

AT+UHFP=<uplink_path_num>,<hf_algorithm_init>,,,,,,,<add_atten>,<min_atten>,<max_atten>,<nr_sw_2>,<nr_u_fak_0>, <nr_u_fak>

If they are not omitted, the value is checked to be within the allowed range.

Parameter	Range	Applicability in LISA module
HF_ALGORITHM_RESTART	0x0000 to 0x07FF	Not Used
STEP_WIDTH	0 to 32767	Not Used
LMS_LENGTH	2 to 400	Not Used
LMS_OFFSET	0 to 400	Not Used
BLOCK_LENGTH	2, 4, 5 and 8	Not Used
RXTX_RELATION	-960 to 960	Not Used

Table 3: Not available parameters range

2.2.3 Parameters for automatic gain control (AGC)

Parameter	Range	Default value	Applicability in LISA module
ADD_ATTEN	-960 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 the calculated attenuation is lower than MIN_ATTEN, then the attenuation is increased to MIN_ATTEN.
MAX_ATTEN	0 to 960	500	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 4: 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 (NR)

The noise reduction operates on the eight 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 apply (attenuation).

Parameter	Range	Default value	Remarks
NR_SW_2	0 to 32767	8192	This is the maximum attenuation that the NR can introduce. 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 the 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	7500	This is the weighting factor for frequency band 0 (0 Hz-250 Hz). Increasing this factor will cause a better noise reduction in this band but also higher distortion of speech. Linear; weighting factor =NR_U_FAK_0 / 32768

Parameter	Range	Default value	Remarks
NR_U_FAK	0 to 16384	7500	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 will cause a better noise reduction in this band but also higher speech distortion. Linear; weighting factor =NR_U_FAK_0 / 32768

Table 5: 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 6: Noise reduction parameters examples

2.2.5 Parameters for echo cancellation

The echo cancellation is a sub band (SB) design, where identical systems on each sub-band perform adaptive linear filtering and the subsequent echo subtraction (see section 2.1). After the echo subtraction the sub-band signals are combined back to a single full spectrum signal. EC uses three sub-bands when the speech channel uses Narrow Band codec, 6 sub-bands with Wide Band codec.

Parameter	Range	Default value	Remarks
EC_BLOCK_LENGTH	1, 2, 4, 5, 8	2	LMS coefficient adaptation block length. It specifies the number of frames during which the adaptive filter coefficients are updated in the AC blocks. It can take only the values 1,2,4,5 and 8 where 1 indicates updating of filter parameters at every frame (160 samples or 20 ms for narrow band), 4 represent updating every four frames. The higher this number, the slower but more accurate the adaptation converges.
EC_NR_COEFF_REAL	2 to 2000	100	Number of filter coefficients in the sub-band EC, for real sub band (in Narrow Band mode: 0-0.8 kHz in Wide Band mode: 0-0.73 kHz).
EC_NR_COEFF_COMPLEX1	1 to 1000	100	Number of filter coefficients in the sub-band EC, for complex sub band 1 (in Narrow Band mode: 0.8-2.4 kHz; in Wide Band mode: 0.73 -2.18 kHz)
EC_NR_COEFF_COMPLEX2	1 to 1000	100	Number of filter coefficients in the sub-band EC, for complex sub band 2 (in Narrow Band mode: 2.4- 4 kHz; in Wide Band mode: 2.18 -3.64 kHz)
EC_NR_COEFF_COMPLEX3	1 to 1000	60	Number of filter coefficients in the sub-band EC, for complex sub band 3 (in Narrow Band mode: Ignored; in Wide Band mode: 3.64 - 5.09 kHz)
EC_NR_COEFF_COMPLEX4	1 to 1000	60	Number of filter coefficients in the sub-band EC, for complex sub band 4 (in Narrow Band mode: Ignored; in Wide Band mode: 5.09 - 6.56 kHz)
EC_NR_COEFF_COMPLEX5	1 to 1000	60	Number of filter coefficients in the sub-band EC, for complex sub band 5 (in Narrow Band mode: Ignored; in Wide Band mode: 6.56 - 8 kHz)

Table 7: Echo Cancellation parameters description

Limit for the sub-band Echo canceller parameters:
<ec_nr_coeff_real>+2*(<ec_nr_coeff_complex1>+<ec_nr_coeff_complex2>+<ec_nr_coeff_complex3>+< ec_nr_coeff_complex4>+ <ec_nr_coeff_complex5>) <2000

3 Procedure for echo canceller tuning

This is a step-by-step procedure to tune the audio path parameters for the removal of the echo heard on the far-end side.

Refer to u-blox AT Commands Manual [1] for more details on the AT commands and their parameters that are used in the tuning procedure (AT+USPM, 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 Turn off all the hands-free algorithm by the AT command:

AT+UHFP=<uplink_path_num>,0x0000,,,,,,,0,0,500,8192,7500,7500,1,100,100,100,60,60,60

2 In case of an hands-free device implementation, set the sidetone to 0 to avoid Larsen effect through the AT command:

AT+USTN=<downlink_path_num>,0

3 Regulate the gain on speaker and microphone used so that speech is not distorted on both uplink and downlink. Tune the gain on the downlink path using AT+USGC command if the speech signal on the speaker is distorted. Tune the gain on the uplink path using AT+UMGC command, if the speech signal from the microphone is distorted.

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.

4 Begin tuning the EC parameters, starting e.g.: with the AT command:

AT+UHFP=<uplink_path_num>,0x0004,,,,,,,0,0,500,8192,7500,7500,1,100,100,100,60,60,60

Parameter	Value	Meaning
<hf_algorithm_init>	0x0004	Only echo cancellation initialization and on
<ec_block_length>	1	Start updating adaptive filter coefficients every frame. This is the quicker convergence time of the coefficients (1-2 s)
<ec_nr_coeff_real>	100	
<ec_nr_coeff_complex1>	100	
<ec_nr_coeff_complex2>	100	
<ec_nr_coeff_complex3>	60	
<ec_nr_coeff_complex4>	60	
<ec_nr_coeff_complex5>	60	

- Change one <ec_nr_coeff_*> parameter at a time and measure resulting echo
- Repeat the previous step to converge at a value that results in minimum echo. Perform these steps for all the <ec_nr_coeff_*> parameters.
- Parameter <uplink_path_num> in this example as well in the following is the index of the uplink path in use. Check the uplink path in use by command:

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

Use <main_uplink> for <uplink_path_num> in all the examples of this procedure.

- The EC parameters are optional. If omitted, the default values 2,100,100,100,60,60,60 are set.
  E.g: AT+UHFP=<uplink_path_num>,0x0004,,,,,,,0,0,500,8192,7500,7500
  sets default EC parameters
- Since AGC and NR are off (only echo cancellation initialized and on), any values in the allowed ranges of the AGC and NR parameters are accepted but not used; only EC parameters are considered. Since the algorithm is adaptive, some seconds are needed to converge after that AT+UHFP command is issued. After the algorithm convergence, a residual echo remains. Try to change EC parameters till there is no residual echo. In very critical case, if the echo never disappears, try to find a minimum residual echo configuration.
- Use higher values of <ec_block_length> for more stable (but slower) convergence.
  E.g.: <ec_block_length>=1 convergence time is 1-2 s
  <ec_block_length>=4 convergence time is 4-6 s
- Use higher values of <ec_nr_coeff_real> and <ec_nr_coeff_complex*> for a long reverberation time.
- Reconsider points 3 to 4, if this command has no effect on the echo
- Parameters <ec_nr_coeff_complex3>, <ec_nr_coeff_complex4>, <ec_nr_coeff_complex5> are used only in WB speech. Test the EC performance both in NB and WB scenarios.
- If EC correctly works, a difference should be heard turning off the EC with the following AT command:
  AT+UHFP=<uplink_path_num>,0x0000,,,,,,,0,0,500,8192,7500,7500

5 Add the Dynamic Echo Suppression (LISA-U1 series only) or Spectral Echo Reduction (LISA-U2 series only) algorithm to remove a residual echo, if present:

AT+UHFP=<uplink_path_num>,0x0404,,,,,,,0,0,500,8192,7500,7500,<ec_nr_coeff_real>,<ec_nr_coeff_complex1>,<ec_nr_coeff_complex2>,<ec_nr_coeff_complex3>,<ec_nr_coeff_complex4>,<ec_nr_coeff_complex5>

Parameter	**Value**	**Meaning**
<hf_algorithm_init>	0x0404	Echo cancellation initialization and on; Spectral echo Reduction initialization and on
<ec_nr_coeff_real>		As in step 4
<ec_nr_coeff_complex*>		As in step 4

6 Add the AGC algorithm to remove a minimal residual echo, if present:

AT+UHFP=<uplink_path_num>,0x0104,,,,,,,0,0,500,8192,7500,7500,<ec_nr_coeff_real>,<ec_nr_coeff_complex1>,<ec_nr_coeff_complex2>,<ec_nr_coeff_complex3>,<ec_nr_coeff_complex4>,<ec_nr_coeff_complex5>

Parameter	Value	Meaning
<hf_algorithm_init>	0x0104	Echo cancellation initialization and on; AGC initialization and on
<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
<ec_nr_coeff_real>		As fond in step 4
<ec_nr_coeff_complex1>		As fond in step 4
<ec_nr_coeff_complex2>		As fond in step 4
<ec_nr_coeff_complex3>		As fond in step 4
<ec_nr_coeff_complex4>		As fond in step 4
<ec_nr_coeff_complex5>		As fond in step 4

It is possible to also add spectral echo reduction initialization e.g. <hf_algorithm_init>=0x0504 (Echo cancellation initialization and on; Spectral Echo Reduction initialization and on; AGC initialization and on).

If residual echo is still present, try to use higher <min_atten> values.

AT+UHFP=<uplink_path_num>,0x0104,,,,,,,50,100,500,8192,7500,7500,<ec_nr_coeff_real>,<ec_nr_coeff_complex1>,<ec_nr_coeff_complex2>,<ec_nr_coeff_complex3>,<ec_nr_coeff_complex4>,<ec_nr_coeff_complex5>

Parameter	Value	Meaning
<add_atten>	50	50*3/32 = 4.7 dB additional AGC attenuation
<min_atten>	100	100*3/32 = 9.4 dB minimum AGC attenuation

Example of AGC settings:

AGC	<add_atten>	<min_atten>	<max_atten>
Weak AGC	0	0	200
Moderate AGC	100	100	500
Strong AGC	200	200	500
No-duplex AGC	500	500	500

Using strong AGC can decrease the performance in double talk scenario and lead to a no-duplex configuration (see dedicated section below)

7 Add the NR algorithm to remove a residual noise on the uplink path (if present):

AT+UHFP=<uplink_path_num>,0x0124,,,,,,,<add_atten>,<min_atten>,<max_atten>,8192,7500,7500,<ec_nr_coeff_real>,<ec_nr_coeff_complex1>,<ec_nr_coeff_complex2>,<ec_nr_coeff_complex3>,<ec_nr_coeff_complex4>,<ec_nr_coeff_complex5>

Parameter	Value	Meaning
EC parameters		As found in step 4, 5 and 6
AGC parameters		
<hf_algorithm_init>	0x0124	Echo cancellation initialization and on; NR initialization and on; AGC initialization and on
<nr_sw_2>	4096	20 log (4096/32767) = -18 dB minimum NR attenuation
<nr_u_fak_0>	7500	7500/32768 = 0.23 weighting factor for frequency band 0
<nr_u_fak>	7500	7500/32768 = 0.23 weighting factor for frequency band 1-7

8 As with the previous step, the NR parameters can be changed until the minimum residual noise remains on the uplink path. To appreciate the NR effect, listen to the uplink speech in silence scenario (neither uplink nor downlink speech, nor echo present) and switch only the NR off/on by these commands:

AT+UHFP=<uplink_path_num>,0x0000,,,,,,,0,0,500,<nr_sw_2>,<nr_u_fak_0>,<nr_u_fak>
AT+UHFP=<uplink_path_num>,0x0020,,,,,,,0,0,500,<nr_sw_2>,<nr_u_fak_0>,<nr_u_fak>

9 Finally, the parameters found can be stored in NVM dynamic parameters profile using the AT&W command (refer to the section 3.1).

3.1 Storing the parameters in the profile

☞ Follow this procedure to save the parameters in the NVM dynamic parameters:

1 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]).
2 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).
3 Save the run-time configuration to Flash memory with the AT+CPWROFF command.
4 Reboot / PWR_ON reset of the device.

4 No-duplex configuration

The no-duplex set-up is recommended for HF systems with high echo coupling and high non-linearity on loudspeaker which make the EC cancellation ineffective.

☞ The no-duplex configuration particularly makes use of the AGC only as muting/un-muting device of the TX path.

The operating conditions of this no-duplex configuration are:

- Far-end user speaking: Tx muted by AGC, high and constant attenuation
- Far-end user silent: Tx un-muted (AGC off), both when near-end user is speaking or silent

4.1 Procedure

1 Configure AGC to high attenuation performance, e.g.:

LISA-U1 series:

AT+UHFP=<uplink_path_num>,0x0104,,,,,,,500,500,500,8192,7500,7500	or
AT+UHFP=<uplink_path_num>,0x0104,,,,,,,960,960,960,8192,7500,7500	

LISA-U2 series:

AT+UHFP=<uplink_path_num>,0x0100,,,,,,,500,500,500,8192,7500,7500	or
AT+UHFP=<uplink_path_num>,0x0100,,,,,,,960,960,960,8192,7500,7500	

Parameter	Value	Meaning
<hf_algorithm_init>	0x0100	Only AGC enabled
<hf_algorithm_init>	0x0104	AGC and AEC enabled
<add_atten>,<min_atten>,<max_atten>	500	500*0.05 = 25 dB add, min, max AGC attenuation
<add_atten>,<min_atten>,<max_atten>	960	960*0.05 = 48 dB add, min, max AGC attenuation

5 External codec management

LISA-U1 and LISA-U2 series modules provide an I^2S digital audio interface to connect an external audio device, e.g. a codec. The Application Processor (AP) should manage the codec.

This section includes an example of architecture for the module / external codec / AP system. In the block diagrams the HW implementation is highly simplified; for more detailed examples of HW implementation, refer to LISA-U series System Integration Manual [2].

For more details about the AT commands used in the examples below, refer to u-blox AT Commands Manual [1].

5.1 LISA-U1 series

Figure 2 shows a possible architecture for the LISA-U1 series module / external codec / AP system.

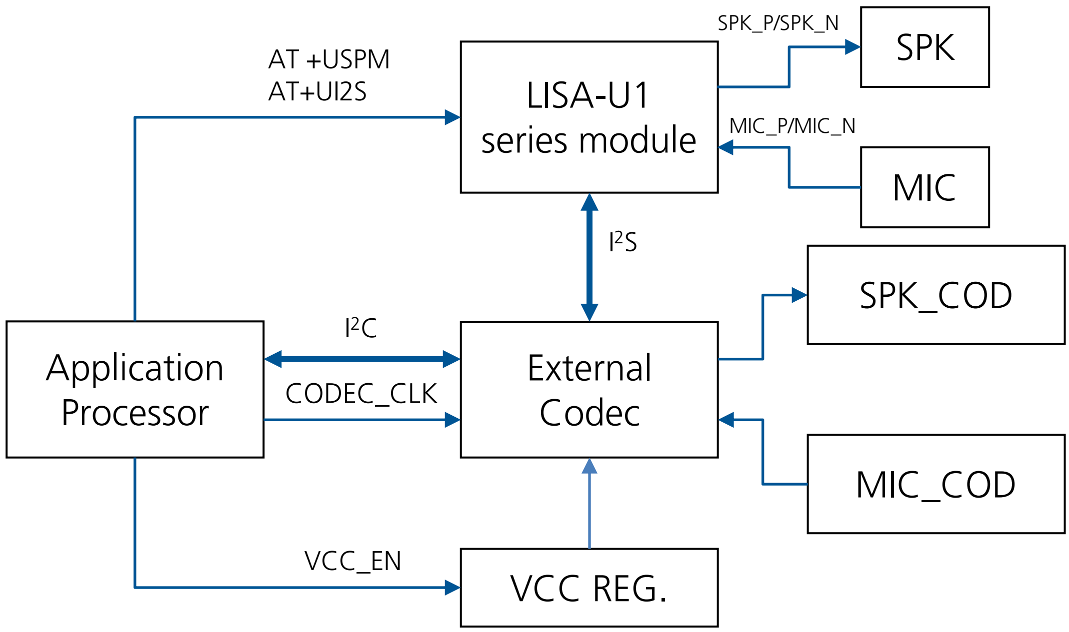

Figure 2: External codec management for LISA-U1 series module

The external codec can be connected to LISA-U1 series module by an I^2S interface. The codec allows the connection of the external microphone and speaker and acts as an A/D and D/A converter.

Generally the external codec needs to be managed, providing it of:

- A power supply provided by an external regulator with possibility to be enabled/disabled by an enabling signal (VCC_EN)
- A master clock signal provided by an external generator (CODEC_CLK signal, e.g.: 13 MHz)
- Some control commands to:
    - set the gains for the codec's amplifiers
    - enable/disable the codec
    - configure the I^2S on the external codec side

  A dedicated interface (e.g.: I^2C) is provided to supports these controls

In the LISA-U1 series module all the above activities to control the codec must be managed by the AP.

Furthermore, the AP should also:

- Configure the I²S on the module side for the compatibility with the I²S settings on the codec side. This can be done by I²S interface that is configurable by AT+UI2S command
- Force the module to route the audio signal toward the codec via I²S interface. The routing of audio signal (e.g.: switching from analog to digital path) is provided by the AT+USPM command

  E.g.: `AT+USPM=2,4,0,0`

5.2 LISA-U2 series

The block diagram for LISA-U1 series also applies to LISA-U2 series. LISA-U2 series modules support additional resources to manage the external codec:

- Master Clock Control AT command +UMCLK: this command provides the codec with a 13/26 MHz clock generated from the module.
- I²S control command +UI2S extension: new connection modes supported by <I2S_port> parameter.
- I²C control commands (+UI2CO, +UI2CW, +UI2CR, +UI2CREGR, +UI2CC). These commands allow sending commands from the module to the codec through a standard I²C interface.
- The AP can monitor the audio activity on the module enabling the +CIEV URCs. The +CIEV URCs on the AT terminal can be generated when the audio activities are running on the module, issuing the command AT+CMER=1,0,0,2,1.

Figure 3 shows a possible architecture for the LISA-U2 series module / external codec / AP system.

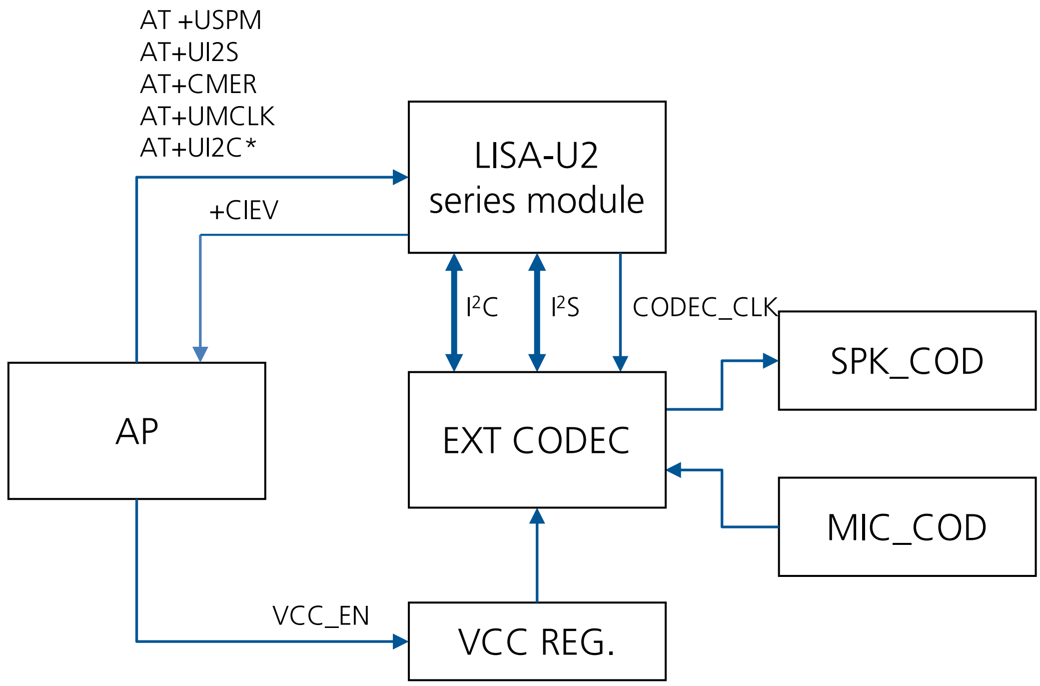

Figure 3: External codec management for LISA-U2 series module

The section 5.2.1 shows an example of the codec management scenario based on this architecture.

The examples proposed are for a Maxim MAX9860 audio voice codec connected to a LISA-U2 series module through the I²S interface as in the EVK-U20 /EVK-U23 evaluation board.

5.2.1 Scenario

- At system start-up, the AP should enable the codec supply by VCC_EN
- Configure the module and the codec with a sequence of AT commands

Command	Response	Description
`AT+USPM=5,5,0,0,2`	OK	AT+USPM=<main_uplink>,<main_downlink>,<alert_sound>,<headset_indication>,<vmic_ctrl> • <main_uplink>=5: Uplink path 5 via I2S1 (5 for e.g. any path via I2S1 is ok) • <main_downlink>=5: downlink path 5 via I2S1 (5 for e.g.; any path via I2S1 is ok) Use audio paths via I2S1. This allows setting of the I^{2}S properties (I^{2}S properties can't be changed while I^{2}S is in use, i.e. the paths are via I^{2}S).
`AT+UI2S=1,1,0,3`	OK	AT+UI2S=<I2S_mode>,<I2S_port>,<I2S_clk_wa>,<I2S_sample_rate>,<I2S_Master_Slave> • <I2S_mode>=1: PCM mode • <I2S_port>=1: Connect I^{2}S to I2Sx connection point • <I2S_clk_wa>=0: Dynamic mode (I2S_CLK and I2S_WA outputs are active and running only while audio path is active) • <I2S_sample_rate>=3: 16 kHz sampling rate • <I2S_Master_Slave>=0: Master mode (default value if parameter is not specified). In master mode I2S_CLK, I2S_WA, I2S_TX are generated by the module as output. I2S_RX is an input signal
`AT+USPM=1,1,0,0,2`	OK	AT+USPM=<main_uplink>,<main_downlink>,<alert_sound>,<headset_indication>,<vmic_ctrl> • <main_uplink>=1: Uplink path 1 via I2S (1 for e.g.; any path via I2S is ok) • <main_downlink>=1: Uplink path 1 via I2S (1 for e.g.; any path via I2S is ok) • Change audio path to I^{2}S. This is the port connected to the external codec on EVB
`AT+UMCLK=2,0`	OK	AT+UMCLK=<mclk_mode>,<enabling_mode> • <mclk_mode>=2: codec master clock at 13 MHz • <enabling_mode>=0: "Audio dependent" mode (the clock is applied to the CODEC_CLK pin only when the audio path is active (audio samples are read on the I2S_RX line and written on the I2S_TX line. For this codec it is not needed to maintain the clock while I^{2}S is not running, since just the voltage supply is needed to make I^{2}C work. Be aware that other codecs could need to maintain the clock running also for register programming. This can be achieved by <enabling_mode>=1: "Continuous" mode
`AT+UI2CO=1,0,0,0x10,0`	OK	AT+UI2CO=<I2C_controller_number>,<bus_mode>,<bit_rate>,<device_address>,<address_width> • Open logical channel for the Maxim external codec on I^{2}C • <I2C_controller_number>=1: Controller 1 • <bus_mode>=0: Bus Mode Standard (0 – 100 kb) • <bit_rate>=0: I^{2}C bit rate is 100 kb/s • <device_address>=0x10: Device Address in HEX format; this address can be found in the coded datasheet • <address_width>=0: 7 bit address

Command	Response	Description
AT+UI2CW="00000000108F202400 00103300210000008A",18	OK	AT+UI2CW=<hex_data>,<nof_byte_to_write> • <hex_data>= first register address, value for 1st register, value for 2nd register, etc. (17 registers values) • <nof_byte_to_write>=18 (register address +17 registers values) • Writing in the register configure the codec (gains, I^{2}S configuration, clock configuration, etc. Refer to Max9860 datasheet for details)
AT+UI2CW="049E",2	OK	As above; write byte 0x9E in register 0x04; See Max9860 datasheet for meanings of registers values
AT+UI2CC	OK	Close logical channel on I^{2}C The codec is now initialized an ready to work when the module starts audio activity; e.g. for audio test
AT+UPAR=0,0,0	OK	A tone with infinite repetitions can be started by command The module enables 13 MHz clock signal on CODEC_CLK pin. I^{2}S is enabled and starts to transmit tone data. The tone is played on the loudspeaker by the codec.
AT+USAR=0	OK	The command stops the tone. The module stops transmission of audio data on I^{2}S and disable 13 MHz clock signal

The codec supply is maintained up also after the codec is not used anymore.

5.2.1.1 Connection to I2S1

If the codec is connected to I2S1 (not I2S), change the initial commands in the example presented in section 5.2.1 with:

Command	Response	Description
AT+USPM=1,1,0,0,2	OK	AT+USPM=<main_uplink>,<main_downlink>,<alert_sound>,<headset_indication>,<vmic_ctrl> • <main_uplink>=1: Uplink path 1 via I^{2}S (1 for e.g.; any path via I2S is ok) • <main_downlink>=1: Uplink path 1 via I^{2}S (1 for e.g.; any path via I2S is ok) Change audio path to I2S. This allows setting of the I2S1 properties (I2S1 properties can't be changed while I2S1 is in use, i.e. the paths are via I2S1)
AT+UI2S=1,3,0,3	OK	AT+UI2S=<I2S_mode>,<I2S_port>,<I2Sclk_wa>,<I2S_sample_rate>,<I2S_Master_Slave> • <I2S_mode>=1: PCM mode • <I2S_port>=3: Connect I2S1 to I2Sx connection point • <I2Sclk_wa>=0: Dynamic mode (I2S_CLK and I2S_WA outputs are active and running only while audio path is active) • <I2S_sample_rate>=3: 16 kHz sampling rate • <I2S_Master_Slave>=0: Master mode (default value if parameter is not specified). In master mode I2S1_CLK, I2S1_WA, I2S1_TX are generated by the module as output. I2S1_RX is an input signal

Command	Response	Description
AT+USPM=5,5,0,0,2	OK	AT+USPM=<main_uplink>,<main_downlink>,<alert_sound>,<headset_indication>,<vmic_ctrl> • <main_uplink>=5: Uplink path 5 via I2S1 (5 for e.g.; any path via I2S1 is ok) • <main_downlink>=5: Uplink path 5 via I2S1 (5 for e.g.; any path via I2S1 is ok) Use audio paths via I2S1. This is the port connected to the external codec on EVB.

Other commands for clock and codec configuration remain as above.

5.2.1.2 External Device Configuration +UEXTDCONF

☞ To be used only with LISA-U2x0-x2S and subsequent versions.

This command can be used for an external device configuration, e.g. an audio codec, at boot time. The setting (on / off) for each supported device is saved in NVM and applied each time the module is powered on. The configuration for each supported device is hard-coded in the firmware.

Command	Response	Description
AT+UEXTDCONF=0,1	OK	• <device_id>=0: Maxim Max9860 audio codec, connected via I^{2}C • <configuration_enable>=1: enabled
AT+CPWROFF	OK	

Once this procedure has been executed, it is no longer necessary to repeat the procedure in the section 5.2.1 after each system boot.

☞ When enabled, at every start-up the module performs the actions corresponding to the following commands:

Command	Description
AT+UMCLK=2,0	Set the external codec master clock at 13 MHz
AT+UI2CO=1,0,0,0x10,0	Open the I^{2}C logical channel (connected to the external codec)
AT+UI2CW="00000000108F20240000103300250000008A",18	Send, via I^{2}C, the specified byte sequence (for external codec configuration
AT+UI2CW="049E",2	Send, via I^{2}C, the specified byte sequence (for external codec configuration)
AT+UI2CC	Close the I^{2}C logical channel

+UI2S must be set with:

- o <I2S_mode>=1: PCM mode
- o <I2S_sample_rate>=0: 8 kHz sampling rate
- o <I2S_Master_Slave>=0: Master mode

☞ This procedure can be used if the codec supply is activated (by VCC_EN) and before the module boot and maintained active afterwards.

☞ <I2S_port> in +UI2S and <main_uplink>, <main_downlink> in +USPM must be set according to the I2S/I2S1 connection, as described in the section 5.2.1.1.

☞ The gains set for the Maxim Max9860 in the +UI2CW command are not optimized for the HW configuration of the EVK-U20 / EVK-U23 EVB. For the EVB configuration, the setting proposed in section 5.2.1 is suggested.

6 DTMF decoder

6.1 About ETSI DTMF

The 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.

The multi-part ETSI Standard ES 201 235 [5] specifies how to apply DTMF signaling to transmitters and receivers. It conforms to the International Telecommunication Union (ITU-T) Recommendation Q.23 [7] and it provides a complete set of requirements for all the 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 the 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

The 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 4. 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.

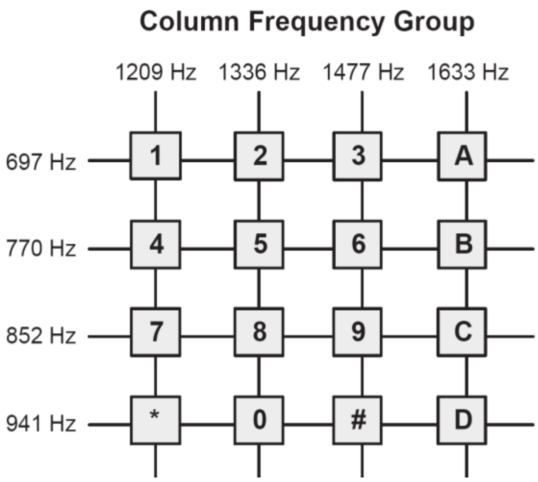

Figure 4: Touch-Tone telephone keypad: a row and a column tone is associated with each digit

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

☞ Not supported by LISA-U1 series or by LISA-U2x0-x1S and previous version.

LISA-U2 series 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 the +UDTMFD AT command is used to configure it (for more details on the command description, refer to u-blox commands manual [1]).

7.1 Implementation

Enable the DTMF decoder via the AT command once per module power cycle and before any call set up, e.g.:

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

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

```
+UUDTMFD: 4
```

Here, 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 [5]. 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. The 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), for instance those caused by 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 a criterion 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 this way, 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 the application.

7.2.1 Decoder configuration

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

The AT command can reconfigure the values 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 8: +UDTMD parameter description and factory-programmed values

The factory-programmed values may vary in 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 the best performance in terms of signal level operating range, and 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 "touchtone-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 [6].

Robust mode advantages: case study

Figure 5 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 clicks (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 false digit repetitions affect the normal mode. 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 detector rejects the bursts and correctly outputs the digits.

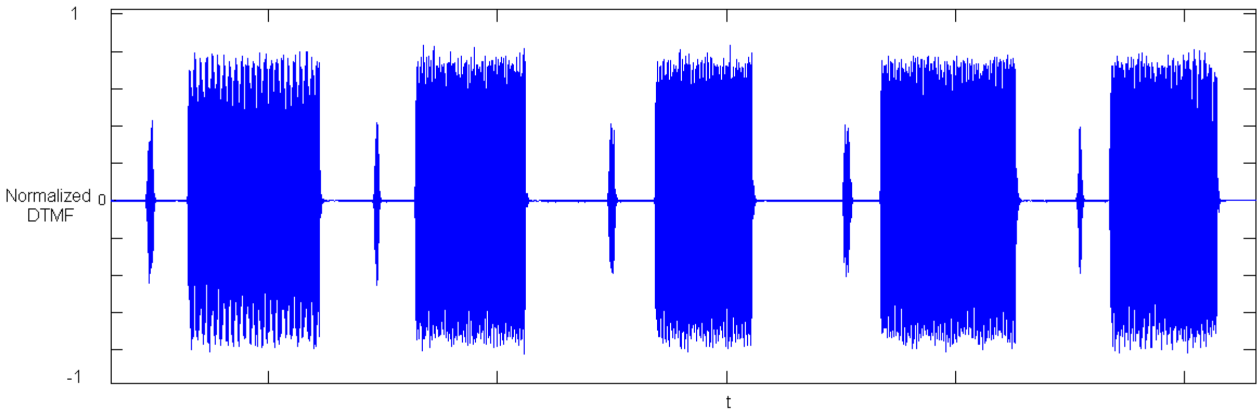

Figure 5: Normalized tones corresponding to digit presses '1','2','3','4','5', with interruptions caused by VoIP-based connection

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

The **<att_cfg>** parameter applies attenuation to 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
```

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
```

Here, 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 is considered unreliable.

The **<threshold>** parameter is the current threshold applied to the signal level in order 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 a 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 meet operating conditions according to the requirements from ETSI ES 201 235-4 [4], for more details refer to the Table 9.

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

Table 9: 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)

The **<immunity>** parameter calibrates the decoder with 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, for instance with low bit-rate codecs.

The default immunity value (14), combined with the robust mode, complies with the ETSI requirements for speech immunity ETSI ES 201 235-4 [4]. According to ETSI ES 201 235-3 [3], ETSI ES 201 235-4 [4], "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 [3] and ETSI ES 201 235-4 [4], 'Speech immunity performance'.

Table 11 provides the tests results with four different detector configurations with respect to speech immunity (default values for the other parameters used). Talk-offs represent the number of false detections during the testing.

7.2.1.4 False digit repetitions (<max_int> parameter)

The 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 [4], 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 10 reports the not configurable signal conditions. The signal conditions and tolerances comply with ETSI ES 201 235-3 [3] and ETSI ES 201 235-4 [4].

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 10: Not configurable factory-programmed signal conditions and tolerances on the u-blox wireless 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

Speech immunity tests have been performed according to ETSI ES 201 235-4 [4] (Paragraph 4.3, 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 11, 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).

<immunity> parameter value	<mode> parameter value	Talk-offs
0	normal mode	9900
0	robust mode	4800
14	normal mode	100
14	robust mode	5 (as in ETSI reference)

Table 11: ETSI Speech immunity tests, with 20 minutes test signal

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

7.3.2 Detection performance

The detection performance measurement and benchmarking was done as in ETSI TR 126 975 [6], Chapter 10 "Performances with DTMF tones", the tests implemented as in 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 necessitate the widely used speech codecs adopted by 2G and 3G wireless networks: **the AMR low bit rate modes are not transparent to DTMF tones** (refer to ETSI TR 126 975 [6] and Table 13).

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 [6] 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 12 and Table 13 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 Table 12 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 12: Results for each experiment (rows) for each decoder configuration (columns)

☞ 100% of detections are 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 Table 13 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**
exp11: -13 dBm with -6 dB (reverse) twist	0/8.1	16.6/0	11.8/0	34.7/0	**35.9/0**

Table 13: Results for each experiment (rows) for each decoder configuration (columns)

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

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

☞ exp11 with artificially added negative twist represents unreal/rare network situations. For more details, refer to section 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 [4] to 12 dB: for more details refer to section 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 the speech immunity performance. See for example the 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 transmitter automatically controls the DTMF signaling pause duration, 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 section 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 section 7.4).

7.4 Configuration examples

The performance estimates of the following configuration examples are given for error free conditions (no speech frame drops). The frame drops may cause false digit repetitions that can be coped with <max_int> parameter configuration as discussed in the section 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 section 7.3.

7.4.2 ETSI-compliant decoder with guaranteed speech channel QoS

To get rid of low-bit rate codecs distortions, it is possible to configure the module to support and make calls only with a reduced speech codec set. The +UDCONF AT command configures the speech codec. For more details, refer to the 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; e.g.

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

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

☞ For the description 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 even 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 guidelines and need actual in-field tuning and validation.

7.4.3.1 Reduced speech immunity

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

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

According to the performance measurements, the 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 operating range reduction improves 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
```

This is a configuration that should have an improved talk-off with respect to section 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
AFB	Analysis Filter Bank
AGC	Automatic Gain Control
AP	Application Processor
AT	AT Command Interpreter Software Subsystem, or attention
DBF	Downlink Biquad Filters command
DES	Dynamic Echo Suppressor
DSP	Digital Signal Processing
EC	Echo Cancellation algorithm
FIR	Finite Impulse Response filter
HF	Hands-free Algorithm
LEM	Loudspeaker-Enclosure-Microphone
LMS	Least Mean Square
MGC	Microphone Gain control command
NLMS	Normalized Least Mean Square
NR	Noise Reduction algorithm
NVM	Non Volatile Memory
PSTN	Public Switched Telephone Network
RX	Receiver
SGC	Speaker Gain control command
SER	Spectral Echo Reduction
SFB	Synthesis Filter Bank
TX	Transmitter
UBF	Uplink Biquad Filters command

Related documents

[1] u-blox AT Commands Manual, Docu No WLS-SW-11000

[2] LISA-U series System Integration Manual, Docu No UBX-13001118

[3] ETSI ES 201 235-3 V1.3.1, Specification of Dual Tone Multi-Frequency (DTMF) Transmitters and Receivers; Part 3: Receivers

[4] 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

[5] Work Items with ETSI Document Number of '201 235', see Work Programme search database, http://www.etsi.org/

[6] 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)

[7] ITU-T Recommendation Q.23: Technical features of push-button telephone sets

[8] ETR 229: October 1995 (GSM 06.08 version 4.0.0), Performance characterization of the GSM half rate speech codec

All these 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
-	Jul. 06, 2012	ague	Initial release (Last revision with old doc number, 3G.G3-CS-12003)
A	Aug. 01, 2013	ague	Added DTMF decoder description for LISA-U2 series

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