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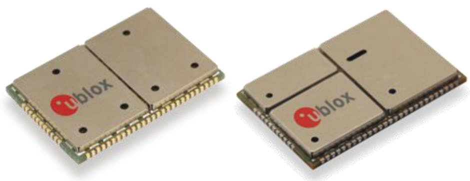
LISA-U1 / LISA-U2 series

Audio

Application Note

Abstract

This document provides information and procedures to resolve audio related problems with LISA-U1 / LISA-U2 series modules.
Moreover procedures for tuning of hands-free algorithm (echo cancelation, automatic gain control, noise reduction) and external audio management are described.



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Contents

1 Introduction

This document provides information and procedures for resolving potential audio related problems with LISA-U1 / LISA-U2 series modules. Examples of procedures, e.g. tuning the hands-free algorithm (Echo cancelation, Automatic Gain Control, Noise Reduction) and management of an external codec, are described.

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 external audio devices (i.e. microphone and loudspeaker) have been connected 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). LISA provides a hands-free algorithm to remove echo. This algorithm is controlled by HF parameters within the uplink audio path in use (refer to AT+USPM and AT+UHFP command in u-blox AT Commands Manual [1]), that are stored in NVM dynamic parameters profile (refer to chapter 3.1 and AT&W command description in u-blox AT Commands Manual [1]).

A step-by-step procedure for choosing parameters to remove echo heard on the far end side, is described in chapter 3. 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 chapter 4.

A description of HF algorithm and parameters meaning is given in section 2.1.

2.1 HF algorithm description

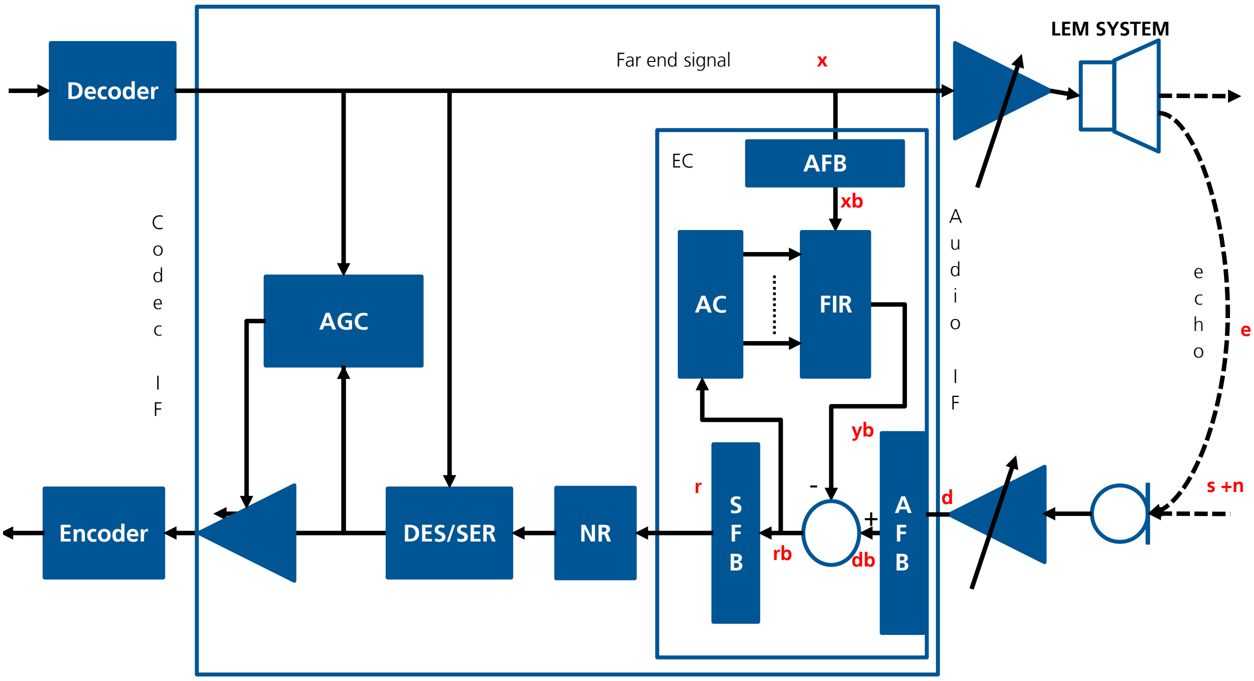

Figure 1: Block diagram for hands-free

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 there is an FIR filter emulating the LEM system behavior in that sub-band, generating 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 coefficients of each FIR filters must be adapted by the Adaptation Control block (AC); 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

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

2.2.1 Parameters for block activation and initialization

HF_ALGORITHM_INIT Range 0x0000, 0x07FF

This parameter is a set of flags that control the activity and initialization of the EC, AGC and NR blocks.

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

Flag	LISA-U1 series	LISA-U2 series
Bit #0 set	Echo Cancelation (EC) initialization	Unused
Bit #1 set	EC restart (without coefficient initialization)	Unused
Bit #2 set	EC on	Echo Cancelation (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 Not Available parameters

The parameters presented in this paragraph 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 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 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 calculated attenuation is lower than MIN_ATTEN, then 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

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	8192	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	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 Cancelation

Echo Cancelation is a sub band (SB) design, where adaptive linear filtering and subsequent echo subtraction is performed by identical systems on each sub-band (see section 2.1). After echo subtraction the sub-band signals are combined back to a single full spectrum signal. EC uses 3 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 4 frames. The higher this number, the slower but more accurate the adaptation converges.
EC_NR_COEFF_REAL	2 to 2000	100	Number of coefficients of the filter 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 coefficients of the filter 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 coefficients of the filter 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	100	Number of coefficients of the filter 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	100	Number of coefficients of the filter 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	100	Number of coefficients of the filter 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 Cancelation 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 Cancelation 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) so that echo heard on the far-end side is removed.

Please 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 Cancelation 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 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 Cancelation 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 for all <ec_nr_coeff_*>
- 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 considered.

  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 Cancelation initialized and on), any values in the allowed ranges of the AGC and NR parameters are accepted but not used; only EC parameters are used. Since the algorithm is adaptive, some seconds are needed to converge after that AT+UHFP command is sent. After algorithm converges, a residual echo remains. Try to change EC parameters till there is no residual echo. In very critical case, if 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 long reverberation time
- Reconsider points 2 to 4, if this command has no effect on 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>,0,0x0000,,,,,,,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 Cancelation 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 Cancelation 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 Cancelation 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

Warning: Using strong AGC can cause bad performance in double talk scenario and lead to a no-duplex configuration (see dedicated chapter 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 Cancelation 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

As with the previous step, NR parameters can be changed until the minimum residual noise remains on the uplink path. To appreciate the effect of the NR, 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>

8 Finally, the parameters found can be stored in NVM 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 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:

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

4.1 Procedure

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

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

Parameter	Value	Meaning
<hf_algorithm_init>	0x0100	Only AGC initialization and on
<add_atten>	500	500*3/32 = 47 dB additional AGC attenuation
<min_atten>	500	500*3/32 = 47 dB minimum AGC attenuation
<max_atten>	500	500*3/32 = 47 dB maximum 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 codec should by managed by the Application Processor (AP).

This chapter 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, please refer to LISA-U series System Integration Manual [2].

For details about AT command used in the examples below, please 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.

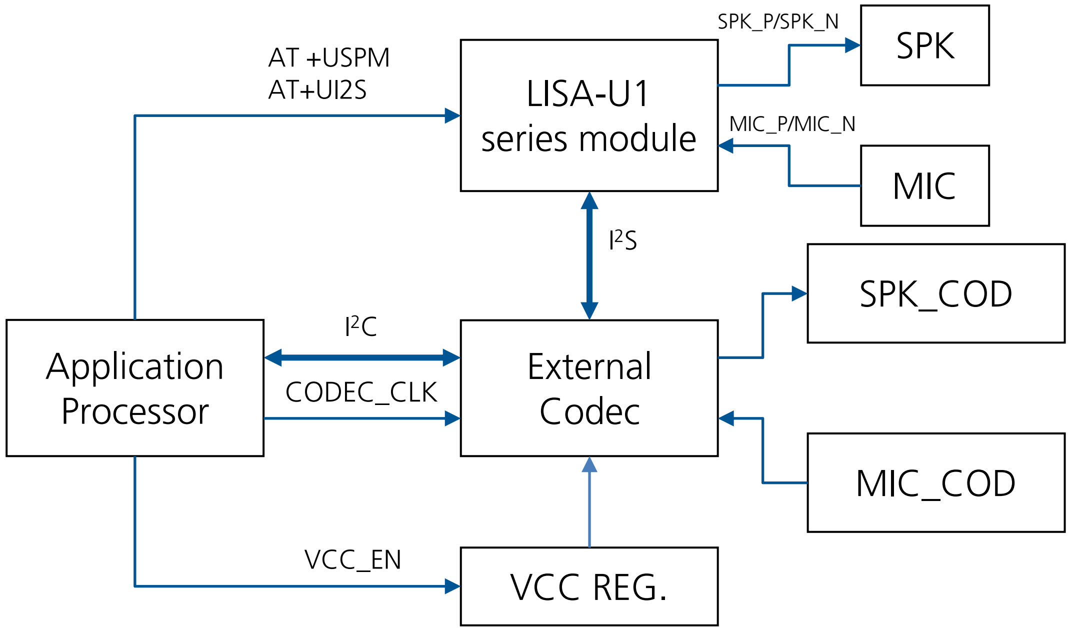

Figure 2: External codec management for LISA-U1 series module

External codec can be connected to LISA-U1 series module by an I^2S interface. The codec allows connection of 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 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 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
- LISA-U2 series modules insert a delay in the startup of ringers, alarms and signaling tones to allow the AP to correctly start up in time the external codec after receiving the +CIEV URC. +CIEV URCs on AT terminal can be generated when audio activities are running on the module, issuing the command AT+CMER=1,0,0,2,1. The delay is 200 ms

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

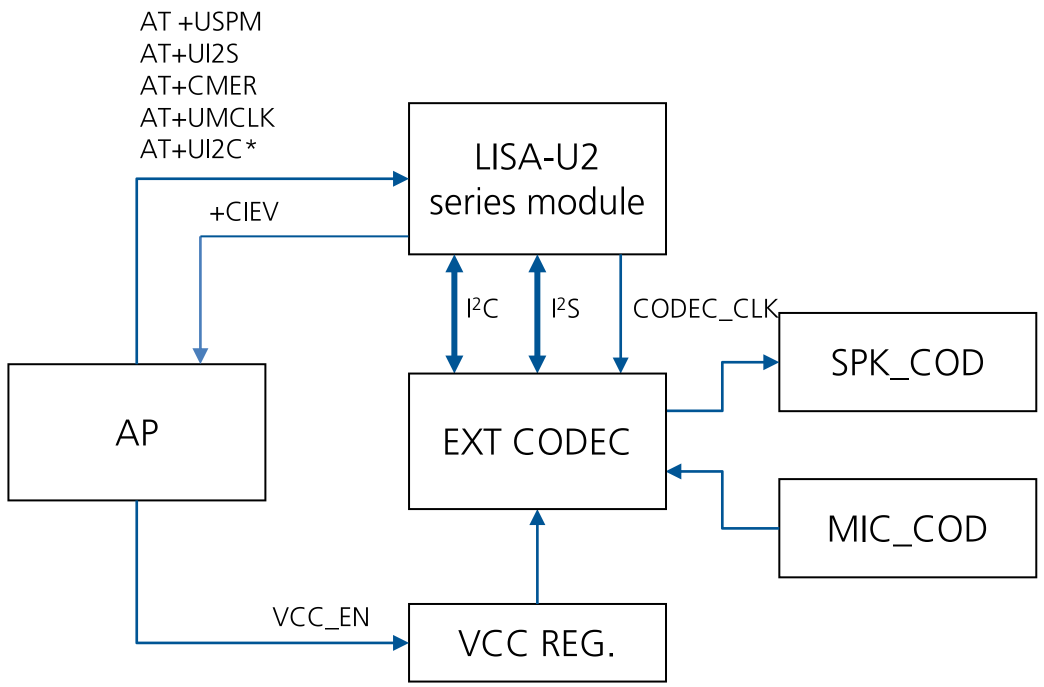

Figure 3: External codec management for LISA-U2 series module

Examples of codec management scenario on this architecture are given in chapter 5.2.1 and 5.2.2.

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: Codec always enabled

- 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^2S properties (I^2S properties can't be changed while I^2S is in use, i.e. the paths are via I^2S).
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^2S 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^2S. 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^2S is not running, since just the voltage supply is needed to make I^2C 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^2C • <I2C_controller_number>=1: Controller 1 • <bus_mode>=0: Bus Mode Standard (0 – 100 kb) • <bit_rate>=0: I^2C 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. Please 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

In case the codec is connected to I2S1 (not I2S), the initial commands in the example presented in 5.2.1 must be changed 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

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^2C • <configuration_enable>=1: Enabled
AT+CPWROFF	OK	

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

☞ When enabled, at every startup the module performs the actions corresponding to the following commands:

- AT+UMCLK=2,0
- AT+UI2CO=1,0,0,0x10,0
- AT+UI2CW="00000000108F2024000010330025000000 8A",18
- AT+UI2CW="049E",2
- AT+UI2CC

+UI2S must be set with:

- <I2S_mode>=1: PCM mode
- <I2S_sample_rate>=0: 8 KHz sampling rate
- <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 chapter 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 chapter 5.2.1 is suggested.

5.2.2 Scenario: Codec enabled / disabled

The codec is setup by AP only when needed and can be shut down after use to reduce power consumption.

The AP should activate the Mobile termination Event Reporting to have an unsolicited message when audio (I^2S) is activated and codec need to be enabled.

Command	**Response**	**Description**
AT+CMER=1,0,0,2,1	OK	Activate the Mobile termination Event Reporting enabling the +CIEV URCs sending from MT to DTE for indications
AT+USPM=5,5,0,0,2	OK	
AT+UI2S=1,1,0,3	OK	
AT+USPM=1,1,0,0,2	OK	
AT+UMCLK=2,0	OK	
AT+UI2CO=1,0,0,0x10,0	OK	For all these steps refer to comments in previous scenario

After this sequence the module is configured. Two distinct scenarios can be considered.

5.2.2.1 AP starts an audio activity

When the AP needs to start an audio activity, e.g. start a speech call, the following activities needs to be performed:

- Enable the Codec supply by VCC_EN
- Enable and configure the codec by +U2ICW commands, e.g.:

Command	**Response**	**Description**
AT+UI2CW="00000000108F202400 00103300210000008A",18	OK	
AT+UI2CW="049E",2	OK	For all these steps refer to comments in previous scenario
ATD<number>	OK	Start audio activity, e.g. start a speech call
		The codec clock and the I^2S automatically starts
		Due to +CMER setting, the module generates +CIEV URCs; these messages can be ignored in this case, since the codec has been already initialized
	+CIEV: 6,1	URC about call in progress indication
	+CIEV: 10,2	URC about outgoing call in dialing indication
	+CIEV: 2,1	URC about signal quality indication
	+CIEV: 2,2	URC about signal quality indication
	+CIEV: 10,3	URC about alerting indication of outgoing call in remote party
	+CIEV: 4,1	URC about indication of sound generation starting
	+CIEV: 10,0	URC about setup indication for end of call
	+CIEV: 4,0	URC about of stop indication for sound generation
ATH	OK	Stop the audio activity, e.g. stop the speech call
		The codec clock and the I^2S are automatically stopped by the module
		AP can disable the codec supply by VCC_EN
		Due to +CMER setting, the module generates +CIEV URCs; these messages can be ignored in this case, since the codec has been already disabled
	+CIEV: 6,0	URC about indication for end of call in progress
	+CIEV: 2,1	URC about indication for signal quality
	+CIEV: 2,2	URC about indication for signal quality

5.2.2.2 Module starts an audio activity

When an audio activity is not started from AP, e.g. an incoming call is received from the module and the ringer is started, the AP should be informed by the module to start the codec wake-up procedure. The Mobile termination Event Reporting can be used to receive an unsolicited message on AP when audio (I^2S) is activated and audio codec needs to be enabled.

Command	Response	Description
		The module receives an incoming call. The module has been already configured by AP at the start up, as described in 5.2.2
		The I^2S path is enabled by the module with the modality set by +UI2S command
		The audio driver enables CODEC_MCLK in the preset mode (13 MHz)
		Due to +CMER setting, the module generates +CIEV URCs
	`+CIEV: 2,1`	URC about signal quality indication
	`+CIEV: 6,1`	URC about call in progress indication
	`+CIEV: 10,1`	URC about incoming call indication
	`+CIEV: 4,1`	URC about sound generation start indication
		The unsolicited messages trigger the AP routine for codec control
		AP enables codec power supply by VCC_EN signal
`AT+UI2CW="00000000108F202400 00103300250000008A",18`	`OK`	Codec configuration triggered by the AP
	`RING`	The ringer is started with a delay of 200 ms after +CIEV URC. This delay allows the correct codec setup before ringer starts. The RING unsolicited message is received by AP. The ringer melody is played by the codec
	`+CIEV: 2,0`	URC about indication of the signal quality
	`RING`	AP answers to the call
`ATA`	`OK`	
		Due to +CMER setting, the module generates +CIEV URCs
	`+CIEV: 4,0`	URC about stop indication of sound generation
	`+CIEV: 10,0`	URC about end indication of call set up
		Speech can be heard through the codec
	`NO CARRIER`	Far end hangs call; the NO CARRIER unsolicited message is received by the AP
		Due to +CMER setting, the module generates +CIEV URCs
	`+CIEV: 6,0`	URC about indication of no call in progress
	`+CIEV: 2,0`	URC about indication of the signal quality
	`+CIEV: 2,1`	URC about indication of the signal quality
	`+CIEV: 2,2`	URC about indication of the signal quality
		Speech channel is closed and I^2S interface is stopped by the module
		The module turns CODEC_MCLK off
		The unsolicited messages trigger the AP routine for codec control
		AP can disable codec power supply by VCC_EN signal

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 Cancelation 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 3G.G2-HW-10002

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 please register on our homepage.

Revision history

Revision	Date	Name	Status / Comments
-	06/07/2012	ague	Initial release

Contact

For complete contact information visit us at www.u-blox.com

u-blox Offices

North, Central and South America

u-blox America, Inc.

Phone: +1 (703) 483 3180
E-mail: info_us@u-blox.com

Regional Office West Coast:

Phone: +1 (703) 483 3184
E-mail: info_us@u-blox.com

Technical Support:

Phone: +1 (703) 483 3185
E-mail: support_us@u-blox.com

Headquarters
Europe, Middle East, Africa

u-blox AG

Phone: +41 44 722 74 44
E-mail: info@u-blox.com
Support: support@u-blox.com

Asia, Australia, Pacific

u-blox Singapore Pte. Ltd.

Phone: +65 6734 3811
E-mail: info_ap@u-blox.com
Support: support_ap@u-blox.com

Regional Office China:

Phone: +86 10 68 133 545
E-mail: info_cn@u-blox.com
Support: support_cn@u-blox.com

Regional Office Japan:

Phone: +81 03 5775 3850
E-mail: info_jp@u-blox.com
Support: support_jp@u-blox.com

Regional Office Korea:

Phone: +82 2 542 0861
E-mail: info_kr@u-blox.com
Support: support_kr@u-blox.com

Regional Office Taiwan:

Phone: +886 2 2657 1090
E-mail: info_tw@u-blox.com
Support: support_tw@u-blox.com