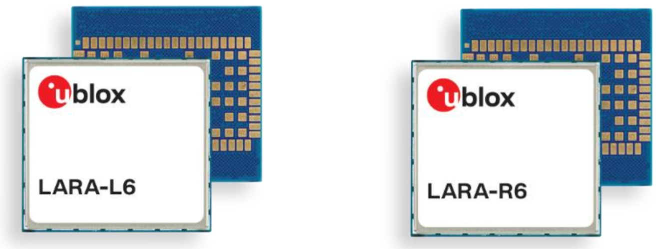
LARA-L6 / LARA-R6 series

Audio interface

Application note



Abstract

This document provides information about the LARA-R6 and LARA-L6 SW audio functionality and application interfaces.

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Document information

Title	**LARA-L6 / LARA-R6 series**	
Subtitle	Audio interface	
Document type	Application note	
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This document applies to the following products:

Product name	
LARA-R6 series	Except for LARA-R6001D, LARA-R6401D, LARA-R6801D
LARA-L6 series	Except for LARA-L6004D, LARA-L6804D

Contents

1 Introduction

This document provides an overview of the audio features, and address specific topics including volume management, audio routing and profiles, speech codecs and external codec management, the DTMF in-band signaling detector [2] / DTMF RTP decoder [6], etc.

Echo cancellation and the audio tuning interface are introduced in section 8.

For details of the AT commands and feature integration, see the AT commands manual [1] and the system integration manual [5].

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 Supported features

Table 1 shows the complete list of supported features and AT commands.

Feature	Sub-features	AT commands	Supported by LARA-R6 "00B"	Supported by LARA-L6 "00B"	Supported by LARA-R6/L6 "01B"
Volume	Microphone muting	`+CMUT`	Yes	Yes	Yes
	Loudspeaker volume	`+CLVL`	Yes	Yes	Yes
	Ringer volume	`+CRSL`	No	**Yes**	Yes
	Players volume	`+CRSL`, `+USGC`	No	No	No
	Alert tone muting	`+CALM`	Yes	Yes	Yes
	Message sound muting	`+UMSM`	Yes	Yes	Yes
	Silent alarm	`+CALA`	No	No	No
Audio routing and profiles	Speech path mode	`+USPM`	Yes	Yes	Yes
Speech codecs	GSM EFR		Yes	Yes	Yes
	GSM FR		Yes	Yes	Yes
	GSM HR		Yes	Yes	Yes
	FR AMR		Yes	Yes	Yes
	FR AMR WB		Yes	Yes	Yes
	HR AMR		Yes	Yes	Yes
	UMTS AMR		Yes	Yes	Yes
	UMTS AMR 2		Yes	Yes	Yes
	UMTS AMR WB		Yes	Yes	Yes
	VoLTE AMR NB		Yes	Yes	Yes
	VoLTE AMR WB		Yes	Yes	Yes
	VoLTE EVS NB		No	No	No
	VoLTE EVS WB		No	No	No
	VoLTE EVS SWB		No	No	No
	VoLTE EVS FB		No	No	No
Supervisory tones	Ringing tone on MOC		Yes	Yes	Yes

Feature	Sub-features	AT commands	Supported by LARA-R6 “00B”	Supported by LARA-L6 “00B”	Supported by LARA-R6/L6 “01B”
	Subscriber busy on MOC		Yes	Yes	Yes
	Call waiting on MTC		Yes	Yes	Yes
	Ringer on MTC (ringtone)		Yes	Yes	Yes
	Incoming SMS		Yes	Yes	Yes
	Alarm tone		No	No	No
	Tones mixed with speech		Yes	Yes	Yes
Players	Pre-defined tones		No	No	No
	Tone generator UL/DL	`+UTGN`	Yes	Yes	Yes
	Ringer selection	`+URNG`	No	No	No
Audio file player / recorder	NB 8 kHz		No	No	No
	WB 16 kHz		No	No	No
	Generic player UL	`+UPLAYFILE`	No	No	No
	Generic player DL	`+UPLAYFILE`	No	No	No
	Custom ringer melody	`+UPLAYFILE`	No	No	No
	Answering machine	`+UPLAYFILE`	No	No	No
	Generic recorder UL	`+URECFILE`	No	No	No
	Generic recorder DL	`+URECFILE`	No	No	No
	Microphone recorder	`+URECFILE`	No	No	No
Speech player / recorder	Speech player UL	`+UAPLAY`	No	No	No
	Speech player DL	`+UAPLAY`	No	No	No
	Speech recorder UL	`+UAREC`	No	No	No
	Speech recorder DL	`+UAREC`	No	No	No
Speech codec management	Speech codec configuration 2G/3G	`+UDCONF=30`	Yes	Yes	Yes
	Codec mode info 2G/3G	`+USPEECHINFO`	Yes	Yes	Yes
	Speech codec configuration VoLTE	`+USPEECHCFG`	Yes	Yes	Yes
	Codec mode info VoLTE	`+USPEECHINFO`	Yes	Yes	Yes
	Codec rate info VoLTE	`+USPEECHINFO`	No	No	No
External codec management	Codec configuration	`+UEXTDCONF`	Yes	Yes	Yes

Feature	Sub-features	AT commands	Supported by LARA-R6 “00B”	Supported by LARA-L6 “00B”	Supported by LARA-R6/L6 “01B”
	Clock for external codec	`+UMCLK`	Yes	Yes	Yes
	Continuous mode	`+UMCLK`	No	No	No
Digital audio interface	I2S PCM modes configuration	`+UI2S`	Yes	Yes	Yes
	Normal I2S configuration	`+UI2S`	Yes	Yes	Yes
	Stereo mode 48 kHz	`+UI2S`	No	No	No
	Sampling rates	`+UI2S`	16kHZ	16kHz	16kHz **& 48kHz**
	I2S always on (continuous mode)	`+UI2S`	No	No	No
	Slave mode	`+UI2S`	No	No	**Yes**
Analog audio interface			No	No	No
Speech enhancement	Car HF speech enhancement		No	No	No
	Alarm panel speech enhancement		Yes	Yes	**Yes (with microphone gain automatic adaptation)**
	Desktop HF speech enhancement		Yes	Yes	Yes
	Digital speech gains UL/DL	`+UMGC`, `+USGC`	No	No	No
	Side tone	`+USTN`	No	No	No
	Equalizers UL/DL	`+UUBF`, `+UDBF`	No	No	No
	Basic tuning AT commands	`+UHFP`	No	No	No
	Extended tuning AT commands	`+UTI`	No	No	**Yes**
	Audio Tuning Tool		No	No	No
	Audio parameters saving/recovery		Yes	Yes	**Yes (OTA support)**
In-Band DTMF	DTMF detector (legacy)	`+UDTMFD` `+UDTMFCFG`	No Yes	No Yes	No Yes
	DTMF detector MODA	`+UDTMFCFG`	No	No	No
	Burst mode	`+UDTMFCFG`	No	No	No
	DTMF generator	`+UDTMFG` `+UTGN`	No Yes	No Yes	No Yes
	Contact ID protocol support	`+UDTMFCFG`	No	No	No
	PCM logging	`+UDTMFCFG`	No	No	No
VoLTE DTMF	DTMF RTP decoder	`+UDTMFCFG`	Yes	Yes	Yes
	Burst mode	`+UDTMFCFG`	No	No	No
	DTMF regenerator (local play on loudspeaker)	`+UDTMFCFG`	Yes	Yes	Yes

Feature	Sub-features	AT commands	Supported by LARA-R6 “00B”	Supported by LARA-L6 “00B”	Supported by LARA-R6/L6 “01B”
	Smart DTMF generator	+UDTMFG	No	No	No
	VoLTE DTMF RTP mode disable	+UDTMFCFG	No	No	No
Audio configuration interface	Audio configuration	+UAUDCFG	No	Yes	Yes
	NVM-RAM mode setting	+UNVMCFG +UDCONF=110	Yes No	Yes No	Yes No
	NVM configuration management commit	+UNVMW	Yes	Yes	Yes
	NVM configuration management reset	+UNVMR	Yes	Yes	Yes
	NVM configuration management factory restore	+UNVMF	Yes	Yes	Yes
Product testing	Test audio IF loopback	+UPAR, +USAR	Yes	Yes	Yes
Sound activity indications	Sound activity indications	+CIEV	No	No	No

Table 1: Supported features by product versions

2 Volume

2.1 Loudspeaker volume

Figure 1 shows the volume control in downlink direction, by the +CLVL AT command.

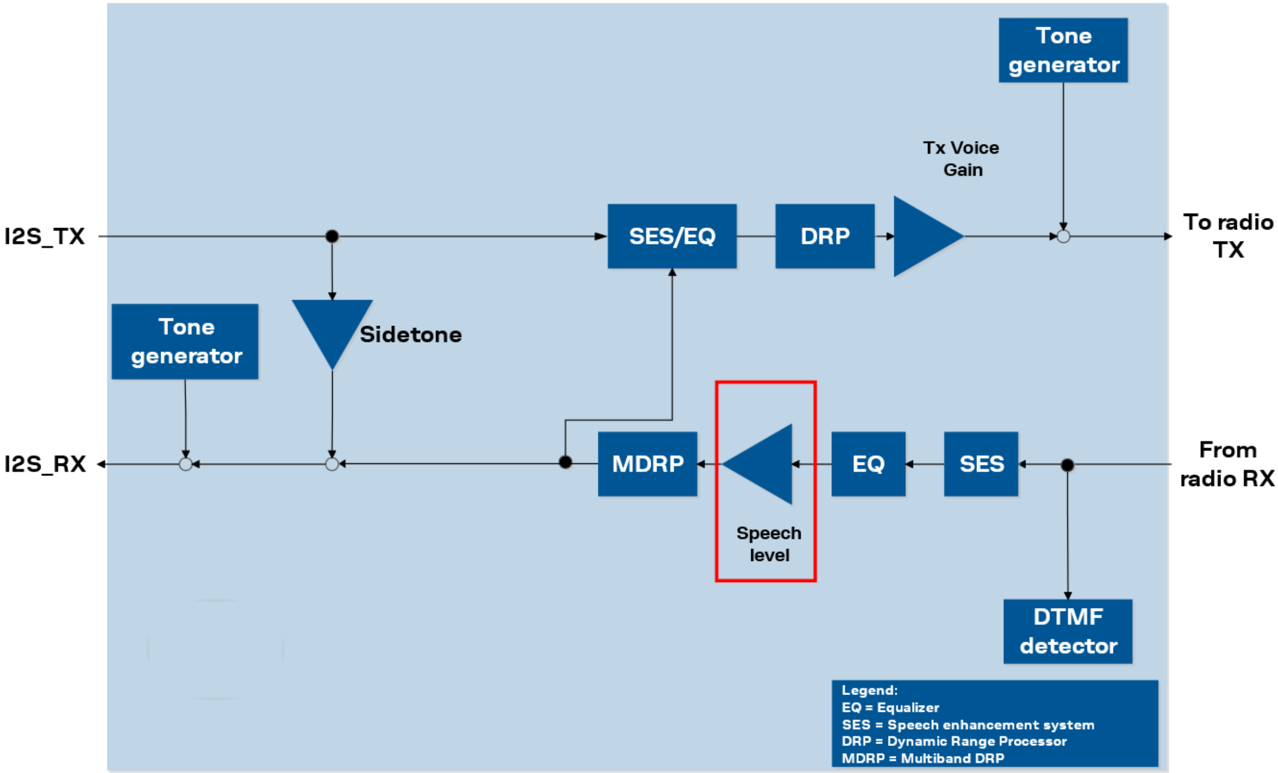

Figure 1: Incoming speech level gain

The +CLVL AT command selects the incoming speech volume during every call:

```
AT+CLVL=[<level>]
```

The parameter is saved in NVM, and it is applied on all profiles.

LARA-R6 "00B" series

The allowed values range from 0 to 6, where 0 means mute, 1 means -15 dB, 6 means 0 dB and the step size is 3 dB. The default and factory-programmed value is 3, i.e., -9dB.

Speech level (dB) vs <level> parameter							
<level>	6	5	4	**3 (default)**	2	1	0 (mute)
dB	0	-3	-6	**-9**	-12	-15	- inf.-

LARA-L6 series and LARA-R6 "01B" series

The allowed values range from 0 to 6, where 0 means mute, 1 means -30 dB, 6 means 0 dB and the step size is 6 dB. The default and factory-programmed value is 4, i.e., -12dB.

Speech level (dB) vs <level> parameter							
<level>	6	5	**4 (default)**	3	2	1	0 (mute)
dB	0	-6	**-12**	-18	-24	-30	- inf.-

The command affects only the downlink speech volume, not the tone generator volume. For example:

Command	Response	Description
AT+CLVL=5	OK	Set the speech volume to -3dB on LARA-R "00B" and -6dB on LARA-L6 and LARA-R6 "01B"

Loudspeaker output can be maximized (to e.g. save cost in external amplifier power) by changing the +CLVL attenuation from -9 dB (LARA-R6 "00B")/-12dB (LARA-L6 series and LARA-R6 "01B") up to 0dB.

On handsfree profiles the actual gain increase at I2S port with respect to factory setting is less than the theorical 9dB/12 dB because of the dynamic range processor (MDRP) that is placed after the gain controlled by +CLVL, as shown in the following table:

Speech level (dB) at I2S port vs <level> parameter for LARA-R6 "00B"							
<level>	**6**	5	4	3 (factory-programmed)	2	1	0
Flat profile(dB)	**0**	-3	-6	-9	-12	-15	-50
HF profile(dB)	**-6**	-7	-7.5	-9	-12	-15	-50

The flat profile on the other side maintains the full dynamic range.

2.2 Ringer sound level

☞ On LARA-R6 "00B" series the command is not supported.

Figure 2 shows the volume control of the ringer in downlink direction, controlled by the +CRSL AT command.

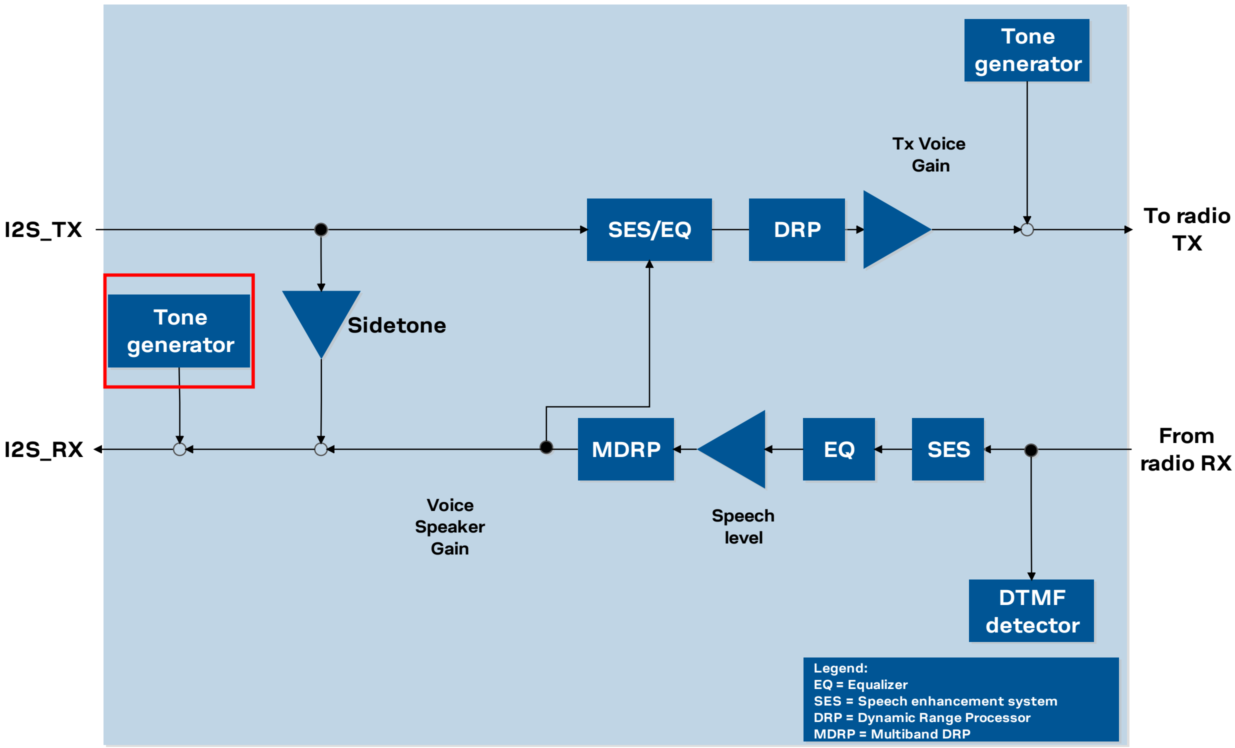

Figure 2: The Ringer is implemented through the downlink tone generator

The ringer is implemented through the generic tone generator in downlink using a 425Hz tone.

The +CRSL AT command selects the sound level for the ringer of an incoming call:

```
AT+CRSL=[<level>]
```

The allowed values range goes from 0 to 6, where 0 means mute. The step size is about 6 dB. The default and factory-programmed value is 5 that is 12dB above the speech level when +CLVL is set to factory. The setting is persistent also after a reboot, it is saved in NVM.

Example to set a new sound level for the ringer:

Command	Response	Description
AT+CRSL=4	OK	Set the ringer volume to level 4
AT+CFUN=16	OK	Module reboot
AT+CRSL?	+CRSL:4	Read the value, the setting is saved in NVM

Example to restore the factory level for the ringer:

Command	Response	Description
AT+CRSL=	OK	Restores factory value
AT+CRSL?	+CRSL:5	Read the value, the factory has been restored

2.3 Microphone muting

The +CMUT AT command configures uplink voice muting during all the voice calls:

```
AT+CMUT=<n>
```

This setting is persistent also after an audio profile switch. The <n> parameter can be:

- 0 (default value): mute off, or
- 1: mute on

Muting acts after the speech enhancement system (SES) block, as shown in Figure 3:

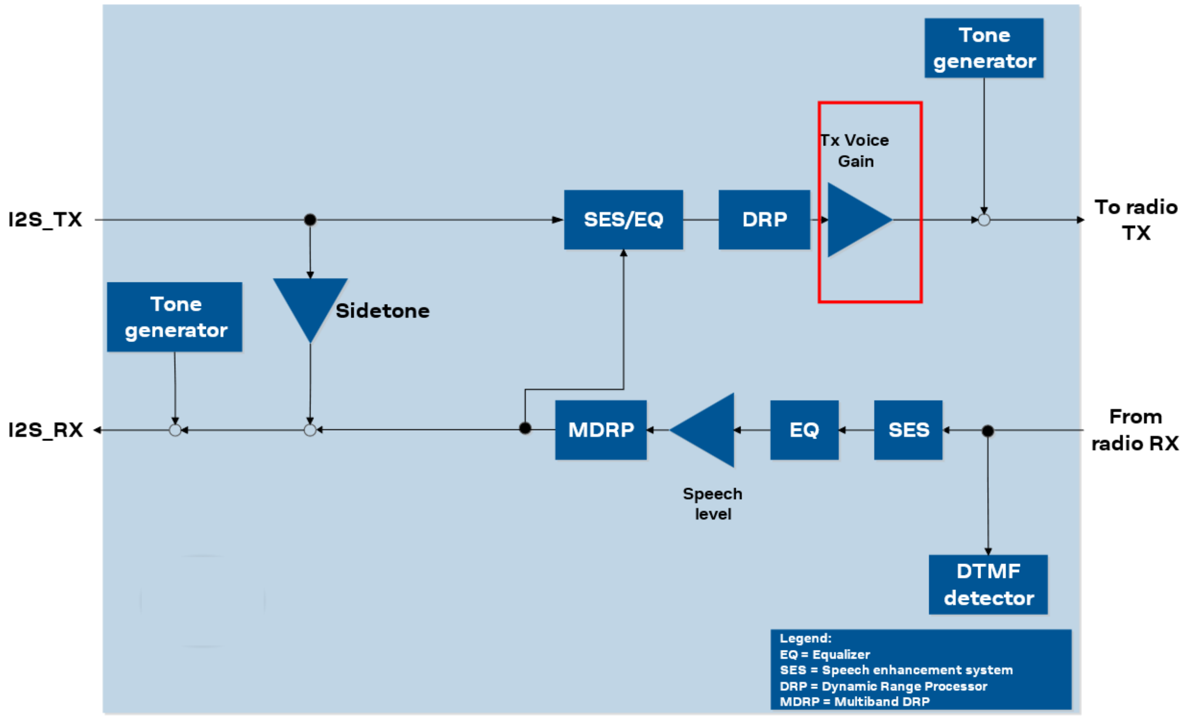

Figure 3: Muting acting after the SES block

Example:

Command	Response	Description
AT+CMUT=1	OK	Mute the uplink voice

3 Supervisory tones

3.1 Enabled supervisory tones

The modules support below supervisory tones:

- Ringing / ring back tone on mobile originated calls (free tone).
- Subscriber busy tone on mobile originated calls.
- Call waiting tone on mobile terminated calls.
- Ringer on mobile terminated calls (ringtone).
- Incoming SMS sound.

The tones played during the call are mixed with speech, except for the busy tone.

3.1.1 Ringing tone

During the alerting phase of an outgoing call, the ringing / ring back tone is generated, either reproduced locally or sent by the network (using in-band tones on 2G/3G calls or early media on VoLTE calls).

Alerting phase is also notified by the +UCALLSTAT: 1,3 URC.

☞ On LARA-R6 "00B" series and only for VoLTE calls, the URC is not sent when the ring back is generated by the network.

3.1.2 SMS alert sound mode

Whenever an SMS is received, the module generates a specific tone to notify the SMS reception.

The generation of this tone can happen in idle state or during an established voice call. It can be disabled by the +UMSM AT command.

It is also possible to enable URC to indicate the SMS reception and the text sent within an SMS notification using the AT commands below:

Command	Response	Description
`AT+CMGF=1`	`OK`	Indicate the format of URCs upon receiving SMS messages.
`AT+CNMI=1,2`	`+CMT: "+393470630039",,"22/05/17 ,11:23:41+08"` `Test SMS`	SMS-DELIVER indications are routed to the AT terminal by the +CMT URC.

The SMS alert sound can also be muted by the customer using the +UMSM AT command:

Command	Response	Description
`AT+UMSM=1`	`OK`	Mute the SMS alert sound.
`AT+UMSM=0`	`OK`	Un-mute the SMS alert sound (default value).

☞ When the SMS alert sound is enabled, i.e. +UMSM: 0, and the module is playing the sound, issuing the +UTGN AT command will return an error result code (+CME ERROR: operation not supported).

3.1.3 Alert sound mode

The ringer alert sound on mobile terminated calls and the SMS alert sound can both be muted using the +CALM AT command as below. The setting is saved in NVM.

Command	Response	Description
AT+CALM=1	OK	Mute both the ringer on MT calls and the SMS alert sound.
AT+CALM=0	OK	Un-mute both the ringer on MT calls and the SMS alert sound.

☞ In silent mode, i.e., `+CALM: 1`, the +UTGN command is fully supported.

☞ When silent mode is disabled, i.e., `+CALM: 0`, and SMS alert tones, waiting tones, or ringer on MT call are playing, issuing the +UTGN AT command will return an error result code (+CME ERROR: operation not supported).

☞ The silent mode, i.e., +CALM: 1, does not affect the status of the +UMSM AT command, which means that the +UMSM read command (AT+UMSM?) might indicate that the SMS alert sound is enabled. Therefore, the application code must not rely on +UMSM status to determine if SMS alert sound is active when silent mode is enabled.

4 Audio routing and profiles

One digital audio path is available, for which the speech is routed from the network to the I2S digital port.

The audio profiles are parameter settings of the audio processing blocks and other audio related parameters configurable by AT commands. They are stored in NVM. The audio profile consists of the speech profile (uplink/downlink) and the I2S profile.

There are four profiles available for the digital audio path:

- **Headset profile**: factory-programmed settings for handset/headset devices
- **Hands-free profile**: factory-programmed settings for hands-free devices
- **Flat profile**: no additional processing on the audio path, all the blocks are disabled.
- **Alarm panel profile**: factory-programmed settings for the alarm panel use case, e.g. when there is very high coupling between the mic and the speaker.

For more details concerning audio profile settings, see the section 8.

The combination of an audio path and an audio profile is called the **audio path mode**. The current audio path mode can be configured using the +USPM AT command. The first parameter sets the audio path type. LARA-R6/L6 series modules supports only the digital path (I2S), which is controlled by the +UI2S AT command. The second parameter specifies the profile type to use. For example:

Command	Response	Description
`AT+USPM=1,1`	`OK`	Set the digital audio path with hands-free profile.

The command can be used to switch the profile in run-time and during a voice call.

5 Player management

The +UTGN AT command generates:

- A custom tone with given frequency, duration, and volume, or
- A single DTMF tone given a DTMF character, duration, and volume.

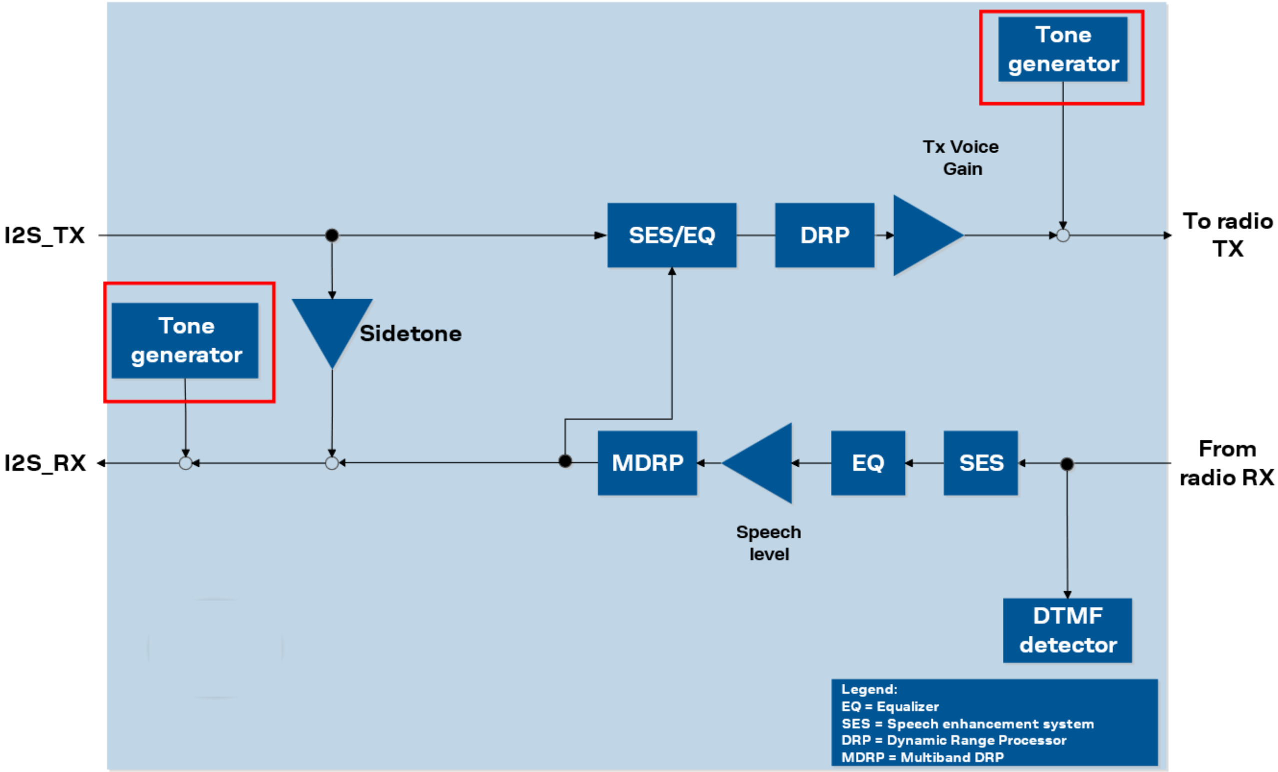

Figure 4: Tone generator used by tone players in the uplink and downlink directions

The +UTGN AT command starts a tone in uplink or downlink on the module tone generator. The frequency or single DTMF tone digit, duration, and volume of the tone must be set:

```
AT+UTGN=<frequency or tone>,<duration>,<volume>[,<UplinkSending>]
```

- The frequency range goes from 300 to 3400 Hz.
- The DTMF digit tones are one-character string values from " 0" – "9", " A"-" D", and "*", "#".
- The duration range goes from 50 to 1360 ms.
- The volume range goes from 0 to 100 where volume 0 means muted, 1 means -40 dBFS, and 100 means 0 dBFS.

The tone can be sent on the downlink or uplink path, configured using the <UplinkSending> parameter.

Command	Response	Description
AT+UTGN=1000,1000,100,1	OK	Start the generation of a 1000 Hz tone with duration of 1000 ms and volume 100, sent on the uplink path
AT+UTGN=?	+UTGN: (300-3400,"0"-"9","A"-"D","*","#"),(50-1360),(0-100),(0-1) OK	AT test command

When the tone stops, the +UUTGN URC is generated.

The tone playing can be stopped using the AT+UTGN=0,0 command. In this case the +UUTGN URC is not generated.

The tone generation is not affected by the “silent mode” (+CALM: 1).

If the +UTGN AT command is issued before the stop of a previous generated tone, i.e., before +UUTGN URC generation, the current tone generation will stop the previously generated tone. No error result code is sent in this case.

During a speech call the generated tone is not mixed with speech; speech is muted while the tone is playing.

The ringer on an incoming call, the alarm tones, and service tones (e.g., the call waiting tone) have priority over the tone generator (+UTGN). Since they are never muted and must be played, the tone generator (+UTGN) is stopped and +UUTGN URC will not be generated.

If the waiting tone and SMS alert tone are playing, the +UTGN AT commands cannot be sent. In this case an error result code (+CME ERROR: operation not supported) is returned.

5.1 Audio loopback by +UPAR/+USAR

The audio loopback for I2S pins and I2S configuration testing purposes is available using the +UPAR AT command. . The +UPAR AT command switch on the I2S HW and +USAR AT command stops the I2S HW.

The loop is implemented between I2S_TX and I2S_RX.

For example:

Command	Response	Description
`AT+UPAR=2,0,0`	`OK`	Start the audio loop, <audio_resource> is audio loop
`AT+UPAR=?`	`+UPAR: (2),(0-66),(0-255)` `OK`	AT test command

The audio loop cannot be issued during an established voice call. If issued, an error result code (+CME ERROR: operation not allowed) will be returned.

A MO or MT call automatically stops the audio loop when the call is established.

The +USAR AT command stops the audio loop as described below:

Command	Response	Description
`AT+USAR=2`	`OK`	Stop the audio loop, <audio_resource> is audio loop
`AT+USAR=?`	`+USAR: (2)` `OK`	AT test command

6 Speech codec management

For the use of AT commands related to speech codecs in combination with the DTMF detector, see section 9.3.2.

6.1 Speech codec information

The +USPEECHINFO AT command provides the speech codec related information and enables the corresponding +UUSPEECHINFO URC. The URC is issued each time the speech codec changes. The information text response to the read command and the URC are issued depending on the <mode> parameter configuration.

When <mode> is set to 1, the URC provides information during calls on 2G and 3G networks only. When <mode> is set to 2, the URC provides information during calls on 2G, 3G, and LTE networks. When <mode> is set to 3, VoLTE uplink codec information is not supported, making <mode>=2 and <mode>=3 equivalent.

The current bitrate indication is not supported for VoLTE codecs (value in URC is always 255).

Speech codec list:

- 0: codec Full Rate Adaptive Multi-Rate
- 1: codec GSM Enhanced Full Rate (12.2 kbit/s)
- 2: codec GSM Full Rate (13.0 kbit/s)
- 3: codec Half Rate Adaptive Multi-Rate
- 4: codec GSM Half Rate (5.6 kbit/s)
- 5: codec Full Rate Adaptive Multi-Rate Wideband
- 9: codec UMTS Adaptive Multi-Rate
- 10 : codec UMTS Adaptive Multi-Rate 2
- 11: codec UMTS Adaptive Multi-Rate Wideband
- 20: codec LTE AMR Narrowband
- 21: codec LTE AMR Wideband

Indication for <codec>=3 (Half Rate Adaptive Multi-Rate codec) is not supported, and it is always reported as <codec>=0 (Full Rate Adaptive Multi-Rate codec).

☞ If there's no audio activity on a channel (e.g.: microphone is disconnected), the speech codec information may not be available.

6.2 Speech codec configuration

6.2.1 Speech codec configuration 2G/3G

The +UDCONF=30 AT command configures the allowed speech codecs to be presented to the network during a voice call setup. The command does not affect VoLTE calls.

The factory-programmed value of <active_codec_bitmap> is 2089. The codecs that can be excluded are:

- Full Rate Adaptive Multi-Rate (FR AMR)
- Half Rate Adaptive Multi-Rate (HR AMR)
- Full Rate Adaptive Multi-Rate WideBand (FR AMR WB)
- UMTS Adaptive Multi-Rate WideBand (UMTS AMR WB)

The new setting is saved in NVM and a power cycle is required to apply the new configuration.

Example:

Command	Response	Description
`AT+UDCONF=30,1`	`OK`	Excludes AMR-WB codecs for 2G and 3G and Half Rate Adaptive Multi-Rate

6.2.2 Speech codec configuration LTE

The +USPEECHCFG AT command configures the allowed speech codecs to be presented to the network during a VoLTE call setup.

Only the AMR WB codec can be disabled, but both AMR WB and AMR NB can be configured with the desired set of bitrates to be used. Codec bitrate configuration is applied on MO calls only. If the configuration is not supported by the network, the call is immediately rejected.

☞ The bitrate mask cannot be applied after a codec change requested by the network during a call.

Examples:

Command	Response	Description
`AT+USPEECHCFG=21,0`	`OK`	Disables AMR-WB codecs
`AT+USPEECHCFG=21,1,510`		Enables AMR-WB for MO calls and excludes AMR-WB 6.60 kbit/s

By default, all AMR-WB and AMR-NB codecs are declared as supported to the network.

The new setting is saved in NVM and a power cycle is required to apply the new configuration.

☞ If the network does not support any of the selected codec modes, the call will not start. This means that if a codec mode has been selected for MT calls and the network does not support that mode, any received call would be discarded.

7 Digital audio interface / External codec management

The modules provide an I2S digital audio interface to connect an external audio device, e.g., an audio codec. The application processor (AP) shall manage the codec.

The digital I2S interface is described in the AT commands manual [1] and the system integration manual [5]. The module supports a single I2S interface configurable using the +UI2S AT command.

This section includes an example of the architecture for the module / external codec / AP system.

u-blox cellular modules support additional resources to manage the external codec:

- **+UMCLK** (master clock control): Provides the codec with a 12.288 MHz clock generated by the module.
- **+UI2S** (I2S control): Selects the most appropriate I2S configuration for the external codec. Two <i2s_mode> configurations are supported by LARA-R6/L6 series.
- **+UI2CO**, **+UI2CW**, **+UI2CR**, **+UI2CREGR**, **+UI2CC** (I2C control): Allow sending commands from the module to the codec through the modem I2C interface.

Figure 5 shows a possible architecture for the LARA-R6/L6 module / external codec / AP system. The HW implementation is highly simplified

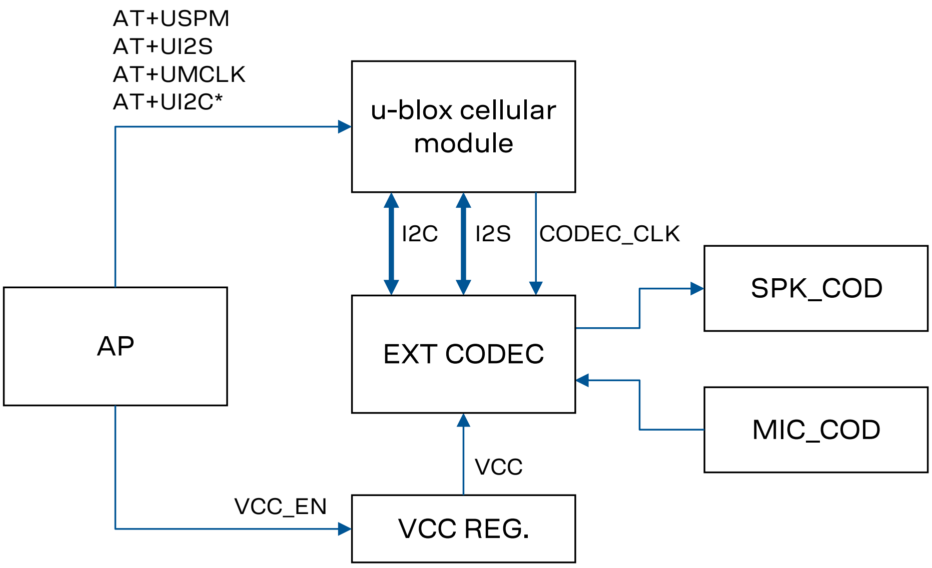

Figure 5: External codec management for LARA-R6/L6 series module

Section 7.1 shows an example of the codec management scenario based on this architecture. The examples are for the audio codec mounted in the evaluation board, i.e., the Maxim MAX9860 audio voice codec.

7.1 Examples of codec management

At system start-up, the application processor (AP) should enable the codec supply VCC provided by a voltage regulator (VCC REG.) via a dedicated VCC_EN pin. The external codec (EXT CODEC) is connected to a speaker (SPK_COD) and a microphone (MIC_COD).

The application processor configures the module and the codec with a sequence of AT commands

Command	Response	Description
`AT+UI2S=14,1,0,3,0`	`OK`	AT+UI2S=<I2S_mode>,<I2S_port>,<I2S_clk_wa>,<I2S_sample_rate>, <I2S_Master_Slave>

Command	Response	Description
		• <I2S_mode>=14: I2S standard mode • <I2S_port>=1: connect I2S 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 the 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=1,0`	`OK`	AT+USPM=<audio_path>,<profile_type> • <audio_path>=1: digital audio path via I2S port • <profile_type>=0: headset profile • Change audio path to I2S. 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 12.288 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 the clock does not need to be maintained while I2S is not running since only the voltage supply is needed to make I2C work. Be aware that other codecs may need to keep the clock running for register programming.
`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 I2C • <I2C_controller_number>=1: controller 1 • <bus_mode>=0: bus mode standard (0 – 100 kb) • <bit_rate>=0: I2C bit rate is 100 kbit/s • <device_address>=0x10: device address in HEX format; this address can be found in the coded datasheet • <address_width>=0: 7 bit address
`AT+UI2CW="0000000010A0003` `03000063300500000008A",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 configures the codec (gains, I2S configuration, clock configuration, etc.)
`AT+UI2CC`	`OK`	Close logical channel on I2C • The codec is now initialized and ready to work when the module starts audio activity, e.g., for audio test
`AT+UPAR=2,0,0`	`OK`	An audio loop can be started by command. The module enables a 12.288 MHz clock signal on the CODEC_CLK pin. I2S is enabled and starts to transmit and receive samples. The mic signal sent to the AFE is looped back and is played on the loudspeaker by the codec.
`AT+USAR=2`	`OK`	The command stops the audio loop. The module stops transmission of audio data on I2S and disables the 12.288 MHz clock signal

The codec supply VCC is maintained even when the codec is no longer in use.

To disable the master clock, issue the AT+UMCLK=1 command. AT+UMCLK=0 is not supported. To enable or disable the master clock, issue the +UMCLK AT command when no call is in progress, otherwise an error result code is issued.

7.1.1 External device configuration +UEXTDCONF

The +UEXTDCONF AT command configures an external device when the module boots up. The only supported external device is the Maxim MAX9860 audio codec.

The setting for MAX9860 codec is stored in the NVM and applied at each module power-on. The setting consists of codec enabling and a data string for codec register programming.

Procedure to enable the MAX9860 codec is:

Command	Response	Description
AT+UEXTDCONF=0,1	OK	• <device_id>=0: Maxim MAX9860 audio codec, connected via I2C • <configuration_enable>=1: enabled • It enables the external audio codec Maxim MAX9860; current <hex_data> string in NVM is maintained (Factory-programmed <hex_data>="0000000010A00030300018330050000008A") ☞ AT+UEXTDCONF=0,1 must not be used during call or while audio loop is enabled (by AT+UPAR=2,0,0) ☞ On LARA-R6 "00B": Factory-programmed value is different: <hex_data>="0000000010A00030300006330050000008A"
AT+CFUN=16	OK	Reboot the module

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

To test if the audio subsystem is correctly configured, execute the sequences below at any time, starting an audio loop between the mic and speaker signal.

Command	Response	Description
AT+UPAR=2,0,0	OK	Start an audio loop. The module enables a 12.288 MHz clock signal on CODEC_CLK pin. I2S is enabled and starts to transmit and receive samples. The microphone signal sent to the AFE is looped back and is played on the loudspeaker by the codec.
AT+USAR=2	OK	Stop the audio loop. The module stops transmission of audio data on I2S and disable 12.288 MHz clock signal

If <configuration_enable>=1, at every system boot, the module performs the actions corresponding to the following commands:

Command	Description
AT+UMCLK=2,0	Set the external codec master clock to 12.288 MHz.
AT+UI2CO=1,0,0,0x10,0	Open the I2C logical channel (connected to the Maxim MAX9860 external codec). The I2C address for the MAX9860 audio codec device is hard-coded in the module's firmware.
AT+UI2CW = <hex_data>,18	Send, via I2C, the <hex_data> string stored in NVM to Maxim MAX9860. The string must start with 00 followed by hex values to be written in codec's register from address 0x00 to 016x.
AT+UI2CC	Close the I2C logical channel.

This sequence of commands forces the MCLK to be enabled and programs the codecs registers.

The sequence is also sent if <configuration_enable> changes from 0 to 1, without the need to reboot the module.

There are two ways to enable the codec. If the codec power supply is always on, the +UEXTDCONF AT command can be used to enable the codec. Instead, if the codec power supply can be turned off, the +UI2C AT commands must be issued to enable the codec every time the codec is turned on, as in section 7.1. This is mandatory because the codec register settings are lost after a power off.

The +UI2S AT command default setting is:

- <I2S_mode>=14: I2S standard modem, i.e., CLK edge TX/RX falling/raising, 1 MSB delay, RX channel WA LOW)
- <I2S port>= 1: only 1 I2S port is supported
- <I2S clk wa>= 0: dynamic mode
- <I2S_sample_rate>=3: 16 kHz sampling rate

- <I2S_Master_Slave>=0: master mode

This setting is compliant with the MAX9860 default setting (normal I2S mode).

The gains set for the Maxim MAX9860 in the +UI2CW AT command are optimized for the headset included in the evaluation kit.

The AT+UEXTDCONF=0,2 command can be issued to configure the Maxim MAX9860 audio codec without enabling the master clock. This is useful if a quartz oscillator is used instead of the MCLK signal.

Command	Response	Description
AT+UEXTDCONF=0,2	OK	• <device_id>=0: Maxim MAX9860 audio codec, connected via I2C • <configuration_enable>=2: enabled; the +UMCLK AT command setting stored in NVM is maintained It enables the Maxim MAX9860 external audio codec; current <hex_data> string in NVM is maintained
AT+CFUN=16	OK	Power off the module

If <configuration_enable> changes from 0 to 2, the module performs the actions corresponding to the following commands at every boot-up:

Command	Description
	+UMCLK setting stored in NVM is maintained
AT+UI2CO=1,0,0,0x10,0	Open the I2C logical channel (connected to the Maxim MAX9860 external codec)
AT+UI2CW=<hex_data>,18	Send, via I2C, the <hex_data> string stored in NVM (for the Maxim MAX9860 external codec configuration)
AT+UI2CC	Close the I2C logical channel

7.1.1.1 Examples on +UEXTDCONF usage

Below are examples of +UEXTDCONF AT command:

Command	Response	Description
Reset parameters to factory and resets the module		
AT+UEXTDCONF=0	OK	Resets parameters to factory settings. Default <hex_data> string is saved in NVM.
AT+UMCLK=1	OK	Set MCLK pin output steady low. ☞ Setting AT+UEXTDCONF=0,1 forces at every start-up +UMCLK: 2,0 in NVM. To undo /remove this setting, explicitly issue the AT+UMCLK=1 command after the start-up. When resetting to factory settings by means of the command AT+UEXTDCONF=0, the master clock mode is not automatically forced back to 1.
AT+CFUN=16	OK	Reset the module.
AT+UEXTDCONF?	+UEXTDCONF: 0,0,"0000000010A00030300018330500000008A" OK	At boot, the external audio codec is not enabled(default) String <hex_data> remains the last saved value (default). ☞ On LARA-R6 "00B" the response is +UEXTDCONF: 0,0,"0000000010A000303000063300500000008A" OK
AT+UMCLK?	+UMCLK: 1,0 OK	+UMCLK setting is not changed; I.e. MCLK is disabled.
Enable the external codec and reset the module		

Command	Response	Description
AT+UEXTDCONF=0,1	OK	Enable external codec MAX9860. <hex_data> string is not changed.
AT+CFUN=16	OK	Reset the module.
AT+UEXTDCONF?	+UEXTDCONF: 0,1,"0000000010A00030300018330500000008A" OK	At boot, the external audio codec is enabled and it is programmed via I2C.String <hex_data> remains the last saved value. On LARA-R6 "00B" the response is +UEXTDCONF:0,1,"0000000010A00030300006330050000008A" OK
	+UMCLK: 2,0 OK	If the external codec is enabled, MCLK is enabled at boot
Check correspondence of MAX9860 registers with the <hex_data> string		
AT+UI2CO=1,0,0,0x10,0	OK	Open the I2C logical channel (connected to the Maxim MAX9860 external codec)
AT+UI2CREGR=0x00,17	+UI2CREGR: 0: 0x0 +UI2CREGR: 1: 0x0 +UI2CREGR: 2: 0x0 +UI2CREGR: 3: 0x10 +UI2CREGR: 4: 0xA0 +UI2CREGR: 5: 0x0 +UI2CREGR: 6: 0x30 +UI2CREGR: 7: 0x30 +UI2CREGR: 8: 0x0 +UI2CREGR: 9: 0x18 +UI2CREGR: 10: 0x33 +UI2CREGR: 11: 0x0 +UI2CREGR: 12: 0x50 +UI2CREGR: 13: 0x0 +UI2CREGR: 14: 0x0 +UI2CREGR: 15: 0x0 +UI2CREGR: 16: 0x8A OK	Read 17 registers MAX9860 starting from address 0x00; hex values match with those in sequence in <hex_data> string (the first byte in the string is the address 0x00 of the first register). On LARA-R6 "00B" the response for register 9 is: +UI2CREGR: 9: 0x6 For details about MAX9860 registers settings, see the datasheet [6]
AT+UI2CC	OK	Close the I2C logical channel.
Enable the codec and power off the module		
AT+UEXTDCONF=0,1	OK	Enable the MAX9860 external codec. <hex_data> string is not changed.
AT+CPWROFF	OK	At next module power on, the module will program and enable the external codec and the scenario will be as in the example with module reset by AT+CFUN=16.
Reset parameters to factory and configure the codec		
AT+UEXTDCONF=0	OK	Resets parameters to factory settings. Default <hex_data> string is saved in NVM.
AT+UEXTDCONF=0,1	OK	If <configuration_enable> changes from 0 to 1, the module configures the MAX9860 external codec with <hex_data> in NVM, without need of reset.
AT+UMCLK?	+UMCLK: 2,0 OK	If <configuration_enable> changes from 0 to 1, MCLK is enabled.

7.1.1.2 Audio profile configuration

Command	Response	Description
Configure the MAX9680 for Alarm panel profile		
AT+USPM=1,3	OK	Set Alarm Panel profile
AT+UI2S=	OK	Set I^{2}S parameters to factory-programmed values (Normal I^{2}S mode)
AT+UEXTDCONF=0,0	OK	Disable MAX9860 external codec. <hex_data> string is not changed.

Command	Response	Description
AT+UEXTDCONF=0,1,"0000000010A0003030001833004E0000008A"	OK	If <configuration_enable> changes from 0 to 1, the module configures the MAX9860 external codec with new <hex_data> without need of reset. Register 12 is set to 0x4E, which sets external codec microphone gain for Alarm Panel profile (For details on MAX9860 registers settings, see the datasheet) ☞ On LARA-R6 "00B" the command should be: AT+UEXTDCONF=0,1,"0000000010A0003030000633004E0000008A"

7.1.1.3 Typical I2S configurations

This section collects examples on how to properly configure both the module and the MAX9680 audio codec for some typical I2S configurations.

Command	Response	Description
Normal I2S mode at 48 kHz, with module as master		
AT+UEXTDCONF=0,0	OK	Disable MAX9860 external codec. <hex_data> string is not changed.
AT+UEXTDCONF=0,1,"0000000010A000303000183300500000008A"	OK	If <configuration_enable> changes from 0 to 1, the module configures the MAX9860 external codec with new <hex_data> without need of reset. For details on register settings refer to MAX9860's datasheet.
AT+UI2S=14,1,0,8,0	OK	Set LARA-R6's I2S to normal I2S mode, 48 kHz, master. ☞ On LARA-R6 "00B" 48 kHz is not supported.
PCM mode at 48 kHz, with module as master		
AT+UEXTDCONF=0,0	OK	Disable MAX9860 external codec. <hex_data> string is not changed.
AT+UEXTDCONF=0,1,"0000000010A000040000183300500000008A"	OK	If <configuration_enable> changes from 0 to 1, the module configures the MAX9860 external codec with new <hex_data> without need of reset. For details on register settings refer to MAX9860's datasheet.
AT+UI2S=30,1,0,8,0	OK	Set LARA-R6's I2S to PCM mode, 48 kHz, master. ☞ On LARA-R6 "00B" 48 kHz is not supported.
PCM mode at 16 kHz, with module as master		
AT+UEXTDCONF=0,0	OK	Disable MAX9860 external codec. <hex_data> string is not changed.
AT+UEXTDCONF=0,1,"0000000010A000040000183300500000008A"	OK	If <configuration_enable> changes from 0 to 1, the module configures the MAX9860 external codec with new <hex_data> without need of reset. Register 6 is set to 0x04, which sets external codec in PCM mode-short sync. ☞ On LARA-R6 "00B" the command should be: AT+UEXTDCONF=0,1,"0000000010A000040000063300500000008A"
AT+UI2S=30,1,0,3,0	OK	Set LARA-R6's I2S to PCM mode, 16 kHz, master
Normal I2S mode at 16 kHz, with module as slave		
AT+UEXTDCONF=0,0	OK	Disable MAX9860 external codec. <hex_data> string is not changed.
AT+UEXTDCONF=0,1,"00000000102000B03600183300500000008A"	OK	If <configuration_enable> changes from 0 to 1, the module configures the MAX9860 external codec with new <hex_data> without need of reset. For details on register settings refer to MAX9860's datasheet. ☞ Due to divider limitation, I2S_CLK generated by MAX9860 cannot be set to 512 kbps (32 bits per frame). This example set it to 1536 kbps (96 bits per frame).
AT+UI2S=14,1,0,3,1	OK	Set LARA-R6's I2S to normal I2S mode, 16 kHz, slave. ☞ On LARA-R6 "00B" slave mode is not supported.

Command	Response	Description
PCM mode at 16 kHz, with module as slave and external 16.384 MHz MCLK.		
`AT+UEXTDCONF=0,0`	`OK`	Disable MAX9860 external codec. <hex_data> string is not changed.
`AT+UEXTDCONF=0,1,"00000000306000A42700183300500000008A"`	`OK`	If <configuration_enable> changes from 0 to 1, the module configures the MAX9860 external codec with new <hex_data> without need of reset. For details on register settings refer to MAX9860's datasheet. Due to divider limitation, I2S_CLK generated by MAX9860 cannot be set to 256 kbps (16 bits per frame) using 12.288 MHz factory MCLK. Just for testing purpose, a different master clock signal can be used; e.g. using 16.384 MHz an I2S clock signal at 256 kHz can be generated.
`AT+UI2S=30,1,0,3,1`	`OK`	Set LARA-R6's I2S to PCM mode, 16 kHz, slave. On LARA-R6 "00B" slave mode is not supported.
`AT+UMCLK=1`	`OK`	MAX9860 cannot be configured in master mode, PCM 16kHz dividing 12.288 MHz factory MCLK signal. Just for testing purpose, it can be provided an external master clock, e.g. 16.384 MHz. In this case, set MCLK pin output steady low.
Normal I2S mode at 48 kHz, with module as slave		
`AT+UEXTDCONF=0,0`	`OK`	Disable MAX9860 external codec. <hex_data> string is not changed.
`AT+UEXTDCONF=0,1,"00000000106000B03600183300500000008A"`	`OK`	If <configuration_enable> changes from 0 to 1, the module configures the MAX9860 external codec with new <hex_data> without need of reset. For details on register settings refer to MAX9860's datasheet.
`AT+UI2S=14,1,0,8,1`	`OK`	Set LARA-R6's I2S to normal I2S mode, 48 kHz, slave. On LARA-R6 "00B" 48 kHz and slave mode are not supported.
PCM mode at 48 kHz, with module as slave		
`AT+UEXTDCONF=0,0`	`OK`	Disable MAX9860 external codec. <hex_data> string is not changed.
`AT+UEXTDCONF=0,1,"00000000106000A42700183300500000008A"`	`OK`	If <configuration_enable> changes from 0 to 1, the module configures the MAX9860 external codec with new <hex_data> without need of reset. For details on register settings refer to MAX9860's datasheet.
`AT+UI2S=30,1,0,8,1`	`OK`	Set LARA-R6's I2S to PCM mode, 48 kHz, slave. On LARA-R6 "00B" 48 kHz and slave are not supported.

8 Speech enhancement

This section shows the speech enhancement system (SES).

The module is delivered with programmed profiles tuned for specific use cases such as:

- **Headset**: Profile with mild echo cancellation settings to cope with low coupling between microphone and speaker signal.
- **Handsfree**: Profile with average echo cancellation settings to cope with mid coupling between microphone and speaker signal conforming to 3GPP TS 26.131 for desktop devices[1].
- **Flat**: All algorithm blocks, e.g., echo canceller and noise suppressor, are disabled.
- **Alarm panel**: Profile with strong echo cancellation settings to cope with high coupling between the microphone and speaker signals. It has been tested on an alarm panel scenario defined by internal requirements. It also enables downlink NR algorithm block if the far-end call center does not remove background noise on the signal captured by the far-end operator microphone.

These profiles can be selected as described in section 4.

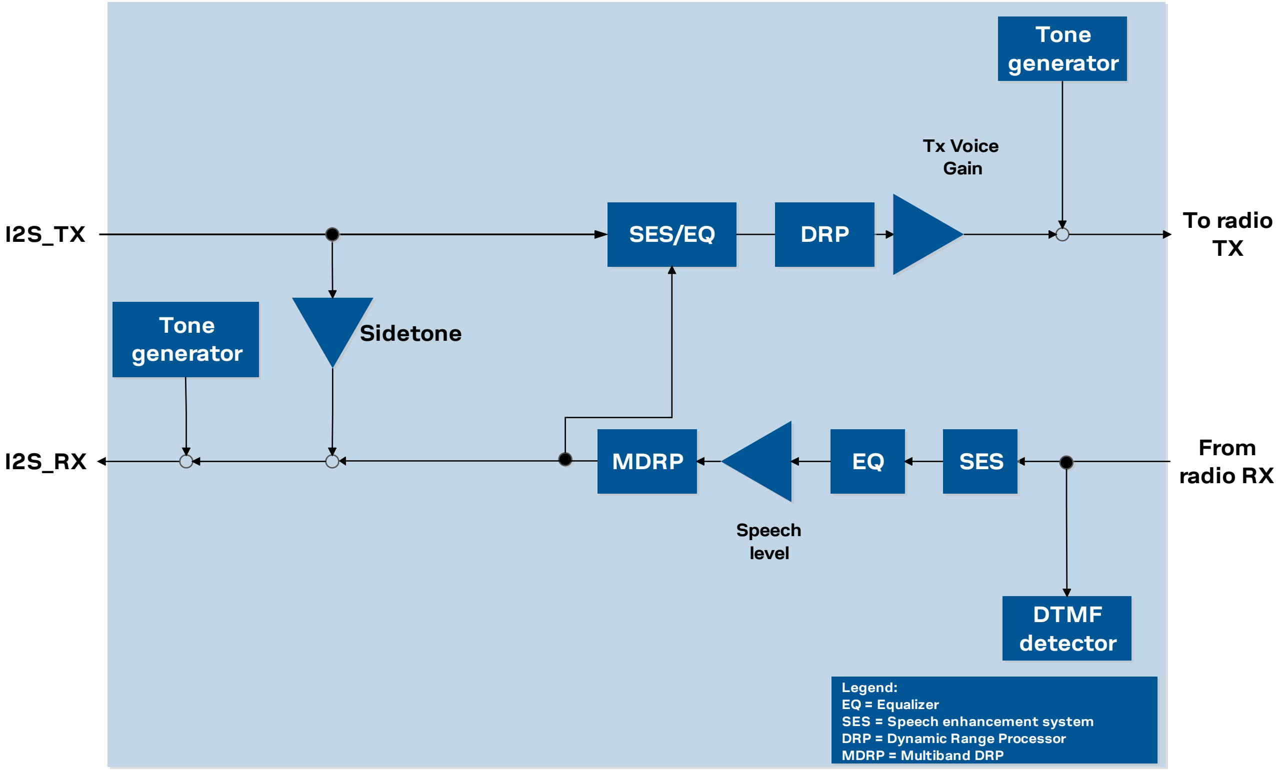

Figure 6: LARA-R6 / LARA-L6 speech enhancement system

8.1 SES processing blocks

The SES blocks perform different kinds of speech processing, which are:

- In uplink, high pass filters (HPFs) acoustic echo canceller (AEC), single microphone noise suppression (NR), single band dynamic range processing (DRP) and adaptive input gain (AIG).
- In downlink, far-end noise suppression (NR), adaptive input gain (AIG) and single band dynamic range processing (DRP).

[1] Conformance test passed on reference device in u-blox audio laboratory.

Each profile exhibits different tuning parameters for the SES blocks to cope with different requirements (e.g., SLR, max RLR, $TCL_{w.}$ etc.) associated to each profile.

The tables below show which speech processing blocks are enabled / disabled for the supported profile for each direction uplink and downlink.

Profile	HPF	AEC	NR	EQ	AIG	DRP
Headset	ON	ON	ON	ON	OFF	OFF
Hands-free	ON	ON	ON	ON	OFF	ON
Flat	OFF	OFF	OFF	OFF	OFF	OFF
Alarm Panel	ON	ON	ON	ON	OFF	ON

Table 2 SES blocks ON/OFF in uplink direction for LARA-R6 “00B” and LARA-L6 “00B” product versions

Profile	HPF	AEC	NR	EQ	AIG	DRP
Headset	ON	ON	ON	ON	OFF	OFF
Hands-free	ON	ON	ON	ON	OFF	ON
Flat	OFF	OFF	OFF	OFF	OFF	OFF
Alarm Panel	ON	ON	ON	ON	ON	ON

Table 3 SES blocks ON/OFF in uplink direction for LARA-R6 “01B” and LARA-L6 “01B” product versions

Profile	DRP	HPF	EQ	AIG	NR
Headset	ON	ON	ON	ON	OFF
Handsfree	ON	ON	ON	ON	OFF
Flat	OFF	OFF	OFF	OFF	OFF
Alarm Panel	ON	ON	ON	ON	ON

Table 4 SES blocks ON/OFF in downlink direction for LARA-R6 and LARA-L6 series

8.1.1 Microphone gain automatic adaptation

Except LARA-R6 and LARA-L6 “00B” product versions, the alarm panel profile integrates an adaptive input gain (**AIG**) on the uplink path to optimize the microphone speech level to the distance of the near-end talker in the range of 35 cm to 250 cm.

8.1.2 Support for external speech enhancement

If the external audio codec connected on I2S interface supports its own SES (i.e. AEC), the internal SES must be disabled. To disable the AEC and all other speech enhancement blocks, select the flat profile using the +USPM AT command with the audio flat profile as below:

Command	Response	Description
`AT+USPM=1,2`	`OK`	Flat profile selection, saved into NVM.

☞ The disabling of a single SES block is not supported.

8.2 SES tuning for customer’s audio device

The profiles can be tuned and customized by u-blox upon customer request to meet the customer’s audio device characteristics and audio quality requirements.

Usually tuning of the customer device requires that a complete prototype is delivered to u-blox audio laboratory. This team will perform an accurate analysis to characterize the audio performance of the device; based on the results, a specific audio configuration will be designed.

Result of tuning is a binary package (i.e. audioconfig.mbn file) that the user must download on the devices via the +UDWNFILE AT command. The download operation substitutes all the profiles parameters and I2S configuration.

For the most common use cases, predefined .mbn audio configuration file are available.

☞ To get the appropriate .mbn audio configuration file for your use case or ask for specific tuning of your device, contact the u-blox office or sales representative.

⚠ Analysis and tuning of audio performances require the availability of a USB port. Thus when designing any device for audio applications, it is strongly recommended to allow access to LARA USB port (e.g test points on LARA USB pins and possibility to disconnect / switch off the host processor).

The package is downloaded as shown in the table below:

Command	Response	Description
`AT+UDWNFILE="audioconfig.mbn",196648,"AUDIO_EXT"`	`OK`	Store the audioconfig.mbn file and apply the 'AUDIO_EXT' tag.

☞ Download operation requires room for the audio configuration file in the USER filesystem (about 200kbytes).

Downloading the audio configuration file will result in the following:

- After a successful download, the audio configuration is validated. If it is not valid (i.e., the initial tag used in the file is not the expected one), the "+CME ERROR: operation not supported" error result code will be provided, the downloaded file will be closed and deleted from USER folder, and the audio configuration file version number verification by +UTI="uaud_save_data?" will show the original version with checksum OK.
- The download validation procedure also performs a file checksum verification at the end of the downloading procedure. If this fails, the "+CME ERROR: operation not supported" error result code will be provided, and the result of this check is stored in NVM.
- The checksum result can be always shown by the AT+UTI="uaud_save_data?" command. If this shows a failure, audio quality cannot be guaranteed since the downloaded audio configuration file is corrupted. In that case it is strongly recommended to restore the original firmware (e.g., with FOTA or EasyFlash; for more details, see LARA-R6/L6 series FW update application note [7]).
- After successful application, the module needs to be restarted to activate the new audio configuration.
- The factory-programmed configuration is not backed up. It can be restored by downloading the original firmware.
- The downloaded file is deleted after processing is successfully completed and can, therefore, not be read out.
- Firmware update through FOTA or EasyFlash will overwrite any previous audio configuration file downloaded using the +UDWNFILE AT command.

The examples below show the usage of +UTI command for checking and setting the version and checksum stats:

Command	Response	Description
`AT+UTI="UAUD_SAVE_DATA:<label>"`	`OK`	Save audio configuration version number for the audio configuration file in NVM.
`AT+UTI="UAUD_SAVE_DATA?"`	`Parameters label: 1.02, Checksum: 1` `OK`	Display the current audio configuration and checksum result of the downloaded audio configuration file using the +UDWNFILE AT command.

8.2.1 Binary package download Over-The-Air (OTA)

Alternatively to package installation through AT+UDWNFILE command, the customer can download the package in the user file system through FTP application and install the package with a dedicated AT command. Contact u-blox support for further information.

8.2.2 Extended tuning AT commands (+UTI)

Simple patches of a profile are possible through UTI AT command. Patches are provided by u-blox upon customer request. Example of simple patch is disabling of a SES block (e.g. noise reduction) or calibration of echo canceller strength. The patch is not stored in NVM and it has to be applied at each boot.

The patches are useful for fast device tuning on the customer's site and preliminary verification of the patches to be applied in the audio configuration binary package that u-blox will deliver to customer.

⚠ The patches can be used only for the product release/firmware version for which have been generated.

8.3 Error reporting URCs

Except LARA-R6 and LARA-L6 "00B" product versions, the audio subsystem errors are reported by URCs. The URCs cannot be disabled.

URCs by Audio Subsystem	Indication meaning
`+UUAUDE: 0`	Success. The operation completed, and there were no errors
`+UUAUDE: -1`	General failure
`+UUAUDE: -2`	Invalid operation parameters
`+UUAUDE: -3`	Unsupported routine or operation
`+UUAUDE: -4`	Unexpected problem was encountered
`+UUAUDE: -5`	Unhandled problem occurred
`+UUAUDE: -6`	Unable to allocate resources
`+UUAUDE: -7`	Invalid handle
`+UUAUDE: -8`	Operation is already processed
`+UUAUDE: -9`	Operation is not ready to be processed
`+UUAUDE: -10`	Operation is pending completion
`+UUAUDE: -11`	Operation cannot be accepted or processed
`+UUAUDE: -12`	Operation aborted due to an error
`+UUAUDE: -13`	Operation was preempted by a higher priority
`+UUAUDE: -14`	Operation requires intervention to complete
`+UUAUDE: -15`	Operation requires immediate intervention to complete
`+UUAUDE: -16`	Operation is not implemented
`+UUAUDE: -17`	Operation requires more data or resources
`+UUAUDE: -18`	Operation is a local procedure call
`+UUAUDE: -19`	Operation timed out
`+UUAUDE: -20`	Driver did not find the handle or an internal resource required to process the request
`+UUAUDE: -21`	Operation cannot proceed due to an improper state
`+UUAUDE: -22`	Audio subsystem returns error status
`+UUAUDE: -23`	Audio subsystem is in reset

9 DTMF detector

9.1 Introduction to DTMF decoder

9.1.1 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 [2] specifies how to apply DTMF signaling to transmitters and receivers. It conforms to the International Telecommunication Union (ITU-T) recommendation Q.23 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.

9.1.2 About DTMF

The dual-tone multi-frequency signaling is a standard in telecommunication systems. In the DTMF scheme, a telephone is equipped with a keypad as shown in Figure 7. 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.

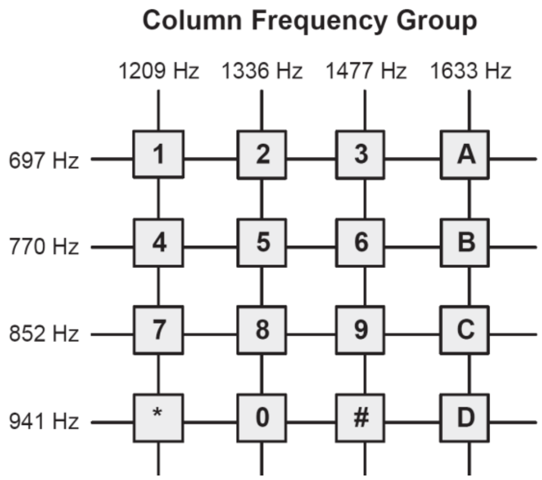

Figure 7 Touch-Tone telephone keypad: a row and a column tone is associated with each digit

9.1.3 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.

9.2 Implementation

The LARA-R6/L6 series module performs DTMF detection on the RX speech channel when the DTMF detector is enabled. Enable the DTMF detector/RTP decoder feature using the +UDTMFCFG the AT command:

```
AT+UDTMFCFG="algo","default"
```

The DTMF detector is started at each call setup. During the call, the DTMF detector/RTP decoder provides URCs for each detected digit. In case DTMFs are received as RTP events (VoLTE calls only), two indications are provided, one for the key press (RTP start event) and optionally one for the key release (RTP stop event).

URC	Description
`+UUDTMFD: A,1`	The digit "A" has been detected, received as in-band tone.
`+UUDTMFD: 4,0`	In VoLTE only: the RTP start event indicating the receiving of digit "4" is started
`+UUDTMFD: 4,2`	In VoLTE only: the RTP stop event indicating the receiving of digit "4" is finished

When the DTMF detector is enabled, URCs with information about in-band/RTP detection are enabled by default without the RTP stop event indication. For URCs configuration and more examples, refer to Section 9.5.

The DTMF detector can be enabled and disabled in any call state.

9.3 In-Band DTMF

9.3.1 Detection rate

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.

Here are some tips on tone characteristics to ensure the best detection performance.

- A low amplitude is preferred, e.g. -26 dBm instead of -6 dBm.
- Long tones, e.g. 200 ms, are better detected. A duration of less than 80 ms may lead to missed detections.
- A pause between two tones is required to avoid double detections.

Short tones can be missed when a low bit rate codec is selected, e.g., AMR-NB 4.75 kbit/s. So, when possible, it is suggested to restrict the codec set.

9.3.2 Codec set restrictions

The +UDCONF=30 AT command configures the allowed speech codec to be presented to the 2G/3G network during a voice call setup. After the voice call setup, the +UUSPEECHINFO URC notifies the actual codec used.

If issued during a call, the command cannot force the use of a new codec.

Below is an example of speech codec configuration and notification.

Command	Response	Description
`AT+USPEECHINFO=1`	`OK`	Enable +UUSPEECHINFO URC

Command	Response	Description
`ATD1234;`	`OK`	Make a call
	`+UUSPEECHINFO: 4`	GSM Half Rate (5.6 kbit/s) codec is used
`ATH`	`OK`	Terminate the call
`AT+UDCONF=30,0`	`OK`	Select Full rate Adaptive Multi-Rate codec
`ATD1234;`	`OK`	Make a call
	`+UUSPEECHINFO: 0`	Full rate Adaptive Multi-Rate codec is used

For the complete description of +UDCONF=30 and +USPEECHINFO AT commands for speech codec configuration and notification, see the u-blox LARA-R6/L6 AT commands manual [1].

9.3.2.1 Suggestions for 2G/3G

Low bit rate codecs (e.g., AMR-WB 6.6 kbit/s) introduce distortions on DTMF signals. Therefore, it is suggested to restrict the codec set to exclude Full Rate Adaptive Multi-Rate Wideband, UMTS Adaptive Multi-Rate Wideband, and Half Rate Adaptive Multi-Rat codecs using the AT+UDCONF=30 command.

☞ Half Rate Adaptive Multi-Rat codec introduces distortions on DTMF signals. It is not notified by the +UUSPEECHINFO URC but can be excluded by AT+UDCONF=30 command.

Command	Response	Description
`AT+UDCONF=30,1`	`OK`	Exclude AMR-WB codecs for 2G and 3G and Half Rate Adaptive Multi-Rat

9.4 VoLTE DTMF

RTP DTMF are DTMF tones sent as RTP events following the RFC 4733 [6].

DTMF tones are encoded into RTP packets with specially marked payloads – named telephone events- (NTEs) – carrying the pressed digit info and sent in the RTP stream.

The first packet must have the marker bit “M” set to 1. All the packets for the same event must have the same timestamp, and the sequence number must be the same as that of the regular audio channel.

The event duration is increased in each packet by the packetization period specified during call setup.

The last packet of the event, with bit “E” set to 1, is sent three times to avoid missing the end of the event if a packet is lost and has the final duration of the event.

In Table 5, the first packet of DTMF digit “9” is sent after 50 ms (packetization time) from the start of the event. Marker bit “M” is set to 1, the end bit “E” is set to 0, and the duration is increased by 400 timestamp units (50 ms).

Time (ms)	Event	M bit	Timestamp	Seq no	Event Code	Duration	E bit
0	“9” starts						
50	RTP packet 1 sent	“1”	0	1	9	400	“0”
100	RTP packet 2 sent	“0”	0	2	9	800	“0”
150	RTP packet 3 sent	“0”	0	3	9	1200	“0”
200	RTP packet 4 sent	“0”	0	4	9	1600	“1”
250	RTP packet 4 sent	“0”	0	5	9	1600	“1”
300	RTP packet 4 sent	“0”	0	6	9	1600	“1”

Table 5 – RTP DTMF example

Subsequent packets do not have either the “M” or the “E” bit set.

The final packet has the end bit “E” set to 1, is sent three times, and the duration of the event is the same in every final packet (1600 timestamp units – 200 ms).

9.4.1 DTMF RTP decoder

The DTMF RTP decoder detects the DTMF event from the RTP stream and sends an indication as soon as the first packet of a new event is detected. If the first packet is lost, the indication is sent when a packet with a new timestamp arrives.

The decoder and the URCs on event detection are enabled together with the in-band DTMF detector (see section 9.3).

9.4.2 DTMF regenerator

When a DTMF tone is sent as an RTP event during a VoLTE call, the audio packets are replaced by the DTMF event packets, resulting in silence on the receiver side.

The AT+UDTMFCFG="regen" command configures the audio regeneration of the DTMF tones locally on the loudspeaker:

```
AT+UDTMFCFG="regen",<mode>
```

Set <mode>=1 to enable the feature.

The DTMF tones are generated using the Generic PCM Player, so this feature and the DTMF generator can be enabled at the same time.

Since the DTMF generator only supports durations of multiples of 20 ms, the event duration is automatically adapted (e.g. an RTP events of 50 ms is reproduced as a tone of 60 ms).

9.5 URCs

The URCs can be controlled with the AT+UDTMFCFG="urc",<enable_urc> command:

Command	Response	Description
AT+UDTMFCFG="urc",0	OK	URC disabled.
AT+UDTMFCFG="urc",1	OK	URCs indicate only in-band tones
AT+UDTMFCFG="urc",2	OK	URCs indicate in-band tones and RTP start event
AT+UDTMFCFG="urc",3	OK	URCs indicate in-band tones and RTP start and stop events

The AT+UDTMFCFG="urc",2 command enables indications related to the in-band or RTP decoding as shown in the following table:

URC	Indication meaning
+UUDTMFD: 4,1	In-band tone detected
+UUDTMFD: 4,0	RTP start event decoded (key pressed)

Table 6: URC examples of AT+UDTMFCFG="urc",2

The AT+UDTMFCFG="urc",3 command enables also the RTP stop event:

URC	Indication meaning
+UUDTMFD: 4,1	In-band tone detected
+UUDTMFD: 4,0	RTP decoding, RTP start event (key pressed)
+UUDTMFD: 4,2	RTP decoding, RTP stop event (key released)

Table 7: URC examples of AT+UDTMFCFG="urc",3

The tone duration may be calculated, only in VoLTE RTP case, as the time between the reception of URCs indicating start and stop events of the same digit.

10 DTMF generator

The DTMF generator (+UTGN) is used to play a single DTMF tone on the uplink or downlink path.

By default, DTMF tones are injected just before the AMR encoder on the uplink path (tones are not affected by the speech enhancement system) or after AMR decoder on the downlink path (tones are affected by the speech enhancement system).

Supported DTMF tones are:

- Tones '0' – '9',
- Tones '*' and '#',
- Tones 'A' – 'D'.

A single DTMF tone can last from 50 ms to 1360 ms. For example:

Command	Response	Description
`AT+UTGN="A",1000,90,0`	`OK`	Inserted tone length is 1000 ms with volume 90% dBFS in downlink.

In downlink, the tone generation can also be performed in IDLE mode. In uplink it can only be performed during an established call, otherwise an error result code (+CME ERROR: operation not allowed) is returned.

A playing tone can be stopped using the AT+UTGN=0,0,0 command.

The generation command does not block the AT interface. The completion of the generation is acknowledged by the +UUTGN:0 URC. If the tone generation is stopped using the AT+UTGN=0,0,0 command, no +UUTGN URC generation will occur.

If the generation of a tone is issued before +UUTGN URC generation of the current tone, the current tone will be stopped, its URC will not be issued, and the newly generated tone will be played. For example, in the sequence of AT commands below, if the `AT+UTGN="#",200,60,0` command is issued before the URC of tone '0' was received, the '0' tone will stop playing.

The example below shows the generation of a sequence of 16 tones with a duration of 200 ms and a volume of 60% dBFS in downlink.

Command	Response	Description
`AT+UTGN="0",200,60,0`	`OK`	Send 1 DTMF tone of duration 200 ms, volume 60, on uplink path.
	`+UUTGN: 0`	Tone is sent in uplink (received 220 ms after the command is issued)
	`Pause for 100 ms`	
Repeat the AT+UTGN command to generate tones: 2,3,4,5,6,7,8,9,0,A,B,C,D,*		
`AT+UTGN="#",200,60,0`	`OK`	Send 1 DTMF tone of duration 200 ms, volume 60, on uplink path.
	`+UUTGN: 0`	Tone is sent in uplink (received 220 ms after the command is issued)

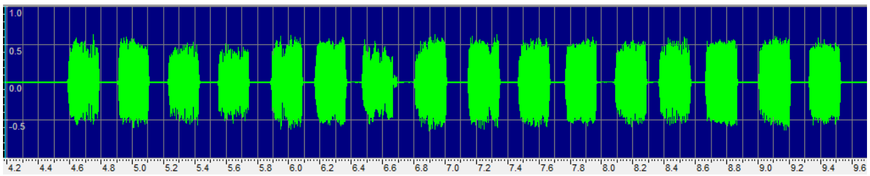

Figure 8: Sequence of 16 DTMF tones as received by remote on GSM full rate speech channel

☞ If the +UTGN command is issued before receiving +UUTGN URC of the previous +UTGN command, only the last URC is printed out.

11 Audio configuration interface

11.1 Audio configuration

☞ The command is not supported on LARA-R6 “00B”.

The +UAUDCFG AT command configures the volume of the local tones.

```
AT+UAUDCFG="tones_volume",<param1>,<param2>,<param3>,<param4>,<param5>,
```

where:

- <param1>: Free tone volume
- <param2>: Waiting tone volume
- <param3>: SMS tone volume
- <param4>: Busy tone volume
- <param5>: RTP DTMF local tone volume

The allowed values ranges go from 0 to 8192, where 0 means mute. The default and factory-programmed value is 1024 for all the local tones. 1024 corresponds to 12 dB above the speech level when +CLVL is set to factory, except for the SMS tone that is 6dB above the speech level. Multiplying the value by two corresponds to increase the volume of about 6dB.

The setting requires a module reboot to be applied.

Example:

Command	Response	Description
`AT+UAUDCFG="tones_volume",2048,2048,2048,2048,2048`	`OK`	Set the local tones volume
`AT+CFUN=16`	`OK`	Module reboot

11.2 NVM mode setting

11.2.1 General concept

The +UNVMCFG AT command sets the NVM/RAM mode for AT command settings stored in NVM. This feature is available only for a sub-set of AT commands with settings persistency in NVM. The AT commands are grouped by feature (e.g. audio commands) and the mode configuration is applied to all the AT group. Group listing is available through +UNVMCFG=? command.

By factory-programmed value, all the AT groups are in NVM mode.

11.2.1.1 NVM mode

In NVM mode, the writing in flash follows the legacy behavior (the behavior of any AT command not handled by +UNVMCFG command) with the following improvements:

- Writing in flash occurs only if the parameter value is changed with respect to the actual value in RAM/NVM (Smart NVM feature)
- The NVM item in flash is created at the first AT command execution only if the parameter is changed with respect to the factory setting.

As in legacy behavior, AT command returning OK indicates that the writing in physical flash has been accomplished.

11.2.1.2 RAM mode

In RAM mode, an AT command with settings in NVM does not write changes to physical flash, but to the RAM mirror only. Writing to flash is performed by a dedicated AT command, i.e., +UNVMW.

When the NVM settings manager of a specified group of AT commands is configured to operate in RAM mode, the following AT commands are available for the synchronization of the RAM mirror, NVM, and factory-programmed values:

- +UNVMW commits NVM settings from RAM to NVM
- +UNVMR resets the RAM settings from NVM
- +UNVMF restores the factory settings into NVM at reboot

☞ In NVM mode, +UNVMW and +UNVMR have no effect since RAM and NVM contents are always coherent.

☞ The NVM items with custom values that differs from the factory values can be listed or deleted through AT FS command operations (i.e., +ULSTFILE, +UDELFILE AT commands) with “UNVM” file tag. See AT commands manual for more information.

☞ If the UNVM item of an AT command is not listed by +ULSTFILE AT command, the factory-programmed settings are in use.

RAM mode is useful for customer devices not requiring NVM capabilities on the u-blox module (e.g., audio settings are configured at each boot by the application processor).

The +UNVMCFG AT command operates on groups of AT commands rather than on a single AT command. One of those groups is the “audio” group which includes the following AT commands:

- +CLVL, +USPM, +UI2S, +UMCLK, +CALM

11.2.1.3 Example of usage on audio AT group

Command	Response	Description
`AT+UNVMCFG="audio",0`	`OK`	Set NVM mode for the audio group
`AT+UNVMCFG="audio",1`	`OK`	Set RAM mode for the audio group
`AT+UNVMCFG=`	`OK`	Restore the factory-programmed value (NVM for LARA-R6/L6 series), for all AT groups

The NVM/RAM mode setting is stored in NVM too, so it is persistent after reboot.

To apply the new NVM/RAM mode, the module needs to reboot.

Examples:

NVM mode (factory-programmed setting)		
Command	**Response**	**Description**
`AT+UNVMCFG="audio",0`	`OK`	Set NVM mode for the audio group.
`AT+CFUN=16`	`OK`	Reboot the module to apply the new mode.
`AT+UMCLK?`	`+UMCLK:1,0` `OK`	Default value: master clock is disabled.
`AT+UMCLK=2`	`OK`	Enable the master clock.
`AT+CFUN=16`	`OK`	Reboot the module.
`AT+UMCLK?`	`+UMCLK: 2,0` `OK`	The master clock is still enabled, the setting has been stored in NVM.

RAM mode		
Command	**Response**	**Description**
`AT+UNVMCFG="audio",1`	`OK`	Set RAM mode for the audio group.
`AT+CFUN=16`	`OK`	Reboot the module to apply the new mode.
`AT+UMCLK?`	`+UMCLK:1,0` `OK`	Default value: master clock is disabled.

RAM mode		
Command	**Response**	**Description**
`AT+UMCLK=2`	`OK`	Enable the master clock.
`AT+CFUN=16`	`OK`	Reboot the module.
`AT+UMCLK?`	`+UMCLK: 1,0` `OK`	The master clock is disabled, the setting has been stored in RAM and it is not persistent. Thus, after reboot the master clock mode is still set to the default value.

11.2.2 NVM configuration management commit +UNVMW

This command commits the current NVM settings of all AT commands belonging to the specified AT command group (e.g., “audio”) from RAM to NVM.

For example:

Command	Response	Description
`AT+UNVMCFG="audio",1`	`OK`	Set RAM mode for the audio group.
`AT+CFUN=16`	`OK`	Reboot the module to apply the new mode.
`AT+UMCLK?`	`+UMCLK:1,0` `OK`	Default value: master clock is disabled.
`AT+UMCLK=2`	`OK`	Enable the master clock, value is stored in RAM only.
`AT+UNVMW="audio"`	`OK`	Copy the setting from RAM to NVM.
`AT+CFUN=16`	`OK`	Reboot the module.
`AT+UMCLK?`	`+UMCLK: 2,0` `OK`	The master clock is still enabled, the setting has been stored in NVM by +UNVMW.

☞ The command has no effect in NVM mode of operation.

11.2.3 NVM configuration management reset +UNVMR

☞ The command has no effect in NVM mode of operation.

This command resets the settings from NVM related to the AT commands managed using +UNVMCFG and then applies them.

For example:

Command	Response	Description
`AT+UNVMCFG="audio",1`	`OK`	Set RAM mode for the audio group.
`AT+CFUN=16`	`OK`	Reboot the module to apply the new mode.
`AT+UMCLK?`	`+UMCLK:1,0` `OK`	Default value: master clock is disabled.
`AT+UMCLK=2`	`OK`	Enables the master clock.
`AT+UNVMR="audio"`	`OK`	Copy the setting from NVM to RAM.
`AT+UMCLK?`	`+UMCLK: 1,0` `OK`	The master clock has been reset at the default value stored in NVM.

11.2.4 NVM configuration management factory restore +UNVMF

This command restores the NVM of all AT commands in the selected group to the factory-programmed values.

After a factory-programmed restore using the +UNVMF AT command, it is recommended to reboot the module. Optionally, for the “audio” AT group, the restored NVM can be synchronized with the RAM copy using the +UNVMR AT command.

For example:

Command	Response	Description
Module boot up		
`AT+UMCLK?`	`+UMCLK:2,0` `OK`	Master clock is enabled (read from NVM at boot).
`AT+UNVMF="audio"`	`OK`	Restore factory-programmed settings for the audio group.
`AT+UMCLK?`	`+UMCLK: 1,0` `OK`	The master clock has been reset to the factory-programmed value.

12 Migration guide

12.1 From LARA-R2 to LARA-R6

Table 8 describes the main differences related to the audio features and sub - features supported by LARA-R2 and LARA-R6 series.

Feature or sub-feature	LARA-R2	LARA-R6
Loudspeaker volume: +CLVL	The allowed values range goes from 0 to 100. Factory-programmed value is 80 **(-4dB)**.	The allowed values range goes from 0 to 6. Factory-programmed value is 3 **(-9dB)**.
Ringer volume: +CRSL		NOT SUPPORTED
Players volume: +CRSL, +USGC		NOT SUPPORTED
Alert tone muting: +CALM	If +CALM is set to 1, the +UTGN command is not functional and returns with error.	+CALM=1 does not affect +UTGN: the tone generator is fully functional.
Silent alarm : +CALA		NOT SUPPORTED
Speech path mode: +USPM	Syntax: AT+USPM=<main_uplink>,<main_downlink>,<alert_sound>,<headset_indication>[,<vmic_ctrl>]	Syntax: AT+USPM=<audio_path>,<profile_type>
Supervisory tones: Alarm tone		NOT SUPPORTED
Pre-defined tones: +UPAR +USAR		Only <audio_resource>=2 (audio loop) is supported
Ringer selection: +URNG		NOT SUPPORTED
Audio file player / recorder: +UPLAYFILE, +URECFILE		NOT SUPPORTED
Speech player/recorder: +UAPLAY, +UAREC		NOT SUPPORTED
Speech codec configuration 2G/3G: +UDCONF=30		The Half Rate codec cannot be excluded. GSM Enhanced Full Rate and GSM Full Rate cannot be excluded.
Codec mode info 2G/3G : +USPEECHINFO		Indication for <codec>=3 (Half Rate Adaptive Multi-Rate codec) is not supported, and it is always reported as <codec>=0 (Full Rate Adaptive Multi-Rate codec).
Codec rate info VoLTE +USPEECHINFO		LTE codec bit rates are not available.
Speech codec configuration VoLTE +USPEECHCFG		Only the AMR WB codec can be disabled. See 6.2.2 for more info.
Codec configuration: +UEXTDCONF	<configuration_enable> range is 0-1.	<configuration_enable> range is 0-2. AT+UEXTDCONF=0,2 configures the external device but does not change the state of the master clock
Clock for external codec: +UMCLK	The clock supported is 13 MHz.	The clock supported is 12.288 MHz. <mclk_mode> = 0 is not supported, the clock is disabled with <mclk_mode> = 1.
Continuous mode: +UMCLK		NOT SUPPORTED

Feature or sub-feature	LARA-R2	LARA-R6
I2S PCM modes configuration: +UI2S		Only one PCM mode is supported.
Normal I2S configuration: +UI2S		Only one normal mode is supported.
Sampling rates: +UI2S	Supports a sample rate up to 48kHz.	LARA-R6 “00B”: Only 16kHz is supported. LARA-R6 “01B”: 16 and 48 kHz are supported
I2S always on (continuous mode): +UI2S		NOT SUPPORTED
Slave mode: +UI2S		LARA-R6 “00B”: NOT SUPPORTED LARA-R6 “01B”: SUPPORTED with appropriated slave version audioconf file.
Car HF speech enhancement		NOT SUPPORTED
Digital speech gains UL/DL: +UMGC, +USGC		NOT SUPPORTED
Side tone: +USTN		NOT SUPPORTED
Equalizers UL/DL: +UUBF, +UDBF		NOT SUPPORTED
Basic tuning AT commands: +UHFP	Customers can design the audio configuration by themselves using specific AT commands.	NOT SUPPORTED
Extended tuning AT commands: +UTI	Audio configuration is set by AT commands.	Audio configuration is provided in the form of a binary file that the customer can download as a common file. LARA-R6 “00B”: UTI command NOT SUPPORTED LARA-R6 “01B”: UTI command SUPPORTED
DTMF detector (legacy): +UDTMFCFG	The <algo_type> supported is “legacy” and must be issued when the call is off.	The <algo_type> supported is “default” and can be issued in any call state.
DTMF detector MODA: +UDTMFCFG		NOT SUPPORTED
Burst mode: +UDTMFCFG		NOT SUPPORTED
DTMF generator: +UDTMFG +UTGN	+UDTMFG implements the DTMF generator. Maximum string length is 64 characters.	+UTGN implements the DTMF generator. Maximum string length is one character.
Contact ID protocol support: +UDTMFCFG		NOT SUPPORTED
PCM logging: +UDTMFCFG		NOT SUPPORTED
DTMF RTP decoder: +UDTMFCFG	<enable_rtp> enables/disables RTP DTMF support.	The RTP DTMF support is enabled by default for VoLTE calls and cannot be disabled.
Burst mode: +UDTMFCFG		NOT SUPPORTED
Smart DTMF generator: +UDTMFG		NOT SUPPORTED
VoLTE DTMF RTP mode disable : +UDTMFCFG		NOT SUPPORTED

Feature or sub-feature	LARA-R2	LARA-R6
Audio configuration: +UAUDCFG	NOT SUPPORTED	LARA-R6 “00B”: +UAUDCFG command NOT SUPPORTED
NVM-RAM mode setting: +UNVMCFG	NOT SUPPORTED	
NVM configuration management commit : +UNVMW	NOT SUPPORTED	
NVM configuration management reset : +UNVMR	NOT SUPPORTED	
NVM configuration management factory restore: +UNVMF	NOT SUPPORTED	
Sound activity indications: +CIEV		NOT SUPPORTED

Table 8: Audio features migration guide from LARA-R2 to LARA-R6

12.2 From TOBY-L2 to LARA-L6

Table 9 describes the main differences related to the audio features and sub - features supported by TOBY-L2 and LARA-L6 series.

Feature or sub-feature	TOBY-L2	LARA-L6
Loudspeaker volume: +CLVL	The allowed values range goes from 0 to 37, where 0 means mute, 37 means 0 dB and the step size is 1 dB. Factory-programmed value is 30 **(-4dB).**	The allowed values range goes from 0 to 6, where 0 means mute, 1 means -30 dB, 6 means 0 dB and the step size is 6 dB. Factory-programmed value is 4 **(-12dB).**
Ringer volume +CRSL	The allowed values range goes from 0 to 5, where 0 means mute, 4 means -8 dB and the step size is 9 dB. Factory-programmed value is 4.	The allowed values range goes from 0 to 6, where 0 means mute, 1 means -30 dB, 6 means 0 dB and the step size is 6 dB. Factory-programmed value is 5.
Players volume: +CRSL, +USGC		NOT SUPPORTED
Alert tone muting +CALM	If +CALM is set to 1, the +UTGN command is not functional and returns with error. If +CALM is set to 1, the service tones and alarm tone are also muted; furthermore the +UPLAYFILE and +UPAR are not functional.	+CALM=1 does not affect +UTGN: the tone generator is fully functional.
Silent alarm +CALA		NOT SUPPORTED
Speech path mode: +USPM	Syntax: AT+USPM=<main_uplink>, <main_downlink>, <alert_sound>,<headset_indication>[, <vmic_ctrl>]	Syntax: AT+USPM=<audio_path>, <profile_type>
Speech codecs VoLTE AMR NB	NOT SUPPORTED	
Speech codecs VoLTE AMR WB	NOT SUPPORTED	
Alarm tone		NOT SUPPORTED
Pre-defined tones: +UPAR +USAR		Only <audio_resource>=2 (audio loop) is supported
Audio file player / recorder: +UPLAYFILE, +URECFILE		NOT SUPPORTED

Feature or sub-feature	TOBY-L2	LARA-L6
Speech codec configuration 2G/3G: +UDCONF=30		The Half Rate codec cannot be excluded. GSM Enhanced Full Rate and GSM Full Rate cannot be excluded.
Codec mode info 2G/3G : +USPEECHINFO	NOT SUPPORTED	
Codec mode info VoLTE: +USPEECHINFO	NOT SUPPORTED	LTE codec bit rates are not available.
Speech codec configuration VoLTE: +USPEECHCFG	NOT SUPPORTED	
Codec rate info VoLTE: +USPEECHINFO	NOT SUPPORTED	
Codec configuration: +UEXTDCONF	<configuration_enable> range is 0-1. +UVGC is used to volume gain control of external Maxim MAX9860 audio codec.	<configuration_enable> range is 0-2. AT+UEXTDCONF=0,2 configures the external device but does not change the state of the master clock
Clock for external codec: +UMCLK	The clock supported is 13 MHz.	<mclk_mode> = 0 and 3 are not supported. The clock supported is 12.288 MHz. "Continuous" mode is not supported.
Continuous mode: +UMCLK		NOT SUPPORTED
I2S PCM modes configuration: +UI2S		Only one PCM mode is supported.
Normal I2S configuration: +UI2S		Only one normal mode is supported.
Sampling rates: +UI2S	Supports sample rates of 8kHz and 16kHz.	LARA-L6 “00B”: Only 16kHz is supported. LARA-L6 “01B”: 16 and 48 kHz are supported
I2S always on (continuous mode): +UI2S		NOT SUPPORTED
Slave mode: +UI2S		LARA-L6 “00B”: NOT SUPPORTED LARA-L6 “01B”: SUPPORTED with appropriated slave version audio configuration file.
Car HF speech enhancement		NOT SUPPORTED
Basic tuning AT commands:	The command +UAPT (Audio Parameter Tuning) is used to audio parameters management for all the signal processing blocks.	NOT SUPPORTED
Extended tuning AT commands: +UTI	NOT SUPPORTED	Audio configuration is provided in the form of a binary file that the customer can download as a common file. LARA-L6 “00B”: UTI command NOT SUPPORTED LARA-L6 “01B”: UTI command SUPPORTED
DTMF detector (legacy)	Implemented by +UDTMFD.	Implemented by +UDTMFCFG.
DTMF generator	NOT SUPPORTED	Implemented by +UTGN.
DTMF RTP decoder	NOT SUPPORTED	

Feature or sub-feature	TOBY-L2	LARA-L6
DTMF regenerator (local play on loudspeaker): +UDTMFCFG	NOT SUPPORTED	
Audio configuration: +UAUDCFG	NOT SUPPORTED	
NVM-RAM mode setting	NOT SUPPORTED	
NVM configuration management commit	NOT SUPPORTED	
NVM configuration management reset	NOT SUPPORTED	
NVM configuration management factory restores	NOT SUPPORTED	
Sound activity indications: +CIEV		NOT SUPPORTED

Table 9: Audio features migration guide from TOBY-L2 to LARA-L6

Appendix

A Glossary

Abbreviation	Definition
AC	Adaptation Control block
AEC	Acoustic echo canceller
AFB	Analysis Filter Bank
AGC	Automatic Gain Control
AIG	Adaptive Input Gain
ANA	Ambient Noise Adaptation
AP	Application Processor
AT	AT Command Interpreter Software Subsystem, or attention
DBF	Downlink Biquad Filters command
DCN	Downlink Comfort Noise Injector
DCP	Downlink Compressor
DES	Dynamic Echo Suppressor
DRP	Dynamic Range Processor
DSP	Digital Signal Processing
EC	Echo Cancellation algorithm
EDL	Echo Delay Line
FIR	Finite Impulse Response
EQ	Equalizer
GLC	Gain Loss Control
HF	Hands-free Algorithm
HPF	High Pass Filter
MDRP	Multiband Dynamic Range Processor
MOC	Mobile Originated Call
MTC	Mobile Terminated Call
NB	Narrow Band
NR	Noise Reduction
NS	Noise Suppressor
NVM	Non-Volatile Memory
PSTN	Public Switched Telephone Network
RX	Receiver / Receive path
SER	Spectral Echo Reduction
SES	Speech Enhancement System
SFB	Synthesis Filter Bank
SGC	Speaker Gain control command
SNS	Spatial Noise Suppressor
SRL	Set Reference Line
TX	Transmitter / Transmission path
UBF	Uplink Biquad Filters command
UCP	Uplink Compressor
UCN	Uplink Comfort Noise Injector
UNS	Uplink Noise (Reduction) Suppressor

Abbreviation	Definition
WB	Wide Band

Related documents

[1] u-blox LARA-L6 / LARA-R6 series AT commands manual, UBX-21046719
[2] Work Items with ETSI Document Number "201 235". See Work Programme search database, http://www.etsi.org/
[3] u-blox LARA-R6 series data sheet, UBX-21004391
[4] u-blox LARA-L6 series data sheet, UBX-21047783
[5] u-blox LARA-L6 / LARA-R6 series system integration manual, UBX-21010011
[6] RTP Payload for DTMF Digits, Telephony Tones, and Telephony Signals RFC4733
[7] u-blox LARA-R6 series firmware update with uFOTA, FOAT and EasyFlash application note, UBX-22008011
[8] maxim integrated MAX9860 16-Bit Mono Audio Voice Codec datasheet, 19-4349; Rev 2; 1/12. Available from the maxim integrated website (https://datasheets.maximintegrated.com/en/ds/MAX9860.pdf)

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

Revision history

Revision	Date	Name	Comments
R01	27-Jun-2022		Initial release
R02	15-May-2024		Document extended for LARA-L6 series and LARA-R6 "01B" product versions.

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