LARA-R6 series

Audio interface

Application note

Abstract

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

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Document information

Title	**LARA-R6 series**	
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This document applies to the following products:

Product name	
LARA-R6 series	Except for LARA-R6001D and LARA-R6401D

Contents

1 Introduction

This document provides an overview of the audio features supported on the LARA-R6 series modules. Most of the features are fully covered in the u-blox LARA-R6 series AT commands manual [1] and in the LARA-R6 series system integration manual [3].

This document addresses the topics that need additional application notes, like volume management, audio routing and profiles, speech codecs and external codec management, the DTMF in-band signaling detector [2] / DTMF RTP decoder [4], etc.

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

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 List of supported features

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

Feature		AT commands	Supported
Volume	Microphone muting	+CMUT	Yes
	Loudspeaker volume	+CLVL	Yes
	Ringer volume	+CRSL	No
	Players volume	+CRSL, +USGC	No
	Alert tone muting	+CALM	Yes
	Message sound muting	+UMSM	Yes
	Silent alarm	+CALA	No
Audio routing and profiles	Speech path mode	+USPM	Yes
Speech codecs	GSM EFR		Yes
	GSM FR		Yes
	GSM HR		Yes
	FR AMR		Yes
	FR AMR WB		Yes
	HR AMR		Yes
	UMTS AMR		Yes
	UMTS AMR 2		Yes
	UMTS AMR WB		Yes
	VoLTE AMR NB		Yes
	VoLTE AMR WB		Yes
	VoLTE EVS NB		No
	VoLTE EVS WB		No
	VoLTE EVS SWB		No
	VoLTE EVS FB		No
Supervisory tones	Ringing tone on MOC		Yes
	Subscriber busy on MOC		Yes
	Call waiting on MTC		Yes
	Ringer on MTC (ringtone)		Yes
	Incoming SMS		Yes
	Alarm tone		No
	Tones mixed with speech		Yes
Players	Pre-defined tones		No
	Tone generator UL/DL	+UTGN	Yes
	Ringer selection	+URNG	No
Audio file player / recorder	NB 8 kHz		No
	WB 16 kHz		No
	Generic player UL	+UPLAYFILE	No
	Generic player DL	+UPLAYFILE	No
	Custom ringer melody	+UPLAYFILE	No
	Answering machine	+UPLAYFILE	No
	Generic recorder UL	+URECFILE	No
	Generic recorder DL	+URECFILE	No
	Microphone recorder	+URECFILE	No

Feature		AT commands	Supported
Speech player / recorder	Speech player UL	+UAPLAY	No
	Speech player DL	+UAPLAY	No
	Speech recorder UL	+UAREC	No
	Speech recorder DL	+UAREC	No
Speech codec management	Speech codec configuration 2G/3G	+UDCONF=30	Yes
	Codec mode info 2G/3G	+USPEECHINFO	Yes
	Speech codec configuration VoLTE	+USPEECHCFG	Yes
	Codec mode info VoLTE	+USPEECHINFO	Yes
	Codec rate info VoLTE	+USPEECHINFO	No
External codec management	Codec configuration	+UEXTDCONF	Yes
	Clock for external codec	+UMCLK	Yes
Digital audio interface	I2S PCM modes configuration	+UI2S	Yes
	Normal I2S configuration	+UI2S	Yes
	Stereo mode 48 kHz	+UI2S	No
Analog audio interface			No
Speech enhancement	Car HF speech enhancement		No
	Alarm panel speech enhancement		Yes
	Desktop HF speech enhancement		Yes
	Digital speech gains UL/DL	+UMGC, +USGC	No
	Side tone	+USTN	No
	Equalizers UL/DL	+UUBF, +UDBF	No
	Basic tuning AT commands	+UHFP	No
	Extended tuning AT commands	+UTI	No
	Audio Tuning Tool		No
In-Band DTMF	DTMF detector (legacy)	+UDTMFD +UDTMFCFG	No Yes
	DTMF detector MODA	+UDTMFCFG	No
	Burst mode	+UDTMFCFG	No
	DTMF generator	+UDTMFG +UTGN	No Yes
	Contact ID protocol support	+UDTMFCFG	No
	PCM logging	+UDTMFCFG	No
VoLTE DTMF	DTMF RTP decoder	+UDTMFCFG	Yes
	Burst mode	+UDTMFCFG	No
	DTMF regenerator (local play on loudspeaker)	+UDTMFCFG	Yes
	Smart DTMF generator	+UDTMFG	No
	VoLTE DTMF RTP mode disable	+UDTMFCFG	No
NVM RAM mode management	NVM-RAM mode setting	+UNVMCFG +UDCONF=110	Yes No
	NVM configuration management commit	+UNVMW	Yes
	NVM configuration management reset	+UNVMR	Yes
	NVM configuration management factory restore	+UNVMF	Yes
Product testing	Test audio IF loopback	+UPAR, +USAR	Yes
Sound activity indications	Sound activity indications	+CIEV	No

Table 1: Supported features

2 Volume

2.1 Loudspeaker volume

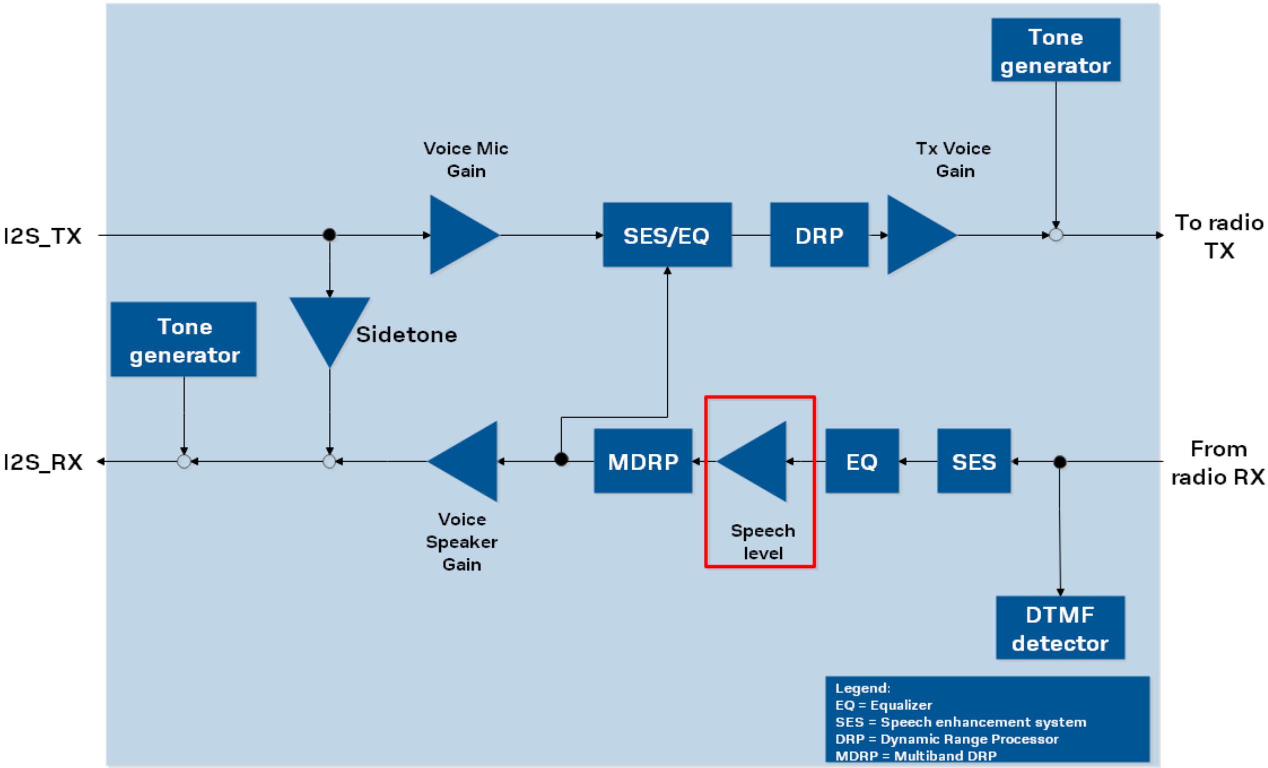

Figure 1: Incoming speech level gain

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

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

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. This setting is persistent also after a profile switch.

The command affects only the speech volume, the tone generator volume is not affected.

For example:

Command	Response	Description
AT+CLVL=4	OK	Set the speech volume to level 4.

2.2 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.

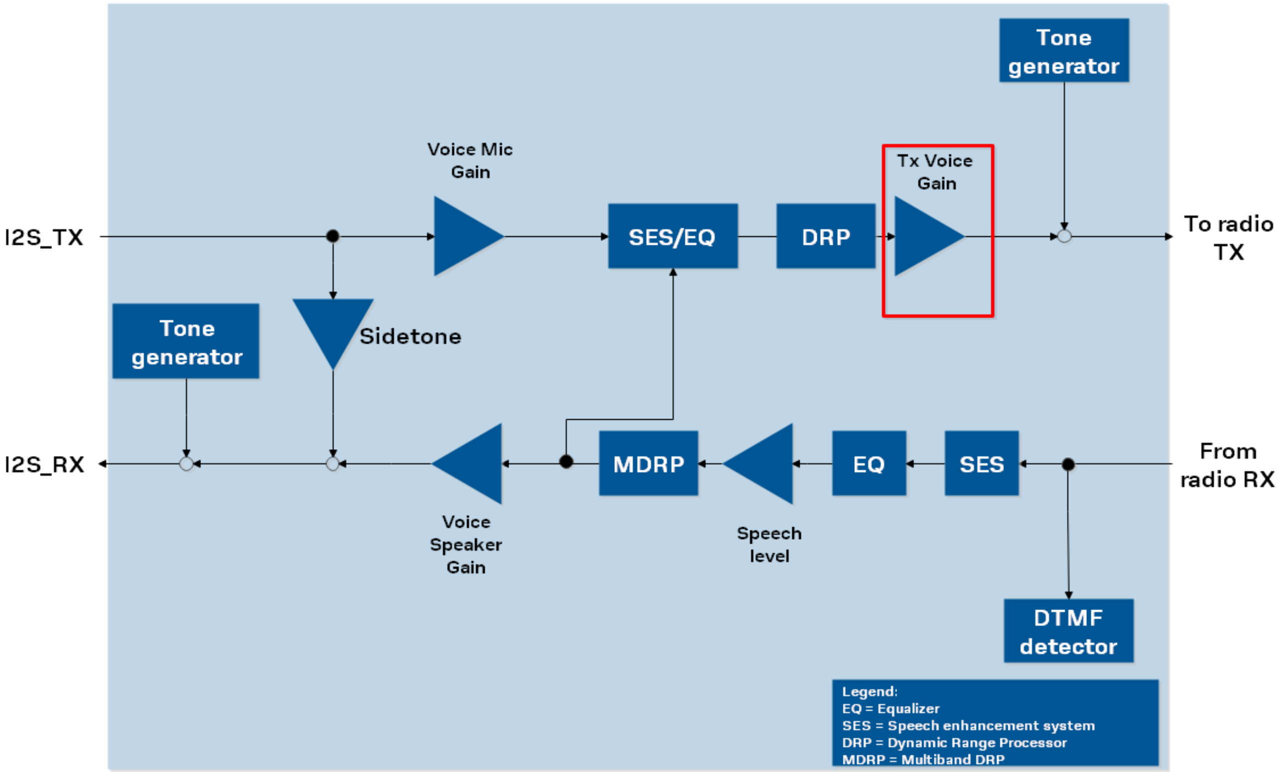

Figure 2: Muting acting after the SES block

Example:

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

3 Supervisory tones

3.1 Enabled supervisory tones

Supported supervisory tones by the module are:

- 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 tone.

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 can be reproduced locally or sent by the network, using in-band tones on 2G/3G calls or early media on VoLTE calls.

During VoLTE calls, the +UCALLSTAT: 1,3 URC can be checked to verify whether the tone is reproduced locally or not:

- If the `+UCALLSTAT: 1,3` URC is present, the tone is played locally.
- Otherwise, the ring back should be generated by the network.

3.1.2 SMS alert sound for incoming messages

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 and 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`	Indicates the URCs format resulting from 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?) may possibly indicate that the SMS alert sound is enabled. Therefore, the application code must not rely on +UMSM status to check if SMS alert sound is active when silent mode is enabled.

4 Audio routing and profiles

4.1 Audio profiles setting

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

There are four profiles available for each audio path:

- **Headset profile**: factory-programmed settings suitable for handset/headset devices
- **Hands-free profile**: factory-programmed settings suitable for hands-free devices
- **Flat profile**: no additional processing on the audio path, all the blocks are disabled.
- **Alarm panel profile**: factory-programmed settings suitable 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 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 voice call.

5 Player management

The +UTGN AT command generates:

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

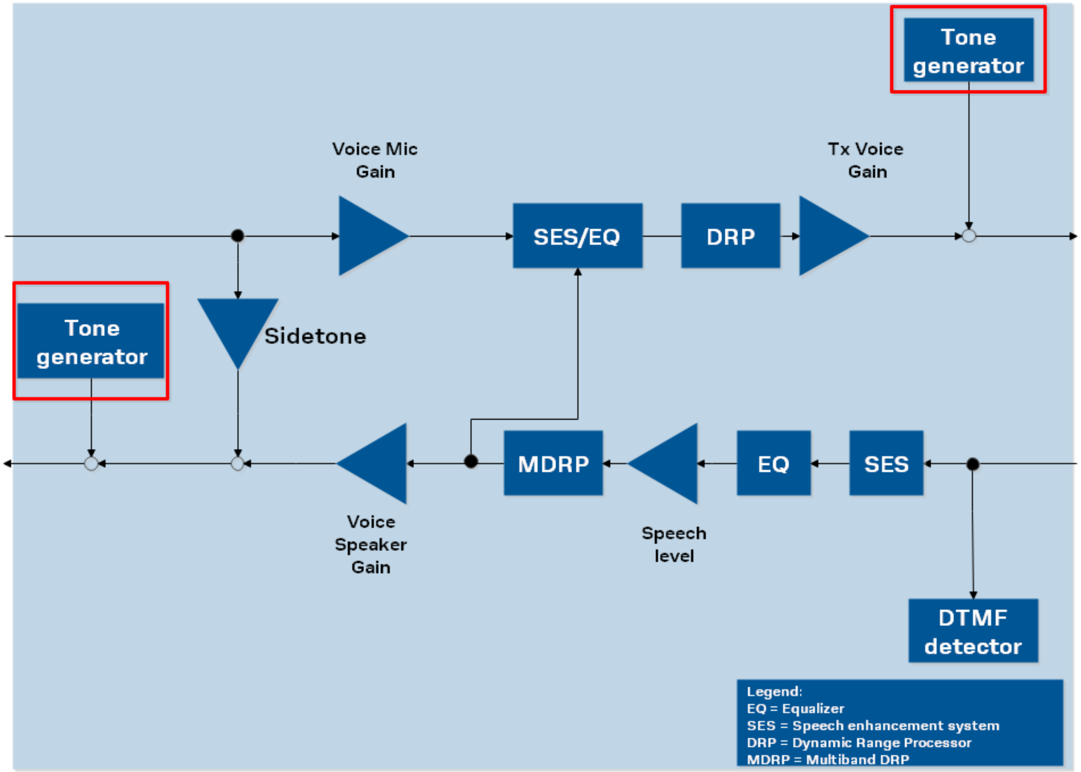

Figure 3: Tone generator used by tone players in the uplink and downlink directions

5.1 Tone generator UL/DL

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 is 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=0,0,0 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.

5.2 Audio loopback by +UPAR/+USAR

The audio loopback for testing purposes is available using the +UPAR AT command. For this platform, only the <audio_resource> audio loop is implemented.

The loop is implemented between I2S_RX and I2S_TX.

For example:

Command	Response	Description
`AT+UPAR=2,0,0`	`OK`	Starts 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.

In the event that an audio loop is started and a MO or MT call is performed, the audio loop stops automatically when the call is established.

To stop a started audio loop, the +USAR AT command is used 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. On LARA-R6 modules, 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).

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.

7 Digital audio interface / External codec management

LARA-R6 series modules provide an I2S digital audio interface to connect an external audio device, e.g., an audio codec. The application processor (AP) should manage the codec.

The digital I2S interface is described in u-blox LARA-R6 series AT commands manual [1] and in the LARA-R6 series system integration manual [3]. 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. In Figure 4, the block diagram of the HW implementation is highly simplified.

For more details about the AT commands used in the examples below, see the u-blox LARA-R6 series AT commands manual [1].

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

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

Figure 4 shows a possible architecture for the LARA-R6 module / external codec / AP system.

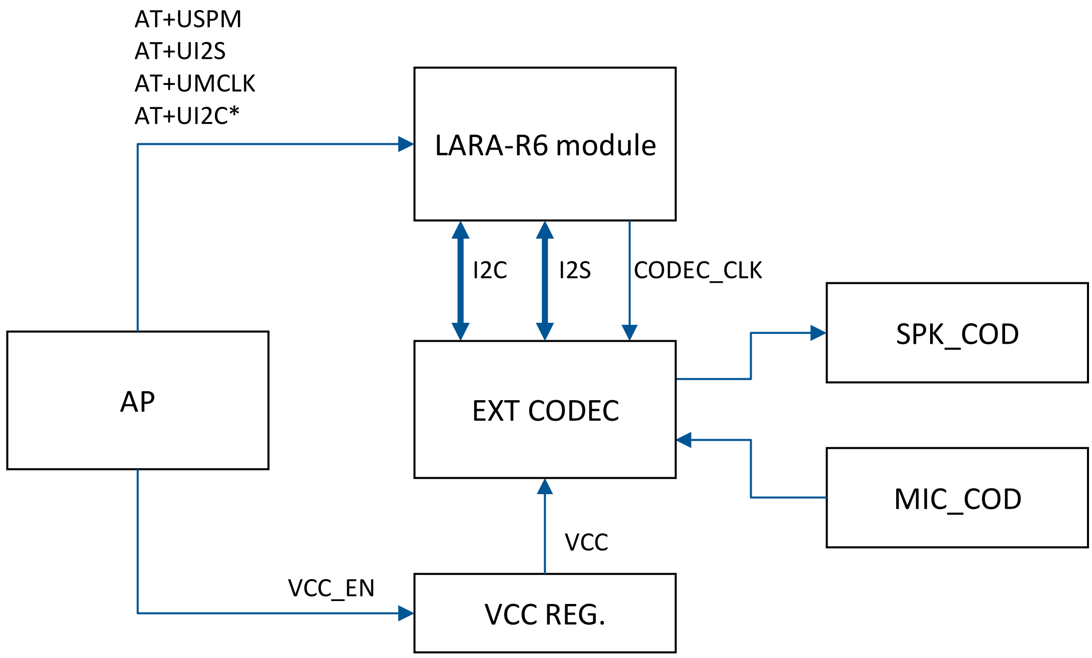

Figure 4: External codec management for LARA-R6 series module

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 Scenario

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> • <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 1^{st} register, value for 2^{nd} 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=0,1 command. AT+UMCLK=0,0 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>="0000000010A000303000063300500000008A")
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

The example below shows the procedure of configuring the external codec with the +UEXTDCONF AT command:

Command	Response	Description
`AT+UEXTDCONF=0`	`OK`	Resets parameters to factory settings. Default <hex_data> string is saved in NVM.
`AT+CFUN=16`	`OK`	Reset the module.
`AT+UEXTDCONF?`	`+UEXTDCONF: 0,1,"0000000010A00030300006330500000008A"` `OK`	At boot, external audio codec is enabled, and it is programmed via I2C.
`AT+UMCLK?`	`+UMCLK: 1,0` `OK`	+UMCLK setting is not changed; e.g., MCLK is disabled.
`AT+UEXTDCONF=0,1`	`OK`	Enable external codec Max9860. <hex_data> string is not changed.
`AT+CFUN=16`	`OK`	Reset the module.

Command	Response	Description
AT+UEXTDCONF?	+UEXTDCONF: 0,0,"0000000010A000303000063300500000008A" OK	At boot, the external audio codec is not enabled(default) String <hex_data> remains the last saved value.
AT+UMCLK?	+UMCLK: 2,0 OK	If the external codec is enabled, MCLK is enabled at boot
		The following commands are just an example of how to 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: 0x6 +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).
AT+UI2CC	OK	Close the I2C logical channel.
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 previous example.
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 MAX8960 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.
		Configure I2S on both the module and the MAX9680 for PCM mode.
AT+UEXTDCONF=0,0	OK	Disable MAX9860 external codec. <hex_data> string is not changed.
AT+UEXTDCONF=0,1,"0000000010A0000400000063300500000008A"	OK	If <configuration_enable> changes from 0 to 1, the module configures the MAX8960 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.
AT+UMCLK=2,0	OK	Enable MCLK.
AT+UI2S=30,1,0,3,0	OK	Set LARA-R6's I2S to PCM mode.
AT+UPAR=2,0,0	OK	Start speech loop; I2S runs in PCM mode.
AT+USAR=2	OK	Stop speech loop; I2S stops.

8 Speech enhancement

This section shows the speech enhancement system (SES) available on LARA-R6 series modules.

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.

These profiles can be selected as described in section 4.

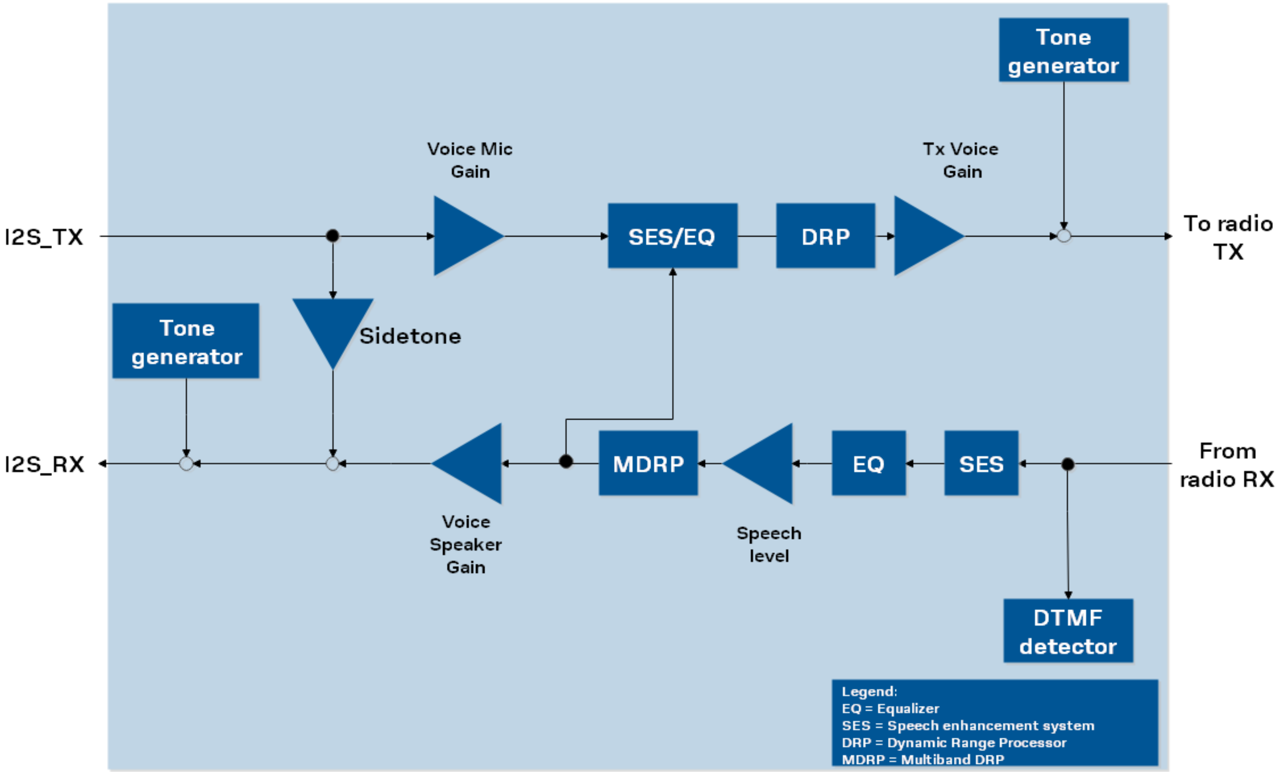

Figure 5: LARA-R6 speech enhancement system

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

- In uplink, AEC and single microphone noise suppression.
- In downlink, far-end noise suppression and single band dynamic range processing.

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.

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:

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

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

The tuning of these profiles can be changed upon customer request and u-blox will support the tuning required by the customer's audio device.

The audio parameter which contains the voice algorithm parameters and I2S settings for all u-blox audio profiles are stored in the audioconfig file which is part of the u-blox binary package.

Upon request, u-blox will deliver a new audioconfig file to fulfil specific customer requirements[2]. Any new audioconfig files can be downloaded into the module using the +UDWNFILE AT command as shown in the table below with the AUDIO_EXT tag:

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

Downloading the audioconfig 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 configuration is not applied, the "+CME ERROR: operation not supported" error result code will be provided, the downloaded file will be closed and deleted, and the audioconfig 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 audioconfig 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 series FW update application note [5]).
- 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 audioconfig 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 audioconfig 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 audioconfig file using the +UDWNFILE AT command.

[2] Tuning of the customer device requires that the customer device is delivered to u-blox audio laboratory.

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 6. 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.

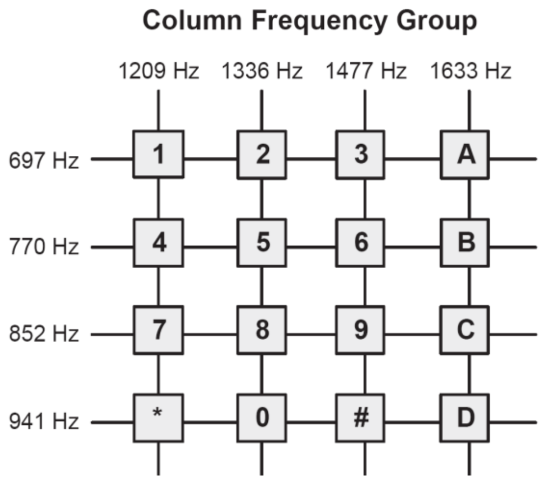

Figure 6 Touch-Tone telephone keypad: a row and a column tone is associated with each digit

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 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:

Command	Response	Description
	`+UUDTMFD: 4,0`	The digit “4” has been detected, it is received as an RTP event.

When the DTMF detector is enabled, URCs with information about in-band/RTP detection are enabled by default.

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
`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 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.

☞ The GSM Half Rate (5.6 kbit/s) codec) introduces distortions on DTMF signals and decreases the detection rate. However, it cannot be excluded.

☞ 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 [4].

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 2, 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 2 – 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`	URC enabled
`AT+UDTMFCFG="urc",2`	`OK`	URC enabled with RTP/in-band indication
`AT+UDTMFCFG="urc",3`	`OK`	URC enabled with RTP/in-band indication and duration indications enabled

When one of the DTMF algorithms is enabled with `AT+UDTMFCFG="algo","default"`, the URCs with RTP/in-band indication are enabled by default (<enable_urc>=2).

Indeed, the `AT+UDTMFCFG="urc",2` command allows to add, near the DTMF tone detected, an indication related to the in-band or RTP decoding as shown in Tabl 3.

URC	Decoding mode
`+UUDTMFD: 4,1`	In-band decoding
`+UUDTMFD: 4,0`	RTP decoding

Tabl 3: URC examples

`AT+UDTMFCFG="urc",3` command allows to add, near the DTMF tone detected, an indication related to the in-band or RTP decoding and the tone duration for the RTP use-case as shown in Table 4.

URC	Decoding mode
`+UUDTMFD: 4,1`	In-band decoding
`+UUDTMFD: 4,0`	RTP decoding, RTP tone start
`+UUDTMFD: 4,2`	RTP decoding, RTP tone stop

Table 4: URC with tone duration example

The tone duration is equal to the time between the RTP tone start and RTP tone stop events. This information is available only for RTP use case.

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 tone '0' – '9', '*' and '#', and '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)

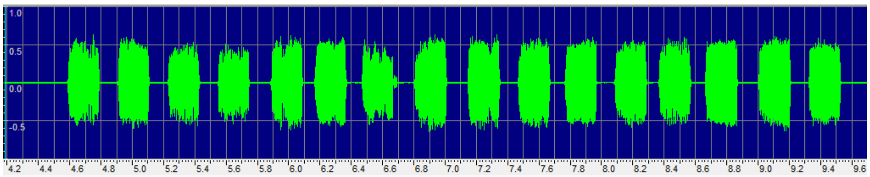

Figure 7: Sequence of 16 DTMF tones as received by remote on GSM full rate speech channel

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

11 NVM RAM mode management

The +UNVMCFG AT command sets the NVM/RAM mode for AT command settings stored in NVM.

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: +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

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

By factory-programmed settings, the "audio" AT group operates in NVM mode.

To list the commands related to each AT group, issue the command: AT+UNVMCFG=?

Example of usage:

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 series)

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.

11.1 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 NVM configuration management reset +UNVMR

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.

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

11.3 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, 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.

Appendix

A Glossary

Abbreviation	Definition
AC	Adaptation Control block
AEC	Acoustic echo canceller
AFB	Analysis Filter Bank
AGC	Automatic Gain Control
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
DSP	Digital Signal Processing
EC	Echo Cancellation algorithm
EDL	Echo Delay Line
FIR	Finite Impulse Response
GLC	Gain Loss Control
HF	Hands-free Algorithm
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
WB	Wide Band

Table 5: Explanation of the abbreviations and terms used

Related documents

[1] u-blox 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 system integration manual, UBX-21010011

[4] RTP Payload for DTMF Digits, Telephony Tones, and Telephony Signals RFC4733

[5] u-blox LARA-R6 series firmware update with uFOTA, FOAT and EasyFlash application note, UBX-22008011

☞ 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	vpah	Initial release

Contact

For further support and contact information, visit us at www.u-blox.com/support.