BG95&BG77&BG600L Series
Voice Application Note

LPWA Module Series

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About the Document

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-	2022-06-01	Nora HE	Creation of the document
1.0	2022-09-30	Nora HE	First official release

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1 Introduction

This document introduces the application of VoLTE and describes the related AT commands of VoLTE, CS voice over GSM (2G) and audio services on Quectel BG95 series, BG77 and BG600L-M3 modules.

1.1. Applicable Modules and Supported Voice Feature

Table 1: Applicable Modules and Supported Voice Feature

Module Series	Module	Supported Network	Supported Voice Feature
BG95	BG95-M1	Cat M1 only	VoLTE
	BG95-M2	Cat M1/Cat NB2	VoLTE
	BG95-M3	Cat M1/Cat NB2/EGPRS	VoLTE/CS voice
	BG95-M5	Cat M1/Cat NB2/EGPRS, Power Class 3	VoLTE/CS voice
	BG95-M6	Cat M1/Cat NB2, Power Class 3	VoLTE
BG77	BG77	Cat M1/Cat NB2	VoLTE
BG600L	BG600L-M3	Cat M1/Cat NB2/EGPRS	VoLTE/CS voice

2 General Overview of IMS and VoLTE

2.1. Introduction to IMS Features

The IP Multimedia Subsystem (IMS) is an architectural framework for delivering IP multimedia services.

2.1.1. Supported Features

The following features are supported in IMS:
- LTE RAT
- Power-saving mode
- IMS registration with or without IPsec
- Single VoLTE call
- SIP forking (up to seven dialogs)
- Audio codecs: AMR-NB, DTMF
- QoS and preconditions
- SMS over IMS in 3GPP format only

2.1.2. Features Not Supported

The following features are currently not supported in IMS:
- Emergency call
- Multi-call, such as conference call, call transfer and call pulling
- SRVCC and CSFB
- Audio codecs: EVS, AMR-WB, and G.711
- VoLTE for internet-connected endpoint (VICE)
- DAN
- Video call, RTT, and TTY
- MWI
- IWLAN
- RCS or Presence

2.2. Introduction to VoLTE Call Flow

VoLTE (Voice over Long-Term Evolution) is based on the IP Multimedia Subsystem (IMS) network, and uses special configuration files for the control and media planes. This enables voice services (at the control and media levels) to be transmitted as data streams over LTE data bearer networks. VoLTE initialization procedures are required before a VoLTE session is established.

2.2.1. Prerequisite for VoLTE Call

1. Attach to LTE network and select domain (all call flows assume voice over IMS).
2. Establish IMS PDN connection and SIP QoS flow. QCI = 5 is used for SIP signaling on IMS default bearer.
3. Complete IMS registration and subscription with IMS CN. IMS feature tags for available services are MMTel IMS Communication Services Identifier (ICSI).

2.2.2. Process of VoLTE Call

The steps to establish a VoLTE session with a remote party are as follows.

1. Attach to LTE network and select domain.
2. Establish IMS PDN connection and SIP QoS flow.
3. Complete IMS registration and subscription.
4. Establish IMS session.
   - Transmit information by SIP signaling.
   - Negotiate codec with SDP.
5. Establish VoLTE QoS call flow.
   - Implement network-initiated QoS procedure.
6. Propose ROHC negotiation.
7. Transmit full-duplex VoLTE data between UE and E-UTRA(N).
   - This data is routed from the E-UTRA(N) by the packet data core network to the remote party involved in the call.
   - Each voice frame is generated by the vocoder and carried as the payload in RTP.
   - RTP packets are encapsulated in UDP packets and carried over IP.
8. Terminate IMS session.
   - Transmit information by SIP signaling.
   - Release and clean up existing VoLTE call(s).
9. Release VoLTE QoS flow.
   - Release network-dependent QoS procedure.
10. Terminate VoLTE feature.
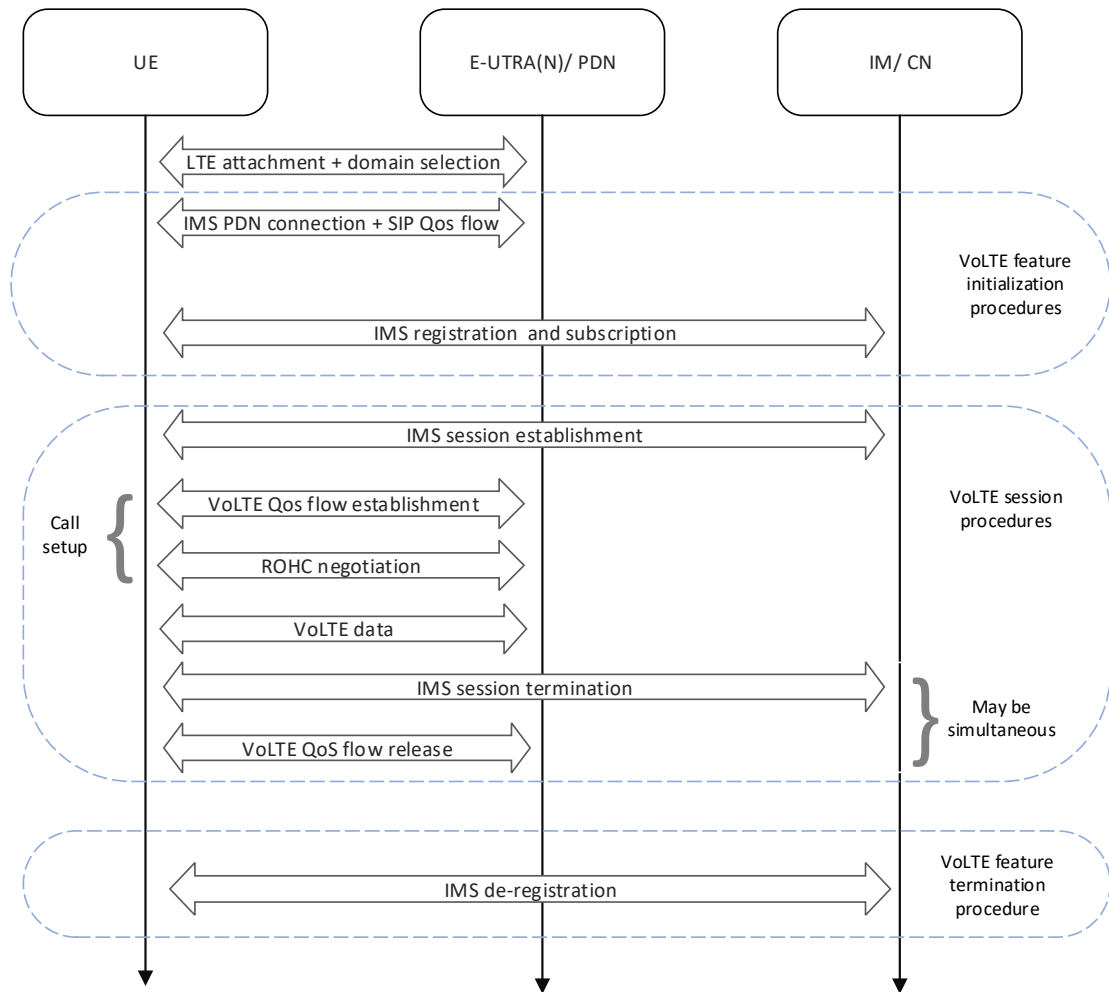
Figure 1: VoLTE Call Flow

3 AT Command Introduction

3.1. AT Command Introduction

3.1.1. Definitions

- **<CR>**   Carriage return character.
- **<LF>**   Line feed character.
- **<...>**   Parameter name. Angle brackets do not appear on the command line.
-     **[...]** Optional parameter of a command or an optional part of TA information response. Square brackets do not appear on the command line. When an optional parameter is not given in a command, the new value equals its previous value or the default settings, unless otherwise specified.
- **Underline**   Default setting of a parameter.

3.1.2. AT Command Syntax

All command lines must start with **AT** or **at** and end with **<CR>**. Information responses and result codes always start and end with a carriage return character and a line feed character: **<CR><LF><response><CR><LF>**. In tables presenting commands and responses throughout this document, only the commands and responses are presented, and **<CR>** and **<LF>** are deliberately omitted.

Table 2: Types of AT Commands

Command Type	Syntax	Description
Test Command	**AT+<cmd>=?**	Test the existence of the corresponding command and return information about the type, value, or range of its parameter.
Read Command	**AT+<cmd>?**	Check the current parameter value of the corresponding command.
Write Command	**AT+<cmd>=<p1>[,<p2>[,<p3>[...]]]**	Set user-definable parameter value.
Execution Command	**AT+<cmd>**	Return a specific information parameter or perform a specific action.

3.1.3. Declaration of AT Command Examples

The AT command examples in this document are provided to help you learn about the use of the AT commands introduced herein. The examples, however, should not be taken as Quectel's recommendations or suggestions about how to design a program flow or what status to set the module into. Sometimes multiple examples may be provided for one AT command. However, this does not mean that there is a correlation among these examples, or that they should be executed in a given sequence.

4 AT Command Description

This chapter introduces the voice and audio related AT commands supported by the modules, including the functions, usages and the scope of AT commands. For some AT commands or URCs mentioned but not described in detail in this chapter, unless otherwise specified, see *document [1]* for more information and usage.

4.1. AT+QCFG="ims"　Query IMS Registration State

This command queries the module's IMS registration state. VoLTE state indicates IMS registration state, that is, VoLTE is ready indicates IMS feature is enabled.

AT+QCFG="ims"　Query IMS Registration State	
Write Command **AT+QCFG="ims"**	Response **+QCFG: "ims",<VoLTE_state>** **OK** Or **ERROR** If there is an error related to ME functionality: **+CME ERROR: <err>**
Maximum Response Time	300 ms
Characteristics	-

Parameter

<VoLTE_state>	Integer type. VoLTE state.
	0　　　　VoLTE is not ready
	1　　　　VoLTE is ready
<err>	Error code. See *Chapter 8* for details.

Example

AT+QCFG="ims"
+QCFG: "ims",1

OK

> **NOTE**
>
> This command is valid only on the firmware version supporting VoLTE.

4.2. ATD   Originate a Call

This command sets up outgoing voice or data call.

ATD Originate a Call	
Execution Command **ATD<n>[<mgsm>][;]**	Response If no dial tone and **ATX2** or **ATX4** is set: **NO DIAL TONE** If busy and **ATX3** or **ATX4** is set: **BUSY** If a call connection cannot be established: **NO CARRIER** If a call connection is established successfully and a non-voice call is to be set up: **CONNECT<text>** TA switches to data mode. **<text>** is output only when **<value>** is greater than 0 in **ATX<value>** parameter setting. When TA returns to command mode after call release: **OK** If a call connection is established successfully and a voice call is set up: **OK** Or **ERROR**
Maximum Response Time	5 s, determined by the network (**AT+COLP=0**).
Characteristics	-

Reference
V.25ter

Parameter

\<n\>	String of dialing strings and optional V.25ter modifiers.
	Dialing strings: **0-9**, *****, **#**, **+**, **A**, **B**, **C**
	Following V.25ter, optional modifiers **,**(comma), **T**, **P**, **!**, **W**, **@** are ignored.
\<mgsm\>	String of GSM modifiers:
	I Activates **CLIR** (Disable presentation of calling number to called party).
	i Deactivates **CLIR** (Enable presentation of calling number to called party).
	G Activates closed user group invocation for this call only.
	g Deactivates closed user group invocation for this call only.
\<;\>	It is required when setting up a voice call. After the call, TA will return to command state.

Example

ATD10086;	//Dial a number.
OK	

NOTE

1. This command may be aborted generally by receiving an **ATH** command or a character during execution. It cannot be aborted during some connection establishments such as handshaking.
2. Parameters **I** and **i** can be omitted only when there is no ***** or **#** code in the dial string.
3. See **ATX** for details about setting result codes and call monitoring parameters.
4. For voice calls, if dialing with **ATD**, TA returns **OK** immediately either after the dialing is completed or the call is established. Otherwise, TA returns **BUSY**, **NO DIAL TONE**, or **NO CARRIER**.
5. Use **ATD** during an active voice call:
   - When a user originates a second voice call while there is already an active voice call, the first call will be automatically put on hold.
   - The current states of all calls can be easily checked at any time by **AT+CLCC**. See *Chapter 4.6* for details.
6. Currently, supplementary services, such as COLP, CLIR or closed user group, are not supported.

4.3. ATA   Answer a Call

This command connects the module to an incoming voice or data call indicated by a **RING** URC.

ATA Answer a Call	
Execution Command **ATA**	Response TA sends off-hook to the remote station. In case of data call, if successfully connected: **CONNECT<text>** And TA switches to data mode. Note: **<text>** is output only when **<value>** is greater than 0 in **ATX <value>** parameter setting. When TA returns to command mode after call release: **OK** In case of voice call, if successfully connected: **OK** If no connection: **NO CARRIER** Or **ERROR** If there is any error related to ME functionality: **+CME ERROR: <err>**
Maximum Response Time	90 s, determined by the network.
Characteristics	-
Reference V.25ter	

Parameter

<err>	Error code. See *Chapter 8* for details.

Example

RING **OK**	//Ring tone.
ATA **OK**	//Answer the call.

> **NOTE**
>
> 1. Any additional commands on the same command line are ignored.
> 2. This command may be aborted generally by receiving a character during execution. It cannot be aborted during some connection establishments such as handshaking.
> 3. See also **ATX**.

4.4. ATH   Disconnect Existing Connection

This command disconnects and terminates existing data call or voice call. **AT+CHUP** is also used to disconnect voice call. See *Chapter 4.7* for details.

ATH Disconnect Existing Connection	
Execution Command **ATH[<n>]**	Response **OK** Or **ERROR** If there is any error related to ME functionality: **+CME ERROR: <err>**
Maximum Response Time	90 s, determined by the network.
Characteristics	-
Reference V.25ter	

Parameter

<n>	Integer type.
	0 Disconnect and terminate existing call from command line.
<err>	Error code. See *Chapter 8* for details.

Example

```
RING                            //Ring tone.
OK


ATH                             //When AT+CVHU=0, hang up the call.
OK
```

4.5. AT+CVHU　Voice Call Hang Up Control

This command controls whether **ATH** can be used to disconnect the voice call.

AT+CVHU　Voice Call Hang Up Control	
Test Command **AT+CVHU=?**	Response **+CVHU: (**list of supported **<mode>**s**)** **OK**
Read Command **AT+CVHU?**	Response **+CVHU: <mode>** **OK**
Write Command **AT+CVHU=<mode>**	Response **OK** Or **ERROR** If there is an error related to ME functionality: **+CME ERROR: <err>**
Maximum Response Time	300 ms
Characteristics	The command takes effect immediately. The configuration is not saved.
Reference *3GPP TS 27.007*	

Parameter

<mode>	Integer type.
	<u>0</u>　　**ATH** can be used to disconnect the voice call.
	1　　**ATH** is ignored with the response **OK** returned only.
<err>	Error code. See *Chapter 8* for details.

4.6. AT+CLCC   List Current Calls of ME

The Execution Command returns the list of all voice calls of ME. If the command is executed successfully, but no call exists, then only **OK** is sent to TE.

AT+CLCC List Current Calls of ME	
Test Command **AT+CLCC=?**	Response **OK**
Execution Command **AT+CLCC**	Response TA returns the list of all current calls of ME. If the command is executed successfully, but no calls exists, then only **OK** is sent to TE. **[+CLCC: <id1>,<dir>,<stat>,<mode>,<mpty>[,<number>,<type>[,< alpha>]]]** **[+CLCC: <id2>,<dir>,<stat>,<mode>,<mpty>[,<number>,<type>[,< alpha>]]]** **[...]** **OK** Or **ERROR** If there is an error related to ME functionality: **+CME ERROR: <err>**
Maximum Response Time	300 ms
Characteristics	-

Parameter

<idx>	Integer type. Call identification number (starting with 1). See *3GPP TS 22.030 subclause 6.5.5.1* for details.	
<dir>	Integer type.	
	0	Mobile originated (MO) call
	1	Mobile terminated (MT) call
<stat>	Integer type. Call state.	
	0	Active
	1	Held
	2	Dialing (MO call)
	3	Alerting (MO call)
	4	Incoming (MT call)
	5	Waiting (MT call)
<mode>	Integer type. Bearer/teleservice.	
	0	Voice

	1	Data
	2	Fax
<mpty>	Integer type.	
	0	Call is not one of the multiparty (conference) call parties
	1	Call is one of the multiparty (conference) call parties
<number>	String type. Phone number in format specified by **<type>**.	
<type>	Integer type. Address type in octet format (See *3GPP TS 24.008 subclause 10.5.4.7* for details). Usually, it has three values:	
	129	Unknown type
	145	International type (contains character "+")
	161	National type
<alpha>	Alphanumeric representation of **<number>** corresponding to the entry found in phonebook.	
<err>	Error code. See *Chapter 8* for details.	

Example

ATD10086;	//Dial a number.
OK	
AT+CLCC	
+CLCC: 2,0,0,0,0,"10086",129	//Dial a number, and the call has been answered.
OK	

4.7. AT+CHUP   Hang up a Call

This command cancels any voice call in the state of Active, Waiting or Held. To cancel data call, use **ATH**. See *Chapter 4.4* for details.

AT+CHUP Hang up a Call	
Test Command **AT+CHUP=?**	Response **OK**
Execution Command **AT+CHUP**	Response **OK** Or **ERROR** If there is an error related to ME functionality: **+CME ERROR: <err>**
Maximum Response Time	90 s, determined by the network.
Characteristics	-

Reference	
3GPP 27.007	

Parameter

<err>	Error code. See *Chapter 8* for details.

Example

RING	//Ring tone.
AT+CHUP	//Hang up the call.
OK	

4.8. AT+QIIC   Read/ Write Codec Through IIC

This command reads and writes the codec through the IIC interface.

AT+QIIC Read/ Write Codec Through IIC	
Test Command **AT+QIIC=?**	Response **+QIIC: (**list of supported **<rw>**s**),(**range of supported **<device>**s**),(**range of supported **<addr>**s**),(**list of supported **<bytes>**s**),(**range of supported **<value>**s)** **OK**
Write Command **AT+QIIC=<rw>,<device>,<addr>,<bytes>[,<value>]**	Response If **<rw>**=0, all configuration parameters are specified: **OK** If **<rw>**=1, **<value>** is omitted: **+QIIC: <value>** **OK** Or **ERROR**
Maximum Response Time	300 ms
Characteristics	The command takes effect immediately. The configurations are not saved.

Parameter

\<rw\>	Integer type.
	0 Write codec through IIC
	1 Read codec through IIC
\<device\>	Hexadecimal integer type with prefix "0x". 7-bit slave device address (address of ALC5616: 0x1B; address of NAU8814, NAU88C10: 0x1A, TLV320AIC3104: 0x18 and MAX9860: 0x10). Range: 0–0xFF.
\<addr\>	Hexadecimal integer type with prefix "0x". Register address. Range: 0–0xFF.
\<bytes\>	Integer type. Length of the read or write byte(s). Value: 1, 2.
\<value\>	Hexadecimal integer type with prefix "0x". Data value. Range: 0–0xFFFF.

Example

AT+QIIC=1,0x1B,0x00,2	//Read register value; slave device address: 0x1B; register address: 0x00; read two bytes.
+QIIC: 0x0021	
OK	
AT+QIIC=0,0x1B,0x00,2,0x0000	//Write register value; slave device address: 0x1B; register address: 0x00; write two bytes; data value: 0x0000.
OK	

NOTE

\<device\> is 7-bit slave device address (read or write bit not included), which can be found in the device datasheet.

4.9. AT+QDAI   Configure Digital Audio Interface

This command configures the digital audio interface.

AT+QDAI Configure Digital Audio Interface	
Test Command **AT+QDAI=?**	Response **+QDAI:** (list of supported **\<io\>**s) **OK**
Read Command **AT+QDAI?**	Response **+QDAI: \<io\>** **OK**

Write Command	Response
AT+QDAI=\<io>	**OK**
	Or
	ERROR
Maximum Response Time	300 ms
Characteristics	The command takes effect immediately.
	The configuration is saved automatically.

Parameter

\<io>	Integer type.	
	0	Analog input and output (for audio codec NAU88C10)
	1	Analog input and output (for audio codec NAU8814)
	2	Analog input and output (for audio codec ALC5616)
	3	Analog input and output (for audio codec TLV320AIC3104)
	4	Analog input and output (for audio codec MAX9860)
	<u>99</u>	Close all external codecs.

Example

```
AT+QDAI=?
+QDAI: (0-4,99)

OK
AT+QDAI?              //Query the current interface configuration.
+QDAI: 1

OK
AT+QDAI=2             //Set the analog output to ALC5616.
OK
```

4.10. AT+CLVL   Configure Loudspeaker Volume

This command configures the volume of the internal loudspeaker of the MT.

AT+CLVL Configure Loudspeaker Volume	
Test Command	Response
AT+CLVL=?	**+CLVL:** (range of supported **\<level>**s)
	OK
Read Command	Response

AT+CLVL?	+CLVL: <level>
	OK
Write Command	Response
AT+CLVL=<level>	OK
	Or
	ERROR
Maximum Response Time	300 ms
Characteristics	The command takes effect immediately.
	The configurations are saved automatically.

Parameter

<level>	Integer type. Volume level with manufacturer specific range (Smallest value represents the lowest sound level). Range: 0–5. Default value: 3.

Example

```
AT+CLVL=?
+CLVL: (0-5)

OK
AT+CLVL?              //Query the current volume level.
+CLVL: 3

OK
AT+CLVL=2            //Set the volume level to 2.
OK
```

4.11. AT+CMUT   Muting Control

This command enables or disables muting in the Tx/Rx direction during a voice call.

AT+CMUT Muting Control	
Test Command	Response
AT+CMUT=?	+CMUT: (list of supported <direction>s),(list of supported <enable>s)
	OK
Write Command	Response

AT+CMUT=<direction>,<enable>	OK
	Or
	ERROR
Maximum Response Time	300 ms
Characteristics	The command takes effect immediately.
	The configurations are not saved and must be set during the call.

Parameter

<direction>	Integer type.	
	0	Tx direction
	1	Rx direction
<enable>	Integer type	
	0	Mute off
	1	Mute on

Example

```
AT+CMUT=?
+CMUT: (0,1),(0,1)

OK
AT+CMUT=1,1          //Mute on in Rx direction.
OK
```

4.12. AT+QPCMCFG   Configure PCM Interface

This command configures the PCM interface.

AT+QPCMCFG Configure PCM Interface	
Test Command AT+QPCMCFG=?	Response +QPCMCFG: (list of supported **<aux_mode>**s),(range of supported **<frame_setting>**s),(range of supported **<quant_type>**s),(list of supported **<bits_per_slot>**s) OK
Read Command AT+QPCMCFG?	Response +QPCMCFG: **<aux_mode>,<frame_setting>,<quant_type>,<bits_per_slot>**

	OK
Write Command **AT+QPCMCFG=<aux_mode>,<frame_setting>,<quant_type>,<bits_per_slot>**	Response **OK** Or **ERROR**
Maximum Response Time	300 ms
Characteristics	The command takes effect after re-running voice call. The configurations are saved automatically.

Parameter

<aux_mode>   Integer type.

          0   Short frame sync

          <u>1</u>   Long frame sync

<frame_setting>   Integer type. Bits width per frame.

          0   8 bits (not supported currently)

          1   16 bits (not supported currently)

          2   32 bits

          3   64 bits

          4   128 bits

          <u>5</u>   256 bits

<quant_type>   Integer type. Quant type in PCM format.

          0   A-law no padding

          1   Mu-law no padding

          <u>2</u>   linear no padding (supported only)

          3   A-law padding

          4   Mu-law padding

          5   linear padding

<bits_per_slot>   Integer type.

          <u>1</u>   16-bit PCM (supported only)

          2   24-bit PCM

          4   32-bit PCM

Example

```
AT+QPCMCFG=?
+QLDTMF: (0,1),(0-5),(0-5),(1,2,4)

OK
AT+QPCMCFG=1,5,2,1              //Configure the PCM interface.
OK
```

> **NOTE**
>
> Default: 2.048 MHz BCLK, 8 KHz (supported only) sampling rate, audio PCM only in master mode.

4.13. AT+QLDTMF   Play Local DTMF

This command plays a local DTMF tone.

AT+QLDTMF Play Local DTMF	
Test Command **AT+QLDTMF=?**	Response **+QLDTMF: (**list of supported **<DTMF_character>**s**),(**range of supported **<duration>**s**),(**range of supported **<vol_gain>**s**)** **OK**
Write Command **AT+QLDTMF=<DTMF_character>, <duration>,<vol_gain>**	Response **OK** Or **ERROR**
Maximum Response Time	300 ms
Characteristics	The command takes effect immediately. The configurations are not saved.

Parameter

<DTMF_character>	String type. DTMF character. Range: 0–9, A–D, *, #.
<duration>	Integer type. DTMF tone play time duration. Range: 0–65535.
<vol_gain>	Integer type. DTMF tone play volume gain. Range: 0–65535.

Example

```
AT+QLDTMF=?
+QLDTMF: (0-9,A-D,*,#),(0-65535),(0-65535)

OK
AT+QLDTMF=2,200,200        //Play "2" DTMF tone whose volume gain is 200 for 200 ms.
OK
```

4.14. AT+VTS   DTMF and Tone Generation

This command sends ASCII characters which cause MSC to transmit DTMF tones to a remote subscriber. This command can only be executed in a voice call.

AT+VTS DTMF and Tone Generation	
Test Command **AT+VTS=?**	Response **+VTS: (**list of supported **<DTMF_character>**s**)** **OK**
Execution Command **AT+VTS=<DTMF_character>**	Response **OK** Or **ERROR** If there is an error related to ME functionality: **+CME ERROR: <err>**
Maximum Response Time	300 ms
Characteristics	-
Reference *3GPP 27.007*	

Parameter

<DTMF_character>	String type. DTMF character. Range: 0–9, A–D. *, #.
<err>	Error code. See *Chapter 8* for details.

Example

ATD12345678900;	//Dial.
OK	
//Call connects.	
AT+VTS=1	//The remote caller can hear the DTMF tone.
OK	

NOTE

This command cannot send multiple tones at one time.

4.15. AT+QTONEDET   Enable/Disable DTMF Detection

This command enables or disables DTMF detection during CS voice or VoLTE call. If this function is enabled, DTMF tones sent by the other side are detected and reported on the assigned serial port.

AT+QTONEDET Enable/Disable DTMF Detection	
Test Command **AT+QTONEDET=?**	Response **+QTONEDET:** (list of supported **<enable>**s) **OK**
Read Command **AT+QTONEDET?**	Response **+QTONEDET: <enable>** **OK**
Write Command **AT+QTONEDET=<enable>**	Response **OK** Or **ERROR**
Maximum Response Time	300 ms
Characteristics	The command takes effect immediately. The configuration is not saved.

Parameter

<enable>	Integer type. Enable or disable DTMF detection. 0 Disable 1 Enable

Example

```
AT+QTONEDET=?
+QTONEDET: (0,1)

OK
AT+QTONEDET=1          //Enable DTMF detection.
OK
```

> **NOTE**
>
> DTMF characters - ASCII
>
> | DTMF | ASCII | DTMF | ASCII |
> |------|-------|------|-------|
> | 0 | 48 | 8 | 56 |
> | 1 | 49 | 9 | 57 |
> | 2 | 50 | A | 65 |
> | 3 | 51 | B | 66 |
> | 4 | 52 | C | 67 |
> | 5 | 53 | D | 68 |
> | 6 | 54 | * | 42 |
> | 7 | 55 | # | 35 |

4.16. AT+QAUDLOOP   Enable/Disable Audio Loop Test

This command enables or disables the audio loop test. AFE loopback is supported to verify the audio hardware connection, as shown in the following figure.
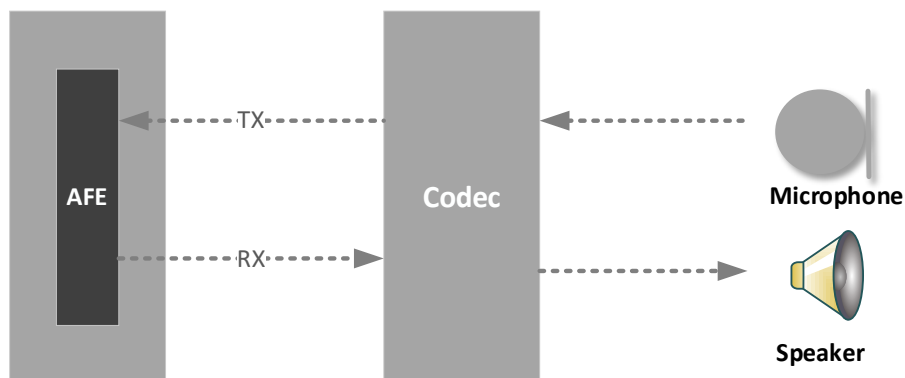
Figure 2: Data Path of AFE Loopback

AT+QAUDLOOP Enable/Disable Audio Loop Test	
Test Command **AT+QAUDLOOP=?**	Response **+QAUDLOOP:** (list of supported**<enable>**s) **OK**

Write Command **AT+QAUDLOOP=<enable>**	Response **OK** Or **ERROR**
Maximum Response Time	300 ms
Characteristics	The command takes effect immediately. The configuration is not saved.

Parameter

<enable>	Integer type. Enable or disable audio loop test.
	<u>0</u> Disable audio loop test
	1 Enable audio loop test

Example

AT+QAUDLOOP=1   //Enable audio loop test, talk into the microphone, and the earphones can hear you.
OK

4.17. AT+QINDCFG   Configure URC Indication

This command configures the URC indication.

AT+QINDCFG Configure URC Indication	
Test command **AT+QINDCFG=?**	Response **+QINDCFG: "all",(**list of supported **<enable>**s),(list of supported **<save_to_NVRAM>**s) **+QINDCFG: "csq",(**list of supported **<enable>**s),(list of supported **<save_to_NVRAM>**s) **+QINDCFG: "smsfull",(**list of supported **<enable>**s),(list of supported **<save_to_NVRAM>**s) **+QINDCFG: "ring",(**list of supported **<enable>**s),(list of supported **<save_to_NVRAM>**s) **+QINDCFG: "smsincoming",(**list of supported **<enable>**s),(list of supported **<save_to_NVRAM>**s) **+QINDCFG: "ccinfo",(**list of supported **<enable>**s),(list of supported **<save_to_NVRAM>**s) **OK**
Write command	Response

AT+QINDCFG=<URC_type>[,<enable>[,<save_to_NVRAM>]]	If the optional parameters are omitted, query the current configuration: **+QINDCFG: <URC_type>,<enable>** **OK** If the optional parameters are specified, configure the URC indication: **OK** Or **ERROR** If there is any error related to ME functionality: **+CME ERROR: <err>**
Maximum Response Time	300 ms
Characteristics	The command takes effect immediately. **<save_to_NVRAM>** determines whether the configuration is saved.

Parameter

<URC_type>	String type. URC type.	
	"all"	Main switch of all URCs. **<enable>** default value is 1 (ON).
	"csq"	Indication of signal strength and signal channel bit error rate change. **<enable>** default value is 0 (OFF). If it is 1, **+QIND: "csq",<rssi>,<ber>** is reported.
	"smsfull"	Indication of full SMS storage. **<enable>** default value is 0. If it is 0, **+QIND: "smsfull",<storage>** is reported.
	"ring"	Indication of **"RING"**. **<enable>** default value is 1.
	"smsincoming"	Indication of Incoming message. **<enable>** default value is 1. Related URCs include **+CMTI**, **+CMT**, **+CDS.**
	"ccinfo"	Indication of voice call state change. **<enable>** default value is 0. If it is 1, the URC indication is enabled and **+QIND: "ccinfo",<id>,<dir>,<state>,<mode>,<mpty>, <number>,<type>[,<alpha>]** is reported (see *Chapter 4.6* for details about the parameters).
<enable>	Integer type. URC indication is ON or OFF.	
	0	OFF
	1	ON
<save_to_NVRAM>	Integer type. Whether to save the configuration into NVRAM.	
	0	Not save
	1	Save

Example

```
AT+QINDCFG=?
+QINDCFG: "all",(0,1),(0,1)
+QINDCFG: "csq",(0,1),(0,1)
+QINDCFG: "smsfull",(0,1),(0,1)
+QINDCFG: "ring",(0,1),(0,1)
+QINDCFG: "smsincoming",(0,1),(0,1)
+QINDCFG: "ccinfo",(0,1),(0,1)

OK
AT+QINDCFG="ring",0       //Disable RING URC report.
OK
AT+QINDCFG="ccinfo",1     //Enable ccinfo URC report.
OK
```

5 Description of URCs

5.1. RING URC

The RING URC indicates an incoming call signal from network.

RING URC	
RING	An incoming call.

5.2. Alert URC

When you make a call by executing **ATD**, there will be a **RING** URC in the remote terminal. In the local terminal, we can get an Alert URC returned instead of RBT (RING BACK TONE).

Alert URC	
+ALERTING: <id>,<number>,<num_type>	Peer side is ringing.

Parameter

<id>	Integer type. Call ID. Call identification number as described in *3GPP TS 22.030*.
<number>	String type. Phone number.
<num_type>	Integer type. Type of phone number.
	129 Unknown type
	145 International type (contains character "+")
	161 National type

Example

```
ATD505010000000045;              //Dial a number.
OK
//Alert URC
+ALERTING: 3,505010000000045,129 //Peer side is ringing.
ATH
OK
```

6 Issue Analysis

If an outgoing call fails, check whether the wrong number is dialed first, and then check whether IMS is registered successfully. If the two mentioned preconditions are correct, further analyze the phonetic problems. Before analyzing, make certain the direction that whether it is an uplink problem or a downlink problem. After identifying the direction, pinpoint the type of the problem, such as intermittence, noise, silence, howling, etc.
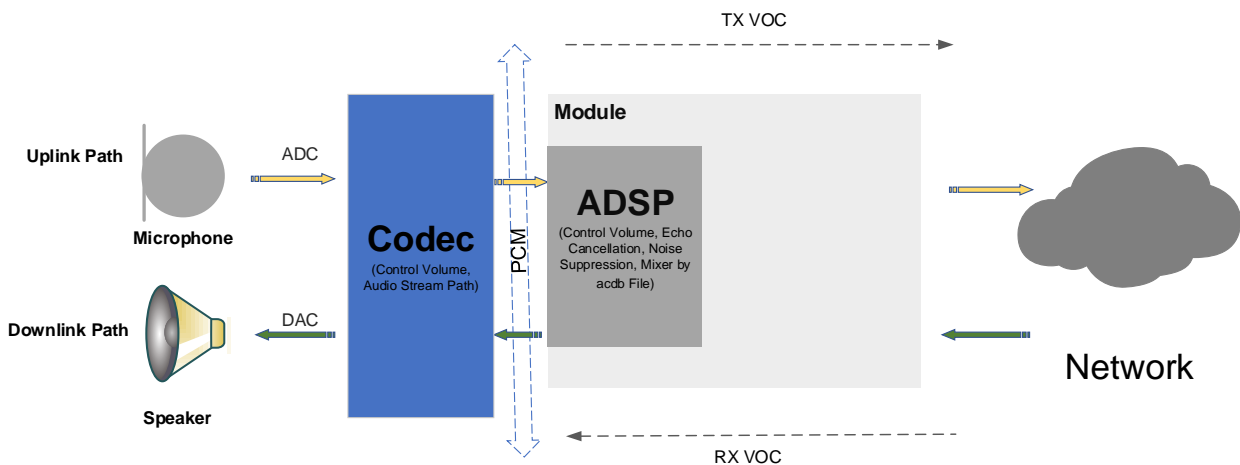
Figure 3: Voice Flow

Factors Affecting Voice Quality:

- Network condition: if the test is conducted in a weak-signal area.
- Channel condition: if the test is conducted in a fading/poor channel or if the channel condition results in CRC errors.
- ADSP processing: if ADSP parameters are set wrongly.
- Packet loss/miss: if the packet is lost in the network or the packet is missed in the system.
- Codec: if an incorrect codec type is selected or a wrong registration parameter (see **AT+QDAI**, **AT+QIIC** and *3GPP TS 26.103*) is configured.
- PCM interface: if unsuitable parameters (see **AT+QPCMCFG**) with a selected codec are configured.

> **NOTE**
>
> BG95 series, BG77 and BG600L-M3 modules do not support adjusting ADSP parameters in real time

by AT command.

6.1. Direction-Related Problem Analysis

Making it clear whether the problem occurred on the uplink or downlink side, on the home side or peer side, is the key step to solving the problem.

Uplink:

If the TX VOC itself has any problem, there is no need to analyze the problem from the perspective of the protocol. If the TX VOC sounds good but the downlink sounds bad on the other end, it is necessary to capture the logs on both sides of the calling and called parties. This scenario can be regarded as the most complex one because there may be a problem with the callee on the remote party, or may be a problem with the local caller. Therefore, we need to compare the logs of both sides and analyze the logs together.

Downlink:

If the RX VOC itself sounds good but the subsequent PCM sounds bad, there is no need to analyze the problem from the perspective of the protocol. You only need to check whether the Codec and PCM are configured correctly. The RX VOC problem is the most common voice problem. Generally, it is necessary to start from the access layer and make a comprehensive analysis based on the current network coverage, wireless environment, signal-to-noise ratio, bit error rate and other factors. In fact, the vast majority of such problems are caused by poor network coverage.

> **NOTE**
>
> Be careful when capturing logs. First, select a dedicated audio log mask to capture the log before calling. Then, keep talking for about a minute. Last, hang up.

6.2. Protocol-Related Problem Analysis

Voice analysis of common protocols is point-to-point analysis. That is, the time point at which the voice is faulty must be determined in the voice file generated by playback. By observing SNR and RSRP methods to measure the wireless environment at that time. When RSRP is lower than -95 dBm, the wireless environment is considered to be average, and when RSRP is lower than -105 dBm, the wireless environment is considered to be very poor. RxQual represents a bit error rate level, which is a very valid statistic for SNR. Generally, if RxQual is greater than 4 or CRC is "Fail", or SNR is less than 0, it is difficult to ensure the voice quality.

6.2.1. VoLTE Issue Analysis

From the image below, we can see that the current network condition is very poor because RSRP is less than -105 dBm, which may affect voice quality. Generally, it is not ideal for users to dial in the circumstance.

Figure 4: Poor Network Condition in VoLTE

6.2.2. CS-Voice over GSM Issue Analysis

From the images below, we can see that RxQual is 7 (greater than 4) in *Figure 5* and CRC is "Fail" in *Figure 6*. All these data represent the poor channel condition under GSM network, under which circumstances users may hear discontinuous sounds when making a call.
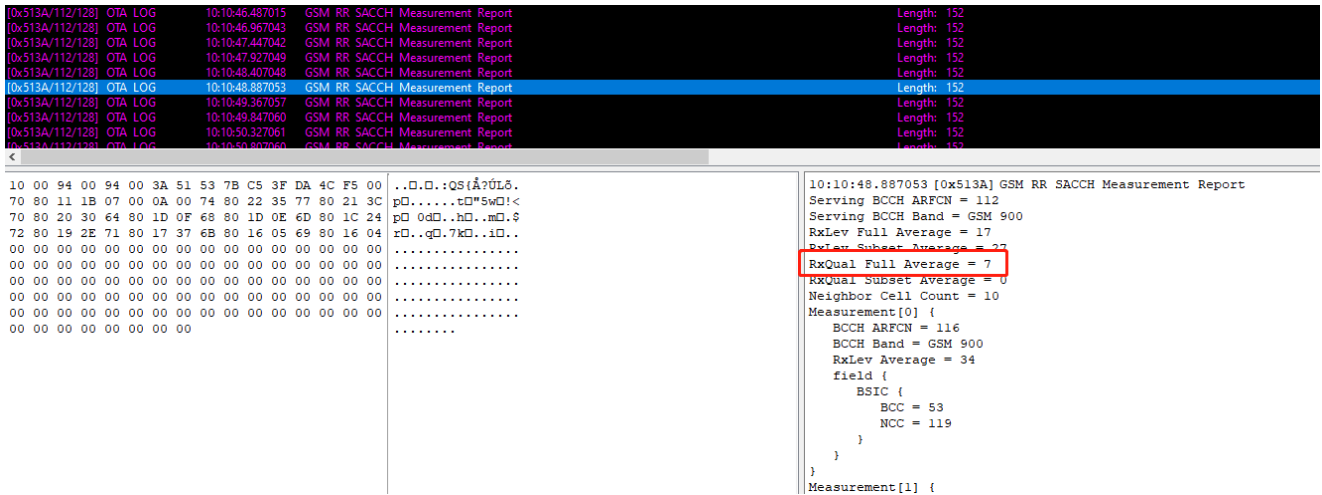
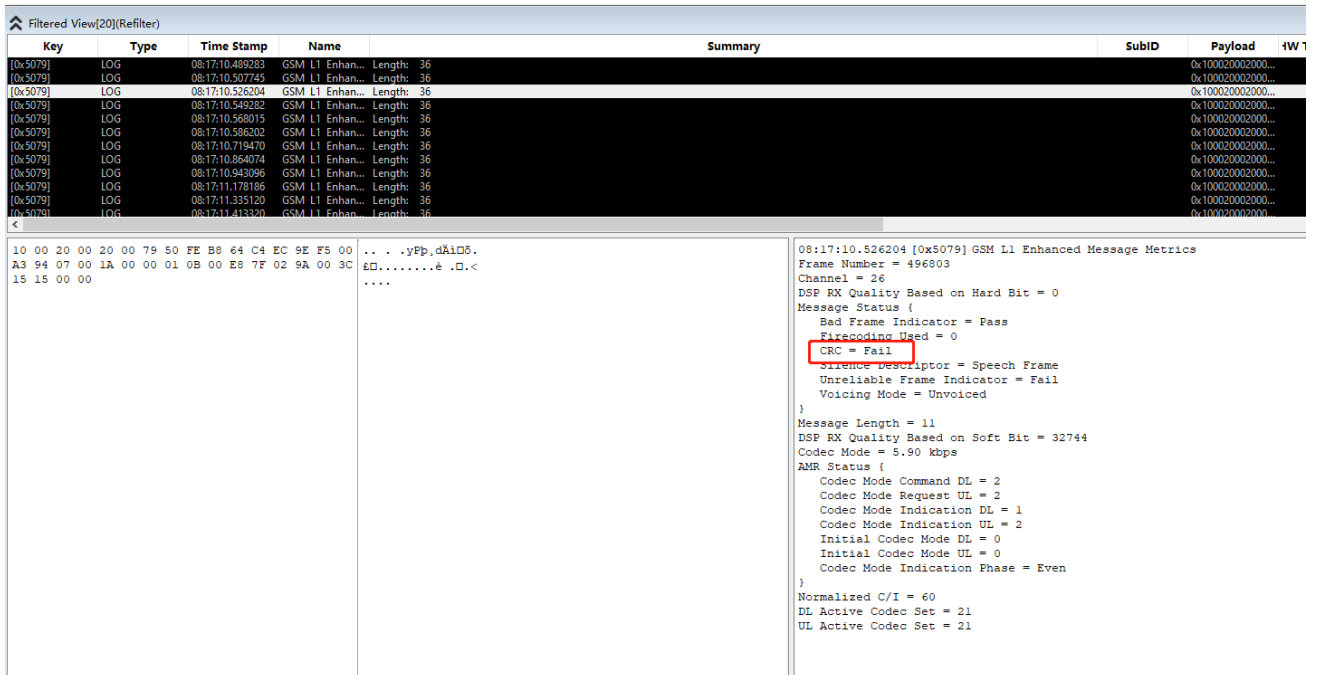**Figure 5: Poor Channel Condition in CS-Voice (1)**

Figure 6: Poor Channel Condition in CS-Voice (2)

7 Appendix A References

Table 3: Related Document

Document Name
[1]   Quectel_BG95&BG77&BG600L_Series_AT_Commands_Manual

Table 4: Terms and Abbreviations

Abbreviation	Description
ADSP	Audio Digital Signal Processor
AMR-NB	Adaptive Multi-rate Narrow Band
AMR-WB	Adaptive Multi-rate Wide Band
CLIR	Calling Line Identification Restriction
COLP	Connected Line Identification Presentation
CRC	Cyclic Redundancy Check
CSFB	Circuit Switched Fallback
DAN	Domain Availability Notification
DTMF	Dual-tone multi-frequency
EVS	Enhance Voice Services
E-UTRA(N)	Evolved Universal Terrestrial Radio Access (Network)
G.711	ITU-T Standard Pulse Code Modulation (PCM) of Voice Frequencies
IMS	IP Multimedia Subsystem
IPsec	Internet Protocol Security
IWLAN	Industrial Wireless LAN

MO	Mobile Originated
MSC	Mobile-services Switching Centre
MT	Mobile Terminated
MWI	Message Waiting Indicator
PDN	Packet Data Network
PCM	Pulse Code Modulation
QCI	QoS Class Identifier
QoS	Quality of Service
RAT	Radio Access Technology
RCS	Rich Communication Suite
ROHC	Robust Header Compression
RTP	Real-time Transport Protocol
RTT	Real-Time Text
SDP	Session Description Protocol
SMS	Short Message Service
SIP	Session Initiation Protocol
SRVCC	Single Radio Voice Call Continuity
TTY	Teletypewriter
UDP	User Datagram Protocol
VICE	VoLTE for Internet-connected Endpoint
VoLTE	Voice over LTE

8 Appendix B Summary of CME ERROR Codes

Final result code **+CME ERROR: <err>** indicates an error related to mobile equipment or network. The operation of **+CME ERROR: <err>** final result code is similar to the regular **ERROR** result code: if **+CME ERROR: <err>** is the result code for any of the commands in a command line, none of the following commands in the same command line is executed (neither **ERROR** nor **OK** result code shall be returned as a result of a completed command line execution). The format of **<err>** can be either numeric or verbose. This is set with **AT+CMEE**.

The following table lists most of general and GRPS related **ERROR** codes. For some GSM protocol failure cause described in GSM specifications, the corresponding **ERROR** codes are not included.

Table 5: Different Coding Schemes of +CME ERROR: <err>

Code of <err>	Meaning
0	Phone failure
1	No connection to phone
2	Phone-adaptor link reserved
3	Operation not allowed
4	Operation not supported
5	PH-SIM PIN required
6	PH-FSIM PIN required
7	PH-FSIM PUK required
10	(U)SIM not inserted
11	(U)SIM PIN required
12	(U)SIM PUK required

13	(U)SIM failure
14	(U)SIM busy
15	(U)SIM wrong
16	Incorrect password
17	(U)SIM PIN2 required
18	(U)SIM PUK2 required
20	Memory full
21	Invalid index
22	Not found
23	Memory failure
24	Text string too long
25	Invalid characters in text string
26	Dial string too long
27	Invalid characters in dial string
30	No network service
31	Network timeout
32	Network not allowed - emergency calls only
40	Network personalization PIN required
41	Network personalization PUK required
42	Network subset personalization PIN required
43	Network subset personalization PUK required
44	Service provider personalization PIN required
45	Service provider personalization PUK required
46	Corporate personalization PIN required
47	Corporate personalization PUK required