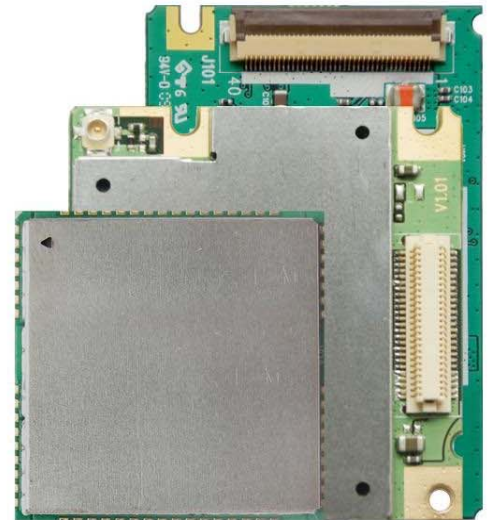




GSM Wireless Module

Audio Design

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0. Revision History

Revision	Date	Author	Description of change
1.00	2009-11-2	Tracy Zhang	Initial
1.1	2010-12-31	Roy Chen	1:Add AT command instructions in Figure19; 2:Chang the back cover image.

1. Introduction

This document provides reference design and advices about audio design of Quectel modules, including peripheral circuits, parameter setting, PCB layout, whole structure, etc. Understanding the details of this document would help customer on developing a good acoustic performance of product.

1.1. Related documents

Table 1: Related documents

SN	Document name	Remark
[1]	Mxx_HD	Hardware design document. Mxx refers to Quectel's GSM modules starting with "M".
[2]	Mxx_ATC	AT commands set

1.2. Terms and abbreviations

Table 2: Terms and abbreviations

Abbreviation	Description
AFE	Audio Front End
BB	Base Band
DC	Direct Current
ESD	Electro-Static Discharge
GPIO	General Purpose Input Output
GSM	Global System for Mobile communications
LDO	Low Drop Out regulator
M1x	Quectel's GSM module starting with "M1" such as M10 and M16
MIC	Microphone
NLP	Non Linear Processing
PA	Power Amplifier
PGA	Programmable Gain Amplifier
PSRR	Power Supply Rejection Ratio
RF	Radio Frequency
SPK	Speaker
TDD	Time Division Duplex
THD	Total Harmonic Distortion
TVS	Transient Voltage Suppressor

1.3. Terms explanation

Table 3: Terms explanation

Terms	Explanation
Echo	In talking mode, the far end could hear its own voice returned from the module side (the near end) with a delay of about several hundreds of milliseconds. It can be caused at the near end by various reasons, such as voice at the receiving circuit coupling to the microphone circuit due to improper circuit, unsealed mechanical structure, very loud voice from receiver or speaker getting into microphone, high sensitivity microphone. Talking quality can be obviously affected when echo problem is present.
Echo Cancellation	A software arithmetic for canceling echo, usually using in handset mode and headset mode
Echo Suppression	A software arithmetic for suppressing echo, usually using in hands-free application. Echo Suppression can reduce severe echo, but it leads to half-duplex or weakening to voice at the extent decided by the acoustic parameter settings.
TDD noise	GSM module generates an RF burst transmission lasting for 576 μ s in every 4.615ms during voice communication, which leads to higher power consumption at this 576 μ s. When the RF radiation energy conducts to the acoustic circuit, the envelope curve of the RF burst could be present due to filtering effect and form a noise in the acoustic circuit like a constant pulse at 217Hz and its harmonic frequencies. Another cause of the TDD noise is from the power supply. The burst consumption of current can cause obvious ripple at the supply voltage at 217Hz. If the ripple at the supply voltage conducts to audio circuit through power supply or ground, TDD noise could be heard at the far end or the near end.

2. Audio Hardware Design

This chapter introduces some audio peripheral circuits, including handset mode application with audio channel 1, hands-free mode application with audio channel 1, headset application mode with audio channel 2, hands-free application with audio channel 2. Moreover, some ways to suppressing TDD noise and reducing echo are expatiated.

Table 4: Audio signals configuration of different models

Model	Audio channel 1 (AIN1/AOUT1)		Audio channel 2 (AIN2/AOUT2)	
	MIC1P/1N	SPK1P/1N	MIC2P/2N	SPK2P
M1x	Differential, internal bias	Differential	Differential, internal bias	Single-end
M20	Differential, external bias	Differential	Differential, internal bias	Single-end

2.1. Peripheral circuits recommended

2.1.1. Audio channel 1

MIC1P/1N and SPK1P/1N are differential signals, and this channel can be applied to handset application or hands-free application. There is a little difference in MIC1P/1N circuit between M1x and M20. Because M20 does not provide internal DC MIC bias, external DC MIC bias circuit must be added in customer host board.

2.1.1.1. Handset application of audio channel 1

Handset application circuit of M1x audio channel 1 is recommended as below:

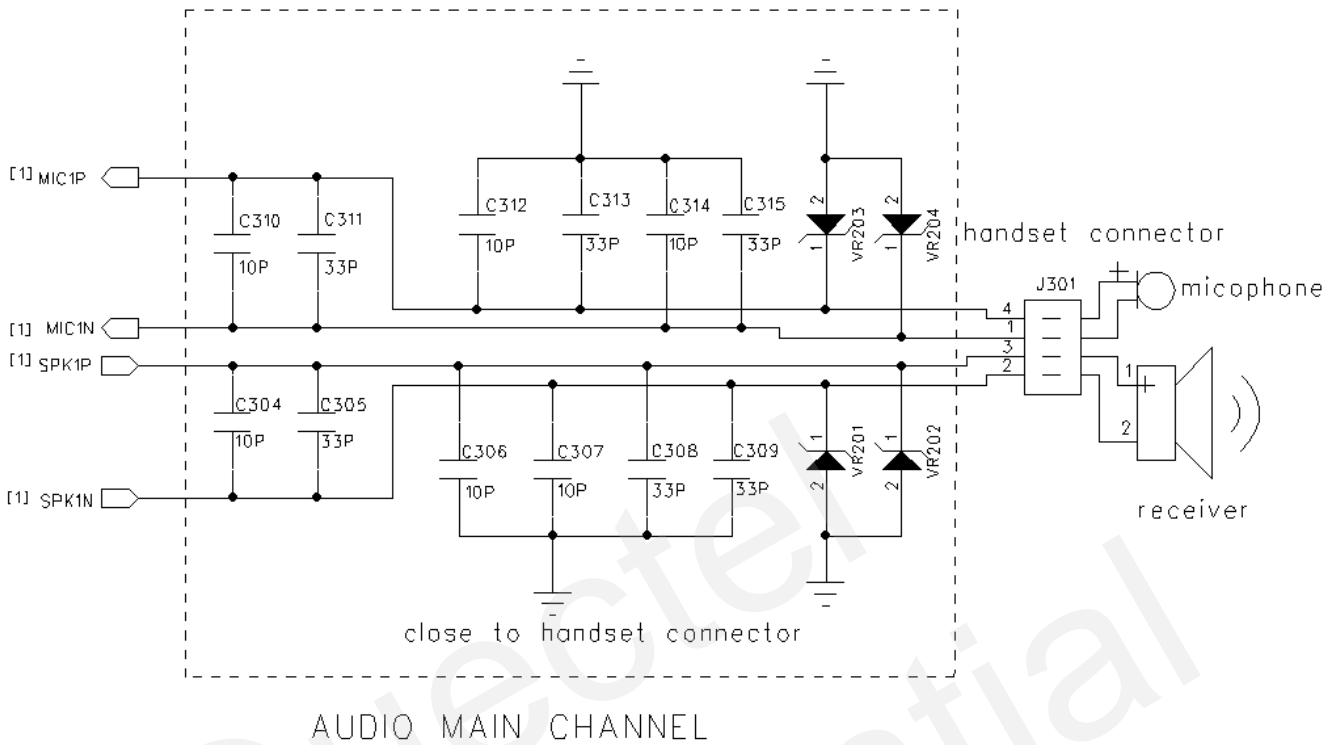


Figure 1: Handset application of M1x audio channel 1

Handset application circuit of M20 audio channel 1 is recommended as below:

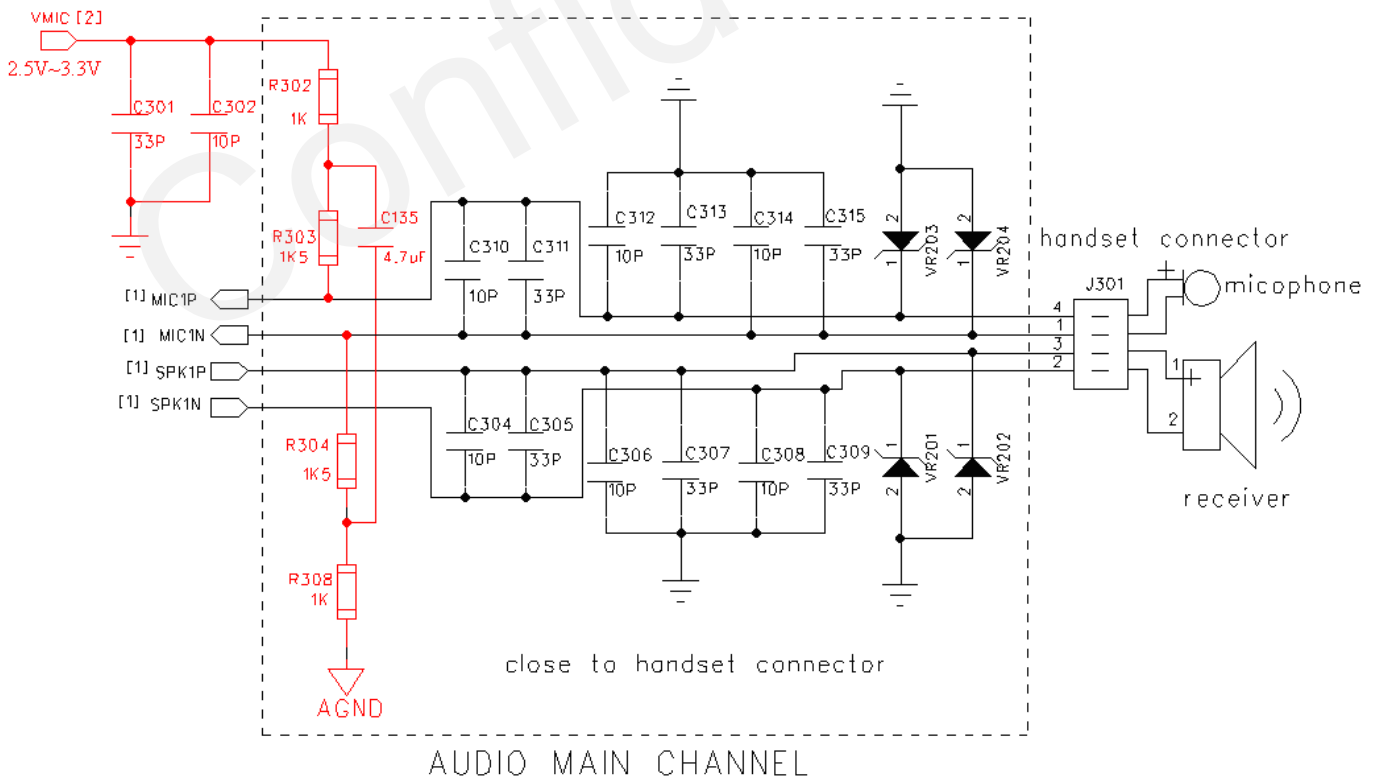


Figure 2: Handset application of M20 audio channel 1

The voltage input scope of MIC bias (VMIC) is 2.5V~3.3V. To avoid interference, customer had better use a separate LDO to supply for VMIC. The recommended circuit of the LDO is shown as below:

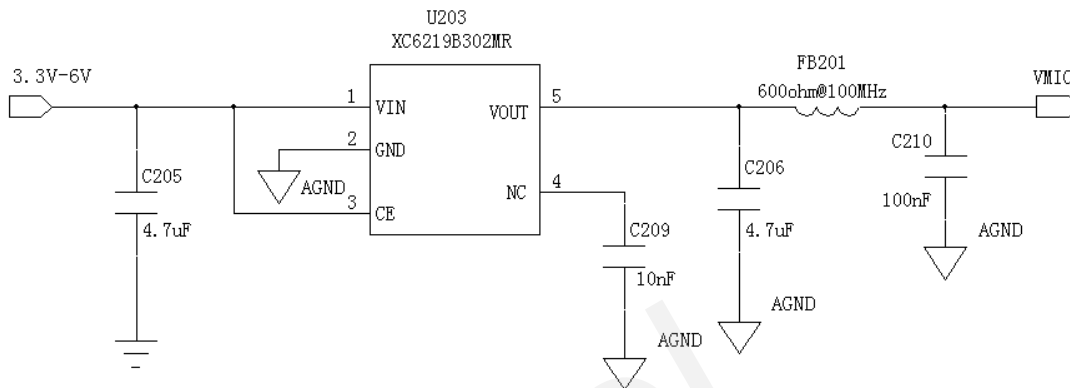


Figure 3: LDO circuit to VMIC of M20 MIC1P/1N

In order to supply a stable bias voltage for audio circuit of MIC, the LDO component should be placed as close as possible to the audio circuit. Ground signal of LDO output should be connected to **analog ground** (AGND), and separately laid trace to AGND pin of module interface.

The 33pF (0603) capacitors in Figure 1 and Figure 2 are applied for filtering out 900MHz RF interference when the module is transmitting at EGSM900MHz. Without placing this capacitor in SPK signals, TDD noise could be heard at the near end of the voice communication. Similarly, without the capacitor in MIC signals, TDD noise could be heard at the far end. Moreover, the 10pF (0603) capacitor here is for filtering out 1800MHz RF interference. However, the resonant frequency point of a capacitor largely depends on the material and production technique. Therefore, customer would have to discuss how to choose the most suitable capacitor with vendor for filtering GSM850MHz, EGSM900MHz, DCS1800MHz and PCS1900MHz separately. The 33pF and 10pF values here are just for reference from Murata Company.

The severity degree of the RF interference in the voice channel during GSM transmitting period largely depends on the application design. In most cases, EGSM900 and GSM850 TDD noise are more severe; while in some cases, DCS1800 TDD and PCS1900 noise are more obvious. Therefore, customer should decide which capacitors to use based on test result. Sometimes, even no RF filtering capacitor mounted, the TDD performance is still acceptable.

TVS components can protect microphone and receiver from ESD damage. They should be laid as close as possible to audio device or audio interface.

The capacitor C135 (4.7µF), R302 (1kΩ) and R308 (1kΩ) in Figure 2 are used to suppress the voltage ripple between VMIC and AGND, thus ensure a cleaner bias supply to the microphone.

2.1.1.2. Hands-free application of audio channel 1

If only hands-free application is required in design, the audio channel 1 is recommended to be

used. Because audio channel 1 supports differential output, it would be helpful to suppress common-mode noise in the audio output channel. **Please be noted that audio channel 1 does not provides ringing function for in-coming voice or SMS calling. To support ringing function, audio channel 2 must be used.** The recommended MIC circuit of hands-free application of audio channel 1 is the same as handset application. M1x doesn't need external MIC bias circuit, while M20 needs it as shown in Figure 2.

The recommended circuits of SPK1P/1N output part in hands-free application are shown in below.

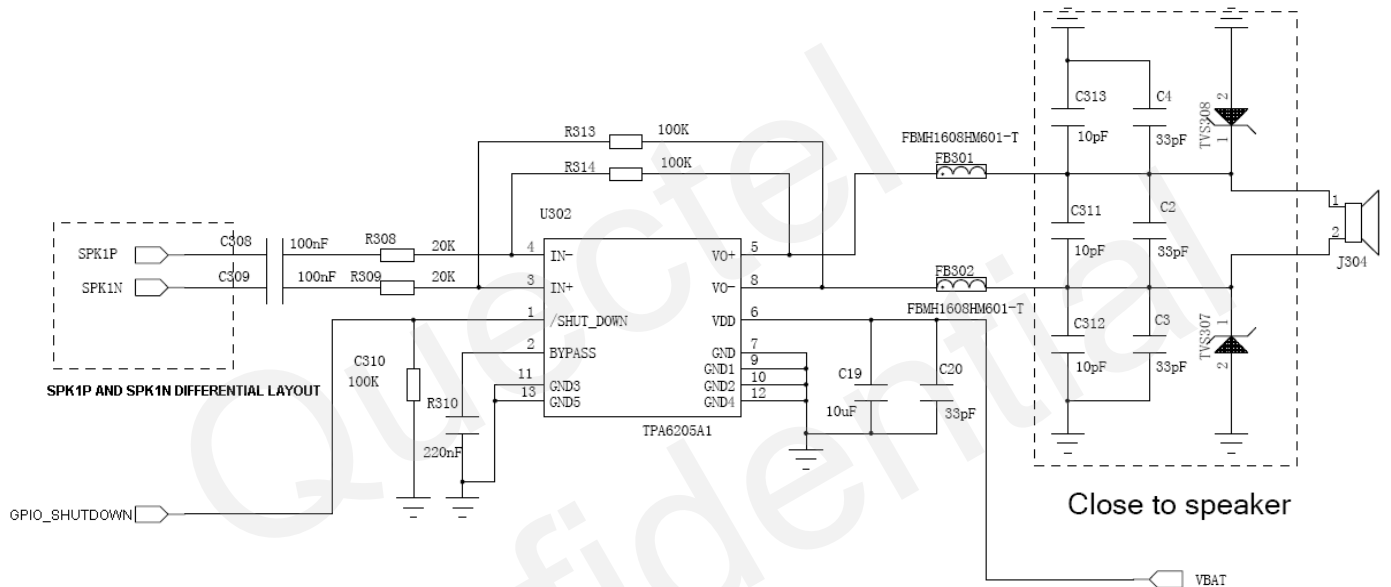


Figure 4: Hands-free application of SPK1P/1N (with TPA6205A1)

Here is U302, TPA6205A1 from TEXAS INSTRUMENTS, which is a audio power amplifier device and can driver a 8 Ω speaker impedance.

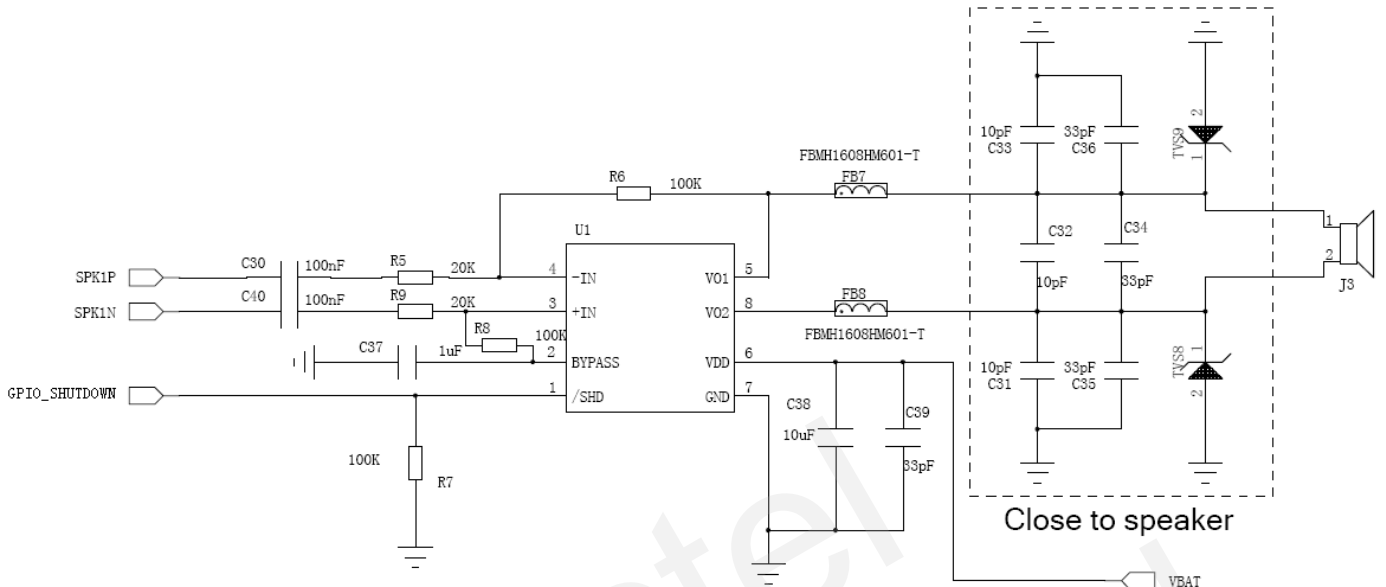


Figure 5: Hands-free application of SPK1P/1N (with LM4990/ NCP2890)

(LM4990 from NATIONAL SEMICONDUCTOR / NCP2890 from ON SEMICONDUCTOR)

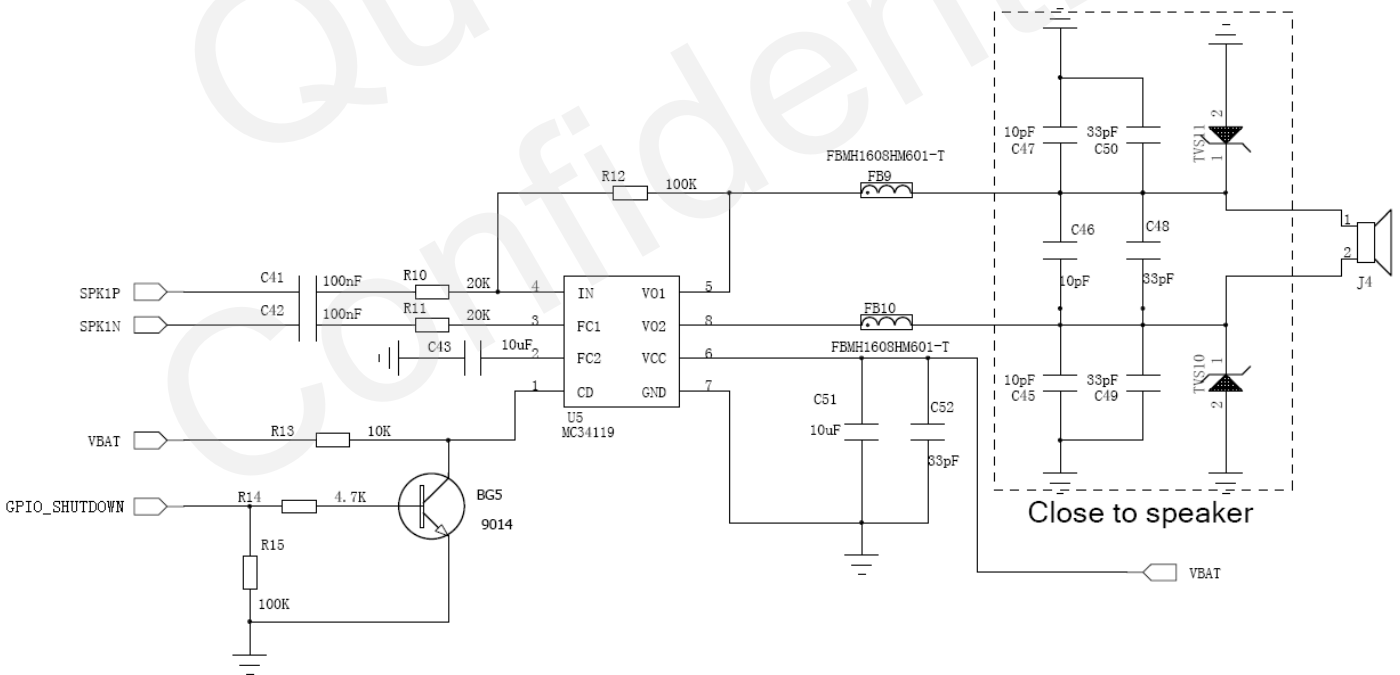


Figure 6: Hands-free application of SPK1P/1N (with MC34119)

(MC34119 from FREESCALE SEMICONDUCTOR)

In all kinds of audio PA chips, customer should choose a model providing a greater PSRR value at 217Hz. TPA6205A1 is strongly recommended.

The pin named GPIO_SHUTDOWN in Figure 4~6 can be connected and controlled by an I/O port

of customer's host MCU.

2.1.1.3. Advice for audio trace of channel 1

As the audio signals are very sensitive and are interfered easily from power supply ripple, unbalanced ground and RF burst radiation, the traces MIC1P and MIC1N, SPK1P and SPK1N in the PCB should follow the layout principle of differential traces. Moreover, these two pairs of signals should be separated from each other by flooding ground to avoid echo interference from SPK signal to MIC signal. An example is shown in Figure 7.

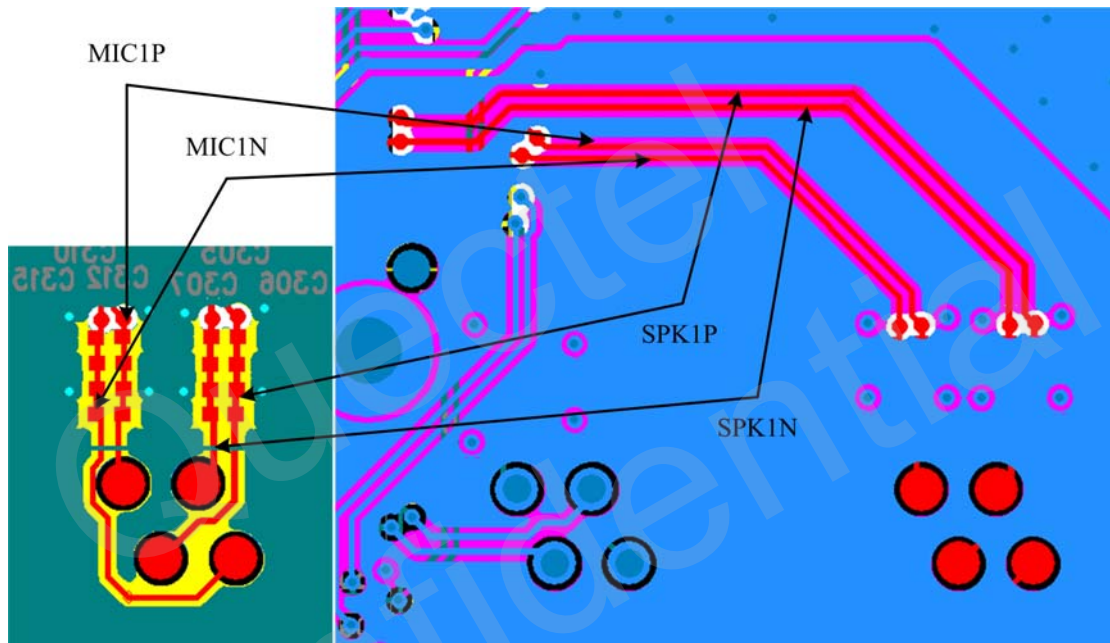


Figure 7: Schematic diagram of audio trace

2.1.2. Audio channel 2

MIC2P/2N is differential signal, while SPK2P is single-end signal. This channel can be applied to headset application or hands-free application.

2.1.2.1. Headset application of audio channel 2

Headset application circuit of M1x audio channel 2 is recommended as below:

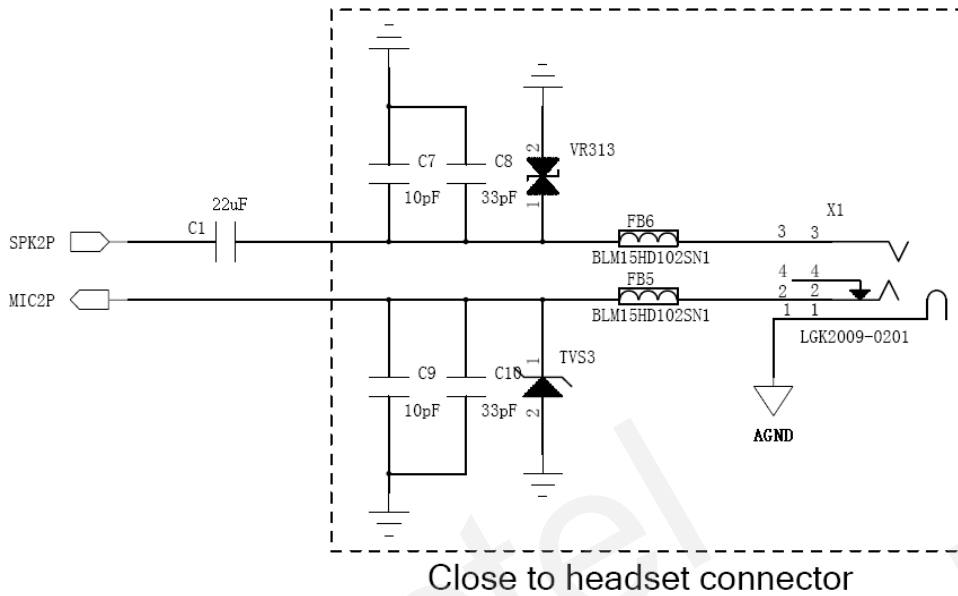


Figure 8: Headset application of M1x audio channel 2

The capacitor C1 (22µF) in Figure 8 is used for DC block. In M20, this capacitor is integrated in the SPK2P line within the module, thus there is no need to add it in the external circuit. The recommended circuit of headset application using M20 audio channel 2 is shown as below:

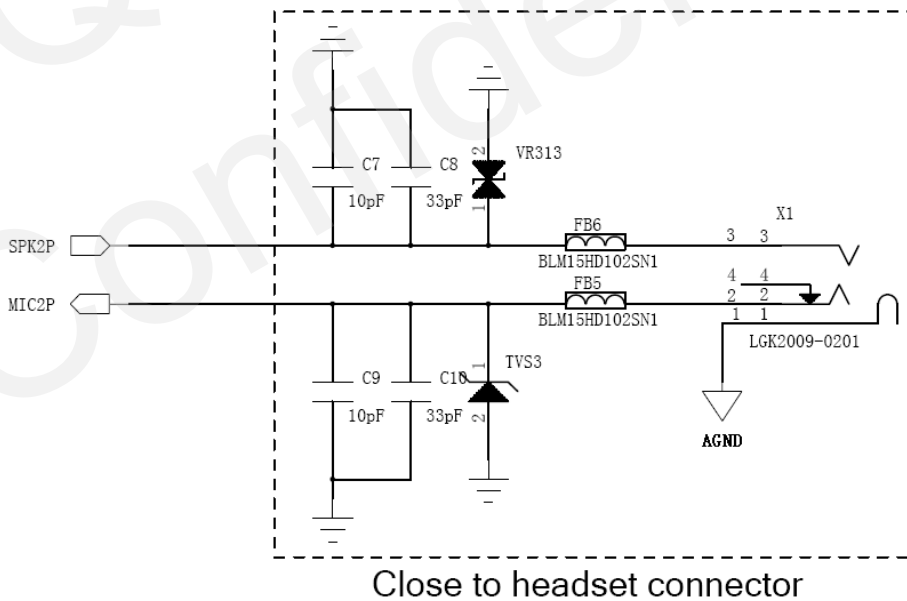


Figure 9: Headset application of M20 audio channel 2

If the earphone inserting detection and answer key detection functions are required in M1x, GPIO0 can be used to detect earphone inserting, and ADC0 can be used to detect pressing answer key. The reference circuit is present as below.

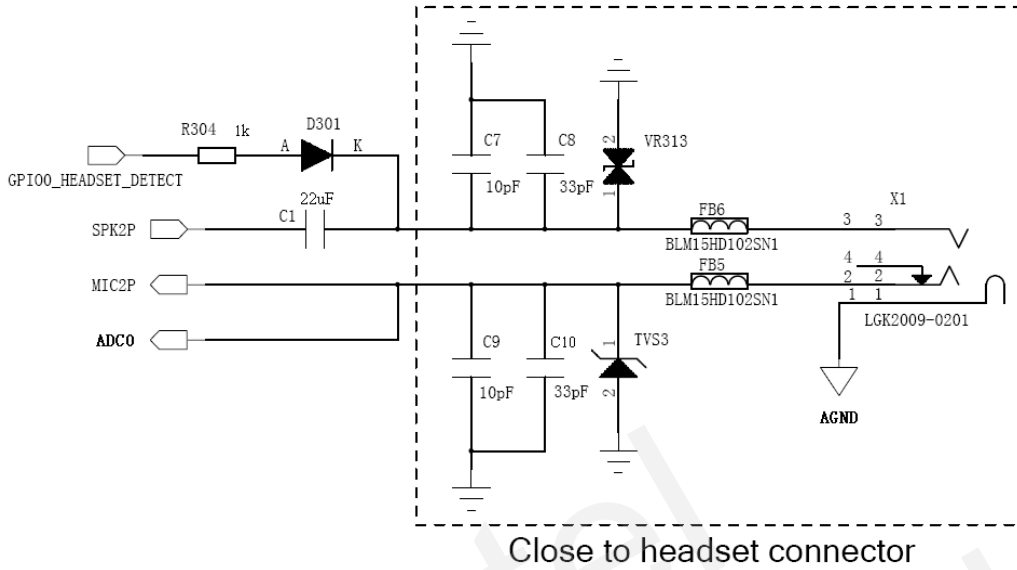


Figure 10: Headset application with earphone detection of M1x audio channel 2

M20 has internal detection circuit for earphone, but associated components are not mounted in default hardware configuration.

The earphone detection is not supported in default firmware of M1x and M20.

2.1.2.2. Hands-free application of audio channel 2

When audio channel 2 is used for hands-free application, the MIC2P/2N circuit is same to MIC1P/1N, and the trace principle in PCB should follow differential mode. Since SPK2P signal is single-ended, the input circuit needs some modification compared to SPK1P/1N. The reference circuit is recommended as below.

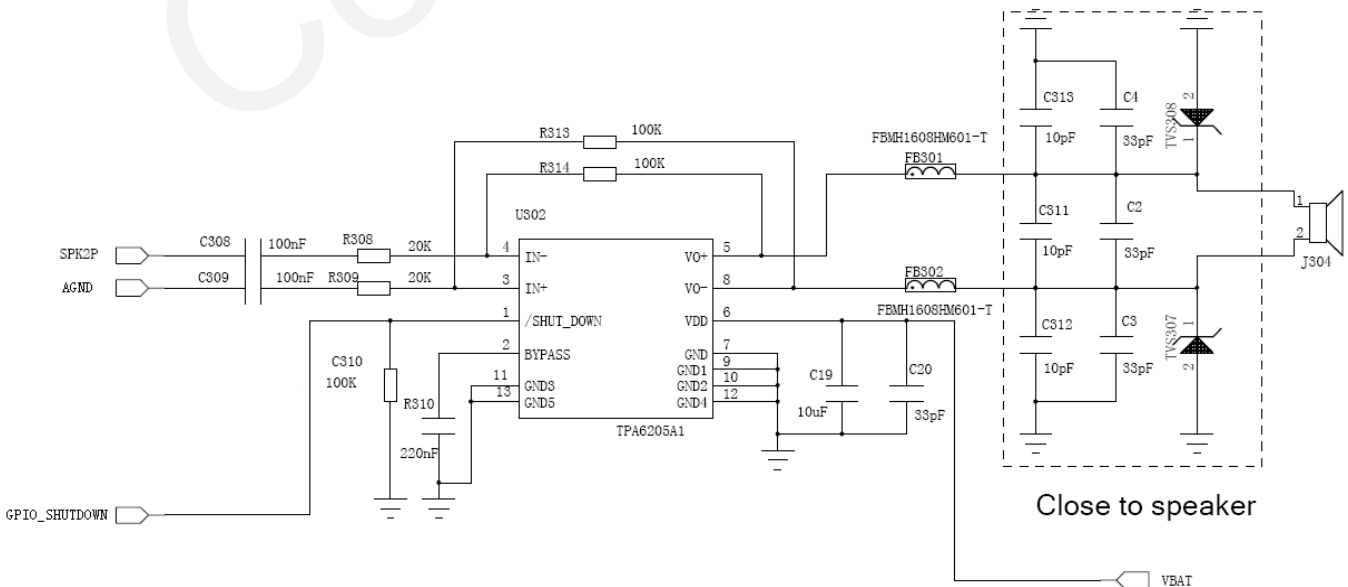


Figure 11: Hands-free application of SPK2P (with TPA6205A1)

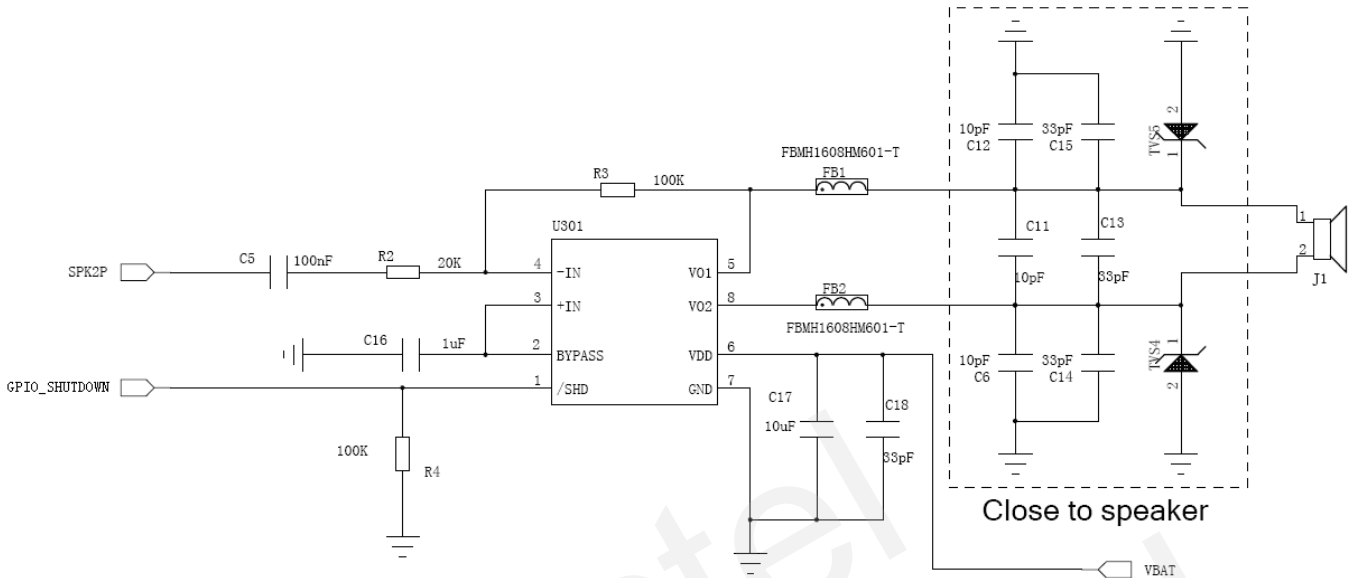


Figure 12: Hands-free application of SPK2P (with LM4990/NCP2890)

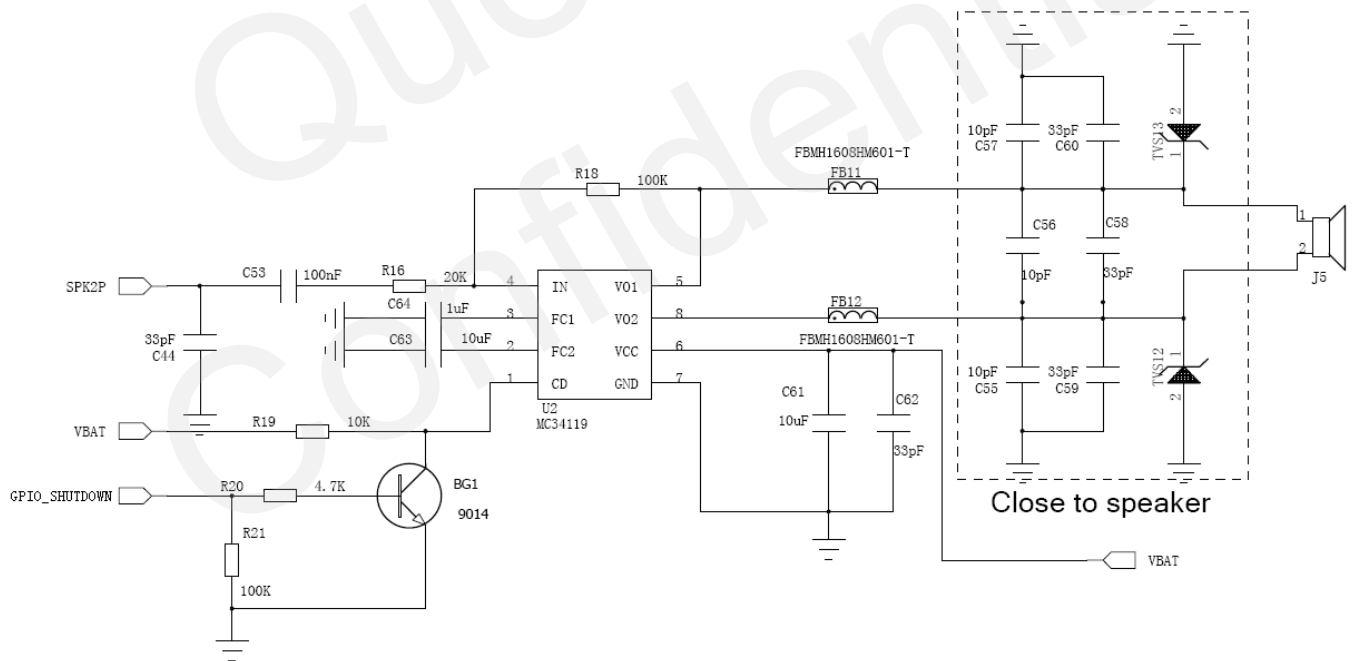


Figure 13: Hands-free application of SPK2P (with MC34119)

2.2. Methods to avoid TDD noise

It is particularly important to avoid and reduce TDD noise in audio design of GSM system. Except the ways mentioned above, such as filter capacitors, magnetic beads and differential traces, other advices are provided below.

- 1) It is strongly recommended to use an electret microphone with two filter capacitors (10pF(0603) and 33pF(0603)) in handset and hands-free application. These two capacitors could largely suppress coupling TDD noise from RF interference.
- 2) Place the RF filter capacitors as close as possible to audio components or interfaces on PCB. Audio traces should be as short as possible, and firstly pass through the filter capacitors and then connect to other pins. One terminal of RF filter capacitors can fully connect main ground of system.
- 3) Flood ground area as large as possible to reduce the ground impedance and improve ground integrity.
- 4) Reduce power voltage ripple, especially the power supply in audio circuits. Customer should use a wide trace from power source (like adapter interface, battery connector, or LDO output pin) to audio power supply. Good antenna matching is also important to reduce power ripple.
- 5) In external audio circuits on customer host board, the filtering capacitors and ESD protection devices should be connected to main digital ground, and other audio components should use AGND from the module interface. Please be noted that the AGND pin should not be connected with main digital ground in host board, or else, it could lead to TDD noise.
- 6) In order to reduce coupling interference, place the GSM antenna as far as possible away from audio components and traces. Microphone is most sensitive part to RF burst radiation. Even a high-quality microphone should be placed at least 5cm away from the antenna.
- 7) Do not make VBAT trace and audio traces parallel and lay VBAT trace as far as possible away from audio traces.

2.3. Suppress echo in mechanical design

It is important to consider how to suppress echo in the equipment with hands-free function or in an application which microphone and speaker are very close to each other.

The mechanical design has significant impact on echo problem. If it is not properly designed, the echo suppressing arithmetic in software could not make up large echo caused by bad mechanical structure, and even force to redesign.

Echo problem could be generated by several paths as shown in Figure 14.

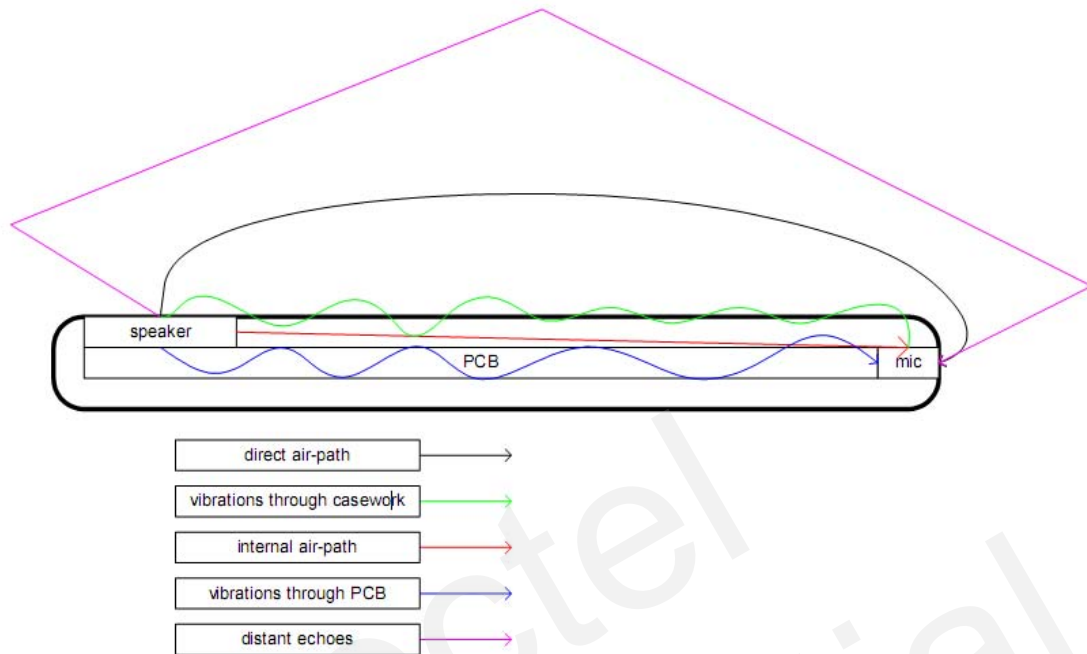


Figure 14: Five echo paths

In these five paths, internal air-path and direct air-path are the two most influential factors. Other three factors (vibrations through casework, vibrations through PCB, distant echoes) are secondary. Some suggestions about how to obstruct direct air-path and internal air-path are listed as follows:

1) Internal air-path

Separating microphone from internal space of chassis by foam or rubber ring can effectively suppress the inner echo interference. Recommended design of microphone socket is shown in Figure 15.

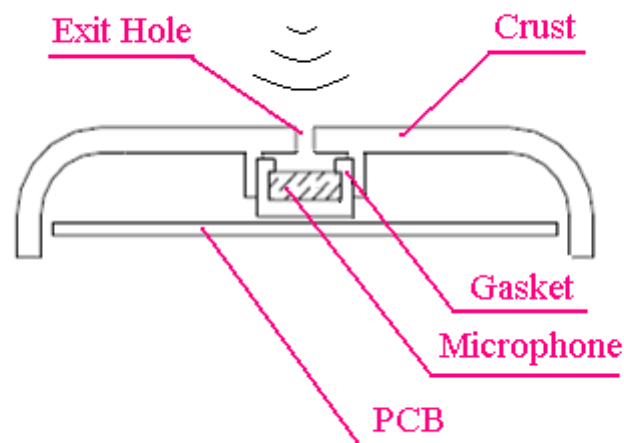


Figure 15: Microphone socket design

The best installation way of microphone socket is that encase microphone by silicone cover except for front cavity, and design a cylindrical hole whose center is the exit hole inside the chassis, make the microphone with silicone cover just fit the cylindrical hole, so as to only let voice enter into microphone from the exit hole, and not from leak of chassis interior. Certain air space room should be reserved in the front cavity of microphone as it is necessary for good microphone performance.

Recommended design of speaker socket is shown as in Figure 16.

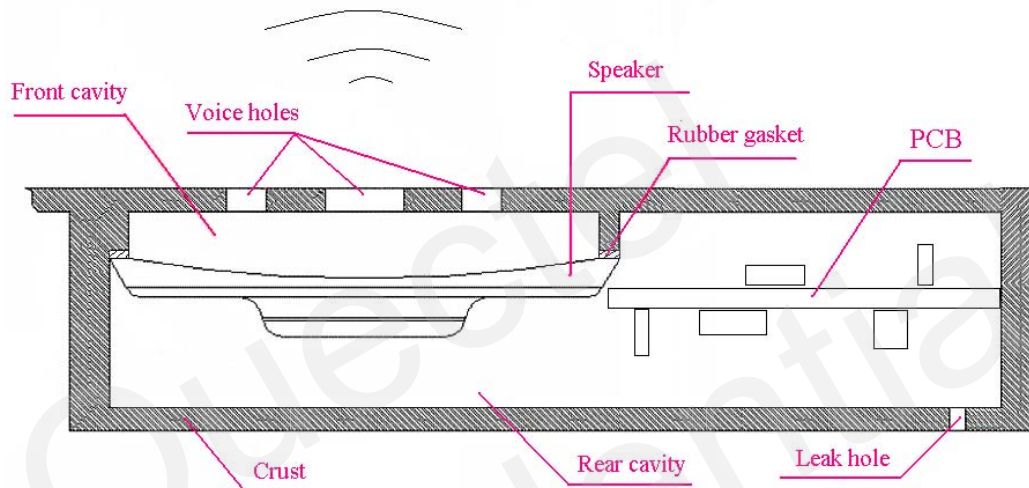


Figure 16: Speaker socket design

A good way to suppressing the internal echo path is to sealing the rear cavity of receiver, which is usually expensive. The rear cavity of receiver and speaker is important for good voice quality. A sealed rear cavity with sufficient space could produce a good voice output. An 8Ω speaker is often big and difficult to give an independently sealed rear cavity for it. However, sealing microphone socket in chassis is always useful. Furthermore, the whole chassis must be sealed as far as possible. If there is any unavoidable leak hole, keep it far away from microphone. If the leak hole is close to microphone, the voice coming from the hole could be picked up by MIC, and then leads to echo at the far end. If the leak hole is close to speaker, the output voice quality could be aggravated at certain extent.

2) Direct air-path

This path can be obstructed through adjusting the holes direction and distance of receiver/SPK and MIC. For the purpose to transfer voice from receiver and SPK into MIC as little as possible, the receiver or SPK and MIC should be installed far away from each other. And their holes direction should not be opened in the same plane. Customer can refer to the structure of SIEMENS desk phone, whose hands-free SPK opening hole locates in left center and MIC opening hole locates in back right bottom corner. Nonetheless, the voice from an 8Ω speaker in hands-free mode is so big that it can always be picked up by the microphone. The echo arithmetic in software must be used to suppress this echo.

2.4. SPK choice

A SPK and receiver with higher sensitivity, flatter frequency response, less THD and impedance of 32 Ω (receiver), 16 Ω (receiver) or 8 Ω (speaker) is recommended. These technical data are often shown in data sheet of SPK and receiver. For SPK, its frequency response and THD performance can be tested by speaker test system. SPK frequency and THD response are shown in Figure 17 and Figure 18.

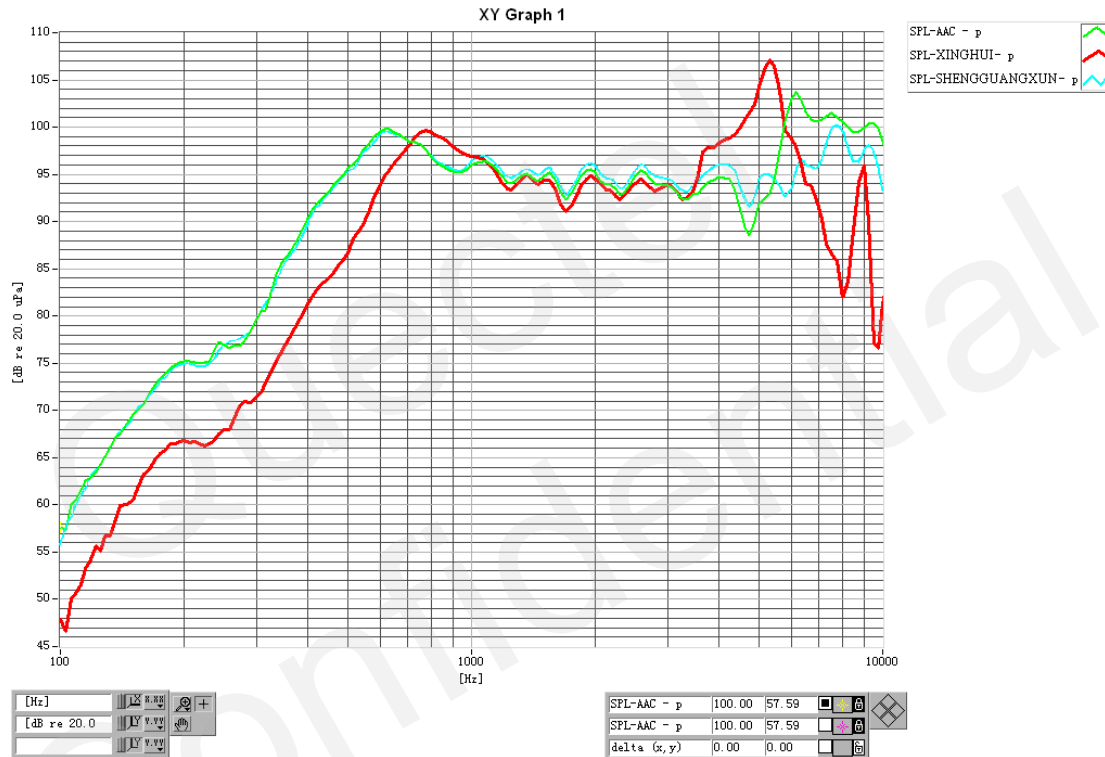


Figure 17: SPK frequency response

(Horizontal axis represents frequency, longitudinal axis represents loudness whose unit is dB)

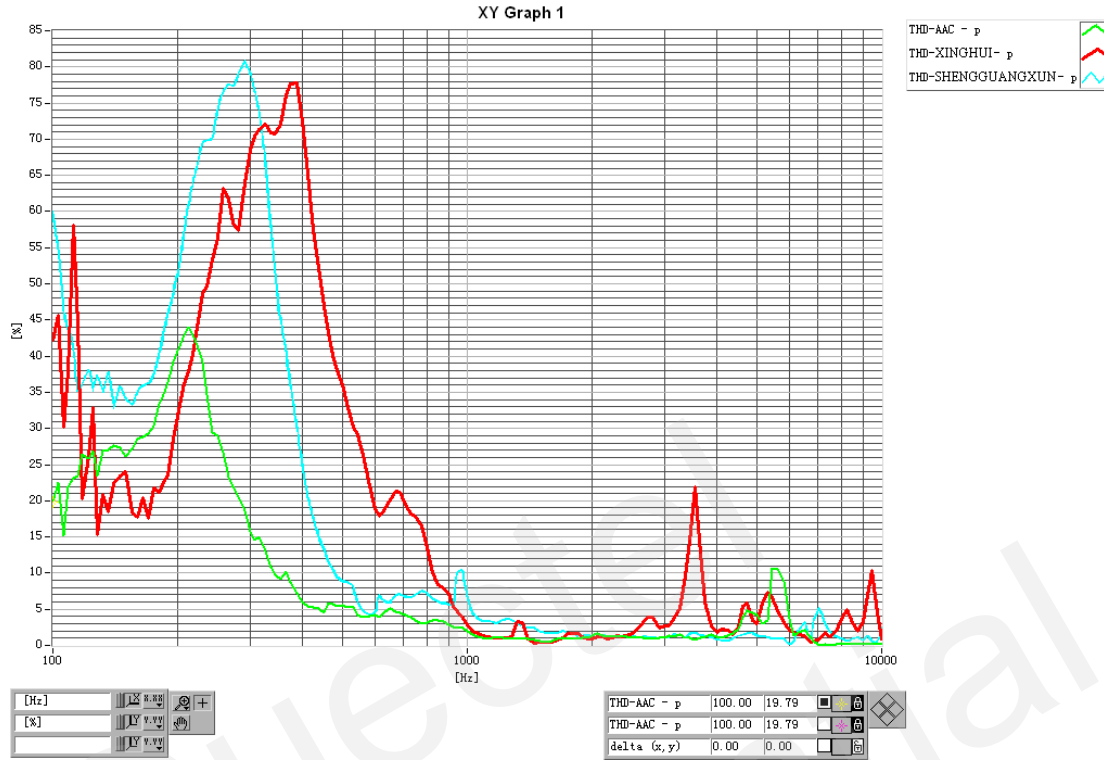


Figure 18: SPK THD

(Horizontal axis represents frequency, longitudinal axis represents distortion whose unit is %)

In Figure 17 and Figure 18, the three colors represent three vendors' products. After comprehensive comparison, the green one performs the best, and the blue one is second, and the red one is the worst.

2.5. MIC choice

It is recommended to use an electret microphone with a sensitivity of $-42\pm 3\text{dB/Pa}$ @ 2V (not less than $-44\pm 3\text{dB}$) and impedance of $2.2\text{k}\Omega$. If RF TDD noise is detected at the MIC, please contact MIC vendor for product with better RF suppression capability. Furthermore, the microphone channel circuit can be optimized to decrease TDD problem at MIC side.

2.6. Instruction of vehicle application

- 1) When the module is used in vehicle application and the hands-free talking function is required, audio channel 1 can be used. (Note that audio channel 1 can't provide ringing for in-coming calling and SMS calling)
- 2) Because the distance between the talker and the MIC is often farer, high sensitivity MIC such as -38dB is recommended.

3) Follow those previous instructions on suppressing echo problem.

Quectel module can run echo cancellation and echo suppression arithmetic simultaneously, so that echo can be largely weakened and talking quality is still excellent even in hands-free module. In addition, the uplink auto-gain controlling mechanism of module efficiently restrains voice clipping at the MIC channel, which is extremely useful for hands-free mode. The downlink noise reduction controlling mechanism is available to reduce environmental noise in the SPK output path, which enhances comfortable feeling to the listener. The next chapter is going to introduce how to set audio parameters in different applications. Vehicle application is regarded as an example with “bad” and severe echo problem.

3. Audio Parameters Introduction and Suggestion

Audio channels of Quectel modules can be divided into audio channel 1 and audio channel 2 in hardware, while it is defined three channels in software, including software channel 0, software channel 1 and software channel 2. Software channel 0 is only applied in hardware channel 1, while both software channel 1 and software channel 2 can be applied in hardware channel 2. When hardware audio channel 2 is used for headset application, software channel 1 must be active for this hardware channel. When hardware audio channel 2 is used for hands-free application, software channel 2 should be active for it and this configuration means two more gains added to the uplink and downlink of hardware audio channel 2. These two gains are constant in software. But software channel 1 has no such gains. Thus software channel 2 is designed for hands-free application of hardware channel 2.

Sketch map of audio channels inside BB chip of module is shown below.

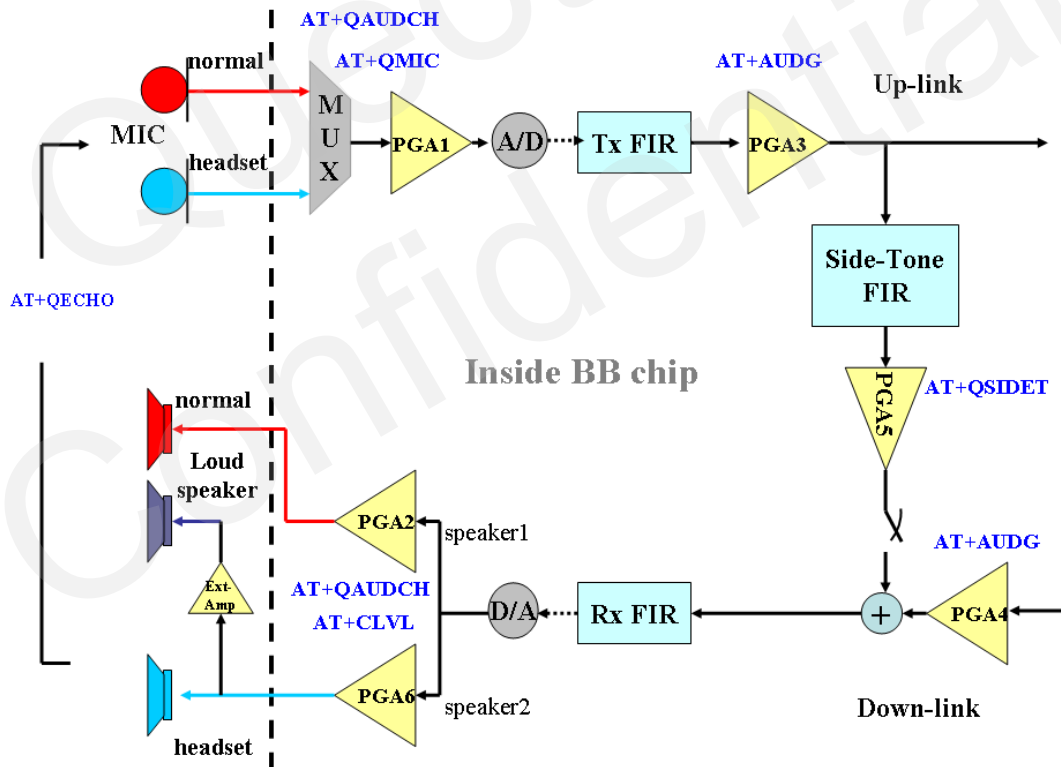


Figure 19: Audio channels inside BB chip

PGA1: Programmable analog gain amplifier on microphone channel

PGA2: Programmable analog gain amplifier on speaker channel in hardware audio channel 1

PGA3: Programmable digital gain amplifier on microphone channel

PGA4: Programmable digital gain amplifier on speaker channel

PGA5: Programmable digital gain amplifier on sidetone

PGA6: Programmable analog gain amplifier on speaker channel in hardware audio channel 2

3.1. Audio parameters introduction

This section will introduce common audio parameters, which can be modified by AT commands.

3.1.1. Change active software channel

AT+QAUDCH is used to set active software channel. Setting to 0 represents activating hardware audio channel 1, setting to 1 or 2 represents activating hardware audio channel 2. The formats of read and write commands are shown in Table 5.

Table 5: AT+QAUDCH introduction

Read Command AT+QAUDCH?	Response +QAUDCH: <active channel> OK
Write Command AT+QAUDCH=[<active channel>]	Response OK (success) ERROR (fail) Parameter <active channel> 0 – Software channel 0 (default) 1 – Software channel 1 2 – Software channel 2
Reference	AT+QAUDCH=1 AT+QAUDCH? Response: +QAUDCH:1 OK

3.1.2. Change audio gain level

3.1.2.1. Change microphone gain level

AT+QMIC is used to modify microphone uplink analog gain level. All the three software channels can be set by this command. The formats of read and write commands are shown in Table 6.

Table 6: AT+QMIC introduction

Read Command AT+QMIC?	Response + QMIC: <gainlevel(software channel 0)>, < gainlevel(software channel 1)>, < gainlevel(software channel 2)>
---------------------------------	--

	OK
Write Command AT+QMIC= <channel> , <gainlevel>	Response OK (success) ERROR (fail) Parameters < channel > 0 – Software channel 0 1 – Software channel 1 2 – Software channel 2 < gainlevel > int: 0 ~ 15
Reference	AT+QMIC=0,4 AT+QMIC=1,9 AT+QMIC=2,8 AT+QMIC? Response: +QMC:4,9,8 OK
Note	Customer should be flexible to set MIC gain level according to different sensitivity of microphones used. For high sensitivity MIC, gain level can be reduced; for low sensitivity MIC, gain level can be increased.

3.1.2.2. Change speaker gain level

AT+CLVL is used to modify speaker downlink analog gain level of active software channel. All the three software channels can be set by this command. Before setting the speaker gain level, **AT+QAUDCH** command must be used firstly to specify a software audio channel. The formats of read and write commands are shown in Table 7.

Table 7: AT+CLVL introduction

Read Command AT+CLVL?	Response +CLVL: <gainlevel(active software channel)> OK
Write Command AT+CLVL=<gainlevel(active software channel)>	Response OK (success) ERROR (fail) Parameter <gainlevel(active software channel)> int: 0~100 Setting to 0 represents mute volume
Reference	AT+QAUDCH=0 AT+CLVL=50 AT+CLVL? Response:

	+CLVL:50 OK
Note	Customer can set speaker gain level according to different loudness of speaker used.

3.1.2.3. Change sidetone gain level

AT+QSIDET is used to modify sidetone gain level of active software channel. The software channel 0 and software channel 1 can be set by this command. Sidetone gain must be closed in hands-free application, so the command is not supported in software channel 2. Note that when hardware audio channel 1 is applied to hands-free, please close the sidetone. Before setting the sidetone gain level, **AT+QAUDCH** command must be used to specify a software audio channel. The formats of read and write commands are shown in Table 8.

Table 8: AT+QSIDET introduction

Read Command AT+QSIDET?	Response +QSIDET(NORMAL_AUDIO): <gainlevel(software channel0)> OK +QSIDET(HEADSET_AUDIO): < gainlevel(software channel 1)> OK
Write Command AT+QSIDET=<gainlevel(active software channel)>	Response OK (success) ERROR (fail) Parameter <gainlevel(active software channel)> int: 0~255 Setting to 0 represents closing sidetone
Reference	AT+QAUDCH=0 AT+QSIDET=80 AT+QSIDET? Response: +QSIDET(NORMAL_AUDIO):80 OK
Note	The sidetone gain level should be set to a suitable value. Too high level leads to local environmental noise; too low level leads to uncomfortable talking feeling.

If customer needs to have a larger input voltage scale in MIC port for special applications such as DTMF tone transfer through MIC channel, some other undocumented AT commands related with acoustic configuration can be utilized to support them. Please contact Quectel for details of these commands.

3.1.3. Echo arithmetic command

AT+QECHO is used to modify echo arithmetic parameters of all three software audio channels. There are five parameters in this command. The formats of read and write commands are shown in Table 9.

Table 9: AT+QECHO introduction

Read Command AT+QECHO?	Response +QECHO:<control word>,<NLP>,<echo suppression>,<noise reduction>,<channel> OK
Write Command AT+QECHO= <control word>,<NLP>,<echo suppression>,<noise reduction>,<channel>	Response OK (success) ERROR (fail) Parameters <control word> 221 in handset or headset application 224 in hands-free application 0 represents disabling the whole echo arithmetic <NLP> int:0~2048 <i>The greater the value, the more reduction of echo</i> 0 means disabling the NLP algorithm <echo suppression> int:0~32767 <i>The smaller the value, the more reduction of echo</i> 0 means disabling the echo suppression algorithm <i>The default value of software channel 1 is 0, because headset application no need to use echo suppression</i> <noise reduction> 849 in handset or headset application 374 in hands-free application <channel> 0 – Software channel 0 1 – Software channel 1 2 – Software channel 2
Reference	AT+QECHO=221,1024,16388,849,0 AT+QECHO? Response: +QECHO:221,1024,16388,849,0 OK
Default parameters of three channels	AT+QECHO=221,1024,16388,849,0 AT+QECHO=221,1024,0,849,1 AT+QECHO=224,1024,5218,374,2

Quectel modules configure the best default parameters of this command to all three software

channels. Usually, customer need not modify any parameter. But in case of some applications such as audio channel 1 being applied in hands-free mode, or failing to completely cancel echo caused by bad circuits or structure, some of parameters can be modified to reduce more echo. Advices of audio parameters in different application are provided in the next section.

3.2. Audio parameters suggestion

Audio parameters are recommended in this section according to customer common applications, including hardware audio channel 1 in handset application and hands-free application, hardware audio channel 2 in headset application and hands-free application. The enumerative parameters of AT+QMIC /AT+CLVL /AT+QSIDET can be changed by customer according to peripheral components performance. The enumerative parameters of AT+QECHO are not recommended to change.

3.2.1. Hardware audio channel 1 in handset application

Whole structure	AT+QMIC=	AT+CLVL=	AT+QSIDET=	AT+QECHO=
Good echo	0,4	60	80	221,1024,16388,849,0
Bad echo	0,2	40	40	221,1024,10756,849,0

3.2.2. Hardware audio channel 1 in hands-free application

Whole structure	AT+QMIC=	AT+CLVL=	AT+QSIDET=	AT+QECHO=
Good echo	0,2	60	0	224,1024,5218,374,0
Bad echo	0,0	40	0	224,1024,2218,374,0

3.2.3. Hardware audio channel 2 in headset application

Whole structure	AT+QMIC=	AT+CLVL=	AT+QSIDET=	AT+QECHO=
Good echo	1,9	30	144	221,1024,0,849,1
Bad echo	1,8	10	80	221,1024,16388,849,1

3.2.4. Hardware audio channel 2 in hands-free application

Whole structure	AT+QMIC=	AT+CLVL=	AT+QSIDET=	AT+QECHO=
Good echo	2,8	35	0	224,1024,5218,374,2
Bad echo	2,7	20	0	224,1024,2218,374,2

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