SIMCOM WCDMA Wireless Module

SIM5360 Audio Application Note

Document Title:	SIM5360 Audio Application Note
Version:	1.01
Date:	2014-06-20
Status:	Release
Document Control ID:	AN SIM5360 Audio Application Note_V1.01

Contents

Figure Index

Version history

Date	Version	Description of change	Author
2014-06-20	1.01	Origin	Libing

1 Introduction

SIM5360 provides some AT commands for audio tuning. This document describes how to design and tune the audio part for best performance of SIMCOM WCDMA wireless module. (SIM5360 represents the series which is stated below.)

2 Scope of the document

This document is intended for the following versions of the SIMCom modules
•SIM5360

3. Audio Application

3.1 PCM interface

Table 1: PCM mode

<table>
<tr><th>PCM mode</th><th>SYNC</th><th>CLK</th><th>MODE</th><th>Format</th><th>Slot</th></tr>
<tr><td>Auxiliary</td><td>8KHz</td><td>128KHz</td><td>Master</td><td rowspan="2">A-law(8 bits)
υ-law(8 bits)
linear(16 bits)</td><td>Only slot 0</td></tr>
<tr><td>Primary</td><td>8KHz</td><td>2.048MHz</td><td>Slave/Master</td><td>0~15(Changed by AT command：at+cpcmslot)
Default：slot 0</td></tr>
</table>

The default PCM interface on power up is the auxiliary PCM interface. Under PCM, the data is output on the rising edge of PCM_CLK and sampled at the falling edge of PCM_CLK. Primary PCM is disabled at power up or when RESET is asserted, but you can use AT command to enable the primary PCM mode.

PCM Interface can be operated in Master and Slave mode. When the PCM interface is configured, PCM Tx data will be routed from the external codec Mic through the DSP encode path in the Module. PCM Rx data will be routed through the DSP decode path to the external codec speaker. When using the PCM Interface, the Module can be set either into Master Mode or Slave Mode.

In Master Mode, the Module drives the clock and sync signals that are sent out to the external codec via the PCM Interface. When in Slave Mode, the external codec drives the clock and sync signals that are sent to the Module.

Configuration Mode can be selected either primary or auxiliary. Primary configuration mode uses 2.048MHz clock and 8kHz short sync clock, and auxiliary configuration mode uses 128kHz clock and 8kHz long sync clock. One important consideration is that Slave mode is only available for use with Primary configuration Mode. PCM formats can also be chosen by AT command.

Note:
1. PCM interface can be control by AT command. For more details please refer to SIM5360_ATC_EN_V1.xx.doc
2. Timing of SIM5360 pcm please refer to SIM5360_Hardware_Design_V1.xx.doc

3.2 Block diagram of codec

The block diagram of the SIM5360 and external audio codec is described in the figure below. .

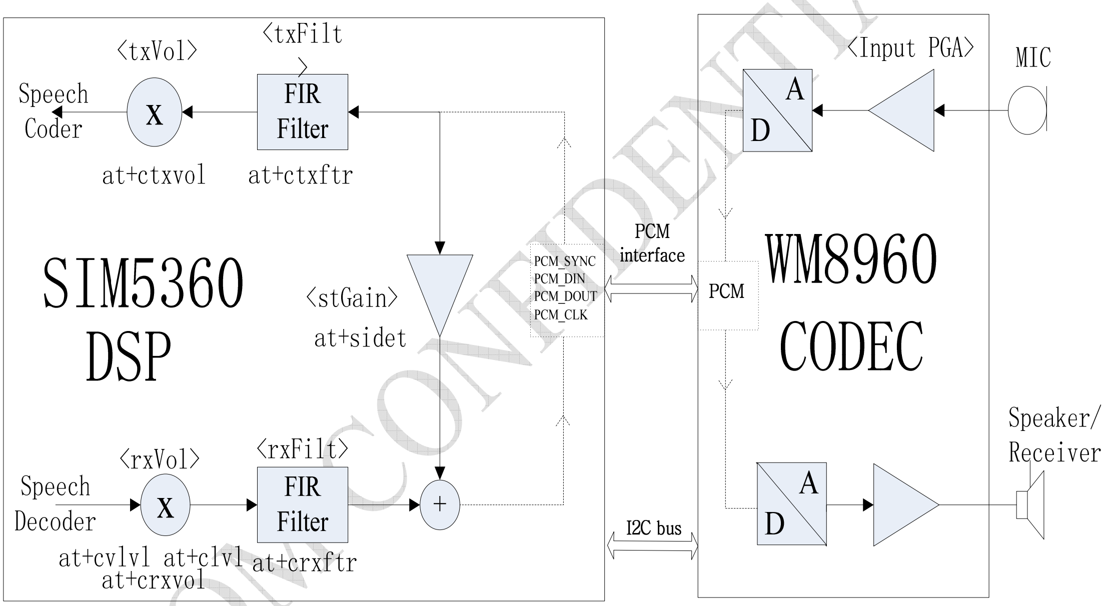


Figure 1: block diagram

3.3 External Audio Codec Application

The following figure is the reference design of SIM5360 PCM interface and external codec IC.

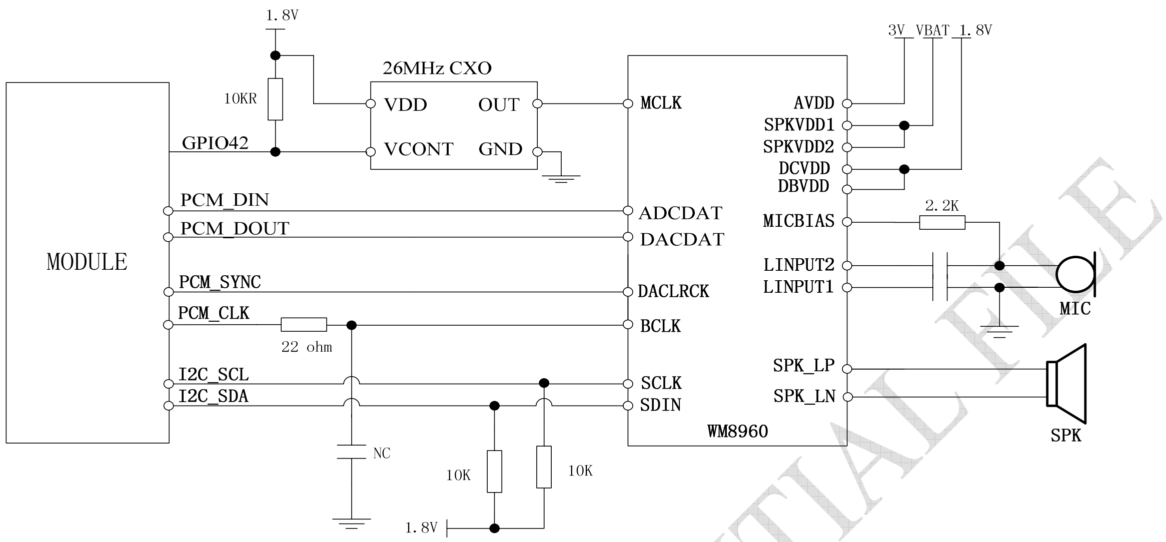


Figure 2: Reference Circuit of PCM Application with Audio Codec

3.4 Audio channel overview

The table below shows the audio channels of SIM5360 wireless module.

Table 2: Audio channels

Module	Audio Channel		Note
SIM5360	Handset: AT+CSDVC =1	Input: LINPUT2, LINPUT1	
		Output: HP_L	
	Headset: AT+CSDVC=2	Input: LINPUT2, LINPUT1	
		Output: HP_L	
	Handfree: AT+CSDVC=3	Input: LINPUT2, LINPUT1	
		Output: SPK_LP,SPK_LN	

4 Hardware Design

4.1 Speaker interface configuration

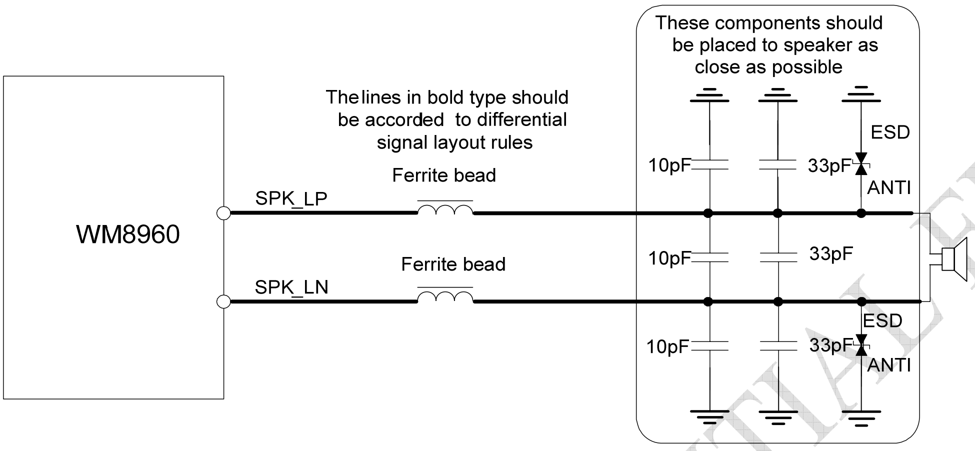


Figure 3: Speaker interface configuration

Because SPK_LP and SPK_LN are outputs of Class-D audio amplifier, optional EMI filtering is shown at Figure 1; these components (two ferrite beads and two capacitors) can be added to reduce electromagnetic interference. If used, they should be located near the SPK_LP and SPK_LN. Considerable current flows between the audio output pins and the speaker, so wide PCB traces are recommended (~ 20 mils). 8Ohm speaker is suggested. And the SPK_LP and SPK_LN should layout differential, and they should be far away from VBAT, RF signals, clock and other high power or high frequency signals.

4.2 Receiver interface configuration

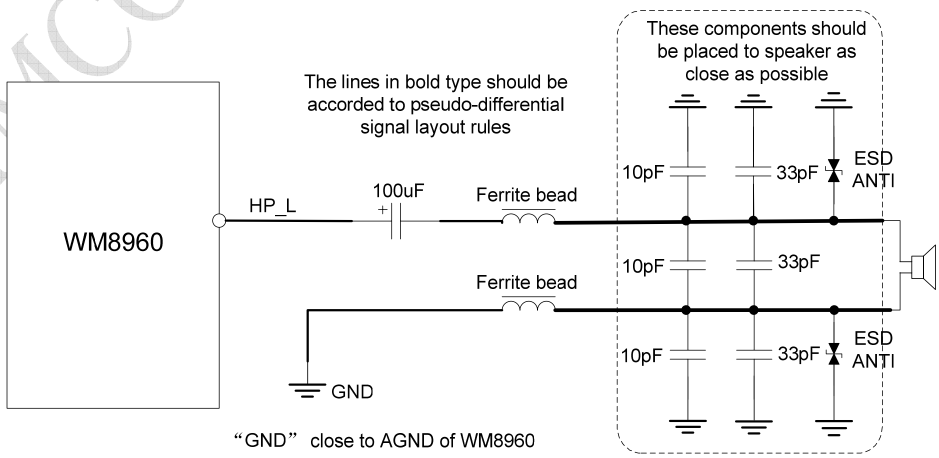


Figure 4: Receiver interface configuration

33p and 10p are suggested to be added beside the 32 Ohm receiver to reduce RF interfere. The width of HP_L line is typical 6 mils to reduce impedance. They should be far away from VBAT, RF signals, clock and other high power or high frequency signals. HP_L and it's return path lines should be layout pseudo-differential.

4.3 Microphone interfaces configuration

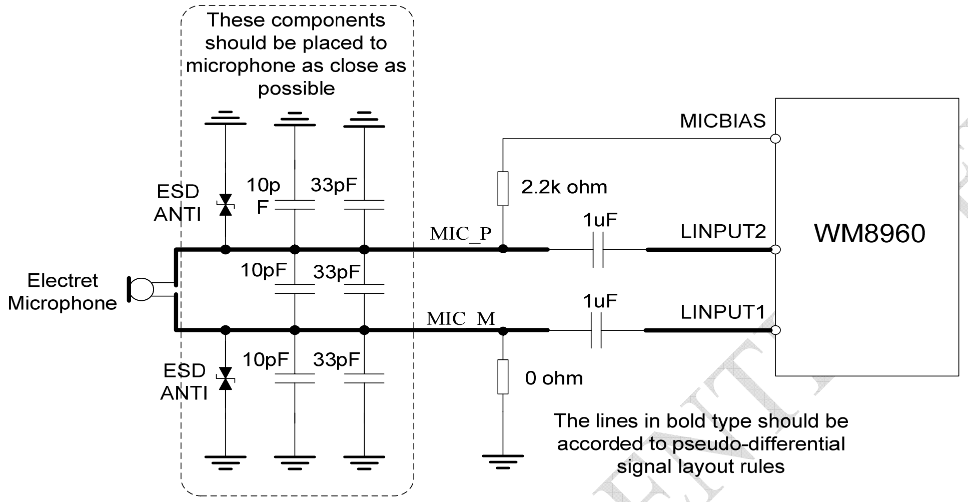


Figure 5: Microphone interface configuration

WM8960 CODEC has integrated internal MIC bais circuit. There is no need to pull the MIC_P and MIC_M up to the external power, because they have been pulled up in the Module. MIC_P and MIC_N should be layout pseudo-differential.

4.4 External MIC bias circuit configuration

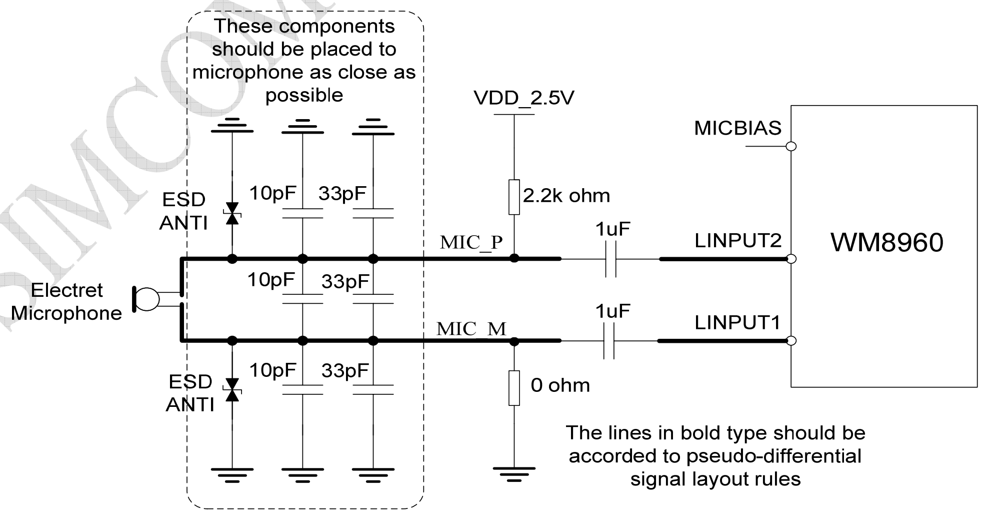


Figure 6: External MIC bias circuit configuration

External MIC bias circuit is described in the figure above. MIC_P and MIC_N should be layout pseudo-differential.

4.5 Referenced electronic characteristic

Table 3: MIC Input Characteristics

Parameter	Min	Typ	Max	Unit
Working Voltage		1.80		V
Working Current	70		400	uA
External Microphone Load Resistance	1.2	2.2		k Ohms

Table 4: Audio Output Characteristics

Parameter			Min	Typ	Max	Unit
Normal Output (HP_L)	Differential	load Resistance	27	32		Ohm
		Output power		70		mW
Auxiliary Output (HPR,HPL)	Single Ended	load Resistance	12	16		Ohm
	Differential	load Resistance	27	32		Ohm
	Single Ended	Output power		21.6		mW

Table 5: Speaker Output Characteristics

Parameter	Min	Typ	Max	Unit
Quiescent Current		6.2		mA
Output power(1KHz)		500		mW

5 Audio Tuning

The audio programming model shows how the signal path can be influenced by varying AT command parameters. Parameters <txVol>, <txFilter>, <stGain> ,<rxVol> and <rxFilter> can be adjusted with corresponding AT commands. For more information on the AT commands and parameters please refer to SIMCOM_SIM5360_ATC_EN_V1.xx.doc

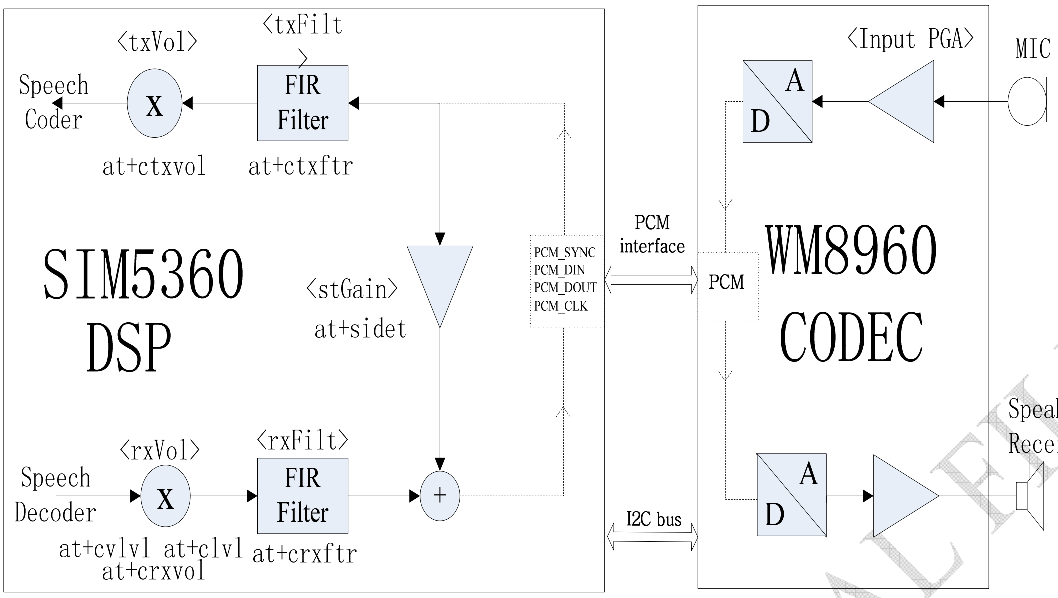


Figure 7: Audio programming model

Main audio parameters can be changed to satisfy users' requirement. Here primary register parameters and related description are listed. User can adjust them through AT command. For more detail please refers to Audio Application Document.

Table 6: Audio parameter

Parameter	Influence to	Range	Gain range	Calculation	AT command
txVol	Digital gain of input signal after ADC	0, 1...65535	Mute, -84...+12dB	20 * log (txVol/ 16384)	AT+CTXVOL
txFilter	Input PCM 13-tap filter parameters, 7 values	0...65535	---	MATLAB calculate	AT+CTXFTR
rxVol	Digital Volume of output signal after speech decoder, before summation of sidetone and DAC	-300…300	dbm	-300…300dbm	AT+CLVL AT+CVLVL AT+CRXVOL
stGain	Digital attenuation of sidetone	0, 1...65535	Mute, -96...0dB	20 * log (stGain/ 16384) -12	AT+SIDET
rxFilter	Output PCM 13-tap filter parameters, 7 values	0...65535	---	MATLAB calculate	AT+CRXFTR

5.1 MIC volume and frequency response

In figure7, one can turn adjust codec part or DSP part parameters to get desired MIC volume or frequency response.

DSP part

<txFilt>: AT+ctxftr (Detail description refer to table 7)

<txVol>: AT+ctxvol (Detail description refer to table 8)

Note: From figure7, user can see that AT+ctxftr, AT+ctxvol can influence sidetone.

5.1.1 AT+CTXFTR Set TX filter

Description

This command is used to set audio path parameter – TX filter, and refer to related hardware design document to get more information.

SIM PIN	References
NO	Vendor

Syntax

Test Command	Responses
AT+CTXFTR=?	+CTXFTR: (list of supported <tx_ftr_*N*>s) OK
Read Command	**Responses**
AT+CTXFTR?	+CTXFTR: <tx_ftr_*1*>,<…>,<tx_ftr_*7*> OK
Write Command	**Responses**
AT+CTXFTR= <tx_ftr_*1*>,<…>,<tx_ftr_*7*>	OK

Defined values

<tx_ftr_*N*>
TX filter level which is from 0 to 65535. (*N* is from 1 to 7)

Examples

AT+CTXFTR=1111,2222,3333,4444,5555,6666,7777
OK

5.1.2 AT+CTXVOL Set TX volume

Description

This command is used to set audio path parameter – TX volume, and refer to related hardware design document to get more information.

SIM PIN	References
NO	Vendor

Syntax

Test Command	Responses
AT+CTXVOL=?	+CTXVOL: (list of supported <tx_vol>s) OK
Read Command	**Responses**
AT+CTXVOL?	+CTXVOL: <tx_vol> OK
Write Command	**Responses**
AT+CTXVOL=<tx_vol>	OK

Defined values

<tx_vol>
TX volume level which is from 0 to 65535.

Examples

AT+CTXVOL=1234
OK

5.2 Receiver or Speaker volume and frequency response

In figure7, one can only turn adjust DSP part parameters to get desired receiver or speaker volume and frequency response. The parameter of codec part of module can not be adjusted.

DSP part

<rxFilt>: AT+crxftr (Detail description refer to table 9)

<rxVol>: AT+cvlvl, (Detail description refer to table 10)

AT+clvl, (Detail description refer to table 11)

AT+crxvol (Detail description refer to table 12)

AT+crxvol is used for fine tuning for <rxVol>. AT+CLVL and AT+CVLVL are used for coarse tuning for <rxVol>. Although we provide some AT for adjust the volume such as CRXVOL. These commands can change the voice levels together, that is to say, all the levels are promoted by these two parameters. But if you want to change each sound level value, you should use command CVLVL.

AT+CVLVL This command changes the sound level values of the command CLVL. Now we provide 8 levels for each audio channel. The level 0 is muted and it can not be changed by CVLVL. Levels 1 to 7 are supported to change the value of sound level. CVLVL command could let you change these four levels. The bigger the number presents the louder the voice. And the range of each level is -5000 to 5000.

5.2.1 AT+CRXFTR Set RX filter

Description

This command is used to set audio path parameter – RX filter, and refer to related hardware design document to get more information.

SIM PIN	References
NO	Vendor

Syntax

Test Command	Responses
AT+CRXFTR=?	+CRXFTR: (list of supported <rx_ftr_*N*>s) OK
Read Command	**Responses**
AT+CRXFTR?	+CRXFTR: <rx_ftr_*1*>,<…>,<rx_ftr_*7*> OK
Write Command	**Responses**
AT+CRXFTR= <rx_ftr_*1*>,<…>,<rx_ftr_*7*>	OK

Defined values

<rx_ftr_*N*>
RX filter level which is from 0 to 65535. (*N* is from 1 to 7)

Examples

AT+CRXFTR=1111,2222,3333,4444,5555,6666,7777
OK

5.2.2 AT+CVLVL Set value of sound level

Description

This command is used to set audio path parameter – RX volume. This command is different from CRXVOL (command CRXVOL will modify the values of all sound levels offset we provided together). You can change the value of each sound level based on your design separately through this command. Please refer to related hardware design document for more information.

SIM PIN	References

NO	Vendor

Syntax

Test Command	Responses
AT+CVLVL=?	+CVLVL: (list of supported <lvl>s),(list of supported <lvl_v-alue>s) OK
Read Command	Responses
AT+CVLVL?	+CVLVL: <lvl_value1>,<lvl_value2>,<lvl_value3>,<lvl_value4>,<lvl_value5>,<lvl_value6>,<lvl_value7>,<lvl_value8> OK
Write Command	Responses
AT+CVLVL=<lvl>,<lvl_value>	+CVLVL: lvl_value OK
	ERROR

Defined values

<lvl>	
	Sound level number which is from 1 to 8.
<lvl_value>	
	Sound level value which is from -5000 to 5000.
<lvl_value1>	
	Sound level value that sound level number equals 1.
<lvl_value2>	
	Sound level value that sound level number equals 2.
<lvl_value3>	
	Sound level value that sound level number equals 3.
<lvl_value4>	
	Sound level value that sound level number equals 4.
<lvl_value5>	
	Sound level value that sound level number equals 5.
<lvl_value6>	
	Sound level value that sound level number equals 6.
<lvl_value7>	
	Sound level value that sound level number equals 7.
<lvl_value8>	
	Sound level value that sound level number equals 8.

5.2.3 AT+CLVL Loudspeaker volume level

Description

Write command is used to select the volume of the internal loudspeaker audio output of the device.

Test command returns supported values as compound value.

SIM PIN	References
NO	3GPP TS 27.007

Syntax

Test Command	Responses
AT+CLVL=?	+CLVL: (list of supported <level>s) OK
Read Command	**Responses**
AT+CLVL?	+CLVL: <level> OK
Write Command	**Responses**
AT+CLVL=<level>	OK
	ERROR

Defined values

<level>
Integer type value which represents loudspeaker volume level. The range is from 0 to 7, and 0 represents the lowest loudspeaker volume level, 2 is default factory value. **NOTE:** <level> is nonvolatile, and it is stored when restart.

Examples

AT+CLVL?
+CLVL:2 *OK*
AT+CLVL=3
OK

5.2.4 AT+CRXVOL Set RX volume

Description

This command is used to set audio path parameter – RX volume, and refer to related hardware design document to get more information.

SIM PIN	References
NO	Vendor

Syntax

Test Command	Responses
AT+CRXVOL=?	+CRXVOL: (list of supported <rx_vol>s) OK
Read Command	**Responses**
AT+CRXVOL?	+CRXVOL: <rx_vol> OK
Write Command	**Responses**
AT+CRXVOL=<rx_vol>	OK

Defined values

<rx_vol>
RX volume level which is from -100 to 100.

Examples

AT+CRXVOL=12
OK

5.3 AT+SIDET Digital attenuation of sidetone

Description

The command is used to set digital attenuation of sidetone. For more detailed information, please refer to relevant HD document.

SIM PIN	References
NO	Vendor

Syntax

Test Command	Responses
AT+SIDET=?	+SIDET: (list of supported <st>s) OK
Read Command	**Responses**
AT+SIDET?	+SIDET:<st> OK
Write Command	**Responses**
AT+SIDET= <st>	OK
	ERROR

Defined values

<st>

Digital attenuation of sidetone, integer type in decimal format and nonvolatile.

Range: from 0 to 65535.

Factory value: HANDSET:2034, HEADSET:1024, SPEAKER PHONE: 0.

Examples

AT+CSDVC=1
OK
AT+SIDET?
+SIDET: 2304 *OK*

5.4 Echo canceller

SIM5360 has AT command*"AT+ CECM"* to adjust echo canceller.

This AT command is used to select the echo cancellation mode. Each audio channel has it's own default echo cancellation mode.

at+cecm=0 : disable EC mode

at+cecm=1 : EC mode recommended for Handset

at+cecm=2 : EC mode recommended for Headset

at+cecm=3 : EC mode recommended for HANDSFREE

at+cecm=4 : EC mode recommended for SPEAKER

at+cecm=5 : EC mode recommended for BT HEADSET

at+cecm=6， EC mode recommended for Speaker phone aggressive echo arithmetic

at+cecm=7， EC mode recommended for Speaker phone medium echo arithmetic

at+cecm=8， EC mode recommended for Speaker least aggressive echo arithmetic

5.5 TDD noise

Making sure the module connect to ground well can help to reduce the TDD noise and improve ESD.

Filtering capacitors and beads are suggested to be added in the audio lines, 33p and 10p can help reduce the 850Mhz/900Mhz and 1800Mhz/1900Mhz RF interfere. If it is signal, the filtering capacitors and beads are suggested to add beside the module pins. If it is output trace, the filtering capacitors and beads are suggested to add beside the handset/ headset/speaker connector.

5.6 Sending and receiving distortion

There are many factors which may influence the sending and receiving distortion.

1. Unsuitable FIR parameters. They can be adjust by AT+CTXFTR, AT+CRXFTR.
2. Too large TDD noise.
3. Unsuitable AGC parameters.

5.7 DTMF distortion

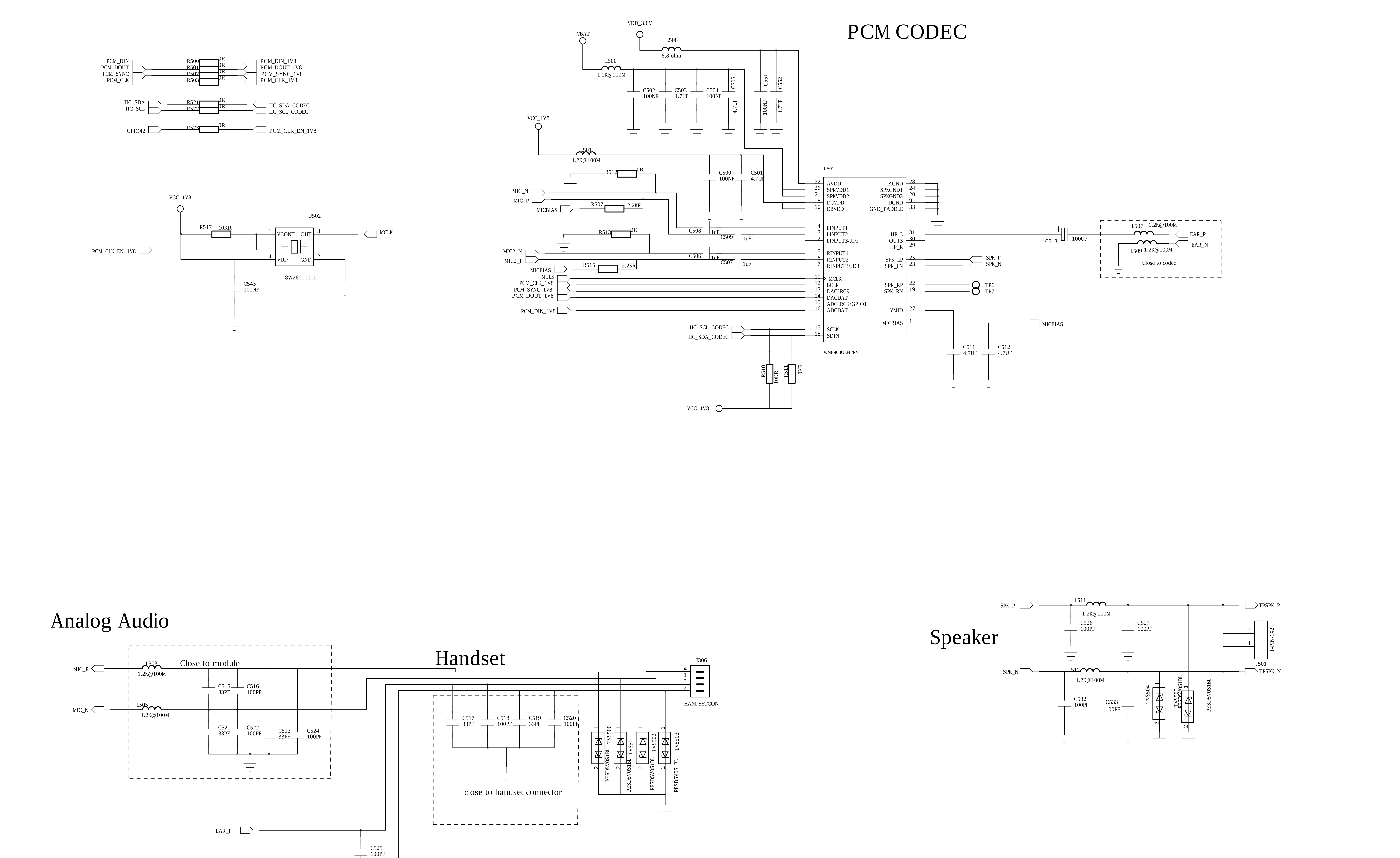
Too large sending and receiving gain may result in DTMF distortion.
If one find DTMF sending distortion which can be measured by the oscillograph in receiving end, one can decrease the codec gain and try again. Other sending parameters can also be adjusted to get better DTMF performance.
If one find DTMF receiving distortion, on can adjust receiving parameters to get better DTMF performance.

6 Layout guide

The audio signals are sensitive to RF signals and power sources (for example Vbat). Please make sure that the audio signals are far away from the RF signals and Vbat. And the output signals and input signals should be kept away from each other by ground. The differential lines should be layout together. And HPL and HPR are not differential signals, so they should be layout separately.
Filtering capacitors and beads are suggested to be added in the audio lines, 33p and 10p can help reduce the 850Mhz/900Mhz and 1800Mhz/1900Mhz RF interfere. If it is signal, the filtering capacitors and beads are suggested to add beside the module pins. If it is output trace, the filtering capacitors and beads are suggested to add beside the handset/ headset/speaker connector.
One can send design to us for checking.

7 Appendix

I. SIM5360 external audio code schematic



SIM Technology		
DRAWN BY <NAME HERE>	PORJECT SIM5360-EVB	TITLE P5_AUDIO
CHECKED BY <NAME HERE>	SIZE A2	VER V1.01
	SHEET 5 of 5	2014-4-30

Contact us:
Shanghai SIMCom Wireless Solutions Ltd.
Add: Building A，SIM Technology Building，No.633，Jinzhong Road，Changning District, Shanghai,P. R. China 200335
Tel: +86 21 3235 3300
Fax: +86 21 3235 3301
URL: www.sim.com/wm