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Version history

Date	Version	Description of change	Author
2016-09-08	1.00	Origin	Ma Honggang

1 Introduction

The SIM7X00 stands for SIM7600, SIM7500 and SIM7100, this document will take SIM7600 as the example.

SIM7600 provides some AT commands for audio tuning. This document describes how to design and tune the audio part for best performance of SIM7600 module.

2 Scope of the document

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

•SIM7600

•SIM7500

•SIM7100

3. Audio application

SIM7100 supports WM8960 and NAU8810 codec, but SIM7600 and SIM7500 only support NAU8810 codec.
When customer select SIM7100 module, customer should use AT+CEODECSEL command to set the codec, AT+CODECSEL=1 set the codec to WM8960, and AT+CODECSEL=2 set the codec to NAU8810 codec.
If customer selects SIM7600 and SIM7500, there's no need to set the codec kind.

3.1 PCM interface

SIM7600C provides a PCM interface for external codec, which can be used in master mode with short sync and 16 bits linear format.

Table 1: PCM specification

Characteristics	Specification
Line Interface Format	Linear(Fixed)
Data length	16bits(Fixed)
PCM Clock/Sync Source	Master Mode(Fixed)
PCM Clock Rate	2048 KHz (Fixed)
PCM Sync Format	Short sync(Fixed)
Data Ordering	MSB

PCM Interface can be operated in Master mode only. 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.
In Master Mode, the Module drives the clock and sync signals that are sent out to the external codec via the PCM Interface.

3.2 Block diagram of audio circuit

The block diagram of the SIM7600 and external audio codec is described in the figure below.

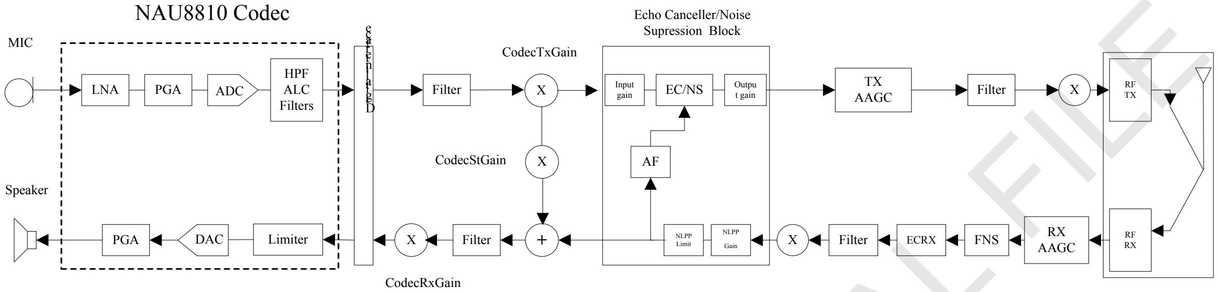


Figure 1: Block diagram

3.3 External audio codec application

The following figure is the reference design of SIM7600 PCM interface and external codec IC.

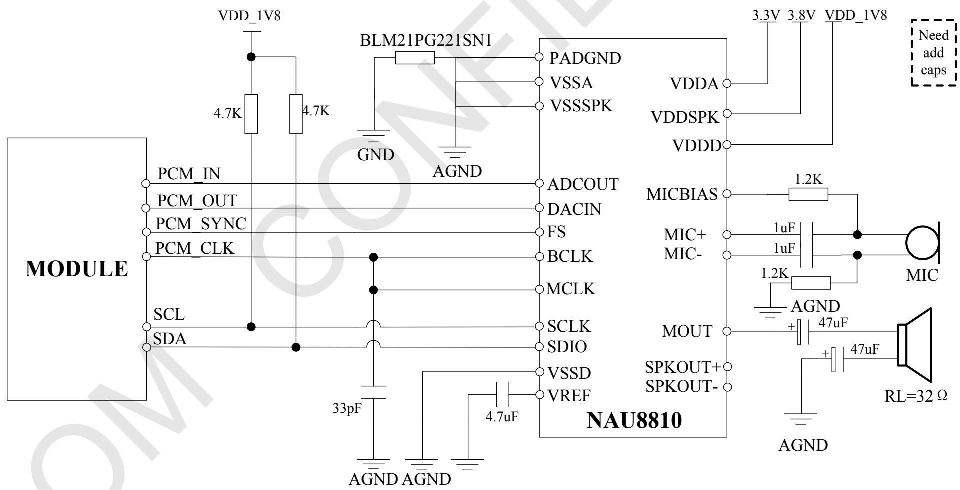


Figure 2: Reference circuit of PCM application with audio codec

3.4 Audio channel overview

The table below shows the audio channels of SIM7600 wireless module.

Table 2: Audio channels

Module	Audio Channel		Note
SIM7600	Handset: AT+CSDVC =1	Input: MIC+, MIC-	
		Output: MOUT	
	Handfree: AT+CSDVC=3	Input: MIC+, MIC-	
		Output: SPKOUT+, SPKOUT-	

Note: NAU8810 codec does not support AT+CSDVC=2.

4 Hardware design

4.1 Speaker interface configuration

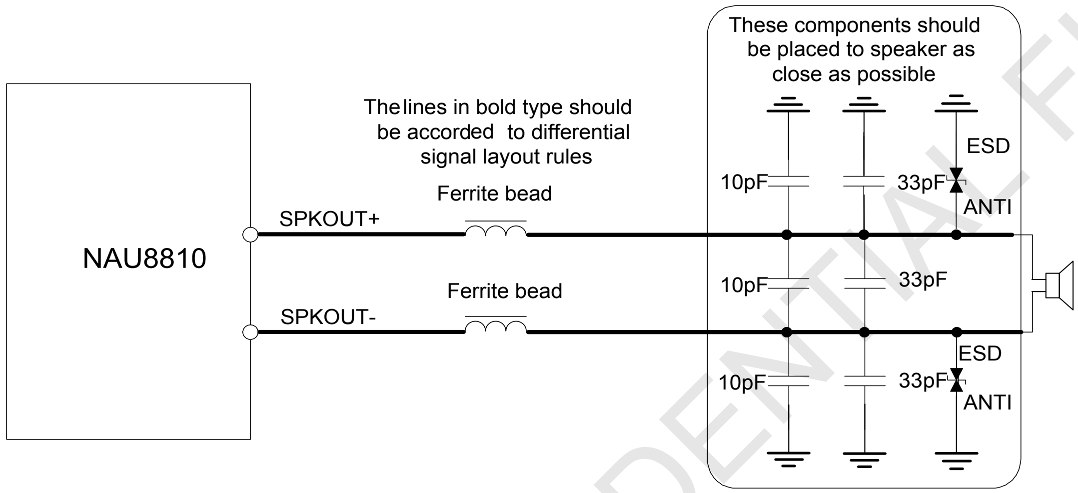


Figure 3: Speaker interface configuration

Because SPKOUT+ and SPKOUT- are outputs of audio amplifier, optional EMI filtering is shown at Figure 3; these components (two ferrite beads and two capacitors) can be added to reduce electromagnetic interference. If used, they should be located near the SPKOUT+ and SPKOUT-. 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 SPKOUT+ and SPKOUT- 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

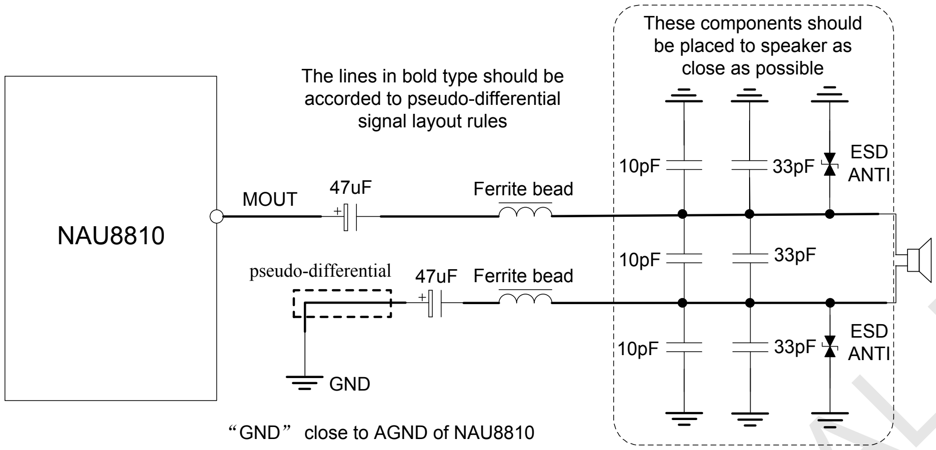


Figure 4: Receiver interface configuration

33p and 10p are suggested to be added beside the 32 Ohm receiver to reduce RF interfere. The width of MOUT 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. MOUT and it's return path lines should be layout pseudo-differential.

4.3 Microphone interfaces configuration

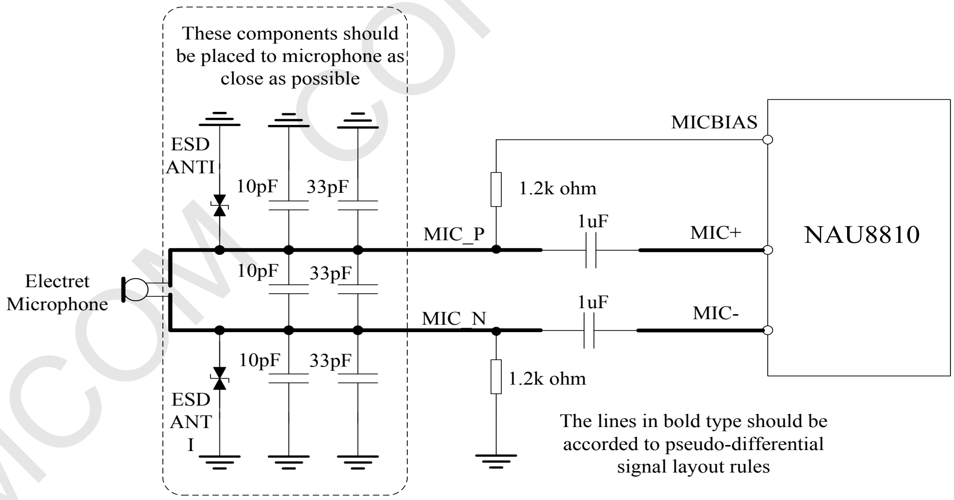


Figure 5: Microphone interface configuration

NAU8810 codec has integrated internal MIC bias circuit. MIC_P and MIC_N should be pulled up to the external power. MIC_P and MIC_N should be layout differential.

4.4 Referenced electronic characteristic

Table 3: MIC input characteristics

Parameter	Min	Typ	Max	Unit
Working Voltage		0.9*VDDA		V
Working Current			3	mA
External Microphone Load Resistance	1.2	2.2		k Ohm

Table 4: Audio output characteristics

Parameter					Min	Typ	Max	Unit
Normal Output (MOUT) VDDSPK=3.3 V	THD	-84dB	Po=20mW	RL=16Ω		20		mW
		-85dB	Po=20mW	RL=32Ω		20		mW

Table 5: Speaker output characteristics

Speaker Output (SPKOUT+, SPKOUT- with 8Ω bridge tied load)								
PARAMETER	SYMBOL	TEST CONDITIONS			MIN	TYP	MAX	UNIT
Signal to Noise Ratio	SNR	VDDSPK = 3.3V,RL = 8Ω				90		dB
		VDDSPK = 1.5*VDDA,RL = 8Ω				90		dB
Total Harmonic Distortion	THD	PO =180mW	RL = 8Ω	VDDSPK=3.3V		-63		dB
		PO =400mW				-56		dB
		PO =360mW		VDDSPK = 1.5*VDDA		-60		dB
		PO =800mW				-61		dB
		PO =800mW				-34		dB
Power Supply Rejection Ratio(50Hz - 22kHz)	PSRR	VDDSPK = 3V				50		dB
		VDDSPK = 1.5*VDDA				50		dB

5 Audio tuning

The audio programming model shows how the signal path can be influenced by varying AT command parameters. Parameters can be adjusted with corresponding AT commands. For more information on the AT commands and parameters please refer to SIMCOM_SIM7600_ATC_EN_V1.xx.doc
All commands mentioned in figure 6 only work in the phone call status, when customer hung up the phone, the parameter would set back to the default value.

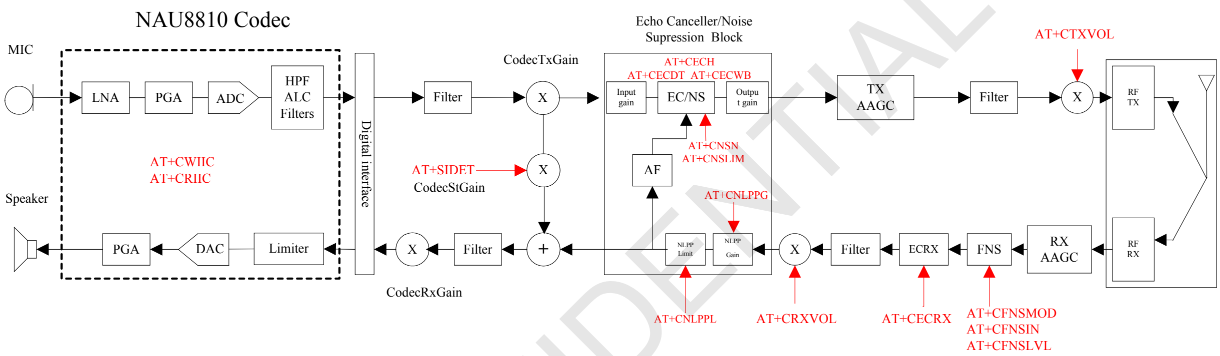


Figure 6: Audio programming model

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.

5.1 Volume and sidetone

In figure6, customer can turn adjust codec part or DSP part parameters to get desired MIC volume, SPK volume and sidetone.

DSP part

<TXVOL>: AT+CTXVOL (Detail description refer to table 6)
<RXVOL>: AT+CRXVOL (Detail description refer to table 6)
<SIDET>: AT+SIDET (Detail description refer to table 6)

Table 7: Audio parameter for volume and sidetone

Parameter	Influence to	Range	Gain range	Calculation	AT command
CTXVOL*	Digital gain of input signal after ADC	0, 1...0xFFFF	Mute, -78...+18dB	0x333= -20db 0xA1F=-10db	AT+CTXVOL

CRXVOL	Digital Volume of output signal after speech decoder, before summation of sidetone and DAC	0, 1... 0xFFFF	Mute, -78...+18dB	0x2000= 0db 0x38E8= 5db 0x6531= 10db 0xB3F3= 15db 0xFE2F= 18db	AT+CRXVOL
SIDET*	Digital attenuation of sidetone	0,1	-	0-close sidetone 1-open sidetone	AT+SIDET

Note:SIM7100 should use AT+CTXVOLEX to set the digital gain of input signal; the "sidet" command only apply to SIM7600 and SIM7500;

5.1.1 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 0xFFFF.

Examples

AT+CTXVOL=0x1234
OK

5.1.2 AT+CRXVOL set RX volume

In figure6, customer can turn adjust DSP part parameters to get desired receiver or speaker volume.

DSP part

<RXVOL>: AT+CRXVOL (Detail description refer to table)

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 0 to 0xFFFF.

Examples

AT+CRXVOL=0x1234

OK

5.1.3 AT+SIDET set 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>
Sidetone value is 0 or 1.

Examples

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

5. 2 Echo canceller

5.2.1 AT+CECM set echo mode

SIM7600 has AT command *"AT+ CECM"* to adjust echo canceller.
This AT command is used to select the echo cancellation mode.
at+cecm=0 : disable EC mode
at+cecm=1 : EC mode recommended for Speaker phone aggressive
at+cecm=2 : EC mode recommended for Speaker phone medium
at+cecm=3 : EC mode recommended for Speaker least aggressive
at+cecm=4 : EC mode recommended for Bluetooth
at+cecm=5 : EC mode recommended for Bluetooth (less aggressive)
at+cecm=6 : EC mode recommended for Bluetooth (least aggressive)
at+cecm=7 : EC mode recommended for HANDSFREE
at+cecm=8 : EC mode recommended for Headset
at+cecm=9 : EC mode recommended for Handset

5.2.2 Set echo parameters of RX path

Usually, AT+CECM command would solve most of echo issues, but when the issue can't be solved, customer could use the following AT commands to tuning the echo parameters to get better voice quality.

Table 8: Audio parameter for echo RX path

Parameter	Influence to	Range	Gain range	Calculation	AT command
CECRX	Echo canceler Rx module	0,1	-	0---Enable feature 1---disable feature	**AT+CECRX**
CNLPPG*	Rx In gain factor for NLPP	0...0x7FFF	-	-	**AT+CNLPPG**
CNLPPL*	Clipping limit value for NLPP	0...0x7FFF	-	-	**AT+CNLPPL**

NLPP means Non-linear pre-processing; NLPPG and NLPPL would work after the CECRX was enabled.

5.2.2.1 AT+CECRX

Description

The command is used to enable or disable the echo function of RX path.

SIM PIN	References
NO	Vendor

Syntax

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

Defined values

<st>
Valid value 0 or 1;

Examples

AT+ CECRX*?*
+ CECRX*: 1* *OK*

5.2.2.2 AT+CNLPPG

Description

The command is used to set the NLPP gain.

SIM PIN	References
NO	Vendor

Syntax

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

Defined values

<st>
Valid value 0….0x7FFF;

Examples

AT+ CNLPPG?
+ CNLPPG*: 0x0800* *OK*

5.2.2.3 AT+CNLPPL

Description

The command is used to set the NLPP limit.

SIM PIN	References
NO	Vendor

Syntax

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

Defined values

<st>
Valid value 0….0x7FFF;

Examples

AT+ CNLPPG*?*
+ CNLPPL*: 0x7FFF* *OK*

5.2.3 Set echo parameters of TX path

Table 9: Audio parameter for echo TX path

Parameter	Influence to	Range	Gain range	Tuning direction	AT command
CECH	Additional muting gain applied in DES during far-end only.	0...0x7FFF	-	Higher value is more muting.	**AT+CECH**
CECDT	Additional muting gain applied in DES during doubletalk.	0...0x7FFF	-	Higher value is more muting.	**AT+ CECDT**
CECWB	This parameter adjusts the aggressiveness of EC in the high band (4 ~ 8 kHz). A higher value is more aggressive and suppresses more high-band echo.	0...0x7FFF	-	Higher value is more muting.	**AT+ CECWB**

5.2.3.1 AT+CECH

Description

The command is used to set mute gain of far-end echo. For more detailed information, please refer to relevant HD document.

SIM PIN	References
NO	Vendor

Syntax

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

Defined values

<st>
mute gain of far-end echo, integer type in decimal format and nonvolatile. Range: from 0 to 0x7FFF.

Examples

AT+CECH?
+CECH: 0x0200 *OK*

5.2.3.2 AT+CECDT

Description

The command is used to set mute gain of echo during doubletalk. For more detailed information, please refer to relevant HD document.

SIM PIN	References
NO	Vendor

Syntax

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

	ERROR

Defined values

<st>
mute gain of echo during doubletalk, integer type in decimal format and nonvolatile. Range: from 0 to 0x7FFF.

Examples

AT+CECDT?
+CECDT: 0x0100 *OK*

5.2.3.3 AT+CECWB

Description

The command is used to set mute gain of echo during doubletalk. For more detailed information, please refer to relevant HD document.

SIM PIN	References
NO	Vendor

Syntax

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

Defined values

<st>
mute gain of echo during doubletalk, integer type in decimal format and nonvolatile. Range: from 0 to 0x7FFF.

Examples

AT+CECWB?
+CECWB: 0x0300 *OK*

5.3 Noise suppression

SIM7600 supports noise suppression function, includes far-end noise and near-end noise.

5.3.1 Near-end noise suppression

Customer could use AT+CNSN and AT+CNSLIM to tune the voice noise of TX path, so the listener would have a better voice call.

Table 10: Audio parameter for Near-end noise suppression

Parameter	Influence to	Range	Gain range	Calculation	AT command
CNSN	Oversubtraction factor and bias compensation for noise estimation	0...0x7FFF	-	0x286D= -10db 0x1CA8= -13db 0x16C3= -15db	**AT+CNSN**
CNSLIM	Controls the maximum amount of noise suppression.	0...0x7FFF	-	-	**AT+CNSLIM**

5.3.1.1 AT+CNSN

Description

The command is used to set Over subtraction factor and bias compensation for noise estimation.

SIM PIN	References
NO	Vendor

Syntax

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

Defined values

<st>
Range: from 0 to 0x7FFF. 0x286D= -10db 0x1CA8= -13db 0x16C3= -15db

Examples

AT+ CNSN?
+ CNSN: 0x0258 *OK*

5.3.1.2 AT+CNSLIM

Description

The command is used to set maximum amount of noise suppression.

SIM PIN	References
NO	Vendor

Syntax

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

Defined values

<st>
Range: from 0 to 0x7FFF.

Examples

```
AT+ CNSN?
+ CNSN: 0x 16C4

OK
```

5.3.2 Far-end noise suppression

An example use-case is illustrated in following figure.The far-end is making a call from a public phone in a noisy environment. This public phone does not have any NS capability implemented on its Tx path. The near-end mobile phone user answers the call but has difficulty understanding the far-end due to noise contamination with the speech signal.

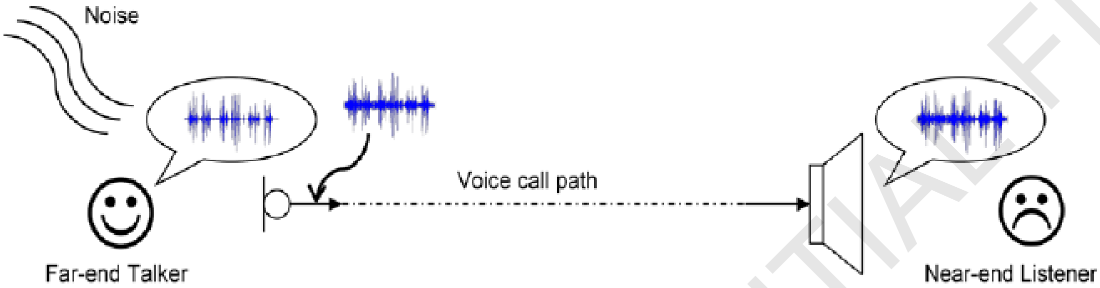


Figure 7: Conversation without FENS

However, with the FENS feature, the near-end mobile device is able to perform nonstationary NS on the far-end signal to improve on the SNR and voice clarity so the conversation can be maintained in an efficient manner, as illustrated in Figure 2-2.

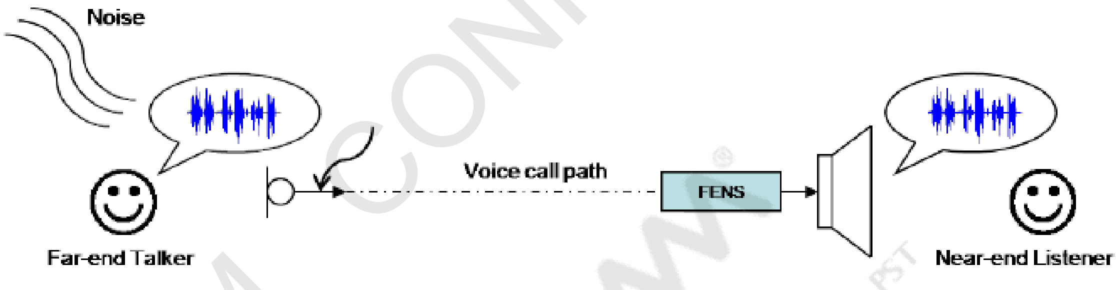


Figure 8: Conversation with FENS

Table 11: Audio parameter for Far-end noise suppression

Parameter	Influence to	Range	Gain range	Calculation	AT command
CFNSMOD	Mode for enabling/disabling submodules	0x00FF 0x0073 0x00F3 0x01FF	-	0x00FF – Maximum NS 0x0073 – Basic stationary NS 0x00F3 – Enhanced stationary NS 0x01FF – Aggressive NS	**AT+CFNSMOD**

CFNSIN	Input gain to FENS module	0x2000...0x7FFF	-	-	**AT+CFNSIN**
CFNSLVL	Target noise suppression level in dB	0...0x7FFF	-	-	**AT+CFNSLVL**

5.3.2.1 AT+CFNSMOD

Description

The command is used to set the mode of Far-end noise suppression.

SIM PIN	References
NO	Vendor

Syntax

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

Defined values

<st>

0x00FF – Maximum NS

0x0073 – Basic stationary NS

0x00F3 – Enhanced stationary NS

0x01FF – Aggressive NS

Examples

AT+ CFNSMOD*?*

+ CFNSMOD*:* 0x0073

OK

5.3.2.2AT+CFNSIN

Description

The command is used to set the Input gain to FENS module.

SIM PIN	References
NO	Vendor

Syntax

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

Defined values

<st>
Range: 0...0x7FFF

Examples

AT+ CFNSIN?
+ CFNSIN*:* 0x1234 *OK*

5.3.2.3AT+CFNSLVL

Description

The command is used to set the target noise suppression of noise suppression.

SIM PIN	References
NO	Vendor

Syntax

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

Defined values

<st>
Range: 0...0x7FFF

Examples

AT+ CFNSLVL?
+ CFNSIN*:* 0x1234 *OK*

5. 4 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 850 Mhz/900Mhz and 1800 Mhz/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.

6 Codec operation (Only applied to SIM7600 and SIM7500)

SIM7600 only support NAU8810 codec by default, customer could set the parameters of codec directly by I2C interface via AT commands.
The module provides AT+CWIIC and AT+CRIIC to write and read the register value of codec, this chapter would introduce the methods of tuning MIC gain, speaker gain, and MOUT gain of codec. These commands only work when the module is making a call.

6. 1 Tuning the MIC gain of codec

Customer could change the PGAGAIN[5:0] 0x2D register value to change the MIC gain of codec, as the following figure shows.

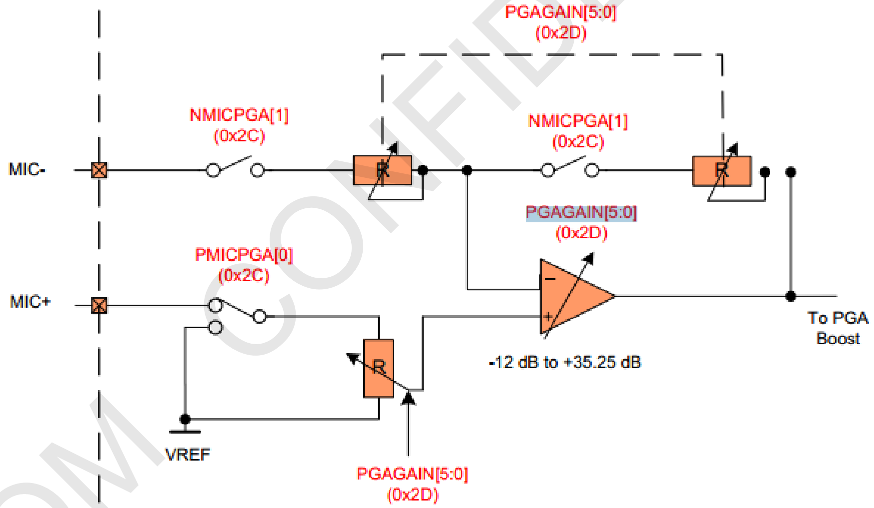


Figure 9: MIC gain register of codec

The default register value of 0x2D is 0x010, PGAGAIN[5：0] is 10000, it is equal to 0db.

Addr	Bit 8	Bit 7	Bit 6	Bit5	Bit 4	Bit 3	Bit 2	Bit 1	Bit 0	Default
0x2D	0	PGAZC	PGAMT	PGAGAIN[5:0]						0x010

Figure 10: Bit config of 0x2D

Programmable Gain Amplifier Gain						
PGAGAIN[5:0]						
B5	**B4**	**B3**	**B2**	**B1**	**B0**	**Gain**
0	0	0	0	0	0	-12.00 dB
0	0	0	0	0	1	-11.25 dB
0	0	0	0	1	0	-10.50 dB
:::	:::	:::	:::	:::	:::	:::
0	0	1	1	1	1	-0.75 dB
0	1	0	0	0	0	0 dB
0	1	0	0	0	1	+0.75 dB
PGA Gain Range -12dB to +35.25dB @ 0.75 increment						
:::	:::	:::	:::	:::	:::	:::
1	1	1	1	0	1	33.75
1	1	1	1	1	0	34.50
1	1	1	1	1	1	35.25

Figure 11: programmable amplifer gain

Customer could tuning the PGAGAIN[5:0] from 0x0 to 0x3F.
AT+CWIIC=0x34,0x5A,0x[0~3F],1

6. 2 Tuning the speaker and MOUT gain of codec

NAU8810 provides a SPKOUT audio channel, customer could tune the SPKGAIN [5:0] and SPKBST [2] to change the speaker volume. The register address is 0x36 and 0x31.

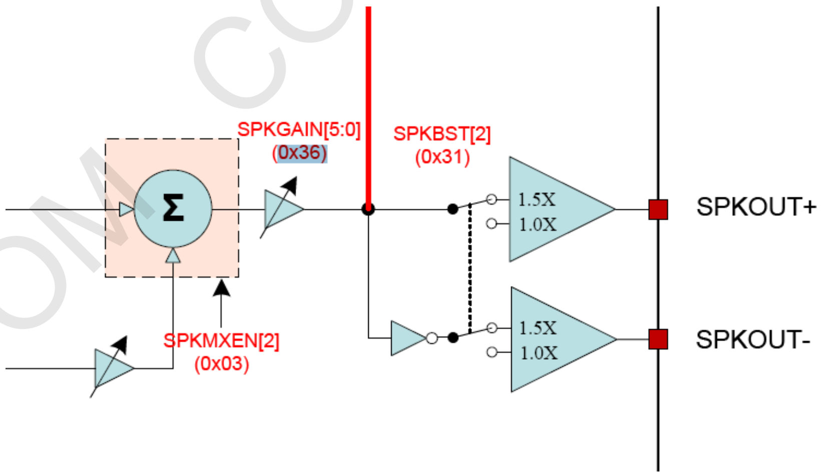


Figure 12: Speaker gain register of codec

Addr	D8	D7	D6	D5	D4	D3	D2	D1	D0	Default
0x36	0	SPKZC	SPKMT	SPKGAIN[5:0]						**0x039**

SPKBST[2]	0x31	Speaker output Boost	0 – (1.0x VREF) Boost 1- (1.5 x VREF) Boost
SPKGAIN[5:0]	0x36	Speaker output Volume	Range: -57dB to +6dB @ 6dB increment
SPKMT[6]	0x36	Speakpr output Mute	0 – Speaker Enabled 1 – Speaker Muted

Figure 13: 0x36 register configuration

The default value of 0x36 is 0x039, it is equal to 000111001, from the figure 14, customer would found that the default speaker gain is 0db, the value could be tuned from 000000000 to 000111111, so the corresponding AT commands is AT+CWIIC=0x34,0x6C,0x[0~3F],1

Speaker Gain						
SPKGAIN[5:0]						
B5	B4	B3	B2	B1	B0	Gain (dB)
0	0	0	0	0	0	-57.0
0	0	0	0	0	1	-56.0
0	0	0	0	1	0	-55.0
...	...	...	...	...	...	...
1	1	1	0	0	0	-1.0
1	1	1	0	0	1	0.0
1	1	1	0	1	0	+1.0
Speaker Gain Range -57 dB to +6 dB @ +1 increment						
...	...	...	...	...	...	...
1	1	1	1	0	1	+4.0
1	1	1	1	1	0	+5.0
1	1	1	1	1	1	+6.0

Figure 14: gain value of Speaker

Customer could set MOUTBST from 0 to 1 to set the MOUTBST to 1.5*vref, so the output gain would increase. The AT Command AT+CWIIC=0x34,0x62,0xA,1
Use "AT+CWIIC=0x34,0x62,0x6,1 ", customer could set the SPKOUT output power to 1.5 times.

Addr	D8	D7	D6	D5	D4	D3	D2	D1	D0	Default
0x31	0	0	0	0	0	MOUTBST	SPKBST	TSEN	AOUTIMP	**0x002**

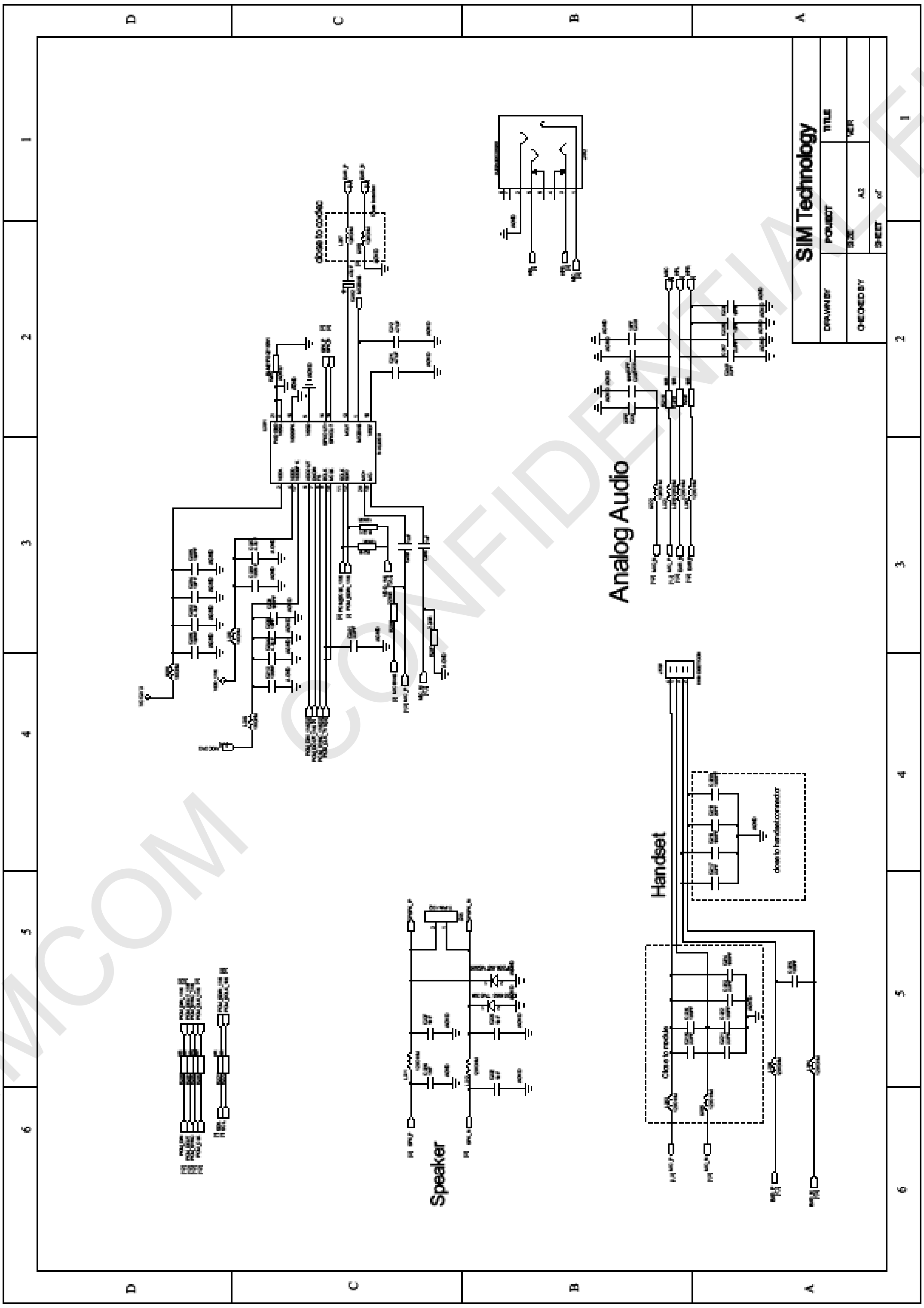
Figure 15: 0x31 register configuration

7 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 850 Mhz/900Mhz and 1800 Mhz/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.

8 Appendix

I. SIM7600 external audio code schematic



Contact us:

Shanghai SIMCom Wireless Solutions Ltd.

Add: SIM Technology Building, No.633, Jinzhong Road, Changning District, Shanghai P.R. China 200335

Tel：+86 21 3235 3300

Fax：+86 21 3235 3020

URL：www.simcomm2m.com