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IS20XX-002 DSP APPLICATION NOTE – INTRODUCTION TO DSP TOOL AND AEC TUNING GUIDE (V1.1)

- **Speaker (Phone) Application**
- **Embedded Car Kit Application**

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Change History

Version	Date	Description of Changes	Author
Draft	Spe./04, 2014	Initial version	YT Lin
1.1	Dec./04, 2014	Add description for audio effects	YT Lin

Index

Figure

Table

1 Introduction

ISSC IS2010/IS2015/IS2020/IS2025-002 chips provide high-performance noise reduction and acoustic echo cancellation with advanced signal processing techniques. Sophisticated voice and audio enhancement functions, including filtering, equalizations and intelligent MIC control, are also provided. Combining with these state-of-the-art processing and these voice and audio functions working together, a simple tuning flow for tuning the AEC and NR accommodated for different applications are given for easy product development as well.

This document will also give the tutorial about how to tune each of the parameters that are provided in the DSP tool step by step to allow system designers to fit DSP features with particularly desired requirements. The DSP configuration tool provides the visual interface to adjust the parameters for all provided voice and audio signal processing functions.

2 Overview

ISSC IS2010/IS2015/IS2020/IS2025-002 chips feature high-performance signal processing that can provide the excellent voice/audio user experience. Included fundamental and optional modules are Stationary Noise Reduction (**NR**), Acoustic Echo Cancellation (**AEC**), Audio Equalization (**EQ**), and High-Pass Filter(**HPF**). In addition, AEC function is only provided at the microphone (**Uplink**) path in the SCO link connection.

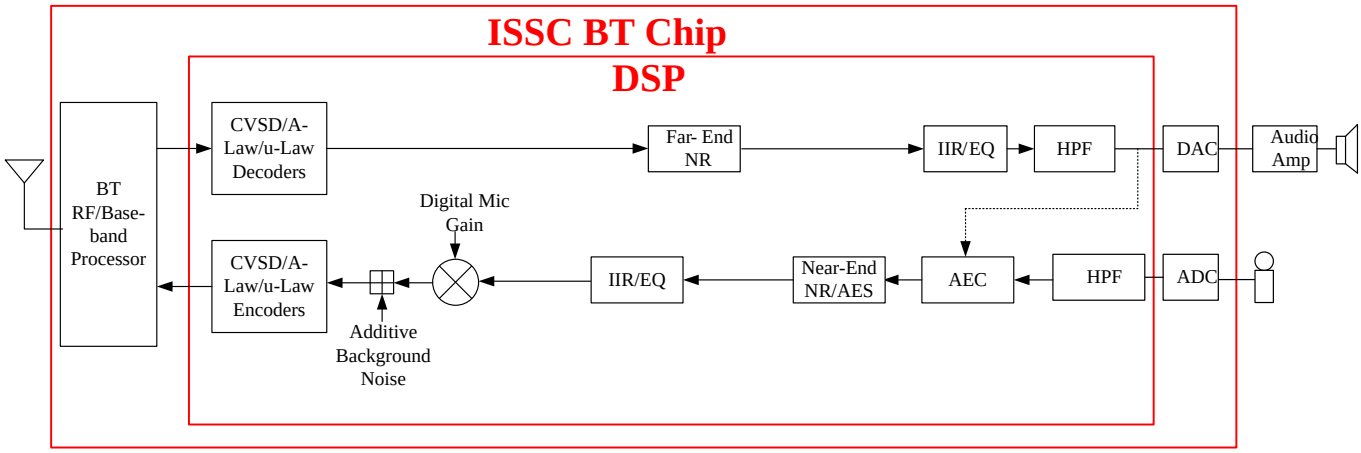
The functionality of each these modules are summarized in Table I.

Table 1: Summary of module functionalities

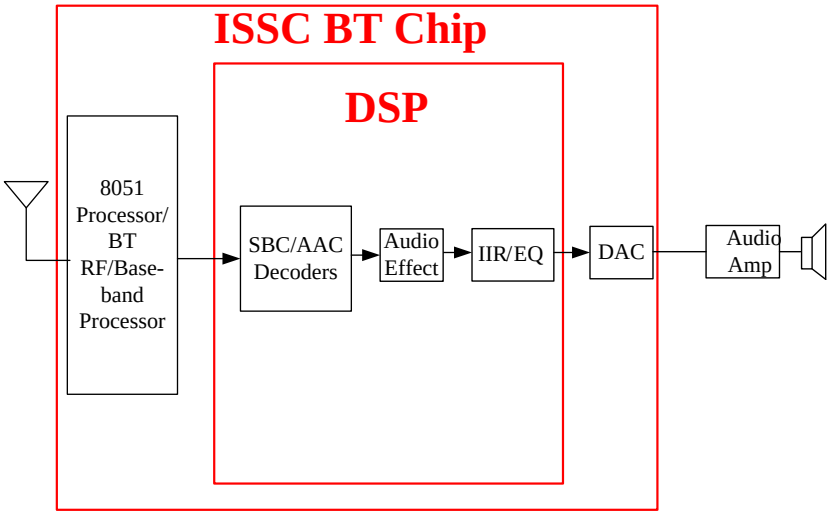
Module	Functionality
AEC	Cancel the acoustic echo coupled into the microphone for the loud speaker output.
NR	Suppress the stationary ambient noise to enhance the voice signal quality.
EQ	Provides the 5-band EQs for both voice and audio applications in order to compensate imperfect frequency response of the adopted microphone or speaker. Defaulted audio effects are also provided to enhance the user experience.
HPF	Provides a low-latency IIR-structured low-pass filter to filter out unwanted low frequency band for both MIC/SPK paths.

3 Processing Flow

Before introducing the signal processing flow, some abbreviations and terminology are defined in advance for avoiding the further confusion. The path, that BT device receives bitstream and pass to DSP for decoding process, can be called as the **downlink**, **downstream**, **far-end** and **speaker** paths in this document. On contrary, the path, that BT device transmit the bitstream that is encoded by the DSP processor, is called as **uplink**, **upstream**, **near-end** and **MIC** paths.



(a)



(b)

Figure 1: The block diagram of the processing flow for the speakerphone applications for (a) speech and (b) audio signal processing.

Figure 1 gives a block diagram of the DSP processing flow for speakerphone/headset applications. The

DSP part is focused on speech and audio processing. The embedded ADC and DAC provide high-fidelity data conversions with 85dB and 98dB SNRs, respectively. The BT and RF/modem processors deal with the medium access control (MAC) and the wireless data transmission. For the speakerphone application, the external audio amplifiers are usually needed to boost the signal to desired volume.

4 Voice Processing Functions

In the main page, one can select the IC version to set up corresponding IC's DSP configurations. Among them, IS2010S and IS2020S are optimized for mono-headset (MHS) and stereo-headset (SHS) applications, respectively. As for speaker/speakerphone, IS2015 and IS2025, which support on-chip class-D amplifiers, are designed to have a high performance and low-cost solutions for one speaker and two speaker applications. Further, to support the requirement for external DSP processing, one can also adopt IS2023 chip for such application.

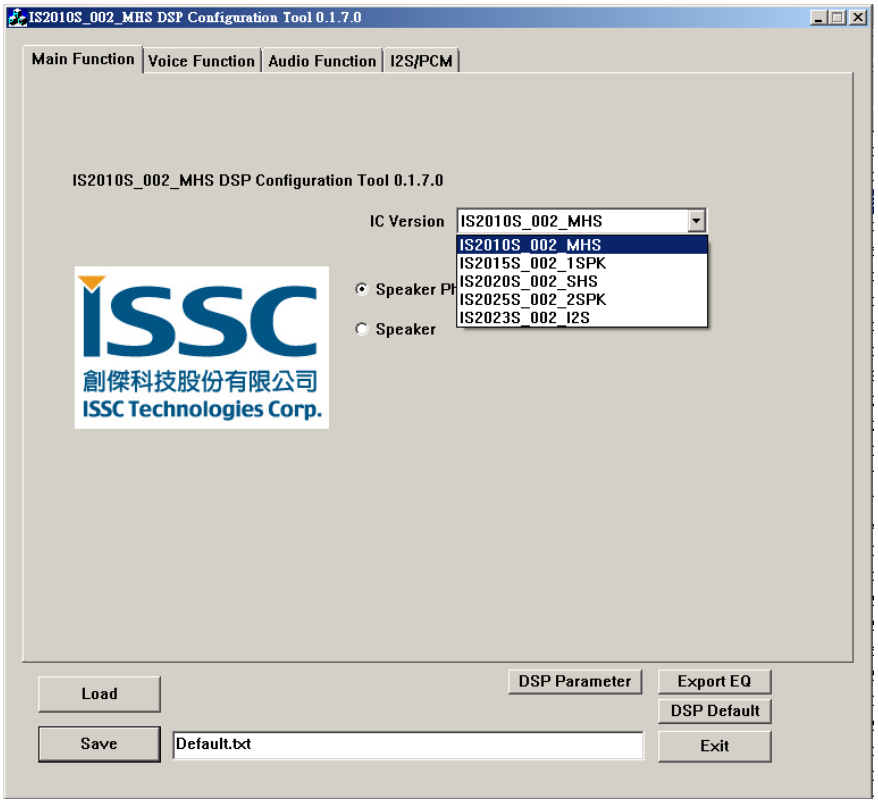


Figure 2: The main function page.

4.1 High-Pass Filter

The high-pass filter provides a low-latency filter option and this is IIR-structured filter.

Function:

Seven selectable cutoff frequencies are available for HPF which is to filter out unwanted low-frequency signal, such as PCB noise, coupled current noise and wind noise, etc. It is a trade-off between speech signal quality and the noise reduction level.

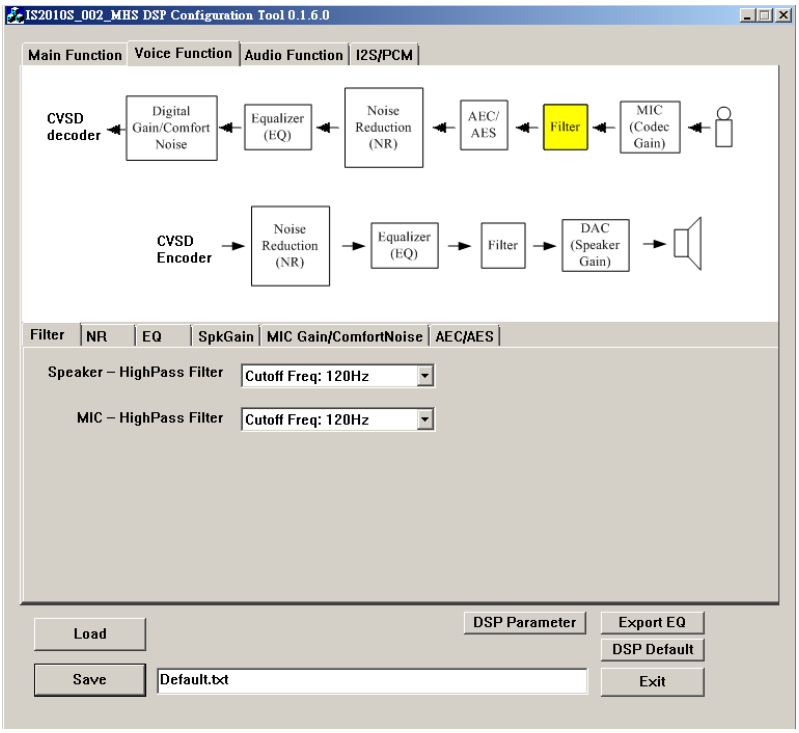


Figure 3: Configuring the High-pass filter parameter in the DSP tool.

EERPOM settings:

- To enable the far-end HPF module

Table 2 EEPROM addresses for enabling HPF at the SCO speaker path.

Addr	Bit 5 at 0x0300
Value	0x1

- To enable the near-end NR module

Table 3:EEPROM addresses for enabling HPF at the SCO MIC path.

Addr	Bit 4 at 0x0300
Value	0x1

- The EEPROM addresses to modify the NR noise suppression levels are

Table 4: EEPROM addresses for configuring NR function

	EEPROM addr	Default	Settings
Nar-end NR	0x0330	0x02	0x00: 50Hz 0x01: 80Hz

Far-end NR	0x032F	0x02	0x02: 120Hz 0x03: 180Hz 0x04: 210Hz 0x05: 300Hz 0x06: 400Hz

4.2 Noise Reduction

The noise reduction (NR) function suppresses stationary noises present in the far-end/downstream and near-end/up-stream signals. With proprietary intelligent voice activity detection (VAD), the NR module can effectively suppress the unwated noise while maintaining satisfactory quality for the speech communication. This function allows both near-end and far-end talkers to experience benefits.

The dualmic functions can only be activated when two microphone are physically enabled for headset applications. Note that, dualmic function cannot be enabled for speakerphone applications and is not designed for dualmic echo cancellation. Dualmic parameters include "DualMic_NRCoef", "Mic_Distance" and "DualMic_Stepsize."

Function:

- **NR suppression level:**

Two selectable parameters for NR configurations shown in Figure 4 are suppression levels for low-frequency (<1000Hz) and high-frequency (from 1000Hz to 4000Hz). The tunable range for both speaker and MIC paths are from 0dB to 21dB. However, in the DSP tool, only the low-frequency NR suppression level is provided while high-frequency suppression levels are determined empirically based on field-test results.

- **DualMic NR Coef:**

This parameter determines the dualmic suppression level. However, the dualmic suppression level is a tradeoff between noise suppression capability and voice quality. If the higher the suppression level is selected, the easier the voice quality is degraded.

- **Mic Distance:**

To define the distance between two microphones such that the dualmic suppression algorithm can achieve its optimal performance. If the distance between two MICs are farther (typically longer than 10cm, or say long-boom headset), one can select for higher noise suppression level since its SNR is higher than short-boom headsets (Typically around 4cm) and vice versa.

- **DualMic Stepsize:**

Denote the convergence time of the dualmic noise suppression algorithm. If choose faster convergence time, then linear noise suppression capability is worse than the slowest convergence time. It

is a tradeoff between faster convergence time and better noise suppression.

EERPOM settings:

- To enable the far-end NR module

Table 5:EEPROM addresses for enabling NR at the SCO speaker path.

Addr	Bit 2 at 0x01df	Bit 0 at 0x01ec	Bit 0-3 at 0x01E1
Value	1	1	0x1

- To enable the near-end NR module

Table 6:EEPROM addresses for enabling NR at the SCO MIC path.

Addr	Bit 1 at 0x01df	Bit 0 at 0x01ec	Bit 4-7 at 0x01E1
Value	1	1	0x1

- The EEPROM addresses to modify the NR noise suppression levels are

Table 7: EEPROM addresses for configuring NR function

	EEPROM addr	Default	Settings
Nar-end NR	0x032D	0x0F	0x00: 0dB suppression 0x06: 6dB suppression 0x09: 9dB suppression 0x0C: 12dB suppression 0x0F: 15dB suppression 0x12: 18dB suppression 0x15: 21dB suppression
Far-end NR	0x032B	0x0F	

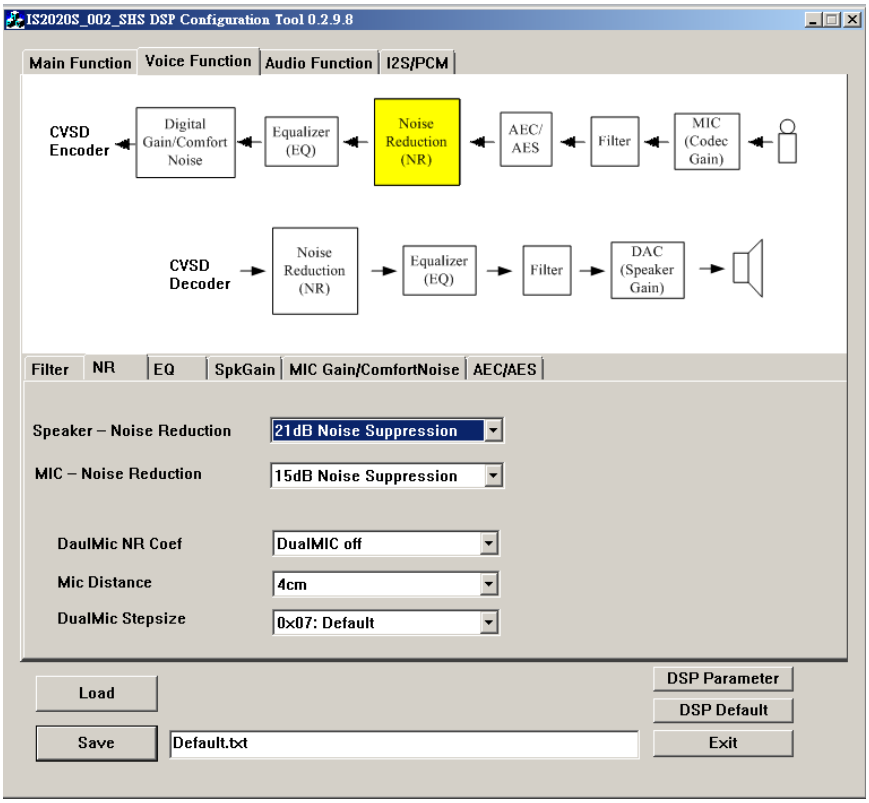


Figure 4: Configuring the NR parameter in the DSP tool.

4.3 Echo Cancellation

In order to cancel and suppress returned echo, which is coupled from the loud speaker output back into the microphones, acoustic echo canceller (AEC) and acoustic echo suppression (AES) functions are respectively needed. The difference between AEC and AES is that AEC can remove the coupled echo while maintaining the desired near-end speech. With a good AEC capability, the full-duplex speech communication can be achieved easily. However, unfortunately, nonlinear echo, caused by ID housing, analog circuit and the echo environment cannot be canceled out by AEC. Hence, what AES does is to intelligently suppress the full-band input signal with the assist of AEC information. By doing so, the desired speech quality would be degraded if the nonlinear echo dominates the MIC input signal.

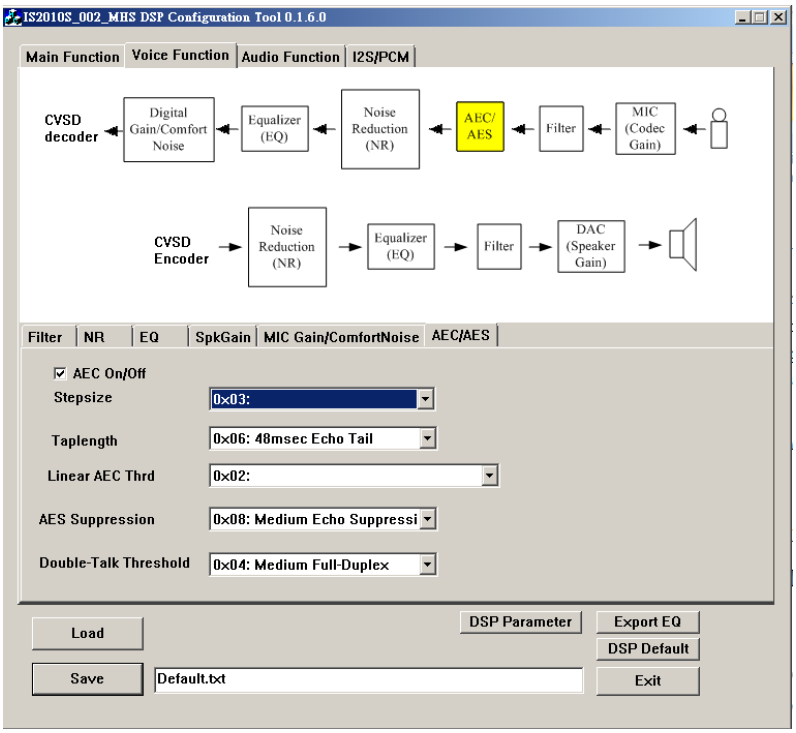


Figure 5: AEC/NR tuning interface in the DSP tool.

Function:

5 adjustable parameters are provided for the echo adjustment and they are .

- **AEC_Stepsize**: The AEC convergence speed. If fast convergence rate is selected, fewer echo is capable to be cancelled out linearly. On the contrary, better linear echo cancellation can be achieved.
- **AECTapLength**: Taplength should be selected to be longer than the echo tail and it is also a tradeoff between echo cancellation (EC) performance, mega-instructions per second (MIPS). If longer taplength is selected, higher MIPS and power consumption by DSP are the result.
- **Double-Talk Threshold**: The full-duplexity of the AEC means the level of how much can the far-end talker can listen to the near-end talker voice while both talkers speak at the same time. Basically, this parameter maps to a threshold that controls the AES to nonlinearly suppress the echo. If this parameter is set to be more favorable to half-duplexity, the double-talk capability degraded more. On the contrary, the residual echo becomes more perceivable at the far-end side.
- **AES_Suppression**: This parameter determines the maximal non-linear echo suppression capability.
- **Linear AEC Threshold**: This parameter determines the linearity threshold of the returned echo. If selecting toward “Worst linearity”, then echo suppression mechanism is easier to kick in while the “Higher Linearity” option allows to have better Full-duplex echo cancellation perforamce.

EERPOM settings:

To enable the AEC, one needs to set as follows:

Table 8: EEPROM addresses for enabling AEC function.

Addr	Bit 0 at 0x01df	Bit 4 at 0x01ec	Bit 0-2 at 0x01E2
Value	1	1	0x01

The AEC's parameters and their corresponding meaning are

Table 9: Configuring the AEC paramters

Parameters	Address	Note	Default Value
AEC_Stepsize	0x031A	0x01: Fastest AEC convergence ~ 0x06: Slowest AEC convergence	0x03
AECTapLength	0x031B	0x01: 8msec Echo Tail ~ 0x0A: 80msec Echo Tail	0x06
Double-Talk Threshold	0x0325	0x00: More Full Duplex 0x0F: Least Full Duplex	0x04
AES_Suppression	0x0324	0x00: 24dB AES Suppression ~ 0x06 :60dB AES Suppression	0x08
Linear AEC Threshold	0x031E	0x70: No Half-Duplex Suppression 0x01: 42dB Suppression	0x02

4.4 Digital MIC Gain

Digital MIC gain provides different digital control functions at the MIC path. "Digital MIC Gain" allows user to boost the volume digitally in case that the analog amplifier of ADC is unable to provide enough gain. As shown in Figure 1(a), the digital boost part of the dynamic MIC Control is placed at the end of all digital signal processing modules.

Notice:

- **Digital MIC gain:** by adjusting the digital MIC gain, one potential issue is that the suppressed echo is going to be amplified as well.

EERPOM settings:

- To enable the Digital MIC gain, one needs to configure the following addresses:

Table 10: EEPROM addresses for enabling the digital boost gain at the SCO MIC path.

EEPROM Addr	0x0318
Value	0x00: 0dB Digital Boost 0x01: 1dB Digital Boost 0x02: 2dB Digital Boost

	0x03: 3dB Digital Boost 0x04: 4dB Digital Boost : 0x14: 20dB Digital Boost

Note that the available selections of the Dynamic Range would be automatically changed by the DSP tools.

4.5 Comfort Noise

The comfort noise is generated by a random number generator and its frequency response is flat across all frequencies. This is to provide a constant noise level to let cell phone's speech codec not inject unwanted noise which would affect far-end listener's using experience.

Function:

One parameter to adjust the configuration of the comfort noise:

- **Background Noise:** This adjusts the level of the comfort noise.

EERPOM settings:

Table 11: EEPROM addresses for enabling the comfort noise at the SCO MIC path

EEPROM Address	Values
0x308 (MSB) /0x309(LSB)	0x7FFF: 0dBc Highest Comfort Noise level 0x4000: 6dBc Comfort Noise level 0x2000: 12dBc Comfort Noise level 0x1000: 18dBc Comfort Noise level 0x0800: 24dBc Comfort Noise level 0x0400: 30dBc Comfort Noise level 0x0200: 36dBc Comfort Noise level 0x0100: 42dBc Comfort Noise level 0x0080: 48dBc Comfort Noise level 0x0040: 54dBc Comfort Noise level 0x0020: 60dBc Comfort Noise level 0x0010: 66dBc Comfort Noise level 0x0008: 72dBc Comfort Noise level **0x0004: 78dBc Comfort Noise level (Recommended)** 0x00002: 84dBc Comfort Noise level 0x00001: No Comfort Noise level

4.6 IIR/EQ

The IIR/EQ provides the flexibility for compensating the imperfect frequency responses of the selected MIC and speaker. This function provides a 5-band customized filter for both MIC or speaker paths.

Figure 7 shows an example about how to select the customized equalizer's coefficients. One can also assign **Custom1** to MIC path and **Custom 2** to SPK paths. Afterwards, one can click into the "**Custom EQ 1**" and then a window shown in Figure 8 is poped out.

In Figure 8, one can type in the desired frequencies and the gain/attenuations. The "**Q**" columns are to configure the cutoff frequencies of each band. An example is shown in Figure 9, the smaller the value of "**Q**" is, the wider the bandpass cutoff frequency is. Buttons, "**M+**" and "**MR**", function as the calculator's "**M+**" and "**MR**". The major purpose for these two buttons is to record the frequency responses for easy analysis comparison.

If want to store the frequency response that just being designed, click "**Save**" in Figure 9 and system would automatically store the current EQ's configuration into a file. Also, if want to restore a frequency response that have been previously designed, click "**Load Response**" to restore the EQ configurations.

The column "Stage" configures how many bands of 1-order IIR filter are used. The fewer the "Stages" are, the lower the MIPS as well as the power consumption is.

EERPOM settings:

In order to enable the EQ for voice application, one needs to configure the following bits:

Table 12: EEPROM addresses for enabling the EQ/IIR functions.

	Bit3 0x1EC	Bit6 of 0x1DF	Bit5 of 0x1DF	0x1E8	0x1E9	Bit 2 of 0x1E1	Bit 6 of 0x1E1
MIC path	1	1	No need	0x07	0xFF	1	No need
Speaker path	1	No need	1	0x07	0xFF	No need	1

The coefficients of the Custom1 and Custom2 are stored from 0x0356 to 0x03FD.

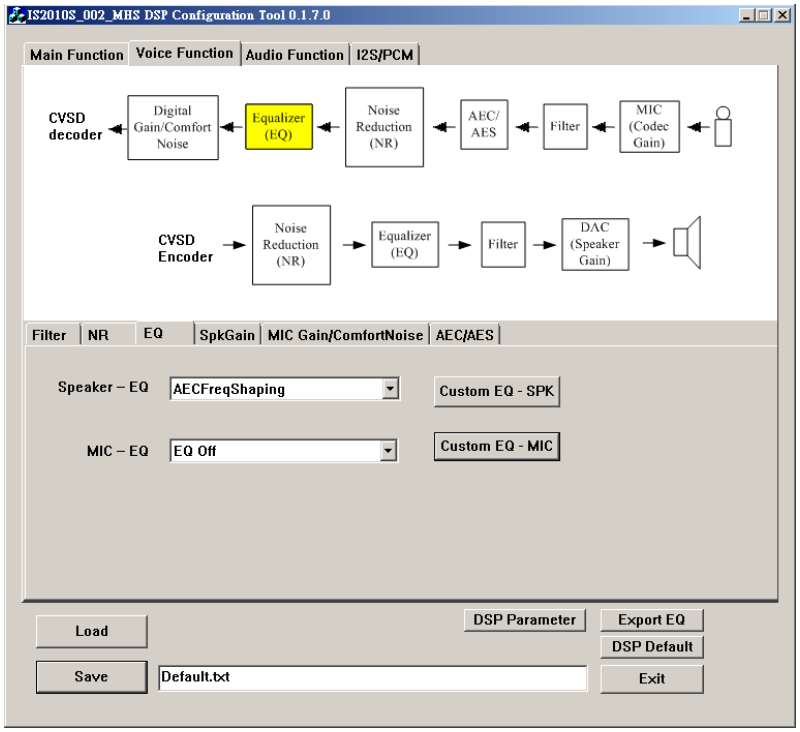


Figure 6: DSP tool interface for configuring the EQ/IIR.

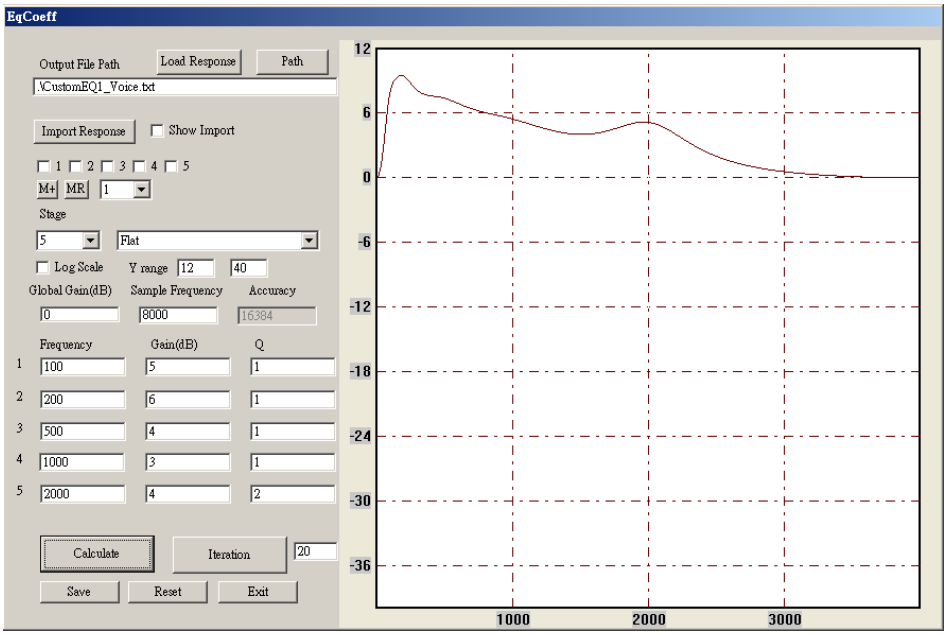


Figure 7: An example of configuring the EQ/IIR functions.

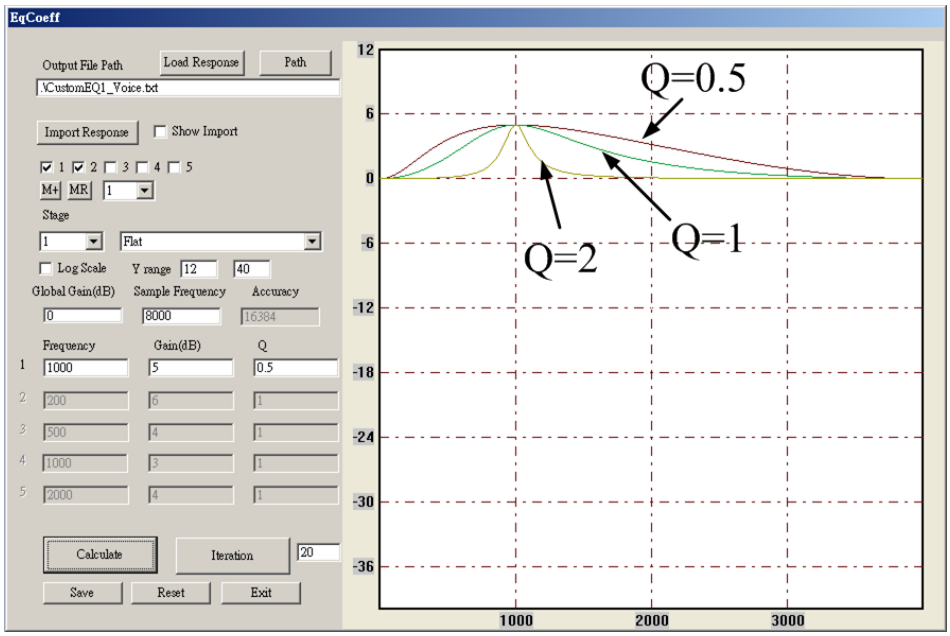


Figure 8: Illustrating the function of the Q factor.

4.7 Speaker/MIC Gain Settings:

The number of speaker gain levels and MIC gain levels are configured in the DSP tool. Figure 10 shows that there are three different number of speaker gain levels are selectable based on particular requirements. Once the number of the speaker gain level is determined, one can choose the corresponding gain for each level.

The difference between each MIC gain level is roughly between 2.7dB ~ 3.4dB per step.

EERPOM settings:

Table 13: EEPROM addresses for MIC and Speaker Gains

	EEPROM address
MIC path	0x00C5
Speaker path	0x018D~0x019C

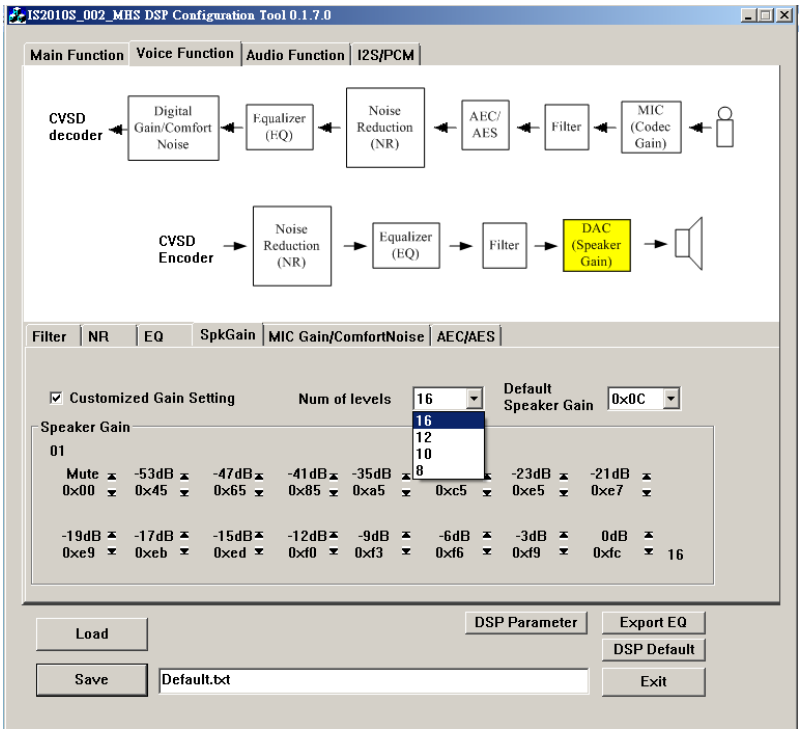


Figure 9: DSP tool interface of configuring the speaker gain in the SCO mode.

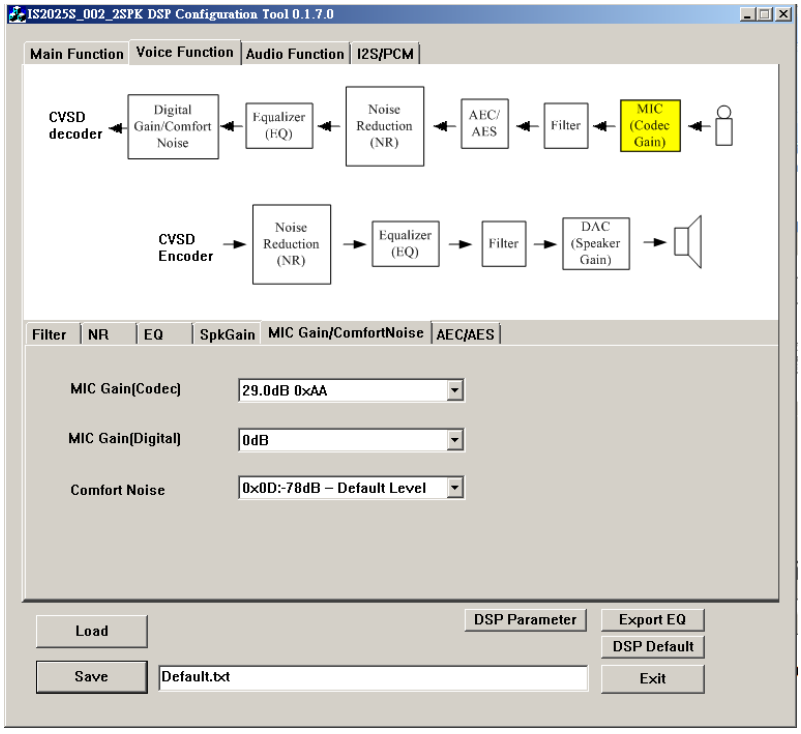


Figure 10: DSP tool interface of configuring the MIC gain in the SCO mode.

5 Audio Processing Functions

5.1 IIR/EQ

The system diagram of the audio signal processing is shown in Figure 1(b). In addition to the SBC decoder, only one IIR/EQ are allowed to process to audio signal. Figure 12 shows the configuration of the IIR/EQ for the audio function. In the column "**EQ Mask Selection**," one can select the adjustable special audio sound effect. Except the option "**Custom EQ,**" one can use an external button to select different sound effects. The procedure to configure "**Custom EQ 2**" is also identical to the IIR/EQ introduced in section 4.7.

EERPOM settings:

In order to enable the EQ for Audio application, one needs to configure the following bits:

Table 14: EEPROM addresses for enabling the IIR/EQ in the audio(or SBC) mode.

Addr	Bit3 0x1EC	Bit5 of 0x1E0	0x1E8	0x1E9
Values	1	1	0x07	0xFF

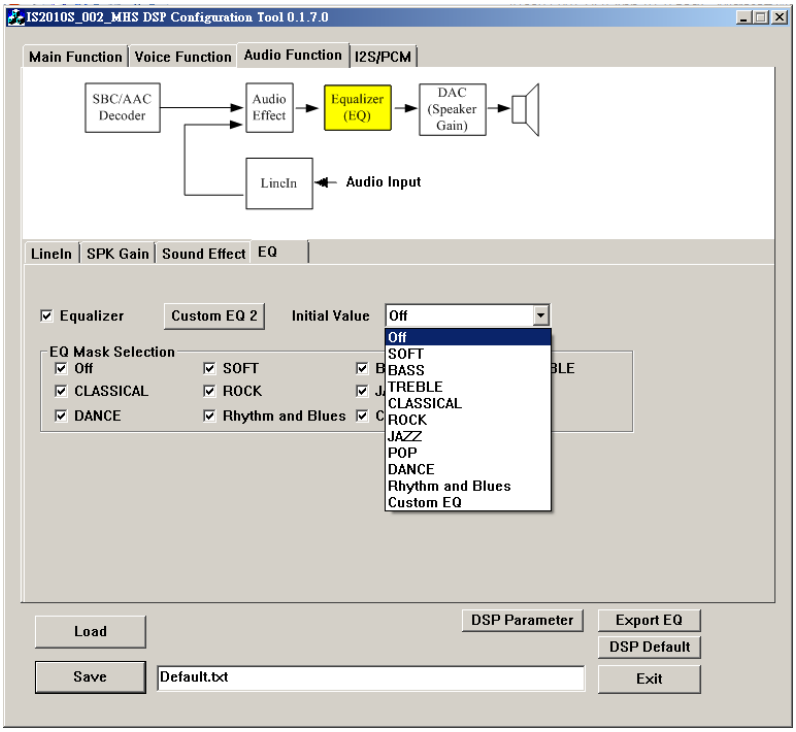


Figure 11: DSP tool interface for configuring the IIR/EQ in the audio (or SBC) mode.

5.2 Speaker Gain Setting / Line-In Gain

Similar to the speaker setting for voice application, the number of speaker gain level is also selectable. The selection procedure can refer to section 4.8.

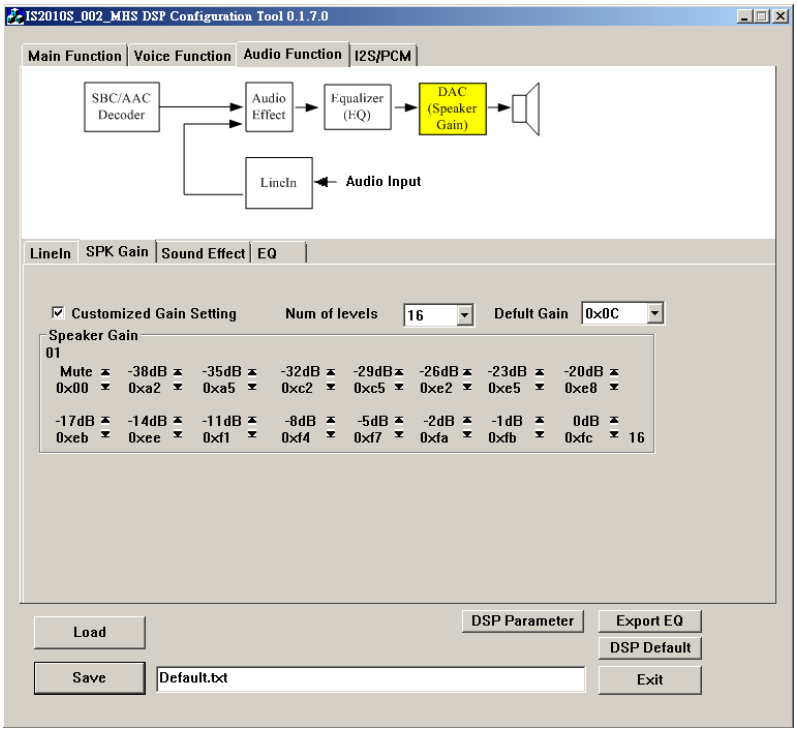


Figure 12: DSP tool interface of configuring the speaker gain in the audio (or SBC) mode.

5.3 Auto PowerOff mode for LineIn Silence detection

This DSP function enables the system to detect power level of the line-in signal. The power level of the line-in signal is calculated digitally and, then, the silence status is reported to the Bluetooth MAC controller.

"**Initial Line-In Gain**" is to configure the line-in gain to amplify the external signal source and playback to the speakers.

"**Silence Detection Threshold**" determines silence power threshold level of line-in signal for auto power-off mechanism.

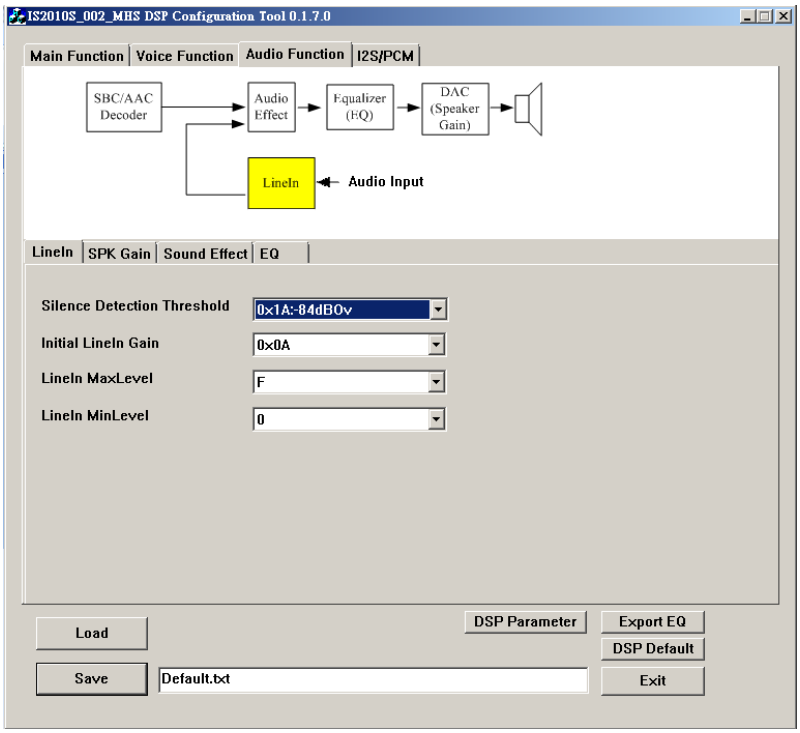


Figure 13: Configurations of LineIn interface.

5.4 Sound effect – Audio Widening (AW), Multi-band Dynamic-Range-Compression (MB-DRC)

AW and MB-DRC are embedded audio signal processing functions, which can provide better audio quality and user experience without needing external digital signal processor, in the IS20XX chip series.

AW function processes audio signal by manipulating the signal played by a close-placed speakers sounding like a farther-placed speakers. In such way, the sound quality can be enriched with better surrounding effects.

MB-DRC function is an automatic volume control. Loud sounds over a certain threshold are reduced in level while quiet sounds remain untreated-- (this is known as downward compression, while the less common upward compression involves making sounds below the threshold louder while the louder passages remain unchanged). In this way, it reduces the dynamic range of an audio signal. This may be done for aesthetic reasons, to deal with technical limitations of audio equipment, or to improve audibility of audio in noisy environments.

Checkboxes of AW and MB-DRC:

Figure 14 shows adjustable parameters for AW and MB-DRC. Checkboxes for MB-DRC and AW should be checked if one wants to enable these functions.

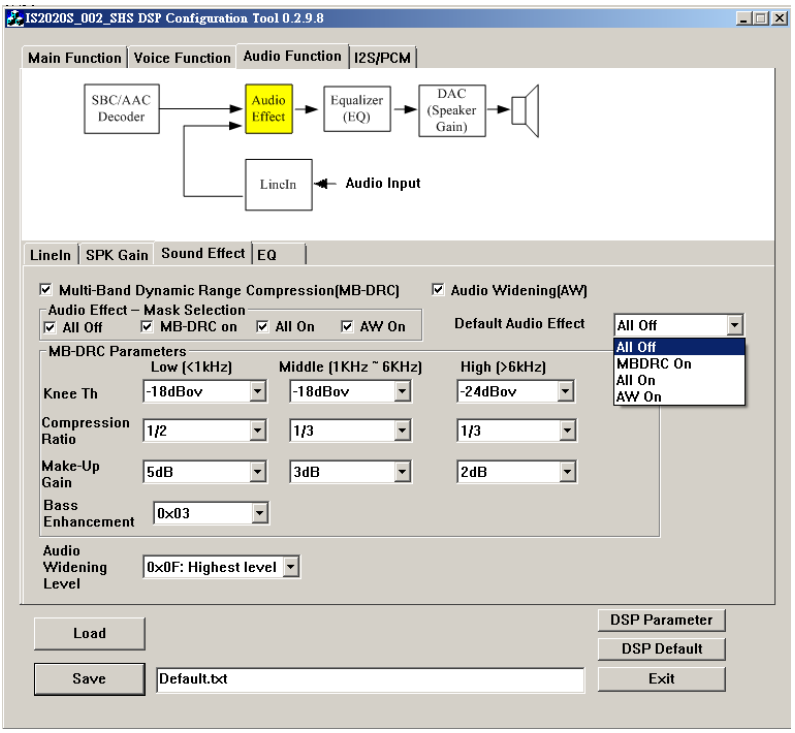


Figure 14: User interface for sound effect configurations.

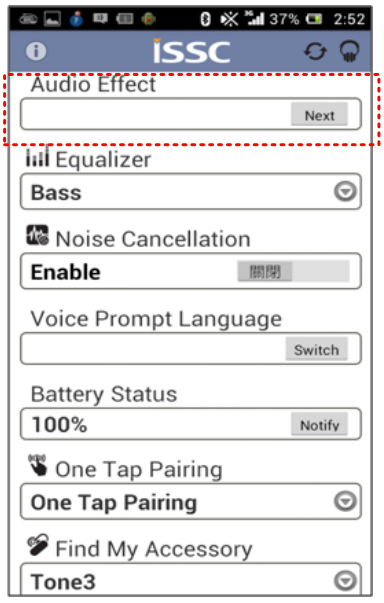


Figure 15: APP-enabled audio effect control.

Audio Effect – Mask Selection:

The “Audio Effect – Mask Selection” checkbox group is to select what combinations of audio effects can be selectable by either external buttons or APP (see Figure 15, but only on Android phone). In the

case given in Figure 14, “All off”, “MB-DRC On” and “All On” are selected. When one presses the “Next” button, the audio effect switches from one to its next. The order of selected audio effect is “All off”→“MB-DRC On”→ “All On”→ “All off”.

Default Audio Effect:

To select the initial audio effect mode after the device is powered on.

MB-DRC Parameters:

Figure 16 shows the general concept of MB-DRC which transform the input signal nonlinearly to its output. Three parameters control the behavior of MB-DRC which are “KneeTh”, “Compression Ratio” and “Make-Up Gain”. To better process the input signal, MB-DRC provides adjustable parameters in three different bands(0~1khz, 1kHz~6kHz and beyond 6kHz). Explanations of these parameters are listed as follows:

Knee TH:

This parameter corresponds to the compression threshold in Figure 16. This parameter constrains that sound level to which the make-up gain is applied.

Compression Ratio (CR):

CR is a compression ratio which compresses the average sound level, exceeding “Knee TH”, of the audio signal. However, if CR is closer to 0, the distortion due to the compression could be generated more easily.

Make-Up Gain:

Make-up gain is the maximal gain applying to audio signal whose average power is between silence threshold and the compression knee. This parameter can boost soft music signal to an audible level especially in a noisy environment.

Bass Enhancement:

This parameter controls the level of bass enhancement which is enabled along with the MB-DRC

function.

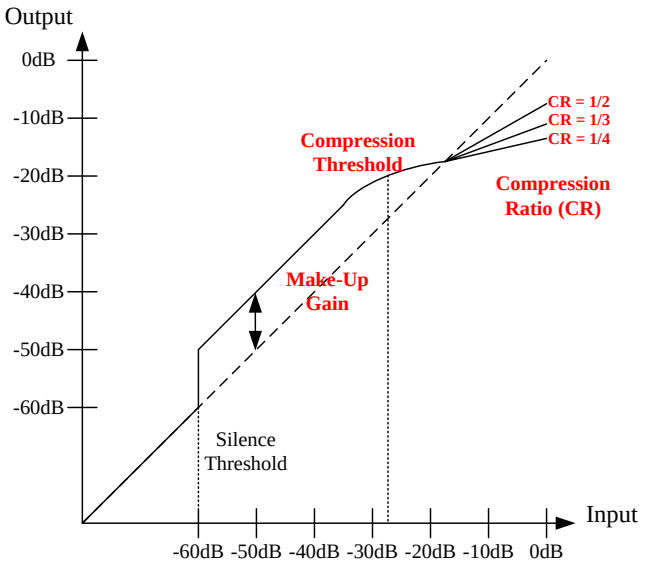


Figure 16: Mapping function of MB-DRC.

AW Parameters:

There is only one available parameter controlling the AW effect.

Audio Widening Level:

This parameter controls the extent of AW effect. It basically is a mixing ratio of original sound and AW-processed sound signal.

6 I2S Digital Output/Input Interface

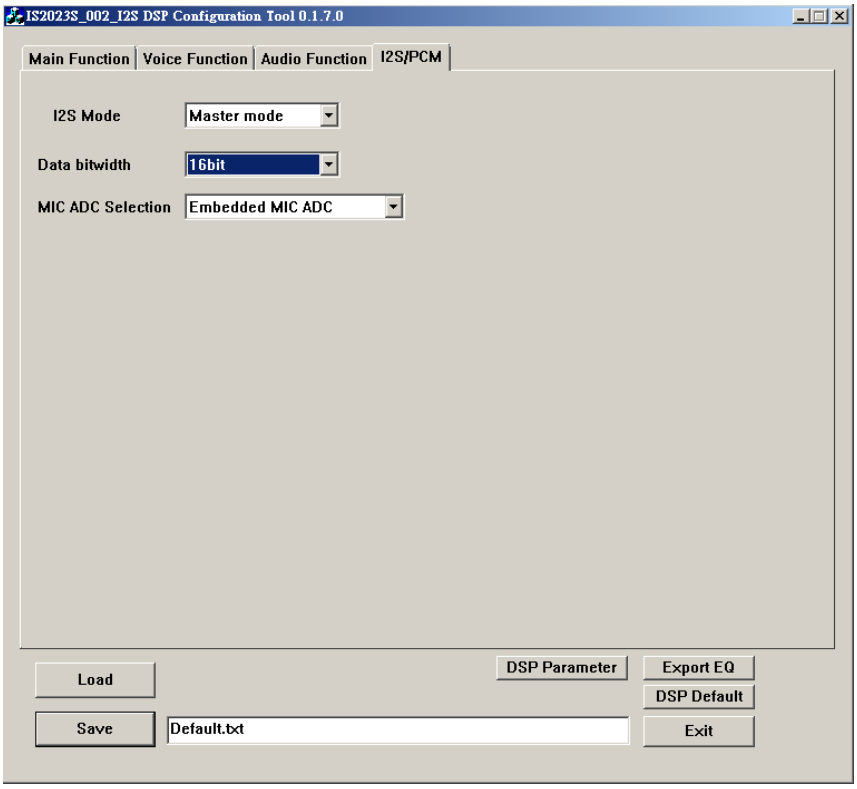


Figure 17: I2S parameter tuning interface for IS2023 chip.

IS2023-002 chip can support the I2S interface for digital input/output. In this document, the hardware wiring issue is discussed but only the FW configuration is introduced. The selectable parameters are given as “I2S Mode”, “Data bitwidth” and “MIC ADC bitwidth” as shown in Figure 17.

Details of these parameters are discussed as follows.

I2S Mode:

- **Master:** IS2023 chip serves as a master to provide clock and frame sync signals for the master/slave data synchronizations as shown in Figure 18(a).
- **Slave:** IS2023 chip serves as a slave to receive clock and frame sync signals from external codec or DSP devices illustrated in Figure 18(b).

Data Bitwidth:

The numbers of bits for DR/DT are expected to receive from or transmit to external codec or DSPs.

- **16bit:**
- **24bit:**

MIC ADC Selection:

If the hands-free function is supported, one must carefully select the ADC configuration

- **Internal ADC:** on-chip ADC is used.
- **External ADC:** external ADC/DSP is selected.

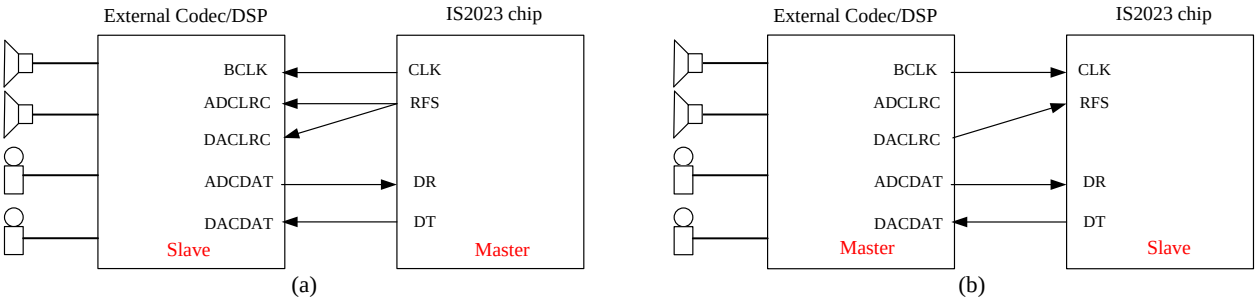


Figure 18: I2S hardware configurations for (a) Master, and (b) Slave modes with external ADC.

7 Guidelines for Tuning Echo Cancellation Performance

This section introduces a guideline to fine-tune the echo cancellation performance. There are usually three requirements, which are **high MIC volume**, **echo-free** and **double-talk** (DT) performance, for the EC tuning. These three requirements contradict with each other. For example, if requiring high volume at the MIC path, echo would be amplifies as well and need to tune some parameters to make echo inaudible which would result in worse DT performance. As shown in Figure 17, step 1 ~ step 5 are basically handling these whole speakerphone/carkit echo issues.

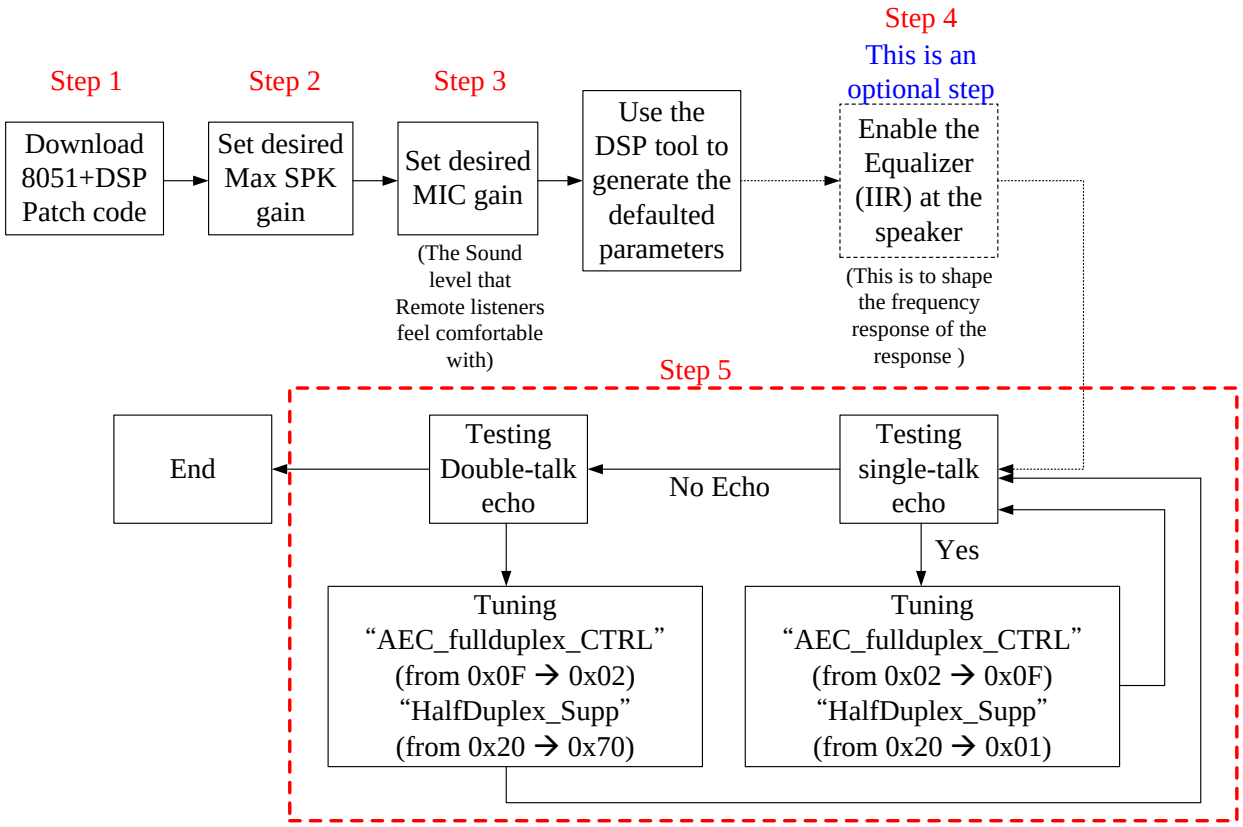


Figure 19: An illustration of the AEC tuning flow.

The details of these steps are explained as follows.

Step 1: First of all, download all of the merged patch code which consists of the settings, for user interface (UI), 8051 and DSP, and merged patch code for 8051 and DSP parts. Some DSP functions may be implemented in the patch code such that download the latest patch code can get in sync with this

document.

Step 2: Before tuning the AEC performance, the desired maximal speaker output level must be determined. More specifically, the recommended speaker output volume should be at least 95dB SPL (sound pressure level) and 100dB SPL for indoor speakerphone and car-kit applications, respectively. Note that the target speaker output volume needs to be determined in the beginning based on the required specification.

Step 3: The principle to adjust the MIC gain to a suitable value. It is not necessarily good to set the MIC to its maximal level because the slope overlaod effect, as shown in Figure 15, caused by the CVSD codec itself would naturally suppress the high frequency parts and makes it look like a low pass filter. This effect would distort the near-end speech and make it not as clear as the softer MIC gain levels.

(See the reference link for more information of the CVSD slope overlaod effect: http://www.datasheetcatalog.org/datasheet/CML/mXxyzvw.pdf)

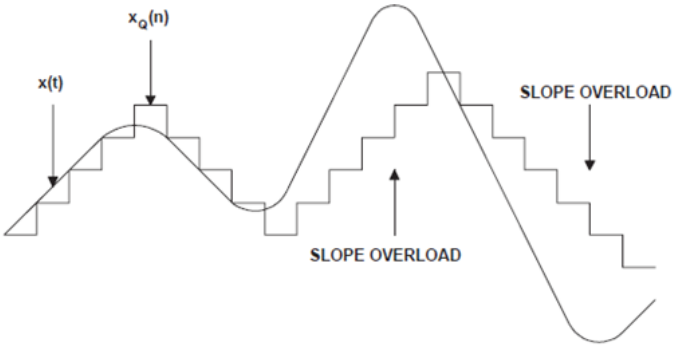


Figure 20: Illustration of the CVSD slope overload effect, where x(t) and $x_Q(n)$ are denoted as the original signal and CVSD encoded/decoded signal, respectively.

Step 4: This is an optional step. The purpose of this step is to shape the frequency response of the speaker output by lowering the low frequency (<1 kHz) and enhancing the high frequency parts (1 kHz to 3 kHz). By doing so, the echo reverberation within the speakerphone/ car kit housing can be reduced such that the linearity of the echo coupled to the MIC input can be better. The echo linearity is highly associated with the AEC performance. An example is given in Figure 14; these settings can is obtained empirically. However, the required frequency shaping may vary in terms of what the speaker and the housing is selected.

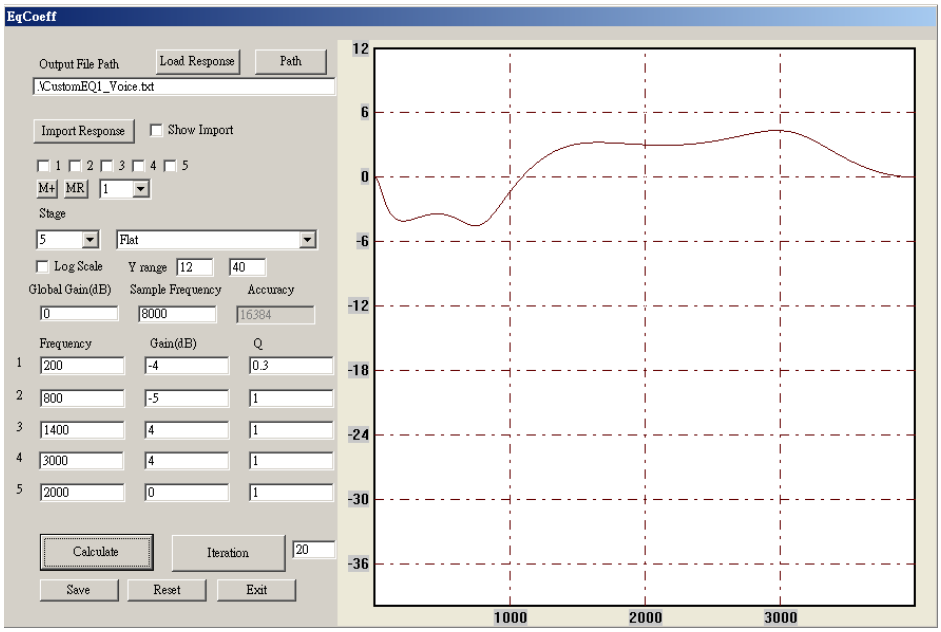


Figure 21: An example of the frequency shaping for signal at the speaker path.

Step 5: This step is to finetune the AEC performance step. It basically breaks into two parts which are single-talk echo and the double talk echo tunings.

- **Single-talk echo**: "**Double-Talk Threshold**" and "**AES Suppression**" introduced in section 4.4 are responsible for tuning the single-talk echo performance. As a rule of thumb, firstly adjust the parameter of "**Double-Talk Threshold**" from 0x01 to 0x0F. If the value of the "**AES Suppression"** is 0x04 and still can't effectively suppress the echo (audible single-talk echo), then, start to fine tune the "**AES Suppression**" from 0x01 toward the value 0x0F.

  Note that: Although the selectable value of "**Double-Talk Threshold**" can be up to 0x0F, these two values are not recommended to suppress the echo since it would distort the MIC speech severely.

- **Double-talk echo**: If the single talk echo can already be effectively suppressed by the default settings of "**Double-Talk Threshold**" and "**AES Suppression**", then, the double-talk performance can be further moved on and finetuned. The first recommended parameter for the double-talk performance is "**AES Suppression**" which is suggested to tune from 0x0F to 0x01(No Half-duplex). If the single-talk echo is still not present for "**Double-Talk Threshold**" at 0x01, the parameter "**AES Suppression**" is then considered to be adjusted from 0x0C to 0x01.

Four possible measures to improve the <u>double-talk</u> performance:

1. *Increase the AEC MIC gain*: Go back to step 3 to increase the MIC gain. Because AES and AEC would suppress echo as well as the double-talk near-end speech, the double-talk performance can be improved if the near-end speech energy is raised up such that the near-end speech may become audible while no single-talk echo is present.

2. *Adjust the frequency shaping shown in Figure 14*: If the speaker output level is very loud and echo-to-speech ratio at the MIC input is too high, one way to improve this is to further suppress the low-frequency part of the speaker output. As a result, the echo-to-speech ration at the low frequency parts of the MIC input is further reduced and could have better double-talk performance.

3. *Use the handsets supporting the full-duplex AEC*: Some handsets, such as Samsung Gallaxy II, and HTC Incredible etc., don't support the full-duplex speech communication while connecting with the BT handfree devices. In this way, one can't obtain satisfactory full-duplex performance while using these handsets.

4. *Allow only one channel output(Assuming for the stereo speakerphone case)*: If the distance between the MIC and one of the speaker channels are very close (<4cm), the AEC tuning for the full-duplexity becomes very difficult. One simple way is to turn off the closer speaker channel output and only allow the other speaker to output such that the full-duplexity can be much easier to be achieved. The User Interface (UI) can configure the number of the desired speaker channel outputs.

8 Simple Manual for the DSP Tool

Figure 20 shows the cover page of the DSP tool. There are four subpages placing on the top of the tool layout. They are "**Main Function**", "**Voice Function**", "**Advanced Voice Function**" and "**Audio Function**" where the later three pages are introduced in previous sections.

This section mainly introduces the functions within the "**Main Function**" subpage.

Two available configurable modes, "**Speaker Phone**" and "**Speaker**" modes, for overall DSP settings are given. The "**Speaker Phone**" mode allows the user to configure the NR/EC/EQ/FIR functions while "**Speaker**" mode is not permitted to do so. Whenever selecting either one of these mode, please click the "**DSP Default**" button to initialize all the defaulted settings.

The button "**Load**" is to load in the EEPROM table that may contain all the UI, all other behavior settings as well as the DSP parameters. For DSP related parameters, the DSP tool can parse them in the load-in file and automatically show its corresponding value in other subpages.

The buttons "**Save**" is to the current settings in the tool and transform them into the EEPROM format. The checkbox "**Write Complete Table**" is selected if want to merge the all UI settings, which are previously loaded in, into the output file. If not selected, only the DSP parameters along with the DSP patch code are stored in the output EEPROM file.

The "**Export EQ**" button is only to export the coefficients of the IIR/EQ for tuning the frequency responses for MIC and speaker.

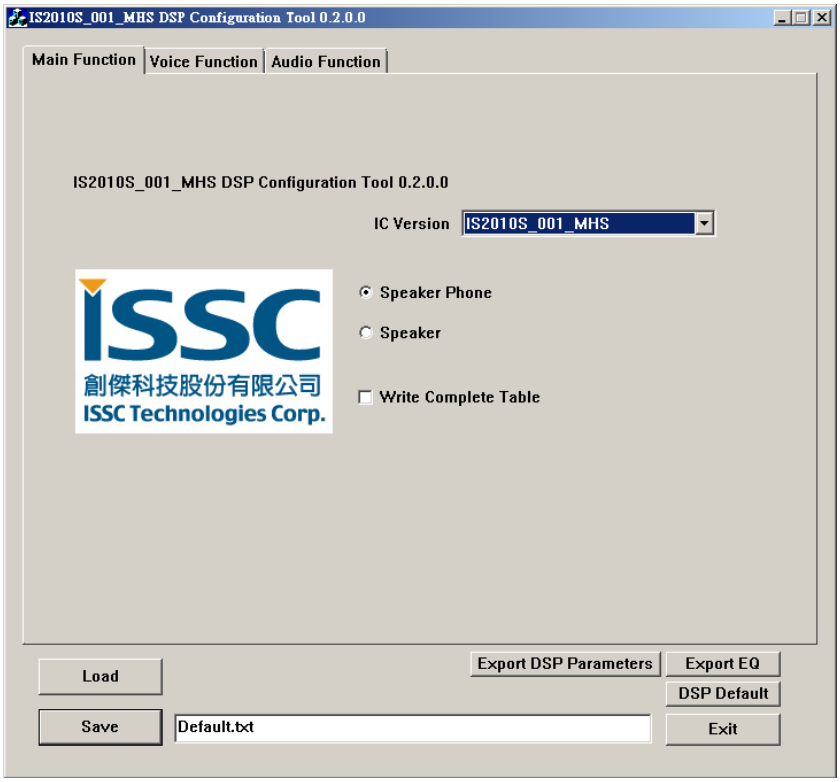


Figure 22: The cover page of the DSP tool.