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# IS206X DSP APPLICATION NOTE – INTRODUCTION TO DSP CONFIGURATION TOOL AND AEC TUNING GUIDE

**(V1.1)**

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

# Change History

| Version | Date | Description of Changes | Author |
|---|---|---|---|
| Draft | Jun./08, 2015 | Initial version | YT Lin |
| Ver. 1.0 | April/04, 2016 | Release version | Allen Lo |
| Ver. 1.1 | Auguest/01, 2016 | Release version- add an explanation for I2S | Allen Lo |

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# 1 Introduction

IS206X 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 audio effect processing, are also provided. A simple tuning flow for finetuning the AEC and NR accommodated for different applications are given for easy product development as well.

This document will also provide the guidance to finetune parameters provided in the DSP configuration 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

IS206X 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.

In addition to NR/AEC function, audio effect functions, including multi-band dynamic-range-compression (**MB-DRC**), virtual bass enhancement (**VB**) and audio widening (**AW**), for A2DP audio streaming are also available to enhance the audio quality for various applications. For mono speaker/speakerphone and stereo headset applications, MB-DRC and VB can be enabled to have much better audio clarity performance. For stereo speaker/speakerphone applications, in addition to turn on MB-DRC and VB, AW is also available to provide much betterlive audio effect for the users.

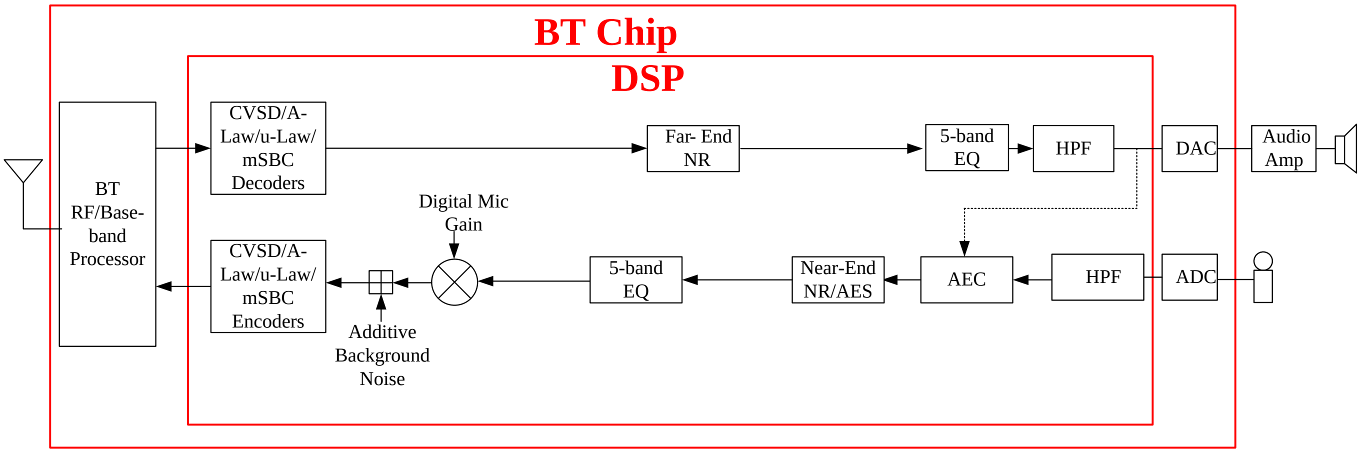
Main functionalities are summarized in Table I.

**Table 1: Summary of module functionalities**

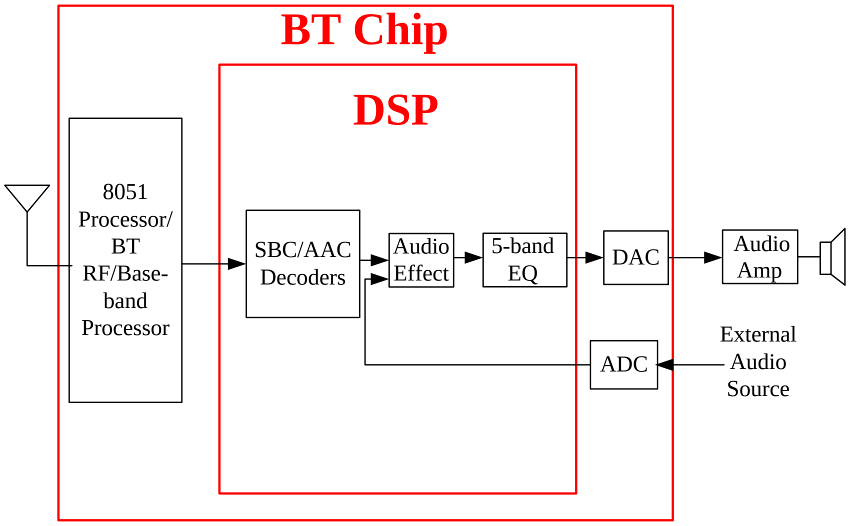
| Module | Functionality |
|---|---|
| AEC | Cancel the acoustic echo coupled into the microphone for the loud speaker output. |
| NR | Suppress the stationary ambient noise to enhance the voice signal quality. |
| EQ | Provides the 5-band EQs for both voice and audio applications in order to compensate 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 providesthe block diagram of the DSP processing flow for speakerphone/headsetapplications. The DSP part is focused on speech en/de-coders and audio decoders along with their corresponding signal processing functions. The embedded ADC and DAC provide high-SNR data conversions with 88dB and 98dB SNRs, respectively. The BT and RF/modem processorsdeal with the medium access control (MAC) and the wireless data transmission. For the speaker/speakerphone application, the external audio amplifiers are usually needed to amplify the audio signal.

In Figure 1(a), CVSD and mSBC speech codecs, whose supported bandwidths are respectively 8 kHz and 16 kHz, are available for Bluetooth speech applications. mSBC is the mandatory speech codec defined in Bluetooth HFP 1.6 profile to provide HD voice quality. Therefore, as long as the host device establishes the BT link via HFP 1.6 profile, mSBC is the speech codec regardless of the cellular network conditions, i.e. 3G network or VoLTE.

As for the audio effect in Figure 1(b), it includes MB-DRC, VB and AW functions followed by the 5-band EQ. The digital line-in loopback mode allows picking up signal from external audio source and feedback into the DSP for audio processing whichalso goes through the audio effect and 5-band EQ functions as the BT A2DP audio streaming. AAC-LC and SBC audio codecs are both supported in this chip series. However, for legal usage of AAC-LC decoder, licensing from “Via Licensing” is necessary before the product launch to market.

# 4 IC and DSPSimple Manual for the DSP Configuration Tool

Figure 2 shows the main page of the DSP configuration tool. Four subpages, “**Main Function**”, “**Voice Function**”, “**Advanced Voice Function**” and “**Audio Function**”,are on the top of the tool layout.

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 the parameters back to their defaulted settings. The button “**Load**” is to load and parse the information of DSP-related parameters in the EEPROM file and automatically show its corresponding value in other subpages.The button “**Save**” is to save current settings in this GUI tool and transform them into the EEPROM format (Please refer to section 9 for simple introduction to EEPROM format). “**DSP Parameter**” button is to export NR/AEC/Sound Effect/Partial EQ coefficients/DSP-system control parameters but not include DSP patch code, I2S coefficients, and remaining EQ coefficients.In the main page, one can also select desired IC number andconfigure its corresponding DSP behavior. Note that product portfolio for mono-headset (MHS), stereo-headset (SHS), mono speakerphone (1SPK), stereo speakerphone (2SPK) and I2S interface applications are all supported in the DSP configuration tool.

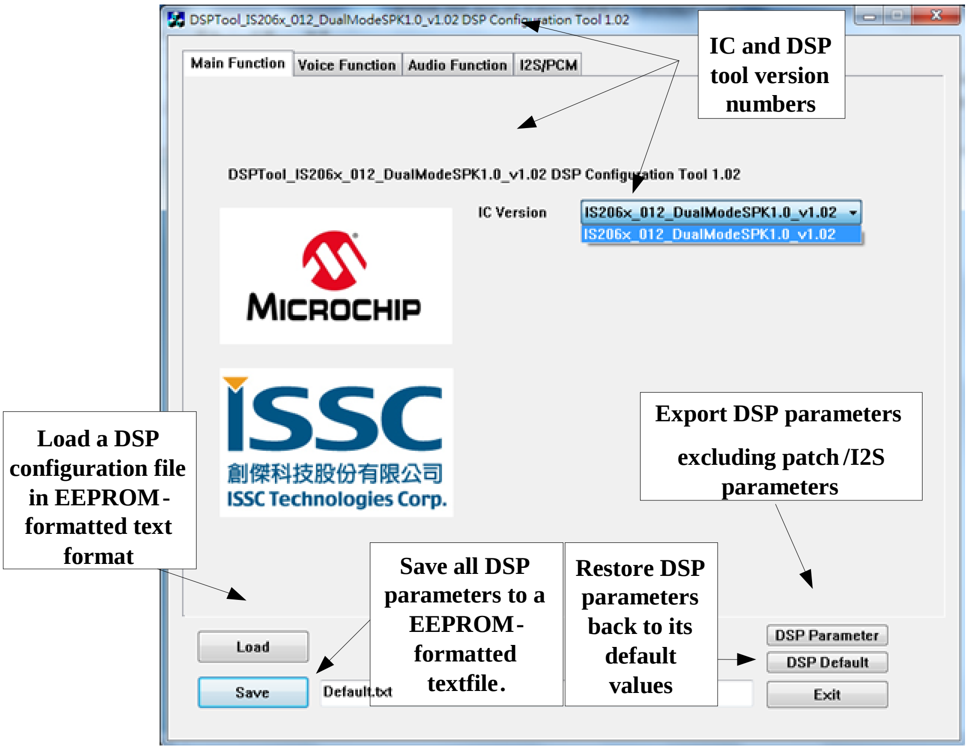



**Figure 2: The main page of the DSP configuration tool.**

# 5 Voice Processing Functions

## 5.1 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. The interface of HPF parameters for both speaker and mic paths are illustrated in Figure 3.

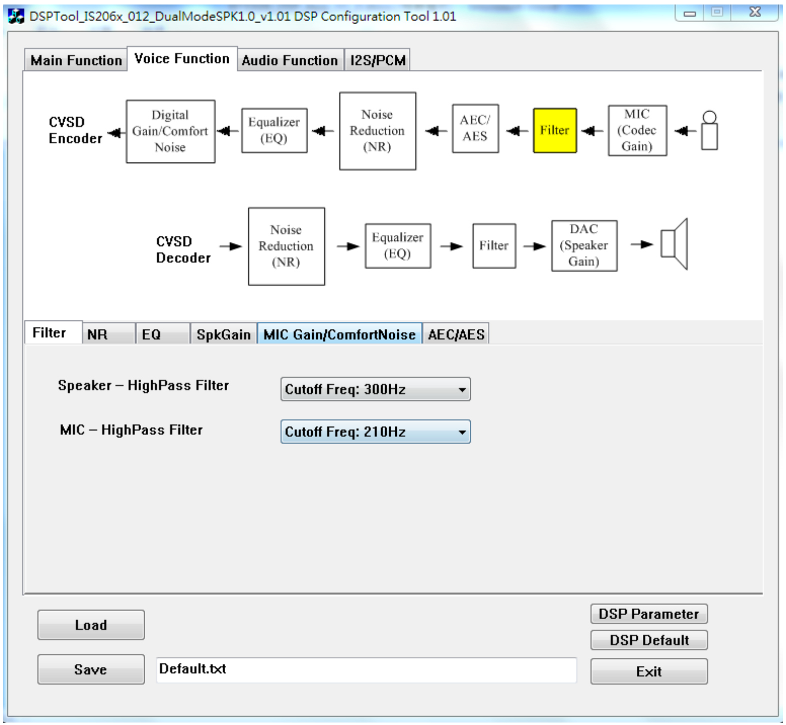



**Figure 3:Configuring the High-pass filter parameter in the DSP configuration tool.**

**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 HPF module, referring to Table 3.

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

| Addr | Bit 4 at 0x0300 |
|---|---|
| Value | 0x1 |

- The EEPROM addresses and settings for HPF are

**Table 4: EEPROM addresses for configuring HPF function**

| | EEPROM addr | Default | Settings |
|---|---|---|---|
| Nar-end HPF | 0x0316 | 0x02 | 0x00: 50Hz<br>0x01: 80Hz<br>0x02: 120Hz<br>0x03: 180Hz<br>0x04: 210Hz<br>0x05: 300Hz<br>0x06: 400Hz |
| Far-end HPF | 0x0317 | 0x02 | |

## 5.2 Noise Reduction

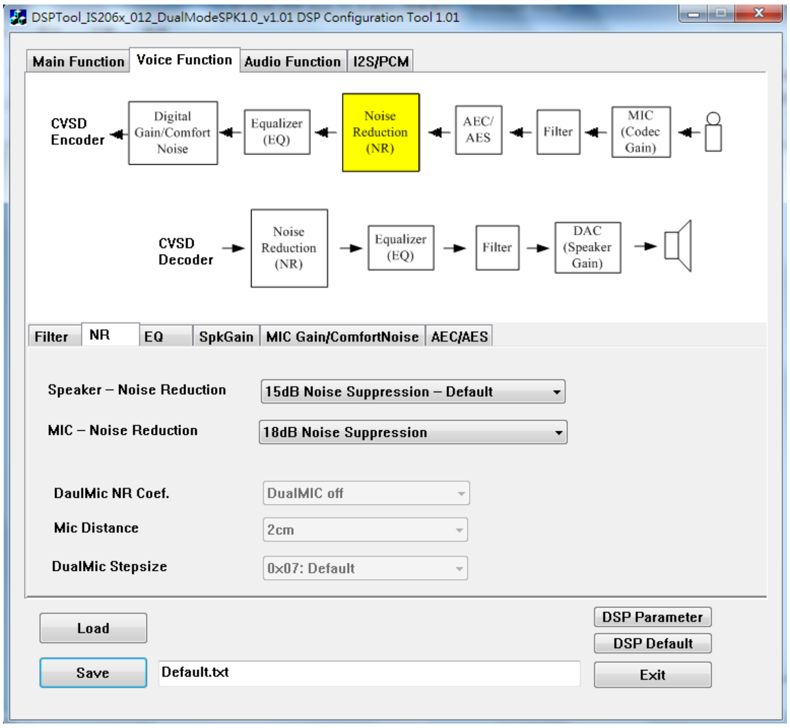



**Figure 4:Interface of the NR and DualMic NR parameters in the DSP configuration tool.**

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.

In Figure 4, the dualmic (dual microphone noise suppression) function 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 the design target 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 configuration 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 dualmicsuppression 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 to achieve its optimal performance of the dualmic suppression algorithm.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 at the main microphone side is much higher than short-boom headsets (Typically around 4cm).

- **DualMicStepsize:**

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**

<table>
<tr><th></th><th>EEPROM addr</th><th>Default</th><th>Settings</th></tr>
<tr><td>Far-end NR</td><td>0x0325</td><td>0x16</td><td rowspan="2">0x7F: 0dB suppression<br>0x40: 6dB suppression<br>0x2D: 9dB suppression<br>0x20: 12dB suppression<br>0x16: 15dB suppression<br>0x10: 18dB suppression<br>0x0B: 21dB suppression<br>0x08: 24dB suppression<br>0x06: 27dB suppression<br>0x04: 30dB suppression</td></tr>
<tr><td>Near-end NR</td><td>0x0327</td><td>0x10</td></tr>
</table>

## 5.3 Echo Cancellation

In order to linearly cancel and suppress returned echo, acoustic echo canceller (AEC) and acoustic echo suppression (AES) functions are both needed. The difference between AEC and AES is that AEC can cancel out the linearly coupled echo while maintaining the desired near-end speech. With good AEC performance, the full-duplex speech communication can be achieved easily. However, unfortunately, nonlinear echo, generated by speakerphone housing, imperfect speaker/microphone devices and undesired echo environment, cannot be canceled out by AEC. Hence, the function of AES is to suppress input signal mixed with desired signal and unwanted echo signal in the frequency domain based on the information of voice activity detection. One needs to carefully select the parameters for AES and double-talk threshold to preventfrom the degradation of desired speech quality in the case ofstrong nonlinear echo or close placement between microphone and speakers.

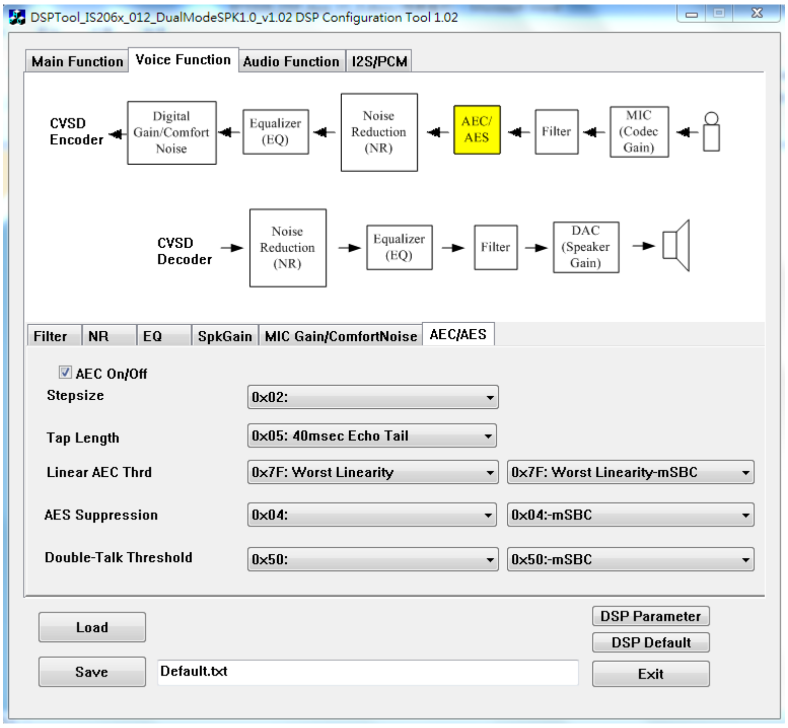



Figure 5: AEC tuning interface in the DSP configuration tool.

**Function:**

In Figure 5, adjustable parameters to finetune the performance of echo cancellationare provided as follows:

- **<u>AEC_Stepsize</u>**: 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.

- **<u>AECTapLength</u>**: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.

- **<u>Double-Talk Threshold</u>**: 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 configured to be more favorable for half-duplexity, the double-talk capability is supposed to degrade more but can remove more residual echo.

  If assuming that x[n] and y[n] are respectively near-end and far-end signal, then the cross-correlation equation between these signals are expressed as:

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$$Echo\_corr = \mathrm{EMA}(corr\_coef)$$

where $corr\_coef = \dfrac{\sum x[n]y[n]}{\sqrt{\sum x[n]^2}\sqrt{\sum y[n]^2}}$ and EMA: exponetially moving average

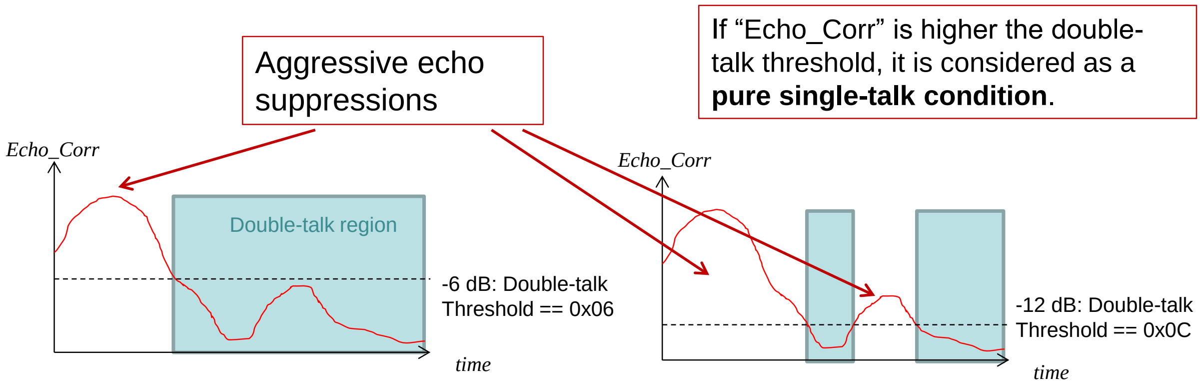


**Figure 6: Illustration of double-talk threshold parameter.**

In Figure 6, it shows that if selecting double-talk threshold as 0x50, the threshold to determine the presence of echo is higher, and, as a result, AES function is less likely to kick in to suppress residual echo. On the contrary, for the case of Double-talk Threshold as 0x2D, this setting is more actively to enable AES function for echo suppression.

- **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 AESfunction is configured to more easily kick in and suppress the residual echo while the "Higher Linearity" option allows having better Full-duplex echo cancellation performance but may contain more echo in the returned signal.

**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** | 0x032D<br>Bit 4~7 | 0x01: Fastest AEC convergence ~<br>0x06: Slowest AEC convergence | 0x03 |
| **AECTapLength** | 0x0311 | 0x01: 8msec Echo Tail ~<br>0x0A: 80msec Echo Tail | 0x06 |
| **Double-Talk Threshold** | 0x0322/0x0339(mSBC) | 0x7F: More Full Duplex~<br>0x1C: Least Full Duplex | 0x50 |
| **AES_Suppression** | 0x0313/0x0336(mSBC) | 0x01: Less Echo Suppression ~ 0x10 : Most Echo Suppression | 0x04 |
| **Linear AEC Threshold** | 0x032E/0x0338(mSBC)<br>Bit 0~3 | 0x7F: Worst Linearity<br>~ 0x03: Best Linearity | 0x7F |

## 5.4 Digital MIC Gain

Digital MIC gain provides different digital control functions at the MIC path. In Figure 7, “Digital MIC Gain” allows user to boostthe 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 |
|---|---|
| MIC Gain (Digital) | 0x00: -19dB Digital Boost<br>0x03: -16dB Digital Boost<br>0x05: -14dB Digital Boost<br>0x07: -12dB Digital Boost<br>: |

| | |
|---|---|
| | 0x13: 0dB Digital Boost<br>0x15: 2dB Digital Boost<br>:<br>0x27: 20dB Digital Boost |

Note that the available selections of the Dynamic Range would be automatically updated according to the MIC gain (Codec) by the DSP configuration tool.

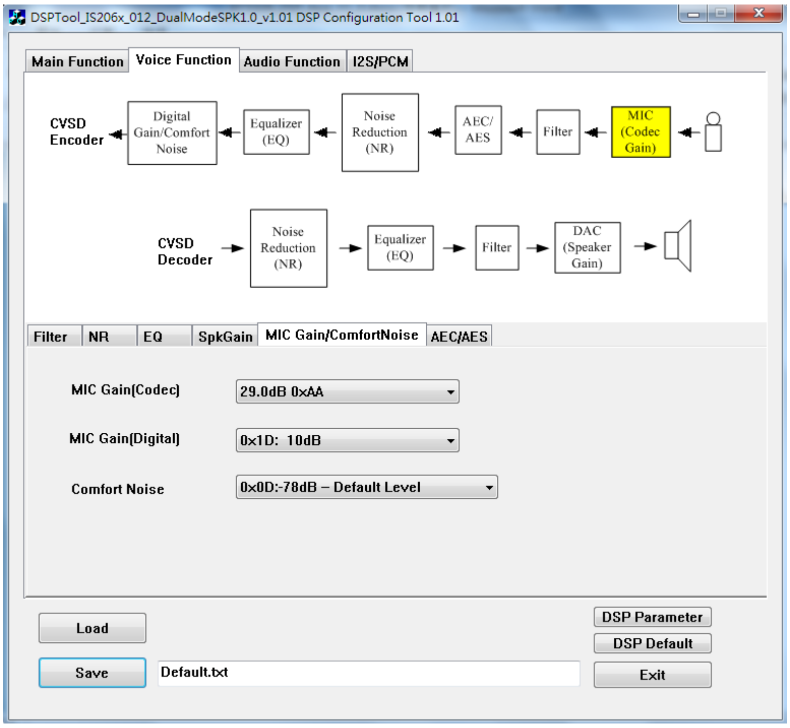


**Figure 7: Parameters for MIC gain and comfort noise.**

## 5.5 Comfort Noise

The comfort noise is generated by a random number generator and its frequency response is flat across all frequencies. The main purpose is to provide a constant noise level to preventthe speech codec algorithm of host cellphonesfrom injecting unwanted noise which could degradespeech clarity at the far-end listener side.

**Function:**

One parameter to adjust the configuration of the comfort noise is as shown in Figure 7:

- **<u>Background Noise:</u>**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 |
|---|---|
| 0x306 (MSB) /0x307(LSB) | 0x7FFF: 0dBcHighest Comfort Noise level<br>0x4000: 6dBcComfort Noise level<br>0x2000: 12dBcComfort Noise level<br>0x1000: 18dBcComfort Noise level<br>0x0800: 24dBcComfort Noise level<br>0x0400: 30dBcComfort Noise level<br>0x0200: 36dBcComfort Noise level<br>0x0100: 42dBcComfort Noise level<br>0x0080: 48dBcComfort Noise level<br>0x0040: 54dBcComfort Noise level<br>0x0020: 60dBcComfort Noise level0x0010: 66dBcComfort Noise level<br>0x0008: 72dBcComfort Noise level<br>**0x0004: 78dBcComfort Noise level (Recommended)**<br>0x00002: 84dBcComfort Noise level<br>0x00001: No Comfort Noise level |

## 5.6 EQ

The 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 and speaker paths.

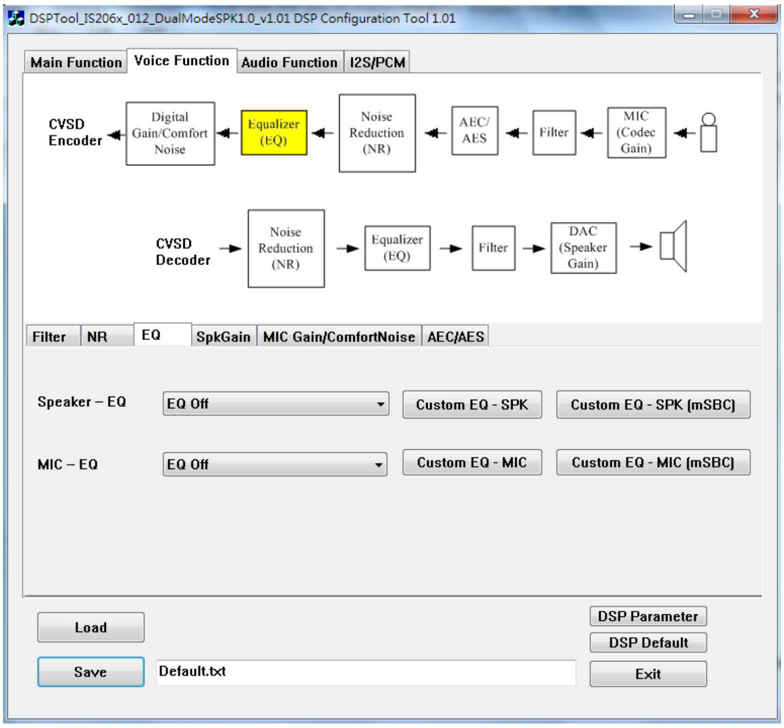


**Figure 8: EQconfiguration interface for both speaker and MIC paths in SCO(or Speech) mode.**

Figure 8 shows the interface to configure equalizers for both speaker and microphone paths as the buttons of“**Custom EQ –MIC**”, and“**Custom EQ–SPK**”. Note that Custom EQ ones with mSBC is configured for the Hands-free profile (HFP) 1.6 supporting the wideband (16khz sampling rate) speech features. One need to configure the EQ coefficients for 8khz and 16khz separately if desiring to support HFP 1.6. By clicking buttons of “**Custom EQ – MIC**” or “**Custom EQ–SPK**”,the EQ configuration window shown in Figure 9 is popped out.

In Figure 10, one can type in the desired frequencies and the gain/attenuations. The “**Q**” (Quality factor) columns are to configure the cutoff frequencies of each equalizer band. An example is shown in Figure 10, the smaller the value of “**Q**” is, and 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 one wants to save 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 functions.**

| | Bit3 0x1EC | Bit6 of 0x1DF | Bit5 of 0x1DF | 0x356/ 0x358(mSBC) | 0x357/ 0x359(mSBC) | Bit 2 of 0x1E1 | Bit 6 of 0x1E1 |
|---|---|---|---|---|---|---|---|
| MIC path | 1 | 1 | No need | 0x0B/0x0D | No need | 1 | No need |
| Speaker path | 1 | No need | 1 | No need | 0x0C/0x0E | No need | 1 |

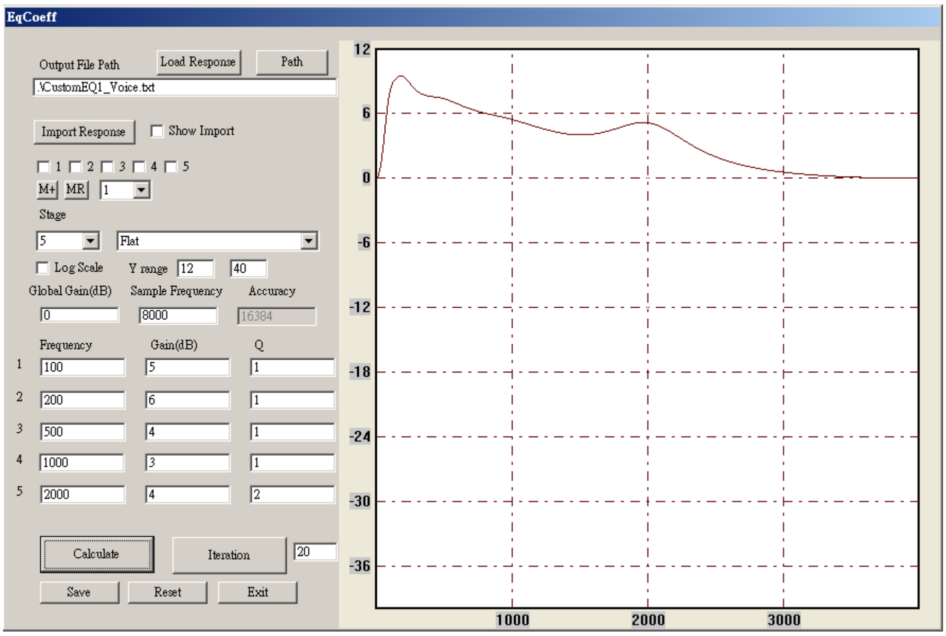


**Figure 9: An example of configuring the EQ functions.**

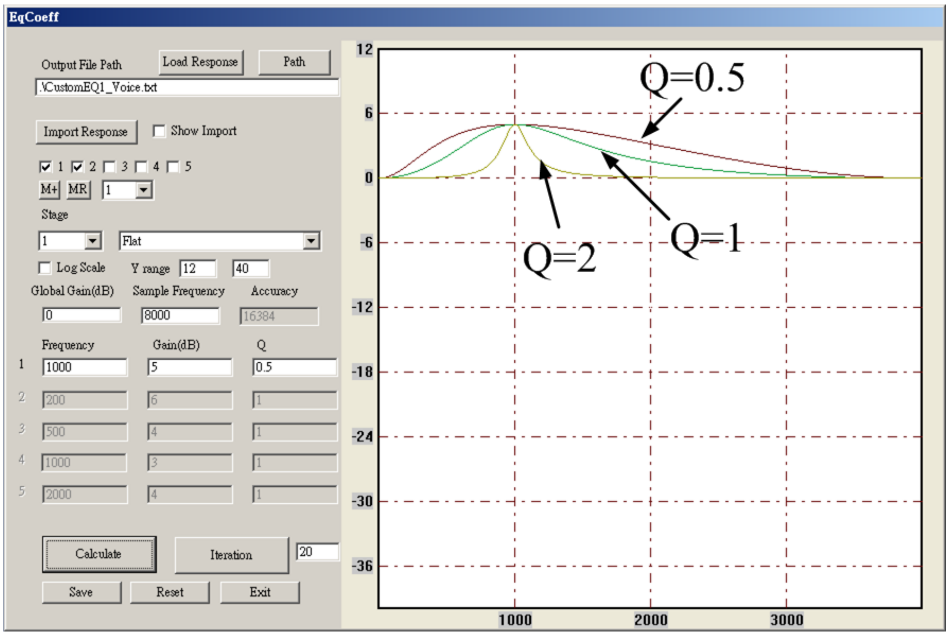


**Figure 10: Illustrating the function of the Q factor.**

## 5.7 Speaker/MIC Gain Settings:

The number of speaker gain levels and MIC gain levels are configured in the DSP configuration tool. Figure 11 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.

In Figure 12, gain difference between each MIC gain level for “**MIC Gain (Codec)**”is ranging 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 |

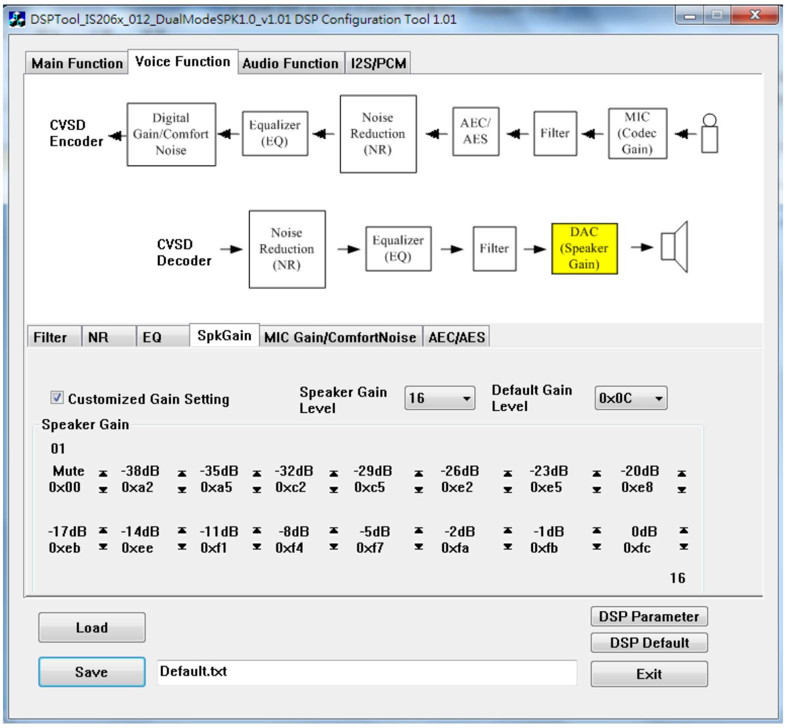


**Figure 11: Interface for the speaker gain configuration in the SCO mode.**

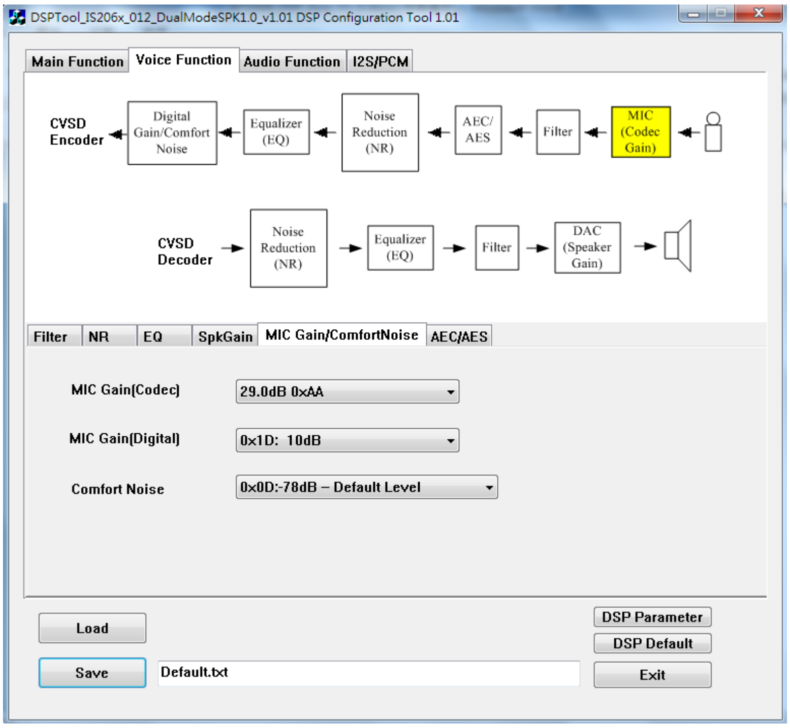


**Figure 12: MIC gain configuration in the SCO mode.**

# 6 Audio Processing Functions

## 6.1 EQ

The system diagram of the audio signal processing is shown in Figure 1(b). In addition to the SBC decoder, only one EQ are allowed to process to audio signal. Figure 13 shows the configuration of the 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 EQ introduced in section 5.6.

**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 EQ in the audio(or SBC) mode.**

| Addr | Bit3 0x1EC | Bit5 of 0x1E0 | 0x1E7 | 0x1E8 | 0x1E9 |
|---|---|---|---|---|---|
| Values | 1 | 1 | 0x0A | 0x07 | 0xFF |

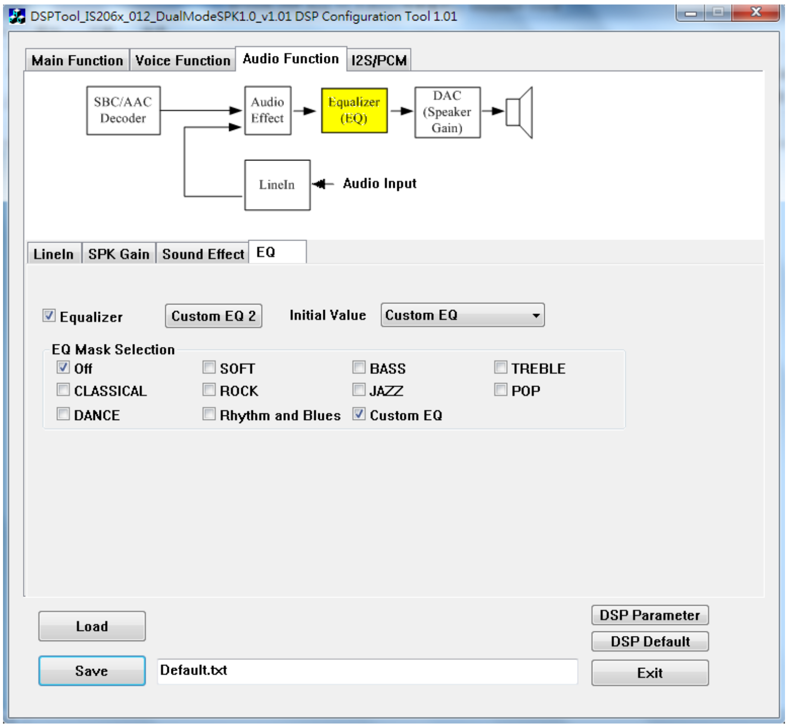


**Figure 13: EQ configuration interface for audio mode.**

## 6.2 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 5.7.

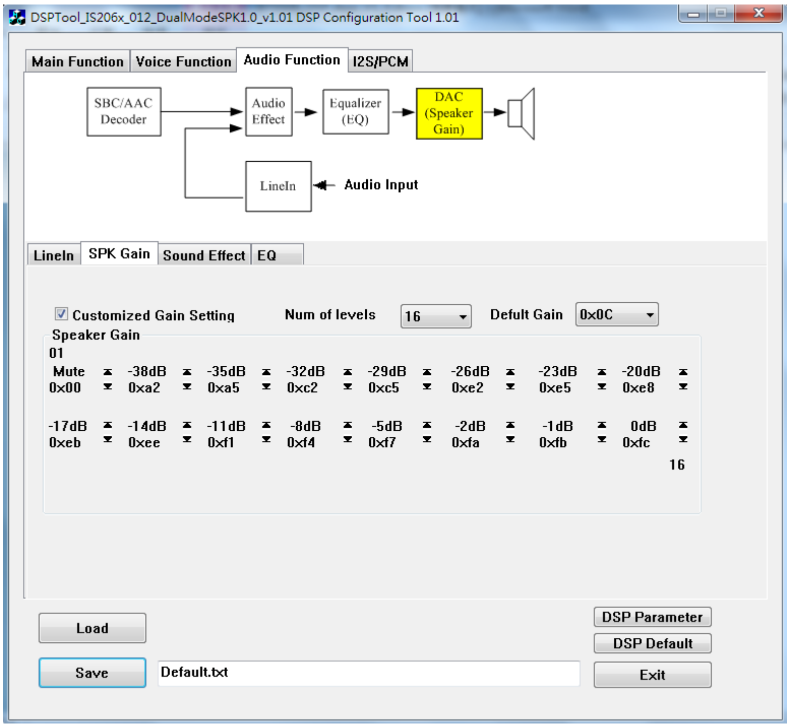


**Figure 14: Configuration interface for speaker gains in the audio mode.**

## 6.3 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 digitallyand,then,the silence status is reported to the BluetoothMAC 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.

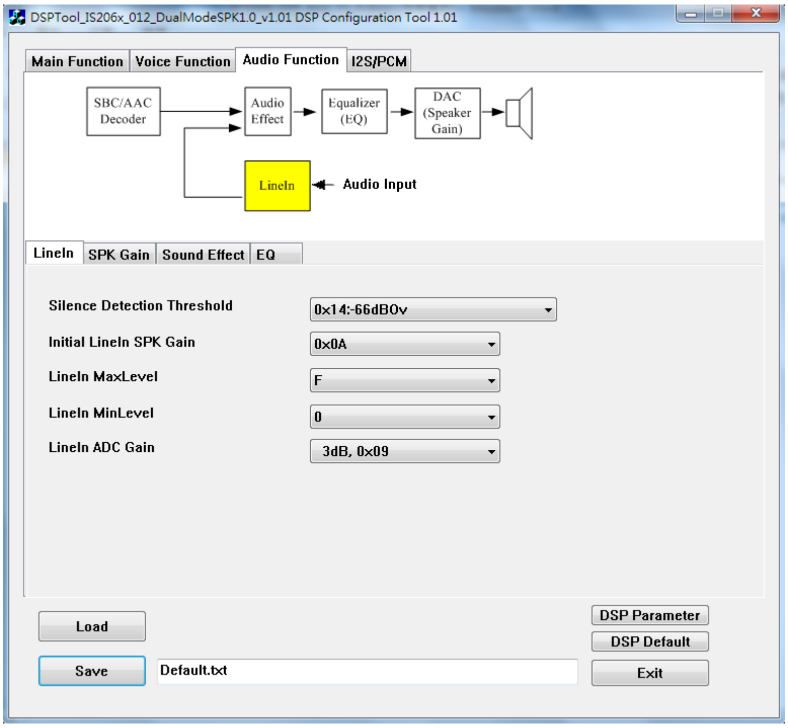


**Figure 15: Line-In(or Aux-In)configuration interface.**

## 6.4 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 processesaudio signal bymanipulating thesignal 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 functionis 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 compressioninvolves 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 15 shows adjustable parameters for AW and MB-DRC. Checkboxes for MB-DRC and AW should be checked if one wants to enable these functions.

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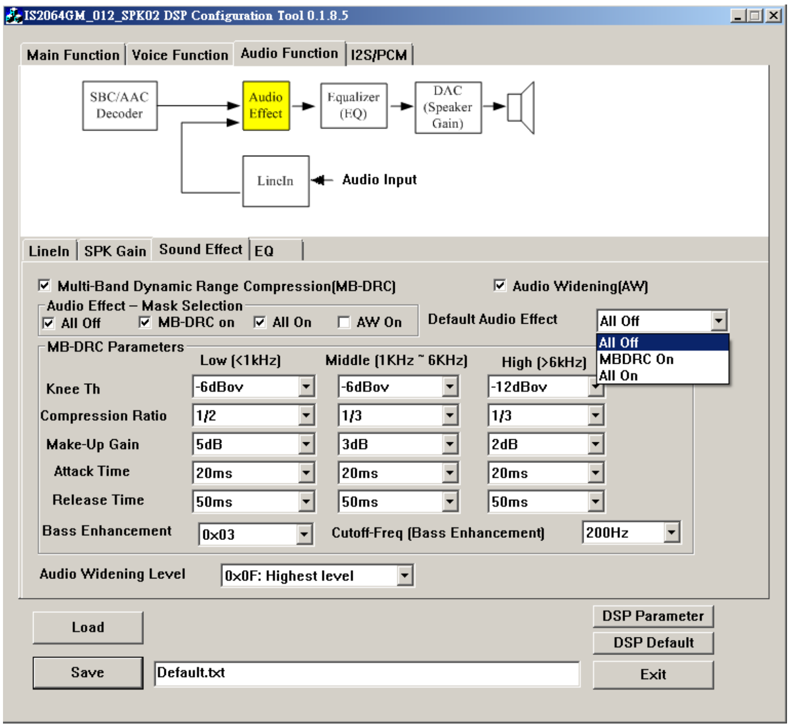


**Figure 16: User interface for sound effect configurations.**

**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. In the case given in Figure 16, “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. Threeparameters 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.

**Attack Time and Release Time:**

The attack and release time parameters are the period of time to hit 1-1/e = 63% of its final value. The difference between attack and release time constants can be illustrated in Figure 17. The attack time is the time period to decrease the signal amplitude exceeding the level of "**Knee TH**" while the release time is the time period to increase the signal amplitude below the level of "**Knee TH**" to its desired "**Make-Up Gain**" level.

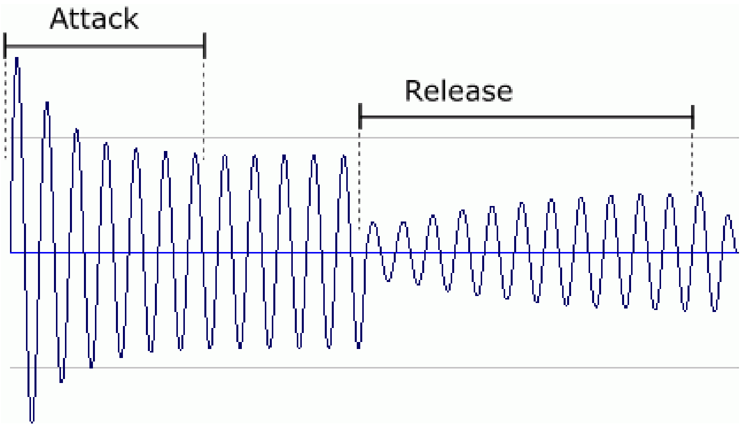


**Figure 17: Illustration of attack and release time parameters.**

**Bass Enhancement:**

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

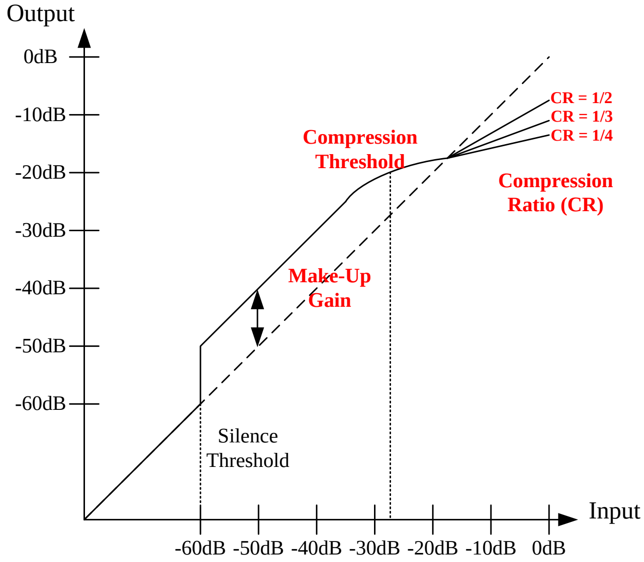


**Figure 18: Mapping function of MB-DRC.**

**AW Parameters:**

As illustrated in Figure 19, AW sound effect is to process the music signalplayed out by speakers placed close apart or inside one housing ID to make it sound like being placed far apart from each other.As shown in Figure 20, virtually generated widening signal are mixed with original signal with various levels of weighting pairs.

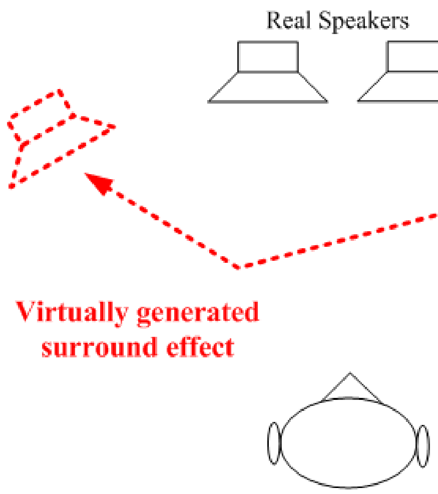


**Figure 19: An illustration of audio widening effect.**

## 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.

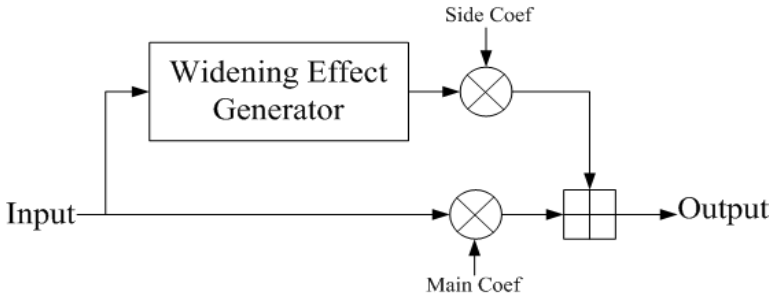


**Figure 20: Block Diagram of AW generator.**

**Table 15: ParametersofAW effect.**

| EEPROM Address | Values |
|---|---|
| 0x353<br>Bit 7~4: SideCoef<br>Bit 3~0: Main Coef | 0x00: SideCoef = 0; Main Coef = 1<br>0x01: SideCoef = 1/16; Main Coef = 1<br>0x02: SideCoef = 2/16; Main Coef = 1<br>0x03: SideCoef = 3/16; Main Coef = 14/16 |

| | |
|---|---|
| | 0x04: SideCoef = 4/16;Main Coef = 14/16<br>0x05: SideCoef = 5/16;Main Coef = 14/16<br>0x06: SideCoef = 6/16;Main Coef = 13/16<br>0x07: SideCoef = 7/16;Main Coef = 13/16<br>0x08: SideCoef = 8/16;Main Coef = 13/16<br>0x09: SideCoef = 9/16;Main Coef = 12/16<br>0x0A: SideCoef = 10/16;Main Coef = 12/16<br>0x0B: SideCoef = 11/16;Main Coef = 12/16<br>0x0C: SideCoef = 12/16;Main Coef = 11/16<br>0x0D: SideCoef = 13/16;Main Coef = 11/16<br>0x0E: SideCoef = 14/16;Main Coef = 10/16<br>0x0F: SideCoef = 15/16;Main Coef = 9/16 |

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# 7 I2S Digital Output/Input Interface

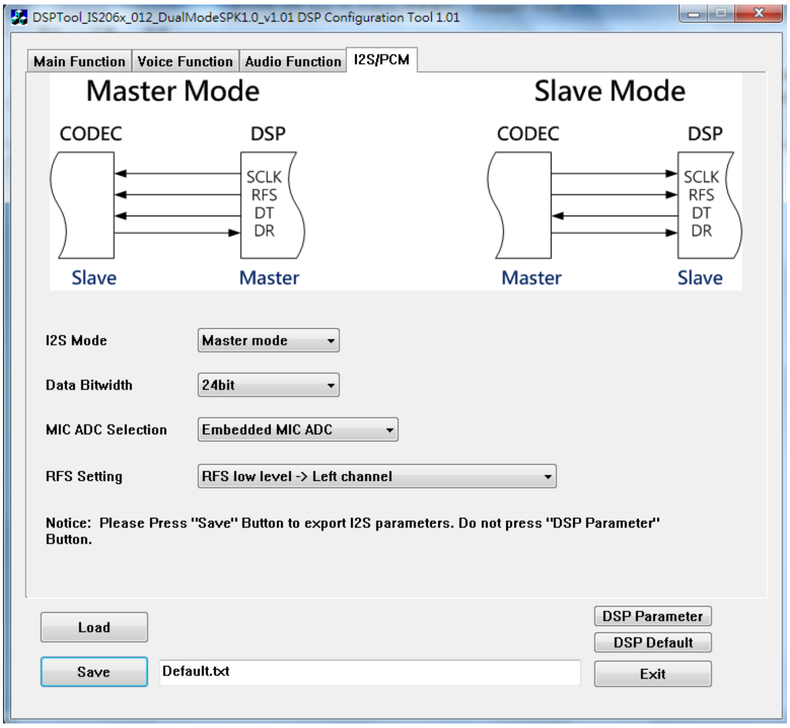


**Figure 21: I2S parameter tuning interface for IS2023 chip.**

IS2023-002 chip or BM23-002bluetooth module can support the I2S interface for digital input/output. In this document, the hardware wiringissue 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 21.

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 23(a).
- **Slave:**IS2023 chip serves as a slave to receive clock and frame sync signals from external codec or DSP devices illustrated in Figure 23(b).
- 

**Data Bitwidth:**

The numbers of bits for DR/DT are expected to receive from or transmit to external codec or DSPs as shown in Figure 22.

- **16bit:**BT supported codec can only have 16-bit resolution.
- **24bit:**Trailing zeros in LSBs is the only way to meet the external DSP that supported 24-bit I2S port

requirement.In this way, the 16bits occupy 16 MSBs and 8 zeros are filled in for LSBs.

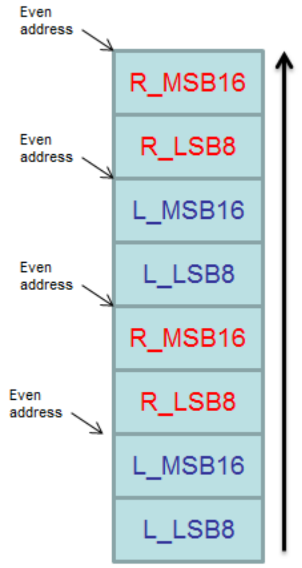


**Figure 22: I2S configurations for 16-bit/24-bit.**

**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.

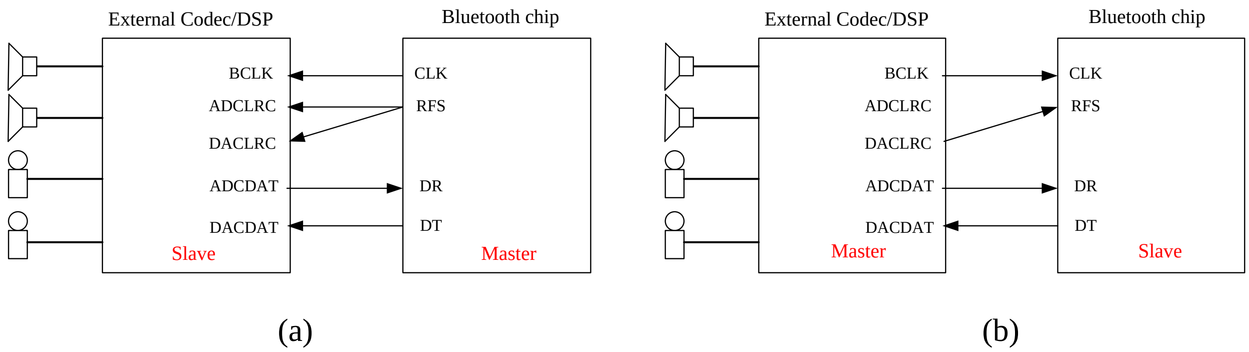


**Figure 23: I2S hardware configurations for (a) Master, and (b) Slave modes with external ADC.**

**RFSSetting:**

This setting determines that either high or low level of the RFS signal represents L channel data for both

- **RFS low level -> Left channel:** please refer to the timing diagram in Figure 24(a).
- **RFS highlevel -> Left channel:** please refer to the timing diagram in Figure 24(b).

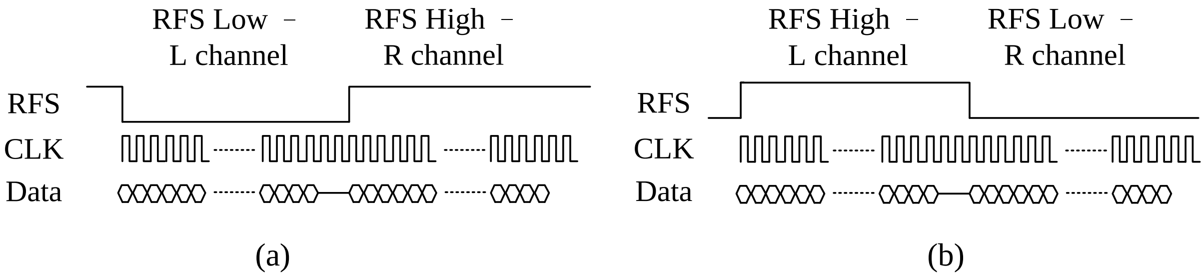


**Figure 24: Low and high levels of RFS denote (a) left channel and (b) right channel.**

# 8 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 25, step 1 ~ step 5 are basically handling these whole speakerphone/carkit echo issues.

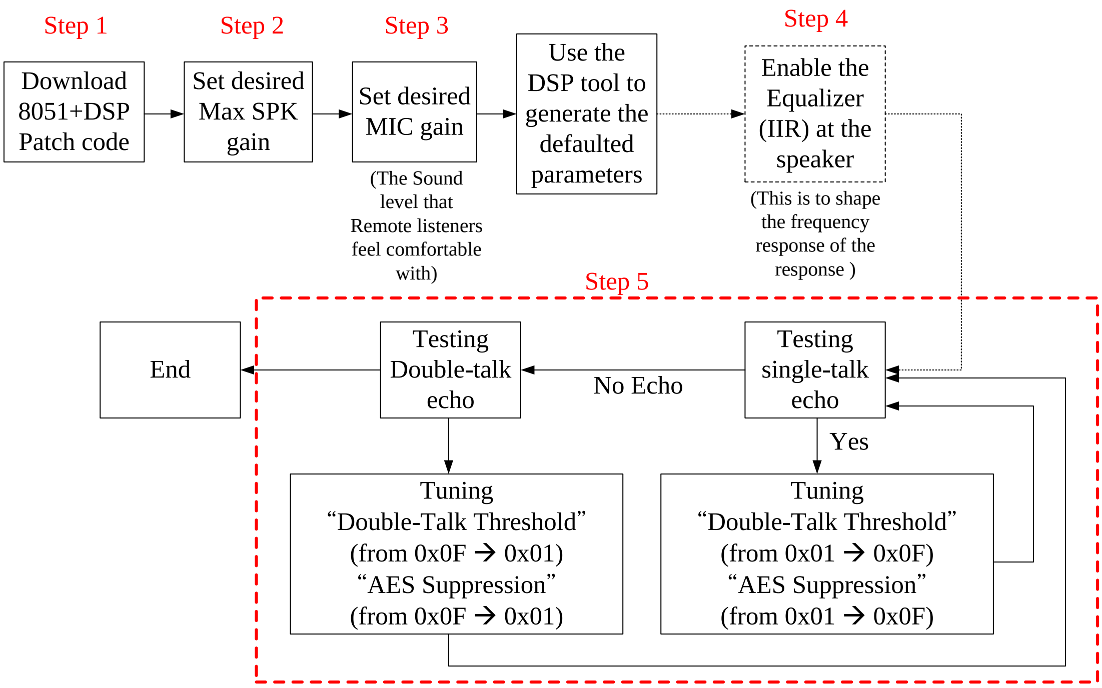


**Figure 25: An illustration of the AEC tuning flow.**

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 26, 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 )

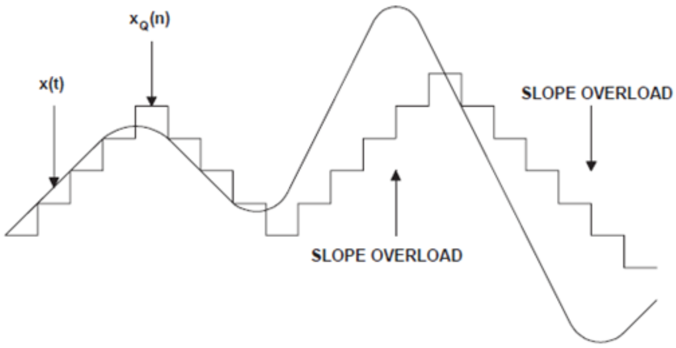


**Figure 26: Illustration of the CVSD slope overload effect, where x(t) and $x_Q(n)$ are denoted as the original signal and CVSD encoded/decoded signal, respectively.**

**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 27; these settings can is obtained empirically. However, the required frequency shaping may vary in terms of what the speaker and the housing is selected.

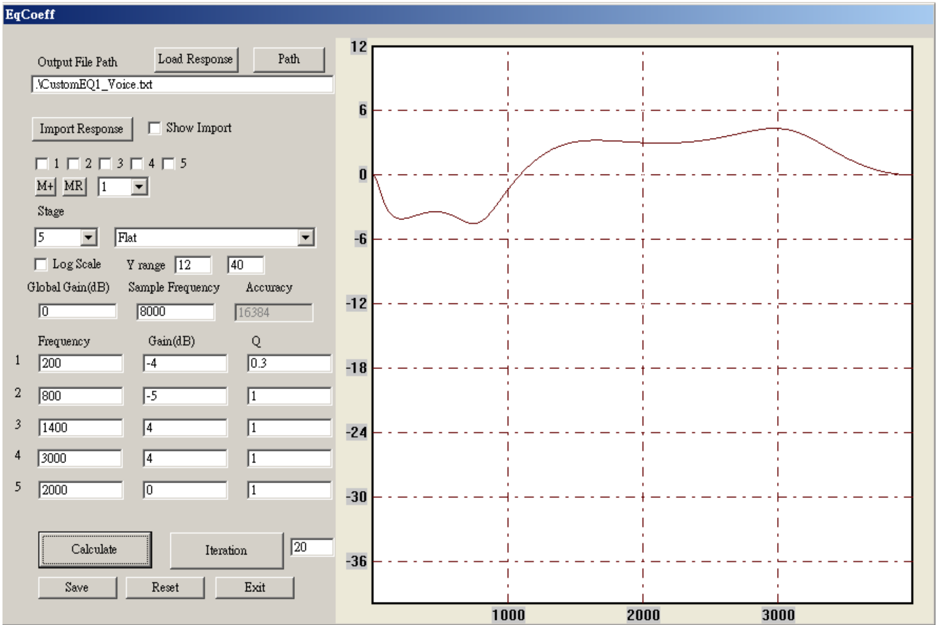


**Figure 27: An example of the frequency shaping for signal at the speaker path.**

<u>**Step 5:**</u> 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.

- <u>**Single-talk echo**</u>: "**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 0x7F to 0x1C. 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 0x1C, these two values are not recommended to suppress the echo since it would distort the MIC speech severely.

- <u>**Double-talk echo**</u>: 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 double-talk 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 27*: 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 channeloutput(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.

# 9 EEPROM File Format

The DSP configuration tool is to generate the EEPROM-formatted file, as illustrated in Figure 28, based on configurable parameters on the GUI.

The header information defines the IC version, DSP configuration tool version, date and checksum of the content. As being brought up in Figure 2, the button “DSP Parameter” only produces the EEPROM content whose address is below 0x400. The full EEPROM table needs to be generated via the button “Save”. Note that, the content stored beyond 0x800, shown in Figure 29, includes the DSP patch code, I2S coefficients, and partial EQ coefficients.

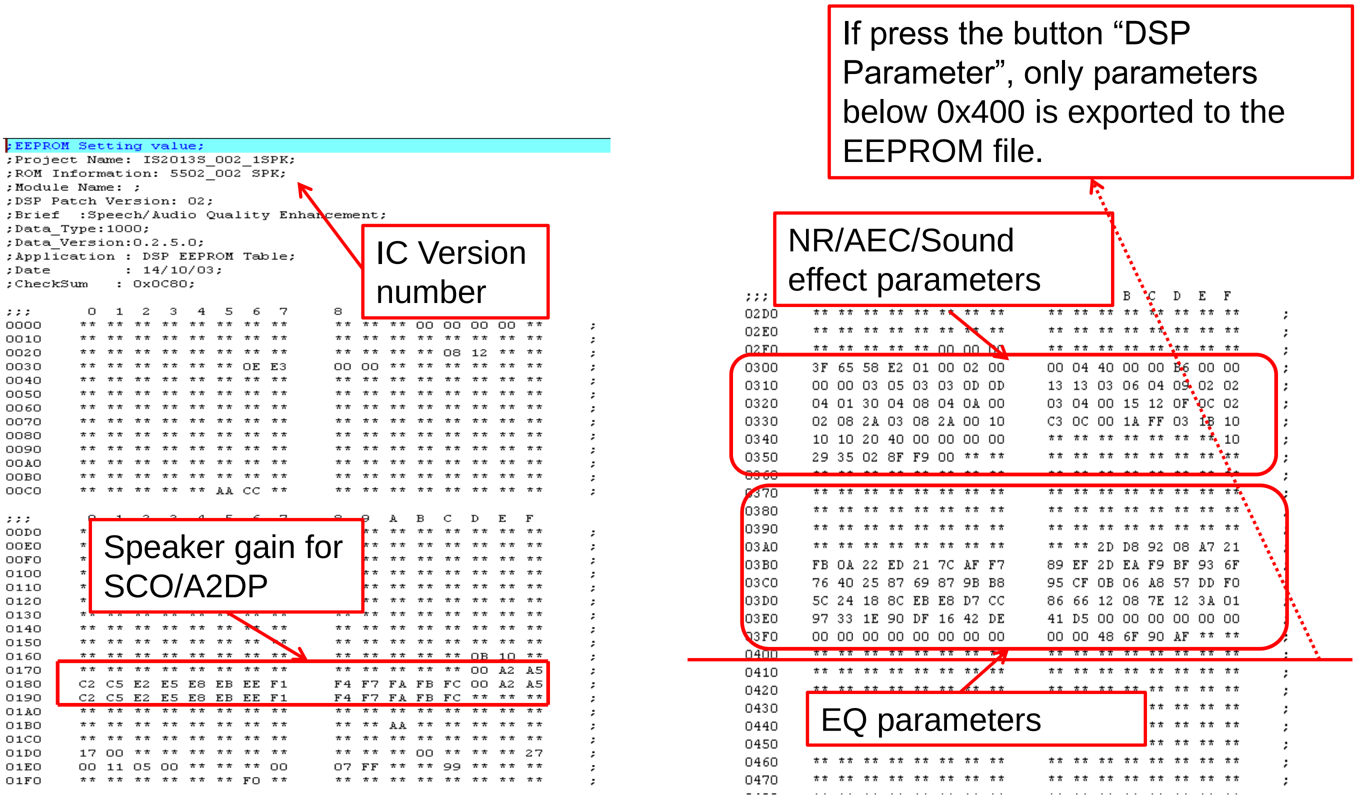


Figure 28: EEPROM format and its definitions.

```
07A0      ** ** ** ** ** ** ** **      ** ** ** ** ** ** ** **      ;
07B0      ** ** ** ** ** ** ** **      ** ** ** ** ** ** ** **      ;
07C0      ** ** ** ** ** ** ** **      ** ** ** ** ** ** ** **      ;
07D0      ** ** ** ** ** ** ** **      ** ** ** ** ** ** ** **      ;
07E0      ** ** ** ** ** ** ** **      ** ** ** ** ** ** ** **      ;
07F0      ** ** ** ** ** ** ** **      ** ** ** ** ** ** ** **      ;
0800      01 00 00 04 07 A0 8A 1D      D0 00 06 CD 00 02 05 00      ;
0810      06 C7 AA 55 08 8E 06 4B      08 E6 6B E8 55 E9 1C 29      ;
0820      14 8C 0C 28 FE 86 90 28      5C E0 C7 28 60 81 C8 28      ;
0830      F3 E9 FC 68 EE 8C 2C 28      00 00 00 00 00 00 00 00      ;
0840      00 00 00 00 00 00 00 00      00 00 00 00 00 00 00 00      ;
0850      00 00 00 00 00 00 00 00      00 00 00 00 00 00 00 00      ;
0860      00 00 00 00 00 00 00 00      00 00 00 00 00 00 00 00      ;
0870      00 00 00 00 00 00 00 00      00 00 00 00 00 00 00 00      ;
0880      00 00 00 00 00 00 00 00      00 00 00 00 00 00 1A 80      ;
0890      1F 83 FD AA 44 00 04 2B      82 AA 1A 80 B0 40 50 00      ;
08A0      90 F1 50 40 06 D0 90 F1      60 47 FF F0 90 F1 70 80      ;
08B0      F1 60 2A 78 AA 1A 82 40      23 9A BF 93 FD AA 0C 0C      ;
08C0      00 22 E0 1F 90 F1 6A 80      F1 54 80 F1 70 22 E0 0F      ;
08D0      90 F1 7A 0C 08 00 87 26      C0 87 26 A1 0C 20 00 3C      ;
08E0      04 05 16 82 2E 60 00 62      20 82 0F 68 00 C1 60 00      ;
08F0      66 20 82 0F 68 00 C5 0C      30 00 81 55 C0 4C 00 04      ;
0900      23 80 0F 1A 85 20 95 39      66 2A E2 AA 1A 84 C0 23      ;
0910      98 DF 1A 83 A0 01 03 FB      00 00 00 00 00 00 01 03      ;
0920      FB 01 03 EC 00 00 00 00      00 00 01 03 EC 23 83 3F      ;
0930      01 83 FA 00 00 00 23 AC      DF 01 83 EA 87 26 A1 40      ;
0940      01 8A 01 84 0A 00 00 00      00 00 00 01 84 3A 3C 04      ;
0950      05 26 18 0F 16 84 6E 60      00 A5 23 FA 0F 2A F2 AA      ;
0960      26 7A 02 42 00 00 23 30      0F 1A 85 23 4C 00 04 91      ;
0970      55 C4 87 26 C0 87 26 A1      3C 04 05 16 85 1E AF FF      ;
0980      F1 AF FF F5 83 23 F7 38      C2 D2 18 86 4F 1A 85 6F      ;
0990      0F 13 03 0F 2C 03 0D 00      BE 0D 00 CF 3A 66 D2 18      ;
09A0      86 CF 1A 85 DF 26 78 0F      38 69 12 18 86 4F 1A 86      ;
09B0      1F 82 FC 5A 40 00 74 2A      E2 AA 1A 86 C1 97 28 E6      ;
09C0      97 2D B6 97 2D B6 97 2E      06 97 2C 96 97 2C 96 97      ;
09D0      2C E6 38 00 17 39 98 A0      43 4A 7A 58 00 A3 49 8C      ;
09E0      DA 58 00 A3 43 29 1A 58      00 A3 47 28 BA 58 00 A3      ;
09F0      4C A2 4A 58 00 A3 4D 8F      0A 58 00 A3 0C 30 00 3B      ;
0A00      FF C7 40 00 14 84 B0 F1      84 B0 F3 39 98 A0 34 92      ;
0A10      00 09 00 01 3C 04 05 1E      8C CF 40 00 14 84 B1 01      ;
0A20      84 B1 03 39 98 A0 34 93      00 09 00 01 3C 04 05 1E      ;
0A30      8C CF 34 00 08 38 00 06      3B FF F7 0C 08 00 0C 20      ;
0A40      00 84 B0 F0 84 B0 F1 84      B1 32 84 B1 53 80 B0 8A      ;
```

The placing order is DSP patch → I2S parameters → EQ coefficients

**Figure 29: EEPROM content beyond 0x800.**