IS2083/BM83 Echo Tuning
AN5220

Introduction

This application note provides general information of the IS2083SoC/BM83 Module, covering noise reduction with high performance and Acoustic Echo Cancellation (AEC) with advanced signal processing techniques, sophisticated voice and audio enhancement functions, filtering, equalization and audio effect processing. In addition, it offers fine tuning the AEC and Noise Reduction (NR) parameters for different applications during product development. The Digital Signal Processor (DSP) configuration tool (IS208x DSP GUI configuration tool) provides step-by-step guidance for setting the DSP features according to desired requirements of the system designers.

The IS2083SoC/BM83 Module is designed for high-performance signal processing, providing an excellent user experience for voice or audio. The fundamental modules of the IS2083SoC/BM83 Module include the following:

- Filter
- Stationary NR
- Spk gain
- MIC gain
- AEC
- I^{2}S
- AES (optional)
- Sound effect (optional)

Note: The user can adjust the optional modules as per the customer requirements.

The AEC function is available during the Synchronous Connection-Oriented (SCO) link connection.

Features

- Supports AEC to Cancel the Acoustic Echo Coupled into the Microphone from the Loud Speaker Output
- Supports NR to Suppress the Stationary Ambient Noise to Enhance the Voice Signal Quality
- Supports Equalizer (EQ) to Provide the 5-Band EQs for Both Voice and Audio Applications to Compensate the Imperfect Frequency Response of the Adopted Microphone or Speaker. Default Audio Effects are Also Provided to Enhance the User Experience.
- Supports High Pass Filter (HPF) to Provide a Low-Latency Infinite Impulse Response (IIR) Structured Low-Pass Filter to Filter the Unwanted Low-Frequency Band for Both MIC and SPK Paths

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1. Quick References

1.1 Acronyms and Abbreviations

Table 1-1. Acronyms and Abbreviations

Acronyms and Abbreviations	Description
ADC	Analog-to-Digital Codec
AEC	Acoustic Echo Cancellation
AES	Advanced Encryption Standard
CVSD	Continuous Variable Slope Delta
DAC	Digital-to-Analog Codec
DSP	Digital Signal Processor
EC	Echo Cancellation
EQ	Equalizer
ES	Echo Suppression
GUI	Graphic User Interface
HD	High Definition
HFP	Hands-Free Profile
HPF	High-Pass Filter
IIR	Infinite Impulse Response
MAC	Medium Access Control
MEMS	Micro-Electro Mechanical Systems
MHS	Mono-Headset
MIPS	Mega Instructions Per Second
1SPK	Mono speakerphone
mSBC	Modified Sub-Band Coding
NR	Noise Reduction
SHS	Stereo-Headset
SCO	Synchronous Connection-Oriented
2SPK	Stereo Speakerphone
SPL	Sound Pressure Level
SNR	Signal-to-Noise Ratio
VAD	Voice Activity Detection

2. Signal Processing Flow

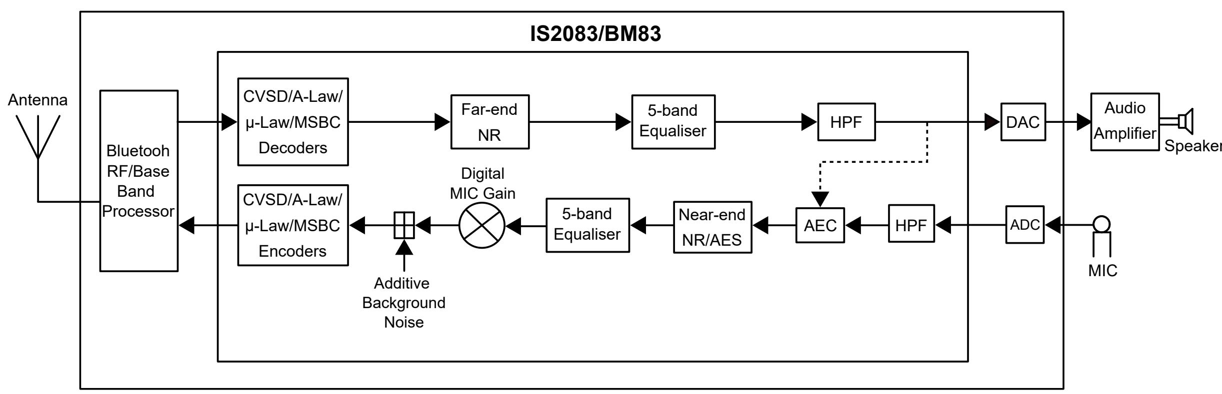
Before introducing the signal processing flow, the user must understand the following terminologies:

- Downlink/Downstream/Far-end/Speaker path – Bluetooth® device receives bitstream and pass to DSP for decoding process
- Uplink/Upstream/Near-end/MIC path – Bluetooth device transmits the bitstream, encoded by the DSP processor

The following figure illustrates the block diagram of the DSP processing flow for speakerphone and headset applications for speech and audio signal processing. The DSP part mainly focuses on speech encoders/decoders and audio decoders along with their corresponding signal processing functions. The embedded Analog-to-Digital Codec (ADC) provides high Signal-to-Noise Ratio (SNR) data conversion with 88 dB, and the Digital-to-Analog Codec (DAC) provides high SNR data conversion with 98 dB. The Bluetooth and RF/modem processors deal with the Medium Access Control (MAC) and the wireless data transmission. For the speaker and speakerphone applications, use the external audio amplifiers to amplify the audio signal.

The speech codecs, Continuous Variable Slope Delta (CVSD) with supported bandwidth of 8 kHz and Modified Sub-Band Coding (mSBC) with supported bandwidth of 16 kHz are available for Bluetooth speech applications (see the following figure). mSBC is the mandatory speech codec defined in Bluetooth Hands-Free Profile (HFP) 1.6 to provide High Definition (HD) voice quality. As long as the host device establishes the Bluetooth link through the HFP 1.6 profile, mSBC is the speech codec regardless of the cellular network conditions (3G network or VoLTE).

Figure 2-1. Speech Signal Processing



3. IS208x DSP GUI Configuration Tool

The following figure illustrates the IS208x DSP GUI configuration tool window with various tabs:

- Main Function
- Voice Function
- Audio Function
- I^2S/PCM

This section provides the functionalities of the **Main Function** tab (see Figure 3-1). It also covers two configurable modes that are used for overall DSP settings. The user needs to select either of these modes:

- Voice Function mode – Allows the user to configure the NR/EC/EQ/FIR functions
- Audio Function mode – Related to audio (music playback)

The following are the buttons on the opening window:

- **Reset** button – To set the parameters to their default settings
- **Save** button – To save the current settings in the Graphic User Interface (GUI) tool and transform them to the EEPROM format

Note: The IS208x DSP GUI configuration tool supports the product portfolio for Mono-Headset (MHS), Stereo-Headset (SHS), Mono Speakerphone (1SPK), Stereo Speakerphone (2SPK) and I^2S interface applications.

Figure 3-1. IS208x DSP GUI Configuration Tool

4. Voice Processing Functions

The voice processing functions use the following features to enhance voice call quality by suppressing ambient noise, echo and other call interferences:

- High-Pass Filter (HPF) for SPK/MIC paths – Removes low-frequency board/background noise
- Noise Reduction (NR)
- Echo Cancellation (EC)/Echo Suppression (ES) – Linear/non-linear echo cancellations
- Comfort noise insertion for MIC paths – Insert comfort noise to maintain stable noise level
- 5-band EQs for SPK/MIC paths – Compensate frequency imperfection
- MIC and speaker gain controller

4.1 High-Pass Filter

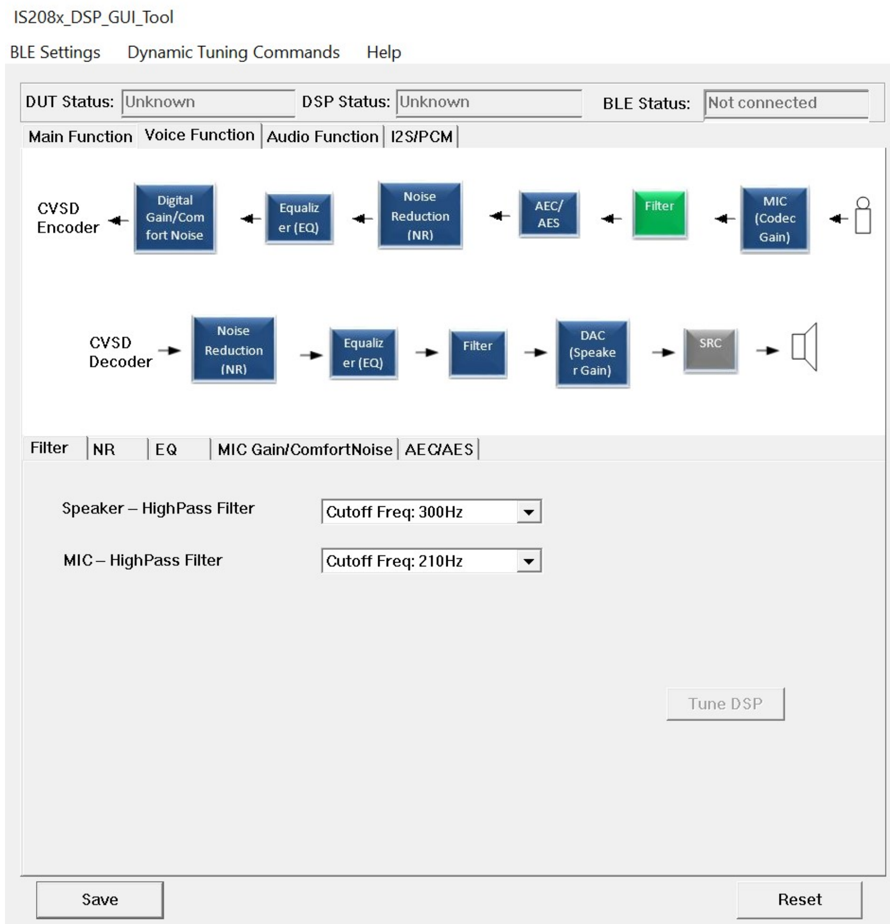
The HPF provides an option for a low-latency and IIR filtering.

The main function of the HPF contains seven selectable cutoff frequencies used to filter the unwanted low-frequency signals, such as PCB noise, coupled current noise, wind noise and more. It is a trade-off between the speech signal quality and the noise reduction level.

Note: The default cutoff frequencies for the speaker is 300 Hz and MIC is 210 Hz.

The following figure illustrates the interface of HPF parameters for both speaker and microphone paths in the DSP tool.

Figure 4-1. Configuration of High-Pass Filter Parameter in DSP Tool

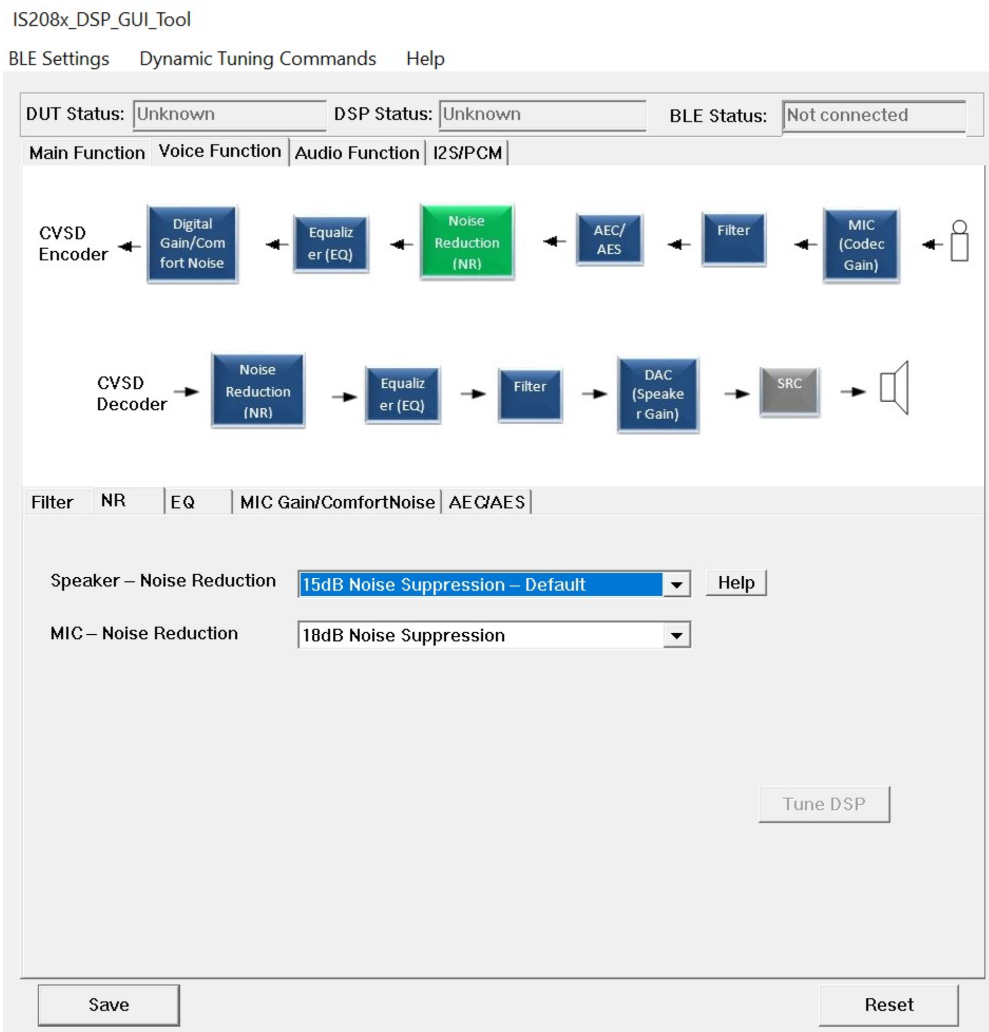


4.2 Noise Reduction

The NR function suppresses the stationary noises present in the far-end/downstream and near-end/upstream signals. The NR module can also effectively suppress the unwanted noise with the proprietary intelligent Voice Activity Detection (VAD), while maintaining satisfactory quality for the voice communications. This function allows both near-end and far-end users to experience the benefits.

The following figure illustrates the interface of the NR and dual MIC NR parameters in the DSP tool.

Figure 4-2. Interface of NR and Dual MIC NR in DSP Tool



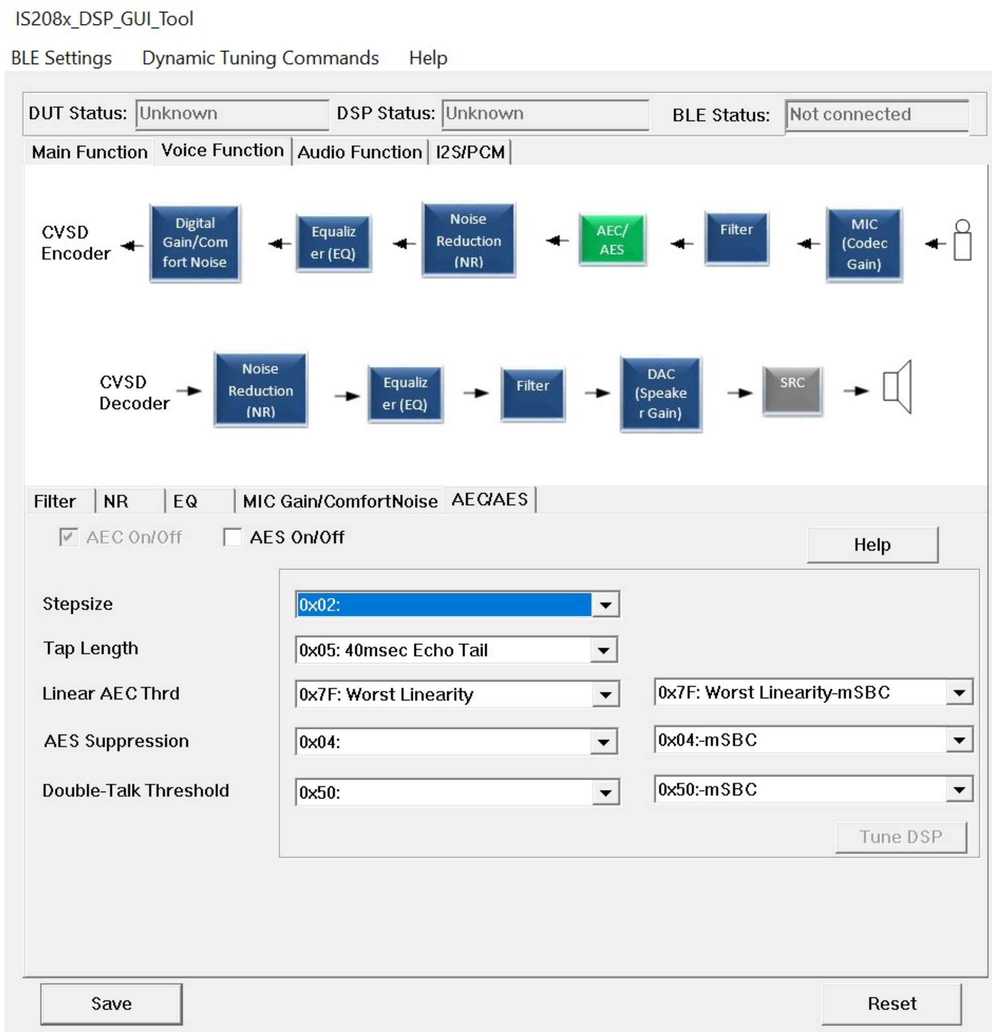
4.2.1 NR Suppression Level

Two selectable parameters for NR configurations are suppression levels for low frequency (<1000 Hz) and high frequency (from 1000-4000 Hz) (see Figure 4-2). The tunable range for both speaker and microphone paths are from 0-21 dB. However, in the DSP configuration tool, only the low-frequency NR suppression level is available, while the high-frequency suppression levels are determined empirically based on the field test results.

4.3 Echo Cancellation

To linearly cancel and suppress the returned echo, both Acoustic Echo Cancellation (AEC) and Acoustic Echo Suppression (AES) functions are required. The difference between AEC and AES is AEC can cancel the linearly coupled echo by maintaining the desired near-end speech. With the better AEC performance, the user can easily achieve full-duplex speech communication. But nonlinear echo generated by speakerphone housing, imperfect speaker/microphone devices and undesired echo environment are not canceled by AEC. Hence, the function of AES is to suppress the input signal mixed with the desired signal and unwanted echo signal in the frequency domain based on the information of the VAD. Therefore, the user must carefully select the parameters for AES and Double-Talk Threshold carefully to prevent from the degradation of the desired speech quality in the case of strong nonlinear echo or close placement between the microphone and speakers. The following figure illustrates the AEC tuning interface in the DSP configuration tool.

Figure 4-3. AEC Tuning Interface in DSP Configuration Tool



The following sections explain the adjustable parameters to fine tune the performance of echo cancellation.

4.3.1 AEC Stepsize

This is the AEC convergence speed. If selecting a fast convergence rate, the lower echo cancels out linearly. On the other side, the user can achieve better linear echo cancellation.

4.3.2 AEC Tap Length

The user must select the tap length that is longer than the echo tail, and also it is the trade-off between EC performance and Mega Instructions Per Second (MIPS). The longer the tap length, the higher the MIPS and DSP power consumption.

4.3.3 Double-Talk Threshold

The full-duplexity of the AEC means the level of how much the far-end user can listen to the near-end user voice, while both users speak simultaneously. In general, this parameter maps to a threshold that controls the AES to suppress the echo non-linearly. If this parameter is configured to be more favorable for half-duplexity, the double-talk capability gets degraded more but removes more residual echo.

If assuming that x[n] is near-end signal and y[n] is far-end signal, the cross-correlation equation between these signals is indicated as follows:

Echo_corr = EMA (corr_coef), where EMA is the exponentially moving average.

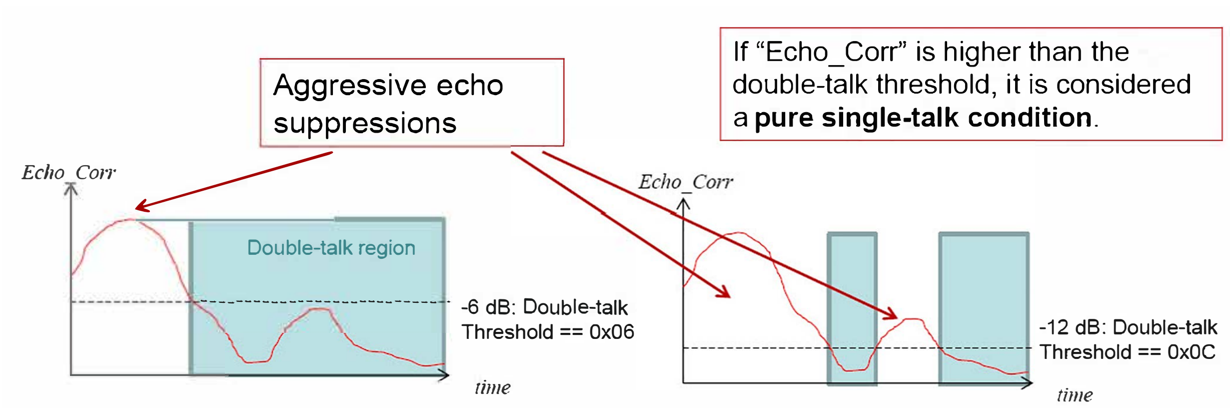
Equation 4-1. Cross-Correlation Equation

$$corr\_coef = \Sigma x[n]y[n] \div \left(\sqrt{\Sigma x[n]^2}\sqrt{\Sigma y[n]^2}\right)$$

If selecting the Double-Talk Threshold as 0x50, the threshold to determine the presence of echo is also high. As a result, the AES function is less likely to get enabled to suppress residual echo. The following figure illustrates the Double-Talk Threshold parameter.

If selecting the Double-Talk Threshold as 0x2D, this setting is more active to enable the AES function for echo suppression.

Figure 4-4. Double-Talk Threshold Parameter



4.3.4 AES Suppression

This parameter determines the maximum nonlinear echo suppression capability.

4.3.5 Linear AEC Threshold

This parameter determines the linearity threshold of the returned echo.

- If selecting the Worst Linearity parameter, the user must configure the AES function to easily enable and suppress the residual echo.
- If selecting Higher Linearity, a better full-duplex EC performance with more echo in the returned signal is allowed.

4.4 Digital MIC Gain

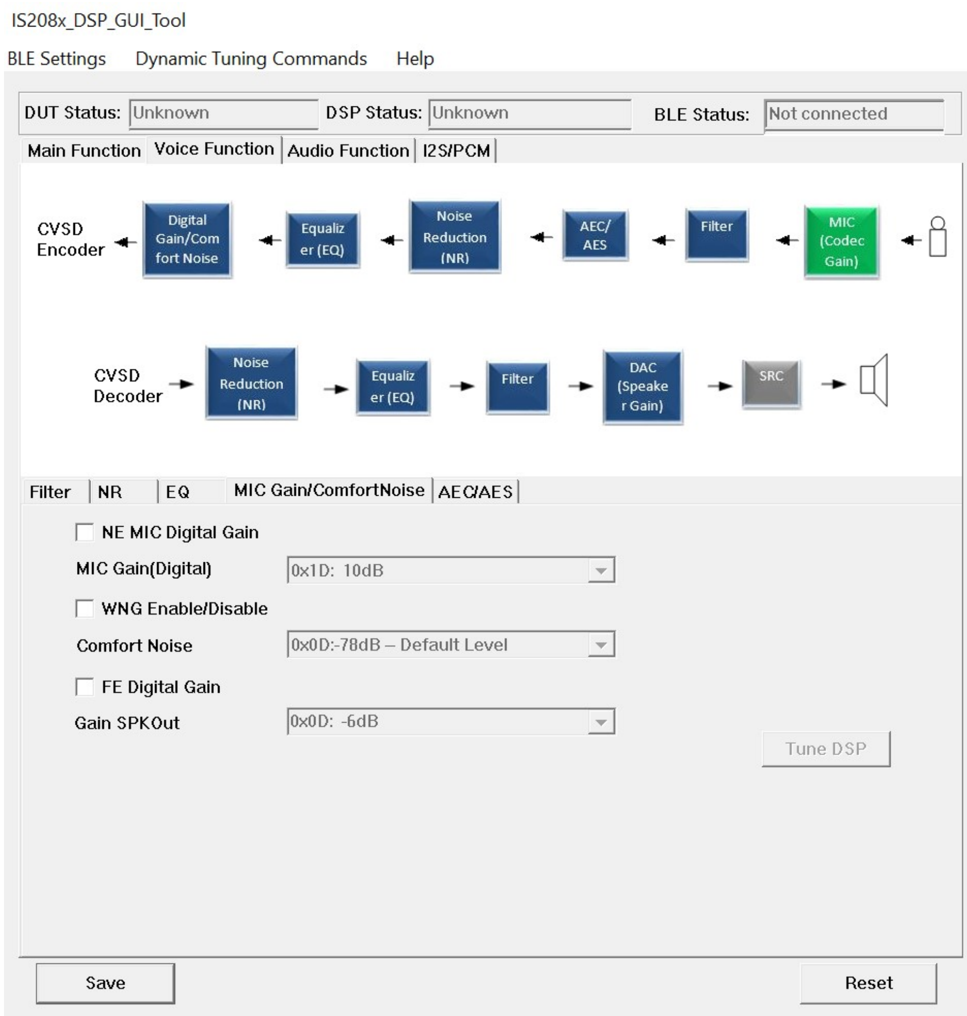
The digital MIC gain provides different digital control functions at the microphone path and allows the user to boost the volume digitally. If the analog amplifier of ADC is unable to provide enough gain, the digital boost part of the dynamic MIC control is placed at the end of all digital signal processing modules (see the following figure).

The user can amplify the suppressed echo by adjusting the digital MIC gain.

Note: The DSP configuration tool automatically updates the available selections of the dynamic range according to the MIC gain (codec).

The following figure illustrates the parameters to adjust the configuration of the digital MIC gain and comfort noise.

Figure 4-5. Parameters for MIC Gain and Comfort Noise



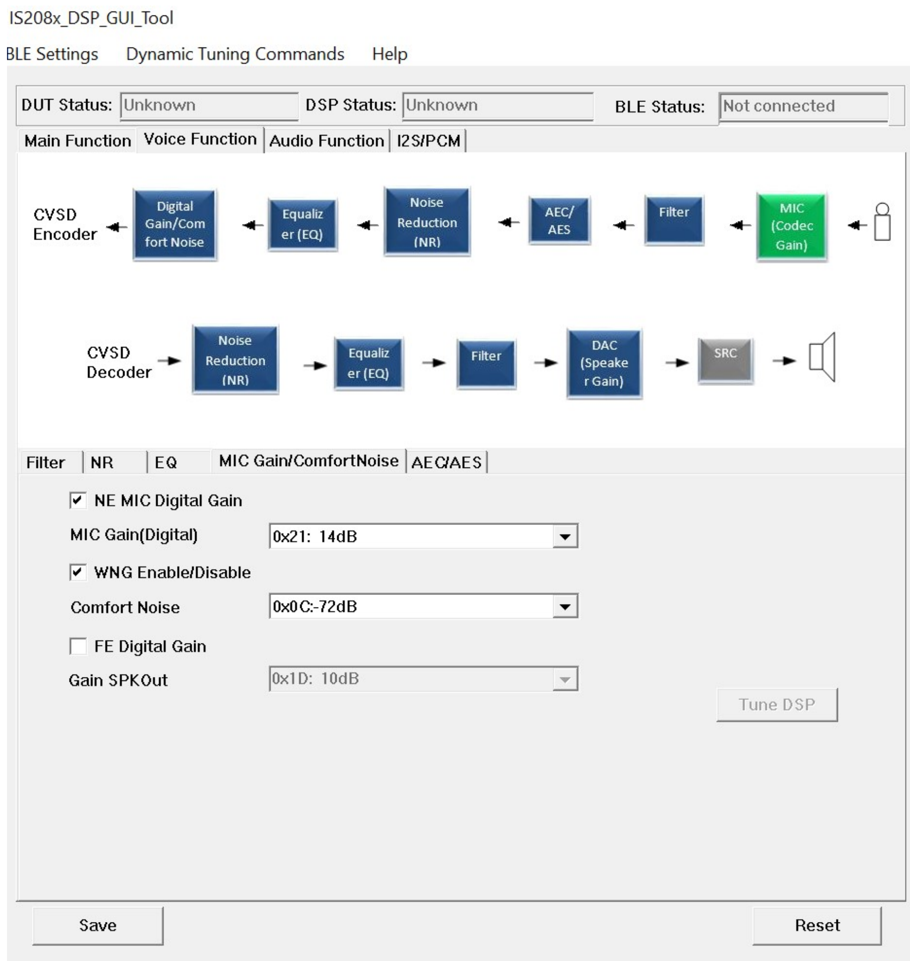
4.5 Comfort Noise

A random number generator generates the comfort noise, and its frequency response is constant across all frequencies. The main purpose is to provide a constant noise level to prevent the speech codec algorithm of host cell phones from injecting unwanted noise, which degrades speech clarity at the far-end listener side.

Note: Background noise adjusts the level of the comfort noise.

The following figure illustrates the parameters to adjust the configuration of the digital MIC gain and comfort noise.

Figure 4-6. Parameters for MIC Gain and Comfort Noise



4.6 Equalizer

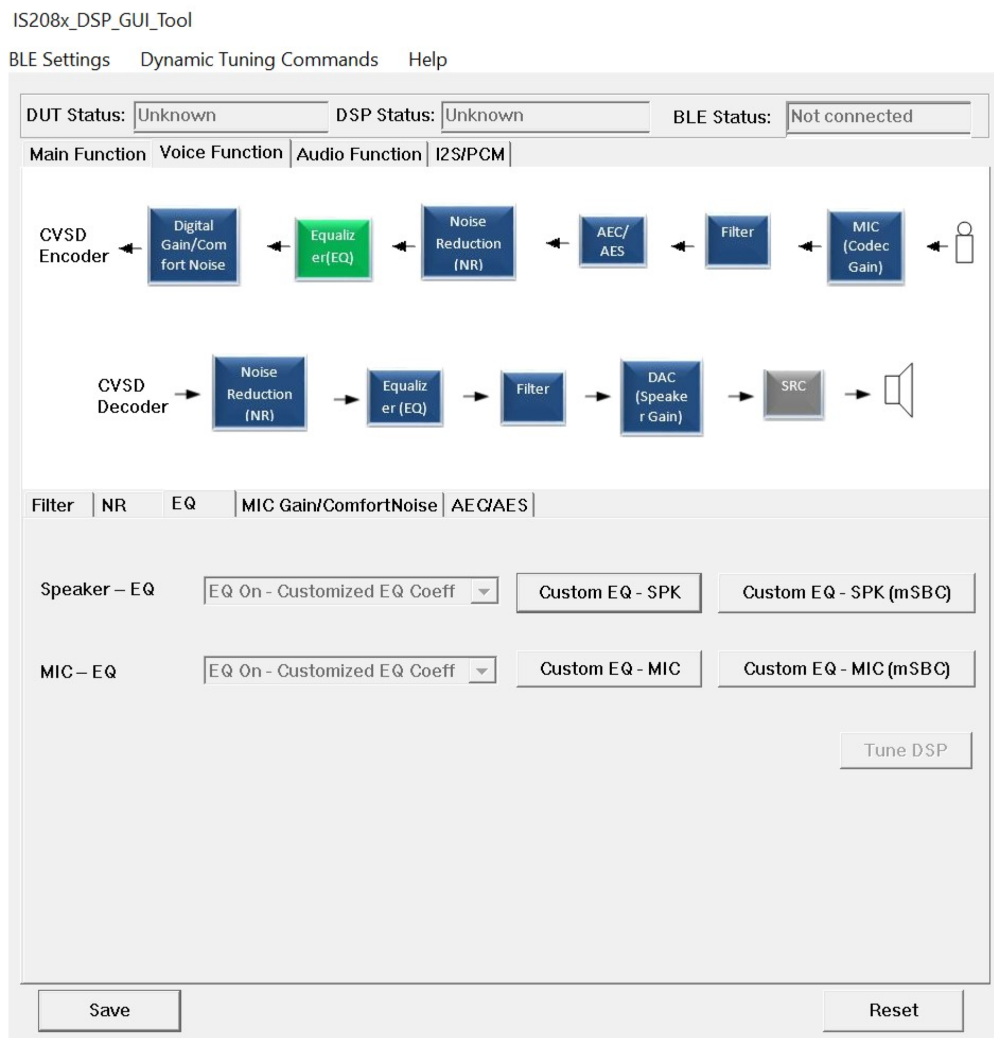
The EQ provides the flexibility for compensating the imperfect frequency responses of the selected microphone and speaker. This function provides a 5-band customized filter for both MIC and speaker paths.

The following figure illustrates the interface to configure equalizers for both speaker and microphone paths using buttons.

- Custom EQ – MIC (for Microphone path)
- Custom EQ – SPK (for Speaker path)

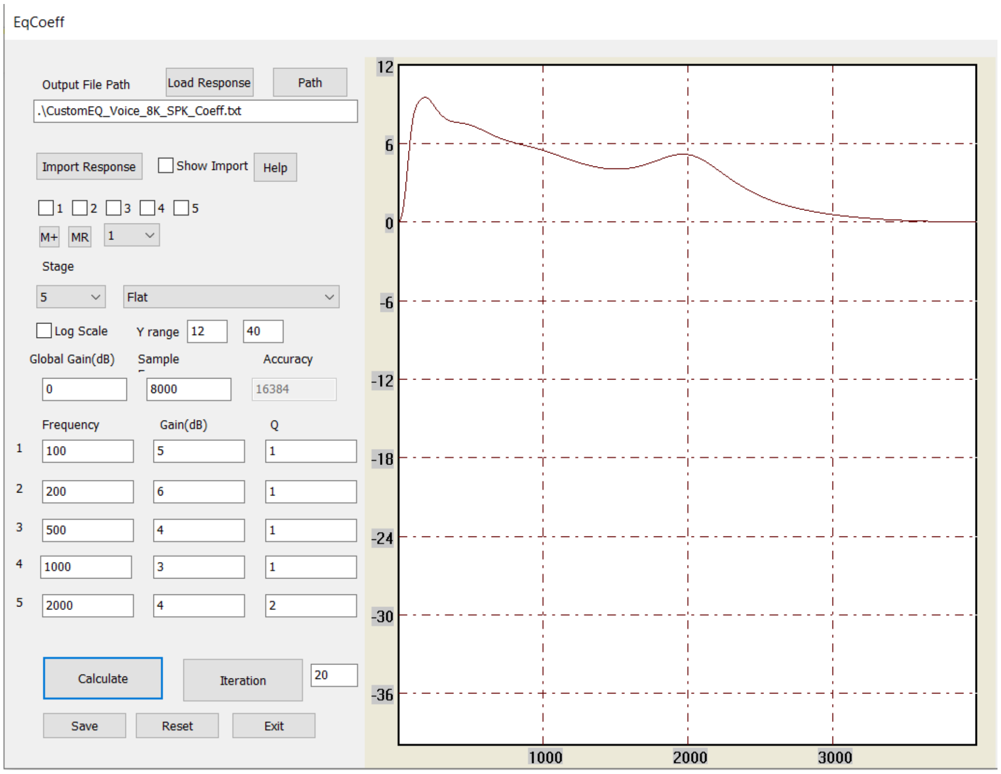
Note: The custom EQs with mSBC is configured for the HFP 1.6, supporting the wideband (16 kHz sampling rate) speech features.

Figure 4-7. EQ Configuration Interface for Speaker and MIC Paths in SCO or Speech Mode



The following figure illustrates how to configure the equalizer functions.

Figure 4-8. Configuring the Equalizer Functions



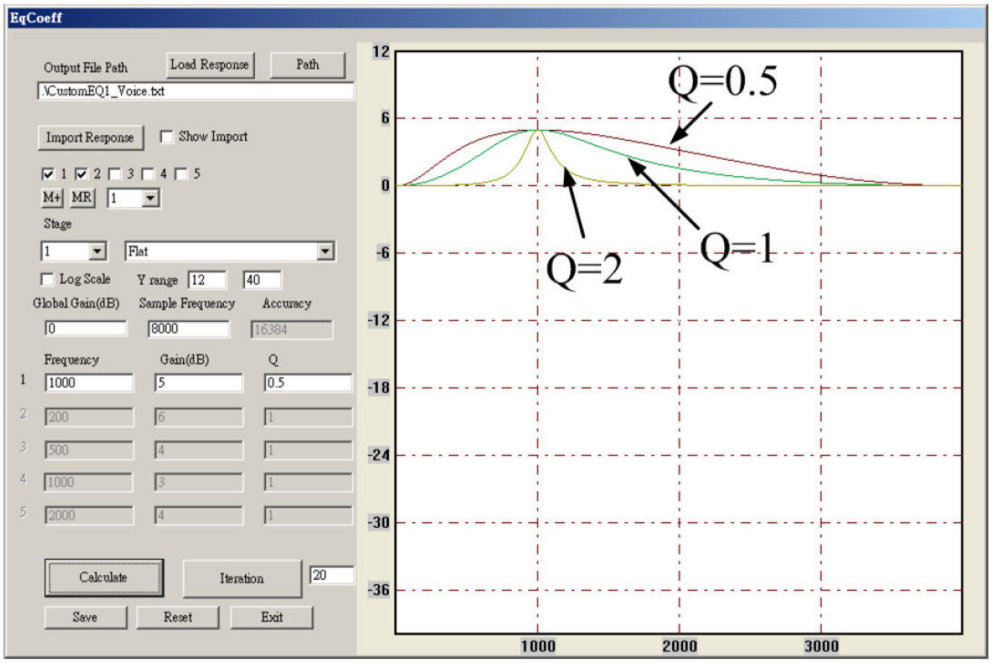
The user must enter the required frequencies and gain/attenuations in the columns. The Quality (Q) factor columns are to configure the cutoff frequencies of each equalizer band. Therefore, if the value of Q is smaller, the wider the bandpass cutoff frequency (see the preceding figure). The **M+** and **MR** buttons function as the calculator's M+ and MR. The purpose of these two buttons are to record the frequency responses for easy analysis comparison.

- Click the **Save** button to save the designed frequency response, and the system automatically stores the current EQ's configuration into a file.
- Click the **Load Response** button to restore the frequency response and EQ configurations from the previous design.

The section "Stage" configures the number of bands of the 1-order IIR filter. The fewer stages, then lower the MIPS and power consumption.

The following figure illustrates the functions of the Q factor.

Figure 4-9. Q Factor Function

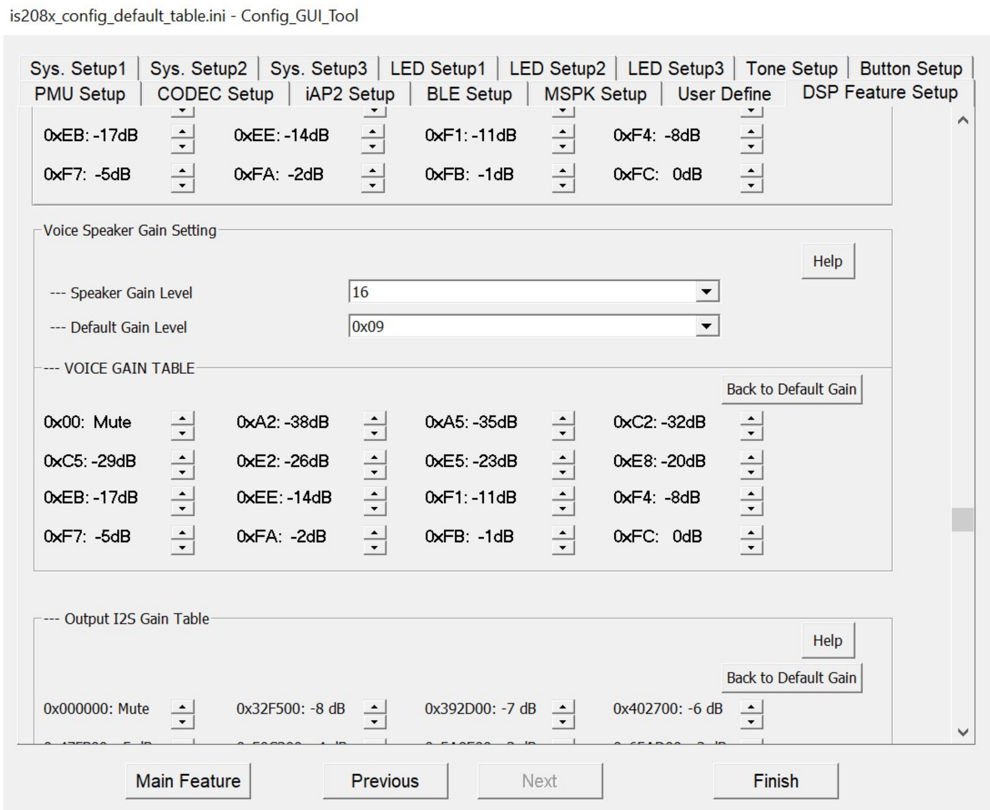


4.7 Speaker/MIC Gain Settings

The DSP configuration tool configures the speaker gain levels and MIC gain levels. There are three different speaker gain levels, and the user can select them based on their particular requirements. After determining the number of speaker gain level, choose the corresponding gain for each level.

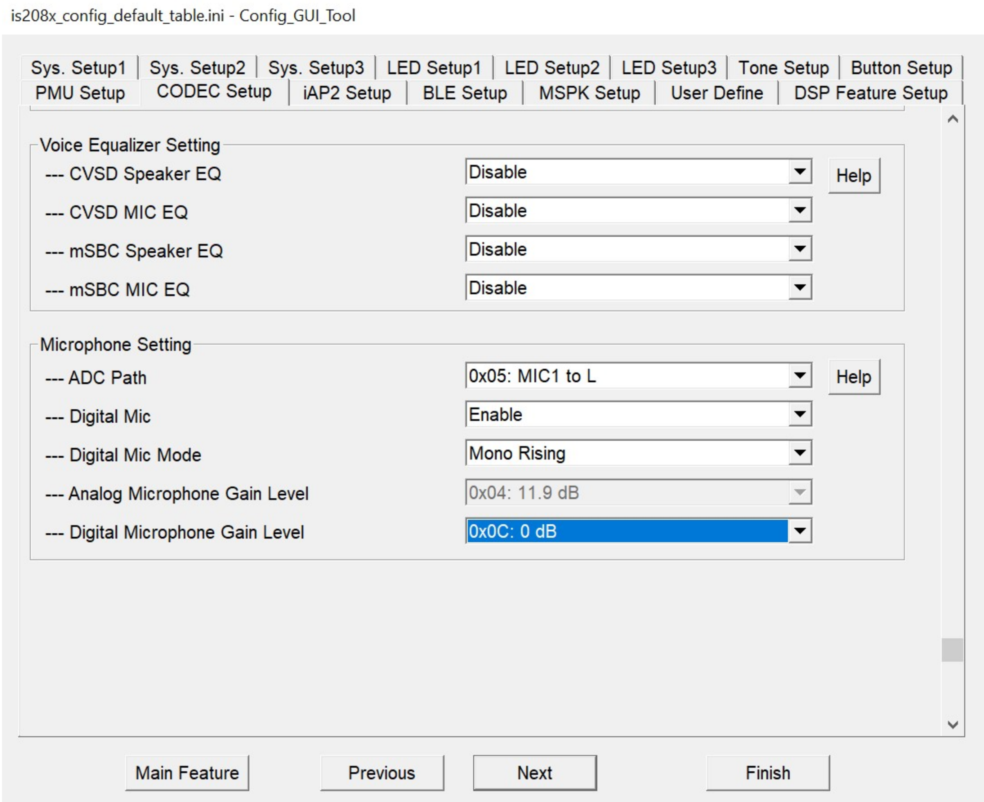
The gain difference between each MIC gain level for MIC Gain (codec) ranges between 2.0-3.0 dB per step. The following figure illustrates the interface for the speaker gain configuration in the SCO mode.

Figure 4-10. Interface for the Speaker Gain in the SCO Mode



The following figure illustrates the MIC gain configuration in the SCO mode.

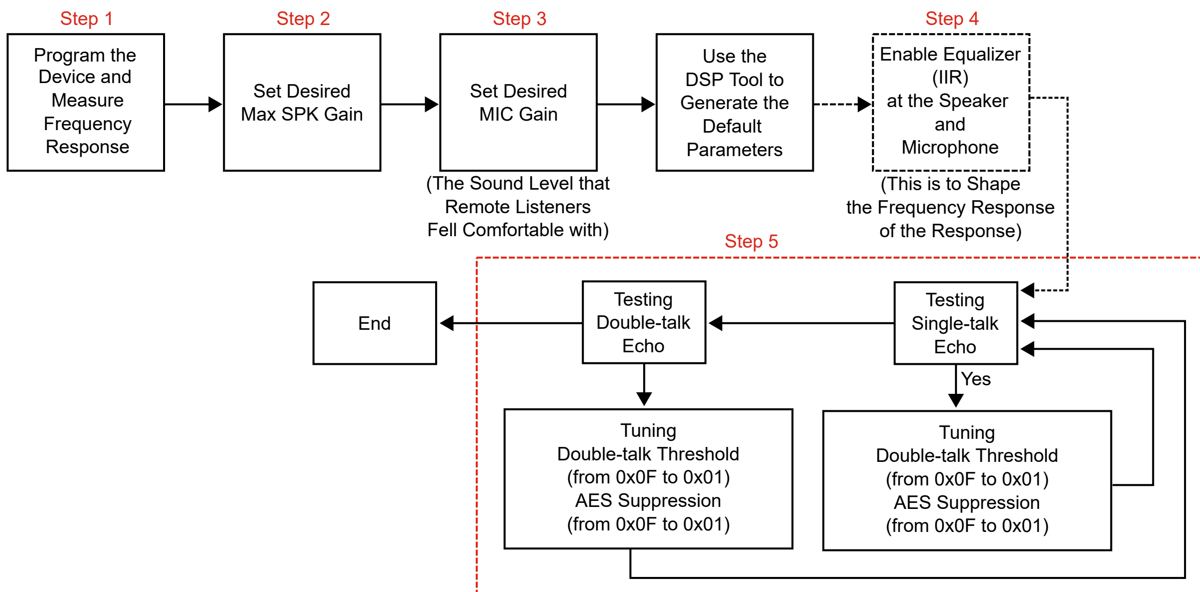
Figure 4-11. MIC Gain Configuration in the SCO Mode



5. Guidelines for Tuning Echo Cancellation Performance

This section introduces a guideline to fine-tune the EC performance. There are usually three requirements for the EC tuning. They are: high MIC volume, echo-free and double-talk performance. These three requirements are different from each other. For example, if there is a need for high volume at MIC path, the echo amplifies. Therefore, tuning of the parameters is required to make echo inaudible, which results in poor double-talk performance. The following figure illustrates the first five steps that basically handle the whole speaker phone/car kit echo issues.

Figure 5-1. Illustration of the AEC Tuning Flow



5.1 Measurement (Step 1)

1. Program the device with the merged file from UI, DSP configuration file and firmware.
2. Assemble the samples.
3. Measure the frequency response of the speaker and microphone of the sample (the recommendation is to use the maximum SPK and MIC gain at this moment).
4. If the frequency response is not good, go back to check the mechanical design and assembly. For more details, refer to 6. MIC and SPK Mechanical Design and Placement.

5.2 Set Receive Path Level (Downlink) Step 2

Before tuning the AEC performance, the user must determine the maximum desired speaker output level; that is, the recommended speaker output volume must be at least 95 dB Sound Pressure Level (SPL) for indoor speakerphones and 100 dB SPL for car-kit applications.

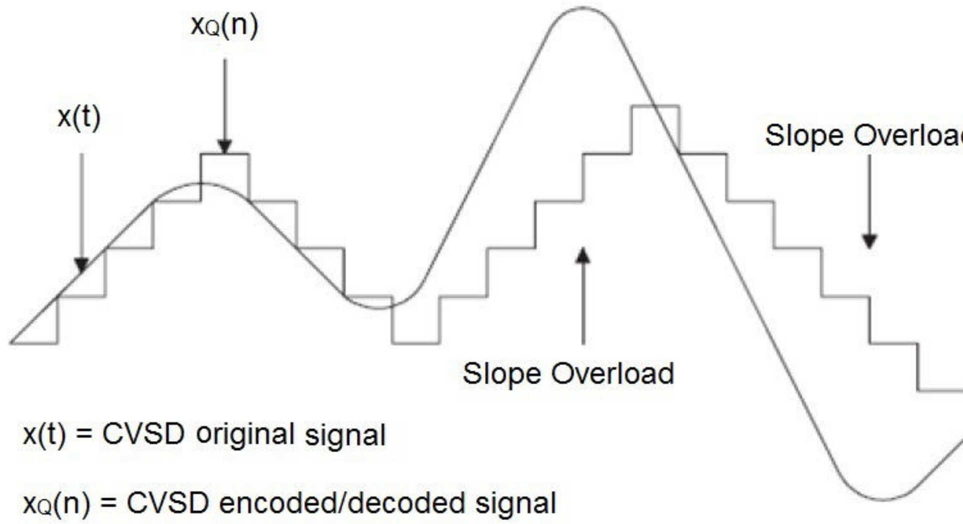
Note: The user must determine the target speaker output volume at the beginning based on the required specification.

5.3 Set Send Path Level (Uplink) Step 3

This is the principle step to adjust the MIC gain to a suitable value. It is not necessary to set the MIC to its maximum level due to the slope overload effect caused by the CVSD slope, which naturally suppresses the high-frequency parts and works as a low-pass filter. The following figure illustrates the CVSD slope overload effect. This effect distorts the near-end speech and makes it not as clear as the softer MIC gain levels.

For more information on the CVSD slope over load effect, refer to the Continuously Variable Slope Delta Modulation: A Tutorial.

Figure 5-2. CVSD Slope Overload Effect



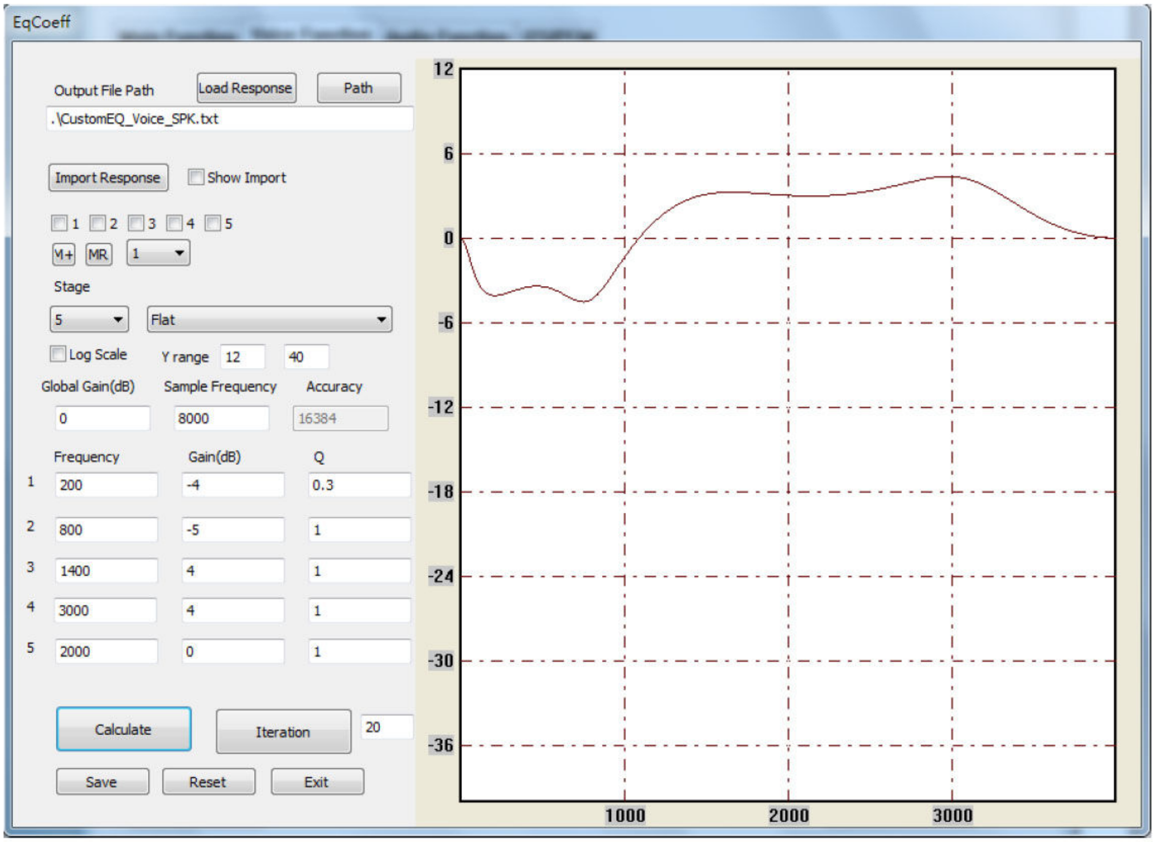
5.4 Tune EQ (Step 4)

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 (1-3 kHz).

Note: This is an optional step.

This helps in reducing the echo reverberation within the speaker phone/car kit housing, so that the linearity of the echo coupled to the MIC input can be better, which is highly associated with the AEC performance. The following figure illustrates an example of these settings, which are obtained empirically. However, the required frequency changes depending on the selected speaker and housing.

Figure 5-3. Example of the Frequency Shaping for Signal at Speaker Path



5.5 Fine Tune Echo (Step 5)

The purpose of this step is to fine tune the AEC performance, and it is divided into two parts:

- Single-talk echo tuning
- Double-talk echo tuning

5.5.1 Single-Talk Echo Tuning

The parameters Double-Talk Threshold and AES Suppression are responsible for tuning the Single-Talk echo performance. For more details, refer to 4.3. Echo Cancellation. Adjust the parameter of the Double-Talk Threshold from 0x7F to 0x1C. If the value of the AES Suppression is 0x04 and still not able to suppress the echo (audible single-talk echo) effectively, start to fine tune the AES Suppression from 0x01 to 0x0F.

Note: While the selectable value of the Double-Talk Threshold is up to 0x1C, the recommendation is to not use these two values to suppress the echo as they can severely distort the MIC speech.

5.5.2 Double-Talk Echo Tuning

If the single-talk echo is able to get suppressed effectively by the default settings of Double-Talk Threshold and AES Suppression, the double-talk performance can be increased further and fine tuned. The first recommended parameter for the double-talk performance is AES Suppression, which it is suggested to tune from 0x0F to 0x01 (no half-duplex). If the single-talk echo does not appear for the Double-Talk Threshold at 0x0F, the user must adjust the AES Suppression parameter from 0x0C to 0x01.

The possible measures to improve the double-talk performance are as follows:

1. Increase the AEC MIC gain – To increase the MIC gain, both AES and AEC suppress the echo during double-talk near end speech. For more details, refer to 5.3. Set Send Path Level (Uplink) Step 3. Improving the double-talk performance involves raising the near-end speech energy to make it audible while eliminating the single-talk echo.
2. Adjust the frequency shaping (see Figure 5-3). If the speaker output level is very loud and echo-to-speech ratio at the microphone input is too high, the way to improve this is to further suppress the low-frequency part of the speaker output. As a result, the echo-to-speech ratio at the low frequency parts of the microphone input is further reduced and can have the better double-talk performance.
3. Use the handsets to support the full-duplex AEC – Some mobile phones do not support the full-duplex speech communication while connecting with the Bluetooth hand-free devices. In this way, the user cannot obtain a 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 microphone and one of the speaker channels is very close (< 4 cm), 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, so that the full-duplexity can be much easier to achieve.

6. MIC and SPK Mechanical Design and Placement

This chapter provides details about guidance on the placement of microphones and speakers in a BM83 design. AEC performance is very dependent on the microphone and speaker placement (plastics design). In most cases, the difference between a design with poor performance and one with great performance is due to the plastics design and not the electronic design.

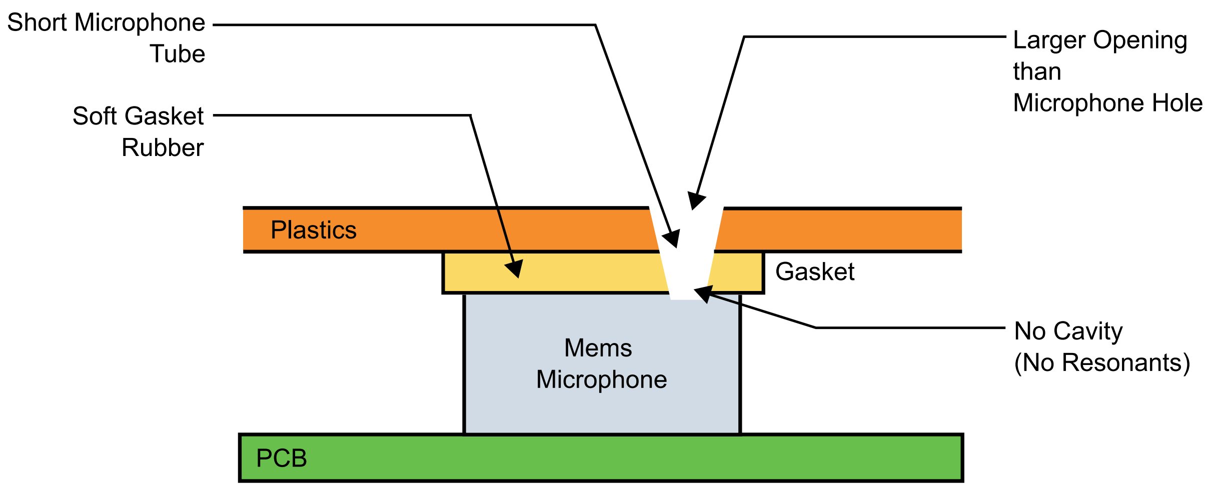
6.1 Microphone Design and Placement

6.1.1 Microphone Mechanical Design

The user must place the microphone in a soft rubber mount to eliminate audio coupling in the plastic enclosure. No part of the microphone must touch anything other than the rubber, and there must be no air gaps in the gasket to minimize vibration coupling. The gasket must not compress too much as it can pinch off the microphone hole.

The following figure illustrates the MEMS microphones; however, the analog microphone must be mounted in the same manner.

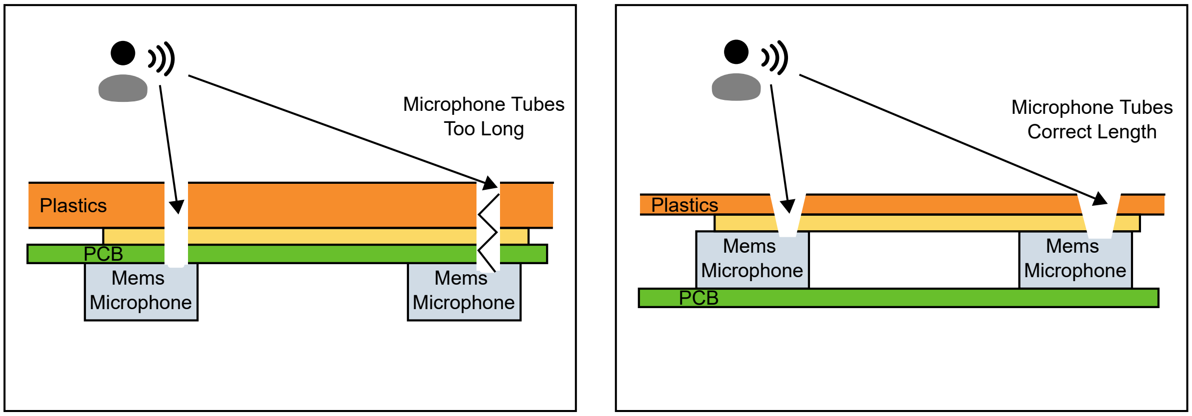
Figure 6-1. MEMS Microphone



Ensure the microphone opening is larger than the microphone aperture and that the microphone is aligned with the microphone hole. If this hole is too small, the microphone is going to pick up more sound from behind when compared to the front (in general, this causes a poor signal-to-echo ratio, which can impact the performance of the AEC).

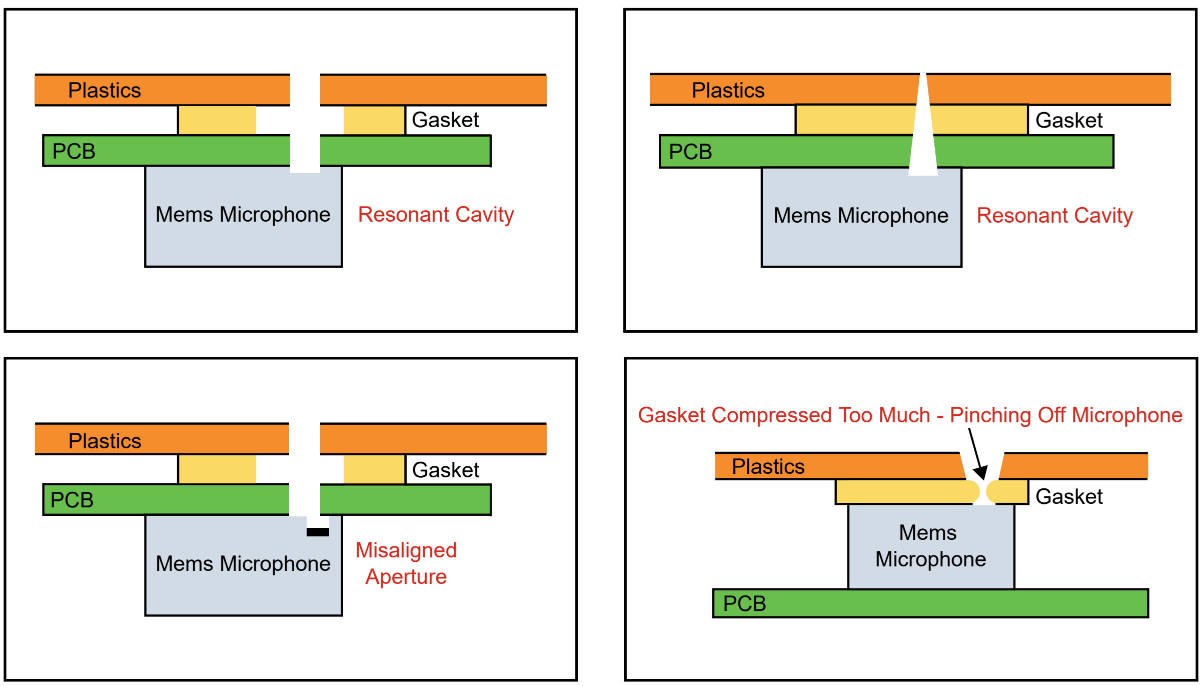
The microphone tube lengths must be kept to a minimum (in general, 0.5 mm) and a maximum of 4 mm to ensure lower frequencies are not attenuated. The user can use the rear-mounted microphone, but must be aware about the addition of the extra tube length by the PCB. The following figure illustrates the microphone design.

Figure 6-2. Microphone Design



The following figure illustrates the common errors when mounting the microphones.

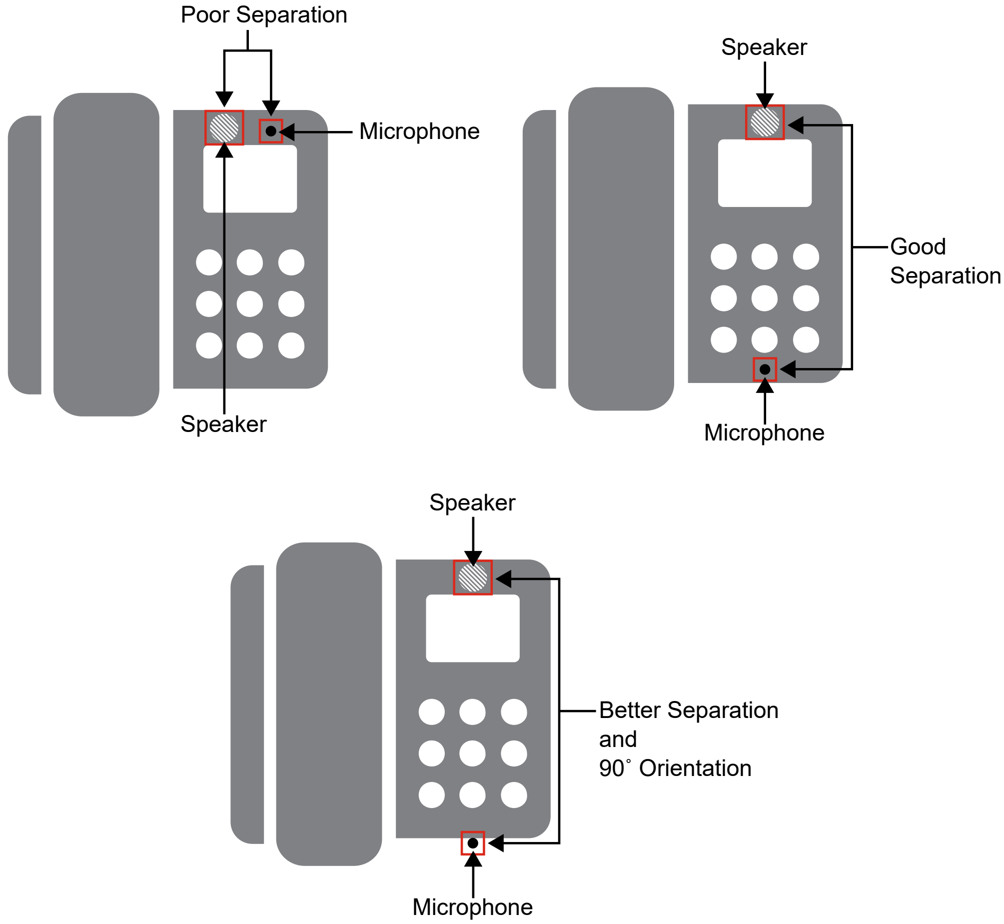
Figure 6-3. Common Errors when Mounting Microphones



6.1.2 Microphone Placement

The user can improve the AEC performance by maximizing the acoustic separation between the speaker and microphone. The microphone must be oriented at 90° with respect to the speaker whenever possible. The following figure illustrates the example microphone and speaker placements.

Figure 6-4. Example Microphone and Speaker Placements



6.2 Speaker Design and Placement

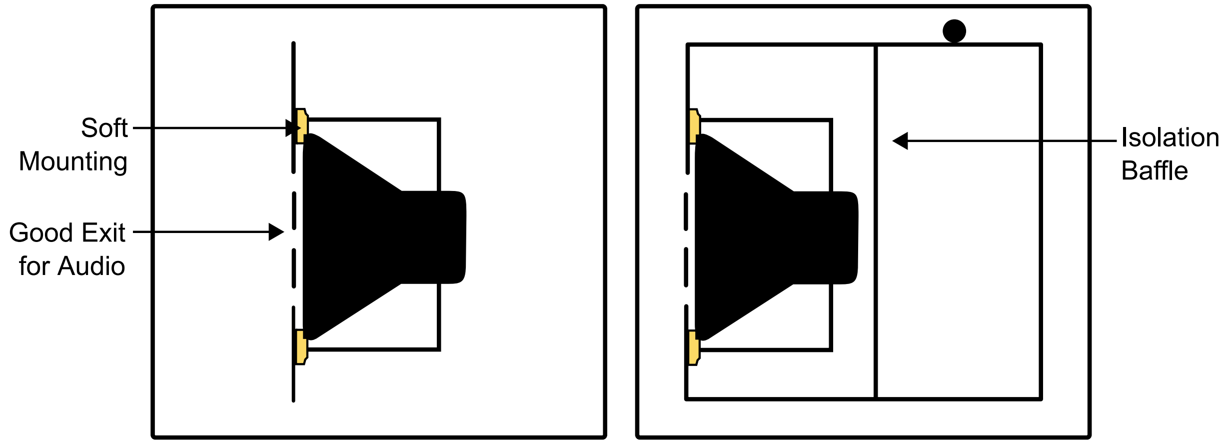
6.2.1 Speaker Mechanical Design

The speaker must be soft-mounted to eliminate audio coupling into the plastics. There must be soft rubber around the ring of the speaker where it is mounted against the front of the device. There must not be any hard mounting on the back of the speaker. The speaker bracket must only press against rubber or foam at the rear of the speaker.

Ensure there are sufficient sound holes placed in front of the speaker to allow a clear path for the sound pressure to the user. Insufficient sound holes can cause more of the sound pressure to be transmitted into the plastic enclosure and cause a reduction in the perceived volume and increase the acoustic coupling.

The acoustic coupling inside the plastics must be minimized using baffling or a box around the speaker itself. A box around the speaker can be very effective at reducing the return echo inside the product, but care must be taken not to impact the bass response of the speaker.

Figure 6-5. Speaker Mechanical Design



If the plastics do not have screw-down points, it can result in vibrations at specific audio frequencies. These vibrations result in high non-linear echo returning to the echo canceller and can greatly impact system performance. If there are any vibrations, it is very important to eliminate them as much as physically possible within the constraints of the phone design. It can be as easy as soft-mounting certain sections or adding an additional screw or two to hold the plastic case together more solidly.

6.2.2 Speaker Placement

For more details about speaker placement, refer to 6.1.2. Microphone Placement.

7. Document Revision History

The revision history describes the changes that were implemented in the document. The changes are listed by revision, starting with the most current publication.

Table 7-1. Document Revision History

Revision	Date	Section	Description
A	01/2024	Document	Initial Revision

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