IS208x Config GUI Tool
User Guide D00R04

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1. INTRODUCTION

This document is the user guide for the Configuration Tool of IS208x Bluetooth Speaker, IS208x_Config_GUI_Tool, which is designed to be installed in Microsoft Windows based PC to provide developers of IS208x Bluetooth Speaker a friendly interface to edit the Speaker product's parameters. There parameters include system setup、BLE setting、LED setting、power setting、buzzer setting、GPIO assignment、etc. Different configurations of parameter setting could empower different capacities and differentiate the products with one another.

2. NOMENCLATURE

- MFB

  MFB (Multiple Function Button) is a specific button that can be used for turning on/off Speaker and the other general-purpose functions.

- Pairing Mode

  If Speaker is in Pairing Mode, it is discoverable and waiting for remote devices to connect to.

- Single Mode

  If Speaker doesn't have a connection or pair with another Speaker, the state of the Speaker is called Single Mode.

- Twin Mode

  The Speaker has a connection with another one or is pairing with it, the state of the Speaker is called Twin Mode.

- Twin Speaker Link

  The connection between a pair of Speakers is called Twin Speaker Link.

- Twin Master

  Twin Master is the Speaker which initiates a Twin Mode Link and it may has a connection with a remote device such as a phone at the same time.

- Twin Slave

  Twin Slave is a Speaker as the slave role of Twin Mode Link, waiting for Twin Master to establish a Twin Mode Link. When a Speaker is the role of Twin Slave, it can't have a connection with a remote device.

- Twin Speaker Pairing

  Twin Speaker Pairing is the process of a Twin Master trying to pair and create a connection with a Twin Slave.

3. TOOL OVERVIEW

There are three parts in IS208x_Config_GUI_Tool: Start Menu, Main Features and Function Settings, those will be introduced in the following sections.

3.1 Start Menu Page

After launching this tool, the very first view you will see is Start Menu. It consists of two blocks, information block and operation block as presented below.

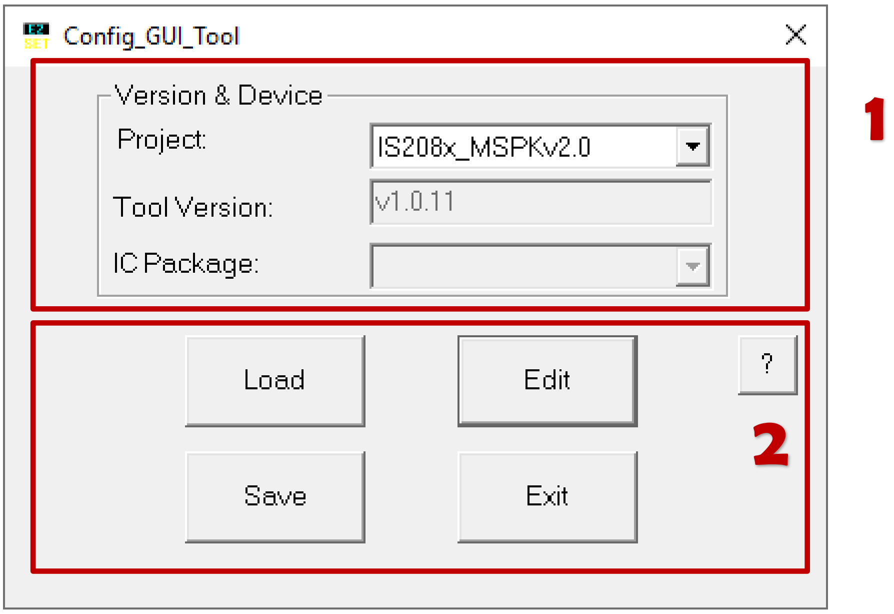


Figure 3.1 Start menu page view

3.1.1 Information Block

After loading a valid config parameter table, it displays the following information.

- **Project**

  Display the Project name.

- **Tool Version**

  Display the version of config tool

- **IC Package**

  Display the IC part number.

3.1.2 Operation

- **Load**

  Load a config parameter table. Before doing anything in this tool, loading a config parameter table is must.

- **Edit**

  Start to edit parameters. This button is disabled before loading a valid config parameter table.

- **Save**

  Save as a user factory table.

✧ **Exit**

Exit the config GUI tool.

3.2 Main Features Page

When you start to edit config parameters, you will see Main Features page as Figure 3.2. At first, this page lists some main features and shows the settings of the loaded config parameter table. Then you can modify it to achieve what you want.

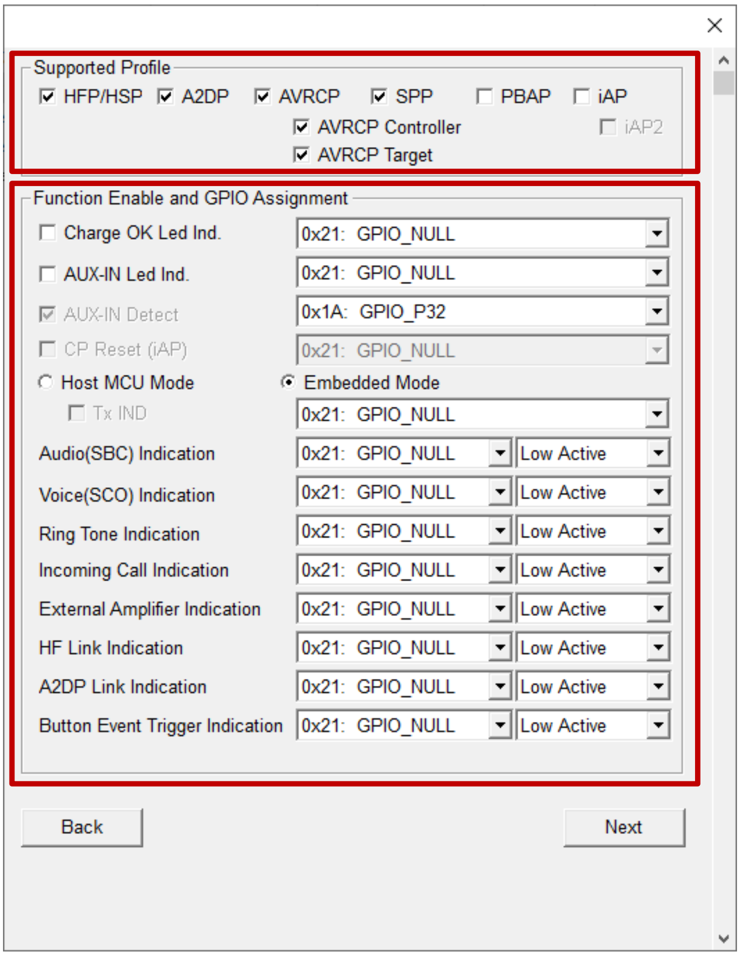


Figure 3.2 Main Feature

3.2.1 Supported Profiles

Determine what Bluetooth profiles the product supports.

○ **HFP (Hands-free Profile)**

Support voice input/output and some call operations.

○ **A2DP (Advanced Audio Distribution Profile)**

Support audio output.

○ **AVRCP (Audio/Video Remote Control Profile)**

Support audio remote control.

✧ **AVRCP Controller**

Support controlling the music players of remote devices such as play/pause, forward and rewind. All supported controls can be triggered by buttons of a Speaker upon the settings of button events.

- ✧ **AVRCP Target**

Support volume synchronization with music players of remote devices. When users adjust volume of music on remote devices, remote devices will send controls to Speakers to synchronize volume.

- ○ **SPP (Serial Port Profile)**

Support SPP profile.

- ○ **iAP (iPod Accessory Protocol)**

Support Apple iAP protocol.

- ✧ **iAP2 Protocol**

Support communications between iOS devices and wireless accessories.

3.2.2 Function Features

- ○ **Charge OK Led Ind.**

Support uses an additional LED3 to indicate charging complete state.

- ○ **Aux- in Led Ind.**

Support uses an additional LED3 to indicate Aux-in status.

- ○ **Aux Line-in**

Support Aux-in function as audio input by an Aux-in wire.

- ○ **CP Reset (iAP)**

Assign GPIO for iAP CP reset pin.

- ○ **Host MCU Mode**

Support application to use HCI UART.

- ✧ **TX Ind.**

TX Ind. is used to indicate MCU that data will be transmitted out.

- ✧ **RX Ind.**

Rx Ind. is used to wakeup Speaker before UART data transmission when speaker is in low power mode.

- ○ **Indication Setting**

Support to indicate some specific conditions and the polarities of indications are configurable.

The Speaker could control the specific pin to indicate something happened such as receiving an incoming call, having a call and so on. And this specific pin may be configured to reflect several states.

- ✧ **Voice (SCO) Indication**

Indicate that the voice link is established.

- ✧ **Audio (SBC) Indication**

Indicate that Speaker is playing audio.

- ✧ **Ring Tone Indication**

Indicate that a ring tone is outputting.

- ✧ **Incoming Call Indication**

Indicate that Speaker is having an incoming call.

✧ **HF Link Indication**

Indicate that the HF link exists between Speaker and the remote device.

✧ **A2DP Link Indication**

Indicate that the A2DP link exists between Speaker and the remote device.

✧ **Button Event Trigger Indication**

Indicate that the Button Indication Toggle function has been triggered.

3.3 Function Settings Page

Pressing the Next button on Main Features described above will pop up Function Settings page. Function Settings provides more detailed configurations. The settings of Main Features will affect some parameters on Function settings page. Base on the settings of Main Features, some detailed configurations may be fixed or limited to a reduced option. For example, once a profile is disabled in the configuration of supported features, all settings related to this profile will be canceled.

And there is a button in bottom left corner that can go back to Main Features if something is needed to be changed. When the editing is done, pressing the Finish button in bottom right corner to go back to Start Menu. Then you can save the editing as a parameter table.

There are so many parameters in Function Settings page you can get more information in the next chapter. Function Settings

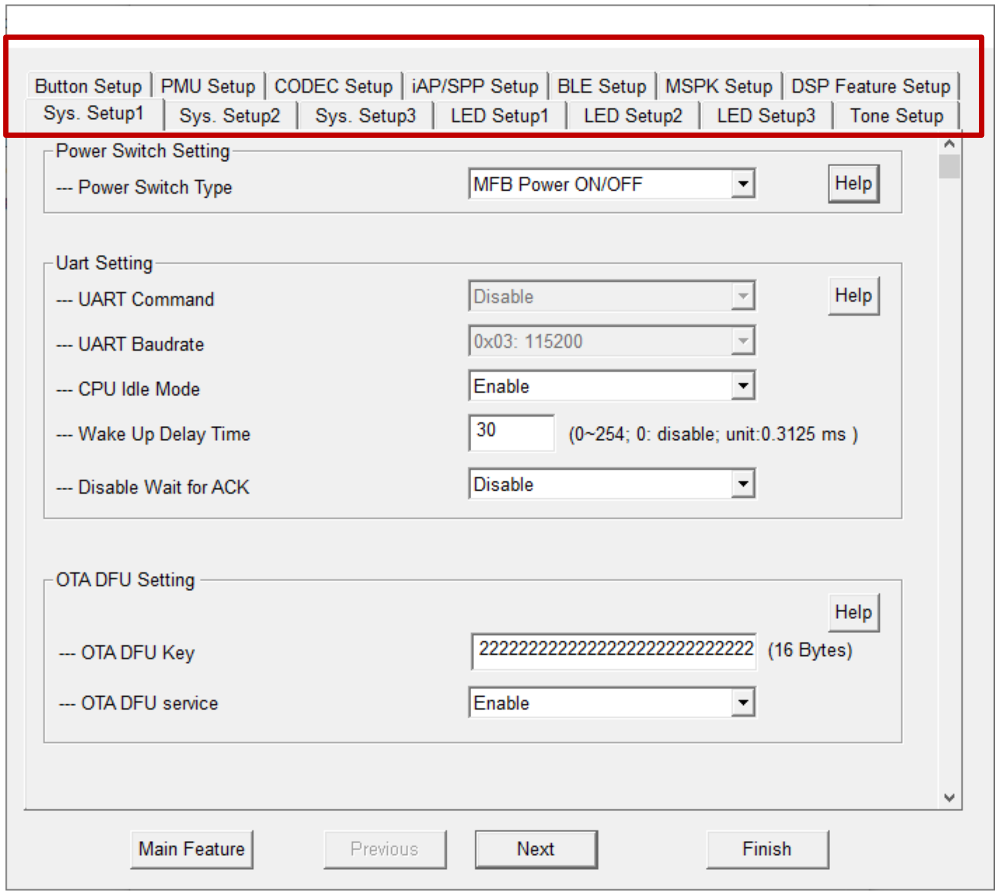


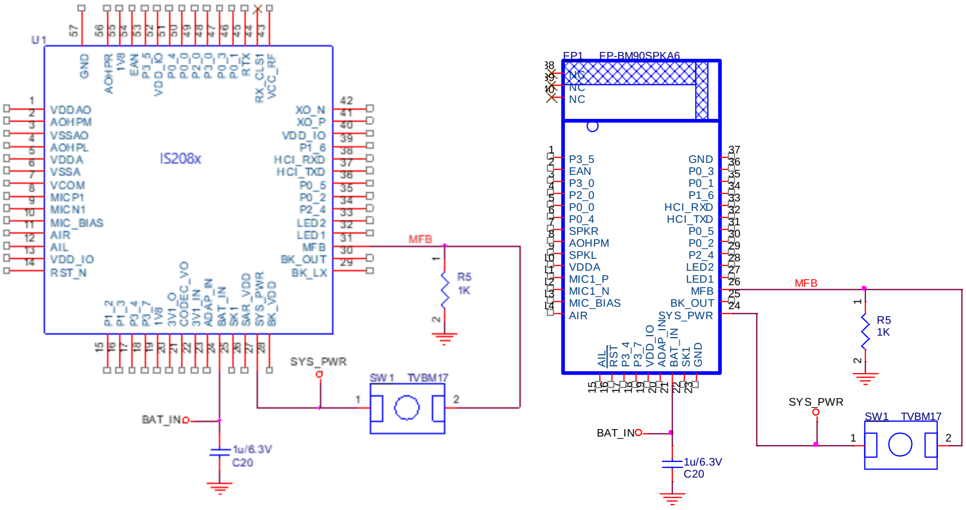
Figure 3.4 Function setting page view

3.4 Power Switch Setting

The power switch configuration has four types and determines which pattern is used in Speaker. These four types are listed as below.

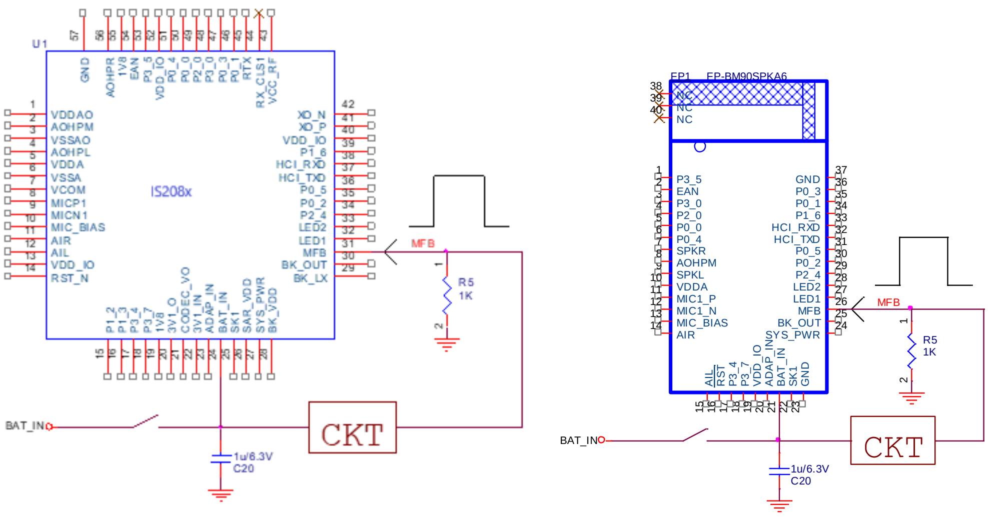
- **MFB Power ON/OFF**

  Turning on and turning off Speaker depends on the duration of pressing MFB button. The related settings are Power On Duration and Power Off Duration. That is, if users keep MFB key pressed over the configurable time, Speaker will be turned on or turned off. Please refer to 4.9.1 Button Press Duration for more information.



- **Power on Directly**

  The Speaker is controlled directly by power source. When Speaker connects to power source such as a battery and a plus to MFB Pin, it is on, otherwise it is off.



3.5 Connection Setting

The Speaker could establish Bluetooth connection with a remote device. The configurations listed below control the behavior regarding Bluetooth connection.

- **Link Back Scheme**

  The scheme of reconnection is to connect to the last device which is the highest priority or the paired devices with priority-based.

- **Infinite Pairing Mode Duration if No Paired Record**

  When Speaker has no paired device record, it keeps being discoverable. If this setting is enabled, above Pairing Mode Duration setting will be ignored.

- **Link Loss Reconnection Duration**

  When Speaker loses Bluetooth connection with a remote device by accident, it will try to link back to the remote device in this duration of time.

- **Link Back When Power On**

  This setting determines Speaker tries to reconnect to a paired remote device or not when it is turned on.

- **Link Specific Profile When Power On**

  Assign a target to reconnect to when Speaker is turned on. The target may be the last disconnected device or a paired device with the specific profile.

- **Link HS Duration**

  The setting is the duration trying to reconnect to the remote device with HF profile.

- **Link A2DP Duration**

  The setting is the duration trying to reconnect to the remote device with A2DP profile.

- **Pairing Timeout**

  The configuration is the duration of Speaker being discoverable. During this period, other devices can discover and pair it. If no device pairs Speaker successfully in this period, Speaker will leave Pairing Mode.

3.6 LED Setting

There are two LEDs named LED1 and LED2 in Speaker. They indicate what the current state is to users. The flash pattern of LEDs for every defined state could be configured. The defined state and related configuration are listed as below.

3.6.1 LED FLASH Parameters

In this section, the XXX represents one of the supported states described in next section.

- **XXX LED Type**

  The setting is the displayed pattern of XXX state. Below lists all supported types.

  - Type0

    LED1 and LED2 are always blind.

  - Type1

    LED1 and LED2 are always bright.

  - Type2

    LED1 is always bright and LED2 is always blind.

  - Type3

    LED1 is always blind and LED2 is always bright.

  - Type4

    LED1 is always blind.

    LED2 is twinkling.

  - Type5

    LED1 is twinkling.

    LED2 is always blind.

  - Type6

    LED1 is always bright.

    LED2 is twinkling.

  - Type7

    LED1 is twinkling.

    LED2 is always bright.

  - Type8

    LED1 and LED2 are twinkling simultaneously.

  - Type9

    LED1 and LED2 are twinkling alternately.

- **XXX LED On Duration**

  The bright duration for twinkling LED.

- **XXX LED Off Duration**

  The blind duration for twinkling LED.

- **XXX LED Interval**

  The configuration is for the interval of a round.

- **XXX LED Count**

  The configuration is for the LED flash times in a round.

  Notice that the value of XXX LED Interval must be more than (XXX LED On Duration + XXX LED Off Duration) * (XXX LED Count).

- **LED1 brightness**

  Configure the brightness of LED1.

- **LED2 brightness**

  Configure the brightness of LED2.

3.6.2 Defined LED State

This section lists all supported states of Speaker those can indicate by LED.

- **Power On LED Flash Setting**

  This LED setting is displayed when Speaker is turned on.

- **Power Off LED Flash Setting**

  This LED setting is displayed when Speaker is turned off.

- **Pairing LED Flash Setting**

  This LED setting is displayed when Speaker is in Pairing Mode.

- **Pairing OK LED Flash Setting**

  This LED setting is displayed when Speaker is paired with a remote device successfully.

- **Link LED Flash Setting**

  This LED setting is displayed if Speaker has a Bluetooth connection with a remote device.

- **Standby LED Flash Setting**

  This LED setting is displayed if Speaker doesn't have any Bluetooth connection.

- **Low Battery Disallow Power On LED Flash Setting**

  This LED setting is displayed Speaker is being turned on under low battery mode (this is disallowed).

- **Twin Mode Pairing Master LED Flash Setting**

  This LED setting is displayed if Speaker is Twin Master in Twin Mode Pairing state.

- **Twin Mode Pairing Slave LED Flash Setting**

  This LED setting is displayed if Speaker is Twin Slave role in Twin Mode Pairing state.

- **Twin Master with BT Connected LED Flash Setting**

  This LED setting is displayed if Speaker as Twin Master has a Twin Mode Link and a Bluetooth connection with a remote device.

- **Twin Master with BT Pairing LED Flash Setting**

  This LED setting is displayed if Speaker as Twin Master has a Twin Mode Link and is in Pairing Mode.

- **Twin Master with BT Link Back LED Flash Setting**

  This LED setting is displayed if Speaker as Twin Master has a Twin Mode Link and is trying to reconnect to a remote device.

- **Twin Master with BT Standby LED Flash Setting**

  This LED setting is displayed if Speaker as Twin Master has a Twin Mode Link and has no Bluetooth connection with a remote device.

- **Twin Slave LED Flash Setting**

  This LED setting is displayed if Speaker is Twin Slave role in an existing Twin Mode Link.

- **Incoming Call LED Flash Setting**

  The LED setting is displayed when Speaker receives an incoming call.

- **Talk LED Flash Setting**

  This LED setting is displayed when Speaker has an active call.

- **AV LED Flash Setting**

  This LED setting is displayed if Speaker is playing music.

- **Low Battery LED Flash Setting**

  The LED setting is displayed periodically if the battery voltage of Speaker is low. The settings related to battery detection is discovered in 4.6 Battery Detection Setting.

- **Link Back LED Flash Setting**

  This LED setting is displayed when Speaker is trying to reconnect to a remote device.

- **Min and Max Volume LED Flash Setting**

  This LED setting is displayed under the operation of increasing the volume of Speaker upto maximum volume and vice versa.

- **Mic Mute LED Flash Setting**

  This LED setting is displayed if the microphone of Speaker is muted.

3.6.3 Charging LED Setting

This section describes the LED settings for charging states. There is an optional indicator LED3 accomplished by a pin to indicate charging complete.

- **Charging LED Option**

  There are two options listed as below to indicate charging state by LED2.

  - LED2 is not affected. That is, it has no indicator for charging state.
  - LED2 is bright.

- **Charging Error LED Option**

  If a fault occurs while charging, the setting determines how to indicate the error..

  - Use the same setting with above Charging LED Option.
  - LED1 is always blind and LED2 flashes.

- **Charging Complete LED Option**

  The patterns of LEDs for charging complete as below.

  - LED1 and LED2 display as normal. No special indicator for charging complete.
  - LED1 flashes once, and then LED1 and LED2 display as normal.
  - LED1 is always bright and LED2 displays as normal.
  - LED1 is always blind and LED2 displays as normal.

- **Extra Charging Complete LED**

  This option determines if using the additional LED3 to indicate charging complete state. Note that it has no twinkling type for LED3.

3.7 Battery Detection Setting

It supports the feature of battery detection on Speaker. There are three stages of voltage levels decided by the settings of Low Battery Warning Level and Battery Shut Down Level. If the battery voltage of Speaker is higher than the value of Low Battery Warning Level, it is at normal battery state. Otherwise it is at low battery state and it may have some warnings by tones or LEDs upon settings to remind users to charge battery. And once the battery voltage is lower than the voltage of Battery Shut Down Level, Speaker starts the scheme of battery protection and shuts down automatically.

- **Battery Detection**

  Enable or disable battery detection.

- **Low Battery Warning Level**

  The setting is the threshold of low battery state. (Range: 3.0v~4.2v)

- **Battery Shut Down Level**

  The setting is the threshold of battery protection and should be lower than the voltage of Low Battery Warning Level. (Range: 3.0v~4.2v)

- **Low Battery Warning Time**

  The setting is the period of the low battery warning. (Unit: 0.64sec)

- **Low Battery Debounce**

  If this option is enabled, the low battery warning will keep until power adaptor in or power cycle.

3.8 Power Save

To save power, Speaker provides the automatic shutdown function excluding some conditions. This section describes the related settings of automatic shutdown function.

- **Auto Power Off Time**

  The setting is the timer to shut down automatically.

- **In Case Of Auto Power Off Inhibit**

  This setting determines some conditions that will stop and reset the timer of automatic shutdown. If Speaker is not at these conditions, the timer starts. And when the timer reaches the value of Auto Power Off Timer described above, Speaker shuts down automatically. The conditions that may disable the shutdown function are listed as below.

  - Pairing Mode

    The Speaker is in Pairing Mode.

  - Music Streaming

    The Speaker is playing music with A2DP link.

  - Line-in Plugged

    The line-in jack of Speaker connects a music input.

  - Link Back

    The Speaker is linking back to a remote device.

  - Charging

    The Speaker is being charged.

  - HF Link

    The Speaker has an HF link.

  - A2DP Link

    The Speaker has an A2DP link.

3.9 Charging Setting

The Speaker supports charging function. In general, there are several stages in charging cycle and it has different ways to charge in each stage. When charging, Speaker detects the voltage of battery to decide in which stage it is. The charging flexibility Speaker supported lists as below.

- **Charging Detect Enable**

  Enable or disable the charging function.

- **Re-Charging as Charge complete**

  Enable or disable the recharging function. When Speaker stops charging due to charging complete for a while, the voltage of battery may fall slowly. If the voltage less than 4.1V and this function is enabled, Speaker charges battery again to keep it full.

- **Voltage Level for each Charging Stage**

  Constant current voltage for each stage.

- **Constant Current Protect Time for each Charging Stage**

  When cc mode overprotective time expired, charging will be completed and stop charging.

- **Maximum Charging Current for each Charging Stage**

  Maximum charging current (unit: mA) for each charging stage.

- **Constant Voltage Protect Time**

  The protection time of Constant Voltage state. If Speaker stays in Constant Voltage state over the time, it raises an error.

- **Stop Working SAR Min**

  The minimum SAR voltage for ambient temperature detection to stop charger.

- **Stop Working SAR Max**

  The maximum SAR voltage for ambient temperature detection to stop charger.

- **Ambient Temperature Charging Detection**

  Enable or disable ambient temperature detection during charging.

- **Stop Charging SAR Min**

  The minimum SAR voltage for ambient temperature detection to stop charger.

- **Restart Charging SAR Min**

  The minimum SAR voltage for ambient temperature detection to restart charger.

- **Stop Charging SAR Max**

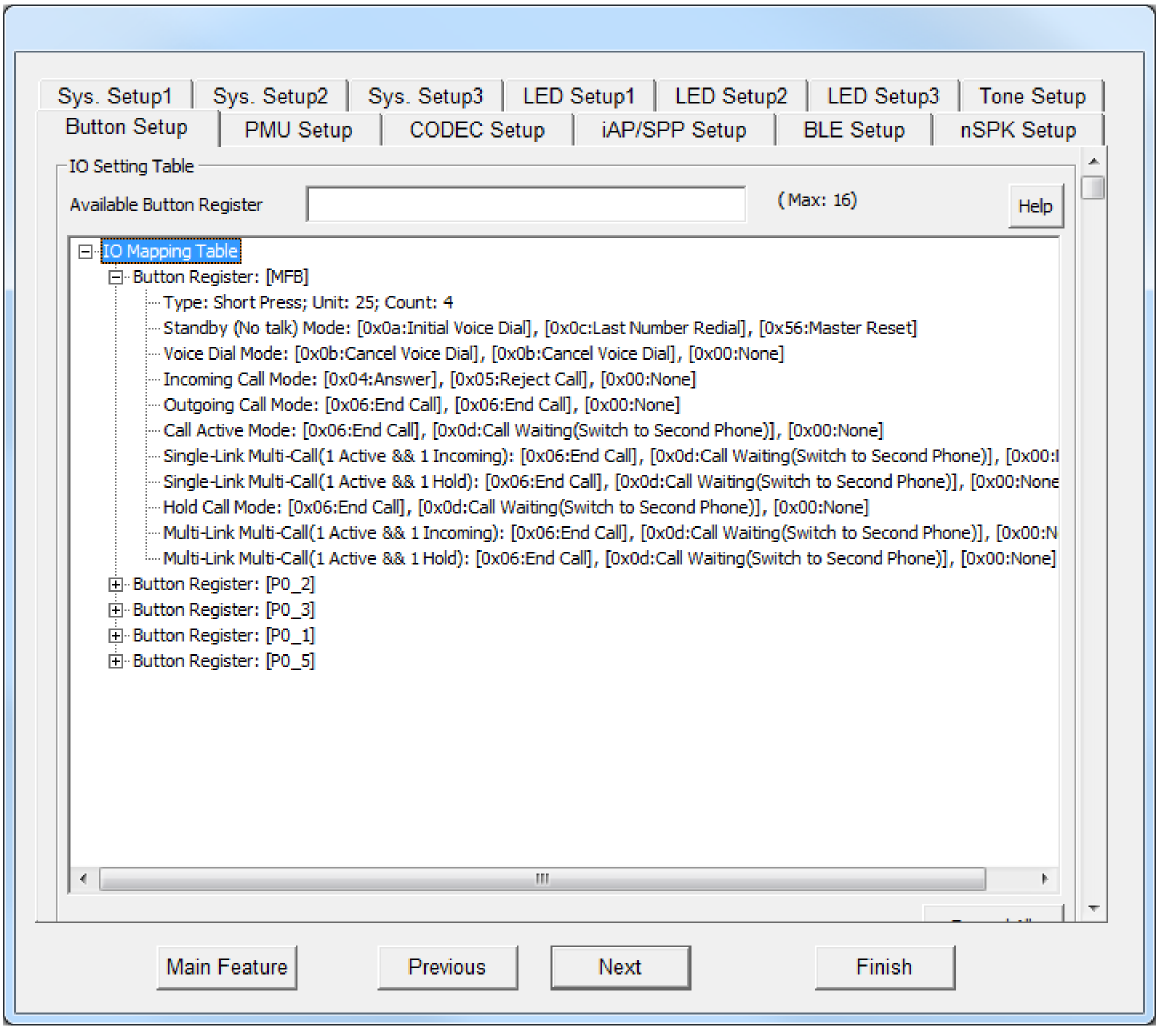
  The maximum SAR voltage for ambient temperature detection to stop charger.

- **Restart Charging SAR Max**

  The maximum SAR voltage for ambient temperature detection to restart charger.

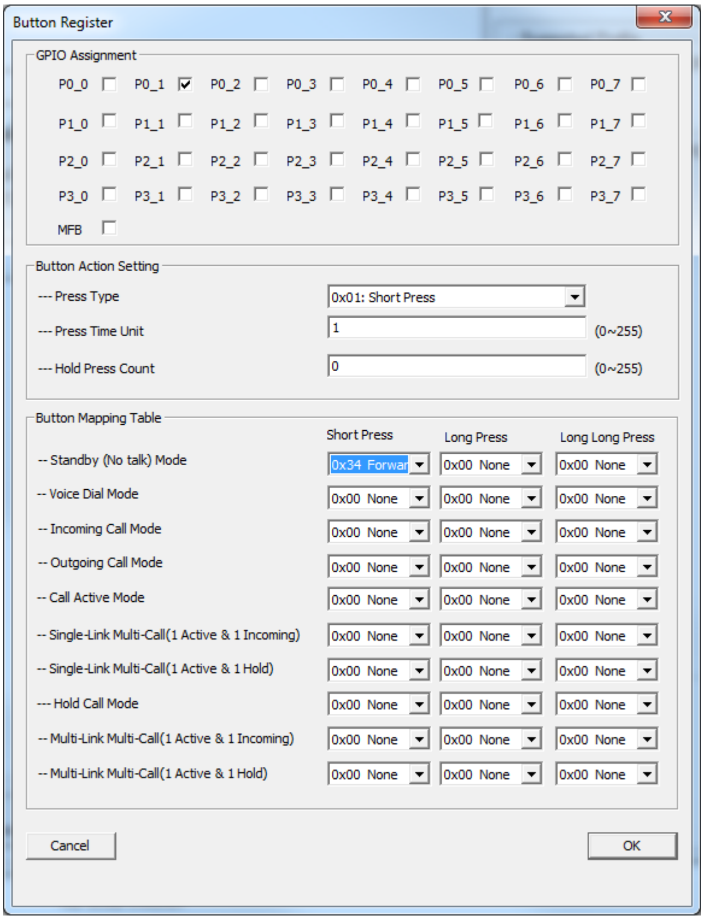
3.10 Button Settings

By operating buttons, Speaker can achieve some functions such as playing music, answering a call and so on. Here it will describe how these buttons work and what operation can be performed.



3.10.1 Function Mapping Table

The Function Mapping Table is to set the functions for the triggered events in all states that divided by call status. There are three kinds of button events in the Function Mapping Table; Short Press, Long Press and Long Long Press. If a function is set to certain button event, it means the function will be performed when this button event is triggered. For example, if the function Forward is set to the Short Press event of P0_1 in Standby (No talk) state, when users short press P0_1, the music will forward to next track. More information about performed functions and call states will be described later.



3.10.2 Button Press Unit

Configure the pressing time for button operations.

- **Press Type**

  There are three kinds of press type; Short Press, Long Press and Long Long Press.

  If pressing a button over this time it will trigger Long Press event, else it will have a Short Press event.

  If pressing a button over long long press time it will trigger Long Long Press event, else it will have a Long Press event.

- **Long Press Time**

  Configure the pressing key of Long Press. (Unit: 80ms)

  If pressing a button over this time it will trigger Long Press event, else it will have a Short Press event.

- **Hold Press Count**

  If pressing a button over long long press time it will trigger Long Long Press event, else it will have a Long Press event.

  (Long Long Press Time = Hold Press Count * Press Time Unit)

- **Long Long Press Time**

  If pressing a button over this time it will trigger Long Long Press event, else it will have a Long Press event.

3.10.3 Call State

It defines several states below upon call status.

- **Standby (No Talk) Mode**

Two conditions belong to this state; Speaker has no connection with a remote device or Speaker has no actions related to call.

- **Voice Dial Mode**

  The remote device connected Speaker is processing voice dialing.

- **Incoming Call Mode**

  The remote device connected Speaker is receiving an incoming call.

- **Outgoing Call Mode**

  The remote device connected -Speaker is dialing.

- **Call Active Mode**

  The remote device connected Speaker has an active call.

- **Single-Link Multi-Call (1 Active & 1 Incoming)**

  The remote device connected Speaker has an active call and a waiting call.

- **Single-Link Multi-Call (1 Active & 1 Hold)**

  The remote device connected Speaker has an active call and a held call.

- **Hold Call Mode**

  The remote device connected Speaker has an hold call.

- **Multi-Link Multi-Call (1 Active & 1 Incoming)**

  The remote device connected two Speaker has an active call and a waiting call.

- **Multi-Link Multi-Call (1 Active & 1 Hold)**

  The remote device connected two Speaker has an active call and a held call.

3.10.4 Button Functions

The functions below may be performed by button operation upon the settings of Function Mapping Table. When pressing a button, the corresponding action will be performed.

- **N/A (0x00)**

  No action.

- **Force End Call(0x02)**

  End the current active call no matter the voice sound is outputted by the local side and remote side.

- **Answer(0x04)**

  Answer the incoming call. Be used in Incoming Call Mode.

- **Reject Call(0x05)**

  Reject the incoming call or cancel the outgoing call. Be used in Incoming Call and Outgoing Call Mode.

- **End Call(0x06)**

  End current active call. Be used in Call Active Mode and the modes relevant to multiple calls.

- **Toggle Mic Mute(0x07)**

  Toggle the muted status of the microphone of Speaker. If the microphone is muted, it will be un-muted and vice versa.

- **Mic Mute(0x08)**

  Mute the microphone. Be used in Call Active Mode and the modes relevant to multiple calls.

- **Mic Unmute(0x09)**

  Unmute the microphone. Be used in call active and the modes relevant to multiple calls.

- **Initial Voice Dial(0x0A)**

  Initial the voice dial function. If Speaker isn't connected with a remote device at this moment, it will try to reconnect to the paired device. Be used in Standby (No Talk) Mode.

- **Cancel Voice Dial(0x0B)**

  Cancel the voice dial function. If Speaker isn't connected with a remote device at this moment, it will try to reconnect to the paired device. Be used in Voice Dial Mode.

- **Last Number Redial(0x0C)**

  Redial the last dial-out number. Be used in Standby (No Talk) Mode.

- **Call Waiting or Switch to Second Phone(0x0D)**

  Hold the active call and answer the waiting call or switch to held call. Be used in Call Active Mode and the modes relevant to multiple calls.

- **Transfer To Phone(0x0E)**

  Voice transferred between Speaker and the remote device. Be used in Call Active Mode and Standby (No Talk) Mode. Because some phones terminate Bluetooth connection when voice is transferred to the phone side, Speaker returns to Standby (No Talk) Mode at this moment. If you want to transfer voice between phone and Speaker by the same operation, this function should be also set to the same button event in Standby (No Talk) Mode. Besides, Initiate Voice Dial(0x0A) or Last Number Redial(0x0C) set in Standby (No Talk) Mode also can transfer voice to Speaker in that condition.

- **Join Two Calls(0x10)**

  Build a conference call. Be used in Single-Link Multi-Call (1 Active & 1 Hold) Mode.

- **Release Waiting Call(0x11)**

  Reject the waiting call or end the held call. Be used in the modes relevant to multiple calls.

- **Release Active Call and Accept Waiting Call**

  End the active call and accept the waiting call or held call. Be used in the modes relevant to multiple calls.

- **Connect HF Link (0x16)**

  Force Speaker to reconnect to a paired device with HF profile.

- **Disconnect HF Link (0x17)**

  Disconnect the HF link.

- **Volume Up (0x30)**

  When Speaker is playing music, it increases the audio volume by one step. And it increases the voice volume when Speaker is talking. Otherwise do nothing.

- **Volume Down (0x31)**

  The conditions of decreasing volume are similar with above Volume UP function.

- **Play (0x32)**

  Two conditions:

  - **Line-in removed:**

    If Speaker has no A2DP link, it will try to establish A2DP connection with a paired remote device. Otherwise it toggles the playing status between play and pause.

- **Line-in plugged:**

Toggle muted status of music.

- **Stop(0x33)**

Stop the music.

- **Forward(0x34)**

Go to the next song.

- **Backward(0x35)**

Go to the previous song or return to the beginning of current song upon the design of the remote device.

- **Fast Forward(0x36)**

Fast forward the songs.

- **Rewind(0x37)**

Rewind the songs.

- **EQ Up(0x38)**

Change the equalizer to next predefined mode if Speaker is playing.

- **EQ Down(0x39)**

Change the equalizer to previous predefined mode if Speaker is playing.

- **Lock Button(0x3A)**

Lock all buttons except the power button.

- **Disconnect A2DP Link(0x3B)**

Force to disconnect the A2DP link.

- **DRC Effect Toggle(0x3C)**

Enable or disable built-in dynamic range control function for audio.

- **Button Indication Toggle (0x58)**

Output the specific pin to high or low when this action is triggered.

The setting of Button Event Trigger Indication described later must be enabled if Speaker needs to support this function. Please refer to 4.14 Indication Function.

- **Master Reset(0x56)**

Reset the paired device history and the volume of voice and audio.

- **Function0 (0x59)**

Trigger the function that is set in the Combined Function0.

- **Function1 (0x5A)**

Trigger the function that is set in the Combined Function1.

- **Function2 (0x5B)**

Trigger the function that is set in the Combined Function2.

- **Function3 (0x5C)**

Trigger the function that is set in the Combined Function3.

- **Pairing Mode (0x5D)**

Enter the Pairing Mode to be discoverable.

- **Indicate Battery Status (0x6A)**

  Display current battery status by LEDs. It divides the voltage to three levels:

  - **Above 4V**

    LED1 brightens 2 seconds.

  - **Between 3.7V and 4V**

    LED1 and LED2 brighten 2 seconds.

  - **Under 3.7V**

    LED2 brightens 2 seconds.

- **Twin Master Role (0x70)**

  Force Speaker to be Twin Master and start Twin Mode Pairing process.

- **Twin Slave Role (0x71)**

  Force Speaker to be Twin Slave and wait for Twin Master to create a link.

- **Twin Speaker Pairing Cancel(0x72)**

  Cancel the Twin Mode Pairing if processing and return to the Single Mode.

- **Twin Speaker Disconnection(0x73)**

  Disconnect the Twin Mode Link and return to the Single Mode if the Twin Mode Link exists.

- **Twin Speaker Pairing Cancel/Disconnection(0x74)**

  Cancel the Twin Mode Pairing or disconnect the Twin Mode Link upon the status of Speaker.

- **Intelligent Twin Speaker Pairing(0x75)**

  Start to process Twin Mode Pairing and try to create a connection between two Speakers upon the presetting group code.

3.11 Tone Setting

The Speaker generates ring tones to notice users in some status and operations.

- **Power On Tone**

  The tone setting is for turning on Speaker.

- **Power Off Tone**

  The tone setting is for turning off Speaker.

- **Enter Pairing Tone**

  The tone setting is for entering the Pairing Mode.

- **Pairing Complete Tone**

  The tone is generated when Speaker pairs with a remote device successfully in Pairing Mode.

- **Pairing Not Complete Tone**

  The tone is generated when Speaker fails to pair with a remote device in Pairing Mode.

- **HF Connected Tone**

  The tone type is for creating the HF link successfully.

- **Link Loss Tone**

  The tone type is for abnormal disconnection.

- **Incoming Call Tone**

  The tone is generated when Speaker is receiving an incoming call.

  - **Incoming Call Tone Repetition**

    Enable or disable that Speaker repeats the incoming call tone periodically when Speaker is receiving an incoming call.

- **Low Battery Tone**

  The tone type is for low battery warning. This tone is generated periodically until the battery voltage is higher than the Low Battery Warning Level configuration.

- **Short Press Tone**

  The tone type is for the button event of Short Press.

- **Long Press Tone**

  The tone type is for the button event of Long Press.

- **Double Click Tone**

  The tone type is for the button event of Double Click.

- **Call Active Tone**

  The tone type is generated when Speaker answers an incoming call.

- **End Call Tone**

  The tone type is for ending the active call.

- **Press Volume Key Tone**

  The tone type is generated when Speaker adjusts volume by button events.

- **Min. & Max. Volume Tone**

  The tone type is generated if increasing the volume of Speaker that has reached maximum volume and vice versa.

- **No Service Tone**

  The tone type is for out of phone call service.

- **Mute Tone**

  The tone type is generated when Speaker has an active call and the microphone is muted.

- **Music Mode Ready Tone**

  The tone type is for creating the A2DP link successfully.

- **Function Alarm Tone**

  The tone type is generated when the NFC function is triggered successfully.

- **Twin Mode Enter Pairing Tone**

  The tone type is generated when Speaker is processing the Twin Mode Pairing.

- **Twin Mode Connected Tone**

  The tone type is generated when the Twin Mode Link is connected successfully.

3.11.1 External Audio Tones

The Speaker provides a way to add external tones. If users want to use their own tones, they can add these tones by the audio converter which is built in the config tool. After adding the tones, the corresponding options will be created in the tone settings above. There is one thing need to be paid attention, all translated data of external tones is stored in the configuration table and will consume the EEPROM space.

There are two kinds of external audios: Voice Prompt and Multi-Tone.

- **Voice Prompt**

  The audio format must be *.wav and meets the conditions listed as below. If not, it may pop up a warning message and fails to add the tone.

  - **Limit the length of the tone to 3 seconds except the length of Power On Tone that limits to 4.5 seconds.**
  - **Sampling rate should be 8k.**
  - **Bit rate should be 16bit.**
  - **The type is mono type.**
  - **The size of translated raw data can't over 9K bytes.**

- **Multi-tone**

  The audio format must be *.mid and comply with the conditions listed as below. If not, it may not play the tone as expectation.

  - **Limit the length of the tone to 3 seconds except the length of Power On Tone that limits to 4.5 seconds.**
  - **The beat clock should be 135.**
  - **The octave range should be 3.**
  - **No multi-chord.**
  - **The smallest musical note is sixteen notes.**
  - **Single soundtrack.**
  - **The size of translated raw data can't over 256 bytes.**

3.11.2 Add and Delete External Tones

The following introduces how to add and delete external audio tones and related functions.

- **Add Tone**

  Click the button of Add Tone to add an external tone. The maximum number of external tones is twenty. If the amount of the external tones reaches this limitation, it won't be allowed to add tones.

- **Tones Selection**

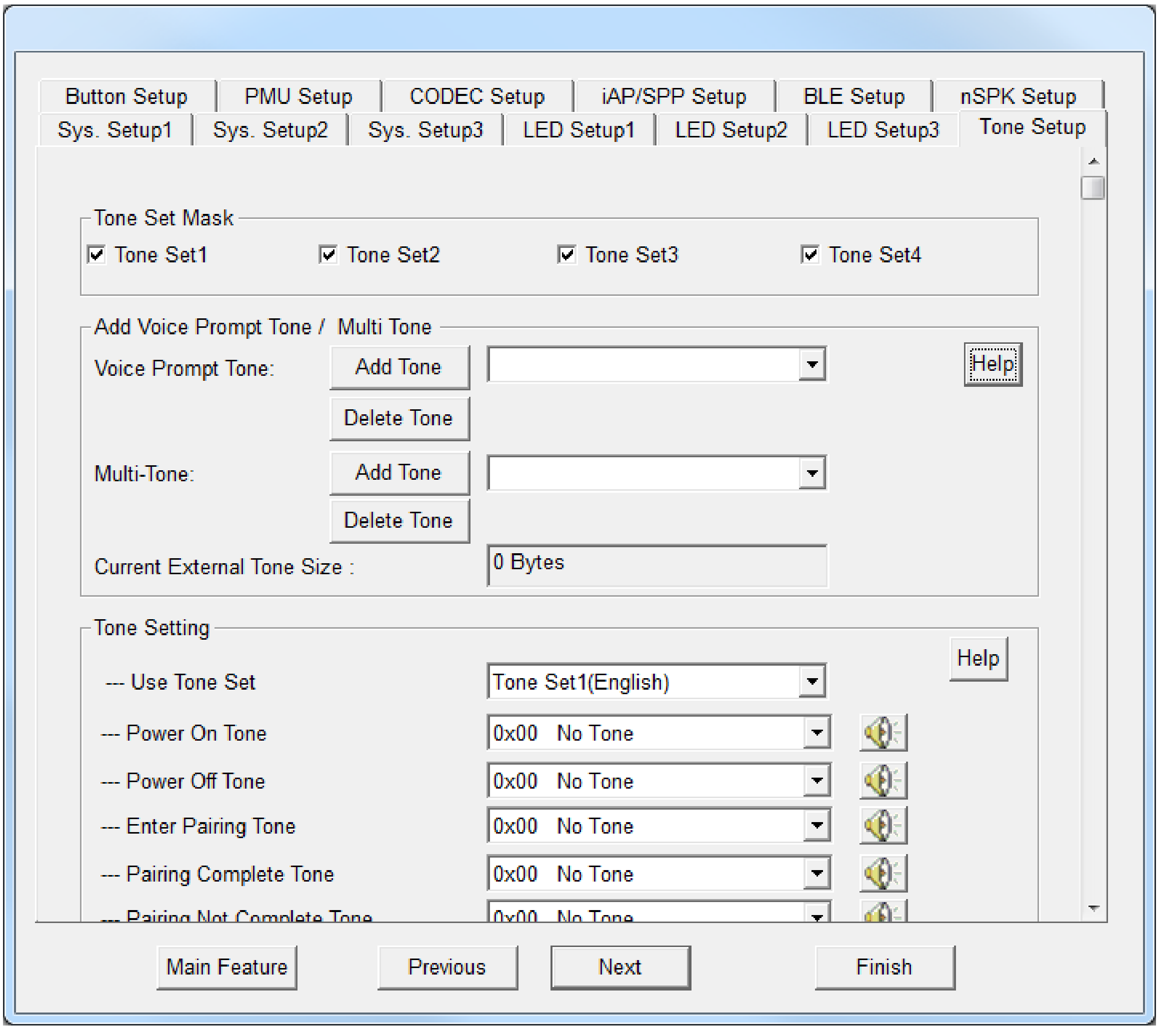
  Display all added external tones.

- **Delete Tone**

  Select a tone in Tones Selection above. Clicking the button of Delete Tone can remove the selected external tone.

- **Current External Tone Size**

  This field displays the needed space of current configuration. The displayed value reflects operation of adding and deleting tones.



3.11.3 Other Tone Settings

Other tone settings are listed as below.

- **Fixed Ring Tone Volume**

  The volume of ring tone is fixed and cannot be changed with the volume of voice.

- **Ring Tone Volume**

  Configure the ring tone volume level if the setting above is enabled.

- **Button Press Alarm**

  Generate the ring tone when pressing any key except the key with the volume adjustment.

- **MFB Long Press Force Alarm**

  Generate the ring tone when triggering Long Press events by MFB key.

3.12 Line-in Setting

The Speaker Supports optional line-in audio input by an Aux-in wire. The related parameters listed as below should be configured if this function is enabled.

- **Line-in/SBC Priority**

This setting determines the priority of audio sources. There are two audio sources that one is from an A2DP link and the other is from line-in input. If these two play simultaneously, Speaker will output the one as high priority upon this setting. For example, if it has two sources at the same time and set line-in high priority, Speaker outputs audio from line-in. And Speaker changes the audio source to A2DP when unplugging the line-in.

Line In > SBC means when system detects LINE_IN plugs in, it will switch to line in mode.

SBC> Line In: means system will still be in A2DP mode even if Line In is plugged in.

- **Line-in Mute/Unmute**

This setting determines whether the line-in audio can be muted. If it is enabled, the muted status is controlled by the button events with Play function described in 4.9.6 Button Functions.

- **Line-in Silence Detection**

Detect silence status for Aux-in signal if this option is enabled. The duration to determine the Aux-in is silence is about 33 seconds.

- **Line-in Indicate Led**

Use the GPIO pin to indicate Aux-in status. The pin is low active, 0 means the speaker is in Aux-in mode.

- **Line-in CSB**

Enable or disable Broadcast/2-SPK Aux-in mode.

- **Line-in Latency**

Line-in latency adjustment (unit: *1/3 ms)

ex: 0xF0 => 240 * 1/3 = 80 ms

- **Initial Line-in SPK Gain**

Default gain level for line in mode.

3.13 Amplifier Control Setting

The Speaker outputs sounds with the external amplifier. To avoid outputting the unexpected noise when the CODEC is turning on or off, it needs to tune the timing of CODEC on and off base on the property of external amplifier.

- **Cut Off Amplifier When Mute**

  This setting determines if turning the external amplifier off or not when the CODEC has no data to process.

- **Off Amplifier T0**

  This setting determines the timer to turn the external amplifier off when the CODEC has no data to process.

  If this value set to 0, then turn off external amplifier directly.

- **Off Amplifier T1**

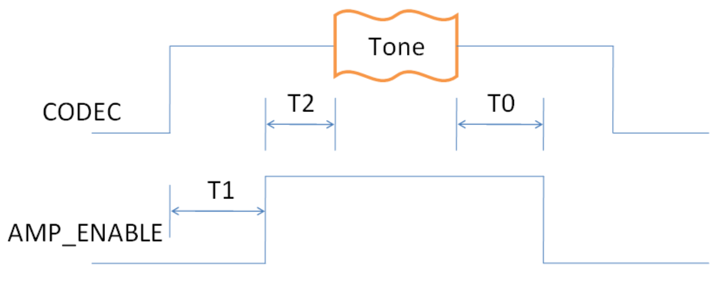
  The timer to enable the external amplifier after the CODEC is activated.

  If this value is 0, then wait 100ms.

- **Off Amplifier T2**

  when AMP_CTRL_T1 expired, then wait at least this time to start play audio.

  Notice: AMP_ENABLE means external amplifier enable pin



3.14 CODEC Function

This section describes the settings related to CODEC. The hardware circuit should be considered when configuring these settings to get good performance.

- **Speaker Output**
  Set the output type as Capless or Single End upon the hardware circuit.
- **Enable LR Sound Channel Swap**
  Swap left and right channel speaker output (only for A2DP)
- **Enable LR Sound Mix**
  Enable or disable to mix the left channel with the right channel to a single channel.
- **DSP Codec Always on Enable**
  Never turn the CODEC off.
- **Close Codec Time**
  When DSP in SBC decode state, and FW do not receive any stream data, it will close CODEC after this time
- **CODEC Output Type**
  Select to use external codec or internal codec.
- **Audio SRC**
  44.1K to 48K for A2DP audio.
- **Voice SRC**
  8K/16K to 48K for ring tone or SCO voice.
- **Tone Stereo**
  Enable or disable the function that single tone and VP output both of L and R channel.
- **Voice Stereo**
  Enable or disable the function that voice outputs both of L and R channel.

3.15 Twin Mode Setup

3.15.1 Twin Speaker Settings

- **Twin Pairing Duration**
  The setting is the maximum duration for Speaker at Twin Mode Pairing.
- **Twin Aux. Pairing Duration**
  The CSB pairing timeout of Slave role in One_Key_Operation. (Unit: 2s)
  When the One_Key_Operation is triggered, it enters CSB pairing as Slave role first and then change to Master role after this timeout occurs.
- **General Twin Mode Group Code**
  The setting is for General type of Twin Mode Pairing Scheme above to discover a matching Twin Slave.
- **Stereo Model ID**
  The setting is used with "General Twin Mode Group Code" for Stereo Mode.
  It will disallow the speakers with different model IDs form the connection of Stereo Mode.
- **Recover MSPK Link As Twin Speaker Link Loss**
  Try to recover the Twin Mode Link when losing the Twin Mode Link abnormally.
- **Default Multi Speaker Mode**
  Select to use stereo mode or concert mode.
- **Concert Mode Resync**
  Enable audio resync function in Concert Mode.
- **Concert Mode Link Status Record**
  Enable/Disable broadcast back to last mode.
- **ADV Policy in Concert Mode Slave**
  The Advertising policy for the Concert Mode Slave.
- **ADV Policy in Stereo Slave**
  The Advertising policy for the TWS Slave.
- **A2DP Latency**
  To set a2dp latency timer. (Unit: ms)
- **Enable AAC Codec in CSB**
  Enable AAC codec in CSB.

3.15.2 Twin Speaker Link Sound Settings

- **Twin Speaker Link Sound Channel**
  The sound channel setting is for the Twin Master and the Twin Slave.

3.15.3 Twin Speaker Pairing Success for Twin Master

- **At Bluetooth Standby State**
  This setting determines which state the Speaker will be if the Speaker as Twin Master has no Bluetooth link and processes Twin Speaker Pairing successfully.
- **At Bluetooth Connected State**
  This setting determines which state the Speaker will be if the Speaker as Twin Master is connected with the remote device and processes Twin Speaker Pairing successfully.
- **At Bluetooth Link Back State**
  This setting determines which state the Speaker will be if the Speaker as Twin Master is reconnecting to the remote device and processes Twin Speaker Pairing successfully.
- **At Bluetooth Pairing State**
  This setting determines which state the Speaker will be if the Speaker as Twin Master is at Pairing Mode and processes Twin Speaker Pairing successfully.

3.15.4 Twin Speaker Pairing Failed for Twin Master

- **At Bluetooth Standby State**
  This setting determines which state the Speaker will be if the Speaker as Twin Master has no Bluetooth link and fails to create the Twin Speaker Link

- **At Bluetooth Connected State**
  This setting determines which state the Speaker will be if the Speaker as Twin Master is connected with the remote device and fails to create the Twin Speaker Link.
- **At Bluetooth Link Back State**
  This setting determines which state the Speaker will be if the Speaker as Twin Master is reconnecting to the remote device and fails to create the Twin Speaker Link.
- **At Bluetooth Pairing State**
  This setting determines which state the Speaker will be if the Speaker as Twin Master is at Pairing Mode and fails to create the Twin Speaker Link.

3.16 Security

- **Enable Simple Pairing**
  Enable or disable Secure Simple Pairing (SSP). If Speaker supports Bluetooth version 2.1 and later ones, it won't be needed to enter fixed PIN code for pairing.
- **PIN Code**
  This setting is four-byte ASCII code which is the fixed PIN code for pairing. The PIN code is used for pairing when the Secure Simple Pairing (SSP) is disabled.

3.17 Misc. Options

- **Enable Pairing as Standby Mode**
  Force the Speaker to be discoverable when it has no Bluetooth link.
- **Enter Pairing When Power On**
  Enter Pairing Mode automatically when the Speaker is turned on.
- **Suspend Streaming When SCO Established**
  If the Speaker has a HFP link and an A2DP link with different remote devices, suspend audio stream when it is outputting voice.
- **Class of Device**
  The Class of Device is used to indicate the capabilities of the Speaker to other devices.
- **Report Battery Status to Smart Phone**
  Report speaker/hs battery status via AT command base on HFP to mobile phone.
- **Link Application**
  Single link: Only one HF link exists.
  Multi-link: Two HF links exist.
- **Always Answer Incoming Call**
  If this option is enabled and HFP connection exists, it will always answer incoming calls automatically.
- **Auto Answer Incoming Call When Link Back**
  When the speaker links back HF link and call status is incoming call, then it will answer the call automatically if this option is enabled.
- **Hang Up a Call When Switch Off**
  Hang up all calls when the speaker is turned off by slide switch.
- **Shut Down Power in Off State**
  Cut off the power when the speaker enters off state.
- **Enter Pairing When Power on Link Back Failed**
  Enable this option to enter pairing automatically if it fails to reconnect to phone when power on.
- **Only Accept Paired Device**
  If this option is enabled, the speaker only accepts the connect request from paired devices when it is not in pairing mode.
- **Disconnect All In Pairing**
  Enable/Disable disconnect all link when SPK in pairing state.
- **Keep BLE In Power Off**
  Enable/Disable keep BLE connection when SPK in power off state.
- **AVRCP Version Option**
  Select to use AVRCP v1.6 or AVRCP v1.3.
- **Auto Unsniff in Data Transmission**
  After system enter sniff mode, it would not auto exit sniff mode if received packets or send packets.

3.18 Introduction to DSP Configuration tool and AEC Tuning Guide

Features

- Supports Acoustic Echo Cancellation (AEC) to cancel the acoustic echo coupled into the micro-phone speaker output.
- Supports Noise Reduction (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/speaker. Defaulted audio effects are also provided to enhance the user experience.
- Supports High Pass Filter (HPF) to provide a low-latency IIR-structured low-pass filter to filter the unwanted low-frequency band for both MIC and SPK paths.
- Supports the over the air (OTA) tuning (dynamic tuning) of the DSP parameters related to audio and voice functions.

Introduction

This section of user guide provides the general information of the IS208x SoC. It also covers noise reduction and acoustic echo cancellation with advanced signal processing techniques, sophisticated voice and audio enhancement functions, filtering, equalizations and audio effect processing. Additionally, for fine tuning the AEC and NR in different applications, during product development. Also, to get guidance on parameters for fine tuning provided in the Digital Signal Processor (DSP) configuration tool in step-by-step format, which allows the system designers to set the DSP features with desired requirements.

The IS208x SoC is designed for high-performance signal processing, which provides excellent voice or audio user experience. The fundamental modules of the IS208x SoC are Filter, Stationary NR, SPK Gain, MIC Gain, AEC/AES and I²S and Sound effects. These two modules are adjusted based on customers requirement. The AEC function is provided at the microphone (uplink) path in the Synchronous Connection-Oriented (SCO) link connection. In addition to NR/AEC function, the audio effect functions, multi-band dynamic-range-compression (MB-DRC), virtual bass (VB) enhancement and (A2DP) audio streaming are used in various applications.

For mono speaker/speakerphone and stereo headset applications, the MB-DRC and VB are enabled to have better audio clarity performance. The AW is enabled for better live surrounding effect for the users.

3.18.1 Signal Processing Flow

Before introducing the signal processing flow, some abbreviations and terminologies are defined in advance to avoid further confusion. In this document, the path that the Bluetooth device receives bitstream and pass to DSP for decoding process, is called as “downlink/downstream/far-end/speaker” path. The path that the Bluetooth device transmits the bitstream, encoded by the DSP processor, is called as “uplink/ upstream/near-end/MIC” path.

Figure 1: Speech Signal Processing

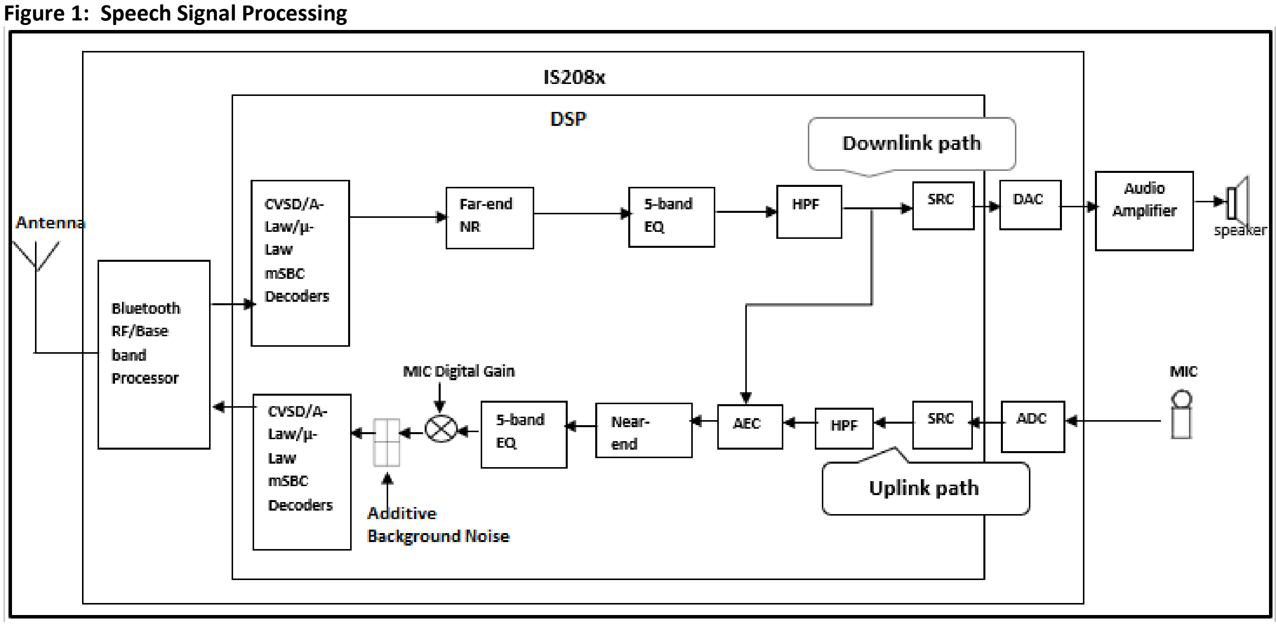


Figure 2: Audio Signal Processing

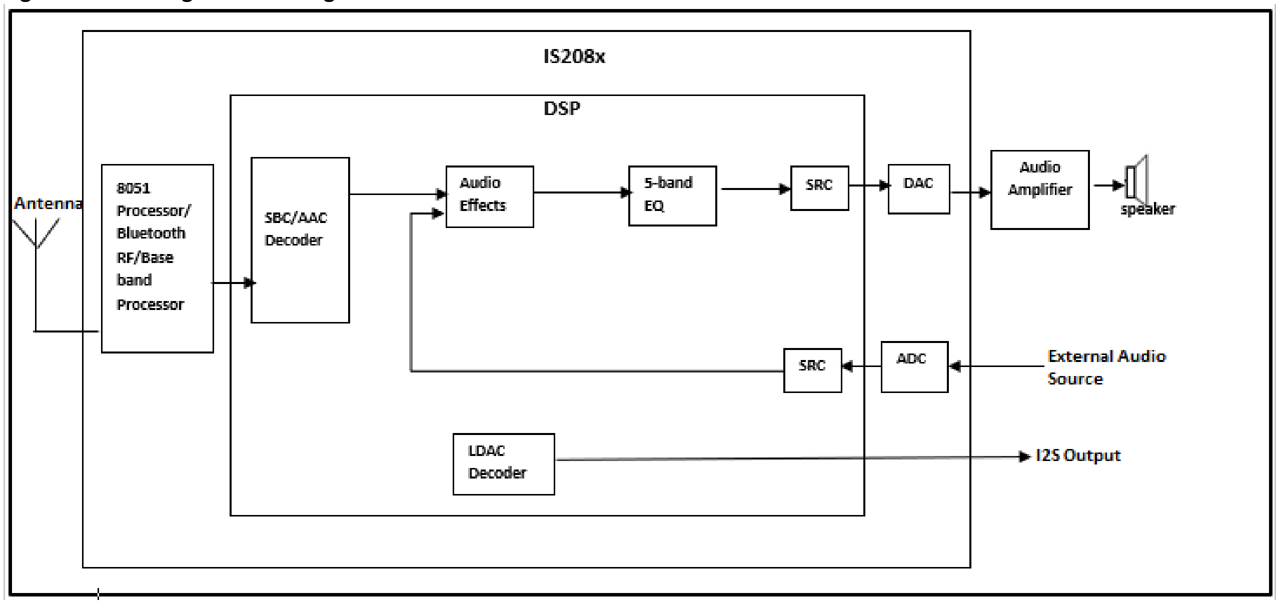


Figure 1 and Figure 2 illustrate the block diagram of the DSP processing flow for speaker-phone and headset applications for speech and audio signal processing. The DSP part is focused on speech encoders/decoders and audio decoders along with their corresponding signal processing functions. The embedded analog-to-digital codec (ADC) provides high signal-to-noise ratio (SNR) data conversion with 90 dB and digital-to-analog codec (DAC) provides high SNR data conversion with 95 dB. The Bluetooth and RF/modem processors deal with the medium access control (MAC) and the wireless data transmission. For the speaker and speaker-phone applications, the external audio amplifiers are used 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, as illustrated Figure 1. mSBC is the mandatory speech codec defined in Bluetooth Hands-free profile (HFP) 1.6 to provide HD voice quality. If the host device establishes the Bluetooth link through HFP 1.6 profile, mSBC is the main speech codec regardless of the cellular network conditions, i.e. 3G network or VoLTE.

In Figure 2, DSP includes MB-DRC, VB and AW functions followed by the 5-band EQ for the audio effect. The digital line-in Loop-back mode allows to pick up the signal from external audio source and feedback into the DSP for audio processing, which goes through the audio effect and 5-band EQ functions, as the Bluetooth A2DP audio streaming. IS208x family support both SBC, AAC-LC, LDAC decoder. For specific details refer to IS208x Silicon Datasheet. However, for legal usage of AAC-LC decoder, the license from "Via Licensing" is necessary before the product is launched to the market.

3.18.2 DSP Configuration Tool

After launching configuration tool and enabling necessary features in DSP Feature Setup page and CODEC Setup page, click on Finish button to launch the DSP Configuration tool as shown in Figure 3(a). Figure 3(b) shows the DSP GUI configuration tool window with various tabs: Main Function, Voice Function, Audio Function, and I^{2}S/PCM.

This tool allows configuring the NR/AEC/EQ/FIR functions.

This tool has two modes one is static mode and other is dynamic mode. In static mode tool will not be connected over the BLE to any DUT but in dynamic mode it will be connected over BLE to DUT for Over the Air (OTA) tuning of DSP parameters.

In tool at the bottom you can see following options for Save and Reset buttons. As shown in Figure 3(c).

- Save button — It can be used to save the current settings present in the Graphic User Interface (GUI) of tool for later reload when required and transform them to the hex file format.
- Reset button — It can be used to load back or reset parameter values to saved values, pass the information of the DSP-related parameters saved in the tool and display its corresponding values automatically in all other tabs of tool.

Save and Reset button will not be visible in dynamic mode and they are visible only in static mode. You can switch between static and dynamic modes in between as many times as you want.

Figure 3(a): Configuration Tool

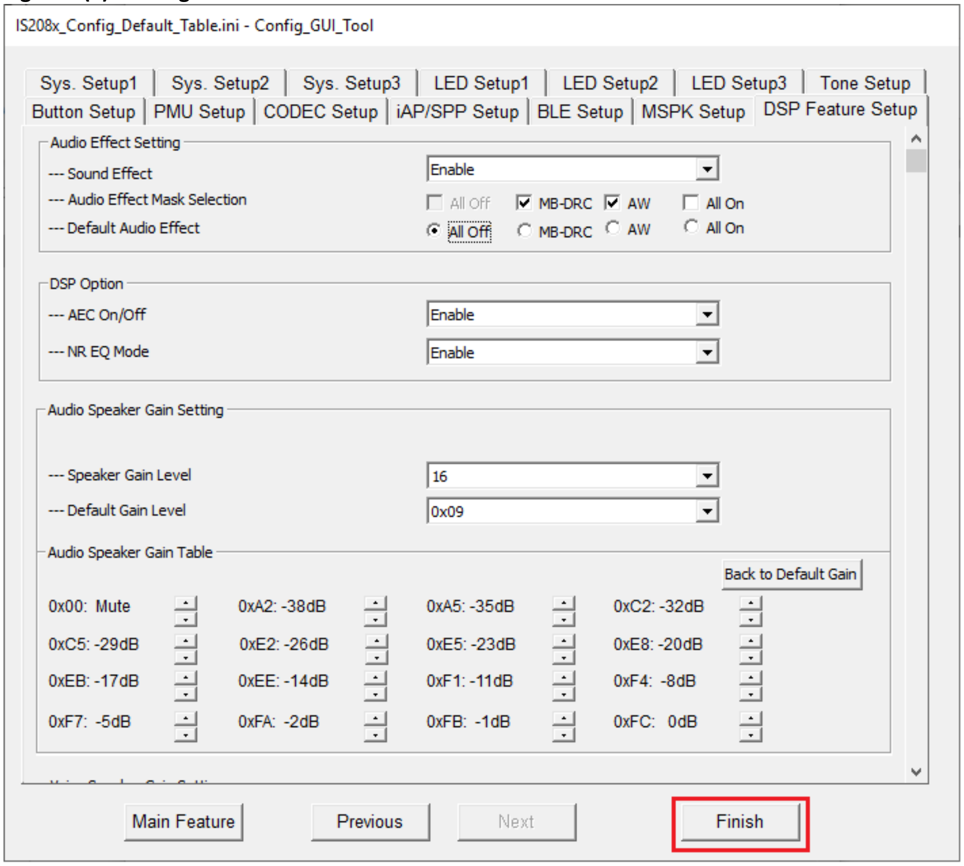


FIGURE 3(b): DSP GUI Configuration Tool

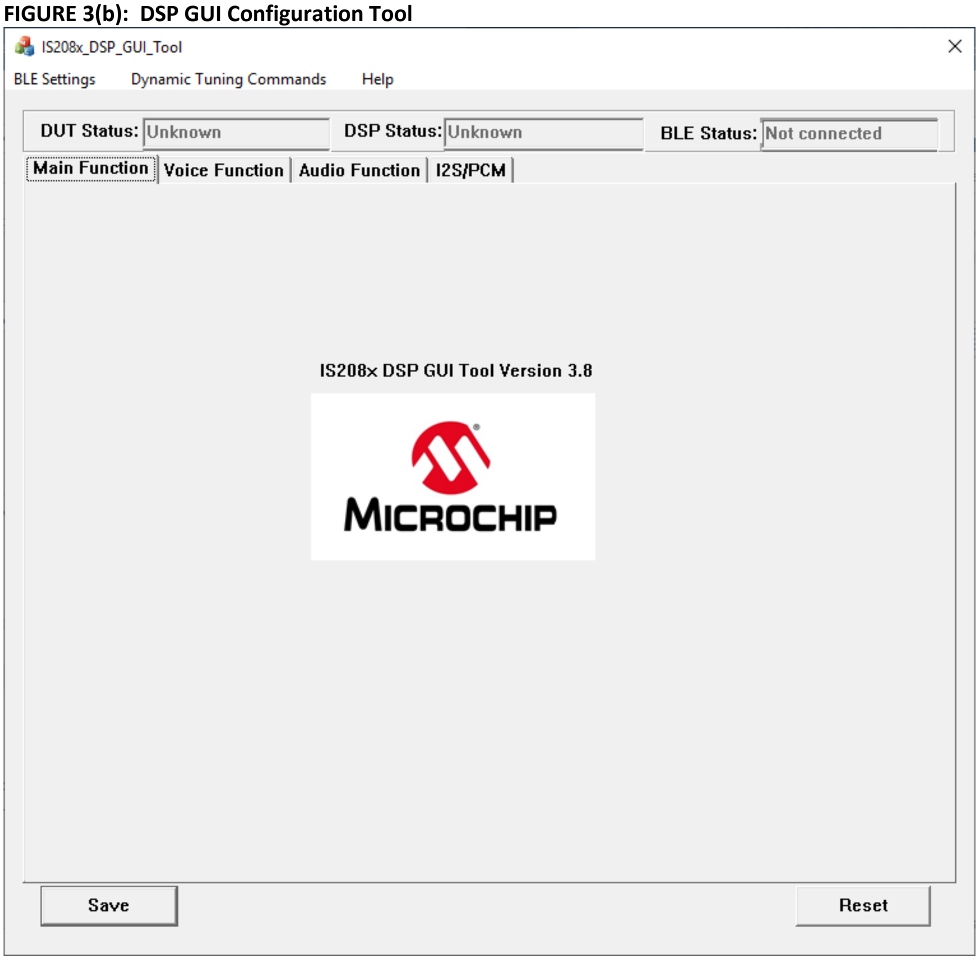
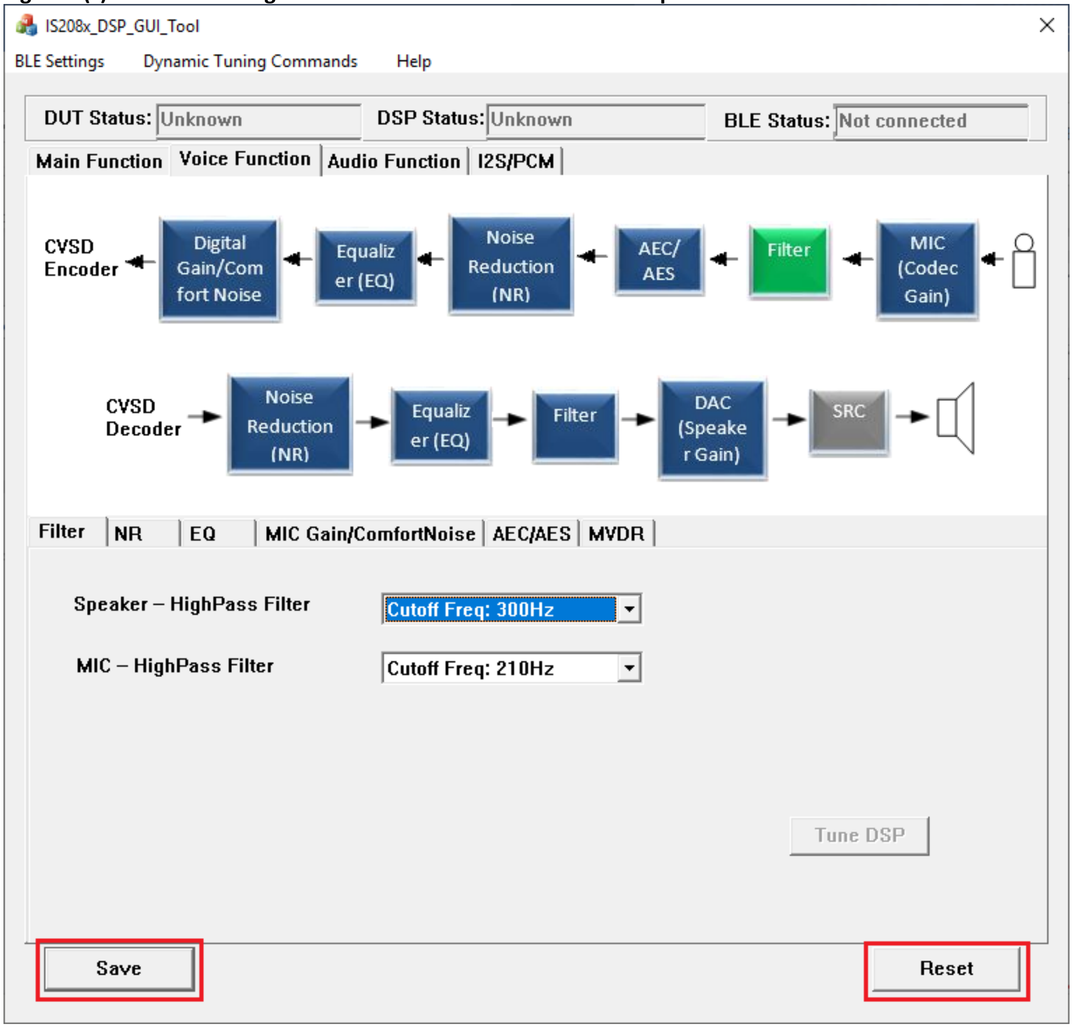


Figure 3(c): DSP GUI Configuration Tool Save and Reset Buttons Options.



3.18.3 Voice Processing Functions

The Voice functions are used to suppress ambient noise, echo and other call interferences during voice calls using following features to increase the quality of voice.

- High-pass filter for SPK/ MIC paths (HPF) — Remove low-frequency PCB/background noise.
- Noise Reduction (NR).
- Acoustic Echo Cancellation (AEC)/ Suppression (AES) — 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.

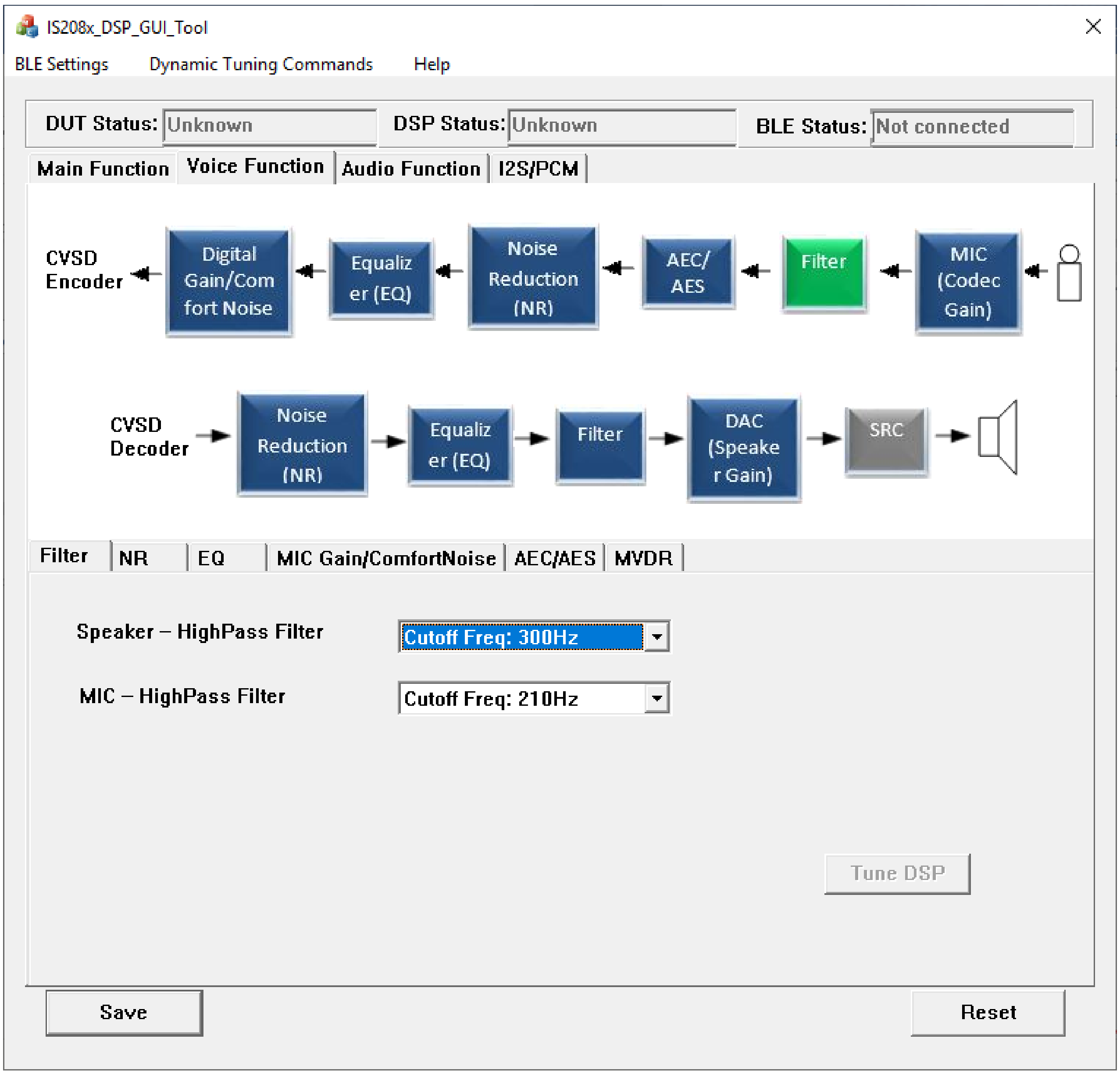
3.21.3.1 High-Pass Filter

The high-pass filter (HPF) provides an option for a low latency and infinite impulse response (IIR) filtering.

The main function of the HPF contains seven selectable cutoff frequencies used to filter out the unwanted low-frequency signals, such as PCB noise, coupled current noise, and wind noise etc. It is a trade-off between the speech signal quality and the noise reduction level. The default cutoff frequencies for the speaker is 300 Hz and MIC is 210 Hz.

Figure 4 illustrates the interface of HPF parameters for both speaker and microphone paths in the DSP tool.

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



Settings for HPF are provided in Table 3-1.

Table 3-1: Configuring HPF Function.

Functionality	Default	Settings
Near-end (MIC path) HPF	0x04: 210 Hz	0x00: 50 Hz 0x01: 80 Hz 0x02: 120 Hz 0x03: 180 Hz 0x04: 210 Hz 0x05: 300 Hz 0x06: 400 Hz
Far-end (SPK path) HPF	0x05: 300 Hz	

3.21.3.2 Noise Reduction

The noise reduction (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.

Figure 5 illustrates the interface of the NR and Dual Mic NR parameters in the DSP tool.

Figure 5(a): Enabling NR Feature in Config Tool to Enable NR In DSP GUI Tool.

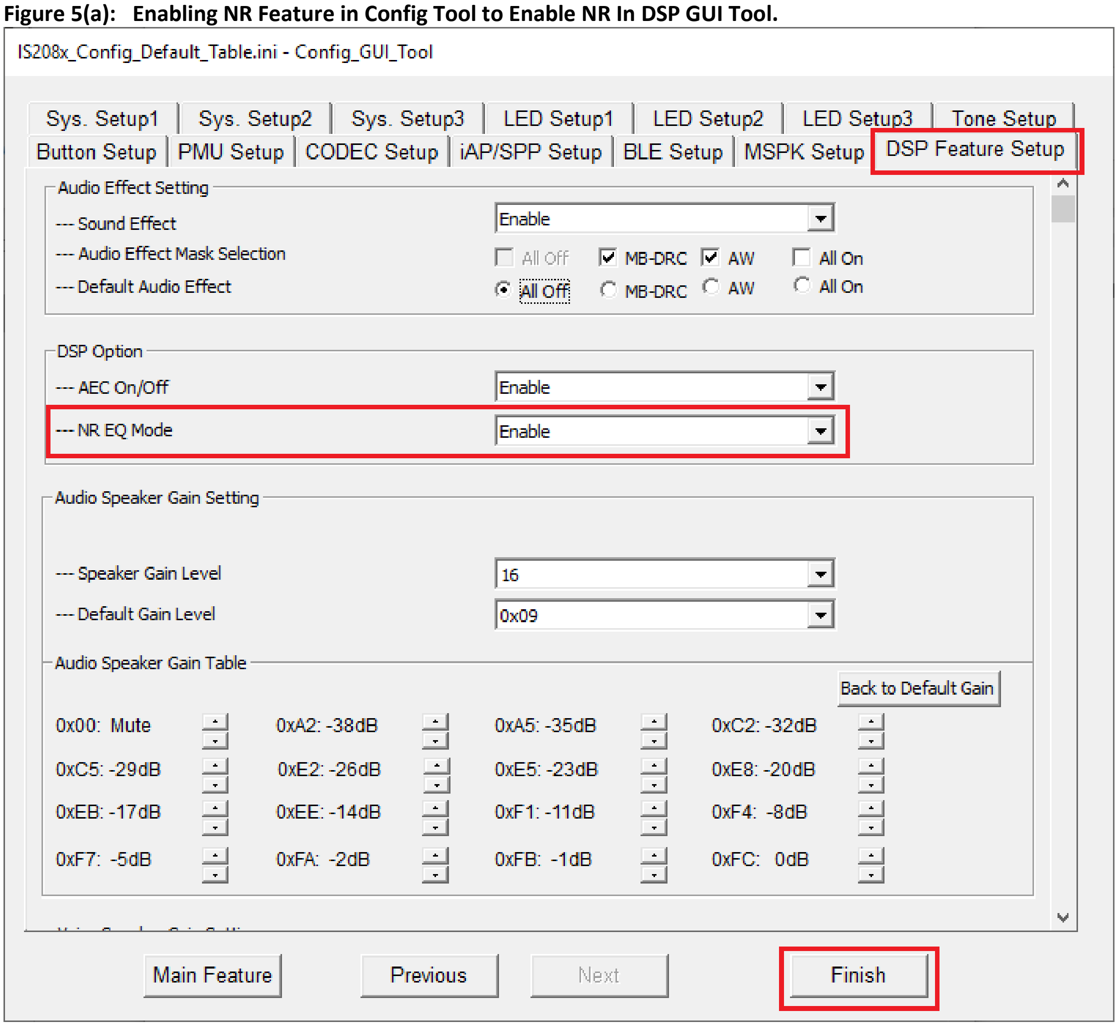
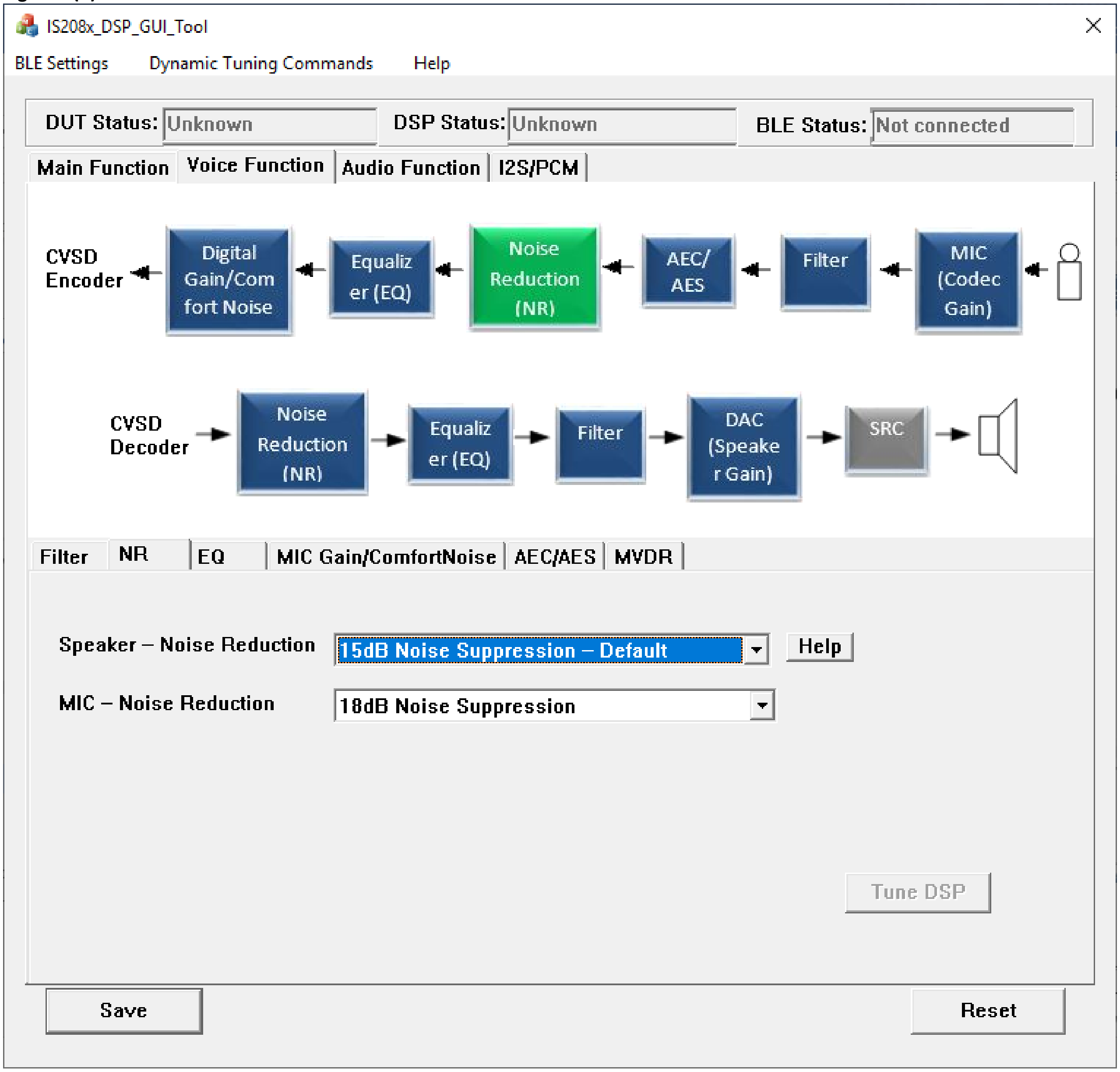


Figure 5(b): Interface of NR Parameters in DSP GUI Tool.



1. NR Suppression Level

Two selectable parameters for NR configurations, illustrated in Figure 5, are suppression levels for low frequency (<1000 Hz) and high frequency (from 1000 Hz to 4000 Hz) with single selection. The tunable range for both speaker and microphone paths is from 0 dB to 21 dB. However, in the DSP configuration tool, only the low-frequency NR suppression level is provided, while the high-frequency suppression levels are determined empirically based on the field test results.

Table 3-2 provides the NR noise suppression levels for selection.

Table 3-2: Configuring NR Function

Functionality	Default	Coefficient level	Settings
Far End (SPK Path) NR	0x16: 15 dB suppression	Low	NR off=>127, 6dB=>64 9dB=>45 12dB=>32 15dB=>22 18dB=>16 21dB=>11 24dB=>8 27dB=>6 30dB=>4
		High	NR off=>127, 6dB=>127 9dB=>64 12dB=>45 15dB=>32 18dB=>22 21dB=>16 24dB=>11 27dB=>8 30dB=>6
Near End (MIC Path) NR	0x10: 18 dB suppression	Low	NR off=>127, 6dB=>64 9dB=>45 12dB=>32 15dB=>22 18dB=>16 21dB=>11 24dB=>8 27dB=>6 30dB=>4
		High	NR off=>127, 6dB=>127 9dB=>64 12dB=>45 15dB=>32 18dB=>22 21dB=>16 24dB=>11 27dB=>8 30dB=>6

3.21.3.3 Echo Cancellation

To linearly cancel and suppress, the returned echo, both acoustic echo canceller (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 full-duplex speech communication is achieved easily. 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 input signal mixed with desired signal and unwanted echo signal in the frequency domain based on the information of voice activity detection. Therefore, select the parameters for AES and Double-Talk Threshold carefully to prevent from the degradation of desired speech quality in the case of strong nonlinear echo, or close placement between the microphone and speakers. Figure 6 illustrates the AEC tuning interface in the DSP configuration tool and DSP GUI Tool.

Figure 6(a): Enabling AEC Tuning Interface in Config Tool.

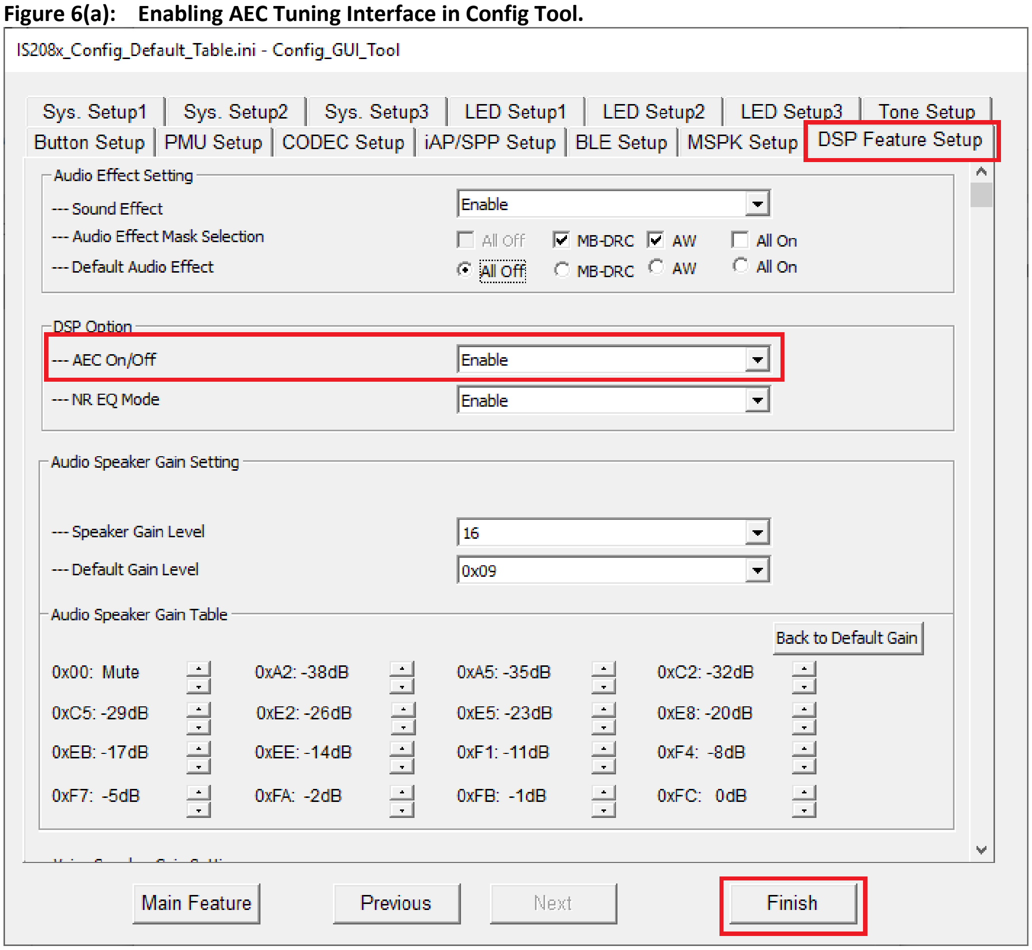
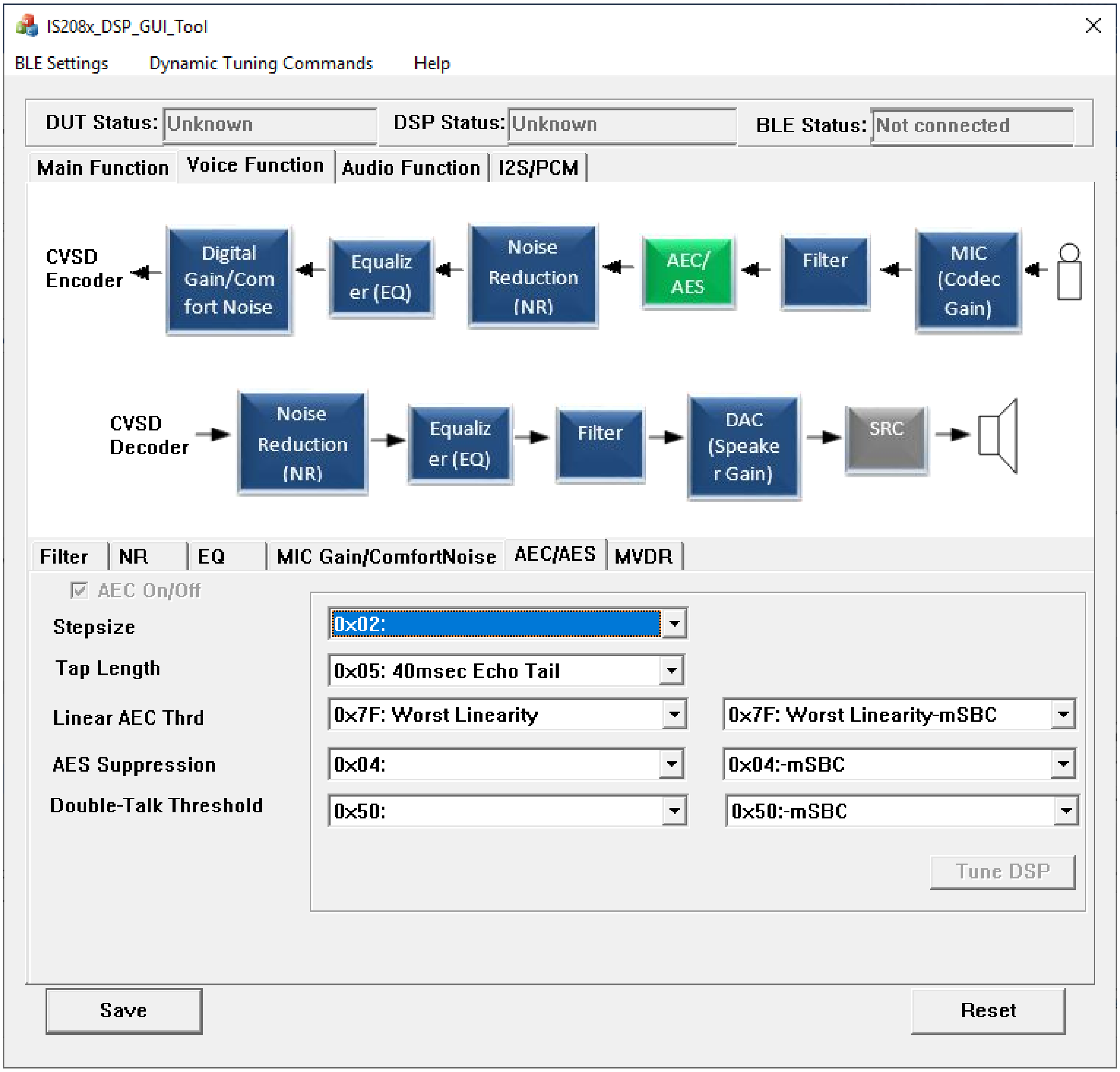


Figure 6(b): AEC Tuning Interface in DSP GUI Tool



The adjustable parameters to fine tune the performance of echo cancellation is provided as follows:

1. AEC Step size

The AEC convergence speed. If fast convergence rate is selected, then lower linear echo cancelation capability. On the other side, if slower convergence rate is selected it takes more time to cancel the echo but with better echo cancellation performance.

2. AEC Tap Length

The tap length must be selected in which the tap length must be longer than the echo tail and it is the trade-off between echo cancellation (EC) performance and million-instructions per second (MIPS). If longer the tap length, then higher the MIPS and DSP power consumption.

Basically, this parameter maps to a threshold that controls the AES to suppress the echo nonlinearly. If this parameter is configured to be more favorable for half-duplexity, then double-talk capability gets degrade more, but removes more residual echo. If we assume that x[n] is near-end signal and y[n] is far-end signal, then the cross-correlation equation between these signals are indicated as follows:

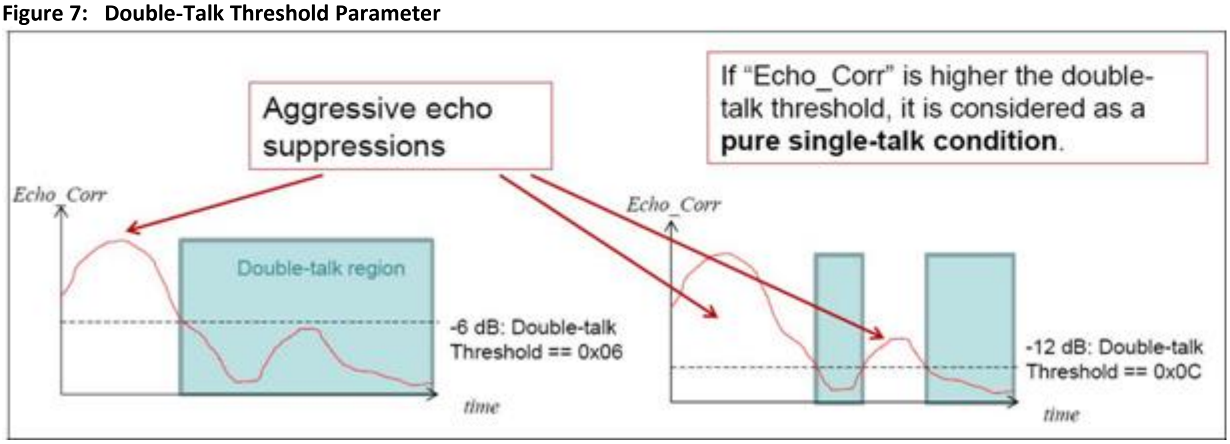
$$Echo\_corr = \mathrm{EMA}(corr\_coef)$$

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

3. Double-Talk Threshold

The full-duplexity of the AEC means the level of how much is the far-end user can listen to the near-end user voice, while both users speak simultaneously.

Figure 7: Double-Talk Threshold Parameter



If the Double-Talk Threshold is selected 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, see Figure 7.

On the other hand, if the Double-Talk Threshold is selected as 0x2D, this setting is more active to enable AES function for echo suppression.

4. AES Suppression

This parameter determines the maximum nonlinear echo suppression capability.

5. Linear AEC Threshold

This parameter determines the linearity threshold of the returned echo. If "Worst linearity" option is selected, then the AES function is configured to easily enable and suppress the residual echo. The "Higher linearity" option allows having better Full-duplex echo cancellation performance with more echo in the returned signal.

The AEC parameters and their corresponding values for selection are provided in Table 3-8.

Table 3-8: Configuring the AEC Parameter

Parameters	Note	Default Value
AEC_Stepsize	0x01: Fastest AEC convergence to 0x06: Slowest AEC convergence	0x02
AECTapLength	0x01: 8 msec Echo Tail to 0x0A: 80 msec Echo Tail	0x05
Double-Talk Threshold	0x7F: Better Full Duplex to 0x1C: Worse Full Duplex	0x50
AES_Suppression	0x01: Less Echo Suppression to 0x10: Most Echo Suppression	0x04
Linear AEC Threshold	0x7F: Worst Linearity to 0x03: Best Linearity	0x7F

3.21.3.4 MIC Digital 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, then the digital boost part of the dynamic MIC control is placed at the end of all digital signal processing modules, as illustrated in Figure 8. The suppressed echo can be amplified by adjusting the MIC digital gain.

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

Figure8(a) illustrates the parameters for MIC gain(codec). Figure 8(b) illustrates the parameters to adjust the configuration of the MIC digital gain and comfort noise.

To enable the MIC Digital gain, configure using given value's in Table 3-9.

Note: For more information about Digital MIC interface with BM83 investigate section 3.22.

Table 3-9: Digital Boost Gain at SCO MIC Path.

Feature	Default	MIC Gain (Digital)
MIC Gain (Digital)	0x1D: 10dB	0x00: -19 dB Digital Boost 0x03: -16 dB Digital Boost 0x05: -14 dB Digital Boost 0x07: -12 dB Digital Boost : 0x13: 0 dB Digital Boost 0x15: 2 dB Digital Boost : 0x27: 20 dB Digital Boost

Figure 8(a): Parameter MIC Gain(codec) in config tool.

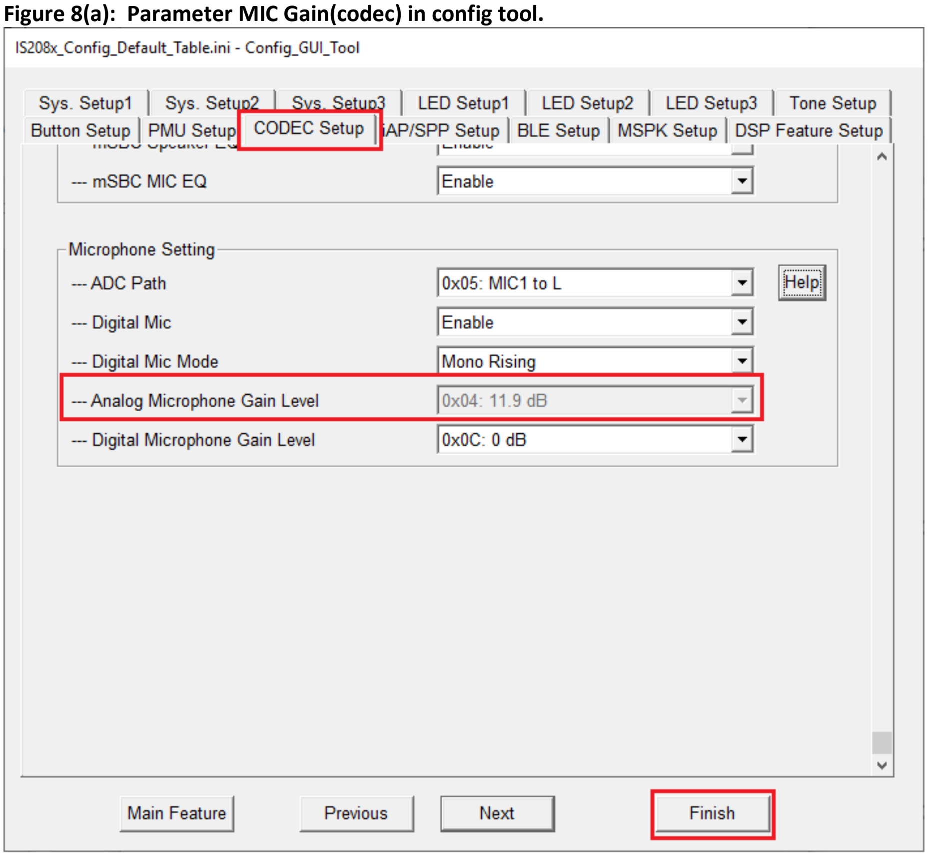
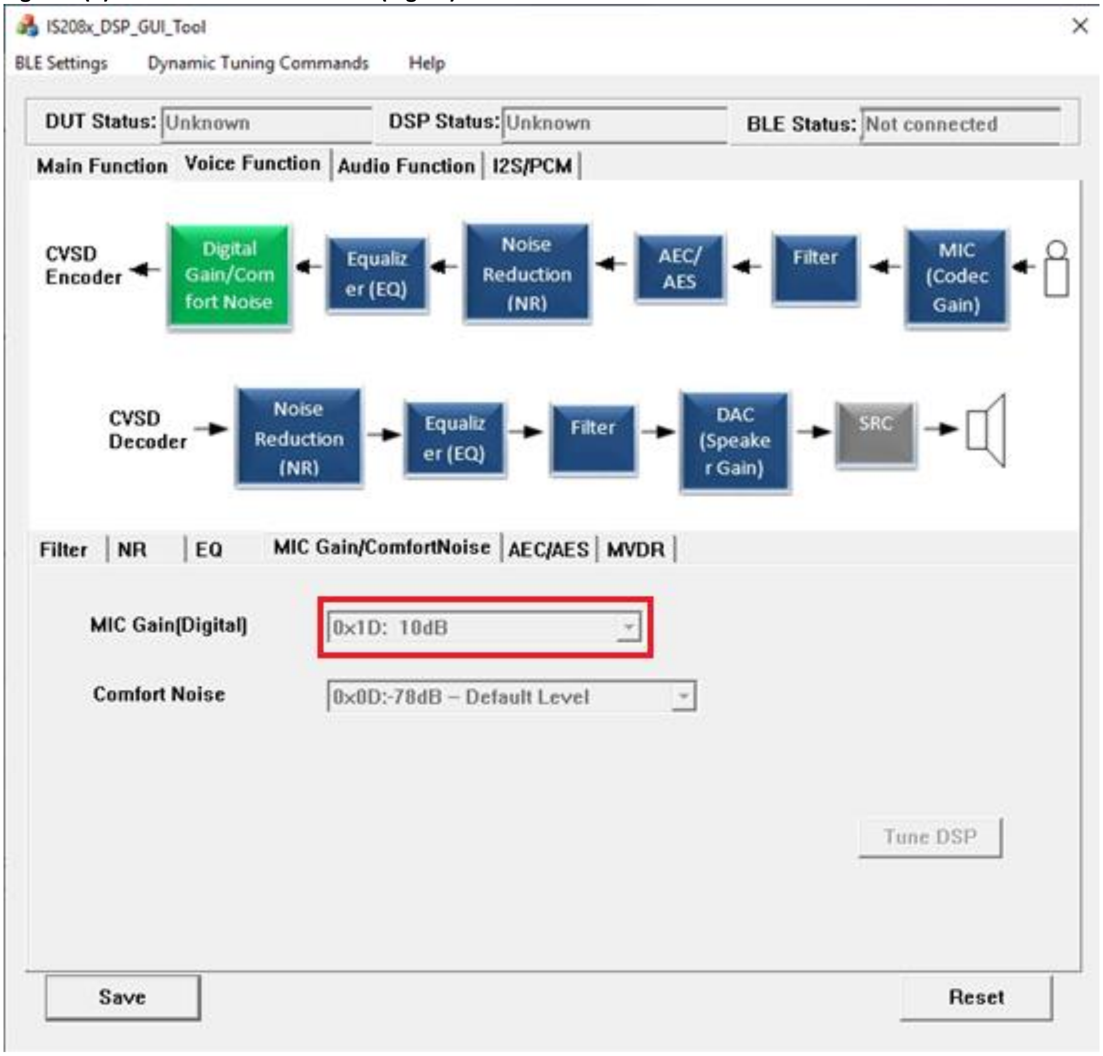


Figure 8(b): Parameters for MIC Gain (digital) in DSP GUI Tool.



3.21.3.5 Comfort Noise

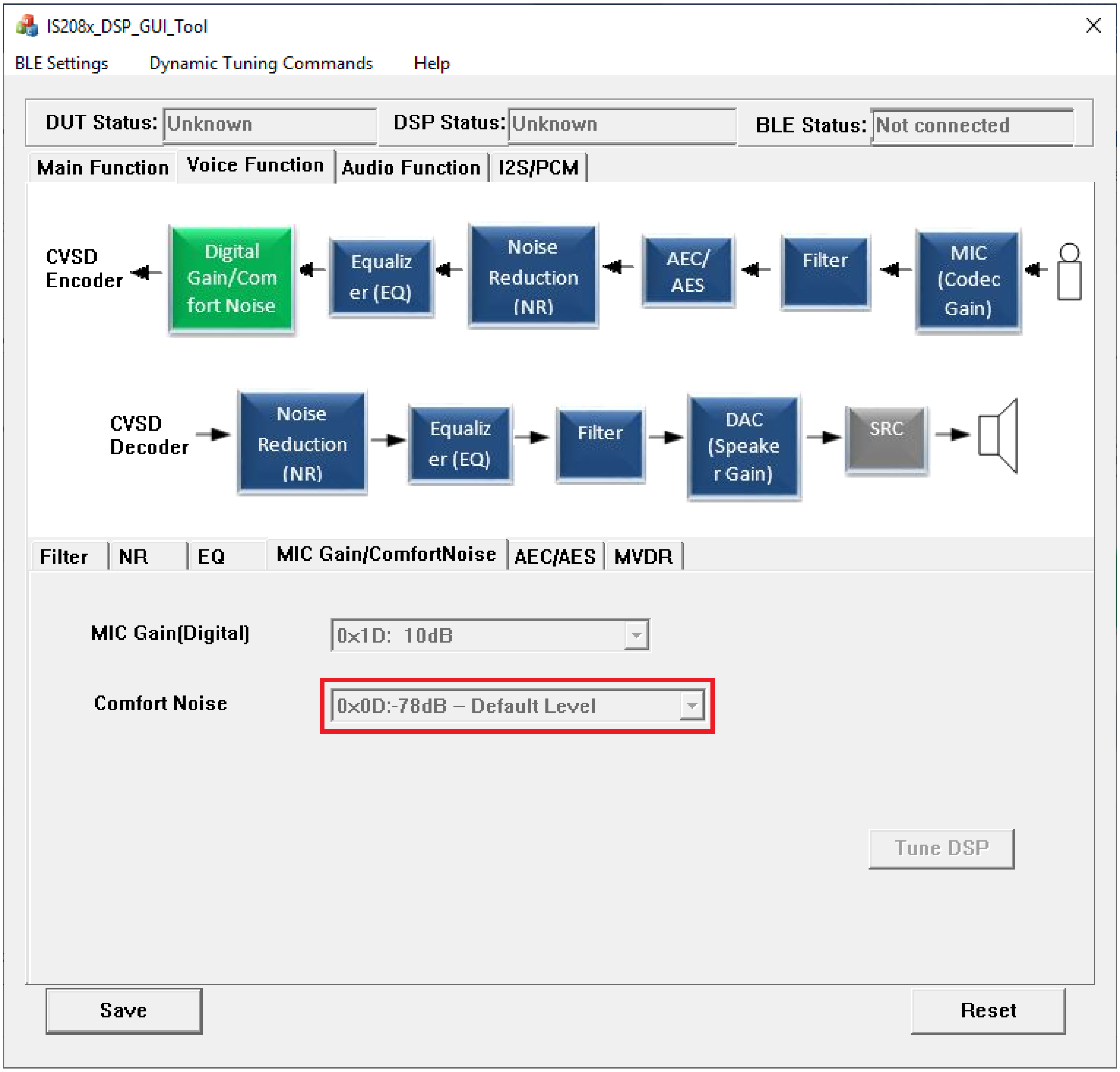
The comfort noise is generated by a random number generator, and its frequency response is constant across all frequencies. The main purpose is to provide a constant noise level to prevent the speech codec algorithm of host cellphones from injecting unwanted noise, which degrades speech clarity at the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

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

Table 3.10: comfort noise values at the SCO MIC path.

Feature	Default value	Range
Comfort Noise	0x0D: – 78dB	0x00: 0dB - Highest Comfort Noise level 0x01: -6dB 0x02: -12dB 0x03: -18dB 0x04: -24dB 0x05: -30dB 0x06: -36dB 0x07: -42dB 0x08: -48dB 0x09: -54dB 0x0A: -60dB 0x0B: -66dB 0x0D: -78dB – default level 0x0E: -84dB 0x0F: -90dB -Lowest Comfort Noise level

Figure 9: Comfort Noise Settings in DSP GUI Tool.



3.21.3.6 Equalizer

The Equalizer (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.

Figure 10 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 Hands-free profile (HFP) 1.6 supporting the wideband (16 kHz sampling rate) speech features. To support HFP 1.6, configure the EQ coefficients separately for 8 kHz and 16 kHz. Click **Custom EQ – MIC** or **Custom EQ – SPK** button to display EQ configuration window, as illustrated in Figure 10.

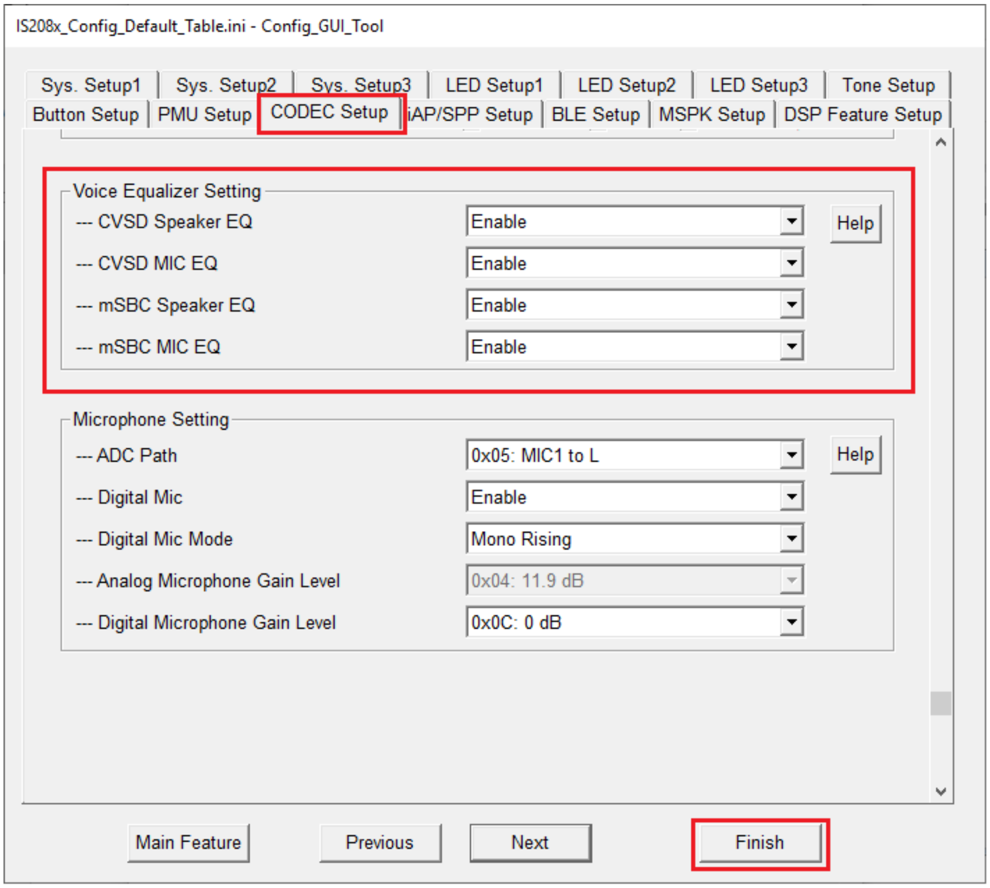


Figure 10(b): EQ Configuration Interface for Speaker and MIC Paths in SCO or SPEECH MODE in DSP GUI Tool.

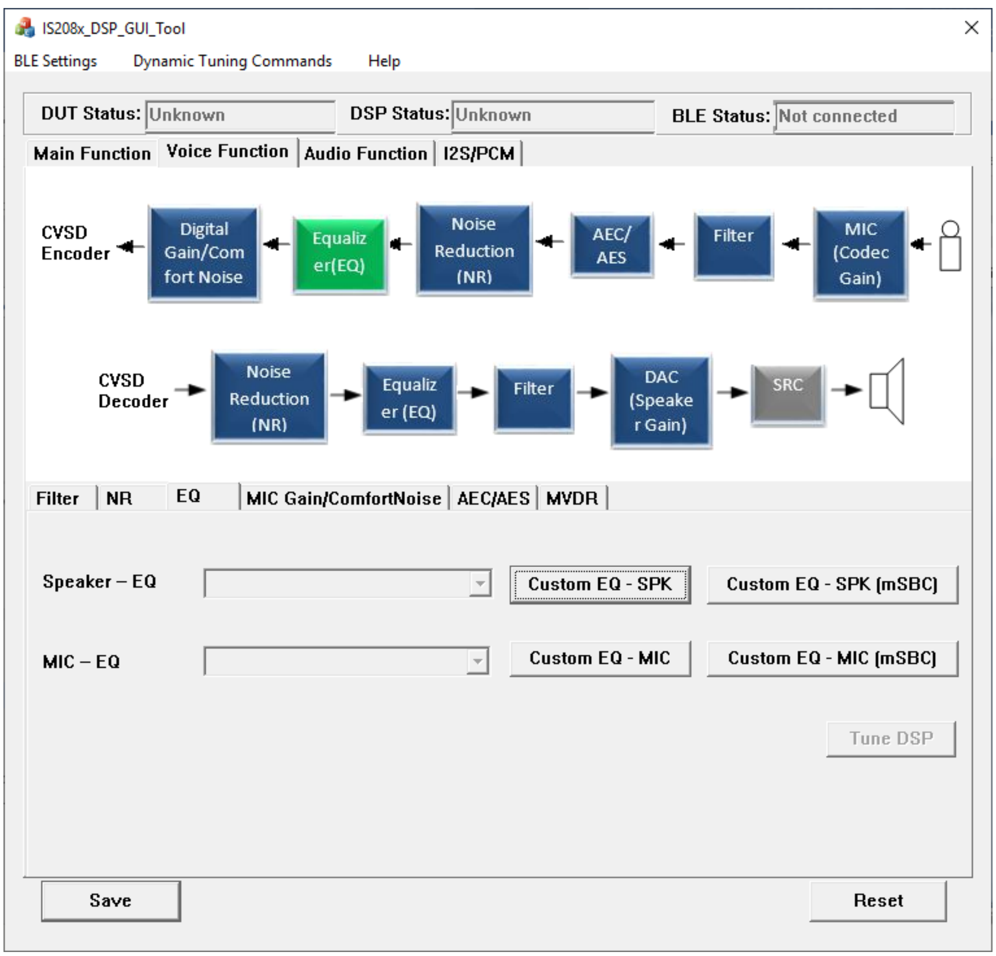
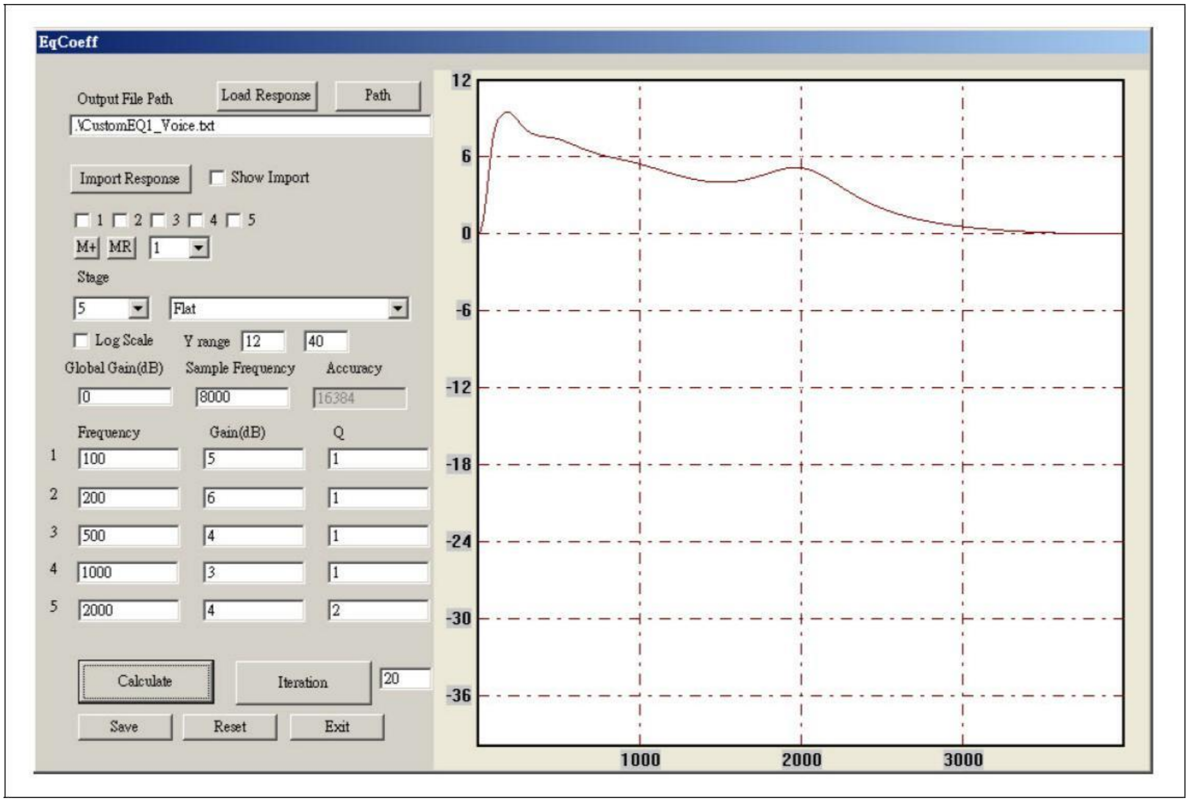


Figure 10(c) illustrates how to configure the Equalizer (EQ) functions.

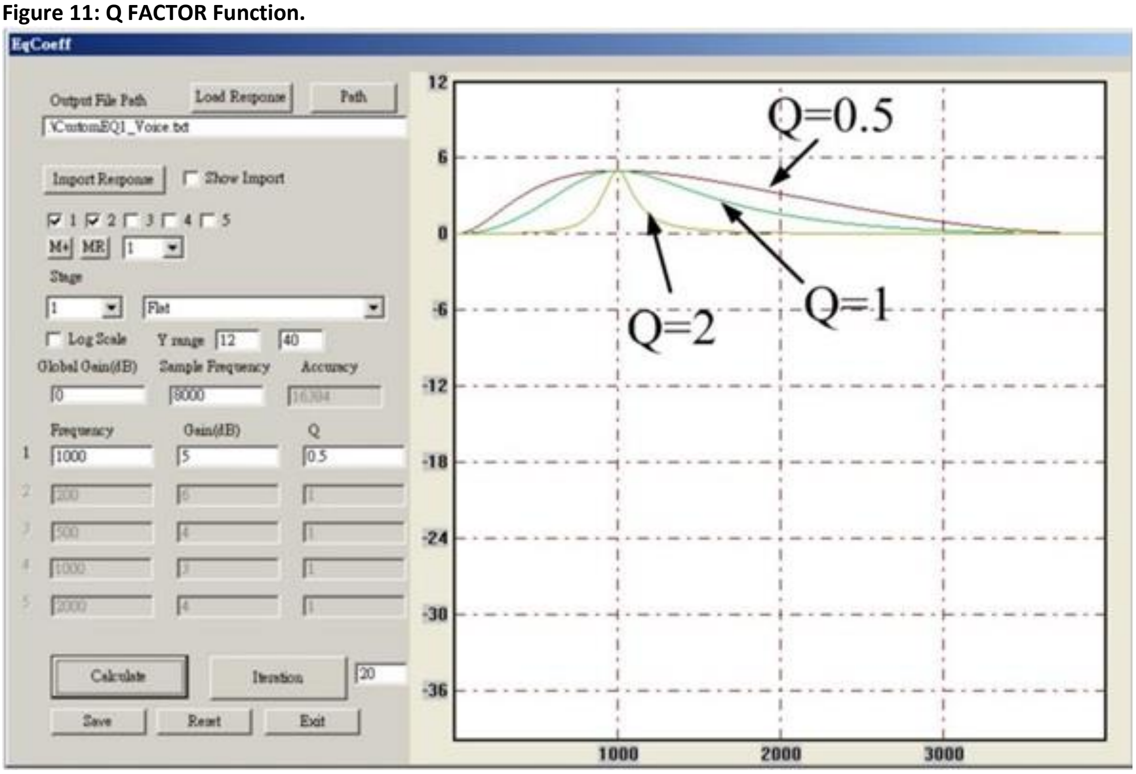
Figure 10(c): Configuring the EQ Functions



Enter the required frequencies and gain/attenuations in the columns. The columns of Quality factor (Q) are to configure the cutoff frequencies of each equalizer band. Consider the value of “Q” is smaller, then wider the bandpass cutoff frequency, as illustrated in Figure-11. 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 **Save** button to save the frequency response that is designed, and the system automatically stores the current EQ's configuration into a file. Click **Load Response** button to restore the frequency response and EQ configurations that was designed previously. The section “Stage” configures the number of bands of 2nd -order IIR filter used. The fewer the “Stages”, then lower the MIPS and power consumption. Figure 11 illustrates the function of the Q factor.

Figure 11: Q FACTOR Function.



3.21.3.7 Speaker/MIC Gain Settings

The speaker gain levels and MIC gain levels are configured in the config tool. There are three different number of speaker gain levels, which are selectable based on requirements. Once the number of the speaker gain level is determined, ensure to choose the corresponding gain for each level.

The gain difference between each MIC gain level for “MIC Gain (codec)” is ranging between 2.7 dB to 3.4 dB per step, as illustrated in Figure 12.

Figure 12: Interface for the Speaker Gain Configuration in the SCO MODE in Config Tool.

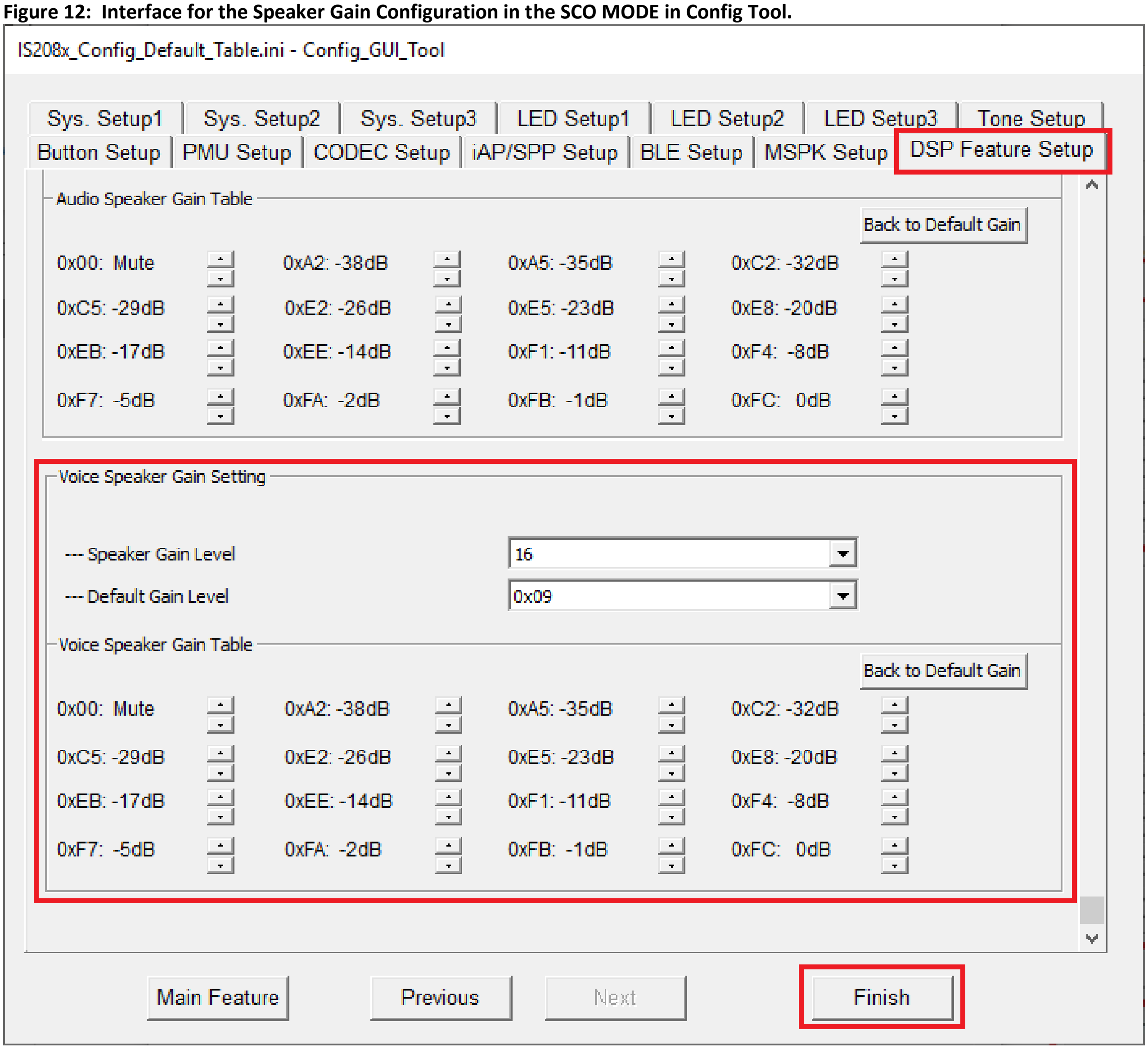
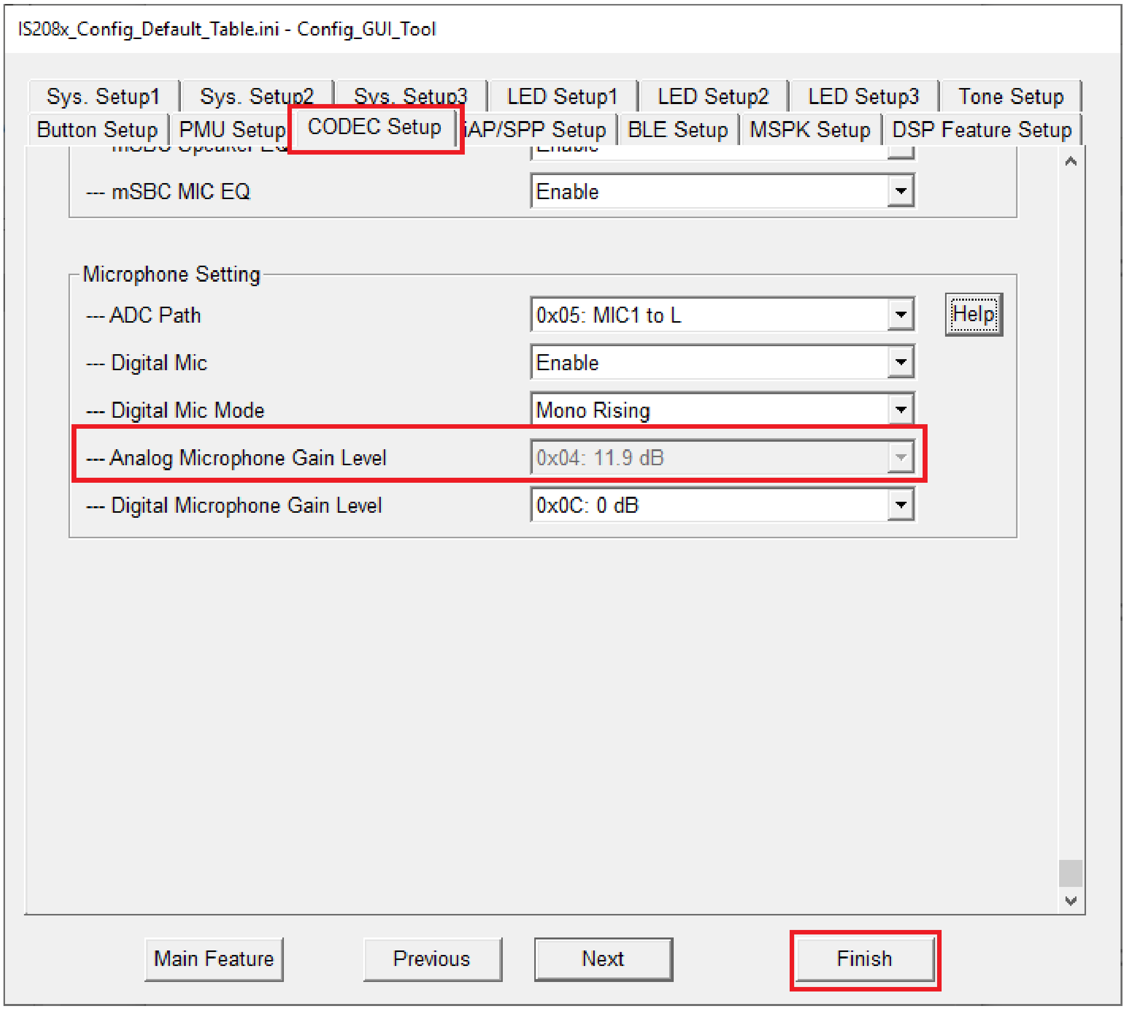


Figure 13 illustrates the MIC gain configuration in the SCO mode.

Figure 13: MIC Gain(codec) Configuration in the SCO MODE in config tool.



3.18.4 Audio Processing Functions

The Audio processing is used to enhance the audio listening experience for following functionalities:

- 5-band EQ for SPK
- Multi-band dynamic range compression (MB-DRC)
- Virtual bass (VB) enhancement — Enhance the bass using psychoacoustic algorithm
- Audio widening (AW) process — Make speakers placed close apart sound like being placed far apart from each other

Note: LDAC works in I2S case only. Audio effects like MB-DRC and equalizer are not available in LDAC playback.

3.18.4.1 Audio Equalizer

Figure 2 illustrates the system diagram of the audio signal processing. In addition to the SBC/AAC decoder, audio effect and EQ are allowed to process the audio signal. Figure 14 illustrates the configuration of the EQ for the audio function. In the column "EQ Mask Selection", select the adjustable special audio sound effect. Except the option "Custom EQ", use an external button to select different sound effects. The procedure to configure "Custom EQ 2" is also identical to the EQ, refer to **3.21.3.6"Equalizer"** section for details.

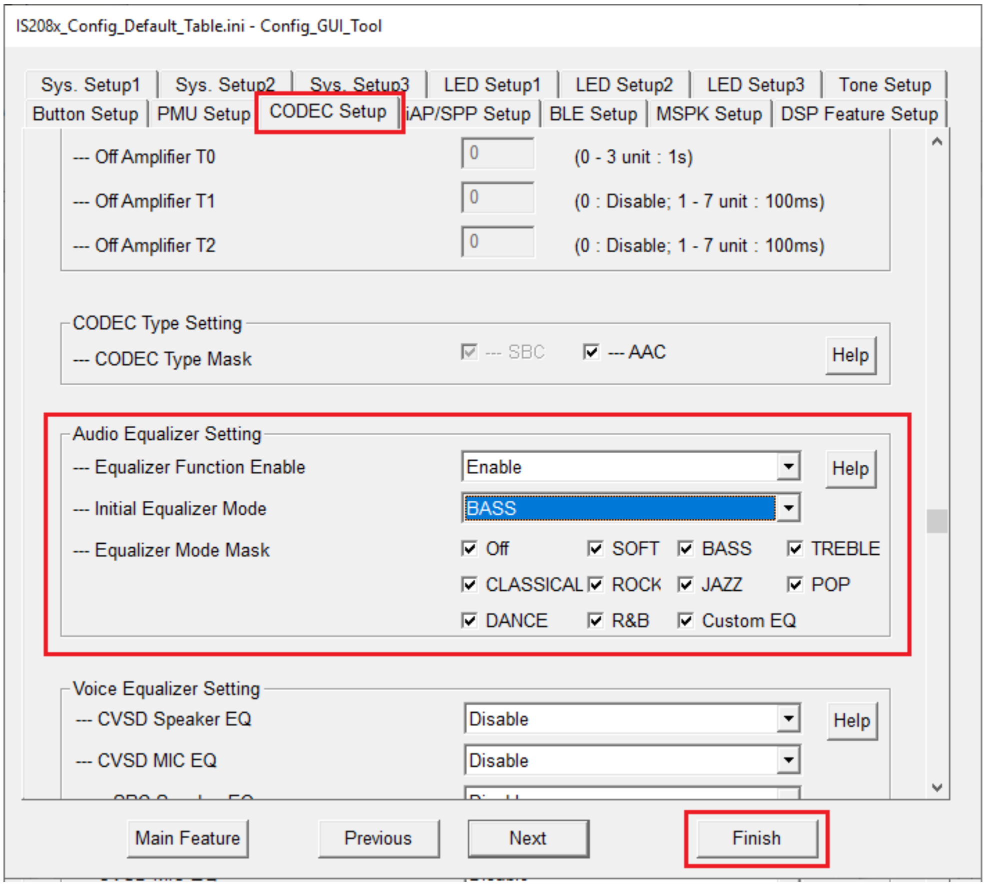
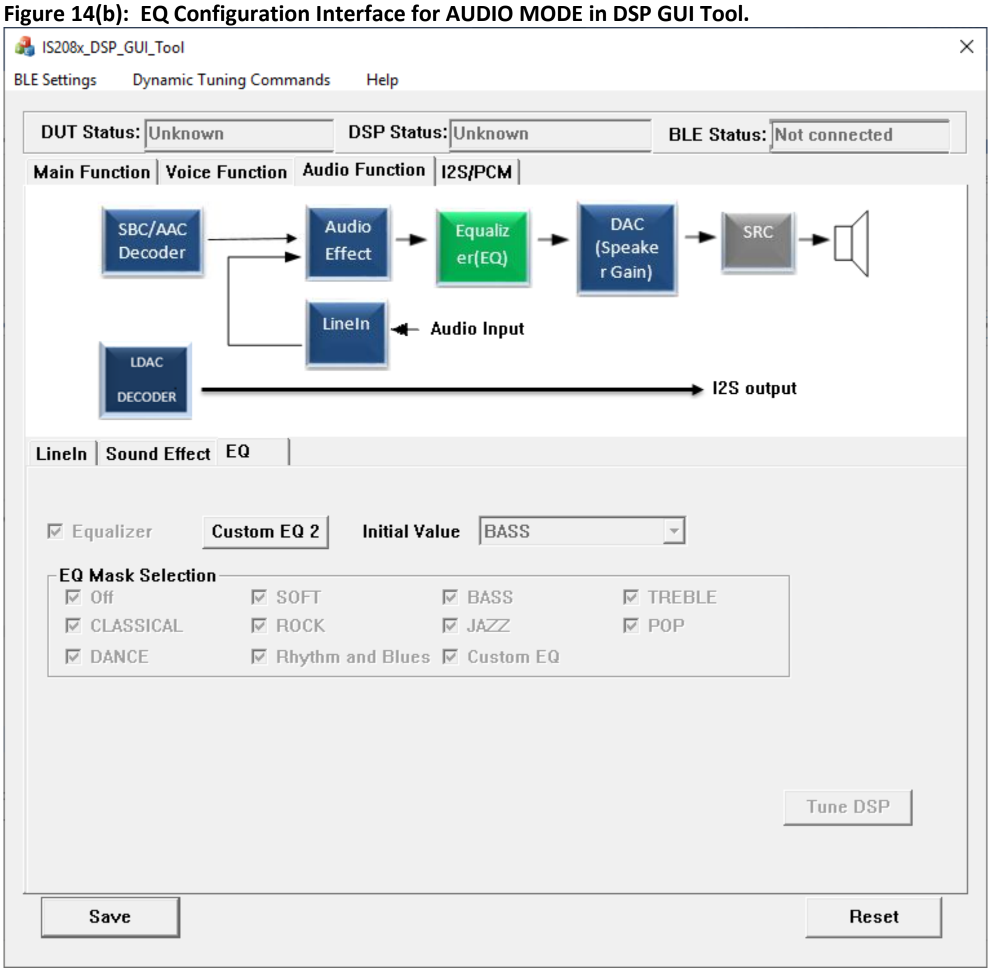


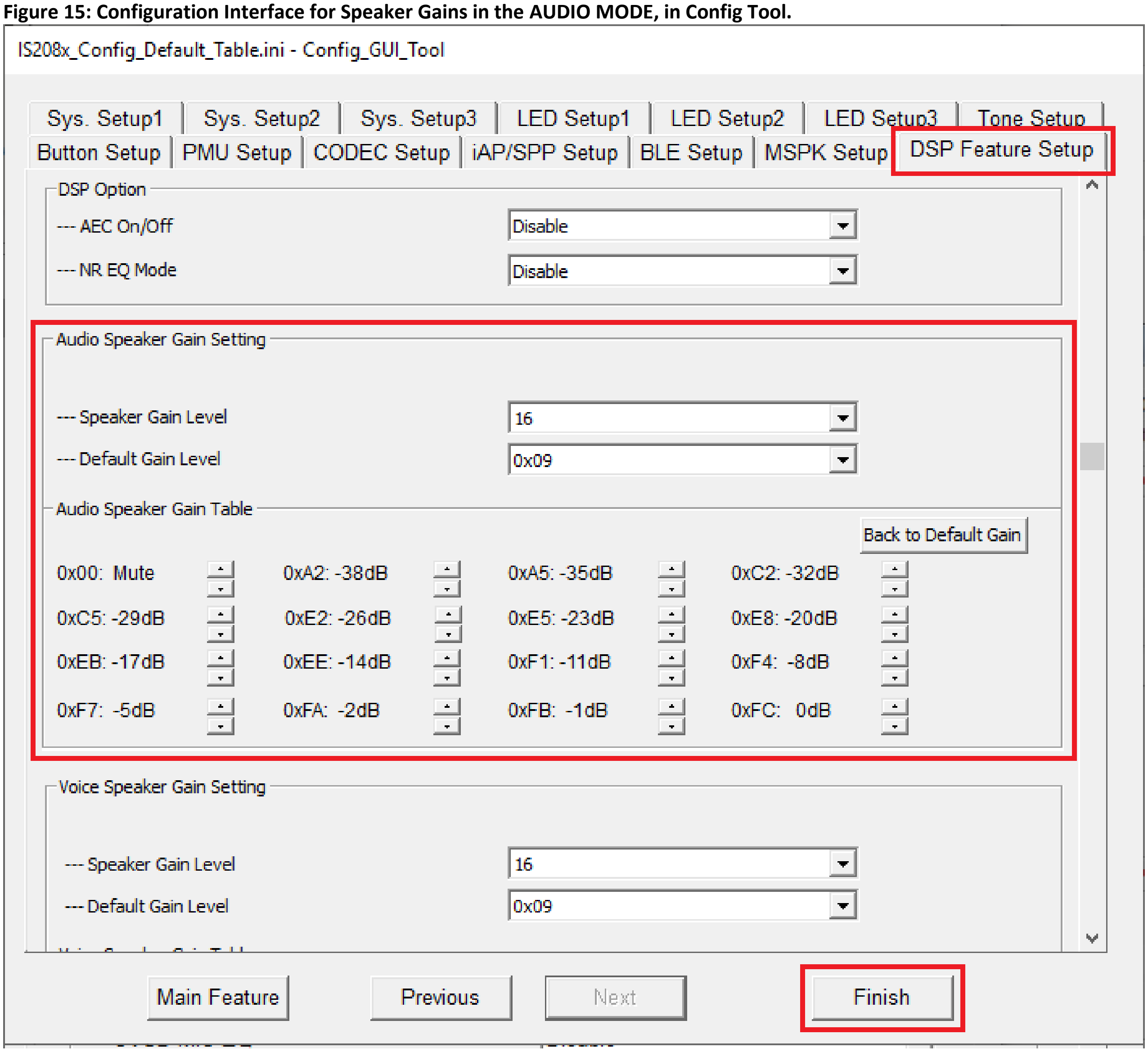
Figure 14(b): EQ Configuration Interface for AUDIO MODE in DSP GUI Tool.



3.21.4.2 Speaker Gain Setting / Line-in Gain

Like speaker setting for voice applications, the number of speaker gain level is also selectable. For selection procedure, refer to **3.21.3.7"Speaker/MIC Gain Settings"**.

Figure 15: Configuration Interface for Speaker Gains in the AUDIO MODE, in Config Tool.



3.21.4.3 Auto Power-off mode for Line-in Silence detection

The DSP function enables the system to detect power level of the line-in signal. The power level of the line-in signal calculated digitally, and the silence status is reported to the Bluetooth MAC controller. The "Initial Line-In SPK Gain" is to configure the line-in gain to amplify the external signal source and playback to the speakers. The "Silence Detection Threshold" determines the silence power threshold level of line-in signal for auto power-off mechanism.

Figure 16(a): Line-In (AUX-In) Configuration Interface in Config Tool.

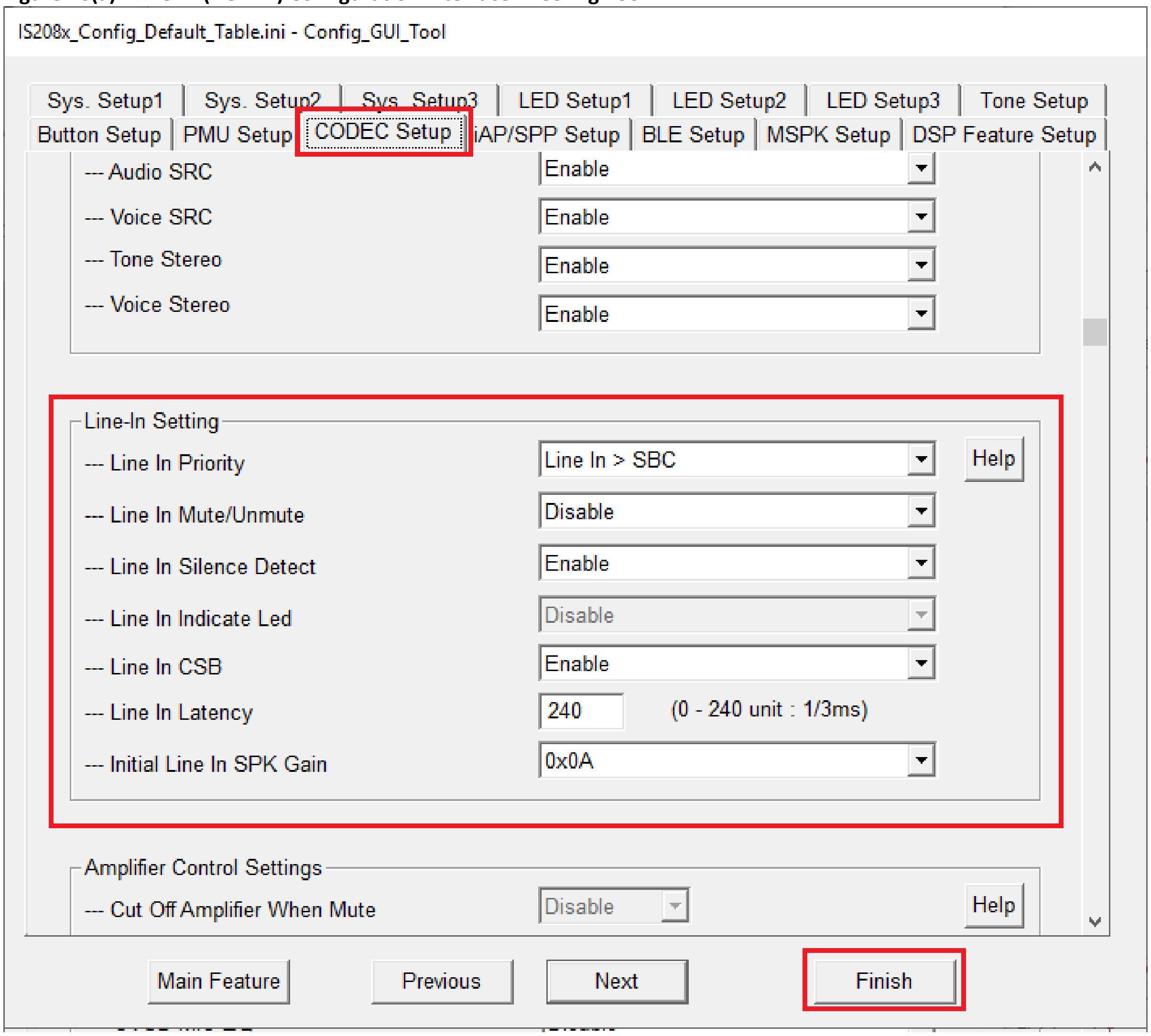
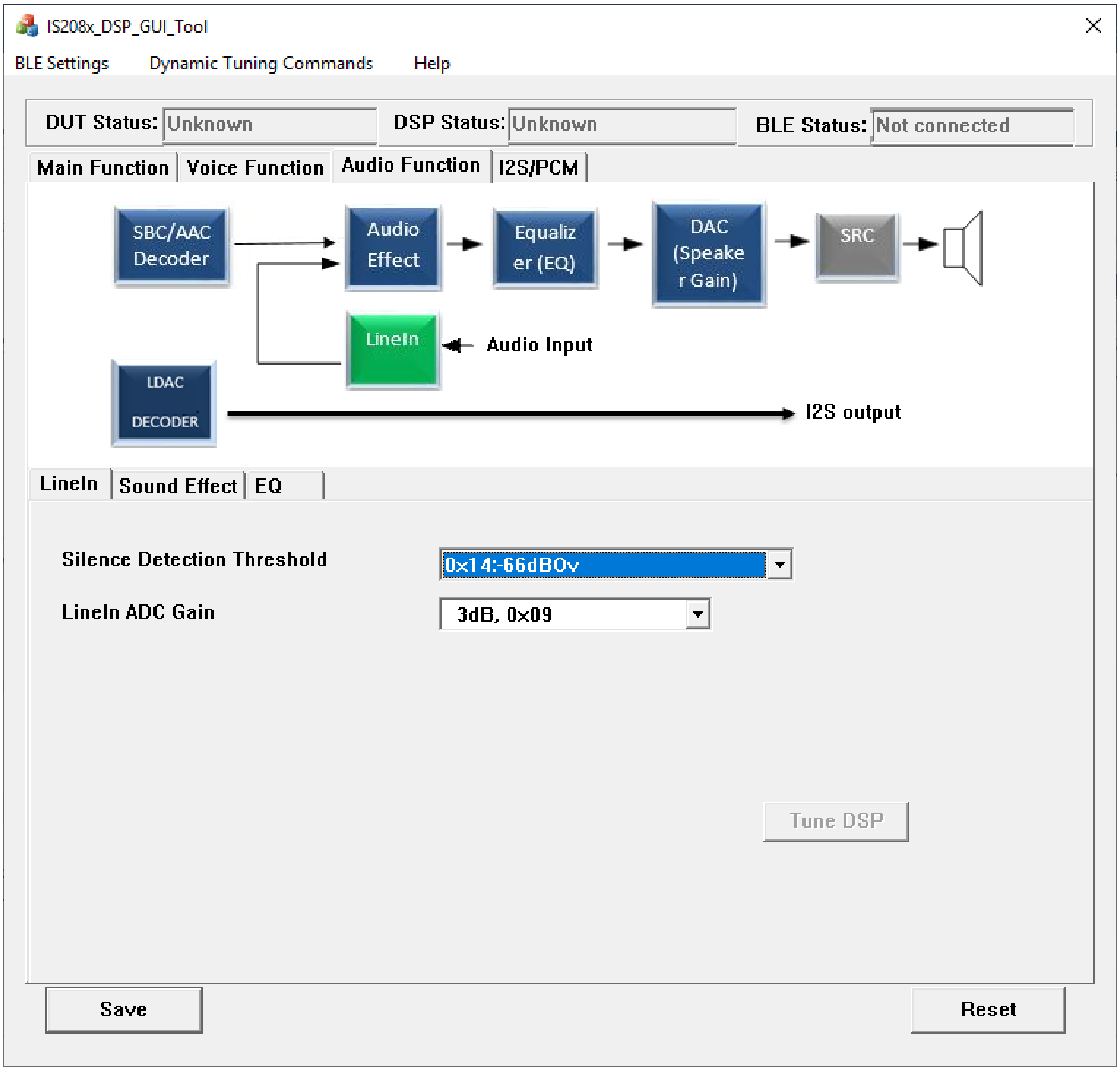


Figure 16(b): Line-In (AUX-In) Configuration Interface in DSP GUI TOOL.



- Parameters of Line-In Silence Detection:

Figure 16(a) illustrates the parameters of Line-in Silence detection.

- Silence Detection Threshold

This parameter determines the silence power threshold level of line-in signal for auto power-off mechanism.

- Initial Line-In Speaker Gain

Firmware initializes with this DAC gain, while playing line-in mode with selected index “0x0A”. The Gain (DB) values for the indexes are mapped to a table extracted from Figure 17. This will increase/decrease the volume in the speaker.

- Line-In Max Level

Firmware uses this level index to constrain maximum DAC gain in line-in mode with selected index “F”. The Gain (DB) values for the indexes are mapped to a table extracted from Figure 17.

- Line-In Min Level

Firmware uses this level index to constrain minimum DAC gain in line-in mode with selected index "0". The Gain (dB) values for the indexes are mapped to a table extracted from Figure 17.

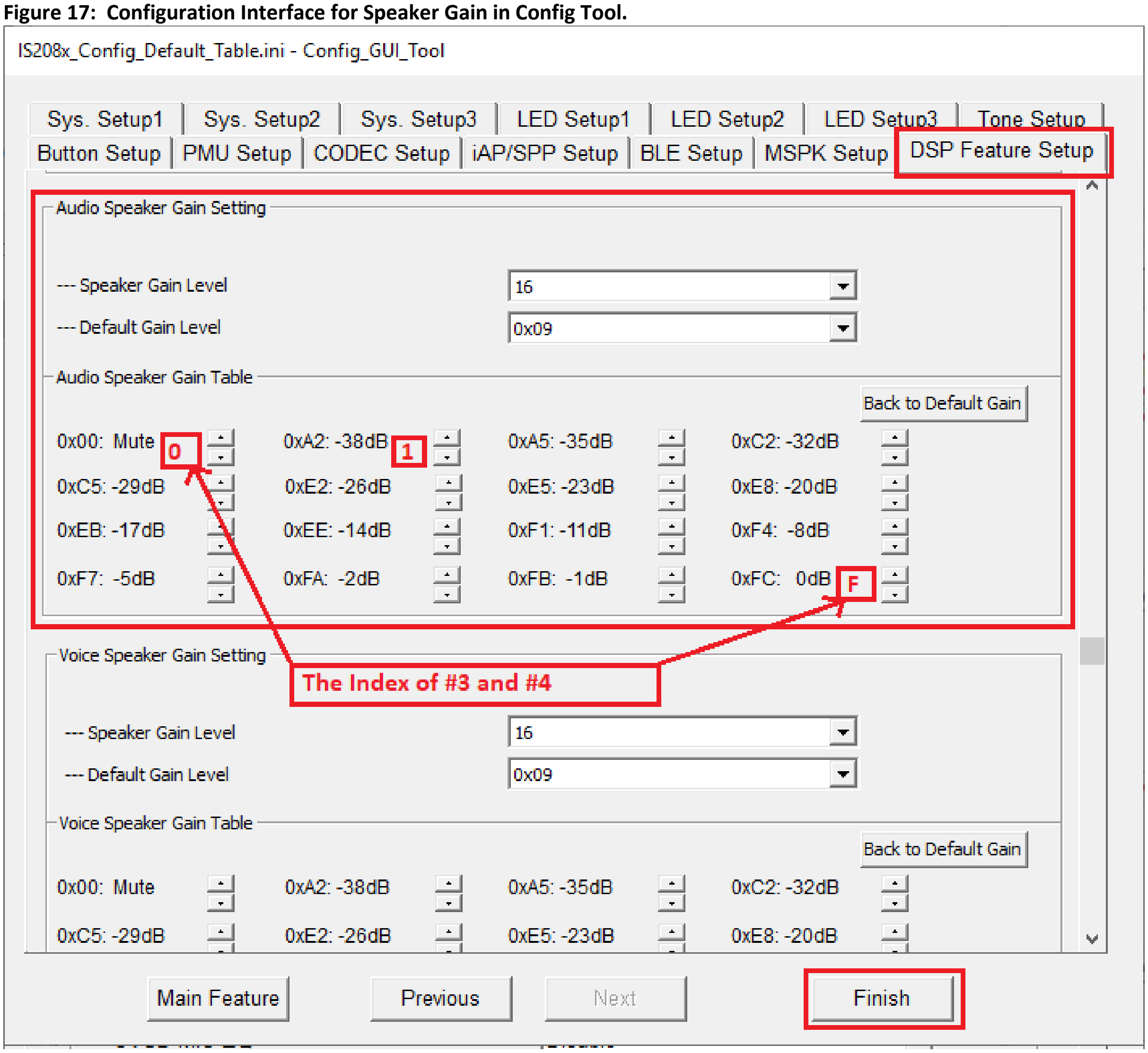
Note: The Parameters "Line-in max and Line-in min levels" hold the speaker gain value with in the minimum and maximum ranges.

- Line-In ADC Gain

Firmware configures the ADC gain, while playing line-in mode with selected index "3dB, 0x09". This controls the volume in Microphone. Line-In ADC Gain is present in DSP GUI Tool under Line-In page under audio function tab as shown in Figure 16(b).

When the line-in mode is connected and there is no audio (the RMS power of noise) below the threshold set in the DSP tool, the DSP sends a signal to MCU. Then MCU sends power down signal for the system. If there is an audio after the power-off, it cannot play as the system is off and it requires a power-on. For details, refer to *BM83x Evaluation Board User's Guide.*

Figure 17: Configuration Interface for Speaker Gain in Config Tool.



The time settings for the system power-off occurs in MCU firmware UI tool, as illustrated in Figure 18.

Figure 18: Auto Power-Off in Config Tool.

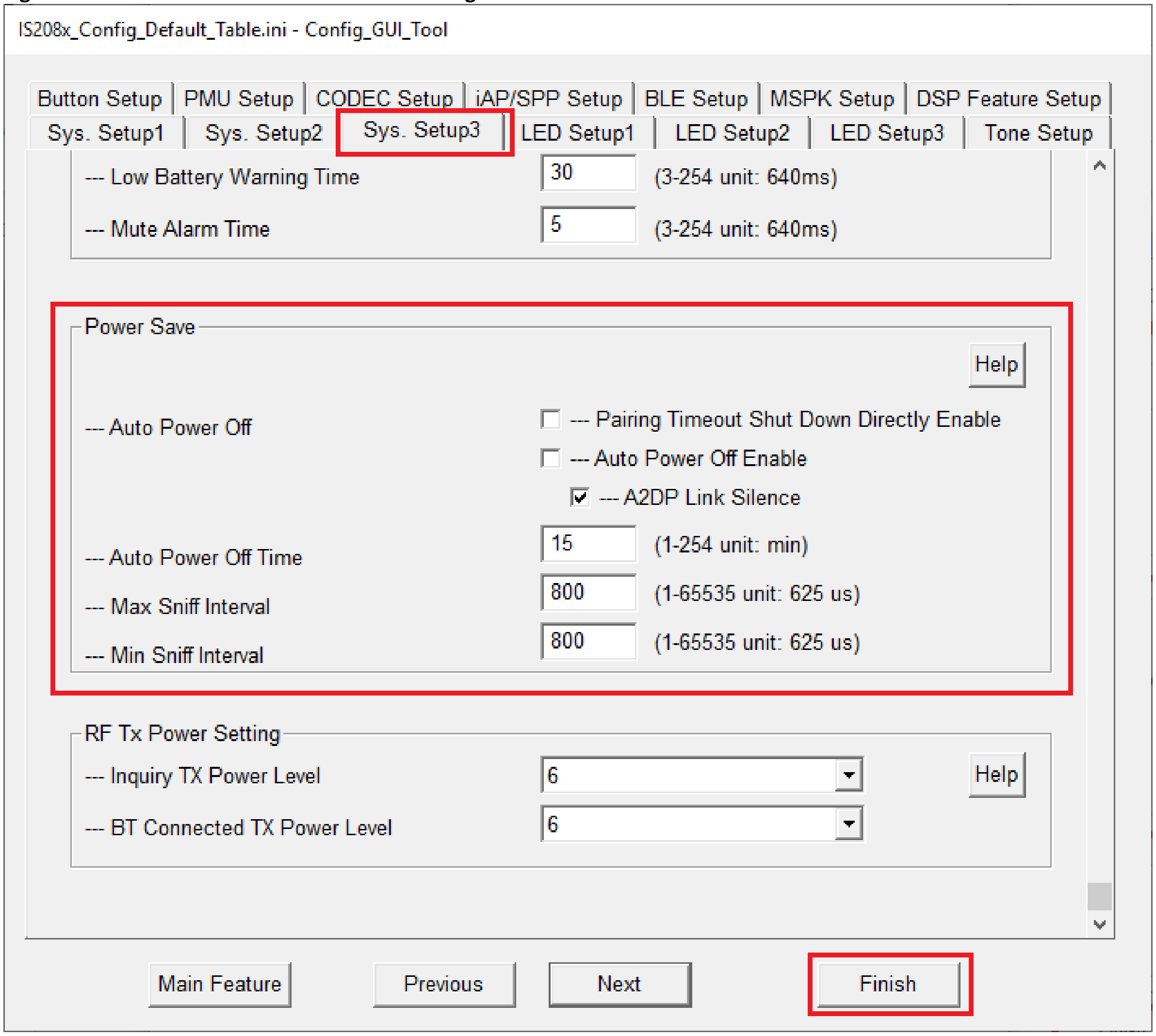
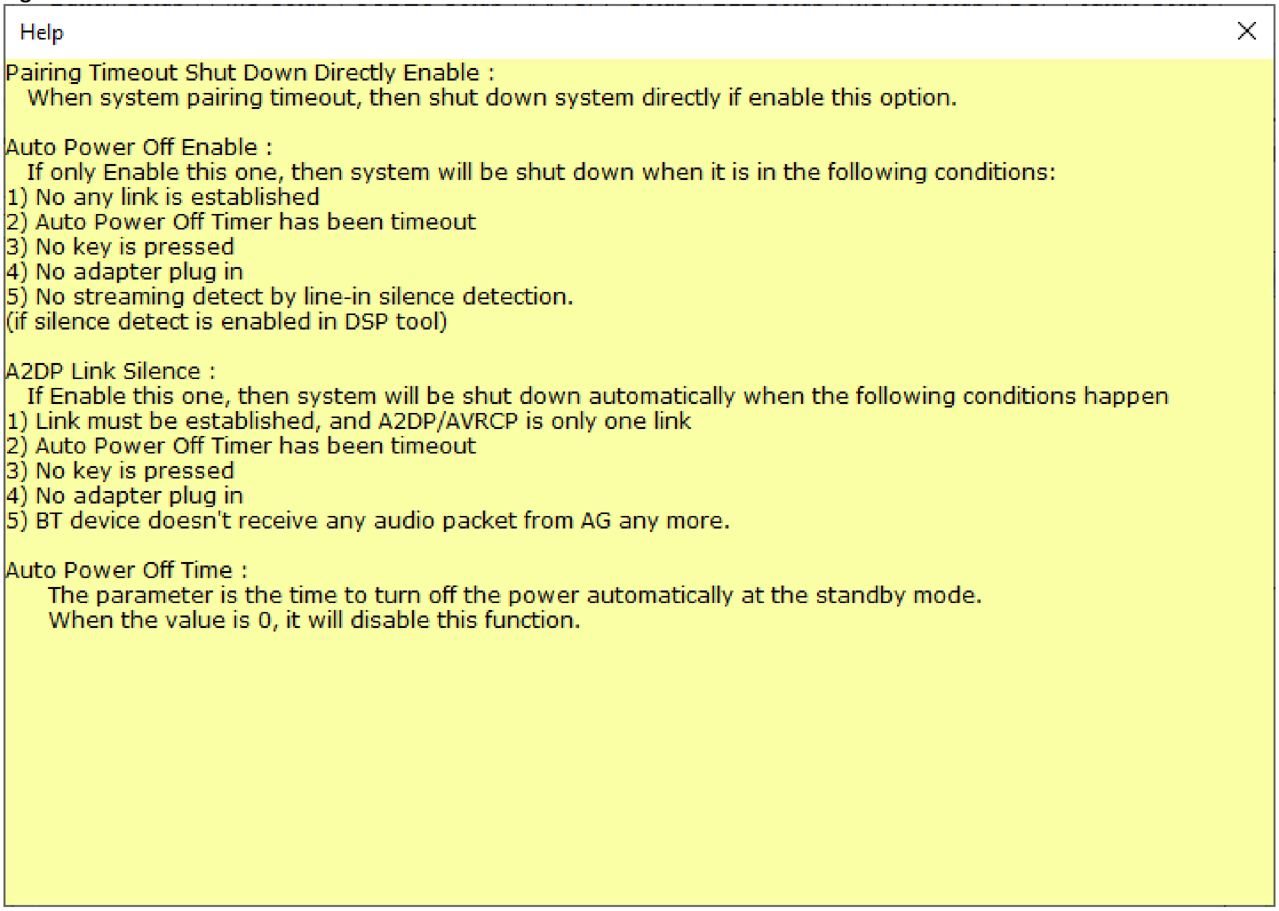


Figure 19: Auto Power-Off HELP



3.21.4.4 Sound effect – Audio Widening and Multiband Dynamic Range Compression

The Audio Widening (AW) and Multiband Dynamic Range Compression (MB-DRC) are the two functionalities with embedded audio signal processing, which provides better audio quality and user experience without using the external digital signal processor in the IS20XX SoC family.

The AW function processes the audio signal by phase filtering the signal played by the closely placed speakers, which sounds like the speakers placed far. In such way, the sound listening experience will be better with surrounding effects.

The MB-DRC function is an automatic volume control and uses the following compression mechanisms. Downward compression reduces the louder sounds over a certain threshold, while quiet sounds remain unaffected. Upward compression increases the loudness of sounds below a certain threshold by leaving the louder sounds unaffected.

Both downward and upward compression reduces the range of an audio signal. This is done for aesthetic reasons to deal with technical limitations of audio equipment, or to improve audibility of audio in noisy environments. 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.

1. Parameters of AW and MB-DRC:

Figure 20 illustrates the adjustable parameters for AW and MB-DRC. Please enable “sound effects” to use existing audio effects. The check boxes for MB-DRC and AW must be checked to enable these functions in config tool as shown in Figure 20(a). Virtual Bass (VB) check box is used to enable/disable Virtual Bass gets available only if MBDRC is enabled in config tool under DSP Feature Setup page as shown in Figure 20(a).

Figure 20(a): Enabling Interface for Sound Effect Configurations in Config Tool.

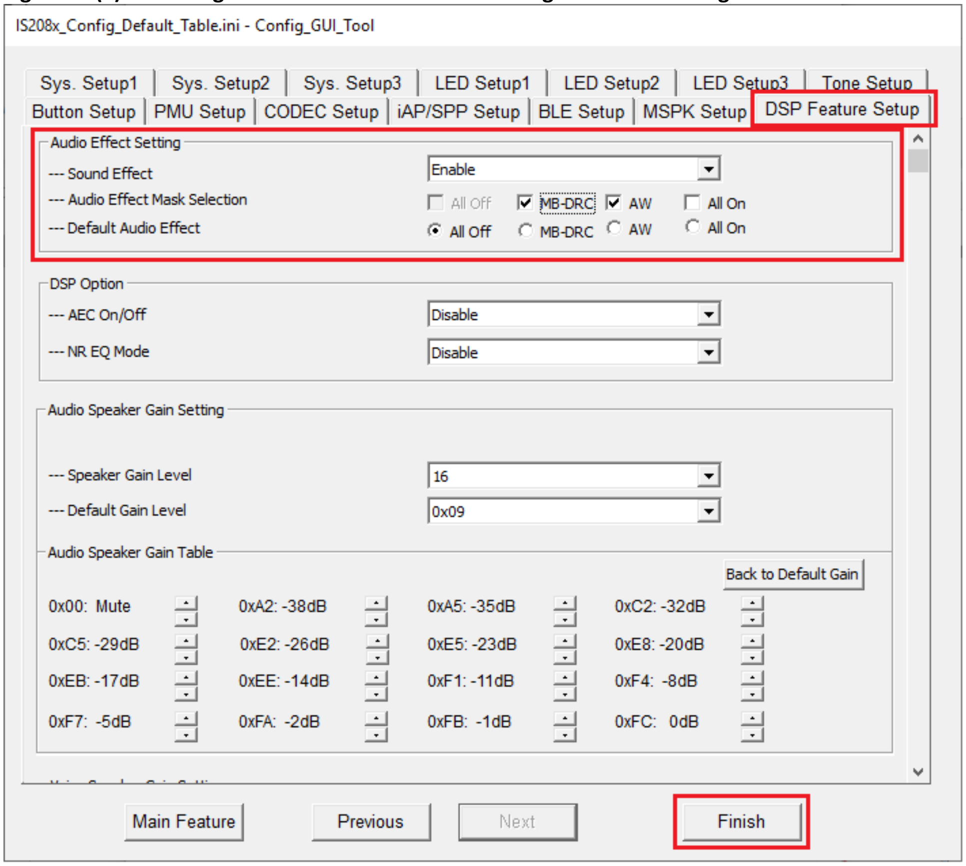
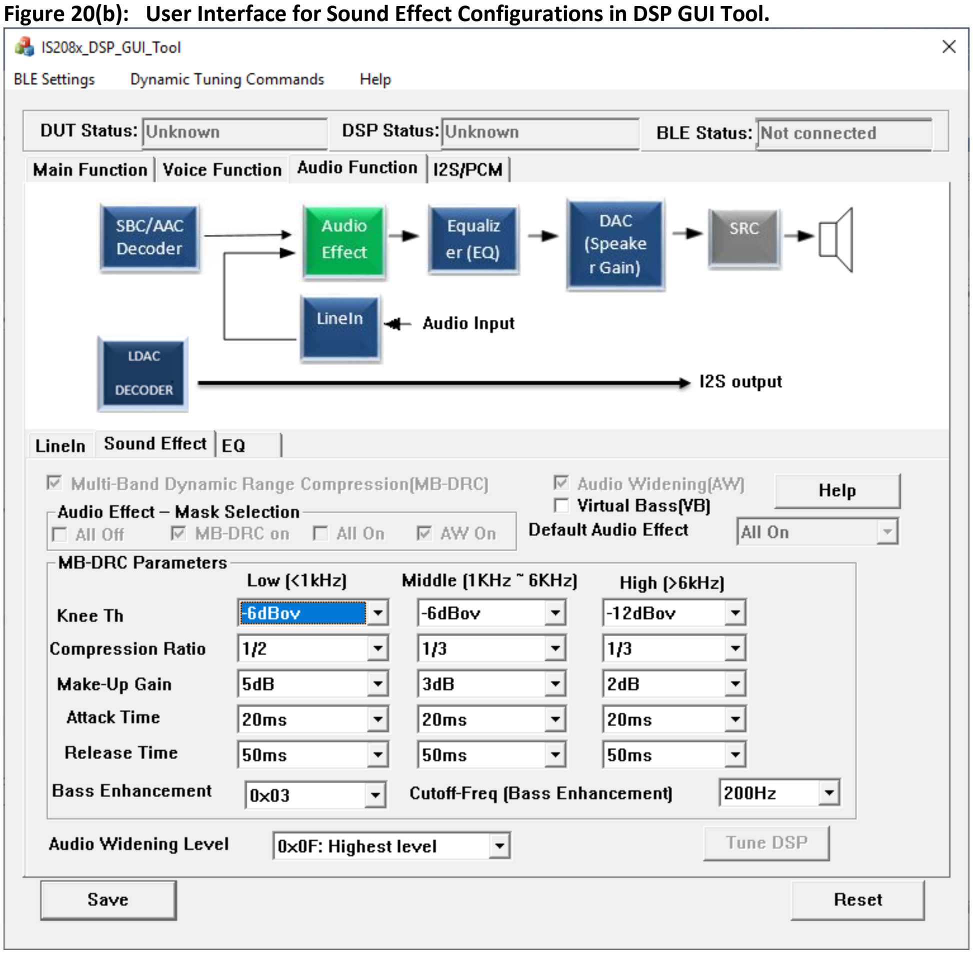


Figure 20(b): User Interface for Sound Effect Configurations in DSP GUI Tool.



2. Audio Effect – Mask Selection

The "Audio Effect - Mask Selection" is to select the combinations of audio effects, that can be selected by following checkboxes All off, MB-DRC, AW and All On.

3. Default Audio Effect

This parameter is to select the initial audio effect mode, after the device is power-on. You can select one of the following options All off, MB-DRC, AW and All On.

4. MB-DRC Parameters

The general concept of MB-DRC is to transform the input signal nonlinearly to its output. The parameters to control the behavior of MB-DRC are "Knee Th", "Compression Ratio", "Make-Up Gain", "Attack Time", "Release Time" and "Bass Enhancement". The MB-DRC provides adjustable parameters in three different bands (0 kHz to 1 kHz, 1 kHz to 6 kHz and >6 kHz) to process the input signal in a better way. The description of these parameters is listed as follows:

- Knee TH

  This parameter corresponds to the compression threshold, as illustrated in Figure 20(b). This parameter constrains the sound level of input to this threshold, above which attenuation of the input signal occurs.

- Compression Ratio

  The term Compression Ratio (CR) compresses the input sound level, exceeding the "Knee Th" of audio signal. However, if CR is closer to 0, the distortion due to the compression will be generated more easily.

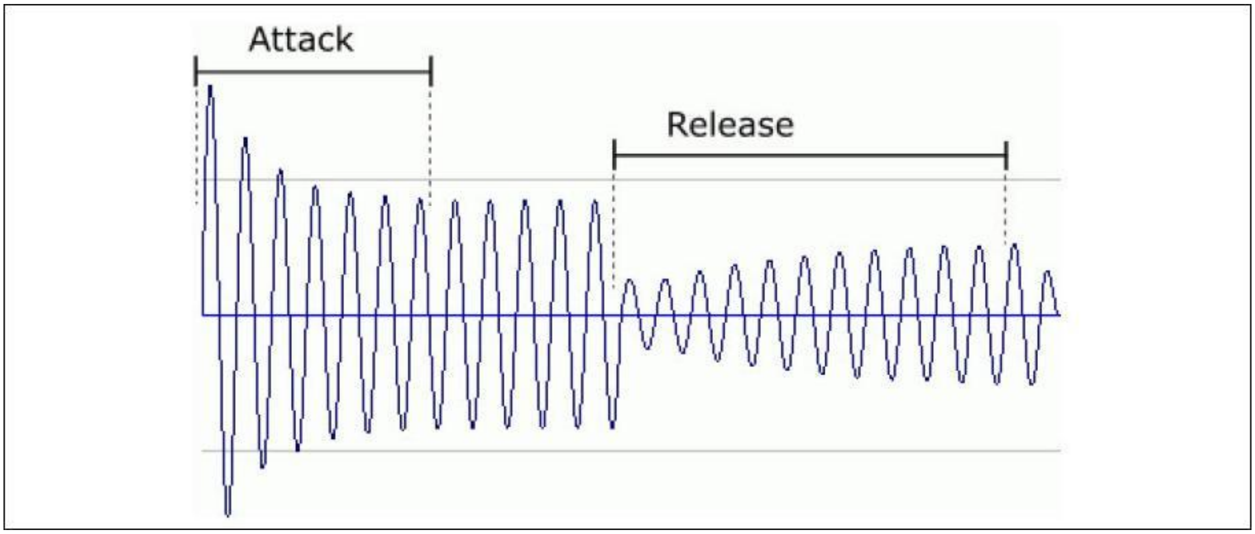
- Make-Up Gain

  Make-up gain is the maximum gain applying to audio signal, in which the average power is between silence threshold and the compression knee. This parameter boosts, the soft music signal to an audible level, especially in a noisy environment.

- Attack Time and Release Time

  The Attack time and Release time parameters are the period, which reaches 1-1/e = 63% of its final value. The difference between attack and release time constants, are illustrated in Figure 21. 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 21: Attack and Release Time Parameter

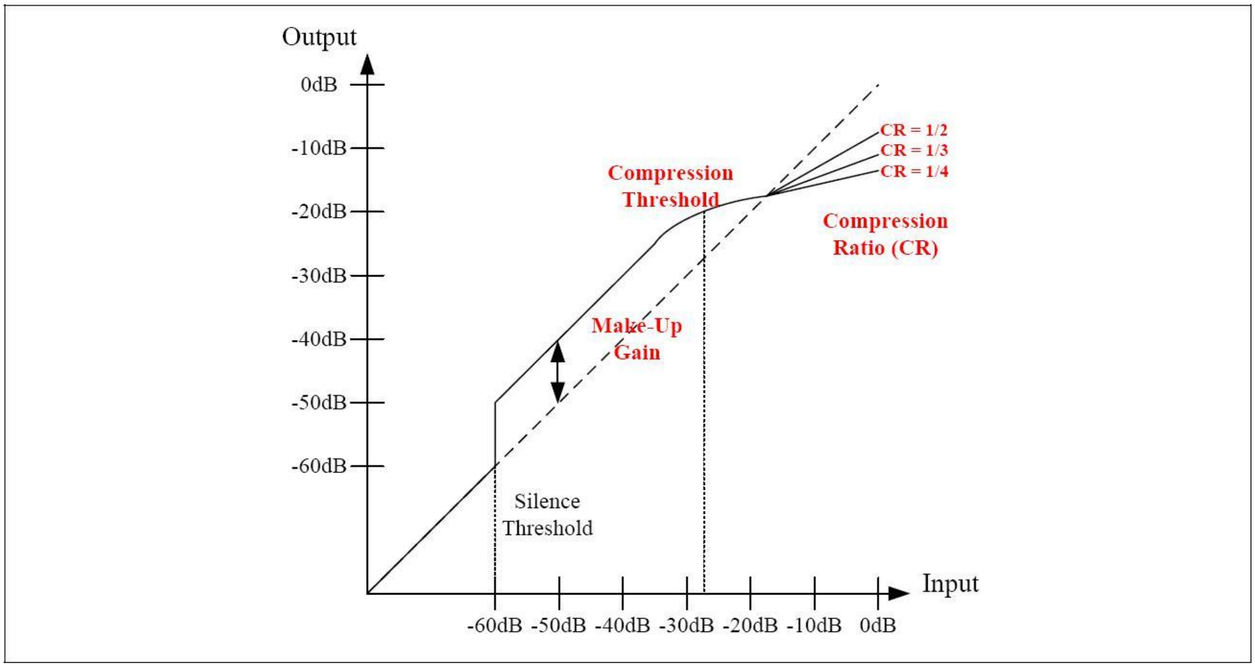


- Bass Enhancement:

This parameter controls the level of bass enhancement, which is disabled by default. Virtual Bass (VB) works along with the MB-DRC function when VB check box is enabled.
Figure 22 illustrates the mapping function of MB-DRC.

Note: Virtual bass creates harmonics and deviates THD+N by 30% around 100HZ to 300Hz frequency range. For high end speakers it is preferred to be disabled.

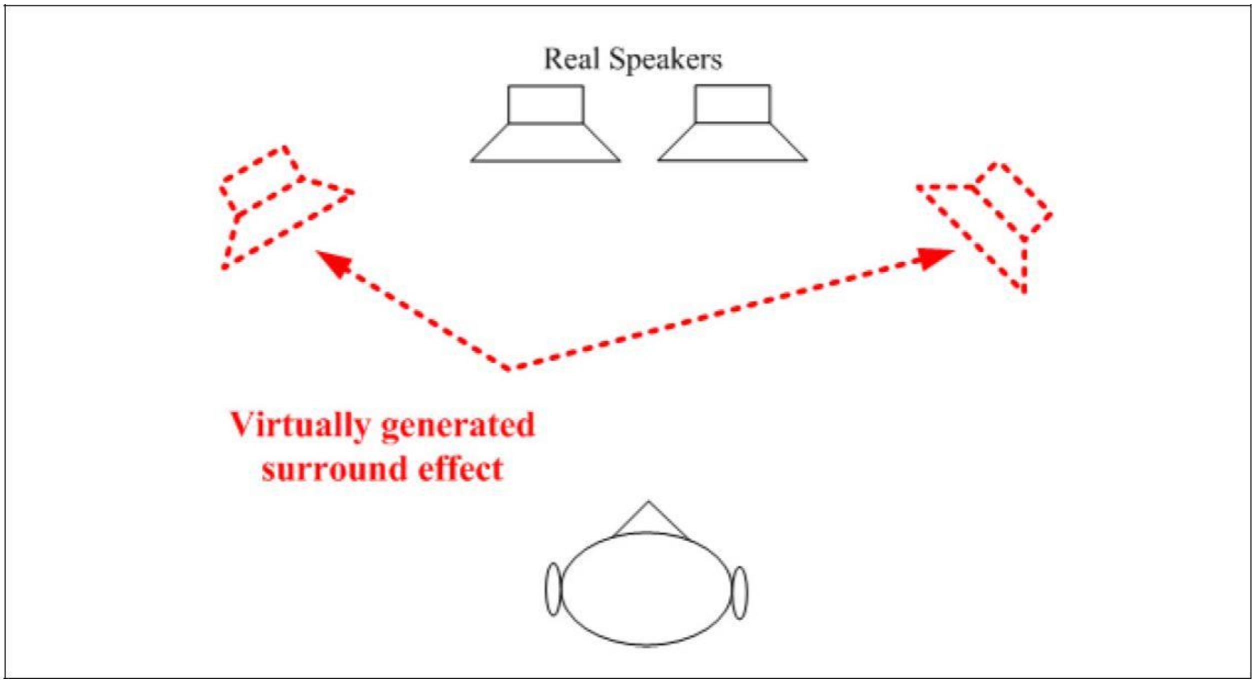
Figure 22: Mapping Function of MB-DRC



5. Audio Widening Parameters:

The Audio Widening (AW) sound effect is to process the music signal played by speakers, placed close apart or inside one housing, to make it sound like being placed far apart from each other, as illustrated in Figure 23. The virtually generated widening signal are mixed with original signal with various levels of weighting pairs, as illustrated in Figure 24.

Figure 23: Audio Widening Effect



6. Audio Widening Level:

This parameter controls the extent of AW effect. Basically, it is a mixing ratio of original sound and AW-processed sound signal. Figure 24 illustrates the block diagram of AW generator.

Figure 24: Block Diagram of AW Generator

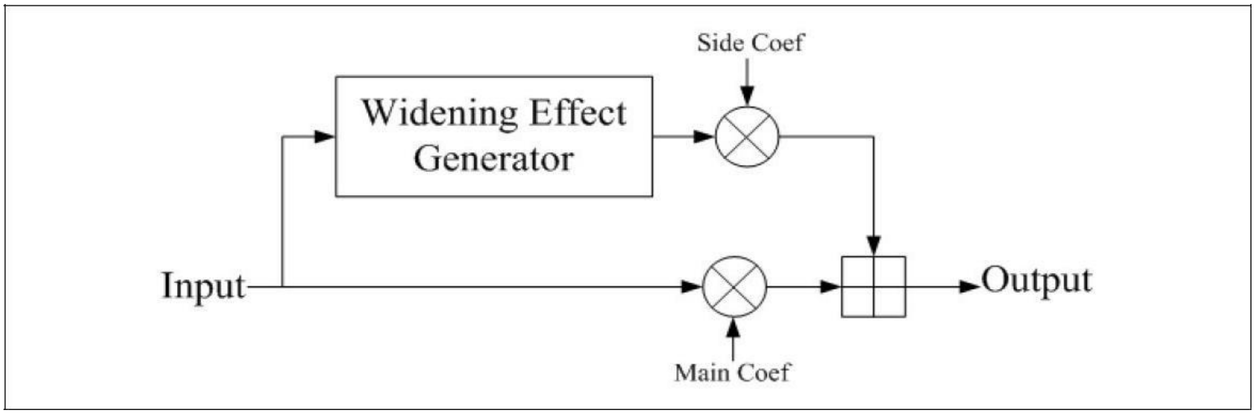


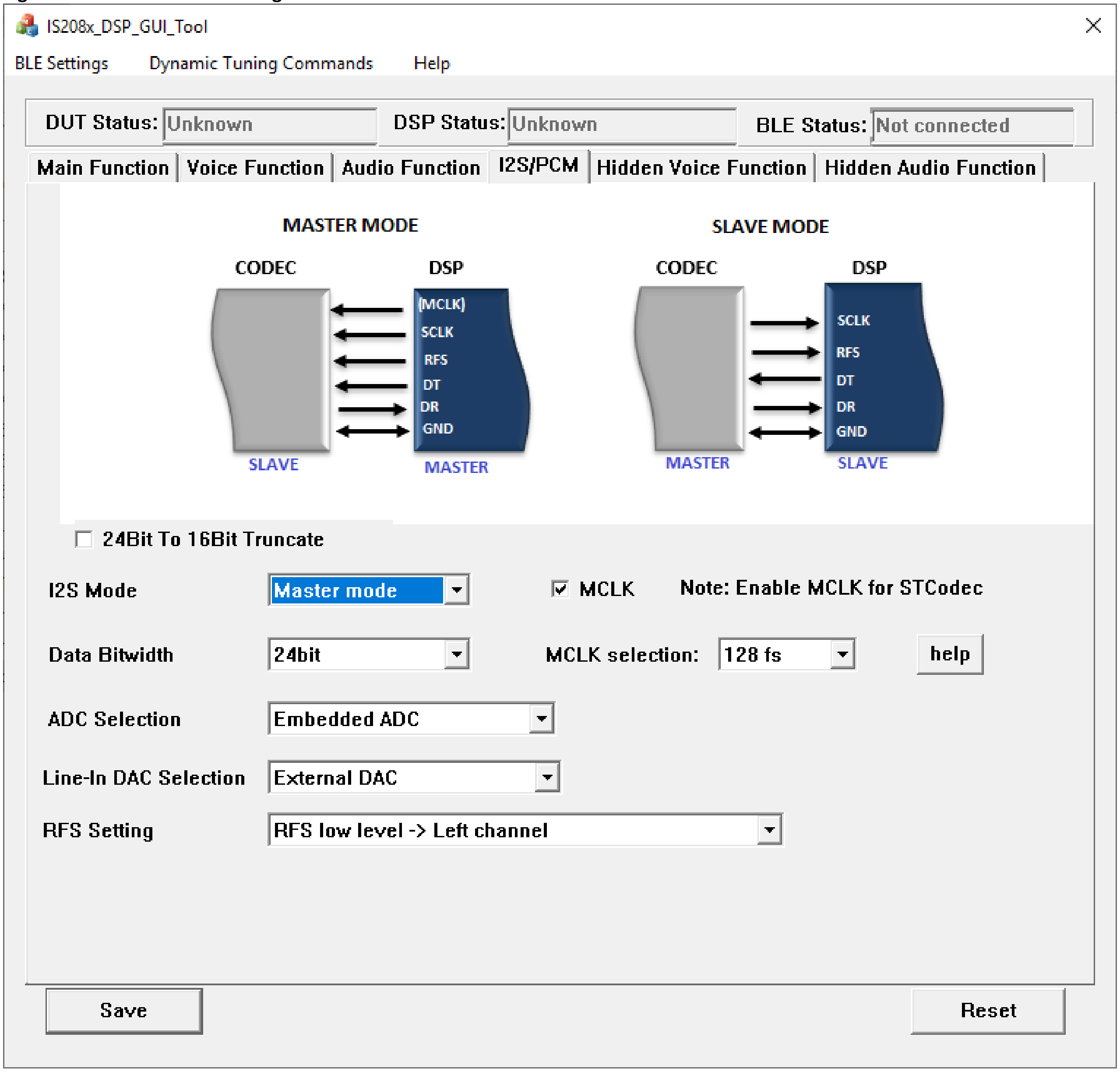
Table 4-2 provides the parameters of AW effect.

Table 4-2: Parameters of AW Effect

Feature	Values
Side Coef and Main Coef	0x00: Side Coef = 0; Main Coef = 1 0x01: Side Coef = 1/16; Main Coef = 1 0x02: Side Coef = 2/16; Main Coef = 1 0x03: Side Coef = 3/16; Main Coef = 14/16 0x04: Side Coef = 4/16; Main Coef = 14/16 0x05: Side Coef = 5/16; Main Coef = 14/16 0x06: Side Coef = 6/16; Main Coef = 13/16 0x07: Side Coef = 7/16; Main Coef = 13/16 0x08: Side Coef = 8/16; Main Coef = 13/16 0x09: Side Coef = 9/16; Main Coef = 12/16 0x0A: Side Coef = 10/16; Main Coef = 12/16 0x0B: Side Coef = 11/16; Main Coef = 12/16 0x0C: Side Coef = 12/16; Main Coef = 11/16 0x0D: Side Coef = 13/16; Main Coef = 11/16 0x0E: Side Coef = 14/16; Main Coef = 10/16 0x0F: Side Coef = 15/16; Main Coef = 9/16

3.18.5 I²S Digital Output/Input Interface

Figure 25: I²S Parameter Tuning Interface in DSP GUI Tool.



The IS208x SoC or BM8x Bluetooth module supports 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 “I²S Mode”, “Data Bitwidth”, “MIC ADC Selection” and “RFS Setting”, as illustrated in Figure 25.

The details of these parameters are discussed as follows.

5.1 I²S Mode

This parameter contains two modes:

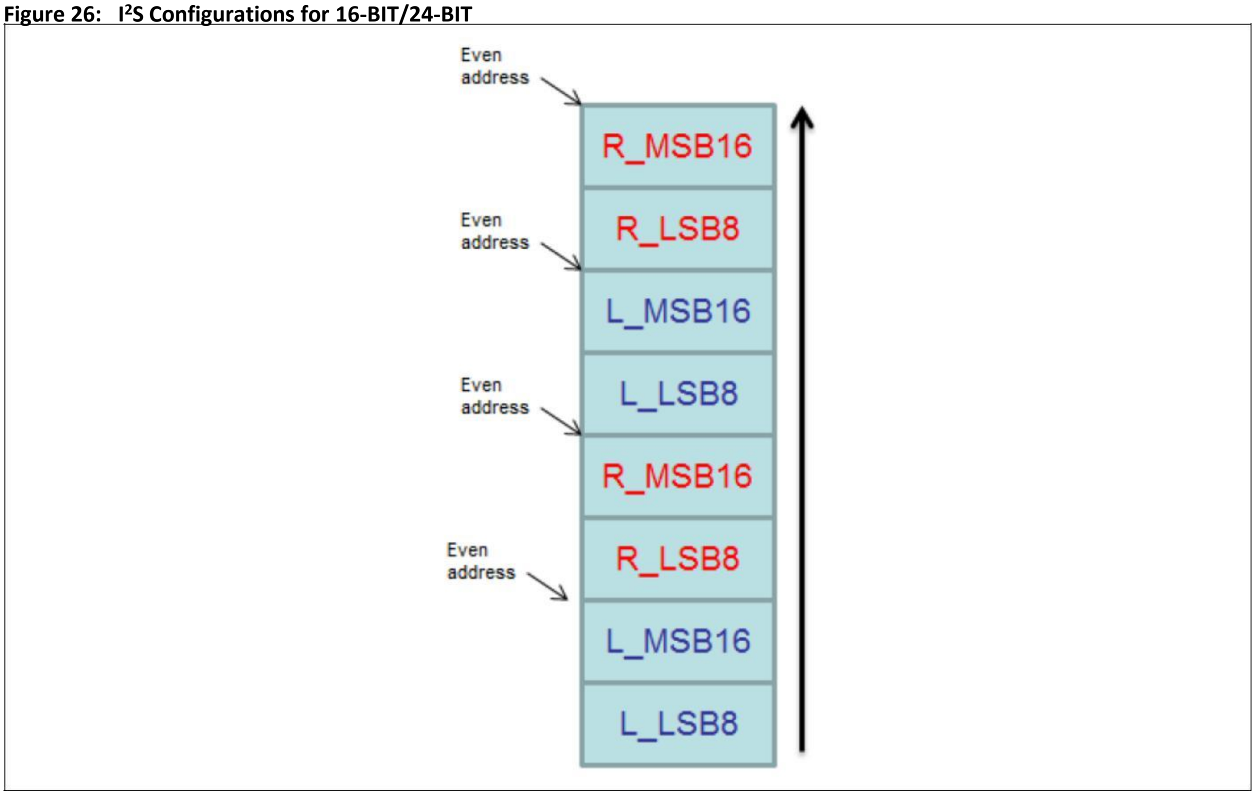
- Master mode - The IS208x SoC serves as a master to provide clock and frame synchronous signals for the master/slave data synchronizations, as illustrated in Figure 27.
- Slave mode - The IS208x SoC serves as a slave to receive clock and frame synchronous signals from the external codec or DSP devices, as illustrated in Figure 28.

3.21.5.2 Data Bitwidth

The numbers of bits for DR/DT are expected to receive from external codec or DSPs, or transmit to external codec or DSPs, as illustrated in Figure 26.

- 16 bits: The Bluetooth supported codec can only have 16-bit resolution.
- 24 bits: The trailing zeros in LSBs is the only way to meet the external DSP, that supports 24-bit I²S port requirement. In this way, the 16 bits are filled in 16 MSBs and 8 zeros in LSBs. (**Note**: Non LDAC case)

Figure 26: I²S Configurations for 16-BIT/24-BIT



3.21.5.3 MIC ADC Selection:

If the hands-free function is supported, select the following ADC configuration.

- Internal ADC: On-chip ADC is used.
- External ADC: External ADC/DSP is selected.

Figure 27: I²S Hardware Configurations for Master Mode

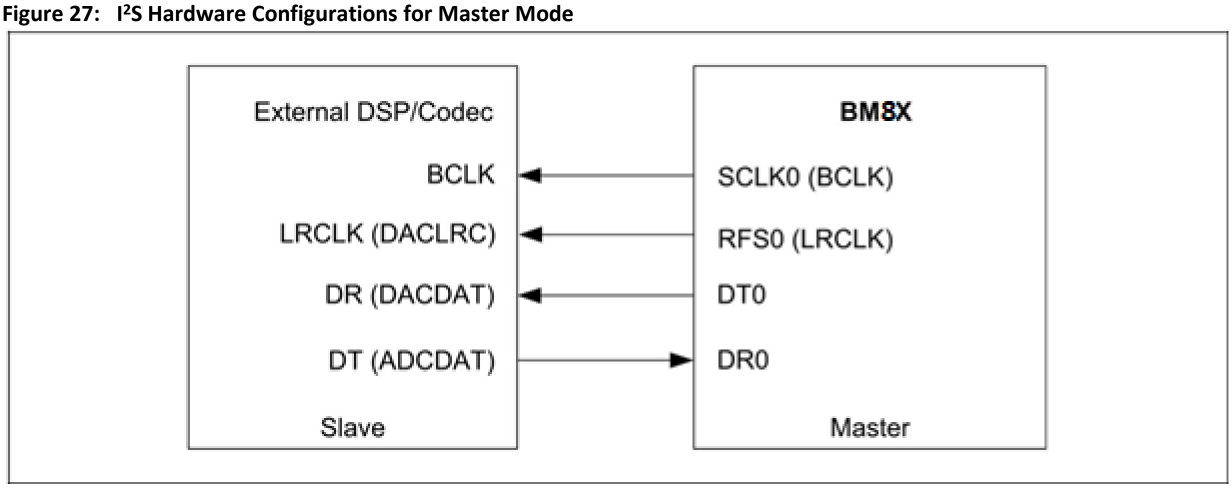
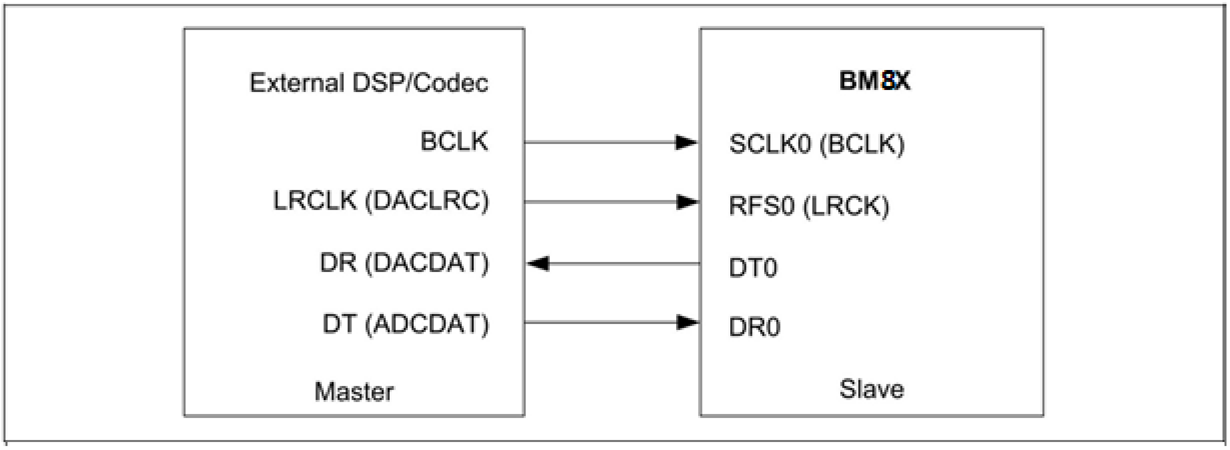


Figure 28: I²S Hardware Configurations for Slave Modes with External ADC



3.21.5.4 Line-In DAC Selection:

Select the following Line-In DAC configuration during line-in input case.

- Internal DAC: On-chip DAC is used.
- External DAC: External DAC/DSP is selected.

3.21.5.5 RFS Setting:

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

- RFS low-level - Left channel: Refer to timing diagram, as illustrated in Figure 29.
- RFS high-level - Right channel: Refer to timing diagram, as illustrated in Figure 30.

Figure 29: Low and High levels of RFS Denote in Left Channel

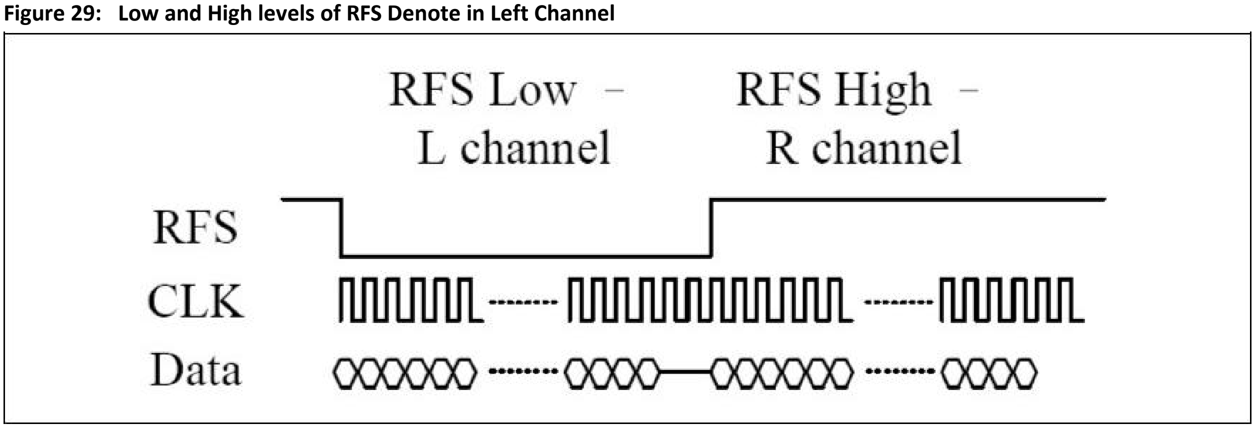
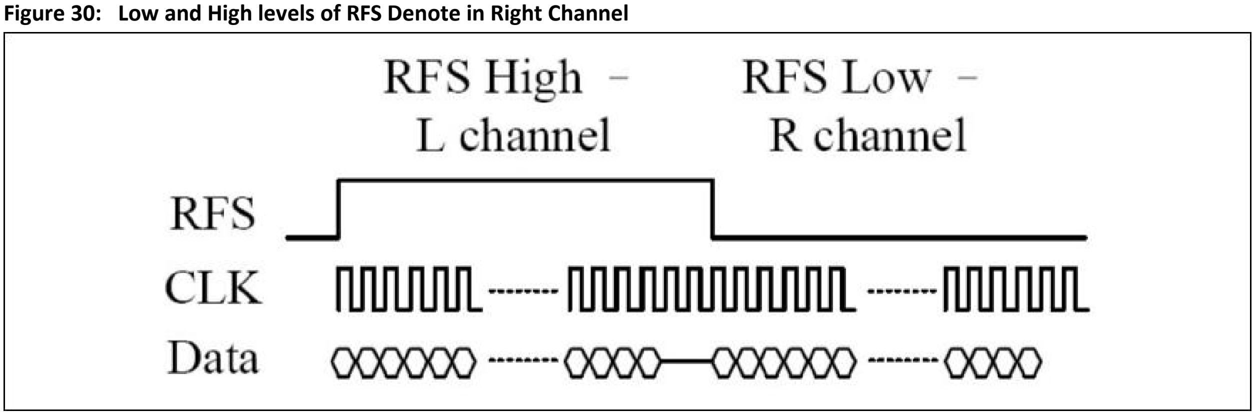


Figure 30: Low and High levels of RFS Denote in Right Channel



3.21.5.4 MCLK Selection:

MCLK has 128*fs and 256*fs selection options. Where fs is sampling frequency and 128, 256 over sampling rates.
MCLK supports following sub sampling rates in each option's:

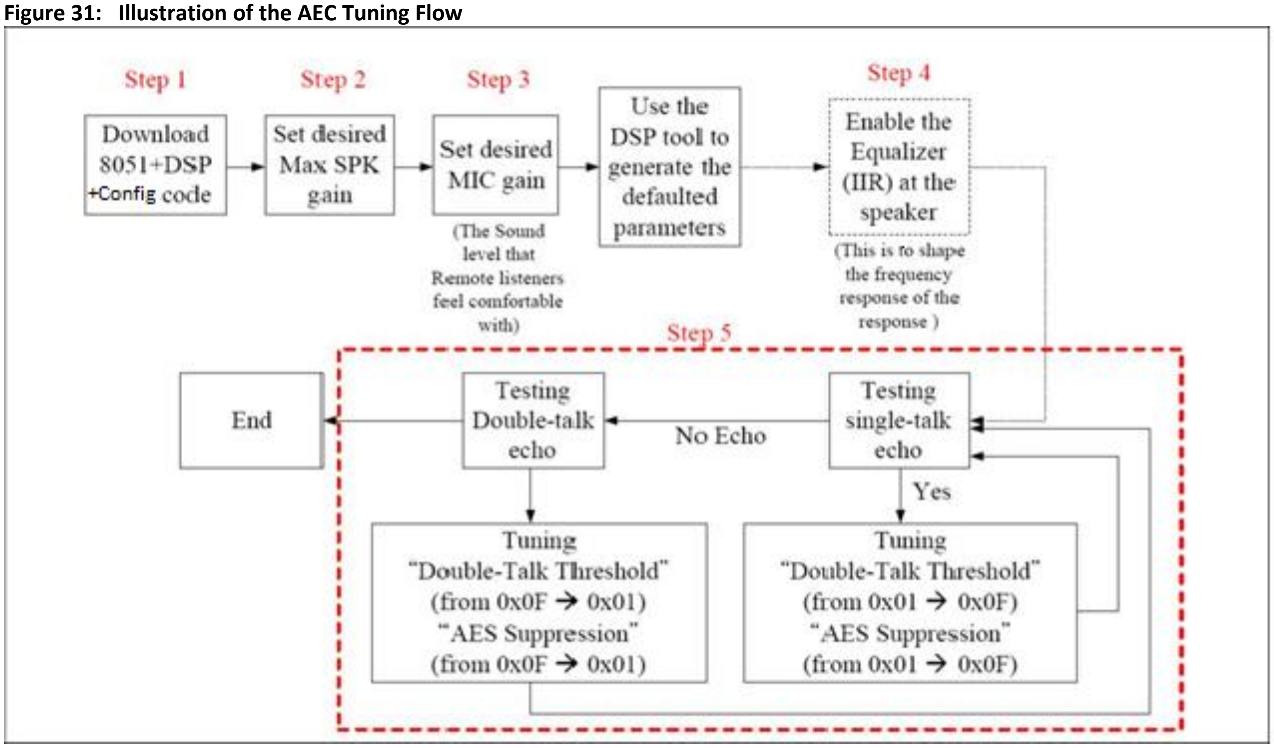
- MCLK 128*fs case:
  For 8KHz,16 KHz,44.1 KHz and 48KHz 128*fs is used for MCLK.
  For 88.2 KHz and 96KHz 64*fs is used for MCLK.
- MCLK 256*fs case:
  For 8 KHz,16 KHz,44.1 KHz and 48KHz 256*fs is used for MCLK.
  For 88.2 KHz and 96KHz 128*fs is used for MCLK.

To enable MCLK selection option choose I2S mode as master mode and enable MCLK check box.
Note: MCLK should always be enabled for STCodec.

3.18.6 Guidelines for Tuning Echo Cancellation Performance

This section introduces a guideline to fine tune the echo cancellation performance. There are usually three requirements for the EC tuning. They are high MIC volume, echo-free and double-talk (DT) performance. These three requirements are different with each other. For example, if the high volume at MIC path is required, the echo will also get amplified. Therefore, tuning of the parameters is required to make echo inaudible, which results in poor DT performance. Figure 31 illustrates the first five steps basically handling the whole speaker phone/car kit echo issues.

Figure 31: Illustration of the AEC Tuning Flow



- **Step 1**

Initially download all the firmware (8051 + DSP) and Config tool hex file.

- **Step 2**

Before tuning the AEC performance, the maximum desired speaker output level must be determined. More specifically, 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.

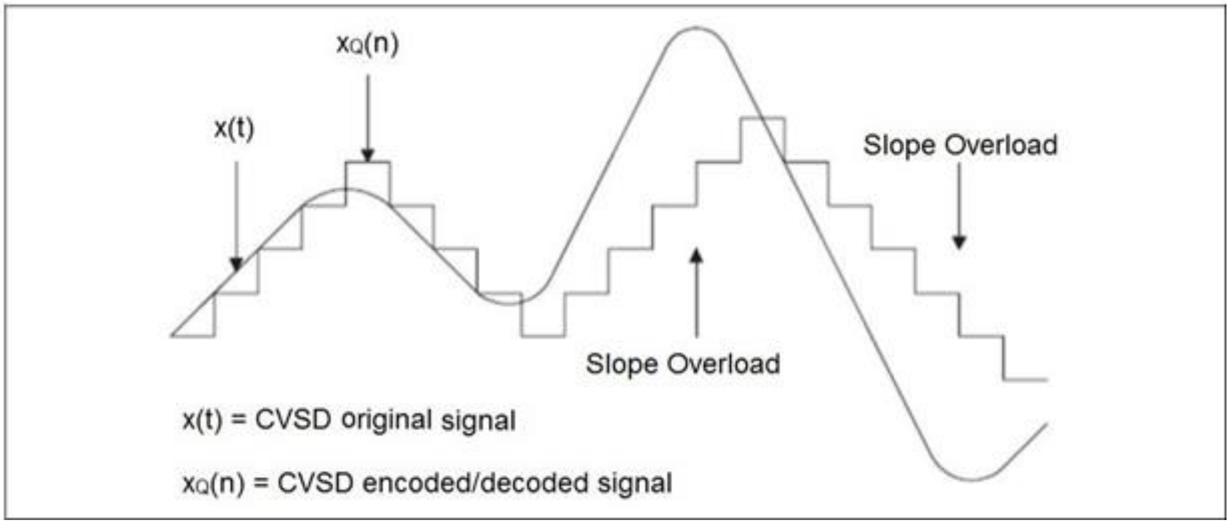
Note: The target speaker output volume must be determined at beginning only, based on the required specification.

- **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 because of the slope overload effect, as illustrated in Figure 32, which is caused by the CVSD codec that naturally suppress the high frequency parts and make it works as a low-pass filter. This effect distorts the near-end speech and make it not as clear as the softer MIC gain levels.

For more information of the CVSD slope overload effect, see http://www.datasheetcatalog.org/datasheet/ CML/mXxyzvw.pdf.

Figure 32: CVSD Slope Overload Effect

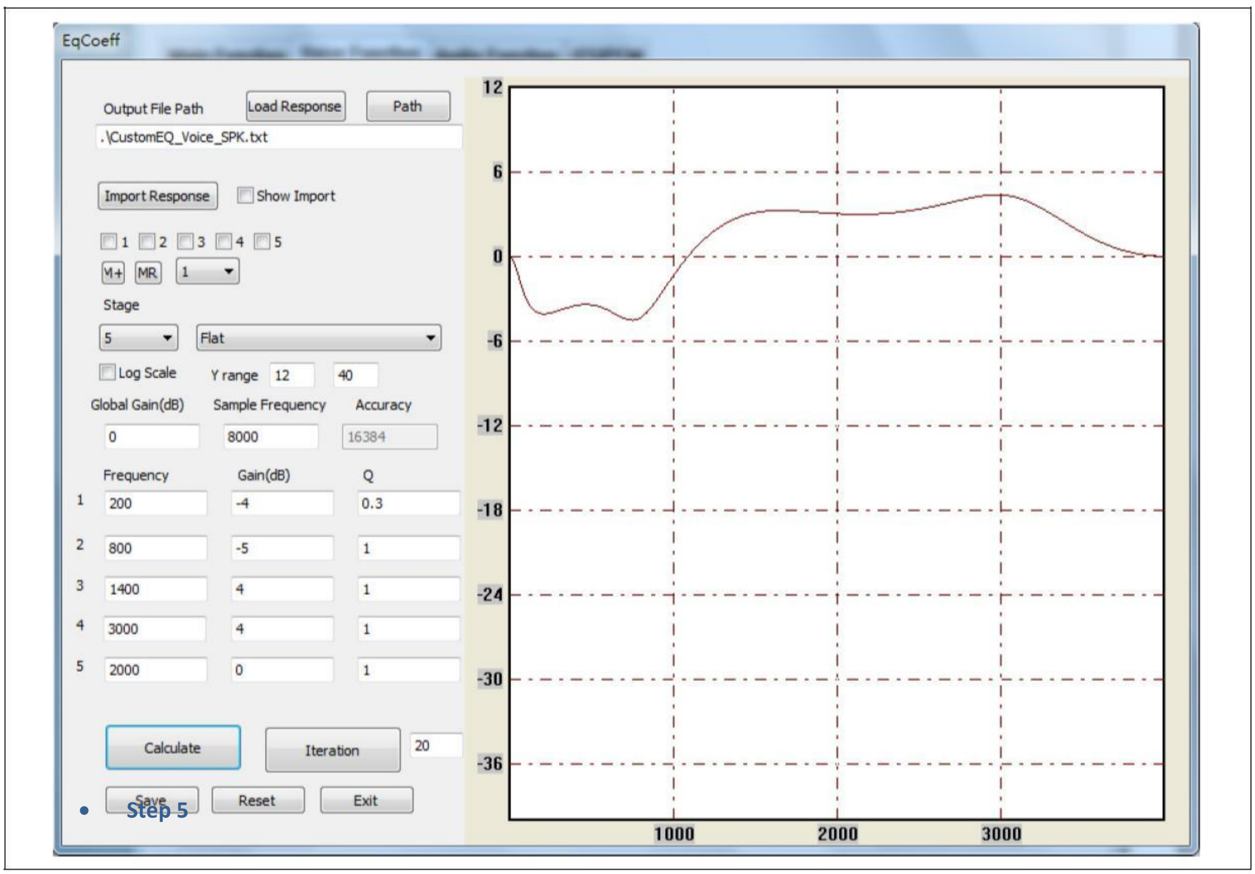


- **Step 4**

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).

The echo reverberation within the speaker phone/car kit housing is reduced in this way, so that the linearity of the echo coupled to the MIC input can be better. The echo linearity is highly associated with the AEC performance. Figure 33 illustrates an example of these settings which are obtained empirically. However, the required frequency gets changed in terms of which speaker and housing are selected.

Figure 33: Example of the Frequency Shaping for Signal at Speaker Path



- **Step 5**

The purpose of this step is to fine tune the AEC performance. Basically, it is divided into two parts: Single-talk echo tuning and Double-talk echo tuning.

a) Single-Talk Echo Tuning

The parameters **"**Double-Talk Threshold" and "AES Suppression" (refer to **3.3"Echo Cancellation"**) are responsible for tuning the Single-talk echo performance. As per thumb rule, first adjust the parameter of "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, then start to fine tune the "AES Suppression" from "0x01" to "0x0F".

Note: Although the selectable value of "Double-Talk Threshold" is up to 0x1C, these two values are not recommended to suppress the echo, since it would distort the MIC speech severely.

b) Double-Talk Echo

If the single-talk echo can get suppressed effectively by the default settings of "Double-Talk Threshold" and "AES Suppression", then the double-talk performance can be increased further 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".

There are four possible measures to improve the double-talk performance:

1. Increase the AEC MIC gain: To increase the MIC gain, refer to **6.3"Step 3"** procedure, as both AES and AEC suppresses echo as the double-talk near-end speech. The double-talk performance can be improved, if the near-end speech energy is raised up, so that the near-end speech can become audible, while no single-talk echo is present.
2. Adjust the frequency shaping, as illustrated in Figure 33: If the speaker output level is very loud and echo-to-speech ratio at the microphone input is too high, then the 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 micro-phone input is further reduced and can have the better double-talk performance.
3. Use the handsets to support the full-duplex AEC: Some handsets, such as "Samsung Galaxy II" and "HTC Incredible" etc, don't support the full-duplex speech communication, while connecting with the Bluetooth hand free devices. In this way, satisfactory full-duplex performance is not obtained, 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 are 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. The User Interface (UI) con-figures the number of the desired speaker channel outputs.

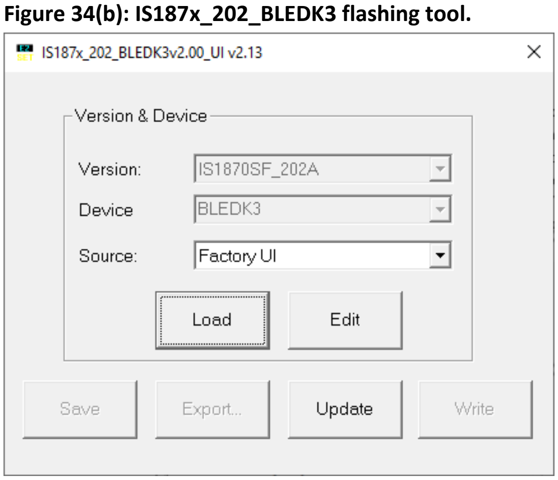
3.18.7 Guidelines for Over the Air (OTA) Tuning of DSP Parameters

3.21.7.1 (Over the Air) OTA Tuning Setup.

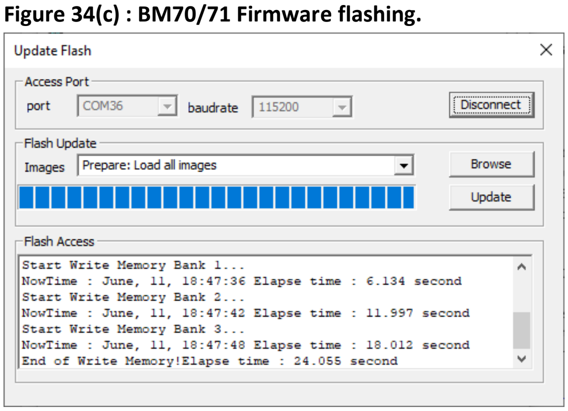
Step 1:

- Select BM70/71 PICTail board that is shown in Figure 34(a) and connect it over the UART cable to the computer. Connected BM70/71 board is used as a dongle at computer side to tune BM83 DSP over BLE connection.
  Figure34(a): BM71 PICTail board

- Put BM70/71 module SW7 button into ON mode and click on reset button on BM70/71. Blue LED should be glowing on BM70/71 module after this.
  Figure 34(b): IS187x_202_BLEDK3 flashing tool.

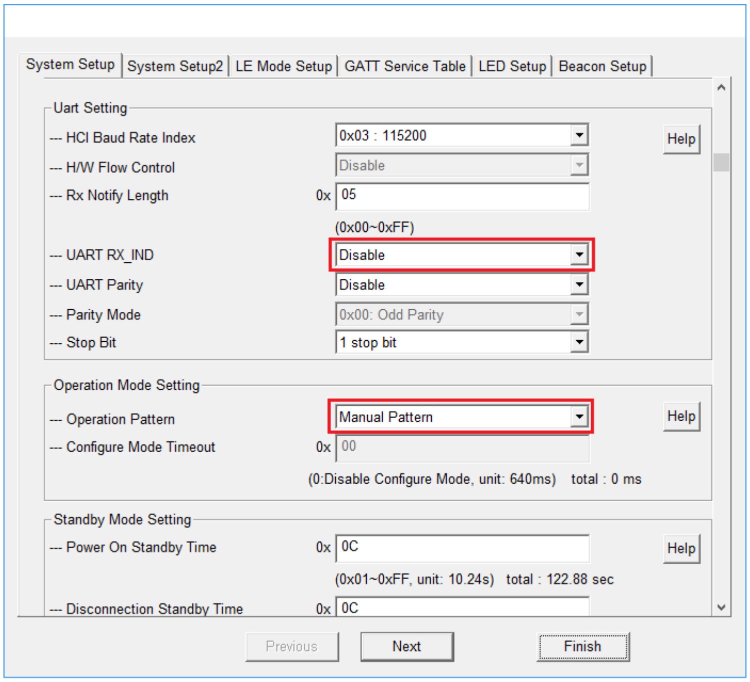


- Open IS187x_202_BLEDK3 flashing tool as shown in figure 34(b). Click on update button to open BM70/71 FW flashing dialog. Select proper comport, select 115200 baud rate and connect to BM70/71. Click on browse button and load hex images with extensions ".H00",".H01",".H02",".H03" in SDK downloaded. And click on update button to flash BM70/71 firmware.
  Figure 34(c) : BM70/71 Firmware flashing.



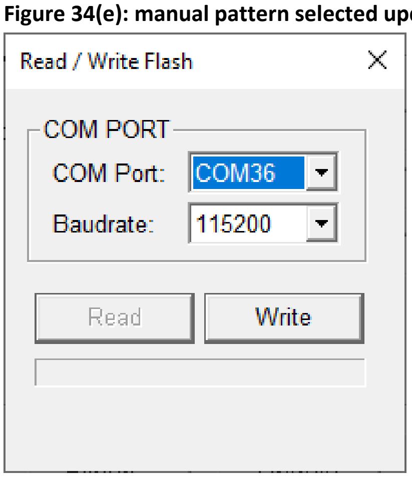
- Click on *Load button in* IS187x_202_BLEDK3 tool -> click *Load Text File button* -> load *"IS1870SF_202_BLEDK3v2.00_UIv2.13(BM70)_DSP_OTA_Tuning.txt"* or *"IS1870SF_202_BLEDK3v2.00_UI v2.13(BM71)_DSP_OTA_Tuning.txt"* file.
- Then click on Edit button and select Manual Pattern mode, disable UART RX_IND options in tool and then click on finish button as shown in Figure 34(d).

Figure34(d): BM71 Manual Pattern mode Selection.



- Click on write button in IS187x_202_BLEDK3 tool as shown in figure 34(b). and Flash these selected options to BM70/71 by selecting proper comport and baud rate and click write button as shown in figure 34(e).

Figure 34(e): manual pattern selected update to flash.



- Once flashing is completed put back SW7 button in MB70/71 to OFF mode and click on reset button. For More information look into Firmware Programming Procedure section in BM71 User Guide Manual given in following link https://www.microchip.com/Developmenttools/ProductDetails/BM-71-PICTAIL.
- After this power up BM83 and enable advertising in BM83.

Note: Generate latest hex file with all the necessary features enabled in config tool, and flash BM83 every time when new config tool is released. This avoids flash data read error in tool, when connected over BLE with BM83.

Step 2: Open DSP GUI config tool in IS208x_Config_GUI_Tool. Click on ***BLE Settings*** in menu bar -> click ***UART and BLE Connect*** to open OTA tuning dialog box as shown in Figure 35(a) and 35(b).

Figure 35(a): OTA tuning setup dialog launch option in menu bar

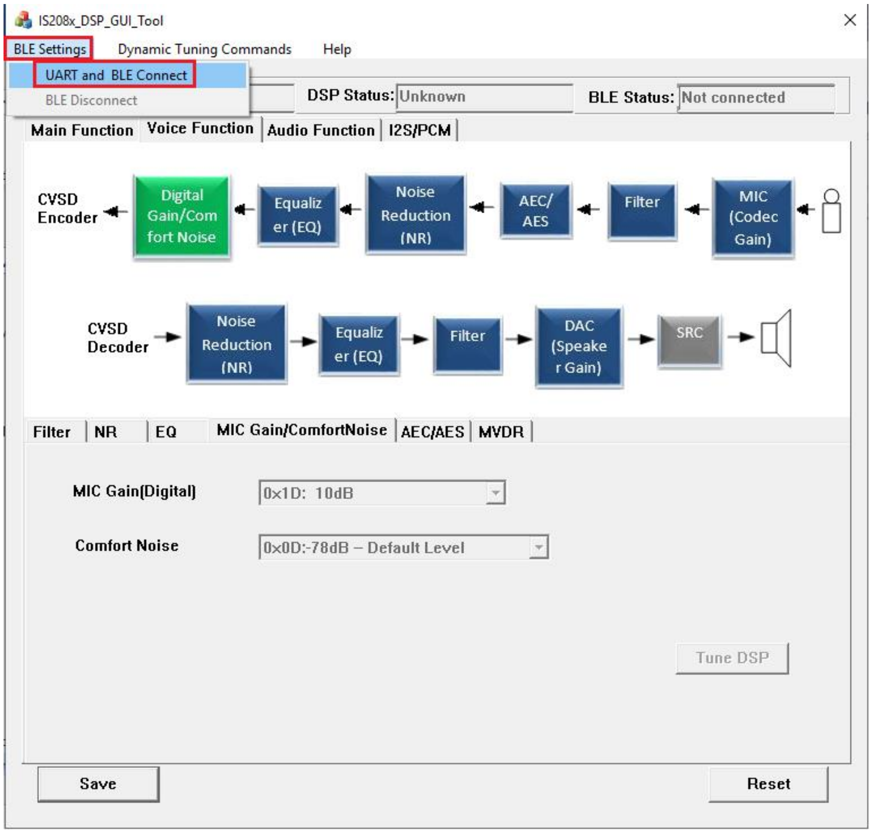
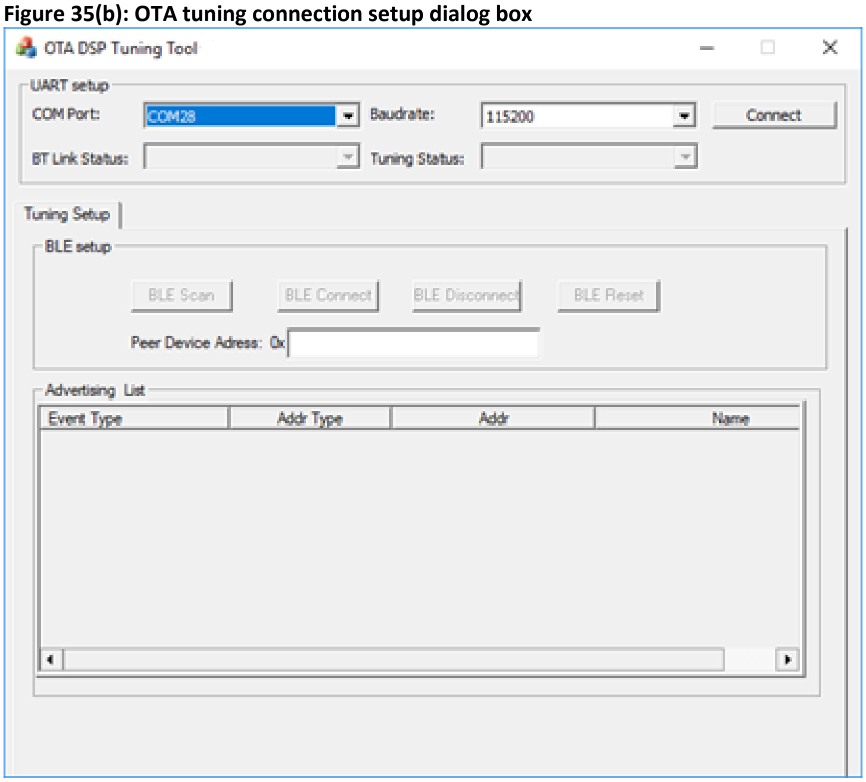


Figure 35(b): OTA tuning connection setup dialog box



Step 3: In OTA tuning dialog select BM71 dongles comport and appropriate baud rate and click on connect to establish UART connection with BM71 dongle. As shown in the Figure 36(a). Once the connection is established you could see the Link status and Tuning status in the status combo boxes as shown in Figure 36(b).

Note: After UART connection if you see BT link status and tuning status shown in red box in Figure 36(b) blank, please click on reset button present in BM70/71 board.

Figure 36(a): UART connection setup with BM71 dongle.

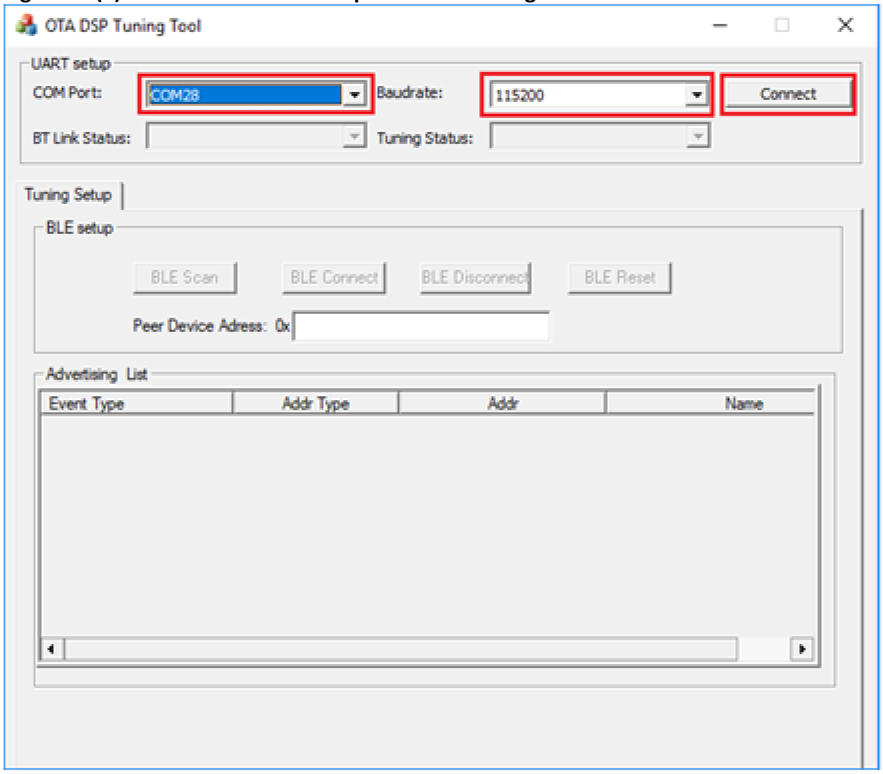
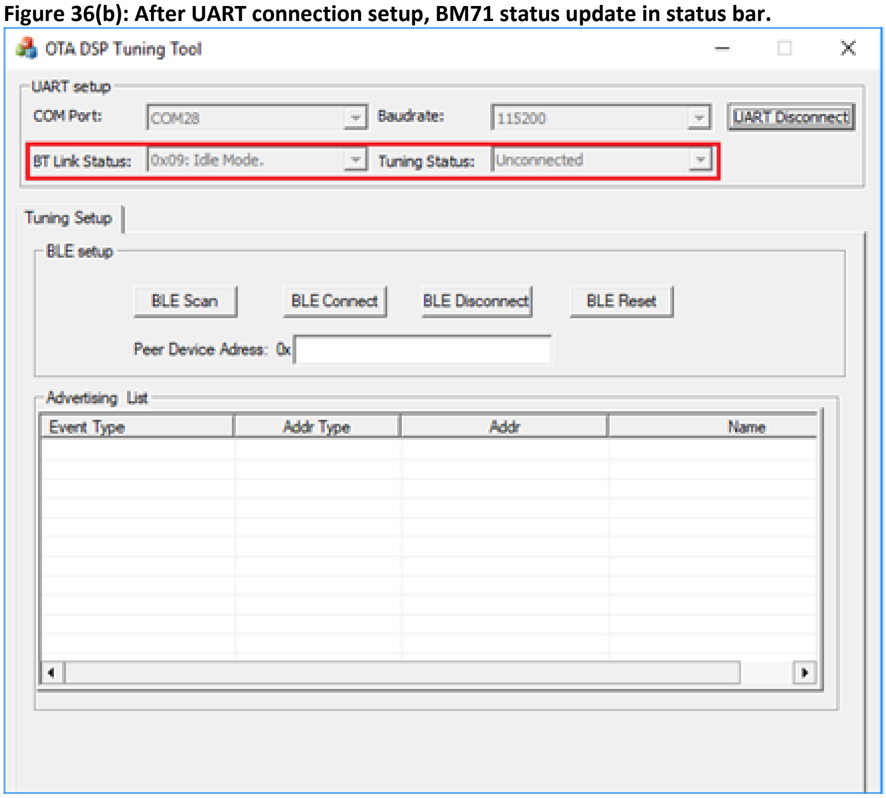


Figure 36(b): After UART connection setup, BM71 status update in status bar.



Step 4: Once UART connection is setup **BLE Scan**, **BLE Connect**, **BLE Disconnect** and **BLE Reset** button will be enabled. Click on BLE scan button to get the list of advertising devices as shown in Figure 37(a). Or you can directly enter peer device BT address directly if you know in advance and click on BLE connect to establish the Bluetooth connection as shown in Figure 37(b).

NOTE: In case of BM70 dongle is not responsive and not able to scan, do software reset by clicking on BLE Reset button and hard reset by clicking on Reset button in BM70/71 board, and try once again scanning. And click on BLE Reset button to enable scan button, if scan button is not enabled at any point of time but this will terminate your BLE connection.

Figure 37(a): Scan for advertising devices.

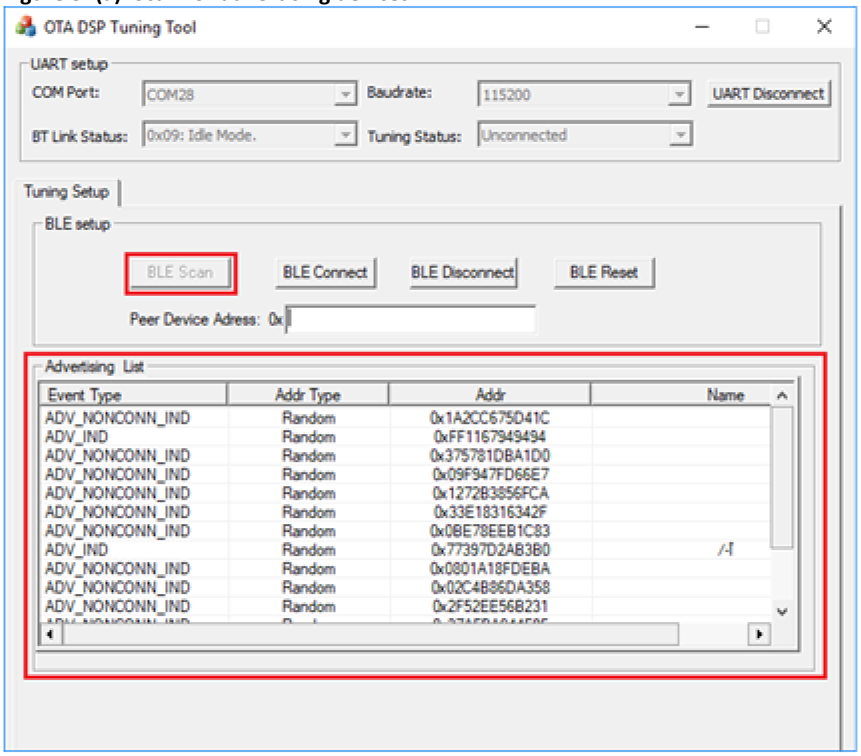
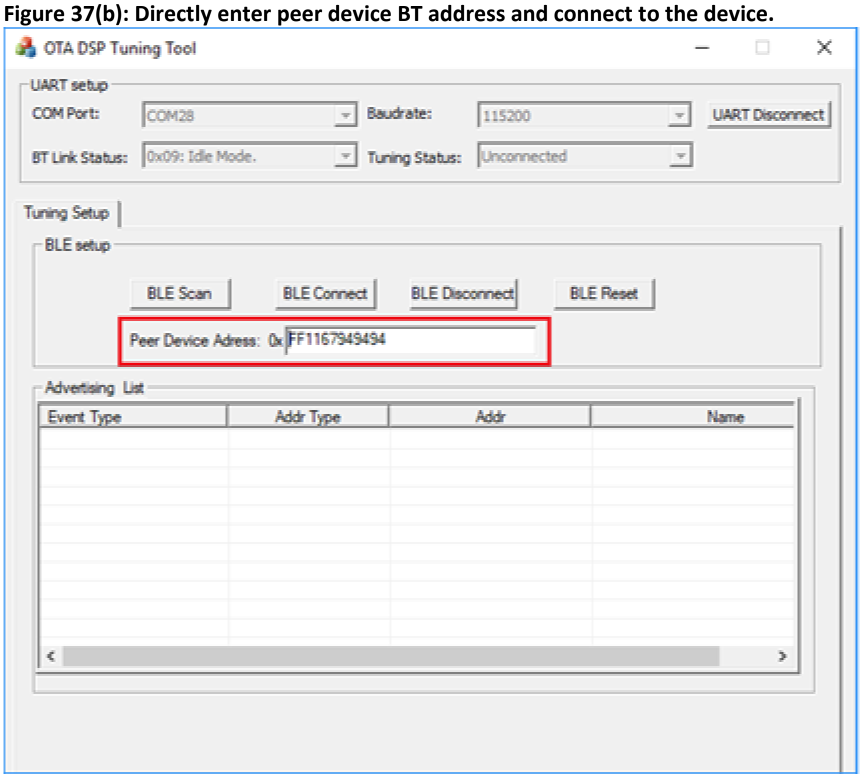


Figure 37(b): Directly enter peer device BT address and connect to the device.



Step 5: To setup the OTA tuning connection, select the device in advertising list and click on BLE Connect button as shown in Figure 38(a). Once the connection is setup you will get pop up saying connection is setup and status bar in main tool window and OTA Tuning dialog gets updated as shown in Figure 38(b).

Figure 38(a): BLE Connection setup.

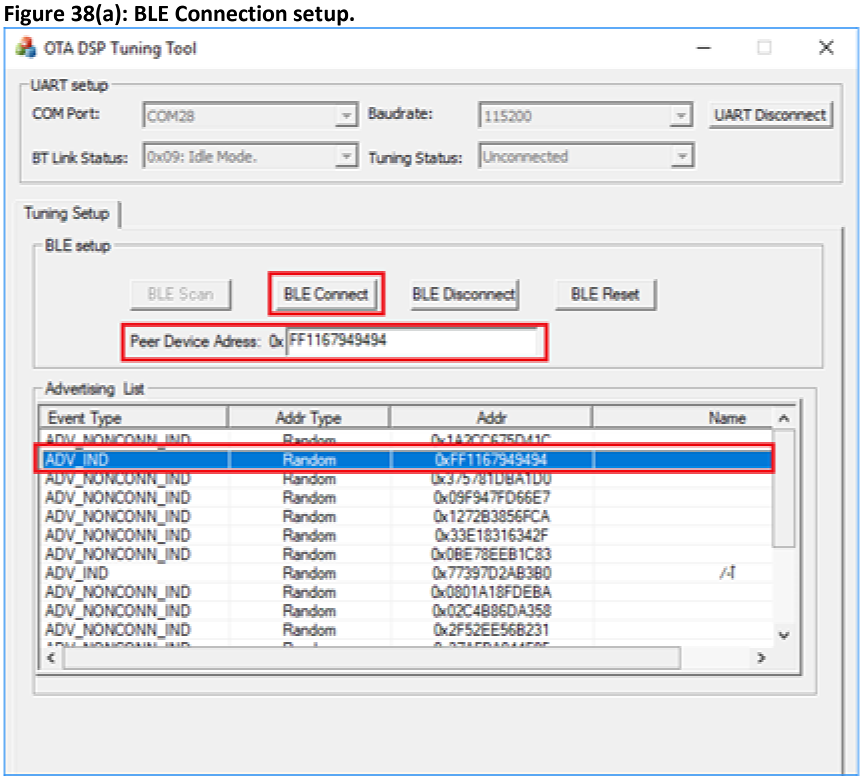
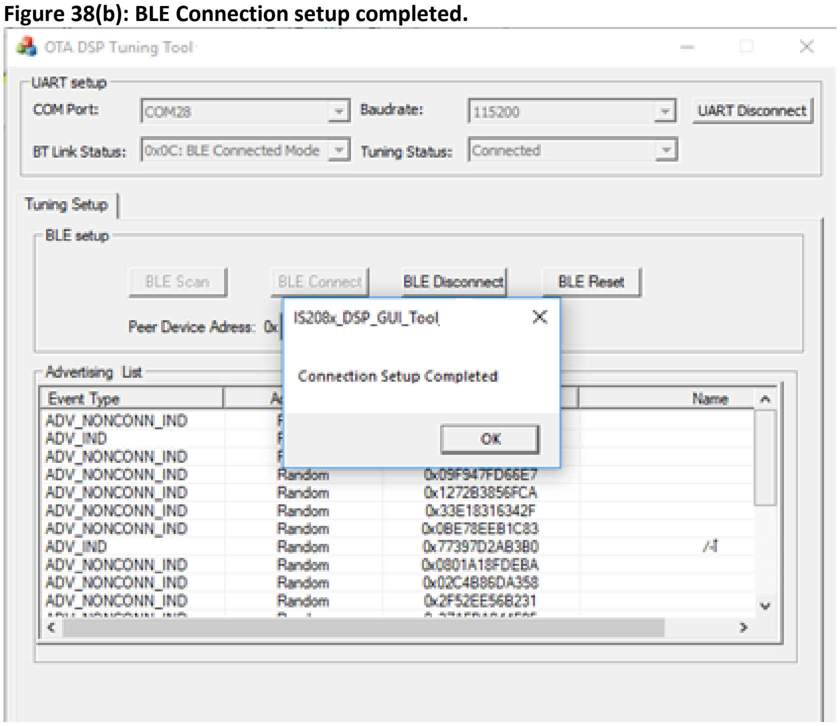
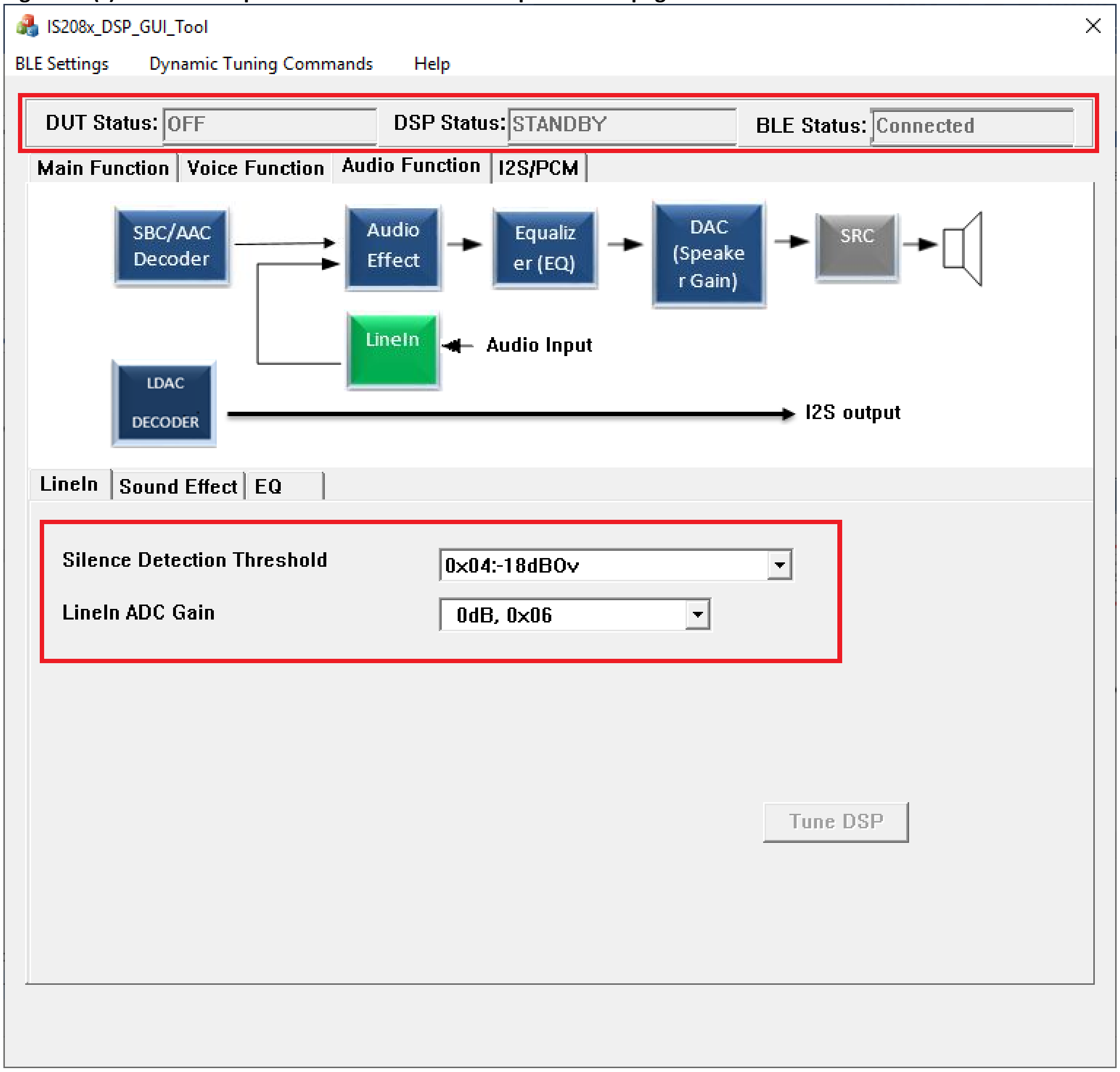


Figure 38(b): BLE Connection setup completed.



Once the Connection is setup, status bar in the main window will get updated showing current status of BLE, DUT and DSP. And reads the device flash and updates GUI with read flash data of BM83 as shown in Figure 38(C).

Figure 38(c): Status bar update and GUI Parameters update in all pages of DSP GUI Tool.



3.21.7.2 Dynamic OTA DSP Tuning

Change any of the parameters in audio and voice tabs to enable the Tune DSP button at the bottom of each page. And click on ***Tune DSP*** button and wait for success or failure notification from the DUT. As shown in Figure 39(a) and 39(b). At once only one Tune DSP of page of tool can be in progress.

NOTE: Voice function pages will be enabled for tuning only after voice call setup and during voice call progress audio function pages will not be enabled. Similarly, audio function pages will be enabled only after audio streaming is setup and voice function pages will be disabled during audio streaming. I2S page parameters OTA tuning is disabled in dynamic mode.

Figure 39(a): Changing Parameter and tuning DSP

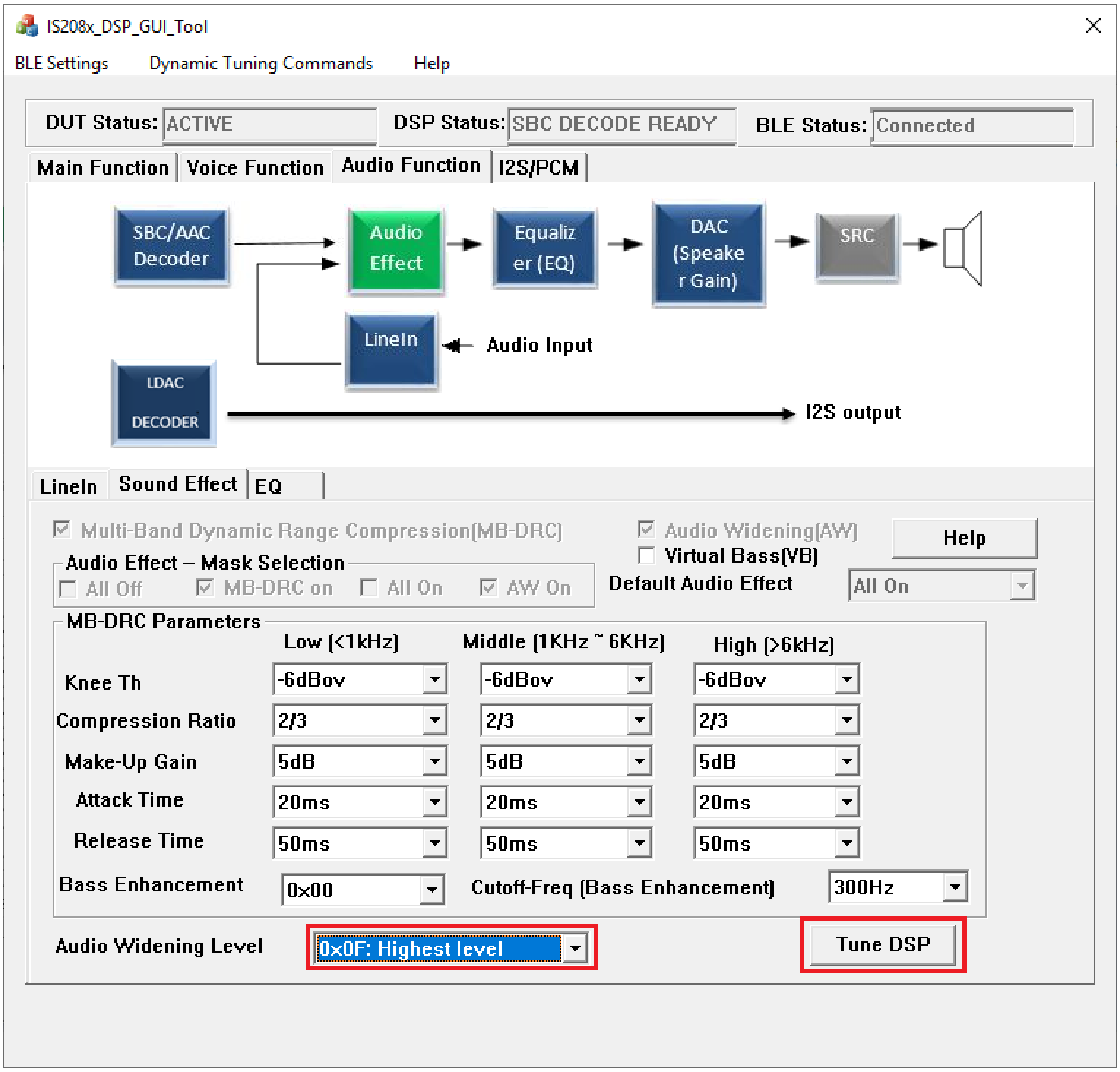
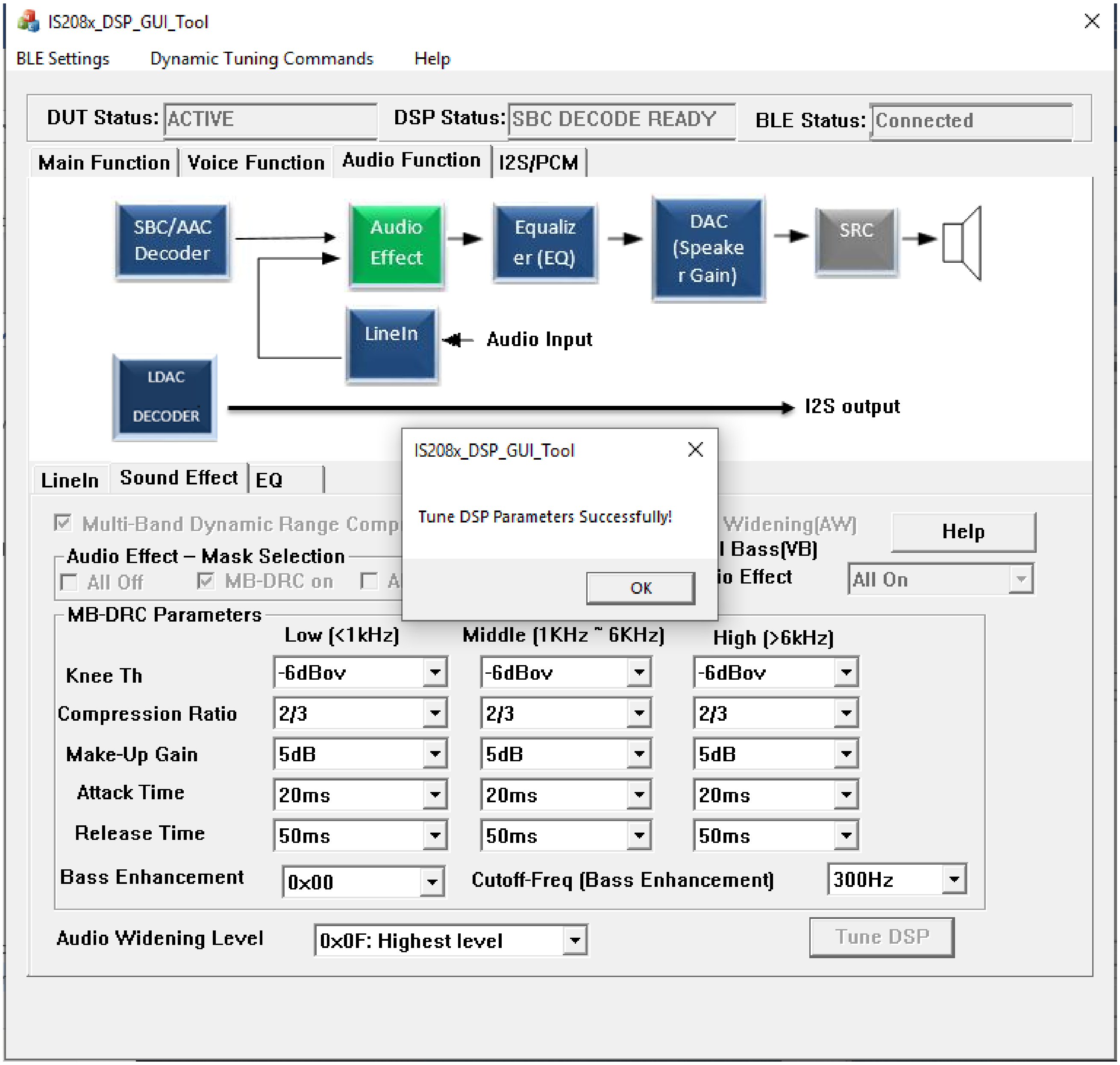


Figure 39(b): Notification from DUT on tuning DSP.



3.21.7.3 Dynamic OTA Tuning Commands

In menu bar under ***Dynamic Tuning Commands*** you will be having different commands like ***Factory Reset***, ***Reset DSP***, ***Reset DUT***, ***Save to Flash*** and ***Reset Parameters***. These will be enabled only in Dynamic tuning mode. Once the connection is disconnected then Dynamic Commands will be disabled. As shown in the Figure 40(a) and Figure 40(b).

- **Reset Parameters:** Resets the configuration data to Flash Configuration Data.

  On Reset Parameter the BLE connection will be disconnected. User must manually reconnect once again with the DUT.

  Note: To reset only selective set of parameters in the tool, you can use "Tune DSP button" present at the bottom of each page. If parameters are changed in multiple pages clicking on "Tune DSP button" in any page takes all the changed parameters from all the pages and sends over the air to tune DSP.
- **Reset DSP:** Resets the DSP.

  Note: The parameters are not reset as part of this procedure. The PC tool user needs to explicitly do "Reset Parameters"

- **Reset DUT:** Resets the DUT.
  On Reset DUT the BLE connection will be disconnected. User must manually reconnect once again with DUT.
  Note: The parameters are reset, as part of this procedure. The tool user needs to explicitly do "Save To Flash", if required to save the parameters to flash.
- **Save to Flash:** Save the configuration parameters to Flash (runtime section). This allows the parameters to be available after power cycle also.
- **Factory Reset:**
  Re-store the Configuration Data back to Factory Settings. This also resets the DUT after restoring the Configuration Data. The PC tuning tool should re-connect to the DUT after power-cycle.

Figure 40(a): Save to Flash Dynamic Tuning Command.

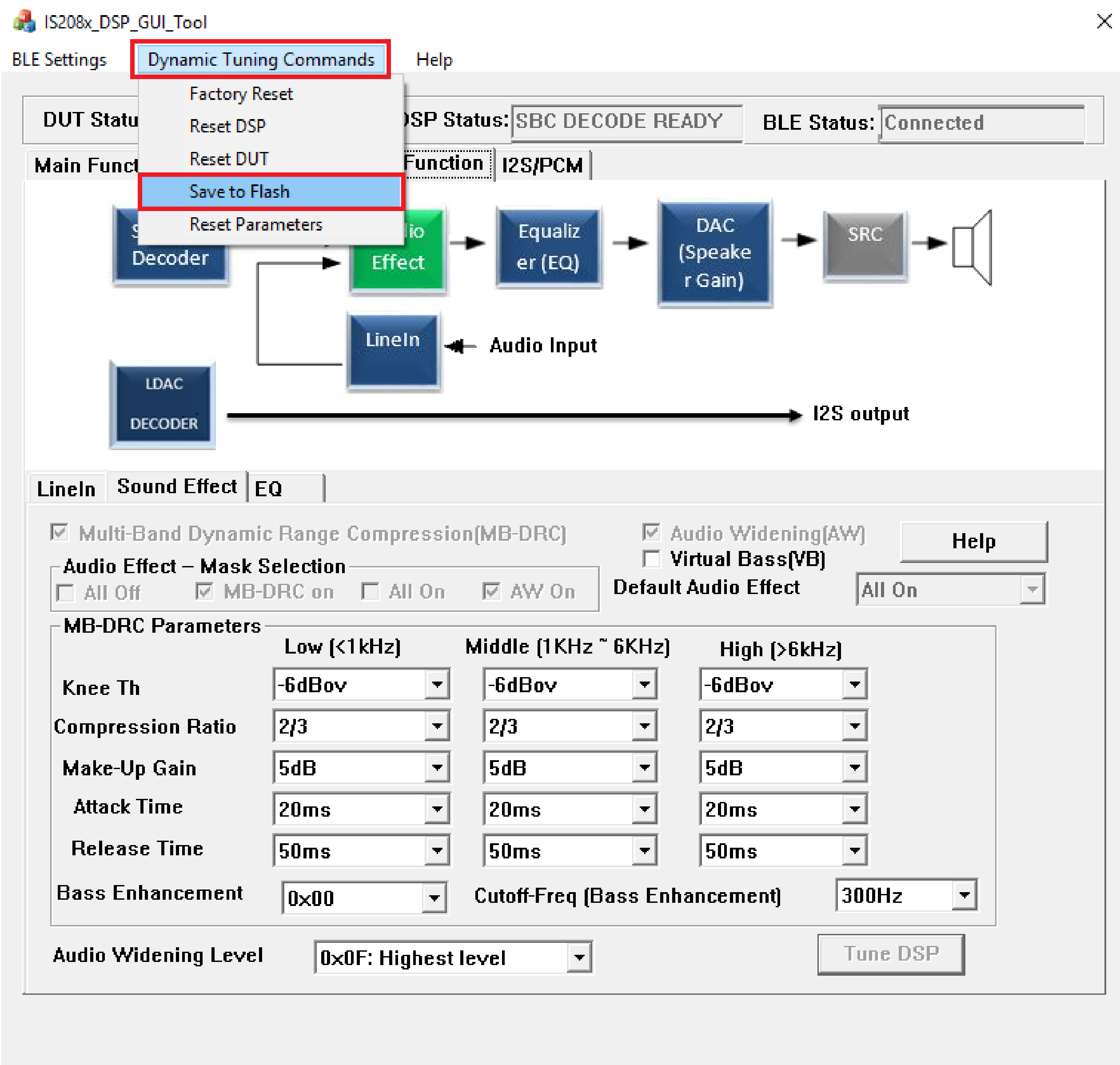


Figure 40(b): Save to Flash Dynamic Tuning Command success notification.

3.21.7.4 OTA Tuning BLE Disconnection

You can use BLE disconnection button present under BLE Settings menu in main window or you can use disconnect button present in OTA tuning dialog box as shown in Figure 41(a) ,41(b) and 42. And BLE Reset button is used for BM71 dongle reset and this will disconnect the BLE connection.

NOTE: In case of BM70 dongle is not responsive and not able to scan, do software reset by clicking on BLE Reset button and hard reset by clicking on Reset button in BM70/71 board. And try once again scanning. Click on BLE Reset button to enable scan button if scan button is not enabled at any point of time, but this will terminate your BLE connection.

Figure 41(a): OTA tuning BLE disconnection option in Setup dialog.

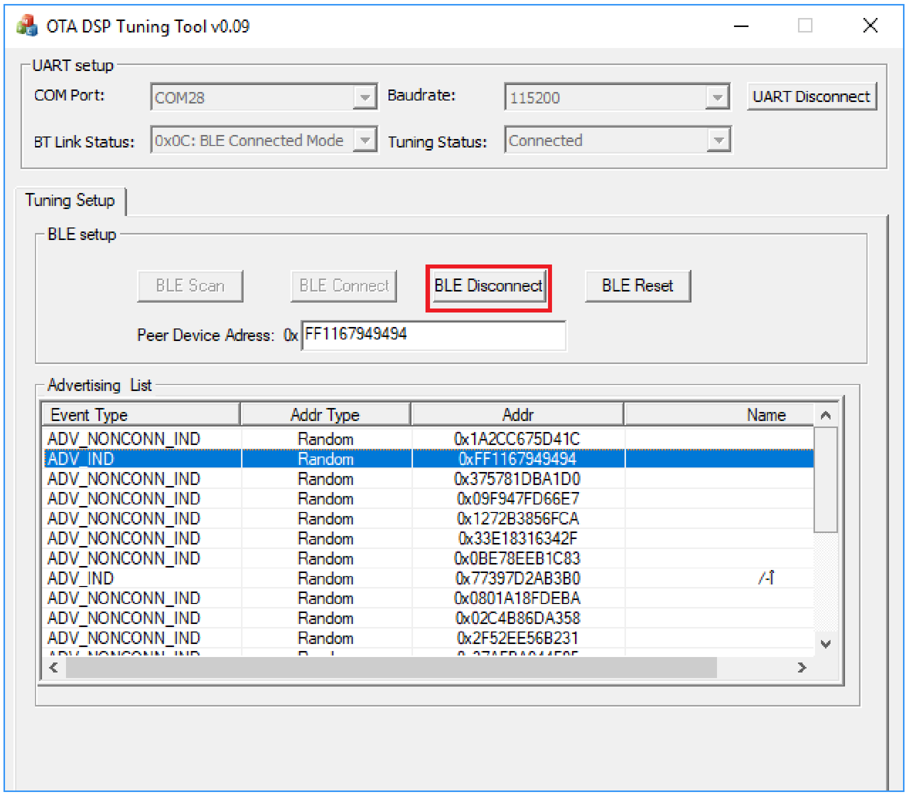


Figure 41(a): OTA tuning BLE disconnection option in main dialog.

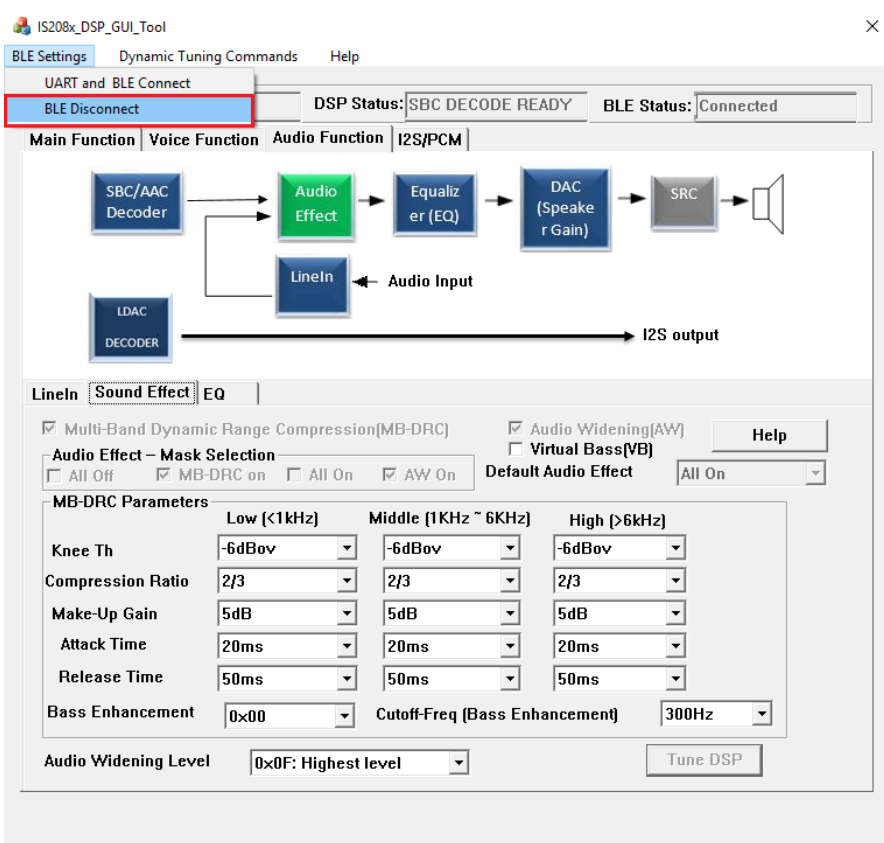
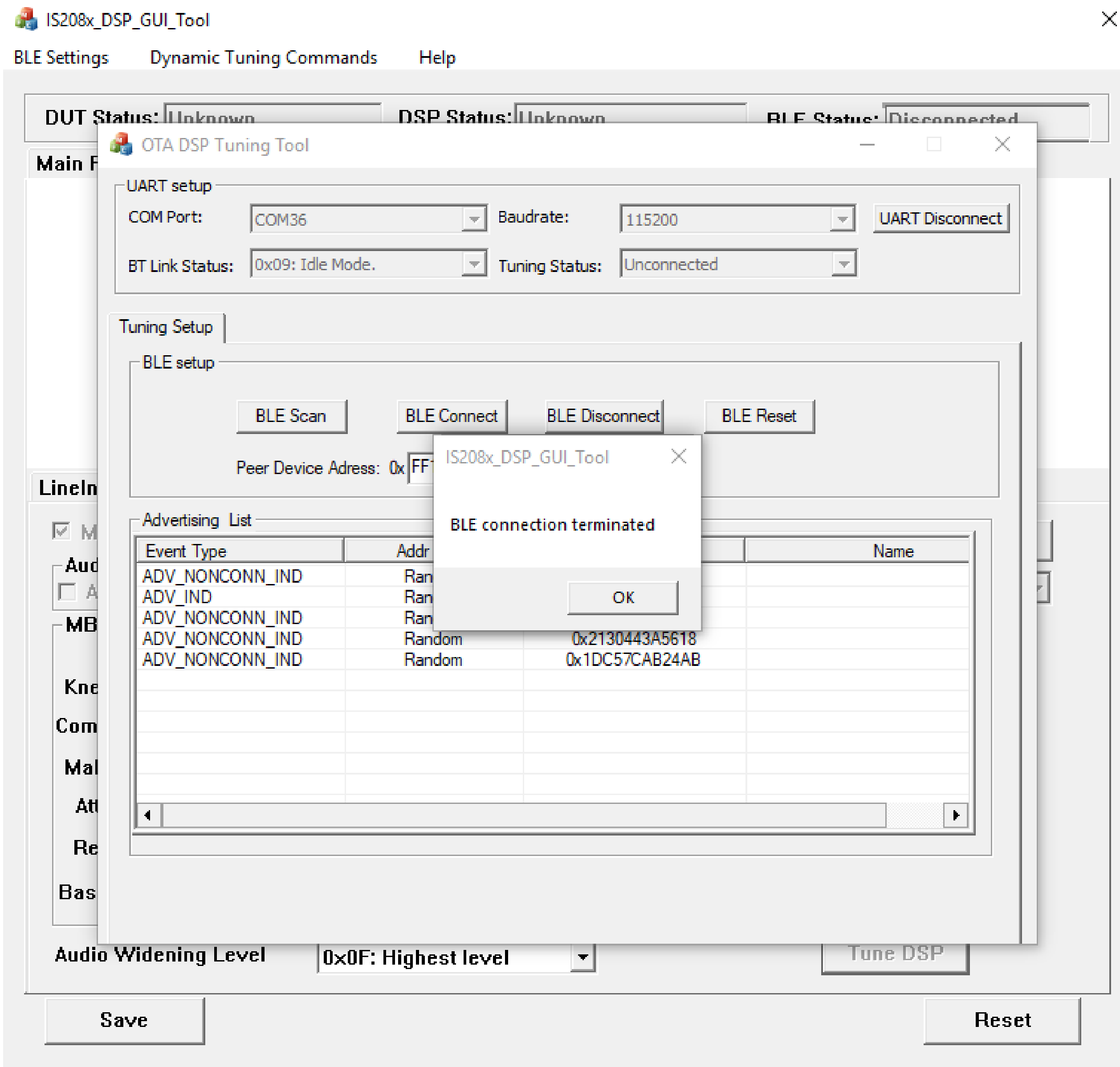


Figure 41(b): Disconnection complete notification


IS208x_DSP_GUI_Tool
BLE Settings
Dynamic Tuning Commands
Help
DUT Status: Unknown
DSP Status: Unknown
BLE Status: Disconnected
OTA DSP Tuning Tool
UART setup
COM Port: COM36
Baudrate: 115200
UART Disconnect
BT Link Status: 0x09: Idle Mode.
Tuning Status: Unconnected
Tuning Setup
BLE setup
BLE Scan
BLE Connect
BLE Disconnect
BLE Reset
Peer Device Adress: 0x
IS208x_DSP_GUI_Tool
BLE connection terminated
OK
Advertising List
Event Type
Addr
Name
ADV_NONCONN_IND
ADV_IND
ADV_NONCONN_IND
ADV_NONCONN_IND
Random
0x2130443A5618
ADV_NONCONN_IND
Random
0x1DC57CAB24AB
Audio Widening Level
0x0F: Highest level
Tune DSP
Save
Reset

3.19 BM83 Digital MIC Interface

3.19.1 Introduction

IS208X chips provide DIGMIC interface to connect PDM (Pulse-density modulation) digital microphones. This topic describes the UI settings and HW connections that need to be configured for interfacing a digital mic to the IS208X. The IS208X supports up to 4 digital Mics, which is equivalent to 2 stereo Mics. BM83 module supports only 2 digital Mics (1 stereo).

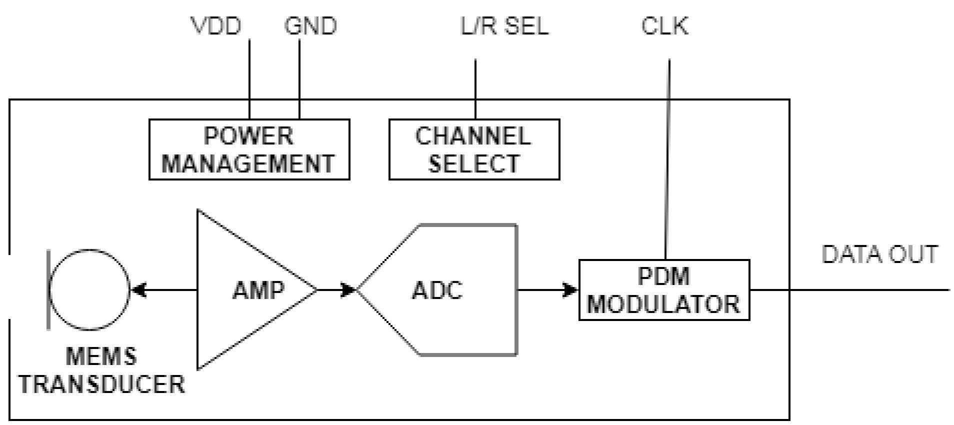
The digital mic path is multiplexed with analog mic path, in the audio codec subsystem. So, the user can choose either analog or digital mic (not both).

3.19.2 Basics of Digital Microphone

Digital microphones are immune to RF noise and electromagnetic interference as they have built-in analog to digital converter, permitting an all-digital audio capture path from the microphone to the DSP processor.

A typical digital microphone is a 5-pin module as shown below in figure 2.1. The MEMS (Microelectromechanical systems) transducer is a variable capacitance that converts the acoustic sound pressure to a voltage. The amplifier buffers this voltage to provide an amplified signal to the PDM converter. CLK controls the sampling rate of the PDM modulator. The PDM modulator converts the analog signal to a series of one-bit data called pulse density modulated signal which is outputted through DOUT. L/R (Left Right) SELECT decides the edge of the clock on which the data gets asserted based on its connection to VDD or GND. Typically when L/R SELECT is connected to VDD the data is asserted on rising edge of the clock and when connected to GND the data assertion is on falling edge of the clock.

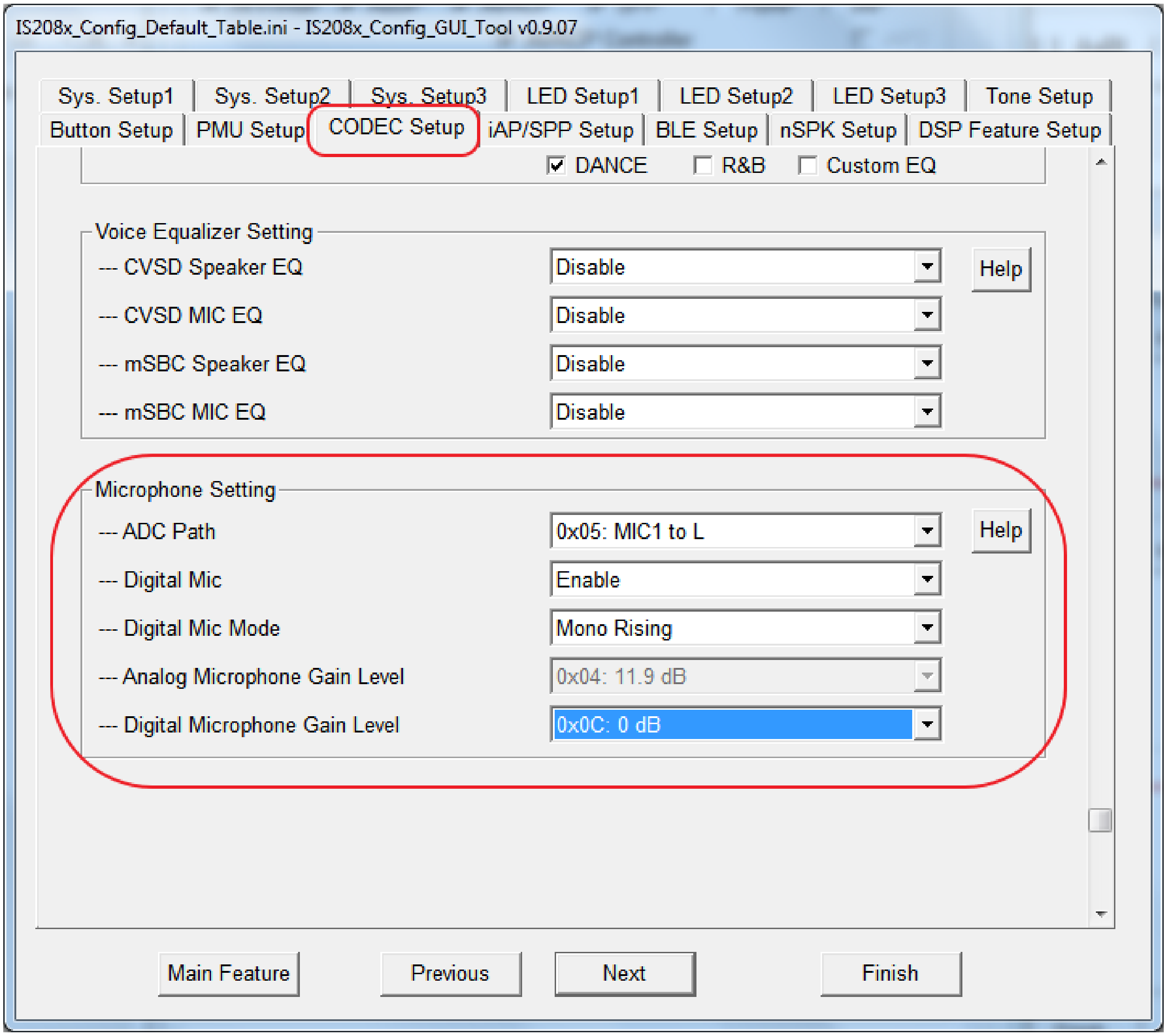
Figure 2.1: A typical Digital Microphone Module



3.19.3 UI Configuration Tool Settings

The digital mic selection and configuration parameters are present in the CODEC setup tab of IS208x_Config_GUI_Tool.

Figure 3.1: Digital mic configuration



1. **ADC Path**: By default, BM83 chooses the left channel for the mic. User can select the right channel, by selecting ‘MIC2 to L’. As this is a common setting for analog and digital mic, if the user selects ‘MIC2 to L’, even in analog mic case analog ‘MIC2’ will be selected instead of analog ‘MIC1’.

2. **Digital Mic:** Choose Enable from the drop-down menu to enable digital mic.

3. **Digital Mic Mode:** Select Mono Rising, to configure the digital mic for rising clock edge data assertion. Select Mono Falling, for falling clock edge data assertion.

4. **Microphone Digital Gain Level:** Default gain is at 0dB. User can increase or decrease the digital mic volume by choosing the values and gain mapping is given in below table.

Value in Hex	Digital Gain in dB
0x1	-54
0x2	-48
0x3	-42
0x4	-36
0x5	-30
0x6	-24
0x7	-18
0x8	-12
0x9	-8.5
0xA	-6
0xB	-2.5
0xC	0
0xD	1.93
0xE	3.52
0xF	4.85

Save the config file when all the settings are updated as desired.

Note: These UI settings should match with the 'Digital Mic Interface' settings mentioned in section 3.22.4.

3.19.4 Digital MIC Interface

This section explains how to interface a digital mic module to the BM83. Connect VDD_DIGMIC pin to VDDIO in J509 by a jumper to power on the DIGMIC module. As dual mic solution is not yet ready user must use only one channel at a time. The digital mic channel interface is brought out on the board through the headers J503 and J502 for left and right channels respectively. Refer to the schematic for the pin information [2]. Jumper settings decide the rising or falling edge mode for the digital mics.

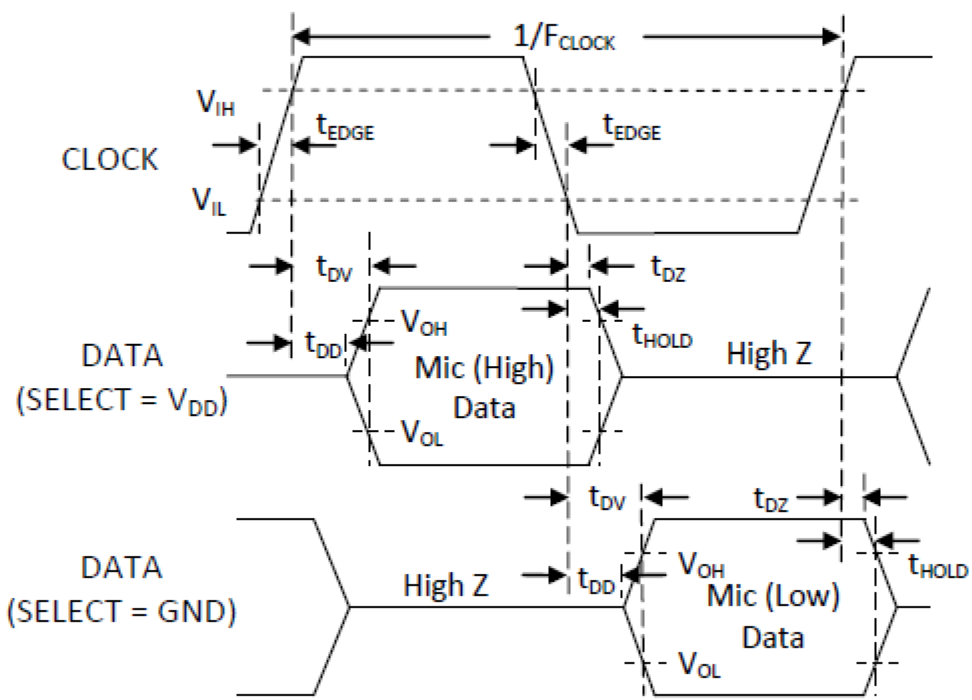


Figure 4.1: Data assertion for rising and falling edge of the clock of a digital mic

The below figure shows DMIC interface to the digital mic left channel of the BT5511_EVB.

Figure 4.1: DMIC1 left channel rising edge configuration.

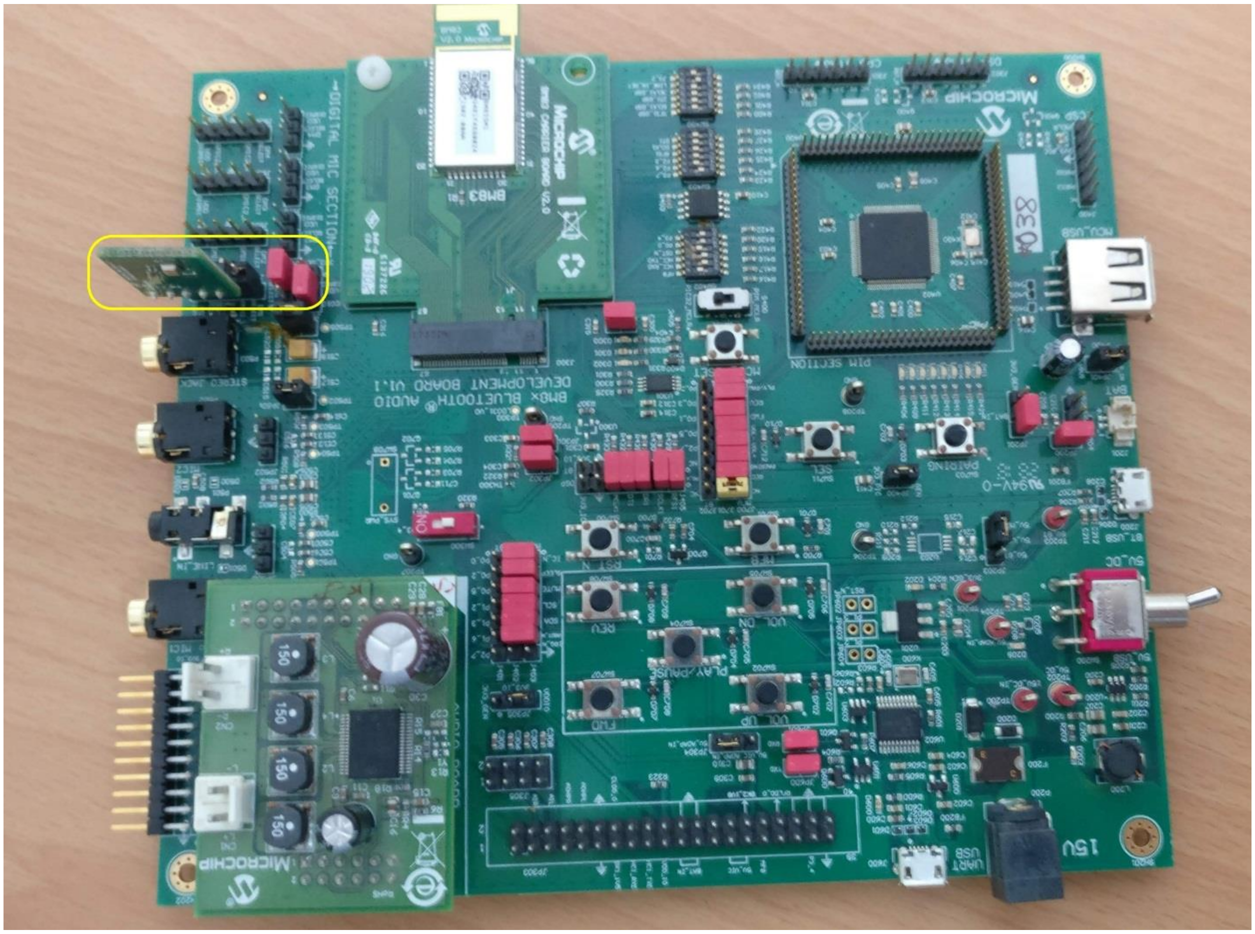
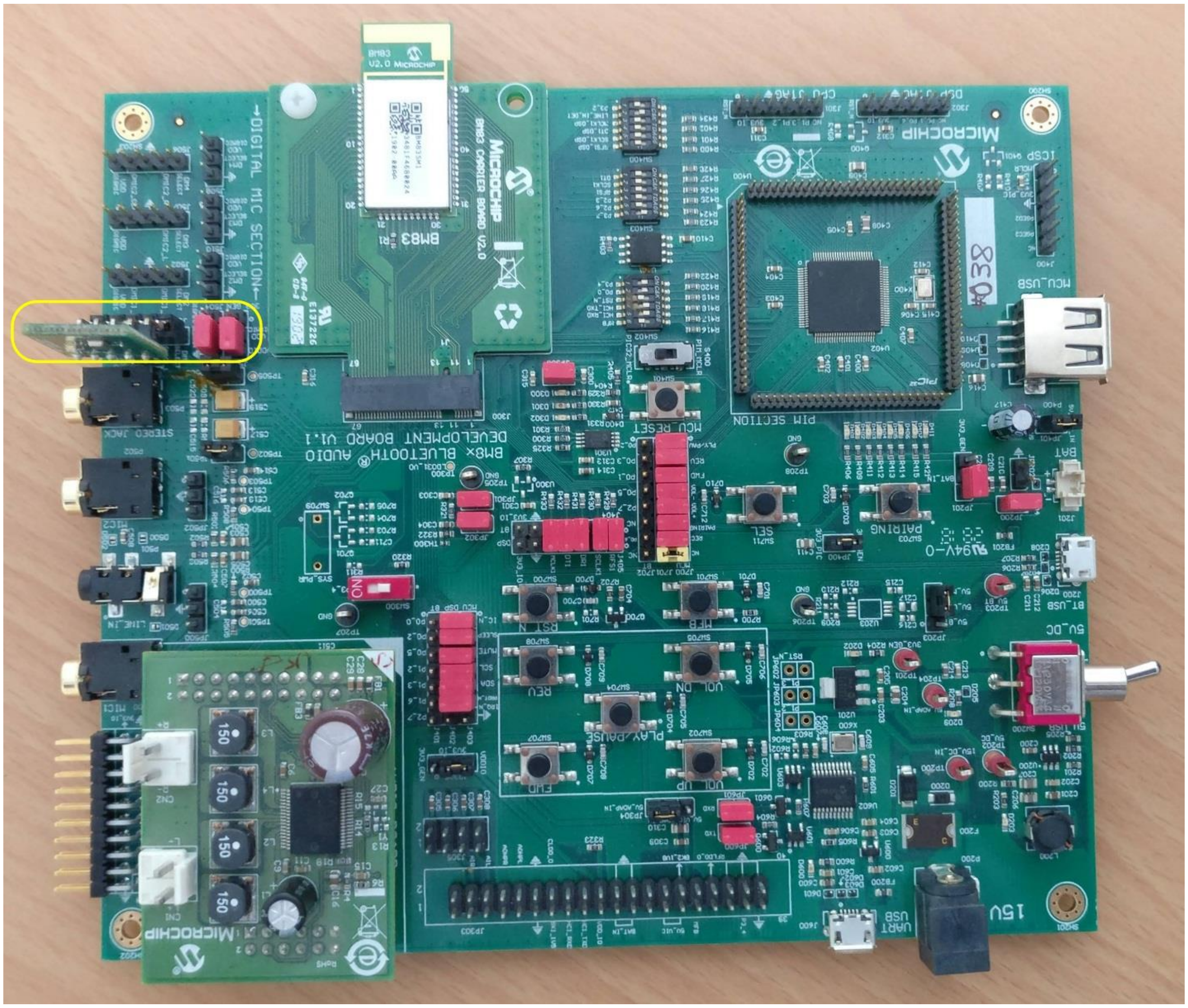


Figure 4.2: DMIC1 left channel falling edge configuration



The below figure explains DMIC interface to the digital mic right channel of the BT5511_EVB.

Figure 4.3: DMIC1 right channel rising edge configuration

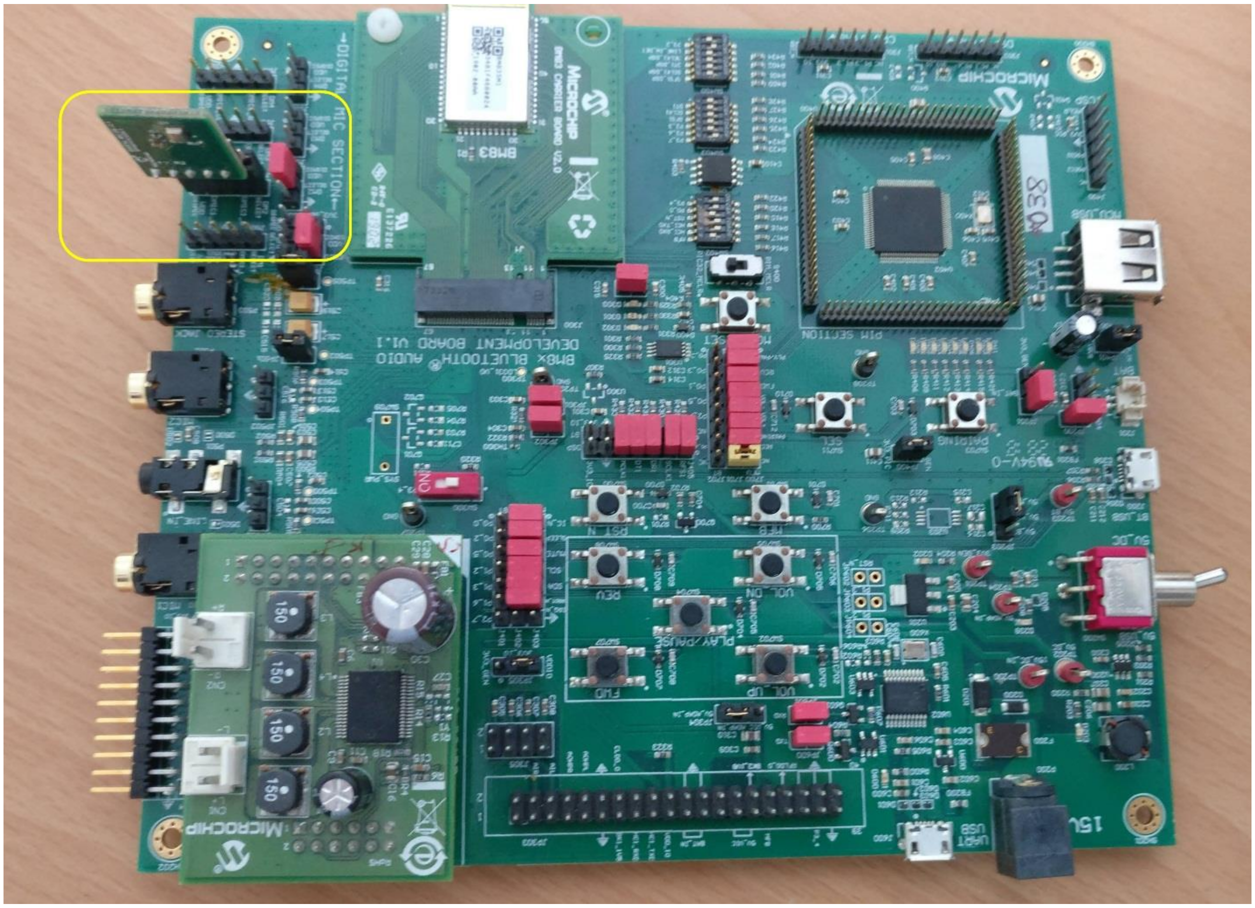
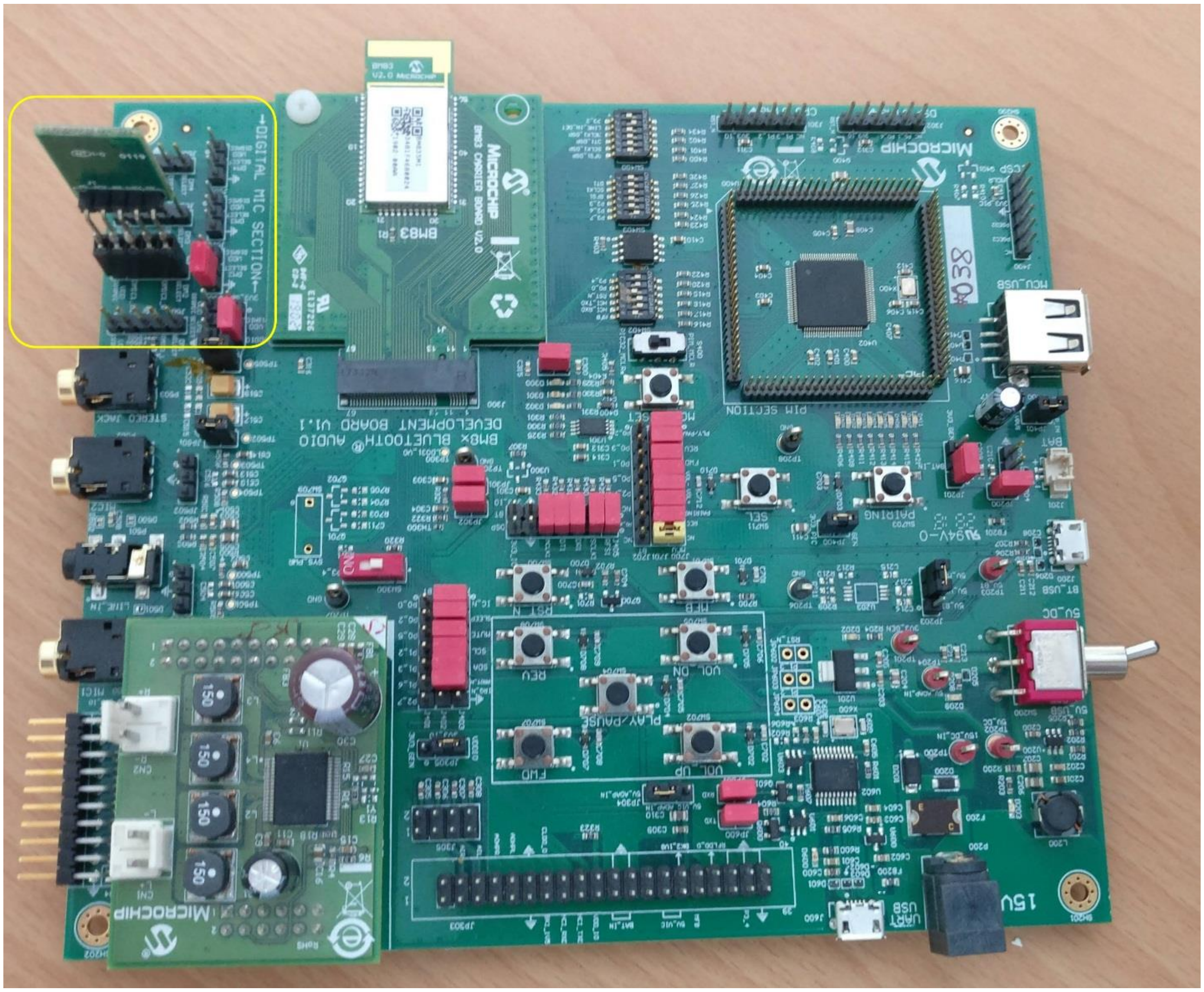


Figure 4.4: DMIC1 right channel falling edge configuration



Note:

1. When using right channel, the user must select the ‘ADC Path’ to ‘MIC1 to R’ as explained in section 3.22.3.

2. User must follow the data sheet provided by the digital mic vendors, to ensure correct pin settings for rising or falling edge configurations.
    a. SiSonic SPH0641 digital mic uses SELECT connected to VDD for data assertion on the rising edge of the clock

3. Ideally, a digital mic that is configured for data assertion on the rising edge of the clock must have valid data in the rising edge of the clock and a high Z state in the falling edge. But some digital mics may hold valid data on both the edges. Thus, even though the SELECT pin is connected to VDD, it will work for both the rising and falling edges of the clock.

4. Ideally, a digital mic that is configured for data assertion on the falling edge of the clock must have valid data in the falling edge of the clock and a high Z state in the rising edge. But some digital mics may hold valid data on both the edges. Thus, even though the SELECT pin is connected to VDD, it will work for both the rising and falling edges of the clock.

3.19.5 Limitations

- As of now, voice features that use two microphones is not ready, the user can't choose stereo mode for digital mic. Even if chosen only one channel will be selected depending on the configurations set in the ADC Path as mentioned in section 3.22.3.
- Most of the PDM digital microphones support up to 4MHz clock frequency. The DIGMIC module support only 128x over sampling rate. To support a 48KHz sampling rate, the DIGIMIC module must be clocked at 6.144 MHz (i.e., 48*128) and hence the External ADC (48KHz) option in the DSP config tool must not be used. Effectively, digital mic works only for 8KHz or 16KHz sampling rate.

4. APPENDIX

4.1 Abbreviations

Full forms for the abbreviations used in this application note.

Abbreviation	Full Form
ADC	Analog to Digital Converter
ANAMIC	Analog Microphone
CH	Channel
DIGMIC	Digital Microphone
DSP	Digital Signal Processor
dB	deci Bell
LR	Left Right
MEMS	Microelectromechanical systems
MIC	Microphone
PDM	Pulse-Density Modulation
RF	Radio Frequency

4.2 Reference

1. BM8x EVB Schematics

5. REVISION HISTORY

Version	Date	History
D00R01	2018/06/20	First Edition of IS208x Config Tool User Guide
D00R02	2019/04/17	Second Edition of IS208x Config Tool User Guide with Contents about DSP GUI Tool and BM83 Digital MIC Interface