DSP Config Tool User Guide

V1.0

2023/4/19

Revision History

Date	Version	Comments	Author	Reviewer
2023/04/18	1.0	Initial release	Ryan Chang	

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1. Introduction

This documentation provides detailed guidance of graphical user interface (GUI) to fine-tune the behavior related to the DSP FW part in RTK 87xx IC series. This GUI tool, the DSP configuration tool, allows developers to configure parameters of built-in signal processing algorithms for voice and audio applications

In addition, this DSP configuration tool also provides user-friendly parameter fine-tuning functions, which can greatly shorten the turnaround time of parameter-tuning, and they are:

- Over-the-air parameter configuration
  - Run-time DSP configurations of embedded voice and audio enhancement functions via the wireless Bluetooth link.

Finally, this tool also supports the generation of DSP configuration files for RTK 87XX toolchain including flash download and mass-production software tools.

The terminology which will be used in the rest of this document is listed in Table 1-1.

Table 1-1: Terminology.

Abbreviations	Descriptions
NR	Noise reduction
AEC	Acoustic echo cancelation
AES	Acoustic echo suppression
AGC	Adaptive gain control
Trans. EQ	Transducer equalizer
MB-AGC	Multi-band automatic-gain control
APT	Audio pass-through
LL APT	Low Latency Audio pass-through
Line-In	A analog/digital input signal looped into DSP and then playback via analog/digital output interface as illustrated in Figure 1-2.
HFP	Hand-free profile, a Bluetooth connection profile dedicated for serving voice communication.
A2DP	Denote advanced audio distribution profile. A Bluetooth profile dedicated for wireless audio streaming.
SCO/eSCO	(Enhanced)-Synchronous-connection oriented link. Bluetooth connection types for HFP.

Far-end	Denote the remote side phone caller in Bluetooth HFP mode as illustrated in Figure 1-1.
Near-End	Denote the user of the Bluetooth audio device as illustrated in Figure 1-1.

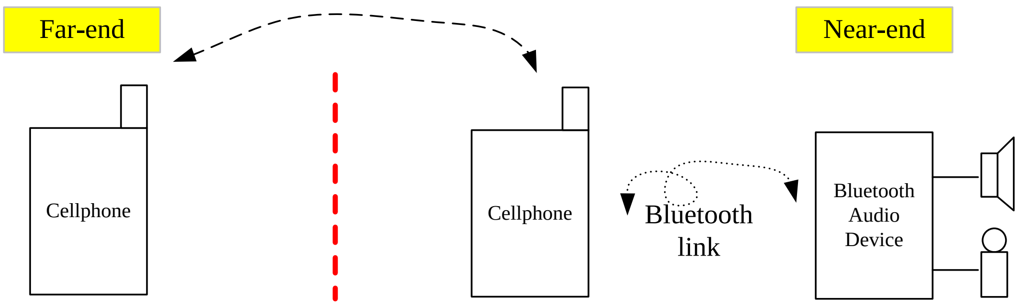


Figure 1-1: Illustration of far-end and near-end sides.

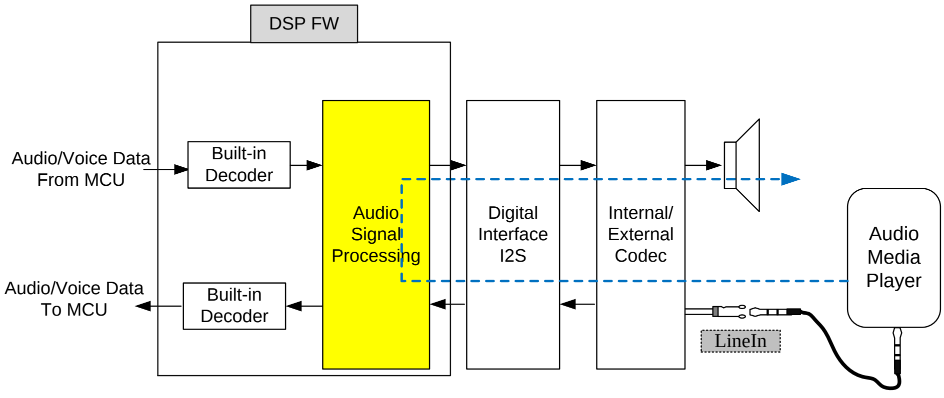


Figure 1-2: Illustration of Line-In loopback application.

2. Getting Started

2.1 Development Flow

Before starting the development, one must make sure that types of IC package and ID number. Subsequently, Figure 2-1 shows the development flow for Bluetooth audio product based on RTL 87XX DSP configuration tool generates configuration binary file for the later MP tool (See *MP tool user guide*) which then merges binary files of UI tool (Also see *UI tool user guide*) generated files, DSP system binary file(s) ,MCU binary file(s), etc.

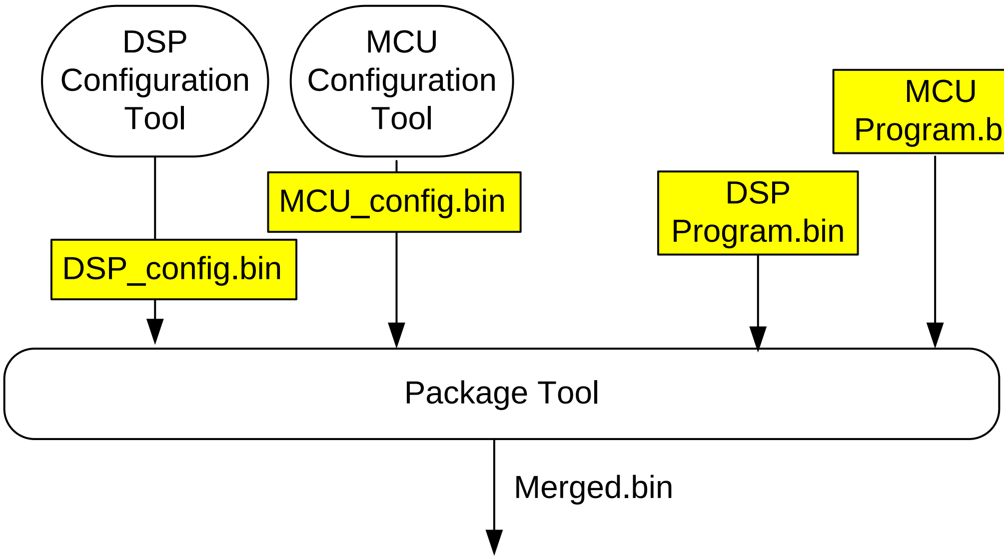


Figure 2-1: Illustration of RTL 87 xx IC development flow

2.2 Bluetooth Connectivity and Its Hardware/Software Installation

Since the function for over-the-air parameter update via Bluetooth link is available in the tool, the developer must make sure that RTL dongle RTL8761AUV or RTL8761AU is on hand and its Bluetooth driver correctly installed in the Windows OS.

To disable the built-in Bluetooth adapter, one can find the Bluetooth device via “My Computer / Right Click / Properties / Hardware / Device Manager / Bluetooth Radio” in the Windows 7 operation system (OS). After disabling the Bluetooth adapter hardware, the removal of Bluetooth driver program is also necessary.

In the case of successful installation of Bluetooth drivers, the "Device Manager" shows the "Realtek Bluetooth 4.0 Adapter" and no driver of other Bluetooth IC vendors shall be still exist because it could possibly affect the operational behavior of Realtek Bluetooth adapter.

Figure 2-2: Illustration of successful driver installation of Realtek Bluetooth Adapter.

3. Overview of DSP Configuration Tool

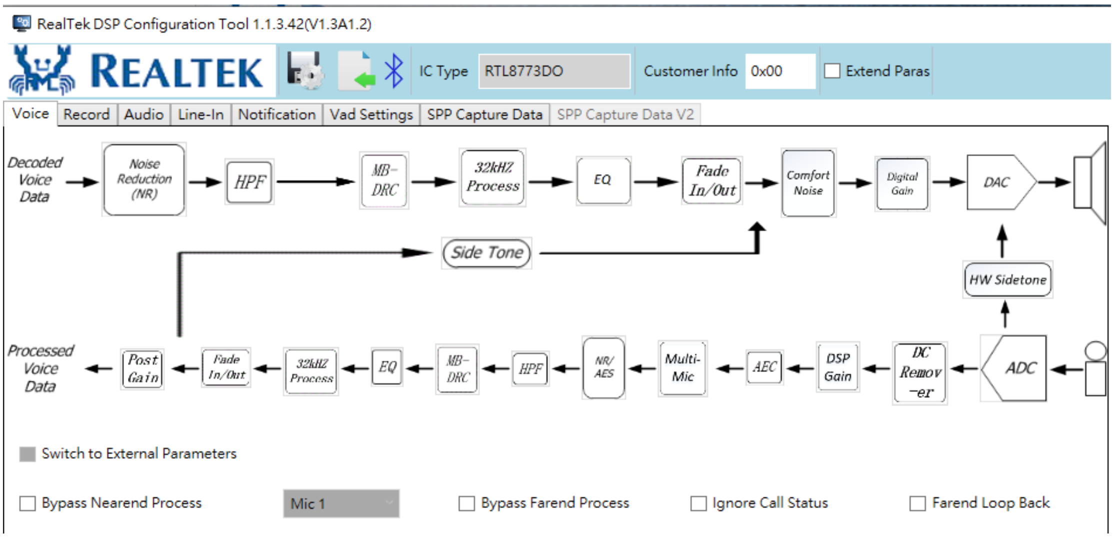


Figure 3-1: GUI of DSP configuration tool.

Figure 3-1 shows that the interface of DSP configuration tool which allows developers to easily configure the DSP algorithmic functions, analog ADC/DAC analog properties and run-time configuration over the Bluetooth link.

NOTE：The layout of main/sub graphic user Interface maybe changed without notice.

3.1 Available Subpages

Commonly used algorithmic parameters of Bluetooth speaker/speakerphone and headset applications, are classified as following subpages:

- Voice (HFP, SCO/eSCO)
- Audio (A2DP)
- Line-In
- VAD Settings
- Notification
- SPP capture

For Voice, Audio and Line-In subpages, parameters and processing flow of proprietary signal-processing algorithms and configurations are displayed and users can easily fine-tune to targeted performance.

3.2 Button Functions

Button functions illustrated in Figure 3-1 are summarized in Table 3-1.

Table 3-1: Function details of buttons in Figure 3-1.

<table>
<tr><th>Button Name</th><th colspan="2">Note</th></tr>
<tr><td>Save Config.</td><td colspan="2">To save DSP configuration in binary format compatible for the input of MP tool.</td></tr>
<tr><td>Import Config.</td><td colspan="2">To import saved DSP configuration. The import function also checks the version of imported DSP configuration compatible to the current DSP configuration tool or not.</td></tr>
<tr><td rowspan="4">Spp Control</td><td>Choose Device</td><td>Choose paired device in list</td></tr>
<tr><td>Connect</td><td>To build up Bluetooth SPP link with selected DUT.</td></tr>
<tr><td>Send Cmd to DSP</td><td>Send out DSP configuration based on current GUI settings to the DSP via Bluetooth SPP link/BT MCU to DSP FW. Be aware that this function needs to be authorized with correct password during connection process. This function is effective only when the DUT (device under test) is in active voice (HFP) or audio (A2DP/Line-In) modes.</td></tr>
<tr><td>EnableMultiLink</td><td>Enbale another BT device connect when doing online tuning with DSP TOOL</td></tr>
</table>

To send parameters in “Audio” tab via SPP, one must make sure that the DSP is playing sounds via A2DP or line-in. Similarly, one can only send parameters of “Voice” tab while the DSP is playing and recording with an HFP link. The parameters sent by “Send Cmd to DSP” are not stored in flash and thus are only effective until the A2DP/line-in playback or the HFP is terminated.

The text editor Customer info for user to save some information, for example, user parameter version number.

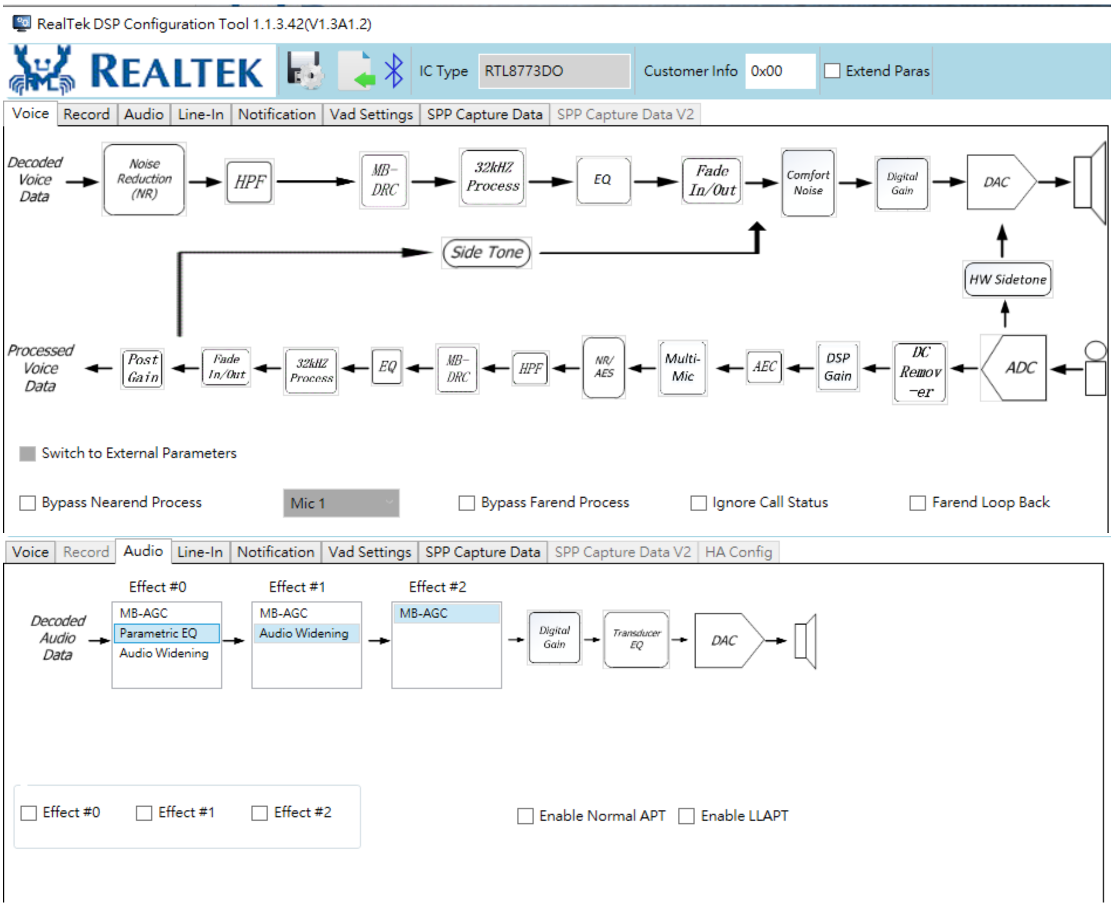


Figure 3-2: Illustration of algorithmic parameters and EQ coefficient transferrable to the DUT via "Send Cmd to DUT" function.

It is also note that, the Bluetooth DUT device shall be active and connected to the Realtek Bluetooth adapter, as shown in Figure 3-3, either in HFP, A2DP or Line-In mode as such the "**Send Cmd to DUT**" operation is effective and can transfer the desired algorithmic parameters and EQ coefficients to the DUT.

Please **NOTE** that whether Effect0~2 is checked or not in Mode#0~3, when "**Send Cmd to DUT**" in Audio or A2DP scenario, the parametric EQ parameters (currently selected EQ index in the parametric EQ interface) will be sent to the DUT. While if DSP configuration binary download to RTL8763Bx, the Mode and Effect must be valid or selected.

Figure 3-3: The Bluetooth DUT shall be connected via the windows interface in advance before connecting the device with DSP configuration tool via SPP link.

4. Voice Processing

This subsection introduces signal processing flow and block diagrams of voice algorithms as shown in Figure 3-1, consisting of NR, AEC, AES, HPF, Transducer EQ and MB-AGC functions. Table 4-1 summarizes the brief descriptions of each voice function. In addition, ADC/DAC gain settings are also configurable in this subpage.

Table 4-1: Functions in voice processing.

Abbreviation	Descriptions
AEC	To linearly remove echo signal mixing in the microphone input without affecting the quality of desired speech signal.
NR	To intelligently suppress stationary noise mixed within desired voice signal. Note that one-mic solution can handle stationary noise suppression while two-mic solutions can suppress stationary and dynamic noise sources.
AES	To apply the non-linear echo suppression to the AEC-processed signal. This could degrade the quality of desired speech signal but can efficiently remove echo from the microphone captured signal where the processed voice quality can be optimized by fine-tuning AES parameters.
Multi-Mic	To suppress noise coming from unwanted directions while preserving the desired voice signal. This function requires two/three physical microphones on the Bluetooth audio device with proper placements and distance to achieve optimal performance.
HPF	To filter out low-frequency signal typically caused by PCB power noise.
MB-AGC	To dynamically control the signal level within desired power ranges in adjustable 1 to 3 frequency bands.
Transducer EQ	This equalizer is used to compensate the imperfect frequency response of microphone and speaker devices.
DAC Gain	To configure DAC gain settings including gain mapping for each level, default gain level for HFP mode and number of gain levels.
ADC Gain	Configure ADC gain to amplify the level of microphone signal. This ADC gain applies the same settings to both L/R channels.

DSP Gain	To configure DSP gain for Primary/Secondary/FB mic
Sidetone	Choose suitable sidetone level to make user hear own voice slightly
HW sidetone	HW sidetone has lower latency however less DSP processing
Comfort noise	Choose suitable comfort noise level to make the phone call more natural

4.1 AEC

Selectable AEC parameters are given in Figure 4-1 and described in Table 4-2.

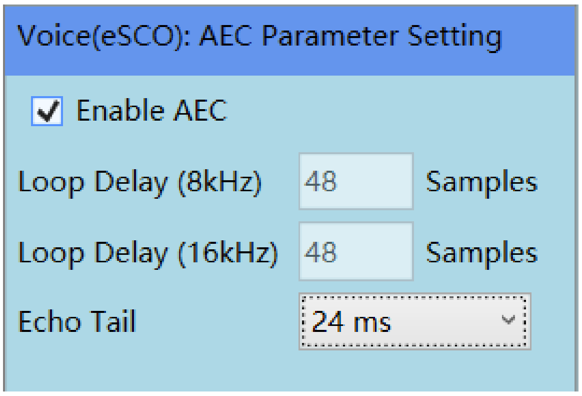


Figure 4-1: AEC parameters.

Table 4-2: Parameters of AEC.

	Note
Loop Delay 8K/ Loop Delay 16K	Denote the loop delay (unit in audio samples per channel) of echo signal passing through DSP output, speaker, the shortest echo propagation path in the air picked up by microphone and finally transfer back to the input of DSP as illustrated in Figure 4-2. If adopting either external ADC or DAC, the loop delay can only be selectable. In this case, the measurement of loop delay can be achieve by using the raw ADC/DAC data capture function, introduced in subsection, to measure the delay.
Echo Tail	Denote the longest echo propagation path as illustrated in Figure 4-2. A rule of thumb to configure the parameter of Echo Tail is that this parameter empirically needs no more than 24msec and 48msec for headset/headphone and speakerphone applications, respectively. Note that if echo tail is set beyond the required echo channel delay, echo

	performance will not be further enhanced due to more induced estimation error of residual echo signal.

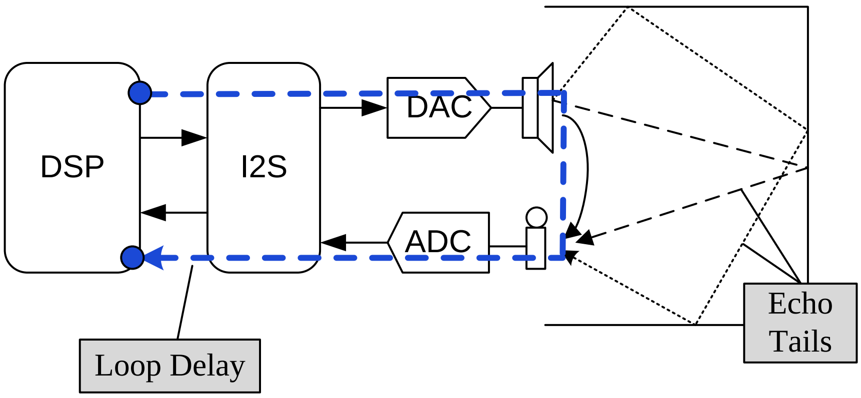
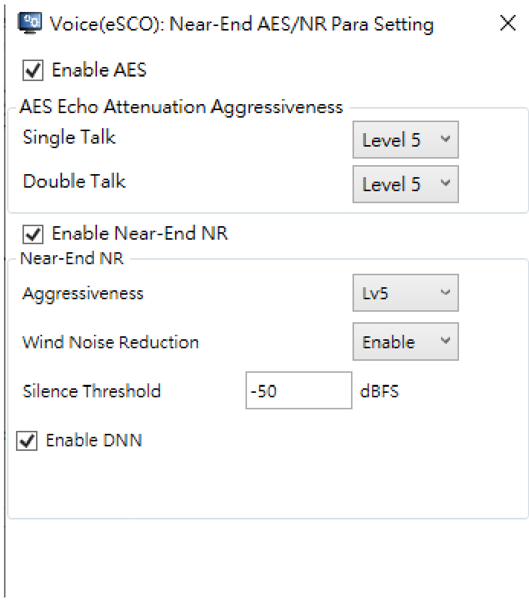


Figure 4-2: Illustrations of loop delay and echo channel.

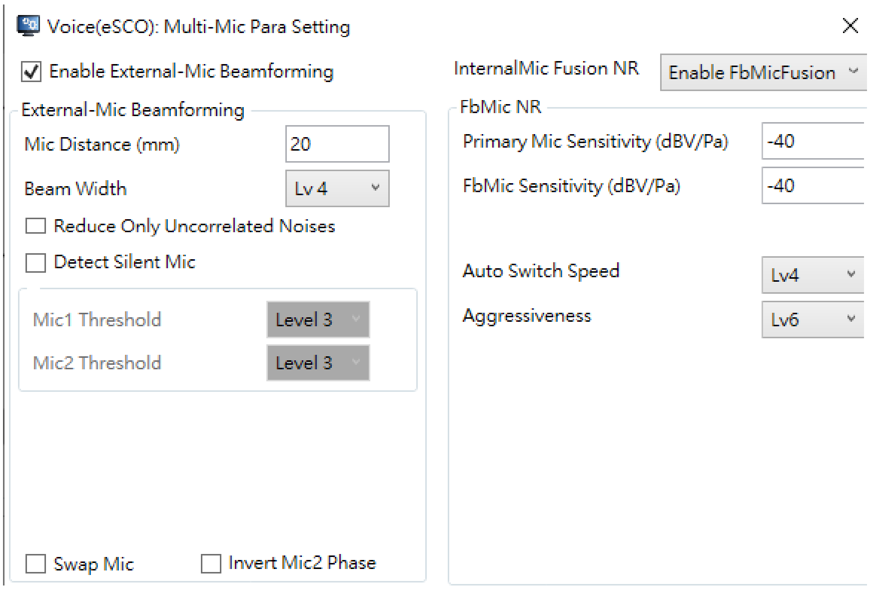
4.2 NR/AES

In this subsection, parameter interface of non-linear voice enhancement functions, NR and AES, are introduced, and Figure 4-3(a) and (b) show the tuning interface of far-end NR and near-end NR/AES functions, respectively. Note that, in the down-streaming (Far-end) signal in SCO(or Bluetooth HFP mode) mode, only mono channel is available according to the Bluetooth specifications and, hence, only one-mic NR function is necessary at the down-streaming (Far-end) path.

Realtek NR technology includes one-mic and two-mic NR functions to process pseudo-stationary and dynamic noise sources, respectively. The definition of pseudo-stationary noise is the signal pattern whose signal power remains relatively stable over a long period of time within a fixed frequency range. Examples of such pseudo-stationary noise are like sounds of car engines, air-conditioners, and white/pink noises, etc. On the other hand, dynamic noise refers to the signal source not holding the pseudo-stationary property and its incident angle is beyond the desired range with respect to the desired speech signal. Configurable parameters of pseudo-stationary NR function are summarized in Table 4-3.



(a)



(b)

Figure 4-3: (a) far-end NR settings (b) NR/AES/Dualmic settings at the near-end.

Table 4-3: Descriptions of NR and AES parameters.

Function	Parameters	Note
NR	**Single-Tone Pass-Through**	To enable tonal signal detection. If enabled, tone detection function can detect one and up to five tones, and prevent them from being suppressed by one-mic NR function. The tone detection function is especially useful in measuring SNR and THD, because the test instrument often uses single-tone, usually at 1KHz, sinusoidal wave

		as the reference signal.
	Max Noise Attenuation	Define the maximal suppression capability of NR function. Note that heavier NR suppression can remove more stationary noise from captured microphone signal in the tradeoff of more voice quality degradation.
	Aggressiveness	Denote the aggressiveness level to suppress noise. “Level 1” is the least aggressive level to suppress noise where “Level 10” is the most aggressiveness NR level. Note that less NR processing can preserve desired speech quality but more likely to have more un-processed ambient noise.
	Wind Noise Reduction	Enable Wind Noise Reduction for outdoor headset
	Enhance Noise Naturalness	Make idle noise more natural
AES	**Echo Attenuation Aggressiveness /Single Talk**	Denote the aggressiveness level to enable AES for echo suppression. “Level 0” is the least aggressive level to kick-in AES where “Level 14” is the most aggressiveness AES level. Note that less AES processing can preserve desired speech quality but more likely to have more un-processed residual echo.
	Echo Attenuation Aggressiveness /Double Talk	Denote the aggressiveness level to enable AES for echo suppression. “Level 0” is the least aggressive level to kick-in AES where “Level 14” is the most aggressiveness AES level. Note that less AES processing can preserve desired speech quality but more likely to have more un-processed residual echo.
Multi-Mic NR	**Mic Distance**	The distance between two microphones. The unit is mm (millimeter). The microphone distance must be in the range of 10mm to 70mm.
	Voice Direction	Denote the desired angle-of-arrival (AoA) direction of desired speech. Figure 4-4 illustrates the definition of

		AoA, θ, with respect to the placement of the microphone array. Selectable AoAs in the DSP configuration tool are given as +/-0, +/-15, +/-30, and +/-45 degree. Note that, due to the symmetric nature of two-microphone arrays, incident angles, θ and $-\theta$, are indistinguishable. Mic 0 is the microphone closer to the mouth.
	Post-Processing	Enabling post-processing of dual-mic NR can suppress more noises at the cost of degrading voice quality.
	Max Noise Attenuation	The maximal noise suppression by post-processing.
	Aggressiveness	To set the aggressiveness of the post-processing of dual-mic NR. Higher aggressiveness level can suppress more noises at the cost of degrading voice quality.
	Primary Mic Sensitivity/FbMic Sensitivity	To set Mic Sensitivity including primary mic and Fb mic
	Auto Switch Speed	When Fb mic detected speech signal,NR will change dual-mic to muti-mic
	Test Mode	To test if each microphone functions normally and if placement of the primary and reference microphone is correct. The placement of microphones is depicted in Figure 4-5. In test mode, dual-mic NR does nothing but output the signal of the user-selected microphone. One can test if the primary microphone functions normally by selecting “Mic 1”. Similarly, to test the reference microphone, select “Mic 2”.
	Swap Mic	With this option checked, the dual-mic NR algorithm takes the microphone connected to MIC2_P, MIC2_N as the main microphone and the microphone connected to MIC1_P, MIC2_N as reference microphone. Check this option only when the PCB layout is wrong.
	Invert Mic2 Phase	With this option checked, the mic 2 signal is multiplied by -1 before processed by dual-mic NR. Check this option only when the microphone polarity is wrongly connected.

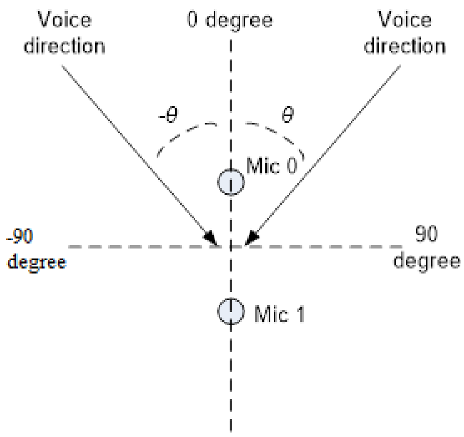


Figure 4-4: Illustration of dual-mic NR voice direction.

Realtek's dual mic algorithm can track the movement of the headsets. However, the most desirable angle shall be assigned by configuring the parameter "**Voice Direction**" as listed in Table 4-3.

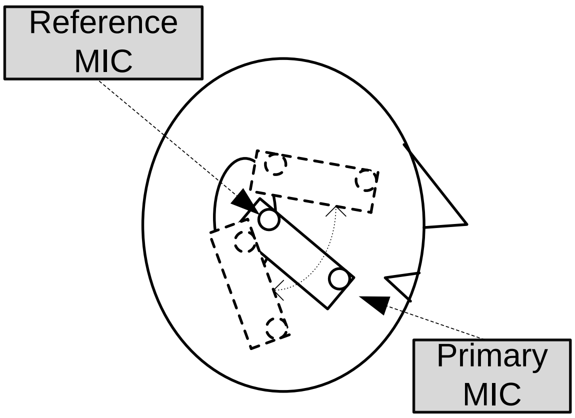


Figure 4-5: Requirement of microphone placement

4.3 Dual-Mic Industrial Design Guide

Microphone placement

The primary mic (connected to MIC1_P, MIC2_N pin) must be placed closer to the mouth while the reference mic (connected to MIC2_P, MIC2_N pin) must be placed closer to the ears as depicted in Figure 4-5, and that the distance between microphones must be in the range of 15mm to 30mm. The opening of the primary microphone should be directed to the mouth and the opening of the reference microphone should be directed away from the face. The structural design of the headset should avoid blocking the direct path from the mouth and the environment to each microphone. The reference microphone is better directed to the left or right of the one who wears the headset, not to the back, in order to receive sounds coming from the mouth directly. See Figure 4-6 for desirable microphone placement and see **Figure 4-7** for the placement that should be avoided.

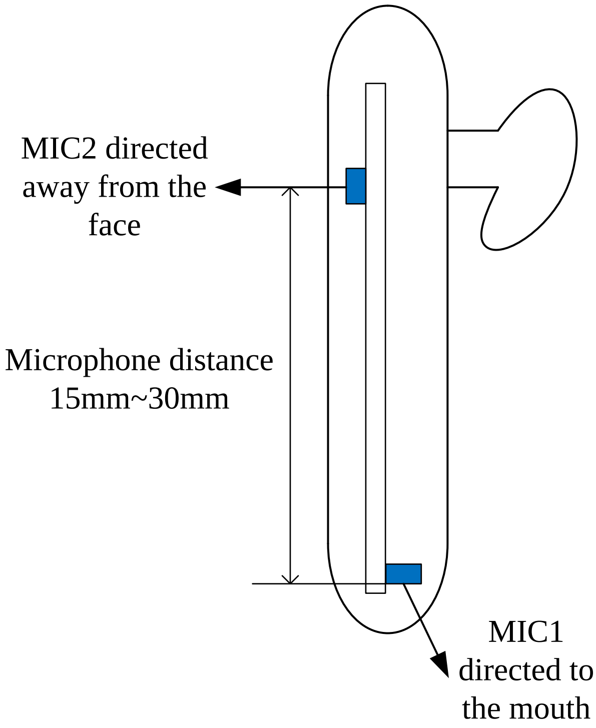


Figure 4-6 Recommended microphone placement

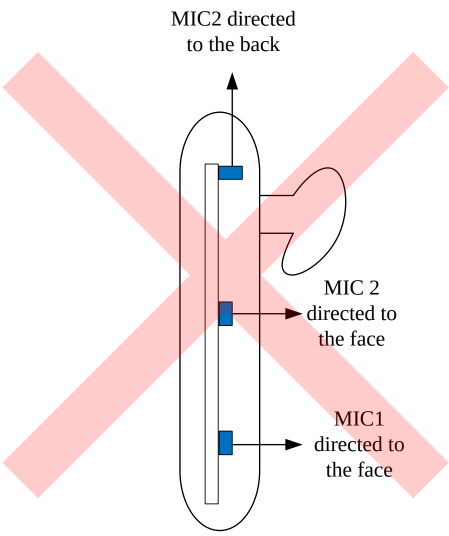


Figure 4-7 Undesirable microphone placement

Recommended microphone specification

Parameter	**Test Condition**	**Requirement**
Directivity		Omni-directional
Sensitivity	1kHz, 94dB SPL	>-45dBV/Pa
SNR		>60dB
THD	94dB SPL	<1%
	114dB SPL	<5%

Other design suggestion

The microphone pairs must be of the same part number. MEMS microphones are suggested since they have smaller production variance than ECM. The microphones should be isolated and sealed. Sealing and isolating microphones is to ensure that each microphone only picks up sounds from its own opening on the cover and to avoid multipath effect. Figure 4-8 shows an example of the acoustic paths that should be avoided. Place microphones closer to the opening on the cover can help to reduce multipath effect. See Figure 4-9. To test if both microphones are isolated properly, one can use the test mode of the dual-mic NR tuning interface to enable only one of the microphones and compare the sound level when the microphone's opening on the cover is sealed (with fingers) and when it's not. When the opening is sealed, the sound level picked up by the microphone should be 20dB (or more) less than the sound level when the opening is not sealed. The speaker should be placed closer to the reference microphone than the primary microphone.

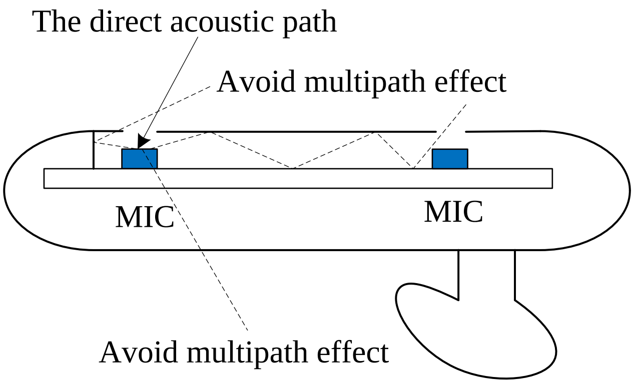


Figure 4-8 The dashed line indicates the multipath effect that should be avoided

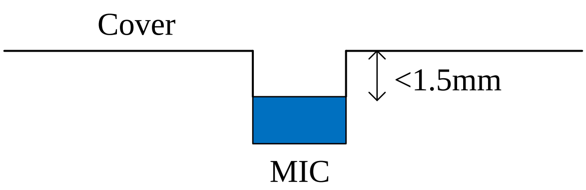


Figure 4-9 Place microphones closer to the opening on the cover

4.4 High-Pass Filter (HPF)

HPF is mainly to remove unwanted low-frequency signal/noise such as power noise generated by the power adapter, and wind noise, etc. Selectable cutoff frequencies range from 90Hz to 300Hz. Note that higher cutoff frequency may also filter out the fundamental pitch frequency of the desired voice signal.

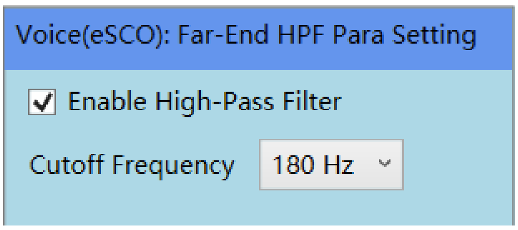


Figure 4-10: Parameter of HPF.

4.5 Transducer EQ

The transducer EQ supports up to 10-stage IIR filter for both speaker/microphone paths and 16KHz sampling rate.

As shown in Figure 4-11, tuning interface of customized EQ is based on the parametric equalizer where the parameter set of each EQ stage includes “Frequency”, “Gain” and “Q” as summarized in Table 4-4.

Please note that the number of desired EQ stages, “Stage” in Figure 4-11, must be selected to physically enable the EQ function. In addition, frequency response of each EQ stage can be displayed in logarithmic scale.

Table 4-4: Parameters of customizable EQ.

	Note
Frequency (Hz)	The desired center frequency of each EQ stage. The unit is Hz.
Gain	The boost/attenuation gain in dB scale at the selected frequency.
Q	The bandwidth of the gain applied at the selected center frequency. If the value of Q is closer to zero, the wider bandwidth it is, and vice versa.
EQ Type	There are 5 EQ types supported. Peaking/Highpass/Lowpass/High Shelving/Low Shelving

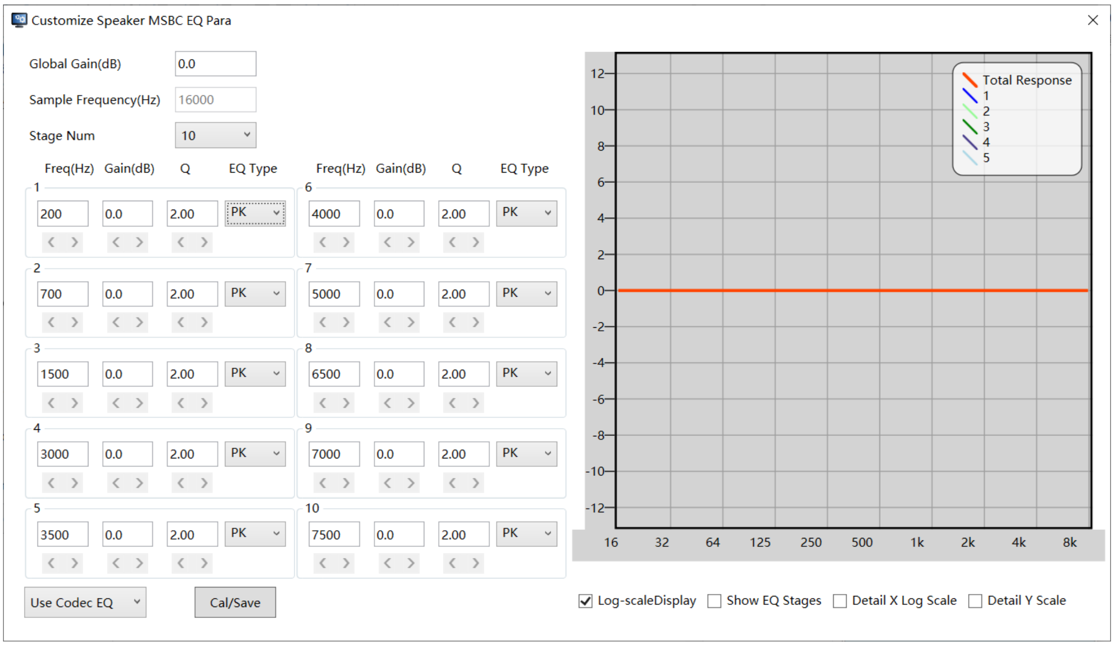


Figure 4-11: Interface of customizable 5-stage EQ.

4.6 Multi-Band Automatic Gain Control (MB-AGC)

In Figure 4-12, MB-AGC function comprises Signal Separator and three AGCs for three frequency bands. Signal Separator function is to separate the input signal into three frequency bands whose frequency boundaries are selectable ranging from 500Hz to 7.5kHz and 500kHz to 7.5kHz of first/second and second/third frequency bands, respectively.

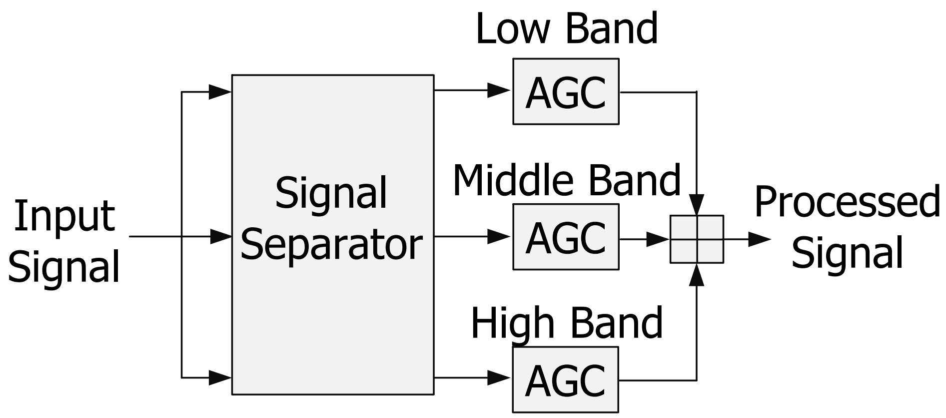


Figure 4-12: Block diagram of MB-AGC function.

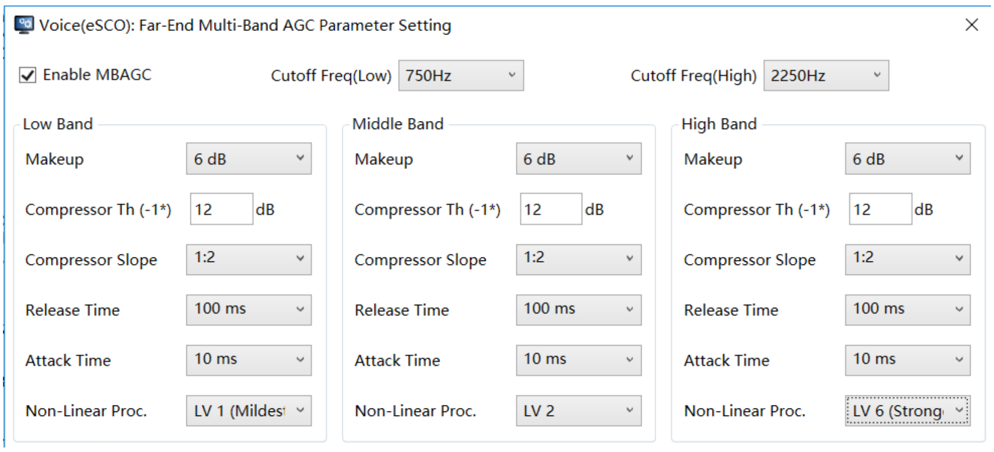


Figure 4-13: Parameter interface for MB-AGC.

AGC function is used to adjust the input signal to its desired power range by attenuating the level of strong peaks, and amplifying weaker peaks. With separated frequency bands, AGC can enhance desired speech signal loudness more efficiently especially for those bands easily being inaudible due to hearing masking effect. Figure 4-13 shows the interface for parameter of MB-AGC and functions of these parameters are summarized in Table 4-5. Among them, the definition of time constant for "Attack Time" and "Release Time" illustrated in Figure 4-14 is a period of time for a system to reach $1 - 1/e = 63\%$ of its final value. The "Attack Time" is the period of time to suppress the signal when input level hits the defined loud volume threshold. During the "Release Time", the MB-AGC function boosts the soft input signal up to the defined

"Makeup Gain". Figure 4-15 shows the illustration of MB-AGC parameters including "Makeup Gain", "Compression Ratio" and "Compression Threshold".

Each frequency band has its own independent set of AGC parameters. One can select the number of frequency bands from 1 to 3 frequency bands. However, more selected frequency bands consume more DSP computing power and hence the current consumption as well. The MB-AGC can also compensate the distorted voice quality by noise reduction and echo cancelation algorithms by balancing signal power in each frequency band.

Table 4-5: Parameters of MB-AGC function.

	Note
Attack Time	The time constant to suppress input signal to desired level as shown in Figure 4-14.
Release Time	The time constant to release from the suppression back to its desired level.
Makeup Gain	The maximal boost gain that AGC function can apply to the input signal.
Compression Threshold	Amplitude threshold to define the boundary between loud and soft signal levels.
Compression Ratio	The ratio to suppress loud audio signal exceeding whose average signal level exceeding the "Compression Threshold". Ex: if the ratio is R, compression threshold is TH dBOv, and input signal level is x dBOv. The output signal level is: Output signal level = $TH + (x - (TH)) / R$.
Cutoff Frequency (Low)	The cutoff frequency to distinguish middle and low bands.
Cutoff Frequency (High)	The cutoff frequency to distinguish middle and high bands.

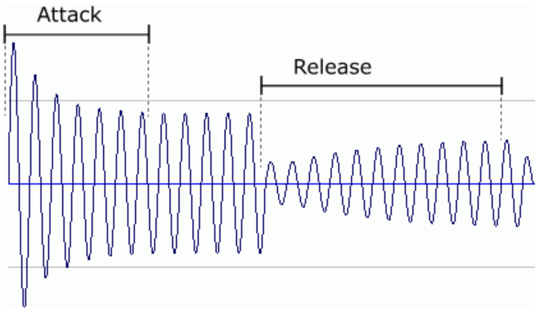


Figure 4-14: illustration of attach and release time of AGC parameters.

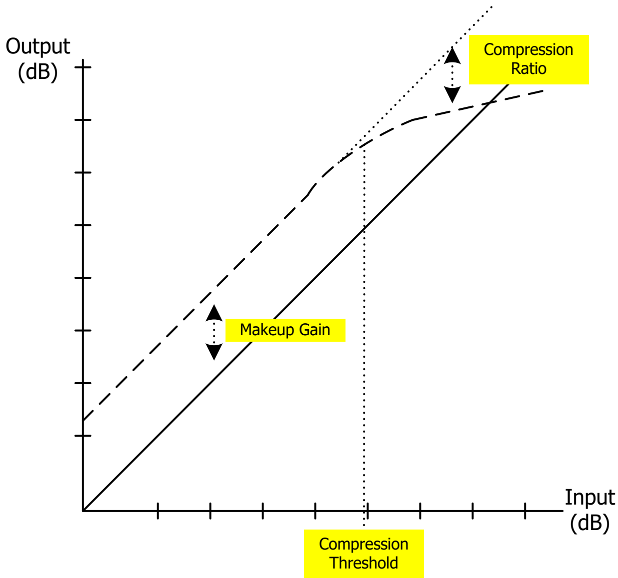


Figure 4-15: illustration of DRC parameters.

The MB-AGC allows developers to select the number of bands with defined cutoff frequencies as illustrated in **Figure 4-16**/4-17/4/18/4-19. One can select one/two/three-bands with various cutoff frequencies.

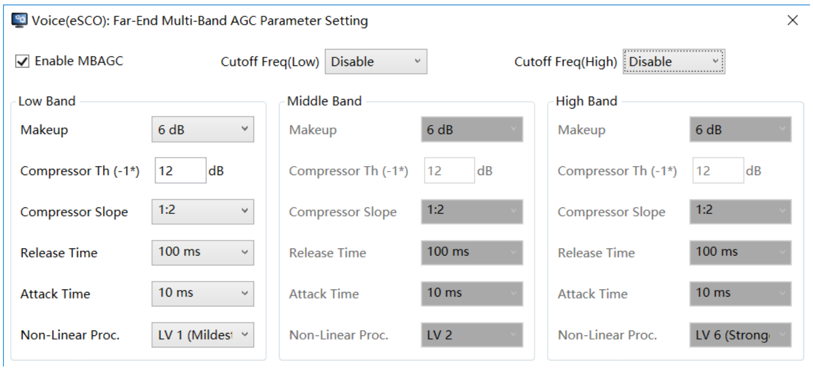


Figure 4-16: One-band setting with cutoff frequency at 1000Hz

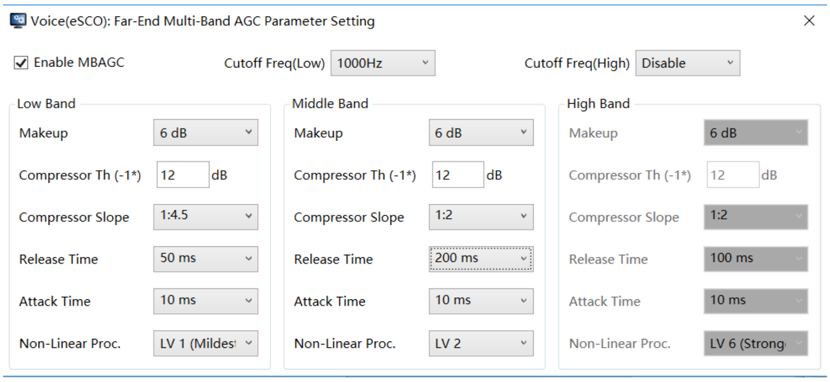


Figure 4-17:Two-band setting with cutoff frequency at 1000Hz

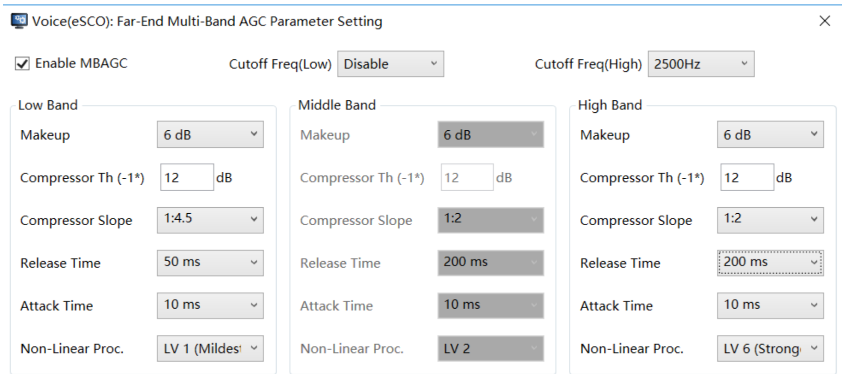


Figure 4-18:Two-band setting with cutoff frequency at 2500Hz

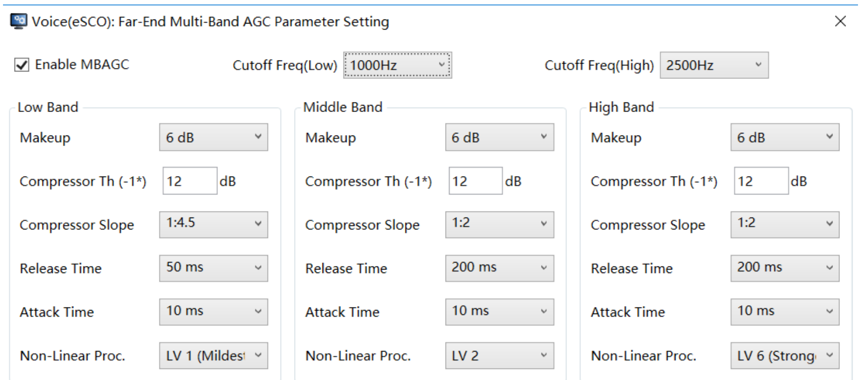


Figure 4-19:Three band setting with cutoff frequencies at 1000Hz and 2500Hz.

5. Audio/Line-In Processing

This chapter discusses configurations of audio processing algorithms designed for various Bluetooth audio applications.

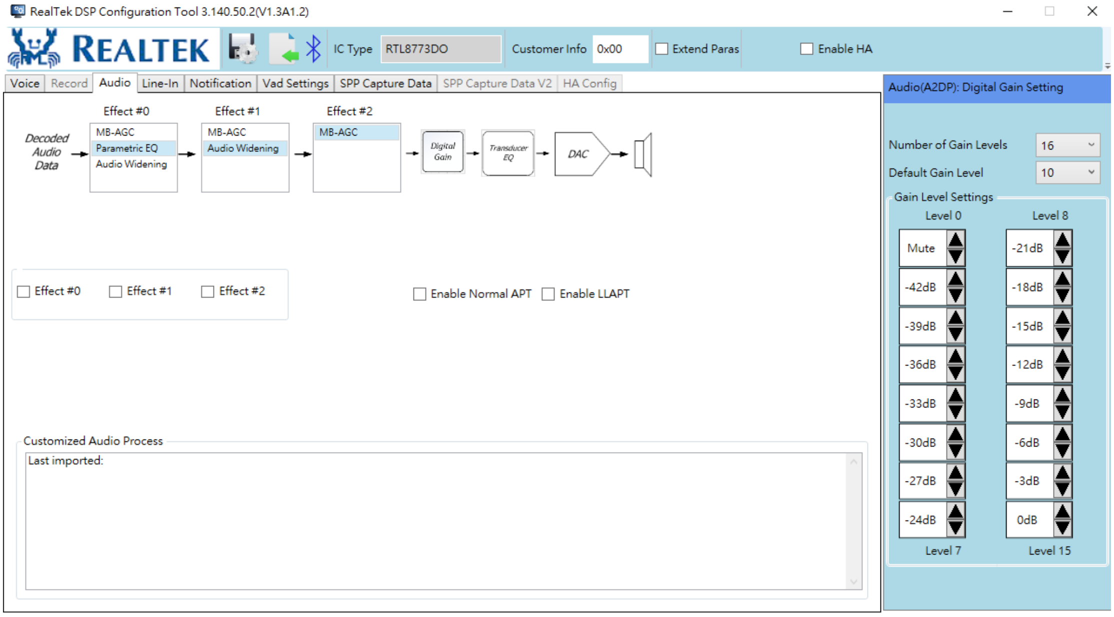


Figure 5-1: Illustration of audio subpage.

Figure 5-1 demonstrates configurations and block diagrams of Realtek's audio processing algorithms in Bluetooth A2DP/Line-In mode. In this subsection, supported audio algorithms include MB-AGC, parametric EQ, audio widening, and audio pass-through (APT) functions. Besides, interfaces to configure transducer EQ for speaker path, ADC and DAC gain are also shown. The detailed description of each block is summarized in Table 5-1.

Table 5-1: Detailed descriptions of audio processing functions.

	Note
MB-AGC	Dynamically control the output volume range of audio signal.
Parametric EQ	To boost/attenuate the specific frequencies to enhance audio signal quality.
Audio Widening	To create widened audio field based on the original audio signal, especially effective for stereo speaker placing very close apart.
Suppress gain	"suppress gain" is a global negative gain for mode change with button.
Normal APT	To mix the processed microphone signal with down-streamed audio data. The purpose of APT is to allow headset users being able to perceive the ambient sound while listening to Bluetooth audio

	streaming to reduce risk of ignoring any alert signal in the ambient environment.
LLAPT	LLAPT has lower latency compared with Normal APT, It is noteworthy that LL APT should be turn on in MCU settings in advanced, LLAPT EQ is designed in ANC design tool

In addition to the algorithmic function, the audio subpage also allows developers to have three different combinations of audio processing algorithms which will be introduced in the rest of this chapter.

In RTL8763 IC, the Line-In mode is to capture audio signal, from external media player, which are then passed to DSP for further processing before converting back to analog signal played out by speaker devices. Based on the DSP architecture design of RTL8763 IC platform, the Line-In mode are designed to share the same parameter set of audio algorithms, including DAC gains, MB-AGC, Parametric/Transducer EQs, Audio Widening and the configurations of sound effect in each selectable mode. However, the only difference is that ADC settings are not following to Audio (A2DP) setting due to much less ADC amplification gain required in Line-In mode than the APT function of Audio(A2DP) mode. Note that the APT mode is not supported in Line-In mode because APT mode needs to have microphone being enabled to capture ambient sound level.

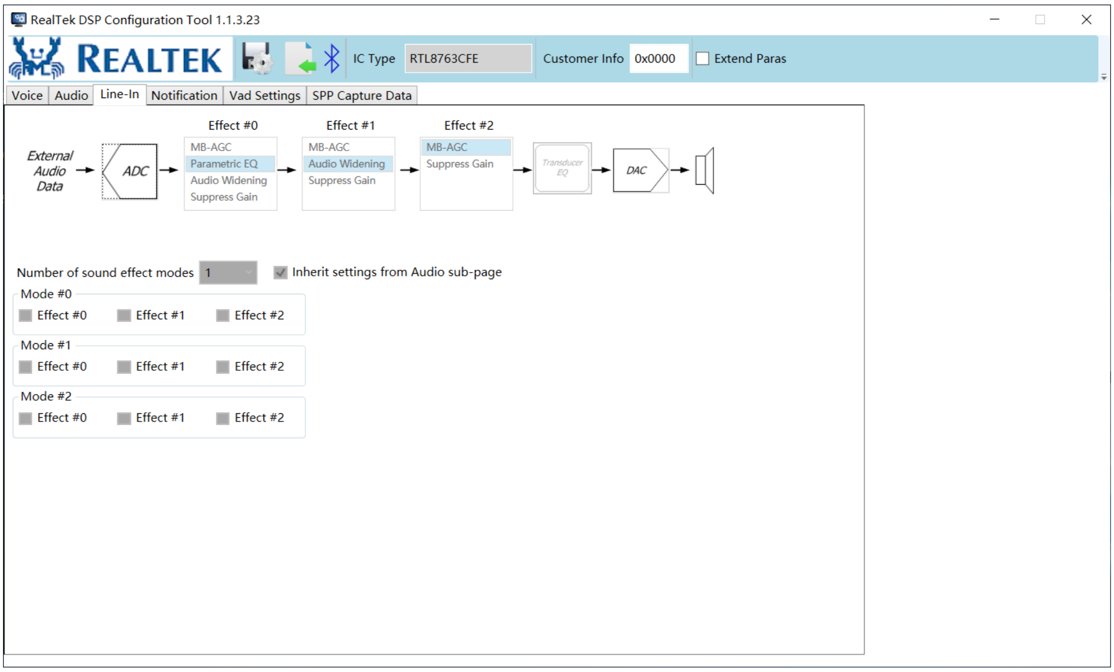


Figure 5-2: Illustration of Line-In subpage.

5.1 Mode/Effect Configurations

Audio processing algorithms can be arranged in terms of processing order in the effect configuration region as in Figure 5-3. The configuration interface of each sound effect will be popped up when it is selected in the list box of each effect. Subsequently, developers also need to configure at most three modes containing different algorithm configurations which are able to be switched from one mode to another during the audio playback with external triggers from, i.e. physical button on evaluation board or cellphone APP.

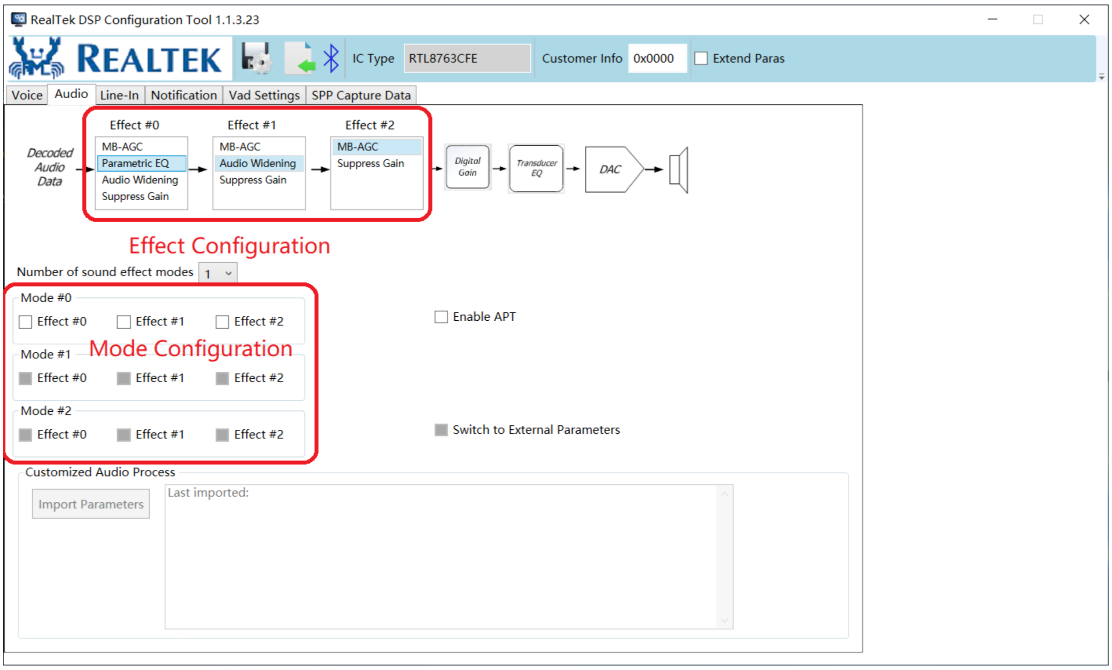


Figure 5-3: Effect and mode configurations in audio process subpage.

5.2 MB-AGC

The parameter tuning interface is identical to the voice processing part, and is enabled by default. The only difference is the selectable cutoff frequencies

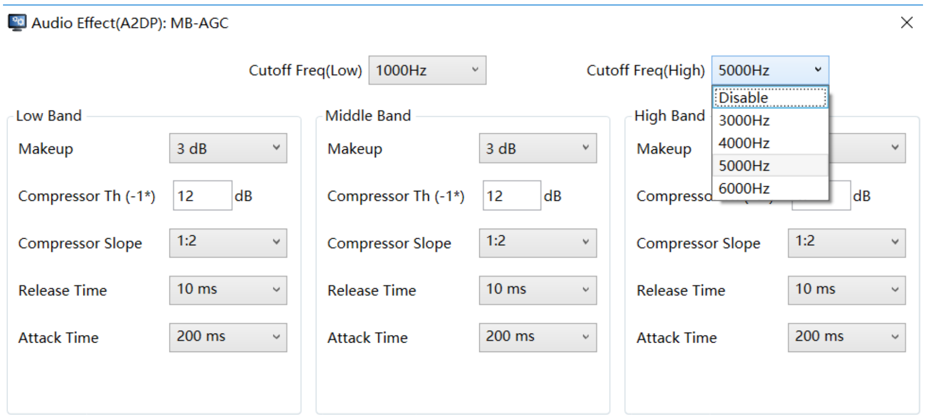


Figure 5-4: Interface of MB-AGC for audio function.

5.3 Parametric and Transducer EQs

Audio transducer EQ default sampling frequency is 44.1KHz, and coefficients of 48KHz sampling rate are automatically generated and EQ coefficients of both sampling rates will be saved in the parameter file while exporting DSP configuration. And the Parametric EQ interface is different from transducer, and parametric EQ stage number can be up to 10.

There are up to 10 user-defined EQ parameters for each IC packages. User-defined EQ parameters can be changed via DSP configuration tool.

When one EQ Index selected in Combox, the corresponding EQ parameter and EQ frequency response is displayed in the DSP configuration tool UI.

If any parameter(stage number/frequency/Q/Gain/EQ type/Global gain) changed, click the button “Apply” to make sure the EQ coefficients are correctly computed

The EQ IsEnable group is for MCU to define which EQ(s) is/are enabled or not. If enabled, end-user can select and active the corresponding EQ.

	Note
Frequency (Hz)	The desired center frequency of each EQ stage. The unit is Hz.
Gain	The boost/attenuation gain in dB scale at the selected frequency.
Q	The bandwidth of the gain applied at the selected center frequency. If the value of Q is closer to zero, the wider bandwidth it is, and vice versa.
EQ Type	There are 5 EQ types supported. Peaking/Highpass/Lowpass/High Shelving/Low Shelving

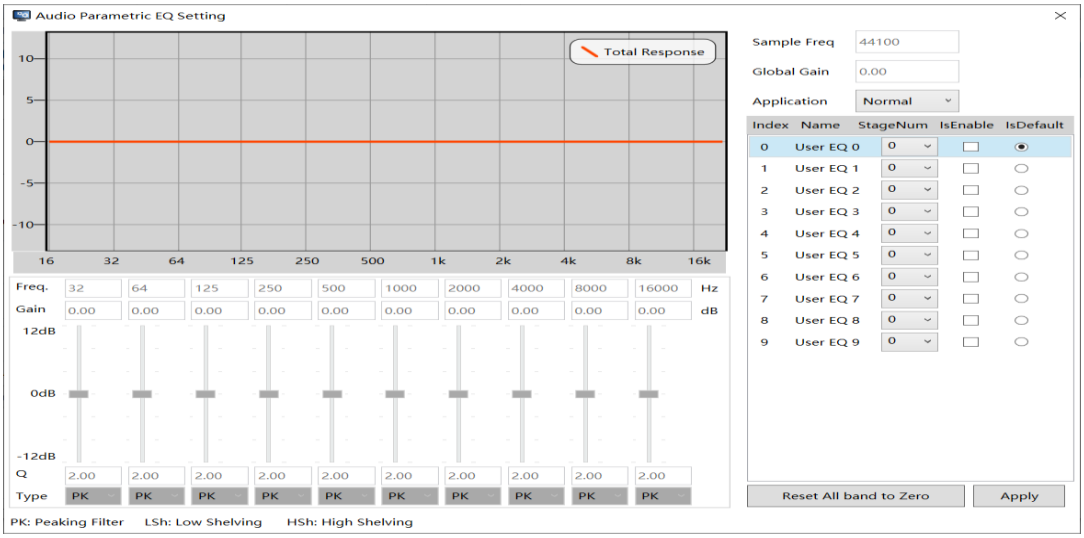


Figure 5-5: Parametric EQ interface for audio.

5.4 Audio Widening

The audio widening function is to expand the audio image especially useful for the narrow audio field generated by stereo speakers placed closely apart. Note that, this function is only useful when the channel number of audio decoder output is more than one channel. A very simple tuning interface is provided as in Figure 5-6.

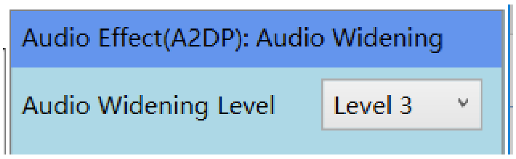
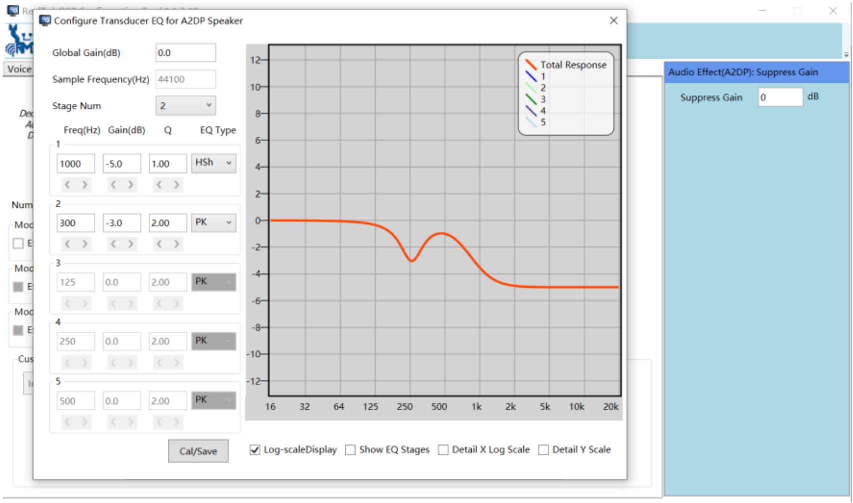


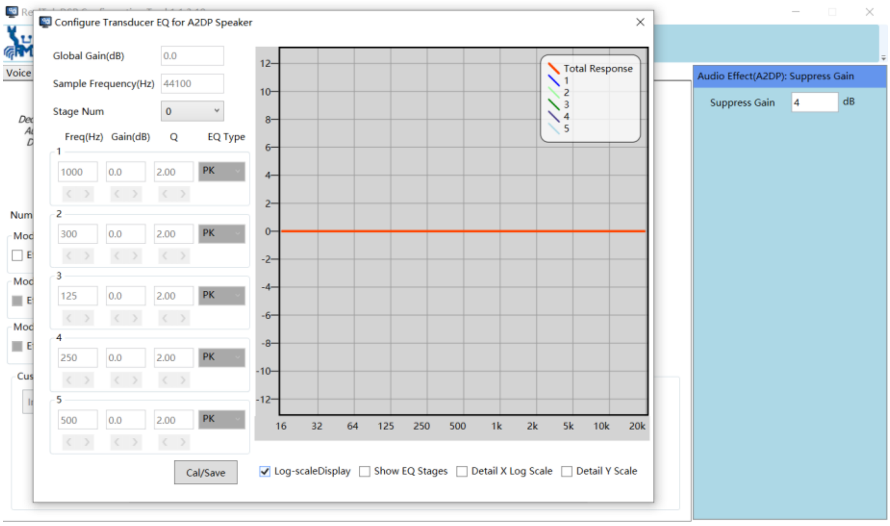
Figure 5-6: Parameter of audio widening function.

5.5 Suppress Gain

“suppress gain” is a global negative gain for mode change with button. For example, the user can set 2 audio modes, one has EQ setting with negative gain to avoid saturation(as Figure 5-7 (a) shows), the other one mode does not have any sound effect setting. In this case, the user may perceive the difference in volume. So we can set the suppress gain in second mode as Figure 5-7 (b) shows to compensate the negative gain.



(a) one example of audio mode setting



(b) the suppress gain usage for other example of audio mode setting

Figure 5-7: use of suppress gain

5.6 Audio Pass-Through (APT)

The APT mainly consists of DC remover, noise reduction, de-howling, and mix gain control functions. The noise reduction function is to suppress the ambient noise. The de-howling function is to dynamically control the microphone input gain to avoid howling effect. The performance of de-howling function is based on the echo attenuation. The echo from the microphone captured signal must be attenuated at least 15 dB. The selectable parameters shown in Figure 5-8 is listed in Table 5-2 for the de-howling function.

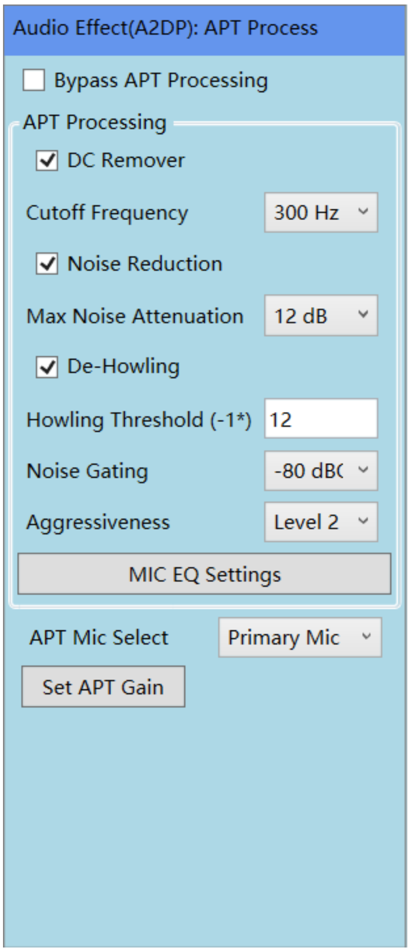


Figure 5-8: Interface of AGC parameter in APT.

Table 5-2: Parameters of de-howling function.

	Note
Howling Threshold	The detected digital level of ADC pick-up input signal. If A2DP playback is active and the digital input level exceeds the selected threshold, de-howling function starts to kick in to lower the ADC gain to avoid the howling from happening.
Noise Gating	If the detected digital input level is smaller than the selected threshold, the signal will be suppressed.
Aggressiveness	Denote the aggressiveness level to suppress howling. “Level 1” is the least aggressive level to suppress howling where “Level 4” is the most aggressiveness level.

In Figure 5-10, the selected “ADC gain” is adjusted to control Mic volume. When the howling effect happened, the selected lower level can reduce that effect.

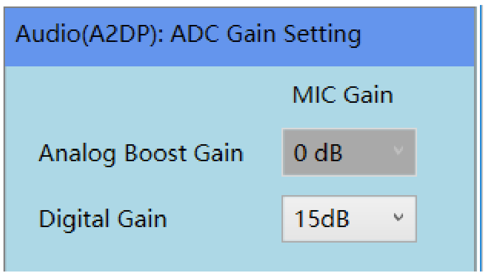


Figure 5-9: Parameters of ADC gain setting.

5.7 Low Latency Audio Pass-Through (LLAPT)

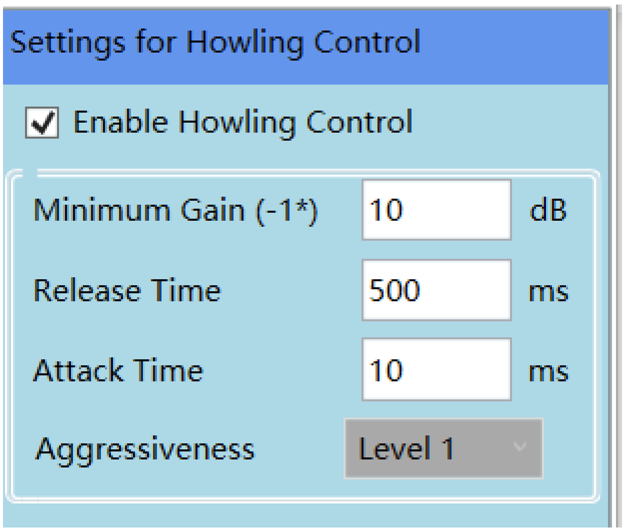


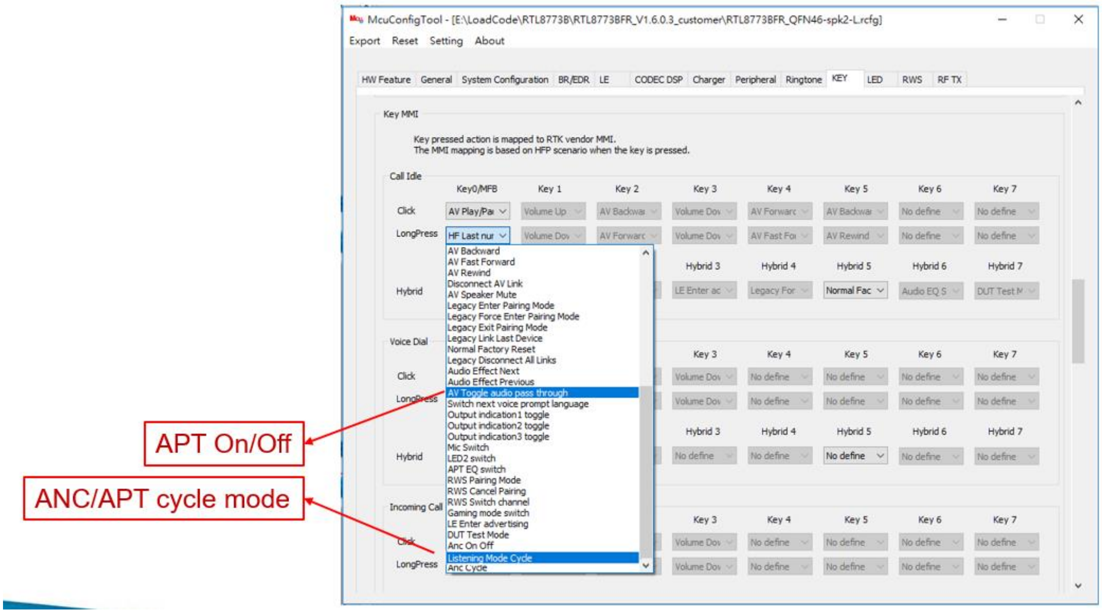
Figure 5-10: Parameters of LLAPT Howling Control setting.

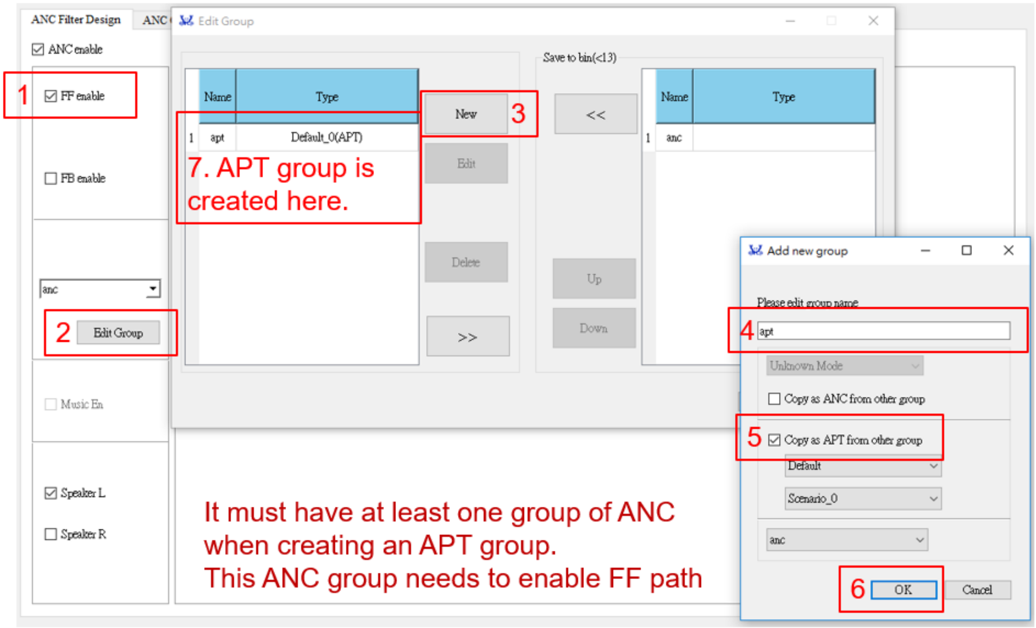
	Note
Minium Gain	The detected digital level of ADC pick-up input signal. If A2DP playback is active and the digital input level exceeds the selected threshold, de-howling function starts to kick in to lower the ADC gain to avoid the howling from happening.
Release TIme	The time constant to release from the suppression back to its desired level.
Attack Time	The time constant to suppress input signal to desired level as shown in Figure 4 14.
Aggressiveness	Denote the aggressiveness level to suppress howling. “Level 1” is the least aggressive level to suppress howling where “Level 4” is the most aggressiveness level.

LLAPT enable and filter design：

It is worth to mention LLAPT and normal APT need MCU setting in advanced before enable in DSP configuration tool

LLAPT filter need to be designed in ANC design tool





More details please see “LLAPT_tuning_guide_draft_20200430_v2”

6. VAD Settings

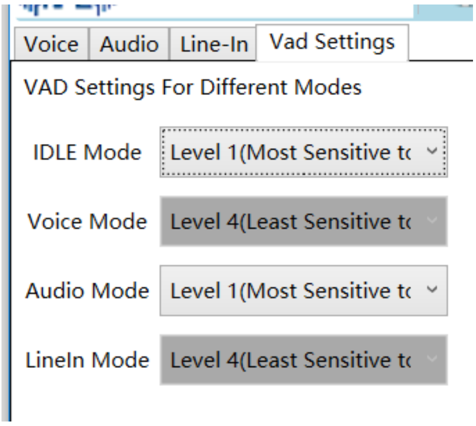


Figure 6-1

VAD is working only in idle and A2DP mode. Level 1 means the module is most sensitive to detect a voice activity, Level 4 means the module is least sensitive to detect the voice activity. VAD is NOT support in voice and line-in mode.

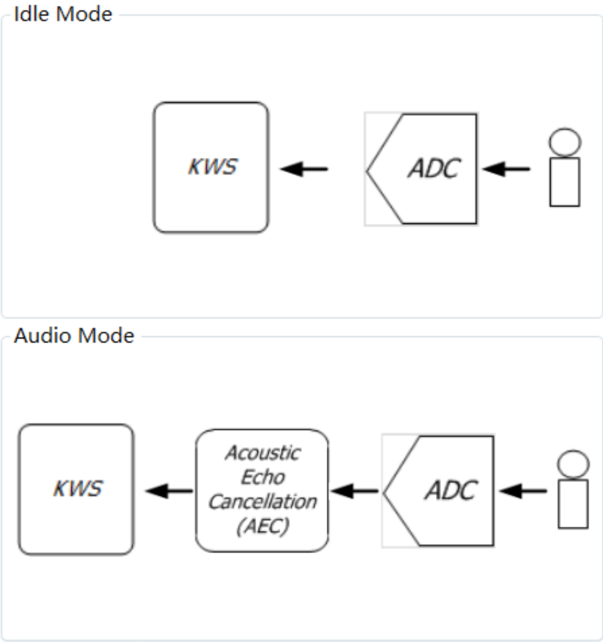


Figure 6-2

ADC gain settings is showed in this VAD settings page, and the ADC gain changed in this VAD settings page will lead to the ADC gain in the *voice* page, and the ADC gain changed in the voice page will lead to the ADC gain in this VAD page.

Recommended Fine-tuning method for Voice Processing

The methodology to fine-tune the performance of voice processing is brought up in this chapter. A good parameter set for NR/AEC functions must abide the following requirements:

- **High microphone/speaker volume.**
- **Echo-free at the microphone path.**
- **Double-talk (DT) performance after processed by AEC/AES algorithms.**
- **Voice clarity at microphone/speaker paths.**
- **Ambient noise suppression without degrading the desired voice quality.**

Figure 0-1 provides recommended fine tuning steps to optimize the performance of voice processing algorithms for speakerphone and headset applications.

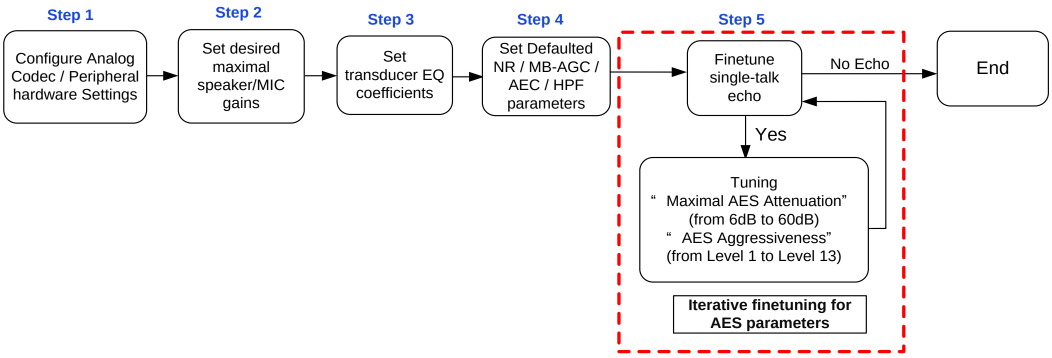


Figure 0-1 illustration of the parameter tuning flow for voice processing.

The details of these steps are explained as follows.

Step 1: The hardware configuration shall be correctly programmed before any parameter fine-tuning for voice algorithms following the programming guidelines

For example, dual-microphone noise reduction is only supported when two physical microphone devices are connected on the hardware.

Step 2: The speaker output of each gain level shall be determined following the principle that lower gain difference (<=2dB) between gain levels at high volume and bigger gain difference (>2dB) between those at softer volume. In addition, the maximal gain level shall be determined at the criteria of acceptable THD performance since poor THD could severely degrade the AEC performance.

A rule of thumb of the THD criteria at the maximal speaker volume shall be less than 5%. Unlike multiple speaker gain levels, only one gain level is needed to be configured for ADC microphone gain. The principle to set a proper ADC gain also follows the less-than-5% THD criteria to have better AEC performance. On the other hand, the slope-overload nature of CVSD codec may distort output signal if the input signal volume is too high. As a result, it is not always good to set the MIC gain to its maximal level. Note that the MIC signal recorded by Realtek BT adapter is 6dB less than the actual level.

Step 3: In light to have the optimal voice quality, the transducer EQ shall be properly configured to compensate the imperfect response of selected speaker and microphone devices. Snapshots illustrated in Figure 0-2 and Figure 0-3 (in Bluetooth standard HPF 1.6) show the frequency masks of processed microphone and speaker paths, respectively.

Frequency (Hz)	Upper Limit	Lower Limit
100	0	-2
6 200	0	-2
7 000	0	-3

Wide Band Mask, Note: All sensitivity values are expressed in dB on an arbitrary scale.

Frequency (Hz)	Upper Limit	Lower Limit
200	0	-2
3 100	0	-2
3 400	0	-3

Narrow Band Mask, Note: All sensitivity values are expressed in dB on an arbitrary scale.
Narrow band stops at 3400 Hz

Figure 0-2: Tolerance mask for the Bluetooth send sensitivity frequency response.

Frequency (Hz)	Upper Limit	Lower Limit
100	0	-2
6 200	0	-2
7 000	0	-3

Wide Band Mask Note: All sensitivity values are expressed in dB on an arbitrary scale.

Frequency (Hz)	Upper Limit	Lower Limit
200	0	-2
3 100	0	-2
3 400	0	-3

Narrow Band Mask Note: All sensitivity values are expressed in dB on an arbitrary scale.
Narrow band stops at 3400 Hz

Figure 0-3: Tolerance mask for the Bluetooth receive sensitivity frequency response.

Step 4: Apply defaulted parameters for MB-AGC, NR, HPF and AEC for both speaker and microphone processing paths and download the DSP configuration file to the IC.

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

- **Single-talk echo**: "**Maximal AES Attenuation**" and "**Aggressiveness**" introduced in section 0 are responsible for tuning the single-talk echo performance. As a rule of thumb, firstly adjust the parameter of "**Aggressiveness**" from Level 1 (Least aggressive) to Level 14 (Most aggressive). If the value of the "**Maximal AES Attenuation"** is 30dB and single-talk echo is not able to be effectively suppressed, subsequently one can continue to finetune the "**Maximal AES Suppression**" from 30dB toward the value 60dB.

  Note that: Although the selectable value of "**AES Aggressiveness**" can be up to Level 14, the voice quality of processed microphone signal is getting to be distorted more easily as “**Aggressiveness**” is getting more aggressive.

Interface for SDK Development

DSP Configuration Tool allows SDK customers to place their parameters in flash in order to, for example, fine tune SDK audio/voice processing algorithms for different products. Figure 0-1 and Figure 0-2 show the interface of importing SDK parameters. To import SDK parameters, firstly, one has to select the correct IC package type. Secondly, check “**Switch to External Parameters**”. The greyed out blocks are algorithms developed by Realtek. Next, click “**Import Parameters**” to import the customized parameters in binary format. The imported parameters are displayed in the textbox. Finally click “Save Config.” to save to the customized parameters.

The saved file, **dsp_config_image-xxx.bin**, should be loaded into flash via MP tool. The MCU reads the customized parameters from flash and send them to the DSP control system. Then, the DSP control system passes the customized parameters to SDK via entry function `RTK_RecvCmd` before initializing SDK audio/voice processing. Take Figure 0-1 for example, one can expected to receive 0x01, 0x23, 0x45, …, 0xEF in `RTK_RecvCmd`.

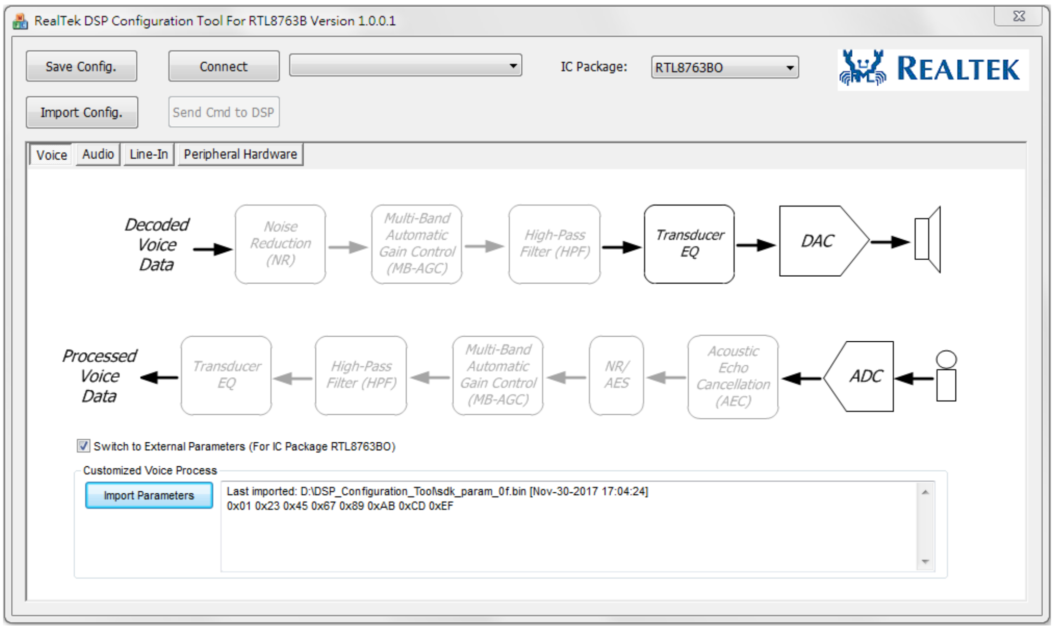


Figure 0-1: Importing Customized Voice Processing Parameters

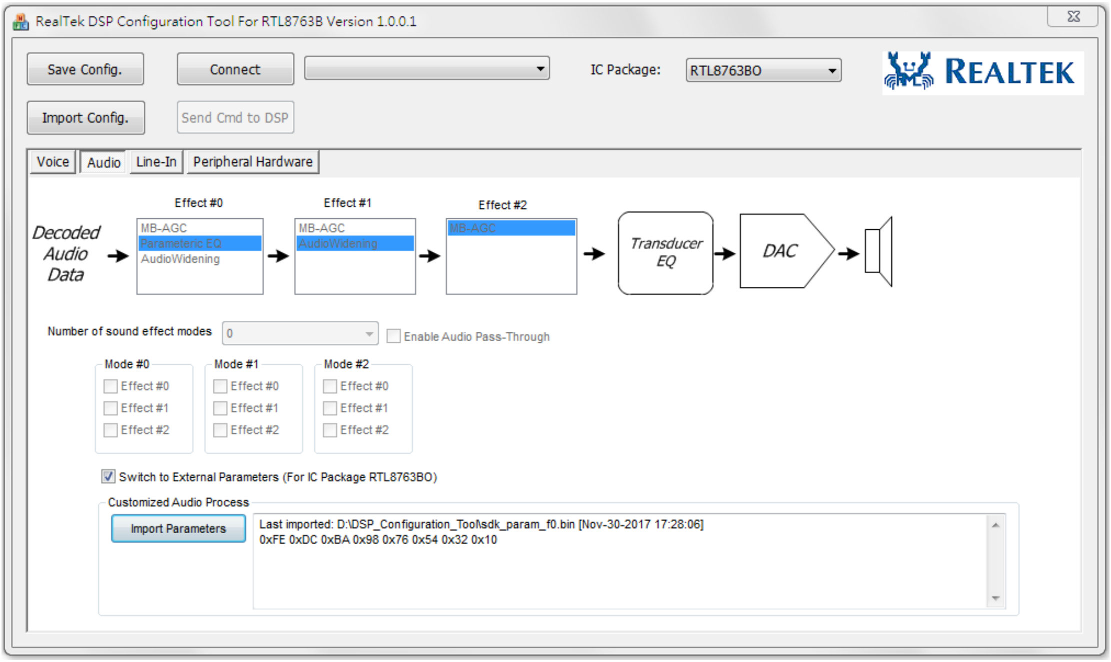
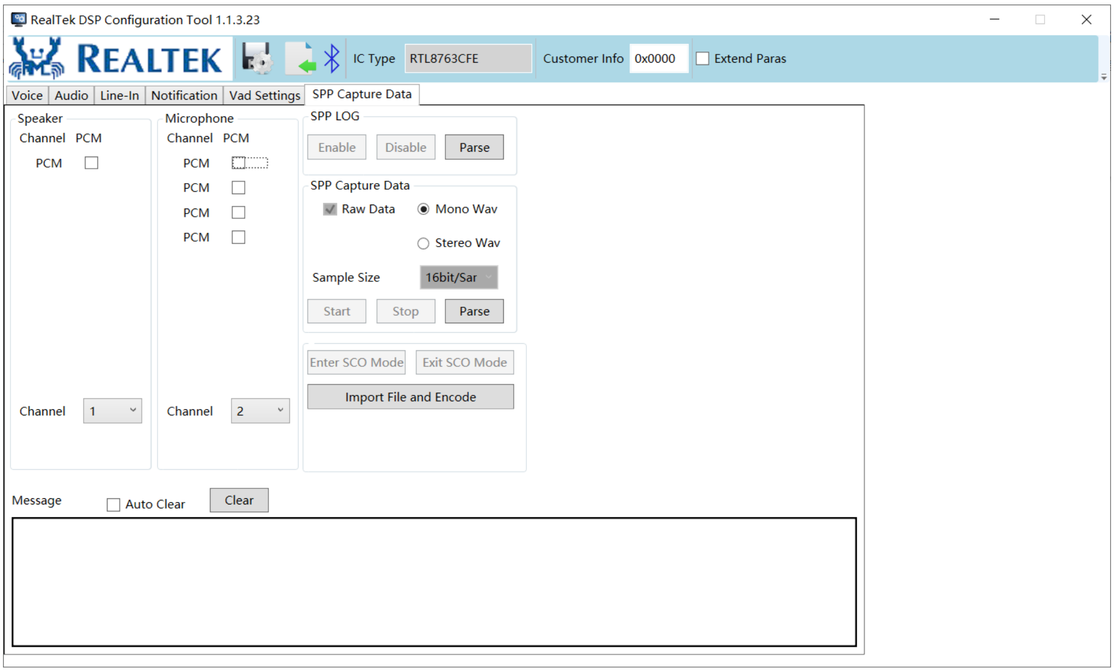
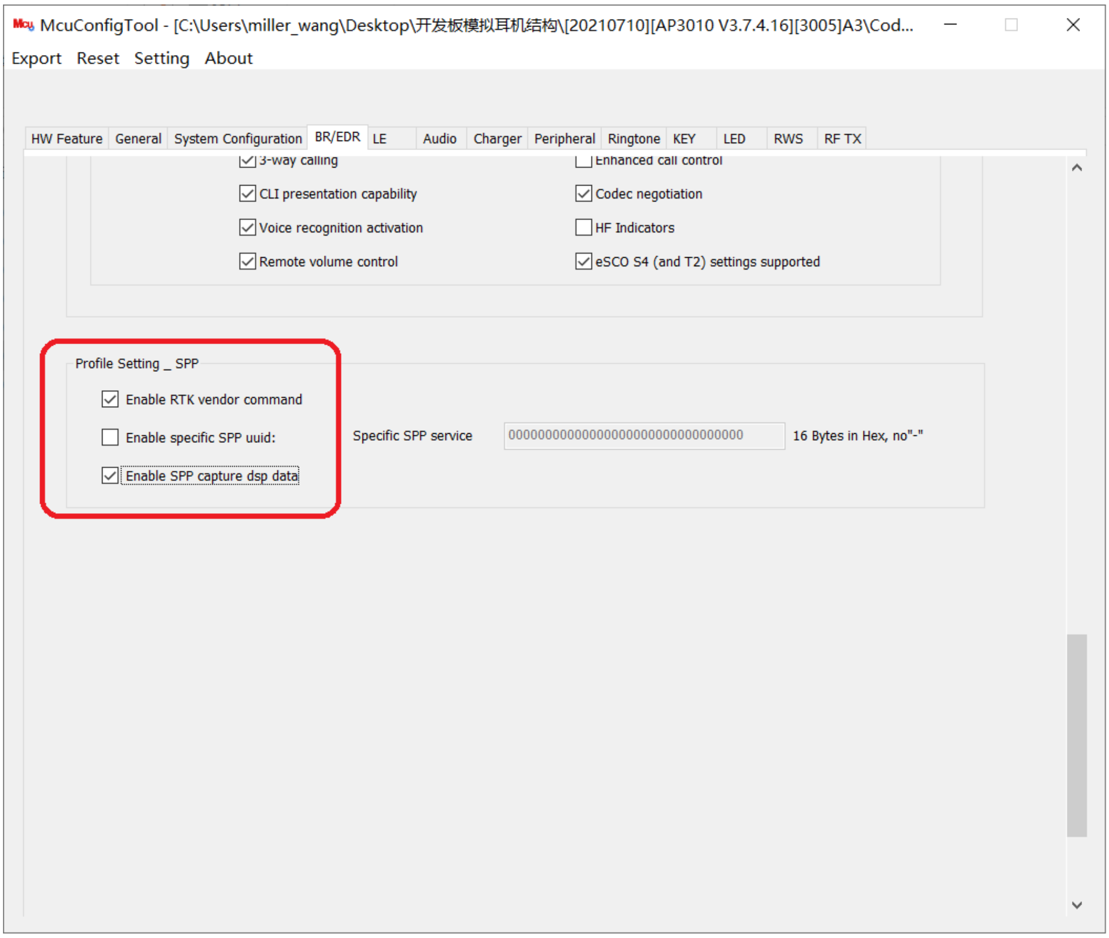


Figure 0-2: Importing Customized Audio Processing Parameters

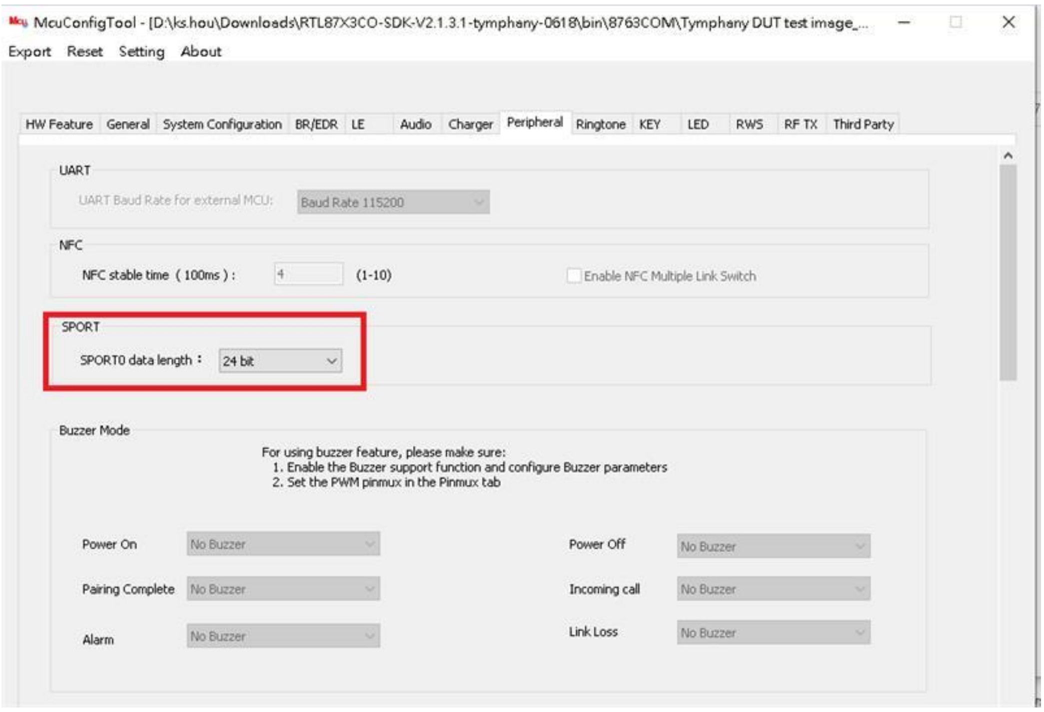
SPP Capture Data

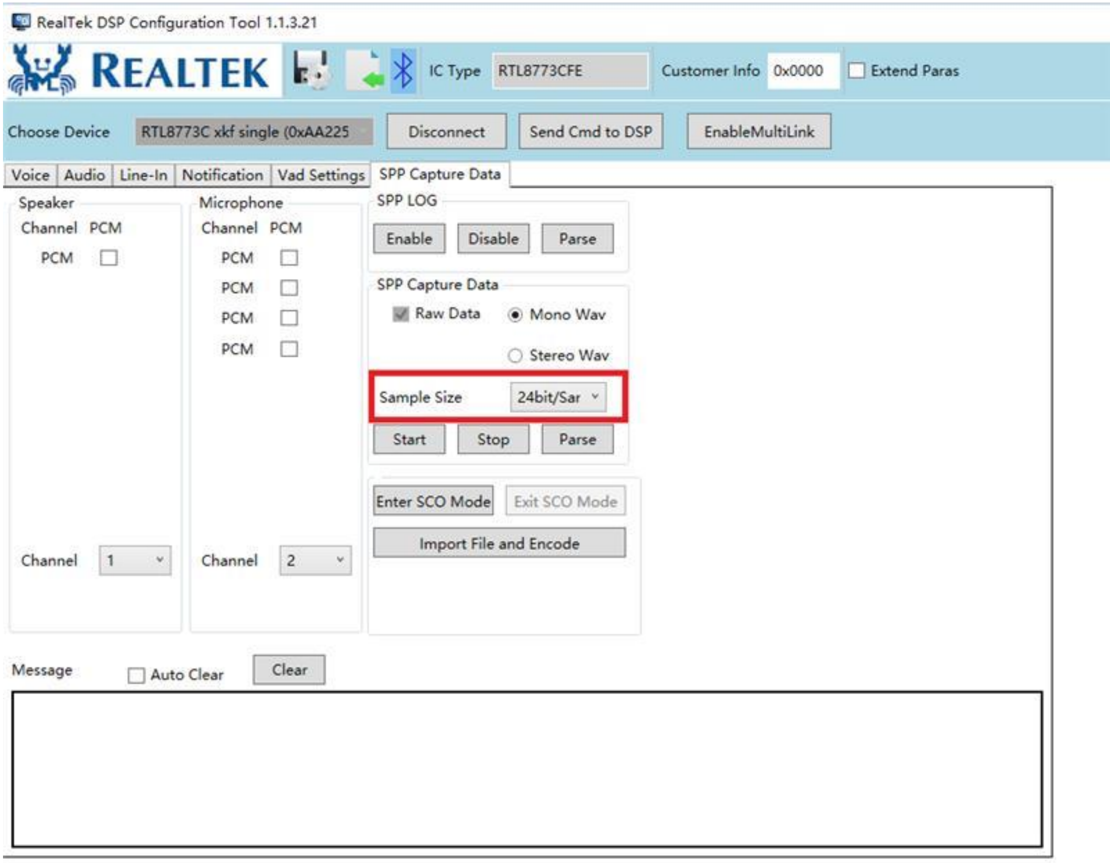


UI setting：MCU UI need support SPP Capture as below：

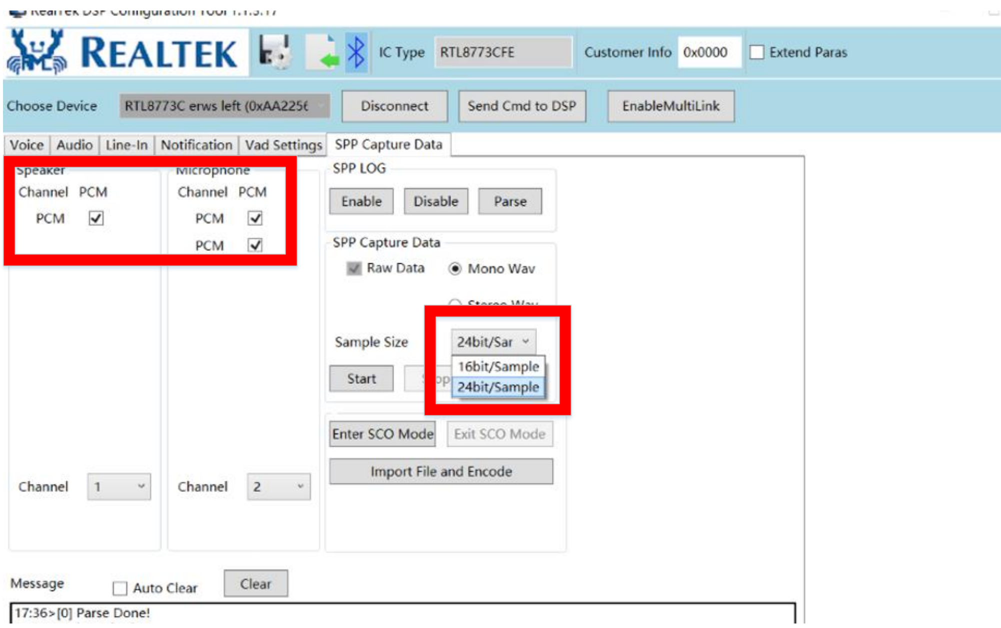


Choose 24Bit in McuConfigTool and Choose SampleSize=24Bit in SPP Capture Data can increase the resolution

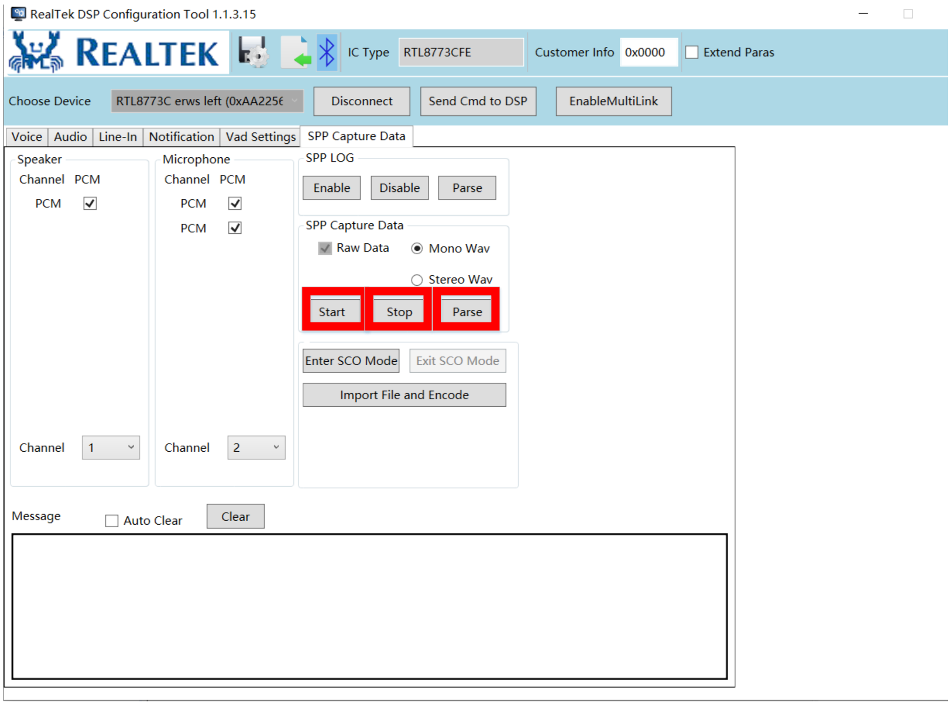




The translate data size should under 960bytes， so the capture channels（Spk+Mic） need to under 3 channels



Press Start to capture data under A2DP or HFP Mode, Captured data will saved in Audio.bin file (Audio.bin saved in path with Dspconfigtool.exe),Press Stop to finish capture ,after the DATA translation complete, press parse to translate raw data to pcm and wav



More details please see: SPP Capture SOP