Rockchip Voice Intercom 3A Algorithm Integration

ID: RK-SM-YF-391

Release Version: V1.1.0

Release Date: 2021-04-06

Security Level: □Top-Secret □Secret □Internal ■Public

DISCLAIMER

Trademark Statement

Rockchip Electronics Co., Ltd.

No.18 Building, A District, No.89, software Boulevard Fuzhou, Fujian,PRC

Website: www.rock-chips.com

Customer service Tel: +86-4007-700-590

Customer service Fax: +86-591-83951833

Customer service e-Mail: fae@rock-chips.com

Preface

Overview

Rockchip Audio Processor (referred to as RKAP) is a set of Rockchip audio processing algorithms. This document mainly introduces the processing flow and related parameter configuration of the voice call 3A algorithm.

Product Version

Name	Version
Introduce of the 3A algorithm	RKAP_3A_V1.2.0

Intended Audience

This document is mainly intended for:

Technical support engineers

Software development engineers

Support benchmarks

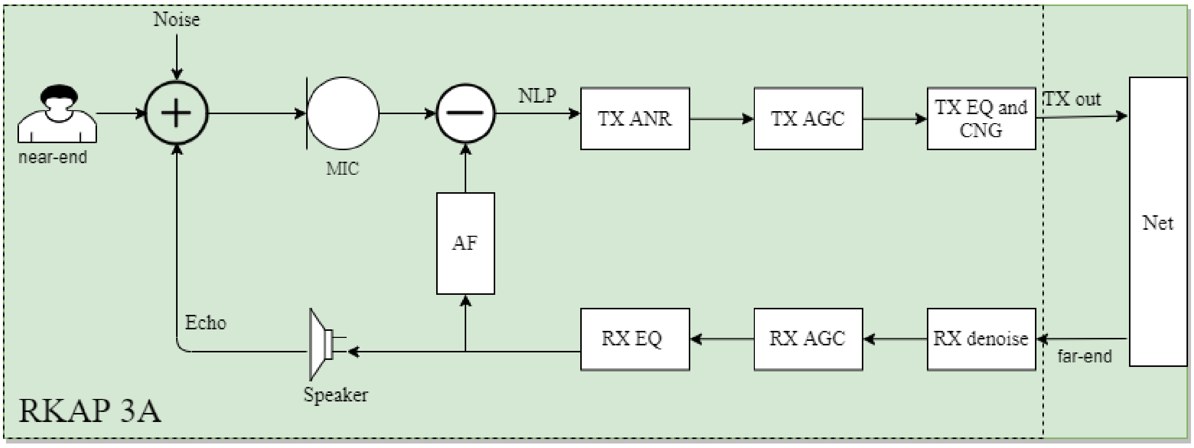
Date	Version	Author	Revision History
V1.0.0	Cherry.Chen	2020-12-09	Initial version
V1.1.0	Cherry.Chen	2021-04-06	add AEC ERLE mode and Hardware test

Contents

1. Introduction

The sound of the loudspeaker is fed back to the microphone for a period of time, and the original sound that is heard by human ear is called echo. The block diagram of the echo generation and the basic working principle of the 3A algorithm are as follows.

1.1 Flow Description



In the above figure, the TX represents the transmitting end, that is, sending data from the near-end to the far-end, and the RX represents receiving; that is, receiving signals from the far-end to the near-end. In the case of speaker playback, the far-end sound played by the speaker spreads over the air to the near-end microphone. If Acoustic Echo Cancellation (AEC) is not applied to it, the far-end can hear the near-end sound at the same time. Hearing what user just said, the worse situation is that when the hands-free device is also used at the remote end, the sounds at both ends stimulate each other, which is easy to produce howling. Therefore, it is necessary to attenuate the echo through the AEC algorithm. The AEC algorithm mainly includes two parts: AF (Adaptive Filter) and NLP (Nonlinear Process), where AF stands for adaptive filter, which calculates the echo in the near-end signal by simulating the echo path, and the NLP means non-linear processing. After AF processing, there is residual echo in the near-end signal, which needs to be suppressed by the NLP algorithm. It can be seen from the figure that in addition to the near-end speech and echo, the signal captured by the microphone also contains environmental noise. Therefore, audio noise reduction (ANR) is also required to remove the environmental noise. In the signals captured by the near-end microphone, after the echo and noise are cancelled, the remaining useful sound gain is low, so the automatic gain control algorithm (Audio Gain Control, AGC) is introduced to gain the useful signal; finally, the EQ (Equalizer) and CNG (Comfort Noise Generation) increase the comfort of the sound and send it to the far-end.

1.2 Testing

From the description in section 1.1, we can know that there are three situations in a two-terminal call:

- **Near-end speaks, far-end does not speak**

  In this case, the system is in the near-end single-speaking state, and the far-end does not speak (the near-end speaker has no sound), so there is no echo. At this time, it is equivalent to the microphone collecting only the near-end sound and noise. Equivalent to the adaptive filter not working.

- **Near-end does not speak, far-end speaks**

In this case, the system is in the far-end single-talking state, and the far-end sound is played through the near-end speaker and then collected by the microphone (the microphone input contains echo and noise). In this state, the adaptive filter converges in a certain period of time After reaching a steady state.

- **Near-end and far-end talk at the same time**

  In this case, the system is in a dual-talk state. In this state, the adaptive filter is prone to divergence, resulting in reduced echo cancellation.

  Therefore, when testing the echo cancellation effect, the above three conditions need to be tested simultaneously.

1.3 Indicators

1.3.1 Objective indicators

It is mentioned in ITU G.168 that Echo Return Loss (ERL): The attenuation of the signal from the receiving port to the transmitting port of the echo canceller. Refers to the loss in the echo channel (near end), that is, the attenuation of the sound from the far end through the echo path and then to the microphone input.

Echo Return Loss Enhancement (ERLE): The attenuation when the echo signal is transmitted through the transmission channel of the echo canceller.

Using ERLE to measure the effect of echo cancellation can be regarded as the average value of echo attenuation achieved over a period of time. The larger the ERLE value, the better the effect, and the value can be recorded in time to count the convergence time:

$$\mathrm{ERLE}(m) = 10 \log 10 \frac{\sum_{j=1}^{q} d^2[(m-1)q+j]}{\sum_{j=1}^{q} e^2[(m-1)q+j]}$$

The \$d(n)\$ is the near-end signal, and \$e(n)\$ is the difference signal, which is the output signal.

1.3.2 Subjective indicators

ITU-TP.800 and P.830 define the subjective test method of MOS score (Mean Opinion Score): different people compare the original corpus and the corpus after audio algorithm processing, score MOS separately, and finally get average value, this is a purely subjective qualitative measurement. The ITU selects the same scores for different ages, genders, and language groups within a wide range of hearing to make the evaluation criteria for voice quality.

MOS Rating	Subjective Opinions	Auditory Perception
4-5	Excellent	Speaking is clear, almost no delay. Smooth communication
3-4	Good	Speaking is clear, and the delay is small. Communication is not smooth and there is a little noise.
2-3	Fair	Speaking is not too clear, and the delay feels obvious. Communication was repeated many times.
1-2	Poor	Speaking is not clear, and the delay is large. Communication was repeated many times.
1 or less	Bad	Hard to hearing clearly, and the delay is large. Difficult to communicate.

2. API Integration

2.1 RKAP_3A_Init()

Functions	**RKAP_Handle RKAP_3A_Init(RKAP_AEC_State *st, RKAP_AEC_TRANS_ENUM transType);**
Input Parameters	st: The structure of st input some basic parameters transType: TX or RX to indicate the current processing flow
Return Values	Handle of TX or RX processing
Function Description	Initialize the 3A algorithm

Note: The value range of each parameter of the structure RKAP_AEC_State is as follows:

```
typedef struct RKAP_AEC_State_S
{
    /* Basic info */
    int swSampleRate;     /* Sample rate, only supports 8000Hz or 16000Hz */
    int swFrameLen;       /* The length of frame, only supports 16ms or 20ms */
    const char *pathPara; /* The path of loading Para.bin */
} RKAP_AEC_State;
```

2.2 RKAP_3A_Destroy()

Functions	**void RKAP_3A_Destroy(RKAP_Handle handle);**
Input Parameters	RKAP Handle
Return Values	None
Function Description	Deinitialize

2.3 RKAP_3A_Process()

Functions	int RKAP_3A_Process(RKAP_Handle handle, short *pfSigIn, short *pfSigRef, short *pfSigOut);
Input Parameters	handle: The handle of initialized TX or RX pfSigIn: the signal of near-end pfSigRef: the signal of far-end for reference pfSigOut: output signal
Return Values	0 means correct return, the others mean error return.
Function Description	The processing of 3A algorithm Note: Generally TX and RX are co-exist, so they need to be initialized and processed separately.

2.4 RKAP_3A_DumpVersion()

Functions	void RKAP_3A_DumpVersion(void);
Input Parameters	None
Return Values	None
Function Description	Print the current algorithm library version

2.5 example

external/rkmedia/examples/rkmedia_audio_test.c

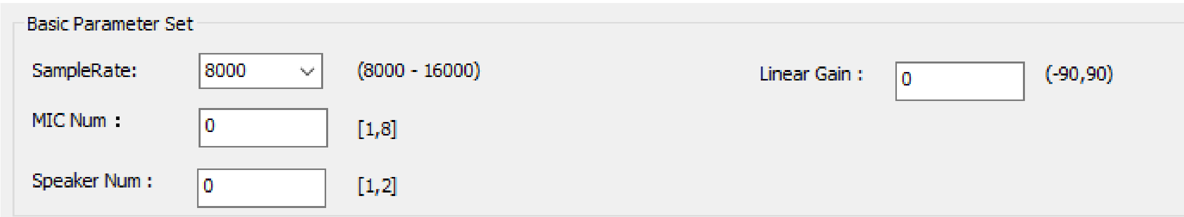
3. RKAP_Para.bin Files

The following with the TX mark indicates the TX process adjustment parameters, and RX is the same. There are many tuning modules involved, so use the Windows tool RKAP_3A_Para_Tool to save the relevant parameters.

3.1 Basic Parameters

Parameter Name	Index	Ranges	Description
SampleRate	0	8kHz or 16kHz	Sample Rate
Mic_Num	5	1~8	The number of Mic-phones
Speaker_Num	6	1~2	The number of reference channels
Linear Gain	7	-90~90 (dB)	Linear gain for TX input signal, the unit is dB

The UI of tool looks as follows:



3.2 AEC Parameters

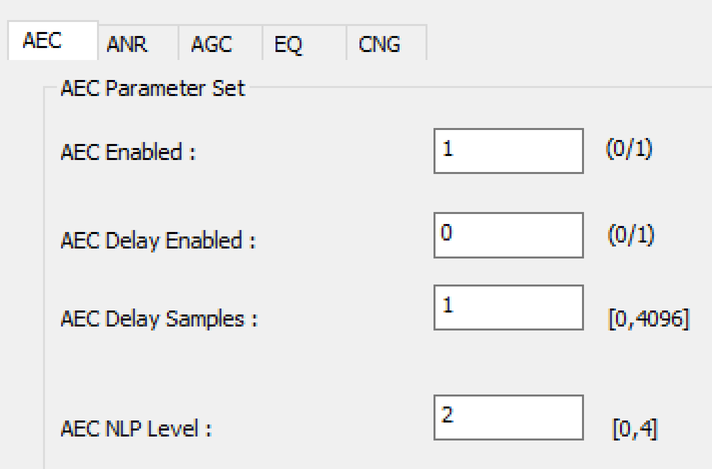
3.2.1 AEC Basic Parameter

Parameter Name	Index	Ranges	Description
AEC Enabled	10	0 or 1	0-off, 1-on
AEC Delay Enabled	11	0 or 1	Usually the AEC Delay is used for software loop back
Default Delay Samples	12	0~4096	When the automatic delay estimation is not enabled and the mic and ref signals have a fixed delay, this value represents the number of samples of the delay.
AEC NLP Level	13	0~4	The degree of inhibition of AEC NLP, 0-4 inhibition is gradually increasing.

Note: AEC NLP Level is the suppression of residual echo by nonlinear processing:

- Level 0 means that the NLP module is turned off, that is, only the echo is adaptively filtered. This situation is suitable for scenarios where the near-end signal and the far-end signal have a linear relationship.
- Level 1 means light intensity to suppress residual echo, suitable for scenes with high speaker quality, good cavity sealing, less nonlinearity introduced by the whole machine vibration, and better structure.
- Level 2 means that the residual echo is suppressed with moderate strength. It is suitable for situations where the speaker, cavity and structure basically meet the requirements, but the linearity is not enough.
- Level 3 means the highest level of echo suppression. This situation is suitable for scenes where the speaker is cheap, the cavity is bad, and the audio index is poor. Using Level 3 can cancel echo, but it is easy to produce over-cutting, sound-cutting, etc., resulting in voice understanding influences.

The UI of tool looks as follows:



3.2.2 AEC ERLE Parameter

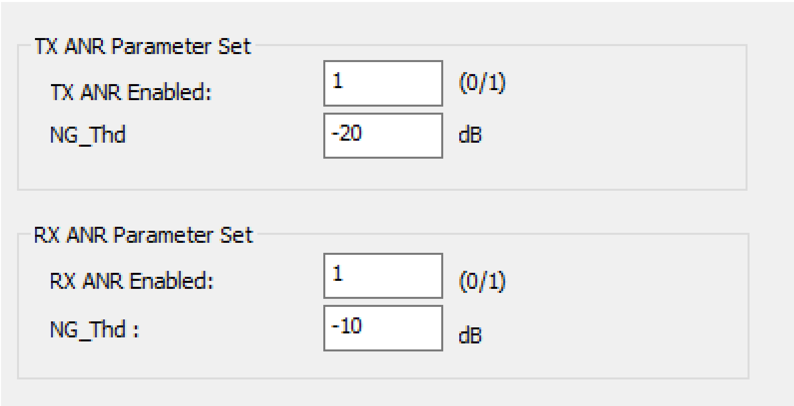
Parameter Name	Index	Range	Default Value	Description
ERLE Enabled	450	0 or 1	0	0-off,1-on
ERLE Smooth	451	0~1	0.92	ERLE smoothing coefficent
ERLE Thd	452	0~1	0.05	ERLE threshold, repressing when ERLE low than THD
ERLE Con Thd	453	1~20	1	ERLE continuation threshold which lower than ERLE THD

3.3 ANR Parameters

Parameter Name	Index	Ranges	Description
TX ANR Enabled	100	0 or 1	0-off, 1-on
TX ANR Gmin	101	-50~-5 (dB)	Gmin represents the noise threshold, that is, the noise energy value when there is no speech.
RX ANR Enabled	110	0 or 1	0-off, 1-on
RX ANR Gmin	111	-50~-5 (dB)	The same as the Gmin of TX ANR

Note: The smaller the value of Gmin, the cleaner the noise is eliminated, but at the same time, the speech with lower energy may be over-eliminated.

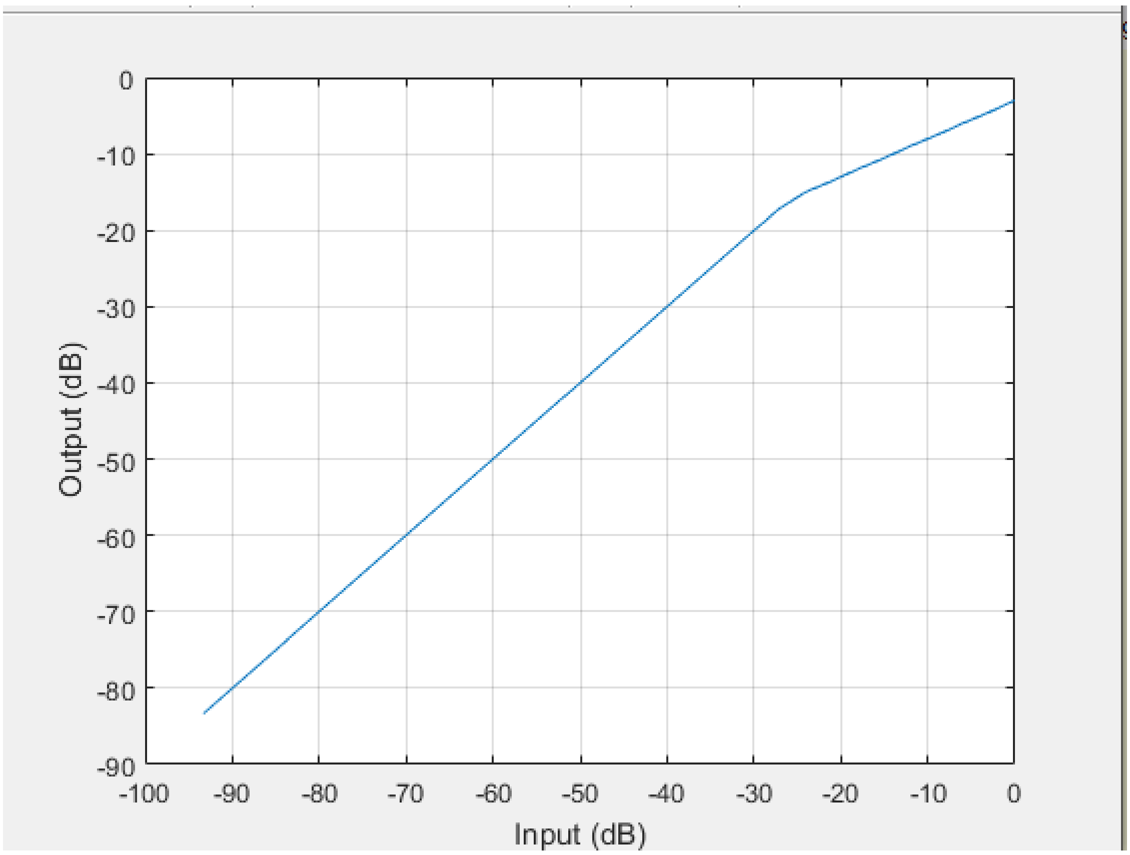
The UI of tool looks as follows:



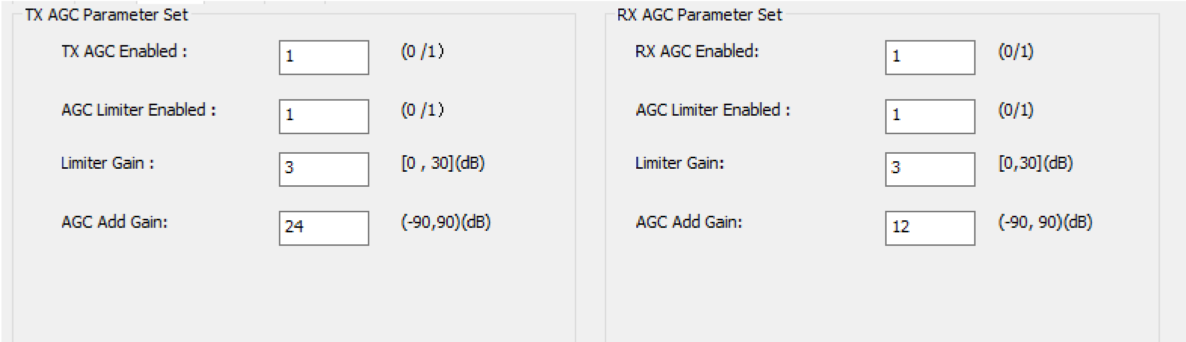
3.4 AGC Parameters

Parameter Name	Index	Ranges	Description
TX AGC Enabled	130	0 or 1	0-off, 1-on
TX AGC Limiter Enabled	131	0 or 1	0-off, 1-on
TX AGC Target Level	132	0~90(dB)	When the limiter is enabled, the TX AGC target level means (-1) times the gain to be limited
TX AGC Add Gain	133	-90~90(dB)	The range of the TX AGC Gain
RX AGC Enabled	150	0 or 1	0-off, 1-on
RX AGC Limiter Enabled	151	0 or 1	0-off, 1-on
RX AGC Target Level	152	0~30(dB)	When the limiter is enabled, the RX AGC target level means (-1) times the gain to be limited
RX AGC Add Gain	153	-90~90(dB)	The range of the RX AGC Gain

The following figure shows the input and output comparison diagram when the target Level is 3dB and the RX AGC Add Gain is 10dB:



The UI of tool looks as follows:



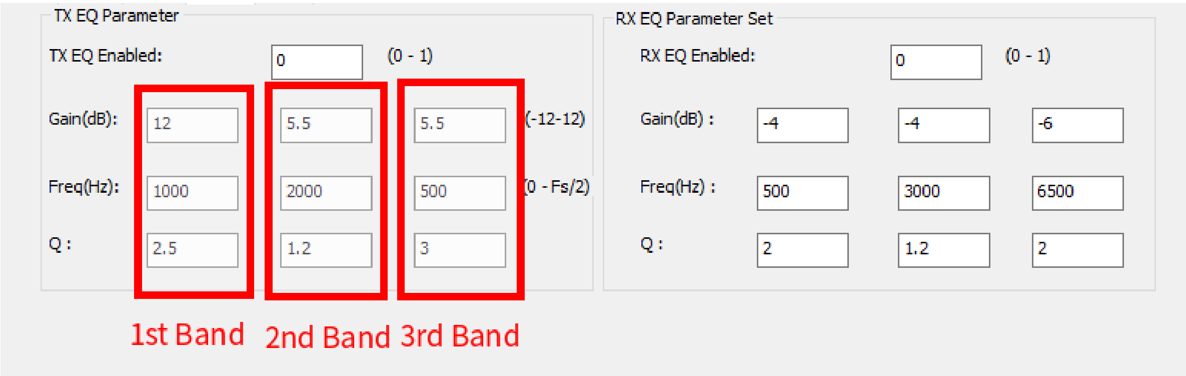
3.5 EQ Parameters

3.5.1 PEAK Filter

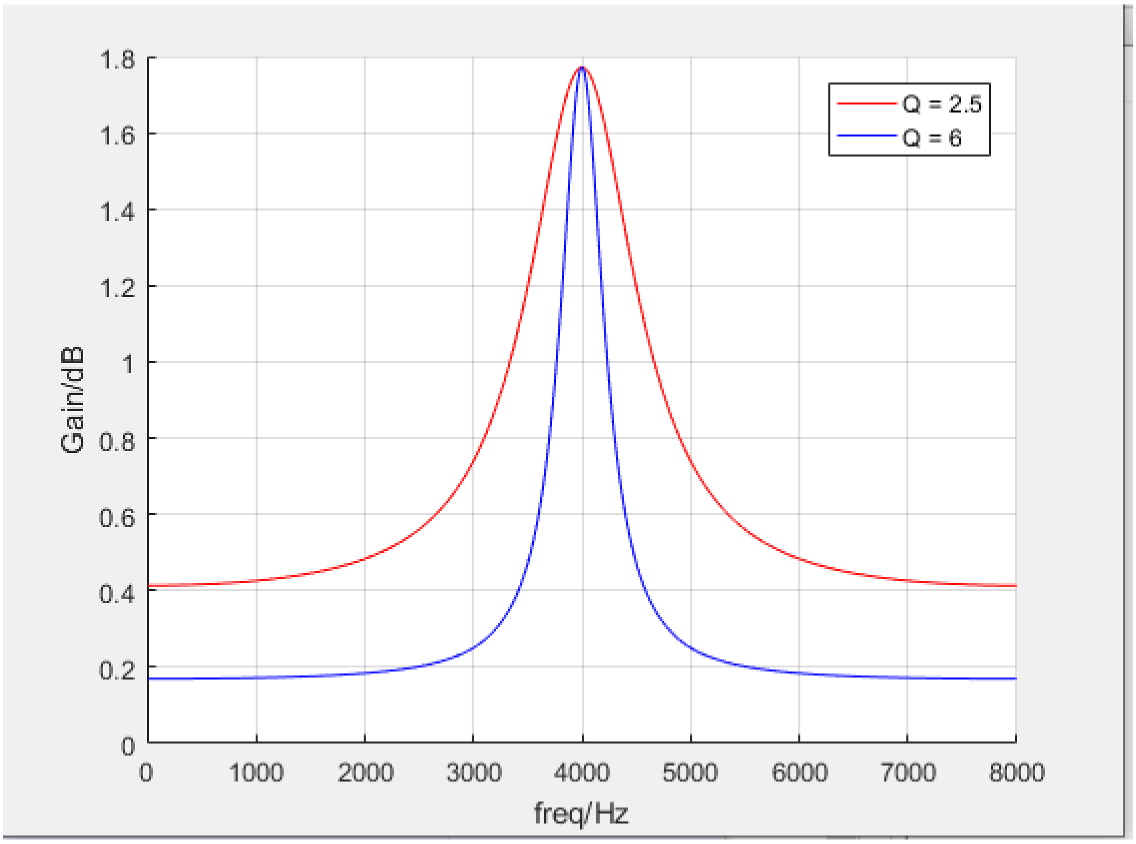
The EQ (Equalizer) in this algorithm is a simple 3-Bands EQ, which aims at the voice comfort after 3A algorithm processing.
The specific parameters are as follows:

Parameter Name	Index	Ranges	Description
TX EQ Enabled	160	0 or 1	0-off, 1-on
TX EQ Freq0	170	(0, Fs/2)	The center frequency of the first band TX EQ
TX EQ Gain0	171	[-12,12] (dB)	The gain of the first band TX EQ, unit: dB
TX EQ Q0	172	(0,10]	The quality factor of the first band TX EQ
TX EQ Freq1	180	(0, Fs/2)	The center frequency of the second band TX EQ
TX EQ Gain1	181	[-12,12] (dB)	The gain of the second band TX EQ, unit: dB
TX EQ Q1	182	(0,10]	The quality factor of the second band TX EQ
TX EQ Freq2	190	(0, Fs/2)	The center frequency of the third band TX EQ
TX EQ Gain2	191	[-12,12] (dB)	The gain of the first band TX EQ, unit: dB
TX EQ Q2	192	(0,10]	The quality factor of the third band TX EQ
RX EQ Enabled	300	0或1	0-off, 1-on
RX EQ Freq0	310	(0, Fs/2)	The center frequency of the first band RX EQ
RX EQ Gain0	311	[-12,12] (dB)	The gain of the first band RX EQ, unit: dB
RX EQ Q0	312	(0,10]	The quality factor of the first band RX EQ
RX EQ Freq1	320	(0, Fs/2)	The center frequency of the second band RX EQ
RX EQ Gain1	321	[-12,12] (dB)	The gain of the second band RX EQ, unit: dB
RX EQ Q1	322	(0,10]	The quality factor of the second band RX EQ
RX EQ Freq2	330	(0, Fs/2)	The center frequency of the third band RX EQ
RX EQ Gain2	331	[-12,12] (dB)	The gain of the third band RX EQ, unit: dB
RX EQ Q2	332	(0,10]	The quality factor of the third band RX EQ

The UI of tool looks as follows:



The figure below shows an example of peak filter which Freq = 4000Hz,Gain = 12, Q = 2.5/6:

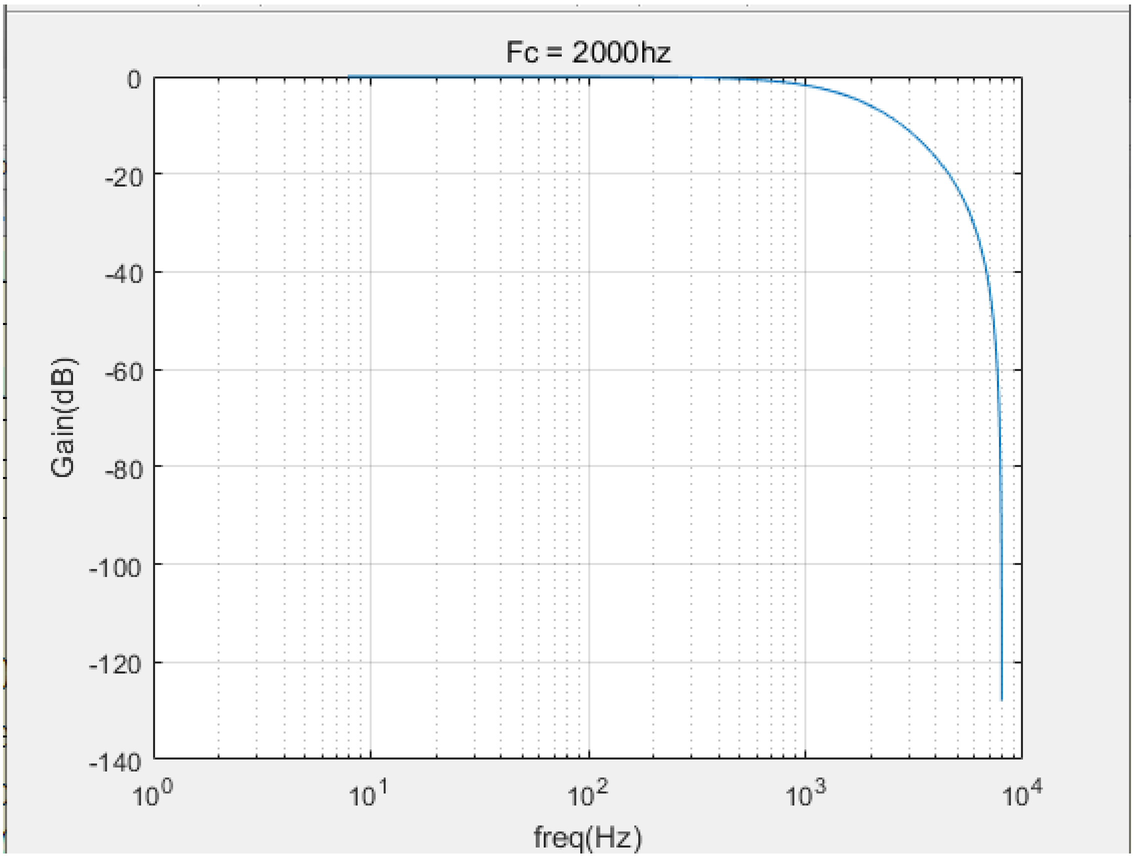


3.5.2 High Pass Filter & Low Pass Filter

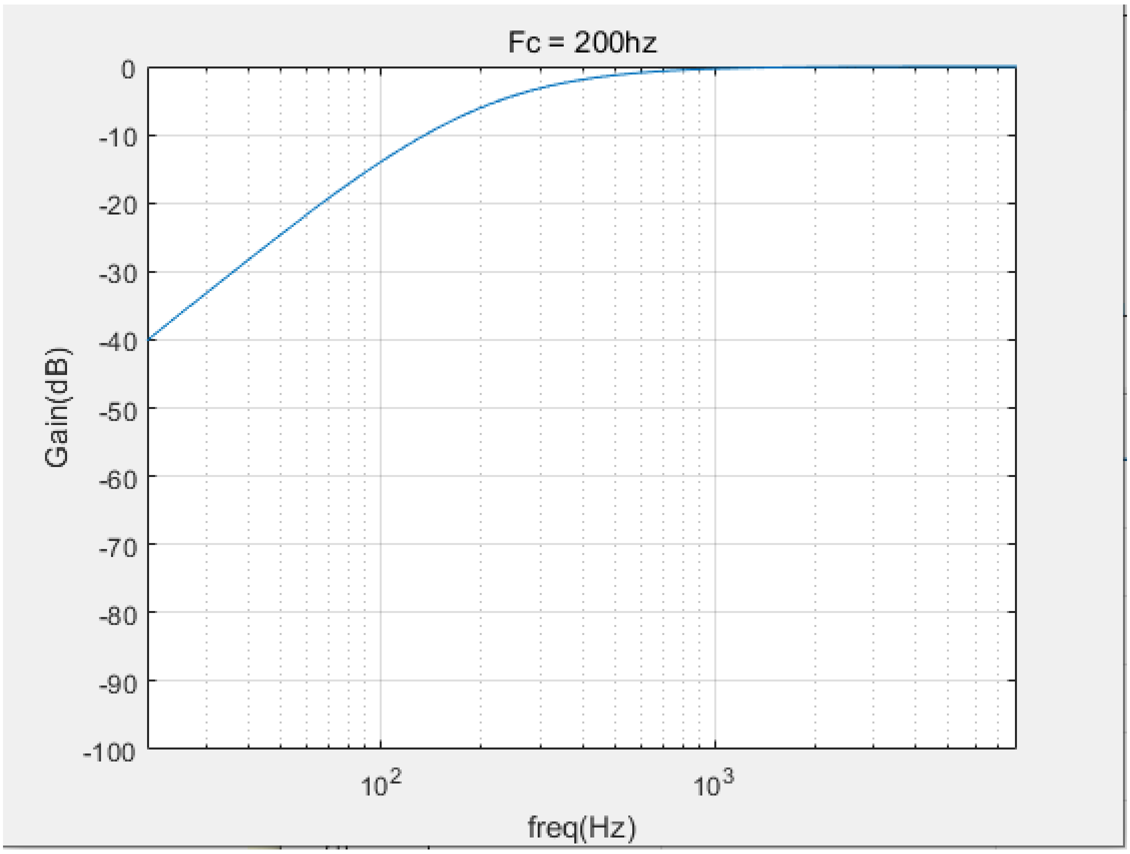
Parameter Name	Index	Range	Description
TX HPF Fc	200	0~Fs/2	TX high pass filter cut-off frequency
TX LPF Fc	201	0~Fs/2	TX low pass filter cut-off frequency
RX HPF Fc	340	0~Fs/2	RX high pass filter cut-off frequency
RX LPF Fc	341	0~Fs/2	RX low pass filter cut-off frequency

Note： if Fc = 0 or Fc = Fs/2 ,the filter is off.

The figure below shows an example of low-pass filter which Fc = 2000Hz:



The figure below shows an example of high-pass filter which Fc = 200Hz:

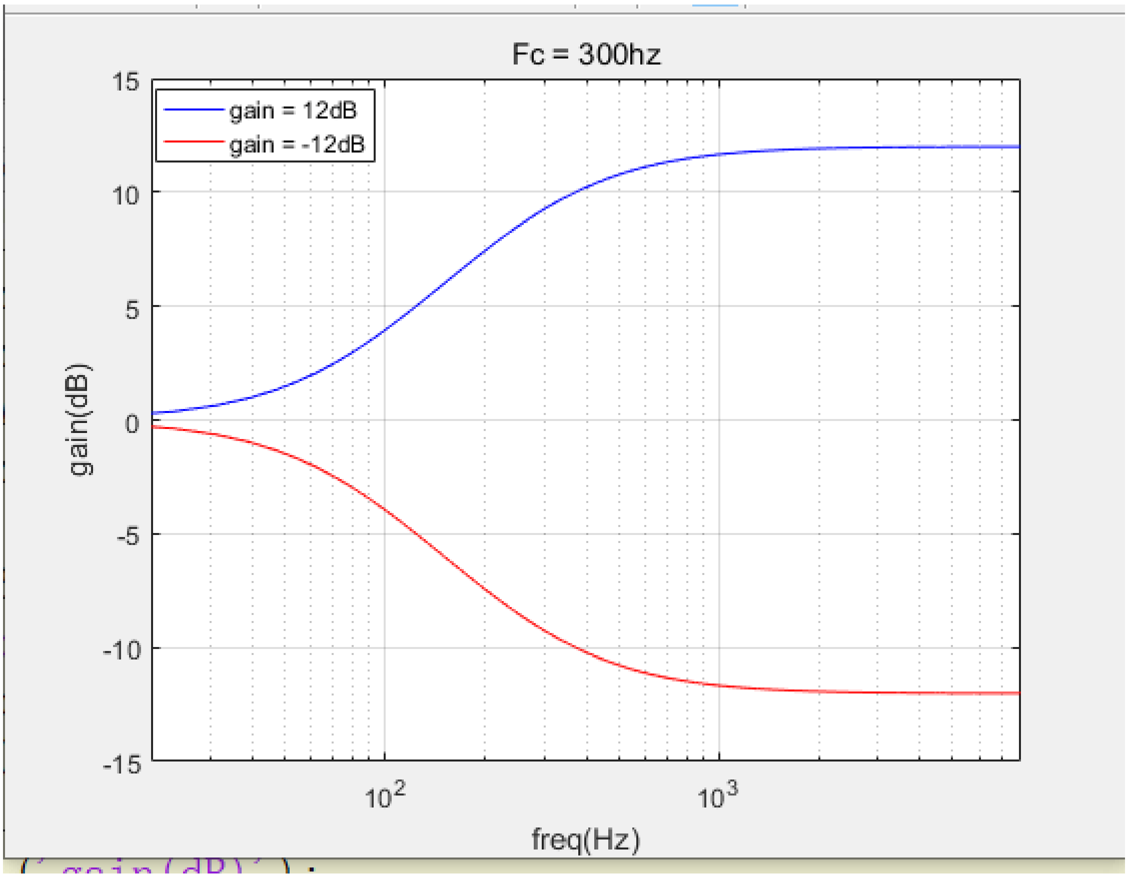


3.5.3 Shelf Filter

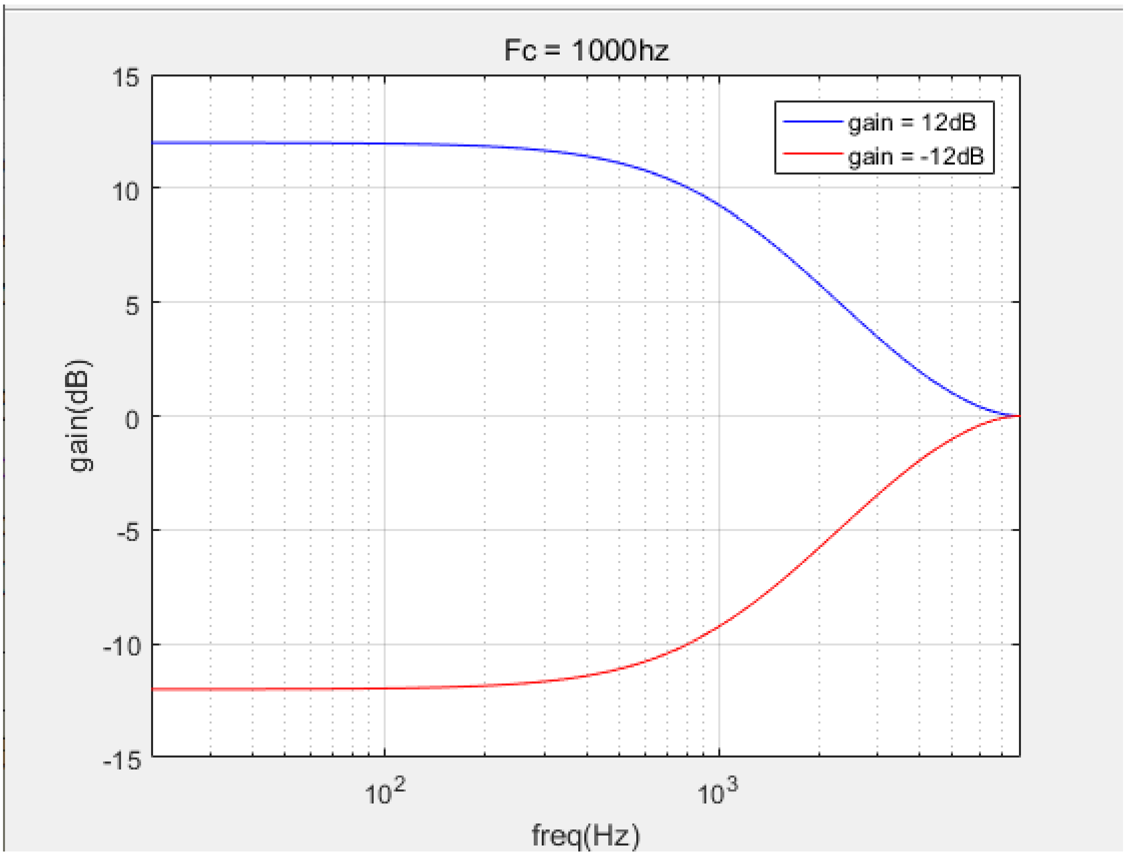
Parameter Name	Index	Range	Description
TX HSF Fc	202	0~Fs/2	TX high shelf filter cut-off frequency
TX HSF Gain	203	-12~12	TX high shelf filter Gain
TX LSF Fc	204	0~Fs/2	TX low shelf filter cut-off frequency
TX LSF Gain	203	-12~12	TX low shelf filter Gain
RX HSF Fc	342	0~Fs/2	RX high shelf filter cut-off frequency
RX HSF Gain	343	-12~12	RX high shelf filter Gain
RX LSF Fc	344	0~Fs/2	RX low shelf ilter cut-off frequency
RX LSF Gain	345	-12~12	RX low shelf filter Gain

Note： if Fc = 0 or Fc = Fs/2 ,the filter is off.

The figure below shows an example of high-shelf filter which Fc = 300Hz,Gain = ±12 :



The figure below shows an example of low-shelf filter which Fc = 1000Hz,Gain = ±12 :



3.6 CNG Parameters

Parameter Name	Index	Ranges	Description
TX CNG Enabled	440	0 or 1	0-off, 1-on
TX CNG Ratio	441		Applying ratio of TX CNG
TX CNG Amp	442		Applying amplitude of TX CNG

4. Hardware Test

4.1 Audio Precision Test

- Environment: AP
- Audio Signal: Sweep Signal
- Purpose: Measure audio quality
- Method: Use AP test microphone signal and reference signal
- Standard: THD+N < 5%

Special Note: if you use RK809 or RK817, recommend hpout+PA method. if you need use SPK, please note that:

(1) RK809 adn RK817 codec ClassD designed for 8ohm.

(2) RK809 adn RK817 codec do not support single-ended to get reference signal.

4.2 Audio Easy Test

- Environment: protetype
- Audio Signal: Sweep Signal
- Purpose: Measure audio quality
- Method: use protetype to play sweep signal and record it

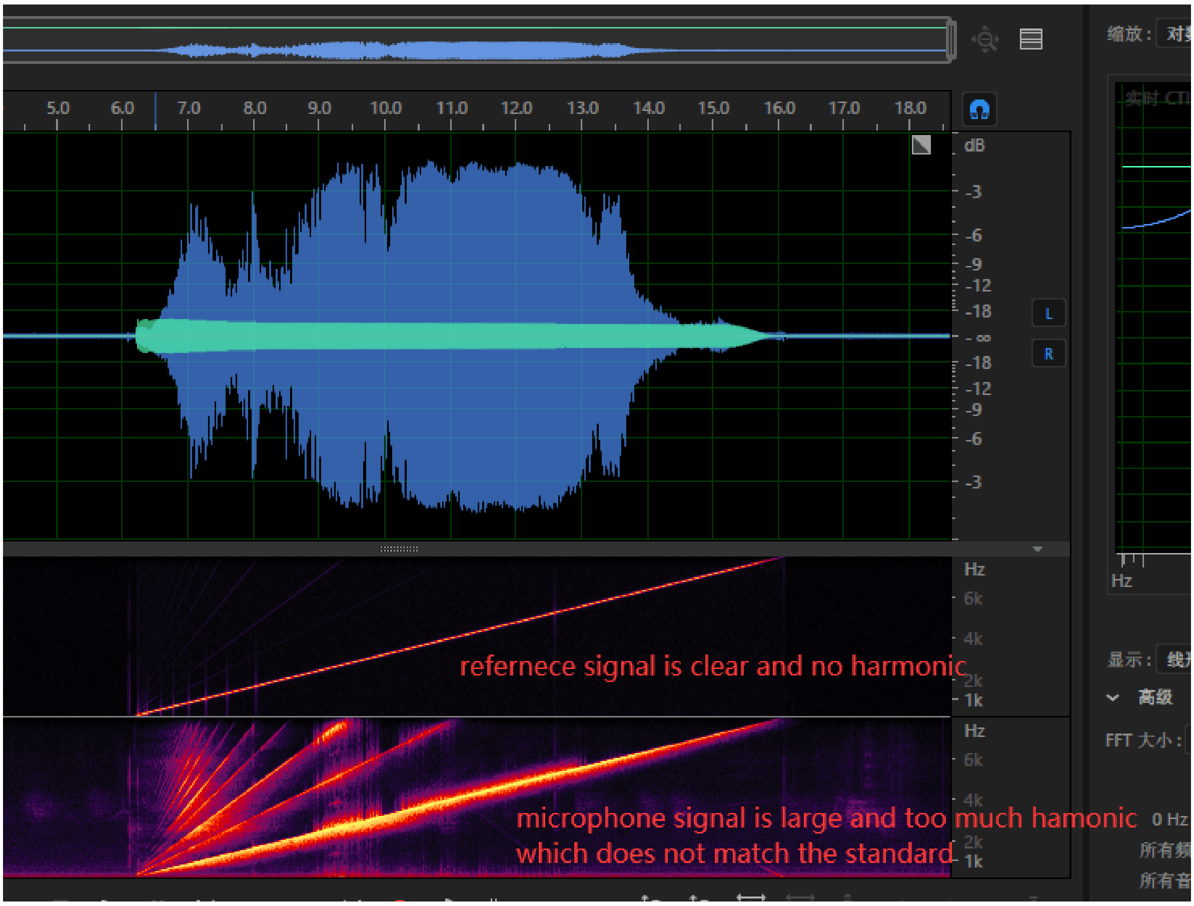
```
arecord -D hw:0,0 -c 2 -f S16_LE -r 16000 /tmp/sweep.wav
sox -b 16 -r 16000 -c 2 -n -t alsa hw:0,0 synth 20 sine 20:8000 // samplerate =
16k, and sweep signal from 20Hz to 8KHz
```

or

```
arecord -D hw:0,0 -c 2 -f S16_LE -r 8000 /tmp/sweep.wav
sox -b 16 -r 8000 -c 2 -n -t alsa hw:0,0 synth 20 sine 20:4000 samplerate = 8k,
and sweep signal from 20Hz to 4KHz
```

then, observe mic signal and ref signal's SNR and THD of sweep.wav.

The figure below shows an example of bad mic signal and good reference signal:



Special Note: Before use RK809/RK817 Codec to record and play, you should make sure record and play path according to hardware. Play path:

```
amixer -c 0 sset 'Playback Path' SPK // Open Codec ClassD
```

or

```
amixer -c 0 sset 'Playback Path' HP // Open Codec HPOUT
```

Record Path:

```
amixer -c 0 sset 'Capture MIC Path' 'Main Mic' // Open codec MIC
```

4.3 Hardware Structure

- The microphone and speakers are unobstructed, and the speakers are not facing objects such as wall or desktop;
- The microphone and the speaker should not be on the same panel, and the distance should be as far away as possible to avoid resonance;
- The sound pickup hole of the microphone should not be directly facing the speaker, it needs to be separated by materials such as sound insulation cotton.

4.4 Sealing Test

- Environment: Anechoic room
- Audio Signal: 0dBFS White Noise
- Purpose: Measure seal of microphone
- Method: The source audio is placed 50cm in the normal direction of the mic external plane, the mic inlet holes are blocked with rubber paste or other things in turn, and the white noise signal is played with high fidelity playback equipment with the volume level of 94dBA. The prototype to be tested is recorded, and the RMS size of mic channel blocked picked-up hole is measured, and the RMS difference between blocked and non-blocked picked-up holes is compared.
- Standard: RMS difference between blocking and not blocking the sound hole is more than 20dB.