Audio Enhancement SW tuning I2, I5, I6, I2M

Version 1.1

REVISION HISTORY

Revision No.	Description	Date
1.0	• Create this document	2019/07/30
1.1	• Add tool use guide and some annotations	2019/08/05

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1.1. Audio Block Diagram

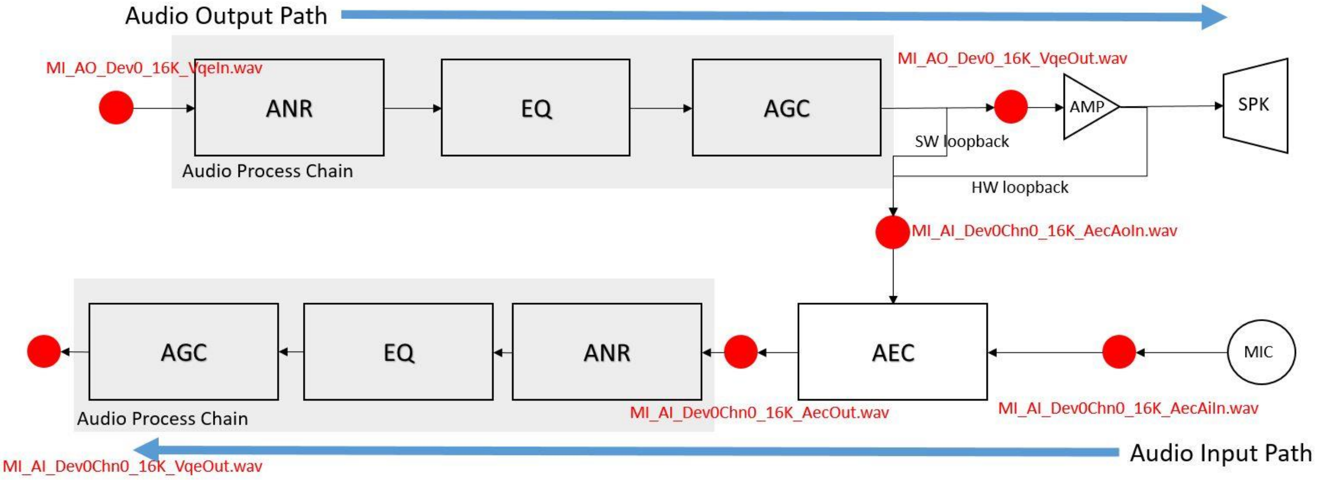

Figure 1: Audio enhancement block diagram (AEC Hardware loopback version and SW loopback can only choose one of them)

1.2. Audio enhancement terms and definition

1.2.1 AEC (Acoustic Echo Cancellation)

Acoustic echo is generated when the sound played out of a speaker device is coupled back to a microphone via direct or indirect paths. Therefore, a talker at the remote end hears his/her own voice back after a tangible delay, which is known as acoustic echo (AE). Acoustic echo cancellation (AEC) refers to the echo cancellation method that prevents the talker from hearing an echo of their own voice.

1.2.2 ANR (Acoustic Noise Reduction)

Acoustic noise reduction can reduce the noise which is continue to appear for a while from environment and keep the speech signal.

1.2.3 EQ (Equalizer)

The equalizer can enhance or attenuate some frequency band energy.

1.2.4 AGC (Automatic Gain Control)

The Automatic gain control can balance the power of signal. It can enhance power if the signal level is too low. It can attenuate power if the signal level is too high.

2. AEC TERMS AND DEFINITION

2.1. Acoustic Echo (AE)

Acoustic echo is generated when the sound played out of a speaker device is coupled back to a microphone via direct or indirect paths. Therefore, a talker at the remote end hears his/her own voice back after a tangible delay, which is known as acoustic echo (AE). Acoustic echo cancellation (AEC) refers to the echo cancellation method that prevents the talker from hearing an echo of their own voice.

The sources of coupling of the speaker to the microphone may include various paths, as follows:

1. Direct path between the speaker and the microphone, if any;
2. Reflections from the surface where the VoIP phone is kept;
3. Reflections from the walls and other objects/people around the VoIP enabled phone;
4. Coupling of sound via the physical enclosure of the phone, in form of vibrations from the chassis;
5. Loopback modes in hardware audio codecs at the audio front end of the phone.

2.2. Near-End Signal (Sin)

The signal recorded from a microphone, refers to the Near-End signal. It may mix the acoustic echo (AE) and the signal of the talker (NS). See Figure 2

2.3. Far-End Signal (Rin)

The source signal of a speaker, refers to the Far-End signal. It may come from the audio file (smart speaker plays out music) or Internet (someone talks from other device). See Figure 2

2.4. Single talk

Play some audio or speech from the speaker only (i.e., Sin = AE + NS, where NS =0). See Figure 2

2.5. Double talk

Play some audio or speech from the speaker and someone talks in front of the microphone
(i.e., Sin = AE + NS, where NS≠0). See Figure 2

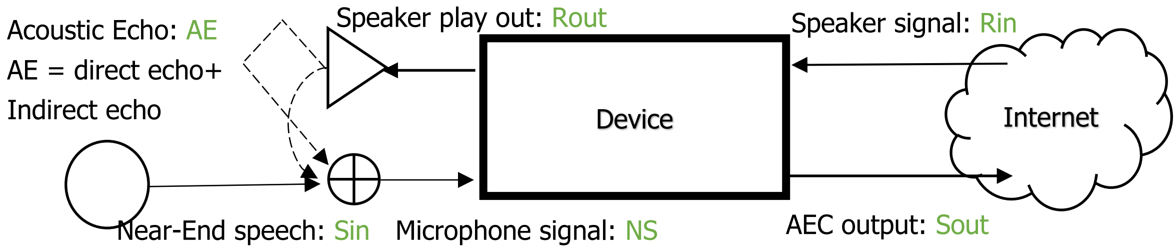

Figure 2: Relationship between the input signal and the output signal

3. AEC SPECIFICATION

Following lists the AEC specification:

1. The distance between the speaker and the microphone is preferably from 4 cm to 10 cm, the further the better.
2. If audio files are used as the sound source, make sure the following:
   A. There is no aliasing in the audio files, see Figure 3
   B. There is no signal clipping in the audio files, see Figure 4
   C. Effective sample rate is 8 kHz or 16 kHz;
   D. The Far-End signal should be fed in AEC library when the echo appears, see Figure 5
3. The delay time between AO and AI must be smaller than 0.128 sec when sample rate is 16 kHz (or 0.256 when 8 kHz), and the timing of Far-End signal should be earlier than Near-end signal, see Figure 6
4. The sample rate of the microphone and the speaker must be the same.
5. The speaker should not be overloaded, otherwise the AEC performance will decrease as nonlinear increases.

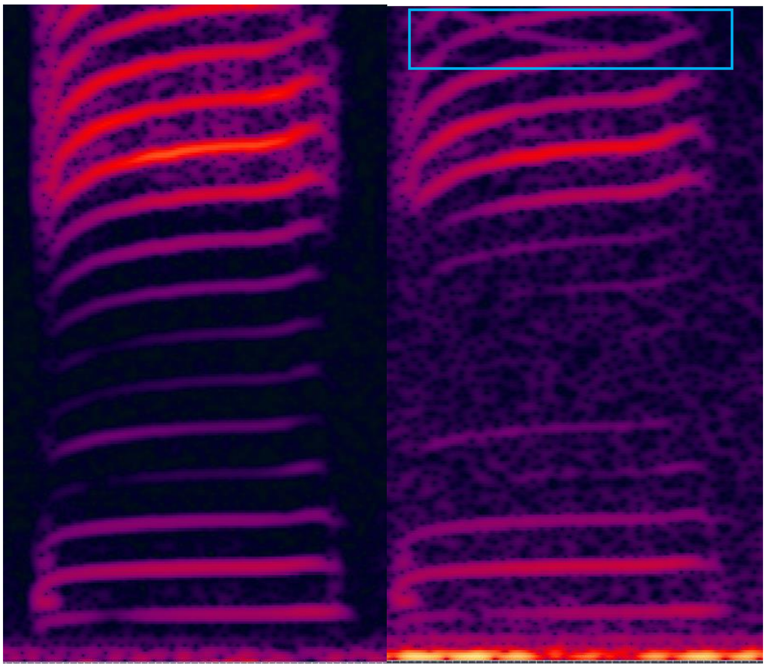

Figure 3: The right hand signal is aliasing (This will affect AEC result.)

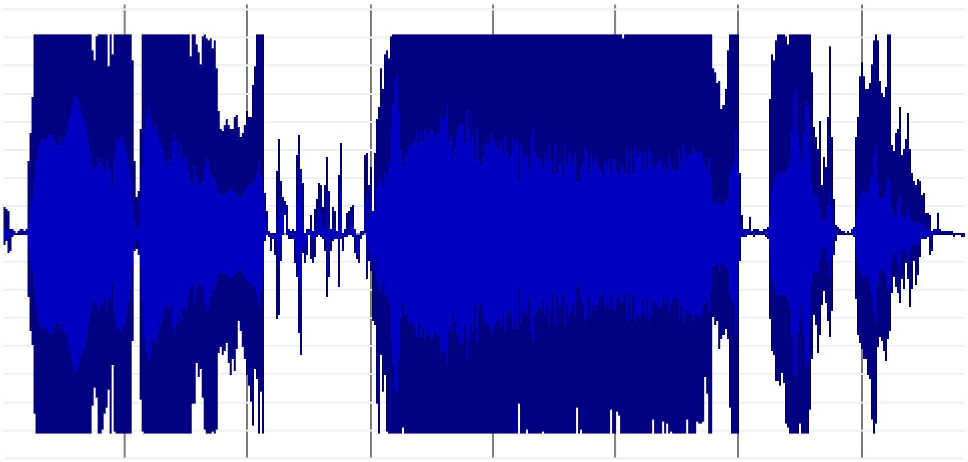

Figure 4: The signal is clipping (This will affect AEC result.)

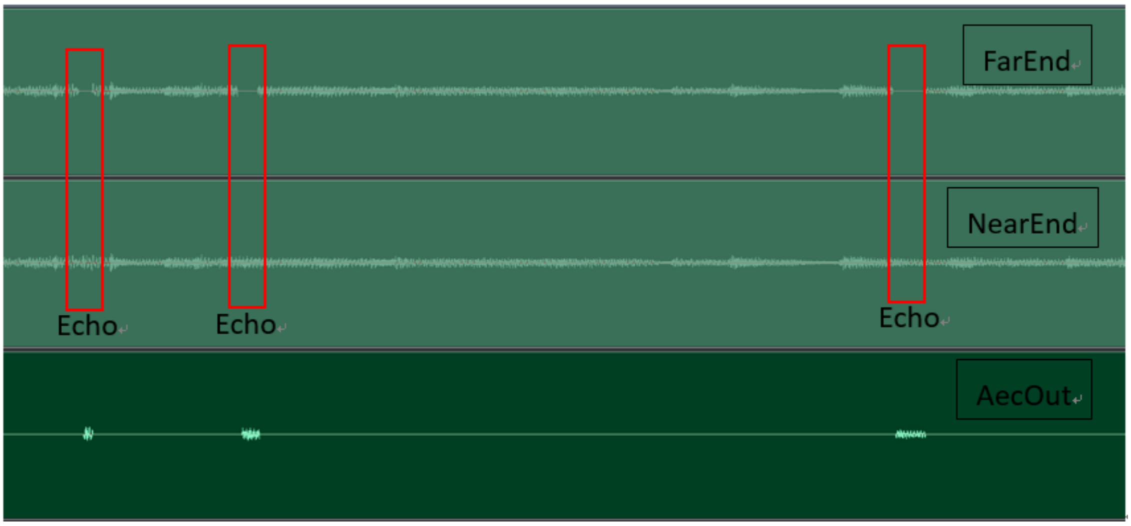

Figure 5: The Far-End signal is not fed in AEC when the echo appears (The result will be affected.)

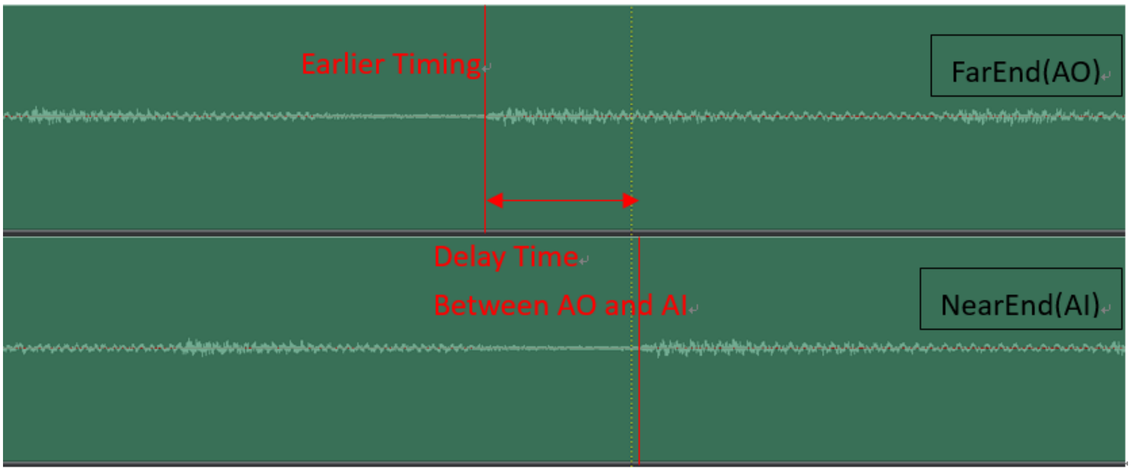

Figure 6: The delay time between AI and AO must be smaller than 0.128 sec when sample rate is 16kHz (0.256 when 8kHz), and the timing of Far-End signal should be earlier than Near-End signal.

4. REAL SET HOUSING ACOUSTIC TEST

4.1. The relationship between AEC quality and housing.

Housing design determines to a large extent the performance of a hands-free device (e.g., a phone with speaker -phone mode). Voice quality can suffer significantly due to poorly designed device enclosure even when the world's best AEC software is used. A few primary recommendations to ensure that hardware does not become the limiting factor in terms of AEC performance are listed in the following sections. For example: wideband speech codec support makes sense only when the microphone and speaker are wideband. Similarly, if the microphone or the speaker induces any non-linearity in the echo path, then no linear model of the echo will be able to cancel the echo effectively, this will cause distortion side effect in full-duplex condition.

4.1.1 Housing design recommend.

- Speaker:

The Rear Enclosure is must. Do not use speaker only in product housing. See the Figure 7.
A tight fitting junction between the front and rear of the speaker enclosure will help to reduce the sound propagation from the rear cavity to the front of the speaker. The speaker should be securely mounted at the interface between the front and rear cavities as the speaker will serve as a part of the rear enclosure structure. The secure mounting of the speaker will also ensure rattling sounds are not created by the speaker and enclosure. High density foam is often used when mounting speaker frames to enclosures to assist in creating a secure and tight fitting configuration. The hole of front cavity should at least 20% of speaker area and spacing from speaker frame 1-2 mm

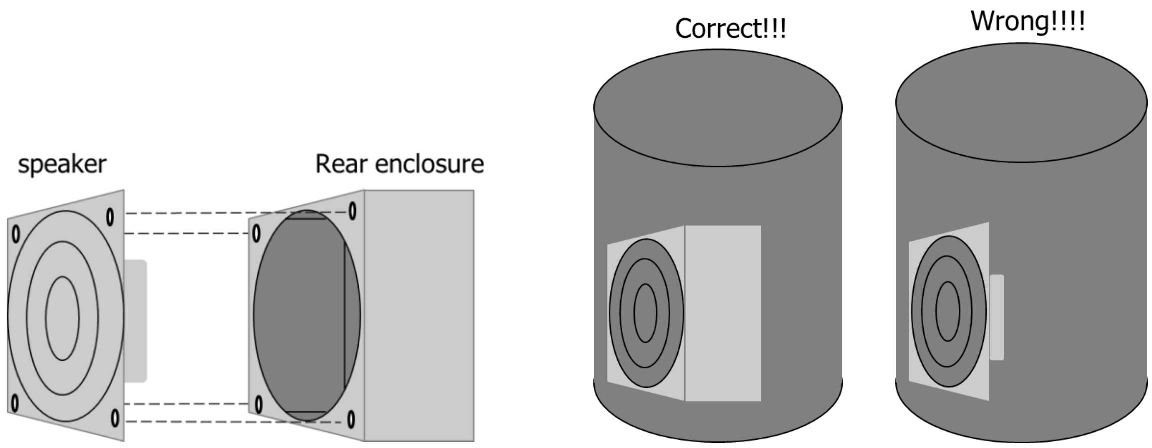

Figure 7: Please device the rear enclosure in speaker. The speaker vendor will design.

- Microphone:

Ensure that the microphone is housed in foam to increase ERL (Echo Return Loss) by reducing speaker-microphone coupling through direct path. Going one step ahead, one could also encase the microphone and surrounding foam in a separate housing within the enclosure. The microphone should line up the hole. See Figure 8

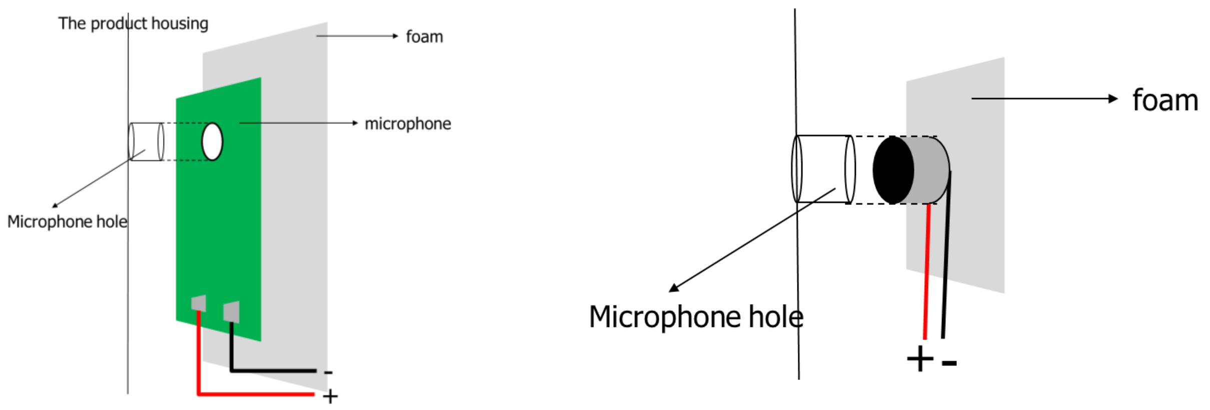

Figure 8: The microphone should line up the hole and ensure that the microphone is housed in foam

- The position of microphone and speaker

The distance between speaker and microphone should as far as possible. If the distance is too close, the acoustic echo recorded from microphone will clipping, to keep AEC performance user should setting the volume of speaker lower, then the sound pressure level will not loud enough. As Figure 9

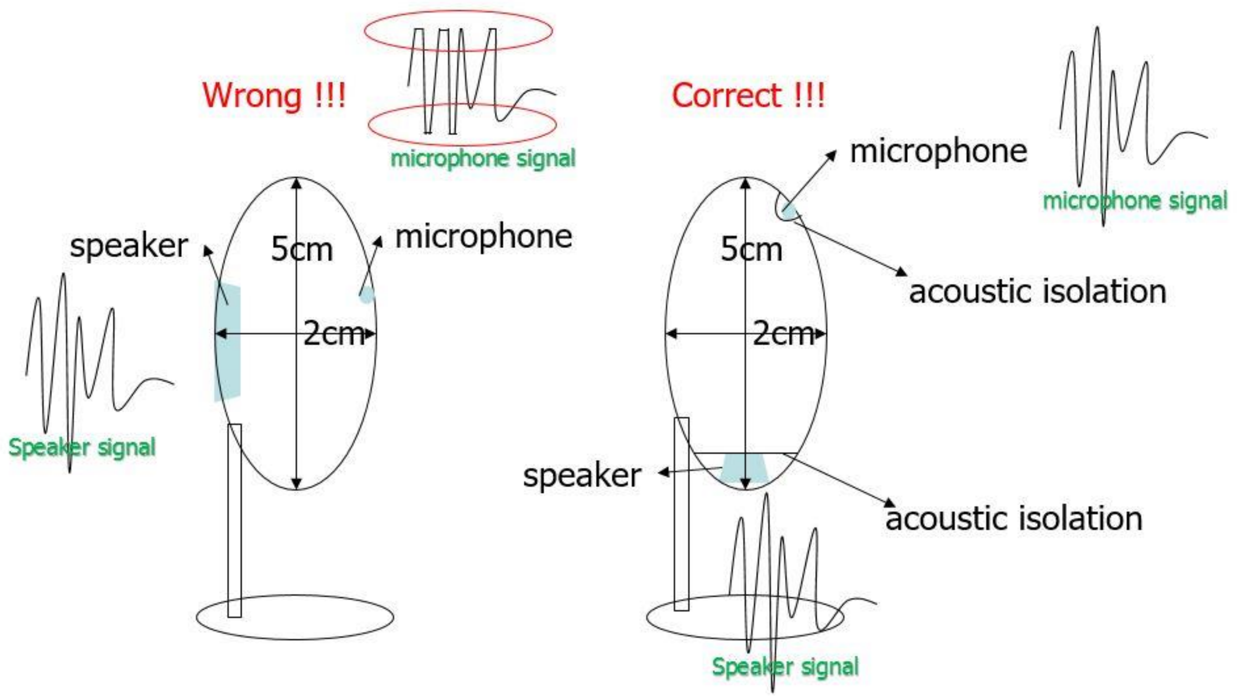

Figure 9: The distance between speaker and microphone should as long as possible or the microphone data will clipping, the audio performance will very poor.

4.2. Objective acoustic audio test

After design a housing of product, user needs to evaluate three objective value for verify the housing is good enough for audio quality.

- Internal Isolation:
  It can measure the isolation inside the housing between speaker and microphone
- Echo Return Loss (ERL):
  It can measure how much power of acoustic echo attenuate from speaker to microphone.

User need to decide the speaker volume (DBA) and microphone level at beginning.
If the speaker is 1W/8ohm, the speaker do not over 70 DBA (measure at 1m) or the audio qualify will very poor.

4.2.1 The measurement of speaker level and microphone level.

The testing environment noise cannot over 40dbA. Please put decibel meter 1m away from DUT speaker and playout a TestFile1.wav (Pink noise) from speaker of DUT, if the level is not loud enough, please adjust AO gain. See Figure 10.

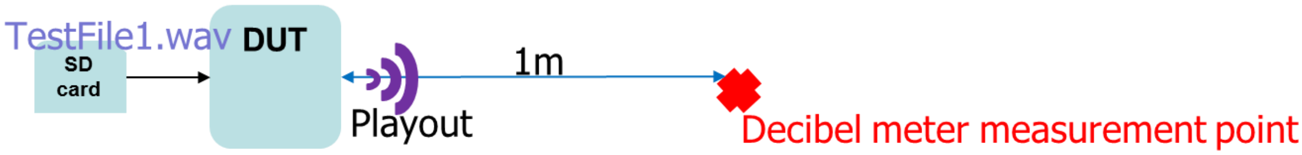

Figure 10: The method for measure speaker level.

After speaker level measurement, user need to measure the microphone level. Please put the decibel meter at DUT microphone position, and put an external speaker 1m away from DUT. Please playout TestFile1.wav from external speaker and measure the decibel meter should show 70DBA. Record the data and save in SD card. The digital level should be -25 DBRMS (recommend). If the microphone level is too low, please adjust the microphone gain. See Figure 11.

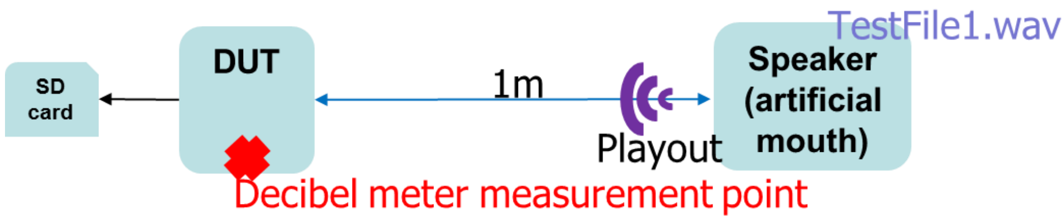

Figure 11: The method for measure microphone level.

NOTICE: The speaker level and microphone level should satisfied: When speaker playout a normal speech the microphone cannot clipping.

4.2.2 Internal Isolation

The value of internal isolation the bigger the better, please make sure this value is bigger than 6db.
Please disable every audio improvement process.
Step 1: Put DUT in a silence environment, the environment noise cannot over 40dbA
Step 2: Playout TestFile1.wav from DUT speaker and record the acoustic echo from microphone at the same time (Record2.wav).
Step 3: Use clay to seal the microphone hole and repeat the step2 (Record3.wav).
Step 4: Internal Isolation = AVGRMS (Record2.wav) – AVGRMS (Record3.wav), AVGRMS is the average of DBRMS.

4.2.3 ERL (Echo return loss)

The value of ERL the bigger the better. Please make sure the value is positive.
Please disable every audio improvement process.
Step 1: Put DUT in a silence environment, the environment noise cannot over 40dbA
Step 2: Playout TestFile1.wav from DUT speaker and record the acoustic echo from microphone at the same time (Record4.wav).
Step 3: ERL1 = AVGRMS (TestFile2.wav) - AVGRMS (Record4.wav), AVGRMS is the average of DBRMS.
Step 4: AI and AO gain set as 0. Please repeat Step2, but playout TestFile3.wav instead of TestFile1.wav and record to Record5.wav
Step 5: Record5.wav cannot clipping at any frequency

4.2.4 The measurement of DBRMS

Download Audacity, and plug in stats.ny, show as Figure 12.

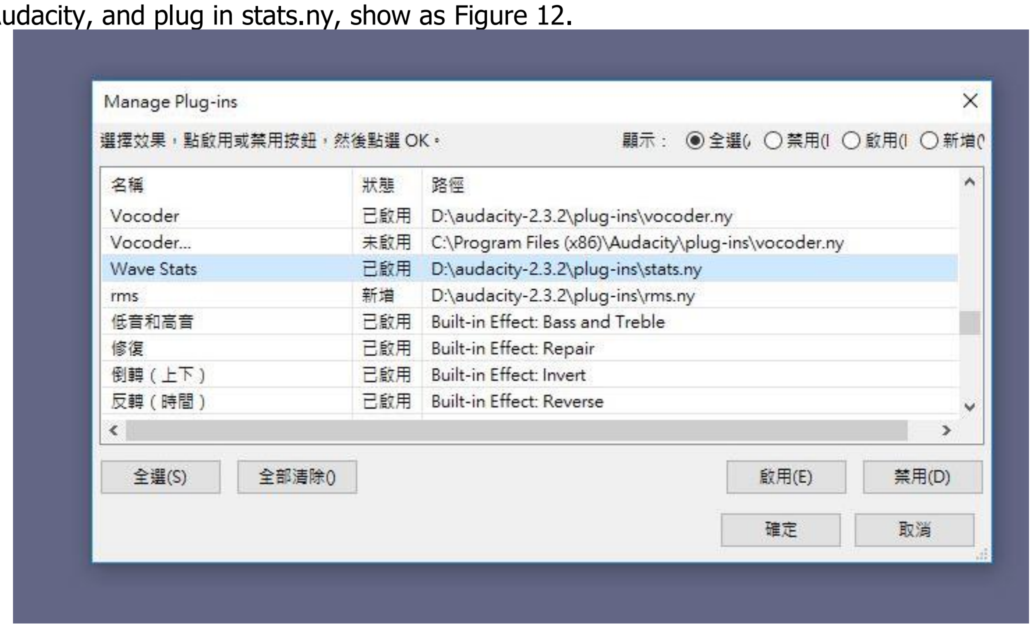

Figure 12: Audacity plug in stats.ny

Import a wave file, select analysis >> wav stats, and you will see a window show as Figure 13. Please adjust as 10 sec.

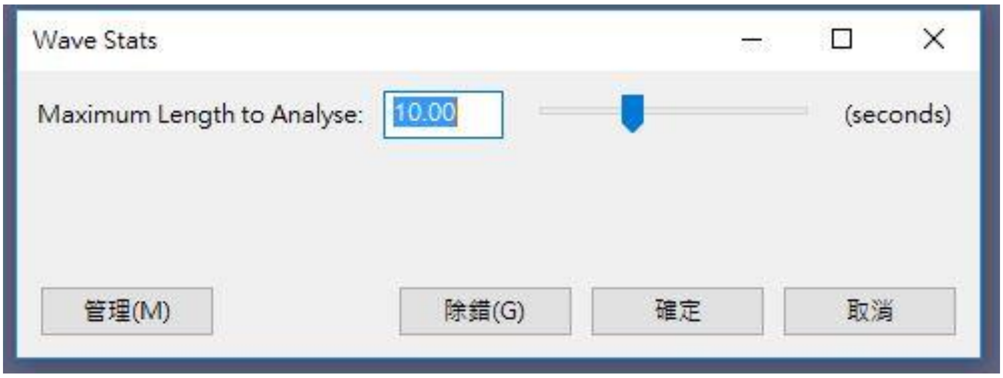

Figure 13: Audacity wave stats.

Next you will see a window like as Figure 14

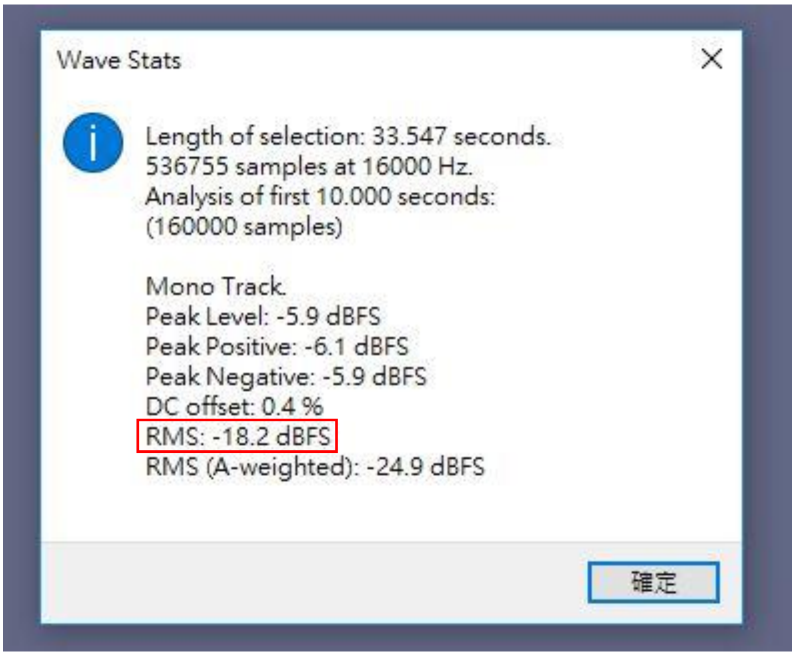

Figure 14: The element of RMS shows the average DBRMS.

4.2.5 The measurement of frequency response

Import a wave file, and select a region you need to analysis, show as Figure 15. Select analysis >> Frequency graph, and you will see the frequency response, show as Figure 16. Please using Hann window and 128 point to analysis frequency response.

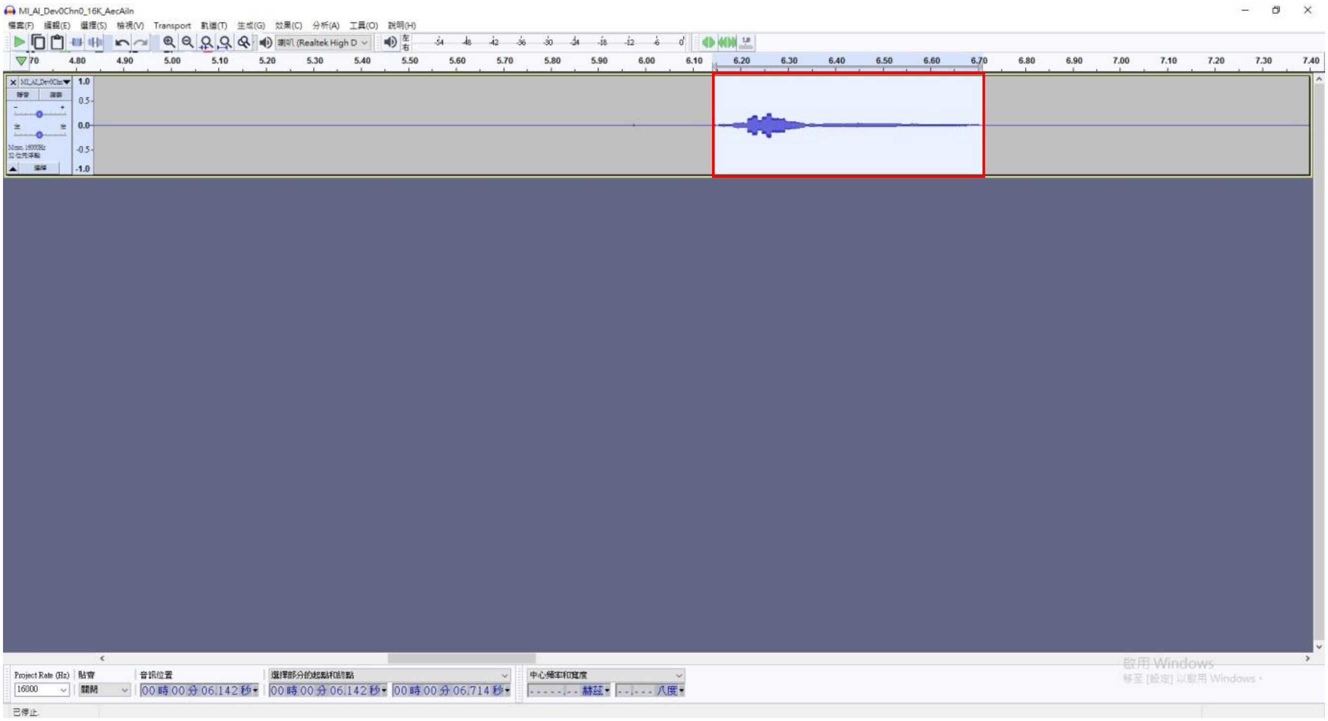

Figure 15: Select the region you needed.

Figure 16: The Frequency response window.

5. WHEN CUSTOMER CANNOT MODIFIED THE HOUSING DESIGN

If the customer cannot modified the housing, the AEC performance will downgrade. The side effect show as

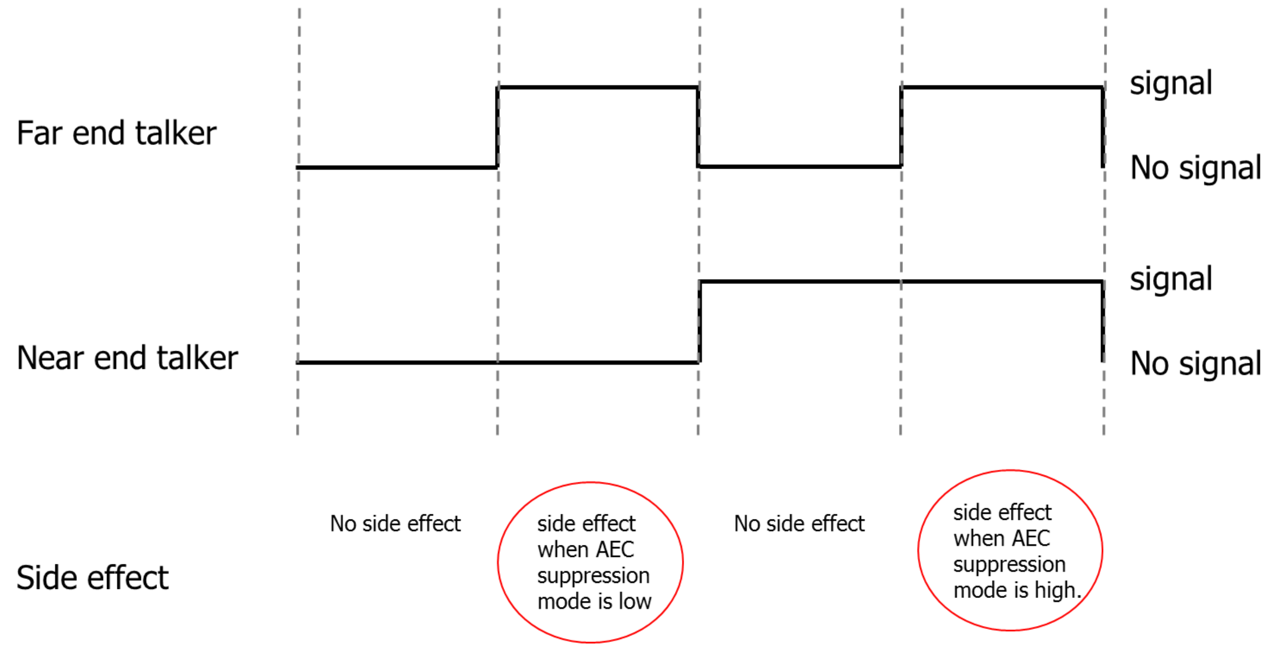

Figure 17: The side effect will appear when the housing cannot modified. It will appear two conditions (red circle) are mutually exclusive.

If the far end talker and near end talker talk at the same time, the suppression mode setting lower, the far end talker can hear the echo, but the suppression mode setting higher the far end talker cannot hear the voice come from near end talker.

Because cannot solve the root cause (housing), no matter what user tuning software, it will still get some side effect.
In this chapter we give some recommendation about software tuning, but user still got some side effect.

5.1.1 AEC tuning

Step 1: AO gain setting 0 db, AI gain setting 0 db.
Step 2: Repeat section 4.2.3 Step4 and get a wave file, Record5.wav.
Step 3: Open the file, Recored5.wav, show as Figure 18. You can see the digital level is still low.
Step 4: Playout a speech wave (TestFile2.wav) file from speaker, and record as Record_S.wav, show as Figure 19 You can see the digital level is still low. User can adjust microphone gain higher.
Step 5: Adjust microphone gain 3db until the Record_S.wav clipping. And choose the microphone gain biggest Level but Record_S.wav not clipping, show as Figure 20

If the housing is bad the microphone gain or the speaker gain very small, please see Section 5.1.2, 5.1.3 and 5.1.4.

Step 6: After fix the AO gain, AI gain and enable some audio function if needed, default AEC suppression mode setting as

MI_U32 u32AecSupfreq[6] = {20,40,60,80,100,120};
MI_U32 u32AecSupIntensity[7] = {4,4,4,4,4,4,4};

Please test single talk first. If there are some residual echo after AEC, please review the PCM wave, MI_AI_Dev0Chn0_16K_AecOut.wav. Check the frequency band of residual echo. If the frequency of residual echo is concentrated (show as Figure 21), please adjust the u32AecSupfreq and u32AecSupIntensity and go to Step 7. Otherwise please set u32AecSupIntensity[7] higher.

Step 7: Calculate the u32AecSupfreq[6] corresponding to the residual echo frequency band according to the point number and sampling rate. Then, turn up the u32AecSupIntensity parameters on the corresponding frequency band.

Refer to Figure 21, the frequency band can be calculated as in the following equation:

$$\frac{residual\ echo\ frequency}{\frac{sampling\ rate}{2}} \times point\ number = \frac{2600\sim2800}{4000} \times 128 \approx 83\sim90$$

Apply AEC suppression mode, as follows.

MI_U32 u32AecSupfreq[6] = {20, 40, 60, 83, 90, 120};
MI_U32 u32AecSupIntensity[7] = {4, 4, 4, 4, 8, 4, 4};

After the adjustment, the residual echo is suppressed, as shown in Figure 22

Step 8: Test the double talk and repeat the Step6 and Step7.

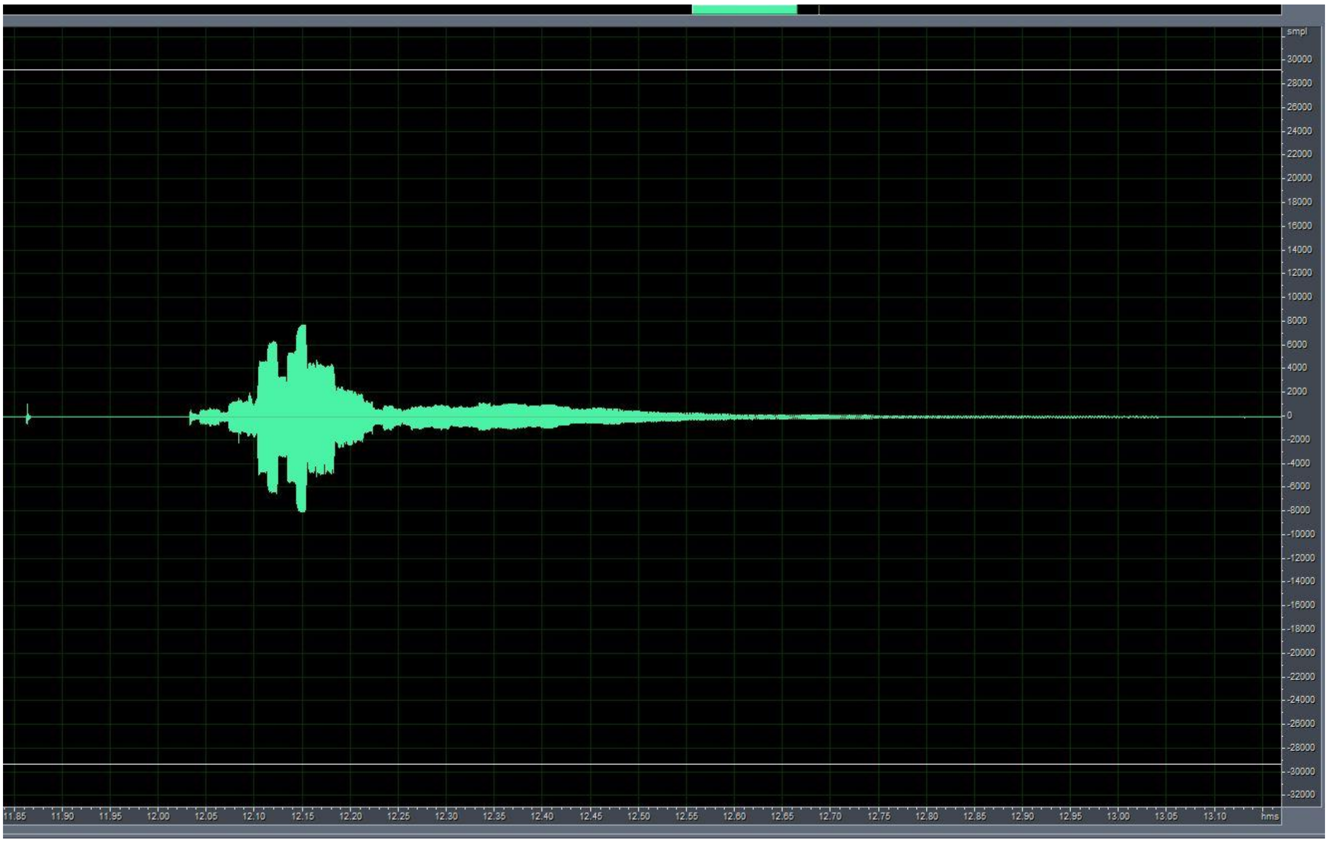

Figure 18: The acoustic echo when play out a sweep wave from speaker.

Figure 19: When microphone level is 0 DB

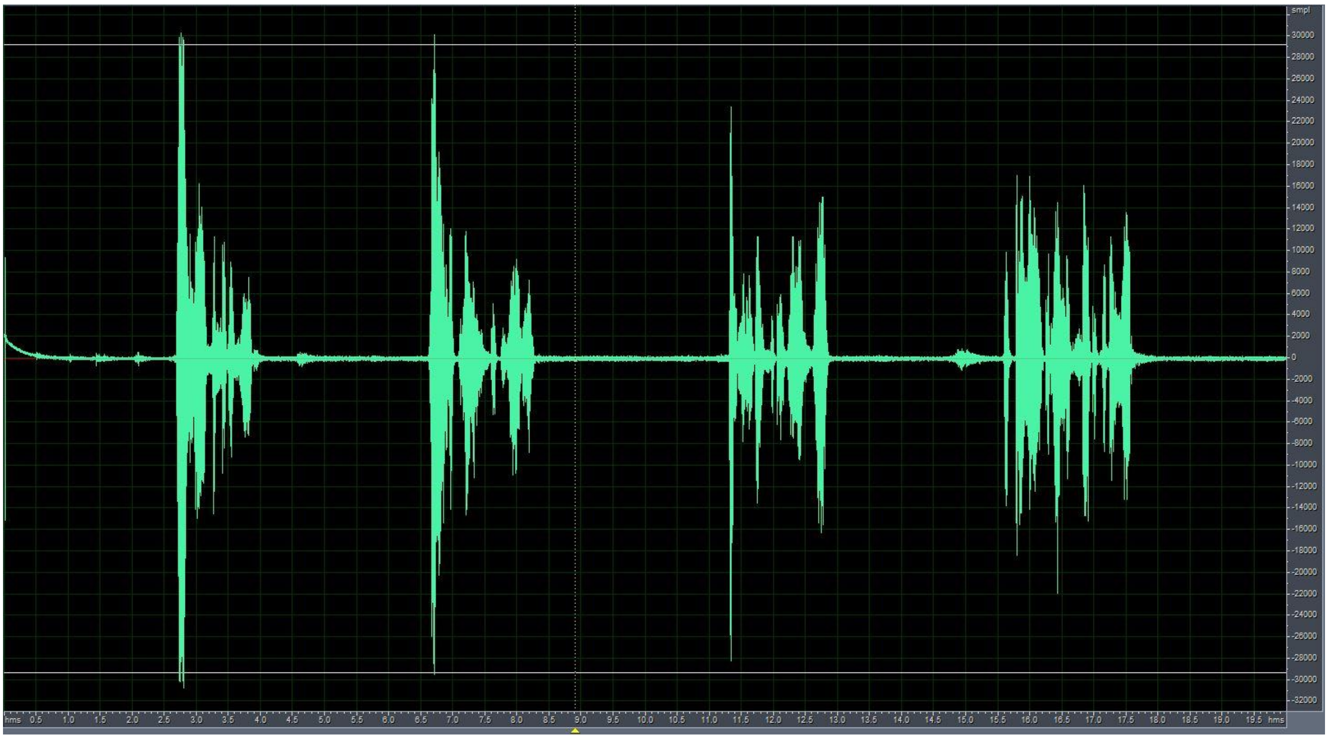

Figure 20: When microphone level adjust to 24DB.

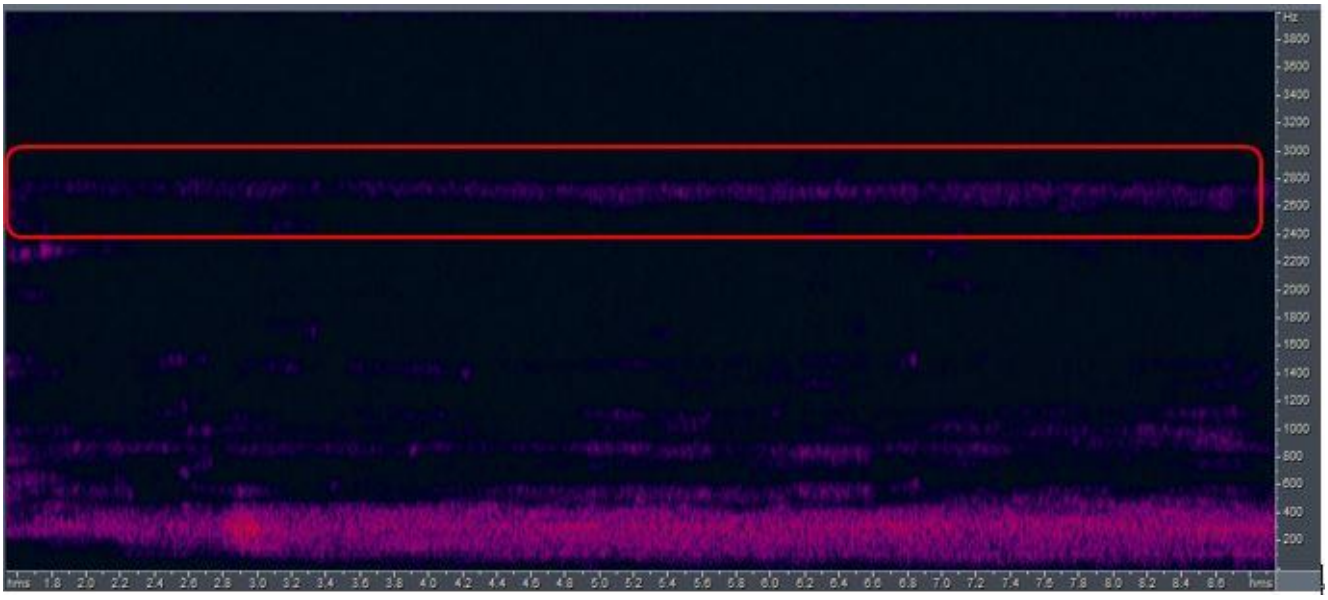

Figure 21: The frequency of residual echo is concentrated at 2600~2800 Hz

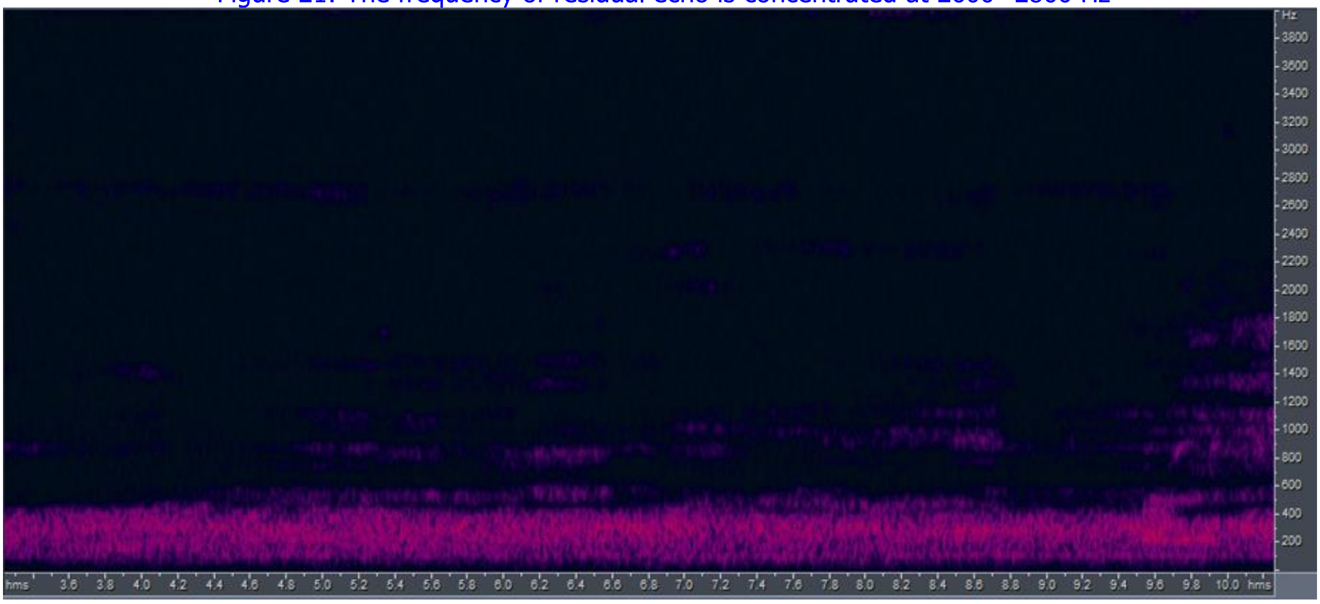

Figure 22: The result after adjust the AEC parameter.

5.1.2 AEC parameter tuning – case 1

If the user finish the SOP in section 5.1.1 , the microphone gain is very small, that's means the isolation of housing is very bad. If customer can fix the housing isolation problem, it is the better way. If the user cannot fix the housing problem, please let customer knows that the side effect is cannot avoid.

If customer can turn down the speaker gain, please turn down the speaker level at first and repeat the section 5.1.1to find the new microphone gain.

Otherwise, please follow the below SOP

Step 1: open the Record5.wav, show as Figure 23. And high light the acoustic echo and transform to frequency domain. Show as Figure 24.

Step 2: Enable the EQ in AO path, and using EQ suppress the power which is higher frequency response. Show as Figure 24. The EQ setting method please see the chapter 12

Step 3: Apply the EQ table in AO path, and repeat the SOP in section 5.1.1

Step 4: If the microphone still very small, please suppress more DB value in Step2. And repeat the Step3.

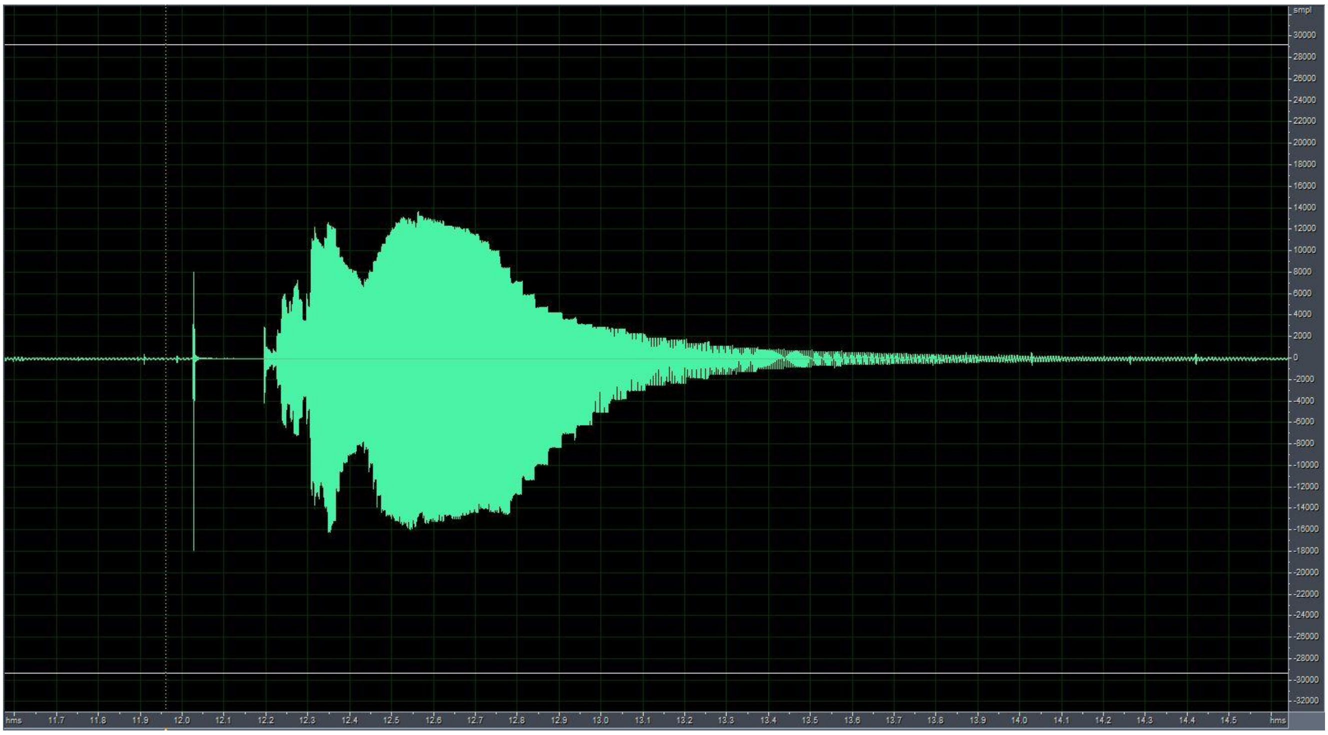

Figure 23: The Acoustic Echo record from microphone when playout sweep.

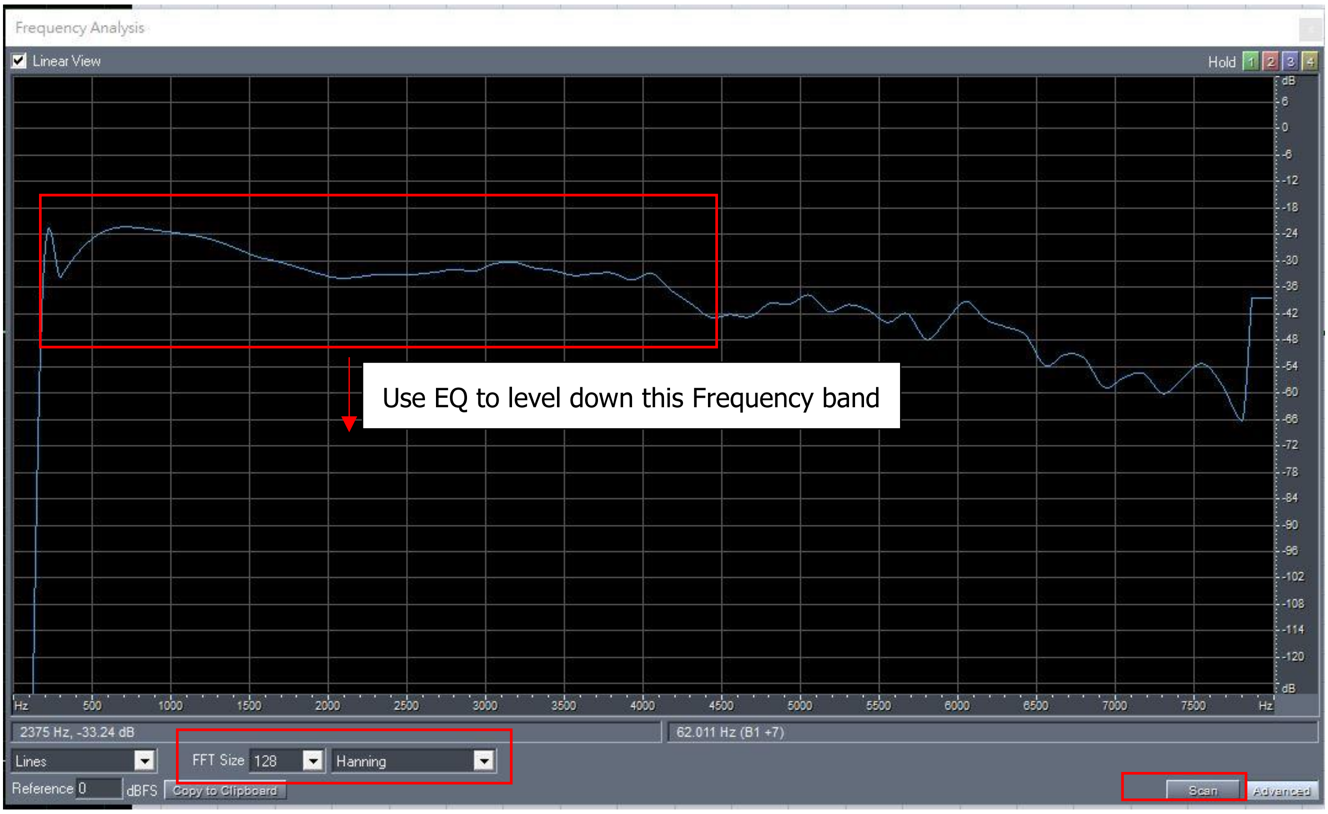

Figure 24: The acoustic echo frequency response.

5.1.3 AEC parameter tuning – case 2

If the using EQ cannot help the AEC result, please turn down the microphone gain and enable the AGC function and NR function in microphone. The AGC tuning method see the chapter 7 and chapter 8. The NR tuning method see chapter 9.

5.1.4 AEC parameter tuning – case 3

If the microphone output is still small, let the acoustic echo is clipping, turn off the hardware loopback first and please set the suppression mode higher. Please let customer know the side effect when AEC is enable, shown as Figure 25.

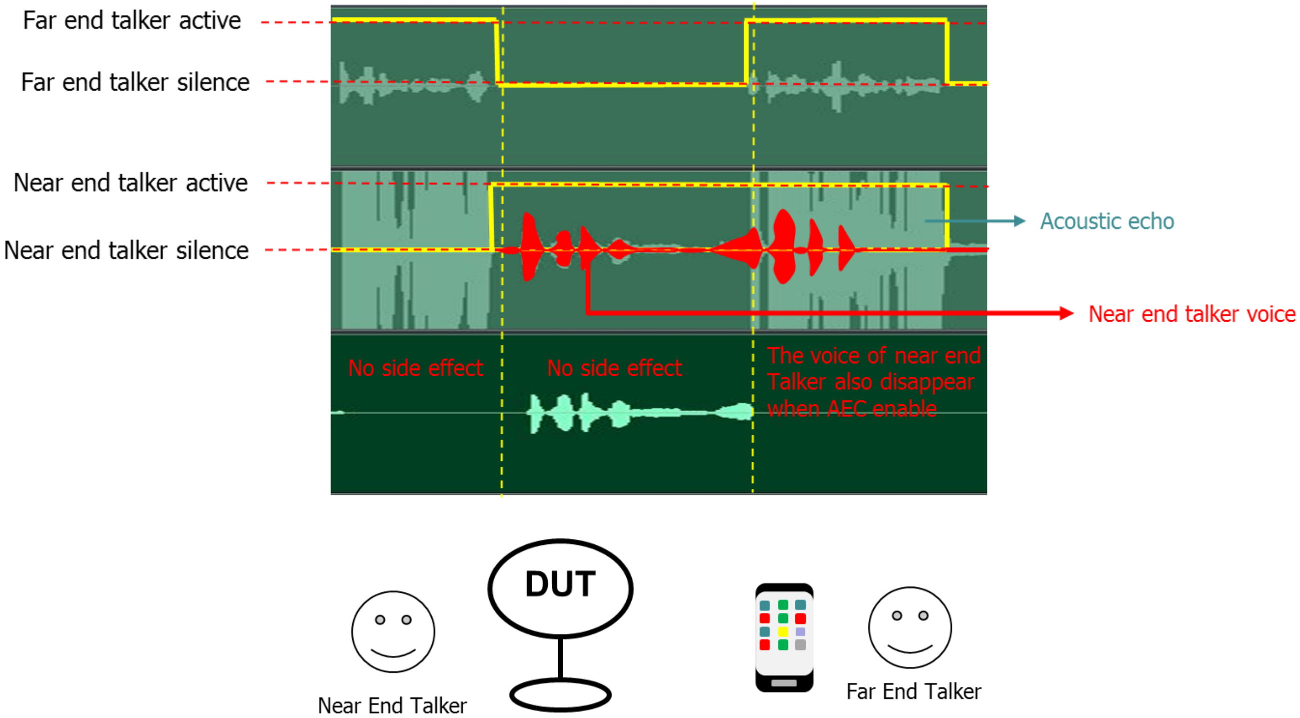

Figure 25: The side effect when far end and near end active at the same time

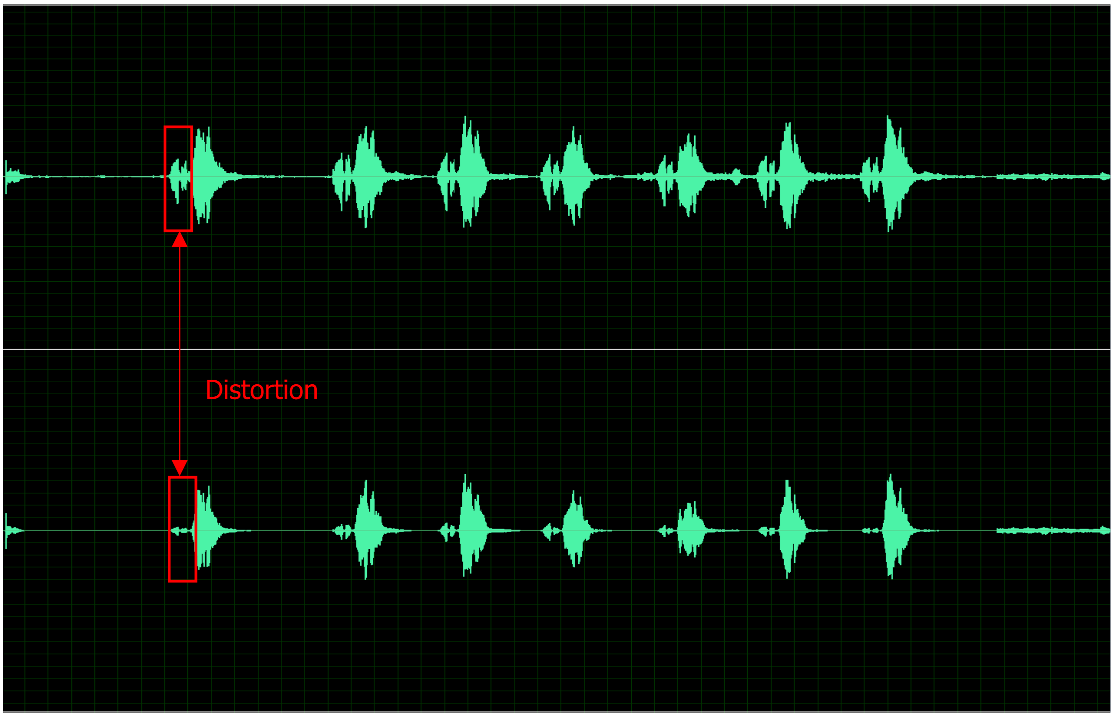

Figure 26: This figure shows the distortion when double talk, the power of near-end speech may decrease

6. AEC RESULT WAVE

6.1. AEC result in different modes (4 and 15)

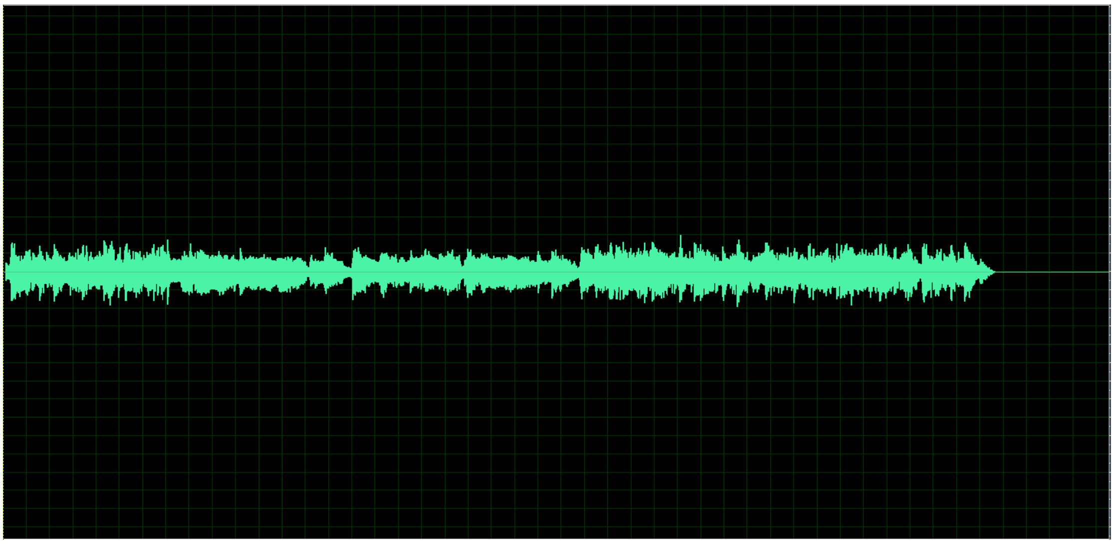

Figure 27: Speaker digital signal (FarEnd)

Figure 28: Microphone digital signal (NearEnd)

Figure 29: AEC digital output signal (AecOut). AEC mode is set as 4

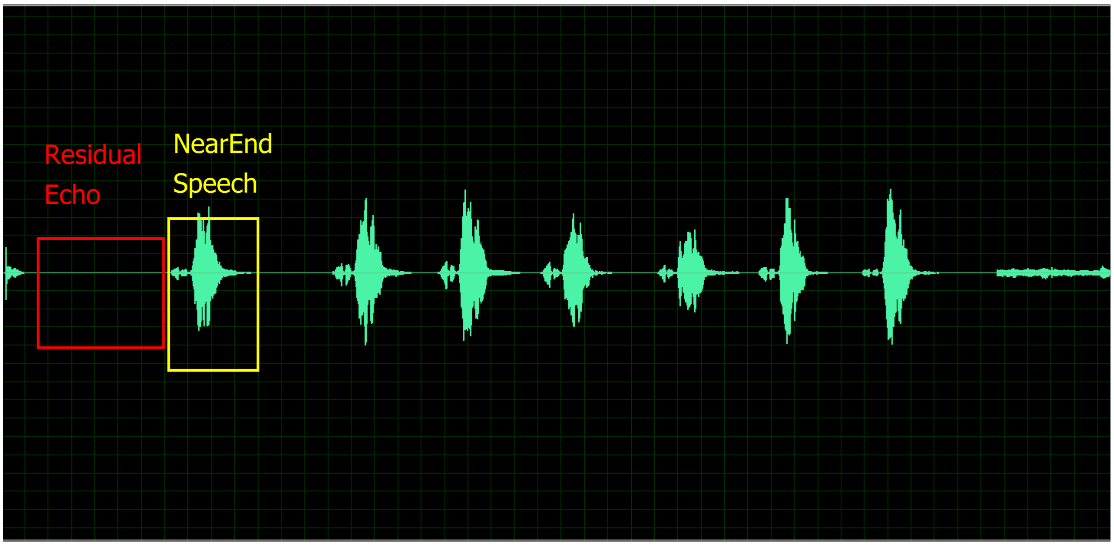

Figure 30: AEC digital output signal (AecOut). AEC mode is set as 15

AEC suppression mode determines how aggressive AEC is. Mode 0 is the most conservative while mode 15 is the most aggressive. The more aggressive the mode is, the fewer residual echoes with the more distortion. We recommend mode 4.

7. AGC TERMS AND DEFINITION

7.1. Compression Ratio Curve

This is the relation (ratio) between input power level (dBFS) and output power level (dBFS). The compression ratio compresses the dynamic range of audio. You can set up to five coordinate points on the relationship coordinates of input and output power level to determine the curve you want. This five points which contain four turning points can determine four different slopes.

For example, if the compression ratio setting as follow:

compression_ratio_input[5] = {-80, -60, -30, -12,0}
compression_ratio_output[5] = {-80, -45, -18, -9, -6}

As Figure 31 show, these parameter represent when input is -80dBFS, the target level would be set -80dBFS; Input is -60dBFS, the target level would be set -45dBFS…and so on.

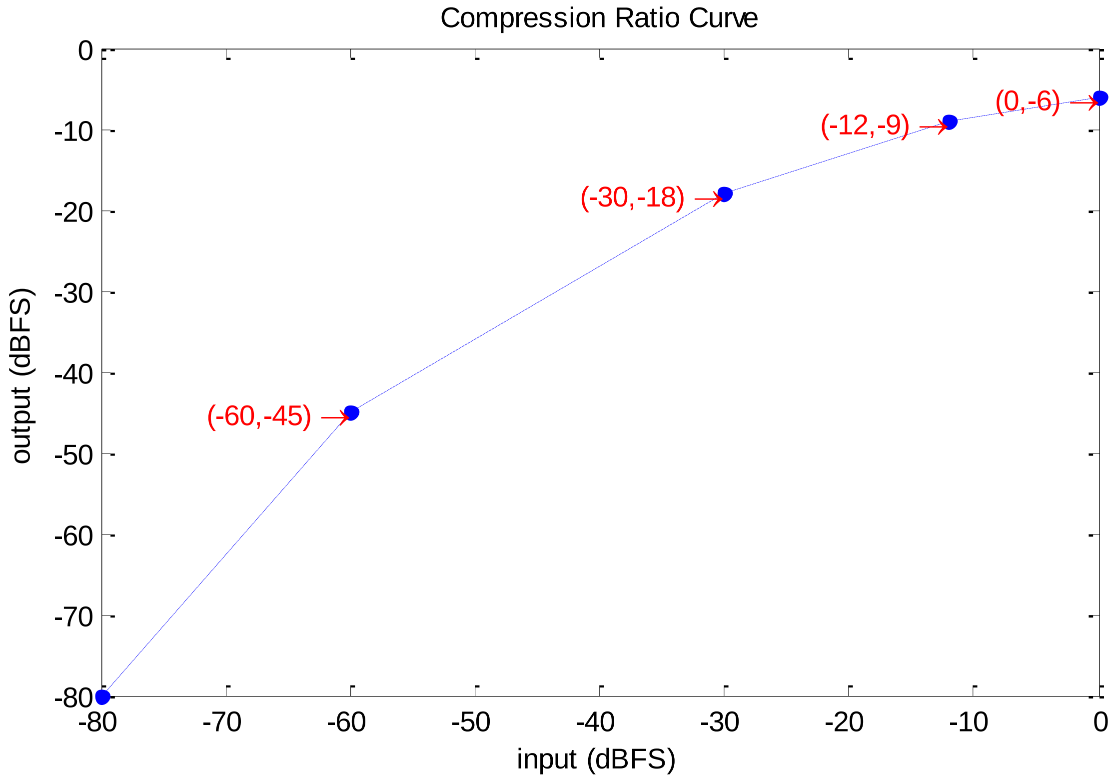

Figure 31: Compression ratio curve

If you don't need 4 turning point, you can set the remaining parameters to zero

compression_ratio_input[5] = {-70, -60, -30, 0,0};
compression_ratio_output[5] = {-60, -50, -10, -3, 0};

As Figure 32 show, this set of parameter only have 3 different slope.

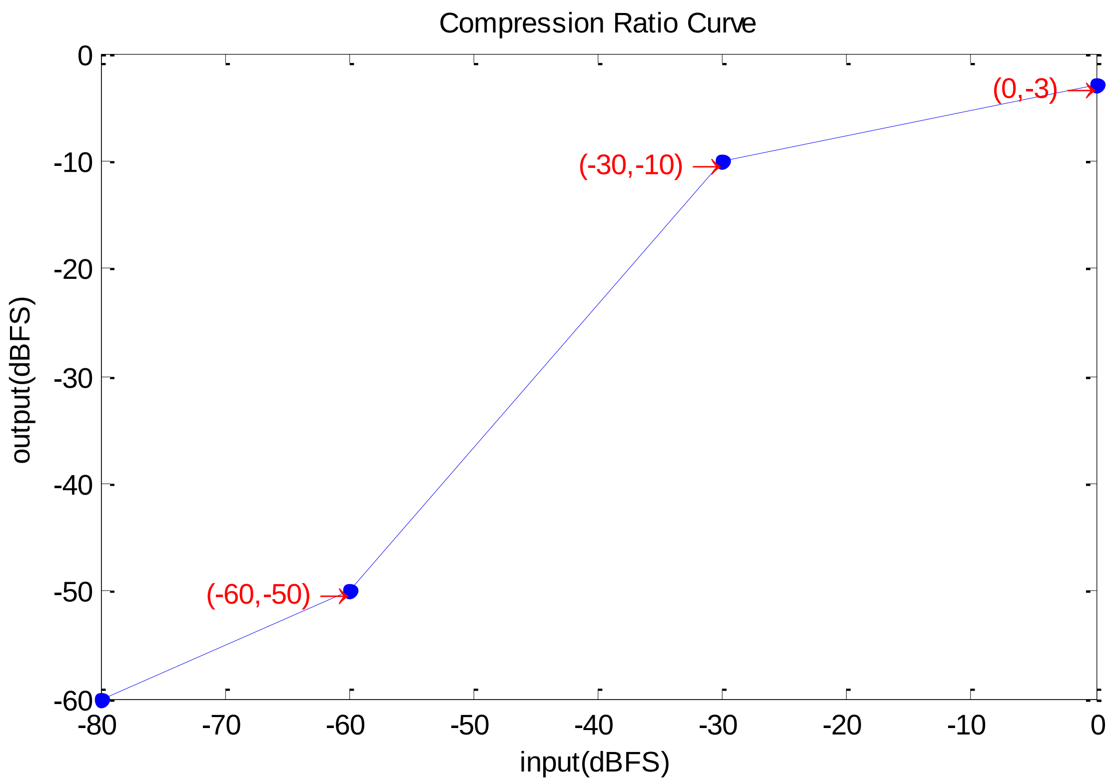

Figure 32: Compression ration curve with 3 different slope

Before setting the compression ratio curve, you must check the input data power level and know how much gain (dB) you want to boost or cutoff or which target power level you want to reach. Besides this, the digital power level doesn't represent the loudness of speaker. The real loudness should be measured by decibel meter.

Suggest that the maximum of compression output is set to -3dBFS for prevent data clipping. If the power level of output data doesn't meet you requirements, you can try boost more gain on low level input power. On the other word, this means increasing the slope of the low power level curve.

For audio input, user should provide our clearly compression ratio curve or how many average energy (dBFS) they want to reach. If need to record wave data in real scenario, please ask user to provide the following information:

1. Obvious distance between speaker and mic (like 1m, 2m, 3m…)
2. Loudness of speaker (normal speech volume about 65-70dBA).
3. Play speech or music, better provide wav file.
   (If user need to play speech, but they don't provide wav file. Please using IEEE_Female_mono_16_kHz or IEEE_269-2010_Male_mono_16_kHz_-18dBFS).

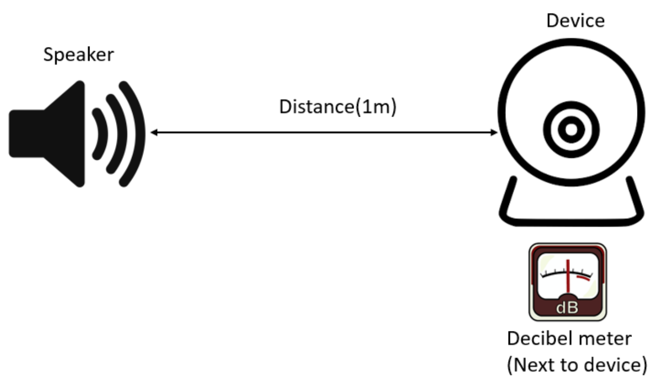

Figure 33: Audio input record schematic diagram

For audio output, user should provide our clearly compression ratio curve or how many dB SPL(measured by decibel meter) they want to reach. Real loudness is about not only power level of digital signal but also hardware device. If the power level of digital data is close to 0dBFS, but speaker still cannot reach the desired volume. Please ask user to change their speaker or modify their loudness spec.

If user ask for data which be measure by decibel meter, please ask user to provide following information:

1. Obvious distance between device and decibel meter (like 1m, 2m, 3m…).
2. Desired loudness of devices.
3. Specified audio file for playing.

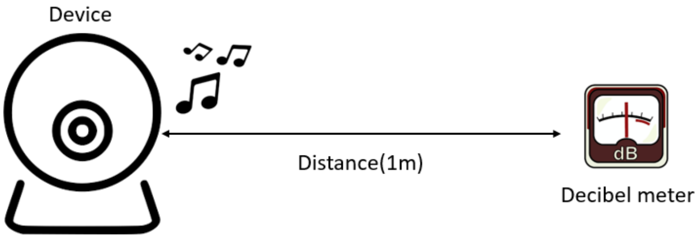

Figure 34: Audio output record schematic diagram

7.2. Target Level

This is the value (dBFS) that sets the maximum allowed output amplitude. After audio row data apply digital gain, our AGC would estimate the maximum whether exceed the target level. If the maximum exceed the target level, the digital gain will be drop down immediately for prevent data clipping. The maximum value of gain dropping will be set by parameter Drop Gain Max

7.3. Noise Gate Threshold

Below this value, AGC will adjust gain until zero to avoid boosting noise signal and clipping of front segment of next voice. If you measure the pure noise signal power level before setting the noise gate threshold, AGC will have better results. The noise gate is preferred over compression ratio curve. No matter how much compression ratio curve is set, the signal will be judged as noise when the power level of input is under the noise gate threshold. Adjustment step is 1dB. Value range: [-80dBFS, 0dBFS]. Recommended value is -60dBFS.

If you don't want adjust gain until zero when power below noise gate, please set noise gate value to -80dBFS. At same time, please set the compression ratio slope of first segment to 1.

For example:

As Figure 35 show, if the power under the noise gate -60dBFS, our AGC will keep gain.

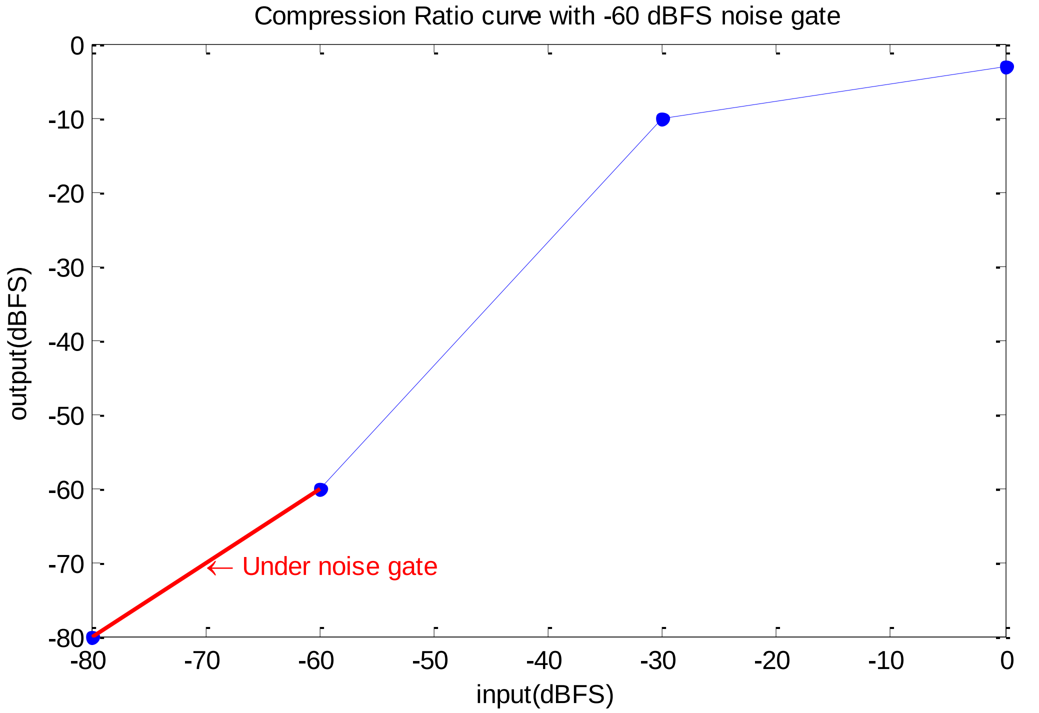

Figure 35: compression ratio curve setting with keeping gain under noise gate

7.4. Attack time

The minimum time between two gain decrements. Please refer to Figure 36 and Figure 37. Adjustment step is 16ms. Recommended value is 1.

7.5. Release time

The minimum time between two gain increments. Please refer to Figure 36 and Figure 37. Adjustment step is 16ms. Recommended value is 10.

P.S. If release time set too short, the gain will increase fast. It may result in the appearance of clipping points.

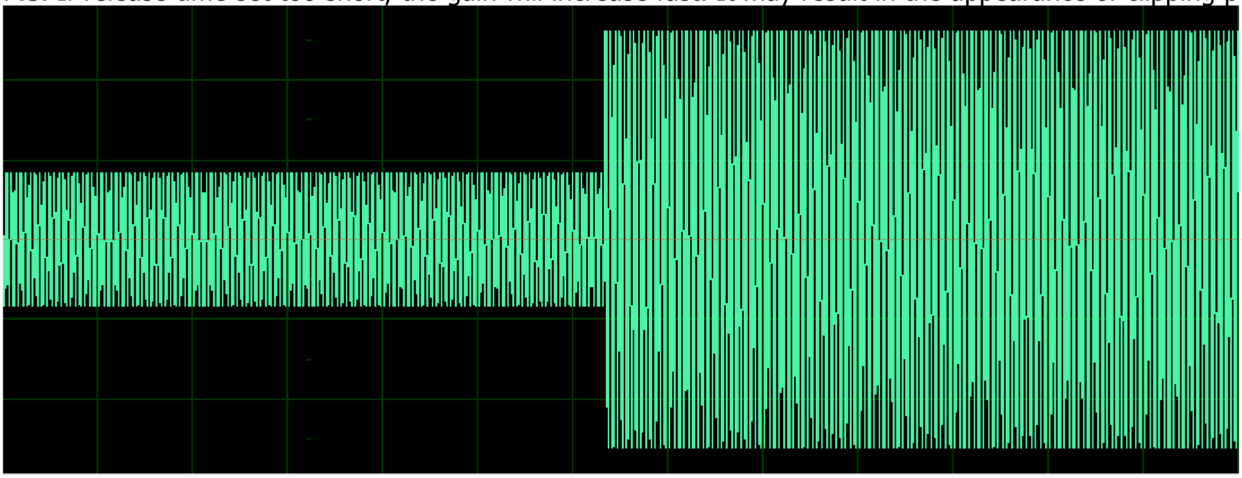

Figure 36: Origin input signal before AGC

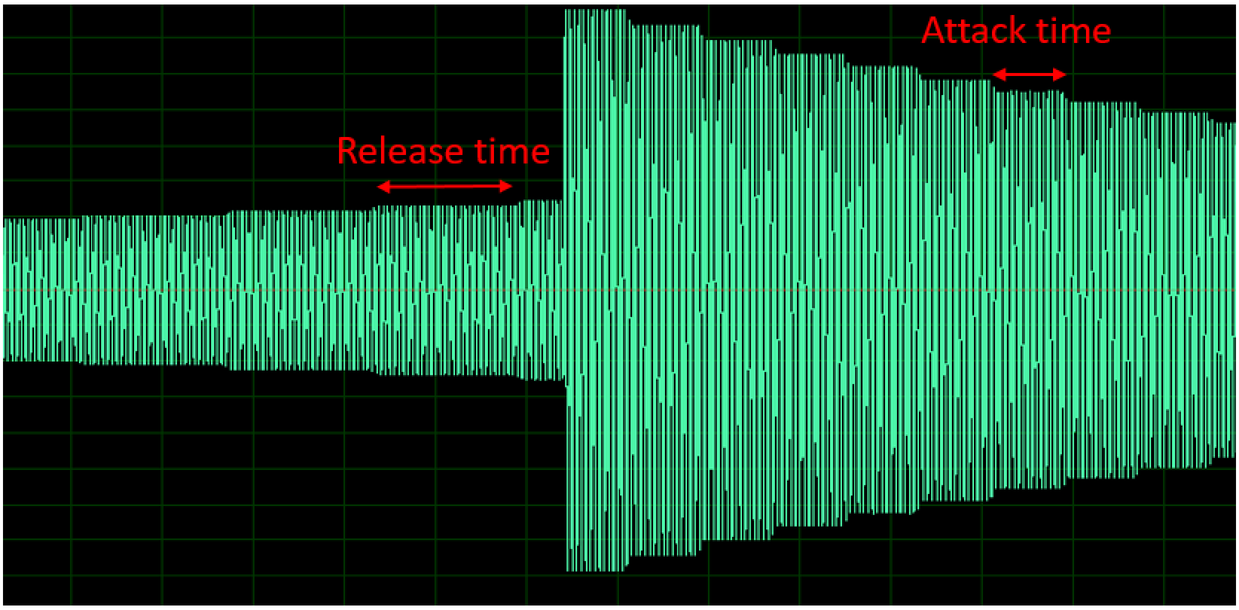

Figure 37: Relation between attack time and release time, output signal after AGC

7.6. Gain info

Define the maximum, minimum and initial gain value of AGC. Adjustment step is 1dB.
Maximum gain: Value range: [0 dB, 60 dB];
Minimum gain: Value range: [-20 dB, 30 dB];
Initial gain: Value range: [-20 dB, 60 dB].

7.7. Drop Gain Max

The maximum of dropping gain to prevent output saturated. Because AGC would smooth data between frame and frame, it doesn't make sure every point no clipping. Suggest that using compression ratio and release time to make sure data no clipping. Please use with Target Level
Adjustment step is 1dB. Value range: [0 dB, 60 dB]. Recommended value is 12 db.
P.S. If the Drop Gain Max value is set too large, the junction between frame and frame may appear obvious pop noise. On the other side if the Drop Gain Max value is set too small, output may have a lot of clipping points. Therefore, this parameter need to be tune in order to meet your requirement

7.8. Noise Gate Attenuation

The percentage of attenuation when input power is under the noise gate. Please refer to Figure 38.
Adjustment step is 1dB. Value range: [0, 100]. Recommended value is 0.
We recommend not to use noise gate attenuation to prevent the output signal sound like interrupted connection. In addition, the tail and start tone of speech will be cut which make speech unnatural.

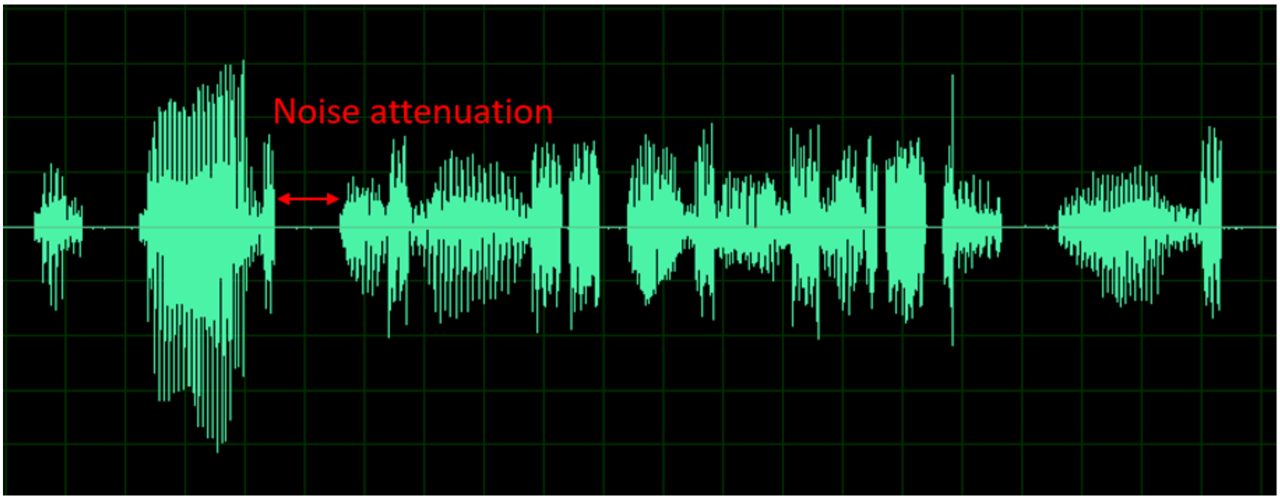

Figure 38: Noise gate attenuation

8. AGC PROBLEM SOLUTION

8.1. Clipping

If the output signal data have too many clipping points, please adjust parameter Drop Gain Max, Compression Ratio Curve and Release Time. Please increase Drop Gain Max or lower the slope of compression curve or increase release time.

The Figure 39 below represents the result of different drop gain max vale. The above one is the result of drop gain max 6dB, and the following is the result of drop gain max 12dB. Drop gain max measurement is too small, resulting in insufficient gain reduction.

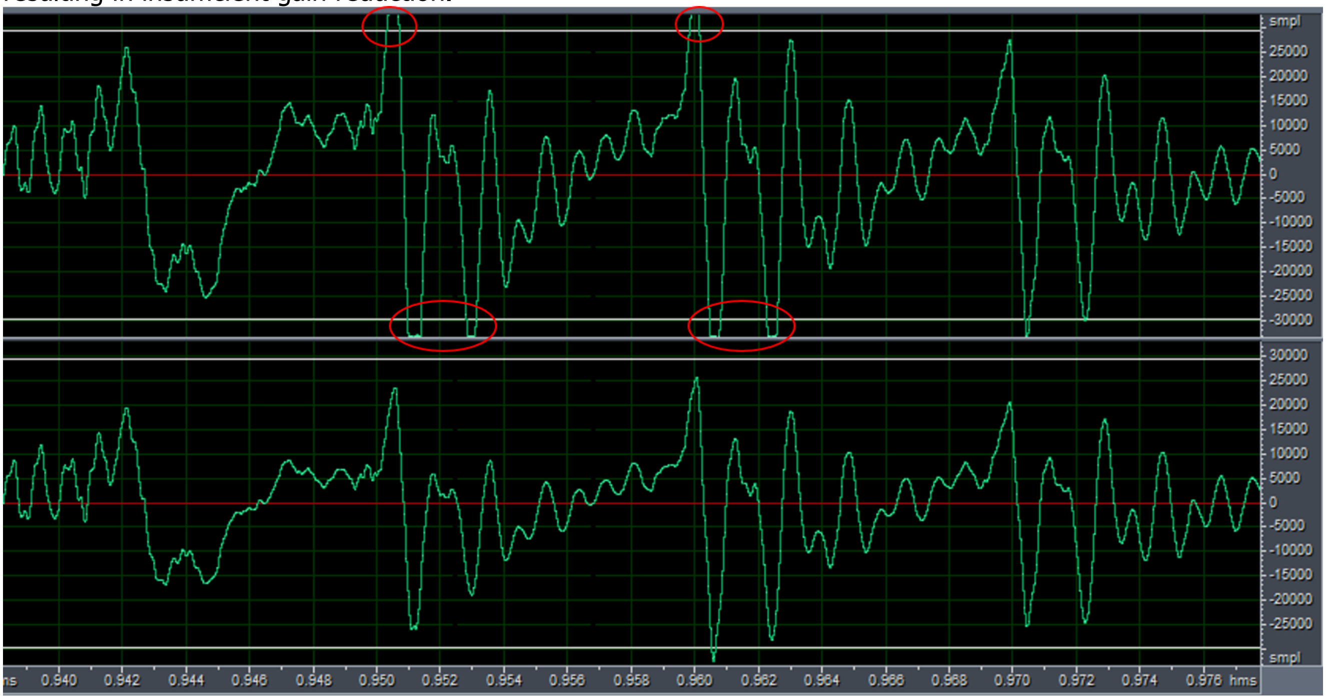

Figure 39: Clipping status with different drop gain value

The Figure 40 represents the result of different compression ratio curve. The above one boost less gain in range from -48dBFS to -25dBFS. It could avoid suddenly increasing of volume.

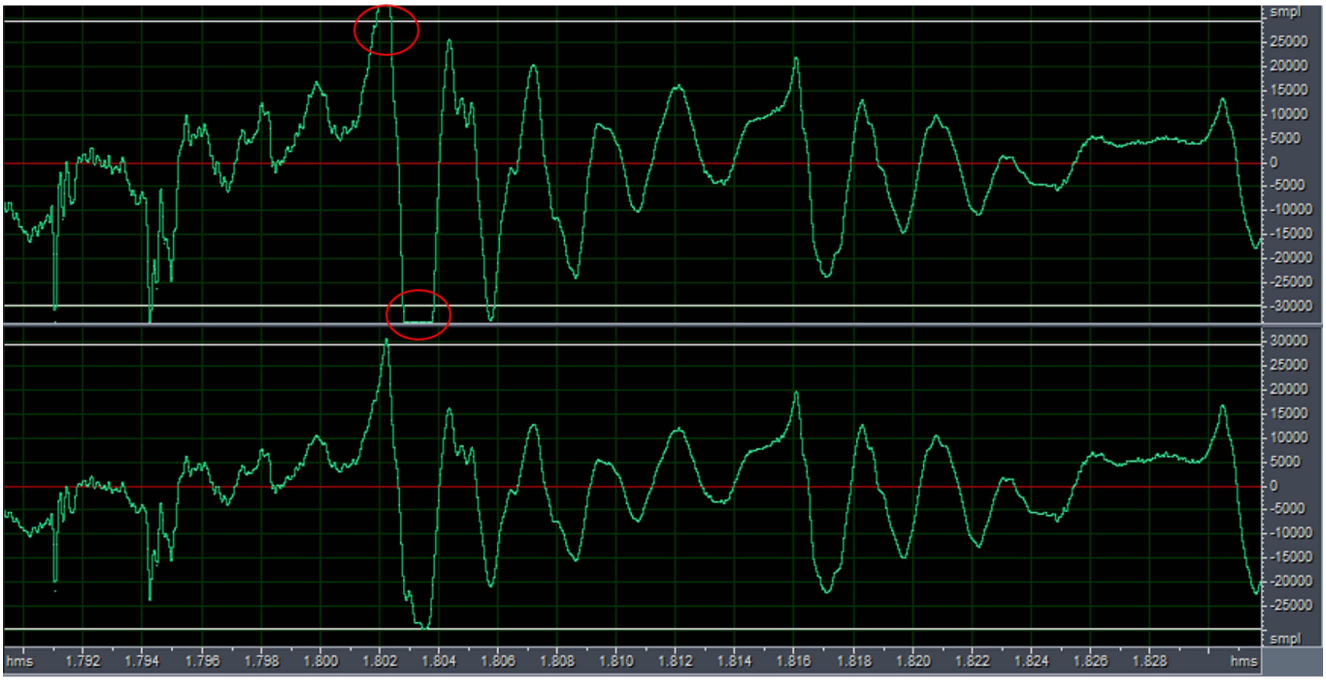

Figure 40: Clipping status with different compression ratio curve

The Figure 41 represents the result of different release time. The appropriate release time is able to effectively reduce the clipping points. However, if the release time is too long, it will result in AGC fail.

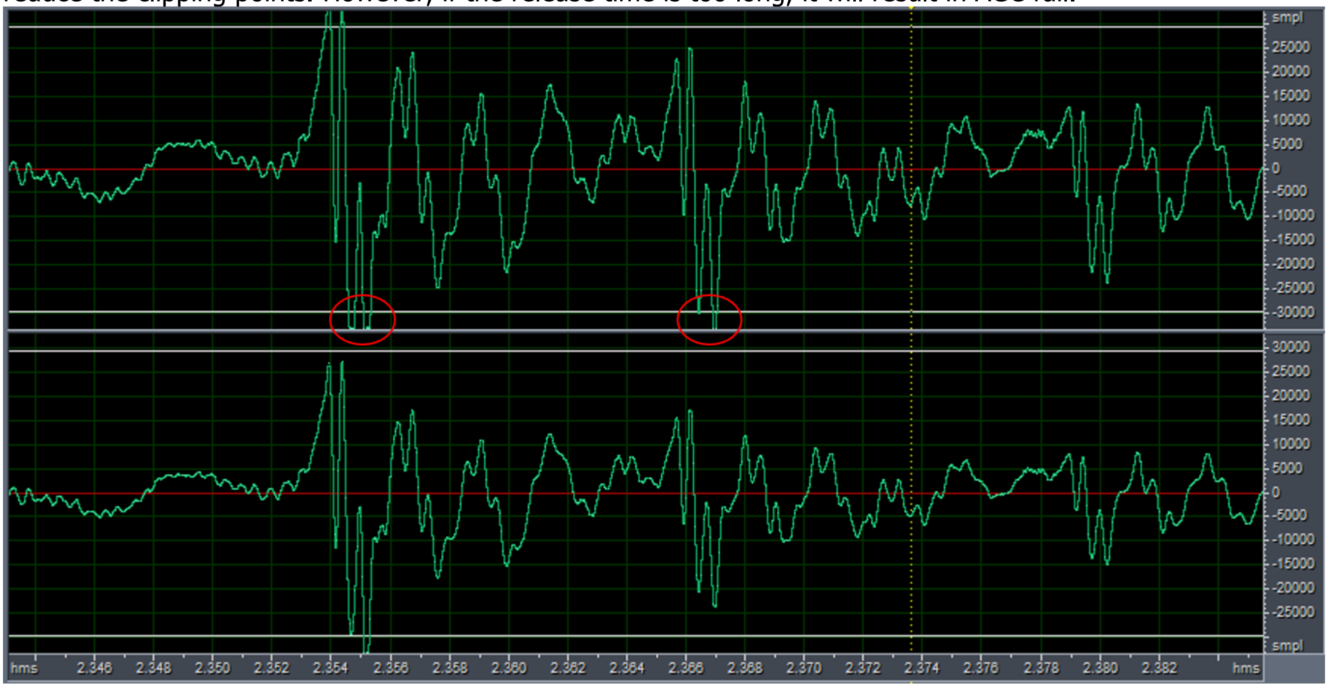

Figure 41: Clipping status with different release time

8.2. AGC does not work

If the output signal does not apply any gain after 0.5 sec, please check Gain info, release time, attack time and compression ratio curve

If the check gain info is not properly set, it will seriously effect AGC apply gain. For safety reasons, please set gain info in larger range.

If noise gate is set too high, the input signal continues below the noise gate. AGC will not apply gain on output

signal. Please check speech power level whether is under the noise gate you set.
If you want to attenuate the signal below the noise gate by percentage, please adjust the configuration, see Section 7.8. But this parameter might cause unnatural voice at the end and starting point. Therefore, when you set this parameter, the noise gate value should be concerned carefully.
If you all saturation above, please dump the input files, output file sent them to SigmaStar FAE.

8.3. Sound too small after AGC

If output average power level doesn't reach the target dBFS you need, please check the compression ratio curve setting, noise gate and release time. Whether release time be set too long or compression ratio curve be set too gentle

9. NR TERMS AND DEFINITION

9.1. NR Mode

NR mode.
Value range: [0, 1, 2]. 0: Default mode for speech; 1: User setting mode; 2: Music mode.

9.2. NR Intensity

NR strength. Intensity 30 is the highest level of noise suppression, but more details losses/damage.
Value range: [0, 30]. Adjustment step is 1. Recommended value is 20.
For NR result of different intensity, please refer to Figure 42 and Figure 43.

9.3. NR Smooth Level

The signal smooth level; the larger value indicates signal smoother.
Value range: [0, 10]. Adjustment step is 1. Recommended value is 10.

9.4. NR Converge Speed

NR speed of noise adaptation; the larger values make NR detect noise faster, but more details losses/damage.
Value range: [0, 1, 2]. Recommended value is 1.

10. NR AND AGC DEFAULT PARAMETER CONFIGURATIONS

10.1. Default Parameter Table

Table 1: NR and AGC default parameter table

Module	Parameter	Scenario (Mode)	
		speech mode	music mode
NR	Intensity	20	10
	Smooth Level	10	10
	Converge Speed	1	1
AGC	Target Level	-20	-20
	Attack Time	1	1
	Release Time	10	5
	Noise Gate	-55	-50
	Compression Ratio	2	4
	Noise Gate Attenuation	0	0
	Maximum Gain	15	20
	Minimum Gain	-5	-10
	Initial Gain	0	0

11. NR RESULT WAVE

11.1. NR result in different intensity (15 and 30)

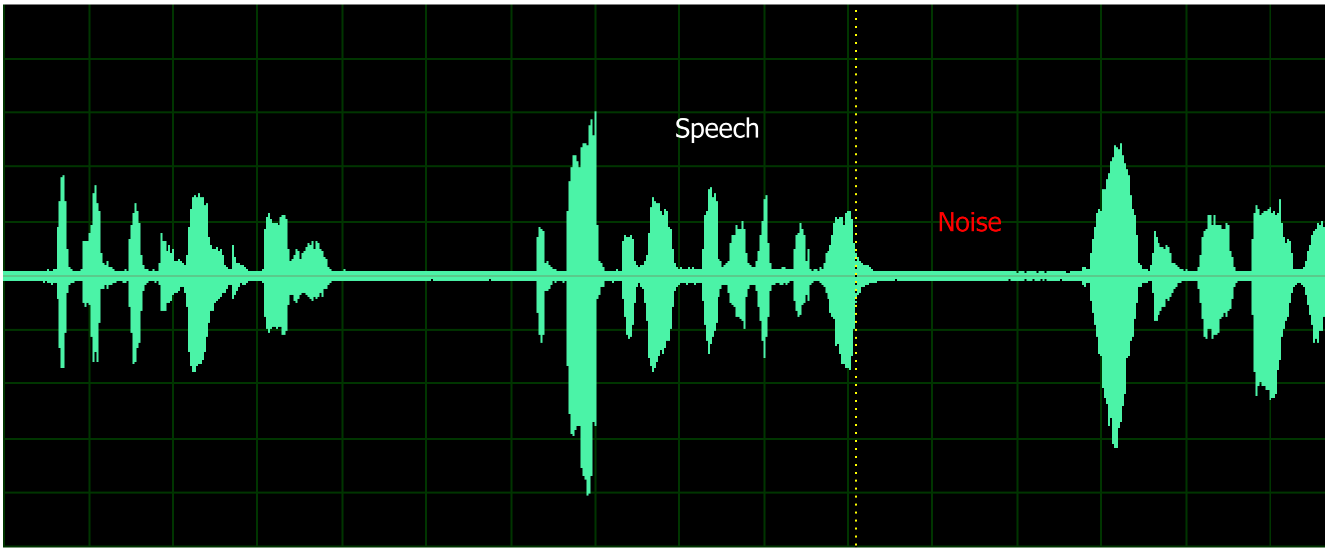

Figure 42: NR result (Intensity: 15)

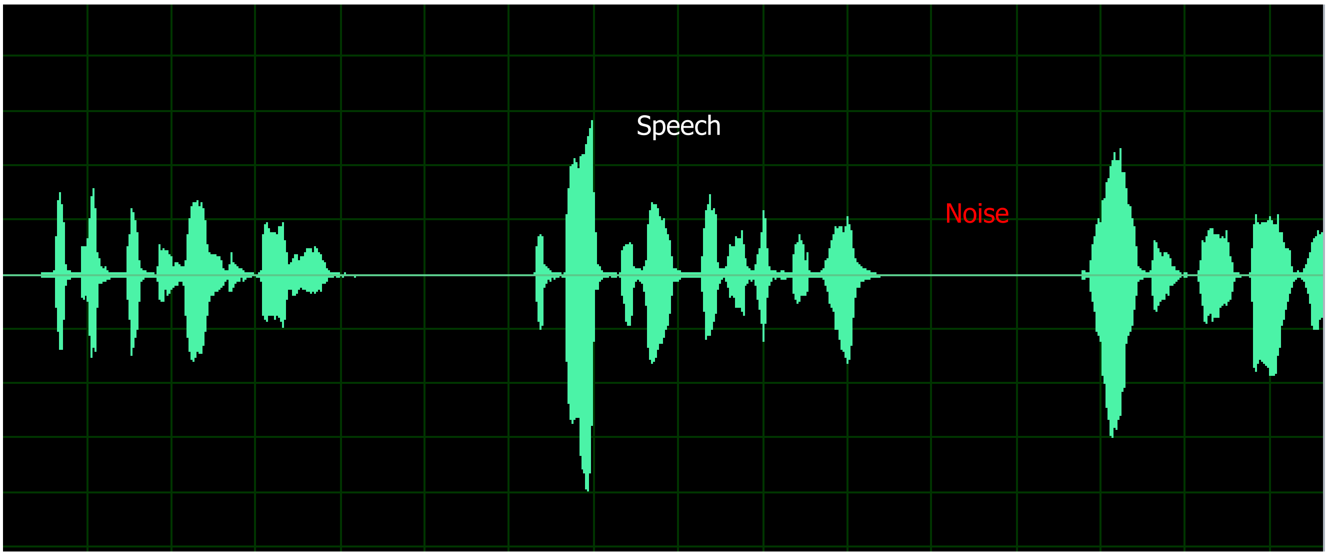

Figure 43: NR result (Intensity: 30)

12. EQ PAREMETER SETTING

Please setting the Array: eq_gain_db[_EQ_BAND_NUM]
This is the Gain adjustment for the EQ frequency bands.
Divide the frequency into 129 equal parts. Every value present

8k/129=62.015 Hz

The 8 kHz frequency band is valid only when the working sampling rate (s32WorkSampleRate) is set to 16 kHz. The value range of each frequency band is [–50, +20] dB.

For example:
We need to decrease -10DB from 1560 Hz to 3000 Hz

Setting the eq_gain_db array index,($\mathrm{floor}(\frac{1560}{62.015})$ to $\mathrm{ceil}(\frac{3000}{62.015})$) as -10 DB