Application Report

SLAA894–May 2019

General Tuning Guide for TAS58xx Family

Alix Wan

ABSTRACT

This application note aims to introduce how to perform tuning the TAS58xx family process flow. A detailed introduction of various process flow coefficient parameters in the flow and a referenced tuning process is provided. In the recommended process flow, software preparation and an overview of the tuning resource in the DSP are introduced. Then, the gain structure and DSP headroom are introduced to obtain expected maximum output power and avoid digital clipper before the tuning. Finally, structures and tuning suggestions of common processing blocks including EQ, DRC, AGL, and clipper, are discussed in detail.

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Trademarks

PurePath is a trademark of Texas Instruments.
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1 Tuning Preparation

1.1 Tuning Setup

Figure 1 shows a PurePath™ Console 3 platform along with the EVM board, which can setup a tuning environment quickly.

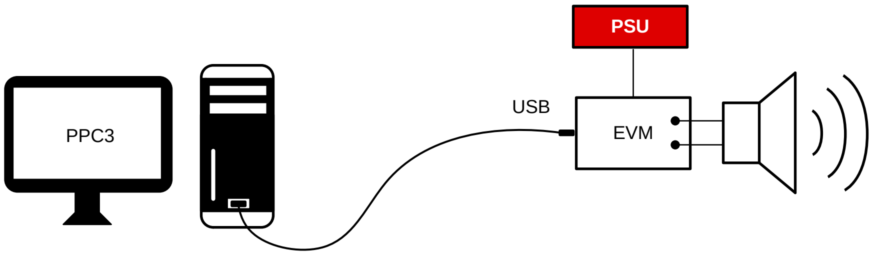

Figure 1. Tuning Setup

After the setup from Figure 1 is ready, most tuning work is performed on TI's PPC3 platform. A quick start-up PurePath™ Console (PPC3) Overview training video is provided. For the TAS5805M, all tuning resources available can be obtained from the *TAS5805M, TAS5806M, and TAS5806MD Process Flows Application Report* (SLOA263). Different process flows can be chosen based on requirements. All are available in the PPC3 platform.

Table 1. Processing Features Comparison Table

FEATURE	PROCESS FLOW 1 (3-Band DRC, 96 kHz, 2.0)	PROCESS FLOW 2 (3-Band DRC & FIR, 48 kHz, 2.0)	PROCESS FLOW 3 (3-Band DRC & FIR, 48 kHz, 2.1)
Maximum Internal Sample Rate	96 kHz	48 kHz	48 kHz
SRC and Auto-detect	Yes	No	No
Supported Input Sample Rates (32 k, 44.1 k, 48 k, 88.2 k, and 96 k)	Yes	88.2 k and 96 k are not supported.	88.2 k and 96 k are not supported.
Biquads for EQ Filtering (Individual Left / Right)	15	15	15
Input Mixer	Yes	Yes	Yes
Click & Pop Free Volume	Yes	Yes	Yes
DRC	3-Band 4" Order Crossover	3-Band 4" Order Crossover	3-Band 4" Order Crossover and 1-Band
Automatic Gain Limiter	Yes	Yes	Yes
Output Clipper	Yes	Yes	Yes
FIR Filter	No	Yes	No
Hybrid PWM Mode	Yes	Yes	Yes

Usually, a specific process flow can be chosen based on the speaker mode, input sampling rate, and required features. Once the process flow is chosen in PPC3, tuning is performed with the GUI in Figure 2.

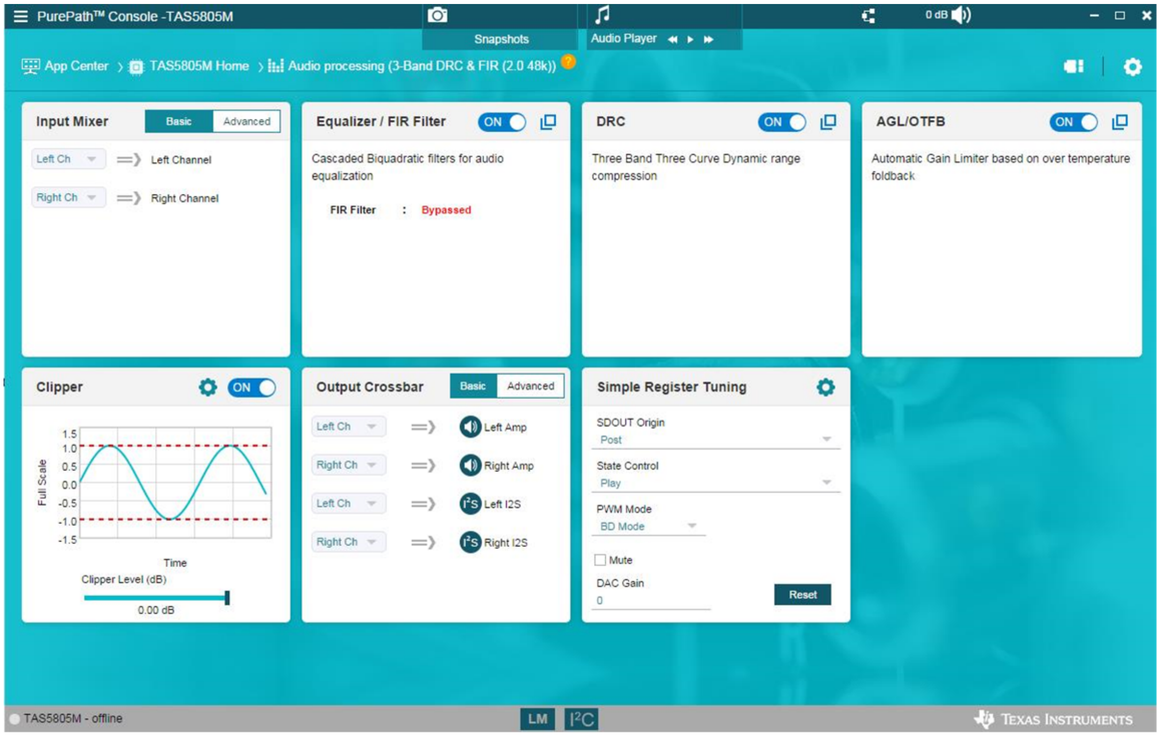

Figure 2. PPC3 Tuning GUI

As shown in Figure 2, this process flow consists of several processing blocks:

- Input mixer
- EQ/FIR
- DRC
- AGL
- Clipper
- Output cross-bar

In fact, all process flows provided in Table 1 are obtained with different configurations of the signal path shown in Figure 3.

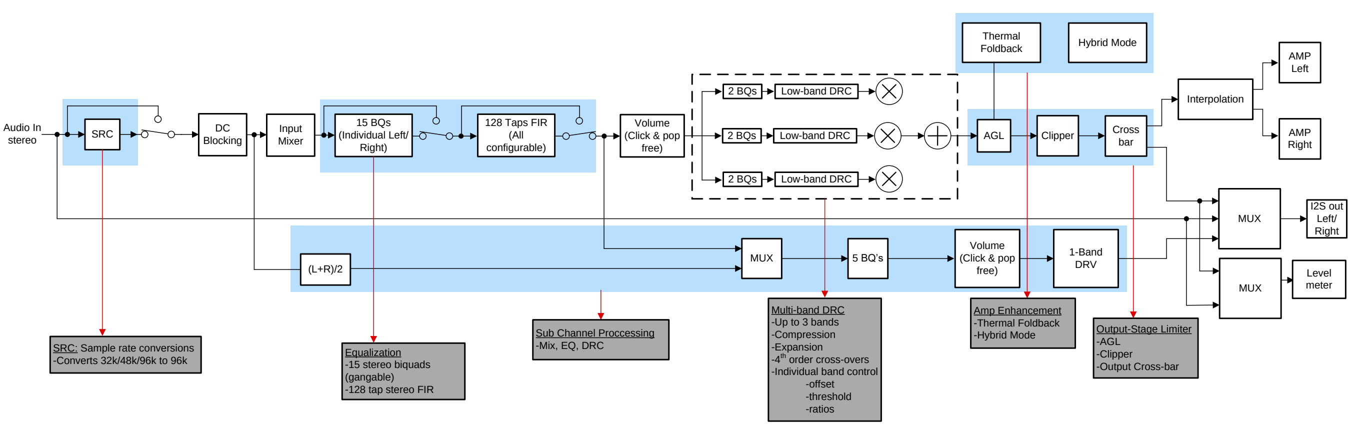

Figure 3. Signal Path of Process Flow

The main signal path includes the following:

- SRC
- DC blocking
- Input mixer
- BQ
- FIR
- Volume
- DRC
- AGL
- Crossbar

Features of these blocks may vary based on the chosen process flow. For the branch signal path of (L+R) / 2, it is usually used to implement 1.1 or 2.1 mode. For a more detailed introduction, see the *TAS5805M, TAS5806M, and TAS5806MD Process Flows Application Report*.

1.2 Gain Structure

As a digital input amplifier, gain structure of the TAS58xx family consists of three main parts:

1. Volume: -110 dB ~ 24 dB
2. DAC gain: -103 dB ~ +24 dB
3. Analog gain: -15.5 dB ~ 0 dB

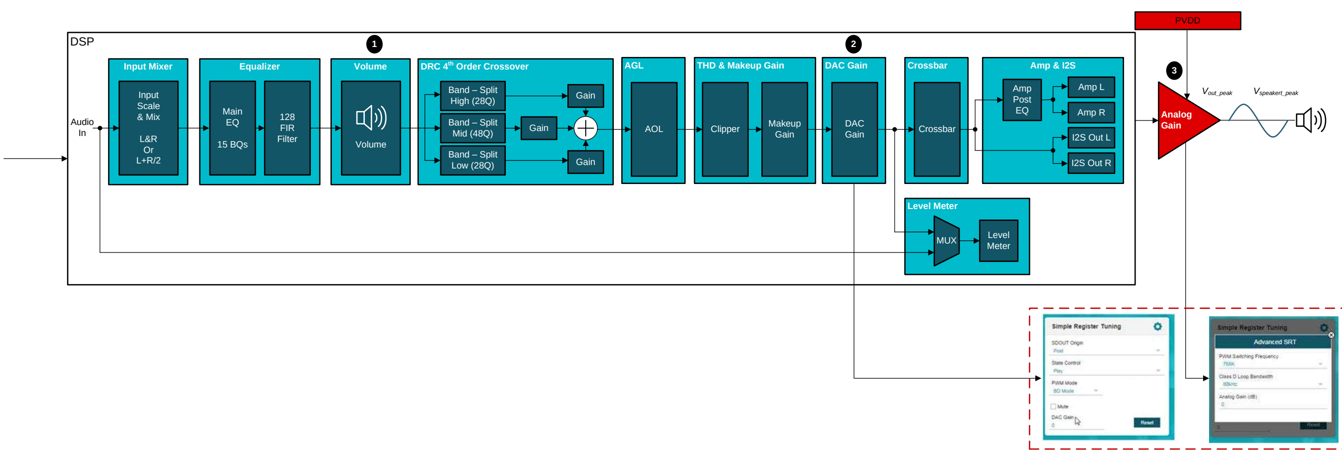

Figure 4. Gain Structure of the TAS58xx Family

Generally, system gain equals digital gain multiplied by analog gain. For digital gain, maximum output of the digital domain is limited to -0.5 dB (the 0.5 dB is the margin for avoiding digital overflow). Full scale (0 dB) analog gain output peak voltage is around 29.5 V. Hence, when digital output is -0.5 dB and analog gain is configured as 0 dB, the output voltage is ~29.5 Vp in theory. Of course, this output peak voltage is limited by PVDD too, which means clipping occurs when output peak voltage in theory exceeds PVDD. For example, if PVDD = 12 V, different gain settings lead to different output as in Table 2.

Table 2. Gain Setting Example

VOLUME	ANALOG GAIN	MAXIMUM Vout (PEAK)	PVDD CLIP OR NOT
0 dB	0 dB	29.5 V	Y (14.75 V > PVDD)
0 dB	-6 dB	14.75 V	Y (14.75 V > PVDD)
-6 dB	-6 dB	7.38 V	N (7.38 V < PVDD)
0 dB	-12 dB	7.38 V	N (7.38 V < PVDD)

Further, the class D output circuit can be simplified to the architecture in Figure 5.

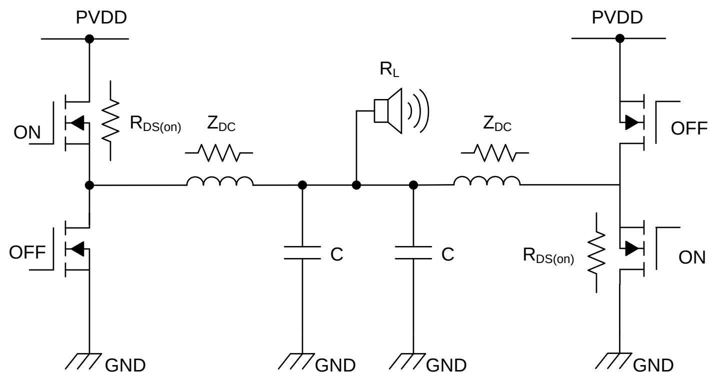

Figure 5. Schematic of Equivalent Output

Voltage on load needs to be calculated with Equation 1.

$$V_{speaker\_peak} \sim = V_{out\_peak} \times R_L / (R_L + 2 \times (R_{DS(on)} + Z_{DC})) \quad (1)$$

where

- $V_{speaker\_peak}$ is the peak voltage on speaker
- Z_{DC} is the impedance of inductor and trace in sum
- $R_{DS(on)}$ is the drain-to-source on resistance of the individual output MOSFETs

For example, if PVDD = 12 V and the load of speaker = 6 Ω, peak voltage clips at around 11.6 V rather than 12 V.

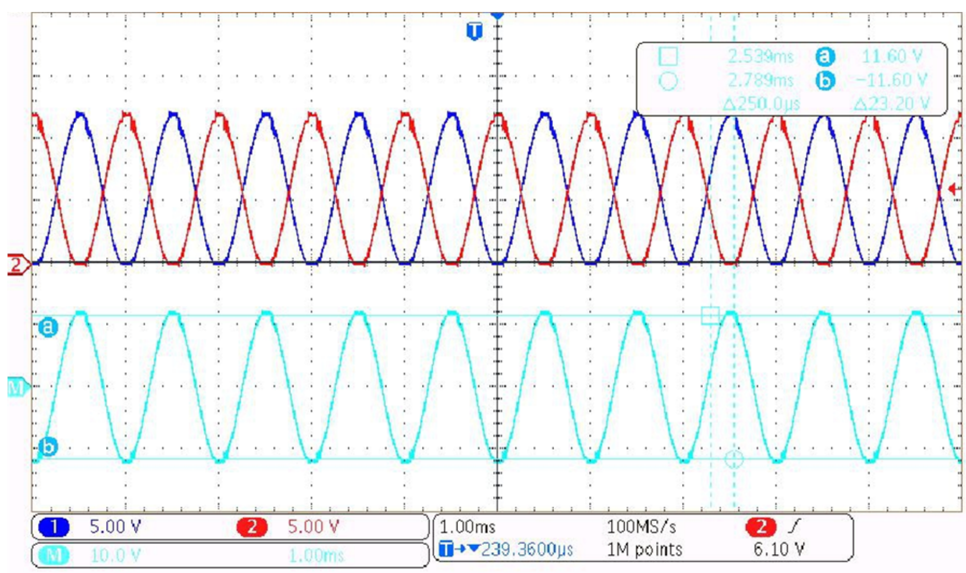

Figure 6. Output Clip

The red curve is the output between the two speaker terminals, which is calculated by (Ch1 - Ch2) as the output is differential. Mostly importantly, considering variations of analog circuitry and the inaccuracy of analog gain, output voltage in practice on load needs to be measured with Audio Precision or an oscilloscope.

2 Headroom of Digital Processing

2.1 DSP Headroom

Tuning is implemented in the DSP of the TAS5805M and all arithmetic is done in the digital domain. In order to avoid digital calculation overflow, headroom is needed to be taken into consideration. Coefficient or audio data in the processing block is usually formatted as 1.31, 5.27, 9.23, whose structure is shown in Figure 7.

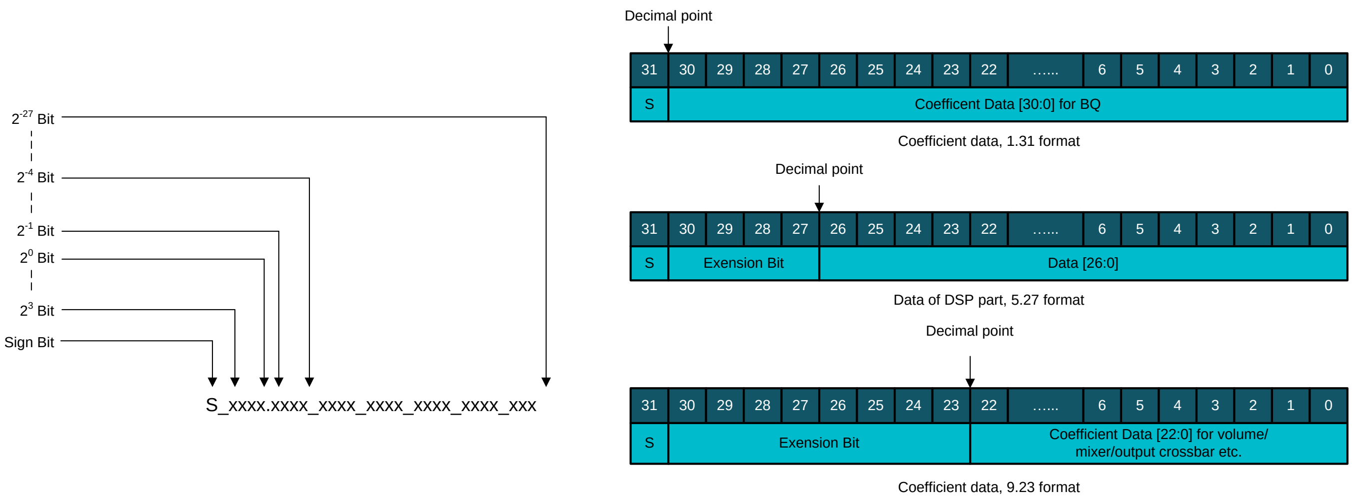

Figure 7. Data and Coefficient Format

Taking volume as an example, a 9.23 format indicates the maximum gain is:

$$\text{Max Gain} = 20 \times \log M\,(2^{(9-1)}) = 48\ \text{dB} \quad (2)$$

Figure 8 shows a data format in the process flow of the TAS5805M for a reference.

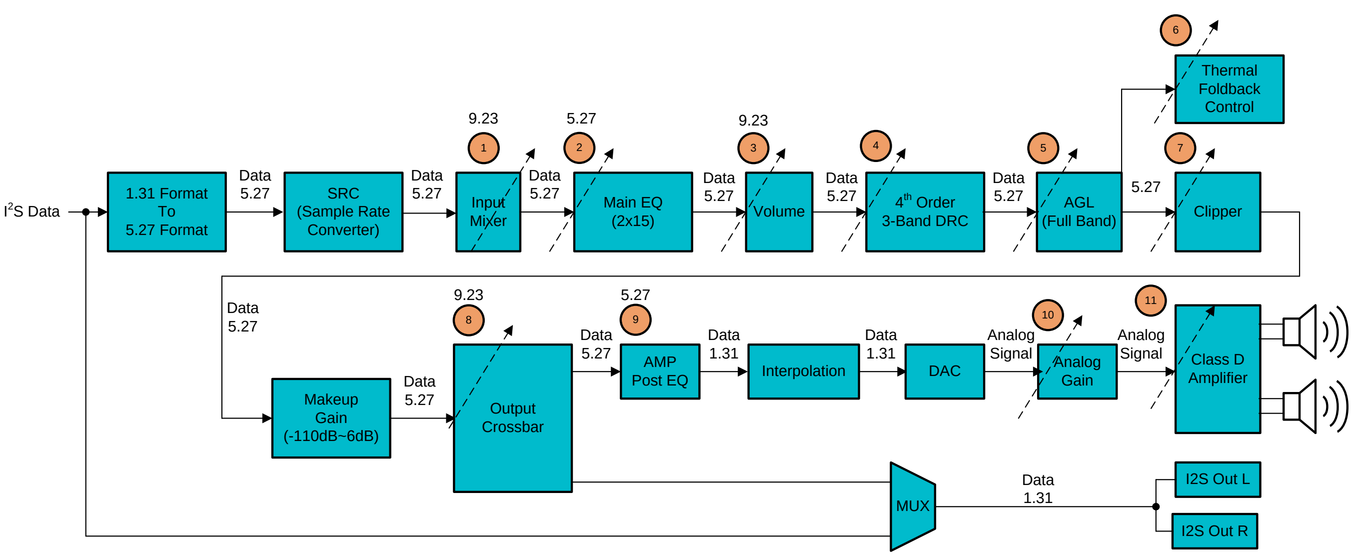

Figure 8. Data Format in the Process Flow

One common challenge is avoiding data overflow during tuning. Three main principles are needed to avoid this problem:

1. The output of digital part of the TAS5805M cannot exceed -0.5 dB to avoid digital clipping.
2. For each block, total digital gain should be less than 24 dB in default (24 dB is the headroom of the gain).

3. The maximum gain of every processing block should be limited by their data format.

For example, assuming that there are n blocks in the processing path including the input mixer, EQ, DRC, and so forth. Their gain is G_i (i = 1, 2, … , n). The level of input signal is Sin.

$$S_{in} \times G_1 \times G_2 \times \ldots \times G_n \leq -0.5 \text{ dB} \tag{3}$$

$$G_1 \times \ldots \times G_n \leq 24 \text{ dB} \tag{4}$$

$$G_i \leq G_{i\_max} \text{ (determined by the data format)} \tag{5}$$

For example, the input mixer of the TAS5805M is 9.23 format, hence maximum gain is $G_{1\_max} = 20 \times \log(2^8) \approx 48$ dB. Similarly, a single Biquad G_2 (5.27) ≤ 24 dB and the volume G_3 (9.23) ≤ 48 dB.

2.2 Headroom Measurement and Tuning Suggestions

As shown in Figure 9, all blocks are in serial. With the audio signal in, digital clipping may occur at the output of each process block. Combined with Figure 9, the engineer needs to consider the gain of each block to avoid digital clipping. In order to better understand the headroom, use Table 3.

Table 3. DSP Headroom Calculation

NO.	INPUT	INPUT MIXER	EQ	VOLUME	CLIPPER	DAC GAIN	ANALOG GAIN/MAXIMUM Vout (PEAK)	Vout (PEAK)/24 V
1	0 dB	0 dB	0 dB	24 dB	0 dB	-30 dB	0 dB/29.5 V	15.2 V/24 V
2	0 dB	0 dB	0 dB	27 dB	0 dB	-33 dB	0 dB/29.5 V	10.8 V/24 V
3	0 dB	27 dB	0 dB	0 dB	0 dB	-33 dB	0 dB/29.5 V	10.8 V/24 V
4	0 dB	0 dB	27 dB	3 dB	0 dB	-33 dB	0 dB/29.5 V	10.8 V/24 V

In order to check whether or not volume has 24 dB headroom, line 1 and line 2 are checked to resemble Figure 9 and Figure 10.

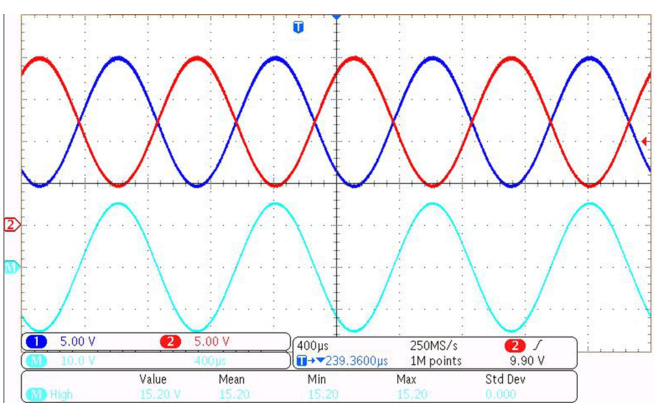

Figure 9. Result of Line 1

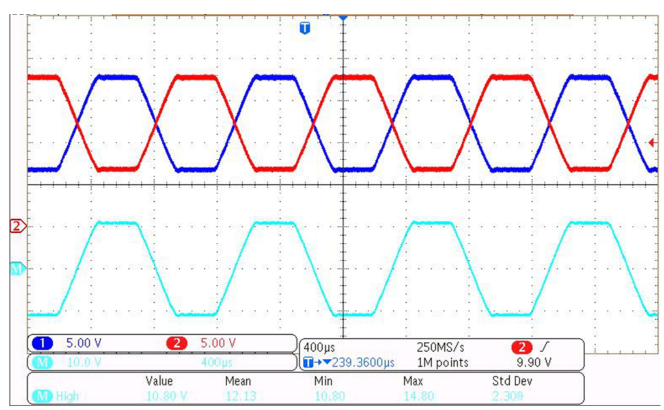

Figure 10. Result of Line 2/3/4

If headroom is not sufficient for the application, negative gain in certain blocks can be used to increase headroom. This is better for dynamic range. Usually, the input mixer can be set to achieve more headroom, then gain compensation can be obtained in the output cross bar as shown in Figure 11.

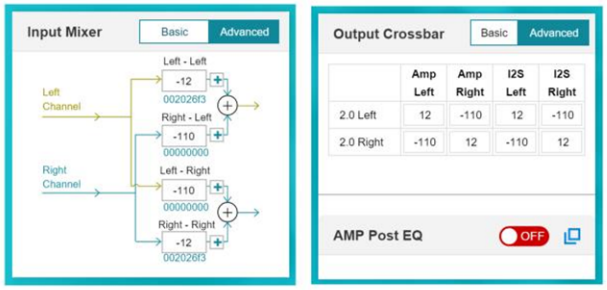

Figure 11. Increase Headroom

3 Recommended Tuning Flow

In general, acoustic engineers need to achieve the desired tuning style while meeting specific technical specifications. These specifications include maximum output power, enhanced bass, and good THD+N. At the same time, they need to ensure that the speaker does not get damaged due to excessive output power. Tuning can be roughly divided into two parts: achieve expected sound effect and ensure there is no sound effect distortion. The following processes are recommended by TI for a basic reference, assuming that the specification of a certain application are like Table 4.

Table 4. Design Specification

SPEC	VALUE
Power Supply / PVDD	12 V
Impendence / R_load	4 Ω
Output power / P_(Out_Max)	2 × 8 W

The following tuning procedure is recommended:

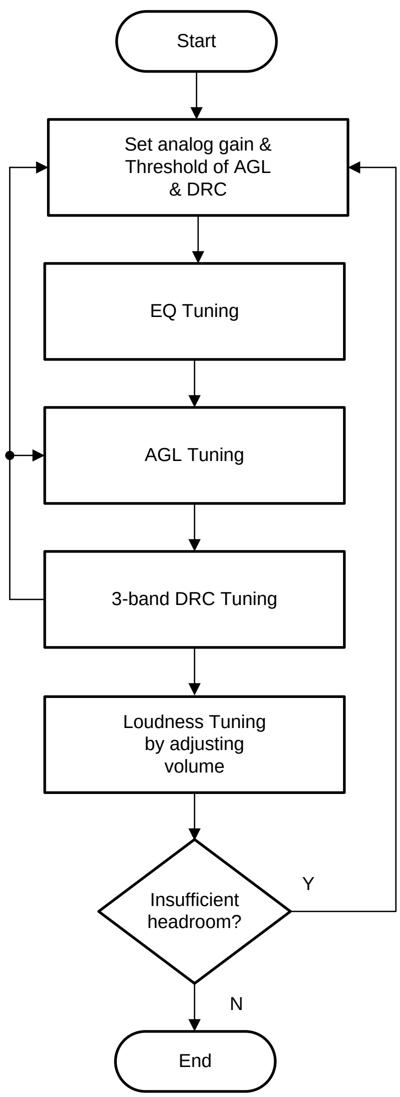

Figure 12. Overview of Tuning Process

The tuning process can be divided into the following steps:

1. In order to achieve expected maximum output power, analog gain and the threshold of AGL and DRC should be set first. In this step, digital clipping and PVDD clipping are needed to be taken into consideration. Further, dynamic range is needed to be noted.
2. To achieve the expected sound effect, it is suggested to tune the EQ with a small audio source without triggering any suppression or clipping.
3. AGL can suppress the full audio band once the input level of any frequency point exceeds the threshold. It is faster than DRC. If the threshold needs to be adjusted, turn to step 1.
4. 3-band DRC can configure the three audio bands separately. If AGL is needed, turn to step 3.
5. Obtain expected loudness by adjusting the volume in the digital domain.
6. If adjusting the volume leads to digital clipping, turn to step 1 to redesign the gain structure.

3.1 Analog Gain Setting

It is recommended to set the analog gain first to meet required maximum output power. Figure 13 shows this basic idea.

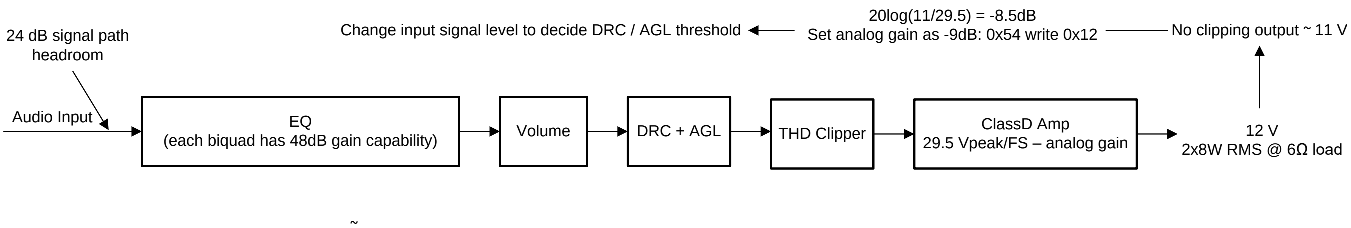

Figure 13. Recommended Analog Gain Setting

As discussed previously, analog gain can determine maximum output peak voltage. In this application, PVDD = 12 V, hence:

$$V_{speaker\_max} \approx PVDD \times R_L / (R_L + 2 \times (R_{DS(on)} + Z_{DC})) = 12\text{ V} \times 4 / (4 + 2 \times (0.18 + 0.023)) = 10.89\text{ V} \quad (6)$$

Maximum output peak voltage is ~10.89 V whose accurate data is needed by measuring experimentally. Analog gain and threshold of AGL and DRC can be obtained by Table 5.

Table 5. Gain Setting

NO.	SPECIFICATION	CALCULATION	COMMENTS
1	Maximum peak voltage on speaker	10.89 V	Maximum P_{Out} = 14.8 W.
2	Output peak voltage at P_{Out_Max} = 2 × 8 W	8 V	No clipping at 2 × 8 W at 4 Ω
3	Choose no clipping output peak voltage.	10.89 V	Range = [8 V, 10.89 V]. Choose 10.89 V for maximum dynamic range.
4	Calculate analog gain at step 3.	-9 dB	20 × log(10.89 V / 29.5 V) = -8.7 dB, set -9 dB.
5	Calculate total gain at 2 × 8 W.	-11.34 dB	20 × log(8 V / 29.5 V) = -11.34 dB
6	Obtain AGL/DRC threshold.	-2.5 dB	-11.34-(-9) = -2.3 dB. Choose -2 dB or -2.5 dB.

For step 3, analog gain changes based off the chosen voltage from 8 V to 10.89 V. In order to achieve maximum dynamic range, it is suggested to choose 10.89 V here, but it may be that the maximum output peak voltage reaches 10.89 V, which may need to sink a maximum current from PVDD. Smaller no clipping output voltage can be chosen to avoid a large current spike but it decreases the sound dynamic range. Further, considering the smallest step of gain and the threshold of AGL and DRC are 0.5 dB. -8.7 dB is needed to be rounded to -9 dB. All these variations need measurement and adjustment in actual application.

3.2 EQ Setting

Usually, the tuning of EQ is based on SPL curve of speaker.

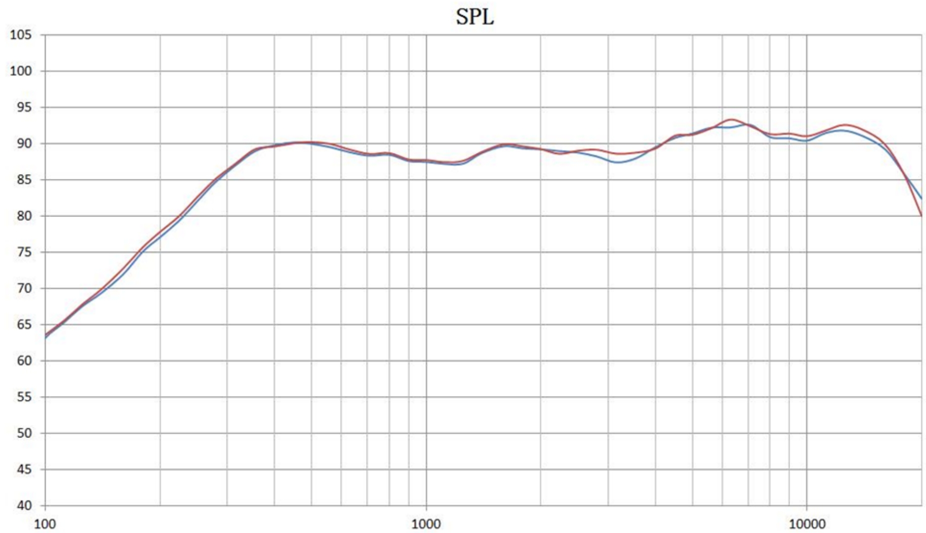

Figure 14. SPL Curve of the Speaker

The SPL curve can be a reference for EQ. A common strategy is to use EQ to flatten the SPL.

Figure 15. EQ Setting

In order to not trigger any suppression or clipping, TI recommends setting input level to -20 dBFS or smaller to achieve the expected sound effect. Each of the 15 BQs has a maximum 20 dB gain, with each connected in serial. In order to avoid digital clipping in EQ, it is suggested to set gain suitably with a limitation of 24 dB headroom for any output of the 15 BQs.

3.3 AGL Setting

The Automatic Gain Limiter (AGL) is a feedback mechanism that can be used to automatically limit the amplitude to the threshold in the full audio band. Audio signal level is sensed sample by sample and compared with the threshold, which usually makes it faster than DRC. A boosted gain or decreased gain is calculated and sent to the alpha filter at the output of the AGL for smoothing the adjusted the gain. Figure 16 shows the PPC3 GUI for AGL.

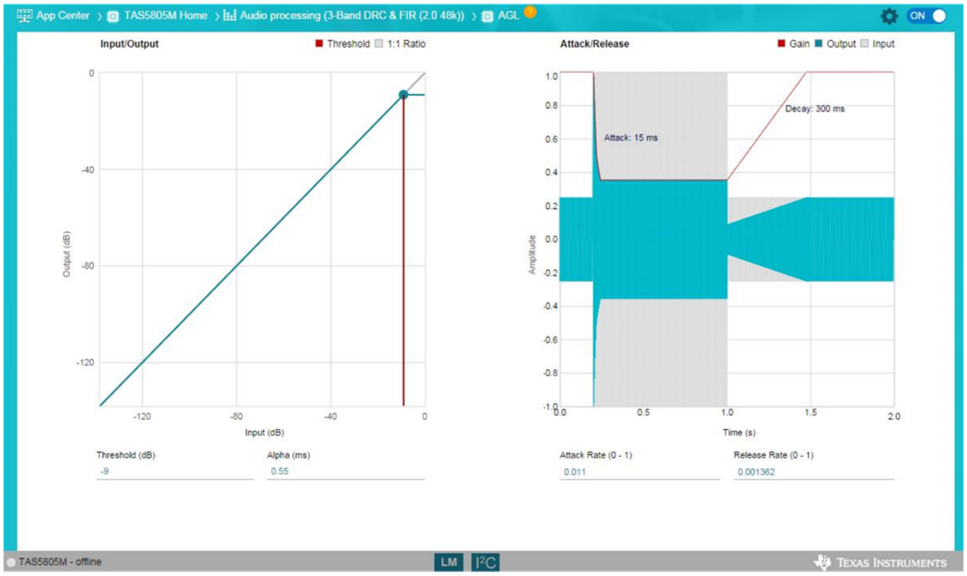

Figure 16. AGL Tuning GUI

The configuration of AGL consists of two parts: input/output and attack/release.

3.3.1 Threshold and Alpha Time

The threshold of AGL determines the limitation level of output, which can be calculated by referring to Table 5. The change of threshold leads to the change of attack/decay time. Alpha time is for the alpha filter and is a non-linear structure. It is suggested to use the PPC3 GUI to obtain attack time considering calculation accuracy limitation. More details are discussed in the following sections. Usually it is suggested to keep Alpha time in default.

3.3.2 Attack Rate and Release Rate

As shown in the PPC3 GUI, attack time and decay time are determined by attack rate and release rate. Attack time indicates the time expense for AGL to limit the output voltage to the threshold. Similarly, decay time shows the recovery time when the input audio level is below the threshold. Figure 17 shows the definitions of the attack and release rate.

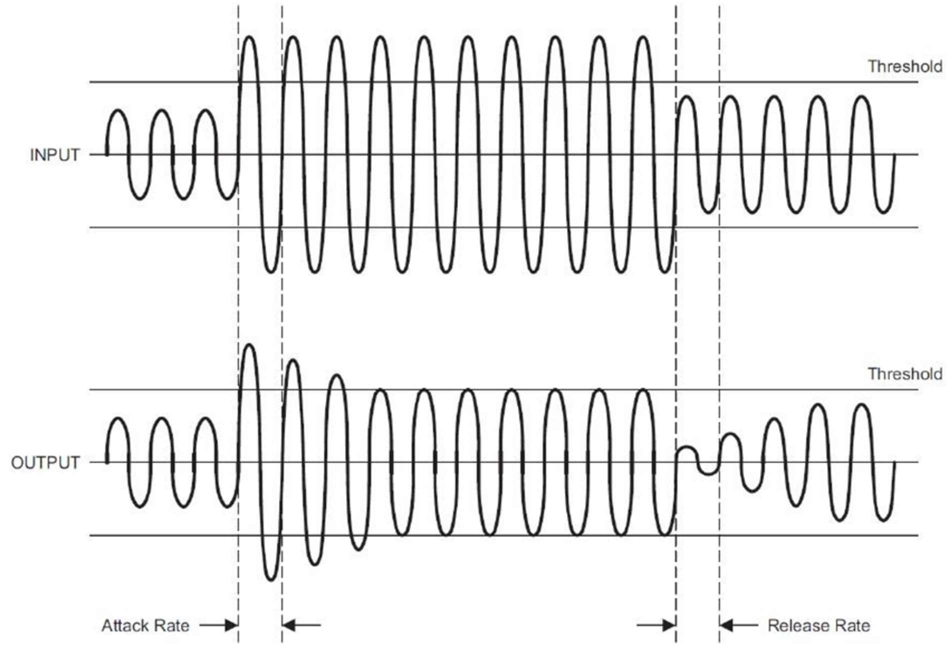

Figure 17. Attack Rate and Release Rate

The corresponding attack time and release time can be obtained by:

Release Rate / samples = 1000 × (User_release_time) / ms) / F_S (7)

Release time = (Gain_release_threshold) - Gain_current) / (Relase Rate / sample × F_S) = (Gain_release_threshold) - Gain_current) / (User_release_rate) (8)

Attack rate / sample = 2 × (1000 × (User_Attack_Time / ms) / F_S + Release Rate / samples) (9)

Attack time = (Amplitude_current - Amplitude_attack_threshold) / (Attack Rate / sample × F_S) + Smooth time

where

- User_release_time and User_attack_time are the expected release time and attack time
- F_S is the input sample rate
- Gain_release_threshold and is the gain when release thershold is reached
- Gain_current is current gain before release threshold is reached
- Amplitude_current is current amplitude of signal
- Amplitude_attack_threshold is the amplitude when attack threshold is reached (10)

Considering the non-linear structure of AGL, smooth time cannot be calculated with certainty, but it can be simulated through GUI.

For release time, it can be calculated with formulas above. Measured release time with different release rate on condition that the threshold is -6 dB, alpha time is 0.55 ms, and attack rate is 0.2 are below:

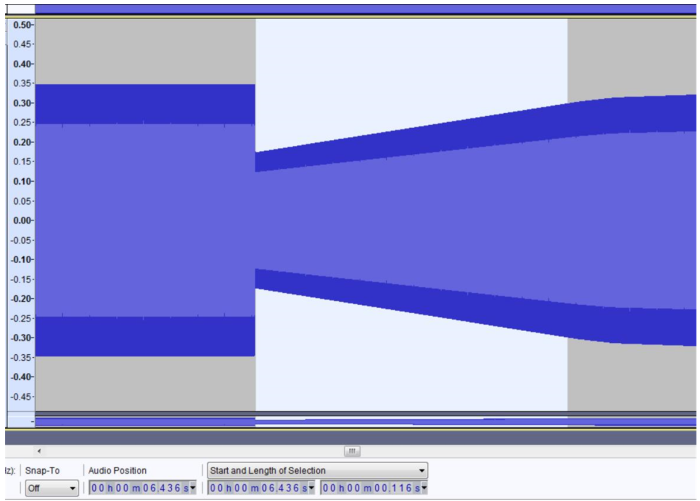

Figure 18. Release Rate = 0.0025

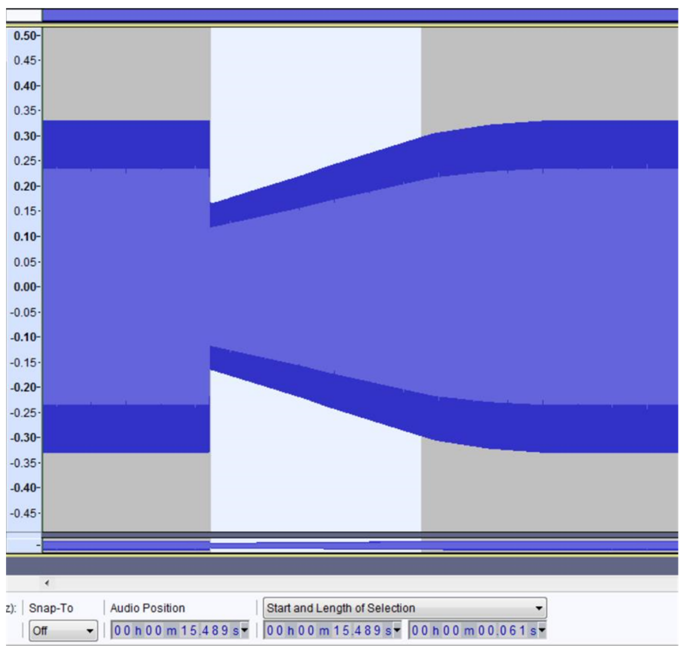

Figure 19. Release Rate = 0.005

3.3.3 Tuning Suggestions for Attack Rate and Release Rate

Usually, AGL is suggested to be tuned with certain fast beating music, such as piano music. The main aim of this is to avoid clipping and changing loudness of output music after AGL is triggered.

The attack rate can be set to 0.2 first and then the release rate can slowly be increased from 0.00001. More decay time helps to eliminate the distortion of piano music but it may cause changing loudness cycle by cycle when AGL is triggered and released. A common solution here is to increase the release time of DRC. A range from 50 ms to 300 ms is common in practical cases.

A larger attack time lets more input audio exceeding the threshold pass AGL while smaller attack time makes the attack behavior more aggressive and suppressive. This may cause loss of loudness. TI recommends tuning AGL to decrease clipping and avoid loss of loudness on condition that full scale fast beating music is played. It is suggested to avoid the gain curve (red curve in the GUI) being below the threshold when attacking. A range from 2 ms to 10 ms should suffice.

With attack behavior, always consider that the attack is followed by a smooth filter. Hence the actual attack time needs to consider the smooth filter time. Due to the nonlinearity of the smooth filter, it cannot be accurately calculated like decay time. It is suggested to adjust the attack rate slowly to achieve expected attack time through the GUI.

After finishing tuning the AGL, there may be clipping in certain frequency zones. Furthermore, DRC can be used to eliminate and reduce the clipping.

3.4 Three-band Dynamic Range Control

The Dynamic Range Control (DRC) is a feed-forward mechanism that can be used to automatically control the audio signal amplitude or the dynamic range within specified limits. The dynamic range control is done by sensing the audio signal peak level using an Alpha filter structure then adjusting region based gain, slope parameters, and so forth.

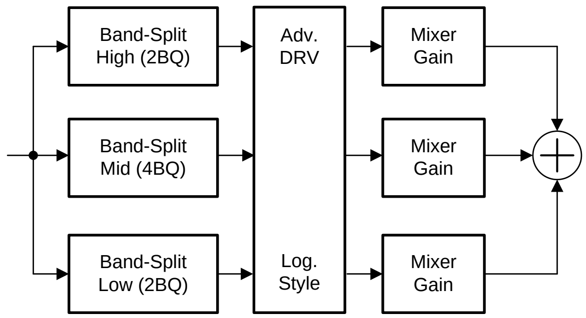

Figure 20. 3-Band DRC

3.4.1 DRC Tuning Interface

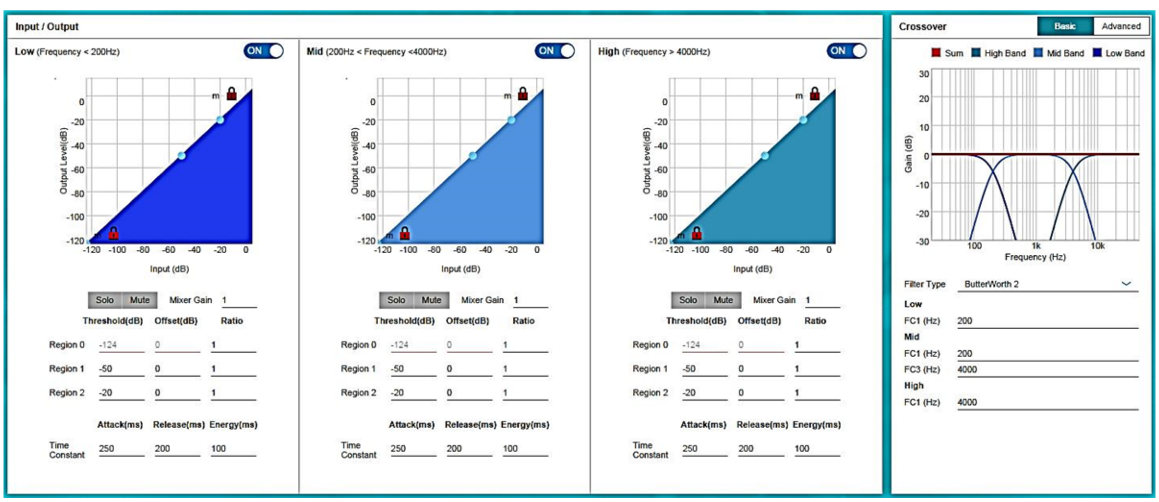

Figure 21. 3-Band DRC Tuning GUI

The DRC Tuning window consists of three identical windows for low, mid, and high frequency bands. Each band has a DRC curve that offers three regions of compression. The points on the DRC curve can be dragged and dropped. Below each DRC plot, parameters such as threshold, offset, and ratio can be manually typed in for each of the three regions. By typing a value and pressing Enter on the keyboard, the DRC curve automatically adjusts to the entered parameter.

The three elements comprising the DRC include a peak detector, a compression/expansion coefficient computation engine, and an attack/decay controller. Correspondingly, three kinds of parameters are needed to be set during the tuning.

3.4.2 Energy Time

For each band, a peak detector derives the peak value of the audio data stream into the DRC by applying an alpha filter structure as shown in Figure 22.

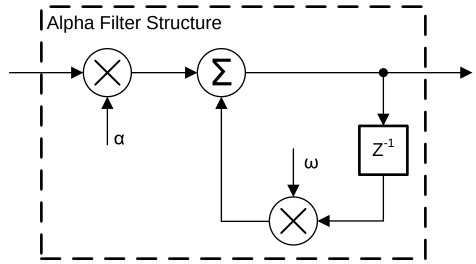

Figure 22. Alpha Filter Structure

$\alpha = 1 - e^{((-1000) / (F_S \times T_{Set}))}$

where

- F_S is sampling frequency (Note that F_S is the internal processing sampling rate rather than input sampling rate)
- e is Euler's number
- T_{set} is the user input energy time constant in the GUI (11)

Since Alpha filter is used to do peak detection from audio data stream, it is vital to keep the energy time constant. This can be calculated from at least one complete cycle of all the frequency tone in the audio data stream. For example, in the low band, the lowest cut-off frequency F_{min} in the bass is usually a specification in system design.

$T\_attack \geq 1 / F_{min}$ (12)

If Equation 12 is true, then the detected peak value is suggested to be computed by at least one complete cycle of every single frequency tone in the whole band.

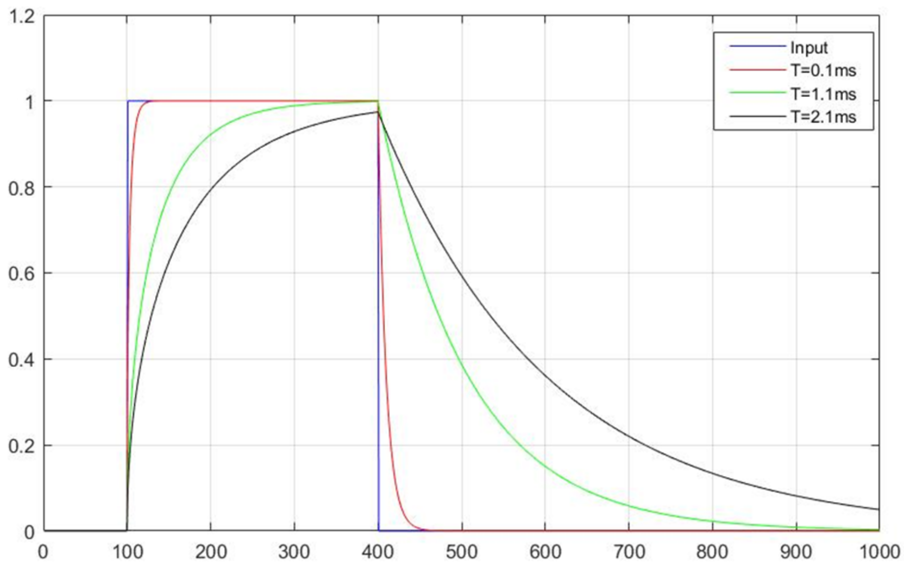

Figure 23. Response of Time Constant Value of Alpha Filter

As can be seen above, when T = 2.1 ms, T_attack is too big to achieve a peak before decreasing, which may cause loudness loss. In addition, T_attack being too small may cause overshot. Generally, TI recommends to set T_attack as:

$T\_attack \geq (2\sim3) \times 1 / F_{min}$ (13)

3.4.3 Attack Time and Release Time

Attack time is defined as time required for the amplitude to decrease to 36.8% of initial value while the release time is defined as time required for the amplitude to increase to 63.2% of its final value.

Based on the threshold T and the compression ratio, once estimated energy time obtained from the peak detector exceeds the threshold, the DRC attacks to adjust the audio signal to settled level. The DRC time constants control the transition time of changes and decisions in the DRC gain during compression or expansion. The energy, attack, and release time constants affect the sensitivity level of the DRC. A shorter time constant usually leads to more aggressive DRC responses.

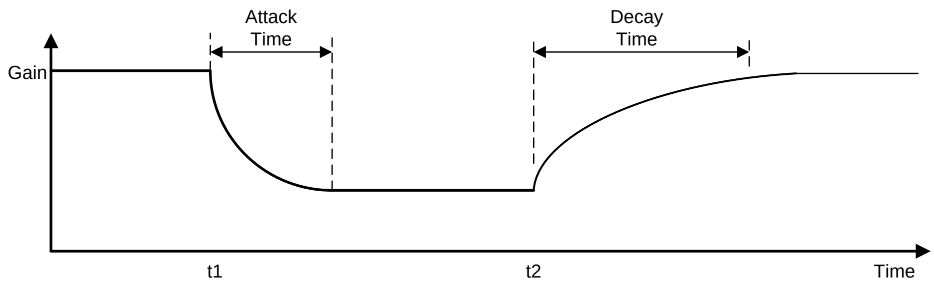

Figure 24. DRC Attack Time and Release Time

When the attack or release is activated, the attack or release behavior is spotted with a setting like Figure 25.

Figure 25. GUI Setting for Attack and Release Time Verification

The following results are obtained:

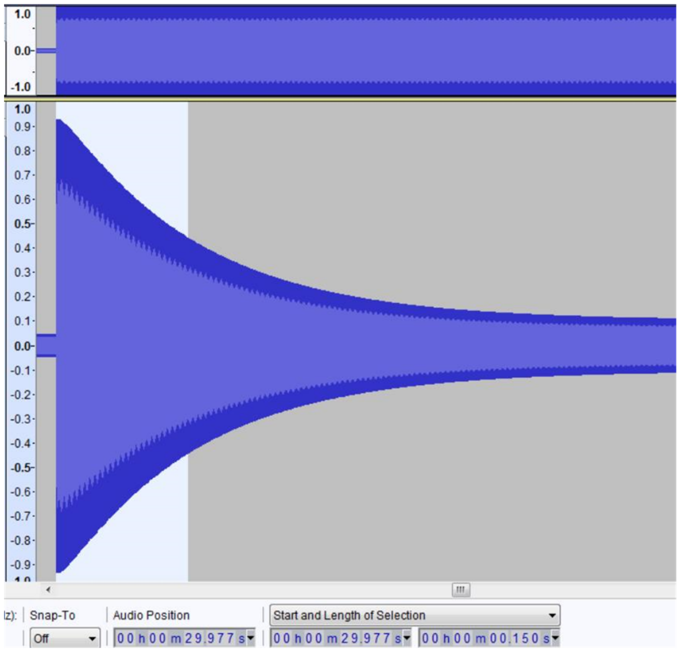

Figure 26. Attack Time

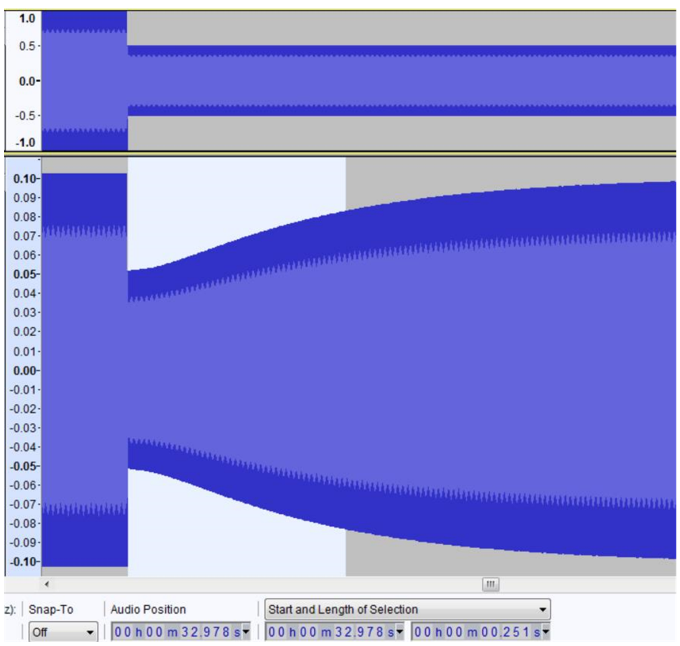

Figure 27. Release Time

As can be seen above, both the attack time and release time meet the following calculations:

T_attack_actual = T_attack + T_energy (14)

T_release_actual = T_release + T_energy (15)

T_attack and T_release are the attack time or release time set in the GUI and T_energy is the energy time for a Alpha filter. As can be seen above, after the Energy time is set, attack time and release time need to take the energy time into consideration for their actual values.

A small attack time may cause the DRC to be too sensitive, which leads to sudden weak loudness. On the other hand, a big attack time may increase the risk of clipping. Besides technical analysis through F_{min} / F_{max} in certain bands, TI recommends to set this value by listening. Similarly, actual release time also includes energy time.

3.4.4 Other Parameters

The DRCs have seven programmable transfer function parameters:

- k0
- k1
- k2
- T1
- T2
- OFF1
- OFF2

The T1 and T2 parameters specify thresholds or boundaries of the three compression or expansion regions in terms of input level. The parameters k0, k1, and k2 define the gains or slopes of curves for each of the three regions. The parameters OFF1 and OFF2 specify the offset shift relative 1:1 transfer function curve at the thresholds T1 and T2, respectively.

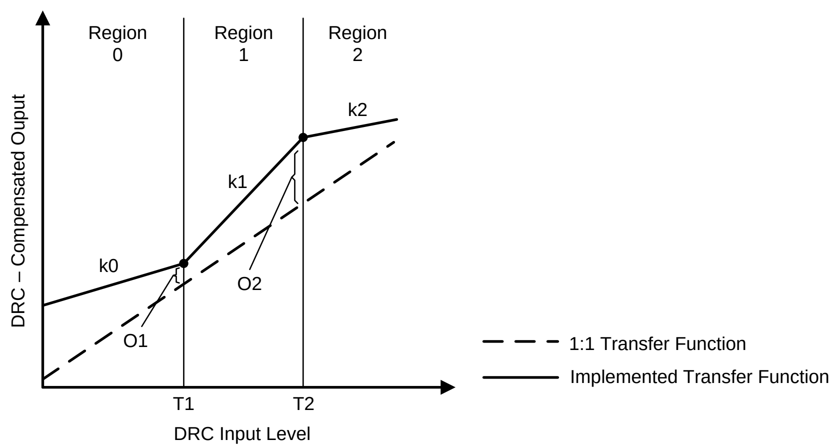

Figure 28. Programmable DRC Transfer Function Structure

The mixer gain of each of the three frequency bands after the DRC are all set to be 1. Since filters may affect phase, a phase reversal or a 180 degree phase shift may be necessary, which calls for using a negative sign on the coefficient to reverse the phase for the second-order LR filter. In the TAS58xx family, the phase reversal is not needed because 2BQ, 4BQ, and 2BQ are separately used in the crossover. After tuning, the response is automatically displayed on the right side of the DRC plot. The Crossover configuration has two tabs. In the Basic tab, only the filter type and cut-off frequencies need to be determined. Go to the Advanced tab if more parameters need to be adjusted.

Due to the flexibility of DRC configuration, it can be used to implement many specific functions. For example,

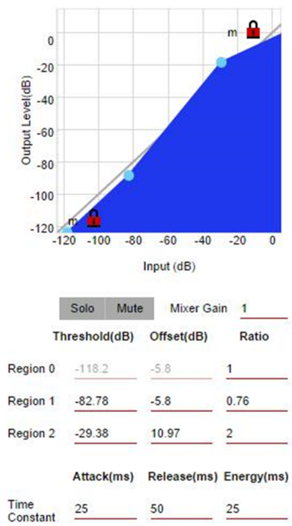

Figure 29. Dynamic Bass Enhancement

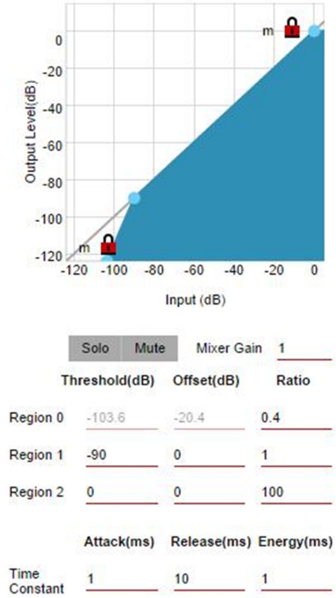

Figure 30. Dynamic Bass Enhancement

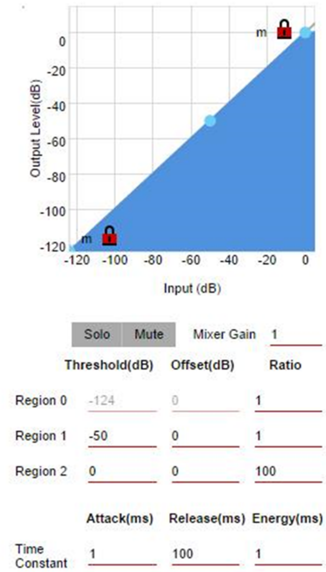

Figure 31. Function as AGL

3.4.5 Crossover

The 3-band DRC is divided by three filters: Low, Mid, and High as shown in Figure 32.

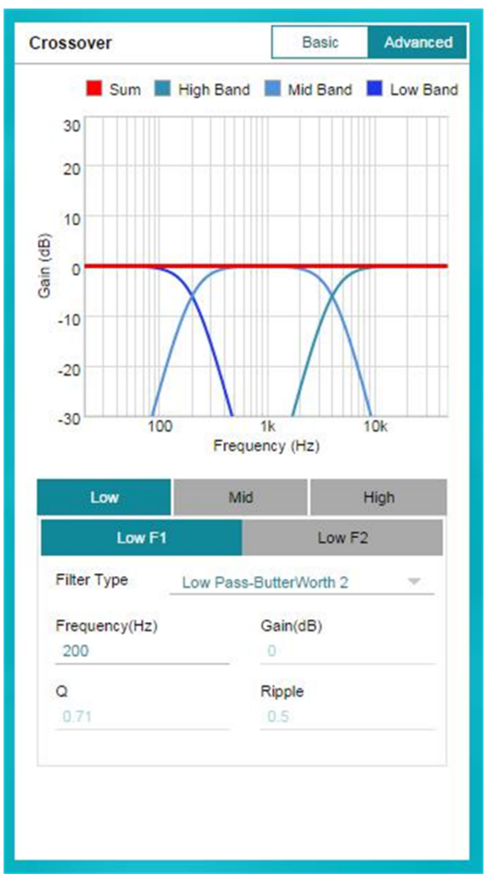

Figure 32. DRC Crossover

As can be seen above, a fourth order filter is provided by F1 and F2 for Low, Mid, and High. Sometimes, it is desired to use a 3-band DRC alone rather than 3-band DRC with AGL. One main concern here is when AGL is turned off, if DRC is triggered, there is a bulge at the junction of the two cross-overs. In this case, AGL can effectively guarantee the flatness of the entire frequency band.

3.4.6 Tuning Suggestions for 3-band DRC

After setting the AGL, it is suggested to tune the DRC from 1-band (Mid-band) DRC like Figure 33.

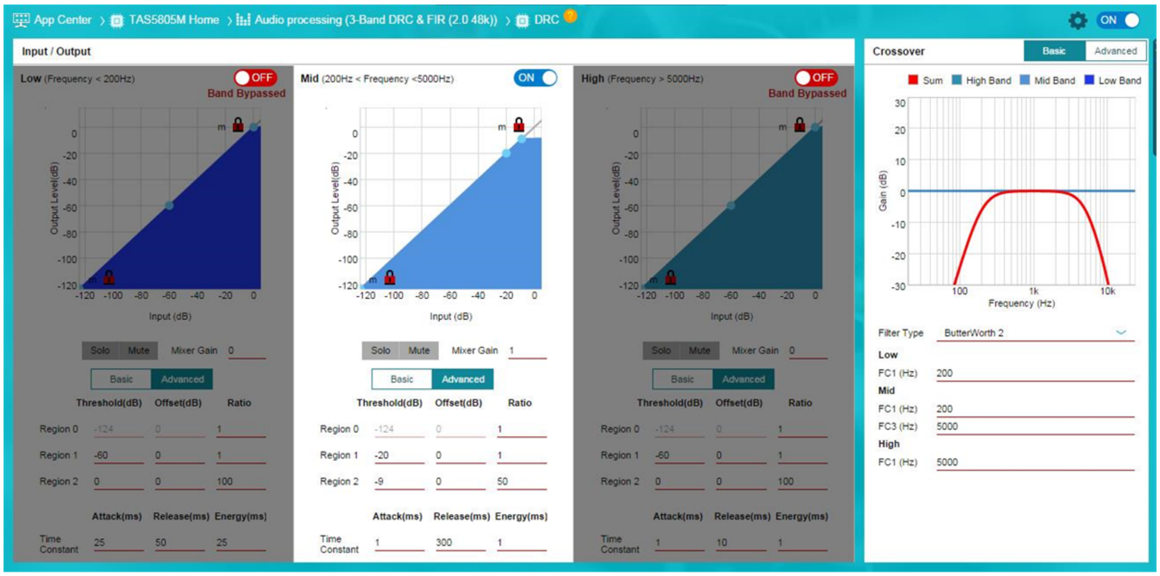

Figure 33. Mid-band DRC

First, energy time can be set based on Figure 34.

$$T\_attack \geq (2\sim3)1 / f\_min \quad (16)$$

Considering Fc1 of the Mid-band is 300 Hz, T_attack > 7~10 ms. Firstly, in order to avoid a sudden change of loudness, 7 ms should be set. Secondly, a 300 ms release time can be set during the tuning of the attack time from 0.5 ms. Listen to the music and change the attack time until the clipping is reduced. After the attack time is set, release time can be reduced until the loudness is satisfactory.

If bass cannot meet the expectation, the low band of DRC can be enabled for more bass like Figure 34.

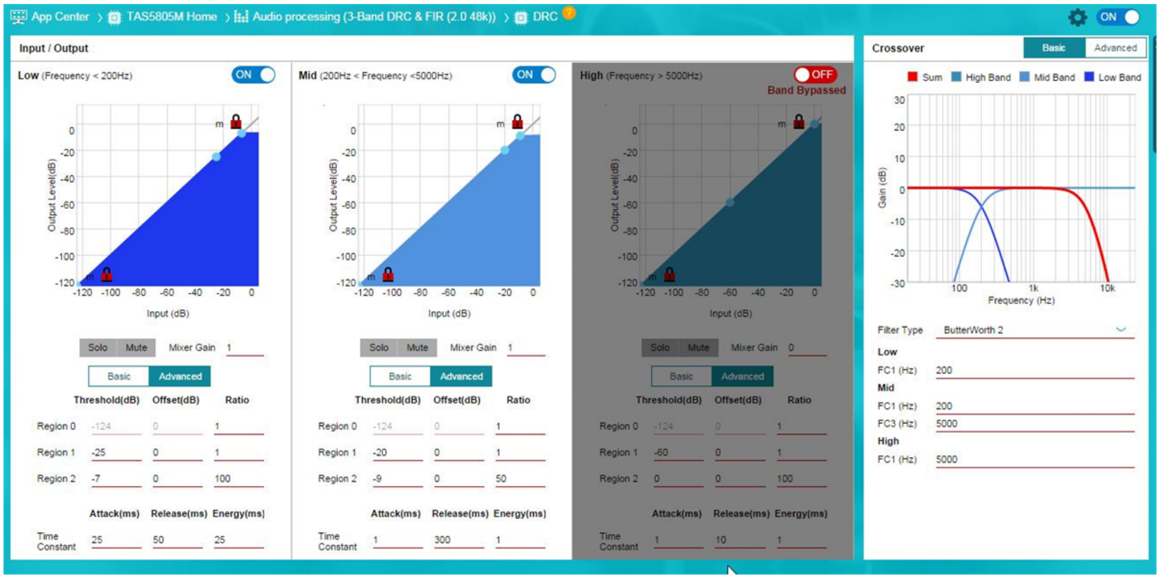

Figure 34. Low-band DRC

The next step is to set the energy time. It is recommended to set the attack and release time to be 50 ms. Then, reduce the attack time to obtain more bass while increasing the attack time to balance the bass and people speech.

Similarly, turn on the 3-band DRC to achieve the desired sound.

Figure 35. High-band DRC

Usually, the threshold of region 2 can be set to the same value as the low band. Use the default value of the time constant for listening at first. If it is not satisfactory, release time can be set as default and attack time can be adjusted until satisfactory.

4 Loudness Tuning

Loudness is usually measured by SPL (Sound Pressure Level). Usually, it is recommended to use volume to obtain the required loudness. If the increased volume leads to digital clipping, decreasing the digital gain as well as the increasing analog gain is suggested to achieve more headroom for volume.

Another common method to increase loudness is to use the clipper, but it introduces the digital clipper. Figure 36 shows the structure of the clipper.

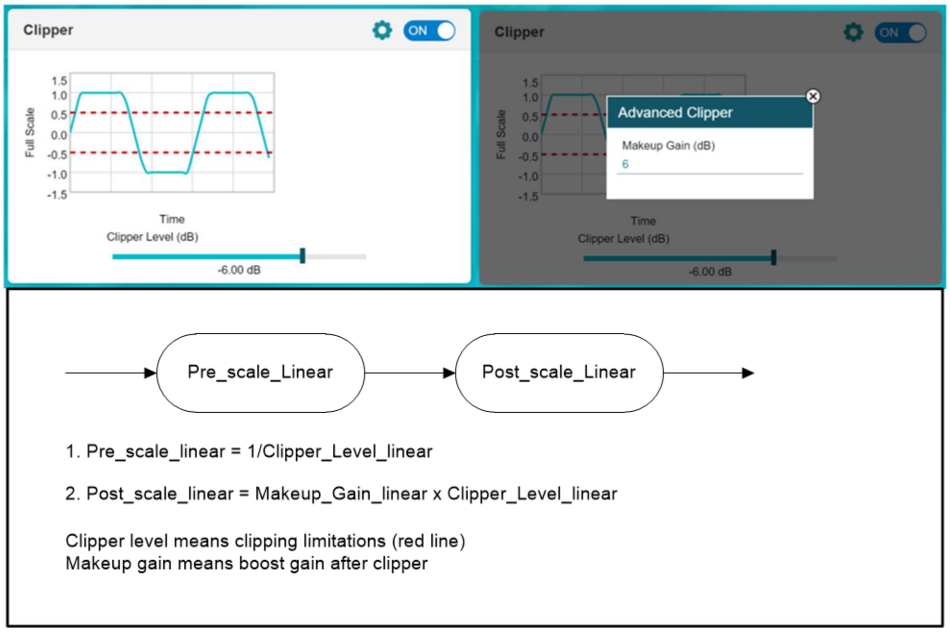

Figure 36. Clipper and Makeup Gain

As can be seen above, the clipper level first limits the digital output to the red line, then makeup gain boosts the signal after the clipper. For example, if the clipper level = -6 dB and makeup gain = 6 dB, then:

1. Pre_scale_linear = 1 / $10^{(-6\ /\ 20)}$ = 2
2. Post_scale_linear = $10^{(-6\ /\ 20)} \times 10^{(-6\ /\ 20)}$ = 1

As the post_scale_linear indicates, the total gain for this structure is 1, hence maximum peak output voltage stays the same. It generates digital clipping with THD attenuation for higher output RMS voltage. TI recommends to do the following:

1. Take AGL_threshold into consideration.
2. Try to set Clipper_Level = AGL_threshold to -3 dB. Check the output THD+N level with a 0 dBFS signal.
3. Fine tune Clipper_Level until the output THD level is acceptable.
4. Configure Makeup_Gain = Clipper_Level – AGL_threshold. The output power is increasing.

Table 6 is collected on the TAS5805M EVM board, 18 V PVDD, 6 Ω load, AGL and -5 dB to avoid PVDD clipping. The input signal is 1 kHz and 0 dBFS signal.

Table 6. Clipper and Makeup Gain to Increase Output Power

	DEFAULT	-5.5 dB CLIPPER + 0.5 dB MAKEUP GAIN	-6dB CLIPPER + 1dB MAKEUP GAIN	-6.5 dB CLIPPER 1.5 dB MAKEUP GAIN	-7 dB CLIPPRT 2 dB MAKEUP GAIN	-8 dB CLIPPER 3 dB MAKEUP GAIN
Output level	11.74 Vrms	12.16 Vrms	12.52 Vrms	12.82 Vrms	13.1 Vrms	13.56 Vrms
THD+N	0.03%	2.7%	5.1%	7.39%	9.6%	13.65%

As can be seen above, considering -5 dB AGL has avoided the PVDD clipper, the steps above ensure more output level with expected THD+N.

Compared with DRC and AGL, the clipper without makeup gain can limit output level without any response time at a cost of worse THD+N because it does not have attack time or release time. In certain applications, for example, when the PVDD current is needed to be strictly limited to a certain value, the clipper can be used to limit the maximum output peak current.

Mailing Address: Texas Instruments, Post Office Box 655303, Dallas, Texas 75265