

Application Report
TAS2563 Tuning Guide



ABSTRACT

The *Smart Amp Tuning Guide* facilitates the quick implementation of microspeaker tuning using a smart amp device such as the TAS2563 without requiring any audio tuning experience. This guide is intended for smart amp applications such as mobile phones, tablets, laptops, and home automation. The quick tuning procedure provides a step-by-step process for rapid tuning for the purposes of speaker matching. *Tuning Elements* provides a more detailed explanation of audio theory and of the useful audio blocks in the device.

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1 Introduction

The task of optimizing the sound pressure level (SPL) and sound quality of a micro-speaker for mass-market, low-power applications can be very challenging. The combination of digital signal processor (DSP), digital-to-analog converter (DAC), and smart amp protection provide the audio engineer with many tools and knobs with which to achieve the best quality sound for the small-form factor speakers popular in portable low-power devices like mobile phones, tablets, laptop computers, and home automation. Micro-speakers are especially limited in maximum SPL and bass response due to the small diaphragm size; however, the smart amp allows the user to tune the speaker to achieve the maximum audio quality from a limited speaker. For professionals in the audio industry, learning from the target audience (customer) as to what their key audio priorities are is important to make adjustments to their preference. However, there are some universal qualities and characteristics of a good balanced tuning. Most customers generally want the same ideal sound:

- As loud as possible
- Strong and tight bass response
- Low distortion
- Warm and smooth sound
- Voice clarity
- Stereo balance
- Flat response
- Speaker protection

Achieving all of these goals is usually very difficult, so tuning is often a matter of determining the customer's priorities. Some customers are willing to sacrifice SPL to achieve a flatter, more-balanced response. Some customers prioritize SPL at the risk of a narrower tone. A well-designed speaker, combined with the smart amp, typically enables a well-rounded sound which can then be fine-tuned to the customer's preference.

This application report makes many references to the PurePath™ Console 3 (PPC3) graphical user interface (GUI) used for both integrated and non-integrated smart amp solutions, such as the TAS2563 EVM. This document does not cover proper installation, navigation, and elementary proficiency with PPC3. This application report mainly focuses on the "Tuning and Audio Processing" tile under the device home page of the GUI and expands on certain tuning tools in the GUI.

This report does not address speaker characterization. While proper speaker characterization is actually quite critical to a good tuning and is integrated into PPC3 and the EVM hardware, this report assumes the speaker parameters, such as the Re , Bl , X_{max} , T_{max} , S_d , Thiele-Small model, and temperature model are measured and loaded correctly into the .ppc3 file before proceeding with the tuning. Perform a full characterization with the combination of PPC3, the smart amp EVM, and the learning board (sold separately). The user can also import parameters from one .ppc3 file to another or enter them manually if acquired by other means.

This report also assumes that the speaker normalization is complete. Even though speaker normalization is not included in the characterization page of the GUI (and is found in the *Audio Processing* section), the user must consider this as a required post-characterization step, which is critical to any good tuning. Normalization allows the smart amp algorithm to compensate for the inherent high-frequency impedance response caused by the Le of the speaker. The volume of the test signal in the wizard can be left at the default value of -12 dB.

This report only addresses the mono-tuning mode, and does not address the dual-mono mode used in stereo applications with multiple devices. The user can apply all of the tuning strategy that this report proposes to each speaker of a stereo application. However, note that stereo applications can benefit from additional tuning such as phase alignment, ganged dynamic range compression (DRC), or linked smart amp protection.

2 Quick Tuning Procedure

A deep understanding of audio and speaker fundamentals, as well as familiarity with the smart amp device, is extremely helpful toward achieving the optimal tuning but takes time and practice to develop. The goal of this application report is to facilitate rapid implementation of quality audio for any speaker using the smart amp technology, with little audio experience required. This procedure excludes detail on any particular feature to focus on the minimum steps required for any good tuning. This document provides details on certain features and functions, which are referenced and located throughout the document.

This particular section focuses on the common example of reference matching, where the user is attempting to match the acoustic response of the target device to a reference sound source. In some cases the customer has a particular reference device, such as a phone, that has the audio performance and character that they want to replicate in their product. This reference is a great place to start for tuning because it gives the tuner some direction of audio priority as well as some objectivity. In this report, the device under tuning refers to the target device and the reference refers to the reference device. Tuning to match only one speaker at a time is preferable; so, use left-only or right-only tracks on the reference device if possible.

TI highly recommends that the user study the algorithm signal chain before tuning to understand the effect and sequential order of each signal path block on the audio signal. TI also recommends to tune the speaker in the quietest room available for both measurement and listening purposes.

2.1 Initialize Tuning

One very important task is to “neutralize” every step of the signal chain before tuning, so as to start with the natural speaker response. As a best practice, determine at all times which steps of the signal chain are enabled and which affect the sound. For instance, the user should bypass all equalization (EQ), dynamic range compression (DRC), smart amp protection, and input gain as much as possible at the beginning of any tuning. Bypass any preprocessing of the audio originating from the system host as well. When using the in-system tuning tool to tune a product through USB, any audio processing from the host processor must be bypassed.

A recurring check that can prevent frustration throughout the process is to ensure that the system gain is set to a known reference value. The soundcard playback volume found in a Windows audio device configuration (often available as a shortcut key on the keyboard) is responsible for setting the system volume in the case of EVM tuning. In the case of a reference device used for comparison, such as a mobile phone, the volume buttons on the phone hardware usually control the phone system volume. Ensure that the system audio setting does not have an artificial limit that prevents the device from achieving the maximum volume. The personal computer (PC) and mobile device scales can often differ in step size, so when comparing two sound sources it is best to simply keep all device system volume at the maximum. Setting the source volumes at the maximum specification allows for the maximum SPL out of the smart amp and ensures a fair comparison to a reference source. This practice also accounts for the artifacts of high-volume tuning, such as unwanted distortion and compression.

2.2 Flatten Protection

The user can only truly bypass the speaker protection section of the algorithm in one of the ROM modes and not in tuning mode. However, the tuning does require tuning mode, where the user cannot bypass excursion and thermal protection but merely avoid it. The default values for the protection tuning in PPC3 are sufficient in most cases. TI recommends a thermal and excursion priority of 1, 1, and 1.

Figure 2-1 shows an example of a flat protection tuning.

Protection Tuning			
<input type="checkbox"/> Feed Forward Protection Only	Low	Mid	High
<input checked="" type="checkbox"/> Enable Output	<input checked="" type="checkbox"/> Enable Output	<input checked="" type="checkbox"/> Enable Output	
Frequency (Hz)	171.97343	800	2000
Thermal Priority	1	1	1
Thermal Speed (ms)	30	30	30
Excursion Priority	1	1	1
Excursion Speed (ms)	100	50	30

Figure 2-1. Flat Protection

2.3 Frequency Response Matching

This subsection addresses how the tuner can use the smart amp tuning elements of input gain, EQ, DRC, and smart amp to match the pink noise response of a reference sound source. At this point the user must implement the EVM or prototype device, PPC3 GUI, a microphone, any audio player software, and an audio analysis software tool, such as RoomEQ Wizard.

Use a prepared playlist of reference pink noise audio tracks here starting with 0 dBFS or -6 dBFS and decreasing to -78 dBFS. Most audio processing software can generate these audio tracks. If the reference device has more than one speaker, the user must utilize a left-only or right-only track to avoid measuring the stereo response of the device. Alternatively, ask the TI audio engineer to provide these files. After acquiring these tracks, copy the files to the reference device and open the tracks in an audio player on the PC connected to the target device. TI recommends to make a playlist from the noise tracks on each device.

2.4 Microphone Setup

For the next step, set up the microphone for mono input and open the audio analysis software to a real- time fast-Fourier transform (FFT) analyzer tool where the input to the FFT data is the microphone data. Arrange the microphone orientation directly toward the speaker port of the reference device with a distance of approximately 10 cm. Play the loudest pink noise track on the reference device and adjust the microphone gain so that there is no clipping or overdrive on the microphone system input. This level can usually be seen in the sound devices recording window on a Windows® operating system (OS). Note the importance of retaining the same system volume, microphone distance, and microphone gain for all comparison testing. The absolute level of the gain or input signal is not important, but it is important to stay consistent. If a microphone audio reference device is available, the microphone can be calibrated using this, but it is not necessary when only making a device comparison measurement.

2.5 Record Reference Device Response

Play each noise track on the reference device and record and save the frequency response for each track. These recordings eventually produce a family of curves similar to that in [Figure 2-2](#). This family of curves define the pink noise response of the reference device and contains a lot of information on the reference sound for use later in the tuning process.

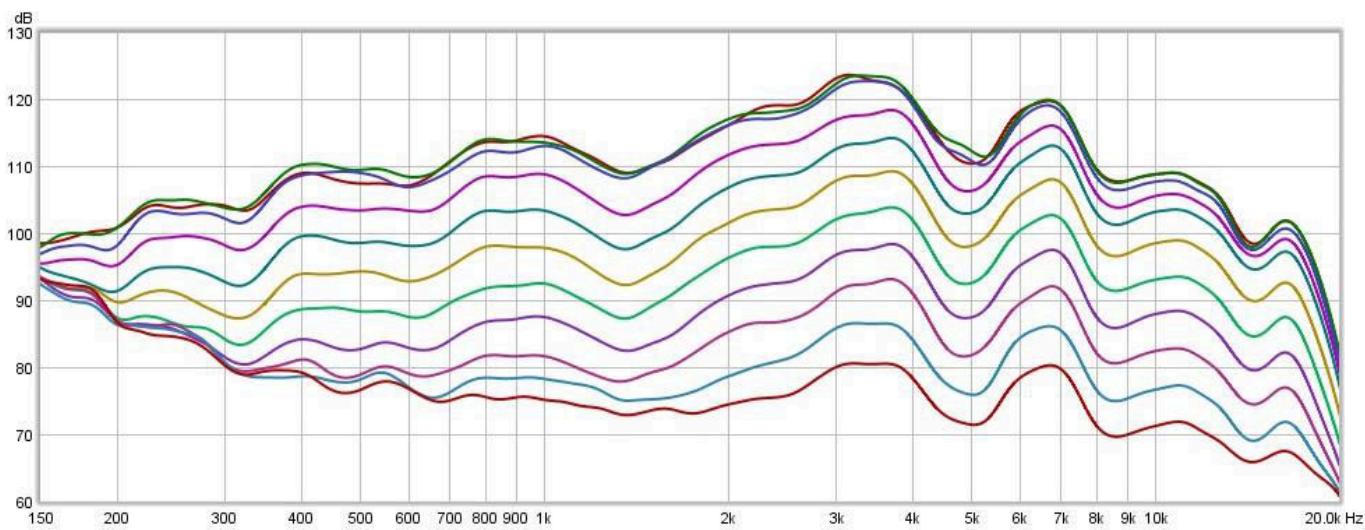


Figure 2-2. Pink Noise Response Curve Family

2.6 Set Input Gain

Now point the microphone toward the target device speaker port at the exact same distance and keep the same gain as before. Play the -48-dB pink noise through the PC audio player, EVM, and target speaker. Use the audio analysis software to overlay the -48-dB pink noise response of the reference device onto

the new target device response. Adjust the input gain of the target device by using the input gain control in PPC3 until the average level of the two plots match as much as possible. This matching does not have to be accurate. Sometimes, the frequency response shape of the two devices is quite different, which makes matching the two curves across the frequency difficult. The point of this step is to only match the general sound level across the entire band, not at all the individual frequency regions.

2.7 Equalizer

The next step is to use the smart amp equalizer to match the shape of the curve across the spectrum as much as possible. The most common biquad to use for adjusting the EQ is the “Equalizer (Bandwidth) or Equalizer (Q Factor)”, which functions as a band-pass filter where the center frequency, gain, and width can be adjusted. Adjustments to the bass shelf, treble shelf, and low-pass filter can also be useful. Continue to play the -48-dB pink noise on the target device and overlap the target device response and the reference device response as before. Use the available biquads to match the target response to the reference response as best possible.

[Figure 2-3](#) shows an overview of the equalizer tool. [Figure 2-4](#) shows an example of low-volume pink noise matching.



Figure 2-3. Equalizer Overview

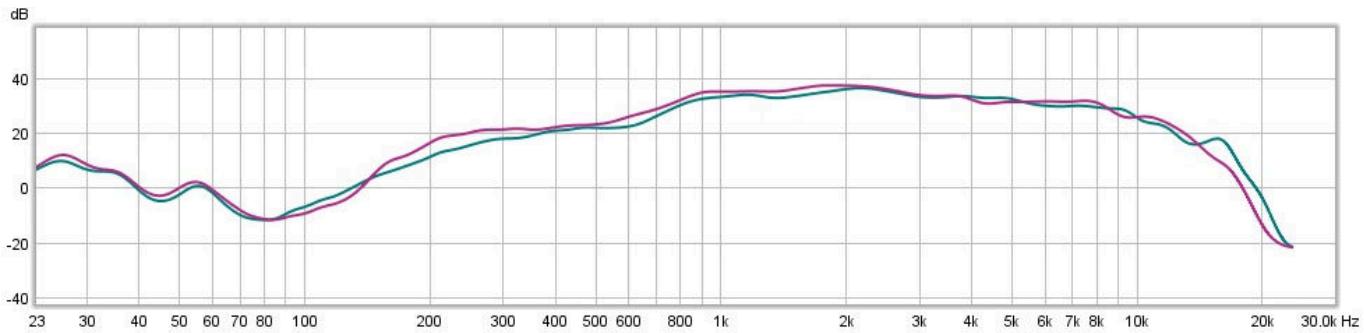


Figure 2-4. Pink Noise Low-Volume Matching

After completing this step, record the target device response for the -48-dB pink noise as well as -78 dB through -24 dB. If the -48-dB target device response matches the reference response, then the remaining plots up to -24 dB are likely to match as well. This occurrence is known as low-volume response, or low-volume tuning, which is not affected by any DRC or speaker protection, but only the input gain and EQ.

2.8 Dynamic Range Compression (DRC)

After completing the low-volume tuning, the next step is to incorporate the DRC for mid- to high-volume tuning. On the DRC page, click the enable button at the top right to enable the global DRC function. Next, click on the Crossover button on the right side to reveal the crossover controls. Set the crossover filter and time constant settings as shown in [Figure 2-5](#) and [Figure 2-6](#).

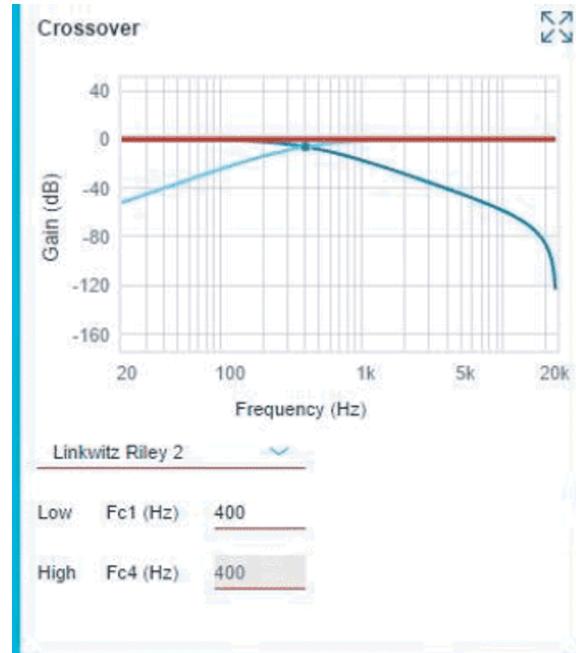


Figure 2-5. DRC Crossover Filter



Figure 2-6. DRC Crossover Time Constants

Next, set the compression input and output configuration for each band as shown in [Figure 2-7](#). Do not change the mixer gain or offset—this is a very harsh limiter that prevents most of the high-volume signal from reaching the protection stage. In fact, this limiter effect can be visible with the pink noise in the audio analysis software.

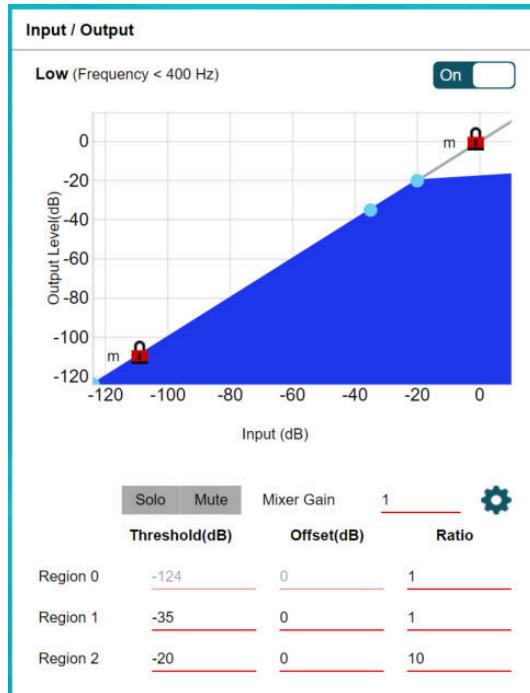


Figure 2-7. DRC Threshold Control

For each band, raise the threshold of region 2 until the response matches the reference device response or ceases to go higher. Now reduce the region 2 threshold again by 5 dB and change the ratio to 5. For region 1, change the ratio to 1.2. The goal is to match the maximum level plot (−6-dB pink noise) to the reference plot for the same level. In the EQ-only region, only 6-dB amplitude intervals occur between each noise track. In the protection region, intervals close to 0 dB occur between each track. Gaps smaller than 6 dB occur in the DRC region. The goal is to try to match these gaps as much as possible.

Figure 2-8 shows an example of the output acoustic pink noise response with the effect of noise floor, DRC, and protection. Figure 2-9 shows the significant power ranges in the audio and the effect of EQ, DRC, and protection.

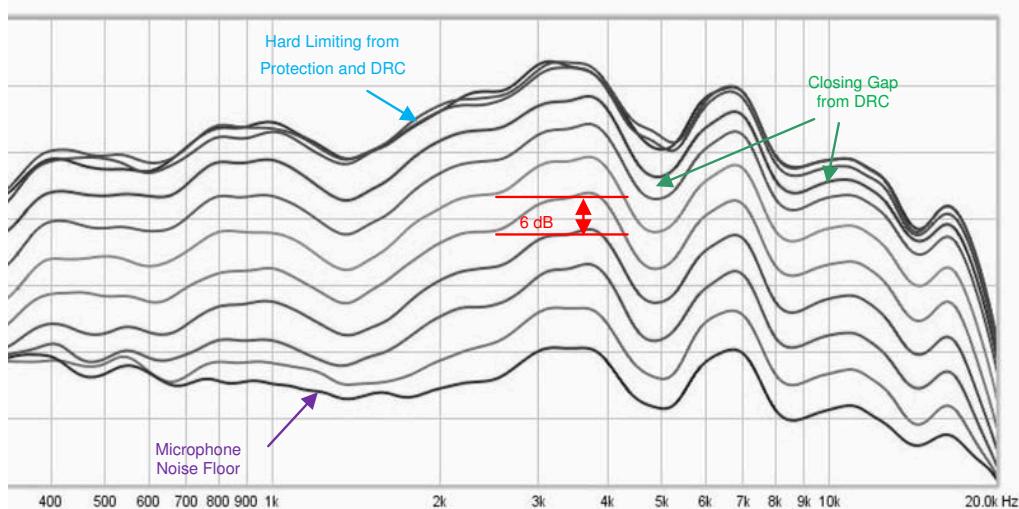
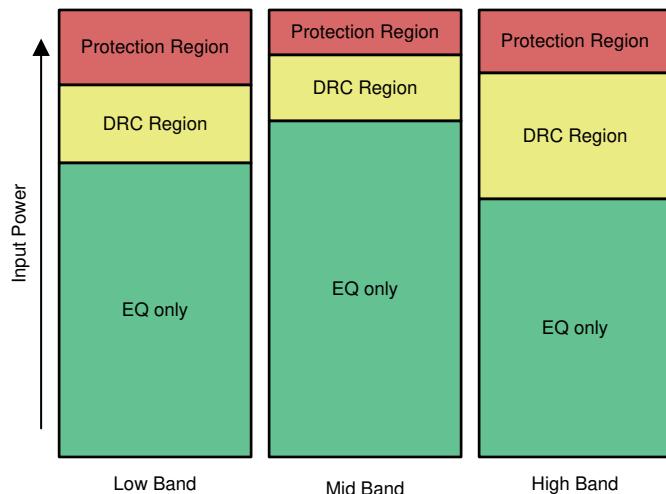


Figure 2-8. Acoustic Pink Noise Response



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Figure 2-9. Input Power Regions

2.9 Smart Amp Protection

When the DRC is enabled and is aggressive enough, the protection rarely initializes and does not dominate the overall color of the sound. However, pushing the speaker output to its limits to achieve a high SPL is ideal. If the speaker response at high volumes cannot achieve the level of the reference device in a certain band with or without the DRC enabled, try to increase the excursion and temperature protection priority for that band.

Remaining aware of the resonant frequency (F_s) and the electrical resistance (R_e) acquired during characterization is always important. Find these parameters on the characterization data tile in the *Audio Processing → Smart Amp* pages of the PPC3. The high-pass filter cutoff for nominal loudness tuning determines the low cutoff frequency of the low-band protection. The user can set the cutoff frequency of the mid-band protection to either include or exclude the resonant frequency. TI recommends to set the mid-band frequency 100 Hz above F_s for matching purposes.

Users typically note that the most excursion occurs in the low band, especially if the low-band cutoff is set to include F_s . As the speaker characterization excursion response in the [Figure 2-10](#) shows, the high band does not have much excursion; therefore, the user can set the low band with a higher priority than the high band. Try 3, 2, and 1 as low-, mid-, high-excursion priorities to see if it increases the low-end response. Likewise, the speaker temperature is dominated by the mid and high bands. Try 1, 2, and 3 as the low-, mid-, and high-thermal priorities. Set the low-band excursion and thermal speed to $10 / (\text{mid-band frequency})$. For instance, if the mid-band frequency is 1000 Hz, set the low-band speeds to 100 ms. Likewise, set the mid-band speeds to $10 / (\text{mid-band frequency})$. Be sure to check whether these settings have any effect on the pink-noise response matching in a negative way.

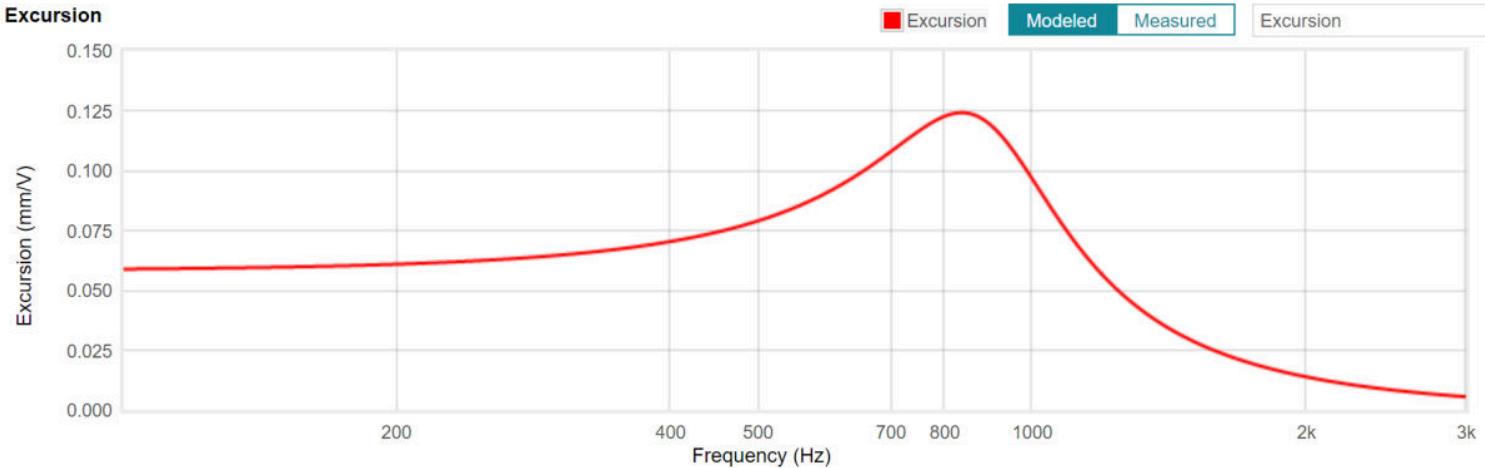


Figure 2-10. Modeled Excursion From Characterization

2.10 Speaker Protection Verification

After setting all of the tuning parameters correctly for the pink noise matching, the next important step is to actually listen to some music and voice tracks to ensure that negative artifacts do not occur. Choose several songs with varying degrees of volume among different genres. Piano, bass, drums, and voice are varieties of noise that serve as good test parameters for worst-case excursion and distortion.

Open the verification page from the bottom of any GUI page (see [Figure 2-11](#)). This interface shows two plots by default: estimated excursion and temperature. Play a very loud and busy track from a playlist and press the *Start* button on the verification page. To pass verification, neither the excursion nor the temperature should exceed their limits as denoted by the red lines. This verification process ensures that the speaker protection is properly working.

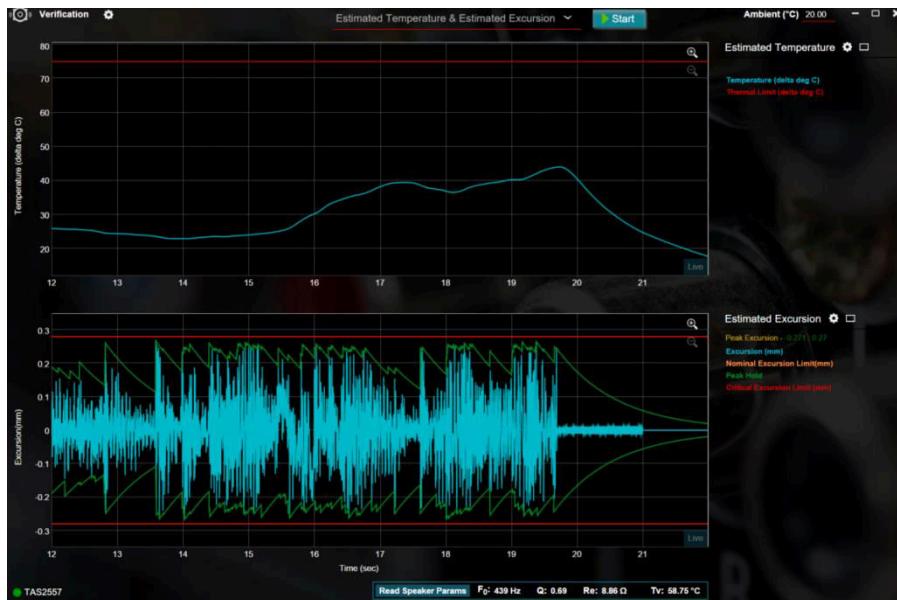


Figure 2-11. GUI Verification Page

2.11 System Signal Chain

Understanding the smart amp signal chain can be very helpful for optimizing sound level and quality while avoiding distortion or undesirable effects from overdriving the audio signal. This signal flow also facilitates the debugging of unwanted artifacts from the tuning. TI generally recommends to tune the speaker in the order consistent with the signal path. The gain stages in the chain are: host system gain, host preprocessing, input gain, EQ, DRC makeup gain, and the class-D gain

[Figure 2-12](#) shows the audio signal chain through the TAS2563.

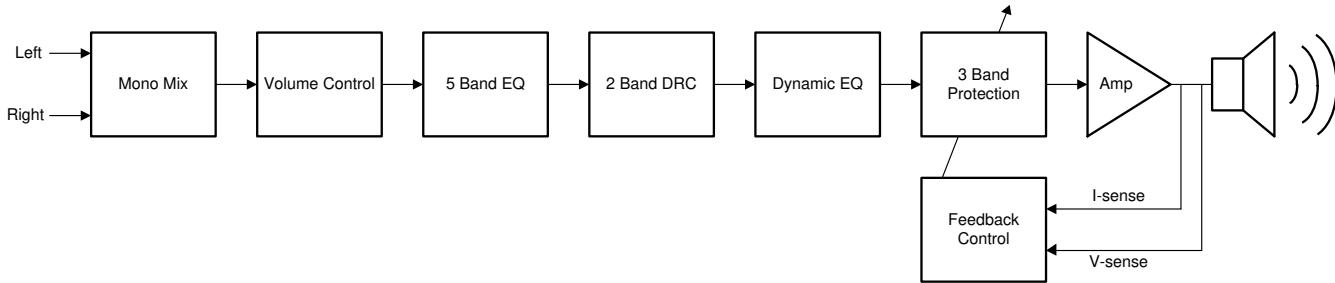


Figure 2-12. Smart Amp Signal Chain

- **Volume Control:**

The Volume Control is input signal. This gain stage has a dramatic effect on the output volume and sound quality. TI recommends to tune this first, especially for reference device matching, and then adjust this value at the end of tuning to help avoid distortion and over-compression. A value of a 0-dB gain is equivalent to bypassing this stage. The output of this gain stage feeds directly into the EQ.

- **EQ:**

The EQ is an optional block of the chain but the most influential part of speaker tuning. The broad spectrum input gain stage greatly influences the input to the EQ block, which is further limited by the DRC stage. The EQ is used to color and shape the general sound of the speaker output. TI recommends to tune this immediately after the input gain and with low-volume input signals as to avoid influencing the sound with the DRC and protection stages. This block feeds into the DRC stage.

- **DRC:**

The DRC is a highly-recommended (but optional) advanced feature of the smart amp. The Input to the DRC block comes from the output of the EQ block. The output feeds into the smart amp protection block.

- **Protection and feedback:**

The protection and feedback is a critical and final stage of the smart-amp-algorithm digital path used for speaker protection. This stage controls the final digital gain in three bands based on the input from the DRC and from the feedback path. If all other stages are bypassed, this stage continues to protect the speaker from overexcursion and overtemperature and also colors the output sound based on the priority settings.

- **Class-D Amp:**

The digital input to the class-D amplifier comes from the output of the DSP, specifically the speaker protection stage. The analog output of the class-D amplifier feeds to the device pin and then to the speaker. The designer can control the class-D amplifier parameters, which are listed under the home page of the GUI, using the "Device Control" tile (see [Figure 2-13](#)). TI does not recommend modifying any of the default settings found here, including the amplifier gain.

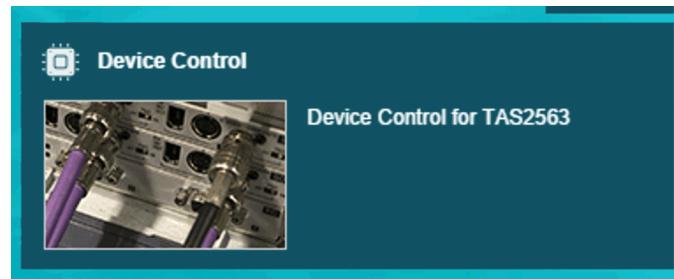


Figure 2-13. Device Control GUI Tile

- Speaker:

The OUT pins of the smart amp device and the Class-D amplifier output feed the input to the speaker. The designer may place an optional ferrite bead in between the OUT pins and the speaker terminal.

2.12 Tuning Elements

2.12.1 Input Gain

Located in the upper-right corner of the *Audio Processing* page, the input gain is a simple digital gain on the input signal. The ideal setting can vary widely between speakers and is usually subject to user preference. For mobile phone and tablet speakers, the input gain is typically set between +2 dB to +12 dB. Setting the input gain too high can result in a loud but very compressed or distorted sound at the output when the input signal is large. Choosing an overly aggressive setting with this gain to achieve a high SPL output is common, but this has negative consequences on sound quality. Toggling the  symbol also allows the user to quickly mute or unmute the input signal.

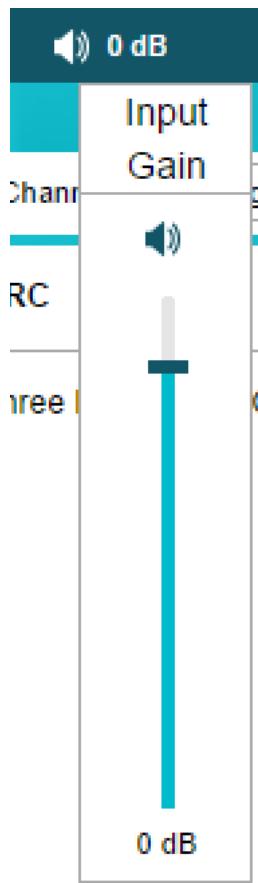


Figure 2-14. Input Gain Control

2.12.2 Equalizer

2.12.2.1 Biquads

Located under the *Audio Processing* page, the biquad stage comprises ten biquad filters which feature an array of filter options such as first-order or second-order low-pass and high-pass, notch, bass and treble shelf, and band pass. The configuration options vary by filter type. The most typical filter is the equalizer, where the range can be set as the bandwidth or Q.

Figure 2-15 shows an example setting for a bandpass filter using the EQ control.

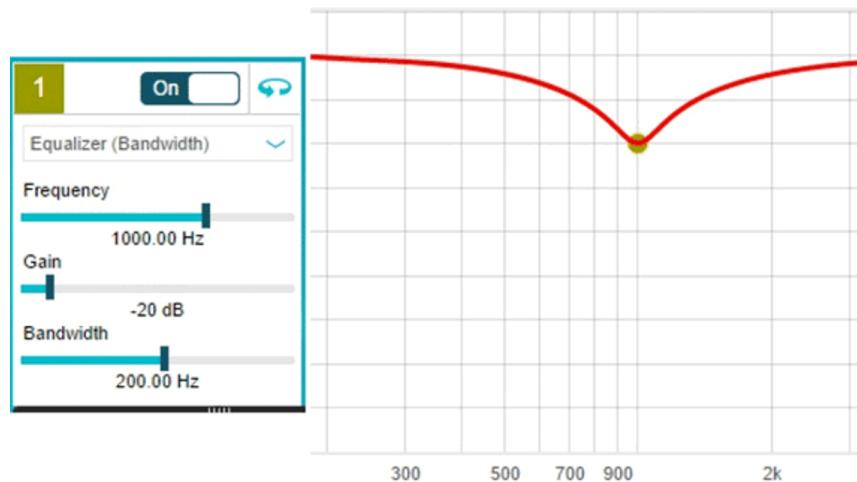


Figure 2-15. Equalizer and Biquad Control

2.12.2.2 Dynamic EQ

Dynamic EQ (DEQ) helps to reduce distortion due to rocking mode/rub-and-buzz, which may occur at a particular frequency by compressing loud signals at that frequency.

The dynamic EQ uses a 128-sample look-ahead, and provides controls for frequency points where dynamic equalization can be activated based on input signal level. Controls for the DEQ include:

- **Center frequency(Hz):** Sets the center frequency of the DEQ Compressor.
- **Q:** Sets the band Q-factor to narrow the bandwidth DEQ compression region
- **Peak Gain Threshold (dB):** Gain Threshold of the limiter below which the gain compressor will be enabled.
- **Peak Gain Compression:** Compression Ratio in the compression region.
- **Peak Limit(dB):** Sets the limit of the output signal at the center frequency.
- **Peak Alpha (ms):** Release time of the DEQ

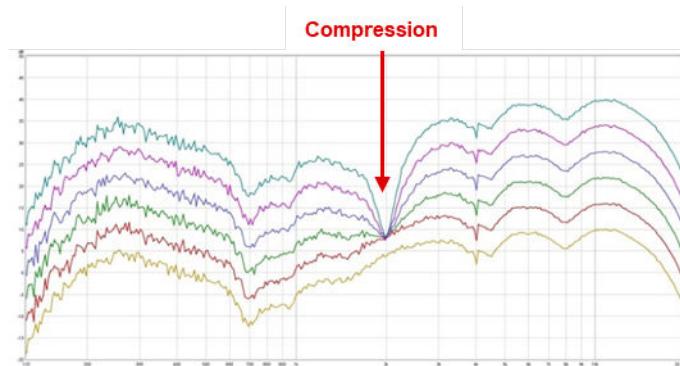


Figure 2-16. Dynamic EQ Example (Single Band DEQ at 2 kHz)

2.12.3 Dynamic Range Control

2.12.3.1 2-band DRC

The smart amp DRC available in the TAS2563 features two distinct bands separated by customizable first- or second-order filter crossovers and also enables configuration of typical compressor parameters such as threshold, ratio, attack, release, cutoff frequencies for each band, and global makeup gain (see [Figure 2-17](#)).

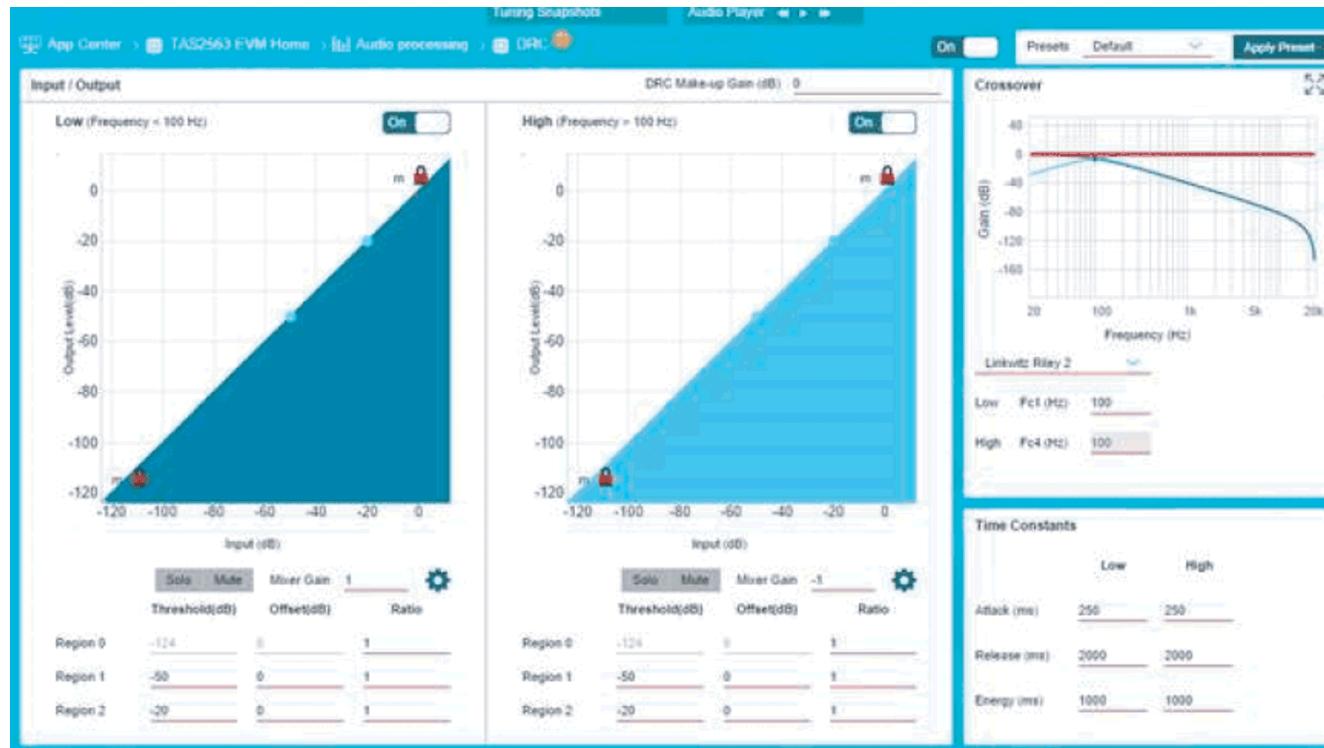


Figure 2-17. 2-Band DRC Control

DRC is a very-powerful audio processing tool which is often used in several segments of the audio industry from professional studio and live environments to small audio electronics. Compression is the process of reducing the dynamic range of an audio signal by attenuating the loud signals and amplifying the soft signals. Compression can help to make audio sound louder without actually increasing the maximum output level of the signal. Because micro-speakers are subject to an excursion limit, but also have a reputation for low SPL output, DRC is a great tool to expand the apparent volume of the speaker sound while helping the protection block to limit the output signal level, and therefore, the excursion.

The typical procedure is to reduce the high-volume amplitude of the signal without affecting the low- volume portions of the signal. When these high-volume peaks are reduced, a makeup gain can be added to then increase the volume of the entire signal. In [Figure 2-18](#), [Figure 2-19](#), and [Figure 2-20](#), signal A is the original, unprocessed audio track which has a few points of peak amplitude that prevent the entire track from being amplified without distortion. When these peaks are amplified past the full-scale value, they can often result in a distorted sound. Signal B shows the result after using a very harsh compression at -6 dB and, as a result, there is no signal content above -6 dB. Signal C then shows the result when signal B is amplified or normalized after the compression stage. Now the signal is at the maximum amplitude for the high-energy (loud) signals, but also the low-energy (quiet) parts of the track are louder than in signal A. If the compression is not too aggressive, the loss of volume in the peaks is negligible and unnoticeable, while the rest of the track sounds noticeably louder.



Figure 2-18. Typical Compression Sequence for Optimal Loudness—DRC Input, No Compression



Figure 2-19. Typical Compression Sequence for Optimal Loudness—DRC Output, Compressed



Figure 2-20. Typical Compression Sequence for Optimal Loudness—DRC Output, Compressed With Makeup Gain

The following are useful terms for compression:

Threshold: The threshold is the input signal level above which the compression engages. Any input signal above this level has a reduced output level. Figure 2-21 shows a compression plot with the threshold at -20 dB. TI recommends a level of -20 dB with a ratio of 4:1; however, this value can result in reduced volume if a makeup gain is not in use. Also note the importance of being able to recognize an “over-compressed” sound. The target audience, and even the end user, are most likely able to recognize this sound, and it is always undesirable. Experiment with an extremely-low threshold and ratio such as -40 dB, 100:1 and become familiar with this sound to avoid it in the future. If the compression is too aggressive (low threshold and high ratio), it sounds over-compressed. If the compression is not aggressive enough (high threshold and low ratio), the output signal feeds into the protection region and sounds harsh.

Ratio: This term refers to the ratio of the input signal to output signal above the threshold. Ratio determines the amount of compression applied to the signal, which is denoted as *input level:output level*. A signal region unaffected by compression has a ratio of 1:1. A typical ratio can be 2:1 to 5:1. A 10:1 ratio or higher is an extreme case of compression known as a “limiter”. A limiter is useful for maximizing the output signal, but is often used after a compressor in the signal chain. The speaker protection region of the smart amp acts more like a limiter than a compressor. Therefore, the DRC is useful to control and limit the dynamics of the signal with more variability and with a softer effect than the protection as to prevent a harsh sound while still maximizing the output level. When the ratio is less than 1, this is called “expansion”, and this increases the gain of low-volume signals while leaving louder signals unaffected. The following [Figure 2-21](#) shows a compression with ratio 2:1 above the threshold. When the input signal level is at 0 dB (full-scale), the output level is at -10 dB.

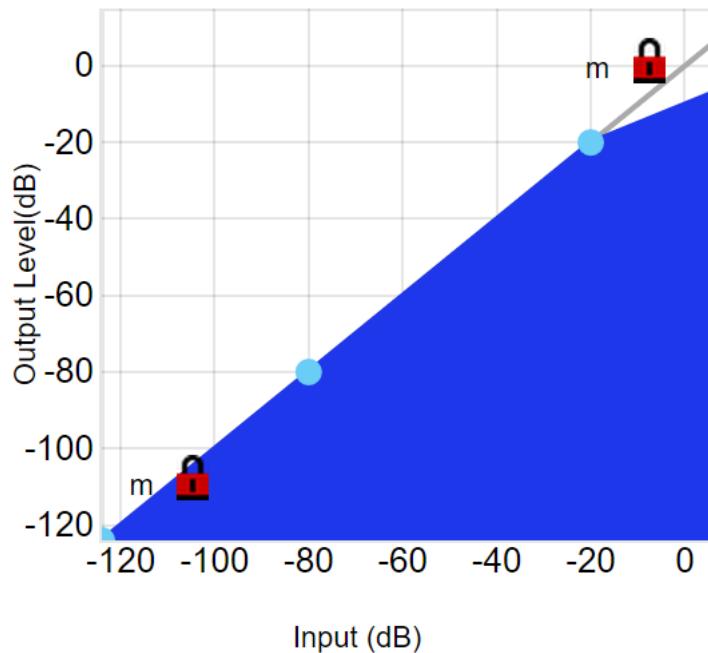


Figure 2-21. DRC Compression Ratio Curve

The following are useful terms for compression:

- **Threshold:**

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- **Ratio:**

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maximizing the output level. When the ratio is less than 1, this is called “expansion” and this increases the gain of low-volume signals while leaving louder signals unaffected. [Figure 2-22](#) shows a compression with ratio 2:1 above the threshold. When the input signal level is at 0 dB (full-scale), the output level is at –10 dB.



Figure 2-22. DRC Time Constant Controls

- Attack, release, and energy time constants:

These controls introduce a degree of control over how quickly the compressor reacts to the input signal. The DRC requires a good estimation of the input signal to function. The DRC is estimated in three steps. The energy time constant calculates the level of the input signal. Next, the input and output relationship is calculated, and finally the DRC control signal is smoothed by the attack and release times. The attack time determines the amount of time to reduce the signal gain after the input signal level has surpassed the threshold level. When the input signal has fallen back below the threshold, the release time determines the time to bring the output signal back up to its original gain. The recommended values for time constants in the smart amp DRC are 10 / flow for the energy, where flow is the low cutoff frequency of the DRC crossover band filter and 0.01 ms for the attack and release time (see [Figure 2-22](#)). For example, if the mid-band-crossover low cutoff is 400 Hz, then set the energy time for the mid-band to 25 ms and the attack and release to 0.01 ms. The user can set the energy for the low-band to 100 ms. The energy time constant, which is not seen in a typical compressor, introduces a delay in the reaction time of the compressor much like attack and release delays. For this reason, the attack and release times are often not required in this DRC, but are available to add additional reaction control.

- Makeup gain:

This gain stage is at the end of the DRC signal chain and is a global gain for the entire audio band. The user may occasionally find benefits in increasing the post-attenuation signal to maximize the output signal and create the apparent volume discussed earlier because one function of the compressor is to attenuate and control high-volume signals to shape and reduce the dynamic range of the whole sound. Making this increase is not always the case in a smart amp design like this because the DRC is followed by the protection stage, which acts like a low-latency limiter. TI recommends to try and balance SPL optimization while also avoiding the protection stage as much as possible. TI also recommends a typical makeup gain of 0 dB to 2 dB. Experiment with balancing the makeup gain with the full-chain input gain from earlier. Sometimes just reducing the input gain and increasing the makeup gain can create a well-balanced, clean, and loud sound. Be sure to note, however, that the makeup stage inherently increases the low-volume output level and therefore, enhances noise.

- Crossover and multi-band compression:

The more primitive compression tools use only a single band for the entire frequency spectrum; however, the smart amp DRC enables a commonly-used professional audio tool called multi-band compression. More bands means more control over the input signal to optimize the sound quality and SPL. Using more bands also allows for high-volume sound shaping in addition to the EQ. For instance, a user can implement a more aggressive high-band compression to reduce certain shrill, high-end sounds in loud input signals or to accentuate the low-end bass region. In the TAS2563 device, the audio signal originating from the EQ block

output is split into three bands using customizable crossover filters. The default values of 400 Hz and 4000 Hz are recommended. However, because the resonance frequency is often in the 700- to 1000-Hz range, another option is to place the low-band cutoff higher than the resonance frequency and use the low-band DRC to control the overexcursion from resonance. The crossover frequencies must always be equal. For example, the low-band cutoff must equal the mid-band low cutoff.

The *DRC* page offers a global enable button. Additionally, each band has an enable button which only bypasses the compression for that band but does not mute the signal from that band. Use the solo and mute buttons below the input and output curve to isolate one band to focus on the compression results. Use the enable buttons to hear the effect of the DRC on the audio quality. [Figure 2-23](#) shows the DRC filter crossover. [Figure 2-24](#) shows the DRC makeup gain control.

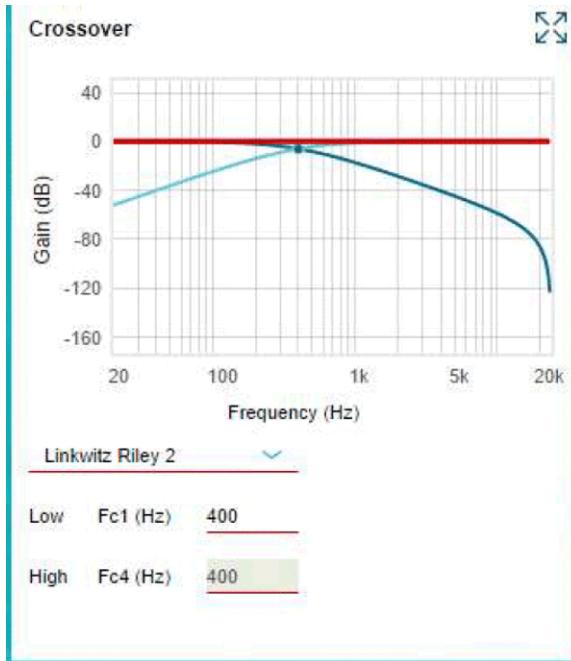


Figure 2-23. DRC Crossover Filter Controls

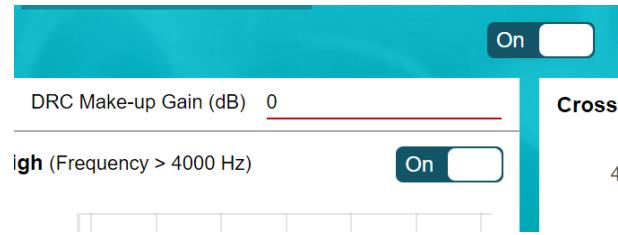


Figure 2-24. DRC Makeup Gain

- **Ganged DRC:**

For stereo applications, where the two-speaker responses are identical, TI usually recommends to gang the DRC for channels A and B in dual mono mode. This ganged DRC configuration indicates that when the input signal level of either channel reaches the threshold value, both channel gains are reduced simultaneously, which enhances the sensation of stereo sound. In some audio tracks, one side of the stereo image is much louder due to panning. Without ganging the two channels, the DRC only reduces the loud side and leaves the quiet side unmodified, which can create unwanted artifacts in the stereo image and disorientation in the listener. If the two speakers are not identical and have very different responses, ganged DRC can also cause negative artifacts in the sound and is usually not recommended. Find the gang feature at the top-right of the *DRC* page.

- **Noise gate:**

Another useful feature of the DRC and a common tool in professional audio is the noise gate. The purpose of the noise gate is to reduce unwanted, low-volume noise by attenuating any signal below a set noise gate threshold. The gain curve in [Figure 2-25](#) shows the effective response of the noise gate and also shows another way to implement the noise gate. The noise gate can help to create the effective sound of a higher-dynamic range by reducing the small-signal response. Experiment with a solo drum track and make an extreme noise gate of a -30-dB threshold and -100-dB attenuation to understand the effect of this feature.

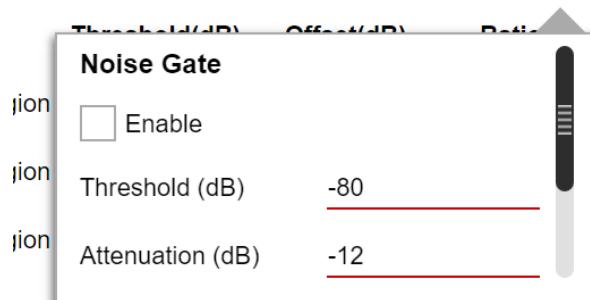


Figure 2-25. Automatic Noise Gate Control

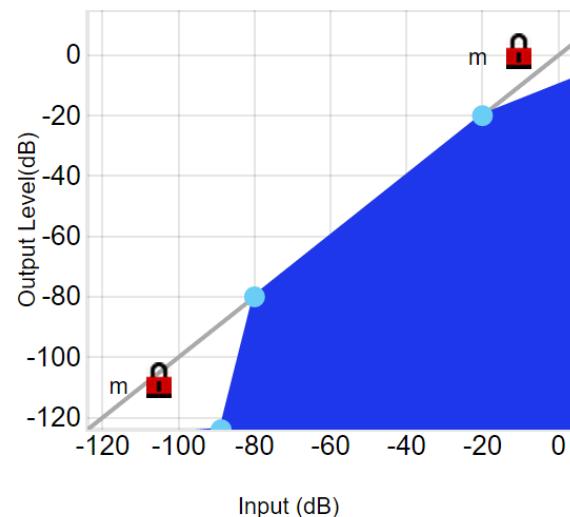


Figure 2-26. Manual Noise Gate Setting

2.12.3.2 DRC Test

In the DRC example data seen in [Figure 2-27](#), the pink noise FFT plots of the DRC output can be seen.

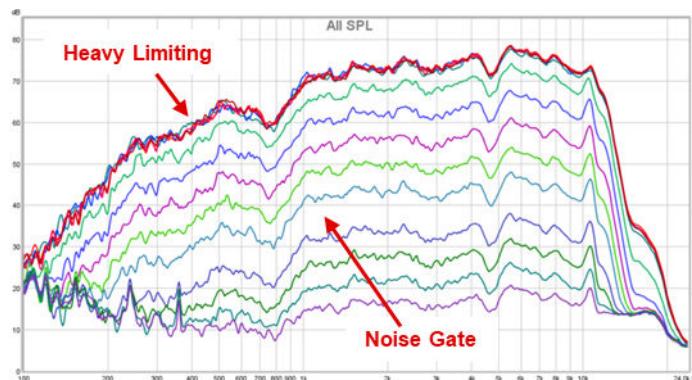
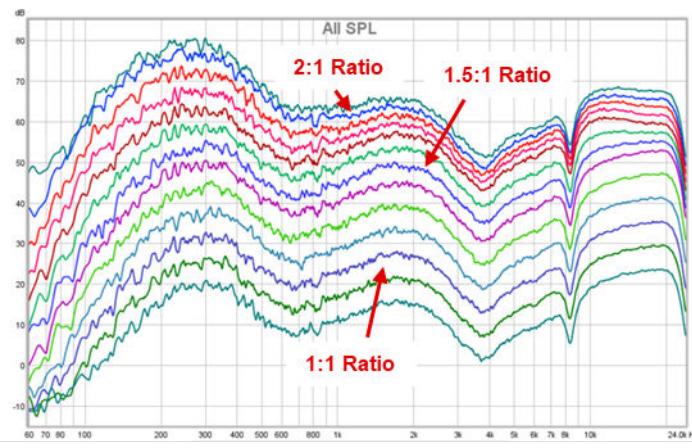


Figure 2-27. DRC Example: Pink Noise Family with DRC Compression and Noise Gate



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Figure 2-28. DRC Ratio Comparison

2.12.4 Smart Amp Speaker Protection

Protection tuning controls the excursion and temperature protection for each band. The speaker protection algorithm uses three bands: low, mid and high. The low-, mid-, and high-frequency limits are adjustable in the *Protection Tuning* tile (see Figure 3-29). Each band has a *Thermal Priority* setting and a *Thermal Speed* setting. Values are automatically filled in for all protection tuning parameters based on the speaker characterization process. The band limits are configurable, either through dragging the vertical red (mid) or green (high) vertical line in the protection tuning graph or by typing in the respective *Frequency (Hz)* fields.

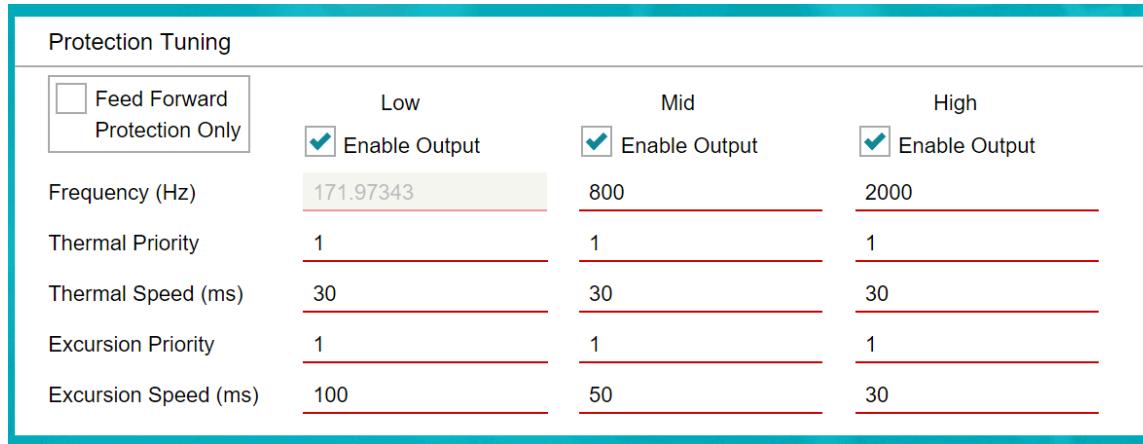


Figure 2-29. Protection Controls

The priorities (for each band) determine the relative attenuation for each band for both excursion and temperature protection. The lower the value relative to the other bands, the higher the attenuation for each band during protection. The valid range for priority is 0.01 to 10. If the priority is the same for each band, the protection attenuation is the same for each band (flat).

Tuning for protection is an iterative process that requires listening to a song at high volume while tweaking the parameters for each band. For optimal bass performance, choose a higher priority for the low band as compared to the mid and high bands—for example, 6 versus 1 and 1 (see Figure 3-30). Tweak the

frequency band limits and priorities to achieve the desired acoustic balance between all three bands while protection is engaged.

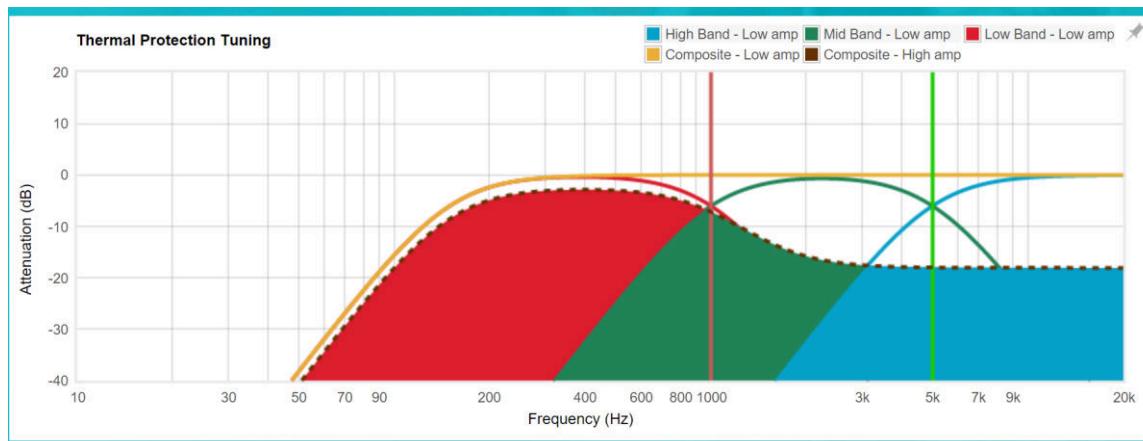


Figure 2-30. Thermal Protection Priority

The speed values for both thermal and excursion determine the attenuation release speed. Lower values deliver more power to the speaker because the attenuation is released quickly if protection is not necessary; however,

this can lead to pulsating artifacts if another excursion or temperature violation is detected in fast succession. Choose the shortest value that does not produce audible artifacts.

The user can disable each band check box for debugging persistent sound artifacts (distortion or pulsating artifacts). For example, if a low-frequency artifact is present, disable the mid and high bands and tweak the priority and speed settings for the low band until the artifact is removed.

If you want to disable the protection, you can set the ThermalLimit=1000 degC and Xmax=10mm.

Characterization Data	
BI	: 0.814 Tm
Mms	: 0.0666 g
Cms	: 0.479 mm/N
Rms	: 0.0812 Ns/m
Xmax (mm)	10
ThermalLimit (Δ°C)	1000

Figure 2-31. Disable Protection

2.13 PDM MIC

TAS2563 also supports 2 pcs PDM MIC recording and has 2 simple digital gains on the input signals.

The Maxim gain range is 30 dB (see [Figure 2-32](#)).

If you need this PDM MIC feature, you need a special version PPC3 Tool to support this feature.

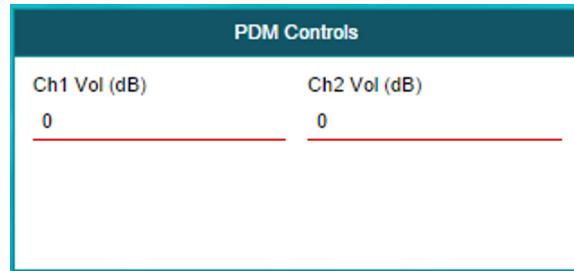


Figure 2-32. PDM Controls

The System Block Diagram is like [Figure 2-33](#). The Host Provide BCLK, WCLK, DIN to TAS2563 while TAS2563 transmit the PDM1 data, PDM2 data and Echo Reference to Host.

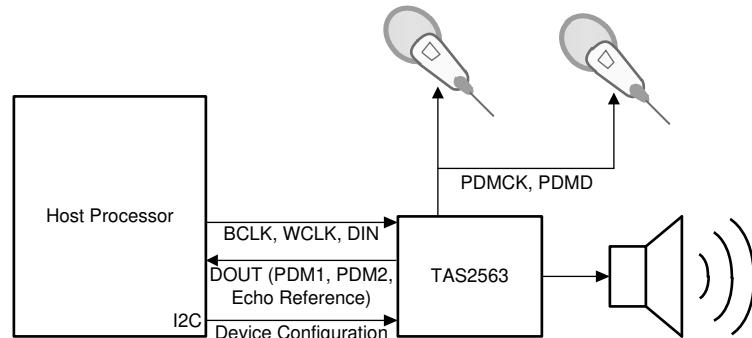


Figure 2-33. Smart Amp Tuning

[Figure 2-34](#) shows the speaker audio signal chain through the TAS2563 if you use PDM Mic version PPC3.

There are 3 bands EQ in the signal chain.

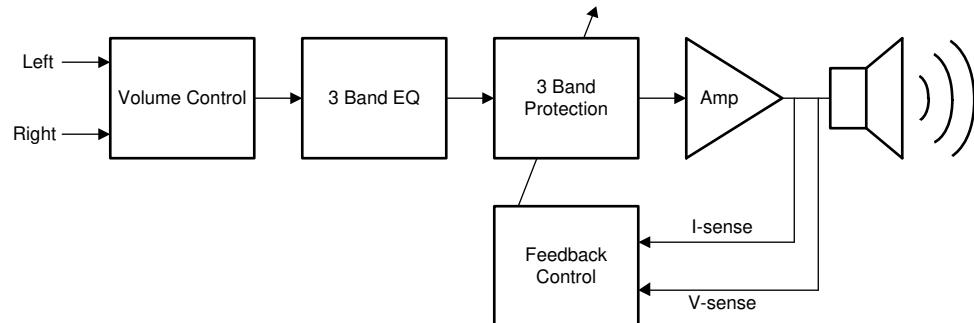


Figure 2-34. Speaker Audio Signal Chain

3 Related Documentation

- Texas Instruments, [*Smart Amp Tuning Guide Application Report*](#)
- Texas Instruments, [*TAS2563 6.1-W Boosted Class-D Audio Amplifier With Integrated DSP And IV Sense Data Sheet*](#)

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