ABSTRACT

The TAS5825M device has a powerful μCDSP audio processing core, which supports several selectable process flows. This application report explains the details of each process flow.

The TAS5825M process flows with standard processing, feature a few advanced audio processing blocks: Dynamic Ranger Control (DRC), Automatic Gain Limiter (AGL), Dynamic Parametric Equalizer (DPEQ) and Spatializer. A 3-band DRC + AGL structure limits the output power of the amplifier for three regions while controlling the peaking that can occur in the crossover region during compression. DPEQ dynamically adjusts the equalization curve that is applied to low-level signal and the curve that is applied to high-level signals. The Spatializer increases the field of sound for a broader and more encompassing audio experience.

The TAS5825M process flows with SmartAmp Processing replaces traditional continuous power design principles and hardware-based speaker protection methods with algorithms that allow significant increases in peak power output, loudness, and sound quality relative to conventional amplifiers. Smart Amp tools allow developers to understand how speakers are performing in the system and then make informed decisions to improve performance. The algorithms, characterization, and tuning tools allow developers to overcome a wide variety of audio challenges.
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## Trademarks

PurePath is a trademark of Texas Instruments.
1 General Overview

1.1 Supported Use Cases

The TAS5825M process flows have been generated based upon several popular configurations, primarily around the number and type of amplified outputs. Table 1 shows the use cases supported by available process flows and the PPC3 GUI.

Table 1. Supported Use Cases

<table>
<thead>
<tr>
<th>Mode</th>
<th>Also Known As</th>
<th>Amplifier Output Configuration</th>
<th>Symbol in PPC3 GUI</th>
</tr>
</thead>
<tbody>
<tr>
<td>2.0</td>
<td>Stereo</td>
<td>One device drives two full-range speakers in stereo.</td>
<td><img src="image" alt="Symbol" /></td>
</tr>
<tr>
<td>2.2</td>
<td>Dual stereo</td>
<td>Two devices drive two-way speakers in stereo. One device drives two tweeters and one device drives two woofers.</td>
<td><img src="image" alt="Symbol" /></td>
</tr>
<tr>
<td>1.1</td>
<td>Bi-amped, dual mono</td>
<td>A single input signal is separated into high- and low-frequency content. One BTL output drives a high-frequency transducer and the other drives a low-frequency transducer.</td>
<td><img src="image" alt="Symbol" /></td>
</tr>
<tr>
<td>2.1</td>
<td>N/A</td>
<td>One device uses 2.0 mode and a separate device uses mono mode.</td>
<td><img src="image" alt="Symbol" /></td>
</tr>
</tbody>
</table>
Table 2 and Table 3 show the processing features of each process flow available in the current PPC3 GUI.

Table 2. Process Flows 1–4

<table>
<thead>
<tr>
<th>Feature</th>
<th>Process Flow 1 (Base and Pro, 96 kHz, 2.0)</th>
<th>Process Flow 2 (2-Band DRC and AGL, 96 kHz, 2.0)</th>
<th>Process Flow 3 (3-Band DRC and AGL, 96 kHz, 2.0)</th>
<th>Process Flow 4 (SmartAMP, 96 kHz, 2.0)</th>
</tr>
</thead>
<tbody>
<tr>
<td>Maximum Internal Sample Rate</td>
<td>96 kHz</td>
<td>96 kHz</td>
<td>96 kHz</td>
<td>96 kHz</td>
</tr>
<tr>
<td>SRC and Auto-detect</td>
<td>√</td>
<td>√</td>
<td>√</td>
<td>√</td>
</tr>
<tr>
<td>Supported Input Sample Rates (32 kHz, 44.1 kHz, 48 kHz, 88.2 kHz, and 96 kHz)</td>
<td>√</td>
<td>√</td>
<td>√</td>
<td>√</td>
</tr>
<tr>
<td>Support for Input Sample Rate 192 kHz</td>
<td>√</td>
<td>√</td>
<td>×</td>
<td>√</td>
</tr>
<tr>
<td>Biquads for EQ Filtering (Individual Left and Right)</td>
<td>15</td>
<td>15</td>
<td>15</td>
<td>15</td>
</tr>
<tr>
<td>Additional Biquad Bank (44.1 kHz, 88.2 kHz)</td>
<td>√</td>
<td>√</td>
<td>×</td>
<td>×</td>
</tr>
<tr>
<td>Input Mixer</td>
<td>√</td>
<td>√</td>
<td>√</td>
<td>√</td>
</tr>
<tr>
<td>Click and Pop Free Volume</td>
<td>√</td>
<td>√</td>
<td>√</td>
<td>√</td>
</tr>
<tr>
<td>Spatializer (Stereo Widening)</td>
<td>√</td>
<td>×</td>
<td>×</td>
<td>×</td>
</tr>
<tr>
<td>Dynamic Biquad</td>
<td>2nd order</td>
<td>4th order</td>
<td>2nd order</td>
<td>2nd order</td>
</tr>
<tr>
<td>DRC</td>
<td>3-band 4th order crossover</td>
<td>2-band 2nd order crossover</td>
<td>3-band 2nd order crossover</td>
<td>×</td>
</tr>
<tr>
<td>Automatic Gain Limiter (AGL)</td>
<td>×</td>
<td>√</td>
<td>√</td>
<td>√</td>
</tr>
<tr>
<td>Smart Excursion, Smart Thermal, and Smart Bass</td>
<td>×</td>
<td>×</td>
<td>×</td>
<td>√</td>
</tr>
<tr>
<td>SmartEQ</td>
<td>×</td>
<td>×</td>
<td>×</td>
<td>√</td>
</tr>
<tr>
<td>Output Clipper</td>
<td>√</td>
<td>√</td>
<td>√</td>
<td>√</td>
</tr>
</tbody>
</table>

Table 3. Process Flows 5–6

<table>
<thead>
<tr>
<th>Feature</th>
<th>Process Flow 5 (Base and Pro, 48 kHz, 2.1)</th>
<th>Process Flow 6 (SmartAMP, 48 kHz, 2.1)</th>
</tr>
</thead>
<tbody>
<tr>
<td>Maximum Internal Sample Rate</td>
<td>48 kHz</td>
<td>48 kHz</td>
</tr>
<tr>
<td>SRC and Auto-detect</td>
<td>√</td>
<td>√</td>
</tr>
<tr>
<td>Supported Input Sample Rates (16 kHz, 32 kHz, 44.1 kHz, and 48 kHz, 88.2 kHz, 96 kHz)</td>
<td>√</td>
<td>√</td>
</tr>
<tr>
<td>Biquads for EQ Filtering (Individual Left and Right)</td>
<td>15</td>
<td>15</td>
</tr>
<tr>
<td>Input Mixer</td>
<td>√</td>
<td>√</td>
</tr>
<tr>
<td>Click and Pop Free Volume</td>
<td>√</td>
<td>√</td>
</tr>
<tr>
<td>Spatializer (Stereo Widening)</td>
<td>√</td>
<td>√</td>
</tr>
<tr>
<td>Dynamic Biquad</td>
<td>4th order</td>
<td>4th order</td>
</tr>
<tr>
<td>DRC</td>
<td>3-band 4th order crossover</td>
<td>2-band 4th order crossover</td>
</tr>
<tr>
<td>Automatic Gain Limiter (AGL)</td>
<td>√</td>
<td>×</td>
</tr>
<tr>
<td>Smart Excursion, Smart Thermal, and Smart Bass</td>
<td>×</td>
<td>√</td>
</tr>
<tr>
<td>SmartEQ</td>
<td>×</td>
<td>×</td>
</tr>
<tr>
<td>Output Clipper</td>
<td>√</td>
<td>√</td>
</tr>
</tbody>
</table>
3 Process Flow 1 (96 kHz, 2.0 Base and Pro)

This process flow supports an internal sample rate of 96 kHz and is therefore considered “true” 96 kHz. It is intended for stereo speakers where the 3-band DRC will use individual coefficients for left and right. It is possible to tune the left and right Biquads (BQs) in the 15-BQ bank individually between left and right.

Figure 1 depicts the signal path of this flow. The blocks in Figure 1 correspond to the functions found in the PPC3 GUI.

Figure 1. Process Flow 1

3.1 SRC

The Sample Rate Converter (SRC) supports 32 kHz, 44.1 kHz, 48 kHz, 88.2 kHz, and 96 kHz input sample rates. These input sample rates are converted to an 88.2 or 96 kHz sample rate.

3.2 Input Mixer

The input mixer is used to mix the left and right channel input signals. Refer to Section 9.1 for more details.

3.3 Equalizer

The equalizer contains 15 independent filters for both left and right channels. Refer to Section 9.2 for more details.

3.4 Volume

This volume block is click and pop free. Refer to Section 9.3 for more details.

3.5 Spatializer

Spatializer is a method to increase the field of sound for a broader and more encompassing audio experience. Refer to Section 9.4 for more details.

3.6 DPEQ

The dynamic parametric equalizer is used to mix the audio signals through two signal paths (low level and high level). These two paths are used with separate equalization properties. A third path monitors the incoming audio and determines the thresholds and mixing characteristics between these two paths. Thus, the mix between the two high- and low-level channels is dynamic in nature and depends on the incoming audio. Refer to Section 9.5 for more details.
3.7 3-Band DRC

The 3-Band DRC can be used to automatically control the audio signal amplitude or the dynamic range within specified limits. Refer to Section 9.6 for more details.

3.8 Clipper

A THD boost and fine volume together can be used for clipping. The THD boost block allows the user to programmatically increase the THD by clipping at an operating point earlier than that defined by the supply rails. Refer to Section 9.11 for more details.

3.9 Crossbar

The crossbar provides the end user with a flexible way to control what finally appears on amplifier outputs and I2S SDOUT. Refer to Section 9.12 for more details.

3.10 Level Meter

The level meter provides the end user with an easy way to study the power profile. Refer to Section 9.13 for more details.
4 Process Flow 2 (96 kHz, 2.0 2-Band DRC and AGL)

This process flow supports an internal sample rate of 96 kHz and is therefore considered a “true” 96 kHz flow. This process flow is similar to Process Flow 1. The differences are: (1) the 3-Band DRC and Spatializer are replaced by 2-Band DRC and AGL, (2) the 2\textsuperscript{nd} order DPEQ becomes a 4\textsuperscript{th} order DPEQ.

Figure 2 depicts the signal path of this process flow. The blocks in Figure 2 correspond to the functions found in the PPC3 GUI.

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4.1 SRC

The Sample Rate Converter (SRC) supports 32 kHz, 44.1 kHz, 48 kHz, 88.2 kHz, and 96 kHz input sample rates. These input sample rates are converted to an 88.2 or 96 kHz sample rate.

4.2 Input Mixer

The input mixer is used to mix the left and right channel input signals. Refer to Section 9.1 for more details.

4.3 Equalizer

The equalizer contains 15 independent filters for both left and right channels. Refer to Section 9.2 for more details.

4.4 Volume

This volume block is click and pop free. Refer to Section 9.3 for more details.

4.5 DPEQ

The dynamic parametric equalizer is used to mix the audio signals through two signal paths (low level and high level). These two paths are used with separate equalization properties. A third path monitors the incoming audio and determines the thresholds and mixing characteristics between these two paths. Thus, the mix between the two high- and low-level channels is dynamic in nature and depends on the incoming audio. Refer to Section 9.5 for more details.

4.6 2-Band DRC

The 2-Band DRC can be used to automatically control the audio signal amplitude or the dynamic range within specified limits. Refer to Section 9.6 for more details.
4.7 AGL

The AGL can also be used to automatically control the audio signal amplitude or dynamic range within specified limits. Refer to Section 9.8 for more details.

4.8 Clipper

A THD boost and fine volume together can be used for clipping. The THD boost block allows the user to programmatically increase the THD by clipping at an operating point earlier than that defined by the supply rails. Refer to Section 9.11 for more details.

4.9 Crossbar

The crossbar provides the end user with a flexible way to control what finally appears on amplifier outputs and I2S SDOUT. Refer to Section 9.12 for more details.

4.10 Level Meter

The level meter provides the end user with an easy way to study the power profile. Refer to Section 9.13 for more details.
5 Process Flow 3 (96 kHz, 2.0 3-Band DRC and AGL)

This process flow supports an internal sample rate of 96 kHz and is therefore considered a “true” 96 kHz flow. This process flow is similar to Process Flow 1. The differences are: (1) the Spatializer is replaced by an AGL; b) the 4th order 3-Band DRC becomes a 2nd order 3-Band DRC.

Figure 3 depicts the signal path of this process flow. The blocks in Figure 3 correspond to the functions found in the PPC3 GUI.

![Figure 3. Process Flow 3](image)

5.1 SRC

The Sample Rate Converter (SRC) supports 32 kHz, 44.1 kHz, 48 kHz, 88.2 kHz, and 96 kHz input sample rates. These input sample rates are converted to an 88.2 or 96 kHz sample rate.

5.2 Input Mixer

The input mixer is used to mix the left and right channel input signals. Refer to Section 9.1 for more details.

5.3 Equalizer

The equalizer contains 15 independent filters for both left and right channels. Refer to Section 9.2 for more details.

5.4 Volume

This volume block is click and pop free. Refer to Section 9.3 for more details.

5.5 DPEQ

The dynamic parametric equalizer is used to mix the audio signals through two signal paths (low level and high level). These two paths are used with separate equalization properties. A third path monitors the incoming audio and determines the thresholds and mixing characteristics between these two paths. Thus, the mix between the two high- and low-level channels is dynamic in nature and depends on the incoming audio. Refer to Section 9.5 for more details.

5.6 3-Band DRC

The 3-Band DRC can be used to automatically control the audio signal amplitude or the dynamic range within specified limits. Refer to Section 9.6 for more details.
5.7 **AGL**

The AGL can also be used to automatically control the audio signal amplitude or dynamic range within specified limits. Refer to Section 9.8 for more details.

5.8 **Clipper**

A THD boost and fine volume together can be used for clipping. The THD boost block allows the user to programmatically increase the THD by clipping at an operating point earlier than that defined by the supply rails. Refer to Section 9.11 for more details.

5.9 **Crossbar**

The crossbar provides the end user with a very flexible way to control what finally appears on amplifier outputs and I2S SDOUT. Refer to Section 9.12 for more details.

5.10 **Level Meter**

The level meter provides the end user with an easy way to study the power profile. Refer to Section 9.13 for more details.
6 Process Flow 4 (96 kHz, 2.0 SmartAmp Processing)

This process flow supports an internal sample rate of 96 kHz and is therefore considered a “true” 96 kHz flow. This process flow enables SmartAmp processing with the three components: SmartBass with morphing, Excursion Limiter, and Thermal Limiter.

Figure 4 depicts the signal path of this flow. The blocks in Figure 4 correspond to the functions found in the PPC3 GUI.

6.1 Input Mixer

The input mixer is used to mix the left and right channel input signals. Refer to Section 9.1 for more details.

6.2 Equalizer

The equalizer contains 15 independent filters for both left and right channels. Refer to Section 9.2 for more details.

6.3 Volume

This volume block is click and pop free. Refer to Section 9.3 for more details.

6.4 Smart Bass, Excursion Limiter and Thermal Limiter

Smart Bass is an intelligent True Bass Alignment algorithm. Smart Bass uses the combination of the speaker model and a desired target response selected by the user to equalize the speaker in the bass region. This target response is critical for the sound character and the user can apply the same target response to very different speakers and get the same sound. Refer to Section 10 to Section 12 for more details.

Based on mechanical, electrical, and acoustical properties of speakers, Excursion Limiter and Thermal Limiter can predict potentially damaging situations, take timely precautions, and therefore, protect speakers from over-excursion and overheating.

6.5 Clipper

The clipper allows the user to programmatically increase the THD by clipping at an operating point earlier than that defined by the supply rails. Refer to Section 9.11 for more details.
6.6 **Crossbar**

The crossbar provides the end user with a very flexible way to control what finally appears on amplifier outputs and I2S SDOUT. Refer to Section 9.12 for more details.

6.7 **Level Meter**

The level meter provides the end user with an easy way to study the power profile. Refer to Section 9.13 for more details.
7  **Process Flow 5 (48 kHz, 2.1 Base and Pro)**

This process flow supports an internal sample rate of 48 kHz. It can accept both 48 and 96 kHz input sample rate but will down-sample the 96 kHz to 48 kHz with a 2× decimator.

**Figure 5** depicts the signal path of this flow. The blocks in **Figure 5** correspond to the functions found in the PPC3 GUI.

**Figure 5. Process Flow 5**

### 7.1 Input Mixer

The input mixer is used to mix the left and right channel input signals. Refer to **Section 9.1** for more details.

### 7.2 Equalizer

The equalizer contains 15 independent filters for both left and right channels. Refer to **Section 9.2** for more details.

### 7.3 Volume

This volume block is click and pop free. Refer to **Section 9.3** for more details.

### 7.4 Spatializer

Spatializer is a method to increase the field of sound for a broader and more encompassing audio experience. Refer to **Section 9.4** for more details.

### 7.5 DPEQ

The dynamic parametric equalizer is used to mix the audio signals through two signal paths (low level and high level). These two paths are used with separate equalization properties. A third path monitors the incoming audio and determines the thresholds and mixing characteristics between these two paths. Thus, the mix between the two high- and low-level channels is dynamic in nature and depends on the incoming audio. Refer to **Section 9.5** for more details.

### 7.6 Crossover

The crossover block is used to set low pass filters on the woofer and high pass filters on the tweeter. Refer to **Section 9.9** for more details.

### 7.7 Phase Optimizer

The phase optimizer allows time aligning the 2.0 (tweeter) path with the 0.1 (woofer) path. Refer to **Section 9.10** for more details.
7.8  **3-Band DRC**

The 3-Band DRC can be used to automatically control the audio signal amplitude or the dynamic range within specified limits. Refer to Section 9.6 for more details.

7.9  **AGL**

The AGL can also be used to automatically control the audio signal amplitude or dynamic range within specified limits. Refer to Section 9.8 for more details.

7.10  **Clipper**

The clipper allows the user to programmatically increase the THD by clipping at an operating point earlier than that defined by the supply rails. Refer to Section 9.11 for more details.

7.11  **Crossbar**

The crossbar provides the end user with a flexible way to control what finally appears on amplifier outputs and I2S SDOUT. Refer to Section 9.12 for more details.

7.12  **Level Meter**

The level meter provides the end user with an easy way to study the power profile. Refer to Section 9.13 for more details.
8 Process Flow 6 (48 kHz, 2.1 SmartAmp Processing)

The process flow is similar to Process Flow 5. The difference is the 3-Band DRC and AGL is removed to free up processing resources for these three components: SmartBass with morphing, Excursion Limiter and Thermal Limiter.

Figure 6 depicts the signal path of this flow. The blocks in Figure 5 correspond to the functions found in the PPC3 GUI.

Figure 6. Process Flow 6

8.1 Input Mixer

The input mixer is used to mix the left and right channel input signals. Refer to Section 9.1 for more details.

8.2 Equalizer

The equalizer contains 15 independent filters for both left and right channels. Refer to Section 9.2 for more details.

8.3 Volume

This volume block is click and pop free. Refer to Section 9.3 for more details.

8.4 Spatializer

Spatializer is a method to increase the field of sound for a broader and more encompassing audio experience. Refer to Section 9.4 for more details.

8.5 DPEQ

The dynamic parametric equalizer is used to mix the audio signals through two signal paths (low level and high level). These two paths are used with separate equalization properties. A third path monitors the incoming audio and determines the thresholds and mixing characteristics between these two paths. Thus, the mix between the two high- and low-level channels is dynamic in nature and depends on the incoming audio. Refer to Section 9.5 for more details.

8.6 Crossover

The crossover block is used to set low pass filters on the woofer and high pass filters on the tweeter. Refer to Section 9.9 for more details.
8.7 **Phase Optimizer**

The phase optimizer allows time aligning the 2.0 (tweeter) path with the 0.1 (woofer) path. Refer to Section 9.9 for more details.

8.8 **Smart Bass, Excursion Limiter and Thermal Limiter**

**Smart Bass** is an intelligent True Bass Alignment algorithm. **Smart Bass** uses the combination of the speaker model and a desired target response selected by the user to equalize the speaker in the bass region. This target response is critical for the sound character and the user can apply the same target response to very different speakers and get the same sound. Refer to Section 10 to Section 12 for more details.

Based on mechanical, electrical, and acoustical properties of speakers, **Excursion Limiter** and **Thermal Limiter** can predict potentially damaging situations, take timely precautions, and therefore, protect speakers from over-exursion and overheating.

8.9 **2-Band DRC**

The 2-Band DRC can be used to automatically control the audio signal amplitude or the dynamic range within specified limits. Refer to Section 9.6 for more details.

8.10 **Clipper**

The clipper allows the user to programmatically increase the THD by clipping at an operating point earlier than that defined by the supply rails. Refer to Section 9.11 for more details.

8.11 **Output Crossbar**

The crossbar provides the end user with a flexible way to control what finally appears on amplifier outputs and I2S SDOUT. Refer to Section 9.12 for more details.

8.12 **Level Meter**

The level meter provides the end user with an easy way to study the power profile. Refer to Section 9.13 for more details.
9 Audio Processing Blocks

9.1 Input Mixer

The input mixer can be used to mix the left and right channel input signals as shown in Figure 7. The input mixer has four coefficients, which control the mixing and gains of the input signals.

![Figure 7. Input Mixer](image)

The Basic tab (see Figure 8) provides the easiest method for configuration in PPC3 GUI.

![Figure 8. Input Mixer (Basic Tab)](image)
Switch to the Advanced tab (see Figure 9) if all the four coefficients need to be adjusted. Note that the four parameters need to be specified in decibels (dB). The Invert options will reverse the sign of the gain values.

9.2 Equalizer

The equalizers are implemented using cascaded “direct form 1” BQs structures as shown in Figure 10.

\[
H(z) = \frac{b_0 + b_1 z^{-1} + b_2 z^{-2}}{a_0 + a_1 z^{-1} + a_2 z^{-2}}
\]  

(1)

All BQ coefficients are normalized with \(a_0\) to ensure that \(a_0\) is equal to 1. The structure requires 5 BQ coefficients as shown in Table 4. Any BQ with coefficients greater than 1 undergoes gain scaling.

<table>
<thead>
<tr>
<th>BQ Coefficient</th>
<th>Coefficient Calculation</th>
</tr>
</thead>
<tbody>
<tr>
<td>B0_DSP</td>
<td>(b_0 / a_0)</td>
</tr>
<tr>
<td>B1_DSP</td>
<td>(b_1 / (a_0 \times 2))</td>
</tr>
<tr>
<td>B2_DSP</td>
<td>(b_2 / a_0)</td>
</tr>
<tr>
<td>A1_DSP</td>
<td>(-a_1 / (a_0 \times 2))</td>
</tr>
<tr>
<td>A2_DSP</td>
<td>(-a_2 / a_0)</td>
</tr>
</tbody>
</table>
The *Equalizer Tuning* window contains 15 independent filters for both left and right channels. They are designed for tuning the frequency response of the overall system. This is where the bulk of the frequency compensation occurs. Complex tuning shapes can be made to compensate for deficiencies in speaker response.

As Figure 11 shows, each filter has quite a few different filter types and can be turned on or off independently. All the changes to these filters are reflected in Figure 11. The composite plot (red) shows the overall frequency response alteration applied to the incoming digital audio data. *Phase*, *Group Delay*, *Impulse Response* and *Pole zero* charts are also available on the right side.

![Figure 11. Equalizer Tuning Window](image)

The equalizers for left and right channels are *ganged* by default, but they can be configured independently by deseleting the *Gang* option.
9.3 Volume

Figure 12 shows the default volume in PPC3 GUI. Note that volume needs to be specified in decibels (dB). Independent volume change for the left and right channels is achieved by deselecting the Gang option.

![Volume](image1)

**Figure 12. Volume**

The volume block is implemented using an alpha filter structure. As Figure 13 shows, when a volume level change is initiated, the volume block will assure a smooth transition to the newly commanded volume level without producing artifacts such as pops and clicks.

![Volume Attack and Decay](image2)

**Figure 13. Volume Attack and Decay**
9.4 Spatializer

Spatializer is a method to increase the field of sound for a broader and more encompassing audio experience. Here, copies of the left and right channels are subtracted from each other. This creates a signal that removes any audio or instrumentation that is shared by both channels. Next, a bandpass filter sets the frequency range for which the effect is active. After which, a level control adjusts the strength of this channel before being reintroduced back into the original left and right channels.

![Figure 14. Spatializer Block Diagram](image)

It is generally not recommended extending the bandpass filter below 300 Hz, since low-frequency content often presents itself in both channels. Extending the bandpass too low results in a loss of bass response. Similarly, extending the bandpass too high can create effects similar to reverb which can blur the spatial cues of music.

In the Spatializer Tuning window (see Figure 15), the pass band can be set as well as the Level which controls the level of the effect.

![Figure 15. Spatializer Tuning Window](image)
For a given piece of end equipment, it may be helpful to create three *presets* from which to choose. This provides the option of choosing the preferred type of spatializing effect. The three settings can vary both the HPF, LPF, and effect intensity and their settings stored in the system processor to be updated upon a button press from the end user.

Three recommended *presets* are available in the GUI:

### Table 5. Recommended GUI Presets

<table>
<thead>
<tr>
<th>Preset</th>
<th>Frequency Range</th>
<th>Level</th>
<th>Comment</th>
</tr>
</thead>
<tbody>
<tr>
<td>Full</td>
<td>300 Hz to 20 kHz</td>
<td>0.75</td>
<td>Reverberant sounding</td>
</tr>
<tr>
<td>Medium</td>
<td>800 Hz to 6 kHz</td>
<td>0.5</td>
<td></td>
</tr>
<tr>
<td>Low</td>
<td>4 kHz to 20 kHz</td>
<td>0.25</td>
<td>Works well in systems with a flat frequency response up to 16-18 kHz.</td>
</tr>
</tbody>
</table>

#### 9.5 DPEQ

The dynamic parametric equalizer mixes the audio signals routed through two paths containing 1 or 2 Biquads each, based upon the signal level detected by the sense path, as shown in Figure 16.

![Figure 16. DPEQ](image-url)
DPEQ

_Energy (ms)_ simply tells the algorithm for how long to average the samples of audio before it determines how it compares to the mixing thresholds. The shorter the time, the faster the mixer reacts to changes in the input signal level. The longer the time, the slower the mixer reacts to changes in level.

The mixing of the two paths (low level and high level) is controlled by setting _Threshold Low (dB)_ and _Threshold High (dB)_. When the averaged signal (as set by the _Energy_) is below the _Threshold Low_, the dynamic mixer sends all of the audio through the low-level path. When the signal is above the _Threshold High_, it is sent through the upper-level path. When the signal is between the two, it is mixed together by the dynamic mixer level.

**Energy Sense**

The sense path contains 1 or 2 configurable Biquads, which can be used to focus the DEQ sensing on a specific frequency bandwidth.

**Low Level EQ**

The low-level path also has 1 or 2 configurable Biquads to establish the EQ curve the audio is sent through when the time average signal is at a low-level. This fully-functional Biquad can be assigned to several filter types. This determines frequency response when low-level is active based on the _Energy_ configuration and the mixing thresholds.

**High Level EQ**

The high-level path, similar to the low-level path, has 1 or 2 Biquads that can set the EQ curve used when the time averaged input signal is above the upper mixing threshold.
9.6 3-Band DRC

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 level using an estimate of the alpha filter energy then adjusting the gain based on the region and slope parameters that are defined. The 3-Band DRC is shown in Figure 18.

The DRC works to reduce the peak of energy if it goes beyond the programmable threshold level. DRC starts an attack event (reduces gain) if energy goes above the threshold. Similarly, it starts a release event if the level goes below the threshold (increases gain back to the original value). Attack and release events occur only when the level remains above or below the threshold continuously during the time-constant time. And the constant time is controlled by the attack and release rate. If the attack or release rate is short, DRC operates frequently. Attack time defines how fast to cut the signal to bring it under the threshold. Similarly, release time defines how fast to release the cut back to normal. The 3-band DRC is comprised of three DRCs that can be split into three bands using the BQ at the input of each band. The DRC in each band is equipped with individual energy, attack, and decay time constants.

This DRC can be used for power limiting and signal compression; therefore, it must be tested with maximum signal levels for the desired application. Use a resistive load for initial testing. However, the speaker used in the end application must be used for final testing and tweaking.
The **DRC Tuning** window consists of three identical windows for low, mid, and high bands. Each has a DRC curve that offers 3 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 3 regions. By typing a value and pressing *Enter* on the keyboard, the DRC curve automatically adjusts to the entered parameter.
DRC Time Constant

Change time constants by entering new values for each band.

*Attack(ms)* determines the attack time of the DRC and *Release(ms)* determines the release time once the windowed energy band passes. *Energy(ms)* controls the time averaging windowing uses to determine the average signal energy; therefore, where the incoming signal compares to the set DRC curve. It is beneficial to have control over the DRC time constant for a given frequency band to avoid beating tones caused by the DRC attack and the incoming signal frequency.

The mixer gain controls the relative gain of each of the 3 frequency bands after the DRCs when they are mixed together. This is used to attenuate one of the frequency bands relative to the others, if needed.

**Make note of the sign of the gain coefficients.** Since filters affect phase, a phase reversal or a 180 degree phase shift may be necessary. Use a negative sign on the coefficient to reverse the phase for the second-order LR filter.

Crossover

Configure the frequency range associated with each of the 3 bands used, where the tuning can take place. 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.

### 9.7 2-Band DRC

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 level using an estimate of the alpha filter energy then adjusting the gain based on the region and slope parameters that are defined. The 2-Band DRC is shown in Figure 21.

![Figure 21. 2-Band DRC](image)

The DRC works to reduce the peak of energy if it goes beyond the programmable threshold level. DRC starts an attack event (reduces gain) if energy goes above the threshold. Similarly, it starts a release event if the level goes below the threshold (increases gain back to the original value). Attack and release events occur only when level remains above or below the threshold continuously during the time-constant time. And the constant time is controlled by the attack and release rate. If the attack or release rate is short, DRC operates frequently. Attack time defines how fast to cut the signal to bring it under the threshold. Similarly, release time defines how fast to release the cut back to normal. The 2-band DRC is comprised of two DRCs that can be split into two bands using the BQ at the input of each band. The DRC in each band is equipped with individual energy, attack, and decay time constants.
This DRC can be used for power limiting and signal compression; therefore, it must be tested with maximum signal levels for the desired application. Use a resistive load for initial testing. However, the speaker used in the end application must be used for final testing and tweaking.

The DRC Tuning window consists of two identical windows for low and high bands. Each has a DRC curve that offers 3 regions of compression.

Below the DRC plot, parameters such as threshold, offset and ratio can be manually typed in for each of the 3 regions. By typing a value and pressing Enter on the keyboard, the DRC curve automatically adjusts to the entered parameter.

**DRC Time Constant**

Change time constants by entering new values for each band.
Attack\(\text{(ms)}\) determines the attack time of the DRC and Release\(\text{(ms)}\) determines the release time once the windowed energy band passes. Energy\(\text{(ms)}\) controls the time averaging windowing uses to determine the average signal energy; therefore, where the incoming signal compares to the set DRC curve. It is beneficial to have control over the DRC time constant for a given frequency band to avoid beating tones caused by the DRC attack and the incoming signal frequency.

The mixer gain controls the relative gain of each of the 3 frequency bands after the DRCs when they are mixed together. This is used to attenuate one of the frequency bands relative to the others, if needed. Make note of the sign of the gain coefficients. Since filters effect phase, a phase reversal or a 180 degree phase shift may be necessary. Use a negative sign on the coefficient to reverse the phase for the second-order LR filter.

**Crossover**

By default, the two-band crossover frequencies are set to 1000 Hz, using second-order Linkwitz-Riley filters. This filter type is chosen because the total sum of the two-band signals has a flat response without having to calculate individual crossover frequencies for unity summation. The crossover frequencies need to be separated far enough in the frequency range from each other to avoid any dip caused by the filter sum response.

**9.8 AGL**

The Automatic Gain Limiter (AGL) is a feedback mechanism that can be used to automatically control the audio signal amplitude or dynamic range within specified limits. The automatic gain limiting is done by sensing the audio signal level using an alpha filter energy structure at the output of the AGL then adjusting the gain based on whether the signal level is above or below the defined threshold. Three decisions made by the AGL are engage, disengage, or do nothing. The rate at which the AGL engages or disengages depends on the attack and release settings, respectively.

Figure 24 shows the AGL Tuning window. By default, the AGL is disabled and it can be enabled by clicking the ON/OFF switch on the top right corner.

**Figure 24. AGL Tuning Window**
Threshold (dB)
This parameter sets the threshold at which the compressor will be activated. Lowering the threshold will cause the compression to be activated at lower volume levels. Once the signal exceeds this threshold, compression will be applied.

Alpha (ms)
This parameter configures the sharpness of the compression knee of the AGL.

Attack Rate (0–1)
This parameter controls how quickly compression will be applied to the signal. Higher values will cause the compressor to respond to signals quickly, while lower values will decrease the response time.

Release Rate (0 – 1)
This parameter controls how quickly compression will be removed from the signal as the signal gets quieter. Higher values will cause the compressor to release from signals quickly, while lower values will decrease the release time.

Figure 25. AGL Attack and Release
9.9 Crossover

The major purpose of a digital crossover is to split the frequencies and then send them off to each individual speaker. The crossover is actually a series of filters, which filter out the frequencies that should not go to each speaker. Usually, low pass filters are set on the woofer and high pass filters are set on the tweeter in the crossover.

The plot in Figure 26 in the Crossover Tuning window shows the response of the woofer and tweeter with crossover filters in place, and the combined response after crossover tuning. Five BQs are available for woofer and tweeter channel. Fine-tune the filters to get the smoothest response around the crossover frequency. Optimally, the crossover sum curve (dark green) is flat and crossover difference curve (light green) has a large dip. If the opposite is seen, it is necessary to invert the phase of BQ1 for the woofer or tweeter.

![Figure 26. Crossover Tuning Window](image-url)
9.10 Phase Optimizer

The phase optimizer allows time aligning the 2.0 (tweeter) path with the 0.1 (woofer) path. A programmable phase delay of up to 16 samples can be achieved for both the tweeter and woofer. The Phase Delay Tuning Window pops up if the icon on the top right of the Crossover Tuning window is clicked.

![Phase Delay Tuning Window](image)

Figure 27. Phase Delay Tuning Window

9.11 Clipper

A Clipper can be used to digitally achieve the specified THD levels without voltage clipping. It allows users to achieve the same THD (for example, 10% THD) for different power levels (15 W, 10 W, 5 W) with same PVCC level.

![Clipper](image)

Figure 28. Clipper

**Clipper Level**
The clipper level controls the signal level at which clipping occurs.

**Makeup Gain (dB)**
The Makeup Gain sets additional gain steps from –110 dB to 6 dB.
9.12 Output Crossbar

The crossbar provides the end user with a very flexible way to control what finally appears on amplifier outputs and I2S SDOUT. The Basic tab provides the easiest way for configuration. Go to the Advanced tab if more parameters need to be adjusted. Note that all the parameters need to be specified in decibels (dB).

**Figure 29. Output Crossbar (Basic Tab)**

**Figure 30. Output Crossbar (Advanced Tab)**
9.13 **Level Meter**

Figure 31 shows the level meter, which uses an energy estimator with a programmable time constant to adjust the sensitivity level based on signal frequency and desired accuracy level. The level meter will appear if the LM icon on the bottom is clicked.

![Figure 31. Level Meter](image-url)
10 Smart Amp

Conventional hardware-based speaker protection matches the continuous power output of the audio amplifier with the speaker output rating and sometimes incorporates high-pass filtering to prevent over-exursion.

If the maximum output voltage limit of a traditional system is based on the average power of a full-scale sinusoid, there is risk of voice coil overheating if a square wave is provided as an input. This is due to the fact that a square wave has 6 dB higher average power than a sinusoid of the same peak amplitude as well as having the presence of higher-frequency components. Conservative designs may then have to trade off sound pressure level (SPL) with reliability.

More advanced methods to control load power include the use of limiters and dynamic range compressors. These methods can protect the speaker; however, peaks may be clipped or greatly reduced, especially on source material with high peak-to-average ratios (PAR).

*PurePath™ Smart Amp* replaces hardware-based speaker protection methods with predictive algorithms, speaker characterization tools, and real-time signal monitoring to increase the peak output of the speaker without damage.

*Figure 32* and *Figure 33* are actual song clips comparing the traditional method (left) against *Smart Amp* (right) to control output power. The dashed lines correspond to the output limit of a traditional system. Note that the average power (Pave) is increased while allowing peaks to cross the output limit.

![Figure 32. Audio Clip A, 22-dB Peak-to-Average Ratio Source](image)

![Figure 33. Audio Clip B, 9-dB Peak-to-Average Ratio Source](image)
The first implementation step of Smart Amp-based audio solutions is characterizing the speaker with TI’s PurePath Console 3 and the PurePath Learning Board. These are powerful, easy-to-use tools designed specifically to simplify system-level characterization, tuning, and implementation. The characterization process creates a digital model of the speaker based on thermal, electro-mechanical and acoustic parameters.

The output of the characterization process is an initial set of coefficients that define the Safe Operating Area (SOA) which establishes the boundaries of maximum speaker diaphragm excursion and voice-coil temperature during operation. If the SOA is set correctly, the audio engineer need not worry about speaker damage during the audio tuning process – depending on how hard the system is pushed – audio might sound more or less desirable, but speaker safety is ensured if configured properly.

Figure 34. PurePath™ Smart Amp Block Diagram

PurePath Smart Amp technology enables significant sound quality and system reliability improvements while reducing component size and cost. The PurePath Console 3 GUI and Learning Board speaker characterization hardware provide simple configuration of advanced properties fully describing the acoustical, electrical, thermal and reliability capabilities of an audio system and simplifying system-level characterization, tuning, and integration.

10.1 Smart Amp Features

Smart Bass
Bass can easily be extended into any alignment automatically. As signal amplitude is increased in the bass region, Smart Bass automatically morphs the response to accommodate for larger excursion.
Smart SPL
High-frequency behavior of the loudspeaker diaphragm cannot be obtained electrically. Similarly, it is difficult to obtain accurate low-frequency acoustical measurements without an expensive anechoic chamber. Smart SPL automatically merges electrical and acoustical measurements to create a full picture of the SPL response.

Smart EQ
Automatically and efficiently tunes high frequencies to deliver a flat response or match a target curve in seconds.

Thermal and Excursion Protection
The Smart Amp algorithm understands the thermal and excursion limitations of the speaker. This allows to drive it at peak levels much louder than conventional amplifiers while keeping the voice coil temperature and excursion within the specified limits. This results in louder audio playback.

10.2 Smart Amp Development Overview
The following steps summarize Smart Amp evaluation, planning, characterization, tuning, and integration:
Step 1. **Obtain Hardware and Software** – Speaker characterization and tuning are performed using the PurePath Console 3 software. The TI Learning Board and the Smart Amp Target EVM are needed in order to fully evaluate and develop with Smart Amp.

Step 2. **Plan for Development** – Developing Smart Amp-based systems for the first time can be different than working with conventional amplifiers. Information obtained during the speaker characterization process often leads to changes to the speaker or enclosure to maximize output and quality.

Step 3. **Obtain Speaker Parameters** – The next step is to understand the characteristics of the speaker to be tested. Once a speaker is characterized, the ppc3 file obtained from this step will be used on the next step.

Step 4. **Tune Speaker** – Once the speaker data is obtained, a speaker can be tuned using the Target EVM by importing a ppc3 file.

Step 5. **End-System Integration** – Smart Amp fundamentally shifts how audio systems are designed. Using the Smart Amp tool set, a designer gathers an in-depth understanding of speaker electro-mechanical, thermal and acoustic parameters. Based on these parameters, Smart Amp algorithms deliver high peak voltage and current to the speaker while protecting the speaker from excessive heat or movement. Increased voltage and current levels lead to changes in the system power design. For these reasons, it is important to understand the power supply requirements early in the design.
11 Loudspeaker Characterization

The main objective of the loudspeaker characterization is to obtain the **electro-mechanical** and thermal parameters and establish the **SOA** of the loudspeaker system. The **electro-mechanical** and thermal parameters are obtained using the **Learning Board**.

The **Learning Board App** has a step-by-step wizard that guides the user through the entire loudspeaker characterization process.

11.1 Characterization Process

To perform a loudspeaker characterization:

1. Connect the **Learning Board** to your PC using the USB cable.
   - a. Provide a power supply
   - b. Do not connect the speakers, yet
2. In **PurePath Console 3**, open the **Learning Board App** and select **New**
3. If the **Learning Board** is shown as offline, click **Connect**
4. Click **Characterization**
5. Follow the step-by-step wizard until the characterization is complete

6. Once complete, the Characterization Summary page is shown

### 11.2 Characterization Summary Page

The Characterization Summary page shows the results of the loudspeaker characterization. To verify the loudspeaker plots, use the controls on the top of the graph to select between Modeled SPL and Excursion or Measured Impedance, Temperature, and SPL plots.

Driver and enclosure parameters are also shown as well as the established Safe Operating Area (SOA).

If desired, click the button to redo the characterization. The button will bring back the Characterization Summary page.
11.3 Saving a Characterization

The characterization data can be saved by clicking the button at the Title Bar and selecting Save. This will output a .ppc3 file. This .ppc3 file can later be imported into the Target EVM App for tuning.
12 Smart Amp Tuning

Tuning is a process involving both subjective and methodical approaches. This section provides guidelines to help establish a baseline to achieve the best possible tuning. The main objectives of the (iterative) audio tuning process, also referred to as ‘voicing’, are:

- Improve bass performance using Smart Bass controls by adjusting:
  - Bass Enhancement (Section 12.2.1)
  - Morphing Control (Section 12.2.2)
  - Harmonic Bass Alignment (Section 12.2.3)
- Improve high frequency response using Equalizer

The Smart Amp Tuning Process is summarized in Figure 36.

![Figure 36. Smart Amp Tuning Process](image)

12.1 Tuning Preparation

Tuning is performed using the TAS5825MEVM:

1. **Connect** the TAS5825MEVM to your PC using the USB cable.
   - Provide a power supply (matching the one to be used in the final system)
   - **Connect** the loudspeaker
2. In PurePath Console 3, open the TAS5825M App.

3. If the board is shown as offline, click Connect.

4. Click Tuning and Audio Processing.

5. Select a SmartAmp Processing (48k or 96k).

6. Import the characterization data that was obtained during the Characterization process by clicking the Import button.
7. Click and perform a **System Calibration**. This process ensures that the **Smart Amp** algorithm is properly scaled based on the amplifier output gain.

![System Calibration](image1)

**12.2 Smart Bass Tuning**

During the characterization process, the low-frequency SPL model of the loudspeaker was obtained. Based on this model, the response is automatically optimized to match popular loudspeaker alignment types (that is Butterworth, Linkwitz-Riley, and so forth). This allows the audio engineer to focus on choosing the desired sound with just a few clicks.

When tuning **Smart Bass** for the first time, it is best to first disable the **Equalizer**.

![Equalizer](image2)

To start tuning **Smart Bass**, enable **Smart Bass** and click the Expand symbol, as shown in the following image.

![Smart Bass](image3)

The **Smart Bass Tuning** page has all the controls needed for **Smart Bass** tuning. Several plots (such as Excursion) are provided as an aid to the tuning process.
12.2.1 Bass Enhancement (Low Volume Tuning)

The main objective is to maximize the bass response as much as possible and tune the bass, as desired. During this phase, it is important to listen at low volume levels only – this is to ensure that thermal and mechanical protection systems do not kick-in. Ensure that the Morphing Control (Section 12.2.2) is set at maximum during this phase.
Table 6. Bass Enhancement Parameters

<table>
<thead>
<tr>
<th>Field</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>Corner Frequency</td>
<td>–3-dB point of the target response. Smaller speaker drivers should use higher corner frequency.</td>
</tr>
<tr>
<td>Order (Slope)</td>
<td>Determines the sharpness of the roll-off towards lower frequencies. For lower corner frequency, choose higher order.</td>
</tr>
<tr>
<td>Type</td>
<td>Selects the alignment type</td>
</tr>
</tbody>
</table>

**Corner Frequency**

The *Corner Frequency* indicates the –3-dB point of a flat response target (indicated by the green curve in the following image). Selecting a proper *Corner Frequency* is important for the overall performance of the speaker system. If the cutoff is set too high, the speaker will have limited bass response. If set too low, energy will be wasted trying to drive frequencies that the speaker will not be able to reproduce and the excursion protection system will be overly active.

TI recommends doing a series of listening tests while adjusting settings.

- Adjust *Corner Frequency* while watching the compensation (red curve) in the response plot window.
- Targeting between a 10- to 20-dB compensation (red curve) often provides the best results.
- Do not exceed the 20-dB line (at least initially).

**Alignment Order and Type**

The *Order and Type* determine the bass roll-off. In other words, it determines what occurs below the corner frequency.

A high-order roll-off cuts bass faster, saving power and limiting speaker excursion that will not produce much SPL. Likewise, *Type* has significant influence on the SPL and energy below the corner frequency.

- Select a higher order if the speaker handles excursion poorly.
- Select a lower order to leave small amounts of low-frequency content in the signal.
- Butterworth is suitable for most applications. For ported or passive radiator systems that can reproduce 60–80 Hz, a Chebyshev alignment works well.
12.2.2 Morphing Control (Mid- to High-Volume Tuning)

*Morphing* determines the headroom dependent balance between *Bass Enhancement* and *Harmonic Bass Alignment*. As excursion and thermal headroom drops with increase in music loudness, the *Morphing* feature gradually and dynamically reduces bass.

From Section 12.2.1

![Morphing Control Diagram]

- **Artifacts?**
  - Yes: Set Morphing Higher
  - No: Set Morphing Lower
- **No Bass?**
  - Yes: Re-Tune Bass Enhancement
  - No: Continue

NOTE: This is an iterative process! It is important to listen to different types of music and at several volume settings (listening levels).

**Morphing Speed (Optional)**

The *Morphing Speed* control determines the aggressiveness on which the *Smart Amp* algorithm adapts to a change of headroom.

Speakers react very differently to morphing speed and unfortunately there is no universal guideline for how to tune this for the best setting. TI recommends experimenting with several settings. This setting may also be left at the default value (0).

- Listen to different music types at moderate to high volume levels
- Listen for audible artifacts such as:
  - **Bass region**: distortion, especially with high transients (such as a kick drum)
  - **Mid/high range**: distortion, modulation artifacts
12.2.3 Harmonic Bass Alignment (Mid- to High-Volume Tuning)

The Harmonic Bass Alignment control determines the aggressiveness of the excursion protection algorithm as speaker headroom is reduced. Some speakers sound great with an aggressive setting (high value) where other speakers, typically of lower quality, will sound harsh and distorted and will require less aggressive setting (lower value).

- Listen to different music types at moderate to high volume levels
- Listen for audible artifacts such as:
  - **Bass region**: distortion, especially with high transients (such as a kick drum)
  - **Mid/High range**: distortion

12.2.4 Excursion Tuning (Optional)

Depending on the type and quality of the speaker (as characterized in the measurement phase), the Peak Excursion SOA setting sometimes needs a post-audio-tuning adjustment for best sound quality. As a typical speaker approaches its $X_{\text{max}}$, the THD tends to rise quickly. This behavior can have an amplifying effect on artifacts.

- In the Max Level Tuning window, change the peak excursion limit value while you listen to audio.
- Reduce it if there are Artifacts at high volume settings.
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