Using the TAS5754/6M and PCM5242 HybridFlow Processor

User's Guide

Literature Number: SLAU577A
January 2015–Revised June 2015

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1 Introduction to PurePath™ HybridFlow Processing

PurePath™ HybridFlow Processing refers to a proprietary method of signal processing created by Texas Instruments to combine the benefits of a fully-programmable digital signal processor (DSP) and the benefits of a fixed-function DSP. This method reduces the negative impact of each approach, such as the complexity and download time of a fully-programmable device and the limited flexibility of the fixed-function device.

1.1 RAM-Based Audio Processing

fully-programmable devices are sometimes referred to as “RAM devices” because the customer is offered a given number of instructions (MIPs) and a certain amount of random access memory (RAM) used to create proprietary DSP programs. A RAM device is known for ultimate flexibility and an experienced DSP programmer can create high-quality audio algorithms. However, the tools required for the product developer to interact with the device must support an almost endless collection of fundamental functions, therefore, it can be quite complex and difficult for the novice. These devices are most commonly stand-alone DSPs or devices with integrated CODECs such as the popular AICxxxx and PCMxxx miniDSP line from TI.

1.2 ROM-Based Audio Processing

Other devices where DSP programs are programmed into the silicon during manufacturing by the silicon vendor are sometimes referred to as “ROM devices” since the DSP program is presented as a ROM image to the customer and cannot be fundamentally changed after the integrated circuit (IC) has been manufactured. The older devices (pre-2014) in the popular TAS57xx line of mid-power audio amplifiers with integrated digital audio processor are all examples of this type of IC. They are characterized as being easy to use. However, since the DSP program is fixed at the time of manufacturer, they are also inflexible. Although most of the TAS57xx devices produced during this time were quite similar, dozens of parts were created in order to meet the need of small changes to the ROM image to make them suitable for various use cases. In some cases, entirely new devices were create to simply move the location of one or two filters used for equalization (Biquads). This disallows the end-customer from reusing previously qualified parts.

1.3 HybridFlow Audio Processing

In contrast, a HybridFlow processor builds several of the larger components commonly used by several target applications into small ROM images which are “called” from within a larger RAM-based DSP program, as required. Essentially, the audio algorithms are primarily defined in ROM but the manner they are used is defined in smaller DSP programs called HybridFlows. These pre-defined HybridFlows increase ease-of-use, while minimizing the download time of the DSP program. Additionally, a single device can now be used to support several types of applications. For instance, the TAS5754/6M is used in stereo (2.0), mono, bi-amped (1.1), and 2.1 applications, with HybridFlows specifically designed for each use-case.

2 General Overview of HybridFlows

In the signal processing path, the HybridFlow processor is placed after the Serial Audio Port and before the Digital-to-Analog Converter (DAC), which precedes the Class D amplifier. This is illustrated in Figure 1.

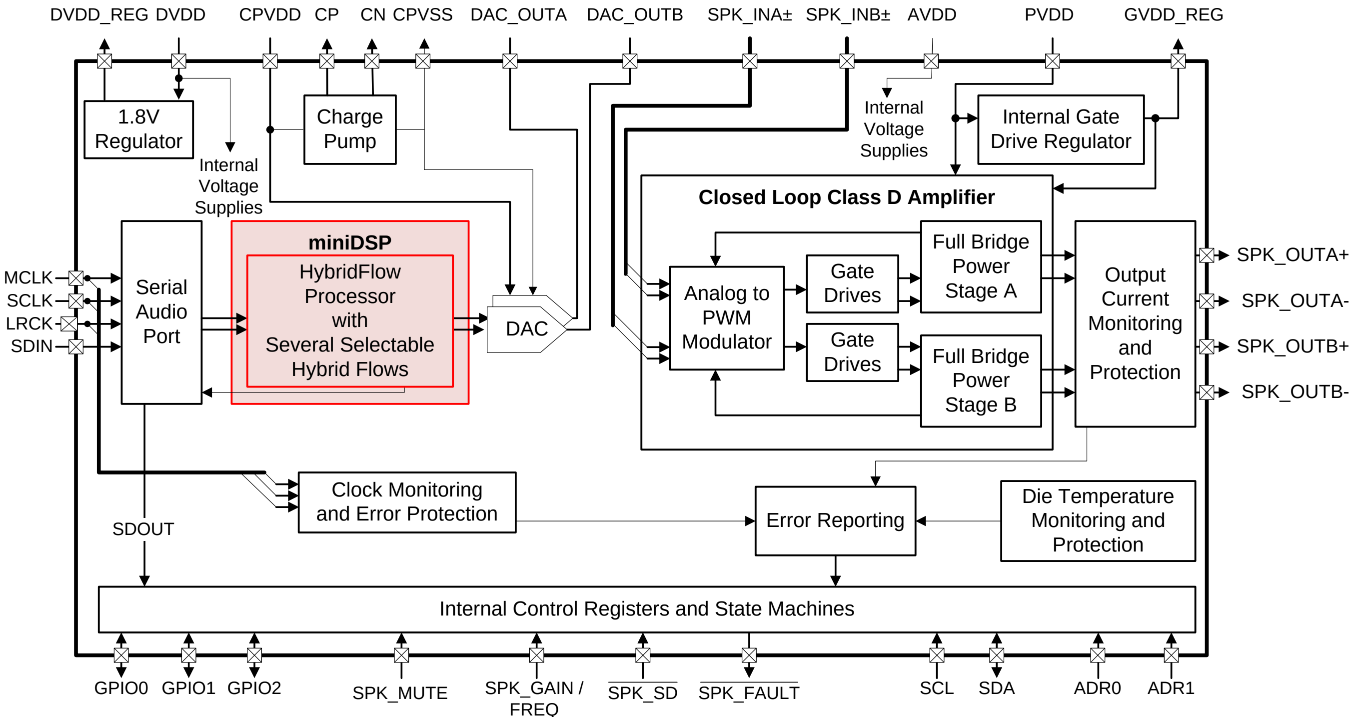


Figure 1. HybridFlow Processor Location in the Signal Path

Generally speaking, the HybridFlow processor is thought of as a 2 input and 2 output digital signal processor. The two input channels can come from a TDM stream, or a traditional I²S, left-justified, or right-justified serial audio input stream and are presented to the miniDSP through the serial audio port. The controls that configure the serial audio port to accept various input formats are detailed in the device data sheet.

2.1 Signal Routing

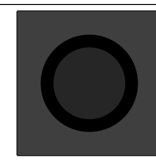
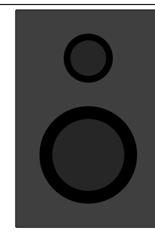
The two output streams of the HybridFlow processor are presented to the DAC and can be any type of signal supported by the HybridFlow. The most straightforward example is that of a signal that is presented as the left frame at the input of the HybridFlow processor and is carried through as the Channel A output. However, this is not necessarily the case for all HybridFlows. For instance, HybridFlow 3 allows selection of one of the two signals, or a mixture of both. It then separates the input signal into the low- and high-frequency portions of that input signal and presents them as two separate signals on OUTA and OUTB. Clearly, the signals emerging from the HybridFlow processor can vary widely in their content and do not necessarily directly relate to a traditional Left and Right input signal scenario. This is the reason that generic *Channel A* and *Channel B* labeling is used when describing these channels. The traditional Left and Right correlation may not apply.

Inside the DAC block, which follows the HybridFlow processor, there are controls to flip the polarity of signals as well as to swap the outputs the Channel A DAC can carry. Channel B input information and Channel B DAC can likewise be configured to carry Channel A content. It is important to comprehend the setting of those controls, as well as those in the HybridFlow itself, in order to understand the signal presented at the output of the amplifier.

2.2 Supported Use Cases

The HybridFlows have been generated based upon several popular configurations, primarily around the number and type of amplified outputs. For the purposes of this document and other collateral related to the TAS5754/6M documents, some terminology bears definition:

Table 1. Supported Use Cases

Mode:	Also Known As:	Amplifier Output Configuration	Symbol
2.0	Stereo	Two independent signals sent via two BTL outputs to two independent speakers	
Mono	0.1 (when used exclusively for low-frequency information for a subwoofer)	A single signal, created from one or both of the two input signals sent via a single output created by placing the two output channels in parallel into a single channel, usually to drive more power	
1.1	Bi-Amped, Dual Mono	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	
2.1	N/A	One device uses 2.0 mode and a separate device uses Mono mode	
2.2	Dual Stereo	Two devices drive four speakers with four independent signals. This is implemented as two devices using 2.0 mode or two devices using 1.1 mode	

Throughout this document, the configurations supported by a given HybridFlow are noted using the symbols in Table 1.

3 Typical Product Development Flow

The following typical product development flow is provided for reference. This may differ from application to application or customer to customer. Nonetheless, this is offered as a work flow suggestion to ensure the most seamless interaction with the tools possible.

1. Identify the end-equipment, target loudspeaker enclosure, and target speaker driver
2. Determine applicable speaker/amplifier configuration (2.0, 2.1, 1.1, and so forth)
3. Choose from available HybridFlows supporting the target configuration
4. Identify which HybridFlows among those identified in step 3 integrate the processing features desired
5. Use the PurePath Console GUI to develop the configuration files for the end equipment
   (a) A *base configuration* .cfg file is created for the most common use case in the application
   (b) Any number of *variant configuration* .cfg files are generated by creating a configuration file consisting of only those changes required for the use case variant

3.1 Register Dumping and .cfg File Generation

Dumping of the registers is used to create a .cfg file containing all of the necessary register writes to achieve the setup in the PurePath Console GUI at the time of the dump. This holds all of the HybridFlow information as well as the coefficients used for each processing block.

1. To execute a register dump, the GUI must be set into *Advanced* by using the *Mode* selection slider on the top right of the GUI window. Once in advanced mode, the *Direct I2C Read/Write* tab should become available where the *Register Dump* tool is found.
2. Register dumping can only occur when the device is in stop mode due to the data buffering of the miniDSP. Once the desired HybridFlow and associated processing coefficients are loaded onto the device, click the stop button to stop the HybridFlow. At this point, the *Register Dump* tool becomes available and the format is chosen.

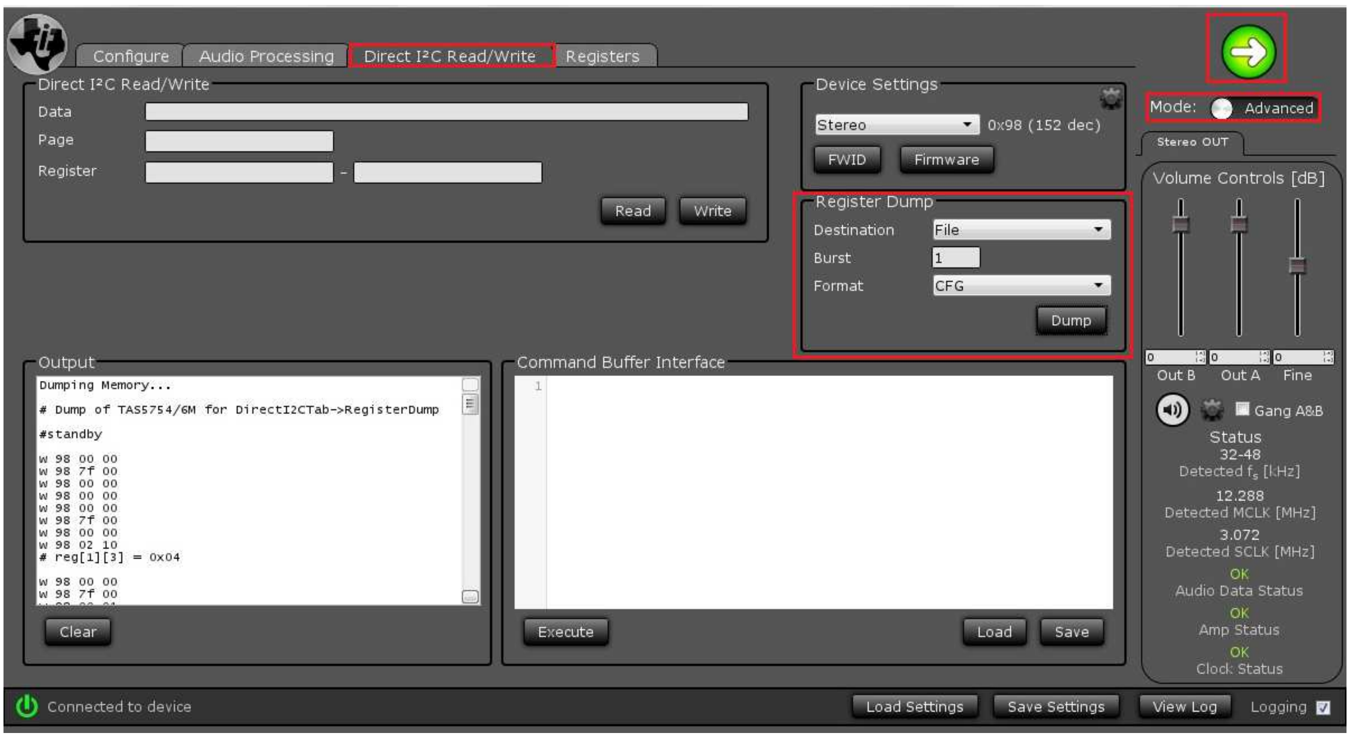


Figure 2. Register Dump Tool

3. Since the HybridFlow must be stopped in order to execute a register dump, the .cfg file also contains a shutdown sequence configuration executed by the PurePath Console GUI. Therefore, the registers controlling the shutdown state of the part must be rewritten in order to use the .cfg file. An example sequence to be added to the end of .cfg file follows:

```
#-------------------------Exit shutdown
# Address Register Data
w 98 00 00
w 98 03 00
w 98 2a 11
w 98 02 00
w 98 3d 30
w 98 3e 30
```

With this appended to the .cfg, the HybridFlow runs as expected.

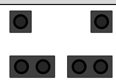
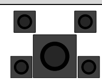
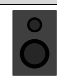
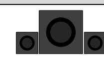
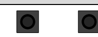
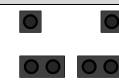
3.2 Using Several HybridFlows in Single Application/End Equipment

A single piece of end-equipment can use more than one HybridFlow with the TAS5754/6M device. An example of this is an end-equipment where the product development engineer wants to use HybridFlow 2 in one user scenario and HybridFlow 6 in another use-case scenario. To cycle between HybridFlows, the device must first be placed into power down, reprogrammed with the configuration file generated for the given HybridFlow via the GUI, and then brought back out of power down. The speed this transition occurs is specified by the download time of the target HybridFlow and is ultimately determined by the bus speed and burst capabilities of the host processor. The download time of the target HybridFlow is shown in each HybridFlow section. A period of no sound should be expected as the device is quickly moved into and out of power down in order to implement the handoff from one HybridFlow to another.

4 HybridFlow Processing Cross Reference

Table 2 shows the processing features available for each of the available HybridFlows:

Table 2. HybridFlow Cross Reference

Feature	HybridFlow 1	HybridFlow 2	HybridFlow 3	HybridFlow 4	HybridFlow 5	HybridFlow 6	HybridFlow 7
Supported Output Configurations							
Typical Target Application	Mid-Level DTVs & General Audio	Mid-Level DTVs & General Audio	Bi-Amped Bluetooth® and Active Speakers	Bluetooth Speakers and Wireless Subs	Hi-End Digital TVs	Docking Stations, All-in-One PC, & General Audio	Mid-Level DTVs, Soundbars, & General Audio
Supported Sample Rate	8–48 kHz	8–48 kHz	8–48 kHz	8–48 kHz	8–192 kHz	8–48 kHz	8–96 kHz
Psychoacoustic Bass Enhancer (PBE)	✓	x	✓	✓	x	x	x
Output Configurations (Stereo/Mono)	Stereo	Stereo	1.1	Mono	Stereo	Stereo	Stereo
DRC Type	3-Band Compander	3-Band Compander	3-Band Compander	3-Band Compander	DRC-Lite	3-Band Compander	3-Band Compander
Biquad Equalizers (In Full-Range Path)	2 × 12	2 × 12	1 × 10 + 1 × 5	1 × 12	2 × 1	2 × 15	2 × 5
PurePath SmoothClip	✓	✓	✓	✓	✓	✓	✓
Sound Field Spatializer (SFS)	x	x	x	x	x	✓	x
Dynamic Dialog Enhancer (DDE)	x	✓	x	x	x	x	x
Dynamic Bass Enhancer (DDE)	✓	✓	✓	✓	x	✓	x
Serial Audio Data Out (Subwoofer, Full-Range)	FR	SW or FR	SW or FR	FR	SW or FR	FR	FR

The symbols used to indicate supported output configurations are simple representations to show the targeted use case for the HybridFlow. Traditional passive crossovers, consisting of passive components to separate the signal sent to a speaker into high- and low-frequency elements can further extend the usefulness of HybridFlows for a given application. For instance, a HybridFlow originally configured to drive a single Mono loudspeaker is used in a 1.1 configuration with the use of an appropriate crossover to separate the signal for a tweeter and midrange speaker. Likewise, a HybridFlow created for a 2.0 system can also be used with a 2.2 system if a crossover is present in the system to separate the stereo signal into high and low-frequency elements.

It is important to note that 2.1 systems are implemented in any 2.0 system driving a pair of speakers by simply adding a TAS5754/6M device using HF4 to create the subwoofer channel. However, in Table 2, 2.1 support is only noted for those devices creating a subwoofer channel internally, allowing a simple digital input device with no processing (like the TAS5760M or TAS5760L) to drive the subwoofer in PBTL mode.

5 HybridFlow 1 (HF1)

Figure 3 depicts the signal path for HybridFlow 1. The shaded tabs correspond to the functions found in the PurePath Console GUI.

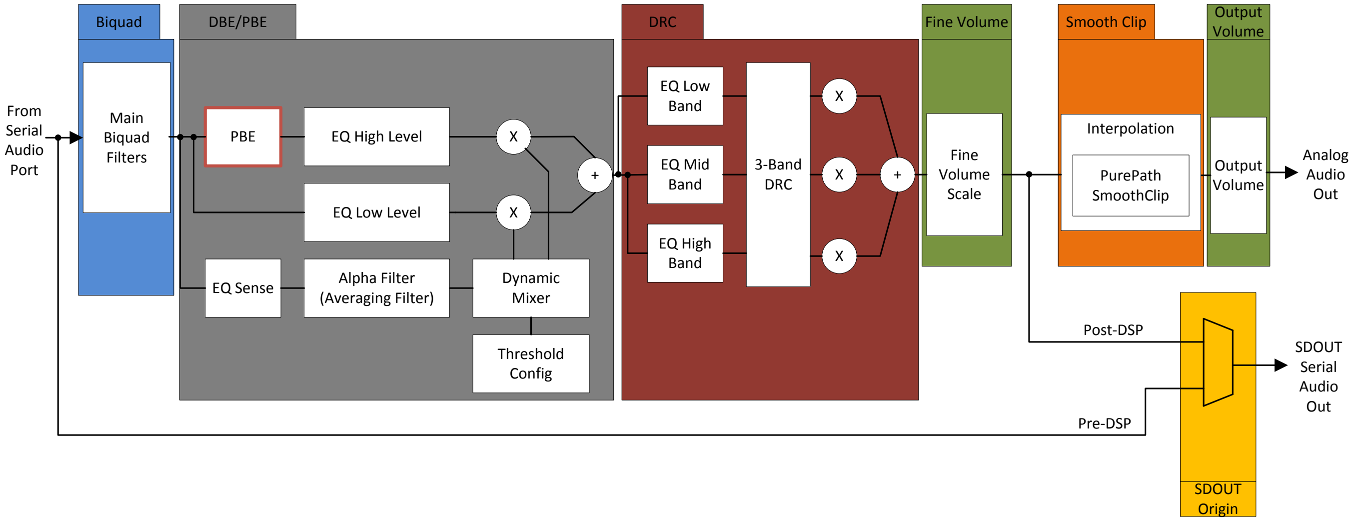


Figure 3. HybridFlow 1 Block Diagram

5.1 Biquad

The Biquad filter block contains 10 independent filters designed for tuning the frequency response of the overall system. This is where the bulk of the frequency compensation occurs; select the filter type, subtype, and other parameters for each of the 10 Biquads. Complex tuning shapes are made to compensate for deficiencies in speaker response with a goal of a flat response over the frequency band of interest.

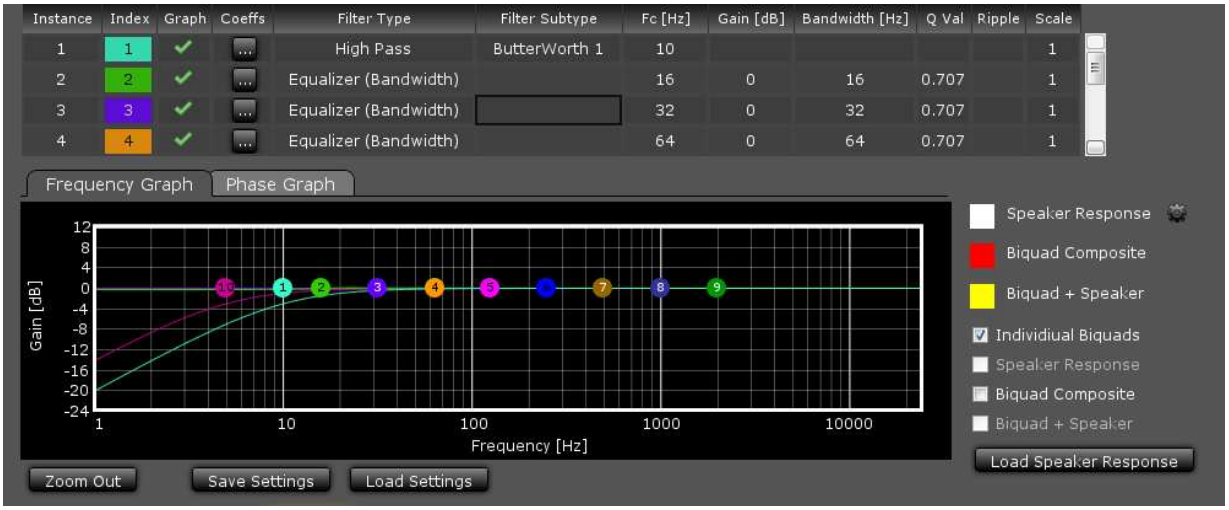


Figure 4. HF1 Biquad Tuning Window

Figure 4 shows the Biquad audio processing window in the PurePath Console GUI. Drag and drop each Biquad into position using the mouse on the plotting window. Other parameters must be typed into the chart above the plotting space. After which, pressing *Enter* on the keyboard causes the change to take effect.

Filter Type and *Filter Subtype* are drop-down menus activated when this space is clicked for the desired Biquad. Remove or add Biquads by clicking on the "graph" space next to the desired Biquad.

The *Phase Graph* shows the phase response for each of the individual Biquads.

The *Biquad Composite* check box shows the overall response based on the position of the individual Biquad filters. This composite view is the frequency response alteration applied to the incoming digital audio data. View this independently by deselecting *Individual Biquads*.

5.1.1 Load Speaker Response

If desired, a measured speaker response is loaded into the Biquad window by clicking *Load Speaker Response*. This aids in tuning the frequency response. With a speaker response loaded, view the overall audio system response by clicking *Biquad + Speaker*. This takes into account the added Biquads as well as the natural response of the speaker.

The file format for the speaker response is .txt. Each line in the file represents a data point containing frequency (in Hz), SPL (in dB), and Phase (in degrees), separated by a space. Any line that does not begin with a number is ignored.

An example of the format follows. Note that the header line, that includes the titles of each column, is ignored since it starts with a "*".

```
* Freq(Hz) SPL(dB) Phase(degrees)
20.142 39.577 -153.765
20.508 39.799 -154.993
20.874 40.023 -156.829
21.240 40.188 -159.066
```

NOTE: A loaded speaker response is not loaded into the HybridFlow, it is only displayed in the PurePath Console GUI as a tuning tool.

5.2 PBE

The next processing block in the HybridFlow is the Psychoacoustic Bass Enhancement (PBE). PBE perceptually increases the bass level using the principal of "missing fundamental", a well-known psychoacoustic effect invoking a perception of the bass frequencies even though the fundamental of those frequencies has been filtered out. The PBE algorithm enhances bass sound for small loudspeakers that are incapable of reproducing bass frequencies efficiently.

Figure 5 shows the block diagram of the PBE block for stereo channels with down-mixed harmonics generation path. The module is composed of input high-pass filters (HPF) and the harmonic generator (containing the harmonic intensity control). The high-pass filter removes frequencies that are irreproducible with the loudspeaker at the high signal level flowing through this path. Those frequencies are attenuated in advance and do not disturb the harmonics generation. This eliminates the irreproducible low-frequency energy in the output signal. The harmonics generator generates harmonics of the low-frequency band selected using the HPF. These are then summed back into the left and right audio paths using the effect intensity control.

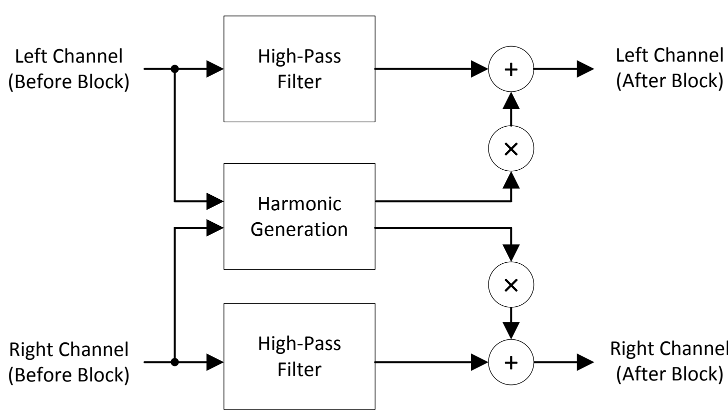


Figure 5. PBE Block Diagram

Figure 6 shows the PBE tuning window in the PurePath Console GUI. There are three controls for the PBE block. The first is a high-pass filter corner frequency *HPF f_s*, determining which fundamental frequencies are removed from the audio output. A second control, called *Harmonic Intensity*, determines how many (the order) Harmonics are created. It is important to note that the 0–100 scale of the Harmonic Intensity is not directly related to the number of harmonics created, but instead a relative number where 100 is the maximum number of harmonics. A value of 0 means no harmonics are added. The third control is *Effect Intensity*. Effect intensity range is from 1.0 to 5.0, in 1-integer steps. This property sets how much of the harmonic content is mixed back into the output signal of the processing block.

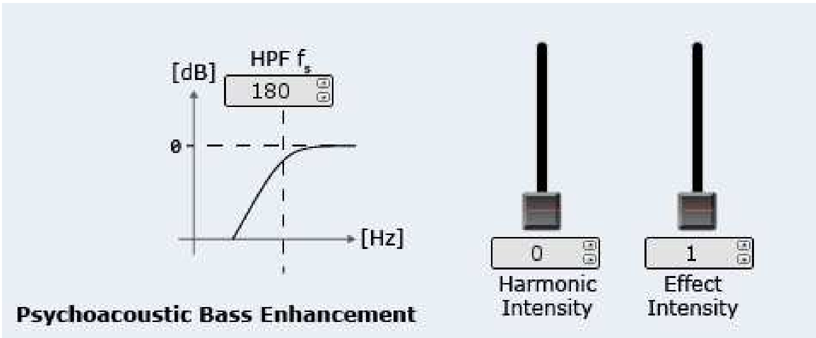


Figure 6. PBE Tuning Window

5.3 DBE

Dynamic Bass Enhancement (DBE) is a processing block that allows for optimizing the bass response of the system. Two signal paths (low level and high level) 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 DBE channels is dynamic in nature and depends on the incoming audio. Figure 7 shows the tuning window in the PurePath Console GUI.

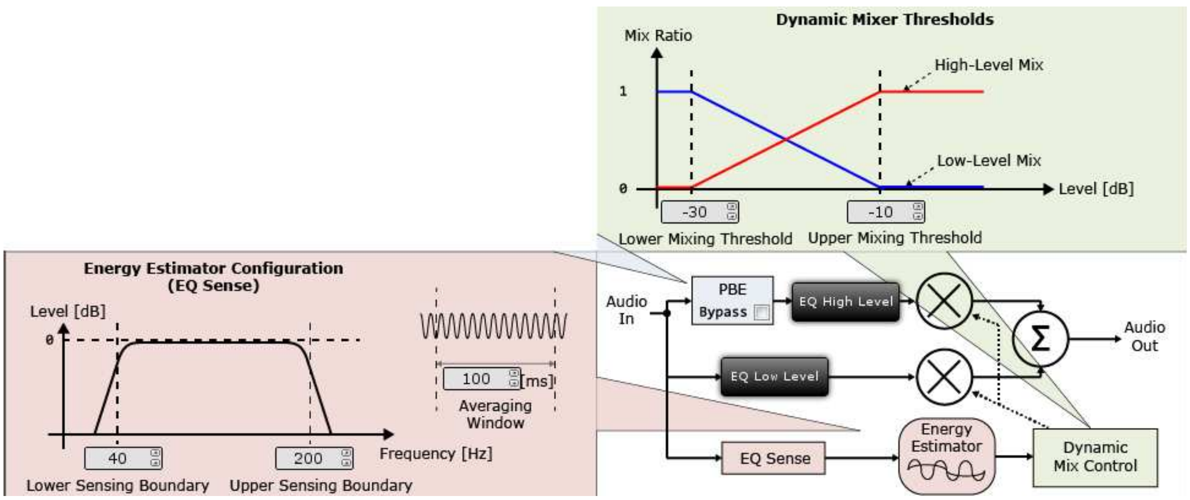


Figure 7. HF1 DBE Tuning Window

5.3.1 Dynamic Mixer Thresholds

The mixing of the two paths (low level and high level) is controlled by setting the *Upper Mixing Threshold* and *Lower Mixing Threshold*. When the averaged signal (as set by the *Averaging Window*) is below the lower mixing threshold, the *Dynamic Mixer* sends all of the audio through the low-level path. When the signal is above the upper mixing threshold, it is sent through the upper-level path. When the signal is between the two, it is mixed together by the *Dynamic Mixer*.

5.3.2 Energy Estimator Configuration (EQ Sense)

Another key configuration for the dynamic mixer block is the EQ sense and alpha filter. In HF1, the EQ sense is a single bandpass filter. The bandwidth of the filter is set by entering in the lower and upper sensing boundary in the GUI. This tells the dynamic mixer which frequency range of the incoming signal to average in order to determine how the signal compares to the mixing thresholds.

The *Averaging Window* or alpha filter works similarly to that of a DRC. It 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.

5.3.3 EQ Low Level

The low-level path contains 2 configurable Biquads to establish the EQ curve the audio is sent through when the time average signal is at a low-level. These fully-functional Biquads can be assigned to several filter types or sub-types. This determines frequency response when low-level is active based on the Energy Estimator Configuration and the mixing threshold.

Click the *EQ LOW LEVEL* button to display the tuning window (Figure 8). The tuning shown for the EQ Low Level was chosen as a bass boost to reduce early onset bass roll off from the speaker. For more details visit Section 5.1, Biquad.

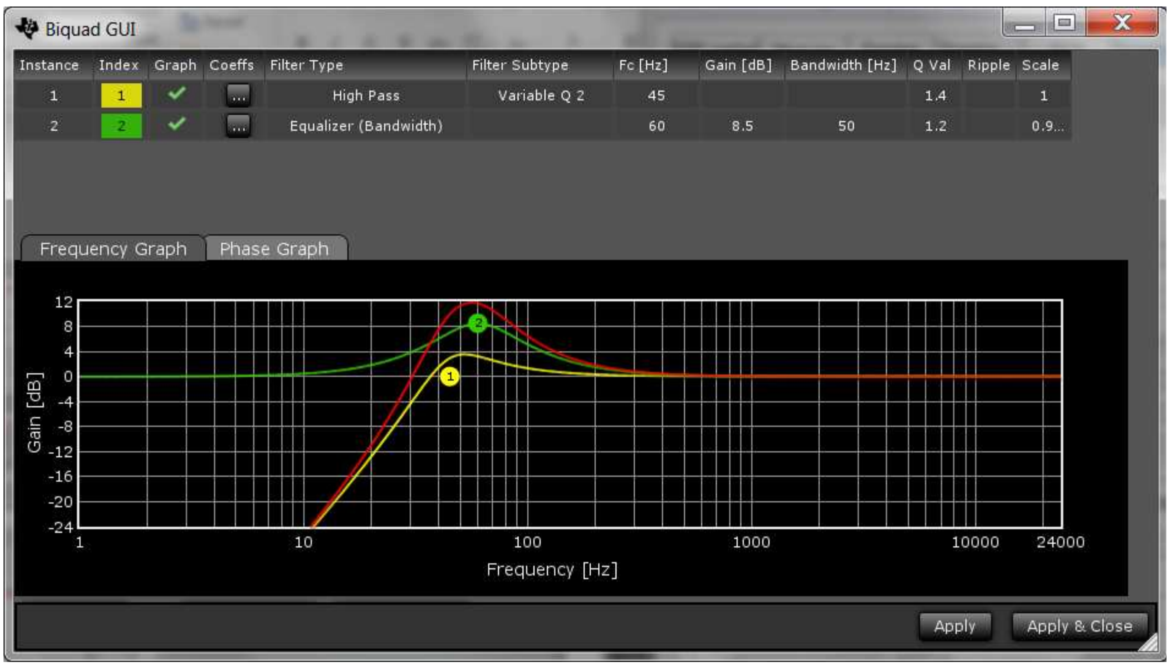


Figure 8. DBE EQ Low-Level Tuning Window

5.3.4 EQ High Level

The high-level path, similar to the low-level path, has two Biquads that can set the EQ curve used when the time averaged input signal is above the upper mixing threshold. However, for HF1, there is also an additional feature allowing harmonic bass (called *Psychoacoustic Bass* or PBE) to be mixed into the output whenever “real” bass must be filtered out. This is explained in section Section 5.2 PBE.

Click the *EQ High LEVEL* button to display the tuning window (Figure 9). The tuning shown for the EQ High Level is a simple high-pass filter with fast roll off. For more details visit Section 5.1, Biquad.

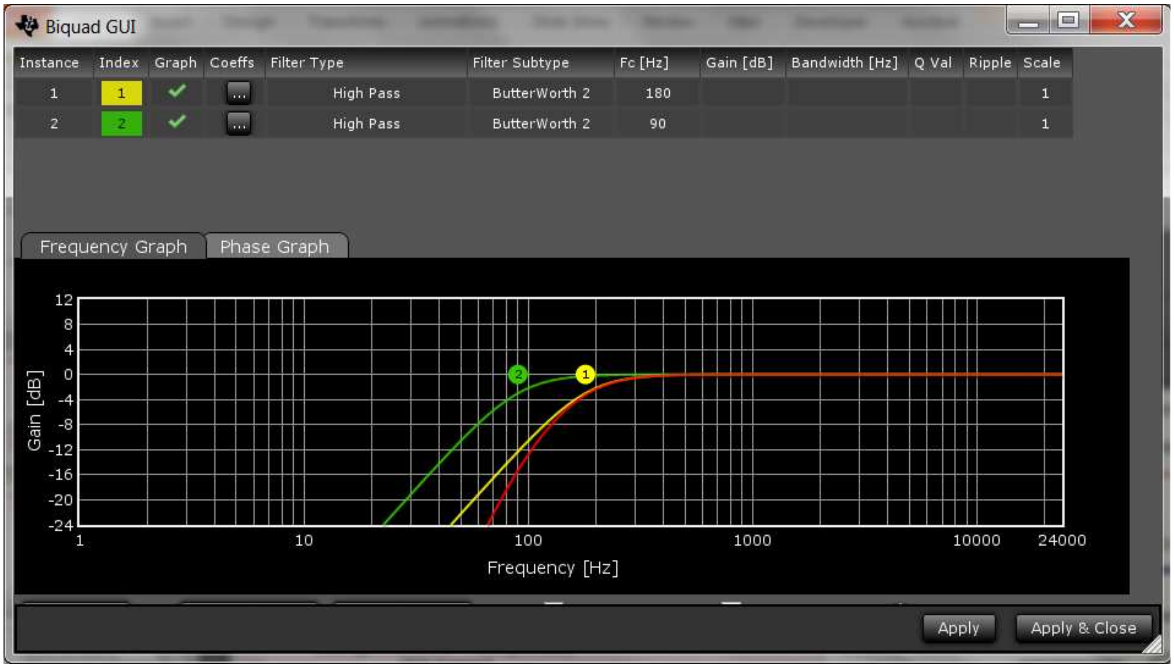


Figure 9. DBE EQ High-Level Tuning Window

5.4 *DRC Standard 3-Band Dynamic Compander*

HybridFlow 1 features a 3-band DRC compander (compression and expansion). The DRC is used for power limiting and signal compression; therefore, it must be tested with maximum signal levels for the desired application. For initial testing, use a resistive load. However, the speaker used in the end application must be used for final testing and tweaking.

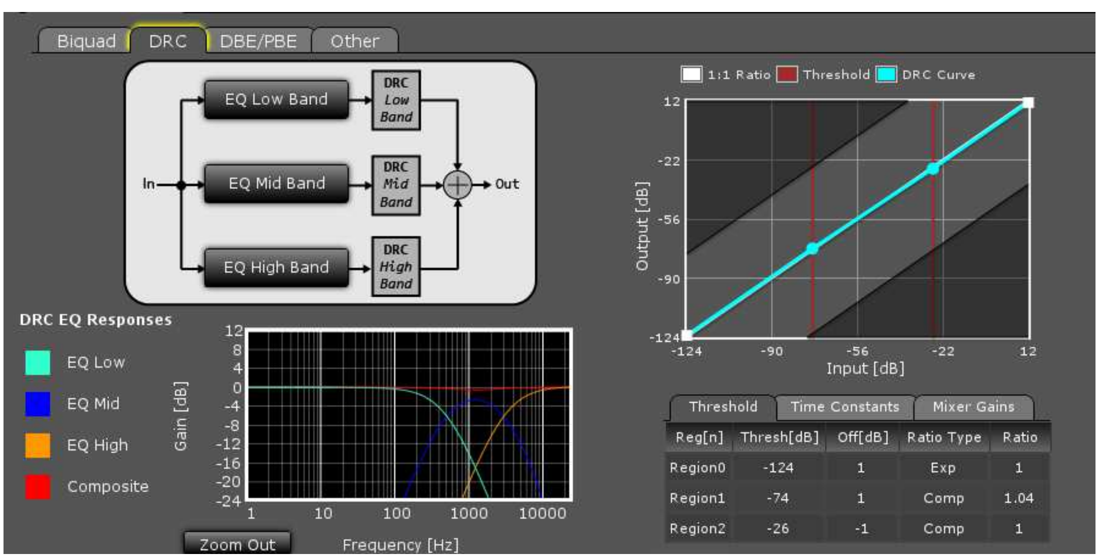
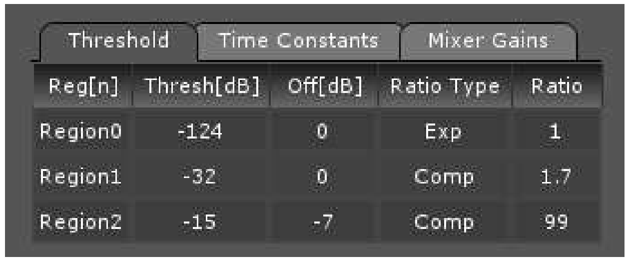


Figure 10. HF1 3-Band DRC Compander Tuning Window

On the right side of the window in Figure 10 is the DRC curve offering 3 regions of compression. The points on the DRC curve can be dragged and dropped. At the top center of the window is the block diagram of the DRC as well as 3 configurable frequency bands (Low, Mid, and High).

5.4.1 DRC Threshold Tab

Below the DRC window, parameters such as threshold, offset, expansion or compression, and the ratio value can be manually entered for each of the 3 regions under the *Threshold* tab. By typing a value and pressing *Enter* on the keyboard, the DRC curve automatically adjusts to the entered parameter.

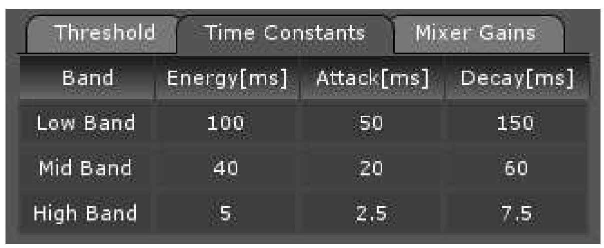


Reg[n]	Thresh[dB]	Off[dB]	Ratio Type	Ratio
Region0	-124	0	Exp	1
Region1	-32	0	Comp	1.7
Region2	-15	-7	Comp	99

Figure 11. HF1 DRC Threshold Control Tab for 3-Band Compander

5.4.2 DRC Time Constants Tab

The standard 3-band compander offers splitting of the incoming audio into 3 frequency bands determined by the user. Although the same DRC curve gets applied to all 3 frequency bands, different attack time constants can be associated with each band to optimize audio quality and speaker protection. Change time constants by clicking on the *Time Constants* tab, as shown in Figure 12, and enter new values for each band.



Band	Energy[ms]	Attack[ms]	Decay[ms]
Low Band	100	50	150
Mid Band	40	20	60
High Band	5	2.5	7.5

Figure 12. HF1 DRC Time Constants Tab for 3-Band Compander

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. *Attack[ms]* determines the attack time of the DRC and *Decay[ms]* determines the release time once the windowed energy band passes.

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.

For example, a very fast time constant on a low-frequency signal may cause the DRC to attack and release before a full cycle of the incoming signal has passed. Then, when the next peak of the wave passes through the DRC, it again attacks and then releases as the peak passes. The DRC continuously attacks and releases rather than enveloping the signal causing audible distortion.

With separate time constants, the standard 3-band compander can still have a very fast time constant at high frequencies and a slower time constant at low frequencies enveloping and compressing the entire audible range quickly and effectively.

5.4.3 Mixer Gains Tab

The mixer gain controls the relative gain of each of the 3 frequency bands 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.

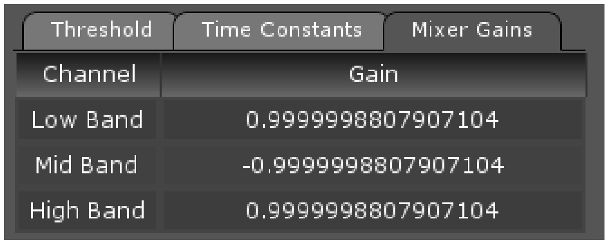


Figure 13. HF1 Mixer Gains Tab for 3-Band Compander

5.4.4 Band Splitting

Configure the frequency range associated with each of the 3 bands used by the *Time Constants* tab by clicking on the *EQ Low Band*, *EQ Mid Band*, and *EQ High Band* buttons. Here a Biquad window appears where the tuning can take place. After tuning, the response is automatically displayed in the 3-Band DRC Compander tuning window on the bottom left.

For more details visit Section 5.1, Biquad.

5.5 Fine Volume

The fine volume control is a digital volume control that allows adjustment between –0.25 dB and 0.25 dB by setting the slider.

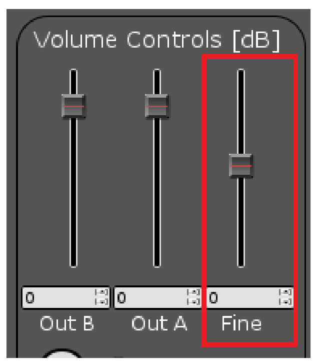


Figure 14. Fine Volume Control

5.6 PurePath SmoothClip

SmoothClip works as a comparator in the digital domain on a sample-by-sample basis. If the incoming audio data word is larger than the set comparator coefficient, the set coefficient is passed until the incoming audio data word is below the set coefficient. This effectively clips the signal. Unlike typical digital clipping that occurs at the sample rate (Fs), SmoothClip operates at very high speeds, minimizing the unwanted distortions associated with digital clipping.

This is often used in conjunction with slower DRC time constants. With a more gradual time constant and compression ratio, the potential for DRC beating or "pumping" is reduced and sound quality and dynamics are improved. However, due to the slow DRC response, a few cycles of incoming audio data that are greater than the set DRC thresholds can pass through. With SmoothClip following a DRC, these cycles can be clipped in a well-controlled fashion to prevent speaker damage until the DRC has attacked the signal.

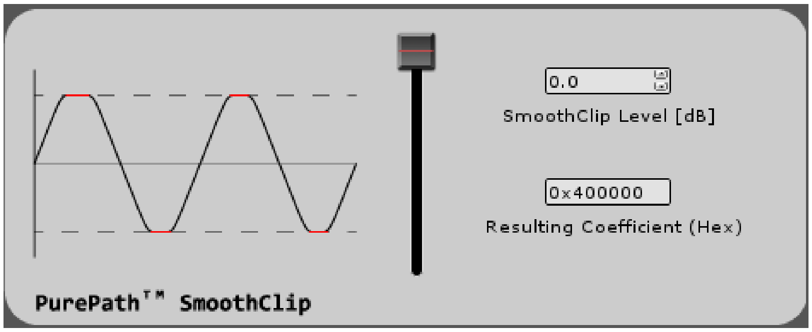


Figure 15. SmoothClip Tuning Window

SmoothClip only has one tuning parameter; the level at which the clipping occurs. TI recommends setting the level by measuring the THD+N at the frequency most boosted by the overall system since this frequency is clipped the most.

5.7 Output Volume

The output volume controls the digital level of both Channel A and Channel B independently from –103 dB to 24 dB by setting the slider. In HybridFlow 1, where Channel A and B are identical, for most applications channels A and B are adjusted together to avoid mismatch on stereo speakers.

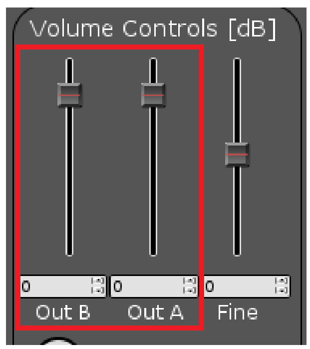


Figure 16. Output Volume Control

5.8 SDOUT Serial Audio Data

In HybridFlow 1 there are 2 choices of digital output sources available on GPIO2 (pin 21 on the TAS5754/6M devices). The first option is Pre-DSP which passes the incoming digital data to SDOUT. There is no processing of the data. The second option is Post-DSP which allows for the fully processed data to be available for use. In post-DSP, Smooth Clip and the digital output volume control have no effect. SDOUT is useful when trying to connect with other digital input devices and amplifiers.

NOTE: SDOUT acts like a MUX; however, it is a hard mixer meaning it mixes only one of the inputs at a time with a 100% mix ratio. This method creates a more efficient DSP MUX device rather than a traditional switching MUX.

6 HybridFlow 2 (HF2)

Figure 17 depicts the signal path for HybridFlow 2. The shaded tabs correspond to the functions found in the PurePath Console GUI.

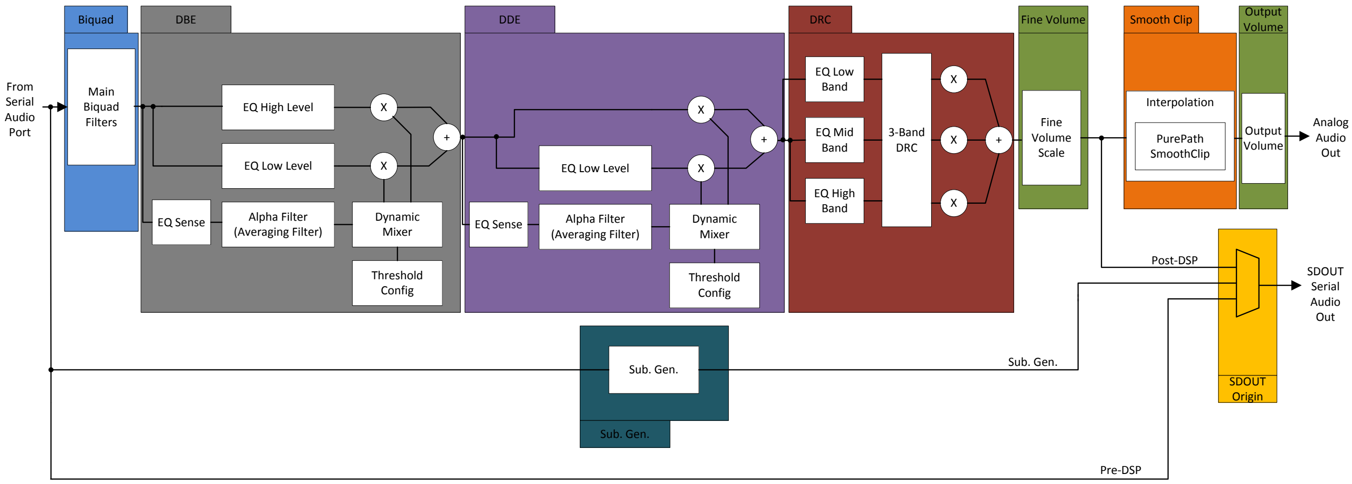


Figure 17. HybridFlow 2 Block Diagram

HybridFlow 2 target applications include stereo audio devices such as DTVs and general audio applications where a subwoofer or low-frequency amplifier is easily added. HF2 supports up to 48-kHz sample rate.

HF2 differs from HF1 in that Dynamic Dialog Enhancer (DDE) processing has been added as well as a third SDOUT origin form Sub Gen. Sub Gen contains audio processing designed for subwoofer applications where digital audio is sent to an external sub amplifier through SDOUT. No additional processing for the sub channel is necessary. HF2 does not have Psychoacoustic Bass Enhancement.

Both Channel A and B are identical and follow the HF2 block diagram. That is, changing the coefficients in any of the processing blocks in Figure 17 automatically applies the change to Channel A and B. The only channel-independent control is the output volume that is set in the PurePath Console GUI.

6.1 Biquad

The Biquad filter block contains 10 independent filters designed for tuning the frequency response of the overall system. This is where the bulk of the frequency compensation occurs; select the filter type, subtype, and other parameters for each of the 10 Biquads. Complex tuning shapes can be made to compensate for deficiencies in speaker response with a goal of a flat response over the frequency band of interest.

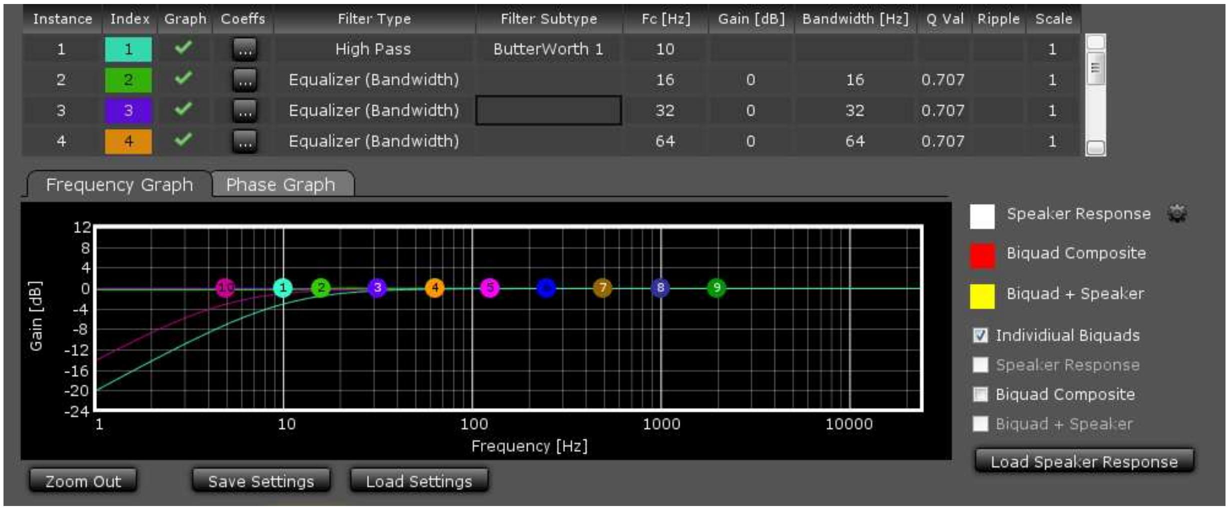


Figure 18. HF2 Biquad Tuning Window

Figure 18 shows the Biquad audio processing window in the PurePath Console GUI. Drag and drop each Biquad into position using the mouse on the plotting window. Other parameters must be typed into the chart above the plotting space. After which, pressing *Enter* on the keyboard causes the change to take effect.

Filter Type and *Filter Subtype* are drop-down menus activated when this space is clicked for the desired Biquad. Remove or add Biquads by clicking on the "graph" space next to the desired Biquad.

The *Phase Graph* shows the phase response for each of the individual Biquads.

The *Biquad Composite* check box shows the overall response based on the position of the individual Biquad filters. This composite view is the frequency response alteration applied to the incoming digital audio data. View this independently by deselecting *Individual Biquads*.

6.1.1 Load Speaker Response

If desired, load a measured speaker response into the Biquad window by clicking *Load Speaker Response*. This aids in tuning the frequency response. With a speaker response loaded, view the overall audio system response by clicking *Biquad + Speaker*. This takes into account the added Biquads as well as the natural response of the speaker.

The file format for the speaker response is .txt. Each line in the file represents a data point containing frequency (in Hz), SPL (in dB), and Phase (in degrees), separated by a space. Any line that does not begin with a number is ignored.

NOTE: A loaded speaker response is not loaded into the HybridFlow. It is only displayed in the PurePath Console GUI as a tuning tool.

6.2 DBE

Dynamic Bass Enhancement (DBE) is a processing block that allows for optimizing the bass response of the system. Two signal paths (low level and high level) 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 DBE channels is dynamic in nature and depends on the incoming audio. Figure 19 shows the tuning window in the PurePath Console GUI.

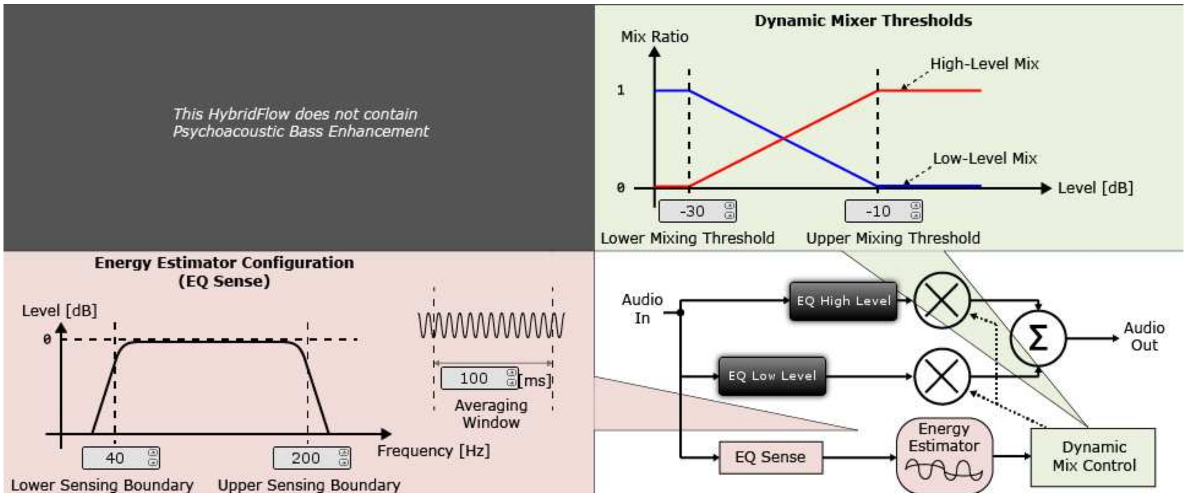


Figure 19. HF2 DBE Tuning Window

6.2.1 Dynamic Mixer Thresholds

The mixing of the two paths (low level and high level) is controlled by setting the *Upper Mixing Threshold* and *Lower Mixing Threshold*. When the averaged signal (as set by the *Averaging Window*) is below the lower mixing threshold, the *Dynamic Mixer* sends all of the audio through the low-level path. When the signal is above the upper mixing threshold, it is sent through the upper-level path. When the signal is between the two, it is mixed together by the *Dynamic Mixer*.

6.2.2 Energy Estimator Configuration (EQ Sense)

Another key configuration for the dynamic mixer block is the EQ sense and alpha filter. In HF2, the EQ sense is a single bandpass filter. The bandwidth of the filter is set by entering in the lower and upper sensing boundary in the GUI. This tells the dynamic mixer which frequency range of the incoming signal to average in order to determine how the signal compares to the mixing thresholds.

The *Averaging Window* or alpha filter works similarly to that of a DRC. It 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.

6.2.3 EQ Low Level

The low-level path contains 2 configurable Biquads to establish the EQ curve which the audio is sent through when the time average signal is at a low-level. These fully-functional Biquads can be assigned to several filter types or sub-types. This determines frequency response when low-level is active based on the Energy Estimator Configuration and the mixing threshold.

Click the *EQ High LEVEL* button to display the tuning window (Figure 20). The tuning shown for the EQ High Level is a simple high-pass filter with fast roll off. For more details visit Section 6.1, Biquad.

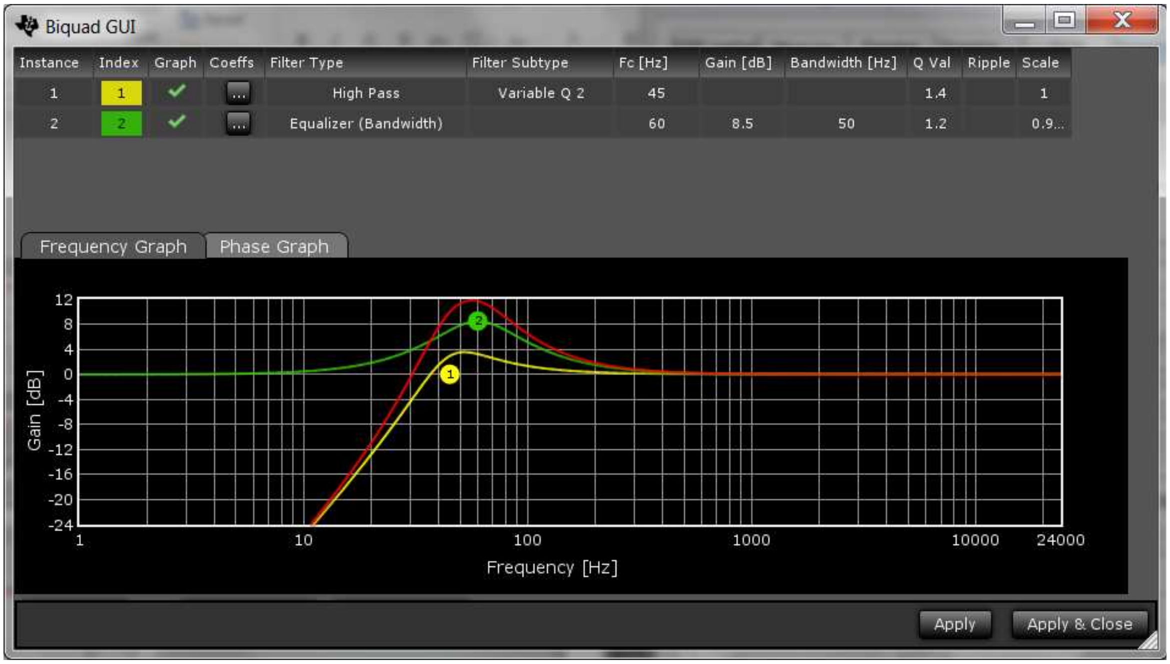


Figure 20. DBE EQ Low-Level Tuning Window

6.2.4 EQ High Level

The high-level path, similar to the low-level path, has two Biquads which can set the EQ curve used when the time averaged input signal is above the upper mixing threshold.

Click the *EQ High LEVEL* button to display the tuning window (Figure 21). The tuning shown for the EQ High Level is a simple high-pass filter with fast roll off. For more details visit Section 6.1, Biquad.

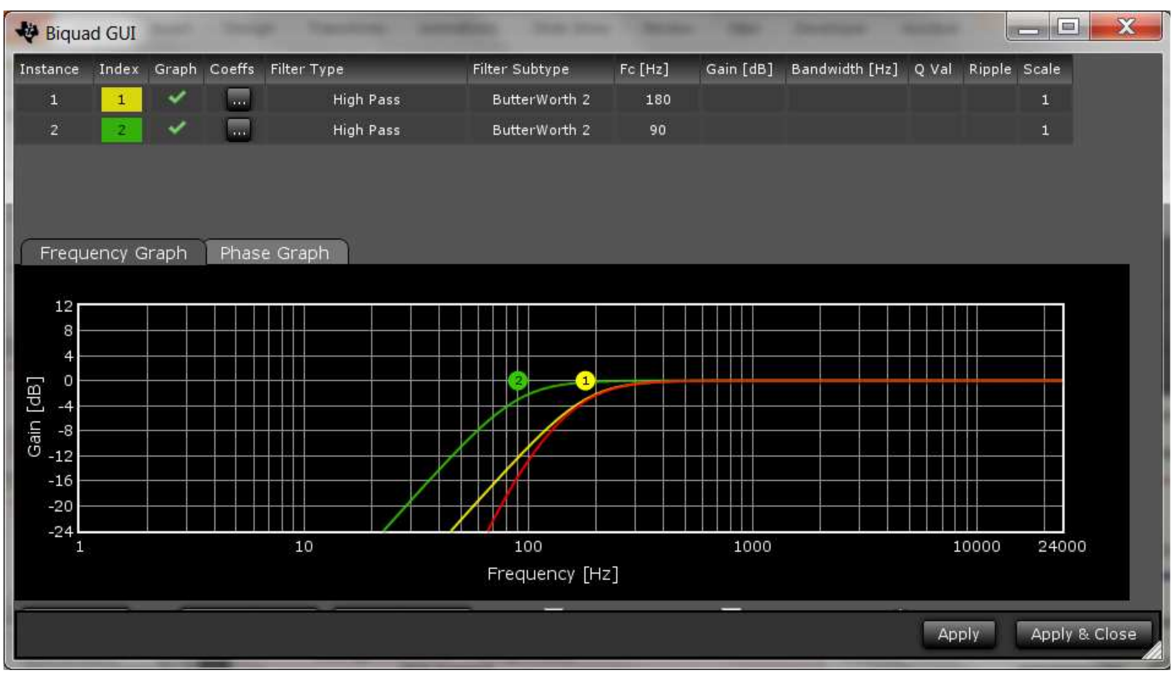


Figure 21. DBE EQ High-Level Tuning Window

6.3 DDE

A dedicated Dynamic Dialog Enhancement (DDE) processing is proprietary to HybridFlow 2, although DBE can be configured as DDE. DDE is similar to DBE except for the frequency range of interest. Here audio filters are added to the fundamental vocal range from 100–400 Hz and to the sibilance range from 6–8 kHz. For DDE, the high-level path is set as a bypass and the low-level contains the filtering. Just like DBE, 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 DDE channels is dynamic in nature and depends on the incoming audio. Figure 22 shows the tuning window in the PurePath Console GUI.

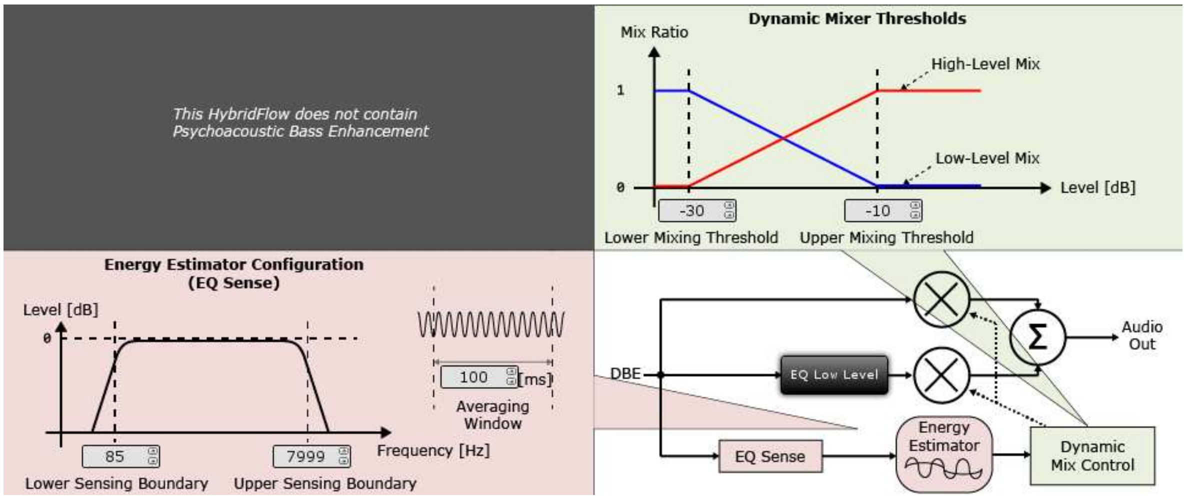


Figure 22. HF2 DDE Tuning Window

6.3.1 Dynamic Mixer Thresholds

The mixing of the two paths (low level and high level) is controlled by setting the *Upper Mixing Threshold* and *Lower Mixing Threshold*. When the averaged signal (as set by the *Averaging Window*) is below the lower mixing threshold, the *Dynamic Mixer* sends all of the audio through the low-level path. When the signal is above the upper mixing threshold, it is sent through the upper-level path. When the signal is between the two, it is mixed together by the *Dynamic Mixer*.

6.3.2 Energy Estimator Configuration (EQ Sense)

Another key configuration for the dynamic mixer block is the EQ sense and alpha filter. In HF2, the EQ sense is a single bandpass filter. The bandwidth of the filter is set by entering in the lower and upper sensing boundary in the GUI. This tells the dynamic mixer which frequency range of the incoming signal to average in order to determine how the signal compares to the mixing thresholds. For DDE, the upper boundary is best set to 20 kHz.

The *Averaging Window* or alpha filter works similarly to that of a DRC. It 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. For vocals, reducing the range allows the Dialog Enhancer to respond more quickly, but listening tests are important to make sure there are no audible artifacts.

6.3.3 EQ Low Level

The low-level path contains 2 configurable Biquads to establish the EQ curve which the audio is sent through when the time average signal is at a low-level. These fully-functional Biquads can be assigned to several filter types or sub-types. This determines frequency response when low-level is active based on the Energy Estimator Configuration and the mixing threshold.

Click the *EQ LOW LEVEL* button to display the tuning window (Figure 23). The tuning shown for the EQ Low Level is chosen as the suggested boost in the vocal range to increase speech intelligibility. For more details visit Section 6.1, Biquad.

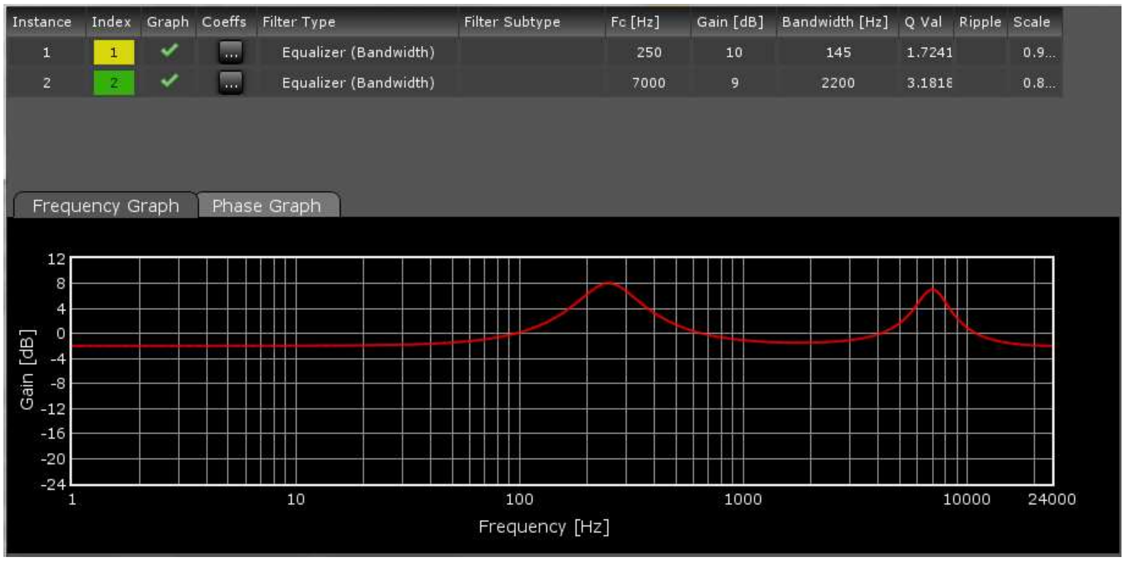


Figure 23. DDE EQ Low-Level Tuning Window

6.4 DRC Standard 3-Band Dynamic Compander

HybridFlow 2 features a 3-band DRC compander (compression and expansion). The DRC is 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.

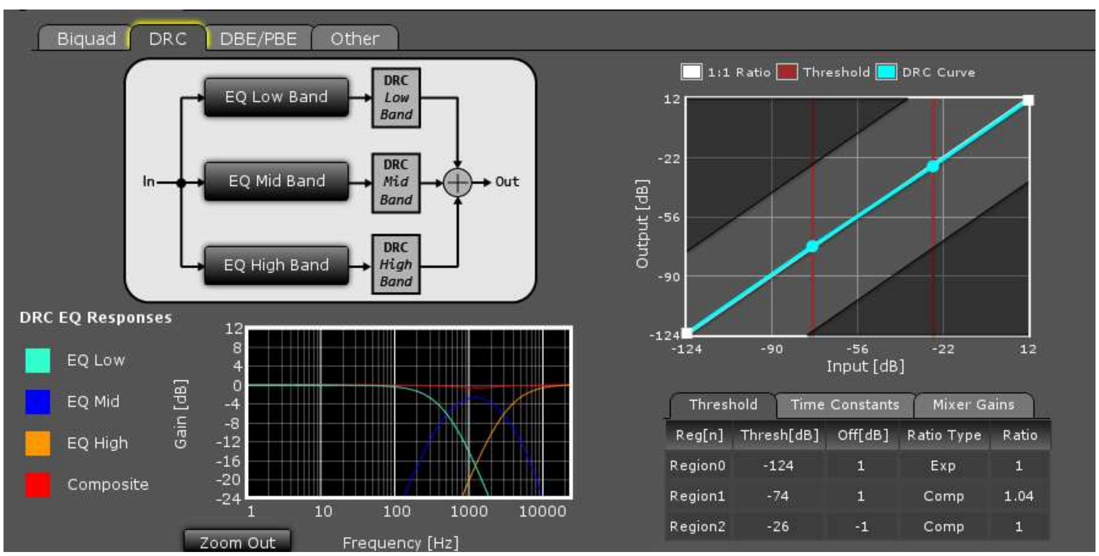
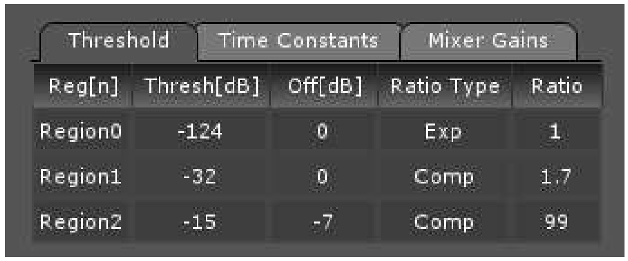


Figure 24. HF2 3-Band DRC Compander Tuning Window

On the right side of the window in Figure 24 is the DRC curve which offers 3 regions of compression. The points on the DRC curve can be dragged and dropped. At the top center of the window is a block diagram of the DRC as well as 3 configurable frequency bands (Low, Mid, and High).

6.4.1 DRC Threshold Tab

Below the DRC window, parameters such as threshold, offset, expansion or compression, and the ratio value are manually entered for each of the 3 regions under the *Threshold* tab. By typing a value and pressing *Enter* on the keyboard, the DRC curve automatically adjusts to the entered parameter.

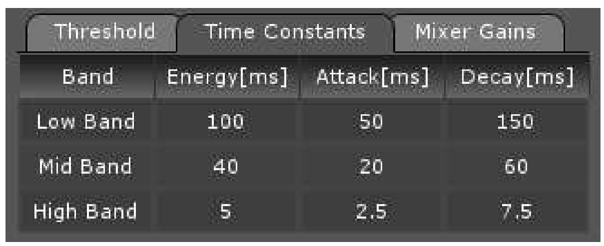

Threshold | Time Constants | Mixer Gains

Reg[n]	Thresh[dB]	Off[dB]	Ratio Type	Ratio
Region0	-124	0	Exp	1
Region1	-32	0	Comp	1.7
Region2	-15	-7	Comp	99

Figure 25. HF2 DRC Threshold Control Tab for 3-Band Compander

6.4.2 DRC Time Constants Tab

The standard 3-band compander offers splitting of the incoming audio into 3 frequency bands determined by the user. Although the same DRC curve gets applied to all 3 frequency bands, different attack time constants can be associated with each band to optimize audio quality and speaker protection. Change time constants by clicking on the *Time Constants* tab (Figure 26) and enter new values for each band.


Threshold | Time Constants | Mixer Gains

Band	Energy[ms]	Attack[ms]	Decay[ms]
Low Band	100	50	150
Mid Band	40	20	60
High Band	5	2.5	7.5

Figure 26. HF2 DRC Time Constants Tab for 3-Band Compander

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. *Attack[ms]* determines the attack time of the DRC and *Decay[ms]* determines the release time once the windowed energy band passes.

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.

For example, a very fast time constant on a low-frequency signal may cause the DRC to attack and release before a full cycle of the incoming signal has passed. Then when the next peak of the wave passes through the DRC, it again attacks and then releases as the peak passes. The DRC continuously attacks and releases rather than enveloping the signal causing audible distortion.

With separate time constants, the standard 3-band compander can still have a very fast time constant at high frequencies and a slower time constant at low frequencies enveloping and compressing the entire audible range quickly and effectively.

6.4.3 Mixer Gains Tab

The mixer gain controls the relative gain of each of the 3 frequency bands when they are mixed together. Use this 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.

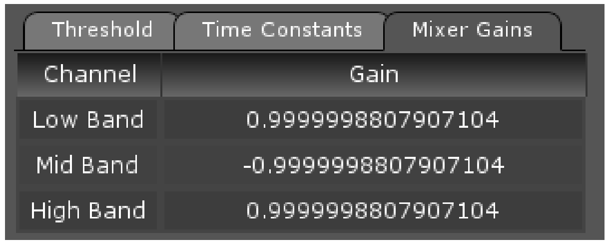


Figure 27. HF2 Mixer Gains Tab for 3-Band Compander

6.4.4 Band Splitting

Configure the frequency range associated with each of the 3 bands used by the *Time Constants* tab by clicking on the *EQ Low Band*, *EQ Mid Band*, and *EQ High Band* buttons. Here a Biquad window appears where the tuning can take place. After tuning, the response is automatically displayed in the 3-Band DRC Compander tuning window on the bottom left.

For more details visit Section 6.1, Biquad.

6.5 Fine Volume

The fine volume control is a digital volume control that allows adjustment between –0.25 dB and 0.25 dB by setting the slider. Both Channel A and B are set simultaneously.

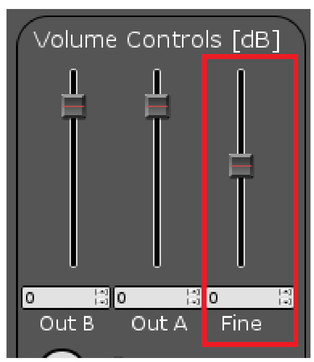


Figure 28. Fine Volume Control

6.6 PurePath SmoothClip

SmoothClip works as a comparator in the digital domain on a sample-by-sample basis. If the incoming audio data word is larger than the set comparator coefficient, the set coefficient is passed until the incoming audio data word is below the set coefficient. This effectively clips the signal. Unlike typical digital clipping which occurs at the sample rate (Fs), SmoothClip operates at very high speeds, minimizing the unwanted distortions associated with digital clipping.

This is often used in conjunction with slower DRC time constants. With a more gradual time constant and compression ratio, the potential for DRC beating or “pumping” is reduced and sound quality and dynamics are improved. However, due to the slow DRC response, a few cycles of incoming audio data that are greater than the set DRC thresholds can pass through. With SmoothClip following a DRC, these cycles can be clipped in a well-controlled fashion to prevent speaker damage until the DRC has attacked the signal.

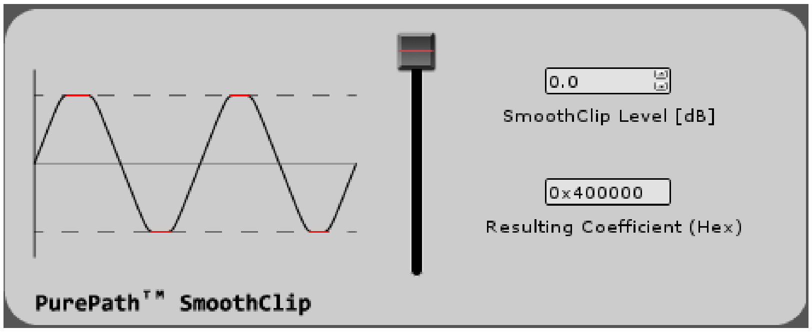


Figure 29. SmoothClip Tuning Window

SmoothClip only has one tuning parameter; the level at which the clipping occurs. TI recommends setting the level by measuring the THD+N at the frequency most boosted by the overall system since this frequency is clipped the most.

6.7 Output Volume

The output volume controls the digital level of both Channel A and Channel B independently from –103 dB to 24 dB by setting the slider. In HybridFlow 2, where Channel A and B are identical, for most applications channels A and B are adjusted together to avoid mismatch on stereo speakers.

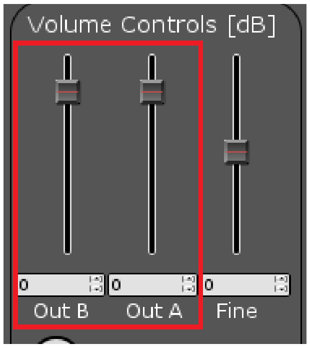


Figure 30. Output Volume Control

6.8 Sub. Gen.

Sub Gen is a digital output-only audio processing block designed for subwoofer output generation. It is a mono audio processor where a copy of the input serial audio port data gets mixed by (A + B) / 2. Therefore, only one processing block is shown in the HybridFlow diagram.

Sub Gen is designed so that the TAS5754/6M devices provide the central processing for both the full-range channels, A and B, as well as a mono subwoofer channel which simplifies the end design. After the Sub Gen tuning has been set, the digital data can be selected through SDOUT and sent to a separate subwoofer amplifier. When using Sub Gen in this manner, the separate subwoofer amplifier requires no audio processing.

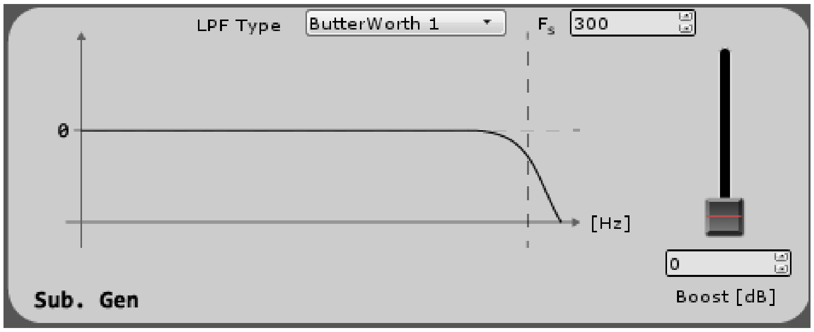


Figure 31. HF2 Sub Gen Tuning Window

Set a low-pass filter at the desired cutoff frequency in the Sub Gen tuning window and select the filter type from the drop-down menu. Add a boost to match the efficiency of a subwoofer speaker channel to the efficiency of the full-range channels. The digital audio can then be passed to SDOUT.

The digital format for Sub Gen is synchronized to the clocking on the TAS5754/6M. For a single cycle of LRCLK, the output word is available twice since it represents a mono signal. There are still separate Left and Right words for each cycle of LRCLK however the words are identical. Therefore, It is up to the designer to choose when to read the word on the device monitoring SDOUT.

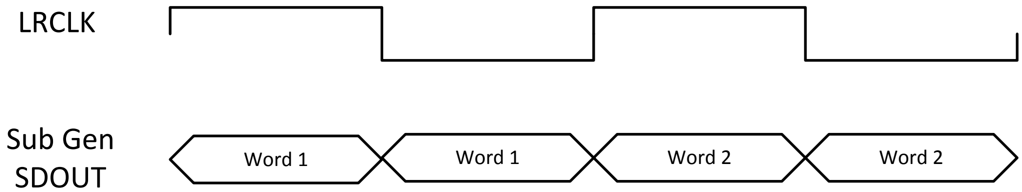


Figure 32. Sub Gen Data Format

6.9 SDOUT Serial Audio Data

In HybridFlow 2 there are 3 choices for digital output sources available on GPIO2 (pin 21 on the TAS5754/6M devices). The first option, Pre-DSP, passes the incoming digital data to SDOUT. There is no processing of the data. The second option, Sub Gen, passes the digital data processed by the Sub Gen block. The third option, Post-DSP, allows for the fully processed data to be available for use. In post-DSP, Smooth Clip and the a digital output volume control have no effect. SDOUT is useful when trying to connect with other digital input devices and amplifiers.

NOTE: SDOUT acts like a MUX; however, it is a hard mixer meaning it mixes only one of the inputs at a time with a 100% mix ratio. This method creates a more efficient DSP MUX device rather than a traditional switching MUX.

7 HybridFlow 3 (HF3)

Figure 33 depicts the signal path for HybridFlow 3. The shaded tabs correspond to the functions found in the PurePath Console GUI.

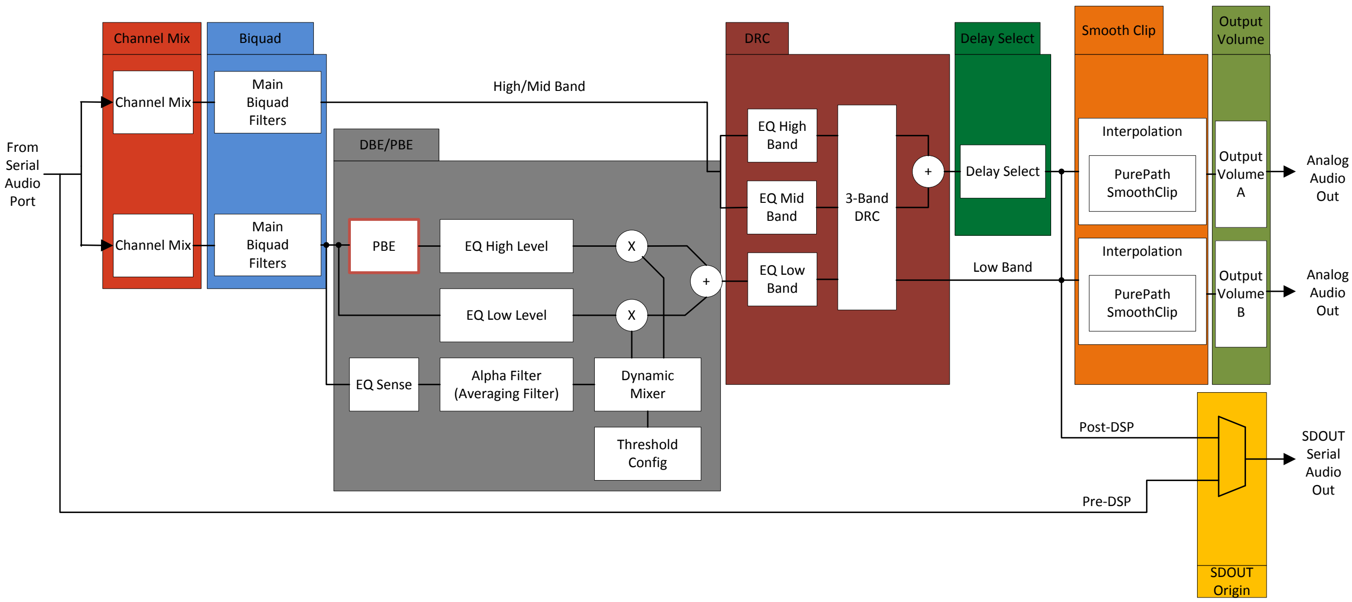


Figure 33. HybridFlow 3 Block Diagram

HybridFlow 3 is unique in that Channel A and Channel B can have different audio processing. In Figure 33, Channel A is the top leg and Channel B is the bottom leg. A target application of HF3 includes 1.1 bi-amping systems for Bluetooth® and active speakers. HF3 supports up to 48-kHz sample rate.

The first processing block is the *Channel Mix* where mixing properties are defined. For a 1.1 system, stereo input data can be mixed into 2 identical mono channels and passed to channels A and B. After the mixer, channels A and B can be processed differently. HF3 is designed so that Channel B is used as a mono subwoofer or low-frequency channel with PBE and DBE in the HybridFlow path. Channel A is used as a mono high-frequency channel. Therefore, Channel A can drive a high-frequency loudspeaker and Channel B a low-frequency loudspeaker for a bi-amplified 1.1 mono system.

7.1 Channel Mix

Channel Mix configures the mixing of the incoming digital stereo audio. The *Med/High Input* is Channel A and the *Low Input* is Channel B. The phase can also be reversed for each of the channels. For a 1.1 system *(L+R)/2* should be selected for both inputs as to create a mono copy of the input digital audio for both Channel A and Channel B.

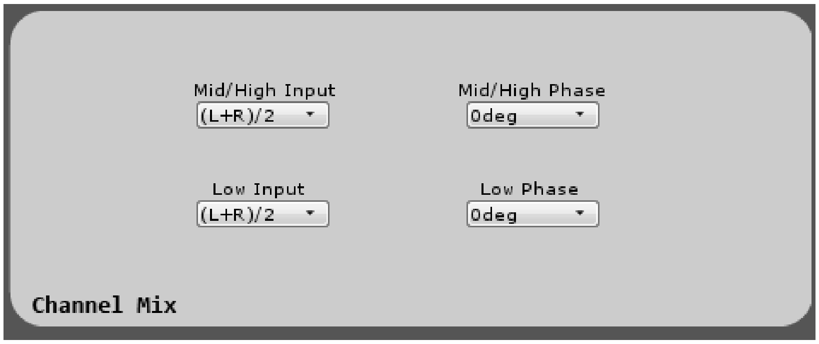


Figure 34. HF3 Channel Mix Window

7.2 Biquad

With HF3 there are two Biquad windows with 5 Biquads for each. These are designed for tuning the frequency response of the overall system. The *High/Mid Band* determines the frequency response for Channel A while the *Low Band* determines the frequency response for Channel B.

This is where the bulk of the frequency compensation occurs; select the filter type, subtype, and other parameters for each of the 5 Biquads. Complex tuning shapes can be made to compensate for deficiencies in speaker response with a goal of a flat response over the frequency band of interest.

Figure 35. HF3 Biquad Tuning Window

Figure 35 shows the Biquad audio processing window in the PurePath Console GUI. Drag and drop each Biquad into position using the mouse on the plotting window. Other parameters must be typed into the chart above the plotting space. After that, pressing *Enter* on the keyboard causes the change to take effect. Select the tuning for Channel A and Channel B by choosing *High/Mid Band* and *Low Band*, respectively, as highlighted in red.

Filter Type and *Filter Subtype* are drop-down menus activated when this space is clicked for the desired Biquad. Remove or add Biquads by clicking on the "graph" space next to the desired Biquad.

The *Phase Graph* shows the phase response for each of the individual Biquads.

The *Biquad Composite* check box shows the overall response based on the position of the individual Biquad filters. This composite view is the frequency response alteration applied to the incoming digital audio data. View this independently by deselecting *Individual Biquads*.

7.2.1 Load Speaker Response

If desired, load a measured speaker response into the Biquad window by clicking *Load Speaker Response*. This aids in tuning the frequency response. With a speaker response loaded, view the overall audio system response by clicking *Biquad + Speaker*. This takes into account the added Biquads as well as the natural response of the speaker.

The file format for the speaker response is .txt. Each line in the file represents a data point containing frequency (in Hz), SPL (in dB), and Phase (in degrees), separated by a space. Any line that does not begin with a number is ignored.

NOTE: A loaded speaker response is not loaded into the HybridFlow. It is only displayed in the PurePath Console GUI as a tuning tool.

7.3 PBE

The next processing block in the HybridFlow is the Psychoacoustic Bass Enhancement (PBE). PBE is only available on Channel B (Low Band). PBE perceptually increases the bass level using the principal of “missing fundamental”, a well-known psychoacoustic effect invoking a perception of the bass frequencies even though the fundamental of those frequencies has been filtered out. The PBE algorithm enhances bass sound for small loudspeakers that are incapable of reproducing bass frequencies efficiently.

Figure 36 shows the block diagram of the PBE block for stereo channels with down-mixed harmonics generation path. The module is composed of input high-pass filters (HPF) and the harmonic generator (containing the harmonic intensity control). The high-pass filter removes frequencies that are irreproducible with the loudspeaker at the high signal level flowing through this path. Those frequencies are attenuated in advance and do not disturb the harmonics generation. This eliminates the irreproducible low-frequency energy in the output signal. The harmonics generator generates harmonics of the low-frequency band selected using the HPF. These are then summed back into the left and right audio paths using the effect intensity control.

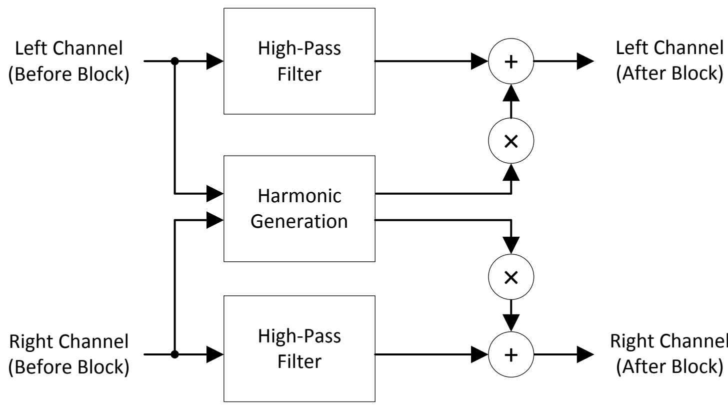


Figure 36. PBE Block Diagram

Figure 37 shows the PBE tuning window in the PurePath Console GUI. There are three controls for the PBE block. The first is a high-pass filter corner frequency *HPF* f_s, determining which fundamental frequencies are removed from the audio output. A second control, called *Harmonic Intensity*, determines how many (the order) Harmonics are created. It is important to note that the 0–100 scale of the Harmonic Intensity is not directly related to the number of harmonics created, but instead a relative number where 100 is the maximum number of harmonics. A value of 0 means no harmonics are added. The third control is *Effect Intensity*. Effect intensity range is from 1.0 to 5.0, in 1-integer steps. This property sets how much of the harmonic content is mixed back into the output signal of the processing block.

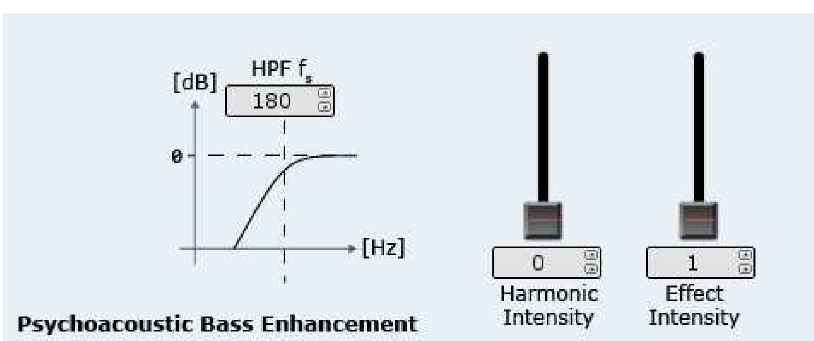


Figure 37. PBE Tuning Window

7.4 DBE

Dynamic Bass Enhancement (DBE) is a processing block that allows for optimizing the bass response of Channel B (Low Band). Two signal paths (low level and high level) 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 DBE channels is dynamic in nature and depends on the incoming audio. Figure 38 shows the tuning window in the PurePath Console GUI.

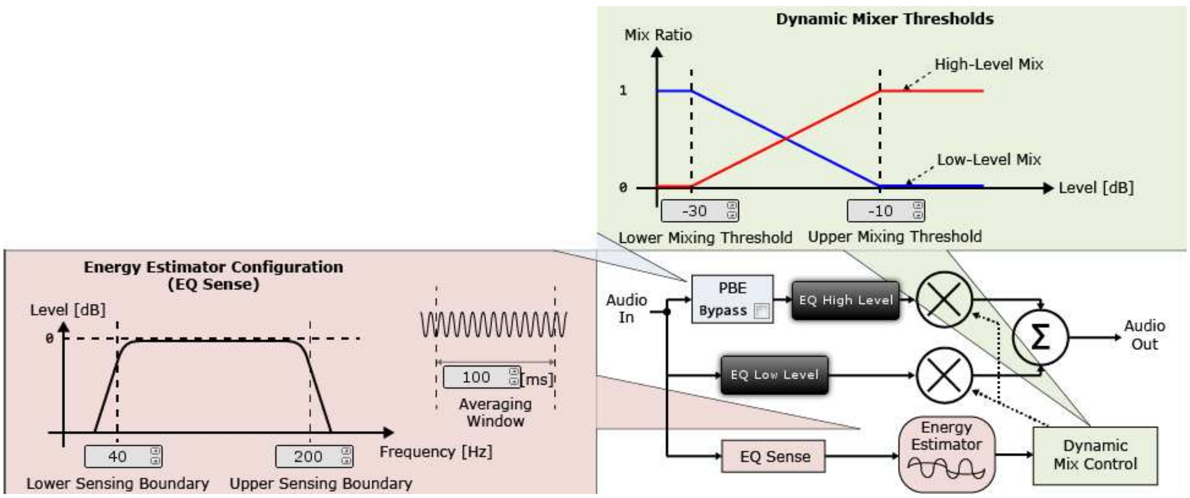


Figure 38. HF3 DBE Tuning Window

7.4.1 Dynamic Mixer Thresholds

The mixing of the two paths (low level and high level) is controlled by setting the *Upper Mixing Threshold* and *Lower Mixing Threshold*. When the averaged signal (as set by the *Averaging Window*) is below the lower mixing threshold, the *Dynamic Mixer* sends all of the audio through the low-level path. When the signal is above the upper mixing threshold, it is sent through the upper-level path. When the signal is between the two, it is mixed together by the *Dynamic Mixer*.

7.4.2 Energy Estimator Configuration (EQ Sense)

Another key configuration for the dynamic mixer block is the EQ sense and alpha filter. In HF3, the EQ sense is a single bandpass filter. The bandwidth of the filter is set by entering in the lower and upper sensing boundary in the GUI. This tells the dynamic mixer which frequency range of the incoming signal to average in order to determine how the signal compares to the mixing thresholds.

The *Averaging Window* or alpha filter works similarly to that of a DRC. It 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.

7.4.3 EQ Low Level

The low-level path contains 3 configurable Biquads to establish the EQ curve the audio is sent through when the time average signal is at a low-level. These fully-functional Biquads can be assigned to several filter types or sub-types. This determines frequency response when low-level is active based on the Energy Estimator Configuration and the mixing threshold.

Click the *EQ LOW LEVEL* button to display the tuning window (Figure 39). The tuning shown for the EQ Low Level is chosen as a bass boost to reduce early onset bass roll off from the speaker. For more details visit Section 7.2, Biquad.

Figure 39. HF3 DBE EQ Low-Level Tuning Window

7.4.4 EQ High Level

The high-level path, similar to the low-level path, has 2 Biquads that can set the EQ curve used when the time averaged input signal is above the upper mixing threshold. However, for HF3, there is also an additional feature allowing harmonic bass (called *Psychoacoustic Bass* or PBE) to be mixed into the output whenever “real” bass must be filtered out. This is explained in Section 7.3, PBE.

Click the *EQ High LEVEL* button to display the tuning window (Figure 40). The tuning shown for the EQ High Level is a simple high-pass filter with fast roll off. For more details visit Section 7.2, Biquad.

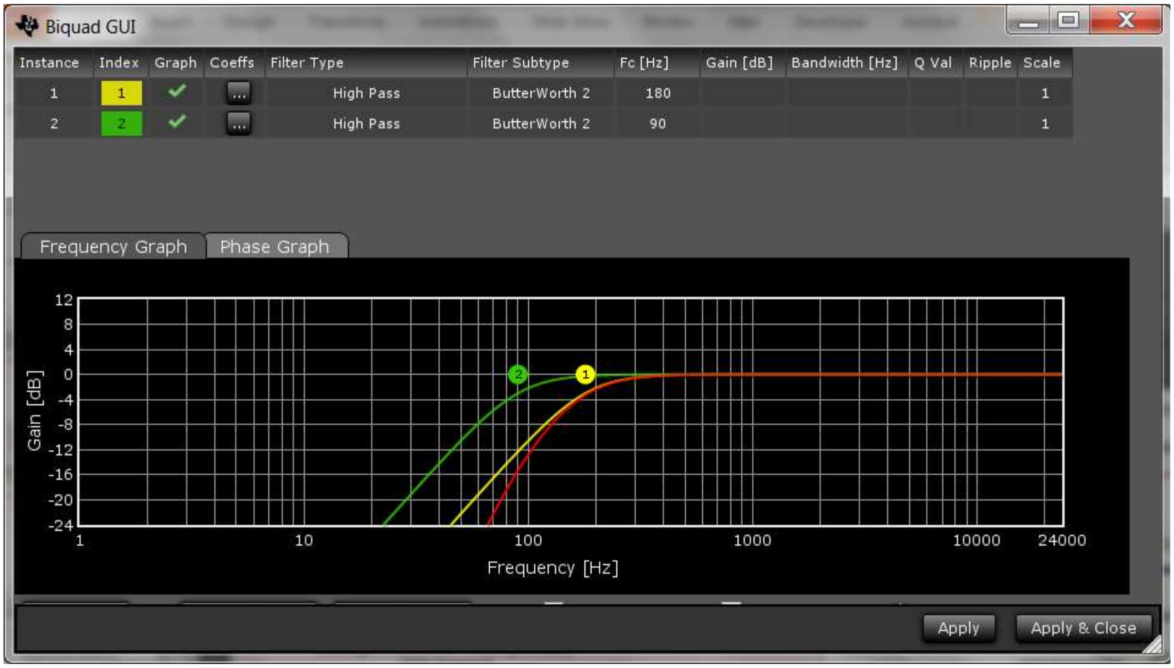


Figure 40. HF3 DBE EQ High-Level Tuning Window

7.5 DRC Standard 3-Band Dynamic Compander

HybridFlow 3 features a 3-band DRC compander (compression and expansion). 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.

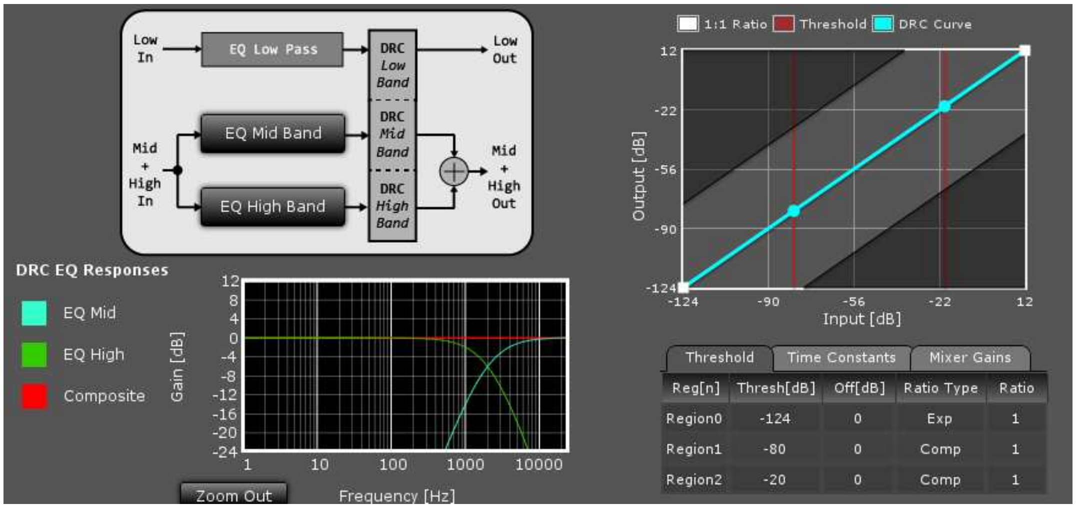


Figure 41. HF3 3-Band HF3 DRC Compander Tuning Window

On the right side of the window in Figure 41 is the DRC curve offering 3 regions of compression. The points on the DRC curve can be dragged and dropped. At the top center of the window is the block diagram of the DRC as well as 3 frequency bands (Low, Mid, and High). In the HF3 DRC, the low-frequency band cannot be defined as it is sourced from Channel B. The Mid and High bands are sourced from Channel A and can be defined.

7.5.1 DRC Threshold Tab

Below the DRC window, parameters such as threshold, offset, expansion or compression, and the ratio value can be manually entered for each of the 3 regions under the *Threshold* tab. By typing a value and pressing *Enter* on the keyboard, the DRC curve automatically adjusts to the entered parameter.

Threshold | Time Constants | Mixer Gains

Reg[n]	Thresh[dB]	Off[dB]	Ratio Type	Ratio
Region0	-124	0	Exp	1
Region1	-32	0	Comp	1.7
Region2	-15	-7	Comp	99

Figure 42. HF3 DRC Threshold Control Tab for 3-Band Compander

7.5.2 DRC Time Constants Tab

The standard 3-band compander in HF3 offers splitting of the incoming audio into 2 frequency bands determined by the user. The third low-frequency band cannot be configured as it comes from Channel B. The Mid Band and High Band are sourced from Channel A. Although the same DRC curve gets applied to all 3 frequency bands, different attack time constants can be associated with each band to optimize audio quality and speaker protection. Change time constants by clicking on the *Time Constants* tab (Figure 43) and enter new values for each band.

Threshold | Time Constants | Mixer Gains

Band	Energy[ms]	Attack[ms]	Decay[ms]
Low Band	100	50	150
Mid Band	40	20	60
High Band	5	2.5	7.5

Figure 43. HF3 DRC Time Constants Tab for 3-Band Compander

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. *Attack[ms]* determines the attack time of the DRC and *Decay[ms]* determines the release time once the windowed energy band passes.

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.

For example, a very fast time constant on a low-frequency signal may cause the DRC to attack and release before a full cycle of the incoming signal has passed. Then when the next peak of the wave passes through the DRC, it again attacks and then releases as the peak passes. The DRC continuously attacks and releases rather than enveloping the signal causing audible distortion.

With separate time constants, the standard 3-band compander can still have a very fast time constant at high frequencies and a slower time constant at low frequencies enveloping and compressing the entire audible range quickly and effectively.

7.5.3 Mixer Gains Tab

The mixer gain controls the relative gain the Mid Band and High Band frequency ranges when they are mixed together. Use this to attenuate one of the frequency bands relative to the others, if needed. The Low band cannot be mixed since it is sourced from Channel B and stays isolated, as such.

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.

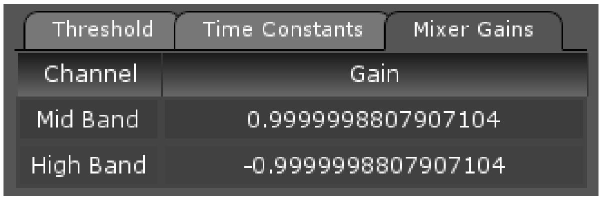


Figure 44. HF3 Mixer Gains Tab for HF3 3-Band Compander

7.5.4 Band Splitting

Configure the frequency range associated with the Mid Band and High Band used by the *Time Constants* tab by clicking on the *EQ Mid Band*, and *EQ High Band* buttons. Here a Biquad window appears where the tuning can take place. After tuning, the response is automatically displayed in the 3-Band DRC Compander tuning window on the bottom left. The frequency range associated for the *Low Band* comes from the previous processing blocks of Channel B.

For more details visit Section 7.2, Biquad.

7.6 Delay Select

HybridFlow 3 offers a Delay Select in series with Channel A. This allows time aligning Channel A with Channel B due to the delay associated with PBE and DBE processing only available with Channel B. Channel A can be delayed by up to 16 samples.

Figure 45. HF3 Delay Select Window

7.7 PurePath SmoothClip

After Delay select, both Channel A and B pass through Smooth Clip which effects both channels equally. Smooth Clip works as a comparator in the digital domain on a sample-by-sample basis. If the incoming audio data word is larger than the set comparator coefficient, the set coefficient is passed until the incoming audio data word is below the set coefficient. This effectively clips the signal. Unlike typical digital clipping which occurs at the sample rate (Fs), SmoothClip operates at very high speeds minimizing the unwanted distortions associated with digital clipping.

This is often used in conjunction with slower DRC time constants. With a more gradual time constant and compression ratio, the potential for DRC beating or “pumping” is reduced and sound quality and dynamics are improved. However, due to the slow DRC response, a few cycles of incoming audio data that are greater than the set DRC thresholds can pass through. With SmoothClip following a DRC, these cycles can be clipped in a well-controlled fashion to prevent speaker damage until the DRC has attacked the signal.

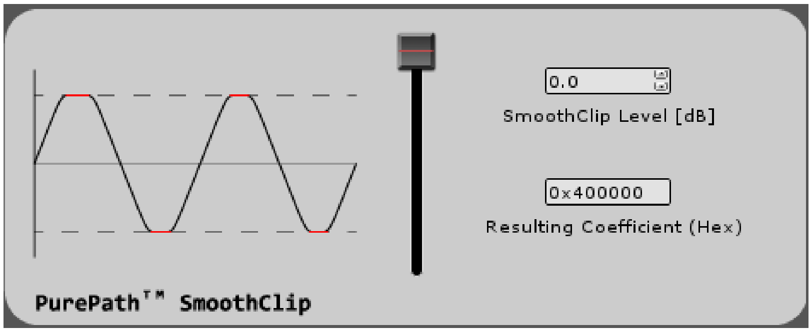


Figure 46. SmoothClip Tuning Window

SmoothClip only has one tuning parameter; the level at which the clipping occurs. TI recommends setting the level by measuring the THD+N at the frequency most boosted by the overall system since this frequency is clipped the most.

7.8 Output Volume

The output volume controls the digital level of both Channel A and Channel B independently from –103 dB to 24 dB by setting the slider. In HybridFlow 3, use the volume control to help match different driver efficiencies of a 1.1 system.

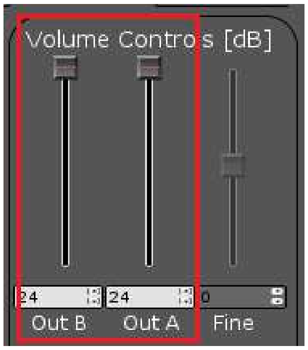


Figure 47. HF3 Output Volume

7.9 SDOUT Serial Audio Data

In HybridFlow 3 there are 2 choices of digital output sources available on GPIO2 (pin 21 on the TAS5754/6M devices). The first option is Pre-DSP which passes the incoming digital data to SDOUT. There is no processing of the data. The second option is Post-DSP which allows for the fully processed data to be available for use. In post-DSP, Smooth Clip and the analog output volume control have no effect. SDOUT is useful when trying to connect with other digital input devices and amplifiers.

NOTE: SDOUT acts like a MUX; however, it is a hard mixer meaning it mixes only one of the inputs at a time with a 100% mix ratio. This method creates a more efficient DSP MUX device rather than a traditional switching MUX.

8 HybridFlow 4 (HF4)

Figure 48 depicts the signal path for HybridFlow 4. The shaded tabs correspond to the functions found in the PurePath Console GUI.

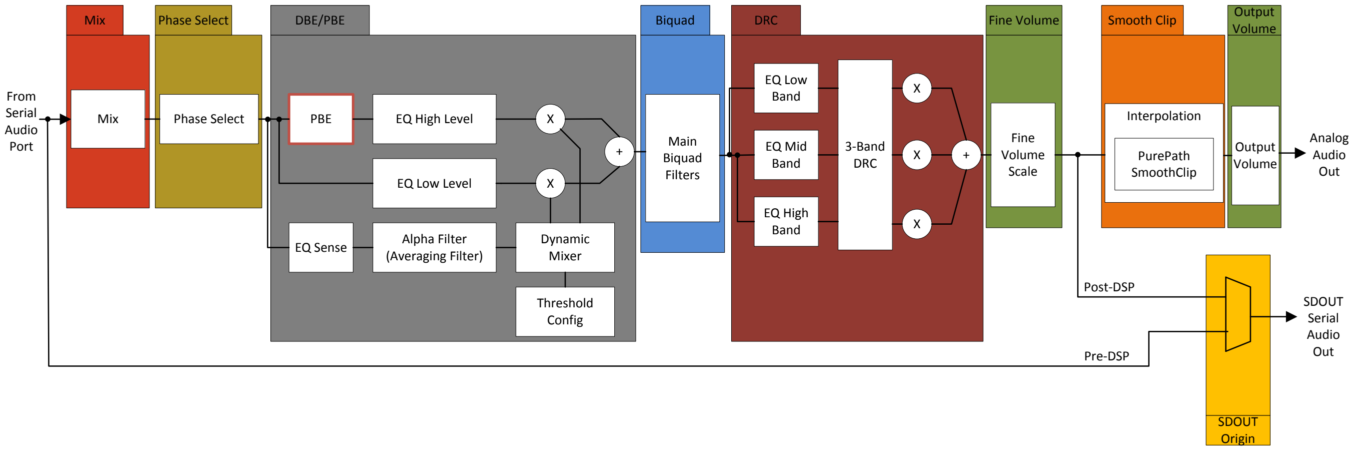


Figure 48. HybridFlow 4 Block Diagram

HybridFlow 4 target applications include mono audio devices such as Bluetooth speakers and wireless subwoofers requiring up to a 48-kHz sample rate. HF4 is a mono only HybridFlow and therefore it can only be loaded onto the mono device in the PurePath Console GUI. This mono path is labeled as Channel C. The first stage of this HybridFlow includes a Mix function which mixes the stereo digital input into a single mono channel of Left only, Right only, or (L+R)/2 audio data.

8.1 Mix and Phase

The mix control selects the data to be used in the rest of the hybrid flow. The choices of audio data are the incoming left signal (L), right signal (R), or a mono mix of left and right (L+R)/2.

Also included in the same tuning window is *Phase* select which allows phase inversion.

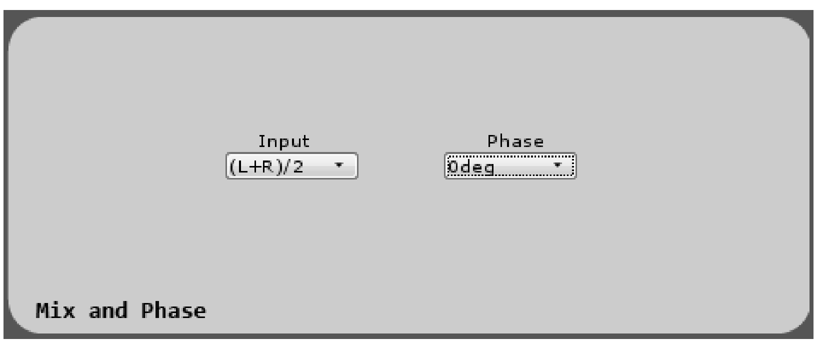


Figure 49. HF4 Mix and Phase Tuning Window

8.2 PBE

The next processing block in the HybridFlow is the Psychoacoustic Bass Enhancement (PBE). PBE perceptually increases the bass level using the principal of “missing fundamental”, a well-known psychoacoustic effect invoking a perception of the bass frequencies even though the fundamental of those frequencies has been filtered out. The PBE algorithm enhances bass sound for small loudspeakers that are incapable of reproducing bass frequencies efficiently.

[Figure 50](#) shows the block diagram of the PBE block for stereo channels with down-mixed harmonics generation path. The module is composed of input high-pass filters (HPF) and the harmonic generator (which contains the harmonic intensity control). The high-pass filter removes frequencies that are irreproducible with the loudspeaker at the high signal level flowing through this path. Those frequencies are attenuated in advance and do not disturb the harmonics generation. This eliminates the irreproducible low-frequency energy in the output signal. The harmonics generator generates harmonics of the low-frequency band selected using the HPF. These are then summed back into the left and right audio paths using the effect intensity control.

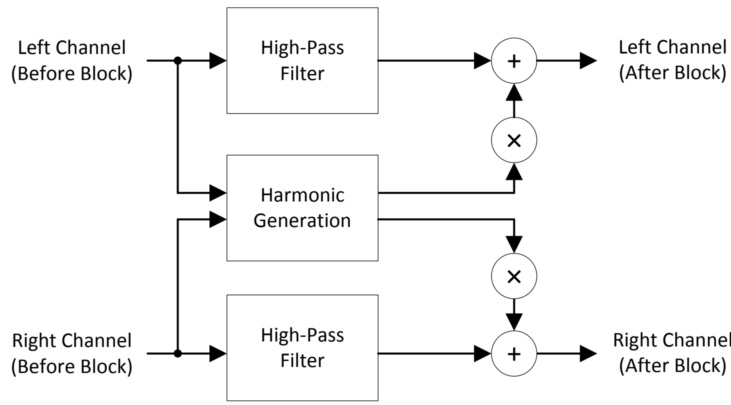


Figure 50. PBE Block Diagram

[Figure 51](#) shows the PBE tuning window in the PurePath Console GUI. There are three controls for the PBE block. The first is a high-pass filter corner frequency *HPF* f_s, determining which fundamental frequencies are removed from the audio output. A second control, called *Harmonic Intensity*, determines how many (the order) Harmonics are created. It is important to note that the 0–100 scale of the Harmonic Intensity is not directly related to the number of harmonics created, but instead a relative number where 100 is the maximum number of harmonics. A value of 0 means no harmonics are added. The third control is *Effect Intensity*. Effect intensity range is from 1.0 to 5.0, in 1-integer steps. This property sets how much of the harmonic content is mixed back into the output signal of the processing block.

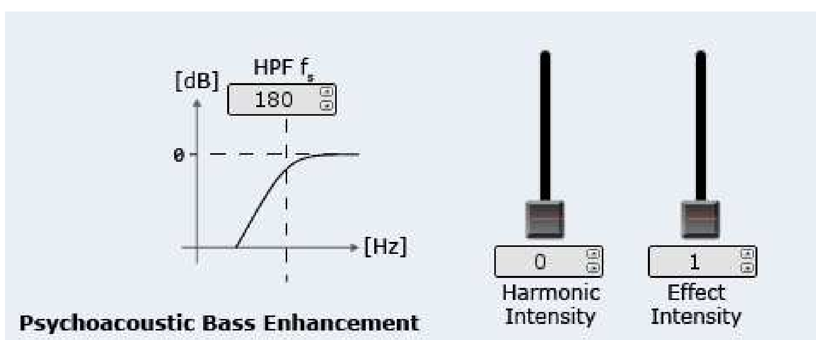


Figure 51. PBE Tuning Window

8.3 DBE

Dynamic Bass Enhancement (DBE) is a processing block that allows for optimizing the bass response of the system. Two signal paths (low level and high level) 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 DBE channels is dynamic in nature and depends on the incoming audio. [Figure 52](#) shows the tuning window in the PurePath Console GUI.

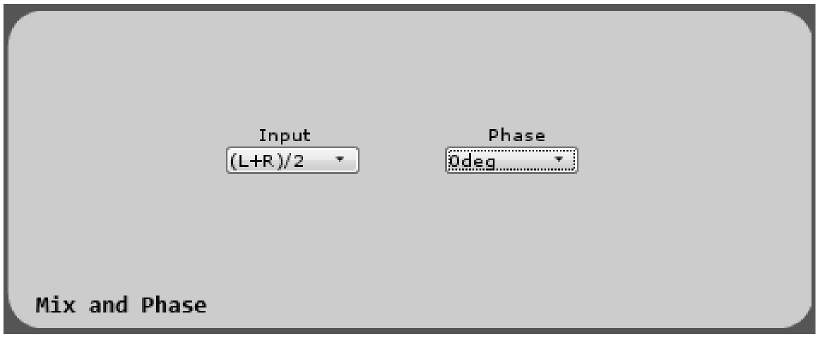


Figure 52. HF4 DBE Tuning Window

8.3.1 Dynamic Mixer Thresholds

The mixing of the two paths (low level and high level) is controlled by setting the *Upper Mixing Threshold* and *Lower Mixing Threshold*. When the averaged signal (as set by the *Averaging Window*) is below the lower mixing threshold, the *Dynamic Mixer* sends all of the audio through the low-level path. When the signal is above the upper mixing threshold, it is sent through the upper-level path. When the signal is between the two, it is mixed together by the *Dynamic Mixer*.

8.3.2 Energy Estimator Configuration (EQ Sense)

Another key configuration for the dynamic mixer block is the EQ sense and alpha filter. The EQ sense is a single bandpass filter. The bandwidth of the filter is set by entering in the lower and upper sensing boundary in the GUI. This tells the dynamic mixer which frequency range of the incoming signal to average in order to determine how the signal compares to the mixing thresholds.

The *Averaging Window* or alpha filter works similarly to that of a DRC. It 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.

8.3.3 EQ Low Level

The low-level path contains 4 configurable Biquads to establish the EQ curve which the audio is sent through when the time average signal is at a low-level. Assign these fully-functional Biquads to several filter types or sub-types. This determines frequency response when low-level is active based on the Energy Estimator Configuration and the mixing threshold.

Click the *EQ LOW LEVEL* button to display the tuning window (Figure 53). The tuning shown for the EQ Low Level is chosen as a bass boost to reduce early onset bass roll off from the speaker. For more details visit Section 8.4, Biquad.

Figure 53. HF4 DBE EQ Low-Level Tuning Window

8.3.4 EQ High Level

The high-level path, similar to the low-level path, has 4 Biquads which can set the EQ curve used when the time averaged input signal is above the upper mixing threshold. However, for HF4, there is also an additional feature which allows harmonic bass (called *Psychoacoustic Bass* or PBE) to be mixed into the output whenever "real" bass must be filtered out. This is explained in Section 8.2, PBE.

Click the *EQ High LEVEL* button to display the tuning window (Figure 54). The tuning shown for the EQ High Level is a simple high-pass filter with fast roll off. For more details visit Section 8.4, Biquad.

Figure 54. HF4 DBE EQ High-Level Tuning Window

8.4 Biquad

In HF4 there are 5 Biquads. This is where the bulk of the frequency compensation occurs; select the filter type, subtype, and other parameters for each of the 5 Biquads. Complex tuning shapes can be made to compensate for deficiencies in speaker response with a goal of a flat response over the frequency band of interest.

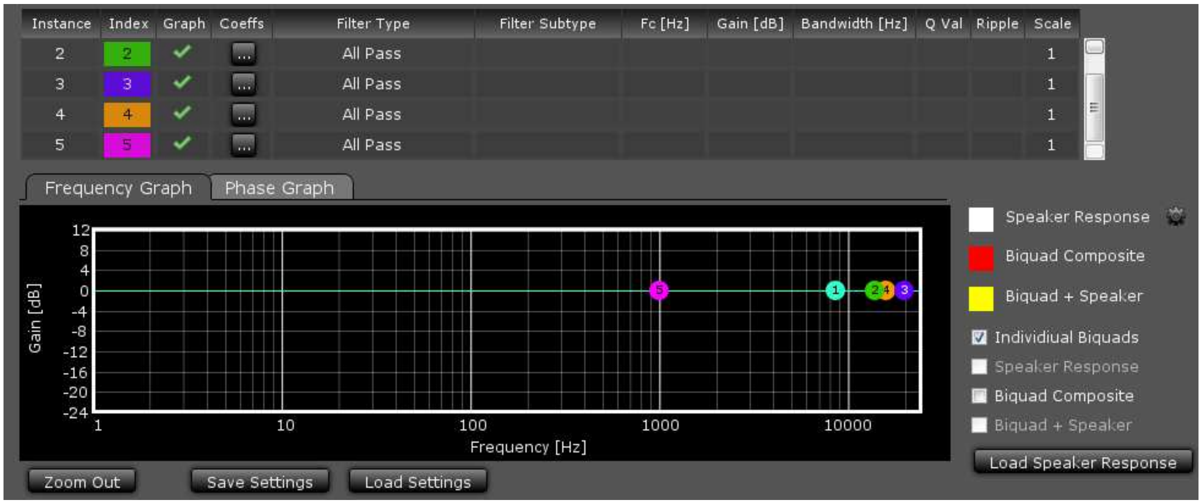


Figure 55. HF4 Biquad Tuning Window

Figure 55 shows the Biquad audio processing window in the PurePath Console GUI. Drag and drop each Biquad into position using the mouse on the plotting window. Other parameters must be typed into the chart above the plotting space. After which, pressing *Enter* on the keyboard causes the change to take effect.

Filter Type and *Filter Subtype* are drop-down menus activated when this space is clicked for the desired Biquad. Remove or add Biquads by clicking on the “graph” space next to the desired Biquad.

The *Phase Graph* shows the phase response for each of the individual Biquads.

The *Biquad Composite* check box shows the overall response based on the position of the individual Biquad filters. This composite view is the frequency response alteration applied to the incoming digital audio data. View this independently by deselecting *Individual Biquads*.

8.4.1 Load Speaker Response

If desired, load a measured speaker response into the Biquad window by clicking *Load Speaker Response*. This aids in tuning the frequency response. With a speaker response loaded, view the overall audio system response by clicking *Biquad + Speaker*. This takes into account the added Biquads as well as the natural response of the speaker.

The file format for the speaker response is .txt. Each line in the file represents a data point containing frequency (in Hz), SPL (in dB), and Phase (in degrees), separated by a space. Any line that does not begin with a number is ignored.

NOTE: A loaded speaker response is not loaded into the HybridFlow. It is only displayed in the PurePath Console GUI as a tuning tool.

8.5 DRC Standard 3-Band Dynamic Compander

HybridFlow 4 features a 3-band DRC compander (compression and expansion). The DRC is 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.

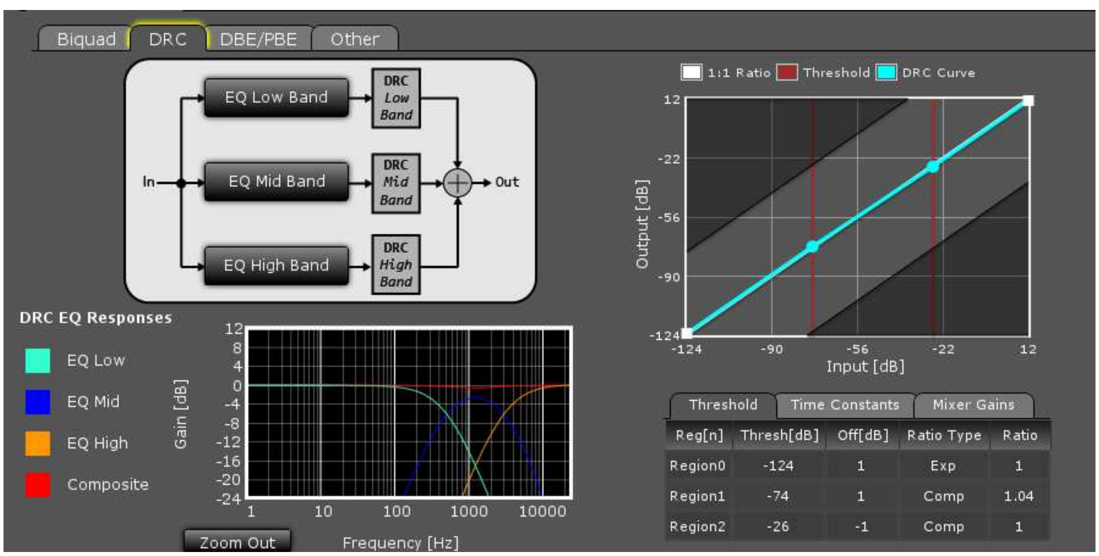


Figure 56. HF4 3-Band DRC Compander Tuning Window

On the right side of the window in Figure 56 is the DRC curve which offers 3 regions of compression. The points on the DRC curve can be dragged and dropped. At the top center of the window is a block diagram of the DRC as well as 3 configurable frequency bands (Low, Mid, and High).

8.5.1 DRC Threshold Tab

Below the DRC window, parameters such as threshold, offset, expansion or compression, and the ratio value can be manually entered for each of the 3 regions under the *Threshold* tab. By typing a value and pressing *Enter* on the keyboard, the DRC curve automatically adjusts to the entered parameter.

Threshold | Time Constants | Mixer Gains

Reg[n]	Thresh[dB]	Off[dB]	Ratio Type	Ratio
Region0	-124	0	Exp	1
Region1	-32	0	Comp	1.7
Region2	-15	-7	Comp	99

Figure 57. HF4 DRC Threshold Control Tab for 3-Band Compander

8.5.2 DRC Time Constants Tab

The standard 3-band compander offers splitting of the incoming audio into 3 frequency bands determined by the user. Although the same DRC curve gets applied to all 3 frequency bands, different attack time constants can be associated with each band to optimize audio quality and speaker protection. Change time constants by clicking on the *Time Constants* tab (Figure 58) and enter new values for each band.

Threshold | Time Constants | Mixer Gains

Band	Energy[ms]	Attack[ms]	Decay[ms]
Low Band	100	50	150
Mid Band	40	20	60
High Band	5	2.5	7.5

Figure 58. HF4 DRC Time Constants Tab for 3-Band Compander

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. *Attack[ms]* determines the attack time of the DRC and *Decay[ms]* determines the release time once the windowed energy band passes.

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.

For example, a very fast time constant on a low-frequency signal may cause the DRC to attack and release before a full cycle of the incoming signal has passed. Then when the next peak of the wave passes through the DRC, it again attacks and then releases as the peak passes. The DRC continuously attacks and releases rather than enveloping the signal causing audible distortion.

With separate time constants, the standard 3-band compander can still have a very fast time constant at high frequencies and a slower time constant at low frequencies enveloping and compressing the entire audible range quickly and effectively.

8.5.3 Mixer Gains Tab

The mixer gain controls the relative gain of each of the 3 frequency bands when they are mixed together. Use this 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.

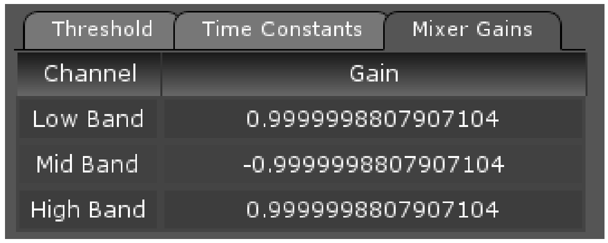


Figure 59. HF4 Mixer Gains Tab for 3-Band Compander

8.5.4 Band Splitting

Configure the frequency range associated with each of the 3 bands used by the *Time Constants* tab by clicking on the *EQ Low Band*, *EQ Mid Band*, and *EQ High Band* buttons. Here a Biquad window appears where the tuning can take place. After tuning, the response is automatically displayed in the 3-Band DRC Compander tuning window on the bottom left.

For more details visit Section 8.4, Biquad.

8.6 Fine Volume

The fine volume control is a digital volume control that allows adjustment between –0.25 dB and 0.25 dB by setting the slider.

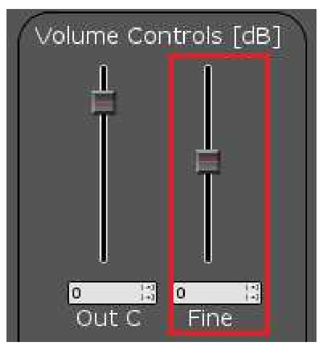


Figure 60. HF4 Fine Volume Control

8.7 PurePath Smooth Clip

SmoothClip works as a comparator in the digital domain on a sample-by-sample basis. If the incoming audio data word is larger than the set comparator coefficient, the set coefficient is passed until the incoming audio data word is below the set coefficient. This effectively clips the signal. Unlike typical digital clipping which occurs at the sample rate (Fs), SmoothClip operates at very high speeds, minimizing the unwanted distortions associated with digital clipping.

This is often used in conjunction with slower DRC time constants. With a more gradual time constant and compression ratio, the potential for DRC beating or “pumping” is reduced and sound quality and dynamics are improved. However, due to the slow DRC response, a few cycles of incoming audio data that are greater than the set DRC thresholds can pass through. With SmoothClip following a DRC, these cycles can be clipped in a well-controlled fashion to prevent speaker damage until the DRC has attacked the signal.

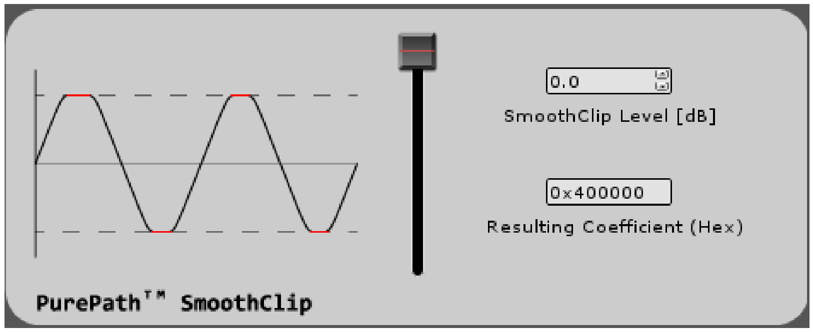


Figure 61. SmoothClip Tuning Window

SmoothClip only has one tuning parameter; the level at which the clipping occurs. TI recommends setting the level by measuring the THD+N at the frequency most boosted by the overall system since this frequency is clipped the most.

8.8 Output Volume

The output volume controls the digital level of Channel C from –103 dB to 24 dB by setting the slider. In HybridFlow 4, Channel C is used since this is the output from the mono device and therefore only once channel is present.

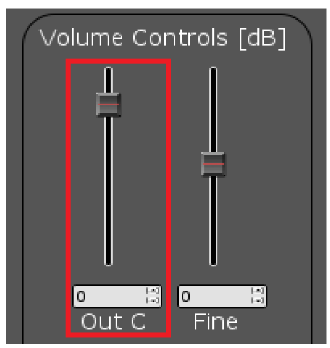


Figure 62. HF4 Output Volume Control

8.9 SDOUT Serial Audio Data

In HybridFlow 4 there are 2 choices of digital output sources available on GPIO2 (pin 21 on the TAS5754/6M devices). The first option is Pre-DSP which passes the incoming digital data to SDOUT. There is no processing of the data. The second option is Post-DSP which allows for the fully processed data to be available for use. In post-DSP, Smooth Clip and the digital output volume control have no effect. SDOUT is useful when trying to connect with other digital input devices and amplifiers.

NOTE: SDOUT acts like a MUX; however, it is a hard mixer meaning it mixes only one of the inputs at a time with a 100% mix ratio. This method creates a more efficient DSP MUX device rather than a traditional switching MUX.

9 HybridFlow 5 (HF5)

Figure 63 depicts the signal path for HybridFlow 5. The shaded tabs correspond to the functions found in the PurePath Console GUI.

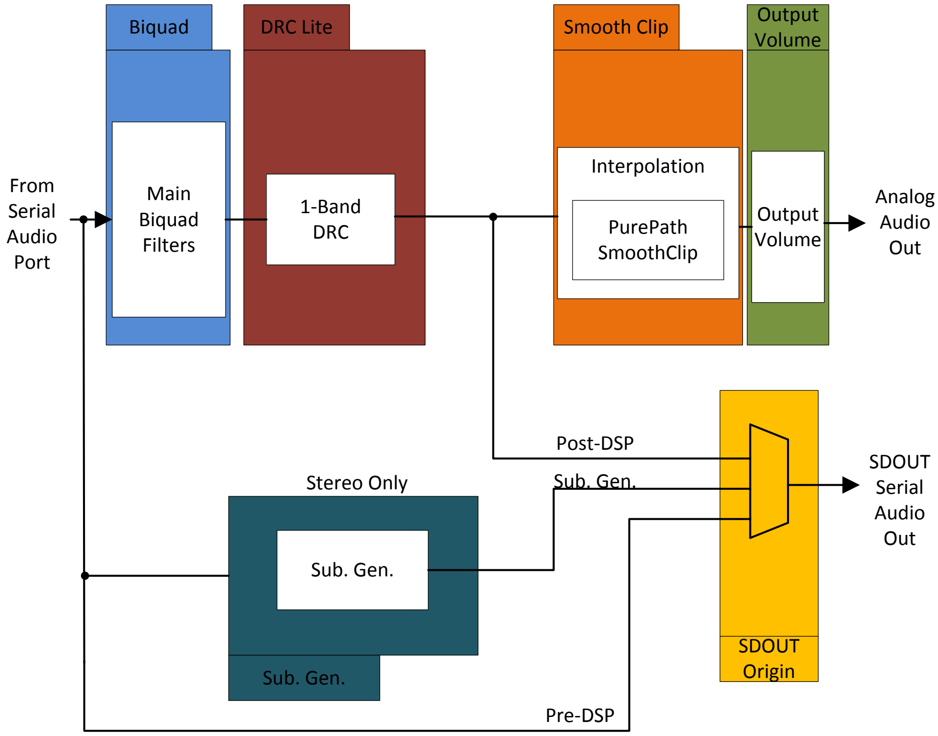


Figure 63. HybridFlow 5 Block Diagram

HybridFlow 5 target applications include high-end DTVs and audio applications requiring up to a 192-kHz sample rate. Simplified audio processing allows for faster sample rates required by high-end audio applications. HF5 is available on the stereo device and the mono device. In stereo operation, Channel A and B are identical and follow the HF5 block diagram. That is, changing the coefficients in any of the processing blocks in Figure 63 automatically applies the change to Channel A and B. The only channel-independent control is the output volume set in the PurePath Console GUI. In mono operation, only one of the incoming digital channels gets passed through the HybridFlow, labeled Channel C. Therefore, to use HF5 on the mono device without losing an audio channel, the digital audio must be premixed into a mono channel before entering HF5. Channel C on the mono device follows the HF5 block diagram with the exception of Sub Gen, which is unavailable.

Loading HF5 on both the stereo and mono device simultaneously allows for a highly configurable 192-kHz, 2.1 system. For this application, the Sub Gen SDOUT output of the stereo device is used as the input into the mono device. Since Sub Gen automatically mixes the incoming stereo digital audio into a (L+R)/2 digital mono channel, using this as an input into the mono device loaded with HF5 increases audio processing of a sub-woofer channel.

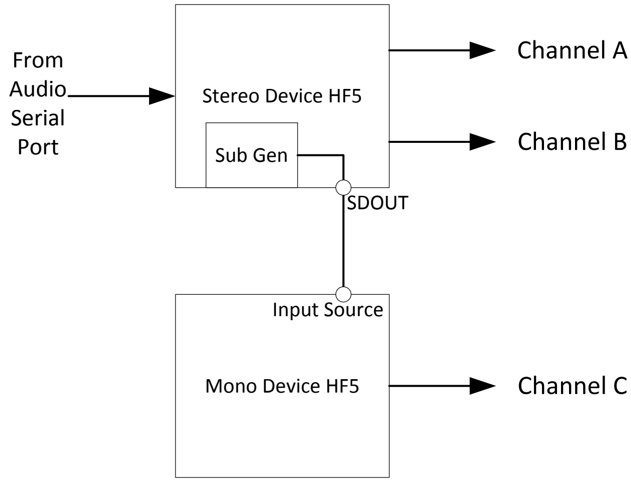


Figure 64. Loading HF5 into both the Stereo and Mono Device

9.1 Biquad

The Biquad filter block contains 1 Biquad filter designed for tuning the frequency response of the overall system. This is where the bulk of the frequency compensation occurs; select the filter type, subtype, and other parameters for the Biquad. Different tuning shapes can be made to compensate for deficiencies in speaker response with a goal of a flat response over the frequency band of interest.

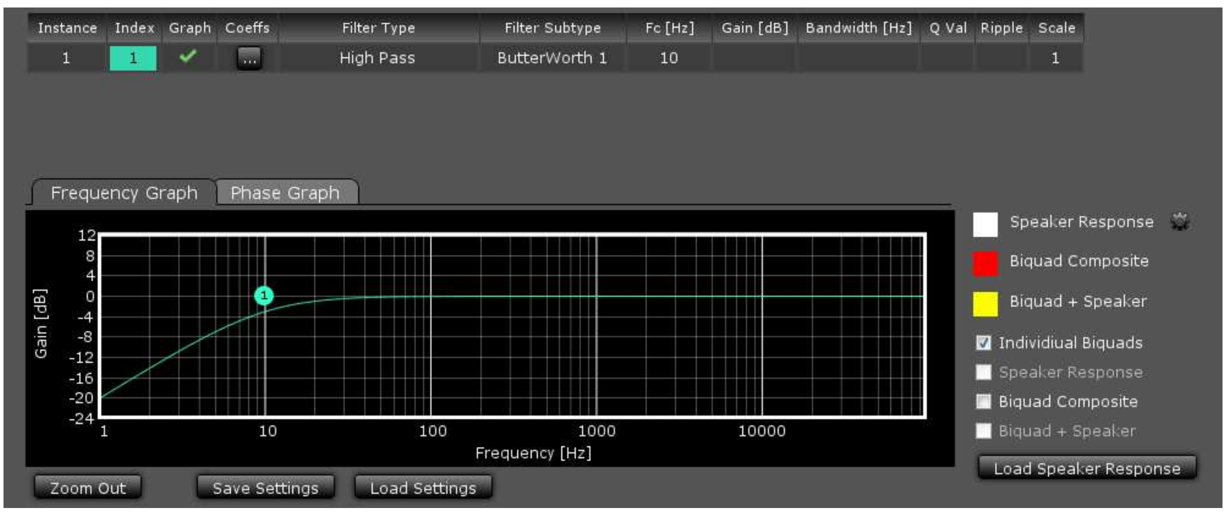


Figure 65. HF5 Biquad Tuning Window

Figure 65 shows the Biquad audio processing window in the PurePath Console GUI. Drag and drop each Biquad into position using the mouse on the plotting window. Other parameters must be typed into the chart above the plotting space. After which, pressing *Enter* on the keyboard causes the change to take effect.

Filter Type and *Filter Subtype* are drop-down menus activated when this space is clicked for the desired Biquad. Remove or add Biquads by clicking on the “graph” space next to the desired Biquad.

The *Phase Graph* shows the phase response for each of the individual Biquads.

The *Biquad Composite* check box shows the overall response based on the position of the individual Biquad filters. This composite view is the frequency response alteration applied to the incoming digital audio data. View this independently by deselecting *Individual Biquads*.

9.1.1 Load Speaker Response

If desired, load a measured speaker response into the Biquad window by clicking *Load Speaker Response*. This aids in tuning the frequency response. With a speaker response loaded, view the overall audio system response by clicking *Biquad + Speaker*. This takes into account the added Biquads as well as the natural response of the speaker.

The file format for the speaker response is .txt. Each line in the file represents a data point containing frequency (in Hz), SPL (in dB), and Phase (in degrees), separated by a space. Any line that does not begin with a number is ignored.

NOTE: A loaded speaker response is not loaded into the HybridFlow. It is only displayed in the PurePath Console GUI as a tuning tool.

9.2 DRC Lite

DRC Lite is a single-band DRC where all audio frequencies have the same time constant associated with the DRC. Change time constants by typing the desired values into the *Time Constants* chart. Three compression ratios are offered (2:1, 4:1, and 8:1), and the threshold can be adjusted from 0 to –60 dB in steps of 1 dB. Once the adjustments are made, the DRC compression curve displays the resulting configuration.

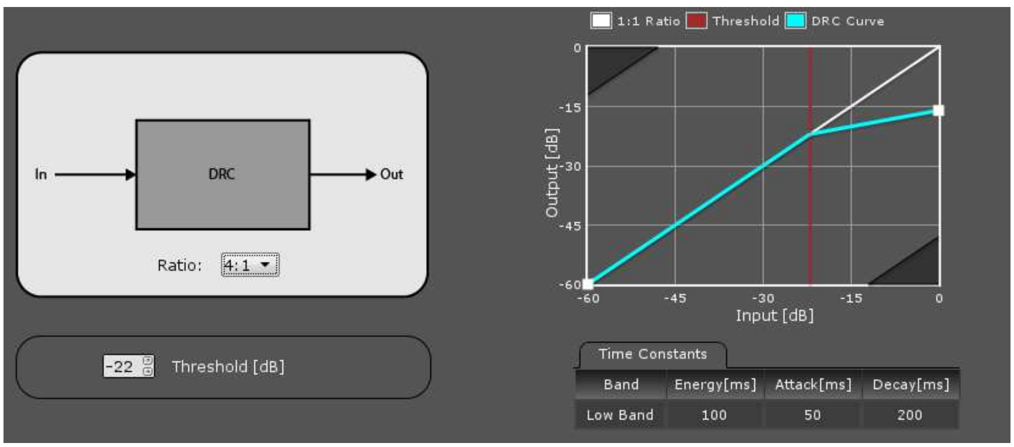


Figure 66. HF5 DRC Lite

9.3 PurePath SmoothClip

SmoothClip works as a comparator in the digital domain on a sample-by-sample basis. If the incoming audio data word is larger than the set comparator coefficient, the set coefficient is passed until the incoming audio data word is below the set coefficient. This effectively clips the signal. Unlike typical digital clipping which occurs at the sample rate (Fs), SmoothClip operates at very high speeds, minimizing the unwanted distortions associated with digital clipping.

This is often used in conjunction with slower DRC time constants. With a more gradual time constant and compression ratio, the potential for DRC beating or “pumping” is reduced and sound quality and dynamics are improved. However, due to the slow DRC response, a few cycles of incoming audio data that are greater than the set DRC thresholds can pass through. With SmoothClip following a DRC, these cycles can be clipped in a well-controlled fashion to prevent speaker damage until the DRC has attacked the signal.

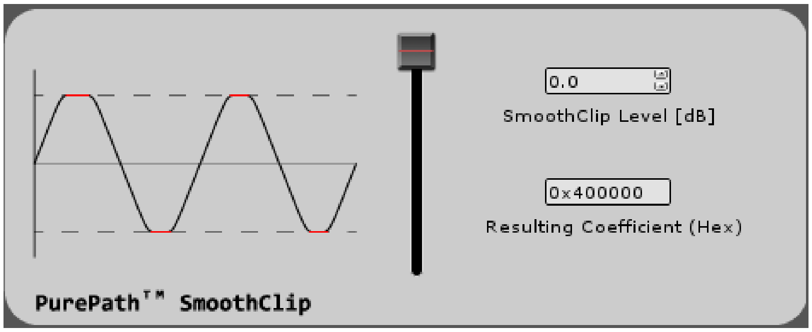


Figure 67. SmoothClip Tuning Window

SmoothClip only has one tuning parameter; the level at which the clipping occurs. TI recommends setting the level by measuring the THD+N at the frequency most boosted by the overall system since this frequency is clipped the most.

9.4 Output Volume

The output volume controls the digital level of both Channel A and Channel B independently from –103 dB to 24 dB by setting the slider. When loaded onto the stereo device, where Channel A and B are identical, most applications require that the volume level be adjusted together to avoid mismatch on stereo speakers. When HF5 is loaded onto the mono device, only Channel C volume is displayed.

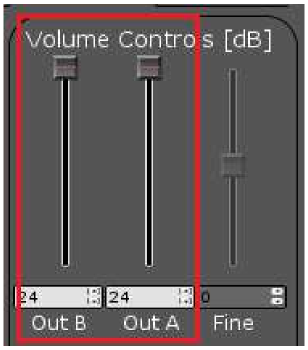


Figure 68. Output Volume Control

9.5 Sub. Gen.

Sub Gen is a digital output-only audio processing block designed for subwoofer output generation. It is a mono audio processor where a copy of the input serial audio port data gets mixed by (A+B)/2. Therefore only one processing block is shown in the HybridFlow diagram.

NOTE: For HF5, Sub Gen is only available when loaded onto the stereo device.

Sub Gen is designed so that the TAS5754/6M devices provide the central processing for both the full-range channels, A and B, as well as a mono subwoofer channel which simplifies the end design. After the Sub Gen tuning has been set, the digital data can be selected through SDOUT and sent to a separate subwoofer amplifier. When using Sub Gen in this manner, the separate subwoofer amplifier requires no audio processing.

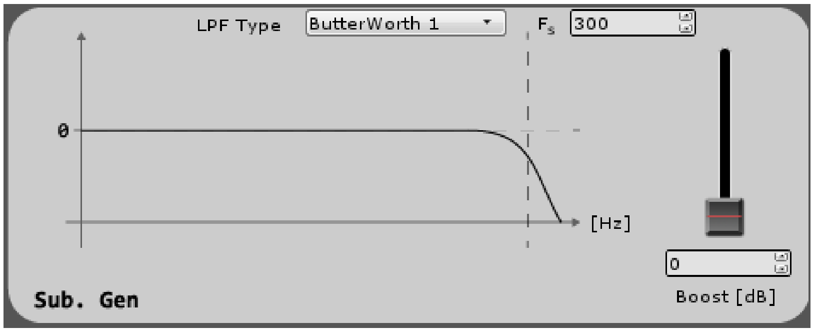


Figure 69. HF2 Sub Gen Tuning Window

Set a low-pass filter at the desired cutoff frequency in the Sub Gen tuning window and select the filter type from the drop-down menu. Add a boost to match the efficiency of a subwoofer speaker channel to the efficiency of the full-range channels. The digital audio can then be passed to SDOUT.

The digital format for Sub Gen is synchronized to the clocking on the TAS5754/6M. For a single cycle of LRCLK, the output word is available twice since it represents a mono signal. There are still separate Left and Right words for each cycle of LRCLK; however, the words are identical. Therefore, it is up to the designer to choose when to read the word on the device monitoring SDOUT.

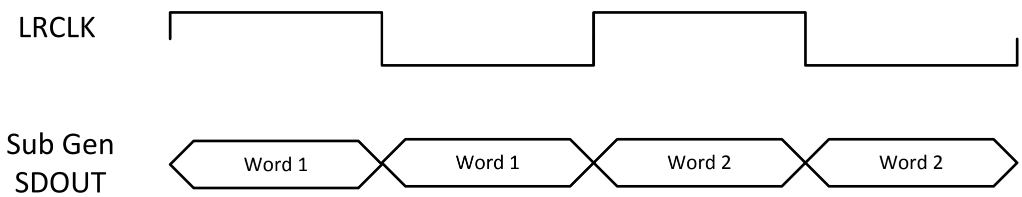


Figure 70. Sub Gen Data Format

9.6 SDOUT Serial Audio Data

In HybridFlow 5 there are 3 choices for digital output sources available on GPIO2 (pin 21 on the TAS5754/6M devices). The first option is Pre-DSP which passes the incoming digital data to SDOUT. There is no processing of the data. The second option is Sub Gen which passes the digital data processed by the Sub Gen block. The third option is Post-DSP which allows for the fully processed data to be available for use. In post-DSP, Smooth Clip and the digital output volume control have no effect. SDOUT is useful when trying to connect with other digital input devices and amplifiers.

NOTE: SDOUT acts like a MUX; however, it is a hard mixer meaning it mixes only one of the inputs at a time with a 100% mix ratio. This method creates a more efficient DSP MUX device rather than a traditional switching MUX.

10 HybridFlow 6 (HF6)

Figure 71 depicts the signal path for HybridFlow 6. The shaded tabs correspond to the functions found in the PurePath Console GUI.

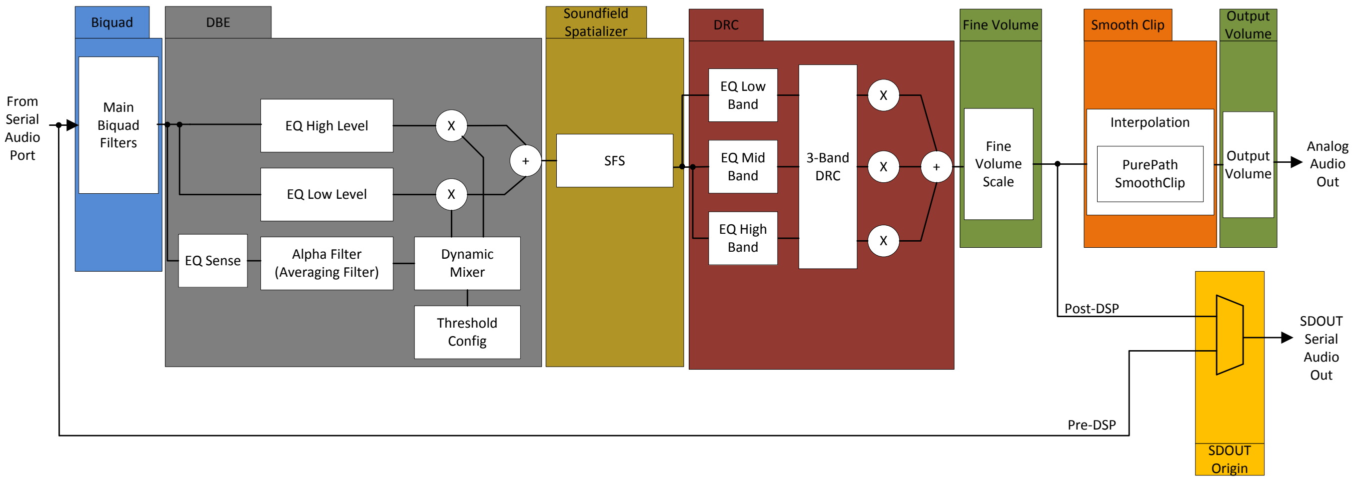


Figure 71. HybridFlow 6 Block Diagram

HybridFlow 6 target applications include stereo audio devices such as docking stations, all-in-one PCs, and general audio applications requiring up to a 48-kHz sample rate. Both Channel A and B are identical and follow the HF6 block diagram. That is, changing the coefficients in any of the processing blocks in Figure 71 automatically applies the change to Channel A and B. The only channel-independent control is the output volume set in the PurePath Console GUI.

10.1 Biquad

The Biquad filter block contains 10 independent filters designed for tuning the frequency response of the overall system. This is where the bulk of the frequency compensation occurs; select the filter type, subtype, and other parameters for each of the 10 Biquads. Complex tuning shapes can be made to compensate for deficiencies in speaker response with a goal of a flat response over the frequency band of interest.

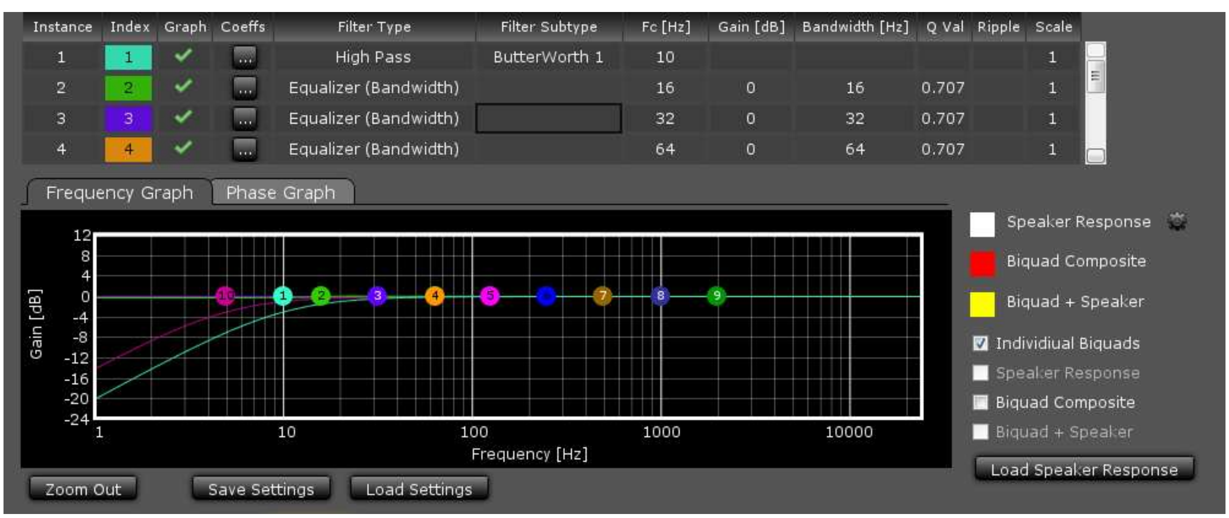


Figure 72. HF6 Biquad Tuning Window

Figure 72 shows the Biquad audio processing window in the PurePath Console GUI. Drag and drop each Biquad into position using the mouse on the plotting window. Other parameters must be typed into the chart above the plotting space. After which, pressing *Enter* on the keyboard causes the change to take effect.

Filter Type and *Filter Subtype* are drop-down menus activated when this space is clicked for the desired Biquad. Remove or add Biquads by clicking on the “graph” space next to the desired Biquad.

The *Phase Graph* shows the phase response for each of the individual Biquads.

The *Biquad Composite* check box shows the overall response based on the position of the individual Biquad filters. This composite view is the frequency response alteration applied to the incoming digital audio data. View this independently by deselecting *Individual Biquads*.

10.1.1 Load Speaker Response

If desired, load a measured speaker response into the Biquad window by clicking *Load Speaker Response*. This aids in tuning the frequency response. With a speaker response loaded, view the overall audio system response by clicking *Biquad + Speaker*. This takes into account the added Biquads as well as the natural response of the speaker.

The file format for the speaker response is .txt. Each line in the file represents a data point containing frequency (in Hz), SPL (in dB), and Phase (in degrees), separated by a space. Any line that does not begin with a number is ignored.

NOTE: A loaded speaker response is not loaded into the HybridFlow. It is only displayed in the PurePath Console GUI as a tuning tool.

10.2 DBE

Dynamic Bass Enhancement (DBE) is a processing block that allows for optimizing the bass response of the system. Two signal paths (low level and high level) 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 DBE channels is dynamic in nature and depends on the incoming audio. Figure 73 shows the tuning window in the PurePath Console GUI.

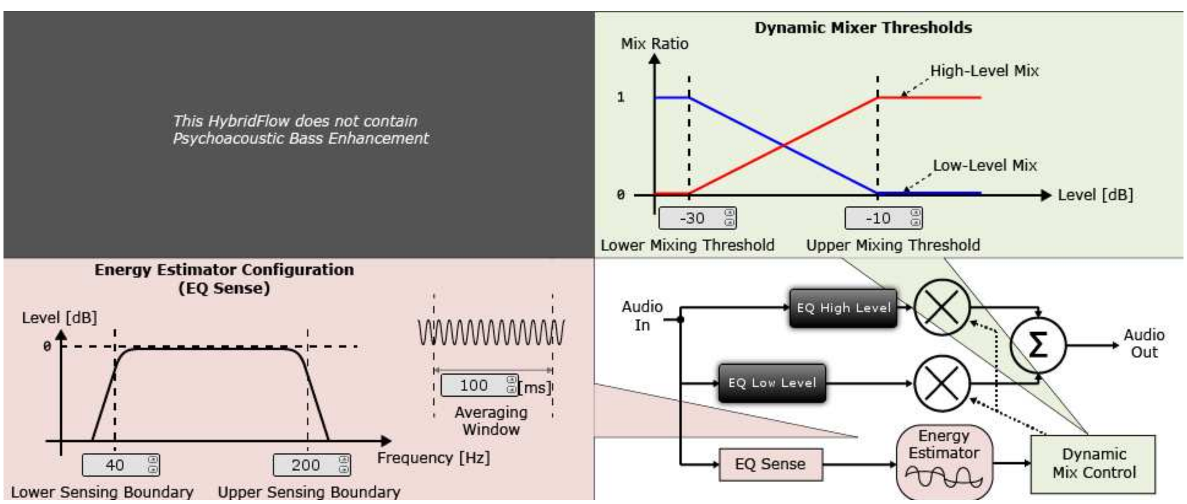


Figure 73. HF6 DBE Tuning Window

10.2.1 Dynamic Mixer Thresholds

The mixing of the two paths (low level and high level) is controlled by setting the *Upper Mixing Threshold* and *Lower Mixing Threshold*. When the averaged signal (as set by the *Averaging Window*) is below the lower mixing threshold, the *Dynamic Mixer* sends all of the audio through the low-level path. When the signal is above the upper mixing threshold, it is sent through the upper-level path. When the signal is between the two, it is mixed together by the *Dynamic Mixer*.

10.2.2 Energy Estimator Configuration (EQ Sense)

Another key configuration for the dynamic mixer block is the EQ sense and alpha filter. In HF6, the EQ sense is a single bandpass filter. The bandwidth of the filter is set by entering in the lower and upper sensing boundary in the GUI. This tells the dynamic mixer which frequency range of the incoming signal to average in order to determine how the signal compares to the mixing thresholds.

The *Averaging Window* or alpha filter works similarly to that of a DRC. It 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.

10.2.3 EQ Low Level

The low-level path contains 5 configurable Biquads to establish the EQ curve which the audio is sent through when the time average signal is at a low-level. These fully-functional Biquads can be assigned to several filter types or sub-types. This determines frequency response when low-level is active based on the Energy Estimator Configuration and the mixing threshold.

Click the *EQ LOW LEVEL* button to display the tuning window (Figure 74). The tuning shown for the EQ Low Level is chosen as a bass boost to reduce early onset bass roll off from the speaker. For more details visit Section 10.1, Biquad.

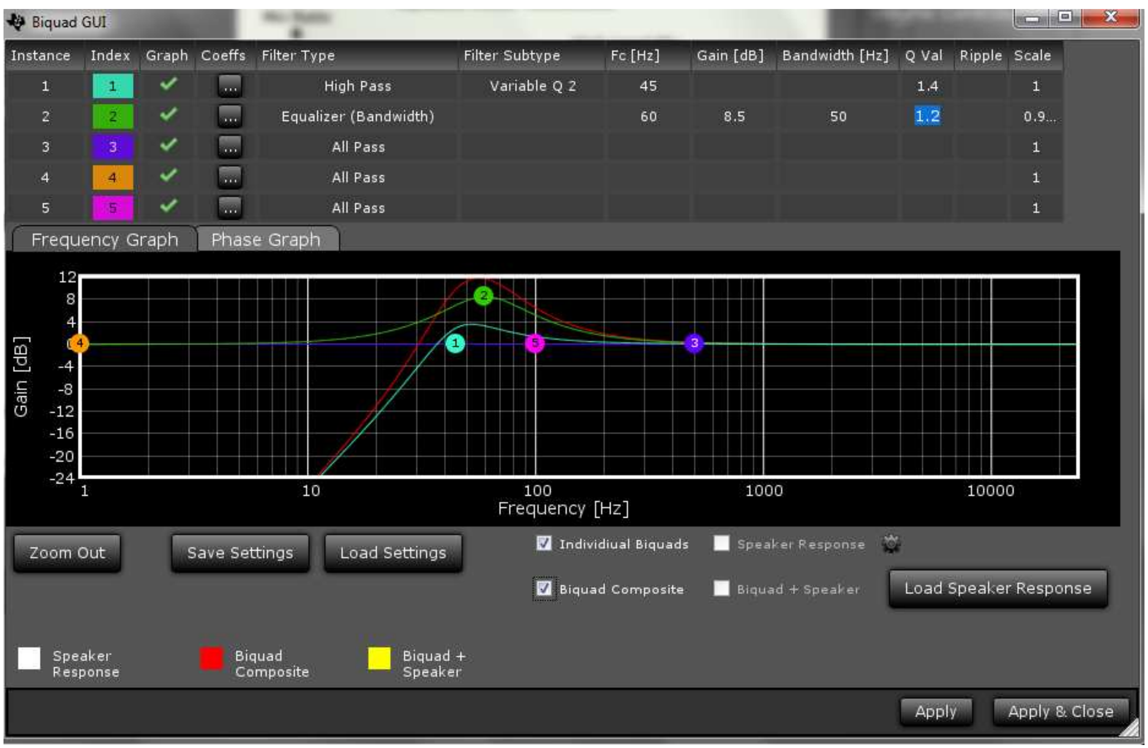


Figure 74. HF6 DBE EQ Low-Level Tuning Window

10.2.4 EQ High Level

The high-level path, similar to the low-level path, has 3 Biquads which can set the EQ curve used when the time averaged input signal is above the upper mixing threshold.

Click the *EQ High LEVEL* button to display the tuning window (Figure 75). The tuning shown for the EQ High Level is a simple high-pass filter with fast roll off. For more details visit Section 10.1, Biquad.

Figure 75. HF6 DBE EQ High-Level Tuning Window

10.3 *Soundfield Spatializer*

Soundfield 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.

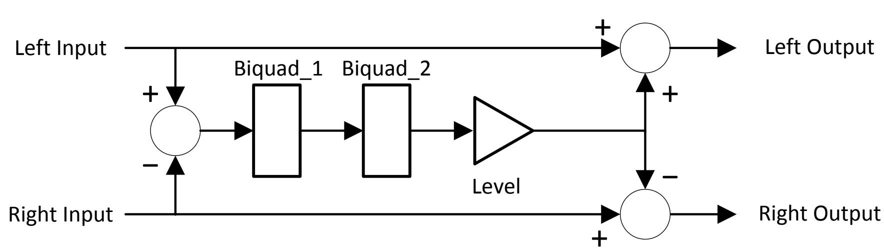


Figure 76. Soundfield Spatializer Block Diagram

It is generally not recommend 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.

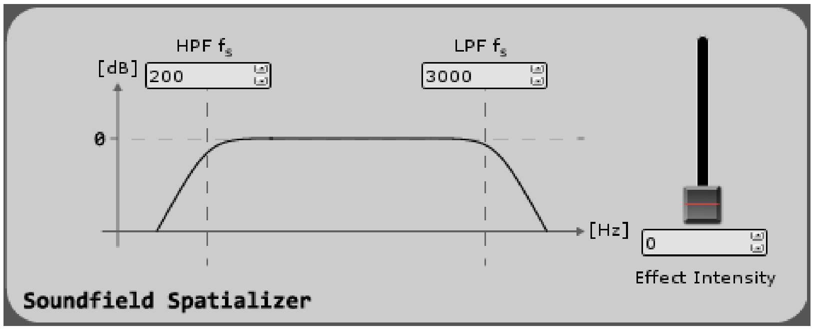


Figure 77. Soundfield Spatializer Tuning Window

In the tuning window, the pass band can be set as well as the *Effect Intensity* which controls the level of the effect. For a given piece of end equipment, it may be helpful to create three "presets" that 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.

10.4 DRC Standard 3-Band Dynamic Compander

HybridFlow 6 features a 3-band DRC compander (compression and expansion). The DRC is 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.

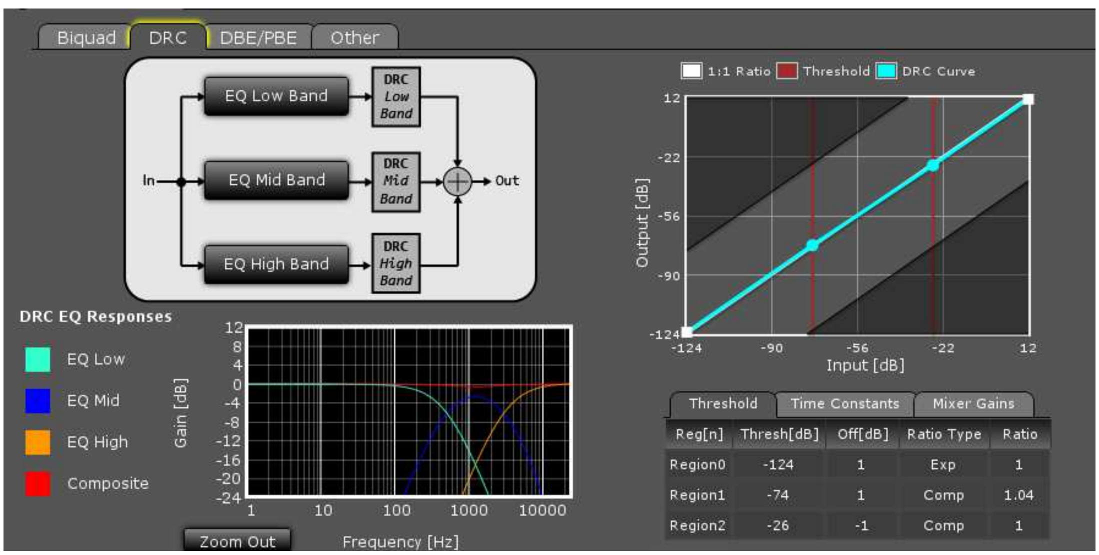
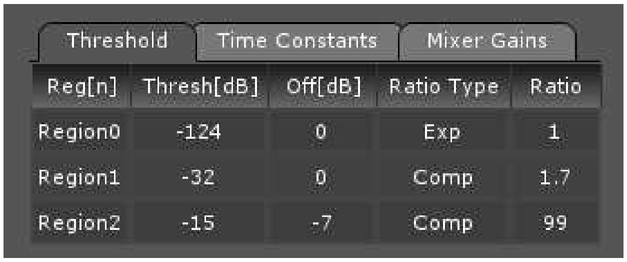


Figure 78. HF6 3-Band DRC Compander Tuning Window

On the right side of the window in Figure 78 is the DRC curve which offers 3 regions of compression. The points on the DRC curve can be dragged and dropped. At the top center of the window is a block diagram of the DRC as well as 3 configurable frequency bands (Low, Mid, and High).

10.4.1 DRC Threshold Tab

Below the DRC window, parameters such as threshold, offset, expansion or compression, and the ratio value can be manually entered for each of the 3 regions under the *Threshold* tab. By typing a value and pressing *Enter* on the keyboard, the DRC curve automatically adjusts to the entered parameter.

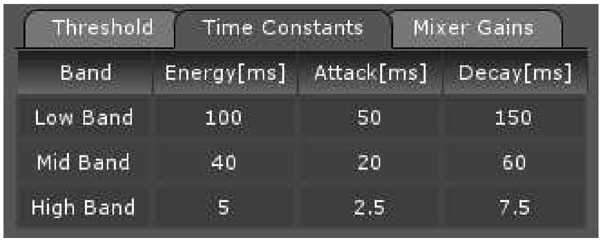


Reg[n]	Thresh[dB]	Off[dB]	Ratio Type	Ratio
Region0	-124	0	Exp	1
Region1	-32	0	Comp	1.7
Region2	-15	-7	Comp	99

Figure 79. HF6 DRC Threshold Control Tab for 3-Band Compander

10.4.2 DRC Time Constants Tab

The standard 3-band compander offers splitting of the incoming audio into 3 frequency bands determined by the user. Although the same DRC curve is applied to all 3 frequency bands, different attack time constants can be associated with each band to optimize audio quality and speaker protection. Change time constants by clicking on the *Time Constants* tab (Figure 80) and enter new values for each band.



Band	Energy[ms]	Attack[ms]	Decay[ms]
Low Band	100	50	150
Mid Band	40	20	60
High Band	5	2.5	7.5

Figure 80. HF6 DRC Time Constants Tab for 3-Band Compander

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. *Attack[ms]* determines the attack time of the DRC and *Decay[ms]* determines the release time once the windowed energy band passes.

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.

For example, a very fast time constant on a low-frequency signal may cause the DRC to attack and release before a full cycle of the incoming signal has passed. Then when the next peak of the wave passes through the DRC, it again attacks and then releases as the peak passes. The DRC continuously attacks and releases rather than enveloping the signal causing audible distortion.

With separate time constants, the standard 3-band compander can still have a very fast time constant at high frequencies and a slower time constant at low frequencies enveloping and compressing the entire audible range quickly and effectively.

10.4.3 Mixer Gains Tab

The mixer gain controls the relative gain of each of the 3 frequency bands when they are mixed together. Use this 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.

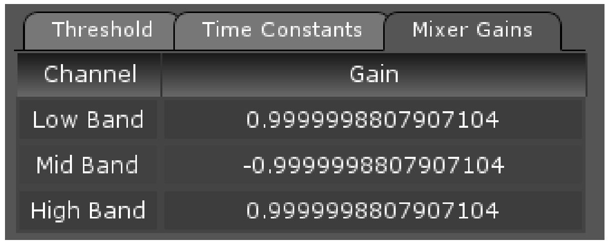


Figure 81. HF6 Mixer Gains Tab for 3-Band Compander

10.4.4 Band Splitting

Configure the frequency range associated with each of the 3 bands used by the *Time Constants* tab by clicking on the *EQ Low Band*, *EQ Mid Band*, and *EQ High Band* buttons. Here a Biquad window appears where the tuning can take place. After tuning, the response is automatically displayed in the 3-Band DRC Compander tuning window on the bottom left.

For more details visit Section 10.1, Biquad.

10.5 Fine Volume

The fine volume control is a digital volume control that allows adjustment between -0.25 dB and 0.25 dB by setting the slider. Both Channel A and B are set simultaneously.

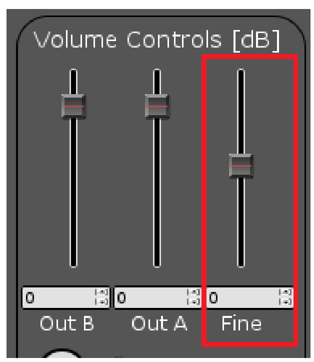


Figure 82. Fine Volume Control

10.6 PurePath SmoothClip

SmoothClip works as a comparator in the digital domain on a sample-by-sample basis. If the incoming audio data word is larger than the set comparator coefficient, the set coefficient is passed until the incoming audio data word is below the set coefficient. This effectively clips the signal. Unlike typical digital clipping which occurs at the sample rate (Fs), SmoothClip operates at very high speeds, minimizing the unwanted distortions associated with digital clipping.

This is often used in conjunction with slower DRC time constants. With a more gradual time constant and compression ratio, the potential for DRC beating or "pumping" is reduced and sound quality and dynamics are improved. However, due to the slow DRC response, a few cycles of incoming audio data that are greater than the set DRC thresholds can pass through. With SmoothClip following a DRC, these cycles can be clipped in a well-controlled fashion to prevent speaker damage until the DRC has attacked the signal.

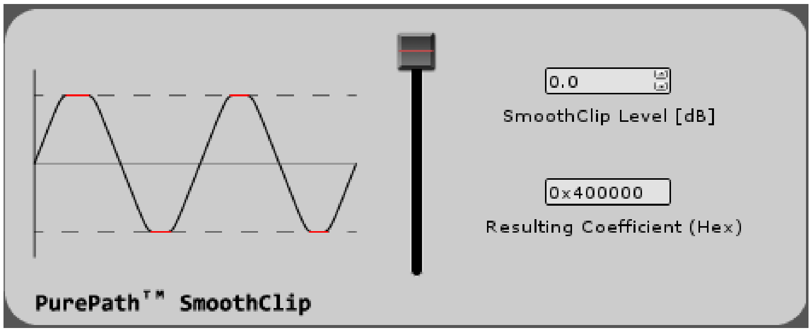


Figure 83. SmoothClip Tuning Window

SmoothClip only has one tuning parameter; the level at which the clipping occurs. TI recommends setting the level by measuring the THD+N at the frequency most boosted by the overall system since this frequency is clipped the most.

10.7 Output Volume

The output volume controls the digital level of both Channel A and Channel B independently from –103 dB to 24 dB by setting the slider. In HybridFlow 6, where Channel A and B are identical, most applications require channels A and B are adjusted together to avoid mismatch on stereo speakers.

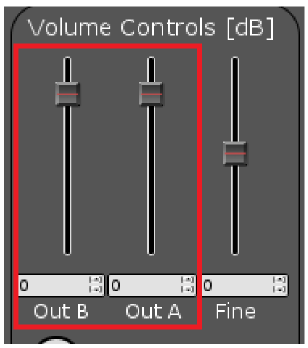


Figure 84. Output Volume Control

10.8 SDOUT Serial Audio Data

In HybridFlow 6 there are 2 choices of digital output sources available on GPIO2 (pin 21 on the TAS5754/6M devices). The first option is Pre-DSP which passes the incoming digital data to SDOUT. There is no processing of the data. The second option is Post-DSP which allows for the fully processed data to be available for use. In post-DSP, Smooth Clip and the analog output volume control have no effect. SDOUT is useful when trying to connect with other digital input devices and amplifiers.

NOTE: SDOUT acts like a MUX; however, it is a hard mixer meaning it mixes only one of the inputs at a time with a 100% mix ratio. This method creates a more efficient DSP MUX device rather than a traditional switching MUX.

11 HybridFlow 7 (HF7)

Figure 85 depicts the signal path for HybridFlow 7. The shaded tabs correspond to the functions found in the PurePath Console GUI.

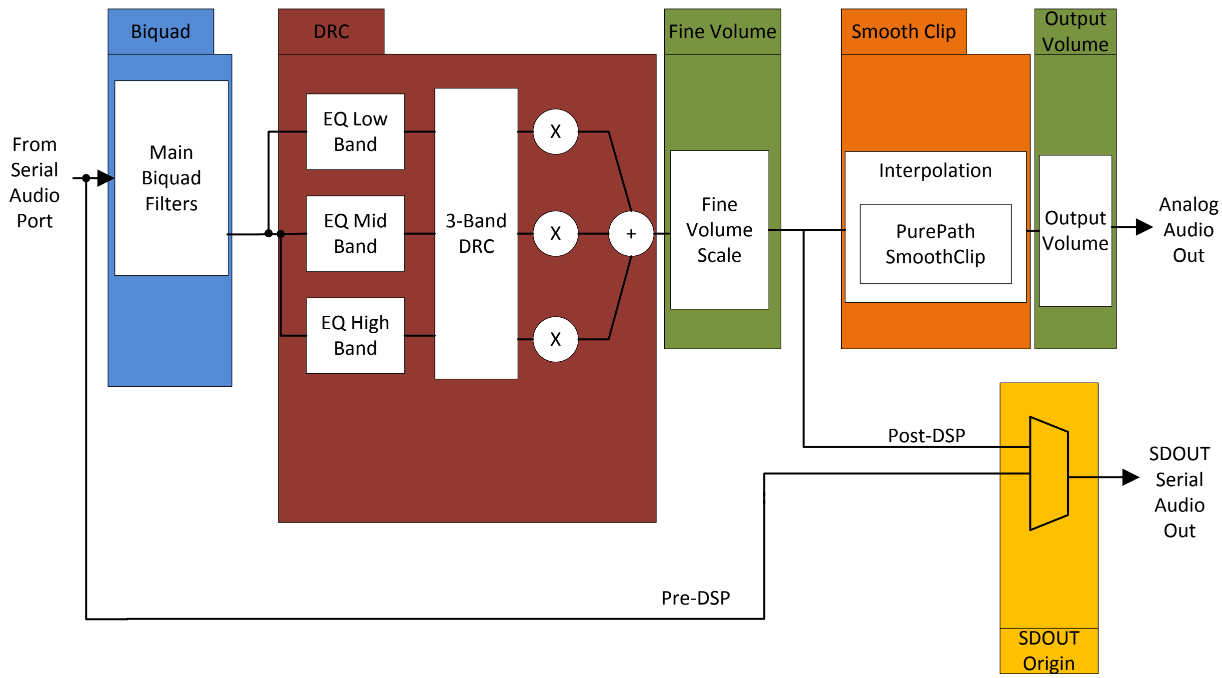


Figure 85. HybridFlow 7 Block Diagram

HybridFlow 7 target applications include stereo audio devices such as soundbars, DTVs and general audio applications requiring up to a 96-kHz sample rate. Simplified audio processing allows for faster sample rates required by high-end audio applications.

HF7 is available on the stereo device and the mono device. In stereo operation, Channel A and B are identical and follow the HF5 block diagram. That is, changing the coefficients in any of the processing blocks in Figure 85 automatically applies the change to Channel A and B. The only channel independent control is the output volume set in the PurePath Console GUI. In mono operation, only one of the incoming digital channels gets passed through the HybridFlow, labeled Channel C. Therefore, to use HF7 on the mono device without losing an audio channel, the digital audio must be premixed into a mono channel before entering HF7.

11.1 Biquad

In HF7 there are 5 Biquads. This is where the bulk of the frequency compensation occurs; select the filter type, subtype, and other parameters for each of the 5 Biquads. Complex tuning shapes can be made to compensate for deficiencies in speaker response with a goal of a flat response over the frequency band of interest.

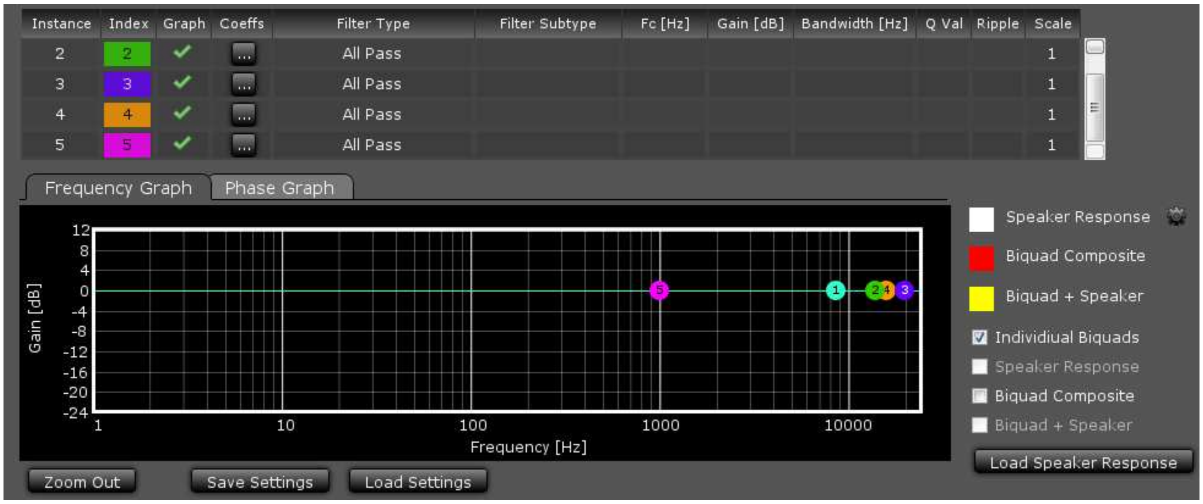


Figure 86. HF7 Biquad Tuning Window

Figure 86 shows the Biquad audio processing window in the PurePath Console GUI. Drag and drop each Biquad into position using the mouse on the plotting window. Other parameters must be typed into the chart above the plotting space. After which, pressing *Enter* on the keyboard causes the change to take effect.

Filter Type and *Filter Subtype* are drop-down menus activated when this space is clicked for the desired Biquad. Remove or add Biquads by clicking on the “graph” space next to the desired Biquad.

The *Phase Graph* shows the phase response for each of the individual Biquads.

The *Biquad Composite* check box shows the overall response based on the position of the individual Biquad filters. This composite view is the frequency response alteration applied to the incoming digital audio data. View this independently by deselecting *Individual Biquads*.

11.1.1 Load Speaker Response

If desired, load a measured speaker response into the Biquad window by clicking *Load Speaker Response*. This aids in tuning the frequency response. With a speaker response loaded, view the overall audio system response by clicking *Biquad + Speaker*. This takes into account the added Biquads as well as the natural response of the speaker.

The file format for the speaker response is .txt. Each line in the file represents a data point containing frequency (in Hz), SPL (in dB), and Phase (in degrees), separated by a space. Any line that does not begin with a number is ignored.

NOTE: A loaded speaker response is not loaded into the HybridFlow. It is only displayed in the PurePath Console GUI as a tuning tool.

11.2 DRC Standard 3-Band Dynamic Compander

HybridFlow 7 features a 3-band DRC compander (compression and expansion). The DRC is 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.

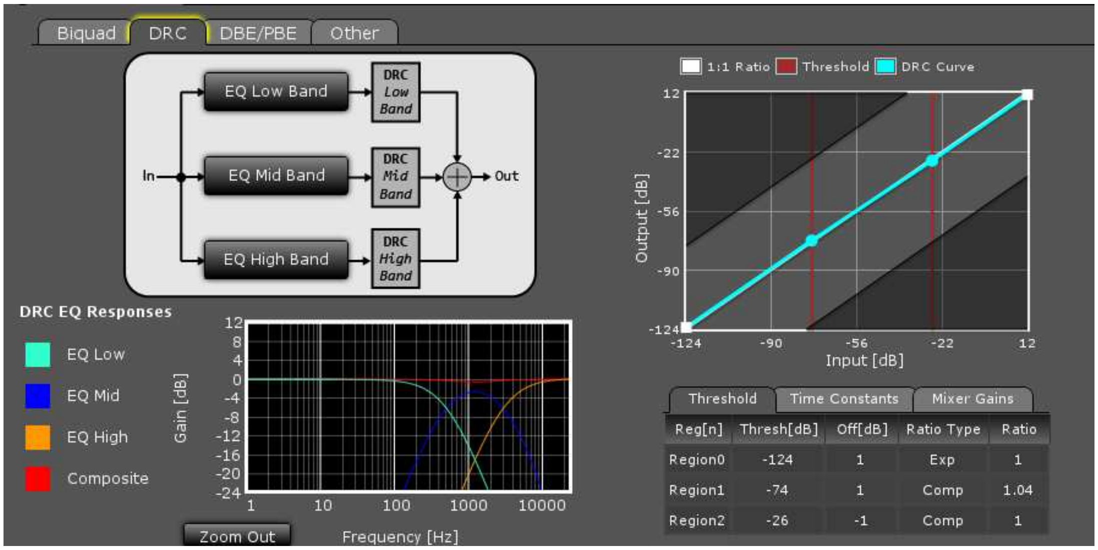


Figure 87. HF7 3-Band DRC Compander Tuning Window

On the right side of the window in Figure 87 is the DRC curve which offers 3 regions of compression. The points on the DRC curve can be dragged and dropped. At the top center of the window is a block diagram of the DRC as well as 3 configurable frequency bands (Low, Mid, and High).

11.2.1 DRC Threshold Tab

Below the DRC window, parameters such as threshold, offset, expansion or compression, and the ratio value can be manually entered for each of the 3 regions under the *Threshold* tab. By typing a value and pressing *Enter* on the keyboard, the DRC curve automatically adjusts to the entered parameter.

Threshold | Time Constants | Mixer Gains

Reg[n]	Thresh[dB]	Off[dB]	Ratio Type	Ratio
Region0	-124	0	Exp	1
Region1	-32	0	Comp	1.7
Region2	-15	-7	Comp	99

Figure 88. HF7 DRC Threshold Control Tab for 3-Band Compander

11.2.2 DRC Time Constants Tab

The standard 3-band compander offers splitting of the incoming audio into 3 frequency bands determined by the user. Although the same DRC curve gets applied to all 3 frequency bands, different attack time constants can be associated with each band to optimize audio quality and speaker protection. Change time constants by clicking on the *Time Constants* tab (Figure 89) and enter new values for each band.

Threshold | Time Constants | Mixer Gains

Band	Energy[ms]	Attack[ms]	Decay[ms]
Low Band	100	50	150
Mid Band	40	20	60
High Band	5	2.5	7.5

Figure 89. HF7 DRC Time Constants Tab for 3-Band Compander

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. *Attack[ms]* determines the attack time of the DRC and *Decay[ms]* determines the release time once the windowed energy band passes.

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.

For example, a very fast time constant on a low-frequency signal may cause the DRC to attack and release before a full cycle of the incoming signal has passed. Then when the next peak of the wave passes through the DRC, it again attacks and then releases as the peak passes. The DRC continuously attacks and releases rather than enveloping the signal causing audible distortion.

With separate time constants, the standard 3-band compander can still have a very fast time constant at high frequencies and a slower time constant at low frequencies enveloping and compressing the entire audible range quickly and effectively.

11.2.3 Mixer Gains Tab

The mixer gain controls the relative gain of each of the 3 frequency bands when they are mixed together. Use this 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.

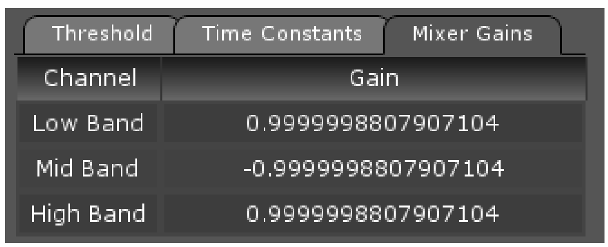


Figure 90. HF7 Mixer Gains Tab for 3-Band Compander

11.2.4 Band Splitting

Configure the frequency range associated with each of the 3 bands used by the *Time Constants* tab by clicking on the *EQ Low Band*, *EQ Mid Band*, and *EQ High Band* buttons. Here a Biquad window appears where the tuning can take place. After tuning, the response is automatically displayed in the 3-Band DRC Compander tuning window on the bottom left.

For more details visit Section 11.1, Biquad.

11.3 Fine Volume

The fine volume control is a digital volume control that allows adjustment between –0.25 dB and 0.25 dB by setting the slider. Both Channel A and B are set simultaneously.

NOTE: When HF7 is loaded into the mono device, Channel C has no fine volume control.

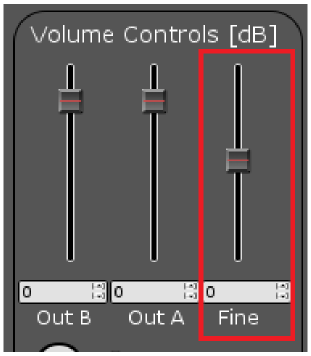


Figure 91. Fine Volume Control

11.4 PurePath SmoothClip

SmoothClip works as a comparator in the digital domain on a sample-by-sample basis. If the incoming audio data word is larger than the set comparator coefficient, the set coefficient is passed until the incoming audio data word is below the set coefficient. This effectively clips the signal. Unlike typical digital clipping which occurs at the sample rate (Fs), SmoothClip operates at very high speeds, minimizing the unwanted distortions associated with digital clipping.

This is often used in conjunction with slower DRC time constants. With a more gradual time constant and compression ratio, the potential for DRC beating or “pumping” is reduced and sound quality and dynamics are improved. However, due to the slow DRC response, a few cycles of incoming audio data that are greater than the set DRC thresholds can pass through. With SmoothClip following a DRC, these cycles can be clipped in a well-controlled fashion to prevent speaker damage until the DRC has attacked the signal.

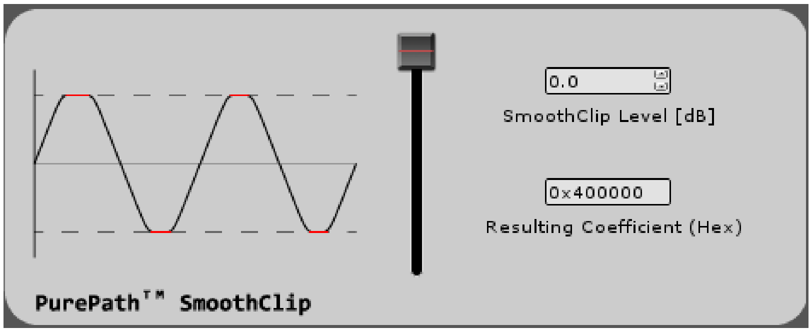


Figure 92. SmoothClip Tuning Window

SmoothClip only has one tuning parameter; the level that the clipping occurs. TI recommends setting the level by measuring the THD+N at the frequency most boosted by the overall system since this frequency is clipped the most.

11.5 Output Volume

The output volume controls the digital level of both Channel A and Channel B independently from –103 dB to 24 dB by setting the slider. When loaded onto the stereo device, where Channel A and B are identical, most applications require channels A and B are adjusted together to avoid mismatch on stereo speakers. When HF7 is loaded onto the mono device, only Channel C volume is displayed.

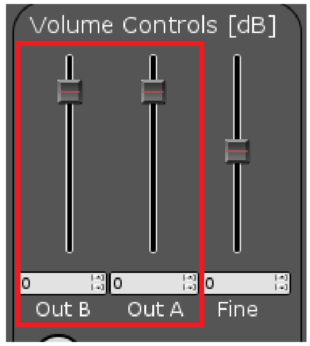


Figure 93. Output Volume Control

11.6 SDOUT Serial Audio Data

In HybridFlow 7 there are 2 choices of digital output sources available on GPIO2 (pin 21 on the TAS5754/6M devices). The first option is Pre-DSP which passes the incoming digital data to SDOUT. There is no processing of the data. The second option is Post-DSP which allows for the fully processed data to be available for use. In post-DSP, Smooth Clip and the digital output volume control have no effect. SDOUT is useful when trying to connect with other digital input devices and amplifiers.

NOTE: SDOUT acts like a MUX; however, it is a hard mixer meaning it mixes only one of the inputs at a time with a 100% mix ratio. This method creates a more efficient DSP MUX device rather than a traditional switching MUX.

---

12 Tips for Initial HybridFlow Tuning

This section discusses some common tips and tricks when beginning to work with any of the HybridFlows previously discussed in this document.

12.1 Recommended Order of Operations for HybridFlow Tuning

1. Use the *Configure* tab in the PurePath Console GUI to setup the desired use case
2. Set the maximum digital gain using the volume controls
3. Use Biquad filters to set the baseline system frequency response
4. Set the DRC and PurePath™ SmoothClip for power limit and clipping characteristics
5. Set low-level DBE/DDE frequency response
6. Set high-level DBE/DDE frequency response
7. Configure Soundfield Spatializer (if applicable)

12.2 Speaker Measurements

Take speaker measurements using a high-quality measurement microphone and computer software.

USB microphones from Dayton Audio are an affordable option, but many others are available. One option is found here: http://www.parts-express.com/dayton-audio-umm-6-usb-measurement-microphone--390-808

There are many speaker measurement software packages available. Room EQ Wizard (REW) software is a good, free program (http://www.roomeqwizard.com/).

In the range from 500 Hz to 20 kHz, measurements are best made in the "near field" (5–10 centimeters) from the system unless measuring in the middle of a large, quite, open room is available. Measurements below 500 Hz usually are quite difficult outside of an anechoic chamber or through half-plane measurements in a large, open, outside area. Data collected below this frequency should be considered untrustworthy outside of an appropriate measurement environment. After the measurements are made, load the results into PurePath Console GUI in the Biquad tuning window until the *Biquad + Speaker* response is relatively flat. Then measure the system again and repeat if necessary.

As mentioned, in the range from 500 Hz and lower, the effects of the room can dominate the frequency response. Measurements in an anechoic chamber produce the best possible results. Large open outdoor areas can also work quite well. Remember that you can also tune to preference. The goal is extended frequency response without emphasis of a particular frequency when the entire system is in use.

12.3 Flattening Speaker Response Using Biquads

One goal of tuning with Biquad filters in any of the HybridFlows should be to help flatten the frequency response of the end system. This could be to remove peaks and dips or extend the frequency response. Since the goal is to flatten the response of the system, measurements should be made of the full working system. It should be noted that the final tuning is usually not "flat" but instead tuned to the target audience and their listening preferences. However, the first step is to achieve a baseline from which further tuning is made. Once a somewhat flat response is obtained, adding more or less of a given frequency range is quite easy.

An end response with variations of ±3–5 dB is considered respectable. Large high Q nulls that don't change with added boost from high Q Biquad EQs are usually the result of a room reflection. If this is the case, change the measurement setup to get an accurate view of that frequency region. These nulls can often be removed by changing the configuration of the test set-up within the room. If you cannot change the measurement setup, leave the response in that region as it is and move on.

Biquads used to attenuate the signal should be placed first. For example, use Biquads 1–5 first for attenuation and 6–10 for boosting the signal. This reduces the risk of clipping due to additive gains of Biquads used for boosting. For example, if a large boost is added using Biquad 1 and the signal is clipped, Biquads 2–10 pass a copy of the clipped signal, even if their gain is reduced.

The data path has +12 dB of headroom. Therefore, try to limit peaking of Biquads to 9–10 dB. Be careful with adjacent peaking Biquads for this reason, since their response is additive if their frequency range is close to one another.

Try to use the fewest number of Biquads to achieve relatively flat results.

12.4 Loading Speaker Response

Load speaker responses into any number of Biquad filter windows available throughout the HybridFlows. The file format for the speaker response is .txt. Each line in the file represents a data point containing frequency (in Hz), SPL (in dB), and Phase (in degrees), separated by a space. Any line that does not begin with a number is ignored.

12.4.1 Example Speaker Response .txt

The following example was exported as a .txt using REW Software and fits the desired format.

```
* Measurement data saved by REW V5.01
* Source: Trace Arithmetic result A + B
* Format: Trace Arithmetic result A + B
* Dated: Dec 11, 2014 9:11:27 AM
* REW Settings:
*  C-weighting compensation: Off
*  Target level: 75.0 dB
* Measurement: A plus B
* Frequency Step: 0.3364563 Hz
* Start Frequency: 20.187378 Hz
*
* Freq(Hz) SPL(dB) Phase(degrees)
20.187 56.148 -21.917
20.524 56.599 -25.916
20.860 57.032 -29.484
21.197 57.422 -32.499
21.533 57.783 -35.027
21.870 58.220 -37.307
22.206 58.728 -39.629
22.543 59.215 -42.191
22.879 59.674 -45.020
```

12.5 Tuning the DRC Standard 3-Band Dynamic Compander

12.5.1 Power Limiting

Power liming is used to limit loudspeaker power to avoid damage at high signal levels.

Hard Power Limiting: Uses compression/expansion ratio of 1:1 until the rated output power of the speaker is reached. After which, a single transition point is used for very hard compression at a level of 25:1 or greater. This should be used in conjunction with a fast attack time and energy estimator to avoid potential over power conditions since there is little to no margin of error.

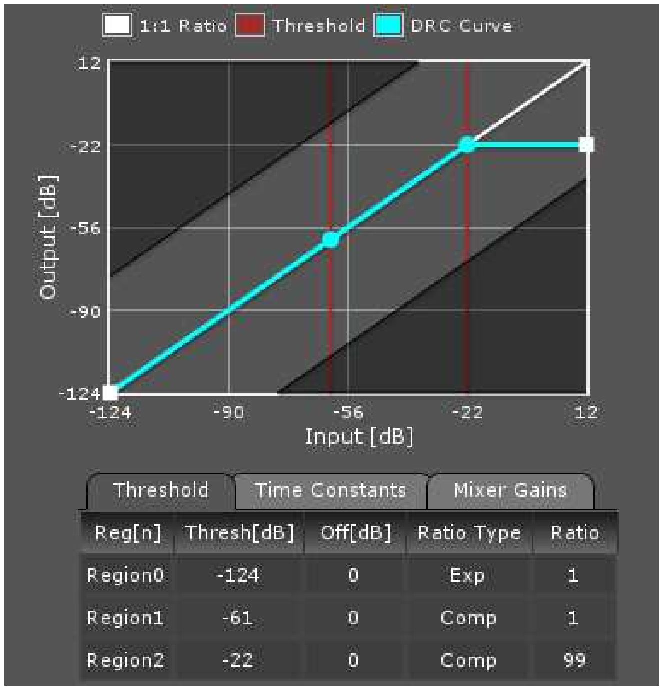


Figure 94. DRC Curve for Hard Power Limit

NOTE: Hard Power Limiting increases chances of DRC “pumping”, where the audio sounds like it is surging as the compressor attacks and releases. Be sure to check the final tuning at high volumes with bass-heavy songs to prevent this occurrence.

Soft Power Limiting: Uses compression/expansion ratio of 1:1 until 0.85 to 0.9 times the rated output power of the speaker is reached. Then a mild compression ratio is used up to the rated loudspeaker power where a second compression ratio of 25:1 or greater is added.

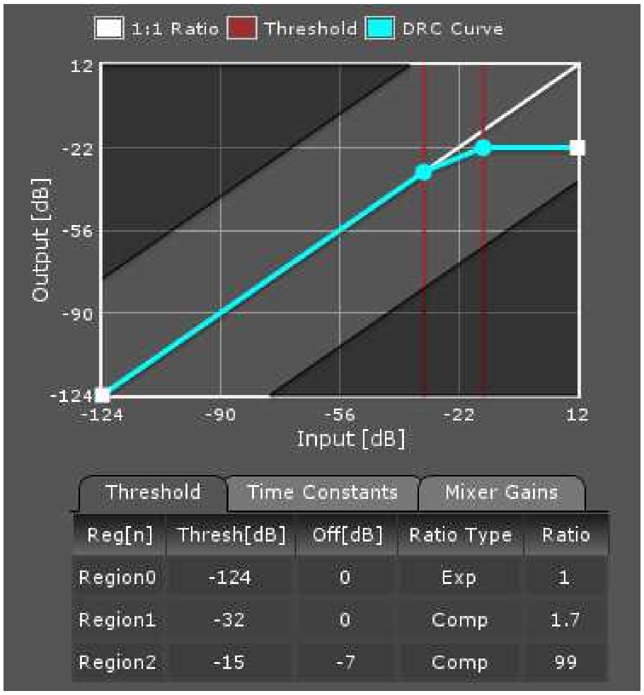


Figure 95. DRC Curve for Soft Power Limit

12.5.2 Recommended DRC Time Constants

DRC time constants are often characterized as fast or slow depending on the design goals. Faster time constants are characterized with slightly higher distortion, while slow time constants have improved distortion but slower response allowing some energy above the set threshold to pass. In the case where the DRC is slow, this can be compensated for by using Smooth Clip to reduce the amplitude of any energy that gets past the DRC before it has attacked the signal.

A good starting point is found by referring to Table 3:

Table 3. Recommended DRC Time Constants

	Fast	Slow
Energy Threshold	5x(1/fmin)	15x(1/fmin)
Attack	2x(1/fmin)	5x(1/fmin)
Decay	10x(1/fmin)	20x(1/fmin)

12.5.3 Recommended Band Splitting

The default band splitting on the 3-Band Dynamic Compander is configured by default as a Linkwitz Rielly 2 filter which simplifies unity summation. The Fc of the filter is set the same for the 2 bands for unity summation. Evaluate the effectiveness of a frequency band by muting the other channels using the Mixer Coefficients. TI recommends testing this for each band.

The recommended starting ranges for band splitting the 3-Band dynamic compander are as follows:

Low Band: 300 Hz and below
Mid Band: 300 Hz to 5 kHz
High Band: 5 kHz to 20 kHz

12.6 Configuring PurePath SmoothClip

When testing the functionality of SmoothClip, a resistive load equal to the target speaker impedance should be used. A sine wave should be the input source set to the maximum expected input level.

The THD + N measured should be less than or equal to the target value when measuring with an input frequency equal to the most boosted frequency. Make sure to look at all of the set Biquads to determine where the most boost is added, then recheck over the entire spectrum.

The use of PurePath SmoothClip triggers Audio Data Status to show “Overflow” since the clip event is recognized by the device as a signal path overflow

12.7 Configuring Psychoacoustic Bass Enhancement (PBE)

Psychoacoustic Bass Enhancement is only effective at high-output levels where traditional bass boosting cannot be applied. Therefore, final tuning should take place after the DRC has been configured using very heavy bass audio content. Too much PBE adds significant distortion since it is adding harmonics. The effect should be noticeable at high output levels but not overwhelming.

It should be noted that the high-pass filters in DBE can filter out the harmonic generation of PBE. It is recommended that the high-level high-pass filter set in DBE is not set to more than 2x cutoff frequency used in the high-pass filter of PBE.

Also, since the PBE block emulates a harmonic generation block, the overflow detection circuit inside the device sometimes triggers when the PBE is active. Generally speaking, some overflow indications are acceptable when playing audio back at a high output level. However, constant indication of overflow or overflow at relatively low output levels indicate that the signal is over-driving the data path and should be attenuated.

12.8 Configuring Dynamic Bass Enhancement (DBE)

12.8.1 Setting High-Level Biquads

It is best to start by setting the high-level path with fast roll-off set at 1.5 × loudspeaker resonance. The roll-off rate is increased by adding a second Biquad one octave below the HPF.

12.8.2 Setting Low-Level Biquads

In the low-level path, some bass boost should be added. If a passive radiator or ported enclosure is used, the $f_{c\ (Boost)}$ should be set at 0.85 × $f_{c\ (Port/PassiveRadiator)}$, or greater. If bass boost is added below this, then the chance of port noise increases. It is recommended to set $f_{c\ (Boost)}$ slightly above the $f_{c\ (Port/PassiveRadiator)}$ and adjust for best sound as a starting point and then “tune to taste” in subsequent listening tests

12.8.3 Setting the Energy Estimator

The bandpass filter determines the range of interest for decision making. The lower boundary of the filter is determined from the DC blocking filter used in the main equalization. The upper boundary is determined from the highpass filter assigned to the High-Level Path. A good starting point is 1.1–1.25 x $f_{c\ (High\text{-}Level)}$.

The *Averaging Window* causes the mix between high level and low level to change faster as the averaging window is reduced. If the averaging window is increased, the mix responds more slowly. The recommended starting point is 10 cycles of 100 Hz or 100 ms.

12.8.4 Setting Dynamic Mixer Thresholds

The lower threshold determines when the mix beings, the upper determines the signal lever when mixing is complete. It is best to use audio test tracks and determine the best range of thresholds though listing tests.

Revision History

Changes from Original (January 2015) to A Revision	Page
• Changed the document title From: Using the TAS5754/6M2 HybridFlow Processor To: Using the TAS5754/6M and PCM5242 HybridFlow Processor	7

NOTE: Page numbers for previous revisions may differ from page numbers in the current version.

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