



Tempo Semiconductor, Inc.

## Application Note 02:

### Configuring Audio Processing Algorithms using the Audio System Configurator (ASC) Tool

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## Revision History

Revision	Date	Author	Comments
0.9	01/10/2017	Keith Lyszowski	First internal release.
1.0	06/07/2017	Keith Lyszowski, David Grant	Updated to use official company template. Updated ASC Tool screenshots.

## 1.0 Purpose

The purpose of this application note is to help the system designer understand the audio processing algorithms that are available in the Wolverine Family (ACS422x00, ACS422Mx00, ACS422x67, ACS422x68), Wookie Family (TSCS25xx, TSCS422xx) and Redwood (TSCS454xx) consumer audio (I2S / I2C) products from Tempo Semiconductor.

## 2.0 Description of Audio Processing Algorithms

The ACS422xXX / ACS422MxXX, TSCS25xx, TSCS422xx and TSCS454xx family of products feature several types of audio processing algorithms intended to improve or alter the way the audio sounds.

Typically, these features are used for speaker compensation, making small speakers sound as good as possible, however, these same audio processing algorithms can also be used to enhance the audio on any output to suit even the end-user's tastes for the content they are enjoying.

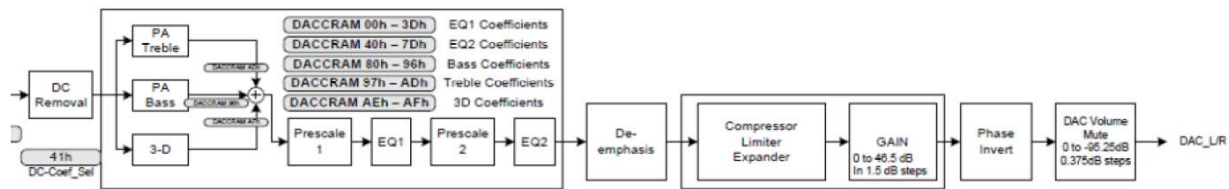
These families of products all feature psychoacoustic bass enhancement, psychoacoustic treble enhancement, 3D spatialization enhancement, two banks of fully parametric EQ / configurable filters, and a wide band dynamic range compressor (DRC).

The latest TSCS25xx, TSCS422xx and TSCS454xx product families also support a multiband compressor / limiter / dynamic range expander.

Each parametric EQ bank features a pre-scaler, 6 bands of left channel processing, and 6 bands of right channel processing.

The two EQ banks can be combined to provide a total of 12 bands of left channel processing and 12 bands of right channel processing.

Below is a diagram of the output audio processing chain:



All the audio processing algorithms are adjustable to suit a system designer's specific solution. The audio processing algorithms can be tuned and adjusted using Tempo's "ASC" tool (Audio System Configurator for Consumer Audio).

This tool is available upon request from Tempo by sending an e-mail to [support@temposemi.com](mailto:support@temposemi.com).

Note, some of the screenshots in this document may not exactly reflect the absolute latest version of ASC tool, that version you currently have or potentially future release versions of the ASC tool. This document will be updated on an as-needed basis.

### 3.0 Prior to Adjusting Any Algorithms

When you first receive the ASC tool, please unzip it. It will ask for a password. The first 4 digits in the file name are the password. For example, for ASC\_0509.zip, the password would be "0509".

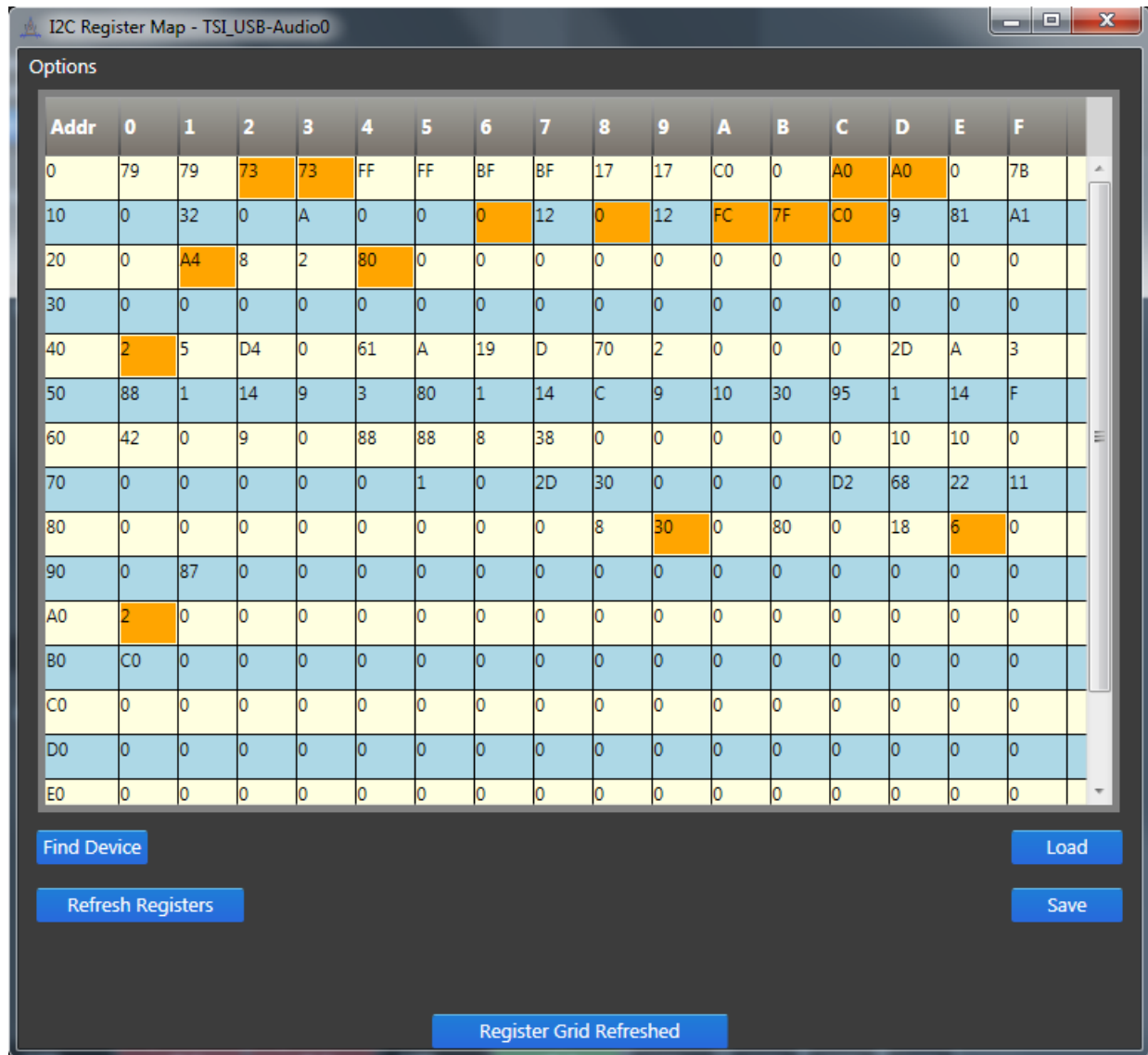
Please extract the zip file to any location that is convenient for you on your main Windows® OS install drive (typically your C:\ drive).

The ASC.exe file can be found inside the extracted folder.

When opening ASC, it is important to right click on the "ASC.exe" executable and choose "Run As Administrator".

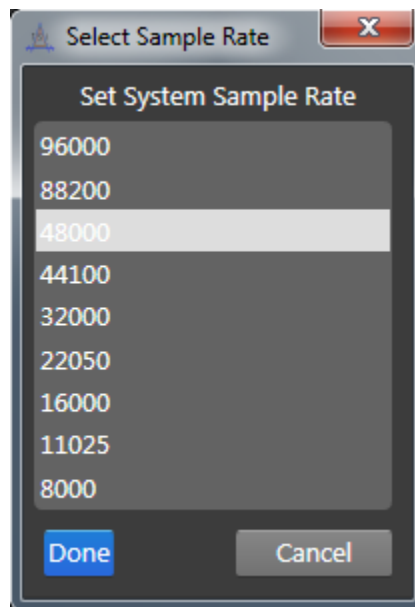
After ASC loads, the designer must initialize the part being configured. This is provided with a ".regv" file that can be loaded from the "I2C" option in the menu choices.

Select "I2C Register" and click on the "Refresh Registers" button. You should now see the values updated with the current settings of the part. Next, click on the "Load" button, and select the correct ".regv" value required for initialization. The modified registers values will appear orange. For an example of this type file, please contact your Tempo representative.



Because Tempo's audio algorithms operate in the frequency domain, it is important that the audio processor's values match the sampling rate of the audio content. Since the sampling rate is determined by the system design (via the I2S clock rates), the ASC tool allows the system designer to set the tool so that all values are calculated properly.

To set the sampling rate once ASC is opened, click "System" in the menu choices and choose "Set Sample Rate". There, the system designer can choose one of the supported sampling rates:



Note that if the sampling rate set in the tool does not match the incoming audio content, then the frequency values appearing in the tool will not be the exact frequencies affected by the algorithms.

*Hint: We recommend only selecting the sampling frequencies that the end-product can support as the exported configuration file (regardless of specific format) from the ASC tool will contain a complete set of register values (coefficients) for every processing algorithm for every sampling frequency. This can result in an increase in the size of the exported file.*

*For example, if you are designing a USB-C headset using the CP2615 and the TSCS25xx Audio Codec, you would only need to select 48000 (48kHz) as this is the only sampling frequency supported by that device, even though the TSCS25xx Audio Codec is capable of supporting all the sampling frequencies listed above.*

In addition, the ASC tool provides a way to keep the temporary modifications made during design testing. There are two function groups located above the frequency response graph:

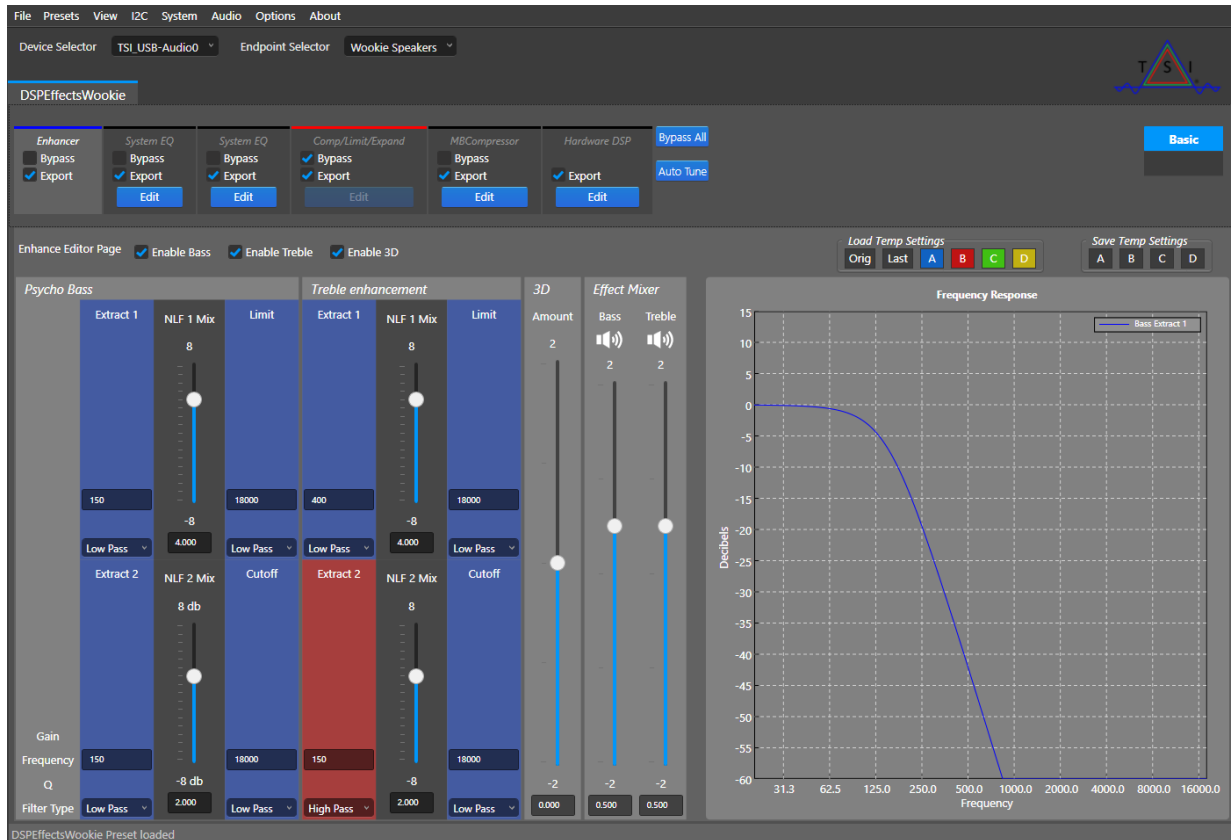


- Load Temp Settings:
  - Orig button - Loads the settings that were preset when this window was opened.
  - Last button - Restores the last configuration entered.
  - Color buttons - Loads previously saved settings from temporary configuration A/B/C/D.
- Save Temp Settings:
  - Saves current settings to temporary configuration A/B/C/D.

These temporary save functions also apply to 3D Stereo, Psychoacoustic Treble & Bass Enhancements, Parametric EQs, Wideband Dynamic Range Compression, and Multiband Compressor / Limiter / Expander\* features.

\*(For all product family devices except the ACS422xXX / ACS422MxXX)

## 4.0 Psychoacoustic Enhancements



### 4.1 Psychoacoustic Bass Enhancement

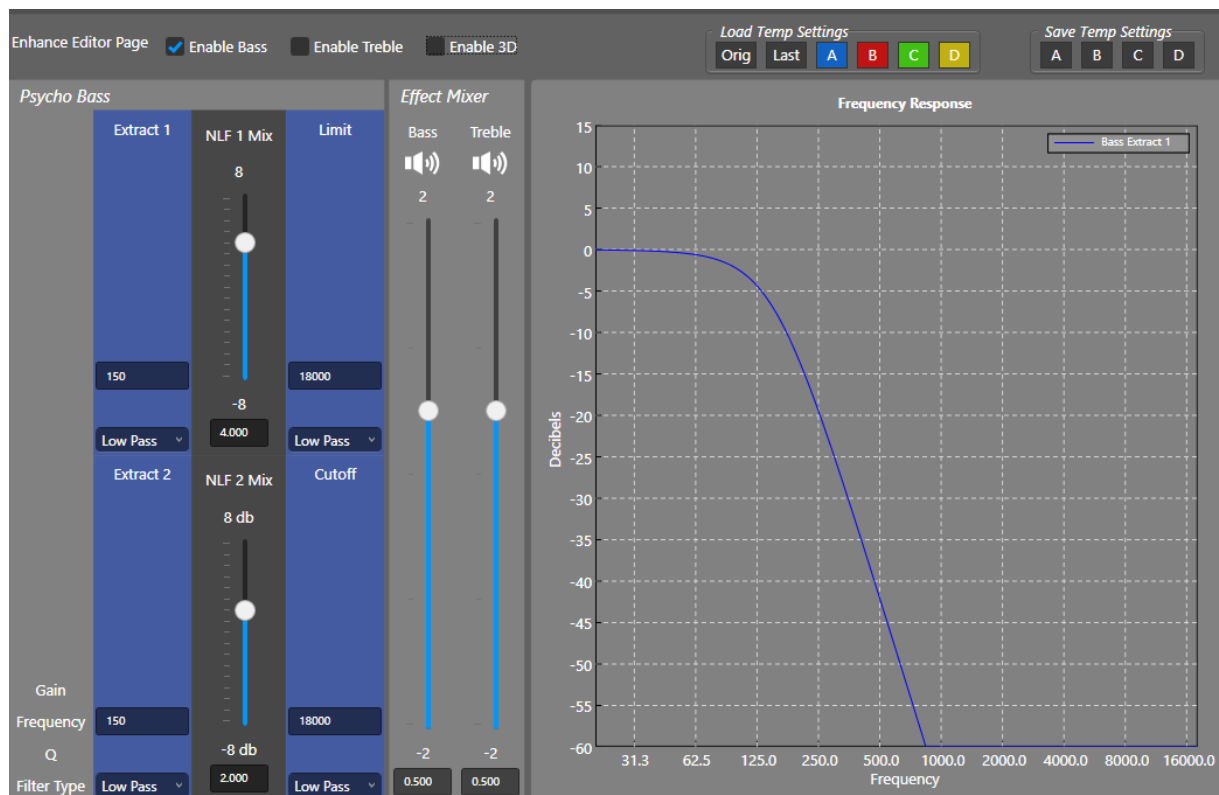
Today's portable electronic devices are making lives more connected to media, information, and each other. However, the trend for smaller, thinner, and lighter devices means that speakers have also gotten smaller. Due to the physics of reproducing sound, small speakers simply cannot reproduce appreciable amounts of bass frequencies. In fact, forcing a small speaker to reproduce bass frequencies that it is not normally capable of reproducing will generally cause speaker failure. To improve the bass response without causing damage to the speakers, Tempo features a psychoacoustic bass enhancement algorithm.

The psychoacoustic bass algorithm relies on a psychoacoustic principle called "missing fundamental". If the human ear hears a proper series of harmonics for a bass note, the listener will hear the fundamental of that series, even if it is not present.

To further explain how this works, assume a tablet device has a speaker solution with a natural low-frequency roll-off of 500Hz. In other words, the speakers cannot naturally

reproduce audible frequencies below 500Hz. Typically, a system designer will apply a high pass filter around 500Hz to prevent the speaker from being exposed to the low frequencies and prevent over-excursion damage, but it will also limit the fidelity experience of the end user. The psychoacoustic bass algorithm can be used to help the end user “hear” bass frequencies that the speaker cannot reproduce. Note that the psychoacoustic bass algorithm is only effective at improving the bass down to about  $\frac{1}{2}$  of the roll-off frequency of the speakers.

Inside the ASC tool, the psycho acoustic algorithm is adjusted by unchecking the “bypass” box in the Enhancer panel and then clicking on “Edit”. There, the system designer can choose to “Enable Bass” via a checkbox. The screen appears like this:



The “Extract 1” and “Extract 2” boxes will choose the bandwidth of frequencies that the bass algorithm will use as the fundamentals when deciding which harmonics to inject. Whenever adjusting the Extract 1 or Extract 2 settings, a graphical representation will appear in the Frequency Response graph.

The “NLF 1 Mix” and “NLF 2 Mix” control the amount of odd and even harmonics injected. The proper ratio of these settings is completely up to the system designer, as the setting can dramatically alter the way the audio sounds. However, satisfactory results are typically obtained with an NLF 1 Mix setting of a moderate positive number (such as 3 or 3.5) and an NLF 2 Mix setting of a small positive number (such as 1).



The Limit setting is typically a low pass filter, and it controls how many harmonics are injected into the audio signal. This value is best chosen by the system designer according to preference, but a typical frequency is 2 to 4 times the value of the high-pass filter frequency.

The Cutoff frequency is typically set to a value close to the high-pass filter setting used by the system designer.

The Effect Mixer controls how much of the “enhanced” signal is applied to the DAC as versus how much “dry” or unenhanced signal. Note that the sum of the values of the Bass, Treble, and Dry mix must equal one if the system designer wishes to maintain unity gain. Any summed value over 1 indicates that digital gain may be applied. While the exact mix of Bass, Treble, and Dry is best determined by the system designer according to acoustical preference, satisfactory results have been obtained with Bass (if used) set to 0.2, Treble (if used) set to 0.2.

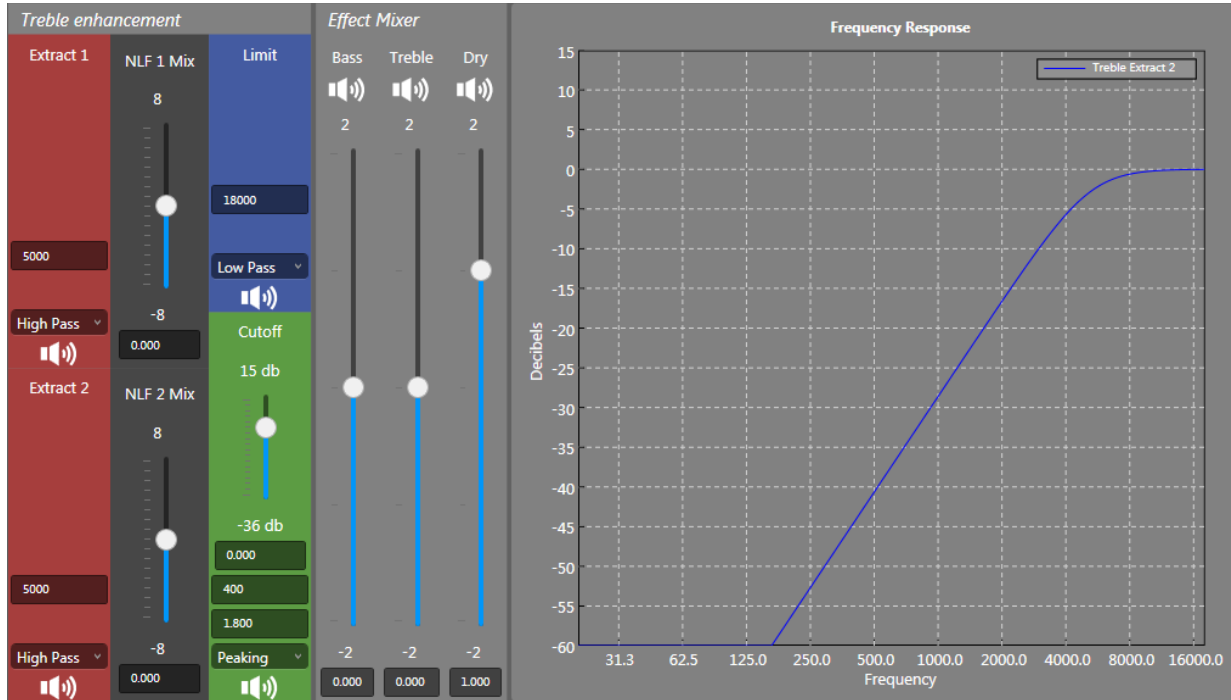
Note: To simplify the design experience when tuning the Psychoacoustic Bass Enhancement, control over the Dry mix parameter is no longer offered in the latest ASC tool apps (starting with release ASC\_0509) as this parameter has been “hardcoded” to a value of 1.

## 4.2 Psychoacoustic Treble Enhancement

One of the mechanisms used to limit the bit rate for compressed audio is to first remove high frequency information before compression. When these files are decompressed, this can lead to dull sounding audio. The Tempo treble enhancement replaces these lost high frequencies.

The enhanced treble function works much like the enhanced bass; however, its intended use is different. The enhanced treble uses a nonlinear function to add treble harmonics to a signal that has limited high frequency bandwidth (such as a low bit rate MP3). In this case, the algorithm makes use of the audio fact that presence of audio between 4K and 8K is a good predictor of audio between 10K and 20K. This implementation extracts the high frequency content that is available in the audio, generates harmonics of those frequencies. These harmonics are then summed back into the original signal, providing a brighter sound.

Inside the ASC tool, the psycho acoustic algorithm is adjusted by unchecking the “bypass” box in the Enhancer panel and then clicking on “Edit”. There, the system designer can choose to “Enable Treble” via a checkbox. The screen appears like this:



The “Extract” and “Extract 2” boxes control the audio that is used to generate the treble distortion. Typically, these are set to 4000Hz and 8000Hz, but the system designer can choose values that best suit the solution.

The “NLF 1 Mix” and “NLF 2 Mix” control the amount of odd and even harmonics injected. The proper ratio of these settings is completely up to the system designer, as the setting can dramatically alter the way the audio sounds. However, satisfactory results are typically obtained with an NLF 1 Mix setting of a moderate positive number (such as 3) and an NLF 2 Mix setting of a smaller positive number (such as 2).

The “Limit” box controls how many high the harmonics are inserted into the frequency band. Typically, this is set to 18,000Hz to prevent aliasing that can cause the speakers to sound harsh.

The “Cutoff” box is not used in this application. Leave the default settings.

As described in detail in the Psychoacoustic Bass section, the “Effect Mixer” controls how much of the “enhanced” signal is applied to the DAC as versus how much “dry” or unenhanced signal. While the exact mix of Bass, Treble, and Dry is best determined by the system designer according to acoustical preference, satisfactory results have been obtained with Bass (if used) set to 0.2 and Treble (if used) set to 0.2.

Note: To simplify the design experience when tuning the Psychoacoustic Treble Enhancement, control over the Dry mix parameter is no longer offered in the latest ASC tool apps (starting with release ASC\_0509) as this parameter has been “hardcoded” to a value of 1.

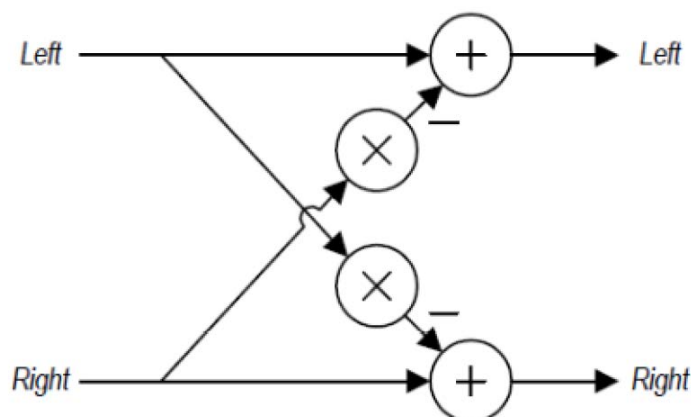
## 4.3 3D Spatialization Enhancement

Another detriment to having small, thin, and light portable devices is that the stereo separation between the left and right channels is usually very small. This is because the left and right speakers are usually placed relatively close together. To enhance the end user’s perceived stereo depth, Tempo has implemented a digital depth enhancement option to artificially increase the separation between the left and right channels, by enabling the attenuation of the content common to both channels.

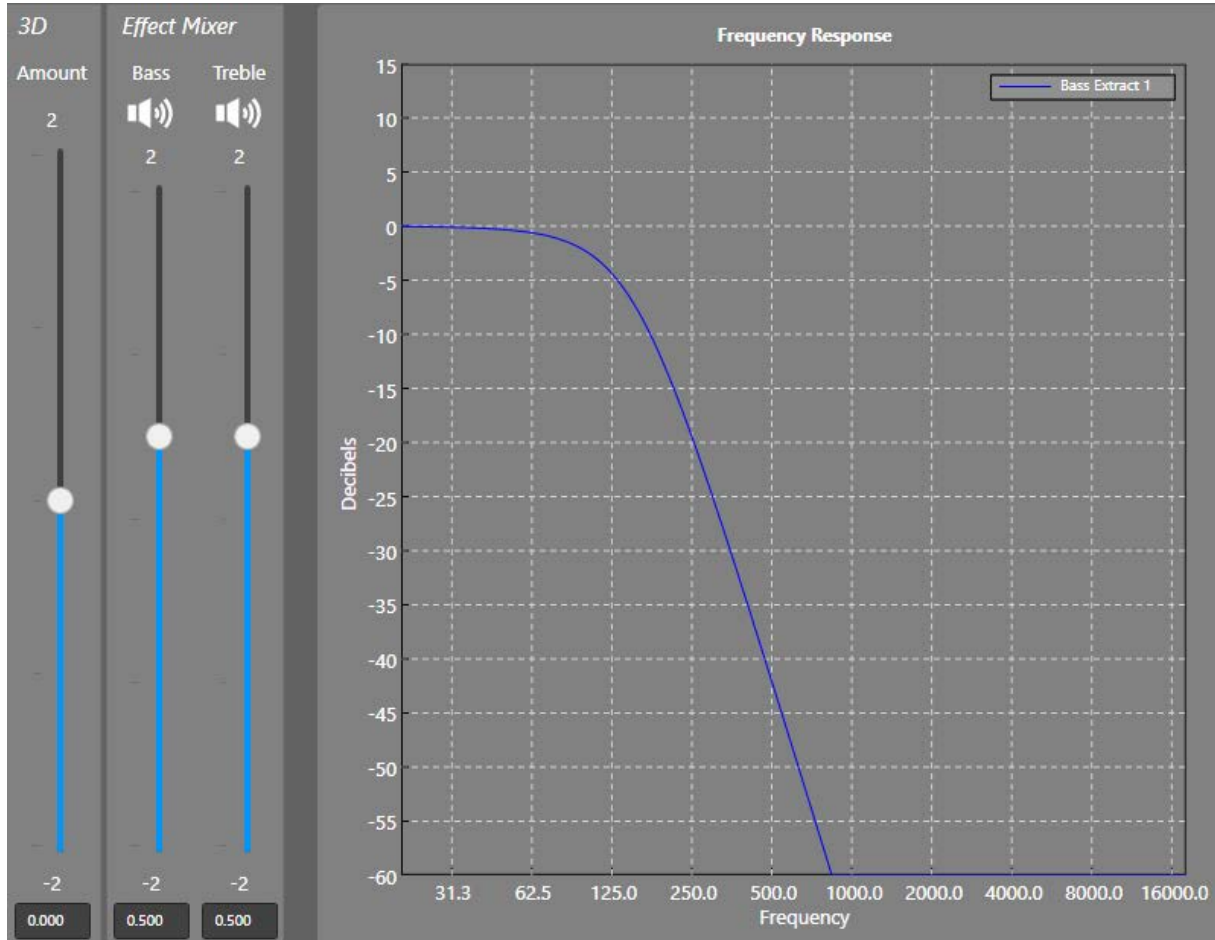
The 3D enhancement algorithm is a tried and true algorithm that uses two principles.

1. If the material common to the two channels is removed, then the speakers will sound more 3D.
2. If the material for the opposite channel is presented to the current channel inverted, it will tend to cancel any material from the opposite channel on the current ear. For example, if the material from the right is presented to the left ear inverted, it will cancel some of the material from the right ear that is leaking into the right ear.

A diagram of the 3D Enhancement algorithm is below:



Inside the ASC tool, the 3D enhancement algorithm is adjusted by unchecking the “bypass” box in the Enhancer panel and then clicking on “Edit”. There, the system designer can choose to “Enable 3D” via a checkbox. The screen appears like this:

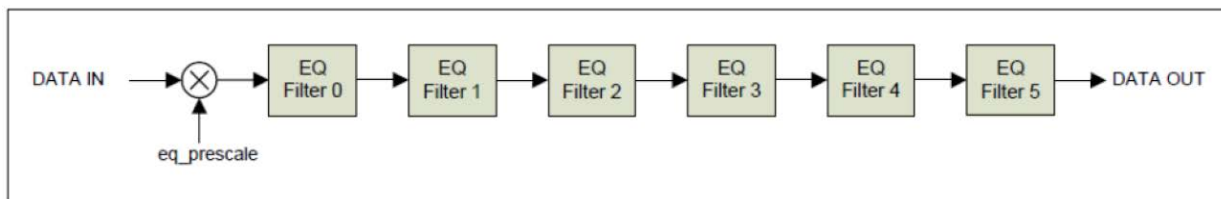


The exact value of the 3D “Amount” is completely adjustable and best determined by the system designer for his/her specific solution. However, satisfactory results are typically obtained by using a value of “-0.500”.

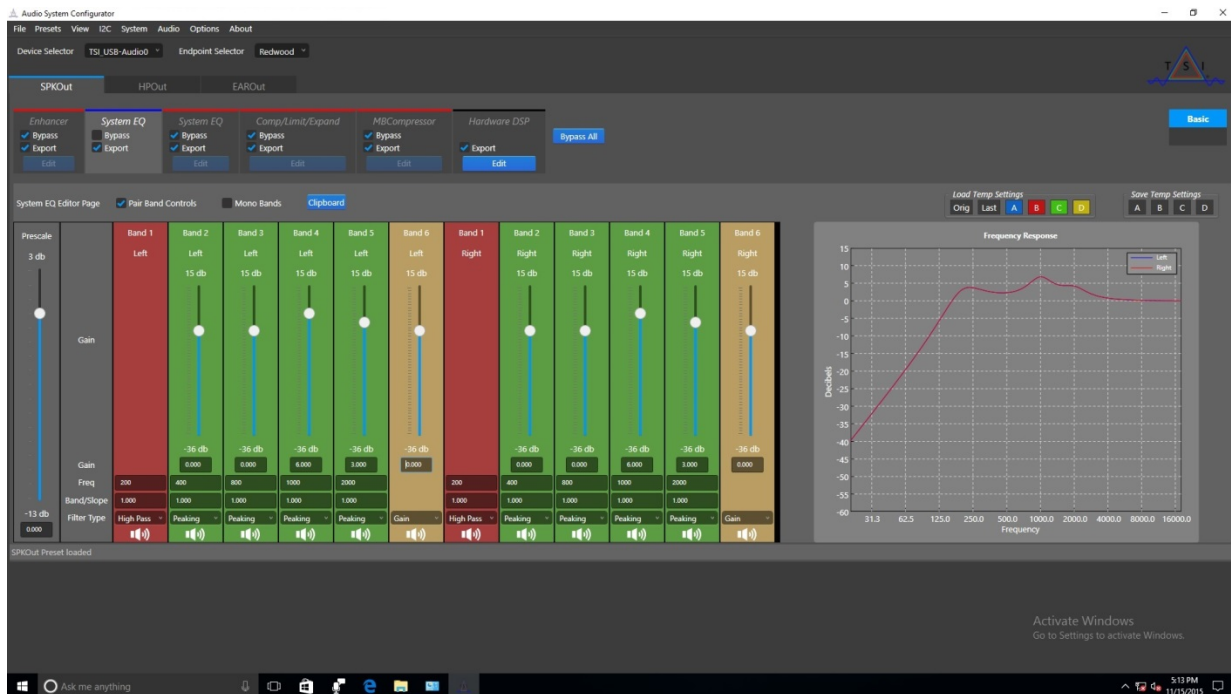
Please note that the intensity of the 3D spatialization algorithm is also dependent upon content. For example, 2 channel Lt, Rt downmixed Dolby® Audio, Dolby Surround, Dolby Pro Logic® or Dolby Pro Logic II encoded content will result in a different 3D Spatialization effect level compared to an Lo, Ro downmixed or stereo content due to difference in the energy content found on the L and R channels for these various mixes.

## 5.0 Parametric Equalization Algorithms

Tempo's ACS422xXX, TSCS25xx, TSCS422xx and TSCS454xx products feature two banks of 12 fully parametric EQ bands (6 for the left channel and 6 for the right channel) with a prescale function. The EQ banks are very flexible, allowing for use of both banks simultaneously on one output path, or allowing for one bank to be dedicated to the amplifier path and the second bank to be dedicated to the headphone path. Additionally, the system designer can choose how many EQ bands to enable for each bank. Please refer to the product's datasheet for further instructions on these features. A graphical representation of each EQ bank is as follows:



Inside the ASC tool, the EQ banks are adjusted by unchecking the “bypass” box in either of the “System EQ” panels and then clicking on “Edit”. The first “System EQ” panel appearing in the signal chain is EQ Bank 1, which the second “System EQ” is EQ Bank 2. If both System EQ panels are not bypassed, then both EQ banks will be processed concurrently. The screen appears like this:



There are two check boxes to the left top of System EQ Editor Page. One is Pair Band Controls and it is checked by default. It means the left and right channels change values simultaneously when user adjusts the band bars. Uncheck it, and the user can change the value of left and right bands separately.

The other check box is for Mono Bands. Check this box and then the user can only change left channel values when adjusting the band bars.

At the right side of the Mono Bands check box is the Clipboard button. This is only for the ASC tool developers to check their registry values.

Note that inside each EQ bank, the “prescale” value applies to both the left and right signals while bands 1-6 apply to the left channel and bands 7-12 apply to the right channel.

The Prescale value is intended to be a digital attenuator that attenuates the digital signal prior to any EQ processing. Since the EQ is a digital function, it is highly recommended to use only attenuation in the EQ bands to avoid exceeding digital full scale. However, some situations may arise where gain is desired in a specific frequency band. In this case, Tempo recommends that the prescale be set to a negative value whose absolute value is greater than or equal to the largest amount of positive gain existing in the EQ bands.

*Please note that Tempo recommends setting the prescale value to “-2dB” even if the EQ bands are attenuation only. This helps prevent any digital clipping that might occur due to artifacts in the IIR filters. To offset the 2dB decimation in the prescale, +2dB of gain can be applied either in one of the EQ bands, the compressor’s make-up gain, or in the gain value for the speaker or headphone output.*

Each EQ band can be assigned one of 8 distinct filters:

- **Peaking** - The most generic form of a parametric equalization filter, a peaking filter allows you to emphasize or attenuate specific frequencies. Tempo’s peaking filters have adjustable center frequencies, adjustable gain or attenuation, and adjustable bandwidth.
- **Low Shelf** - This is a common shelving filter that allows the system designer to choose a specific frequency and then attenuate or gain the content below that chosen frequency.

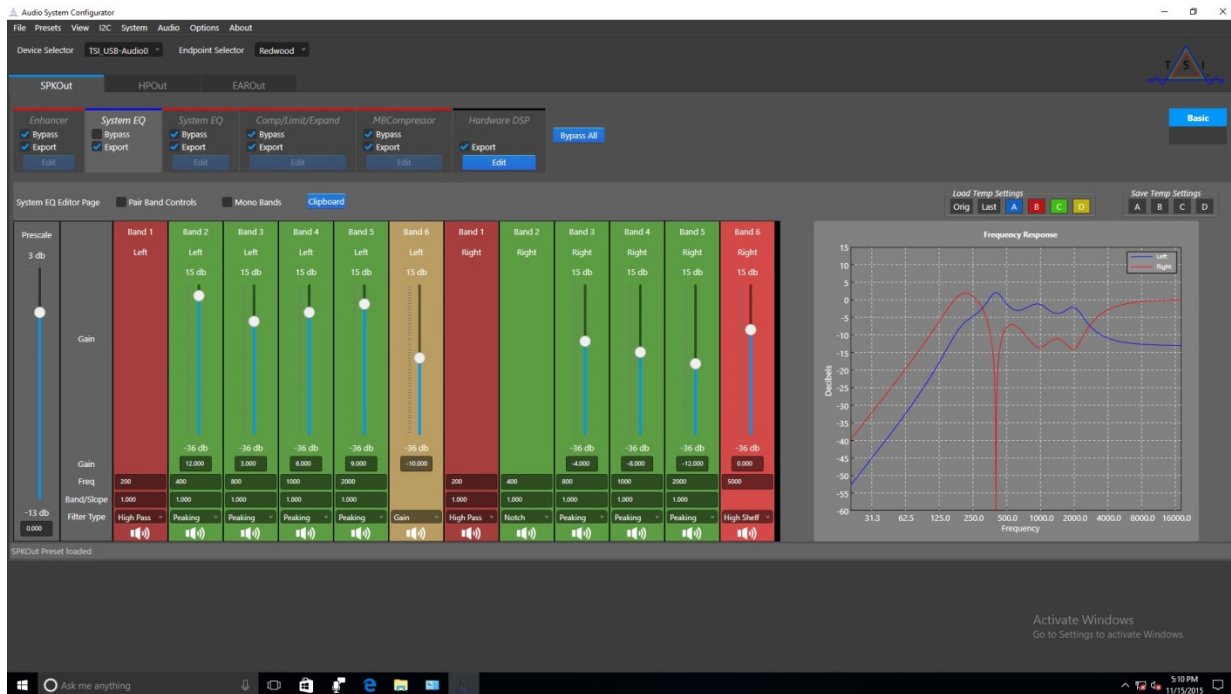
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- **High Shelf** - This is a common shelving filter that allows the system designer to choose a specific frequency and then attenuate or gain the content above that chosen frequency.
  - **Low Pass** - This is a 2nd order low pass filter with an adjustable -3dB frequency.
  - **High Pass** - This is a 2nd order high pass filter with an adjustable 3dB frequency.
  - **Bandpass** - This is a bandpass filter with an adjustable center frequency and adjustable bandwidth.
  - **Notch** - Like a peaking filter with a very narrow bandwidth, the notch filter is typically used to filter out a very specific frequency, such as AC line noise.
  - **Gain** - This is a filter that applies a universal gain or attenuation to the incoming content.

If the system designer is more familiar with using Q Factor instead of bandwidth, then he/she can convert the bandwidth values into a "Q" value using the following equation:

$$Q = f0/BW, \text{ where } f0 \text{ is the center frequency of the filter}$$

Whenever a change is made to a filter, the graphical representation of that change is reflected in the frequency response graph on the bottom right of the system. The left channel frequency response appears in blue and the right channel appears in red. Note that you can adjust the X and Y axis values of the frequency response graph by right-clicking on the graph.

Below is an example of some various EQ filter types and the associated values entered the ASC tool:



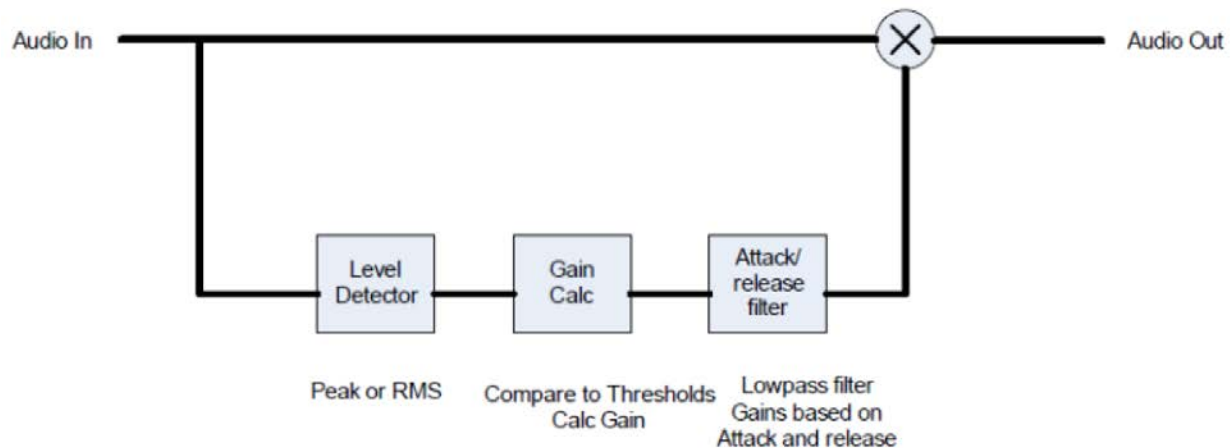
## 6.0 Dynamic Range Compression

### 6.1 Compressor Algorithm

One method for improving the overall SPL or loudness in small-speaker devices is to reduce the amount of dynamic range in the audio content. If the delta in volume between the loud explosions and the soft vocal scenes in a movie soundtrack is reduced, the overall system volume is perceived to be louder. This is accomplished by using a “dynamic range compressor” or compressor for short.

Tempo has implemented a single band compressor that will reduce the overall signal gain by a programmable ratio any time the incoming signal exceeds a programmable threshold. Additionally, Tempo’s compressor allows for a programmable “attack” and “release” time, which provides a degree of control over how quickly the compressor acts. A basic block diagram of the compressor is shown below:





As shown in the diagram, there are three main control blocks:

- **Level Detector** - The level detector, detects the level of the incoming signal. Since the compressor is designed to work on blocks of signals, the level detector will either find the peak value of the block of samples to be processed or the rms level of the samples within a block.
- **Gain Calculation** - The gain calculation block is responsible for taking the output of the level detector and calculating a target gain based on that level and the compressor and expander thresholds. The compressor recalculates the target gain value every block, typically every 10ms.

The gain calculation operates in 2 regions:

- o **Linear Region** - If the level is higher than the expander threshold and lower than the compression threshold, then the gain is 1.0.
- o **Compression Region** - When the level is higher than the compressor threshold, then the compressor is in the compression region. The gain is a function of the compressor ratio and the signal level.
- **Attack/Release Filter** - To prevent objectionable artifacts, the gain is smoothly ramped from the current value to the new value calculated by the gain calculation block. This is achieved using a simple tracking low pass filter to smooth out the abrupt transitions.

## 6.2 Limiter Algorithm

As the speakers in today's consumer electronics are small, they are also prone to damage from over excursion and over power. One way to protect these small speakers

is to apply a limiter algorithm. Limiting is an extreme example of the compressor where, as the input signal level is increased, the gain is decreased to maintain a specific output level. Limiters may also be used to provide hearing protection on the headphone output, intentionally limiting the output level before full scale is reached.

The Limiter function will limit the output of the DSP module to the Class-D and DAC modules. If the signal is greater than the programmable threshold, it will saturate at the threshold as the final processing step within the DSP module. It is best to think of a Limiter as essentially a compressor with a very high compression ratio.

The exact settings for the limiter are very dependent on the system solution, speakers chosen, and audible preferences.

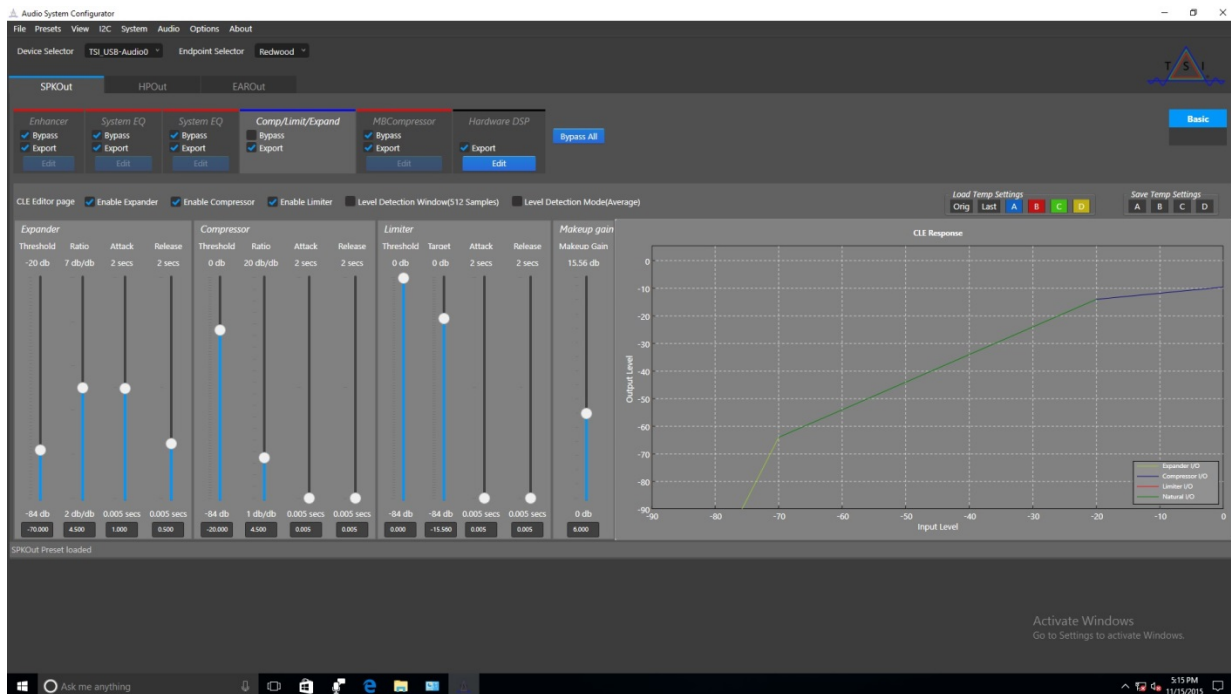
## 6.3 Expander Algorithm

The expander is a close relative of the compressor. Rather than using signal dependent gain to restrict the dynamic range, the expander uses signal dependent gain to expand the dynamic range. Thus, if a signal level is below a particular threshold, the expander will reduce the gain even further to extend the dynamic range of the material.

As recording studios have continually increased the amount of compression of recorded music and as “compressed audio” such as MP3 has increased in popularity, the dynamic range of the music has been reduced significantly. An expander can help to restore some of the lost dynamic range, improving the end user’s experience when playing back highly compressed MP3’s, satellite radio, internet radio, or other media sources.

Inside the ASC tool, the Dynamic Range Compressor is adjusted by unchecking the “bypass” box in the “Comp/Limit/Expand” panel and then clicking on “Edit”. The system designer can then enable the specific functions by clicking the appropriate checkbox.

The screen appears like this:



A window to the right of the Compressor/Limiter/Expander adjustment section shows the graphical response of the compressor.

### The Compressor algorithm has five parameters:

- **Threshold** - This is the point at which the limiter will begin limiting the output. The value is in dB relative to full scale. Any audio signal that exceeds this threshold will be limited.
- **Ratio** - This is the ratio at which the compressor will operate. For example, if the ratio is set to 3.000, then the input signal must exceed the threshold by 3dB for the compressor to allow 1dB of output gain.
- **Attack Time** - This is the amount of time it will take for the signal to reach the compressed level once the signal has exceeded the threshold. Essentially, this is the compressor ramping time.

- **Release Time** - This is the amount of time it will take for the signal to return to normal after the signal is below the compressor's threshold.
- **Makeup Gain** - This control allows the system designer to bring the total signal output back up to a more useable level if it has been decreased by higher compression settings.

#### The Limiter algorithm has four parameters:

- **Threshold** - This is the point at which the limiter will begin limiting the output. The value is in dB relative to full scale. Any audio signal that exceeds this threshold will be limited.
- **Target** - This is the point to which the limiter will limit the output level. Typically, this value is set to the same point as the threshold.
- **Attack Time** - This is the amount of time it will take for the signal to reach the limited level once the signal has exceeded the threshold. Essentially, this is the limiter ramping time.
- **Release Time** - This is the amount of time it will take for the signal to return to normal after the signal is below the limiter's threshold.

#### The Expander algorithm has four parameters:

- **Threshold** - This is the point below which the expander will begin reducing the output. The value is in dB relative to full scale. Any audio signal that falls below this threshold will be reduced in gain.
- **Ratio** - This is the ratio at which the expander will operate. For example, if the ratio is set to 3.000 and the input signal falls below the threshold by -3dB, the expander will only output -9dB.
- **Attack Time** - This is the amount of time it will take a signal to reach the expanded level after the signal has crossed below the expander's threshold. Essentially this is the expander's ramping time.
- **Release Time** - This is the amount of time it will take a signal to return to normal after the signal has crossed above the expander's threshold.

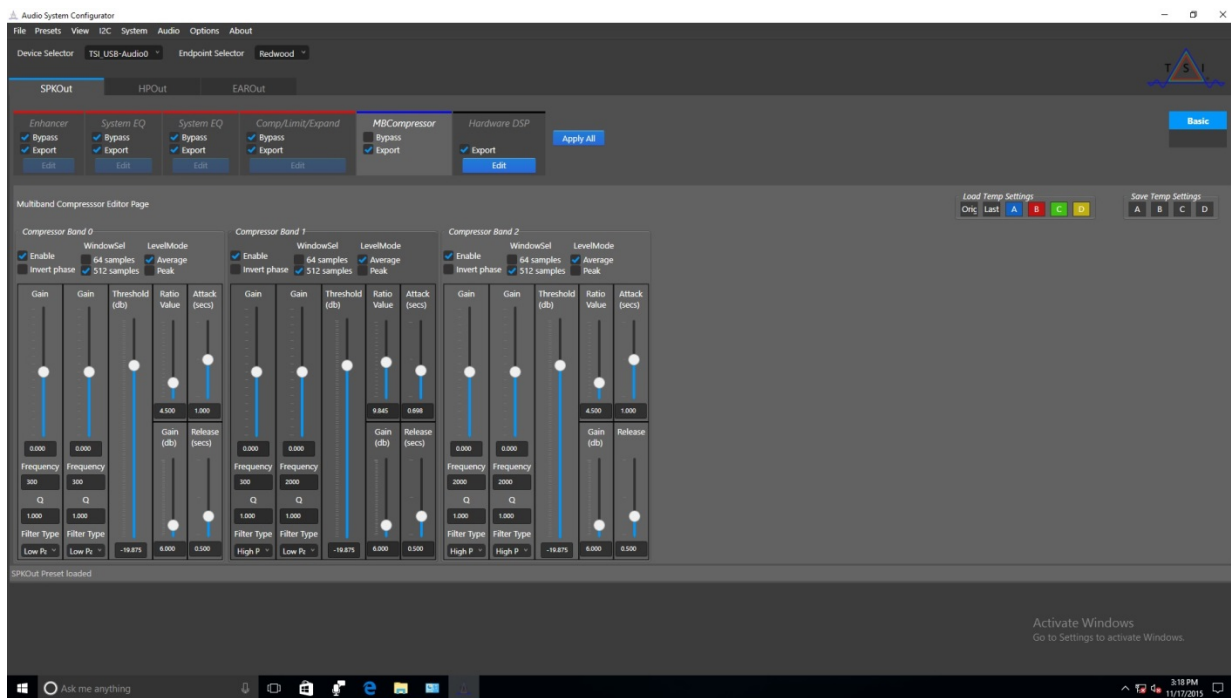
## 7.0 Multiband Compressor

The compressor algorithm affects the audio content over all frequencies, but the multiband compressor provides an extra two EQ bands, to allow the designer to select the frequency range that they would like to change.

The parameters of multiband compressor are the same as the compressor algorithm, and the audio designer can use the extra two EQ bands to select or limit the range of frequencies that they would like to affect.

Inside the ASC tool, the MBCompressor is adjusted by unchecking the “bypass” box in the “MBCompressor” panel and then clicking on “Edit”. The system designer can then enable the specific functions by clicking the appropriate check boxes.

The screen appears like this:



It provides three Compressor Bands: 0, 1 and 2. Their functions are all the same so the designer can use any one or all of them to adjust.

There are several check boxes at the top of the controls:

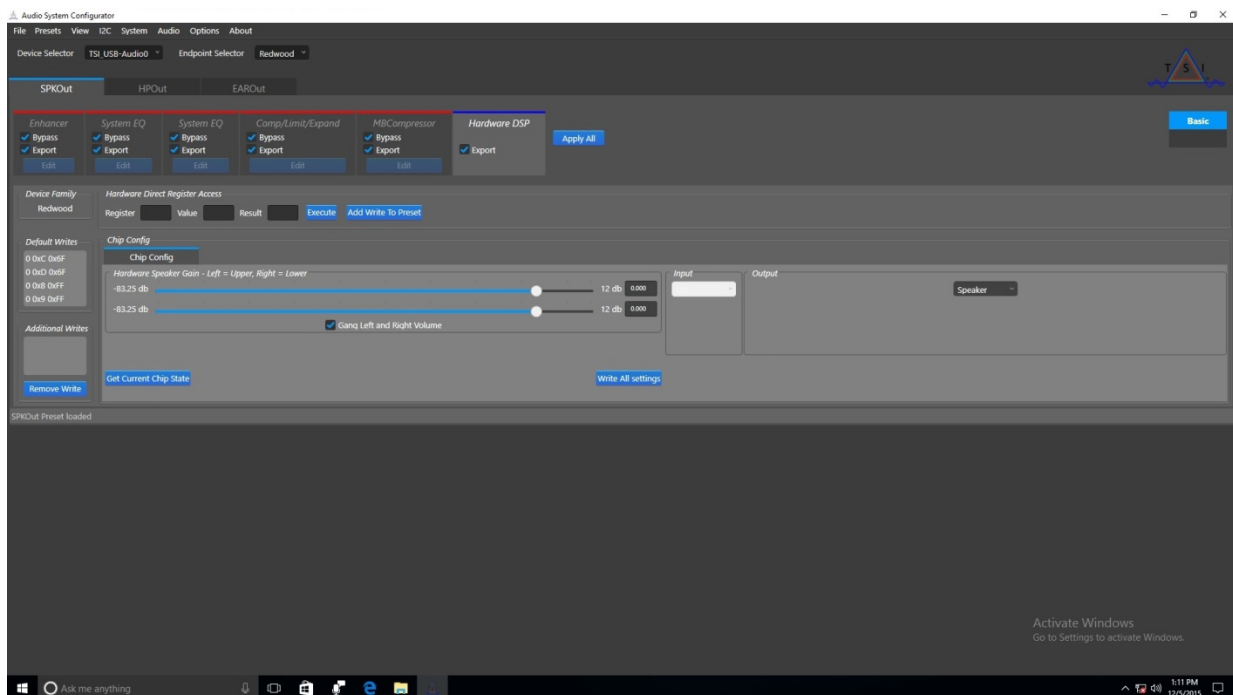
- **Enable** - This will enable/disable the individual compressor.
- **Invert Phase** - This will invert the phase of the audio affected by the compressor.

- **WindowSel** - Determines the numbers of samples that the designer wishes to use. There are selections for 64Hz or 512Hz.
- **LevelMode** - Chooses between using the average RMS or the peak value of the samples within the range of frequencies.

## 8.0 Hardware DSP

This screen allows the designer to check or set the functions by registry address.

It appears like this:



- **Device Family** - This will show the codec name or part number.
- **Default Writes** - This will show default states of input path and output gain values.
- **Additional Writes** - Enter the register address and values of the functions that you wish to modify here.
- **Remove Write** - Clicking this button will erase the register address and values in the additional writes area.
- **Hardware Direct Registry Access** - The designer can observe the values of specific registers here.

- **Chip Config** - Designer can adjust the speaker gain of the left and right channels simultaneously, or separately, if the Gang Left and Right Volume box is unchecked.
- **Write All Settings** - After adjusting the speaker gain values, press this button to write them into the codec registers.
- **Get Current Chip States** - the designer can get the value of current speaker gain registers after pressing this button.
- **Input** - The designer can select input path here. This depends on codec capabilities.
- **Output** - The designer can select output path here. This also depends on codec capabilities.

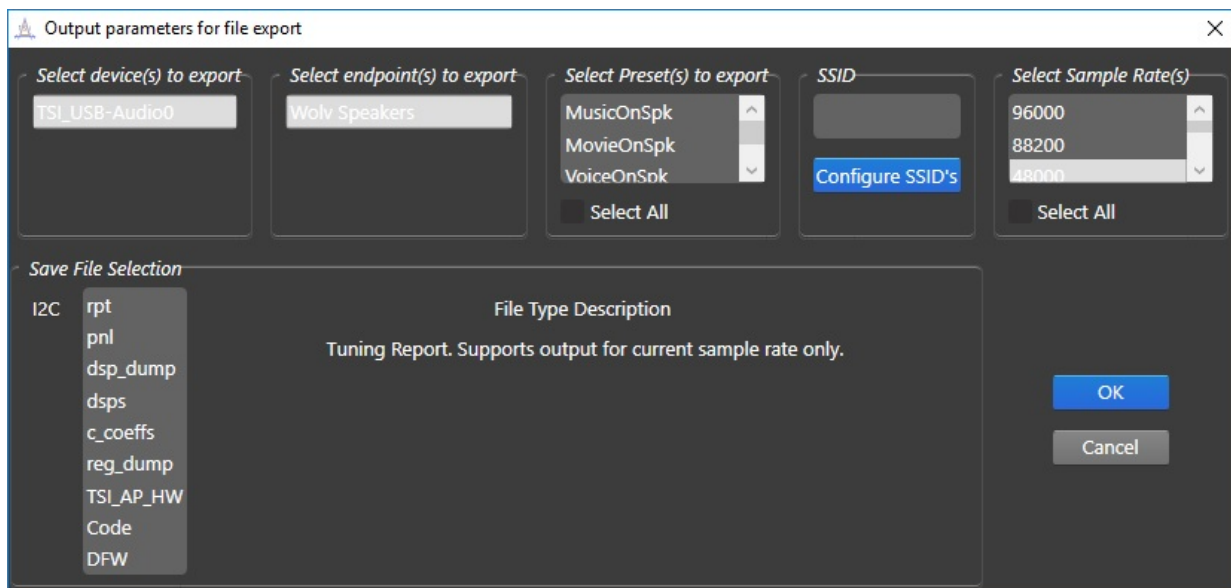
## 9.0 Putting it All Together

The Tempo Semiconductor ACS422xXX, TSCS25xx, TSCS422xx and TSCS454xx products are very versatile parts with regards to audio processing. While this document has focused on each individual feature, it is important to note that any or all of the features can be enabled simultaneously. As a system designer, you can “choose your own adventure.” However, be cautious and aware of how the processing is affecting the total output level, phase, and fidelity of the part.

## 10.0 Saving the Settings

The ASC is a very versatile and powerful tool, allowing the system designer to perform real-time adjustments of the audio processing algorithms. It is important to note that the tool does provide several output formats for the system designer. Once a design is complete, or if the designer chooses to save the work in progress, he/she can click on the “File > Export presets to file” button in the top left of the ASC tool to save the work.

The screen appears like this:



The output choices are:

- **Select device(s) to export** - This depends on the codec and is fixed.
- **Select endpoint(s) to export** - This depends on the codec and is fixed.
- **Select Preset(s) to export** - The designer can select the endpoint to export presets. The “Check Select All” checkbox will export the presets of all endpoints.
- **SSID** - The designer can add or remove a SSID by pressing the “Configure SSID’s” button.
- **Select Sample Rate(s)** - ASC provide sample rates of 8000, 11025, 16000, 22050, 32000, 44100, 48000, 88200 and 96000. Check the “Select All” checkbox to export presets for all sample rates.



The formats for preset exporting are:

- **RPT** - This is a text based report of all the settings included in the current work in progress.
- **PNL** - This is ASC's "native" file format. This is the only file format that can be imported by ASC. This file will allow ASC to immediately resume where it left off.
- **DSP\_DUMP** - This file contains the memory dump of the current state of ASC (EQ coefficients, enhancement parameters, etc).
- **DSPS** - This file contains the memory dump of the current state of ASC (EQ coefficients, enhancement parameters, etc). This file is formatted for use inside a C-code program.
- **C\_COEFFS** - Like DSPS, this file contains the memory dump of the current state of ASC (EQ coefficients, enhancement parameters, etc). This file is formatted for use by Tempo's "devregtest.jar" direct register access tool.
- **REG\_DUMP** - This file contains the dumped data of register addresses and values only.
- **TSI\_AP\_HW** - This is a Bios Snippet file (saved as Hex File) of the HW DSP presets only. 26 registers and 176 DACCRAM locations. Also writes a ".dsp\_dump" file.
- **Code** - This is a raw coefficient file and registry dump in C code format with comments indicating EQ0, EQ1, Enhancer memory, etc.
- **DFW** - This file contains raw coefficients in HEX format.

The system designer of a tablet or microcontroller device will need to format the output from the ASC application to be usable by their specific code used in their system.



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