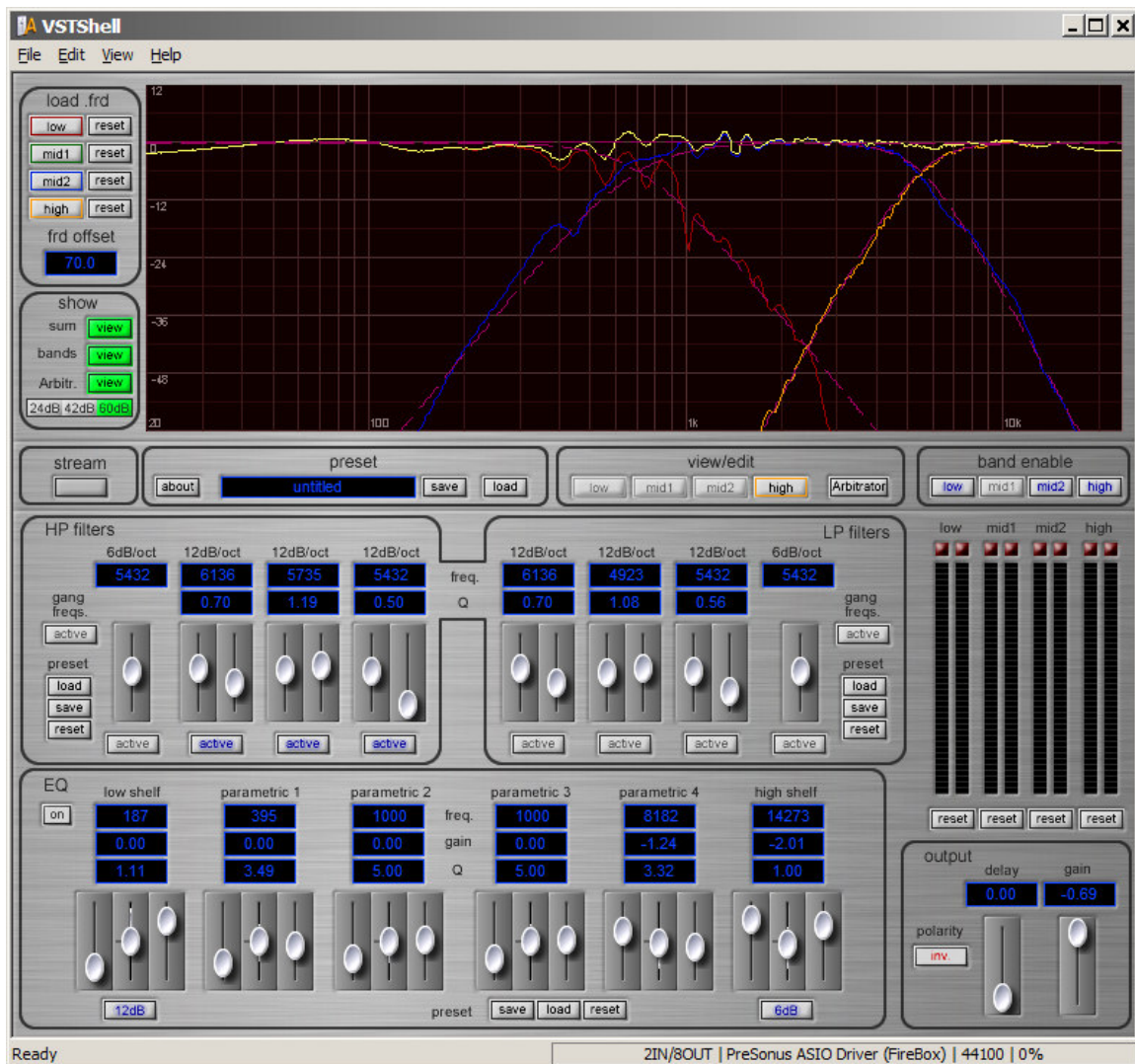


# **LOUDSPEAKER FREQUENCY ALLOCATOR**

## **USER GUIDE**

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## Introduction

The Loudspeaker Frequency Allocator is an innovative PC and Mac application that performs the duties of an active 4-way stereo loudspeaker crossover. It is a VST compatible plug-in, which means that it requires a host application for the connection to computer sound cards. It comes packaged with such a host- called VSTShell, but it can function within a variety of VST hosts such as dedicated audio editing programs (Steinberg Cubase and Nuendo, Sony Vegas, 12 Tone Systems Sonar) and audio processing hosts such as the Console.

### \*A word about the VSTShell.

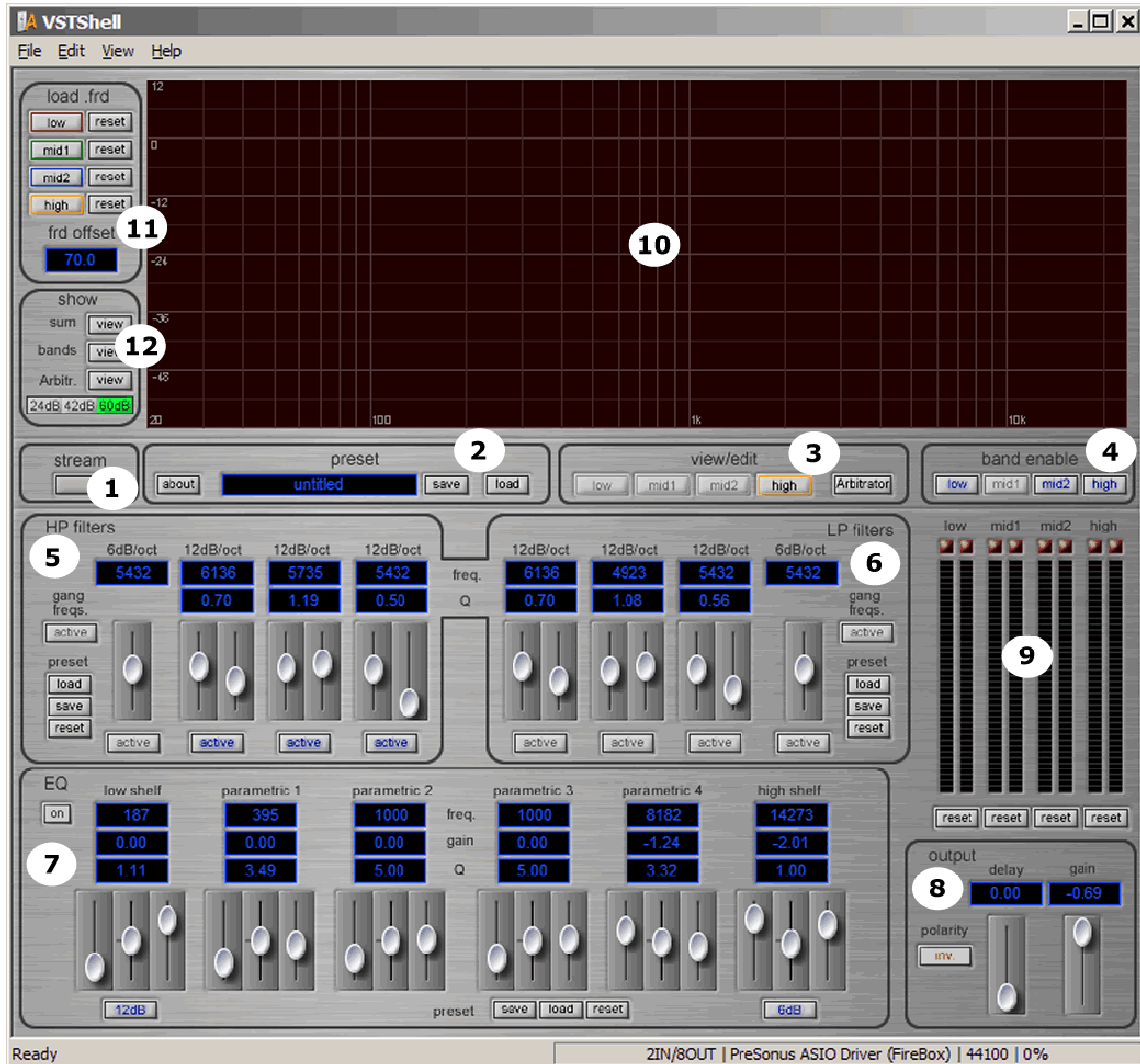
VSTShell only works with sound cards that have ASIO drivers. If your sound card is not ASIO compatible, you can still use the Allocator, but you will have to load it within a VST host that is capable of accessing non ASIO sound cards. **The Console** (available for purchase from [www.console.jp](http://www.console.jp)) is such a program. The Allocator has been tested inside the Console and it has shown to be fully compatible.

The Frequency Allocator's crossover section is preceded by a unique processor called The Phase Arbitrator. The Phase Arbitrator changes the phase relationships of the audio material it processes without affecting its frequency response. The special function of the Arbitrator is that it rolls the phase in the opposite direction from a traditional crossover. An analog crossover (and its digital IIR-based counterpart) exhibits a phenomenon called group delay. It means that after filtering, frequencies below the filter's cutoff frequency are delayed in relation to frequencies above the cutoff point. This results in smeared transients, which can be audible.

The Phase Arbitrator actually pushes frequencies below the cutoff point **ahead** of the frequencies above the cutoff point. When the Allocator/loudspeaker combination performs its "normal" phase distortion, it puts the phase relationships back in order! We can say that the Arbitrator section does a **preventive damage control**. But, in order to work properly, the two sections need to be dialed in to identical or at least very close settings. The Tutorial section of this manual explains the procedure for achieving transient perfect reproduction from your loudspeaker.

## User Controls

### Frequency Allocator Screen:



1. **Stream.** This button turns the audio stream on and off.
2. **Preset.** This is for global presets. Every setting **including any imported frd** files gets saved in this preset.
  - "Save" opens a standard dialog box where the user can name the presets and navigate to a location where the presets are to be saved.
  - "Load" opens a standard dialog box where the user can navigate to and recall presets.
  - "About" switches to the page with a quick start guide and the Allocator serial number.

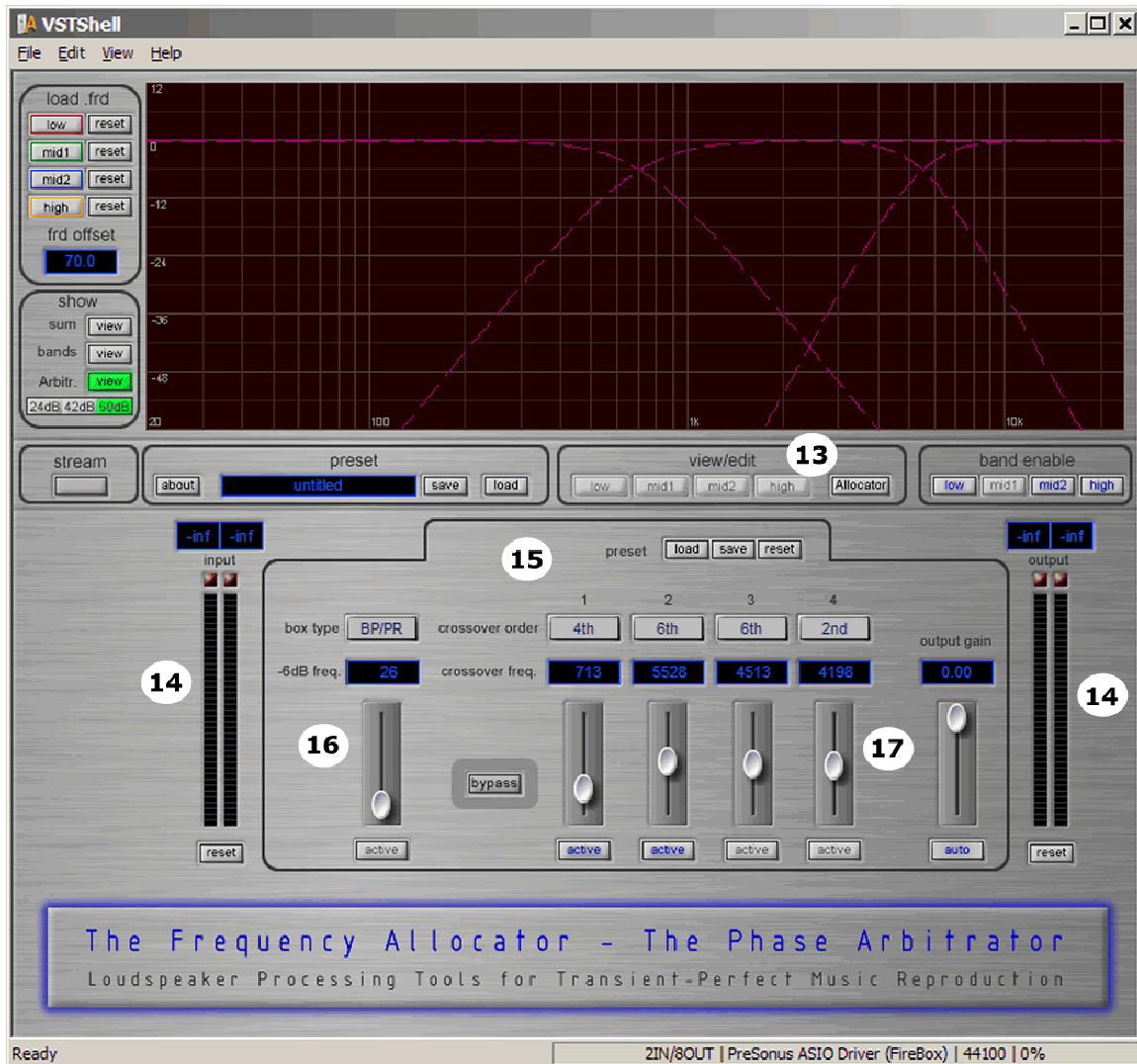
3. **View/Edit.** These buttons switch between individual Allocator bands.
  - "Low" shows the settings for section 1
  - "Mid1" shows the settings for section 2
  - "Mid2" shows the settings for section 3
  - "High" shows the settings for section 4
  - "Arbitrator" switches to the Phase Arbitrator settings page
4. **Band Enable.** This section turns the individual bands and their graphs on and off.
  - "Low" turns on section 1
  - "Mid1" turns on section 2
  - "Mid2" turns on section 3
  - "High" turns on section 4
5. **High Pass Filters.** Each High Pass section of the Allocator has one 6dB/oct filter and three 12dB/oct filters. Each of the four filters has its own on/off switch called "active". The switches change colors to indicate when they are on or off.
  - "Gang Freqs". This switch when in "active" state gangs together the frequency values of all filters in the section. This is helpful for sliding a higher order filter (consisting of more than just one 6dB/oct or 12dB/oct filter) in frequency.
  - "Load". This switch opens a standard dialog box where the user can load a predetermined filter response. Allocator comes with Butterworth and Linkwitz-Reiley filter presets up to 7<sup>th</sup> order (42dB/oct) already stored in the Presets folder.
  - "Save". This switch opens a standard dialog box where the user can save favorite High Pass filter responses.
  - "Reset" turns off all the individual filters and returns the frequency and Q settings to a "neutral", factory setting.
  - Sliders. 6dB/oct filter has only the frequency slider; 12dB/oct filters have frequency and Q value sliders. **When dialing in values using the sliders hold down the Shift key for finer mouse resolution. Values can also be typed in directly in the individual displays.**
6. **Low Pass Filters.** Each Low Pass section of the Allocator has one 6dB/oct filter and three 12dB/oct filters. Each of the four filters has its own on/off switch called "active". The switches change colors to indicate when they are on or off.
  - "Gang Freqs". This switch when in "active" state gangs together the frequency values of all filters in the section. This is helpful for sliding a higher order filter (consisting of more than just one 6dB/oct or 12dB/oct filter) in frequency.
  - "Load". This switch opens a standard dialog box where the user can load a predetermined filter response. Allocator comes with Butterworth and Linkwitz-Reiley filter presets up to 7<sup>th</sup> order (42dB/oct) already stored in the Presets folder.
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- Sliders. 6dB/oct filter has only the frequency slider; 12dB/oct filters have frequency and Q value sliders. **When dialing in values using the sliders hold down the Shift key for finer mouse resolution. Values can also be typed in directly in the individual displays.**
7. **EQ section.** This is the equalizer section for each crossover leg. It consists of 6 fully parametric equalizer bands- one low shelf, one high shelf and four peaking filters. Each band has a gain range of +/-20dB and frequency range of 20Hz to half the sampling frequency Allocator is processing at. So for 44.1kHz sampling rate the top frequency value is 22,050Hz
    - "On/Off". This button enables and disables the whole EQ section.
    - "12dB/6dB" These two buttons indicate the two available modes of the shelving filters. 12dB slopes are shorter and steeper (with the Q slider adjusting the steepness) and appropriate for the Baffle Step compensation, 6dB slopes are more gradual and appropriate for dipole woofer equalization. Either mode can be appropriate for the High Shelf- depending on the application.
    - "Load". This switch opens a standard dialog box where the user can load a predetermined EQ response.
    - "Save". This switch opens a standard dialog box where the user can save favorite EQ settings.
    - "Reset" returns all frequency and Q settings to a "neutral", factory setting. **When dialing in values using the sliders hold down the Shift key for finer mouse resolution. Values can also be typed in directly in the individual displays.**
  8. **Output Section.** Each Allocator band has an output gain slider (0dB to -inf. range), delay setting (0msec. – 50msec. range) and polarity switch. **See the note about delay settings in the Tutorial Section (p. 22-23).**
  9. **Meter Section.** Each band has two bar graph meters- one for left and one for right channel. Above each meter is a clip indicator that lights up when the output levels exceed 0dBFS.
    - "Reset" buttons turn off the clip indicators for a given pair of meters.
  10. **Graphing area.** This area displays frequency response data for all legs of the Allocator as well as the Arbitrator's filter pairs.
  11. **Load FRD.** This section lets the user import **F**requency **R**esponse **D**ata files that are then merged with (actually multiplied by) the corresponding Allocator crossover leg response. This lets the user visually tweak the final **acoustic** response of the transducer served by the given crossover filter.
    - "Low" This switch opens a standard dialog box where the user can load an frd file of the transducer served by the "Low" band.
    - "Mid1" This switch opens a standard dialog box where the user can load an frd file of the transducer served by the "Mid1" band.
    - "Mid2" This switch opens a standard dialog box where the user can load an frd file of the transducer served by the "Mid2" band.
    - "High" This switch opens a standard dialog box where the user can load an frd file of the transducer served by the "High" band.
    - "Reset" buttons remove the frd data from the corresponding bands and show the electrical response of the filters only.

- "frd offset" This window lets the user type in an offset in dB's that might be needed depending on how the frd files were acquired. If they indicate the absolute transducer sensitivity (usually in the 80dB-100dB/2.83V/1m range), the offset will have to be applied so that the maximum values fall in the 0dB range (Allocator graphs are set to display +12dB to -60dB range at the most). If the frd files were acquired using a non absolute level system they usually fall in the 0dB to -90dB range to begin with and will not require an offset to display properly. **It is important that all frd files used in a design come from the same source- preferably from the same measurement session as to eliminate relative sensitivity errors. Frd files should at the very least be "normalized" to reflect the same driving voltage. If the relative sensitivity is not properly accounted for, the graphs displayed by the Allocator will not reflect the actual frequency response of the loudspeaker. Garbage in- garbage out principal applies here.**
12. **Show Section.** Switches in this section control what is displayed and how.
- "Sum" view button turns on and off the summed frequency response graph
  - "Bands" view button turns on and off individual Allocator legs response graphs
  - "Arbitrator" view button turns on and off the Phase Arbitrator's filters display.
  - "24dB" button switches the range of the display to +12dB to -24dB.
  - "42dB" button switches the range of the display to +12dB to -42dB.
  - "60dB" button switches the range of the display to +12dB to -60dB.
- These three "zoom" ranges make it easy to tweak different parts of the design



## Phase Arbitrator Screen



### 13. View/Edit section.

- "Allocator" button switches back to the Frequency Allocator view. The "low", "mid1", "mid2" and "high" buttons are actually a part of the "Allocator" button. Clicking on any them takes you to the Allocator page. **It does not** automatically select the clicked band. You will have to select the desired band once in the Allocator view (unless of course it's the one you worked on last- it will still be selected).

### 14. Input and output meters.

These meters indicate the levels at the input of the sound card and the levels leaving the Phase Arbitrator that are being fed to the Allocator for filtering.

- Peak Indicators give numerical display of the signal strength
- "reset" buttons turn off the clip indicator "LED's" and also reset the Peak Indicator numerical values

15. **Preset Save, Recall and Reset.** These buttons allow for storing and recalling of the settings in the Arbitrator section. The reset button returns the settings to a neutral "factory" position.
16. **Box filter.** This control is dedicated to correction of the box roll-off induced phase error.
- The type button lets the user select "Sealed", Vented" and "BP/PR" (Band Pass/ Passive Radiator) for box type. These correspond to a 2<sup>nd</sup>, 4<sup>th</sup>, and 6<sup>th</sup> order crossover filter.
  - "-6dB frequency" setting is for dialing in the -6dB point of the loudspeaker low frequency response. Many manufacturers list -3dB point in their spec. Dial in the listed -3dB frequency multiplied by 0.64 for sealed, 0.8 for vented and 0.88 for band-pass box to arrive at the -6dB setting.
  - "Active" button enables the filter.
17. **Crossover filters.** These four identical phase correction filters let the user dial in the settings for up to 5-way speakers. In most instances as few as one or two will be used.
- \*In order for the correction to work properly the Arbitrator settings should closely reflect the **acoustic response** of the loudspeaker's drivers. When the View Arbitrator button is on, these filter curves also serve as guides for optimizing of the frequency response of the Allocator section. This approach works best when the frd files are imported and the Allocator graphs display the acoustic response of the speaker drivers.
- "Crossover order" setting lets the user dial in phase correction for crossovers up to 8<sup>th</sup> order (48db/octave)
  - "Crossover frequency" is adjustable from 20Hz to half of the sampling frequency used (for 44.1kHz sampling the top range is 22.05kHz, for 96kHz sampling the top range is 48kHz)
  - "Active" buttons enable individual filter sections.
  - "Output Gain" slider lets the user attenuate the signal leaving the Arbitrator section. It can be used as a Master Volume for the whole crossover.
  - "Auto" button. This control is there as a measure for avoiding digital "overs" or clipping. Because the Arbitrator changes the phase relationships of the audio material it processes, the peak levels of the wave forms are affected. In some cases as much as 6dB to 9dB of overshoot is possible. The "auto" button when enabled compares the input and output levels for 2 seconds and applies a corresponding attenuation on the output. This control doesn't necessarily prevent all "overs" because they are dependent on the harmonic content of the audio being processed. Attention should be paid to the peak indicators in both sections of the Frequency Allocator program.

## VSTShell Settings

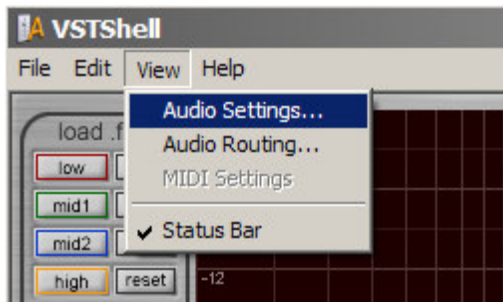
### Important!

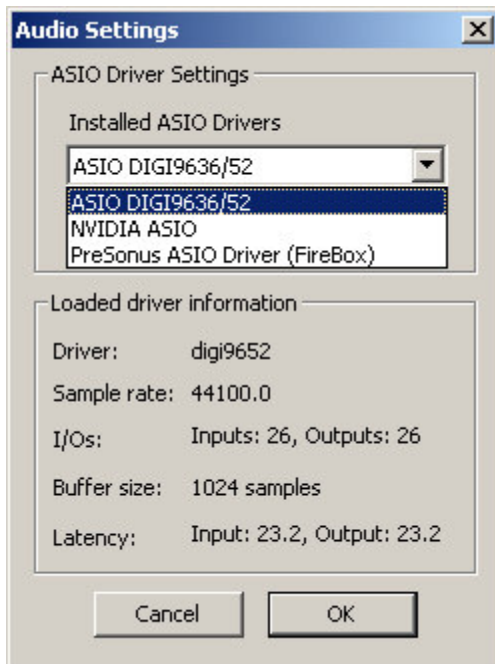
**Do not make live connections to your loudspeakers before dialing in the Allocator settings! Keep the amplifiers turned off and master volume controls down until you are 100% sure of your settings and crossover band assignments. I will not accept any responsibility for damaged transducers. It's a good idea to have a full range speaker with a quick disconnect for checking signals at the outputs of the amplifiers before connecting actual tweeters and sensitive midrange transducers to the amplifiers.**

With the warnings out of the way, let's first look at the way the VSTShell works. After installation, open the folder where you unzipped all the files and double-click on the Allocator.exe icon. (You might want to create a shortcut to it on your desktop for easy access later)

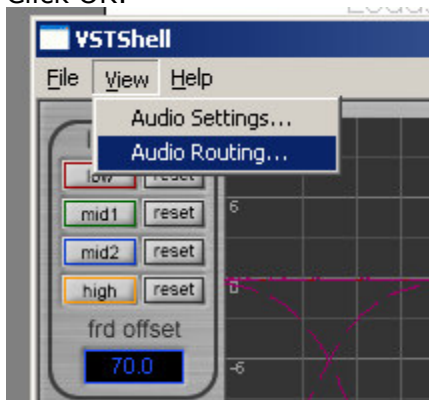
You will initially be presented with a registration screen. Enter your serial number in the fields provided and click "Apply". Click on the "Edit" button to go to the main Allocator page.

In the View menu select "Audio Settings"

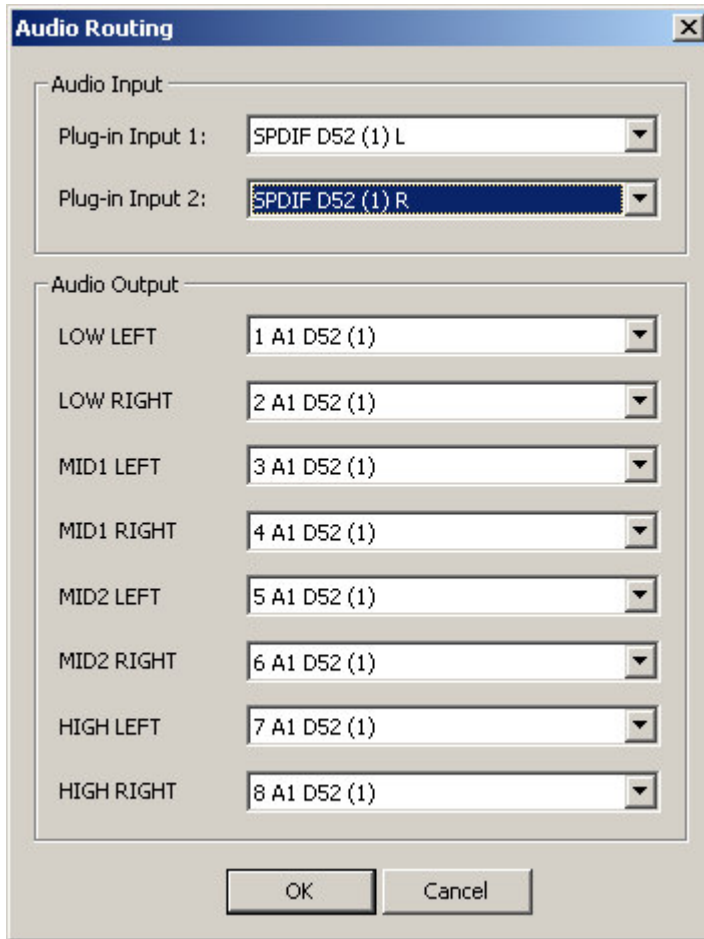




If you have more than one ASIO card in your system you will be able to select the one you want to use with Allocator here. Select the desired sample rate. In most systems you will leave the Minimal buffer size box unchecked. Click OK.



Go to the View> Audio Routing menu.



In the drop boxes assign your ASIO card's inputs and outputs to the Allocator inputs and outputs. Click OK.

Some ASIO cards will at first generate error messages in the VSTShell. Please ensure that the sound card's ASIO driver is set correctly. The sampling frequency, synchronization, routing and possibly other proprietary functions can at first confuse the VSTShell. When that happens restart the VSTShell, go through the Audio Settings setup once again and confirm the selections by clicking OK. If the settings are valid, one or two restarts usually makes everything work as expected.

Things to look out for:

- Is another application accessing/controlling the card at the same time?
- Is the card set to external sync, but no sync signal is present?
- Is the card set to operate at one sampling frequency but the VSTShell is set to a different frequency?

If you are not using the VSTShell as the host, refer to your particular program's manual for the procedure of assigning inputs and outputs.

The Allocator's logical outputs follow the low, mid1, mid2 and high assignment for outputs 1L+R through 4L+R.

## Tutorial

### The Phase Arbitrator Operation

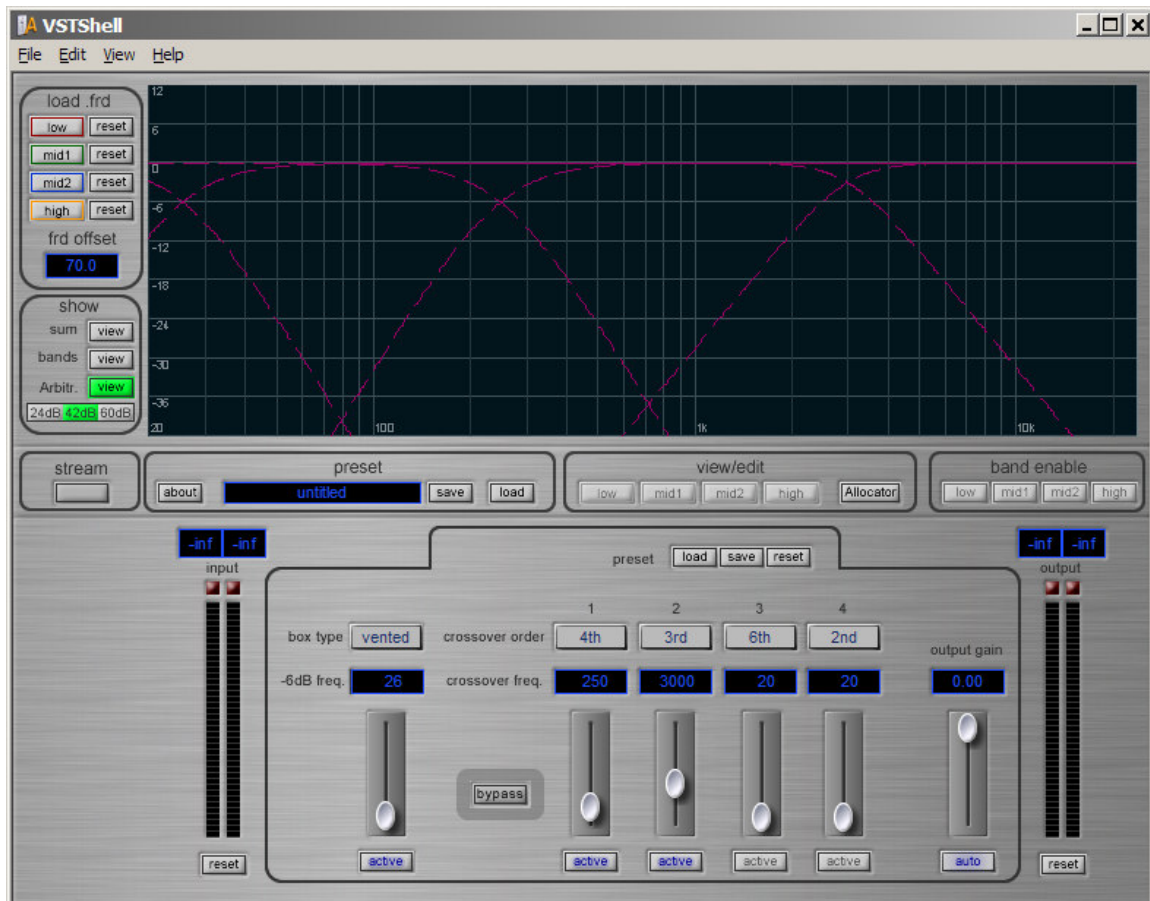
Now you are ready to start designing your crossovers. We will start with the Arbitrator settings.

If you are not there yet, click on the button called "Arbitrator" in the View area of the interface.

**\* An important note. When referring to loudspeaker crossovers I mean the final acoustic response, which is a combination of the electrical (or DSP in our case) filter and the natural response of the loudspeaker driver. Very seldom the electrical filter response corresponds to the final acoustic response of the loudspeaker. For that to happen the loudspeaker driver needs to have a very flat frequency response well over one octave passed the electrical filter's roll off point. If the electrical filter roll off point is close to the natural acoustic roll off point of the driver, the two interact resulting in final acoustic slopes that are steeper than the electrical response alone. For best results you should have your loudspeaker drivers' frequency and phase response measurements available as frd files.**

When building your crossover from scratch decide here first what your loudspeaker's design will be. Is it a 2-way, 3-way or maybe a 3-way with a subwoofer? Is it supposed to be a 3<sup>rd</sup> order crossover at 2000Hz or maybe a 6<sup>th</sup> order at 2500Hz? You will have to use your best judgment based on your driver selection and type of loudspeaker you are building. Nothing prevents you from storing a few global presets and switching back and forth between them to compare the sound of 3<sup>rd</sup> order filter versus 4<sup>th</sup> or 6<sup>th</sup> order at the same or different frequencies. As you work with the different crossovers your experience will grow and lead you to better designs down the road.

So, with the Arbitrator view on, in the "show" section click on Arbitrator "view" button to view the Phase Arbitrator filters. Enable one band per crossover point. In the screen shot below I decided to make a 3-way loudspeaker with the woofer crossing to the midrange at 250Hz with 4<sup>th</sup> order filters and the midrange crossing to the tweeter at 3000Hz with 3<sup>rd</sup> order filters.

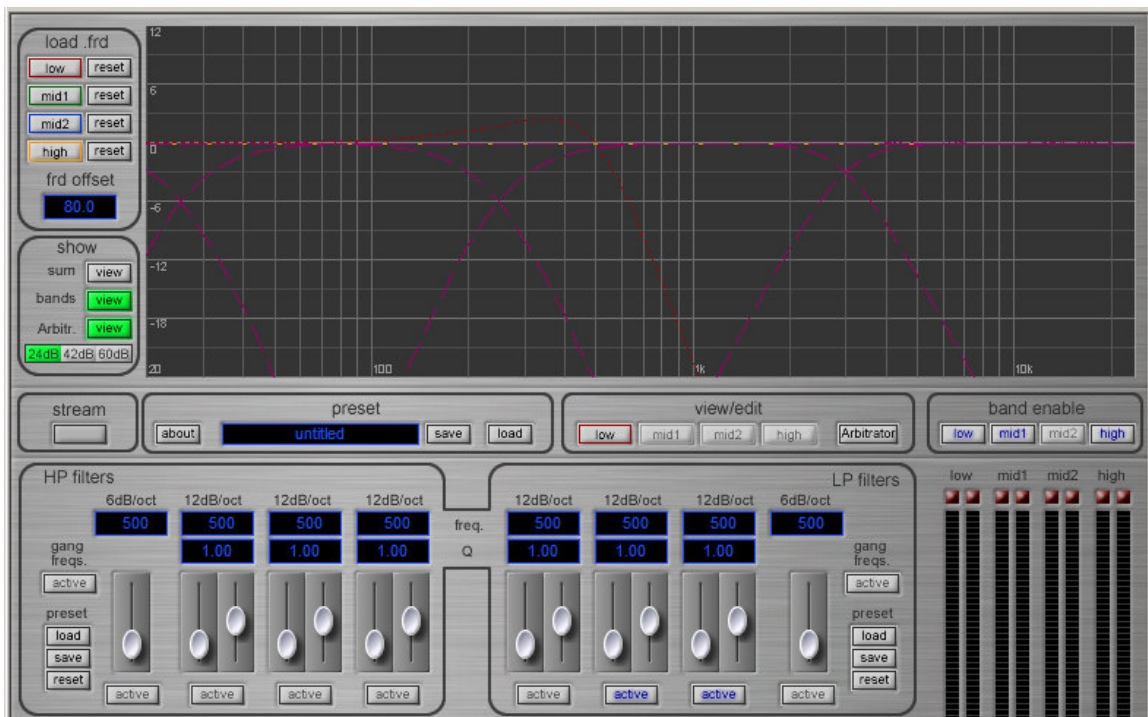


With that first decision made it's time to switch to the Frequency Allocator and start dialing in a crossover.

## The Frequency Allocator Operation

If you don't have measurements for your drivers you are left with your ears, driver spec sheets, intuition and guessing skills. You will want to start with Allocator settings that closely reflect the Arbitrator settings. So, for our above laid out speaker you would enable three crossover legs- low, mid1 and high.

- Then, first switch the View/Edit section to Low.
- Enable two Low Pass 12dB/oct filters by clicking on the "active" buttons below their sliders. We decided on 4<sup>th</sup> order crossover, so it would seem appropriate to use two 12dB/oct filters to arrive at that goal (side note: each "filter order" is defined as 6dB/oct attenuation).
- In the "Show" section enable the "bands" view button (it's supposed be green). This will show a red "low" filter response trace. (see the graph below)



- Start by adjusting the frequency sliders of the active filters toward 250Hz. You can move them simultaneously by activating the "gang freqs." button. Hold the Shift button for finer resolution of the sliders. The response will not exactly follow the Arbitrator curve until you adjust the Q settings of the filters. To arrive at a textbook Linkwitz-Reiley alignment, set both Q sliders to 0.71. This should make the curves agree very closely.
- Of course you could just load a LR4 preset by clicking on the "load" button in the LP section and pointing to the Ir4.lhp file in the LP Filters folder and then typing in 250 in the frequency display of any filter. I just wanted you to get an idea of manually tweaking filters first.
- Switch the view/edit section to mid1 and repeat the procedure for mid1 band. In mid1 band you will want to dial in the High Pass section to LR4 at 250Hz

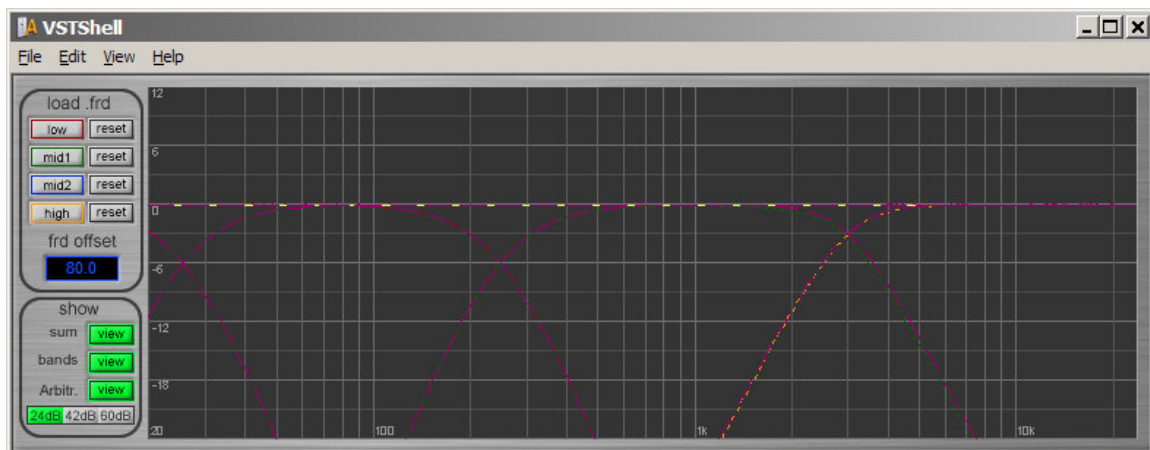


and the Low Pass section to B3 (Butterworth 3<sup>rd</sup> order) at 3000Hz. You can again do it manually, or just recall the presets.

- Switch to "high" in the view/edit section and dial in a B3 High Pass filter at 3000Hz. This should bring all the individual **electrical** responses to our goal. But, if you enable the "show sum" button the combined response will not be flat. It will have a slight bump at 3000Hz. This is due to a phase shift of the mid1 band related to the High Pass filter. See below:



So what do we do? We can use the EQ section of mid1 and high legs to straighten out the combined response. I use the shelving sections that are tuned in well below and well above the pass bands of the mid1 and high sections to manipulate the phase response of the leg without affecting the frequency response in the pass band. For example, the high leg gets a low shelving filter at 215Hz with a cut of 15.5dB and the mid1 band gets a high shelf at 12500Hz with a cut of 2.5dB. This brings the combined frequency response back to flat while closely preserving the individual frequency responses of the filters inside the all important first 24 dB window.



How does it affect the Arbitrator/Allocator interaction? Very little. The deviation of the final phase response from perfect in this example is less than 5 degrees.

Now it's time to listen to our first design. Save your work as a main preset with a name like "My speaker 1" and turn on the Stream button.

- At this time your amps should still be off.
- In the Arbitrator view turn the output down to at least -40dB.
- Apply a signal to the selected input. Pink noise or music is OK.
- Keep your individual drivers **disconnected** from the amplifier for now.
- Connect a "sacrificial" full range speaker that can take some power (at least 30W) to the output of the amplifier that is driven by the left low output.
- Turn on the amp. Listen to the speaker to ensure that indeed the low frequencies are coming out of the output. If the signal is very weak, turn up the Arbitrator until you can hear the signal. If everything is as expected switch the full range speaker to the right low output.
- Repeat for the mid1 and high outputs.
- Only when absolutely sure that correct frequencies at safe levels are coming out of the outputs, connect your expensive drivers to the amps.

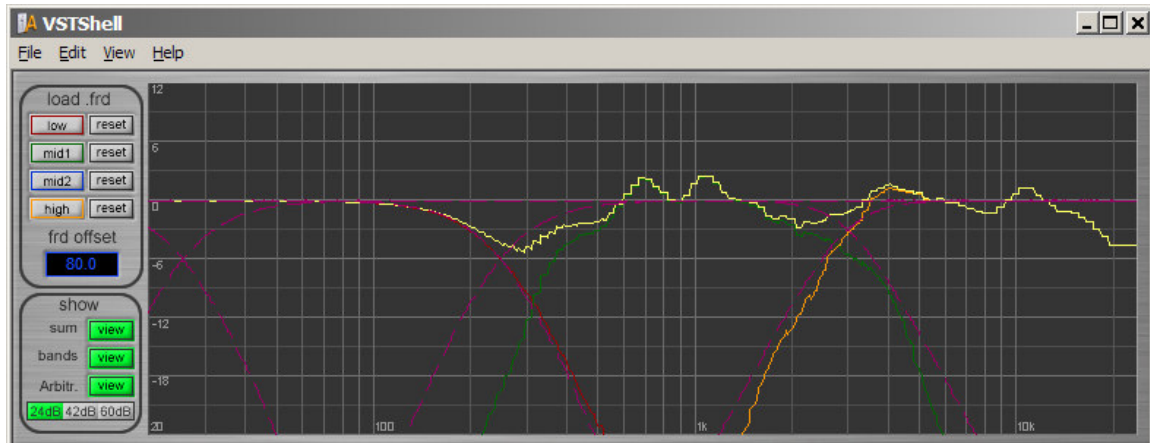
At this time your speaker will most likely sound rather wrong. This is foremost due to a sensitivity mismatch. Tweeters and midranges are usually more sensitive- play louder with similar signal applied- than woofers. You should either adjust your amps (if you have individual volume controls per channel or pair of channels) or attenuate the individual outputs in the Allocator. If you have access to the spec sheets of your drivers look at the sensitivity listing. If the tweeter is listed as 91dB/1m/2.83V and the woofer is listed as 86dB/1m/2.83V you should start by attenuating the tweeter by 5dB (91-86). Same for the midrange driver. It could be listed as high as 96dB/1m/2.83V. You would have to attenuate it by 10dB to bring it in the same range as the woofer. This will ruin your pretty graph, but there is no way around it- unless you have actual frequency response graphs for your drivers **that are normalized to the same stimulating signal strength**.

This brings us to the fun part of laying out a crossover using imported frd files.

Let's see how importing real world measurements into our first design affect the simulation. By pressing on individual bands buttons in the "load frd" section we can point to real world measurements that include absolute sensitivity, frequency and phase response information about our drivers. Here are 3 frd files placed in the design. These came from examples used in a program called lspCAD (highly recommended loudspeaker design tool). I adjusted the frd offset to 80dB to bring the curves fully in the window



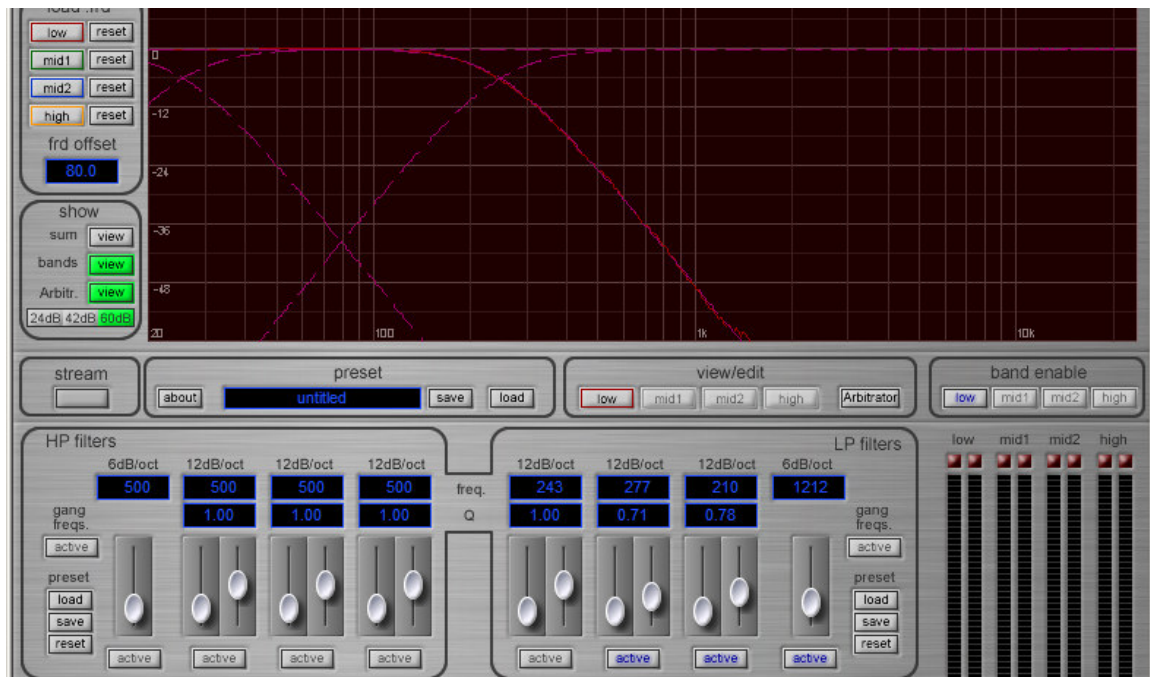
As you can see this textbook crossover is not going to work well with real world drivers. Even when the differences in sensitivity are accounted for, the combined frequency response still looks rather bad and the acoustic curves are way off the Arbitrator target.



Let's try to adjust our DSP filters so that the final acoustic response follows the Arbitrator curves closely. We start with the low band.

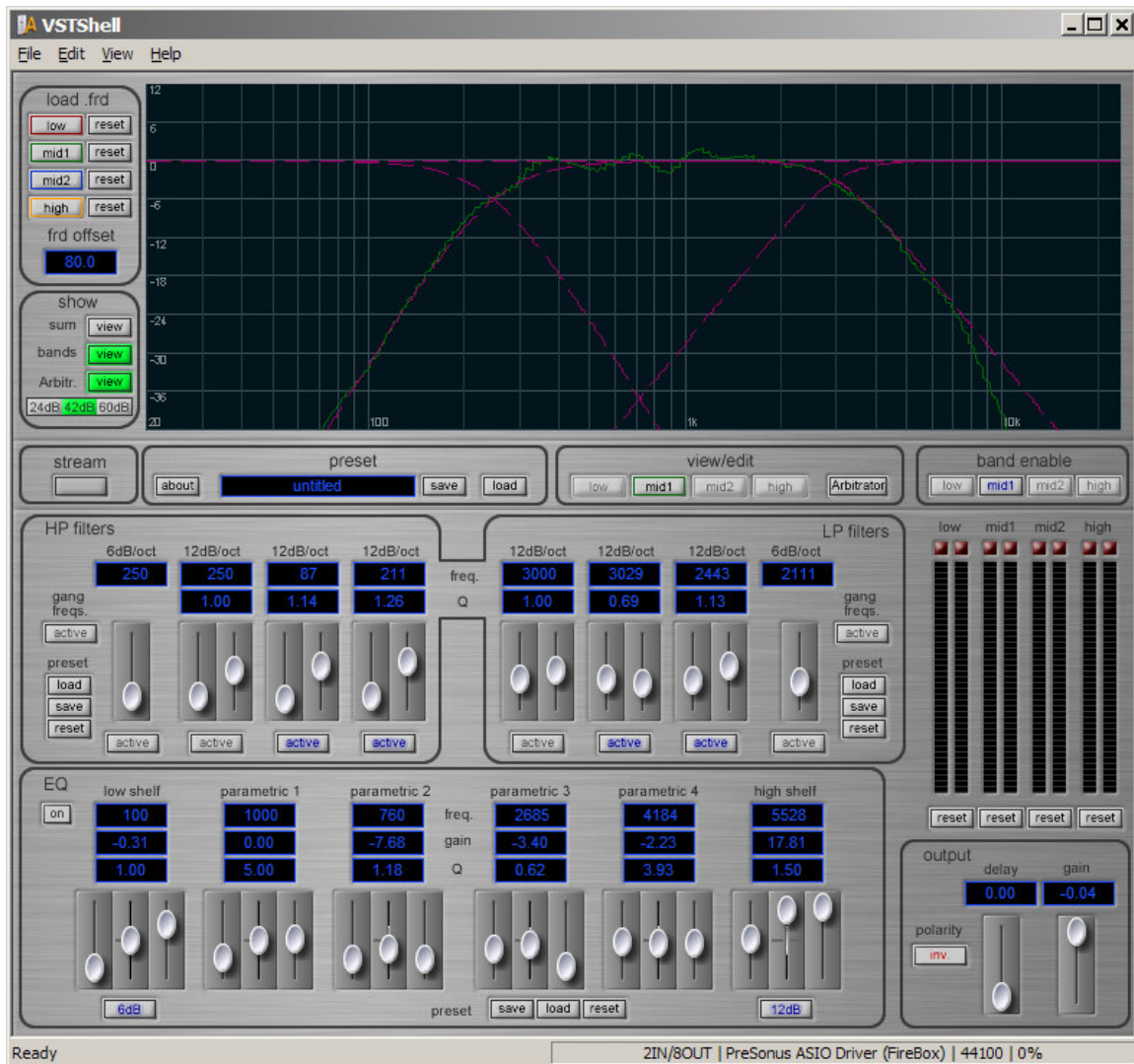
Turn off the mid1 and high bands in the "band enable" section so that we can concentrate on one part of the graph at a time.

- First, let's forget about the textbook presets and un-gang the frequencies
- We tweak the LP filters until a good agreement between the red low curve and the Arbitrator curve is found. Make sure you use all three views 24dB, 42dB and 60dB as they expose different errors. With a little tweaking of frequencies and Q values as well as an addition of a 6dB filter and one parametric filter we arrive at this setting:



That's much better. Let's tweak the midrange.

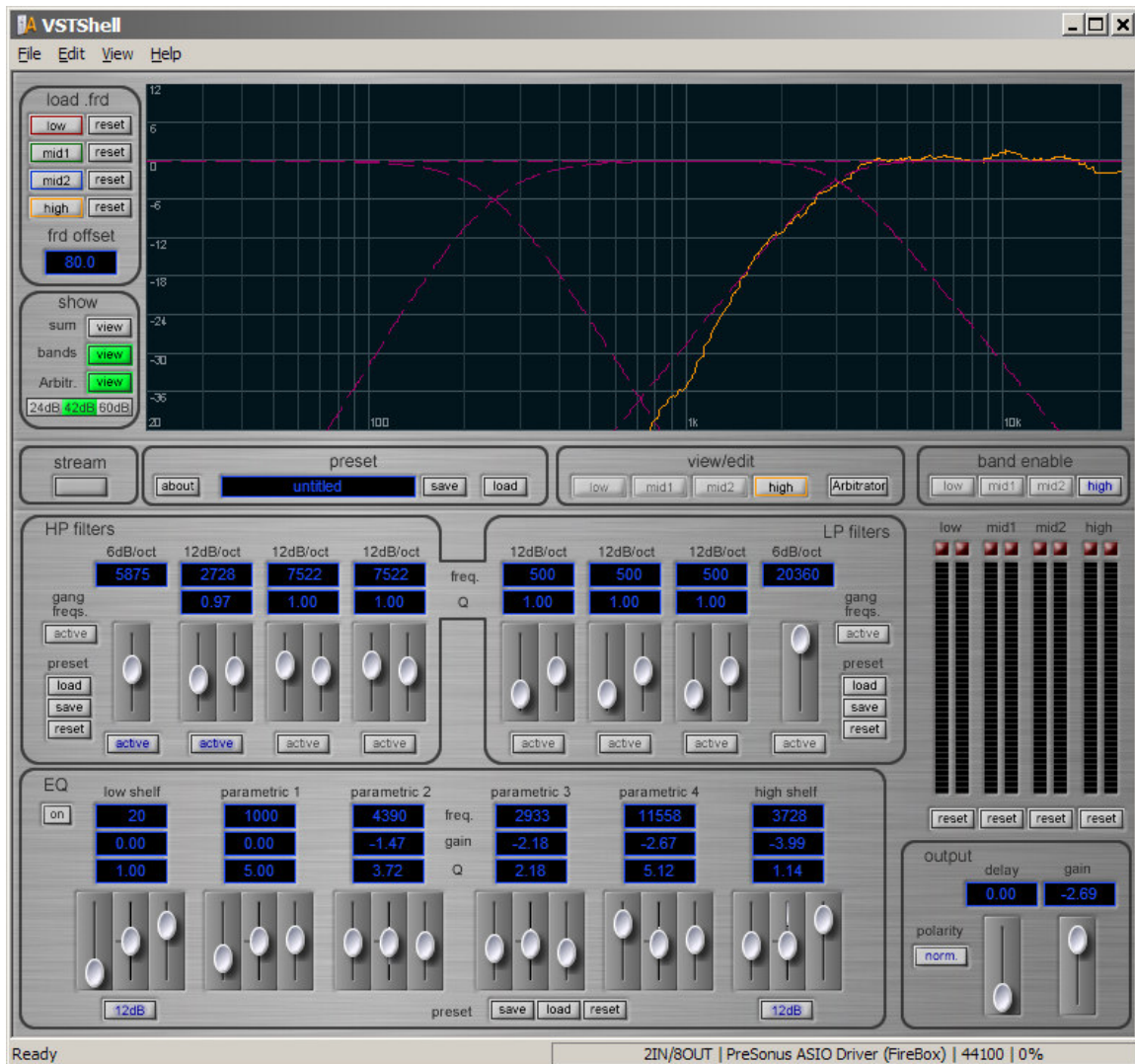
- Turn on the mid1 band and switch to it in the view/edit section. You can keep or turn off the low band for now. After a few minutes of tweaking we arrive at this response:



Notice that it took quite a bit of high shelf boost to arrive at the goal of 3<sup>rd</sup> order LP at 3000Hz. This tells us that the midrange driver is most likely not well suited for that particular design. Maybe going with steeper slopes or lower frequency would be a better choice here.



Next we tweak the high band:



This tweeter might not be a good candidate for third order acoustic slope either. It follows the curve down to -24dB, but then it wants to roll off much faster. Applying a boost at low frequencies could prove counterproductive as it might lead to unnecessary stress of the tweeter. But for the sake of the discussion let's leave the filter the way it is. The summed response looks fairly good. We should save this design as a preset.



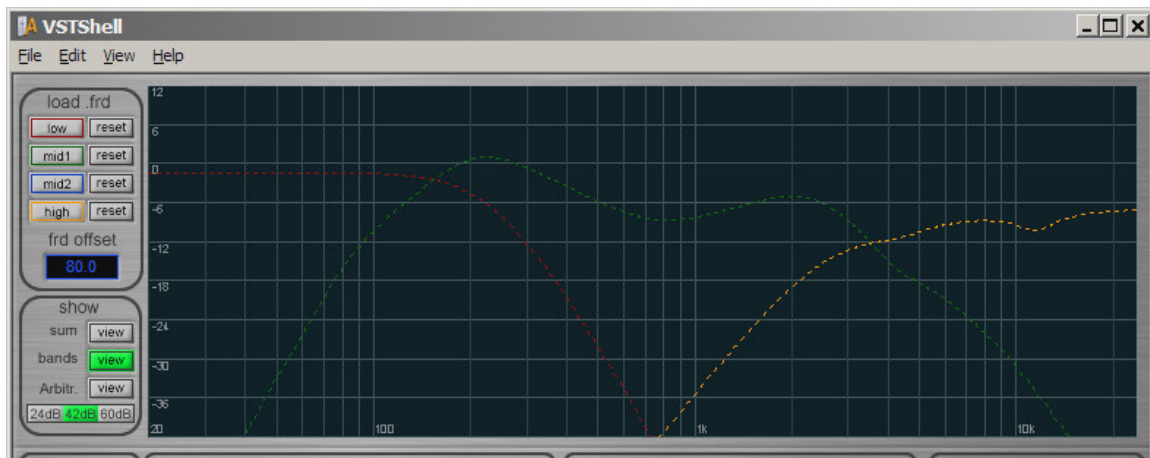
Click on the "save" button in the preset area and give it a name like "My speaker 2".

**A tip: The frd files if used in the design are saved with the presets. You can email them to friends who can help you with your crossovers.**

Listening to it should give you clues as what might need to change. But if you like what you hear at this stage, you could be set. Incorporating a bit of baffle step compensation by turning the lows up by 2dB-4dB could prove beneficial.

Let's take a look at the electrical curves that yield this acoustic response.

We click on the "reset" buttons next to the load frd buttons to remove the frd files from the graphs.



Do you think you would come up with these curves by just listening to your speaker and eyeballing the electrical response graphs? Well, that's what it takes to make a good measuring speaker given a set of drivers.

**\*A note about the delay settings.** Computations used for calculating the graphs are performed using complex numbers- the phase of the filters and the phase from measurements are taken in account. **The phase related to absolute delay setting is not used in the computations and therefore not reflected in the displayed graphs.** The computations assume equal distance from each transducer to the listener. If you are designing a loudspeaker for a particular point in space, measure

or compute the distances from the drivers to the design point and dial in the appropriate delay to compensate for differences in the time of arrival. For example, if your tweeter is 2 meters away from your ears but the woofer is 2.5 meters from your ears, add 0.5m worth of delay to the tweeter, which corresponds to 1.45msec to bring things into alignment.

Use the following numbers when calculating delay:

1" = 0.074msec. So the smallest available delay- 0.01msec corresponds to 0.1356" or 0.344cm.

Another way to design a crossover, while taking in account driver positioning in 3D space is to use a dedicated loudspeaker/crossover design software package such as lspCAD ([www.ijdata.com](http://www.ijdata.com)) or SoundEasy (<http://www.interdomain.net.au/~bodzio/>). Both of them have the capability of simulating all of the functions of the Frequency Allocator. Both of them have powerful optimizers that will fit the filter parameters to predetermined goals. Settings derived in those programs would have to be manually typed into the Allocator.

There are also free tools available for download from the FRD Consortium (<http://www.pvconsultants.com/audio/frdgroup.htm>) that allow for acquisition and manipulation of frd files as well as various loudspeaker related simulations. Most of them require Microsoft Excel to work.

## Frequently Asked Questions

**Q: How do I handle a mono subwoofer?**

A: In VSTShell assign both low outputs to the same physical ASIO output. This will perform the mono summing and adjust the output level by 6dB to prevent digital "overs".

**Q: Can I use the Allocator as a "live music" loudspeaker processor?**

A: The Allocator has a hard coded latency of 8192 samples. This corresponds to about 175msec at 44.1 kHz sampling frequency. So, the short answer is No. The Frequency Allocator is designed to work as a playback loudspeaker processor.

**Q: Why such a big latency. Aren't you using IIR filters?**

A: Yes, the Frequency Allocator and the Phase Arbitrator use IIR processing, but the Arbitrator algorithm, which processes audio in reverse needs to accommodate fairly large group delays so that even steep filters at low frequencies can be corrected. To assure high fidelity, we decided to err on the side of caution. The Arbitrator will correct phase errors down to 10Hz.

**Q: Where can I get frd files for my drivers?**

A: a) You can "scan" them into your computer by using a program called SPL Trace available from FRD Consortium- if you have a spec sheet for your driver that shows its frequency response.

b) You can ask around on the boards- maybe someone has the measurements as frd's and is willing to share.

c) You can also download a free program called Speaker Workshop that has a measurement utility built in. Also, lspCAD and SoundEasy (commercial speaker design packages) come with measurement programs.

d) buy a dedicated measurement system (an option for the serious hobbyists and audio professionals)

**Q: When I start the Allocator I hear a lot of clicks and "stutter". How do I get rid of it?**

A: As mentioned above, the Allocator processes audio in 8192 sample blocks. For most efficient operation the ASIO buffers should be set as close as possible to 8192. Some computers can handle the long blocks with shorter ASIO buffers. But a safe setting for a fast computer is at least 1024 samples.

**Q: I increased my ASIO buffers to max and I still hear clicks and pops. Also, the Task Manager reports very high CPU usage.**

A: There are a few system configurations (processor/chip set/ASIO sound card) that suffer from this problem. At this time (May 2006) we have no solution, but we are actively investigating this problem. Please visit our forum for up to date information regarding this.

**Q: I use a passive 2-way (3-way) loudspeaker with a subwoofer. I would like to use The Allocator as the crossover and the Arbitrator as the phase correction for the whole system. How do I set it up?**

A: You would use the low and high (or mid1, mid2) bands. You would set the Allocator high band to the frequency you want the sub to take over at (usually around 80-100 Hz) and dial in a HP slope that is somewhat shallower than the subwoofers LP slope (to compensate for the natural roll off of the small woofer which will augment the final filtered response). If the subwoofer has its own Low Pass filter, leave the Arbitrator's Low output unfiltered. But, if you can bypass the subwoofers filter, use the Allocator filters for better control. In the Arbitrator you would enable the box filter, the sub-satellite filter (80hz 4<sup>th</sup> order or similar) and one filter for each crossover point of your passive loudspeakers. It helps to know their **acoustic** crossover point(s) and type - is it a second order design, 3<sup>rd</sup>, 4<sup>th</sup>, etc? Try to match the Arbitrator to the crossovers of the satellite.

**Q: Where can I get more information about active crossovers?**

A: Please visit our forum at [www.thuneau.com/forum/](http://www.thuneau.com/forum/) or ask your questions at other loudspeaker design centric forums such as <http://www.madisound.com/cgi-bin/discuss.cgi>? and <http://www.diyaudio.com/>.