Loudspeaker Phase Arbitrator

User Guide

Jan Luszczek June 27, 2006

Introduction 2
Description 2
Background and theory of operation. 3
Controls 8
Operation 9
The Volume control and Auto button. 11

Introduction.

Almost all people enjoy music on one level or another. Some like music so much that they will go to great lengths to ensure that their listening pleasure is a pure as possible. A lot of money and effort is expanded to hear recordings with the absolutely greatest possible fidelity. There are products that claim to bring the performers right in the listeners' living rooms through magic of technology or skill of the equipment designer.. While electronics and various tweaks bring tiny perceived changes to the sound, the source of greatest errors in the playback chain- the loudspeaker remains the decisive factor in final reproduction quality. In order to maximize the loudspeaker's performance engineers traditionally result to using multiple transducers that are specialized in reproducing a particular range of frequencies. Large woofers excel at low frequencies, while small tweeters are great for the very high frequencies. The frequencies are distributed to the transducers using a network called crossover. Crossovers are typically built out of capacitors, coils and resistors, but some are made using semiconductors and work on small signal level that is then distributed to the loudspeakers using one amplifier channel per transducer. There are also crossovers that work on digitized signals inside computers or dedicated Digital Signal Processors (DSP for short). The most popular DSP crossovers simulate the workings of analog parts. All of those crossovers share a common characteristic. By filtering of frequencies they introduce an error in the time domain. Frequencies below the filter's cut off point lag behind the higher frequencies. The steeper the filtering action the more lag is introduced in the signal. While at higher frequencies the lag can be only on the order of 1 to 2 milliseconds, crossovers in the lower midrange and bass frequencies can introduce lags of as much as 10 milliseconds. An additional factor in loudspeaker's reproduction is the fact that the woofer in a box naturally rolls off low frequencies due to its limited efficiency at low frequencies. From physics point of view it is another high pass filter with its own phase error. A typical sealed box makes the low bass lag behind treble by 15 to 20 milliseconds, the most popular type of enclosure- so called bass reflex can easily introduce 30 or more milliseconds of lag to low frequencies. We all got used to this time smear and perceive the bass reproduction of the loudspeakers as "normal". While some scientists came up with ways to negate the phase error caused by crossover slopes, I'm not aware of anyone trying to fix the group delay of loudspeaker woofers in their boxes.
Enter the Loudspeaker Phase Arbitrator.

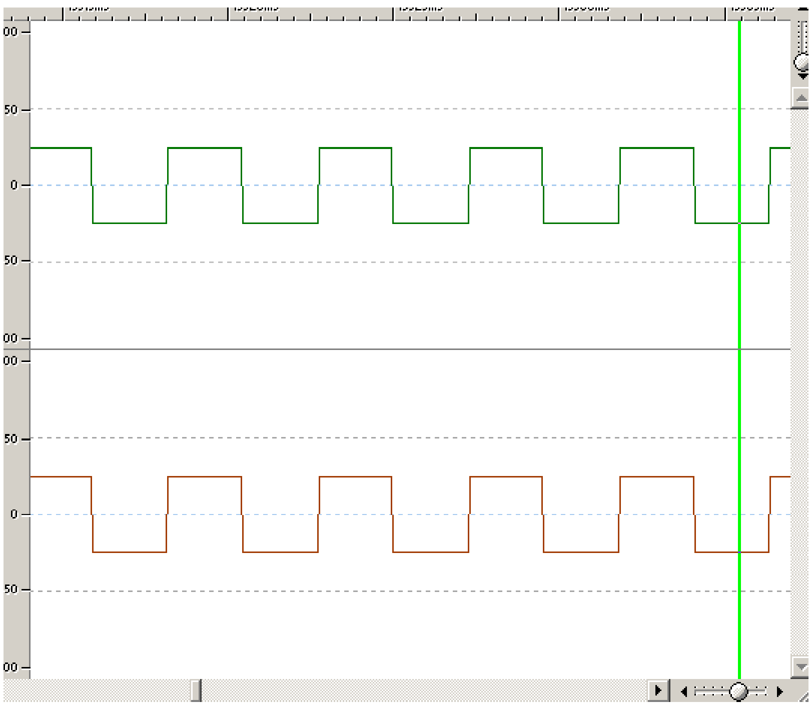
Description

The Loudspeaker Phase Arbitrator is a Windows program designed to process digital audio. Its main function is to induce in the audio stream a phase error that is complementary and opposite to the phase roll caused by a given loudspeaker's crossovers and natural frequency response roll-off on either end of the spectrum. The Arbitrator achieves the task by simulating textbook crossover filters (ones that sum flat in frequency domain) and passing audio through them in reverse. Audio stream coming out of the Arbitrator actually leads with low frequencies. The loudspeaker, as explained above, naturally delays the lower frequencies through its crossovers and woofer/box

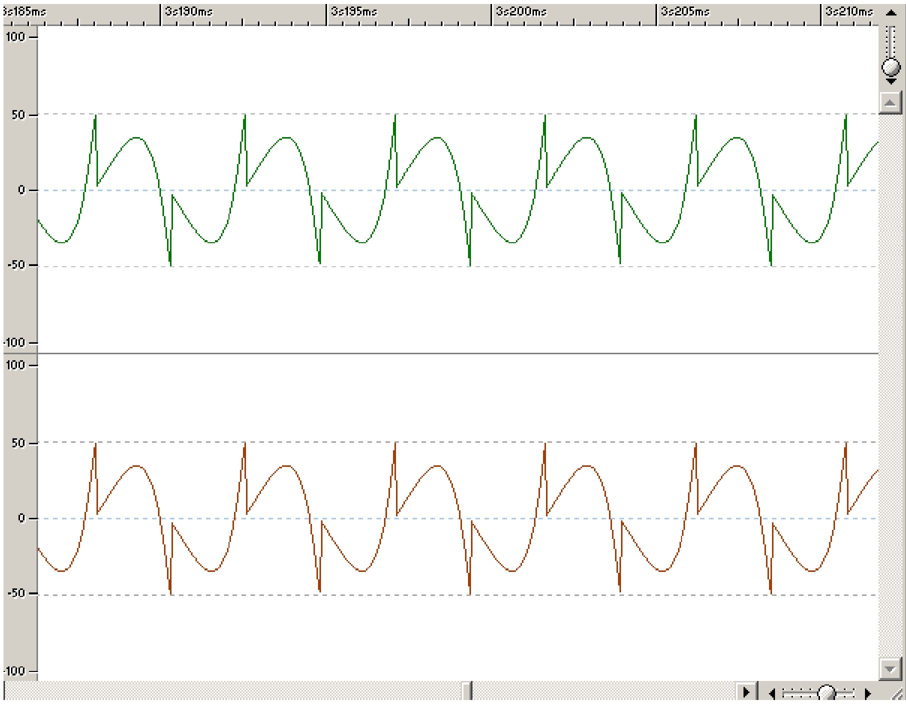
action. The net effect, if dialed in correctly, is nearly perfect timing of all frequencies in relation to each other.
As mentioned before, the Phase Arbitrator leaves the frequency domain untouched- therefore the final frequency response of the system is the same as before Arbitrator was introduced. Any characteristics that make a loudspeaker special, such as its frequency response, power response and polar patterns (how it radiates frequencies into the room) stay perfectly preserved. The only difference is that impulses are reproduced as such and not smeared in time.

Background and theory of operation.

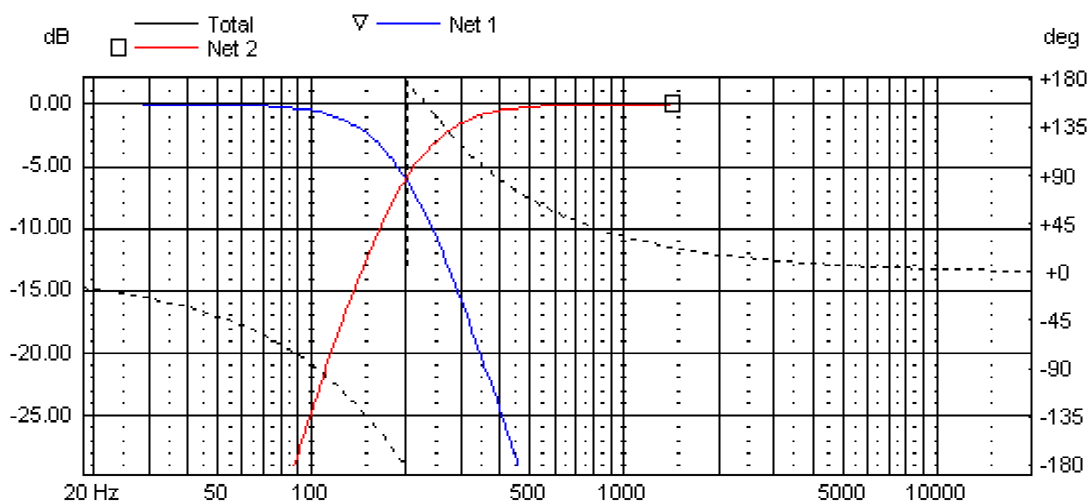
Let’s take a look at how a typical loudspeaker reproduces impulses and square waves- a good visual indicator of the loudspeaker’s performance in the time domain. No one actually listens to square waves and single clicks but the graphs are revealing and easily comprehensible.



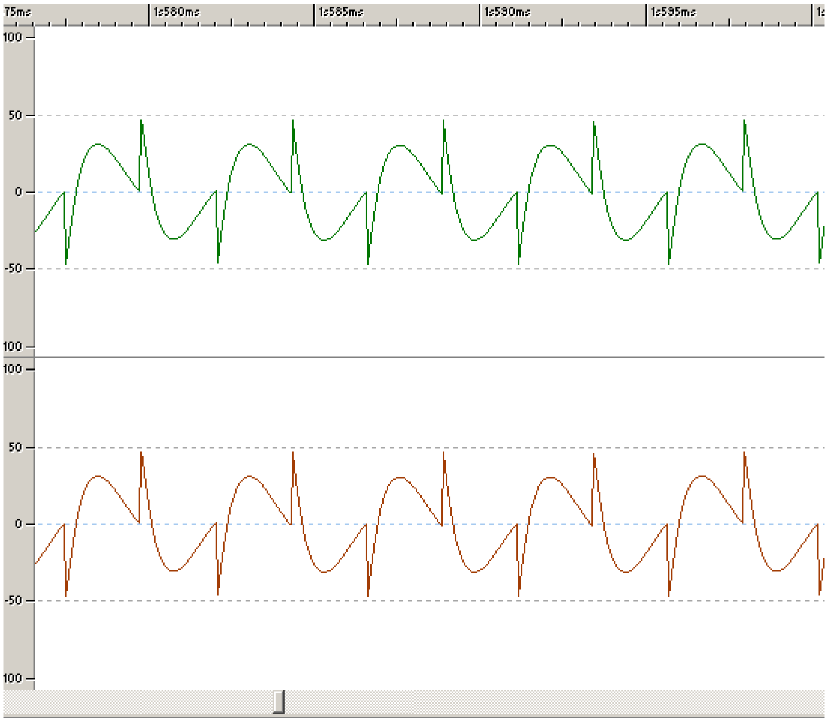
This is a perfect 220Hz square wave generated in a program called Wavelab.



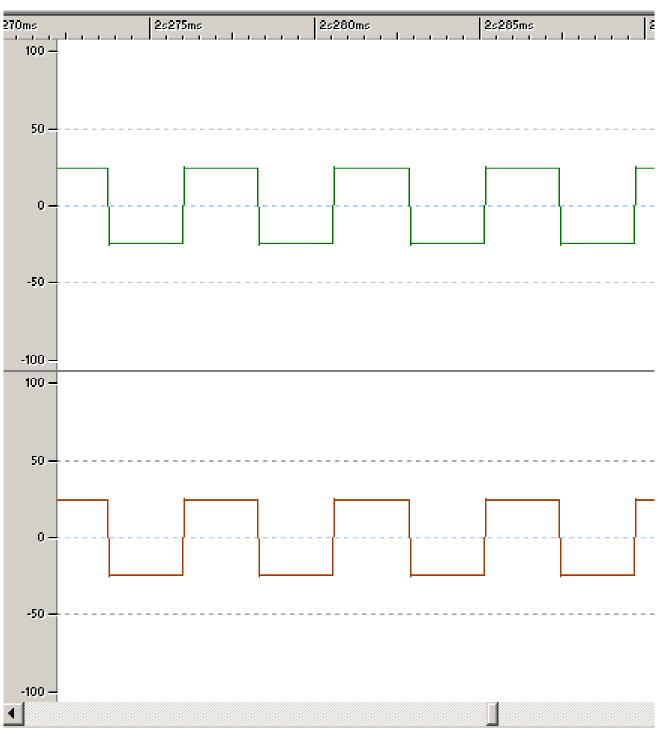
2. Here is the same wave processed by the Phase Arbitrator set to Linkwitz-Reiley 4th order crossover at 200Hz. The crossover's response looks like this:



When we process our square wave in IspCAD (filter function) to see what it will look like after it passes through the crossover we get this result:



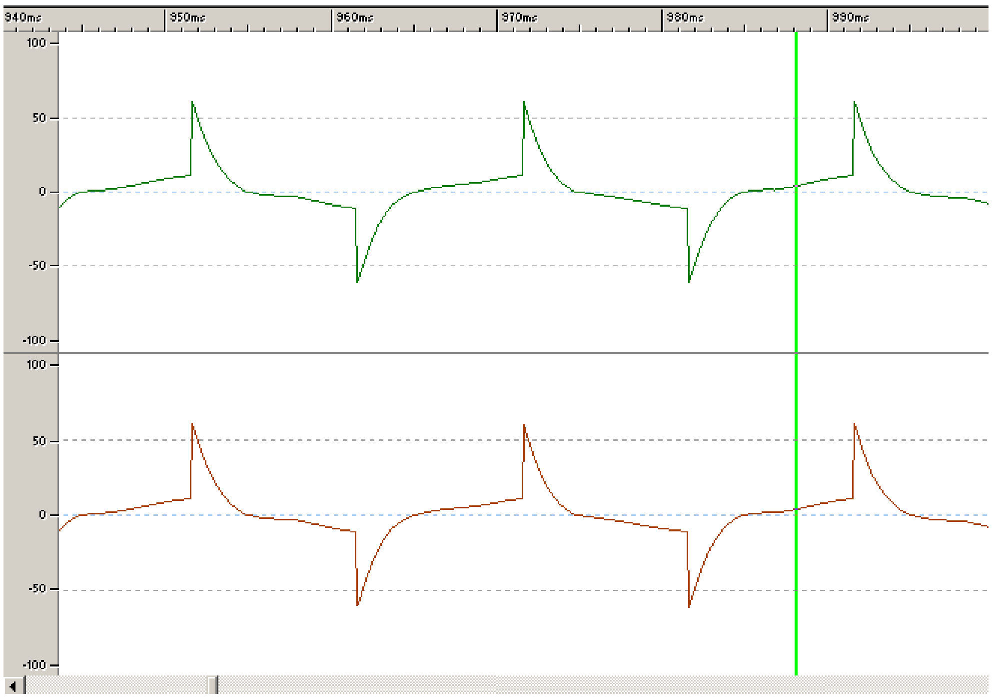
But, when we process in IspCAD the Arbitrator processed square wave from picture 2 we get this:



We are getting perfect square waves out of a speaker with LR-4 crossover by using the Phase Arbitrator! The process can be simply described as **preventive damage control**.

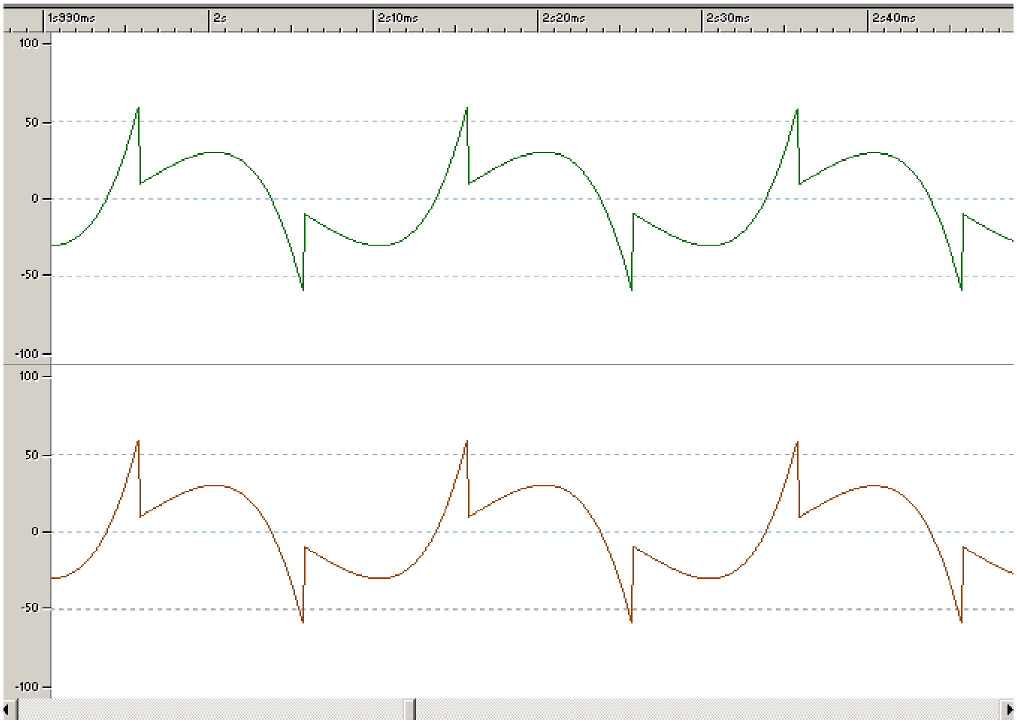
Let's take a look at what the box correction does to our waves.

Let's assume we have a vented box that rolls off at 40Hz with Q of 0.71 (nice Butterworth alignment).
I simulated such box in IspCAD and processed a 50Hz square wave through it.
Here is what the wave looks like:



Because the speaker has little power in low frequencies it cannot deliver the pressure to keep the wave flat. The attack of each half cycle is sharp, but the sustain part decays due to limited LF ability.

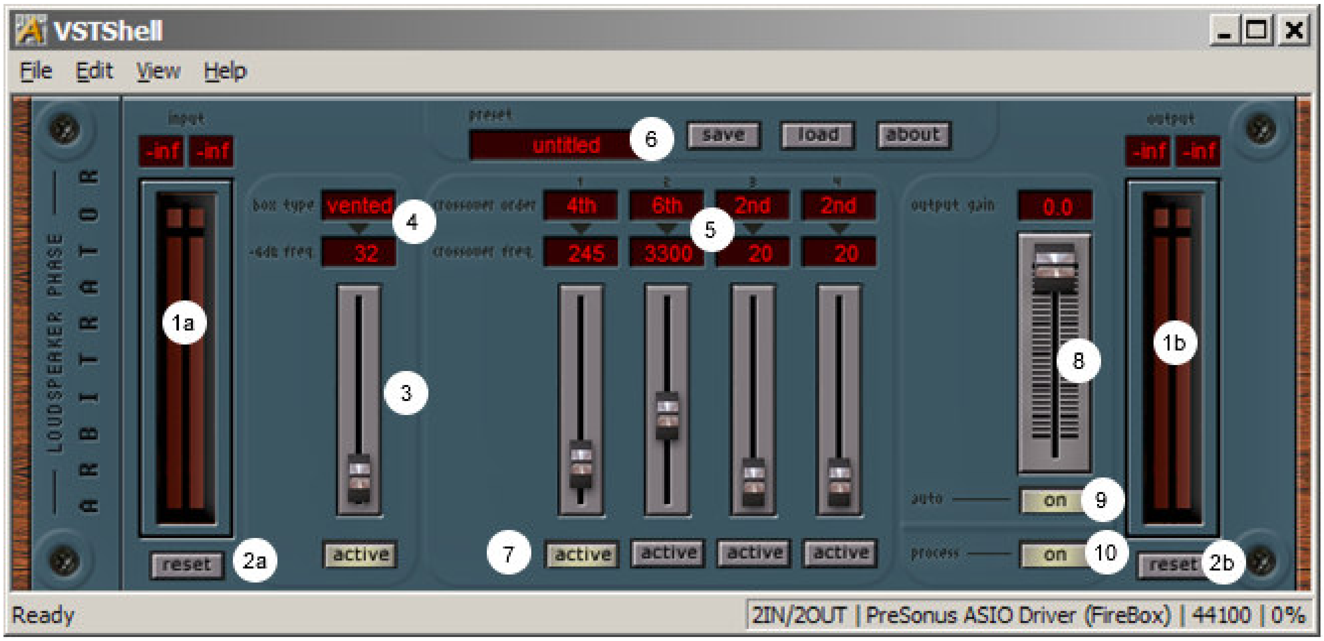
Let's pre-process a 50Hz square wave with Arbitrator set to vented box at 40Hz. Here is the wave:



Now let's see what happens to the box response when fed the output of Arbitrator:

The speaker still cannot pressurize the room, but the output resembles a square wave much better.
This aspect of the Arbitrator is very noticeable. The bass just feels tight and correct when dialed in to match the actual speaker used.

Controls

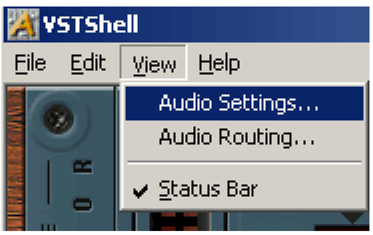


1a- Input meter with peak level indicators above it.
1b- Output meter with peak level indicators above it.
2a- Reset button for the peak indicators.
2b- Reset button for the peak indicators
3- Frequency control slider for the box correction section with a numerical readout directly above it.
4- Box type drop down control. The options are sealed, vented and Band Pass/Passive Radiator.
5- Individual crossover point controls. Each crossover point can vary in frequency from 20 to 24000hz and available crossover slopes range from 2nd order (12dB per octave) to 8th order (48dB/oct). Select them by clicking directly in the crossover order readout box.
6- Preset control section. You can save and recall settings for multiple loudspeakers here.
7- Crossover/box correction filters on-off buttons. Activate only crossover bands that your speaker actually has. Arbitrator allows for correction of systems up to 5-way (4 crossover point).
8- Output gain. Use this slider to control the volume on the output of the Arbitrator. In normal operation this level will be somewhere between -0dB and -6dB. More about it later.
9- Auto gain button. Enabling this button compares the input and output peak levels for 2 seconds and applies a corresponding gain reduction in the background. The output gain slider is not affected. See the Operation section for explanation of this function.
10- Phase correction global on/off button. Use it to compare before and after sound. This button leaves the volume control in place so that a meaningful comparison at the same volume can be made.

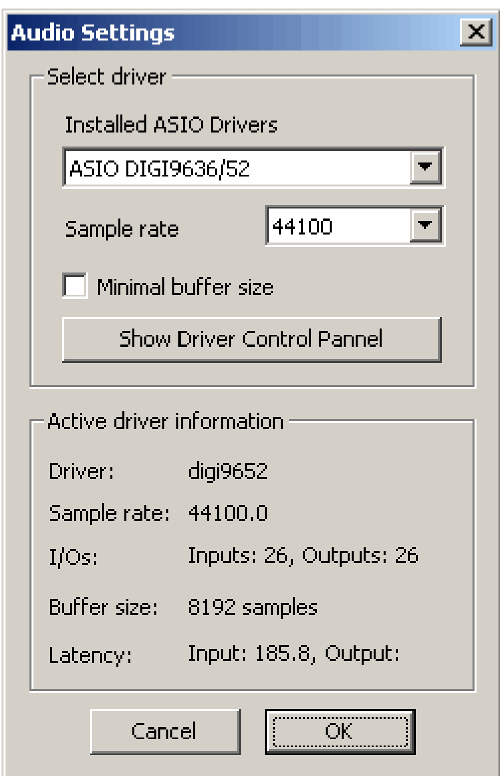
Operation

Before you can use the Arbitrator you should have a high quality sound card with native ASIO drivers. That means they were written by the manufacturer especially for this card and are guaranteed to adhere to the published ASIO standard. These are usually professional music recording and production cards. Generic ASIO bridges, such as ASIO4All software are not guaranteed to work with the Arbitrator- but they might depending on the setup.
When you first start the Arbitrator by double clicking on the Arbitrator.exe icon you will have to configure your system to accept the audio where you are feeding it and to output it where you expect it to come out of.

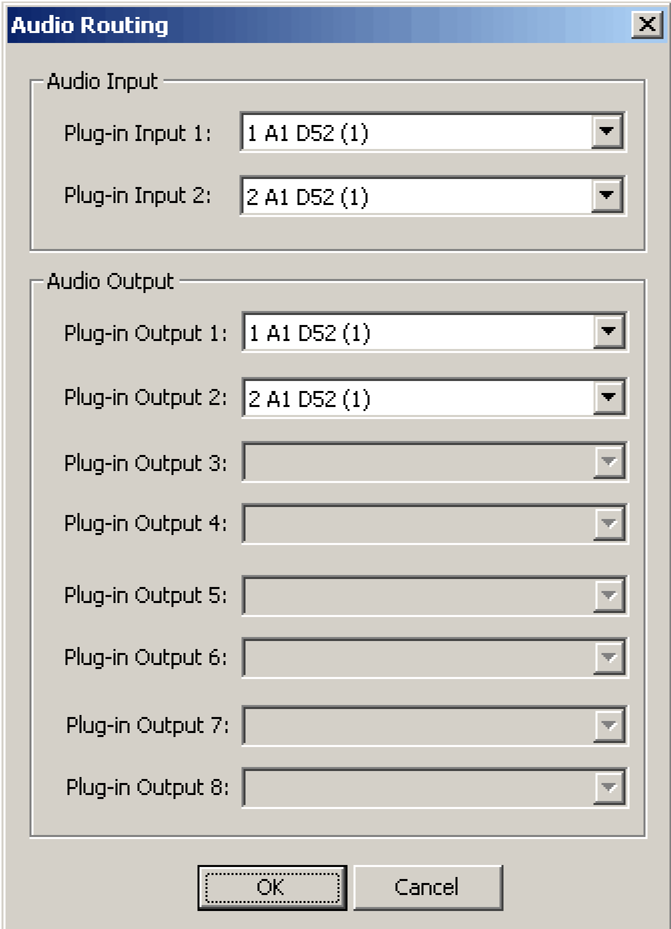
Click on View>Settings:



And open the control panel:



All of your installed ASIO drivers will be listed in the drop down box. From here you can open the sound cards Control panel and make adjustments to routing and sampling frequency etc. Once that is set open the Audio Routing window.



Here you can pick the stereo input and output out of all that might be available on the card. Click OK and you should hear sound coming out of the assigned output. Of course you have to feed some audio into the input first.

The settings are very basic. What you want to do is come very close to reflecting your speaker's characteristics. What is its –6dB point? Is it sealed or vented or maybe it's a small band-pass subwoofer? Dial in the box settings and move on to the crossover points. If you have a setup with a sub, you would dial in the first crossover point somewhere around 80-100 Hz and select 4^{th} order (seems to be the crossover of choice for most powered subs), but you should make sure that it indeed is 4^{th} order by looking at the User Manual or asking the manufacturer.
If it were a small 2-way speaker you would only enable one crossover point in addition to the box correction. You should make sure that your setting reflects the final **acoustic crossover slopes and point, not only the electrical crossover slope**. As indicated above, from the physics point of view, acoustic roll off has the same effect on phase as an electric filter. Since filters "stack up" and combine into steeper slopes, your speaker might have a second order (12dB per octave) electric crossover on the tweeter, but due to the tweeter's natural roll off, the combined, final response is that of a 4^{th} order crossover. The phase shift you are trying to correct with the Arbitrator is dictated by the final acoustic response of the speaker's crossovers. Many manufacturers list their crossovers as acoustic, but some do not. A call or email to the manufacturer with the

request should suffice in such case. Looking at posted measurements gives a good clue as to the crossovers as well. Each order of filters is 6dB per octave. Look at the slopes and find how many dBs does the response attenuate going from the crossover point to half (for tweeters) or twice (for woofers) the crossover frequency. If the tweeter is −6dB at the crossover point of 2500Hz and −30dB at 1250Hz it is a 4th order crossover. If it were −24dB at 1250Hz it would be a 3rd order crossover (18dB per octave slope). If you see mixed slopes, go by the Low Pass filter slope, as it's the dominating one in the listening experience. If in doubt see us at the Product Support forum and we'll try to figure out the best setting for your loudspeaker.

The Volume control and Auto button.

The Phase Arbitrator changes the waveforms of audio it processes- as seen in the pictures above. It's important to keep in mind that not only square waves are affected, but real music as well. In process of shifting frequencies the peak levels of the digital audio material are changed (the RMS levels remain consistent though- the music is perceived with the same loudness as before processing). Often, audio material that is recorded "hot" to begin with will peak above the maximum allowed level for the converters. That's why in order to avoid digital clipping you should lower the output of the Arbitrator. We incorporated an Auto level function in the Arbitrator. When enabled, Arbitrator takes note of the peak levels coming in and peak levels going out of the processor for 2 seconds and applies a corresponding gain reduction in the background. That gain reduction is not reflected on the volume slider.
The small problem with this approach is the fact that if the music has a certain frequency content during the sampling period it might not trigger enough reduction and you still get "overs" later when a particular combination of frequencies in music stacks up just right. Keep an eye on the peak indicators and apply additional attenuation as needed.
Some music with my crossover settings hardly ever goes into overs, some triggers as much as 8dB overs.
Another way is to find a particularly offending passage and play it a few times, while turning the auto function off, resetting the output meters and turning the auto function back on.

But if you don't want to use the auto button, most times an attenuation of 6dB is enough to prevent 99% of "overs".
Today's 24 bit converters are very good and attenuating of the signal by 6dB will not result in any appreciable loss of dynamic range, but can actually improve the sonics of "hot" material. Many converters and output electronics tend to sound better when not operated close to their absolute RMS limits.

The Process on/off button appears before the volume control in the signal chain. This allows for a meaningful comparison between processed and unprocessed signal. Our ears would normally favor the louder source and attenuation in the Arbitrator would make it seem like it hurts the sound when actually the opposite is true.

System Requirements

MS Windows XP computer
Pentium 3 or better processor 1GHz or faster
(for 48kHz processing of up to 5 way crossovers)
Pentium 4 2GHz recommended for 96kHz operation
256MB of memory
ASIO sound card- models with available long buffer settings preferred
The optimum buffer setting is 8192 samples- equal to Arbitrator's processing block.