Legacy®

Owner's Operating Manual

Thank you for purchasing the **Legacy V loudspeaker system**!

The Legacy V is a full range loudspeaker system utilizing the present state of the art in driver, crossover, amplifier and acoustic radiation control technologies.

The Legacy speaker system is designed, assembled and tested in Springfield, Illinois by a dedicated group of engineers, craftsmen and music lovers. The custom processor is designed and fabricated in Lund, Sweden under the direction of Bohmer Audio. Please take a few moments to learn more about the features and controls of these instruments to assure full enjoyment.

Owners Record

Thank you for selecting a Legacy Loudspeaker System. These handcrafted instruments will provide you with many years of listening enjoyment. The serial number is located on the rear of each unit. You can record this number in the space provided below. Refer to this when calling your dealer regarding this product:

Model: V Speaker System

Serial No: ____________________

Date of Purchase: ________________

Share your Legacy speakers with the Legacy community!

Post your Legacy experience and system photos at Facebook.com/LegacyAudio

Like the page to continue receiving the latest Legacy announcements.

Uncrating

Each V speaker has been carefully packaged in a specially designed crate to ensure they reach you safely and undamaged.

You will find the Top, Front and Back panels labeled on each crate as well as the Left/Right speaker designation.

Screws that are to be removed will be painted black for your reference.

Unpacking Instructions:

Step 1: First remove the top panel **B**. Due to the height of the crate, this will likely require a step ladder.

Step 2: Once the top panel is removed, remove internal cross bracing **C**. There are screws located on each end of the brace.

Step 3: Next, remove the front panel **A** and braces **D, H**.

Step 4: Then, remove the two side panels **F, G**.

Step 5: Finally, remove the remaining back panel **E,** leaving the speaker resting on the base **I**. Due to the substantial weight of the loudspeakers,

we suggest that ***at least* two persons** assist in removing each speaker from the base.

Step 6: Remove any additional foam/bagging. Use caution when moving to the final listening position. If a dolly is used, cover with a soft material such as a blanket to prevent any scratches.

Note: You will find the associated top rear speaker grille bonnets, Wavelet processor unit and cables each boxed separately.

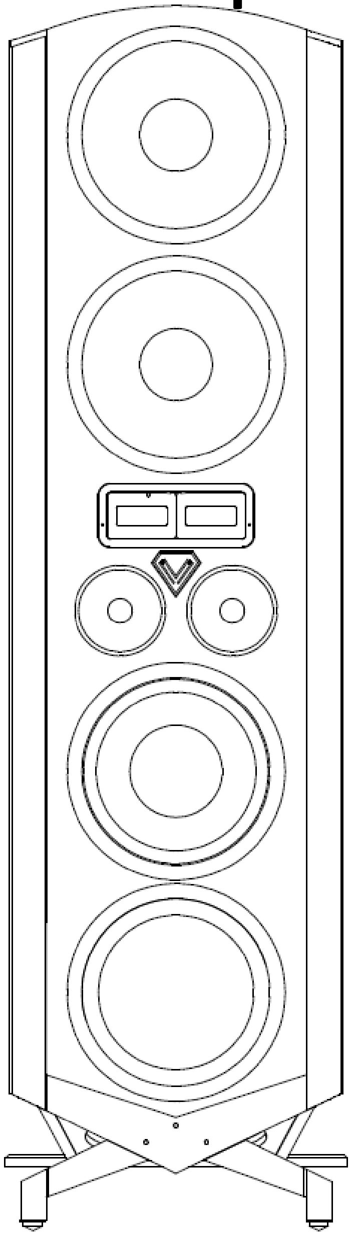

B
D
A
F


C
G
E
J
H
I


C

Inside the Wavelet processor carton packaging you'll find a hard-case containing a measurement microphone, a 25 ft. XLR cable and a mic calibration plot. At the opposite end of the carton you will find a power supply with a five pin locking connector and its 115 V AC cable, a compact remote volume control, a Wi-Pi network connector and a SanDisk USB memory stick. Also included is a micro USB cable to connect your computer or media server to the Wavelet.

V Speaker Specifications



System Type: Frequency and time domain optimized four-way directivity controlled array.

T**weeter:** Dual 4” AMT Ribbons configured in post convergent array

Upper Midrange: Dual 6” curvilinear with phase plug configured in dipolar array

Midrange/Midbass: Dual 14” Carbon/pulp, neodymium motor, cast frame, dipolar patter

Bass: 12” Aluminum diaphragm, Aura neodymium motor, sealed enclosure

Subwoofer: 12” Alumium very low frequency radiator driving three 10” mass loaded pneumatic radiators

Low Frequency Alignment: Compound B6, B2 handing up to dipole

Frequency Response (Hz, +/- 2dB): 16Hz-30kHz

Impendance: 4 Ohm upper range

Sensitivity: 98 dB @ 2.83Volts / 1m in room

Recommended Amplification: Sub and bass sections are powered with 1500 Watts of internal power, two external channels of 30 Watts or greater required

Crossover: 80Hz, 400Hz, 3kHz

Binding Posts: 2 pair of external binding posts, 2 XLR balanced inputs

Cabinet Dimensions HxWxD (Inches): 72 x 18.75 x 19 **Cabinet Weight:** 226lbs

Wavelet

At the heart of your V loudspeaker system is the Wavelet processor. Wavelet is a control preamp, a premium DAC, a digital crossover with time alignment for each driver section and an acoustic correction system that will literally 'learn' your room.

True digital flexibility

SPDIF, TosLink: up to 96kHz/24bit

USB: All file formats up to 96kHz/24bit are sent directly from the PC to the Wavelet without any conversion. Higher resolution files such as PCM and DXD can be readily played back using software such as J-River (select: Greater than 192kHz under DSP STUDIO.)

Analog Friendly

Already have a favorite DAC with a volume control? How about a big vinyl collection? Analog lovers can take advantage of balanced XLR or unbalanced RCA inputs without concern of digital artifacts. An apodizing circuit corrects for the pre-ringing native to CODECs. Wavelet has adequate headroom to handle these higher level signals while functioning as a crossover and compensating for room resonances. Wavelet processes at 56 bits of depth in a domain more than one trillion times finer in resolution than that of a 16 bit CD.

The Wavelet is by design upgradable. The unit can download firmware updates directly when you choose.

While the **Wavelet** offers high quality/ low noise balanced inputs to accept the analog output from any SACD player, ideally one would instead stream PCM versions of these files to the processor via the USB input. Within software such as J-River, the 1 bit DSD is converted to 64 bit PCM at 1/8 the sample rate. The total amount of data from this conversion grows by 8x, so the process is effectively lossless / perfect. The conversion is necessary as DSD is inefficient for sophisticated DSP operations.

In the J-River software this configuration is located in the Player -> DSP Studio - Output Format section. Setup all sample rates up to and including 96ks to “No Change”. Then set input 176400 to output 88000, 192000 to 96000 and greater than 192000 to 96000. DSD is converted automatically in the software to PCM. Once you have PCM, it will be 64bit @ 352.8 kHz for DSD, and 64bit @ 705.6 kHz for DSD 2x. The option 'Greater than 192kHz' in DSP Studio > Output Format should be selected.

Why PCM?
This is a format issue and has nothing to do with the **Wavelet** in particular. As professional studios rely on PCM based equipment such as Pro Tools to mix, pan and balance recordings, the vast majority of SACDs are in fact mixed in PCM, or mixed analog and recorded in PCM. They may then be converted to DSD for SACD mastering stages. The PCM format is far more efficient where DSP is in use. Recently a newer studio format, DSD-wide, has been developed to allow DSP operations that can be down-converted to DSD for SACD production. Pro DAWs such as SADiE are now using this technology.

PCM is the universal format of studios and digital signal processing. Most recordings released in other formats were actually recorded in PCM format.

Wavelet Preamplifier/DAC/Crossover/Room Correction Processor

Inputs

Analog

- Two pairs of Stereo balanced inputs on XLR connectors.
  Input sensitivity without attenuation 0 dBFS[*1] = 1 dBV[*2], input impedance 20 kOhm.
  Analog attenuation available in three steps of -3 dB, -6 dB and -12 dB for an input sensitivity of respectively 0 dBFS = 4 dBV, 7 dBV or 12dBV.
- Two pairs of Stereo unbalanced inputs on RCA connectors.
  Input sensitivity without attenuation 0 dBFS[*1] = 1 dBV[*2], input impedance 100 kOhm.
  Analog attenuation available in three steps of -3 dB, -6 dB and -12 dB for an input sensitivity of respectively 0 dBFS = 4 dBV, 7 dBV or 12dBV.
- One XLR Measurement microphone input, 48 Vdc Phantom power.

Digital

- Asynchronous USB audio, 24 bit, 44.1 – 96 ks.
- SPDIF, 24 bit, 44.1-192ks
- TosLink, 24 bit, 44.1-96ks

Outputs

Analog

- 8 balanced output channles on 8 XLR connectors. 0 dBFS[*1] = 8 dBV[*2] , 33 Ohm output impedance. An analog output level increase of 6 dB is available through internal jumpers offering 0 dBFS = 14 dBV
- 8 unbalanced output channels on 8 RCA connectors. 0 dBFS[*1] = 8 dBV[*2], 33 Ohm output impedance.

Digital

- SPDIF, 24 bit, 96 kHz
- TosLink, 24 bit, 96kHz

Control Interface

- Ethernet, TP-Cable & WLAN

Processing

DSP

- Analog Devices, internal processing sample rate 96ks, bit depth 56 bits

  The Bohmer Correction is a loudspeaker in-room energy-time alignment that optimizes the loudspeaker room acoustic transfer function in both frequency and predominantly time domain. Working with revolutionary new algorithms is starts with a psychoacoustically based measurement method. Alignment errors are then optimized individually, not resorting to the common crude bulk correction over the entire frequency spectra. The Algorithms use psycho acoustic reasoning for alignment and correction of the loudspeaker room transfer function. The correction improves sound quality in the whole room, provides improved transient response, clarity & soundstaging and gives a relaxed sound without rough edges or any booming.

Physical

Dimensions

445 mm W x 301 mm D x 95 mm H / 17.52” W x 11.85” D x 3.74 H   Weight 6.1 kg / 13.5 lbs

Speaker Connections

The standard V system provides two channels of amplification internally for the subwoofer and bass drivers. You will need to supply two additional channels for the lower midrange (60 watts minimum) and midrange/treble (30 watts minimum) of each speaker. Additional channels of internal amplification can be provided by special order. Two professional grade 15 ft. balanced XLR cables are provided for each speaker.

<u>Wavelet processor output connections:</u>

1 Left speaker subwoofer amplifier (internal 1000 watts)

2 Left speaker bass (internal 500 watts)

3 Left lower midrange amplifier (user provided)

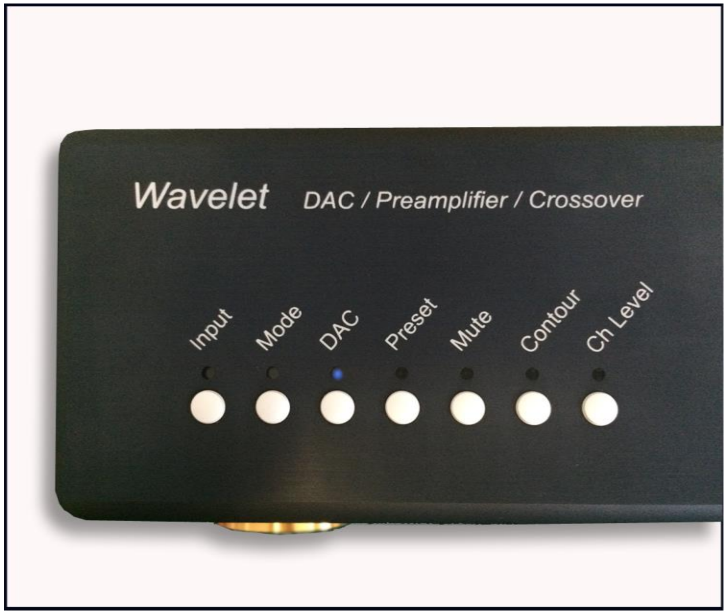
4 Left midrange/treble amplifier (user provided)

5 Right speaker subwoofer amplifier (internal 1000 watts)

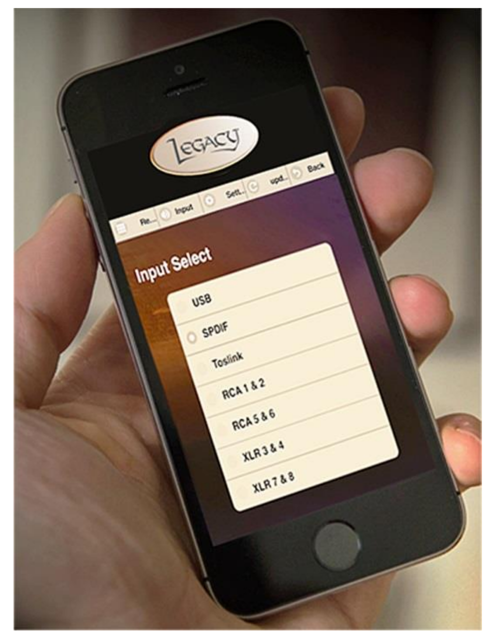
6 Right speaker bass (internal 500 watts)

7 Right lower midrange amplifier (user provided)

8 Right midrange/treble amplifier (user provided)



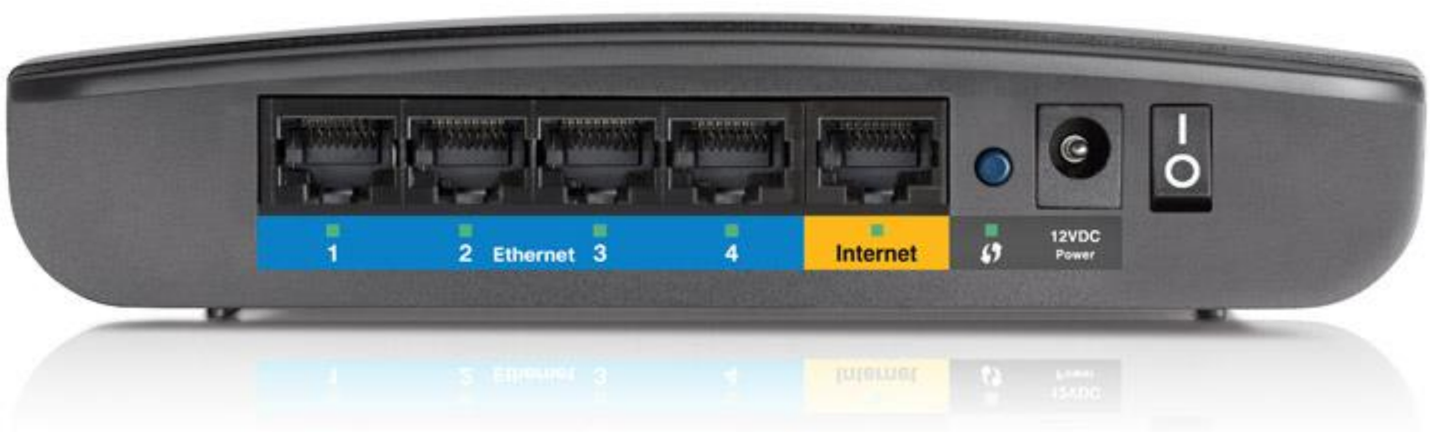
Ready to *take control* of your V system?



Your Wavelet processor has a serial number located on the decal adhering to the top of the unit. There is also a permanent sticker on the rear corner of the unit which indicates this number. The Wavelet can connect to the network via WiFi or Ethernet.

Wavelet features can be remote controlled via iPad, iphone, or other mobile device. In order to perform room correction it is essential that you connect the unit to the internet. To maximize enjoyment, we recommend dedicating one of these handheld devices to your music system, thus keeping your phone calls, and messages from interfering directly with your listening and control experience. If you don't have an extra device, consider picking one up used. We love controlling the system with an iPad Mini sized screen.

You can also utilize a low cost additional router to enhance or expand local connectivity. Units such us shown are useful for this purpose. Just connect the internet input of the new router to your modem tied to the incoming cable line.



To benefit from Wavelet's room correction and wireless remote features please connect to the internet.

1. Let's teach the Wavelet your network name and password. With the power supply connected
   a) Insert the red and black Sandisk USB stick into your computer and click to view contents.
   b) You will see a wifi-conf.txt you can open with your editor. Enter your network name and password accordingly.

   #ssid + password for AP-mode
   ssid=network name
   password=your password

   c) Now resave this file with changes over the original file on the USB stick.
   d) With the Wavelet processor off, insert the USB stick into the slot on the rear panel, then press the **Standby** button on the front panel to turn on. Wait at least 90 seconds for the unit to reboot and upload the text file.
   e) Power off by pressing Standby again and then enchange the Sandisk stick for the WiPi USB network adapter in the same USB slot. Then press Standby finally to reboot.

2. From your mobile device or computer browser, enter **bohmeraudio.com/setup.html**
A **Wifi Setup** form will appear. At the bottom of the form enter the serial number of your Wavelet without spaces. For this Wavelet unit enter:

BAP Serial:

3. Now click on **GO TO WEB INTERFACE** and the remote control feature will be activated and appear.

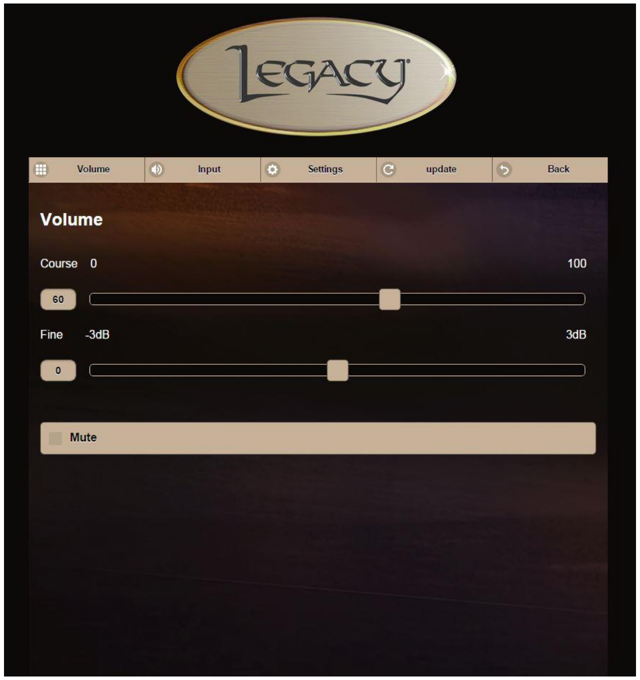
wifi-conf - Notepad

File Edit Format View Help

```
#Choose if raspberry pi is in accespoint-mode or WiFi mode
#Observe that minimum number of characters for password is 8
#Use only lowercase for ssid and password
#Observe that the system must be restarted twice in order
#to get accesspoint mode to provide ip-numbers to clients
#Observe that static addresses can be configured here (see below)

accespoint-mode=false

#If accespoint-mode=true
#   ssid will be the name of your accesspoint
#   password will be the password for the accesspoint
#   lines starting with '#' are omitted by the configuration script
#   Observe that only one pair of ssid and password should be used,
#   all other ssid and passwords should start with '#'
#   Example: If you switch to accesspoint mode, you put an '#' in
#   front of ssid and password used for AP-mode. And take away the
#   '#' in front of the ssid and password used for accesspoint mode.


#Wifi settings and accesspoint settings (only ssid + password is used for accesspoint)

#ssid + password
ssid="legacyspeaker"
password="a08278082b"

#WPA2
proto=RSN
key_mgmt=WPA-PSK
pairwise=CCMP
group=CCMP
auth_alg=OPEN

#WPA1
#proto=WPA
#key_mgmt=WPA-PSK
#pairwise=TKIP
```

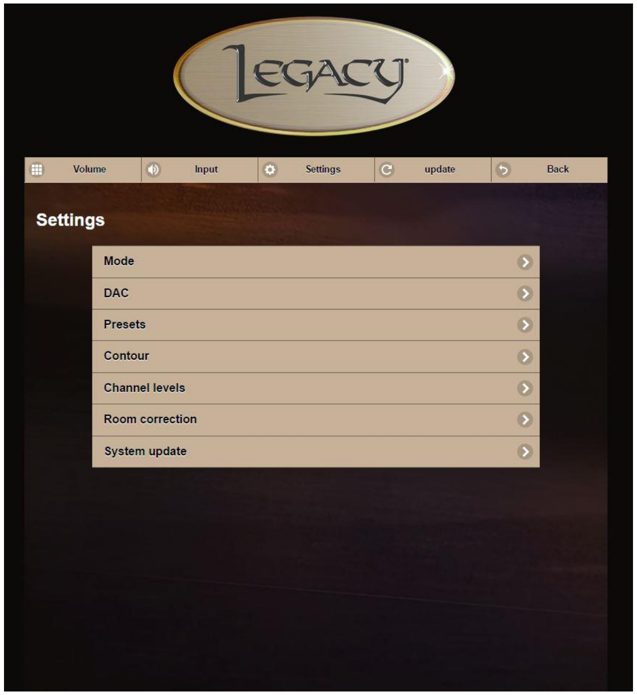
This is what the wifi-conf.txt file looks like. You will enter your network name and password between the quotes as shown, and then save over the wifi-conf.txt file that exists on the removable SanDisk memory stick.

When you insert the stick into the Wavelet USB port with the power off (standby), the unit will load into memory the new network name and password after being powered on for 1 minute.

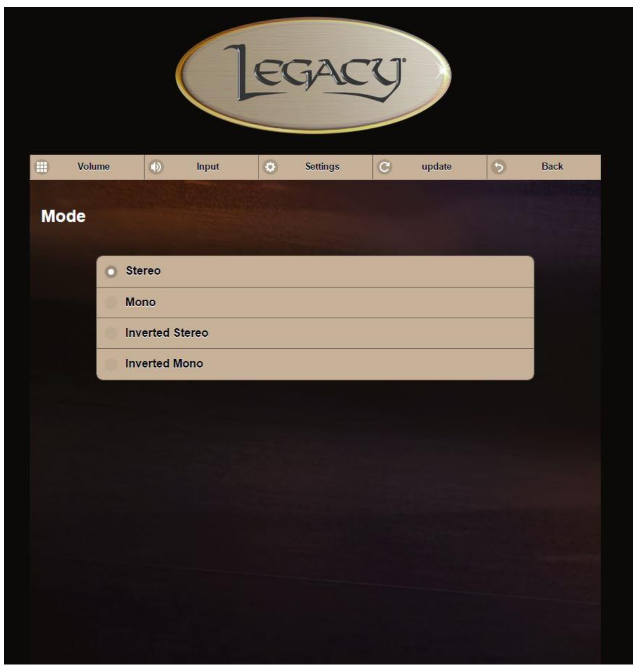

Volume
Input
Settings
update
Back
Volume
Course 0
100
60
Fine -3dB
3dB
0
Mute


Volume
Input
Settings
update
Back
Input Select
USB
SPDIF
Toslink
RCA 1 & 2
RCA 5 & 6
XLR 3 & 4
XLR 7 & 8

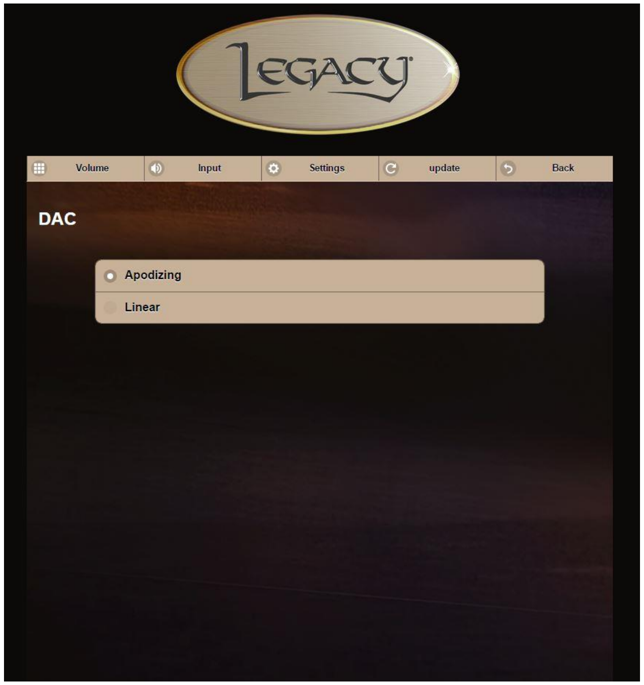
Input



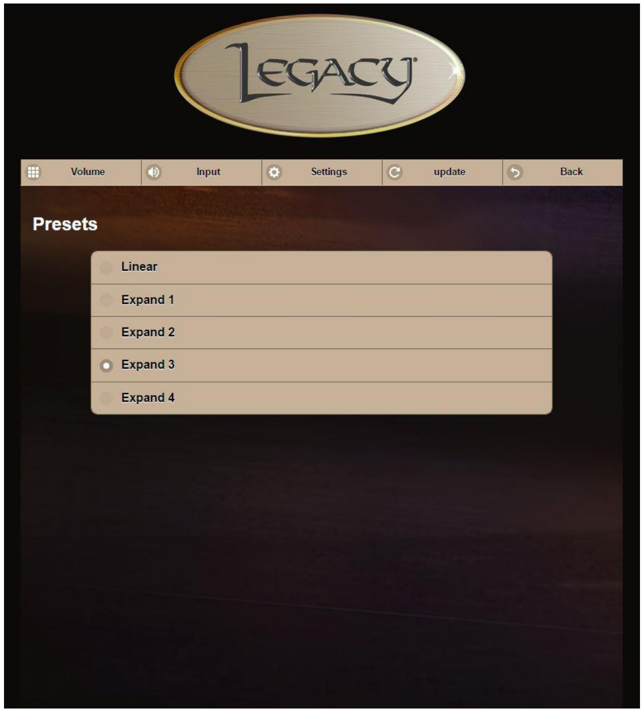
Settings



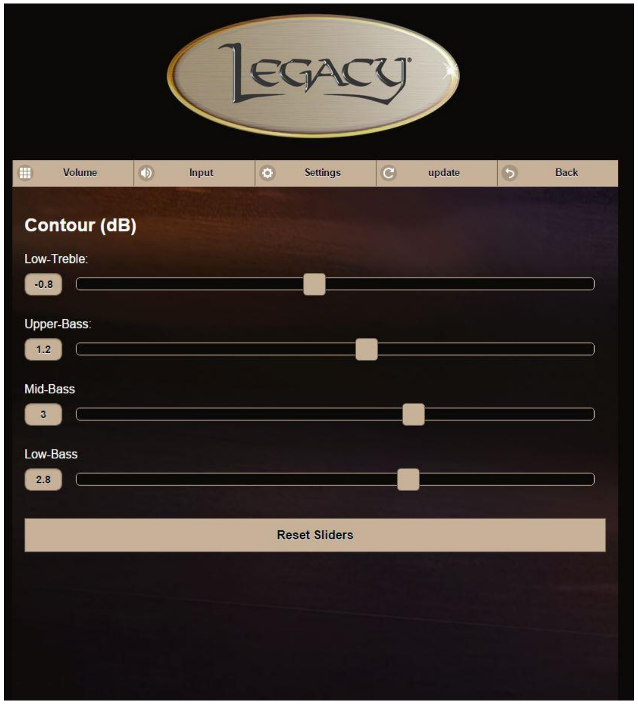
Mode


LEGACY
Volume
Input
Settings
update
Back
DAC
Apodizing
Linear

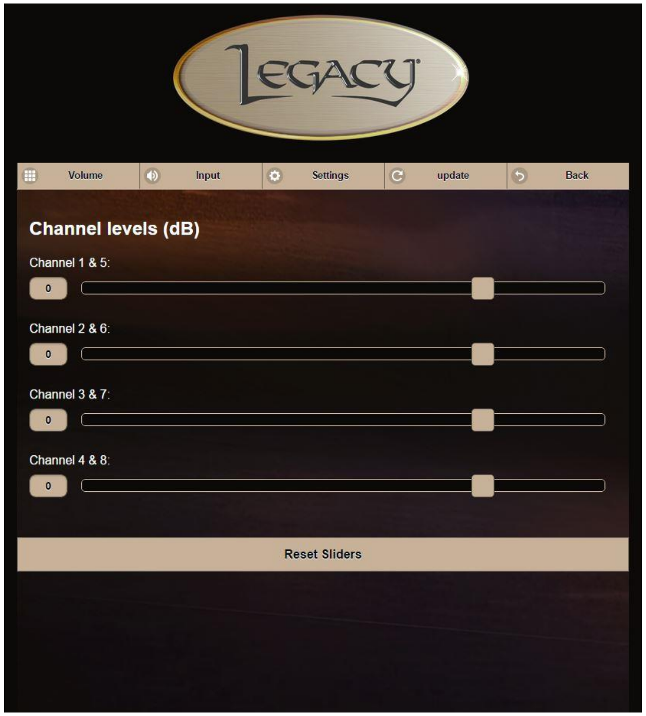
DAC


LEGACY
Volume
Input
Settings
update
Back
Presets
Linear
Expand 1
Expand 2
Expand 3
Expand 4

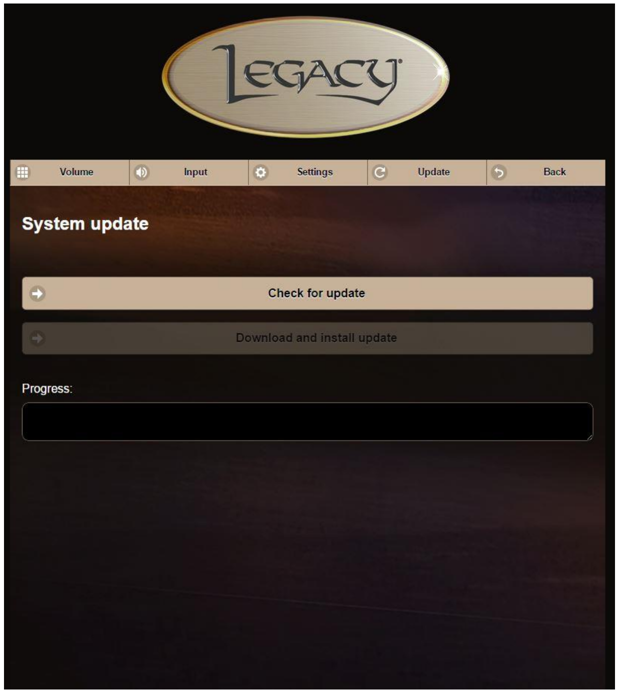
Presets



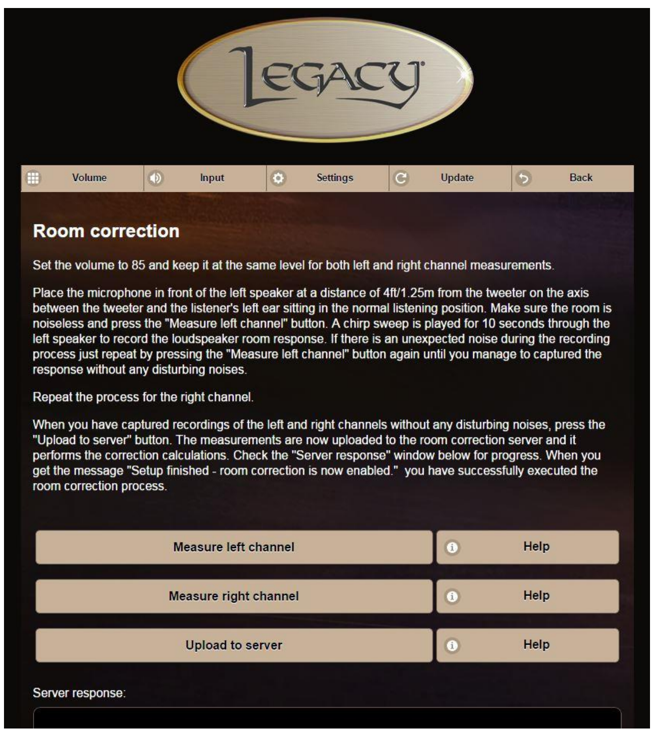
Contour



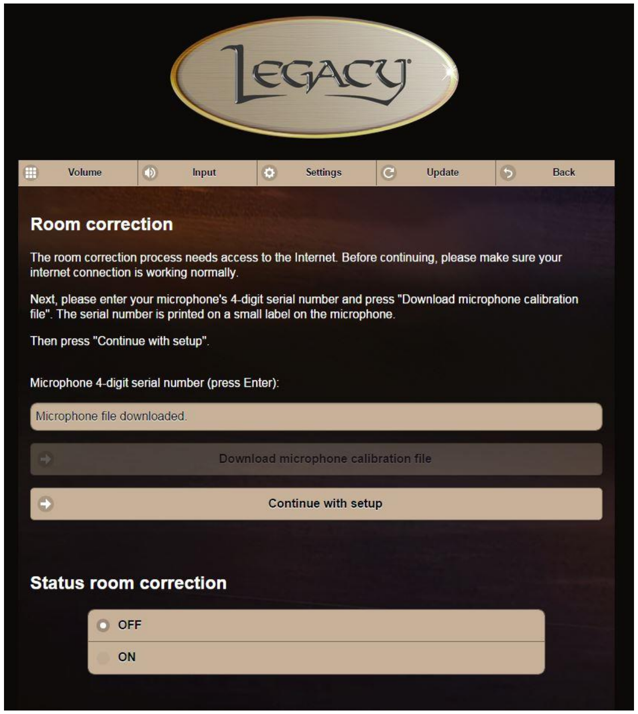
Channel Levels



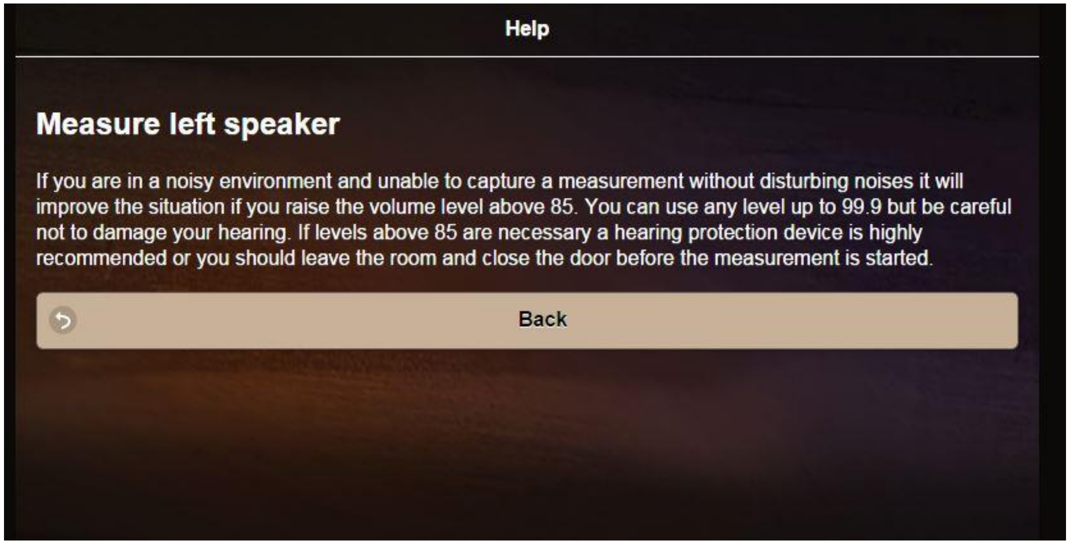
System Update



Room Correction pg 1



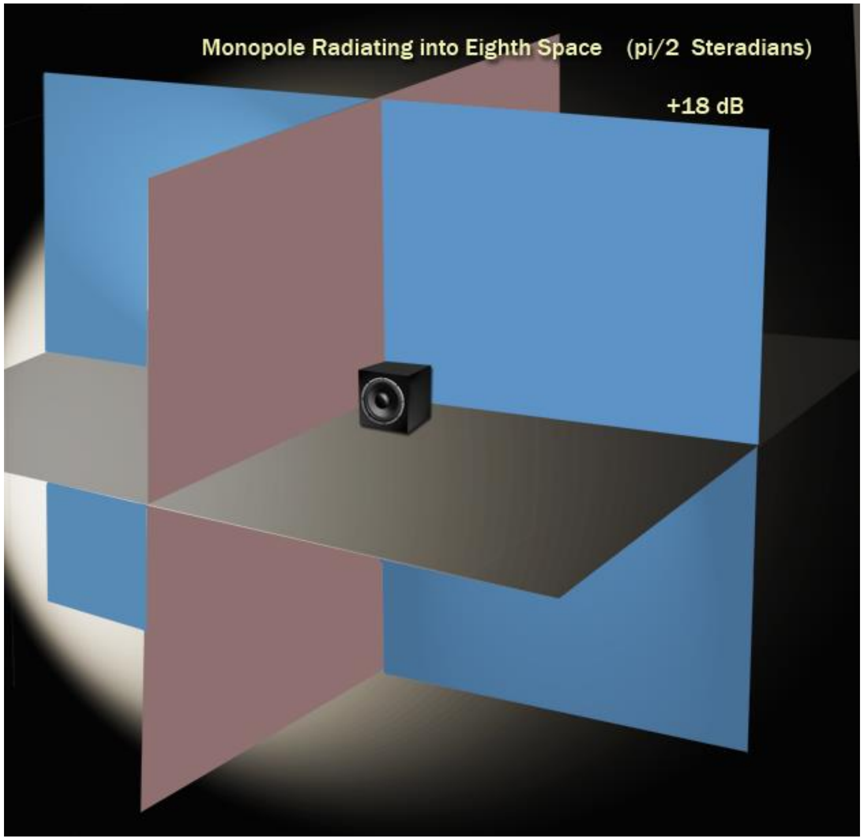
Room Correction pg.2



Room Correction Help Screen

Loudspeakers continue to be the weakest link in the playback system by an order of magnitude and more. It's easy to understand why. Driver elements must combat startup inertia. Once in motion they fight inertia again to stop, continuing to oscillate post signal.

It's well known that distortion is proportional to diaphragm travel requirements. It's also true that travel requirements increase by a factor of 4 with each octave downward, so a speaker's radiating surface area must expand with wavelength to keep distortion low and directivity uniform. Keep in mind the wavelength at 20 Hz is *1000 times* greater than at 20kHz.

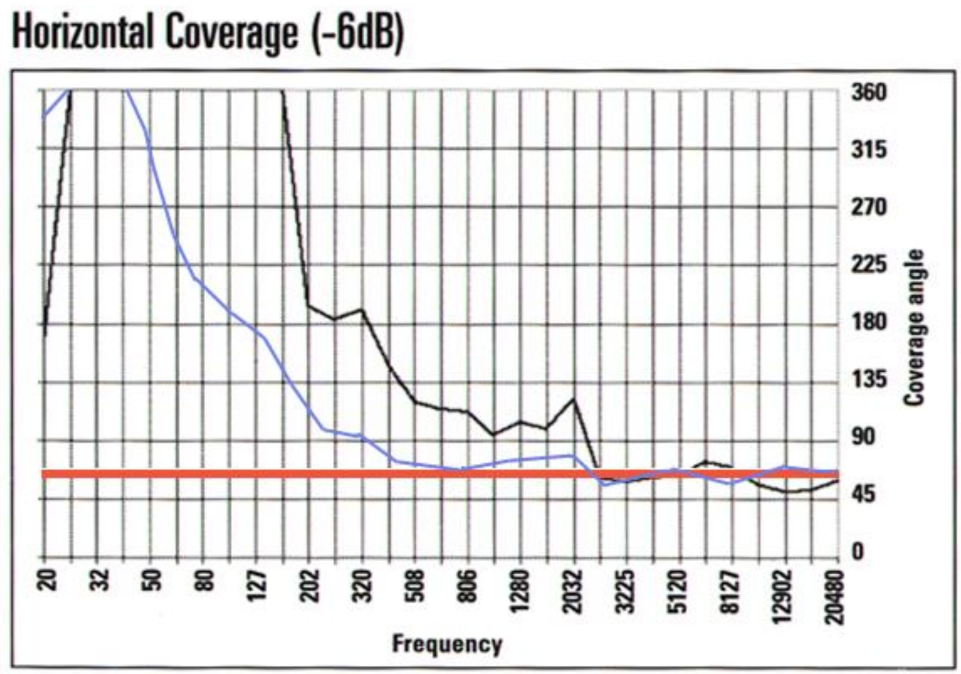


When a wavelength is greater than room dimensions, reflections have little phase shift and rooms can reinforce (pressurize) deepest bass frequencies in a beneficial way. Without this help, most speakers would be practically inaudible in the first octave.

However as loudspeakers radiate at slightly higher frequencies the reflections are no longer in phase and colorations due to errors in the time domain become very significant. The multiple wavefront arrivals to the listener provides an experience quite unlike the original source material. Furthermore, these reflections cannot be equalized conventionally without respect to the time domain without introducing further damage.

Controlling the Directivity of Sound

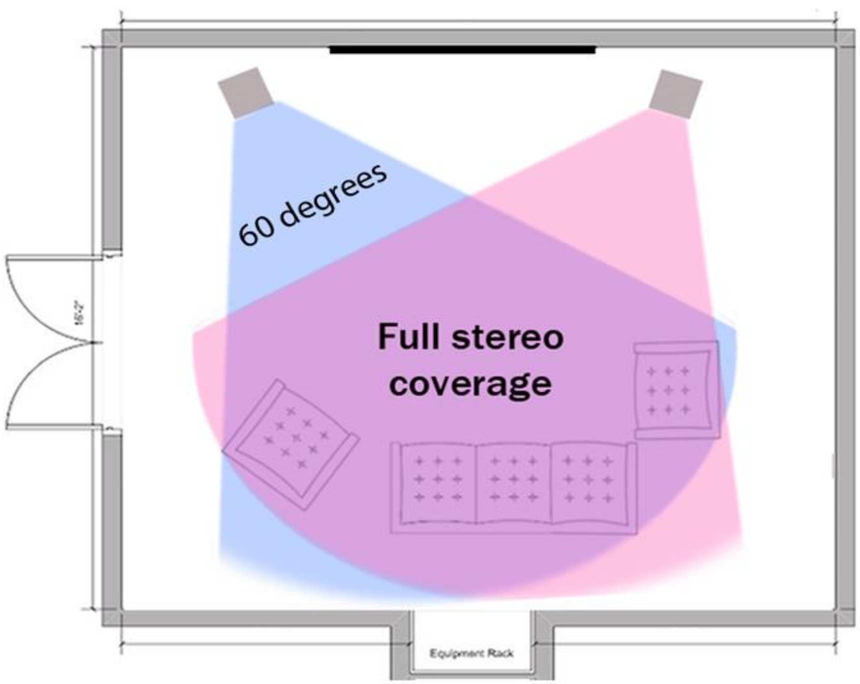
While Legacy Audio has continued to improve directivity in loudspeaker designs for more than 2 decades, speakers with deviant radiation patterns continue to exist in the marketplace. While their high frequency drivers tend to be cardioid in pattern, lower frequencies progressively become omni-directional. This is particularly true of speakers with small bass drivers. By 500 Hz most speakers are radiating about 180 degrees, increasing midrange flutter by adding early reflections.



The red line indicates ideal uniform directivity for a very desirable 60 degree horizontal radiation pattern. The black line is typical of a three-way speaker design. The Legacy V system (blue line) employs acoustic directivity control and DSP to reduce lateral room energy and resonances far lower in the spectrum. The area between the blue and black lines represents the energy reduction as a function of frequency that the V system affords.

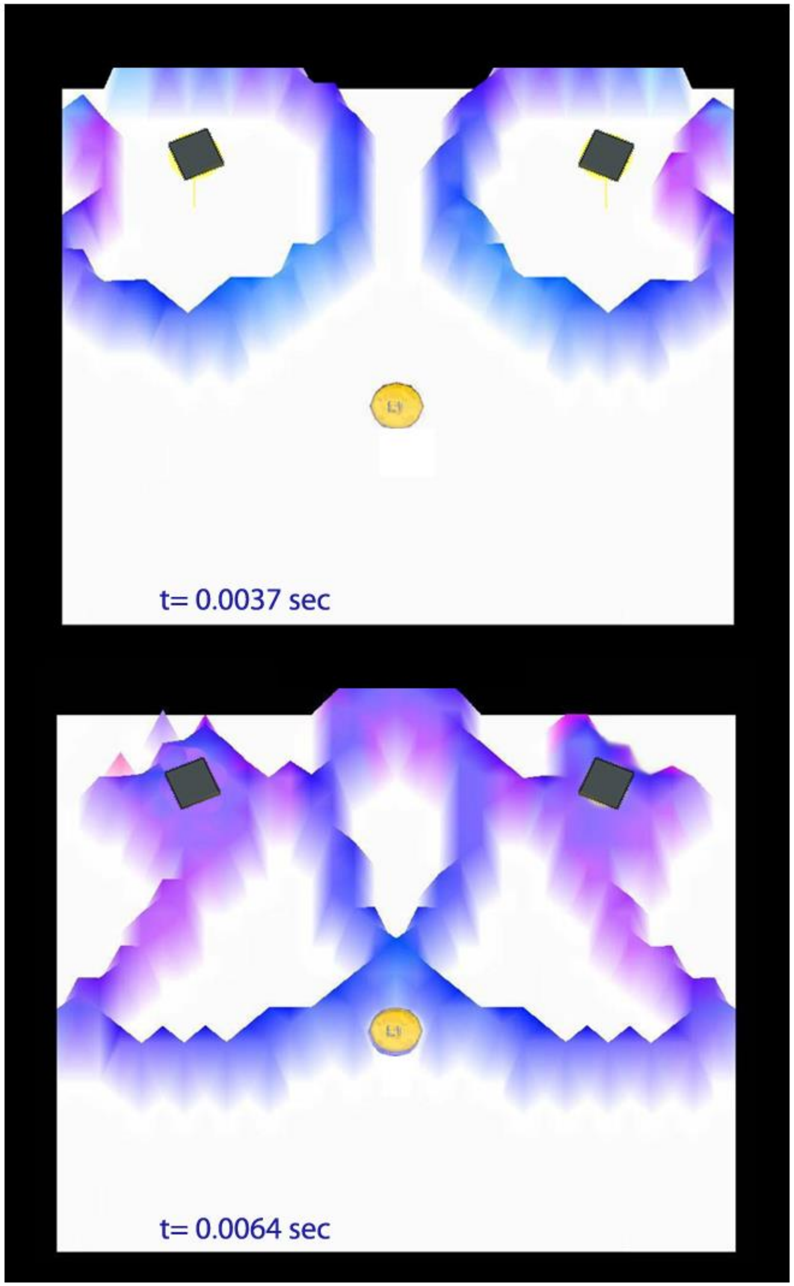
Ideally, speaker systems would fully cover the listening area while avoiding early reflections. This is desirable at all frequencies to avoid masking the key temporal cues of the recording venue.

Even the best loudspeakers do not radiate uniformly with frequency. This lack of directivity control greatly skews the room power balance, reduces clarity and drives room resonant modes heavily. Note that even in this broad listening setup with two side chairs and a sofa, a radiation angle greater than 60 degrees for each of the loudspeakers is simply not required.   What we will demonstrate next is why wider coverage is actually quite detrimental.



Below about 1500Hz there is a phase difference between the sound waves entering the ears, thus providing acoustic localization cues (interaural time differences, or ITD). At frequencies greater than 1500 Hz, the wavelength is shorter than the distance between the 2 ears, and phase sensitivity diminishes. The brain now relies on acoustic head shadows produced across the face to provide level differences to each ear (ILD) and cues for the localization of this sound. If the speaker system projects sound onto the room boundaries creating a diffuse field before it can reach the listener directly, these fragile ITD and ILD cues will be altered and scrambled.

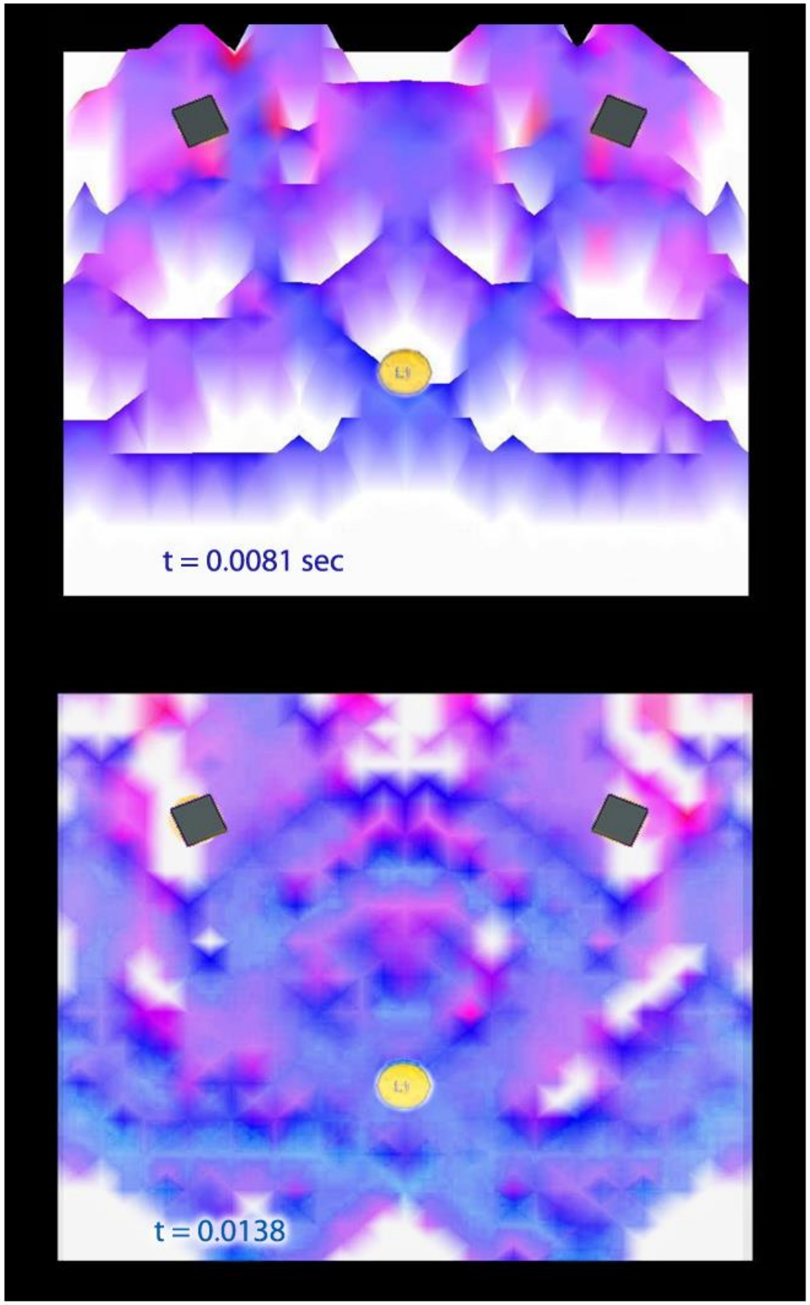
The key to imaging is preserving the interaural time and level differences. The shadowing to the far ear from the source is quite fragile in a reflective environment. Maintaining directivity in the speaker radiation pattern is essential to preserve this content. Legacy V, Whisper, and Aeris each offer unique directivity control to provide greater clarity, better localization and resonance reduction.



Why is Room Correction Necessary?

To the left is a 2 dimensional simulation of a 1ms wave pulse from a pair of conventional speakers into a room similar to the above. Because the dispersion exceeds 60 degrees, undesirable energy from each speaker is reflected back into the room within a few thousandths of a second. This reflected energy is out of sync with the original signal.

A few milliseconds later, the first wave-front is about to reach the listener, while the reflected energy is close behind. These early reflections alter the original tonal balance. As they occur within the fusion time window, the brain cannot separate the sounds.



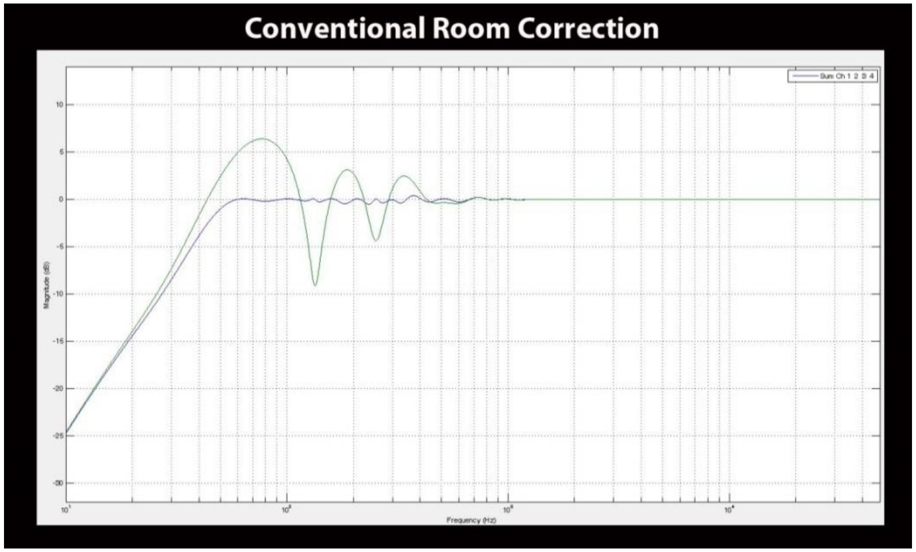
After less than 1/100th of a second, the room has developed a complex wave pattern with energy varying with room position. The listener is now awash in a series of wave-fronts which will soon reflect off the wall behind.

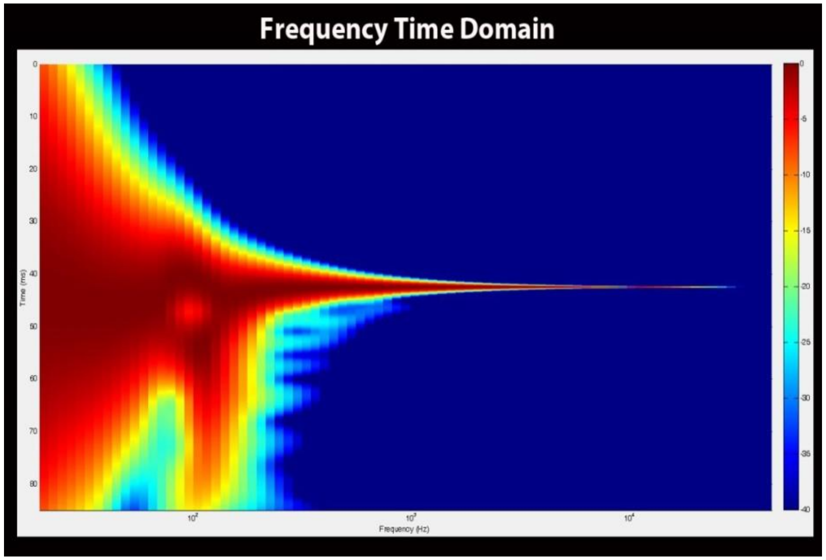
A mere 5 milliseconds later, the initial direct wave-front has now reflected off the rear wall and has made its way back to the listener.

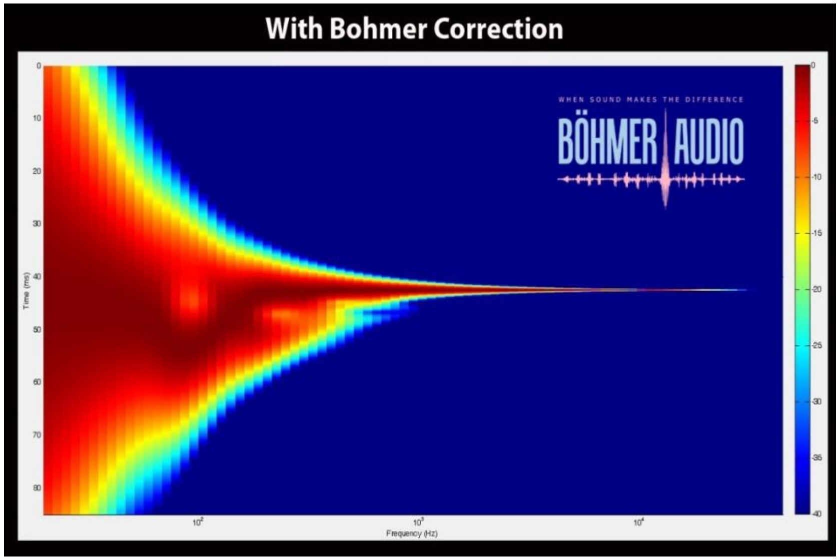
The listener will perceive this reflection as additional bass energy, though a standing wave has not had time to develop.

It is a common misconception that such low frequency excess energy is merely the result of inevitable resonances within the room, when a large portion can be attributed to initial reflections.

To the right is the irregular frequency response of a speaker on axis in the presence of room boundaries. The smoothed curve is the result of applying conventional room correction methods. Below left is the impact of the correction on the Fequency-Time domain. Below right is the same wavelet plot with the Bohmer correction method.





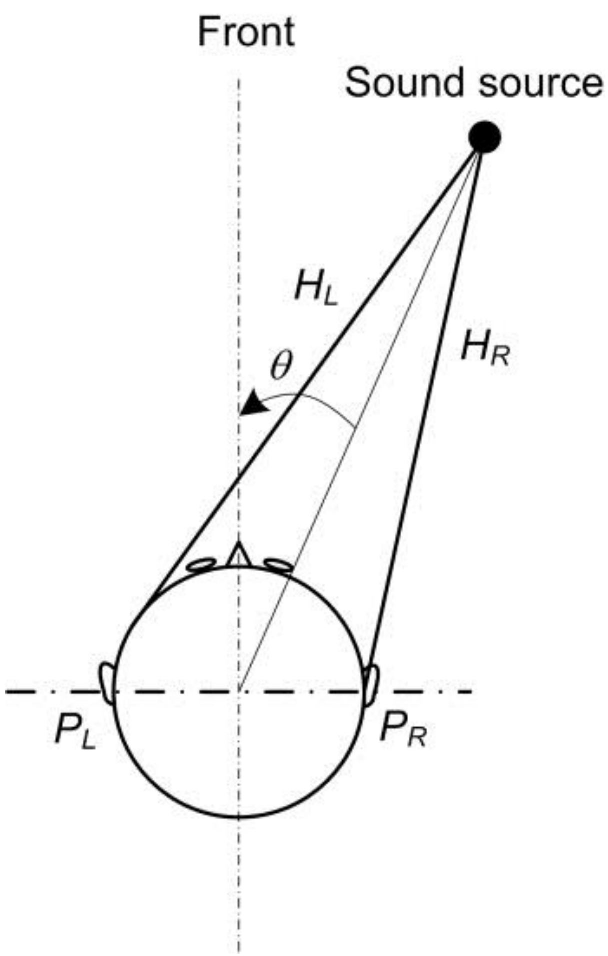


The Engineer’s Perspective thoughts from Legacy Chief Designer, Bill Dudleston

- The listening room greatly diminishes left/right separation as frequency decreases, reducing or destroying spatial cues at each ear.
- Reflections superimpose an additional room environment onto the listening experience, masking the original recording venue which we strongly desire to preserve.
- Resonances introduce tonal colorations at low frequencies, hindering transient response.
- Pre-ringing in digital codecs have previously prevented even high resolution recordings from sounding truly analog in nature.

We have two receptors (ears) which enable us to differentiate the position of sound sources (Haas). The two ears and the brain work together to determine the relative distance and angular position of the source in the free-field. Much has been written about the head-related transfer function and interaural crosstalk. Some applications have been developed for headphone playback (Bauer, Smythe). Some have argued that we need to eliminate the crosstalk from each loudspeaker to the far ear, even to the point of building a dividing wall that meets the face separating the two ears (Glasgal). This would imply a purely binaural recording process as a standard, which is certainly not the norm. Others have suggested an electronic crosstalk cancellation signal (Polk, Carver), and even a higher order cancellation that even reduces residual information introduced by the crosstalk cancellation itself (Griesinger). None of these methods specifically address the errors that room boundaries introduce to stereo loudspeaker playback. They also ignore the fact that stereo playback and the HRTF is actually dependent on the proper crosstalk for spatial impression (Blumlein, Lipshitz)

Spatial Impression (SI, Barron) is primarily a low-frequency phenomenon, depending mostly on the lateral sound energy below 400Hz arriving at the listener's head between 10 and 100ms after the direct sound. This frequency dependence of SI is a significant addition to the work of Schroeder and Ando on the importance of minimizing inter-aural cross correlation for the subjective attractiveness of sound. By manipulating the spatial properties as a function of frequency the recording engineer has a chance to control factors in the sound which influence spaciousness, depth, richness, envelopment without affecting factors which influence placement.



It is the lateral sound energy which creates pressure differences between the two ears of a front-facing listener at frequencies below 700Hz. The easiest way to measure or think about lateral sound is in terms of the lateral or Y axis sound velocity, which one can measure with a sideways facing figure of eight microphone.

Lateral sound velocity obviously determines both localization and spaciousness, since without it we would hear only mono. However. See Blauert. When the velocity and pressure are uncorrelated or vary rapidly in phase with frequency, a sense of spaciousness appears to result. As will be shown later, lateral velocity is generated by the L-R or side signal in a stereo recording, even in a reverberant room, although the room can play havoc with the phase.

It is only when the lateral velocity (L-R) and the pressure are phase correlated that accurate localization is possible.

It is only when velocity and pressure are uncorrelated that spaciousness is heard.

Understand foremost that velocity is a vector, implying a path. If we cannot preserve the path to the listener, both localization and spaciousness will be reduced and diffusiveness will be the result. The ratio of these components is the key to imagery and apparent clarity.

Early reflections are reduced by providing broadband uniform directivity loudspeaker designs (Dudleston) thus improving localization.

Likewise correlated spatial information in the recording is more audible because of reduced masking effects of early room reverberation and an increase of ambient information beyond 14ms.

Low frequency resonances can be greatly reduced without introducing error in the time domain (Bohmer, Dudleston) with modern DSP.

While the amplifier is very important, the loudspeaker is traditionally <u>the</u> limitation in the playback chain. To address this, the V system with Wavelet processor corrects each driver section individually, compensates and aligns them in the time domain, then examines the summation acoustically and fits it to the target curve. Lastly it samples the radiation into the room and provides the final correction, removing time domain errors and resonances as old as 40ms. What's most amazing about this is that after the *final* measurement is made, it factors all these steps into one single computation. After a few moments of iterations and regression calculations, it provides the resulting coefficients to compensate in real time- on the fly - from that moment forward -the least possible error in amplitude without compromising the time domain at all. That is what makes this process so unique.

The process can be accomplished in home with a press of a button once amplifier sensitivities (gain levels) are matched. Distortion levels from the speaker will now be largely determined by the amplifiers chosen to drive it.

The target curve for the system is specified by yours truly. Understand that every speaker has a radiation pattern that is as unique as a thumb print. As the designer, I am confident the target curve should be based on how the sound arrives to you, the listener, in your room, and how much of the total sound is diffuse reflection. More than 30 years of psychoacoustics research is weighted in the result.

In summary, we have successfully attacked these four nemeses of the stereo experience.

- The V system with Wavelet reconstructs the low frequency separation occurring within the first 14ms.
- The V system's directivity pattern prevents early reflections from masking the recording venue.
- The Wavelet processor eliminates resonances throughout the listening field over a 40ms window.
- The Wavelet processor virtually eliminates digital pre-ringing of brick wall filters via apodization.

We will also be offering the Wavelet processor in time for some of the other Legacy speakers. But before we move on to our next conquest, let's kick back a bit and savor the results of this one …

Cheers, Bill

CE Declaration of Conformity

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States that this product is in conformity with the with the essential requirements and other relevant provisions of:

Low Voltage Directive 2006/95/EC
EMC Directive 2004/108/EC

WEEE Compliance

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