



Owners Manual For The
VALOR
Loudspeaker System

THANK YOU FOR CHOOSING LEGACY AUDIO



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

The system is designed, assembled and tested in Springfield, Illinois by a dedicated group of engineers, craftsmen and music lovers.

Please take a few moments to learn more about the features and controls of these instruments to assure full enjoyment.

Table of Contents

Registration	Page
Owners Record	4
The Cabinetry / Our Commitment	5
Warranty	6
Setup	
Placement	7
Hooking up Cables	8-9
Amplification	10-12
Speaker Connections	13-14
Technology	
Behind the Design	15-16
Stereo Unfold Methodology	17
Specifications	18
Wavelet	19-25
Wavelet Setup	26-43
Imaging & Room Correction	44-47

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 the unit. Record this number in the space provided below. Refer to this when calling your dealer regarding this product.

Model: VALOR

Serial No: _____

Date of purchase: _____

Register your product at www.legacyaudio.com/register

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.

The Cabinetry / Our Commitment

Handcrafted

Beneath the surface of VALOR's elegant exterior lies rigid MDF construction. Interlocking joinery maximizes the strength of the cabinet parts.

Each cabinet is impeccably finished on all exposed surfaces with select veneers. The exquisite finish is hand-rubbed several times to assure a patina at home with the most elegant decor.

Our Commitment

A great deal of forethought, love and satisfaction is instilled in each piece of Legacy workmanship. We take pride in getting to know many of our customers on a first name basis.

Your purchase of this product is backed by the renowned "Legacy Satisfaction Guarantee".

Warranty

Legacy Audio supports its customers and products with pride. We cheerfully warrant our loud-speaker products we manufacture from defects in materials and workmanship for a period of seven (7) years. Electronic components such as internal amplifiers and digital processors are covered for three (3) years. Please register your product with Legacy Audio. Should you require service Legacy will require a proof of purchase in order to honor the warranty - so please keep your receipt.

- The warranty applies to the original owner and is not transferable.
- The warranty applies to products purchased from an "Authorized Legacy Dealer".
- The warranty on active components such as digital processors or internal amplifiers is limited to three (3) years of coverage.
- The warranty on dealer stock will extend for a maximum of two years from invoice.

The warranty does not cover transportation costs of product to or from the customer, distributor or dealer, or related shipping damage.

Exclusions from Warranty

The following situations or conditions are not covered by the Legacy Audio warranty:

- Accidental damage, electrical abuse or associated equipment failure.
- Use inconsistent with recommended operating instructions and specifications
- Damage caused by modification or unauthorized service
- Costs associated with the removal and reinstallation of defective products. Consequential damage to other products.
- Normal wear such as fading of finishes due to sunlight.

Speaker Placement

The Legacy VALOR system is designed to afford maximum flexibility in seating arrangements and yield a large listening sweet spot by preserving directional cues and eliminating unwanted room sound. The Legacy Wavelet, included with VALOR, allows the speakers to operate in a variety of different positions and room setups. Assuming a listening distance of about 10-12 feet, begin by placing the speakers about 10 feet or more apart and 1-3 feet from the wall behind them. A slight toe-in is recommended. Because of the unique room correction abilities of the VALOR system, adhering to the exact recommended placement guidelines is not as critical as it is with conventional speaker systems. Your dealer and Legacy Audio will be able to look at your room setup and recommend the best positioning for your VALOR system.

Hooking Up Cables

The ideal conductor would have negligible resistance, inductance and capacitance. The table below shows how a few actual speaker cables measure up.

Cable	Ω s/ft	pF/ft	μ H/ft
12 ga.	0.0033	24	0.21
14 ga.	0.0048	17	0.13
16 ga.	0.0079	16	0.18
18 ga.	0.0128	28	0.21



Capacitance is considered insignificant in each cable because its effect is well out of the audio bandwidth; inductance can be decreased (at the expense of increased capacitance) by keeping the conductor pair closely spaced.

How long would a cable have to be before inductance effects would impinge on the audio spectrum? Approximately 300 feet of 12 gauge would be required to establish a corner frequency of 20 kHz with an 8 Ohm loudspeaker. As you see, inductance is not a problem for most of us.

Hooking Up Cables

What about phase shift due to frequency dependent travel times down the speaker cable? Measurements show that 100 Hz waves will be delayed about 20 billionths of a second behind 10 kHz waves when traveling to the end of a 10 foot speaker cable. Since the cilia of the ear requires 25,000 times longer than this just to transmit phase information, phase shifting is obviously not the primary concern when considering speaker cables.

What about resistance? Finally we are getting somewhere. Resistance is the controlling factor of the amplifier/loudspeaker interface. Excessive resistance can cause major shifts of speaker crossover frequencies. The lower the impedance of the loudspeaker, the greater the effects of series resistance. A 20 foot run of 18 gauge cable can cause up to 10% deviations of crossover center frequencies. That same 20 feet can un-damp your damping factor and reduce your systems' output by onehalf decibel.

In summary, there are no perfect cables. The best way to approximate the ideal would be to keep loudspeaker leads as short as is practical.

Amplification

Ideally the loudspeaker would be among the first components selected when assembling a playback system. This would allow the user to choose an amplifier capable of delivering adequate amounts of current into the frequency dependent load presented by the loudspeaker. However, when upgrading a system, audiophiles may find themselves matching their new loudspeakers to their existing amplification. For this reason, extensive measures have been taken to ensure that each Legacy speaker system represents a smooth, non-reactive load to virtually any amplifier.

Often there is much confusion regarding amplification and loudness levels. It should be understood that the role of the amplifier goes beyond that of driving loudspeakers to a given sound pressure level. The amplifier should be able to CONTROL the loudspeakers across the entire music spectrum. This means that parameters such as damping factor (values greater than 60 are acceptable) and dynamic headroom should not be overlooked when comparing amplifiers.



Amplification

How much power will your new speakers need? That ultimately depends on your listening environment and musical tastes. As little as five watts per channel should drive them to a level satisfactory for background music. A typical 45 watt per channel receiver may fill a room with the compressed mid-band energy of "heavy metal," but seem to lack weight or control with classical recordings. Some audiophiles feel that 200 watts per channel is the bare minimum to avoid audible clipping distortion when reproducing music at "live" playback levels. Your Legacy speakers are designed to take advantage of "high-powered" amplifiers, so don't be afraid to put them through their paces.

How much is too much power? Rarely is a drive unit damaged by large doses of music power. More often than not the villain is amplifier clipping distortion. Even through decades of refinement, loudspeakers are still notoriously inefficient transducers, requiring huge amounts of power to recreate the impact of the live performance. Typically less than 1% of electrical power is converted into acoustic output. (For example, an omnidirectional transducer with an anechoic sensitivity of 90 dB @ 1w/1m has a full space efficiency of only 0.63%)

Amplification



When an amplifier is unable to fulfill your loudspeakers demands, a damaging harmonic spike may be leaked to the high frequency drivers.

Another important point regarding loudness is that the dB scale is a logarithmic one. This means that a 150 Watt amplifier will potentially sound only twice as loud as a 15 Watt amplifier. If all of this discussion of power and loudness seems a bit abstract, consider the example below.

The average acoustical power developed by a person speaking in a conversational tone corresponds to a mere 0.00001 Watts. The power that would be developed by the entire population of the city of New York speaking at once would barely illuminate a single 100 Watt light bulb.

Speaker Connections

The standard VALOR system provides four channels of amplification internally for the subwoofers, bass drivers, midwoofer and ambient array. You will need to supply one channel of amplification of 60 watts or greater, for the midrange and tweeter section of each speaker. VALOR can be built with internal amplification for the entire speaker, or with fewer internal amplifiers - should the listener prefer to use external amps. Three professional grade 15 foot balanced XLR cables are provided for each speaker.

Wavelet processor output connections:



- 1 Left speaker subwoofer amplifier (internal 1000 watts)
- 2 Left speaker bass (internal 750 watts)
- 3 Left speaker midwoofer (internal 500 watts)
- 4 Left midrange/tweeters amplifier (user provided)
- 5 Right speaker subwoofer amplifier (internal 1000 watts)
- 6 Right speaker bass (internal 750 watts)
- 7 Right speaker midwoofer (internal 500 watts)
- 8 Right speaker midrange/tweeters amplifier (user provided)

Speaker Connections



Behind the Design

with Chief Designer, Bill Dudleston

Much like a directional microphone the VALOR's output is carefully shaped into a cardioid pattern to reduce early reflections to side and rear. This sculpting occurs as two of the 14" dipolar 'open air' drivers acoustically combine with a third closed back driver. The central 14" is a coaxial from Italy where a 2" polyester/titanium HF driver sets in a precision machined aluminum throat that aligns output within the 3.5" voice coil of the woofer. Placement is dead center of the triple 14" driver array where a horizontal bridge with a central lens splays a pair of 4" AMT super tweeters. The drivers' output angles trade off intensity to maintain uniform output with varying listener positions.

The two 12" subs, front and bottom loaded are hefty dual voice-coil units with aluminum cones and 30 lb. motor structures. The two 12" passives are on the rear of the cabinets and utilize a patented dual symmetric suspension and a damped diaphragm with 4" peak to peak travel.



Time vs. Phase

There are some often misunderstood aspects of phase, polarity and time. We know we can hear the acoustic cancellations and summations that occur when soundwaves interfere in space on the path to our ears. This interference introduces frequency response changes. However, put on a set of headphones and flip polarity on one channel and you might be in for a surprise. You might logically expect the bass to become thin or weak. But the brain does not process this phase difference in the way that you might think when the waveforms have not interfered acoustically. No audible comb filtering is introduced.

Another noteworthy aspect of phase, timing and polarity is you cannot compensate a time delay with any amount of phase shift. Time delay represents an infinite number of phase shifts at an infinite range of frequencies. Similarly, a waveform cannot be truly out of phase unless it's a single frequency (think sine or cosine wave). Therefore, the terms "reverse polarity" and "180 degrees out of phase" are not interchangeable when discussing music waveforms.

Proper understanding of the time domain finds that equalization introduces time smear .

Music Reproduction: It's all in the Timing

The stereo effect is a fragile illusion based on the interference of the acoustic arrivals at the left and right ear. Timing and amplitude differences over the audible frequency bandwidth for the arrival to each ear is the basis of the head related transfer function (HRTF).

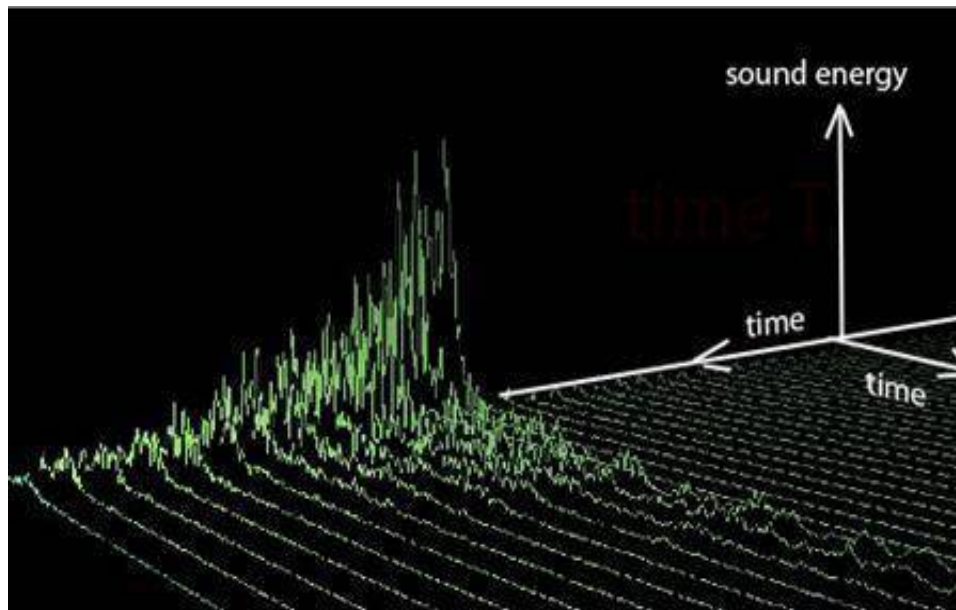
The brain has an uncanny ability to process and triangulate acoustic source locations from these combined arrivals. However, this is not the only way that we localize sounds. In fact, completely covering one ear does not prevent you from localizing sound. Give it a shot.

However, the perception of sound from only one ear will radically reduce one's ability to map out the acoustic environment and will make speech seemingly less articulate in even a moderately reverberant environment. It is key to have sound arrive to both ears for the brain to selectively focus on a source and while working to subjectively reduce the masking background noise.

Stereo Unfold Methodology

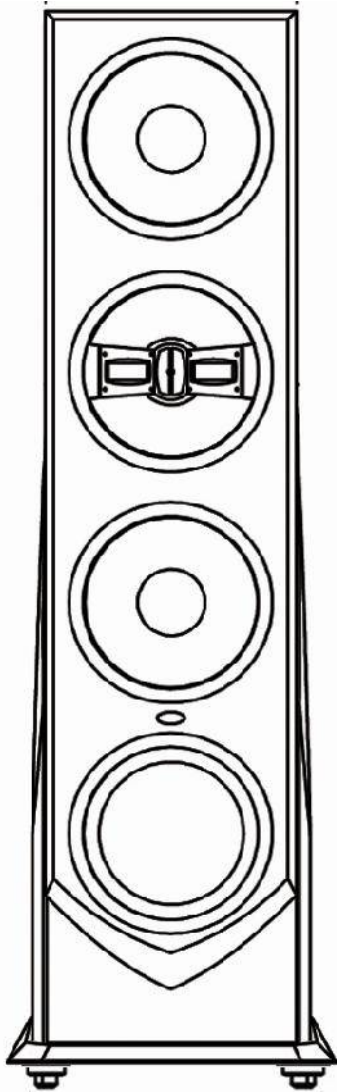
The 3D measurement below allows us to view how a particular transient (in this case a simple clap of hands) decays over time within the recording environment. Shown on the axis pointing left is the composite energy that is the direct plus diffuse energy. The diffuse energy of the recording environment is shown isolated on the axis pointing right. Observe that for each instant there is a unique tail of reverberant information for this recording channel.

STEREO UNFOLD separates the diffuse energy of each stereo channel and restores it to the natural level and time relationship within the particular listening room. This allows the direct energy to be articulated more clearly and the directional vectors to be analyzed by the brain as matrixed in the left, right arrivals.



New STEREO UNFOLD Technology™ advances the art of music reproduction by restoring the natural timing and level relationships of direct and late arrivals. The spectral balance of ambient information is returned to a natural level while early reflections that cloud spatial information are minimized. This is made possible by first reducing the masking effect the listening room has on the actual recording environment by applying the Bohmer Room Correction which realigns acoustic arrivals to the listeners in the time domain. STEREO UNFOLD Technology then examines and identifies the related ambient components following the initial wave-front and restores this diffuse field information to the proper level in your listening room.

Specifications



Application:	Sequential sound field reconstruction system, auto-setup with calibrated microphone
System Type:	8 Driver, 4 way system with specialized 3 driver ambient array
Tweeter:	Dual 4" AMT bridge-mounted in post convergent array
Midrange:	1.5" coaxial, titanium/polyester diaphragm, precision waveguide
Midwoofer:	14" carbon/pulp curvilinear cone, neo motor, dipolar
Bass:	Dual 14" carbon/pulp curvilinear cone, neo motor in super cardioid array
Subwoofer:	Dual 12" aluminum diaphragms, 480oz motors, cast frame, 3" dual 4 layer voice coils
Passive Radiator:	Dual 12" patented symmetrically loaded with 2" travel
Low Freq. Align:	Hybrid cardioid pattern, dual rear radiators, down-firing sub
Inputs:	1 pr binding posts for upper range, 2 XLR balanced for bass, sub, 1 XLR for STEREO UNFOLD
Internal Amp:	Subs- 1kW, Bass- 750W, Mid- 500W, Ambient Array- 500W
Recom. Amp:	1 external channel of 60 watts or greater required for high frequencies
Freq. Response:	12Hz-30kHz
Impedance:	XLR 10k, binding posts 4 ohms
Sensitivity:	100.5 dB (2.83V @1m)
Crossover:	65, 800, 6k
Cabinet Size:	Cabinet 67" H x 16.25" W x 18" D, Base 1.5" H x 20.75" W x 20.75" D
Weight:	288 lbs each

Wavelet

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 (optical/TosLink, coaxial): 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 96kHz to “No Change”. Then set input 176,400 to output 88,000, 192,000 to 96,000 and greater than 192,000 to 96,000. 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.



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.





Wavelet Preamplifier/DAC/Crossover/Room Correction Processor

Inputs

Analog

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

Digital

- Asynchronous USB audio, 24 bit, 44.1 – 96 kHz, PCM up to 352.8 kHz.
- SPDIF, 24 bit, 44.1-96kHz
- TosLink, 24 bit, 44.1-96kHz

Outputs

Analog

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

Digital

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

Control Interface

- Ethernet, TP-Cable & WLAN

Processing

DSP

- Analog Devices, internal processing sample rate 96kHz, 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 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

Optimizing Analog Outputs

When using the analog inputs it is useful to optimize the input levels according to how they are being used. This will maximize the signal to noise ratio and provide optimal gain while preventing distortion from input overload.

Outboard DAC to Wavelet Analog Inputs Using Wavelet as your Preamplifier

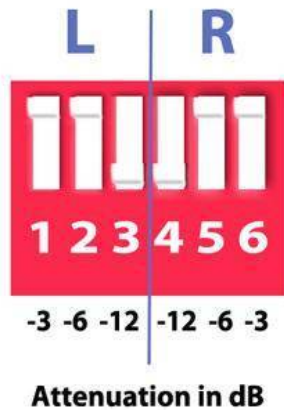
The Wavelet has adequate internal gain for virtually any outboard DAC output. In some cases you may wish to decrease the input sensitivity of the Wavelet to accommodate an external DAC with high output level, especially if you are using one with a volume control. Each stereo pair of analog inputs of the Wavelet can be adjusted downward in -3, -6, or -12dB steps as shown below. The goal is to set the analog channel's input sensitivity so that adequate volume level is achieved from your system without any audible clipping of the inputs. When adjusted properly, the Wavelet's blue front panel display will usually be in the range of 85dB for average listening levels. When using the Wavelet's digital inputs such as the SPDIF from a CD transport, adjustments are not required - thus there are no provisions for such. Some computer hosted media software may introduce an extra level control stage to the USB output. In this case it is best to configure the software settings so a typical listening level is accomplished with Wavelet displaying 85dB.

Outboard Preamp Driving Analog Inputs Using Wavelet for Crossover/Room Correction

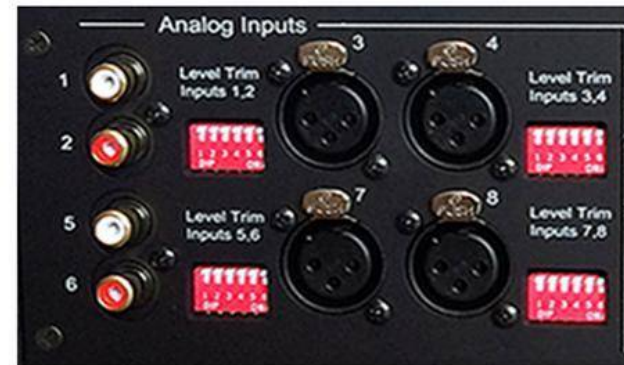
If you are relying on an external preamplifier to provide the both the master gain and level control into the Wavelet, you will need to optimize the inputs sensitivity of the Wavelet's analog inputs. The key is be certain that the highest volume from the preamp does not overload the analog inputs. We recommend beginning

with the -12dB setting initially as shown below. Further attenuation of -3, -6dB is available if needed. While you can also further reduce the volume setting from your preamp, the goal is totally avoid clipping at all levels. Again try to establish a Wavelet level setting of about 85dB for typical listening. With a little care you will have minimized your noise floor and prevented input saturation. Rarely does the Wavelet need to be set above 90dB when optimizing to the volume settings familiar from your preamplifier.

Rear Panel View of Resistive Switches



The settings shown indicate an attenuation of 12dB on both left and right channel inputs for the analog input pair chosen.



Ready to *take control* of your VALOR with Wavelet?



Introduction

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.

What happens if my internet goes out? Don't worry- the Wavelet remote functions continue to work and room correction settings are maintained, even without an internet connection. An internet connection is only necessary for the initial setup and all settings are retained inside of Wavelet- even if the internet is out your system will work as designed.

Wavelet WiFi Internet Setup

1. To connect your Wavelet to the internet, please open a browser on your PC or Mac and visit <http://bohmeraudio.com/setup.html>



Wifi setup

The First step below is only required once to connect your Wavelet to your WiFi network. The Second step is used to find your Wavelet on the network if you're not using a fixed IP address. Perform the second step on the computers, pads and phones you wish to use as remote controls for the Wavelet.

First step

1. Insert the supplied USB memory stick into your computer
2. Enter your WiFi network name (SSID) and your network password in the appropriate fields.
3. **This step is only necessary if you wish to use a fixed IP address for your Wavelet.** Tick the box "Fixed IP address" to set a fixed IP address in your Wavelet. Enter the fixed IP address you want to use in the field for IP address. If you use a different net mask and gateway than the suggested default values you need to edit these fields as well.
4. Press the "Download wifi-conf.txt file" button to download a "wifi-conf.txt" configuration file.

Connect your Wavelet to your WiFi network

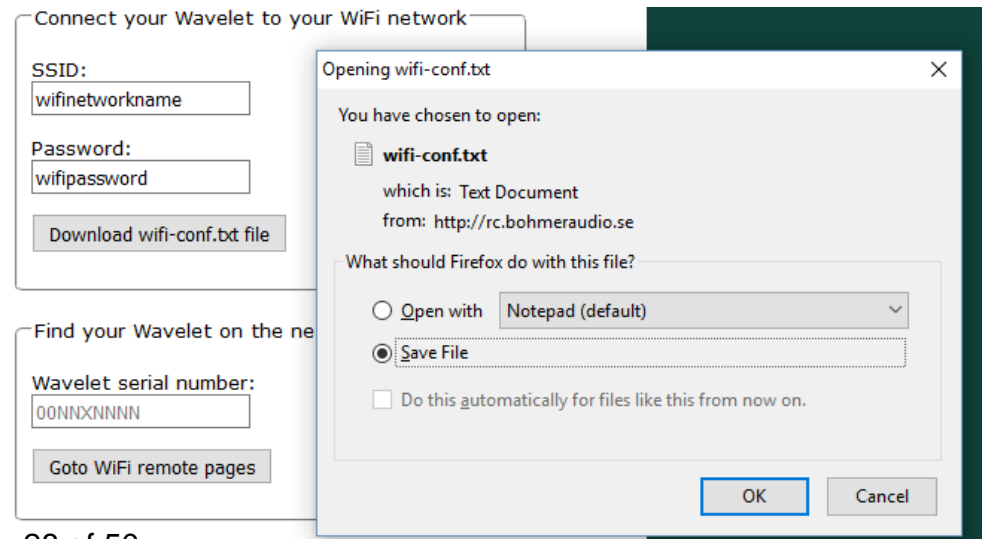
SSID:

Password:

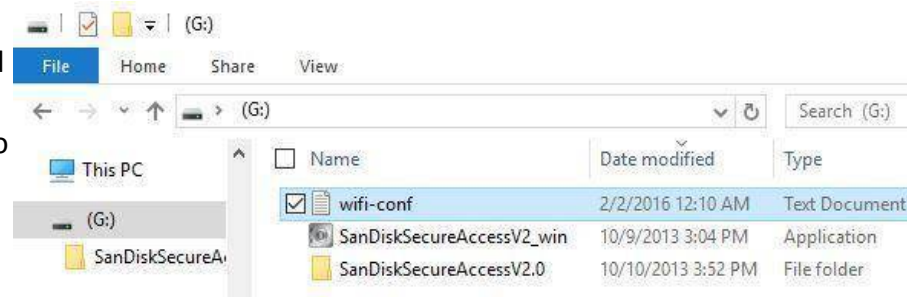
Find your Wavelet on the network

Wavelet serial number:

- a. Enter your SSID- this is your Wifi Network name.
- b. Enter your Wifi Network Password and click "Download wifi-conf.txt file"
- c. Insert the included SanDisk USB memory stick into your computer.
- d. When prompted to "Open" or "Save File" choose "Save File" click "OK".



e. Copy the "wifi-conf.txt" file to the root directory of the SanDisk USB memory stick. Depending on your browser settings, you might find the file downloaded to your Desktop or Downloads folder. After locating it, right click the file and choose copy. Navigate to the SanDisk USB memory stick and paste the file into the main "root directory" so that when you click on the SanDisk USB drive from your computer, the file is visible and *not* placed within another folder. You may now remove the SanDisk USB memory stick from your computer.



- f. Power off your Wavelet. Insert the SanDisk USB memory stick into the port located at the back of Wavelet labeled WLAN USB Stick. Power Wavelet back on and wait 1 minute. The system is saving your wifi network to memory.
- g. After 1 minute, power Wavelet off, remove the SanDisk USB memory stick and insert the WiPi WiFi Stick into the port located at the back of Wavelet labeled WLAN USB Stick.
- h. Turn Wavelet on and wait for 1 minute- confirm that the Blue LED on the WiPi WiFu Stick is lit or lit with occasional flashing.
2. Now we will open the WiFi remote. Please use the device you wish to control Wavelet with- we recommend an iPad or iPhone. Any smart phone or smart device will work.
- First, make sure that your device is connected to the same WiFi network your Wavelet is connected to. (On Apple devices, click Settings, Wi-Fi, and choose the appropriate network)
 - In your internet browser (Safari on Apple devices) enter <http://bohmeraudio.com/setup.html> in the address bar.
 - Your Wavelet Serial Number is located on the back of the Wavelet, on a sticker near the USB port. Enter it into the box labeled "Wavelet serial number" without any spaces and with capitalized letters. Click ok and click "Goto WiFi remote pages"

**-Volume
-Balance**



- d. You will now see the Volume Control Page- please bookmark this page for future use. The bookmark button on Apple devices is the square and arrow at the bottom of the screen that looks like this:



Firefox users can click the Star to bookmark their remote control.

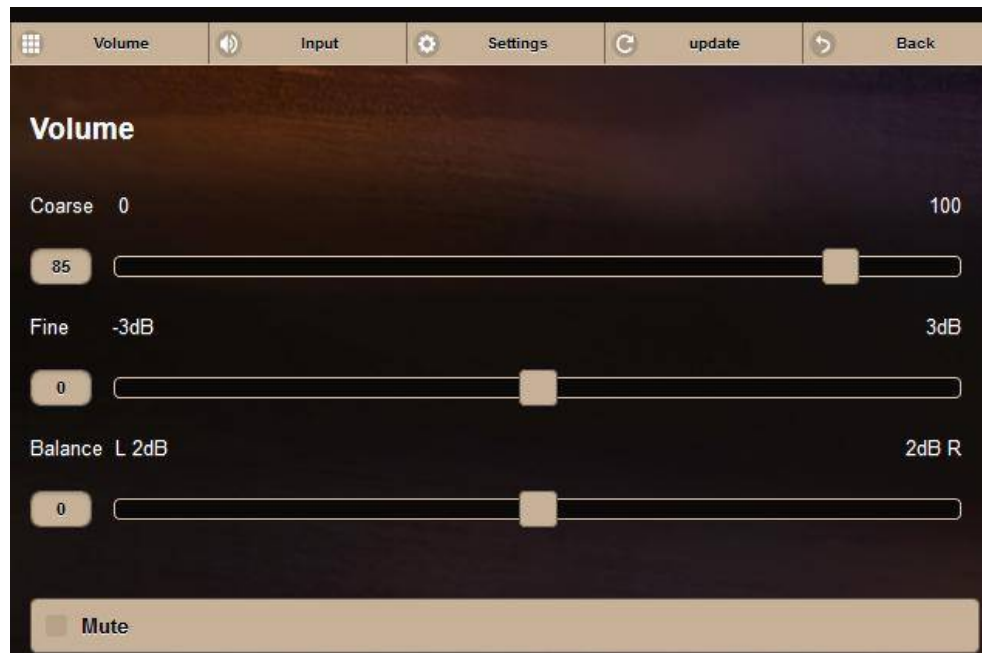
**TIP You can always return to your remote by visiting and entering your serial number <http://bohmeraudio.com/setup.html>
You do NOT need to re-enter your Internet information, only the serial number**

- e. Confirm your device is talking to Wavelet by adjusting the volume slider from your device and watch the Wavelet volume change remotely. Congratulations! You can also control your Wavelet from additional devices by repeating Step 2 for each device you wish to use.

Volume Adjustments Before Running Room Correction on Your Wavelet

1. To prepare for running room correction, please set the Wavelet volume to 85. The “Fine” slider is useful for making precise adjustments. Touch the “Fine” slider and move it to the far right- notice when you release, the volume increases by 3dB, and when moved slightly to right, increases by .1 dB increments. You can use the Coarse adjustment is useful for making large adjustments. The “Balance” slider can be used to adjust the center image of your speakers. At the extreme left or right, the slider provides a 2dB boost. Please note that the balance should be be set to 0 for running room correction.

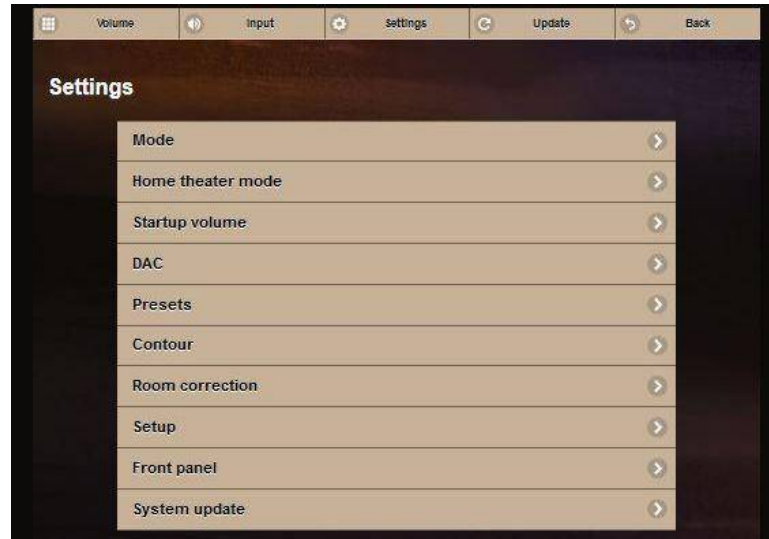
**-Volume
-Balance**



TIP On any page of the Wavelet, click the “update” button at the top to refresh your handheld device and display the settings that are inside Wavelet.

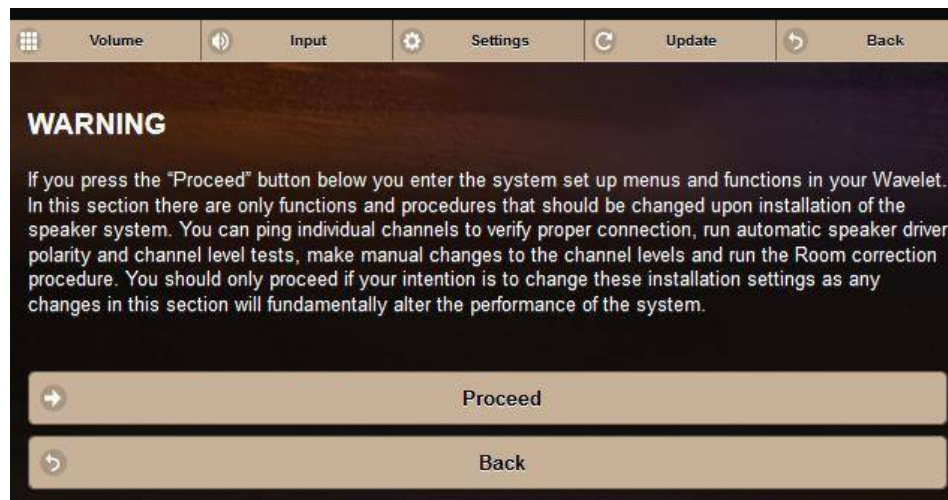
- a. First we will verify that all connections are correct by clicking the “Settings” button at the top. Then choose “Setup” and Click Proceed

-Settings

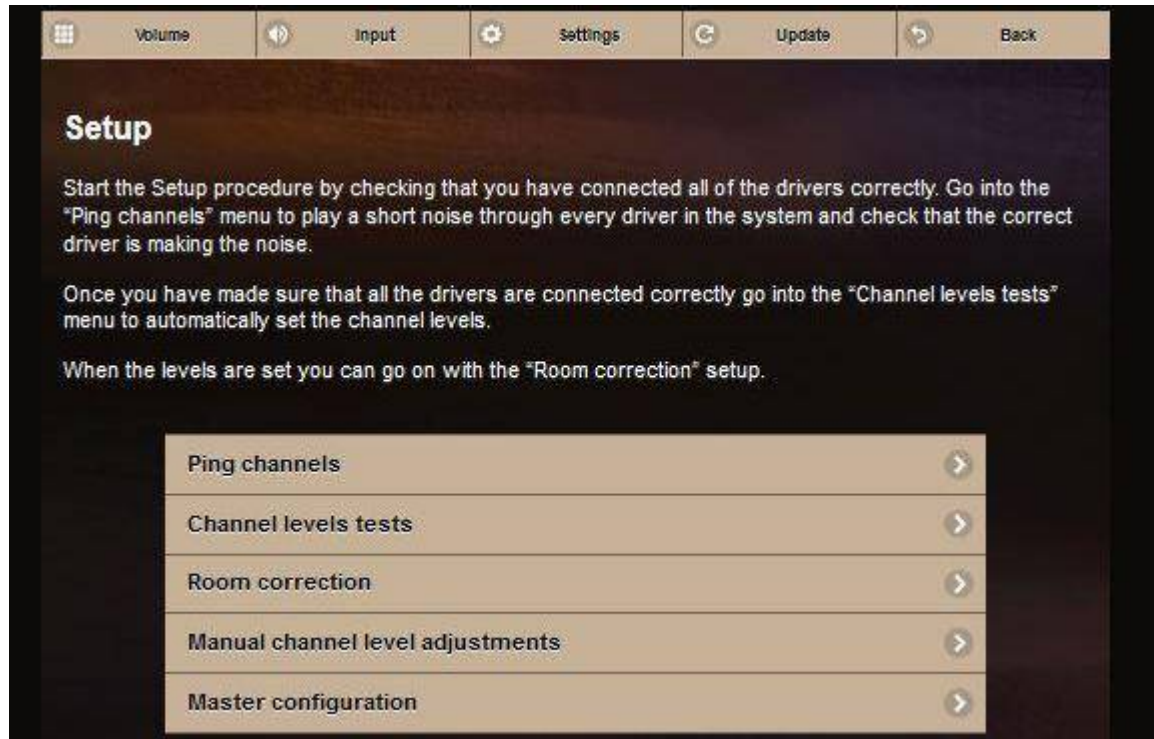


- b. Choose “Setup” and click “Proceed”

-Settings
-Setup

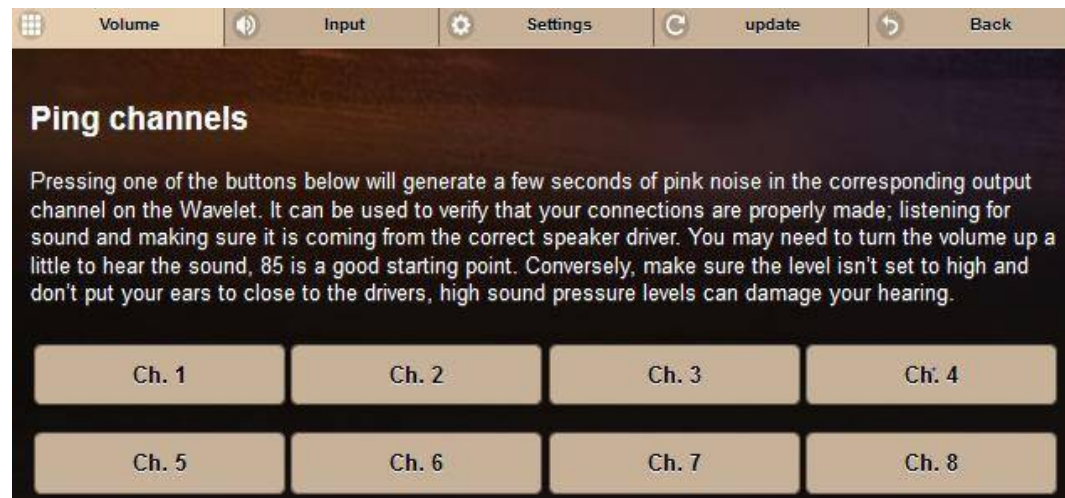


- Settings
- Setup
- Proceed

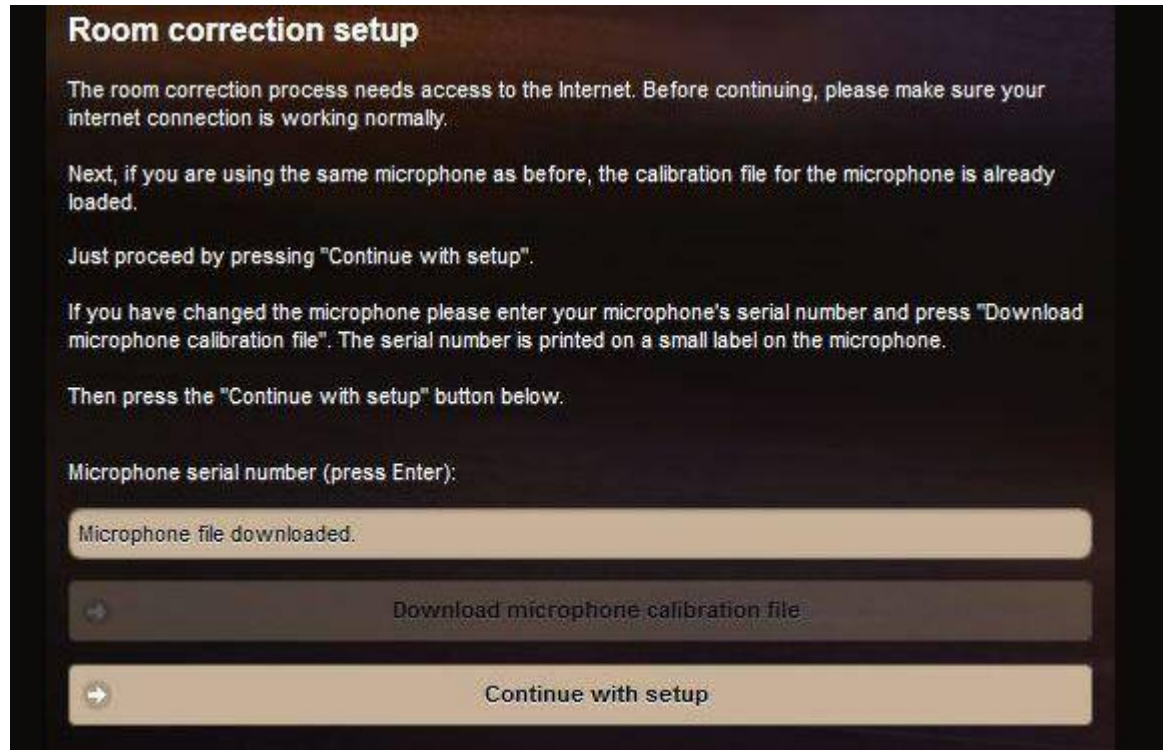


- c. Select "Ping channels." You can now follow the on screen prompt and click the corresponding channels to confirm that sound is coming from the appropriate drivers. Please refer to the connection section of this manual for appropriate channel connections.

- Settings
- Setup
- Proceed
- Ping channels

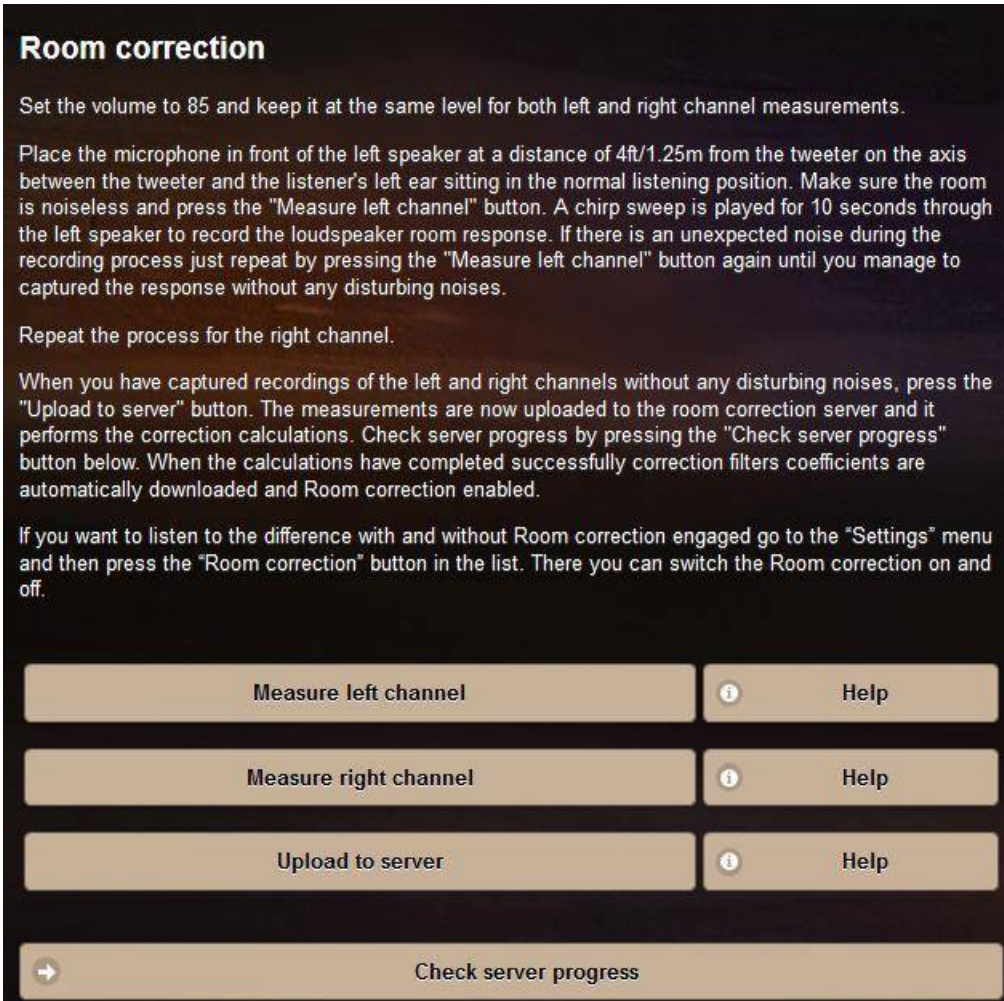


- Settings
- Room Correction



- Your microphone has a serial number on it, please click Settings, Room Correction, enter it into the Wavelet app and click "Download microphone calibration file".
- Once it is downloaded, click Continue with Setup. (Note, if you have already downloaded the microphone calibration file, you will *not* need to download it again.)

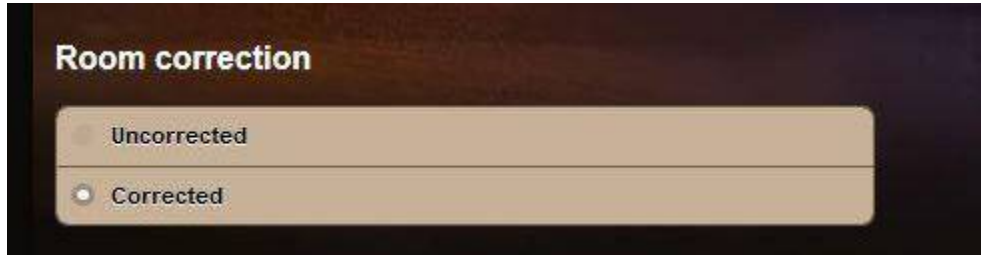
- Settings
- Room correction



2. Click settings at the top of the screen, and choose "Room correction".
 - a. With the microphone at tweeter center height, 48" away from the left speaker, perpendicular to the speaker baffle, choose "Measure left channel".
 - b. Please wait for the process which lasts about 10 seconds. If there is an unexpected noise, please repeat the process by clicking "Measure left channel" again.
 - c. Look at the bottom of the screen- if you see an error such as "Volume too low," please check the connections and/or increase the level and try again.
 - d. This process may take 5-10 minutes. Please be patient, you will be prompted when the process is complete.
 - e. If you receive an error message that the levels are too low, please check that both ends of the microphone cable are firmly docked, and repeat the process, raising the volume if necessary.
 - f. Repeat steps a-e for the right channel
 - g. Click "Upload to server"- this sends the measurements to the room correction device which calculates the ideal settings for your speakers and automatically sends them back to your Wavelet. You can observe progress at the bottom of the screen. This process may take 5-10 minutes. You will be prompted when the process is complete.

- f. Upon completion, click the “Back” button at the top. Confirm that Room Correction is set to “Corrected” You are now ready to enjoy your Wavelet system! You can turn the room correction on and off during your listening. Access the Room Correction by clicking "Settings" at the top of the screen, and choosing "Room Correction"

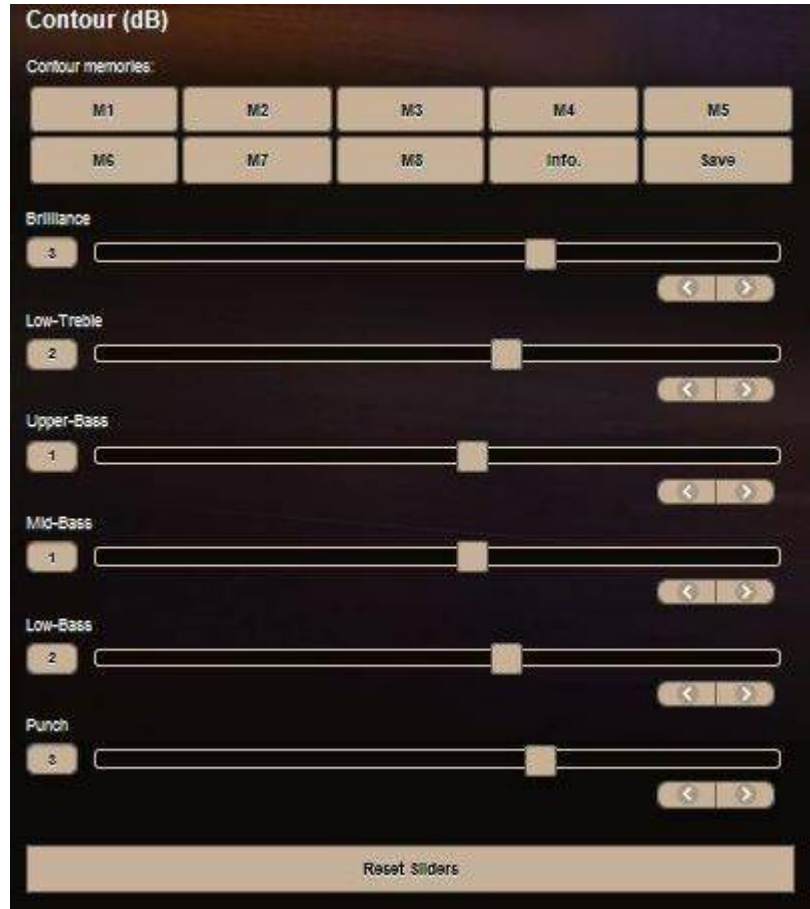
-Settings
-Room correction



Adjusting your Wavelet

After room correction, you can further tailor the sound of your Legacy speakers in your room via the “Contour” page. Access this page by clicking “Settings” and choosing “Contour”.

-Settings
-Contour



The sliders can be adjusted in the same way as the volume control- by placing a finger on the center slider. Moving to the right creates a boost (more volume) and moving to the left creates a cut (less volume) in the given frequency band. A handy “Reset Sliders” button is available at the bottom to reset the Contour Sliders.

Individual tonal preferences can vary with mastering engineers as well as listener preference. For this reason it is great to have the flexible control over voicing of the playback system that Wavelet provides. The Wavelet's carefully designed contour controls allow gentle response shelving without phasiness or ringing.

Brilliance: controls the "air" and definition of a recording above 10kHz

Low-Treble: adjusts the brightness or forwardness above 3kHz

Upper-Bass: adjusts the fullness or bloom of vocals, cello, etc. below 300 Hz

Mid-Bass: determines the apparent speed of decay of bass frequencies.
Reducing will tighten, slight boosts will warm below 150 Hz

Low-Bass: adjusts the overall weight or heaviness below 75 Hz

Punch: controls the drive or impact felt from the rhythm at 55 Hz

TIPS

1. Try a boost of +2.5 dB in the brilliance, a low treble setting of -0.5 dB, a low-bass setting of +2 dB with the punch slider set at +2.5 dB. Now adjust mid-bass by ear until it seems most natural without excessive thickness.

2. Want a tube-like warmth? Manually increase the output on channels 2 and 6 by 1.2 dB and then trim the brilliance contour to -1.0 dB. Fine tune the depth by adjusting the low-treble contour.

The most recent slider settings are retained permanently in the Wavelet memory- even if you turn the unit off, Wavelet will remember your settings.

You can also adjust the output levels of the individual channels.

- Settings
- Setup
- Manual channel level adjustments



Input Selections

You can wirelessly switch between inputs on your Wavelet by selecting “Input” at the top and choosing your desired source. USB allows Wavelet to connect to a computer and play back high resolution audio. USB, SPDIF and Toslink allow for digital connectivity and reduces the need for an extra AD conversion- a valuable performance advantage.

4 pairs of stereo analog inputs allow Wavelet to easily interface with your preamplifier, transport or other devices- making it both flexible and letting you retain all of the color you might enjoy from your gear- tube, solid state or digital.



-Input

Mode & DAC Adjustments

Wavelet allows you to listen in Stereo, Mono and Inverted modes. You can access these settings by choosing “Settings” at the top and selecting “Mode”



-Settings
-Mode



-Settings
-DAC

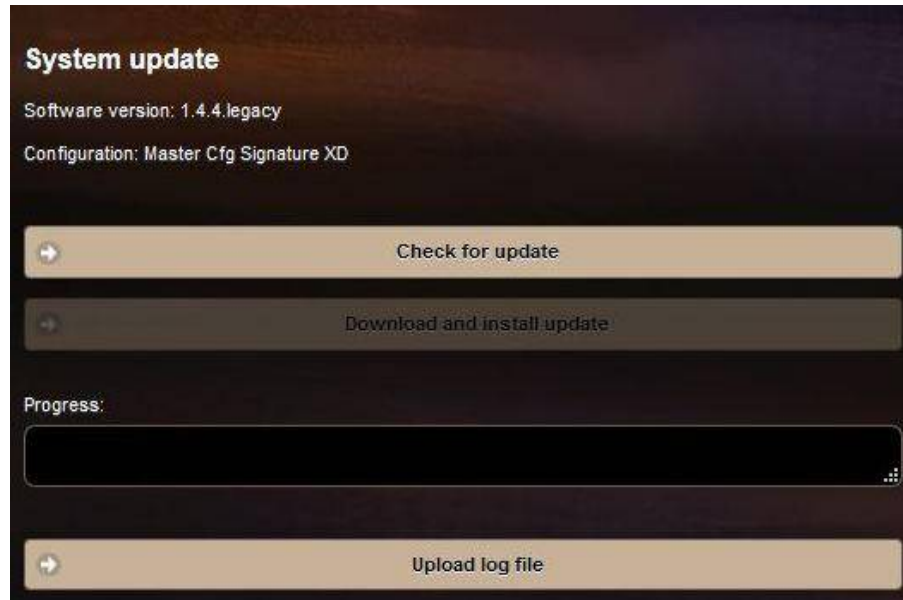
The Wavelet features an apodizing circuit that corrects for the pre-ringing native to CODECs. Analog lovers can take advantage of balanced XLR or unbalanced RCA inputs without concern of digital artifacts. Turn on the apodization circuit by choosing “Settings” at the top, clicking “DAC” and selecting “Apodizing”. “Linear” bypasses the apodization circuit.

Updating the Wavelet System

Wavelet is, by design, easily updated via the app. This allows the device to be updated with new added features as they are developed.

To check for updates, choose “Settings” at the top, select “System update” and click “Check for update”

- Settings
- System update



If an update is available, you can click “Download and install update”. If the system is updated, it will display “System is up to date.”

Presets

- Settings
- Presets



Wavelet with Valor features Stereo Unfold technology that can be activated by clicking Presets and choosing "Restored."

SUT works by unmasking the material that is already in the recording (whether it is an analog or digital signal). It does not add anything to the material- instead it allows your ear to hear more of the information that already exists in the recording. You can disengage Stereo Unfold by choosing "Standard" to compare the difference.

For more Stereo Unfold information, please [click here](#).

Updating the Wavelet System

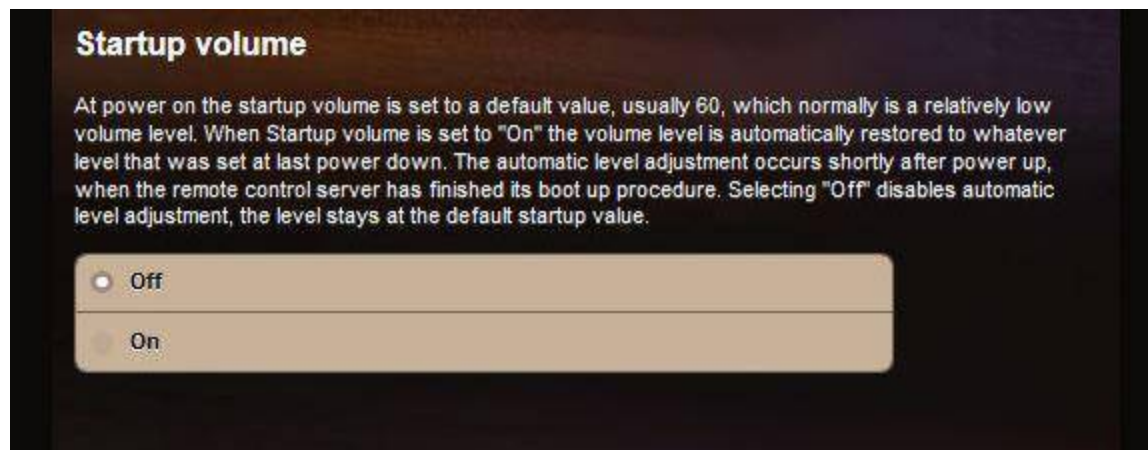
To adjust the brightness on the front panel of your Wavelet, choose "Settings" at the top, select "Front panel" and adjust the slider left or right.

-Settings
-Front panel

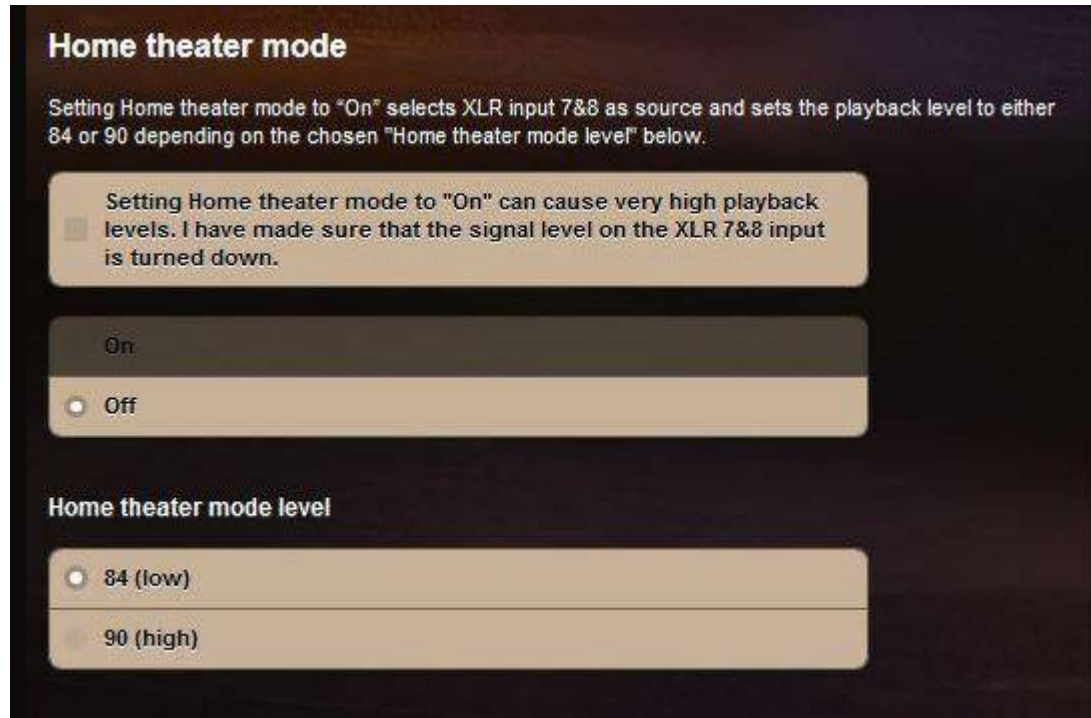


At power on the startup volume is set to a default value, usually 60. If you would like Wavelet to recall the last volume level you listened at before turning the power off, please switch "Startup volume" to "On".

-Settings
-Startup volume



- Settings
- Home theater mode



Wavelet can be easily used in a home theater (and simultaneously a 2 channel) system, and you can take advantage of the Wavelet's room correction for your left and right speakers in your home theater system, just like you do during stereo playback. Turning Home Theater Mode on will bypass the Wavelet volume and allow you to control the overall volume of your surround sound system from your receiver (or preamplifier).

Achieving Ideal Imaging In The Room

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



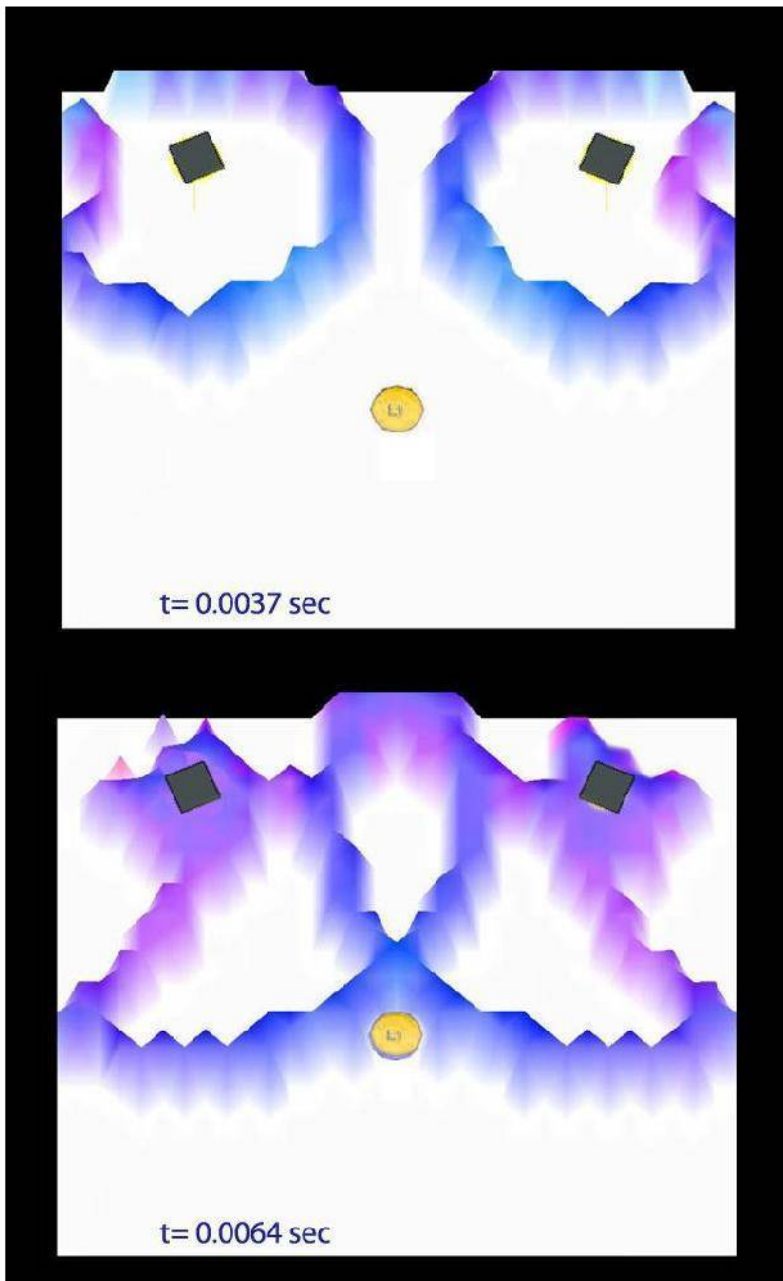
differences to each ear (ILD) and cues for the localization of this sound. If the speaker system projects sound onto the room boundaries creating a diffuse field before it can reach the listener directly, these fragile ITD and ILD cues will be altered and scrambled.

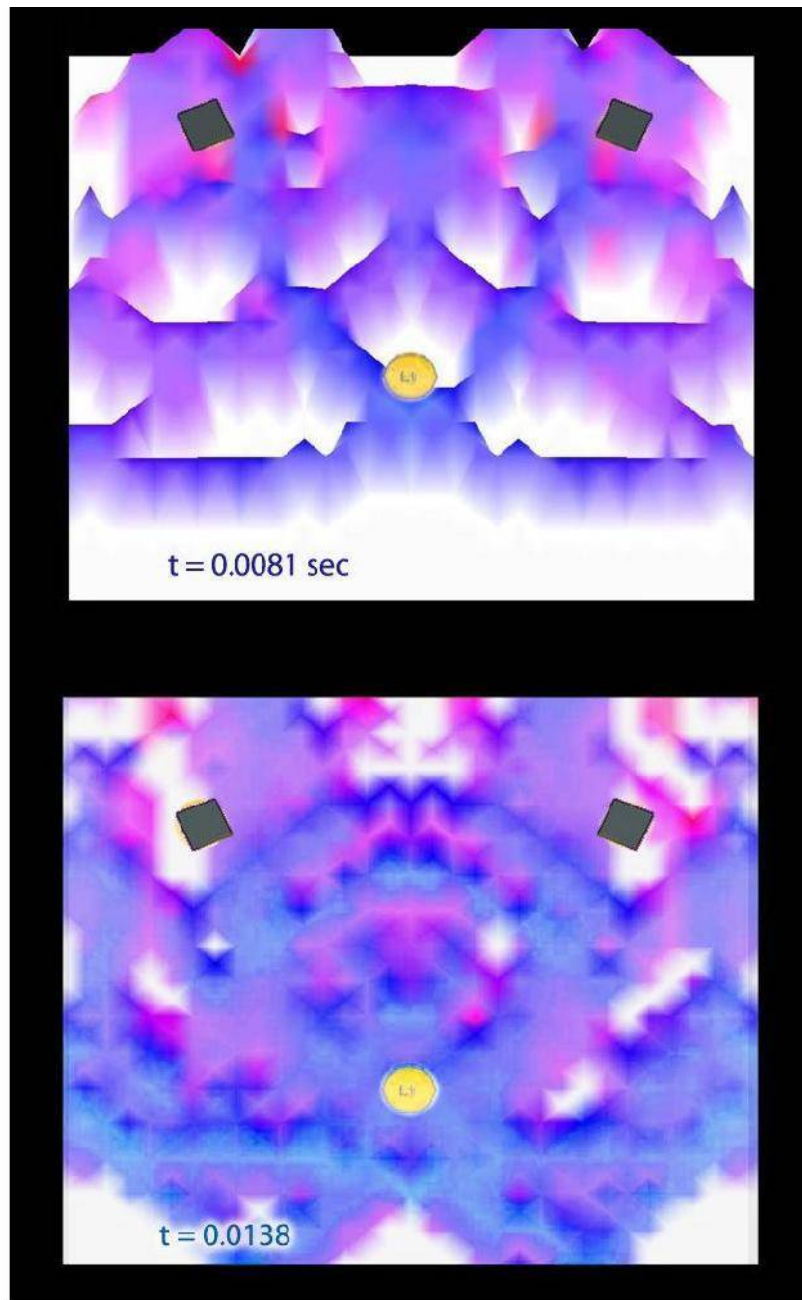
The key to imaging is preserving the interaural time and level differences. The shadowing to the far ear from the source is quite fragile in a reflective environment. Maintaining directivity in the speaker radiation pattern is essential to preserve this content. Legacy's VALOR, V, Whisper, and Aeris designs 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.





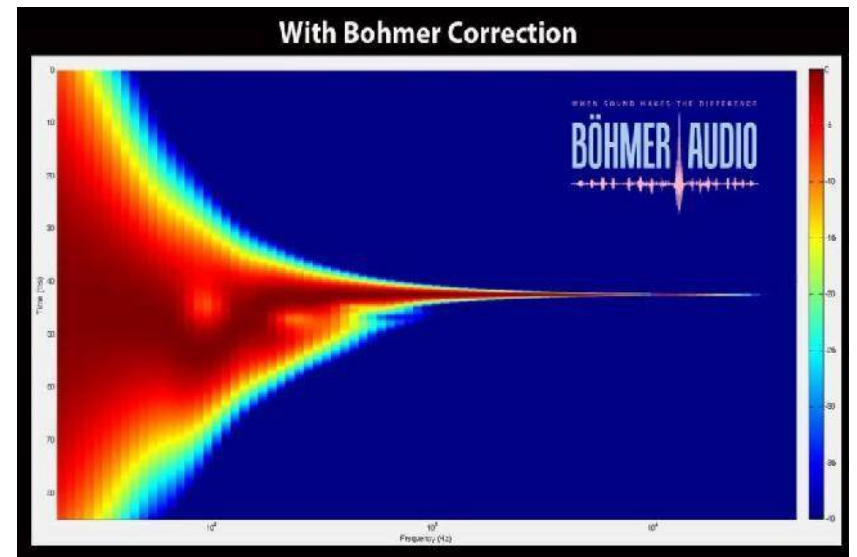
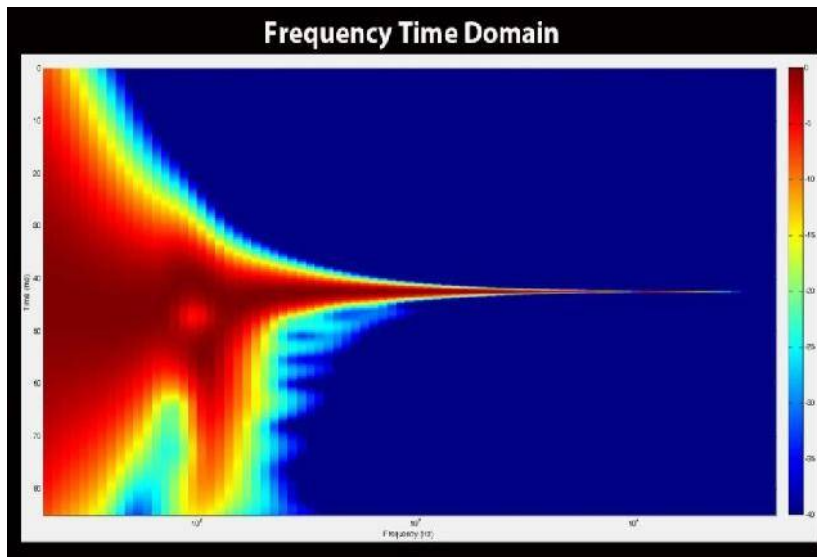
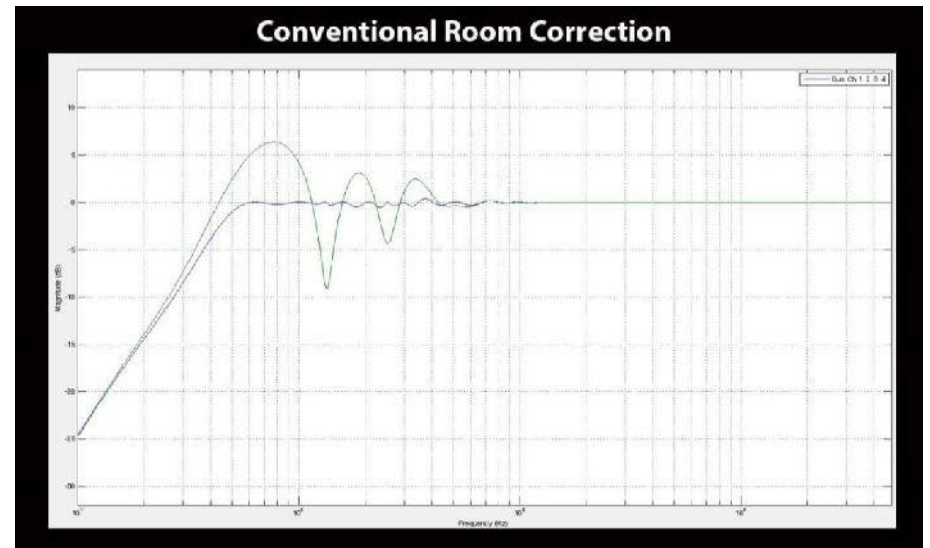
After less than $1/100^{\text{th}}$ of a second, the room has developed a complex wave pattern with energy varying with room position. The listener is now awash in a series of wave-fronts which will soon reflect off the wall behind.

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 Frequency-Time domain. Below right is the same wavelet plot with the Bohmer correction method.



CE Declaration of Conformity

Legacy Audio

3023 E. Sangamon Ave.
Springfield, IL 62702 USA
800-283-4644

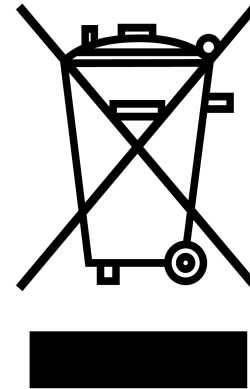
States that this product is in conformity with the with the essential requirements and other relevant provisions of:

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



All information contained in this manual is accurate to the best of our knowledge at the time of publication. In keeping with our policy of ongoing product improvement, we reserve the right to make changes to the design and features of our products without prior notice.

WEEE Compliance



Product Disposal -
Certain international, national and/or local laws and/or regulations may apply regarding the disposal of this product. For further detailed information, please contact the retailer where you purchased this product or the Legacy Audio Distributor in your country. A listing of Legacy Audio Distributors can be found on the Legacy Audio website www.legacyaudio.com or by contacting Legacy Audio at: 3023 E. Sangamon Ave., Springfield, IL 62702, USA—Phone: +1 217 544-3178.

Notes:



©2019 Legacy Audio
3023 E Sangamon Ave.
Springfield, IL 62702
Phone: 800-283-4644
Fax: 217-544-1483