



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](https://www.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.

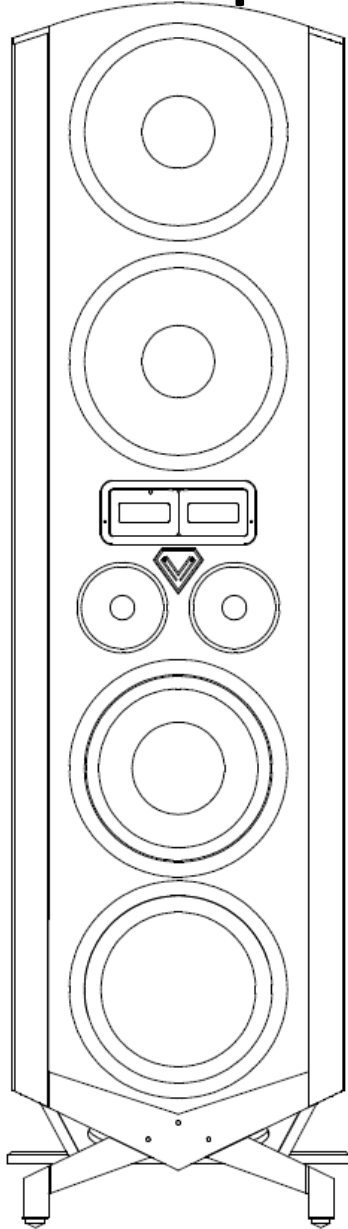




Inside the Wavelet processor carton packaging you'll find a hard-case containing a measurement microphone, a 25 ft. XLR cable and a mic calibration plot. At the opposite end of the carton you will find a power supply with a five pin locking connector and its 115 V AC cable, a compact remote volume control, 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.

Tweeter: 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 pattern

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

Subwoofer: 12" Aluminum very low frequency radiator driving three 10" mass loaded pneumatic radiators

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

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

Impedance: 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

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.



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, 44.1-192kHz
- 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 it 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.



Wavelet processor output connections:

- 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)

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

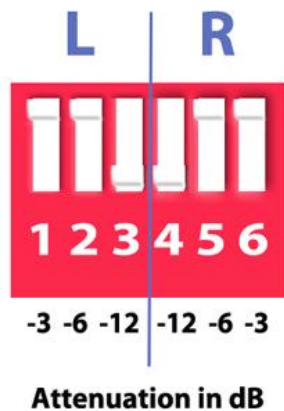
The Wavelet has adequate internal gain for virtually any outboard DAC output. In some cases you may wish to decrease the input sensitivity of the Wavelet to accommodate an external DAC with high output level, especially if you are using one with a volume control. Each stereo pair of analog inputs of 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

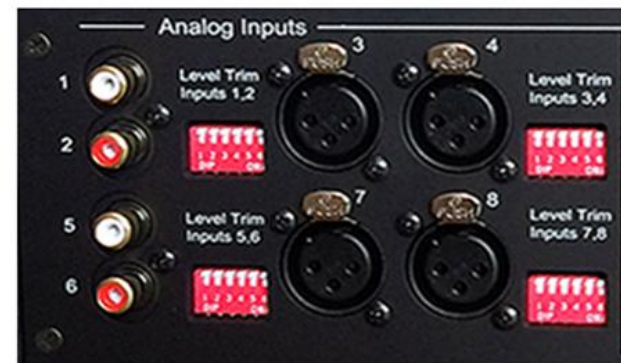
If you are relying on an external preamplifier to provide the both the master gain and level control into the Wavelet, you will need to optimize the inputs sensitivity of the Wavelet's analog inputs. The key is be certain that the highest volume from the preamp does not overload the analog inputs. We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

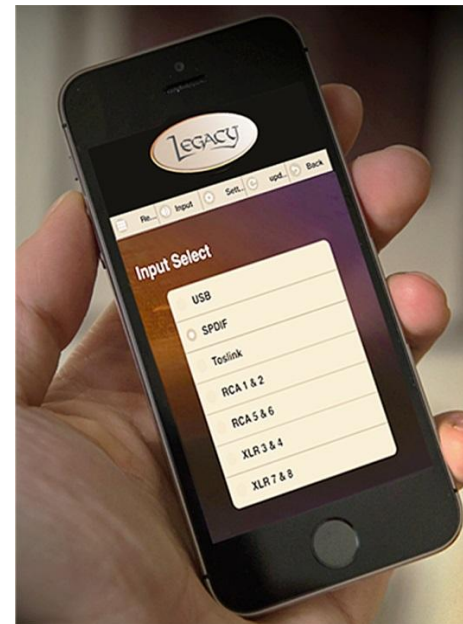
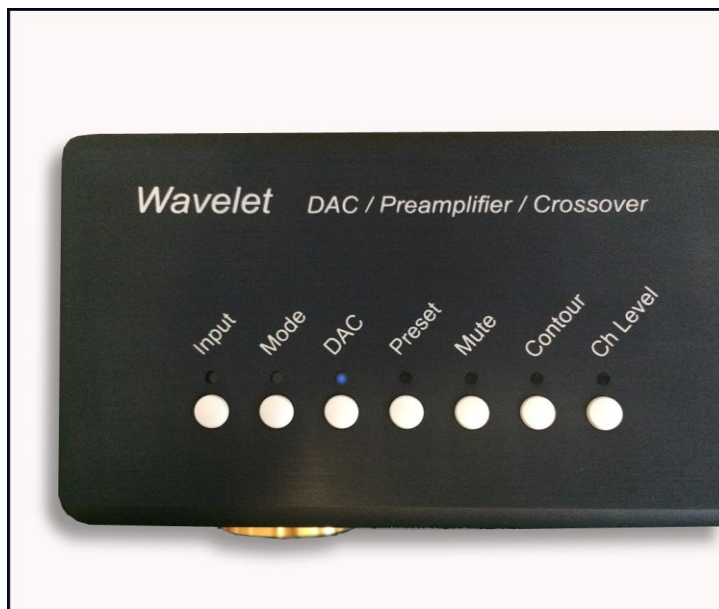
Rear Panel View of Resistive Switches



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 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

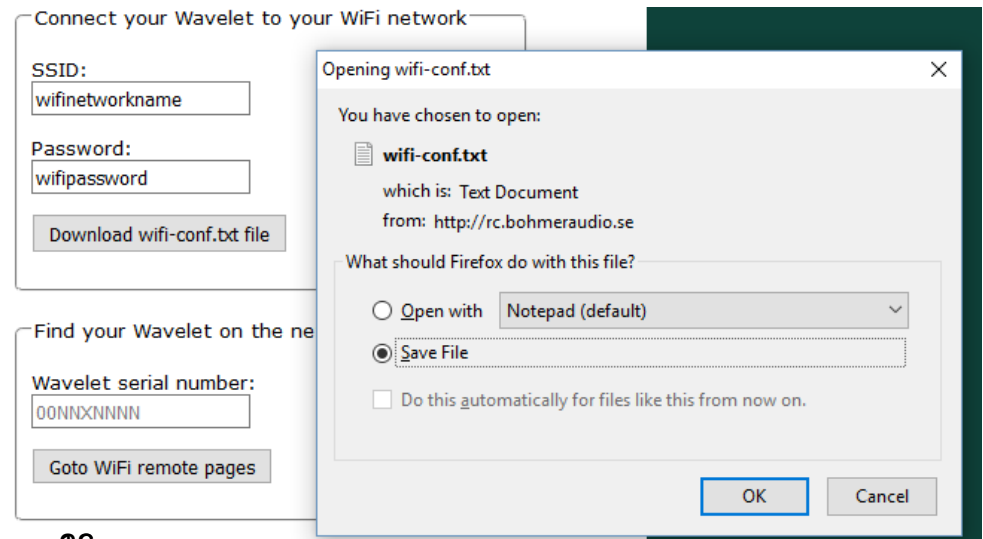
SSID:

Password:

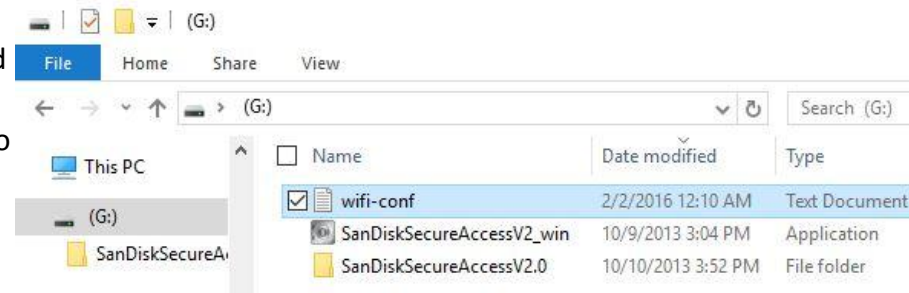
Find your Wavelet on the network

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".

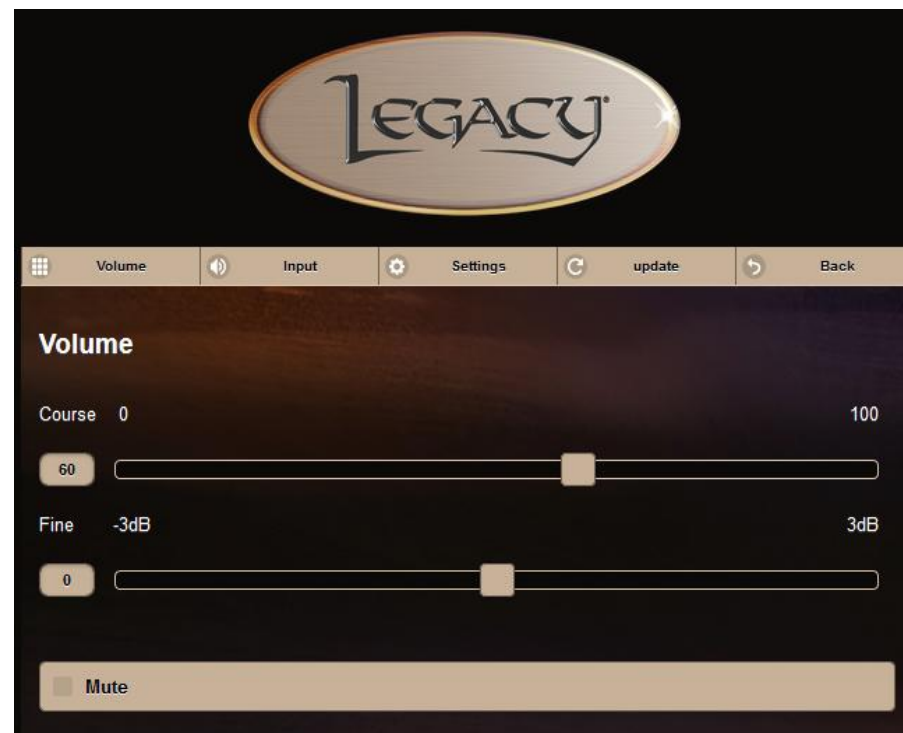


- e. Copy the “wifi-conf.txt” file to the root directory of the SanDisk USB memory stick. Depending on your browser settings, you might find the file downloaded to your Desktop or Downloads folder. After locating it, right click the file and choose copy. Navigate to the SanDisk USB memory stick and paste 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.
 - a. 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)
 - b. In your internet browser (Safari on Apple devices) enter <http://bohmeraudio.com/setup.html> in the address bar.
 - c. 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



- 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.

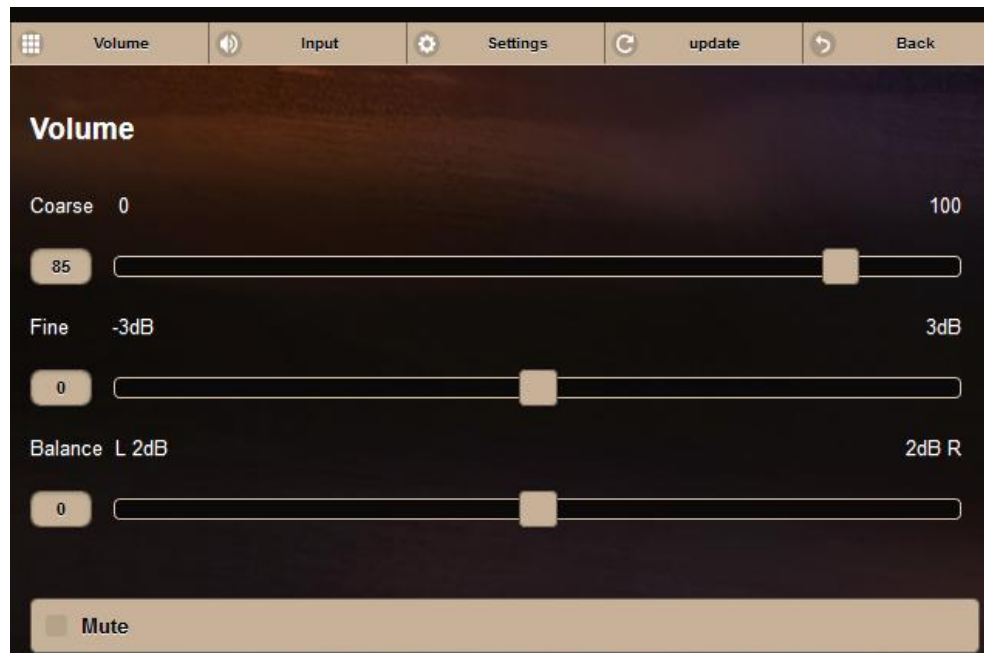
TIP You can always return to your remote by visiting and entering your serial number <http://bohmeraudio.com/setup.html>
You do NOT need to re-enter your Internet information, only the serial number

- 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.

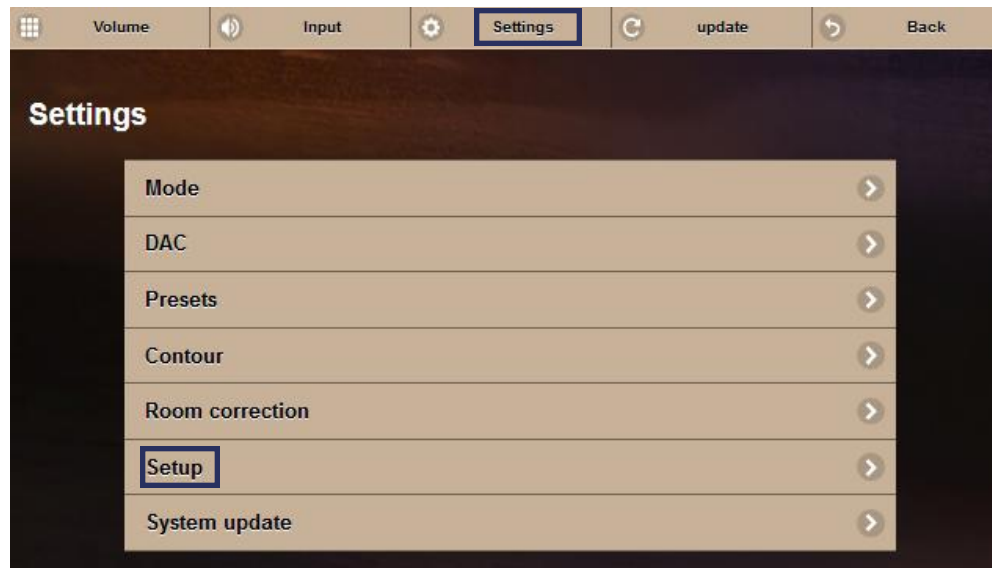
-Volume



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.

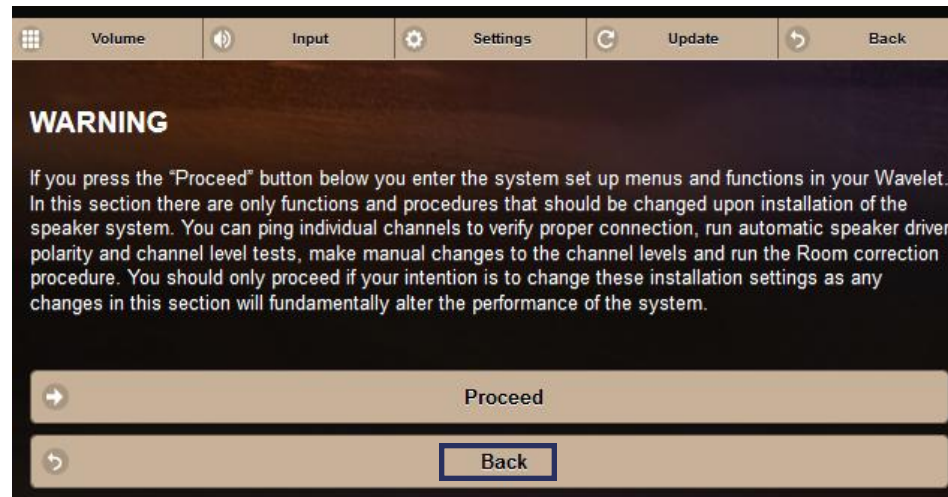
- a. First we will verify that all connections are correct by clicking the “Settings” button at the top. Then choose “Setup” and Click Proceed

-Settings

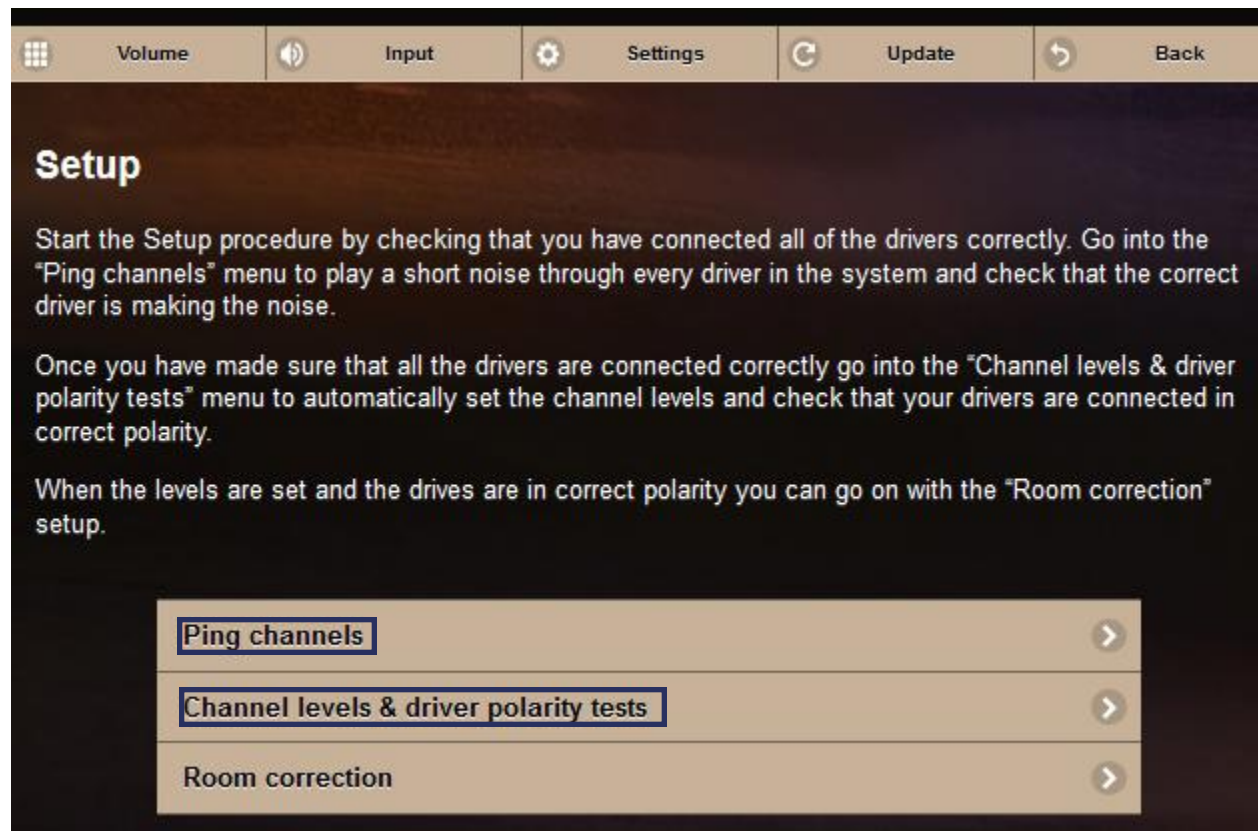


- b. Choose “Setup” and click “Proceed”

-Settings
-Setup

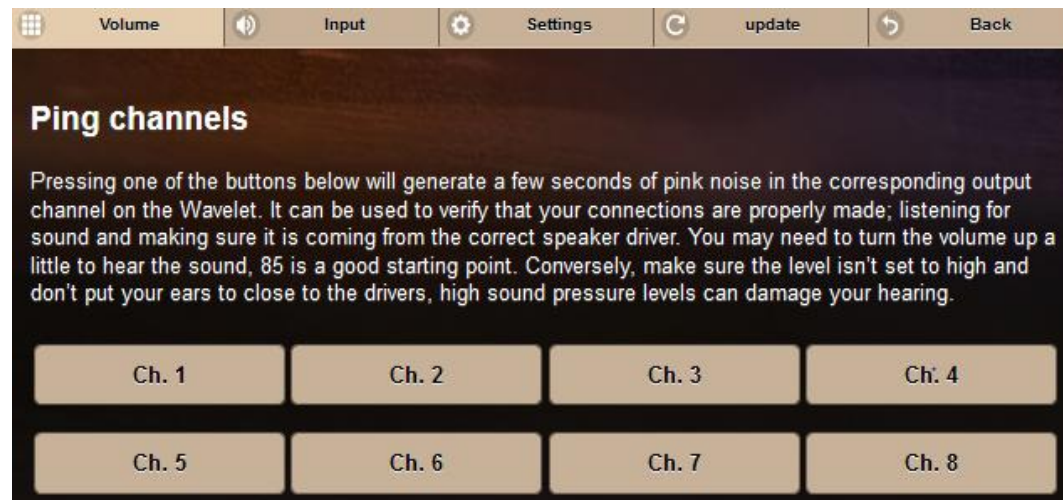


- Settings
- Setup
- Proceed



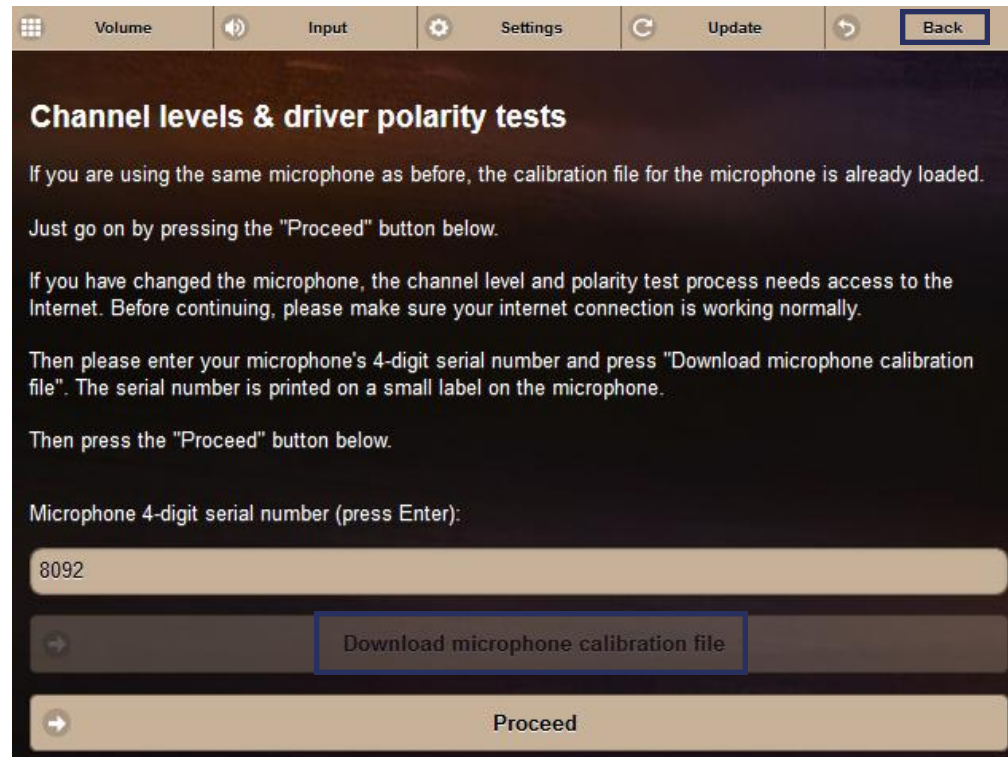
- c. Select "Ping channels." You can now follow the on screen prompt and click the corresponding channels to confirm that sound is coming from the appropriate drivers. Please refer to the connection section of this manual for appropriate channel connections.

- Settings
- Setup
- Proceed
- Ping channels

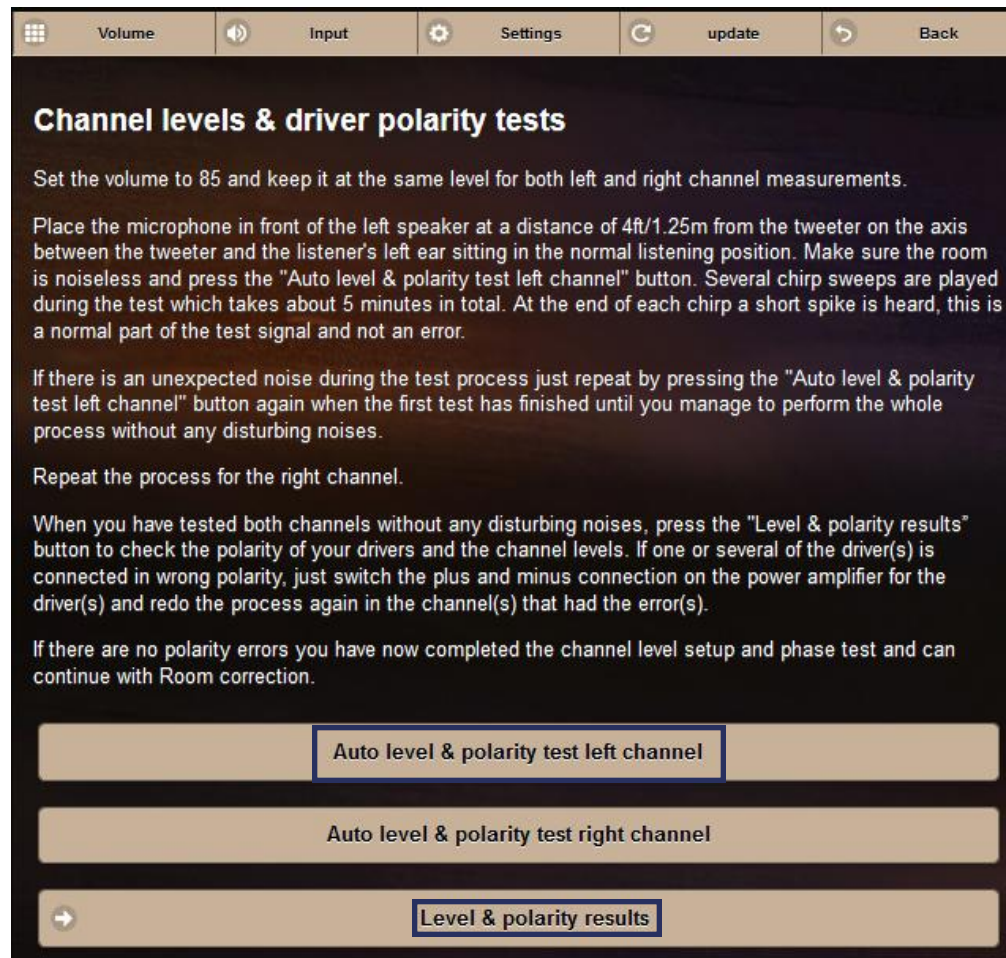


- d. Wavelet is capable of setting the channel levels for you- this is useful for balancing your amplifier(s) with the internal amplifier(s). To do so, click the **back button** and choose **"Channel levels & driver polarity tests"**.
- e. Place the microphone that shipped with Wavelet at tweeter center height, 48" away from the left speaker, perpendicular to the speaker baffle.

-Settings
 -Setup
 -Proceed
-Channel levels & driver polarity tests



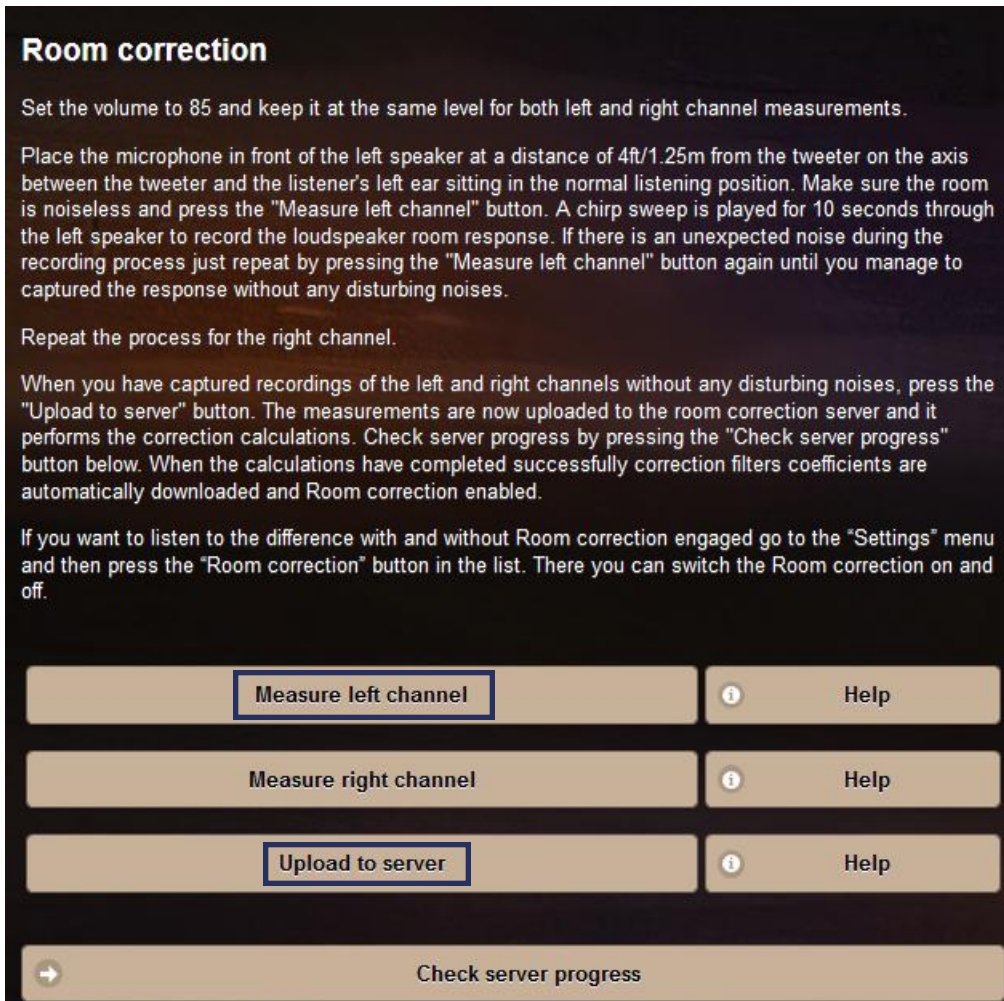
- f. Your microphone has a 4-digit serial number on it, please enter it into the Wavelet app and click **"Download microphone calibration file"**.
- g. Once it is downloaded, click proceed. (Note, if you have already downloaded the microphone calibration file, you will *not* need to download it again.)



- Settings
- Setup
- Proceed
- Channel levels & driver polarity tests

- h. Follow the on screen prompts. With the volume set to 85, select **'Auto level & polarity test left channel.'**
 - i. If you receive a message that the levels are too low, first check that both ends of the mic cable are firmly docked and repeat the test. Increase volume if necessary.
 - j. After completing the left measurements, move the microphone to the right speaker, keep the volume the same, and repeat step i for the right speaker
- Note: this process can take about 5 minutes per channel while Wavelet calculates the settings. Once they are set, Wavelet remembers the settings even after power down, so you do not have to repeat this process.
- k. You can view the results by clicking **"Level & polarity results"**. You have now completed the channel level setup and phase test and can continue with room correction.

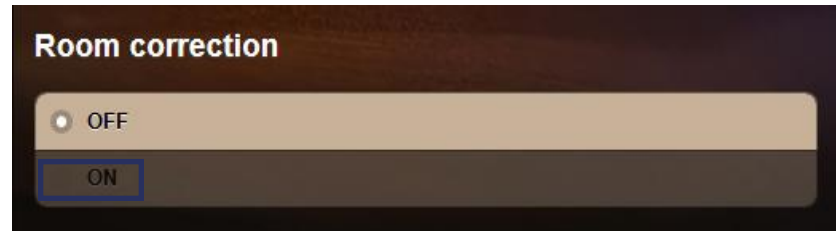
- Settings
- Room correction



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 “ON” 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



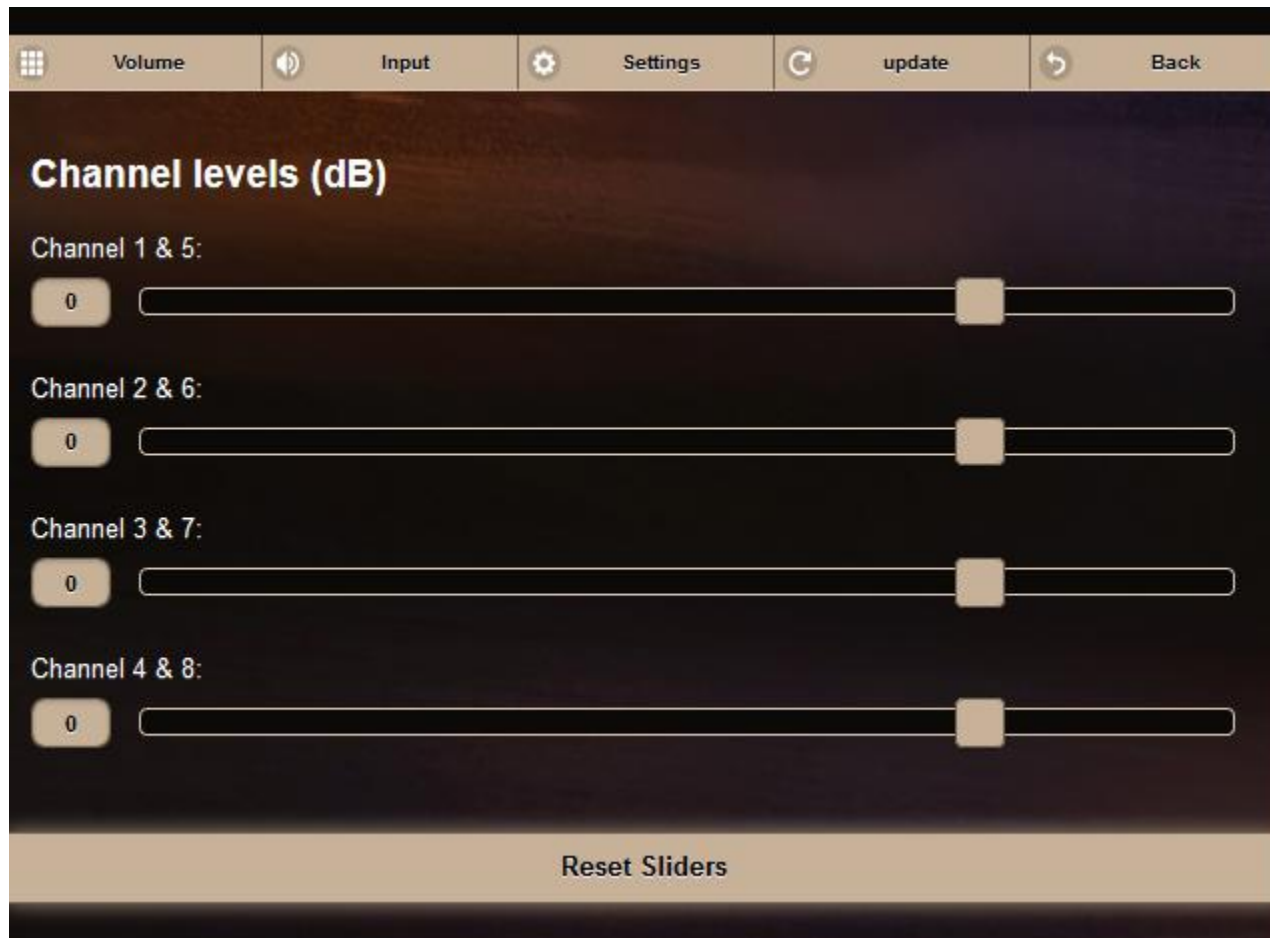
The sliders can be adjusted in the same way as the volume control- by placing a finger on the center slider. Moving to the right creates a boost (more volume) and moving to the left creates a cut (less volume) in the given frequency band. A handy “Reset Sliders” button is available at the bottom to reset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

TIP 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. If you performed automatic channel adjustment (as explained in this manual) you will have settings already applied. You are welcome to further adjust this, if you desire.

TIP This is useful if, for example, you are using stereo Legacy Audio subwoofers with Aeris on channels 3 and 7- you can use this slider to control the volume of your subwoofers!

- Settings
- Setup
- Channel levels



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

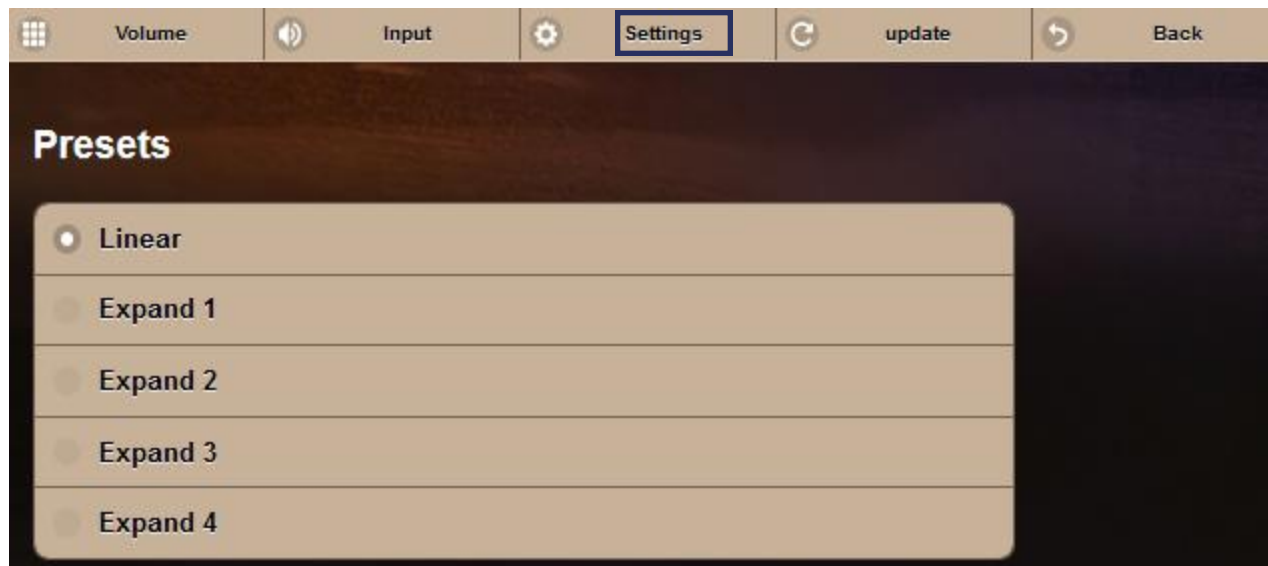


-Settings
-DAC

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.

Presets

-Settings
-Presets

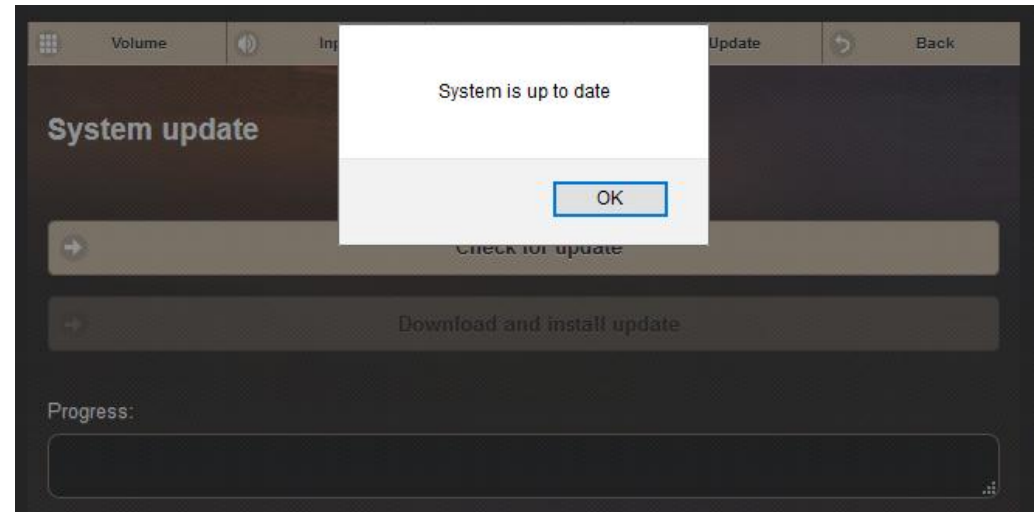
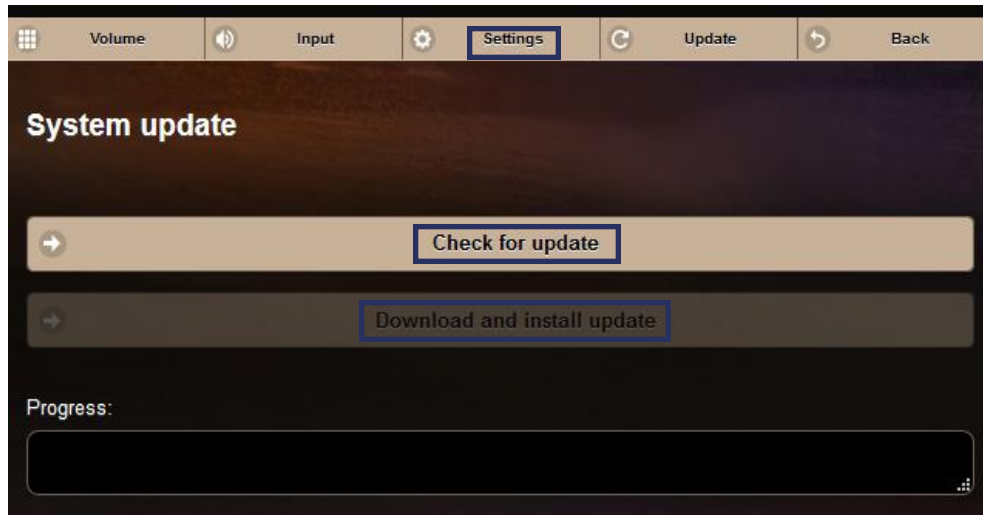


Wavelet features a number of presets to allow you further control of your listening experience. Access these presets by choosing “Setting” at the top of the screen and clicking “Presets”.

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 requested and developed.

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



-Settings
-System update

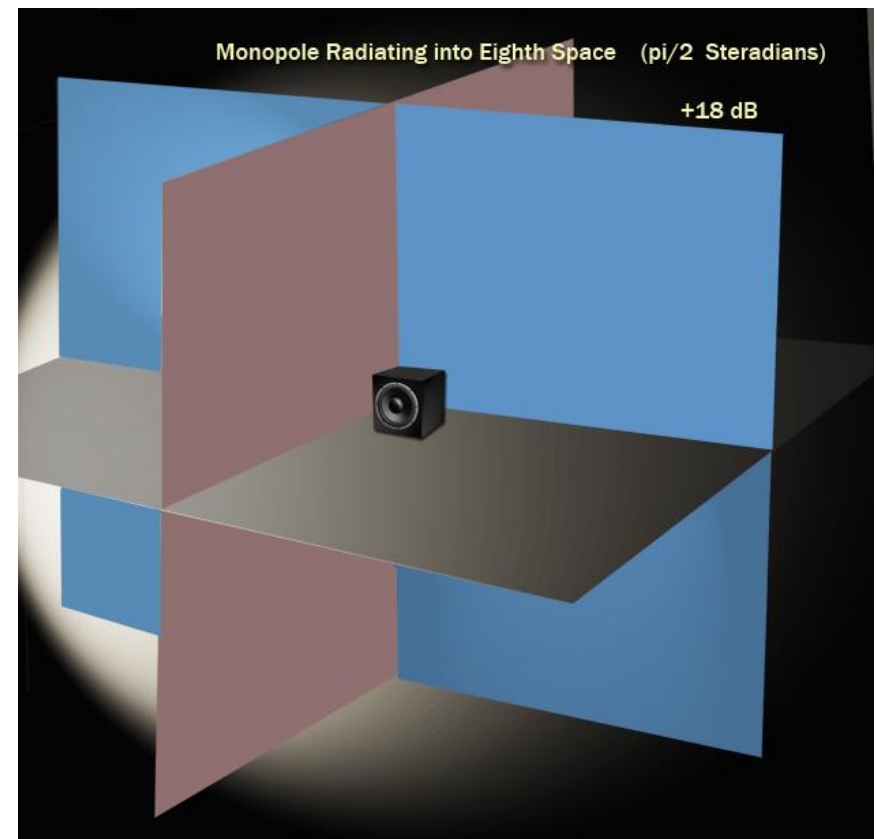
If an update is available, you can click “Download and install update”. If the system is updated, it will display “System is up to date.”

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



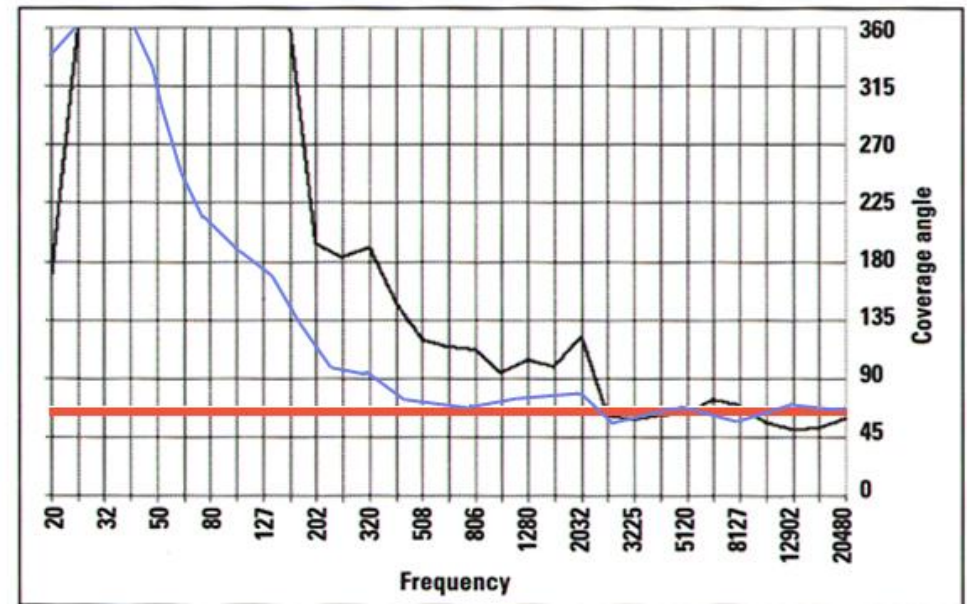
without introducing further damage.

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 omnidirectional. 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.

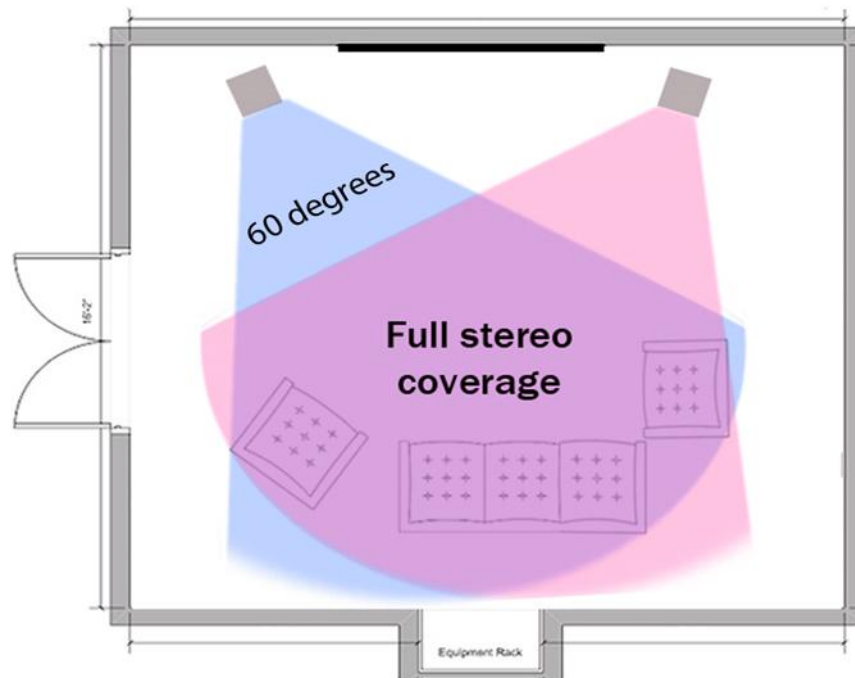
The red line indicates ideal uniform directivity for a very desirable 60 degree horizontal radiation pattern. The black line is typical of a three-way speaker design. The Legacy V system (blue line) employs acoustic directivity control and DSP to reduce lateral room energy and resonances far lower in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

Horizontal Coverage (-6dB)



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.

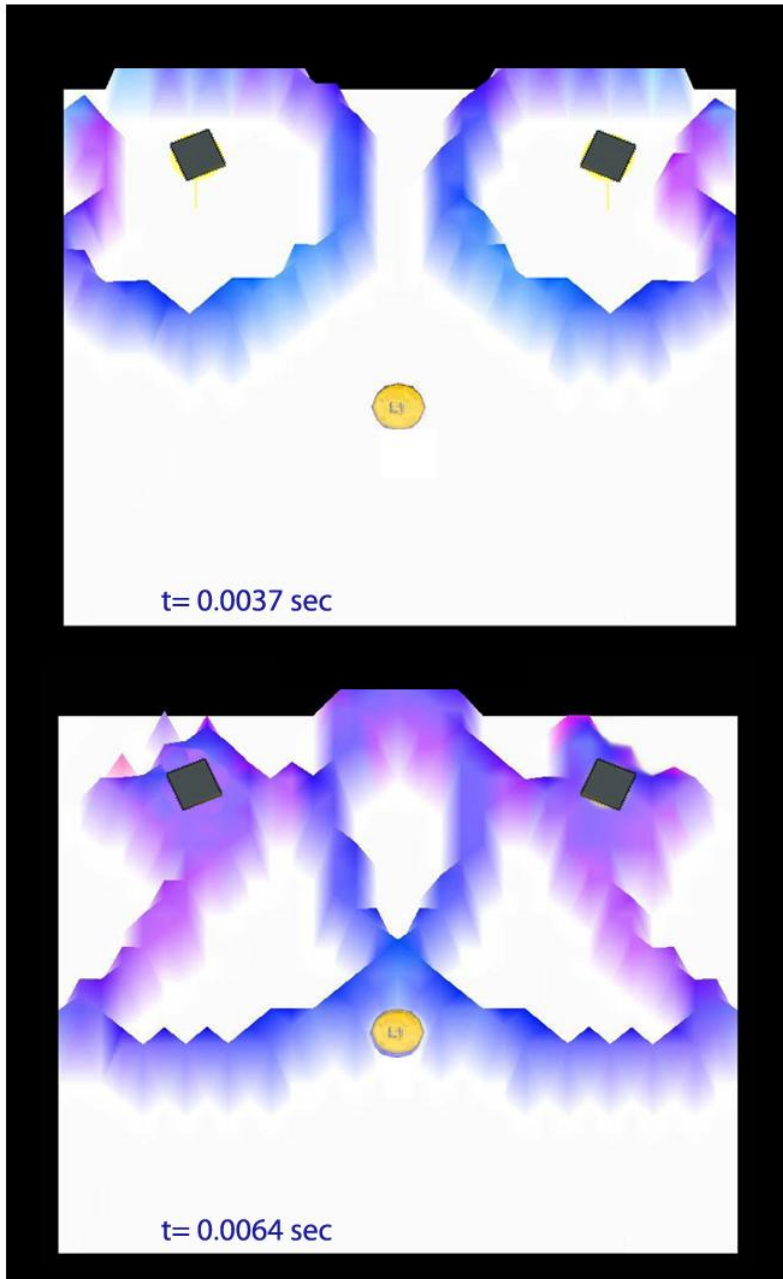


Below about 1500Hz there is a phase difference between the sound waves entering the ears, thus providing acoustic localization cues (interaural time differences, or ITD). At frequencies greater than 1500 Hz, the wavelength is shorter than the distance between the 2 ears, and phase sensitivity diminishes. 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.

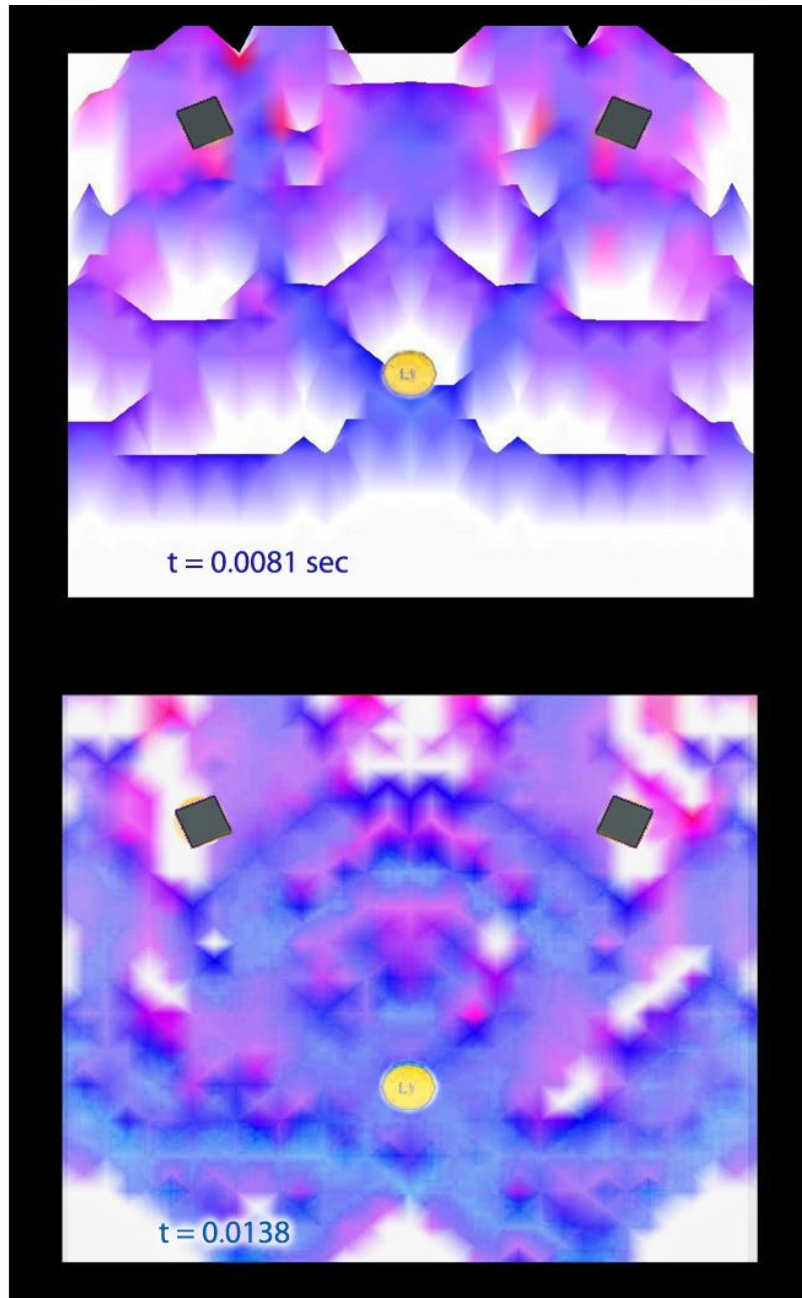
The key to imaging is preserving the interaural time and level differences. The shadowing to the far ear from the source is quite fragile in a reflective environment. Maintaining directivity in the speaker radiation pattern is essential to preserve this content. Legacy V, Whisper, and Aeris each offer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



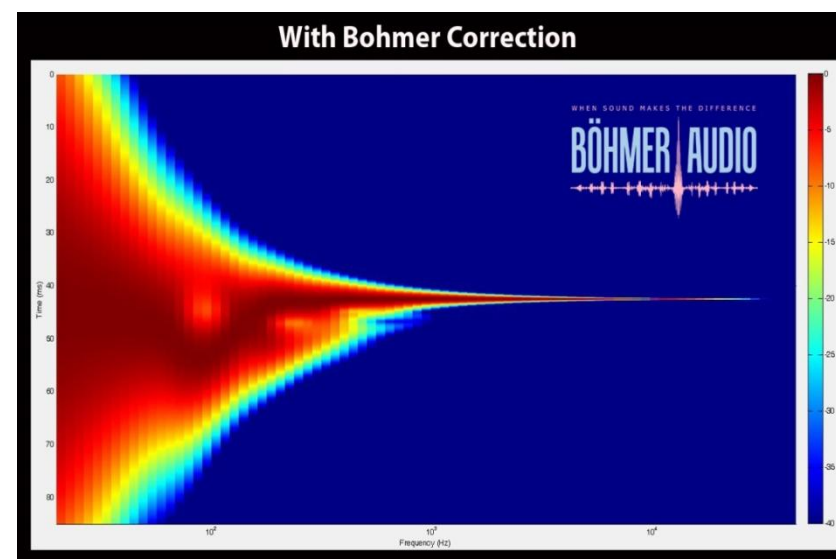
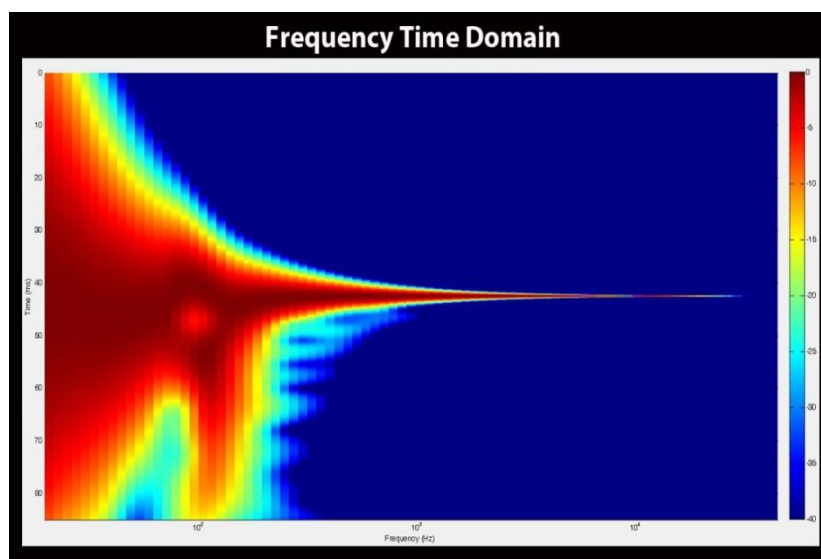
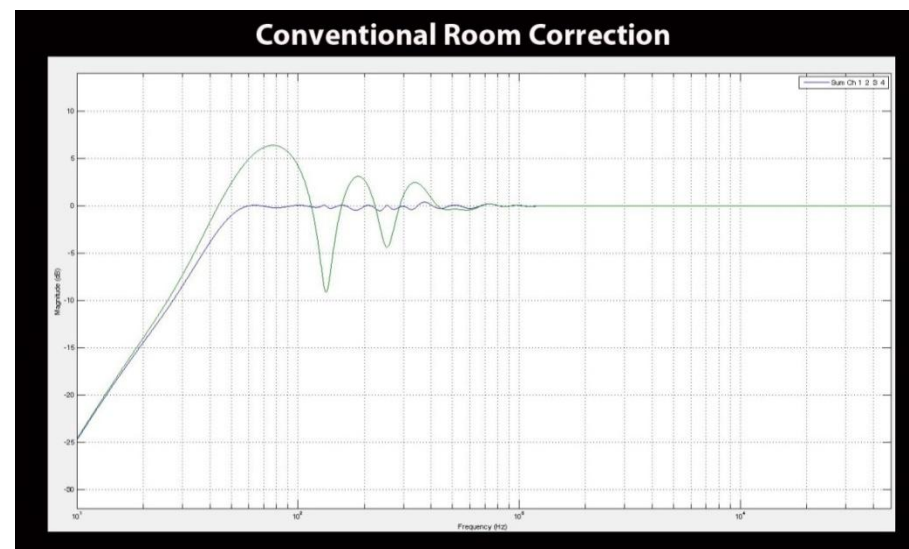
After less than $1/100^{\text{th}}$ of a second, the room has developed a complex wave pattern with energy varying with room position. The listener is now awash in a series of wave-fronts which will soon reflect off the wall behind.

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.



The Engineer's Perspective

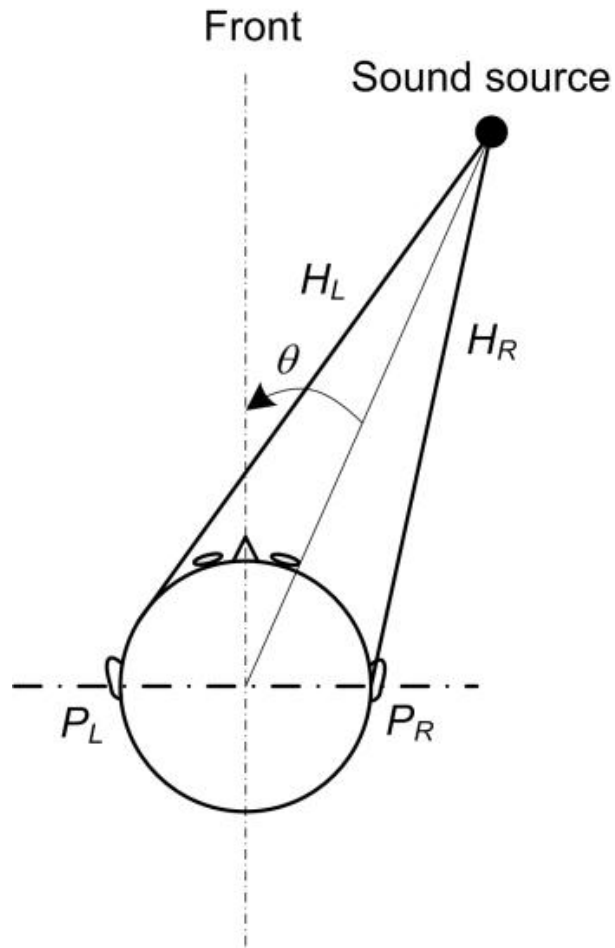
thoughts from Legacy Chief Designer, Bill Dudleston

- 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 the 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

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