LS1 Playback System Manual

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Introduction

Congratulations on your purchase of the Grimm Audio LS1 playback system. The LS1 was originally conceived as a professional studio monitor for mixing and mastering, but people soon discovered it as a unique high-end quality audio system for audiophiles and music lovers. We made sure that the LS1 feels equally at home in studios and living rooms, by covering the needs of both use cases in functionality and sound quality.

In this manual we assist you with setting up the LS1 playback system in all possible contexts. You will learn how to connect sources and how to control the settings. Furthermore we like to explain what makes the LS1 playback system so different compared to most other loudspeakers; why it looks, functions and most importantly sounds the way it does.

As an introduction into the 'LS1 world', let's first outline three music source concepts for the LS1 playback system. You can receive music from:

1. The Grimm Audio MU1 music streamer, via the proprietary LS1 cat5 connector .
2. A computer, by using our LS1i USB interface and LS1r remote control; or by using our UC1 USB interface & monitor controller. These connect via the proprietary LS1 cat5 connector.
3. Sources that are connected straight into the analog or digital XLR inputs of the LS1, such as a CD transport, phono preamp (like our PW1), streamer, analog or digital monitor controller, etc.

Volume changes and source selection inside the LS1 DSP (Digital Signal Processor) can be set via the LS1r remote control, the MU1 or the UC1 USB interface & monitor controller. You may use either the physical controls on these devices, or software controls. For the latter we offer the

MU1 web interface GRUI, or the LS1 Control application which runs on the computer that is connected to the LS1. Additionally an LS1 also responds to volume instructions from the music application that runs on the MU1 or from the Operating System of the connected computer in case you are using a Grimm Audio UC1. This means that the volume slider of your player software will actually control the volume inside the LS1 DSP. Controlling the volume there has the advantage that it is applied in the last stage before conversion to analog, so digital audio is kept at the highest resolution throughout the signal chain.

Note: if you are about to unpack and assemble your LS1 set, please consult the separate “LS1 Assembly and Install” manual. If you have SB1 or LS1s subwoofers and like to add them to your system, please read the “SB1 and LS1s” manual.

Note: if you want to connect your system first and read about the LS1 backgrounds and fine tune instructions later at your leisure, please start reading the “LS1 System Setup” chapter of this manual.

The LS1 concept

Grimm Audio‘s LS1 was designed as a loudspeaker that is analytically precise and musically pleasing at the same time. When we first developed the LS1, we had the professional user in mind. In a studio you need a monitor with uncolored sound (a flat frequency response) that is non fatiguing to listen to for a whole day of work. Soon the LS1 was embraced by audiophiles and music lovers since it let them experience all recordings in their purest form, hiding no details and offering a strong emotional connection to the performance of the musicians.

By integrating top quality digital signal processing, converters and amplifiers in one of the speakers' legs, we were able to tune the complete reproduction chain including speakers and filters to perfection. The need for external amplification and DA conversion is eliminated, and the streamlined signal path optimally preserves sound quality. As a consequence, the LS1 is not just a loudspeaker, but a complete audio system that only needs an analog or digital source to perform.

Since we aim for a neutral frequency response, avoiding so called ’voicing‘, we tested the prototypes in a large anechoic room with a calibrated measurement microphone. It made us decide to measure and calibrate every single LS1 during production, so the response of every loudspeaker is individually corrected.

All crossover filtering and frequency response equalisation is done digitally in the DSP (’Digital Signal Processor’) of the LS1 electronics. Digital processing offers a much higher degree of correction than possible in the analog domain, and without the distortion associated with analog components. At the same time, we are aware that digital processes have their own sonic limitations, and they need to be addressed. Over the years, Grimm Audio became an acclaimed developer of ground breaking DSP algorithms that redefined the sound of digital audio. These algorithms found their way into the systems of all our customers through regular software updates.

By now it is clear that the heart of the LS1 is formed by digital processing, with a separate DA converter and amplifier per driver. This means that digital sources can be fed directly into the DSP. Analog sources are first converted to the digital domain by a very high quality Analog to Digital converter. These sources also benefit from the unique optimization that we achieve in the digital domain.

The LS1 acoustical design

The DSP in the LS1 takes care of three important tasks: response correction of the drivers, crossover between the drivers and phase correction of the crossover. Let's look at the response correction in more detail now. And remarkably this means we first need to step back into the original analog domain of acoustics. The spatial radiation behavior of a loudspeaker cannot be corrected electrically so it should be taken care of first, in its physical design.

The theory behind the LS1's acoustical design originates back to the 1930s and '40s. Cabinets in those days were flat but wide, which moved the so-called ’baffle-step‘ frequency to lower than 250 Hz. Below the baffle step frequency, sound is radiated equally in all directions. Above the baffle step the sound becomes more forward directed, thereby attenuating reflections against the wall behind the speaker and the side walls. Acoustics says that below approximately 250 Hz the frequency response is dominated by ‘room mode’ resonances. Above that range the direct sound from the loudspeaker ideally reaches the listener with the least amount of interfering reflections. This is achieved by aiming a wide baffle loudspeaker away from a nearby reflective side wall.

Progress since the 40's has learned that edge diffraction of the cabinet deteriorates the stereo image, hence the sides of the LS1 cabinet are strongly rounded by the legs. As a bonus, the shallow cabinet moves the resonance along the depth axis inside the box significantly above the crossover frequency of the low frequency driver. This helps to avoid boxiness.

The next step in the design phase was the driver selection. We settled on an outstanding pair of Seas units. The tweeter’s exceptional performance in the higher mid range allows for a fairly low crossover frequency that benefits vertical dispersion. Its small constant directivity wave guide ensures an even spread of high frequency sound. Even dispersion of the LS1 is also helped by the lack of cone breakup effects in the magnesium woofer below the crossover frequency. The woofer's long linear excursion permits generous sound pressure capability down to 40Hz for the two-way configuration. Looking at the other LS1 models, the woofer of the LS1a model was selected since it possesses many of the assets of the magnesium one. The Carbon and Beryllium tweeters of the LS1c and LS1be take the performance of the LS1 to the next level by eliminating virtually all break-up resonances and the associated phase distortion in the tweeter.

Adding an LS1s subwoofer takes performance up a notch by extending the response down to 20 Hz and further increasing attainable loudness. By choosing our motion feedback controlled SB1 subwoofer, the low frequency performance of the LS1 reaches unparalleled levels. Distortion is lowered by more than 25 dB and the intrinsic resonance of the system is lowered to 15 Hz, which is below the hearing range. More information about the SB1 is offered below and in the SB1 manual.

The LS1 Electronics

With an acoustically optimized response as a starting point, we apply DSP equalization to flatten the response of the woofer and tweeter. Thanks to the cabinet design only limited DSP corrections are required to linearize amplitude and phase responses. Part of these are individually tuned during production. To help the operator in acquiring the optimal equalisation setting for these corrections, we developed a unique Computer Aided Calibration software tool in-house. It calculates thousands of correction curves and presents the best solutions for the system. Finally, the acoustic sum is phase corrected to linear phase. Measurements of the step response confirm the effectiveness of this strategy.

When LS1s or SB1 subwoofers are added, the LS1 becomes a true 3-way system. The DSP again takes care of all filtering, including the crossover and its phase correction, which has a dramatic impact at low frequencies. This results in a 3-way system that sounds very coherent.

Let’s now look in more detail at the electric signal path. Digital input signals are first re-clocked by a high quality ASRC circuit which is based on the same oscillator as the legendary Grimm CC1 clock. Analog inputs are converted to digital with a high performance AD converter. The signal then enters a high performance ‘fixed point’ DSP, which has a data path of 48 bits wide with a 76-bit accumulator. Each driver then has its own high quality DA converter. The output of the DA converters is directly fed into 250 W (woofer) and 100 W (tweeter) Hypex Ncore power amplifiers. The SB1 subs have a 400 W amp each. Hypex Ncore power amplifiers have extreme low harmonic distortion and are the perfect choice for the LS1 due to their linearity and excellent transient response. Throughout the whole circuit, careful attention has been paid to component selection like capacitors and resistors. Critical parts such as power supply regulators are of our own discrete design. Once our circuits measure well, we start listening to tweak the last parts by ear.

And this explains why the LS1 is not only neutral and precise but also a joy to listen to and musically pleasing. We started with smart concepts and measurements but we did not stop there. We listened. And doing so we developed a loudspeaker system that is equally at home in a studio as well as in a living room.

Optional subwoofers: SB1 and LS1s

For high-fidelity playback of music with substantial low-frequency content, such as large church organs or modern electronic music, additional acoustic power in the low-frequency range may be necessary. While the standard two-way LS1 configuration can reproduce a full range of sound at adequate levels for most music, it can benefit from low-frequency support when handling significant sub-50 Hz content at high playback levels.

The ‘LS1’ preferences tab in the LS1 Control software and the MU1 GRUI web page allows users to adjust the low-cut frequency of a two-way LS1 system to 40-45 Hz, enabling the LS1 to deliver high sound levels as a two-way system. Lowering the cut-off to 35 Hz or even lower allows for extended low-frequency reproduction, albeit at the cost of reduced maximum achievable sound levels and increased distortion.

Grimm Audio developed the LS1s and SB1 subwoofers for a further improvement of low frequency output. As an optional extension they are designed to integrate seamlessly, both acoustically and visually, with the LS1. The LS1s design achieves high sound pressure levels at low frequencies while maintaining a flat frequency, time, and phase response. It uses a moderately large, high performance driver with a very long linear excursion to allow a huge volume displacement. A powerful 400W amplifier, paired with the LS1’s DSP, compensates for the frequency response of the driver in its relatively compact cabinet.

We realized that low frequency performance in general is limited by two things: distortion that results from the large cone excursions needed for low frequency playback, and the intrinsic resonance of a driver that’s mounted in a box. To offer the definitive solution to both issues, we developed the SB1 subwoofer that is controlled by a digital Motional Feedback (MFB) system.

MFB technology uses a sensor on the cone of the low-frequency driver to measure the actual cone acceleration. The cone acceleration is equivalent to the generated sound pressure, which is the audible audio signal. By feeding this measured signal back to the amplifier, the actual acoustic signal can be compared to the intended signal, and deviations are corrected. This technique was originally developed for consumer loudspeakers by Philips in the 1970s. Through this approach they were able to reduce distortion by 12 dB and at the same time lower the resonance frequency for an extended low-frequency response.

Grimm Audio’s digital version of Motion Feedback further enhances this concept. It provides increased accuracy, expanded dynamic range, and vastly improved acoustic correction capabilities. Our digital MFB system reduces harmonic distortion by more than 25 dB, and lowers the speaker resonance to as low as 15 Hz. As a result the modest sized SB1 sounds as if a much larger subwoofer is playing, with more control than any passive loudspeaker can offer.

In certain challenging acoustic environments, it may be beneficial to position the subwoofers separately from the main speakers. The LS1’s design allows the LS1s or SB1 subwoofers to be moved from the main system’s foot plate, enabling independent placement. If room acoustics are particularly difficult, consider placing the subwoofer closer to the side walls than the main speaker.

The preference pane of the LS1 Control software and the MU1 GRUI web interface provides parameters for subwoofer gain and delay, which can be adjusted to correct alignment issues that may arise from detached placements of the sub and main loudspeaker. While we do not offer case-specific placement recommendations, we encourage you to experiment in search of the optimal setup. For further details, please see the chapter “Software Control Settings.”

Room acoustics

It is important to keep in mind that one always hears the loudspeaker plus its acoustic environment. Although the LS1 has superior acoustical characteristics, positioning it correctly in a room remains of great importance. Let‘s have a brief look at room acoustics first and then see how positioning the LS1 influences its performance.

There are three main factors that affect speaker placement: first reflections, low-frequency coupling, and room modes (resonances).

First reflections

First reflections are sound waves that bounce off nearby surfaces before reaching the listener. They affect the stereo image and the color of the sound. Lateral first reflections influence the stereo image. Since high frequencies are dominant in acoustic imaging, adding fabric acoustic absorption like curtains at the reflection spots on the left and right walls helps a lot. Since the LS1 has a wide baffle that projects the sound over a wide frequency range, aiming the loudspeaker away from the reflective spot helps too (this is often called ‘toe in’). Another major type of reflection is via the floor and ceiling. These reflections interfere less with the stereo image, but do color the sound. Absorbing the floor bounce with a carpet helps.

A special case is reflections against the back of an obstacle in front of the listener, like a computer screen in a music studio, bouncing the sound toward the front wall and from there back to the listener. In those cases, absorbing the reflection at either the obstacle or the wall cures the problem. Due to the wide baffle of the LS1, which focuses the sound, reflections on the wall behind the listener tend to have slightly more impact than with other loudspeakers. Please consider adding absorption or sound scattering objects like bookcases against the wall behind you.

First reflections also introduce phase distortion in the early sound if the walls are too close. Psycho acoustics tells us this effect is important down to 250 Hz. One wavelength of a 250 Hz tone lasts 4 ms so this is the time span that certainly needs to be ’clean‘ of reflections. Sound travels app. 1.4 m in that time. The rule of thumb is therefore that you need to have at least 80 cm distance from the side walls (1.5 m is better). Because of the wide baffle, the distance to the wall behind the LS1 is less critical. In fact it makes the LS1 unique since it can be placed very close to the wall without coloration, and this can actually be the best position in some cases. Feel free to experiment with the distance to the front wall and experience the impact it has on both the lower mid response and the stereo imaging. In case you place the LS1 pair close to the wall, make sure to apply some acoustic absorption on the wall between the loudspeakers so the stereo image is preserved best.

Low-frequency coupling

At frequencies below 250 Hz (bass and lower mid), the LS1 radiates sound in all directions. Sound travelling backwards and sideways reflects against the walls and adds up to the direct sound. Although the ear is less sensitive to the resulting phase errors at these low frequencies, the impact on the low frequency energy itself can be dramatic since at certain frequencies the reflected sound can cancel the direct sound. A real solution is to apply low frequency absorption behind the loudspeaker or in the corners of the room so the reflection is attenuated. If that is not possible, spend some time shifting the speakers back and forth until the optimal compromise is found. In this respect the LS1 is not different from any other standard loudspeaker. What ís different is that the wide baffle allows the LS1 to be placed close to the wall behind it, as mentioned in the previous paragraph. The reflection will then not cancel the direct sound at any frequency, but just amplify it over the full low end. The low frequency shelf filter in the LS1 Control software and the MU1 GRUI allows to compensate for excessive low frequency energy when needed.

A note about subwoofer positioning: the LS1s or SB1 subwoofers can be taken off the main speakers’ footplates. In challenging acoustic environments, placing the subwoofer closer to the front or side wall may improve low-frequency performance. Adjust the subwoofer's gain and delay settings in the software to align it with the main speakers.

Room modes

Room modes are resonances that occur at specific frequencies and on specific locations due to the room's dimensions. All listening rooms resonate at low frequencies (below app. 200 Hz). Tones bouncing back and forth between walls interfere, and at certain frequencies they form node and antinode patterns, so called standing waves, which means that certain tones sound silent in one place in the room but are very loud at other spots. These resonances are called room modes and since they are dependent on the dimensions and the weight of the walls, they vary from room to room. Some rooms therefore sound ‘better‘ than others. Whether a certain room mode is triggered or not is dependent on the position of the source and of the listener. So it makes sense to spend some time positioning the LS1's and listening seats carefully to find the optimal compromise. Small changes can have a significant impact on the perceived bass response. Consult a specialized acoustician if you have a real strong room mode problem in your room that cannot be solved without acoustic treatments.

DSP is sometimes used for ’room correction‘. We rather not use the DSP in the LS1 for this since the effects are highly dependent on the position of the listener. To understand this, let’s again look at the three acoustic factors: 1. First reflections are strongly dependent on the listeners‘ position, so there is no way to correct these with DSP. This is certainly true above 200 Hz where the wavelength is short (< 1.5m). 2. The low frequency coupling effect can partly be compensated by DSP (please use the LS1 EQ for that). However in the region around 100 Hz where energy can be attenuated when a strong reflection is out of phase, DSP will not help since a louder tone will still be canceled by its reflection. Careful placement is the only cure. 3. Finally, room modes are also strongly dependent on the position of the listener. Theoretically it would be advantageous to attenuate peaks in the room response that reside in the whole listening area, but all DSP solutions for this we tried so far compromised the LS1's performance too much. Of course you may come to different conclusions, so please feel free to experiment with room EQ DSP. But please also consider solving problematic room modes with acoustic treatments like passive or active bass traps.

LS1 positioning

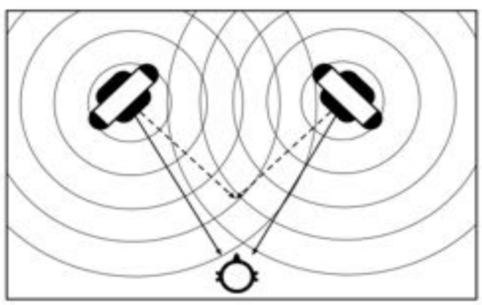
Correct placement of your loudspeakers is essential for achieving the ultimate goal of reproducing the music as it was intended by the artists. During production in the studio the producer listens in a so-called ’equal triangle‘ setup where the corners of the triangle between the two loudspeakers and the listener are all 60 degrees and so all sides of the triangle are of equal length. When possible, please aim at this arrangement for your ‘sweet spot listening chair’. The distance between the loudspeakers ideally is between 2.5 and 3m, with an equal distance between the listening seat and the loudspeakers.

To achieve the best stereo image, it is important that both the left and right loudspeaker are in a comparable acoustic environment. For example, if one loudspeaker is placed close to a side wall and the other has free space on both sides, the low frequencies of the speaker near the wall will be enforced by reflections and therefore the stereo image shifts to that side. A quick and easy check is to sit at the listening position and have an assistant talk to you from both the (intended) speaker positions. If his or her voice sounds equal in both positions, you are fine. If the voice is boomy in one position and bright in the other, it’s recommended to search for better positioning. Listen to the direct sound from the assistant as well as the ‘room sound‘. In fact, strange as it may seem, you can hear these effects too when speaking yourself at the loudspeaker positions and listen carefully. If you cannot find equal sounding positions for the loudspeakers in your room, and for instance one speaker has to stand in a corner and the other along the wall, you may use the LS1 EQ to equalize the difference in the low end response between both speakers. Please consult the EQ paragraph of the Software Control Settings chapter in this manual.

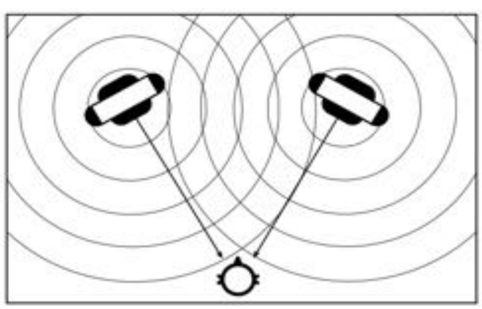
‘Toe-in’ or not

The LS1 cabinet is wider than usual, so the effect of turning the loudspeaker around its vertical axis is larger than with most other systems. The off-axis response on the front side of the LS1 is extremely even (the LS1 sounds just as good slightly off-axis as on-axis), but it becomes gradually softer over a large frequency range when the angle is larger. We can use this effect to our advantage. In a classic arrangement the loudspeakers face the listener. The angle of the loudspeakers relative to the wall behind them is 30 degrees. The superb off-axis performance of the LS1 offers an alternative setup where both loudspeakers are turned slightly further inward, or ’toe in‘. We advise an angle relative to the wall of 45 degrees. The listener then sits at 15 degrees off-axis of the loudspeakers, like you can see in the LS1 setup images.

LS1 setup with 45° angle



LS1 setup with 30° angle



This ’toe in‘ setup has two advantages:

1. For seats outside the sweet spot, the stereo image is better preserved. The reason is that for a left seat the sound of the left loudspeaker will reach the ear more early. This causes a shift to the left in the stereo image. But at the same time, the listener is more on-axis of the right speaker and more off-axis of the left speaker. The right loudspeaker therefore becomes slightly louder and the image shifts back to the right again. The compensation is not perfect, but the net effect is a wider area in which you can enjoy a nice stereo image.

2. The reflection of the left loudspeakers‘ sound to the nearby left wall (and vice versa) becomes attenuated a bit. Since this reflection path is usually relatively short, it often interferes with the stereo image. Therefore attenuating that reflection improves the imaging. An easy trick to check if this reflection is too loud is to listen to a mono sound on one loudspeaker only. Close your eyes and point in the direction of the sound. Now open your eyes again and look at your finger. If it is pointing towards the center of the loudspeaker everything is fine. If you point a little bit off-center in the direction of the wall, the reflection is too loud. If turning the LS1's to a toe-in position does not cure the problem, you will need to apply acoustic absorption at the reflective spots. Note that by turning the left LS1 like this, its reflection on the right wall becomes relatively louder. Usually this is not a problem but rather an advantage because this reflection comes later in time. Psychoacoustics says that discrete later reflections help the brain to extract the ’ambience‘ information from the recording.

In about 20% of the rooms, the classic setup where the LS1's face the listener still sounds better. This is usually the case in fairly wide rooms or in rooms with professional acoustic treatments where left and right reflections are very well balanced. Please accept this as an invitation to experiment and listen in your own environment.

Once you have decided on which of the two setups sounds best, please run the LS1 control software and set the ’toe in‘ angle in the preferences to your selected 45 degrees or 30 degrees angle. This applies a small correction to the highest treble, making sure the frequency response of the direct sound is perfectly linear at the listening position.

LS1 System Setup

Hardware connections

In this chapter we explain how to connect all elements of your playback system. We will first introduce a few possible system setups before we guide you through the configuration of the LS1 itself.

For all setups it is important to understand that an LS1 has internal digital DSP processing for the crossover, driver corrections, volume control, equalizing and source selection. This DSP processor is controlled externally by Grimm Audio hardware devices such as the MU1, LS1r remote or UC1 USB interface, or by the LS1 Control software that runs on a connected computer.

Note: The previous section contains further information and tips on how to position your loudspeakers in their acoustic environment.

Note: a separate “LS1 unpacking and install” manual contains detailed information on how to unpack and assemble your LS1 system.

In the picture the connection plate of an LS1 is shown. From left to right:

- An analog audio input on XLR.
- A digital audio input on XLR.
- A cat5 ‘Control In’ input for digital audio and control data in Grimm Audio’s proprietary format.
- A cat5 ‘Control Out’ output that must be connected to the cat5 ‘Control In’ input of the next LS1 in the chain.
- A filtered analog subwoofer output on XLR.
- A switch to select the left or right channel for this LS1.
- The mains power connector and power switch.

Subwoofer connection

When a subwoofer is added, the LS1 system becomes a true 3-way loudspeaker that can play lower, louder and with less distortion. Every LS1 has a dedicated subwoofer output, which can be controlled from within the LS1 Control software and the MU1 GRUI. Often a Grimm Audio SB1 or LS1s subwoofer is used. It is also possible to connect any 3rd party subwoofer to this output. In all cases a high quality digital crossover is applied. However, our unique crossover phase compensation which offers a perfect time domain behavior is only available when using Grimm Audio subwoofers.

To add a subwoofer, connect an analog XLR cable from the Subwoofer output of the LS1 to the analog input of the subwoofer.

Note: More information about the SB1 and LS1s subwoofers is available in the previous section.

Loopback connector

A note for owners of first generation LS1 systems with a serial number lower than 03-2.004.xxx. These systems require the use of a ‘loopback connector’, as shown in the picture. It was supplied with your system. The loopback connector must be placed in the ‘control out’ cat5 output of the second LS1 in the signal chain. Without the loopback connector the LS1 Control software and MU1 GRUI will not be able to communicate with these LS1 systems. In case you misplaced your loopback connector, please contact us.

Loopback connector.

1. LS1 with MU1 Streamer

This chapter shows you how to connect a MU1 to an LS1 system. In the picture you see the connections on the back of the Grimm Audio MU1. From left to right:

- Three digital outputs on RCA and XLR (S/PDIF out, AES out 2, AES out 1). With an LS1, these are only used in a surround setup, or if you have a headphone DAC. In the latter case, please consult the MU1 software manual for information about how to configure a 'second out' in your MU1 so you can conveniently switch between stereo LS1 and headphone playback. If you are setting up a surround system of LS1's with a MU1, detailed setup information is given in the MU1 hardware manual.

- A proprietary cat5 connector 'LS1 Control Out' carrying digital audio and control data. It links the MU1 to the first LS1 in the chain.

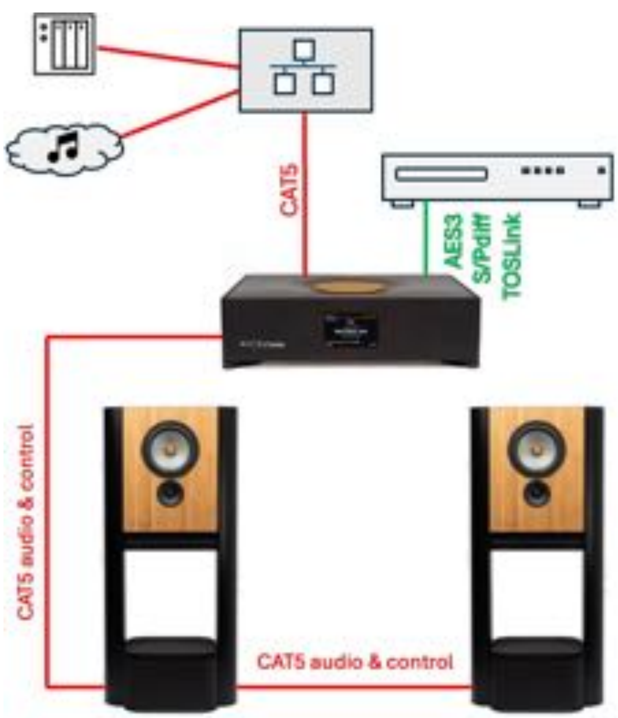
- Three digital audio inputs on XLR, RCA and Toslink (AES in, S/PDIF in, Optical in).

- A USB connector for external storage.

- An ethernet connector to connect the MU1 to your computer network.

- A 3.5mm jack for the MU1's IR remote control extension cord.

- The mains power connector.



LS1 and MU1 digital connections.

LS1 and MU1 analog connections.

The wiring diagrams show how to interconnect a MU1 to an LS1 and several sources. In this typical 'hifi' configuration, the LS1 speakers are connected to the MU1 Music Streamer and each other via cat5 cables. These cables carry both audio and control signals in a proprietary format. The MU1 acts as the central hub of your home audio system, controlling the LS1's and interfacing with various audio sources, such as:

1. LS1 Audio and Control wiring: the first LS1 loudspeaker in the chain receives digital audio and control information from the MU1's 'LS1 Control Out' output via a proprietary cat5 cable into its 'Control In' input. Another cat5 cable is connected from the 'Control Out' output of the first LS1 to the 'Control In' input of the second LS1.

2. Network Connection for Streaming. The MU1 is linked to a computer network via a standard Ethernet cable. This network connection enables access to a tablet, smartphone or laptop on the same network that controls the system and the music playback. It also offers access to music that is stored locally on your network or on the server of a streaming service. For more information about setting up your MU1, please consult the MU1 Software manual.

3. Analog inputs: every LS1 has an XLR input for connecting analog sources. For example a phono preamplifier such as the Grimm Audio PW1 can be connected directly to the LS1 to enjoy vinyl playback.

4. Digital inputs: the AES XLR, SPDIF RCA and Toslink inputs on the MU1 allow for connection of digital audio sources such as CD Players, AV receivers or smart TV's. The

audio from these sources is dejittered and upsampled, just like with streaming sources. Additionally a digital audio source can be connected to the AES XLR input of the first LS1. Note that the audio on this input will not benefit from the MU1's low jitter and (possible) upsampling.

5. Digital outputs: the AES 1 and 2 outputs can be used either for surround playback, or for connecting a (headphone) DAC (via a "2nd Out" function). For information about how to set up such a surround system, or control the 2nd Out function, please consult the MU1 Software Manual.

In this setup the MU1 serves as the main playback device. It offers source selection and volume control, plus it facilitates audio playback of physical and streaming sources across the network. The MU1 functions can be set via the main knob on top or via its 'GRUI' web page. The GRUI also offers access to all LS1 settings, that are described in the LS1 Control chapter of this manual.

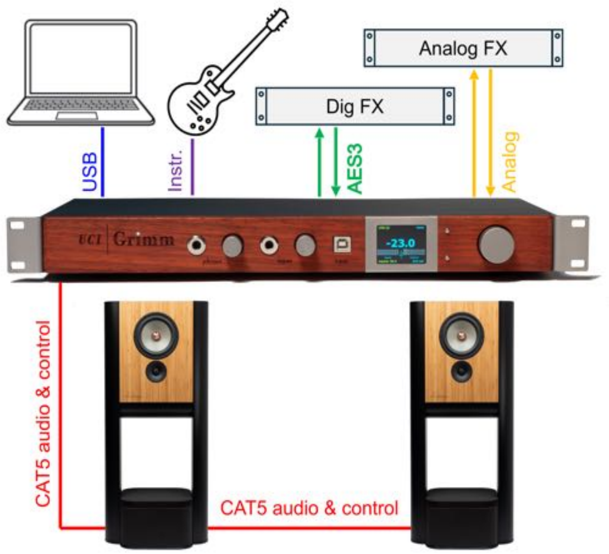
Note that the GRUI does not facilitate firmware updates of an LS1. In case you are notified through for instance the Grimm Audio Newsletter that new firmware for your LS1 is available, please use the Grimm Audio USB Control Cable that came with your system to connect an LS1 to a Mac, Windows or Linux computer. For more information, please read the LS1 Control Software chapter of this manual.

2. LS1 with UC1 Universal Converter

In professional studios, the UC1 audio interface & monitor controller forms a perfect partner for an LS1 system. In the picture you see the back of the Grimm Audio UC1. The connections from left to right are:

- A proprietary 'LS1 Control Out' cat5 connector carrying digital audio and control data. It links the UC1 to the first LS1 in the chain.
- A digital audio output on XLR.
- Two digital audio inputs on XLR.
- A USB connection to the main computer (usually running the DAW software).
- Wordclock in- and outputs.
- Six analog audio outputs on XLR

- Two analog audio inputs on XLR.



LS1 and UC1 digital and analog connections.

In this professional audio setup, the LS1 loudspeakers are connected to the Grimm Audio UC1 USB interface & monitor controller via cat5 cables. The UC1 serves as the central 'hub' of the studio, handling multiple essential functions, including USB audio interface with AD/DA conversion, word clock source/destination, headphone amplification, and studio monitor control.

The LS1 and UC1 together form a complete monitoring system for studios. It offers the following connections and functions:

1. LS1 Audio and Control wiring: the first LS1 loudspeaker in the chain receives digital audio and control information from the UC1 'LS1 Control Out' via a (proprietary format) cat5 cable to its 'Control In' input. Another cat5 cable is connected from the 'Control Out' output of the first LS1 to the 'Control In' input of the second LS1. The main knob of the UC1 can control the volume inside the LS1 DSP via this cat5 wiring.
2. USB Digital Audio Workstation interfacing: the UC1 is connected to a computer via USB and functions as a mastering quality sound card. This connection enables playback and recording with a DAW, allowing digital audio to be directly routed to the LS1 speakers and to all other inputs and outputs of the UC1.
3. USB control of all LS1 functions via the Grimm Audio LS1 Control software for Mac, Windows and Linux. The extensive LS1 Control functions are described in the "LS1 Control software" chapter of this manual.
4. Analog inputs:
    - Instrument Input: the UC1 has a high-impedance instrument input on the front. Since it is a stereo connector, it doubles as an unbalanced stereo analog input.

- A stereo XLR input on the back of the UC1 for input of a stereo analog effects loop, or of a generic analog source.
- An analog monitoring XLR input on each LS1. It can be enabled via the LS1 Control software and has no USB routing to the DAW.

5. Analog outputs:
   - Three pairs of analog outputs: one for the analog effects loop, one for a 2nd speaker and one additional output pair (used for analog main output when there's no LS1 connected).
6. Digital inputs:
   - Two stereo AES XLR inputs: one for a digital effects loop and one generic input.
   - A secondary two channel USB input on the front.
   - A digital monitoring AES input on the first LS1. It can be enabled via the LS1 Control software and has no USB routing to the DAW.
7. Digital output:
   - A stereo AES XLR output for a digital effects loop or as a generic output. It can also be used as main output in stead of the LS1 output or analog outputs.
8. Word Clock connection: the UC1 offers a mastering quality word clock output on BNC. It can also slave to a word clock input signal on BNC.
9. Metering: convenient metering of all input and output channels is offered on the UC1 front display.
10. Studio Monitor Control: the UC1 functions as a studio monitor controller, that can switch between auditioning the audio on the LS1, on a built-in mastering quality headphone amplifier or on a secondary loudspeaker set.

More information about the UC1 can be found in the UC1 manual.

3. LS1 with LS1r Remote and LS1i USB interface

The combination of an LS1 system with the LS1r Remote and LS1i USB interface can be considered the 'classic' system. This is how the LS1 system was originally conceived.

Here is a picture of the back of the LS1r remote control. The connections from left to right are:

- Two digital audio inputs on XLR.

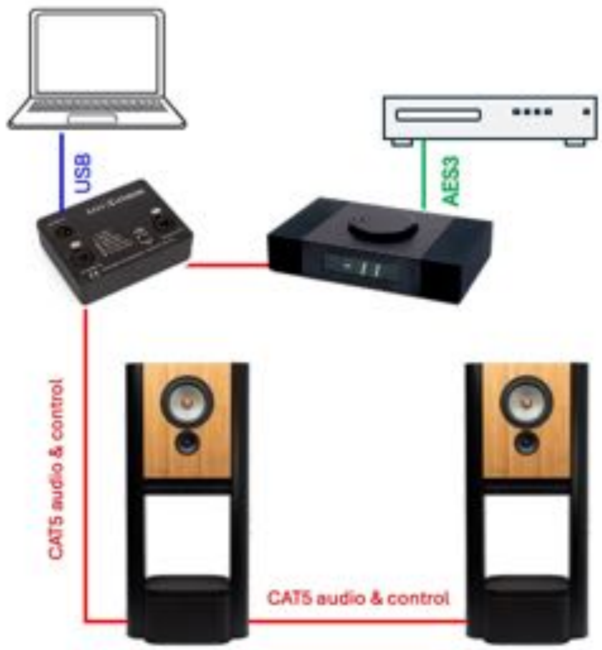
- A proprietary ‘Remote Out’ cat5 connector carrying digital audio and control data. It links the LS1r to the first LS1 in the chain, or to the LS1i.

This picture shows the connections of the LS1i USB interface. From left to right:

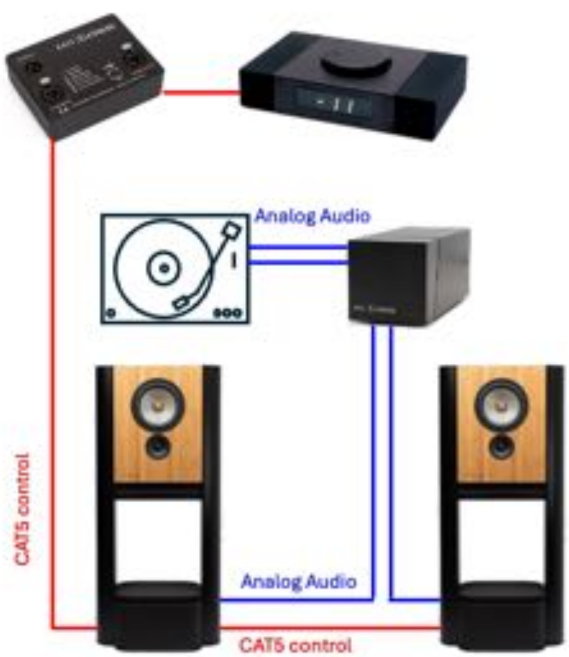
- A digital AES/EBU audio output on XLR. This output carries the same digital audio signal that is send to the LS1 via USB, without any volume control. In a professional application it can be connected to a hardware digital audio meter so the engineer has a visual indication of the sound that is being monitored.

- A proprietary ‘LS1 Control Out’ cat5 connector carrying digital audio and control data. It links the LS1i to the first LS1 in the chain.

- A proprietary ‘Remote Control In’ cat5 connector that receives the LS1r remote’s audio and control data.



LS1r and LS1i digital connections.



LS1r and LS1i analog connections.

This configuration can be used in both hifi and professional audio settings. The LS1i USB Interface and LS1r Remote Control together serve as a central hub, managing audio from a computer (or USB music streamer) and digital audio sources, delivering the audio and control metadata to the LS1 speakers with an efficient, streamlined setup for both playback and control. Together they offer the folowing connections and functions:

1. LS1 Audio and Control wiring: the LS1r is connected to the “Remote Control in” connector of the LS1i via a proprietary cat5 cable. The “LS1 Control Out” of the LS1i is connected to the ‘Control In’ cat5 input of the first LS1 and the ‘Control Out’ cat5 output of the first LS1 is connected to the second LS1 ‘Control In’ cat5 input with a second cable. Via these cat5 wires the LS1r can control the volume inside the LS1 DSP’s and perform source selection.
2. USB connection to a DAW or Music Streaming computer: the LS1i is connected to a computer via a USB cable. This connection enables the LS1i to function as a USB sound card (or ‘USB DAC’), allowing digital audio playback from the computer directly to the LS1 speakers through the LS1i. It also facilitates control of all LS1 functions via the LS1 Control software for Mac, Windows and Linux. All LS1 Control functions are described in the LS1 Control Software chapter of this manual.
3. Analog inputs: analog sources can be connected directly to the analog XLR inputs of the LS1’s. For example a phono preamplifier such as the Grimm Audio PW1 can be connected directly to the LS1 to enjoy vinyl playback.
4. Digital inputs: two stereo AES XLR inputs on the LS1r and one stereo AES XLR input on the first LS1. AES digital audio sources, such as CD players, can be connected to the LS1r remote control via a digital audio cable. In case your source is S/PDIF instead of AES, Grimm Audio offers a S/PDIF to AES conversion cable.
5. Digital output: the LS1i has an AES XLR output that carries the same digital audio signal as is sent to the LS1 from the LS1i or LS1r, at the original audio level (without any volume change). This output can for instance be connected to a digital audio meter so you can visually monitor all audio that is played on the LS1. Please note that digital or analog sources that are directly connected to the LS1 cannot be monitored via this AES output.

3.2. LS1 with just an LS1r Remote

An LS1 system can also be controlled with just an LS1r remote. It can adjust the system’s volume and select sources. You may connect two digital sources to the remote and one to the first LS1 in the chain. Analog sources can be connected to both LS1's.

When using just an LS1r to control an LS1 set, setting up the system requires you to temporarily use the “Grimm Audio USB Control Cable” that came with your system. To do so, disconnect the LS1r and connect the USB Control Cable instead. This cable facilitates control data communication between the LS1 system and a computer that runs the LS1 Control software. After the setup you can reconnect the LS1r. For more information about setting up an LS1 system, please read the LS1 Control Software chapter of this manual.

3.3. LS1 with just an LS1i USB interface

An LS1 system can be used with just an LS1i USB interface. Volume control and source selection is then done via the LS1 Control software that runs on a Windows, Mac or Linux PC. The available sources are USB on the LS1i, digital in on the first LS1 and analog in on both LS1's.

4. LS1 with just a Grimm Audio USB Control Cable

In the picture the Grimm Audio USB Control Cable is shown. Its USB connector should be plugged into the computer that runs the LS1 Control software. The cat5 connector at the other side should be connected into the ‘Control In’ cat5 input of the first LS1 in the chain.

The same LS1 Control software functionality that you obtain with the LS1i is offered when using the Grimm Audio USB Control Cable, except that it does not provide USB audio playback and you cannot combine the USB Control Cable with an LS1r Remote control. But in the most basic setup of an LS1 system, you can control it with just the Grimm Audio USB Control Cable and the LS1 Control software. The available sources are digital in on the first LS1 and analog in on both LS1's.

Surround setup

An extraordinary feature of the LS1 ecosystem is that it gives full control of a 5, 7 or even more channel surround system. The Grimm Audio MU1 music streamer can replay 5 channel surround files on its physical outputs, which is unique for audiophile streamers. When combining the MU1 with 5 LS1's, full control of all speakers from within the MU1 GRUI web interface is possible. To facilitate this, all speakers should receive the control information via daisy chaining the cat5 wiring to all LS1's in the system. Please refer to the MU1 hardware manual for information about how to wire the system.

Surround configurations of LS1's may also be controlled via an LS1i USB interface. In this setup, the LS1 Control software can even control up to 7 LS1 loudspeakers. Again, control data needs to be received by all LS1's in the system by daisy chaining the cat5 wiring. The order of connecting the channels is as follows:

1 LS1’s: Interface -> Center

2 LS1’s: Interface -> Front 1 -> Front 2

3 LS1’s: Interface -> Front 1 -> Front 2 -> Center

4 LS1’s: Interface -> Front 1 -> Front 2 -> Rear 1 -> Rear 2

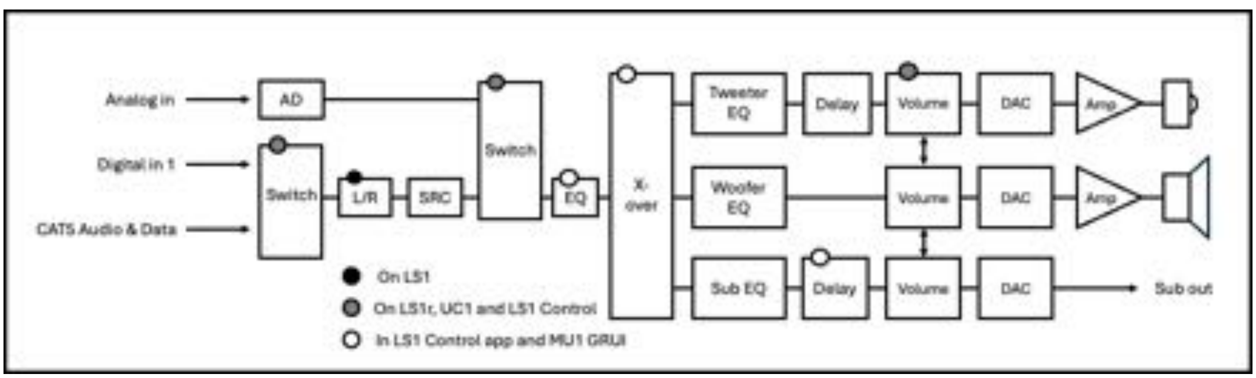
5 LS1’s: Interface -> Front 1 -> Front 2 -> Center -> Rear 1 -> Rear 2

6 LS1’s: Interface -> Front 1 -> Front 2 -> Surround 1 -> Surround 2 -> Rear 1 -> Rear 2

7 LS1’s: Interface -> Front 1 -> Front 2 -> Center -> Surround 1 -> Surround 2 -> Rear 1 -> Rear 2

LS1 Configuration and Control

Now you connected all cables, let’s have a look at how to configure and control the LS1 system. You need to keep in mind that the LS1 is a complete playback system, not just a speaker. The block diagram below illustrates the internal processes involved. In the first stage, signals from various sources are selected, some are connected directly to the LS1’s and others to the LS1r Remote, the LS1i USB interface or the MU1 streamer. Next, a switch on the connector plate assigns the LS1 to play either the left or right channel of a digital stereo pair.



A sample rate converter then ensures that the DSP can linearize the speaker’s frequency response up to very high frequencies. The analog input enters the system after this converter. In the following step an equaliser offers some sonic tailoring to taste. The crossover then splits the audio range into two or three bands, depending on the settings. Next, EQ's are applied to the individual drivers that are calibrated individually during production. A delay compensates for the acoustic positions of the tweeter and woofer. Just before the DAC stage, digital volume control is applied; this placement ensures that all prior processing takes place at maximum resolution.

Many LS1 settings are user-adjustable. In the diagram, the dark dot marks the L/R switch on the LS1's connector plate. Grey dots indicate functions managed through the LS1r remote control, the LS1 Control software or the MU1 GRUI web interface. White dots show functions that are adjustable solely via the LS1 Control software or the MU1 GRUI web interface.

Controlling your system

Every LS1 has one hardware switch on its connector plate. This 'L/R' switch determines whether the left or right channel of a digital AES or cat5 signal is picked up by the speaker. Please start your setup by putting the L/R switch of every loudspeaker in the correct position: 'left' on the left speaker and 'right' on the right speaker.

For control of all other LS1 settings, we offer three options:

1. The hardware controls of a MU1 music streamer, UC1 USB interface or Grimm Audio LS1r remote control.
2. The software controls of the LS1 Control software that runs on a Windows, Mac or Linux computer with a connected Grimm Audio USB interface such as an LS1i, UC1 or Grimm Audio USB Control Cable.
3. The software controls of the GRUI web interface of a MU1 that is connected to your LS1 system. The GRUI offers the same type of control as the LS1 Control software.

We will first have a detailed look at the hardware control options in the following pages.

Control of an LS1 with the MU1 streamer

An MU1 is the central hub in your living room audio system. It connects your LS1 to digital audio sources, to various streaming platforms, to music on an internal storage, a NAS or a connected USB drive. The analog and digital inputs of the LS1 can also be used. Source selection and volume control is all done via the MU1's main knob on top. Turning the knob instructs the LS1 to change the volume in its internal DSP. Press-hold and turning the knob selects one of the sources that are connected to the MU1 or the LS1. Alternatively, you may use a Grimm Audio or generic infrared remote with the MU1 to change the volume or select sources. As a third option, the MU1 built-in web page, called GRUI, can be used to control the LS1. The GRUI not only offers volume control and source selection but also gives you access to all LS1 settings. More information about the MU1 and the GRUI can be found in the MU1 Software manual.

Control of an LS1 with the UC1 Universal Converter

The LS1 can be connected to a UC1 Universal Converter for a turnkey professional studio setup. The Grimm Audio UC1 then functions as a USB interface and monitor controller in one. The UC1 is designed to integrate seamlessly with LS1 loudspeaker systems. Its large knob controls the volume inside the LS1 DSP, and offers source selection by press-hold and turning. If you run the LS1 Control software on the connected computer, the UC1 will relay all control data to the LS1 so software control of the LS1 system is possible. The next chapter outlines all the options of the LS1 Control software.

Control of an LS1 with the LS1r Remote

Combined with an LS1r Remote, the LS1 is a complete playback system with built-in amplifiers, converters, volume control, and source selection of 1 analog and 3 digital sources (even 4 when adding the LS1i USB interface). Conceptually, the LS1r can be seen as a digital preamplifier extension for the LS1.

LS1r Remote Functions

The LS1r Controller provides straightforward control, with a display showing volume level, mute (or dim) mode, and the selected input source. The main knob on top controls three functions:

1. Volume Control. Turn clockwise to increase volume, counterclockwise to decrease.
2. Mute or Dim function. A short press mutes or dims (10dB or 20dB attenuation) the signal. This function is set in the LS1r menu or the LS1 Control software. Press again to deactivate.
3. Input Source Selection. Press and hold the knob while turning to select a source. The display shows the current input; release the knob to confirm the selection.

This streamlined design simplifies source selection and volume control, enhancing ease of use without compromising sound quality. Alternatively the LS1r can be controlled by a Grimm Audio or generic IR (infrared) remote control for volume control, source selection and mute.

LS1r Remote setup

Some settings of your LS1r Controller can be adjusted to suit your preference. This can be done via the LS1r Remote built-in menu system or (if you combine the LS1r with an LS1i USB interface) via the LS1 Control software (described in the next chapter).

To setup the LS1r Controller via its menu system, press - hold the large knob for about eight seconds. During these eight seconds you will see three dots appear at the bottom of the display, until the display briefly shows ‘ESE ’ (which stands for ’Extra Settings‘). Now you are in setting mode and you can release the knob. The display will show the first setting that can be adjusted to taste.

This first setting is ‘bri’ or the display brightness. Alternating it shows the current setting. By turning the knob you can change this setting.

The next setting is selected by briefly pressing the knob. You will then see ‘CAL’. This setting calibrates the 0.0 volume point of the LS1 remote. If you turn the knob clockwise it shows ‘SET’, meaning it has set the 0.0 point to the current playback volume level. If you turn the knob counterclockwise, the ‘def’ indication tells you the 0.0 reference point has been set back to its default position.

By pressing the knob again you will see a ‘P’ in the display, informing you that you are now in ’IR Programming‘ mode and the device is waiting to receive commands from your IR handheld remote.

The Grimm Audio LS1r Controller is equipped with an infrared (IR) sensor capable of receiving IR commands. This allows you to utilize many types of handheld IR remote controls to control the main functions of the LS1r. Supported formats are RC5, RC6, and Sony protocols. You may of course also use Grimm Audio’s own IR remote control.

Grimm Audio IR remote.

IR controllable settings are volume, input source selection, and mute. You can step through the available controls by turning the knob:

“Volume up: Volume down: Volume mute: Channel up: Channel down: ”

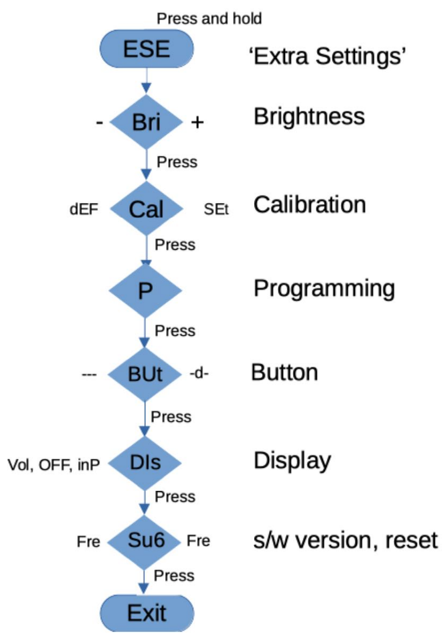
When a function is shown on the display according the above symbols, press the button of your handheld remote control that you want assign it to. If the transmitted format of your remote control is accepted, ‘PPP’ will show in the display. This means the command is stored and linked to the IR handheld remote button. You can repeat this action for the other buttons, and also for the same function to overwrite it. When a received code is already in use for another function, the display will show ‘P--’ and the command is not stored.

By pressing the knob once more, you will go to the next setting. This setting determines the function of a press on the knob in normal operation. The display shows ‘BuT’ for ’button‘: Turning the knob let’s you select whether this function is mute (‘---’) or dim (‘-d-’).

The next setting after another press of the knob is ‘dis’ or ’display‘. The options here are: ‘vol’, ‘off’ and ‘inp’, which means you can choose if you want your display to show the ‘volume’, be ‘off’ or show the ‘input’.

Pressing the knob now moves you to the next and last setting, which shows the software version of the LS1r Controller. ‘SV9’ for instance means ’Software Version 9‘. If you turn the knob in this menu you give a ‘Factory Reset’ and the display shows ‘fre’. If you instead press the knob once more, you will be back at the first ’brightness‘ setting.

You can leave the programming mode of the LS1r Controller by pressing and holding the knob for about 8 seconds. You will see three dots appear and when all dots are lit, the display turns back to normal mode again, briefly showing the ‘ESE’ message.



LS1 Control Software

The LS1 Control software is a standalone application that runs on a Windows, Mac or Linux computer. It is designed to control the LS1 via an LS1i USB interface, a UC1 Universal Converter or the Grimm Audio USB Control Cable. This software is needed during the LS1 setup and firmware updates, and can control the LS1 main functions. It may be used instead of or in conjunction with an LS1r Remote Control.

Please download the LS1 Control software for your Windows, Mac or Linux PC from the downloads tab of the LS1 web page and install the software. Note that if we release an update of the LS1 Control software, this will be notified via our newsletter, so we recommend to subscribe to our newsletter in the footer of our home page. You may then use the ’Check for update’ function in the LS1 Control menu of the application, which will guide you to the right place in our website again.

When you run the LS1 Control software, it will automatically recognize a Grimm Audio USB interface that is connected to your computer and start communicating with the connected LS1's. It will read the internal settings and check the firmware status. We recommend first time and hifi users to keep the ‘Control Options’ in the ‘LS1 Control’ menu (Mac) or ‘View’ menu (Windows/Linux) set to ’HiFi‘. The main display will then look like in the picture below.

In this main window the LS1 Control software offers access to the basic functions of the LS1:

At the top the current volume setting is shown. Use the slider to set your preferred listening level. You may also click on the + or – icons to nudge the volume up or down. When double clicking the slider dot the volume will jump to the ‘Reset Volume’ value that’s stored in the ‘Advanced’ section of the LS1 Settings page.

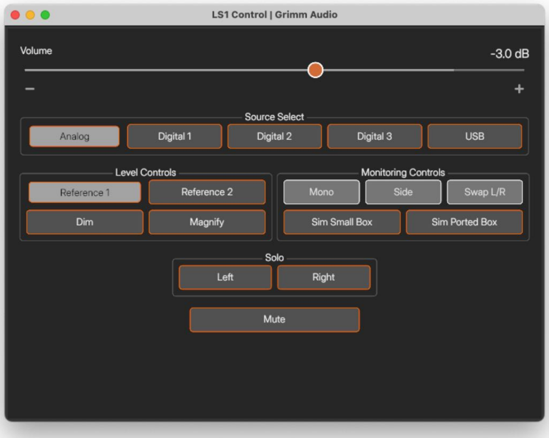
An LS1r Controller or UC1 in the system will show the same level as the LS1 Control on its display; it will follow the slider and the slider will also follow the LS1r or UC1 main knob setting.

A source can be selected with the radio buttons in the Source Select field. The indicated name can be changed in the settings menu.

Clicking the Mute button mutes the audio. Pressing the button again deactivates the mute function. When Mute is enabled, the power LED’s of the LS1’s will fade.

Pro settings

If you enable the Control Option ’Pro‘ from the ’LS1 Control‘ menu (Mac) or ‘View’ menu (Windows/Linux), three more fields with settings and options become available.



In the section “Level Controls” you can select what the ‘0 dB’ reference is for the volume control. You may choose from two reference levels, they can be set in the LS1 Settings. The volume level display shows how much above or below that reference playback level you are. The factory default setting for 0 dB in both reference levels is 79 dBC for a -20 dBFSrms pink noise signal, which is defined as ’K14‘ by mastering engineer Bob Katz.

When clicking an already selected Reference button, the volume slider will jump to the 0 dB reference volume. Note that when changing from Reference 1 to Reference 2 or vise versa (when clicking the non-selected button), the system does not change the output level, it just changes the display to reflect the volume setting with respect to the reference level you have set.

In the same “Level Controls” section you find the buttons ‘Dim’ and ‘Magnify’. The dim button dims the output by the value that’s set in the settings of the LS1 (default is -10 dB). The Magnify option does the opposite. It magnifies or boosts the output by 20 dB to track down audio artifacts or low level sounds like hum. It also turns on a limiter so the audio can never become loud. If you play music when using the Magnify function, an attenuated and distorted sound will be heard. This is normal. Use Magnify only for checking very low level sounds. When Dim or Magnify is enabled, the power LED’s of the LS1’s will fade.

The “Mixing Controls” section offers a few tools for checking the content of a mix. ‘Mono’ gives you L + R; ‘Side’ gives you L - R; ‘Swap L/R’ swaps the left and right channel; ‘Sim Small Box’ simulates a small bookshelf loudspeaker; ‘Sim Ported Box’ simulates a ported (“bass reflex”) loudspeaker. When any of these functions is engaged, the power led’s of the LS1’s will fade.

The Mono function sums both sides of the stereo signal into one mono signal, which is then played back on both speakers. If you want to listen to a ’real‘ mono source, you can additionally press either the ’L‘ or ’R‘ button of the Solo Section, which mutes one of the speakers. You will then hear both sides of the stereosignal mixed and played back on one speaker.

Note: this function is also a great tool when setting up your LS1's. Listen to the mono sum of your music in turn on the left and right speaker and move your loudspeakers until both sound identical.

The Side function calculates the difference between the left and right signals and plays them on both speakers. This is a very useful tool for checking the stereo content during recording or mixing and for monitoring MS processing during mastering.

The Swap L/R button swaps the left and right speaker channels, a trick used by mixing engineers to temporarily have a fresh look on the stereo balance of the mix. If the left and right speakers are physically reversed and you want to permanently swap them, please do not use this swap button but use the channel switches on the LS1 connection panels in stead.

The Sim Small Box button activates a digital simulation of a small box shelf type speaker. The Sim Ported Box button actives a simulation of a ported (“bass reflex”) speaker system. These functions can help the audio engineer to determine how a mix ’translates‘ to other types of speaker systems. When the simulators are engaged, the power led’s of the LS1’s will fade as a notification, since these functions are not meant to be used during normal listening.

In the “Solo” section you can solo a loudspeaker (the other channel will mute). Solo can be combined with the Mono or Side buttons and even ‘Sim Small Box’ button of the ‘Mixing Controls’ section to get an idea of how your mix will sound on a mono loudspeaker with limited frequency range. This may help you to make your mixing or mastering decisions. In solo mode, the power led’s of the LS1's will blink.

LS1 Settings

To change the settings of your LS1, choose ‘Preferences’ from the main menu of the LS1 Control application. The preferences panel will then open:

On the left you can see the main volume control and source selection. These are still active, so you can choose your input and change the volume while working on the settings.

At the top of the preferences page there are buttons that you can use to identify your speakers. By pressing the ‘L’ or ‘R’ box the corresponding speaker’s power led will light up strongly for 3 seconds, while all the others are turned off. This way you can quickly check if left/right is correctly configured. A second press on the button disables the channel check. Below the speaker button some basic information is shown. On top you see a possible volume offset for the loudspeaker (to correct an acoustic imbalance). The next line shows whether an EQ is engaged. The third line shows if an LS1 is configured as 2-way or 3-way (with subwoofer) system.

Global Settings

By opening the “Global Settings” menu you can see how many speakers are connected in your system and in which order. The firmware version of the LS1's is also indicated. Next, you can set the analog input gain. This control defines an offset to the main volume control so you can set the analog inputs to be equally loud as the digital inputs when switching between them.

The next options is for 3-way use only: the subwoofer phase correction can be set to accurate (= linear phase) with app 40 ms latency, or ‘Low Latency’. We recommend to only select the low latency mode in three use cases: 1. when using the LS1 in an AV system that cannot delay the video for 40 ms (= app. 1 frame), so lip sync is maintained; 2. when using the LS1 as a monitor for your musical instrument; 3. when manually placing markers during an editing session.

Lastly in this menu you can set the power led brightness to your preference.

Front Setup

In the “Front Setup” section you can set the parameters for the front speakers, starting with the toe-in angle. You can set this to the traditional 30 degrees toe-in (speakers aimed at the listener’s ears), or the 45 degrees toe-in that Grimm recommends. When the speakers are angled more inward, nearby side wall reflections are attenuated and the stereo image improves (see the “LS1 Positioning” paragraph in the “The LS1 Concept” chapter of this manual). This setting will determine what the linear frequency response axis is: 0 degrees or 15 degrees off-axis. A high shelf EQ in the DSP is set accordingly. We invite you to experiment with the toe-in angle and adjust this setting accordingly.

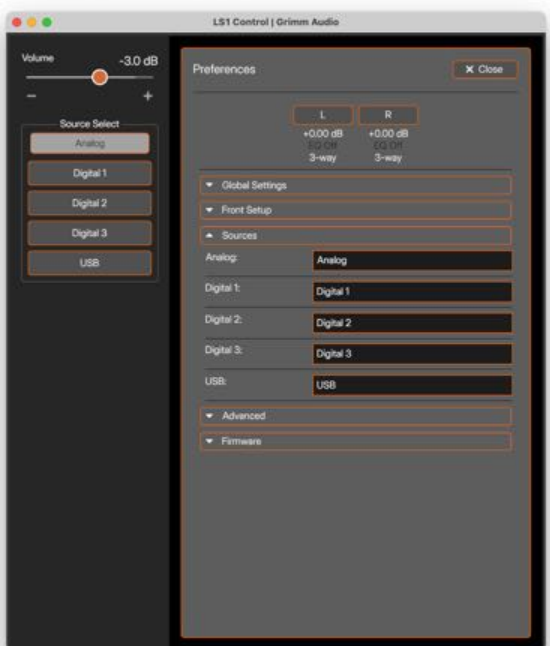
Next is a section with high shelf and low shelf equaliser and volume offset (‘balance control’). We recommend to try and solve sound color and channel imbalance issues first with speaker placement and acoustic treatments before adjusting the equaliser and changing the volume offset. And in general use as little EQ as possible. The high shelf EQ is tuned in such a way that it precisely compensates for the off-axis roll off of the (wide dispersion) tweeter of the LS1. It is the same EQ as used for the 30 or 45 deg toe-in angle setting. The low shelf EQ is tuned at the ‘baffle-step’ of the LS1, which is the frequency below which the loudspeaker radiates mostly omnidirectionally. The channels of the low shelf EQ can be unlinked and set independently. This can be helpful for instance when one loudspeaker is placed in a corner and the other channel just along the front wall. Below the baffle-step frequency sound waves bend around the loudspeaker. As a result, in the corner of the room more reflected energy adds up to the sound than along the wall. Adjust both EQ sliders until the low frequency levels on both sides sounds equally loud. Note that it might be useful to turn on the Mono switch and flip the Solo button between left and right when judging this balance.

To change a full spectrum imbalance in your room, you can use the Volume Offset sliders. You need to first unlink them to change the balance.

In the next section, the subwoofer type can be chosen. Turn it off for a 2-way system, select ‘Grimm’ if you have an LS1s, SB1 or LS1s-DMF subwoofer, and select 3rd-party if you have a subwoofer from another manufacturer. With the Sub Gain sliders the amount of energy below the selected crossover frequency (70 Hz in Grimm mode) can be set. The channels may be linked or not. The Sub Delay setting is intended for when the subwoofers are not placed on the LS1 footplate but in a different location. If the distance is not too large, the time alignment can be compensated (note that the speed of sound is app. one foot per ms).

Lastly the low cut (high pass) frequency of the system can be set. In 3-way mode we recommend a full range setting of 20 Hz, unless a strong 20-30 Hz room mode is excited. In 2-way mode we recommend a low cut of 35 to 40 Hz to ensure a high enough maximum SPL.

Sources

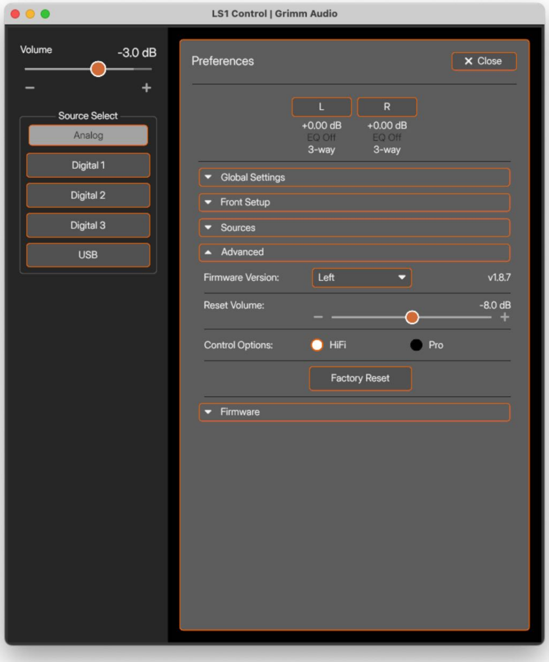


In the “Sources” section each source can be renamed to your liking by clicking the black text field.

LS1r
In case you connected an LS1r remote control, the menu item “LS1r” will appear.

The first setting determines the function of the main button. It will either mute or dim the output. Next is the choice of what the LS1r display shows: the volume setting, the selected source or nothing. Lastly the LS1r display brightness can be adjusted.

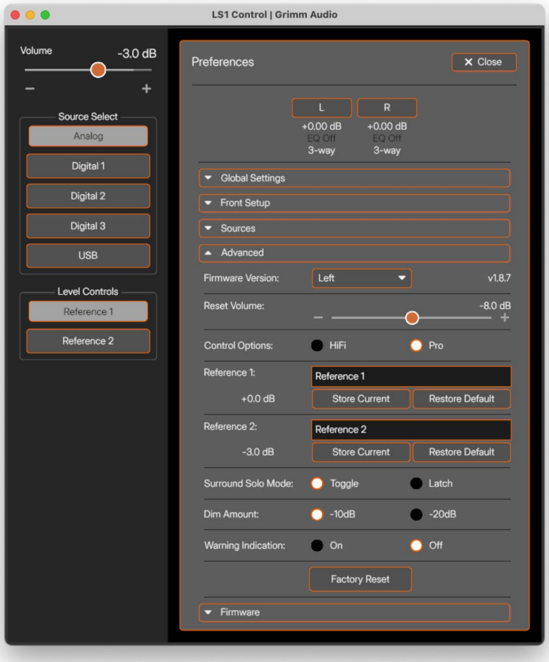
Advanced



The “Advanced” section has two views, for ‘HiFi’ and ‘Pro’ Control Options. In Hifi mode, at the top you can check the firmware of every LS1 in your system via a dropdown menu. In the next line a slider named ‘Reset Volume’ is shown. The value that you set here is the level that the volume slider jumps to when double clicking the volume bar in the main window. We recommend to not set this value too high to prevent inadvertently setting a high playback volume.

Below the Reset Volume, you can choose between the HiFi or Pro user interface when starting the LS1 Control. And lastly there is a ‘Factory Reset’ button. This button will reset all LS1 settings to the default value, turning off EQ’s etc.

In case you selected the ‘Pro’ Control Option, a few more settings appear.



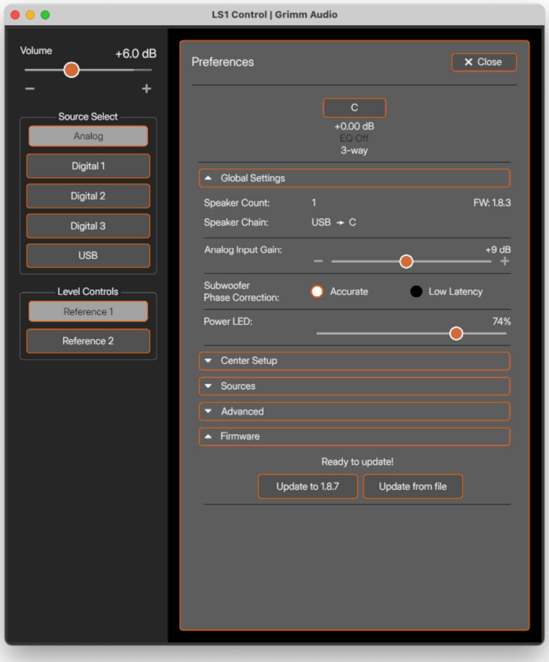
The ‘Reference 1’ and ‘Reference 2’ levels can be set here. This works as follows: first set the volume at the intended acoustical reference level (by ear or with a sound level meter), then click on ’Store Current’ for Reference 1 or Reference 2. In case you like to revert to the factory default setting of 79 dBC for a -20 dBFSrms pink noise signal, click the ‘Restore Default’ button. If you like you can change the name of the reference in the black text entry field.

‘Surround Solo Mode’ is a function that’s only applicable when you have connected more than 2 LS1's in your system. You may choose between a Solo Toggle (clicking another channel mutes the active one) or a Solo Latch (when clicking another channel the active one stays on – clicking an active channel mutes it).

‘Dim amount’ determines whether the Dim button turns the volume 10 dB or 20 dB down.

When the ‘Warning Indication’ is turned on, the power led’s of the LS1's will flash when the DSP signals that the system is driven outside its linear playback region. In a mastering studio it can be helpful to have an indication that the sound may have some distortion. However, please note that virtually all loudspeakers show distortion when driven hard. The LS1 is actually quite unique in that it maintains a relatively low distortion over large range of playback levels, above which it will suddenly show more distortion. Many loudspeakers already start distorting at lower levels, and then continue to gradually add more distortion. They will sound ‘loud’, but it will be harder to judge a mix on them at moderate playback levels. Since it is likely that at slightly elevated playback levels you will hit the warning level occasionally, we recommend to only turn this feature on when you wish to be warned. Otherwise the flashing can become annoying.

Firmware



Lastly there is a “Firmware” section that is used to update the LS1 firmware. Before performing an update, please read the firmware update manual. It can be found in the download page of the LS1 Control software and LS1 firmware. The quick reference is: Connect only one LS1 to the LS1i (without the LS1r), UC1 or the Grimm Audio USB Control Cable. The LS1 Control software comes with a built-in LS1 firmware version, which is indicated in the left button. If you have been notified that a newer firmware version is available you may download the firmware update file from the LS1 web page download tab and open it with the LS1 Control software by clicking the ‘update from file’ button. Repeat this procedure for every LS1 in your system. We strongly recommend to read the LS1 update manual for more detailed information about updating your LS1 firmware.

Presets

In the File menu of the application you will find the options ‘Open Preset’, ‘Save Preset’ and ‘Save Preset As’. An LS1 Control preset saves all settings. Opening such a preset restores all these settings. This for instance facilitates switching between personal EQ preferences in a studio that is used by several engineers.

UI Scale

A unique feature of the LS1 Control software is that the size of the window can be adjusted to taste. In the View menu of the application you can choose a size between 0.5 x and 2 x the standard size. In the Window menu of the application you can choose to keep the LS1 Control window ‘Always on Top’, which means it will always stay visible as an overlay in other applications. By combining ‘Always on Top’ with a small window size the LS1 controls are always available for you. This is a beloved feature of professional engineers.

Surround setup

Up to 7 LS1 loudspeakers can be controlled and setup with the LS1 Control software.
(This chapter is work in progress).

LS1 control via the MU1 GRUI

When using a MU1 with the LS1, its GRUI web page offers a similar functionality as the LS1 Control software, except for firmware updates. The visual interface of the GRUI has even almost the same layout, so you may refer to the LS1 Control description of this manual. For more in depth information, please read the MU1 software manual.

Specifications

Frequency response:
20 Hz - 27 kHz +0.5dB/-3dB.
30 Hz - 20 kHz +/- 0.5dB (unsmoothed!) Deviation from linear phase: <10 Degrees.
Amplifier power:
250 W (woofer); 100 W (tweeter); 400 Watt (per Sub).
Power consumption:
Idle power, drawn from mains: 16W per LS1.
LS1i Current drawn from USB: 250mA. Maximum allowed USB cable length: 2m.
Max SPL: (about) 105dB / 1m.
Max analog input level: 19 dBu.
Signal to noise ratio: 114dB (unweighted).
Crossover functions:
1550 Hz - 4th order Linkwitz Riley, sum corrected to linear phase.
70 Hz - 2nd order Linkwitz Riley (3-way system) sum corrected to linear phase in normal mode, not corrected in ’low-latency‘ mode.
Supported sample rates:
44.1 - 192 kHz.
Via USB with LS1i USB interface: 44.1 - 384 kHz ('DXD'), DSD64 and DSD128.
Level:
The default ’0‘ setting of the LS1 remotes volume setting gives 79 dBC at the listening position with a -20 dBFS reference noise (’K14‘).
Latency:
7.3 ms (2-way and in ’low-latency‘ mode).
40 ms (3-way).

Dimensions:
Height: either 1150mm for Hifi use (standard sofa height) or 1450mm for studio use.
Width, Depth: 520mm x 160mm.
Internal cabinet volume: 14 liters.