POWER CONNECTIONS

Depending on your model, three or four power connections are provided at the Phoenix power plug. They are labelled **R** for Remote input (turn-on), **+** for +12v, and **–** for Ground as well as a fourth connection labelled **P.** The P pin is a +12v external trigger intended to be connected to a toggle switch and is used to send phone calls to the center channel when Matrix is selected in the GUI. It can also be used to switch between upmix and stereo when Matrix is deselected in the GUI.

The SRP is designed to be start/stop capable and will operate from below 6v. The power section is reverse voltage protected and handles surge voltages in excess of +45v during cranking and load dump.

The SRP remote input should be wired to power on before the amplifier and shut down after the amplifier, so it is recommended to connect the SRP's **R** connection to the same trigger that turns on the system DSP. This is usually the headunit source or the key turn-on. This helps prevent any turn-on noise from making it to the speakers. The SRP has a 3-second turn-off delay after power is removed from the **R** pin to help ensure quiet shut-down.

Units equipped with the **P** terminal are configured to bypass the upmix when +12v is applied to this terminal. When active, the bypass sends mono information (the phone call) to the center speaker and difference information (background music that may be playing) to the left and right channels with zero latency. This prevents the caller from hearing an echo during the call. The user-provided switch should be a latching switch. The bypass will remain in effect as long as +12v remains present at the **P** terminal. Do not use a momentary switch for this feature.

This functionality can be changed with the new GUI firmware. Starting with GUI firmware version 1.0.0, the user can select pin function. The available options are Upmix/Matrix which provides the legacy phone functionality listed above, or Upmix/Stereo. This option allows a physical switch to control upmix for one seat and two seat tuning configurations. Both Stereo and Matrix are low latency options and either can be used for phone calls without the upmix delay.

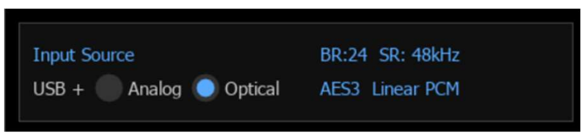
AUDIO CONNECTIONS

The analog audio inputs and outputs are located on the same side of the unit and are clearly labelled. The input section is fully balanced and employs a TI® SoundPlus™ differential input line receiver with exceptionally high common mode rejection, high impedance and ultra-low noise and THD. The input features a virtual ground circuit to eliminate ground loop noise. An Ohm meter will read zero ohms from the input signal ground to the power ground. The analog outputs are configured for single ended operation. In its default configuration the SRP can accept a 5v P/P input and provide 4V RMS output (11.3v P/P).

In addition to the analog inputs, the SRP can function as a driverless USB DAC through the USB Type B connector. Plugging an Apple® or Android® phone, tablet, DAP or PC into the USB input will allow direct audio playback through the SRP. The source device will see the SRP as an audio device, and the source volume control will control the audio level at the DAC ensuring that the full signal is sent to the SRP when supported by the source. The volume tracking functionality can be disabled in the GUI application to suit user preference. The USB connection *does not* charge the connected device while it is plugged in.

When being used as a USB DAC, the analog inputs are simultaneously active. If audio is present at both the USB and analog input at the same time, both sources will be mixed to the output. This situation should be avoided by either unplugging or stopping the USB source or by turning off the analog source. Use only one of the available inputs at a time or distortion is likely.

Your SRP is configured with an optical input. It will accept optical signals from 16bit 44.1Khz up to 24bit 192Khz. Optical input is resampled to the internal sample rate and bit depth for processing (24bit 48Khz). Your SRP will only lock to stereo 2-track (left and right) signals and will not decode DSD or Dolby Digital over the optical input. You can either use the analog inputs and USB or you can use the optical input and USB by selecting your preferred input in the GUI. It is recommended that you power the processor off and back on after changing your selection, but it is not strictly required.



Two LEDs are provided on the endplate.

LED A will illuminate green when a USB device has an active connection with the SRP. That can be a phone, tablet, or computer. The USB connection will pass audio and enable the SRP to function as an up-mixing DAC. The USB also functions as the data connection for the GUI application.

When used with an optical input, LED B (the blue LED) indicates that the optical path is synchronized with a valid optical signal from the source. The input setting in the GUI must be set to OPTICAL to use an optical input. Some sources will send a signal with digital silence between tracks and so LED B may remain illuminated when the system is on. Other sources do not send a keep-alive signal between tracks and LED B will turn off when music is not playing. When a valid signal is locked, the GUI application will provide the bit depth, bit rate, and framing data from the optical stream in text to the right of the selector. This LED does not illuminate when the SRP input is set to ANALOG.

The SRP does not have a power LED. Plugging a USB device into the SRP will illuminate the green LED A if the device is powered, and the blue LED B will illuminate if there is an optical signal present. With analog input, no LED are illuminated. If you are uncertain, plug in a USB device.

System Configuration

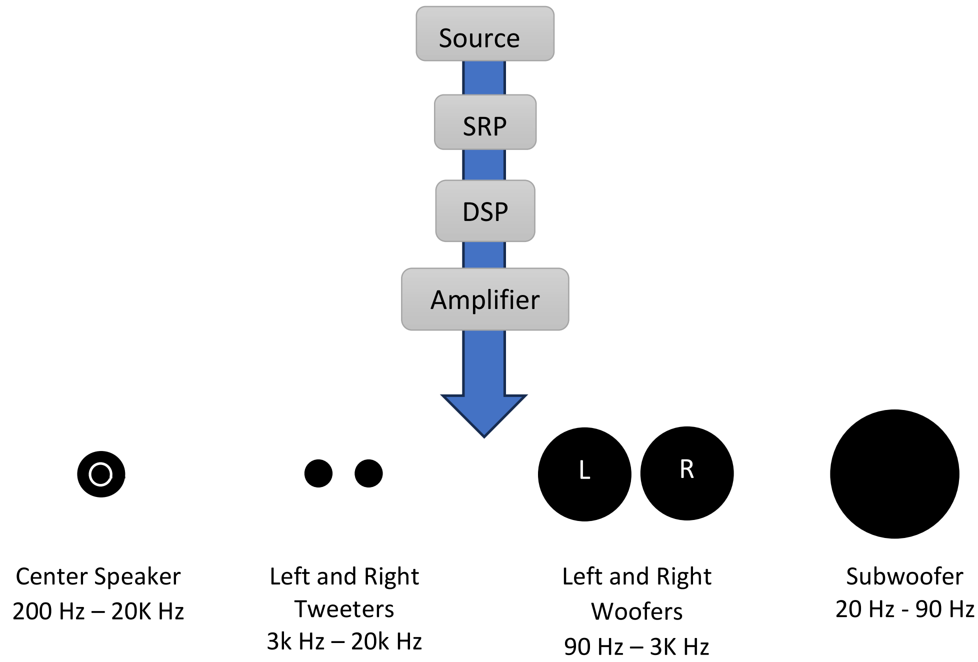
The illusion of a stereo phantom image relies on inter-aural time differences, amplitude matching, and phase consistency. In a room the stereo image is easy to perceive by listening from the apex of the stereo triangle. It is known to be a little more challenging in a car.

The SRP creates two stereo triangles. One is created between the listener, the left speaker and the center speaker. A second is created between the listener, the right speaker and the center speaker. The left of center (LoC) phantom image is created by the Left speaker and Center speaker playing the same information and arriving at the listener at the same amplitude, phase, and time. The left speaker and the right speaker no longer contain common information capable of creating a phantom image. All common (correlated) information is sent by the SRP to the center channel speaker and removed from the left and right speakers. The left and right speakers no longer create a phantom image for either listener.

Unlike a one seat stereo configuration, aiming a speaker at the nearest listener is not beneficial. Instead, the ideal is for the nearest speaker to sound slightly quieter than the more distant speaker. It is generally best to aim the speaker across the width of the car at the opposite seat (aiming the left speaker in the direction of the right seat listener) for speakers that will operate outside of the range of their omnidirectional pattern control. For speakers whose passband is entirely omnidirectional, speaker aiming is more about how the speaker interacts with the environment because the environment may force pattern control.

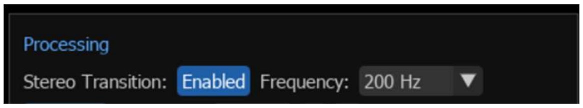
What follows is an example 7-channel system configuration. It is not intended as a recommendation or an ideal, just an example. The two-way plus sub setup was selected for simplicity in this example, but you can configure your system however you like. An upmixed system must have a left channel, a right channel, and a center channel and at least left and right channels should be capable of playing full range with the help of a subwoofer.

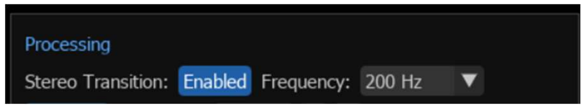
Below, you see an active speaker system with a two-way left and right channels and a single subwoofer. The center channel consists of a small two-way configuration in the dashboard playing 200Hz to 20kHz, and the side speakers consist of a woofer playing 90Hz to 3kHz and tweeter playing 3kHz to 20kHz. The system is configured as shown in the graphic below. Again, this is only an example for the sake of this discussion.



- The source supplies two channels to the SRP. This could be analog, optical or USB.
- The SRP sends three analog channels to the system DSP.
- The system DSP is configured with the crossovers as shown in the graphic. Those crossover points could be at any frequency the speaker supports.
- The system DSP mixing matrix should be set to sum L and R into the subwoofer. **Do not** sum Left, Right and Center into the subwoofer.

The small center channel speaker cannot play down to 90Hz like the side mid-woofers can, so the low frequency information present in the music cannot be reproduced from the center speaker. The sound information below 200Hz (the center speaker high-pass) and above 90Hz (the subwoofer low-pass) that is part of the center channel signal must be routed to the left and right woofers. That is the function of the stereo transition setting in the SRP.





The stereo transition setting in the SRP will route the portion of the center signal that the center speaker cannot reproduce over to the side speakers. Setting the stereo transition to 200 Hz in the SRP allows all of the information below 200Hz to be played back as part of the left and right channels. All of the information above 200Hz will be routed to the left, right and center speakers as is appropriate, and all of the information below 200Hz will be routed to the left and right channels. The mixing matrix in the system DSP should be configured to sum the left and right signals and send them to the subwoofer, and the 90Hz low pass crossover on the subwoofer and 90Hz high pass on the mid-woofers should be configured as it would in a stereo configuration.

If you did not enable the SRP stereo transition at 200Hz, then no speaker would play the center information between 200Hz and 90Hz and the vocals in the center would lack body and warmth. Also, any center-panned low bass would never make its way to the subwoofer. Tracks with off-center bass would sound correct, but tracks with center panned bass would sound thin. Do not disable the stereo transition setting unless you have Lef, Right and Center full range speakers with dedicated L, R, and C subwoofers.

Unless the Left, Right and Center channels are all full range and play from 20Hz to 20Khz, the transition setting must be turned on. To my knowledge the only car that meets this requirement is the MSE NASCAR which has a left, center, and right channel subwoofers. If you aren’t tuning the NASCAR, turn the transition setting on, and set the frequency to the center channel highpass.

Center channel speaker selection is vitally important. The center of the soundstage in many recordings is home to the vocalist, drum kit, and bass guitar. In a stereo environment those phantom images are the result of combined energy from the left and right channels. The combined surface area, power handling, and excursion capabilities and overall dynamic capability are challenging to replicate in a single center channel speaker. It is likely that you will find that the highpass of smaller midranges will need to be raised to handle the additional energy of high output reproduction.

Along with power handling, the ability of your car’s acoustic environment to keep as much of the left and right midbass in phase in either seat from the transition frequency towards the subwoofer lowpass frequency are a major factor in center speaker selection. Since that frequency range will be entirely in stereo, keeping the left and right midbass in phase and of equal amplitude will help improve focus and size in the chest of the singer and reduce image wandering on lower guitar, bass, piano, etc. The ‘easy way out’ is to use a very large center speaker capable of playing all the way down to the sub transition, but that is not strictly required. Every car and midbass installation is different, and midbass installed with longer path lengths than in the doors helps this significantly.

Small center speakers benefit from midbass that can be kept in phase with each other from either seat. Large center speakers fill in the hole when that simply isn’t possible due to the cabin acoustics and installation requirements.

Selecting a center channel speaker capable of playing as low as the side mid-woofers is ideal, but most factory speaker locations do not allow you to put a 5.25 or 6.5 (or larger) speaker there. The custom installation of a center mounted woofer (where possible) can considerably improve the placement of male vocalists and provide a tight, controlled image in the mid-bass region. If a center mid-woofer is installed, then it is possible to move the stereo transition in the SRP down to the high-pass frequency of the newly installed center mid-woofer. The center midwoofer – if present – doesn't need to be located in the top of the dashboard. Just like the left and right woofers which are often mounted in the doors, a low-mounted center woofer can be extremely effective without requiring a big hole in the dashboard!

The transition frequency is not related to the left and right midbass. The transition frequency is set at or near the center channel highpass wherever that is. That may fall at a midbass crossover boundary or it may fall in the middle of the midbass passband. Either is just fine. The transition frequency is NOT A CROSSOVER. It a bass management feature that is defined by the capability of the center speaker and decides where the low frequency information gets sent for playback.

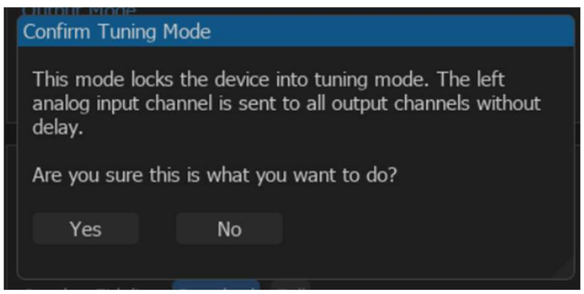
Caution!

Care must be taken not to exceed the excursion capabilities of the center channel speaker or damage will result. Select crossover frequencies appropriate to the speakers in use. The center channel speaker must reproduce the same energy as two speakers produced in stereo. This works the center speakers much harder than either left or right speaker in stereo. The center channel is very important in a 3-channel car and does a lot of work.

SYSTEM TUNING

The following general process for tuning a system that includes an SRP is intended as a starting point. Every car and system DSP is different, and each car will present unique challenges. **Before you begin, download the panned reference pink noise tracks (FLAC) from the website to facilitate tuning.**

- Verify system operation.
- Set system gains from source to amplifier outputs allowing appropriate gain overlap.
- Set initial crossover settings to protect all speakers within their operating passband using your system DSP. This includes the center speakers.
- Set the stereo transition frequency in the SRP. In most cases you should set the SRP stereo transition frequency at or just below the center channel highpass frequency. All audio below this frequency will be output in stereo and not be sent to the center speaker. Set up your system DSP to sum left and right channels at 50% for your subwoofer as you normally would. If this crossover is left disabled, the subwoofer will have little to no output on music that pans sub-bass to the center of the soundstage. This is true for most music, so this setting is important.
- Bypass the SRP. This can be done using the Tuning Mode function in the SRP, or you can wire your input source directly to the system DSP.



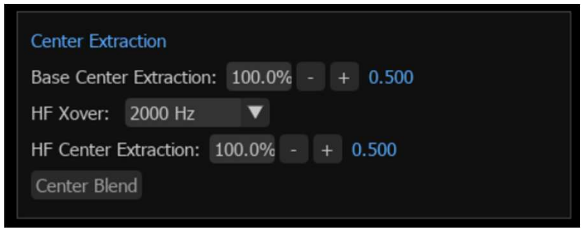
The SRP has significant latency from input to output, and Tuning Mode reduces that latency to a minimum to allow for impulse response measurements. Attempting to take impulse measurements with the SRP in normal playback mode will fail to resolve a peak because of the inherent delay.

- Mute the Left and Right outputs of the system DSP. Using a L+R signal, perform center channel time alignment from either seat. Align the speakers with each other, not with left speakers or the right speakers so that the center channel speakers arrive at either seat together.
- Tune the center channel to the chosen house curve. You can use either seat for this step. We recommend the seat without a steering wheel for comfort when tuning inside the car or the seat of your choice if using mic arrays. The left seat and right seat are in different acoustic environments and may have slightly different response curves due to the pedals, steering wheel, gauge cluster, etc. If you are tuning frequency response through the SRP (not bypassed or in tuning mode) then the low frequency response will be limited due to the stereo transition frequency setting in the SRP. This is expected. The rest of the energy is being sent to the left and right channels as a result of the Transition Frequency setting, and this gets tuned as part of the left and right channel tuning.

- Mute the Center and Right outputs of the system DSP. Using a left channel signal, perform left channel time alignment from the left seat. Tune the left channel to the chosen house curve from the left seat. Using best judgement, match the curve as closely as possible to the curve obtained in the center.
- Mute the Center and Left outputs of the DSP. Using right channel signal, perform right channel time alignment from the right seat. Tune the right channel to the chosen house curve from the right seat. Using best judgement, match the curve as closely as possible to the curve obtained in the center.
- Set the delay of the left channel speaker group so that left and center have the same time of arrival in the left seat. Set the delay of the right channel speaker group so that right and center have the same time of arrival in the right seat. The left delay and right delay are usually the same or very similar in this step. They should not be wildly different. Use whichever method suits your equipment for this step – Smaart, JL Max, REW, or a tape measure.
- Return wiring to normal or de-select Tuning mode in the SRP. Un-mute the DSP outputs.
- Using the L, LoC, C, RoC and R Pink Noise tracks that can be downloaded from the rivenaudio.com website, verify that the perceived loudness of each location is equivalent. Adjust center output level to the best approximation of equal perceived loudness. Move to the opposite seat to verify your choices. If the previous tuning steps were completed accurately these should be very close to correct.
- From the left seat, play the LoC pink noise test track. While listening, visualize the focus and location of the pink noise. Adjust the left side group time delay and – if necessary – the left output level to locate the pink noise at the optimal LoC location. The changes at this stage should be small. If you are delaying or adjusting levels by a significant amount, something is probably wrong. Double-check speaker polarity and phase. The pink noise should appear directly in front of you and may be somewhat diffuse or left-biased below the stereo transition set point.
- From the right seat, play the RoC track. Visualize the focus and location of the pink noise. Adjust the right side group time delay and – if necessary – the right output level to locate the pink noise at the optimal RoC location. If you are delaying or adjusting levels by a significant amount, something is probably wrong. Double-check speaker polarity and phase.
- From the left seat, play the 5 position tracks in sequence and evaluate relative loudness and location of each track. Move to the right seat and evaluate the relative loudness and location of each track. Within reason, and based on choices made during installation, the goal will be mirror images of staging, placement, and balance.

At this stage you should have the primary building blocks in place to begin refining what you hear.

Assuming the other GUI user controls had been left in their default configuration, both Base Center Extraction and HF Center Extraction should still be set to 100% focus. This usually gives an overly prominent center image. The stage may sound like alternating bright and dim lights at the 5 position locations.



The Base Center Extraction covers the range from the transition frequency setting to the HF Xover frequency which is defaulted to 2Khz. The HF Center Extraction adjusts focus above the HF Xover selection.

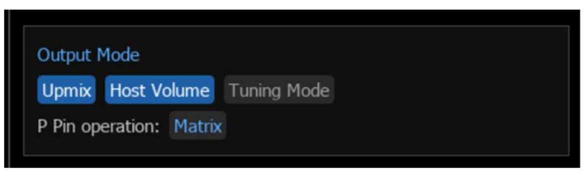
For many cars with factory-like installs, setting the Base Center Extraction value to 96% softens the center image slightly by reducing the degree of upmix. This setting is very car specific, but for most cars 100% is just too focused and puts too much energy in center. Reducing this level makes the center extraction slightly more stereo and spreads some of that energy back into the image. Center level goes down slightly, and the phantom image energy goes up slightly. The change in energy source moves the phantom images OUT towards Left and Right slightly. The lower this setting, the wider the placement of the phantom images. The phantom images move to the outside because this setting is shifting energy away from the center speaker into the side channels and the image moves to the new energy balance. Slightly increasing the amplitude of center or slightly reducing delay on the left and right groups moves the phantom images inboard.

This setting is split into a high and low value, and those values have a configurable center frequency. A setting at the default 2khz lets you soften the vocal range below 2k and tighten the range above 2k to retain high frequency focus while softening the image below that. When properly set for your environment, the center should be focused, but ‘in’ the mix; not in front of it. This control decides where in the soundstage the energy is placed. The lower the value, the more widely distributed that energy is. If the setting gets very low, you may need to increase the level of the center channels to bring the vocal back up in the mix. The center width setting and center volume must be balanced for your specific environment.

The image from left to right should be more uniform and the phantom images should have more energy and be better placed between far left and right. At this stage you begin to turn your attention to phantom image focus. Focus in these locations is usually driven by the focus and placement in the seat associated with the phantom. Left seat for the left of center and right seat for the right of center. The far side of-center phantom image has the least favorable geometry, and the focus of that image is strongly dependent on speaker placement and cross-car reflections, so maximum effort on the near-side phantom from each seat will usually approach the best-case focus to the far side phantom. Getting all 5 image locations in perfect location, focus and level from all listening positions is a combination of every decision from installation through tuning, so do not be surprised if the far side phantom is good but not perfect.

Further refinement may be possible using the tools you'd normally use to approach focus in a one-seat configuration. Using 1/3 octave barks between pairs (left and center being a pair) can help improve frequency-steered focus, and careful phase alignment can improve focus as well. Each of these interacts slightly with your other controls, so the deeper you go towards perfection, the more challenging maintaining the balance can be. Store presets often, and A/B between presets regularly to help judge direction and improvement. Although an initial image and placement is very easy, pinpoint refinement and perfect tonality is a challenge. There are many ways to achieve the end result, and the method described here may not be the best for your tuning style and approach. Experiment, and above all have fun!

Other GUI controls



The Output Mode section has 4 controls.

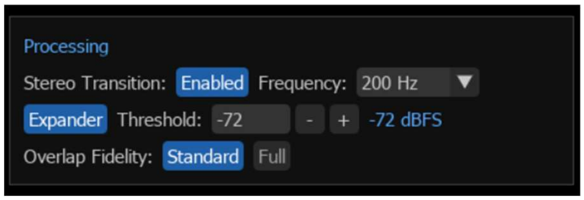
The first control, Upmix, is a permanent toggle that survives reboots. If this toggle is left in Upmix, then the SRP will reboot into Upmix mode. If it is Stereo, the SRP will reboot into STEREO mode.

The Host Volume control is only active when the input is USB. If Host Volume is enabled, then the volume control of the USB device directly controls the DAC output volume. If it is disabled, then the output of the SRP is always fixed at 100% and volume must be controlled somewhere else – like at the System DSP controller.

The Tuning Mode toggle. For the SRP Analog, this sends the Left Analog Input to the Left, Right and Center analog outputs with 1 sample block latency (128 samples at 48Khz). This is a permanent toggle that survives reboots, so the system will sound awful if you leave it in Tuning Mode after tuning.

The P Pin Operation toggle determines what the SRP will do when +12v is applied to the P pin. With no input to the P pin (grounded or floating), the SRP will do whatever the Output Mode button says. That should be set to Upmixed, so the SRP will upmix when nothing is applied to the P pin. When +12v is applied and the mode is set to Matrix, then the SRP will send any audio to the center channel with no latency. This is designed to allow clear, immediate phone calls over the stereo system and gets rid of the echo and sense you're talking over satellite. If the P pin Mode is set to Stereo, then the SRP will output direct-stereo. This can be used for phone calls or it can be used in conjunction with your system DSP to toggle one-seat and two-seat tunes. Doing this requires that BOTH the system preset and the SRP are in the correct mode at the correct time. If you have questions on how to implement this, please reach out to me at tluliak@rivenaudio.com and we'll talk through it.

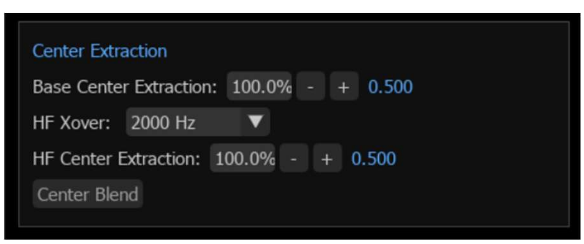
The Processing Section has 5 controls.



The first control, **Stereo Transition**, has been discussed above in Tuning. It should always be Enabled, and any correlated information below the set Frequency will be sent to the Left and Right channels.

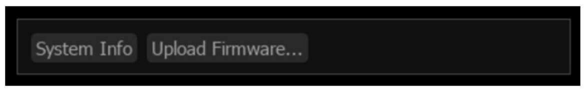
The Expander control is a downward expansion dynamics processing. The threshold of -72dB is generally good for USB and Optical inputs where the signal level is always known to be referenced to 0dB. Analog input is variable, so when the input is Analog or volume-controlled Optical and the volume level is very low, you will hear the muting effect at -72dB. Turn the level DOWN towards the floor at -96dB if that happens. If that remains inadequate, disable the function altogether. Below the Threshold, a 12dB knee reduces output at -1.5dB per dB. If the input signal is dropping 1dB, the output signal will drop at 1.5dB, so it makes quiet signals quieter. The purpose of this control is to expand the dynamic range by making the quiet parts quieter and increasing the attack rate of louder content without changing the frequency response of the system. The -96dB floor is the 16-bit dynamic range lower limit but we have 24bits of dynamic range within the SRP's processing, so this helps use more of the available dynamic range of the device and reduce background noise. If in doubt, disable it.

The Overlap Fidelity control chooses the mathematical specifics of the internal FFT processing engine. The Standard mode has the best time domain response. It uses a 50% overlap windowing scheme that reduces time-domain smearing on impulse-like inputs like a drumstick hitting the rim of a snare drum. The alternative Full setting uses 75% overlap to provide a smoother time domain response with slightly more pre-ring on fast transients. This is a preference setting, and the Standard setting is most correct.



The Center Extraction section is discussed in detail in the tuning section above. The noteworthy holdout is the Center Blend button.

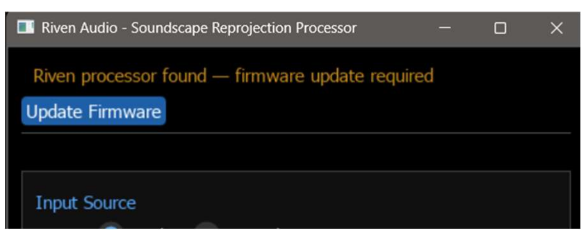
The Center Blend button is disabled by default. In a stereo system, human physiology causes the sound from the right speaker to be heard at the right ear and by the left ear moments apart, modified by a transfer function generated by the shape of the listener's head and ears. The same sound is experienced from the left speaker when the sounds are correlated. Any time the same sound is reproduced by two or mode sources, a comb filter develops. That comb filter is an inherent part of the sound of a phantom center image. With the SRP, the center image comes from a single speaker and lacks the comb filtered center signature we're used to hearing with a stereo phantom center. Engaging this feature enables an internally derived comb filter that approximates the stereo phantom comb. It sets the physical center back into the mix slightly and sounds 'more like stereo'. More specifics on the exact mechanism at play here is included in the hovertext for this control. This is a preference setting and defaults to off. **DO NOT TUNE WITH THIS ENABLED and then try to correct the combs it is intentionally generating!**



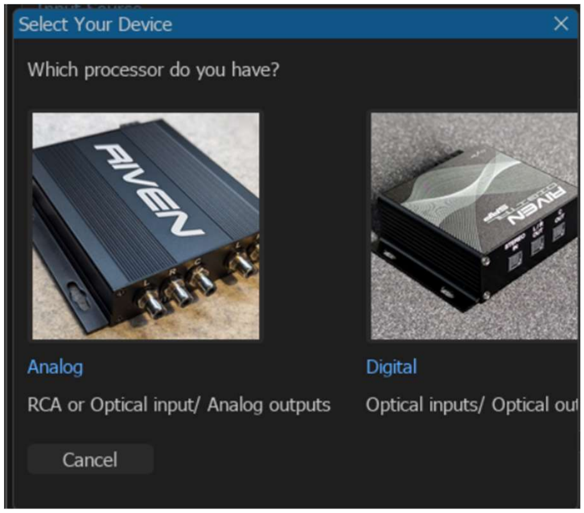
The System Info button embeds SRP operating information into the SRP log for manufacturer diagnostics purposes. The Upload Firmware button starts the firmware upload process. The software ships with firmware for both the Analog and Digital SRP, and will prevent you from loading the incorrect firmware for your device. If you manage to defeat that functionality and install the wrong firmware, please contact tluliak@rivenaudio.com and I'll get you back on track. It is extremely difficult – but not completely impossible – to damage an SRP with a firmware installation failure, so self-recovery of most faults is possible. If something went wrong do not panic, contact me, and we'll get you back to listening to music.

Firmware Transition – Pre-GUI to GUI

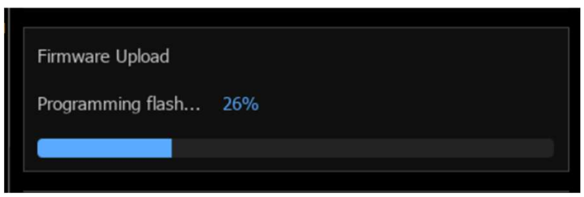
The first time you update from the old firmware to the new GUI firmware, the GUI recognizes your device and provides a wizard-like update interface.



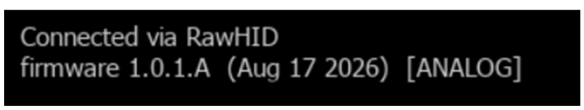
Clicking the blue Update Firmware button begins the process.



You will be asked to select your device. Click on the photo that matches your hardware. The firmware upload will begin immediately.



The SRP will reboot, and you should be greeted with 'processor connected' at the top of the UI and a notification of your current firmware at the bottom.



This indicates your firmware version, the build date, and the device type – ANALOG or DIGITAL.

Internal Diagnostics Logging

The SRP now has very sophisticated hardware diagnostics logging running on the device. Occasionally you may log into your SRP and notice a banner indicating the device experienced a problem and requests that you email a log file to Riven Audio. Almost all of these errors are non-critical in nature, and your log files help me learn what your SRP experiences to further improve the software and firmware that makes this spectacular device work. In those exceedingly rare cases that your log file shows an actual error, I may reach out to you with a fix or patch to correct it or replace your processor if the diagnostics indicate hardware may be experiencing problems.