Danville Signal Processing, Inc.

dspNexus 2/8

DSP Audio Processor

P/N A.03743A

User Manual

Version 1.0

Danville Signal Processing, Inc.
Product Name User Manual

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Overview

The dspNexus 2/8 combines the traditional role of a modern stereo preamplifier (control center) with a powerful DSP crossover. It is intended for applications where audio quality is of paramount importance.

We envision a product that will never apologize for its audio performance. Most key components are constructed as replaceable modules. Today's best DSPs, ADCs and DACs are tomorrow's second best. To this end, each module has readback capability so that as new modules are created, the dspNexus can automatically determine what was plugged into the motherboard and adapt accordingly. The dspNexus is software upgradeable without special programming tools. You just need your Windows PC.

You can think of a dspNexus as a device with multiple stereo inputs, a powerful DSP engine and eight high performance DACs that will be implementing an active DSP crossover This means that the dspNexus needs to be programmed with a crossover design that is optimized for a specific target loudspeaker system.

Crossover designs are created with DSP Concepts' Audio Weaver audio development platform. Audio Weaver is a Windows-based graphical design environment tool that uses optimized DSP software modules to execute production quality DSP software that runs directly on the dspNexus without the need to write computer code.

This manual covers the dspNexus 2/8 as implemented as a complete ready to operate product. A rotary encoder or an IR remote is used for volume control, source selection and adjustment of other parameters with information presented in the LCD display.

Most dspNexus 2/8 systems are resold by OEMs with optimized software that supports their product needs. This manual covers most of the basic features of the dspNexus 2/8, however there might be custom features implemented in the version of the dspNexus that you are using that are specific to the OEM provider. If you have an OEM version, the splash screen will typically have their name and product.

If you are a loudspeaker designer, you will be creating the DSP crossover design for your target. This means that you will be installing Audio Weaver on your local Windows computer. Tutorials, example signal processing canvases, training and other documentation about Audio Weaver are not covered in the manual.

The dspNexus 2/8 is also available in bare board configurations for integration by OEMs into custom housings. The OEM integration manual covers bareboard configurations with wiring details, physical dimensions and similar information needed to implement the boards into a custom enclosure.

Versions

The dspNexus is a growing platform. This manual is largely focused on the current released implementations. Since many of the subcomponents are module based, there are alternatives available. Early dspNexus versions can be field upgraded.

Current Models

All current models of the dspNexus use the fifth generation ADSP-21569 SHARC DSP. This DSP is twice as fast as the original DSP used in the dspNexus and also includes an independent FIR coprocessor.

dspNexus 2/8 with AK4499EX based DACs

P/N A.03743A-569-4499

This is the premium version of the dspNexus. The output DACs are based on industry leading state of the art AKM AK4499EX DACs.

dspNexus 2/8 with AK4493SEQ based DACs

P/N A.03743A-569-4493

This is the standard version of the dspNexus. The output DACs are based AKM AK4493SEQ DACs. This version still performs very well with a lower price point. In our view, the AK4493SEQ DAC is better than competing devices from other manufacturers.

dspNexus 2/8 with AK4499EX based DACs (4 channels populated)

P/N A.03743A-569-4499-4

This version is identical to the premium version with fewer output channels. The output XLR connectors are still available on the back panel allowing this unit to be upgraded in the future by adding additional modules. It is primarily sold aimed at customers that are only supporting two way systems or perhaps adding subwoofers to an existing passive system.

Legacy Models

Earlier dspNexus 2/8 systems used a fourth generation ADSP-21469 SHARC DSP. Most of these systems also used the AK4493SEQ based DACs. These systems can all be field upgraded with simple tools (module swaps) and software upgrades.

Hardware

The dspNexus 2/8 is powered by an Analog Devices' SHARC floating point DSP as implemented on a Danville dspblok DSP module. There are a number of pin compatible dspbloks that will work with the dspNexus. These choices will be discussed later in this manual.

As important as the DSP is to the platform, the audio quality is ultimately determined by the analog electronics, the data converters and the clock sources.

The dspNexus 2/8 has a variety of selectable input sources including:

- Balanced Analog (Stereo)
- USB Audio Class 2
- S/PDIF
- Bluetooth aptX HD
- Phantom Powered Microphone

The dspNexus 2/8 has the following outputs:

- Balanced Analog (Eight Channels)
- USB Audio Class 2
- S/PDIF
- Headphone Amplifier

Each interface will be discussed in detail.

If you take off the cover of the dspNexus, you will quickly notice that it is very module based.

Removing the Cover

Remove front panel allen (socket head) M3 screws (4)
Remove headphone nut (11mm)
Lay front panel down (it is connected to a ribbon cable)

This approach was chosen because the dspNexus is intended to be a living platform. Clearly, it might seem obvious why you might want ADC, DAC and DSP modules, since each of these are likely to be upgradable in the future. It also makes sense with other components such as the power supply, headphone amplifier and display.

Another advantage of modular construction is that variations of the dspNexus are more easily accommodated. This may be of more interest to OEM customers, but we are always open to suggestions from all users.

Motherboard

The Motherboard connects all the modules together. It has a variety of functions including control and I2S routing, local power supplies, low phase jitter master clock and distribution and expansion connections.

Minimal clock jitter is of prime importance for high quality audio. If a clock has excessive jitter, a perfect data converter would be transformed into something mediocre. The right data at the wrong time is the wrong data! The dspNexus 2/8 uses a very low phase jitter clock with very careful routing to minimize reflections to each device. The SHARC DSP has eight ASRCs that provide jitter attenuation far better than a low noise PLL is capable of achieving. This is very helpful for rate conversions (generally upsampling) and is absolutely imperative for S/PDIF and Bluetooth streams that are intrinsically noisy from a jitter perspective.

The Master Clock on the motherboard is 49.152 MHz. Normally the dspNexus ADC and DACs operate with a 192k sampling rate although 384k or 96k is possible.

There are really very few user oriented items to consider.

There is a DIP switch that is located near the DSP module. All the switches should be set in the OFF position which means they are oriented toward the front panel. The other combinations should be considered factory settings and should not be changed unless instructed by Danville.

If you are replacing a module, you should remove the IEC power cord or make sure the power switch on the rear entry is in the 0 (Off) position to insure that there is no power on the motherboard. This includes standby current that exists in Sleep mode (see Operation Section). Modules require the removal of four screws and then can be removed by carefully lifting the module from multiple sides so that you do not bend pins or damage the mating connector.

DSP Module

The DSP Module is a Danville dspblok that is preprogrammed with a licensed version of DSP Concepts' Audio Weaver. It also includes a Danville bootloader and a supporting Audio Weaver platform file that connects Audio Weaver to the hardware.

Danville dspbloks have been manufactured for over 20 years in some form or another. Each dspblok used in the dspNexus supports USB Audio Class 2, external DDR SDRAM and non volatile memory. The heart is the dspblok is a fifth generation Analog Devices SHARC DSP. The SHARC has long been the gold standard of high performance audio floating point DSP processing.

In the dspNexus design, the SHARC DSP is the master processor. All other devices including the display microcontroller are subservient to the SHARC once the dspNexus is fully powered. The exception to this is when the dspNexus is in Sleep Mode. In this case, the SHARC and almost everything else is powered down with the exception of the display microcontroller.

The DSP Module firmware is upgradable via the USB port using a Windows computer. Danville will provide instructions and files if this is needed.

Over time, it is probable that the DSP Module (dspblok) could be updated with a faster DSP. Of course, this assumes that a faster DSP would have an advantage in your crossover design.

ADC Module (Analog Input)

The dspNexus analog inputs are balanced and uses a standard female XLR connector that is very familiar to the professional community segment and increasingly common to higher performance consumer audio. Balanced audio should have been used consistently for audio a long time ago, but unfortunately it wasn't. This is starting to change with higher end audio electronics and we support this direction.

The analog input circuit is a low noise balanced input with very high common mode rejection. The gain is programmable in eight 6dB steps to support a range of inputs from +4 dBu professional audio levels to moving magnet and high output moving coiling phono cartridges.

The ADC module uses an AKM AK5578, considered by many to be the best ADC on the market. This ADC is actually an 8 channel ADC that is configured to use four ADCs per channel. It has a S/N and DR of 127 dB. This is about 30 dB better than a CD and 57 dB better that the theoretical noise floor of a perfect vinyl record before it is pressed.

The differential input impedance is 47K with switchable capacitance loading. There are several use cases for the analog input, line level inputs, microphone inputs and phono inputs. In general, you want input levels to be set so that the ADC will not overload but use most of its dynamic range to maximize the signal to noise ratio. Levels are selected via the front panel controls.

Line Level Inputs

Line level in audio circuits is usually defined as having a standardized gain such as -10 dBV (unbalanced consumer) or +4 dBu (balanced professional). It gets a little more complicated when balanced audio is used with some consumer audio products since -10 dBV could be the level from hot to cold or hot to ground which would be 6 dB different.

Setting the input gain to 0 dB is appropriate for +4 dBu balanced inputs and +12 dB for unbalanced -10 dBV inputs.

Some customers are going to need to interface with single ended (unbalanced) sources (RCA phono). This can be done in two ways, the "OK" way and the better way. In both cases, you want the center HOT of the phono to connect to pin 2 (HOT) of the XLR connector. Ground of the phono connects to both pin 3 (COLD) and pin 1 (GND) of XLR. The difference between the two methods is where the pin 2 and pin 3 connections occur.

WARNING !!!

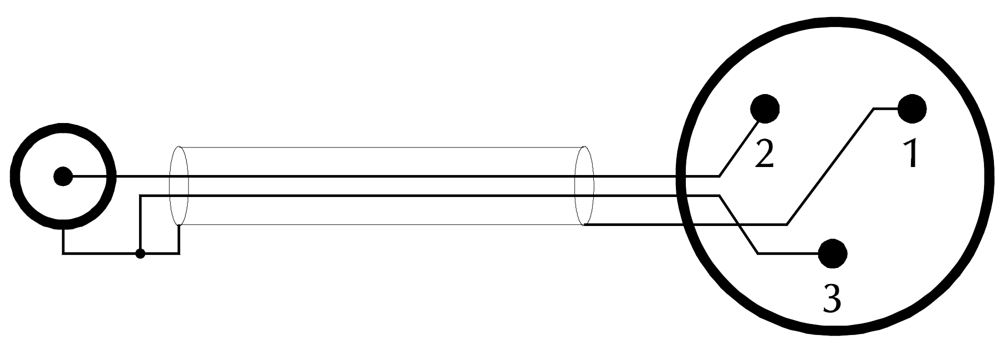
This wiring method is only valid for inputs.
Unbalanced outputs are wired differently so that Cold (Pin 3) is not driving Ground

OK method:

This is likely a RCA to XLR adapter connected directly to the dspNexus The connecting cable would be a phono to phono shielded cable. The advantages of the balanced input is largely eliminated in this configuration. It is convenient, but certainly not ideal.

Better method:

Use a two conductor shielded cable. You connect the RCA ground connection to one of the internal wires and the shield. The shield and internal wire are isolated on the XLR side. The picture illustrates this method.



Microphone Inputs

There are two ways to connect microphones. If you have a +48V phantom microphone, you should use the front panel XLR. This actually uses another ADC located on the motherboard and disconnects the ADC module. The alternative is to use an external microphone / preamplifier combination. This may or may not have gain. Follow the same procedure as line inputs and just set the gain accordingly.

If you use the front panel phantom powered microphone input, you can use an inexpensive measurement microphone such as a Dayton Audio EMM-6, Behringer ECM8000 or PreSonus PRM1. You can also use a high performance measurement microphone from GRAS and ACO Pacific. These companies make phantom powered microphone preamplifiers that work with compatible microphone cartridges.

A stereo ADC dedicated to the microphone input is on the motherboard. It samples at 192 kHz. One of the eight DAC outputs is assigned to the remaining channel. This makes it easy to measure the relative time delay of each driver since the two channels are time aligned.

You can also use the Analog Input on the back panel for measurements. In this case, the right channel will be the microphone input (without 48V phantom power) and the left channel can be used as a reference. You would need to patch cables for the reference inputs in this case. You might use the analog inputs if you have a measurement microphone/preamplifier that is not phantom powered.

Generally, test programs will use the USB UAC2 interface to connect to an external computer for the measurement functions.

Phono Inputs

In this section we are focusing on a digital RIAA preamplifier implemented in the DSP and configuring the Analog Input accordingly. You can certainly use an external analog phono preamplifier. In this case, refer to the Line Level Input discussion.

This section of the manual concentrates on the hardware aspects of phono cartridge / turntable interfacing and not the equalization / signal processing that will be implemented by the dspNexus.

Phono cartridges are generally one of three types, moving magnet (MM), Moving iron (MI) and moving coil. (MC) Moving magnetic cartridges usually have a nominal output sensitivity of 4-5mV and expect an input resistive load of 47K. Moving coil cartridges have much less output and should be loaded into a small impedance. In general, the analog input gain will be set to the maximum gain of 42dB and in some cases addition gain will be added in the signal processing section. Keep in mind that the S/N of the ADC is very high with respect to the intrinsic noise floor of the cartridge and vinyl records.

Moving Magnet (MM):

Some MM cartridges benefit from capacitive loading. This causes the resonance of the cartridge to fall which tends to boost the frequency response below the resonant frequency, mostly in the 10 to 20 kHz region. Some manufacturers rely on this effect. The inductance of the phono cartridge factors into the exact resonant frequency. If you add enough capacitance, the resonance will fall into the audio band below 20 kHz. This will tend to boost levels in the upper midrange at the expense of the last octave (10-20 kHz region). You may or may not like this effect. Let's just agree that optimal capacitance loading is a bit subjective.

You can adjust the load capacitance of the ADC module using DIP switches. This will be discussed below. The capacitance load will be the input capacitance of the preamplifier and the cable capacitance (~100pF).

Since cartridge inductance is important as well as your high frequency hearing, you may not hear much difference as you change capacitance loading. It is probably useful to see what the cartridge manufacturer recommends for their situation. One more thing to remember, if the vertical tracking angle of your cartridge is too high, the sound will also be a bit bright at the cost of the bass. There is a goldilocks zone with VTA.

Moving Iron (MI):

Grado makes moving iron cartridges that have high output (HO) and low output (LO) versions. You can treat the high output type as essentially MM and ignore capacitance loading. The LO type have a sensitivity of 0.5mV which means that make up gain should be added in the signal processing block. The 47K input resistive impedance is fine in both cases.

High Output Moving Coil (MC)

Moving coil cartridges vary considerably with respect to both gain and optimal loading. If the output of a moving coil cartridge is sufficiently high and the load impedance is not too low, then it is reasonable to interface a cartridge directly into the dspNexus with the addition of a smaller load resistor added to the connector. For example a Denon 103 has a sensitivity of 0.25mV and a reasonable load resistance of 400 ohms. You can add a 402 ohm metal film load resistor across pins 2 & 3 of the XLR connector on the cable as an input load. The input capacitance should be small. Gain will be made up in the signal processing section in this case.

Low Output Moving Coil (MC)

Low output moving coil cartridges need extra gain from an external step up transformer or head amplifier. These devices are usually designed to load directly into MM preamplifiers and therefore the 47k of the dspNexus will work fine. In this configuration, you will not add additional capacitance loading.

Connecting your turntable

Your phono cartridge is essentially just a coil. In most cases, the shield of an RCA Phono is isolated from the turntable chassis. You want to use a cable with a twisted pair for the cartridge, and a shield for GND. Connect the Phono center pin to XLR Pin 2 (HOT), Phono GND to XLR Pin 3 (COLD) and the turntable chassis ground to XLR Pin 1 (GND). These connections should be made at the turntable end of the cable in a similar manner as the "Better" method" in the Line Input Section. If your cartridge/tonearm is actually grounded, you might use the "Better" method as an alternative.

Capacitive Loading

The ADC Module has a pair of DIP switches that adjust the capacitance loading at the input. These serve two purposes, RF suppression and phono cartridge loading. Keep in mind that cables add additional capacitance and small stray capacitance has been ignored in the table.. Switches 1,2 & 3 are used for capacitive loading of phono cartridges. Switch 4 is used for RF filtering. They are mutually exclusive so not all combinations are desirable.

Capacitance (pF)	SW1	SW2	SW3	SW4
0	Off	Off	Off	Off
50	Off	Off	On	Off
100	Off	On	Off	Off
150	Off	On	On	Off
220	On	Off	Off	Off
270	On	Off	On	Off
320	On	On	Off	Off
370	On	On	On	Off
RF Filtering	Off	Off	Off	On

DAC Modules (Analog Outputs)

The dspNexus analog outputs are balanced and uses a standard male XLR connector. Each DAC has individual 3dB analog attenuators. The full scale range of the DAC should ideally correspond to the maximum voltage swing you want from your amplifier. This will maximize signal/noise of the DAC.

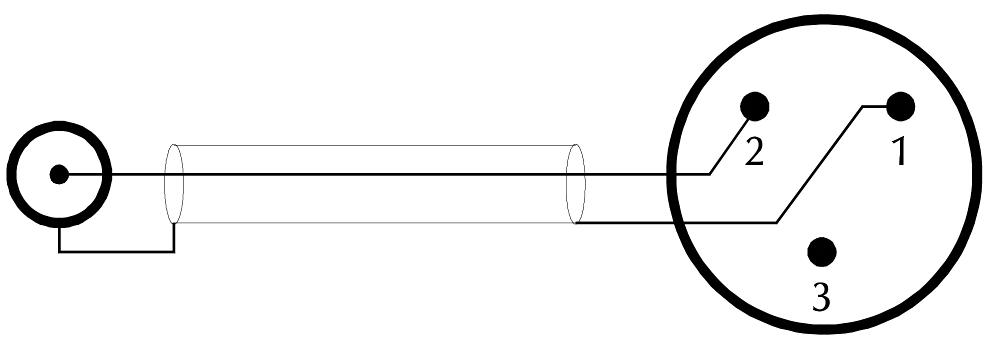
With woofers, this is often the maximum power of the amplifier, but sometimes you may want to restrict the level below full scale.

Full scale (0 dB) corresponds to about +22 dBFS which is appropriate for +4 dBu professional audio levels. This is likely too hot for most power amplifiers. Unfortunately, there is no standard voltage gain standard for power amplifiers. Typical gains are 20 to 30 dB which probably means that the attenuators should start about -6 to -9dB

In addition to the analog attenuators, there are finer adjustments so that amplifier gain can be set in 0.25dB increments. This is useful when mixing amplifier types or setting up subwoofers. Assuming that you are using the dspNexus as a crossover, remember that adjusting gain of the individual DACs will change the crossover levels independently for each driver.

Amplifier Connections

Ideally, your power amplifiers have balanced inputs which is preferred if you have a choice. Unbalanced connections can also be made but the cables are wired differently than input cables. Both the HOT and COLD outputs are driven in the dspNexus. This means that you DO NOT connect COLD (Pin 3) to GND (Pin 1) in an unbalanced circuit or you will effectively be shorting the COLD output to GND.



Recommended DAC Channel Assignments

Since the dspNexus is probably configured as a crossover, each output will be assigned to a different driver. It is important to make sure that cables are not crossed or you could be driving low frequency, high level content into a tweeter.

The dspNexus 2/8 convention with is to use channels 1-4 for the right speaker and channels 5-8 for the left speaker, starting with the highest frequencies in the lowest channel.

Here is a 3 way example with optional subwoofers:

Ch 1	Right Tweeter
Ch 2	Right Midrange
Ch 3	Right Woofer
Ch 4	Subwoofer (Mono) or Right Subwoofer
Ch 5	Left Tweeter
Ch 6	Left Midrange
Ch 7	Left Woofer
Ch 8	Subwoofer 2 or Left Subwoofer

Here is a 2 way example with optional subwoofers:

Ch 1	Right Tweeter
Ch 2	Right Woofer
Ch 3	Subwoofer 3
Ch 4	Subwoofer (Mono) or Right Subwoofer
Ch 5	Left Tweeter
Ch 6	Left Woofer
Ch 7	Subwoofer 4
Ch 8	Subwoofer 2 or Left Subwoofer

Headphone Amplifier

Sometimes people want to listen to music when others may not want to share their experience (perhaps they're sleeping?). Since most of the supporting circuits already exist in the dspNexus, this was really just a matter of creating a small power amplifier with appropriate signal routing. The dspNexus has a very good headphone amplifier.

The headphone amplifier is engaged when a headphone is plugged into the front panel 6.3mm (¼″) phone jack. DAC channels 1 & 2 are rerouted to the headphone amplifier with independent signal processing and gain settings. The analog outputs are automatically muted on the back panel.

Power supply

The dspNexus has a universal power supply that accepts 85 to 265 VAC inputs at 50/60 Hz. There is no need to configure the mains voltage. The power entry uses a standard IEC power inlet. You may need to acquire your own mains cable depending on your location since mains termination is not globally standardized.

The IEC power entry has a power switch. This cuts all power to the dspNexus. Most of the time, power is managed via the front panel controls. In this case, the unit will be either fully operational or in a low power sleep mode.

In sleep mode, there is a low power microcontroller that manages power to most of the dspNexus. In sleep mode, the DSP, ADC, DACs and most other circuits are powered off.

Display, Remote & Encoder

The encoder/display module is used to control the dspNexus, but it also has other functions that may not be as apparent.

There is a low power microcontroller that is always on when the dspNexus has power applied via the (power inlet switch on). The microcontroller is used to actually turn on power to the main dspNexus electronics including the DSP and converters. This allows the dspNexus to idle in Sleep mode and consume very little power when not actively playing music.

In addition to the rotary encoder, the dspNexus has an IR receiver that utilizes an Apple compatible remote to control the dspNexus. The remote is the easiest way to operate a dspNexus. Operation is discussed in the next major section of this manual.

There is also a Bluetooth module located on the display module. This module supports aptx-HD for near CD quality audio streaming as well as A2DP that is universal, but lower quality. The dspNexus automatically upsamples and jitter attenuates the audio stream to 192k. We consider Bluetooth operation to be convenient, but certainly not high performance audio.

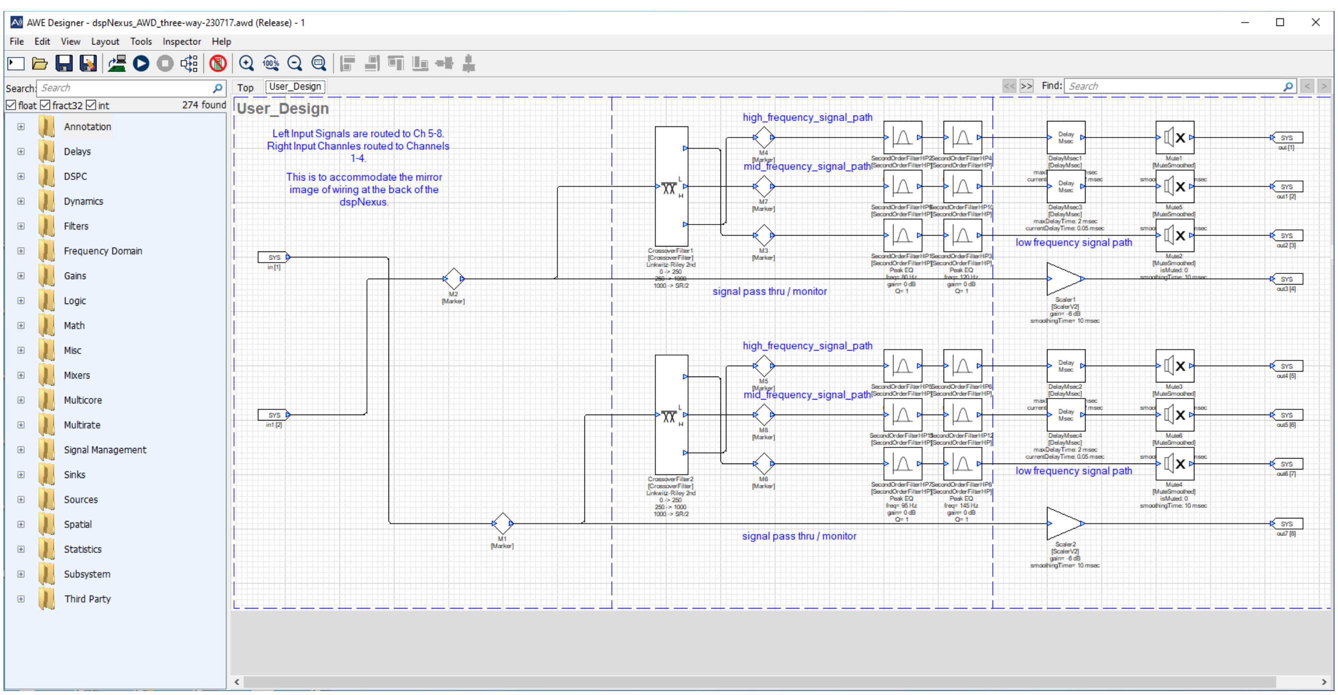
Software

The dspNexus comes pre-installed with DSP Concepts' Audio Weaver. Danville has a licensing agreement that allows you to design your DSP crossover without buying additional tools or licensing. All you need is a Windows computer.

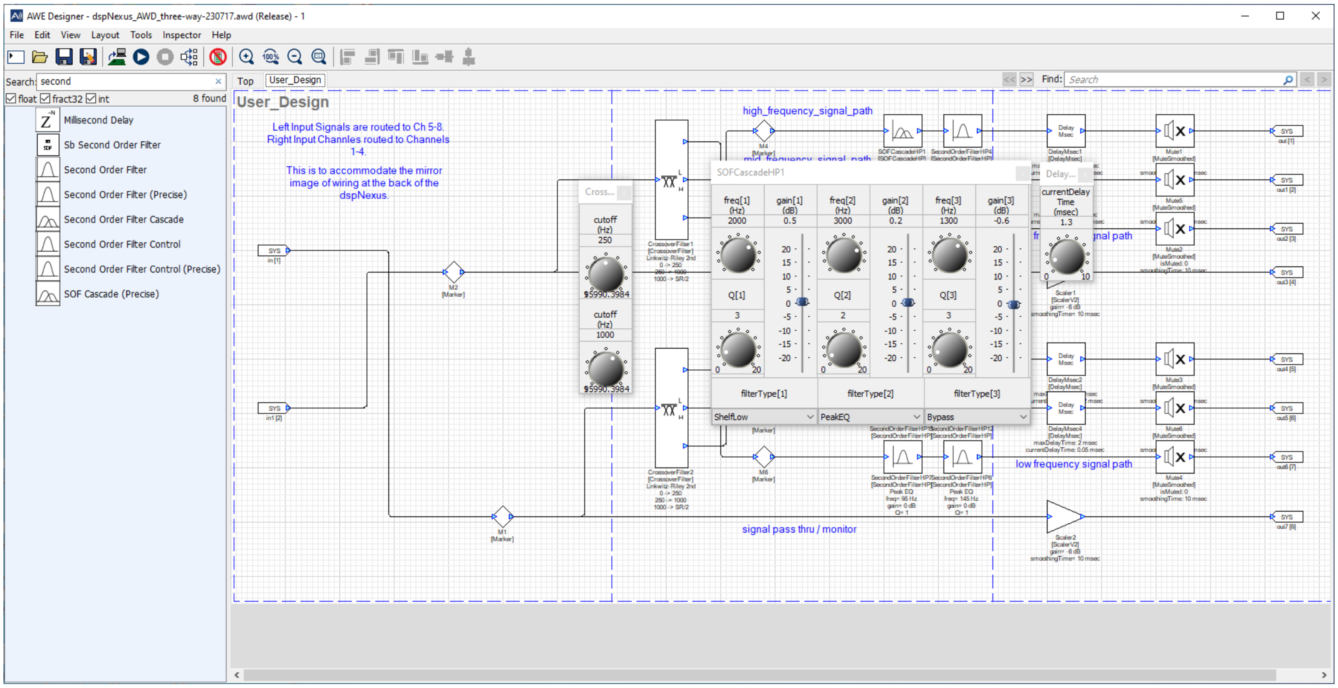
In many cases, an optimized crossover may already have been designed by your speaker manufacturer. If this is your situation, you can safely ignore this section. You crossover already runs on the internal DSP and adjustable parameters will be accessible via the Remote/Encoder Display.

Audio Weaver is a graphical design program that includes hundreds of optimized modules. Systems are designed in a drag and drop fashion to create signal flow. Audio Weaver runs natively on the SHARC DSP with optimized production quality modules. You design the signal processing system without the need for custom programming. Once you have your layout, you can adjust parameters in real time. For example, you can create a crossover design, make some measurements and listening tests, and then adjust parameters to fine tune the system without interruption of the test source or music.

DSP Crossover design might seem a bit daunting at first. As part of our support program, we will help you get started by providing example designs, training videos and a private cloud based support directory where you can share your specific configurations with our staff. We can also answer questions via video conferencing. In addition to the mechanics of using the program, we might discuss signal flow architectures choices such as IIR versus FIR, bandsplitting options (Butterworth versus Linkwitz-Riley) and other useful features.



The above design called an Audio Weaver canvas illustrates a typical crossover design for a one input, four output system. This one uses second order IIR filters. FIR type filters are also available.



If you double click on a module, an inspector box opens up. In the example, we opened three inspectors.

Inspectors allow you to adjust parameters for each module. These can be changed in real time so that you can listen to changes as they are made. The inspector in the middle (SOF CascadeHP) demonstrates an 6th order IIR filter made from three biquad sections. This module could have been configured to be much larger or smaller depending on your requirements. You can see the parameters that are adjustable in this instance. The inspector would have looked a bit different for other configurations.

One of the features we really think is important is that Audio Weaver includes a complete set of high performance filter modules. The SOF CascadeHP is one of these modules. Precision is extremely important when creating filters with high Qs and low frequency corners, a.k.a woofer adjustments. There can be a 60dB signal to noise difference between an ordinary filter and the special high performance filters built into Audio Weaver.

Audio Weaver also supports subsystems. A subsystem is a module that expands into another sheet when you double click on it. It is useful when you want to create a fairly complex component and then not need to look inside it all the time. A very special subsystem is needed in dspNexus to configure the hardware. It is included the example illustration (module in the lower left corner) and discussed in later in this manual.

Installation

There are a few basics that almost all dspNexus systems are going to need to consider. You are likely to have many power amplifiers and you are probably going to connect to an external computer. This section addresses these issues.

Since the dspNexus is likely implementing a DSP crossover it is very important that amplifier assignments are implemented so that tweeters do not see low frequency, high power content. This was discussed in the hardware section. We suggest that you use color coded XLR connectors to help avoid cable mishaps. In a new installation, you should power up the woofers first, then midrange to make sure that the right content is going to each driver.

Power amplifiers have very different sensitivities. If you are integrating different amplifiers or adding subwoofers, you are going to need to level balance the system. This can be done through the setup controls using the Remote or Encoder in 0.25dB steps. The procedure is discussed in the Setup Operation Section of this manual. Subwoofer integration will also need time alignment adjustments since this will be room dependent.

If you are connecting to an external music server or computer, you are going to need a USB Audio Class 2 driver. In a Windows machine, you will need to install a device driver as a minimum. If you are using Audio Weaver, there will be additional software installation instructions.

Amplifier Connections

As discussed in the Hardware Section, the convention is to use lower numbered outputs as the higher frequency channels. For example, channel 1 is assigned to the highest frequency driver (tweeter). Since the dspNexus is generally supporting a stereo system, the sequence repeats at Channel 5.

If your power amplifier only has single ended inputs (RCA), then connect the center pin to the Hot (Pin 2) connection and the Ground return to Pin 1. If for some reason, your amplifier is inverted, you could also use Pin 3 for the Hot connection. This will flip the phase.

WARNING !!!

Never connect Pin 2 or Pin 3 to Ground. You will drive the DAC output into a near short since both halves of the balanced outputs are driven.

Ideally, you have balanced inputs on your amplifiers. In this case, you will just need shielded XLR male to female cables. If your cables are not color coded, you might want to wrap colored tape on the ends.

The dspNexus is capable of driving professional audio levels which is usually too hot for most power amplifiers. In general, you want the maximum level of the DAC is be near the full scale voltage of the mating power amplifier. This will maximize the available signal to noise and headroom of the system. The DACs have internal 3dB attenuators and additional 0.25dB steps to optimize these settings.

Input Connections

Wiring considerations are discussed in the hardware section.

Control: Operation

While it is possible to run the dspNexus from a computer using Audio Weaver, this section is going to assume that you have a crossover designed and operating already and that you are now using the dspNexus as a control center. In this case, you will be using either the Remote or the front panel Encoder. The Encoder is located next to the display. The Remote is a seven button Apple compatible, IR transmitter that allows you to control the dspNexus from your listening position. Both devices control the same parameters, but the Remote is easier and probably more convenient.

Control is split up into two main areas: Operation and Setup. Operation essentially selects the source and adjusts the volume. It is the mode you use 99% of time and therefore easy to access. Setup is used to configure the dspNexus to the operating environment of the room and companion equipment and to set user preferences.

Display/Encoder/Remote Operation

The display/encoder consists of a 2x16 alphanumeric display and a rotary encoder for control. The encoder rotates continuously and also has a pushbutton shaft switch.

There is also an IR receiver that allows alternative control from the Remote.

Ergonomically, the Encoder and Remote work in very similar fashion. Both controls can operate at the same time.

The Encoder functions are as follows:

- Rotating clockwise increases the selection or moves toward the end of a list.
- Rotating counterclockwise decreases the selection or moves toward the start of a list.
- Normal Click (pushing like a mouse click) is a normal selection, works like an Enter key.
- Doubleclicking (similar to a mouse) toggles mute.
- Long Click (holding the encoder button down for 2 seconds).

The Remote functions are as follows:

- The Up and Down Arrows increase/decrease the level (i.e. Volume).
- The Left and Right Arrows rotate through the selections (i.e. Menu).
- Center functions like an Enter Key. There is no doubleclick functionality.
- The Pause Button functions like a doubleclick (mute).
- Menu is a hot key that eliminates the need for the Source Selection mode.

There are three main modes and one temporary mode (Splash) when using the Encoder:

- Splash / Sleep
- Volume
- Source Selection
- Settings

This reduces to two main modes and the Splash screen when using the Remote.

- Splash / Sleep
- Volume / Source
- Settings

Splash

When you first power up the dspNexus, a small microcontroller starts up prepares the dspNexus for further activity. At this point, the power consumption is low because most of the dspNexus is still unpowered. Pressing the **Center** button on the Remote or engaging the Encoder will continue the wakeup process.

Once the dspNexus is fully powered up and the signal processing functions are loaded, the mode changes to Volume. You can return to sleep my rotating the Source selection and selecting Sleep.

Volume

The volume mode is the normal operating mode. It displays the volume setting in 0.5dB steps. 0 dB is maximum setting. It also displays the selected source. Volume level is independent for each source.

Getting the Full Scale Level Optimized

If the 0 dB full range setting is not loud enough, it probably means that your output level settings are set too low. This is adjustable in the Setup section. You should also make sure that your music server or computer settings are not adding significant attenuation.

If levels are only low with one of the source types, make sure there is sufficient input gain for the source. For example, if you are using a phono stage, does your input have enough gain?

One way of thinking about the 0dB setting is that this is the maximum level you want your system to play. When this condition is met, you have maximized the available dynamic range and signal to noise ratio of your system, Rotating the encoder changes the volume for the selected source. Alternatively, you can use the Up or Down on the Remote. Doubleclicking the encoder or pressing the Pause button of the remote causes the system to mute or unmute for all sources. When the system is muted, it is shown on the display.

Example Screens:

V	o	l	u	m	e			-	2	0	.	0		d	B
S	/	P	D	I	F										

V	o	l	u	m	e						M	u	t	e	d
S	/	P	D	I	F										

If the headphone is plugged in, the signal routing will mute the outputs on the back panel. The signal processing and the volume setting for the outputs are independent from the headphone. It is possible to have filtering for the headphones if it is implemented in the Audio Weaver signal chain. On the display, the speaker icon is replaced with a headphone symbol.

Headphone Change Example Screen:

V	o	l	u	m	e			-	2	4	.	5		d	B
S	/	P	D	I	F										

Source Selection

The dspNexus has four stereo input sources, Analog, USB Audio Class 2, S/PDIF and Bluetooth. Each input can be selected from the Display/Encoder or Remote. Operation is a little different when using the Remote or the Encoder.

Remote Source Selection

The Left and Right Arrows change the selection. The display looks largely the same with the source changing along with the corresponding volume setting. This is the preferred way to change sources.

Encoder Source Selection

The Encoder needs to switch contexts, you press the Encoder knob quickly (like a computer single click mouse). This will change the display screen to Source Selection.

Source Selection Screens:

S	o	u	r	c	e		S	e	l	e	c	t	i	o	n
A	n	a	l	o	g										

S	o	u	r	c	e		S	e	l	e	c	t	i	o	n
S	/	P	D	I	F										

Rotating the knob will scroll through the available sources. Press the Encoder knob again will toggle back to the Volume Mode of the display

Return back to Volume Screen:

V	o	l	u	m	e			-	2	0	.	0		d	B
S	/	P	D	I	F										

Source Variations

Even though there are only four main sources, the Source menu gives you many more options. You determine which choices you want to have displayed in the Setup section.

For some users, this will mean scrolling through many available choices. For example, a vinyl enthusiast may want want the option to switch to Mono for an old 78 pressing. Occasionally, a recording can have a phase inversion (polarity) , that can be switched as needed.

Alternatively, you may have a system that doesn't even use all the available sources. For example, someone who only uses digital sources may wish to bypass the Analog Input from the menu selection and certainly many users are will opt to skip the Mono or Invert options to keep the menu selections to a smaller set.

There is also a source choice that is really not a source. This is the Sleep selection. This shuts down the dspNexus and puts it into its low power state. It is the preferable way to turn off the dspNexus as opposed to the Power switch on the Back Panel.

Each source and its options are discussed next.

Analog (Phono)

The Analog inputs are balanced with Gain, Polarity and Equalization (EQ) configured in Setup. If the EQ is set to Flat, the displayed name is Analog. If the EQ is set to RIAA, the name is changed to Phono.

Usually, you will just want to listen in a standard stereo configuration. Nevertheless you have additional choices. These are as follows:

- Analog (Phono)
- Analog (Phono) Invert
- Analog (Phono) Mono
- Analog (Phono) Invert Mono

USB UAC2

The USB Audio Class 2 (UAC2) input is probably the most used selection with the dspNexus. It is the preferred connection to a music server or streamer from an external computer. Most content will already be in PCM format with various bit depths and sample rates. If your content was provided as 44.1k, 16 bits (CD quality), let the dspNexus handle the upsampling automatically. You won't get more effective bandwidth or resolution (blame the universe we live in if you like), but it will work seamlessly. Any DSD content needs to be converted in the music server to PCM since DSD is useless as an input source for additional DSP signal processing.

Once again will likely want to listen in a standard stereo configuration. Nevertheless you an additional choice as follows:

- USB UAC2
- USB UAC2 Inverted

S/PDIF

S/PDIF and its close cousins, AES3 and Toslink are becoming less common in modern audio systems. One good use for S/PDIF would be as a connection to your television. This will likely take an Toslink to Coaxial adapter or an EARC adapter. The sample rate is limited to 96k or less which is generally not a limitation where S/PDIF is encountered. Since the dspNexus manages the volume settings, S/PDIF inputs can be used even when external volume controls are not available. This is very common in consumer products where S/PDIF is considered just a digital line out feature.

Once again you an additional choice as follows:

- S/PDIF
- S/PDIF Inverted

Bluetooth APTx-HD

Bluetooth Audio is available for convenience. Perhaps your friend has a song on a phone that he or she wants to share. Bluetooth has a variety of profiles. Phones are fairly universal with A2DP support. This is about MP3 quality. Some phones also support APTx-HD. This profile has near CD quality performance.

Phantom Powered Microphone

There is an XLR connector on the front panel for phantom powered microphones. Do not plug in devices that cannot support 48V biasing. They may not survive. If you are making measurements, the best way is to use the Measurement selections in the Setup menu.

Sleep

This selection puts the dspNexus in a low power mode. To power up the dspNexus do one of the following::

- Press the Center (Enter) button on the Remote
- Turn the Encoder knob
- Press the Encoder knob

Control: Setup – General

The dspNexus 2/8 Settings Mode is infrequently accessed and the most complex of the control functions. Its main purpose is to allow you to configure the dspNexus optimally for the environment where it is expected to operate. It still works under the principle that there is a working signal processing design that is implementing a crossover, but that there are still. parameters that need adjusting outside the signal processing design. Many of these are operating preference choices, room dependent features, power amplifier integration and product information.

Setting Menus

Since the settings cover a broad rang of topics, this manual is going to divide the various setting categories into separate sections in the manual. The main menus are as follows:

- Measurement
- Room EQ
- Active Sources
- ADC Settings
- DAC Settings
- Line Out Configuration
- Frequency Response Tailoring
- Info
- Exit

Access & Operation

Access to the Setup functions is a little different when using the Remote or the Encoder. The Remote is the easier method. Remote and Encoder functions can be used interactively. When Setup is entered and left inactive, it will timeout (exception – Measurement) and return to the Main Operating Mode (Volume & Source). There is also an Exit from the Main Menu.

Remote Setup Operation

Press the Menu button on the Remote. This will bring you to the main menu of Setup. If you press this button again, you will return to the Main Operating Mode (Volume & Source).

First Setting Main Menu Screen:

M	a	i	n		M	e	n	u							
M	e	a	s	u	r	e	m	e	n	t					

Remote Functions

The Remote functions are as follows:

- The Up and Down Arrows increase/decrease values
- The Left and Right Arrows rotate through the selections.
- Center functions like an Enter Key.
- The Pause Button is ignored
- Menu moves up one menu level. At the Main Menu level, Setup is exited.

Encoder Setup Operation

Press the Encoder knob and hold for several seconds. This is known as a long click. This will bring you to the main menu of Setup. If you press and hold this button down again (long click), it functions like the Menu button on the Remote and you will return to the Main Operating Mode (Volume & Source).

First Setting Main Menu Screen:

M	a	i	n		M	e	n	u							
M	e	a	s	u	r	e	m	e	n	t					

Encoder Functions

The Encoder functions are as follows:

- Rotating clockwise increases the selection or moves toward the end of a list.
- Rotating counterclockwise decreases the selection or moves toward the start of a list.
- Normal Click (pushing like a mouse click) is a normal selection, works like an Enter key.
- Doubleclicking is ignored.
- Long Click moves up one menu level. At the Main Menu level, Setup is exited.

Control: Setup – Measurement & Room EQ

The dspNexus 2/8 Measurement and Room EQ functions are used with a measurement microphone in conjunction with an external speaker design tool such as Room EQ Wizard. The are two methods that are available: Measurement and Room EQ.

Measurement is the most flexible method, but adds a bit more complexity. Many users will find that Room EQ may be all they need. You can think of Room EQ as Measurement Lite.

There are many software programs for making measurements. The most popular program is Room EQ Wizard. REW is free software that you can find at roomeqwizard.com .

Measurement Microphones

On the front panel of the dspNexus there is an XLR for 48V phantom powered microphones. In general, you want a microphone with a small capsule and a calibration curve. The microphone does not necessarily need to have great signal to noise, stability or accurate sensitivity to make speaker measurements. It does need a wide frequency response which is why a small capsule is important.

Low cost microphones such as the Dayton Audio EMM-6 or Behringer ECM-8000 are acceptable for most situations. Better measurement microphones are available with phantom powered options from Earthworks, GRAS and ACO Pacific. If you use the Measurement function (Not Room EQ), you can also use a microphone/preamplifier that uses the Analog Inputs from the back panel. In this case, you are not providing phantom supply directly from the dspNexus.

Control: Setup – Active Sources

Active Sources determines which Sources will be selectable during normal operation. For many users, this pares down the choices to the subset that they actually care about.

Upon entry into the Active Source Menu, the first source is displayed and its current setting. Press the Center button (Enter) on the Remote or the knob on the Encoder if you want to change states (On or Off). This will cause the current state to blink. The Left or Right Arrows on the Remote or turning the Encoder will change the state. Press Enter again to select. Use the Left or Right arrows or turn the Encoder to continue the process. There is an Exit at the end of the list.

Analog/Phono

There are multiple choices with the Analog/Phono inputs. Phono is displayed when the EQ in ADC Settings is RIAA, otherwise Analog is displayed.

- Analog (Normal Stereo)
- Analog Invert (Stereo, Inverted Polarity)
- Analog Mono (Mono)
- Analog Mono Invert (Mono, Inverted Polarity)

> Inverted Polarity assumes that the input source is normally in phase. In rare cases, the external input source could be inverted. This should be corrected via ADC Settings. This will make the Normal, non-inverted state correct.
>
> Most content will be played using the Analog selection.

If you are in the Measurement Menu and use the Analog Input on the back panel, these settings are ignored.

We expect that most users will turn off the Invert and Mono selections to keep the Operating Menu more compact. If you only use digital content, you might turn all of the Analog selections off.

USB UAC2

USB Audio Class 2 has a Normal and Inverted option.

S/PDIF

S/PDIF has a Normal and Inverted option. Unless, you are using a TV as an input source, we suggest that you use USB when using an external computer or music server.

Bluetooth aptX-HD

S/PDIF has a Normal and Inverted option. Unless, you are using a TV as an input source, we suggest that you use USB from an external computer or music server.

Microphone

S/PDIF has a Normal and Inverted option. Unless, you are using a TV as an input source, we suggest that you use USB from an external computer or music server.
Settings Mode

Control: Setup – ADC Settings

The ADC Settings tailor the input source requirements to best take advantage of the wide dynamic range of the ADC Module.

ADC Level

The ADC has programmable gain amplifiers (PGAs) that accommodate a large range of audio analog sources from professional audio levels to the high gain requirements of phono preamplifiers or microphones.

ADC Level controls a PGA in eight 6dB steps. Ideally, you want your input to use most of the ADC full scale range without ever clipping. If the sensitivity is too low, you lose bits. Fortunately, the range is forgiving because the S/N of the ADC is very high.

The names of the levels just reflect typical source types. All inputs are actually balanced. You can wire them with single ended (Unbalanced) sources (refer to the Hardware section). For example the 0 dB setting references professional +4dBu balanced audio which has a nominal level of +4dBu and a full scale of +22dBu.

- Pro +4dBu 0 dB Professional Audio Balanced
- Balanced 6 dB
- SE -10dBV 12 dB Consumer Unbalanced
- Balanced 18 dB
- Balanced 24 dB
- Balanced 30 dB Use for Moving Magnet Phono
- Balanced 36 dB
- Balanced 42 dB Use for High Output Moving Coil Phono

ADC Polarity

In rare cases, the polarity of your external source (an odd phono preamplifier?) might be inverted. If this is the case, you flip it here. This is not used to flip a recording that was flipped. This should be done in the Active Sources Menu.

ADC EQ

Flat EQ means that there is no extra filtering of the incoming source, RIAA is used to implement an RIAA curve. The accuracy of the RIAA curve is very closely matched to the theoretical resonse uo to 20 kHz.

Control: Setup – DAC Settings

The DACs have individual programmable attenuators that accommodate the broad range of amplifier sensitivities. The maximum output level can also be set to support+4 dBu professional line out requirements.

Amplifier Considerations

Power amplifiers may have balanced or unbalanced inputs. If you have both options available, use the balanced inputs. If your amplifier is limited to having only unbalanced inputs, refer to the Hardware Section of this manual for correct wiring.

Amplifier gain is not standardized. This means one amplifier type may output a different level than another with the same input drive. Its not always easy to compare amplifier specifications in this regard either. For example, balanced inputs have hot and cold inputs whereas single-ended (unbalanced) inputs only have a hot input. This can create a 6dB difference depending on how one views the input drive.

From a maximum signal to noise – dynamic range perspective, you want each DAC to use all of its available bits to output the loudest sound you ever want from its connected driver. This might mean that full scale of the DAC should drive the amplifier to its full scale output, but not always. You may not want an amplifier to be able to drive a tweeter to full scale output if the tweeter can't handle the full scale output of the amplifier.

Another consideration is that at some point, the amplifier will clip or the DAC will clip (or the speaker is being seriously over driven) . There is a balancing act between conditions. An amplifier that is clipping is a bad sounding amplifier regardless of topology. Obviously, you want to have sufficient power for your situation. In an active loudspeaker, you will likely need much larger amplifiers for the woofers than the higher frequency drivers. Assuming that your drivers can handle full output swing of the mating power amplifier, we recommend that the DACs should clip just a little before the power amplifier. The DACs will recover much faster than a power amplifier going into saturation.

Crossover Considerations

If you are designing your own custom crossover, you can optimize the DAC/amplifier attenuator settings for each driver to maximize the signal to noise performance of your system. This method works well when the amplifiers are identical and each driver is precisely located with respect to each other (like a typical full range speaker).

Keep in mind, that the DACs have a very large dynamic range, so that these considerations are not as nearly important as it might seem. As long as the relative levels between drivers are correct, you should be fine.

Subwoofer Considerations

If you have subwoofers that are located independently from your mains, you have a few parameters that are important to adjust. In the general situation, the location of the subwoofers is unknown with respect to the mains when the crossover is created. It is room dependent.

You want to time align the subwoofer to the mains. If you ignore this parameter at some frequency you will likely have less bass! The reason is that the bass from the subwoofer will be anti-phase to the bass from the mains. The solution is that you add delay to the closer speaker which is often the mains.

Suppose the subwoofer is located behind the main loudspeaker. It this case, the sound takes longer to arrive to your listening position. You can't fix this with the subwoofer even if it has controls since you can't go back in time. The solution is that you can delay the mains. The next step is that you may need to adjust the relative level of subwoofer to the mains. If the subwoofer is farther away, it may need a little gain boost to make it the correct level.

The dspNexus has controls that lets you adjust the level in 0.25 dB steps and the delay from 5us to 21 ms. Sound travels a little more than one foot per second, so the woofer / mains separation could be a maximum distance of about 23 feet. When the subwoofer is aligned correctly, it location tends to disappear.

DAC Level - All

DACs are independently adjustable. With this setting, you can adjust all of the levels at the same time. This will overwrite individual settings. You might start with this function initially and then modify specific DACs afterwards. The steps are in 3dB increments. A good initial setting for amplifiers might be -12dB. This will depend on your amplifier sensitivity.

DAC Level - 1-8

This setting allows individual increments of each DAC in 0.25dB steps. Use the Center button to select whether you want to change the level of the current channel or change channels. The blinking <> will point to the selection.

First Setting DAC Level 1-8 Screen:

D	A	C		L	e	v	e	l				1	-	8	
C	h		1		<			-	6	.	2	5		d	B

This points to the Channel. Using the Left or Right Arrow changes the channel.

Second Setting DAC Level 1-8 Screen:

D	A	C		L	e	v	e	l				1	-	8	
C	h		1		>			-	6	.	5	0		d	B

After an Enter, this now points to the Level. Using the Left or Right Arrow changes the level.
The image assumes that you changed the level.

Next Setting DAC Level 1-8 Screen:

D	A	C		L	e	v	e	l				1	-	8	
C	h		1		<			-	6	.	5	0		d	B

After an Enter, this now points back to the Level. Using the Left or Right Arrow changes the level.

Next Setting DAC Level 1-8 Screen:

D	A	C		L	e	v	e	l				1	-	8	
C	h		2		<			-	6	.	0	0		d	B

The image assumes that you changed the channel. The level reflects the current value for channel 2

Last Setting DAC Level 1-8 Screen:

D	A	C		L	e	v	e	l				1	-	8	
E	x	i	t				D	A	C		L	e	v	e	l

If you go past the last channel you arrive at the Exit option, Pressing Menu will also exit.

DAC Delay Units

Delay units can be calculated in meters, feet or milliseconds. Room temperature is assumed for distance conversions. The actual delay is converted internally to number of sample delays: (1/192000) ~ 5us increments.

DAC Delay All

Delay units can be calculated in meters, feet or milliseconds. Room temperature is assumed and the actual delay is converted in internal units of number of sample delays (1/192000) ~ 5us.

DAC Delay 1-8

This setting allows individual increments of each DAC in approximately 5us steps. Use the Center button to select whether you want to change the delay of the current channel or change channels. The blinking <> will point to the selection.

First Setting DAC Delay 1-8 Screen:

D	A	C		D	e	l	a	y				1	-	8	
C	h		1		<		0	.	0	0	0	0		m	s

This points to the Channel. Using the Left or Right Arrow changes the channel.

Second Setting DAC Delay 1-8 Screen:

D	A	C		D	e	l	a	y				1	-	8	
C	h		1		>		0	.	0	0	0	0		m	s

After an Enter, this now points to the Delay. Using the Left or Right Arrow (or Encoder) changes the delay. The Encoder has a velocity feature that makes changing these parameters easier.

Next Setting DAC Delay1-8 Screen:

D	A	C		D	e	l	a	y				1	-	8	
C	h		1		>		1	.	1	5	1	0		m	s

The delay has now changed for Channel 1

Next Setting DAC Delay 1-8 Screen:

D	A	C		D	e	l	a	y				1	-	8	
C	h		2		<		0	.	0	0	0	0		m	s

The image assumes that you changed the channel. The delay reflects the current value for channel 2

Later Setting DAC Delay 1-8 Screen:

D	A	C		D	e	l	a	y				1	-	8	
C	h		2		>		4	.	8	3	3	3		m	s

Two steps: Entered to point to delay and then changed the delay value.

Last Setting DAC Delay 1-8 Screen:

D	A	C		L	e	v	e	l				1	-	8	
E	x	i	t				D	A	C		D	e	l	a	y

If you go past the last channel you arrive at the Exit option, Pressing Menu will also exit.

Polarity All

This setting sets all DACs to Absolute Polarity or Inverted Polarity.

Polarity 1-8

This setting allows the polarity of each DAC to be set separately. It works the same way as DAC Level or DAC Delay.

Headphone Level

This setting adjusts the DAC attenuator in 3 dB steps when using the headphone amplifier located on the front panel. It is an independent the attenuator settings when the same DAC is used as an output on the back panel. Use this setting to optimize the full scale range of the DAC when using your headphones since sensitivity can vary considerably.

Control: Setup – Line Out Settings

In the dspNexus, we generally think of the outputs as being the eight balanced analog outputs driven by the DACs. There are other outputs available as well. These are stereo and tend to be used outside of the typical crossover features. In many cases, these are going to be used for measurement purposes. They also could be used for expansion into another system. There are three auxiliary outputs addressed in this section of the manual:

- USB Audio Out
- S/PDIF Out
- Bluetooth Out

USB Audio Out

Level

You can set up the line output level of the USB in 32 steps of 0.5dB.

Track Volume

If the Track Volume is On, the level will scale with the Volume setting, otherwise the output is fixed.

S/PDIF Out

S/PDIF

Level

You can set up the line output level of the S/PDIF in 32 steps of 0.5dB.

Track Volume

If the Track Volume is On, the level will scale with the Volume setting, otherwise the output is fixed.

Loopback

If the Track Volume is On, the level will scale with the Volume setting, otherwise the output is fixed.

Sample Rate

48k, 96k, 192k

Bluetooth Out

Level

You can set up the line output level of the USB in 32 steps of 0.5dB.

Track Volume

If the Track Volume is On, the level will scale with the Volume setting, otherwise the output is fixed.

Control: Setup – Frequency Response

Audio Weaver can be set up to include a pair of low frequency and high frequency shelf filters. The idea is that some users may want simple tailoring of the frequency response after a crossover has been designed. In many cases, this function may not be implemented in the crossover and therefore these settings will be ignored.

Low Freq Shelf

Five settings are available (-2.0 dB, -1.0 dB, 0 dB, +1.0 dB, +2.0 dB). The corner frequency is determined by the Audio Weaver design.

High Freq Shelf

Five settings are available (-2.0 dB, -1.0 dB, 0 dB, +1.0 dB, +2.0 dB). The corner frequency is determined by the Audio Weaver design.

Control: Setup – Info

The Info section reports the internal configuration of the dspNexus and the software versions of internal software.

LCD Version

2.5.0.0

This is the version of code running on the display microcontroller.

OEM Version

0.0.0.0

Platform Version

0.0.0.0

ADC

AK5578

This is the ADC converter used for the Analog Input Module. Currently, only one type is used in the dspNexus. ADC

DAC1, DAC2, DAC3, DAC4

AK4493 or AK4499

This is the DAC converter used for each Analog Output Module.

DSP Module

dspblok 21569uac2 or dspblok 21469uac2

This is the DSP Module used in the dspNexus 2/8.

Product Warranty

Danville Signal Processing, Inc. products carry the following warranty:

Danville Signal Processing products are warranted against defects in materials and workmanship. If Danville Signal Processing receives notice of such defects during the warranty period, Danville Signal Processing shall, at its option, either repair or replace hardware products, which prove to be defective.

Danville Signal Processing software and firmware products, which are designated by Danville Signal Processing for use with our hardware products, are warranted not to fail to execute their programming instructions due to defects in materials and workmanship. If Danville Signal Processing receives notice of such defects during the warranty period, Danville Signal Processing shall, at its option, either repair or replace software media or firmware, which do not execute their programming instructions due to such defects. Danville Signal Processing does not warrant that operation of the software, firmware, or hardware shall be uninterrupted or error free.

The warranty period for each product is one year from date of installation.

Limitation of Warranty:

The forgoing warranty shall not apply to defects resulting from:

- Improper or inadequate maintenance by the Buyer;
- Buyer-supplied software or interfacing;
- Unauthorized modification or misuse;
- Operation outside the environmental specification of the product;
- Improper site preparation and maintenance.

Exclusive Remedies:

The remedies provided herein are the Buyer's sole and exclusive remedies. In no event shall Danville Signal Processing, Inc. be liable for direct, indirect, special, incidental or consequential damages (including loss of profits) whether based on contract, tort, or any other legal theory.