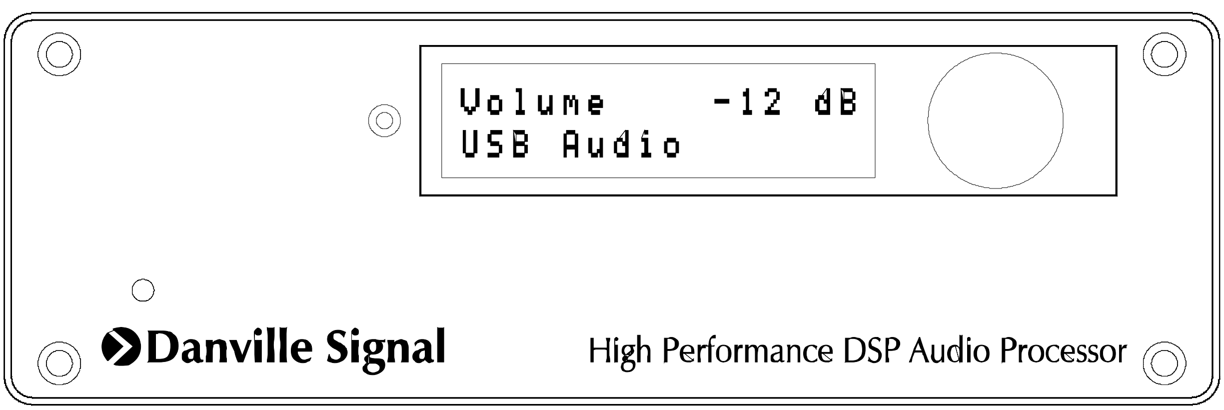
Danville Signal Processing, Inc.

dspMusik™ 2/8

High Performance DSP Audio Processor

P/N A.03708D
P/N A.03708E



User Manual

Version 2.10

Danville Signal Processing, Inc.
dspMusik™ 2/8
User Manual

Danville Signal Processing, Inc. strives to deliver the best product to our customers. As part of this goal, we are constantly trying to improve our products. Danville Signal Processing, Inc., therefore, reserves the right to make changes to product specification or documentation without prior notice. Updated operating manuals and product specification sheets are available at our website for downloading.

This manual may contain errors, omissions or "typo's". Please send your comments, suggestions and corrections to:

Danville Signal Processing, Inc.
29687 82nd Avenue Way
Cannon Falls, MN 55009

Trademark Notice

Contact Information

Danville Signal Processing, Inc.
29687 82nd Avenue Way
Cannon Falls, MN 55009

E-mail: dsp@danvillesignal.com
Web Site: http://www.danvillesignal.com

Voice: 507-263-5854

Fax: 877-230-5629

Table of Contents

Overview

The dspMusik™ 2/8 is intended for applications where audio quality is of paramount importance. Applications include high performance DSP based crossovers for studio monitors and high end consumer audio systems. It is also well suited for multichannel decoders and algorithm development and demonstration.

Loudspeaker manufacturers can use the dspMusik for as an external standalone crossover or as a development system for Danville's embedded dspCrossover™ semi custom embedded DSP crossovers used in powered monitors

One of the advantages of the dspMusik 2/8 for algorithm developers is that the hardware performance does not mask the quality of the algorithm. This allows you to demonstrate your intellectual property at its best.

This manual covers the dspMusik 2/8 as implemented as a complete ready to operate product. It is housed in an extruded aluminum box with an LCD display and encoder. The 2x16 character alpha-numeric LCD display and companion encoder are used for volume control, source selection and adjustment of other parameters. It can also be controlled via IR using an Apple Remote. The Apple Remote is available from Apple resellers for about $20.

The dspMusik 2/8 is also available in bare board configurations for integration by OEMs into custom housings. There are many options including switching amplifier interfaces, LCD displays and 8 channel expansion. This manual covers the complete standalone version. The OEM integration manual covers bareboard configurations with wiring details, physical dimensions and similar information needed to implement the boards into a custom enclosure.

Most dspMusik 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 dspMusik 2/8, however there might be custom features implemented in the version of the dspMusik 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.

Hardware

The dspMusik 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 dspMusik. 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 dspMusik 2/8 supports 2 analog inputs, an S/PDIF receiver, USB Audio Class 2 and 8 analog outputs. All analog I/O is available as balanced +4dBu professional or balanced/single ended –10dBV consumer audio levels. The choice is based on software configuration. The sample rate can be as high as 384k with 24 bit conversions.

Four state-of-the-art AKM AK4495 DACs are used for the 8 analog outputs. Each channel has balanced anti-imaging filters to attenuate clock feed through and high frequency images. The output level is balanced (or single ended) with levels as high as +4dBu (+22dB FS). Each DAC has analog attenuators to optimize the full scale performance of the converter.

The analog input circuit is a low noise balanced input with very high common mode rejection. The optional ADC module uses an AKM AK5397, considered by many to be the best ADC on the market. It has a S/N in excess of 127dB. This is 20-30dB better that most home theater oriented converters.

All of the data converters have a variety of internal filter choices. This include both sharp cutoff and minimum phase types. Minimum phase filters are favored for most applications, because they avoid pre echo artifacts.

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 dspMusik 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 streams that are intrinsically noisy from a jitter perspective.

The dspMusik 2/8 uses a standard IEC connector and can be powered with any 100-240 VAC 50/60 Hz mains source. All that is needed is the appropriate mains cord set. This makes the dspMusik 2/8 functional in international markets without special configuration settings.

Software

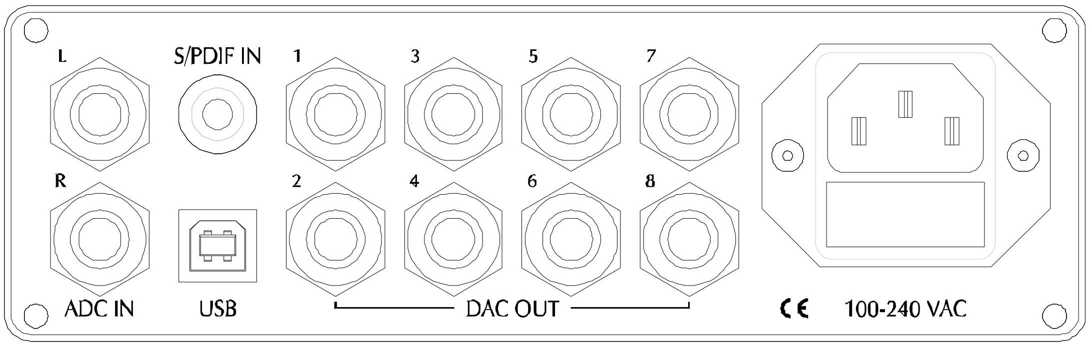
There are two primary ways to program the dspMusik 2/8.

Most customers will opt for DSP Concepts' Audio Weaver. This is a graphical design program that includes as many as 400 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, a loudspeaker designer 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. A run time version of Audio Weaver is included with the dspMusik 2/8. This lets you design and distribute your application without buying additional tools or licensing. All you need is a Windows computer.

If you have an OEM version of the dspMusik 2/8, chances are this is how most of your system software was implemented. The specific layout and tuning was done by the OEM, so that their system performs optimally.

The alternative method would be traditional programming in C or assembly. In this case, you would use either Analog Devices' Visual DSP++ or CrossCore Embedded Studio. You would also want an emulator, either an ICE-1000 or ICE-2000. The older ADI emulators will also work with any of the dspblok 214xx boards. Danville supplies example code with device drivers for the converters and a basic talk thru shell program to get you started.

I/O Configuration



The dspMusik has three stereo input sources, Analog, USB Audio Class 2, and S/PDIF. Each input can be selected from the display/encoder or Apple remote and have independent gain settings. This is helpful when the levels from each source can be quite diverse.

For example, you might have the Analog input connected to a phono preamplifier, the USB connected to your music server and the S/PDIF to a television. Each of these sources could have very different output levels. When you switch between each source, the volume level can remain consistent. This also means that turning the volume down for one source does not adjust the others. Mute is universal. All sources are muted together.

There are 8 balanced Analog outputs. These are usually assigned for specific purposes by the software. For example, in a DSP crossover, some outputs are assigned to tweeters, others, to woofers, etc. If you have an OEM version of the dspMusik, make sure you consult the OEM's instructions for the correct assignments of each output.

Balanced Audio - ADC & DAC Level Setting:

The dspMusik 2/8 has both input PGAs (programmable gain amplifiers) and output attenuators to adjust the audio signals to optimum levels. The goal is to maximize signal level without clipping. These switch in approximately 6dB steps (a little different for the first step to accommodate dBu to dBV differences).

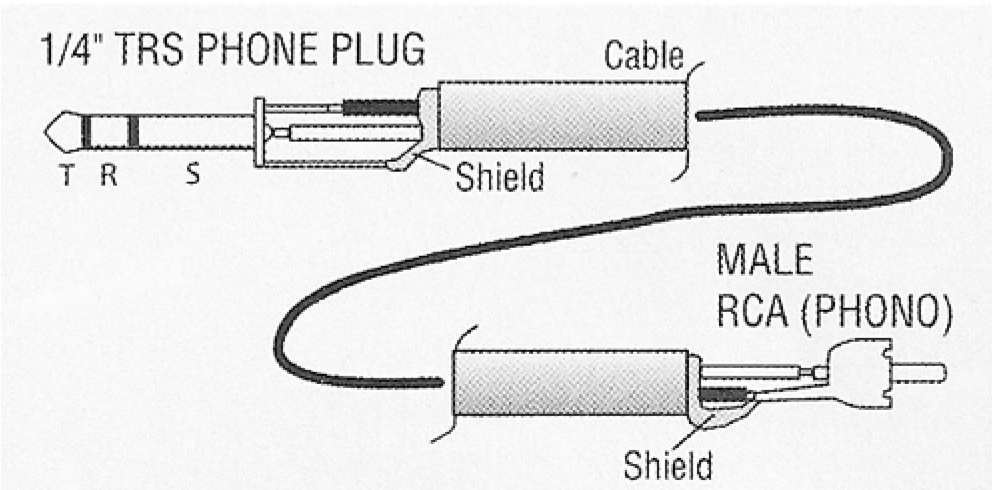
Nominal audio levels are defined by standard in only two ways:

- Professional level (Balanced) -4dBu nominal (+22dBu full scale)
- Consumer level (Unbalanced) –10dBV nominal (+8dBV full scale)

This means that signal levels are much higher with professional audio equipment (about 11.75dB). We don't amplify inputs and we amplify outputs to create these levels.

Since professional level is defined as balanced and consumer level is defined as unbalanced, it gets a little more complicated.

Consumer level is usually unbalanced. This means that a proper input cable (maybe from a phono preamplifier) should connect Center (source) to Hot (dspMusik, tip), and Ground (source) to Ground and Cold (ring) on the dspMusik TRS. The best way is with a three conductor cable with separate wires (shield & twisted pair) connected at the source end.



Unbalanced outputs ONLY use the Hot and Gnd on the dspMusik. The Cold connection is left open. Otherwise, The Cold output will drive into a short (Ground).

When making input connections, a less optimum but certainly convenient interconnect method is to use a phono to 2 conductor (mono) phone plug adapter. This shorts the Cold and Ground connections at the TRS input of the dspMusik. DO NOT DO THIS FOR UNBALANCED OUT CONNECTIONS! It will short the Cold Output to Ground! Output cables for unbalanced circuits leave Cold open.

In the professional audio case, levels and interfaces are consistent. With consumer gear, everything gets a little sketchy. Some consumer gear now has balanced audio connections available. The problem is that –10dBV is no longer is clearly defined. If balanced connections are used, should levels be measured from Hot to Cold or alternatively, Hot to Ground? This is a 6 dB difference and different manufacturers have different ideas.

Another issue that comes up is power amplifier gain. Most power amplifiers have a gain of between 20 and 30 (26dB to 31dB). This can be referenced from balanced or unbalanced inputs. This means that a large power amplifier might need to about 3.5V peak to swing to full range. If the output gain is insufficient, the amplifier will not swing to maximum power. In general, you use the settings that allow the amplifier to go full range and not more. This will usually be the Consumer –10dBV output settings wired as balanced if the amplifier has balanced inputs and single ended (unbalanced) if it does not.

dspMusik Naming Conventions

Input Levels:

• Pro Audio +4 dBu	
• Balanced –4 dBV	5.75 dB Gain
• Consumer –10 dBV	11.75 dB Gain
• Balanced –16 dBV	17.75 dB Gain

Output Levels:

• Pro Audio +4dBu	
• Consumer –10dBV	5.75 dB Attenuation (same level as Balanced –4 dBV)
• Balanced –10dBV	11.75 dB Attenuation
• Balanced –16dBV	17.75 dB Attenuation

On some displays, the levels may be labeled as +6 dB, +12 dB, etc for Inputs and –6 dB, -12dB etc for Outputs.

When we are referring to Consumer Input and Output, the assumption is that we are wiring the cables in an unbalanced configuration. For inputs, this means that we do not have an out of phase signal available for the Cold input. A +4dBu (Pro) balanced signal is the nominal level between Hot and Cold. A –10dBV Consumer level is the nominal level between Hot to Gnd.

The “Balanced” labels are expected nominal levels from Hot to Cold. A Balanced –10dBV signal would be 6dB lower than the Consumer –10dBV level since Hot to Gnd is only half of the signal.

It is also possible that a manufacturer decided that it would provide a balanced output where the level is –4 dBV. This would allow the unbalanced circuit (leaving Cold disconnected) to be –10dBV. It would have twice the voltage swing when the balanced wires are used. This means that the input circuit should have less gain in this wiring configuration.

This is admittedly confusing. The good news is that if you are interfacing pro audio equipment, you just set the levels to the Pro settings. If you are interfacing to most consumer equipment (RCA Phono) or PC soundcards; make your cables as discussed above and use the consumer settings.

dspMusik Display/Encoder 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 an Apple Remote.
The Apple remote is assumed to be the later style (silver) that has 7 buttons.

Ergonomically, the Encoder and Apple 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 Apple Remote functions are as follows:

- Up or Right Arrows function the same as a clockwise rotation of the encoder.
- Down or Left Arrows function the same as a counterclockwise rotation of the encoder.
- Center functions like a normal click or long click. 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 discussed below.

There are three main modes and one temporary mode:

- Splash
- Volume
- Source Selection
- Settings

Splash Mode

The splash screen typically displays company name and product and lasts for about 5 seconds.

During this time, all the nonvolatile settings from the dspMusik are transferred to the display processor. This is how the display/encoder knows the current source selection, volume settings, etc. This process is invisible on the display. During this time the encoder is ignored.

Volume Mode

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

Rotating the encoder changes the volume for the selected source. Alternatively, you can use the Up, Down, Left or Right buttons of the Apple 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 Screen:

V	o	l	u	m	e		-	1	0	0	.	0		d	B
S	/	P	D	I	F						M	u	t	e	d

Source Selection Mode

Source selection is invoked by a normal click of the encoder when starting in Volume Mode. Rotate the encoder to the desired selection and press the encoder again. This will accept the new source setting and return to Volume Mode. Alternatively, you can use the Apple Remote Menu button to toggle through the selections. This is quicker and bypasses the Source Selection Mode (or returns to Volume mode if already in Source Selection mode).

Example Screen:

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

Settings Mode

Settings are a collection of configurations, information or test features that are used occasionally. Different dspMusiks will likely have a different collection of choices since dspMusik displays are routinely customized for specific OEM requirements.

Since Settings are accessed infrequently, the assumption is that its access should be somewhat slower and decidedly intentional. To enter Settings, press the encoder for several seconds (Longclick) until the display toggles to Settings. Then release the button. You can also do this with the Center button of the Apple Remote.

S	e	t	t	i	n	g	s								
A	D	C		L	e	v	e	l							

You can rotate the encoder (or use the Apple Remote arrows) to move through the various selections. Generally, pressing the encoder (Normal Click) will move you to a submenu or return you back to the main Settings menu.

A	D	C		L	e	v	e	l							
P	r	o		A	u	d	i	o			+	4	d	B	u

It some cases, the selection is just informational.

V	e	r	s	i	o	n									
2	.	0	.	0	.	0									

There is one odd settings choice – Apple Remote Tester. This is not accessible directly from the Apple Remote. Its purpose is to test the Apple Remote which is the reason why the Apple Remote doesn't directly access it. Otherwise, Apple Remote works as an alternative to the Encoder for all the other items in the Settings Mode.

A	p	p	l	e		R	e	m	o	t	e				
<	P	r	e	s	s		A		B	u	t	t	o	n	>

To leave Setting go to the last selection – Exit and press the encoder. This takes you back to the normal operation – Volume Mode.

S	e	t	t	i	n	g	s								
E	x	i	t												

dspblok Options

The dspMusik 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 dspMusik. Historically, the dspMusik has been supplied with either the dspblok 21479u8a or the dspblok 21469+USB. The dspblok 21479u8a supports USB Audio Class 2 and operates at 250MHz. The dspblok 21469+USB does not support USB Audio Class 2. It used the USB port for control only.

The USB audio dspbloks have a companion processor that supports USB Audio Class 2 (UAC2). These are supplied with Windows device drivers and are supported natively by Apple OS X and Linux.

The latest USB audio dspbloks are the dspblok 21489uac2 and dspblok 21469uac2. These combine the best features of the previous dspbloks and are now the recommended choices for the dspMusik 2/8. Both of these products take advantage of XMOS's late generation XU208 processor that provides the USB Audio Class 2 support.

dspblok 21469uac2

The dspblok 21469uac2 is powered by the Analog Devices ADSP-21469 operating at 450MHz. It is the most powerful 4th generation Analog Devices' DSP. It also includes external DDR2 SDRAM. The dspblok 21469uac2 is the best choice for systems where large FIRs or very long delays are going to be implemented. This would be likely for large space acoustics applications.

dspblok 21489uac2

The dspblok 21489uac2 is powered by the Analog Devices ADSP-21489 operating at 400MHz. It is almost as fast as the dspblok 21469uac2 at a lower cost. Since the dspblok 21489uac2 does not include external SDRAM, it tends to the best choice for small space acoustics applications, for example studio monitors and high performance consumer loudspeakers. There is still plenty of internal RAM for delay compensation and moderate size FIRs as well as the ability to process a very large number of IIR filters.

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.