Danville Signal Processing, Inc.

DSP-8200s

Tone Suppression Instrument

Operating Manual

Version 1.2

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

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Specifications

Power Supply:

Voltage:	9 to 15VDC, 2.1mm DC coaxial, center pin positive
Maximum Ripple:	500mVpp
Power:	Approx 1.7W operating. 140mA at 12VDC nominal
Protection:	Fused input, series diode to prevent reversal, and common mode choke to reduce conducted emissions

Environment:

Temperature Operating:	-10 to +50deg C
Temperature Storage:	-20 to +70C (limited duration at high temperatures)
Humidity:	0 to 95 % RH, NON-CONDENSING
Dimensions:	175mm deep x 106mm wide x 38mm high without DIN bracket
Weight:	0.5kG
Electromagnetic Emissions:	FCC Part 15 Class B, EN 55022 (CISPR 22) within Australian/New Zealand EMC Framework (C-TICK) * Emissions are substantially below Class B specification * Edition 5 CISPR removes requirements for ferrite suppressors being fitted to cables that exist the unit including power input leads
MTBF:	Demonstrated >80,000 hours based on recent batch of 50 similar units

Audio Input:

Configuration:	Balanced transformer input with failsafe bypass.
Impedance:	Balanced 600 ohms termination or 5k bridging (internal jumper), AC coupled
Maximum Input Level:	+10dBm
Dynamic Range:	Greater than 66dB
Electronic Level Adjustment:	40dB, 1 dB steps

Audio Output:

Configuration:	Balanced transformer isolated drive, AC coupled
Output Impedance:	Approx 300 ohms
Maximum Output Level:	+10dBm
Electronic Level Adjustment:	70dB, 1dB steps
Failsafe Bypass:	Relay contact with variable resistive attenuator to match thru/bypass level. Attenuator may be bypassed for unity gain operation

Control:

Input Type:	2 x Opto, LED via 3k3 resistor 5 to 30V will active input
Output Type:	2 x Opto transistors, with optional internal 5V pullup resistor Max switching voltage: 70 volts Max switching current: 10mA
	* Limited by current drive (device max current is 150mA)
USB:	Interface for control, configuration and upgrades
Startup Delay:	5 seconds to allow field s/w upgrade bootloader to wait

Tone Suppression:

Audio Quantization:	24 bits
Processor Type:	32 Bit Floating Point DSP
Operating Delay:	~ 7.2mS
Tone Removal Time:	25mS (-20dB); less than 35mS (-30dB)
Thru Mode Distortion:	<3% into terminated load at 0dBm output
Frequency Response:	300 to 4000Hz. +/- 1dB

Overview

Digital Signal Processing

Digital Signal Processing (DSP) is a powerful and complex method of analyzing and modifying analog signals, such as speech. Speech signals have fairly well known and predictable characteristics; however, these characteristics are quite complex.

By converting the analog signal to a digital signal, a digital signal processor with a special program can analyze the characteristics of the analog signal. The digital signal processor can then modify the digital signal to enhance the desired characteristics and to remove undesirable characteristics such as noise or tones. The processed signal is converted back to an analog signal and sent on to a speaker or headphone.

Automatic Tone Suppression

The Automatic Tone Suppression mode of the DSP-8200s removes heterodynes (tones) caused during multiple aircraft-to-aircraft and multiple aircraft-to-controller communications. It also attenuates unwanted signaling tones that may be inadvertently applied to normal monitoring channels. This tone suppression wipes out potential hearing-impairing test and interference tones in milliseconds.

The automatic tone suppression function of the DSP-8200s operates by examining a characteristic of the audio signal called correlation, and dynamically filters out undesired tones from the signal. The degree of correlation is relative. Tones are highly correlated as compared to speech signals. The DSP-8200s uses this difference to separate speech from tones. The DSP-8200s can attenuate tones by as much as 50 dB depending on the specific characteristics of the incoming signal.

The amount of tone suppression is dependant on many factors. There are many parameters that can be adjusted to increase the amount of tone suppression, the speed of the tone suppression and the quality of the resulting speech. Speech can be modeled as a combination of pure tones and broadband noise. For example, if we removed most the white noise components of “sit”, we would have “it” or even more drastically “i…” (soft i). In a similar fashion, removing the pure tone components from voice causes the vowels to become distorted.

The Danville tone suppression algorithm has been tuned to protect hearing by quickly suppressing test tones while at the same time passing speech with minimal degradation. One of the important attributes of this algorithm is that the tone suppression is best when the level of the tone(s) is high; the stronger the tone, the better the suppression. A lower level tone will not be suppressed as well, but low-level tones don’t cause hearing damage either.

The Danville tone suppression algorithm is unique. It was been used successfully since 1998 to eliminate tone incidences in air traffic control installations. In comparisons with more

conventional tone suppression algorithms, the Danville algorithm provides about 10dB more suppression for the same voice quality.

Bandpass Filtering

The DSP-8200s provides high performance bandpass filtering from 300 to 3400 Hz.
Since the DSP-8200s is intended for voice communication systems, signals outside of the voice band generally contribute to noise. The bandpass filter is also part of the Automatic Tone Suppression mode.

Level Adjustment

The DSP-8200s has an input programmable amplifier (PGA) and an output attenuator. They are both adjustable in 1dB steps. This allows the input levels and output levels to be optimized for the specific interfacing requirements.

Indicators

Overload and Normal Level LEDs are provided on the front panel of the DSP-8200s. There is also a heartbeat LED that flashes to indicate that the DSP-8200s is powered and operational.

Digital I/O

The DSP-8200s has an opto-isolated input for bypass control and an opto-isolated output for fault indication.

Connectors

The DSP-8200s uses a DB-9F connector for inputs and a DB-9M connector for outputs.

Operation

Operating Modes

The DSP-8200s has three operating modes: Tone Suppression, Bandpass and Talk-Thru. The operating mode is selected via software command. Software commands are described later in this document.

Tone Suppression Mode (factory default) automatically notches tones and also includes a 300-3400 Hz bandpass filter. For optimum performance, the nominal input level is adjusted so that the normal level indicator is ON and the overload level indicator occasionally flashes.

The Bandpass Mode includes the 300-3400 Hz bandpass filter but does not include tone suppression. Since all modes use the same hardware, the bandpass mode is useful as a test mode since test tones are notched by the tone suppression mode. If the DSP-8200s functions properly in the bandpass mode, it will work properly in all modes.

The Talk-Thru Mode is just a loopback from the audio input to the audio output.

In all operating modes, the input levels and output levels are adjustable in 1dB steps via software commands.

In addition to the three operating modes, the DSP-8200s also has a Bypass mode. The DSP-8200s uses relays to automatically loopback the audio from input to output. Bypass can be configured for unity gain or attenuated by adjusting an internal trimpot. If unity gain bypass is desired, two jumpers are placed at the JH3-0dB positions. If attenuated bypass is desired, a jumper is placed at the JH3-Var position. RV2 is then used to set the amount of attenuation.

Bypass mode occurs in one of three conditions: the DSP-8200s is unpowered, the optoisolated digital input is active, or the DSP-8200s is in reset.

Configuration

The DSP-8200s uses 600/600 Ohm transformers for both the input and output circuits.

The input load impedance can be either 600 Ohms or Hi-Z. If the desired input impedance is 600 Ohms, a shorting jumper (0.100 inch) is placed across JH3 in the 600 position. This would typically be the case when the DSP-8200s is driven by a balanced 600 Ohm audio circuit. If the input is being driven by a low impedance single ended source, JH3-600 should be left open.

The output can drive a 600 Ohm balanced circuit or a Hi-Z single ended circuit. Since these loads are quite different, there is a software command that specifies which type of load is present. This affects the output level calibration.

The opto-isolated output is active when the DSP-8200s is operational.

Connections

The DSP-8200s uses a DB-9F for its input connections. The audio input should be connected using Audio In+ and Audio In-. The ground return is not necessary but may be used for noise shielding. If it is used, we recommend leaving one end open on this connection. The digital input is active when a level of 5 to 12 VDC is applied across Opto In.

The DSP-8200s uses a DB-9M for its output connections. The audio output should be connected using Audio Out+ and Audio Out-. The ground return is not necessary but may be used for noise shielding. If it is used, we recommend leaving one end open on this connection. The digital output is an open collector that needs to be pulled up to 5 to 12V at the receiving end.

Power is supplied via an external 9 to 12V DC power supply. A standard 2.1/5.5mm coaxial power jack is the power input.

Inputs:

Connector: DB-9F

Pin Assignments:

1	Reserved
2	Audio In+
3	Opto In+
4	Reserved
5	Ground Return
6	Reserved
7	Audio In -
8	Opto In-
9	Reserved

Outputs:

Connector: DB-9M

Pin Assignments:

1	Reserved
2	Audio Out+
3	Opto Out+ (Collector)
4	Reserved
5	Ground Return
6	Reserved
7	Audio Out-
8	Opto Out- (Emitter)
9	Reserved

Software Commands

The DSP-8200s is configured via software commands. These commands can be sent via a standard terminal program using simple ASCII (text) commands.

The control port uses a USB connection (B style connector). A computer running Microsoft Windows 2000 or XP is required. Danville supplies a device driver that emulates a standard COM port connection.

In addition to storing the desired User Configuration, the USB port can also be used to upload firmware updates. The DSP-8200s uses non-volatile eeprom memory to store its configuration and flash memory to store its operating firmware. Upon power-up, the DSP-8200s will operate with its previous software settings.

Software Command Structure

All command start with either a "W" or an "R". All letters are case insensitive and converted to uppercase by the firmware. The parser ignores spaces.

A command is terminated by a 0, <CR> or <LF>.

Here is the complete summary of user commands:

RM, WM	Read/Write Operating Mode (0 = Tone Suppression, 1 = Talk Thru, 2 = Bandpass, etc)
RIG, WIG	Read/Write Input Gain (0 to +40 in 1dB steps)
ROA, WOG	Read/Write Output Attenuation (-6 to +73 in 1dB steps)
ROL, WOL	Read/Write Output Load (0 = 600 Ohm or 1 = Hi Z load)

The tone suppression algorithms are input level sensitive. The input gain should be adjusted so that the Normal LED is generally lit when someone is talking and will flash occasionally on the Overload LED.

In a typical setup, the input gain might be 10dB. If unity gain is desired into a Hi Z load, here are the commands:

```
WIG 10
WOA 10
WOL 1
```

All of the user commands take effect immediately and are also written to the EEProm. When the DSP-8200s is powered up, the last settings are used. You can get the settings by using the R (read commands).

Commands

Mode Command (WM, RM):

The Mode command is used to select the algorithm in the DSP-8200s. In normal operation it is 0.

Mode 0:	Tone Suppression
Mode 1:	Talk-Thru (The input gain, output attenuation and output load settings are active)
Mode 2:	Bandpass (Same as Talk-thru with a 300-3400 Hz filter)

Example:	WM 0	Set operating mode to tone suppression

Input Gain Command (WIG, RIG):

The Input Gain command is used to adjust the input gain in 1 dB steps before the ADC.

Format:	WIG d	where d is 0 to 40
	RIG	will return RIG d

Example:	WIG 0	Set input gain to 0 dB

Output Attenuation Command (WOA, ROA):

The Output Attenuation command is used to adjust the output attenuation in 1 dB steps after the DAC. A negative value is actually gain.

Format:	WOA d	where d is -6 to 73
	ROA	will return ROA d

Examples:	WOA 6	Set output attenuation to 6 dB
	WOA -6	Set output gain to 6 dB

Output Load Command (WOA, ROA):

The Output Load command is used to normalize the output level to either a 600 ohm load or a high impedance load.

Format:	WOL 0	600 Ohm Load
	WOL 1	Hi Z load
	ROA	will return ROA 0 or ROA 1

Troubleshooting & Checkout

The DSP-8200s signal processing modules require no adjustments or calibration. All signal processing functions are performed within the DSP processor.

The tone suppression mode of the DSP-8200s does not easily lend itself to traditional measurement techniques. These algorithms have been designed for human speech and interfering noise or tones. As such, these algorithms modify themselves dynamically with changing speech or noise conditions.

The easiest method to verify correct operation of a DSP-8200s module is to reconfigure the mode to bandpass. Since all signal processing functions are performed exclusively by mathematical calculation, a DSP-8200s module that passes this test will also operate correctly in the other operating modes.

WARNING:

THIS PROCEDURE SHOULD NEVER BE PERFORMED IN AN OPERATING COMMUNICATION CHANNEL!

IT IS INTENDED FOR OUT-OF-CIRCUIT TESTING ONLY.

TEST TONES WILL PASS THROUGH THE DSP MODULE IN THIS TEST!!

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 three years 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, accidental or consequential damages (including loss of profits) whether based on contract, tort, or any other legal theory.

Declaration of Conformance, EMC

FCC Class B Conformance

This equipment complies with Part 15 of the FCC Rules. Operation is subject to the following two conditions:

- This device may not cause harmful interference.
- This device must accept any interference received, including interference that may cause undesired operation.

This equipment has been tested and found to comply with the limits for a Class B digital device, pursuant to Part 15 of the FCC Rules. These limits are designed to provide reasonable protection against harmful interference in a residential installation. This equipment generates, uses and can radiate radio frequency energy and, if not installed and used in accordance with the instructions, may cause harmful interference to radio communications. However, there is no guarantee that interference will not occur in a particular installation. If this equipment does cause harmful interference to radio or television reception, which can be determined by turning the equipment off and on, you are is encouraged to try to correct the interference by using one or more of the following measures:

- Reorient or relocate the receiving antenna.
- Increase the separation between the equipment and receiver.
- Connect the equipment into an outlet on a circuit different from that to which the receiver is connected.
- Consult the dealer or an experienced radio/TV technician for help.

CISPR 22 Class B,<C-TICK>

This device complies with EMC directive of the European Community and meets or exceeds the following technical standard: EN 55022

-Limits and Methods of Measurement of Radio Interference Characteristics of Information Technology Equipment.

This device complies with CISPR 22 Class B.