

POWERSOFT KDSP USER MANUAL

EDITION Feb. 2010

Rev. 2.1.1

All copyright and industrial rights in this document and in the technical knowledge it contains are owned by Powersoft and/or the third parties rightfully concerned.

No part of this document nor any data herein shall be disclosed, reproduced or used for any purpose whatsoever without the prior written consent of POWERSOFT as foreseen by the law.

Drawings and specifications are subject to change.

All trademarks and registered trademarks are the property of their respective holders.

Printed in Italy.

VOID

SUMMARY

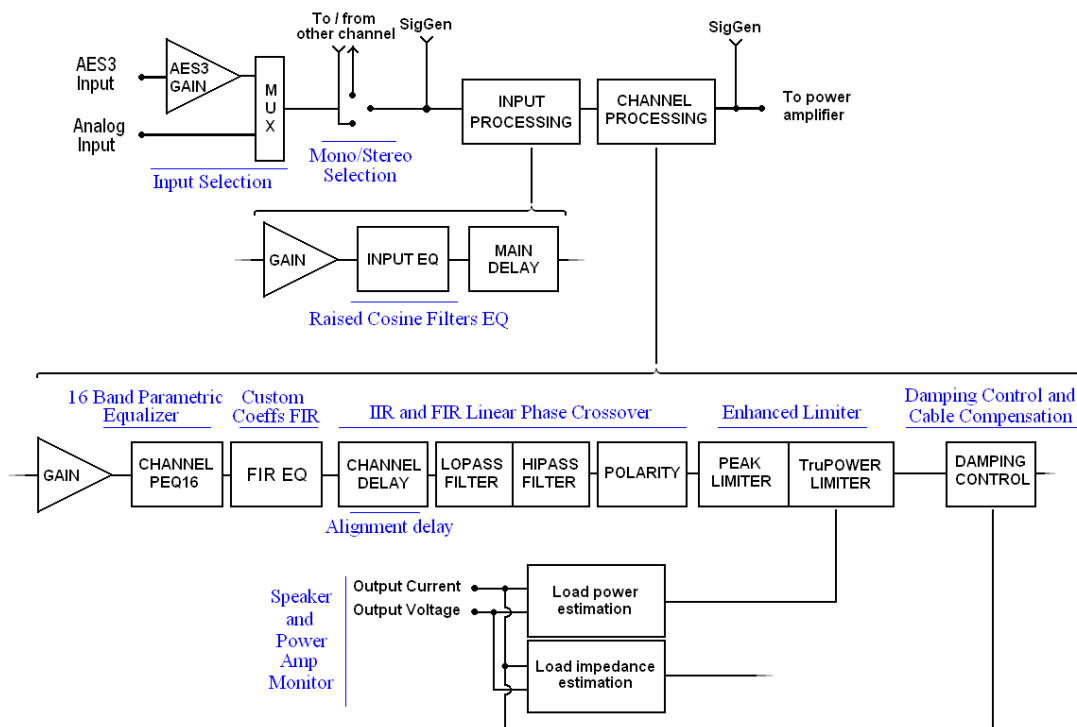
1. INTRODUCTION	6
2. DSP PROCESSING CHAIN	6
3. KDSP SETTINGS	7
KDSP SUBMENU	7
RESTORE FLAT	7
COMMON SETTINGS	7
SOURCE SELECTION	7
AES3	8
SOUND VELOCITY	8
CH1, CH2, CH1+2 SETTINGS	8
EQs	8
LP FILTER and HP FILTER	9
POLARITY	10
CHANNEL DELAY	10
GAIN	10
PEAK LIMITER	10
POWER LIMITER	10
DAMPING CONTROL	11
CHANNEL SETTINGS COPY	12
INPUT EQ	12
4. ARMONÍA SOFTWARE CONTROLLED FEATURES	13
INPUT PROCESSING	13
OUTPUT FIR EQUALIZER FILTER	13
ADVANCED LOAD IMPEDANCE MEASUREMENT	13
ARBITRARY WAVEFORM SIGNAL GENERATOR	13
5. HOWTO	14
SETUP A CROSSOVER	14
SETUP AES3 INPUT WITH ANALOG BACKUP	14
SETUP A BRIDGED CONFIGURATION	14
ADDENDUM: CHANGES IN FIRMWARE VERSION 4.0.2-2000	15

1. INTRODUCTION

The KDSP is a digital signal processing add-on board for the Powersoft K-Series amplifiers. KDSP can be used to optimize the performances of your system by means of powerful crossovers and equalizers. High reliability is guaranteed in any conditions through the use of advanced limiters, while loudspeakers working parameters are continuously monitored and reported on the remote control software. This documents describes the features and operating instructions of the KDSP board.

2. DSP PROCESSING CHAIN

The KDSP board embeds a very advanced sound processor. There are some features found only in high level sound processor, and some new functionalities that can be implemented only if a complete DSP/amplifier integration is done, as for the Powersoft KSeries amplifier. A complete block diagram of the processing chain is shown below.



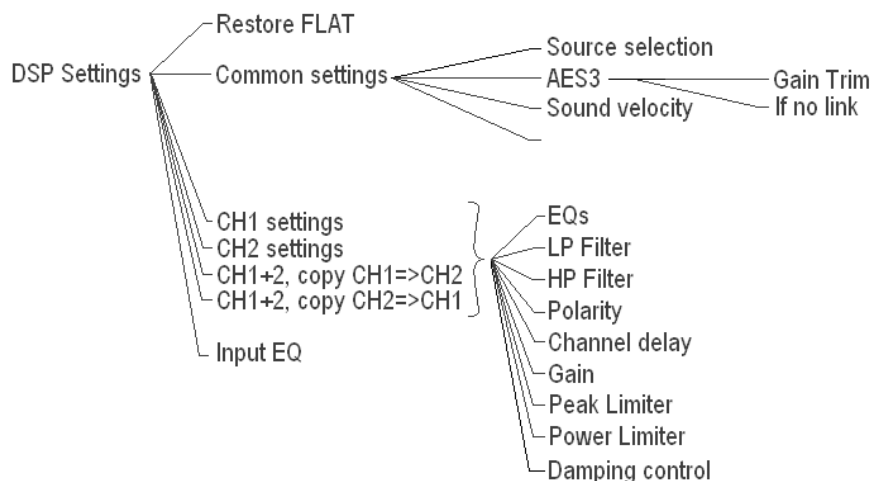
The features of the current DSP release are:

- ✓ AES3 digital input
- ✓ high performance (128 dB SNR) analog input: get the optimal quality whatever is your level
- ✓ high performance (122 dB SNR) analog output: no quality loss with DSP inserted in the path
- ✓ latest generation SHARC DSP with 40 bit floating point signal path for noiseless processing
- ✓ enhanced IIR filtering routine optimized for the highest SNR
- ✓ up to 2 seconds of main delay
- ✓ PEQ16: fully parametric equalizer, 16 band, each with 8 different filter shapes (peaking, hi/lo shelving, hi/lo pass EQ, band pass, band stop, all pass)
- ✓ FIR filter equalizer with 256 user defined coefficients

- ✓ alignment delay with high temporal resolution
- ✓ crossover with IIR Butterworth, Bessel and Linkwitz-Riley filters
- ✓ advanced crossover with FIR linear phase brickwall filters
- ✓ advanced load impedance estimation
- ✓ peak limiter with lookahead
- ✓ 4096 sample arbitrary waveform signal generator for system testing
- ✓ NEW: arbitrary frequency response input equalizer, up to 256 raised cosine filter cells
- ✓ NEW: true power limiter to avoid burning voice coils while getting the maximum performances from drivers
- ✓ NEW: **patented** damping control and cable resistance compensation: get optimally damped bass response with any cabling length

3. KDSP SETTINGS

KDSP SUBMENU



RESTORE FLAT

This command resets to default values all the DSP parameters, after a confirmation request. Flat frequency response, 0 dB gain, + polarity, no delay, no limiters, no damping control, input processing disabled.

The parameters belonging to input signal selection, such as the stereo/parallel input mode and the AES3 gain trim are NOT modified.

COMMON SETTINGS

These parameters are applied (identical) to both channels.

SOURCE SELECTION

This menu allows to choose the input selection mode.

In “Stereo Mode” the signal on CH1 IN is processed and then goes to CH1 OUT, and that on CH2 IN is processed and then goes to CH2 OUT.

In “Parallel from CH1” the signal on CH1 IN goes on both CH1 and CH2 out, through independent DSP processing. In similar way, in “Parallel from CH2” mode the source is the signal on CH2 IN.

AES3

This menu controls the AES3 input options.

GAIN TRIM

This menu sets the gain to be applied at the signal coming from the AES3 digital input. Setting 0 dB makes a full-scale digital signal equivalent to an analog input signal of 20 dBu.

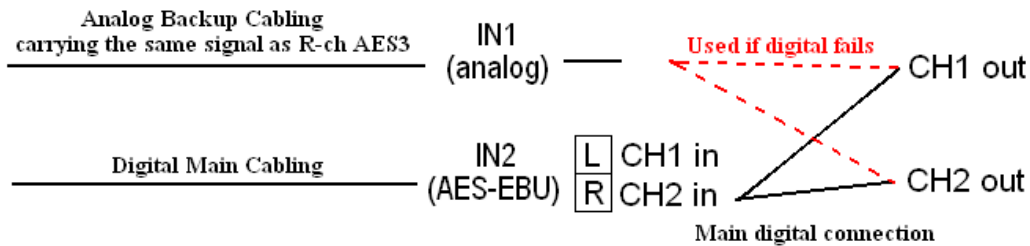
IF NO LINK

This menu controls the behaviour if the AES3 signal connection fails or becomes unreliable (>1% of errors). You can choose either “mute” or “analog”.

If “mute” is selected, when the AES3 connection fails, the amplifier will mute the output.

If “analog” is selected, when the AES3 connection fails, the amplifier will take the signal from the analog input. The switching is done in real time in order to avoid any glitch. If the input levels are correctly matched between analog input and AES3 input (use AES3 GAIN TRIM parameter), the switching between AES3 and analog will be inaudible.

When you use the analog input backup, you must run the amplifier in parallel mode to select only one digital input channel and you have to connect the corresponding analog signal cable in the ANALOG IN1 plug. There is only one working way to cable the amplifier:



Analog backup mode connection: the amplifier runs with AES3 input selected, in parallel mode. In the above example diagram, the R channel of the AES3 stream is selected, which is the “Parallel from CH2” choice from Source Selection submenu. Also the L channel can be used, by selecting “Parallel from CH1”. This signal goes, through processing, to both amplifier outputs. A cable with an analog signal identical to that provided by AES3 should be connected to IN1 of the amplifier for backup purposes. The level should be accurately trimmed, by using the AES3 gain trim setting, or by changing the analog signal level.

SOUND VELOCITY

This option sets the sound velocity used for time to distance conversions throughout the local interface. It can be set from 320 m/s to 360 m/s.

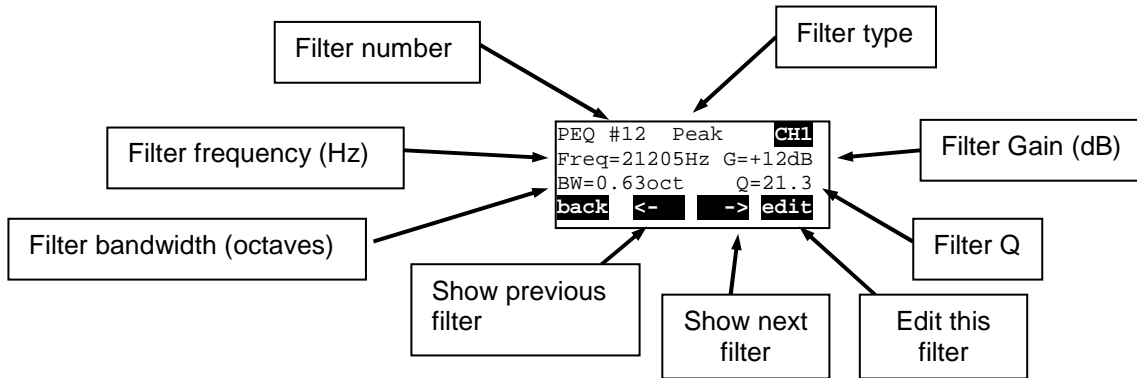
CH1, CH2, CH1+2 SETTINGS

In all these submenus, the channel (or the channels) that are being edited are shown on the top-right corner: CH1, CH2, or 1+2 when you are editing both channels at once.

EQs

This menu enters the parametric equalizer input interface, that lists the 16 parametric filters one by one. The current filter number is shown on the left of the first line. By pressing the arrows, you can move

from one filter to the next or to the previous. The parameters are sintetized on the display: you can check the settings of the whole equalizer by simply move through all the 16 filters.



By clicking “edit”, you can change the settings for the selected filter.

```

Edit PEQ #12 CH1
Active = ON/OFF
Frequency(Hz)= 11205
Gain(dB)= +12
Q factor= 7.3
Bandwidth(oct)=0.23
Type = Peaking
back  v  ^  edit
    
```

You can enable or disable (flat response) the filter.

You can choose the filter type from the following types:

- ✓ peaking (Peak)
- ✓ low (LSh) and high (HSh) shelving (from 3 to 15 dB/oct)
- ✓ low pass EQ (LPEQ) and high pass EQ (HPEQ)
- ✓ bandstop (Notch)
- ✓ bandpass (BandP)
- ✓ allpass (AllP)

You can change the frequency (20 to 20000Hz, 1/96 octave step), the gain (only if filter is peaking or shelving, from -15 to +15 dB, 0.1 dB step), the quality factor (for all filters but shelving, 0.1 to 30, 0.1 step), and the slope (only for shelving, from 3 to 15 dB/oct).

Peaking/Bandpass/Bandstop/Allpass

```

Q Factor CH1
[=== ] 2.3
BW = 0.59 oct
back  v  ^  edit
    
```

Lowpass/Highpass EQ

```

Q Factor CH1
[=== ] 2.3
Gain@f0 = 7.2 dB
back  v  ^  edit
    
```

Shelving

```

Slope CH1
[=== ] 0.75
9.0 dB/oct
back  v  ^  edit
    
```

Q/Bandwidth, Q/Gain and shelving slope dual mode input.

LP FILTER and HP FILTER

This menu sets up the crossover filters. There are 2 available filters, one lowpass filter and one highpass filter. By using both, you will obtain a bandpass response. Both traditional IIR and brickwall linear phase FIR are implemented.

The classic IIR crossover filters can be selected as a Butterworth shape, Bessel shape and Linkwitz-Riley shape. In the first 2 cases, the Freq. parameter defines the -3 dB point, in the latter, the -6 dB point. The slope is freely selectable from 6 dB/octave (1st order filter) to 48 dB/octave (8th order filter). The FIR filters can be selected as normal (FIR) or enhanced (FIR+). The enhanced version gives a higher rejection of out of band signals, at the expense of a little (<10 degrees) phase nonlinearity. In both cases, the minimum working frequency is related to the desired latency, and with standard settings is limited to about 400 Hz. So it is better to use the FIR filters to crossover upper midranges or mid-high drivers, for which the phase coherency is a key point.

```
HP FILTER CH1
Active = ON
Freq.(Hz) = 1200
Slope (dB/oct) = 48
Shape = Butterworth
back v ^ edit
```

```
LP FILTER CH1
Active = ON
Freq.(Hz) = 3307
Slope (dB/oct) = 24
Shape = Link.-Riley
back v ^ edit
```

```
HP FILTER CH1
Active = ON
Freq.(Hz) = 1307
Slope (dB/oct) = 24
Shape = FIR+
back v ^ edit
```

POLARITY

This menu allows to reverse the signal polarity. It is equivalent to a 180° phase change.

CHANNEL DELAY

This menu allows to time-align two different transducers on the two amplifier channels. The selectable delay varies from 0 to 30 ms (about 10 meters), with one sample step ($1/96000^{\text{th}}$ second or 10.4 us, about 3.5 mm) step.

GAIN

This menu changes the channel gain, from -40 dB to $+15$ dB, with a 0.1 dB step.

PEAK LIMITER

This menu controls the output peak limiter. The peak limiter decreases the gain in order to reduce the maximum output voltage. The parameters are the voltage threshold at which the gain begins to get reduced, the attack time, that is the response time of the limiter intervention and the decay time, that is the time constant after which the gain is restored to the nominal gain. You should in first setup the time parameters, and then you should adjust the threshold voltage. When editing the threshold, the display shows in real-time the gain reduction due to the limiting function ($GR = xx.x$ dB), together with the limiting voltage referred to the input.

The limiter is conceived to pass-through inalterated the signal peaks lasting less than the attack time parameter, in order to exploit the exceptional dynamics of the K-Series amplifier.

The limiter has also a lookahead buffer, in order to soften the clipping and to minimize the distortion, giving a superior sonic performance. The lookahead time is 0.5 ms.

```
PEAK LIMITER CH1
Active = ON
Threshold(Vpk) = 200
Attack (ms) = 10
Decay (ms) = 300
back v ^ edit
```

```
Threshold CH1
[=====] 140 Vpk
GR=11.1dB In=12.3dBu
ok + - fast
```

POWER LIMITER

This menu controls the output power limiter. The power limiter decreases the gain in order to reduce the power delivered to the load.

In “TruePower” operating mode, the unique ability to measure the load current gives the opportunity to estimate the actual active power output by the amplifier.

Given the low efficiency of electromechanical transducers (<10%), almost the same level of power is transformed into heat into the voice coil. This limiter is intended to avoid burning the voice coils of the drivers while exploiting their maximum performances.

In “Power @ 8 ohm” mode, the output power is estimated by measuring the RMS of the output voltage, assuming an 8 ohm load. This mode allows to create a settings that works quite well for any number of paralleled connected speakers. For example, if you set the threshold to 150W, you will get 150W with 8 ohm load (1 cabinet), 300 W with 4 ohm load (2 cabinets) and 600 W with 2 ohms load (4 cabinets).

The parameters are the power threshold in W at which the gain begins to get reduced, the attack time, that is the response time of the limiter intervention and the decay time, that is the time constant after which the gain is restored to the nominal gain. You should select as power threshold the true power handling of your speaker (it is usually between one third and one half of the declared RMS or AES power handling, see the table below), with time constants related to the thermal inertia of your speaker (the mass to be heated). So, huge 4” VC subwoofers could require some seconds of attack and some seconds of release time, while tiny 1” VC tweeters may need less than 1/10 of second of attack time.

Consider that if you use more than one speaker, the limiter power threshold must be multiplied by the number of connected speakers, but only in the TruePower mode.

When editing the threshold, the display shows in real-time the gain reduction due to the combined peak/power limiting function (GR = xx.x dB) and the power really delivered to the load (Pavg=xxxxW).

```

POWER LIMITER  CH1
Mode = TruePower
Threshold(W) = 120
Attack (ms) = 1500
Decay (ms) = 6000
back  V  A  edit
    
```

```

Threshold  CH1
[== ] 40 W
GR=7.2dB  Pavg= 45W
ok  +  -  fast
    
```

DRIVER	POWER THRESHOLD	ATTACK TIME	DECAY TIME
1” VC tweeter	10...20 W	100 ms	300 ms
1.5” VC tweeter	20...30 W	150 ms	300 ms
2” VC horn driver	20...40 W	200 ms	400 ms
3” VC horn driver	30...50 W	300 ms	500 ms
4” VC horn driver	40...60 W	500 ms	3000 ms
2” VC midrange	30...100 W	500 ms	3000 ms
3” VC midbass	50...150 W	1000 ms	5000 ms
4” VC woofer	100...200 W	2000 ms	5000 ms
4” VC subwoofer	150...250 W	4000 ms	8000 ms
6” VC massive subwoofer	250...500 W	6000 ms	10000 ms

Suggested power limiter settings for common used drivers.

The power limiter should be used to protect the drivers from burning, and at working levels it normally shouldn’t activate. Check the gain reduction, it should not raise over 2-4 dB even in the loudest piece of music, in order to obtain the optimal sound. Please note that a common musical signal has very loud peaks, but a small average level, while a continuous tone has a much higher average power even if it has a modest impact on the hear.

DAMPING CONTROL

This unique and **patented** feature allows to add a “virtual” resistor in series with the amplifier output. The value of this resistor can also be negative in order to get the correct damping factor with any cabling

you use. For example, if you use a 10 meters cable for powering the subs, you will add a parasitic series resistance (0.3 ohm for instance). By enabling the damping control, you can add a negative resistance that compensates the cable resistance.

WARNING: when enabled, a lowpass filter cutting around 400 Hz is automatically inserted into the chain. This feature is intended to be used only for subwoofer applications.

AWG	Sq mm	Length	Resistance
16	2 x 1.5 mm ²	5 m	0.13 Ohm
16	2 x 1.5 mm ²	10 m	0.26 Ohm
16	2 x 1.5 mm ²	20 m	0.52 Ohm
14	2 x 2.5 mm ²	5 m	0.08 Ohm
14	2 x 2.5 mm ²	10 m	0.16 Ohm
14	2 x 2.5 mm ²	20 m	0.32 Ohm
12	2 x 4 mm ²	5 m	0.05 Ohm
12	2 x 4 mm ²	10 m	0.10 Ohm
12	2 x 4 mm ²	20 m	0.20 Ohm

Typical cabling resistance.

Another advantage is that you can take in account the voice coil rise in resistance due to heating, getting a correctly damped bass response in the average working condition. For example, if you will run hard your subwoofer amplifiers during your performance, you should add a negative resistance between 1 and 2 Ohms to keep the correctly damped response. This value is correct for an 8 Ohm typical subwoofer, remember to divide it for the number of paralleled speakers on the K amplifier output. Take into account that if you use the system at a lower level, you should select a smaller negative resistance to avoid an overdamped response.

Average power / rated power	Power compression	Equivalent series resistance for 8 Ohm driver
10%	1.4 dB	1.0 Ohm
20%	2.0 dB	1.4 Ohm
50%	2.8 dB	2.1 Ohm
100%	4.5 dB	3.8 Ohm

Typical rise in resistance due to voice coil heating.

Note the huge amount (3.8 Ohm) when the driver reaches its thermal limit.

CHANNEL SETTINGS COPY

By selecting the “CH1+2, copy CH1=>CH2” submenu, all the CH2 DSP settings are cloned from the CH1, and then the parameters of both channels are changed at the same time.

The same for “CH1+2, copy CH2=>CH1”.

INPUT EQ

This menu allows to turn on / turn off the input processing block. This can be useful to reset the amplifier to the original “output processing only” behavior even without using the software, discarding all the input settings due to the Armonía software editing. As an hint, it is better to save amplifier presets with this setting turned off. The user can load the preset and be sure that only the output processing is enabled. The burden of reenabling and setting the input processing is left to the remote control software.

4. ARMONÍA SOFTWARE CONTROLLED FEATURES

These features are controlled only by using Powersoft Armonía, the new remote control software. Local editing of those features is not supported due to the huge parameter set needed.

INPUT PROCESSING

The input processing block consists of a gain control (+15 dB...-40 dB range), a delay control (0 ... 2 s with single sample stepping) and a powerful equalizer.

The new input equalizer implements an arbitrary frequency response defined by using up to 256 raised cosine filter cells. It is used mainly to equalize the whole system or a part of it depending on various conditions such as environment, sound style (artistic eq) and other corrections not related to single transducer. The direct advantage is the replacement of multiple system equalizers used for main PA as well as stage monitors.

Armonía software uses those cells to implement multiple levels amplifier grouping in a very flexible way. Please refer to the Armonía user manual for details on grouping.

Input equalizer settings are stored in the current settings memory in order to reload them at power on, but they can't be stored in a local memory preset or smartcard preset.

OUTPUT FIR EQUALIZER FILTER

Modern driver equalization techniques involve the usage of filters with well defined impulse response, optimized for improving linearity and yielding an accurate time response. This translates to using a FIR equalizer with user defined coefficients. KDSP board supports an output FIR filter defined by 256 coefficients at 96 kHz sample rate, or up to 2.67 ms of impulse response. Those coefficients are loaded by Armonía from a text file and transferred to the appropriate amplifier channel.

Amplifier stores those coefficients in the current settings of the channel, and can save them in a local memory preset or a smartcard preset, since they are related to the transducer (output) processing.

ADVANCED LOAD IMPEDANCE MEASUREMENT

Since the DSP acquires the current and voltage delivered by the amplifier output to the load, it has been possible to estimate in real time the load impedance curve plotted against frequency. The measurement is done from 20 Hz up to 20 KHz and can be displayed by using the Armonía software.

ARBITRARY WAVEFORM SIGNAL GENERATOR

A signal generator has been added in order to play an arbitrary signal either through the processing chain or directly to the output of the amplifier. The generator plays in looping mode up to 4096 samples downloaded by means of the Armonía software. This could be used to generate noises, tones and pulses of various kinds and at frequencies up to the upper bandwidth limit of the amplifier, in order to test system behavior and monitor load impedance without the need of an external signal generator.

5. HOWTO

SETUP A CROSSOVER

For example, midrange on CH1 and high frequency driver on CH2.

Edit CH1, go to LP Filter, set Active=ON, set Frequency = 2000 Hz, set Shape=Linkwitz-Riley, set Slope=24dB/oct. You could also experiment with shape=FIR.

ON OFF* back v ^ sel	Frequency CH1 [=====] 2000 Hz 100 1k 10k back - + fast	Butterworth CH1 Bessel Link.-Riley* FIR Lin Phase FIR+ Lin Phase back ^ v fast	Slope (dB/oct) CH1 [=====] 24dB/oct back - + fast
----------------------------------	---	---	---

Final result:

```

LP FILTER   CH1
Active = ON
Freq.(Hz) = 2000
Slope (dB/oct) = 24
Shape = Link.-Riley
back   v   ^   edit
    
```

Edit CH2, go to HP Filter, set Active=ON, set Frequency = 2000 Hz, set Shape=Linkwitz-Riley (or FIR), set Slope=24dB/oct. You could also experiment with shape=FIR.

ON OFF* back v ^ sel	Frequency CH2 [=====] 2000 Hz 100 1k 10k back - + fast	Butterworth CH2 Bessel Link.-Riley* FIR Lin Phase FIR+ Lin Phase back ^ v fast	Slope (dB/oct) CH2 [=====] 24 back - + fast
----------------------------------	---	---	---

Final result:

```

HP FILTER   CH2
Active = ON
Freq.(Hz) = 2000
Slope (dB/oct) = 24
Shape = Link.-Riley
back   v   ^   edit
    
```

On CH2, edit Ch Delay to temporal align midrange with highs horn.

On CH2, edit Gain to align levels.

SETUP AES3 INPUT WITH ANALOG BACKUP

Refer to IF NO LINK command.

SETUP A BRIDGED CONFIGURATION

If you already have a preset made for a non-bridged configuration, lets say for the CH2 for example, you should load the preset, then choose to edit “CH1+2, copy CH2=>CH1”. The CH1 now is identical to CH2. With this command you can also edit both channels at the same time.

Otherwise, you can right away choose “CH1+2, copy CH1=>CH2” or “CH1+2, copy CH2=>CH1”, then edit both channels at once.

Remember that, in bridged mode, the 2 DSP channels should be identical to get the right results.

ADDENDUM: CHANGES IN FIRMWARE VERSION 4.0.2-2000

To fully support the Armonía software and the amplifier grouping functionalities, some major changes were made to the firmware of the KDSP board.

The main differences from previous firmware version are:

- Power Limiter now works also in Power @ 8 ohm mode, in addition to TruePower mode
- Implementation of an input processing block that consists in a gain control, a delay and an arbitrary response equalizer based on up to 256 Raised Cosine Filter blocks
- Load impedance vs frequency curve measurement for advanced load monitoring
- Output FIR filter with used defined filter coefficients for custom equalization
- Arbitrary waveform signal generator

As a backside effect, be aware of the following considerations:

- Main delay cannot be edited locally anymore
- System latency has been increased to 5 ms in order to accommodate the input equalizer block
- When you update the firmware from a previous version, all the local memory presets are lost. Also smartcard presets can't be used anymore. Please take care of storing them on your PC using the PowerControlManager software, and then import them with the Armonía software.