AMPLIFIER INSERTION MODULE
WITH DSP TECHNOLOGY

AKB-400DSP
Best.-Nr. 32.0010

INSTRUCTION MANUAL

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Product description

Features

- 2x amplifier
- Switch mode power supply
- Balanced audio in
- Balanced audio through loop
- Two channel active filtering
- Fully user-configurable filters
- Firmware update by USB
- DSP supply voltage dependant clip limiter

Applications

This module can be used for the following setups:

- Active 2-channel system
- Active power sub-woofer

And with 2 modules it is also possible to set up a 3-channel system with one module for the separate Subwoofer, and a second module for the top in a 2-channel system.

Connections

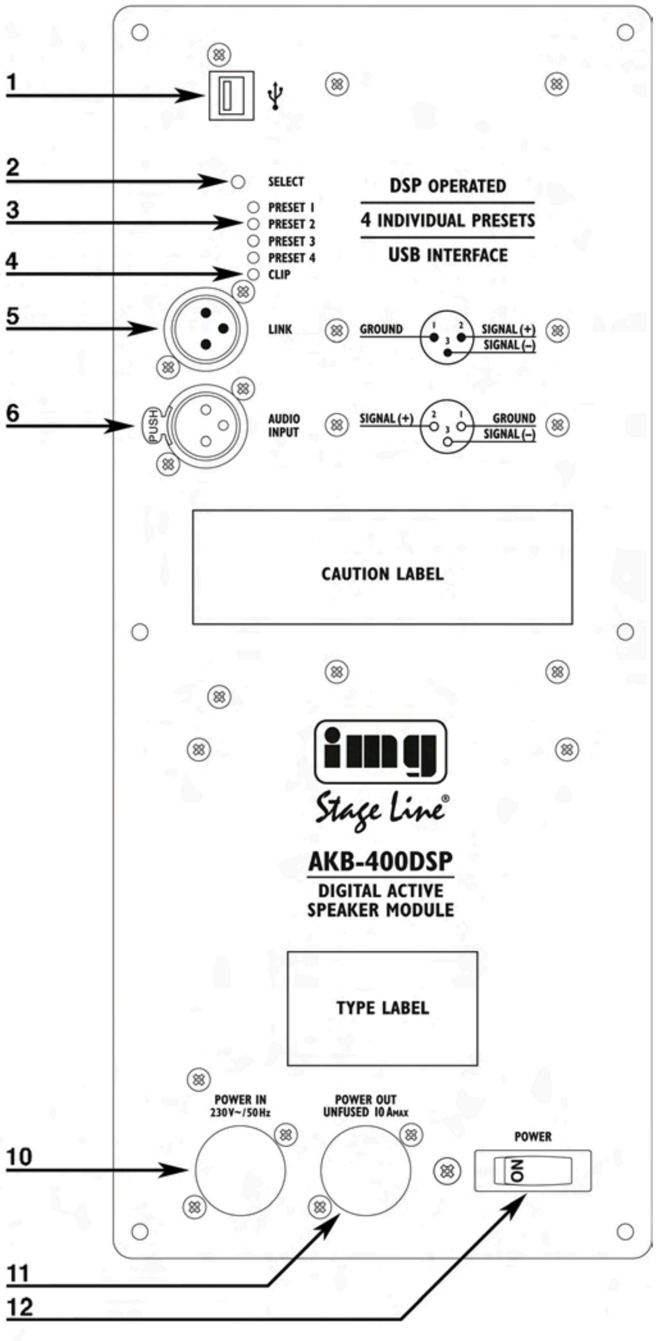

Figure 1

Overview of the connections on the front of the AKB-400DSP

	Function
1	USB input
2	Select button
3	Preset LED's
4	Clip detect LED
5	Analogue Link
6	Analogue Audio Input
10	Power input
11	Power output
12	On/Off switch

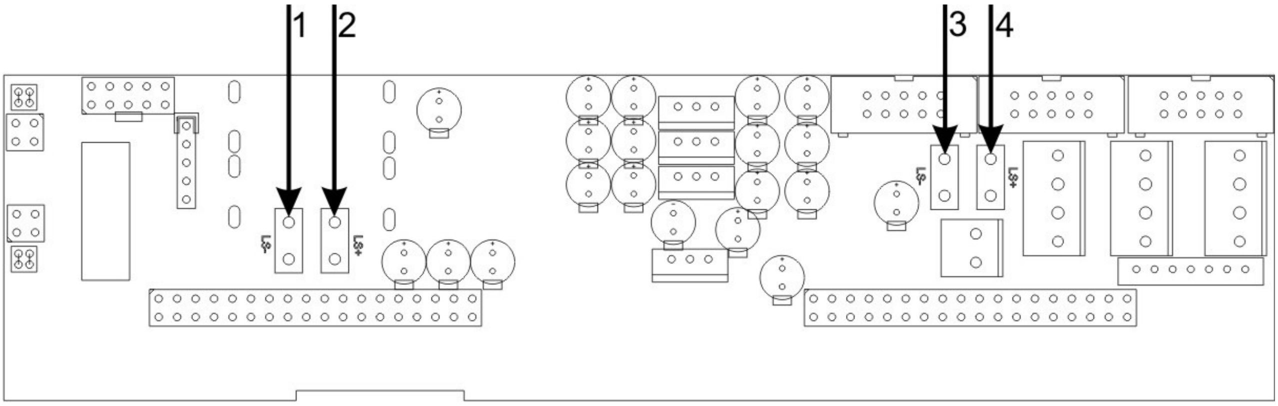

Figure 2

Overview of the connections on the back of the AKB-400DSP

	Function
1	J16, Speaker -
2	J14, Speaker +
3	J17, Speaker -
4	J15, Speaker +

* For bridged mode use J14 and J15. Signal on J14 is inverted

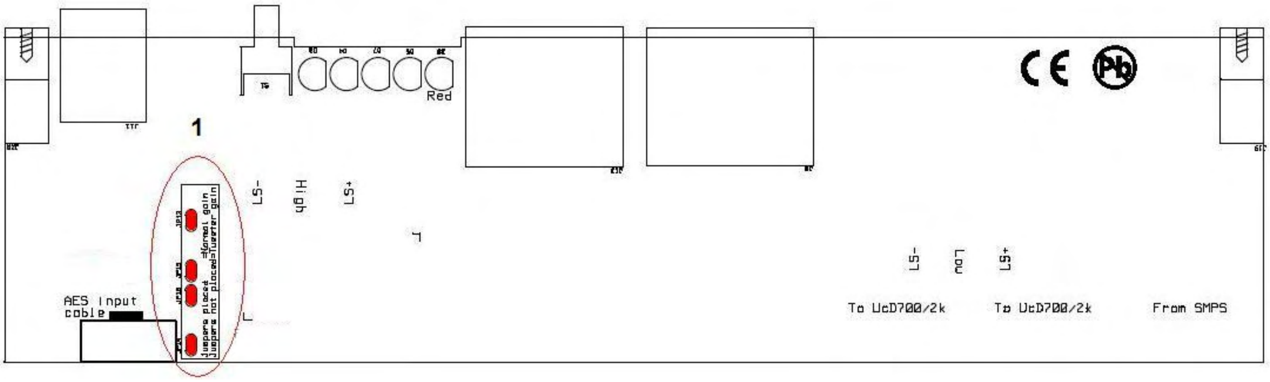

Figure 3

	Function
1	Jumpers placed is normal function
	Jumpers not placed lowers High channel gain with 10dB, this must be corrected in “Filter design” software under Tools Options

System information

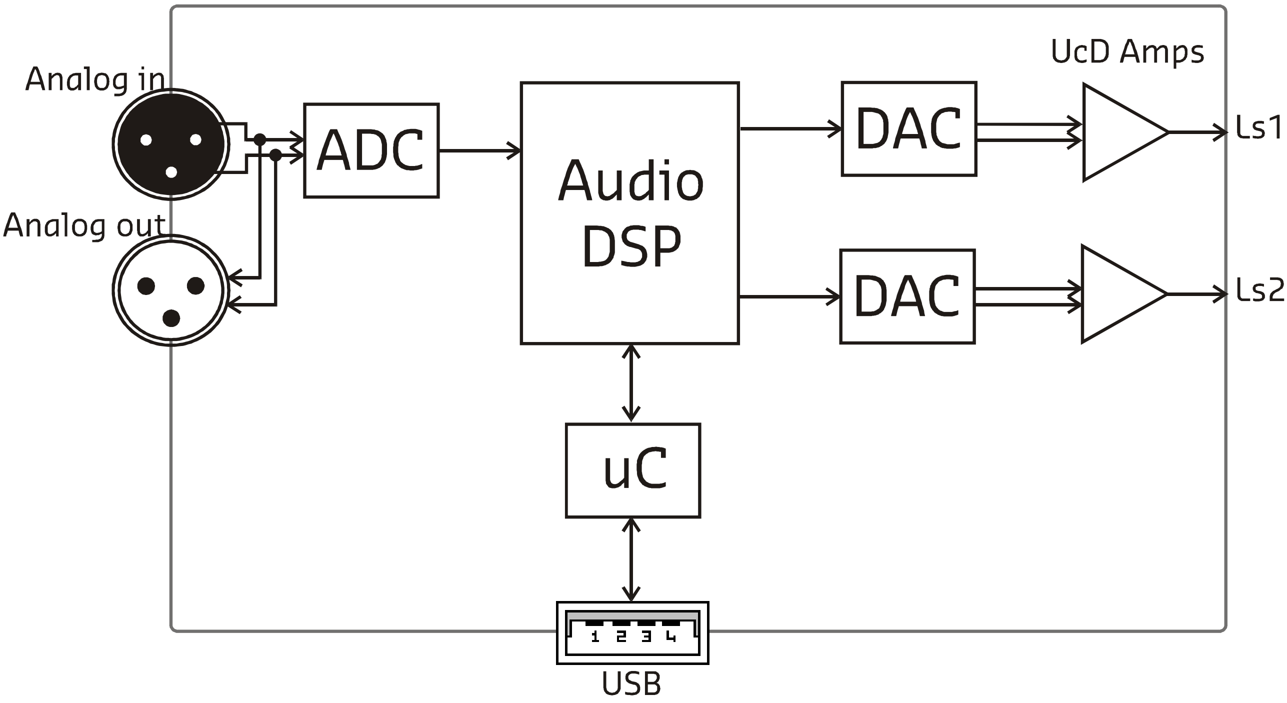

Figure 4: Block diagram

Two-channel active amplifier with DSP filter control.

Description

The AKB-400DSP is a plate amp for use in powered speaker systems. As an active speaker controller, a AKB-400DSP can form the basis of a powerful active two-way monitor.
Figure 4 shows the audio path of the AKB-400DSP
The module has two 400Watt UcD modules implemented.
The supply voltage is provided by a Switched Mode Power Supply module.

A PC controls the AKB-400DSP through the USB port. This connection is used to upload the configuration and filter settings. It will also be possible to update the firmware through USB.

Audio performance data

MBW=20kHz, unless otherwise noted. All filters set to unity. Noise levels unweighted.

Item	Symbol	Min	Typ	Max	Unit	Notes
Input level	Vin		+18		dBu	Vin max
Output level	LS1			400*	W	Into 4 Ohms
	LS2			400*	W	Into 4 Ohms
Signal/Noise	SNR		101		dB	ADC
			107		dB	DAC
Total harmonic distortion+Noise	THD+N		-90		dB	Without UcD400
DSP sampling rate	Fs		48		kHz	
Delay per channel		0	0	15000	us	Set in software
Supported digital sampling	Fs	32, 44.1, 48, 88.2, 96,			kHz	All input rates con-

rates		192				verted to 48kHz
Gain			+16		dB	volume at 0dB
Analogue Latency			1,12		ms	
AES Latency			3,58		ms	

* Note: Total power for the AKB-400DSP is limited to 500Watts total

Hardware Architecture

The standard version of the AKB-400DSP has an analogue balanced audio input.This analogue signal is converted to digital, processed by the DSP and then converted back to analogue. A microcontroller controls the DSP and communicates with the PC and can set the presets. An unbalanced source may also be applied, this signal is divided over the two input DAC’s to get better performance.

Controllers

The AKB-400DSP can only be controlled through USB.When the module is powered on it will automatically start up with the last settings. All setting, like volume and preset, are stored after a change is made.

Clip limiter

The AKB-400DSP has a build in clip protection. The module measures the supply voltage and uses this value for the limiter. Figure 5 shows the response graph of the limiter.
The hold time is fixed to 100ms and the decay is set to 100dB/s.

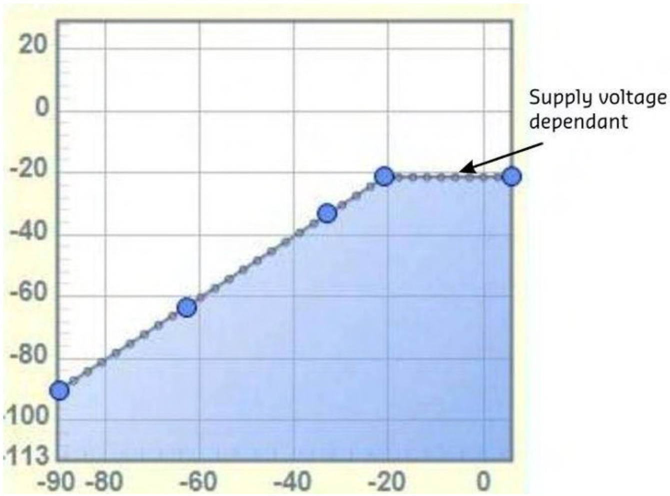

Figure 5

When the limiter for either channel is active the Clip led will be lit.

Thermal protection

All modules have a build-in thermal protection. This protection lowers the output gain 1 dB per degree starting at plate temperature of 70 degrees Celsius.
The outputs will be completely muted when the temperature rises over 90 degrees.

Product overview

Before you can use the AKB-400DSP in your particular setup, you first have to set the right settings. This is done by pc software, called DSP filter design.
Please do not connect any speakers to your system yet!

Hardware part

The AKB-400DSP can be used in 2 ways, as a 2-channel setup or as a one channel bridged version for more power. This is selected in software.
The module can store up to 4 presets. A preset holds the complete filter-settings of both channels. By this the user is able to make different system settings for its speakers, for example for different rooms and setups.
All of the presets are empty from factory setup, therefore the filters must be downloaded first. See section “examples”.

* note that for bridged mode both LS+ connections must be used, see section **Connections** for more information

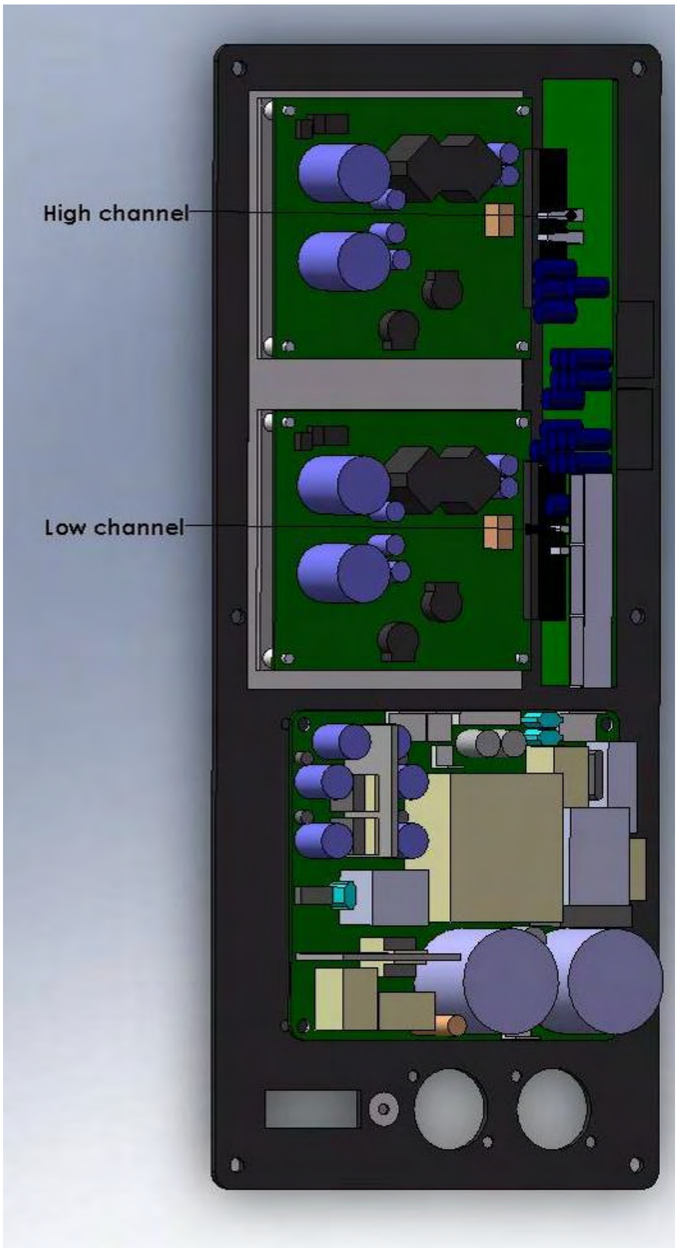

Figure 6

Software installation

System requirements:

- Pentium class or higher
- 64MB RAM
- USB1.0 or higher

Tested on Windows XP, Vista, Windows 7 and Windows 8.

All files are compressed in the setup.zip file. This zip file contains 1 DLL file for communication and an .EXE file.

1. Unzip the setup.zip file on your hard disk
2. Open the "filter design.EXE" by double clicking the file

Control panel

Now the program is ready for use. You will see the following window, called the control panel.

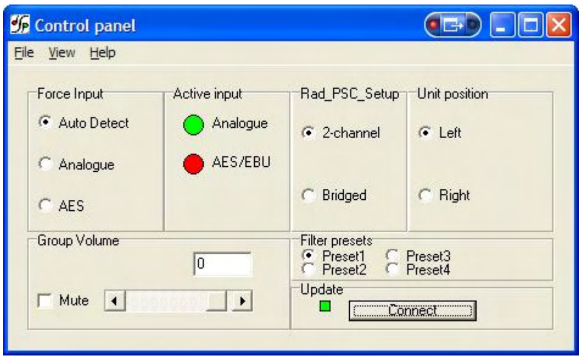

Figure 7

When the program is running connect your module to your pc by the USB cable. Windows will automatically detect and install the HID-device. When the installation is done the connected the connection light will turn green. You can also manually make a connection by the "connect" button, on the bottom right of the control panel.

Now you may adapt the settings of the module real time through USB. On the left side in the control panel you see the Force input and active input groups. This shows you the settings of the audio inputs and lets you control these inputs (only for digital version).
Audio can be provided analogue and digital. The selection of any of these inputs can be automatic, "auto detect". This means that when any of the inputs is presented there will be switched to this channel automatically. This is done by setting the analogue input as the standard source. During analogue in the other two inputs are scanned for any valid signal. The first channel that contains any valid audio is selected and becomes the new input source. But if you put in the USB cable but you want to use the analogue input you can force the analogue input to be used. This can be done for both inputs.
When the module selects one of the inputs this is shown by the active input lights.
Under the Setup select part the system setup can be selected. There can be chosen between:

2-channel	Bridged
Master on the left	Master on the left
Master on the right	Master on the right

* The channel assignment in analogue mode makes no difference because there is only one input channel

The last thing that can be controlled in this window is volume. This can be done by setting the scroll-bar to a desired position or by typing the value in the volume field, the value is send when pressed "enter".
When the presets are installed they may be selected, the module will switch to the selected preset. The presets can be filled in the "Filter design" section.

Filter design

When you want to make some filters for your module they can be designed in the "Filter design". Under view there can be switched between the control panel and the filter design window.
The following pages will give you a widespread instruction of the possibilities of the program.

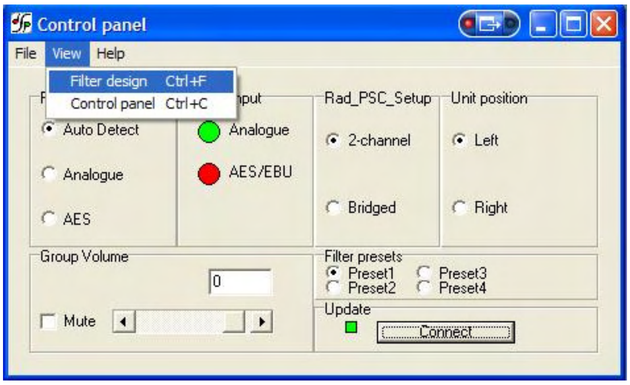

Figure 8

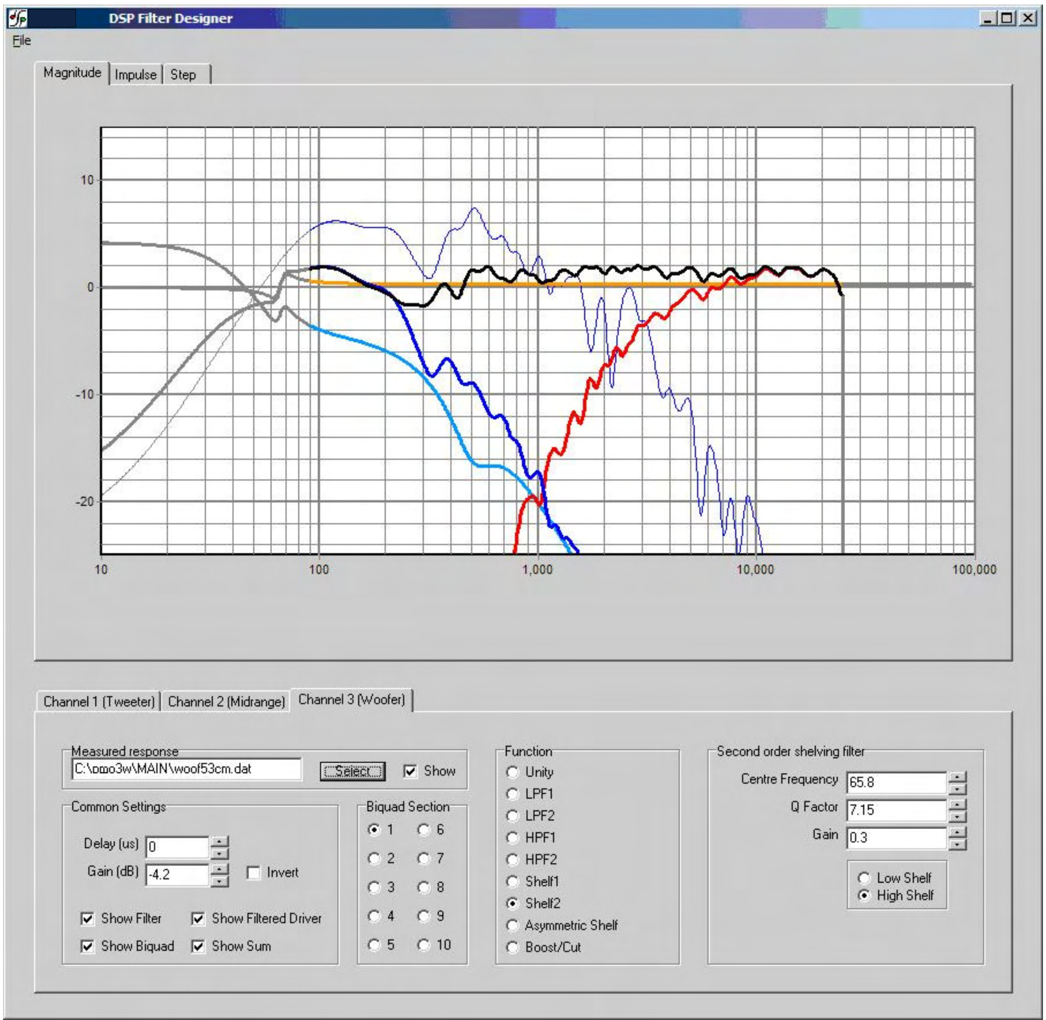

Figure 9

Graph Area

The magnitude tab shows the imported driver responses, filters, individual biquads, individual filtered driver responses and the sum.

Colour	Function
Blue, thin	Measured woofer response
Green, thin	Measured midrange response
Red. thin	Measured tweeter response
Blue, thick	Filtered woofer response
Green, thick	Filtered midrange response
Red, thick	Filtered tweeter response
Light blue, thick	Response of filter, selected channel only
Orange, thick	Response of selected biquad
Black, thick	Sum response

Having all of these on at the same time quickly produces an intractable mess so these graphs can be separately enabled or disabled in the filter definition area.
The impulse or step tabs show the time domain response of the imported drivers and the sum response, and are used to demarcate the anechoic portion.

Filter Definition Area

The two channel tabs, labelled Tweeter and Woofer are functionally identical. The top left frame is used to import response files. The "select" button opens a file. The "show" checkbox turns display of the measured graph on or off. The Common Settings box controls global gain (for each channel), delay, and the visibility of plots.
The amplified channels have up to 12 biquads, selected using the "Biquad Section" radio buttons in the middle. To the right is a settings area specific to the type of function selected. Unused biquads are set to unity.
The selected biquad is edited by selecting a function and setting relevant parameters.

Settings Window

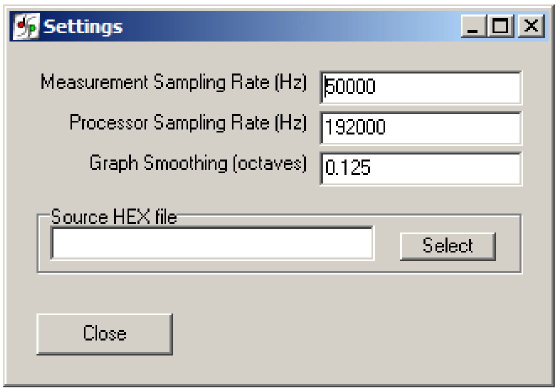

Figure 10

The settings window is under File > Settings... Measurement sampling rate sets the sample rate used in the imported response files (typically 48kHz). Processor sampling rate is that of the DSP hardware. Note that this setting does not control the sampling rate of the hardware. Rather, it informs the filter design application of what that sampling rate is. In short, leave this at 46,875kHz.
The select button opens a file dialogue box witch is not used in this application.

Work flow

Measurement

Measuring using the DSP unit set to "flat"

Perform impulse response measurements for each driver separately. Save the entire impulse record – truncation can be done later on the filter design program.

Measuring using an external amplifier

A separate amplifier may also be used for measuring the drivers, provided the amplifier's output impedance is as low as the DSP unit's.

Importing response data

Select the tab for the channel you want to import and click "select". The filter designer expects the impulse response measurement as a text file with one sample per line.

There is no restriction on the absolute gain of the impulse response data. The only thing that matters is that the absolute gain be the same for all three measurements. The filter designer computes a gain offset based on all loaded responses to centre them collectively on the vertical scale.

Truncating response data

Switch to the impulse or step response graph.

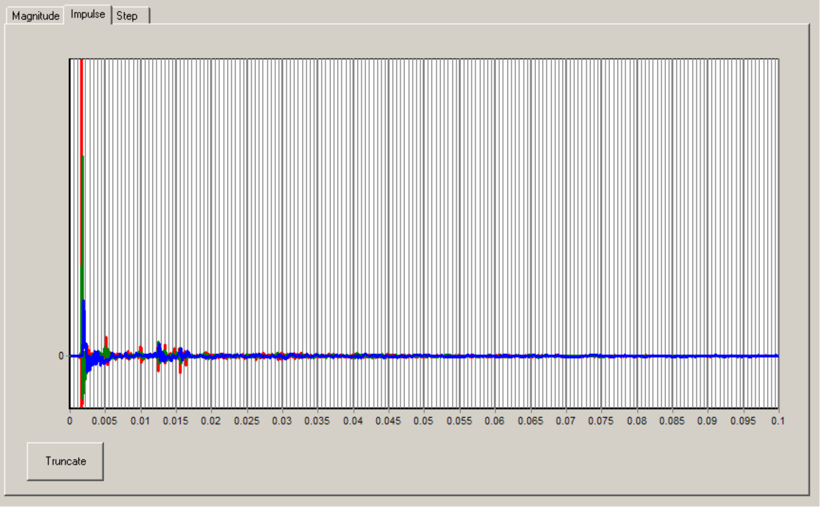

Figure 11

The first echo is apparent at 5ms. Zoom in until you see only the anechoic portion of the impulse response. Dragging the mouse, left-button down, from left to right marks a zoom area. Dragging from right to left zooms out. Dragging with the right button down pans the plot left and right.

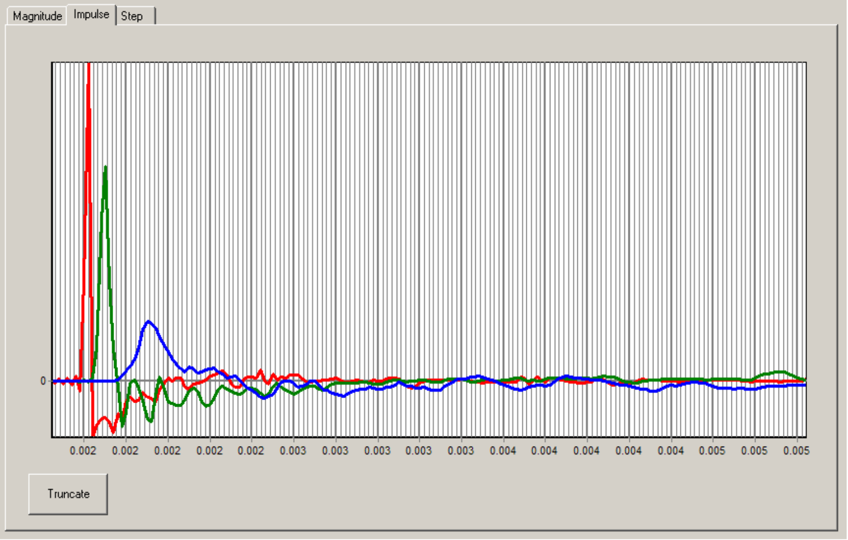

Figure 12

Click "truncate". Anything currently outside the display is drawn in grey and not processed. You will notice that in the frequency graph a portion of the low-frequency response is also drawn in grey. This is to remind the user that insufficient information is available to make *any* correction below

this frequency. A way of obtaining quasi anechoic low-frequency measurements is making close-up measurements. Working with such measurements requires a good deal of interpretation but it is doable.
Important: Avoid making any corrections for which no anechoic data is available. If reflections are included, they are guaranteed to dominate the measurement at low frequencies, and you will end up making corrections for circumstances that are highly specific to the room in which the measurement was made. Power-response data can only be made in a proper echo chamber or preferably, by collating a large number of anechoic off-axis measurements. A reverberant measurement in a normal live room just won't do.

The steps of loading and truncating data can be repeated at any time. This can be particularly practical when combining close-up and far-field measurements during the filter design phase. The window in Figure 12 shows the result of this. The small knot of corrections made around 70Hz is based on close-up data first loaded separately. The LF section of the test mule is quite smooth apart from one internal standing wave.

Designing filters

Biquad function	Parameters	Use
Unity	-	Section is not used
LPF1	Cut-off frequency (always -3dB)	First order lowpass
LPF2	Cut-off frequency (asymptotically) Q	Second order lowpass. A Q of 0.71 corresponds to butterworth. 0.5 corresponds to LR2. Two identical sections with a Q of 0.71 form an LR4 filter.
HPF1	Cut-off frequency (always -3dB)	First order highpass
HPF2	Cut-off frequency (asymptotically) Q	Second order highpass. A Q of 0.71 corresponds to butterworth. 0.5 corresponds to LR2. Two identical sections with a Q of 0.71 form an LR4 filter.
Shelf1	Centre Frequency (halfway point) Gain Direction	First order shelf. Useful for baffle-step correction
Shelf2	Centre Frequency (halfway point) Gain Q Direction	Second order shelf. Useful for correcting internal cabinet resonances and for the midband peak/dip combo of most mid-woofer speakers.
Asymmetric Shelf	Pole frequency and Q Zero frequency and Q	Equalising the bottom end of closed-box woofers with large magnets
Boost/Cut	Centre frequency Q gain	Dip/peak filter. For peaks, Q is defined by the poles. For dips, Q is defined by the zeros. Thus the same filter with opposite gains will cancel.

The first step is equalising the magnitude responses of the drivers flat over their entire useable frequency range.
The weapons of choice are shelving filters, and boost/cut sections. A sharp peak followed by an equally sharp dip can be corrected using a second-order shelving filter with a high Q.
Exercise care when deciding what to correct. When correcting for diffraction errors, do not exceed a Q of 3 lest the cure be worse than the ailment. Errors that are caused inside the driver, or internal cabinet resonances that emanate through the same diaphragm, may be corrected ruthlessly – provided the measurement has sufficient resolution to pin them down.
As a rule of thumb, sharp dips are diffraction artefacts while sharp peaks are caused by the drivers themselves. Exceptions are room resonances (if the response is not correctly truncated) and diffractions on repetitive patterns.

The second step is designing the actual crossover filters. All the usual strategies work. Delaying higher frequency drivers with respect to lower-frequency ones is a powerful alternative to using asymmetric slopes and yields substantially improved coherence through the crossover region.

Download

Under download the user can press "load DSP" to download its designed filters to the module, but only for the current used preset!

Firmware update

Every module has the ability to update its firmware, when img Stage Line® provides a new firmware version. The firmware can be simply updated by USB, the same for master and slave module.
Under option "download" you can find "Firmware update". When this option is selected the user can select the new firmware file. This is a complete hex file provided by img Stage Line®, no adapts can and may be made by the user! After you selected the file you will enter the bootloader, the device will reconnect and shows version 99.99 in the status bar.
Now you need to select the file again to load the new firmware into the module.
After the file is selected the update begins.

DO NOT DISCONNECT THE MODULE AT THIS POINT!

When the progress bar is filled the update is completed.
On a CRC error the update is automatically restarted, after three errors the update is aborted.

Note that the new firmware does not have any filters installed, so the filters must be reloaded with the correct values.

Safety precautions

The AKB-400DSP is supplied with a switched mode power supply. The following chapter explains the safety precautions which have to be made for these kinds of products.

The AKB-400DSP operates at mains voltage and carries hazardous voltages at accessible parts. These parts may never be exposed to inadvertent touch. Observe extreme care during installation and never touch any part of the unit while it is connected to the mains. Disconnect the unit from the mains and allow all capacitors to discharge for 10 minutes before handling it.

This product has no serviceable parts other than the on-board fuse. Replace the fuse only with the same type and rating (250V T5AL).

All parts enclosed by the dotted line below carry hazardous voltages. This includes parts on the top and the bottom of the board. When the AKB-400DSP is mounted in a tight space there needs to be at least 6mm clearance or a layer of insulation with a minimum thickness of 0.5mm between the top of the transformer and the housing.

Standard the AKB-400DSP is supplied with 10mm spacers to mount the AKB-400DSP onto the chassis. This creates the mandatory 6mm clearance from the bottom side of the PCB to the chassis without the need for additional insulating material.

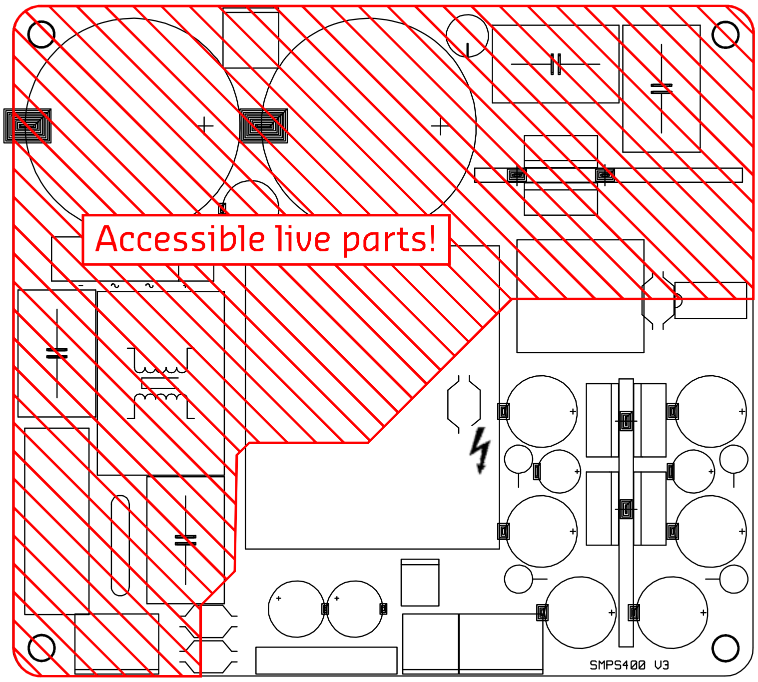

Figure 13

Instructions For Installation

Warning: To reduce the risk of fire or electric shock, do not expose this apparatus to rain or moisture.

Warning: Disconnect the unit from the mains and allow all capacitors to discharge for 10 minutes before handling it.

Warning: Please follow the mounting guidelines.
1. Do not build the module on a conductive surface.
2. Keep at least 60mm between the module front plate and the back of the enclosure, see Figure 14.
3. Do not lead any single insulated speaker-cable below the indicated line in Figure 14. With double insulated cable you are allowed to lead the cables below this line.

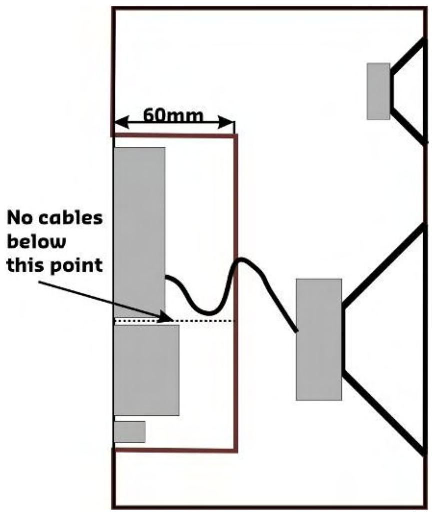

Figure 14

This symbol indicates the presence of hazardous voltages at accessible conductive terminals on the board. Parts that are not highlighted in red (picture above) may carry voltages in excess of 140VDC!

1. Read these instructions.
2. Keep these instructions.
3. Heed all warnings.
4. Follow all instructions.
5. Do not use this apparatus near water.
6. Protect the power cord from being walked on or pinched particularly at plugs, convenience receptacles, and the point where they exit from the application.
7. Only use attachments/accessories specified or approved by the manufacturer.
8. Unplug this apparatus during lightning storms or when unused for long periods of time.
9. Refer all servicing to qualified service personnel. Servicing is required when the apparatus has been damaged in any way, liquid has been spilled or objects have fallen into the apparatus, the apparatus has been exposed to rain or moisture, does not operate normally or has been dropped.
10. Don't run any cables across the top or the bottom of the AKB-400DSP
Apply fixtures to cables to ensure that this is not compromised.
11. Observe a minimum distance of 6mm maintain clearance with all possible conducting parts (housing etc.). All parts enclosed by the dotted line below carry hazardous voltages. This includes parts on the top and the bottom of the board.
12. Natural convection should not be impeded by covering the AKB-400DSP
(apart from the end applications housing).

Recommended Operating Conditions for the AKB-400DSP

Item	**Symbol**	**Min**	**Typ**	**Max**	**Unit**	**Notes**
High Line Input Voltage	V_B	180	230	264	Vac	
Low Line Input Voltage	$V_{B\,FP}$	90	115	132	Vac	
Line Input Frequency	f	47		63	Hz	

General Performance data for the AKB-400DSP

Item	**Symbol**	**Min**	**Typ**	**Max**	**Unit**	**Notes**
Max Output Power	P_R	600	-	-	W	See Note 1
Max Audio Output Power @ 20Hz into amplifier load	P_{RALF}	400	-	-	W	See Note 2
Switching frequency	F_{SW}	80	100	120	kHz	
Maximum power consumption	P_{max}			800	W	See Note 3

Note 1: Output Power delivered to a resistive dummy load (generally the only specification supplied by other AKB-400DSP manufacturers).
Note 2: An audio amplifier actually draws twice the RMS power from the power supply. At high frequencies the secondary storage output caps are capable to provide this power. At very low frequencies however the AKB-400DSP is responsible for delivering this peak power to the amplifier.
Note 3: Limited by over current protection.

Mains selection

The AKB-400DSP is factory set to operate at a mains voltage of 230Volt.
The user is able to change this setting to 115Volt. Instructions must be followed!
Before changing the mains settings disconnect the unit from the mains and allow all capacitors to discharge for 10 minutes before handling it.
Update the front plate with the updated information by checking the 115V or 230V setting.

J1: Mains Voltage Input Selection. Connector type: JST-B2P-VH

Pin	Function
1,2	Not Connected = 230Vac Mains; Connected = 115Vac Mains

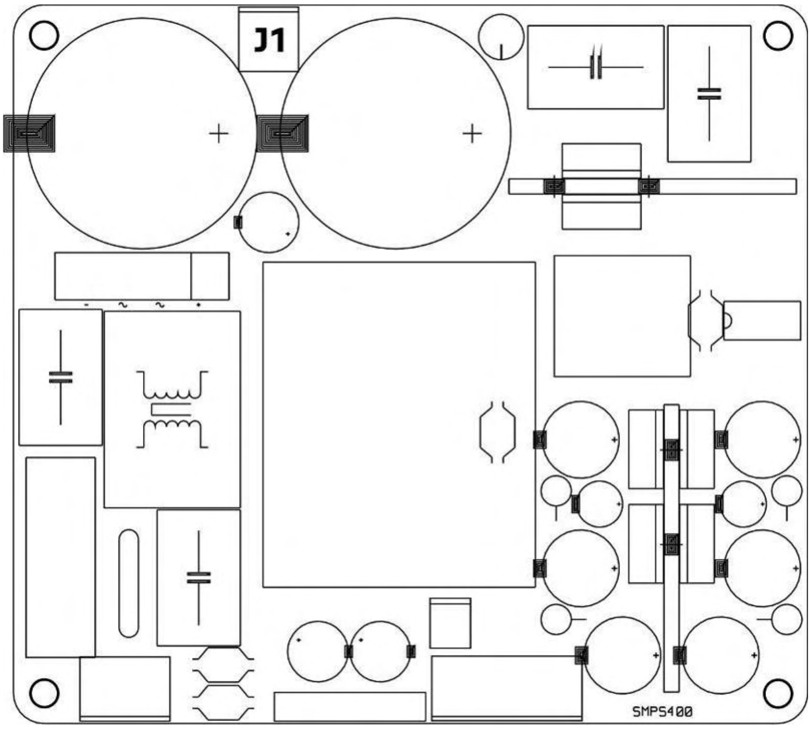

Figure 15

Technical data

Supply voltage	230Volt AC/50Hz +/- 10% (switch able to 115Volt 60Hz)
Dimensions	142mmx330mmx55mm (WxHxD)
Plate thickness	2,5mm
Weight	1,1 kg
Clearance	>9,5mm

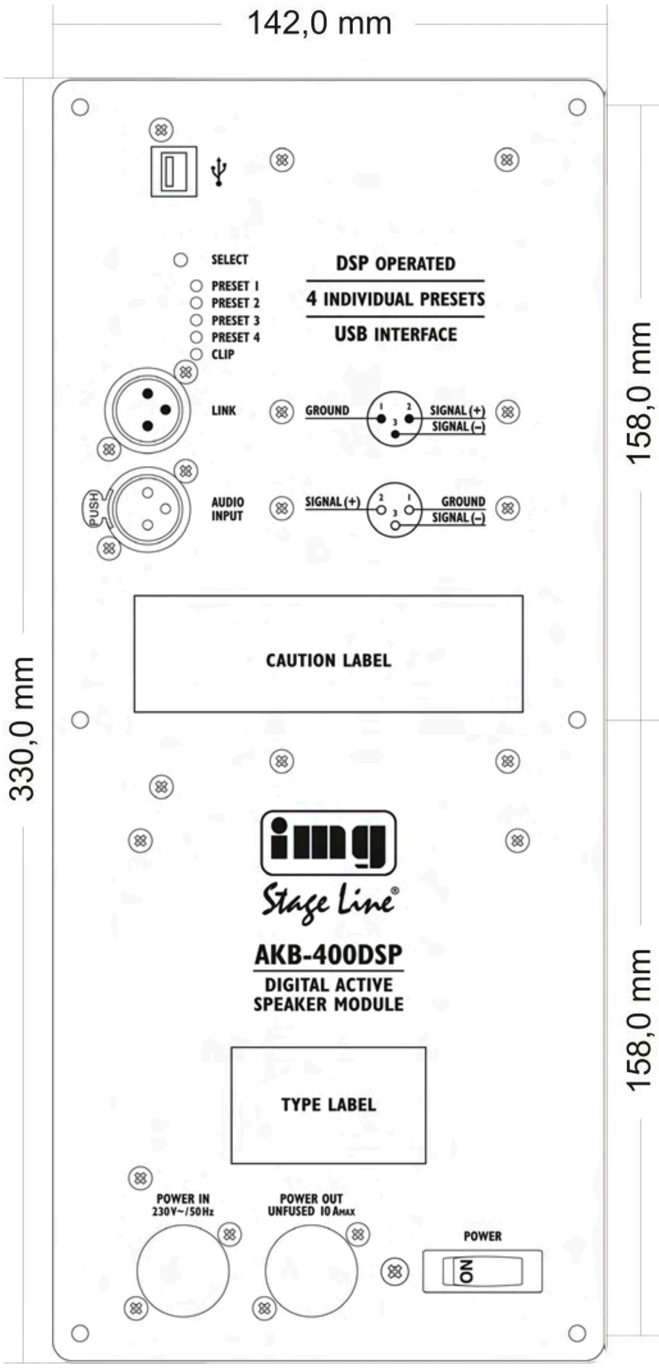

Figure 16: Plate amp 2xUcD400Watt

Revision	PCB Version	Description	Date
R0	V0.1	- Initial Draft.	21-04-2009
R1	V0.2	- Hard- and software part added	06-10-2009
R2	V1	- Pinout description added - Clip limiter added	23-06-2010
R3	V1	- Safety instructions added - Installation instructions added - Mains selection added	29-06-2010
R4	V2	- Explain button functions on digital version added - Temperature protection added	18-11-2010
R5	V3	- High channel gain change added	12-07-2011