



stage accompany

## DS50 USER MANUAL

# DIGITAL SERIES DS50

Digital Power Amplifier  
User Manual



*That's How Music Should Sound !*



## DS50 USER MANUAL

Published by:

**Stage Accompany**

Training & Documentation

Haven28

2984 BR Ridderkerk, The Netherlands

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First printing, May 2012

Printed in The Netherlands

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Stage Accompany reserves the right to alter specifications without prior notice.



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## 1 Quick Introduction to the DS50

Thank you for choosing a Stage Accompany Power Amplifier. Making this choice has guaranteed you many years of troublefree amplification.

If you have to start using the DS50 straight away and do not have the time to read the complete manual, make sure that you at least read the following:

- Ensure that you have a reliable, well earthed power source. The required current for the DS50 can be found in the table in paragraph 3.1.
- Connect the DS50 to your signal sources via the <INPUT> connectors. Connect this amplifier to other units via the <LINK> outputs using short signal leads. Connect your single signal source to the channel 1 input if you use the DS50 in dedicated active presets.
- Never make connections to the amplifier or to any preceding equipment with the amplifier switched on!
- Switch the DS50 on using the <POWER ON/OFF> switch.
- Select the desired preset and input sensitivity by means of the <SELECT ROTATOR >.

### Important

- For full-range operation at a 4 ohms load or higher please select preset 1 <2500 2500>  
For full-range operation at a load below 4 ohms please preset 2 <2000 2000>

The DS50 is now ready for use.



## 2 Introduction

The Stage Accompany DS50 is a dual channel, digital power amplifier and comprises the following components:

- Balanced input stages.
- Two digital amplifiers which deliver each 1600W into 8 ohms, 2500W into 4 ohms and 2000W into 2 ohms.
- 64-bit 96kHz high quality DSP controller
- Advanced protection circuits.

The audio inputs and link outputs are connected internally without electronics. This is done to prevent any loss of sound quality. With an input impedance of 25 kohms, it is possible to connect a maximum of 30 DS50 amplifiers to a 600 ohms source like a mixing console. If more than 30 units need to be connected to a single source, a separate signal driver must be used.

The DS50 is protected against the connection of speaker impedances less than 2 ohms or even short circuiting of the output. In this fault condition the power supply lines to the amplifier module will be interrupted.

The DS50 has a number of features, including:

- DSP processing.
- Two channel AES/EBU input.
- Dual auto-negotiation RJ45 ports.
- Switched mode power supply.
- Soft start to reduce power on inrush current.
- Class D operation.
- DDC (Dynamic Damping Control).
- Variable speed DC fan.
- Temperature, DC, HF and short circuit protection.



## DSP processing

The built in DSP module is a high quality four input, two output DSP processor with all the necessary features for a professional amplifier. The unit features two analogue and one two channel AES/EBU inputs with full configuration and real-time monitoring via PC controlled by Ethernet and USB interface.

## Two channel AES/EBU input

Two XLR-3 AES/EBU connectors for input and buffered output are mounted at the rear.

## Dual auto-negotiation RJ45 ports

An internal high performance 10/100Mbps Ethernet switch module provides a simple network connection to your workgroup or a server.

## Switched mode power supply

The amplifier stages are individual powered by a low weight high efficiency high power audio SMPS, optimized from the first phase of design to final impletation to realize the low EMI signature required of the most demanding audio applications.

## Soft Start

The soft start circuit is active when the DS50 is turned on. The mains inrush current, which normally is in excess of 100 A is limited to a safe value. However, take care not to switch on too many DS50 amplifiers at the same time, or you risk that the main fuses of your distribution may trip.

## Class D operation

The high performance class D output stages features a flat frequency response irrespective of load impedance, nearly frequency-independent distortion behaviour and very low radiated and conducted EMI.

## DDC

DDC (Dynamic Damping Control) is a special way of measuring the cone movement of a loudspeaker. The voltage induced in the voice coil is sensed and fed back to the amplifier's feedback circuit. Hence, a virtually infinite damping is achieved, which results in an exceptionally tight sound production.

## DC fan

The built in DC fan has a continuous variable speed. Its control circuit monitors the temperature of the output devices and the temperature of the power supply. The speed of the fan is calculated out of this information. This system ensures that the fan always runs at optimum speed and thus produces less noise.

## Temperature, DC, HF and Short Circuit Protection

The amplifier and the loudspeakers are protected against all possible damages.

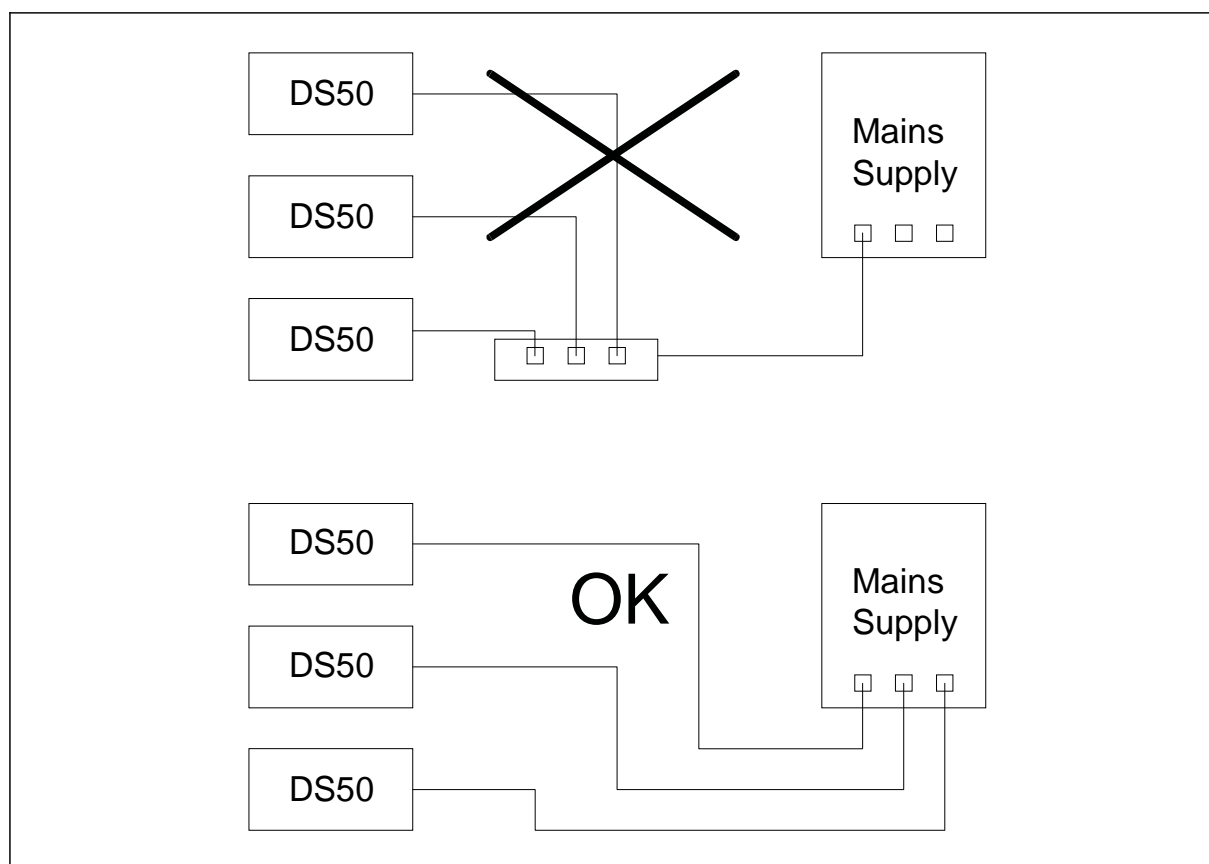


## 3 Connections

Only three connections need to be made: mains power, audio input signals, and loudspeaker leads. Do not make any connections to the amplifier or to any preceding equipment with the amplifier switched on!

### 3.1 Mains Power Connection

The DS50 is internally selectable for 110-120/220-240V 50/60Hz mains voltage. The mains voltage is configured at the factory according to standard used in the country of destination. The exact voltages are stated at the rear of the amplifier. Switching between mains voltages has to be done by an authorised Stage Accompany service centre. Connecting to the wrong voltage is dangerous and may damage the amplifier. Always ensure that you use a correctly grounded power supply. If more than one DS50 is to be used, it is advisable to connect each one separately to the nearest mains supply in stead of connecting several DS50s using an adapter block (see figure 3-1).



**Figure 3-1** DS50 connections to the mains supply.



This will prevent unnecessary power loss in the mains cables. If circumstances do not allow this, it is advisable to connect the adapter block directly to the power supply using a short cable and then connect the individual DS50s to it using longer leads. Although this method is not ideal, it does limit power loss as much as possible.

Ultimately, it is possible to place the adapter block near the DS50s, but if a great deal of power is required from the unit, the final current provided is far from optimum: loss of current leads to unrecoverable loss of output power and therefore sound pressure.

**Warning :**

Always disconnect the DS50 from the power supply before operating on the fuse holders! Replace a blown fuse only with a new one of the same type and value!

**Current Table**

Below, the current consumption of the DS50 is tabulated for two selectable mains voltages. The power consumption shown applies 1/8 of maximum output power pink noise in 2 ohms, which is representative for typical music program with occasional clipping. If your speaker system has an impedance of 4 or 8 ohms you may multiply the currents by 0.5 or 0.25.

No. DS50s	Mains Voltage (V)	
	110-120	220-240
Current Consumption (A)		
1	18	9
2	36	18
3	54	27
4	72	36
5	90	45
6	108	54
7	126	63
8	144	72
9	162	81

**Table 3-1** Current consumption as a function of the mains voltage at 1/8 of maximum output power with pink noise in 2 ohms.

The mains voltage may NOT deviate more than -20 % to +10 % of the nominal value. The proportional mains voltage cable loss may not exceed 10 %. The following formula may be used to calculate the maximum mains cable length in meters for 220-240 V models:

$$\text{Maximum Length (ML)} = \frac{G * 45}{N}$$

The two input variables are:

N - number of DS50s supplied through the cable

G - cable gauge in square millimeters

For 110-120 V models the following formula applies:

$$\text{Maximum Length (ML)} = \frac{G * 11.25}{N}$$

This means that the cable length is proportional to the cable gauge, while it is inversely proportional to the number of DS50s supplied through the cable.

On this and the next page, per mains voltage a table is shown with the maximum mains cable length in meters, given the cable gauge in square millimeters and the number of DS50s supplied through the cable.

110-120 V		Cable Gauge (mm <sup>2</sup> )					
		1.5	2.5	4.0	6.0	10.0	16.0
No. of DS50s	1	17	27	45	67	112	180
	2	9	14	22	34	57	90
	3	6	9	15	22	37	60
	4	4	7	12	17	28	45
	5	3	6	9	13	22	36
	6	3	4	7	12	19	30
	7	3	4	6	9	16	25
	8	2	3	6	7	13	20
	9	2	3	5	7	12	19
	10	2	3	5	7	12	16

**Table 3-2** Maximum cable length as a function of the cable gauge and the number of DS50s supplied through the cable

220-240 V		Cable Gauge (mm <sup>2</sup> )					
		1.5	2.5	4.0	6.0	10.0	16.0
No. of DS50s	1	45	112	180	270	450	720
	2	34	57	90	135	225	360
	3	22	37	60	80	150	240
	4	16	28	45	60	112	180
	5	13	22	36	54	90	144
	6	12	16	30	45	75	120
	7	9	16	25	39	64	103
	8	9	13	22	34	57	90
	9	7	12	19	30	49	79
	10	7	12	18	27	45	72

**Table 3-3** Maximum cable length as a function of the cable gauge and the number of DS50s supplied through the cable.

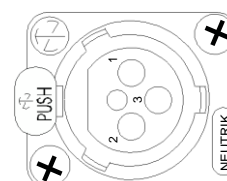
### 3.2 Audio Connections

The XLR connectors (analog audio, AES/EBU in and link) of the DS50 are wired as follows:

Pin 1 = shield

Pin 2 = normal phase (signal+ or “hot”)

Pin 3 = inverted phase (signal- or “cold”)



Always use high quality XLR connectors and shielded signal cables. Using the signal link connectors, up to 30 DS50s can be linked without a problem. If more than 30 DS50s need to be interconnected, a separate signal driver should be used.

The input stages are electronically balanced. To make optimum use of the balanced input stages even in unbalanced situations, use balanced signal cables and connect pin 3 to pin 1 at the output of the preceding equipment. If this is not possible, connect pin 3 to pin 1 at the input of the DS50.

The digital input of the DS50 offers two input channels on one electronically balanced XLR-connector and will accept AES/EBU and S/PDIF input signals of 16 to 24 bit 44.1, 48, 88.2 or 96kHz.

### 3.3 Loudspeaker Connections

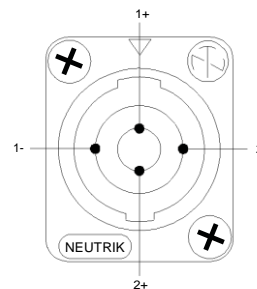
Due to the DDC feature, the DS50 has 4 output terminals per channel instead of 2. The speakon connector is wired as follows:

Pin 1+ = “hot” or Signal+

Pin 1- = “cold” or Signal-

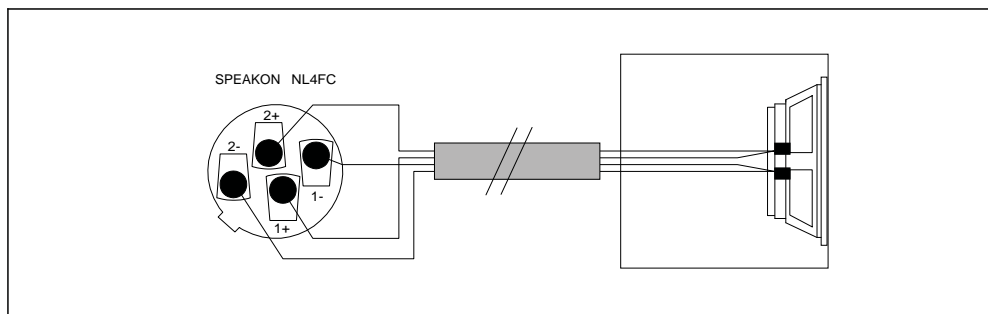
Pin 2+ = “DDC hot” or DDC+

Pin 2- = “DDC cold” or DDC-



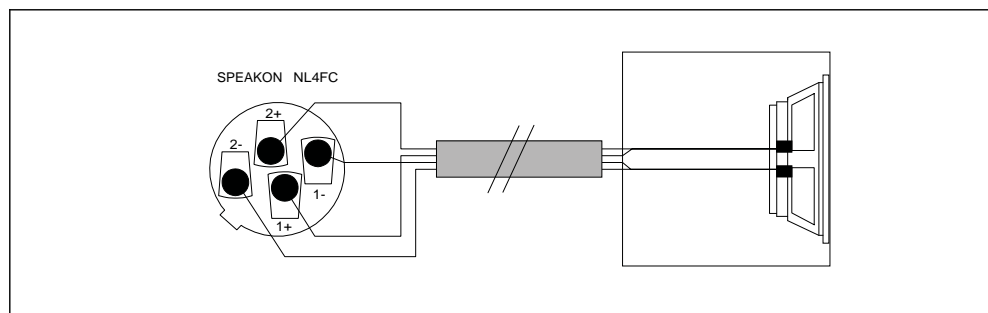
For optimum performance, the amplifier to loudspeaker connections should be made as shown in figure 3-2.

The signal+ and DDC+ (Dynamic Damping Control) terminals are linked together at the



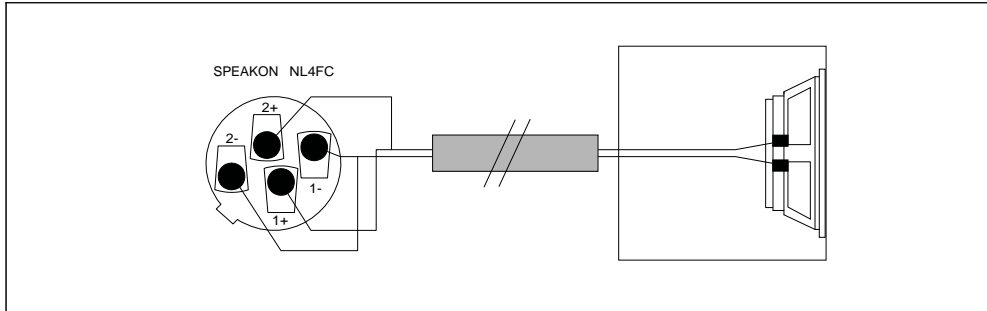
**Figure 3-2** DS50 4-terminal loudspeaker connection.

loudspeaker + terminal. The signal- and DDC- terminals are linked at the loudspeaker - terminal. A maximum damping is obtained with these connections. If your speaker system is not prepared for a four terminal connection, link the DDC and the signal leads at the input of the enclosure (see figure 3-3).



**Figure 3-3** DS50 2-terminal loudspeaker connection

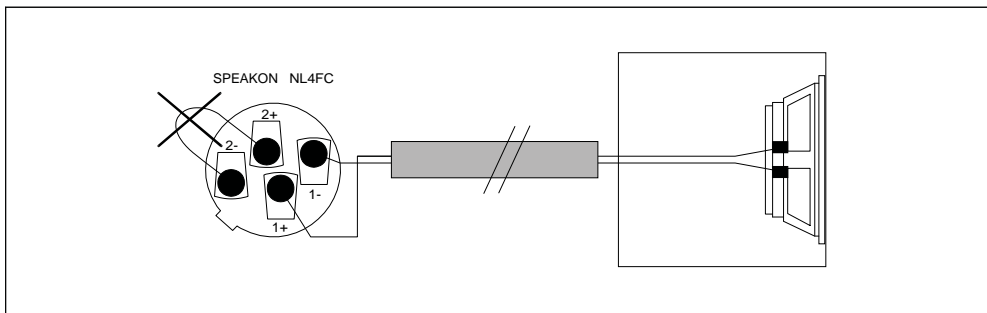
If you don't want to use the DDC feature at all, the DDC and the signal leads should be linked at the amplifier output in the speakon connector (see figure 3-4). The DDC terminals can also be left unconnected but then the performance will be less.



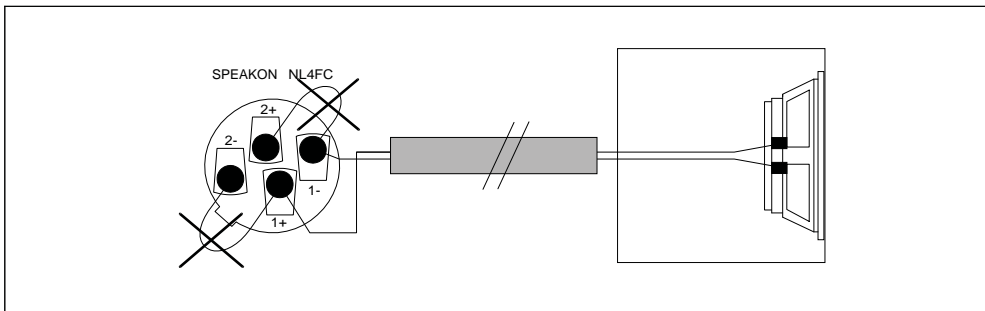
**Figure 3-4** DS50 loudspeaker connection without DDC

**Warning !** Never short-circuit the DDC terminals, and never connect the signal+ output terminal to the DDC- terminal or vice versa (see figure 3-5). Since you are disabling the amplifier's feedback network, signal gain will be almost infinite. The smallest input signal will result in full power output with the possibility of square waves damaging your loudspeakers!

Therefore, check your cables regularly. we also advice you to use special 4-wire loudspeaker cable: 2 x 3.0 mm<sup>2</sup> and 2 x 0.75 mm<sup>2</sup>.

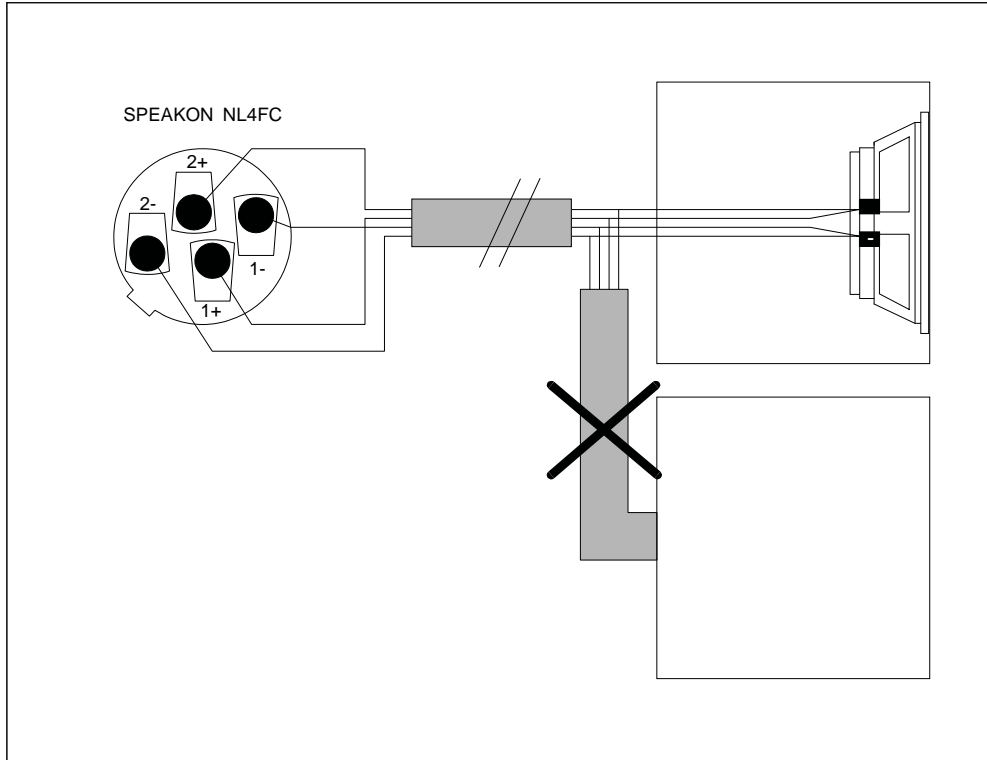


**Figure 3-5** Dangerous DS50 output connection



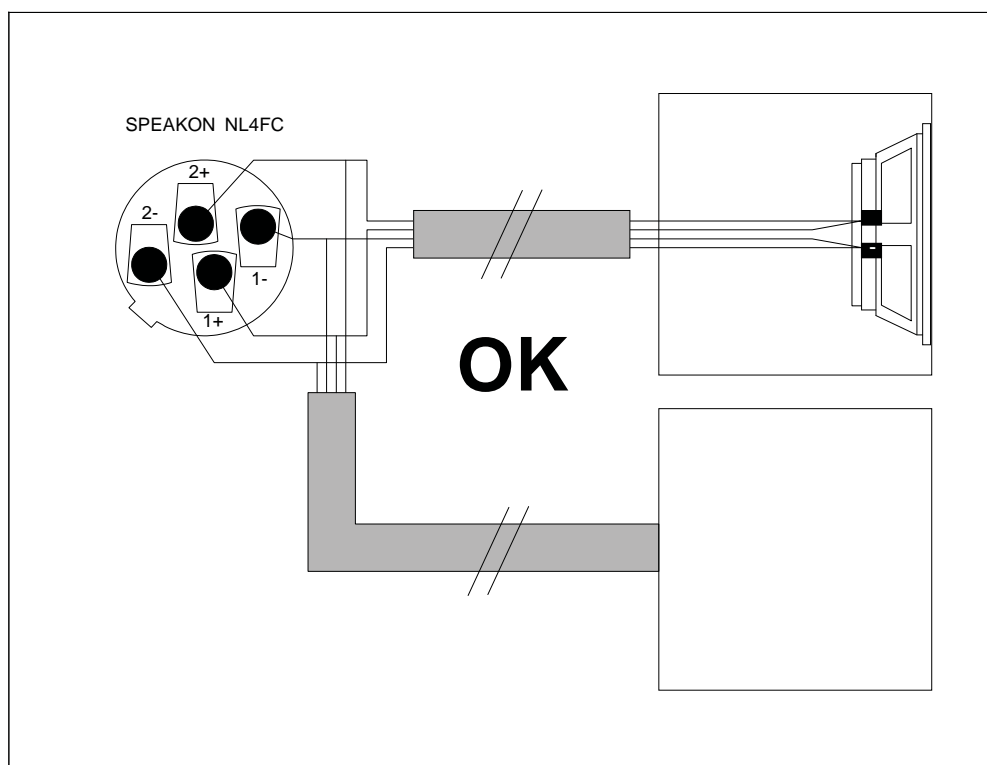
**Figure 3-6** Dangerous DS50 output connection

When more speakers are going to be connected to one amplifier, split the output and DDC leads as soon as possible behind the output terminals of the DS50 (see figure 3-7 on this and figure 3-8 on the next page).



**Figure 3-7** Try to avoid this way of connecting several enclosures to the DS50.





**Figure 3-8** Correct way of connecting several enclosures to the DS50

The amplifier can cope with any load of 2 ohms or higher. This means that you may connect a maximum of two 4 ohms, four 8 ohms or eight 16 ohms loudspeakers per channel in parallel. Linking the loudspeakers from one enclosure to another has two major disadvantages:

1. The loudspeaker cable from the amplifier to the first enclosure conducts the current for all enclosures, which means extra losses.
2. The DDC only works for the first enclosure.

Splitting the leads at the amplifier output terminals solves these problems (see figure 3-8). The DDC system corrects the performance loss due to connectors, loudspeaker cables and so on. However, it cannot prevent loss of power. So only use cables of minimal  $2 \times 1.5 \text{ mm}^2$  or more for the loudspeaker leads and  $2 \times 0.75 \text{ mm}^2$  for the DDC leads.



An indication of the proportional cable power loss (percentage of the amplifier's total output power) can be calculated using the following formula:

$$\text{Proportional Power Loss} = \frac{100}{1 + \frac{29.4 * G * R}{L}}$$

where the variables represent:

- L - cable length in meters
- G - cable gauge in square millimeters
- R - DC resistance of the loudspeaker(s) in ohms

This formula is based on the DC resistance in stead of the complex impedance of the loudspeaker(s). However, the approximation can be used to get a good impression of the loudspeaker cable power loss. Per loudspeaker impedance a table is shown in tables 3-4 to 3-7 with the proportional power loss, given the loudspeaker cable gauge in square millimeters and the cable length in meters.

16Ω		Cable Gauge (mm <sup>2</sup> )						
		1.5	2.5	3.0	4.0	6.0	10.0	16.0
Cable Length (m)	10	1.4	0.8	0.7	0.5	0.4	0.2	0.1
	20	2.8	1.7	1.4	1.1	0.7	0.4	0.3
	30	4.1	2.5	2.1	1.6	1.1	0.6	0.4
	40	5.4	3.3	2.8	2.1	1.4	0.8	0.5
	50	6.6	4.1	3.4	2.6	1.7	1.1	0.7
	100	12.4	7.8	6.6	5.0	3.4	2.1	1.3

**Table 3-4** Proportional power loss as a function of loudspeaker cable gauge and cable length (%).

		Cable Gauge (mm <sup>2</sup> )						
		1.5	2.5	3.0	4.0	6.0	10.0	16.0
<b>8Ω</b>								
Cable Length (m)	10	2.8	1.7	1.4	1.1	0.7	0.4	0.3
	20	5.4	3.3	2.8	2.1	1.4	0.8	0.5
	30	7.8	4.9	4.1	3.1	2.1	1.3	0.8
	40	10.2	6.4	5.4	4.1	2.8	1.7	1.1
	50	12.4	7.8	6.6	5.0	3.4	2.1	1.3
	100	22.1	14.5	12.4	9.6	6.6	4.1	2.6

**Table 3-5** Proportional power loss as a function of loudspeaker cable gauge and cable length (%).

		Cable Gauge (mm <sup>2</sup> )						
		1.5	2.5	3.0	4.0	6.0	10.0	16.0
<b>4Ω</b>								
Cable Length (m)	10	5.4	3.3	2.8	2.1	1.4	0.8	0.5
	20	10.2	6.4	5.4	4.1	2.8	1.7	1.1
	30	14.5	9.3	7.8	6.0	4.1	2.5	1.6
	40	18.5	12.0	10.2	7.8	5.4	3.3	2.1
	50	22.1	14.5	12.4	9.6	6.6	4.1	2.6
	100	36.2	25.4	22.1	17.5	12.4	7.8	5.0

**Table 3-6** Proportional power loss as a function of loudspeaker cable gauge and cable length (%).

		Cable Gauge (mm <sup>2</sup> )						
		1.5	2.5	3.0	4.0	6.0	10.0	16.0
<b>2Ω</b>								
Cable Length (m)	10	10.2	6.4	5.4	4.1	2.8	1.7	1.1
	20	18.5	12.0	10.2	7.8	5.4	3.3	2.1
	30	25.4	16.4	14.5	11.3	7.8	4.9	3.1
	40	31.2	21.4	18.5	14.5	10.2	6.4	4.1
	50	36.2	25.4	22.1	17.5	12.4	7.8	5.0
	100	53.1	40.5	36.2	29.8	22.1	14.5	9.6

**Table 3-7** Proportional power loss as a function of loudspeaker cable gauge and cable length (%).

## 4 Operation

The DSP (loudspeaker management) part of the DS50 offers a wide range of tools for sound system design and setup. These tools can make your system more efficient and better sounding, but to get the best possible sound it is important to use these tools properly. The following section explains the various functions of the DS50.

**Please note that if your DS50 amplifier is incorrectly setup you will not be using your system to its fullest potential and could be harming your speakers.**

**Always select the right Stage Accompany preset for your SA-system! (see 4.4)**

We encourage you to configure the device via PC or MAC program for maximum ease. Working from the device is possible though.

### 4.1 System check

Before plugging the DS50 in, always make sure that the power supply matches the product specification voltage. Install this device on a flat, stable surface, not bend or curved.

Do not supply power before all components of the system are set up and connected properly. After connection to the correct mains voltage, the DS50 can be switched on using the <POWER ON/OFF> switch.

After a start-up period of a few seconds the LCD display will show the last used preset.



### 4.2 Input setup

To enable operation of the control panel you first need to unlock the DS50. To unlock or lock the DS50 please refer to chapter 4.4.3 (page 4-9).

Press the MENU button to load the dedicated preset of the currently used loudspeaker system. Rotate the encoder (SELECT button) to search for the desired preset and push the encoder to select. Load preset by rotating the encoder to YES and pushing the encoder.

Now the right preset has been loaded you are free to make your own settings.

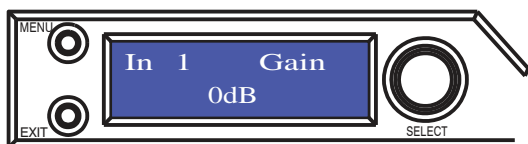
You are free to save your presets at preset number 7 or higher, preset nr. 1 through 6 are reserved for factory settings.

Press the SELECT button to select the input channel you wish to edit. To toggle through the different items, press the SELECT button and rotate the encoder.



### 4.2.1 Gain

Adjust the gain by dialing the rotary encoder. The gain is adjusted in steps of 0.25dB. Smaller steps (0.01dB) can be set via the PC or MAC interface.



### 4.2.2 Input Selection

The DS50 inputs are fixed. Input 1 is routed from the analog XLR-input of channel 1, channel 2 is routed from the analog XLR-input of channel 2, channel 3 and 4 are routed from the digital AES/EBU XLR-input on the backside of the DS50.

### 4.2.3 Delay

Set the delay time by turning the encoder. The delay time on the front will be shown in ms, by controlling the DS50 with StageControl the displayed unit can be changed between ms or s, mm or m, feet, inches, or mils.



### 4.2.4 Low Pass Filter

Adjust the Low Pass Filter frequency by turning the rotary encoder. You can switch the low pass filter off by turning the rotary encoder up (clockwise) until the frequency passes 20kHz. Press the rotary encoder to change the filter type. Select the type by turning the rotary encoder. You can choose from: Butterworth 6dB, Bessel 6dB, Butterworth 12dB, Bessel 12dB, Linkwitz Riley 12dB, Butterworth 18dB, Bessel 18dB, Butterworth 24dB, Bessel 24dB. Higher order filters may be set by adding filter sections in the PEQ blocks (see below). Enable or disable the Low Pass filter by rotating the encoder between on and off.



### 4.2.5 High Pass Filter

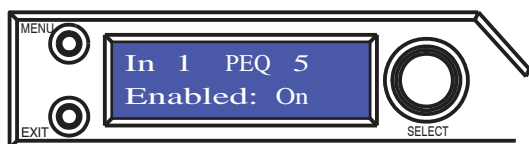
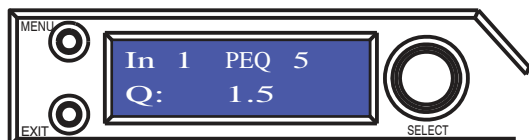
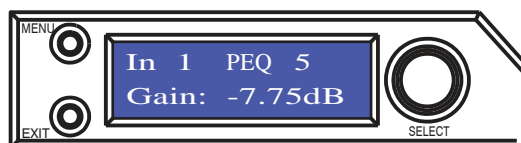
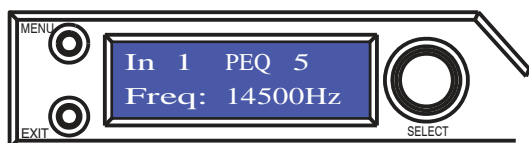
Adjust the High Pass Filter frequency by turning the rotary encoder. You can switch the high pass filter off by turning the rotary encoder down (counterclockwise) until the frequency passes 20Hz. Press the rotary encoder to change the filter type. Select the type by turning the rotary encoder. You can choose from: Butterworth 6dB, Bessel 6dB, Butterworth 12dB, Bessel 12dB, Linkwitz Riley 12dB, Butterworth 18dB, Bessel 18dB, Butterworth 24dB, Bessel 24dB. Higher order filters may be set by adding filter sections in the PEQ blocks (see below).

Enable or disable the High Pass filter by rotating the encoder between on and off.



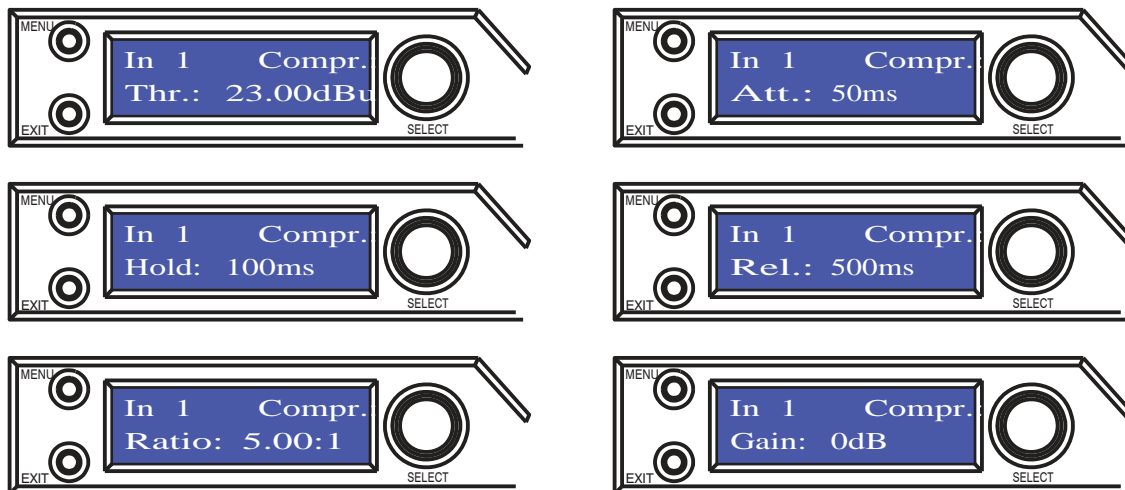
### 4.2.6 Parametric Equalizer (PEQ)

There are 10 bands of parametric equalization. Each band can be adjusted freely over the complete frequency range of 20Hz to 20kHz. Adjust the frequency by dialing the rotary encoder. Press the rotary encoder to select the parameters. The available parameters are: Frequency (20Hz to 20kHz), Gain (-12dB to +12dB), Q (0.2 to 25), Enabled (on or Off), Type (Bell, High Shelf, Low Shelf, Notch, All Pass, Band Pass, High Pass, Low Pass). For the Shelving filters, the Q value sets the steepness of the filter in dB/Oct.



### 4.2.7 Compressor

The Compressor is a true RMS compressor. Turn the rotary encoder to set the threshold. Press the rotary encoder to select the parameters. The available parameters are: Threshold, Attack, Hold, Release, Ratio and Makeup Gain.



### 4.2.8 Limiter

The limiter is a zero-attack peak limiter. Only the threshold and release can be set. Press the rotary encoder to select the parameters. The release value is displayed in dB per second.



### 4.2.9 Channel Link

By linking 2 channels, the settings are guaranteed to be identical for both channels, except for mixer and mute. Turn the encoder to set the Channel link on or off.

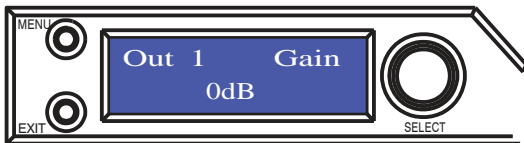


### 4.3 Output Setup

Press the SELECT button to select the output channel you wish to edit. To toggle through the different items, press the SELECT button and rotate the encoder.

#### 4.3.1 Gain

Adjust the gain by dialing the rotary encoder. The gain is adjusted in steps of 0.25dB. Smaller steps (0.01dB) can be set via the PC or MAC interface.



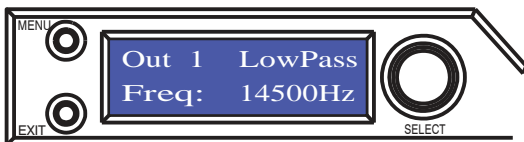
#### 4.3.2 Delay

Set the delay time by turning the encoder. The delay time on the front will be shown in ms, by controlling the DS50 with StageControl the displayed unit can be changed between ms or s, mm or m, feet, inches, or mils.



#### 4.3.3 Low Pass Filter

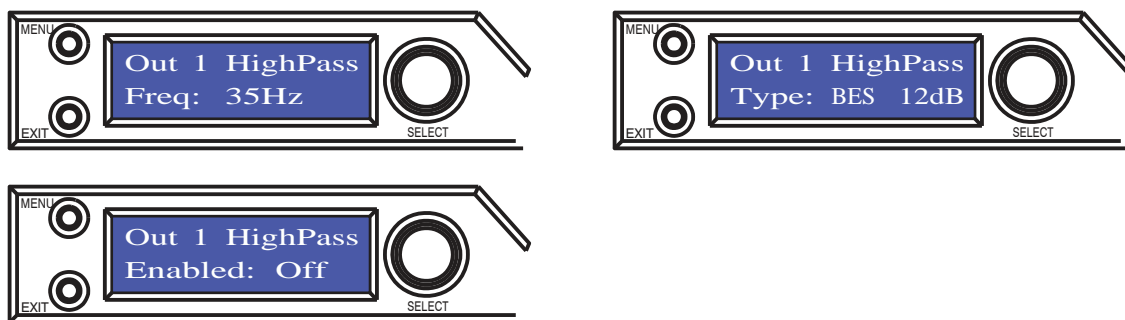
Adjust the Low Pass Filter frequency by turning the rotary encoder. You can switch the low pass filter off by turning the rotary encoder up (clockwise) until the frequency passes 20kHz. Press the rotary encoder to change the filter type. Selct the type by turning the rotary encoder. You can choose from: Butterworth 6dB, Bessel 6dB, Butterworth 12dB, Bessel 12dB, Linkwitz Riley12dB, Butterworth 18dB, Bessel 18dB, Butterworth 24dB, Bessel 24dB. Higher order filters may be set by adding filter sections in the PEQ blocks (see below). Enable or disable the Low Pass filter by rotating the encoder between on and off.



### 4.3.4 High Pass Filter

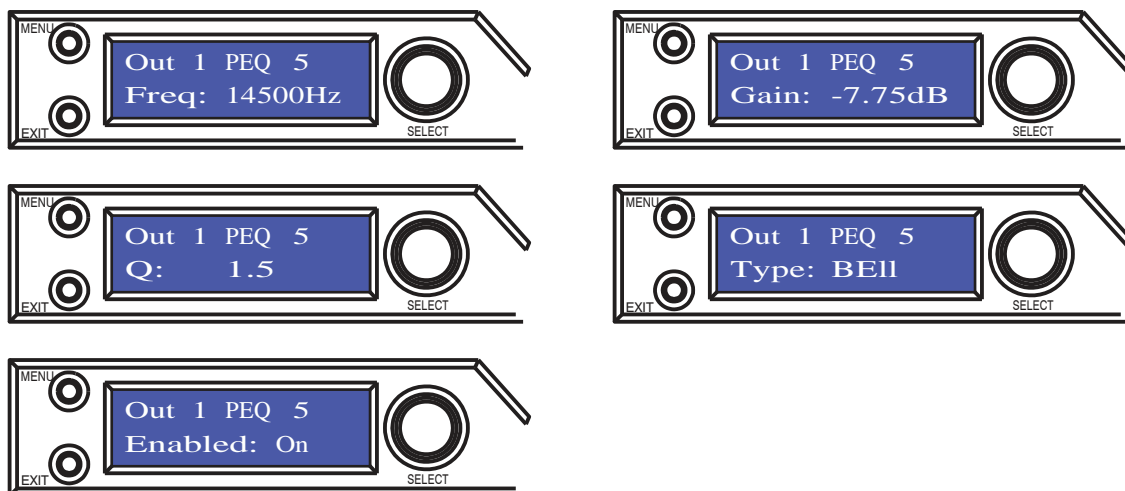
Adjust the High Pass Filter frequency by turning the rotary encoder. You can switch the high pass filter off by turning the rotary encoder down (counterclockwise) until the frequency passes 20Hz. Press the rotary encoder to change the filter type. Select the type by turning the rotary encoder. You can choose from: Butterworth 6dB, Bessel 6dB, Butterworth 12dB, Bessel 12dB, Linkwitz Riley12dB, Butterworth 18dB, Bessel 18dB, Butterworth 24dB, Bessel 24dB. Higher order filters may be set by adding filter sections in the PEQ blocks (see below).

Enable or disable the High Pass filter by rotating the encoder between on and off.



### 4.3.5 Parametric Equalizer (PEQ)

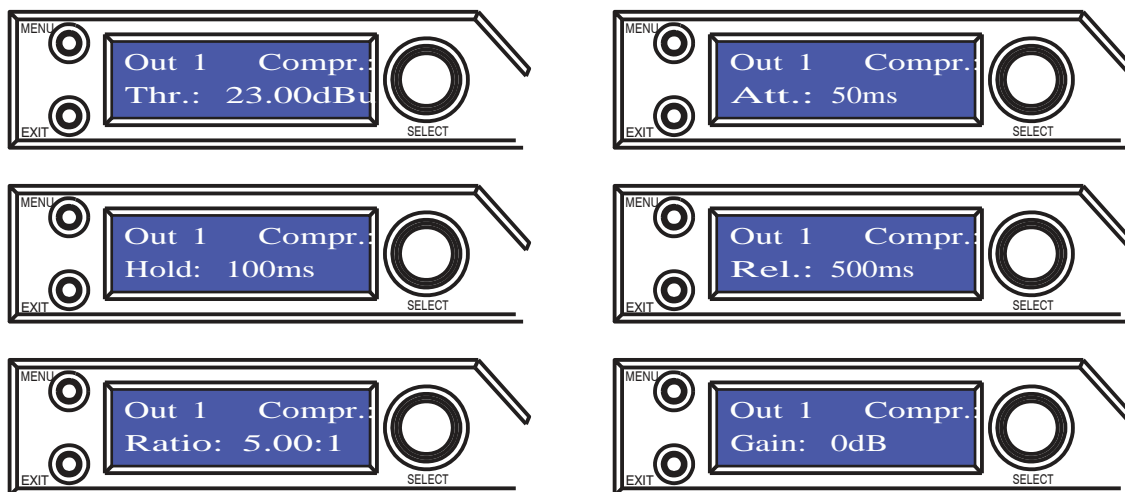
There are 10 bands of parametric equalization. Each band can be adjusted freely over the complete frequency range of 20Hz to 20kHz. Adjust the frequency by dialing the rotary encoder. Press the rotary encoder to select the parameters. The available parameters are: Frequency (20Hz to 20kHz), Gain (-12dB to +12dB), Q (0.2 to 25), Enabled (on or Off), Type (Bell, High Shelf, Low Shelf, Notch, All Pass, Band Pass, High Pass, Low Pass). For the Shelving filters, the Q value sets the steepness of the filter in dB/Oct.





### 4.3.6 Compressor

The Compressor is a true RMS compressor. Turn the rotary encoder to set the threshold. Press the rotary encoder to select the parameters. The available parameters are: Threshold, Attack, Hold, Release, Ratio and Makeup Gain.



### 4.3.7 Limiter

The limiter is a zero-attack peak limiter. Only the threshold and release can be set. Press the rotary encoder to select the parameters. The release value is displayed in dB per second.



### 4.3.8 Phase Inversion

Turn the rotary encoder to switch phase inversion on or off.



### 4.3.9 Channel Link

By linking 2 channels, the settings are guaranteed to be identical for both channels, except for mixer and mute. Turn the encoder to set the Channel link on or off.



## 4.4 System Menu

Push the Menu button to enter the system menu. Push the Menu button again to toggle through the menu items. The available menu items are:

### 4.4.1 Load Preset

Turn the rotary encoder to select the preset you wish to load. Push the rotary encoder to select it, then turn it again to select "YES" and confirm again by pushing the rotary encoder knob. The preset is loaded and all settings are applied immediately. All settings that were in the unit prior to loading the preset will be erased. **This action can not be undone!!** The presets contain all filter, dynamics, gain settings etc; in other words, everything that makes out a loudspeaker configuration. Presets do NOT contain the name of the unit, network configuration, automatic standby delay, user access rights and passwords.

### 4.4.2 Save Preset

Turn the rotary encoder to select the location of the preset. If that location is not empty, the preset in that location will be overwritten. **This action can not be undone!!** The system will ask you to give a name to the preset name. Select a character in the cursor position by turning the rotary encoder; pushing the rotary encoder confirms the character selected and moves the cursor to the next character to edit. Pushing the exit/ESC button erases the last confirmed character. Once the name is set up, push the menu button again. If you want to continue storing the preset, select Yes by turning the rotary encoder and confirm by pushing it. To cancel, push the exit button. Now your preset is saved in the selected location. The presets contain all filter, dynamics, gain settings etc; in other words, everything that makes out a loudspeaker configuration. Presets do NOT contain the name of the unit, network configuration, automatic standby delay, user access rights and passwords. You can save your presets at preset nr. 7 or higher, nr. 1 trough 6 are reserved for factory settings.



### 4.4.3 Access Level

The DS50 has the option of locking away the front panel controls to avoid tampering of settings by unauthorized persons. When switching on the DS50 for the first time all the functions are locked.

To enable operation of the control panel you first need to unlock the DS50. To unlock the DS50 control panel Press the MENU button, rotate the SELECT button (encoder) counterclockwise until the display shows Unlocked. Press the SELECT button and the system will ask you to enter a password. Select a character in the cursor position by the rotary encoder: pushing the rotary encoder confirms the character selected and moves the cursor to the next character to edit. Pushing the exit/ESC button erases the last confirmed character. The Password is automatically checked after each entered character, and will exit the system menu when the right password has been entered. The default password is "SA".

To lock the DS50 control panel Press the MENU button for 3 times, and the display will show Acces level: Unlocked. Rotate the SELECT button (encoder) clockwise to Locked and press the SELECT button. The system will show you the last used password. You are free to use this or change it to another password by rotating and pressing the SELECT button. Press MENU twice to confirm and set the password.

**ATTENTION:** Make sure to remember the password! When the unit is locked and you forgot the password, it is not possible to unlock it without contacting your local service representative.

### 4.4.4 Version Information


Press the MENU button four times to enter the version information menu. By pushing the rotary encoder, the display toggles through some version information about the unit as well as some parameters like IP address, MAC address etc.

## 4.5 Indicators

Each channel of the amplifier is equipped with the following indicators:

- Power Amp On: These LEDs will light when the power amp of the related channel is switched on and the electronic circuits are stable. In case of an error in power supply or main stage the LED will blink or swithes off. In that case the amplifier is damaged and should be repaired by an authorised Stage Accompany service center.
- Signal present: these LEDs will light when a input level of more than 40 dB under the level that would cause maximum power is detected. The signal present LEDs work dependently of the settings of in- and output of the DSP settings. These LED's indicates the present of input signal at the power amp stages.
- Clip: this LED will light when the output level of the amplifier has reached its limit. The indication is correct for every connected impedance. When the clip LEDs flash frequently, distortion is to be expected.

## 5 StageControl



**Getting started**

**Supported operating systems**

**Installation instructions**

**Start communication**

**Quick configuration**

### Getting started

This document describes how to install and use the StageControl software on your computer for the Stage Accompany Digital Series amps. Installing StageControl gives you the possibility to remotely setup, monitor and update (firmware) your SA amp(s) from the Digital Series through USB or internet.

### Supported operating systems

The following operating systems are supported:

- Microsoft Windows XP with Service Pack 2 (32-bit and 64-bit)
- Microsoft Windows Vista (32-bit and 64-bit)
- Microsoft Windows 7 (32-bit and 64-bit)
- Microsoft Windows 8 (32-bit and 64-bit)
- MAC OS X 10.5.8 or higher

### Installation instructions:

#### Microsoft Windows:

Download the installation package from our website and open it.

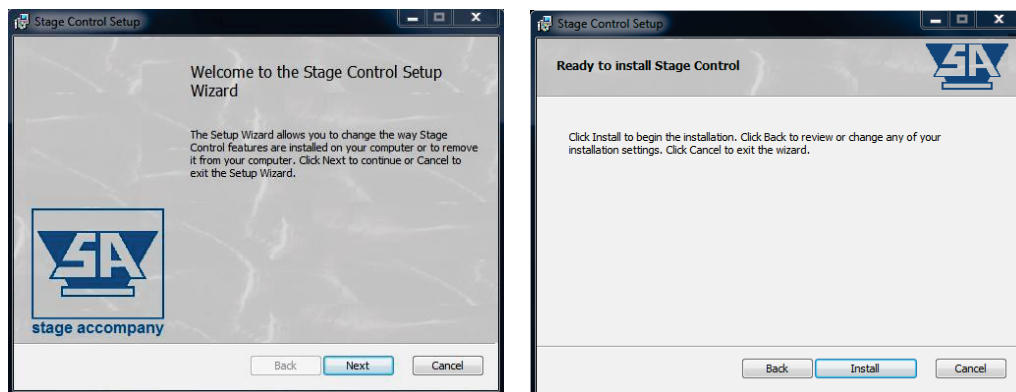
Microsoft Windows 32-bit:

[http://www.stageaccompany.com/support/downloadnew/stage\\_control\\_32bit.msi](http://www.stageaccompany.com/support/downloadnew/stage_control_32bit.msi)

Microsoft Windows 64-bit:

[http://www.stageaccompany.com/support/downloadnew/stage\\_control\\_64bit.msi](http://www.stageaccompany.com/support/downloadnew/stage_control_64bit.msi)

The installer will guide you through the installation procedure. It is not necessary to remove any older version; it will be removed automatically, unless the previous version is older than 1.9.93. In this case, you must uninstall the previous version manually.



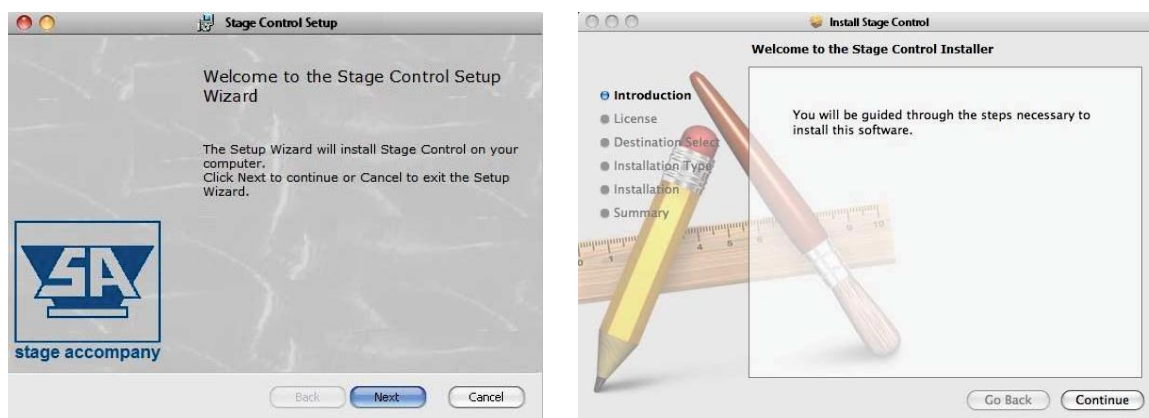
Apart from the main application, also the binary firmware files for all supported SA Digital Series amps will be installed, as well as factory preset files (if applicable). After installation start the software by double-clicking the desktop icon, or by selecting: *Start – Programs – Stage Accompany – StageControl.*

**MAC OS X:**

Download the installation package from our website and open it.

[http://www.stageaccompany.com/support/downloadnew/stage\\_control.mpkg.zip](http://www.stageaccompany.com/support/downloadnew/stage_control.mpkg.zip)

The installer will guide you through the installation procedure. It is not necessary to remove any older version; it will be removed automatically.



After installation start the application by double-clicking *StageControl.app* in your *Applications* folder.

**Note:**

It's not possible to upgrade the firmware of your SA Digital Series amp(s) if you are using MAC OS X, this is by design. Upgrading the Digital Series amps hardware can only be done from a PC with the Windows version of StageControl.

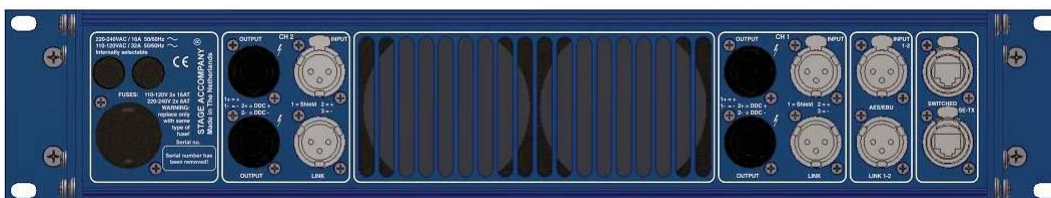
## Start communication

There are two ways to remotely control and monitor your SA Digital Series amp from your PC;

1. through USB, using the Neutrik USB connector located at the right on the front of the SA Digital Series amp.

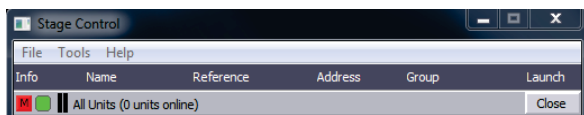


2. through Ethernet (LAN), using one of the two Neutrik etherCON connectors located right at the back of the SA Digital Series amp.



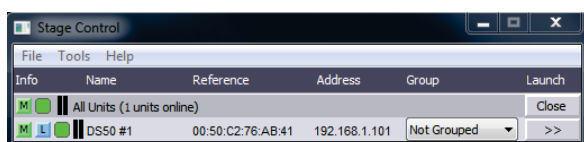
To configure or monitor your SA Digital Series amp(s) you do the following;  
Double-click the *Stage Accompany StageControl* configuration icon on your desktop to start the software.

If no SA Digital Series amps are discovered during start-up the screen looks like the figure below.



To let the program discover your SA Digital Series amp(s) you do the following;  
Connect your SA Digital Series amp(s) to your PC using the USB port on the front or, if your PC is connected to a LAN, use one of the two switched 10/100Base-TX Ethernet ports on the back of the amp.

Because the StageControl software is constantly polling for new devices newly connected devices will automatically be discovered and added to the device list (see figure below).



After discovering each device can be given a user defined name and can be placed in an user defined group.

### Note:

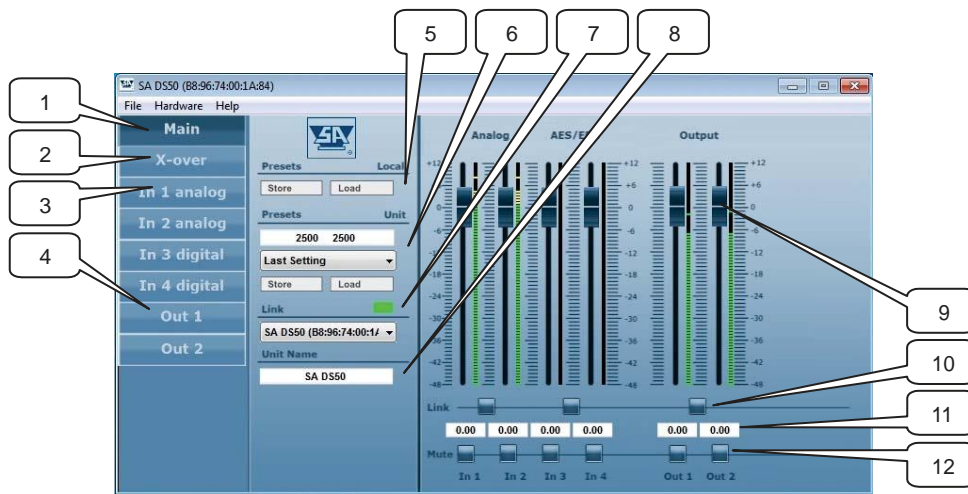
The amps IP-address can be manually entered (through the configuration screen; *Hardware – Configure – Network Settings*) or automatically provided by a DHCP server. If no DHCP server is available on the network the Digital Series can also communicate through an APIPA address. (Automatic Private IP Addressing) The software will only discover devices which are connected to the same network and in the same subnet.



### Quick configuration

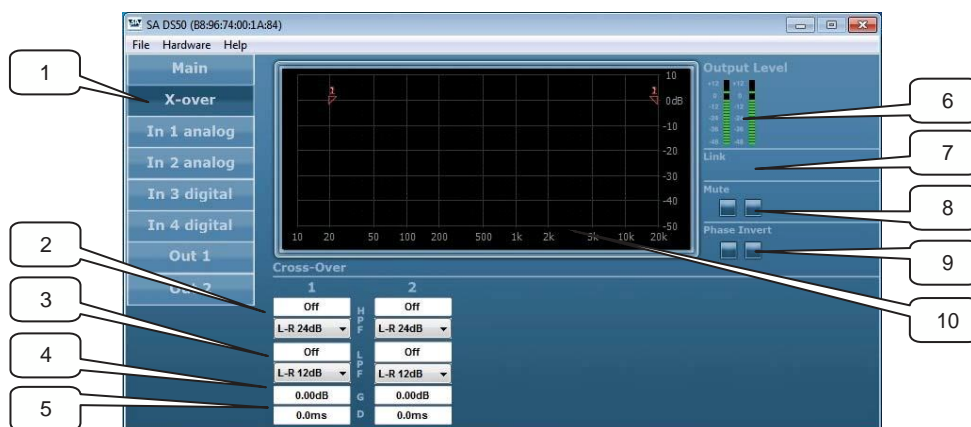
If you have to start configuring your SA Digital Series amp(s) straight away and do not have time to read the complete manual, make sure that you at least read the following; When you have connected the amp(s) to your network and successfully discovered the amp(s) via the software, you can open the configuration on the amp by *Launch* ( >> ) the appropriate amp on your device list. Your opening screen will look similar to the screen below:

### Main Window



1. Main window tab: pressing this tab will show the above window.
2. X-over tab: clicking on this tab will open the cross-over window.
3. Input tabs: clicking on these tabs will open the corresponding input window.
4. Output tabs: clicking on these tabs will open the corresponding output window.
5. Preset: choose load or store to manage presets.
6. Choose a preset or name a preset in these fields.
7. In the link fields you can check the connection to your amp and the device name.
8. You can rename the linked amp by entering a new name in this field.
9. Input/output faders: you can set the input and output levels by sliding these faders with your pointing device.
10. You can link adjacent inputs/outputs by engaging this radio button. When engaged the faders can then be moved simultaneously.
11. The corresponding levels can be read or entered directly in these fields.
12. Finally each in/out can be muted here individually.

## X-over Window



1. X-over tab: clicking on this tab will open the cross-over window.
2. High Pass Filter slope and frequency fields: you can enter the desired high pass filter frequency here and choose the slope. The filter types that can be chosen are Butterworth, Bessel and Linkwitz Riley. The filter slopes that can be chosen are 6, 12, 18 and 24 dB/octave.
3. Low Pass Filter slope and frequency fields: you can enter the desired low pass filter frequency here and choose the slope.
4. Gain: the output level of each output can be set here. This way you can adjust the output to match the driver/enclosure sensitivity and amplifier gain.
5. Delay: each output has its own delay setting to adjust the relative driver/enclosure positions.
6. Output level meters: you can monitor the outputs here.
7. Link buttons and indicators: clicking on these buttons will link channels, all settings of both channels are then synchronized. When two adjacent channels are linked, the buttons lights up.
8. Mute buttons: clicking these buttons mute the corresponding channel. When a channel is muted the corresponding button lights up.
9. Phase invert: to compensate for phase behavior of the chosen filter slope, loudspeaker or loudspeaker positioning, the phase of each output can be inverted individually.
10. Frequency response window: this window gives you a graphical representation of the cross-over settings.

### Note:

A full discussion on the merits of various filter slopes goes beyond the scope of this manual, but in a nutshell it is as follows:

Butterworth slopes have the highest Q or sharpest knee without an overshoot. They are generally advisable for low cut of stage monitors etc..

Bessel slopes are chosen because of their phase characteristics.

Linkwitz-Riley crossover are chosen for the flat energy balance between drivers.

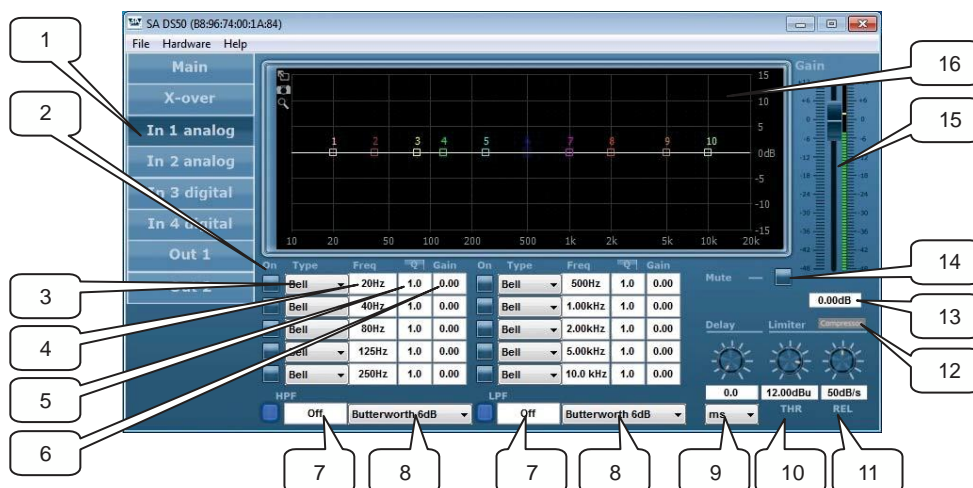
Linkwitz-Riley crossover are chosen for the flat energy balance between drivers.

Steeper slopes have smaller crossover regions, smaller crossover regions have less directivity problems, less lobing. But steeper slopes have more phase variation.

Butterworth, Bessel, Linkwitz-Riley have each their own response characteristics and 6 dB to 24 dB slopes. You can use them depending on what you value in your system, and what aspect is most important to you.



## Input Window



1. Input tab: clicking on this tab will open the input processing window
2. PEQ on/off button: clicking these buttons enable the corresponding parametric equalizer (PEQ). When a PEQ is enabled the corresponding button lights up.
3. EQ type: here you can choose Bell, Notch, high shelving, low shelving All Pass, Band Pass, High Pass or Low Pass equalizers.
4. Center or shelving frequency: here you can set the equalizer frequency.
5. Q-factor: the Q factor is a value of how sharp the equalizer-curve is. A high value means a sharp curved EQ. This is either a narrow band or a sharp increase or decrease with shelving equalizers. Use your ears to judge and check with the frequency response windows of the software.
6. Gain: set the amount of boost or cut in this field.
7. High pass and low pass filter: also known as low-cut (sub-sonic) and high-cut respectively, these filters remove frequencies below or above the chosen frequency.
8. Slope: this determines the steepness of the filter. The same types and slopes are available at the cross-over settings.
9. Delay: the maximum delay permitted is 1000ms, in steps of 0.1ms. It can be displayed in feet, inch, m, mil, mm, ms and s.
10. Limiter threshold: this is the level above which the limiter becomes active. Set this level to the maximum desired output.
11. Limiter release: this is the setting to determine how fast the limiter function (gain reduction) is returned to normal (no limiting). This value is time-dependent.
12. Input Gain: this is the field to set the absolute value of the gain setting.
13. Compressor tab: clicking on this tab will open the compressor processing window..
14. Mute button: input channel mute button.
15. Input Gain: this is the input channel gain setting fader and input level meter.
16. Frequency response window: this window gives a graphical representation of the cross-settings.

### Note:

When using EQ; use narrow EQ mostly to cut and not to boost.

Use narrow EQ (high Q) for correction and low Q (wide EQ) for getting the desired sound.

## Output Window

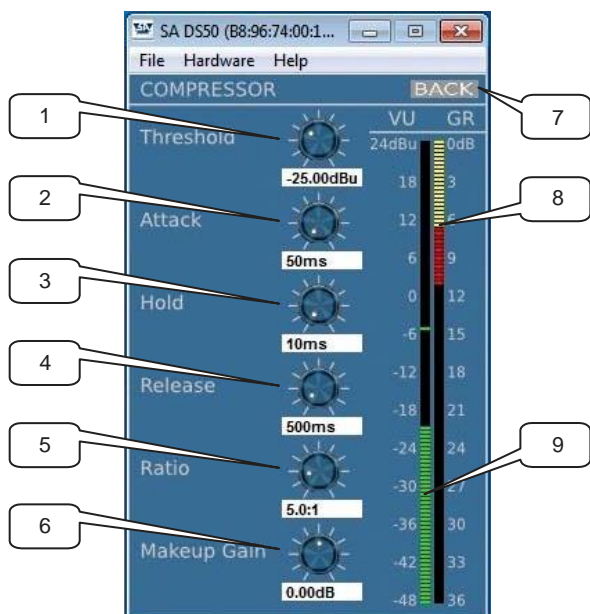


1. Output tab: clicking on this tab will open the output processing window.
2. Mixer: the ratio of each available input that is routed to the output.
3. Invert button: clicking this buttons inverts the signal to the output.
4. Compressor tab: clicking on this tab will open the compressor processing window.

The output window gives access to an independent set of EQ's, high pass / low pass filters, delay, compressor and output limiter.

The buttons and functions of the limiter and EQ, HPF and LPF are the same as at the input menu. Having independent input and output processing is very powerful. It gives the user the the output EQ to the used driver/enclosure, and the input EQ of the source. Different sources may need different equalizer settings.

## Compressor Window



1. Threshold: Used to set the minimum signal level at which the compressor circuit begins to function.
2. Attack: Adjust the amount of time the compressor takes to reach full gain reduction.
3. Hold: Adjust the minimum time that the compressor will process the signal for.
4. Release: Adjust the length of time the compressor takes to return the signal to it's original level.
5. Ratio: Controls the amount Gain reduction in proportion to the amount of signal over the selected threshold level.
6. Makeup Gain: Controls the amount of output level.
7. Back: Clicking on this tab returns you to the selected output channel.
8. Output level meter: Shows the output level.
9. Gain reduction meter: Indicates the amount of attenuation.

The Compressor window gives access to several ways of protecting your loudspeaker system depending on type and power handling.

### Notes:

- The StageControl configuration software can be used in offline mode (Demo mode) to learn and prepare configurations, without the need to be connected to a Digital Series amp through USB or Ethernet. To use StageControl in demo-mode go to; *Tools-Enter Demo Mode* (password: Demo).
- An .iso CD image is also available at our website, the CD image contains all latest versions (Windows and Mac) of StageControl, as well supporting documentation and an *Autostart* feature with automatic 32/64-bit system detection.  
[http://www.stageaccompany.com/support/downloadnew/stage\\_control.iso](http://www.stageaccompany.com/support/downloadnew/stage_control.iso)
- Please visit our website for more detailed information about the Stage Accompany Digital Series amplifiers.  
<http://www.stageaccompany.com/products/electronics/electronics.html>

## 6 Recommendations for Optimum Use

- Ensure that the quality of the mains supply is good enough and that it can supply the required peak currents. In addition, good earthing is necessary to prevent noise, hum and safety problems.
- Be sure that the DS50 is built into a space with sufficient circulation of fresh air. A temperature that is constantly too high leads to switching on and off of the thermal protection and considerably reducing the lifetime of the amplifier.
- When built into a flightcase, ensure yourself that there are sufficient possibilities for the heated air to leave the flightcase. When more DS50s are built into one flightcase, install an extra fan in the case. Air flow of the DS50 goes from back to front.
- Do not make connections to the amplifier or to any proceeding equipment with the amplifier switched on
- A faulty fuse must ALWAYS be replaced by a new one of the same type and value! A different fuse can cause permanent damage to the amplifier. All internal fuses are of an automatic type, so they do not need to be replaced. Always disconnect the DS50 from the power supply before operating on the fuse holder! When a replaced fuse immediately blows again, the amplifier is defect and should be repaired by an authorised Stage Accompany service centre. Do not try to replace the fuse again because you may risk further damage to the amplifier.
- Repairs to the DS50 should only be carried out by any autorised Stage Accompany service centre. Never dismantle the DS50 yourself, you may risk damaging vital components: moreover, the guarantee becomes irrevocably invalidated.
- Read this manual carefully. Tryout various functions described in this manual. In this way you will soon become familiar with the many possibilities that the DS50 offers. It is just as important that your colleagues who will use the DS50 also read the manual! With a good knowledge of the working of the DS50, it is possible to fully utilize its superior sound quality in every situation.

**GOOD LUCK!**



## 7 Technical Specifications

Efficiency	: 92% @ full power
Max input level	: +23dBu Input
impedance	: 25k $\Omega$ Frequency
range (@ 100W into 8 $\Omega$ )	: 10Hz – 40kHz Gain : 34dB
Channel separation	: > 90dB @ 1kHz
Total harmonic distortion	: 0.05% 20Hz – 20kHz, Pout < Pmax/2 0.05% 20Hz – 20kHz, Pout = 1W
Output noise	: 30uV unweighted, 20Hz – 20kHz
Damping factor	: 10.000 @ 1kHz, 8 $\Omega$
Output power @ 1kHz, < 1% THD	: 2 x 1600W @ 8 $\Omega$ 2 x 2500W @ 4 $\Omega$ 2 x 2000W @ 2 $\Omega$
Amplifier section	: flat load independent frequency response
Audio and data connections	: 1 x Neutrik USB connector (located at the front) 2 x XLR-3 female connector for input signal 2 x XLR-3 male connector for signal link 4 x SpeakON NL4 for output signal 1 x 32A PowerCON connector for mains power 2 x EtherCON connector (switched) 2 x XLR - 3 AES/EBU connector
DSP section	: up to 10 parametric filters on in- and output Butterworth, Bessel, L-R filters up to 24dB/oct. limiter on each input and each output
Mains supply voltage	: stated at the rear of the amplifier
Power consumption	: 3500VA
Housing	: 19" Rack Mount, 2HU high, 415mm deep
Physical dimensions (h x w x d)	: 88 x 482 x 415 mm. (without connectors and handles) (3.47 x 19.0 x 16.3 in)
Weight	: 12.5kg (27.3lb)
Warranty	: 3 years

## 8 Warranty

Stage Accompany warrants to the initial purchaser of each new Stage Accompany product, from the date of purchase by the original purchaser until the end of the warranty period that the product is free of defects in materials or workmanship.

The warranty period on all Stage Accompany products is five years from the date of the first consumer purchase, with the exception of:

- all electrical products: three years from the date of the first consumer purchase;
- cone assemblies in the loudspeaker and diaphragms of the Ribbon Compact Drivers: one year from the date of the first consumer purchase;
- movable parts, such as castors, locks, handles, hinges, fans etc. one year from the date of the first consumer purchase;

To validate warranty, the purchaser must always keep the original receipt of purchase to establish the date of purchase.

Appeal on warranty will be voided in case:

- of defects caused by influence from the outside, accident, misuse, neglect or influence of water;
- the serial number on the product has been defaced, altered or removed;
- of damage due to shipment;
- of damage resulting from neglecting instructions listed in the user manual;
- of damage caused by incorrect, abnormal handling or abuse during delivery;
- the unit has been repaired (or shows signs of repair) by someone not authorised by Stage Accompany
- the original receipt of purchase cannot be presented whenever warranty service is required;
- the cause of damage is unknown.



Shipment of the product to an authorised Stage Accompany service centre is at the risk and responsibility of the customer. Stage Accompany will pay for all labour and material expenses for all repairs covered by the warranty. Stage accompany will not pay for the costs of shipment to the service centre or factory. However, Stage Accompany will pay for the costs of return shipment to the customer if repairs are covered by warranty.

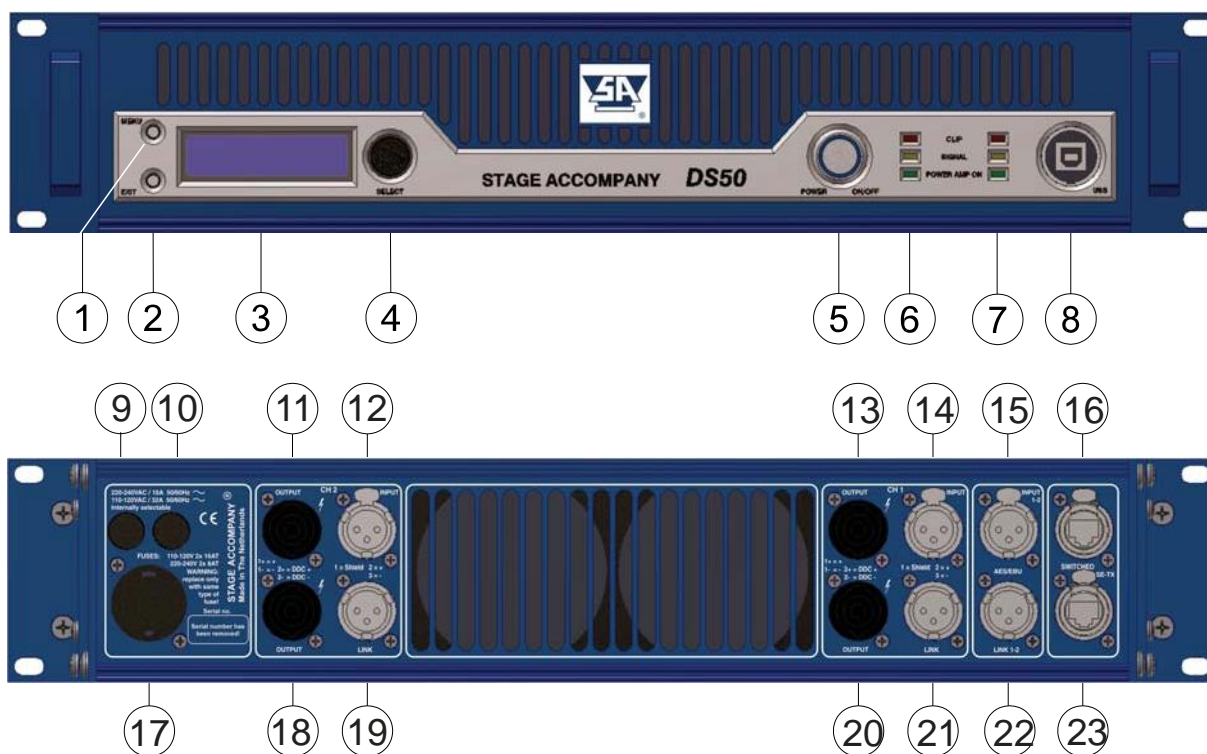
**Caution:**

Warranty work can only be performed at authorised Stage Accompany service centres or at the Stage Accompany factory. Every repair or attempted repair by a non-authorised party will void the warranty.

Stage Accompany reserves the right to alter specifications without prior notice.



## 9 Overview



- ① Function Button "MENU"
- ② Function Button "EXIT"
- ③ LCD Display
- ④ Rotary Encoder / Function "Button Select"
- ⑤ Power Switch
- ⑥ Power Amp On, Signal and Clip LEDs (Channel 1)
- ⑦ Power Amp On, Signal and Clip LEDs (Channel 2)
- ⑧ USB connection
- ⑨ Fuse (Channel 2)
- ⑩ Fuse (Channel 1)
- ⑪ Output (Channel 2)
- ⑫ Input (Channel 2)
- ⑬ Output (Channel 1)
- ⑭ Input (Channel 1)
- ⑮ AES/EBU digital input
- ⑯ Switched Ethernet connection
- ⑰ AC mains connection
- ⑱ Output (Channel 2)
- ⑲ Link output (Channel 2)
- ⑳ Output (Channel 1)
- ㉑ Link output (Channel 1)
- ㉒ AES/EBU buffered digital output
- ㉓ Switched Ethernet connection