GAE
Digital PA Master DSC28

User‘s Guide

User's Guide

Digital PA Master DSC28 (from Serial-No. E04-0110)

Table of Contents

Reference to EC statement of conformity

This document confirms that the product **GAE Digital PA Master DSC28** bearing the CE label meets all requirements in the EMC directive 89/336/EEC laid down by the Member States Counsel for adjustment of legal requirements. Further more the product complies to the rules and regulations of the electromagnetic compatibility of devices from 30. August 1995

This product bearing the CE label complies with the following harmonised or national standards:

EN 55022/-11/-14; EN 61000-3-2; EN 61000-3-3; EN61000-4-2; EN 61000-4-3; EN61000-4-4; EN61000-4-5; EN 61000-4-6; EN 61000-4-11

The authorised declaration and compatibility certification lies with the manufacturer and can be viewed on request. Responsible as manufacturer is:

opal audio vertrieb GmbH, Engerstraße 47, D-33824 Werther, Tel. 05203-236/237, Fax 238

The awarding of the CE label confirms the compliance with legal directives issued for the manufacture and marketing of electronic and electrical devices. As such the CE label is not a "seal of quality" but rather proof that the device bearing the CE label is conform with the electromagnetic compatibility standards laid down in the above named testing regulations.

Liability- and guarantee conditions

Declaration of liability. opal audio vertrieb GmbH accepts no liability for damage to loudspeakers, amplifiers or other devices that become damaged through the use of the DSC28. This applies to the regular, as well as the improper or negligent start-up and/or installation of the DSC28. Also, opal audio vertrieb GmbH accepts no claims in tort, even from third parties, based on speculations of alleged restricted or absence of function of the DSC28 (e.g. cancellation of events).

Product guarantee. Beyond the framework of the legal requirement, opal audio vertrieb GmbH guarantees the DSC28 to be free from defects in material and workmanship for a period of 24 months after date of purchase. As valid evidence for the beginning of the period of guarantee is the date of an official GAE-distributor's issued invoice. As manufacturer, opal audio vertrieb GmbH will replace faulty parts and restore defect modules within the period of guarantee, if the defect has appeared under normal operating conditions. The evaluation of a guarantee claim is acknowledged after our inspection, provided that the device has been returned freight and carriage paid and in the original packaging. Excluded from guarantee are faults incurred by improper electrical or mechanical connection or as a result of transport or accident. This guarantee is voided by any unauthorised repair attempts or by the removal or alteration of the device's serial number.

Contents of packaging

The standard packaging of the DSC28 contains:

- 1 GAE Digital PA Master DSC28 with customer specific configuration
- 1 1.5m-connection cable, Sub-D9/DIN-5-pole, for the connection to a serial PC-interface (COM#)
- 1 replacement fuse M1A (1A, medium-slow behaviour)
- 2 lateral mounting elements
- 1 front and 1 back mounting element
- PE-cover
- User's Guide
- single-sheet documentation with serial number, preset-configuration, output stages-configuration etc.

Important notices

🕮.... denotes: Refer to further information in the chapter specified by the symbol.

💣 denotes: Please regard this warning as especially critical.

Before the initial start-up of the device, please read the following indications and warnings:

- Read the User's Guide carefully. It contains numerous pointers for the proper use of the device.
- It cannot be excluded, that this user's guide shows typographical failings or misprints, it is however regularly checked and proof corrections can be requested in the form of future updates.
- Modifications which serve the purpose of technical improvement of the device may be carried out without prior notification.
- Keep the original packaging of your DSC28, so that, in the case of returning the device for maintenance, it can be shipped originally packed. We reserve the right to replace non-original packaging on returning the device to the owner. In this case the packaging will be invoiced to the customer.
- Always pay attention to the sufficient cooling of the device during operation. This especially applies when installing in racks above other heat generating devices.
- Never pull the mains plug by means of the mains cable. Always pull the plug itself. Be certain, that the mains cable does not become crushed or damaged by sharp edges and never replace a damaged mains cable yourself.
- During operation and storage always protect the device from dust, moisture and direct sunlight.
- Only clean the device with a dry linen cloth. In the case of strong soiling this can be moistened with water and a small amount of household detergent. Never use cleaning-agents containing solvents to clean the device.
- Use only high-quality cable material to connect the device.

- 💣 Leave all repair- and maintenance-work to qualified technical personnel. Any future guarantee claims will be invalidated by unauthorised manipulation.
- 💣 The opening of the device is not required for operation as there are no user adjustable components located within the casing. Solely to release the remote-controllable switch-on function (🕮 Chap.18) will necessitate the removal of the device cover. Never forget, to disconnect from the mains before opening the cover.
- 💣 The device including the mains cable and -plug may not be altered or redressed. The operation with an opened enclosure is not permitted.
- 💣 Always ensure the correct grounding of the device via the mains-plug. Never cover the grounding terminal of the plug by means of insulation material !
- 💣 Mains fuses cannot prevent an unexpected malfunction of electrical components, rather they should protect the user and its environment from damage. For this reason never try to substitute the mains fuse by any other than the specified M1A type (1A, medium behaviour). Never try to repair or bypass a blown mains fuse.
- 💣 The substitution of power amplifiers in a system driven by the DSC28 may only be carried out without further consideration, when the performance data and amplification factors of the new amplifiers are totally comparable to those of the original amplifiers. If that is not the case, possible losses to the tonal behaviour and to the safety of the speaker components in the system must be taken into account.
- 💣 High sound pressure levels can lead to irreparable injures to the human hearing. In the region of the threshold of pain even physical impairment of the entire organism cannot be exempted. Modern sound systems are designed for high sound reproduction levels and as such, when improperly handled, can cause injury to the human hearing organs. Never expose anybody, not even yourself, to extreme high volume levels over a longer period of time.

Initial start-up

The **Digital PA Master DSC28** is a **system-controller** for the driving of pre-specified loudspeaker systems. For this reason the manufacturer has supplied this unit with **presets** consisting of frequency cross-over's (🕮 Chap.14), system correction (🕮 Chap.14) and limiter (🕮 Chap.15) for one or more systems. One of these presets is loaded after connection of the device to the power source, so that provided that it has been properly connected, the system is immediately ready for use (🕮 Chap.16).

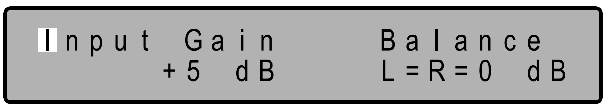
The controller is the link between numerous signal sources and editing devices and the power stage of the system (amplifiers/transducers). One of its top priority duties is the protection of the speaker components as well as the avoidance of excessive distortion. For this reason the manufacturer has exactly calibrated the DSC28 to match "your" power amplifiers (🕮 Chap.12, Chap.14).

To start-up, proceed as follows:

- 💣 Be sure that the controller's configuration matches your power amplifier set.
- Secure the device in a rack with four bolts and connect the in- and output-ports according to the connector panel requirements (🕮 Chap.16).
- Connect the controller with the mains voltage. The unit operates safely within a range of 85V...265V. The performance can even be conditionally maintained down to around 50V, however the output paths will become muted at $\leq$ 75V. Devices with a serial number up to -0030 recognise and store as to whether they are connected to the European net (230/240V) or to the American net (110/115V) and mute at voltages of $\leq$ 155V or $\leq$ 75V respectively.
- After about 4 seconds the device is ready for operation and releases the protective mute to the 8 audio outputs. Now you are automatically located at the menu-point **Input Gain**, and able to carry out the first important settings.
- 💣 During the initial start-up make sure that you only operate your system at low nominal signal levels. Double check the correct speaker link-up of the particular paths of your system (🕮 ?User's Guide of the speaker manufacturer?).
- Should you have not reached this first menu-point, read (🕮 Chap.17 • Initialise).
- Match the power amplifiers to the source device by adjusting the **Input Gain** by means of the increment dial and, parallel to this, whilst observing the VU-meters and the DSC28's output limiter display, move the master fader of your mixing console to the desired full operating-level of your system. Because of the high maximum dynamic-range of the analog-inputs of 28dBu, an adjustment setting of about 30dB for +4dBu-systems is normal.

SET-0: Input Gain, Balance

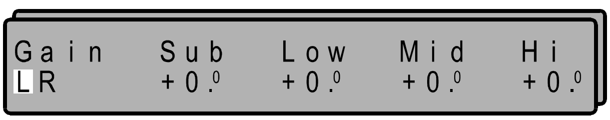
Further menus: ↓ SET-1..5, ↑ SET-6



- At this stage you may possibly want to adjust the bass-level. Change into the menu **Output Gain** for the output-paths (6 times DOWN-key) and press the RIGHT-key to choose the required path (**Sub** and/or **Low**)

SET-2: Output-Gain

Further menus: ↓ SET-3..5, ↑ SET-1..0+6



- If your system is correctly connected (🕮 Chap.16), you can now perform your first event. Further system settings should only be carried out after reading the respective points of the User's Guide.

LED-Display

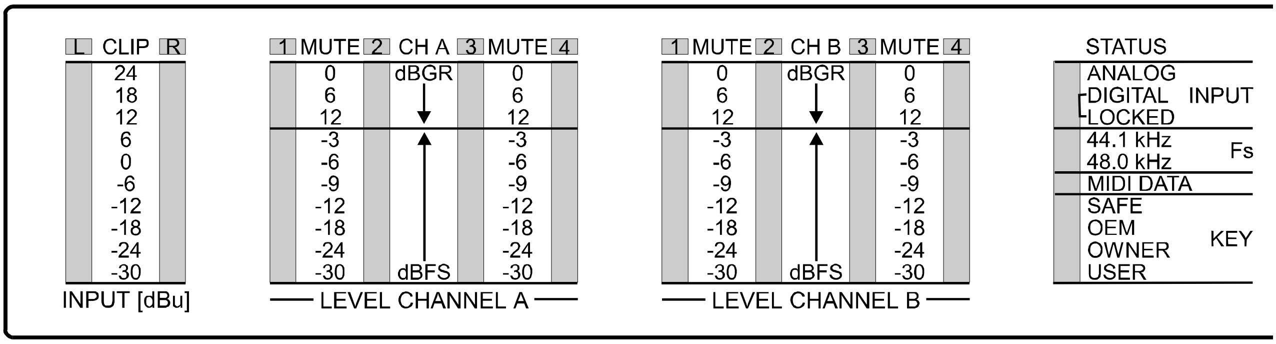


Fig. 4.1 LED-Display

INPUT L + R. The actual input-level to the analog-inputs is displayed by the input-bargraphs in dBu (0dBu≡ 0.775VRMS). A corresponding audio-signal drives the digital-input to the full without considering the word-width of the source and as such is incapable of being overdriven. For this reason the clip-displays are only present on the analog-inputs. The threshold-value lies below 1kHz at ≥ 28dBu (28dBu ≡ 19.47V_{RMS}, 27.53V_{Peak}) and above 10kHz due to pre-emphasis at 18dBu.

LEVEL CHANNEL A + B. The output-bargraphs show (green) the digital output-level before conversion in dBFS (FS ≡ Full Scale) as well as (red) the limiting action in dBGR (GR ≡ Gain Reduction). The 0dB-position has the dual task of signifying full digital-level (≡ 0dBFS) and the advent of limiting (> 0dBGR), a higher digital- as well as analog output-level is not possible. The MUTE-displays indicate inactive outputs.

Due to the level-matching of the analog output-stages to the amplification-value of the connected amplifiers, the exact maximum-power, pre-determined by the loudspeaker components, can be supplied to the loud-speaker-system. Because of this, the output-bargraphs are an accurate visual-check of the level at which the connected PA-system is being driven.

Attention! The power amplifiers should meet or exceed the power requirements of the connected components. If the power amplifier is matched without headroom-reserve there is a risk that, during critical mains-voltage fluctuations, the necessary DSC28-limiters do not come into effect before the amplifier is overdriven. (This is the normal situation for amplifiers with insuficient power.) In this case it is unfortunate that the excellent tonal-qualities of the DSC28-peak-limiter, compared to the signal-limitation possibilities of the respective amplifier, remain unused. Should you however, use inadequately dimensioned amplifiers, it is important to pay attention to the use of amplifier-integrated clip-limiters (for the sake of both your system components and your ears).

STATUS. This LED-bargraph displays information regarding internal settings of the DSC28.

INPUT. Here the activated input is indicated: **ANALOG**, **DIGITAL** (/ **DIGITAL INSERT**) **LOCKED.**

Fs. Indicates the sample-frequency **44.1kHz**/**48kHz**. The DSC28 operates with a sample-frequency of 44.1kHz. A sample-rate of 48kHz can be factory configured for special purposes. The DIGITAL INPUT is able to receive AES/EBU signals with a sample rate between 32kHz and 96kHz. These are then converted to 44.1kHz by means of a sample rate converter.

MIDI DATA. Indicates data-reception via the MIDI- or RS-232-inputs and data-transmission via the MIDI-output.

KEY. One of four available security- levels for the prevention of unauthorised operation is indicated.
Attention! Security-level selection of is not yet supported by the software. (stand:01/2001)

Increment-dial, Function-keys and VF-Display

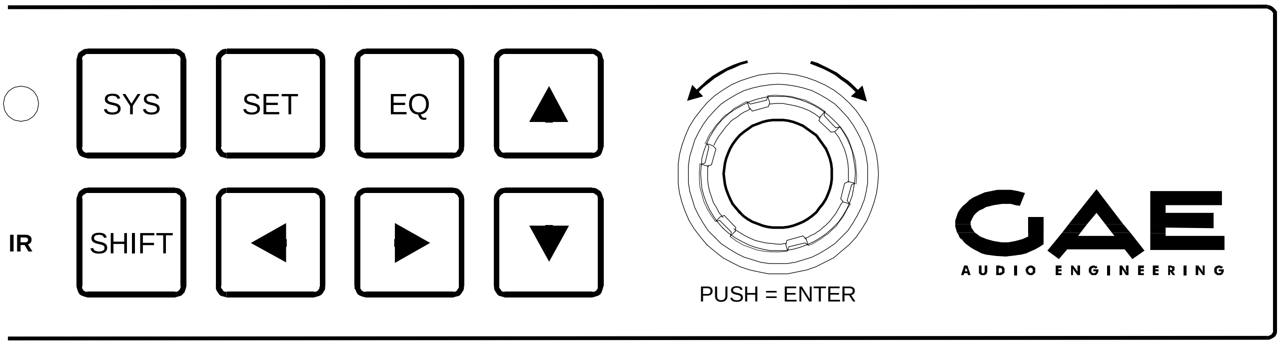


Fig. 5.1 control panel

SYS — Access to the **menu-area SYS**, system-settings.

SET — Access to the **menu-area SET,** speaker set-up. The menu SET-2, Input Gain & Balance, appears approx. 4seconds after power-up.

EQ — Access to the **menu-area EQ,** IIR-equaliser.

▲ ▼ — Selects single menus within the menu-areas. Holding the button depressed enables quick-step-through. (Symbols: **UP-key: ↑ DOWN-key: ↓**)

◀ ▶ — Moves the (blinking) **Cursor █** within a single menu to an adjustable parameter; also page change, if a menu consists of more than one page. (**█ X** denotes cursor position X, counted from left to right) Holding the button depressed enables quick-step-through. (Symbols: **LEFT-key: ← ; RIGHT-key: →**)

SHIFT — **Shift function-key** for all other tip-keys and the increment-dial.

Increment-dial — **Parameter-selection** by turning the dial (Symbol: **↙O↘**). Simultaneously pressing whilst turning the dial enables larger-value steps. For smaller-value steps press the SHIFT-key whilst turning the dial.
Confirm setting by pushing the dial (PUSH = ENTER, Symbol: ⊙). A flashing point in the display (top right) warns that a parameter-value has been changed and requires confirmation before becoming operational.

IR — Infra-red remote-control-activity. An optional IR-receiver is situated behind the small round window.
Attention! This function is available as an optional extra.

VF-Display — A vacuum-fluorescence-display with two, 24-character lines enables the visual control of all operation-measures carried out at the user-controls. The display's blue coloration allows easy distinction of the characters, even under adverse lighting conditions.

█	Cursor, active menu
●	Is blinking?, requires confirmation by ⊙, ENTER
| →	Further menus, right
← |	Further menus, left
← →	Further menus, right as well as left

Boot-display, first-parameters and menus

Boot-display

Further menus after approx. 4 secs

```
   ITA RWTH Aachen
DIGITAL PA MASTER + EQ
```

This display appears after power-up. The basic hard- and software-concept for the GAE DSC28 was developed within the framework of a research and development arrangement with the Institut für Technische Akustik at the Rheinisch-Westfälischen Technischen Hochschule, Aachen, **ITA RWTH Aachen**.

First-parameters. On shipping the device, all positions in all menus of the system-controller DSC28 are occupied by first-parameters. The device is as-such instantly ready for operation. This condition can be recalled via the function **Initialise** (🕮 Chap. 17). However –to prevent misunderstanding– the function Initialise is not necessary for the initial power-up (🕮 Chap.3). All menus shown within this guide are shown with these first-parameter settings. The relevant symbols are explained in the following two examples:

- **Min ↙○ X ○↘ Max** denotes, starting from the first-parameter, value **X,** clockwise rotation of the increment-dial increases values up to **Max,** respectively counter-clockwise rotation decreases values down to **Min**.
- **L➔ ↙○ LR** denotes, starting from the first-parameter **LR,** the parameter **L➔** can be adjusted by counter-clockwise rotation of the increment-dial.

After the first parameter-adjustment, the new value of the underlined first-parameter becomes the basis for further parameter-adjustments.

Menu-areas. The three menu-areas: System-settings **SYS,** Speaker set-up **SET** and IIR-Equaliser **EQ** can be selected by the pertinent function-keys **SYS, SET** and **EQ**. Approx. 4 seconds after power-up the menu-area SET is automatically activated with SET-0, █ **1**, Input Gain.

Multi-page-menus are located under SYS and EQ and are explained later in this guide.

2-page menus are located under menus SET and EQ, and have a common function, which can be illustrated by using the mute-menu as an example: a link-function for the simultaneous operation of both identical channels of the controller (Channel I ≡ Left and Channel II ≡ Right), is adjustable at the most-left-hand cursor-position █ **1**. **LR** denotes linked-operation-mode, **L➔** denotes the adjustment is only valid for Channel I. The right-arrow indicates that, beyond the far-right cursor-position, █ **5** the second page of the menu, █ **6..10,** can subsequently be accessed. **🡐R** denotes the adjustment is only valid for channel II. Here the left-arrow indicates, that beyond the far-left cursor-position, █ **6,** the first menu-page, █ **5..1**, can be accessed.

```
Mute   Sub   Low   Mid   Hi
L→     No    No    No    No
```

█ **1** █ **2** █ **3** █ **4** █ **5**

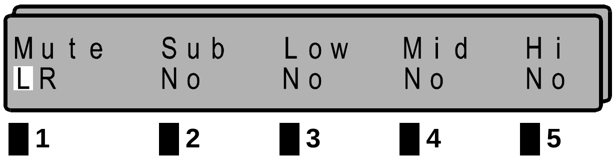
Menu-page 1. Cursor-position **L➔,** █ **1.** The link-function **LR** can be selected by turning the increment-dial, **L➔ ○↘ LR**

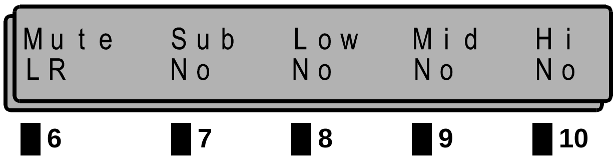
```
Mute   Sub   Low   Mid   Hi
←R     No    No    No    No
```

█ **6** █ **7** █ **8** █ **9** █ **10**

Menu-page 2. Cursor-position **🡐R,** █ **6.** The link-function **LR** can be selected by turning the increment-dial, **🡐R ○↘ LR**

This 2-page-menu-function remains even during linked **LR** condition. To reach page 2 from page 1 move the cursor with the right-key beyond the far-right cursor-position, █ **5.** █ **6..10** can now be accessed although it is not possible to distinguish the channels as both indicate **LR**. Only when the link-function is cancelled, the distinction appears as **L➔** (link-function was cancelled at page 1), or as **🡐R** (link-function was cancelled at page 2).





Menu-page 1. Cursor-position **LR, █ 1.**
The link-function can be cancelled by turning the increment-dial, **L➔ ↙O LR**

Menu-page 2. Cursor-position **LR, █ 6.**
The link-function can be cancelled by turning the increment-dial, **➔R ↙O LR**

If under **L➔** and/or **R🡐**, changes are made, these will be written to the channel from which the link-function **LR** is re-activated.

Menus under SET, Speaker Setup

SET-0: Input Gain, Balance

```
Input Gain          Balance
    +5 dB           L=R=0 dB
```

Further menus: ↓ SET-1..5, ↑ SET-6
Cursor-Position: █ 1 █ 2

1, Input Gain: -83dB ↙O +5dB O↘ +45dB, 1dB steps. Adjust the PA to the source-device with the increment-dial by adjusting the Input Gain. For this, whilst observing the VU-meters and the DSC28's output-limit display bring the master-fader of your mixing-console to the desired position for full-level-operation of your system. Because of the analog-inputs' high dynamic-range-limit of ≤ 28dBu, an level of nearly 30dB for +4dBu-systems is normal.

2, Balance: R-16dB ↙O L=R=0dB O↘ L-16dB, 1dB steps. To adjust to different levels or to compensate a level-difference between the two channels.

SET-1: Master Delay

```
      Master Delay
 0.0 ms            0.0 m
```

Further menus: ↓ SET-2..5, ↑ SET-0 + 6

Min-Delay O↘ 1000ms, 340m, resolution 0.363ms. The minimum value corresponds to the intrinsic run-time of the controller. This delay-time is the sum of the different signal-processing times required by such system-components as the AD- and DA-converters, the filters allocated to the converters, down- and over-sampling-filter as well as the pre-viewing limiter. Added to these are the loudspeaker-system-dependent delays derived from the system-eqalisation (filter-slopes, adjusted amplitude- and phase-frequency-response) as well as time-alignment-requirements. (📖 Chap.13 & 14)
Delay-times greater than the preset-dependant Min-Delay-value can be set with the increment-dial.

SET-2: Output-Gain (Wege)

```
Gain   Sub    Low    Mid    Hi
LR     +0.0   +0.0   +0.0   +0.0
```

Further menus: ↓ SET-3..5, ↑ SET-1..0 + 6
Cursor-Position: █ 1+6 █ 2+7 █ 3+8 █ 4+9 █ 5+10

1, Channel I-Link: L→ ↙O LR; 6, Channel II-Link: ←R ↙O LR. To cancel the channel-link-function. (📖 Chap.6/1 ● 2-page-menus).

2..5 and 7..10, Gain, Sub, Low, Mid, Hi: -18.0db ↙O 0.0dB O↘ +6.0dB, 0.5dB steps. With this control it is possible if necessary, to balance the individual loudspeaker-paths. Situated in front of the limiters the Output-Gain-controls are not suitable for level-adjustment of a power-amplifier with a different amplification factor than the originally assigned amplifier, because the limiter-threshold is not affected by this adjustment.

SET-3: Output-Mute

```
Mute   Sub    Low    Mid    Hi
LR     No     No     No     No
```

Further menus: ↓ SET-4..5, ↑ SET-2..0 + 6
Cursor-Position: █ 1+6 █ 2+7 █ 3+8 █ 4+9 █ 5+10

1, Channel I-Link: L→ ↙O LR; 6, Channel II-Link: ←R ↙O LR. To cancel the channel-link function. (📖 Chap.6/1 ● 2-page-menus).
2..5 and 7..10, Mute, Sub, Low, Mid, Hi: No O↘ Yes. The upper (red) LEDs of the Output-Level-Displays indicate the mute-status of each output.

SET-4: Output-Phase Invert

Further menus: ↓ SET-5, ↑ SET-3..0 + 6

```
P.Inv  Sub   Low   Mid   Hi
LR     No    No    No    No
```

Cursor-Position: ▌1+6 ▌2+7 ▌3+8 ▌4+9 ▌5+10

1, Channel I-Link: L➔ ⬋○ LR; 6, Channel II-Link: ←R ⬋○ LR. To cancel the channel-link-function. (📖 Chap.6/1 ● 2-page-menus).

2..5 and 7..10, Phase Invert, Sub, Low, Mid, Hi: No ○⬊ Yes. With this controll you can easily invert the phase of an individual output path. This can be of assistance in achieving optimal bass reproduction when working with bass systems of different designs and positioning to one another (e.g. Low: System-Bass; Sub: zusätzliches Bassystem).

SET-5: Output-Delay (Paths)

Further menus: ↑ SET-4..0 + 6

```
Delay  Sub   Low   Mid   Hi
LR     0.0   0.0   0.0   0.0
```

Cursor-Position: ▌1+6 ▌2+7 ▌3+8 ▌4+9 ▌5+10

1, Channel I-Link: L➔ ⬋○ LR; 6, Channel II-Link: ←R ⬋○ LR. To cancel the channel-link-function. (📖 Chap.6/1 ● 2-page-menus).

2..5 and 7..10, Delay, Sub, Low, Mid, Hi: 0.0msec ○⬊ 92.1msec, resolution 0.363msec. The values in the display are rounded up/down to one decimal. Through the use of the output-delays run-time corrections can be made especially in the bass region when Mid/high packs and Bass systems are operated at some distance to each other (e.g. Mid/highs with some Bass support flown and an additional centre-bass cluster under the centre of the stage). Do not attempt to use the output-delays for time-allignment purposes of loud-speaker components in a GAE Mid/High system. The time-allignment correction between the single components within a system is an integral part of the preset and as such already adjusted to its optimum. For this reason the parameters of the menu Delay-Link (📖 SET-11) should also not be altered. Failure to observe this point will result in impairment to the sound character and the dispersion properties of the Mid/High system.

Menus above SET-Menu 0

SET-6/0: Setup

Further menus: ↓ EQ-0..5, ➔ SET-6/1

```
Setup (→ Store)          ●
000 Default Program
```

000 ○⬊⬊ 014. Change the momentary set-up (stored parameter set of the DSC28) with a different set-up, by selecting the required set-up and confirming choice by pressing ⦿ **ENTER**. The new set-up is only made available after the ENTER-key has been pressed.

➔ Store. With the ➔RIGHT-key move to this menu-point to store the momentary settings and a neme to a memory position.

↑ ←. These keys have no function in this menu.

SET-6/1: Save Setup as

This function cannot be interrupted !

```
Save Setup as:
Name 18 Zeichen
```

⬋⬋O **Character** O⬊⬊ The momentary set-up (parameter set of the DSC28) can be given a name (max. 18 characters) and stored to one of 15 memory positions (000...014).
🡅 **Space** 🡅 **Character A** 🡅 **Character a.** With the UP-key the three indicated start-positions for character-selection with the increment dial are reached. Pressing the DOWN-key steps back through the list.
SHIFT ➔. This combination inserts a space at the cursor position.
➔ **Character 1** ➔ **Character 2** ➔ **...** ➔ **Character 18.** With the RIGHT-key all 18 character-positions are reached so as to adjust them by means of the increment-dial. Depressing the LEFT-key enables the return to the previous positions. The active position is underlined with a cursor.
⦿ **ENTER.** Confirm the selected name by pressing the ENTER-key. The third page of the menu is now activated, in which you can store the adjusted EQ-Set-up to a memory location.
Warning! This function cannot be interupted. The menu can only be exited by pressing the ENTER-key.

SET-6/2: Save Setup to

This function cannot be interrupted !

```
Save Setup to:
000 Default Program
```

000 O⬊⬊ **015.** Select one of the 15 memory positions (000...014), to which the momentary configuration is to be stored. The name shown in the display represents the stored set-up at this memory position which will be over-written. On leaving the factory all memory positions are occupied by **Default Setup** containing the original first-parameter set.
🡇 🡅 🡄 **➔.** These keys have no function in this menu.
⦿ **Enter.** Confirm the number of the selected memory position with the ENTER-Taste. Enter also returns the display to the first page of this menu point.
Warning! This function cannot be interupted. The menu can only be exited by pressing the ENTER-key.

Menus under SHIFT+SET, Speaker Setup

SET-7: Preset

Further menus: 🡇 SET-8..11

```
Preset                ●
Name 18 Zeichen
```

Preset 1 O⬊ **Preset 2** O⬊**.. Preset X.** As a system-controller, the DSC28 is configured with presets for one or several loudspeaker-systems by the manufacturer. A preset consists of: frequency-cross-over (🕮 Chap.14), system-equaliser (🕮 Chap.14) and limiter (🕮 Chap.15). The last selected preset is loaded automatically after power-up so that the system is immediately ready for operation, provided that it is properly connected (🕮 Chap.16). Before selecting any preset, read the manual of the loudspeaker-system carefully! For systems comparable with the **GAE Director** and comprising Top, Bass and additional Subbass (4-Way), the memory of the DSC28 can hold up to approx. 12 presets.
⦿ **Enter.** Confirm your selection by pressing the ENTER-key!

SET-8: Input Select

Further menus: ↓ SET-9..11, ↑ SET-7

```
Input Select         •
Analog
```

Analog **○↘ Digital ○↘ Dig. Insert.** Select Analog for an analog source, Digital for a digital source that transmits with 32kHz...96kHz Sample-Frequency. Ensure that the clock-pulse generator of the DSC28 is operating correctly: Status-LED Locked must light.
⦿ **Enter.** Confirm your selection by pressing the ENTER-key!

The digital in- and output of the device operate to the AES/EBU-protocol.
The digital output always supplies the digital form of the input-signals with 44.1kHz Sample Rate. The digital input is equipped with a sample rate converter and “understands” sample rates between 32kHz...96kHz.

SET-10: Limiter Release

Further menus: ↓ SET-11, ↑ SET-9..7

```
Limiter Release
    60dB/s
```

Limiter Release: 10dB/s ↙○ 60dB/s ○↘ 250dB/s, 1dB/s steps. The value of 60dB/s as the standard-value for the limiter release-behaviour results from practical experience. Decrease/increase this value for longer/shorter Release-times. Attack- and Hold-times are constant (📖 Chap.15).

SET-11: Delay Link

Further menus: ↑ SET-10..7

```
Delay Link
  Mid+Hi
```

Off ↙○ Mid + Hi ○↘ Low + Mid + Hi. For the operation with the GAE-Director-System the Delay Link should always be set at Mid + Hi. This setting ensures that in the Menü Output Delay (📖 SET-7) the 2-way active Director Top can only be commonly altered.

Menus under EQ, IIR-Equaliser

To enable the tuning of your system to differing acoustical situations the DSC28 is equipped with a full-parametric equaliser. Situated prior to the cross-over-network the user has access to 14 EQ-bands per channel. The EQ is of the IIR-filter type and can simulate analoge filters, adjustable at the user-interface. The following filter-types can be selected: Bell (≡ Peak), Low-Shelving 6 and 12dB/oktave (≡ LS 6 and LS 12), High-Shelving 6 and 12dB/oktave (≡ HS 6 and HS 12), Low-Pass 6 and 12dB/oktave (≡ LP 6 and LP 12), High-Pass 6 and 12dB/oktave (≡ HP 6 and HP 12).

Attention! Only use the high- and low-pass types of EQ for additional band-limitation on the lower and upper end of the total transfer-range. (Or if possible, do not use them at all.) Be aware that a system-equalising cross-over exactly adjusted to the loudspeaker-system is already present within the preset. Use the Bell-type filters to compensate the tonal-discoloration caused by room acoustics. Especially within the bass-range the possibilities of balancing the single loudspeaker-paths through use of the Output-Gain-controls should first be applied (🕮 Chap.7/2).

Attention should be given to one restrictive particularity which appears when using shelving-filters: If for example, a 12dB/octave high-shelving-filter with a cut-off frequency of 5kHz and a Q-factor of 0.7 is selected, the user-interface will accept no gain-setting higher than +5dB. A higher value is only obtainable by reducing the Q-factor or raising the cut-off frequency. The device sets these limits due to the calculated filter-coefficients required for a higher boost exceeding the DSP's (Digital Signal Processors) processing-range and as such can not be produced.

EQ-0: EQ-Gain, -On/Off

```
EQ     Volume            EQ
         -3 dB           On
```

Further menus: 🡻 EQ-1..14

Cursorposition: █ 1 █ 2

█ **1, Gain: -24dB ⬋○ -3dB ○⬊ 0dB, 1dB steps.** Should the activated EQ-bands be solely used for frequency reduction then a value of 0dB is recommended. Should however, frequency-boosts also be applied then a gain-reduction in accordance with the highest boost-value is recommended.

█ **2, On/Off: Off ○⬊ On.** EQ -on -off. At EQ-off, all filters are non-effective, the adjusted EQ-Gain however remains effective.

EQ-1: EQ 1

```
EQ 1   Type     Q     Hz     dB
LR     Peak     2.0   30.0   +0.0
```

Further menus: 🡻 EQ-2..14, 🡹 EQ-0

Cursor-Position: █ 1+6 █ 2+7 █ 3+8 █ 4+9 █ 5+10

█ **1, Kanal I-Link: L➔ ⬋○ LR;** █ **6, Kanal II-Link: 🡸R ⬋○ LR.** To cancel the channel-link-function. (🕮 Chap.12/1 ● 2-page-menus).

█ **2+7, Type: HP12, HP 6, LP12, LP 6 ⬋○ Peak ○⬊ LS 6, LS12, HS 6, HS12.** Adjustments to an EQ only become affective after depressing the increment-dial (ENTER*).*

█ **3+8, Q: 0.1 ⬋○ 2.0 ○⬊ 6355, steps: 0.1 (Min - 3.0); 0.2 (3.0 - 6.0); 1 (6.0 - 10.0); 2 (10.0 - 50.0); 5 (50.0 - 200); 10 (200 - 1000); 20 (1000 - Max).** For a general step-increment of 0.1 additionally press the SHIFT-key during the Q-selection.

█ **4+9, Hz: 1.0Hz ⬋○ 30Hz ○⬊ 20.0kHz, steps: 1Hz (1.0 - 100.0Hz); 2Hz (100.0 - 150.0Hz); 5Hz (150.0 - 300.0Hz); 10Hz (300.0 - 600.0Hz); 20Hz (600.0 - 1.00kHz); 50Hz (1.00 - 5.00kHz); 100Hz (5.00 - 20.0kHz).** For a general step-increment of 0.5Hz additionally press the SHIFT-key when selecting the filter-frequency.

■ **5+10, dB: -99.0db ↙O <u>0.0dB</u> O↘ +12.0dB, 1dB steps.** For a general increment of 0.1dB additionally press the SHIFT-key during the level-selection.

A higher step-increment during the parameter-selection can be achieved by the simultaneous turning and pressing of the increment-dial.

In EQ-menu 1, the operation of EQ 1has been described. EQs 2-14 are shown to document their first-parameter (default) settings. The operation of all EQs is identical.

<u>EQ-2:</u> EQ 2

Further menus: ↓ EQ-3..14, ↑ EQ-1..0

```
EQ 2  Type    Q     Hz    dB
LR    Peak    2.0   60.0  +0.0
```

<u>EQ-3:</u> EQ 3

Further menus: ↓ EQ-4..14, ↑ EQ-2..0

```
EQ 3  Type    Q     Hz    dB
LR    Peak    2.0  120.0  +0.0
```

<u>EQ-4:</u> EQ 4

Further menus: ↓ EQ-5..14, ↑ EQ-3..0

```
EQ 4  Type    Q     Hz    dB
LR    Peak    2.0  250.0  +0.0
```

<u>EQ-5:</u> EQ 5

Further menus: ↓ EQ-6..14, ↑ EQ-4..0

```
EQ 5  Type    Q     Hz    dB
LR    Peak    2.0  500.0  +0.0
```

<u>EQ-6:</u> EQ 6

Further menus: ↓ EQ-7..14, ↑ EQ-5..0

```
EQ 6  Type    Q    kHz    dB
LR    Peak    2.0  1.00   +0.0
```

<u>EQ-7:</u> EQ 7

Further menus: ↓ EQ-7..14, ↑ EQ-6..0

```
EQ 7  Type    Q    kHz    dB
LR    Peak    2.0  2.00   +0.0
```

<u>EQ-8:</u> EQ 8

Further menus: ↓ EQ-8..14, ↑ EQ-7..0

```
EQ 8  Type    Q    kHz    dB
LR    Peak    2.0  4.00   +0.0
```

<u>EQ-9:</u> EQ 9

Further menus: ↓ EQ-9..14, ↑ EQ-8..0

```
EQ 9  Type    Q    kHz    dB
LR    Peak    2.0  8.00   +0.0
```

<u>EQ-10:</u> EQ10

Further menus: ↓ EQ-11..14, ↑ EQ-9..0

```
EQ10  Type    Q    kHz    dB
LR    Peak    2.0  16.0   +0.0
```

<u>EQ-11:</u> EQ11

Further menus: ↓ EQ-12..14, ↑ EQ-10..0

```
EQ11  Type    Q     Hz    dB
LR    LS12    0.7  40.0   +0.0
```

<u>EQ-12:</u> EQ12

Further menus: ↓ EQ-13..14, ↑ EQ-11..0

```
EQ12  Type    Q    kHz    dB
LR    HS12    0.7  5.00   +0.0
```

<u>EQ-13:</u> EQ13

Further menus: ↓ EQ-14, ↑ EQ-12..0

```
EQ13  Type    Q     Hz    dB
LR    LS12    0.7  40.0   +0.0
```

<u>EQ-14:</u> EQ14

Further menus: ↑ EQ-13..0

```
EQ14  Type    Q    kHz    dB
LR    HS12    0.7  5.00   +0.0
```

<u>Menüs unter SHIFT EQ</u>

<u>EQ-15:</u> EQ Noise Shaper Select

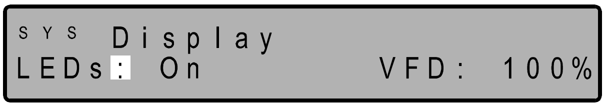
Further menus: keine

```
EQ Noise Shaper Select
1. Order Error Feedback
```

None ↙○ <u>1. Order Error Feedback</u> ○↘ Lipsh. 3 Taps Modified E ○↘ Lipsh. 5 Taps Improved E. Re-quantisation of the 48-Bit audio-signal to 24-Bit for further proceedings. Adjustments here are only audible at very low levels.

Menus under SYS, System adjustments

SYS-0: Display, LEDs and VFD



Further menus: ↓ SYS-1..2

Cursor position: █ 1 █ 2

█ **1, LEDs: Off ↙○ On.** Here the LED-display can be turned on or off. The status LEDs remain on.

█ **2, VFD: 25% ↙○ 50% ↙○ 75% ↙○ 100%.** Select the desired brightness of the vacuum-fluorescence-display.

Midi-Settings. This menu runs over four pages and serves the adjustment of several parameters for the exchange of data via the serial- or MIDI-interfaces between several controllers or from a PC to one (or several) controller(s) (🕮 Chap.19).

SYS-1/0: Midi-Settings

Further menus: ↓ SYS-2, ↑ SYS-0
→ SYS-1/1

Cursor position: █ 1 █ 2

█ **1, Channel: 1 ○↘↘ 16.** Assign the controller-address (basic channels 1-16), at which data can sent and received.
⦿ **Enter.** Confirm your selection with ENTER!

█ **2, Out: OutOnly ○↘ Out/Thr ○↘ Loop.** OutOnly is a pure MIDI-output. Out/Thr is a MIDI-output with an additional Soft-Thru-function, which passes the signals received at the MIDI-input to the MIDI-output. Loop is a MIDI-output which passes-on all incoming data on channels other than the devices addressed channel. This mode links several DSC28-controllers in a closed MIDI-loop (🕮 Chap.19).

SYS-1/1: Midi-Settings

Further menus: ↓ SYS-2, ↑ SYS-0
← SYS-1/0, → SYS-1/2

Cursor position: █ 3

█ **3, Baudrate: MIDI (31250) ↙○ RS-232 (9600).** Use RS-232 and the supplied cable to remote control the device with the help of a PC via one of it's serial-interfaces (COM#) or to re-load presets or to update the operating system. Use MIDI, to link several DSC28-controllers by means of a standard MIDI-cable (🕮 Chap.19).
⦿ **Enter.** Confirm your selection with ENTER!

SYS-1/2: Midi-TX Filters

Further menus: ↓ SYS-2, ↑ SYS-0
← SYS-1/1

Cursor position: █ 4 █ 5

█ **4, ParaChg: <u>Off</u> ○➘ On .** If this value is set on On, the device sends all performed parameter-adjustments out. As such several device (Slaves) can be commonly operated by one device (Master).
⦿ **Enter.** Confirm your selection with ENTER!

█ **5, TX-Chn: <u>1</u> ○➘➘ 16 ○➘ OCM.** The transmitting channel for parameter-adjustments can be selected by this parameter. As well as channels 1-16 an Omni-Channel-Mode (OCM) can be selected. Parameter adjustments via this channel are accepted by the connected devices independent of thier channel-address.
⦿ **Enter.** Confirm your selection with ENTER!

<u>SYS-2:</u> AES Stat. Samplerate

```
SYS AES-Stat. Samplerate
   .........  44.095 kHz
```

Further menus: ↑ SYS-1..0

Shows the measured internal sample rate.

<u>Menüs unter SHIFT SYS</u>

<u>SYS-3:</u> Version No. 1.23c

```
SYS Version-Nr. 1.23c
Reboot   Initialise
```

█ 1 █ 2

Further menus: ↓ SYS-2, ↑ SYS-5..10 + 2

█ **1, Reboot.** This command resets all DSPs. The last loaded program- and data-set is re-loaded and executed. The outputs of the device are muted during this operation.
⦿ **Enter.** Execute function.

█ **2, Initialise.** This command calls up the menu SYS-4.
⦿ **Enter.** Execute function.

<u>SYS-4:</u> Initialise System?

```
Initialize System?
F1-EXIT F2-ClrRAM
```

Further menus: none
Cursor position: none

F1-Exit. F1 is the SYS-Key. The devicewill not be initialised.

F2-ClrRAM. F2 is the SET-Key. This command returns the device to it's defined initial setting at the time of shipping. All values return to the so-called First-Parameters values, to which this guide refers when explaining the menus. All adjustments performed by the user are lost. The device is ready for the initial start-up. **Attention!** Even presets re-loaded over the serial-interface are lost and have to be re-loaded after this action (🕮 Chap.17). Never apply this command, without being aware of the consequences.

Unit Description

The **GAE DIGITAL PA MASTER DSC28** is a state-of-the-art **digital System-Controller.** Its role as an integral system component, even at the development stage of a loudspeaker system, clearly sets it apart from other loudspeaker management devices available on the market. By utilising the DSC28, elementary system-parameters (e.g. of a mid/high unit) can be decisively influenced during early development stages. The user-accessible presets represent as such, the outcome of intensive development and determine, together with the loudspeaker components and the power amplifiers, the tonal and power handling behaviour of the sound system as a combined single-unit, even in its different configurations.

This product philosophy was already successfully realised by GAE in their drive devices (e.g. system controller BF1, programmable via interchangeable preset cards). System-relevant functions are stored as recallable parameter-sets and important, user-definable variables are freely available. As such, even in the digital-age of signal processing, the **totally user-programmable cross-over** as a central driving device for GAE-loudspeaker systems remains a "second-choice" commodity.

Function. The GAE DSC28 combines the functions of cross-over, equaliser, delay and limiter in a 1 rack-space enclosure. It has been conceived as a remote controllable (RS232/MIDI), 2-channel control-unit for up to 4 way, high power, sound system applications, with two analog inputs, a digital in- and output (AES/EBU) which can also be used as a digital insert and four analog outputs 1 [SUB], 2 [LOW], 3 [MID], 4 [HIGH] per channel. A 2x14-band, full parametric EQ, situated prior to the cross-over network enables the comfortable system-tuning to room acoustics.

Input-dynamic. Without the necessity for analog level-matching the GAE DSC28 achieves, with the help of dual-rate conversion and a combination of analog pre-emphasis / digital de-emphasis, a dynamic range of 130 dB with 28 dBu maximum input level (@ < 1kHz).

Output-dynamic. The output level of each of the outputs of the DSC28 can be individually matched to the input sensitivities of the connected power amplifiers. DSP-controlled noise shaper- and dither-stages with 1st Order-Error-Feedback (DA-Server), serving the DA-converter of the R2R- instead of the Delta Sigma-type, are used. This results in an extraordinary dynamic range of ≥114 dB!

Noise, dynamics and headroom. The signal-to-noise ratio of the DSC28 is -96dBu (unweighted, 22Hz - 22kHz) at a maximum output-level of 18dBu. The lowering of the output-level when adjusting to a power amplifier leads to a corresponding reduction of the noise-level. At the same time, the best possible digital-resolution is attained; these procedures – unusual in a digital device - make the DSC28 even superior to an analog standard-controller. Important modules within the digital range operate with 48-bit-accuracy. Where a reduction from 48-bit to 24-bit occurs, a noise-shaper with 1st Order-Error-Feedback is always implemented so as to guarantee minimum noise-level and distortion, especially when driven with low level signals.

Peak-limiter. The DSC28 fully incorporates the improved possibilities made available to limiter circuitry by digital technology. An “in-advance” signal-analysis of the output-signal allows pre-determined threshold excesses to be detected and the level to be decreased over a fixed time-factor with a matched time-constant. Over-shooting peaks are automatically reduced to the exact value determined by the threshold-level. This procedure enables the exact adjustment of the limiter-thresholds to the absolute maximum-rating of the amplifiers, and the connected loudspeakers. Strong transient-impulses additionally benefit from a pre-masking effect which allows an inaudible processing-time before the occurrence of the impulse. The subsequent hold-time prevents level-modulation in the directly ensuing passages. The “pre-viewing” signal-analysis also allows a much better generation of the control-signal without worsening the attack-time of the limiters. High-frequency distortion of the wanted-signal by the control-signal are thus, as good as eliminated.

RMS-limiter. The RMS-limiters of the DSC28 protect the loudspeaker components by simulating the thermal time-constants of the voice-coils and magnet-materials.

Delay. A master-delay for the adjustment of the signal delay-time is situated prior to the filter-network. A further delay is inserted to each output path for time-alignment adjustment of the individual loudspeaker components and are part of the correcting-filter network. The intrinsic-delay of the DSC28 depends on the path and is between 5 and 7ms, and determined by AD/DA conversion, down-/oversampling and the "pre-viewing" limiter-concept. All further delays occur during the signal-manipulation of filters and are especially dependent on the run-time behaviour of the loudspeaker-system, the cross over slopes and filter-frequency-thresholds.

Filter. The DSC28 calculates crossovers and speaker-equalisation as FIR-filters (finite impulse response).This type of filter requires more computing-power in comparison to digital devices processing with IIR-filters, which simply simulate analog filters within the digital domain. For this reason the computing power has been optimised by the use of down- / oversampling modules in each path/frequency-bandwidth, with the exception of the HIGH-path. This results in band-width limitations of the SUB-, LOW- and MID-paths. FIR-technology allows the realisation of equalising-filters with linear-phase-behaviour which has the advantage of equal delay-times for every frequency range of a complex signal. Unfortunately linear-phase response, down to the lowest frequency-levels lead to extended signal run-times, unacceptable for live situations. This can be compensated by attenuation to minimum-phase (“analog“) response below a specific frequency. For the filtering of an active-system with moderate crossover slopes (< 36 dB/octave), a minimum-phase response up to 120 Hz and a linear-phase response ≥ 120Hz, a controller delay of 30ms has to be considered: intrinsic controller delay of approximately 7ms + filter-delay of approximately 23ms. Due to its configuration with FIR-filter technology, the use of the GAE DSC28 as a user-programmable cross-over allowing subsequent simulation of analog (IIR)-filters is not intended.

Equaliser. (📖 Chap.8). Situated prior to the frequency crossover, the user has 14 bands of full parametric EQ for each channel at his disposal. These are of the (analog) IIR-filter type, adjustable at the user interface. The following filter-types can be selected: Bell, Low-Shelving, High-Shelving, Low-Pass and High-Pass.

Summary. The DSC28 is a non-compromising, system-controller concept operated via a standardised user-interface allowing a choice of presets and the control of user-parameters. These presets are pre-defined and made available as relevant operation-parameters by the loudspeaker-system manufacturer. As such, the DSC28 is ideally suited for **OEM-application**. Designed for use with selected GAE-applications the DSC28 can, on request, be programmed to provide the necessary parameters for the driving of all types and makes of loudspeaker-systems.

The parametric equaliser, completely configurable by the user, substitutes the need for an additional 19“-device and facilitates the uncomplicated minor modifications necessary when adjusting an optimally corrected loudspeaker system to room acoustics.

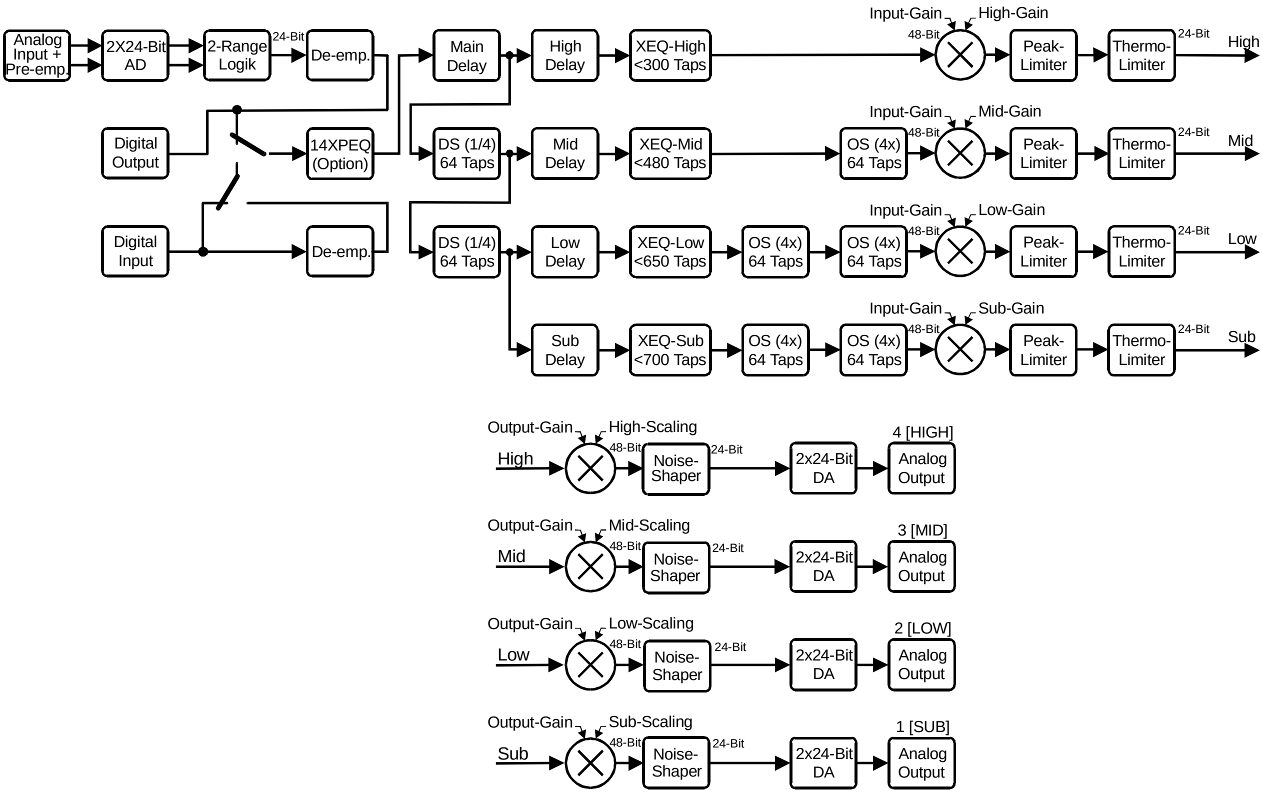
The GAE DSC28 concept not only allows for better efficiency in design and realisation of future system developments in loudspeaker technology. Even today the employment of the DSC28 with its FIR filter technology offers astounding advantages in the driving of existing systems.

The audio-quality of the signal-processing, determined mainly by the converters but also influenced by the analog-circuitry in the signal-path, is engineered to a maximum with regards to present technical possibilities, and this at an acceptable price/performance ratio.

The **GAE DIGITAL PA MASTER DSC28** offers state-of-the-art solutions not only to future sound reinforcement applications.

Block diagram and component description.

Fig. 11.1 The schematic diagram shows the complete signal flow of one channel of the DSC28.



Analog input. After passing through an analog pre-emphasis filter, the analog input leads to two different amplifier stages with different gain-rates, which drive both inputs of the 24-Bit Stereo AD-converter.

Dual-range-converting. In the digital domain of the Signal Processing (SP) block, the DSP switches and adapts both AD-channels, to complete the dual-range AD conversion, reproducing the analog input-signal in ("real") 24-Bit resolution. Following this the De-emphasis filter is calculated and the digital-signal is available for further processing. In this way a dynamic range of 127dB is achieved, which is much higher than that available from the newest generation of 24-Bit single-converters.

Digital in/output. The 24-Bit-AES/EBU-formatted digital output is permanently fed with the AD-converted data stream. As well as via the analog inputs, the audio signals can be entered directly by means of an AES/EBU-digital input. If requested by the digital-signal flow's attendant files, a De-emphasis-Filter can be introduced. The combination of in- and output can additionally be used as a digital insert. The device to be linked should be capable of 24-Bit-processing, otherwise a lamentable digression of the Dual-Range-Converter's prime values will be the result.

Parametric Equaliser (PEQ). The following 2 X 14-band parametric equaliser requires an additional Motorola DSP56009/81. In order to compensate boosting-filters, the function EQ-Gain is integrated to reduce the input signal-level. An additional limiter-step at the PEQ's output prevents digital overflow and a noise-shaper attenuates quantization faults of this full 48-bit operating EQ when re-quantisising to 24-bit.

Down-sampling. Of the following four signal-paths only the designated high-frequency path requires the maximum bandwidth for data-processing, allowing the sampling-rate of the remaining three paths to be reduced by down-sampling by the factors 4 (MID) and 16 (SUB and LOW). Keeping the number of filter-coefficients constant, the reduction of the sampling rate enables the increase of the length of the filter by the same factor and at the same time, dramatically reduces the amount of computing power necessary.

Down-sampling is carried out by means of 2 FIR low-pass filters with 64 taps. After low-pass filtering, the signal with the reduced sampling-rate can be subjected to further processing. For this application the filter-characteristic has been contrived for maximum rejection-band-damping (>120dB) so as to prevent disturbances due to aliasing effects.

Delays. A master-delay for the adjustment of the entire system-delay is situated prior to the crossover-network, whilst a further delay stage is situated in each of the output paths providing time-align compensation between the individual speaker components and forming part of the correcting filter network. The basic delay of the DSC28 is approx. 5ms, effected by A/D-, D/A- conversion, down-/over-sampling and the pre-viewing limiter concept. All further delays originate from the signal processing within filters and depend particularly on the run-time behaviour of the loudspeaker-system, filter slopes and frequency limits. The controller's entire basic delay-time for the loaded preset is displayed in the Master-Delay menu.

Crossover-network and equaliser (XEQ). Next on the signal path are the actual crossover-networks and equalisers for the individual paths (SUB, LOW, MID, HIGH). A processing-power of 80 Mips, made available to the controller's XEQ-filters by two Motorola DSP56009/81-processors for both channels, is intelligently distributed between the individual paths. Considering the respective sampling rate the resulting values are listed in the table below.

Path-no.	Path-name	Down-sampling Factor	Length of filter, taps	Length of Filter Taps (eff.)	Sample frequency kHz	Frequency limit (approx.) kHz	Frequency resolution, Hz
1	SUB	16	700	11 200	2.756	1	3.9
2	LOW	16	700	11 200	2.756	1	3.9
3	MID	4	480	1 920	11.025	4	23
4	HIGH	1	300	300	44.100	20	147

Over-sampler. After XEQ-filtering the sampling rate can be returned to it's original value by means of one, respectively two, 4x over-samplers. This process takes place in the reversed order to down-sampling. For the paths LOW and SUB two 4x over-samplers are connected one after the other. Analog to the down-samplers, the low-pass filters are equipped with 64 taps and are optimised for maximum rejection-band-damping.

Intrinsic frequency response. Ripples within the transmission-range of the down- and over-sampling filters, as well as the amplitude- and phase-response of the controller's analog and converter modules are taken into consideration and are inversely re-compensated during the generation of the system-dependant filter-coefficients.

Input-Gain. The following Input-Gain function has a wide-level adjustment range of -83...+45dB and can therefore be used as a volume-control. In the first instance this function enables the adaptation of the feed sources to the PA-system connected to the controller's output ports.

Output-gain (Paths). The next function is the gain-function of the individual paths with a level adjustment-range of -18...+6dB. With the 0dB-position as reference, the individual levels of the actively driven loudspeaker-groups, particularly within the bass-range, can be balanced to each other according to personal preference. The sensitivity of the individual loudspeakers varying within their SPL-range have already been inversely taken into consideration by scaling during the calculation of the filter-coefficients. (🕮 Chap.14)

Limiter-system. Two limiter-functions per path follow. A pre-viewing operation-mode involving a short delay of 1.5ms has been realised for the peak-limiter whereas a thermo-limiter protects the loudspeakers against thermal overstrain. During the creation of the XEQ-filter-coefficients each transducer is matched exactly to the performance specifications of the connected power amplifier. The protection circuitry within the controller intervenes according to this data. (🕮 Chap.15)

Multi-path. The so-called multi-path-variation of the controller is not shown in the block diagram. Two, respectively max. three paths of the controller can be summed post-limiters and made available at a common output. This allows the opportunity to create correcting-filters even for loudspeaker-systems with passive cross-overs. The XEQs and the frequency-range limiter of the single passive drivers are individually configured. Finally the single edited paths are summed together, led to a further peak-limiter-level and made available at one of the outputs. Possible multi-path-variations and their assigned outputs are as follows:

LOW + MID → 3 [MID]
MID + HIGH → 4 [HIGH]
LOW + MID + HIGH → 4 [HIGH]

Presets. A preset consists of a parameter-set configured by the loudspeaker-system manufacturer, which is selected as a whole and activated for the system by means of the controller. The DSC28 preset consists of:

- system-correction in amplitude and phase under consideration of sound-pressure and -power values of the loudspeaker components as well as the controller's intrinsic response. The equaliser is calculated in respect of a target function (e.g. linear from Freq. X to Freq. Y).
- a band-pass filter-structure based on the cross-over function of the individual paths including the dispersion behaviour within the cross-over range as well as the performance-data of the loudspeaker components.
- delay compensation (time-alignment) of the PA's loudspeaker components, whose acoustical focal points lie on different vertical lines.
- limiter-thresholds related to the performance-specifications of the connected loudspeaker components and power amplifiers.
- output-scaling-factor, which guarantees an optimal balance between the analog default-level amplification of the DSC28, the amplification factor of the connected power amplifier, the limiter-thresholds and the highest possible digital-resolution of the signal-processing.

Output-gain. Before being passed to the DA-converter, the signal can, if necessary be reduced in common-level (all four paths) by the output-gain within a range of 0...-24dB. This function operates like a volume-regulator, but can be better interpreted as a 'Safe'-regulator, as it's location after the limiters enables the setting of the maximum possible output-power of the PA-system.

DA-converter. A State-of-the-art, 24-bit Stereo, DA-converter of the Delta-Sigma-type is incorporated into each path and switched in parallel. The resulting dynamic range is ≥114dB. A 2nd-order low-pass provides the reconstruction of the digital data-flow of over-sampled converter-output signals.

Analog-output. The output-amplifiers of the DSC28 are electronically balanced and of low-impedance. Connection-wise they can be regarded as a transformer output, i.e. the output-signal only flows between the two active output-poles (🕮 Chap. 16). At a maximum output-level of 18dBu the output can be loaded with ≥ 375Ω. The result, remarkable for a digital device, is an output-dynamic value of ≥114dB. The matching of the different amplifier-values is performed in three maximal possible output-level steps (18, 12, and 6dBu) and receive fine-adjustment during XEQ-filter scaling. As such, the superior dynamic values and the particularly high resolution of the digital-signal at the DA-converter-inputs are maintained even when connecting to high-power amplification.

Gain architecture and dynamics

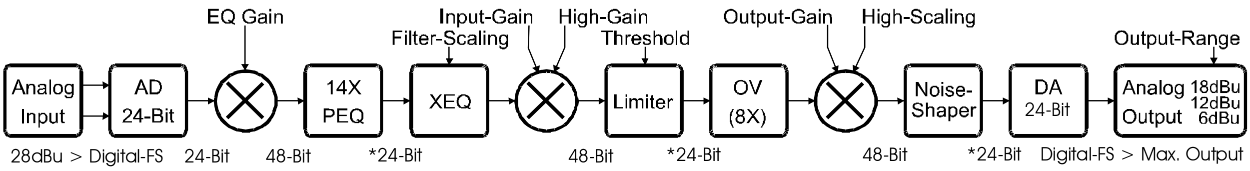


Fig. 12.1 Gain architecture. Fig.6.1 is an extract of the block-diagram and shows the signal-flow using the example of the HIGH-path. Only modules wholly or partly relevant to gain are considered. (The * at *24 bit indicates: reduction from 48-bit to 24 bit by means of a noise-shaper with 1st Order-Error-Feedback).

Dual-range-AD. The introduction of a dual-range-AD-converter enables the input-dynamic to be increased by approximately 17dB. The basic idea of the dual-range-AD is that two separate AD-converters reproduce the input-signal simultaneously with different pre-amplification. In the case of an overloading of the amplified channel a subsequent "intelligent" switch, within the digital-range, switches to the less sensitive channel. To guarantee a high signal-processing quality, the characteristic differences of the two identical (but not ideal) converter modules against the set-point deviating results of the analog amplifiers with different amplification factors as well as the DC-offsets and general long term stability have to be taken into consideration. This takes place during an additional adjustment of these parameters. Introduced to the analog input adjustment of the converters is a pre-emphasis-filter causing a treble-boost with a standardised frequency response, denoted by the two time-constants 50µs and 15µs. This pre-distortion is reversed by a de-emphasis-filter during the digital signal-processing. This treble boost affects the converters' set noise, which was not increased by the pre-emphasis-filter. Comparing the noise-level of a white-noise signal before and after the de-emphasis-filter, a difference of 5.5dB is seen which corresponds to the dynamic-gain, independent of using the dual-range-principle, won with this procedure,. The increase to the high frequencies results in a reduction of the range of high-level adjustment (Max. 28dBu - 10dB @20kHz) however this represents no problem to musical signals within the normal spectral range.

PEQ. Additionally available to the user is a parametric equaliser with 14 bands per channel, situated prior to the crossover-network. These are of the ('analog') IIR-filter type and are adjustable via the user-interface. The following filter-types can be selected: Bell, Low-Shelving 6 and 12dB/octave, High-Shelving 6 and 12dB/octave, Low-Pass 6 and 12dB/octave, High-Pass 6 and 12dB/octave. The preceding EQ-gain function is designed to attenuate the input-signal to compensate a possible boosting by the filters. The signal leaves the gain-stage with 48-bit-accuracy. The entire PEQ calculation including the filter-coefficients is processed at this accuracy. A limiter is introduced to the output of the PEQ to prevent possible overflow. With the help of a noise-shaper with Error-Feedback attenuating any quantization faults (distortions and noise) of this module, the signal is reduced back to 24-bit accuracy.

High- and low-pass filter-types should only be used for additional bandwidth limitation on the lowest and highest end of the total transmission range, however, where possible, their employment is discouraged. A complete equaliser-crossover-network, adjusted to the working range of the connected PA system, has already been configured within the employed presets via XEQ. Bell-type filters are used to compensate for tonal discoloration caused by room-architecture and acoustics. Firstly we recommend however, especially compliant to the bass-range, the balancing of the individual outputs by means of the output-gain controls.

Attention should be made to a restrictive particularity, which appears when using shelving-filters. When a high-shelving-filter adjusted with 12dB/octave, limit frequency of 5kHz and a Q-factor of 0.7 is set, the user-interface accepts no gain-setting higher than +5dB. A higher value can only be obtained by reduction of the Q-factor or by increasing the threshold-frequency. These restrictions are set by the device as the calculation of filter-coefficients for a higher boost-level would exceed the DSPs' (Digital Signal Processors) value range and as such are not available.

XEQ. (📖 Chap.14) The influence of the XEQ-filter on the gain-structure is characterised by two features. Firstly the differences of efficiency of a PA-system's individual sound-transducers, which leads to a relative scaling of the filters while authoring the coefficients. Secondly, the highest appearing peak of the amplitude-frequency-response of a part-filter (path) requires the further general scaling of all filters to the threshold of the digital system (Digital-Full-Scale).

Limiter. (📖 Chap.15) An overshoot of the maximum-rating of the output-signal is detected by the peak-limiters which reduce the amplification over a pre-determined time with optimally adjusted time-constants. Over-shooting peaks are automatically reduced to the exact value determined by the threshold-level. This procedure enables the exact adjustment of the limiter-thresholds to the absolute maximum-rating of the amplifiers, and the connected loudspeakers. This requires however an exact adjustment of all subsequent modules to these values. The reference values for the limiters are the performance-specifications of the loudspeaker components and the power amplifiers. The RMS-limiter protects the loudspeaker components in a similar way, referring however to the absolute maximum-thermal-rating.

DA-server and -converter. The DA-converter transforms the digital signal-flow with as little dynamic-loss as possible into an equivalent analog-signal. Unfortunately the DA-converters are the weakest elements of the signal-chain and therefore determine the output-noise-level. For this reason they are integrated into a digital/analog gain-structure. The function output-scaling provides an optimal digital driving of the converters (≡ Digital Full Scale). It is included within the generation of the XEQ-filter-coefficients. The reference figures for the output-scaling are the amplifying-factors of the power amplifiers and the analog output-driver-stages of the DSC28 (≡ Output Range).

Analog-output. The adjustment to different amplifier-ratings is performed here in three output-gain-stages of maximum possible output-levels of 18, 12 and 6dBu. The noise-level caused by the DA-converter as main noise-source, is correspondingly reduced.

Gain-regulator. The function of the gain-regulators is resolved from their operating-range and their position within the signal-chain.

- **EQ-gain.** This function decreases the input signal to compensate filter boosting. Adjustment range 0...-24dB.
- **Input-gain.** This function has a wide-range level-adjustment of -83...+45dB and as such can be used as a volume-control. It is however, intended to allow the matching of the driving source-signal to the PA-system connected to the controller outputs, particularly when using the analog-inputs. Due to the high driving-level of the analog-inputs (≤ 28dBu) an adjustment value of around 30dB for a +4dB-system is considered normal.
- **Path-gain.** This function, represented here by high-gain, allows any necessary, slight level-adjustments of the individual paths in relation to each other. The adjustment range is -18...+6dB. Placed ahead of the limiters this function is not(!) able to perform level-adjustment for amplifiers with differing amplification factors than those originally assigned.
- **Output-gain.** This function allows the parallel decrease of the output-levels of all paths after the limiters. As such it is possible to reduce the possible maximum-output-level of the PA-system and so rightly earns the title of "Safe"-control. The adjustment range is 0...-24dB.

Dynamics. The following noise- and dynamic-values are stated unweighted, the measurement bandwidth is 22Hz...22kHz. The high input-dynamic of 130dB at an input-level of 28dBu is realised using dual-range-conversion at the input, as well as the implementation of analog pre-emphasis/digital de-emphasis filters. Input-dynamic is defined as the ratio between the driving limit of the non-amplified AD-channel and the noise-level of the amplified channel. The DA-converters dominate the transient-dynamics of the device with their low signal-to-noise-ratio of 114dB, whereby a Dynamic-overflow of 130dB - 114dB = 16dB is generated which is added to the limiter-function as additionally usable headroom. The digital signal-processing is adjusted to these values in such a way as to always ensure that it operates with sufficient accuracy (48-bit instead of 24-bit accuracy; Implementation of the Noise-Shaper with 1st order error feedback during reduction from 48-bit to 24-bit). As such the digital re-quantization-noise is always lower than the noise-level of the converters and other analog modules. The great advantage of noise-shaper implementation is the reduction of distortion, appearing when amplifying low signal-levels. This improves the controller's tonal behaviour. Analog controllers offer dynamic-ranges of around 120dB, in relation to a usual maximum output-level of 26dBu. The resulting noise-level is -94dB whereas the DSC28 is comparable value of -96dB is reached at an output-level of 18dBu. The reduction of the output-level by adjustment to a connected amplifier leads to a corresponding decrease of the noise-level.

Amplifier-adaptation. As already mentioned, the adjustment of the DSC28's output-stages to match the amplifier to be connected, is performed by the digital/analog gain-structure. An example of mis-adjustment shows the advantages of this procedure. A power amplifier provides the requested power for the connected loudspeaker-component at an input-voltage of 0dB. The necessary adjustment of the DSC28's output-level is not carried out and, to adjust, the controller's output-scaling must be set at -18dB. In the following table these values are compared to the values at "correct" adjustment. The result is a worsening of the output-noise-level and the dynamic-range by 18dB, but also a worsening of the digital resolution by 3bit, the value corresponding to 18dB. Therefore the converter may only be driven with 21-bit, to ensure that the power-amplifier-output is not overdriven.

Adjustment	Output-Scaling	Output-Level (max)	Output-Level (eff.)	Noise-level (DA)	Noise-level (Out)	Resolution (digital)	Output-dynamic
correct	0dB	0dBu	0dB	-96dBu	-114dBu	24-Bit	114dB
incorrect	-18dB	18dBu	0dB	-96dBu	-96dBu	21-Bit	96dB

Together with the function output-scaling and the parallel switching of a stereo DA-converter per path, the matching to the amplifying-factor of the connected power-amplifiers provide to date (January 2001) an unsurpassed, excellent dynamic-value and moreover, a constant high-resolution of the digital signal-flow at the DA-converter inputs of the controller. Due to the step-adjustment of the output-range parameter in three 6dB-steps, a mis-adjustment is still possible, but only to a maximum of 6dB.

Exchanging an amplifier. When exchanging an amplifier, as well as ensuring identical power-specifications it is important that amplification-factors are also equivalent, otherwise an adjustment of the two gain-factors output-scaling and output-range is inevitable. The value for output-scaling is part of the coefficient-set of the XEQ-filters and as such, is an integral component of the preset. The output-range value is set by a four-position jumper within the analog area of the output-circuitry. Further more it must be considered that the limiter-threshold values, also components of the preset, have to be in an exact defined relationship to both output values and that, by this balance, the output-bargraph-chains represent the exact driving-state of the whole PA-system. If this balance is disturbed by the replacement of one or more amplifiers with different power-specifications and/or amplifying-factors in several paths of the PA-system, as well as tonal and other disadvantages, the safety of the loudspeaker-components is endangered by the implementation of wrong limiters, especially when the amplification-factors are higher than the original ones. If necessary a simultaneous adjustment to higher amplification-levels can be carried out by the function output-gain, provided that this corresponds to all paths at the same time. An adjustment to amplifiers with lower amplification-factors however is not possible, and as such, the proper (guaranteed) capacity of the loudspeaker-system can not be fully realised.

💣 **The replacement of power-amplifiers to a PA-system may only be performed without further consideration, when power-specification and amplification-factors are absolutely identical; otherwise encasing losses in tonal responce and loudspeaker-component safety have to be taken into account.**

Run-time behaviour

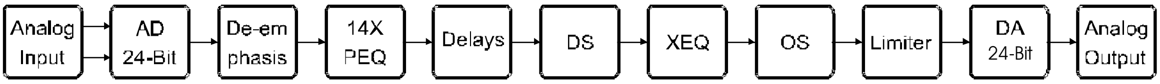


Fig. 13.1 Run-time behaviour. Fig. 13.1 shows the signal flow as an extract from the block-diagram, using one of the three lower-frequency-paths as an example. Only modules influencing the run-time behaviour are considered.

Run-time. Every digital controller adds run-time to the signal passing through the signal-processing ICs, causing an intrinsic-delay which is inevitably higher than that caused by an analog device. This applies also to the digital device which is simply simulating an analog device. The signal is delayed when leaving the digital device, even if within the examined audio frequency-range, no signal modulation is performed, i.e. the signal is "simply" AD-converted, passes through the DSP and is finally DA-converted. The minimum run-time in the chain of AD-converter (no dual-range-principle), DSP-chip and DA-converter, required by all digital controllers is approx. 1.5ms. Of this time 98.5% is caused by the work of the two converters. Further digital functions within the DSC28, e.g. down- and over-sampler, the realisation of the dual-range-principle and the limiters require additional run-times. Efforts to reach better signal-quality within the digital domain have to be offset against the ensuing delay-times. The intention is to optimise theses times in order to keep them as short as possible. The run-time of a digital-controller does not necessarily need to have a constant value, and the sum of the run-times in parts of the DSC28's signal-flow is frequency-dependent. Subsequently the delay-causing modules of the DSC28 and their basic run-time are divided into 3 interdependent groups.

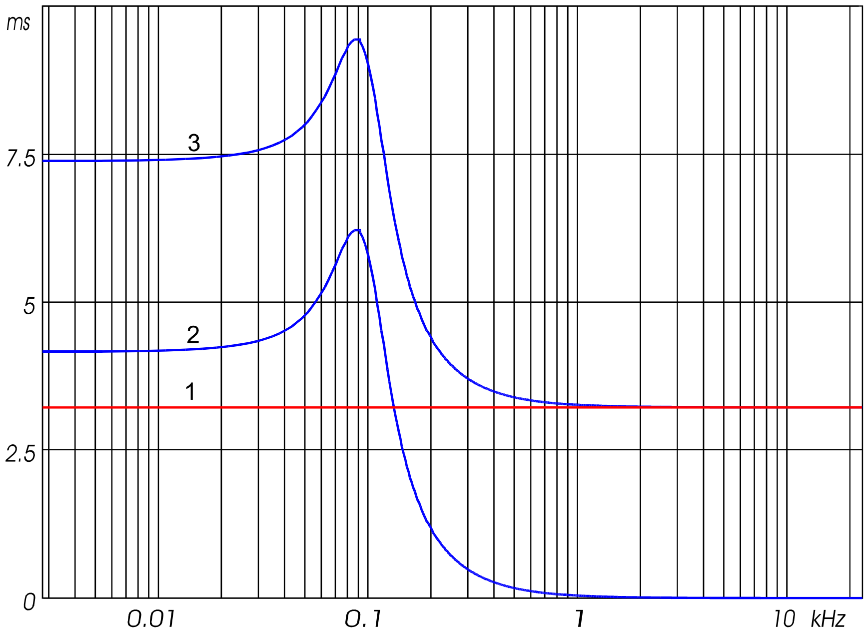


Fig. 13.2 Basic run-time behaviour. Curve 1, Linear-phase. All passing frequencies sustain the same delay-time. (A digital delay device shows such a behaviour). Curve 2, Minimum-phase. The run-time is dependent on frequency; it runs towards zero above the frequency-threshold (an example of minimum-phase objects are analog-filters and speaker-components). The run-time shown represents a Butterworth low- or high-pass, 24dB/octave, 100Hz cut-off frequency. Curve 3, Combination. The shown run-time could belong to a digital controller, which calculates the Butterworth-filter by using the linear-phase portion of curve 1.

The run-time-frequency-response (group-delay) is generally favoured to the phase-frequency-response as the phase-representation at wide frequency ranges and higher run-times is not interpretable. At a delay-time of 5ms and a frequency of 1kHz the phase amounts to 1800°, at 10kHz it amounts to 18000°. The different run-time behaviours are denoted by the abbreviations L-Ph for linear-phase, M-Ph for minimum-phase and L/M-Ph for combination-phase.

1. Intrinsic run-time behaviour. The run-time behaviour of the necessary modules for the function and operating-principles of the controller which build the basic requirement for the use of further modules:

Analog-input	High- and low-pass, pre-emphasis, anti-alias	M-Ph
AD-converter	Conversion, decimation, dual-range-principle	L-Ph
De-emphasis	inverse to pre-emphasis results	L-Ph
Down-sampler	decrease of sampling rates	M-Ph
Over-sampler	increase of sampling rates	M-Ph
DA-server	Interpolation	L-Ph
DA-converter	Conversion	L-Ph
Analog output	Reconstruction	M-Ph

2. Run-time behaviour of the equaliser. The run-time behaviour of the important modules for the actual task of system-equalising:

Alignment	Run-time compensation of loudspeaker-components	L-Ph
XEQ	Band-passes and system equaliser	L-Ph...L/M-Ph...M-Ph
Limiter	Pre-viewing limiter-concept	L-Ph

3. Additional run-times. The run-time behaviour of modules, that can be influenced by the user-interface:

PEQ	Equaliser (dependant on configuration)	M-Ph
Master-Delay	Additional run-time for all paths	L-Ph
Path-Delay	Additional run-time for single paths	L-Ph

Basic run-time. In summary it can be stated that a basic run-time is present in each of the controller's 4 paths. Because of the different down-/over-sampling values within the individual paths, a deviating linear-phase behaviour with minimum-phase segments results. As well as the amplitude-frequency-response, the run-time portion of the controller's intrinsic-behaviour has to be considered when creating the system's equalising-network. The total basic run-time of the DSC28 is strongly dependant on the type of system-equalisation employed and as such dependant on the loudspeaker-system which is to be connected. The linear-phase segment for an activated preset is represented within the menu Master Delay. The additional group 3 run-times, generated by the user, are not added to the total basic run-time.

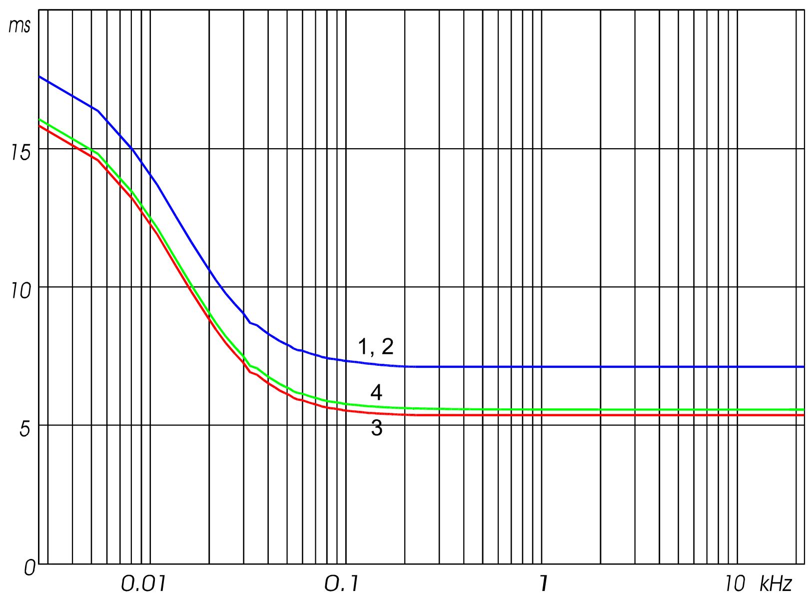


Fig. 13.3 Intrinsic run-time behaviour. The basic run-time of the four paths is represented by the values of the linear-phase segment of the delays, including the delay of the "pre-viewing" limiters (1.5ms): SUB + LOW: 7.2ms, MID: 5.4ms, HIGH: 5.6ms. The minimum-phase segment and the run-time increase towards lower frequencies and are caused by the 1st order high-pass filter within the analog-input module (Coupling-capacitor).

Presets

Presets. A DSC28-preset consists of a set of parameters configured by the loudspeaker-system manufacturer. The preset can be selected and loaded in the controller which then runs the system. To create a preset the following parameters have to be taken into account:

System equalisation. The interpreted and if necessary, edited acoustic pressure- and acoustic-power measurements of the speaker components, the known intrinsic response of the controller, as well as a preset target-function for the desired resulting frequency-response of the loudspeaker-combination, are the default parameters for the equalisation of the amplitude. Differences in sensitivity of the loudspeaker components are considered. (Filter-scaling)

Band-pass structure. The band-pass structure is derived with the help of target-band-passes with attention to the crossover behaviour of the loudspeaker components to each other, including their dispersion behaviour within the crossover-range, as well as their power-specifications.

Run-time behaviour. The use of FIR-filter technology allows the run-time behaviour of the crossover to be separated from the amplitude-frequency-response. A run-time compensation (time-alignment) between speaker components with acoustical centres on different vertical axis' is included.

Limiter. The limiter-thresholds are selected respective of the power-specifications of the connected loudspeaker components and power amplifiers (🕮 Chap.15).

Output-scaling. Output-scaling-factors are an integral part of the parameter-set. These guarantee an optimum balance between the amplification of the analog output-level of the DSC28, the amplification-factor of the connected power amplifier, the limiter-thresholds and the highest possible digital-resolution of the signal-processing.

FIR-filter technology. The calculation of FIR-filters (Finite-Impulse-Response) requires significantly more computer processing-power when compared to digital devices operating with IIR-filters (Infinite-Impulse-response) and as such merely simulating analog filters within the digital domain. For this reason the available processing-power is optimally matched to each of the paths/frequency-ranges (except the HIGH-path) by down-/over-sampling-modules. However, this process requires that a bandwidth-limitation of the useful frequency range of the SUB- and LOW -paths to approximately 1kHz and the MID-path to approximately 4kHz be implemented. FIR-filtering allows the realisation of correction-filters with linear-phase behaviour, which has the benefit of equal delay to all frequency segments of a signal. Inconvenient is that linear-phase behaviour down to the lowest frequencies of the transmission-range causes extended run-time which is unacceptable in live-performance situations. This can be compensated by attenuation to minimum-phase ("analog") response below a specific frequency. For the filtering of an active system with moderate crossover slopes (<36dB/octave), minimum-phase response up to 120Hz and linear-phase response ≥120Hz, a controller delay of 30ms has to be considered: intrinsic controller delay of approximately 7ms + filter-delay of approximately 23ms.

Parameter-set construction. In combination with the measurement system MF from the ITA (Institut für technische Akustik) at the RWTH (Rheinisch-Westfälische Technische Hochschule, Aachen) which includes software for parameter-set construction, the DSC28 serves the loudspeaker system-developer as a universal tool for creating XEQ-filter and protection functions. As such it is a direct constituent in the development of a loudspeaker system. The completed parameter-sets of the XEQ-filters can be written into the Flash-RAM of the controller by means of the RS-232- or MIDI-interface. The result can immediately be evaluated by measurement and hearing assessments. On completion of system-development several parameter-sets, e.g. for different stacking-variations, can be written into the Flash-RAM. Further sets can be added via the RS-232 interface. The user can choose between these parameter-sets in the preset menu.

Example. The following seven illustrations convey an impression of the results of system-equalisation using the 3-way GAE DIRECTOR system. The chosen target-function is merely a superlative possibility of this loudspeaker system. The presets of the DSC28-devices, that are shipped with GAE DIRECTOR-systems, include a minimum-phase system-equalisation to a linear target-function up to approximately 1kHz. Above 1kHz the delay-time behaviour is linear-phase. The Minimum-phase portions, which remain only very slight down to 200Hz, can be seen in Figure 14.7.

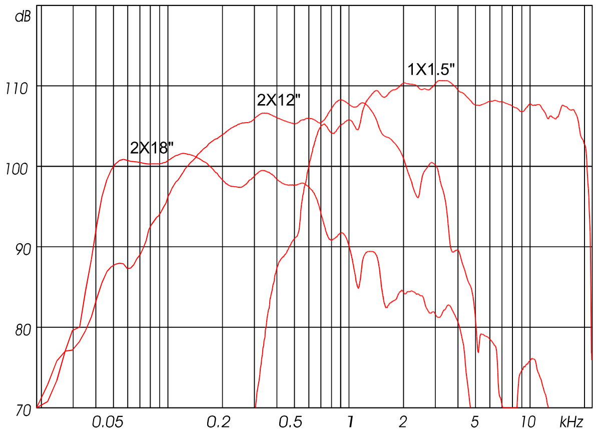


Figure 14.1 Loudspeaker-system, single measurements of the components, sensitivity @ 1W, 1m

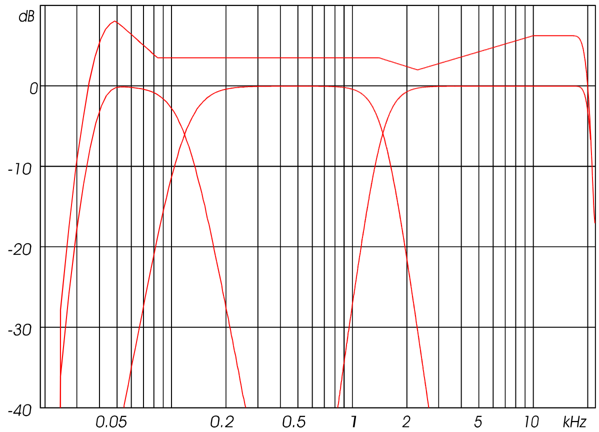


Figure 14.2 Target-functions, system (weighted) and band-passes

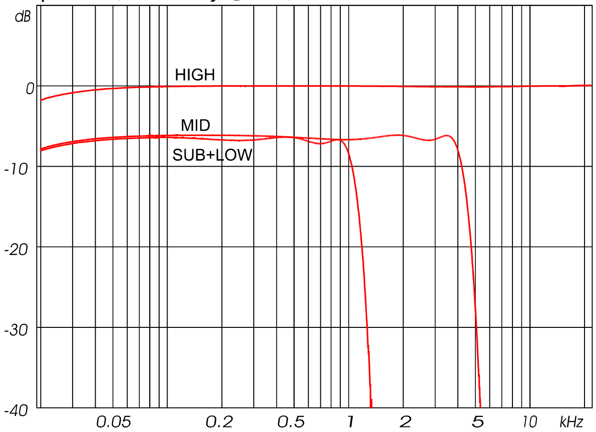


Figure 14.3 Controller-intrinsic-response, amplitude

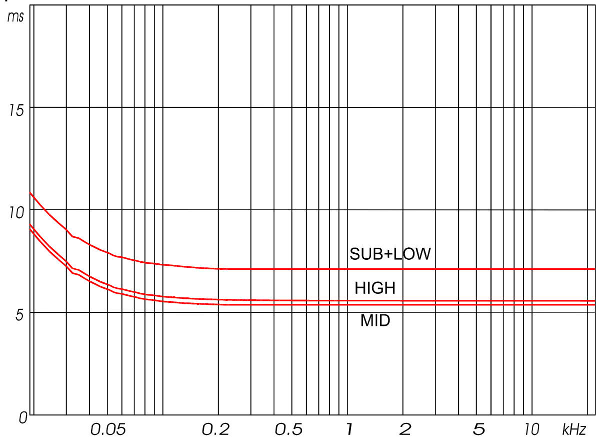


Figure 14.4 Controller-intrinsic response, run-time

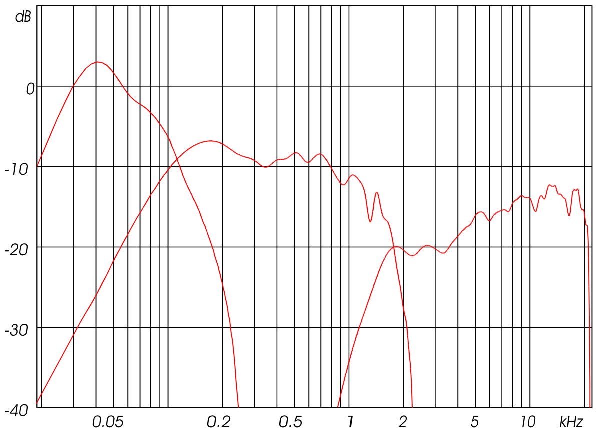


Figure 14.5 Controller-outputs, level

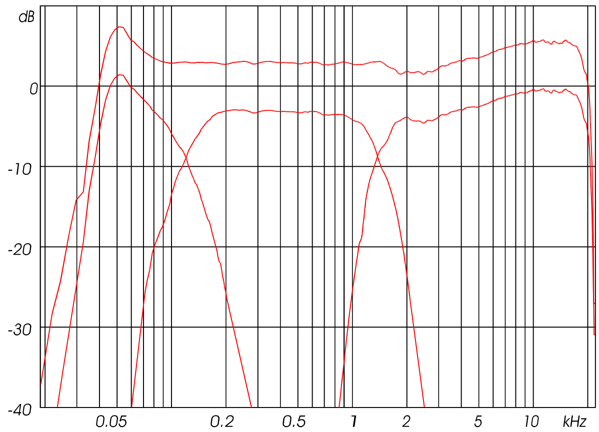


Figure 14.6 Equalised system, total frequency response, single frequency responses

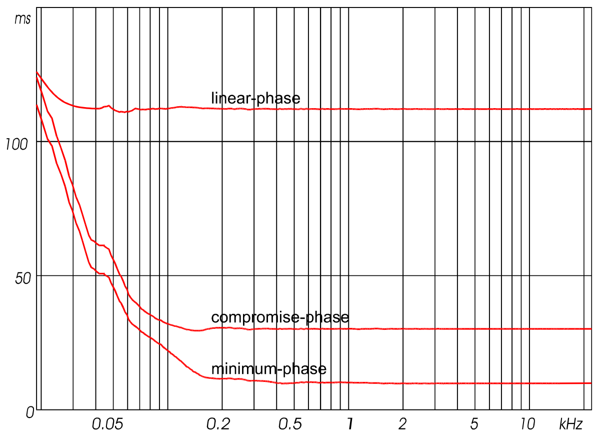


Figure 14.7 Equalised system, 3 examples of total delay response

In a minimum-phase system the run-time increases strongly to low frequencies which is also the case in analog controllers. If a correction of the run-time to linear-phase-behaviour is required, the delays of the higher frequencies have to be increased to that of the longest run-time (at the lowest frequency). The result is a total-signal-delay and for live-situations unacceptable. A compromise-correction allows linear-phase behaviour to the lower frequencies, which converts to minimum-phase behaviour below a cut-off frequency of 100Hz.

Limiter

All components of a signal-processing chain have a limited dynamic range, which is restricted by noise at the lower end and by the operating limit at the higher end. Analog signal-processing components can be cost-effectively dimensioned with sufficient headroom, to make sure that no clip-distortions can occur. In contrast, such an over-dimensioning of a PA-system (power amplifiers, loudspeakers) is hardly economic and as such, the operation of a PA-system within it's critical range is the normal situation. An exceeding of the limits can not only lead to high distortions, but even to the destruction of the components. Consequently the surveillance and control of the signal level belongs to the elementary functions of a signal controller. Only by this means can a PA-system be reliably operated within the admissible load-range. The module described as limiter is the last component before the power amplifiers within the signal-processing chain. It is able to effectively intercede, particularly in active multi-path systems, due to its assigned magnitudes for each path being individually matched to the driven speaker component.
The necessary level-attenuation carried out by the limiter is based on two different types of loudspeaker-overloading.

Peak-limiter. The peak-limiter prevents mechanical overload caused by too high acceleration forces, resulting in excessive material stress, the disruption of membrane and cones in tweeters and cone-drivers, the destruction of the voice-coil caused by hitting the pole plate (mainly cone-loudspeakers) or the breakage of the connecting wires of the voice-coil. Furthermore the task of suppression of disturbing distortions, produced by the loudspeaker or by the clipping of the power amplifier is undertaken.
The basis for the loudspeakers admissible load is it’s peak load-capacity. The characteristic variables of a peak-limiter are the threshold, at which the level-reduction starts, as well as three time-constants: the attack-time, at which the speed of the level reduction is defined, the release-time, which defines the speed of releasing the gain reduction and the hold-time, which defines the period over which the level-reduction is held after the threshold-excess is ended.

RMS-limiter. RMS- or thermo-limiters prevent thermal overload, particularly the burning of voice-coils caused by constant too high a power supply. The basis for the loudspeakers admissible load are the thermal permanent-load capacity and two time-constants: a short one for the thermal-capacity of the voice-coil and a long one for the larger thermal-capacity of the magnet-material and the loudspeaker-chassis. The characteristic variables of an RMS-limiter are the threshold, at which level-limitation starts and the emulation of the loudspeaker's thermal-time-constants.

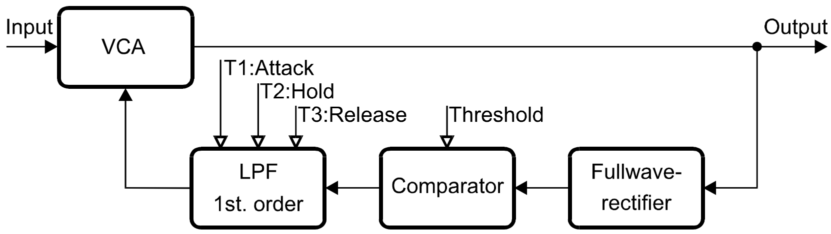


Fig. 15.1 Analog peak-limiter

'Analog' peak-limiter. The block diagram above (Fig. 15.1) shows a peak-limiter circuit, found in most analog limiter devices. A VCA (Voltage-Controlled-Amplifier) controls the adjustment of the level-reduction. The necessary control voltage for the VCA is determined by comparing the default threshold with the rectified output voltage. If the output signal exceeds the threshold, the current increase at the comparator-output is passed to the VCA-control-input by means of an RC-circuit, whose time-constant defines the attack-time (T1). Should the threshold-exceeding voltage no longer be present, the amplification is released back to the output-value depending on the release-time-constant (T3). Before this, the level reduction determined by the attack is held for the period of the hold-time (T2). The hold-function is missing in most of the commonly used analog limiter-systems.

Problematic is, that the comparator's erratic output voltage, smoothed only by a low-pass of 1st-order, corresponds directly to the VCA-control voltage. The only weakly-attenuated high-frequency portions of the control-signal are directly multiplied by the wanted signal and as such, produce additional distortion which leads to an audible "crackle" when using critical source-material with little high frequency content. Low-pass-filters of a higher order would cause too long a run-time delay and therefore a limiter-response which is too slow. Even the low-pass of "only" 1st-order has the effect that, directly after an amplitude-jump the limiter is not able to react fast enough causing an inevitable short-term excess (overshoot) of the default threshold.

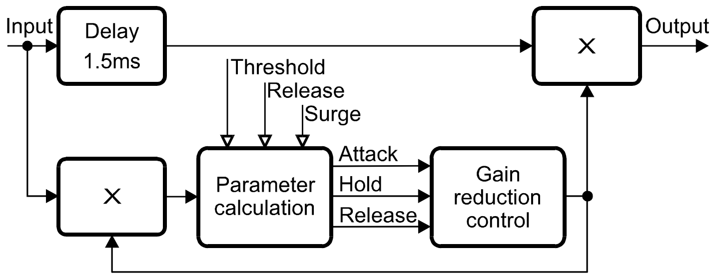


Figure 15.2 Pre-viewing digital peak-limiter

Pre-viewing peak-limiter. Digital signal-processing also allows extended possibilities for the limiter-concept. The figure 15.2 shows the principle of the DSC28's peak-limiter. Here a multiplication-block corresponds to the VCA of an analog limiter. The peak-limiters of the DSC28 work with "pre-viewing" signal-analysis. This allows an "in-advance" registration of impending threshold-excess values and the reduction of the level over a set period with optimally adjusted time-constants. In this process the peak is attenuated to the exact threshold-level, a controlled overshoot of the value is allowed. This procedure allows the limiter threshold to be set at the exact value of the connected amplifiers/loudspeakers. Even sudden jumps in signal-level no longer cause overloading. For strong transient-impulses the effect of pre-masking can additionally be utilised to ensure that the control-time before the impulse is not perceptible. A subsequent hold-time prevents level-oscillations in the directly subsequent signal-paths. The pre-viewing signal-analysis is enabled by means of a 1.5ms delay within the signal path. The control of the gain-reduction defines in which operating phase the limiter is in: attack-, hold- or release-phase. An attack-operation is started when the input-level exceeds the limiter-threshold. The threshold is an element in the digital/analog gain-structure of the amplifier-adjustment and as such, an integral part of a preset, not adjustable by the user. Depending on the degree of threshold-excess, the resulting action is the amplification-reduction over a fixed time-period with a time-constant adjusted to the peak in dB/s. Should a further peak emerge during the attack-operation, the time-constant is re-calculated. The subsequent hold-phase of 20ms is re-started, should during this phase, the input-signal again rise to within 1dB of the limiter-threshold. Is neither limiter- nor hold-threshold reached, the amplification is restored to it's original value dependant of the release-time-constant. The release-time-constant in dB/s is the only user-adjustable limiter parameter.

Controlled overshoot. Conventional power amplifiers differ mainly in circuitry-details, through the use of MOSFET- or BIPOLAR- transistors as well as by different approaches to power-supplies. Conventional power-supplies are unstabilised transformer/rectifier/capacitor-concepts. Modern, switched-mode power-supplies, which are built-up with small, ferrite core-transformers may be either stabilised or unstabilised, whereby the introduction of PFC (Power Factor Control; sinusoidal mains-current-pull) to this type of power-supply ensures that they are always stabilised. When loading a power-supply with current flow to the loudspeakers the amplifier-supply-current inevitably sinks. The amplifier-design determines how much and at which rate the supply-voltage drops when full output-power is suddenly demanded. Here a compromise between highest available-impulse-power and continuous output-stability is necessary. Considering the signal-statistics of common program-material which an amplifier has to deal with, it seems illogical to construct the power-supply as "hard" as possible i.e. with infinitely small internal-resistance, as this leads to an unnecessary high power-loss to the amplifier.
Very helpful during the analysis of an amplifiers signal-statistic is the so-called crest-factor. Usually stated in dB the crest-factor relates the peak-power of a signal to its average performance. A pure square-wave modulation has therefore a crest factor of 0dB, whilst a sine-wave signal amounts to 3dB. With uncompressed wide-band music and speech signals it lies in excess of 10dB.
After the signal-separation to different frequency ranges and following strong compression by the output-limiters in active multi-path systems, the smallest crest-factors can be observed within the bass-range. The amplifiers for this frequency- range must not only provide the highest output-power, but must also have the "hardest" power-supply i.e. with minimum internal-resistance, compared to the amplifiers for the other paths. Even with the roughest compression to "unnatural" signals of non-musical nature does the crest-factor hardly ever sink below 6dB within the bass-channel. This is only a difference of 3dB when compared to a pure sine-wave signal. Therefore the supply-voltage should be able to "collapse" by this amount without causing the amplifier any difficulty in terms of permanent power-capacity at defined peak power and with standard

source-material. Most power-amplifier manufacturers design their power-supplies to be stable in terms of permanent-load-capacity has a sign of quality. Contrary to this however, are the concepts providing extreme relationships between impulse- and permanent-load-capacity which is without doubt, efficient for use within the Mid-/HF-range.

If the threshold of the peak-limiter coincides with the amplifiers permanent-load-capacity, short impulses will be limited to this capacity, even though the amplifier might easily be able to reproduce them without compression. Clip-distortion, resulting from the short-term-requirement of the maximal available impulse power by signal-peaks are either not discernible or - e.g. when hitting the bass-drum - even lead to the desired 'kick'-sound. Consequently, according to the "hardness" of the power-supply, significant power-reserves remain unused. At a peak-/permanent-load-capacity ratio (DHR = Dynamic headroom) of 3dB, half of the amplifiers impulse-power lies neglected. For this reason a further duty of the pre-viewing peak-limiter is the increase of the limiter-threshold value with a time-constant feedback according to the ability of the amplifier (Surge). Unfortunately most amplifier-specification sheets have no information regarding power-supply behaviour and as such must be estimated for limiter-programming purposes.

Connection hints

Due to it's self-explanatory nature, an illustration of the device's rear-panel has been dispensed with. The connection-ports and respective important notices are explained below:

ANALOG-INPUTS

Electronically balanced (🕮 Chap.11/1), source-loading: 20kΩ

CHANNEL A - [LEFT] and CHANNEL B - [RIGHT]

3-pin-female-XLR 1: Shield 2: + Phase 3: - Phase

ANALOG-OUTPUTS

Electronically balanced (🕮 Chap.11/3), internal-resistance: 20Ω, maximum load-capacity: ≥ 375Ω @ 18dBu maximum output-level. The DSC28-outputs can be considered, connection-wise as transformer-outputs.

CHANNEL A [LEFT] and CHANNEL B [RIGHT]

4-paths 1 - [SUB] 2 - [LOW] 3 - [MID] 4 - [HIGH]

3-pin-male-XLR 1: Shield 2: + Phase 3: - Phase

AES/EBU-INPUT

Transformer-balanced, source-loading: 110Ω

3-pin-female-XLR 1: Shield 2: + Phase 3: - Phase

AES/EBU-OUTPUT

Transformer-balanced, internal resistance: 110Ω.

3-Pol-Male-XLR 1: Shield 2: + Phase 3: - Phase

MIDI/RS232-INPUT

Consolidated input due to the respective hardware-conventions. Due to the twofold occupation of this socket the MIDI-Input does is not conform to the standard, as pin 2 , necessary for the RS232-connection, is not connected.

5-pin-DIN 1: RS232-Tx 2: GND 3: RS232-Rx 4: MIDI + 5: MIDI -

MIDI-OUTPUT

Output conform to the MIDI-convention standard.

5-Pol-DIN 1: free (NC) 2: GND 3: free (NC) 4: MIDI + 5: MIDI -

REMOTE ON

This socket facilitates the remote-on function by means of an ac- or dc-voltage of between approx. 12...24V. The remote-on function has to be released before this function is available. (🕮 Chap.19)

FUSE T1A

💣 Mains fuses cannot prevent an unexpected malfunction of electrical components, rather they should protect the user and its environment from damage. For this reason never try to substitute the mains fuse by any other than the specified M1A type (1A, medium slow behaviour). Never try to repair or bypass a blown mains fuse.

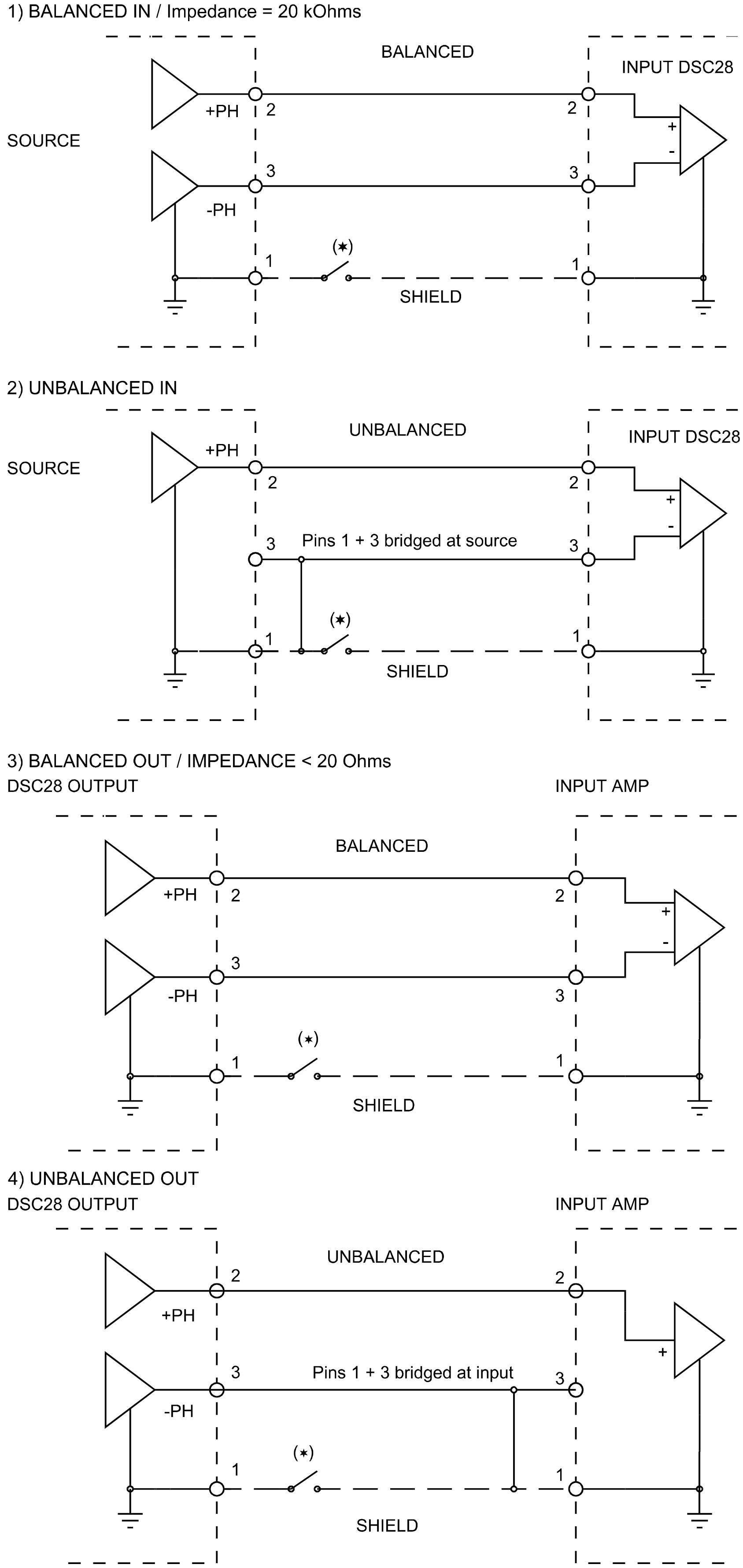


Fig.16.1 Connection-hints

(✱) Eventually necesary for the cancellation of ground-loops.
The active output of the DSC28 can be considered as a transformer-output.
Only use 2-core, shielded LF-cable!

Re-loading of presets, initialise

Memory structure. A Flash-ROM (read-memory only) and a battery-buffered RAM (read-/write-memory) each hold the operating-systems and -parameters of the controller. Because the data-safety of an ROM is essentially higher than that of a RAM, this two-memory-system ensures that, even in the case of a data-loss in the RAM the full functionality of the controller can quickly be restored. A maximum of two presets as well as the first-parameters are always available from the ROM. The lithium backup-battery for the RAM has a manufacturer guaranteed minimum life-expectancy of 10 years and as well as this is over-dimensioned in capacity. The following diagram lists the memory structure:

EPROM
First-Parameters
µC-Program
2 Presets
DSP-Program

RAM
The first-parameters copied from the EPROM converted to user-parameters
8 - 12 presets can be down-loaded via the RS232-interface

Initialise. In case of a data-loss in the RAM, parameters adjusted by the user, as well as possible down-loaded presets are lost. Defect parameters are overwritten with the first-parameters in the initialisation process, lost presets have to be re-loaded. Should the scheduled self-initialisation of the operating-system not be performed (indicated by an error message and the command ‘Press Enter’), one of the following initialisation possibilities is provided. If the initialisation is not successful, the reason is a different or further failure. In this case the device should be returned to the manufacturer for maintenance.

1. Follow the directions in 🕮 Chap.9 ● Menus under SHIFT-SYS, System adjustments ● SYS-4: Initialise. Should the data-defect disturb the operation of the device, proceed as in 2.
2. Remove the device from the mains-supply, press the ENTER-key, now reconnect the device to the mains and only release the ENTER-key if the boot-display is substituted by the Input-Gain-menu SET-0. This procedure does not last longer than 4 seconds.

Updating the operating system. An update of the DSC28 operating software can be loaded to the Flash-ROM. Due to data/operating safety of GAE-systems this software includes at least one preset so that in the event of an eventual RAM-data loss, the continued operation of a connected GAE system can be maintained after initialising the device. Due to the fact that each preset contains information about the users power amplifiers, the complete operating system is dependant on the amplifier configuration.

Re-loading of presets. A further 8-12 presets can be loaded to the battery-puffered RAM. The exact number depends on the size of the blocks. After a successful transmission the transfered presets are available in the Preset-List and can be recalled via the menu SET-7 (🕮 Chap.7/3). Immediately after initiation and during the whole procedure of preset-block transmission the message ‘Appending Module..." is shown in the top line of the controller's VFD-display. If transmission fails, check that the RS232-interface of the DSC28 is active. (🕮 Chap.9/1 ● SYS-1/1: MIDI-Settings ● Baud rate)

The supplied Sub-D9/DIN-5-pin-cable establishes the RS232 connection to the serial interface (COM#) of a PC. For GAE-Systeme, the necessary files in the form of a self-extracting ZIP-file can be requested from the manufacturer.

Activating the remote on/off switching

The socket REMOTE ON facilitates the remote on/off-switching of the device by means of an auxiliary-voltage (ac- or dc-voltage of between approx. 12...24V). To activate this function proceed as follows:

- First remove the controller from the mains-supply, then open the cover.
- To avoid damage caused by electrostatic-charge, touch the device's casing and at the same time an earthed object such as the earth-contact of a Schuko-socket outlet, the metal case of another, still connected device or a central-heating radiator..
- Please refer to figure 18.1: Open the marked contact.
- Close the cover and only then reconnect the device to the mains-supply. Examine the function of the remote on/off-switching. Turning on the auxiliary-voltage switches on the controller.

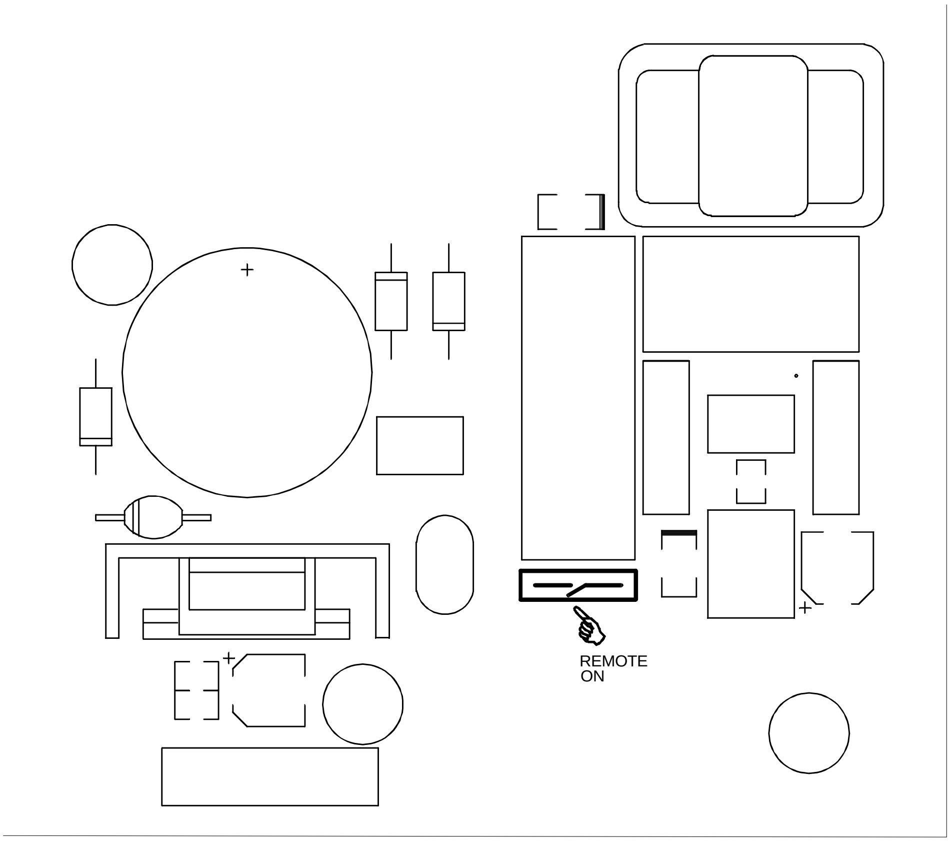


Fig. 18.1 Contact for remote on/off-switching

Parameter-control for several devices

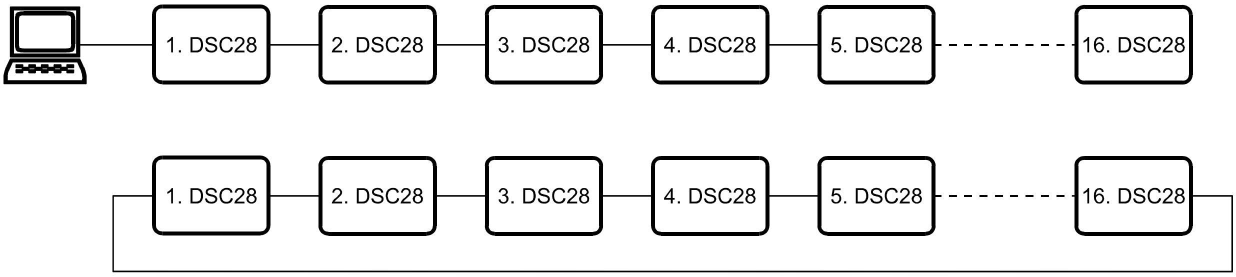


Fig. 19.1 Driving-chain / -loop

Several controllers can be linked in a chain or a loop via the MIDI-in- and outputs, to enable e.g. the adjustment of all the connected devices from a single device. Within such a chain the first device may also be a PC.

Perform the necessary adjustment within the menu SYS MIDI-Settings. The list below is a summary of the MIDI-Settings. (🕮 Chap.9/1 ● SYS-1 ● MIDI-Settings).

█ **1, Channel: 1 ○➘➘ 16.** Assign the controller with ist transmit/receive channel (basic channels 1-16), with which data can sent and received.
⊙ **Enter.** Confirm your selection with ENTER!

█ **2, Out: OutOnly ○➘ Out/Thr ○➘ Loop.** OutOnly is a pure MIDI-output. Out/Thr is a MIDI-output with an additional Soft-Thru-function, which passes the signals received at the MIDI-input to the MIDI-output. Loop is a MIDI-output which passes-on all incoming data on channels other than the devices addressed channel. This mode links several DSC28-controllers in a closed MIDI-loop.

█ **3, Baudrate: MIDI (31250) ↙○ RS-232 (9600).** Use RS-232 and the supplied cable to remote control the first device with the help of a PC via one of it's serial-interfaces (COM#). Note that the RS-232-connection cannot bridge long distances. Use MIDI, to link several DSC28-controllers.
⊙ **Enter.** Confirm your selection with ENTER!

█ **4, ParaChg: Off ○➘ On .** If this value is set to On, the device sends all performed parameter-adjustments out. As such several device (Slaves) can be commonly operated by one device (Master).
⊙ **Enter.** Confirm your selection with ENTER!

█ **5, TX-Chn: 1 ○➘➘ 16 ○➘ OCM.** The transmitting channel for parameter-adjustments can be selected by this parameter. As well as channels 1-16 an Omni-Channel-Mode (OCM) can be selected. Parameter adjustments via this channel are accepted by the connected devices independent of their channel-address.
⊙ **Enter.** Confirm your selection with ENTER!

RS-232-Kabel. The supplied RS-232-connection-cabel is wired as follows:

Pin No.	SUB-D9- 2	DIN5- 1	RS-232-Tx
Pin No.	SUB-D9- 5	DIN5- 2	RS-232-GND
Pin No.	SUB-D9- 3	DIN5- 3	RS-232-Rx

MIDI-cable. MIDI-DIN5-pin-standard-cable has following wiring:

Pin No.	1: not connected	2: GND	3: not connected	4: MIDI +	5: MIDI -

Due to it's twofold occupation the RS-232/MIDI-Input socket does not conform to the standard, as pin 2 (RS-232-GND), necessary for the RS-232-connection, is not connected.

Remote control

A PC remote-control-software is not available.

Technical Specifications

Analog inputs.............electronically balanced, input impedance 20kΩ
maximum input level: 28dBu @ < 1kHz, 18dBu @ 20kHz
noisefloor: -100dBu; dynamic range 127dB
(all measurements linear rated 22Hz - 22kHz)

Converting...................per channel a Dual-Range-24-Bit Delta-Sigma AD-Converter with pre-emphasis, 50/15µs; sample rate: 44,1kHz

Analog outputs........... electronically balanced, output impedance 20Ω
maximum output level: +18dBu into 375Ω; noisefloor: ≤ -96dBu;
dynamic range ≥ 114dB (all measures linear rated 22Hz - 22kHz)
totally THD+N: < 0,005% at operating limit
output-range-steps: 18, 12, and 6dBu maximum output level

Converting...................per output a 24-Bit Stereo-Delta-Sigma converter in parallel mode

Digital in/out............... input: 24-Bit AES/EBU with/without pre-emphasis, sample rate converter 32kHz...96kHz
output: 24-Bit AES/EBU, sample rate: 44.1kHz
also useable as digital insert (AD⇒DigOut ; DigIn⇒Controller)

Remote control........... input: RS-232 and MIDI, output: MIDI; baud rate: 9600 and 31250

Intrinsic run-time.........incl. down-/over-sampling, AD-/DA-converter and limiter: 5...7ms, path dependant

XEQ-filter..................... FIR-filter, group delay dependant on chosen presets

PEQ-filter..................... IIR-filter (quasi analog behaviour)

Limiter......................... <u>peak-limiter</u> with 1.5ms pre-view and controlled-overshoot to fully utilise the PA-impulse reserves, 48-Bit signal processing, precision-adjustment and extremely low distortion
<u>rms-limiter</u> with voice-coil- and magnet temperature-modelling

Operating.................... foil keyboard with eight tip-keys, incremental-dial with additional ENTER-function

VF-display................... 2 lines, 24-character-vacuum-fluorescent-display, blue

LED-display................

2 X input (-30 ...+24dBu) each 10 X LED green	+ 2 X clip (≥ +28dBu) + each 1 X LED red
8 X output (-30 ... 0dBFS + 0 ...12dBGR) each 7 X LED green + 3 X red	+ 8 X mute + each 1 X LED rot

Power supply............. universal-input 85...265V, < 30VA, over-voltage-protection, safety fuse M1A, M = medium, switched-power-supply is self-protected.

Dimensions................. 19“ / 1U, 260mm (10.25“) depth

Weight......................... 3.75kg (net without packaging)

Chapter 22

Addresses, bibliography, miscellaneous

Literature, studies and dissertations referring to the subject:

- Entwicklung eines Signalprozessors zur Entzerrung von Lautsprechern
  A. Goertz
  Diplomarbeit am ITA, RWTH Aachen 1990
- Entzerrung von Lautsprechern mit einem Signal-Prozessor-System in Echtzeit
  A. Goertz ; D. Leckschat
  Tagungsband DAGA 1990
- Digitale Lautsprecherentzerrung
  A. *Goertz ; D. Leckschat*
  Tagungsband VDT 1990
- Nichtlineare Entzerrung von Lautsprechern in Echtzeit mit einem Signalprozessor
  A. Goertz ; W. Klippel ; D. Leckschat
  Tagungsband DAGA 1991
- Verbesserung der Wiedergabequalität von Lautsprechern mit Hilfe von Digitalfiltern
  D. Leckschat
  Dissertation am ITA, RWTH Aachen 1992
- Vergleich verschiedener Verfahren zur digitalen Lautsprecherentzerrung
  A. Goertz
  Tagungsband DAGA 1992
- Aufbau und Inbetriebnahme einer digitalen Frequenzweiche mit einem Signalprozessorsystem
  S. Müller
  Diplomarbeit am ITA, RWTH Aachen 1992
- Einsatz digitaler Limiter in Beschallungsanlagen
  A.Goertz ; S.Müller
  Tagungsband DAGA 1995
- Klangeigenschaften digital entzerrter Lautsprecher
  K. H. Pflaum
  Diplomarbeit am ITA, RWTH Aachen 1995
- Optimierung der Entzerrerübertragungsfunktion für Lautsprechersysteme durch Berücksichtigung psychoakustischer Effekte
  A. Goertz ; K.H. Pflaum
  Tagungsband DAGA 1996
- Aufbau eines erweiterten digitalen Controllers für Lautsprecher
  J. Kleber
  Diplomarbeit am ITA, RWTH Aachen 1996
- Programmierung eines digitalen Equalisers auf der Basis von FIR-Filtern
  Norbert Blissenbach
  Studienarbeit am ITA, RWTH Aachen 1996
- Verbesserung der Wiedergabequalität von Lautsprechern mit Hilfe von Digitalfiltern
  D.Leckschat
  Dissertation an der RWTH Aachen 1992
- Untersuchung zur digitalen Lautsprecherentzerrung mit Hilfe binauraler Meßtechnik
  P. Niggemann
  Diplomarbeit am ITA, RWTH Aachen 1997
- Entwicklung und Erprobung einer digitalen parametrischen Filterbank
  R. Thaden
  Diplomarbeit am ITA, RWTH Aachen 1997
- Digitale Signalverarbeitung für Lautsprecher
  S. Müller
  Dissertation am ITA, RWTH Aachen 1999

Consulting + MF-Meßsystem:

Audio + Acoustics Consulting
Dr. Ing. Anselm Goertz
In der Linen 21
D-52134 Herzogenrath
Fon 02407-565111
Fax 02407-565112
Mobil 0171-9338402
Email Anselm.Goertz@t-online.de

is a product of
opal audio vertrieb GmbH
Engerstraße 47, D-33824 Werther, Fon ++49-(0)5203-236/-7, Fax -238
Internet: http://www.gae.de , Email: info@gae.de