LAWO

NETWORKING
AUDIO
SYSTEMS

zirkon User Manual

Digital Broadcast Audio Mixing Console



Release: V 1.2
Issue: 2003-10-24

interstage
Phistersvej 31, 2900 Hellerup, Danmark
Telefon 3946 0000, fax 3946 0040
www.interstage.dk
- pro audio with a smile

Copyright

All rights reserved. Permission to reprint or electronically reproduce any document or graphic in whole or in part for any reason is expressly prohibited, unless prior written consent is obtained from the Lawo AG.

All trademarks and registered trademarks belong to their respective owners. It cannot be guaranteed that all product names, products, trademarks, requisitions, regulations, guidelines, specifications and norms are free from trade mark rights of third parties.

All entries in this document have been thoroughly checked; however no guarantee for correctness can be given. Lawo AG cannot be held responsible for any misleading or incorrect information provided throughout this manual.

Lawo AG reserves the right to change specifications at any time without notice.

© Lawo AG, 2002, 2003

Contents

CONTENTS **3**
PREFACE **5**
Manual Structure 5
Typographical Markers 6
Conventions of the Notes 7
WARNING NOTES **9**
INTRODUCTION: THE ZIRKON SYSTEM **11**
Features of the zirkon system 11
Components of the zirkon system 12
System block diagramme 13
CONTROL ELEMENTS, WHAT IS WHERE? **15**
Channel module 15
Central module 17
OPERATING CONCEPT **19**
Basic operating concept 19
Channel module *19*
Central module *20*
Assigning a channel strip with sources 21
Summing bus assignment 22
Audiocontrol 23
Pre-amplification (GAIN) mono sources *24*
Pre-amplifying (GAIN) line sources stereo in general *26*
Equalizer *26*
Dynamics section (dyn) *28*
Limiter (lim) *31*
Storing of configurations, PFL 31
Storing a setup *32*
Loading a setup *32*
PFL mode *33*
Storing of snapshots *34*
Loading of snapshots *34*
Radio automation 35
MAINTENANCE AND SERVICE **37**
System page (http://) 37
DSP card information *38*
Sync Information *38*
Slot Information *38*
Panel Information *38*
Supplementary Information *38*

Control Panel Modules: Configuring and Testing 39
Channel module 950/41: 39
CAN SPEED *40*
FADER KAL *40*
TESTS *41*
POWER TEST *41*
KEY TEST *41*
SELF TEST *42*
Central module 950/42: 43
CAN SPEED *44*
POTI KAL *45*
KEY TEST *46*
SELF TEST *46*
POWER TEST *46*
Cleaning the operating surface 47
Replacing the lithium battery 47
Defective modules 47
ACCESSORIES **49**
Power supply unit for console modules 49
Redundant PSU for the system unit 49
19” Push button module (KS16) 49
GLOSSARY **51**

Preface

This manual is subdivided into individual, self-contained chapters that can be studied independently from each other.

Basically, every chapter is structured in a general introduction which is followed by topics that include description or instruction sequences.

Parallel descriptions or interrelations between the described elements or functions are cross-referenced.

Manual Structure

The manual structure is explained by the following brief descriptions of the individual chapters' contents.

Chapter 1 Warning Notices

Includes general warnings about the handling of low-voltage devices and first aid in case of electric shock.

Chapter 2 Introduction about the zirkon

Offers a brief overview over the features and the components as well as a block diagram that illustrates the system structure.

Chapter 3 Control Surface Modules - What is where?

Explains buttons, light emitting diodes (LEDs) and rotary controls of the surface modules.

Chapter 4 Operating Concept

Describes the operating elements and gives instructions for operation.

Chapter 5 Maintenance and Service

Shows how to obtain system information and how maintenance and cleaning is carried out; it also gives instructions for smaller tests.

Chapter 6 Accessories

Lists accessories e.g. a redundant power plug and high-quality LCD push-button strips.

Chapter 7 Glossary

Lists the used technical terms.

Typographical Markers

For easy handling and orientation with this manual at hand you will find here some explanations about the most important typographical markers.

Chapter

The headlines are marked by bold and large font. The headlines structure the topics to guarantee fast orientation.

1. Instruction

The instructions are listed with numbers. These numbers serve for showing you the reasonable sequence of actions. This way you have always an overview over the complete instruction sequence.

Result

The results are written in italic letters. They inform about intermediate as well as final results to give you the chance to identify errors at an early stage.

Marginal Note

Marginal Note

The marginal note serves as eye-catcher. Next to the marginal note you will find useful tips, notes or warnings. About warnings you will learn more in the paragraph conventions.

Conventions of the Notes

This symbol is followed by useful tips!



Tip

This symbol signifies that something is pointed out to you. Notes include useful suggestions or cross-references to records not included in this document.



Note

This symbol appears when something must be observed strongly. Not observing the warnings can cause heavy damages of the device and perilous bodily injuries. Please absolutely observe notes with this symbol.



Warning Notes

Attention:

Read all notes with this “attention” symbol before installing the device. Observe all these notes during installation. Please observe also all other operation and application notes.



Attention:

Check the device and the accessories with regard to transport damages.

A device damaged mechanically or by the intrusion of liquids must not be connected to the mains supply or must be disconnected from it immediately by pulling the power plug.



Attention:

The device must be grounded.

If the device is not operated with a mains power supply unit please use only such line cords with three-leads as provided with the delivered product.



Attention:

By removing parts of the casing, shields, etc., live parts are laid open. Therefore you must strictly observe the following safety instructions:



- Manipulations inside the device must be carried out by qualified service personell only under observance of the applicable regulations.
- Inside the device are no user-servicable parts.

Attention:

Before removing parts of the casing the device must be switched off and disconnected from the mains supply in any case.



- Before opening the device the power supply capacitor must be discharged with a suitable resistor.

- Components that carry heavy electrical loads like power transistors and resistors should not be touched until cooled off to avoid burns.

Attention:



Servicing unprotected and operating devices can be carried out only by instructed qualified personell at their own risk. The following instructions must be observed:

- Never touch bare wires or circuitry
- Use insulated tools only
- Do not touch metal semi-conductor casings as they can bear high voltages.

Attention:



First Aid (in case of electric shock):

- Separate the person as quickly as possible from the electric power source:
  - Switch off the equipment
  - Unplug or disconnect the mains cable
  - Push the person away from the power source by using dry insulating material (such as wood or plastic)

Attention:



If the person is unconscious:

- Check the pulse

Reanimate the person if respiration is poor.

- Lay the body down and turn it to one side (recovery position), call for a doctor immediately.

After having sustained an electric shock, always consult a doctor.

Attention:



Do not touch the person or his/her clothing before power is turned off, otherwise you stand the risk of sustaining an electric shock as well!

Introduction: the zirkon system

The zirkon system offers a solution for small and mid-sized broadcasters to operate fully digital studios. The components are also suited for remote studios and the operation in OB vans.

Features of the zirkon system

For these applications, the zirkon provides the following essential features:

- Fully modular design

  Allows favourable spares pooling and fast maintenance.

- Intuitive operation

  Also by non-trained staff.

- Digital and analogue interfaces

  Derive from high-quality mixing consoles; their number can be increased easily (up to system limits).

- Control panel size is scalable in 4-fader-steps.

- Flat control panel for fitting into a table.

- Signal processing in the system unit

  is performed in a 19'' system frame with 3RU or 6RU which is designed for installation in equipment rooms. This avoids noises by cooling and heat dissipation.

- The system unit can be operated also without control panel.

  In case of a control panel failure, no audio interruption is caused. After replacing the defect part, all operating elements and displays are restored. The work can continue trouble-free.

- Graphical PC surface

  A graphical PC user interface allows to programme the system very customer-specifically. Functions like talkback, redlight, studio switch and many other wishes can be realised.

- Motor faders are standard

  The motor faders allow the transparent operation in combination with a radio automation system (connection of automation via serial protocol, RS422).

- CAN bus connection

  Control panel and system unit are connected via CAN bus. 19'' 1RU button panels (KS16) can be also integrated.

- Simple installation

  The system is cabled via RJ45 plug-in connectors.

- 56/64-channel MADI interface

  The system can be extended by this interface. The interface is designed redundantly.

Components of the zirkon system

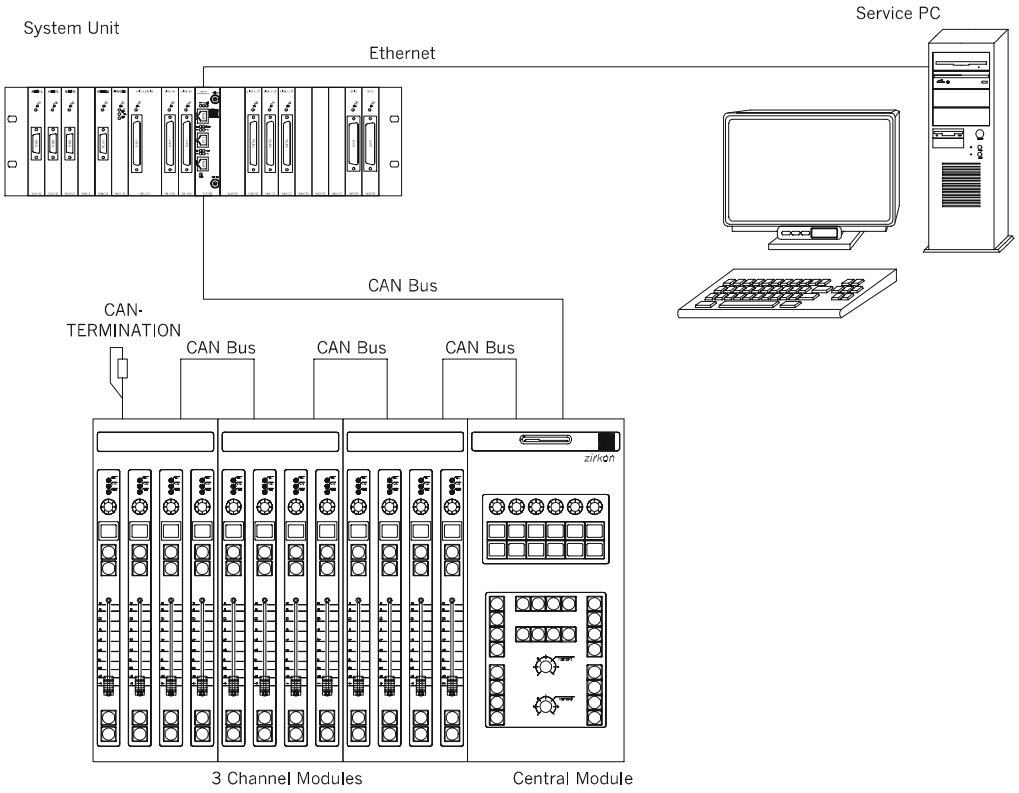
Fundamentally, a system consists of one System Unit and at least two control panel modules (Central Module and Channel Module).

The **system unit** includes the masterboard and a number of analogue and/or digital interfaces. The motherboard performs the signal processing. The interfaces serve for integrating the studio environment.

With the **central module** you operate monitoring functions, many programmable special functions and the signal processing of individual input signals. The **channel module** includes four channel strips. Each channel strip is fitted with one motor fader and five buttons. You can connect up to nine channel modules to the system.

System block diagramme

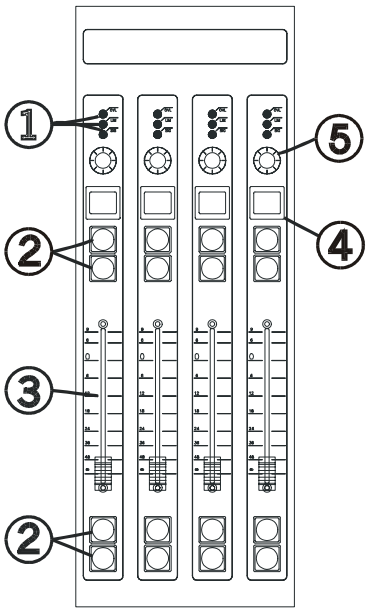
The block diagramme shows a simple zirkon system with system unit, central module and three channel modules. The service PC allows an easy configuration but is not needed for operating the console.



Control elements, what is where?

All control elements are positioned on the central module or the channel modules. Thus all functions are provided on the clearly designed control panel.

Channel module



All channel strips in the channel module are designed identically. This allows to assign any kind of signals (microphone/line/transmission line) to any channel strip.

(1) Three LEDs at the top indicate the signal status, thus providing a quick overview.

- You identify an overload when the top diode blinks or when it is on. In this case please adjust the input signal gain to not jeopardise the signal quality.
- The middle diode shows the activity of the integrated active limiter.
- The bottom lamp indicates whether an input signal is supplied. It lights up in the signal clock and has a responding threshold of -20dB.

(2) The operation of the four freely programmable buttons is defined by the selected operating modes. If you have questions about the configuration please ask your network administrator!

(3) The middle section of the channel strip carries the faders. These are motor faders fitted with tracks of conductive plastic. The faders are maintainance-free. The scale ranges from -∞ to +9.

(4) The respective channel is indicated by the self-labelling button (access button).

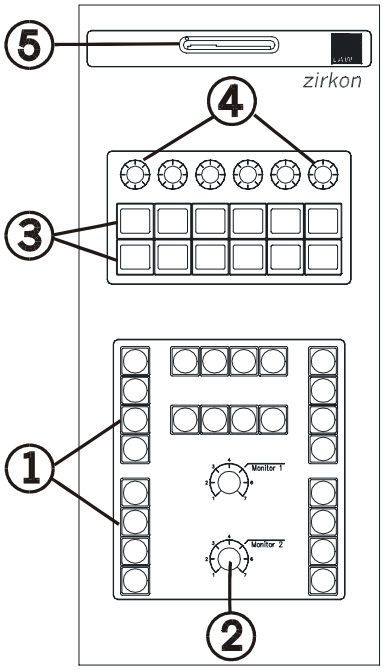
- The button label shows the source name (input signal) and the time. This is a big help in on-air scheduling, particularly for devices without a time display (recorders, microphones, record players, etc.).

(5) The rotary control above the access button serves for the setting of any parameter in the channel strip.

- When handling the rotary control the display of the access button changes. It shows no longer source name and time but the rotary control function and the set value instead.
- The rotary control is used for all sources to set the gain in the channel. This way, inputs with different volume can be adapted to the mixing console. For microphone channels, the gain before digital conversion can be set.

  Pushing the rotary control allows the setting of panorama/balance.

Central module



This control panel module can be integrated in every zirkon system only once. It includes buttons for operating monitoring, talkback equalizer and dynamics. Furthermore it is possible to programme buttons for the summing bus assignment. Established configurations can be managed with the built-in chipcard reader/writer.

(1) These 24 freely programmable buttons are ordered in six groups with four buttons each to establish functional groups. This allows to separate e. g. monitoring selection and talkback optically and with regard to position.

(2) The bottom section is equipped with two freely programmable volume controls. The two rotary controls can be programmed for setting the monitoring level as well as for functions like monitoring, headset volume, loudspeaker volume etc.

(3) The top section includes twelf self-labelling (access) buttons. In the default status these keys are assigned with the pre-configured functions. They can be freely programmed and allow e. g. the assignment of channels to summing buses or special operations (remote function with foldback).

- With selecting a channel via the access button of a channel module the buttons of the central module are labelled according to the source. (See chapter “Operating concept - audio control“)

(4) With the rotary controls positioned vertically above the access buttons you can set parameters offered according to the source.

- The settings are explained in detail in the chapter “Operating concept - audio control”).

(5) A chipcard reader/writer is integrated in the top section of the central module.

- With this reader/writer you can store your own settings and load them next time.

For more about using the chipcard please consult the respective chapter.

Operating concept

The operating concept of the zirkon system has two aims:

- Easy and confident operation of the on-air system also when under pressure - also by untrained staff
- Integration of customer-specific functions in the mixing console which is standardised to the greatest possible extent

These aims were achieved by doing without a complex channel-strip operation. Only those functions indispensable for a radio programme are positioned on the channel strip, all others are operated on the central module. This allowed a clear design of the channel strips in the channel modules which guarantees an easy access to the mixing console also for untrained users.

Basic operating concept

Channel module

Dominating operating elements of the zirkon are the faders. Lawo offers motor faders that make it easy to work with radio automation systems (SAS). When using a SAS the fader position indicates always the actual situation.



The 0dB position of the faders is clearly marked. The maximum position of the faders is at +9dB which matches the headroom of analogue and digital audio interfaces.

- Only use the range between 0dB and +9dB to adjust signals of a particularly low level to the other mixing console signals.

Attention:

When you simply open the respective fader to +9dB while mixing a radio programme there will be distortions!

The motorfaders allow to drive fully automated fades. This works via radio automation or a programmed START button. The user can always watch the actual fader position and can intervene if required.

With activating a programmed START button the fader drives immediately to the marked 0dB position. This guarantees a precise play-back of programme elements.

Central module



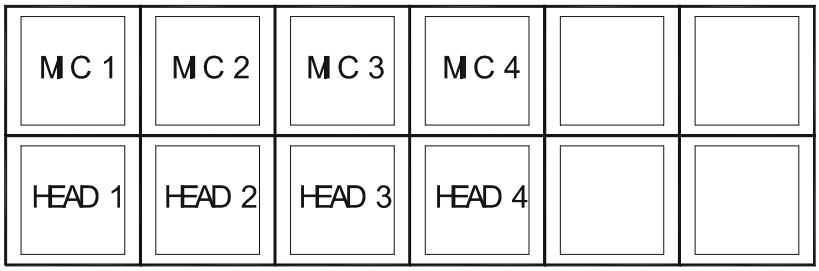
With the access button you release functions in the central module that are not supplied in the channel strip (**Button shows red**). These functions refer always to the currently selected channel strip. Press the access button again to deselect the channel strip.

Assigning a channel strip with sources

According to configuration, the channel strips are pre-assigned with signals/sources. If you want a different signal to be assigned to the channel strip please proceed as follows:

1. Keep the LCD button in the channel strip pressed (> 5 sec)!

After about 3-4 secs the LCD display of the central module changes. The display in the LCD control section shows all available sources (11 sources max.)



2. Press the LCD button with the required source!

The source is now assigned to the channel strip.

3. Releasing the button in the channel strip before selecting a new source establishes the previous status!

I cannot assign a new source! ?

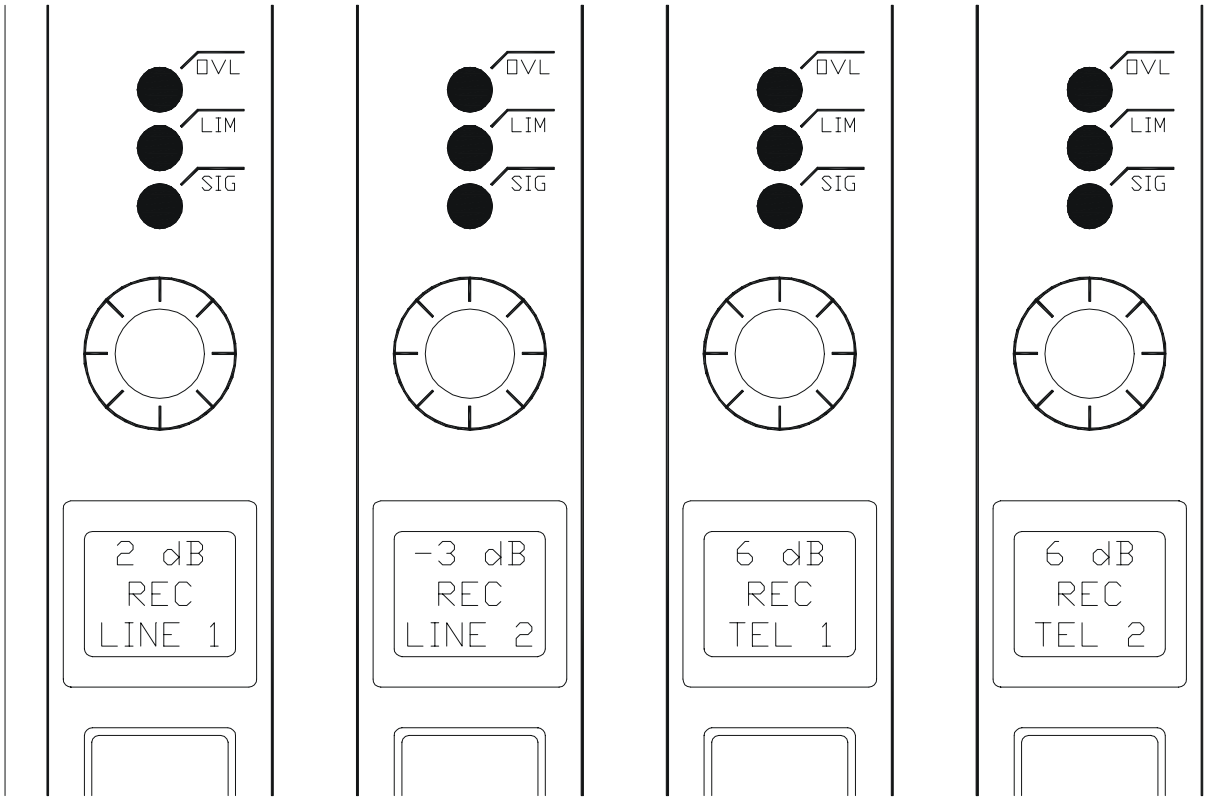
A channel strip is open or the source is already assigned!

➔ Close the fader of the channel strip and carry out the steps 1 and 2 again! For security reasons it is not possible to select a new source while the fader is open.

➔ Assign a source not yet assigned!

Summing bus assignment

Depending on how your zirkon system is programmed you can perform a summing bus assignment.



1. Press the summing definition button (e. g. AUX DEF/ MAIN DEF)!

You are now in the mode for the summing bus assignment.

2. Press the channel-strip LCD button of the source you want to select.

The source can be now configured for the summing bus.

3. Use the rotary control in the channel strip for setting the amplification (AUX SEND LEVEL)

The signal is now ready for assignment. The following signalling apply:

Green: not assigned (OFF)

Red: assignment pre fader (PRE FADER) –

The input signal is directly added to the respective sum after a possible processing (EQ/DYN/LIM) but without being affected by the fader position.

Yellow: assignment after fader (POST FADER).

Audiocontrol

After assigning a source to a channel strip you can now set the audio parameters.

1. Press the LCD button of the channel strip!

The access buttons in the central module change their labels. The following parameters for signal processing are available:

- Pre-amplification (GAIN)
- Equalizer (EQ)
- Compressor (COMP)
- Limiter (LIM)

2. Press the LCD labeled with the parameter you want to adjust!

The access button shows now the assigned parameters with the currently set value.

3. Turn the rotary control positioned above the LCD button!

The settings of the parameter change.

Note

Depending on which source you selected only certain parameters are available. For example, you can only adjust the analogue input amplification for analogue microphone channels.

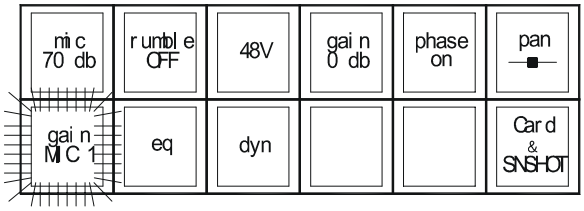
The following signalling rule applies for the audio control if not declared otherwise in the following text:

GREEN: ready for operation

RED: active

Pre-amplification (GAIN) mono sources

The button GAIN (bottom left on the button section) is special within the select-buttons for the signal processing: The bottom line shows in addition the name of the source accessed for being edited.



1. Press the button “gain”

The button shows red. The top button row is labelled as shown above. You can now set the parameter.

The first three buttons (MIC, RUMBLE und 48V) are only available for souces that include a **microphone amplifier**.

The rotary control positioned above the button MIC sets the input amplification before the analogue/digital conversion.

The setting range is –70dB to 0dB.

Here it is also possible to use microphone inputs for line signals. This is in parallel operation with the rotary control of the channel strip (for sources of the type microphone)

The highpass filter RUMBLE is operated the same way.

- The cut-off frequency is selected with a rotary control and can be set to 40, 80 or 140Hz.

2. Push the rotary control positioned above the button “RUMBLE”!

The button shows red. The high-pass filter is active on this channel.

The phantom power supply for condenser microphones is set the same way.

3. Press the rotary control positioned above the button 48V!

The button shows red. The phantom power is active.

The parameter GAIN adjusts, in contrast to MIC, the input gain digitally, i. e. after the analogue/digital conversion.

The setting range is -30 ... +9 dB.

With this parameter you can equalise different volume levels of the sources to the mixing console so that in the fader positions of 0dB all sources reach a more or less equal volume. The operation runs parallel with that of the rotary control in the channel strip.

4. Press the button GAIN!

The value is reset to the normal 0dB-position.

The function PHASE cannot be set via the rotary control.

5. Press the button “phase”!

For mono sources a phase reverse of 180° is carried out. For stereo sources only the phase of the right channel is reversed by 180°!

This function allows to correct a wrong cabling or setting up special microphone arrangements. The normal status of this function is OFF.

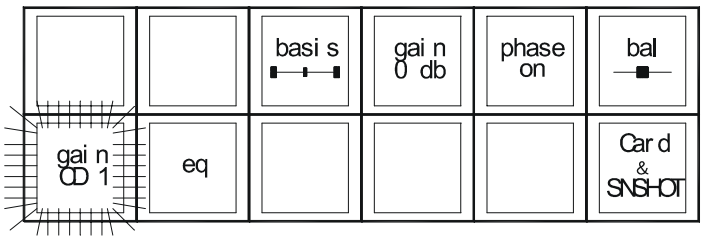
The rightmost button is labeled according to the type of source either with balance BAL (stereo) or panorama PAN (mono).

6. Turn the rotary control positions above the button "pan” resp. “bal”!

The channel strip is positioned in the stereo sum.

Pre-amplifying (GAIN) line sources stereo in general

The selection for stereo sources is similar to that of the mono sources. One difference lies here in the function BALANCE (bal) instead of PANORAMA (pan).



In addition there exists here the possibility to adjust the stereo base width (basis).

1. Turn the rotary control positioned above the button “basis”!

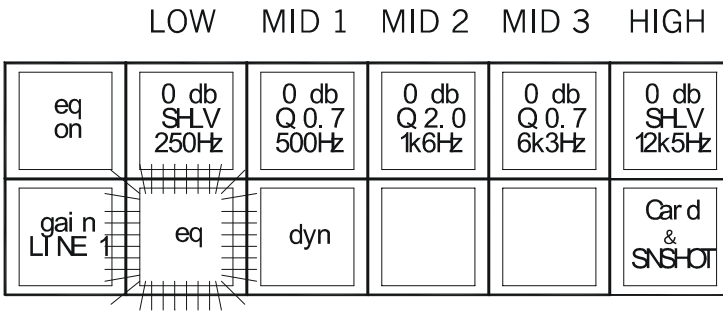
The stereo base width changes.

Equalizer

If an equalizer is programmed for the selected source or sum the audio control shows the button “eq”.

1. Push the button “eq”!

All parameters appear in the top row (see illustration).



How to activate the equalizer:

2. Press the button EQ ON.

The equalizer is activated within a source.

The individual parameters are operated as follows:

3. Rotate the rotary control to change the amplification.

4. Push and turn the rotary control to change the frequency.

This is signalled by an inverse display of the frequency parameter.

5. Press the access buttons (MID 1-3) to switch the function of the rotary control over to the settings of the Q factor.

Here, the parameter is also displayed inverse. Pushing the LCD button again re-establishes the basic status.

6. With LOW and HIGH you alternate between the filter modes!
   LOW: HP – SHLV /HIGH: LP – SHLV

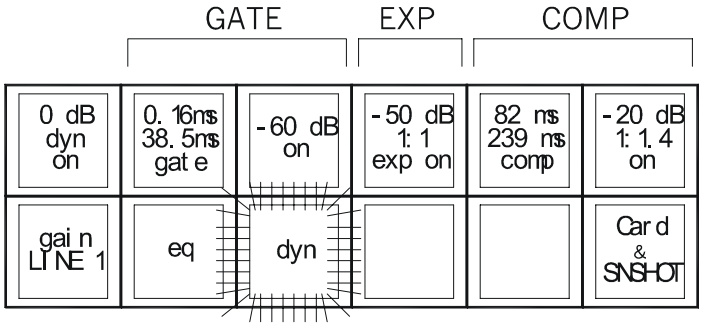
As the modes LP and HP are pure filter functions, only frequencies can be set.

Note

In the equalizer section the source format is irrelevant as for stereo sources the parameters are automatically coupled.

Dynamics section (dyn)

A dynamics section may be integrated for a source or sum as further sound designing module. The dynamics section includes a gate, an expander and a compressor.



1. Press the button “DYN ON”!

The dynamics section is active.

The individual sound designing modules (gate, exp und comp) must be switched on separately.

With the rotary control you further adjust the output signal level. The setting is also displayed on the button.

2. Press the button “gate-on”!

The “GATE” is activated.

3. Turn the rotary control positioned over the button “gate”!

The attack time of the gate is set.

4. Push and rotate the rotary control positioned over the button “gate”!

The release time of the gate is set.

5. Push the button “exp on”!

The expander is activated.

6. Turn the rotary control positioned above the button “exp on”!

The threshold parameter is changed.

7. Push and turn the rotary control!

The ratio is set!

The time constant is set like in the gate section.

The compressor of the zirkon is of RMS-near character. This allows to give a more compact impression of voices and noises. For using it also in a running programme, the characteristic line is rotated around the working point during the ratio setting (see diagramme). This allows the setting without the danger of level breakdown by displacement of the characteristic line.

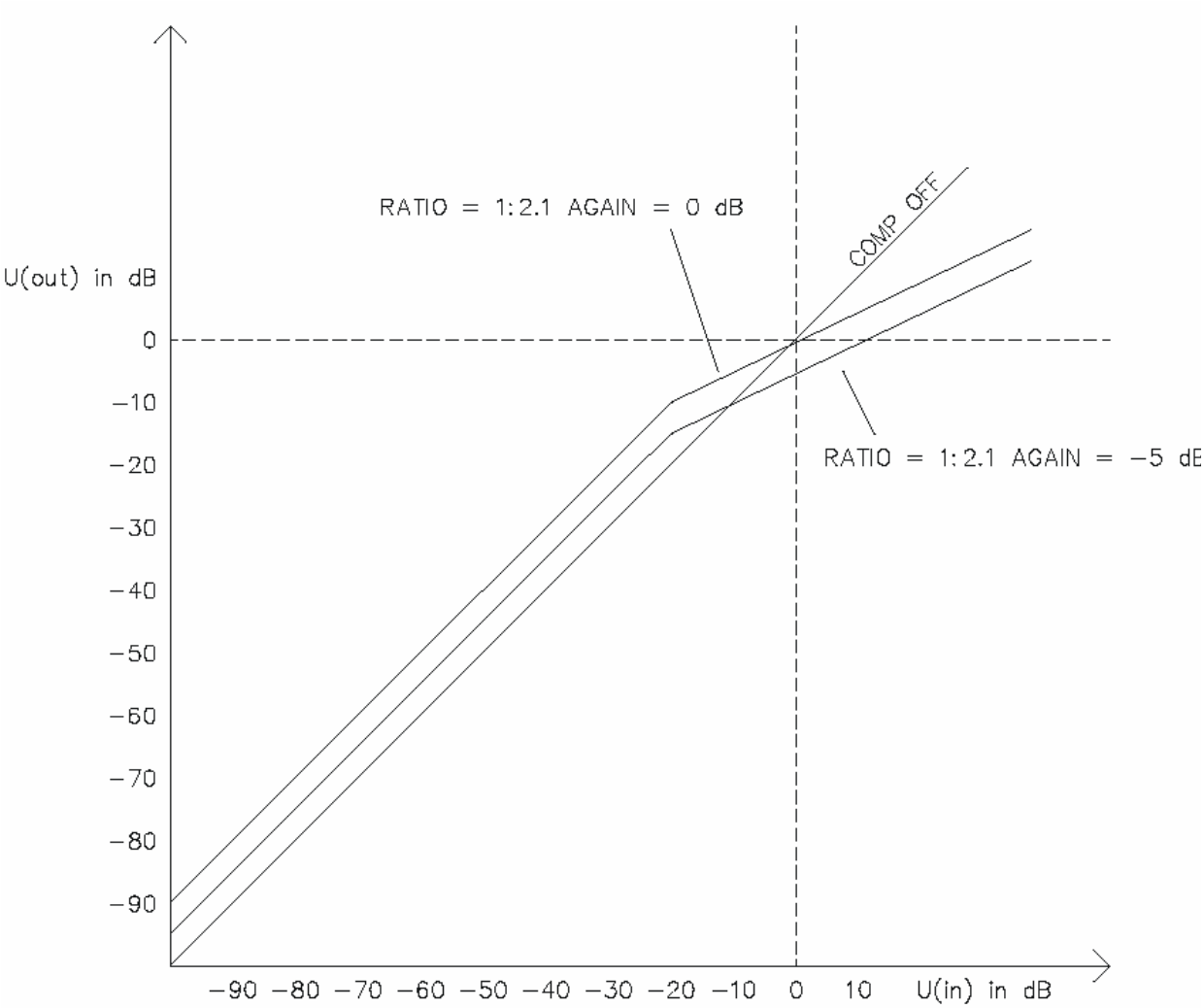
8. Turn the rotary control positioned above the left compressor button!

The attack parameter is set.

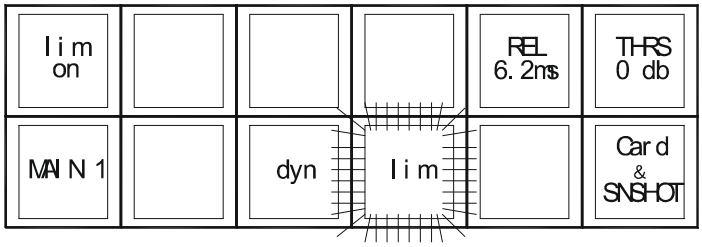
9. Push and rotate to change the release time.

10. Push the right button to activate the compressor.

11. Turn the rotary control positioned above the right button to change the threshold value.


RATIO = 1:2.1 AGAIN = 0 dB
COMP OFF
RATIO = 1:2.1 AGAIN = −5 dB
U(out) in dB
0
−10
−20
−30
−40
−50
−60
−70
−80
−90
−90 −80 −70 −60 −50 −40 −30 −20 −10 0 10 U(in) in dB

Limiter (lim)



The limiter can be programmed in sources and sums like the previously described sound designing modules. Preferably, the limiter should be placed within summing buses to prevent overmodulation. The characteristics are that of a transient limiter for effectively limiting volume peaks.

1. Push the button “lim on”!

The limiter is activated!

2. The parameter release time is set with the rotary control positioned above the button REL!

3. With the rotary control positioned above the button THRS you change the threshold value!

Storing of configurations, PFL

The zirkon allows you to store and recall your setups. The zirkon offers the possibility to store the setups internally or on chipcards. With this function you can minimise your preparation time.

Every system comes with four chipcards.

Each chipcard can store one setup; internally you can store up to five setups. Every setup includes the following settings:

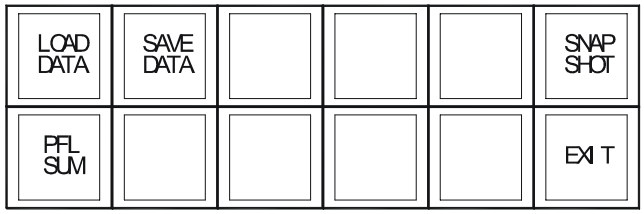
- Control surface assignment
- Settings and modes of the AUX send level
- Settings of the sound designing modules (EQ, DYN, LIM)

Storing the sound designing modules facilitates to save different settings for different presenters.

Speedy insertion of the chipcard starts the reading. This is indicated by the illuminated LED on the chipcard reader.

1. Insert the chipcard speedily into the chipcard reader!

The read process starts (LED is on). The following display appears on the LCD buttons:



Storing a setup

With SAVE DATA you store the current settings.

1. Push the button “SAVE DATA”!

The current settings are stored!

Loading a setup

With LOAD DATA you can load a setup. As for channel-strip assignments, open channels are not overwritten. This allows to load a new surface while on air. Channels open during the loading of a setup can be manually loaded afterwards.

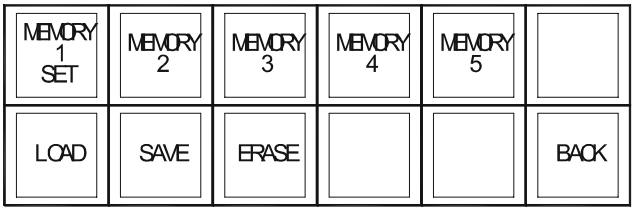
1. Push the button „LOAD DATA“!
2. Select a setup you want to load!

The required setup is loaded.

If LOAD DATA is not displayed either the setup on the card is of a different system or the card does not contain data.

PFL mode

1. As long as a chip card is in the reader you can also set the PFL mode with the left bottom button.



The following modes are offered:

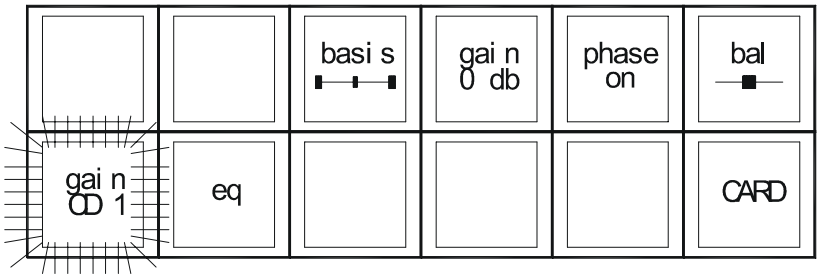
- PFL: pre-listen releasing. Opening the channel cancels the active PFL in this channel.
- PFL SUM: pre-listen sum. Opening the channel cancels the active PFL in this channel.
- PFL R: pre-listen releasing. With opening the channel changes the mode to “prepared”. After closing the channel the mode “pre-listen”.
- PFL SUM R: pre-listen sum. Apart from that like PFL R

2. This menu is left using EXIT.

If you want to return to this menu and the card is still in the reader:

3. Push the LCD button in the channel strip!

The central module shows the following labeling:

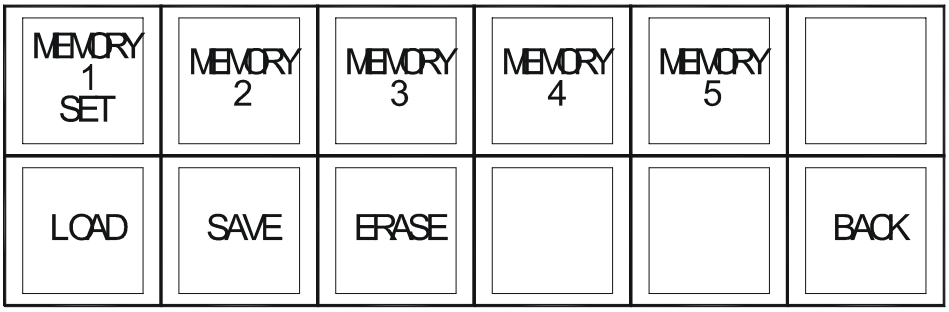


4. Press the button “CARD”!

You are now in the PFL mode menu.

Storing of snapshots

The management of the internal snapshot is reached via the button SNAPSHOT in the main menu. The button section displays the following labeling:



1. Press the button “SAVE”!
2. Select the required memory (MEMORY 1-5)!

The snapshot is stored.

Loading of snapshots

1. Push the button “LOAD”!
2. Select the required memory (MEMORY 1-5)!

The snapshot is loaded.

Radio automation

The serial interface (RS422) at the Central Board of the System Unit allows to connect a radio automation system. This is an easy way to hand over the faderstart functions of the console to the system, to activate a fader automatically with starting a playlist or to use one of the many special functions (programme switch, redlight, unoccupied cycle time, etc.) in the context of console and radio automation.

Activating these features requires the programming of the source names which have to be exactly the same in the zirkon system and the radio automation. With starting the radio automation, the system occupies these sources on the mixing console. From this time on the radio automation controls these channel strips and cooperates with the user on the control panel.

The actual functions of the radio automation depend very strongly on the radio automation system. Partly, the functions are restricted to faderstart, other systems control even the programme switch.

The serial protocol is available on request.

Maintenance and service

System page (http://)

The zirkon system allows to monitor the system status via a website. To this end the master board includes an integrated web server.

Important

The following points must be observed when establishing the link with a “crosscable”:

- TCP/IP – address (see configuration)
- The option “Use of a Proxy Server” in the web browser must be deactivated. Otherwise the access is prevented by looking for a “Proxy”.
- No link of the type telecommunication header record or VPN should be initialised automatically by the computer.

When integrating the zirkon in the radio station network please consult the network administrator.

The system page is reached this way:

1. Enter the IP address in your web browser! Example: http://192.168.101.240 (settings ex works)

The system page is opened.



The system page shows static and current data. The current data are grouped under the heading “snapshot”. This information refers always to the time of the last page update.

DSP card information

Local Date and **Local Time** show the system time and the date. The addition NTP indicates that a connection with a NTP server is established.

Every master board works with three voltages. The current value can be found out under the entries Voltage 1, 2, and 3. Nominally, **Voltage 1** is 1.8 V, **Voltage 2** is 3,3 V and **Voltage 3** is 12 V.

Temperature displays the temperature of the master board in Celsius.



Note

The temperature of 947/81 should not exceed 45° C.

This indicator does not give information about the temperature in the rest of the frame or in the rack.

It can be only used to find out a tendency.

Sync Information

SyncCard: Madi: Wordclock: Intern: present enable active

Slot Information

This table represents the plug-in cards of the DALLIS frame. Under “slot” you find the number of the slot. Card ID signifies the module number. Card type gives a short term of the card type.

Panel Information

Supplementary Information

This part shows general statistic system information. **Deskname** displays the console name that is entered in the configuration. **Last Restart** shows date and time.

Control Panel Modules: Configuring and Testing

The modules 950/41 and 950/42 offer a service menu for servicing and configuring. Before reaching this menu please carry out the following settings:

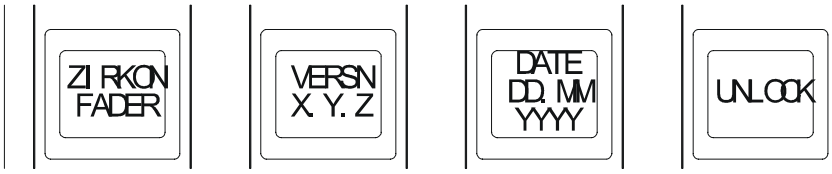
1. Unplug the RJ45 cables (CAN bus) from the control panel modules!
2. Unplug the power supply units of the modules!
3. Set the hex switch to the position “0”!

The modules are prepared for the tests.

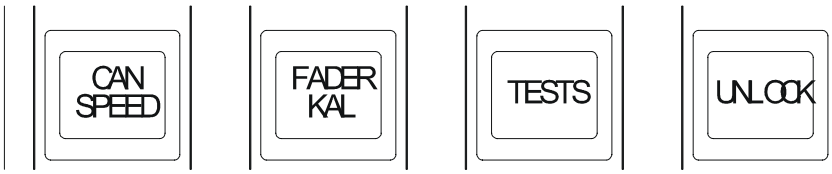
4. Plug in the power supply units again!
5. Connect the modules with the RJ45 cable again!

Depending on the module type the test menu appears now.

Channel module 950/41:

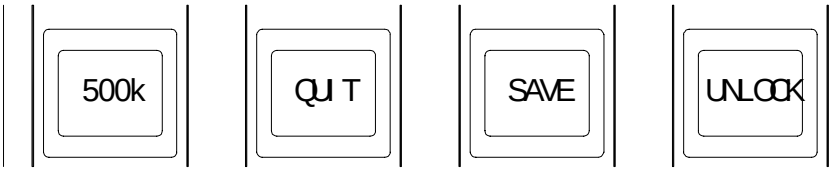


- **VERSN** shows the software number.
- **DATE** is the date the software was created.
- **UNLOCK** leads to further menus.



- **CAN SPEED**
  With the menu “CAN SPEED” you can display and change the speed of the bus.
- **FADER KAL**
  Calibrating the motor faders.
- **TESTS**
  Further menu for different system tests.

CAN SPEED



The left LCD button shows the current speed of the CAN bus interface.

1. Turn the rotary control to adjust the speed! (500k, 100k, 10k)

Note

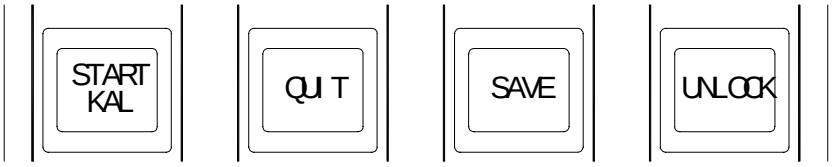
All components of a CAN bus must be set to the same rate or no communication is possible.

2. Push the “SAVE” button to adopt the selected settings!

3. Push the “QUIT” button to reach the main menu!

FADER KAL

Tip



The automated calibrating of faders is only necessary after replacing a fader. After an operating time of about two years a fader calibration is recommended. Apart from that it is not necessary. This way you start the calibrating procedure.

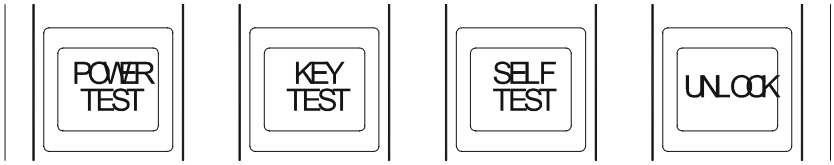
1. Press the button “START KAL”!

The calibrating starts. The calibrating is finished when the display of the “START KAL” button changes to “FADER READY”.

2. Push the button „SAVE” to adopt the new measured results!

3. Push the button “QUIT” to reach the main menu!

TESTS



In this sub menu you can choose from the following tests.

POWER TEST

This test switches all lamps on. It is integrated for the Lawo system test.

1. Press the button “POWER TEST” to start the test!

All lamps light up.

2. Press the button “UNLOCK” to abort the test!

KEY TEST

This test checks the function of the buttons.

1. Press the button “KEY TEST” to start the test.

The test is running. Please follow the instructions:

2. Push all buttons of the channel module one after the other!

The LEDs under the buttons light up.

3. Turn the rotary control positioned above the LCD buttons!

The LEDs flicker according to the rotation direction.

4. You quit the test by pressing the button “UNLOCK”!

On the third button from the left appears the label “QUIT”..

5. Confirm the button “QUIT”!

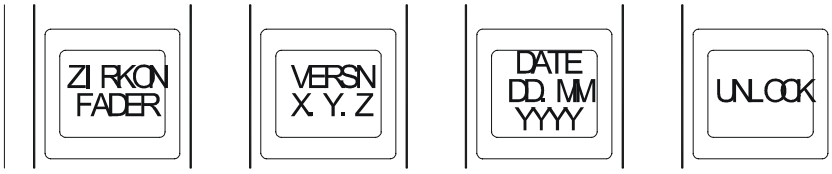
SELF TEST

1. Push the button “SELF TEST”!

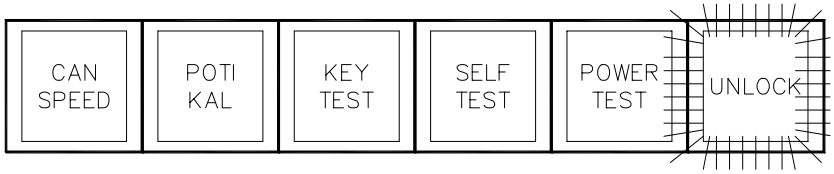
This is an automated test. The motor faders drive up and down, all LEDs light up one after the other, and on the LCD displays all pixels are switched on in lines one after the other.

2. Quit the test by pushing the button “UNLOCK”!

Central module 950/42:

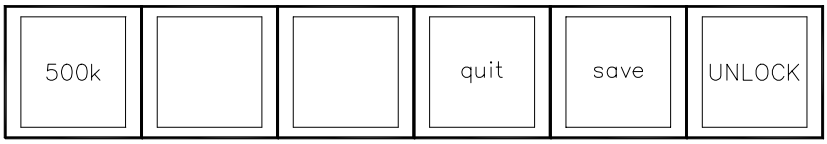


- **VERSN** shows the software number.
- **DATE** is the date the software was created.
- **UNLOCK** leads to further menus.



- **CAN SPEED**
  With the menu “CAN SPEED” you can display and change the speed of the bus.
- **POTI KAL**
  Calibrating the potentiometers.
- **KEY TEST**
  This test checks the function of the buttons.
- **SELF TEST**
  This test checks the LCD-buttons.
- **POWER TEST**
  This test checks all lamps.

CAN SPEED



The left LCD button shows the current speed of the CAN bus interface.

1. Turn the rotary control to adjust the speed! (500k, 100k, 10k)

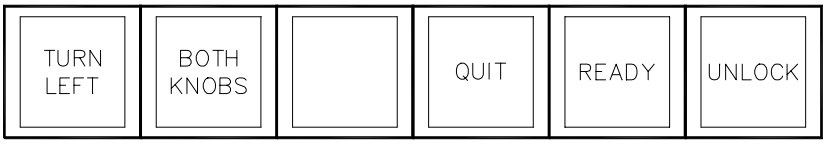
Note

All components of a CAN bus must be set to the same rate or no communication is possible.

2. Push the “SAVE” button to adopt the selected settings!

3. Push the “QUIT” button to reach the main menu!

POTI KAL



Tip

The calibrating of potentiometers is only necessary after replacing a potentiometer. After an operating time of about two years a potentiometer calibration is recommended. Apart from that it is not necessary. Start the calibrating procedure as soon as the above LCD-display appears.

1. Turn the potentiometer “Monitor 1” and “Monitor 2” anti-clockwise to the left dead-stop!
2. Push the buttons “UNLOCK” and READY” simultaneously!

The new measurement is accepted.

3. Turn the potentiometers “Monitor 1” and “Monitor 2” clockwise to the right dead stop!
4. Push the buttons UNLOCK and “SAVE” simultaneously.

The calibrating is finished.

KEY TEST

This test checks the function of the buttons.

1. Press the button “KEY TEST” to start the test.

The test is running. Please follow the instructions:

2. Push all buttons of the channel module one after the other!

The LEDs under the buttons light up.

3. Turn the rotary control positioned above the LCD buttons!

The LEDs flicker according to the rotation direction.

4. You quit the test by pressing the button “UNLOCK”!

On the fourth button from the left appears the label “QUIT”.

5. Confirm the button “QUIT”!

SELF TEST

1. Push the button “SELF TEST”!

This is an automated test. All LEDs light up one after the other, and on the LCD displays all pixels are switched on in lines one after the other.

2. Quit the test by pushing the button “UNLOCK”!

POWER TEST

This test switches all lamps on. It is integrated for the Lawo system test.

1. Press the button “POWER TEST” to start the test!

All lamps light up.

2. Press the button “UNLOCK” to abort the test!

Cleaning the operating surface

For cleaning the control panel of the console please use technical cleaning cloths with the following features:

Important

- Antistatic
- Fluff-free
- Without emery pieces

Warning!

Liquids must not enter the electronic circuitry!

Do not clean the mixing console with sharp instruments!

We use the cleaning cloths “Technical Cleaner” by Coolike (http://www.coolike.com).

Replacing the lithium battery

The master board in the system unit is fitted with a lithium battery. This battery guarantees that your data remain saved also when the system is switched off. As a precaution to prevent malfunctioning we recommend to replace the battery after about eight years. For this replacement, please send the respective central board to your service point.

Defective modules

Warning!

The devices contain no user-serviceable or configurable parts!

Therefore:

- Do not open the devices. Maintenance works inside the devices are not necessary.

In case of an evident hardware defect please send the defective device with a detailed description of the fault (see enclosures) to your service point.

Note

Modules sent to our service-points that are not defective cause much work and high costs. Please check yourself carefully beforehand whether the failure is caused rather by an erroneous configuration, operation or connection. Thank you!

Accessories

Power supply unit for console modules

For the supply of central module and channel module high-quality power supply units (PSU) are available. These are adapted especially to the supply of these console modules and perform permanent operation. The PSU is delivered with country-specific adapters for many socket-systems.

Redundant PSU for the system unit

The 3RU as well as the 6RU system unit can integrate an optional second PSU. This works in parallel operation and takes over the full load in case of failure of the other PSU. The PSUs are hot-pluggable and can be replaced while the system is working.

19'' Push button module (KS16)

In addition to the central module and up to 9 channel modules, the zirkon can be extended via the CAN bus of the system by up to five 19'' 1RU push button modules of the Lawo diamond programme. These devices include 16 self-labelling and freely programmable buttons. One button can be defined as "SHIFT" button so that alternatively 15 programmable buttons are available in two layers.



Furthermore, this control panel offers eight opto-couplers, eight relays (contact: 1 x shutter) and four connectors for potentiometers. All functions can be programmed freely. Thus, this control panel is suited optimally for remote talkback panels, matrix controls and remote controls, or as problem-solver for difficult cases of application.

The power is supplied with 230V by the integrated PSU.

Glossary

Access	= to access, to take into access For editing a channel it has to be taken into **access** and thus be made the active channel that can be edited in the central control section.
AFL	After Fader Listen This mode includes the relative volume while monitoring and allows the sound evaluation and settings of the channel.
Assign	= to assign, assignment Assigning a DSP channel to a fader resp. the direct or indirect summing bus assignment for the channel in **access**. The operation assign is based on the fact that the required channel is in access.
ATM	Asynchronous Transfer Mode (Packing of signals in small portions commonly used and highly standardised network protocol)
Attack- Time	1) Musical instruments differ in the time (attack time) their tone needs to develop the full level. 2) The attack time defines the duration over which the level of the input signal is measured. The more time was chosen the more delayed the module (gate, expander, compressor, limiter) reacts.
Auto on	All modules selected for automation are in the status **Auto** on.
Aux	Auxiliary = reserve Aux and number signifies an auxiliary channel which can be used as additional output e. g. to effect modules.
Aux- Master- Channel	= Auxiliary master channel This channel type offers an additional sum output signal. The input of the aux master channel is fix-connected with the respective aux summing bus. and cannot edited by the matrix. The signal processing includes the same range like a summing channel with insert point and direct out. The output signal of the aux master is available at the direct out. It is positioned at the end of the signal processing chain. It is not possible to access a summing bus from an aux master channel.
Aux- Return	Auxiliary return = auxiliary *input* which is used as return for the effect signal of an effect module. An aux return offers usually no or only a simplified sound regulation and cannot be assigned to an aux send channel. For separate control of the effect signal and assignment to a monitor the aux return should be used for effects only when better equipped channels are occupied.
Aux- Send	Auxiliary send = auxiliary *output* Auxiliary output of a mixer for any part of any channel. With the auxiliary outputs you define the effect parts and the monitor level independently for any channel. The auxiliary output is linked with the input of the effect processor or the input of the monitor amplifier (or its equalizer). A mixer includes in every channel strip several aux fader for the mix on the auxiliary outputs. They are positioned PF or AF.
Balance	Ratio of the level between left and right channel of a stereo signal. Weighting the signal to the left means the diminishing of the level on the right.

Band pass filter	Serial connected high and low pass filter that lets pass only frequences between these two limiting frequency, i. e. a frequency band of a defined bandwidth.
Bandwidth	Frequency band or range of frequences that can pass a system or device e. g. a filter given in Hertz. Often octave is given as unit whereby the next higher octave equals the doubling of the frequency.
Bargraph	Optical display instrument in shape of a LED bar.
Bypass	Mode for evading a signal processor. The bypass switches off the module and the signal passes unprocessed.
Channel	Channels are defined as mono or stereo. Apart from channel types used for the processing of audio signals exists also the type master that containes no individual audio signal but is used for controlling assigned slave channels.
Channel cassette	Operating elements in the channel strip are combined in the channel cassettes (e. g. Unit Processing).
Control panel	Part of the mixing console that includes the control modules, display instruments, operating elements, etc.
Core	Core of the system (Lawo core includes routing system and DSP)
Crosstalk attenuation	Problem caused mainly by electromagnetic induction. The proper signal of a channel appears attenuated also in neighbouring channels. The quality of the separation is given in dB, i. e. the difference between the level of the proper signal and the crosstalk-level.
Cue point	Point of time during automated operation at which a certain signal is to be played.
Cueing	Monitoring or a tape during the modes fast forward wind or rewind.
dB Dezibel	dB describes ratios, like percentage does. As long as dB is not completed by a reference, e. g. dB(A) it works as a factor, i. e. a number multiplied with another.
dBr	Relative voltage level according to the following equation: dBr=20lg U2/U1
dBu	Measure of the absolute voltage level based on 0,775 Volts. 0 dBu equals 0,775 V (RMS). The radio house norm level of +6 dBu is 1,55 V The studio level of +4 dBu is 1,228 V
Delay	Modulation effect based on the delayed play-back of the input signal. The signal is delayed and combined with the original signal.
Direct- Out	The direct out supplies the signal of a channel directly. The pick-up is carried out usually after mute switch, which means also after fader. It is positioned at the end of the signal flow in the channel strip (after insert point and sound control.
Drop- Out	Interruption of the audio signal caused by an error in the signal transfer or recording.
DSP	Digital Signal Processor Central processing unit in a digital signal processor on a board. The DSP is used for processing of data in realtime, which means it can permanently receive, process and give out data without interruption. This allows also a permanent reverberation.

Dynamic parameters	Real audio parameters like e. g. fader or equalizer, the dynamics unit or the pan/bal module.
Equalizer	An equalizer is used for adjust imbalances of the frequency response of the amplitude (equalising non-linearities of the frequency response).
Fader	Potentiometer often for gain adjustments
FC	Free Control Control elements that can be individually assigned by the user.
Filter	Device or circuitry which changes the frequency response of the amplitude (⇒ Equalizer)
Floor	This value defines by which amount the input signal is diminished when it falls below a certain threshold.
fps	Frames per second The ⇒ SMPTE timecode subdivides the time in hours, minutes, seconds, and frames.
Gain	The gain control serves for the adjustment of the level to the requirements, i. e. an increase of the gain leads to an amplification, whereas a reduction leads to an attenuation.
GPI/O	Actually GPI→General Purpose Interface
GUI	Graphical User Interface Screen unit for performing a multitude of settings and configurations.
Headroom	Modulation reserve, usually at +9dB (Germany)
High pass filter	Frequences above a defined limiting frequency can pass this filter type, frequences below this value are attenuated with a certain edge steepness.
High pass filter	Filter in the input of microphone amplifiers and gain faders to suppress rumbling noises that occur when stepping noises penetrate as impact sound the mic housing via the mic tripod.
Hold	Gate and expander allow to keep open the ⇒VCA for a certain time after the falling below the threshold. This parameter is named hold; the respective time is the hold time.
HP	Horizontal pitches ⇒ 5,08 mm resp. 0,2 inch
Input- channel	This channel type is used to edit supplied external sources individually. The input channel receives its source signal via matrix. The output signal is usually routed with the console-internal summing buses to groups, sums or also tracks.
Insert point	Connection (combined input and output) that routes an amplified signal into an effect device and back again into the signal path. Insert return = Input of the insert point Insert send = Output of the insert point
IT	Internationales Tonspurverfahren = international sound
Last access	Function to return to the channel accessed last.
Level control	Serves for displaying the input signal (input level) to be set by gain control or fader that level peaks do not cause distortions but the basic noise (signal to noise ratio) is drowned.
LFE	Low Frequency Effect = a channel for certain surround formats.

Locator	Remote functional unit at a tape recorder that allows an automated finding of certain tape positions. With the locator you define the tape positions (locator points or locator positions) at which for certain tracks the tape recorder switches from play to record and vice versa while playing.
Low pass filter	Frequencies below a defined limiting frequency can pass this filter type, those above are attenuated with a certain edge steepness.
MADI	Multi-channel Audio Interface; digital interface for combining audio signals of 56 channels. This format serves for the transfer between digital mixing consoles and DASH recorders.
Main sum- channel	This channel type serves for generating the main out signal. The input of the main sum channel is connected fix with the respective summing bus. The signal processing has practically the same range like an input channel incl. insert point or direct out. The output signal of the main sums is supplied at the direct out which is positioned at the end of the signal processing chain. From a main sum channel it is not possible to access summing buses (ASSIGN bus assignment).
Master	Operating element that rules over other elements of the same type by definition and assignment.
Merge mode	Mode in which two data streams or tracks of MIDI data are merged to one track or one signal.
Meter	= Display instrument
MIDI	Musical Instrument Digital Interface Manufacturer spanning digital interface which is seen as the standard for the realtime data transfer between different musical instruments.
Module	With regard to the zirkon it signifies the combination of display instrument, LED bar and operating element(s) for setting and monitoring parameters.
Monitor	Loudspeaker box used by musicians or presenters on stage or in the studio for monitoring themselves.
Monitor channel	This channel type is used for integrating optimally a connected multichannel machine with the console. For this purpose the monitor channel includes two signal paths: The summed signal of the track summing bus is supplied to the multitrack machine via direct output of the channel and the matrix (track send). The output signal of the machine (track return) is routed via matrix to the input of the monitor channel and allows the editing of the monitor mix. With the console-internal buses, this signal is routed to groups or sums.
Notch- Filter	Generates a steep-edged notch in the frequency response of the amplitude and thus a high Q factor with a frequency controlled freely. This filter type is well suited for suppressing disturbing noises with a clearly defined frequency.
On air	= Radio programme is being broadcasted
Overload	System is strained at the input with too large a level → non-linear distortion
Panning	Also stereo tremolo

	Effect which causes the signal to move automatically within the stereo basis (left-right).
Panning table	Table which contains already defined points of the signal within the stereo basis.
Panorama	Stereo basis The stereo effect is generated when identical sounds are perceived in the centre between two loudspeakers. The level ratio between the two channels (position of the mono signal within the stereo basis) defines the panorama position of a signal.
Peak- type- filter	also bell-type filter The filter curve corresponds with a Gauß curve or bell. Amplification or attenuation are performed around the filter frequency. The bandwidth is defined by the Q factor. Bell type filters are typically used for mid-range controls and parametric equalizers.
PFL	Pre Fader Listening The activated channels can be monitored with the headphones without being influenced by the position of the channel fader. As with the PFL function the channels remain unchanged, they are used for controlling during the concert or a recording.
Phantom power	Power supply for condenser microphones. The mixing console supplies the microphone with the necessary power of 48V via the symmetrical audio cable.
Q- Faktor	The bandwidth of a bell filter which is calculated as dimensionless ratio of middle frequency and bandwidth. The larger the ratio the better is the quality and the smaller the controlled frequency band.
Ratio	Compression or expansion ratio in a dynamic processor.
Release- Time	Time the level needs to die away completely.
Reset	= Warmstart Resets the system to the status after switching it on.
Return	Return line or auxiliary input of the effect signal, e. g. reverberator.
Routing	Signal routing The routing of a way from input to output of a mixing console.
RU	Rack Units ⇒ 44,45 mm resp. 1,75 inch
Sampling rate	Speed during collecting the measured values during the sampling
Send	⇒ Aux- Send Send channel that serves for editing by external signals.
Sensing	Function: multiple use of an operating element by touching it
Shelving- Filter	Shelving filters are used usually for bass and treble controls. The frequency response of this filter type is characterised by lower edges and resembles with an increase to a steep and with a decrease to a softly hanging cow's tail.
Sigma	Signalmatrix Tool for programming project specific special functions, i. e. freely programmable buttons can be individually defined by sigma programming.
Slope	Rise or fall within a curve with regard to an envelope.
SMPTE	Abbreviation for Society of Motion Picture and Television Engineers Standardised protocol for the synchronisation of audio and video

	technology ⇒Timecode
SRC	Sample Rate Converter
Sum	Signal mixed together with a mixing console from different input signals with the purpose of further editing.
TDM	Time division multiplex, picking up a signal from a data stream
Telephone hybride	Device for the level adjustment of a 2-wire phone line (tone line with two wires for bidirectional talk and for one message or comment line) to a telephone mixing console without acoustic feedback.
Threshold	Parameter that defines for the compressor the gain above which a gain reduction is triggered. For gate and expander it is the level below which a reduction is triggered.
Tiny channel	Audio channel without signal processing.
Unit	This general expression signifies the complete range of all units of a console that are accessable (channel, fader, ...)
VCA	Abbreviation for Voltage Controlled Amplifier.The VCA serves lesser for the amplification of audio signals but rather for their exactly dosed attenuation.

Sources:

Jürg Jecklin, „Musikaufnahmen”, 2 Auflage, Franzis Unterhaltungs-Elektronik

Michael Dickreiter, „Handbuch der Tonstudiotechnik” Band 1/2” , 5 Auflage, K.G. Saur

Jan Friedrich Conrad, „Taschenlexikon”, PPV

Jan Friedrich Conrad, „Recording”, PPV

Friedrich Guthmann, „Lexikon der Tontechnik”

„KEYBOARDS”, Magazin für Musik und Computer, Ausgaben 12/00, 01/01, 02/01

User Manual mc² series (Version 1.7)

Lawo AG
Am Oberwald 8
D-76437 Rastatt
Tel. +49 (0) 7222.1002-0
Fax. +49 (0) 7222.1002-24
E-Mail: info@lawo.de
Web: www.lawo.de

interstage
Phistersvej 31, 2900 Hellerup, Danmark
Telefon 3946 0000, fax 3946 0040
www.interstage.dk
- pro audio with a smile