

## WARNING

Do not expose this appliance to rain or moisture. To reduce the risk of electrical shock, do not remove cover. There are no user-serviceable parts inside. Do not attempt to repair the unit yourself; this voids the right of warranty. Refer servicing to qualified personnel.

### CE-Product

This product is in conformity with the requirements of the CE directives.



This product should only be installed and used in installations as specified in this manual and should only be used with the ancillary equipment and options and in the right environment according the recommendations.

The supplier will not take any responsibility of damage to or induced by this product caused by using the equipment in any not specified application or connecting items or equipment, in any way or environment other than specified.

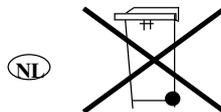
## General Information

### Cleaning

Clean the outside of this product with a soft cloth. Do not use any cleaning detergents like alcohol, white spirit or ammonia based fluids to clean this product. This can severely damage the finish of the product.

### For waste separation:

At the end of the lifecycle of this product or the batteries dispose it in accordance with local procedures for disposal of hazardous or chemical waste. Do not dispose the unit or batteries in the ordinary way, but if possible hand it in as Small Chemical Waste.



Thank you for the purchase of this *D3 digital mixing desk*. First read this manual carefully before starting to use the apparatus. Please keep this manual so you can refer to it at a later stage.

Functions and / or layout of the touch screen of the D3 can vary due to different software versions. More information is available on <http://www.eela.nl/D3TI>

## **GENERAL REMARKS:**

The D3 is designed in a way that most of the functionality is self-explaining and easy to understand by the experienced radio studio operator.

As all software based equipment the functionality and screen layout can change with new versions. This manual is based on hardware revision "D".

Due to the flexibility and multi purpose nature of the unit some settings need extra attention especially the setup of hybrid connections and the integration in an automated environment.

Additional information can be found on the internet: <http://www.eela.nl/D3TI>

## **Software version 1.07**

### **INDEX**

Control surface	5
Main screen / meters	6
System settings	7
Presets	9
Channel settings	10
General specifications	13
Connections	14
GPIO pinout	16
Cleanfeeds, N-1, foldback, outside source return, cough signal	17
Setup examples, Control room microphone, studio microphone, hybrid	18

# **Eela Audio D3 table top / 19" digital mixing console: Quick start manual**

All digital "universal" broadcast mixing console with 24 inputs, 8 input faders, 2 output faders and a touch screen color display. The unit is design for modern radio applications and suited for simultaneous dual use:



## **APPLICATIONS:**

The Eela Audio D3 is designed for use in modern radio and television broadcast applications for On-Air and or production studios.

### **Table top / 19" digital mixing console**

**24 inputs, 2 main outputs, 4 configurable stereo outputs**

**8 input faders**

**2 main output faders**

**Color touch screen for settings, E.Q., dynamics, configuration**

**Channel on "Start" button above each fader**

**Broadcast audio and logical functions**

**Adjustable delay in the main outputs for up to 100 msec.**

## CONTROL SURFACE:



8 Channel input faders with 3 pushbuttons with LED:

Selecting the top grey button opens the channel window in the touch screen display. The red LED in the left top corner of the button will lit indicating the channel displayed in the main screen. A fast blinking LED indicates an incoming call from a hybrid and a slow blinking LED indicates a hybrid "off hook" state.

The yellow button is PFL, pre fade listening, the LED indicates an active state. PFL can be mixed or individual.

Blue button is the 'START' button and LED. This function can be activated in the 'settings' menu of each channel. If active the LED will blink fast if pressed indicating a ready status, faderstart command will be fired upon opening of a fader. Or the other way round, the LED will blink fast upon opening of the fader and the faderstart command will become active after pressing the 'start' button. A steady LED indicates the active state of the faderstart. If 'BTN start' in the settings menu is not selected the start command is the normal faderstart and the LED is steady lit upon opening of the fader. This LED will blink slowly if an external MUTE signal is applied.

3 LED's next to the fader:

LED A and B indicate the destination of the signal, output A and / or output B and are green when the fader is closed and turn red upon opening of the fader. In case the fader is overruled by an external command the LED's will be red also!

LED "O" will turn red in the event of an overload of the input electronics. The led will turn green if there is a signal of at least - 25dB present at the input and box 'SIGN Led' is checked in the channel settings menu. See page 11, from software version 1.05 onwards.

Main output faders A and B, can be switch off or used for other outputs.

Rotaries for volume control of headphones and monitor loudspeakers.

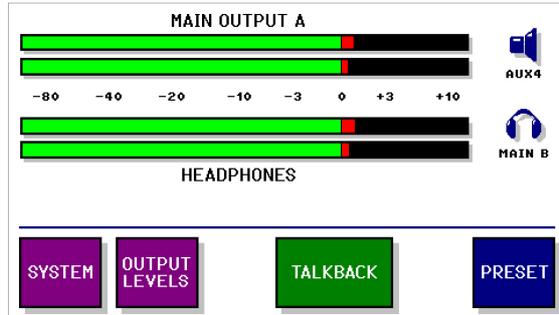
Headphones connector on front and on the back in parallel.

Rotary / selector for use in conjunction with the touch screen.

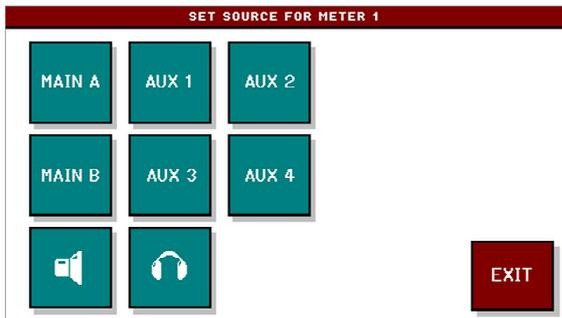
Touching a volume bar or settings box the background changes color and you can adjust with the help of the rotary. Volume bars turn white and a change is momentary executed. The backgrounds of the setting boxes turn yellow and you can change the setting with the rotary. You have to confirm the change by pushing on top of the rotary.

**Eela Audio D3 table top / 19" digital mixing console:**

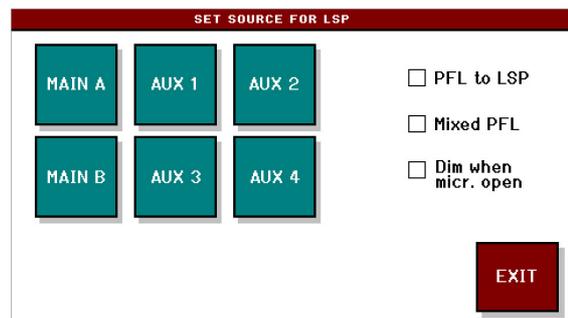
**MAIN SCREEN / METERS:**



Both meters, Loudspeaker and headphone icons hide a **source selector**. You can select the source for each item by touching the meters / icons and the source selector will open:



Source selector for meters



Source selector for speaker / headphones

Selecting the speaker or headphones icon in the 'meter selector' the meter will follow the monitor source.

In addition you can touch the check-boxes in the monitor screen for the other functions:

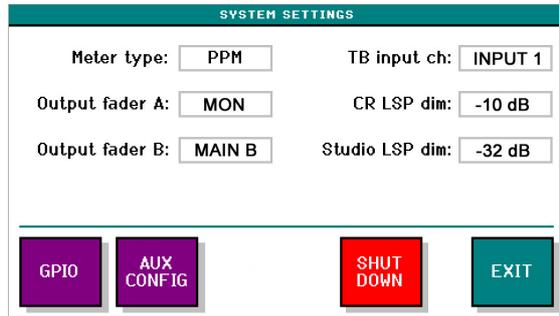
PFL to LSP routes the PFL signal to the main monitor speakers if a PFL button of one of the modules is active.

Mixed PFL or individual PFL.

Dim when (Control Room) microphone opens instead of mute. The dim value can be adjusted in the next screen. Opening a fader of a channel configured as CR microphone will standard mute the CR loudspeaker. Selecting the option "Dim when micr. open" will dim the monitor loudspeakers i.o. mute them.

**SYSTEM:**

Touching the SYSTEM button opens the main settings menu:

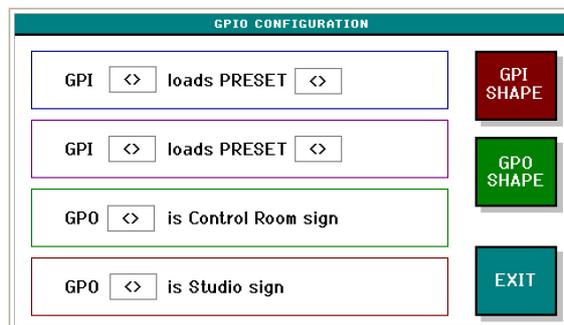


You can touch the value in the boxes to activate them, the background will turn yellow; with the rotary knob you can change the settings. Please note, these settings need to be confirmed by pressing op top of the rotary. Currently only one meter type (PPM characteristic) is available so this setting cannot be changed.

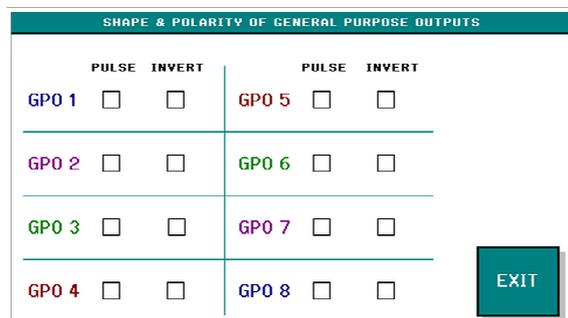
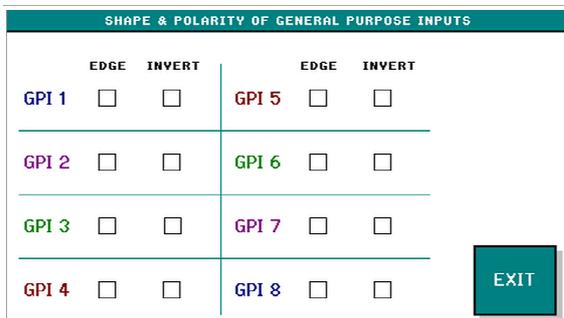
You can assign the output faders as a master to one of the outputs or simply disable the fader. "TB input ch" is the microphone input that is used for TalkBack purposes e.g. the DJ microphone. The unit does not have a separate built-in TalkBack microphone but you can select any of the inputs. You can set the dim level for control room and studio loudspeakers.

**GPIO:**

The D3 has 8 opto isolated outputs and 8 opto isolated switching inputs on board. A further 16 GPI/O are available on the EA963 extender. The GPIO button takes you into the main screen GPIO menu.



Inputs can be used to remotely load presets just with the help of a contact closure. Red-light signaling outputs for Studio and CR can be selected to feed a lamp relays system.



Here you can also set the type and shape of the GPIO's: pulse or continuously, inverted or not... NOTE THAT AN INVERTED GPI WILL BE ACTIVE WHEN NO CONNECTION IS MADE!! E.g. if you have assigned a GPI to an input mute that channel will be muted if the switch is not connected! If set to "EDGE" the GPI signal will toggle the function 'on' on the first pulse and 'off' on the second. You can step trough the pages for the settings of the GPI and GPO 9 to 24 on the EA963 extender.

## AUX CONFIG:

The master settings for the 4 aux outputs:

AUXILIARY OUTPUTS CONFIGURATION			
AUX1:	<input checked="" type="checkbox"/> Pre	<input type="checkbox"/> Post	<input type="checkbox"/> Studio monitor
AUX2:	<input type="checkbox"/> Pre	<input checked="" type="checkbox"/> Post	<input checked="" type="checkbox"/> Studio monitor
AUX3:	<input type="checkbox"/> Pre	<input checked="" type="checkbox"/> Post	<input type="checkbox"/> Studio monitor
AUX4:	<input checked="" type="checkbox"/> Pre	<input type="checkbox"/> Post	<input type="checkbox"/> Studio monitor

EXIT

Master AUX Configuration menu sets the AUX for 'Pre' or 'Post'. If a Studio monitor check box is active the Studio monitor DIM function is applied to this AUX upon opening a "Studio Microphone" fader.

## SHUT DOWN:

The unit has a power switch in the back. It is recommended to use this switch to shut off the unit. If the unit is flush mounted in a table or built in a 19" rack where you cannot reach this switch you can shut down (sleep mode) the unit in software. By holding down the 'settings' rotary for 3 seconds the unit will power up again.

Please note: From this state the unit will also switch on if the (mains) power was down and on again!

## OUTPUT LEVELS: (MAIN SCREEN)

Selecting the OUTPUT LEVELS button in the main screen opens an overview of all output levels.



Touching a bar selects it and allows for level setting with the rotary. Turning the rotary will immediately change the level, no need to confirm the setting by pushing the rotary. In this way you can still set the outputs levels of the main outputs without the faders assigned to these outputs.

Please note: If a master fader is assigned to an output the level cannot be changed with the rotary. Headphones and CR monitor rotaries are permanently assigned to these outputs.

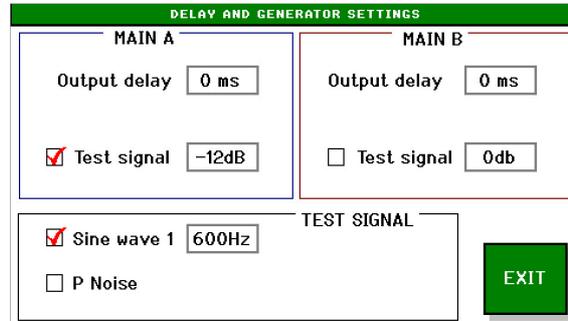
**NO OUTPUT FROM THE MIXING DESK:** This screen is the first to look at!

Example: If you switch off a master fader while the fader is in closed position the main level of this output will also be low. You can turn the volume up as described above.

## OPTIONS OUTPUT DELAY AND GENERATOR

Software version 1.05 or higher.

Selecting the OPTIONS button in the output levels screen opens the settings for delay and tone generator.

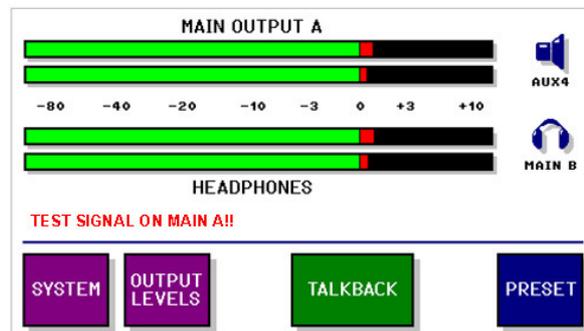


You can insert a delay in each of the stereo outputs channels for up to a maximum of 100 msec. This delay can be used to compensate for the latency in video processing systems to maintain synchronization of picture and audio. Another application is the delay of a (group of) loudspeakers in small PA systems.

In order to accurately calculate the delay, please take the latency of the signal in the D3 in account: From analogue in to analogue out this is approx. 1,5 msec., from digital in to digital out is max. 4 msec.

PLS NOTE: The delay is introduced in the main outputs only, you will not hear this delay in the monitor outputs.

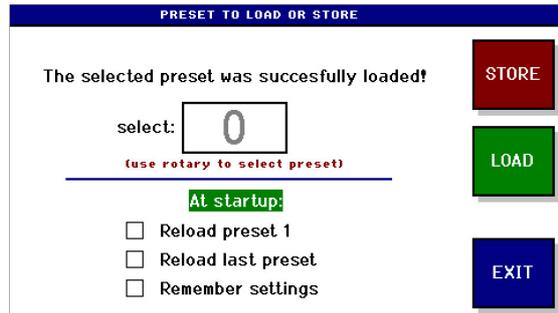
The sine wave test signal is adjustable in frequency or you can select the pink noise generator. This signal is mixed to the output. The volume of this signal can be adjusted with the rotary from – 30 dB to +10 dB but this also depends on the output volume control (fader). If switched on a warning message is displayed in the main meter screen:



The test signal can be inserted in the A or B output.

## PRESETS: (MAIN SCREEN)

The bleu **PRESETS** button in the main screen opens the presets window:



You can store / recall up to 32 presets.

**ALL SETTINGS ARE SAVED INCLUDING ALL PARAMETERS FOR EACH INPUT SEPARATLY!**

You can set all levels, switches for each INPUT individually e.g. for each AUX 3 x 8 = 24 settings are saved.

With check boxes you can indicate how the unit starts at power-up.

In the SYSTEM / GPIO menu you can assign a GPI to load up to 2 presets with external contact closures.

PRESET 1 can only be stored / overwritten with the help of a computer and a small program connected to the USB bus. In this preset you can store your basic settings in a protected way.

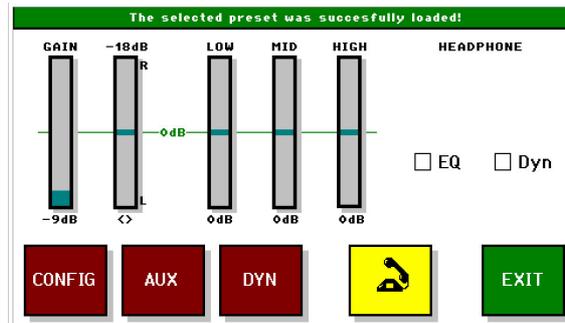
All settings are stored in memory with a battery back-up. This battery will last at least 5 years under normal conditions. The battery is a widely available type CR2032 or equivalent that is commonly used in PC's for the same purpose. A qualified technician can change this (internal) battery.

In the event of a firmware upgrade all settings will be lost including PRESET 1. It is recommended to write down your basic settings and keep these in a safe place.

## CHANNEL SETTINGS:

### Input select:

Each input fader has a "pfl" and "select" and "start" button above the fader. Briefly pushing the grey "select" button opens the channel settings window in the touch screen display:



Touching one of the bars (gain, balance/pan or e.q.) changes the color and with the help of the rotary you can make the necessary changes. The e.q. is limited to + and - 12 dB.

The small meters can be used to set the GAIN of an input but be aware: The meter follows the headphones source so PFL of the channel must be active and in the main headphones menu "PFL to headphones" box must be checked for a correct reading.

E.Q. and dynamics can be switched on / off in the check-boxes.

The yellow telephone icon is only present when the channel is used as a Telco channel (hybrid). With this button you can open the "Telco" window to switch the external hybrid on/off hook and dial a number. The unit has a built-in DTMF generator. (No internal hybrid!) A normal hybrid is transparent for these DTMF tones.

Pressing the CONFIG button opens the configuration menu for this channel. Settings are stored individually for each of the 3 inputs of a channel.

## The channel CONFIG button:

CONFIGURATION FOR INPUT 1 (MICROPHONE IN)

MIC LINE A LINE B Type: CR micr

Config: Mono

PHANTOM  INV L  BTN HPF: 100Hz

SIG Led  INV R TB output: AUX1

GPIO  MAIN A  MAIN B EXIT

### Please note: Some settings can only change when fader is closed!

For input 1 – 6 you can select between MIC, LINE A and LINE B. For input 7 and 8 this is AES/EBU, LINE A and LINE B. The functions of the other boxes are:

With 'Type' you will set the logic behavior of the channel:

'CR microphone', Monitor speaker will MUTE or DIM upon opening of this channel fader as well as activate the CR red-light.

'Studio microphone', Studio monitor will DIM upon opening of this channel fader as well as activate the STUDIO red-light.

The amount of DIM can be set in the SYSTEM setting. Also a line input can behave as a microphone input in case you are using an external microphone processor.

'Hybrid' In case you connect a telephone hybrid to this input. It will reveal the yellow telephone icon in the 'select' mode of the channel offering the possibility to switch a hybrid on and off hook as well as a DTMF keypad to dial a telephone number.

In case you are using another type of communication device or a telephone system that does not need or support the Telco switching or dialing you can set the channel as 'Line input'.

'Config': Can be 'stereo', 'swapped L+R', Left to mono, Right to mono, mono to both. Although displayed this is not active in MIC mode.

'HPF' the frequency of the high-pass filter can be set.

'TB output', if needed an aux output can be coupled to this input for Talkback purposes. E.g. a feed to a presenter headphone or TB to a hybrid or other "Outside Source". If set this will display the green TalkBack" button in PFL mode. Note: if mixed PFL is used Talkback becomes active for all channels in PFL mode that have an AUX coupled to them!

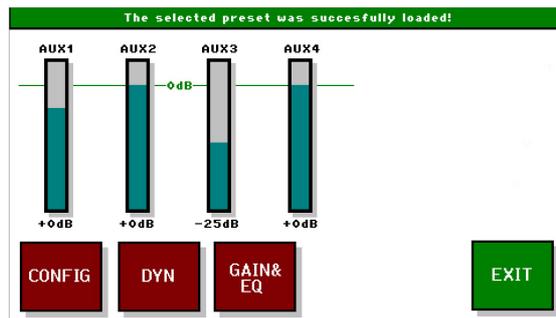
Check boxes INV L and INV R: phase reverse function. To reverse the phase of a microphone select both. Careful: Left or Right phase reversed and input 'Config' in mono will cancel the signal!

Only in MIC Mode: 48 V PHANTOM power switch. Although present in each MIC input screen the 48V phantom powering is centrally switched for all microphones inputs!

Check boxes MAIN A and MAIN B: Selects the main output for the channel signal. The corresponding LED next to the fader will light; green with fader closed and this will turn red upon opening of the fader. A red LED also indicates if a signal is routed to an output in the event a fader is externally overruled.

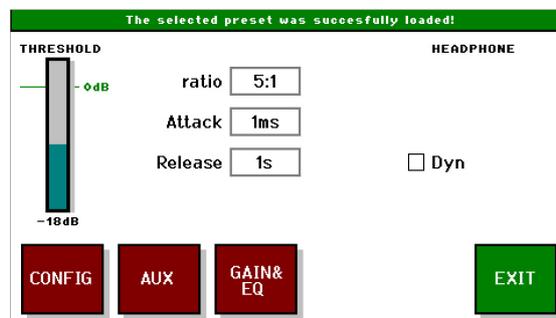
The third LED next to each input fader marked "O" is input overload indication, this will light red in case of an overload. The check box 'SIGN Led' activates the "Input signal present". The led will turn green if there is a signal of at least – 25dB present at the input. (software version 1.05)

## The channel AUX & DYN:



For each channel you can adjust the levels for the 4 AUX sends by touching the bar and using the rotary.

Also the characteristics of the dynamics can be set here:

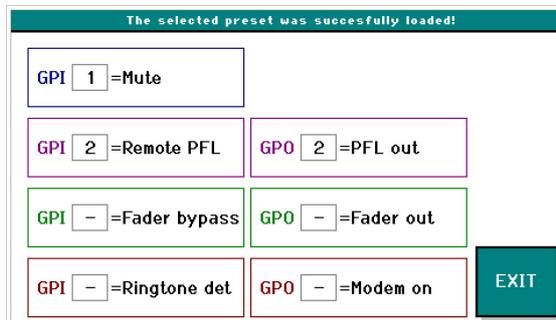


By selecting the ratio and turning the rotary clockwise the limiter function will be available.

The red buttons let you switch between the windows of the selected channel

## The channel GPIO:

The **GPIO** button opens the configuration menu for the General Purpose inputs and outputs.

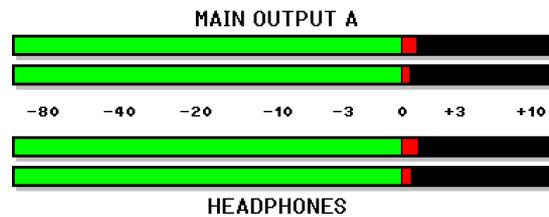


Inputs are: External mute, remote Pfl on, Fader bypass, and if activated as 'hybrid' also 'ringtone detect'. Fader bypass switched the channel on to unity gain and 'faderstart on' independent of fader setting.

Outputs can be: Faderstart, Pfl. Out and if activated as 'hybrid' also modem (hybrid) on.

The type and shape of the I/O's can be set in the main systems menu.

## METERS:



Currently only active PPM characteristics.

The standard alignment of the mixer outputs:

0 PPM is + 6 dBu at the balanced analog outputs

+10 PPM is 0 dBFS



## General specifications:

The D3 comes with 19" rack mounts, unit can be rack mounted without the need for extra space above the unit to allow room for connectors.

6 microphone inputs, balanced on XLR, switchable 48 Volt phantom powering,  
2 AES/EBU digital inputs, balanced XLR with sample frequency converters, range: 22kHz.> 200 kHz.  
2 x 8 analogue line inputs on RCA cinch connectors,  
2 stereo main outputs on balance XLR connectors, nominal output level +6 dBU.  
2 AES/EBU main digital outputs 48 kHz sample rate on XLR,  
4 stereo Aux. / cleanfeed / foldback outputs on 6,3 mm Jack, nominal output level 0 dBU.  
Monitor output on Jack, nominal output level 0 dBU.  
Headphones output on the front panel and on the back on 6,3 mm jack.  
USB serial connection,  
Both RJ45 connectors are for future expansion.  
D-type 25 pin female with 8 opto isolated inputs and 8 outputs. (Option: 2 x 8 GPI/O in 19" rackmount)  
Adjustable delay in the main outputs for up to 100 msec.

## Power Supply:

12 .. 24 Volts DC, approx. 1A, on XLR-4

Unit comes with wide range mains adapter, 100 .. 240 V AC, 47/63 Hz,

## Connections:

Connections are "industry standard":

### Inputs:

Microphone, AES/EBU balanced	XLR female	Pin 1 gnd; pin2 +; pin3 –
Line inputs, unbalanced	RCA Cinch	

### Outputs:

Main analog outputs L and R balanced	XLR male	Pin 1 gnd; pin2 +; pin3 –
Main digital ARS/EBU outputs balanced	XLR male	Pin 1 gnd; pin2 +; pin3 –
Aux 1..4 unbalanced	6,3 mm jack	Tip left; ring right; sleeve gnd
Monitor loudspeaker unbalanced	6,3 mm jack	Tip left; ring right; sleeve gnd
Headphones front and rear	6,3 mm jack	Tip left; ring right; sleeve gnd

LAN RJ45 connector for future use

DBUS RJ45 connector for future expansion: The EA963 is a 19" 1U A rack unit with and additional 2 x 8 GPI/O contacts as well as 4 solid state AC relays for switching lamps.

USB standard connector for:

## Firmware updates:

Locking of preset 1. In order to store your basic configuration in preset one you can use the update program to protect preset one against accidental overwriting.

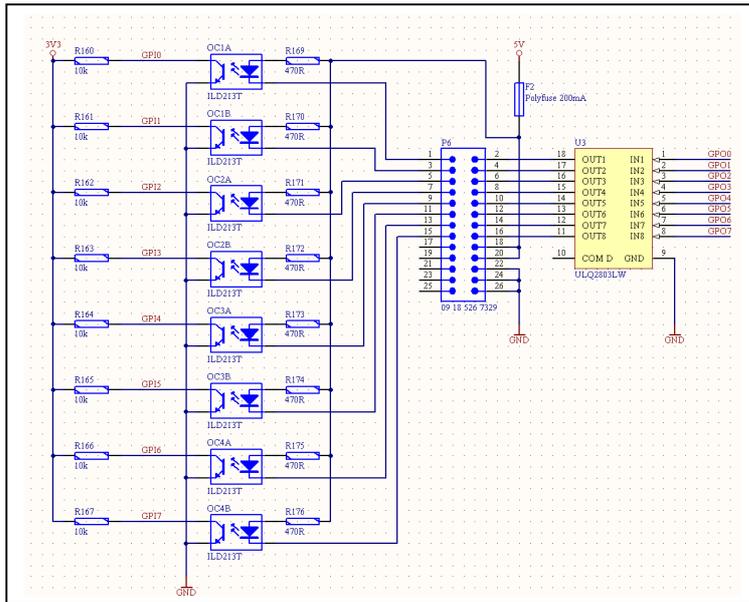
You can download the special update program and updates from: <http://www.eela.nl/D3TI>

The Eela Audio CAPS2 radio automation program interfaces with the D3 via the USB port. For details see the CAPS2 manual.

## Optional units:

EA963	19" 1U GPIO expansion box, with 4 solid state relays, 16 opto isolated general purpose inputs and 16 outputs. 4 outputs have solid state relays 24..230V AC The EA963 is controlled by the DBUS output of the D3.
EA924	19" 1U Quad input balancing unit, 4 x 2 high quality balanced input amplifiers, unbalanced out.
EA815	Electronic telephone hybrid, single or dual 19" version available.

### GPIO schematic overview:



Pin no.	Description
1	GPI_1
14	GPO_1
2	GPI_2
15	GPO_2
3	GPI_3
16	GPO_3
4	GPI_4
17	GPO_4
5	GPI_5
18	GPO_5
6	GPI_6
19	GPO_6
7	GPI_7
20	GPO_7
8	GPI_8
21	GPO_8
9	NC
22	+5V
10	NC
23	+5V
11	NC
24	GND
12	NC
25	GND
13	NC

25 pin D-type connector pin layout.

## Cleanfeeds, N-1, foldback, outside source return, cough signal:

There is a lot of confusion mainly about the names of the same kind of signals so let's take a closer look:

### CLEANFEED or N-1 or OUTSIDE SOURCE:

Basically this is the (mono) signal that goes back to a caller into a telephone hybrid. The signal consists of a mix of all signals in a program MINUS the signal that comes FROM the hybrid INTO the channel of the mixing desk. This signal from the hybrid back into the mix will cause a feedback and that is something we do not want. In radio broadcast mixing desks this signal can be a separate POST FADE AUX bus without the incoming signal from the hybrid.

In some analog mixing desks this is not a separate bus but the signal is made from the total sum and mixing this with a phase reversed version of the unwanted signal. This is OK as long as it is limited in frequency range so it is mainly used for telephone hybrids.

Also upon switching on the TalkBack the signal is replaced by the microphone signal in order to talk to the caller off air without the public hearing this conversation. If the damping of the hybrid allows this can also be done on-air.

In the D3 a dedicated STEREO full bandwidth AUX bus is used for this purpose. Other than only hybrids also full bandwidth stereo communication devices can be used (ISDN Codec).

With this system you can couple 2 or more D3 mixing desks in separate rooms together without the danger of feedback.

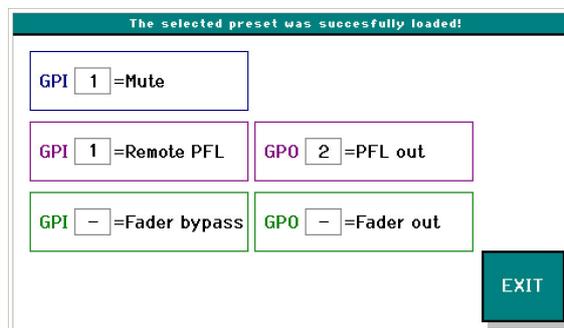
A special mode is putting the AUX-es in "pre-fader mode", this allows incoming callers to talk to each other off-air: CONFERENCE mode. This can be useful when 2 or more reporters are active that need to know about each others activities. However they should be aware when they are on and off-air!

### HEADPHONES FOLDBACK:

The signal that is fed to the headphones of a presenter. Basically the total mix of the main signal. In order to allow the technician behind the mixing desk to talk to the presenter this signal is replaced by the microphone signal if the TalkBack button is pushed.

With the D3 an AUX bus is used for this purpose and you can even built a dedicated mix e.g. lower volume of the music contributions. It is not a cleanfeed signal as the presenter headphones need the own signal present as the presenter needs to hear his/her own voice!

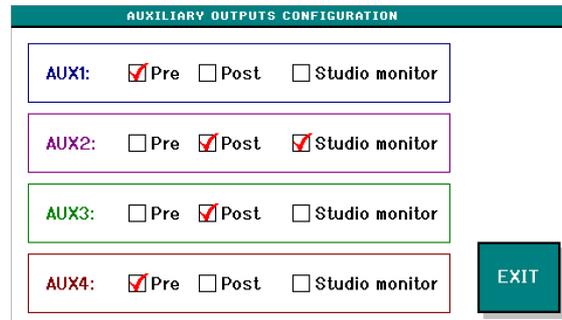
### COUGH:



Mute and Remote PFL selected to the same GPI.

The cough signal is engaged with a pushbutton in the studio. Pushing this button switches the microphone off the main mix and switches on the pfl. In this way the presenter can talk to the technician off- air. In the D3 you can connect an external switch to one GPI and select both "mute" and "remote pfl" of the microphone channel to the same GPI. If an external "mute" is applied the START button LED will blink slowly.

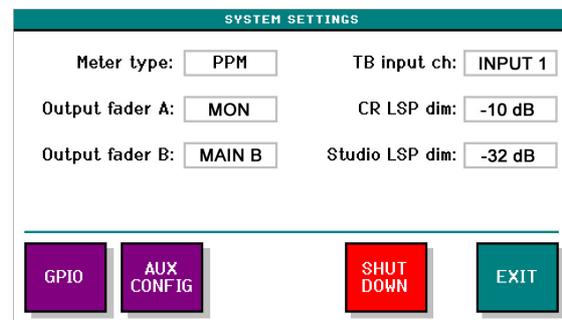
## STUDIO MONITOR FOLDBACK:



The AUX output configuration screen in the main menu.

The signal that is fed to the loudspeakers in the studio room. This signal can be adapted depending on your way of working. In some radio studio's (or TV) people don't use headphones. With an AUX you can make a mix of all post fade signals EXCEPT the studio microphones. The signal can be loud if no microphones are in use and dimmed upon opening of one of the studio microphones. The amount of dimming can be adjusted in the D3. Also this signal is replaced by a TalkBack signal upon pushing the TB button.

As the D3 mixing desk can be used in a true "split" mode e.g. an automated program is playing on air and at the same time making a recording you have to be able to select which signals must be present in the CLEANFEED, FOLDBACK or RETURN. That is why an post fade AUX is used for this purpose and you have to open up the AUX volume controls of ALL channels that must be present in this mix and close the others.



D3 System settings Studio and CR LS dim.

## TALKBACK:

Talkback is only possible when a microphone is set to be the TalkBack microphone in the SYSTEMS menu 'TB input ch:' (and this channel is set to MIC input and a microphone is present)

The TALKBACK button is only visible if a channel PFL button is active of a channel that has the 'TB aux' set to a certain AUX output. Touching this green Talkback button it will turn red and the signal of this AUX is replaced by that of the TalkBack microphone. Active as long as it is pressed.

If 'mixed pfl' is set and more than one channel with TB to Aux are in PFL mode than the TalkBack signal is send to these Aux outputs simultaneously.

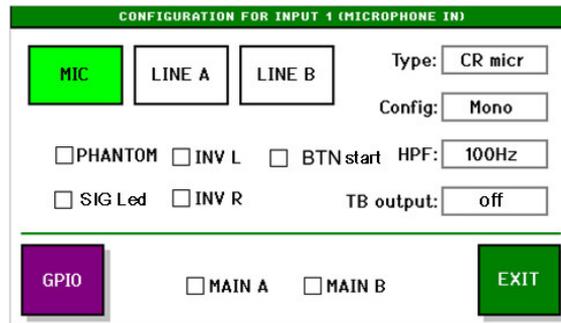
## Setup examples:

Setting up a CONTROL ROOM MICROPHONE:

Connect the control room microphone to one of the MIC inputs.

Briefly pushing the grey "select" button that opens the channel settings window in the display.

Push the red "config" button on the touch screen that will open the settings window:



Select the input being microphone, and briefly touch the "Type" box, the background will turn yellow, with the rotary select "CR micr" and push the rotary button to confirm the selection.

If you are using a condenser microphone select the box "PHANTOM", please note that if you select "PHANTOM" in one channel the 48 Volts will be switched on in all microphone channels.

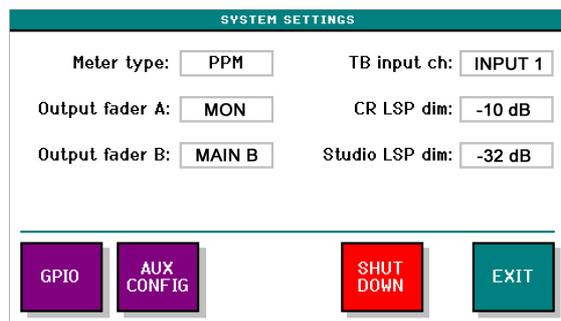
Briefly push the "TB-output" box and with the rotary select "off" as in most cases you do not need a TB channel assigned to a CR microphone. Confirm the selection by pushing the rotary.

Select the "MAIN A" and or the "MAIN B" box depending on the destination of the signal.

NB. The "Config" button mono/stereo has no effect in MIC mode.

Briefly touch the grey "select" button again to return to the main screen with meters.

In the main screen select "SYSTEM".



Select the box "CR LSP dim" and with the help of the rotary select amount the monitor speakers will dim upon opening the CR microphone fader in order to prevent feedback. Same for the "Studio LSP dim".

In case you will need to setup a TalkBack system to other sources you can use your CR microphone as a TB microphone:

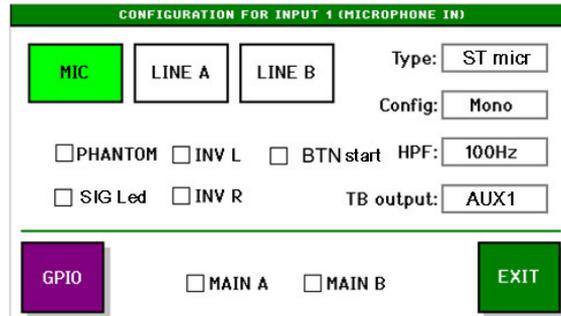
Select the box "TB input ch" and with the help of the rotary select the channel on which the CR microphone is connected. This will then be your TB microphone.

Setting up a STUDIO MICROPHONE with TB facility:

Connect the studio microphone to one of the MIC inputs.

Briefly pushing the grey "select" button that opens the channel settings window in the display.

Push the red "config" button on the touch screen that will open the settings window:



Select the input being microphone, and briefly touch the "Type" box, this will turn yellow and with the rotary select "Studio micr" and push the rotary button to confirm the selection.

If you are using a condenser microphone select the box "PHANTOM", please note that if you select "PHANTOM" in one channel the 48 Volts will be switched on in all microphone channels.

Select the "MAIN A" and or the "MAIN B" box depending on the destination of the signal.

You can assign one of the "AUX" outputs of the mixing desk as a return feed for the headphones of a presenter. Briefly touch the "TB-output" box and with the rotary select the "AUX" that you link to this channel as a feedback to the headphones of the presenter. Confirm the selection by pushing the rotary. As an example select "AUX1".

Of all inputs you want the presenter to hear adjust the volume of "AUX1" to the desired level and don't forget the master volume of AUX1 in the Levels screen.

Please note all AUX levels depend on and are stored with the selected INPUT so if you switch inputs on a channel you have adjust that level also!

To setup a "cough" circuit see details in chapter "COUGH" on page 18.

Briefly pushing the grey "select" button above the fader again to return to the main screen with meters.

## Setting up a HYBRID INTERFACE:

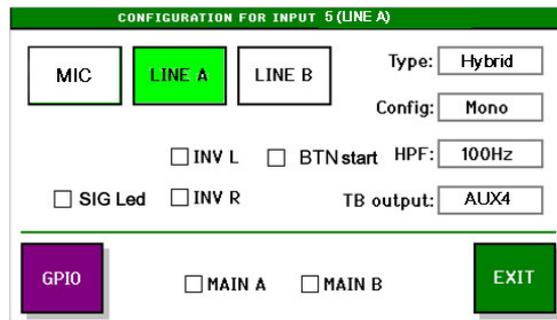
Connect the hybrid to one of the line inputs in this case we will use channel 5 input LINE A. In most cases a telephone hybrid is mono so connect its output to both inputs and set the 'config:' to mono. You can also connect a hybrid to L or R int that case select Mono R or Mono L.

For this example we are using AUX4 as return to the hybrid so connect the AUX4 output to the input of the hybrid with both L and R connected.

We will be using GPI 1 for "ring detect" and GPO 1 for "hybrid ON" (modem). Connect the hybrid accordingly.

Briefly pushing the grey channel 5 "select" button that opens the channel settings window in the display.

Push the red "config" button on the touch screen that will open the settings window:



Select the input being LINE A, and briefly touch the "Type" box, the background will turn yellow, with the rotary select "hybrid" and push the rotary button to confirm the selection.

Briefly push the "TB-output" box and with the rotary select "AUX4" being your return or Cleanfeed. Confirm the selection by pushing the rotary.

Select the "MAIN A" and or the "MAIN B" box depending on the destination of the signal.

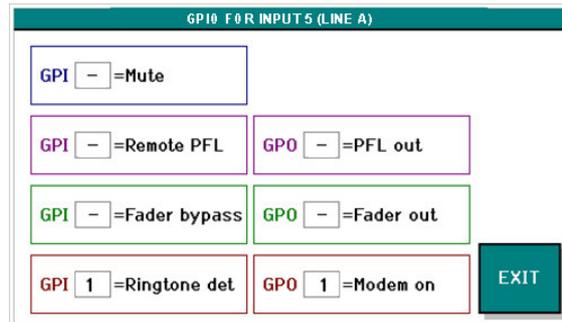
EXIT this screen and go to the AUX settings and make sure the level of AUX 4 is completely turned down to mute.

Of all inputs in all other channels that are part of or will be part of the return signal open AUX4 to the desired level, in most cases nominal level.

Please note all AUX levels depend on and are stored with the selected INPUT so if you switch inputs on a channel you have adjust that level also!

Setup your Talkback microphone in the SYSTEMS menu 'TB input ch:' as described earlier.

## GPIO



Select the purple GPIO button and set your 'Ringtone detect' GPI and 'Modem on' both to '1'. Standard for an Eela Audio hybrid the shape and type of the GPIO's are continuous so do not select any checkboxes in the main GPIO settings menu (see page 7)

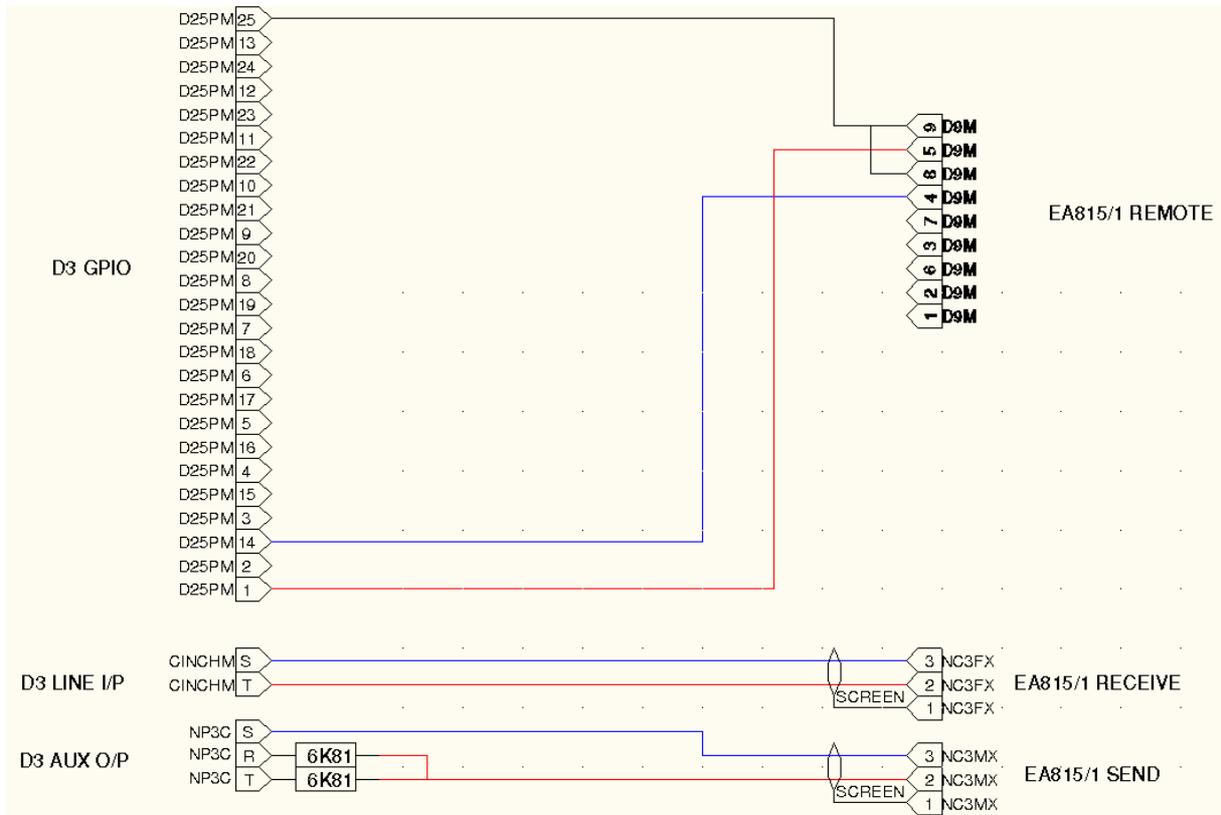
An incoming call is indicated by a fast flashing LED in the grey "select" button. In case this LED is flashing slowly it indicates an "Off Hook" / ON state of your hybrid.

In "Input select" mode of this channel 5 a yellow box with a telephone is visible. Touching this icon will open the telephone screen to put the hybrid on and off hook and to dial a telephone number. Dialing is done with an internal DTM tone generator and most hybrids are transparent for these tones.

Make sure not to use the same GPI or GPO on other channels for other purposes!

You can connect up to 4 hybrids this way each using another AUX for the return signal. Make sure the AUX volume of the corresponding input is closed and all others open.

D3 / EA815/1 telephone hybrid cable diagram:



EA815/1 hybrid remote is connected to GPIO 1

D3 Remote PC software:

PC software to be used for firmware update and store settings in PRESET 1.



PC remote software to save settings in preset 1.