OPERATIONS MANUAL

b40 series

b40
b41
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4ch digital audio leveller

b46

LEVEL MAGIC ™

release 3.0
LM2

jünger audio

JÜNGER AUDIO - Studiotechnik GmbH
Justus-von-Liebig-Str. 7, D - 12489 Berlin, GERMANY
Tel.: +49 30 6777210, Fax.: +49 30 67772146

www.junger-audio.com

FOREWORD

0

Thank you for buying and for using the 4-channel Digital Audio Level Processor b46.

Not only you have aquired the latest generation of digital dynamic range processing, but also a piece of equipment which is unique in its design and specification.

Please read this manual carefully to ensure you have all the information you need to use the 4-channel Digital Audio Level Processor b46.

The unit was manufactured to the highest industrial standards and went through extensive quality control checks before it was supplied.

If you have any comments or questions about installing, setting-up or using the b46, please do not hesitate to contact us.

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CONTENTS

FUNCTION DESCRIPTION

2

2.1 BASIC DESCRIPTION

The digital dynamics processor b46 is a professional studio device that is performing automated levelling of digital audio signals.

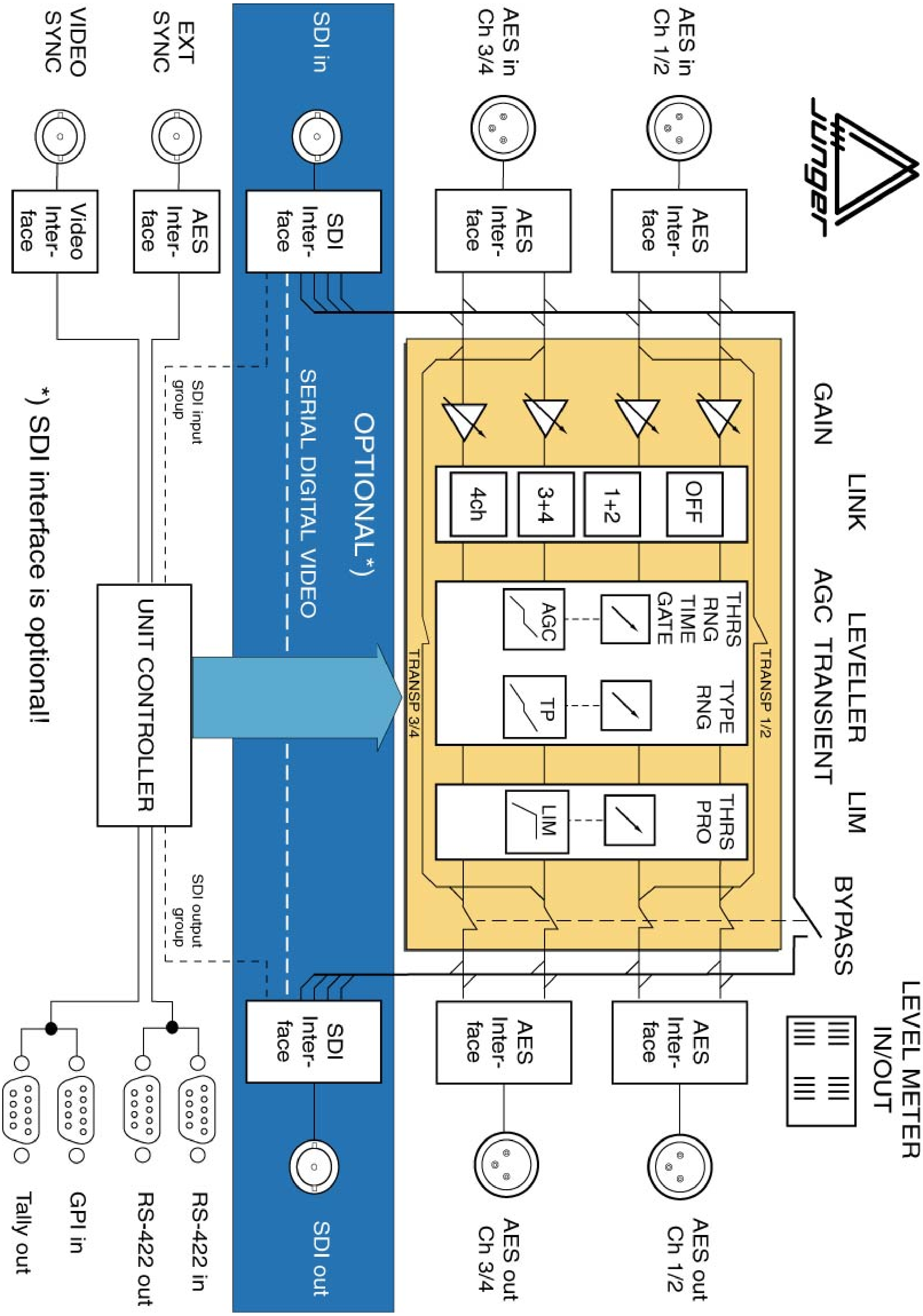
The dynamic range processor principles developed by Jünger Audio enable level managing devices like compressors, AGC and limiters to be produced with exceptionally high audio quality, without coloration, pumping, breathing, distortion or modulation effects sometimes associated with this type of processor.
In short, almost inaudible processing - with ease of use. The outstanding quality of the processing is based on the Multi-Loop dynamic range control principle in combination with adaptive controlled processing algorithms developed by Jünger Audio.

The unit is easy to operate and requires only a limited number of settings to be made by the user to achieve optimum results. All other parameters necessary for inaudible processing are continuously automatically controlled in response to changes in the programme signal.

features

- 4-channel digital audio levelling processor
- various link modes: 4-ch, stereo 1/2 or 3/4, ch1...4 independent
- adjustable input gain (channel independent) -15...+15 dB
- adaptive controlled audio levelling processing
  AGC, Transient Processor, Limiter
- user friendly preset and recall function (10 presets)
- pairwise bit transparent mode input to output
- extern sync mode, AES/EBU or VIDEO (or SDI if optional SDI-interface is present)
- RS-422 interface for serial remote
- GPI interface for parallel remote control, tally output

2.2
BLOCK DIAGRAM



2.3 AUDIO SIGNAL PROCESSING

All signal processing is done in the digital domain by Texas Instruments floating point signal processors. The use of 32 bit word length for calculation ensures that there is no deterioration in signal quality, even if an audio signal with a maximum word length of 24 bit is input into the processing of the unit.

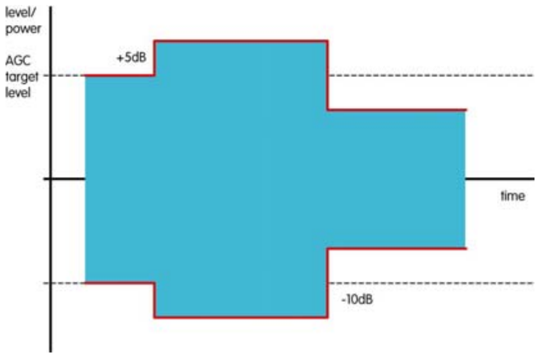
2.3.1 GAIN

GAIN means linear amplification of input signals. The input gain can be changed in steps of 0.1 dB , within a range from -15...+15 dB. Adjustment of GAIN is channel independent.

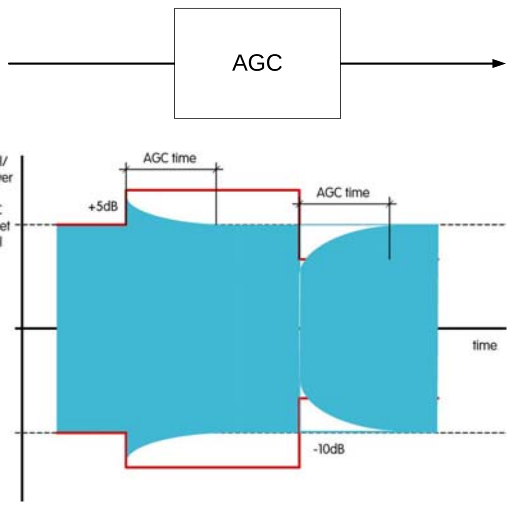
2.3.2 AUDIO LEVELLER LEVEL MAGIC ™

Level Magic ™ is a unique algorithm to make automated audio levelling possible.

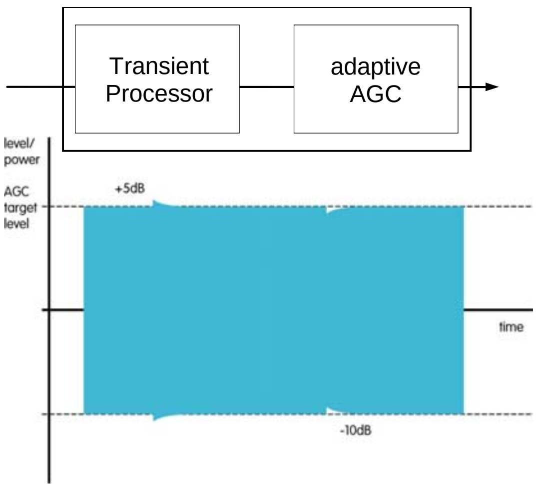
Input level change
Pic. 2 is showing a theoretical level change of +5dB and –5dB around program level.



Working with AGC
In pic.3 a conventional AGC is used to adjust the level. As we can see the AGC needs a certain time to react, that is necessary for mostly inaudible gain correction. But that's too long to get a proper correction of the input level change.



Level Magic ™
Level Magic ™ is a unique combination of a transient processor and an adaptive AGC process. The transient processor can fill the lack of level control against the slow acting AGC. The total gain of Level Magic ™ is the addition of the gain by the transient processor and the gain of the AGC.



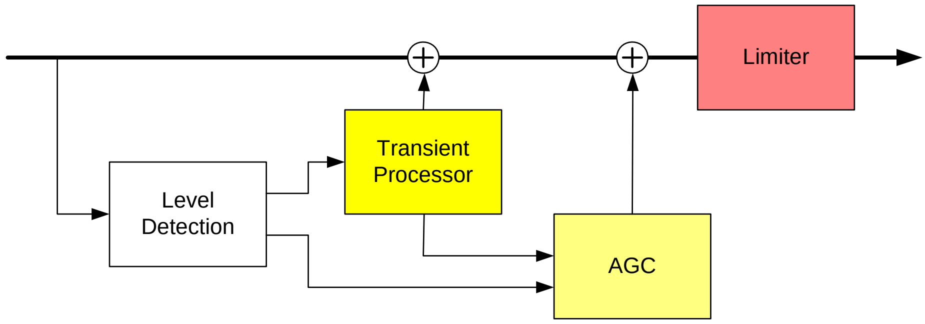
Adjustment procedure

The Level Magic ™ process needs to be setup in three steps

- select one of the default presets for your apllication (see preset description in chapter 5)
- adjust the operation level and peak level referring to standards that are using for your application
- if the default preset is not giving satisfying results change the parameters indivdually

Process description

Level Magic ™ is using a unique combination of QP and RMS level detectors to analyze the incoming audio signal. In comparing QP and RMS measurement results we can find out how much transients are coming in. Dependent on that the necessary resulting gain is controlled in relation between transient processor and AGC.



Transient processor is doing fast gain change and the AGC is doing slow gain change (depending on settings). The way how Level Magic is acting on the audio is mostly determined by balancing between slow and fast gain changing process. The AGC should be set in a way that the gain change is mostly inaudible (1dB per 5 seconds or slower). The Transient Processor should be set that incoming level jumps are reduced but originally dynamic range is not changed too much. As more possible gain by the Transient processor as more reduction of the dynamic range is coming with.

SOFT level control:	AGC	range …15dB, time >=2min
	Transient	range …4dB, soft process
MID level control:	AGC	range …12dB, time >=1min
	Transient	range …6-8dB, mid process
HARD level control:	AGC	range …10dB, time >=40sec
	Transient	range …10dB, hard process

Parameter description

Parameter description:

AGC

OP-level	operation level, target level for the AGC and for the Transient Processor
Range	max. gain by the AGC
Time	time to reach the max. gain change
Gate	threshold level that the AGC stops dynamic gain change and is moving gain slowly to the long term average gain change value

Transient Processor

Process	a combination of level ratio and release characteristic for the fast gain change (soft, mid, hard)
Range	max. gain by the Transient Processor

2.3.3 LIMITER

The static characteristics of the b46 limiter usually refers to a digital output level of 0 dBFS (dB Full Scale). This is useful for most applications of the dynamics processor as the on-following digital recording system is supposed to be balanced down to the final bit.
For applications using headroom the output level of can be adjusted within **0 ... -20 dBFS** in steps of 0.1 dB. The limiter threshold determines the maximum output level.
The static characteristics fo limiter (solid) at a limiter threshold of -12dBFS are illustrated in fig. 6.

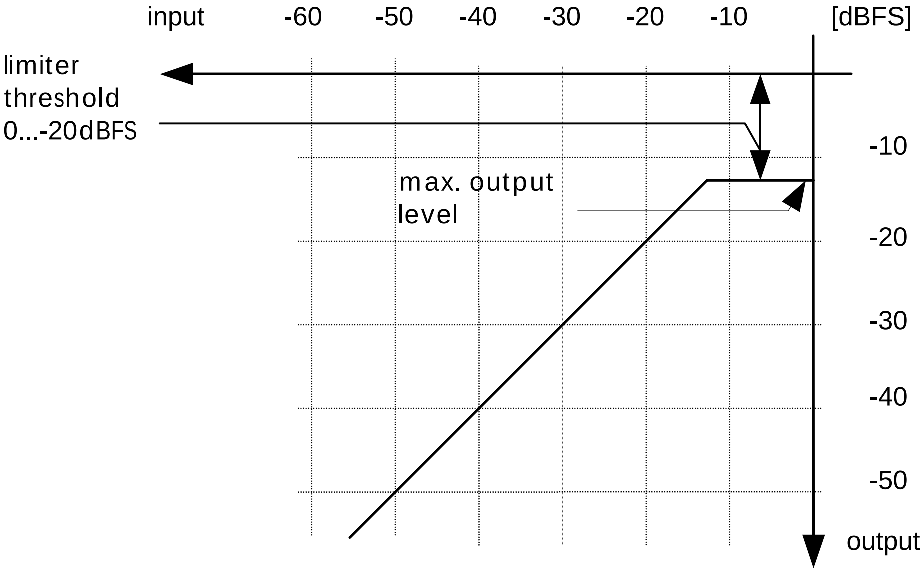


fig. 6: basic function: limiter

For the dynamics functions a **signal delay** of approx. 2 ms is built in. This delay makes it possible to arrange the algorithm of the limiter in such a way that the control mechanism is activated before maximum level is reached (look ahead limiter). Within the rise time of the signal the peak level is recognised and the maximum is calculated in such a way that full scale level is reached precisely without causing clipping.

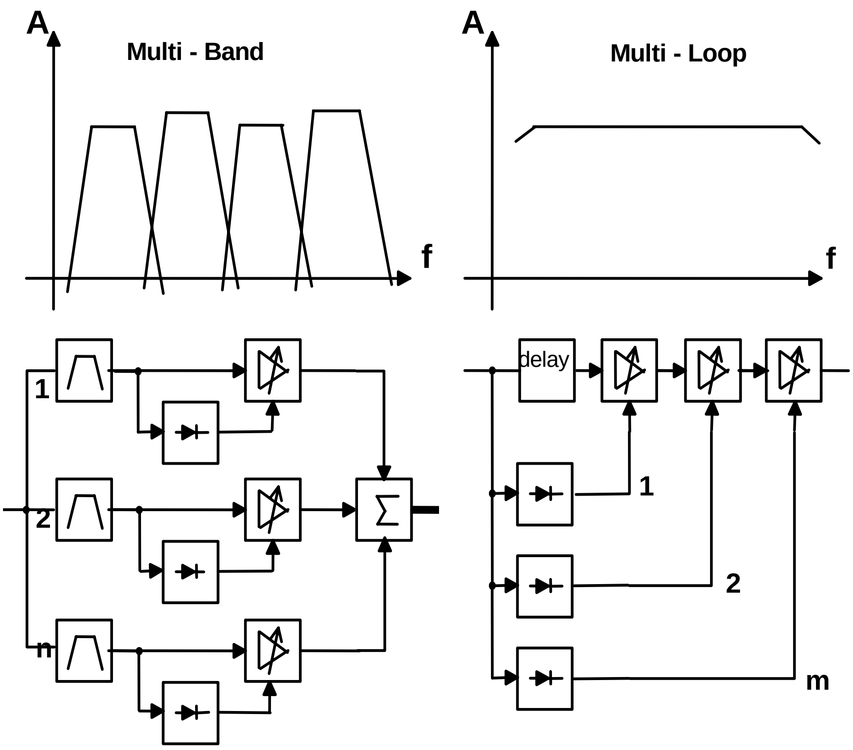
2.3.4 TRANSPARENT MODE

In case that the input signal (audio pair 1/2 or/and 3/4) is not audio (but AC-3, Dolby E, MPEG..) the input can be feeded directly to the related output bit transparent (no bit changes). The unit is switching to *transparent* automatically if *"non audio"* flag in the Channel Status Bit of the AES signal is set. Otherwise transparent mode can be set manually by the user.

2.4 THE JÜNGER AUDIO DYNAMICS PROCESSOR PRINCIPLE

A change in the dynamic range of an audio signal is a non-linear process. The gain of a dynamic range processor is not constant as it is with the gain of a linear amplifier. The gain varies in time depending on the input signal and depending on the specific control algorithm of the dynamics processor. These variations in the gain, which represent the real control process, should take place without any bothersome side effects.
The dynamic range processor principle developed by Jünger Audio makes it possible to realise dynamics processors (compressor, limiter, expander) with very high audio quality, without signal discolouration, pumping or breathing, without distortion and modulation products - in

short, with almost inaudible processing - and they are very easy to use. The Jünger Audio dynamics processors work according to a **Multi-loop principle**, operating with an interaction between several frequency linear control circuits. This is quite different to the popular multiband structure which changes the sound.



The resulting attack and release times of the Multi-loop-system are variable and adapted to the evolution of the input signal. This allows relatively long attack times during steady-state signal conditions but also very short attack times when there are impulsive input transients.
The Multi-loop structure also permits a short **time delay** between the control circuit and the gain changing element. The gain control circuit has time to preview the signal and become active before it reaches the output. This is particularly important for the limiter, which provides a precisely leveled output signal absolutely free of overshoots (clipping).

2.4.1 PROGRAM

For some of the control parameter it is possible to define a limited range of time constant values which is allowed for the adaptive dynamic range algorithms. Inside this range the time constants can be varied by the adaptive processing. Setting the range of time constant values may be sometimes useful, to get the best signal processing performance regarding specific programme material.

Parameter related to the transient response of the control circuit are important for distortionfree processing. These time constants are allways adaptive controlled without remarkable limitation of parameter range. This is caused by the presence of transient pulses in allmost each kind of programme material. The algorithm has to guarantee best reaction for fast increasing level of transient signals anytime even if classical music with slow dying out characteristic is processed. In all

cases the attack time of the limiter for very short transients is zero. Especially the release time of the control circuit has more influence to the increase of loudness as any other parameter. The ranging of time constants in processing time groups reflects this fact. The range for processing time shows influence on release time parameter mostly.
The selection of the parameter **PROGRAM** changes the range of time constant values as follows:

PRO	processing time	corresponds to preset
0	2 ms to 0.2 sec	
1	5 ms to 0.5 sec	LIVE
2	10 ms to 0.8 sec	
3	15 ms to 1.2 sec	SPEECH
4	30 ms to 2.5 sec	POP
5	50 ms to 3.5 sec	
6	70 ms to 5.0 sec	UNIVERSAL
7	100 ms to 6.0 sec	
8	150 ms to 8.0 sec	CLASSIC
9	250 ms to 10.0 sec	

2.4.2 INFLUENCE OF SIGNAL DELAY TIME

The audio signal delay through the dynamics processor is approx. 2ms due to delaying of the audio signal using internal memory. A small delay is deliberately introduced to the audio signal in order to allow limiter and compressor algorithms which can 'preview' the audio signal before changing it. That is the signal curve can be changed before maximum level is reached. This delay must be considered before attempting to mix signals processed by the dynamics processor with other undelayed signals.
When mixing together a delayed signal and a direct signal there may be cancellation of the signal waveform at some frequencies and re-inforcement of the waveform at other frequencies (comb filter effect). Corresponding 2ms delay of direct signals should therefore be carried out before mixing them with delayed processed signals.

3.7 REMOTE CONTROL

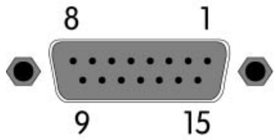
3.7.1 GPI REMOTE CONTROL (PARALLEL REMOTE)

The digital audio level processor b46 can be remote-controlled by means of parallel GPI contacts.

use : remote-controlled changeover of presets

connector: D-SUB 15pin, female

Pin assignments



Pin	Signal name	Logic	I/O	Functions
1	PRESET1	L	I	recall preset1
2	PRESET2	L	I	recall preset2
3	PRESET3	L	I	recall preset3
4	PRESET4	L	I	recall preset4
5	not used	L		
6	BYPASS	L	I	bypass on
7	Transp12	L	I	Input 1/2 transparent
8	Transp34	L	I	Input 3/4 transparent
9	SDI12	L	I	Input 1/2 on SDI
10	SDI34	L	I	Input 3/4 on SDI
11	not used			
12	not used			
13	not used			
14	Common pin			External voltage feed
15	+5V		O	Test power source

Electrical specification:

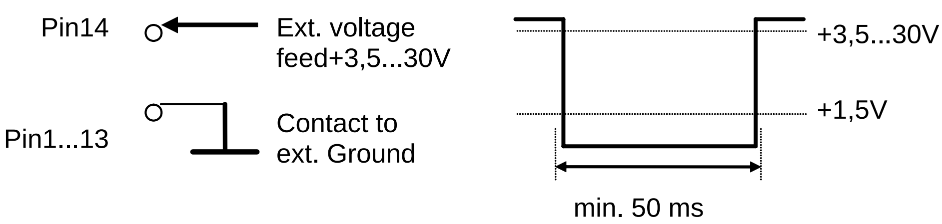
GPI input potential free by opto-coupler, **low active**

OFF: +3.5…+30V between GPI input and pin14

ON: less then 1.5V

min 50ms

Note: If using an external voltage feed it has to be connected to pin 14! External Ground is switching the GPI on any of the inputs.
An internal voltage feed is available on pin 15. Ground is available from the shield of the connector only! By using the internal voltage feed there is no electrical isolation given anymore.



LOCATION OF PARTS AND CONTROLS

4

4.1. FRONT PANEL

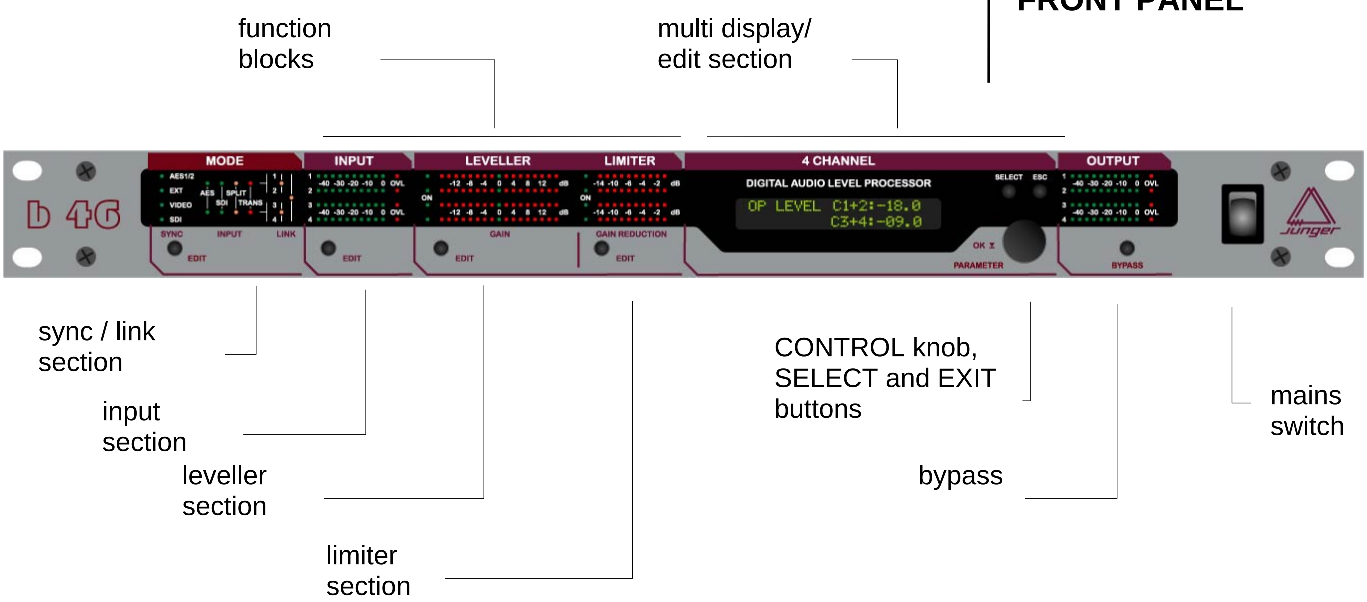


fig1: front panel b46

CONTROL ELEMENTS

MODE	selection/adjustment of sync, link and SDI split mode
INPUT	selection/adjustment of input and SDI parameter (group selection)
LEVELLER	selection/adjustment of leveller parameter
LIMITER	selection/adjustment of limiter parameter

CONTROL knob	selection (push) and adjustment (turn) of processing parameter
SELECT/ ENTER	selection of channels (while editing process parameters) selection of utility menus (ENTER) for recall and store of presets
ESC	exit of adjustment menus and return to level display
GAIN	selection/adjustment of gain parameter
BYPASS	switch for general bypass of the unit

4.2.
REAR PANEL

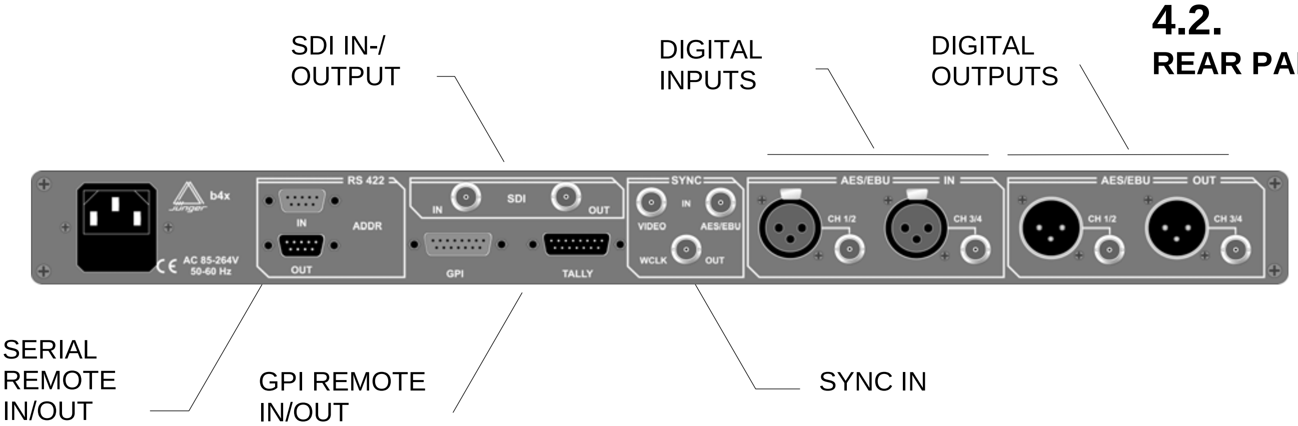


fig. 2: rear panel b46

POWER INPUT
IEC mains input connector 85-264V, 50/60 Hz with integrated fuse

REMOTE
serial remote interface RS-422
connector: 9pin SUB-D, input - female, output - male

GPI
parallel remote interface
TALLY-out open relais contact
connector: 15pin SUB-D, male
GPI-in +3,5…+30V potential-free
connector: 15pin SUB-D, female

SYNC
AES/EBU input for ext. sync signal (AES 3 format, 75 Ohm, unbal)
connector: BNC socket
VIDEO input for video sync signal (blackburst, 75 Ohm, unbal)
connector: BNC socket
W-CLOCK output for wordclock sync signal, TTL level, unbal.
connector: BNC socket

SDI IN / OUT (only if installed!)
Input/output for serial digital video (ITU-R BT.601, SMPTE 272M-A)
with embedded audio
Format: 270 Mb/s, 525/625 line rate, 75 Ohm,
connector: BNC socket

DIGITAL IN
input for AES/EBU standard format
connector: XLR female panel jack
1- ground, 2-3 signal, balanced
connector: BNC socket 75 Ohm, unbalanced

DIGITAL OUT
output for AES/EBU standard format
connector: XLR male panel jack
1- ground, 2-3 signal, balanced , 4 Vpp
connector: BNC socket 75 Ohm, unbalanced, 0.5V pp

4.3 SWITCHES AND JUMPERS FOR CONFIGURATION

Some basic settings are to select by switches on the rear panel or by switches and jumpers at the internal circuit boards of the unit. These settings can occur general changes for operation and should made by qualified engineering staff only.

Rear panel

Selection of the device address for serial remote, 16 device addresses selectable

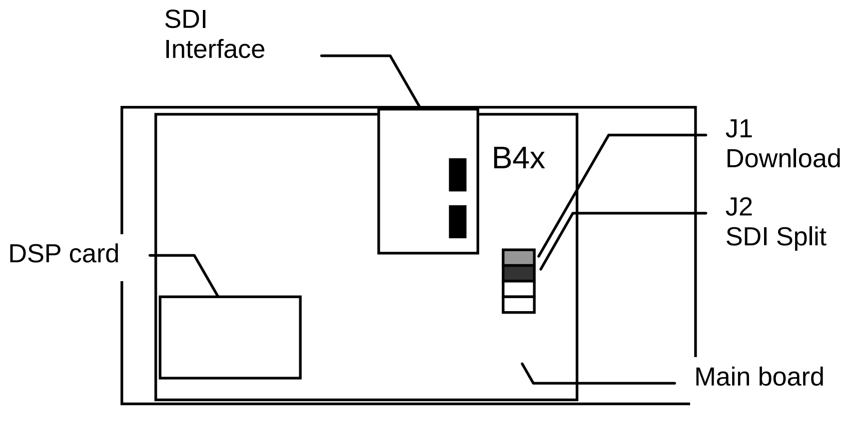
Note: Within a line of remote controlled units every device needs a different address! The selected address is valid after next power-on reset of the unit.

Internal

To set any internal jumper or switches it is necessary to open the unit.

PLEASE DO NOT MAKE ANY ALTERATIONS WITH THE MAINS STILL CONNECTED TO THE UNIT!

Loosen the screws on the top cover and remove. Then you can see all jumper and switches as shown in the drawing below. After setting of jumper or switches reassemble the unit in opposite order.



4.4 CONFIGURATION OF SDI INTERFACE

The 4-channel processors of b40 series fitted with SDI-interface are compatibel with the standard SMPTE 272M-AB. They support 48 kHz synchronous audio sampling with 20 bit word length.
The standard allows up to four groups each of four mono audio channels. (Usually used by most of D-VTR's and other equipment is Group 1 with 48 kHz synchronous sampling.)

Group selection and other settings are to configure with settings by front panel operation (mode section).

OPERATION

5

5.0 DESCRIPTION OF OPERATIONS

The use of the digital dynamics processor b46 is very easy.

The setup or the programming of the digital dynamics processor b46 is made by adjustment of various parameters and settings.

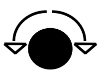
The description is made related to the functions in the menus.

- 5.1 adjustment of parameters
- 5.2 gain / loudness display
- 5.3 mode menu
- 5.4 input menu
- 5.5 leveller menu
- 5.6 limiter menu
- 5.7 utility menu
- 5.8 recall and storage of presets
- 5.9 editing of presets
- 5.10 list of factory presets

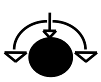
Following syntax is used:

SYMBOL	ACTIVITY
describes how to use button or rotary knob	describes action or function of button or rotary knob

push



turn



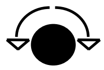
push + turn

5.1 ADJUSTMENT OF PARAMETERS in all menus

After selection of one of the utility or function menus by pushing any of the EDIT- buttons or the SELECT button one can adjust displayed parameters.


push

CONTROL switches between parameter selection and parameter adjustment mode, selected parameter or value is highlighted by arrows on display


turn

CONTROL change of parameter selection or adjustment of selected parameter value (see menu explanation)

Each time SELECT button is pushed it opens next utility menu. **If a function menu is opened (after pushing related EDIT button) the SELECT button changes the channel selection.** After finishing of settings ESC button switches back to main level display. All settings are stored as current adjustment automatically.

5.2 GAIN /LOUDNESS DISPLAY

The Loudness display becomes available if the loudness mode is switched ON (see 5.5, 5th menu item). The loudness display is showing short term loudness for the two channel pairs in numerique value in LKFS units. In case there is no loudness mode selected only the Gain display becomes available after pushing the SELECT button.
Gain display shows gain setting for all channels. You can jump to the gain menu from any other edit menu by pushing ESC button. The character on the left hand side of the display shows the selected loudness measurement method. “I” stays for ITU.1770 mode.
Adjustments are made by turning&pushing CONTROL knob as described previously (see 5.1).

```
  I12   -24.0    I34  -27.0

O12    -23.0   O34  -29.0
```

Ixx:	shortterm Loudness in LKFS of the input channel pair
Oxx:	shortterm Loudness in LKFS of the output channel pair

```
 GAIN  1:  0.0    3:  0.0

I >M:<   2:  0.0    4:  0.0
```

M:	master control, ganging level settings for all channels following channel 1
GAIN 1…4:	channel independent -15.0 ... +15.0 dB
I:	If the “**I**” shows up left hand of “**M:**” ITU weighting is turned on (see 5.5 – leveller menu)

5.3 MODE MENU

Mode menu shows sync setting and input selection.
Adjustments are made by pushing and turning CONTROL knob (see 5.1). Return to level display with EXIT.

SYNC<	LINK	
AES	1 + 2	3 + 4

SYNC MODE:	selection of sync signal input	
	CH 1/2	- sync on digital input 1/2
	EXT	- sync on external sync input
	VIDEO	- sync on video sync input
	SDI	- sync on SDI input
LINK MODE:	all channels independent or following link combinations: 1+2, 3+4, 1+2 & 3+4	

5.4 INPUT MENU

Input menu shows input setting of the unit. There are two windows available by pushing INPUT EDIT button once or twice.
Adjustments are made by pushing and turning CONTROL knob (see 5.1). Return to level display with EXIT.

1. menu

IN12<	TR12	IN34	TR34
AES	off	SDI	on

INxx:	selection of signal input	
	AES	digital input AES/EBU
	SDI	SDI input (embedded audio)
TRxx:	selection of transparent input	
	on/off	for bit transparent path between
	auto	input and output (for Dolby E)
		If set to AUTO the path is switched to transparent automatically if the non-audio flag in the AES/EBU or SDI signal is set.

2. menu (just if SDI interface is present)

SDI GROUPS:	> IN<	OUT
	1	1

IN:	selection of SDI group for deembedding input signals 1...4
OUT:	selection of SDI group for embedding output signals 1...4

5.5 LEVELLER MENU

Leveller menu shows leveller settings for selected channel . There are more windows available by pushing EDIT button of LEVELLER section repeatedly.
Adjustments are made by pushing and turning CONTROL knob (see 5.1). Return to level display with EXIT.

1. menu

```
PR  CH   >LVL<   LDTARGET

 01    2    ON        -24.0
```

PR:	number of current preset
CH:	selected channel (change with SELECT)
LVL:	leveller on/off
LDTARGET:	loudness target in LKFS (if ITU BS.1770 is ON) or operating level in dBFS (if ITU BS.1770 is OFF)

2. menu

```
PR  CH   >ZEROUP< ZERODN

01     2      +0dB        -0dB
```

ZEROUP :	Zero Zone treshold above loudness target
ZERODN:	Zero Zone threshold below loudness target

3. menu

```
PR  CH  >AGCMXGAIN<  TIME

 01    2       10dB          40s
```

AGCMXGAIN:	max. gain by the AGC
TIME:	AGC control time

4. menu

```
 PR  CH                    >FREEZE

01     2                     -50dBFS
```

FREEZE:	freeze threshold level for the AGC in dBFS

5. menu

```
 PR  CH  >TPMXGAIN<  RESP

 01    2     10dB          MID
```

TPMXGAIN:	max. gain by the Transient Processor (TP)
RESP:	TP response (slow, mid, hard)

6. menu

PR	**>ITU BS.1770<**
01	**off**

ITU BS.1770: Weighting of the leveller processing according to ITU BS.1770. If turned **on,** the Loudness Target becomes processing reference (instead of Operating Level). By default the box is changing Operating Level less 6dB to determine Loudness Target (and vice versa).
All measurment definition by ITU (see BS.1770 document for details).

7. menu

PROCESSING THR
-60

PROC THRESHOLD:
If the input signal is below this threshold all remaining GAIN will be taken out.
If the input signal returnes above threshold previous gain is applied again.
Pls. note this is not a PRESET parameter, but a global setting for the box!

5.6 LIMITER MENU

Limiter menu shows limiter settings for selected channel . Adjustments are made by pushing and turning CONTROL knob (see 5.1). Return to level display with EXIT.

PR	CH	>LIM<	THRS	PRO
01	2	ON	-9.0	1

PR: number of current preset
CH: selected channel (change with SELECT)
LIM: limiter on/off
THRS: limiter threshold level -20 ... 0 dBFS
PRO: selected program-preset for adaptive controlled algorithms

The selection of the parameter **PRO** in the limiter edit menu changes the range of time constant values as follows:

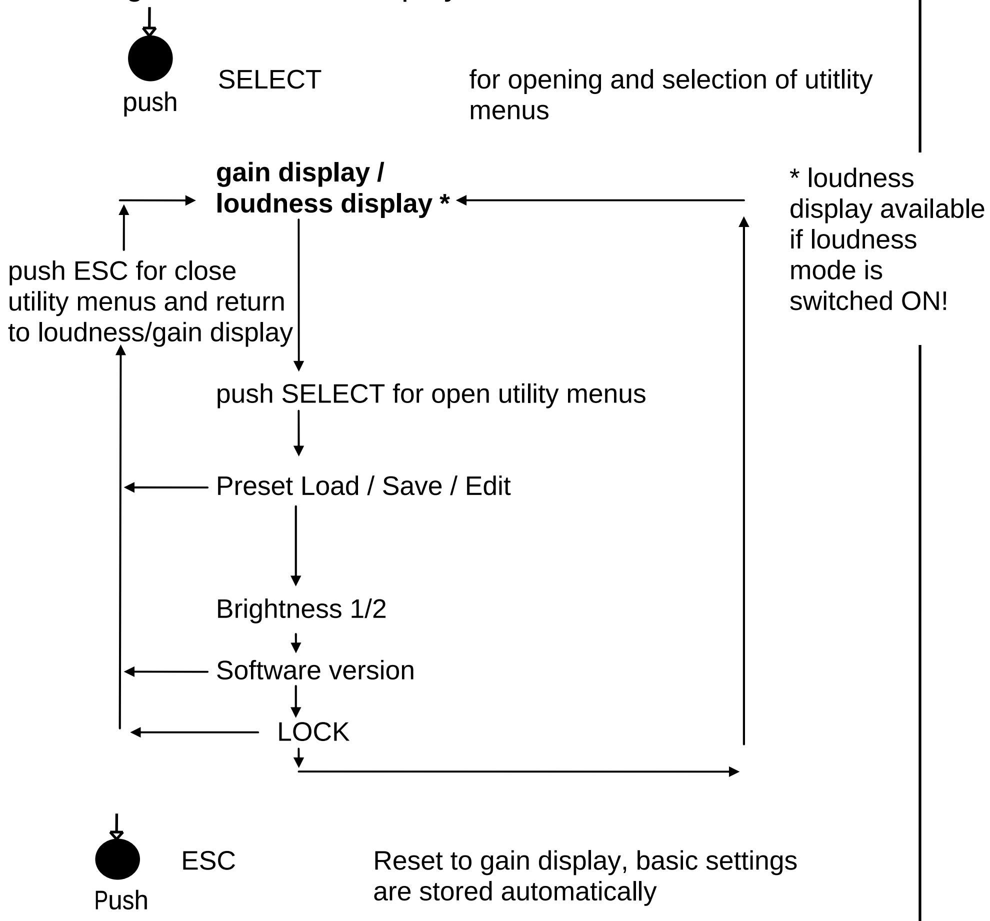
PRO	adaptive processing time	corresponds to preset
0	2 ms to 0.2 sec	
1	5 ms to 0.5 sec	LIVE
2	10 ms to 0.8 sec	
3	15 ms to 1.2 sec	SPEECH
4	30 ms to 2.5 sec	POP
5	50 ms to 3.5 sec	
6	70 ms to 5.0 sec	UNIVERSAL
7	100 ms to 6.0 sec	
8	150 ms to 8.0 sec	CLASSIC
9	250 ms to 10.0 sec	

The basic Multi-Loop principle of Jünger Audio dynamics processors operates with adaption of dynamic range control parameters to the incoming audio signal. That means permanently analysis and calculation of attack times, release times , thresholds and interaction parameters of several frequency linear control circuits.
(please refer to chapter 2 also)

Changing of PRO defines a limited range of time constant values which is allowed for the adaptive dynamic range algorithms. Inside this range the time constants can be varied by the adaptive processing. Setting the range of time constant values may be sometimes useful, to get the best signal processing performance regarding specific program material.

5.7 UTILITY MENU

For opening and selection of UTILITY menus when loudness/gain menu is on display.



BRIGHTNESS1:	display brightness when active (in use)
BRIGHTNESS2:	display brightness when in display save Mode (screen saver)
Software version	C: controller firmware version D: dsp firmware version
LOCK OFF/ON	LOCK ON: all front pannel knobs are locked, except the bypass-button To enter into configuration you have to enter the password (factory default 1234). The password can be changed by choosing the digit, pressing the knob, turniong the knob and choose the wanted number and pressing the knob again to confirm.

5.8 RECALL AND STORAGE OF PRESETS

All individual settings for the function blocks can be stored as presets. 10 presets are storable into the unit.

If the gain display is not visible push ESC button to switch back to gain display.

push	SELECT	selection of PRESET menu
turn	CONTROL	selection of LOAD or SAVE menu
push	SELECT	change to PRESET selection menu
turn	CONTROL	selection of preset number for loading or saving preset
push	SELECT/ ENTER	executes loading or saving of preset, exit preset menu

Push any other button for leaving the preset menu without loading or saving presets.

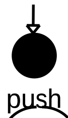
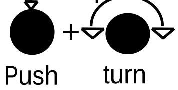
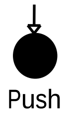
5.10 shows some useful PRESETS that are already coming as factory preset for applications with different audio formats:

5.9 EDITING OF PRESETS

All individual settings for the function blocks can be stored as presets. 10 presets are storable into the unit.
These presets can be changed off-line, that means without influencing running audio on the machine.
If the gain display is not visible push ESC button to switch back to gain display and then:

push	SELECT	selection of PRESET menu
turn	CONTROL	selection of EDIT menu
push	SELECT	change to PRESET selection menu
turn	CONTROL	selection of preset number for editing preset
push	SELECT/ ENTER	executes editing of presets incl. gain Setting. A blinking “E” in the display shows that EDIT mode is valid.

If all changes are done push the ESC button to switch back to gain display.

	SELECT	selection of SAVE or EXIT
	CONTROL	selection of preset number
	CONTROL	selection of preset number

Select EXIT for leaving the EDIT menu without saving presets.

5.10
PRESET LIST

Parameter	Range	Setup/ Preset	1…10 *TV uni*	
MODE Sync	1/ 2, EXT, Video	SETUP	-	
MODE Link	4 schemes	PRESET	1+2 & 3+4	
INPUT In	AES/SDI* for 1/2 and 3/4	SETUP/ GPI	-	
INPUT Transp	ON/OFF/ AUTO	SETUP/ GPI	-	
LEVELLER LVL	ON/OFF	PRESET	ON	
LEVELLER ITU.1770	ON/OFF	PRESET	ON	
LEVELLER Loudness Target	-40…0 LKFS	PRESET	-24	
LEVELLER Zero Zone Up	0…+6 dB	PRESET	0	
LEVELLER Zero Zone Down	-6…0 dB	PRESET	0	
LEVELLER AGC Max Gain	0…40 dB	PRESET	10	
LEVELLER AGC Time	1s…2h	PRESET	40s	
LEVELLER Freeze Level	-60…-20 dBFS	PRESET	-50	
LEVELLER Transient Proc Max Gain	0…15 dB	PRESET	10	
LEVELLER Transient Proc Response	3 modes soft, mid, hard	PRESET	Mid	
LIMITER LIM	ON/OFF	PRESET	ON	
LIMITER Threshold	0…-20.0 dBFS	PRESET	-5	
LIMITER Program	5 modes live, speech, uni, pop, classic	PRESET	uni	
Processing Threshold	-80…-40 dBFS	SETUP	-60	

BOOT DISPLAY AND TROUBLE SHOOTING

6

6.1 BOOT DISPLAY

display	meaning / explanation
AUDIO LEVEL PROCESSOR	display of model
C: x.x	display of loaded controller software version
D: x.x	display of loaded dsp software version

6.2 ERROR MESSAGES AND TROUBLE SHOOTING

display	error / message	remedies
NO SYNC	no sync at sync input!	■ connect the sync input (selectable in SYNC field) with valid input signal ➢ CH 1/2: sync on DIGITAL IN CH 1/2 ➢ EXT: sync on SYNC AES/EBU ➢ VIDEO: sync on SYNC VIDEO ➢ SDI: sync on SDI input
NO SDI!	SDI input selected, no valid SDI signal received!	■ check the availability of SDI data stream or ■ select another input

6.3 INITIALIZATION THE UNIT

Should have remained the device no more operable and/or in the program execution stand, recommends itself an initialization the device.
During initialization, all storage areas important for the program and registers are loaded with the factory setup and the program is restarted.

Any button is to be held pressed in order to initialize the device during switch-on of the device until the program started. To the start of the program and at the completion of the displays (how described in 7.1), the device is ready for operation with the factory setup.

After an initialization of the device, all user presets and adjustments are erased and/or overwritten by the factory setup!

APPLICATION NOTES

7

In digital video recording technology four digital audio channels are the standard configuration. This channel capacity is used increasingly in production and post-production for surround sound, providing mix options and for multi-lingual productions.

Quite often it is necessary to make corrections or changes to the audio which until now required the use of an expensive digital audio mixer. These tasks can now be easily solved with the Jünger Audio range of digital audio toolboxes. Simple processing for up to four digital audio signals may be carried out quickly and efficiently.

7.1
B40 SERIES WITH SDI-INTERFACE (SD or HD available)

Using the SDI versions (SDI=Serial Digital Interface, digital component video format with 270Mb/s transmission) b40 series can process embedded audio.
The standard allows up to four groups each of four mono audio channels. Usually used by most of D-VTR's and other equipment is Group 1 with 48 kHz synchronous sampling. Synchronous means that the audio clock is genlocked to the associated video. Each channel can have up to 20 bits of resolution per audio sample.
The 4-channel processors of b40 series fitted with SDI-interface are compatibel with the standard SMPTE 272M-A. They support 48 kHz synchronous audio sampling with 20 bit word length.

The Jünger Audio SDI interface provides for one group of four audio channels to be extracted from or inserted into the SDI data stream. To address a specific channel group the group selection is possible (see 4).

The b46 provides an optional SD- **or** HD-SDI board. When you switch on the device the plugged in interface will be indicated in the display

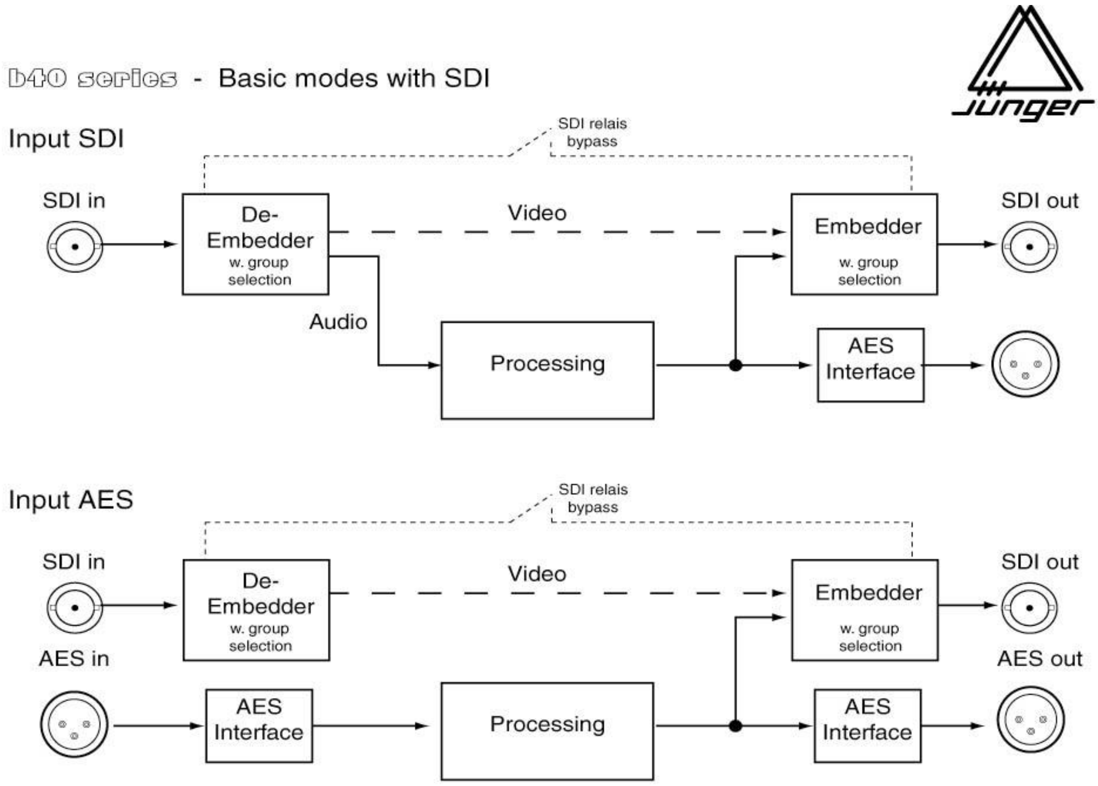
FEATURES

- Bypass relay for SDI IN >SDI OUT
- Bit transparent for coded data streams (e.g. DOLBYE/20bit)
- De-embedder: user selectable de-embedding of one group
- Embedder: user selectable embedding to one of 4 groups
- SDI-SYNC: SDI input can be the clock source of the device

- For HD-SDI: Multi-Format HD/SD operation with auto detection

7.2 BASIC WORKING MODES WITH SDI

For the basic working mode the input of the digital audio processing can be selected between AES/EBU or SDI (serial digital video with embedded audio). The processed signals are present at both outputs always - at AES/EBU and SDI.
There are two additional working modes using the SDI interface. SDI Bypass is bypassing the SDI data stream. In this case only extracted audio is processed and available at AES output. In Split Mode the audio path is splitted. Embedded audio can be processed with external equipment via AES interface.
Following illustration shows working modes:



TECHNICAL SPECIFICATIONS

8

digital signal processing

sample rate :	48 kHz
audio data format :	24 bit (AES/EBU), 20 bit (SDI)

DIGITAL IN/OUT

digital in- / outputs

AES/EBU

connector :	XLR,110 Ohm, balanced
	BNC, 75 Ohm, coaxial
input format :	AES professional, AES consumer
output format :	same as input format

SDI *(only for SDI version)*

SDI in- / outputs (optional)

SD-SDI

VIDEO :

standard:	SMPTE 272 M–A,	270 Mbit	SD-SDI
connection:	BNC, 75 Ohm, coaxial		
signal level:	800mV ±10%		
equalisation:	300m (Belden 8281 , 270 MHz)		
return loss:	>15 dB		

supported video standards:

SD 525/59.94	SMPTE 125M
SD 625/50	SMPTE 125M

AUDIO :

audio data format :	20 Bit, transparent for C-Bit and U-Bit according to AES3
audio sample rate :	48 kHz synchronous to video-carrier
latency :	(deembedder + embedder)
	SD : < 2,6 msec

GENERAL :

power supply :	+5V DC
consumption :	approx. 500 mA
dimension :	3RU, 4HP, 160mm depth (EUROPA size pcb)
temperature :	10°C to 40°C
humidity :	90%, non condensing

HD-SDI

technical specifications

VIDEO :

standard:	SMPTE 299M	1,485 Gbit	HD-SDI
	SMPTE 272M–A, C	270 Mbit	SD-SDI
connection:	BNC, 75 Ohm, coaxial		
signal level:	800mV ±10%		
equalisation:	130m (Belden 1694A, 1.485GHz)		
	300m (Belden 8281 , 270 MHz)		
return loss:	>15 dB (1.485 GHz)		

supported video standards:

HD 720/60	SMPTE 296M	HD 1080/25	SMPTE 274M
HD 720/50	SMPTE 296M	HD 1080/24	SMPTE 274M
HD 720/30	SMPTE 296M	HD 1080/50	SMPTE 295M
HD 720/25	SMPTE 296M	HD 1035/60	SMPTE 260M
HD 720/24	SMPTE 296M		
HD 1080/60	SMPTE 274M	SD 525/59.94	SMPTE 125M
HD 1080/50	SMPTE 274M	SD 625/50	SMPTE 125M
HD 1080/30	SMPTE 274M		

all HD-standards are supported also with their 1/1001-frame-rates

AUDIO :

audio data format :	24 Bit, transparent for C-Bit and U-Bit according to AES3
audio sample rate :	48 kHz synchronous to video-carrier (SD and HD) 32 kHz ... 48 kHz asynchronous to video-carrier (HD only)
latency :	(deembedder + embedder) HD : < 800µsec SD : < 2,6 msec

GENERAL :

power supply :	+5V DC
consumption :	approx. 1.000 mA
dimension :	3RU, 4HP, 160mm depth (EUROPA size pcb)
temperature :	10°C to 40°C
humidity:	90%, non condensing

sync in- / outputs

SYNC IN

AES/EBU

connector :	BNC, 75 Ohm, coaxial
level :	0,5 ... 5 Vpp
input format :	AES professional, AES consumer

VIDEO

connector :	BNC, 75 Ohm, coaxial
level :	0,5...1 Vpp
input format :	Blackburst or PAL/NTSC composite video

REMOTE

remote control

serial remote interface RS-422 in/out

	level :	TTL
	connector :	9 pin SUB-D male/female
GPI parallel remote		
	level :	+3…+30V, H-active, optocoupler
	connector :	15 pin SUB-D female
Tally Out	level :	normally closed relais contacts
	Contact rating:	1A 24 VDC, 0,5 A 125 VAC max. 30 W 62,5 VA max. 60 VDC, 125 VAC
	connector :	15 pin SUB-D male

GENERAL

power consumption :	appr. 15 VA
dimensions :	19“, 1 RU, 250 mm depth
weight :	appr. 5 kg

9

WARRANTY AND SERVICE INFORMATION

JÜNGER AUDIO grants a two-year warranty on the

4-channel digital audio leveller b46

If the unit has to be serviced, please send it, ideally in the original box, to:

JÜNGER AUDIO - Studiotechnik GmbH

Justus-von-Liebig-Str. 7

D - 12489 Berlin
GERMANY

Tel.: (*49) -30-677721-0
Fax.: (*49) -30-677721-46

KONFORMITÄTSERKLÄRUNG

DECLARATION OF CONFORMITY

Geräteart:	Digitaler Dynamikprozessor
Type of equipment:	digital dynamics processor

Produkt / Product: b46

Das bezeichnete Produkt stimmt mit den Vorschriften folgender EU-Richtlinie(n) überein:
The aforementioned product complies with the following Europaen Council Directive(s):

89/336/EWG (geändert durch 91/263/EWG und 92/31/EWG)
(changed by 91/263/EWG and 92/31/EWG)
Richtlinie der Rates zur Angleichung der Rechtsvorschriften der Mitgliedsstaaten über die elektromagnetische Verträglichkeit
Council Directive 89/336/EC on the approximation of the laws of the Member States relating to electromagnetic compatibility

Zur vollständigen Einhaltung dieser Richtlinie(n) wurden folgende Normen herangezogen:
To fully comply with this(these) Directive(s), the following standards have been used:

EN 55022 :1987
EN 50082-1 :1993

Dieser Erklärung liegt zugrunde:	Prüfbericht(e) des EMV-Prüflabors
This certification is based on:	Test report(s) generated by EMC-test laboratory
MEB Messelektronik Berlin	Kalibrier- und Prüflabor accredited EMC laboratory
Aussteller / Holder of certificate:	Jünger Audio Studiotechnik GmbH Justus-von-Liebig-Strasse 7 D - 12489 Berlin

Berlin,	18.03.2003	....
(Ort/Place)	(Datum/Date)	(Rechtsgültige Unterschrift / Legal Binding)

JÜNGER AUDIO - Studiotechnik GmbH
Justus-von-Liebig-Str. 7, D - 12489 Berlin, GERMANY
Tel.: +49 30 6777210, Fax.: +49 30 67772146

www.junger-audio.com