b42

b42

4 ch dynamics processor

Manual

release 2.1.1 / 2007-09-04

FOREWORD

0

Thank you for buying and for using the 4-channel Digital Dynamics Processor b42.

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 Dynamics Processor b42.

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 b42, please do not hesitate to contact us.

1

CONTENTS

FUNCTION DESCRIPTION

2

2.1 BASIC DESCRIPTION

The digital dynamics processor b42 is a professional studio device that processes the dynamic range of digital audio signals.

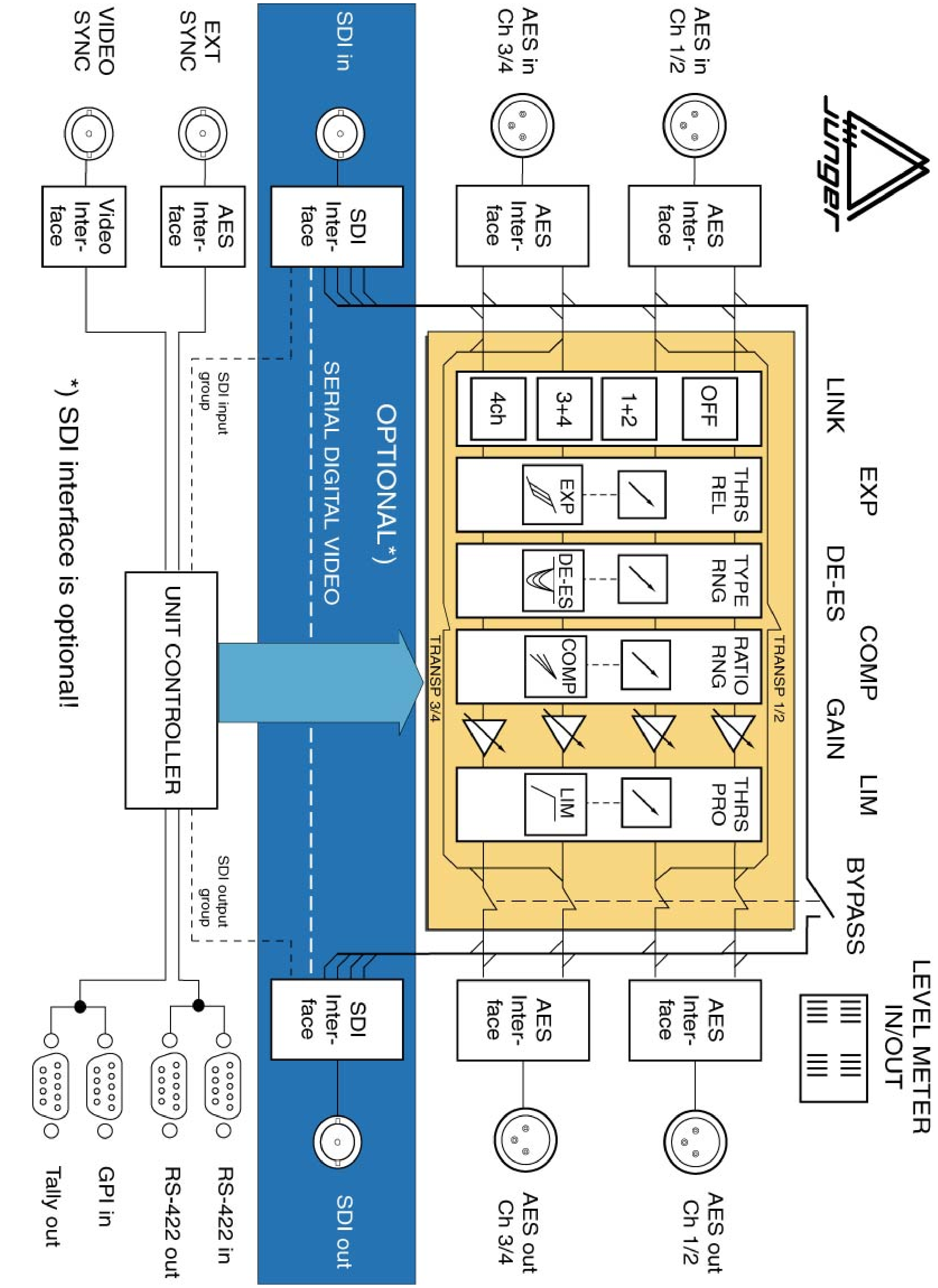
The dynamic range processor principles developed by Jünger Audio enable compressors, limiters and expanders to be produced with exceptionally high audio quality, without coloration, pumping, breathing, distortion or modulation effects sometimes associated with this type of processor.
In short, almost inaudible processing - with ease of use. The outstanding quality of the processing is based on the Multi-Loop dynamic range control principle 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 dynamics 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 dynamic range processing
  - expander on/off, THRS -50...-20 dBFS, REL ...
  - de-esser on/off, TYPE male/female, RNG -20...0 dB
  - compressor on/off, RATIO 1,0:1...4,0:1, RNG 0...15 dB
  - limiter on/off, THRS 0...-20 dBFS, PROgram 1..4
- 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 or output signals. The input or output 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 EXPANDER

Below an adjustable threshold level an **expander** can be activated which can lower the amount of noise signals.

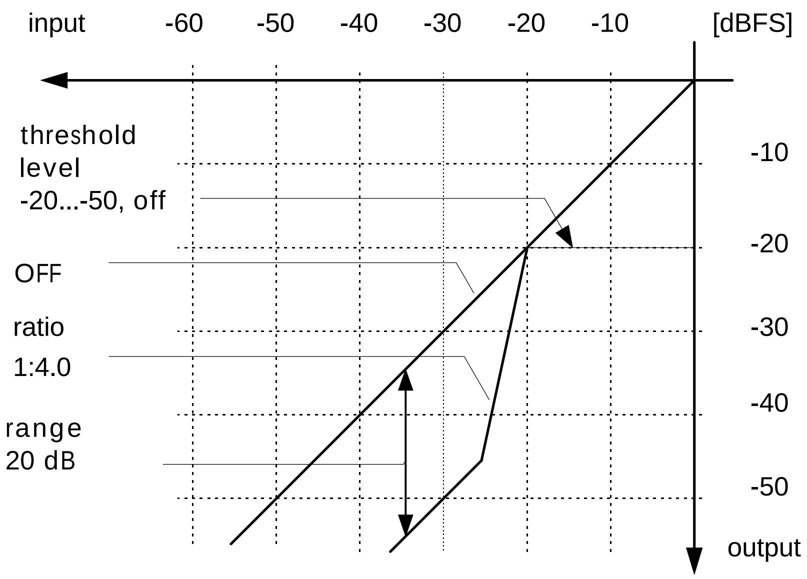


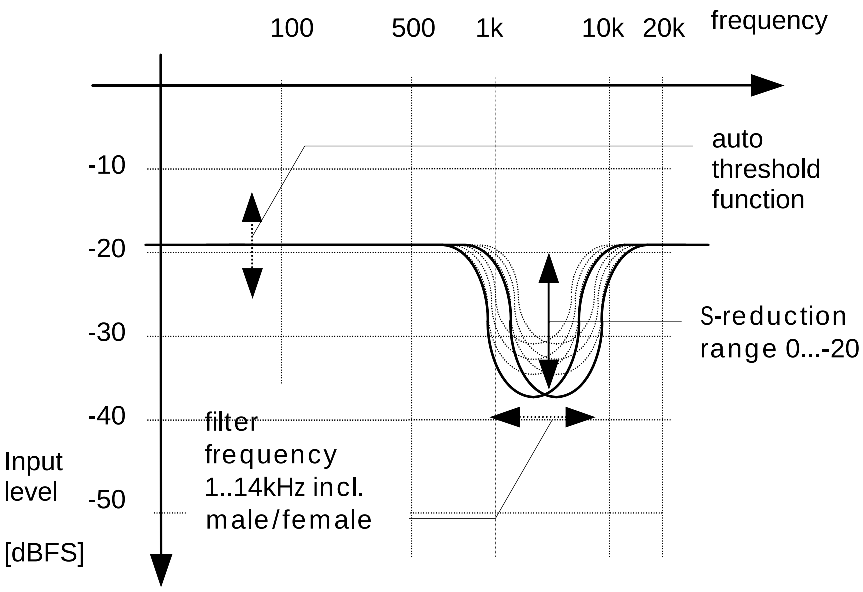
fig. 2: static characteristics: expander

2.3.3 DE-ESSER

The **de-esser** is a special processing function to reduce S-frequencies of speakers. This can be done either by using a compressor with frequency selective side chain, or by dynamic filtering of voice signals.

The de-esser of the b42 uses a sophisticated **dynamic filtering** algorithm for the reduction of S-frequencies. The dynamic filter makes it possible to reduce these frequencies without influencing other spectral parts, and works independent of the signal level.
The critical S-frequencies are different for female and male voices.
Only two basic adjustments are necessary for the de-esser - filter frequency and the amount of s-reduction (range).
All other parameters which are necessary for effective de-esser function are controlled by the audio signal itself.
The threshold of the de-esser is automatically set and follows the signal power level. The reduction of S-frequencies can be controlled by setting the range parameter from 0...-20dB.

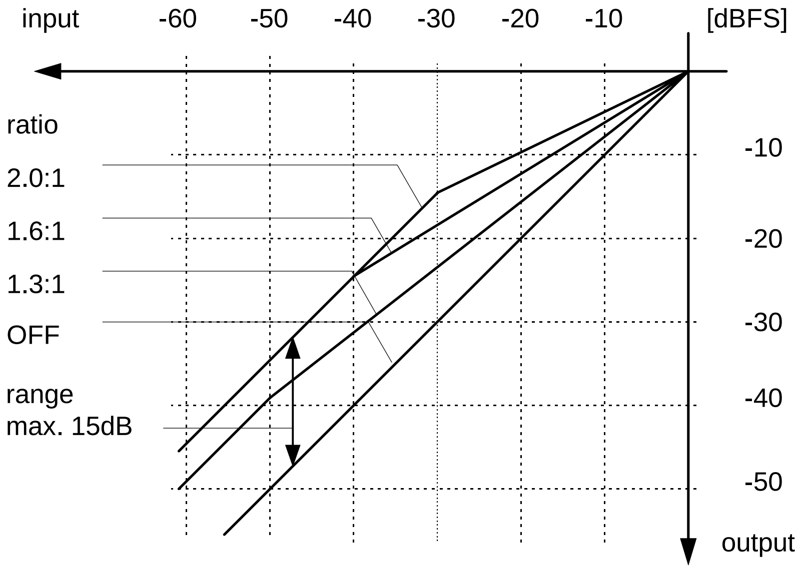
fig.3:
basic function:
de-esser



2.3.4 COMPRESSOR

The **compression** of the programme signal takes place evenly over the entire input level range and not only at the upper end above a certain threshold level. Dynamic structures of the input signal (e.g. dynamic evolutions) are converted proportionally so that even after compression the ratios are maintained, only slightly condensed, leaving on the whole a transparent, seemingly uncompressed sound impression.

fig.4:
static
characteristics:
compressor



Compression (reduction of the dynamic range of the input signal to match the dynamic range of the storage or of the transmission system) is partly achieved by increasing the level of low level signals, the lowest of which might otherwise be below the noise floor of the audio system.
The lower the input signal level the higher the additional gain applied to that input signal by the compression processing will be.

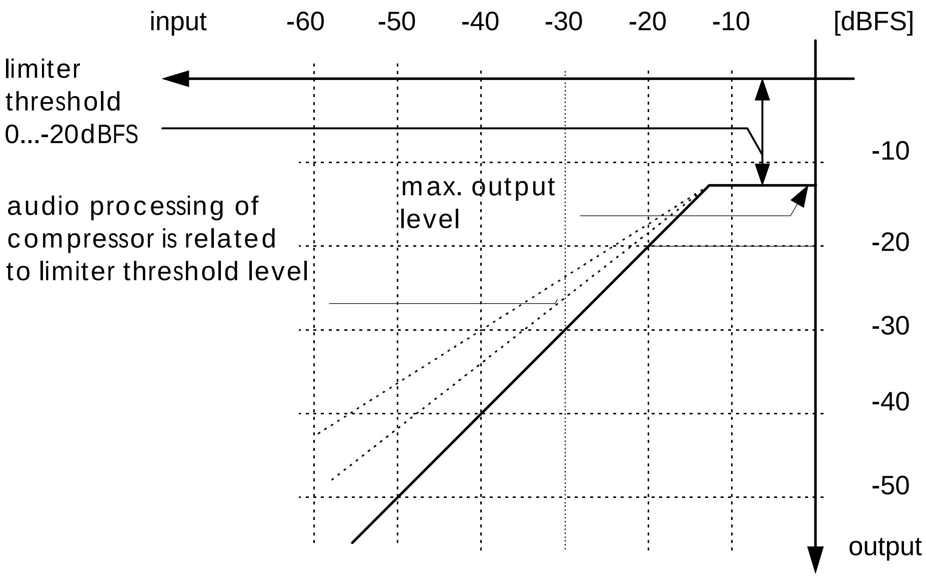
Independent of the compression ratio , the **gain of the compressor (range)** can be limited (maximum 1 dB to 15dB), so that there should be no inadmissible increase of background noises during signal pauses (e.g. live atmos, air-conditioning, hum and noise).

2.3.5 LIMITER

The static characteristics of the b42 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) and compressor (dotted) at a limiter threshold of -12 dBFS are illustrated in fig. 5.

fig. 5: 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.

2.3.6 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	

2.3.7 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.

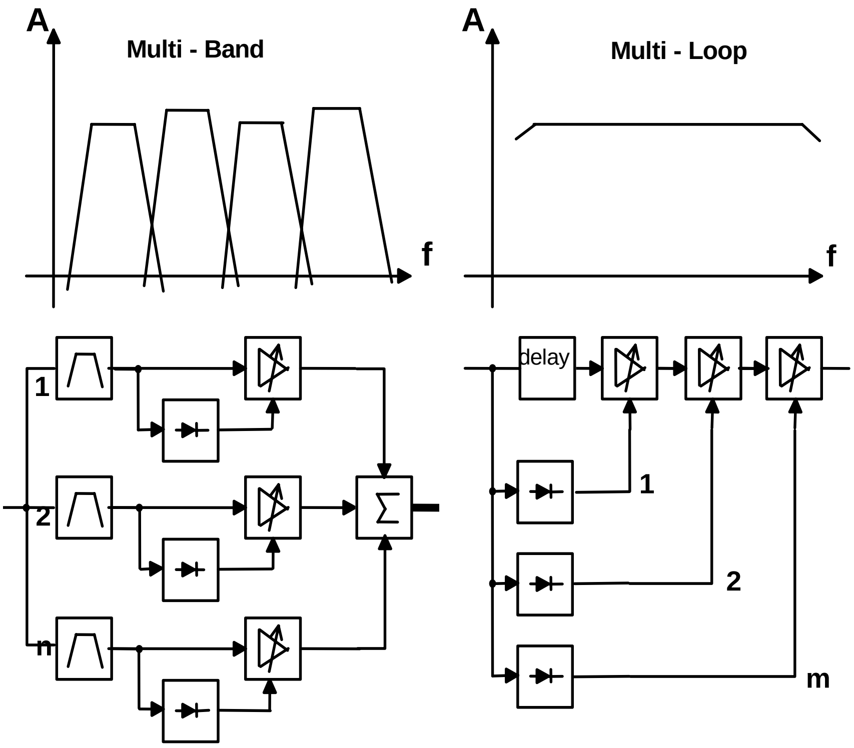
2.4 THE JÜNGER AUDIO DYNAMICS PROCESSOR PRINCIPLE

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 SELECTION OF PARAMETERS TO INCREASE LOUDNESS

Signal compression and the loudness enhancement of the digital audio signal can be achieved by combining two dynamic range control processes: firstly, the **compression** achieved by increasing small and medium signal levels and secondly, **linear amplification** combined with the inaudible **limitation** of individual, remaining peak levels with the limiter.

In the gain change mode the operation of compressor and limiter can be observed on the display. For smaller signal levels the compressor causes additional amplification which however decreases the higher the signal level is . With full scale levels the compressor is practically ineffective so that even an increase of the RATIO will have no effect.
If you now increase the linear amplification GAIN, individual peak levels are raised above the limiter threshold and limited inaudibly. All other signal components can however be increased. If the gain is too large also medium levels are treated by the limiter, which means that the limiter then reduces the signals continually and again reduces the additionally applied amplification.

The display for Limiter-Gain-Reduction should be in the region of 0....-6 dB and should not light up red continuously, so that a dynamic limitation only applies to signal peaks.
Then the signal compression and therefore also the increase of loudness is at its most effective.

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.

INSTALLATION

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3.1 UNPACK THE UNIT

The digital dynamics processor b42 was carefully packed in the factory and the packaging was designed to protect the equipment from rough handling. Please examine carefully the packaging and its contents for any signs of physical damage, which may have occured in transit.

3.2 POWER SUPPLY

The digital dynamics processor b42 is a device under the safety category *Schutzklasse 1* in keeping with the VDE 0804 standards and may only used with power supply installations built according to regulations.
Check the voltage details printed at the rear panel are the same as your local mains electricity supply.

3.3 CONNECTIONS

The dynamics processor b42 is equipped with standard connectors (see also chapter 3).
Before connecting the digital dynamics processor b42 switch the power off at all connected units.

3.4 RACK MOUNTING

The digital dynamics processor b42 is made as standard 19“ unit (EIA format). It occupies 1 RU (44 mm height) space in a rack.
Please allow at least additional 3“ depth for the connectors on the rear panel.
When installing the unit in a 19“ rack the rear side of the unit needs some support, especially for mounting in flight cases.

3.5 OPERATION SAFETY

The digital dynamics processor b42 should not be installed near units which produce strong magnetic fields or extreme heat. Do not install the filter processor directly above or below power amplifiers.
If, during operation, the sound is interrupted or displays no longer illuminate, or if abnormal odor or smoke is detected immediately disconnect the power cord plug and contact your dealer or Jünger Audio.

3.6 SYNCHRONIZATION OF DIGITAL OUTPUT

The digital dynamics processor b42 has a digital signal output only. To the problem-free combination of following digital devices, the digital signal processing can be locked to an external clock reference. The selection of the corresponding input is made in the SYNC MODE menu. If the chosen sync input is connected with the sync signal, this signal is used for synchronization automatically. The digital output signal can be clocked with the following clock frequencies:

CH 1/2 locks with the clock frequency of the input signal at digital input CH 1/2 (AES/EBU, 48 kHz)
EXT SYNC locks with the clock frequency at the external sync input (AES/EBU, 48 kHz)
VIDEO locks with the clock at the Video sync input (internal 48 kHz)
SDI VIDEO locks with the clock at the SDI input (internal 48 kHz)

Both digital outputs CH 1/2 and CH 3/4 are locked with same clock frequency.

Note: SDI sync is available only if SDI interface is installed!

3.7 REMOTE CONTROL

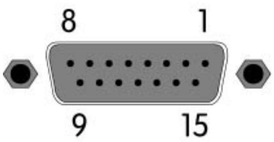
3.7.1 GPI REMOTE CONTROL (PARALLEL REMOTE)

The digital dynamics processor b42 can be remote-controlled by means of parallel GPI contacts.

use : remote-controlled changeover of presets

connector: D-SUB 15pin, female

Pin assignments

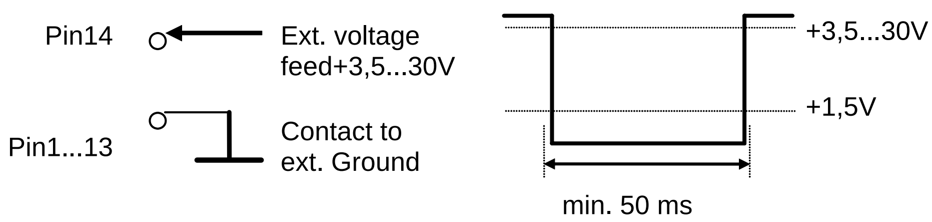


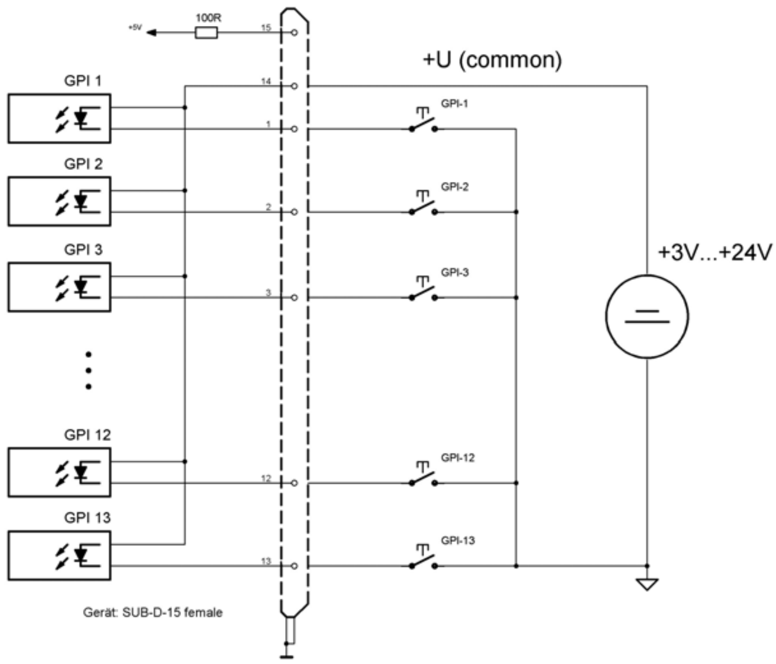
Pin	Signal name	Logic	I/O	Functions
1	PRESET1	H	I	recall preset1
2	PRESET2	H	I	recall preset2
3	PRESET3	H	I	recall preset3
4	PRESET4	H	I	recall preset4
5	PRESET5	H	I	recall preset5
6	PRESET6	H	I	recall preset6
7	PRESET7	H	I	recall preset7
8	PRESET8	H	I	recall preset8
9	MUTE	H	I	Muting outputs
10	BYPASS	H	I	bypass on
11	not used			
12	not used			
13	not used			
14	Common pin			External ground
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.





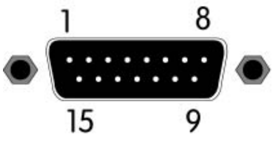
3.7.2 TALLY OUT

The digital dynamics processor b42 can transmit specific device statuses via parallel Tally lines.

use: Control of the remote-controlled changeover of presets

connector: D-SUB 15pin, male

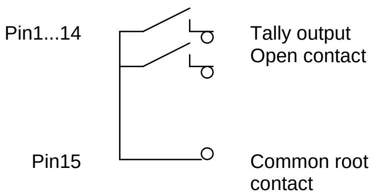
Pin assignments

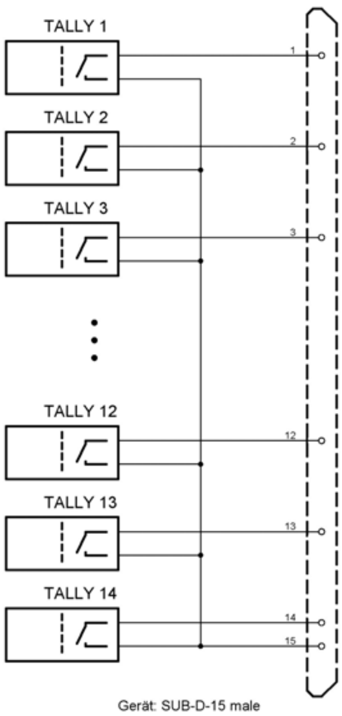


Pin	Signal name	I/O	Functions
1	T1 open contact	O	preset1 recalled
2	T2 open contact	O	preset2 recalled
3	T3 open contact	O	preset3 recalled
4	T4 open contact	O	preset4 recalled
5	T5 open contact	O	preset4 recalled
6	T6 open contact	O	preset4 recalled
7	T7 open contact	O	preset4 recalled
8	T8 open contact	O	preset4 recalled
9	T9 open contact	O	mute
10	T10 open contact	O	bypass
11	T11 open contact	O	Not used
12	T12 open contact	O	Not used
13	T13 open contact	O	Not used
14	T14 open contact	O	Not used
15	root		Common root contact

Electrical specification:

Tally output	type:	normally open relais contacts
	Contact rating:	1A 24 VDC, 0,5 A 125 VAC
		max. 30 W 62,5 VA
		max. 60 VDC, 125 VAC





3.7.3 SERIAL REMOTE CONTROL (RS-422)

The digital dynamics processor b42 can be remote-controlled by means of serial remote RS-422.

use : remote-controlled changeover of presets

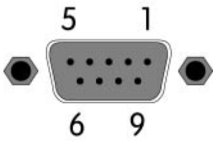
protocol: available on request

connector: D-SUB 9pin, input - female
output - male

Pin assignments

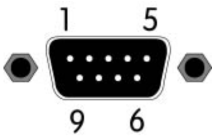
The cable is wired 1:1 completely, the shield of the cable must be connected on both ends!

REMOTE IN



Pin	Signal name	Functions
1	DSR + out	Data set ready
2	DSR - out	
3	SENSE in	Interrogation Remote
4	RXD + out	Receive data
5	RXD - out	
6	DTR + in	Data terminal ready
7	DTR - in	
8	TXD + in	Transmit data
9	TXD - in	

REMOTE OUT



Pin	Signal name	Functions
1	DSR + in	Data set ready
2	DSR - in	
3	GND	GND
4	RXD + in	Receive data
5	RXD - in	
6	DTR + out	Data terminal ready
7	DTR - out	
8	TXD + out	Transmit data
9	TXD - out	

Electrical specification:

signal in-/outputs TTL-level

LOCATION OF PARTS AND CONTROLS

4

4.1. FRONT PANEL

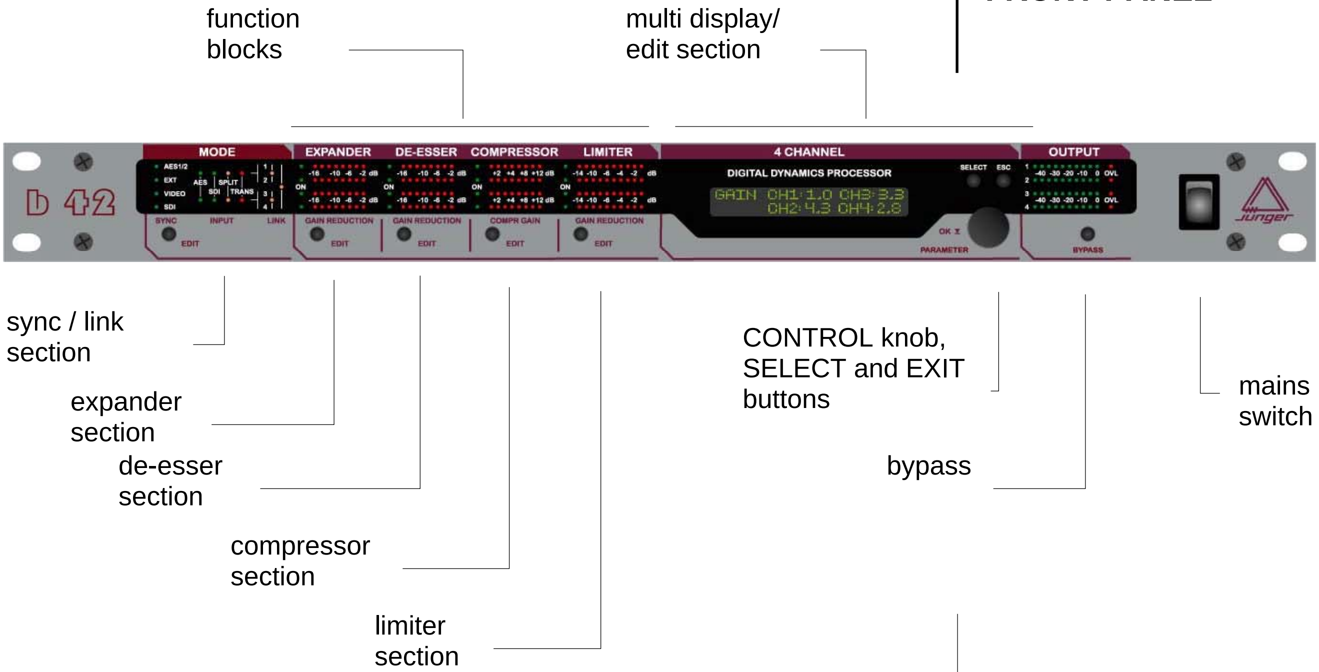


fig1: front panel b42

CONTROL ELEMENTS

MODE	selection/adjustment of sync, input, link and SDI parameter (group selection)
EXPANDER	selection/adjustment of expander parameter
DE-ESSER	selection/adjustment of de-esser parameter
COMPRESSOR	selection/adjustment of compressor parameter
LIMITER	selection/adjustment of limiter parameter
4 CHANNEL	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

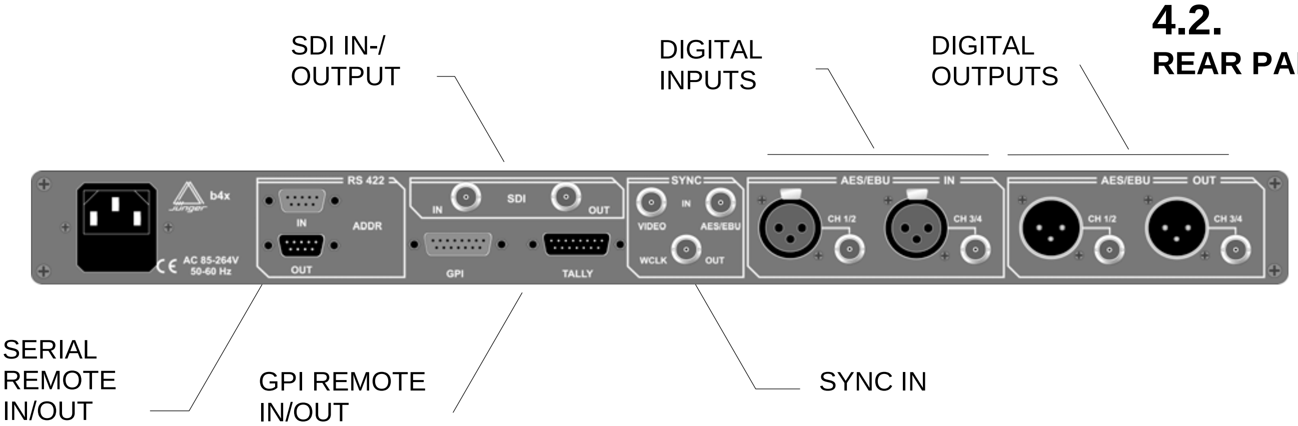


fig. 2: rear panel b42

POWER INPUT
IEC mains input connector 100-240V, 50/60 Hz with integrated fuse

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

GPI
paralle 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

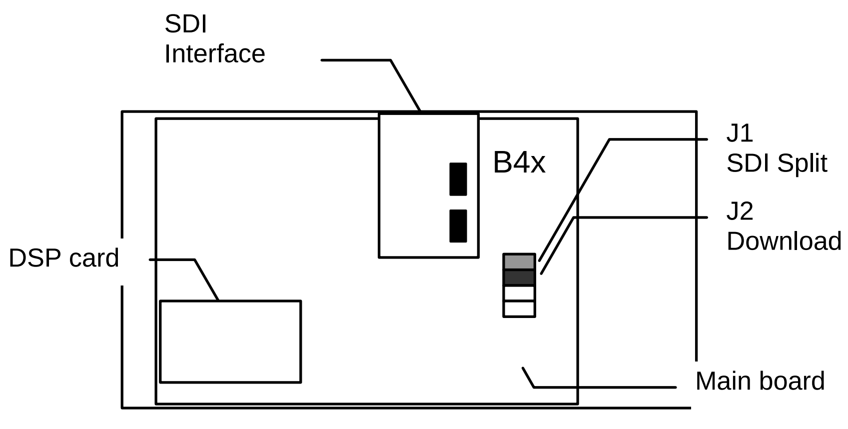
Selection of the device address for serial remote, 16 device addresses selectable
Note: Within a line of remote controlled units every device needs a different address! The selected address is valid after next power-on reset of the unit.

Internal

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

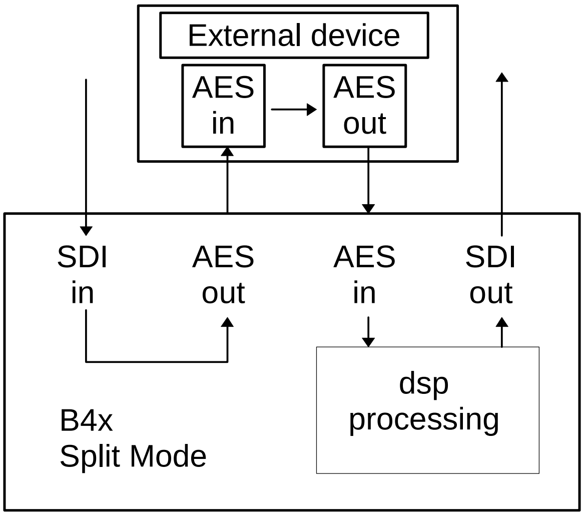
PLEASE DO NOT MAKE ANY ALTERATIONS WITH THE MAINS STILL CONNECTED TO THE UNIT!

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 SELECTION OF SDI SPLIT MODE

Units with SDI interface can be used in SDI split mode:
Audio in path SDI input > AES output
Audio out path SDI output > dsp processing > AES output
(see also 2.5)



The selection of split mode (SDI DIRECT) is made by switch SPLIT on in the input (MODE) edit menu.

4.5 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 b42 is very easy.

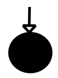
The setup or the programming of the digital dynamics processor b42 is made by adjustment of various parameters and settings.

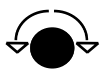
The description is made related to the functions in the menus.

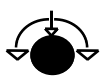
5.1 adjustment of parameters
5.2 level display
5.3 mode menu
5.4 expander menu
5.5 de-esser menu
5.6 compressor menu
5.7 limiter menu
5.8 utility menu
5.9 recall and storage of presets

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

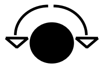

push + turn

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 LEVEL DISPLAY

Gain display shows gain setting for all channels. You can jump to the gain menu from any other edit menu by pushing ESC button.
Adjustments are made by turning&pushing CONTROL knob as described previously (see 5.1).

```
GAIN   1:  0.0     3:  0.0

>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

5.3 MODE MENU

Mode menu shows sync setting, input and link setting of the unit. There are two windows available by pushing MODE 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

SYNC<	LINK		SPLIT
AES	1 + 2	3 + 4	off

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 (only if SDI input or split mode is selected, no sync LED lits!)

LINK MODE: all channels independent or following link combinations:
1+2, 3+4, 1+2 & 3+4, 1+2+3+4

SPLIT AES/EBU + SDI input in split mode (see 4.4)

2. 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 signal is set.

3. 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

5.4 EXPANDER MENU

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

PR	CH	>EXP<	THRS	REL
01	2	ON	-32	SLOW

PR: number of current preset
CH: selected channel (change with SELECT)
EXP: expander on/off
THRS: threshold level -50 ... -20 dBFS
REL: release time slow, mid, fast

5.5 DE-ESSER MENU

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

PR	CH	>DEES<	TYPE	RNG
01	2	ON	MALE	-8.0

PR: number of current preset
CH: selected channel (change with SELECT)
DEES: de-esser on/off
TYPE: de-essing characteristic male / female
RNG: range of s-frequency reduction -20 ... 0 dB

5.6 COMPRESSOR MENU

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

PR	CH	>CMP<	RATIO	RNG
01	2	ON	1.5	6

PR: number of current preset
CH: selected channel (change with SELECT)
CMP: compressor on/off
RATIO: compressor ratio 1.0:1 ...4.0:1
RNG: compression range (maximum compression gain) 0 ... +15 dB

5.7 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	

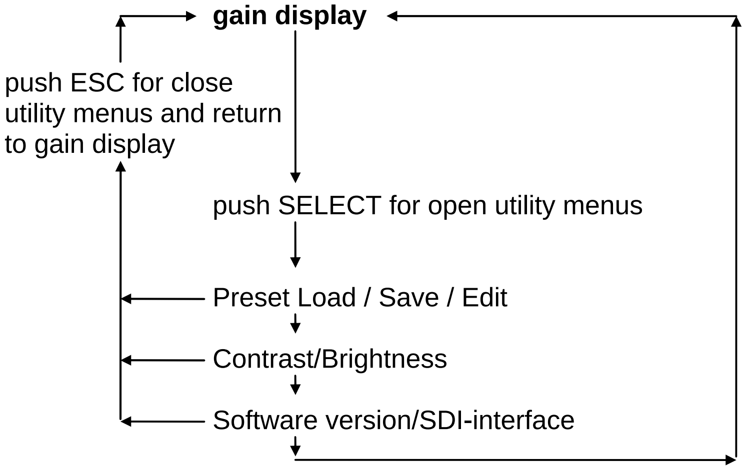
The basic Multi-Loop principle of Jünger Audio dynamics processors operates with adaption of dynamic range control parameters to the incoming audio signal. That means permanently analysis and calculation of attack times, release times , thresholds and interaction parameters of several frequency linear 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 programme material.

5.8 UTILITY MENU

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

push	SELECT	for opening and selection of utitlity menus



Push	ESC	Reset to gain display, basic settings are stored automatically

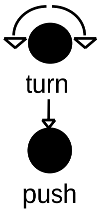
CONTRAST:	display contrast 00-07
BRIGHTNESS:	display brightness 00-07
Software version	C: controller firmware version
	D: dsp firmware version
SDI - interface:	SD-/HD – SDI interface

5.9 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 display is not showing level meter push EXIT button to switch back to level display.

push	SELECT	selection of PRESET menu
turn	CONTROL	selection of LOAD or SAVE menu
push	SELECT	change to PRESET selection menu



CONTROL	selection of preset number for loading or saving preset
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.

In **6. Application notes** some useful PRESETS that are already coming as factory preset for applications with different audio formats are shown.

5.11 PRESET LIST

Preset #1
PROGRAM **UNIVERSAL**
GAIN : **+3.0 dB**
LIMITER: LIM = **ON** , THRSH = **- 9** , PRO= **6**
COMPRESSOR: COPM = **ON** , RATIO = **1.5** , RNG = **10**
DE-ESSER : DEES = **OFF**
EXPANDER: EXP = **ON** , THRSH = **- 40**, REL= **mid**

Preset #2
PROGRAM **POP**
GAIN : **+5.0 dB**
LIMITER: LIM = **ON** , THRSH = **- 9** , PRO= **4**
COMPRESSOR: COPM = **ON** , RATIO = **2.0** , RNG = **12**
DE-ESSER : DEES = **OFF**
EXPANDER: EXP = **ON** , THRSH = **- 40**, REL= **fast**

Preset #3
PROGRAM **SPEECH**
GAIN : **+4.0 dB**
LIMITER: LIM = **ON** , THRSH = **- 9** , PRO = **3**
COMPRESSOR: COPM = **ON** , RATIO = **2.5** , RNG = **10**
DE-ESSER : DEES = **ON** , TYPE = **MALE,** RNG = **16**
EXPANDER: EXP = **ON** , THRSH = **- 30**, REL= **fast**

Preset #4
PROGRAM **LIVE**
GAIN : **+3.0 dB**
LIMITER: LIM = **ON** , THRSH = **- 9** , PRO= **2**
COMPRESSOR: COPM = **ON** , RATIO = **1.8** , RNG = **12**
DE-ESSER : DEES = **OFF**
EXPANDER: EXP = **ON** , THRSH = **- 30**, REL= **mid**

Preset #5
PROGRAM **MOVIE**
GAIN : **+4.0 dB**
LIMITER: LIM = **ON** , THRSH = **- 9** , PRO= **5**
COMPRESSOR: COPM = **ON** , RATIO = **3.0** , RNG = **15**
DE-ESSER : DEES = **OFF**
EXPANDER: EXP = **ON** , THRSH = **- 35**, REL= **mid**

Preset #6
PROGRAM **CLASSIC**
GAIN : **+3.0 dB**
LIMITER: LIM = **ON** , THRSH = **- 9** , PRO= **8**
COMPRESSOR: COPM = **ON** , RATIO = **1.3** , RNG = **10**
DE-ESSER : DEES = **OFF**
EXPANDER: EXP = **ON** , THRSH =**- 50**, REL=**slow**

BOOT DISPLAY AND TROUBLE SHOOTING

6

6.1 BOOT DISPLAY

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

7.1
B4x SERIES WITH SDI-INTERFACE optional SD/HD

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.

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-AB. 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 b40 provides an optional SD- **or** HD-SDI board. When you switch on the device the plugged in interface will be indicated in the display

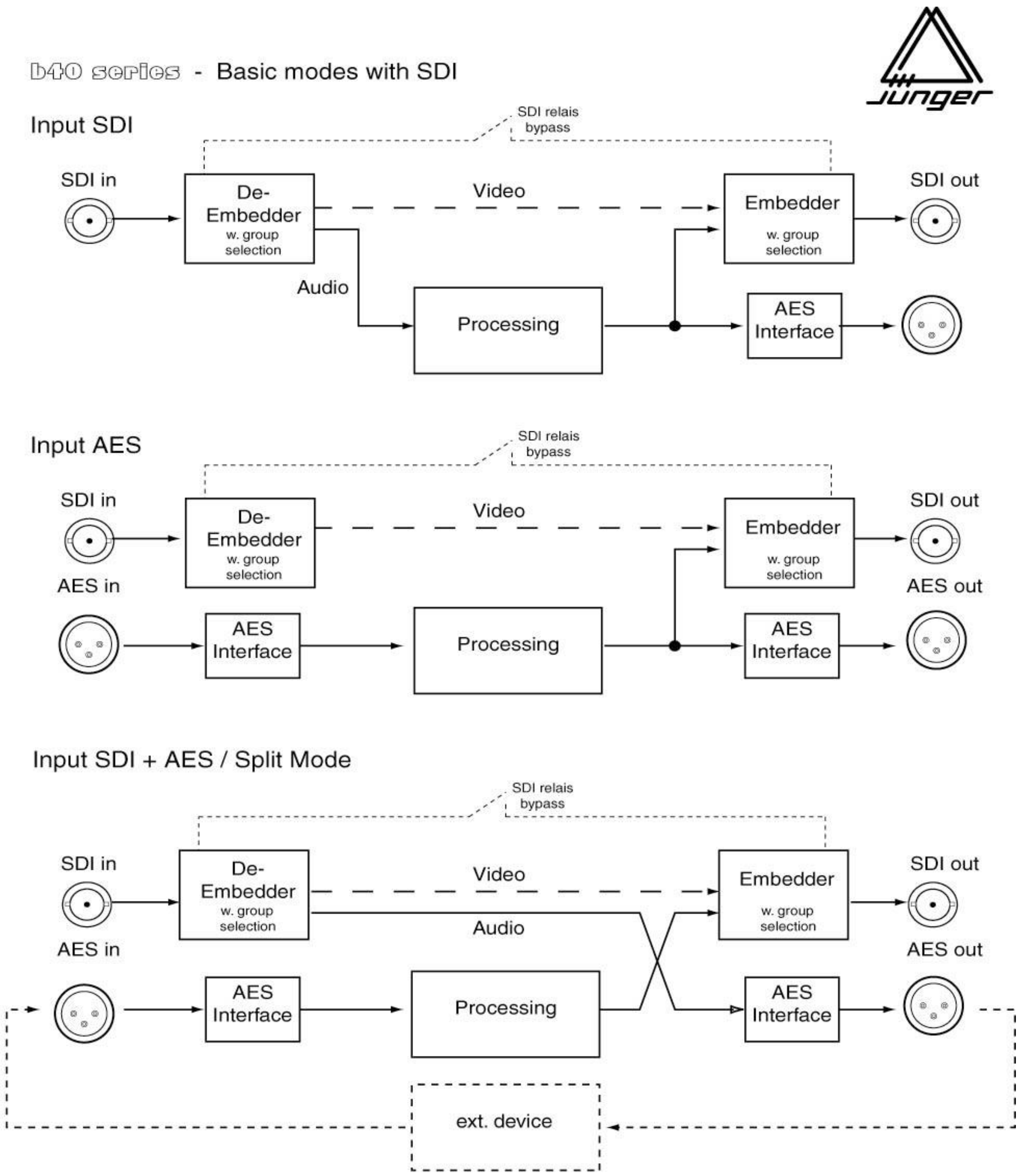
FEATURES

- 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- / outputs

DIGITAL IN/OUT

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)

connector :	BNC, 75 Ohm, coaxial
data rate :	270 Mb/s, 525/625 Line rate
format :	serial digital component video 4:2:2 with embedded audio (ITU-R BT.601, SMPTE 272M-A)
level:	800 mV +/- 10%
equalisation :	appr. 180 m (Belden 8281)
audio data:	4 channels, 20 bit
features :	SDI relais bypass independent group selection for de-embedder and embedder

SDI ***(only for SDI version)***

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...1 Vpp
input format :	Blackburst or PAL/NTSC composite video

---

remote control

REMOTE

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
optional :	programmable remote control brc

WARRANTY AND SERVICE INFORMATION

9

JÜNGER AUDIO grants a two-year warranty on the

4-channel digital dynamics processor b42

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

b42

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