OPERATIONS MANUAL

voice processors

v01
v02
v03
v05

digital voice processor

v02

release 3.0

jünger audio

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INTRODUCTION

0

The Voice Processor v02 is a high quality digital voice processor. It should serve as a convenient microphone preamplifier combined with sophisticated digital signal processing for improvement of voice recordings. It has inputs for microphone and furthermore digital signal sources and has analogue and digital outputs.

1

CONTENTS

BASIC DESCRIPTION

2

2.1 BASIC DESCRIPTION

The Voice Processor V02 is a high quality digital voice processor. It should serve as a convenient microphone preamplifier combined with sophisticated digital signal processing for improvement of voice recordings. It has inputs for microphone and furthermore digital signal sources and it has analogue and digital outputs.

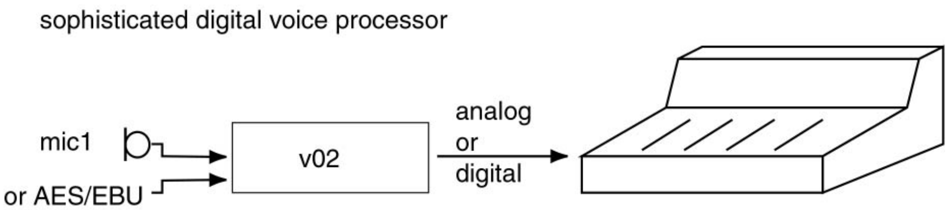
The digital voice processor guarantees the desired sound settings of voice recordings over the microphone input. Processing of recorded voices can be made using the additional digital signal input or the high level analogue input.

The device operates very simply, personal presets can be stored and loaded with the aid of rapid memory key. The unit is very easy to use and requires only a minimum selection of parameters in order to provide effective processing. Other parameters are controlled and optimised adaptively by the program signal, minimising the set-up time.

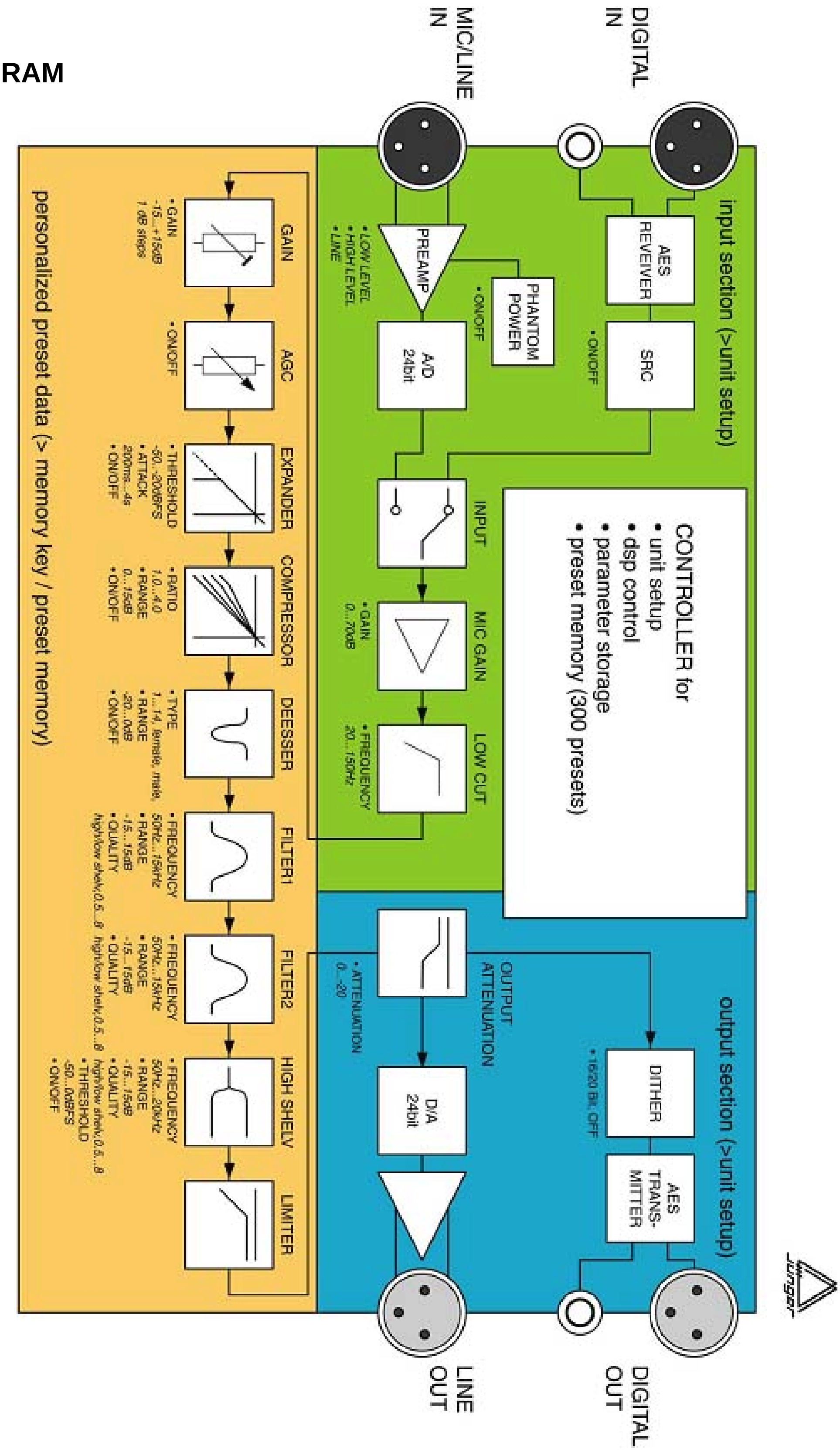
Digital Voice Processor V02 is designed for use in studio and production area. It is an ideal completion to Voice and Monitor Processor V01 regarding funtionality, operation and storage of individual parameters.

* microphone input with 24 bit A/D converter
* digital voice processor (Gain, AGC, expander, compressor, de-esser, filter, limiter)
* free running digital input with sample rate conversion
* digital limiter prevents clipping and overlevel
* digital and balanced analog output
* storage of personal presets with the aid of rapid memory key (also usable for V01)

2.2 APPLICATION EXAMPLE



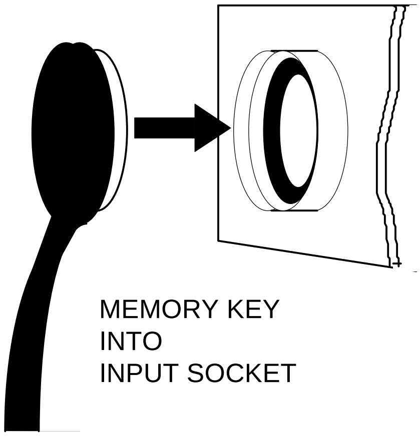
2.3
BLOCK DIAGRAM



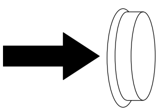
2.4 USING THE MEMORY KEY

The memory key is a small touch memory. This is a semiconductor memory chip housed in a stainless steel container. This steel container is mounted in a plastic key fob.

When a memory chip becomes a label, information is available immediately on the spot. As the memory key is moved from point to point, information is transferred free from restrictions of a wired network. Even with various keys prepared with specific setup datas for different people, relevant information or setup about a particular user is selected by one touch.



Read or write data is transfered by memory key to the VAMP2 memory socket on the front panel of the unit. Push the key into the socket and move it to the edge until a beep sounds. The data is then loaded from or to memory chip.



MEMORY KEY
INTO
INPUT SOCKET

There are two different kind of memory keys available:

user key	normal memory key (RAM) for storage of individual preset data
administrator key	special memory key (ROM) for opening edit functions while unit is in edit lock mode

Up to 100 contents of such keys (100 times 3 presets) can be stored into the data memory of the unit. See also chapter 7.

2.5 AUDIO SIGNAL PROCESSING

All signal processing is in the digital domain by Texas Instruments floating point signal processors. The use of 32 bit word length 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.

2.5.1 GAIN / AUTOMATIC GAIN CONTROL / MIC GAIN

GAIN means linear amplification of input signal. The selection of gain levels takes place in steps of 1 dB and has a range from 0...20 dB.

Automatic gain control AGC has a range of up to 40dB. Input signal is analyzed by measurement of RMS power to control AGC range. AGC is adjusted for an average output level of -12dBFS. AGC can be switched off.

Mic gain is an additional amplification for the mic input only. Gain of analogue mic preamplifier and digital mic gain together are the total gain for mic input. If digital input or if analogue line input is selected mic gain is switched off automatically.

2.5.2 DYNAMICS

principle of Jünger Audio dynamic range processing

The dynamic range processing requires only a limited number of manual settings to be made by the user to achieve optimum results. All other parameters necessary for inaudible processing of the dynamic range are continuously automatically controlled in response to changes in the audio signal.

The increases in signal density and loudness are entirely free from the processing noises typical of dynamic range processors such as "pumping" and "breathing", or signal discolouration.

The Jünger Audio dynamics processors work according to a „Multi-Loop Principle“, operating with an interaction between several frequency linear control circuits. The resulting attack and release times of this system are variable and are adapted to the envelope 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.

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.

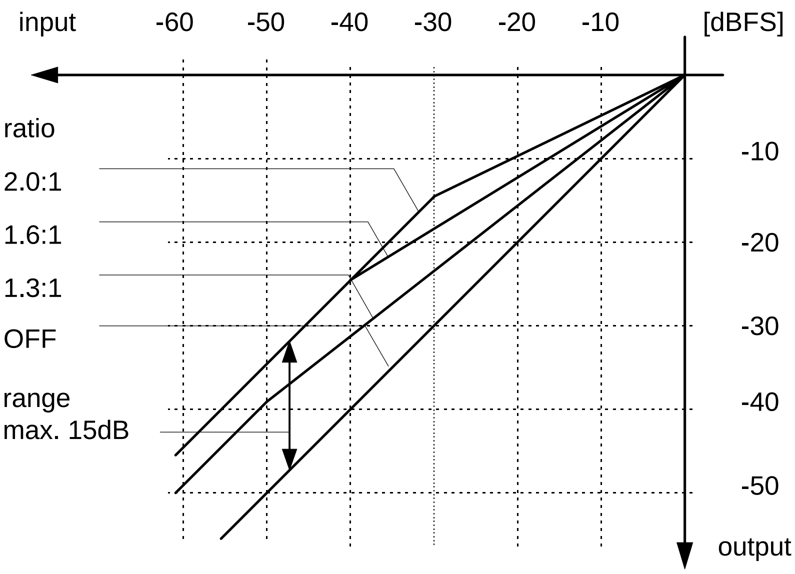


fig. 4: static characteristics: compressor

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 , a **maximum gain of the compressor** (RANGE) can be set, so that there is no inadmissible increase of background noises during signal pauses (e.g. live atmos, air-conditioning, hum and noise).

expander

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

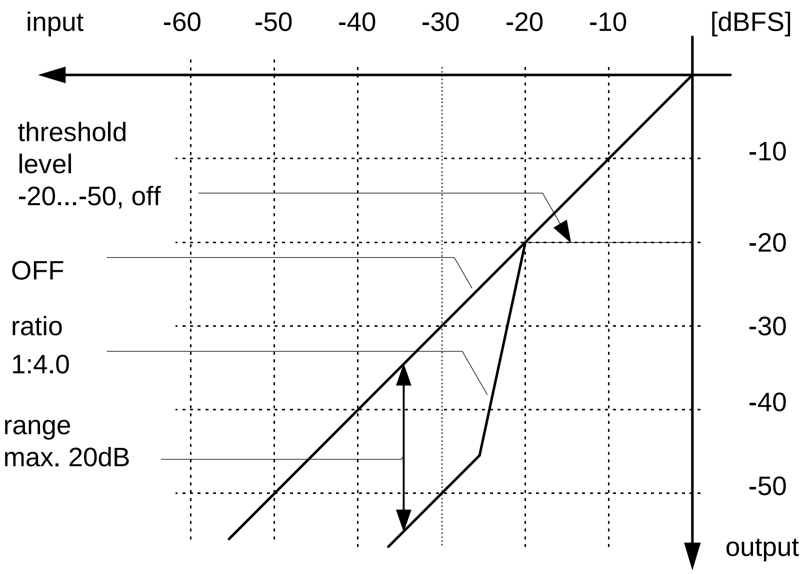


fig. 5: static characteristics: expander

The RELEASE TIME influences the dynamic characteristic of the expander and is to be chosen in accordance with the program.

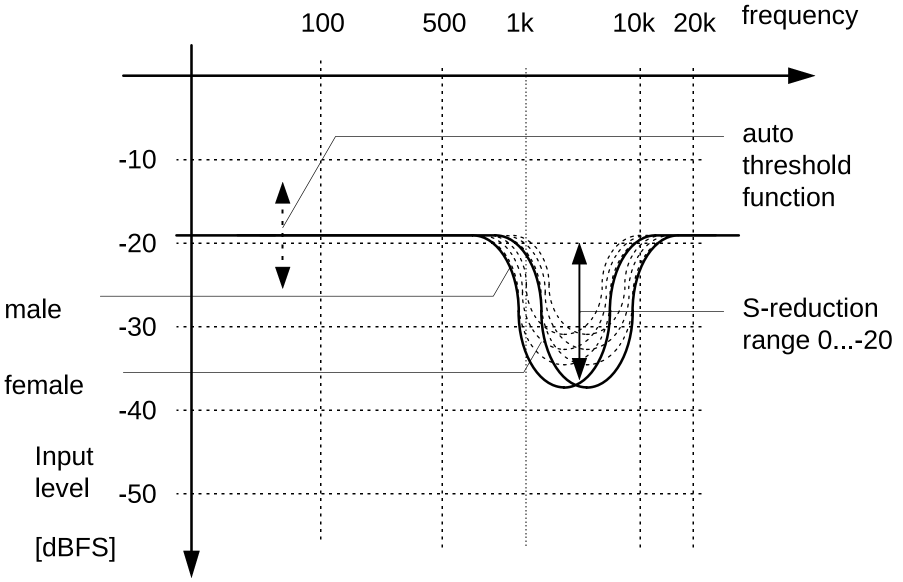
de-esser

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 VAMP2 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 it works independent of the signal level.

Typical critical S-frequencies are different for female and male voices. Therefore on can adjust either frequency value or female/male selection. Only two basic adjustments are necessary for the de-esser – frequency selection and the amount of 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. 6:
basic function:
de-esser



2.5.3
FILTERS

The digital voice processor offers various filters for the creation of the desired sound setting for voice recordings.

low cut filter

The low cut filter is a high pass filter third order (-18dB/octave). Centre frequency (40…150 Hz) is adjustable.

low and mid peak filter

Both peak filters are designed as parametric equalizer. Centre frequency (50...500Hz), gain (-15...+15dB) and Q are adjustable. The frequency control is used to select the centre frequency around which the gain control will boost or attenuate sounds. The Q control affects how frequency selective the gain control is when boosting or attenuating. On a high Q the gain control has influence for a small range of frequencies. With low Q the gain control will boost or cut a whole range of frequencies.

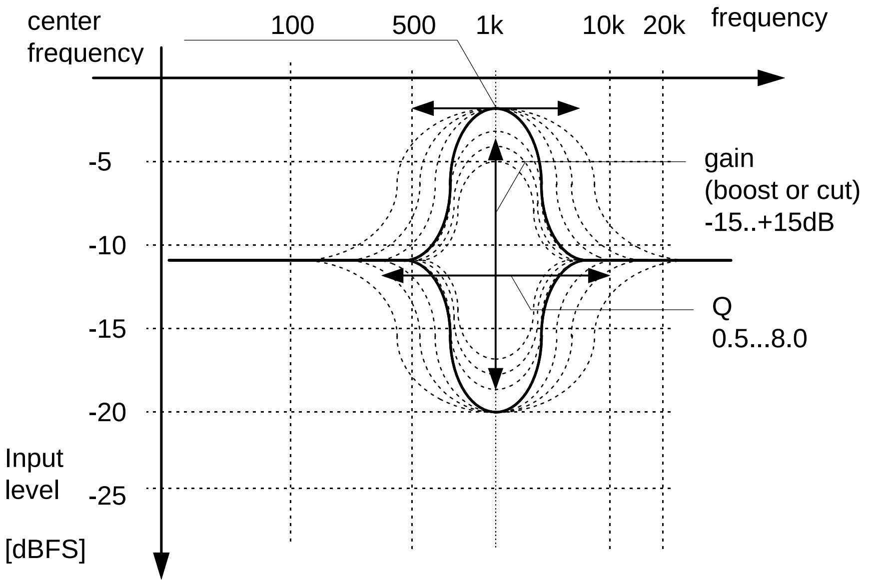


**fig. 7:
basic function:
low and mid
peak filter**

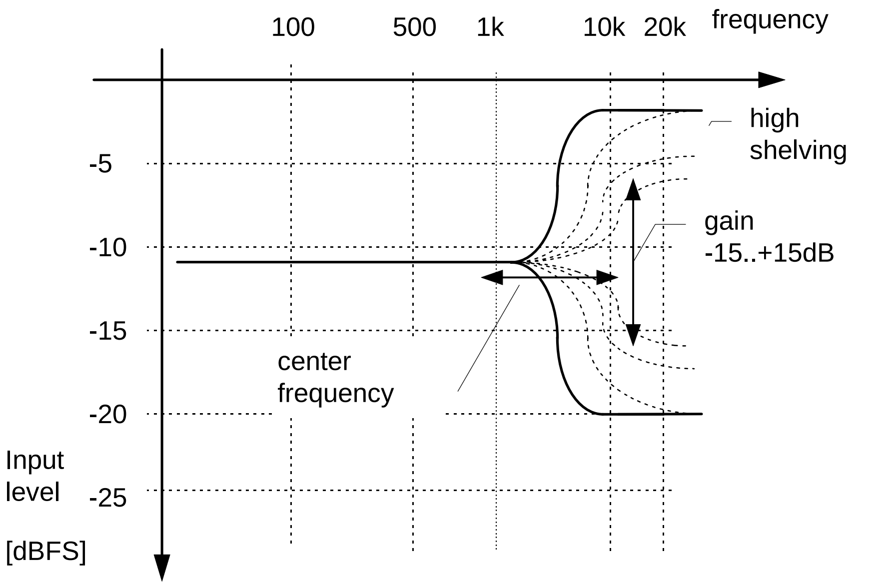


**fig. 8:
basic function:
hi-shelf filter**

high shelfing filter

The high shelfing filter is a low pass filter with shelfing response. Centre frequency (1.000...15.000Hz) and gain (-15...+15dB) are adjustable.

high cut filter

The high cut filter is a low pass filter third order (-18dB/octave). Centre frequency (1.000...15.000Hz) is adjustable.

2.6 VOICE PROCESSOR INPUT

The voice processor inputs can be selected from following inputs:

MIC in	analogue input (low level input, mic gain active)
LINE in *(mic in)*	analogue input (high level input, no mic gain)

Left channel is active input signal!

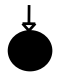
DIGITAL input (high level input, no mic gain)

AES/EBU-CH1 (left) is active input signal!

Output signal is present as mono signal at both output-channels anytime!
(see also 5.3!)

2.7 PHANTOM POWER

Microphone input can be used with phantom power supply of 48 Volts.
Phantom power supply can be switched on and off (in EDIT mode only).
If LINE or DIGITAL input is selected phantom power is switched off automatically.



push
3sec

PRESET 1 for display phantom power status

display: PHANTOM ON or OFF

2.8 OUTPUT DITHER

The digital output signal has a word length of 24 bit. The information of a 24 bits signal is not more storable linear in most cases. One must shorten 24 bits data word to 20 or 16 bit word length.

In order to receive a better quality during cut down the data to 20 or 16 bit one must redithering the material. This is done by calculating random numbers (dither signal) and add a different random number to every sample. Then it will be cut off to 16 bit. As a result, the bit with least weight (LSB) is put in such a way that it corresponds best to the information of the last bits following available ones no more and makes less distortions as hissing in the signal.

2.9 OUTPUT ATTENUATION

All adjustments of audio processing are referred to 0 dBFS. The digital limiter has a fixed threshold level of 0 dBFS. In order to be able to change the nominal digital output level of the unit one can adjust output attenuation. Output attenuation is done after limiter processing and determines maximum digital output level.
Example: If you want to distribute your processed material meeting the EBU recommendations (recommended distribution level -9 dBFS) you have to adjust output attenuation to -9dB. The digital output level will never come over -9dBFS.

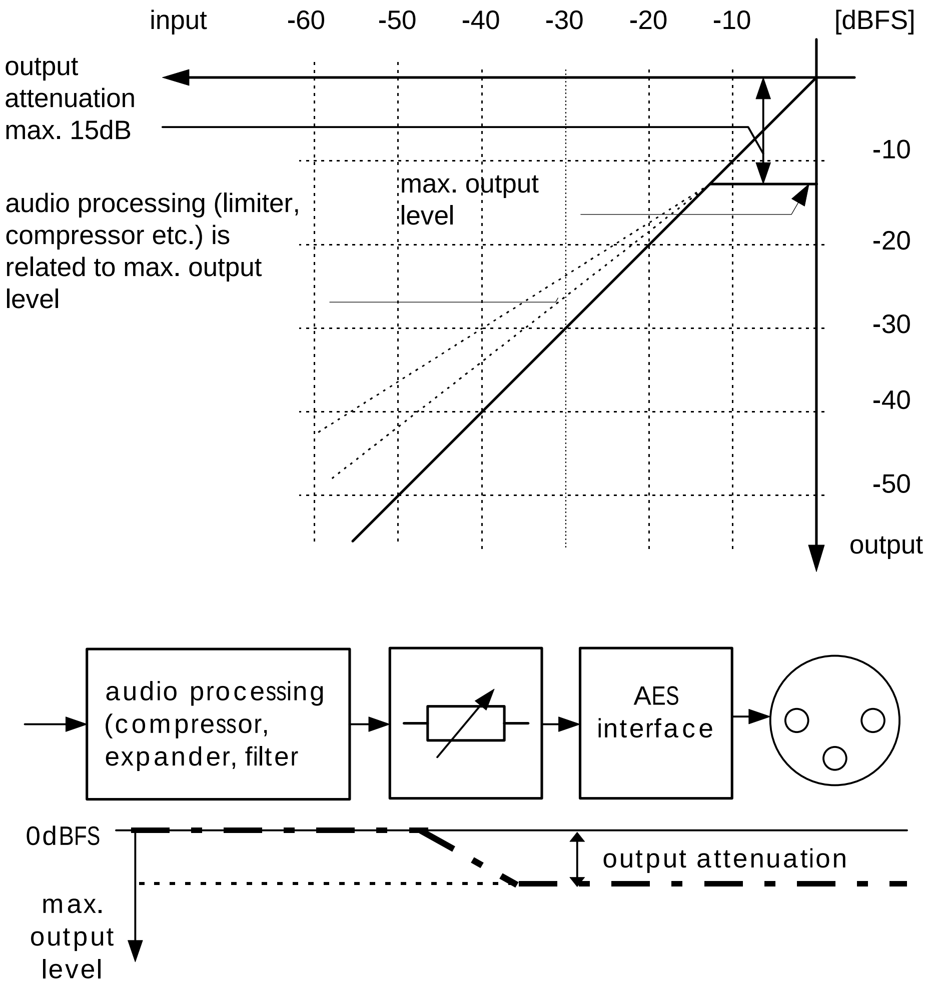


fig. 9: output attenuation

INSTALLATION

3

3.1 UNPACK THE UNIT

The digital voice processor V02 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 voice processor V02 is manufactured under the safety category *Schutzklasse 1* in keeping with the VDE 0804 standards.
Check the voltage details printed at the rear panel are the same as your local mains electricity supply.

3.3 CONNECTIONS

The unit is equipped with standard connectors (see chapter 4). Before connecting the voice processor V02 switch the power off at all connected units.

3.4 RACK MOUNTING

The V02 is made as standard 19“ unit (EIA format). It occupies 1 RU (44 mm height) space in a rack.
Please allow at least addititonal 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.

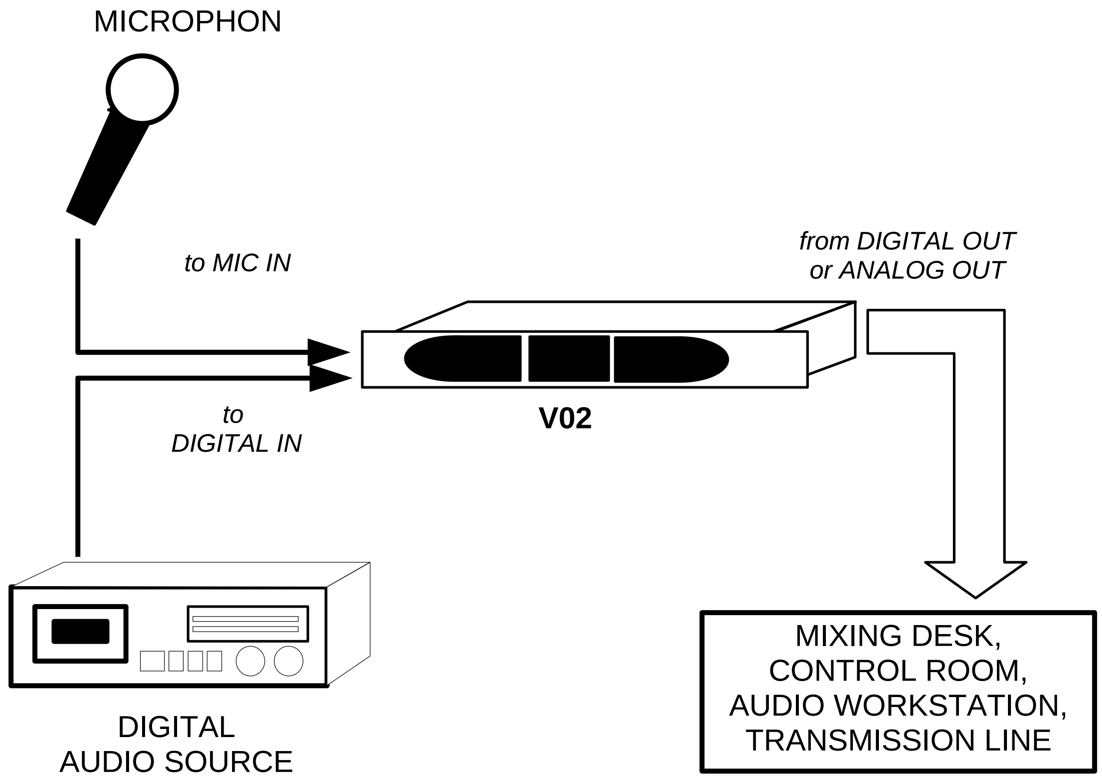
3.5 OPERATION SAFETY

The voice processor 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 BASIC INSTALLATION

Drawing below shows basic installation of Voice Processor V02. More detailed informations about possible connections you will find in chapter 4 - Location of parts and controls.

Fig. 10 : basic installation scheme for V02



3.7 SYNCHRONIZATION OF DIGITAL OUTPUT

The voice processor V02 has a digital output. Digital processing can be locked either by internal or external reference. Selection is made automatically. Sync input has priority over internal reference. Following sample rates at DIGITAL OUT are available:

INT 48 kHz — not synchronized (clocked by internal reference), only active if external sync is not connected

EXT SYNC — synchronized by signal at external sync input (wordsync or AES/EBU, 32...50 kHz)

3.8 SERIAL AND PARALLEL REMOTE CONTROL

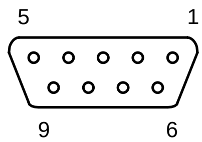
Digital Voice Processor V02 can be remote-controlled by serial remote control (RS-232, RS-422 or CAN bus) and by GPI contact.

Use:	RS-232	Control from personal computer
	RS-422	Loading setup data from pc
	CAN	Loading preset data from pc
		serial remote control
	GPI	mic off (cough button), preset selection

parallel GPI/Tally

Connector: D-SUB 9pin

Pin assignment of the connector



Pin	Signal name	Functions
1	GPI 1: MIC OFF	muting microphone input
2	GPI 2:	not used
3	GPI 3: PRESET 1	calling preset
4	GPI 4: PRESET 2	calling preset
5	GPI 5: PRESET 3	calling preset
6	TALLY1:	not used
7	TALLY2:	not used
8	TALLY3:	not used
9	GND	ground

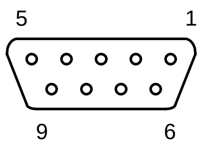
Electrical specifications:

GPI command input	ON:	connection to ground
	OFF:	open

SERIAL 1&2

Connector: D-SUB 9pin

Pin assignment of the connector



Serial remote interfaces are to configure by using jumpers on main board (see chapter5).

Following combinations of serial remote standard selection are possible:

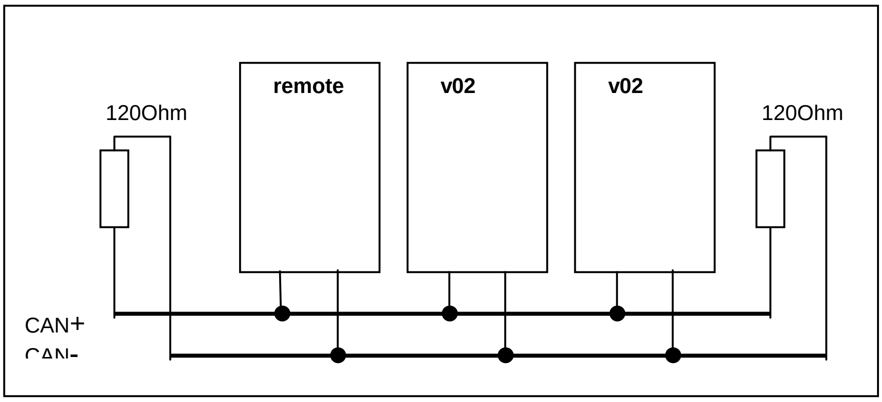
remote 2 \ remote 1	CAN	RS232	RS422	RS485
CAN	X	X	X	X
RS232	X			
RS422	X			
RS485	X			X

Depending on selected remote interface standard following pin out is used at 9pin SubD-connectors:

9pin SubD pin	CAN	RS232	RS422	RS485
1			RX-	DX+
2	CAN-L	TXD	RX+	DX+
3		RXD	TX+	
4			TX-	
5	GND	GND	GND	GND
6	GND	GND	GND	GND
7	CAN-H			
8				
9				

CAN-Bus Termination

CAN-bus (Controller Area Network) has to be terminated at both ends of the bus chain. Therefore termination switch on C8800 module can be used.



Electrical specifications:

inputs / outputs TTL-level

Interface Protocol

optional, on request

3.9 SELECTION OF DEVICE ADDRESS FOR SERIAL REMOTE CONTROL

The digital Voice processor V02 has serial remote control interfaces in the RS-232, RS-422 and in the CAN-Format.

The device address necessary for remote control in the RS-422 or CAN Format is adjusted in the Edit menu in the point DEVICE ADDR. The selection limits to 32 addresses (0...32). The adjusted address is immediately valid.

Up to 32 devices V02 can be remote-controlled with remote control V02 Remote.

LOCATION OF PARTS AND CONTROLS

4

4.1. FRONT PANEL

All control elements are giving direct access.
In menu modes the alphanumeric display above the related button or rotary knob shows the specific function.

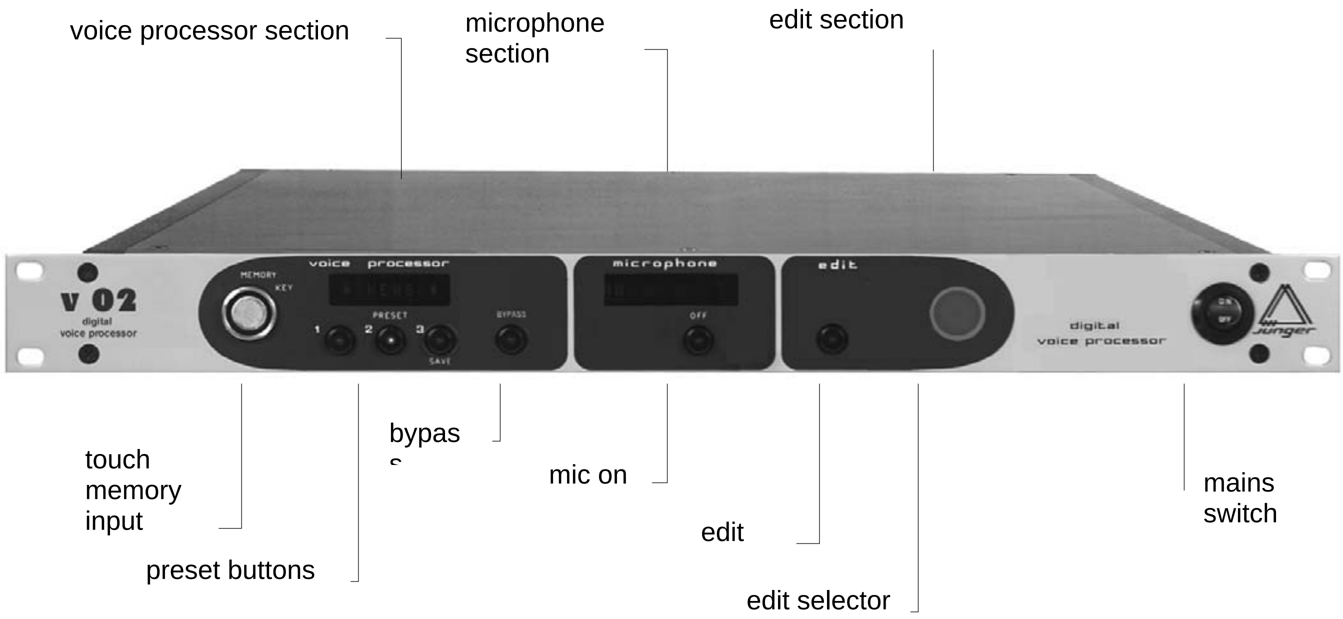


fig. 11: front panel VAMP2

VOICE PROCESSOR SECTION

MEMORY KEY	input for touch memory key for loading of personal presets
PRESET 1...3	selection of preset 1...3
DISPLAY	is showing preset name

MIC SECTION	MICRO ON	to switch off and on voice processor mic input (cough button)
	DISPLAY	to show either the level of microphone input or parameter for editing
EDIT SECTION	EDIT	to switch EDIT mode on and off
	ADJUST	selection and adjustment of edit parameters

4.2. REAR PANEL

fig. 12: rear panel V02

POWER INPUT

IEC mains input connector 230 V, 50 Hz (UK: 240 V, 50 Hz; JAPAN: 100 V, 60 Hz; USA: 127 V, 60 Hz) with integrated fuse

REMOTE

serial remote interface 1
connector: 9pin SUB-D, male
serial remote interface 2
connector: 9pin SUB-D, female
GPI/TALLY TTL level
connector: 9pin SUB-D, female

SYNC

input for sync signal (AES or wordsync)
input: BNC socket; XLR female panel jack

DIGITAL IN

input for AES/EBU standard format
input: XLR female panel jack
1- ground, 2-3 signal, balanced
input for S/PDIF standard format
input: RCA socket

DIGITAL OUT

output for AES/EBU standard format
output: XLR male panel jack
1- ground, 2-3 signal, balanced, max. 5 Vpp
output for S/PDIF standard format
RCA socket

ANALOG OUT

voice processor output, analogue
output: XLR male jacket
1- ground, 2- hot, 3- cold, floating balanced

MIC IN

microphone input, analogue
input: XLR female jacket
1- ground, 2- hot, 3- cold, floating balanced

JUMPER AND SWITCHES

5

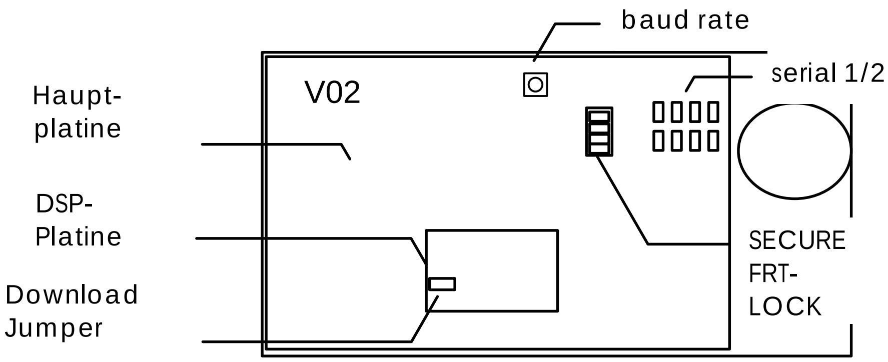
5.0 LOCATION OF JUMPER AND SWITCHES

Basic working modes can be set by using jumper or switches. These settings can occur general changes for operation and should made by qualified engineering staff only.

To set any 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.



5.1 LOCKING LOCAL CONTROL OPERATIONS

Local control of preset selection and mic button can be locked on two ways. This is to avoid doubled controlling while remote operation (guaranteed exclusive remote control).
The edit mode is not influenced by this lock function. To lock edit mode SECURE jumper is to activate.

Setting the FRT-LOCK jumper on main board is deactivating local operation permanently.

If the Digital Voice Processor V02 is combined with remote control V02 Remote, the local control of the device can be locked by setting in edit menu. As soon as the device is combined with a remote control, one finds the menu point FRONT. as a first menu choice. It offers two possibilities - UNLOCKED and LOCKED. If the local control should be stopped, LOCKED must be chosen.
The adjustment is only effective while the device is combined with remote control! It also survives during use of the device without remote control.

5.2 LOCKING EDIT MODE

The EDIT mode can be locked to prevent unallowed change of setup parameter. Set the jumper SECURE on the mainboard to lock edit mode.

Following functions are disabled in EDIT disable position:
open EDIT mode by pressing buttons

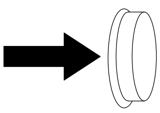
- editing of any setup parameters
- saving of parameters into memory keys
- editing of input names

Following functions are enabled in EDIT disable position:

- changing voice processor input by pressing buttons
- transfer of parameter sets via RS-232
- loading of presets from memory keys

5.3 STARTING EDIT MODE BY USING ADMINISTRATOR KEY

ADMINISTRATOR KEY is a memory key with special coding. This key allows opening of the EDIT mode if unit is in EDIT LOCK status. ADMINISTRATOR KEY should be used by qualified engineering staff only.



MEMORY KEY
INTO
INPUT SOCKET

Contact ADMINISTRATOR memory key with input socket until beep sounds.
Unit is open in the EDIT mode automatically.

5.4 CONFIGURATION OF SERIAL REMOTE INTERFACE

The serial remote interfaces can be configured with the help of jumpers on mainboard (see also chapter 3).

Following combinations of serial remote standard selection are possible:

remote 2 \ remote 1	CAN	RS232	RS422	RS485
CAN	X	X	X	X
RS232	X			
RS422	X			
RS485	X			X

The speed of the data transfer depending on the used standard can be adjusted using the BAUD RATE switch on main board.

5.5 CHANGING THE SENSITIVITY OF MIC INPUT

The input sensitivity of mic input is switchable. The input sensitivity is to change by selection of VOICE PROCESSOR INPUT parameter. Two selections are possible - MIC or LINE.

In position MIC one can connect analogue signals with level not higher as +6dBu.

In position LINE input level until +15dBu can be processed without clipping. The phantom power and mic gain are switched off! After adjustment of low gain value one can place the VAMP unit into insert path of analogue mixing desks.

SETUP

6

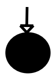
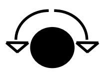
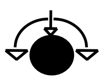
6.0 DESCRIPTION OF SETUP OPERATIONS

The setup of the Voice and Monitor Processor V02 is made by editing of various parameters.

The description is made related to the functions in the EDIT mode.

6.1 starting and closing EDIT mode
6.2 starting EDIT mode by using administrator memory key
6.3 editing of parameters
6.4 saving edited parameters
6.5 automated calibration

Following syntax is used:

SYMBOL	NAME	ACTIVITY
describes how to use button or rotary knob  push  turn  push + turn	describes the name of button or rotary knob NAME name, as printed on front panel NAME name, as displayed on alphanumeric display above button or knob	describes action or function of button or rotary knob

DISPLAY:

describes the status or information shown on the display

NAME	status/name
<NAME>	sort of status/name on display

6.1 STARTING AND CLOSING EDIT MODE

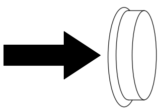
Open EDIT mode to adjust internal voice processor parameters and for closing edit mode.

push

push	EDIT	for opening and closing EDIT mode
	display:	NAME PRESET* [* = number of last selected preset] EDIT LED is flashing

6.2 STARTING EDIT MODE BY USING ADMINISTRATOR KEY

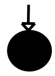
ADMINISTRATOR KEY is a memory key with special coding. This key allows opening of the EDIT mode if unit is in EDIT LOCK status. ADMINISTRATOR KEY should be used by qualified engineering staff only.



MEMORY KEY INTO INPUT SOCKET	Contact ADMINISTRATOR memory key with input socket until beep sounds. Unit is open in the EDIT mode automatically.

6.3 EDITING OF PARAMETERS

For editing of voice processor parameters. All changes are made on-line, this means all parameter changes are audible in real time.


push

PRESET 1...3 selection of preset for parameter editing


turn

ADJUST for selection of parameter for editing

front .... *(only if connected to V02 Remote))*
↓
presetname
↓
gain
↓
expander
↓
compressor
↓
de-esser
↓
low-peak filter
↓
mid-peak filter
↓
hi-shelf filter
↓
hi-cut filter
↓
low-cut filter
↓
voice processor input

*digital selection *)* → SRC → output attenuation

*line selection *):* → output attenuation

mic selection → mic cal, mic gain, pre amp → phantom power → agc → output attenuation

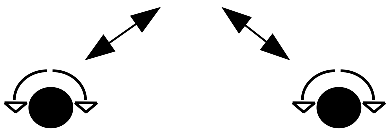
output attenuation
↓
output dither
↓
device

*) *mic gain off*
phantom power off

When the parameter is selected by turning of ADJUST knob push ADJUST knob for enter parameter edit mode. Now one can change the parameter value by turning ADJUST knob again. Pushing the ADJUST knob again switches the unit back to parameter selection mode.

The following table shows parameter value setup range.

Push ADJUST for switching between
parameter selection mode and parameter edit mode



turn   turn

turn ADJUST for parameter selection

turn ADJUST for parameter adjustment

tab.1:
editing parameter V02

PARAMETER	VALUE	STEPS	CLASS	MEMORY
FRONT	LOCKED/ UNLOCKED		SETUP w. Remote Control	
PRESET**NAME**	NAME	8 character	PRESET *)	MEM KEY
GAIN	-15...+15dB	1dB	PRESET *)	MEM KEY
EXPANDER, **THRS**	off, -50...-20dB	2dB	PRESET *)	MEM KEY
EXPANDER, **REL**	200ms...4s	0,01s/1s	PRESET *)	MEM KEY
COMPR. **RAT**IO	off, 1.3, 1.6, 2, 3, 4		PRESET *)	MEM KEY
COMPR. **RANG**E	0...15dB	1dB	PRESET *)	MEM KEY
DE-ESSER	male/female, 1…14	1 kHz	PRESET *)	MEM KEY
DE-**ES**SER, **RNG**	-20...0dB	1dB	PRESET *)	MEM KEY
LOW-PEAK, FREQ	50...500Hz	2Hz	PRESET	MEM KEY
LOW-PEAK, GAIN	-15...+15dB	1dB	PRESET	MEM KEY
LOW-PEAK, Q	0.5...8	0.5	PRESET	MEM KEY
MID-PEAK, FREQ	50...500Hz	5Hz	PRESET	MEM KEY
	500Hz...5.0kHz	50Hz	PRESET	MEM KEY
	5.0kHz...15.0kHz	500Hz	PRESET	MEM KEY
MID-PEAK, GAIN	-15...+15dB	1dB	PRESET	MEM KEY
MID-PEAK, Q	0.5...8.0	0.5	PRESET	MEM KEY
HI-SHELF, FREQ	1.0...15.0kHz	100Hz	PRESET	MEM KEY
HI-SHELF, GAIN	-15...+15dB	1dB	PRESET	MEM KEY
HI-CUT, FREQ	off, 1.0...15.0kHz	1kHz	PRESET	MEM KEY
LOW-CUT, FREQ	40 (60…150)Hz	20 Hz	SETUP	UNIT
VP - INPUT	MIC/DIGITAL		SETUP	UNIT
MIC CAL	WAIT/ACTIVE/OFF		SETUP **)	UNIT
MIC GAIN	0...70dB	1dB	SETUP **)	UNIT
PRE AMP	LOW LVL/ HIGH LVL		SETUP **)	UNIT
PHANTOM POWER	ON/OFF		SETUP **)	UNIT
AGC	ON/OFF		SETUP **)	UNIT
SRC	ON/OFF		SETUP **)	UNIT
OUTPUT ATT	-15...0dB	0.1dB	SETUP **)	UNIT
OUTPUT DITHER	16/20/24 Bit		SETUP **)	UNIT
DEVICE	ADDR	01..32	SETUP **)	UNIT

Notes: *) display of activity in right window
**) parameter not storable in presets. Parameters are stored in the setup memory and are valid for all presets.

To compare adjustments with unprocessed signal push either

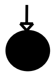
push — EDIT button — for bypassing selected parameter while parameter editing

or

push — BYPASS — for bypassing all parameters while parameter editing

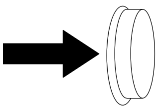
6.4 SAVING EDITED PARAMETERS

All edited parameters will be saved into preset memory automatically.
To get more flexibility regarding preset storage and preset administration it is possible to use memory keys (touch memory keys) for parameter storage. These memory keys store all preset parameters for one set of three presets. In this way it is very easy to guarantee optimum setup for every person using the V02 without any editing or adjustment.
Parameter transfer to memory key can be made in EDIT mode only (see 6.1)



push
3sec

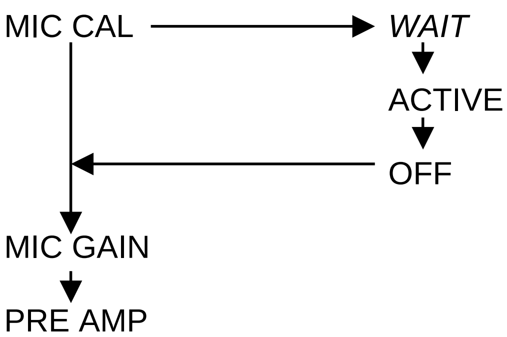
push 3sec	PRESET 3 (SAVE) *display:* SAVE MEM	for activating preset save function LED in button blinks



	MEMORY KEY INTO INPUT SOCKET	contact memory key with input socket until beep sounds, all parameters of 3 actual presets are saved into memory key. Unit returns to normal working mode automatically.

6.5 AUTOMATED CALIBRATION

With the menu selection MIC CAL the unit is doing an auto calibration procedure for setting of PRE AMP and adjustment of MIC GAIN. If MIC CAL is activated you should input the highest expected audio level (use the speech of the loudest announcer). After a few seconds the unit has detected optimum settings. With the following menu items MIC GAIN and PRE AMP you can check and readjust the settings.
If you are pushing MIC ON button while adjusting MIC GAIN you will see level meter for orientation and optimum adjustment.
(see chapter 8.5 also)



OPERATION

7

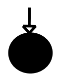
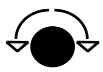
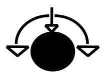
7.0
DESCRIPTION OF USER OPERATION

The operation of the Voice and Monitor Processor V02 is very easy and needs only a few settings to be made by the user.

The operation descriptions are related to the functions in OPERATION mode.

7.1 loading personal voice processor preset data
7.2 selection of presets
7.3 saving of presets into internal memory
7.4 loading of presets from internal memory
7.5 muting microphone input (cough button)
7.6 changing voice processor input

Following syntax is used:

SYMBOL	NAME	ACTIVITY
describes how to use button or rotary knob  push  turn  push + turn	describes the name of button or rotary knob NAME name, as printed on front panel NAME name, as displayed on alphanumeric display above button or knob	describes action or function by use of button or rotary knob

DISPLAY:

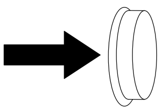
describes the status or information shown on the display

NAME	status/name like displayed
<NAME>	sort of status/name on display

7.1 LOADING PERSONAL VOICE PROCESSOR PRESET DATA BY USING MEMORY KEY

All parameters necessary for personal voice processor settings are stored into presets. Three presets for different parameters (microphones, studios or speakers) are available.

To get more flexibility regarding preset storage and preset administration it is possible to use personal memory keys (touch memory keys) for parameter storage. These memory keys are store all preset parameters for one set of three presets. In this way it is very easy to guarantee optimum setup for every person using the V02 without anymore editing or adjustment (see also 2.4).



MEMORY KEY
INTO
INPUT SOCKET

contact memory key with input socket until beep sounds, all parameter for 3 personal presets are transfered into unit preset memory automatically.

7.2 SELECTION OF PRESETS

For selection of one of three personal presets

push — PRESET 1...3 — selection of preset 1..3 as active preset for voice processor adjustment

display: <preset name> or default PRESET 1...3

7.3 SAVING OF PRESETS INTO INTERNAL MEMORY

It is possible to store up to 100 sets of 3 presets each into an internal data memory of the unit.

push push — Push two preset buttons at the same time! It opens memory menu.

turn — MIC ON / EDIT — selection of memory number (LOAD)

push — ADJUST — selection of SAVE mode (NAME)

The default name is „MEMO *number*“. To change the name use the selector knob. Pushing the knob is selecting the characters place and turning is changing character setting.

push — ADJUST — selection of SAVE mode (SAVE)

push — ADJUST — for saving current set of presets into selected memory → “SAVE OK”

Push any other preset button to leave this menu without storage of presets.

7.4 LOADING OF PRESETS FROM INTERNAL MEMORY

It is possible to store up to 100 sets of 3 presets each into an internal data memory of the unit.

push push		Push two preset buttons at the same time! It opens memory menu.
turn	ADJUST	selection of memory number for loading
push	ADJUST	for loading 3 presets from selected memory as current set of presets → „LOAD OK“

Push any other preset button (preset 1,2 or 3) to leave this menu without loading of presets.

7.5 MUTING MICROPHONE INPUT (COUGH BUTTON)

For muting microphone signal.

push	MICRO ON	switch off microphon signal (mute)
	display: OFF	,green LED in button is switched off

To release mic muting push MICRO ON button again!

The mute function is available during use of other inputs also.

7.6 CHANGING VOICE PROCESSOR INPUT

For changing of voice processor input (see also 2.6).

push	EDIT	for open EDIT mode
turn	ADJUST	for selection of VP-INPUT
push	ADJUST	for switching to parameter edit mode
turn	ADJUST	for changing of VP-INPUT MIC/LINE - analog input DIGITAL - AES/EBU input
push	EDIT	for leaving edit mode, return to normal working mode

Note: If unit is locked (EDIT disable) voice processor input is the only one parmeter selectable in EDIT mode. All other parameter are locked!

Caution! Please, you also read 5.3, 8.4 and 8.5 for the adaptation of the input sensitivity of analog input!

APPLICATION NOTES

8

8.1 DIFFERENT MEMORY FOR PRESET DATA AND FOR UNIT SETUP DATA

In setup menu one can edit various voice processor parameters. Most part of these parameters are user oriented values. These parameter values can be stored in actual preset memory and can be transferred for storage into memory keys. The rest of the parameters will be stored into the permanent setup memory. Parameter values stored in the permanent setup memory are important for the basic function of the unit. They can't be stored into memory keys and they will not be changed by loading new user preset data by using memory keys. Permanent setup data can only be changed in edit mode or by loading new unit setup file from PC via the serial remote interface.
The following table shows different kinds of setup parameters and their storage.

tab.2: parameter storage VAMP2

parameter	preset memory	setup memory	memory key transfer
presetname	x		x
gain	x		x
expander	x		x
compressor	x		x
de-esser	x		x
low filter	x		x
mid filter	x		x
hi-shelf filter	x		x
hi-cut	x		x
v p input		x	
phantom power		x	
agc		x	
mic gain		x	
pre amp		x	
output attenuation		x	
output dither		x	
device address		x	

V02 mic input can be used with +48V phantom power supply.
Phantom power supply for mic input is selectable in setup menu.

8.2 DISPLAY OF PHANTOM POWER STATUS

Phantom power supply status can be displayed by pushing PRESET 1 button 3 seconds or more. This display function is available independent of edit lock mode.

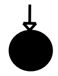
push
3sec

PRESET 1 — switches unit to phantom power supply display mode

display: PHANTOM ON or OFF

8.3 DISPLAY OF SOFTWARE VERSION

For displaying software version. The knowledge of software version can be helpful in consulting technical support regarding technical problems.



push
3sec

PRESET 2 — switches unit to software version display mode

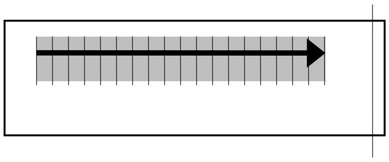
display: VERSION D:XX C:YY

xx - version number of dsp software
yy - version number of controller software

8.4 DISPLAY OF AUDIO LEVEL

In normal working mode MIC display is showing audio level at analog MIC-input. The displayed level is **input level + gain**! There are following conditions to see:

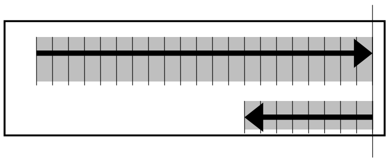
<u>normal input condition</u>



0dBFS

input level (MIC)
+ gain (mic gain, gain)
1 bar = 5dB

<u>raised level of processed material, limiter is working</u>



0dBFS

input level + gain
+ limiter gain reduction

<u>to much input level</u>

0dBFS

input is clipping!
ADC is overloaded!
Change PRE AMP!
(LOW LVL to HIGH LVL!)

8.5 SELECTION OF PARAMETERS TO INCREASE LOUDNESS

When working with analog inputs it is very important not to overload the A/D convertor (ADC), in order to ensure that the ADC always provides accurate linear conversion of the analogue input signal to the digital audio signal which is used for internal processing. **Display message CLIP should never be visible.**

There are two different mic preamplifiers working in VAMP2.

Mic preamp1	analog gain	+40dB
	clipping level	-25dBu
	working, if PRE AMP = LOW LVL	
Mic preamp2	analog gain	+8dB
	clipping level	+6dBu
	working, if PRE AMP = HIGH LVL	

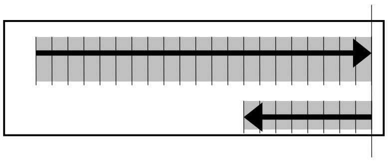
If the MIC input is clipping one has to select the HIGH LVL preamplifier (clipping level +6dBu). If the MIC input is clipping furthermore one has to reduce loudness of recorded speech (or increase distance between speaker and microphone).

The analog line level at input of the VAMP2 should be set so that the maximum possible studio output level which will occur in practice must not overload the A/D converter (maximum clipping level = +15dBu, if line sensitivity is selected!).

Signal compression and the loudness enhancement of the digital audio signal resulting from it 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 (mic gain + gain)** combined with the inaudible **limitation** of individual, remaining peak levels with the limiter.

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...2...4 bars and should not be active 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.
(Note: Filtering with boost in overlapping bands is raising level of processed audio material remarkable. Sometimes it could be necessary to reduce gain after setting of the filters to reduce limiter activitiy.)

TECHNICAL SPECIFICATIONS

9

VOICE PROCESSOR

mic in

input:	electronical. floating balanced
input impedance:	5 kOhm
max. input level:	+12 dBu, if LINE input is selected
(clipping level)	+6 dBu, if MIC input is selected and PRE AMP = HIGH LVL
	-25 dBu, if MIC input is selected and PRE AMP = LOW LVL
gain (mic gain + gain):	0 ...+90 dB, in 1 dB steps
frequency response:	40 Hz ... 15 kHz
CMRR:	-90 dB @ 50 Hz
	-85 dB @ 15kHz
THD + N:	-102 dB @ max. inp. level
equiv. inp. noise:	@ 200 Ohm
	RMS: -129 dBFS
	A-wght: -131 dBFS
	QP-CCIR : -116,5 dBFS

analogue out

output:	electronical balanced
output impedance:	50 Ohm
output level:	+11...+22 dBu adjustable, @ 0 dBFS
frequ. resp.:	20 Hz ... 20 kHz , (-0.5 dB)
with calibration	0 dBFS = +15 dBu, 0dB gain
THD:	<0.005 %
dynamic range:	93 dB
noise :	-88 dBu (RMS)
	-79 dBu (QP-CCIR)

digital input

format :	AES/EBU, S/PDIF, EIAJ-340
input:	XLR, balanced , 110 Ohm, 2 Vpp
	RCA, unbal. , 75 Ohm, 0.5 Vpp
input sample rate:	30 .. 50 kHz
	asynchronous sample rate conversion, 16 bit
	dynamic range: 94 dB
	(not transparent for C- or U-bits)

digital out

format :	AES/EBU, (professional or consumer channel status selectable)
output:	XLR, balanced , 110 Ohm, 5 Vpp
	RCA, unbal. , 75 Ohm, 1 Vpp
output sample rate:	48 kHz or locked to EXT. SYNC

sync

format :	AES/EBU or Wordclock, auto detection
sample rate:	30 .. 50 kHz
sync. level:	>3 Vpp
input impedance:	1 kOhm
connector:	BNC

WARRANTY AND SERVICE INFORMATION

10

JÜNGER AUDIO grants a two-year warranty on the

digital voice processor V02

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

JÜNGER AUDIO - Studiotechnik GmbH

Justus-von-Liebig-Strasse 7

D - 12489 Berlin
GERMANY

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

KONFORMITÄTSERKLÄRUNG

DECLARATION OF CONFORMITY

Geräteart :	**digital voice processor**
Type of equipment :	**digital voice processor**

Produkt / Product : **v02**

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/EEC and 92/31/EEC) Richtlinie der Rates zur Angleichung der Rechtsvorschriften der Mitgliedsstaaten über die elektromagnetische Verträglichkeit Council Directive on the approximation of the laws of the Member States relating to electromagnetic compatibility
73/23/EWG	(geändert durch 93/68/EWG) (changed by 93/68/EEC) Richtlinie des Rates vom 19. Februar 1973 betreffend elektrische Betriebsmittel zur Verwendung innerhalb bestimmter Spannungsgrenzen Council Directive of February 19th 1973 concerning electircal equipment for operation within certain voltage limits

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
EN 60065 :	2002

Dieser Erklärung liegen zugrunde :	Prüfbericht(e) des EMV-Prüflabors Interne Vorschriften zur Sicherheits-Prüfung
This certification is based on :	Test report(s) generated by EMC-test laboratory Internal regulations for safety check
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)	(Herbert Jünger, Geschäftsführer/Managing Director)

professional audio products

digital dynamics processors	d01, d02 accent1, accent2
digital filter processor	e07
surround products	multichannel digital dynamics processor ORION 5.1 level controller 206
digital voice processing	voice and monitor processor v01 digital voice processor v02 dual channel voice processor v03 digital voice processor v05
digital desktop mixer	mix4
transmission signal processing	digital transmission processor d07 digital transmission limiter mpx01
4channel processors b40series	digital audio toolbox b40 digital audio limiter b41 digital dynamics processor b42 digital audio toolbox b43 SDI audio converter / router b44 digital audio delay b45
digital audio modular processing system C8000	
SDI interface modules SDI20	

jünger audio

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