

## d07

### digital transmission processor

Manual

release 2.0.2 / 2015-10-08





## FOREWORD

The logo consists of the text 'd07' in a white, sans-serif font, centered within a solid black square. A thin vertical line is positioned to the left of the square.

Thank you for buying and using the transmission processor d07.

You have not only acquired 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 d07.

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, please do not hesitate to contact us.

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# FUNCTIONAL DESCRIPTION



The mature technology of the digital transmission processor d07 has now been supplemented by the Level Magic™. The d07 is originally designed for optimised levelling of program signals for FM broadcast and TV transmission and reliable protection of transmission paths against overload at the output of studios, OB vans as well as satellite up-links. The device operates fully digitally and, beside AES/EBU interface, it makes use of high end 24bit A/D converters so that digital dynamic processing is possible for analog as well as digital signals.

Besides the previous levelling by means of an AGC, Compressor and Peak-Limiter referring to the peak- level for achieving the maximum loudness and energy of the signal, it's now possible to use the more signal wave adaptive new technology of the Level Magic™ with two reference levels: An Operating-Level for signal processing by the AGC and the Transient processor and the peak-Level for the Brickwall-Limiter.

In both operating methods you can factor the adaptive pre-emphasis, the maximum value for the peak frequency deviation and the MPX-Limiter into levelling of the signal (see application notes).

The dynamic range processor principles developed by Junger 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 Junger Audio.

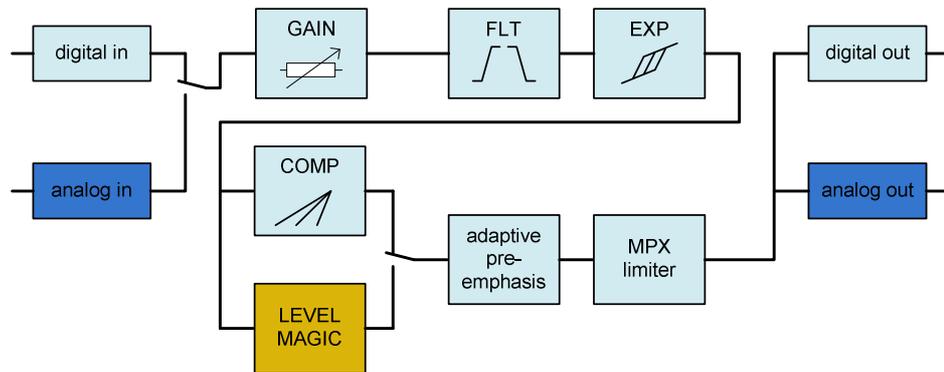
The unit is easy to operate and requires only a limited number of settings to be made by the user to achieve optimum results. All other parameters necessary for inaudible processing are continuously automatically controlled in response to changes in the programme signal.

## 1.1 BASIC DESCRIPTION

## features

- 2-channel digital audio amplifier and limiter for broadcast transmission signals
- new Level Magic™ technology
- audio signal processing also in consideration of pre-emphasis and MPX-Power
- Digital I/Os, AES/EBU format
- 24bit ADV and DAC
- Stereo and dual channel mode
- Parallel and serial remote control
- Equal functionality to the processor C8007 of the C8000 modular system

## 1.2 BLOCK DIAGRAM



Further description of the processor principles see Application notes B!

# INSTALLATION

The digital audio level processor d07 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 occurred in transit.

The digital audio level processor d07 is a device under the safety category *Schutzklasse 1* in keeping with the VDE 0804 standards and may only be 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.

The digital audio level processor d07 is equipped with standard connectors (see also chapter 3). Before connecting the digital audio level processor d07 switch the power off at all connected units.

The digital audio level processor d07 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.

The digital audio level processor d07 should not be installed near units which produce strong magnetic fields or extreme heat. Do not install the audio 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.



## **2.1 UNPACK THE UNIT**

## **2.2 POWER SUPPLY**

## **2.3 CONNECTIONS**

## **2.4 RACK MOUNTING**

## **2.5 OPERATION SAFETY**

## 2.6 SYNCHRONIZATION OF DIGITAL OUTPUT

The digital transmission processor d07 has a digital signal output. For 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 sync source is made in the SYNC MODE menu during setup. If the chosen sync input is connected with the sync signal, this signal is used for synchronization automatically. All sync sources can be used for locking A/D-converters at the analogue inputs as well. The digital output signal can be clocked with the following clock frequencies:

- INTERNAL** locks both the A/D-converters and the digital output with the internal reference 44,1 or 48 kHz. Digital inputs are connected via sample rate converter
  - AES INPUT** locks with the clock frequency of the input signal at digital input CH 1/2 (AES/EBU, 44,1...48 kHz)
  - EXT AES** locks with the AES signal at the sync input (AES, 44,1...48 kHz) Digital inputs are connected via sample rate converter
  - EXT WCLK** locks with the word clock signal at the sync input (WCLK, 44,1...48 kHz) Digital inputs are connected via sample rate converter
- optional:
- EXT VIDEO** locks with black burst at sync input (internal 48 kHz) Digital inputs are connected via sample rate converter

## 2.7 AUDIO CONNECTIONS

The analog audio inputs are RFI filtered and analogue outputs are balanced and floating like transformer coupled devices. All the audio connectors are via rear panel mounted connectors. Standard XLR connectors are used. These are always wired to the AES standard:

pin 1	X	Screen screen
pin 2	L	Live audio 0°
pin 3	R	Return audio 180°

Balanced connections are preferred whenever the other equipment provides balanced inputs/outputs. All line level connections should be wired with twin screened cable for low noise and reliability. The screens of the cable should be connected at one end only. Input cable screening therefore needs to be derived from the signal source end as pin 1 is ground lifted at low frequencies for the inputs.

If the equipment driving the digital audio level processor d07 has unbalanced outputs then you will need to add a wire jumper such that the screen connection of Pin 1 of the XLR is shorted to Pin 3.  
If the equipment being connected to the d07 have only unbalanced inputs, then we recommend still to use a balanced (i.e. 2 core shielded cable) cable where Pin 1 and Pin 3 are connected in the cable ends away from the digital audio level processor d07.

The digital audio toolbox b40 can be remote-controlled by means of parallel GPI contacts.

**use :** remote-controlled changeover of presets

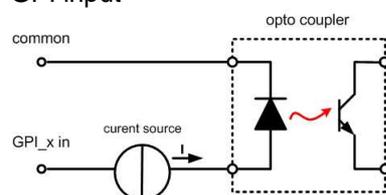
**connector:** D-SUB 15pin, female

Pin assignment of the connector :

Pin	Signal name	Functions
1	GPI1 in	Defined by d07 config
2	GPI2 in	Defined by d07 config
3	GPI3 in	Defined by d07 config
4	GPI4 in	Defined by d07 config
5	GPI5 in	Defined by d07 config
6	GPI6 in	Defined by d07 config
7	GPI7 in	Defined by d07 config
8	GPI8 in	Defined by d07 config
9	+ 5V	110 $\Omega$
10	GPI1/GPI2 common	
11	GPI3 common	
12	GPI4 common	
13	GPI5 common	
14	GPI6 common	
15	GPI7/GPI8 common	
Shield	GND	

Electrical specification:

GPI input

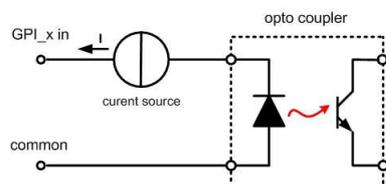


potential free by opto-coupler in line with a current source

**ON:** +3.5...+30V between **GPIx** input and **GPx common**

**OFF:** less then 1.5V between **GPIx** input and **GPIx common**

For serial numbers **S/N 182 and higher** (HW Revision 2 and higher) the polarity of the GPI inputs has been changed. to make use of the internal **ground based** auxiliary 5 V for "low active" switching.



**ON:** -3.5...-30V between **GPIx** input and **GPx common**

**OFF :** less then 1.5V between **GPIx** input and **GPIx common**

Signal duration must be at least 50msec.

**Note :** An internal auxiliary voltage feed of +5V is available on pin 9 via a 110  $\Omega$  resistor. Ground is available from the shield of the connector only! When using the auxiliary voltage feed, there is no electrical isolation given anymore and the risk to inject unwanted noise is high!

**Important Note :** You must take care about the polarity of the external voltage applied to the GPIs. **Wrong polarity** may **destroy electronic components** and may **cause fire** inside the d07!

## 2.8 REMOTE CONTROL

### 2.8.1 GPI REMOTE CONTROL (PARALLEL REMOTE)

## 2.8.2 TALLY OUT

The digital audio level processor d07 can transmit specific device statuses via parallel Tally lines.

use: monitoring of the d07 status

Connector : D-SUB 25pin, female

Pin assignment of the connector :

Pin	Signal name	Functions
1	Tally 1 normally closed	
2	Tally 1 normally opened	Defined by d07 config
3	TALLY 2 common	
4	Tally 3 normally closed	
5	Tally 3 normally opened	Defined by d07 config
6	TALLY 4 common	
7	Tally 5 normally closed	
8	Tally 5 normally opened	Defined by d07 config
9	Tally 6 common	
10	Tally 7 normally closed	
11	Tally 7 normally opened	Defined by d07 config
12	TALLY 8 common	
13	+ 5V	110 Ohm
14	TALLY 1 common	
15	Tally 2 normally closed	
16	Tally 2 normally opened	Defined by d07 config
17	TALLY 3 common	
18	Tally 4 normally closed	
19	Tally 4 normally opened	Defined by d07 config
20	TALLY 5 common	
21	Tally 6 normally closed	
22	Tally 6 normally opened	Defined by d07 config
23	TALLY 7 common	
24	Tally 8 normally closed	
25	Tally 8 normally opened	Defined by d07 config
	Shielded	

### Electrical specifications:

GPO (Tally) potential free relay contact  
 common / normally closed / normally opened  
 24V - 1A  
 125V - 0,5A  
 $P_{max} = 62,5VA$

**Note :** An internal auxiliary voltage feed is available on pin 9 via a 110Ohm resistor. Ground is available from the shield of the connector only!  
 When using the auxiliary voltage feed, there is no electrical isolation given anymore and the risk to inject unwanted noise is high!

The d07 can be remote-controlled by means of serial remote RS-232/422 or via the CAN-bus.

use : remote-controlled changeover of presets

protocol: available on request

Connector : D-SUB 9pin, female

### 2.8.3 SERIAL REMOTE CONTROL (RS-422)

Pin assignment of the connector in serial interface mode :

Pin	Signal name	Functions
1	Rx +	RS422
2	TxD	RS232
3	RxD	RS232
4	NC	not used
5	GND	Ground
6	Rx -	RS422
7	NC	not used
8	Tx -	RS422
9	Tx +	RS422

Pin assignment in CAN-bus mode :

Pin	Signal name	Functions
1	NC	Not used
2	CAN-I	CAN-bus low signal
3	NC	Not used
4	NC	Not used
5	GND	Ground
6	GND	Ground
7	CAN-H	CAN-bus high signal
8	NC	Not used
9	NC	Not used

This connector has multiple functions which may be internally changed by connectors and jumpers. The factory default format setting is RS232 and the it is connected with the serial interface of the LAN Controller.

By using a terminal program (115kB/sec. 8,N,1 no flow control) you may communicate with the consol of the LAN Controller, e.g. to change the IP configuration of the device.

## 2.9 LAN INTERFACE

Connector : RJ 45 with status LEDs

Pin assignment of the connector :

Pin	Signal name	Functions
1	TX +	Ethernet send
2	TX -	Ethernet send
3	RX +	Ethernet receive
4		
5		
6	RX -	Ethernet receive
7		
8		
9		

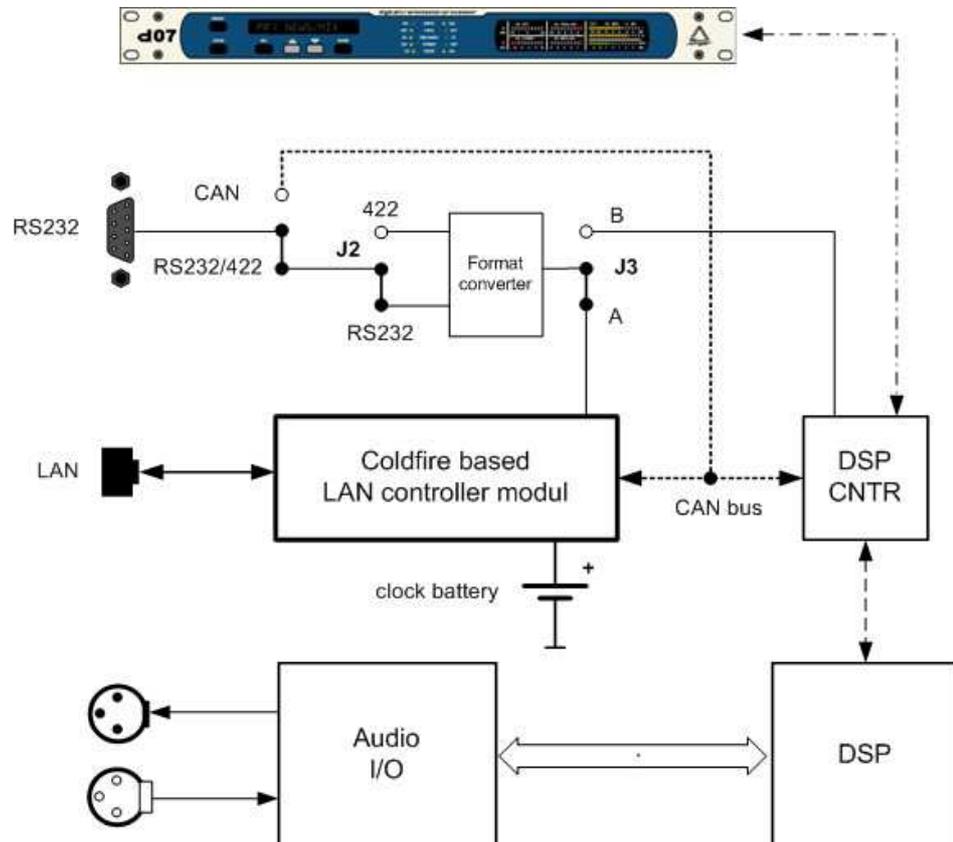
Electrical specifications: 100Mbit/s auto negotiation port

Application remarks :

This port allows the remote control of the d07 by **TCP/IP over Ethernet**. Setting up the network configuration is described in B 6.

The LAN Controller of the d07 has a web server which offers a graphical user interface (GUI). For proper operation you need IE7 or FireFox 2.0. There you input the IP address of the d07 as an URL.

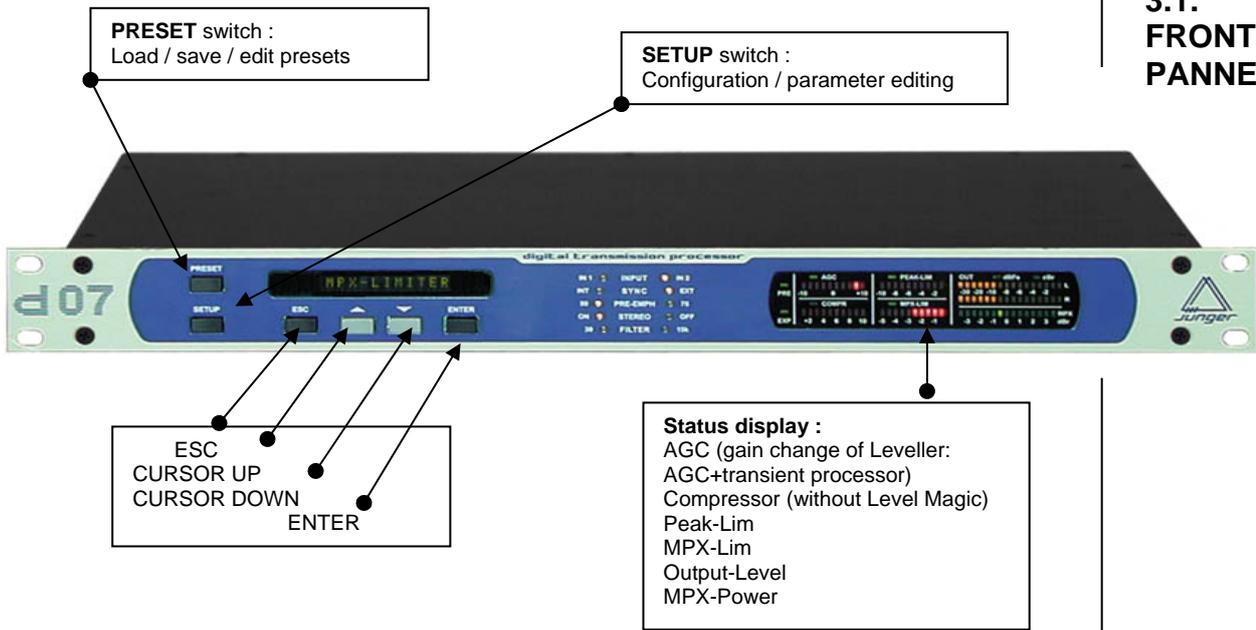
d07 control block diagram :



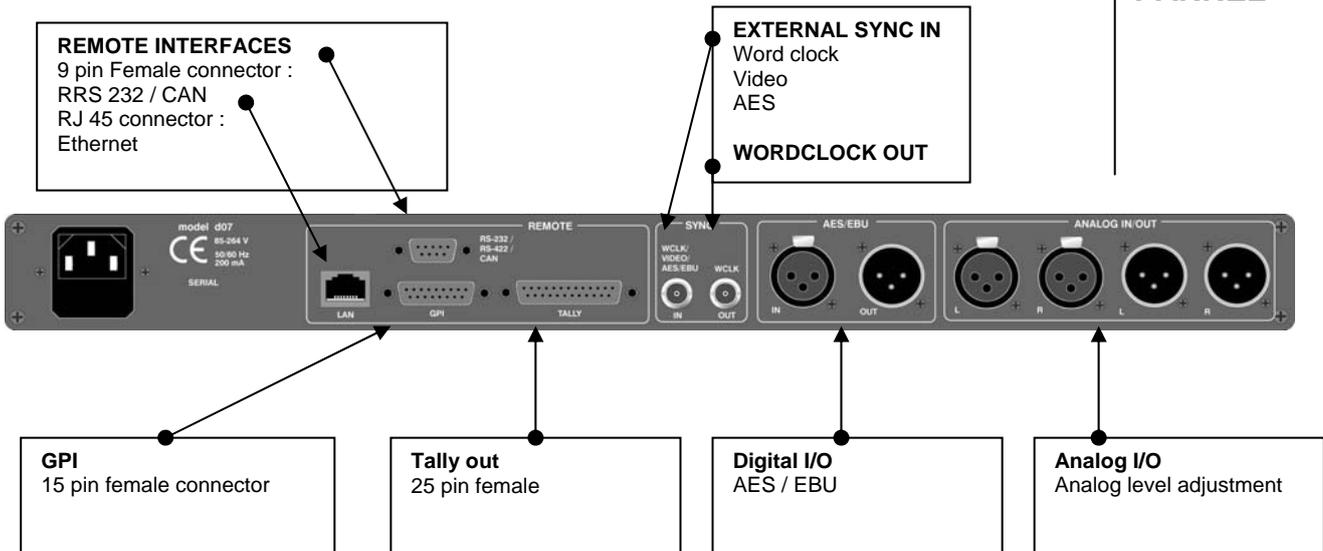
# LOCATION OF PARTS AND CONTROLS



## 3.1. FRONT PANNEL



## 3.2 REAR PANNEL



### **POWER INPUT**

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

### **REMOTE**

serial remote interface RS-422 (232)  
connector: 9pin SUB-D, female

### **GPI**

parallel remote interface  
TALLY-out open relais contact  
connector: 25pin SUB-D, female  
GPI-in +3,5...+30V potential-free  
connector: 15pin SUB-D, female

### **SYNC**

SYNC IN input for ext. sync signal (AES 3 format, 75 Ohm, unbal)  
or video sync signal (blackburst, 75 Ohm, unbal) or  
wordclock sync signal, TTL level, unbal  
connector: BNC socket

WCLK OUT output for word clock (system clock of d07)  
connector: BNC socket

### **DIGITAL IN**

input for AES/EBU standard format  
connector: XLR female panel jack

### **DIGITAL OUT**

output for AES/EBU standard format  
connector: XLR male panel jack

### **ANALOG IN/OUT**

Analog input to 24 bit A/D-converter  
Input floating balanced, XLR connector female  
Analog output from 24 bit D/A-converter  
Output floating balanced, XLR connector m

Some basic settings can be made by switches and jumpers at the internal circuit boards of the unit. These settings can occur general changes for operation and should be made by qualified engineering staff only.

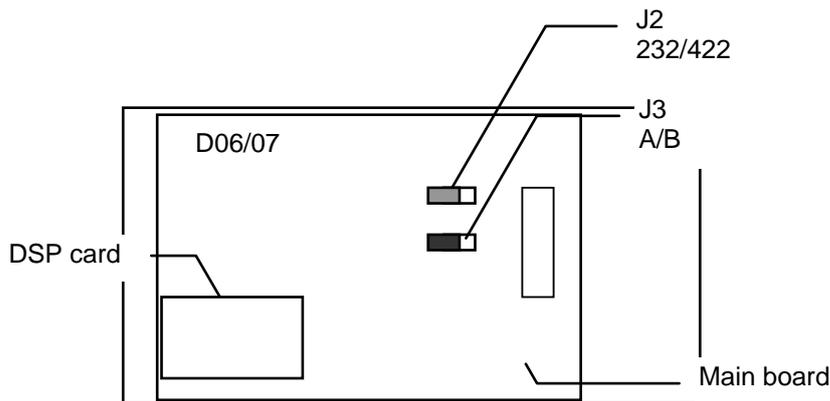
### **Internal**

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.

## **3.3 Switches and Jumpers for Configuration**

## Selection of the serial remote interface



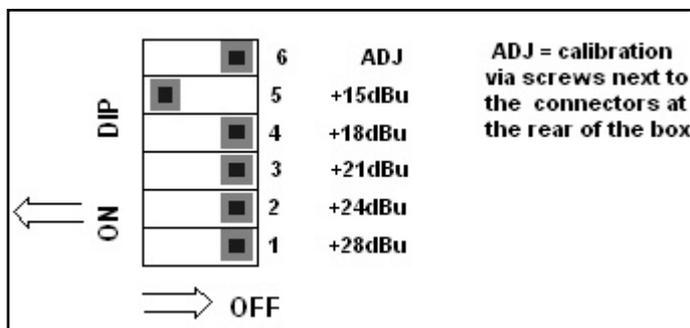
J2 – RS-232 / RS-422: Selection of the serial remote interface (see 3.8.3)

J3 – A / B: for factory use only, set to A

## calibration of the analog in-and outputs

At our factory the d06 is calibrated German broadcast standard +15dBu = 0dBFS. If you want to use a different referencer standard (say +24dBu = 0dBFS) you can change the setting via dip switches on the main board of the d06.

→ The switches are near the analog input and output hardware on the PCB.



With factory setting of +15dBu=0dBFS the dip switch "+15dBu" will be turned ON. To change the setting to another standard you just have to slide this dip switch to the right (OFF) and slide the needed dip switch that corresponds to the reference standard you are using to the left (ON).

→ Make sure that there is always just ONE dip switch turned ON!  
Exception: They may all be OFF if you are using a CUSTOM reference level

### **CUSTOM Reference Level (using Switch 6 ADJ)**

If none of these standard reference settings correspond to your needs you can set the reference to a CUSTOM level by adjusting the input sensitivity by the two potentiometers (L and R) next to the analog input and output connectors at the rear of the box.

→ This should only be done by experienced engineers with measuring instruments!

To set the reference level manually, follow these steps:

1. Set all dip switches to "OFF" except #6, ADJ. –set it to ON
2. Adjust the potentiometers to the desired CUSTOM reference (setting). For this you need to feed the analog input with a known reference level and measure the digital output.  
Make sure that the DSP processing is bypassed, otherwise there could be DSP gain active!
3. When the adjustments are complete, 'capture' the custom settings by setting dip switch #6 to OFF.  
Otherwise your reference level could be changed by accident at the potentiometers.

rear of the box. This should only be done by experienced engineers with measuring instruments! After having adjusted the input level bring the dip switch back to the "OFF" position.

# Operation

The d07 transmission processor is very easy to use.

You can do all settings with the keys on the front of the device, via remote or software.

The d07 uses two groups of data, in the **CONFIG** area to set up device functions (selection of an input, synchronization or input of a device name) and in the **PARAMETER** area to control the signal processing (setting up processing parameters, MPX-limiter, etc.).

**SETUP** directs to the set up menus. Here one can select between **CONFIG** for device related settings and **PARAMETER** for signal processing settings.

**PRESET** directs to loading, saving and editing of one of the user **PRESETs**.

All current data is kept in a non volatile memory. Therefore they are immediately available when turning on the device. All changes to a parameter are effective immediately. Pressing **ENTER** will approve it. By pressing **ESC** the changes will be rejected and the display returns to the parameter menu.

Loading of a **PRESET** will immediately carry over the values into the operating memory. By clicking free cross fade they will be effective immediately.

There is the possibility to edit and store **PRESETs** in the back ground, without taking them over into the operating memory to prepare the device for a different kind of program that is awaited.

**NAVIGATION** through menus of the d07 is done by the

**▲** **▼** **ENTER** **ESC** keys :

**ENTER** - directs into the next possible navigation level

- exit settings

- stores altered values

- directs back to the previous navigation level

**ESC** - abort (changes are rejected)

- back to the previous navigation level

**▲** - scrolling through the navigation level

and **▼** - switching status like ON or OFF

**▼** - altering of values



## 4.1 FRONT PANEL OPERATION Function of the d07 keys

### 4.1.1. NAVIGATION

### 4.1.2. MAIN DISPLAY

When you switch on the d07 the main display will show you the current **controller software** and **hardware version** of your device. You can always have a look on it by pressing **ENTER** if you are not in a menu (**SETUP** or **PRESET**).

After a few seconds the main display switches to showing the input level (R/L).

By pressing  you can step through the following status details:  


Display	Description
<b>MAIN DISPLAY</b>	
D07 C:xx D:yy	Device, Controller software, DSP version
L: xxx R: xxx	Input Level channel L/R in DBFS/dBr (CONFIG out meter)
D07 DEVICE	Device name, 16 characters possible(CONFIG device name)
PRESET x: yyy	Shows current preset
MPX-POWER -14.9dB	Shows actual MPX-Power of the output signal
AGC 0.0 dB	Instantaneous value of the AGC gain
BAL R: 0.0dB	

### 4.1.3. MENU Preset / Setup

The chart on the following pages gives an overview over the menu structure, its parameters and the available ranges:

Menu item	Value/range	Description
<b>PRESET*</b>		
LOAD PRESET	Preset 1-4	<i>User presets</i>
	TV U	<i>TV universal</i>
	R U	<i>Radio universal</i>
	R SP	<i>Radio Sports</i>
	TV L	<i>TV live</i>
	TV M	<i>TV movie</i>
	R CL	<i>Radio classic</i>
EDIT PRESET	Preset 1-4	<i>Here you can change the settings of your individual presets</i>
SAVE PRESET	Preset 1-4	<i>Here you can save your individual preset in one of 4 available user presets</i>

\* When you switch on your d07 or initialize it, the "initialize" preset is loaded. Before you start checking the parameters of the d07 you should load one of the factory presets. Choose the one most fitting to the genre of your program (see list below).

SETUP		
<b>CONFIG</b>		
INPUT	INPUT 1: analog	Select the input mode according to your input signal (analog/digital AES/EBU)
	INPUT 2: digital	
LEVEL MAGIC	OFF/ON	switch Level Magic off/on
PASSWORD	1 2 3 4 (factory preset)	Set your own password to lock the device
LOCK	OFF/ON	Device can be protected against accidental changes while transmission operation
OUT METER	dBFS/dBr	Select between the display of relative output level in dBr to the setup value of the limiter or absolute value in dBFS
DEVICE NAME	16 characters possible	Set your individual device name
CAN ID	00-99	Device address for the CAN-bus for the remote control (optional)
TALLY 1-8	off, preset 1-4, stereo, Exp, Compr, Lim, Preemp, Clip, Input2, Bypass	8 TALLY outputs are carried out as relay change –over switches. One of 12 states of the d07 can be allocated to them
GPI 1-8	off, preset 1-4, stereo, Input2, Bypass	GPI input are carried out as opto coupler driven by a current source. One of seven predefined states of the d07 can be remote controlled by them
SYNC	Video, Wclk, Ext AES, Input AES, INT 44.1, INT 48	Selection of the SYNC source

PARAMETER			
GAIN		-20 to +20dB	Setting the initial gain
STEREO		OFF/ON	For stereo operation with the d07 the control circuits of the dynamic sections can be linked
FILTER		30Hz	Switching off/on 30Hz low cut filter
		15kHz	Switching off/on 15kHz-FIR-Filter
PREEMPHASIS	PRE MODE	OFF/50µs/75µ	The d07 controls high frequency signal components to adapt the audio signal to the predefined pre-emphasis.
	PRE THRES		Setting has to be equal/bigger than the Limiter Thresh (see B 5.3)
PEAK-LIMITER	LIM THRESH		determines the max output level of the d07
	LIM PROG	Pop, speech, uni, live, classic	Characteristic of the LIMITER can be adapted to the program material
MPX-LIMITER	POWER	-4 to +4dB / OFF	Limits the power of the MPX-signal
	PROCESSING	soft/mid/hard	soft –moderate mpx limiter performance as integrated in d07 firmware up to version 1.9.0 mid – faster performance of mpx limiter hard – very fast reaction to mpx power changes
EXPANDER	EXP THRESH	OFF/-20 to -60dBFS	Levels lower than EXP TRESH will be lowered => Enhancement of S/N ratio, active after AGC
	EXP RANGE	OFF/-0 to -20dBFS	
	EXP ATTACK	0,2 to 4sec	
AUTO-BALANCE	BAL RANGE	0 to 20dB	In stereo mode the balance between the two channels can be controlled automatically (reference: left channel), e.g. valuable if there are level changes on the transmission path
	BAL TIME	1,2,5,10,20,40sec/min, 1h, 2h	

LEVEL MAGIC ON			
LEVELLER	OPERATING LEVEL	-40...0dBFS	Desired target level for the levelling process. Reference Level for the Transient Processor and Leveller
	TR P RANGE	0...15dB	Determines the maximum gain change applied by the Transient Processor when there are fast input level changes. Large range values are reducing the dynamic range, especially in combination with the transient program "hard"
	TP PROG	soft/mid/hard	This parameter describes the characteristic of gain change by the transient processor. It has to be chosen dependent on your program genre. If there are just a few level changes or you want to keep the original dynamic range best (e.g. classical music), you have to choose "soft". For mixed program "mid" should be best in most cases. And for live venues (sport etc.) with frequent unexpected level changes the adjustment "hard" is required.
	SILENCE GATE = AGC gate	-60... -20dBFS	If the input level falls below this threshold level, the gain change of the AGC freezes immediately (transient processor still active). After appr. 20 seconds input level below silence gate the current gain change is slowly moving to the longterm average gain.
	LEV TIME = AGC time	10s...2h	Describes the time of development for the AGC to reach the maximum possible gain change (range value). The ratio of gain change should never be faster then 3 seconds for 1 dB!! We are recommending a setting of 4...5 seconds for 1dB gain change by the AGC. Therefore the AGC time is basically determined by the AGC range value. A range setting of 10 dB requires a time setting of minimum 40 seconds.
	LEV RANGE = AGC range	0...40dB	Determines the maximum gain change applied by the AGC. AGC Range must be bigger then the expected difference between the average input level and the operating level. If there is for example an average input level of -23dBFS and your OP-Level is -18dBFS, the AGC needs at least a range of 5dB. In most cases an AGC range of 10dB is a good choice
LEVEL MAGIC OFF			
COMPR	COMPR RATIO	OFF/ 1.1 to 4.0	<i>Compresses the audio signal =&gt; increase of loudness, complicated settings are not necessary because there are diverse adaptive control algorithms, which can be accommodated in an optimal way with the parameter COPM program</i>
	COMPR PROG	live, pop, speech, uni, classic	
	COMPR RANGE	0 to 20dB	

## 4.1.4. FACTORY PRESETS

	Initiali ze preset	P5 TV uni	P6 radio uni	P8 Radio spec h	P8 TV live	P9 TV Movie	P10 Radio classic
<b>SETUP</b>							
<b>CONFIG</b>							
INPUT	analo g						
LEVEL MAGIC	on	on	on	on	on	on	on
PASSWORD	1234						
LOCK	off						
OUT METER	dBFS						
DEVICE NAME	device d07	Grey marked parameters are not saved in the preset!!!					
CAN ID	00						
TALLY 1-8	off						
GPI 1-8	off						
SYNC	INT 48						

PARAMETER								
GAIN		0.0	0.0	0.0	0.0	0.0	0.0	0.0
STEREO		off	on	on	off	off	on	on
FILTER	30Hz	off	on	on	on	on	on	on
	15kHz	off	on	on	on	on	on	on
PREEMPHASIS	PRE MODE	off	off	off	off	off	off	off
	PRE THRES	0.0	0.0	0.0	0.0	0.0	0.0	0.0
PEAK-LIMITER	LIM THRESH	0.0	-9.0	0.0	0.0	-9.0	-9.0	0.0
	LIM PROG	uni	uni	uni	speech	live	uni	classic
MPX-LIMITER	MPX-POWER	off	off	off	off	off	off	off
	MPX- PROCESSING	soft	soft	soft	soft	soft	soft	soft
EXPANDER	EXP THRESH	off	off	off	off	off	off	off
	EXP RANGE	off	off	off	off	off	off	off
AUTO- BALANCE	EXP ATTACK	0.5	0.5	0.5	0.5	0.5	0.5	0.5
	BAL RANGE	0.0	0.0	0.0	0.0	0.0	0.0	0.0
	BAL TIME	1min	1min	1min	1min	1min	1min	1min

LEVEL MAGIC ON								
LEVELLER	OPERATING LEVEL	0	-18	-9	-9	-18	-18	-9
	TR P RANGE	0	10	10	15	10	6	3
	TP PROG	Soft	Mid	Mid	Hard	Hard	Mid	Soft
	SILENCE GATE = AGC gate	-40	-50	-50	-40	-50	-50	-60
	LEV TIME = AGC time	1 min	40sec	40sec	20sec	20sec	2min	2min
	LEV RANGE = AGC range	0	10dB	10dB	10dB	10dB	15dB	10dB
<b>LEVEL MAGIC OFF</b>								
COMPR	COMPR RATIO							
	COMPR PROG							
	COMPR RANGE							

### 4.1.5. User PRESETS

	Initialize preset	P1	P2	P3	P4
<b>SETUP</b>					
<b>CONFIG</b>					
INPUT	analog				
LEVEL MAGIC	on	on	on	on	on
PASSWORD	1234				
LOCK	off				
OUT METER	dBFS				
DEVICE NAME	Device d07			grey marked parameters are not saved in the preset!!!	
CAN ID	00				
TALLY 1-8	off				
GPI 1-8	off				
SYNC	INT 48				

PARAMETER					
GAIN		0.0			
STEREO		off			
FILTER	30Hz	off			
	15kHz	off			
PREEMPHASIS	PRE MODE	off			
	PRE THRES	0.0			
PEAK-LIMITER	LIM THRESH	0.0			
	LIM PROG	uni			
MPX-LIMITER	MPX-POWER	off			
	MPX-PROCESSING	soft			
EXPANDER	EXP THRESH	off			
	EXP RANGE	off			
	EXP ATTACK	0.5			
AUTO-BALANCE	BAL RANGE	0.0			
	BAL TIME	1min			

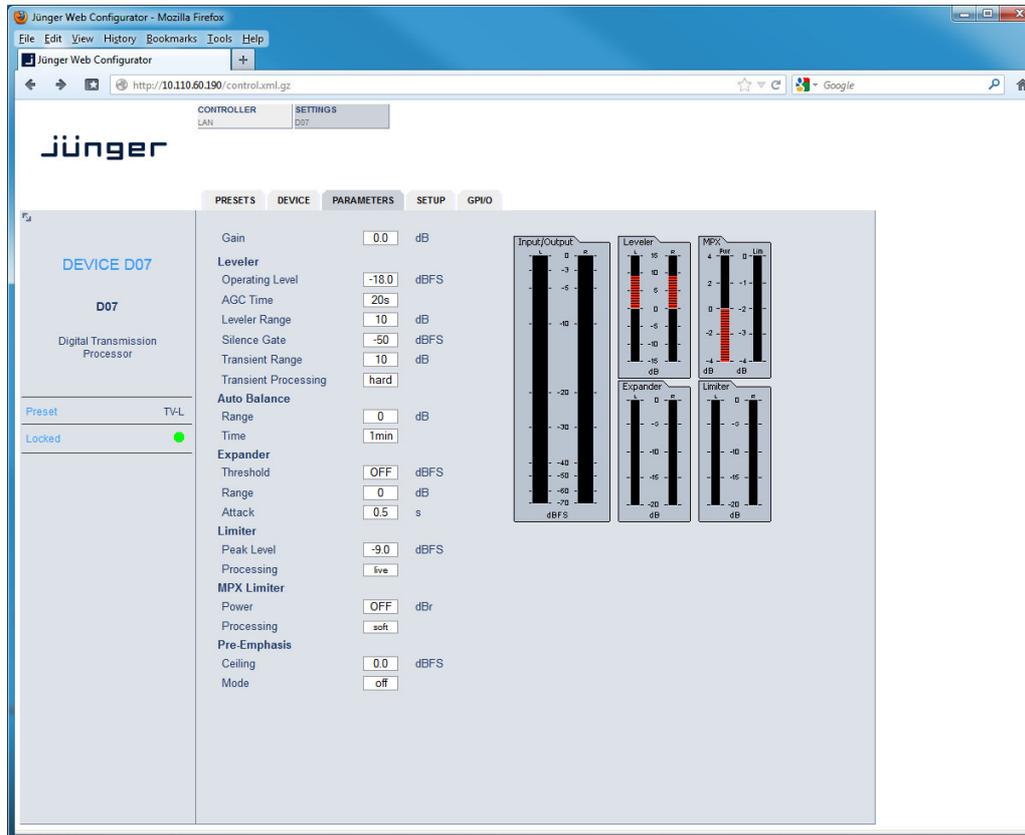
LEVEL MAGIC ON					
LEVELLER	OPERATING LEVEL	0			
	TR P RANGE	0			
	TP PROG	Soft			
	SILENCE GATE = AGC gate	-40			
	LEV TIME = AGC time	1 min			
	LEV RANGE = AGC range	0			
LEVEL MAGIC OFF					
COMPR	COMPR RATIO				
	COMPR PROG				
	COMPR RANGE				

## 4.2. Web interface of the d07

## 4.2 Webinterface

After setting the IP address of the **d07** (See chapter B6 Network integration) you can control the **d07** via web browser besides front panel control.

Just type the IP address of your **d07** as an URL into the web browsers: "http:// IP-address" and press<Enter>, you will get the following page:



The **d07** has two different controllers on board. One is the network interface module and the other one is the DSP controller that also serves the front panel display. Both communicate internally via CAN bus. (see chapter 2.9).

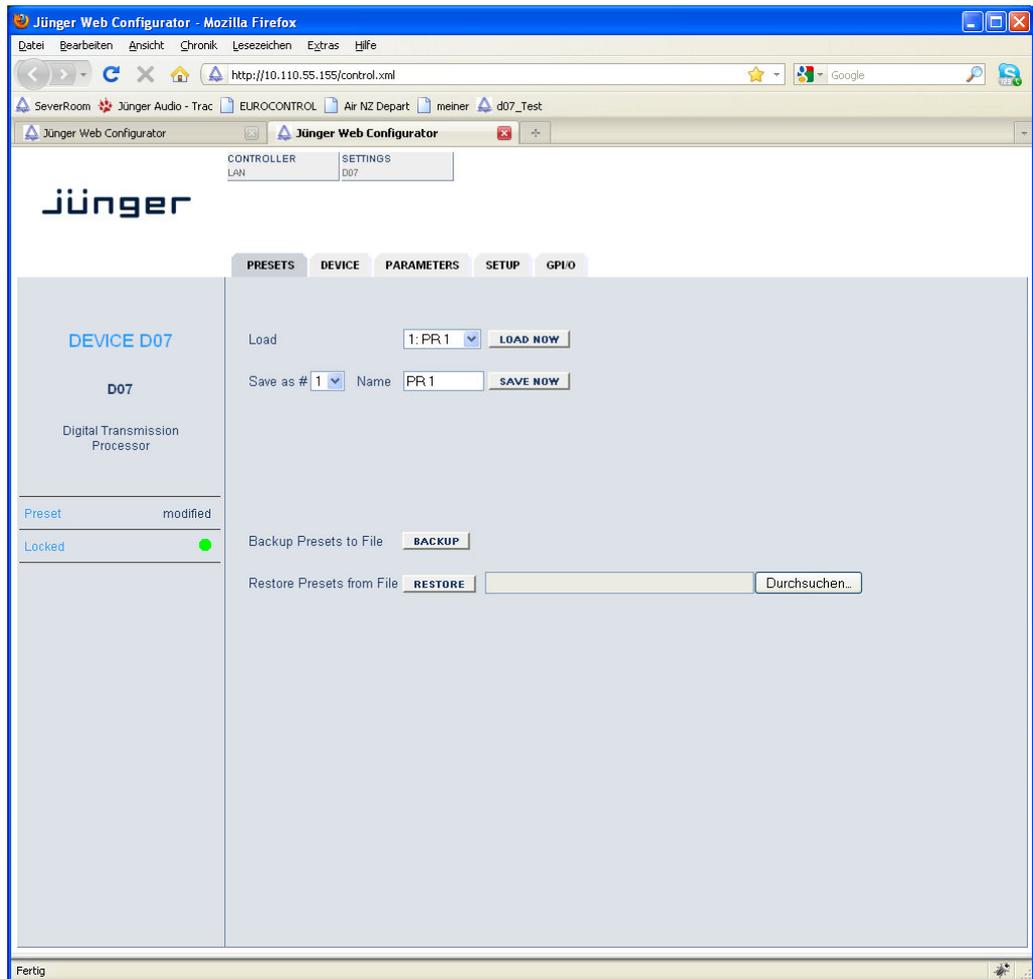
On top of the browser window you see two main buttons: **CONTROLLER** and **SETTINGS**. Via the **CONTROLLER** button you can setup the network interface (see chapter B6).

### 4.2.1 SETTING > PARAMETERS

(See above) here you can change the parameters online. The front panel display will follow these settings. I.e. if at "GAIN: 0.0 dB" is displayed the front panel and you change the value in the above field (see little slider) to 1.0dB, the front panel will also show 1.0dB and vice versa.

**For detailed description of the parameters see chapter 4.1.**

## 4.2.2 SETTING > PRESETS



The **d07** has **4 Presets**. These Presets are named **PR 1** to **PR 4** by default. The status window at the left hand side shows the name of the active preset. The phrase **“modified:”** will appear in line with the Preset name, if a preset parameter was changed by the operator.

**Load** select a preset by name and press **<LOAD NOW>**

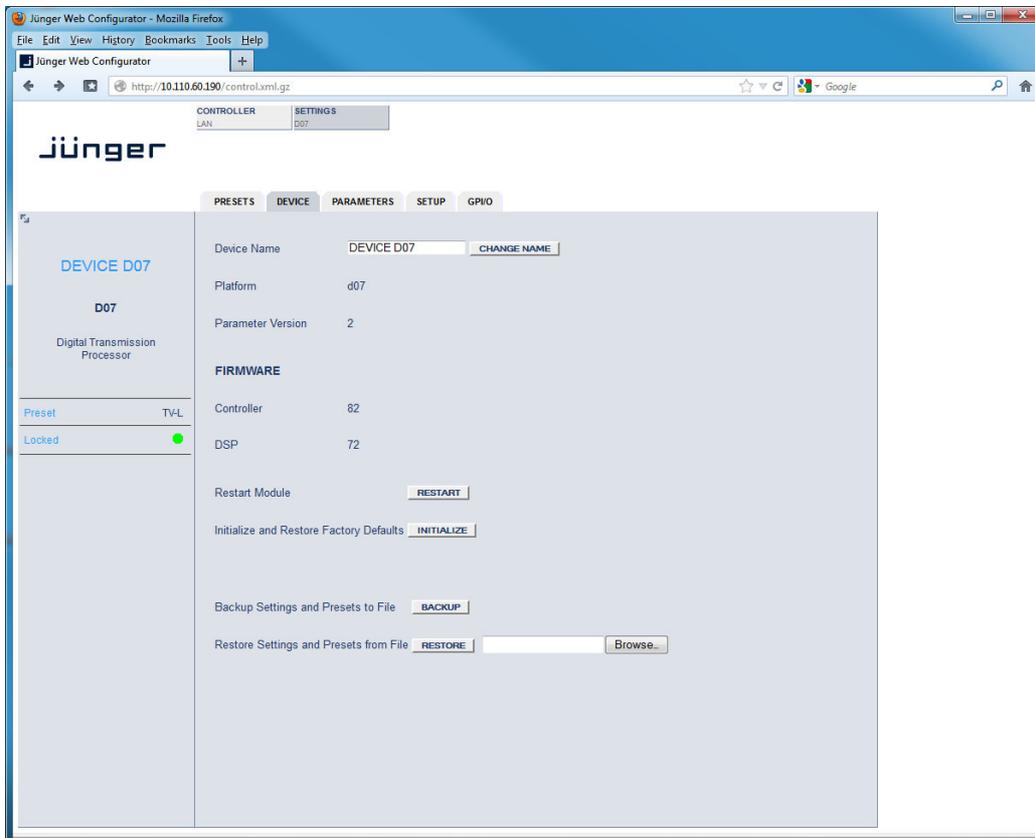
**Save as Preset #** select a preset memory number

**Name** assign the preset a **4 digit name** and press **<SAVE NOW>**

**Preset Clipboard** copy the active preset to a clip board, The data may be used by other modules inside the same frame.

**Backup/Restore Presets to/from file** **!!! Only Parameters of the tab “Parameters” are stored/restored. Parameters of the tab “Setup” must be configured manually. Parameters “High Pass 30Hz”, “Low Pass 15kHz”, and “Stereo Link” must be stored again within the Presets. !!!**

## 4.2.3 SETTING > DEVICE



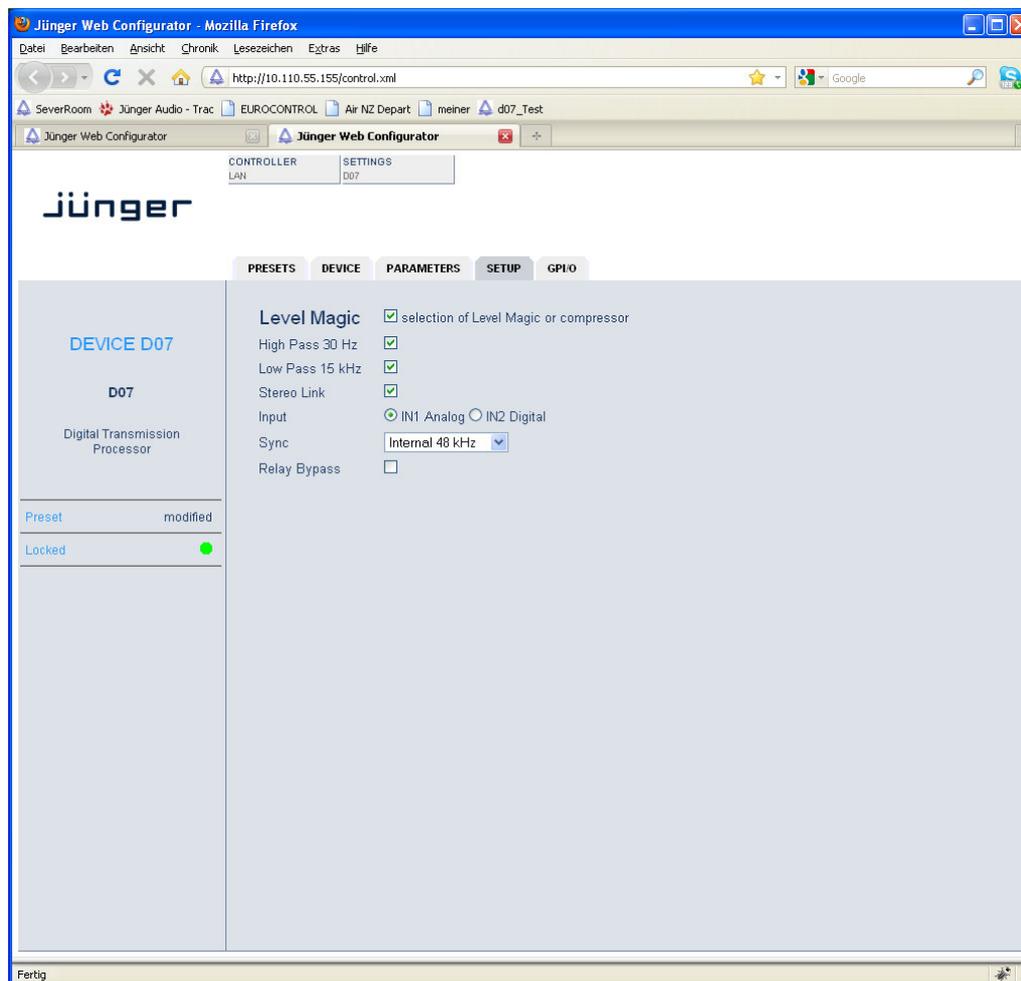
On the **DEVICE** tab you can assign a 16 digit name to the module, perform a warm start by pressing **<RESTART>** or initialize the module to factory default settings by pressing **<INITIALIZE>**.

You can **BACKUP / RESTORE** all module settings and parameters including presets as well as the actual set of parameters used by the module controller. Also on display here is the information for the actual installed firmware.

**!!! The backup/restore function only covers the Presets including the Parameters of the tab "Parameters" and the "GPI/O" Settings. Parameters of the tab "Setup" must be checked and configured manually. Parameters "High Pass 30Hz", "Low Pass 15kHz", and "Stereo Link" must be stored again within the Presets. !!!**

Besides this, the device page provides info about the hardware platform, the used parameter version and **firmware** versions of the **Controller** and the **DSP**.

## 4.2.4 SETTING > SETUP



### Level Magic

selects the main operating mode. The processing can be used as the well known Compressor / Expander / Limiter combination or as a Level Magic processor

### High Pass 30Hz

will activate a 30Hz high pass filter

### Low Pass 15kHz

will activate a 15kHz low pass filter

### Stereo Link

defines if the control loops of both channels are linked together for proper stereo operations

### Input

defines whether the **IN1 analog input** or the **IN2 digital input** is selected

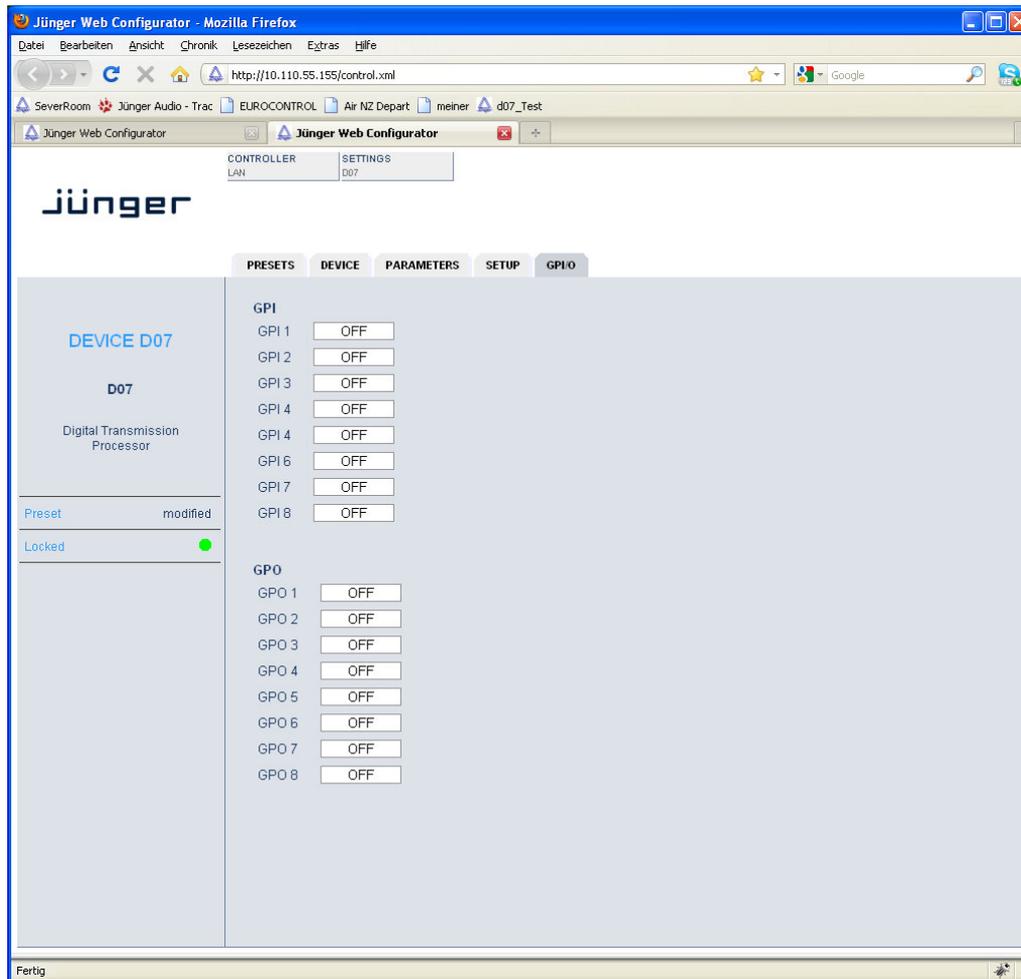
### Sync

defines the source for synchronizing the **d07** if it works in the digital domain

### Relay Bypass

The audio inputs and outputs have relay bypass function which will be engaged if power fails. One may also turn this bypass on by clicking into the check box

## 4.2.5 SETTING > GPI/O



**GPIs** are useful if you want to recall settings remotely e.g. by presets.  
The **d07** has **8 physical GPIs** (opto coupler) which can be assigned functions of the **d07**.

**GPOs** (Tallies) are useful if you want to monitor status information of the **d07**.  
The **d07** has **8 physical GPOs** (relays) which can be assigned conditions of the **d07**.

## 4.2.6 CONTROLLER > SYSTEM CONFIG

The screenshot shows the Jünger Web Configurator interface. The browser address bar displays `http://10.110.55.155/control.xml`. The page title is "Jünger". The main navigation bar includes "CONTROLLER LAN" and "SETTINGS D07". The "SYSTEM CONFIG" tab is active, with other tabs for "BACKUP / RESTORE", "SOFTWARE UPDATE", and "REBOOT CONTROLLER".

**Controller**

Device Controller

Image Version  
rel\_1\_9\_0b\_1he\_8463

**DEVICE**

Device Name: do7 Main Program  
Device Location: Rack 21  
System Contact: info@junger-audio.com

**PASSWORDS**

Password checking enabled

Change password for: operator

Password: [input]  
Repeat password: [input]

**NETWORK**

IP Address: 10.110.55.155  
Netmask: 255.255.0.0  
Gateway: 10.110.0.1

**METERING**

UDP Port Range Start: 49152  
UDP Port Range End: 65535

**SERVICES**

Maintenance Interface via RPC  
 Telnet Server

### DEVICE

you can assign the d07 a unique Device Name as well as a Device Location and a System Contact e-mail address for future applications

### PASSWORDS

if Password checking is set to enable a password for the administrator and for the operator can be assigned

### NETWORK

you can change the IP configuration of the d07 (see also chapter 6).

### METERING

in order to get UDP packets through a fire wall you can define UDP port address range here

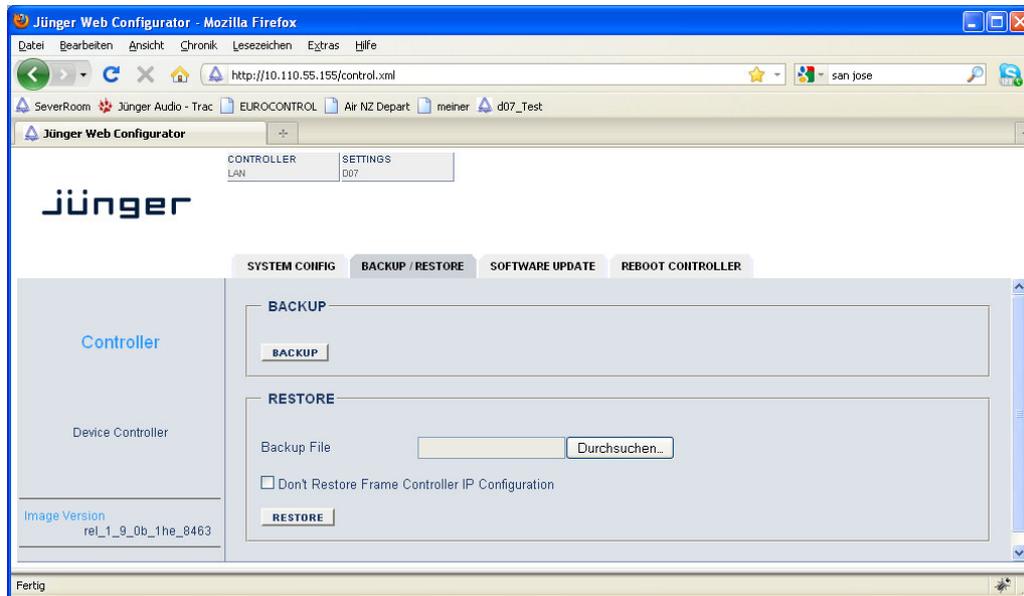
### SERVICES

Maintenance interface via RPC:  
**for internal use only**

#### Telnet Server:

if you want to reach the console interface of the LAN controller via telnet you must enable this feature here

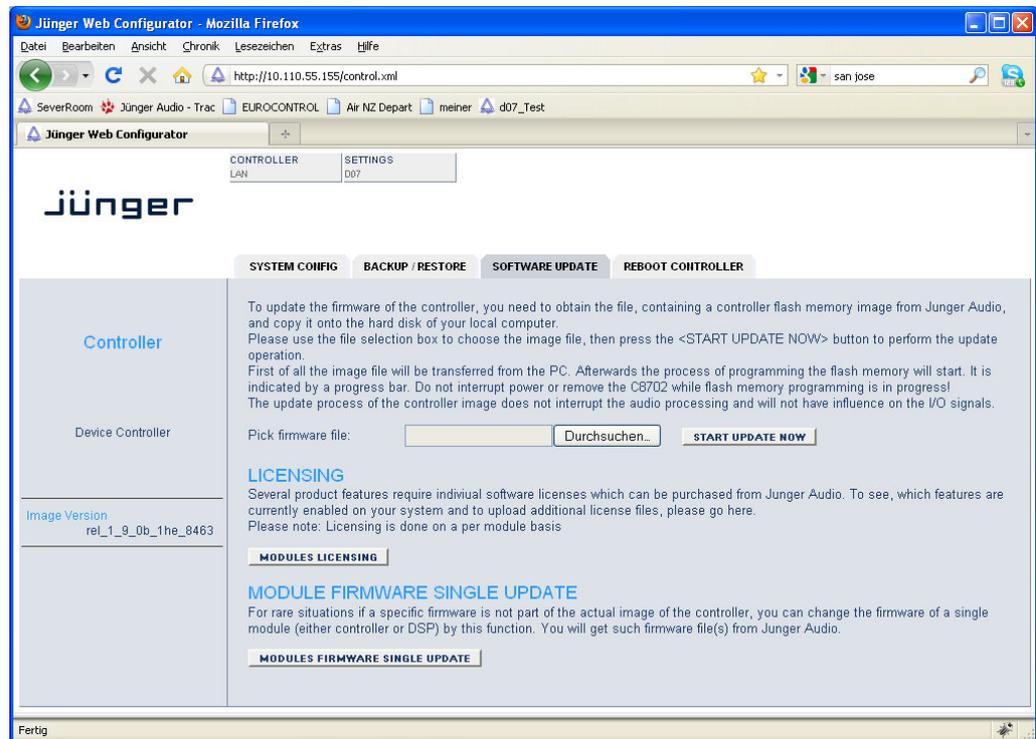
## 4.2.7 CONTROLLER > BACKUP / RESTORE



Here you can get a backup file of the d07. Simply press **<BACKUP>** and the network controller will gather all information and will present it as an XML file for download to your local PC. If you want to restore your d07 from a previous backup, select the file from your PC and press **<RESTORE>**. If you take the backup from a different d07 you must check: **"Don't Restore Frame Controller IP Configuration"** in order to keep the IP settings for the d07 you are about to restore.

**!!! The backup/restore function only covers the Presets including the Parameters of the tab "Parameters" and the "GPI/O" Settings. Parameters of the tab "Setup" must be checked and configured manually. Parameters "High Pass 30Hz", "Low Pass 15kHz", and "Stereo Link" must be stored again within the Presets. !!!**

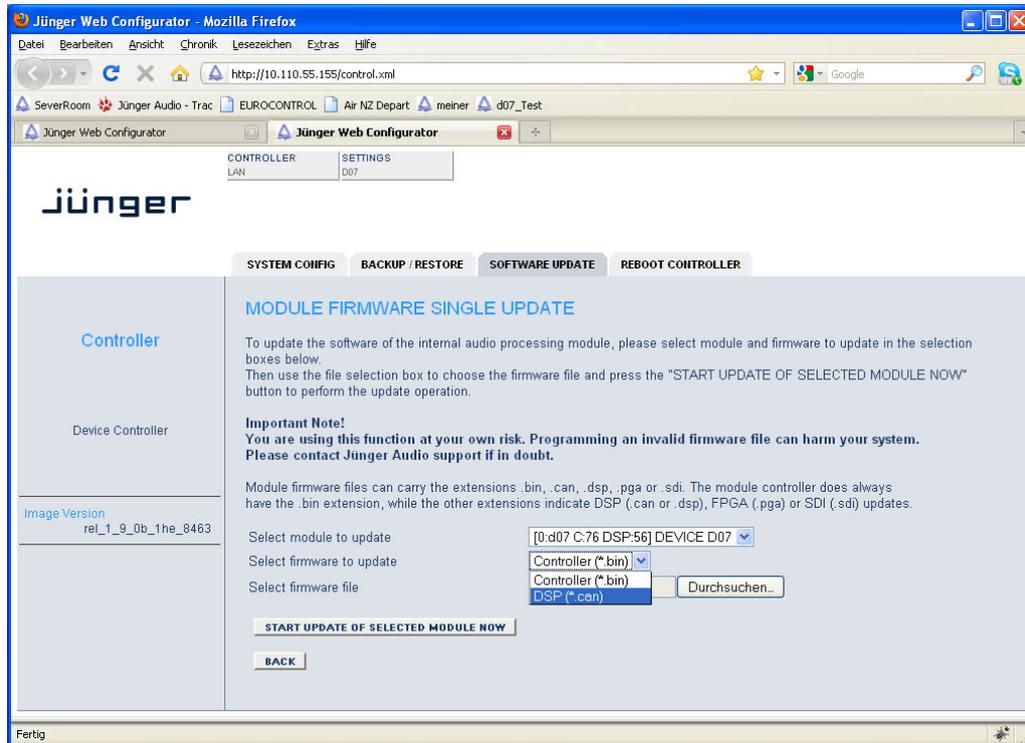
## 4.2.8 CONTROLLER > SOFTWARE UPDATE



In the top part of the above window you can update the firmware of the network controller. You must "Pick the firmware file" from your PC and press <START UPDATE NOW". The image file of the firmware will be moved to the network controller and will be burned into the program Flash memory afterwards.

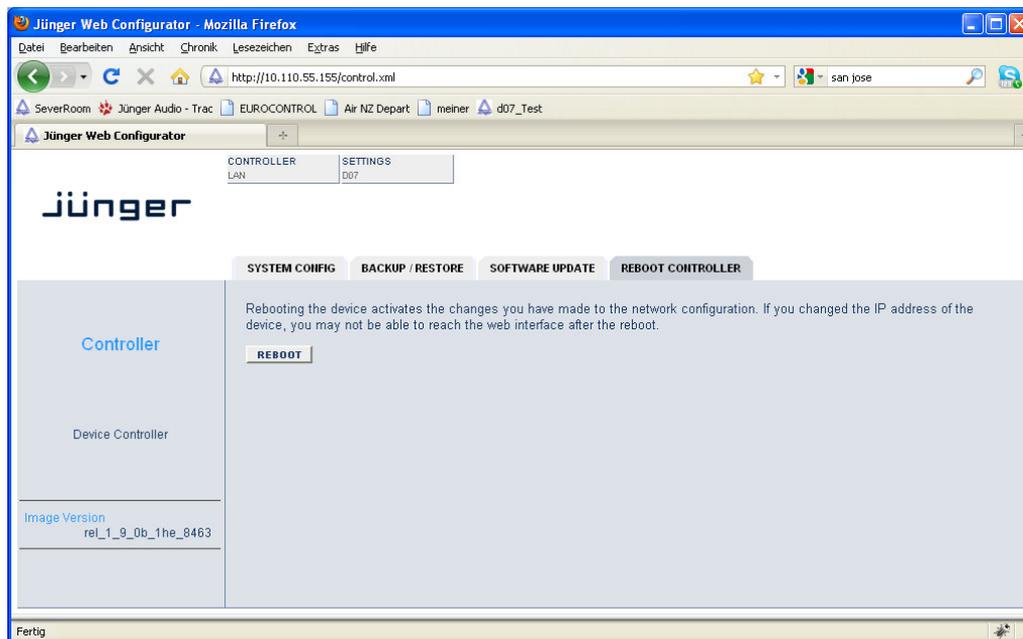
You can also update the DSP controller and the DSP itself by pressing

### <MODULES FIRMWARE SINGLE UPDATE>



You must select either Controller (\*.bin) or DSP (\*.can) and select the respective file from your PC. When done press <START UPDATE OF SELCTED MODULE NOW> And the firmware will be installed to your d07.

## 4.2.9 CONTROLLER > REBOOT CONTROLLER



Network controller can be rebooted.

# BOOT DISPLAY AND TROUBLE SHOOTING



display	meaning / explanation
JUENGER AUDIO D07	display of model
C: x.x D: x.x	display of loaded controller software version display of loaded dsp software version

display	error / message	remedies
SYNC ERROR!	no sync at sync input!	<ul style="list-style-type: none"><li>■ connect the sync input (selectable in SYNC field) with valid input signal</li><li>➤ INPUT: sync on DIGITAL IN CH 1/2</li><li>➤ EXT AES: sync on SYNC AES/EBU</li><li>➤ EXT VIDEO: sync on SYNC VIDEO</li><li>➤ EXT WCLK: sync on word clock</li></ul>

## 5.1 BOOT DISPLAY

## 5.2 ERROR MESSAGES AND TROUBLE SHOOTING

### 5.3 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 and registers important for the program are loaded with the factory setup and the program is restarted.

Any button (exceptional >>MENU, Peak up>>) is to be held pressed in order to initialize the device during power-on of the device until the program has started. To the start of the program and at the completion of the displays (how described in 6.1), 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!**

# TECHNICAL SPECIFICATIONS



sample rate 44.1/48 kHz  
audio data format 24 bit

**digital signal processing**

---

## DIGITAL IN/OUT

### AES/EBU

connector XLR, 110 balanced  
input format AES professional, AES consumer  
output format same as input format

**digital in- / outputs**

channel status bits:

digital input -> digital output transparent  
analog input -> digital output fixed channel status bits  
(professional/48kHz sample frequency/2ch mode/24 bit audio)

## ANALOG IN/OUT

### ANALOG IN

Resolution 24bit  
sample rate 44.1...48kHz  
dynamic range 110dB (RMS)  
114dB (A-weighted)  
THD+N <0.002% @ max. input level  
frequency response 20Hz...20kHz (FS=48kHz) (+/-0.5dB)  
CMRR -100dB @ 50Hz  
max. input level +22dBu @ 0dBFS  
input impedance 10 kOhm, floating balanced  
connector XLR, 1-screen, 2-live, 3-return

**analog in- / outputs**

### ANALOG OUT

Resolution 24bit  
sample rate 44.1...48kHz  
dynamic range 108dB (RMS)  
110dB (A-weighted)  
THD+N <0.002% @ max. input level  
frequency response 20Hz...20kHz (FS=48kHz) (+/-0.5dB)  
max. output level +22dBu @ 0dBFS  
output impedance 30 Ohm, floating balanced  
connector XLR, 1-screen, 2-live, 3-return

**sync  
in- / outputs**

**SYNC IN**

WCLK	connector	BNC, 75Ohm, coaxial
	level	TTL-level
	input format	Wordclock
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

**WCLK OUT**

WCLK	connector	BNC, 10kOhm, coaxial
	level	TTL-level
	output format	Wordclock

**remote control**

**REMOTE**

serial remote interface	RS-232 in/out
connector	9 pin SUB-D female
serial remote interface	RS-422
connector	9 pin SUB-D male, optional TCP/IP

GPI parallel remote	level	opto coupler, 3..24V control voltage
	connector	15 pin SUB-D female
Tally Out	level	relais contact
	connector	25 pin SUB-D female

**general**

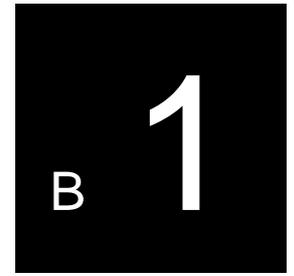
power consumption	appr. 15 VA
dimensions	19", 1 RU, 250 mm depth
weight	appr. 5 kg

## **B – APPLICATION NOTES**

- 1. The Junger Audio Dynamics Processor Principle**
- 2. The Junger Audio Compressor & Expander Principle**
- 3. The Junger Audio Processing Presets**
- 4. LEVEL MAGIC**
- 5. GUI**
- 6. Network Configuration for TCP/IP Operation**
- 7. FM-Transmission (d07)**
  - 7.1. General**
  - 7.2. MPX-Limiter**
  - 7.3. Pre-emphasis**
  - 7.4. terms and definitions**



# THE JUNGER AUDIO DYNAMICS PROCESSOR PRINCIPLES



Changing the dynamic range of an audio signal is inherently a non-linear process. Unlike an ordinary line amp, the gain of a dynamic range processor is not constant – it varies with time depending on the specific control algorithm of the dynamics processor and the changing amplitude of the input signal. These variations in the gain, which represent the real control process, should take place without any bothersome side effects to the audio signal itself, effects such as pumping, signal distortion, sound coloration, or noise modulation. In other words, they should be inaudible.

The setting of the attack time parameter of a dynamics element effects how the unit will react to rapid amplitude changes in the audio signal. A long attack time leads to overshoots (and consequent distortion) because the system is not fast enough to reduce the gain. A short attack time minimizes the chance of overshoots, but the more rapid gain changes in such cases have audible side effects such as "clicks" and other modulation artifacts.

## Traditional Compressor and Limiter Designs

Traditional compressor and limiter designs only have one control circuit with one attack time and one release time. They must be adjusted manually by the user to optimal settings for processing with as little disturbance as possible through a process of trial and error. A lot of experience and a lot of time is necessary to get acceptable results. These settings, once found, are only the right choice for a certain program signal and must be changed for other program types.

## Multi-band designs

These units split the audio frequency spectrum into several frequency bands. The attack and release times are set independently for each frequency band, giving independent processing for each band. The problem with this multi-band approach comes when the outputs of each band's processor are combined together to produce the output audio. The spectral balance of this output signal is always different from the input. The balance of high, mid, and low frequencies is inherently disrupted, which is particularly objectionable when the signals are music, as in commercials, concerts, etc.

## Multi-Loop designs

The Junger Audio Dynamics Processors work according to a [Multi-loop principle](#). The various loops each work over the entire frequency spectrum. They work in parallel, each with a different set of attack and release parameters. Each loop develops a control signal which is then summed with the controls from the other loops to produce a single gain control signal applied to one gain control element. Please see the figure below.

### Look Ahead/Signal Delay

The digital implementation of the [Junger Multi-loop design](#) also permits a very short time delay (approx. 2ms) to be introduced in the audio signal path. It lets the gain changing elements “look ahead” and determine the correction needed. This is applied to the delayed signal just in time to control even the fastest transients. That is particularly important for the limiter, which provides a precisely levelled output signal absolutely free of overshoots (clipping).

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.

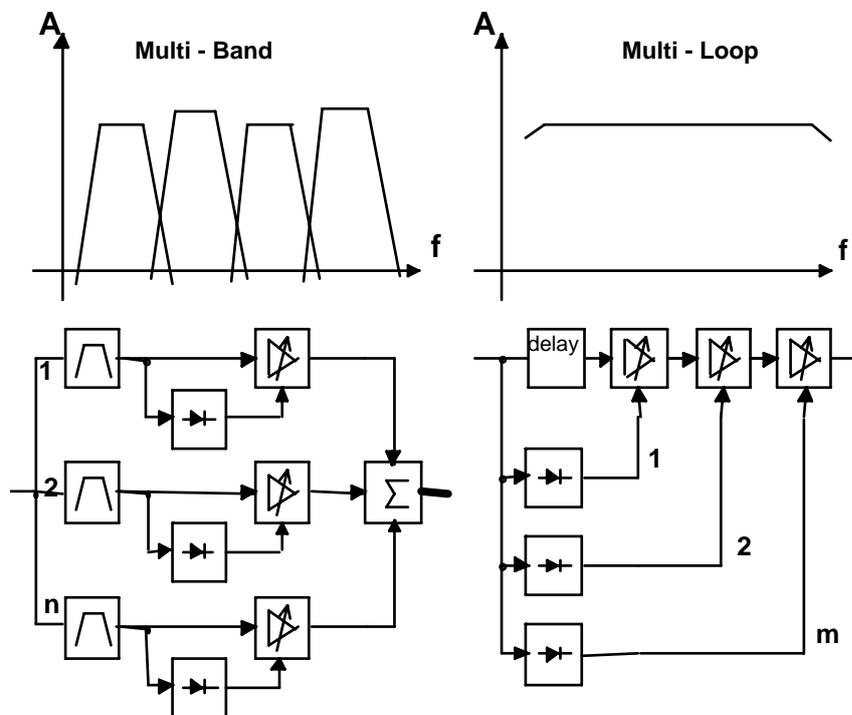
### Adaptive Dynamic Range Control

The proprietary algorithms in the Junger System also allow the automatic adjustment of the attack and release times according to the evolution of the input signal over time. This is called Adaptive Dynamic Range Control. By monitoring the waveform of the incoming audio, the System can set relatively long attack times during steady-state signal conditions but very short attack times when there are impulsive transients.

### The Best Performance

The dynamic range processor principles developed by Junger Audio make it possible to realize dynamics processors (compressor, limiter, expander) with very high audio quality, without signal coloration, pumping or breathing, and without distortion and modulation products.

In short, they offer the best possible performance – inaudible dynamics control.



# THE JÜNGER AUDIO COMPRESSOR & EXPANDER PRINCIPLE



Compression is defined as the reduction of the dynamic range of the input signal to match the dynamic range of the storage system, transmission system, and/or the listening environment. Typical approaches to this task often result in audible artifacts and lack luster performance. The Jünger approach takes an atypical approach that avoids these difficulties.

In the Jünger system, compression of the program signal takes place over the entire input level range, not just the upper end above a certain threshold level.

Compression is partly achieved by increasing the level of low level signals. The lower the input signal level, the higher the additional gain applied to that input signal by the compressor. As the level of the input signal rises, the amount of gain applied is reduced. Please see the figure to the right.

Dynamic structures of the entire range of the input signal amplitude are converted proportionally. Even after compression the dynamics of incoming audio are maintained, only slightly condensed, leaving a transparent, seemingly uncompressed sound impression.

The gain of the compressor (called 'range') can be limited from 1 dB to 15dB **to prevent unacceptable increase in back grounds during signal pauses (e.g. ambience).**

To help eliminate unwanted very low level noise (air-conditioning, hum, and noise), a Jünger Expander can be used. Below an adjustable threshold level, the expander will attenuate the level of the incoming signal: as the incoming signal drops so low that it is 'in the noise' the Expander reduces the signal level further to 'hide' the noise.

The Range parameter sets the maximum attenuation applied when the incoming signal level is within 6dB below the Threshold level.

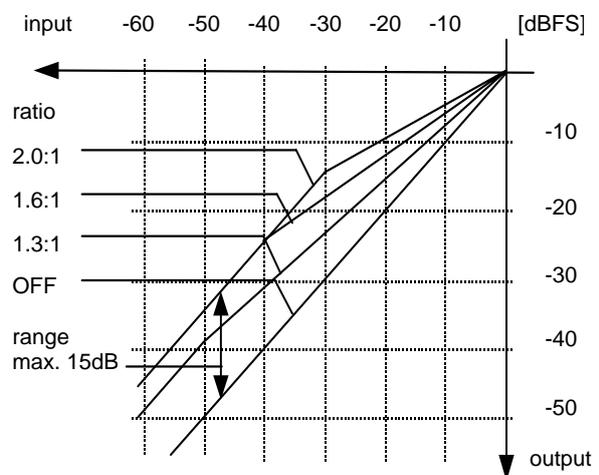


Figure 1: Compressor characteristics

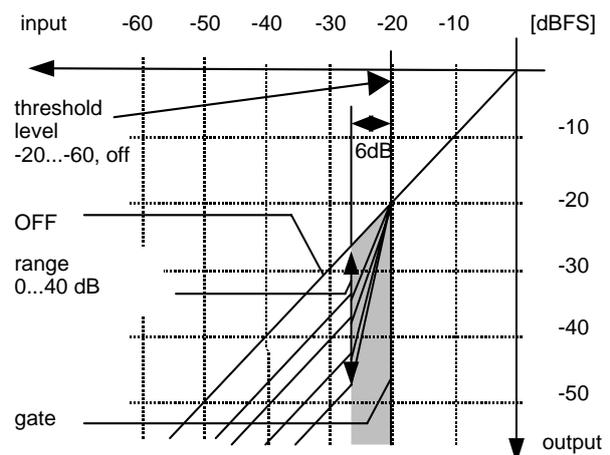
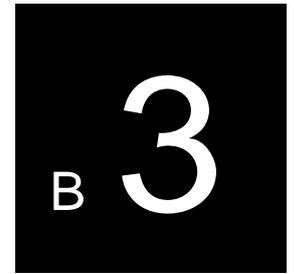


Figure 2 Expander characteristics



# THE JUNGER AUDIO DYNAMICS PROCESSING PRESETS



For some of the control parameter it is possible to define a limited range of time constant values which are 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.

Parameter related to the transient response of the control circuit are important for distortion-free processing. These time constants are always adaptive controlled without remarkable limitation of parameter range. This is caused by the presence of transient pulses in almost each kind of program material. The algorithm has to guarantee best reaction for fast increasing level of transient signals anytime even if classical music with slow dieing 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.

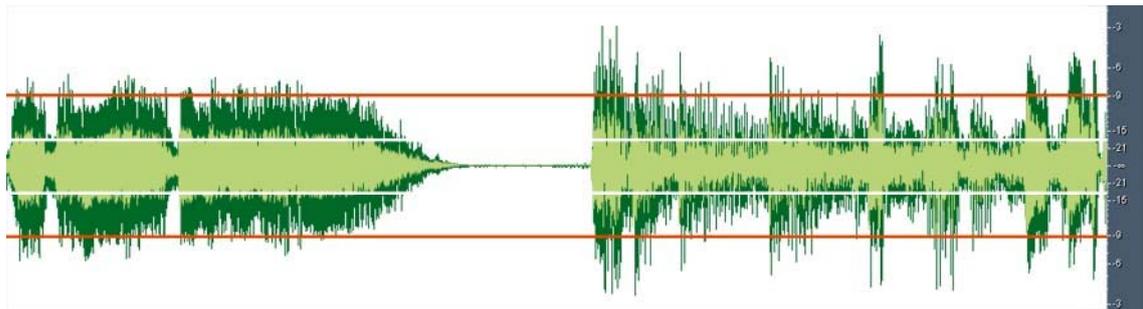
The selection of the parameter PROCESS/PROGRAMM (limiter, compressor) changes the range of time constant values as follows:

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	



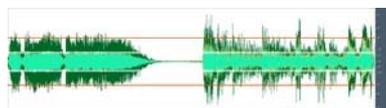


# LEVEL MAGIC™



## Introduction & Reference guide

1. Release 2005  
Junger Audio  
Berlin



## LEVEL MAGIC™

a sophisticated new adaptive level control algorithm capable of adjusting the right audio level from any source at any time.

Program suppliers and broadcasters alike have long been plagued by ‘surprise’ level changes when switching from one source to another. Not only peak levels but also average operating levels can vary wildly from one source to another, wreaking havoc with unattended operation.

Level Magic™ from Junger Audio relies on a sophisticated new adaptive level control algorithm capable of adjusting the right audio level from any source at any time. Automated Gain Control + Transient Processing + Peak Limiting for continuous unattended control of any program material.

### > The audio signal is levelled to the desired Operating Level instantly!

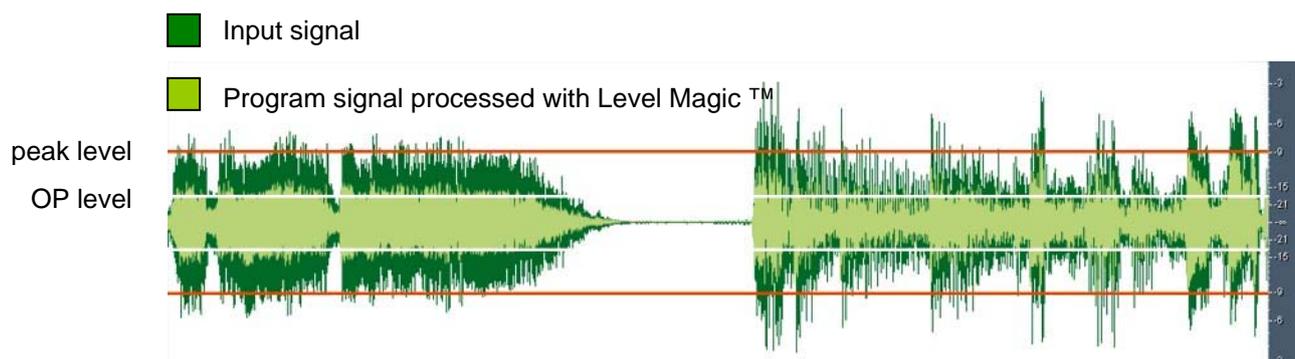
With Level Magic™, the desired Operating Level and Peak Level are dialled in once and thereafter, Level Magic™ will give continuous control, regardless of the source -- without touching the sound of the audio material. No breathing, no pumping, no spectral changes. Just well controlled dynamics!

### > Unpleasant level jumps are eliminated

Level changes from different feeds, level differences between different program parts or even loudness problems in broadcasting – Level Magic™ will take care of them automatically, with a result the Listener will want to hear. Major application fields include playout for multichannel broadcasting for satellite and cable distribution, program transfers with audio level changes, ingest stations and any situation where continuous control of audio level is important.

### > Overmodulation is prevented by a Brickwall-Limiter

The Junger Audio brickwall limiter guarantees precise peak limiting without any distortion. For any kind of program signal and anytime.



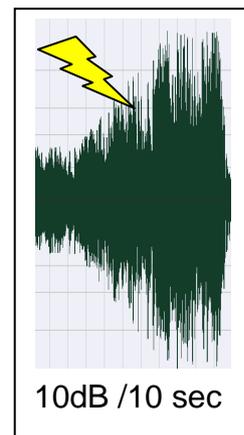
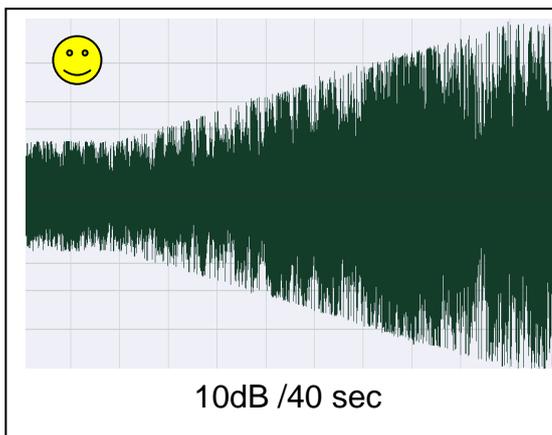
To understand the principle of the new algorithm and the adjustment of **LEVEL MAGIC** it is necessary to keep some psychoacoustic aspects in mind

1. We do not perceive level changes if they happen in a certain period of time dependent on the absolute value of the level change.

That means any slow level changes are not perceptable by the human ear.

If for example the audio level rises from -20dBFS to -10dBFS within one minute you won't realize it, unless the level gets over an bearable value or the audio masks other sources you would like to listen to.

But if the same level change happens in 10 seconds it will be very noticeable!

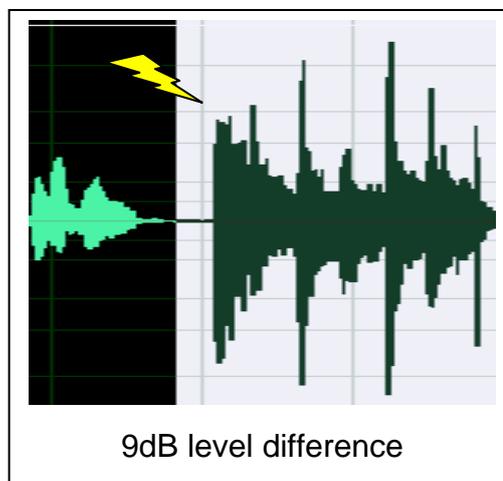
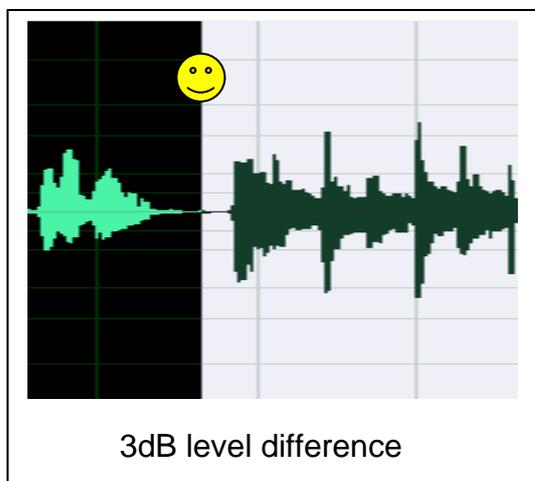
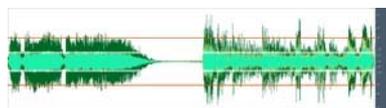


That explains very clear why it is most important that **AGC** may not work too fast (1dB/4-5sec)!! A fast acting AGC would cause perceptable level changes. But we are looking to get an mostly inaudible levelling procedure.

If a fast level adjustment is required (because of transients), this is done by the **Transient Processor**.

2. Level jumps rising over a certain absolute value are very unpleasant for our ears.

Of course, it depends on the type of audio material and consequently on its loudness which absolute value of level change really annoys. A jumping level of 6dB is remarkable. A quick level change of 10...12dB becomes annoying for the human ear! So it's necessary to avoid major level changes. The transient processor of Level Magic is a solution for that.



The [transient processor](#) immediately reduces or raises the level of a new program part so that level jumps over 10...12dB are eliminated.

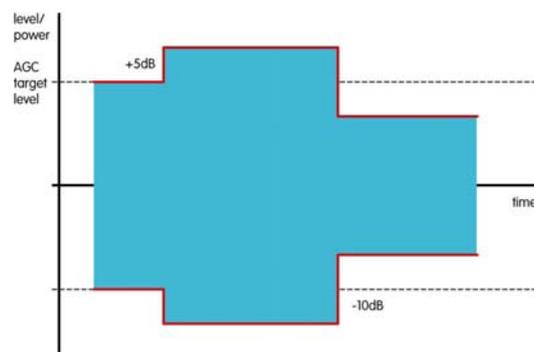
What makes the [Level Magic](#) different from previous dynamics processors?

A compressor/limiter combination of a known dynamics processor by Junger Audio is always controlling the audio level in relation to the limiter threshold. In result no headroom is more existing and the signal is developed to reach a 100% output level. This characteristic is useful to reach maximum levelling for audio disk mastering as well as to reach 100% modulation for FM transmitters.

In compare to that Level Magic™ is serving two different levels – operating level and peak level. Between operating and peak level we will find the so called “headroom” for peaks that are still coming with the audio signal, even if this is level controlled related to the operating level. Level Magic™ is a unique algorithm to make automated audio level control possible. It is a combination of an adaptive AGC (automated gain control) with a transient processor and a brickwall limiter. The combination of an AGC circuit with a transient processor is the key to get a satisfying output level control for any kind of input level changes.

### Input level change

The picture is showing a theoretical level change of +5dB and -5dB around operating level.



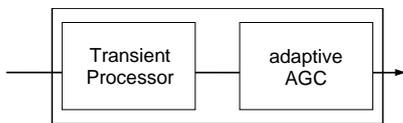
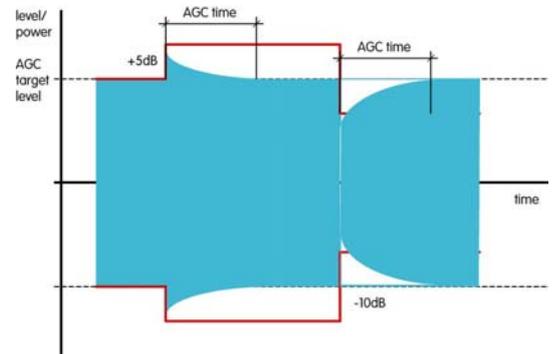
## d06 d07 b46 C8007 C8046

## Introduction & Reference guide



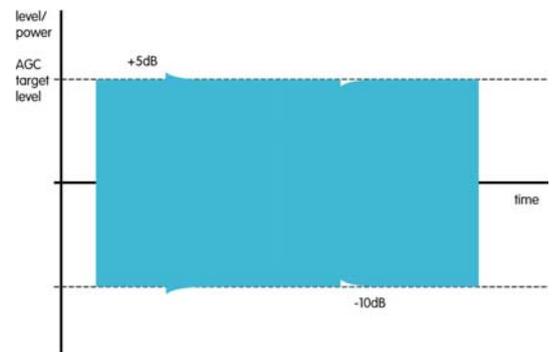
### Working with AGC

In this picture a conventional AGC is used to adjust the output level. As we know the AGC must work slow to perform a mostly inaudible gain change. In result control on the output level is not giving a proper correction of the input level change.



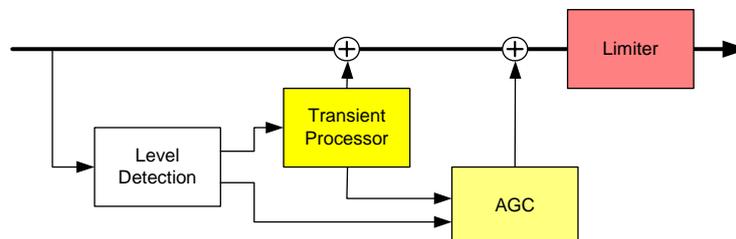
### Level Magic™

Level Magic™ is a unique combination of a transient processor and an adaptive AGC process. The transient processor can fill the lack of fast level control left by 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.

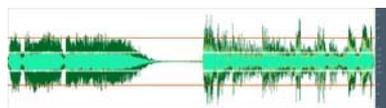


### Block Diagram

Level Magic™ is consisting of adaptive AGC + Transient Processor + Brickwall limiter. Independent on the leveller circuits the brickwall limiter is taking care on the peak level. For the leveller (AGC + Transient Processor) Jünger Audio 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 (fast process) and AGC (slow process).



The characteristic of the Level Magic™ level control is mostly determined by the settings of the Transient Processor. Transient processor is doing fast gain change and the AGC is doing slow gain change (depending on settings). Always 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.



Parameters of **AGC**  
**Transient processor**  
**Limiter** & brief description

Parameter	Range	Description
<b>LEVELLER</b>		
<b>Operating Level</b>	-40...0dBFS	Desired <b>target level</b> for the levelling process. Reference Level for the Transient Processor and the AGC
<b>AGC range</b>	0...40dB	Determines the <b>maximum gain change</b> applied by the <b>AGC</b> . AGC Range must be bigger then the expected difference between the average input level and the operating level. If there is for example an average input level of -23dBFS and your OP-Level is -18dBFS, the AGC needs at least a range of 5dB. In most cases an AGC range of 10dB is a good choice
<b>AGC time</b>	10s...2h	Describes the <b>time of development</b> for the AGC to reach the maximum possible gain change (range value). <b>The ratio of gain change should never be faster then 3 seconds for 1 dB!!</b> We are recommending a setting of 4...5 seconds for 1dB gain change by the AGC. Therefore the AGC time is basically determined by the AGC range value. A range setting of 10 dB requires a time setting of minimum 40 seconds.
<b>AGC gate</b>	-60... -20dBFS	If the input level falls below this threshold level, the gain change of the AGC freezes immediately. Transient processor is still active. After appr. 20 seconds input level below silence gate the current gain change is slowly moving to the longterm average gain.
<b>Transient program</b>	soft/mid/hard	This parameter describes the characteristic of gain change by the transient processor. It has to be chosen dependent on your program genre. If there are just a few level changes or you want to keep the original dynamic range best (e.g. classical music), you have to choose "soft". For mixed program "mid" should be best in most cases. And for live venues (sport etc.) with frequent unexpected level changes the adjustment "hard" is required.
<b>Transient range</b>	0...15dB	Determines the <b>maximum gain change</b> applied by the <b>Transient Processor</b> when there are fast input level changes. Large range values are reducing the dynamic range, especially in combination with the transient program "hard"
<b>LIMITER</b>		
<b>Limiter Threshold (Peak Level)</b>	0...-20dBFS	<b>Reference Level</b> for the <b>Brickwall</b> Limiter. The range between the Operating Level and the Peak Level is the level headroom and should be 6...9dB.
<b>Limiter program</b>	0...9	Characteristic of the limiter, mostly reflecting release of the limiter reduction. 0 – very fast, 9 – very slow.

### Quick Start with Level Magic

For the first use of the Level Magic™ unit it's advisable to start with one of the factory presets (6 available). Some individually needed changes in the settings can be saved later in one of the 4 user presets.

- Select the preset meeting the application you are looking for mostly.
- Check if operating level and peak level are meeting your standard. If this is not the case readjust them and save your settings in one of the available user presets.
- If after having worked with different presets you think the desired setting is between two factory presets, compare them in the following table and look for the differences.

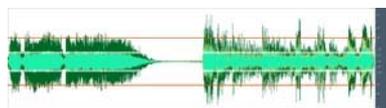
Always take in mind that the balance of both levelling processes is determined the audio performance of the box mostly! As more available maximum gain by the transient processor as more boosting the level control is. As less available maximum gain by the transient processor as more sensible the level control will be applied.

Because of the use of adaptive controlled processing algorithms and considering the fact, that the AGC setting must meet the slow gain change requirement, just a few variations are left. Mostly changeable parameters to play with are Transient Processor Range in accordance with Transient Processor Program. The recommendation is:

Description of the processing result	Smooth levelling, preserving dramatic content	Normal standard level control	Boosting level control, decrease of dynamic range
Content application	Movie Sound, Classical Music	Any kind of audio material	Live audience, Speech dominated program
Transient Program	Soft	Mid	Hard
Transient Range	3...5	6...8	9...12
Limiter Program	6...8	3...5	1...2

Level Magic™ is creating the level headroom between the operating level and the peak level. For almost any audio material used for broadcast transmission the headroom should be 6...9dB.

With this rule it should be easy to find the settings for the limiter. Even if the operating level is -20dBFS and therefore a technical headroom of 20dB is available it doesn't make sense to use it. More than 10dB headroom are increasing the dynamic range of the audio material for broadcast transmission too much.

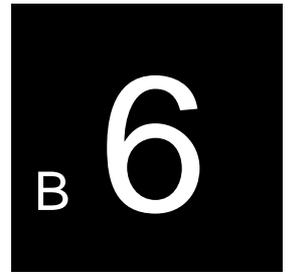


Overview on the **Factory Presets**

Parameter	Factory presets					
	Radio Classical	TV Movie	TV Live	Radio Speech	Radio universal	TV universal
LEVELLER	ON	ON	ON	ON	ON	ON
<b>OP-level</b>	<b>-9dBFS</b>	<b>-18dBFS</b>	<b>-18dBFS</b>	<b>-9dBFS</b>	<b>-9dBFS</b>	<b>-18dBFS</b>
<b>AGC Range</b>	10dB	15dB	10dB	10dB	10dB	10dB
<b>AGC Gate</b>	-60dBFS	-50dBFS	-50dBFS	-40dBFS	-50dBFS	-50dBFS
<b>AGC Time</b>	2min	2min	20s	20s	40s	40s
<b>Transient Program</b>	Soft	Mid	Hard	Hard	Mid	Mid
<b>Transient Range</b>	3dB	6dB	10dB	15dB	10dB	10dB
LIMITER	ON	ON	ON	ON	ON	ON
<b>LIMITER Threshold</b>	0dBFS	-9dBFS	-9dBFS	0dBFS	0dBFS	-9dBFS
<b>LIMITER Program</b>	6	4	1	2	4	4
Max. total gain change	13dB	21dB	20dB	25dB	20dB	20dB

# NETWORK INTEGRATION

## of Jünger Audio devices



d07  
d06  
**Level Magic LT**  
**C8000-modules via C8702 LAN Controller**

To operate the Junger audio devices via web browser you have get an Ethernet connection between the DEVICE and the PC. If you are not familiar with the network setup, please consult an administrative person for assistance and read this chapter carefully!

There are two ways to communicate with the device via Ethernet:

1. You can connect the device to the **LAN** your PC is part of (if there is one existing already)
2. You can connect the device directly to your PC using an **Ethernet crossover cable**.

**In both cases network settings of the device or your PC or even both have to be changed and matched.**

The default network configuration of the Jünger devices is:

IP Address: printed on a label on top of the Ethernet connector at the rear of the device  
Subnetmask: 255.255.0.0.  
Gateway: 10.110.0.1.

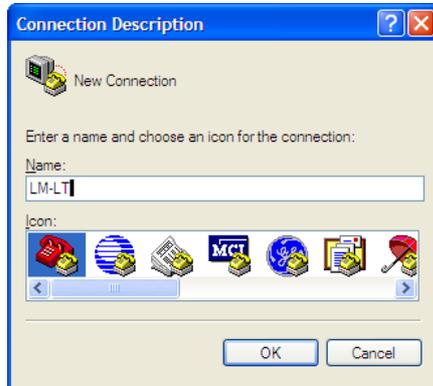
### 1. Integration into existing LAN

When you want to integrate the device into an existing LAN you have to change its IP-address, the subnet mask and the gateway address respectively. You will get valid settings from your network administrator.

You can do that two ways:

- A)** Connecting the device over a **serial cable** to your PC and change the network configuration with a terminal program (e.g. HyperTerminal included in Windows installation).
- B)** Disconnect your PC from your LAN, match your PC's network setup to the IP settings of the device and connect the PC via **Ethernet crossover cable**, change the device's network configuration. Then change your PC's configuration back again and connect both PC and the device to the LAN.

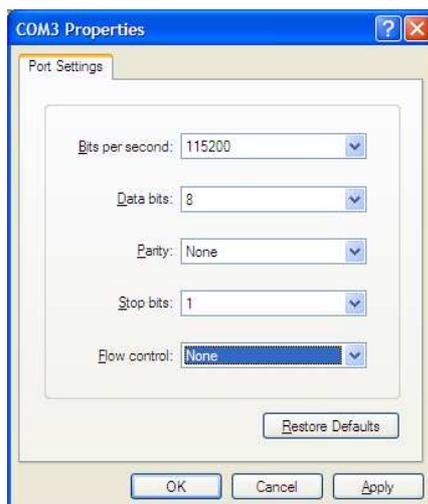
- A) Connect the device over a 9 pin serial cable (connected 1 to 1) to your PC.  
Start your terminal program (e.g. Start -> Programs -> Accessories -> Communications -> HyperTerminal).



Enter a name of your choice and press OK

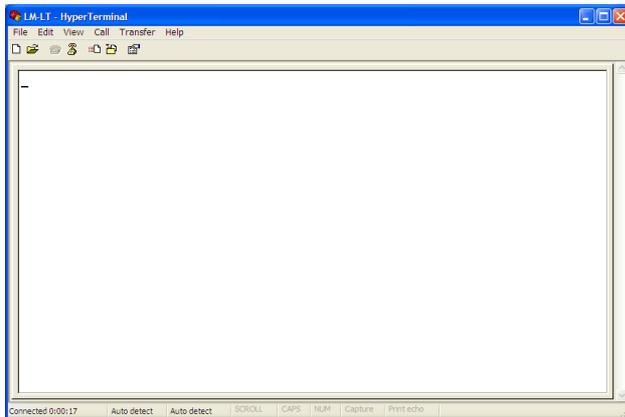


Choose the connection port you are working with and press OK

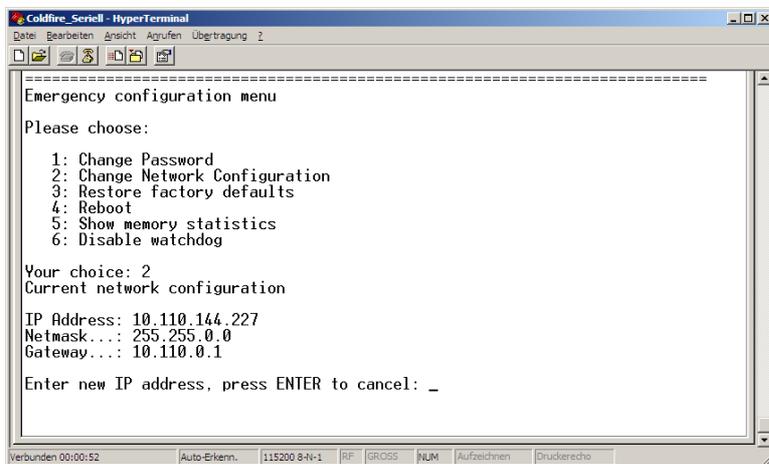


Set the COM settings as they are shown in the window above and press OK.

You will get to the Hyper terminal window:



Press <ENTER> and you will get the following information of the device:

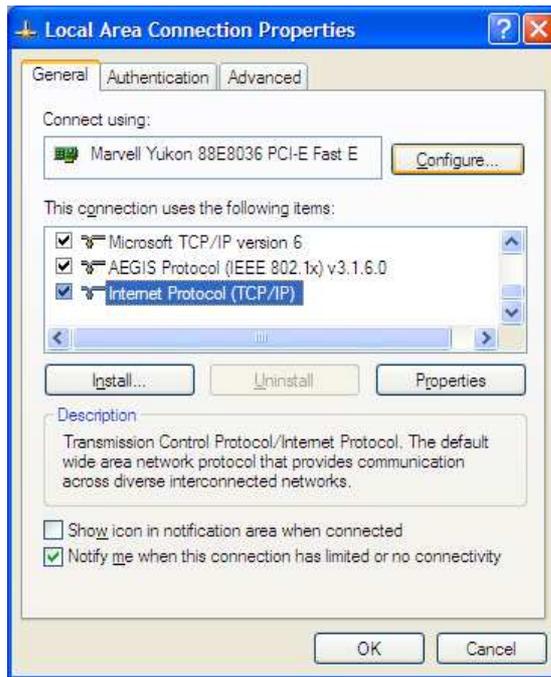


Now you can change the network configuration so that it fits into your LAN.

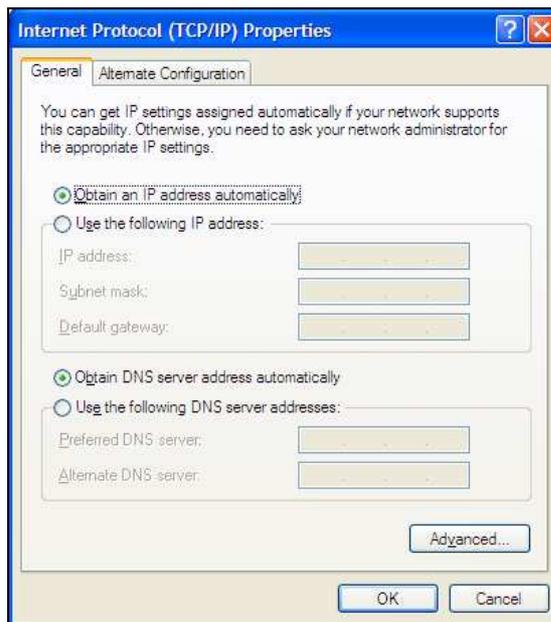
You might have changed the IP-address of the device, so please renew the label at the rear of the device, otherwise it will cause confusion.

When you ever initialize the device the default IP-address and network configuration will be active again. In case of need you can read the default IP-address always on the controller inside the device!

- B)** You can also change the IP address of the device over Ethernet connection. Disconnect your PC from the LAN, connect it to the device directly via **Ethernet crossover cable**. Change the network configuration of your PC (write down the current settings, you need them later to reconnect to your LAN!) via "Local Area Connection Properties" (Windows: Start -> Settings -> Network connections -> Local Area Connections)



Scroll in the list and choose Internet Protocol (TCP/IP).  
Make sure that the 'check box' for this item is checked, and then click on PROPERTIES.



In this example, the Ethernet TCP/IP is set to 'Obtain an IP address automatically.'

If, in your case, it is set to  
'Use the following IP address,'  
**jot down the current settings on a piece of paper**  
(IP address, Subnet Mask, and Default gateway, if used).  
You will need them later to restore the IP address of the PC to what it is required  
to work on your LAN.

Then change the settings in order to be able to communicate with the device. You **must** choose an IP-address different from that of the device.

Example:

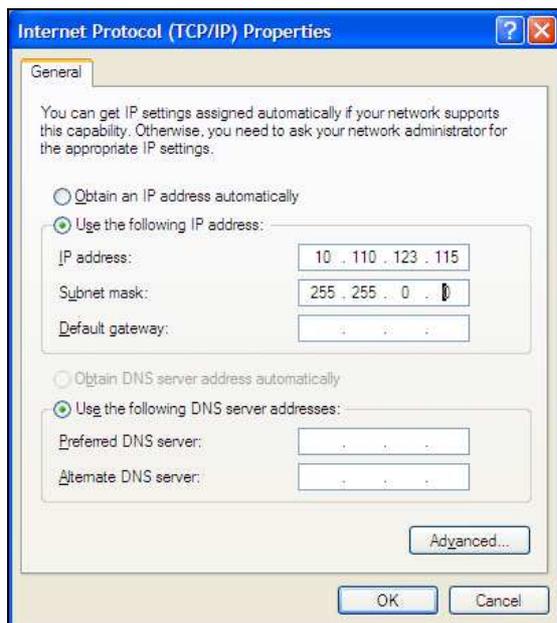
E.g. if the default address of the device is :

IP Address: 10.110.123.114

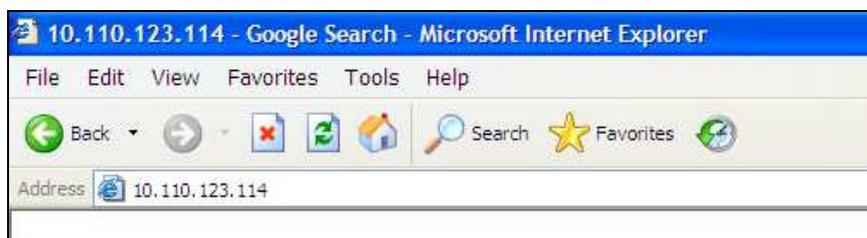
Netmask: 255.255.0.0.

Gateway: 10.110.0.1.

You could take **10.110.123.115** as the IP-address (a number close to but not the same as the device's address) and the same subnetmask. The gateway is not important when you are using an Ethernet crossover cable.



When you have changed the settings press OK. Now you will be able to communicate from your PC to the device by a web browser (IE, FireFox) via the Ethernet crossover cable. Just type in the device's IP-address into your browser:



Then you will enter the device's web pages:

By clicking on CONTROLLER > SYSTEM CONFIGURATION you will be able to change the device's network configuration according to the settings of the LAN you want to use.



The screenshot shows a web interface for network configuration. It has a title 'NETWORK' and three input fields: 'IP Address' with the value '10.110.55.155', 'Netmask' with '255.255.0.0', and 'Gateway' with '10.110.0.1'. Below these fields is a button labeled 'CHANGE NETWORK CONFIGURATION'.

After having changed the settings click CHANGE NETWORK CONFIGURATION and after that reboot the controller (CONTROLLER > REBOOT CONTROLLER).

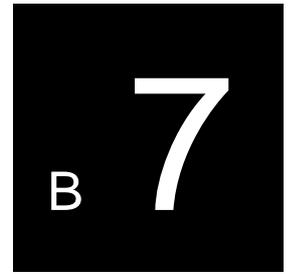
**Important note! Rebooting the device activates the changes you have made to the network configuration. If you have changed the IP address of the device, you may not be able to reach the web interface after the reboot.**

Now you have to change the settings of your PC network configuration again and connect both the PC and the device to the LAN you want to use. Then you will be able to communicate with the device over web browser via the chosen IP-address.

## 2. Using a crossover cable

You simply must connect your PC to the device by an Ethernet crossover cable and set the PC's IP settings according to the default network configuration of the device. I.e. you must give it an IP address and a subnetmask that matches the device IP setup (see above). A gateway address is not necessary because there is no gateway for cross over cable interconnection.

# FM-TRANSMISSION – d07



There are two important parameters for FM-Transmission

- The frequency deviation, determined by the peak level of the signal
- The MPX-Power, determined by the energy of the signal

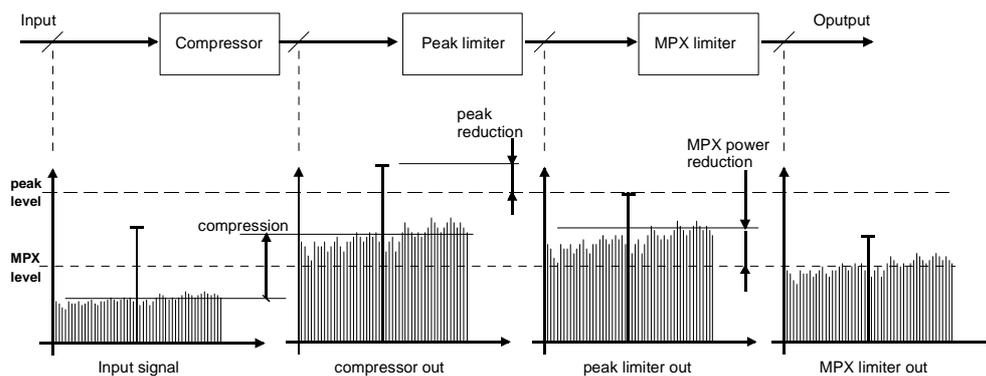
International standards regulate the maximum values that have to be kept tight so that neighbouring transmitters will not be disturbed.

- Frequency deviation  $\pm 75\text{kHz}$
- MPX-Power 0dBr

The job of the d07 processor is standard compliance controlling of FM signal energy within permissible peak deviation.

The d07 achieves this by the knowledge of the mathematical relationship between the MPX power, FM modulation and pre-emphasis. I.e. it is not necessary to feed the d07 with an stereo multiplex signal.

The compression of the program signal causes an increase of the signal energy and, therefore, more loudness, but also more modulation power (MPX power). Too much compression will cause the permissible value of the multiplex power to be exceeded and the MPX limiter must reduce the total signal. As a consequence, the peak levels as well as the average levels are reduced, leading to a reduced loudness, as outlined below:



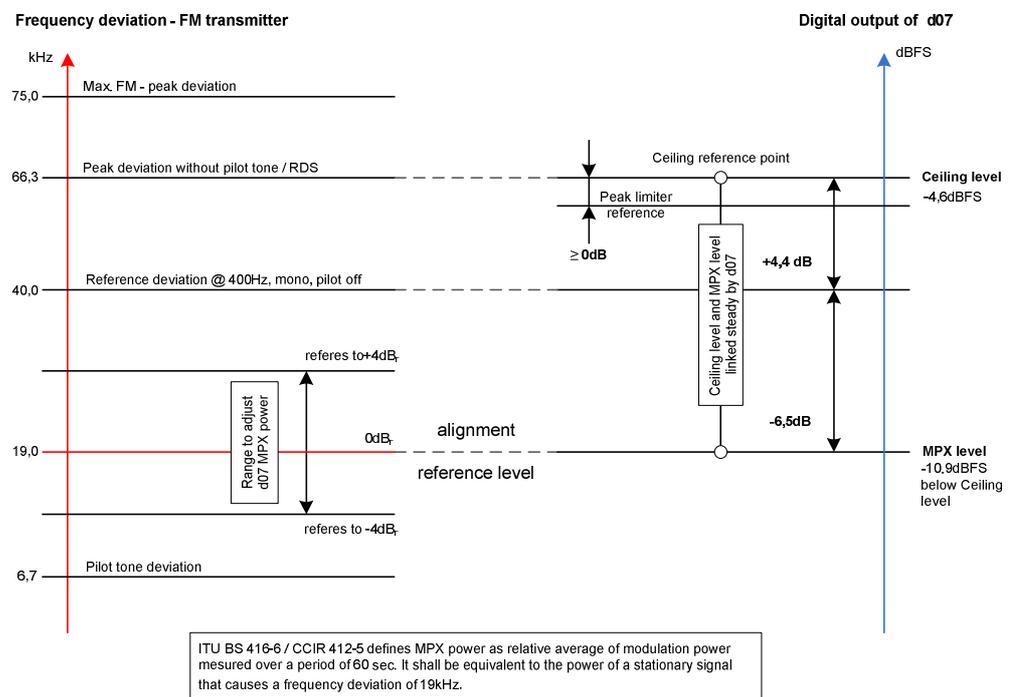
It is necessary to break up this “vicious circle” by optimal setting of GAIN and compression. The optimization of parameters should be done in a way that neither the peak limiter nor the MPX-limiter permanently show GAIN-REDUCTION. The program signal limitation should only occur briefly in order not to deprive it of all dynamic properties. The average modulation power can only be optimized by tweaking linear GAIN and the COMPRESSOR as well.

## 7.1. General

## 7.2. MPX-Limiter

## Level diagramm

For alignment of the d07 it is important to consider the level diagram of the overall transmission chain (see below):



## Peak Level

The ceiling level determines the maximum output level of the d07. This value is the most important setting parameter. It serves at the same time as the reference point for the MPX limiter and the adaptive pre-emphasis.

An MPX level of 0dB<sub>r</sub> lies 10.9dB below the ceiling level of the d07 and is fixed to it. This value is given by the relation of multiplex power level of 0dB<sub>r</sub> (19kHz frequency deviation) to peak level (75kHz frequency deviation). For special applications the MPX level can be changed by +/- 4dB respectively.

## Operating Level

The operating level in standard studio environment is not the peak level but the nominal level of +6dBu e.g. (analog) or -9dBFS (digital). Keep in mind the integration time of level meters when discussing operating level vs. peak level for program signals. The operating level will also be used to line up the FM transmitter by feeding it a 400Hz mono signal. This shall cause a nominal modulation deviation of 40kHz (Pilot tone and RDS carrier turned off). The headroom of 4.4dB left to the peak level (peak deviation respectively) can be exploited for temporary signal peaks or for the pre-emphasis of higher signal frequencies.

## MPX-Limiter set-up hints

With a given nominal level the threshold of the peak limiter may not exceed this nominal level by more than +4.4dB because otherwise the peak frequency deviation level will be exceeded. Therefore the transmission path must be aligned in a way that the nominal output level of the d07 will generate a signal at the output of a MPX-encoder that causes a deviation of 40kHz. Afterwards the peak limiter of the d07 must be set so that the peak deviation is limited to 75kHz.

The dynamically performance of the MPX limiter can be changed with the parameter PROCESSING (soft, mid, hard). The best MPX limiting can be achieved by varying this parameter, depending on the program characteristic.

To enhance the signal to noise ratio for FM transmission, Pre-Emphasis on the transmission end and De-Emphases on the receiving end is used. Higher frequency signal components are raised following a standardized filter curve and leads to an increased drive of the transmitter. This level pull up must be considered for the limitation of peak levels.

In the d07 the signal reaches a dynamic low path filter **after** a broad band limiter. The cut-off frequency of this high shelf filter is controlled adaptive and time depending. The attenuation of the filter for high signal components is controlled in a way that it compensates exactly for the increase of high frequencies by Pre-emphasis on the transmitting end. This alteration of the frequency response is only effective temporarily and will practically not be recognizable for normal program material because the processing time is below the integration time of the human ear.

The threshold of the limiter can now be set in a way that maximum deviation is achieved for low frequencies. Higher frequency components will be reduced if necessary and do not cause an overshooting of the peak deviation.

If the program material has a lot of high frequency components the activity of the adaptive filter is of course more frequent and will eventually be audible.

If the threshold of the peak limiter is not set to maximum but some dB's below, the resulting headroom can be used for higher frequency components.

By changing the limiter reference level, the operating point for the dynamic filter will be set. The curves below show how the cut-off frequency of the dynamic filter varies depending on a given headroom.

Headroom = Ceiling – Limiter peak level



Due to the different adjustable thresholds for peak limiter and adaptive pre-emphasis, an optimisation in regard to maximum level and sound balance can be achieved. This optimisation should also be done under consideration of the multiplex power because higher frequency components caused by a level increase on the transmitter side will make a higher contribution to the overall power.

### 7.3. Pre-emphasis

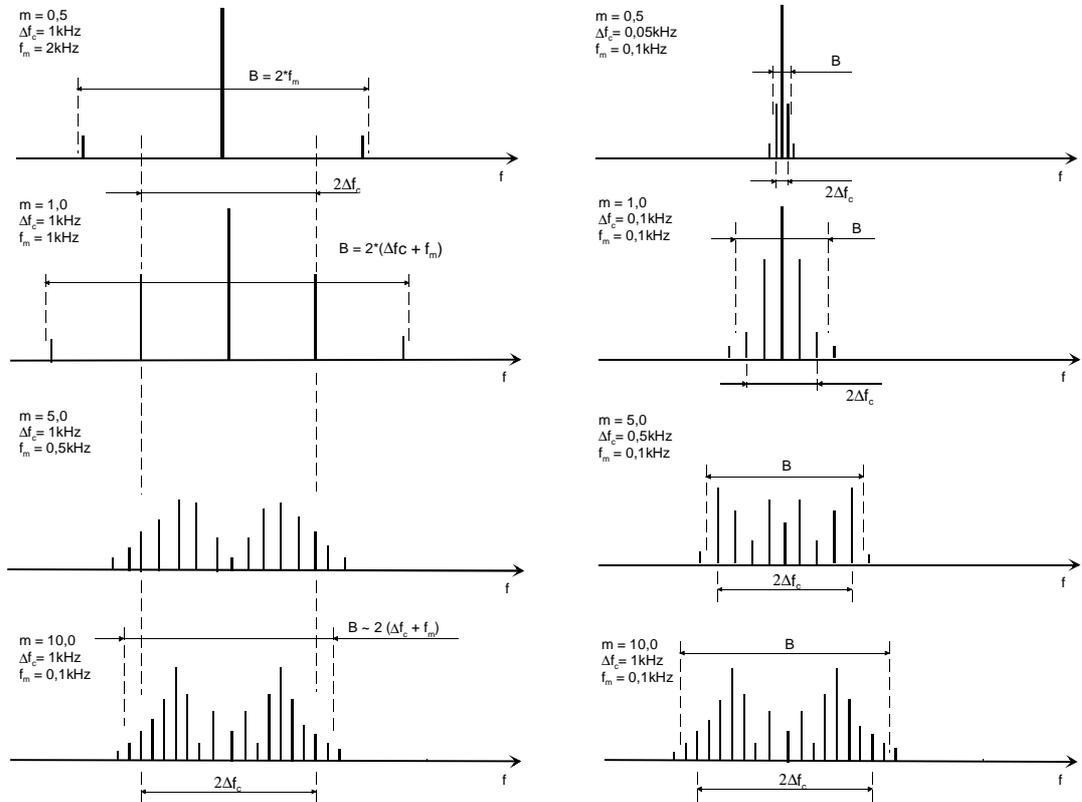
#### Pre-emphasis Ceiling

#### Pre-emphasis set-up hint

## 7.4. Terms and definitions

### Frequency modulation

The sketch below shows the principal influence of the modulation index  $m = \Delta f_c / f_m$  to the spectrum of an FM signal. On the left side the frequency deviation  $\Delta f_c$  (amplitude of the modulating signal) is constant but the modulating frequency  $f_m$  changes. On the right hand the modulating frequency  $f_m$  is constant but the frequency deviation  $\Delta f_c$  is changing :



You can show that a bandwidth of approx  $B \sim 2 \cdot (\Delta f_c(\text{max}) + f_m(\text{max}))$  is needed for FM modulation for high rejection of high frequency signals. With a maximum frequency deviation of 75kHz and 15kHz cut-off-frequency of the modulating signal  $B$  will be  $\sim 180\text{kHz}$ . Based on this fact the planning of transmitter positions and power takes place for area-wide feed. To prevent disturbance in adjacent channels the maintaining of the frequency deviation needs the highest attention.

### Frequency deviation $\Delta f_c$

Value of deviation of the mean frequency from the transmitting frequency of a FM transmitter caused by the amplitude of the modulating signal.

### Peak Deviation $\Delta f_c(\text{max})$

Maximum frequency deviation allowed for an FM transmitter. Defined by the ITU to 75kHz

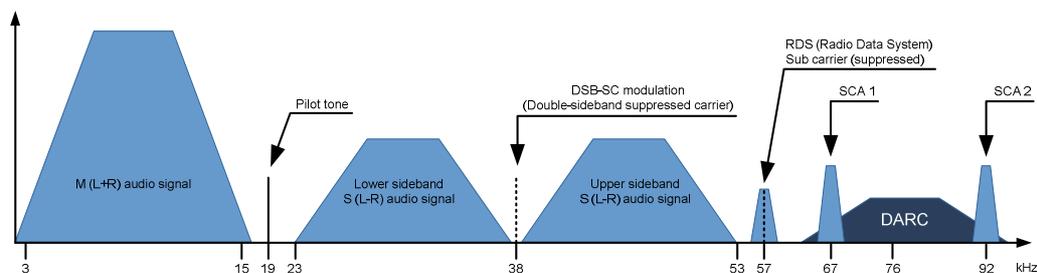
## Pre-Emphasis

Increase of high frequencies to enhance the signal to noise ratio for FM transmission equals to a time constant of a high pass filter of 50µs in Europe and 75µs in America. It will be laterally reversed at the receiver end. This value does not comply with today's spectrum of audio signals. But it is not possible to change it because of the innumerable receivers. Therefore modern audio signals may easily over drive an FM transmitter, causing nasty interferences.

## Adaptive Pre-Emphasis

Function of the d07 that makes it possible to dynamically accommodate the level of an audio signal to the high frequency boost of an FM transmitter. The effect is that over driving by high frequency signal components in critical programs will be avoided (s.a. pre-emphasis).

## MPX-Spektrum



## MPX-power

Overall power of the Multiplex signal from the sum of components of the MPX spectrum :

Monaural signal	$(M=L+R)/2$	30Hz	to	15kHz	base band
Auxiliary carrier		19kHz	approx	9%	of overall power
Double side band suppressed		23kHz	to	38kHz	lower side band
Carrier modulation	$(S=L-R)/2$	38kHz	to	53kHz	upper side band
RDS signal					
DARC signal					





d07



Jünger Audio GmbH  
Justus-von-Liebig-Straße 7  
12489 Berlin  
Germany



phone: +49 30 6777 21 0

fax: +49 30 6777 21 46

info@jungeraudio.com

[www.jungeraudio.com](http://www.jungeraudio.com)