Operating Manual

# *Trans-DIGI 2009*

***Digital Broadcast Audio Processor***

**Software version 6.08**



**Phobos**


**CAUTION:** TO REDUCE THE RISK OF ELECTRICAL SHOCK, DO NOT REMOVE COVER (OR BACK).
NO USER SERVICEABLE PARTS INSIDE. REFER SERVICING TO QUALIFIED SERVICE PERSONNEL.

**WARNING:** TO REDUCE THE RISK OF FIRE OR ELECTRICAL SHOCK,
DO NOT EXPOSE THIS APPLIANCE TO RAIN OR MOISTURE.

**WARNING: THIS APPLIANCE MUST BE EARTHED**

**IMPORTANT:**

DO NOT EXPOSURE APPLIANCE TO RAIN OR MOISTURE. IT PREVENTS RISK OF FIRE OR SHOCK HAZARD.
DO NOT OPEN EQUIPMENT, HIGH VOLTAGE INSIDE. REFER SERVICE TO QUALIFIED SERVICE PERSONNEL ONLY.
NEVER INSTALL APPLIANCE IN HOT CLOSED PLACES OR ON DIRECT SUNLIGHT. FREE AIR CIRCULATION MUST BE AROUND REAR AND SIDE PANELS.
DO NOT EXPOSURE APPLIANCE TO RAIN OR MOISTURE. IT PREVENTS RISK OF FIRE OR SHOCK HAZARD.

**DO NOT PLACE APPLIANCE TO PLACES WHERE IS:**

-EXTREMELY HOT OR COLD (OVER 40 CELSIUS OR LESS THEN 5 CELSIUS)
-DUST
-HIGH MOIST
-VIBRATIONS
-DIRECT SUNLIGHT

IF SOME TROUBLE APPEARS IN RUN TIME, OR SOME DOUBT ABOUT NORMAL OPERATION APPEARS, **DISCONNECT PROCESSOR FROM ITS MAINS IMMEDIATELY** AND CONTACT SERVICE ORGANIZATION OR THE MANUFACTUROR.
APPLIANCE **HAS NOT POWER SWITCH**.

**THE MANUFACTURORIS NOT RESPONSIBLE FOR DAMAGES OR INJURIES CAUSED BY USING THIS PRODUCT OTHER WAYS THEN DESCRIBED IN THIS MANUAL.**

**Quick start guide is at the end of this manual**

## Basic information about equipment    Phobos Model 2009

### 1 Case

- Housing: 19", 2 unit, 320 mm depth
- Weight: .. kg
- Chassis: passive steel CD5B: surface is highly conductive
- Made out of one piece of steel
- Cover: passive steel CD5B
- Cover is mounted with 12 screws
- Excellent electromagnetic compatibility
- Durable front panel, plastic vinyl surface
- PCB with SMD
- IEC power plug

### 2 Front panel

- LCD display (2 lines) for control and adjustments, Rotary encoder for adjustments
- Power control display
- Keyboard for adjustments

### Rear panel

- IEC power plug
- LED indication of analog overload controlled by DSP
- BNC output MPX (2x)
- BNC output Pilot 19 kHz
- port Remote RS 232 - Canon 9 Female for programming, USB port
- XLR line outputs (L and R)
- XLR digital input AES/EBU (Fs 32, 44.1, 48 kHz), AES digital output with 48 kHz sampling rate, internal or external sync
- XLR line outputs (L and R)
- AUX port for external reset (connect pin 5 to pin 3) and day/night switch (connect pin 8 to pin 3)
- Grounding screw M5

### 3 PCBs

- Dual sided PCB's with SMD-components.
- All new DSP circuits from Analog Devices SHARC, 6 times more and bigger power then the DSP's in the Trans DIGI 2007
- A/D a D/A converters Analog Devices, 24 bits
- Reliable components
- Excellent electromagnetic compatibility
- Low noise analog circuits
- Sophisticated algorithms of signal processing, 48 bits internal enumeration precision
- Remote control via RS 232 port and Windows software

## 4Basic characteristic **(quick start guide – see end of manual)** Phobos Model 2009

The Phobos Audio Trans-DIGI 2009 DXC is a DSP based modulation audio processor for AM, FM and TV broadcasting. The processor works **purely mathematically**. Signal processing like filtering, compression, limiting etc. is realized **by calculations** on input samples. The processor works with floating point DSP circuits. DSP's following instructions stored in the program memory. The program memory can be upgraded and the structure of processing can be changed in the future. All processing structures are then defined purely by software like computer software. The processor works with two independent levelers and a total of five-band limiters. The Trans Digi 2009 has a build-in stereophonic encoder with 2 MPX outputs capable of feeding 2 independent transmitters. Processor adjustments can be made via RS 232 or USB using a PC, or by the keyboard and display on the front.

Parameters are stored in EEPROM **non-volatile** memory.
Windows 32-bit software enables adjustments, storage of parameters into presets and files and uploading of these settings.

Parameters can be adjusted in a wide range from slightly processed to a highly competitive sound.
Circuit design and used algorithms interlocks clear processing without perceptible distortion.
The output limiters and clippers are controlling the peak deviation and enable the use of maximum deviation (+/-75 kHz).
A stereo encoder with FIR input filtering generates an output waveform via discrete D/A conversion. It enables very good spectral purity and stereo separation.

The processor uses two microcontrollers and 2 DSP's. The first microcontroller is for communication with a PC, keyboard, display and DSP's, the second microprocessor is for controlling the stereo encoder.

## 5Principle of operation

Please refer to the block diagram of the Trans-DIGI 2009 at the end of this manual.
The input section consists of the de–symmetrical block. Then the A/D converter follows. The A/D conversion works with 24-bit resolution and a 32 kHz sampling rate. Sampling on 32 kHz is optimal for FM use. Audio signals on FM must be frequency limited to 16 kHz; 32 kHz is enough to carry all audio information transmitted via FM. Moreover, the power of the DSP's is invested into sound algorithmic optimalisation.

Quality and complexity of sound algorithms makes sound and overshot protection on processor's output. Instead of arguments about high sample rates, we rather invest all DSP power into quality of the audio processing itself. Result of this is great pilot spectral protection without overshoots. Some of our competitors are using 48 or 96 kHz sample rate. It means (in case of 96 kHz) they need more then 3 times bigger power of the DSP's. More because there is the need of filtration of the output audio signal to maximal 16 kHz bandwidth. Some our competitors do not apply output filtration. Result is an overshot in the stereo encoder. They need strong composite clipping to delete these over-shots. Pilot protection is very low in this case. Spectral purity in region of sub-carrier (57 kHz) is poor. It means RDS coverage can be reduced and stereo separation harmfully affected.
**In our conception signal is spectrally limited to 16 kHz**. **We are only using the safety clipper on the composite outputs**. As a result, you got a high pilot protection and high protection in the sub-carrier region (57 kHz).

24 bits Input A/D converter itself works with **over-sampling and input FIR** filtration. It is insensitive for spectral components over 16 kHz on input.

AES-EBU digital input block contains a SRC (sampling rate converter). This converter converts inputs with sampling rates between 32-48 kHz to an output Fs of 32 kHz on which the DSP's are working. Digital de-emphasis (50/75 us) is also incorporated.

**DSP's runs program** (processing algorithms). Internal calculations are made with 48-bit precision.

After the DSP's, the signal goes to the output D/A converter. This D/A uses an over-sampling and digital **FIR** (finite impulse response) filtering. This kind of filtration provides minimal overshot and high spectral purity over 16 kHz. This is a big advantage, because the following stereo encoder has no input filtering on 19 kHz. Without filtering, the encoder generates no overshot on its output signal. Reproduction of the D/A output is near perfect. The pilot spectral purity of –75 dB is achieved (relatively to full output) or –55 dB (relatively to pilot signal level).

Processing parameters are send from the main microcontroller to the DSP's. Data is stored in a non-volatile EEPROM memory. The microcontroller communicates with the keyboard, the display and/or a PC. After every start of the Trans-DIGI 2009 or after a change of processing parameters, data is send from the microcontroller to the DSP's.

The following explanation is based on processing structure, in other words, **what the audio signal is doing inside the DSP's**. This explanation is based on the block diagram (on the end of this manual). In contrast with the analog appliance block, the diagram is not describing the physical structure of the processor, but rather the structure of the audio processing software. All operations are made within numeric domains.

**The first block of the signal processing** (not indicated in the block diagram) is a high-pass IIR filter with its –3 dB point at 30 Hz. This eliminates sub-acoustic components and DC components from the input signal. The audio spectrum is limited to 16 kHz (by input FIR filter of A/D converter) on other side.

**Defeat-able dynamic spatial enhancer** follows. The enhancer works with a controlled Side/Mid ratio. The amount of stereophonic information is controlled to achieve optimal results. If an original music sample has less stereophonic components, the enhancer boosts the stereo width image. If the original sample is already width enough, the enhancer reduces the stereo width image. Side/Mid information is affected (S/M should be less then 1). The enhancement and the stereo/mono ratio is user adjustable.

**The Next block** splits the input signal into two bands. The crossover frequency is user adjustable (LFT parameter) from 50 to 200 Hz. A low band leveler then processes the low band (30 Hz – LFT Hz). The amount of processing in this leveler is affected by the Bass Density control. The output from the low band leveler is connected to the Optimizer spectral balance control and then into the Optimizer (summing point). In case that **TURBO** processing is switched ON, the Bass process is doubled (x2).

**Processing signals above 50-200 Hz** (above LFT) is much more complex. The signal is processed in the high-band (main) leveler first. Both levelers (Bass and main) have three controls. Two of them are the same for both. These are Gate, Level and Leveler Range. Gate level determines the input signal level relatively to A/D clipping (or full-scale dBFs). If the input signal level comes under the gate level, the gain of both levelers is frozen. The Gate level can be adjusted within a range of –10 to –30 dB (under full scale of input digital representation).

The second common parameter for both levelers is the Leveler Ranges control. This parameter sets the range in which levelers adjust gain (for both levelers). If you adjust –20 dB, the levelers can increase gain maximally to 20 dB. If you adjust –6, levelers can increase gain maximally to 6 dB.

Third control is Bass and Master release. This control is **independent** for both levelers. Adjustment of release time affects the speed of the levelers gain riding. The Bass leveler is slower then the main leveler. The reason for this has to do with acceptable distortion, induced by leveler action.

In case of just **speech, Bass-Main coupling** automatically takes place, resulting in clear voice processing, free from Bass "booming" or over processing. The Bass Density (Bass on voice) control enables the user to control this coupling. If the Bass density is set to low, voice is without strong Bass. If Bass Density is set to high, Bass processing on voice is strong, especially male voices are reproduced with strong Bass.

Loud and weak sections in program material **have similar levels** after levelers.

After the main leveler, there is a Density control. The amount of density affects the amount of processing in the limiter structure. Between density control and limiters, **pre-emphasis** is applied to the signal. Pre-emphasis (boost of high audio frequencies according to CCIR standards) is 50μS in Europe or 75μS in the USA. The use of pre-emphasis increases the signal to noise ratio of receiver resulting in better sound quality.

The signal is then split into 4 bands. In every band there is an EQ control. This control separately enables the control of the load in every 4 bands. The crossover frequency between band 3 and 4 is selectable (from 3.8 to 7 kHz) using the **HFT user control**. The 4-band limiter has adjustable release times for every band. An adjustable feedback controlled expander in the highest band (HFT – 15 kHz) follows the limiter. An expander prevents noise modulation when processing noisy program material.

In case **TURBO** processing is switched ON, the limiter structure is doubled (x2).

**The Optimizer.** The Optimizer is a summing point + master limiter + master clipper). Signals coming from all 5 bands are summed together. The summing relations are adjustable. The signal then goes to the Optimizer drive control. The Optimizer drive control determines the drive of the output limiter. **The Output limiter** takes care of the final output level setting. **A Master clipper** follows the master limiter. The Adjustable Master clipper removes all unwanted over-shots from the signal.

The relation between limiting and clipping affects the **character of sound**; it will be discussed onwards in this manual.

A feedback controlled **main expander** is placed after the final output of the limiter structure. If the output level falls under the adjusted level, the main expander increases the signal to noise ratio by expanding low-level signals.

Finally, the signal is filtered by an **FIR filter** and then converted by a D/A converter. This concept provides the highest pilot protection, high spectral purity above 16 kHz and no over-shots.

The following stage is the **internal stereo encoder (stereo generator)**. It is strongly recommended, if possible, to use this internal encoder. This internal encoder consists of **switched discrete** A/D converters. These converters are running on 4 times (152 KHz) higher frequencies then the 38 kHz chopping frequency. This "over-sampling" enables the Trans Digi 2009 to have Low Pass (LP) output filters with a high cutoff frequency. Such filters practically do not have any affect frequency characteristic and group delay in the band under 53 kHz. Good stereo separation is then certain for all frequencies in audio band.

Because the input audio filters (FIR) have a **flat characteristic of group delay**, output clipper is not necessary. We only have installed a safety output clipper + adjustable composite clipper. This action leaves the spectrum of the composite signal without impurities. Pilot signal is protected and a good separation and effective stereo coverage is thus available. The Pilot signal is summed with the output of the LP filters at the summing point. After this summing point, the signal goes into the **output amplifiers**. The output amplifiers are capable of driving 600-ohm loads. Both outputs can be adjusted separately. Levels are adjustable in a wide range. An AES-EBU digital output is also available. The encoder has a separate **output of the pilot signal (SYNC)**. The level of this signal however is fixed. All MPX and SYNC outputs and the SCA input of the stereo encoder are on BNC type connectors.

**Two microcontrollers are in control of the whole system**. First a system microcontroller. This microcontroller controls the main board, DSP's, display, keyboard, rotary encoder and communicates with a PC via RS 232 or USB. The second microprocessor is the stereo encoder controller. It controls all the switching and timing inside the encoder.

# 6Installation and Connection to other equipment

Installation into an audio rack is recommended.  Free air circulation around the processor must be possible.

Output from a mixing desk (or output from a line buffer/splitter) is connected to the XLR-3 inputs resp. the AES/EBU input in case of a digital signal. The composite output BNC connectors are connected to the input of an FM exciter, composite link (STL) or RDS encoder. If the RDS encoder needs synchronization input, connect the 19 kHz pilot signal (SYNC), onto the SYNC input of the RDS encoder.
Recommended is connecting the MPX output to the loop-through input of the RDS encoder. Loop the RDS encoders output (MPX + RDS) into an FM exciter, STL etc.

If the link to a transmitter can only be feed by L/R stereo line, use the XLR line outputs.

Connect an RS 232 9 pin cable (1:1) to the RS 232 port of the Trans Digi 2009, and on other side to a free COM port at the PC. On the other hand, you can also use the USB connector on both processor and PC. In such case USB drivers are necessary. USB drivers for Phobos products are downloadable from www.phobosaudio.cz , section of downloads.

## Installation of the processor in the Air-chain

The processor can be connected to a transmitter via MPX output or by the L/R line outputs. MPX connection is highly preferable.

An overshot free signal from internal stereo encoder is available **on this output**. The main advantage is using the internal FIR filters and internal pre-emphasis. The processor itself must be installed as close to the transmitter or multiplex transmission line as possible.

## Warning, important!!!

**If using analog inputs, it is necessary to adjust the input sensitivity properly.** After the A/D conversion, the processor cannot remove distortion caused by A/D overloading, or systematically low loading. It means, you have to take **maximal care** about input sensitivity adjustment. Input sensitivity switches are on the rear panel and are independent for each channel L and R.

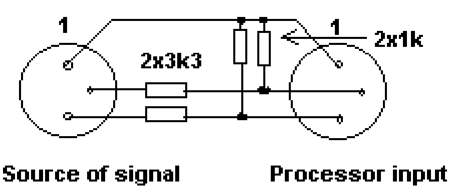
Start with all the input sensitivity switches on the rear panel switch in the "UP" position.

The first step is to load the processor from a mixing desk with full level program material. It means "faders up", or maximal output level, which really exists on the output of mixing desk during normal on-air operation.

Now, switch piecemeal the input sensitivity switches (+3,+6,+9,+12 dB) to the low position. At the same time, observe the "OVER" LED. If the LED starts blinking or slightly lights, stop switching and return to the previous position. This operation must be repeated for both (L and R) channels. If all is OK, the configuration of the input sensitivity switches will be the same for both channels. If not, you have to check your mixing desk output, cables and (optional) preamplifiers in the studio chain for symmetry of output levels.

**It is recommended to check the configuration of the input sensitivity switches on different program materials and moderators**.

If the input signal has a too high level, in this case it can't  be impossible to reduce the input sensitivity sufficiently. In other words, all input sensitivity switches are in UP position and LED's are still blinking. In this case, you have to reduce the output level of the signal source (mixing desk etc.…). If you cannot reduce the output level, you have to add attenuation circuit between the signal source and the processor. A schematic diagram of a simple –10 dB attenuation circuit is on the fig.



### Adjustment in case of MPX connection (highly preferable)

One MPX output must be connected to the MPX input of an FM exciter or multiplex line. Activate the test tone of the processor. Because the level of this signal is 88% of total modulation, you have to adjust the deviation caused by this signal to 66 kHz. This adjustment can be done by increasing the MPX output level control on rear panel, or by the sensitivity control of the exciter. If you incorporate a transmission line, set the load of this line to minus 1,1 dB under maximal level.

After this basic adjustment, test the deviation with real program. Set the deviation at precisely 75 kHz peak on real program (If you can, use an FM modulation monitor or a spectrum analyzer).

### Adjustment in the case of L and R line outputs (not recommended)

Connect both output lines from the processor to the L and R inputs of a transmission line or transmitter. Activate the test tone of the processor. Because the level of this signal is 85% of the maximal level of the line outputs, you have to adjust deviation caused by this signal (and pilot from an external stereo encoder) to 66 kHz. This adjustment can be done by increasing the L and R output level controls on rear panel, or by adjusting the sensitivity control of the line input or external stereo encoder. If you incorporate a transmission line, set loading of this line to –1,4 dB under maximal level.

After this basic adjustment, test the deviation with real program. Finally set the deviation precisely to 75 kHz peak on real program.

### Adjustment when using pre-processing and the Trans Digi 2009 at your transmitter

If you are using a non-composite STL (which you have to feed with L and R audio), cable modulator or music landlines and want to use the Trans Digi 2009 at your transmitter in order to get maximum protection of the transmitter, be sure your studio pre-processor is set to a rather small amount of processing.
In this case, you have to make sure to set the leveller of the Trans Digi to no more than about 60 -70 (approximately. 0dB input to the AD converters) when feeding 0dBm in combination with the input sensitivity switches on the back (see this manual). If the leveller is working to hard, this will result in breathing and pumping, which sounds very unnatural and causes listeners fatigue.

For example: If you use a Compellor® or leveller, set the amount of processing to no more than about 6 -10 dB. When using a simple compressor/limiter set the attack time to slow and make the release time very long in order to get a sound that sounds open and not heavily processed.

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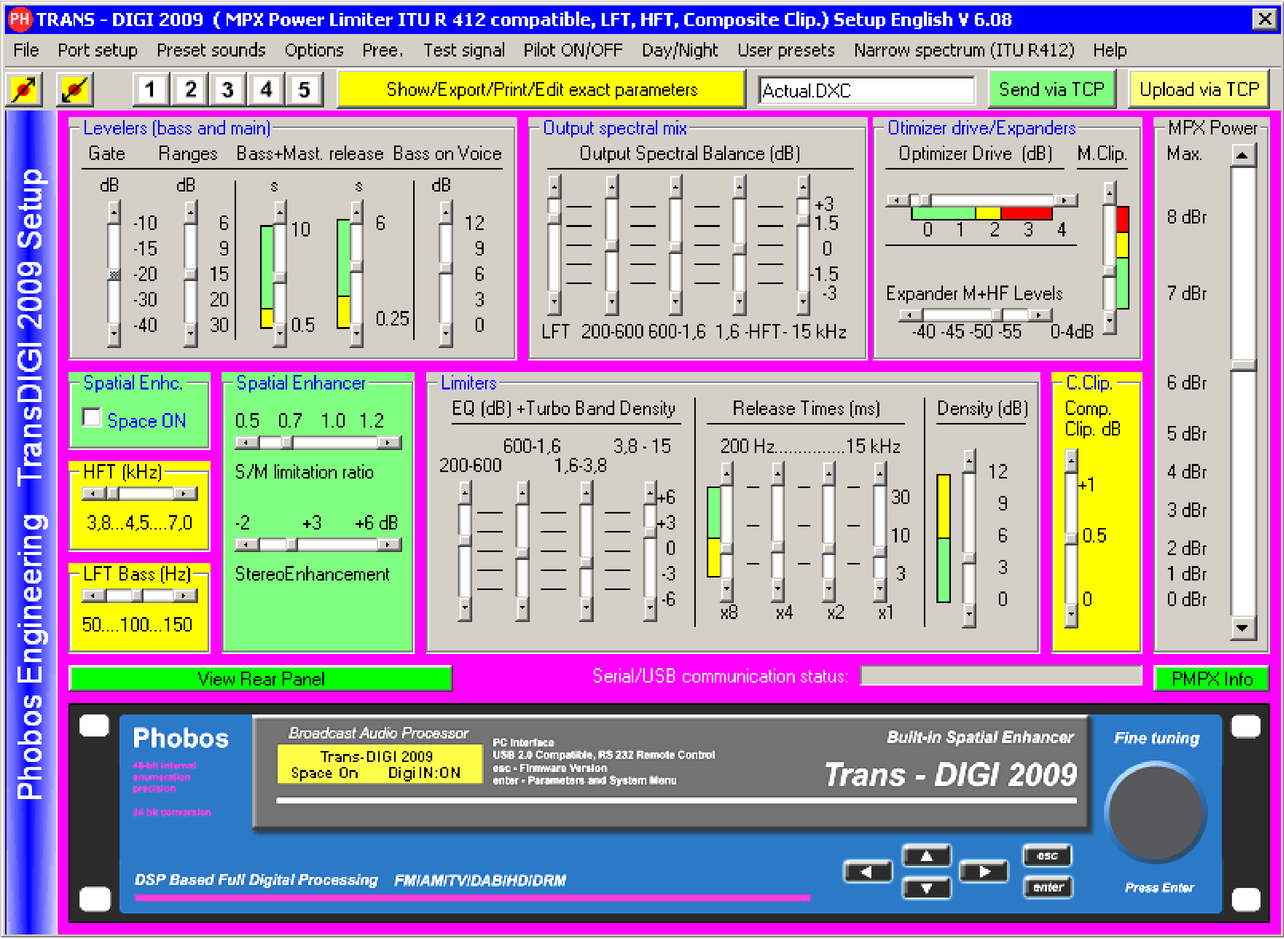
## 7Software installation (PC with Windows© 95/98/2000/ME/XP or NT)

Insert the CD with the **PHOBOS ®** software into your CD drive of PC or download PC control program from www.phobosaudio.cz .

Run setup.exe.

Start the program from the Program folder. The next window appears:




## 7.1 Setup from a PC

In the main window of the application, You can see the processing structure and all the available controls. You have three types of controls (check boxes in menus, push buttons in icon line and sliders in the main window). You can move the controls with a mouse (click for check boxes and push buttons, drag for sliders).

In menu **„File"** you can save the file, upload the file, and exit the program.

In menu **„Port Setup"** you can select a communication port. If you have not installed an RS 232 port on your computer (notebook), communication via USB is also possible. For using USB, you need a reduction cable USB → RS 232 or USB → USB and a Windows© driver. In this case select one of the ports 3 or 4 (typically 3). Cable and driver can be ordered at most computer shops or from Phobos Audio. **USB drivers can be found on the software CD or at the Phobos web site**.

In menu **„Preset Sounds"** you can find some basic sound presets. You can use these presets for quick set-up or for further experiments.

In menu **„Options"** you can find the input selector (Digital ON/OFF – Digital OFF or Analog). Another option is enabling of the Day/Night switching over AUX connector on the rear panel. **Turbo** is also activated here.

In menu **„Pre"** you can switch between 50µS or 75µS pre-emphasis. **In Europe** standard (according to CCIR) is 50µS, in **North America** 75µS.

Menu **„Test Signal"** enables to start the test signal 1 kHz at the left channel. The signal is 100% of the maximal output level at the left output, or 100% of the maximal output level at the MPX output. This signal is also useful for channel identification, deviation pre-adjustment and basic check of multiplex characteristics (because spectral components of 1 kHz, 19 kHz and 38 kHz are present at the multiplex output). This test signal is generated digitally in DSP's.

Menu **Pilot ON/Off** enables mute of pilot signal (19 KHz) at the multiplex output.

Menu **"Day/Night"** enables saving of the actual configuration into the Day/Night internal preset and recalling of these presets. Second function is enabling or disabling this control by switch. The switch must be connected at the AUX connector or rear panel of processor (connect pin 8 to pin 3).

Menu **"User Presets"** enables save and recall of actual configurations into four internal User Presets.

Menu **Narrow spectrum (ITU R-412)** enables to switch on regime with enforced compatibility with ITU R-412. Slightly reduce the stereo effect. Switch to ON only if regulation authority test output spectrum and some reduction of bandwidth is required.

Menu **„Help"** inform us about starting of this help and gives basic information about program and communication parameters used (1200,N,8,2).

Button **"Show/Export/Print/Edit Exact parameters"** switch to the table of parameters. You can **see the exact values** of all parameters (0-255). **Press Print** for printer output. **Press Export** for export parameters to a **commented text file**. You can **edit parameters by** click on a parameters value, press Del **and type a new value (0-255).** Then press the **"Send Edited Parameters to Main Setting Panel"** button. The new parameters **will be** sent to the main application window.

**Under the Menu bar, you find the Icon bar.**

The first icon with an up-arrow enables (by mouse click) you to send parameters from the PC to the processor.

The second icon with a down-arrow enables uploads the processors parameters to the PC's software.

Push buttons named 1-5 are preset buttons. If you press some of this buttons, you can immediately change the setting of the processor. The appropriate file named **preset1.DXC - preset5.DXC are uploaded from the hard disk**. You can create these preset files using the File menu, if you save the file named preset1.DXC – preset5.DXC.

Button **Show (Print)/Edit Exact parameters** is at the right of the icons. It skips to a new window were you can see the exact values of all parameters. Printing is also possible. If you edit parameters, you can send them back to the main window by pressing **Send edited parameters to main panel** button.

**Send an Upload via TCP** allows sending and uploading data via TCP/IP to distant site. A TCP/IP receiver/RS232 **converter module** is necessary.  This converter is available from a third party.

You can see the **information line** right at the icons. The name of the actual setting (DXC file) is indicated at this line. The Actual.DXC is opened after the program starts. This is a file that was saved at the moment of program exit. It means, the program starts in the same configuration as you last used it (the position of controls is not lost during program exit time).

## Review of the controls

Review of controls follows. Every control is discussed here for some program types.

### 7.2 <u>**Gate lev / -10...-30 dB /**</u>

Sets the gate level in dB relatively to full loading (0 dBFs). Setting for low levels (example -30dB) enables the leveler to act in a wider range of input levels. If respect for the original dynamic relations is the first goal of your processing, best is setting this parameter to higher level (example –10 dB). Gate level is common for both levelers, main and Bass. For „Hit, Pop or Rock" format set -25 -30dB. For classical music set -10 -15dB.

## 7.3 **Lev. ranges / 6 – 20 dB /**

Sets the range of the levelers action in dB. More action reduces the original dynamic of the program strongly. If low dynamic reduction is the goal of your processing, select a low range (example 6 dB). For big reduction of dynamic, select 20 dB. In case of „Hit, Pop or Rock" format select 20dB. For classical music, select 6-10dB. The action of the levelers is also affected by control of the Gate level.

## 7.4 **Bass + Master release / 0.5 - 10 s (Bass), 0.25 - 6 s (master) /**

Sets the release time of the levelers. The release time is the time for 10 dB gain advance (if Gate is not activated).

Setting to low time results in better tracking of input signal level variations. We recommend using this setting in case of "Hit, Pop or Rock".

Setting to mid and high release times is good for universal program and classical music.

Action of the levelers is affected by Level **ranges and Gate level too**.

**If set Bass release time to minimal time (0.5 s), it increases Bass punch.** It could be **very positive for "Hit, Pop or Rock" formats**. You can successfully combine minimal Bass release time with medium (2–3 s) main release time for this formats.

We cannot recommend adjust highly different release times for main and Bass leveler for universal and classical format

## 7.5 **Bass density / 0 – 12 dB /**

Density for Bass affects the amount of Bass. This control enables the drive Bass leveler from 0 to +12dB.

For original sound character on Bass we recommend set this control to 0 – 3 dB. **For "hit pop or rock" format set 6 – 12 dB, for big Bass punch (in combination with Bass release to 0,5 s).**

## 7.6 **Density / 0 – 12 dB /**

Sets the amount of loading of the 4-band limiter and then the density of the sound.

Density is the amount of sound compression in the 4-band limiter.

If density is high / **6 – 12 dB** /, sound is highly limited in the 4-band limiter. Dynamic of the original program is strongly reduced. All is loud.

**Spectral normalization** takes place. Every program material after processing has a similar audio spectrum. It is a typical setting for loud "pop, hit or rock" format. For long term listening we recommend to not use density over 9 dB, to avoid listening fatigue. Only use such setting if maximal loudness is the only goal.

Middle density / **3 – 6 dB** /, is good for standard pop or universal program. Spectral normalization is weaker, listening fatigue is not a problem. Such sound is "funny" and transparent. It can be successfully combined with setting the Bass density to (example) 9 dB and Bass release to 0,5s. In case of pop format, it gives pleasant sound with good Bass punch.

Low density / **0 – 3 dB** /, is good for universal program or for classical music. Spectral normalization is practically out of game. Such sound is not loud, but is very natural.

## 7.7 **Limiters EQ / - 6...+ 6 dB /**

Sets the amount of loading of the bands in the 4-band limiters. In other words, you can selectively affect the "density" in the entire band. Sound character can be changed by this way. For pop format you can apply +3dB in band 3,8-15 kHz and +2dB in band 1,6-3,8 kHz, for example. Some listeners prefer an attenuation of 1-3 dB in band 200-600 Hz.

Recommended setting of all EQ controls is 0 dB for universal program. For classical program you can set 1,6-3,8 kHz to 1-2 dB and 3,8-15 kHz to –2 dB.

## 7.8 Release times / 3 – 30ms,x2,x4,x8 /

Sets the release times for the limiters in the 4-band limiter in ms. The release time affect dominantly the character of limitation of the entire band. If you, for example, set all bands in the middle position, only the 3,8-15 kHz band to minimum (3 ms), high frequencies will be subjectively boosted. This boost is based on increase of average , not peak level of high frequencies. In Such way, it is an effective tool for affecting the sound character without change of the real spectral balance and without change of peaks.

You can set band 3.8-15 kHz for low release time (3-10ms) for „pop" format. The same (in fewer amounts) for 1.6-3.8 kHz band. Such setting guaranties open and funny mid and high frequencies.

We recommend set all this controls to mid position for universal and classical format.

## 7.9 Optimizer drive / 0 – 4 dB /

The optimizer is the summing point + the output limiter. This control sets the amount of loading for the limiter section of the optimizer. If 0 dB is set, the limiter is activated only very occasionally. Such setting produces lower loudness, but without risk of spectral gain intermodulation. Setting to +4 dB produces relatively high loudness, but wit some distortion and intermodulation risks. Setting of this control coheres closely with setting of the Master clipper. Final sound set-up must be based on limiting or clipping, not both. If you set the Optimizer drive to more then 2 dB, you cannot set the Master clipper to more then 1-2 dB and vice versa. Generally, sound based on clipping is more naturally, but with more distortion risk. Many listeners prefer clipping–based sound.

## 7.10 Optimizer spectral balance / - 3.....+ 3 dB /

Sets the summing balance in the sum section of the optimizer. This balance affects the spectral balance of output signal.

We recommend to use this controls canny and avoid strongly boosting high frequencies (also depends on Release time 3,8-15 kHz and Limiters EQ 3,8-15 kHz). Boost of high frequencies of audio can cause reduction of effective coverage of transmitter, especially on car radios.

## 7.11 Expanders Main and HF / – 40....- 55 dB /

Sets the expansion levels for HF or respectively Main expanders. The expander's action causes improvement of the signal to noise ratio, especially when noisy program material is processed.

Setting for high levels (-20, resp. –30 dB) cause massive expanders action. For noisy material it is OK, but for good quality material it cause canceling of reverberation and low-level high frequencies. We recommend to set both controls to region –40 -45dB (HF) resp. –50-55 dB (Main).

## 7.12 Master Clipper

The master clipper is the output-clipping device. It prevents the processors output from over-shoots. Setting of this control depends on setting Optimizer drive control as was discussed in the Optimizer drive section. You cannot set both controls (Optimizer drive and Master clipper) to high levels simultaneously. Sound must be based on limiting or clipping, not both. **Clipping based sound is more natural,** limiting based sound sounds more "processed" but has lower distortion.

## Other controls (Model 2009 specific controls, not present in Model 2007)

### MPX Power

Sets desired MPX power**. Set to +9 db for no limit.** If limitation is required by regulation authority, set desired level. Acts in multiband limiter.

### S/M limitation

Sets limit for side to mid signal if space is on. Recommended setting is under 1. Higher ratio can negatively affect coverage of the FM transmitter.

### Stereo Enhancement

Sets gain of side channel in space. Recommended setting is less the +4 dB. Higher gain can negatively affect coverage of the FM transmitter.

### LFT (Bass Shift)

Control for Bass peaking. Set to 50 Hz for deep Bass, big Bass punch. Set to 200 Hz for less Bass punch. Optimal setting depends on target listeners and listening devices. For home HI-FI and good car equipment with subwoofer, you can use 50 Hz. If your target is portable radio, use 100 – 150 Hz. Setting to 100 Hz is a good compromise for all receivers on the market.

### HFT (HF Shift)

Control for high audio frequencies. Set to 3,8 kHz for natural processing on high frequencies. Set to 7 kHz for soft high frequencies (OK for dome tweeters in HI-FI systems or good car equipment). For portable radio set no more then 5 kHz. 3,8 - 5 kHz is a good compromise for all receivers on the market.

### Composite Clipping (Comp Clip)

Control for amount of intelligent composite clipping. Set to 0 dB for no composite clipping. You can use composite clipping without any artefacts up to +0,75 dB. +1dB generates louder sound with some very small clipping artefacts.

This parameter is mot memorised inside presets, you must set it in parameters menu (if set from panel of 2009) or manually adjust from PC program.

## Potentiometers on rear panel

Adjustment of the output levels for line (L and R) and composite (1 and 2) outputs.

## 7.13 Setting via keyboard

**(First, it is necessary to unlock the keyboard**. This can be done by pressing the arrow marked keys arrow up, arrow left, arrow down in less then 1 sec. after each other.)

All adjustments can be **done without a PC**, via keyboard and display on front panel.

Setting via keyboard is **easy and user friendly**. We recommend studying the processor structure (from block diagram in section 6 of this manual) prior to make adjustments.

**The Display** informs you about the parameters you are adjusting. The basic display inform us about selected input (Digi means AES/EBU, Analog means line XLR) and Space status (Spatial Enhancer ON/OFF). The first display line has some additional functions. If digital input is activated and no digital signal is on input (no synchronization), the processor automatically switches to analog input. The first line of the display informs us: **DIN Unlock>Analog In**. If synchronization on the digital input appears, the processor automatically switches back to digital input and the display returns to basic status. Analog input then can be use as reserve in case (for example) digital AES/EBU line failure.

If analog input is activated, the first line of the display informs you about overloading. At every moment in which some (L or R) channel of input A/D converter is overloaded, display inform: **Overload !!! If such information appears, it is necessary to re-adjust the input sensitivity of the processor !!**
Overloading of the A/D converter causes non-removable distortion and degrades the output sound quality drastically.
If you want adjusting parameters, **first is necessary to unlock keyboard.** It can be done by pressing keys arrow up, arrow left, arrow down in less then 1 s after each to other. The keyboard is then unlocked 3 minutes from the last touch. After 3 minutes, the keyboard locks automatically.
By pressing **ESC,** the number of the actual firmware version is displayed:

**By pressing ENTER you enters menu for system (presets, spatial enhancer, input select, test tone…..):**

```
Parameter Setup  ^
System Setup     >
```

By pressing right arrow (>) you enter to system functions, by pressing up arrow to menu parameters.
**If you enter system functions (>), the following appear on the display:**

```
Factory presets ?
press   ENTER
```

By **pressing ENTER,** you enter the factory presets menu. The first preset is Natural sound. Display looks like:

```
     Natural  Sound
Enter select    < > next
```

**By pressing ENTER,** you activate Natural Sound preset, by pressing arrows left/right listing in factory presets (9 presets, the same as in the PC program). By pressing ESC, you enter the basic menu. Other system functions are in User presets.
By pressing ENTER enters menu recalling the User presets. User presets are presets definable by user. Processor has four user presets plus Day and Night preset. First preset is User preset 1.
Display shows:

```
    User presets ?
  press   ENTER
```

```
  Spatial Enhancer
        ON
```

**To recall User preset 1** press ENTER. By arrows left/right you can scroll over User presets 1-4. Press ESC for return to main menu. Saving of User presets is located in Parameter Setup menu.
Next menu is the **spatial enhancer menu**:

Selection between **ON/OFF** by arrows left/right. By pressing ENTER save status into memory.
The same for **Digital input ON/OFF** menu. If Digital input is OFF, Analog input is ON.
Menu **Test Tone is similar**. Enables Start/Stop test tone 1000 Hz in the left channel. The level of the test tone is 85% of the maximal level at the line output and 88% of the maximal level at the MPX output.

```
Day/Night switch
enable OFF
```

Next menu is **Day/Night** switch:
This menu enables switching between day and night setting. Switching is made via pins 1 and 5 of the AUX connector on rear panel. If these pins are connected together, night setting takes place. Day and night are presets in the USER PRESET menu.

**By pressing the up arrow, you enter the parameters menu.**
The display is similar for all parameters. Two examples:

**The first line** (on the left side) is the parameters name, on the right side is **parameter value**. This value

```
 Gate Level           104
-30  ■■■■■■■        -10dB
```

```
Density                124
 0  ■■■■■■■■■■       12 dB
```

is in the range of 0-255 for all parameters. In the second line is the indication units and the range for the given parameter. Between them is a bar, indicating the value of the parameter graphically. Because the bar only has 12 segments, the exact value is on the right side of first line.
**Left/right arrows realize parameter adjustments**. After adjustment, press ENTER. At the moment of **pressing ENTER**, the parameter is saved into memory and **you can hear the sound with new parameters.**
**For** the next parameter press up/down arrow.

**Save to user preset enables you to us save new parameters into User preset 1-3. Select of preset number by left/right arrows. Save into selected preset by ENTER.**

## Quick starting guide (refer this manual for details)

**Unpack** and inspect the processor carefully (**do not connect the power line** if mechanically damaged or wet)

**Connect the output of a mixing desk or studio** to the analog or digital AES/EBU inputs

**Connect Composite Out 1** to a transmitter or a Composite Studio Transmitter Line (highly preferable)

**Or**

**Connect L, R** analog outputs to L,R transmission lines to an FM Transmitter (not preferable)

Connect **power** (use only 3 wires power cord, grounding is mandatory, **appliance MUST BE EARTHED**)

**Unlock keyboard** (press up, left and down arrow and then ENTER in less then 1 seconds)

Enter System menu (right arrow), **Factory presets (by Enter)** – select preset (example TURBO Disco... or other) and press Enter (data is loaded only if you press Enter !!)

By pressing down arrow **go to Digital Input** menu, select Digital Input ON or OFF (Off means input line is analog), press Enter !

If analog input L and R line from a mixing desk are used, it is **necessary to adjust the input gain** by switches at the rear panel. Play music from your desk with +6 dB level relatively to normal output level. Switch DIPswitches down (+3 dB, +6 dB....). At some moment LED (L and/or R) start blinking, the A/D converter is overloaded. Select switches positions with no overloading in both channels. Test in normal conditions with music and voice, the overload LED's must not blink (Overloading is indicated on the display too, Overload!! appears at moments of overload L or R or both)

**Adjust the output level** by pots on composite 1 out (rear panel) to reach **-/+ 75 kHz peak deviation** in case of composite output is used. If use L and R outputs, set L and R output pots for proper output level.

**Listen to the radio and select the optimal preset for you**. (Turbo Sprinkle is OK for first experiments, is similar to Orban 8400 sound, HOT City and HOT AC is similar to Omnia 6 EX sound, Golden Dance and other presets not called TURBO are backwards compatible with Model Trans-DIGI 2007, presets called + 5 dBr sample..... are presets with limited MPX power – it means not so loud, use only if regulation authority need it). Description of the presets is on the next page of this manual. **Preset is loaded before pressing Enter !**

**Congratulation**, you are on-air with the Trans Digi 2009, we recommend to you study all pages of this Operating Manual.

## Description of Factory presets

**Golden Dance** – backwards compatible with model 2007, generates MPX Power approximately. 6,5 dBr
Spatial enhancer is OFF, relatively loud preset good for **disco** or pop formats. In model 2007, this is the loudest preset.

**Urban Pop** - back compatible with model 2007, generates MPX Power aprox. 6,5 dBr
Spatial enhancer is OFF, relatively loud preset good for hit and **pop** formats. In model 2007, it is the second loudest preset.

**Adult Pop** - back compatible with model 2007, generates MPX Power aprox. 5,5 dBr
Spatial enhancer is OFF, preset good for **pop** formats if target is listeners 35 - 45 years old. Middle loudness, gently processed sound.

**TURBO !! Universal** – new 2009 TURBO preset, gentle processed sound with middle loudness,
MPX Power aprox. 5,5 dBr. Intended for universal program with using of new TURBO processing structure.

**TURBO !! Pop** - new 2009 TURBO preset, middle processed sound with middle loudness,
MPX Power aprox. 6 dBr. Intended for universal **hit and pop** program with using of new TURBO processing structure.

**TURBO !! Disco** - new 2009 TURBO preset, loud but clearly processed sound with high loudness,
MPX Power aprox. 7 dBr. Intended for disco or modern pop format with using of new TURBO processing structure.

**TURBO !! Sample +5 dBr MPX Power** - new 2009 TURBO preset, sound processed to achieve a target MPX power +5 dBr in all situations. Use only if regulation authorities apply such limitation of MPX power. Use 2009 new TURBO processing structure.

**TURBO !! Sample +3 dBr MPX Power** - new 2009 TURBO preset, sound processed to achieve a target MPX power +3 dBr in all situations. Use only if regulation authorities apply such limitation of MPX power. Use 2009 new TURBO processing structure.

**Sample +3 dBr MPX Power** - classical 2007 preset, sound processed to achieve a target MPX power +3 dBr in all situations. Use only if regulation authority applies such limitation of MPX power. Use 2007 classical structure.

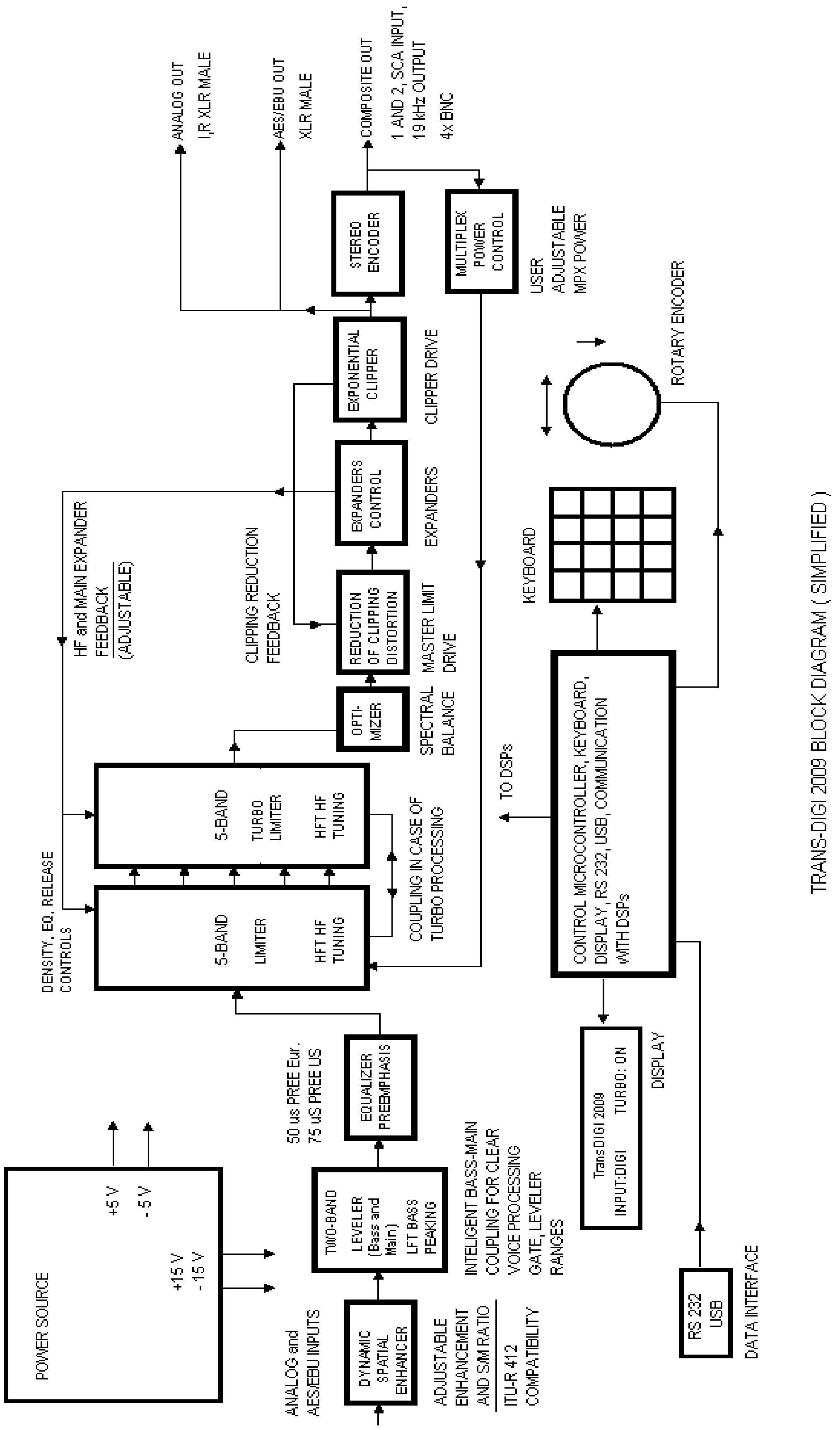
**HOT AC !!** – new 2009 preset, sounding **close to Omnia 6 EX HotAC** preset. Natural sound with aproximately. 6,5 dBr of MPX Power

**HOT City !!** - new 2009 preset, loud, clear, naturally processed, with excellent highs and bass punch.
**Sound is similar to Omnia 6 EX HOT** presets. MPX Power aprox. 6,8 dBr.

**TURBO !! Nova** - new 2009 TURBO preset, soft, clear, open sound, without hearable artefacts of processing and with lot of mid range. OK for small radio**.**
MPX Power aprox. 6,5 dBr.

**TURBO !! Sprinkle** - new 2009 TURBO preset, soft, clear, open sound, without hearable artefacts of processing. OK for universal or **very clear pop, rock or disco** format. **Similar to ORBAN 8400/8500 sound**
MPX Power aprox. 6,5 dBr.

## Block diagram of Trans-DIGI 2009 (simplified)



# Product warranty

## Warranty:

Warranty 2 years from the date of purchasing. Warranty does not cover mechanical damage or damage from electrostatic discharge (electrical storm....)

## Manufacturer:

Phobos Engineering s.r.o.
Podnikatelská 565
Praha 9 – Běchovice
190 11

www.phobosaudio.cz

Czech Republic (CR)