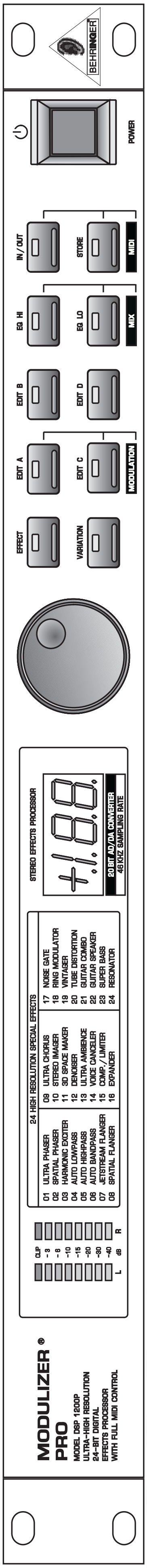
MODULIZER® PRO DSP1200P



User´s Manual

Bedienungsanleitung

E **D**

Version 1.0 November 1998

BEHRINGER®

www.behringer.de

C E

EG-Declaration of Conformity

BEHRINGER®

Spezielle Studiotechnik GmbH

acc. to the Directives 89/336/EWG and 73/23/EWG

We, **BEHRINGER INTERNATIONAL GmbH**
Hanns-Martin-Schleyer-Straße 36-38
D - 47877 Willich

Name and address of the manufacturer or the introducer of the product on the market who is established in the EC

herewith take the sole responsibility to confirm that the product:

MODULIZER PRO DSP1200P

Type designation and, if applicable, Article-N°

which refers to this declaration, is in accordance with the following standards or standardized documents:

☒ **EN 60065**	☒ **EN 61000-3-2**
☒ **EN 55020**	☒ **EN 61000-3-3**
☒ **EN 55013**	☒ **EN 55022**

The following operation conditions and installation arrangements have to be presumed:

acc. to Operating Manual

BEHRINGER
INTERNATIONAL GmbH
Hanns-Martin-Schleyer-Str. 36-38
D-47877 Willich-Münchheide II
Tel.-Nr. 0 21 54 / 92 06-0
Fax-Nr. 0 21 54 / 92 06-30

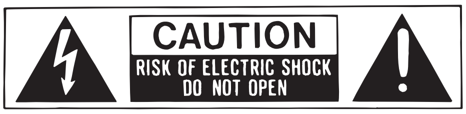
B. Nier, President Willich, 01.11.1998

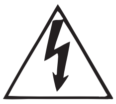
Name, address, date and legally binding signature of the person responsible

SAFETY INSTRUCTIONS

CAUTION: To reduce the risk of electrical shock, do not remove the cover (or back). No user serviceable parts inside; refer servicing to qualified personnel.

WARNING: To reduce the risk of fire or electrical shock, do not expose this appliance to rain or moisture.





This symbol, wherever it appears, alerts you to the presence of uninsulated dangerous voltage inside the enclosure - voltage that may be sufficient to constitute a risk of shock.

This symbol, wherever it appears, alerts you to important operating and maintenance instructions in the accompanying literature. Read the manual.

DETAILED SAFETY INSTRUCTIONS:
All the safety and operation instructions should be read before the appliance is operated.

Retain Instructions:
The safety and operating instructions should be retained for future reference.

Heed Warnings:
All warnings on the appliance and in the operating instructions should be adhered to.

Follow instructions:
All operation and user instructions should be followed.

Water and Moisture:
The appliance should not be used near water (e.g. near a bathtub, washbowl, kitchen sink, laundry tub, in a wet basement, or near a swimming pool etc.).

Ventilation:
The appliance should be situated so that its location or position does not interfere with its proper ventilation. For example, the appliance should not be situated on a bed, sofa rug, or similar surface that may block the ventilation openings, or placed in a built-in installation, such as a bookcase or cabinet that may impede the flow of air through the ventilation openings.

Heat:
The appliance should be situated away from heat sources such as radiators, heat registers, stoves, or other appliance (including amplifiers) that produce heat.

Power Source:
The appliance should be connected to a power supply only of the type described in the operating instructions or as marked on the appliance.

Grounding or Polarization:
Precautions should be taken so that the grounding or polarization means of an appliance is not defeated.

Power-Cord Protection:
Power supply cords should be routed so that they are not likely to be walked on or pinched by items placed upon or against them, paying particular attention to cords and plugs, convenience receptacles and the point where they exit from the appliance.

Cleaning:
The appliance should be cleaned only as recommended by the manufacturer.

Non-use Periods:
The power cord of the appliance should be unplugged from the outlet when left unused for a long period of time.

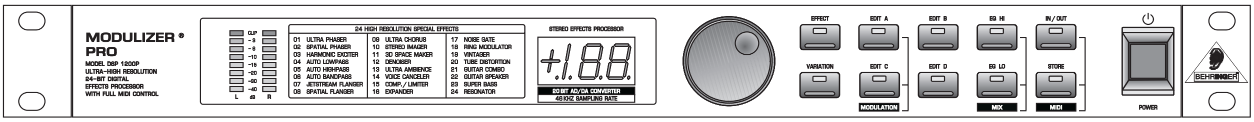
Object and Liquid Entry:
Care should be taken so that objects do not fall and liquids are not spilled into the enclosure through openings.

Damage Requiring Service:
The appliance should be serviced by qualified service personnel when:
- The power supply cord or the plug has been damaged; or
- Objects have fallen, or liquid has been spilled into the appliance; or
- The appliance has been exposed to rain; or
- The appliance does not appear to operate normally or exhibits a marked change in performance; or
- The appliance has been dropped, or the enclosure damaged.

Servicing:
The user should not attempt to service the appliance beyond that is described in the Operating Instructions. All other servicing should be referred to qualified service personnel.

MODULIZER PRO

Ultra-high performance Digital Multi-Effects Processor powered by a 24-bit high-speed Digital Signal Processor (DSP)



▲ Cutting edge effects such as 3D Processor, Stereo Imager, Lo-Fi, Super Bass, Ring Modulator, Voice Canceler etc.

▲ 24 breathtaking effects such as Ultra Phaser, Harmonic Exciter, Auto Filter, Jetstream Flanger, Ultra Compressor, Guitar Combo, Speaker Emulation Resonator and more

▲ Hundreds of effect variations plus two individual parameters and separate low and high EQ section

▲ True stereo processing performance allows separation of left and right channels in the stereo field for open-sounding enhancement of the sound sources

▲ Free MODULIZER software allows for total remote control via PC (download at www.behringer.de)

▲ 20-bit A/D and D/A converters with 64/128 times oversampling for ultra-high headroom and resolution

▲ Internal 24-bit processing with professional 46 kHz sampling rate

▲ Servo-balanced Inputs and Outputs on gold plated XLR and TRS jack connectors for high signal integrity

▲ 100 user preset memories to store programs for instant recall

▲ Accurate eight-segment LED level meters simplify level setting for optimum performance

▲ ”Future-proof” software-upgradeable architecture

▲ Full MIDI capability allows real-time parameter control and program selection

▲ High-quality components and exceptionally rugged construction ensures long life and durability

▲ Internal power supply design for professional application

▲ Manufactured under the ISO 9000 management system

DSP1200P

FOREWORD

Dear Customer,

Welcome to the team of MODULIZER PRO users and thank you very much for expressing your confidence in BEHRINGER products by purchasing this unit.

It is one of my most pleasant tasks to write this letter to you, because it is the culmination of many months of hard work delivered by our engineering team to reach a very ambitious goal: making an outstanding device that will become a standard tool used by studios and P.A. companies. The task to design the MODULIZER PRO certainly meant a great deal of responsibility, which we assumed by focusing on you, the discerning user and musician. It also meant a lot of work and night shifts to accomplish this goal. But it was fun, too. Developing a product usually brings a lot of people together, and what a great feeling it is when everybody who participated in such a project can be proud of what we've achieved.

It is our philosophy to share our joy with you, because you are the most important member of the BEHRINGER family. With your highly competent suggestions for new products you've greatly contributed to shaping our company and making it successful. In return, we guarantee you uncompromising quality (manufactured under ISO9000 certified management system) as well as excellent technical and audio properties at an extremely favorable price. All of this will enable you to fully unfold your creativity without being hampered by budget constraints.

We are often asked how we can make it to produce such high-grade devices at such unbelievably low prices. The answer is quite simple: it's you, our customers! Many satisfied customers means large sales volumes enabling us to get better conditions of purchase for components, etc. Isn't it only fair to pass this benefit back to you? Because we know that your success is our success, too!

I would like to thank all people whose help on “Project MODULIZER PRO“ has made it all possible. Everybody has made very personal contributions, starting from the designers of the unit via the many staff members in our company to you, the user of BEHRINGER products.

My friends, it's been worth the trouble!

Thank you very much,

Uli Behringer

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1. INTRODUCTION

With the BEHRINGER MODULIZER PRO you have acquired an extremely powerful and versatile multi-effects processor, which besides first-class modulation effects is equipped with many other algorithms. Despite the large number of effects, variations, and editable parameters, the DSP1200P is easily and intuitively operated owing to its logically structured layout.

Filters have been enjoying a renaissance during the past few years and not only in dance music. We have implemented a variety of resonant filter types in the MODULIZER PRO. For example, you can use amplitude-controlled and LFO filters giving your sounds an interesting color. Since each filter parameter is MIDI-controllable in manual mode, you can even record song-related filter changes in a sequencer program and play them back later on. So, even timing-based filter settings can be achieved.

So-called lo-fi effects simulating, for example click sounds produced by vinyl records or the noise of older tape recordings, are particularly suitable for techno, house and hip-hop productions. »Space« effects can be produced with the Ring Modulator which also doubles as a separate sound generator.

For guitars, we combined specific distortion and preamp variants with speaker simulations that enable you to create an excellent sound even without actually using a speaker cabinet during the recording session. But the MODULIZER PRO also functions as a multi-effect processor for guitar combo or rack-mount amps producing, for example phaser and various wah-wah and auto-wah effects.

To give you direct access to all modifiable parameters, the MODULIZER PRO has four edit control elements next to the VARIATION button that allow for editing these parameters directly and intuitively. Having selected a parameter you can change its value with the jog wheel. Of course, all effect parameters are MIDI-controllable in real time.

Additionally, the DSP1200P includes such “hip” algorithms as lo-fi, tube distortion and ring modulation. Please read this manual carefully to be able to fully exploit the effects and features of the innovative BEHRINGER effect algorithms implemented in the MODULIZER PRO.

☞ **The MODULIZER PRO can distort signals to a great extent. When you are scrolling through the presets, you may find programs with considerably differing output levels. Always reduce the volume of subsequent devices (e.g. amps, speakers, etc.) to a minimum level to protect these devices against possible damage.**

Despite the enormous and compute-intensive work to be done in the DSP1200P by a “dual-engine” 24-bit processor, the MODULIZER PRO can be operated easily and conveniently. All parameter changes can be made with the jog wheel (rotary control). 100 memory locations are available.

A very special feature are the high-low filters which can be edited freely. They are available for direct selection in each preset. Use the filters to adapt the “sound” of your presets to the room acoustics, which is particularly useful in live situations when every second counts.

☞ **The following operational manual will introduce you to the BEHRINGER MODULIZER PRO and its various functions. After reading the manual carefully, make sure it is always on hand for future reference.**

1.1 The design concept

The MODULIZER PRO is a mighty tool for the processing of audio signals. You can use it like a real musical instrument and owing to the direct access to all major parameters, which are also controllable via MIDI, your imagination is the only limit. We recommend that you read the descriptions of all effects thoroughly, so that you can make efficient use of the MODULIZER PRO’s vast sound potential. Or to use a wise old saying from the studio scene: “The best device is only as good as the person using it!”

With its clearly structured user interface the DSP1200P invites you to try and experiment with the preset effects programs which we programmed very carefully and right down to the last detail. Select them simply by rotating the encoder (jog wheel). Once selected, a preset is activated only after a pause of two seconds (i.e. when the point in the display disappears).

☞ **Presets selected via MIDI, however, are activated without delay!**

The MODULIZER PRO not only boasts a logic and straightforward user interface, its technical features too are quite impressive. Pro-level signal processing is ensured by the following components:

▲ Extremely low-noise and high-precision 20-bit AD/DA converters.

▲ A professional 46 kHz sampling rate guarantees high signal resolution with a frequency response of 20 Hz through 20 kHz.

▲ The 24-bit processor provides lots of computing power (dual-engine software) for real-time effect modulation.

▲ Like all BEHRINGER products, the MODULIZER PRO uses exclusive top-quality components and circuits.

With its complete MIDI implementation the DSP1200P can be integrated in any MIDI system. A MIDI software editor will soon be available and enables you to program the MODULIZER PRO from your personal computer, and the MIDI interface allows for transmitting data from the DSP1200P and store them on an external storage medium. For example, you can use sys-ex dumps to send all presets and settings to your sequencer program and reload them from there whenever you want.

The philosophy behind BEHRINGER products guarantees a no-compromise circuit design and employs the best choice of components. Top-quality 20-bit AD/DA converters which belong to the best components available owing to its outstanding specifications and excellent sonic characteristics. A 24-bit DSP is used as the heart of the MODULIZER PRO. It performs the precise calculations needed for the processing of the complex algorithms. Additionally, the MODULIZER PRO uses metal-film resistors and capacitors with very tight tolerances, high-grade switches, low-noise operational amplifiers (type 4580) as well other selected components.

The MODULIZER PRO DSP1200P uses SMD technology (Surface Mounted Device). These subminiature components adapted from aerospace technology allow for an extreme packing density to further improve the overall reliability. Additionally, the unit is manufactured in compliance with the ISO9000 certified management system.

1.2 Before you begin

Your BEHRINGER MODULIZER PRO was carefully packed in the factory and the packaging was designed to protect the unit from rough handling. Nevertheless, we recommend that you carefully examine the packaging and its contents for any signs of physical damage, which may have occurred in transit.

☞ **If the unit is damaged, please do not return it to us, but notify your dealer and the shipping company immediately, otherwise claims for damage or replacement may not be granted. Shipping claims must be made by the consignee.**

The BEHRINGER MODULIZER PRO fits into one standard 19" rack unit of space (1 3/4"). Please allow at least an additional 4" depth for the connectors on the back panel.

☞ **Be sure that there is enough space around the unit for cooling and please do not place the MODULIZER PRO on high temperature devices such as power amplifiers etc. to avoid overheating.**

The mains connection of the MODULIZER PRO is made by using a mains cable and a standard IEC receptacle. It meets all of the international safety certification requirements. Please make sure that all units have a proper ground connection.

☞ **Before you connect your MODULIZER PRO to the mains, please make sure that your local voltage matches the voltage required by the unit! (see chapter 5 for details)**

☞ **Please ensure that only qualified persons install and operate the MODULIZER PRO. During installation and operation the user must have sufficient electrical contact to earth. Electrostatic charges might affect the operation of the MODULIZER PRO!**

As a standard the audio inputs and outputs on the BEHRINGER MODULIZER PRO are fully balanced. If possible, connect the unit to other devices in a balanced configuration to allow for maximum interference immunity. The automatic servo function detects unbalanced connections and compensates the level difference automatically (6 dB correction).

The MIDI links (IN/OUT/THRU) are made over standardized DIN patch cords. The data communication is isolated from ground by opto couplers.

1.3 Control elements

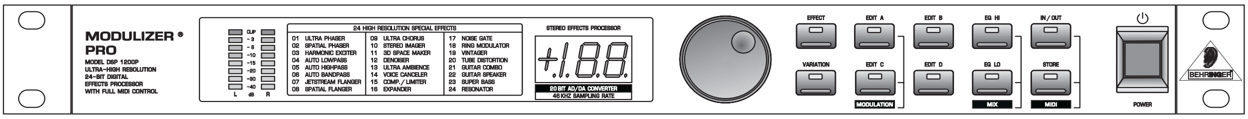


Fig. 1.1: MODULIZER PRO front panel

The BEHRINGER MODULIZER PRO is equipped with ten parameter keys, one jog wheel (rotary control), and an LED display. The two channels can be monitored with the 8-stage LED meters.

1.3.1 Front panel control elements

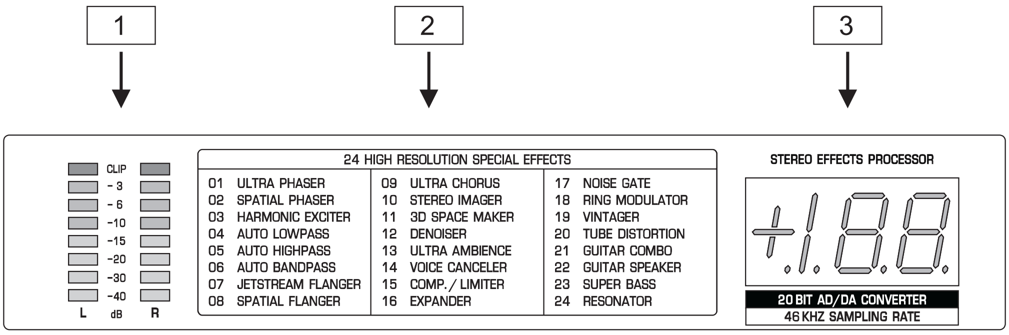


Fig. 1.2: Display section DSP1200P

1 The two *LED* chains read the input signal level in dB, referenced to the internal digital maximum.

☞ **Please note that the nominal level of the MODULIZER PRO can be selected with the +4 dBu / -10 dBV switch located on the back panel.**

2 The *effect table* gives you an overview of the 24 different effect algorithms.

3 After power-up, the *LED* display reads the number of the preset last used. This clearly legible, 2½ digit numeric display has plus/minus indicators to show that parameters are being incremented or decremented in Edit mode.

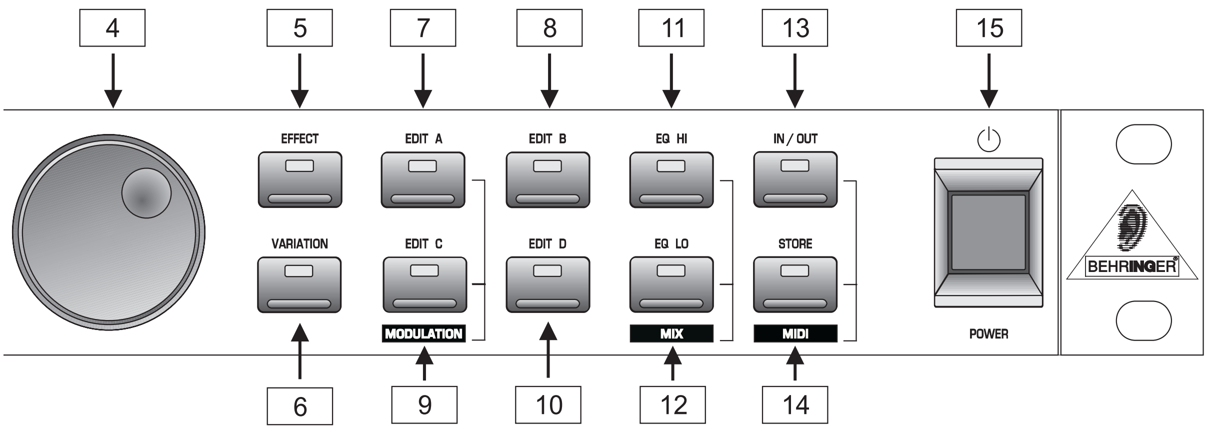


Fig. 1.3: Function keys and jog wheel

4 With the *jog wheel*, a continuous rotary control, you can freely edit the selected parameters. Turn the wheel clockwise to increase the values, or counterclockwise to reduce them.

☞ **As long as none of the edit functions to the right of the jog wheel has been selected, you can use the wheel to select a program directly, which is shown by a dot lighting up in the display. While this dot is on, you can select a program though its settings will not take immediate effect. When the jog wheel has not been touched for one second, the LED in the display disappears and the program is loaded.**

5 Use the *EFFECT* key to directly select one of the 24 basic effect algorithms with the jog wheel.

☞ **Whenever a new algorithm is selected, all parameters are reset to default values.**

6 The *VARIATION* key allows you to select the most important parameter of each algorithm, for example LFO speed for algorithm 1, i.e. a phaser then can be modified with the jog wheel.

7 In each preset you can edit at least three and mostly four parameters in addition to the preset VARIATION. Use the *EDIT A* key to select the first parameter. The exact parameter assignment can be seen from the parameter list printed on the enclosure top and in the appendix.

8 Use the *EDIT B* key to select another parameter which is to be altered.

9 With the *EDIT C* key you can select the third parameter.

10 The *EDIT D* key allows you to modify the fourth parameter if one is given.

☞ **With modulation effects, when the LFO is set to zero, the modulation stops and can be set manually or via MIDI. To adjust it manually press EDIT A and EDIT C simultaneously. To control it via MIDI you can use controller # 56. If control send is activated, the MODULIZER PRO sends the actual LFO state, again using controller # 56. When the LFO is started again it begins on that same value. This applies to all effect algorithms in which a LFO is used, except the Ultra Chorus**.

11 To give your programs the finishing touch, the MODULIZER PRO incorporates two filters. Use the *EQ HI* key to raise or lower the high-frequency portions of the effect program.

12 The *EQ LO* key activates a filter which processes the low-frequency portions of your preset. Pressing both EQ LO and EQ HI will activate the Mix-Mode (See next paragraph).

13 The *IN/OUT* key enables you to bypass the DSP1200P. The green LED lights up as soon as the MODULIZER PRO is activated. Depending on the Mix mode adjusted, this key can also be used to activate the Mute function. Additionally, the green LED starts flashing whenever MIDI data is being received.

14 Use the *STORE* key to save the edited program to a user preset as shown in the display. 100 user presets are available on the DSP1200P. Press the key once to select a memory location (number), then press it again to store the preset. Pressing both *IN/OUT* and *STORE* will put the MODULIZER PRO in MIDI mode (see next paragraph).

15 Use the *POWER* switch to switch the MODULIZER PRO on or off.

1.3.2 Key combinations

To protect the DSP1200P against user errors, three important edit commands have been implemented as a series of key combinations. For example, in normal operating modes, the presets cannot be reset to their factory defaults, so as to keep your own programs safe. Please proceed as follows to reinitialize the preset default settings:

▲ **Press and keep the keys EFFECT and STORE *before powering up* the MODULIZER PRO. Then switch on the DSP1200P and keep the two keys pressed for about one second. The programs are counted up and reset to their original default settings.**

The MODULIZER PRO provides two methods to mix the input and the effect signals (*External Mix* and *Internal Mix* mode). Select External Mix mode to use the DSP1200P with a mixing console: in this mode all presets are set to 100% effect intensity, i.e. you can use the aux return busses of your console to add the processed signal to the original signal. In External Mix mode the IN/OUT key is used to bypass the unit. Here’s how to enter Mix Extern mode:

▲ **With the unit switched on, press the Mix mode key combination, i.e. the keys EQ LO and EQ HI.**

The MODULIZER PRO enters Mix mode. When the display reads two dashes, the DSP1200P is in External Mix mode, and when a figure is read, Internal Mix mode is selected. To toggle between the two modes, simply press both EQ keys for about 1 second.

In Internal Mix mode you can use the jog wheel to freely select the effect intensity in each preset within a range from 0% to 100%, a highly useful feature, for instance, to insert the DSP1200P in the effect loop of a guitar amp. Good results can be achieved with settings between 20% and 50%.

Another key combination can be used to enter MIDI mode. With the MODULIZER PRO switched on, proceed as follows:

▲ **Press and hold the keys IN/OUT and STORE for about two seconds, the DSP1200P automatically enters MIDI mode. Use the IN/OUT key to step through the various MIDI parameters. Press any other key to quit MIDI mode.**

1.3.3 Back panel

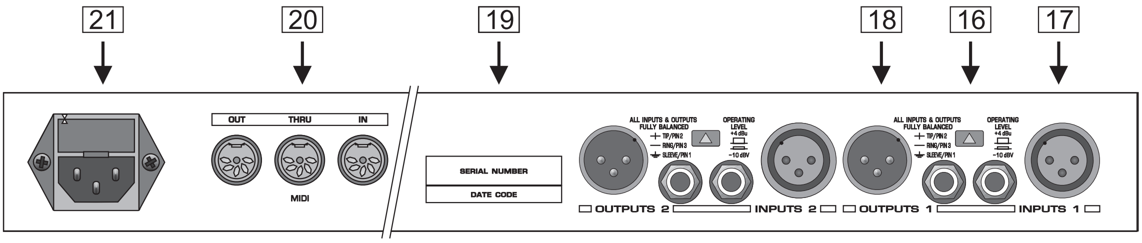


Fig. 1.4: Back panel connectors and control elements

16 Use the *OPERATING LEVEL* switch to adapt the MODULIZER PRO to different operating levels. You can select a -10 dBV semi-pro level used for home recording and a +4 dBu level used in professional studios. The level indicators on the front panel are automatically adapted to read the selected nominal level, i.e. an optimum operating range of the meters is always guaranteed.

17 These are the MODULIZER PRO's analog *INPUTS.* The MODULIZER PRO has both XLR and jack inputs and outputs. Each XLR and jack set are wired parallel and can be used either balanced and unbalanced.

18 These are the MODULIZER PRO's analog *OUTPUTS*. Also on balanced or unbalanced XLR or TRS jacks.

19 These are the MODULIZER PRO's MIDI connectors (*MIDI OUT / THROUGH / IN*). Via these connectors total remote control is possible.

20 Please take the time to make shure that the *SERIAL NUMBER* is printed correctly in the space provided on the enclosed Warranty Registration Card. Put the instruction manual in a safe place and return the completed Warranty Registration Card to us within 14 days of purchase, making sure that the dealer stamp has been acquired.

21 This is the *MAINS CONNECTOR / FUSE HOLDER / VOLTAGE SELECTOR.* Before you connect the unit, please make sure that the displayed voltage corresponds to your Mains supply. Please note that the AC voltage selection is defined by the position of the Fuse Holder. If you intend to change the operating voltage, remove the Fuse Holder and turn it by 180 degrees before you reinsert it. Matching the two markers monitors the selected voltage. Please note that, depending on the mains voltage supplied to the unit, the correct fuse type and rate must be installed (see chapter 6.5 "SPECIFICATIONS"). Please use the enclosed mains cable to connect the unit to the mains power supply.

☞ **Please note that not all appliances can be used with different mains voltage ratings. Please check the description on the back of the unit and the box.**

1.4 The effect algorithms

In a digital effects device all effect programs are based on algorithms computed by a Digital Signal Processor (DSP). How does this work? A DSP can perform an enormous number of binary computations in a minimum amount of time. The binary computations that are used to generate an effect as part of a program are determined by a so-called algorithm which represents a rule for computing numerical values that are exactly specified for each effect type. For example, tube distortion algorithms differ from chorus algorithms in their programming. Plainly speaking: each effect is based on a specific algorithm which processes the input signal (previously converted from analog to digital). All of this work is done by the DSP. Once the effect has been generated and added to the input signal, the digital music signal is converted back to analog by means of a D/A converter.

MODULATION FX

1 Ultra Phaser

A phaser is one of the MODULIZER PRO's classic modulation effects. It is quite popular for guitar sounds and keyboard pads, but was also extensively used during the 70's for other instruments, such as electric pianos.

From a technical point of view, a phaser is a modulation effect producing a multi-stage phase shift between the direct and effect signals, which results in a modulated comb filter effect. Depending on how they are set up, phasers can be used to slightly modulate or heavily distort the signal spectrum. Consequently, the sound they produce is a bit like that of a regularly modulated filter. Although most often used as a single-instrument effect, there are well-known examples when phasers are used on mix signals, so do not hesitate to experiment with it.

The parameters of the phaser:

VARIATION: "Speed": determines the LFO speed between 0 (see following **Note**) and 9.9 Hz.

EDIT A: "Intensity": adjusts the number of all-pass filters.

EDIT B: "Depth": is used to set the modulation depth.

EDIT C: "Feedback": determines how much of the output signal is fed back to the input.

☞ **When the LFO is set to zero, the modulation stops and can be set manually or via MIDI. To adjust it manually press EDIT A and EDIT B simultaneously. To control it via MIDI you can use controller # 56. If control send is activated, the MODULIZER PRO sends the actual LFO state, again using controller # 56. When the LFO is started again it begins on that same value. This applies to all effect algorithms in which a LFO is used, except the Ultra Chorus.**

2 Spatial Phaser

As its name implies, this is basically a phaser which produces an impressive stereo effect by giving the sound room and transparency. It is so to speak a high-tech version of the classical phasing effect. The editable parameters are identical to those of the phaser.

SPECIAL FX

3 Exciter

The *exciter* adds artificially produced harmonics to the original signal, increases its presence and loudness (the subjectively perceived volume), but does not raise the signal in level.

The parameters:

VARIATION: "HP-Shape": allows for raising the levels of the frequencies around the adjusted threshold.

EDIT A: "Tune": varies the frequency above which the exciter produces harmonics.

EDIT B: "Harmonics": this parameter enables you to adjust the intensity of the generated harmonics.

☞ **As the tonal results produced with exciters are initially quite impressive, a common mistake is to overdo the effect.**

☞ **We recommend that you make frequent A/B comparisons (IN/OUT) between the original and the processed signals. Rule of thumb: sound-enhancing effects should be "missed" when absent instead of directly audible. Remember, less is more.**

SPECIAL

E

4 **Auto Lowpass**

5 **Auto Highpass**

6 **Auto Bandpass**

Filters are used to modify the frequency response of a signal. Lowpass filters cut off high frequencies, while high-pass filters attenuate the low-frequency portions of a signal. A resonance parameter allows for raising the level of the frequencies around the cut-off frequency.

A band-pass filter provides both low and high-frequency attenuation, while the adjustable frequency band between the two cut-off points is "passed". The Q (=quality) parameter controls the bandwidth of the passband.

To avoid distortion, the output level is automatically reduced as set with the resonance parameter.

The parameters of the filter effects:

VARIATION: "Mode": lets you select the filter mode:

"L1": LFO-modulated 12 dB/oct. filter.

"L2": LFO-modulated 24 dB/oct. filter.

"A1": signal-envelope-modulated 12 dB/oct. filter.

"A2": signal-envelope-modulated 24 dB/oct. filter.

EDIT A: "Frequency": determines the filter's cut-off frequency.

EDIT B: "Resonance": adjusts the quality/resonance of the filter.

EDIT C: "Depth": controls the modulation depth applied to the cut-off frequency.

EDIT D: "Speed": controls the LFO speed between 0.004 and 9.9 Hz for modes L1 and L2, or the envelope follower speed for modes A1 and A2.

MODULATION

7 **Jetstream Flanger**

Originally, flanging effects were produced by two tape recorders synchronized to each other. Both machines played back the same signals (e.g. a guitar solo). By applying just a little bit of pressure to the **flange** of the supply reel of one tape recorder you can slow down the playback speed. The resulting delay produces phase shifts which in turn lead to a cancellation of certain frequencies, which is called <u>comb-filter effect</u>. Since the pressure applied to the reel is not constant, the canceled frequencies and thus the sound spectrum are varying continuously. In today's age of digital technology you don't have to use your fingers any longer. The MODULIZER PRO mixes the original signal with a delayed copy of the signal. The resulting cancellations are varying as the delay time is being modulated from an LFO. Depending on the maximum delay time and LFO frequency you can produce sometimes dramatic sound effects which become increasingly "metallic" as the effect intensity is raised. The sound resembles that of a flying jet plane (Jetstream). In addition to a positive feedback, the DSP1200P also allows you to add negative feedback which results in a "tube-like" sound.

The parameters:

VARIATION: "Speed": adjusts the LFO speed between 0.004 and 9.9 Hz.

EDIT A: "Delay": this parameter sets the maximum delay time (1-20 ms).

EDIT B: "Depth": adjusts the modulation depth.

EDIT C: "Feedback": controls the amount of effect signal that is fed back to the input.

EDIT D: “Bandlimit”: this parameter enables you to filter the feedback path, which is useful for feedback settings of up to 100%.

☞ **Tip: there is no limitation as to the range of instruments that can be processed with a flanging effect. Check out vocal sounds too, but remember that less is often more!**

8 Spatial Flanger

Includes an additional stereo effect giving the audio material more room acoustics information. The parameters are the same as the jetstream flanger.

9 Ultra Chorus

Think of a string quartet with each musician playing the same notes. As a matter of fact, however, musicians can never play exactly the same, which results in a sequence of slightly detuned and delayed signals. In a chorus, copies of the original signal are delayed by 20-40 ms, slightly detuned and modulated by an LFO. The MODULIZER PRO features no less than 8 voices, which corresponds to a group of 8 musicians playing simultaneously.

The parameters:

VARIATION: “Speed”: controls the LFO speed between 0.1 and 5 Hz.

EDIT A: “Delay”: this parameter adjusts the delay time (1-80 ms).

EDIT B: “Depth”: sets the modulation depth.

EDIT C: “Stereo Width”: positions the individual voices on the stereo basis.

EDIT D: “Wideness”: controls the effect width by detuning the individual voices against each other.

☞ **Tip: The Ultra Chorus widens the audio material, for example to enhance thin-texture keyboard pads, give vocals more room acoustics information and guitar sounds a dreamy character.**

10 Stereo Imager

This effect is used to process stereo mix signals, by splitting up the signal in middle and side information (MS matrix). Both components can be raised separately in level and positioned on the stereo basis. Additionally, the side signal can be shifted in phase to enhance the stereo effect even further.

The parameters:

VARIATION: “Crossover Frequency”: adjusts the phase shift onset.

EDIT A: “Gain”: allows for adjusting level corrections (-6 ... +6 dB).

EDIT B: “Spread”: sets the intensity of phase shift and thus the stereo width.

EDIT C: “Mono Pan”: positions the mono components on the stereo basis.

EDIT D: “Stereo Center”: positions the stereo components on the stereo basis.

☞ **With the MIX function the ratio of center vs. side information is controlled**

☞ **When applying increasing stereo width to heavily reverberated audio material, the reverb can sound unnatural and too intense.**

11 3D Spacemaker

Similar to the Stereo Imager, this algorithm processes the stereo components. Through the use of several psycho-acoustic phenomena, it also generates a kind of “envelope” around the room acoustics information and thus allows for 3D sound with only 2 speakers.

The parameters:

VARIATION: “In-Out”: selects the type of 3D function (with/without emphasis on middle signals).

EDIT A: “Gain”: allows for adjusting level corrections (-6 ... +6 dB).

EDIT B: “Spread”: controls the “depth” of the sound image.

EDIT C: “Crossover Frequency”: adjusts the 3D effect onset.

In this algorithm the MIX function controls the ratio of middle vs. side signals.

☞ **When you increase stereo width on heavily reverberated audio material, the reverb can sound unnatural and too intense.**

DYNAMICS

12 Denoiser

As their name implies, denoisers are used to eliminate or at least reduce noise and other interference. This algorithm also includes a noise-gate and an amplitude-controlled low-pass filter. Use this algorithm of the MODULIZER PRO to eliminate unwanted noise tails.

The parameters:

VARIATION: “Gate Threshold”: determines the threshold below which the gate cuts off the signal.

EDIT A: “Gate Hold”: determines the time after which the gate closes once the signal has fallen below threshold (50-1,000 ms).

EDIT B: “Gate Release”: the gate’s release time depends on the sound, e.g. percussive material requires much quicker release times than, for example, keyboard pads with long decays (1-800 ms).

EDIT C: “Lo Pass Frequency”: controls the cut-off frequency of the low-pass filter.

EDIT D: “Lo Pass Depth”: adjusts the amount by which the signal level influences the low-pass filter.

ROOMS

13 Ultra Ambience

Room reverb consists of early reflections followed by the actual reverb. Unlike the DSP1000P which emulates both components, this effect simulates only the first 15 early reflections. Since our hearing uses these reflections to determine the room size, they can be used to generate a subtle and impressive kind of compression of the audio material, without modifying the signal with long reverb tails. This effect is particularly suited for vocals and also for drum instruments.

The parameters:

VARIATION: “Pre-delay”: determines the predelay time between original signal and first reflections (0-200 ms).

EDIT A: “Size”: adjusts the room size.

EDIT B: “Wall Damp”: this parameter controls the damping factor of the “walls”.

EDIT C: “Stereo Width”: adjusts the stereo width of the early reflections.

EDIT D: “Reflections”: sets the number of early reflections (1-15).

☞ **Ambience is a very popular effect for vocal sounds, as it provides voices with a warm and rich sound and brings them to the foreground when mixing.**

SERIAL

14 Voice Canceler

This effect is particularly suited for karaoke or the rehearsal of lead vocals. The Voice Canceler eliminates mono (midrange) vocal tracks from stereo recordings. The low-frequency range, which in most cases is also recorded in mono, remains unaffected. Backing vocals which tend to be recorded in stereo will remain unaffected. Signals with a lot of stereo information (e.g. chorus effects etc.) will not be eliminated completely.

The parameters:

VARIATION: “Bass Frequency”: this parameter controls the maximum frequency of bass components to be retained in the mix signal.

EDIT A: “Gain”: adjusts the level of the output signal by +/-6 dB.

EDIT B: “Bass Pan”: adjusts the panorama of the low-frequency range.

EDIT C: “Treble Pan”: controls the stereo balance of the mix signal.

DYNAMICS

15 Compressor / Limiter

In broadcasting and recording studios signal levels often exceed the headroom of signal-processing devices and must therefore be reduced in their dynamics to avoid distortion. This is done with compressors or limiters. Although these devices perform similar functions they differ in one essential aspect:

Limiters limit signals abruptly above a certain threshold, while compressors provide a “smooth” control process over a wider range. The limiter monitors the signal continuously and reduces its dynamics as soon as it surpasses the threshold. Any signal level exceeding the threshold is immediately cut back to a safe value.

Compressors, too, monitor the program material and work with a threshold. However, they don’t control the signal abruptly as limiters do, but continuously. Once the signal has exceeded the threshold it is smoothly reduced in level, independently of the amount of excess level. The compressor side-chain implemented in the DSP1200P has a soft-knee characteristic.

The parameters:

VARIATION: “Ratio”: controls the ratio of input vs. output level for all signals surpassing the threshold. If set to maximum, the DSP1200P works as a limiter.

EDIT A: “Threshold”: adjusts the compressor threshold from -60 dB to 0 dB.

EDIT B: “Output gain”: this parameter allows you to raise or lower the output signal in level by max. 24 dB.

EDIT C: “Attack”: the attack control determines the time the compressor needs to respond to signals that are surpassing the threshold (5-200 ms).

EDIT D: “Release”: controls the time the compressor needs to restore the original level, once the signal has dropped below threshold (50-500 ms).

☞ **In all dynamics algorithms the Mix function is disabled: because a compressor processes the entire signal, any other operating mode would make no sense!**

16 Expander

Many audio signals are limited in their dynamics by nature. For example, recordings made outdoors usually suffer from a high level of background noise (traffic noise, wind, etc.). Guitar pick-ups, amplifiers, etc. can produce high noise levels or other sounds that inevitably limit the dynamic range of the wanted signal. Background noise of this kind is inaudible as long as the level of the processed signal is considerably higher than the noise floor and hence “masks” the interference noise.

Expanders are used to effectively enlarge the dynamic range of signals by attenuating signals with small amplitudes, which also reduces the background noise level.

The parameters:

VARIATION: “Ratio”: this parameter determines the ratio of input vs. output levels for all signals below threshold.

EDIT A: “Threshold”: adjusts the expander threshold within a range from -60 dB to 0 dB.

EDIT B: “Output Gain”: allows for raising/lowering the output signal by max. 24 dB.

EDIT C: “Attack”: controls the time the expander needs to respond to signals that are below threshold (5-200 ms).

EDIT D: “Release”: sets the time the expander needs to restore the original signal level (1:1) (50-500 ms).

☞ Similar to the compressor, there is no need for a Mix function here.

17 Gate

Noise-gates can be used in a variety of applications, both on stage and for miking instruments in the studio. For example, they can be used to suppress feedback (e.g. from vocal mics) or to fade out vocal signals (plus background noise) during pauses. In this case, the gate must reopen very quickly, so that subdued syllables can be heard. Noise-gates are often used to record or mix drum sets, so as to avoid possible crosstalk-induced phase problems.

The parameters:

VARIATION: “Threshold”: determines the threshold below which the gate cuts off the signal.

EDIT A: “Hold”: sets how long signals below threshold can still pass the gate, to allow for a smooth control process (1-1,000 ms).

EDIT B: “Range”: determines the degree of attenuation when the gate is closed. If set to maximum, the signal is faded out completely.

EDIT C: “Attack”: sets how fast the gate opens when the signal has surpassed the threshold (1-100 ms).

EDIT D: “Release”: determines how fast the gate closes after the hold time has expired (1-800 ms). The release time depends on the signal envelope, percussive sounds (short decay) need much quicker release times than, for instance, sustained keyboard pads.

☞ No Mix function.

18 Ring Modulator

This effect enables you to distort audio signals in their character. Similar to ultra-short-wave radio, the signal is multiplied with a carrier frequency, so that a frequency modulation (FM) is produced.

This effect is excellently suited for vocals (robot voice). If set to S0...7 the DSP1200 acts like a sound generator because now it uses a 1k sine generator (instead of the input signal) as its modulation source.

You can adjust the basic carrier frequency, LFO speed and modulation depth. The “bandlimit” parameter limits the frequency range of the effect signal, while “random” and “sine generator” use an additional 8-step slewing rate. Additionally, you can edit the basic carrier frequency, modulation depth and LFO speed.

The parameters:

VARIATION: “FM Modulation Mode”:

“L”: carrier frequency is modulated by LFO.

“E”: carrier frequency is modulated by the signal.

“R0...7”: carrier frequency is modulated by a random generator.

“S0...7”: 1k sine generator / carrier frequency is modulated by random generator.

EDIT A: “Frequency”: adjusts the carrier frequency.

EDIT B: “Speed”: controls the LFO speed (L), the envelope follower speed (A) or the slew rate of the random generator (R and S)*).

EDIT C: “Modulation depth”: adjusts...yes, the modulation depth.

EDIT D: “Bandlimit”: this is a subsequent low-pass filter that is used to cut off harsh high-frequency portions. If set to 0 the filter is bypassed, and as you are increasing the value, the amount of high-frequency portions in the output signal is being reduced.

☞ Caution! If used improperly, the Ring Modulator effect can damage your hearing or speakers, as it produces very high-frequency signal portions (particularly if the sine generator is used which works independently of the input signal)!

19 Vintager®

Digital technology has been trying for years to produce ever more high-quality, low-noise and brilliant sounds, but most recently more and more people have been going “back to the roots” looking for the warmth of “old” analog sounds. The techno/dance community swears by vinyl anyway, and many a music lover misses the flair of good old vinyl records and tape machines. The latest trend is called lo-fi.

We have taken this trends into account by creating the Vintager effect. Your recordings will sound like 8-bit material and produce all those clicks and noises you know from old records!

A TR-808/TR-909-like drum loop sounds really hot only when it’s fat and dirty!

The parameters:

VARIATION: “Clicks Level”: adjusts the level of clicks found on old vinyl records.

EDIT A: “Noise Level”: controls the noise intensity.

EDIT B: “Noise BP”: adjusts the sound color of the noise.

EDIT C: “Crack Level”: simulates cracks in the record and adjusts their volume.

EDIT D: “Hi Cut”: turn up this parameter to cut the brilliance of the audio material.

20 Tube Distortion

This effect simulates the sound of three different types of tubes. When you raise the input signal level (e.g. guitar) to overdrive an analog tube, as found in valve guitar amplifiers, harmonics are added to the original signal. As distortion increases (also called saturation), the original signal starts to distort and the guitar sound gets this typical rock music volume and freshness.

The parameters:

VARIATION: “Tube Type”: use this parameter to select the tube type.

EDIT A: “In Gain”: raises the input signal to reach the sound-modifying areas of the tube’s characteristic curve.

EDIT B: “Lo Cut”: controls a high-pass filter preceding the tube (high frequencies pass).

EDIT C: “Hi Cut”: controls a low-pass filter preceding the tube (low frequencies pass).

EDIT D: “Bandlimit”: controls a band-pass filter after the tube.

☞ **Try using the tube distortion effect with a kick drum. From dance to R&B — anything goes!**

21 Guitar Combo

This effect simulates the audio properties of a complete guitar amp. So, you can connect your bass/guitar to a preamp and then to the MODULIZER PRO, from where the signal is sent to a mixing console or recording machine. This algorithm simulates not only two tube stages but also the amp’s cabinet plus speaker.

The parameters:

VARIATION “Type”: controls the basic characteristics.

EDIT A: “In Gain”: raises the input signal to reach the sound-modifying areas of the tube’s characteristic curve.

EDIT B: “Drive”: controls the amount of distortion.

EDIT C: “Presence”: adjusts the sound’s presence by raising high-frequency components.

EDIT D: “Speaker”: this parameter simulates two types of speakers (incl. cabinet). If set to “0”, the “speaker” is bypassed.

22 Guitar Speaker

This effect simulates three different speaker types. Types 1 and 2 are typical guitar amp speakers, while type 3 represents a “multimedia” speaker. Additionally, you can use a parametric EQ to fine-tune the sound image.

The parameters:

VARIATION: “Speaker Type”: selects one of three speaker types.

EDIT A: “Peak Frequency”: controls the center frequency of the parametric equalizer.

EDIT B: “Peak Q”: adjusts the bandwidth of the parametric equalizer.

EDIT C: “Peak Gain”: sets the amount of boost or cut of the parametric equalizer.

EDIT D: “Hi Cut”: turn up this parameter to cut the brilliance of the audio material.

☞ **The multimedia speaker enables you to check your recordings for compatibility. Mix-downs should sound as transparent and pleasant with small speakers as they do with pro-level studio monitors. If you use high-grade studio speakers to mix your recordings, it may happen that, for example, the bass range loses the power it had in the studio when you play back the material on your ghetto blaster at home: often the smaller speakers simply cannot produce the same sound as the huge hi-end monitors in the studio.**

23 Super Bass

This is a completely new type of bass exciter effect. In contrast to usual bass exciters that add subharmonics, this exciter adds specific harmonics to the original signal to generate a psycho-acoustic effect of deep bass signals. The audience has the impression of hearing an additional subbass.

The algorithm of this effect takes advantage of the fact that the human sense of hearing is used to a natural succession of harmonics (i.e. fundamental frequency, octave, fifth, etc.) and even reconstructs fundamental tones that are not part of the original signal. The Super Bass effect creates bass harmonics on the low-end signals. The listener hears the original bass and these “natural harmonics” and perceives a fundamental frequency one octave lower than the original frequency. This effect is obtained without increasing the actual power output of the system significantly. Especially small loudspeaker systems benefit from this effect and can sound a lot “bigger” than they actually are.

A kick drum processed with the Super Bass effect gets the right punch to make itself heard in the mix-down. Bass processors are quite popular in dance/techno music, for example, you can give synthesizer bass lines much more power.

The parameters:

VARIATION: “Frequency”: adjusts the cutoff frequency of the crossover network.

EDIT A: “Density”: this parameter controls the density of the processed bass signal.

EDIT B: “Ratio”: determines how much the processed bass signal gets compressed.

EDIT C: “Bass Level”: controls the low-frequency response of the original signal (which can even be faded out completely).

24 Resonator

A resonator simulates a system that oscillates at one frequency only and hence amplifies this frequency.

The resonator implemented in the DSP1200P can be modulated in its resonance frequency, with positive and negative feedback of up to 100%. This effect is available in three different modes which can be adjusted with the VARIATION parameter:

VARIATION: “Resonator frequency mode”:

“L”: resonator frequency is modulated by the LFO.

“E”: resonator frequency is modulated by the signal amplitude.

“R0...7”: resonator frequency is modulated by the random generator (additional 8-step slew rate is available in random generator mode).

EDIT A: “Frequency”: controls the basic resonator frequency.

EDIT B: “Speed”: controls LFO speed*) in mode L, envelope follower speed in mode A and random oscillator speed in modes R0-R7.

EDIT C: “Depth”: determines the modulation depth.

EDIT D: “Feedback”: controls the resonance intensity.

2. OPERATION

2.1 Effects structure

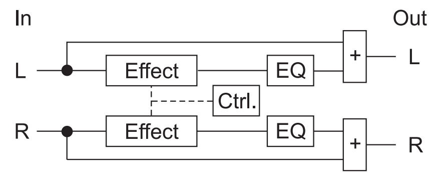
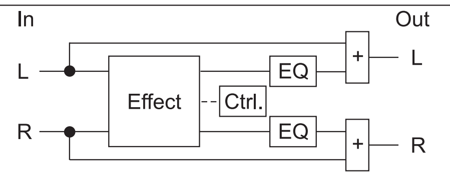
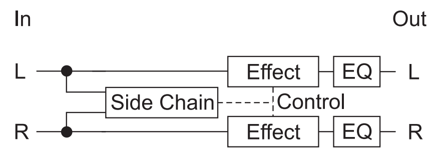
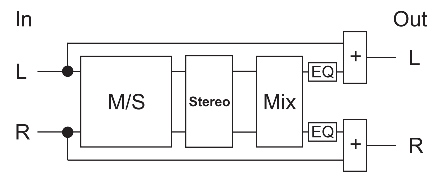
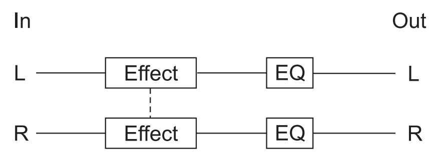
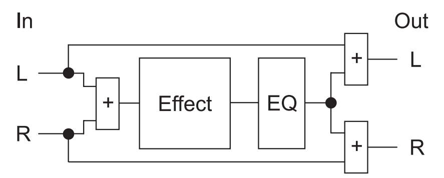
Algorithm Type	Effect Structure
Phaser, Exciter, Filter, Flanger, Ambience, Ring Modulator, Vintager, Tube Distortion, Reson.	
Chorus	
Denoiser, Compr/ Limiter, Expander, Gate	
Stereo Imager, 3D Spacemaker, Voice Canceler	
Guitar Speaker, Super Bass	
Guitar Combo	

Fig. 2.1: Effects structure

The following chapter will familiarize you with the operation of your BEHRINGER MODULIZER PRO. It will provide you with the basic understanding of how to obtain the desired results and should encourage you to experiment with the DSP1200P. Please bear in mind that the technical specifications of a unit may determine its limits, but the skill of the audio engineer determines how much use can be made from it. So do not hesitate to play with the unit, just like a musician does with an instrument, in order to explore the full functionality of your MODULIZER PRO.

2.2 Selecting presets

The MODULIZER PRO stores 100 user-definable presets. After power-up, the unit automatically recalls the preset last used. To select another preset, use the jog wheel to enter the preset number of your choice. Turn the wheel clockwise to increment the preset number, or counterclockwise to decrement it.

☞ **Please note that the MODULIZER PRO generally activates the newly selected presets only after about one second, which is indicated by a dot in the lower right corner of the display. After loading the data, the MODULIZER PRO enables the preset and the dot disappears. This brief interruption avoids the direct activation of every preset, as you scroll through the preset list with the jog wheel. Thus, the MODULIZER PRO makes sure that no "unwanted" programs are loaded unintentionally. Additionally, you can rotate the jog wheel at high speed and still have the time to specifically select the preset of your choice, instead of any of its "neighbors".**

E

2.3 Editing programs

Editing programs is easy on the MODULIZER PRO. Basically, all essential parameters can be selected directly via the keypad and edited with the jog wheel. The list to the left of the display summarizes the effect algorithms that the MODULIZER PRO can generate. Just press the EFFECT key to recall these basic algorithms and directly select them with the jog wheel. With the VARIATION key you can modify the selected effect in full detail, because each variation not only comprises one parameter but a set of several parameters. Thus, you can use the various variations to tailor the sound of an effects program to suit your specific needs. The EDIT A and B keys enable you to edit essential single parameters of the selected effects program, while the EQ LO and EQ HI keys allow for adapting your own presets to match specific room acoustics or sound preferences. Finally, you can also save the edits made to the preset.

2.4 Saving programs

Use the STORE key to save an edited preset. Basically, all parameter changes can be saved. Whenever you're editing a preset, the display starts flashing to indicate that the edits will be saved only when you confirm them by pressing the STORE key twice. Example:

▲ You recall a program for editing. Then you edit the preset as desired using the function keys and the jog wheel. During this process, the flashing STORE key reminds you that the preset settings have been changed but not saved yet. Press the STORE key once. The display reads the current preset number and starts flashing. To keep the original preset, use the jog wheel to select another preset that can be overwritten. Press the STORE key again to save the edits to the selected preset. If you wish to overwrite the original preset, simply press the STORE key twice (after editing) to save all changes you have made.

☞ **Whenever you have edited a preset and pressed the STORE key twice, all previous settings in this preset are erased and overwritten with the new parameter values. However, if you wish to keep the original preset, use the jog wheel to select another preset *before* you press the STORE key a second time.**

2.5 MIDI control

Use the MIDI key combination to select the MIDI parameters you wish to adjust. For this purpose press and keep the IN/OUT and the STORE keys for about two seconds. All parameters can be edited with the jog wheel and the IN/OUT key. The MIDI menu includes five pages which you can select by pressing the IN/OUT key several times.

On the first page you can select the MIDI channel. The display reads a small “c” (= channel). The jog wheel adjusts a channel from 1 through 16. To switch off the MIDI function simply select the “0” value (displayed as “-”).

On the second page you can select MIDI Omni mode, i.e. the unit transmits/receives on all 16 MIDI channels. The display reads “O” (=Omni). Use the jog wheel to activate (“1”) or deactivate (“0”) Omni mode.

The third page allows for configuring controller commands. On its right-hand side, the display reads a capital “C” (=Controller). The jog wheel selects one of the following four controller modes:

Display	Mode
0	No controller data is transmitted
1	Controller data is received but not transmitted
2	Controller data is transmitted but not received
3	Controller data is transmitted and received

Tab. 2.1: Controller settings

The fourth page gives you access to the program change setup. The display reads a capital "P" (=Program). Here, too, four modes can be selected with the jog wheel, as follows:

Display	Mode
0	Program changes are not transmitted
1	Program changes are received but not transmitted
2	Program changes are transmitted but not received
3	Program changes are transmitted and received

Tab. 2.2: Program change settings

The fifth page of the MIDI menu shows the "store enable" flag represented by a capital "S" in the display. The value "0" disables the reception of controller #60, and therefore protects the user presets from being modified via MIDI. Accordingly, the value "1" enables MIDI controller #60 so that you can modify or replace presets with a remote MIDI device or a sequencer. In this case the actual settings will be stored directly to the location that corresponds to the controller value.

☞ **Attention! Since the "store enable" mode allows you to access memory locations directly via MIDI, it is possible that stored presets will be replaced or altered if controller #60 messages are sent on the same MIDI channel. The purpose of this mode is to facilitate MIDI backup and restore operations without express confirmation at the MODULIZER PRO. It is therefore recommended to disable (flag=0) this mode as soon as the intended data transfer has ended. This is done automatically when you switch off the MODULIZER PRO.**

On the sixth, and presently the last, page you can access the "System Exclusive" functions. This is indicated by a "d" (for dump) in the display. To the left of this "d" a number is displayed:

- 0 means that no SYSEX data will be sent or accepted.
- 1 will enable the MODULIZER PRO to receive data. When STORE is pressed the unit will wait for data, this is shown by flashing dots (LEDs) in the display. The MIDI button LED flashes signaling that SYSEX data is being received.
- 2 will enable the MODULIZER PRO to send a "bulk dump". Start your sequencer and press STORE on the DSP1200 to start the transmission.

To load these settings again, select 1, press STORE and start your sequencer. If you press IN/OUT again, you will leave the MIDI setup. You can at all times press any other key to leave the MIDI setup directly.

If you press the IN/OUT key again on the sixth page, the MODULIZER PRO quits MIDI setup mode.

☞ **During a bulk dump all audio functions of the MODULIZER PRO will be deactivated.**

The full-featured MIDI implementation of the MODULIZER PRO allows for easily integrating the MODULIZER PRO into any MIDI system.

▲ MIDI IN

Any MIDI data sent to the MODULIZER PRO (sequencer, MIDI footswitch, etc.) is received via the MIDI IN jack. For example, when you wish to use the MODULIZER PRO as an effects devices for your guitar rack, you can connect the MIDI IN jack to a MIDI footswitch that allows for selecting program presets. If your rack includes another MIDI effects devices (e.g. a multi-effects processor), the data sent from the MIDI footswitch can be routed via the MODULIZER PRO's MIDI THRU jack to your multi-effects processor.

▲ MIDI THRU

The MIDI THRU jack is used to loop through incoming MIDI data, i.e. any control data received at the MIDI IN of the MODULIZER PRO can be transmitted via the MIDI THRU jack to other MIDI devices/instruments.

▲ MIDI OUT

The MIDI OUT jack allows for transmitting MIDI data that originate from the MODULIZER PRO. We are currently developing a software editor which will allow for storing single items of the MODULIZER PRO's internal data on an external medium, by using controller commands. Thus, it will be possible to archive MODULIZER PRO settings and presets on a computer, sequencer or MIDI data recorder. Both MIDI Control Change and MIDI Program Change commands will be transmitted when you edit or recall filter settings. Detailed information on this future control are available from our BEHRINGER hotline (+49(0)2154-920666), our international distributors and/or our Internet homepage http://www.behringer.de.

E

2.5.1 "Modulation"-controller

It is even possible to modulate some of the LFO effects manually or via MIDI. You can access the modulation value pressing EDIT A and EDIT C at the same time while LFO speed is set to zero. The modulation status can now be controlled with the jog wheel; i.e. you can set and change the delay used by the MODULIZER PRO manually to create a Doppler effect. If you try to change the modulation parameter while the LFO speed is not zero the display will show a dot in the lower right corner signaling that no changes can be made.

☞ **With this function you can also determine the starting point for a modulation effect. When the LFO is started again it begins on the value set by the jog wheel.**

You can also control the modulation via MIDI by a sequencer for instance. To control it via MIDI you can use controller # 56. If control send is activated, the MODULIZER PRO sends the actual LFO state, again using controller # 56. As with manual control, the MODULIZER PRO will only accept controller values if the LFO speed is set to zero. This applies to all effect algorithms in which a LFO is used, except the Ultra Chorus**.**

3. APPLICATIONS

The BEHRINGER MODULIZER PRO is a highly flexible device that can be used for a wide variety of applications. Prior to a presentation of the MODULIZER PRO's many uses, please note the following remarks on how to set signal levels correctly.

3.1 Level setting

Take care to set levels properly on the MODULIZER PRO! Low levels deteriorate the dynamics of the music signal, which results in a poor, weak and noisy sound. On the other hand, excess levels overdriving the converters in the MODULIZER PRO should also be avoided. Digital distortion is (unlike its analog counterpart) very unpleasant to hear as it does not occur gradually but abruptly.

Use the input level meter of the MODULIZER PRO to adjust the input signal to about -10 dB. Make sure that the CLIP LEDs never light up!

3.2 Using the MODULIZER PRO in the aux bus

By using the MODULIZER PRO in an aux bus of your mixing console you can feed the channel signals of one, several or even all console channels into the MODULIZER PRO, i.e. for each channel you can use the aux busses to separately determine the reverb levels of, for instance, various drum sounds: while lots of reverb is applied to the snare drum, the effect intensity could be reduced in the channels assigned to the tom-toms. To use the MODULIZER PRO in the aux bus, the unit must be wired as follows:

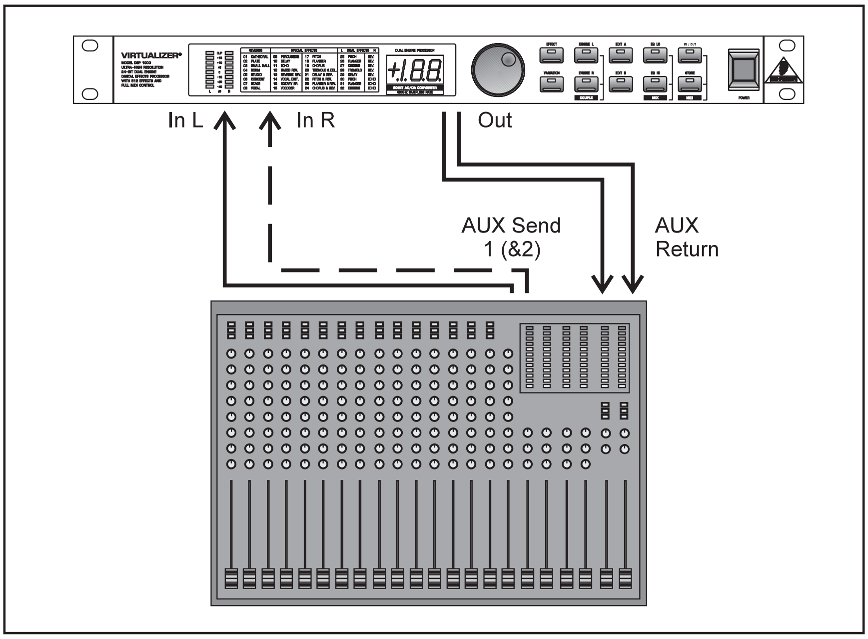


Fig. 3.1: Wiring aux busses

Connect the two Input phone jacks with the Aux Send outputs of the mixing console, and the Output jacks of the MODULIZER PRO with the Aux Return inputs of the console. If you want to use the BEHRINGER MODULIZER PRO in mono or dual mode, connect one audio channel (left or right) to one aux bus.

☞ **Turn back the volume on your amplifier to protect your equipment against damage. All devices you wish to interconnect should remain switched off until they are wired correctly.**

Let's suppose you wish to use the MODULIZER PRO in a live application, interfaced with the f.o.h. mixer, to enhance the guitar sound with a subtle chorus effect.

▲ Connect the MODULIZER PRO to the aux bus of your mixing console (fig. 3.1). Connect the units to the mains and adapt the operating level(s) if necessary. Switch on the MODULIZER PRO and set the levels appropriately (see 3.1). Press the Mix combination to make sure that the unit is set to Mix-Extern mode. Press the EFFECT key and use the jog wheel to select and thus activate the chorus effect (#19). Slowly turn up the aux bus level until the effect portion added to the guitar signal suits your needs. Subsequently, you can make all necessary fine-adjustments. We assume that you wish to edit the modulation frequency of the chorus effect: press the VARIATION key and set the modulation frequency with the jog wheel. To set the modulation delay, press the EDIT-A key, while the modulation depth of the chorus effect can be set by pressing the EDIT-B key. Having edited all parameters as desired, you can save the edits to the original (or any other) preset.

3.3 Using the MODULIZER PRO in the insert path

Basically, you can also insert the MODULIZER PRO in a channel, master output or subgroup of your mixing console. Use a dedicated insert cable. Inserting the MODULIZER PRO in a single channel will be useful only if you wish to process a specific signal (e.g. vocals) with the MODULIZER PRO, or if any other insert facilities of your mixing console are already in use.

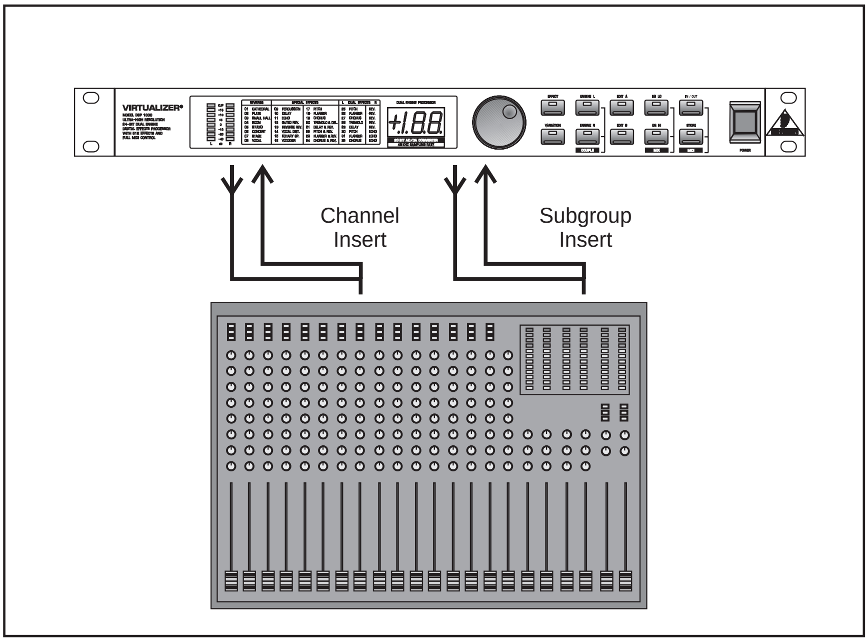


Fig. 3.2: Wiring the MODULIZER PRO in the insert path

With certain special effects, for example the limiter, you should insert the MODULIZER PRO rather than connecting it to the aux bus. Use a dedicated insert cable to connect the left audio channel with a channel insert on your mixing console (fig. 3.2).

3.4 Using the MODULIZER PRO as an effects device for instruments

With its extensive MIDI implementation the MODULIZER PRO can also be used, for instance, as a multi-effects device in a guitar rack. Of course, you can wire it both in stereo and mono.

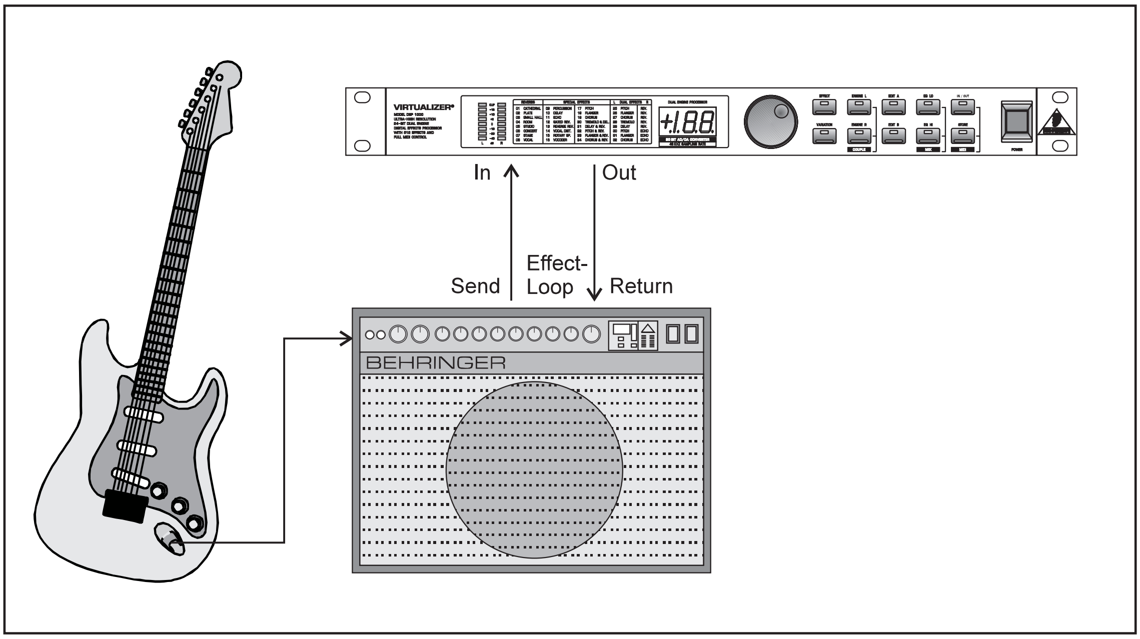


Fig. 3.3: Connecting the MODULIZER PRO to a guitar amp (send/return-mono)

The following hints illustrate the MODULIZER PRO's versatility if used with a guitar amp. Basically, the MODULIZER PRO should be inserted between the preamp and the power stage. Almost all guitar amps have an insert or effect loop to send the preamp signal of the guitar amp to the audio inputs of the MODULIZER PRO. The MODULIZER PRO processes the preamp signal and sends it back via the guitar amp's return bus (power amp in), from where it is routed to the power stage. When you use a stereo rack system for amplification, you can wire the MODULIZER PRO in stereo. Connect the preamp to the audio inputs of the MODULIZER PRO, and the audio outputs (left/right) to one channel each of the power amp (left/right).

☞ **Since most guitar amps only have a serial insert loop, you should make sure that the MODULIZER PRO is set to Mix-Intern mode. In this mode you can control the effect intensity applied to the guitar signal. However, if your amp features a parallel effect loop which allows for adding the effect-signal portion (similar to an aux path in a mixing console), we recommend that you use the MODULIZER PRO's Mix-Extern mode. In this case, the effect intensity present at the output of the MODULIZER PRO is 100%, and you can use the effect loop to determine the amount of effect added to the guitar signal.**

Instrumentalists can benefit from a variety of advantages offered by the MODULIZER PRO's MIDI implementation. For example, you can use a MIDI footswitch board to send program change commands via MIDI. Connect the MIDI OUT jack of your MIDI board to the MIDI IN jack on the MODULIZER PRO. If the MODULIZER PRO fails to respond to the program change commands sent from the MIDI board, check the MIDI channel settings. Consult the user's manual of your MIDI board to find out on which channels program change commands are transmitted (usually in Omni mode). Set the MIDI channels appropriately in MIDI mode (see 6.3) and enable the MODULIZER PRO to receive program change commands.

If your MIDI board features a controller or allows you to connect controller pedals, you can even change parameter settings via MIDI while playing. For instance, you can freely change the effect intensity from 0-100% while playing (Contr. 27, Value 0-100). Set the controller for Mix-Intern mode (Contr. 30, Value 0) so that it can be used to increase the effect intensity. In this way, guitar solos can be enhanced with chorus and delay effects, while the effect intensity is gradually reduced when playing rhythm. You can even control the function of the IN/OUT switch to bypass the MODULIZER PRO when an unprocessed signal is needed. Basically, all MIDI devices that are capable of transmitting MIDI controller commands, e.g. keyboards/sequencers, will allow for using these features.

The MODULIZER PRO may also be inserted between the outputs of a keyboard and the inputs of a mixing console. If required, adapt the levels with the Operating Level switch.

3.5 Using the MODULIZER PRO in a MIDI system

With its built-in MIDI interface the MODULIZER PRO can be integrated into any MIDI system, where it transmits and receives both program change and controller change information to perform program changes via MIDI from a sequencer or any other MIDI device. Wire and set up the MODULIZER PRO as shown below:

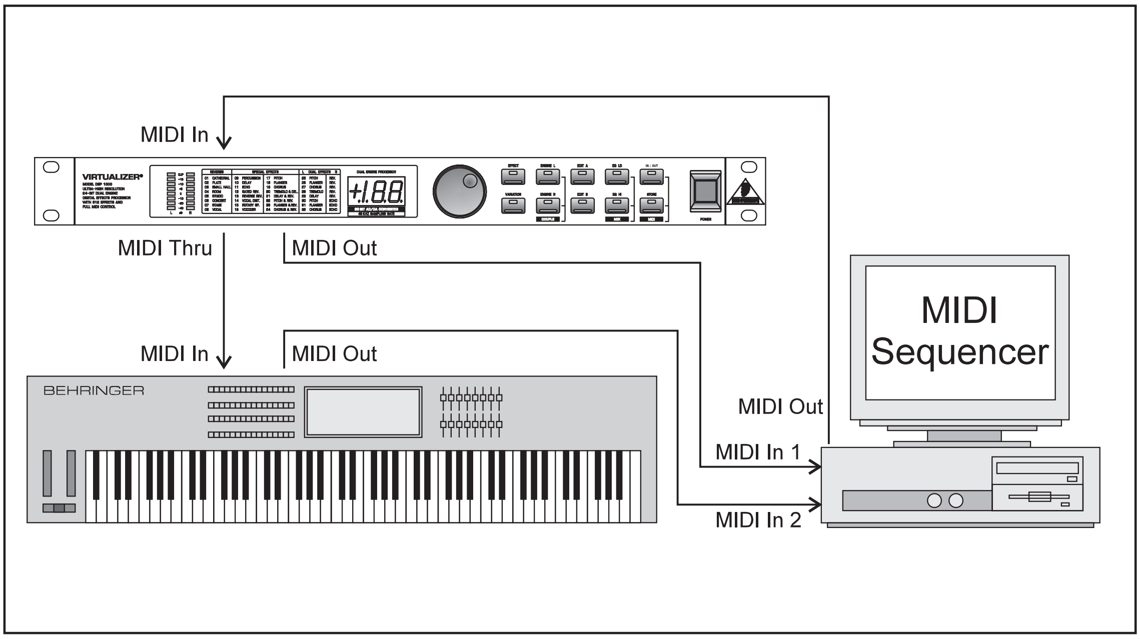


Fig. 3.4: Connecting the MODULIZER PRO via MIDI to a sequencer/computer and a keyboard (option)

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3.6 Saving data via MIDI

The MODULIZER PRO's MIDI implementation also allows for archiving one or several presets on an external storage medium. Proceed as follows:

Connect the MIDI OUT jack of the MODULIZER PRO to the MIDI IN jack of a MIDI data recorder (e.g. sequencer). Press the STORE and IN/OUT keys simultaneously to enter MIDI mode. Set program change mode to 0 and controller change mode to 3. Now quit MIDI mode by pressing the STORE key. Use the jog wheel to select the preset whose data you wish to save. When the preset is activated its parameters are transmitted as controller data and can be recorded on a sequencer or similar device. Repeat this routine until all presets of your choice have been sent to the external data recorder.

To load archived data back into the MODULIZER PRO, you must enable controller reception in MIDI mode (see 2.5). Then, start the sequencer to automatically transmit each preset data set back to the MODULIZER PRO. Press the STORE key, select a program location to store the data and then again press the STORE key. If you want to automate MIDI store functions you must enable the store mode, to switch on the reception of controller #28. This allows you to directly store any modification of the actual preset on the preset number that is transmitted with the controller. You can also restore a complete preset that has previously been recorded with a MIDI sequencer on the same location it had before.

4. TECHNICAL BACKGROUND

4.1 Digital audio processing

In order to convert an analog signal - e.g. music - into a series of digital words, a so-called "Analogue to Digital Converter" or ADC is used. The converter functions by viewing the signal entering it a given number of times over a period of time, e.g. 44,100 times per second, giving a rate of 44.1 kHz, and in each case measuring the signal amplitude, and giving it a numerical value. This form of measuring the signal regularly over a period of time is known as "sampling", the conversion of the amplitude into a numerical value, quantizing. The two actions together are referred to as digitizing.

In order to carry out the opposite - the conversion of a digitized signal into its original analogue form - a "Digital to Analogue Converter" or DAC is used. In both cases the frequency at which the device operates is called the sampling rate. The sampling rate determines the effective audio frequency range. The sampling rate must always be more than twice the value of the highest frequency to be reproduced. Therefore, the well known CD sampling rate of 44.1 kHz is slightly higher than twice the highest audible frequency of 20 kHz. The accuracy at which quantization takes place is primarily dependent on the quality of the ADCs and DACs being used.

The resolution, or size of digital word used (expressed in bits), determines the theoretical Signal/Noise ratio (S/N ratio) the audio system is capable of providing. The number of bits may be compared to the number of decimal places used in a calculation - the greater the number of places, the more accurate the end result. Theoretically, each extra bit of resolution should result in the S/N ratio increasing by 6 dB. Unfortunately, there are a considerable number of other factors to be taken into account, which hinder the achievement of these theoretical values.

If you picture an analog signal as a sinusoidal curve, then the sampling procedure may be thought of as a grid superimposed on the curve. The higher the sampling rate (and the higher the number of bits), the finer the grid. The analog signal traces a continuous curve, which very seldom coincides with the cross points of the grid. A signal level at the sampling points will still be assigned a digital value, usually the one closest to the exact representation. This limit to the resolution of the grid gives rise to errors, and these errors are the cause of quantizing noise. Unfortunately, quantizing noise has the characteristic of being much more noticeable and unpleasant to the ear than "natural" analog noise.

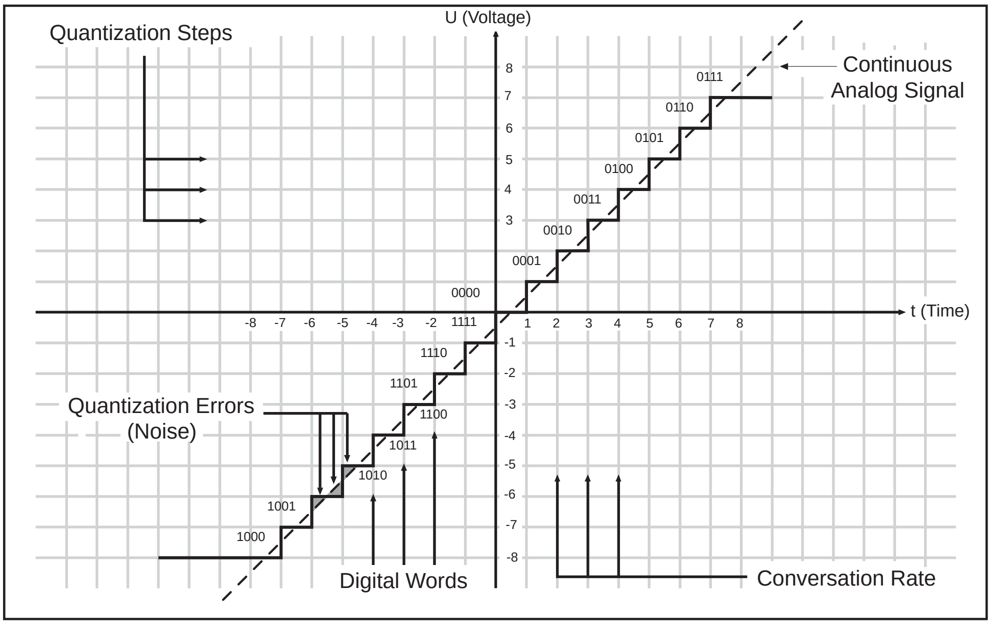

Fig. 4.1: Transfer diagram for an ideal linear ADC (2's complement representation)

In a digital signal processor, such as the DSPs in the MODULIZER PRO, the data will be modified in a number of ways, in other words, various calculations, or processes, will be done in order to achieve the desired effect on the signal. This gives rise to further errors, as these calculations are approximations, due to their being rounded off to a defined number of decimal places. This causes further noise. To minimize these rounding off errors, the calculations must be carried out with a higher resolution than that of the digital audio data being processed (as a comparison, an electronic calculator may operate internally with a greater number of decimal places than can be shown on its display). The DSPs in the MODULIZER PRO operate with a 24 bit resolution. This is accurate enough to reduce quantizing noise to levels which are usually below the audible threshold. However, when using extreme equalizer settings, some quantizing side effects may be detected.

Digital sampling has one further, very disturbing effect: it is very sensitive to signal overload. Take the following simple example using a sine wave. If an analog signal starts to overload, it results in the amplitude of the signal reaching a maximum level, and the peaks of the wave starting to get compressed, or flattened. The greater the proportion of the wave being flattened, the more harmonics, audible as distortion, will be heard. This is a gradual process, the level of distortion as a percentage of the total signal rising with the increase of the input signal level.

Digital distortion is quite different, as illustrated by this simplified example. If we take the situation where a 4 bit word has the positive maximum value of 0111, and add to it the smallest possible value of 0001 (in other words, the smallest increase in amplitude possible), the addition of the two results in 1000 - the value of the "negative" maximum. The value is turned on its head, going instantly from positive max to negative max, resulting in the very noticeable onset of extreme signal distortion.

4.2 Reverberation and reflection

In a concert hall the sound the listener hears comprises both the source signals (e.g. acoustical instruments, P.A. system) and thousands of reflections of these "primary signals", which bounce off floor, ceiling and walls to reach the ear after a short delay. These reflections represent thousands of echoes of the direct signal, which are not perceived any longer as single echoes but - due to their sheer number - as reverberation. Basically, the reflected signal portions reach the ear later than the source signal, and the very fact that they do not arrive from the same direction as the direct signal (see fig. 4.2), makes it possible to hear "spatial information", i.e. to perceive the direct signal as it is "embedded" in the room acoustics.

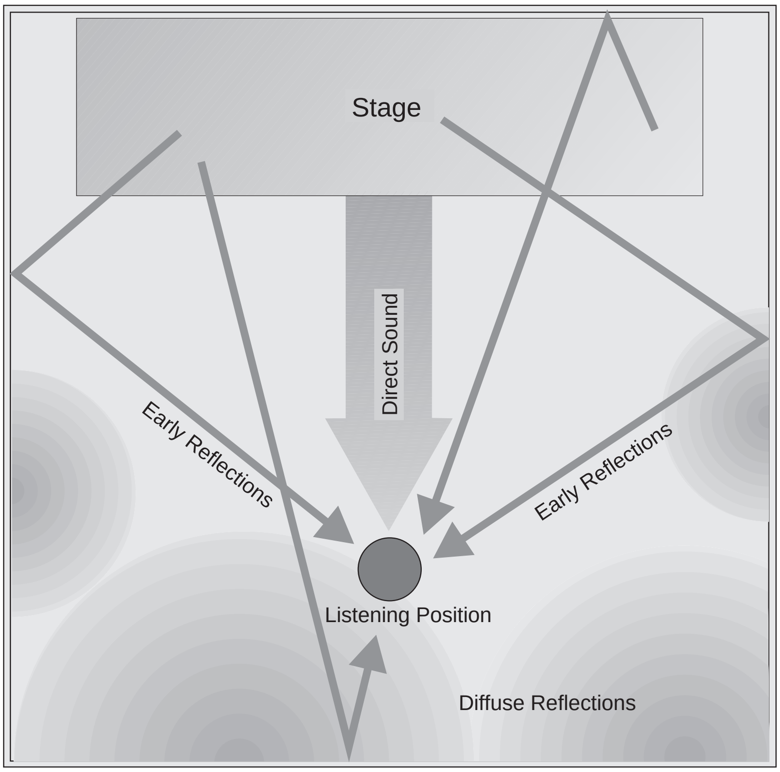

Fig. 4.2: Direct and reflected sounds reaching the listener's ear.

Spatial information is an important means of orientation, because human hearing is also used to determine the position of a sound source. In certain situations, this capability can be very useful or even of vital importance. The fact that we can actually "hear" the size of a room shows how strongly developed the human sense of hearing actually is. Based on the reflectivity of a room, we can also distinguish (though we often don't know how) the materials it consists of. In large rooms with high tiled walls reverberation is generally very dense and needs some time to decay, while a small room with many objects in it (furniture, carpets, etc.) features very short reverberation often not even perceived as such. Nevertheless, this extremely short reverb does exist, which is the reason why many designers of reverb devices (such as our VIRTUALIZER PRO) implement several basic reverb types and give them specific room names. It is quite natural, for example, that a reverb preset called "Cathedral" produces a long and highly dense reverb, while a "Room" program usually represents the acoustics of a room that is much smaller in size.

In addition to the capability of human hearing to determine the direction from where a sound phenomenon arrives, we can also hear modulations of acoustic events. Of importance in this context is the frequency of the modulated signals. Frequency modulations below 100 Hz are virtually inaudible. However frequency modulations can clearly be heard when occurring in the midrange frequency band, due to the "sensitivity" of human hearing. The ear immediately detects changes in midrange frequencies, while its sensitivity to frequency modulation in the extreme low end of the frequency spectrum is reduced.

Frequency modulation can also be used to produce "wanted" effects. The popular chorus effect, for instance, is basically the sum of a variety of frequency modulations. The original signal is slightly delayed in the chorus algorithm, then added again and modulated by means of an oscillator. Subsequently, modulating frequencies (of different pitch) are applied to the original signal, which produces the well-known "floating" chorus sound. Basically, frequency modulation is the starting point for all kinds of chorus-type effects: by simply adding the delayed signal, without modulating the original, you can produce a delay effect. Since chorus effects use very short delay times, the resulting delay effect is not perceived as such. However, when you increase the delay time, there is a clear gap between original and effect signals, and delay becomes audible.

4.3 Audio dynamics

By employing current modern analog technology it is possible to manufacture audio equipment with a dynamic range of up to 130 dB. In contrast to analog techniques, the dynamic range of digital equipment is approximately 25 dB less. With conventional record and tape recorder technology, as well as broadcasting, this value is further reduced. Generally, dynamic restrictions are due to noisy storage in transmission media and also the maximum headroom of these systems.

4.3.1 Noise as a physical phenomenon

All electrical components produce a certain level of inherent noise. Current flowing through a conductor leads to uncontrolled random electron movements. For statistical reasons, this produces frequencies within the whole audio spectrum. If these currents are highly amplified, the result will be perceived as noise. Since all frequencies are equally affected, we term this white noise. It is fairly obvious that electronics cannot function without components. Even if special low-noise components are used, a certain degree of basic noise cannot be avoided.

This effect is similar when replaying a tape. The non-directional magnetic particles passing the replay head can also cause uncontrolled currents and voltages. The resulting sound of the various frequencies is heard as noise. Even the best possible tape biasing can “only“ provide signal-to-noise ratios of about 70 dB, which is not acceptable today since the demands of listeners have increased. Due to the laws of physics, improving the design of the magnetic carrier is impossible using conventional means.

4.3.2 What are audio dynamics?

A remarkable feature of the human ear is that it can detect the most wide ranging amplitude changes - from the slightest whisper to the deafening roar of a jet-plane. If one tried to record or reproduce this wide spectrum of sound with the help of amplifiers, cassette recorders, records or even digital recorders (CD, DAT etc.), one would immediately be restricted by the physical limitations of electronic and acoustic sound reproduction technology.

The usable dynamic range of electroacoustic equipment is limited as much at the low end as at the high end. The thermal noise of the electrons in the components results in an audible basic noise floor and thus represents the bottom limit of the transmission range. The upper limit is determined by the levels of the internal operating voltages; if they are exceeded, audible signal distortion is the result. Although in theory, the usable dynamic range sits between these two limits, it is considerably smaller in practice, since a certain reserve must be maintained to avoid distortion of the audio signal if sudden level peaks occur. Technically speaking, we refer to this reserve as “headroom“ - usually this is about 10 - 20 dB. A reduction of the operating level would allow for greater headroom, i.e. the risk of signal distortion due to level peaks would be reduced. However, at the same time, the basic noise floor of the program material would be increased considerably.

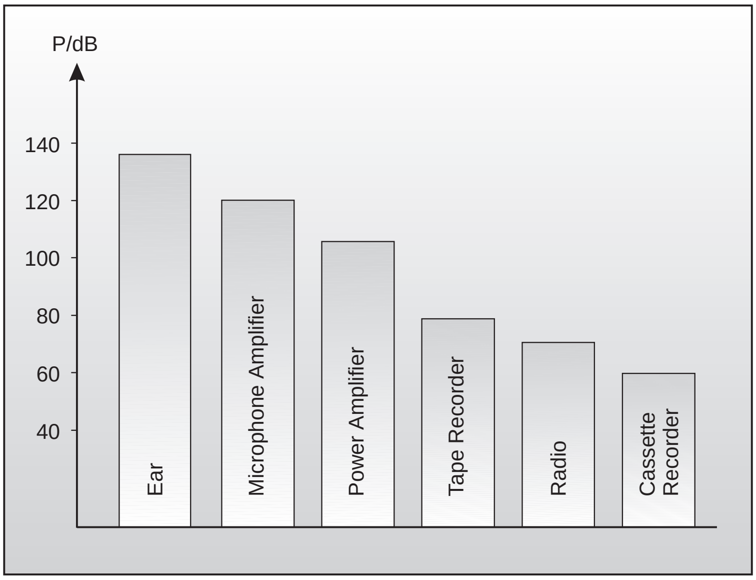

Fig. 4.3: The dynamic range capabilities of various devices

It is therefore useful to keep the operating level as high as possible without risking signal distortion in order to achieve optimum transmission quality.

It is possible to further improve the transmission quality by constantly monitoring the program material with the aid of a volume fader, which manually levels the material. During low passages the gain is increased, during loud passages the gain is reduced. Of course it is fairly obvious that this kind of manual control is rather restrictive; it is difficult to detect signal peaks and it is almost impossible to level them out. Manual control is simply not fast enough to be satisfactory.

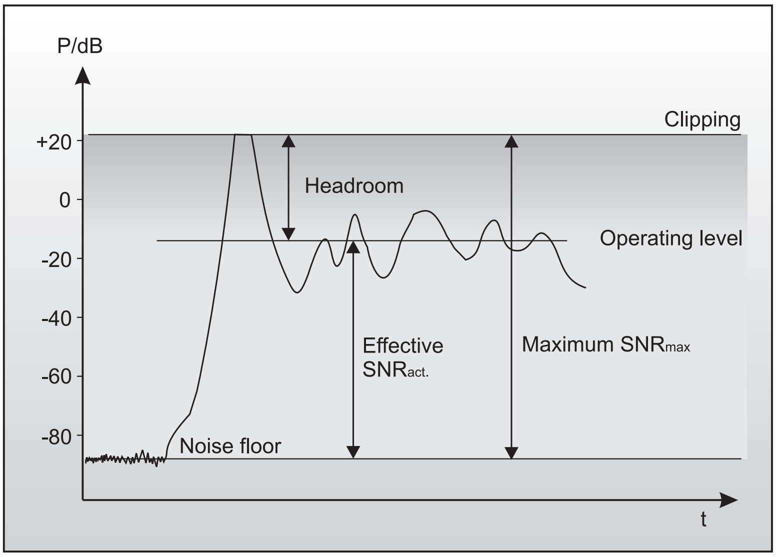

Fig. 4.4: The interactive relationship between the operating level and the headroom

The need therefore arises for a fast acting automatic gain control system which will constantly monitor the signals and which will always adjust the gain to maximize the signal-to-noise ratio without incurring signal distortion. This device is called a compressor or limiter. This system is a part of the BEHRINGER MODULIZER PRO.

4.3.3 Compressors/limiters

With broadcasting and recording, signal peaks can easily lead to distortion due to the high dynamic range of microphones and other musical equipment. Compressors and limiters reduce the dynamics by means of an automatic gain control. This reduces the amplitude of loud passages and therefore, restricts the dynamics to a desired range. This application is particularly useful with microphones, to compensate for level changes.

Although compressors and limiters perform similar tasks, one essential point makes them different: limiters abruptly limit the signal above a certain level, while compressors control the signal "gently" over a wider range. Both continuously monitor the signal and intervene as soon as the level exceeds a user-adjustable threshold. Any signal exceeding this threshold will be immediately reduced in level.

Limiters reduce the output level to the adjusted threshold whenever the input signal exceeds this point. With compression, in contrast to the action of a limiter, the signal is reduced in gain relative to the amount the signal exceeds the threshold. The output of a compressor will still rise if the input level is increased, while the maximum output of a limiter will always be equal to the threshold level.

Generally, threshold levels for compressors are set below the normal operating level to allow for the upper dynamics to be musically compressed. For limiters, the threshold point is set above the normal operating level so that it only intervenes to protect subsequent equipment from signal overload.

The speed, or rather time settings used can differ greatly depending on use. Although both limiter and compressor use very short attack times, the release time of a compressor are in the 100 ms region whereas a limiter uses release times of seconds. To be exact: The release time is a time constant of an exponential function. It is the time it takes the gain reduction to decrease by 63.2 % (= 8.7 dB).

Because fast level changes are more noticeable than slow changes, long release time are used where unobtrusive signal processing is required. In some cases however, the principal goal is to protect devices as loudspeakers and power amplifiers. In those cases a short release time is more appropriate to ensure that the limiter only intervenes when it is needed and the level returns to normal as soon as possible.

Long release times are better suited when the limiter should remain "inaudible" for instance with broadcasting or club applications or when a signal is transferred to (analog) tape. Please note that when using slow release times you should switch to the level meter menu where the functioning of the limiter can be monitored.

4.3.4 Expanders/noise-gates

Audio, in general, is only as good as the source from which it was derived. The dynamic range of signals will often be restricted by noise. Synthesizers, effects devices, guitar pickups, amplifiers etc. generally produce a high level of noise, hum or other ambient background hiss, which can disturb the quality of the program material.

Normally these noises are inaudible if the level of the desired signal lies significantly above the level of the noise. This perception by the ear is based on the “masking“ effect: noise will be masked and thus becomes inaudible as soon as considerably louder sound signals in the same frequency band are added. Nevertheless, the further the level that the desired signal decreases, the more the noise floor becomes a disturbing factor. Expanders or noise-gates offer a solution for this problem: these devices attenuate signals when their amplitudes drop, thereby fading out the background noise. Relying on this method, gain controlling amplifiers, like expanders, can extend the dynamic range of a signal and are therefore the opposite of a compressor.

In practice, it is shown that an expansion over the entire dynamic range is not desired. With an expansion ratio of 5:1 and a processed dynamic range of 30 dB, an output dynamic range of 150 dB will be the result, exceeding all subsequent signal processors, as well as human hearing. Therefore, the amplitude control is restricted to signals whose levels are below a certain threshold. Signals above this threshold pass through the unit unchanged. Due to the continuous attenuation of the signals below this threshold, this kind of expansion is termed “downward“ expansion.

The noise-gate is the simplest form of an expander: in contrast to the expander, which continuously attenuates a signal below the threshold, the noise-gate cuts off the signal abruptly. In most applications this method is not very useful, since the on/off transition is too drastic. The onset of a simple gate function appears very obvious and unnatural. To achieve inaudible processing of the program material, it is necessary to be able to control the signal’s envelope parameters. This is part of the many features of the MODULIZER PRO.

4.4 Artificial harmonics generation

By 1955 an American, Charles D. Lindridge, had already invented the first “EXCITER” (a unit that EXCITES upper harmonics), when he presented a unit for “improving the sound of music and speech”. He enriched signal sources with artificially generated upper harmonics and found that both sound quality, transparency and perceived positioning of musical instruments could be considerably improved using this effect. He was granted an American patent on his circuit design under the number US 2 866 849.

Compared to modern technology, Lindridge’s circuit was anything but fully developed, however, it featured many of the aspects found in today’s modern circuit designs. Psycho-acoustic discoveries and greater knowledge, gathered over the years, have allowed for new and improved circuit designs, through the use of advanced technology.

Vacuum tubes also produce harmonics as a result of distortion caused by saturation. In general, the saturation (overdriving) of transistor and tube-based circuits results in different types of distortion. Distortion caused by tubes is generally considered to be more pleasant and warm, often enhancing the quality rather than deteriorating it. With the MODULIZER PRO various tube types and their specific sound can be emulated.

4.5 Tube technology

A closer look at developments and trends in audio technology shows that tubes are currently enjoying a renaissance, in a time when even amateur musicians are free to use digital effects processors and recording media, and ever more affordable digital mixing consoles are becoming a natural part of the equipment of many semi-professional studios. The manufacturers try with ever new algorithms to get the most out of DSP’s (Digital Signal Processors), the heart of any digital system.

Still, many audio engineers, particularly old hands often prefer using both old and new tube-equipped devices. As they want to use their warm sound character for their productions, they are ready to accept that these “goodies” produce a higher noise floor than modern, transistor-based devices. As a consequence, you can find a variety of tube-based microphones, equalizers, preamps and compressors in today’s recording and mastering environments. The combination of semiconductor and tube technologies gives you the additional possibility of using the best of both worlds, while being able to make up for their specific drawbacks

Tubes do not however have the same task in a recording studio as in an overdriven guitar amp, where the considerably higher saturation of the tube(s) leads to a full and often deliberate modification of the input signal (in many cases combined with a heavy increase in noise floor levels). In the studio more subtle effects are needed. Here, tube circuits add life to the signal's tonal character and increase its power to make itself heard. Often, tubes also increase the signal's perceived loudness (in relation to the unprocessed signal), i.e. the perceived loudness goes up although the volume level remains the same. This is because the dynamic range of the applied audio signal is limited by the tube circuit, while the amplitude of the signal with the lowest loudness is raised. Thus, increasing tube saturation produces a slight compression effect over the entire dynamic range.

A similar effect can be perceived when analog tape is saturated. This saturation effect also compresses the recorded audio material and produces additional harmonics.

The BEHRINGER MODULIZER PRO is equipped with tube emulation algorithms that due to extensive research deliver very subtle and natural sound effects. These newly developed algorithms are capable of delivering both the subtle nuances of a tube as the strong coloration as produced by heavy saturation.

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5. INSTALLATION

Your BEHRINGER MODULIZER PRO was carefully packed in the factory and the packaging was designed to protect the unit from rough handling. Nevertheless, we recommend that you carefully examine the packaging and its contents for any signs of physical damage, which may have occurred in transit.

☞ **If the unit is damaged, please do not return it to us, but notify your dealer and the shipping company immediately, otherwise claims for damage or replacement may not be granted. Shipping claims must be made by the consignee.**

5.1 Rack mounting

The BEHRINGER MODULIZER PRO fits into one standard 19" rack unit of space (1 3/4"). Please allow at least an additional 4" depth for the connectors on the back panel. Be sure that there is enough air space around the unit for cooling and please do not place the MODULIZER PRO on high temperature devices such as power amplifiers etc. to avoid overheating.

5.2 Mains connection

The mains connection of the MODULIZER PRO is made by using a mains cable and a standard IEC receptacle. It meets all of the international safety certification requirements.

☞ **Please make sure that all units have a proper ground connection. For your own safety, do not remove the ground connection within the units or at the supply, or fail to make this connection at all.**

Before you switch on the unit, check that it is configured to match your AC mains voltage requirements. If it does not comply, then it is necessary to switch the operating voltage to the correct supply requirements BEFORE turning on the unit, otherwise the unit could be severely damaged. You will find this combined fuse holder/voltage selector at the back, adjacent to the IEC receptacle. **IMPORTANT: This does not apply for general export models which are built for one operating voltage only.**

The AC voltage selection is defined by the position of the fuse holder. If you intend to change the operating voltage, remove the fuse holder and twist it by 180 degrees before you reinsert it. Matching the two markers monitors the selected voltage.

☞ **If the unit is switched to an other operating voltage, the ruse rating must be changed. See the technical specifications in the appendix.**

A safety fuse protects the unit from serious defects. If the fuse blows, this is a warning sign and always indicates that the circuit is overloaded. The fault must always be repaired before the fuse is replaced. If the safety fuse is faulty and needs replacing after the unit is repaired, please make sure that you replace it only with the identical type and rating. NEVER use fuses of different ratings or cover faulty fuses with aluminium foil. This can cause fire and electric shocks and will endanger your life and the lives of others.

5.3 Audio connections

As standard, the BEHRINGER MODULIZER PRO is installed with electronically servo-balanced inputs and outputs. The new circuit design features automatic hum and noise reduction for balanced signals and thus allows for trouble-free operation, even at high operating levels. Externally induced mains hum etc. will be effectively suppressed. The automatic servo-function recognizes the presence of unbalanced connectors and adjusts the nominal level internally to avoid level differences between the input and output signals (correction 6 dB).

☞ **Please ensure that only qualified persons install and operate the MODULIZER PRO. During installation and operation the user must have sufficient electrical contact to earth. Electrostatic charges might affect the operation of the MODULIZER PRO!**

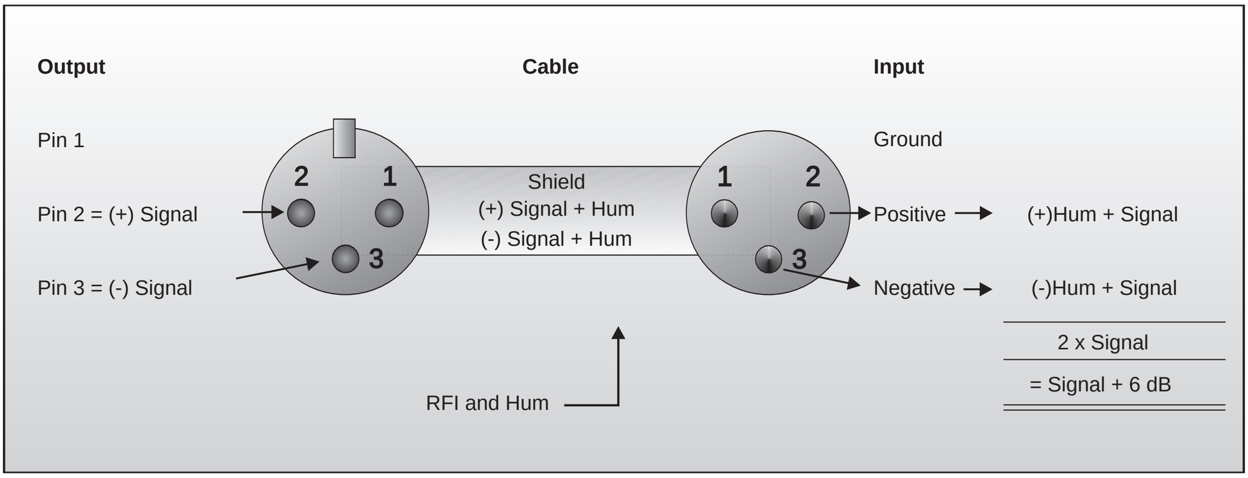

Fig. 5.1: Compensation of interference with balanced connections

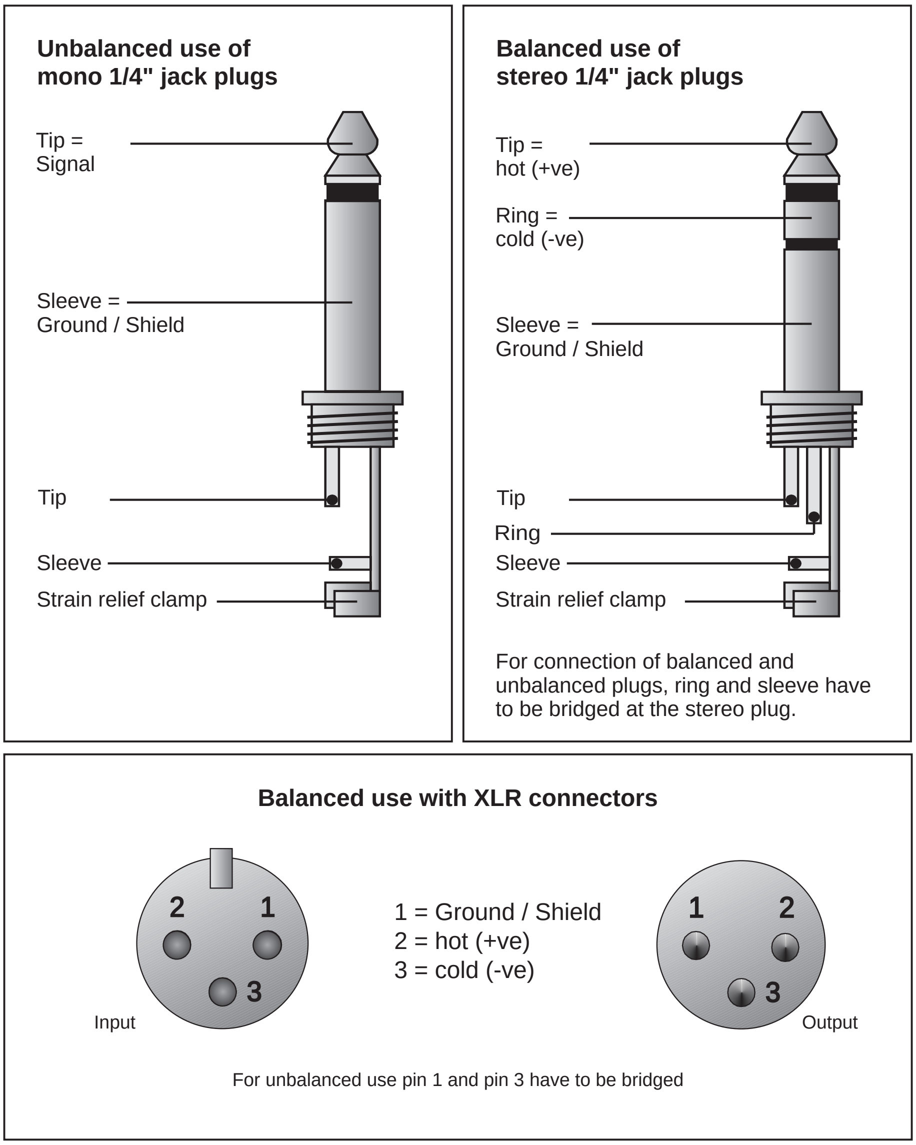

Fig. 5.2: Different plug types

5.4 MIDI connections

The MIDI standard was developed in the early 1980s to allow electronic musical instruments from different manufacturers to communicate with each other. The use of MIDI has developed over the intervening years to the stage where it is now common to find complete recording studios operating entirely on a MIDI basis. The centerpiece in such a studio is usually a computer running a sequencer software which not only controls various keyboards, samplers and sound modules, but can also run the programming of outboard effect devices, typically digital reverberation and delay units. The MODULIZER PRO may be controlled in real time in this studio environment.

☞ **MIDI for Musical Instruments Digital Interface.**

The MIDI connectors found on the rear panel are of the universally used 5 pin DIN type. You require suitable MIDI cables to connect the MODULIZER PRO to other MIDI devices. Normally complete cables will be purchased for this use, you can of course make your own, using a high quality cable with two cores and shielding (like microphone cable), with as connectors two good 180 degree DIN plugs. Pin 2 (center) is connected to the cable’s shield, pins 4 and 5 (left and right next to 2) carry the two cores, pins 1 and 3 are not used. MIDI cables should have a maximum length not exceeding 45 feet.

5.5 Operating level switch

To adapt the MODULIZER PRO to the used operating level, the unit can be switched between homerecording level (-10 dBV) and professional level (+4 dBu). Use the LED bars on the frontpanel to determine the optimal setting. The Level indicators should read somewhere between -10 and -6 dB, while the Clip LED should not light up at all.

6. APPENDIX

6.1 Parameter overview

No.	EFFECT	VARIATION	Edit A	Edit B	Edit C	Edit D
1	**Ultra Phaser**	LFO Speed *	Intensity	Depth	Feedback	
2	**Spatial Phaser**	LFO Speed *	Intensity	Depth	Feedback	
3	**Harmonic Exciter**	HP Shape	Tune	Harmonics		
4	**Auto Lowpass**	Mode (LFO, Auto 12/24 dB)	Frequency	Resonance	Mod. Depth	Env. / LFO Speed *
5	**Auto Highpass**	Mode (LFO, Auto 12/24 dB)	Frequency	Resonance	Mod. Depth	Env. / LFO Speed *
6	**Auto Bandpass**	Mode (LFO, Auto 12/24 dB)	Frequency	Resonance	Mod. Depth	Env. / LFO Speed *
7	**Jetstream Flanger**	LFO Speed *	Delay	Depth	Feedback	Band Limit
8	**Spatial Flanger**	LFO Speed *	Delay	Depth	Feedback	Band Limit
9	**Ultra Chorus**	LFO Speed	Delay	Depth	Stereo Width	Wideness
10	**Stereo Imager**	Xover Freq.	Gain	Spread	Mono Pan.	St. Center
11	**3D Space Maker**	In / Out	Gain	Spread	Xover Freq.	
12	**Denoiser**	Gate Thresh.	Gate Hold	Gate Rel.	LP Freq.	LP Depth
13	**Ultra Ambience**	Pre Delay	Size	Wall Damp	Stereo Width	Reflections
14	**Voice Canceler**	Bass Freq.	Gain	Bass Pan.	Treble Pan.	
15	**Comp./Limiter**	Ratio	Threshold	Out Gain	Attack	Release
16	**Expander**	Ratio	Threshold	Out Gain	Attack	Release
17	**Noise Gate**	Threshold	Hold	Range	Attack	Release
18	**Ring Modulator**	Mode (LFO, Auto, RND)	Carrier Freq.	LFO / Speed *	Mod. Depth	Band Limit
19	**Vintager**	Clicks Level	Noise Level	Noise BP.	Buzz Level	Signal BP.
20	**Tube Distortion**	Tube Type	In Gain	Low Cut	High Cut	Band Limit
21	**Guitar Combo**	Type	In Gain	Drive	Presence	Speaker
22	**Guitar Speaker**	Speaker Type	Peak Freq.	Peak Q	Peak Gain	HF Cut
23	**Super Bass**	Frequency	Density	Ratio	Bass Level	
24	**Resonator**	Mode (LFO, Auto, RND)	Frequency	LFO / Speed *	Mod. Depth	Feedback

* User/controller defined fixed/start value when set to 0

Tab. 6.1: Parameter overview for the different effect types

6.2 Variation table

No.	Effect	Variation Range	Edit A Range	Edit B Range	Edit C Range	Edit D Range	Mix Range
1	Ultra Phaser	0..127 *)	1..8	0..127	0..127	-	0..100
2	Spatial Phaser	0..127 *)	1..8	0..127	0..127	-	0..100
3	Harmonic Exciter	0..127	0..127	0..127	-	-	0..25 1.)
4	Auto Lowpass	L1, L2, A1, A2	0..127	0..127	0..127	0..127 *)	0..100
5	Auto Highpass	L1, L2, A1, A2	0..127	0..127	0..127	0..127 *)	0..100
6	Auto Bandpass	L1, L2, A1, A2	0..127	0..127	0..127	0..127 *)	0..100
7	Jetstream Flanger	0..127 *)	1..128	0..127	-100..+100	0..127	0..100
8	Spatial Flanger	0..127 *)	1..128	0..127	-100..+100	0..127	0..100
9	Ultra Chorus	0..127	1..128	0..127	0..127	0..127	0..100
10	Stereo Imager	0..127	-6.0..+6.0	0..127	-100..+100	-100..+100	0..100 2.)
11	3D Space Maker	1,2	-6.0..+6.0	0..127	0..127	-	0..100 2.)
12	Denoiser	0..127	0..127	0..127	0..127	0..127	-
13	Ultra Ambience	0..127	0..127	0..127	0..127	1..15	0..100
14	Voice Canceler	0..127	-6.0..+6.0	-100..+100	-100..+100	-	0..100 2.)
15	Comp./Limiter	0..24	-60..+0	-24 ..+24	0..127	0..127	-
16	Expander	0..24	-60..+0	-24 ..+24	0..127	0..127	-
17	Noise Gate	0..127	0..127	0..127	0..127	0..127	-
18	Ring Modulator	L,E,R0..7,S0..7	0..127	0..127 *)	0..127	0..127	0..100
19	Vintager	0..127	0..127	0..127	0..127	0..127	0..100
20	Tube Distortion	1,2,3	0..127	0..127	0..127	0..127	0..100
21	Guitar Combo	1..3	0..127	0..127	0..127	0,1,2	0..100
22	Guitar Speaker	1,2,3	0..127	1.0..10.0	-12.0..+12.0	0..127	0..100
23	Super Bass	0..127	0..127	0..6	0..127	-	-
24	Resonator	L, E, R0..7	0..127	0..127*)	0.127	-100..+100	0..100

Tab. 6.2: Variation range

1.) Exciter: Insert: Dry = 0% / Wet = 100%

2.) Mix = M-S-Matrix (Dry = Mono / Wet = Stereo) / Insert function is cancelled.

***)** Additional Modulation-Controller When LFO SPEED = 0

E

6.3 MIDI implementation

MIDI Implementation Chart				
Function		**Transmitted**	**Recognized**	**Remarks**
Basic Channel	Default	OFF, 1 - 16	OFF, 1 - 16	memorized
	Changed	OFF, 1 - 16	OFF, 1 - 16	
Mode	Default	1,2,3,4	1,2,3,4	
	Messages	X	X	
	Altered	X	X	
Note Number		X	X	
	True Voice	X	X	
Velocity	Note ON	X	X	
	Note OFF	X	X	
After Touch	Key´s	X	X	
	Ch´s	X	X	
Pitch Bender		X	X	
Control		O 50 - 62	O 50 - 62	see add. Table
Progr. Change		O (0-99)	O (0-99)	
	True #	1-100	1-100	
System Exclusive		X	X	
System Common	Song Pos	X	X	
	Song Sel	X	X	
	Tune	X	X	
System Real Time	Clock	X	X	
	Commands	X	X	
Aux Messages	Local ON/OFF	X	X	
	All notes OFF	X	X	
	Active Sense	X	X	
	Reset	X	X	
Notes				

O = YES, X = NO
Mode 1: OMNI ON, POLY
Mode 2: OMNI ON, MONO
Mode 3: OMNI OFF, POLY
Mode 4: OMNI OFF, MONO

Tab. 6.3: MIDI implementation chart

Parameter Name	Display Range	Midi-Controller	Controller Range	LEDs				
				Couple	Left	Right	IN on	IN off
Effect	1..24	50	0..23					
Variation	dep. on effect*	51	0..xxx					
Edit A	"	52	"					
Edit B	"	53	"					
Edit C	"	54	"					
Edit D	"	55	"					
Modulation Controller	0..127	56	0..127					
EQ Low	-16..16	57	0..32					
EQ High	-16..16	58	0..32					
Mix	0..100	59	0..100					
Store	1..100	60	0..99					
In/Out		61	0=Out, 1=In				1	0
External/Internal Mix	--/(Mix) "--"=Ext	62	0=Ext, 1= Int					

* nF = no function

Tab. 6.4: Controller functions with MIDI

6.4 Default settings

No.	Effect	Variation	Edit A	Edit B	Edit C	Edit D	Mix
1	Ultra Phaser	51	6	100	127	-	70
2	Spatial Phaser	60	8	80	90	-	70
3	Harmonic Exciter	15	90	60	-	-	24
4	Auto Lowpass	A2	60	70	127	29	80
5	Auto Highpass	L2	20	110	127	40	80
6	Auto Bandpass	A2	70	60	127	10	80
7	Jetstream Flanger	40	4	100	+100	50	50
8	Spatial Flanger	60	8	127	+100	95	50
9	Ultra Chorus	10	60	100	127	20	50
10	Stereo Imager	100	-2.0	+0.0	+0.0	50	60
11	3D Space Maker	1	+3.0	80	20	-	60
12	Denoiser	40	40	10	75	127	-
13	Ultra Ambience	30	127	52	127	15	40
14	Voice Canceler	85	+2.0	+0	+0	-	100
15	Comp./Limiter	12	-26 dB	+6 dB	30	70	-
16	Expander	3	-26 dB	0 dB	50	60	-
17	Noise Gate	60	40	20	20	30	-
18	Ring Modulator	E	10	20	80	50	50
19	Vintager	84	10	31	84	50	100
20	Tube Distortion	2	50	35	10	20	100
21	Guitar Combo	1	30	40	80	1	100
22	Guitar Speaker	1	50	1.0	+0.0	30	100
23	Super Bass	30	35	6	10	-	-
24	Resonator	E	10	20	80	+50	50

Tab. 6.5: Default settings

6.5 Preset parameters

No.	Name	Preset	Var.	Edit A	Edit B	Edit C	Edit D	EQ-LO	EQ-HI	Mix
1	Space Phaser	2	50	8	90	110	nF	2	0	70
2	Mr. Excite	3	115	95	90	nF	nF	2	2	25
3	Auto Bandpass	6	L2	16	74	127	95	5	0	75
4	Speaker I	22	3	80	20	80	68	8	0	100
5	Jet Flanger	7	40	5	75	80	127	3	0	75
6	Vintager I	19	79	106	81	87	75	0	3	100
7	Ultra Chorus I	9	36	19	95	127	127	0	-3	75
8	Stereo Imager I	10	38	5	44	0	0	0	0	67
9	Compressor I	15	8	-30	13	2	90	0	0	nF
10	Ambience I	13	10	90	0	90	15	2	0	100
11	Bass II	23	30	50	0	40	nF	6	3	nF
12	Speech Coder	18	E	0	20	70	50	0	2	100
13	High Resonance Phasing	5	L2	20	110	127	40	16	16	80
14	Auto Bass Mute Effect	5	L1	0	105	127	85	6	6	100
15	Guitar Phaser	1	71	8	100	102	nF	0	6	70
16	Guitar Chorus	9	25	65	127	107	110	6	6	50
17	Guitar Combo I	21	3	127	127	60	1	3	-1	100
18	Guitar Combo II	21	2	127	60	50	2	3	-1	100
19	Guitar Combo III	21	1	127	127	90	1	2	0	100
20	Mr. Tube	20	3	127	127	50	127	-2	3	100
21	Phaser !	1	51	6	100	127	nF	2	0	70
22	Exciter I	3	70	65	60	nF	nF	0	0	25
23	Auto High Pass	5	L1	20	127	127	50	0	0	80
24	Special Jet Flanger	8	50	25	127	-50	70	4	3	75
25	Ultra Chorus II	9	20	10	60	127	127	0	0	70
26	Wooden Hall	13	77	8	55	0	12	3	-3	40
27	Ring I	18	L	64	64	64	0	0	0	25
28	Ring II	18	L	110	0	64	40	0	0	30
29	Resonator	24	L	30	64	64	20	0	0	50
30	Bass II	23	20	40	6	30	nF	0	0	nF
31	Resonator II	24	L	32	60	80	50	3	3	75
32	Spatial Phaser	2	74	8	80	90	nF	0	0	65
33	Jetstream Flanger	7	40	4	100	100	50	0	0	50
34	Spatial Flanger	8	60	8	127	100	95	0	0	50
35	Space Maker I	11	2	33	69	51	nF	0	0	51
36	Fat Funky Phaser	1	40	8	45	110	nF	3	3	100
37	Space Phaser	2	43	8	55	127	nF	3	3	100
38	Funky "Oi"-Phasing	2	127	4	50	127	nF	3	3	100
39	Chaos Phaser	2	127	8	23	127	nF	6	6	100
40	Spacy Trance Flanger I	7	60	6	115	100	50	6	6	50
41	Spacy Trance Flanger II	7	55	20	127	100	65	6	6	50
42	Sirene Flanger	7	42	22	97	96	25	6	6	65
43	Acid Flanger	7	111	40	127	98	15	6	6	50
44	Speed Up Flanger	7	64	128	127	98	0	6	6	50
45	Motorbike	8	40	60	75	100	14	6	6	50
46	3D Space Flanger	8	42	9	80	94	70	6	6	70
47	Low Filtered Wah	4	L2	46	80	67	114	16	16	100
48	Low Filtered Reso-Wah	4	L2	95	80	86	124	16	16	100
49	Percussion Peak	6	L2	127	60	38	38	6	6	70
50	Midband Wah	6	L2	65	75	66	80	12	12	75

Nr.	Name	Preset	Var.	Edit A	Edit B	Edit C	Edit D	EQ-LO	EQ-HI	Mix
51	Auto Midband Wah	6	A2	55	73	127	15	16	16	80
52	Chorus I	9	10	55	127	127	58	3	3	60
53	Chorus II	9	30	20	127	40	0	3	3	40
54	Chorus III	9	60	78	127	127	127	6	6	50
55	Stereo Imager II	10	100	50	127	0	0	3	3	60
56	Stereo Imager III	10	80	50	127	-100	100	3	3	60
57	Ambience II	13	110	119	0	127	3	3	3	40
58	Ambience III	13	30	30	10	110	15	3	3	40
59	Ambience IV	13	20	60	25	100	8	3	3	40
60	Gate	17	82	5	5	0	3	3	3	nF
61	Ring III	18	L	8	30	70	35	6	6	55
62	Ring IV	18	L	85	127	65	30	6	6	55
63	Ring V	18	E	65	60	68	0	6	6	55
64	Ring VI	18	R2	98	70	15	0	6	6	55
65	Ring VII	18	R7	39	70	15	58	6	6	55
66	Ring VIII	18	S1	50	54	76	67	6	6	10
67	Vintager II	19	90	90	20	70	40	3	3	60
68	Vintager III	19	100	107	77	88	50	3	3	60
69	Tube I	20	1	110	0	30	0	3	3	25
70	Tube II	20	3	120	0	20	0	3	3	25
71	Tube III	20	3	127	127	60	90	3	3	25
72	Combo I	21	1	127	90	90	1	3	3	100
73	Combo II	21	2	100	40	38	2	3	3	100
74	Combo III	21	3	110	110	120	0	3	3	35
75	Combo IV	21	3	80	65	80	2	3	3	50
76	Speaker II	22	3	100	100	120	80	3	3	70
77	Speaker III	22	3	103	60	70	8	3	3	70
78	Speaker IV	22	1	127	10	60	70	3	3	70
79	Bass III	23	0	50	0	80	nF	6	3	nF
80	Resonator III	24	L	50	50	127	90	3	3	50
81	Resonator IV	24	L	106	120	110	90	3	3	50
82	Resonator V	24	L	80	95	70	70	3	3	50
83	Resonator VI	24	E	80	95	70	70	3	3	50
84	Resonator VII	24	E	30	110	60	30	3	3	50
85	Resonator VIII	24	E	64	120	4	94	3	3	50
86	Resonator IX	24	E	1	75	36	-72	3	3	50
87	Compressor II	15	6	-44	14	0	90	0	0	nF
88	Denoiser	12	20	40	88	90	64	4	0	nF
89	Ambience V	13	30	30	50	80	15	4	0	100
90	Ambience VI	13	30	0	50	80	15	4	0	100
91	Ambience VII	13	15	64	0	90	10	2	0	100
92	Stereo Imager IV	10	70	-10	127	0	10	0	0	60
93	Space Maker II	11	2	20	90	90	nF	0	2	60
94	Space Maker III	11	2	30	127	80	nF	0	2	75
95	Voice Canceler	14	40	60	0	0	nF	8	0	30
96	Vintager IV	19	98	119	80	92	60	-2	-6	75
97	Vintager V	19	110	127	74	64	0	0	0	50
98	Bright Bass	23	75	64	4	90	nF	0	0	nF
99	Sub Bass	23	0	70	3	127	nF	0	0	nF
100	Tube IV	20	2	127	80	10	66	3	-2	100

Tab. 6.6: Preset parameters

6.6 Specifications

Analog Inputs	
Connectors	XLR and 1/4" jack
Type	RF filtered, servo balanced input
Impedance	60 kOhms balanced, 30 kOhms unbalanced
Nominal Operating Level	-10dBV to +4dBu
Max. Input Level	+16 dBu at +4 dB nominal level, +2 dBV at -10 dBV nominal level
Analog Outputs	
Connectors	XLR and 1/4" jack
Type	Electronically servo-balanced output stage
Impedance	60 Ohms balanced, 30 Ohms unbalanced
Max. Output Level	+16 dBu at +4 dB nominal level, +2 dBV at -10 dBV nominal level
System specifications	
Bandwidth	20 Hz to 20 kHz, +0/-3 dB
Noise	>94 dBu, unweighted, 20 Hz to 20 kHz
THD	0.0075 % typ. @ +4 dBu, 1 kHz, Gain 1
Crosstalk	< -76 dBu
MIDI Interface	
Type	5-Pin-DIN-Socket IN / OUT / THRU
Digital Processing	
Converters	20-bit Sigma-Delta, 64/128-times Oversampling
Sampling Rate	46,875 kHz
Display	
Type	2 ½-digit numeric LED-Display
Power Supply	
Mains Voltages	USA/Canada ~ 120 V AC, 60 Hz
	U.K./Australia ~ 240 V AC, 50 Hz
	Europe ~ 230 V AC, 50 Hz
	General Export Model ~ 100-120 V AC, ~ 200-240 V AC, 50-60 Hz
Fuse	100-120 V AC: **125 mA (slow-blow)** 200-240 V AC: **63 mA (slow-blow)**
Power Consumption	10 Watts
Mains Connection	Standard IEC receptacle
Physical	
Dimensions (H * W * D)	1 3/4” (44.5 mm) * 19” (482.6 mm) * 7 1/2” (190.5 mm)
Net Weight	± 2 kg
Shipping Weight	± 3 kg

BEHRINGER is constantly striving to maintain the highest professional standards. As a result of these efforts, modifications may be made from time to time to existing products without prior notice. Specifications and appearance may differ from those listed or shown.

7. WARRANTY

§ 1 WARRANTY CARD

To be protected by this warranty, the buyer must complete and return the enclosed warranty card (signed/stamped by retail dealer) within 14 days of the date of purchase to BEHRINGER INTERNATIONAL (address see § 3). Failure to return the card in due time (date as per postmark) will void any extended warranty claims.

§ 2 WARRANTY

1. BEHRINGER INTERNATIONAL warrants the mechanical and electronic components of this product to be free of defects in material and workmanship for a period of one (1) year from the original date of purchase, in accordance with the warranty regulations described below. If any defects occur within the specified warranty period that are not caused by normal wear or inappropriate use, BEHRINGER INTERNATIONAL shall, at its sole discretion, either repair or replace the product.

2. If the warranty claim proves to be justified, the product will be returned freight prepaid by BEHRINGER INTERNATIONAL within Germany. Outside of Germany, the product will be returned at the buyer's expense.

3. Warranty claims other than those indicated above are expressly excluded.

§ 3 RETURN AUTHORIZATION NUMBER

1. To obtain warranty service, the buyer must call BEHRINGER INTERNATIONAL during normal business hours BEFORE returning the product (Tel.: +49 (0) 21 54 / 92 06 66). All inquiries must be accompanied by a description of the problem. BEHRINGER INTERNATIONAL will then issue a return authorization number.

2. The product must be returned in its original shipping carton, together with the return authorization number, to the following address:

BEHRINGER INTERNATIONAL GmbH
Service Department

Hanns-Martin-Schleyer-Str. 36-38

D - 47877 Willich-Münchheide

3. Shipments without freight prepaid will not be accepted.

§ 4 WARRANTY REGULATIONS

1. Warranty services will be furnished only if the product is accompanied by an original retail dealer's invoice. Any product deemed eligible for repair or replacement by BEHRINGER INTERNATIONAL under the terms of this warranty will be repaired or replaced within 30 days of receipt of the product at BEHRINGER INTERNATIONAL.

2. If the product needs to be modified or adapted in order to comply with applicable technical or safety standards on a national or local level, in any country which is not the country for which the product was originally developed and manufactured, this modification/adaptation shall not be considered a defect in materials or workmanship. The warranty does not cover any such modification/adaptation, irrespective of whether it was carried out properly or not. Under the terms of this warranty, BEHRINGER INTERNATIONAL shall not be held responsible for any cost resulting from such a modification/adaptation.

3. Free inspections, maintenance/repair work and replacement of parts are expressly excluded from this warranty, in particular if caused by inappropriate use. Likewise, the warranty does not cover defects of expendable parts caused by normal wear of the product. Expendable parts are typically faders, potentiometers, switches and similar components.

4. Damages/defects caused by the following conditions are not covered by this warranty:

- ▲ misuse, neglect or failure to operate the unit in compliance with the instructions given in the user or service manuals.
- ▲ connection or operation of the unit in any way that does not comply with the technical or safety regulations applicable in the country where the product is used.
- ▲ damages/defects that are caused by force majeure or by any other condition beyond the control of BEHRINGER INTERNATIONAL.

5. Any repair carried out by unauthorized personnel will void the warranty.

6. Products which do not meet the terms of this warranty will be repaired exclusively at the buyer's expense. BEHRINGER INTERNATIONAL will inform the buyer of any such circumstance. If the buyer fails to submit a written repair order within 4 weeks after notification, BEHRINGER INTERNATIONAL will return the unit C.O.D. with a separate invoice for freight and packing. Such cost will also be invoiced separately when the buyer has sent in a written repair order.

§ 5 WARRANTY TRANSFERABILITY

This warranty is extended exclusively to the original buyer (customer of retail dealer) and is not transferable to anyone who may subsequently purchase this product. No other person (retail dealer, etc.) shall be entitled to give any warranty promise on behalf of BEHRINGER INTERNATIONAL.

§ 6 CLAIM FOR DAMAGES

Failure of BEHRINGER INTERNATIONAL to provide proper warranty service shall not entitle the buyer to claim (consequential) damages. In no event shall the liability of BEHRINGER INTERNATIONAL exceed the invoiced value of the product.

§ 7 OTHER WARRANTY RIGHTS

This warranty does not exclude or limit the buyer's statutory rights provided by national law, in particular, any such rights against the seller that arise from a legally effective purchase contract.

BEHRINGER INTERNATIONAL GmbH, Hanns-Martin-Schleyer-Str. 36-38, D-47877 Willich-Münchheide II
Tel. +49 (0) 21 54 / 92 06-0, Fax +49 (0) 21 54 / 92 06-30