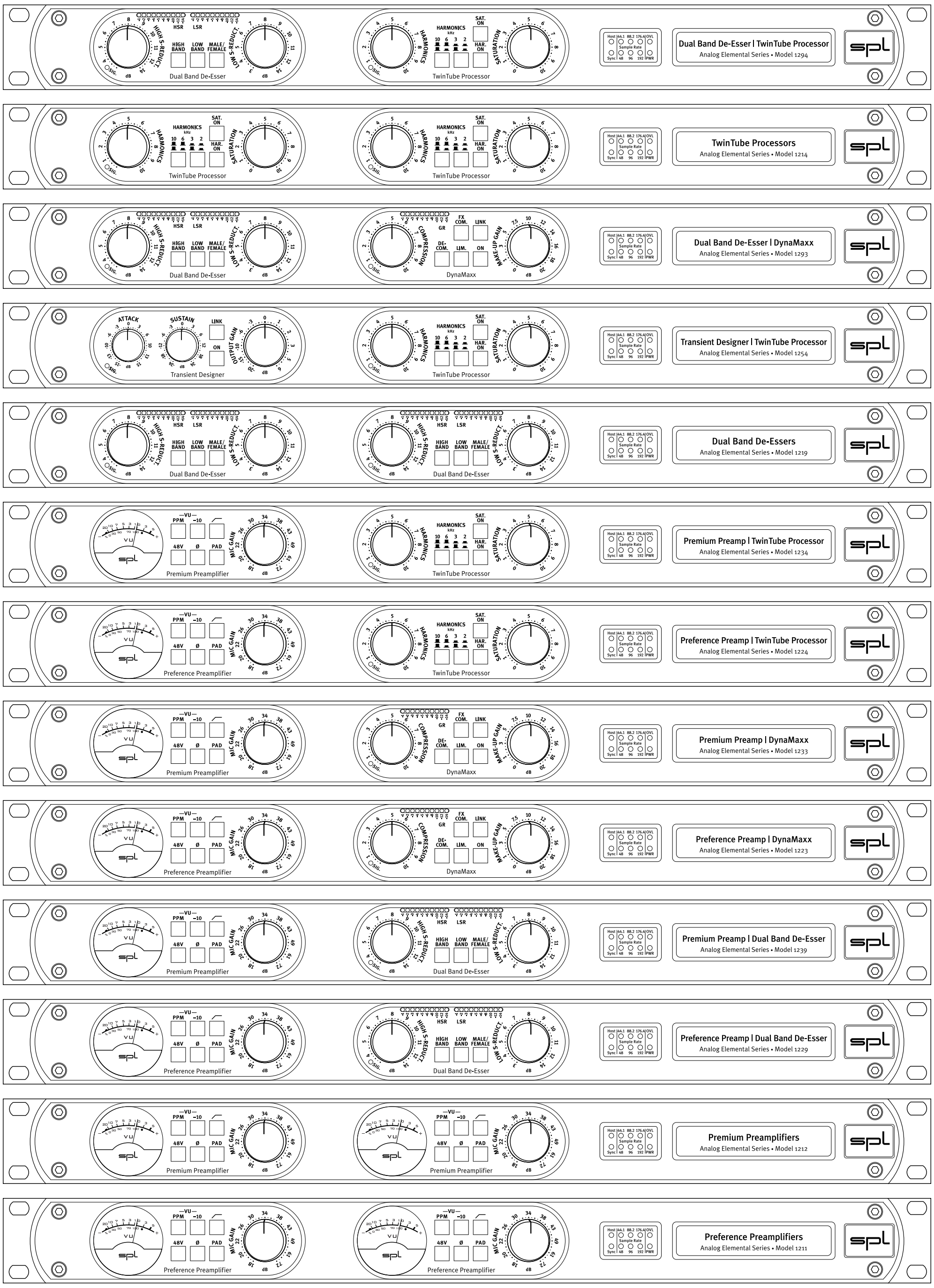
Manual



Analog Elemental Series

Models 1211, 1212, 1214, 1219, 1223, 1224, 1229, 1233, 1234, 1239, 1254, 1293, 1294

Twin Module Series featuring the modules
Premium Mic Pre, Preference Mic Pre, Dual-Band De-Esser, DynaMaxx, Transient Designer und TwinTube

Content

Content

Manual Analog Elemental Series

Version 1.0 – 6/2012

Developers: Wolfgang Neumann, Enzo Triolo, Jens Gronwald

This manual contains a description of the product. It in no way represents a guarantee of particular characteristics or results of use. The information in this document has been carefully compiled and verified and, unless otherwise stated or agreed upon, correctly describes the product at the time of packaging with this document.

Sound Performance Lab (SPL) continuously strives to improve its products and reserves the right to modify the product described in this manual at any time without prior notice.

CE Conformity

The construction of the Analog Elemental Series is in compliance with the standards and regulations of the European Community.

Notes on Environmental Protection

At the end of its operating life, this product must not be disposed of with regular household waste but must be returned to a collection point for the recycling of electrical and electronic equipment. The wheelie bin symbol on the product, user's manual and packaging indicates that. The materials can be re-used in accordance with their markings. Through re-use, recycling of raw materials, or other forms of recycling of old products, you are making an important contribution to the protection of our environment. Your local administrative office can advise you of the responsible waste disposal point. WEEE Registration: 973 349 88.

Contact

SPL electronics GmbH
Sohlweg 80, 41372 Niederkruechten, Germany
Phone +49 (0) 21 63 98 34 0, Fax +49 (0) 21 63 98 34 20
E-Mail: info@spl.info, Website: spl.info

Symbols and Notes

IN THIS MANUAL A LIGHTNING SYMBOL WITHIN A TRIANGLE WARNS YOU ABOUT THE POTENTIAL FOR DANGEROUS ELECTRICAL SHOCKS – WHICH CAN ALSO OCCUR EVEN AFTER THE MACHINE HAS BEEN DISCONNECTED FROM A POWER SOURCE.

AN EXCLAMATION MARK (!) WITHIN A TRIANGLE IS INTENDED TO MAKE YOU AWARE OF IMPORTANT OPERATIONAL ADVICE AND/OR WARNINGS THAT MUST BE FOLLOWED. BE ESPECIALLY ATTENTIVE TO THESE AND ALWAYS FOLLOW THE ADVICE THEY GIVE.

The symbol of a lamp directs your attention to explanations of important functions or applications.

Attention: Do not attempt any alterations to this machine without the approval or supervision of SPL electronics GmbH. Doing so could nullify completely any and all of your warranty/guarantee rights and claims to user support.

Please note and retain this manual. Carefully read and follow all of the safety and operating instructions before you use the machine. Be doubly careful to follow all warnings and special safety instructions noted in this manual and on the unit.

Connections: Only use the connections as described. Other connections can lead to health risks and equipment damage.

Water and humidity: Do not use this machine anywhere near water (for example near a wash basin or bath, in a damp cellar, near swimming pools, or the like). In such cases there is an extremely high risk of fatal electrical shocks!

Insertion of foreign objects or fluids: Never allow a foreign object through any of the machine‘s chassis openings. You can easily come into contact with dangerous voltage or cause a damaging short circuit. Never allow any fluids to be spilled or sprayed on the machine. Such actions can lead to dangerous electrical shocks or fire!

Opening the unit: Do not open the machine housing, as there is great risk you will damage the machine, or – even after being disconnected – you may receive a dangerous electrical shock!

Electrical power: Run this machine only from power sources which can provide proper power in the range from 100 to 250 volts. When in doubt about a source, contact your dealer or a professional electrician. To be sure you have isolated the machine, do so by disconnecting all power and signal connections. Be sure that the power supply plug is always accessible. When not using the machine for a longer period, make sure to unplug it from your wall power socket and from the guitar amp.

Cord protection: Make sure that your power and guitar amplifier signal cords are arranged to avoid being stepped on or any kind of crimping and damage related to such event. Do not allow any equipment or furniture to crimp the cords.

Power connection overloads: Avoid any kind of overload in connections to wall sockets, extension or splitter power cords, or to signal inputs. Always keep manufacturer warnings and instructions in mind. Overloads create fire hazards and risk of dangerous shocks!

Lightning: Before thunderstorms or other severe weather, disconnect the machine from wall power (but to avoid life threatening lightning strikes, not during a storm). Similarly, before any severe weather, disconnect all the power connections of other machines and antenna and phone/network cables which may be interconnected so that no lightning damage or overload results from such secondary connections.

Air circulation: Chassis openings offer ventilation and serve to protect the machine from overheating. Never cover or otherwise close off these openings. Never place the machine on a soft surface (carpet, sofa, etc.). Make sure to provide for a mounting space of 4-5 cm/2 inches to the sides and top of the unit when mounting the unit in racks or on cabinets.

Controls and switches: Operate the controls and switches only as described in the manual. Incorrect adjustments outside safe parameters can lead to damage and unnecessary repair costs. Never use the switches or level controls to effect excessive or extreme changes.

Repairs: Unplug the unit from all power and signal connections and immediately contact a qualified technician when you think repairs are needed – or when moisture or foreign objects may accidentally have gotten in to the housing, or in cases when the machine may have fallen and shows any sign of having been damaged. This also applies to any situation in which the unit has not been subjected to any of these unusual circumstances but still is not functioning normally or its performance is substantially altered. In cases of damage to the power supply and cord, first consider turning off the main circuit breaker before unplugging the power cord.

Replacement/substitute parts: Be sure that any service technician uses original replacement parts or those with identical specifications as the originals. Incorrectly substituted parts can lead to fire, electrical shock, or other dangers, including further equipment damage.

Safety inspection: Be sure always to ask a service technician to conduct a thorough safety check and ensure that the state of the repaired machine is in all respects up to factory standards.

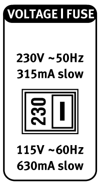
Cleaning: In cleaning, do not use any solvents, as these can damage the chassis finish. Use a clean, dry cloth (if necessary, with an acid-free cleaning oil). Disconnect the machine from your power source before cleaning.

Scope of Delivery and Packaging

The scope of delivery comprises the Analog Elemental Series unit, the power cord, the guarantee card and this manual.

Please keep the original packaging. In case of a service procedure the original packaging ensures a safe transport. It also serves as a safe packaging for your own transports if you do not use special transportation cases.

Hook Up



Be very careful to check that the rear chassis power selection switch is set to the correct local line voltage position before using the unit (230 V position: 220-240 V/50 Hz, 115 V position: 110-120 V/60 Hz)! When in doubt about a source, contact your dealer or a professional electrician.

Before connecting any equipment make sure that any machine to be connected is turned off. Follow all safety instructions on page 5 and read further information on connections on page 10 onwards.

Placement

Place the unit on a level and stable surface. The unit's enclosure is EMC-safe and effectively shielded against HF interference. Nonetheless, you should carefully consider where you place the unit to avoid electrical disturbances. It should be positioned so that you can easily reach it, but there are other considerations. Try not to place it near heat sources or in direct sunlight, and avoid exposure to vibrations, dust, heat, cold or moisture. It should also be kept away from transformers, motors, power amplifiers and digital processors. Always ensure sufficient air circulation by keeping a distance of 4-5 cm/2 inches to the sides and top of the unit.

Introduction

The Analog Elements

Our research and development efforts are always ruled by two maxims: innovation and ease of use. In many occasions both approaches get interleaved so that a good product idea is, at the same time, very easy to use, when implemented as a musically sound solution. Hence, it is no surprise that our „one knob" devices, like the SPL De-Esser and DynaMaxx compressor, have been successful for so many years and are considered classics among our products.

Even though originally conceived in 1U 19-inch format, the possibility of applying the modular concept to our preamps and processors, in order to satisfy the wishes of customers with very specific needs, has always been lingering in our minds. The introduction of the RackPack Series finally allowed for a modular system that can accommodate six different modules in three frame options:

- Premium Mic Pre — High-end microphone preamplifier with transformer
- Preference Mic Pre — Reliable preamplifier ideal for multi-channel applications
- Dual-Band De-Esser — Second generation of our unique De-Esser with phase cancellation technology
- DynaMaxx — Second generation of the best set & forget compressor with signal-dependent automation
- Transient Designer — Second generation processor by the original inventor of the transient tool
- TwinTube — Tube processor for saturation and presence effects

Key to the selection of modules was the idea of offering an elemental collection of combinable analog components for music production. Some modules were developed from scratch with this in mind (Premium and Preference preamps, TwinTube), while some others were enhanced and adapted to the modular concept (which resulted in the second generation of our beloved DynaMaxx, Dual-Band De-Esser and Transient Designer processors). In order to put to use all the work invested in developing our analog elements and translate it into single units, we created the Analog Elemental Series.

The Next Generation Signal Processors

The development of the processors brought with itself the following advances:

- On all three processors that were revised the operating voltage was increased from +/-15 to +/-18 volts. This allows us to enhance the performance of a circuit, considering that the basis of every circuit is the voltage/performance ratio. Thus, dynamic range, signal-to-noise ratio and THD can be improved.
- Dual-Band-De-Esser: two bands for more precise and flexible processing.
- DynaMaxx: modifications to allow for the optional fitting of a transformer.
- Transient Designer: additional output level control — an important enhancement to the device's operation which allows for an immediate control of level differences.

Concept and Categories of the Analog Elemental Series

Every Analog Elemental unit combines two modules. Depending on the application, the 13 resulting units can be sorted into three product categories:

- **Preamplifiers** consisting of two Premium or Preference modules — the outstanding RackPack Preamps in 1U 19-inch units.
- **Channel Strips** consisting of preamp and processing modules — perfect front ends with optimally matched processing for vocals or instruments.
- **Processing Units** to complement analog outboard equipment.

The Analog Elemental concept makes it possible to build any imaginable configuration with SPL's elemental processing tools. Every module combination results in an exceptional and unique unit. We have summarized here the most important aspects of the different configurations.

Dual-Channel Preamplifiers

Premium Preamplifiers
It is not easy to make high-quality recordings. An exceptional sonic foundation to work upon is a must. This can be achieved with microphones and preamps whose impulse response and transduction capabilities are not limited — just like the SPL Premium Preamps. Created by SPL founder Wolfgang Neumann, these classic solid-state masterpieces featuring an integrated transformer seduce anybody that lends an ear to them. The two-channel unit resulting from the combination of two Premium Preamp Modules could withstand comparison to any product, regardless of their price. Model Number: 1212.

Preference Preamplifiers
Our goal when developing the Preference Preamps was to match professional sound quality to the highest reliability possible. Our decade-long experience designing and manufacturing microphone and instrument preamps guarantees the best price-performance ratio. The results can be heard in all sorts of situations — regardless of whether it is live or in the studio, the SPL Preference Preamps provide a rock-solid foundation for any production. And they are also perfectly suited to replace the integrated preamps of audio interfaces and mixers. Model Number: 1211.

Channel Strips

Premium or Preference Preamp & DynaMaxx
Do you play or have to record instruments often? There are lots of ways to do it, but then again, it all comes down to the end result. Our way to achieving outstanding results is arguably the shortest one, considering that it consists of only two knobs: „Gain“ (preamplification) and „Compression.“ This applies to vocal recordings as well. The high-end alternative to the Preference/DynaMaxx Combo, Model 1223, is the Premium/DynaMaxx unit, Model 1233.

Premium or Preference Preamp & Dual Band De-Esser
The combination of a Premium or Preference Preamp with the Dual-Band De-Esser preserves the sound character and timbre of voices, even with extreme settings. An unbeatable combination for live applications. The high-end alternative to the Preference/Dual-Band De-Esser Combo, Model 1229, is the Premium/Dual-Band De-Esser unit, Model 1239.

Premium or Preference Preamp & TwinTube
The combination of a preamp with the TwinTube effects processor allows for fascinating sound shaping possibilities. Two totally independent tube effects — Saturation and Harmonics — make vocals sound more sonorous and rootsy, while highlighting their presence and emphasizing the harmonics. Presence, authority, cutting edge, glaze... take full advantage of the character of the vocals. The high-end alternative to the Preference/TwinTube Combo, Model 1224, is the Premium/TwinTube unit, Model 1234.

Processing Units

Dual-Band De-Essers
Those who need two Dual-Band De-Esser units will find in this combination the perfect 1U 19“ solution. In case more units are needed, a RackPack frame is the best way to go (3U 19“ rack frames that can host four or eight Analog Elemental Series modules).

Dual Band De-Esser & TwinTube
This combination effectively expands any sound engineer‘s processing needs: De-Essing and TwinTube processors are ideal for vocal tracks and are the perfect complement to any respectable preamp. Model Number: 1294.

Dual-Band De-Esser & DynaMaxx
A very nice processing combination for vocals. Being able to have the two most straightforward one-knob SPL processors under the same hood is not only exciting but also comforting — it is very easy to abandon oneself to their magic and prowess. Model Number: 1293.

TwinTube Processors
Considered one of the world‘s finest tube effects processors, the TwinTube provides the two single most important artifacts generated by tubes: Saturation and Harmonics. The warmth, presence and suppleness analog tube and coil filtering provide can enrich any production, regardless of the music genre. Model Number: 1214.

Transient Designer & TwinTube
The second generation Transient Designer (with output level control) and the exceptional TwinTube together in one processing unit — a combination that can certainly fill the gaps of many studios when it comes to analog processing arsenal. This is a perfect example of something that is much more than the sum of its parts: the Transient Designer allows to clearly emphasize sound parts that can then be processed with the TwinTube. Very effective on all sorts of percussive signals, the tube sound complements perfectly the Transient Designer and opens up an unprecedented amount of sound shaping possibilities. Model Number: 1254.

Optional Features

- Optional for all processing modules: Lundhal transformers at the input and output stages. Premium preamplifier modules integrate input and output transformers. Preference preamplifier modules cannot be fitted with transformers. Transformers are not visible from the outside and are only available as a factory option during purchase. Retrofitting is not contemplated. More information on transformers below.
- Optional for every Analog Elemental unit: additional digital output via a converter module. For more information on the Converter Module refer to page 9.
- Additional functions for Channel Strip modules: channel strip switcher on the rear. When engaged, this switch internally forwards the output signal of the first module directly into the input of the second module, making external cabling unnecessary.

Information on Transformers

All processing modules can be equipped with input and output transformers. Exceptions: Preference Mic Pre is not available with transformers, Premium Mic Pre always comes with inout and output transformers.

We think a good part of the „warmth“ that is commonly associated with vintage gear comes from transformers. With transformers the low end and lower mids sound rounder, full-bodied with more punch. The top end gets a silky touch and benefits from improved presence without sounding boosted. Reasons are reduced odd harmonics (which produce harsh top end impressions) and a slower characteristic compared to electronic stages which causes a more voluminous sound.

We recommend transformers especially for vocals while electronic stages can be better for highest precision in signal transmission (transients), but in the end it‘s a question of personal taste, applications or for example which mics are in use.

Transformers transmit the signal by induction. This means there is no static connection (like a wire) to transmit the signal. This protects the unit from damage caused by power failures in the input/output connections. In addition, hum problems are effectively avoided if balanced connections are used throughout. That‘s why transformers can be very interesting for live units or in units used for facilities that need highest operational safety (broadcast, sound reinforcement). Another notable advantage for many live and recording applications is that very long cable lengths are possible with no loss in signal quality. At the end of the day, it depends on personal taste, main applications or for example on the microphone and other equipment if transformers are required or not.

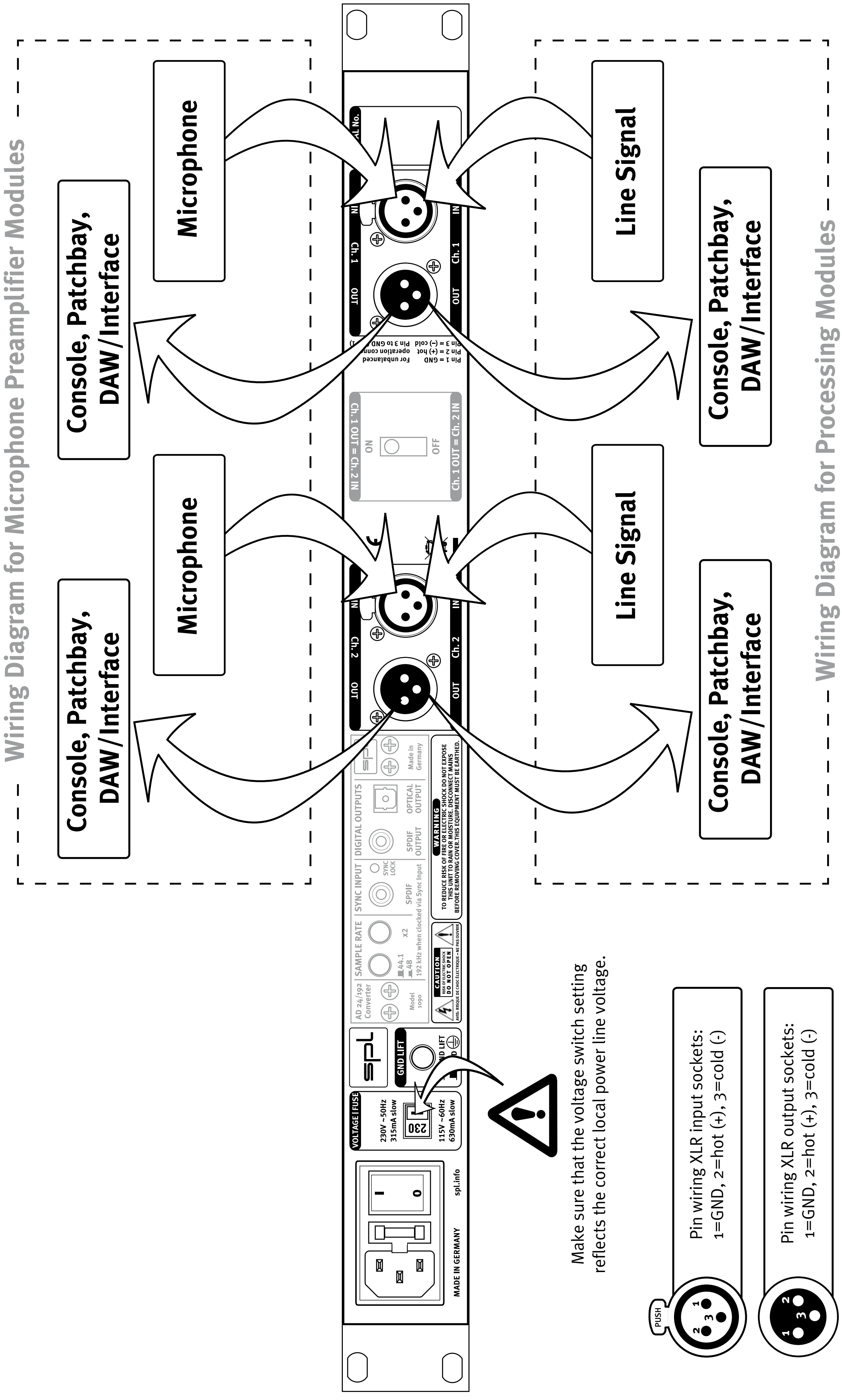
Common Technical Specifications

Power Supply	Toroidal transformer
Fuses	230 V AC, 50 Hz: 315 mA
	120 V AC, 60 Hz: 630 mA
Voltage Selector	115V/230V
Power Consumption	@ 230 V: 9,1 W/10,8 VA
	@ 115 V: 5,6 W/7,1 VA

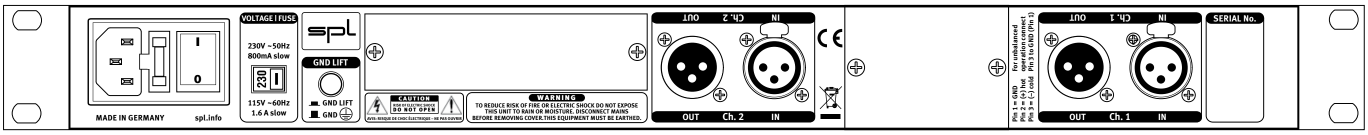
Dimensions and Weight

Housing (W x H x D)	482 x 88 x 320 mm (depth includes knobs and sockets)
Weight	ca. 4,6 kg
Weight with Premium Module(s):	ca. 4,9 kg
Weight with TwinTube Module(s):	ca. 4,8 kg

Rear Panel | Wiring Diagram

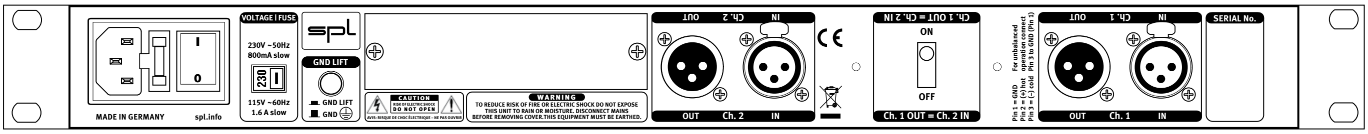


Depending on their configuration and additional features, the Analog Elemental Series modules can have three different rear panels. The additional elements are the optional AD converter (digital output) and the channel strip switch, which internally forwards the output signal of the first module to the input of the second module, ruling out the need for external cabling.



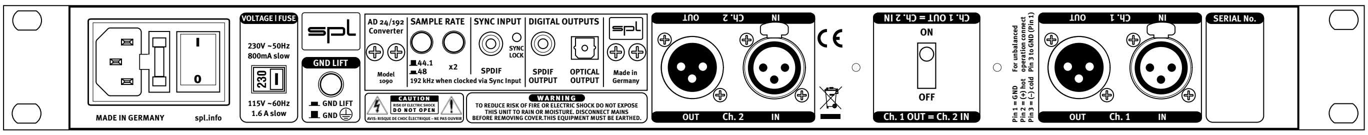
Rear Panel 1

Without converter or channel strip switch (configuration example: two preamp module).



Rear Panel 2

Without converter, with channel strip switch (configuration example: channel strip consisting of preamp and processing module).

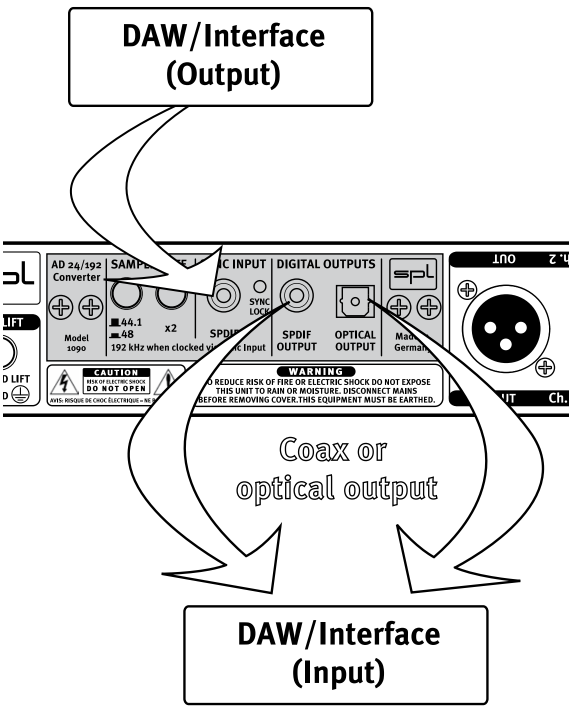


Rear Panel 3

With converter and channel strip switch (configuration example: channel strip consisting of preamp and processing module, including digital output).

Optional Converter: Wiring Diagram

The optional AD converter provides an additional digital output. The diagram shows how to connect it to an interface. For more detailed information refer to the converter module user's manual.



Signal Connections

Turn off the unit before connecting or disconnecting any cable or equipment to it. Otherwise you risk the possibility of damaging your ears or equipment.

Input and Output Electronics

The input and output electronics are based upon bridge circuits that keep the signal flow constant, regardless of malfunctioning equipment and power outages (power fail safety by relay hard bypasses). The bridge and insert circuits rely on high-quality relays. Contact surfaces are gold-plated to provide better conductivity and encapsulated to avoid external influences due to climate or atmospheric conditions.

XLR sockets

All signal connections are made via balanced XLR connectors. Inputs are always female and accept male connectors; outputs are always male. All in all, a very comprehensible principle.

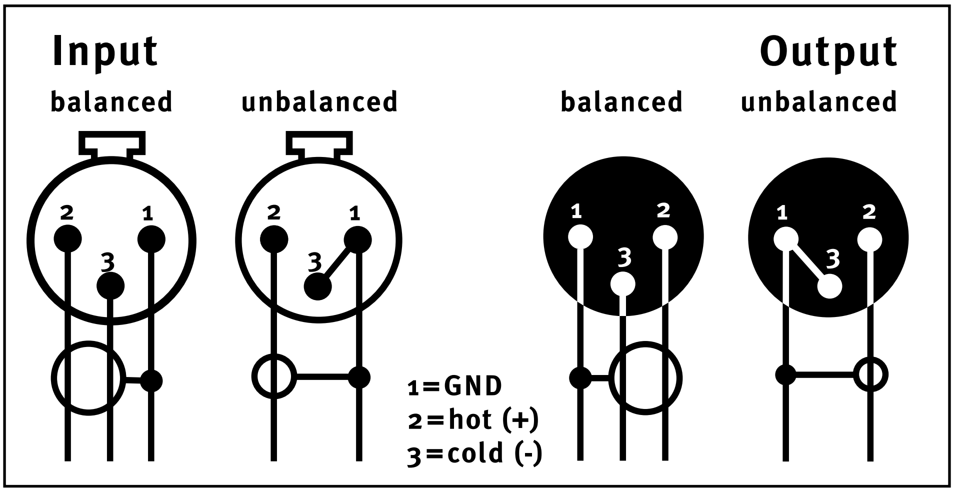
Balanced connections

It is impossible to exclude interferences when a single audio signal is transmitted. Shielding is effective against electric, but not against electromagnetic influences. Motors, transformers, and alternating current can always induce interferences. But even if the transmission would succeed, differences in ground potentials between driver and receiver would produce disturbances.

In balanced connections a reference signal with reversed polarity is transmitted additionally to the audio signal through a second wire. The ground signal is routed separately through a third wire. Input and output stages are drivers and receivers, and the receiving stage can suppress interferences by subtracting the difference between audio and reference signal.

Unbalanced connections

Unbalanced connections from and to RCA or 1/4“ TS sockets can be made without adaptors to the balanced XLR sockets. The correct wiring is important. The diagram shows the pin configuration of the XLR sockets and how to correctly connect them for unbalanced connections:

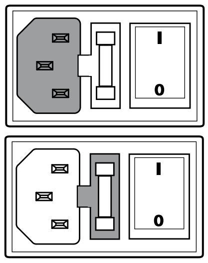


Connections to RCA sockets are always unbalanced, a wiring to jack connectors can be both balanced (1/4“ TRS/stereo jack) or unbalanced (1/4“ TS/mono jack). We recommend to use individually configured cables from XLR to RCA or jack sockets instead of adaptors. You can get cables in any needed configuration from audio dealers. With the diagram above, the dealer can ensure to provide the appropriate cable for your application.

Power connection and fuse

Connect the power cord to the rear MAINS INPUT socket. Transformer, power cord and case connection conform to VDE, UL and CSA requirements.

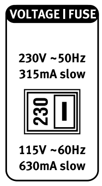
The fuse is accessible from outside and placed right behind the flap right from the socket. Fuse ratings are 315 mA slow blow (230 volts) or 630 mA slow blow (115 volts).



Voltage Selector

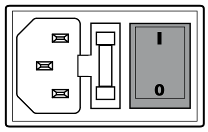
The rear panel VOLTAGE selector sets the local line voltage (115 V position: 110-120 volts/60 Hz, 230 V position: 220-240 volts/50 Hz). The diagram shows the correct switch position for 230 V power supply.

BEFORE you connect electrical power make sure that the VOLTAGE selector setting reflects the correct local power line voltage.



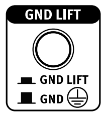
Power switch

Use the POWER switch on the rear panel to turn the unit on or off. The VU-meters on the front panel will light on as soon as you turn the unit on, regardless of the position of the ACTIVE push button. Thus, they fulfill a second function as power indicators.



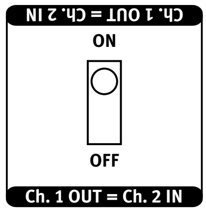
GND Lift

The rear panel GND LIFT switch eliminates hum by separating the internal ground from the unit's housing ground. Hum can, for example, result when this unit's housing has a common ground connection with other devices that might have a different ground potential. The switch is usually deactivated to retain the shielding of the housing.



Channel Strip Switch (Ch. 1 OUT = Ch.2 IN)

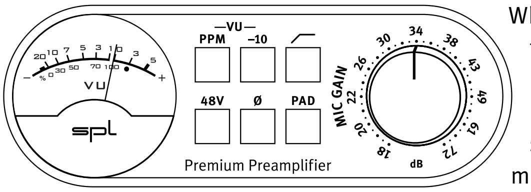
The channel strip switch connects two modules internally with each other, as long as the signal flow allows for it. For instance, with an Analog Elemental unit consisting of a preamp and a processing module (Channel Strip configuration). When the switch is set to „ON“, the output signal of the preamp feeds the input of the processing module directly. This simplifies the signal flow and makes it unnecessary to connect the output and input of the modules externally via cables.



The Modules

Each of the Analog Elemental Series units combines two of the following modules. For more information on the design of the devices, refer to the Introduction on page 8. Here you will find a detailed description of the single modules.

Premium Mic Pre | Introduction

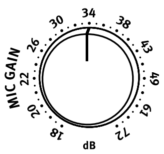


When recording acoustic instruments or vocals, using a microphone is inevitable. The actual output level of a microphone is very low and therefore has to be boosted to studio or line level (0 dB) with a preamplifier. Sometimes signals have to be boosted by a factor of 2000 or more. As a consequence, the resulting sound quality provided by the preamp is of paramount importance, so a good microphone preamp that does not overdrive is the definitive requirement in order to record acoustic instruments or vocals with sufficient dynamics and untainted sound.
The section “technology“ from page 19 explains how the Premium Mic Pre complies with these requirements on the highest level and how it delivers premium results even in difficult and critical set-ups.
Even the less technically interested users get an impression of the great efforts it takes in development, circuitry design, selection of parts and assembly to create a product on the Premium Mic Pre’s quality level. From there we deduce a superior price/performance ratio – emphasizing on the performance which lends the Premium Mic Pre a world-class sound quality.

Features

- The Premium Mic Pre offers preamplification values of up to +72 dB with lowest noise operation and a high common mode rejection.
- Input stage and output are equipped with Lundahl transformers (High Performance Series).
- A VU meter with two modes for average levels (VU) and peak levels (PPM) displays the output levels.
- A very stable phantom power supply (48 V) is provided to power condenser microphones.
- The polarity of the microphone can be switched with the phase reverse switch.
- A high-pass filter protects against low frequency interferences.
- The signal LED indicates that an input signal is present

Premium Mic Pre | Control Elements



Mic Gain

With the Mic Gain control you can regulate the preamplification of the microphone signal. It ranges from +18 dB to +72 dB. The input stage can handle input levels of up to 10 dB. The value set with the Mic Gain control defines the output level.

When you set the Mic Gain you have to consider the type of microphone that you are using (dynamic or condenser microphone) as well as its sensitivity. The sensitivity of a dynamic microphone is at around 2 mV/Pa whereas the sensitivity of a condenser microphone can be up to 20 mV/Pa. The result is a difference in output of 20 dB.

You should also consider the sound pressure level of the sound source, the distance of the microphone to it and the acoustics of the room when you set the preamplification.

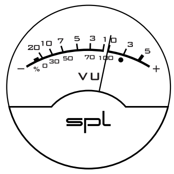
About Leveling

Initially you should always deactivate the -10 dB button to have correct values displayed by the VU meter (see “-10 dB” below). Now turn up the Mic Gain control until the VU meter displays maximum levels between 0 dB and +3 dB. At this level you don’t risk any overdrive when you experience sudden and unexpected peaks in the level of the source. Always remember that the VU meter only shows average values and that a peak level can be much higher (up to +10 dB). If necessary, turn on the PPM mode to see the actual peak levels.

Usually levels of around 0 dB and +3 dB are safe. If, however, you experience very high peaks already at minimum Mic Gain values (i.e. drums, brass instruments), you can activate the PAD function (see “PAD” on page 18). The input level is now reduced so that you can regulate Mic Gain again in a useful range.

If you know in advance that the level will be very consistent you can always turn up Mic Gain. In this case you can activate the -10dB button to have more headroom in the display.

VU Meter



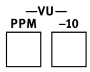
The VU meter displays the output level of the preamp. The gauge indicates levels from -20 dB to +5 dB. If necessary you can lower the sensitivity by 10 dB so that the gauge goes up to +15 dB output level (see “-10dB” below and “About Leveling” above).

The VU meter is custom made to meet SPL specifications and assures a balanced optical perception thanks to it’s optimized ballistics.

An especially interesting feature is the option to switch between two display modes: VU mode and PPM mode. The VU mode (VU=Volume Unit) displays average levels, thus provides information on the overall loudness. The PPM mode (PPM=Peak Program Meter) displays the peak levels.

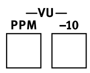
The integration time of the display complies with the BBC requirements. In VU mode the rise time up to 0 dB is approx. 300 ms. In PPM mode the rise time up to 0 dB is about 2 ms and the release time down to -20 dB is at a “slow” 1,5 seconds. This calibration ensures to display even short peaks up to about +6 dB since the needle does not have to travel the entire distance of the gauge every time.

PPM



The PPM button allows to switch the VU metering characteristics from VU display mode (button is deactivated and not illuminated) to PPM display mode (button is active and illuminated). A/D converter display also make use of the PPM mode. Monitoring peak levels is most important to avoid overloading the converter and to prevent audible distortion. Peak levels are always above the average levels. It may make sense in most cases to also press the -10 dB button to prevent the needle from getting stuck on the right side of the gauge.

-10



With this button you can change the sensitivity of the VU display. If you press the -10 dB button (button is illuminated), the sensitivity of the display is lowered by 10 dB. With the needle for example at 0dB, a value of +10 dB is displayed. Now you can read values of up to +15 dB.

You always have to add these 10 dB to the displayed values, for example if you compare these values with those of an A/D converter. Remember that A/D converters show peak levels, not average levels. For comparison press the PPM button to activate the PPM mode.

48V

48 V Phantom Power Supply

The Premium Mic Pre provides 48 volt phantom power for microphones requiring external current (generally condenser microphones). Such microphones are dependent upon a clean, consistent and noise-free power supply for optimal operation and audio quality. The Premium Mic Pre continuously delivers precisely 48 V and a maximum current of 14 mA, which will power all microphones.

IMPORTANT: All microphones with balanced, ground-free outputs can be used with the phantom power activated. Unbalanced microphones may only be used with phantom power deactivated.

Phantom power should only be activated when using microphones that require it. Please be sure to deactivate phantom power with all other microphones (including tube microphones which are supplied from their own power supply, thus need no extra phantom power supply).

ALWAYS FOLLOW THESE RULES WHEN WORKING WITH PHANTOM POWER – THE INPUT STAGES OF THE PREMIUM MIC PRE CAN BE DAMAGED IF YOU DO NOT ACT ACCORDINGLY:

1. First connect the microphone to the Premium Mic Pre.
2. Then activate phantom power – you can start working now.
3. When you have finished recording, turn the phantom power off first.
4. **DO NOT disconnect the microphone from the Premium Mic Pre UNTIL phantom power has been switched off for a minute and all residual current is discharged.**

PAD

Pad

The Pad function allows you to attenuate the input signal by 20 dB so that you can process even very high levels, i.e. from drums or brass instruments.

If the VU meter shows levels above +3 dB even while Mic Gain is set very low (and the -10 dB button deactivated), the time has come to press the Pad button. It illuminates when it is activated.

ø

Phase Reverse

With the phase reverse button you invert the polarity of the microphone signal. When not pressed (button is not illuminated) the polarity is in phase. After pushing the button (button is illuminated) the polarity is inverted.

The phase reverse feature comes in very handy if you have to switch the polarity of the XLR input according to the polarity of the microphone or the microphone cable. The pin wiring of the XLR sockets are as follows: Pin 1 = ground, Pin 2 = hot (+), Pin 3 = cold (-).

Sometimes it is useful to switch the polarity of a microphone, for example in the case of M/S miking. A second classic application is the miking of a snare drum with two microphones that are placed above and below the snare: Since both drum heads move in the same direction when the drum is played, the microphones are out of phase. Switch the polarity of the bottom mic and you avoid any cancellations when you join both signals in the mix.

High-Pass Filter

The high-pass filter, also known as rumble filter, helps to eliminate any interferences within the lowest frequencies. The first order filter operates smoothly with 6 dB per octave, starting from 75 Hz with -3 dB. This characteristic usually helps in most cases and has the least sonic disadvantages. If you need to filter on a more extreme scale, even second order filters (12 dB/octave) are overstrained frequently and sonic disadvantages become more and more apparent. In those cases a variable filter is the means of choice.

Triple Stage Preamp

The preamplifier circuitry of the Premium Mic Pre is composed of three stages: a transformer, a discrete differential amplifier stage and an instrument preamplifier all contribute (in different shares) to the main amplification. The three stage setup firstly ensures a load distribution that minimizes the risk of overloads. Secondly, each stage can be optimally configured through select components and sophisticated circuits – a decisive advantage over single IC circuitries where the main amplification is achieved by just one stage.

In addition, the potentiometer (as the control element of the differential amplifier stage) only has to cover a range of approximately 68 dB while the maximum amplification is at around 80 dB. In practice this means that even very high amplification values are still outside of the extreme control range positions – which are critical for any potentiometer.

Stage 1 – The Input Transformer

Inherent part of the Premium Mic Pre design are the input and output transformers of the High Performance Series by Lundahl. These transformers are lavishly handcrafted. They replace common input and output balancing stages. These classic components offer a very high sound quality and common mode rejection but they are also very reliable and provide connections and signals of superior quality.

Lundahl transformers have a very high reputation in terms of manufacturing quality and the resulting longevity as we have seen proven impressively throughout many years of experience.

In any microphone preamplifier the input transformer is of special relevance as it is an integral part of the preamp circuitry: it contributes to the main amplification through a passive and permanent boost of, in this case, 6 dB. The advantage of passive amplification over active amplification is that it does not add any noise. A second advantage of integrating this passive amplification is a signal processing that is, as a matter of principle, lower in overdrive and noise throughout all following stages simply because the rest of the electronic circuit is charged with 6 dB less at any setting of the amplification.

In addition, transformers ensure a galvanic isolation, preventing any disturbing or damaging voltages from being carried in any of the two directions. Electromagnetic, high frequency or digital interference has no more influence. Problems with humming e. g. in a live environment that have been caused by differences in the potentials between the stage and the FOH do not occur. Even a voltage that has accidentally or through technical failure been connected to a ground line cannot be transmitted. So transformers can exclude even mishaps or problems in an installation reaching the categories “improbable” till “unbelievable” ...

Further, the phantom powering of microphones does not require any condensers in the preamp socket which has further sonic advantages.

From our personal listening impression we can recommend transformers in any case. The advantages in operational safety can not be overestimated especially in critical or complex installations for studio, live or broadcast applications.

Stage 2 – The Discrete Differential Amplifier

From the input transformer, the signal is routed to a discrete differential amplifier based upon a quad parallel transistor circuitry. This parallel circuit of eight single transistors reduces noise remarkably through distribution of load.

The discrete differential amplifier is the central amplifying stage; here the amount of the amplification is controlled by a current-carrying source. Control systems that are triggered by current rather than voltage have the main advantage that possible negative effects of the potentiometer will not affect the audio signal.

The discrete differential amplifier offers a maximum amplification of up to 68 dB. A servo drive circuit actively eliminates DC offsets of the differential amplifier. Servo drive circuitry minimizes DC offsets more effectively than conventional solutions that utilize capacitors by setting the DC offset to almost 0 mV. In addition, the active servo drive solution has sonic advantages over passive capacitors as it tends to produce less distortions.

Stage 3 – The Instrumentation Amplifier

An instrumentation amplifier is following the discrete differential amplifier to produce the output signal's voltage. Functionally a summing amplifier it eliminates possible disturbing voltages and also contributes to the main amplification with an additional +6 dB.

Foil and Styroflex Capacitors

Only the best MKP and styroflex capacitors are used in the various circuits. They sound more open and dynamic in contrast to the conventional ceramic capacitors.

Output Stages

The Premium Mic Pre is based upon a transformer output capable of driving very long cables (up to a few hundred yards, depending on the capacity of the cables and the input stages on the other end). The maximum output level is +22 dBu.

Premium Mic Pre | Technical Specifications

Audio

Frequency Range		10 Hz- 68 kHz (-3 dB)
CMR		-87 dBu
(@ 1kHz with -30 dBu Input level and 30 dB Gain)		
THD&N @ 1 kHz		
Input Level	Gain	THD&N
-30 dBu	30 dB	0,0071%
-60 dBu	60 dB	0,078%
S/N Ratio	Gain	A/N Ratio A-weighted
	80 dB	-48,3 dBu
	60 dB	-64,2 dBu
	30 dB	-89,0 dBu
E.I.N. *(Equivalent Input Noise)*:		-128,3 dBu
Dynamic Range		111,0 dB

Input

XLR socket, transformer balanced	
Impedance unbalanced	ca. 2,0 kOhm
Impedanz balanced	ca. 4,0 kOhm
Max. Input Level	+10 dBu, +30 dBu with Pad activated

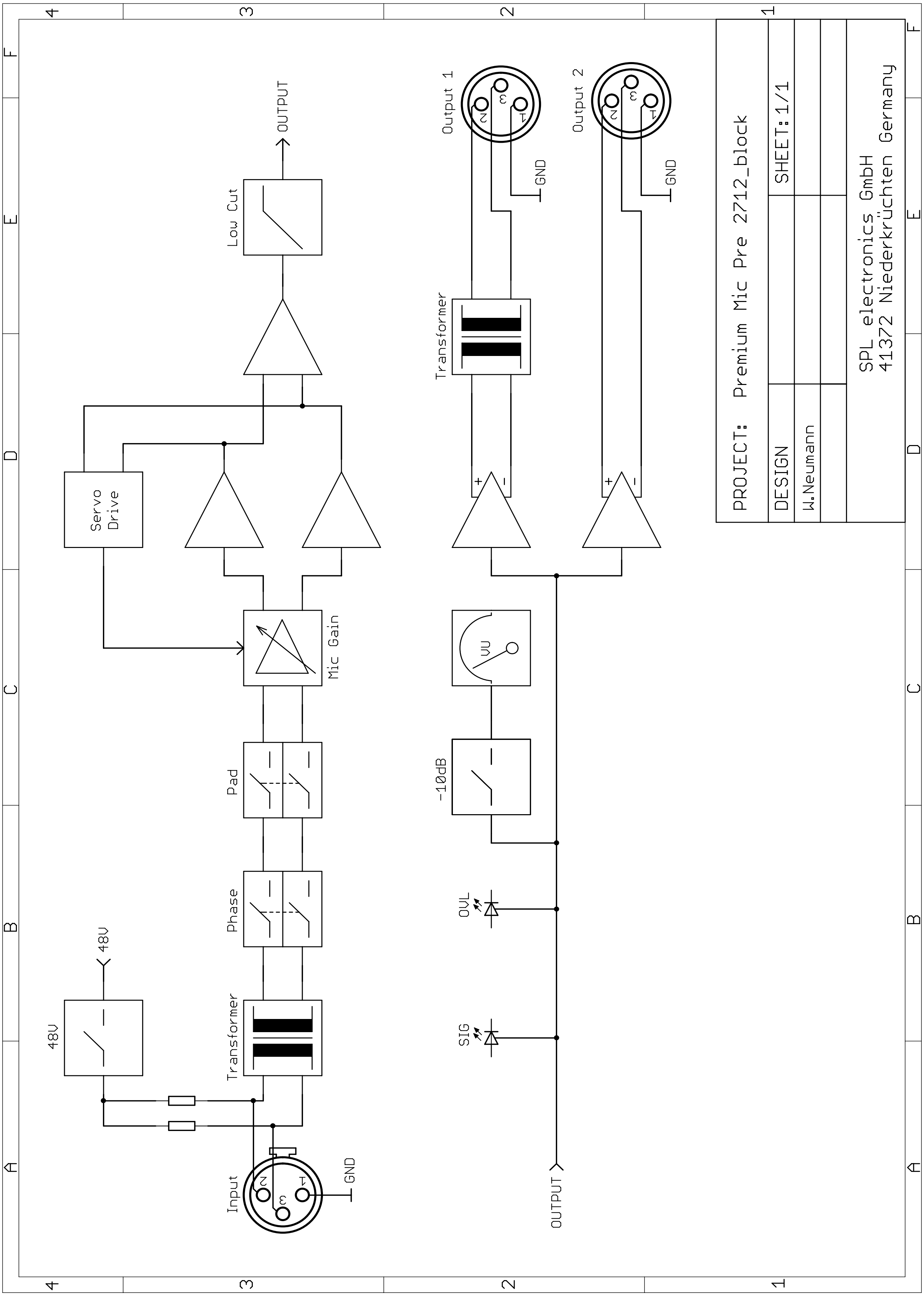
Output

XLR socket, transformer balanced	
Impedance unbalanced	ca. 60 Ohm
Impedance balanced	ca. 120 Ohm
Max. Output Level	+22 dBu

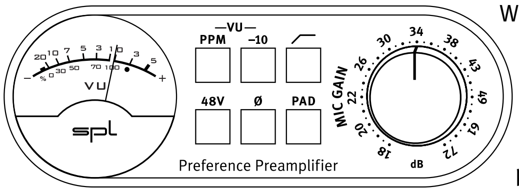
Control Elements

Mic Gain Range	18 dB – 72 dB
Pad Switch	-20 dB
High-Pass Filter	fg = 75 Hz (-3 dB)
Phantom Power Supply	48 V

0 dBu = 0,775 V. Sugbject to change without notice.


Input
GND
48V
48V
Transformer
Phase
Pad
Mic Gain
Servo Drive
Low Cut
OUTPUT
SIG
OVL
-10dB
VU
OUTPUT
Transformer
Output 1
GND
Output 2
GND
PROJECT: Premium Mic Pre 2712_block
SHEET: 1/1
DESIGN
W.Neumann
SPL electronics GmbH
41372 Niederkrüchten Germany

Preference Mic Pre | Introduction



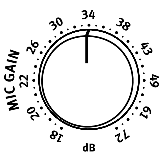
When recording acoustic instruments or vocals, using a microphone is inevitable. The actual output level of a microphone is very low and therefore has to be boosted to studio or line level (0 dB) with a preamplifier. Sometimes signals have to be boosted by a factor of 2000 or more. As a consequence, the resulting sound quality provided by the preamp is of paramount importance, so a good microphone preamp that does not overdrive is the definitive requirement in order to record acoustic instruments or vocals with sufficient dynamics and untainted sound.

The section “Technology” on page 24 explains how the Preference Mic Pre meets these requirements.

Main Features

- The Preference Mic Pre offers preamplification values of up to +72 dB with lowest noise operation and a high common mode rejection.
- A VU meter with two modes for average levels (VU) and peak levels (PPM) displays the output levels.
- A very stable phantom power supply (48 V) is provided to power condenser microphones.
- The polarity of the microphone can be switched with the phase reverse switch.
- A high-pass filter protects against low frequency interferences.

Preference Mic Pre | Control Elements



Mic Gain

With the Mic Gain control you can regulate the preamplification of the microphone signal. It ranges from +18 dB to +72 dB. The input stage can handle input levels of up to 18 dB. The value set with the Mic Gain control defines the output level equally for both Output 1 and Output 2.

When you set the Mic Gain you have to consider the type of microphone that you are using (dynamic or condenser microphone) as well as its sensitivity. The sensitivity of a dynamic microphone is at around 2 mV/Pa whereas the sensitivity of a condenser microphone can be up to 20 mV/Pa. The result is a difference in output of 20 dB.

You should also consider the sound pressure level of the sound source, the distance of the microphone to it and the acoustics of the room when you set the preamplification.

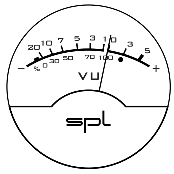
About Leveling

Initially you should always ensure to have deactivated the -10 dB button so that the VU meter displays correct values (see “-10 dB” onthe next page). Now turn up the Mic Gain control until the VU meter displays maximum levels between 0 dB and +3 dB. At this level you don’t risk any overdrive when you experience sudden and unexpected peaks in the level of the source. Always remember that the VU meter only shows average values and that a peak level can be much higher (up to +10 dB). If necessary, turn on the PPM mode to see the actual peak levels.

Usually levels of around 0 dB and +3 dB are safe. If, however, you experience very high peaks already at minimum Mic Gain values (i.e. drums, brass instruments), you can activate the PAD function (see “PAD” on page 24). The input level is now reduced so that you can regulate Mic Gain again in a useful range.

If you know in advance that the level will be very consistent you can always turn up Mic Gain. In this case you can activate the -10dB button to have more headroom in the display.

VU Meter



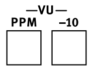
The VU meter displays the output level of the preamp. The gauge indicates levels from -20 dB to +5 dB. If necessary you can lower the sensitivity by 10 dB so that the gauge goes up to +15 dB output level (see “-10 dB” below and “About Leveling” on the previous page).

The VU meter is custom made to meet SPL specifications and assures a balanced optical perception thanks to it’s optimized ballistics.

An especially interesting feature is the option to switch between two display modes: VU mode and PPM mode. The VU mode (VU=Volume Unit) displays average levels, thus provides information on the overall loudness. The PPM mode (PPM=Peak Program Meter) displays the peak levels.

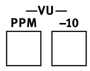
The integration time of the display complies with the BBC requirements. In VU mode the rise time up to 0 dB is approx. 300 ms. In PPM mode the rise time up to 0 dB is about 2 ms and the release time down to -20 dB is at a “slow” 1,5 seconds. This calibration ensures to display even short peaks up to about +6 dB since the needle does not have to travel the entire distance of the gauge every time.

PPM



The PPM button allows to switch the VU metering characteristics from VU display mode (button is deactivated and not illuminated) to PPM display mode (button is active and illuminated). A/D converter display also make use of the PPM mode. Monitoring peak levels is most important to avoid overloading the converter and to prevent audible distortion. Peak levels are always above the average levels. It may make sense in most cases to also press the -10 dB button to prevent the needle from getting stuck on the right side of the gauge.

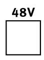
-10 dB



With this button you can change the sensitivity of the VU display. If you press the -10 dB button (button is illuminated), the sensitivity of the display is lowered by 10 dB. With the needle for example at 0dB, a value of +10 dB is displayed. Now you can read values of up to +15 dB.

You always have to add these 10 dB to the displayed values, for example if you compare these values with those of an A/D converter. Remember that A/D converters show peak levels, not average levels. For comparison press the PPM button to activate the PPM mode.

48 V Phantom Power Supply



The Preference Mic Pre provides 48 volt phantom power for microphones requiring external current (generally condenser microphones). Such microphones are dependent upon a clean, consistent and noise-free power supply for optimal operation and audio quality. The Preference Mic Pre continuously delivers precisely 48 V and a maximum current of 14 mA, which will power all microphones.

IMPORTANT: All microphones with balanced, ground-free outputs can be used with the phantom power activated. Unbalanced microphones may only be used with phantom power deactivated.

Phantom power should only be activated when using microphones that require it. Please be sure to deactivate phantom power with all other microphones (including tube microphones which are supplied from their own power supply, thus need no extra phantom power supply).

ALWAYS FOLLOW THESE RULES WHEN WORKING WITH PHANTOM POWER – THE INPUT STAGES OF THE PREMIUM MIC PRE CAN BE DAMAGED IF YOU DO NOT ACT ACCORDINGLY:

1. First connect the microphone to the Preference Mic Pre.
2. Then activate phantom power – you can start working now.
3. When you have finished recording, turn the phantom power off first.
4. **DO NOT disconnect the microphone from the Preference Mic Pre UNTIL phantom power has been switched off for a minute and all residual current is discharged.**

PAD

Pad

The Pad function allows you to attenuate the input signal by 20 dB so that you can process even very high levels, i.e. from drums or brass instruments.

If the VU meter shows levels above +3 dB even while Mic Gain is set very low (and the -10 dB button deactivated), the time has come to press the Pad button. It illuminates when it is activated.

ø

Phase Reverse

With the phase reverse button you invert the polarity of the microphone signal. When not pressed (button is not illuminated) the polarity is in phase. After pushing the button (button is illuminated) the polarity is inverted.

The phase reverse feature comes in very handy if you have to switch the polarity of the XLR input according to the polarity of the microphone or the microphone cable. The pin wiring of the XLR sockets are as follows: Pin 1 = ground, Pin 2 = hot (+), Pin 3 = cold (-).

Sometimes it is useful to switch the polarity of a microphone, for example in the case of M/S miking. A second classic application is the miking of a snare drum with two microphones that are placed above and below the snare: Since both drum heads move in the same direction when the drum is played, the microphones are out of phase. Switch the polarity of the bottom mic and you avoid any cancellations when you join both signals in the mix.

High-Pass Filter

The high-pass filter, also known as rumble filter, helps to eliminate any interferences within the lowest frequencies. The first order filter operates smoothly with 6 dB per octave, starting from 75 Hz with -3 dB. This characteristic usually helps in most cases and has the least sonic disadvantages. If you need to filter on a more extreme scale, even second order filters (12 dB/octave) are overstrained frequently and sonic disadvantages become more and more apparent. In those cases a variable filter is the means of choice.

SSM 2019

The Preference Mic Pre is suitable for all common dynamic and condenser microphones. It works along the principles of an instrumentation amplifier which is a technology that is also used in measurement and medical equipment due to its high common mode rejection of stray pick-ups. It is based on the semiconductor SSM 2019 which sounds more balanced than available alternatives. Low noise and distortion values as well as a broad bandwidth and a fast slew rate are its forte.

Servo Drive Design

The main focus during the development of the Preference Mic Pre was its acoustic transparency and its natural representation of the source signal. Reducing DC offsets in the audio signal paths is a decisive part of this design job.

DC offsets are produced in relatively large amounts especially when amplifying a microphone signal because the signal is amplified by extreme factors from the pico volt range to 0dB nominal level. DC offsets impair the signal quality as they induce noise and distortion that lead to a diffuse sound.

The Preference Mic Pre's servo drive design minimizes these problems much more effectively than the common solution based on capacitors by setting the DC offset to almost 0 mV. In addition to that, the active servo drive circuitry tends to produce less distortions than passive capacitors.

The servo drive design incorporates three operational amplifiers. The SSM 2019 is the first op-amp. The second op-amp acts as a sensor to detect voltage differences. These differences are then compensated for in a third op-amp working as summing stage.

Of course an elaborate active servo drive circuitry is a more expensive solution than simply using passive capacitors, but especially in a microphone preamplifier this effort pays off with improved sound quality.

Foil And Styroflex Capacitors

Only the best MKP and styroflex capacitors are used in the various circuits. They sound more open and dynamic in contrast to the conventional ceramic capacitors.

Output

The output stage is electronically balanced and capable of driving very long cables (up to a few hundred yards, depending on the capacity of the cables and the input stages on the other end). The maximum output level is +22 dBu.

Audio

Frequency Range		10Hz to › 200 kHz
CMR		-84 dBu
(@ 1 kHz with -30 dBu input level and 30 dB gain)		
THD&N @ 1 kHz		
Input Level	Gain	THD&N
-30 dBu	30 dB	0,0035%
-60 dBu	60 dB	0,047%
S/N Ratio	Gain	S/N Ratio A-weighted
	72 dB	-57,0 dBu
	60 dB	-69,0 dBu
	30 dB	-91,7 dBu
E.I.N. (Equivalent Input Noise)		-129,0 dBu
Dynamic Range		114,0 dB

Input

XLR socket, elektronically balanced

Impedance unbalanced	ca. 1,6 kOhm
Impedance balanced	ca. 3,2 kOhm
Max. Input Level	+18 dBu, +38dBu with Pad activated

Output

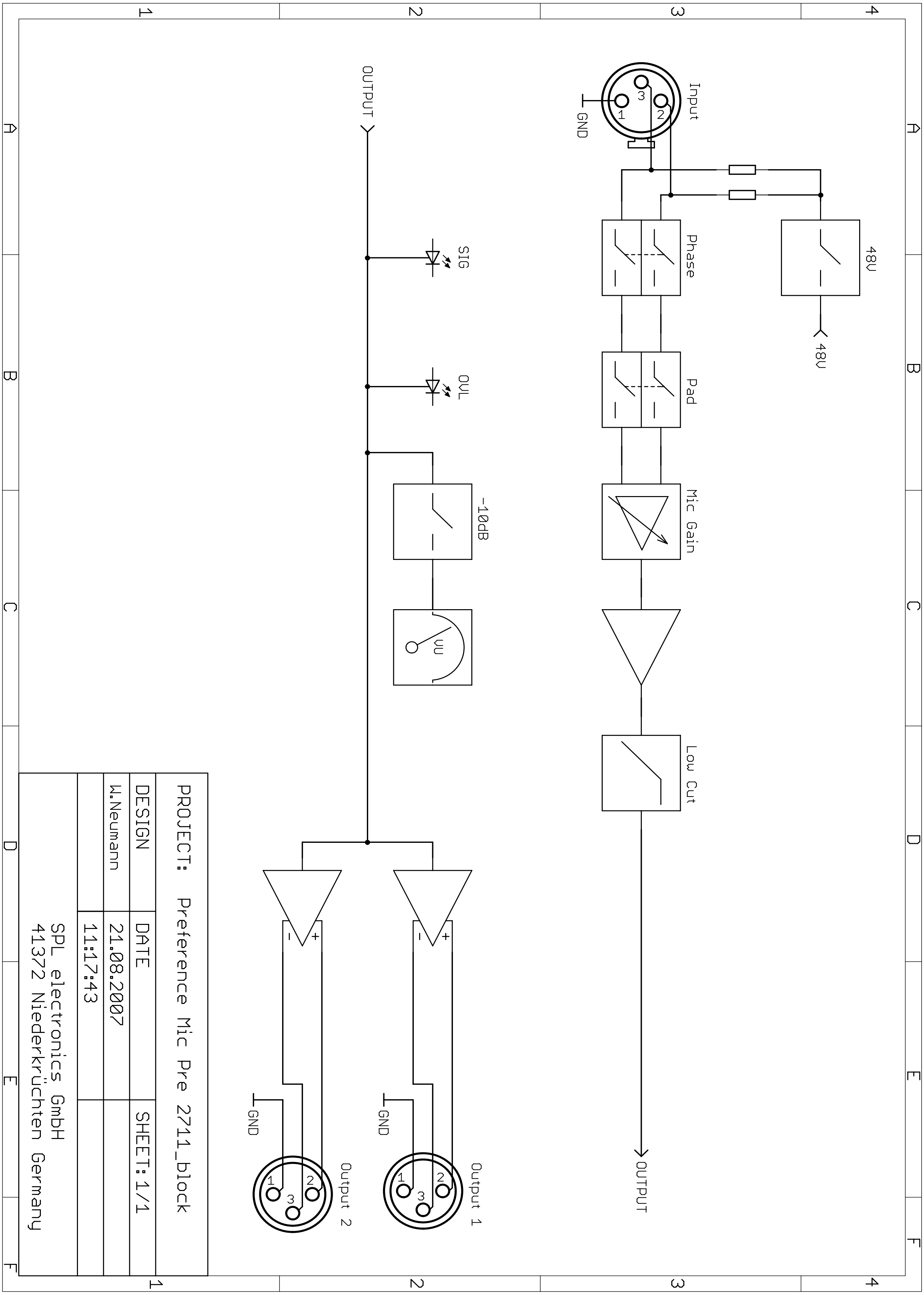
XLR socket, elektronically balanced

Impedance unbalanced	ca. 75 Ohm
Impedance balanced	ca. 150 Ohm
Max. Output Level	+22 dBu

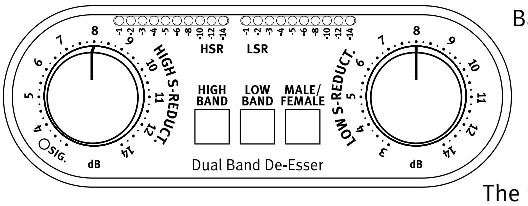
Control Elements

Mic Gain Range	18 dB - 72 dB
Pad Switch	-20 dB
High-Pass Switch	fg = 75 Hz (-3 dB)
Phantom Power Supply	48 V

0 dBu = 0,775 V. Technical changes subject to change without notice.


Input
GND
48V
Phase
Pad
Mic Gain
Low Cut
OUTPUT
SIG
OVL
-10dB
VU
Output 1
Output 2
PROJECT: Preference Mic Pre 2711_block
DESIGN
DATE
SHEET: 1/1
W.Neumann
21.08.2007
11:17:43
SPL electronics GmbH
41372 Niederkrüchten Germany

Dual-Band De-Esser | Introduction



Back in the 1990ies, we developed an alternative way to process signals in order to reduce sibilance based on phase cancellation. Unlike traditional compression methods, this procedure is much more unobtrusive and simplifies control to one single parameter. SPL's De-Esser quickly became a standard reference among recording studios, broadcast stations and live sound engineers.

The most commonly used technology to remove sibilance is based on compression. In addition to determining the threshold, the center frequency for processing must also be set. The processing range can be up to two octaves in order to effectively address all possible problems across the frequency spectrum. This results in one of the most critical disadvantages: the wide range of frequencies being processed leads to undesired effects such as nasalization and lisper.

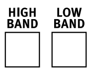
The SPL De-Esser works on the principle of phase cancellation to remove unwanted sounds. And it also adds automatic sibilance detection, which allows processing to be limited only to the range where sibilant sounds are present. The result is a neutral-sounding, unnoticeable but highly effective processing that never requires fine tuning level and frequency settings. This way de-essing has the least possible influence on the voice's timbre, avoiding side effects like nasalization and lisper. Operation is limited to adjusting the processing intensity with one single control. The SPL De-Esser is a safe and precise tool to solve sibilance problems, without having to compromise sound quality nor the hassle of permanently readjusting settings.

The Dual-Band De-Esser

The Dual-Band De-Esser module expands on this concept by making use of two frequency bands that can be used independently or jointly.

- Two de-esser stages increase processing effectiveness without introducing any audible artifacts
- Focused processing with high and low bands
- Input signals are automatically adjusted so that processing is uniform, regardless of the distance between source and microphone
- Male/Female modes that adapt processing in the lower band to male or female voices

Dual-Band De-Esser | Control Elements



HI-Band On, Low-Band On

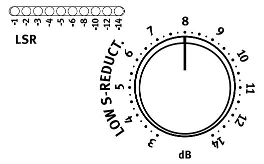
Use the HI-BAND ON button to turn the HIGH-S-REDUCTION on or off. Use the LOW-BAND ON button to turn the LOW-S-REDUCTION on or off. The buttons light up when engaged.

You can use the two processing stages separately or jointly. They are connected in series as independent de-esser modules. The low-band de-esser is set first in the chain.

If both de-essers are engaged, there is interaction between them: a signal already processed with the low-band de-esser is different from the raw material that the high-band de-esser would otherwise process. That is the reason why the readings of the high band's SR LEDs change when the low-band de-esser is engaged while the high-band processor is active.

Hard-Bypass: the Dual-Band De-Esser module features power outage protection based on relay-controlled hard-bypass circuits to always guarantee signal flow from input to output. In order to keep signal flow constant, the bypass is automatically activated whenever a voltage drop or failure is detected.

Low S-Reduction



Use the LOW S-REDUCTION control to adjust the intensity of the sibilance reduction in the lower frequency range. The center frequency for sibilance recognition is set at 7.6 kHz in FEMALE mode and 6.4 kHz in MALE mode. For more information, please refer to section “MALE/FEMALE“ below. The bandwidth of the low-band de-esser is 1.44 kHz.

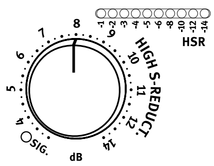
Scale values for the filter are displayed in dB. The actual reduction values, i.e. after phase cancellation, are displayed in the lower SR LEDs. Thus, when the control is set to 3 dB, actual reduction might only be of around 1 dB.

The SR LEDs display sibilance reduction between -1 dB and -14 dB, first in 1 dB increments and from 6 dB on in 2 dB increments.

In practice, for most applications, the best results are usually achieved when LOW S-REDUCTION is set between 3 and 7.

High S-Reduction

Use the HIGH S-REDUCTION control to adjust the intensity of the sibilance reduction in the upper frequency range.



The center frequency for sibilance recognition is set at 11.2 kHz with a 3 kHz bandwidth.

Scale values for the filter are displayed in dB. The actual reduction values, i.e. after phase cancellation, are displayed in the upper SR LEDs. Thus, when the control is set to 3 dB, actual reduction might only be of around 1 dB.

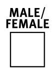
The SR LEDs display sibilance reduction between -1 dB and -14 dB, first in 1 dB increments and from 6 dB on in 2 dB increments.

Please note that the MALE/FEMALE button has no effect on the high-band de-esser.

In practice, for most applications, the best results are usually achieved when HIGH S-REDUCTION is set between 3 and 7.

Male/Female

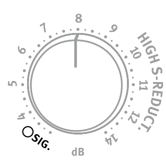
The MALE/FEMALE button allows you to adjust the low-band de-esser to the type of voice being processed. When engaged, the mode selected for the low-band de-esser is FEMALE, otherwise the de-esser works in MALE mode. The mode selected affects the center frequency for sibilance recognition: in FEMALE mode it is set at 7.6 kHz, while in MALE mode it is set at 6.4 kHz.



These values have been determined by practical experience, so that the processor adapts better to gender. Nevertheless, you cannot take for granted that these settings will suit every single male and female voice. Consider the MALE/FEMALE function as an additional tool to help you set the low-band de-esser more precisely according to your needs. Always trust your ears to find the best settings.

Signal-LED (SIG.)

The SIG. LED indicates that an audio signal reaches the input with a level above -20 dB. This LED helps the operator especially in complex setups to determine immediately whether the Dual-Band De-Esser actually receives any signal.



Audio

Frequency Range:	10 Hz-80 kHz
CMR:	›60 dBu
@ 1 kHz, 0 dBu input level and unity gain	
S/N Ratio:	-106 dBu
A-weightedt	
Dynamic Range:	128 dB
THD&N:	0,01%
@ 1 kHz, 0 dBu input level and unity gain	

Input

XLR socket, electronically balanced, optionally transformer balanced

Impedance	ca. 20 kOhm
Max. Input Level	+22 dBu
Nominal Input Level	+4 dBu

Output

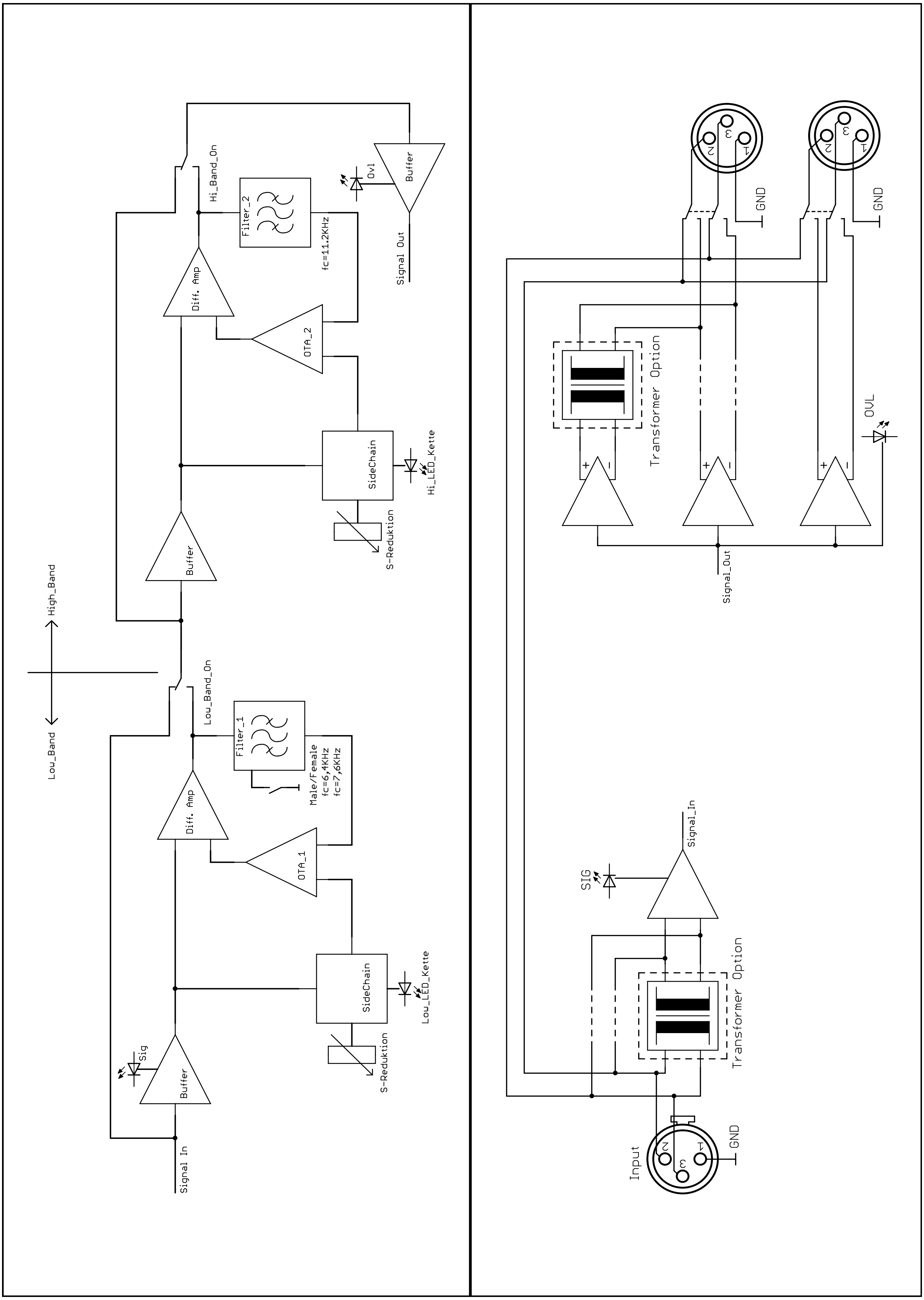
XLR socket, electronically balanced, optionally transformer balanced

Output Impedance	75 Ohm/›600 Ohm with transformer
Max. Output Level	+22 dBu

Control Elements

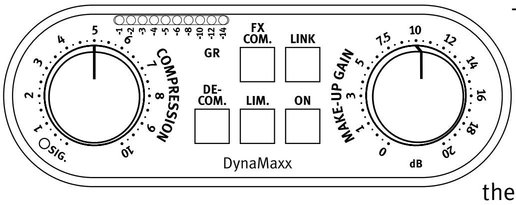
Signal-LED	-20 dBu
Overload-LED	+19 dBu (peak hold 1,5 seconds)

0 dBu = 0,775 V. Technical changes subject to change without notice.



Signal In
Buffer
Sig
Diff. Amp
OTA_1
SideChain
Low_LED_Kette
S-Reduktion
Filter_1
Male/Female
fc=6,4KHz
fc=7,6KHz
Low_Band_On
Low_Band
High_Band
Buffer
Diff. Amp
OTA_2
SideChain
Hi_LED_Kette
S-Reduktion
Filter_2
fc=11,2KHz
Hi_Band_On
Ovl
Buffer
Signal Out
Input
GND
Transformer Option
SIG
Signal_In
Transformer Option
Signal_Out
OVL
GND
GND

DynaMaxx | Introduction



The DynaMaxx probably is the most consequent interpretation of a “set & forget“ compressor concept. In keeping with the SPL design philosophy, it incorporates many features not found on standard compressors, and a unique control system makes is surprisingly easy to use.

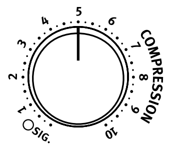
The DynaMaxx also offers a de-compression mode which may be used to counter the effects of overcompression in previously processed source material.

Why is DynaMaxx so different? Though auto attack and release functions are nothing new, in the DynaMaxx design, the time constants are automated in a very musical way. DynaMaxx optimizes all time constants in real time during processing so that the compression characteristics are continually matched to the source material.

The DynaMaxx circuitry actually uses two of the excellent THAT 2181 VCAs in SPL’s Double VCA-Drive mode configuration, which doubles the operating range while increasing transparency and reducing distortion. Another benefit of the DynaMaxx circuitry is that high compression ratios do not affect high frequency detail – high amplitude, low-end bass can be controlled without introducing pumping or other negative side-effects. Similarly, complex stereo sources can be processed easily and very musically.

The DynaMaxx has numerous applications in recording and mixing, as well as for live applications, and because of the level of intelligent processing within the unit, there are only two controls to adjust per channel, making operation very intuitive.

DynaMaxx | Control Elements



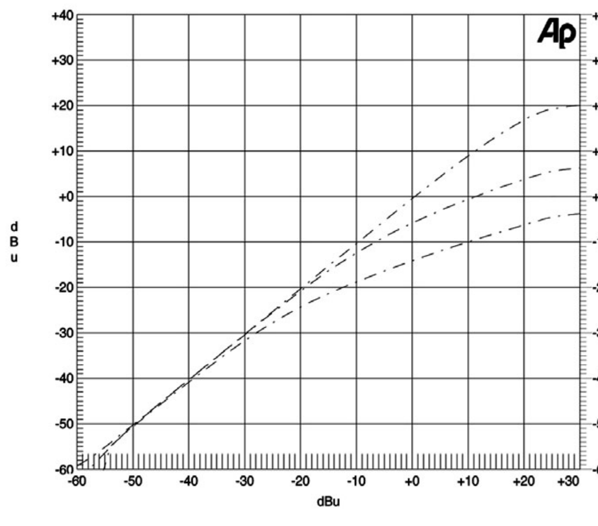
Compression

The COMPRESSION control sets the compression intensity by varying both Threshold and Ratio simultaneously. The further the COMPRESSION control is turned clockwise (which lowers the Threshold and increases the ratio), the more processing is extended to lower level signals. The fully clockwise position is equivalent to a Ratio of 3:1.

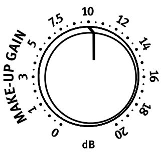
The exact processing values are always depending on the source material. Loud signal parts will always be processed with higher compression rates than quieter signal parts. The DynaMaxx employs soft-knee compression characteristics, following the input signal‘s structures dynamically.

For gentle compression results, turn the COMPRESSION control clockwise until the Gain Reduction LED ladder shows a peak level reduction of between 2 and 3 dB. Turning the Compress control further clockwise will lower the Threshold to include more low-level signals into the compression process. At the same time, compression ratio is increased up to a maximum of 3:1.

The following diagram shows the gain control curve in normal compression mode with three different COMPRESSION settings. For better readability of the graph, the gain has been kept at odB, so no compensation for the gain reduction is applied.



Make-Up Gain

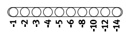


The MAKE-UP GAIN control compensates for any level decrease when applying COMPRESSION. The higher the processing intensity the lower the overall output level, the more Gain will be required to restore the same peak level.

Turn the MAKE-UP GAIN control clockwise until the peak level of the input is the same as the peak level of the output. Use the GAIN REDUCTION LEDs for a precise adjustment. The control has a range of 20dB.

If the DE-COMPRESSION mode is activated, the MAKE-UP GAIN control compensates for the increase in peak level. Note that in this mode, turning the Gain control clockwise decreases the output level.

Gain Reduction LEDs



The 10-digit LED ladder meter is capable of displaying gain changes to a resolution of 1dB over the range -1 dB to -14 dB.

When applying COMPRESSION to the audio signal, the LED meter displays the amount of gain reduction taking place. From here you can see how much gain to compensate for with the MAKE-UP GAIN control.

When the DE-COMPRESSION mode is activated, the LED display shows the gain increase imparted to high-level signals.

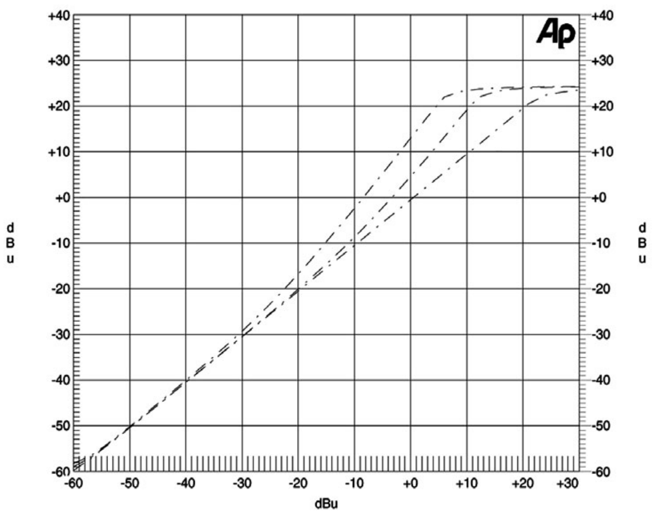
DE-COM. (De-Compression)

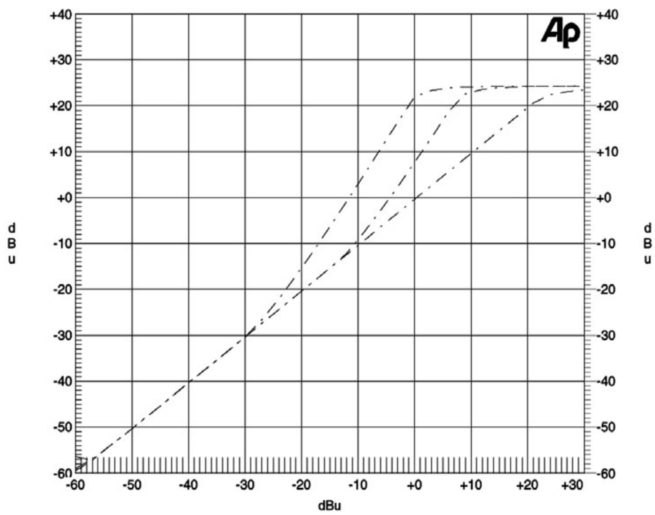
The De-Compression function inverts the operation of the compressor to produce new dynamic headroom, which may be used to increase the dynamic range of a (previously over-compressed) signal.

The process may also be used to expand the dynamic range of other sources, such as drum or synthesizer samples. Especially when creating loops, overcompressed samples can result in lifeless sounds. With the DynaMaxx you can simply de-compress the signals again. Interesting effects can also be achieved by applying different settings to the channels of a stereo source.

IMPORTANT: While MAKE-UP GAIN increases the output level in standard mode, this function is also inverted in DE-COMPRESSION mode – turning the control clockwise reduces the output level.

The following diagrams show different characteristic curves in DE-COMPRESSION mode. If the Limiter is activated in DE-COMPRESSION mode, the intensity of processing is increased remarkably. Especially peaks are expanded more intensively.





FX Com. (Effect Compression)

The FX Com. mode allows to switch off the DynaMaxx's perfectly automated dynamics processing. In this mode a fixed release time of 60ms is applied, but the Attack time is still automated. This mode increases the perceived loudness in comparison with the normal Compression mode, and audible compression artifacts can be generated for creative purposes.

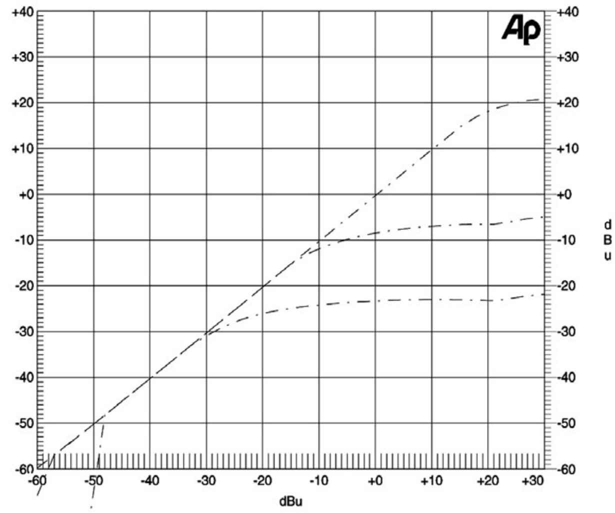
The gain of the audio signal is restored back to normal shortly after the 60ms Release time has passed, creating deliberate breathing and pumping effects. The LED display provides a visual impression of the increased processing speed. The higher the CompressION control setting, the more intense those effects will become.

Lim. (Limiter)

In limiter mode the DynaMaxx only processes the peak levels but leaves, in contrast to the compressor mode, the gain structure of low-level signals unchanged. Limiter mode is useful when working with 'clipping-sensitive' media or units and it improves the utilisation of digital headroom as well as the bit resolution.

The COMPRESSION control sets the threshold in LIM. mode. The further the control is moved clockwise, the lower the threshold, and all signals above the threshold will be submitted to the limiting process as a fixed ratio of ∞:1.

The limiter is based upon a soft-knee limiting characteristic. In comparison with hard limiting characteristics, soft limiting is more unobtrusive and sonically more natural. When a peak level exceeds the threshold, the level isn't suddenly reduced, as would be the case with a hard limiting characteristic, but rather the soft limiting starts its processing earlier. This way, peak levels are limited far more smoothly once the threshold is reached. The following measurements illustrate various soft-knee limiting curves:



On

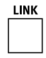
With the ON button you can turn the device on or off. The ON button is illuminated when the device is active.

Relay hard bypass circuits ensure signals to be directly switched from input to output in the case power failures – this "Power Fail Safety" feature guarantees signal flow in any situation.

If you operate a DynaMaxx module in LINK-Mode, the ON button of the master module controls both modules. The LINK and ON buttons of the DynaMaxx module that is currently in LINK mode will thus be illuminated as well—although they have not been activated.

DynaMaxx | Control Elements

Link



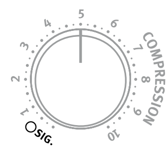
You can operate two DynaMaxx modules at a time by activating the LINK mode. You can only run two modules in the LINK mode at a time. Main applications may be in stereo processing.

When operated in LINK mode, both modules are operated by the master module's control voltage. This ensures a coherent stereo operation. In LINK mode, all controls of the master module (left module) — including the ON and LINK switch — control the slave module. The master also controls all switch illuminations on the slave. All controls of the slave module are inactive in LINK mode.

NOTE: the GAIN REDUCTION LED Display of the slave module is switched off in Link mode.

If you press the LINK button of the slave module without establishing the LINK mode through the master module, you will find that the LINK LED of the master module will not illuminate. This tells you that you have pressed the wrong LINK button since the controls of the master module control both devices in LINK mode.

Signal-LED (SIG.)



The SIG. LED indicates that an audio signal reaches the input with a level above -20 dB. This LED helps the operator especially in complex setups to determine immediately whether the DynaMaxx actually receives any signal. .

DynaMaxx | Technology

This section deals with the technical background of the DynaMaxx and explains why we felt it necessary to do some things rather differently to the way other manufacturers do them. We will also explain the benefits of the DynaMaxx design.

Generally, the function of an audio compressor is to compress the dynamic range of the source signal by altering the gain of its signal path in response to the relative level of the signal as compared to an arbitrary threshold level. In effect, this process can be thought of as providing additional gain to low-level signals or reducing gain in the presence of high-level signals.

Why are conventional compressors unsatisfactory?

One of the main difficulties in setting up a compressor is choosing the time constants Attack, Decay, Sustain, and Release (in short: ADSR). The most important time constants are Attack and Release – in the context of gain control, Decay and Sustain are mostly fixed and they tend to play a minor role. The inappropriate choice of time constants is mainly responsible for the familiar pumping or breathing effects. Threshold and Ratio are easier to set up, but when high compression ratios are used, high frequencies tend to become over-compressed, which makes the material sound dull or lacking in presence. It is normal to set up compressors based on the loudest peak in the program material, in which case the parameters are optimized for this moment in time only. The rest of the time, the time constants are less than optimal. Taking into consideration that each musical instrument has varying attack and release times, depending not only on the character of the instrument, but also on the way it is played, you can see that it is almost impossible to choose a set of fixed parameters that will be correct for an entire piece of music. Setting up a conventional compressor perfectly for recording a vocal part is nearly impossible. However, perfect gain control is extremely important with digital systems to make the best possible use of the available digital headroom while preventing clipping. Vocals can have an extremely wide dynamic range, so for the best results, it may become necessary to re-adjust the compressor settings while recording.

The DynaMaxx solves these problems by the use of intelligent automation. It optimizes Attack and Release times on the fly, which is why the control system is so simple. Both Threshold and Ratio are combined within the Compress-control, and the advanced Double VCA technology maintains signal clarity, even at high levels of compression.

Full-Band versus Multi-Band

Some compressors try to solve the problem of high frequencies being modulated by low frequency compression by moving to a split band system, so why don't we do that?

Multi-band compression seems like a good idea to overcome the pumping effects caused by heavy bass compression also causing high frequency sounds to be pulled down in level. For example, with a regular compressor, you may be compressing a bass-drum but the release time is set a little too long with the result that the following hi-hat gets ducked in level. Multi-band technology splits the original signal into two or more bands to be processed individually, and in this way, heavy gain reduction at the bass end doesn't affect the level of the high frequencies.

Sounds like a good idea, but the elaborate level of automation and the double VCA circuitry allow to dispense with multi-band techniques – and that provides two main advantages:

1. Neutral Sound
Multi-band technology has a significant sonic short-coming: due to different levels of processing within the various bands, each band's output may be changed in phase response, so that when the bands are recombined, the signal tends to have reduced dimensions and sounds incoherent and colored.

2. Time Saving
DynaMaxx offers simplicity of control. With Multi-band systems you have to set all the time constants plus Gain, Threshold, and Ratio for each band. With a fully manual Four-Band-compressor, this would mean 20 parameters to set up per channel. DynaMaxx needs 2 – and provides better sound results.

Attack Time Automation

First it's necessary to see what happens when a compressor is used with a fixed attack time setting. For example, the sound of an e-bass can either come in smoothly (especially with fretless basses), or with a very fast transient attack when slapping. If the attack time is set to minimum (very short), the compressor is able to catch the peak of the transient attack but any following notes will suffer increased transient distortion because the control voltage within the compressor rises further as successive notes are processed. This behavior is sometimes described as 'surfing', and can be overcome by setting a lightly longer attack time, but now some peaks get through because they are faster than the compressor's attack time. Including a separate Peak-Limiter to catch those fast transients has disadvantages: if this is done using two VCA stages, the signal undergoes more quality degradation than is desirable, but even with designs that use the same VCA for both compression and limiting, you still end up with more controls than necessary.

The DynaMaxx doesn't need a separate peak limiter, because it detects and controls very fast transients automatically. Processing is so fast that all signals stay within the soft-knee curve for a more natural sound.

The attack time can be reduced in the instant of a percussive hit or bass guitar slap to a minimum of 50 microseconds. As soon as the peak is passed, the attack returns to a longer time constant (up to 10ms of first Attack time circuitry), and in this way, both pumping and distortion are avoided.

The ability of DynaMaxx to respond so quickly is clearly valuable when complex stereo mixes are being treated. If, for example, a snare drum peak occurs, DynaMaxx rapidly changes to a very short attack time so that the snare hit keeps its original transient characteristics rather than sounding 'softened'. Following signals are compressed using longer attack times immediately.

The DynaMaxx uses 12dB/octave filtering instead of the 6dB/octave filtering used in standard compressor side-chains so as to increase the precision and speed of transient detection and processing. The signal-dependent adjustment of the attack time parameter is too fast to be audible. If the DynaMaxx is operated in its normal compression mode, breathing and pumping effects are avoided.

Release Time Automation

Again, it is beneficial to look at what happens when the Release time is too short: In this case the compressor will restore normal gain conditions as soon as the peak has passed, and this rapid increase in gain is what we call breathing. Breathing is especially annoying during a part with soft strings or low level layered sounds – whenever a peak comes along, the strings duck down with it, only to pump up again when the peak is over. Sonically the sound image swims and pumps while the subjectively perceived loudness remains at a low level.

The DynaMaxx uses a special technique to overcome these problems. First it monitors the average music level. If a loud transient sound occurs (bass drum, snare and so on) that also has a big gain step, a very short Release time is set and the signal level is reduced to the calculated average music level, not to the threshold! If the compressor reduces the level to the Threshold (which is what standard compressors do), you would again hear pumping effects, because of the way low-level signals are lifted in gain.

The Release time is controlled depending on the difference between the actual peak and calculated average signal levels. If a large difference is detected, DynaMaxx will set a faster Release time, whereas if only a small level difference is detected, DynaMaxx will release more slowly.

Threshold And Ratio

Threshold and Ratio are both set by the COMPRESSION control; using a low setting for the COMPRESSION control causes only the peak levels to be compressed because the threshold is relatively high. For more compression, turning the control clockwise has the effect of lowering the threshold to include more low-level signals in the processing.

As the threshold is lowered, the ratio is simultaneously increased, and the fully clockwise position is equivalent to a ratio of around 3:1. However, the threshold and ratio values are not static, but rather vary depending on the attack and level characteristics of the source signal. Peak levels are automatically compressed with a higher ratio to maintain control over maximum signal levels, but still using an unobtrusive soft-knee compression characteristic.

Audio

Frequency Response	10 Hz - 200 kHz
CMR *(@ 1 kHz, 0 dBu input level, unity gain)*	›70 dBu
THD *(0 dBu input level, unity gain)*	0,06%
Noise *(A-weighted)* *(Make Up Gain 0 dBu)*	-93 dBu
Dynamic Range	115 dB

Input

XLR connection, electronically balanced, optionally transformer balanced

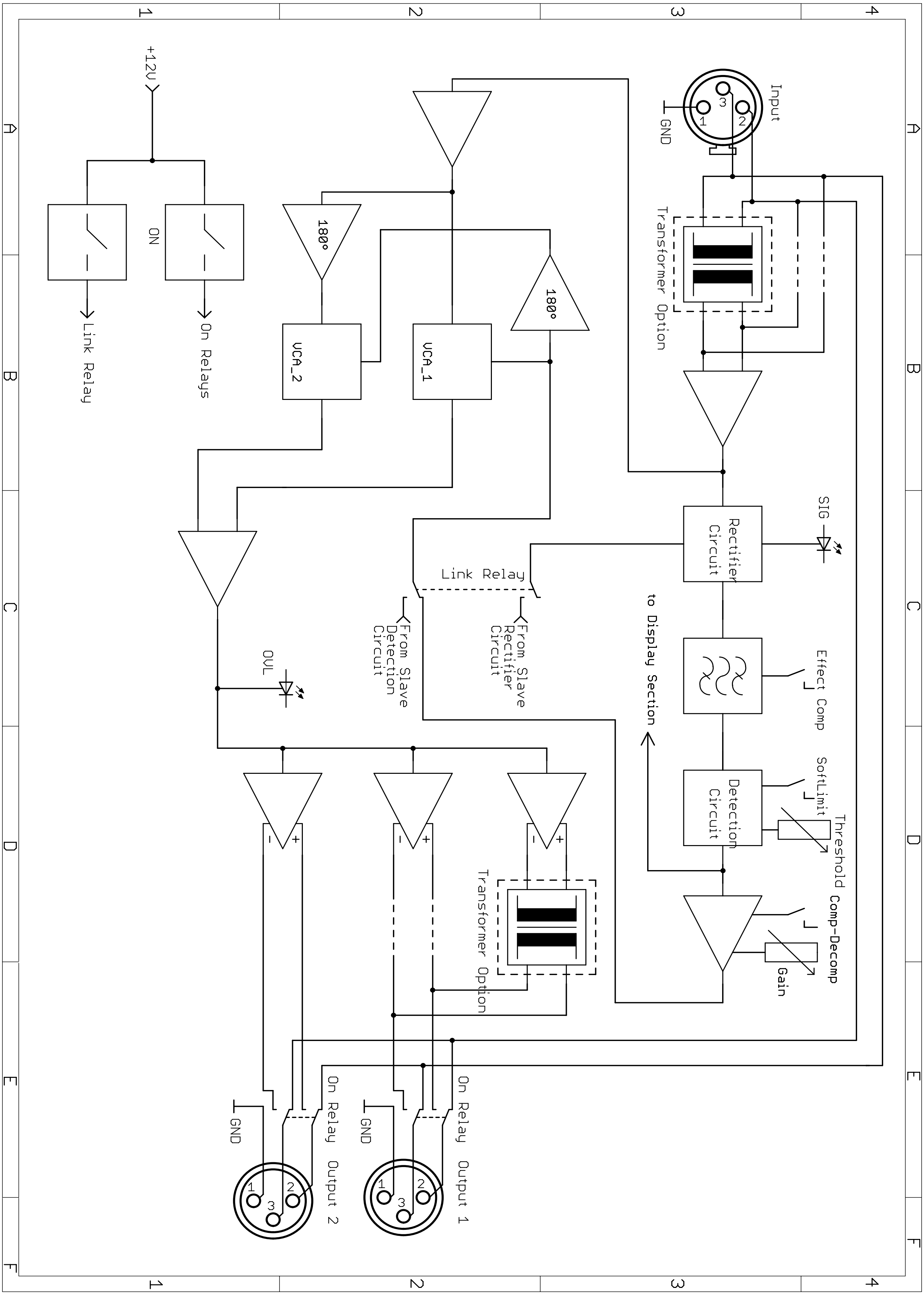
Impedance	ca. 20 kOhm
Max. Input Level	+21 dBu

Output

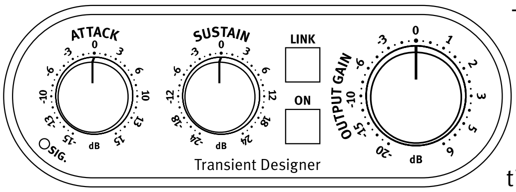
XLR connections, electronically balanced, optionally transformer balanced

Impedance	ca. 150 Ohm
Max. Output Level	+22 dBu

0 dBu = 0,775 V. Specifications subject to change without notice.


+12V
ON
Link Relay
On Relays
VCA_2
VCA_1
180°
180°
Input
GND
Transformer Option
SIG
Rectifier Circuit
Link Relay
From Slave Detection Circuit
From Slave Rectifier Circuit
to Display Section
OVL
Effect Comp
SoftLimit
Detection Circuit
Threshold
Comp-Decomp
Gain
Transformer Option
On Relay
GND
Output 2
On Relay
GND
Output 1

Transient Designer | Introduction



The Transient Designer provides a revolutionary concept for level-independent dynamic processing. In this it differs in principle from common compressors that are based upon processing signals of a specific level. Working with the Transient Designer is very simple: Attacks can be amplified or attenuated and sustain may be prolonged or shortened. However, the possibilities for studio and live application are seemingly endless.

Technical foundation is SPL‘s Differential Envelope Technology (DET) which allows level-independent dynamic processing by calculating differences in generated envelopes. These envelopes are always tracking the curve of the original signal to provide optimal results in every moment of the music.

Thanks to the level-independent processing DET the setting of a threshold is needless. Other common controls of dynamic processing, such as ratio or parameters for time-constants are automated and optimized adaptively in a musical manner according to the characteristics of the input signal.

So only two controls per channel are required to allow the user to completely reshape the attack and sustain characteristics of a sound: attack can be amplified or attenuated by up to 15dB while sustain can be amplified or attenuated by up to 24 dB.

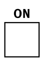
Thus, in a very obvious and simple way, the Transient Designer opens a whole new dimension in dynamic processing with entirely new, stunning and vast possibilities for dynamic manipulation and processing that cannot even be duplicated with several daisy-chained, conventional compressors or other dynamics devices.

A new feature of the Transient Designer module is the output gain control that enables you to compensate changes in the level of the processed signal. This ensures a simple and safe adjustment of levels for any following device – especially A/D converters.

The Transient Designer uses the excellently specified THAT 2181-VCAs, which are particularly natural and transparent sounding and renowned for minimal distortion values. The 2181 processes highest amplitudes without damping of high frequencies or reducing bass.

The module is equipped with a relay hard bypass to ensure the switching audio signals directly from input to outputs in the case of a power failure. Signal flow is always maintained.

Transient Designer | Control Elements

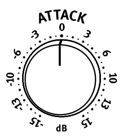


On

With the ON button you can turn the device on or off. The ON button is illuminated when the device is active.

Relay hard bypass circuits ensure signals to be directly switched from input to output in the case power failures – this “Power Fail Safety“ feature guarantees signal flow in any situation.

If you operate a Transient Designer module in LINK-Mode, the ON button of the master module (left) controls both modules. The LINK and ON buttons of the Transient Designer module that is currently in LINK mode will thus be illuminated as well—although they have not been activated.



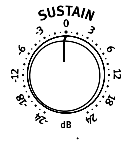
Attack

With the ATTACK control you can amplify or attenuate the attack of a signal by up to 15 dB.

The ATTACK control circuitry uses two envelope generators. One follows the shape of the original curve and adapts perfectly to the dynamic gradient. The second envelope generator produces an envelope with a slower attack. From the difference of both envelopes the VCA control voltage is derived. Positive ATTACK values emphasize attack events, negative ATTACK values smooth out the attack envelopes of sound events.

For an extensive description and explanation of the possible applications of the ATTACK control please refer to “Applications” on pp. 42-43.

Sustain



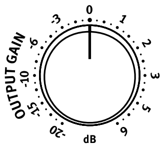
With the SUSTAIN control you can amplify or attenuate the sustain of a signal by up to 24 dB.

The SUSTAIN control circuitry also uses two envelope generators. One follows the shape of the original curve and adapts perfectly to the dynamic gradient. The second envelope generator produces an envelope with a longer sustain. From the difference of both envelopes the VCA control voltage is derived. The gradient of the control voltage matches the time flow of the original signal.

Positive SUSTAIN values lengthen the sustain, negative SUSTAIN values shorten the sustain.

For an extensive description and explanation of the possible applications of the SUSTAIN control please refer to “Applications” on pp. 42-43.

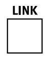
Output Gain



The OUTPUT GAIN control allows you to reduce the output signal by up to -22dB or boost it by up to +6dB. This ensures that following devices receive an optimized level. The centered 12-o‘clock position of the control equals odB output gain.

IMPORTANT: In LINK mode, the master module does NOT transfer OUTPUT GAIN settings to the slave-module. You ALWAYS have to set the output values of both modules manually.

Link Mode

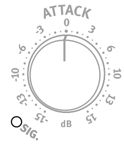


You can operate two Transient Designer modules at a time by activating the LINK mode. You can only run two modules in the LINK mode at a time. Main applications may be stereo processings.

When operated in LINK mode, both modules are operated by the master module’s control voltage. This ensures a coherent stereo operation. In LINK mode, all controls of the (left) master module—including the ON and LINK switch—control the slave module. The master also controls all switch illuminations on the slave. All controls of the slave module are inactive in LINK mode. **IMPORTANT:** The only exception is the Output Gain control which always has to be set manually at each module.

If you press the LINK button of the slave module without establishing the LINK mode through the master module, you will find that the LINK LED of the master module will not illuminate. This tells you that you have pressed the wrong LINK button since the controls of the master module control both devices in LINK mode.

Signal LED



The SIG. LED indicates that an audio signal reaches the input with a level above -20 dB. This LED helps the operator especially in complex setups to determine immediately whether the Transient Designer actually receives any signal.

The Transient Designer is ideally suited for use in professional recording, in project or home studios and sound reinforcement applications.

For the first time you can manipulate and control the attack and sustain characteristics of a signal regardless of level in the most intuitive and simple way. Usually equalizers are used to separate instruments in a mix – the tonal aspect of the signal is considered, but not the temporal aspect. The Transient Designer opens this further dimensions in signal processing. By manipulating the attack and sustain curves of a sound event, the mix can be made to sound more transparent. Instruments can be mixed at lower levels while still maintaining their positions in the mix—but occupying less space.

During a remix or in general after miking you can arrange new positions of instruments. Reduce ATTACK and increase SUSTAIN to move signals back into the mix that are too present. Additionally, the FX parts of too dry signals are strengthened.

Applied to single instruments or loops the Transient Designer allows you to create entirely new sounds and effects.

The following examples are given as suggestions and examples. The described procedures with specific instruments can of course be transferred to others which are not mentioned here.

Drums & Percussions

Processing drum and percussion sounds is probably the Transient Designer's most typical range of application, both from samples to live drum sets:

- Emphasize the attack of a kick drum or a loop to increase the power and presence in the mix.
- Shorten the sustain period of a snare or a reverb-flag in a very musical way to obtain more transparency in the mix.
- When recording a live drum set, shorten the toms or overheads without physically damping them. Usual efforts to damp and mike are reduced remarkably. Since muffling of any drum also changes the dynamic response, the Transient Designer opens up a whole new soundscape.
- Miking live drums is considerably faster and easier because you can correct the apparent 'distance' of the microphone by simply varying the ATTACK and SUSTAIN values.
- The Transient Designer is a perfect alternative to noise gates in live drum miking. Adaptively reacting to the duration of the original signal, the sustain is shortened more musically than with fixed release times and a drumset is freed from any crosstalk quickly and effectively.
- Create unusual dynamic effects including new and interesting pan effects. For example, patch a mono loop through two channels of the Transient Designer and pan fully left and right in the mix. Process the left channel with increased ATTACK and reduced SUSTAIN while you adjust the right channel the opposite way and you get very special stereo loop sounds. You have to try this to appreciate what it sounds like, but expect to hear a lot of unusual stereo movement.
- Enjoy an amazingly simple integration of drum sounds into a mix. If the acoustic level of a snare is expanded to approximately +4 dB by increasing the attack value, the effective increase of peak levels in the overall mix is merely about 0.5 dB to 1 dB.

Guitars

Use the Transient Designer on guitars to soften the sound by lowering the ATTACK. Increase ATTACK for in-the-face sounds, which is very useful and works particularly well for picking guitars. Or blow life and juice into quietly played guitar parts.

Distorted guitars usually are very compressed, thus not very dynamic. Simply increase the ATTACK to get a clearer sound with more precision and better intonation despite any distortion.

Heavy distortion also leads to very long sustain. The sound tends to become mushy; simply reduce SUSTAIN to change that. If you, however, want to create soaring guitar solos that would make even David Gilmour blush, just crank up the SUSTAIN control to the max and there you go.

With miked acoustic guitars you can emphasize the room sound by turning up SUSTAIN. If you want the guitars to sound more intimate and with less ambience, simply reduce SUSTAIN.

Bass: Staccato vs. Legato

Speaking of bass: Imagine a too sluggishly played bass track ... you may not have to re-record it: Reduce the SUSTAIN until you can hear clear gaps between the downbeats—the legato will turn into a nice staccato, driving the rhythm-section forward.

Ambience

A common problem especially with tracks that are recorded and mixed in different studios: Backings lack of ambience, and finding a reverb that “matches” takes time ... so simply emphasize the original ambience by turning up the Transient Designer’s SUSTAIN control.

And the opposite problem, too much ambience, is similarly simply solved with the opposite processing — just reduce SUSTAIN.

Keyboards & Sampler

Sounds in keyboards and samples are usually highly compressed and maintain only little of natural dynamics. Increase the ATTACK values to re-gain a more natural response characteristic. The sounds occupy less space in the mix and appear more identifiable even at lower volumes.

Post Production

When dealing with overdubs in movies you can easily add more punch and definition to effect sounds from any sample library.

The same applies to outdoor recordings that suffer from poor microphone positioning—simply optimize them afterwards.

Mastering

Like with any good thing, you also have to know where not to use it. For example, using a Transient Designer in mastering is not recommended, as it is rarely a good idea to treat a whole mix at once. Instead, treat individual elements within the mix.

Audio

Frequency Response	10 Hz - 70 kHz
CMR *(at 1 kHz with 0 dBu input level/unity gain)*	-80 dBu
THD&N *(0 dBu input level/unity gain)*	0,019%
S/N Ratio *(A-weighted)*	-89 dBu
Dynamic Range	111,0 dB

Input

XLR connector, electronically balanced, optionally transformer balanced

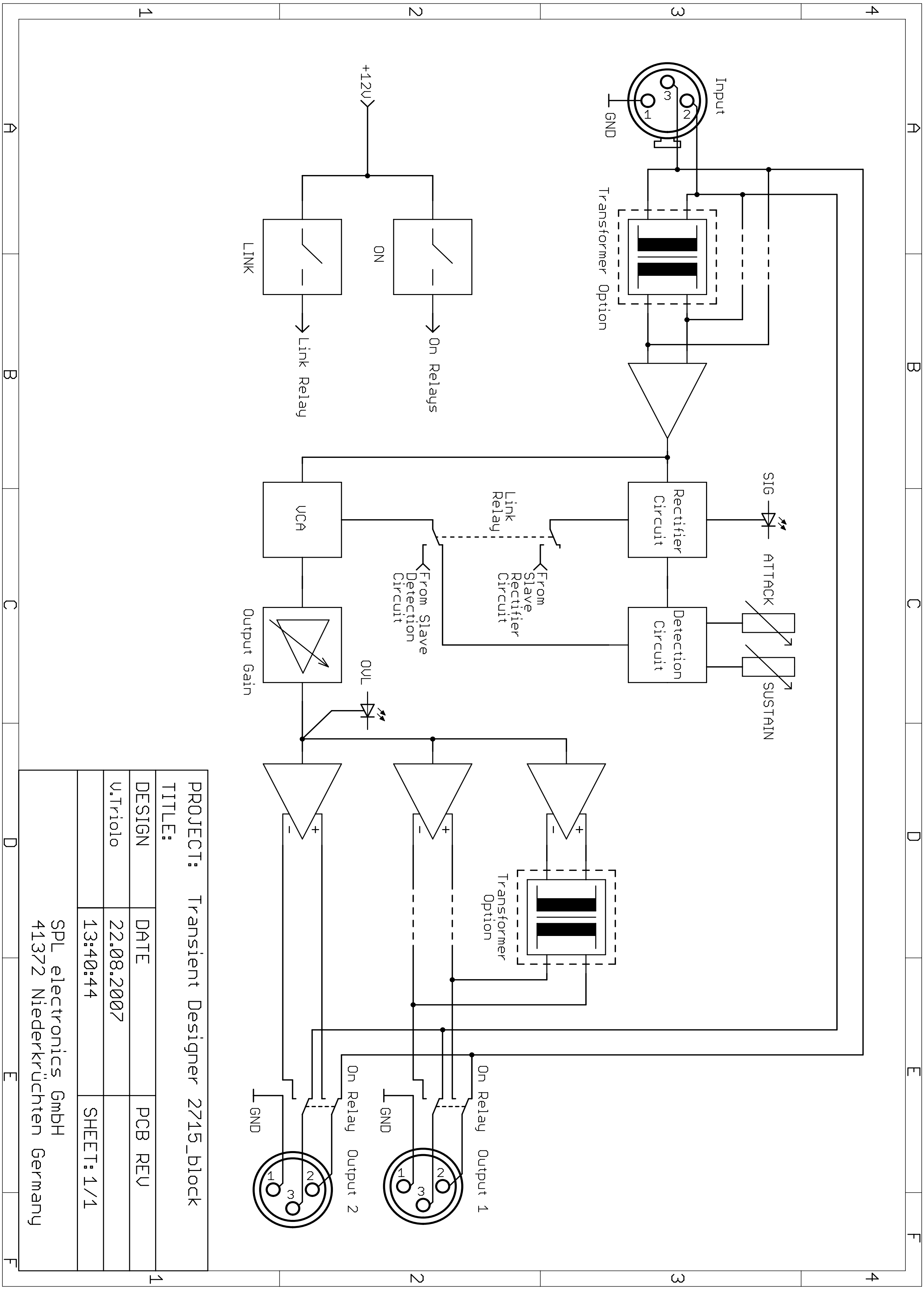
Impedance	ca. 20 kOhm
Max. Input Level	+21 dBu

Outputs

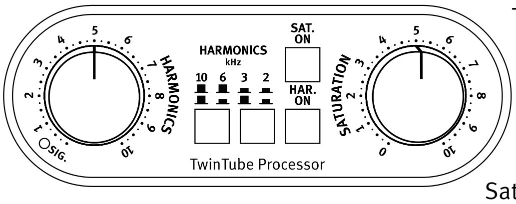
XLR connectors, electronically balanced, Output 1 optionally transformer balanced

Impedance	ca. 150 Ohm
Max. Output Level	+22 dBu

0 dBu = 0,775 V. Specifications subject to change without notice.


Input
GND
Transformer Option
+12V
ON
LINK
On Relays
Link Relay
SIG
Rectifier Circuit
Link Relay
VCA
From Slave Rectifier Circuit
From Slave Detection Circuit
ATTACK
Detection Circuit
SUSTAIN
Output Gain
OVL
Transformer Option
On Relay
Output 1
On Relay
Output 2
GND
GND
PROJECT: Transient Designer 2715_block
TITLE:
DESIGN
DATE
PCB REV
V.Triolo
22.08.2007
13:40:44
SHEET:1/1
SPL electronics GmbH
41372 Niederkrüchten Germany

TwinTube | Introduction



The TwinTube module is the first ever combination of two essential tube effects into a single processor, that is, saturation effects along with harmonics processing. Both stages work separately from each other and are based on an individual tube circuit, therefore each module employs two tubes. The effects can therefore not only be applied both individually or separately, but also in common.

Saturation effects are generated through the tube being pushed to and beyond its normal operating limits. In contrast to semiconductors, a tube thus pushed to such levels does not clip from a certain level, approaching more gradually its level limits and thereby producing its typical tonal result, which in audio signal processing can have such often profitable aural effects—on one hand (and depending on the amount applied), from subtle to extensive harmonic distortion and on the other hand, a compaction of the sonic event, that is, a limiting effect that exhibits a pleasant, rounded or soft sound.

Acoustically and also in its range of applications this can be compared very well with tape saturation effects. Harmonic distortion and limiting are the generally known, “classic” tube effects, which are today cornerstones of sound processing.

But other less known and potentially important effects are a tube’s ability for improving presence and spacial qualities through its processing of specific regions of the overtone series.

A special circuit comes into play for overtone/harmonic processing that involves a combined coil/condenser system working in conjunction with the tube. The control reacts dynamically to the audio signal and thereby processes both overtones as well as a signal’s phase structure.

The processing of the phase structure influences the moments of acoustic perception and occurs in microsecond time divisions – it has to do nothing with the cancellations one associates with 180-degree signal shifts. A decisive factor in resultant tonal quality is the alignment of level relationships the overtone spectrum. Such overtone “enrichment” does not operate on the generator principle of exciters (wherein distortion is added to the original signal). In this case the TwinTube harmonics control effects a rather a more equalized overtone structure resulting in a sound which in effect appears much more in the foreground, but without doing so through extreme level changes. Thus, for example, a voice appears immediate apart from the overall mixture, “sitting” clearly outlined in the mix’s foreground.

TwinTube | Control Elements

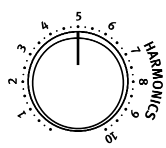
HAR. ON/SAT. ON

Activating HAR ON engages the HARMONICS control. Likewise you engage SATURATION with the switch SAT. ON. When turned on, these switches illuminate.

One of the both processing stages can be applied therefore respectively separately. Similarly you can process the signal with both stages at the same time.

Warm-up phase: A warm-up phase for both tube stages begins after powering on the unit, independent of the HAR. ON or SAT. ON switches. During the warm-up phase HAR ON and SAT. ON flash alternately. The warm-up phase can last between 20 seconds and a minute. During the warm-up phase the entire module remains in a bypass mode.

Hard-Bypass: For power fail safety, the TwinTube module employs a relay hard bypass design. It guarantees a direct switching from the inputs to the outputs in case of a power failure to maintan the signal flow.



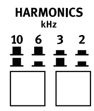
HARMONICS Control

With the HARMONICS Control you can adjust the overtone processing intensity. The available level range of a chosen harmonics frequency (see “HARMONICS Switch” on the next page) lies between 0 and 15 dB.

Overtones are marked also as “Harmonics”. The HARMONICS control results in an enrichment of the overtone range for a chosen fundamental tone (see “Introduction” above). The tonal result is an intensification of the presence that produces a fresher, silky and more brilliant aural image. Similarly, the signal’s spacial qualities gain in intensity.

TwinTube | Control Elements

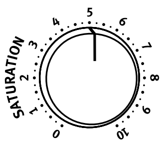
HARMONICS Switch



With the HARMONICS switch you choose the frequency range of the fundamental tone area that should be processed with the HARMONICS control. There are four available frequency ranges with respective center frequencies of 9.8 kHz, 6.6 kHz, 2.8 kHz and 1.9 kHz. For reasons of space, we have rounded frequency values for the values on the front panel. The bandwidths of the filters are listed under “Specifications, Harmonics Filter” on page 50.

The frequency ranges are based upon our experiences and test series to offer ideal processing ranges for wide variety of instruments (and vocals of course). You may find further information under „Operation/Application Examples” below.

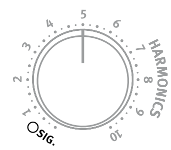
SATURATION Control



As the name implies, the SATURATION control determines the degree of the tube saturation. At the most extreme level this SATURATION control output level may increase the overall output level by approximately 6 dB.

The SATURATION control offers a wide range of effects intensity from most subtle to brutal harmonic distortion. Aside from harmonic tube distortion, the accompanying tube limiting effect should also be considered. For further details on the saturation effect refer to the “Introduction” on the previous page.

Signal LED



The SIG. LED indicates that an audio signal reaches the input with a level above -20 dB. This LED helps the operator especially in complex setups to determine immediately whether the TwinTube actually receives any signal.

TwinTube | Applications

Here we refer to only two significant examples, of course without suggesting completeness. We would like to simplify starting with the TwinTube by sharing some thoughts and experiences in these applications areas. The effects and results described here can be applied to many other instruments—nothing should keep you away from using the TwinTube without restriction.

Vocals

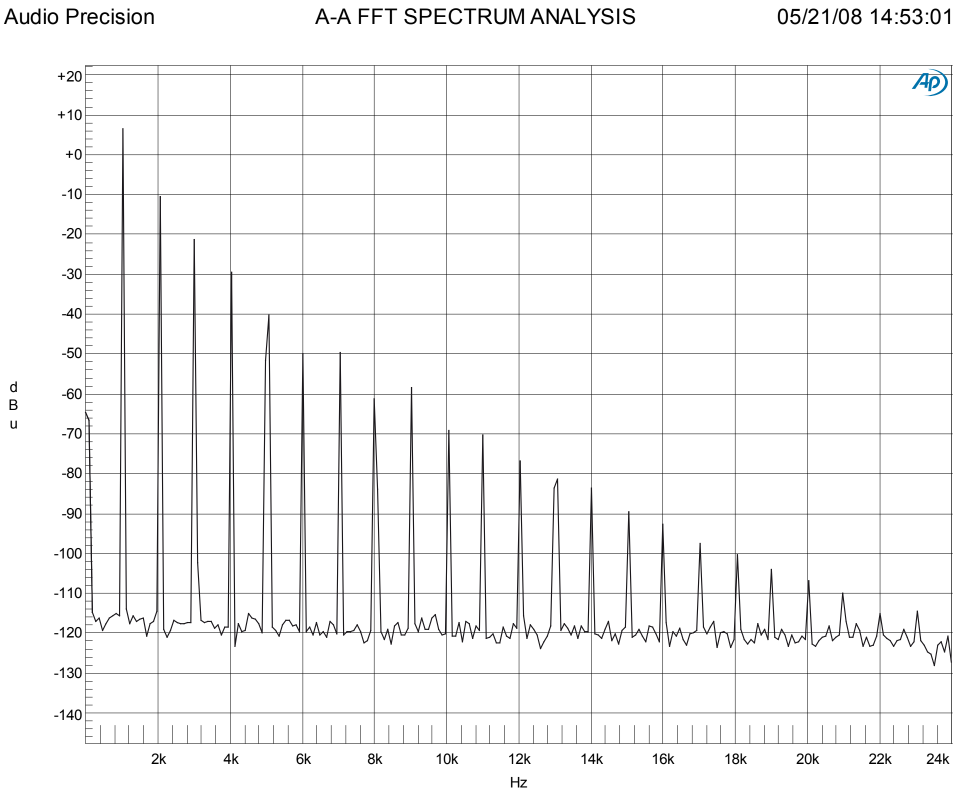
Optimizing vocal tracks is a highlight among the processing applications of the TwinTube. Often further EQing is not necessary anymore in order to lift a voice from a mix and get it up front. A recommendation for female voices: HARMONICS switch to 6, HARMONICS control to about 2 o’clock, SATURATION to about 12 o’clock.

With these settings the described effect should be clearly audible and from here individual tracks can be optimized. With female voices we suggest trying HARMONICS switch settings 6 and 10 while switching between 2, 3, and 6 with male voices.

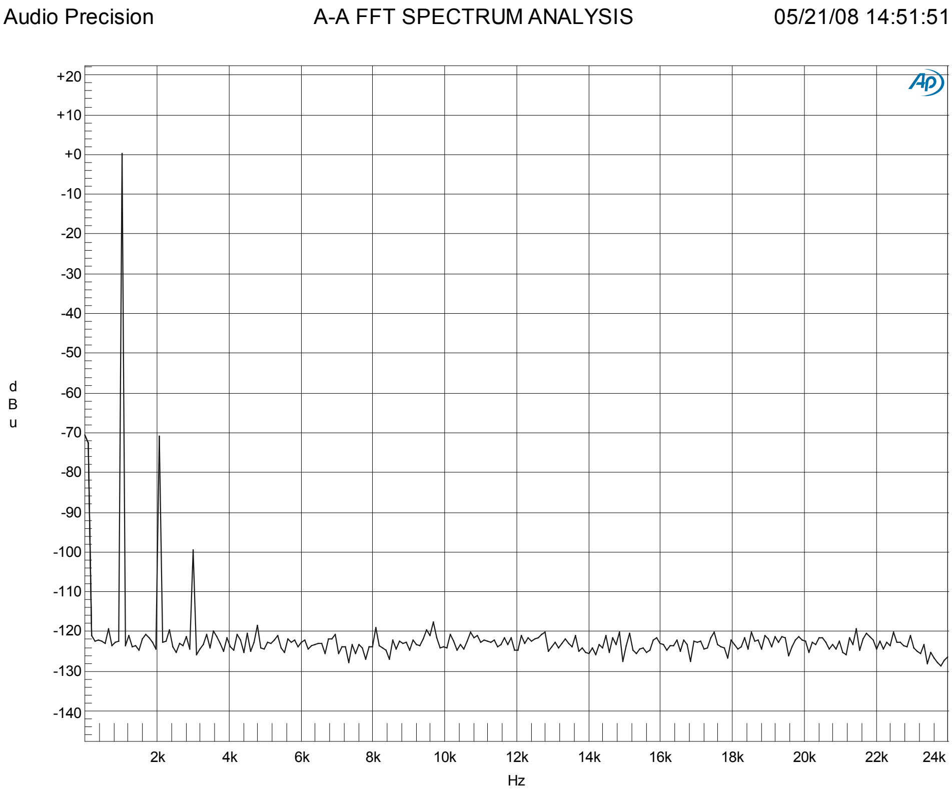
Acoustic Guitars

The success in treating e-guitars depends on previous recording and processing gear and techniques—if tube amps and further effects were already applied, it is hard to foresee how much the TwinTube can contribute when optimizing or designing a sound.

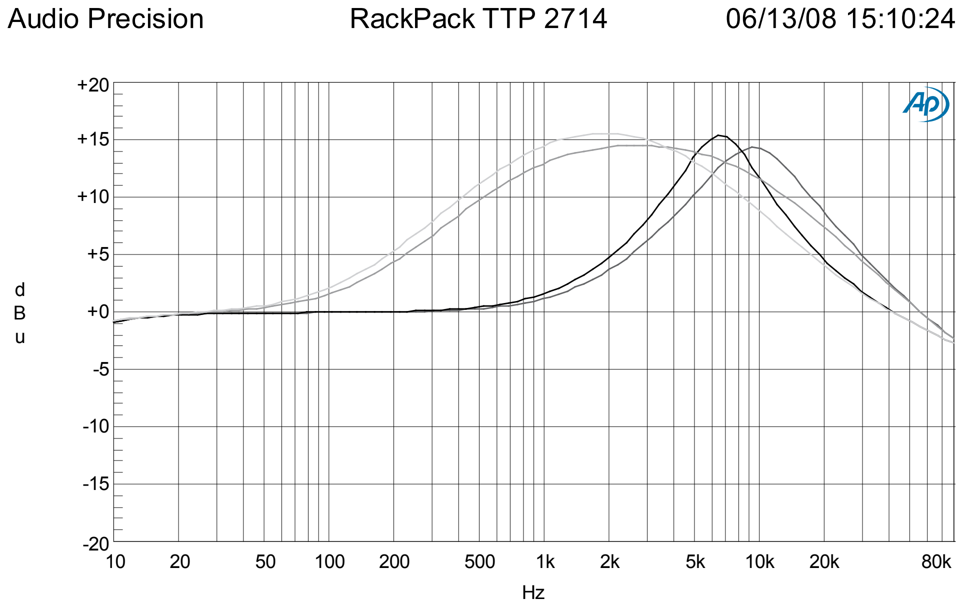
In contrast to e-guitars, there is a huge potential in processing acoustical guitar tracks. Picking sounds can be intensified, in general tube saturation and limiting can improve loudness and condenses the sound. Presence is emphasized and the instrument cuts through a mix much better without raising levels too much. A well chosen amount of harmonic distortion always adds some roughness which may often be a nice touch in several playing styles.



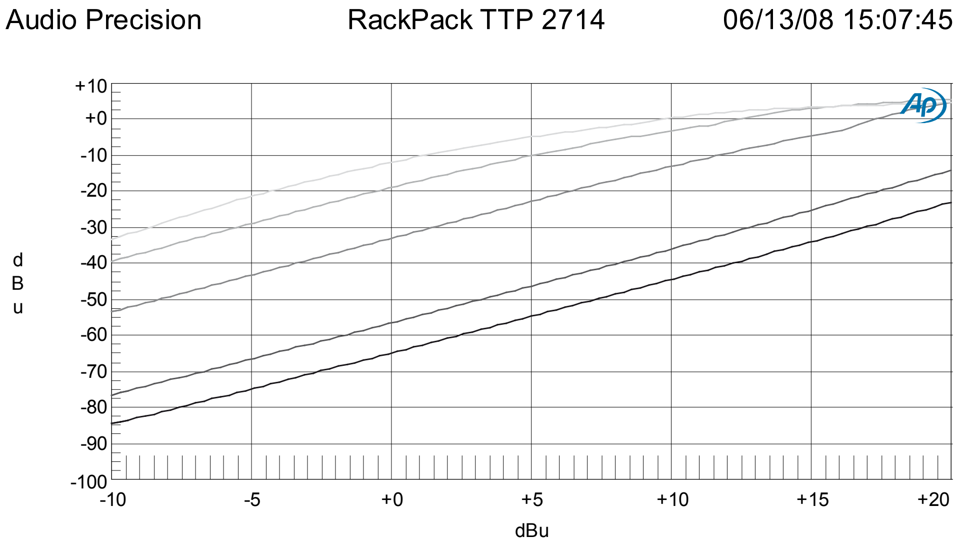
Fast Fourier Transformation (FFT): 1 kHz/0 dBu input signal, SATURATION to maximum.



FFT: 1 kHz/0 dBu input signal , SATURATION to minimum.



HARMONICS control at maximum.



THD vs. input level of the saturation stage at different control intensities (lower curve: SATURATION control set to 0, upper curve: SATURATION set to 20).

AUDIO

Harmonics Stage

Frequency Range	10 Hz-80 kHz
THD	0.1 %
@ 1 kHz, 0dBu input level, unity gain	
Noise, A-weighted	-87 dBu
Dynamic Range	103 dB
Maximum Input Level	+18 dBu

Harmonics Filter

Filter 1	Center Frequency: 9,8 kHz Bandwidth: 9.6 kHz Maximum Gain: +15 dB
Filter 2	Center Frequency: 6.6 kHz Bandwidth: 5 kHz Maximum Gain: +15 dB
Filter 3	Center Frequency: 2.8 kHz Bandwidth: 9 kHz Maximum Gain: +15 dB
Filter 4	Center Frequency: 1.9 kHz Bandwidth: 4.7 kHz Maximum Gain: +15 dB

Saturation Stage

Frequency Range	10 Hz-77 kHz
THD	0.02 % @ *1 kHz, 0dBu input level, unity gain* ca. 15 % @ *0dBu input level and maximum saturation*
Noise, A-weighted	-96 dBu
Dynamic Range	114 dB
Maximum Input Level	+21 dBu

AUDIO – CUMULATIVE (Harmonics and Saturation Stage)

Frequency Range	10 Hz-77 kHz
CMR	-74 dBu
@ 1 kHz, 0dBu input level, unity gain	
Noise, A-weighted	-87 dBu
Dynamic Range	105 dB

Input

XLR connector, electronically balanced, opt. transformer balanced

Max. Input Level	+20 dBu
Impedance	ca. 20 kOhm

Output

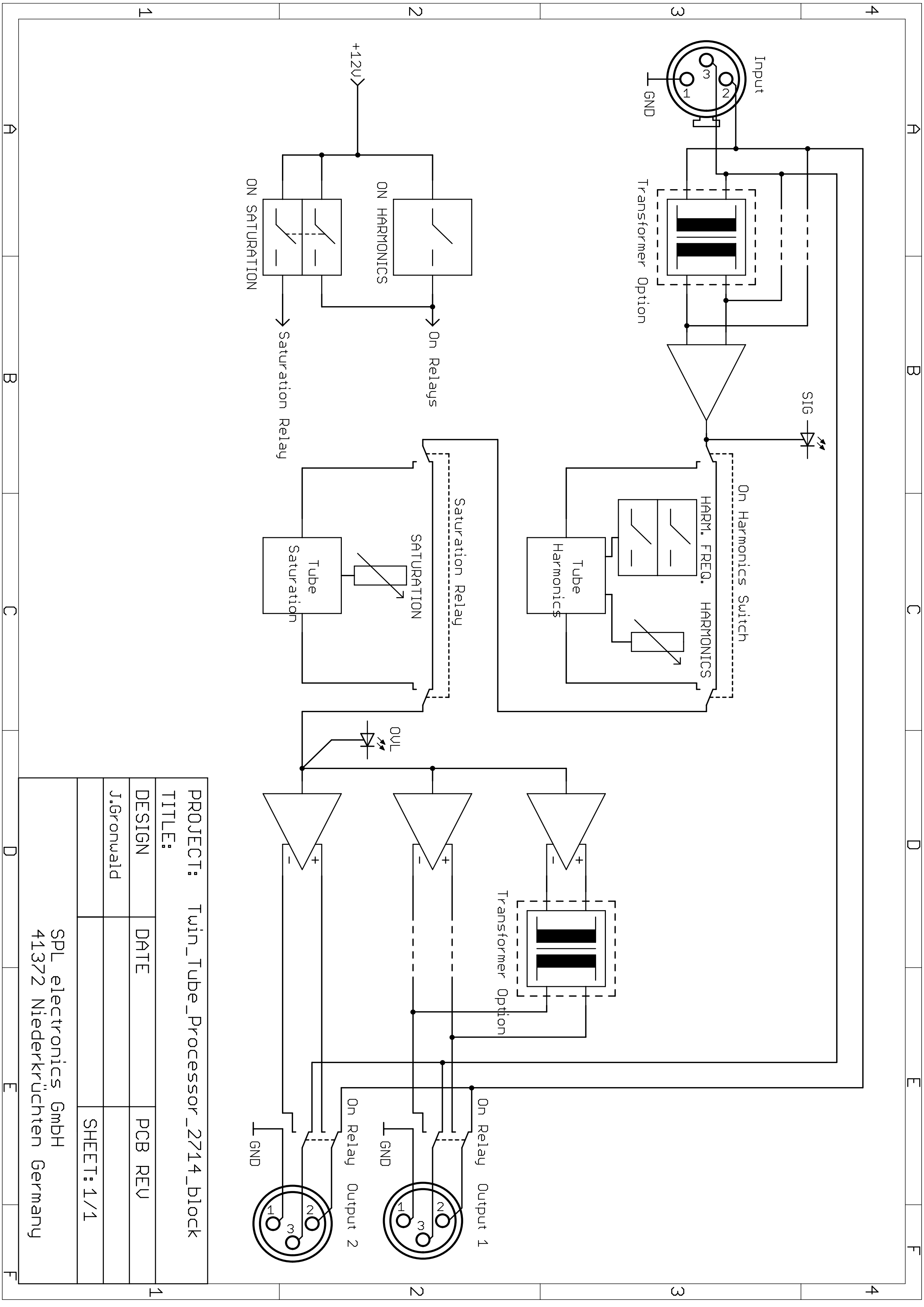
XLR connector, electronically balanced, optionally transformer balanced

Impedance	ca. 150 Ohm
Maximum Output Level	+21 dBu

Control Elements

Signal-LED	-20 dBu
Overload-LED	+18 dBu (peak hold 1,5 seconds)

0 dBu = 0,775 V. Specifications subject to change without notice.



Input
GND
Transformer Option
SIG
On Harmonics Switch
HARM. FREQ.
HARMONICS
Tube Harmonics
Saturation Relay
SATURATION
Tube Saturation
+12V
ON HARMONICS
ON SATURATION
On Relays
Saturation Relay
OVL
Transformer Option
On Relay
Output 1
GND
On Relay
Output 2
GND
PROJECT: Twin_Tube_Processor_2714_block
TITLE:
DESIGN
DATE
PCB REV
J.Gronwald
SHEET:1/1
SPL electronics GmbH
41372 Niederkrüchten Germany