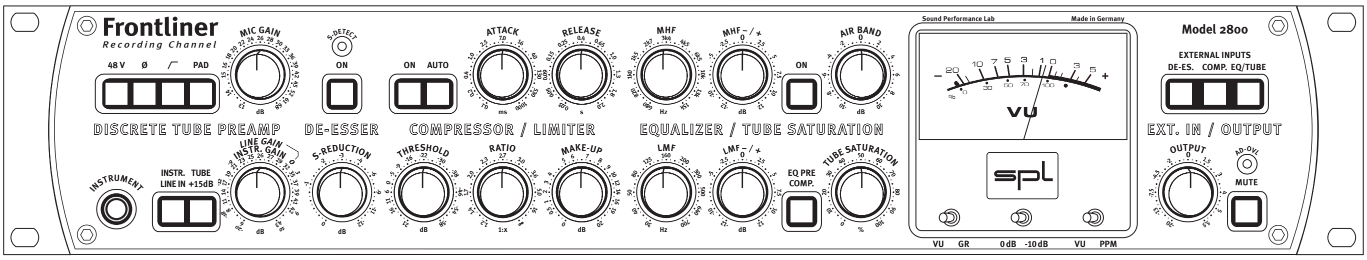
Manual



Frontliner

Model 2800

Recording channel with separate module inputs and outputs

Version 1.0– 6/2009

Designer: 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.

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The construction of the Frontliner, Model 2800, 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

Content

Symbole 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.

Scope of Delivery & Packaging

The scope of delivery comprises the Cabulator, the external power supply, 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.

Important Security Information

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!

Important Security Information

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.

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 pages 4 and 5 and read further information on connections on pages 8, 9 an 10.

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

Our preceding recording channels Track One and Channel One have already been successful units by combining fast and easy operation with high processing quality. These channel strips particularly benefit from previous developments for single processors: the DynaMaxx compressor/limiter delivered the essentials for the respective channel strip stage, the unique SPL De-Esser is also available as a single, dual-channel unit – and the preamplifier modules were also derived from existing products.

The development of new units in the last years inevitably evoked the wish to conceive a new channel strip which would benefit from our latest achievements.

Now the Frontliner comprises the same discrete, hybrid semiconductor/tube preamplifier design like the GoldMike MK2. The compressor module is based upon the Kultube providing a full set of classic controls. We also integrated the Kultube's highlight, a signal-dependent automation of attack and release parameters which can still be influenced manually – with this option, manual and automatic control can be merged to always find the best settings.

The EQ section provides two semi-parametric filters for low and mid band as well as a bell filter delivering a silky and smooth top end. This filter set offers a perfect frequency first aid kit and is rounded off with a Tube Saturation control.

In contrast to semiconductors, a tube does not clip from a certain level, but approaches its limits by increasingly producing harmonic distortions. The sound effects from saturating a tube can often be used to improve the audio signal. The sound results resemble tape saturation effects and increase the density and perceived energy in a very pleasant way.

The Frontliner concept

The Frontliner comprises a hybrid semiconductor/tube preamplifier with microphone, line and instrument input, a de-esser, a compressor/limiter and an EQ section with tube saturation stage. In addition to its primary function as a recording channel with all tools onboard to process a signal prior to storing, its modules can also serve for high-end analog processing during mixing.

The enumeration of the Frontliner's modules also makes clear that the task of a channel strip is relatively complex. There is not one specialized device for one job, but a unit that combines a lot of technology for the preamplification and processing of audio signals. That is why especially the Frontliner benefits very much from SPL's innovations for targeted, efficient work. The SPL De-Esser, for example, performs a highly complex processing, but can simply be adjusted with one single knob.

The straight and clear front panel design also supports fast and safe operation.

Analog plug-ins

In the last years we have been busy with modular concepts. One result is the RackPack series: a RackPack frame hosts up to eight 3U high modules for free placement and (inter-) connection.

But we have also transferred the modular idea to the Frontliner. Each Frontliner module disposes of its own inputs and outputs. Sophisticated switching options allow to integrate Frontliner's modules into a studio environment as if they were stand-alone devices. You can also group modules in any combination and you can determine inserts at each module's input.

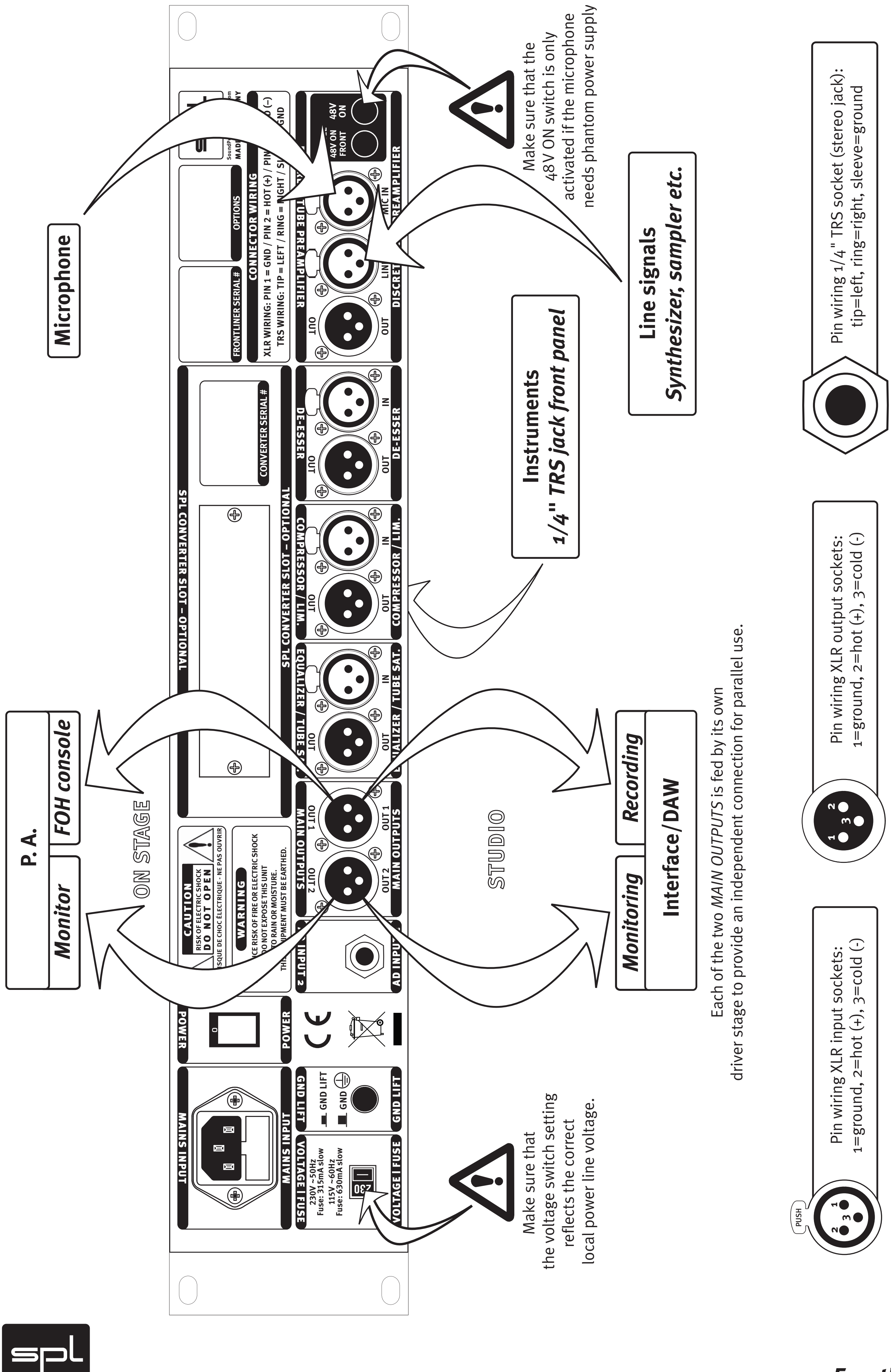
If you imagine the Frontliner modules as analog plug-ins, the additional value suddenly becomes apparent: this channel strip also is a versatile analog processing station. With one single unit, especially DAW based studios benefit from a first-class recording front-end and have access to high end analog processors for the main signal processing tasks.

Special technical features

- Three separate, individually optimized inputs for microphones, line signals and instruments.
- Discrete semiconductor/tube hybrid preamplifier to combine the advantages – high dynamic range and low noise operation of semiconductors with the musical qualities of tube sounds (appealing top end, beautiful spacial impression). Discrete construction is based upon single transistors instead of integrated circuits (ICs) from industrial production and ensures a higher level of optimization.
- 250 volts tube circuits in the preamplifier and for tube saturation stages.
- High-grade operational amplifiers from Burr-Brown at critical positions.
- The SPL De-Esser applies the method of phase cancellation for reducing sibilants. This innovative approach works much more neutral than traditional compression methods – a unique method for unobtrusive results and extremely fast operation.
- The compressor is operated by SPL's double VCA drive circuitry. A differential stage eliminates side effects and the half load per VCA dramatically reduces THD.
- The operational amplifiers for the compressor circuitry meet military specifications. Therefore the regulating voltages are extremely precise – a precondition for high signal processing quality.
- Selected condensers for the EQ filters with pleasant sound characteristics. Especially with vocals, amplified frequencies are not sounding too bright, and on the other hand reductions can tame harsh sounds without sounding dull.
- Central star ground wiring scheme minimizes influences that could affect the ground paths. The audio ground is separated from the ground of the remaining equipment. This leads, in the truest sense of the word "clean", to considerably improved tonal quality.
- The power supply is built around two toroidal transformers, one for the audio voltages and one for the other supply voltages (tube heating, LEDs, micro controller etc.) Mutual interferences are excluded and lavishly over-dimensioned capacities ensure absolutely stable supplies for all audio circuitries.

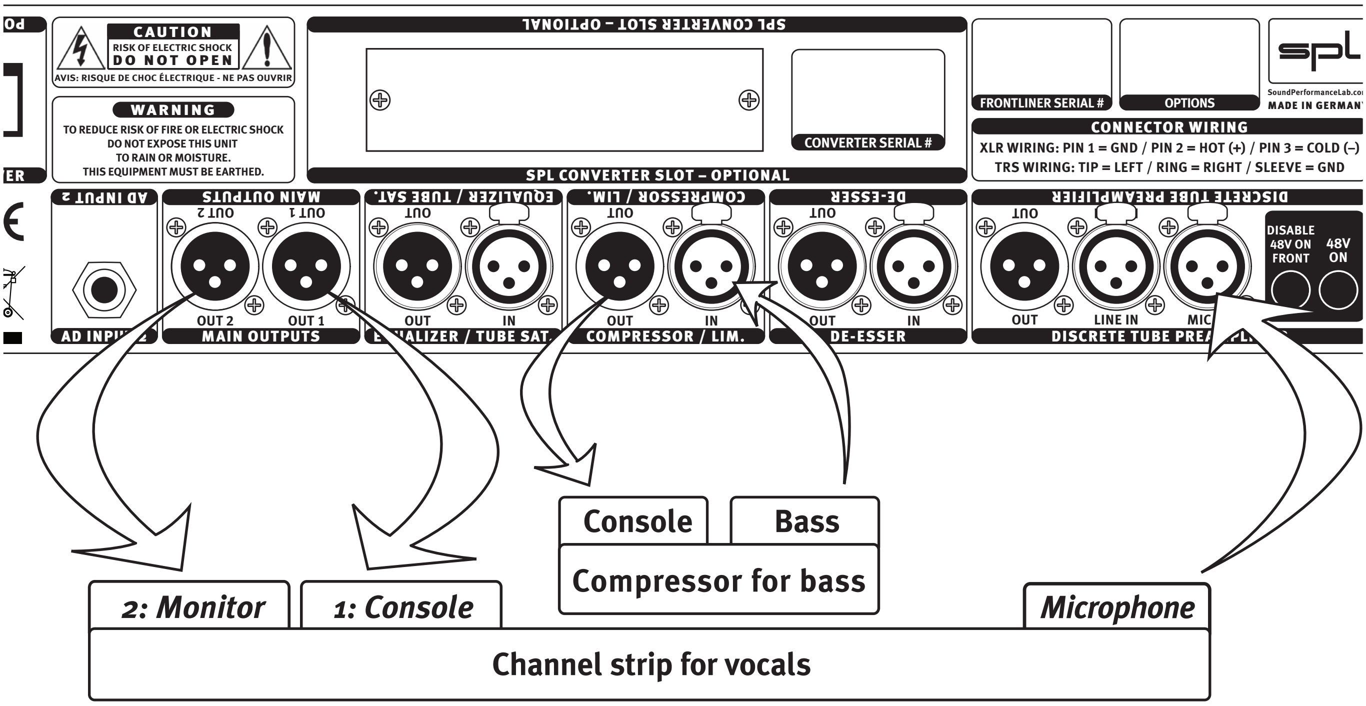
Options

- Lundahl input transformer for the microphone input. The input transformers add ca. 14 dB gain (depending on the microphone). This must be added to the scaled values. The additional passive gain relieves the preamplifier electronics at any gain level. Microphone input transformers are highly recommended for ribbon microphones, as they are demanding a lot of amplification.
- OUT 1 from the MAIN OUTPUTS can optionally be equipped with an output transformer from Lundahl.
- The Frontliner can optionally be equipped with a digital output 24bit/96 kHz AD converter card, model 2376). The Frontliner is a mono channel strip. An additional socket at the rear panel (AD Input 2) allows to feed the dual-channel converter with feed a second signal source.
- Refer to page 27 for further information on the optional equipment.



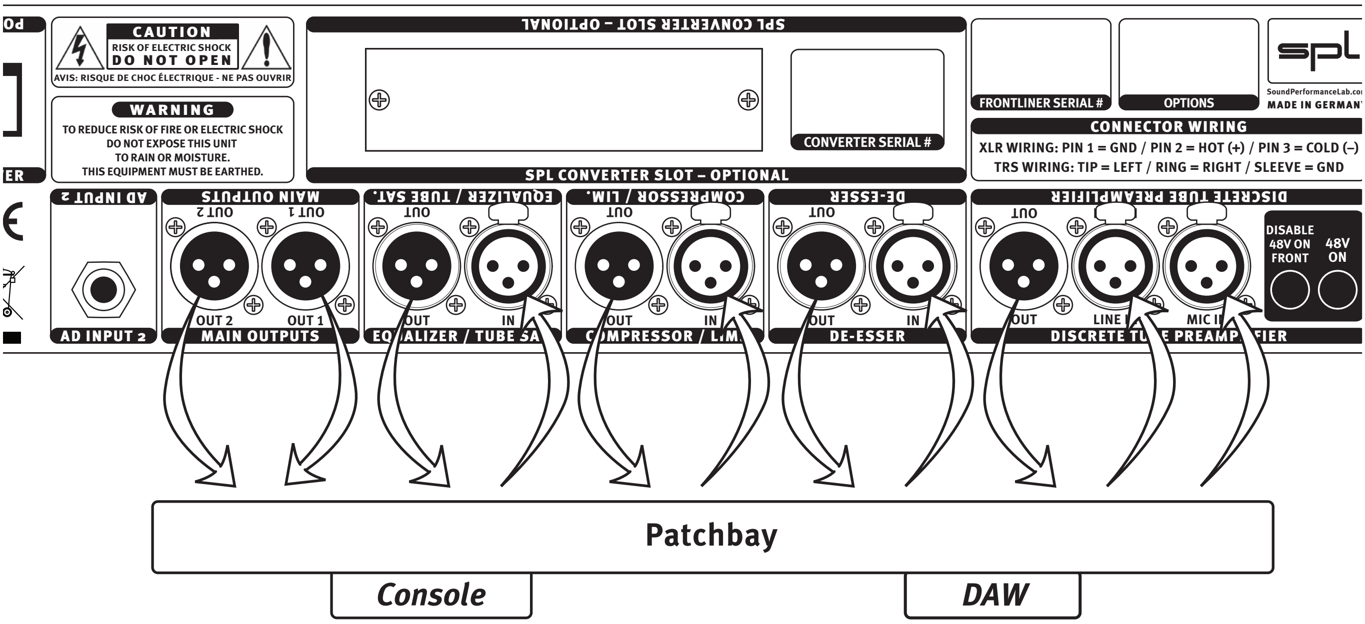
Microphone
Make sure that the 48V ON switch is only activated if the microphone needs phantom power supply
Line signals
Synthesizer, sampler etc.
Instruments
1/4" TRS jack front panel
P. A.
Monitor
FOH console
ON STAGE
STUDIO
Monitoring
Recording
Interface/DAW
Each of the two *MAIN OUTPUTS* is fed by its own driver stage to provide an independent connection for parallel use.
Make sure that the voltage switch setting reflects the correct local power line voltage.
Pin wiring 1/4" TRS socket (stereo jack): tip=left, ring=right, sleeve=ground
Pin wiring XLR output sockets: 1=ground, 2=hot (+), 3=cold (-)
Pin wiring XLR input sockets: 1=ground, 2=hot (+), 3=cold (-)
PUSH
TUBE PREAMPLIFIER
DISCRETE PREAMPLIFIER
MIC IN
48V ON FRONT
48V ON
CONNECTOR WIRING
OPTIONS
FRONTLINER SERIAL #
CONVERTER SERIAL #
SPL CONVERTER SLOT – OPTIONAL
DE-ESSER
COMPRESSOR / LIM.
EQUALIZER / TUBE SAT.
MAIN OUTPUTS
OUT 1
OUT 2
AD INPUT 2
POWER
GND LIFT
MAINS INPUT
VOLTAGE I FUSE
230V ~50Hz Fuse: 315mA slow
115V ~60Hz Fuse: 630mA slow
CAUTION
WARNING

Live application: channel strip for vocals, one module used separately



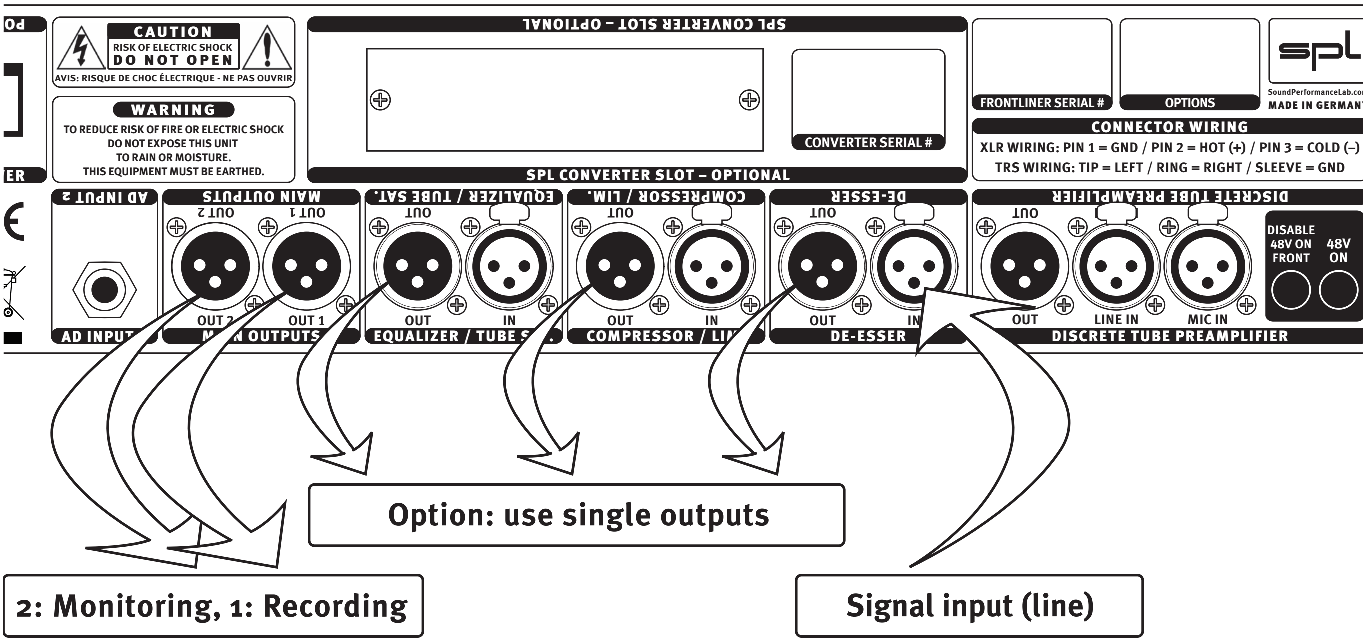
An example for a live setup: The compressor module is separated from the channel strip. The vocal signal can be processed with all remaining modules, while the compressor is used for processing bass. In the same way, each module can be taken out of the channel strip, e. g. using the De-Esser separately while processing an acoustic guitar with the (remaining) channel strip and so on. Detailed information on page 25.

Studio application: all inputs and outputs are routed to a patchbay



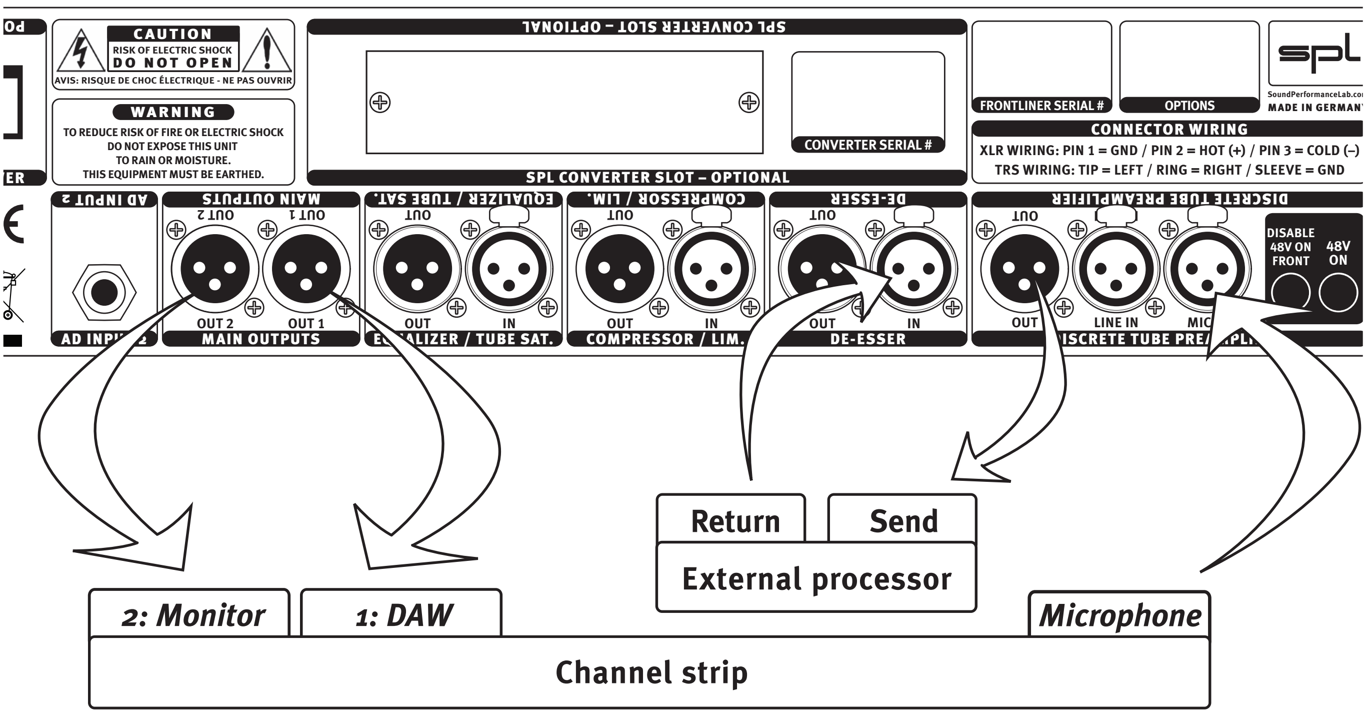
Our recommendation for studio wiring: all inputs and outputs are connected to a patchbay so that each Frontliner module can be routed with maximum flexibility within a studio environment. Connections to a console or to DAW interfaces for example can always be configured to meet the respective requirements. Aside from the switching options that the Frontliner provides the sequence of modules can be determined in any way with the patchbay.

Excluding a group of modules from the channel strip



This example shows the exclusion of de-esser, compressor and EQ/tube saturation as a group. The input signal is connected to the input of the first module. If all three processors are excluded as a group, the output signal of the group is present at the MAIN OUTPUTS. If a dual processor group is excluded, the output of the last module is always the group output. In the example above the preamplifier can be used separately via its own I/Os. Further information on page 25.

Determining inserts for external processors



You can determine inserts between every processing module. In the example above, an insert is determined between preamplifier and de-esser. The insert is routed via preamplifier output and de-esser input as send and return. Refer to page 26 for further information.

Signal connection

Switch off the unit before you begin the process of making the first or any subsequent connections. Neglecting this can damage either or both your ears and your equipment.

1/4" TRS socket

The TRS socket AD INPUT 2 supports both balanced (1/4" TRS/stereo jack connector) and unbalanced connections (1/4" TS/mono jack connector). Refer to page 13 for further information on that input.

XLR sockets

All XLR sockets are balanced inputs or outputs. Input sockets are always female for plugging in male connectors, output sockets are always male for female connectors. All in all a comprehensible principle.

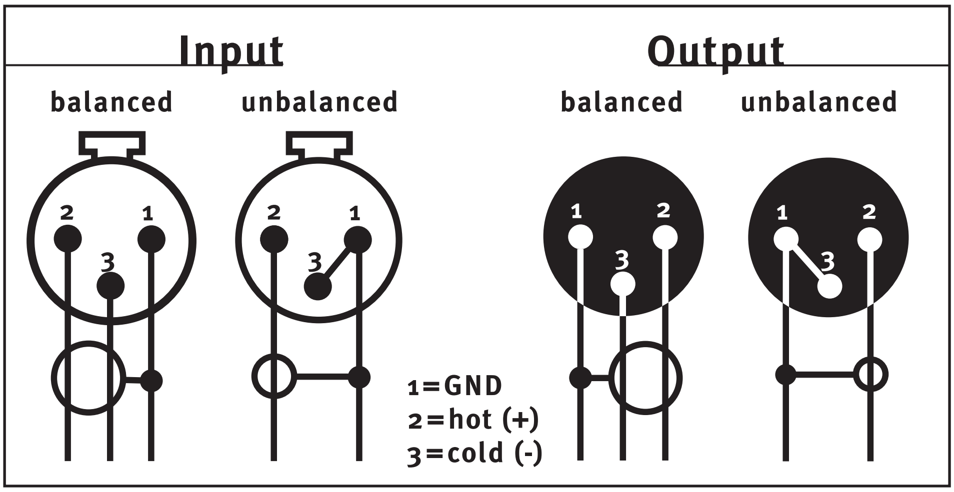
Balanced connections

It is impossible to exclude all 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.

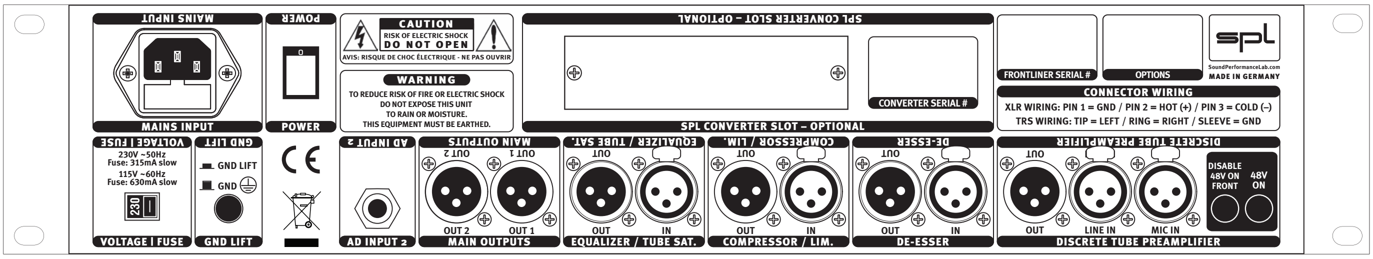
In balanced connections a reference signal with reversed polarity is transmitted additionally to the audio signal through a second wire. The ground signal is routed separately through a third wire. Input and output stages are drivers and receivers, and the receiving stage can suppress interferences by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

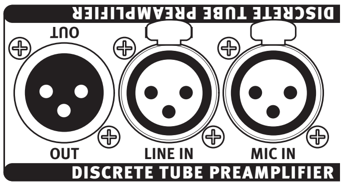


Connections to RCA sockets are always unbalanced, a wiring to jack connectors can be both balanced (1/4" TRS/stereo jack) or unbalanced (1/4" TS/mono jack). We recommend to use individually configured cables from XLR to RCA or jack sockets instead of adaptors. You can get cables in any needed configuration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



Separate inputs and outputs

Each Frontliner module is equipped with its own input and output stage. An obvious purpose is to route the signal from any output, e. g. right after the preamp to store the pure recording signal. Further applications such as using single or grouped modules separately or inserting external processors are depicted on pages 9 and 10. Inserts can be determined before each processing module input.



Preamplifier: MIC IN

You can connect any kind of microphone to the MIC IN socket (dynamic, condenser, tube and ribbon microphones). 48 volts phantom power, which is required for some microphones, can be activated with the 48V switches on the front or rear panel. Please read the important notes in chapter “48V” on page 15. The microphone input can also be equipped with an optional transformer (refer to page 27, “Information on I/O transformers”).

Preamplifier: LINE IN

Use the balanced LINE IN socket for high-level signals with impedances lower than 1 kOhm, e. g. D/A converters, synthesizers or samplers. We recommend connection to a patchbay for easier access.

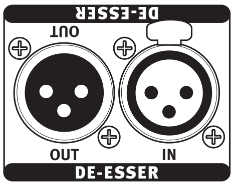
The maximum input level of the LINE IN is +22 dBu.

High impedance sources (above 1 kOhm), such as e-guitars and basses, acoustic guitars with pick ups and so on, must be connected to the instrument input (see pages 16 and 17).

IMPORTANT: The line input is deactivated if the instrument input is in use.

Preamplifier: OUT

The electronically balanced, analog preamplifier output provides the preamplified signal prior to any processing. If you determine an insert between preamplifier and de-esser, the preamplifier output serves as insert send.

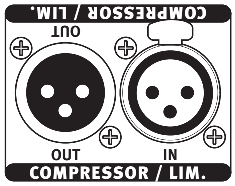


De-Esser: IN and OUT

The analog de-esser input and output sockets are electronically balanced.

Normally the de-esser input signal is the internal preamplifier signal. If you are using the de-esser module separately as single module, you connect the source and receiving units to the de-esser inputs and outputs. If you are using the de-esser module as part of a group of modules, you connect the source and receiving units to the group inputs and outputs. If you determine an insert between preamplifier and de-esser, the de-esser input serves as insert return.

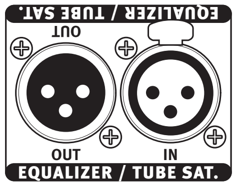
Compressor: IN and OUT



The analog compressor input and output sockets are electronically balanced.

Normally the compressor input signal is the internal de-esser signal (or the EQ signal if EQ PRE COMP. is activated). If you are using the compressor module separately as single module, you connect the source and receiving units to the compressor inputs and outputs. If you are using the compressor module separately as part of a module group, you connect the source and receiving units to the group inputs and outputs. If you determine an insert between de-esser and compressor, the de-esser output is the insert send and the compressor input is the insert return.

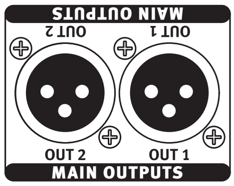
Equalizer: IN and OUT



The analog equalizer/tube saturation input and output sockets are electronically balanced.

Normally the EQ input signal is the internal compressor signal (or the de-esser signal if EQ PRE COMP. is activated). If you are using the equalizer module separately as single module, you connect the source and receiving units to the equalizer inputs and outputs. If you are using the equalizer module separately as part of a module group, you connect the source and receiving units to the group inputs and outputs. If you determine an insert between compressor and equalizer, the compressor output is the insert send and the equalizer input is the insert return.

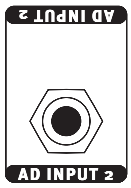
MAIN OUTPUTS: OUT 1 and OUT 2



The output signal of the channel strip is supplied by the electronically balanced, analog outputs OUT 1 and OUT 2. The splitted outputs allow to feed recording and monitoring devices simultaneously, or, in a live application, monitoring and P. A. paths. Each output is driven by its own stage to exclude mutual influences. OUT 1 can optionally be equipped with a transformer (refer to page 27, “Information on I/O transformers”).

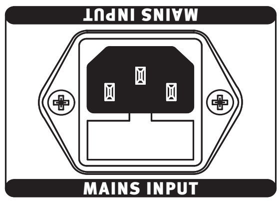
If you have excluded all three processing modules from the channel strip as a group, the group’s output signal is supplied by the MAIN OUTPUTS. In this case, the preamplifier output signal is only present at the preamp’s output. In all other cases the final output signal of the channel strip is always present at the MAIN OUTPUTS.

AD Input 2



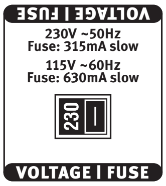
The Frontliner is a mono channel strip, but the optional A/D converter card 2376 is a dual-channel device. Therefore a second (external) signal can be converted with the converter card, if it is connected to the AD INPUT 2. If no signal is connected to the A/D INPUT 2, the output signal of the Frontliner is routed to both converter channels. The maximum input level for the converter is +12dBu (=0 dBFS).

Power connection and fuses



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

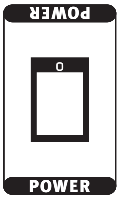
The Frontliner’s power supply houses two fuses, one for the neutral wire and one for the phase conductor. The fuses are accessible from outside and placed right behind the flap below the socket. Fuse ratings are 315 mA slow blow (230 volts) or 630 mA slow blow (115 volts).



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!

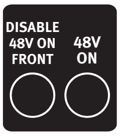


Power switch

The rear panel POWER switch activates the unit, confirmed by the illuminated VU meter.

Switch on the unit only after you have checked the correct setting of the rear VOLTAGE SELECTOR and 48V phantom power supply front and rear switches.

When you activate the Frontliner, the unit commences the warm-up mode to heat the tubes. The warm-up cycle takes between 45 seconds (if the unit was active shortly before) and approximately one minute at cold start. During warm-up the MUTE switch flashes. The MAIN OUTPUTS and the output to the optional converter card are muted, all EXTERNAL INPUTS keys are deactivated. The MUTE switch stops flashing when warm-up is finished. The MUTE switch does not illuminate if it was not depressed before. It illuminates permanently after warm-up when it was activated before.



Phantom power supply/rear panel switches

48 Volt phantom power is only needed for microphones requiring external current (generally condenser microphones). Other microphones can be damaged if they are used with activated phantom power, which may damage the Frontliner, too. Please read the notes on using phantom power and how to activate it under “48 V” on page 15 and “Activating Phantom Power” on page 16.

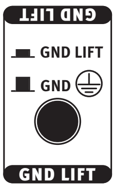
Since phantom power is critical to operational safety, the Frontliner offers additional switches on the rear panel to exclude accidental use of the 48 V switch on the front.

If both rear panel switches are not engaged, you can activate and deactivate phantom power with the 48 V switch on the front.

The switch to the right labelled 48 V ON activates phantom power independently from the front panel switch setting – now phantom power can no longer be switched off with the front panel switch.

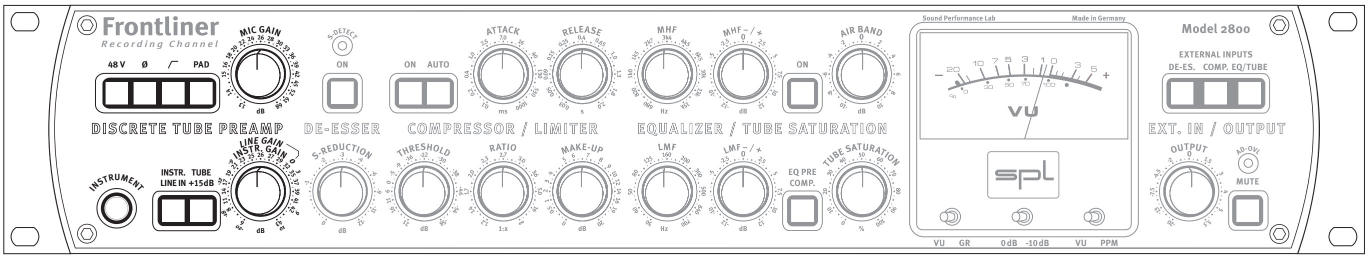
The left rear panel switch labelled DISABLE 48 V ON FRONT deactivates the 48 V switch on the front, now phantom power can no longer be activated accidentally with the front panel switch.

The 48 V switch on the front is always illuminated if phantom power is active.

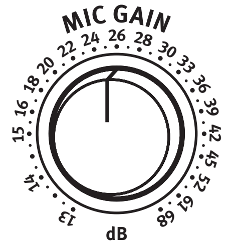


GND Lift

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.



MIC GAIN



The MIC GAIN control determines the level of preamplification. The preamplification values cover a range from +13 dB up to + 68 dB. If an optional microphone input transformer is fitted the scaled values are to be increased by ca. +14 dB (depends upon microphone. Further information on page 27, “Information on I/O transformers”). The potentiometer is a RK 27 from ALPS, a high-grade model with antilogarithmic characteristic and an especially high resolution between 20 dB and 40 dB to allow for sensitive control within this most important range.

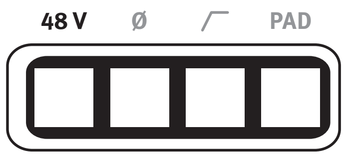
MIC GAIN adjustments

Amplifying a mic signal to line level should principally be done solely with the MIC GAIN control. Deactivate all processing modules in the first place. The OUTPUT level control is used to adjust the proper drive level for subsequent equipment, so initially it should be set to 0 dB (12 o’clock position). The 0 dB/-10 dB switch should be set to 0dB as well to see the overall output level (also see “0 dB/-10 dB” on page 23).

Now turn up the MIC GAIN control until the VU shows maximum values between 0 dB and +3 dB. There is still enough headroom to prevent clipping at varying input levels. Note that the VU meter shows average levels instead of peak levels (which can be up to 10 dB higher). If necessary, switch the VU to PPM mode to see peak levels (see “VU/PPM switch” on page 23).

In recording high-level signals such as snare and kick, etc. it may be necessary to engage the PAD switch (see “PAD” on page 16). You can drive the MIC GAIN harder if you know that unexpected input level changes are unlikely to occur. Choose the -10dB setting with the 0 dB/-10 dB switch to extend the VU meter range accordingly.

48 V



The 48 V switch activates phantom power for condenser microphones with built-in amplifiers. Phantom power should only be activated when using microphones that require it.

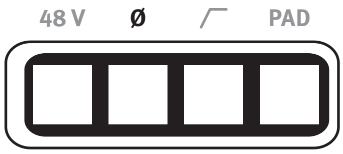
VERY IMPORTANT: All microphones with balanced, ground-free outputs, can be used with the phantom power activated. Please be sure to deactivate phantom power with all other microphones. Unbalanced microphones may only be used with phantom power deactivated.

Since phantom power is critical to operational safety, the Frontliner offers additional switches on the rear panel to exclude accidental use of the 48 V switch on the front. The function of these rear panel switches is described on the previous page.

Activating phantom power

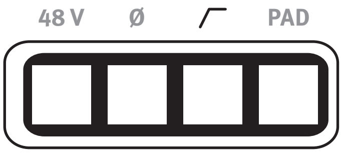
PLEASE ALWAYS FOLLOW THESE INSTRUCTIONS TO ACTIVATE AND DEACTIVATE PHANTOM POWER (ALSO WHEN CHANGING MICROPHONES). THE FRONTLINER’S INPUT STAGE CAN BE DAMAGED IF YOU IGNORE THESE PROCEDURES!

1. Connect the microphone to the Frontliner.
2. Now activate phantom power to use the microphone.
3. When finished, first deactivate phantom power.
4. **Wait at least one minute after deactivation of phantom power before disconnecting the microphone!** This ensures residual current will be discharged.



Phase reverse

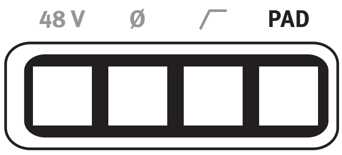
The phase reverse function reverses the polarity of the microphone signal, inverting the phase (by 180°) to correct phase-inverted signals caused by multiple signal sources. A voice-over artist, for example, hears himself during recording through the headphones and simultaneously through the bones in his head. Phase inversion will cause an unnatural sound, and even minimal variations in distance to the microphone will cause drastic variations in the sound. Phase inversion is also commonly encountered when using multiple microphones on a single sound source. We recommend checking for correct polarity before each recording. Phase reverse can be applied to the preamplifier microphone and line inputs.



High-pass filter

This switch activates the high-pass filter, which operates from 85 Hz downwards with 6 dB/octave (often also called a “rumble filter“). The filter prevents the amplification of unwanted low-frequencies. Compared to 12 dB/octave filters, the 6 dB filter works relatively moderate, but very musical.

The high-pass filter can be applied to all three preamplifier inputs (microphone, line, and instrument).



PAD

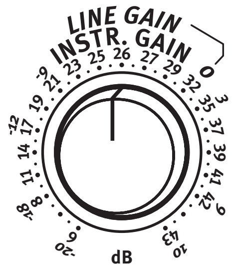
The PAD switch attenuates the microphone input by 20dB. High-level input signals can be attenuated in order to prevent over-driving the preamplifier. The Frontliner has a minimum pre-amplification of +13 dB. But with signals from drums or brass this may already be too much. Engaging PAD attenuates the minimum pre-amplification (MIC GAIN control fully counter clockwise) to -7 dB, thus providing a useful gain control range for loud input signals.

INSTRUMENT input

The INSTRUMENT input jack is placed on the front for easy access. It should be used to connect instruments like e-bass and guitars, acoustic guitars with pick-ups, etc. The Instrument input features a 1 MOhm (one mega Ohm) input impedance. Line signals with lower impedances, such as from D/A converters, samplers, synths, etc. should be connected to the rear LINE IN socket.

IMPORTANT: As long as an instrument is plugged into front INSTRUMENT input, the rear panel LINE IN input is deactivated.

LINE GAIN/INSTR. GAIN



This control determines the level of preamplification for signals connected either to the rear LINE IN or to the INSTRUMENT input on the front. You activate the respective input with the INSTR./LINE IN switch which is described below.

Gain range for line signals reaches from -20 dB to +10 dB. The attenuation allows to also process very high levels.

Instrument signals can be amplified between +6 dB and +43 dB.

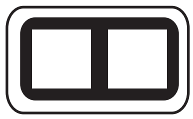
LINE GAIN and INSTR. GAIN adjustments

For optimal leveling of line or instrument signals, deactivate all processing modules and set the OUTPUT level control to 0 dB (12 o'clock position). The VU meter's 0 dB/-10 dB switch should be set to 0dB to see the correct output level (also see "0 dB/-10 dB" on page 23).

Now turn up the LINE GAIN/INSTR. GAIN control until the VU shows maximum values between 0 dB and +3 dB. There is still enough headroom to prevent clipping at varying input levels. Note that the VU meter shows average levels instead of peak levels (which can be up to 10 dB higher). If necessary, switch the VU to PPM mode to see peak levels (see "VU/PPM switch" on page 23).

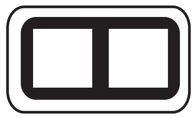
Levels between 0 and +3 dB usually are safe. Attenuate very high input levels with the LINE GAIN control. Switch the VU meter to -10 dB display mode to extend the metering range accordingly.

INSTR./LINE IN



With this switch you select between the microphone (off) and line or instrument inputs (on). The rear mic and line inputs can remain connected, regardless of which input is selected. You can choose the line input as source as long as the instrument input is not being used.

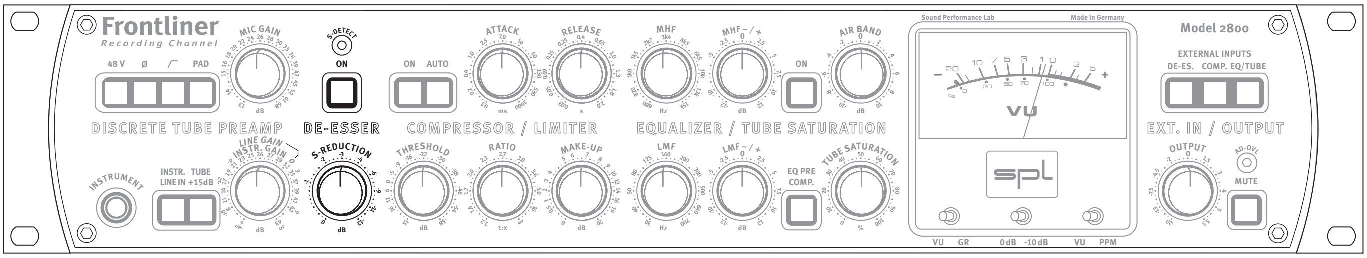
TUBE + 15 dB

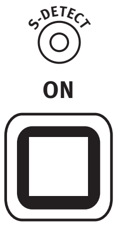


The TUBE +15 dB switch increases the degree of tube preamplification. The switch changes from standard tube preamplification (+6 dB) to +15 dB. In each setting the total output level of the preamp is automatically accommodated so that output levels do not have to be re-calibrated when the tube is driven hotter – therefore MIC GAIN settings do not have to be changed.

Engaging the TUBE +15 dB switch has an impact on the overall sound, as it directly changes the sonic character of the tube. The higher the input level, the more you will notice the effects of the tube's typical even harmonic distortion.

Driving the tube with +15 dB creates subtle to modest presence effects which can put the signal up front in a mix. For decent effects you should not apply too high MIC GAIN values with the TUBE +15 dB switch engaged.





ON

The first processing module is the de-esser, which removes disturbing sibilants when required. The de-esser module is activated with the ON button. The S-DETECT LED above the ON switch shows that S-sounds are being detected.

S-REDUCTION

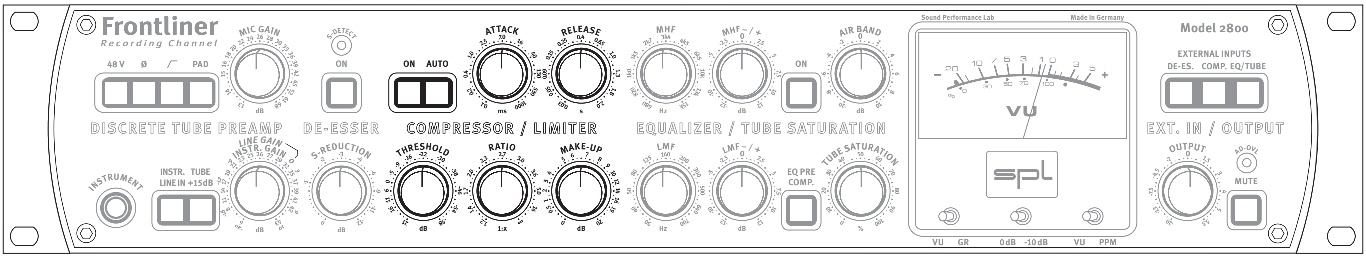
With the S-REDUCTION control you determine the intensity of S-sound reduction. The processing is related to the levels of the entire frequency spectrum (see next section), therefore it is more intensive with extreme S-sound levels than with those of lower levels. This equalizes levels of sibilants in the output signal.

SPL De-Esser technology

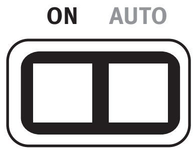
In contrast to common de-essers based upon compressor techniques the SPL De-Esser makes use of the phase cancellation principle. It employs filters that process only the reducible ”S-frequencies” but do not interfere with the remainder of the spectrum. The S-frequencies are detected automatically, the phase is inverted and mixed with the original signal. This method of operation has distinct advantages because it is unobtrusive and helps retain the original tonal quality. Compressor-typical side effects such as lisping or nasal tones do not occur. Finally its operation is as simple as pulling on the hand brake.

The reduction is accomplished by comparing the average level with the individual S-sounds: the de-esser functions only when the S-noise level exceeds the average level of the entire frequency spectrum. This means for example that original S-sounds with a certain S-portion are not processed whereas those that are too loud, or do not effectively contribute to the sound, are reduced – but the character of the voice remains unchanged.

A further specialty is the integrated Auto Threshold function which makes processing independent of the input level. Even when the speaker or singer does not maintain a constant distance to the microphone, processing is retained at the pre-set S-reduction value. Conventional systems are dependent on the input level and work more intensively as the distance to the microphone is reduced. As a result, the SPL De-Esser does not need to be monitored and re-adjusted permanently to keep processing constant – and it can always be applied before the compressor, as changing its position would not be an advantage. That is why an accordant switching function is not necessary.



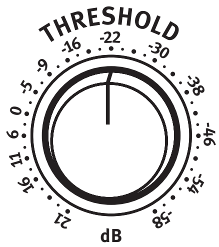
ON



The On button activates the compressor/limiter module. Set the VU/GR switch to GR (=gain reduction) to see the processing intensity (for further information read ”MAKE UP” on page 20).

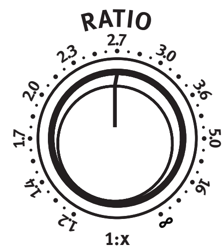
Usually the signal flow follows the design of the Frontliner, so the input signal normally comes from the de-esser. However, with the EQ PRE COMP. switch the compressor/limiter module can be switched behind the equalizer. This allows the compressor to be used as a final compressor or limiter after the equalizer (see „EQ PRE COMP” on page 15).

THRESHOLD



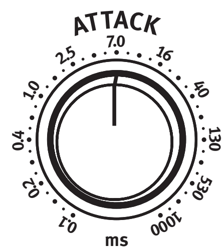
The THRESHOLD control determines the compressor threshold value, thus from which level the compressor starts working. The value scale for the THRESHOLD control indicates the level in dBu and ranges from 21 dB (fully left) to -58 dB (fully right). Turning the control to the right intensifies processing: fully left provides no compression, and fully right provides compression starting at a level of approximately -58 dBu.

RATIO



The RATIO control is used to set the ratio between the original signal and the compressed signal. A ratio of 1:5 (RATIO control set to 5.0) means that a level increase of 5 dB on the input results in an output level change of 1 dB. The more the control is turned right, the more ‘dense’ the sound becomes. The compressor works as a limiter when RATIO is turned fully right to infinity.

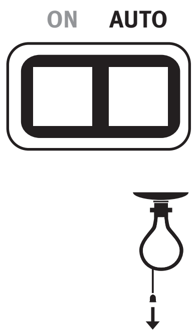
ATTACK



The ATTACK control determines how fast the compressor reacts. When turned fully left this time is 0.1 milliseconds, and turned fully right the time is approximately 1000 milliseconds (one second). Sudden, fast and impulsive attacks are called transients, they are produced for example by hitting a drum. We identify natural sounds by transients, so they are enormously important for sound impressions.

With very short ATTACK times the compressor reacts to every level jump and all transients are processed. With slower settings, transients become more and more audible – a drum kit for example can achieve more presence in a mix and sound faster with such settings. The best ATTACK settings are not always easy to find, since you usually need to find a compromise. Very fast settings run the risk of producing audible distortions – especially in the case of low frequencies, since the compressor now tries to catch each waveform rise. The corresponding control signal assumes a “saw-tooth” form which distorts the audio signal. To suppress this effect, you would have to again increase the attack time until distortion no longer occurs – but without missing the aim of processing.

The Frontliner offers a switchable automation feature for the ATTACK and RELEASE parameters to achieve perfect results when you are not satisfied with fix settings (see next page).



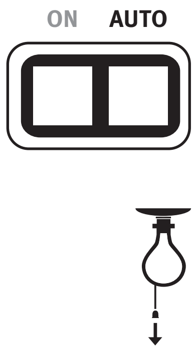
ATTACK with cruise control: AUTO mode

The disadvantage of a usual ATTACK control: the setting refers to one critical moment, but it is not optimal in every moment. But when you switch to AUTO mode, the ATTACK settings interactively respond to the input signal curves. Moreover, the automation can still be influenced manually. With the ATTACK control you can determine the range in which the automation sets it's values. Much like a cruise control resumes up to the set speed, the AUTO mode does the same for the (automated) ATTACK time settings. The more you turn ATTACK to the right, the broader is the range for the automation to set time values – the ATTACK setting you have chosen is the maximum ATTACK time the automation applies.

RELEASE

The RELEASE control determines the time the compressor needs to get back to the initial value after a level reduction. Turned fully left this time is ca. 0.03 seconds, or ca. 2 seconds for fully right.

Just like setting the ATTACK it can be difficult to find perfect RELEASE settings – for the same compromising reason. Fast transients need short RELEASE times, but processing each and every dynamic change bears the danger to produce rough, distorted sounds. Longer RELEASE times sound smoother, but now transients may slip through. Again the AUTO mode can help to achieve better results.

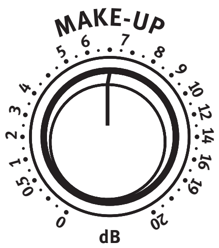


RELEASE with cruise control: AUTO mode

The AUTO mode for RELEASE first determines the average level of the signal. Then the automation refers to this level to achieve a balanced control characteristic: short and loud peaks can be processed very fast, but especially with complex material (summed signals) not every small peak near the average level is included into the processing.

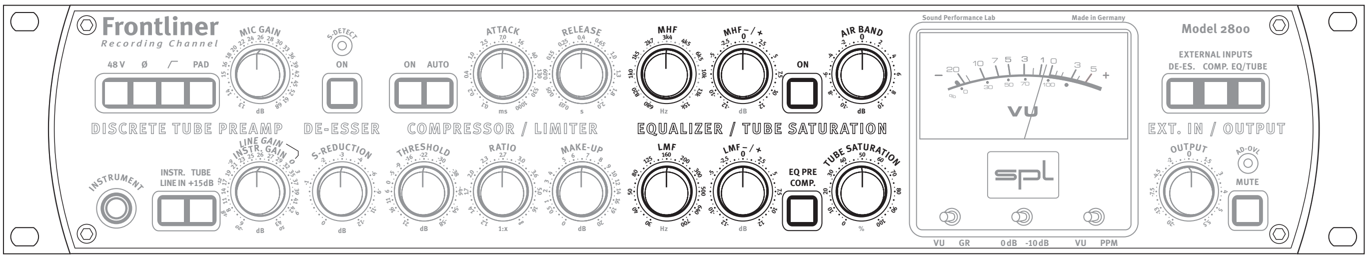
The RELEASE automation can be influenced manually, too: the shorter the RELEASE time ist set, the smaller is the range in which the processing follows the signal. Thus, with short RELEASE times the processing follows very precisely and nearly "sticks" to the signal curve. Short RELEASE times are for example adequate for percussive signals. If you apply longer RELEASE times, the range is getting broader. Medium to slow settings can be recommended for summed signals.

The AUTO switch activates ATTACK and RELEASE automation simultaneously. It is always very effective to gain maximum loudness with minimum consequences in a musical sense. This applies to all kinds of signals, from percussive sounds to vocals and instruments, and aside from single tracks especially also to the complex structures of summed signals ... not to mention a kind of signal that's as difficult as it is common, called "untrained voice".



MAKE UP

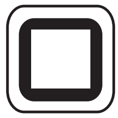
With the MAKE UP control you can make up the output gain that's lost through compression. It changes the output level within a range of 0 dB to +20 dB. Setting is very easy when you switch the VU/GR switch below the VU meter to "GR" (GR=gain reduction). If for example the loudest peal causes a maximum reduction of -9 dB, you adjust the MAKE UP control to +9dB. Switch the compressor out now to perceive the loudness gain.



ON

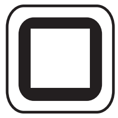
The ON switch adds the Equalizer/Tube Saturation module to the signal path. Usually the input signal comes from the compressor module. With the EQ PRE COMP. key you can switch the EQ module in front of the compressor, so that the input signal for the EQ module comes from the De-Esser module.

ON



EQ PRE COMP.

EQ PRE COMP.

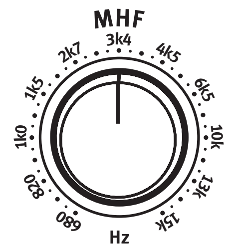


The EQ PRE COMP. key reverses the sequence of Compressor/Limiter and Equalizer: When the key is pressed the Equalizer operates in front of the Compressor/Limiter; when not pressed the succession remains unchanged.

Application examples: use of the compressor module as a limiter to maintain a stable output level. If the EQ was to be used again after limiting it could not be guaranteed that the output level would not alter. Or reduce frequencies with the EQ to prevent too excessive reactions within the compressor, respectively vice versa: emphasize frequencies to let the compressor react more intensively.

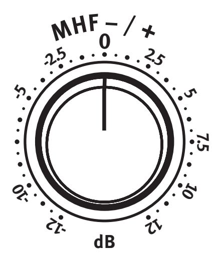
The EQ PRE COMP. key only switches the equalizer module, but not the tube saturation stage. This function can also be used if compressor and EQ are taken out of the channel strip unit as a group. In this case the signal connected to the compressor inputs is directly routed to the EQ inputs. The EQ PRE COMP. function can not be activated if the compressor or EQ module is is excluded from the channel strip via EXTERNAL INPUTS and operated as a single module or if an insert was determined via the compressor or EQ inputs. The EQ PRE COMP. key flashes, if the function is not accessible for one of the mentioned reasons.

MHF (mid-/high filter)



The MHF control serves to set the center frequency of the semi-parametric mid/high filter. The control range is between 680 Hz and 15 kHz, so the filter covers a broad range of 4.5 octaves for processing of both mid and high frequencies.

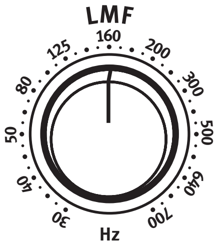
MHF -/+ (cut/boost MHF)



The MHF -/+ control determines the boost, or cut of the mid/high filter; the maximum values range up to +/- 12 dB. The filter utilizes the proportional-Q principle. In other words the bandwidth (=Q) is dependent on the selected boost or cut. Higher boost or cut values narrow the bandwidth; with lower boost or cut values the bandwidth broadens. This filter characteristic permits a musically more sensible processing of the frequency spectrum than with constant-Q filters: if a more thorough setting has been chosen this will lead to a precise definition of the frequency range to be processed. This in turn minimizes influences from adjacent ranges. This filter construction permits the complete scope, from selective removal of accentuated frequencies through to character giving accentuations, to be effectively and quickly covered.

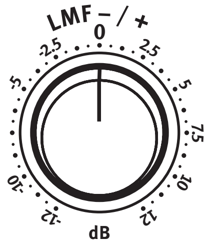
Recommendation on frequency settings: To find the frequency which is to be processed as quickly and accurately as possible, firstly adjust the MHF -/+ control to the maximum position. Subsequently the relevant frequency should be sought. Because the filter at maximum setting works with the smallest bandwidth, the frequencies can be heard most distinctly at this setting, making them easier to locate. Finally the desired MHF +/- setting can be applied after the frequency is determined with MHF.

LMF (low/mid filter)



The center frequency of the half-parametric low/mid filter is set with the LMF control. The adjustable frequency range lies between 30 and 700 Hz so that this filter covers a range of about 4.5 octaves, allowing it to be used from the deepest bass to the lower mid range. This together with the mid/high filter ensures that the entire frequency spectrum is covered.

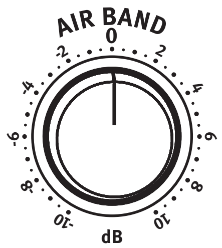
LMF -/+ (cut/boost LMF)



The LMF-/+ control determines the cut or boost of the LMF filter; the maximum values are +/- 12 dB. The LMF filter also operates to the proportional-Q principle, in other words the bandwidth is dependent on the selected boost or cut. The LMF filter can be applied in many ways. Examples are to accentuate the fundamental sound of a voice, to cut “boom frequencies” and for placement of instruments such as bass guitar, bass drums or synthesizers, both during recording or subsequently when mixing.

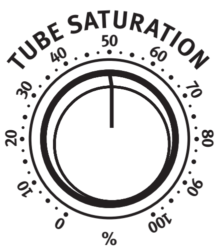
Recommendation on frequency settings: To find the frequency which is to be processed as quickly and accurately as possible, firstly adjust the LMF -/+ control to the maximum position. Subsequently the relevant frequency should be sought. Because the filter at maximum setting works with the smallest bandwidth, the frequencies can be heard most distinctly at this setting, making them easier to locate. Finally the desired LMF +/- setting can be applied after the frequency is determined with LMF.

AIR BAND



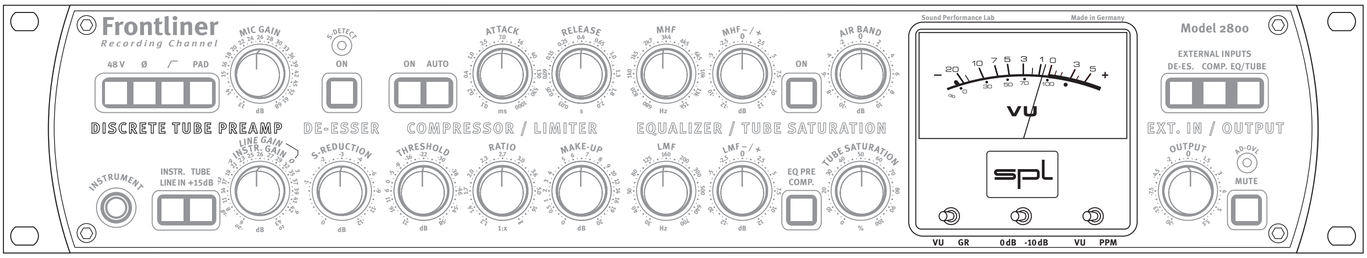
The high frequency filter in the equalizer module is described as the AIR BAND. A coil/capacitor filter with so called bell characteristics and a center frequency of 17.5 kHz comes into operation here. At this frequency the maximum possible accentuation is +10 dB, the maximum possible damping is -10 dB. The soft and natural tonal characteristic of the coil-capacitor filter lends itself extremely well to provide clarity and, well ... air to vocals in the upper frequency range, thereby improving their presence. On the other hand harsh sounds can be lent a more pleasant sound characteristic through damping.

TUBE SATURATION



With this control you determine the amount of tube saturation. The output level is accommodated automatically, in extreme settings the level increases by only 6 dB. Therefore you can easily dial in decent to expressive harmonic distortions with the TUBE SATURATION control by turning just one knob.

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.



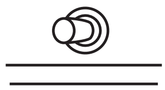
VU meter

The VU meter displays the output level (behind the OUTPUT control). 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 “0 dB/-10 dB” below).

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, as these characteristics represent ideal conditions for reading both VU and PPM values.

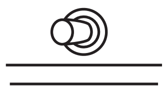
VU/GR switch



VU GR

With the VU/GR toggle switch you can select if the VU meter displays the output level (VU) or the gain reduction (GR) caused by compression. In GR mode the needle jumps to the 0 dB mark and moves from there to the left, the more the compressor takes action. The displayed values correspond to the level reduction. If the loudest peak causes the compressor to execute a -9 dB reduction, the MAKE UP control should be set to values around +9 dB to compensate for (see “MAKE UP” on page 20 for further information).

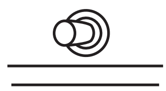
0 dB/-10 dB switch



0 dB -10 dB

With this toggle switch you can change the sensitivity of the VU display. If you select the -10 dB setting, the sensitivity of the display is lowered by 10 dB. With the needle for example at 0 dB, a value of +10 dB is displayed. Now you can read values of up to +15 dB, thus read much higher levels. Switching the VU sensitivity is important with high levels. If you have for example levelled a signal to 0-3 dB during recording, an EQ processing can easily result in a level increase above 5 dB – now the sensitivity switch allows to keep following the exact output level values.

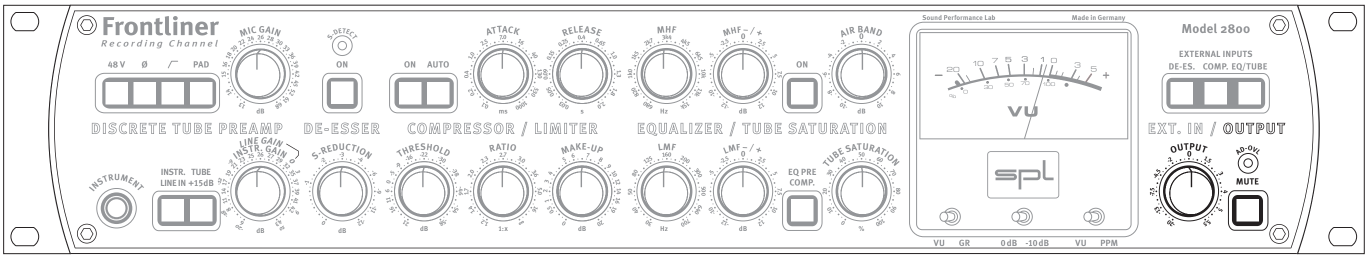
VU/PPM switch

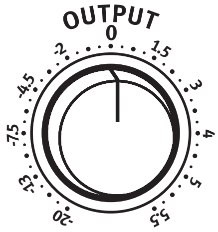


VU PPM

The VU/PPM toggle switch allows to switch the VU metering characteristics from VU display mode to PPM display 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.

A/D converter displays 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. At times it may make sense to also select the -10 dB setting to prevent the needle from getting stuck on the right side of the gauge.





OUTPUT

The outgoing signal can either be dampened to -20 dB or further amplified by +5.5 dB with the OUTPUT control to provide optimal drive to the subsequent units or the optional A/D converter 2376. The individually selected output level is shown in the PPM display mode of the VU meter. Before a recording commences the OUTPUT control should be set to 0 dB – the uninfluenced values from the OUTPUT control are then legible and available for adjustment of the preamplifier levels (also refer to “MIC GAIN adjustments” on page 15).

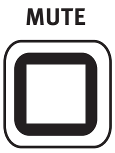
AD OVL-LED

This LED is activated only if the optional A/D converter 2376 is fitted. It indicates that the converter 2376 is overloaded. For information on the converter 2376 please read page 27, “24/96 AD converter, model 2376“.

The AD OVL-LED never illuminates if the unit is not equipped with the optional converter.

If the optional converter is fitted, the AD OVL should not illuminate. Otherwise reduce the output level until it does not illuminate anymore.

If you are using external A/D converters you must use their level meters to control the input levels.



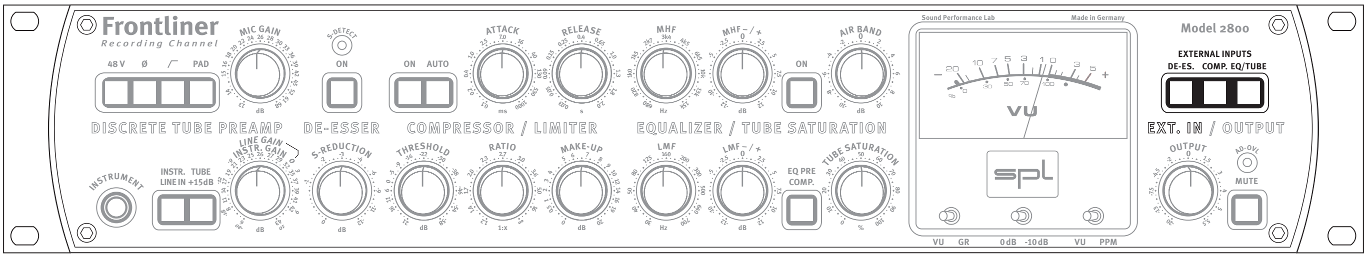
MUTE

The MUTE switch mutes the output signal at both MAIN OUTPUTS and at the output to the optional converter 2376.

An instance could be allowing the musician to practice for a while and then, when ready, freeing the signal path and commencing recording by deactivating the MUTE switch.

Since only the output signal is muted, you can still adjust gain settings etc. (with the VU meter).

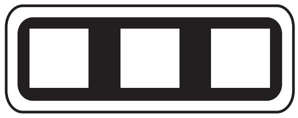
During the warm-up phase the MUTE switch flashes. The MAIN OUTPUTS and the output for the optional converter card are muted, all EXTERNAL INPUTS keys are deactivated. The MUTE switch stops flashing when the warm-up mode is finished. The MUTE switch will not illuminate if it was not activated before switching in the unit and it will illuminate permanently after warm-up when it was activated before.



EXTERNAL INPUTS

EXTERNAL INPUTS

DE-ES. COMP. EQ/TUBE



A main feature of the Frontliner is the modular concept. That's why each module has its own input and output. Aside from the standard application – using the Frontliner as a recording channel – you can also integrate each module independently into your studio or live setup, just like an analog plug-in. With the EXTERNAL INPUTS keys you can apply all switch functions to exclude single modules or dual and triple module groups from the channel strip unit and you can determine inserts between the modules. See pages 9 and 10 for wiring diagrams with application examples.

Let's start with the standard configuration: No EXTERNAL INPUTS key is engaged. You can use the entire channel strip: preamplifier, de-esser, compressor and EQ with tube saturation stage for one signal. The EQ can be switched in front of the compressor with the EQ PRE COMP. key.

Excluding a single module: Press the respective EXTERNAL INPUTS key and hold it for about one second.

The key illuminates permanently and the module can now be used with its own inputs and outputs. With the de-esser as example, the signal now runs from the preamplifier directly into the compressor, while the de-esser can be used as an autarchic device. Each processing module (de-esser, compressor, and equalizer with tube saturation) can be taken out of the channel strip unit in this way. EQ and tube saturation stage are always treated as one module. If all modules are excluded as single modules, the preamp signal is routed directly to the MAIN OUTPUTS – the Frontliner now is a pure preamplifier.

Excluding a group of two or three modules: Press the respective EXTERNAL INPUTS keys of two or three modules simultaneously and hold them for about one second.

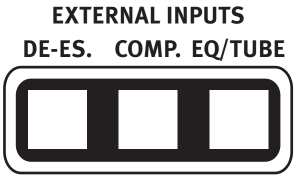
Pressing and holding the EXTERNAL INPUTS keys of two or three modules simultaneously defines a group and excludes this group from the channel strip unit. The input of the first module is the group input. All module outputs can be used, the output of the last module is the group output. With the example of de-esser and EQ as an excluded dual module group, the Frontliner signal now runs from the preamplifier directly into the compressor and from there to the MAIN OUTPUTS. The de-esser/EQ module group could be used for example to treat vocals from another preamp, while the Frontliner is used to record and process (compression) an acoustic guitar. The successful determination of a group is confirmed by pulsing illumination of the respective EXTERNAL INPUTS keys. Like the signal flow the pulse moves from left to right. If all three modules are excluded from the channel strip unit as a group, the group input is the de-esser's input, but the output signal is also present at both MAIN OUTPUTS. The preamplifier signal in this case is only present at the preamp's output.

Ungroup: Press and hold the EXTERNAL INPUTS keys for modules to be grouped for about 1 s.

The module group is ungrouped when the pulsed illumination goes out. Pressing only one key within in a group also ungroups the grouped modules, but the respective module is also excluded from the channel strip unit.

Notes on the EQ PRE COMP. key: This function can also be applied if compressor and EQ are excluded as a group. In this case the input signal connected to the compressor input sockets is directly routed to the EQ inputs. The EQ PRE COMP. function is not available if either the compressor or EQ module is used as a single module or if an insert is determined at the compressor or EQ inputs. The EQ PRE COMP. key flashes if you press the key and the function is not available for one of the above reasons.

Please also note the block diagram on page 32 and the table on page 33 for details about switching configurations.



Determining inserts

You can also determine inserts with the EXTERNAL INPUTS keys in order to integrate further, external devices into the channel strip unit. Inserts can be determined before each processing module. If you for example wish to insert a unit between preamplifier and de-esser, the preamplifier output is the insert send channel and the de-esser input is the insert return.

To determine an insert e. g. between the preamp and de-esser module, press the EXTERNAL INPUTS DE-ES. key shortly once. The key illuminates pulsating. The de-esser input is now active, but the signal flow does not leave the channel strip – in contrast to the exclusion of a module (group). The same applies to all three modules.

Key lock

The EXTERNAL INPUTS keys have a determinant effect on the signal flow. Hence a little mistake can have a major consequence which one may want to exclude for example in a live situation. That's why the EXTERNAL INPUTS keys can be locked.

Lock the EXTERNAL INPUTS keys: Press the DE-ES. and EQ PRE COMP. keys simultaneously and hold them for about two seconds.

All EXTERNAL INPUTS keys blink four times and are locked now. The EQ PRE COMP. key and all other switches are not locked.

Unlock the EXTERNAL INPUTS keys: Again press the DE-ES. and EQ PRE COMP. keys simultaneously and hold them for about two seconds.

The EXTERNAL INPUTS keys blink two times to confirm successful unlocking.

The key lock does not change a switching configuration (exclusion of single or grouped modules, determination of inserts). If the unit is switched off with activated key lock, the key lock remains active after powering up the unit again. When the warm-up cycle is finished, the EXTERNAL INPUTS keys flash four times as if the key lock would be activated for the first time to remind you of the current status.

Keys and switches

The status of all EXTERNAL INPUTS and EQ PRE COMP. keys is always retained also after powering off the unit.

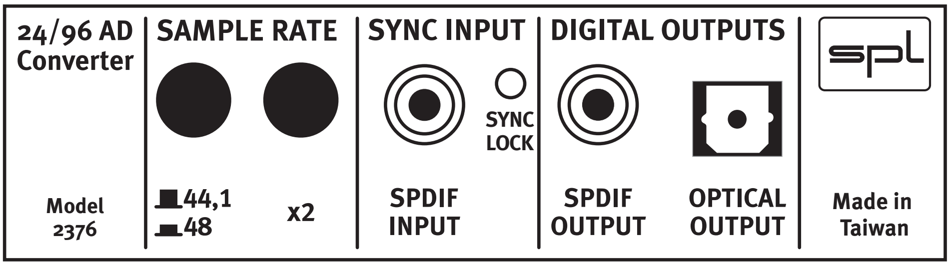
The EXTERNAL INPUTS and EQ PRE COMP. functions are operated by keys which switch relays. All other switches keep their status mechanically.

Faint switching noise can not be avoided when operating the EXTERNAL INPUTS keys, as the switching occurs during signal flow with relays. However, relays sound better and have less THD at higher levels than semiconductor-based switches, so we accepted the switching noise – particularly because an EXTERNAL INPUTS configuration usually is not altered during recording.

Please note that you can order products with optional equipment from all dealers, even if they do only list standard product versions, for example in an online store. Please contact your dealer or SPL before you place an order. Optional equipment can also be installed after sales.

Available option for the Frontliner, model 2800:

- 24 bit/96 kHz A/D converter (user installation possible).
- Lundahl input and output transformers (installation only by qualified technicians or SPL).



24/96 AD converter, model 2376

The optional converter card provides a digital output. Output signals are delivered via a S/P-DIF output through two sockets: one RCA socket and one optical socket. The converter provides 24 bit signals. All common sample rates can be selected (see below). Highly accurate quartz oscillators ensure a clean, low-jitter master clock.

SAMPLE RATE: The A/D converter allows you to select among the four most common sample rates of 44.1, 48, 88.2 and 96 kHz. The 44.1/48 button selects one of the two basic sample rates (out: 44.1 kHz; in: 48 kHz). The x2 button doubles these sample rates to select 88.2 or 96 kHz respectively.

DIGITAL OUTPUTS: The converted S/P-DIF signal is routed in parallel to the RCA and optical outputs. The signal is in professional format with no sample rate data in the status block.

SYNC INPUT: Since this is an AD converter, **the SYNC INPUT is no audio signal input**. The SYNC INPUT allows you to feed the converter with an external sample rate. Connect an S/P-DIF output from your master source (e.g. DAW interface) to the SYNC input. The AD converter will automatically switch to the same sample rate that is received. The A/D converter 2376 is not equipped to accept Word Clock synchronization.

The yellow Sync Lock LED illuminates when a valid sync signal is present at the SYNC INPUT and the converter is automatically synchronized to the external sample rate.

To prevent interference, the internal oscillators are automatically disabled when an external clock signal is present. If the sync signal is no longer present (e.g. in the case of a dropout), the converter automatically reverts to the sample rate selected via the converter’s control switches.

Information on I/O 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 microphones are in use.

Used in SPL preamps or channel strips, the input transformers add ca. 14dB gain (depending on the microphone). This must be added to the scaled values. The additional passive gain relieves the complete unit permanently at any gain level. The higher gain levels are also beneficial with ribbon microphones. That’s why the input transformer is more important in preamps, but to benefit from all possible sonic effects and full operational safety, both input and output should be equipped with transformers.

Inputs & Outputs

Electronically balanced instrumentation amplifiers, optionally transformer-balanced

Sockets/inputs	
Microphone input	XLR
Line input	XLR
Instrument input	6,35 mm (1/4") TRS (stereo jack)
De-Esser input	XLR
Compressor input	XLR
EQ-/Tube Saturation input	XLR
AD Input 2	6,35 mm (1/4") TRS (stereo jack)
Sockets/outputs	
Preamplifier output	XLR
De-Esser output	XLR
Compressor output	XLR
EQ-/Tube Saturation output	XLR
Main Outputs	XLR
Input impedances	
Microphone input	ca. 1,9 kOhms unbalanced/ca. 3,8 kOhms balanced
Line input	ca. 20 kOhms unbalanced/ca. 40 kOhms balanced
Instrument input	ca. 1 MOhm
De-Esser input	ca. 20 kOhms unbalanced/ca. 40 kOhms balanced
Compressor input	ca. 20 kOhms unbalanced/ca. 40 kOhms balanced
EQ-/Tube Saturation input	ca. 20 kOhms unbalanced/ca. 40 kOhms balanced
Output impedances	
Preamplifier output	ca. 600 Ohms unbalanced/ca. 1,2 kOhms balanced
De-Esser output	ca. 600 Ohms unbalanced/ca. 1,2 kOhms balanced
Compressor output	ca. 600 Ohms unbalanced/ca. 1,2 kOhms balanced
EQ-/Tube Saturation output	ca. 600 Ohms unbalanced/ca. 1,2 kOhms balanced
Main Output OUT 1	ca. 75 Ohms unbalanced/ca. 150 Ohms balanced
Main Output OUT 2	ca. 75 Ohms unbalanced/ca. 150 Ohms balanced
Maximum input level	
Microphone input	@ 13 dB gain: +9 dBu, PAD activated: +29 dBu
Line input	+22 dBu
Instrument input	+11,5 dBu
De-Esser input	+21 dBu
Compressor input	+21 dBu
EQ-/Tube Saturation input	+21 dBu
Maximum output level	21,5 dBu
Gain control ranges	
Microphone input	+13 dB to +68 dB
Line input	-20 dB to +10 dB
Instrument input	+6 dB to +43 dB

Measurements

Frequency range *(-3 dB)*

Channel strip	10 Hz to 50 kHz
Microphone preamplifier	‹10Hz to 50kHz
Line preamplifier	‹10Hz to 50kHz
Instrument preamplifier	‹10Hz to 50kHz
De-Esser	‹10Hz to 140kHz
Compressor	‹10Hz to 180kHz
Equalizer/Tube Saturation	‹10Hz to 60kHz

Common mode rejection *(@1 kHz, 0 dBu input level and unity gain)*

Microphone preamplifier	› 80 dB
Line preamplifier	› 75 dB
Instrument preamplifier	› 75 dB
De-Esser	› 75 dB
Compressor	› 75 dB
Equalizer/Tube Saturation line	› 75dB

Total harmonic distortion *(in %, @ 1 kHz, 0 dBu input level and unity gain)*

Channel strip	0,03
Microphone preamplifier	0,03
Line preamplifier	0,03
Instrument preamplifier	0,03
De-Esser	0,002
Compressor	0,007
Equalizer/Tube Saturation Line	0,01

Signal to noise ratio *(A-weighted)*

Microphone preamplifier (VU=30 dB)	-91,5 dB
Line preamplifier	-94,7 dB
Instrument preamplifier	-95,5 dB
De-Esser	-97 dB
Compressor	-97 dB
Equalizer/Tube Saturation Line	-91 dB

Dynamic range *(unweighted)*

Channel strip	110 dB (microphone input › main outputs, @ 30 dB)
Channel strip	112 dB (line input › main outputs)
Channel strip	112 dB (instrument input › main outputs)
Microphone preamplifier (@ 30 dB)	110 dB
Line preamplifier	114 dB
Instrument preamplifier	105 dB
De-Esser	115 dB
Compressor	115 dB
Equalizer/Tube Saturation Line	112 dB

E.I.N. microphone preamplifier	-127 dBu *(@ 68 dB gain, Rs=150 Ohms, 20-22 kHz)*

Power supply	Two toroidal transformers
Audio transformer	20 VA
Transformer other currents	15VA
Fuses	230 V AC, 50 Hz: 315 mA/120 V AC, 60 Hz: 630 mA
Voltage selector	115V/230V
Power consumption	29 W

Dimensions and weight

Housing (W x H x D)	482 x 88 x 261 mm *(Depth with controls and sockets)*
Weight	5,7 kg/ca. 12,57 lbs *(w/o converter & I/O transformers)*

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

Copy master: recall settings

Artist:

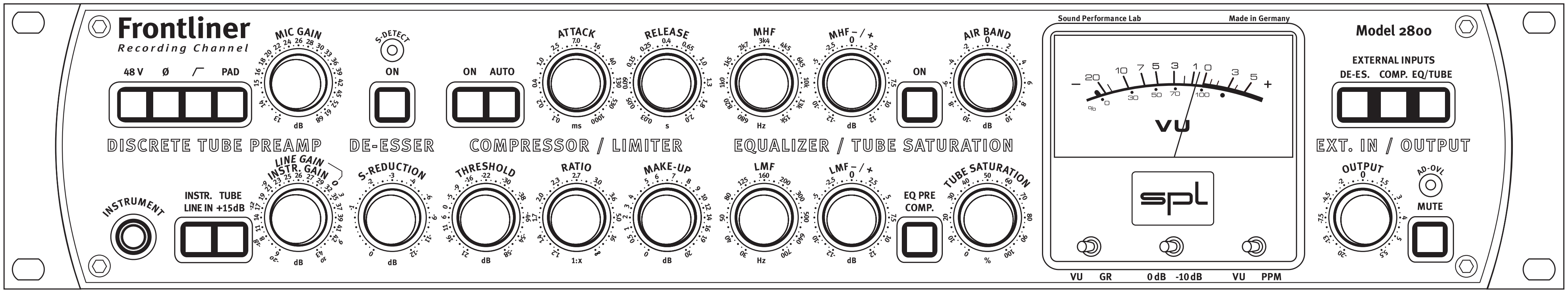
Engineer:

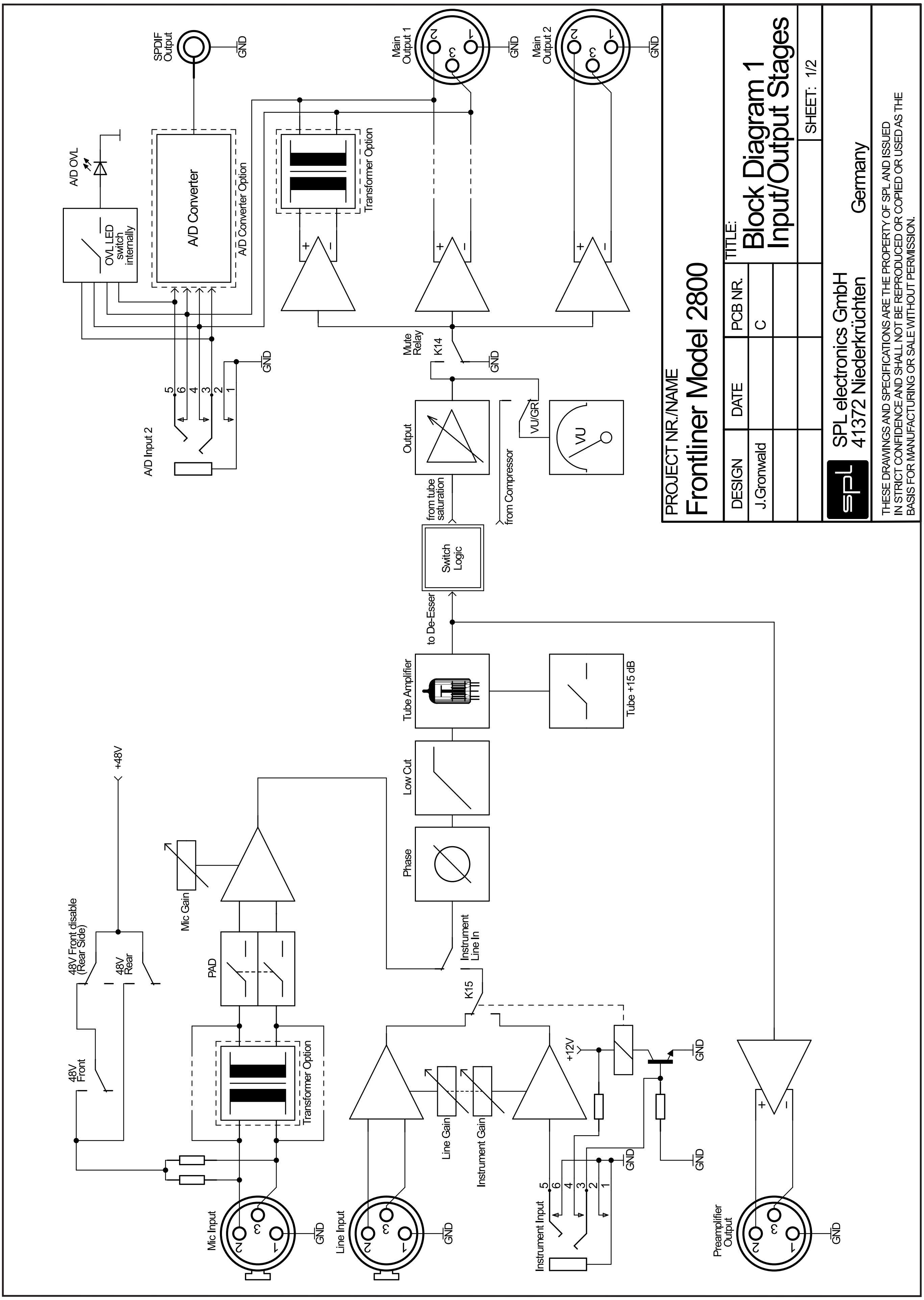
Album/Gig:

Track(s)/Groups:

Title:

Date:




PROJECT NR./NAME
Frontliner Model 2800
DESIGN
J.Gronwald
DATE
PCB NR.
C
TITLE:
Block Diagram 1
Input/Output Stages
SHEET: 1/2
SPL electronics GmbH
41372 Niederkrüchten
Germany
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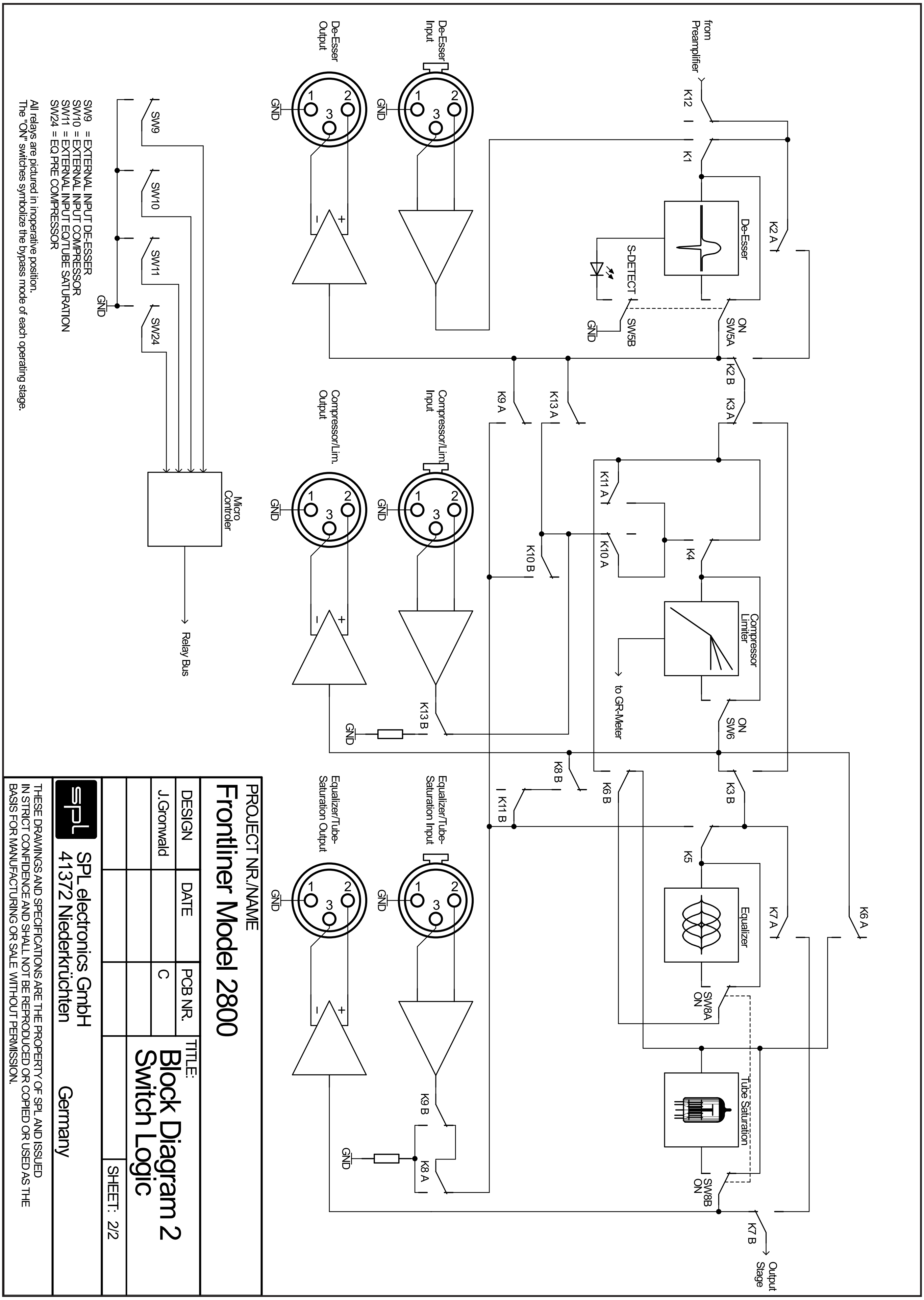
SPDIF Output
GND
Main Output 1
Main Output 2
A/D OVL
OVL LED switch internally
A/D Converter
A/D Converter Option
Transformer Option
Mute Relay
K14
A/D Input 2
Output
from tube saturation
from Compressor
VU/GR
VU
Switch Logic
to De-Esser
Tube Amplifier
Tube +15 dB
Low Cut
Phase
Instrument Line In
K15
+48V
48V Front disable (Rear Side)
48V Rear
48V Front
Mic Gain
PAD
Line Gain
Instrument Gain
+12V
Mic Input
Line Input
Instrument Input
Preamplifier Output


from Preamplifier
K12
K1
K2 A
De-Esser
S-DETECT
SW5B
ON SW5A
GND
De-Esser Input
De-Esser Output
SW9
SW10
SW11
SW24
SW9 = EXTERNAL INPUT DE-ESSER
SW10 = EXTERNAL INPUT COMPRESSOR
SW11 = EXTERNAL INPUT EQ/TUBE SATURATION
SW24 = EQ PRE COMPRESSOR
All relays are pictured in inoperative position.
The "ON" switches symbolize the bypass mode of each operating stage.
K2 B
K3 A
K9 A
K13 A
K11 A
K10 A
K10 B
K4
Compressor Limiter
to GR-Meter
ON SW6
Compressor/Lim. Input
Compressor/Lim. Output
Micro Controler
Relay Bus
K13 B
K8 B
K6 B
K3 B
K11 B
K5
Equalizer
K7 A
K6 A
SW8A ON
Tube Saturation
SW8B ON
K7 B
Output Stage
Equalizer/Tube-Saturation Input
Equalizer/Tube-Saturation Output
K9 B
K8 A
PROJECT NR./NAME
Frontliner Model 2800
DESIGN
DATE
PCB NR.
J.Gronwald
C
TITLE:
Block Diagram 2
Switch Logic
SHEET: 2/2
SPL electronics GmbH
41372 Niederkrüchten
Germany
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SW9 = EXTERNAL INPUT DE-ESSER
SW10 = EXTERNAL INPUT COMPRESSOR
SW11 = EXTERNAL INPUT EQ/TUBE SATURATION
SW24 = EQ PRE COMPRESSOR

Press each "External Input" button seperately and hold for 1 second.

SW9	SW10	SW11	SW24	K1	K2	K3	K4	K5	K6	K7	K8	K9	K10	K11	K12	K13
0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0
0	0	0	1	0	0	1	0	0	1	0	0	0	0	0	0	0
0	0	1	0	0	0	0	0	1	0	1	0	0	0	0	0	0
0	0	1	1	equals line above, EQpre compressor not available in this configuration												
0	1	0	0	0	0	1	1	0	0	0	0	0	0	0	0	0
0	1	0	1	equals line above, EQpre compressor not available in this configuration												
0	1	1	0	0	0	1	1	1	0	1	0	0	0	0	0	0
0	1	1	1	equals line above, EQpre compressor not available in this configuration												
1	0	0	0	1	1	0	0	0	0	0	0	0	0	0	0	0
1	0	0	1	1	1	1	0	0	1	0	0	0	0	0	0	0
1	0	1	0	1	1	0	0	1	0	1	0	0	0	0	0	0
1	0	1	1	equals line above, EQpre compressor not available in this configuration												
1	1	0	0	1	1	1	1	0	0	0	0	0	0	0	0	0
1	1	0	1	equals line above, EQpre compressor not available in this configuration												
1	1	1	0	1	1	1	1	1	0	1	0	0	0	0	0	0
1	1	1	1	equals line above, EQpre compressor not available in this configuration												

Press "External Input" buttons simultaneously and hold for 1 second.

SW9	SW10	SW11	SW24	K1	K2	K3	K4	K5	K6	K7	K8	K9	K10	K11	K12	K13
0	1	1	0	0	0	1	1	1	0	1	1	0	0	0	0	0
0	1	1	1	0	0	1	1	1	1	1	1	0	1	1	0	0
1	1	0	0	1	1	1	1	0	0	0	0	0	0	0	0	1
1	1	0	1	equals line above, EQpre compressor not available in this configuration												
1	0	1	0	1	1	0	0	1	0	1	0	1	0	0	0	0
1	0	1	1	equals line above, EQpre compressor not available in this configuration												
1	1	1	0	1	1	1	1	1	0	0	1	0	0	0	1	1
1	1	1	1	1	1	1	1	1	1	0	1	0	1	1	1	1

Press "External Input" buttons simultaneously and hold for 1 second. "X" = additional external activation of a single module.

SW9	SW10	SW11	SW24	K1	K2	K3	K4	K5	K6	K7	K8	K9	K10	K11	K12	K13
X	1	1	0	1	1	1	1	1	0	1	1	0	0	0	0	0
X	1	1	1	1	1	1	1	1	1	1	1	0	1	1	0	0
1	1	X	0	1	1	1	1	1	0	1	0	0	0	0	0	1
1	1	X	1	equals line above, EQpre compressor not available in this configuration												
1	X	1	0	1	1	1	1	1	0	1	0	1	0	0	0	0
1	X	1	1	equals line above, EQpre compressor not available in this configuration												

Create insert points by pressing "External Input" button shortly.

SW9	SW10	SW11	SW24	K1	K2	K3	K4	K5	K6	K7	K8	K9	K10	K11	K12	K13
0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0
0	0	0	1	0	0	1	0	0	1	0	0	0	0	0	0	0
0	0	1	0	0	0	0	0	1	0	0	0	0	0	0	0	0
0	0	1	1	equals line above, EQpre compressor not available in this configuration												
0	1	0	0	0	0	0	1	0	0	0	0	0	0	0	0	0
0	1	0	1	equals line above, EQpre compressor not available in this configuration												
0	1	1	0	0	0	0	1	1	0	0	0	0	0	0	0	0
0	1	1	1	equals line above, EQpre compressor not available in this configuration												
1	0	0	0	1	0	0	0	0	0	0	0	0	0	0	0	0
1	0	0	1	1	0	1	0	0	1	0	0	0	0	0	0	0
1	0	1	0	1	0	0	0	1	0	0	0	0	0	0	0	0
1	0	1	1	equals line above, EQpre compressor not available in this configuration												
1	1	0	0	1	0	0	1	0	0	0	0	0	0	0	0	0
1	1	0	1	equals line above, EQpre compressor not available in this configuration												
1	1	1	0	1	0	0	1	1	0	0	0	0	0	0	0	0
1	1	1	1	equals line above, EQpre compressor not available in this configuration												