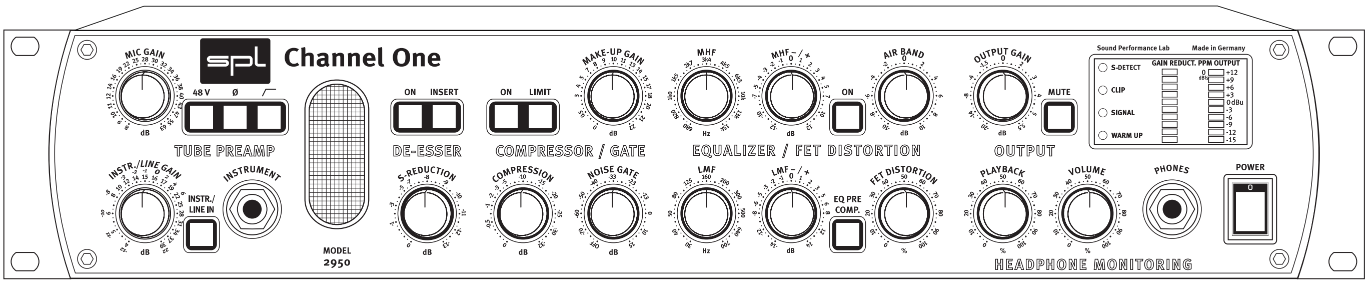
Manual



Channel One

Model 2950

Channel Strip

Version 1.1– 6/2009

SPL electronics GmbH

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The construction of the Channel One, Model 2950, 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

Symbols and Notes

N 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 Channel One, 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!

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 about the rear sockets and switches on pages 8, 9 an 10.

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

Many audio engineers know SPL's specialized audio tools, following our philosophy "one product for one task". This aims at fast and simple operation in conjunction with high processing quality to ensure highest musical performance.

With the Channel One we have produced a fully-featured channel strip which for the greater part is based on the processing concepts already known in other products, such as the SPL De-Esser and the DynaMaxx compressor. The very complex task of a channel strip profits particularly from the innovative techniques that have always allowed the operation of SPL equipment to be efficient and objective.

To a high degree the usual recording day is determined by a series of opposing time limits – the singer/speaker desires a trouble-free and efficient recording; however, if technical preparation takes a long time because of unsuitable equipment, time will be lost, increasing the costs and souring the working environment. The Channel One in all cases however allows fast production without any loss of professional precision and diligence.

The Channel One consists of a transistor/tube pre-amplifier with microphone-, line- and instrumental inputs, a de-esser, a compressor/limiter with noise gate, an equalizer (EQ) section and a latency-free headphone monitor.

Principles

So the Channel One has all tools on board for recording a track – along with the preamp it offers the most needed processors for corrective and creative sound design.

To maximize user friendliness all modules have been reduced to the most important regulating and switching facilities. Fast and effective operation is in no way impeded, quite the opposite – it is supported. And more time remains for the creative tasks.

From the outset great value was placed on high flexibility. An example are the three separate inputs for microphone, line signals or instruments, each of which has been optimized to its function.

A twin triode tube is utilized in the process at two positions – one immediately after the preamplifier stage and the other at the end of the chain, so that the processed signal passes the tube stage twice. This construction combines the advantages of the transistor preamplifier stage (high performance with minimal distortion and low noise) with the improved musical expression of the tone produced by tubes.

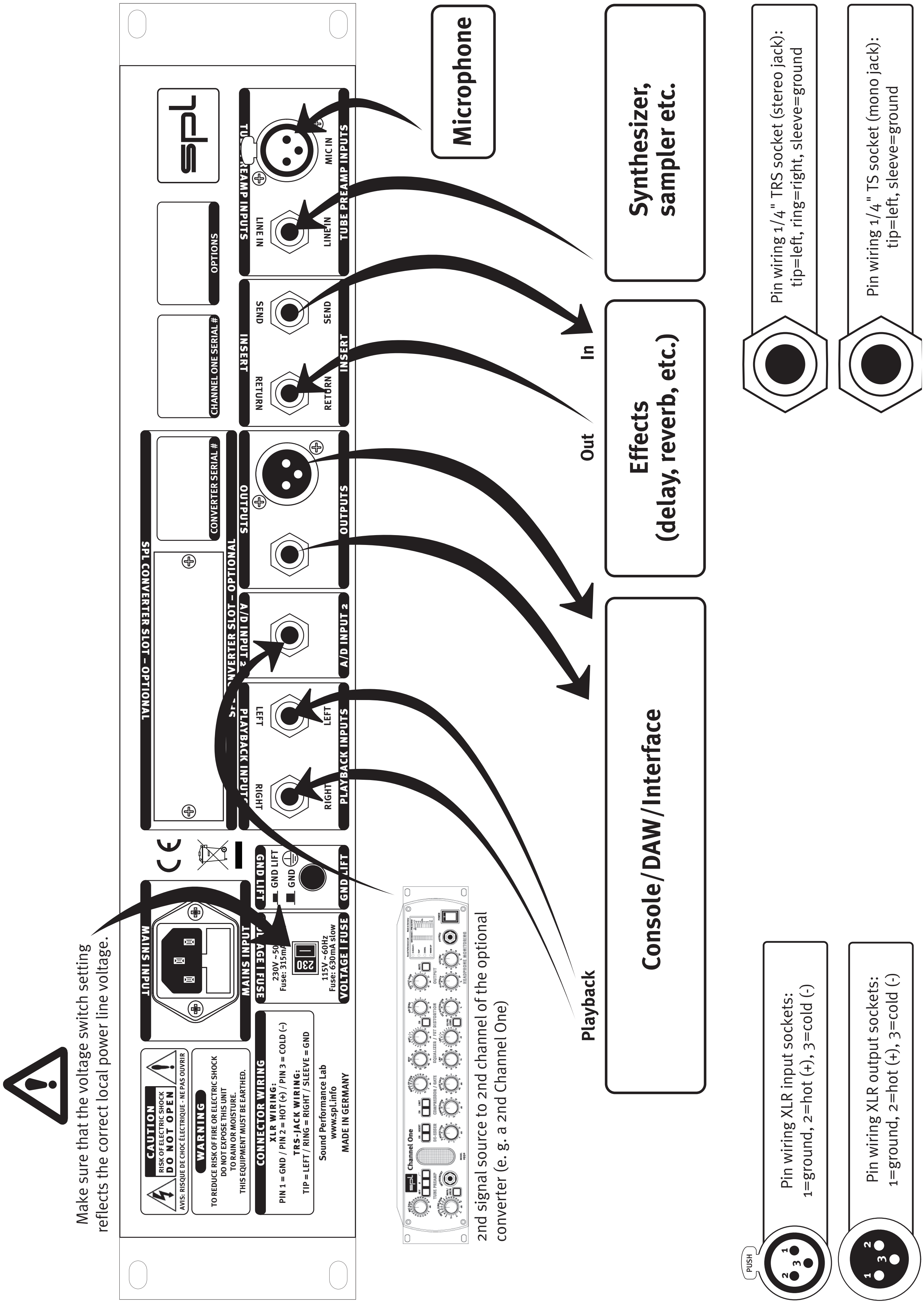
The microphone input can optionally be equipped with an input transformer from Lundahl. The input transformer delivers a fivefold amplified microphone level to the preamplifier. This additional amplification reduces the equivalent load to the preamplifier electronics. The balanced outputs can also be equipped with a Lundahl transformer.

The optional 24 bit/96 kHz AD converter module provides digital outputs. An additional input socket on the Channel One may feed a second signal to the AD converter.

Displays and LEDs for signal level, gain reduction, s-detection, clip warning, warm-up status and signal presence are combined and contained in a single display area to be monitored at a glance.

A special feature of the PCB layout is the central star ground wiring: Disturbing influences that could affect the ground paths are minimized by separating audio-ground from the remaining equipment. This leads, in the truest sense of the word "clean", to considerably improved tonal quality.

The scatter free toroidal transformer supplies the equipment with the necessary voltages and forms the basis for a clean electrical supply to all parts of the circuitry.


Make sure that the voltage switch setting reflects the correct local power line voltage.
2nd signal source to 2nd channel of the optional converter (e. g. a 2nd Channel One)
Microphone
Synthesizer, sampler etc.
Effects (delay, reverb, etc.)
In
Out
Console/DAW/Interface
Playback
Pin wiring 1/4" TRS socket (stereo jack): tip=left, ring=right, sleeve=ground
Pin wiring 1/4" TS socket (mono jack): tip=left, sleeve=ground
Pin wiring XLR input sockets: 1=ground, 2=hot (+), 3=cold (-)
Pin wiring XLR output sockets: 1=ground, 2=hot (+), 3=cold (-)
TUBE PREAMP INPUTS
MIC IN
LINE IN
INSERT
SEND
RETURN
OUTPUTS
A/D INPUT 2
PLAYBACK INPUTS
LEFT
RIGHT
GND LIFT
VOLTAGE I FUSE
MAINS INPUT
OPTIONS
CHANNEL ONE SERIAL #
CONVERTER SERIAL #
SPL CONVERTER SLOT – OPTIONAL
230V ~50Hz
115V ~60Hz Fuse: 630mA slow
CAUTION
RISK OF ELECTRIC SHOCK DO NOT OPEN
AVIS: RISQUE DE CHOC ÉLECTRIQUE - NE PAS OUVRIR
WARNING
TO REDUCE RISK OF FIRE OR ELECTRIC SHOCK DO NOT EXPOSE THIS UNIT TO RAIN OR MOISTURE. THIS EQUIPMENT MUST BE EARTHED.
CONNECTOR WIRING
XLR WIRING:
PIN 1 = GND / PIN 2 = HOT (+) / PIN 3 = COLD (–)
TRS-JACK WIRING:
TIP = LEFT / RING = RIGHT / SLEEVE = GND
Sound Performance Lab
www.spl.info
MADE IN GERMANY
PUSH

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 sockets

The TRS sockets LINE IN, INSERT SEND/RETURN, OUTPUTS und AD INPUT 2 support both balanced (1/4" TRS/stereo jack connector) and unbalanced connections (1/4" TS/mono jack connector). The PLAYBACK INPUTS sockets only support unbalanced connections.

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.

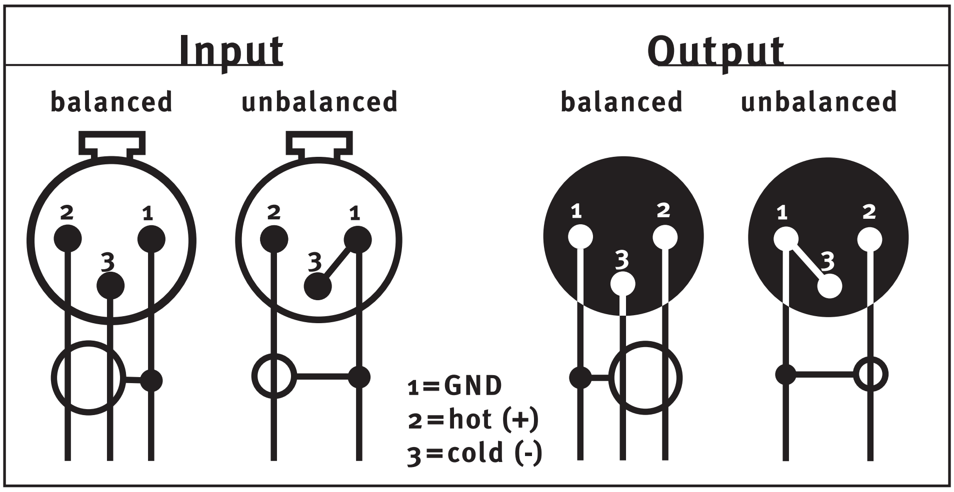
Balanced connections

It is impossible to exclude all interferences when an audio signal is transmitted through a single wire. 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.

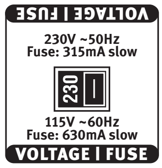
Unbalanced connections

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.

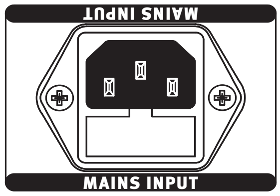
VOLTAGE



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 to the right 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!

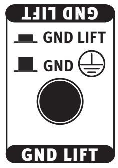
MAINS INPUT – power connection and fuses



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

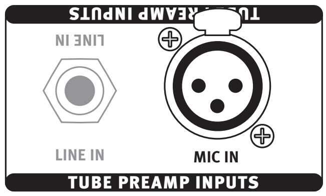
The MAINS INPUT socket also houses the fuse. It is accessible from outside and placed right behind the flap below the socket. Fuse ratings are 315 mA slow blow (230 volts) or 630 mA slow blow (115 volts).

GND Lift



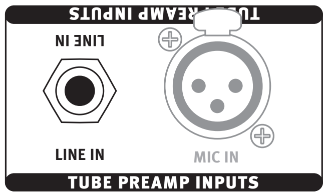
The rear panel GND LIFT switch eliminates hum by separating the internal ground from the unit's housing ground. Hum can, for example, result when this unit's housing has a common ground connection with other devices that might have a different ground potential. The switch is usually deactivated to retain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

MIC 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 switch on the front panel. Please read the important notes in chapters "48 V" and "Activating phantom power" on page 11. The microphone input can also be equipped with an optional input transformer (see page 29, "Information on I/O transformers").

LINE IN



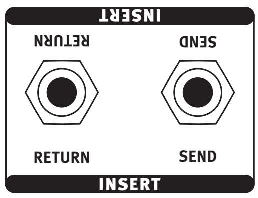
Use the balanced LINE IN socket for high-level signals with impedances lower than 1 kOhm, e. g. D/A converters, synthesizers or samplers. We recommend connection to a patchbay for easier access.

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

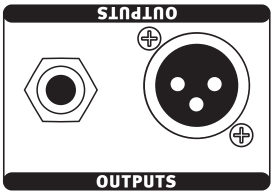
High impedance sources (above 1 kOhm), such as e-guitars and basses, acoustic guitars with pick-ups and so on, must be connected to the INSTRUMENT input.

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

INSERT



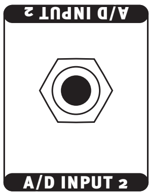
The balanced INSERT connectors (SEND and RETURN) are used to integrate further units into the signal path of the Channel One. The SEND connector is placed behind the de-esser, the RETURN connector is located in front of the compressor. This also allows to record the pre-amplifier signal via the SEND connector while another input signal can be fed into the Channel One's compressor or EQ sections for further processing.



OUTPUTS

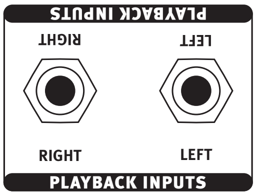
The OUTPUTS deliver balanced output signals. An output transformer can be equipped optionally (see page 29).

Since both connectors are working in parallel, unbalancing one connector also unbalances the other one. If for example a mono jack connector is plugged into to the jack socket, the XLR socket is operating unbalanced as well. Depending on the impedances of the connected devices, a parallel use of both outputs can reduce the signal level. Therefore, we recommend to use either the XLR or the 1/4" TRS output socket.



A/D INPUT 2

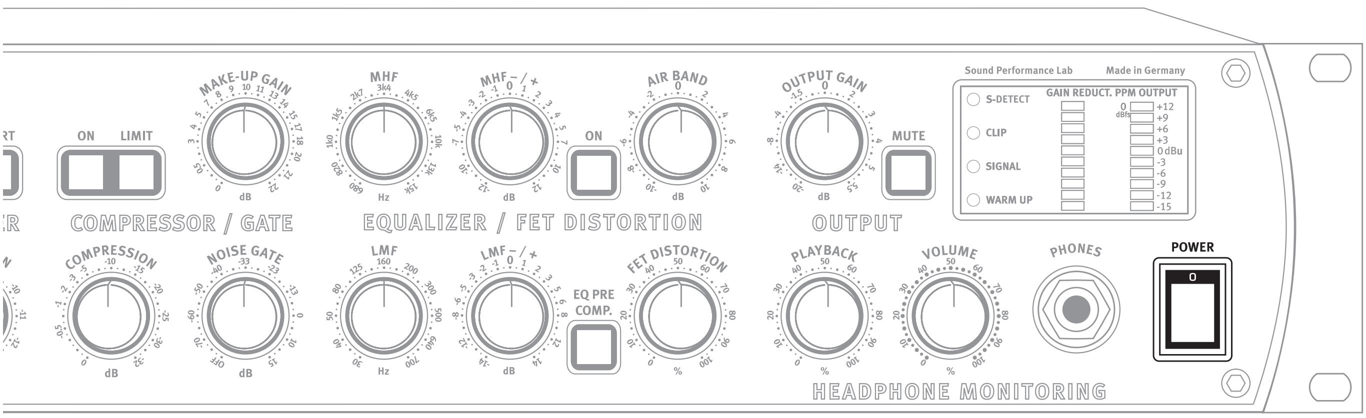
The Channel One is a mono channel strip, but the optional A/D converter card 2376 is a dual-channel device. Therefore a second (external) signal can be converted with the converter card, if it is connected to the AD INPUT 2. If no signal is connected to the A/D INPUT 2, the output signal of the Channel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



PLAYBACK INPUTS

The playback signal is connected to the unbalanced PLAYBACK INPUTS to create a headphone monitoring mix. If a mono playback signal is available, only the LEFT connector must be connected. The signal will then be present on both channels. The RIGHT connector should be used if only one channel should appear on one side of the headphones. In contrast to all other connectors the PLAYBACK INPUTS are unbalanced.

Front Panel *POWER switch*

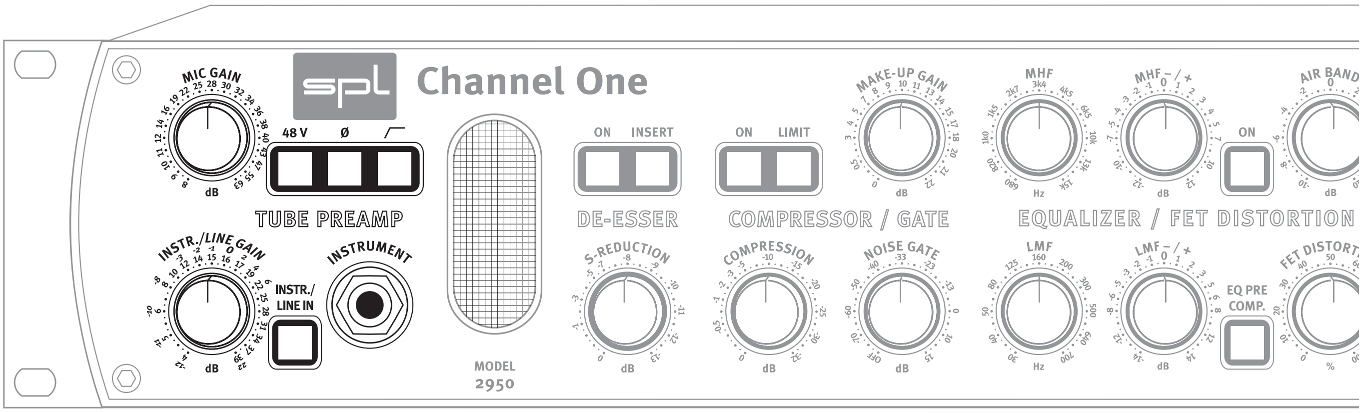


POWER

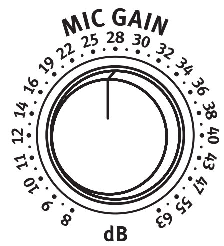
The front panel POWER switch activates the unit, confirmed by the illuminated switch.

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

When you activate the Channel One, the unit commences the warm-up mode to heat the tubes. The warm-up cycle takes between 15 and 30 seconds. The WARM UP LED in the display area illuminates during warm-up mode and the Channel One is ready to operate when the WARM UP LED turns off.



MIC GAIN



The MIC GAIN control determines the level of preamplification. The preamplification values cover a range from +8 dB up to + 68 dB. Further information on page 12, „GAIN adjustments“. If the optional microphone input transformer is installed, the scaled values are to be increased by ca. +14 dB (depends upon microphone, see page 29, “Information on I/O transformers”).

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.

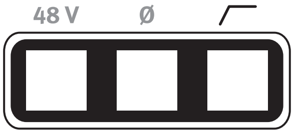
Activating phantom power

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

1. Connect the microphone to the Channel One.
2. Now activate phantom power to use the microphone.
3. After recording 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.

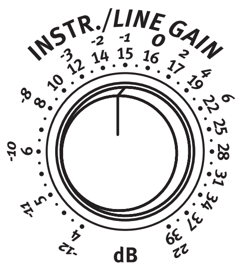
Phase reverse

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 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 recording.



High-pass filter

This switch activates the high-pass filter (often also called a “rumble filter”), which operates from 50 Hz downwards with 12 dB/octave. The filter prevents the amplification of unwanted low frequencies. Compared to 6 dB/octave filters, the 12 dB filter works more intensively, thus more effectively – therefore the threshold is set to a low 50 Hz.



INSTR./LINE 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 -12 dB to +22 dB. The attenuation allows to also process very high levels. The 0 dB mark is highlighted in the line gain scale – this facilitates to find the setting where a line level signal is processed at unity gain.

Instrument signals can be amplified between +4 dB and +39 dB.

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.

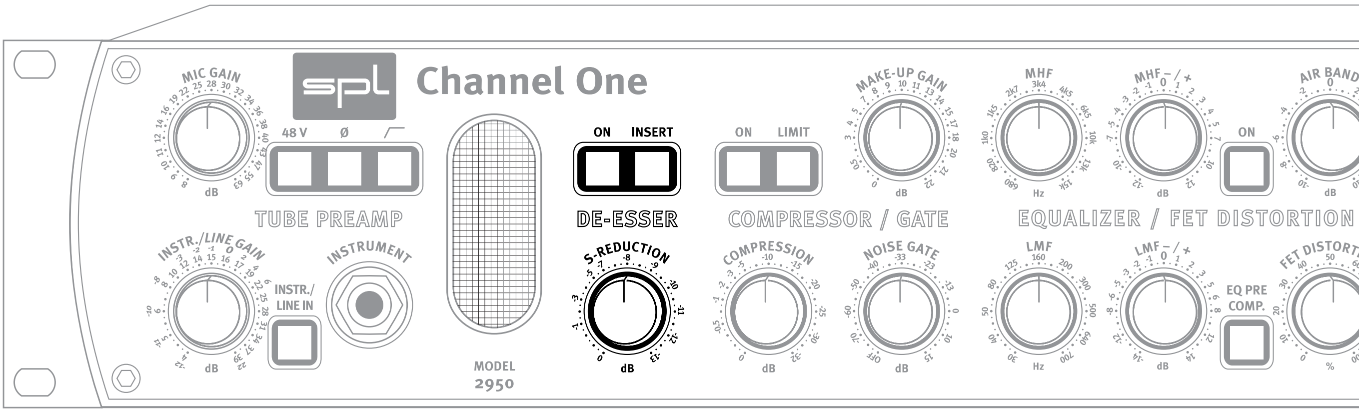
INSTRUMENT

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.

Gain adjustments

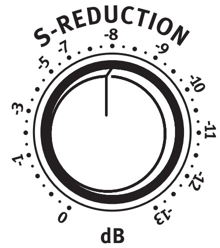
For perfect levelling of the preamplifier firstly switch off all other modules (de-esser, compressor, EQ) and set the OUTPUT control to 0 dB. The signal can now be levelled with the assistance of the PPM OUTPUT display. To achieve a good working level the values should range between 0 and +3 dB. At these levels an optimal drive level and enough headroom for further processing (e. g. adding level in the EQ stage) is guaranteed. The Clip LED will warn you of potential peaks; if during recording the CLIP LED illuminates, the gain value is to be reduced accordingly.



ON

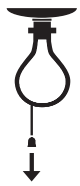
The first processing module is the de-esser, which removes disturbing sibilants when required. The de-esser module is activated with the ON button. The S-DETECT LED in the display area shows that S-sounds are being detected. It is independent from the S-REDUCTION control and always informs about detected sibilants – attracting your attention to a possible need for regulation (also see „S-DETECT“ on page 22).

S-REDUCTION



With the S-Reduction control you can determine the intensity of S-sound reduction. Because processing is undertaken from comparison with the level of the entire frequency spectrum (see next section) the processing is more intensive with extreme S-sound levels than with those of lower levels. This processing method achieves a consistent level of the remaining sibilants in the output signal.

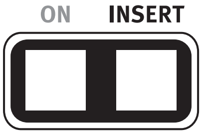
SPL De-Esser technology



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.



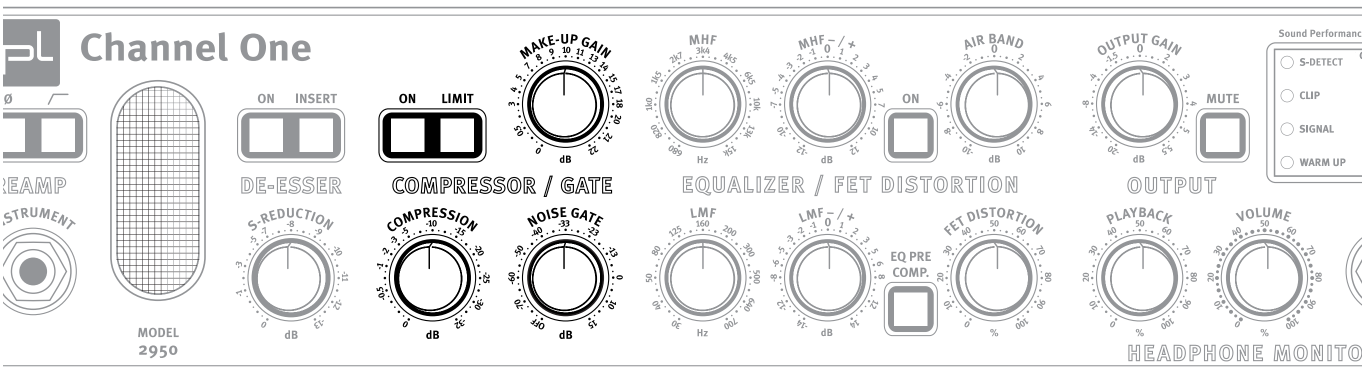
INSERT

The insert button activates any attachments of external equipment for effects, such as delay or reverb, that are connected to the INSERT SEND/RETURN loop on the rear of the Channel One. This binds them into the signal chain, thereby enhancing the processing capabilities ad infinitum.

The INSERT point is located between the de-esser and compressor. This allows to use the pre-amplifier stage/de-esser combination of the Channel One separately from the compressor/EQ combination. This broadens the range of uses enormously, because in this manner the Channel One can be used as two independent units.

As long as units are not connected to the INSERT loop, the signal flow is not interrupted, even when the INSERT button is pressed. The most flexible method of use with the balanced designed insert sockets is to be achieved by connection to a patch bay.

Control Elements *Compressor*



ON

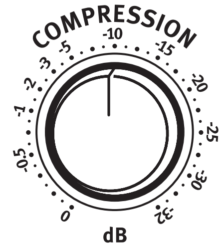
The ON button activates the compressor/limiter/noise gate module. At the same time the GAIN REDUCT. display shows the processing intensity (see “GAIN REDUCT.” on page 23).

Usually the signal flow follows the design of the Channel One and for this reason the input signal normally arises from the de-esser or, when activated, from the INSERT. However, with the EQ PRE COMP. switch the EQ module can be switched in front of the compressor module. This allows it to be used either as a final compressor or limiter (further information in the section “EQ PRE COMP.” on page 17).

LIMIT

The LIMIT switch turns the compressor into a limiter. The COMPRESSION control serves the purpose of controlling the threshold. The Limiter does not function as a peak limiter, in other words there is no guarantee that all peaks are included. It is therefore advisable when modulating a subsequent unit that a headroom of 2 to 4 dB remains. Now peaks can be intercepted very well and unobtrusively. Peak limiters have a system-based disadvantage in producing audible distortions considerably sooner, so with regard to both sound quality and recording safety, we think the soft limiter mode is the better choice for a recording channel strip.

COMPRESSION



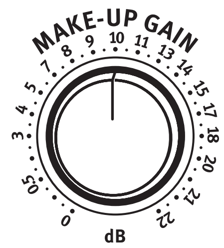
The COMPRESSION control sets the intensity of compression. Turning the control clockwise increases compression. The working area spans between 0 dB (fully left) and -32 dB (fully right).

The compressor applies the so-called “soft-knee” characteristic, which means it starts processing earlier than with hard-knee curve (see page 26, diagram 1, curve B). Hard-knee compressors can sometimes gain more loudness, but they process abruptly and the danger to ruin a recording with compression artifacts is much higher. On the other hand the soft-knee compressor always helps very well to keep levels under control and ensures highest recording safety – and if there is a desire to gain further loudness, the signal can still be processed after recording.

At maximal compression it operates with a ratio of 1:2.5 between input and output signal – very effective dynamic processings are achievable with unobtrusive sound characteristics.

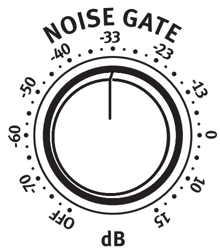
The exact development of the compressor curve is portrayed in the diagram 1 on page 26. When setting the COMPRESSION rate the GAIN REDUCT. display in the display area is of great assistance. The effect on the selected COMPRESSION rate is scaled in 1.5 dB steps. Depending on signal source and dynamic structure the reduction values should lie between 4 and 8 dB to restrict higher peaks and to optimize the operation of the subsequent recording system.

MAKE UP GAIN



With the MAKE UP GAIN control you can restore the overall level reduction caused by compression or limiting. With assistance of the GAIN REDUCT. display setting the MAKE UP GAIN control is very easy: If the maximal reduction value caused by the loudest tone amounts to -9 dB, for instance, the MAKE UP GAIN control is also to be set to the value +9 dB. If the compressor/limiter is now switched off the achieved gain in loudness will be audible.

NOISE GATE



The NOISE GATE control reduces soft disturbances during signal pauses. When turned fully counter clockwise the noise gate is switched off. By turning the control in a clockwise direction the threshold value increases. This means that the noise gate closes relatively earlier.

The processing span of the NOISE GATE control is between -100 dB/OFF (gate control turned fully counter clockwise) and + 15 dB (gate control turned fully clockwise). The noise gate is therefore operable over the complete dynamic range.

The noise gate functions very stably: the point at which it opens lies 6 dB above the point at which it closes again (hysteresis of 6 dB). Definite closure and opening is therefore assured – the most feared characteristic of “fluttering” is excluded. Even critical signals are cleanly processed.

The noise gate’s release time is set automatically. The automation, which depends upon the program, adjusts itself to the musical piece, thereby ensuring optimal (undetectable) opening and closing.

SPL compressor technology

In the compressor/limiter section of the Channel One the parameters for the time constants (attack and release) are set automatically and adapt themselves to the changing conditions of the input signal, far better than can ever be achieved by manual adjustments. The transient and final oscillation behavior of voices and instruments are constantly changing and at times are so erratic that a manual control will only achieve good average values, which at critical moments can produce disadvantageous effects (e. g. distorted sounds, “pumping”, etc).

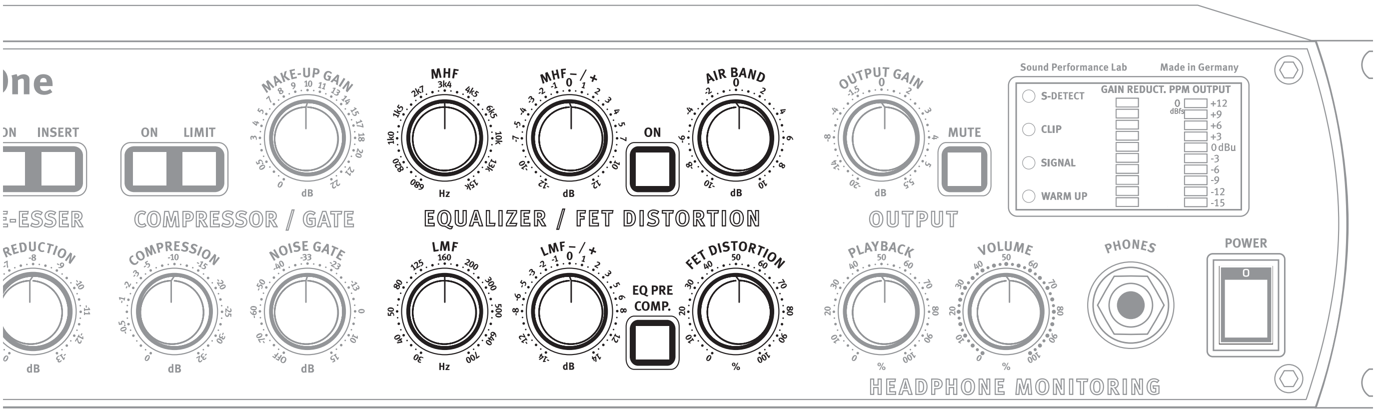
If for example the compressor has to react very quickly to harsh P or T noises it must also be capable of reacting slowly to softer tones – otherwise distortion occurs. Accordingly the Channel One compressor regulates the level of large fluctuations faster than smaller ones; tones of longer duration are automatically processed with a longer attack time to prevent distortions.

The automatic setting of the release times is dependent on the input signal, too. Fast and large level fluctuations are correspondingly processed with shorter time constants than minor fluctuations in order to limit the distortion of the audio signal as far as possible. Overall this technique provides the optimal solution between fast, unobtrusive control response and the least distortion of the audio signal. The result is a natural and transparent sound impression.

A further technical specialty of the circuitry contributes to the high audio quality of the compressor in the Channel One: SPL’s double VCA drive. One VCA receives the in-phase, the other the out-of-phase signal. Subsequently the signal is passed through a differential amplifier. The effect of this circuitry is that distortion products and offset fluctuations can be removed – the product of the differential of both signals means that possible interference is canceled out. The original information is however further amplified by 6 dB. In addition the VCAs provide relief to each other because they share their loads. They do not even run the danger of operating in the saturation range, which ensures to avoid offset noises, audible as clicks or pops.

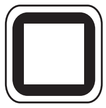
SPL’s double VCA drive circuitry overall displays vastly improved distortion values so that a distinctly clearer and more transparent sound impression is achieved than with conventional circuitry. Voices and instruments are given a considerably more natural and dynamic timbre whereas ”muffled” tones are not audible.

The compressor characteristics are portrayed on page 26.



ON

ON

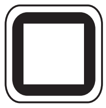


The ON button inserts the equalizer/FET distortion module into the signal path. Under normal circumstances the input signal comes from the compressor. With the EQ PRE COMP. button the equalizer can be switched in before the compressor so that the input signal is received from the de-esser or insert.

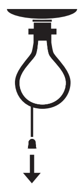
IMPORTANT: To avoid irritations at the beginning of a recording it is recommended to deactivate the equalizer/FET distortion module. If not, tonal changes could occur immediately and furthermore, in the case of the DISTORTION control, additional distortions.

EQ PRE COMP.

EQ PRE COMP.



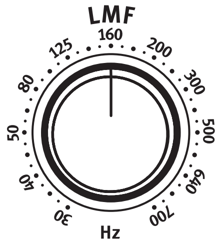
The EQ PRE COMP. switch reverses the sequence of compressor/limiter and equalizer (EQ). When the switch is pressed the equalizer operates in front of the compressor/limiter; when not pressed the succession remains unchanged. This function permits very flexible operation with the Channel One when it is necessary to resolve recurring problems or to create special sounds.



The following examples describe when the equalizer is to be switched in front of the compressor/limiter.

When over-accentuation of instruments or voices is registered within certain frequency ranges these ranges can first be reduced with the EQ. The signal can subsequently be compressed more easily.

A further sensible application is the use of the compressor module as a final 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.

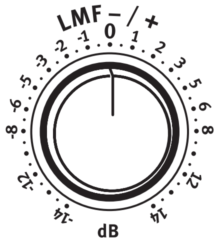


LMF

The center frequency of the half-parametric bass filter is set with the LMF control (low/mid frequencies).

The adjustable frequency range lies between 30 Hz 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 MHF filter ensures that the entire frequency spectrum is covered.

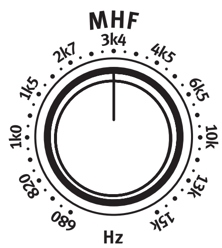


LMF -/+

The LMF-/+ control determines the boost or cut of the LMF filter; the maximum values lie between +/- 14 dB. The LMF filter also operates to the proportional-Q-principle, in other words the bandwidth is dependent on the selected boost or cut. 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 far preciser definition of the frequency range to be processed. This in turn minimizes influences from adjacent ranges.

The boost or cut values, in relation to the bandwidth, lie somewhat higher than with the MHF filter. The bandwidth is therefore narrower at maximum boost than with the MHF filter for even more precise filtering. The exact curve of the LMF filter is shown in diagram 4 on page 27.

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 bass emphasized instruments during recording or subsequently when mixing etc.

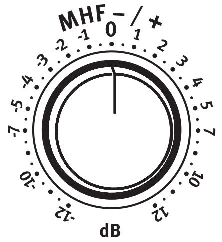


MHF

The center frequency of the semi-parametric mid/high frequency filter is set with the MHF control.

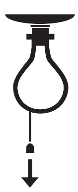
The frequency range can be set between 680 Hz and 15 kHz so that this filter covers a range of 4.5 octaves and can be equally employed in the lower mid as well as the high range.

This together with the LMF filter ensures that the entire frequency spectrum is covered.



MHF -/+

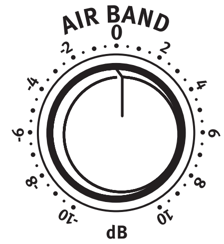
This control determines the boost, or cut of the MHF filter; the maximum values lie between +/- 12 dB. The MHF filter utilizes the proportional-Q-principle, too: the higher the boost or cut values are set, so the bandwidth becomes narrower; by low boost or cut values the bandwidth increases (the exact curves of the MHF filter can be seen in diagram 3 on page 27). The filter construction permits the complete scope, from selective removal of accentuated frequencies through to character giving accentuations of an instrument, to be effectively and quickly covered.



Recommendation on frequency settings for LMF and MHF

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.

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 property, 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.

The characteristics of the AIR BAND filter are shown in diagram 2 on page 26.

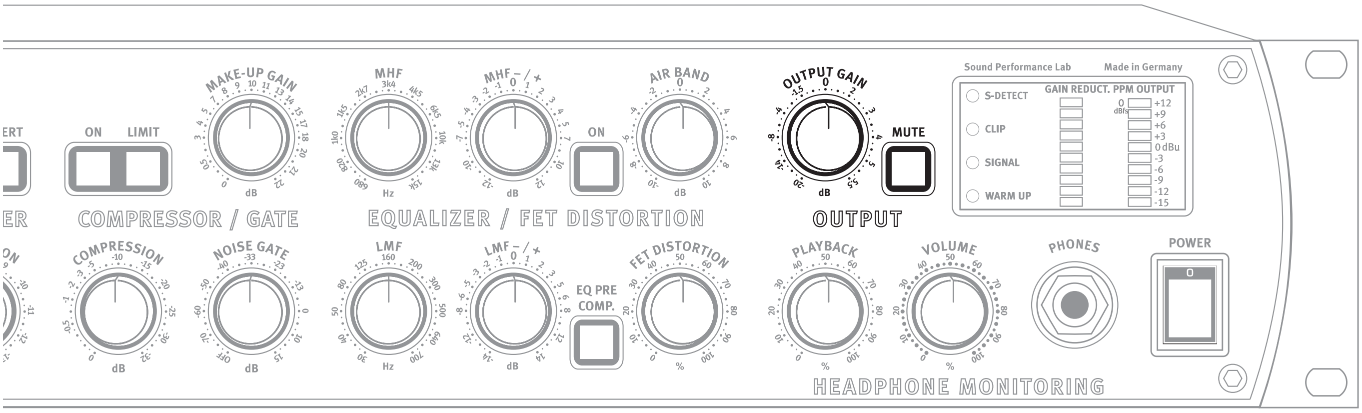
FET DISTORTION

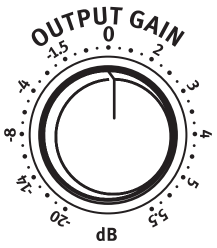
The DISTORTION control offers the capability of applying distortions to signals. The distortions are infinitely variable from 0% through to distinctly perceptible harmonics. The distortion stage is located in front of the equalizer so that the newly created spectrums can also be processed with the EQ.

A (more or less over-driven) field-effect transistor forms a part of the distortion circuitry. It's characteristic curve is similar to a tube and sounds distinctly "warmer" than a pure diode-distortioner.

The signal level is of utmost importance to the operating mode of the DISTORTION module. To achieve useful results the level should lie in the range 0 to + 6 dB. Over and above this the results are strongly dependent on the condition of the input signal and its spectrum. The processing of sinewave-like signals (e. g. e-piano, vocal, guitar) is audible much earlier than signals with predominant harmonical contents (e. g. snare drum, hi hat etc). It is recommended that time and effort is taken to find the correct setting.

IMPORTANT: To avoid irritations at the beginning of a recording it is recommended to deactivate the EQ, and in particular set the DISTORTION control to 0%. If not, tonal changes will occur immediately and furthermore, in the case of the DISTORTION control, additional distortions.

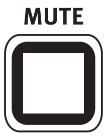




OUTPUT GAIN

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 AD converter. The selected output level is shown on the PPM OUTPUT display in the display field.

Before a recording commences the OUTPUT control should be set to 0 dB (12 o'clock position): the uninfluenced values are then legible and available for adjustment of the preamplifiers levels.

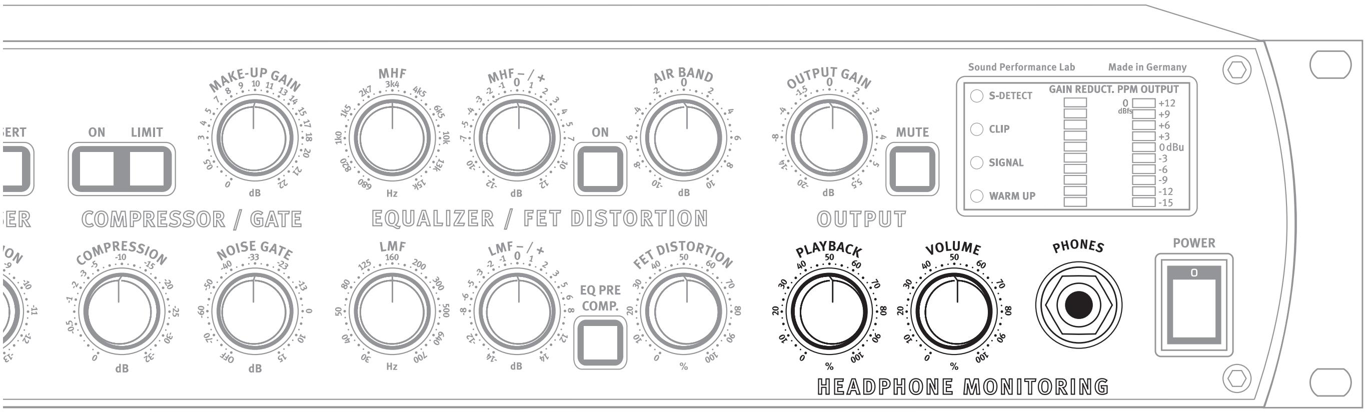


MUTE

The Mute switch mutes the output signal; when activated, the PPM OUTPUT display does not show any values.

An instance of a sensible application could possibly be when the output signal of the Channel One, together with the playback signal, are reproduced via the studio monitors during a recording session. When subsequently the recorded take is monitored it becomes possible to hear extraneous singing or comments arising from the singer. It is therefore advised to press the MUTE switch to permit listening to a clean recording. Do not forget to deactivate the Mute switch before continuing recording.

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



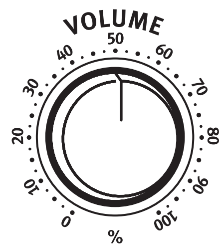
An individual monitoring mix is important for best working conditions and especially a good intonation. That is why the Channel One is equipped with a headphone monitor section, providing a headphone amplifier and a mixing stage to generate an individual mix for the musician with playback and recording signals.

In general, the headphone monitor section can of course always serve for direct monitoring of the recording via headphones. Another practical use of the headphone monitor module is to monitor the signal quality directly to locate and eliminate possible interference rapidly.

PLAYBACK

The PLAYBACK control regulates the volume of the playback signal which is passed to the musician. There are two methods of passing the mono playback signal: The first is to pass the music to both ear pieces of the headphone in which case PLAYBACK INPUT LEFT must be connected. On the other hand some musicians want to hear the playback signal through only one ear piece so they can hear themselves directly with the other ear (playback signal only without microphone signal). In this instance connect PLAYBACK INPUT RIGHT and set the VOLUME control to off (also see page 10, PLAYBACK INPUTS).

VOLUME



The VOLUME control regulates the volume adjustment of the microphone, instrument or line signal. The setting is independent to that of the OUTPUT control or MUTE switch, which means the volume in the headphones does not alter although the output value of a modulation has changed.

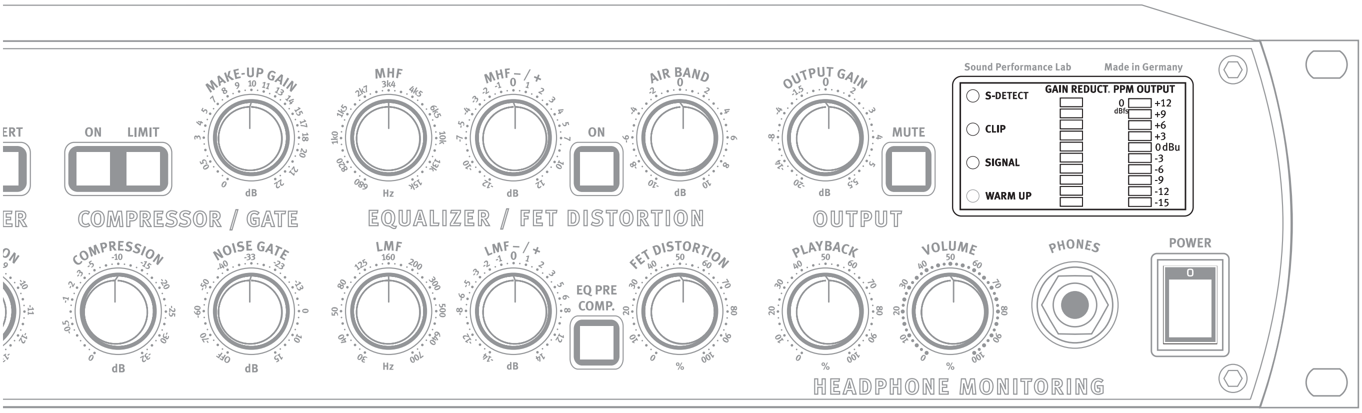
TIP: When working with hard disc systems or digital mixing consoles latency may be present. Flanging or phasing effects occur if the musician receives the monitor signal with a time lag. It is therefore recommended, to obviate latency, that the monitor signal passes directly from the headphone monitor to the headphones. It should be remembered that the recording signal has not been picked up again by the playback signal because phase quenching can occur when the same signal is mixed by both the PLAYBACK and VOLUME controls.

PHONES

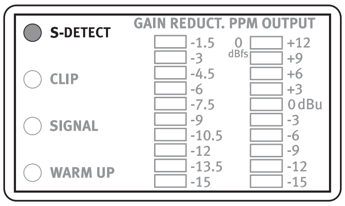
Connect headphones to the standard 1/4" (TRS) stereo jack plug on the front panel. The low-resistance input allows for connection of all usual headphones.

The layout is: Tip =left channel, Ring = right channel, Sleeve = ground.

Make sure that the plug is firmly seated for a solid connection. Reduce volume level before you remove or plug in a headphone (or when switching headphones). **NEVER plug in a mono 1/4" jack (TS) to the headphone output. The use of a mono 1/4" will lead to a short-circuit that will destroy the amplifier stage. Standard headphone connectors always have stereo plugs, and thus a correct connection will be assured when you only connect headphones directly. Double check that you use stereo 1/4" TRS plugs when you connect headphones via patchbays or extension cables etc.**

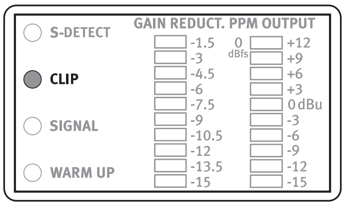


All status and level displays are concentrated in a central display area, so all important information can be perceived at a glance.



S-DETECT

The S-DETECT LED shows when sibilants have been detected. It is only active when the de-esser is switched on, but it is independent from the S-REDUCTION control. So if you turn on the De-Esser, you are always informed about detected sibilants and a possible need for regulation.

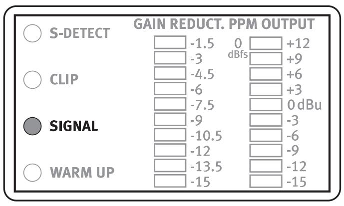


CLIP

The CLIP LED shows overload in the unit. The clipping level of the LED lies approximately 2 dB below the internal full scale (conforms to + 19 dBu). The CLIP LED should flash as seldom as possible.

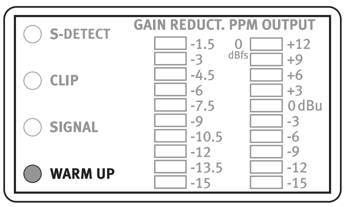
At all relevant points of the signal flow the display gets read off: behind the preamplifier, behind the compressor/limiter, behind the EQ and behind the output control. All possible causes for overload can be directly checked (overdriven microphone/instrument/line gain, an excessive make up value in the compressor/limiter, too much boost in the EQs or too high output level).

Possible causes of overload can be quickly detected by simply switching off the modules individually. If overloads occur during recording the quickest remedy is to gradually reduce the respective gain control in the preamplifier.



SIGNAL

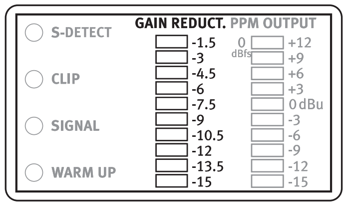
The SIGNAL LED illuminates when a signal is being received at the preamplifier. This provides a quick method of checking that a signal source is correctly connected. All levels above -50 dB are covered.



WARM UP

The WARM UP LED gives an indication regarding the warm up phase of the tube stage. When the LED is extinguished the Channel One is ready for operation; during warm-up the output signal level is low and sounds distorted.

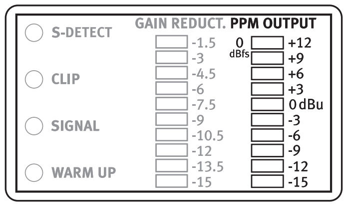
GAIN REDUCT.



The GAIN REDUCT. display provides information about the processing being undertaken with the compressor/limiter or the noise gate. The level changes, perhaps caused by compression, are scaled in 1.5 dB steps. The display is activated when the compressor/limiter module is switched on.

Noise gate operation is visualized by illumination of all GAIN REDUCT. LEDs when the signal level lies under the gate threshold setting.

PPM OUTPUT



The PPM OUTPUT display shows the peak reading of the output level (calibrated to 0 dB) and is present at the rear outputs. This display also serves to level the preamplifier. The value “0dBFS” marked on the left side represents the maximum level of the optional AD converter which should not be exceeded (further information is given in the manual of the AD converter and on page 29).

Although the values of the PPM OUTPUT display only cover up to + 12 dB sufficient headroom remains internally (approximately 6 dB) so that the output value can exceed this limit without causing clipping. The range of optimal noise performance lies between 0 and + 9 dB.

Power Supply

Built around a toroidal transformer, the power supply ensures a minimal electromagnetic field with no hum or mechanical noise. The power supply‘s output side is filtered by an RC circuit to extract noise and hums caused by your power service. 6000µf capacitors smooth out the positive and negative half waves.

The phantom power is derived from a separate winding in the transformer, a precise current regulator a clean phantom power of 48 volts. Our high quality 0.1% / 6,81 kOhm resistors ensure the pristine quality of the phantom power supply.

The 250 Volt power supply for the tube stage is filtered with 300 µF to minimize hum.

Further information on page 9.

Microphone input

Frequency range: (100 kHz = -3 dB)	10 Hz-100 kHz	
Common mode rejection: (@ -20 dBu)	1 kHz: -80 dB/10 kHz: -78 dB	
THD & N:	Gain:	A-weighted:
	20 dB	-97,1 dBu
	40 dB	-91,1 dBu
	65 dB	-69,4 dBu
Dynamic range:	118 dB	

Line/instrument input

Frequency range: (-3 dB)	10 Hz-100 kHz	
Common mode rejection: (@ 0 dBu, LINE IN only)	1 kHz: -80 dB/10 kHz: -78 dB	
THD & N:	Gain:	A-weighted:
	5 dB	-99,4 dBu
	20 dB	-97,2 dBu
	42 dB	-79,4 dBu
Input impedance:	Line: 20 kOhm / Instrument: 1 MOhm	
Maximum input level:	Line: +22 dBu / Instrument: +14 dBu	
Dynamic range:	119 dB	

Output

Maximum output level XLR/TRS:	+20 dBu
Output impedance:	‹50 Ohm

Dimensions & weight

Standard-EIA-19"/2U housing	482 x 88 x 210 mm
Weight	4,15 kg/ca. 9,15 lbs

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

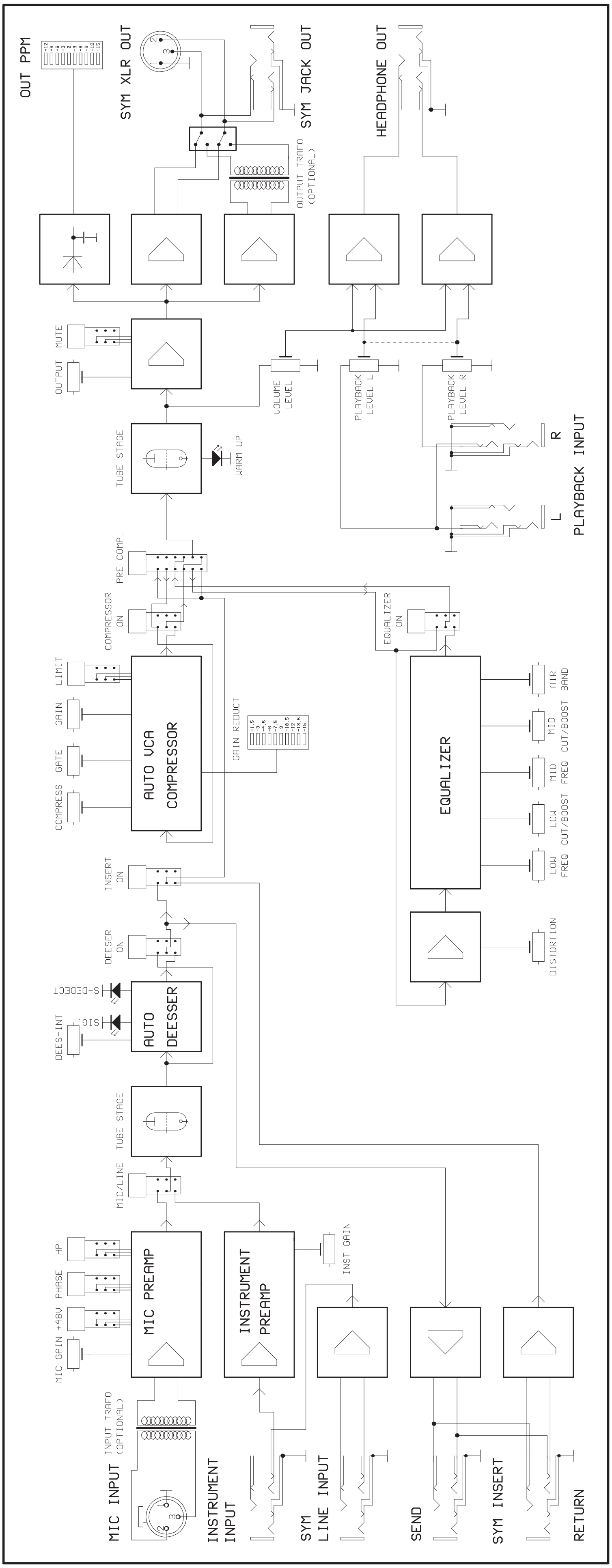

OUT PPM
SYM XLR OUT
SYM JACK OUT
HEADPHONE OUT
OUTPUT TRAFO (OPTIONAL)
MUTE
OUTPUT
VOLUME LEVEL
PLAYBACK LEVEL L
PLAYBACK LEVEL R
TUBE STAGE
WARM UP
R
L
PLAYBACK INPUT
PRE COMP.
COMPRESSOR ON
EQUALIZER ON
LIMIT
GAIN
GATE
COMPRESS
AUTO VCA COMPRESSOR
GAIN REDUCT
EQUALIZER
AIR BAND
MID CUT/BOOST
MID FREQ
LOW CUT/BOOST
LOW FREQ
INSERT ON
DEESER ON
S-DEDECT
SIG
DEES-INT
AUTO DEESSER
DISTORTION
TUBE STAGE
MIC/LINE
HP
PHASE
+48V
MIC GAIN
MIC PREAMP
INSTRUMENT PREAMP
INST GAIN
INPUT TRAFO (OPTIONAL)
MIC INPUT
INSTRUMENT INPUT
SYM LINE INPUT
SEND
SYM INSERT
RETURN

Diagram 1:
compressor characteristics

Reference curve A displays the relation between input and output.

Curve B shows the curve characteristics of the compressor. The soft knee characteristic is clearly visible.

Curve C portrays the limiter's curve characteristics.

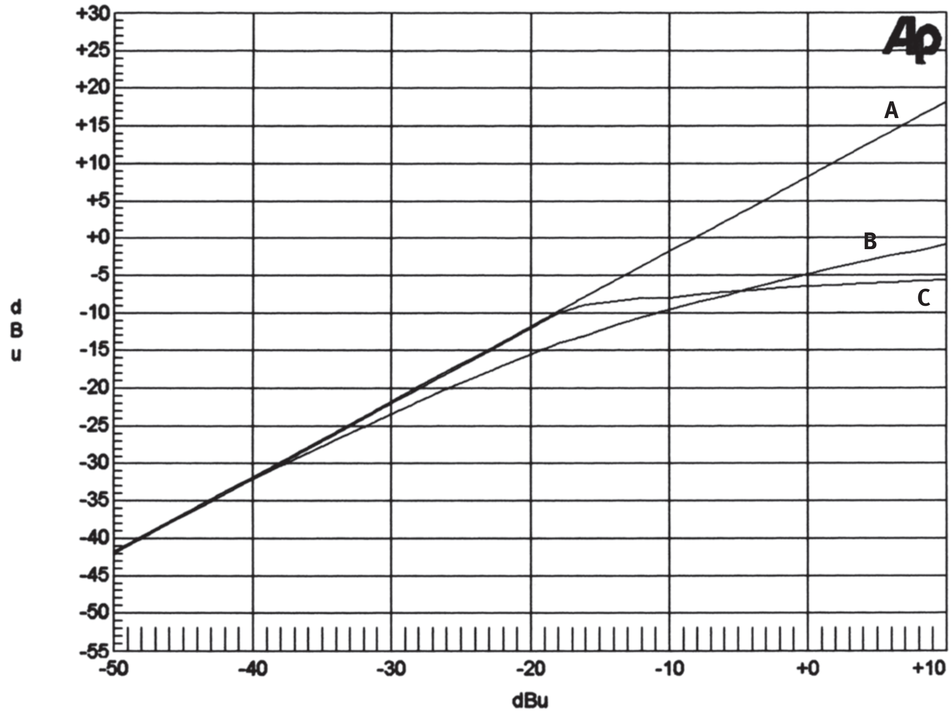
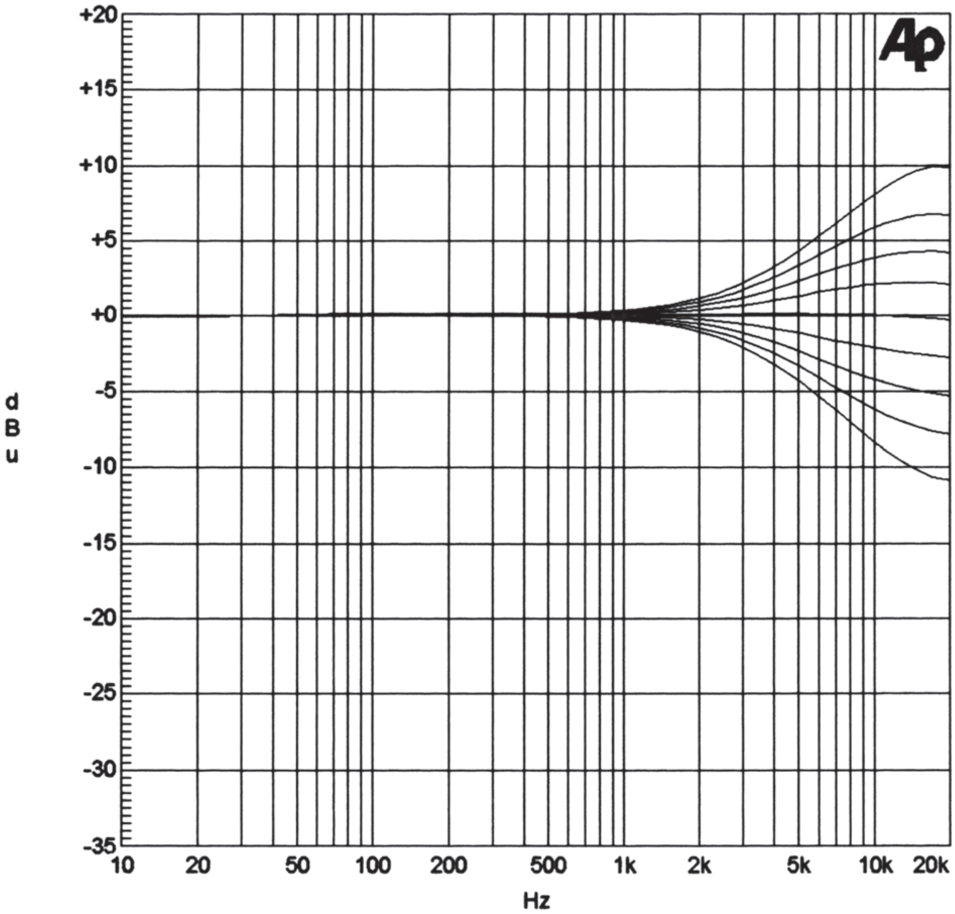


Diagram 2 shows various cut and boost settings of the air band filter.



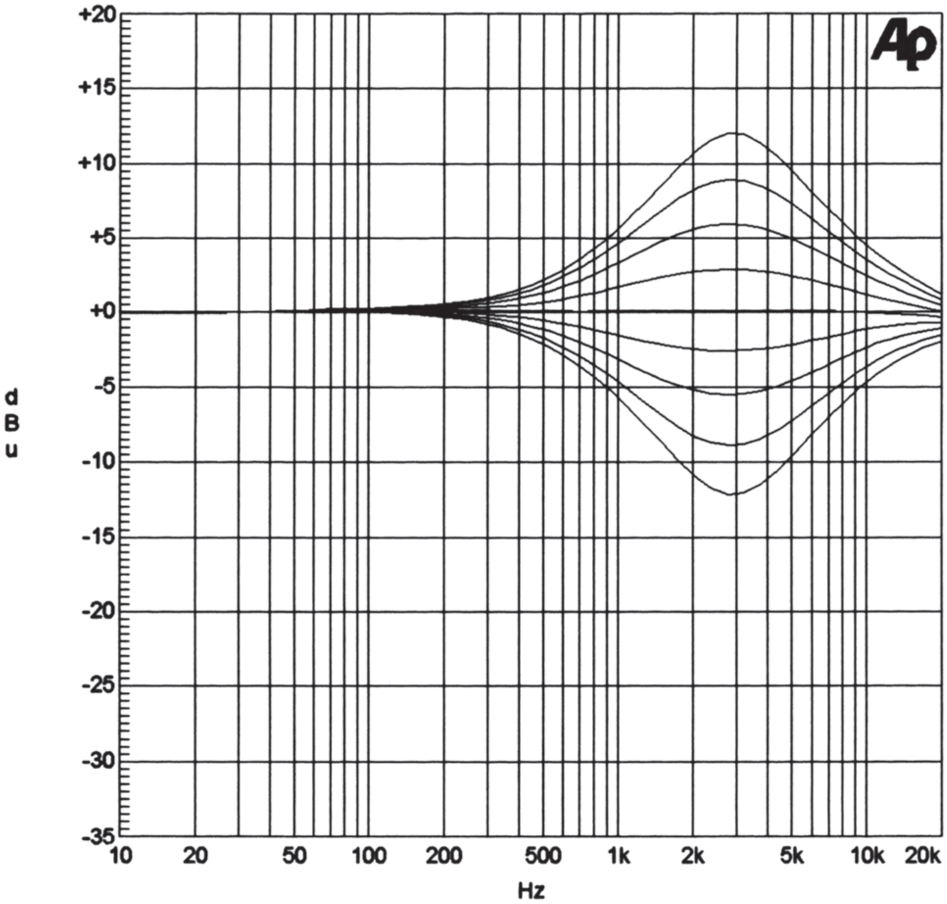


Diagram 3 displays various cut and boost settings of the MHF filter at 3 kHz.

The proportional-Q characteristic is distinctly visible.

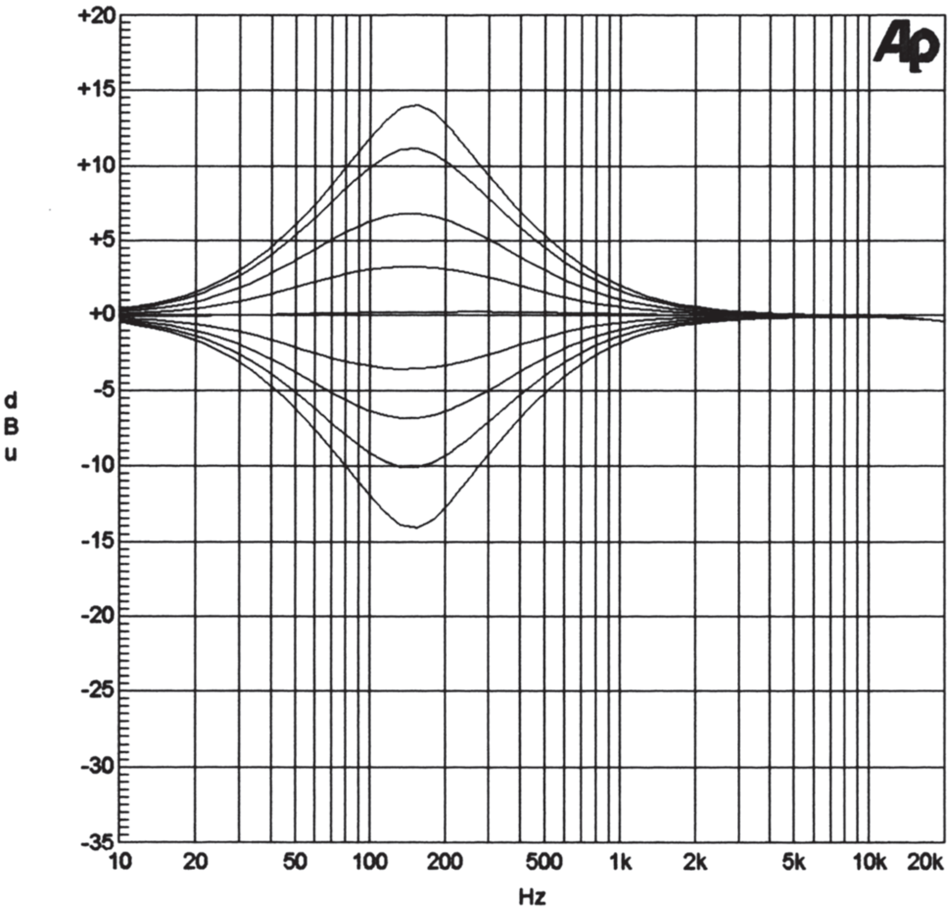


Diagram 4 displays the curves of the LMF filter.

Various cut and boost settings at 150 Hz.

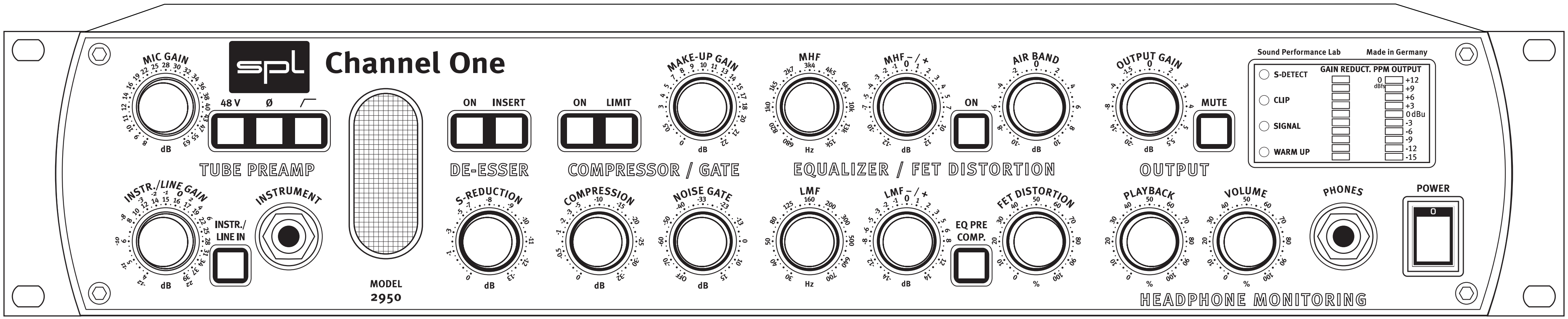
Again the proportional-Q characteristic is clearly visible.

Copy master: recall settings

Artist: | **Engineer:**

Album/Gig: | **Track(s)/Groups:**

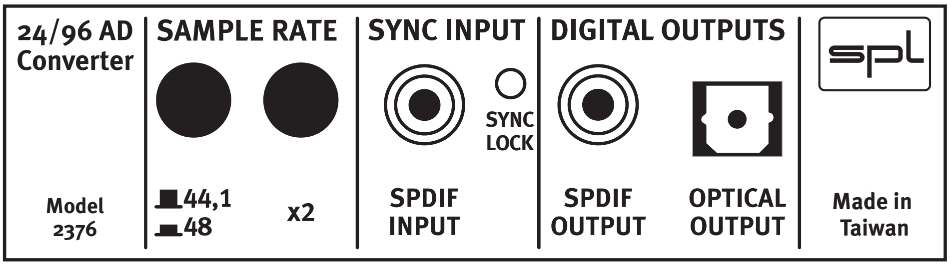
Title: | **Date:**



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 Channel One, model 2950:

- 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.