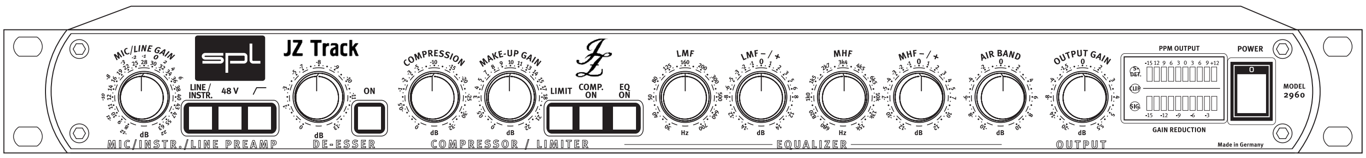
JZ Microphones

Manual



JZ Track

Model 2960

Channel Strip

Version 1.0– 1/2014

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.

JZ Microphones continuously strives to improve its products and reserves the right to modify the product described in this manual at any time without prior notice. This document is the property of JZ Microphones and may not be copied or reproduced in any manner, in part or fully, without prior authorization by JZ Microphones.

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The construction of the JZ Track 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

Symbols and Notes

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

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

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

Attention: Do not attempt any alterations to this machine without the approval or supervision of JZ Microphones. 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 JZ Track, the internal power supply with separate power cable, the guarantee card and directions to find 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

With the JZ Track we have produced a complete channel strip which for the greater part is based on the processing concepts already successfully realized in other products made by SPL. 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. As opposed to this, the JZ Track always ensures fast production without any loss of professional precision and diligence.

Principles

The JZ Track is a channel strip that excels primarily in two aspects: it is disarmingly easy to use and offers outstanding sound qualities. This concept is ideally suited for all kinds of vocal and instrument recordings in studio, broadcast or live applications.

The controls are reduced to the necessary basics to ensure highest user-friendliness. Therefore, working with the JZ Track can be dramatically time-saving which is most important especially in live situations.

Due to its excellent sound quality the JZ Track is a highly recommended alternative to built-in console pre-amps and processing tools. All the modules are immediately at hand for fast interactions. Recording a voice and providing clarity, detail and intelligibility is a question of seconds.

The JZ Track consists of preamplifiers optimized for all kinds of microphones and instruments, SPL's acclaimed De-Esser, a compressor/limiter, an equalizer (EQ) section and an output stage for perfectly feeding following units.

The microphone input can optionally be equipped with a transformer input stage. These transformers already amplify the microphone signals by factor 5 – the preamps are relieved by this factor at any gain setting. The balanced output can also be equipped with optional transformers, read page 22 for further information.

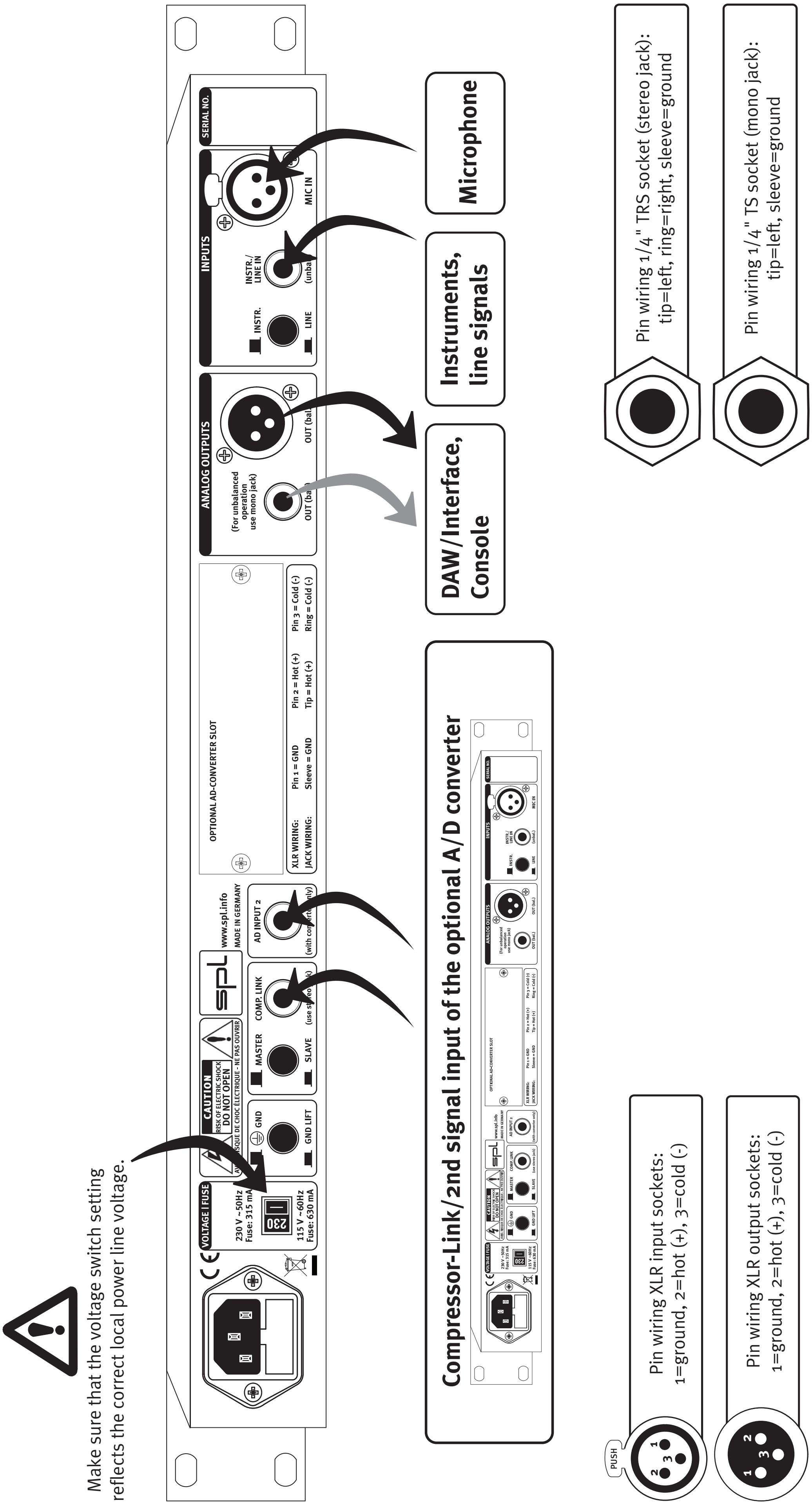
The compressor can be linked with a second JZ Track compressor for coherent stereo operation of two units.

A central display shows metering for output level and gain reduction and all other status LEDs.

Another option is a digital output via a 24 bit/96 kHz A/D converter card (see page 22 for further information).

A special feature of the printed circuitry layout is the central star ground wiring scheme: Disturbing influences that could affect the ground paths are minimized in that 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 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.
Microphone
Instruments, line signals
DAW/Interface, Console
Compressor-Link/2nd signal input of the optional A/D converter
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 (-)
PUSH
SERIAL NO.
INPUTS
MIC IN
INSTR./ LINE IN
INSTR.
LINE
ANALOG OUTPUTS
OUT (bal.)
(For unbalanced operation use mono jack)
OPTIONAL AD-CONVERTER SLOT
XLR WIRING: Pin 1 = GND Pin 2 = Hot (+) Pin 3 = Cold (-)
JACK WIRING: Sleeve = GND Tip = Hot (+) Ring = Cold (-)
www.spl.info
MADE IN GERMANY
AD INPUT 2
COMP. LINK
MASTER
SLAVE
CAUTION
RISK OF ELECTRIC SHOCK DO NOT OPEN
RISQUE DE CHOC ELECTRIQUE - NE PAS OUVRIR
GND
GND LIFT
VOLTAGE | FUSE
230 V ~50Hz Fuse: 315 mA
115 V ~60Hz Fuse: 630 mA
230

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/TS sockets

The INSTR./LINE IN socket only supports unbalanced connections (1/4” TS/mono jack connector). All other sockets are support both balanced (1/4” TRS/stereo jack connector) and unbalanced connections (1/4” TS/mono jack connector).

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.

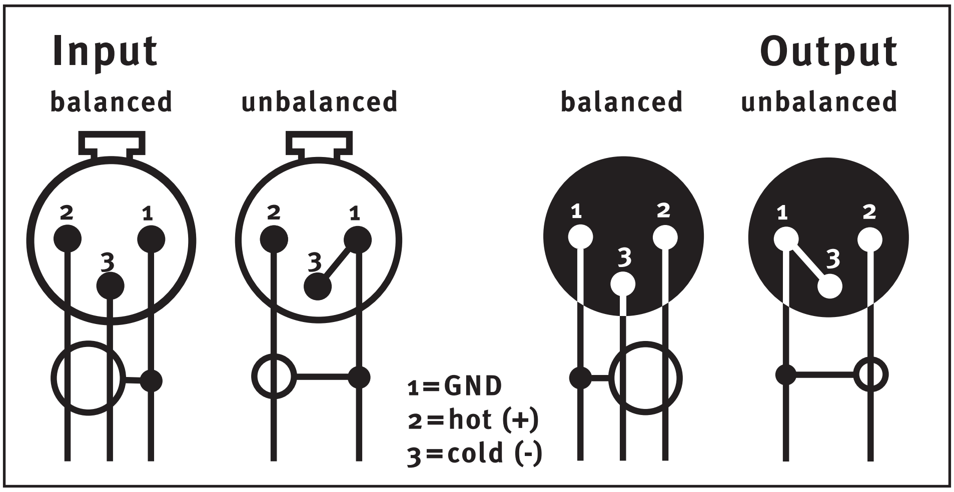
Balanced connections

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 also 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 connec-



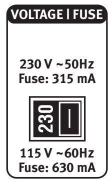
tions:

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

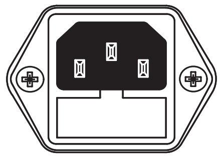
The rear panel VOLTAGE selector sets the local line voltage (115 V position: 110-120 volts/60 Hz, 230 V position: 220-240 volts/50 Hz). The diagram to the right shows the correct switch position for 230 V power supply.

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



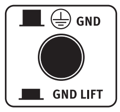
Power connection and fuses

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



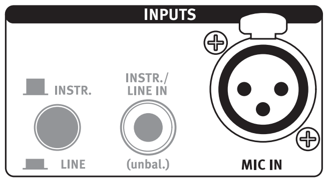
GND Lift

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 22, "Information on I/O transformers").

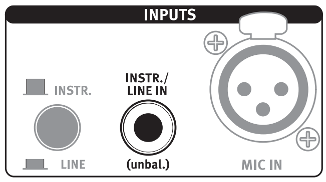


INSTR./LINE IN

Use the balanced INSTR./LINE IN socket for high-level line or instrument signals.

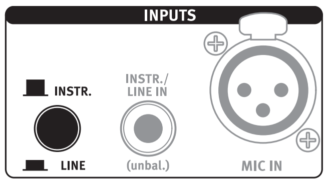
Line signals usually have impedances lower than 1 kOhm, examples are sources like D/A converters, synthesizers or samplers. Instrument signals, such as e-guitars and basses, acoustic guitars with pick-ups and so on, are high impedance sources (above 1 kOhm).

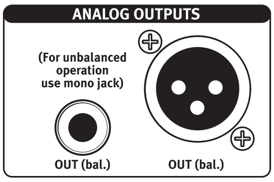
The maximum input level of the INSTR./LINE IN is +28 dBu in LINE mode and +12 dBu in INSTR. mode.



INSTR./LINE switch

This switch serves to match to levels and impedances of line and instrument signals. Press the switch if you have connected line signals, do not press the switch when you connect instrument signals.

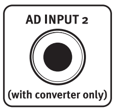




ANALOG OUTPUTS

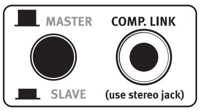
The ANALOG OUTPUTS deliver balanced output signals. An output trans-former can be equipped optionally (see page 29).

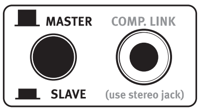
Since both connectors are working in parallel, unbalancing one connector also unbalances the other one. If for example a mono jack connector is plugged into to the jack socket, the XLR socket is operating unbalanced as well. Depending on the impedances of the connected devices, a parallel use of both 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 JZ Track 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 JZ Track is routed to both converter channels. The maximum input level for the converter is +12dBu (=0 dBFS).





COMP. LINK, MASTER/SLAVE

The COMP. LINK (compressor link) feature allows to operate two JZ Track compressors with one control signal to ensure coherent stereo results.

The MASTER/SLAVE switch determines which unit operates as master and which unit is being controlled as slave.

CAUTION – PLEASE NOTE: Never switch two connected units to MASTER! Both units would try to control each other - in worst cases, damaging the units can not be excluded.

Therefore, always follow this procedure when activating COMP. LINK:

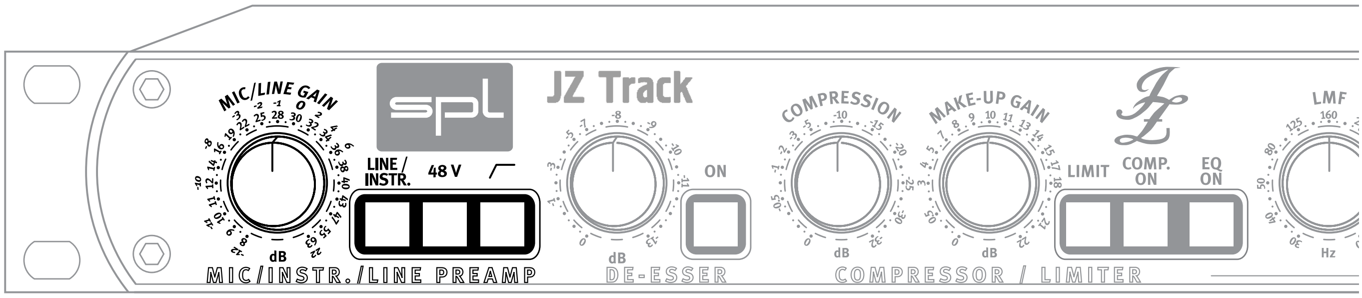
1. Determine the master device (switch not pressed).

2. Determine the slave device (switch engaged).

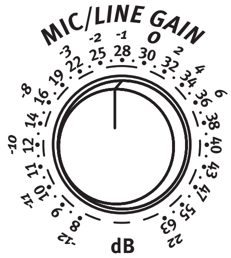
3. NOW connect both COMP. LINK sockets with a stereo jack wiring. Always determining the device status before connecting the COMP. LINK sockets excludes a mutual control activity.

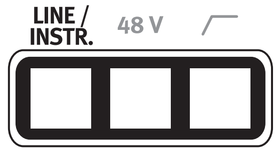
In COMP. LINK mode the master device controls the COMPRESSION, MAKE UP and LIMIT controls of both compressors. The respective controls on the slave unit are deactivated. The COMP. ON switch is not slaved. The GAIN REDUCTION metering of the master unit now is the master display for both units.

If the two units are to be used separately again, the wiring must be disconnected and the slave unit must be set to master again.



MIC/LINE GAIN, LINE/INSTR.

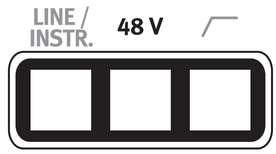




The MIC/LINE GAIN control determines the level of preamplification for microphone and line or instrument signals. In standard mode you control microphone preamplification. Engage the LINE/INSTR. switch to control line or instrument signals.

The preamplification values for microphone signals cover a range from +8 dB up to + 63 dB. If the optional microphone input transformer is installed, the scaled values are to be increased by ca. +14 dB (also see page 22, “Information on I/O transformers”).

The preamplification values for line and instrument signals cover a range from -12 dB to +22 dB.



48 V

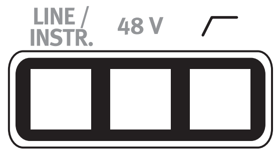
The 48 V switch activates phantom power for condenser microphones with built-in amplifiers. Phantom power should only be activated when using microphones that require it.

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

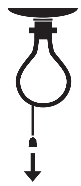
Please always follow these instructions to active and deactivate phantom power – also when changing microphones. The input stage of the JZ Track can be damaged if you ignore these procedures!

1. **Connect the microphone to the JZ Track.**
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.**



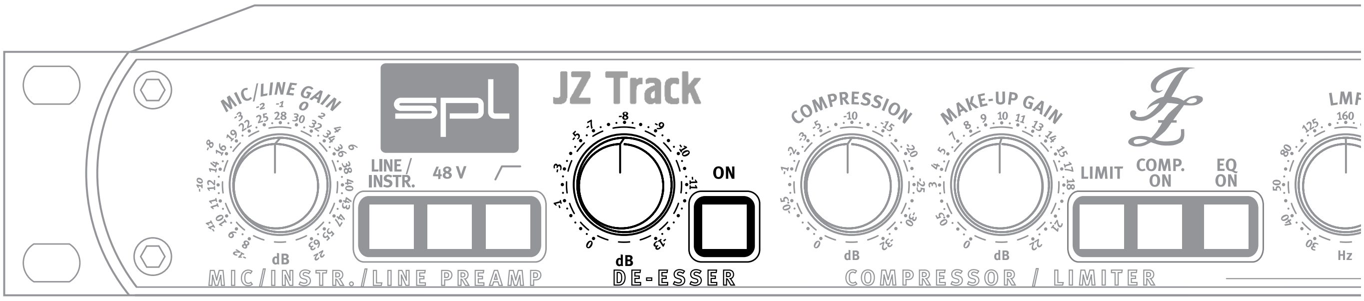
High pass

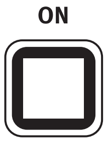
This switch activates the high-pass filter (often also called a “rumble filter”), which operates from a low 50 Hz downwards with a high slope of 12 dB/octave. Therefore, amplification of unwanted low frequencies is avoided effectively while influences on vocal frequencies are not to be expected.



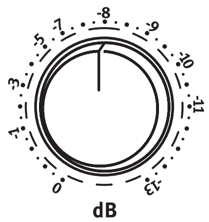
Gain adjustments

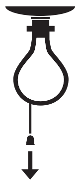
For perfect levelling of the preamplifier firstly switch off all other modules and set the OUTPUT control to 0 dB. The signal can now be levelled with the assistance of the PPM OUTPUT display. To achieve a good signal level the values should range between 0 and +3 dB. At these levels an optimal drive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

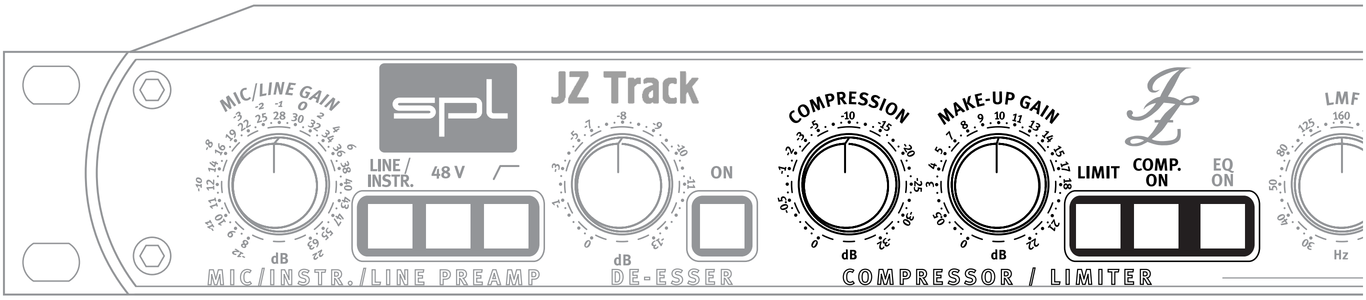




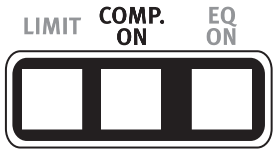
CLIP LED illuminates, the gain value is to be reduced accordingly.



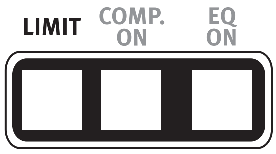




ON

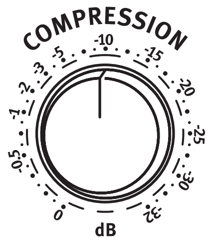


The first processing module is the de-esser, which removes disturbing sibilants when required. The de-esser module is activated with the ON button. The S-DET. LED in the display area shows that S-sounds are being detected. It is independent from the De-Esser control and always informs about detected sibilants – attracting your attention to a possible need for regulation (also see „S-DET." on page 17).



De-Esser control

This control serves to 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.

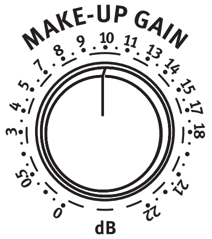


SPL De-Esser technology

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.

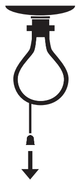
The reduction is accomplished by comparing the average level with the individual S-sounds: the de-esser functions only when the S-noise level exceeds the average level of the entire frequency spectrum. This means for example that original S-sounds with a certain S-portion are not processed whereas those 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 de-essing 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.



COMP. ON

The COMP. ON button activates the compressor module. At the same time the GAIN REDUCTION display shows the processing intensity (also see "MAKE UP GAIN" on page 14 and "GAIN REDUCTION" on page 17).



LIMIT

The LIMIT switch turns the compressor into a limiter. The COMPRESSION control now serves the purpose of controlling the threshold. The Limiter applies a "soft" characteristic and does not function as a peak limiter (see page 20, diagram 1). In other words there is no guarantee that all peaks are limited. It is therefore advisable when driving a subsequent unit that a head-room of 2 to 4 dB remains. 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

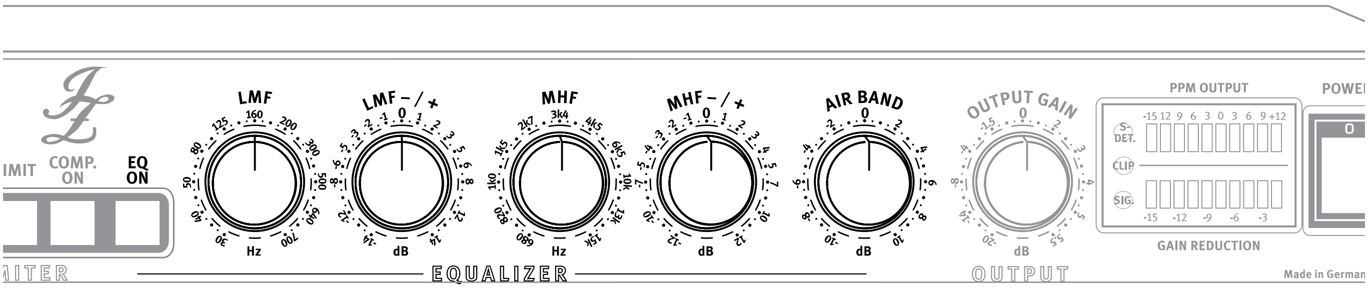
The COMPRESSION control sets the intensity of compression. Turning the control clockwise increases compression. The working area spans between 0 dB (fully left) and -32 dB (fully right).

The compressor applies the so-called "soft-knee" characteristic, which means it starts processing earlier than with hard-knee curve (see page 20, 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.

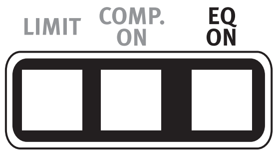
At maximal compression it operates with a ratio of 1:3 between input and output signal – very effective dynamic processing are achievable with unobtrusive sound characteristics.

When setting the COMPRESSION rate the GAIN REDUCTION 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.

IMPORTANT: Make sure that the rear COMP. LINK switch is set to MASTER (not engaged). Otherwise the compressor does not operate. The COMP. LINK switch only serves to determine the SLAVE unit when operating two devices in COMP. LINK mode (for further information see „COMP. LINK" on page 10).

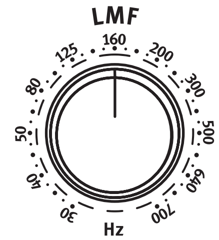


MAKE UP GAIN



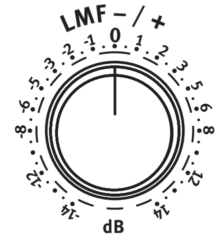
With the MAKE UP GAIN control you can restore the overall level reduction caused by compression or limiting. With assistance of the GAIN REDUCTION display setting the MAKE UP GAIN control is made easy: If the maximal reduction value caused by the loudest tone amounts to -9 dB, for instance, the MAKE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

SPL compressor technology



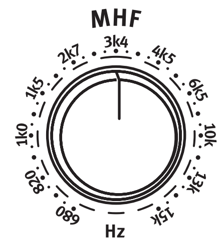
In the compressor/limiter section of the JZ Track the parameters for the time constants (attack and release) are set automatically and adapt themselves to the changing conditions of the input signal, far better than can ever be achieved by manual adjustments. The transient and final oscillation behavior 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 JZ Track 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.



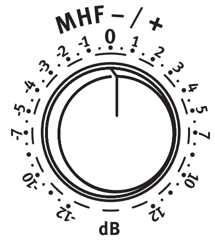
The automatic setting of the release times is dependent on the input signal, too. Fast and large level fluctuations are correspondingly processed with shorter time constants than minor fluctuations in order to limit the distortion of the audio signal as far as possible. Overall this technique provides 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 JZ Track: 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 20.



EQ ON

The ON button inserts the equalizer module into the signal path. The input signal comes from the compressor.

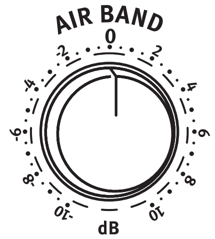
Tip: Deactivating the equalizer module at the beginning of a recording can avoid irritations. Otherwise and especially with intensive EQ settings tonal changes could occur immediately.

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 -/+

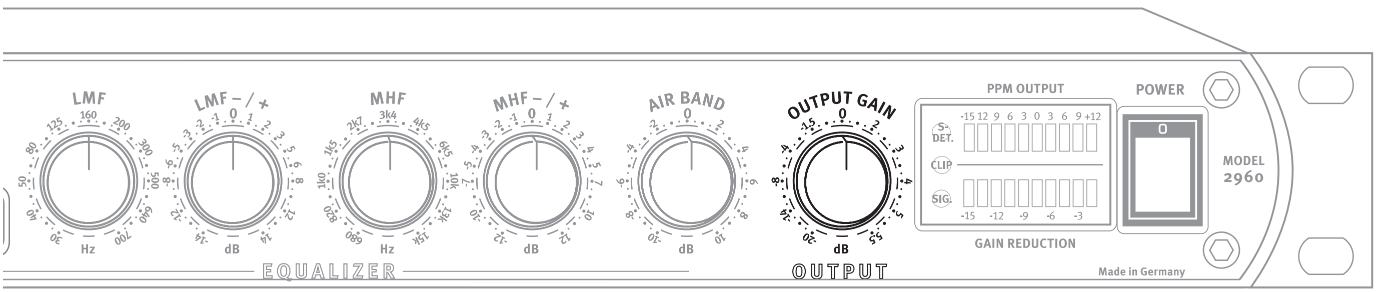
The LMF-/+ control determines the boost or cut of the LMF filter; the maximum values lie between +/- 14 dB. The LMF filter also operates to the proportional-Q-principle, in other words the bandwidth is dependent on the selected boost or cut. This filter characteristic permits a musically more sensible 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.

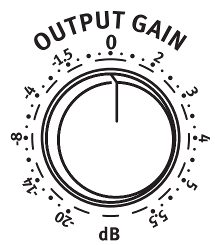
The boost or cut values, in relation to the bandwidth, lie somewhat higher than with the MHF filter. The bandwidth is therefore narrower at maximum boost than with the MHF filter for even more precise filtering. The exact curve of the LMF filter is shown in diagram 2 on page 20.

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.

Control Elements *Output Gain*

MHF

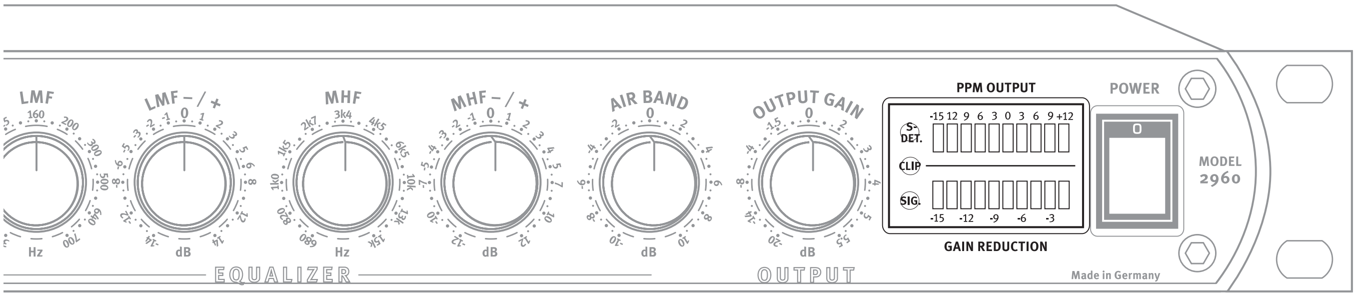




The center frequency of the semi-parametric mid/high frequency filter is set with the MHF control.

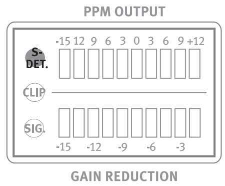
The frequency range can be set between 680 Hz and 15 kHz so that this filter covers a range of 4.5 octaves and can be equally employed in the lower mid as well as the high range.

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



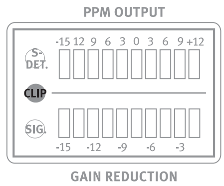
MHF -/+

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 21). 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.

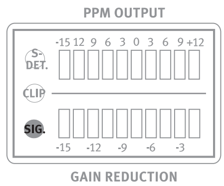


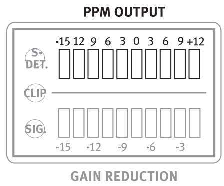
AIR BAND

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 4 on page 21.

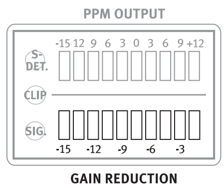




OUTPUT GAIN

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. (see „Gain adjustments“ on page 11).

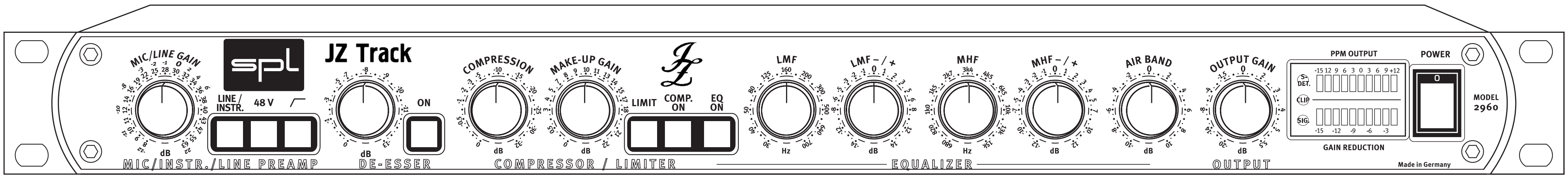


Copy master: recall settings

Artist: **Engineer:**

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

Title: **Date:**



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. 6000 µf 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%/6,81kOhm resistors ensure the pristine quality of the phantom power supply.

Further information on page 9.

Specifications

Microphone Input

Frequency response	10 Hz-200 kHz (-3 dB)	
Maximum input level	+8 dBu	
Common mode rejection at -20 dBu	1 kHz: -80 dB/10 kHz: -68 dB	
THD & N	Amplification:	A-weighted:
	20 dB	-97,5 dBu
	40 dB	-91,0 dBu
	65 dB	-69,6 dBu
Dynamic response	115dB	

Instrument Input

Frequency response	10 Hz-180 kHz (-3 dB)	
THD & N	Amplification:	A-weighted:
	7 dB	-98,4 dBu
	20 dB	-95,8 dBu
	42 dB	-77,2 dBu
Input impedance	Line: 12kOhm, Instrument: 1 MOhm	
Maximum input level	Line: +28 dBu/Instrument: +12 dBu	
Dynamic response	115 dB	

Outputs

Max. output level	+20 dBu
Output impedance	‹ 50 Ohm

Power supply

Toroidal transformer	15 VA
Fuses:	315 mA (230 V/50 Hz)
	630 mA (115 V/60 Hz)

Dimensions & Weight

Stand.-EIA-19 inch/1 U housing	482 x 44 x 210 mm
Weight:	4.15 kg/ca. 9.15 lbs

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

Diagram 1: compressor curves

Line A displays the ratio between input and output

Line B shows the curve characteristics of the compressor

Line C protrays the limiter's curve characteristics

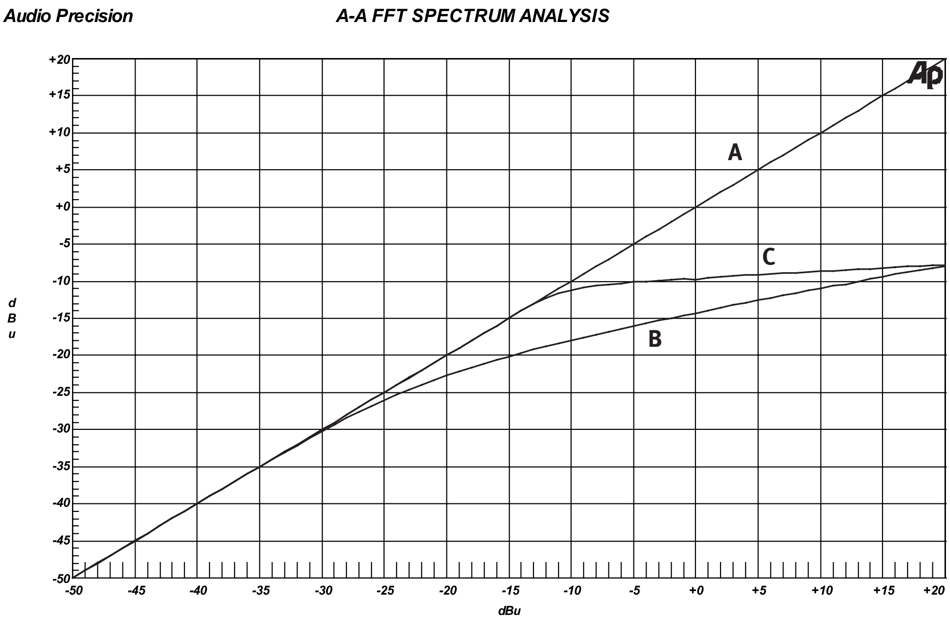
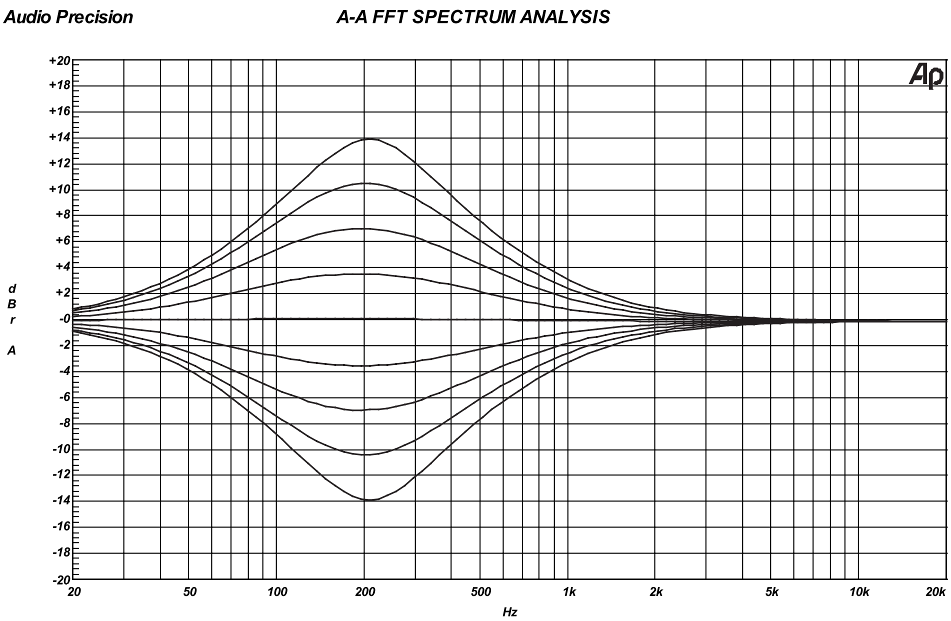


Diagram 2: various LMF settings around 200 Hz



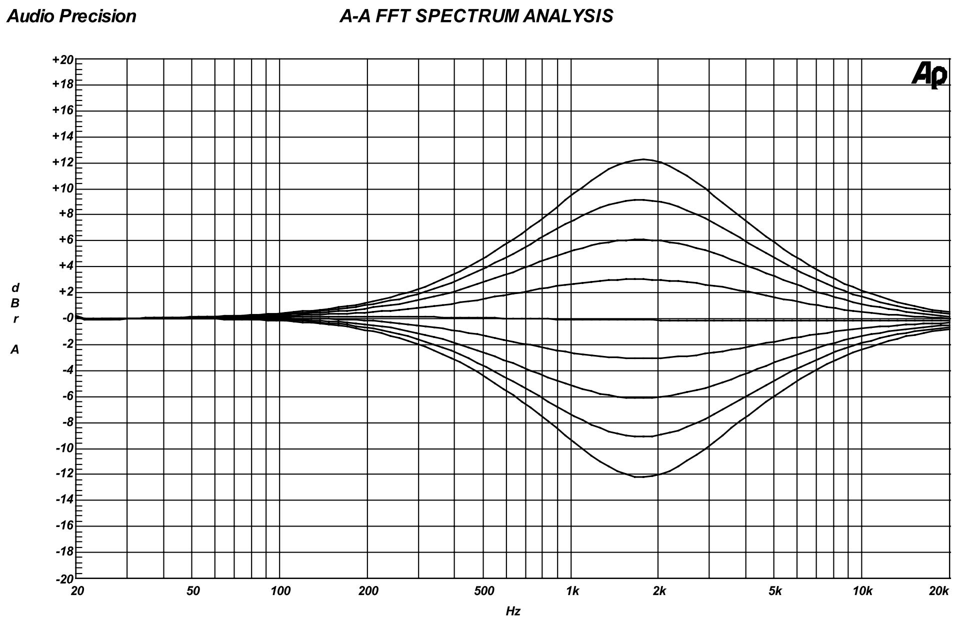


Diagram 3: various cut and boost settings of the MHF filter at around 1,8 kHz

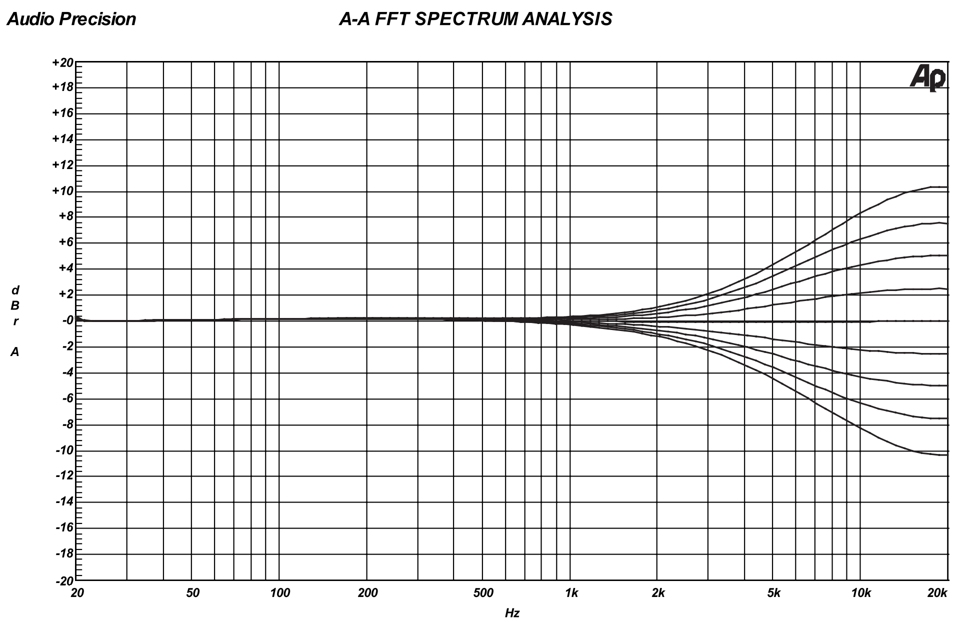
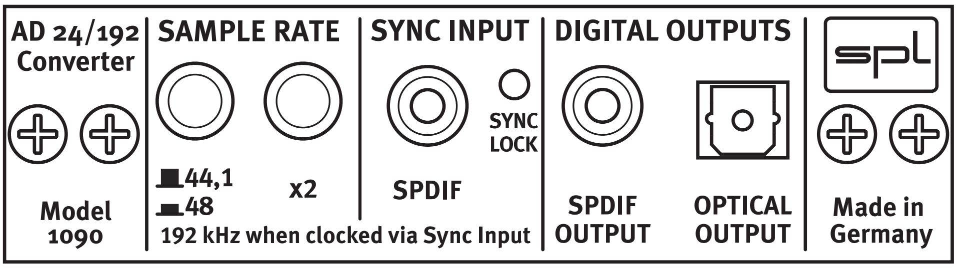


Diagram 4: various cut and boost settings of the Air Band filter

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 JZ Microphones before you place an order. Optional equipment can also be installed after sales.

Available option for the JZ Track:

- 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.