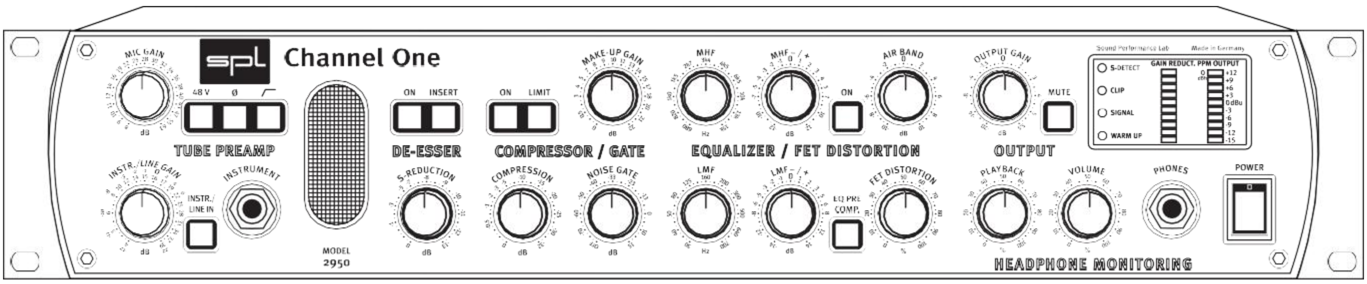
用户手册



ChannelOne

Model 2950

Channel Strip

版本 1.1– 6/2009

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 roduct at the time of packaging with this document.

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The construction of the Channel One, Model 2950, is in compliance with the standards and regulations of the European Community.
2950型的结构符合欧洲共同体的标准和规定。

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.
在其使用期限结束时，本产品不得与普通家庭垃圾一起处理，而必须送回电气和电子设备的回收点。产品、用户手册和包装上的轮式垃圾箱标志已表明。这些材料可以按照它们的标记重新使用。通过重新使用，回收原材料，或其他形式的旧产品回收，你就为保护我们的环境做出了重要贡献。您当地的行政部门有责任可以告诉您的废物处理点。

WEEE Registration: 973 349 88

Symbols and Notes 符号和说明

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. 在本手册中，三角形内的闪电符号警示您：有可能发生危险的电击，即使在机器与电源断开后也会发生。

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. 三角形内的感叹号（！）是为了让您了解必须遵守的重要操作建议和/或警示。要特别注意这些，并始终遵循它们提供的建议。

The symbol of a lamp directs your attention to explanations of important functions or applications. 一个灯的符号引导你注意重要功能或应用的解释。

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. 注意：未经“SPL electronics GmbH”批准或监督，不要试图对本机进行任何改动。这样会使您所有的保修/担保权利和用户支持要求完全失效。

Scope of Delivery & Packaging 交付范围和包装

The scope of delivery comprises the Channel One, the external power supply, the guarantee card and this manual. 交货范围包括Channel One、外部电源、保证卡和本手册。

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. 请保留原包装。在维修过程中，原包装可以确保安全运输。如果你不使用特殊的运输箱，它也可以作为你自己运输的安全包装。

Important Security Information 重要安全信息

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.

请注意并保留本手册。在使用机器之前，请仔细阅读并遵守所有的安全和操作说明。请必须注意遵守本手册和设备上注明的所有警告和特殊安全说明。

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

连接：只能使用所述的连接方式。其他连接可能导致健康风险和设备损坏。

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!

水和湿度：不要在靠近水的地方使用本机（例如靠近洗脸池或浴缸，在潮湿的地下室，靠近游泳池等类似的地方）。在这种情况下，有极高的致命电击的风险!

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!

插入异物或液体：千万不要让异物通过机器的任何底盘开口。你很容易接触到危险的电压或导致破坏性的短路。千万不要让任何液体洒在或喷在机器上。这种行为可能会导致危险的电击或火灾!

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!

打开设备：不要打开机器外壳，因为有很大的风险，你会损坏机器，或者--即使是在断开连接后--你可能会受到危险的电击!

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.

电力供应：本机只能从能提供100至250伏范围内适当电源的电源运行。如果对电源有疑问，请联系你的经销商或专业电工。为了确定你已经隔离了机器，请断开所有的电源和信号连接，要确保电源插头始终处于可接触状态。当较长时间不使用机器时，确保从墙上的电源插座和吉他放大器上拔下插头。

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.

电线保护：确保你的电源线和吉他放大器的信号线被安排好，以避免被踩到或任何形式的压接和与此相关的损坏。不要让任何设备或家具对电线进行压接。

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! >

电源连接过载：在与墙面插座、延长线或分线器电源线或信号输入的连接中，避免任何形式的过载。始终牢记制造商的警告和说明。过载会造成火灾和危险的电击风险! >

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.

雷电: 在雷暴或其他恶劣天气前，请将机器与墙面电源断开（但为了避免雷击，在暴风雨中不要断开）。同样，在任何恶劣天气之前，要断开其他机器和可能相互连接的天线和电话/网络电缆的所有电源连接，这样，不会因为这些次级连接而导致雷电损害或过载。

Air circulation: Chassis openings offer ventilation and serve to protect the machine from over-heating. 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.

空气流通：底座的开口提供了通风，并用于保护机器不会过热。切勿覆盖或关闭这些开口。不要把机器放在柔软的表面（地毯、沙发等）上。当把机器安装在机架或柜子上时，请确保在机器的侧面和顶部留出4-5厘米/2英寸的安装空间。

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.

控制和开关：只能按照手册中的描述，进行控制和开关操作。在安全参数之外的，不正确地调整会导致损坏和产生不必要的维修费用。切勿使用开关或液位控制来实现过度或极端的变化。

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.

修理: 当你认为需要维修时，或当湿气或异物可能意外进入外壳时，或当机器可能跌落并有任何损坏的迹象时，请拔掉设备的所有电源和信号连接，并立即联系合格的技术员。这适用于任何情况，即设备没有受到任何这些异常情况的影响，但仍然不能正常工作或其性能有很大改变。

在电源和电线损坏的情况下，首先考虑关闭主断路器，然后再拔掉电源线。

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.

替换/替代零件：请确保任何维修人员使用原厂更换的零件或与原厂规格相同的零件。不正当的替换零件可能导致火灾、电击或其他危险，包括进一步的设备损坏。

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.

安全检查：一定要请维修技术人员进行彻底的安全检查，确保维修后的机器状态在各方面都达到出厂标准。

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.

清洁：在清洁时，不要使用任何溶剂，因为这些溶剂会损坏底盘的表面。使用干净的干布（如有必要，使用无酸清洁油）。在清洁之前，请将机器与电源断开。

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.

在使用本机之前，要非常小心地检查后机箱的电源选择开关，是否设置在当地正确的线路电压位置（230 V位置：220-240 V/50 Hz，115 V位置：110-120 V/60 Hz）！如果对电源有疑问，请联系经销商或专业电工。

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.

在连接任何设备之前，请确保要连接的任何机器都已关闭。遵循第4页和第5页的所有安全说明，并阅读第8、9和10页有关后部插座和开关的进一步信息。

Placement
摆放位置

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.

将设备放置在一个水平和稳定的表面上。本设备的外壳是EMC安全的，有效地屏蔽了HF干扰。尽管如此，你应该认真考虑你放置设备的位置，以避免电干扰。它的位置应该是你可以很容易地接触到它，但也有其他考虑。尽量不要把它放在靠近热源或阳光直射的地方，并避免暴露在振动、灰尘、热、冷或潮湿的环境中。它还应远离变压器、电机、功率放大器和数字处理器。始终确保足够的空气流通，与设备侧面和顶部保持4-5厘米/2英寸的距离。

Introduction 介绍

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.

许多音频工程师了解SPL的专业音频工具，遵循我们的理念 "一个产品，一个任务"。这旨在快速简单的操作与高质量处理的相结合，以确保最佳的音乐性能。

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.

通过Channel One，我们生产了一个齐备的通道带，它在很大程度上是基于在其他产品中的处理概念，如SPL De-Esser和DynaMaxx压缩器。一个非常复杂的通道带的任务是受益于创新技术，这些技术一直使SPL设备的操作高效和客观。

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.

在很大程度上，通常的录音日是由一系列对立的时间限制决定的--歌手/发言人希望有一个无故障和高效的录音；然而，如果因为不合适的设备而使技术工作准备花费很长时间，这会损失时间，还会增加成本并使工作环境变糟。然而，Channel One在任何情况下都可以快速制作，而不会损失任何专业的精确度和价值。

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.

Channel One包括一个带有麦克风、线路和乐器输入的晶体管/电子管前置放大器，一个去咝咝声器，一个带有噪声门的压缩器/限幅器，一个均衡器（EQ）部分和一个零延迟的耳机监听器。

Principles 规范

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.

因此，Channel One拥有录制音轨的所有工具--与前级放大器一起，它提供了您最需要的处理器，用于纠正和创造性的声音设计。

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.

为了最大限度地方便用户，所有的模块都被减少到最重要的调节和开关设施。快速和有效的操作不会受到阻碍，恰恰相反，它得到了支持。还有更多的时间留给了创造性的任务。

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.

从一开始，我们就非常重视高度的灵活性。例如，三个独立的麦克风、线路信号或乐器输入，每一个都根据其功能进行了优化。

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.

在这个过程中，有两个位置使用了双三极管--- 一个紧接在前置放大器阶段之后，另一个在链的末端，这样处理后的信号就会经过电子管阶段两次。这种结构结合了晶体管前级放大器的优点（高性能，失真最小，噪音低）和电子管产生更好的音乐表现力。

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.

麦克风输入可以选择配备Lundahl 的输入变压器。输入变压器向前置放大器提供五倍放大的麦克风电平。这种额外的放大作用减少了前置放大器的等效负载。平衡输出也可以配备一个Lundahl 变压器。

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.

可选的24位/96kHz的AD转换器模块提供数字输出。Channel One上的一个额外的输入插座可以向AD转换器提供第二个信号。

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.

信号电平、增益降低、S-检测、剪辑警告、预热状态和信号存在的显示和LED灯结合在一起，被显示在同一区域，可以一目了然地进行监听。

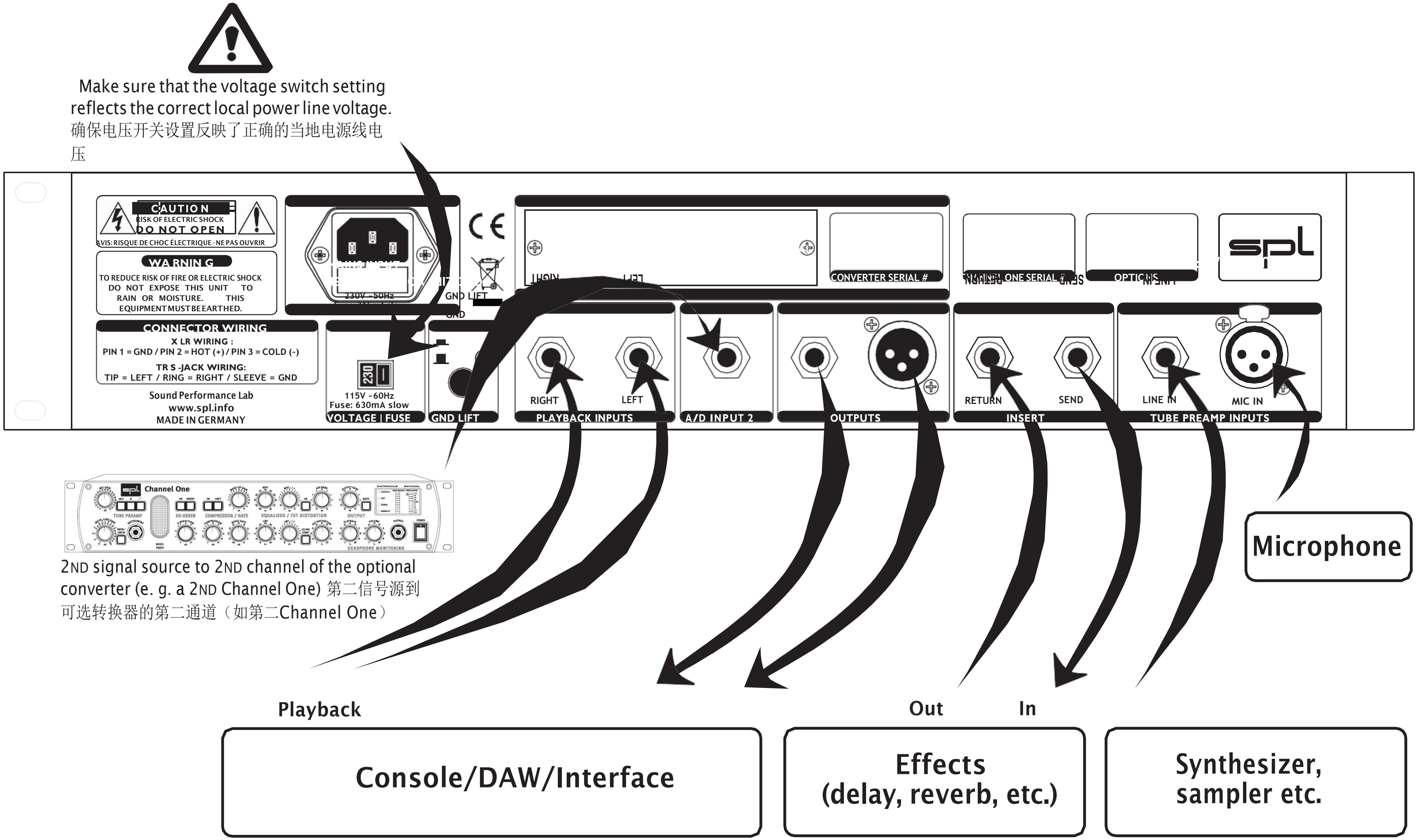
A special feature of the PCB layout is the central star ground wiring: Disturbing influences that could affect the ground paths are minimized by separating audio-ground from the remaining equipment. This leads, in the truest sense of the word ”clean”, to considerably improved tonal quality.

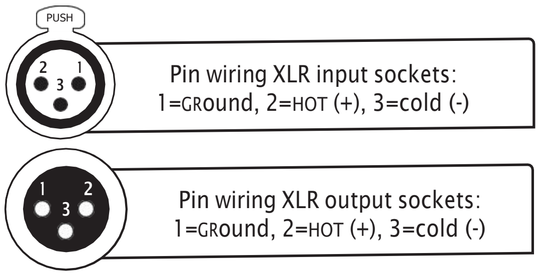
PCB布局的一个特点是星形接地线。通过将音频地与其余设备分开，将可能影响接地路径的干扰影响降到最低。这样使音质的有显著的改善。

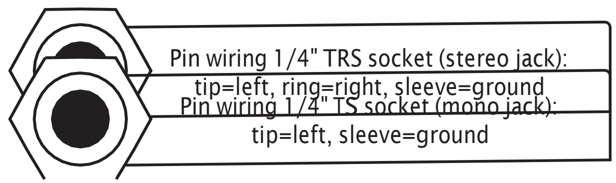
The scatter free toroidal transformer supplies the equipment with the necessary voltages and forms the basis for a clean electrical supply to all parts of the circuitry.

环形变压器为设备提供必要的电压，并为电路的所有部分提供清洁的电力供应奠定基础。

Rear Panel


Make sure that the voltage switch setting reflects the correct local power line voltage.
确保电压开关设置反映了正确的当地电源线电压
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
230V ~50Hz
GND LIFT
115V ~60Hz
Fuse: 630mA slow
VOLTAGE | FUSE
GND LIFT
CONVERTER SERIAL #
CHANNEL ONE SERIAL #
OPTIONS
RIGHT
LEFT
PLAYBACK INPUTS
A/D INPUT 2
OUTPUTS
RETURN
SEND
INSERT
LINE IN
MIC IN
TUBE PREAMP INPUTS
Channel One
2ND signal source to 2ND channel of the optional converter (e. g. a 2ND Channel One) 第二信号源到可选转换器的第二通道（如第二Channel One）
Microphone
Playback
Out
In
Console/DAW/Interface
Effects (delay, reverb, etc.)
Synthesizer, sampler etc.


PUSH
Pin wiring XLR input sockets:
1=GROUND, 2=HOT (+), 3=cold (-)
Pin wiring XLR output sockets:
1=GROUND, 2=HOT (+), 3=cold (-)


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

Signal connection
信号连接

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.

在开始进行第一次或任何后续的连接过程之前，请关闭设备。省略这一项对你的耳朵和你的设备可能会有损伤。

1/4'' TRS sockets 插座

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.

TRS插座LINE IN、INSERT SEND/RETURN、OUTPUTS和AD INPUT 2同时支持平衡（1/4 "TRS/立体声插口）和非平衡连接（1/4 "TS/单声道插口）。PLAYBACK INPUTS插座只支持非平衡连接。

XLR sockets 插座

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.

所有XLR插座都是平衡输入或输出。输入插座总是母插座，用于插入公插座的连接器，输出插座总是公插座，用于母插座的连接器。总而言之，这是一个合理的原则。

Balanced connections 平衡连接

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.

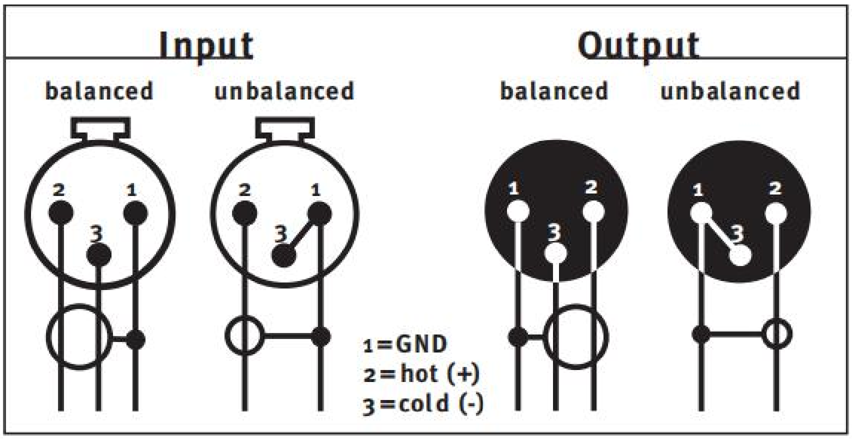
当音频信号通过单线传输时，不可能排除所有的干扰。屏蔽对电的影响是有效的，但对电磁的影响则是无效。电机、变压器和交流电总是能引起干扰。但即使传输会成功，驱动器和接收器之间的地电位差异也会产生干扰。

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. 在平衡连接中，一个极性相反的参考信号通过第二条线与音频信号一起传输。地面信号通过第三根线单独传输。输入和输出级是驱动器和接收器，接收级可以通过减去音频和参考信号之间的差异来抑制干扰。

Unbalanced connections 非平衡连接

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:

从RCA或1/4 "TS插座到非平衡的连接可以不需要适配器就能实现平衡XLR插座的连接。正确的接线很重要。图中显示了XLR插座的引脚配置，以及如何正确连接它们进行非平衡连接：



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.

与RCA插座的连接总是不平衡的，与插口的接线可以是平衡的（1/4 "TRS/立体声插口）或不平衡的（1/4 "TS/单声道插口）。我们建议使用从XLR到RCA或插口的单独配置的电缆，而不是适配器。从音频经销商那里你可以得到任何需要的电缆配置。通过上图，经销商可以确保为你的应用提供合适的电缆。

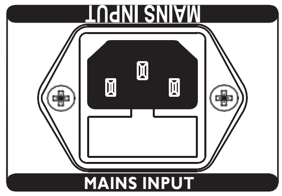
VOLTAGE 电压

The rear panel VOLTAGE SELECTOR sets the local line voltage (115 V position: 110-120 volts/6o Hz, 230 V position: 220-240 volts/50 Hz). The diagram to the right shows the correct switch position for 230 V power supply.

后面板的VOLTAGE SELECTOR设定了当地的线路电压（115V位置：110-120伏/6O赫兹，230V位置：220-240伏/50赫兹）。右图显示了230V电源的正确开关位置。

BEFORE you connect electrical power make sure that the VOLTAGE SELECTOR setting reflects the correct local power line voltage! 在你连接电源之前，请确保电压选择器的设置反映了正确的当地电源线电压!

MAINS INPUT – power connection and fuses 电源连接和熔断器



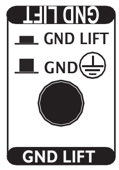
Connect the power cord to the rear MAINS INPUT socket. Transformer, power cord and case connection conform to VDE, UL and CSA requirements.

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).

将电源线连接到后面的主输入插座上。变压器、电源线和外壳连接符合VDE、UL和CSA要求。

MAINS INPUT插座还装有保险丝。它可以从外面接触到，并放在插座下面的挡板后面。保险丝的额定值为315mA（230伏）或630mA（115伏）。

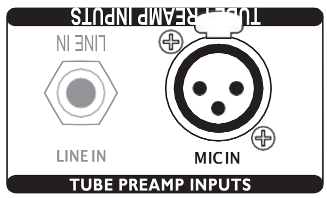
GND Lift



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.

后面板的GND LIFT开关通过将内部地线与设备外壳的地线分开来消除嗡嗡声。例如，当该设备的外壳与其他可能具有不同地电位的设备有共同的接地连接时，就会产生嗡嗡声。该开关通常是停用的，以保持外壳的屏蔽作用。

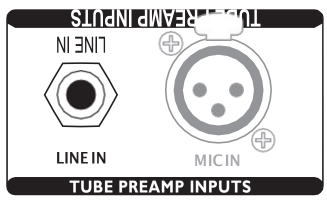
MIC IN



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”).

你可以将任何类型的麦克风连接到MIC IN插座上（动圈、电容、电子管和带状麦克风）。某些麦克风需要48伏幻象电源，可以通过前面板上的48伏开关激活。请阅读第11页 "48V " 和 "Activating phantom power " 两章的重要说明。麦克风输入还可以配备一个可选的输入变压器（见第29页，"关于I/O变压器的信息"）。

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.

对于阻抗低于1 kOhm的高电平信号，请使用平衡的LINE IN插座。

例如，D/A转换器、合成器或采样器。我们建议将其连接到跳线架上以方便使用。

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

线路输入的最大输入电平是+22dBu。

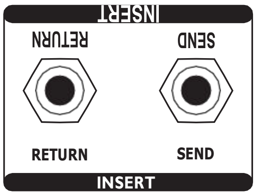
High impedance sources (above 1 kOhm), such as e-guitars and basses, acoustic guitars with pick-ups and so on, must be connected to the INSTRUMENT input.

高阻抗信号源（超过1千欧），如电子吉他和贝司，带拾音器的原声吉他等，必须连接到乐器输入。

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

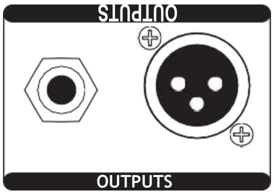
重要提示：如果正在使用仪器输入，线路输入将被停用。

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.

平衡的INSERT连接器（SEND和RETURN）用于将更多的单元集成到Channel One的信号路径中。SEND连接器被放置在除颤器的后面，RETURN连接器则位于压缩器的前面。这允许通过SEND连接器记录前置放大器的信号，同时另一个输入信号可以被送入Channel One的压缩器或EQ部分进行进一步处理。



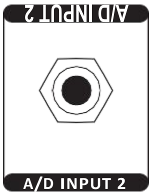
OUTPUTS

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.

OUTPUTS提供平衡输出信号。可以选择配备一个输出变压器（见第29页）。

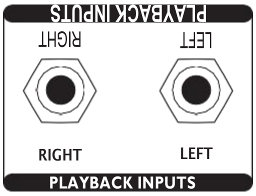
由于两个连接器都是并联工作的，所以一个连接器的不平衡也会使另一个不平衡。例如，如果将一个单声道的插座插入插座，那么XLR插座也是不平衡工作的。根据所连接设备的阻抗，两个输出口的并联使用会降低信号电平。因此，我们建议使用XLR或1/4 "TRS输出插座。



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).

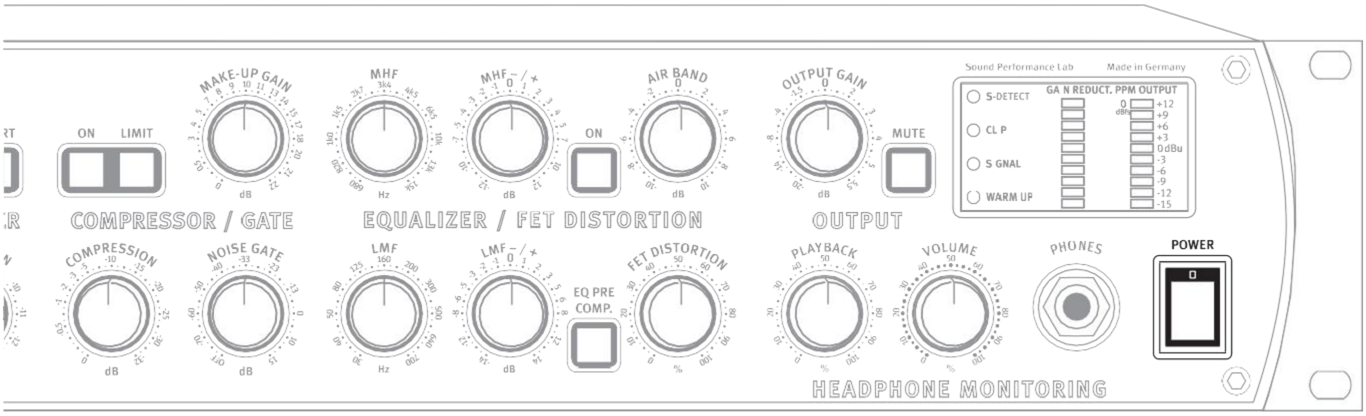
Channel One是一个单声道通道带，但可选的A/D转换卡2376是一个双通道设备。因此，如果将第二个（外部）信号连接到AD INPUT 2，就可以用转换器卡进行转换。如果没有信号连接到A/D INPUT 2，Channel One的输出信号将被输送到两个转换器通道。转换器的最大输入电平是+12dBu（=0dBFS）。



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.

回放信号被连接到非平衡的PLAYBACK INPUTS，形成耳机监听混音。如果有单声道的回放信号，只需连接LEFT接口。这样信号就会出现在两个通道上。如果只有一个通道出现在耳机的一侧，就应该使用右面的连接器。与所有其他的连接器相比，回放输入是非平衡的。



POWER 电源

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

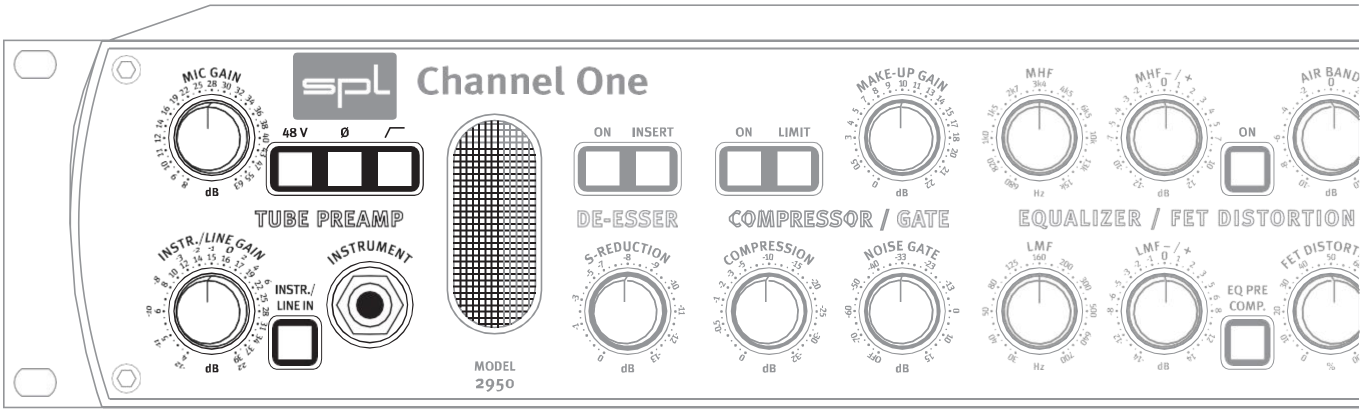
Switch on the unit only after you have checked the correct setting of the rear VOLTAGE SELECTOR and 48V phantom power supply front switch.

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.

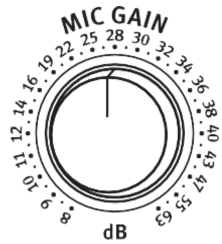
前面板上的POWER开关可以激活设备，通过开关亮起的确认开关状态。

只有在检查了后面的电压选择器和48V幻象电源前开关的正确设置后，才可以打开设备。

当你启动Channel One时，设备开始预热模式以加热电子管。预热周期需要15至30秒。在预热模式中，显示区的WARM UP LED会亮起，当WARM UP LED熄灭时，Channel One就可以开始工作了。



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").

MIC GAIN控制决定了前置放大的水平。前置放大值的范围从+8dB到+68dB。更多信息见第12页，"GAIN调整"。如果安装了可选的话筒输入变压器，比例值将增加约。+14dB（取决于话筒，见第29页，"关于I/O变压器的信息"）。

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.
48V开关激活了内置放大器的电容式麦克风的幻象电源。只有在使用需要幻象电源的麦克风时才应激活。

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.
非常重要的是：所有具有平衡、无地输出的麦克风都可以在激活幻象电源后使用。请确保停用所有其他麦克风的幻象电源。非平衡的麦克风只能在停用幻象电源的情况下使用。

Activating phantom power 激活幻象电源

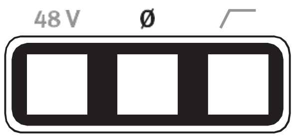
PLEASE ALWAYS FOLLOW THESE INSTRUCTIONS TO ACTIVATE AND DEACTIVATE PHANTOM POWER (ALSO WHEN CHANGING MICROPHONES). THE INPUT STAGE OF THE CHANNEL ONE CAN BE DAMAGED IF YOU IGNORE THESE PROCEDURES!

请始终遵循这些说明来激活和停用幻象电源（在更换麦克风时也是如此）。如果你忽略了这些步骤，Channel One的输入级可能会被损坏！

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.

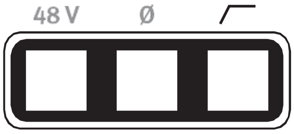
1. 将麦克风连接到Channel One。
2. 现在激活幻象电源用来麦克风使用。
3. 录音后先停用幻象电源。
4. 停用幻象电源后，至少要等一分钟才能断开麦克风的连接! 这样可以确保剩余的电流会被放掉。

Phase reverse 相位反转



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.

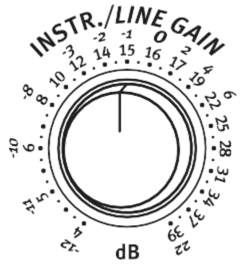
相位反转功能可以扭转传声器信号的极性，将相位反转（180°），以纠正由多个信号源引起的相位反转信号。例如，一个配音艺术家，通过耳机听到自己的声音，同时回荡在脑海中。相位颠倒将导致不自然的声音，甚至与麦克风距离的最小变化也会导致声音的剧烈变化。当在一个音源上使用多个麦克风时，也经常会遇到相位颠倒的情况。我们建议在录音前检查极性是否正确。



High-pass filter
高通滤波器

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..

这个开关可以激活高通滤波器（通常也被称为 "隆隆滤波器"），它的工作频率为50Hz以下，12dB/octave。该滤波器可以避免不需要的低频被放大。与6dB/octave滤波器相比，12dB滤波器的工作强度更大，所以更有效--因此阈值被设置为低于50Hz。



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.

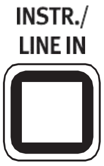
Gain range for line signals reaches from -12 dB to +22 dB. The attenuation allows to also process very high levels. The 0 dB mark is highlighted in the line gain scale – this facilitates to find the setting where a line level signal is processed at unity gain.

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

这个控制器决定了连接到后面的LINE IN或前面的INSTRUMENT输入的信号的前置放大水平。你可以通过INSTR./LINE IN开关来激活相应的输入，下面会介绍。

线路信号的增益范围从-12dB到+22dB。衰减也允许处理非常高的电平。0dB的标记在线路增益刻度中被突出显示--这有助于找到线路电平信号以统一增益处理的设置。

仪器信号可以在+4dB和+39dB之间进行放大。



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.

通过这个开关，你可以在麦克风（关闭）和线路或乐器输入（打开）之间进行选择。无论选择哪种输入，后部的麦克风和线路输入都可以保持连接。只要不使用乐器输入，你就可以选择线路输入作为音源。

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.

乐器输入插孔被放在前面，便于使用。它使用来连接乐器，如电子低音提琴和吉他，带拾音器的原声吉他等。乐器输入具有1 MOhm（一兆欧姆）的输入阻抗。阻抗较低的线路信号，如来自D/A转换器、采样器、合成器等，应连接到后面的LINE IN插座。

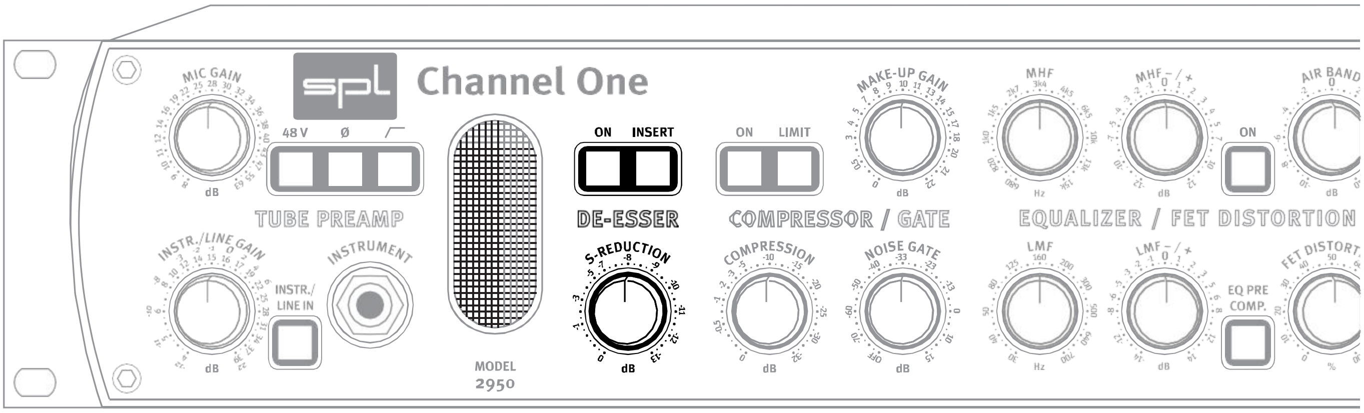
IMPORTANT: As long as an instrument is plugged into front INSTRUMENT input, the rear panel LINE IN input is deactivated.

重要的是：只要有乐器插入前面的乐器输入口，后面板的LINE IN输入口就会被停用。

Gain adjustments 增益调整

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.

为了使前置放大器达到完美的水平，首先关闭其他所有模块（去咝咝声、压缩器、均衡器），并将OUTPUT控制设置为0dB。现在可以在PPM OUTPUT显示的帮助下对信号进行调平。为了达到一个良好的工作水平，数值应该在0到+3dB之间。在这些电平上，可以保证有一个最佳的驱动水平和足够的净空来进行进一步的处理（例如在EQ阶段增加电平）。Clip LED会警告你潜在的峰值；如果在录音过程中，Clip LED亮起，增益值就会相应减少。

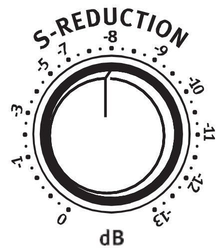


ON

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).

第一个处理模块是去咝咝声，在需要的时候可以去除干扰性的咝咝声。用 "ON " 按钮激活去咝咝声模块。显示区的S-DETECT LED显示出正在检测S音。它独立于S-REDUCTION控制，并会始终通知检测到的咝咝声 - 引起你对需要调节的注意（也见第22页 "S-DETECT"）。

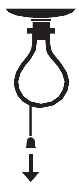
S-REDUCTION



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.

通过S-Reduction控制，可以确定S-声音的还原强度。因为处理是通过与整个频谱水平的比较来进行的（见下一节），对极端的S音水平的处理比对那些较低的水平的处理更密集。这种处理方法使输出信号中剩余的咝咝声达到了一致的水平。

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.

与基于压缩器技术的普通去咝咝声相比，SPL去咝咝声利用了相位消除原理。它采用的滤波器只处理可还原的 "S型频率"，而不干扰频谱的其余部分。S-频率被自动检测，相位被反转并与原始信号混合。这种操作方法有明显的优势，因为它不显眼，并有助于保留原始音质。压缩机典型的副作用，如口齿不清或鼻音，都不会发生。最后，其操作就像拉手刹一样简单。

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.

减少是通过比较平均水平和单个S音来完成的：只有当S音水平超过整个频谱的平均水平时，除噪器才发挥作用。这意味着，例如，具有一定S比例的原始S音不会被处理，而那些过于响亮的，或对声音没有有效贡献的S音则被减少 -- 但声音的特征保持不变。

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.

另一个特点是集成的自动阈值功能，使处理不受输入电平的影响。即使演讲者或演唱者与麦克风没有保持恒定的距离，处理也会保留在预先设定的S-还原值。传统的系统依赖于输入电平，当与传声器的距离减少时，其工作强度更大。因此，SPL De-Esser 不需要长期监听和重新调整，以保持处理的恒定 - 而且它总是可以在压缩机之前应用，因为改变它的位置不会是一个优势。这就是为什么不需要一个和谐的切换功能。

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.

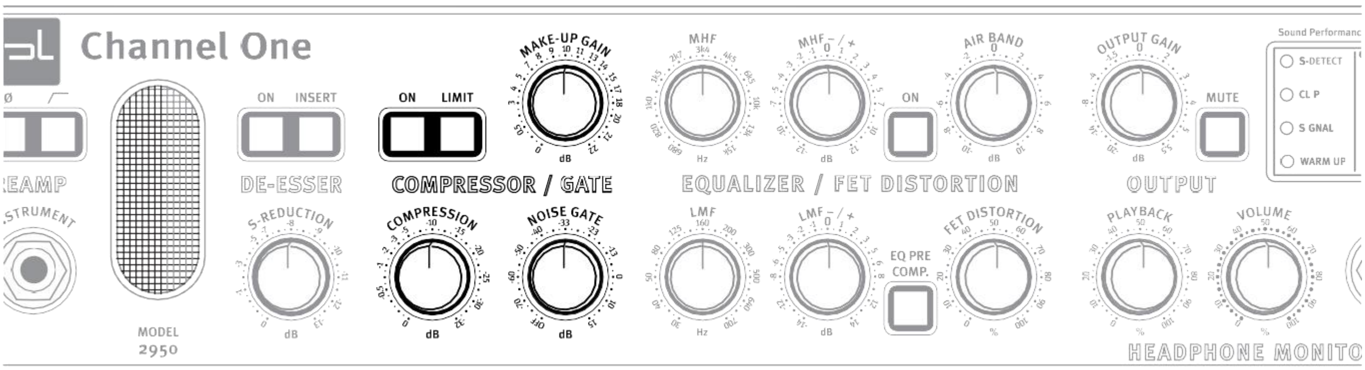
INSERT按钮可以激活任何连接Channel One后面的 INSERT SEND/RETURN回路的外部设备的效果，如延迟或混响。这就把它们绑定到信号链中，从而无限地增强处理能力。

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.

INSERT点位于去噪器和压缩器之间。这使得Channel One的前置放大器/咝声器组合可以与压缩器/均衡器组合分开使用。这扩大了使用范围，因为在这种方式下，Channel One可以作为两个独立的单元使用。

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.

只要单元没有连接到INSERT回路，即使按了INSERT按钮，信号流也不会中断。使用平衡设计的插入式插座，最灵活的方法是通过连接到跳线槽来实现。



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).

ON按钮激活了压缩器/限幅器/噪声门模块。同时，GAIN REDUCT.显示屏显示出处理强度（见第23页的"GAIN REDUCT."）。

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).

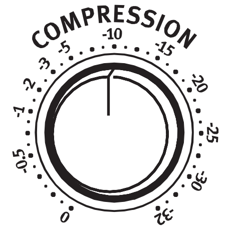
通常情况下，信号流是按照Channel One的设计进行的，为此，输入信号通常来自去咝咝器，或者在激活时来自INSERT。但是，通过EQ PRE COMP.开关，EQ模块可以切换到压缩器模块的前面。这使得它可以作为最后的压缩器或限制器使用（进一步的信息见第17页 "EQ PRE COMP. "一节）。

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.

LIMIT开关将压缩器变成一个限制器。COMPRESSION控制的作用是控制阈值。限制器不具有峰值限制器的功能，换句话说，不能保证所有的峰值都被包括在内。因此，在对后续单元进行修改时，最好保留2-4dB的净空。峰值可以被很好地拦截，而不受干扰。峰值限制器有一个基于系统的缺点，就是会更快地产生可听失真，所以就音质和录音保险而言，我们认为软限制器模式是录音通道带的更好选择。

COMPRESSION 压缩器



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).

COMPRESSION（压缩）控制设置压缩的强度。顺时针转动该控制钮可以增加压缩。工作区域在0dB（完全靠左）和-32dB（完全靠右）之间。

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.

压缩机应用所谓的 "软膝 "特性，这意味着它比硬膝曲线更早开始处理（见第26页，图1，曲线B）。硬膝压缩器有时可以获得更多的响度，但它们的处理很突然，而且用压缩假象破坏录音的危险性也更大。另一方面，软膝压缩器总是能很好地控制电平，并确保最高的录音安全性--如果希望获得更多的响度，信号在录音后仍然可以被处理。

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.

在最大压缩量时，输入和输出信号之间的比例为1:2.5，可以实现非常有效的动态处理，而且声音特征不明显。

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.

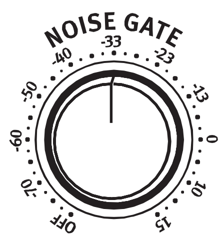
压缩器曲线的具体发展情况见第26页的图1。在设置压缩率时，显示区的GAIN REDUCT.显示有很大帮助。对所选压缩率的影响是以1.5dB为单位的。根据信号源和动态结构的不同，压缩值应该在4-8dB之间，以限制较高的峰值并优化后续录音系统的操作。

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.

通过MAKE UP GAIN（补足）控制，你可以恢复由压缩或限制引起的整体电平降低。在GAIN REDUCT.显示的帮助下，设置MAKE UP GAIN控制非常简单：例如，如果最大声的音造成的最大降低值为-9dB，MAKE UP GAIN控制也要设置为+9dB。如果现在关闭压缩器/限幅器，就可以听到所获得的响度增益。

NOISE GATE 噪声门



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.

噪声门控制可以减少信号暂停时的软干扰。当完全逆时针转动时，噪声门被关闭。通过顺时针方向转动该控制，阈值增加。这意味着噪声门会相对较早地关闭。

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.

dB

噪声门控制的处理范围是在-100dB/OFF（门控制完全逆时针旋转）和+15dB（门控制完全顺时针旋转）之间。因此，噪声门可以在整个动态范围内操作。

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.

噪声门的功能非常稳定：它打开的点比它再次关闭的点高6dB（滞后6dB）。因此，明确的关闭和打开是有保证的，最令人担心的 "颤动 "特性被排除。即使是关键信号也能得到干净的处理。

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.

噪声门的释放时间是自动设置的。取决于程序的自动化，根据音乐作品进行自我调整，从而确保最佳的（不可察觉的）开启和关闭。

SPL compressor technology
SPL 压缩机技术

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).

在Channel One的压缩器/限幅器部分，时间常数（attack和release）的参数是自动设置的，并根据输入信号的变化情况自行调整，远比手动调整的效果好。声音和乐器的瞬态和最终振荡行为是不断变化的，有时非常不稳定，手动控制只能获得良好的平均值，在关键时刻会产生令人失望的效果（如失真声音，"抽吸" 等）。

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.

例如，如果压缩器必须对刺耳的P或T噪音作出非常迅速的反应，那它也可以对较柔和的音调作出缓慢的反应，否则就会发生失真。因此，Channel One压缩器对大的波动水平的调节要比小的波动快；持续时间较长的音调会自动用较长的attack（启动）时间来处理，以防止失真。

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.

Release时间的自动设置也是取决于输入信号的。为了尽可能地限制音频信号的失真，用比小波动更短的时间常数对快速大电平波动进行相应处理。总的来说，这种技术在快速、不显眼的控制响应和最小的音频信号失真之间提供了最佳解决方案。其结果是一个自然和清澈的声音印象。

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.

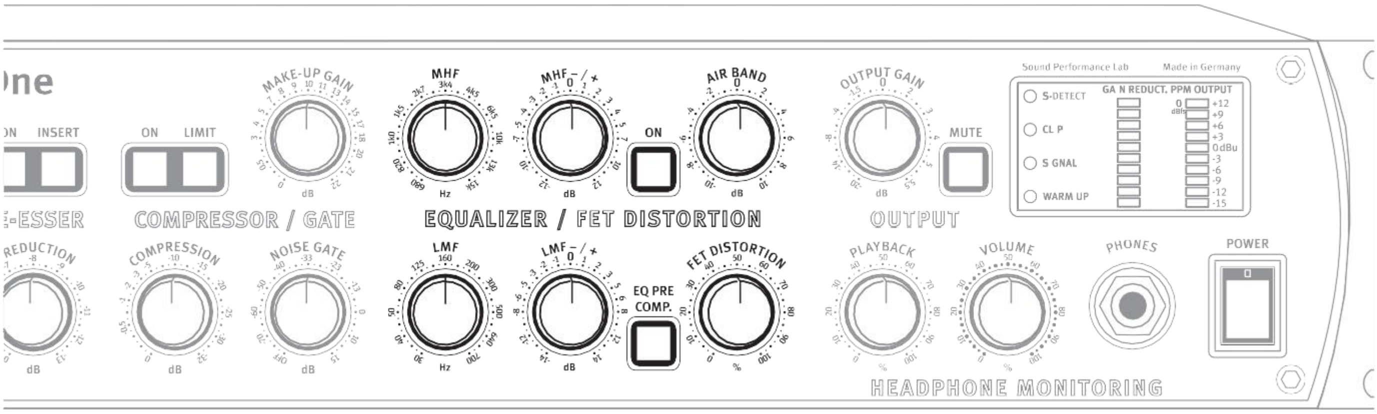
电路的另一个技术特点有助于Channel One中压缩器的高音频质量：SPL的双VCA驱动。一个VCA接收同相的信号，另一个接收非同相的信号。随后，信号通过一个差分放大器。这个电路的效果是，去除失真和偏移波动--两个信号的差分乘积意味着干扰被抵消了。然而，原始信息被进一步放大了6dB。此外，VCA还能相互缓解，因为它们共享负载。它们甚至没有在饱和范围内运行的风险，这确保了避免偏移的噪音，咔嚓声或爆裂声。

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.

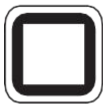
SPL的双VCA驱动电路总体上显示非常明显的失真值改进，因此与传统的电路相比，实现了更清晰和更清澈的声音印象。声音和乐器被赋予了一个相当自然，动态的音色，而 "消音" 的音调则听不到了。

The compressor characteristics are portrayed on page 26.

压缩器的特性在第26页有描述。



ON 打开



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.

ON按钮将均衡器/FET失真模块插入信号路径中。在正常情况下，输入信号来自于压缩器。通过EQ PRE COMP.按钮，可以在压缩器之前被切换进来，使输入信号可以从去噪器或插入器中接收。

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.

重要提示：为了避免在录音开始时出现刺激，建议停用均衡器/FET失真模块。否则，音调可能会立即发生变化，此外，在DISTORTION控制的情况下，还会出现额外的失真。

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.

EQ PRE COMP.开关颠倒了压缩器/限幅器和均衡器（EQ）的顺序。当按下开关时，均衡器在压缩器/限幅器之前工作；不按下开关时，顺序保持不变。当需要解决重复出现的问题或创造特殊的声音时，可以非常灵活的对 Channel One进行操作。

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

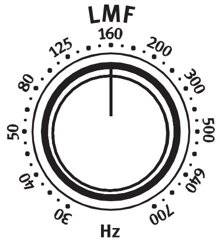
下面的例子说明均衡器什么时候需要在压缩器/限幅器前切换。

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.

当乐器或声音在某些频率范围内出现过度强调时，可以先用均衡器缩小这些范围。随后可以更容易地对信号进行压缩。

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.

另一个合理的应用是将压缩器模块作为最后的限制器来使用，以保持稳定的输出电平。如果在限制之后再使用EQ，就不能保证输出电平不会改变。



LMF

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

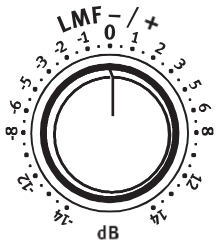
半参数低音滤波器的中心频率是用LMF控制（低/中频）来设定的。

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.

可调节的频率范围在30Hz和700Hz之间，因此这个滤波器涵盖了大约4.5个八度的范围，允许它从最深的低音到较低的中音范围使用。

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

这与MHF滤波器一起确保涵盖整个频谱。



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.

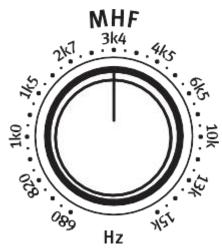
LMF-/+控制决定了LMF滤波器的提升或削减；最大值在+/-14dB之间。 LMF滤波器也按照proportional-Q-principle工作的，换句话说，提升或削减取决于所选择的带宽。与恒定Q值的滤波器相比，这种滤波器的特性允许对频谱进行更合理的处理：如果选择一个更彻底的设置，这将导致对要处理的频率范围的更精确的定义。这反过来又将邻近范围的影响降到最低。

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.

与带宽有关的提升或削减值，比MHF滤波器要高一些。因此，在最大升压时，带宽比MHF滤波器更窄，可以进行更精确的过滤。LMF滤波器的确切曲线见第27页的图4。

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.

LMF滤波器可以在许多方面得到应用。例如：突出声音的基音，减少“boom” 的频率，用于在录音或随后的混音等过程中，强调放置低音的仪器。



MHF

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

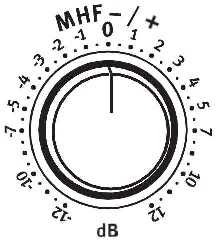
参数中/高频滤波器的中心频率是通过MHF设置控制的。

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.

频率范围可以设置在680Hz和15kHz之间，这样滤波器就涵盖了4.5个八度的范围。

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

4.5个八度，可以在中低频和高频范围内同样使用。这与LMF滤波器一起确保了整个频谱的涵盖。



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.

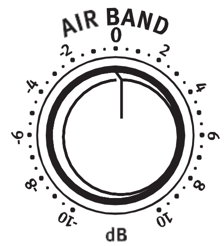
这个控制决定了MHF滤波器的提升或削减；最大值在+/-12dB之间。MHF滤波器也利用了比例-Q原则：提升或削减值设置得越高，带宽就越窄；提升或削减值越低，带宽就越大（MHF滤波器的确切曲线可以在第27页的图3中看到）。滤波器的结构允许完整的范围，从有选择地去除突出的频率，到赋予乐器突出的特点，都可以有效而迅速地涵盖。

Recommendation on frequency settings for LMF and MHF
关于LMF 和 MHF 频率设置建议

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.

为了尽可能快速、准确地找到要处理的频率，首先将MHF-/+控制调节到最大位置。随后应该寻找相关的频率。因为在最大设置下，滤波器的工作带宽是最小的，在这个设置下可以听到最明显的频率，更容易被找到。最后，在用MHF确定了频率后，可以应用所需的MHF+/-设置。

AIR BAND 空气感频段



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.

均衡器模块中的高频滤波器被描述为空气带。具有钟形特性的线圈电容滤波器，中心频率为17.5kHz，在这里开始工作。在这个频率上，最大可能的强调是+10dB，最大可能的阻尼是-10dB。

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.

线圈电容滤波器所具有的柔和自然的音调特性，使其在高频率范围内为人声提供清晰和......空气感，此频段会增添明亮效果。另一方面，刺耳的声音可以通过阻尼使其具有更悦耳的声音特性。

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

AIR BAND滤波器的特性在第26页的图2中显示。

FET DISTORTION FET 失真

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.

DISTORTION（失真）控制提供了对信号施加失真的能力。失真是无限的，从0%到明显可感知的谐波。失真阶段位于均衡器的前面，因此新产生的频谱也可以用均衡器处理。

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.

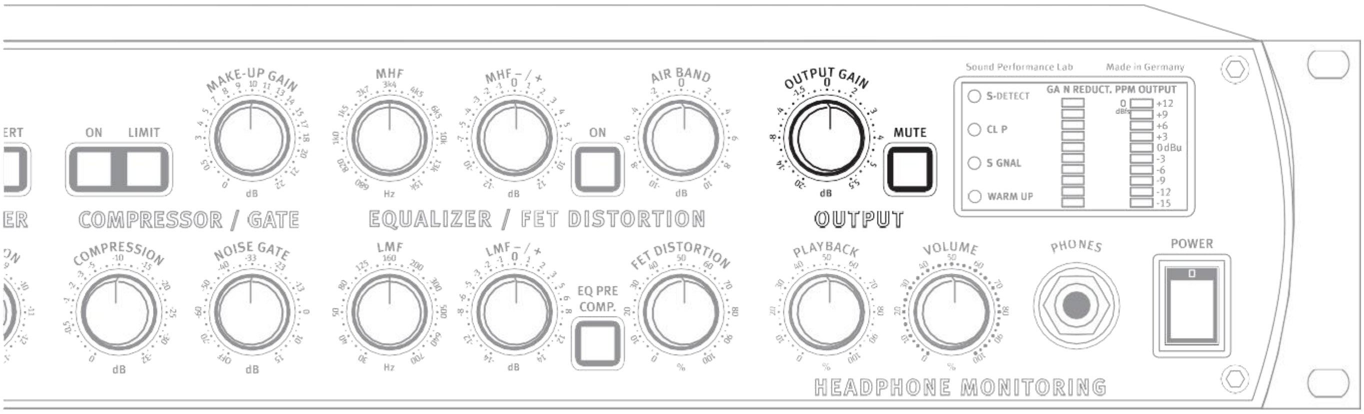
一个（或多或少过载）场效应晶体管构成了失真电路的一部分。它的特性曲线近似于电子管，听起来明显比纯二极管失真器 "更质感"。

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.

信号电平对DISTORTION模块的工作模式是最重要的。为了获得有用的结果，信号电平应该在0到+6dB的范围内。超过这个范围，结果在很大程度上取决于输入信号的条件及其频谱。类似正弦波的信号（如电子琴、人声、吉他）的处理比以和声为主的信号（如小鼓、踩镲等）更早被听到。建议花点时间和精力来找到正确的设置。

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.

重要的是：为了避免在录音开始时的刺激，建议取消EQ，特别是将DISTORTION控制设置为0%。否则，音调会立即发生变化，此外，在DISTORTION控制的情况下，还会产生额外的失真。



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.

输出的信号可以被抑制到-20dB，也可以通过OUTPUT控制进一步放大到+5.5dB，为后续单元或可选的AD转换器提供最佳驱动。选定的输出电平显示在PPM OUTPUT的显示区域内。

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.

在开始录音之前，OUTPUT控制应该被设置为0dB（12点位置）：然后未受影响的数值是清晰的，可用于调整前置放大器的电平。

MUTE

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

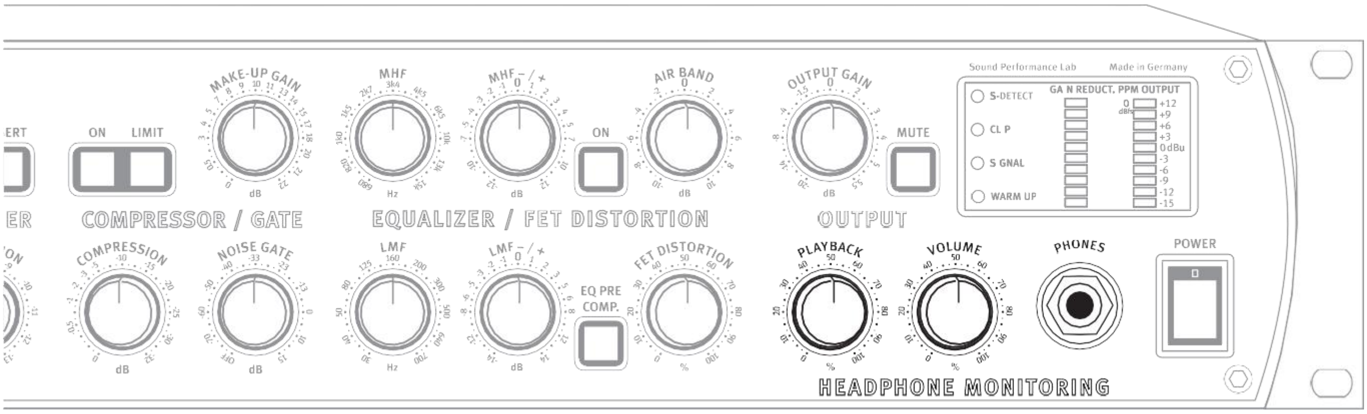
静音开关使输出信号静音；激活后，PPM OUTPUT显示器不显示任何数值。

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.

一个合理的应用实例可能是，在录音过程中，Channel One的输出信号和回放信号一起通过录音室的监听来重现。当随后对录制的片段进行监听时，就有可能听到不相干的歌声或歌手的评论。因此，建议按下静音开关，以允许聆听干净的录音。在继续录音之前，不要忘记关闭静音开关。

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.

另一个例子是，让音乐家排练一段时间，然后在准备好后，释放信号路径，通过关闭静音开关开始录音。



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.

对于最佳的工作条件，特别是良好的音调，这时单独的监听混音是非常重要的。这就是为什么Channel One配备了一个耳机监听部分，提供了一个耳机放大器和一个混音台，为音乐家生成一个带有回放和录音信号的单独混音。

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.

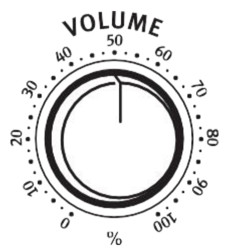
一般来说，耳机监听部分当然总是可以通过耳机直接监听录音。耳机监听模块的另一个实际用途是直接监听信号质量，以迅速定位和消除可能的干扰。

PLAYBACK 回放

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).

PLAYBACK控制可以调节传递给音乐家的播放信号的音量。有两种方法来传递单声道的重放信号：第一种是将音乐传给耳机，这种情况下必须连接PLAYBACK INPUT LEFT。另一方面，有些音乐家希望只通过耳机一侧听到重放信号，这样他们就可以用另一只耳朵直接听到自己的声音（只有重放信号，没有麦克风信号）。在这种情况下，连接右面的播放输入，并将音量控制设置为关闭（也见第10页，播放输入）。

VOLUME 音量



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.

VOLUME（音量）控制可以调节麦克风、乐器或线路信号的音量调节。该设置与输出控制或静音开关的设置无关，这意味着尽管一个调制器的输出值发生了变化，但耳机中的音量不会改变。

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.

提示：当使用硬盘系统或数字调音台工作时，可能会出现延迟。如果音乐家收到的监听信号有时间滞后，就会出现“Flanging”或相位效应。因此，为了避免延迟，建议监听信号直接从耳机监听器传到耳机上。请记住，录音信号没有被回放信号再次拾起，因为当同一信号被PLAYBACK和VOLUME控制混合时，会发生淬灭相位。

PHONES 耳机

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.

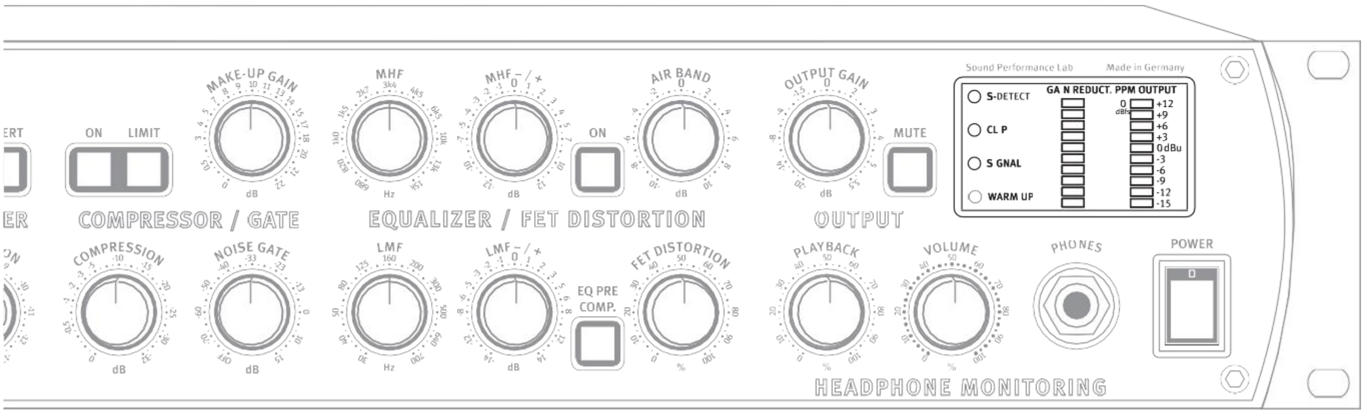
将耳机连接到前面板上的标准1/4"（TRS）立体声插孔插头。低电阻的输入允许连接所有常用的耳机。

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

它的设计是：Tip=左声道，Ring=右声道，Sleeve=地线。

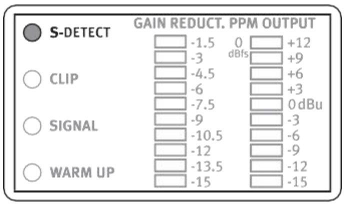
Make sure that the plug is firmly seated for a solid connection. Reduce volume level before you remove or plug in a headphone (or when switching headphones). **NEVER plug in a mono 1/4" jack (TS) to the headphone output. The use of a mono 1/4" will lead to a short-circuit that will destroy the amplifier 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.**

确保插头牢固固定，以确保连接牢固。在移除或插入耳机前（或切换耳机时），请降低音量。千万不要把1/4 "单声道插孔（TS）插到耳机输出上。使用 1/4 "单声道会导致短路，从而破坏功放级。标准的耳机连接器总是有立体声插头，因此当你直接连接耳机时，可以保证正确的连接。当你通过跳线或延长线等连接耳机时，要仔细检查你是否使用1/4 "立体声TRS插头。



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

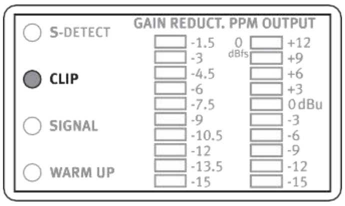
所有的状态和水平显示都集中在一个中央显示区域，所以所有重要的信息都可以一目了然。



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.

S-DETECT LED显示检测到嘧声的时间。它只有在降噪开关打开时才有效，但它独立于S-REDUCTION控制。因此，如果你打开了减震器，你将始终被告知检测到的嘧嘧声和可能需要的被调节的可能。



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.

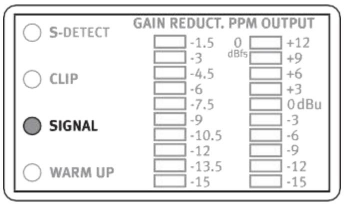
CLIP LED显示了设备的过载。该LED灯的削波电平大约比内部满刻度低2dB（符合+19dBu）。CLIP LED应该尽可能少地闪动。

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).

在信号流的所有相关点上，显示屏会被读出：在前置放大器后面，在压缩器/限制器后面，在均衡器后面，在输出控制后面。所有可能导致过载的原因都可以直接检查出来（过载的麦克风/声道/线路增益，压缩器/限幅器中过多的补偿值，均衡器中过多的提升或过高的输出电平）。

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.

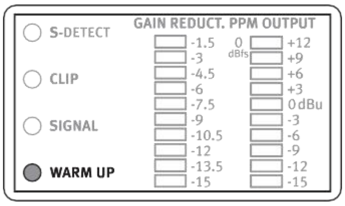
可能的过载原因可以通过简单地单独关闭模块来快速检测。如果在录音过程中发生过载，最快的补救措施是逐渐减少前置放大器中各自的增益控制。



SIGNAL 信号

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.

当前置放大器接收到信号时，SIGNAL LED会亮起。这提供了一个快速检查信号源是否正确连接的方法。所有高于-50dB的电平都被涵盖。

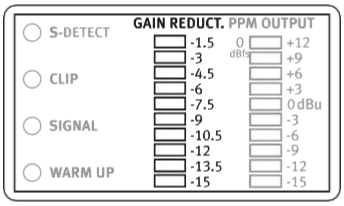


WARM UP （暖机）

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.

WARM UP (暖机) LED显示了电子管级的预热阶段。当LED灯熄灭时，Channel One已经准备好进行操作；在预热期间，输出信号电平很低，听起来有失真。

GAIN REDUCT.



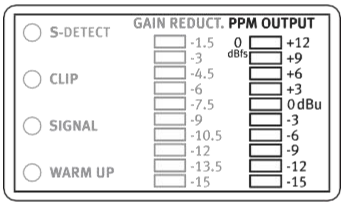
The GAIN REDUCT. display provides information about the processing being undertaken with the compressor/limiter or the noise gate. The level changes, perhaps caused by compression, are scaled in 1.5 dB steps. The display is activated when the compressor/limiter module is switched on.

GAIN REDUCT 显示提供了有关正在进行的压缩器/限幅器或噪声门处理的信息。可能由压缩引起的电平变化是以1.5dB为单位的。当压缩机/限幅器模块被打开时，显示被激活。

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

噪声门的操作通过所有的GAIN REDUCT.LED灯的点亮来显示。当信号电平低于门槛设置时，LED灯就会亮起。

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).

PPM OUTPUT显示屏显示输出电平的峰值读数（校准为0dB），并出现在后部输出上。这个显示也是用来给前置放大器调平的。左边标注的数值 "0dBFS "代表了可选的AD转换器的最大电平，不应该被超过（进一步的信息在AD转换器的手册和第29页上给出）。

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.

虽然PPM OUTPUT的显示值只覆盖到+12dB，但内部仍有足够的余量（大约6dB），所以输出值可以超过这个极限而不引起削波。最佳的噪声性能范围是在0和+9dB之间。

Power Supply

电源供应

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.

围绕一个环形变压器，该电源确保了一个最小的电磁场，没有嗡嗡声或机械噪音。电源的输出端由一个RC电路过滤，提取由你的电源服务引起的噪声和嗡嗡声。6000μf的电容器使正负半波变得平滑。

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.

幻象电源来自变压器的一个独立绕组，一个精确的电流调节器，一个干净的幻象电源为48伏。我们高质量的0.1%/6,81 kOhm的电阻确保了幻象电源的原始质量。

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

Further information on page 9.

用于电子管级的250伏电源用300μF滤波以减少嗡嗡声。更多信息见第9页。

规格介绍

Microphone input 麦克风输入

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

Line/instrument input 线路/乐器输入

Frequency range 频率范围: (-3 dB)	10 Hz-100 kHz	
Common mode rejection 共模抑制: (@ 0 dBu, LINE IN only)	1 kHz: -80 dB/10 kHz: -78 dB	
THD & N:	Gain增益:	A-weighted:
	5 dB	-99,4 dBu
	20 dB	-97,2 dBu
	42 dB	-79,4 dBu
Input impedance 输入阻抗:	Line 线路: 20 kOhm / Instrument: 1 MOhm	
Maximum input level 最大输入电平:	Line线路: +22 dBu / Instrument: +14 dBu	
Dynamic range 动态范围:	119 dB	

Output 输出

Maximum output level XLR/TRS 最大输出电平:	+20 dBu
Output impedance 输出阻抗:	‹50 Ohm

Dimensions & weight 尺寸和重量

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

Note: 0 dBu = 0,775 V. Specifications are subject to change without notice.
Note: 0 dBu = 0,775 V. 规格如有变化，恕不另行通知

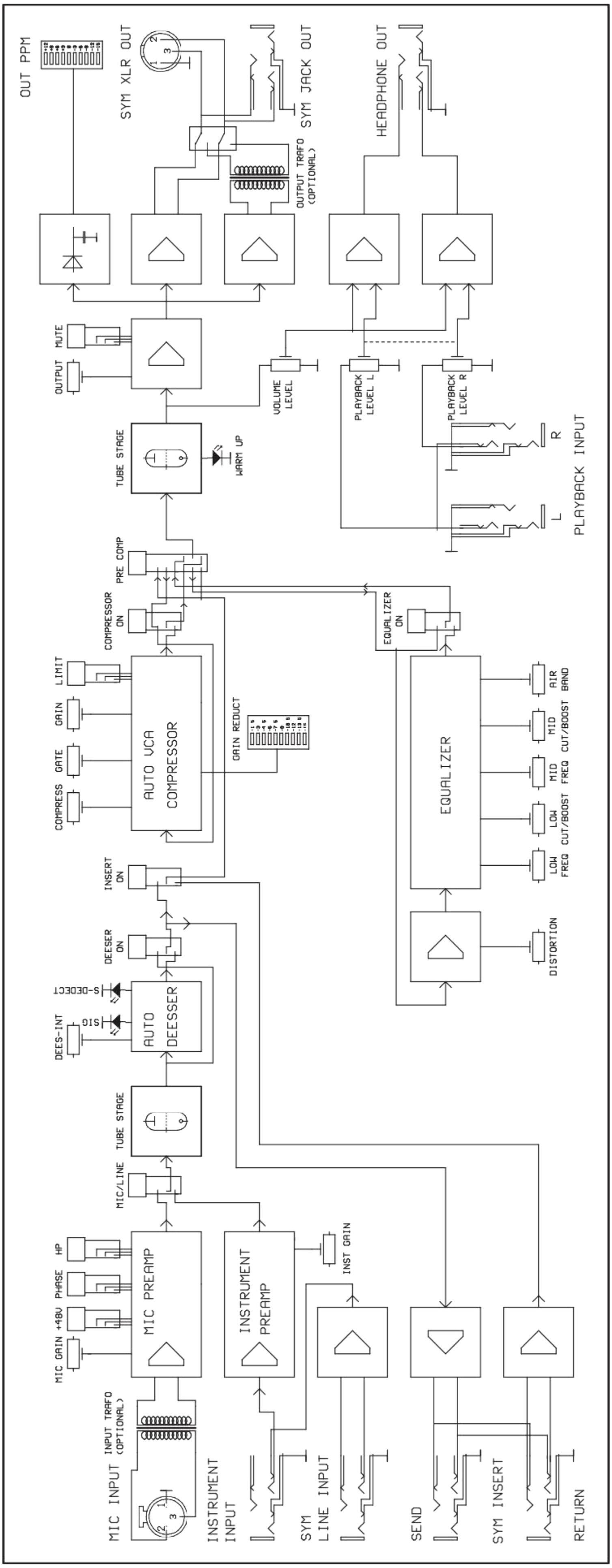

OUT PPM
SYM XLR OUT
SYM JACK OUT
HEADPHONE OUT
OUTPUT TRAFO (OPTIONAL)
MUTE
OUTPUT
TUBE STAGE
WARM UP
VOLUME LEVEL
PLAYBACK LEVEL L
PLAYBACK LEVEL R
R
L
PLAYBACK INPUT
PRE COMP
COMPRESSOR ON
EQUALIZER ON
LIMIT
GAIN
GATE
COMPRESS
AUTO VCA COMPRESSOR
GAIN REDUCT
EQUALIZER
AIR BAND
MID CUT/BOOST
MID FREQ
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 图1: compressor characteristics
压缩器特征

Reference curve A displays the relation between input and output.

参考曲线A显示输入和输出之间的关系。

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

曲线B显示了压缩机的曲线特性。软拐点特性清晰可见。

Curve C portrays the limiter's curve characteristics.

曲线C描绘了限制器的曲线特性。

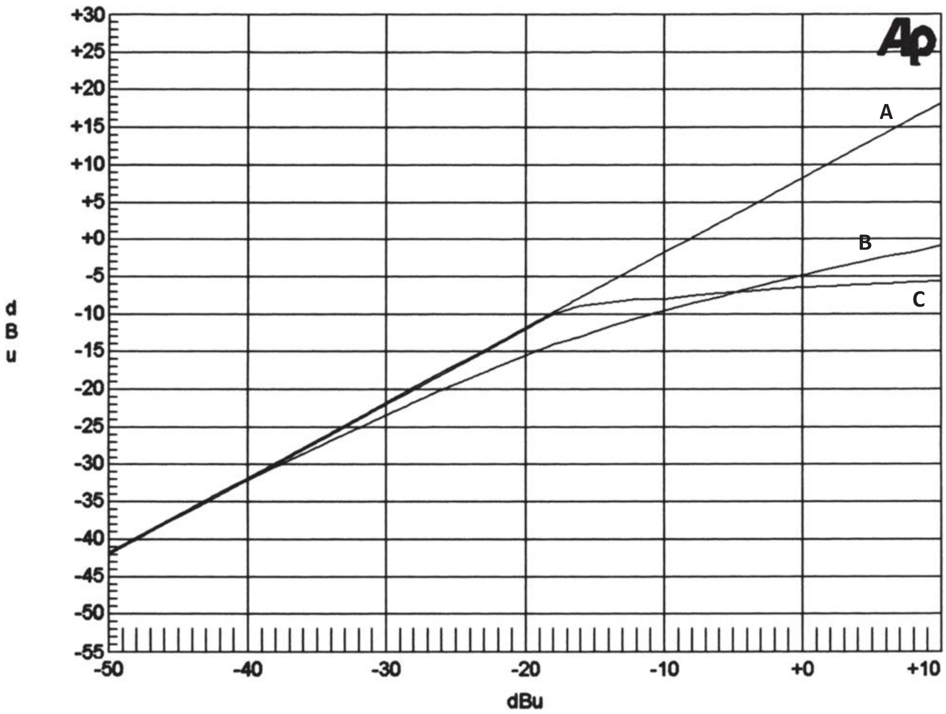
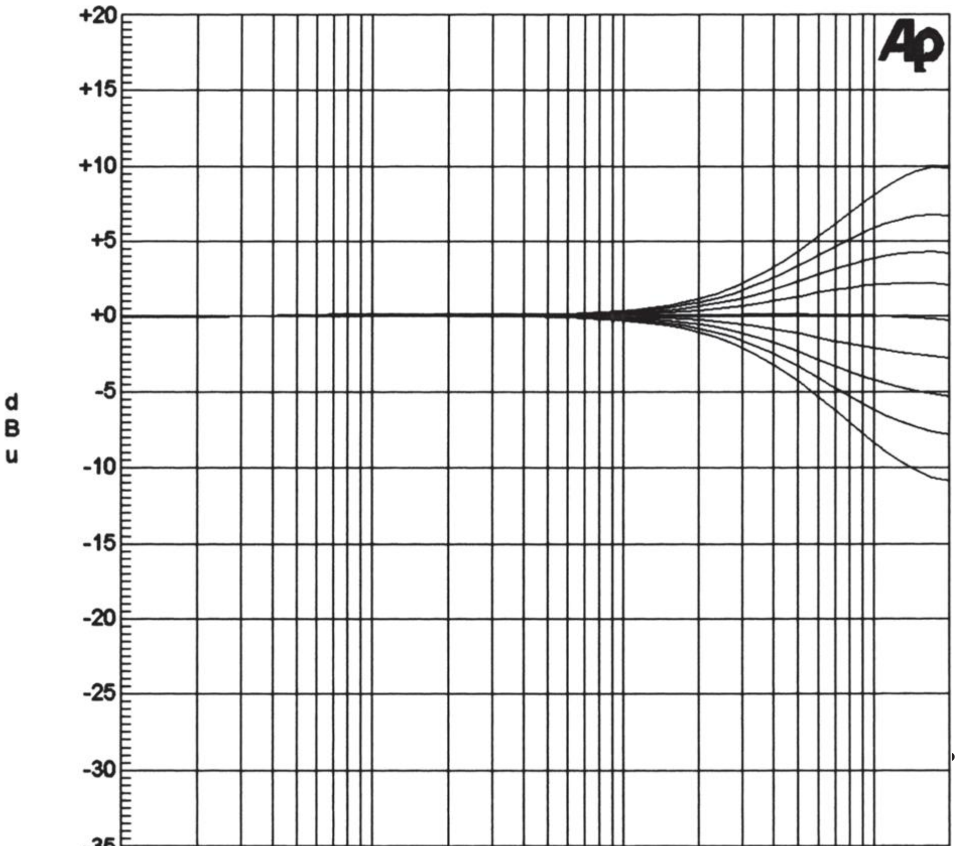


Diagram 2 shows various cut and boost settings of the air band filter.

图2 显示了空气带过滤器的各种削减和提升设置。



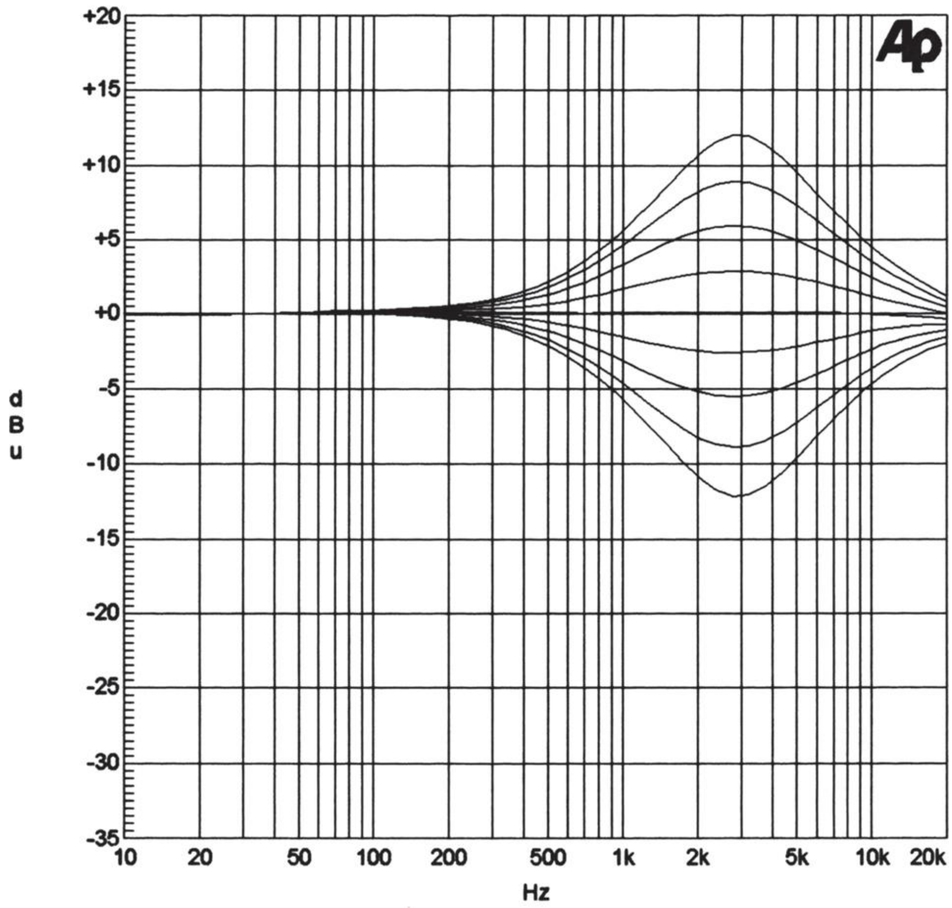


Diagram 3 displays various cut and boost settings of the MHF filter at 3 kHz.

The proportional-Q characteristic is distinctly visible.

图3显示了MHF滤波器在3kHz时的各种削减和提升设置。

proportional-Q特性是明显可见的。

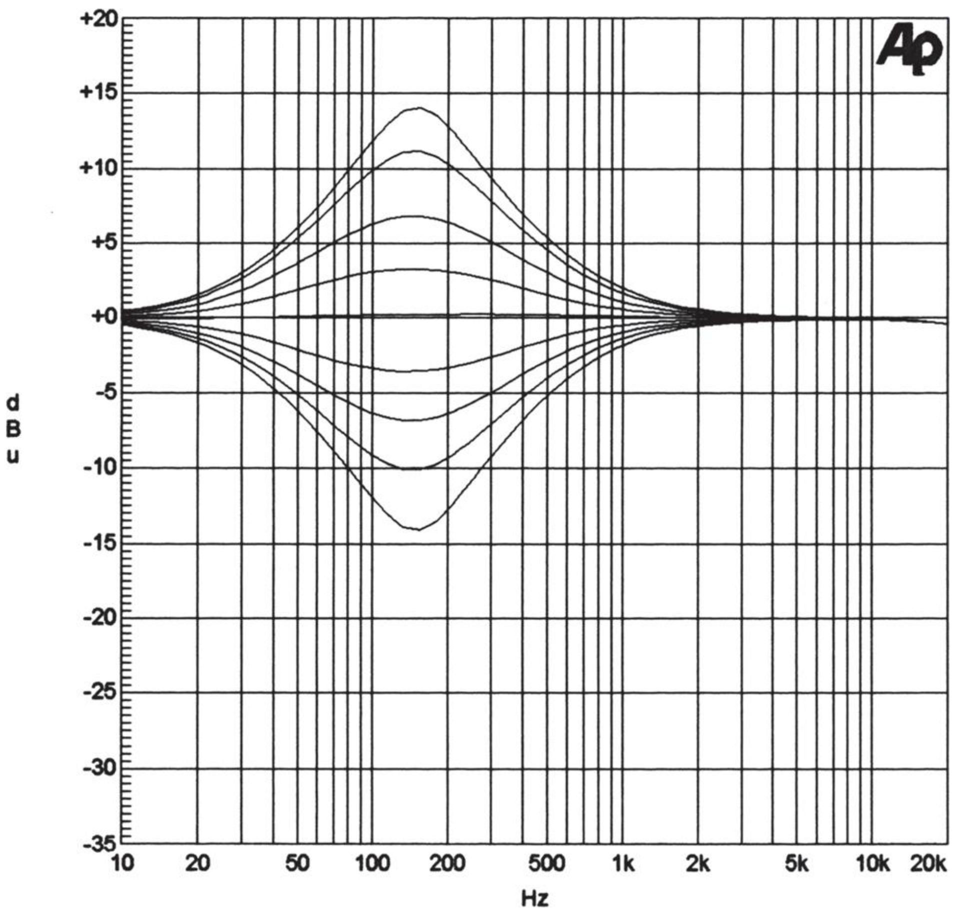


Diagram 4 displays the curves of the LMF filter.

图4显示了LMF滤波器的曲线。

Various cut and boost settings at 150 Hz.

在150赫兹的各种削减和提升设置。

Again the proportional-Q characteristic is clearly visible.

同样，proportional-Q 的特征也是清晰可见的。

Copy master: recall settings

Artist:

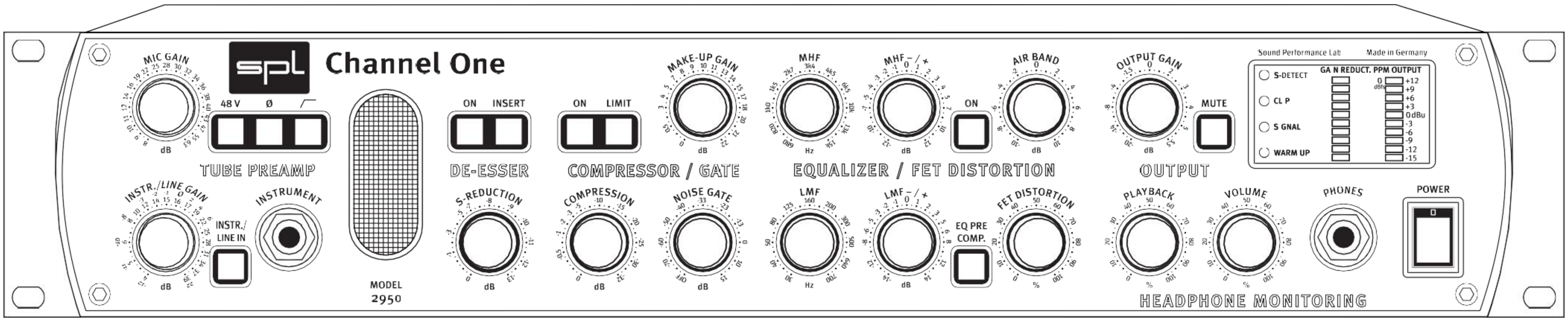
Engineer:

Album/Gig:

Track(s)/Groups:

Title:

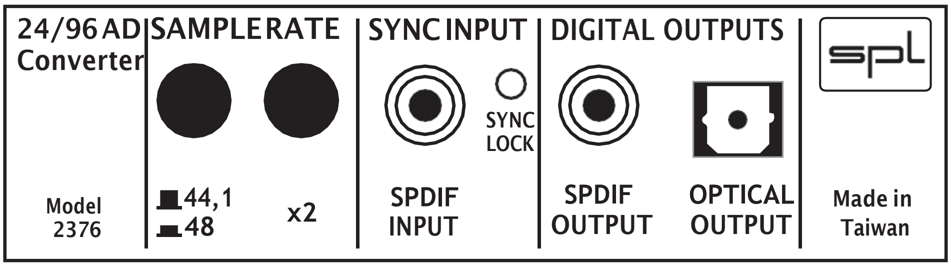
Date:



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

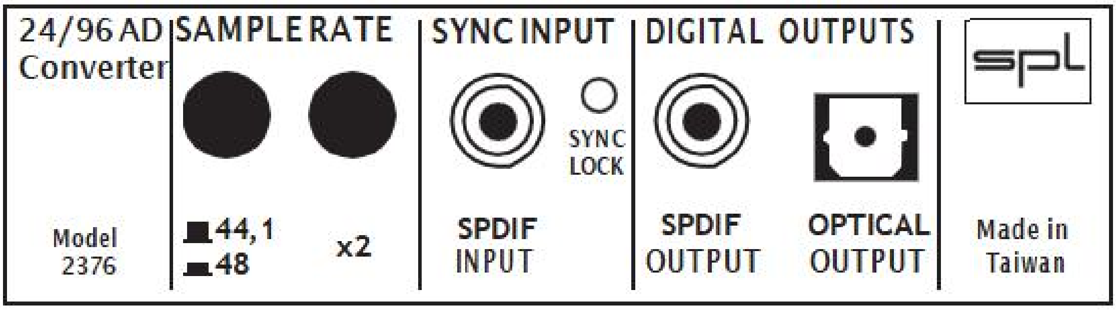
We think a good part of the "warmth" that is commonly associated with vintage gear comes from transformers. With transformers the low end and lower mids sound rounder, full-bodied with more punch. The top end gets a silky touch and benefits from improved presence without sounding boosted. Reasons are 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.

请注意，你可以从所有经销商处订购带有可选设备的产品，即使他们只列出标准产品版本，例如在网上商店。在您下订单之前，请联系您的经销商或SPL。可选设备也可以在售后安装。

Channel One 的可用选项，型号为2950。

- 24位/96kHz的A/D转换器（用户可以安装）。
- Lundahl 输入和输出变压器（只能由合格的技术人员或SPL安装）。



24/96 AD转换器，型号2376

可选的转换器卡提供了一个数字输出。输出信号通过两个插座的S/P-DIF输出：一个RCA插座和一个光纤插座。该转换器提供24比特信号。所有常见的采样率都可以选择（见下文）。高精度的石英振荡器确保了干净、低抖动的主时钟。

采样率：A/D转换器允许你在44.1、48、88.2和96kHz这四个最常见的采样率中进行选择。44.1/48按钮可以选择两个基本采样率中的一个（输出：44.1kHz；输入：48kHz）。x2按钮将这些采样率加倍，分别选择88.2或96kHz。

数字输出：转换后的S/P-DIF信号被平行路由到RCA和光纤输出。该信号为专业格式，状态块中没有采样率数据。

同步输入：由于这是一个AD转换器，同步输入不是音频信号输入。SYNC INPUT允许你向转换器提供外部采样率。将你的主源（如DAW接口）的S/P-DIF输出连接到同步输入。AD转换器将自动切换到接收到的相同采样率。A/D转换器2376没有配备接受字时钟同步的功能。

当SYNC INPUT出现有效的同步信号时，黄颜色的同步锁定LED会亮起，转换器会自动同步到外部采样率。

为了防止干扰，当有外部时钟信号出现时，内部振荡器会自动禁用。如果同步信号不再存在（例如，在掉线的情况下），转换器会自动恢复到通过转换器的控制开关选择的采样率。

关于I/O变压器的信息

我们认为，通常与复古设备相关的 "warmth " 有很大一部分来自变压器。有了变压器，低音和中低音听起来更圆润，饱满，更有冲击力。高音部分听起来并没有增强感，反而会从中受益，音质更加有质感。原因是减少了奇数谐波（产生刺耳的高端印象），而且与电子级相比，有一个较慢的特性，导致更多的体积感。我们推荐变压器，特别是用于人声，而电子级可以更好地实现信号传输的最高精度（瞬态），但最终这是一个个人品味、应用或例如使用的麦克风的问题。

在SPL前置放大器或通道条中使用，输入变压器会增加约14dB的增益（取决于麦克风）。这必须添加到比例值中。额外的无源增益使整个单元在任何增益水平上都能永久地缓解。较高的增益水平对带状麦克风也有好处。这就是为什么输入变压器在前置放大器中更为重要，但为了从所有可能的声音效果和充分的操作安全中获益，输入和输出都应该配备变压器。