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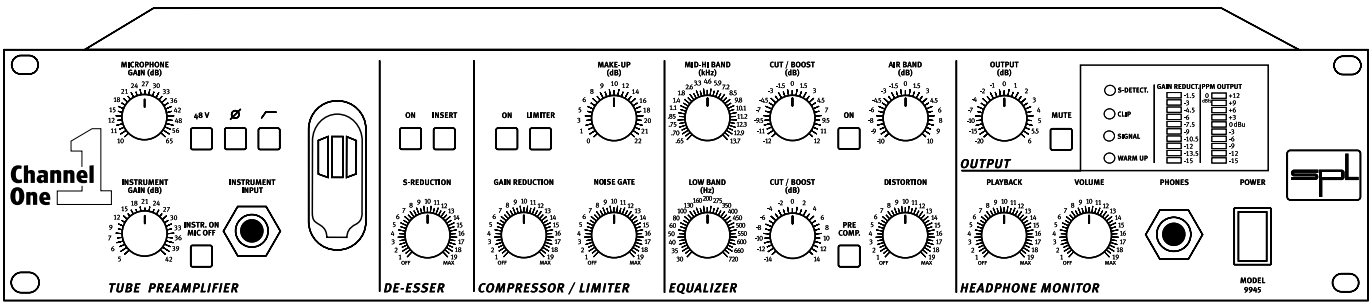
Manual



Channel One

Model 9945

Channel Strip

Version 1.1 - 5/2000

R & D: Ruben Tilgner

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Introduction

SPL is mainly known for the development of highly specialized audio-tools. Our philosophy, "one product for one task", is aimed at fast and simple operation in conjunction with high processing quality, to ensure highest musical performance.

With Channel One SPL have produced a complete channel strip which for the greater part is based on the processing concepts already successfully realized in other products. 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. The ususal recording day is to a high degree determined by a series of opposing time limits – the "highly paid" singer/speaker desires a quick recording; however, if technical preparation takes a long time because of unsuitable equipment, time will be lost, increasing the costs and souring the working environment. The Channel One in all cases however allows fast production without any loss of professional precision and diligence.

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

Principles

The Channel One has all necessary tools on board to prepare a recording for a digital recording system. It offers many possibilities for sound processing – the total bandwidth of subtle corrections to low-fidelity sounds is available.

The versatility and completeness of the unit permits its use in additional areas, over and above the purely "recording channel" facility, for instance in mixdown, as high quality single outboard EQ or compressor.

To maximize user friendliness and for clarity 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´s supported. More time remains to work creatively.

Great value was placed from the outset on high flexibility. An example of this is the 3 separate inputs for microphone, line-signals or instruments, each of which has been optimized to its function.

A twin triode tube is utilized in the process at 2 points – one immediately after the amplifier stage and the other at the end of the chain, so that the processed signal passes once again through the tube stage. This construction combines the advantages of the transistor pre-amplifier stage (high performance with minimal distortions and low noise levels) with the improved musical expression of the tone produced by tubes.

The microphone input can be optionally provided with Lundahl input transformers. The input transformers deliver a five times enhanced microphone level to the pre-amplifier, an amplification which reduces the equivalent load to the electronic pre-amplifier. The balanced outputs can also be equipped with Lundahl transformers, which deliver an even warmer and fuller sound.

Principles

A 24 Bit/96 kHz AD/DA module can optionally be provided to enable immediate digital recording. With this AD/DA converter a complete digital insert is also possible. An additional socket on the Channel One serves the purpose of delivering a further signal source to the AD converter.

In order to quickly determine all signal conditions, displays for level, gain reduction, S-detection, clip, warm-up and signal are combined and contained in a clearly defined display area.

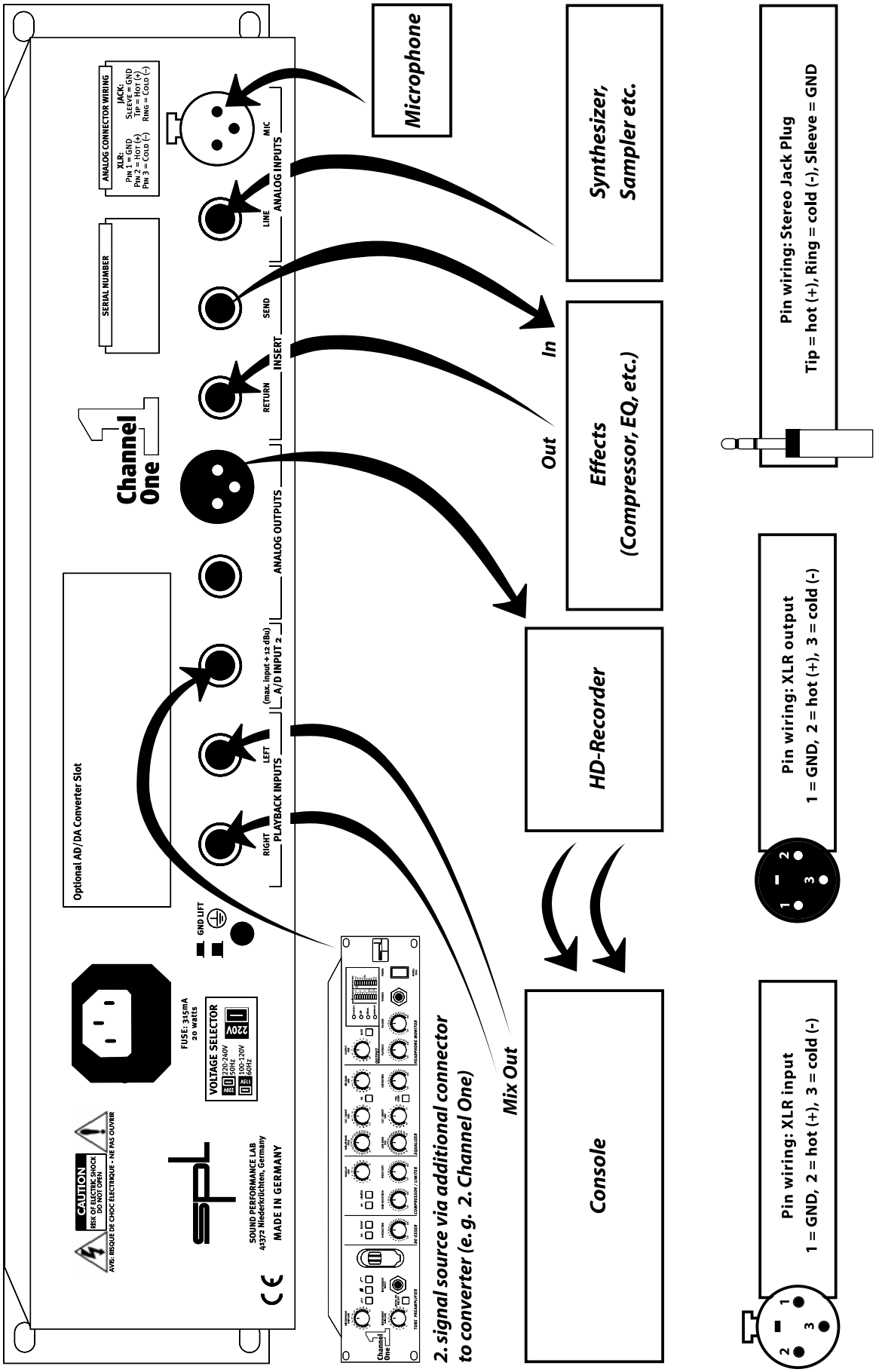
A special feature of the printed circuitry layout is the central star ground wiring scheme:

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, manufactured to SPL tolerances, supplies the equipment with the necessary voltages and forms the basis for a clean electrical supply to all parts of the circuitry.

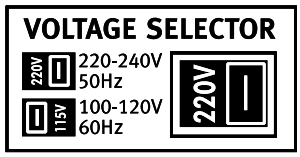
Hookup

Carefully select a place for setting up the Channel One. The unit should be situated away from heat sources and direct sunlight. Avoid installation in environments exposed to vibrations, dust, heat, cold or moisture. Keep the unit away from transformers or motors or any other unit that could generate large variations in power supply or cause electrical interferences. Do not install the unit in proximity to power amplifiers or digital processors. You may consider placing it in a rack containing other analog gear. Such placement can prevent interference from Word Clock, Smpte, MIDI, etc.

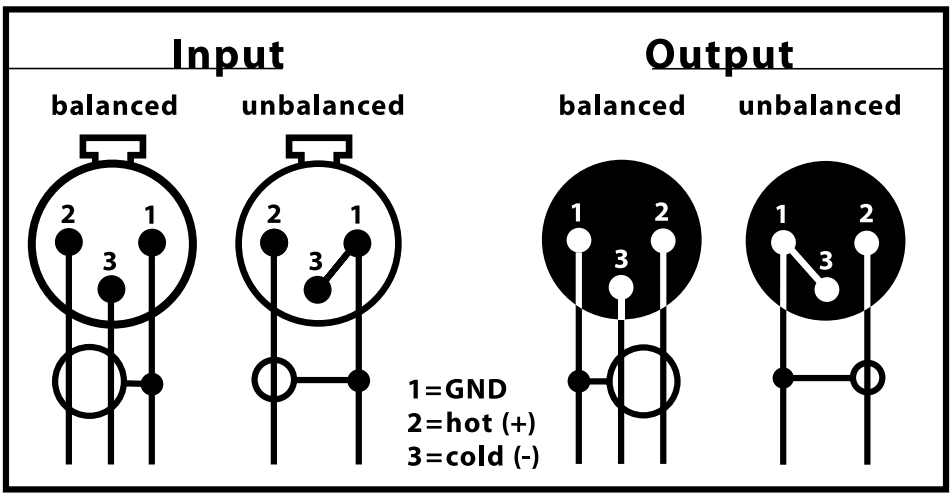
- Do not open the case. You may risk electric shock and may damage your equipment.
- Leave repairs and maintenance to a qualified service technician. Should foreign objects fall inside the case, contact your authorised dealer or support person.
- To avoid electric shock or fire hazards do not expose your unit to rain or dampness.
- In case of lightning unplug the unit. Please unplug the cable by pulling on the plug only; never pull on the cable.
- Never force a switch or knob.
- To clean the case use a lint-free cloth. Avoid cleaning agents as they may damage the chassis. Manufactured in standard 19" EIA format, it utilises two rack units.
- Please support the back of the unit whenever it is being mounted into a 19" rack (especially important when touring).


Microphone
Synthesizer, Sampler etc.
Effects (Compressor, EQ, etc.)
HD-Recorder
Console
Out
In
Mix Out
Channel One
Optional AD/DA Converter Slot
ANALOG CONNECTOR WIRING
XLR: Pin 1 = GND Pin 2 = Hot (+) Pin 3 = Cold (-)
JACK: Sleeve = GND Tip = Hot (+) Ring = Cold (-)
SERIAL NUMBER
MIC
LINE
ANALOG INPUTS
SEND
RETURN
INSERT
ANALOG OUTPUTS
(max. input +12 dBu) A/D INPUT 2
LEFT
RIGHT
PLAYBACK INPUTS
GND LIFT
FUSE: 345mA 20 WATTS
VOLTAGE SELECTOR
220V
220-240V 50Hz
100-120V 60Hz
CAUTION
RISK OF ELECTRIC SHOCK DO NOT OPEN
ATTN: RISQUE DE CHOC ELECTRIQUE - NE PAS OUVRIR
SOUND PERFORMANCE LAB
43372 Niederkrüchten, Germany
MADE IN GERMANY
2. signal source via additional connector to converter (e. g. 2. Channel One)
Pin wiring: Stereo Jack Plug
Tip = hot (+), Ring = cold (-), Sleeve = GND
Pin wiring: XLR output
1 = GND, 2 = hot (+), 3 = cold (-)
Pin wiring: XLR input
1 = GND, 2 = hot (+), 3 = cold (-)

Again, while Channel One's housing is EMV-proof and protects against HF-interference, placement of the unit is very important since it amplifies microphone signals as well as other unwanted signals. Before connecting the Channel One or any other equipment turn off all power. Adjust the voltage setting on the back so that it corresponds with the power conditions.



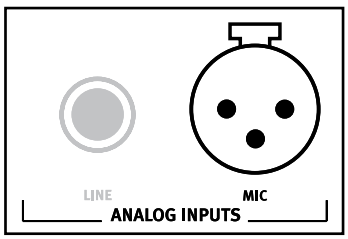
The following graph shows the correct wiring for connecting unbalanced signals to the balanced XLR connectors:



Unbalanced signals with mono jack plugs may easily be connected to the balanced jack connectors without level differences (see "Analog Outputs" on page 8).

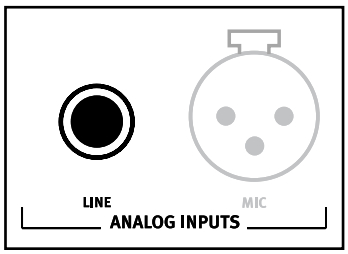
Mic connector

The Mic connector is used to plug in microphones of any type (dynamic, condenser or tube microphones etc.). If 48 V phantom power is required for some mics, switch on the 48 V button. For further information please read "48 V-phantom power" on page 9.



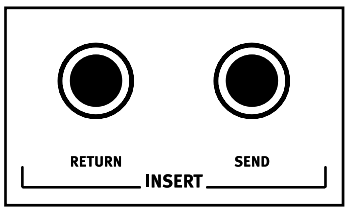
Line connector

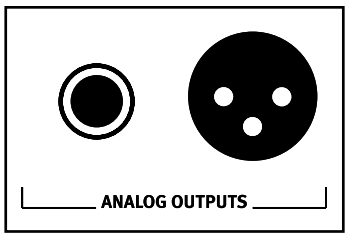
The balanced Line connector serves to connect line level equipment. It is recommended to route the Line input to a patchbay. This allows easy and fast selection of various line signal sources.



Insert connectors

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/Limiter. 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/Limiter/EQ sections for further processing.

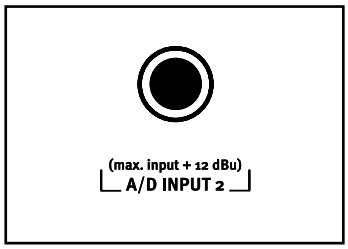




Analog Outputs

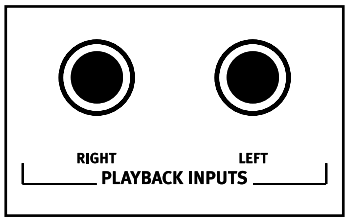
The Analog Outputs deliver balanced output signals. Lundahl output transformers can be equipped optionally.

Since both connectors are working in parallel, unbalancing one connector also unbalances the other one. If for example a mono jack plug is connected to the jack connector, the XLR connector is switched to unbalanced operation as well.



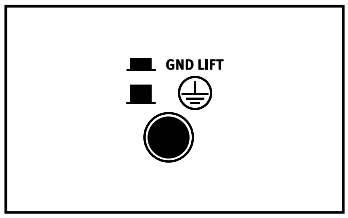
A/D Input 2

This connector serves to feed a further signal to the optional AD/DA converter. Two different signals can be converted at the same time. If no signal is fed to the A/D Input 2 connector, the output signal of the Channel One is routed to both channels of the converter. The maximum input level should not exceed +12 dBu to avoid clipping of the converter (+12 dBu represents the digital full scale level, 0 dBfs).



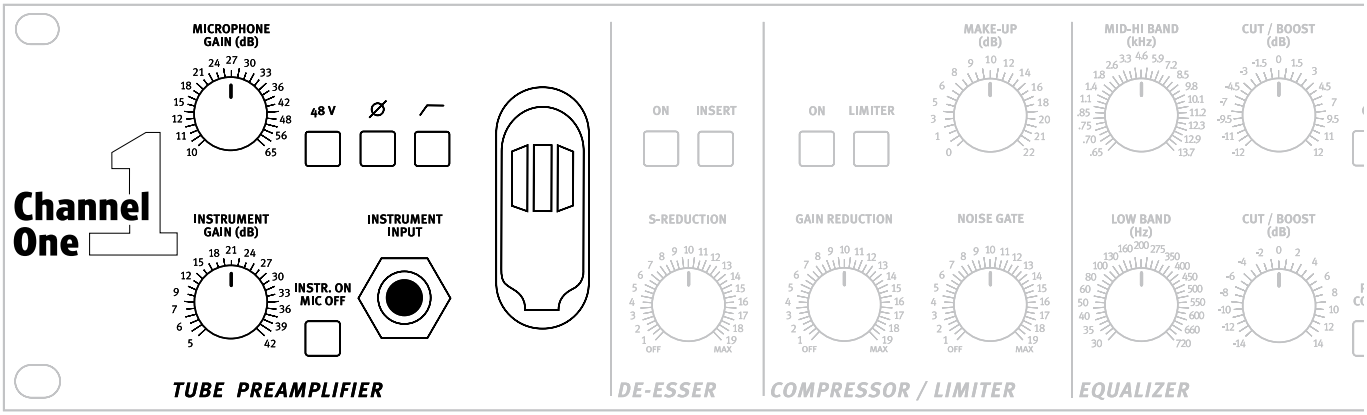
Playback Inputs

The playback signal is connected to the unbalanced Playback Input jacks to direct it to the Headphone Monitor. 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.



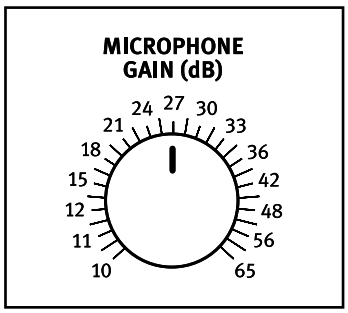
GND Lift

The GND Lift switch separates internal ground from chassis ground. The switch should be activated to eliminate ground loop humming which may occur if the Channel One is connected to units with another ground potential.



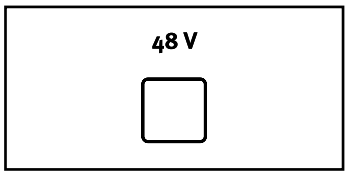
Microphone Gain

The Microphone Gain control determines the preamplification of the microphone signal. The preamplification values extend up to + 65 dB. If Lundahl input transformers are fitted the scale values are to be increased by + 14 dB. Please refer to "About levelling" on page 10 for further information.



48 Volt phantom power

The 48 Volt phantom power in the Channel One serves to supply condenser microphones which are equipped with in-built preamplifiers. A precise construction and disturbance free electrical supply are the main requirements for their trouble free operation. In the Channel One the voltage is maintained at a precise 48 V and delivers a maximum current of 14 mA. This is sufficient for all types of microphones.

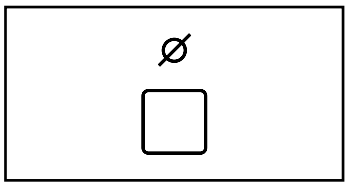


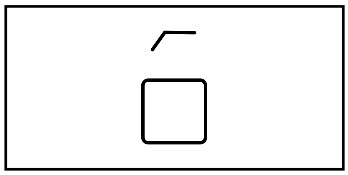
WARNING: All microphones with balanced, ground free output (including tube microphones) can be operated with phantom power switched on. The following procedure is to be adhered to: Firstly connect the microphone to the Channel One, then switch on the phantom power – you can now commence work. When recording has been completed firstly switch off the phantom power then wait 30 seconds before disconnecting the microphone from the Channel One. This allows residual voltages to be discharged.

Phantom powering is only used with condenser microphones. With any other type of microphone it is to be switched off ! An unbalanced microphone is not to be used with phantom power switched on!

Phase Reverse

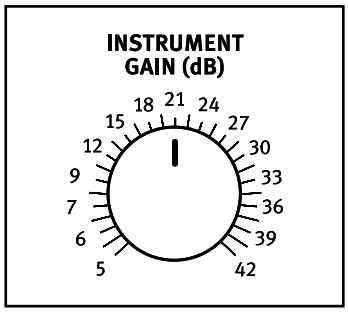
The phase inversion function reverses the polarity of the microphone signal. When the button is pressed the phase is rotated through 180°. The phase inversion function is often useful, for instance to correct a headphone monitor signal which is possibly wrongly phased. A singer/speaker can actually hear himself during recording as well as over the headphones. Incorrect polarity leads to an unnatural tone and to drastic tonal changes, if the distance to the microphone is varied. We recommend that you check the polarity and correct it if necessary before commencing recording.





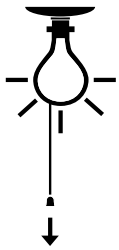
Highpass

The Highpass filter is used to eliminate disturbing low frequencies. These disturbances could impair the following processing or AD conversion. The cut-off frequency at 50 Hz avoids influences to vocals. The roll-off is 12 dB/octave.



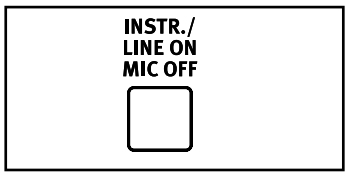
Instrument Gain

The Instrument Gain control determines the preamplification of the inputs Line and Instrument. The signals can be preamplified from + 5 dB up to + 42 dB. By making use of the balanced line input the amplification is reduced by about 18 dB to enable even very "loud" signals to be processed. For further information please read the next section "About levelling".



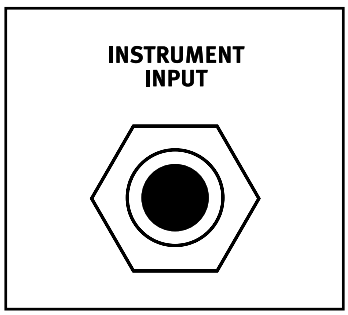
About levelling

For perfect levelling of the preamplifier firstly switch off all other modules (De-esser, Compressor/Limiter, 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 +6 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 preamplifying value is to be reduced accordingly.



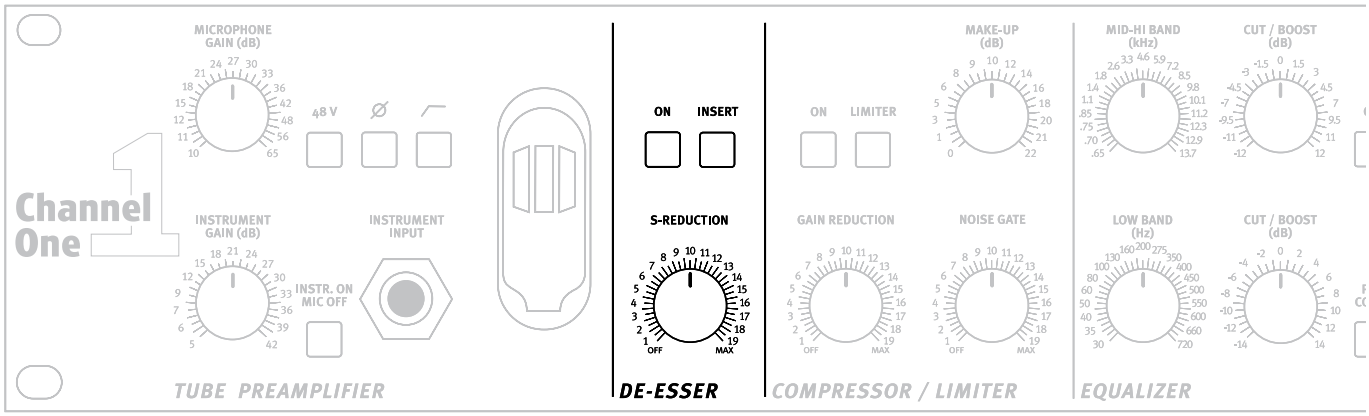
Instrument/Line On – Mic Off

This button allows selection of the input source. The microphone signal is available for processing when the button is not pressed; when pressed the Instrument/Line signal is activated. As long as the Instrument Input at the front of the unit is not in use the Line signal option on the back of the unit is automatically available.



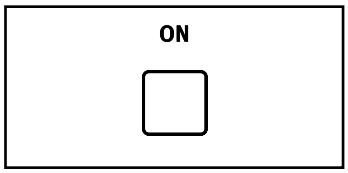
Instrument Input

The Instrument Input on the front of the unit is designed for the connection of electric as well as acoustic guitars with their own pick up etc. This input is of high impedance and designed for high amplifications. It is possible to connect instruments with line levels, such as keyboards, samplers, drum machines to the "Instrument Input" to permit these to be processed quickly with the Channel One, however it is preferable to connect line level equipment to the "Line Input" at the back of the unit, ideally via a patch bay. Line Input provides a more stable level (see section "Instrument Gain") while the connection via a patch bay offers the most flexible routing method combined with the advantage of insensitivity to disturbance characteristic of balanced wiring.



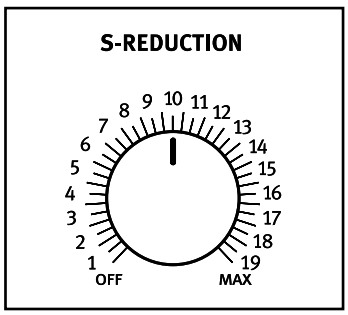
On

The first module behind the pre-amplifier stage is the De-Esser, which immediately removes disturbing S-sounds when required. The De-Esser module is activated when the button is on. The S-Detect. LED in the display will show that S-sounds are being detected regardless of the selected S-Reduction value, in other words even when the button is switched off detection is still shown in the display.



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 "Technical information ...") the processing is more intensive with extreme S-sound levels than with those of lower levels. After processing the output signal has a consistent S-sound level.



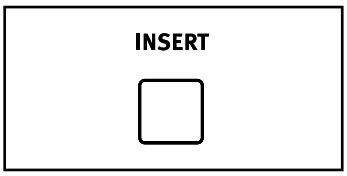
Technical information regarding the De-Esser

In contrast to common de-essers that influence a frequency band of about 2 octaves with compressor techniques the Auto-Dynamic De-Esser utilizes filters that process only the reducible "S-frequencies" but do not interfere with the remainder of the spectrum. The S-frequencies that lie in the unpleasant range are automatically recognized, the phase is inverted and mixed with the original signal. In this manner the disturbing frequency is quenched and the hissing noise reduced. 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 entire 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 determinate S-portion are not processed whereas those that are too loud, or do not effectively contribute to the sound, are reduced – the character of the voice remains unchanged.

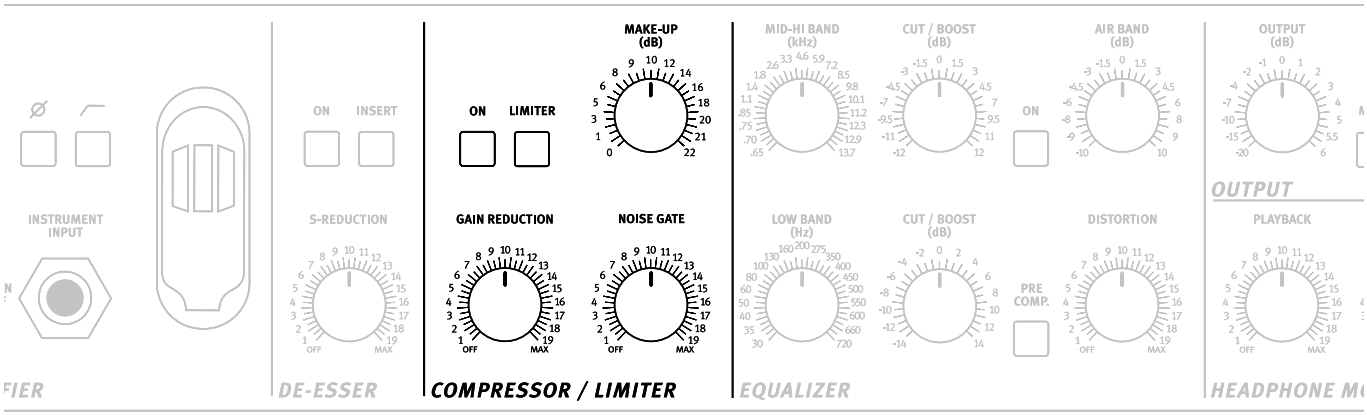
A further specialty is the integrated auto-threshold-function which makes processing independent of the input level. Even when the speaker or singer does not maintain a constant distance to the microphone, processing is retained at the pre-set S-reduction value. Conventional systems are dependent on the input level and work more intensively as the distance to the microphone is reduced.



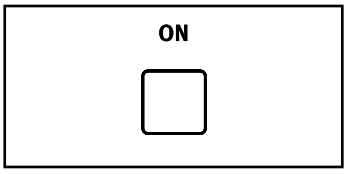
The insert button activates any attachments of external equipments for effects, equalizing or compression that are connected to the Insert Send/Return loop on the rear of the Channel One. This binds them into the signal chain thereby enhancing the processing capabilities ad infinitum.

The Insert point is located between the De-Esser and compressor. This allows the ability to use the pre-amplifier stage/de-esser combination of the Channel One separately from the compressor/EQ combination which, because in this manner the Channel One can be used as 2 independent units, broadens the range of uses enormously.

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.



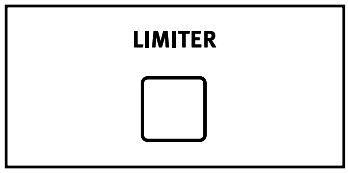
On



The On button activates the Compressor/Limiter/Noise Gate module. At the same time the Gain Reduction display shows the processing intensity (see section "Gain Reduction" on page 13).

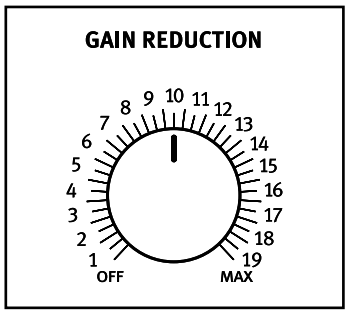
Usually the signal flow follows the design of the Channel One and for this reason the input signal normally arises from the De-Esser or, when activated, from the Insert. However with the Pre Comp. switch function of the Equalizer module the Compressor/Limiter can be switched behind the Equalizer. This allows it to be used either as an final compressor or limiter. (Further information in the section "Pre Comp." on page 15).

Limit



The Limit button switches the Compressor to limiter mode. The Gain Reduction 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 intercepted. It is therefore advisable when modulating a subsequent unit that a headroom of 2 to 4 dB remains. Peak limiters have a system-based disadvantage in that audible distortions are heard considerably sooner.

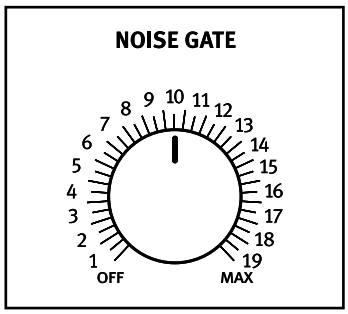
Gain Reduction



The Gain Reduction control sets the intensity of compression. Turning the control clockwise increases compression. The working area spans between + 20 dB (counter clockwise limit) and -50 dB (clockwise limit).

The compressor applies the so-called "soft-knee" characteristic, which means that quiet passages are processed at a lower compression ratio than louder passages. At maximal compression it operates with a ratio of 1:2.5 – very effective dynamic limits are achievable when inconspicuous characteristics are to be processed. The exact development of the compressor curve is portrayed in the diagram 1 on page 24. When setting the compression rate the Gain Reduction display in the display field 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.

Noise Gate

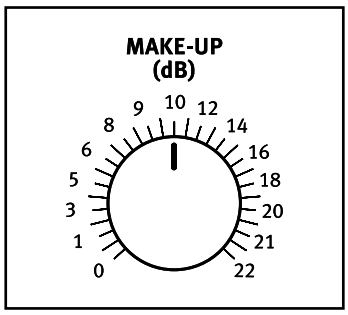


The Noise Gate control monitors the noise gate by which soft disturbances are reduced during signal pauses. When turned fully counter clockwise the noise gate is switched off. By turning the control in a clockwise direction the threshold value increases. This means that the Noise Gate closes relatively earlier.

The processing span of the Noise Gate is between –100 dB (gate control turned fully counter clockwise) and + 18 dB (gate control turned fully clockwise). The Noise Gate is therefore operable over the complete dynamic range.

With a hysteresis of 6 dB the noise gate functions very stably: the point at which the Noise Gate opens lies 6 dB above the point at which the Noise Gate closes again. Definite closure and opening is therefore assured – the most feared characteristic of "fluttering" is excluded. Even critical signals are cleanly processed.

The release-time setting takes place automatically. The automation, which depends upon the program, adjusts itself to the release time of the musical piece thereby ensuring optimal (undetectable) opening and closing.



Make Up

With the Make Up control the level reduction caused by compression or limiting can be restored. With assistance of the Gain Reduction display in the display field setting the Make Up control is very easy: If the maximal reduction value caused by the loudest tone amounts to -9 dB, for instance, the Make Up 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.

Technical Information regarding the Compressor/Limiter

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 (distortion and artifacts).

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/Limiter 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.

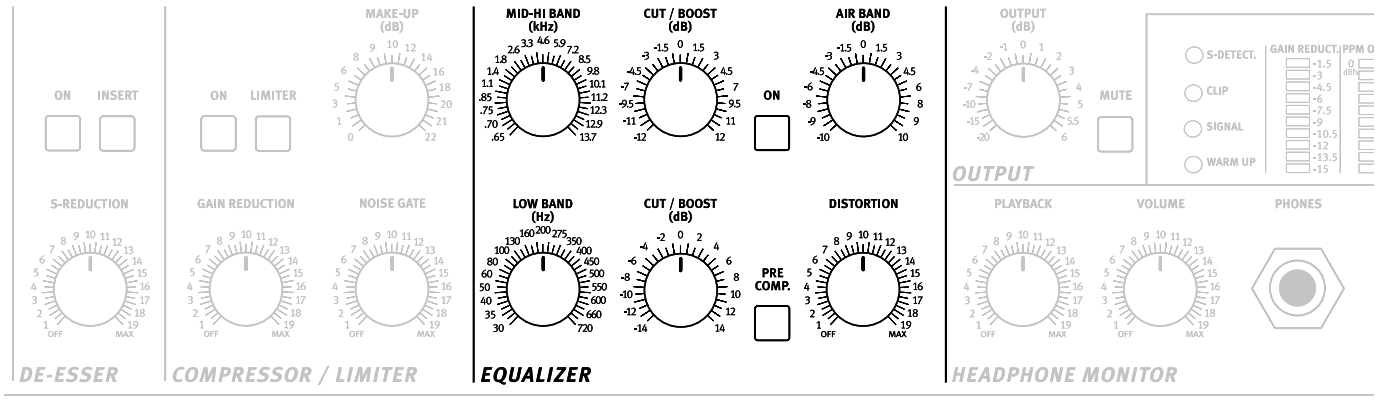
info

Even the control of the release time is dependent on the input signal. 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/Limiter in the Channel One: the Double-VCA-Drive®. Two That 2181 VCAs are utilized, one 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 are removed. The product of the differential of both signals (simply stated) 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 run the danger of operating in the saturation range – this would lead to offset noises, audible as clicks or pops.

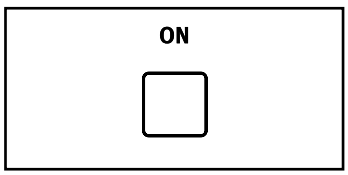
The Double-VCA-Drive® circuitry overall displays vastly improved distortion values so that a distinctly clearer and more transparent sound impression is achieved than with conventional circuitry. Voices and instruments are given a considerably more natural and dynamic timbre whereas "muffled" tones are not audible.

The Compressor/Limiter characteristics are portrayed on page 24.



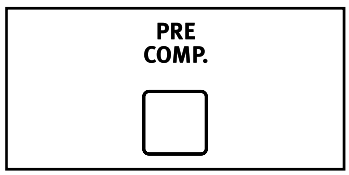
On

The On button inserts the Equalizer module into the signal path. Under normal circumstances the input signal comes from the compressor. With the Pre-comp button the Equalizer can be switched in before the Compressor/ Limiter so that the input signal is received from the De-Esser or Insert.



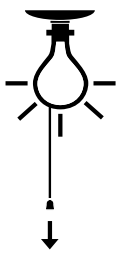
Pre Comp.

The Pre Comp. button reverses the sequence of Compressor/Limiter and Equalizer: When the button is pressed the Equalizer operates in front of the Compressor/Limiter; when not pressed the succession remains unchanged. This function permits very flexible operation with the Channel One when it is necessary to resolve recurring problems or to create special sounds.



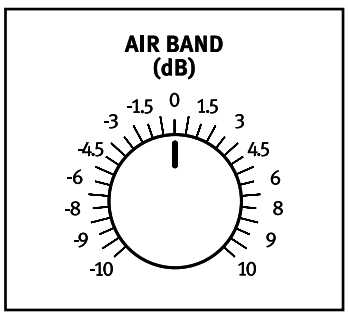
The following example describes when the Equalizer (EQ) is to be switched in front of the Compressor/Limiter.

When over-accentuation of instruments or voices is registered within certain frequency ranges these ranges should first be reduced with the EQ. The signal can subsequently be compressed more easily. If not done in this sequence the compressor would react very strongly to these ranges; subsequent equalization would mean that the compression would be clearly audible (the problem frequencies would then be too soft). A further sensible application of the Pre Comp. function is the use of the compressor module as a limiter to maintain a stable output level. If the EQ was to be used again after limiting it could not be guaranteed that the output level would not alter.

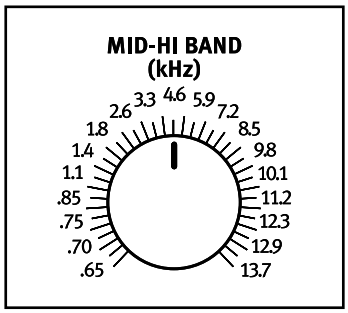


Air Band

The high frequency filter in the equalizer module is described as the "Air Band" and serves the processing of the frequency range of 2 and 20 kHz. 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 characteristics of the Air Band filter are shown in diagram 2 on page 24.

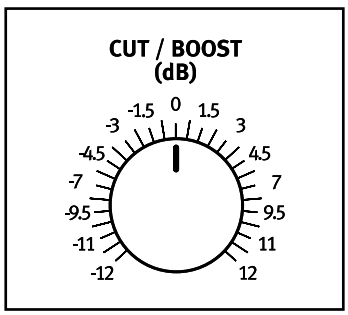


The "soft" and natural tonal property, characteristic of the coil-capacitor filter, lends itself extremely well to provide clarity to vocals in the upper frequency range thereby improving their presence. On the other hand harsh sounds can be lent a more pleasant sound characteristic through damping.



Mid-Hi Band

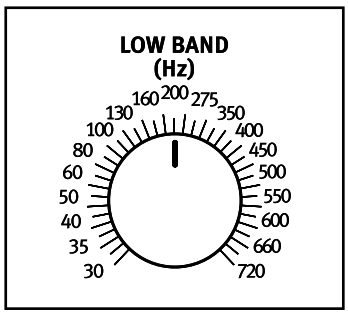
The center frequency of the semi-parametric mid high frequency filter is set with the Mid-Hi Band control. The frequency range can be set between 650 Hz and 13.7 kHz so that this filter covers a range of 4.5 octaves and can be equally employed in the lower mid as well as the high range.



Cut/Boost (Mid-Hi)

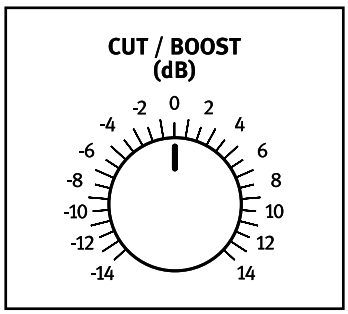
The Cut/Boost control determines the boost, or cut of the Mid-Hi filter; the maximum values lie between +/- 12 dB. The Mid-Hi filter utilizes the proportional-Q-principle. In other words the bandwidth is dependent on the selected boost or cut. The higher the boost or cut values are set, so the bandwidth becomes narrower; by low boost or cut values the bandwidth increases (the exact curve of the Mid-Hi filter can be seen in diagram 3 on page 25). 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.

This 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.



Low Band

The center frequency of the half-parametric bass filter is set with the Low Band control. The adjustable frequency range lies between 30 Hz and 720 Hz so that this filter covers a range of about 4.5 octaves, allowing it to be used from the deepest bass to the lower mid range. This together with the Mid-Hi filter ensures that the entire frequency spectrum is covered.

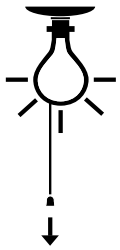


Cut/Boost (Low)

The Cut/Boost control determines the boost or cut of the Low Band filter; the maximum values lie between +/- 14 dB. The Low Band filter also operates to the proportional-Q-principle, in other words the bandwidth is dependent on the selected boost or cut. With the Low Band filter the factor with which the relationship of the boost or cut values, in relation to the bandwidth, is determined lies somewhat higher than with the Mid-Hi filter. The bandwidth is therefore marginally narrower at maximum boost than with the Mid-Hi filter. The exact curve of the Low Band filter is shown in diagram 4 on page 25.

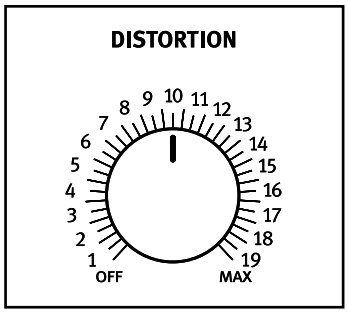
The Low Band filter can be applied in many ways. Examples are; to accentuate the fundamental sound of a voice, to cut "boom frequencies" and for placement of bass emphasized instruments such as bass guitar, bass drums- or synthesizers during recording or subsequently when mixing etc.

Recommendation on frequency settings: To find the frequency which is to be processed as quickly and accurately as possible the Cut/Boost control should firstly be adjusted to the maximum position. Subsequently the relevant frequency should be sought. Following this the required boost or cut can be set with the Cut/Boost control. 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.



Distortion

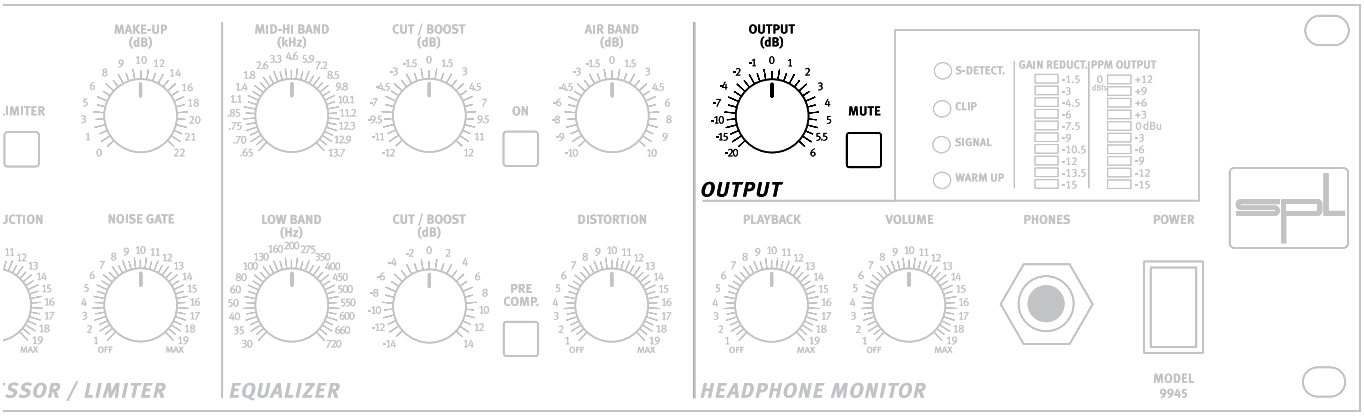
The Distortion control offers the capability of applying distortions to signals. The distortions are infinitely variable from Off through to distinctly perceptible harmonics. The distortion stage is located in front of the equalizer so that the even newly created spectrums can be processed with the EQ.

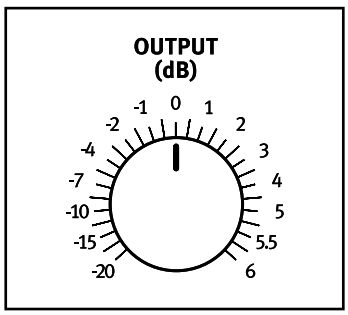


The overmodulated field-effect transistor which forms a part of the distortion circuitry has a similar characteristic curve as a tube and sounds distinctly "warmer" than a pure diode- distortioner.

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

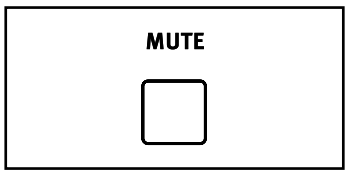
IMPORTANT: To avoid exasperation during recording it is recommended that the EQ controls, in particular the Distortion control, are initially set to Off or 0! If not, tonal changes will occur immediately and furthermore, in the case of the Distortion control, additional distortions.





Output

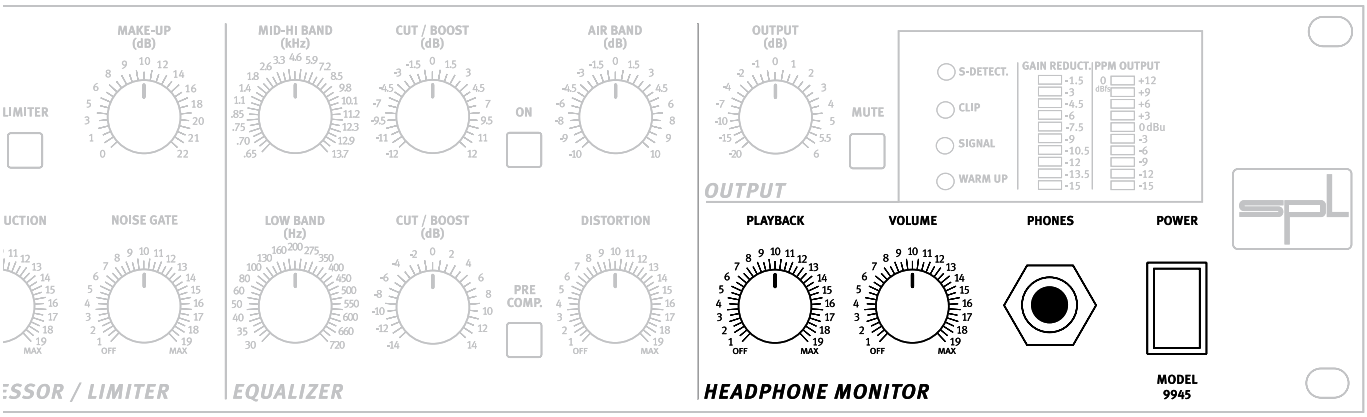
The outgoing signal can either be dampened to –20 dB or further amplified by +6 dB with the Output control to provide optimal drive to the subsequent units or the optional AD/DA converter. The individually selected output level is shown on the PPM-Output display in the display field. Before a recording commences the Output control should be set to 0: The uninfluenced values from the Output control are then legible and available for adjustment of the pre-amplifiers levels.



Mute

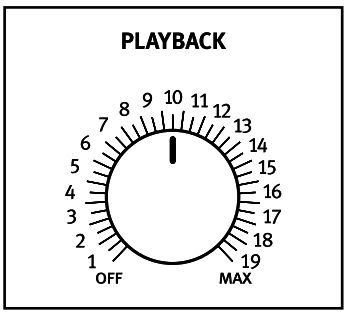
The Mute switch mutes the output signal; when activated the PPM-Output display does not show any values. An instance of a sensible application could possibly be when the output signal of the Channel One, together with the playback signal, are reproduced via the studio monitors during a recording session.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

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



Playback

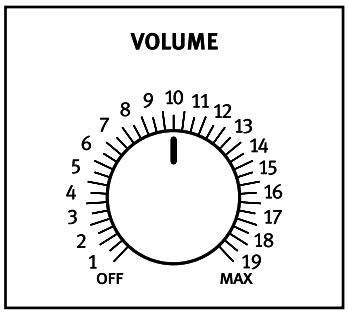
The Playback control regulates the volume of the playback signal which is passed to the musician. There are two methods of passing the mono playback signal: The first is to pass the music to both ear pieces of the headphone in which case "Playback Input Left" must be connected. On the other hand some musicians want to hear the playback signal through only one ear piece so they can hear "directly" with the other ear. In this instance connect "Playback Input Right" and set the Volume control to Off.



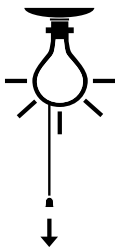
Volume

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

With the Playback and Volume controls an individual mix for the headphones is achievable: it is advisable to ask the musician before recording starts whether he can hear himself and the playback adequately. The best conditions for good intonation, stemming from a relaxed working environment, prevail. Another practical use of the headphone monitor module is to monitor the signal quality directly to locate and eliminate possible interference rapidly.

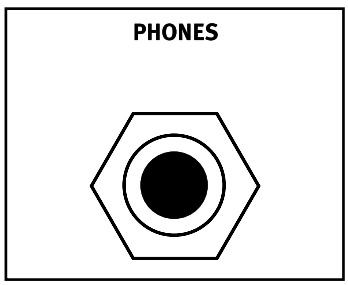


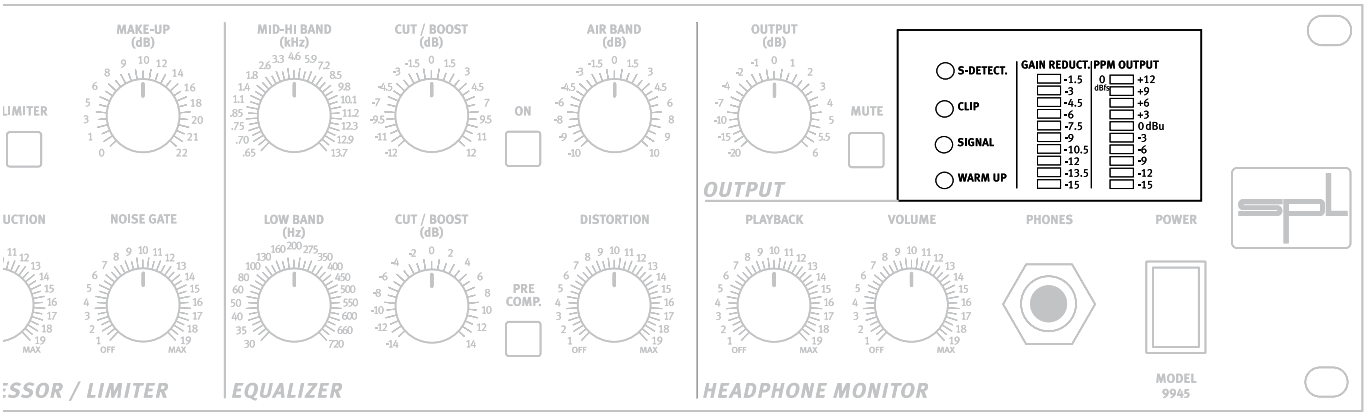
Recommendation on using the Headphone Monitor: When working with hard disc systems or digital mixing consoles latency may be present. Flanging or phasing effects occur if the musician receives the monitor signal with a time lag. It is therefore recommended, to obviate latency, that the monitor signal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

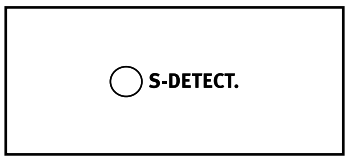


Phones

The Phones socket is provided for the connection of stereo headphones. The high quality headphones amplifier is of low impedance and has low distortion values so that all current types of headphones can be connected to monitor the signals with highest possible audio fidelity.

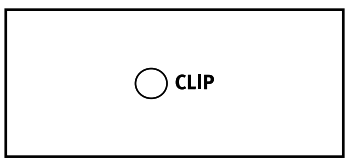






S-Detect.

The S-Detect. LED shows when S-sounds have been detected. It is only active when the De-Esser is switched on and is independent from the selected S-Reduction setting.

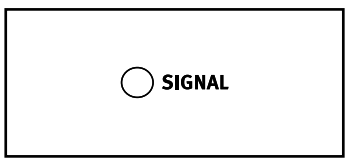


Clip

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

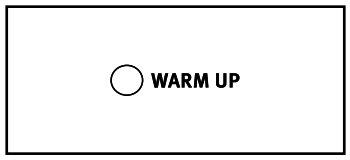
At all relevant points of the signal flow the display gets read off: behind the Pre-amplifier, behind the Compressor/Limiter, behind the EQ and behind the Output control. All possible causes for overload can be directly checked (overdriven Microphone/Instrument/Line Gain, an excessive Make Up value in the Compressor/Limiter, too much boost in the EQs or too high output level).

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



Signal

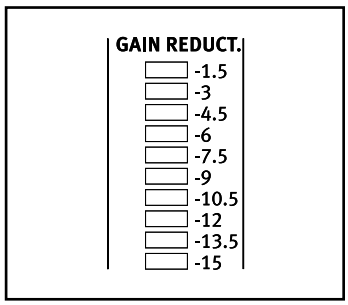
The Signal LED illuminates when a signal is being received at the Pre-amplifier. This provides a quick method of checking that a signal source is correctly connected. All levels above -50 dB are covered.



Warm Up

The Warm Up LED gives an indication regarding the warm up phase of the tube stage. When the LED is extinguished the Channel One is ready for operation; it is possible that before this has occurred the output signal is low and sounds distorted.

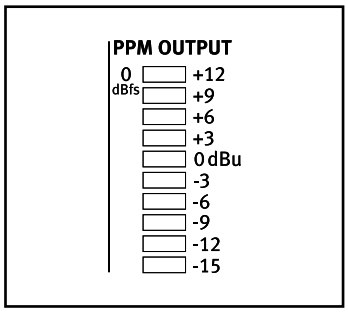
Gain Reduct.



The Gain Reduct. display provides information about the processing being undertaken with the Compressor/Limiter or the Noise Gate. The level change, perhaps caused by compression, are scaled in 1.5 dB steps. The display is activated when the Compressor/Limiter module is switched on.

Noise Gate operation is visible because all Gain Reduct. LEDs illuminate when the signal level lies under the gate threshold setting.

PPM Output



The PPM Output display shows the peak reading of the output level (calibrated to 0 dB) and is present at the analog outputs on the rear of the unit. This display also serves to the pre-amplifying Gain. The value "0dBFS" marked on the left side represents the maximum level of the optional AD/DA converter which should not be exceeded. (Further information is given in the directions to the AD/DA converter).

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

Power Supply

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

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

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

The supply voltage can be set to 230 V/50 Hz or 115 V/60 Hz . Check your country's power requirements for the appropriate setting. An AC power cord is included to feed the IEC-spec, 3-prong connector. Transformer, AC cord and IEC-receptacle are VDE, UL and CSA approved. The main fuse is rated at 315mA.

Chassis ground and AC ground can be physically disconnected by the "Ground Lift" switch (GND LIFT). This helps to eliminate hums.

Measurements

Microphone Input

Frequency Response: ..... 10 Hz-100 kHz
(100 kHz = -3 dB)

Common Mode Rejection: ..... 1 kHz: -80 dB / 10 kHz: -78 dB
(@ -20 dBu)

THD & N: .....

Amplification:	A weighted:
20 dB	-97,1 dBu
40 dB	-91,1 dBu
65 dB	-69,4 dBu

Dynamic Response: ..... 118 dB

Line / Instrument Input

Frequency Response: ..... 10 Hz-100 kHz
(100 kHz = -3 dB)

Common Mode Rejection: ..... 1 kHz: -80 dB / 10 kHz: -78 dB
(@ 0 dBu, only Line Input)

THD & N: .....

Amplification:	A weighted:
5 dB	-99,4 dBu
20 dB	-97,2 dBu
42 dB	-79,4 dBu

Input Impedance: ..... Line: 20 kOhm / Instrument: 1 MOhm

Max. Input Level: ..... Line: +22 dBu / Instrument: +14 dBu

Dynamic Response: ..... 119 dB

Outputs

Max. Output Level XLR / Jack: ..... +20 dBu

Output Impedance: ..... <50 Ohm

Dimensions

Housing ..... Standard-EIA-19"/2 U
... 482 x 88 x 210 mm

Weight ..... 4,15 kg

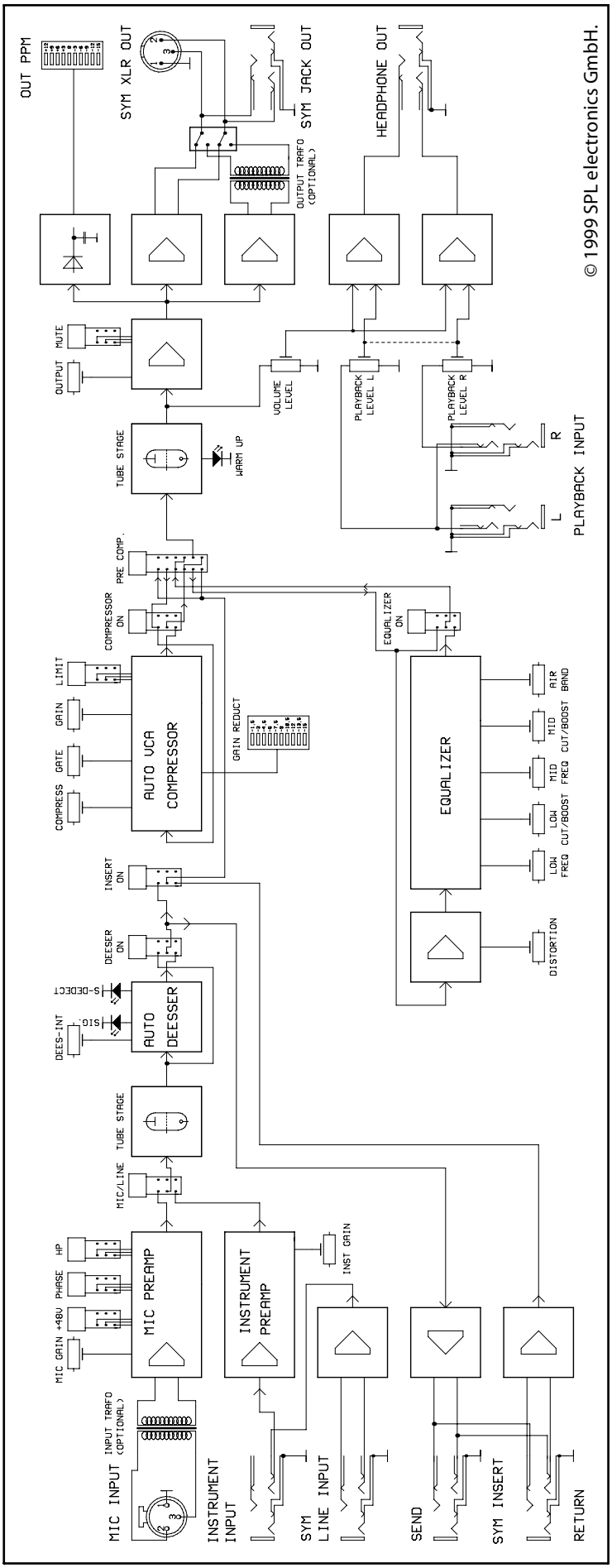

OUT PPM
SYM XLR OUT
SYM JACK OUT
OUTPUT TRAFO (OPTIONAL)
HEADPHONE OUT
MUTE
OUTPUT
VOLUME LEVEL
PLAYBACK LEVEL L
PLAYBACK LEVEL R
R
L
PLAYBACK INPUT
TUBE STAGE
WARM UP
PRE COMP.
COMPRESSOR ON
EQUALIZER ON
LIMIT
GAIN
GATE
COMPRESS
AUTO VCA COMPRESSOR
GAIN REDUCT
EQUALIZER
AIR BAND
MID CUT/BOOST
MID FREQ
LOW CUT/BOOST
LOW FREQ
DISTORTION
INSERT ON
DEESER ON
S-DEDECT
SIG.
DEES-INT
AUTO DEESSER
MIC/LINE
TUBE STAGE
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
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Diagram 1 shows various curve characteristics for the Compressor/Limiter

The reference curve A displays the relation between input and output.

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

Curve C portrays the limiter's curve characteristics.

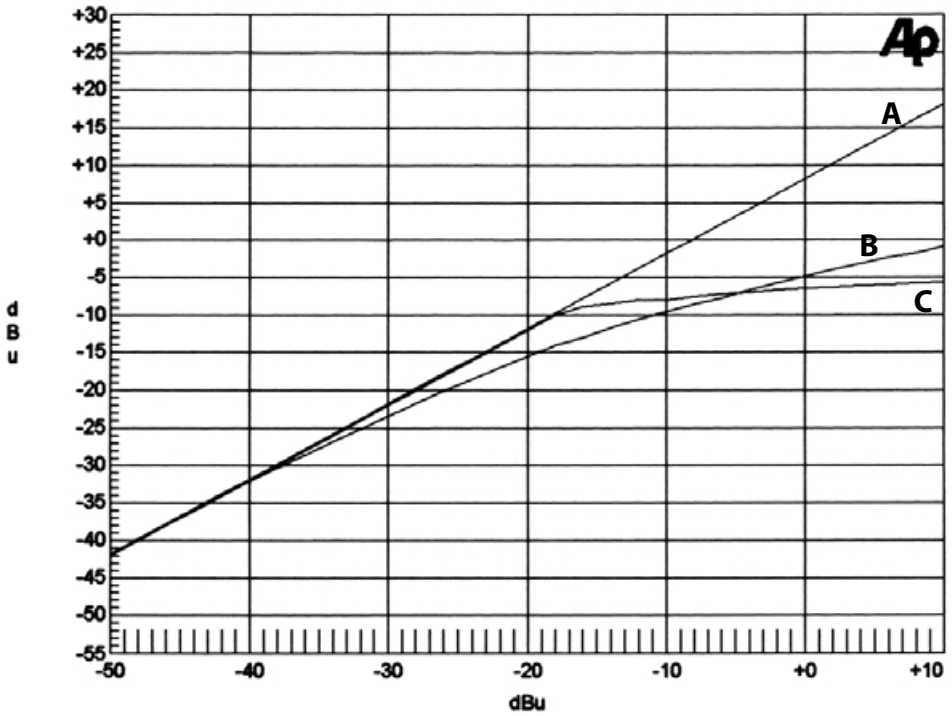
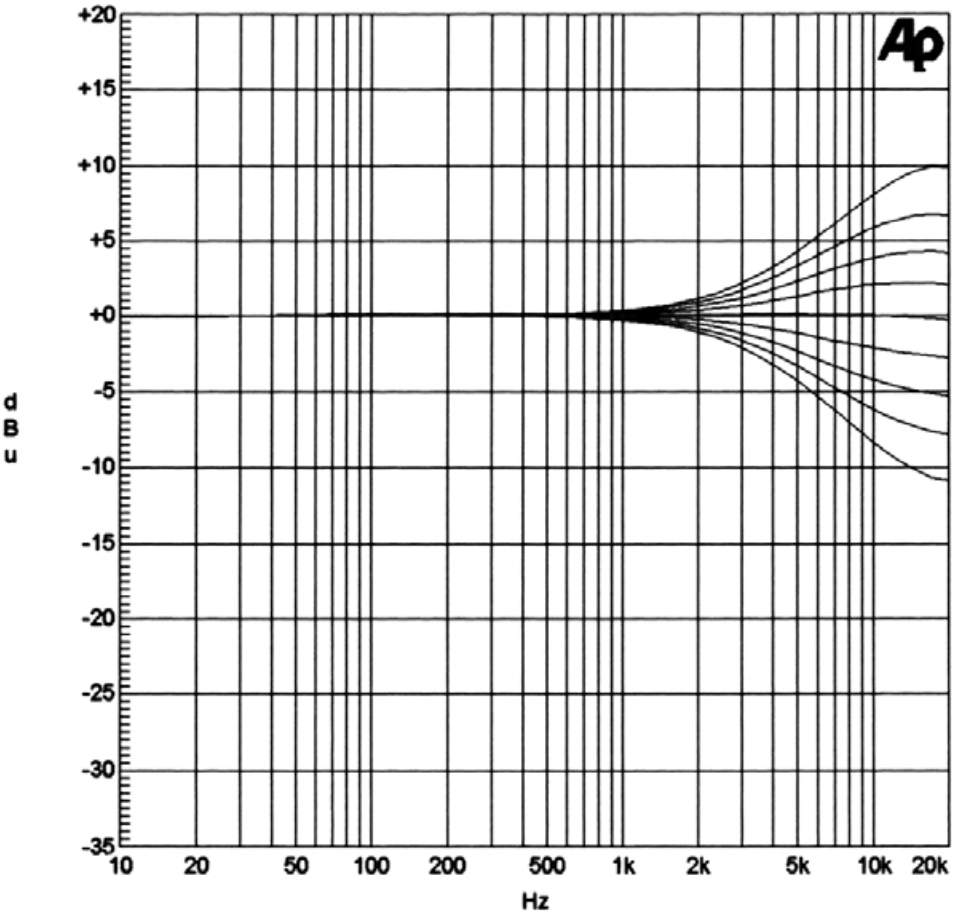


Diagram 2 shows various cut and boost settings of the Air Band filter.



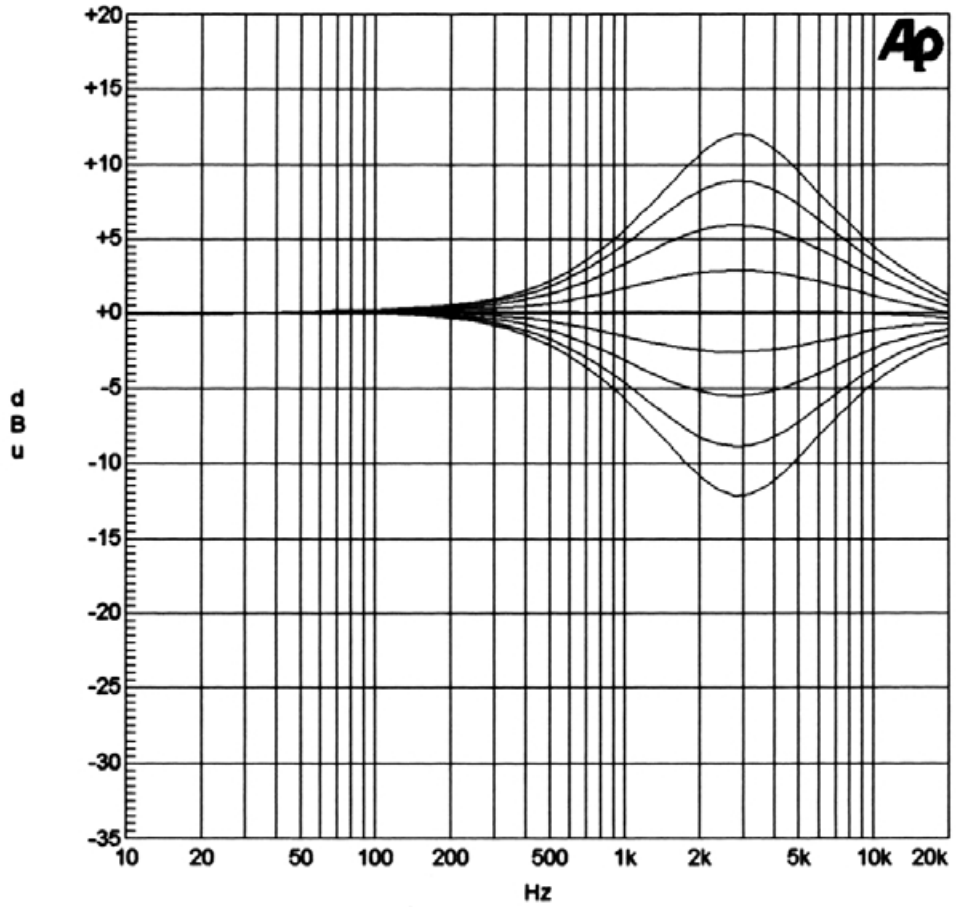


Diagram 3 displays various cut and boost settings of the Mid-Hi filter at 3 kHz.

The proportional-Q characteristic is distinctly visible.

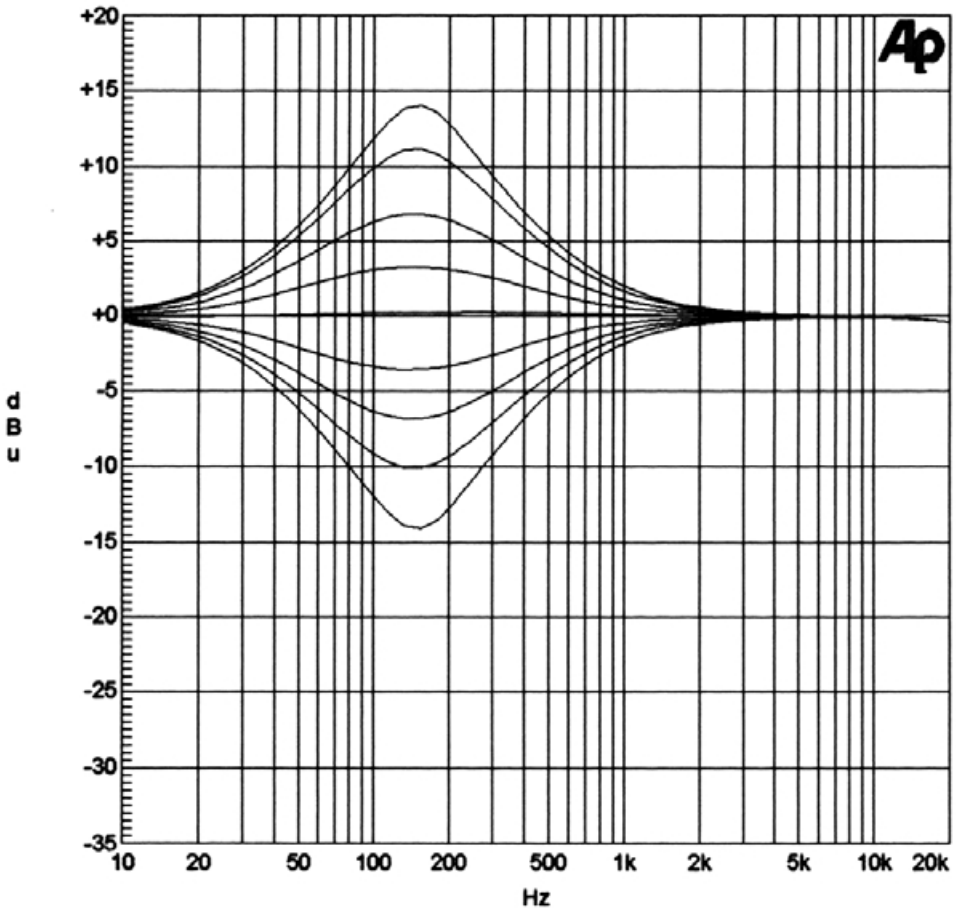


Diagram 4 displays the curves of the Low Band filter.

Various cut and boost-settings at 150 Hz.

Again the proportional-Q characteristic is clearly to see.

Warranty

SPL electronics GmbH (hereafter called SPL) products are warranted only in the country where purchased, through the authorized SPL distributor in that country, against defects in material or workmanship. The specific period of this limited warranty shall be that which is described to the original retail purchaser by the authorized SPL dealer or distributor at the time of purchase.

SPL does not, however, warrant its products against any and all defects:

1) arising out of materials or workmanship not provided or furnished by SPL, or 2) resulting from abnormal use of the product or use in violation of instructions, or 3) in products repaired or serviced by other than authorized SPL repair facilities, or 4) in products with removed or defaced serial numbers, or 5) in components or parts or products expressly warranted by another manufacturer.

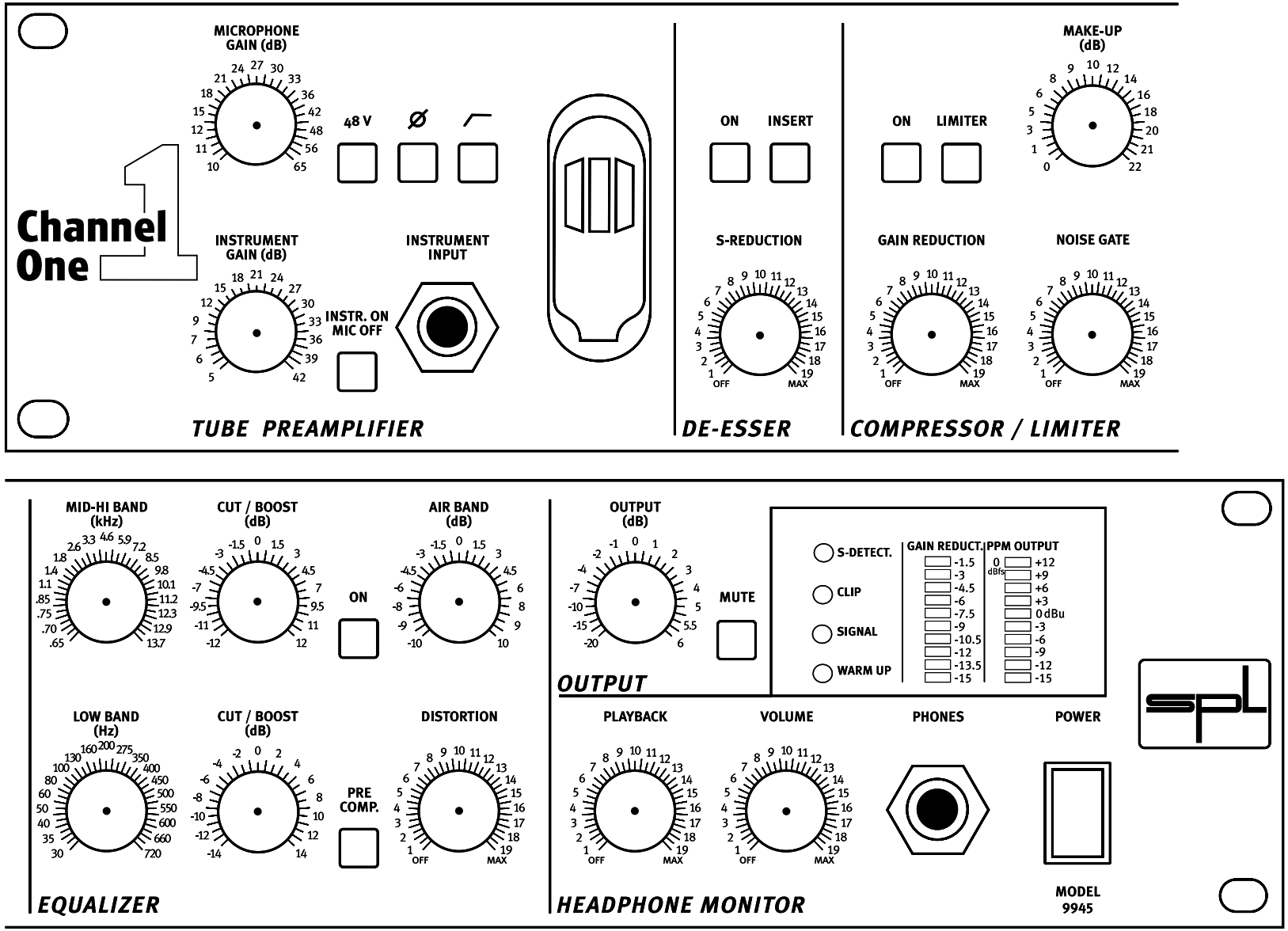
SPL agrees, through the applicable authorized distributor, to repair or replace defects covered by this limited warranty with parts or products of original or improved design, at its option in each respect, if the defective product is shipped prior to the end of the warranty period to the designated authorized SPL warranty repair facility in the country where purchased, or to the SPL factory in Germany, in the original packaging or a replacement supplied by SPL, with all transportation costs and full insurance paid each way by the purchaser or owner.

All remedies and the measure of damages are limited to the above services. It is possible that economic loss or injury to person or property may result from the failure of the product; however, even if SPL has been advised of this possibility, this limited warranty does not cover any such consequential or incidental damages. Some states or countries do not allow the limitations or exclusion of incidental or consequential damages, so the above limitation may not apply to you.

Any and all warranties, express or implied, arising by law, course of dealing, course of performance, usage of trade, or otherwise, including but not limited to implied warranties of merchantability and fitness for particular, are limited to a period of 1 (one) year from either the date of manufacture. Some states or countries do not allow limitations on how long an implied warranty lasts, so the above limitations may not apply to you.

This limited warranty gives you specific legal rights, and you may also have other rights which vary from state to state, country to country.

SPL electronics GmbH
41372 Niederkruechten, Germany



Production: **Artist:** **Date:**

Engineer:

Copy Master