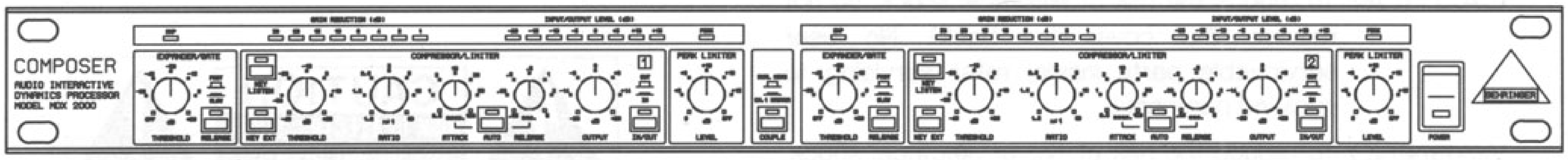

COMPOSER

The Audio Interactive Dynamics Processor Model MDX 2000

VERSION 2.1 July 1992

Text and layout: Dipl.-Ing. Ulrich Behringer

Translation: Paul Christopher Beckett and Sharon McKoy

BEHRINGER®

Spezielle Studiotechnik GmbH

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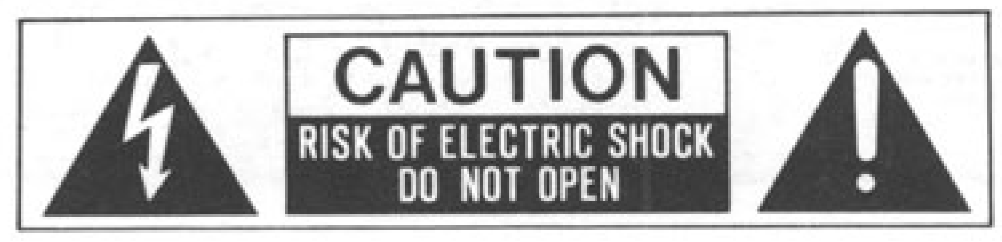
SAFETY INSTRUCTIONS

CAUTION: To reduce the risk of electrical shock, do not remove the cover (or back). No user serviceable parts inside; refer servicing to qualified personnel.

WARNING: To reduce the risk of fire or electrical shock, do not expose this appliance to rain or moisture.



This symbol, wherever it appears, alerts you to the presence of uninsulated dangerous voltage inside the enclosure - voltage that may be sufficient to constitute a risk of shock.

This symbol, wherever it appears, alerts you to important operating and maintenance instructions in the accompanying literature. Read the manual.

DETAILED SAFETY INSTRUCTIONS:

All the safety and operation instructions should be read before the appliance is operated.

Retain Instructions:

The safety and operating instructions should be retained for future reference.

Heed Warnings:

All warnings on the appliance and in the operating instructions should be adhered to.

Follow instructions:

All operation and user instructions should be followed.

Water and Moisture:

The appliance should not be used near water (e.g. near a bathtub, washbowl, kitchen sink, laundry tub, in a wet basement, or near a swimming pool etc.).

Ventilation:

The appliance should be situated so that its location or position does not interfere with its proper ventilaton. For example, the appliance should not be situated on a bed, sofa rug, or similar surface that may block the ventilation openings: or placed in a built-in installation, such as a bookcase or cabinet that may impede the flow of air through the ventilation openings.

Heat:

The appliance should be situated away from heat sources such as radiators, heat registers, stoves, or other appliance (including amplifiers) that produce heat.

Power Source:

The appliance should be connected to a power supply only of the type described in the operating instructions or as marked on the appliance.

Grounding or Polarization:

Precautions should be taken so that the grounding or polarization means of an appliance is not defeated.

Power-Cord Protection:

Power supply cords should be routed so that they are not likely to be walked on or pinched by items placed upon or against them, paying particular attention to cords and plugs, convenience receptacles and the point where they exit from the appliance.

Cleaning:

The appliance should be cleaned only as recommended by the manufacturer.

Non-use Periods:

The power cord of the appliance should be unplugged from the outlet when left unused for a long period of time.

Object and Liquid Entry:

Care should be taken so that objects do not fall and liquids are not spilled into the enclosure through openings.

Damage Requiring Service:

The appliance should be serviced by qualified service personnel when:

- The power supply cord or the plug has been damaged; or
- Objects have fallen, or liquid has been spilled into the appliance; or
- The appliance has been exposed to rain; or
- The appliance does not appear to operate normally or exhibits a marked change in performance; or
- The appliance has been droped, or the enclosure damaged.

Servicing:

The user should not attempt to service the appliance beyond that is described in the Operating Instructions. All other servicing should be referred to qualifield service personnel.

FOREWORD

Dear Customer,

we thank you for the confidence that you have shown in the Behringer company by purchasing the Behringer COMPOSER. Not only have you acquired the latest generation of dynamics processor, but also a piece of equipment which is unique in its design and specification.

Please study this manual carefully in order to make full use of the extensive capabilities of the Behringer COMPOSER.

The unit was manufactured to the highest industrial standards and went through extensive quality control checks before it was supplied.

However, should you have any reason for complaint, please do NOT return the unit to us, but proceed as outlined in chapter 11.0 "WARRANTY"!

The Behringer company wishes you every success in the use of your new COMPOSER!

BEHRINGER Specialized Studio Equipment

Dipl.-Ing. Ulrich Behringer (President)

Due to the high quality of parts and materials used, we offer with this product a

5 year warranty.*

Please note that this warranty is only valid if the enclosed warranty registration card is fully completed and returned to us within 8 days of purchase. You will find full details of our warranty terms on Page 11 - 1 of this manual.

* subject to our General Terms of Business

COMPOSER

THE AUDIO INTERACTIVE DYNAMICS PROCESSOR

1.0 INTRODUCTION

1

In purchasing the new COMPOSER Model MDX 2100, you have acquired an extremely efficient and universal dynamics processor, which combines the most commonly used dynamic functions within a compact stereo unit: every channel has its own independent Compressor/Limiter, an Expander/Gate and a Peak Limiter. The precision and flexibility of the functions are the main outstanding features of this high end unit.

Despite the extremely complex internal circuitry, the unit has a control surface which is clearly laid out and easy to understand. The internal design of the unit, together with its external sidechain path, gives the user unsurpassed creative flexibility when processing sound.

Advanced Behringer Technology

The new Behringer COMPOSER Model MDX 2100 contains several new circuit designs which make the unit the ultimate dynamics processor: compared to its predecessor MDX 2000 the unit incorporates some improvements: the Couple function relates now to both input signals (True Stereo) and the audio quality has been drastically refined due to a total redesign of the internal circuitry.

IKA (Interactive Knee Adaption) Compressor

The new IKA *(Interactive Knee Adaption)* circuit successfully combines the traditional "Hard Knee" compressor concept with the "Soft Knee" feature. For the first time, this programme adaptive ability satisfies the requirements of inaudible and musical programme compression, as well as providing creative and effective dynamics processing.

The ratio value given on the front panel is not reached until the signal level exceeds the threshold level by at least 10 dB, which means that the compression curve characteristic runs very smoothly. This gives programme compression of a quality previously unknown: even with extreme compression ratios or in the processing of critical music (classical music etc.), the programme material remains musical, clear and to a large extent, free of pumping or breathing effects and other side effects, which are found in conventional compressors.

The IKA circuit allows the COMPOSER to achieve a level of unmatched performance that until now has been unknown in the studio or in live use.

IRC (Interactive Ratio Control) Expander

A basic problem in the use of a compressor is the fact that the noise floor is highly amplified during quiet sections or when there are music pauses. This effect is exaggerated when the compression ratio is inappropriate. In order to eliminate this problem, one would normally use an additional expander or gate. The noise is then simply faded out in the quiet sections.

However, simple expanders, even when they are used correctly, drastically cut signals below the preset threshold. This effect becomes more noticeable during the transition from signal to noise floor. This can mean, that the start or end of words can be cut on a vocal track.

A newly developed IRC *(Interactive Ratio Control)* Expander has been integrated into the COMPOSER. The ratio of which, is automatically adjusted, dependent on the programme material. The result is an expander which is less critical of adjustment and which is more tolerant in the presence of those signals which appear slightly above the noise floor. Because of its new IRC circuit design, the Behringer COMPOSER's Expander/Gate section can be used as an independent unit to eradicate noise and offers almost limitless possibilities within this application.

IGC (Interactive Gain Control) Peak Limiter

A further remarkable feature of the Behringer COMPOSER is the IGC *(Interactive Gain Control)* Limiter, an intelligent combination of a *clipper* and a *programme limiter.* Above an adjustable threshold the peak limiter begins to function and restricts signal peaks radically *(clipper).* If however, the threshold of the limiter was surpassed for more than a few milliseconds, the IGC circuit automatically kicks in and reduces the level of the overall output signal so that no audible distortion occurs *(programme limiter).*

After the level falls below the threshold, the signal returns to the original value after a period of about 1 second. This IGC circuit proves to be extremely valuable as much for live work (loud speaker protection) as for digital situations, where any extreme signal peaks would exceed the maximum headroom and therefore would cause severe problems.

The following instructions should familiarize you with the special terms used first, so that you can get to know all the functions of the unit. After you have read the instructions carefully, please put them away safely, so that you can refer to them again if necessary.

1.1 TECHNICAL BACKGROUND

By the means of today's modern analogue technology, it is possible to manufacture audio equipment with a dynamic range of up to 125 dB. In contrast to analogue technique, the dynamic range of digital equipment is approximately 25 dB less. With conventional records and tape recorder technology, as well as broadcasting, this value is further reduced. Mostly, dynamic restrictions are due to noisy storage in transmission media and due to the maximum headroom of these systems.

1.1.1 Noise As A Physical Phenomenon

All electrical components produce a certain level of inherent noise. Current flowing through a conductor leads to uncontrolled random electron movements. For statistical reasons, this produces frequencies within the whole audio spectrum. If these currents are highly amplified, the result will be perceived as noise. Since all frequencies are equally affected, we term this *white noise*.

It is fairly obvious that electronics cannot function without components. Even if special low-noise components are used, a certain degree of basic noise cannot be avoided.

This effect is similar when replaying a tape. The *undirectional* magnetic particles passing the replay head can also cause uncontrolled currents and voltages. The resulting sound of the various frequencies are heard as noise. Even the best possible tape biasing can "only" provide signal-to-noise ratios of about 70 dB, which is not acceptable today since the demands of listeners have increased. For physical reasons, improving the design of the magnetic carrier is impossible using conventional means.

1.1.2 What Are Audio Dynamics?

A remarkable feature of the human ear is, that it can detect the most wide ranging amplitude changes - from the slightest whisper to the deafening roar of a jet-plane. If one tried to record or reproduce this wide spectrum of sound with the help of amplifiers, cassette recorders, records or even digital recorders (CD, DAT etc.), one would immediately be restricted by the physical limitations of electronic and acoustic sound reproduction technology.

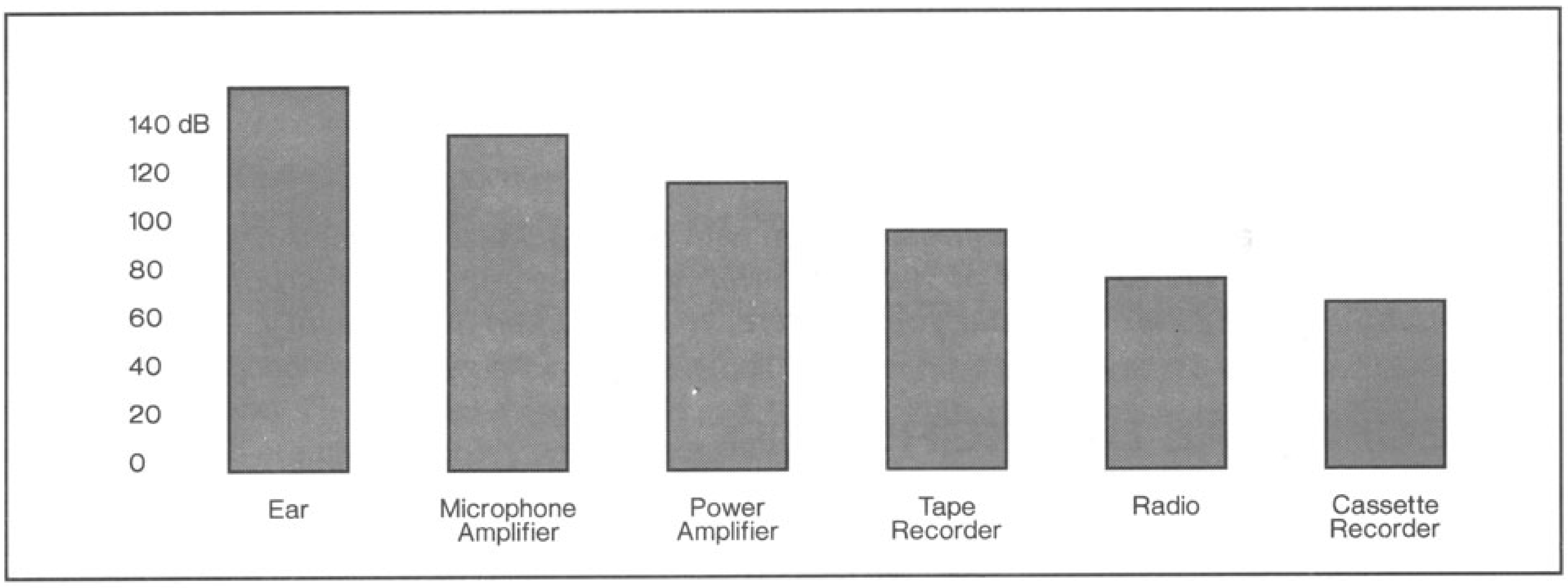


Fig. 1 The dynamic range capabilities of various devices

The usable dynamic range of electro-acoustic equipment is limited as much at the low end as at the high end. The thermal noise of the electrons in the components results in an audible basic noise floor and thus represents the bottom limit of the transmission range. The upper limit is determined by the levels of the internal operating voltages; if they are exceeded, audible signal distortion is the result. Although in theory, the usable dynamic range sits between these two limits, it is considerably smaller in practice, since a certain reserve must be maintained to avoid distortion of the audio signal if sudden level peaks occur. Technically speaking, we refer to this reserve as "headroom" - usually this is about 10 - 20 dB. A reduction of the operating level would allow for greater headroom, i.e. the risk of signal distortion due to level peaks would be reduced. However, at the same time, the basic noise floor of the programme material would be increased considerably.

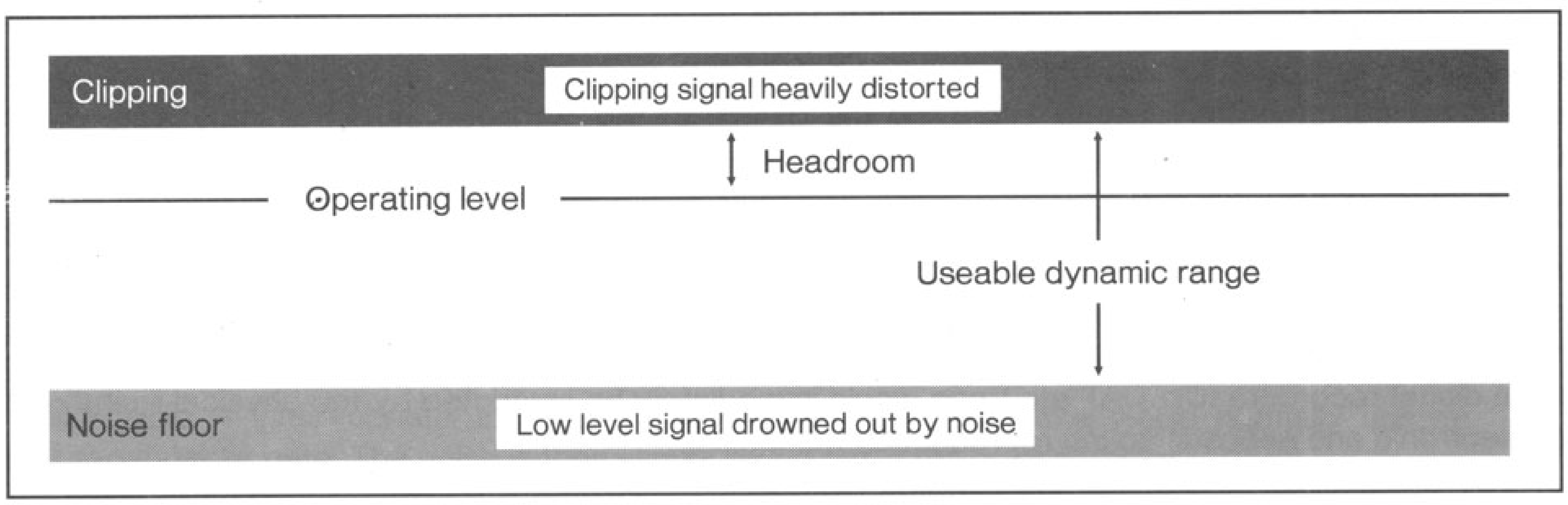


Fig. 2 The interactive relationship between the operating level and the headroom

It is therefore useful to keep the operating level as high as possible without risking signal distortion in order to achieve optimum transmission quality.

It is possible to further improve the transmission quality by constantly monitoring the programme material with the aid of a volume fader, which manually levels the material. During low passages the gain is increased, during loud passages the gain is reduced. Of course it is fairly obvious that this kind of manual control is rather restrictive; it is difficult to detect signal peaks and it is almost impossible to level them out. Manual control is simply not fast enough to be satisfactory.

The need therefore arises for a fast acting automatic gain control system which will constantly monitor the signals and which will always adjust the gain to maximize the signal-to-noise ratio without incurring signal distortion. This device is called a *compressor* or *limiter*. This system is a part of the Behringer COMPOSER.

1.1.3 Compressors/Limiters

By measuring the dynamic range of musical instruments in live recording situations, you will experience that extreme amplitudes will occur which will often lead to overload in subsequent signal processing equipment. Especially in broadcasting and in record cutting techniques, these signal peaks can lead to heavy distortion. To avoid this kind of distortion or, for example, to avoid loudspeaker being damaged by overload, *Compressors* or *Limiters* are used.

The principle function used in these devices is dependent on an automatic gain control as mentioned in the previous section, which reduces the amplitude of loud passages and therefore restricts the original dynamics to a desired range. This application is useful especially in microphone recording technique, to compensate for level changes which are caused by varying microphone distances.

Although compressors and limiters perform similar tasks, one essential point makes them different:

limiters abruptly limit the signal above a certain level, while compressors control the signal "gently" over a wider range. A limiter continuously monitors the signal and intervenes as soon as an adjustable threshold level is exceeded. This level is called the threshold. Any signal exceeding this threshold will be immediately reduced back to the adjusted threshold level.

A *compressor* also monitors the programme material continuously and also has a certain threshold level. However, in contrast to the limiter, signals exceeding the threshold are not reduced abruptly but gradually. Above the threshold, the signal is reduced in level, relative to the amount the signal exceeds this point.

Generally, threshold levels for compressors are set below the normal operating level to allow for the upper dynamics to be musically compressed. For limiters, the threshold point is set above the normal operating level in order to provide reliable signal limiting and thus protects subsequent equipment.

1.1.4 Expanders/Noise-Gates

Audio in general is only as good as the source from which it was derived. The dynamic range of signals will often be restricted by noise. Synthesizers, effects devices, guitar pickups, amplifiers etc. mostly produce a high level of noise, hum or other ambient background hiss, which can disturb the quality of the programme material.

Normally these noises are inaudible if the level of the desired signal lies significantly above the level of the noise. This perception by the ear is based on the "masking" effect: noise will be masked and thus becomes inaudible, as soon as considerably louder sound signals in the same frequency band are added. Nevertheless, the further the level that the desired signal decreases, the more the noise floor becomes a disturbing factor.

Expanders or *noise-gates* offer a solution for this problem: these devices attenuate signals when their amplitudes drop, thereby fading out the background noise. Reliant on this method, gain controlling amplifiers, like expanders, can extend the dynamic range of a signal and are therefore the opposite of a compressor.

In practice, it is shown that an expansion over the entire dynamic range is not desired. With an expansion ratio of 5:1 and a processed dynamic range of 30 dB, an output dynamic range of 150 dB will be the result, exceeding all subsequent signal processors, as well as human hearing. Therefore, the amplitude control is restricted to signals, whose levels are below a certain threshold. Signals above this threshold pass through the unit unchanged. Due to the continuous attenuation of the signals below this threshold, this kind of expansion is termed "downward" expansion.

The *noise-gate* is the simplest form of an expander: in contrast to the expander, which continuously attenuates a signal below the threshold, the noise-gate cuts off the signal abruptly. In most applications this method is not very useful, since the on/off transition is too drastic. The onset of a simple gate function appears very obvious and unnatural. To achieve an inaudible processing of the programme material, it is necessary to be able to control the signal's envelope parameters.

2.0 THE DESIGN CONCEPT

2

2.1 HIGH QUALITY COMPONENTS AND DESIGN

The philosophy behind Behringer products guarantees a no-compromise circuit design and employs the best choice of components.

The operational amplifiers BE027/BE037 developed by Behringer, which are used in the COMPOSER, are exceptional. They boast extreme linearity and very low distortion characteristics. The most important aspect of the COMPOSER design is a radical VCA implementation which results in outstanding technical specification and excellent performance. To complement this design the choice of components includes high tolerance metalfilm resistors and capacitors, 41 detent potentiometers, gold plated relay contacts and several other stringently selected elements.

Before final calibration the unit is "burnt in", which means that the unit is placed in a special oven for 24 hours in order to stabilize and artificially age the unit. This guarantees several years of constant performance specifications. The burn-in test conforms to military guidelines.

2.1.1 Two Independent Channels

The Behringer COMPOSER has two identical channels which may be used independently or may be linked together by the COUPLE switch.

2.1.2 Failsafe Relays

Failsafe relays have been incorporated into the design of the Behringer COMPOSER, which automatically and silently bypass the unit in the event of power supply disconnection or failure. These relays are also active at switch-on to isolate the COMPOSER until the power rails have settled, thus preventing the possibility of a potentially damaging switch-on thump.

2.1.3 The "Operating Level" Switch

In order to adapt the COMPOSER to various operating levels, an operating level switch for each channel has been installed on the back of the unit. This switch allows you to choose between the home recording level (-10 dBV = +0.316 V) and the studio level (+4 dBu = +1.22 V) and permits the COMPOSER to be used to maximum effect in both areas. The switch also sets the level meter to its respective reference.

2.2 INPUTS AND OUTPUTS

2.2.1 Balanced Inputs And Outputs

As standard, the Behringer COMPOSER is installed with electronically servo-balanced inputs and outputs. The new circuit design features automatic hum and noise reduction for balanced signals and thus allows for trouble-free operation, even at high operating levels. Externally induced mains hum etc. will be effectively suppressed.

The automatic servo-function recognizes the presence of unbalanced connectors and adjusts the nominal level internally to avoid level differences between the input and output signals (correction 6 dB).

2.2.2 Transformer Balanced Outputs (Optional)

In contrast to electronic balancing, the use of transformer-balanced outputs offers the advantage of galvanic separation between units. Electrical potential differences and ground loops in audio installations do not therefore impair the performance of the units.

The transformer-balanced outputs, commonly used in radio and TV engineering, can also be fitted retrospectively upon request. The Behringer transformer OT-1 is designed to the highest exacting standards and is available as an accessory.

3.0 BLOCK DIAGRAM

3

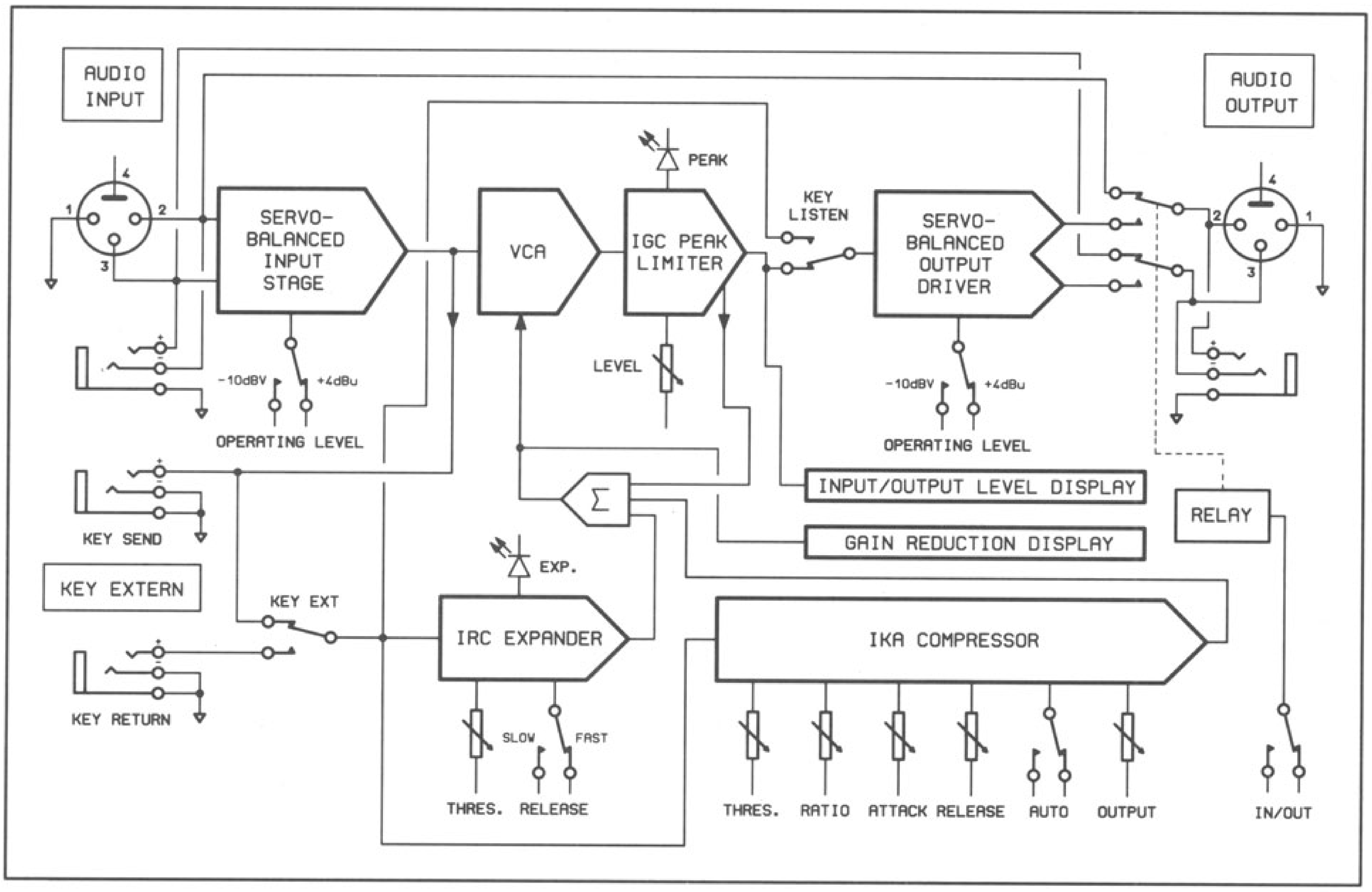


Fig. 3 Block diagram of the Behringer COMPOSER

AUDIO Chain

The input signal passes firstly through an electronically balanced input stage and is then directed through the VCA *(Voltage Controlled Amplifier)* which actually governs the dynamic process. It then passes through the Peak Limiter section. The subsequent output stage balances the signal and leads it via the IN/OUT-relay to the output connectors.

SIDECHAIN Path

The audio signal is simultaneously directed to the KEY SEND output and across the KEY EXTERNAL switch to the Expander/Gate and Compressor sections. Further processing of this input signal results in a conversion of the audio signal into rectified control voltages in each section, which are then summed by an amplifier to control the VCA.

"EXTERNAL KEY" Facility

An external signal can be fed into the KEY RETURN connector, which allows the unit to be controlled externally. By engaging the KEY EXT switch, the Behringer COMPOSER can be used for example as a frequency selective compressor (de-esser etc.) or the Expander/Gate section can be used as an externally triggered noise-gate.

The KEY LISTEN switch installed in the audio chain before the output stage, allows the audio signal to be monitored and facilitates the adjustment of external units.

4.0 INSTALLATION

4

Your Behringer COMPOSER was carefully packed in the factory and the packaging was designed to protect the unit from rough handling. Nevertheless, we recommend that you carefully examine the packaging and its contents for any signs of physical damage, which may have occurred in transit.

If the unit is damaged, please do not return it to us, but notify your dealer and the shipping company immediately, otherwise claims for damage or replacement may not be granted. Shipping claims must be made by the consignee.

4.1 RACK MOUNTING

The Behringer COMPOSER fits into one standard rack unit of space (1 3/4"). Please allow at least an additional 4" depth for the connectors on the back panel. Be sure that there is enough air space around the unit for cooling and please do not place the COMPOSER on high temperature devices such as power amplifiers etc. to avoid overheating.

4.2 CONNECTORS

The COMPOSER can be installed using either XLR or standard 1/4" jacks. Although the inputs and outputs are fully balanced, the automatic servo-function allows them to operate with unbalanced sources/loads.

The key connectors are fitted with 1/4" jack plugs and their inputs and outputs are unbalanced. The tip of the jack plug is in phase with pin 2 of the audio input connection and carries the signal.

4.2.1 Impedances

The input has an impedance of 80 kOhms and can be driven by most input sources. If a device's output requires a load of 600 Ohms (provided with most output transformers), a 600 Ohm resistor should be tied across pins 2 and 3 on the input connector.

As standard, the output of the COMPOSER is electronically balanced (transformer balancing optional) and has an output impedance of 40 Ohms. When driving transformer coupled loads, it may be necessary to create a 600 Ohm source impedance. For this purpose, install two 287 Ohm resistors (tolerance: 1%) in series with pins 2 and 3.

4.2.2 The Unbalanced/Balanced Operation

90% of all mistakes in audio installations can be attributed to incorrect and defective audio connections! In order to utilize the Behringer COMPOSER to its full potential, please pay special attention to the following section.

For better understanding, the technical difference between *unbalanced* and *balanced* systems must be clarified:

The Unbalanced System

Unbalanced operation is characterized by a single conductor shielded cable with the center conductor carrying the signal and the shield at ground.

The Balanced System

A balanced operation is defined as a two conductor shielded cable, where each of the two center conductors carry the signal but of opposite phase. They have equal but inverted potential differences from that of ground.

The advantage of the balanced system is based on the effect, that the differential amplifier in a subsequent device suppresses all equal phased noise, which has been induced during its transmission down the cable link, but the original signal will be amplified and will retain all its original integrity.

In this way, audio signals can be transmitted without interference or loss across long distances.

Balanced or unbalanced systems require different wiring. Please read the next section carefully and pay close attention to the correct wiring requirements of the units in the audio chain.

4.2.3 The Correct Wiring For Balanced Operation

If the unit preceding the COMPOSER uses *output balancing,* we recommend that you use balanced audio connections. This will avoid interference such as mains hum etc.

For maximum hum rejection, you should avoid common grounding, which means, grounding the COMPOSER's input and output.

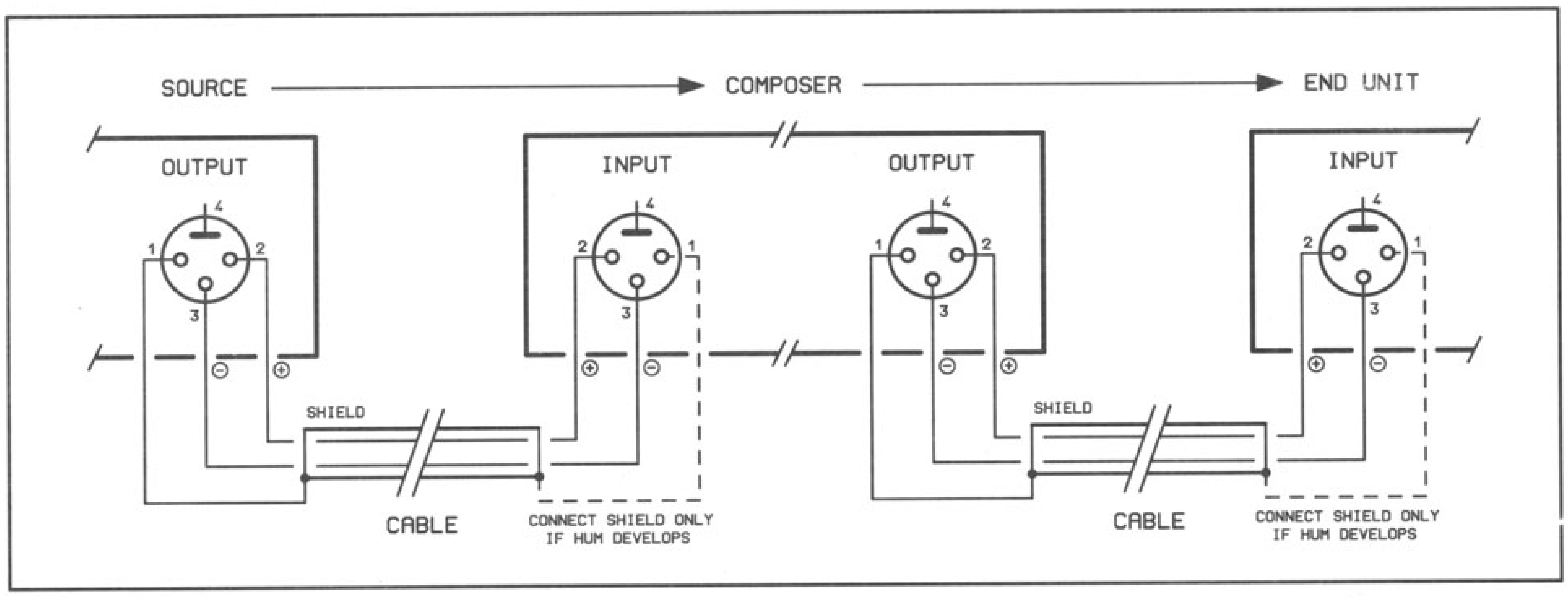


Fig. 4 The correct wiring of the balanced system

We recommend that you connect the shield of the input cable with the ground of the signal source, making sure that the shield is *not* connected to the COMPOSER's input connector.

At the output, the shield of the cable is connected to the ground of the COMPOSER, but making sure that the shield of the corresponding cable's end is *not* connected to the ground of the subsequent unit.

Generally speaking, the shield connection will be tied to the *source* units, but *not to destination* units. Please avoid at all costs linking pin 1 and 4 of the XLR connector.

If you still develop hum, it may be helpful in some cases to connect the shield on the input of the subsequent device also.

4.3 OPERATION WITH XLR CONNECTORS

4.3.1 Balanced Operation With XLR Connectors

The Behringer COMPOSER also uses XLR connectors. We recommend in accordance with the internationally agreed IEC 268-12 standard that pin 1 = ground (sleeve), pin 2 = positive input and pin 3 = negative input. We advise that you adhere to this standard, in order to provide compatibility with preceding or subsequent units and to maintain phase coherence with the KEY SEND output.

The following figure 5 shows the correct connection for the balanced input wiring, whereas figure 6 shows the correct connection for the balanced output wiring for the COMPOSER. Please note that you can distinguish between the figures by observing the shield connections.

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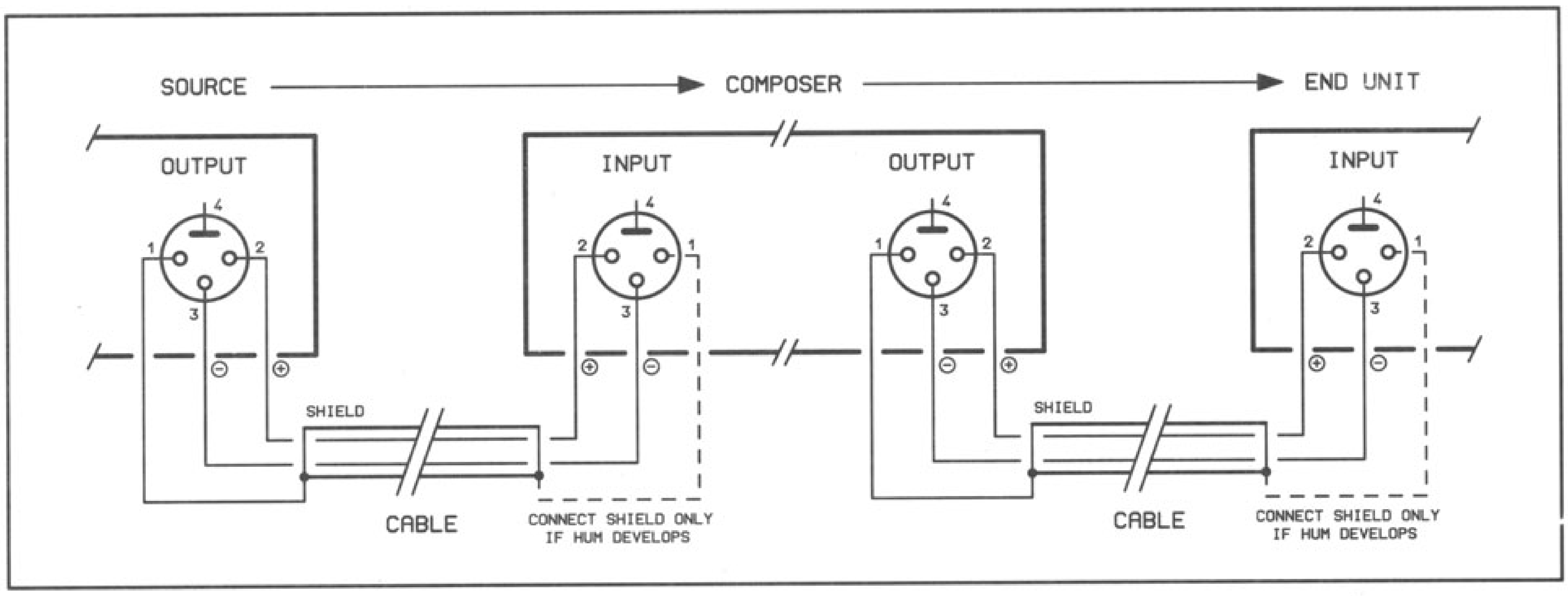

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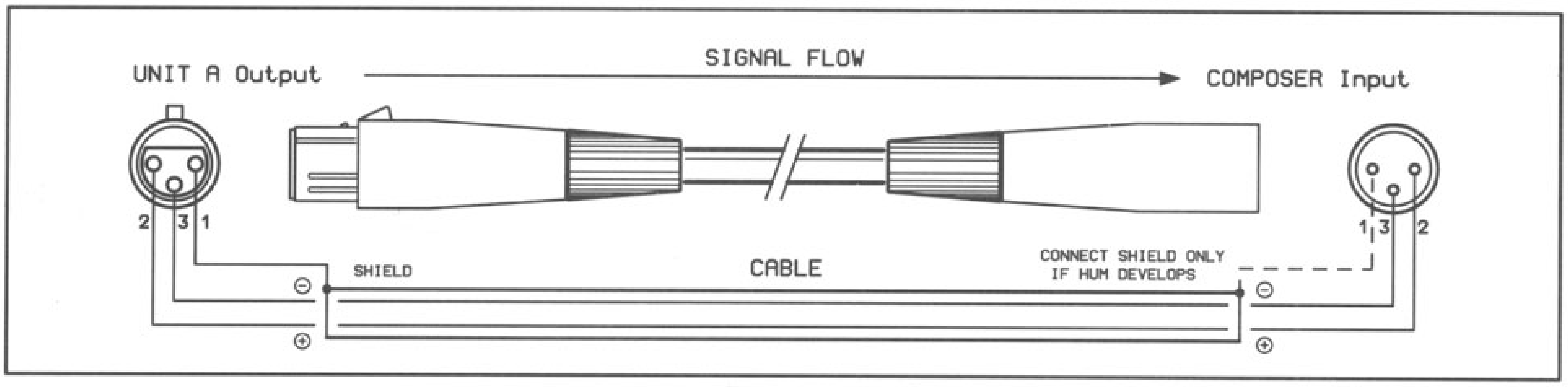


Fig. 5 Balanced COMPOSER *input wiring* using XLR connectors

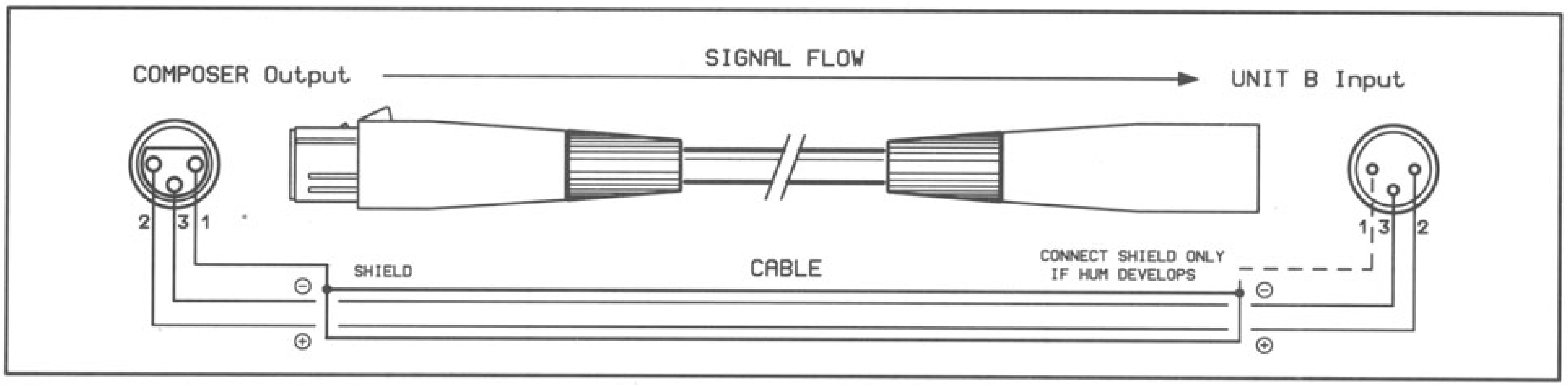


Fig. 6 Balanced COMPOSER *output wiring* using XLR connectors

4.3.2 Unbalanced Operation With XLR Connectors

Although the COMPOSER is equipped with electronically balanced inputs and outputs, it can also function unbalanced. The automatic servo-function recognizes the connection of unbalanced connectors and compensates for the 6 dB level difference which occurs, when used with unbalanced connections.

If unbalanced operation is required, please connect pin 3 to pin 1 (ground) of the XLR connector. As a result pin 2 carries the positive (+/hot) signal. If pin 3 and pin 1 are not joined, the negative input will be "open" resulting in a drastic deterioration in the signal-to-noise ratio.

This applies to both the input and output connections. Please note that in this application, the cable shield has to be connected on both ends.

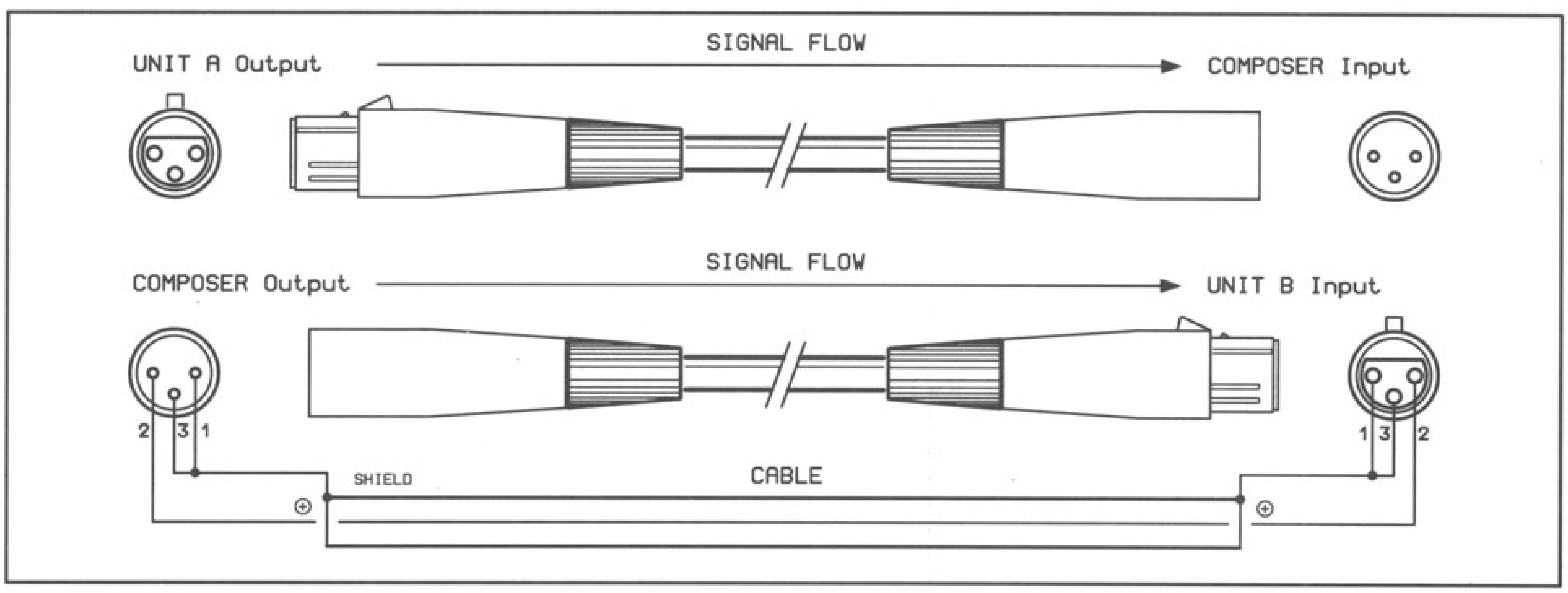


Fig. 7 Unbalanced *input and output wiring* using XLR connectors

4.4 OPERATION WITH 1/4" JACK PLUGS

The Behringer COMPOSER can also be used with standard 1/4" jack plugs. Please refer to the following sections for correct wiring:

4.4.1 Balanced Operation With 1/4" Jack Plugs

If the unit preceding the COMPOSER uses output balancing or if the unit subsequenting the COMPOSER uses input balancing, than we recommend the following adaptations. Figures 8 and 9 show the correct connection for *stereo jack to jack* operation.

Figure 8 shows the correct way to connect the balanced input, whereas figure 9 shows the correct way to connect the COMPOSER's output. Please note that you can distinguish between the figures by observing the shield connections.

If you still develop hum, it may be helpful in some cases to connect the shield on the input of the subsequent device also.

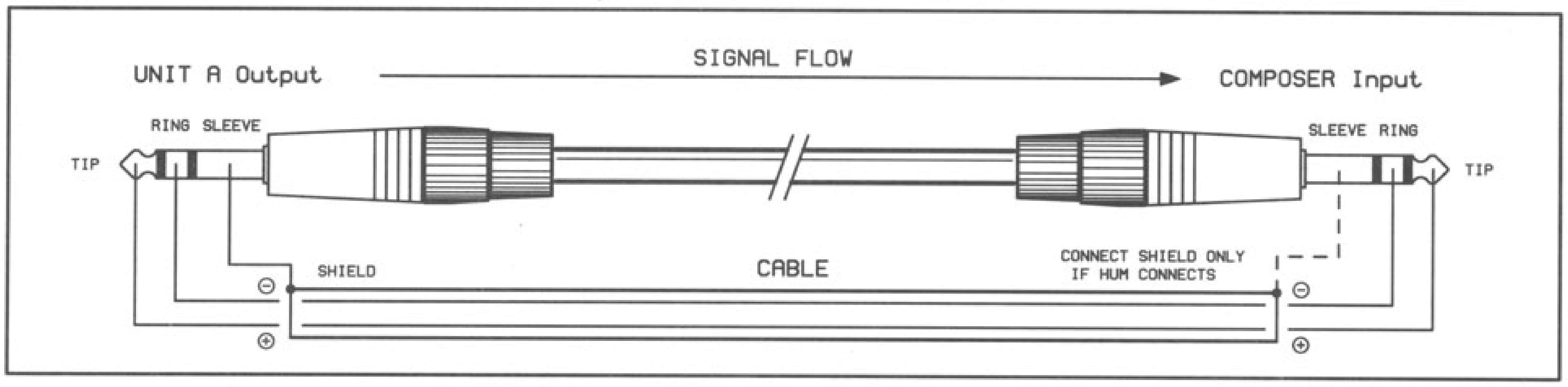


Fig. 8 Balanced COMPOSER *input wiring* with 1/4" jack connections

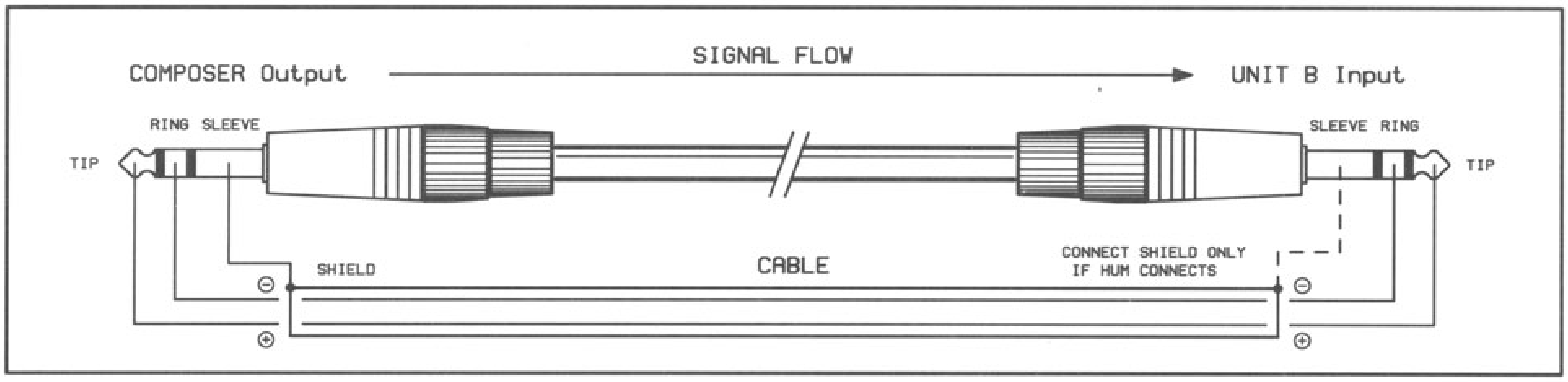


Fig. 9 Balanced COMPOSER *output wiring* with 1/4" jack connections

4.4.2 Unbalanced Operation With 1/4" Jack Plugs

In applications that do not require balanced connections, we recommend that you use a single conductor shielded cable with two *mono jack plugs.* Please make sure that the shield is connected at both ends.

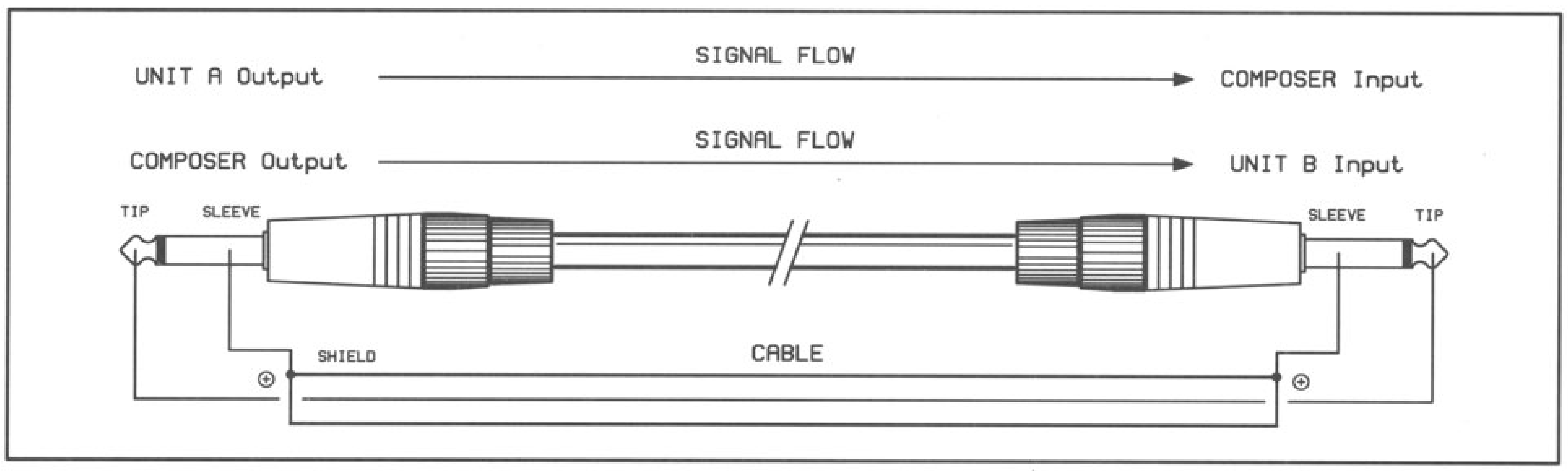


Fig. 10 Unbalanced *input and output wiring* with 1/4" jack connections

4.5 MAINS CONNECTION

The mains connection of the COMPOSER is made by using a mains cable and a standard IEC receptacle. It meets all of the international safety certification requirements.

Please make sure that all units have a proper ground connection. For your own safety, it is advisable not to remove the ground connection within the units or at the supply or fail to make this connection at all. The audio ground of the COMPOSER is internally capacitor de-coupled, to isolate it from the supply earth. It is therefore not advantegeous to attempt ground loop problem solving using this method.

4.5.1 Operating Voltage Switch

Before you switch on the unit, check that it is configured to match your AC mains voltage requirements. If it does not comply, then it is necessary to switch the operating voltage to the correct supply requirements BEFORE turning on the unit, otherwise the unit could be severely damaged. You will find this switch at the back, adjacent to the IEC receptacle.

4.5.2 Safety Fuse Replacement

A safety fuse protects the unit from serious defects. If the fuse blows, this is a warning sign and always indicates that the circuit is overloaded. The fault must always be repaired before the fuse is replaced.

If the safety fuse is faulty and needs replacing after the unit is repaired, please make sure that you replace it only with the identical type and rating. NEVER use fuses of different ratings or cover faulty fuses with aluminium foil. This can cause fire and electric shocks and will endanger your life and the lives of others.

5.0 CONTROLS

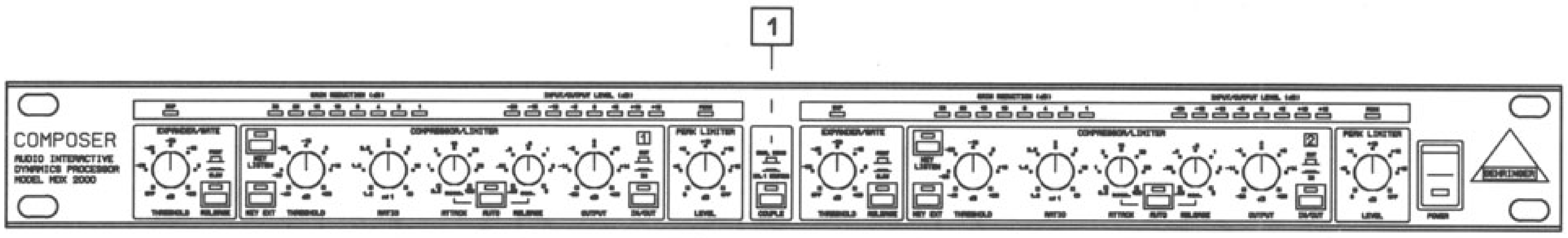


Fig. 11 The control surface of the COMPOSER

The Behringer COMPOSER has two identical channels. Each channel is equipped with 5 push button switches, 7 rotary controls and 18 LEDs. The COUPLE switch is for stereo operation:

1 COUPLE switch

The COMPOSER converts to stereo mode by engaging the COUPLE switch, where the left channel assumes the control of both audio paths, whereby the control voltage of channel 2 will be replaced with that of channel 1. By depressing the COUPLE switch, you overide all the controls and switches of channel 2 with the exception of the IN/OUT and the KEY LISTEN switch. Channel 1's controls completely take over the functions of channel 2.

5.1 EXPANDER/GATE SECTION

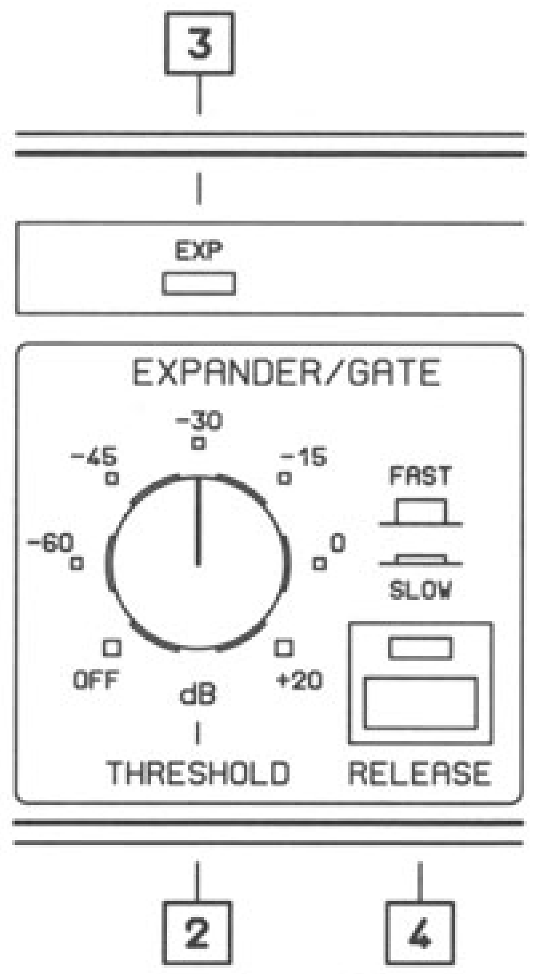


Fig. 12 Control surface of the Expander/Gate section

2 THRESHOLD control

This control sets the level below which expansion occurs. The range of this control lies between -70 and +20 dB.

3 EXP LED

This LED illuminates when expansion occurs.

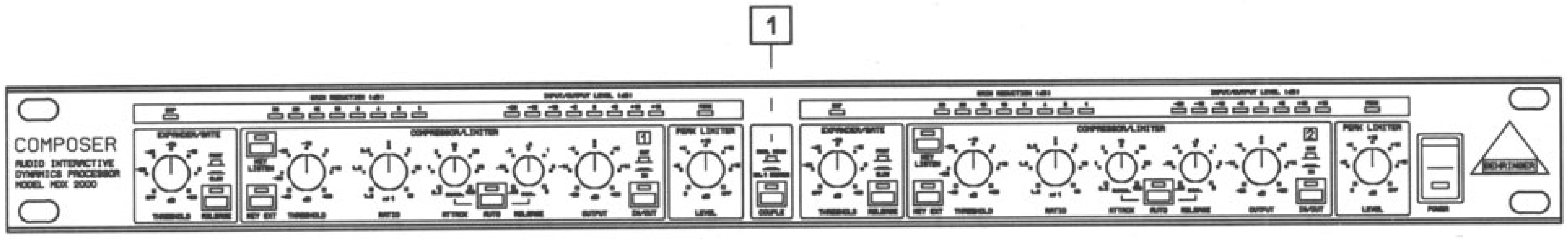
4 RELEASE switch

So that the Expander/Gate can be adjusted to the programme material, you can choose between a slow or fast release time. When this switch is depressed, the expander responds with a SLOW release.

As a general rule percussive material with little or no ambience is processed using the FAST release mode, whereas signals with long decay or with heavy ambience require the SLOW release mode.

5.0 CONTROLS

5



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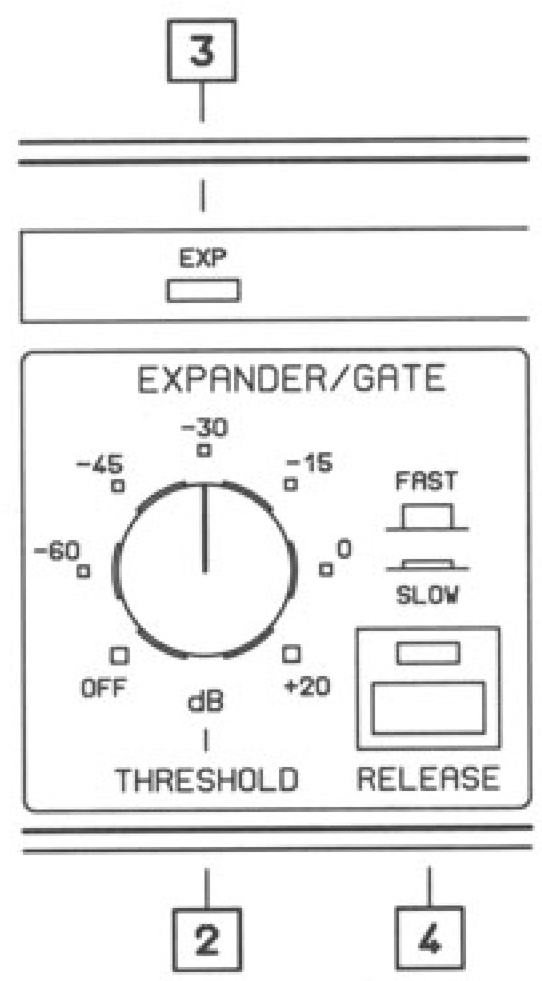

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5.2 COMPRESSOR SECTION

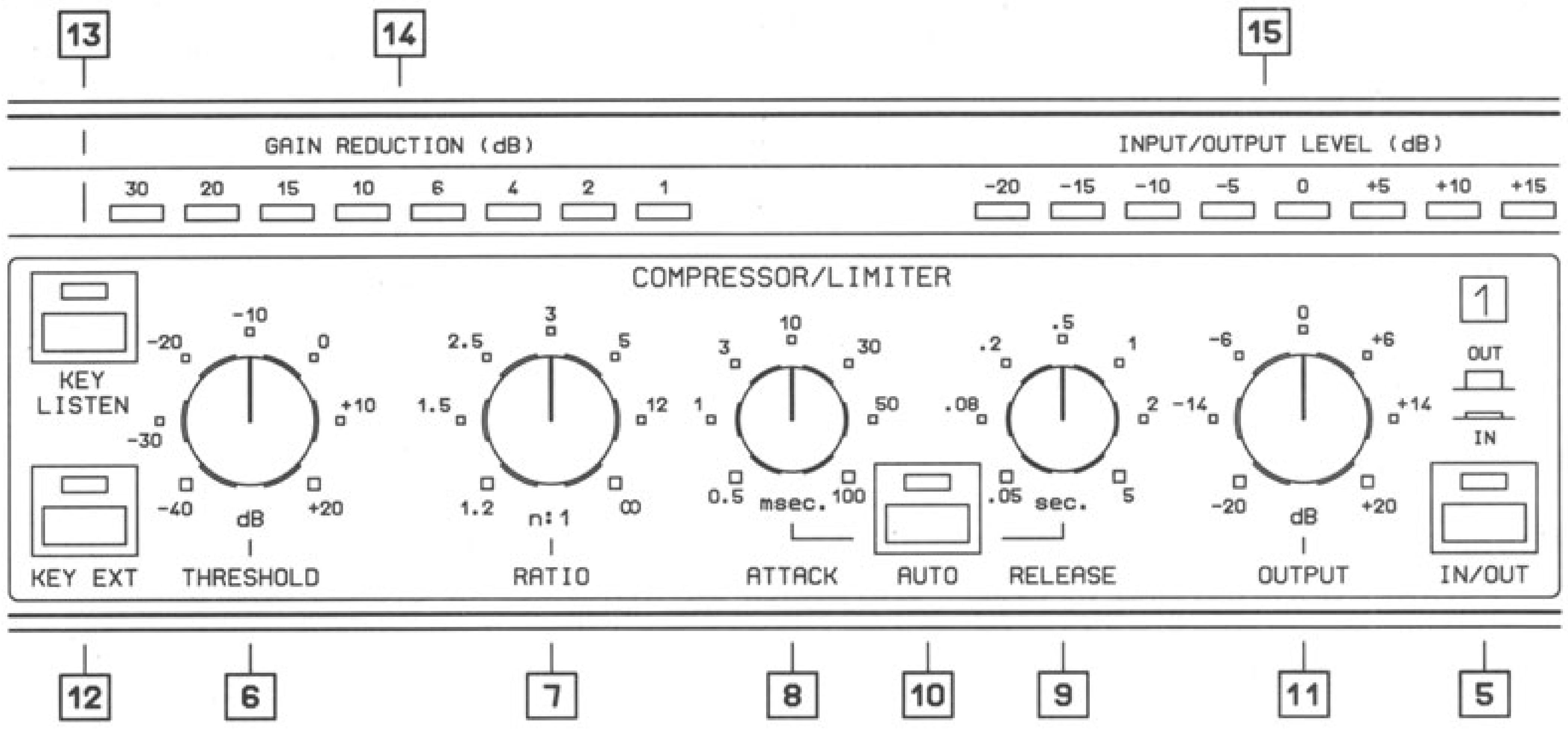

Fig. 13 Control surface of the Compressor section

5 IN/OUT switch

This switch activates the relay and engages the corresponding channel. The switch has a "Hard Bypass" function. This means that when the switch is not depressed (OUT) or the unit is turned off, the input to output connections are direct. The IN/OUT switch is used to make direct A/B comparisons between source material and the processor's effected signal.

6 THRESHOLD control

This control sets the threshold point for the compressor section. It has a range of -40 to +20 dB. The "Soft Knee" characteristic is applied to the signal exceeding the threshold point by a maximum of 10 dB. Above 10 dB, the signal would experience "Hard Knee" compression.

7 RATIO control

The RATIO control determines the ratio between the input and output level for all signals exceeding the threshold point by more than 10 dB. The control range can be adjusted from 1.2:1 to infinity:1.

8 ATTACK control

The ATTACK control determines the rate, by which the compressor responds to the signal which exceeds the threshold. This control can be adjusted from 0.5 to 100 milliseconds.

9 RELEASE control

The RELEASE control determines the rate, that the compressor returns to unity gain, after falling below the threshold level. This control can be adjusted from 0.05 to 5 seconds.

10 AUTO switch

By activating the AUTO switch, the ATTACK and RELEASE controls are disabled and the attack and release rates are automatically derived from the programme material. This function allows for unobtrusive musical compression of signals or mixes with widely varying dynamics.

11 OUTPUT control

The OUTPUT control allows for the increase or decrease of the output signal by a maximum of 20 dB. Thus, a level loss due to the compression or limiting process can be compensated for.

Please note when using the LEVEL control of the Peak Limiter section, that the OUTPUT control of the Compressor section preceedes the Peak Limiter section. If the OUTPUT control is set too high, this can result in continuous peak limiting (see item 16 "LEVEL control").

12 KEY EXT switch

When activated, this switch severes the connection between the audio input and the sidechain path, whilst at the same time allowing an external signal to be sourced at the KEY RETURN jack on the rear panel.

13 KEY LISTEN switch

Using this switch will enable you to connect the key control signal to the audio output, whilst at the same time muting the audio input. This function provides you with the ability to monitor the key signal, that is returned via inserted equalisers or other external processors. The KEY LISTEN function will assist you with tuning equaliser parameters for example.

Please note when the KEY LISTEN switch is engaged, the audio processing facility of the respective channel is disabled. When this function is active, a visiual indication will be provided by the switches LED, which will blink.

14 GAIN REDUCTION meter

The 8-stage GAIN REDUCTION meter informs you of the actual gain reduction and displays this in a range of 0 to 30 dB.

15 INPUT/OUTPUT LEVEL meter

This 8-stage INPUT/OUTPUT LEVEL meter constantly monitors the level of the input or output signal level depending on the position of the IN/OUT switch. The displayed level range lies between -20 to +15 dB. When the switch is in the OUT position, the meter monitors the input signal, whereas in the IN position the output signal is monitored. The meter is referenced to an operating level using the switch provided at the rear of the unit which is able to select between -10 dBV and +4 dBu.

5.3 PEAK LIMITER SECTION

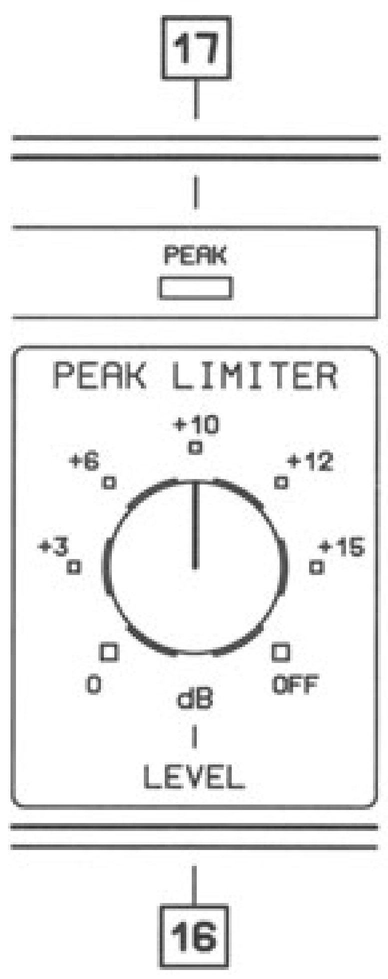

Fig. 14 Control surface of the Peak Limiter section

16 LEVEL control

This LEVEL control sets the absolute point by which the output signal is not allowed to go beyond. This limiter responds with unparalleled speed ("zero" attack) and is able to control even the fastest peak signals without allowing any audible distortion. If the output signal is excessively high, it will cause the limiter to identify clipping. If this occurs for more than 20 ms the overall output gain will be reduced for a period of 1 second. This will avoid heavy and audible distortion.

If you use the Peak Limiter as a protective device, to prevent harmful transients appearing on the output, the LEVEL control should be adjusted relative to the OUTPUT control of the Compressor section. This will ensure that the Peak Limiter section only operates when absolutely necessary. If you deliberatly drive the Peak Limiter excessively, creative effects can be achieved.

17 PEAK LED

When the Limiter function occurs, the PEAK LED illuminates.

5.4 THE BACK PANEL LAYOUT OF THE COMPOSER

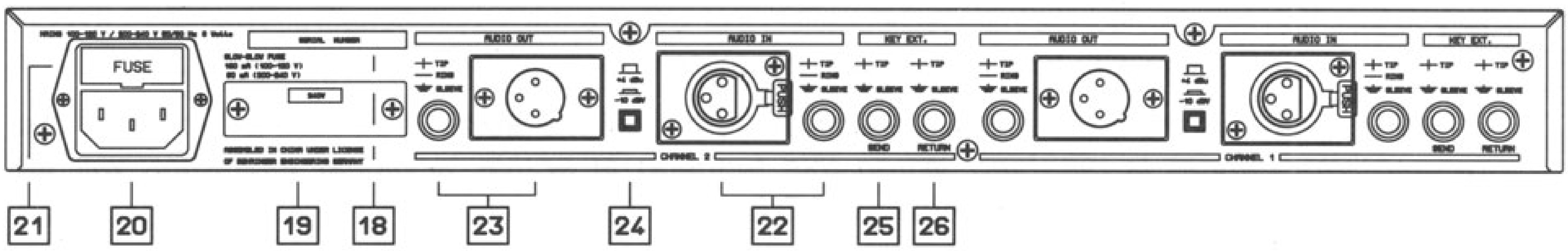

Fig. 15 The back panel layout of the COMPOSER

18 SERIAL NUMBER

Please take the time to make a note of the serial number in the space provided on the enclosed warranty registration card. Put the instruction manual in a safe place and return the completed warranty registration card to us within 8 days of purchase, making sure that the dealer stamp has been aquired.

19 OPERATING VOLTAGE SWITCH

Before you connect the unit, please make sure that the displayed voltage corresponds to your mains supply.

20 MAINS CONNECTOR

Please use the enclosed mains cable to connect the unit to the mains power supply.

21 FUSE HOLDER

Please note that, depending on the mains voltage supplied to the unit, the correct fuse type and rate must be installed.

22 AUDIO IN

These are the COMPOSER's audio inputs.

23 AUDIO OUT

These are the COMPOSER's audio outputs.

24 OPERATING LEVEL switch

This switch allows the COMPOSER to be adapted to various operating levels. You can choose between the home recording level (-10 dBV) or the professional studio level (+4 dBu). This automatically changes the metering of the unit to represent the nominal levels and permits the COMPOSER to work to its optimum dynamic range.

25 KEY SEND

This is the key signal output for the connection of external units.

26 KEY RETURN

This is the key signal input for the connection of external units.

6.0 OPERATION

6

6.1. EXPANDER/GATE SECTION

As already described in chapter 1.1.3, a downward expander automatically reduces the overall level for all signals *below* an adjustable threshold. The expander therefore operates in an opposite way to that of a compressor/limiter. Expanders generally function with a flat ratio curve, so that the signal continually fades.

Noise-gates however, can be seen as "high ratio" expanders. If the signal falls below the threshold, it is radically attenuated.

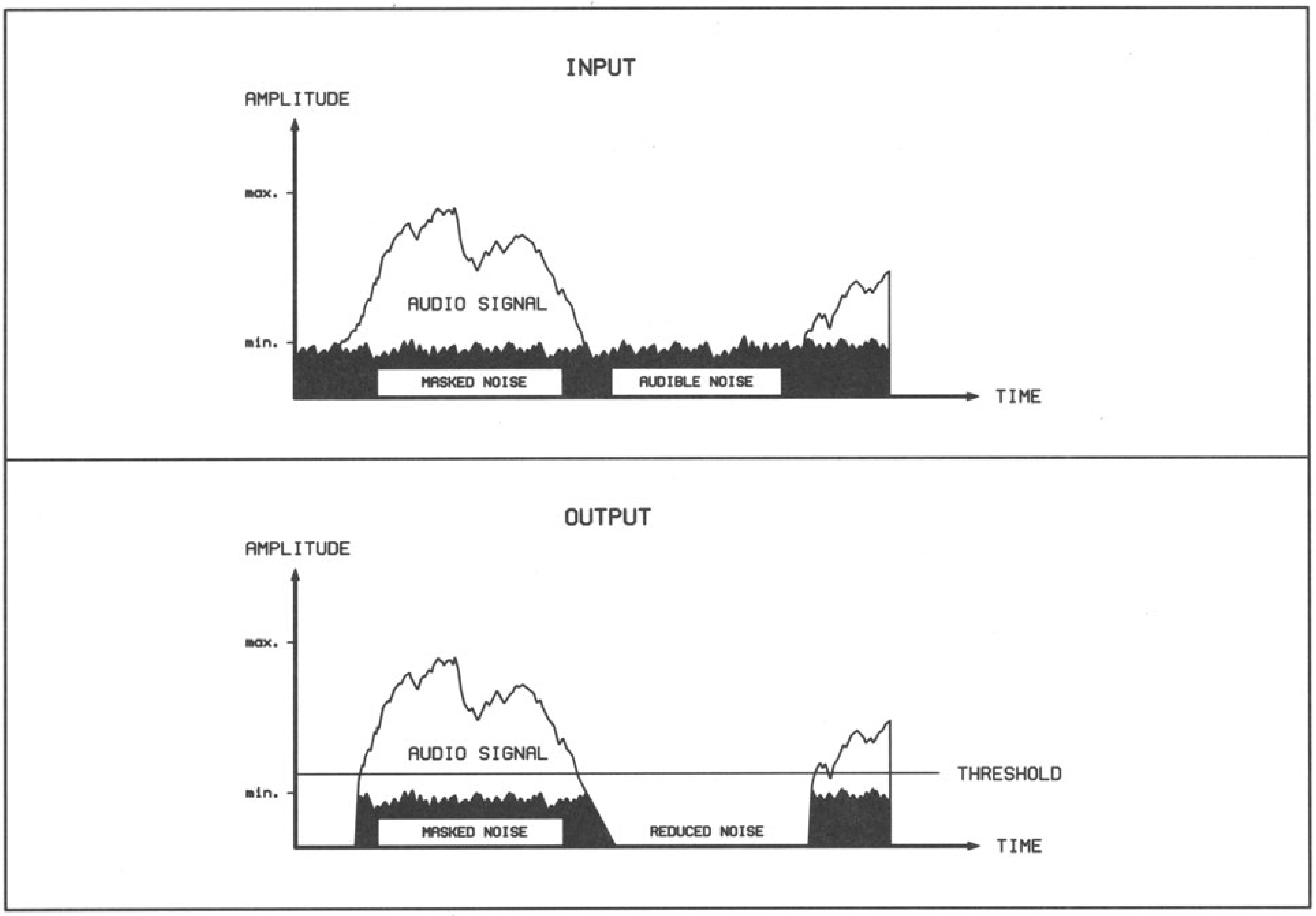

Fig. 16 Function of an expander

The Behringer COMPOSER is equipped with a newly developed IRC *(Interactice Ratio Control)* Expander, the ratio of which is automatically adjusted dependent on the programme material. The response characteristics of conventional expanders tend to cut into the signal abruptly and the result of this is unacceptable most of the time. Gain changes become audible.

The IRC Expander is therefore equipped with a soft, interactive non-linear ratio curve, which is best suited to the human hearing. Critical signals in the vicinity of the threshold level are processed with a minute expansion ratio, whereas signals that reduce in level will be subjected to an increasingly higher ratio, which will result in greater attenuation.

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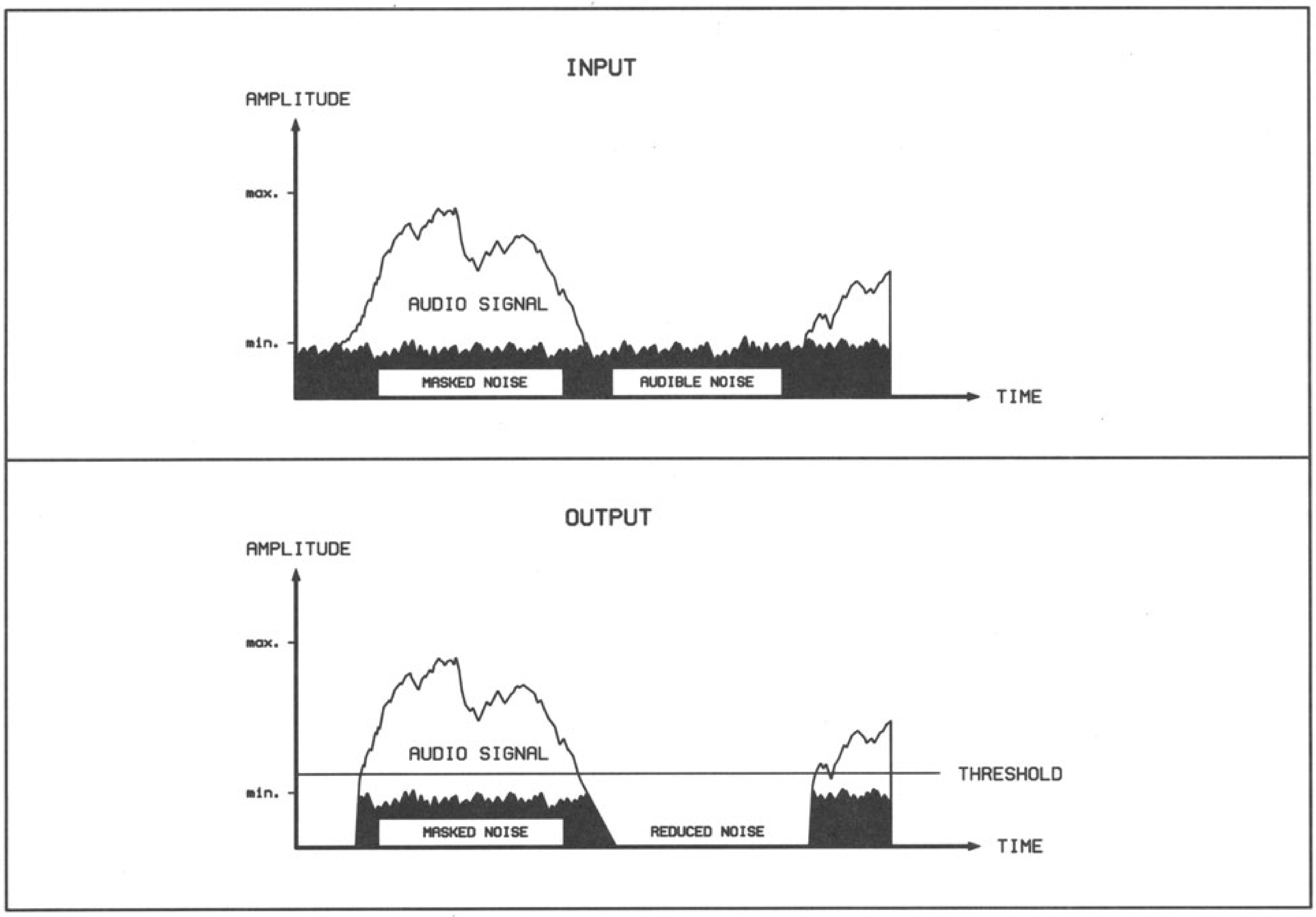

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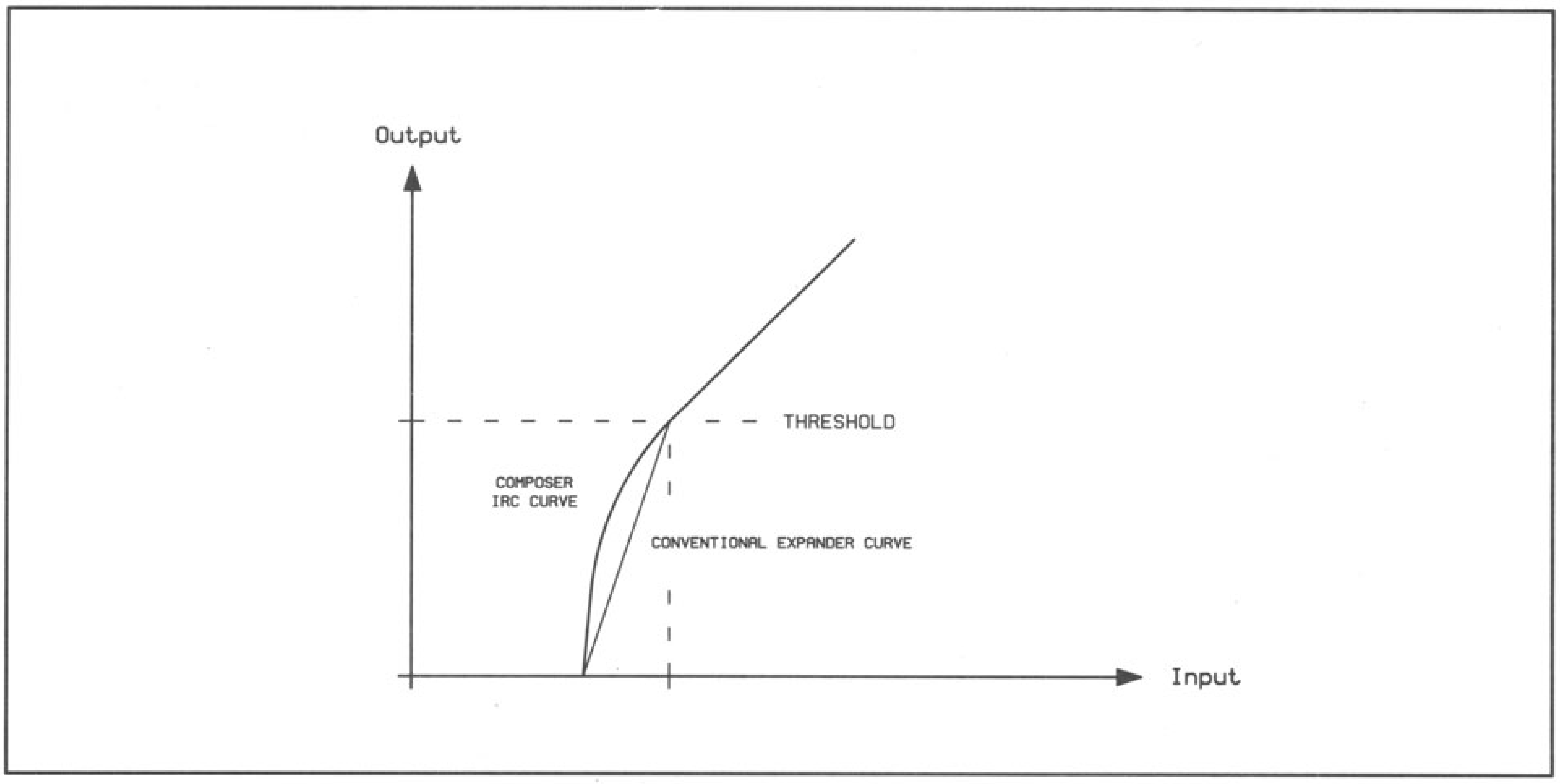


Fig. 17 IRC curve characteristics of the Expander

The result is expansion, which is less critical to adjust and which is more tolerant of useable signals, whose level is slightly above that of the noise floor.

THRESHOLD adjustment

The THRESHOLD control of the Expander/Gate stretches across a very wide range and therefore applies to all working levels. If the THRESHOLD control is turned anticlockwise, the Expander/Gate section is inoperative.

ATTACK time

The quality of an expander/gate is essentially determined by a fast attack time. It is defined as the amount of time that the expander/gate needs to come back to unity gain once the signal has exceeded the threshold. An extremely quick attack time is necessary for very fast transients, e.g. for handclaps or percussive instruments, so that the expander does not loose the initial transients and affect the sound.

RELEASE time

Another parameter is the release time: this determines the time that an expander/gate requires to attenuate the signal by a certain amount, after it has fallen below the threshold.

The most suitable release time is entirely dependent on the programme material. In order to adapt to the programme material, the Expander/Gate can be adjusted for a slow or fast release time. When the switch is engaged, the unit works with a SLOW release. The gain change is 1 dB per 100 ms in this mode. In fast mode, it is 100 dB of attenuation per 100 ms.

6.2 COMPRESSOR SECTION

In the Behringer COMPOSER, control of the dynamics process is achieved by means of a high quality VCA *(Voltage Controlled Amplifier)* with an operating range of more than 60 dB, i.e. the input signal level can be reduced or increased within a range of 60 dB. Input signal levels below the adjusted threshold are not reduced. However, as soon as the input signal exceeds the threshold level, dynamics control is activated. The amount of compression (gain reduction) is proportional to the amount by which the input signal exceeds the threshold level.

THRESHOLD control

The THRESHOLD control determines the point at which a certain input level causes the level reduction to commence. For instance, if the operating level is +10 dB and the THRESHOLD control is set to +3 dB, up to 7 dB (10 - 3 dB) can be compressed. If the input level is the same and the control is set to -10 dB, the maximum compression is 20 dB (10-[-10] =20 dB). The THRESHOLD control has an operating range of -40 to +20 dB.

If the THRESHOLD control is set fully clockwise, this corresponds to a threshold level of +20 dB. In practice, this threshold level cannot be reached as the unit would overdrive. This control setting allows the Compressor section to be put out of action, if for example, the Expander/Gate or the Peak Limiter sections are to be used independently.

The degree and type of compression is determined not only by the THRESHOLD control, but also by the subsequent RATIO, ATTACK and RELEASE controls and by the AUTO switch.

RATIO control

The RATIO control determines the ratio of change in output level compared to input level, for all signals exceeding the threshold. The scale of the ratio is calibrated in dB on the front panel. It indicates the increase in input level required, to produce a 1 dB increase in output level.

In chapter 1.1.1, we described the function of a compressor by comparing it to a volume fader, where signal peaks are controlled manually, in order to avoid distortion due to signals exceeding the threshold.

There are two ways to realise this level control: either the output level is limited in such a way that no signal can exceed a pre-defined maximum or the output level is reduced above the threshold, so that signal peaks may well exceed the threshold, but are reduced proportionally. The extent of this change in dynamics is determined by the RATIO control.

A ratio of 1:1 indicates, that the output signal will correspond to the input signal i.e., there is no level change. A ratio of 2:1 indicates, that for every 2 dB increase in input level above the threshold, there will be a corresponding increase in output level of 1 dB. A ratio of 10:1 indicates, that for every 10 dB increase in input level above the threshold, there will be a corresponding increase in the output level of 1 dB etc. If the RATIO control is set fully clockwise, this corresponds to a ratio of infinity:1. This means that all input levels are reduced to the threshold point and are thus kept constant.

It is worth observing, that a hard or infinite ratio limit has applications in certain specialised situations, but in general, this setting is neither appropriate nor necessary as this would cause audible side effects.

The effect of the RATIO control can be shown on a graph which plots input level versus output level. It clearly shows that below the threshold, the compressor acts purely as a linear amplifier.

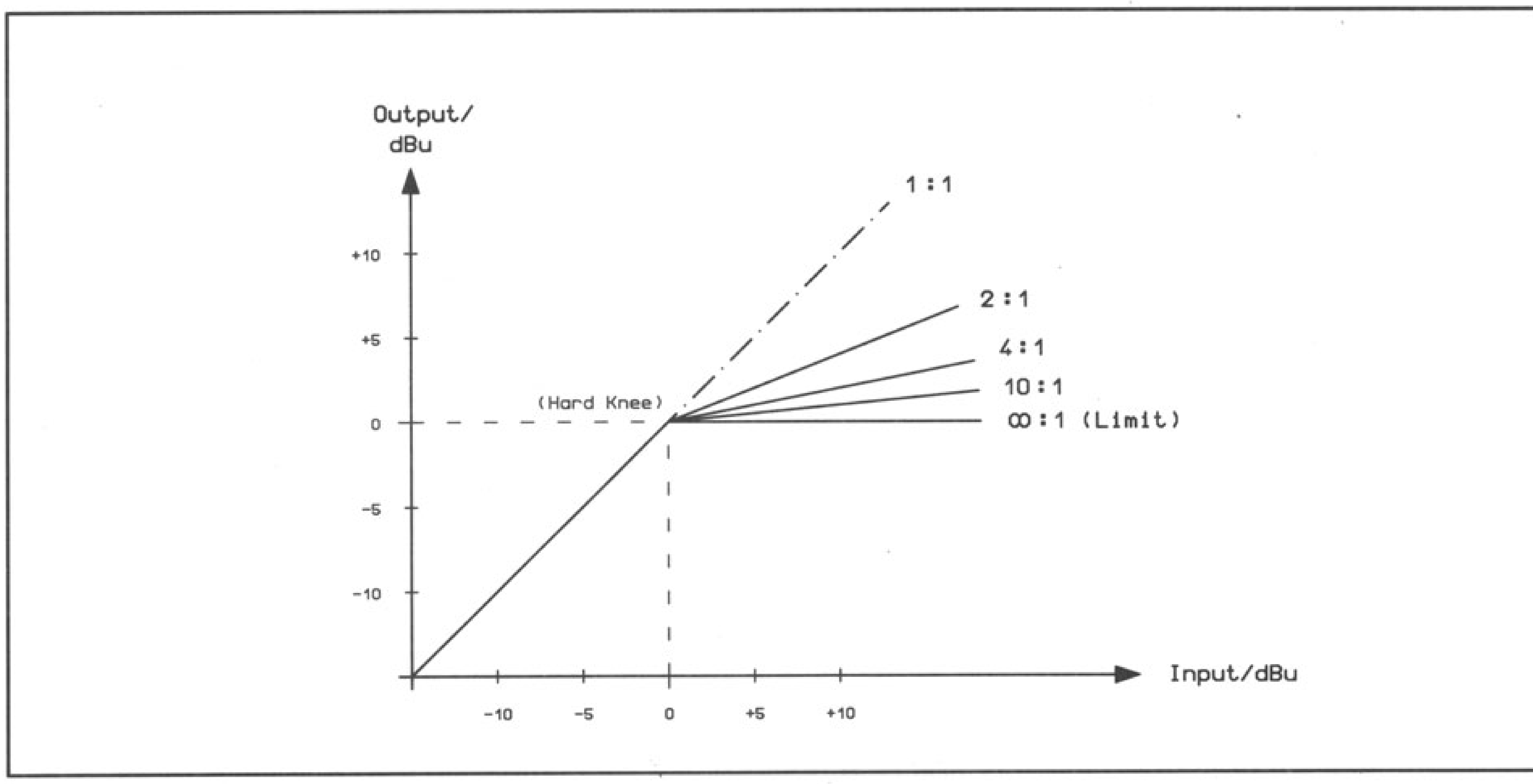


Fig. 18 The ratio of input to output level with reference to different ratio settings

In applications where gentle compression is required, it is advantageous to change from a non-compressed to compressed state in a very gradual manner to create inaudible transitions.

The Behringer COMPOSER has been configured so that for low settings of the ratio control and low levels of compression, the transition is "soft" and for increased ratio settings and high levels of compression, the transition becomes "harder". This programme dependent control curve is called "Interactive Knee Adaption", which resembles how this function appears on the following figure. The IKA curve characteristics permit inaudible compression ("Soft Knee") for low levels of ratio and gain reduction, whilst allowing harder compression ("Hard Knee") with extreme control and gain reduction if required.

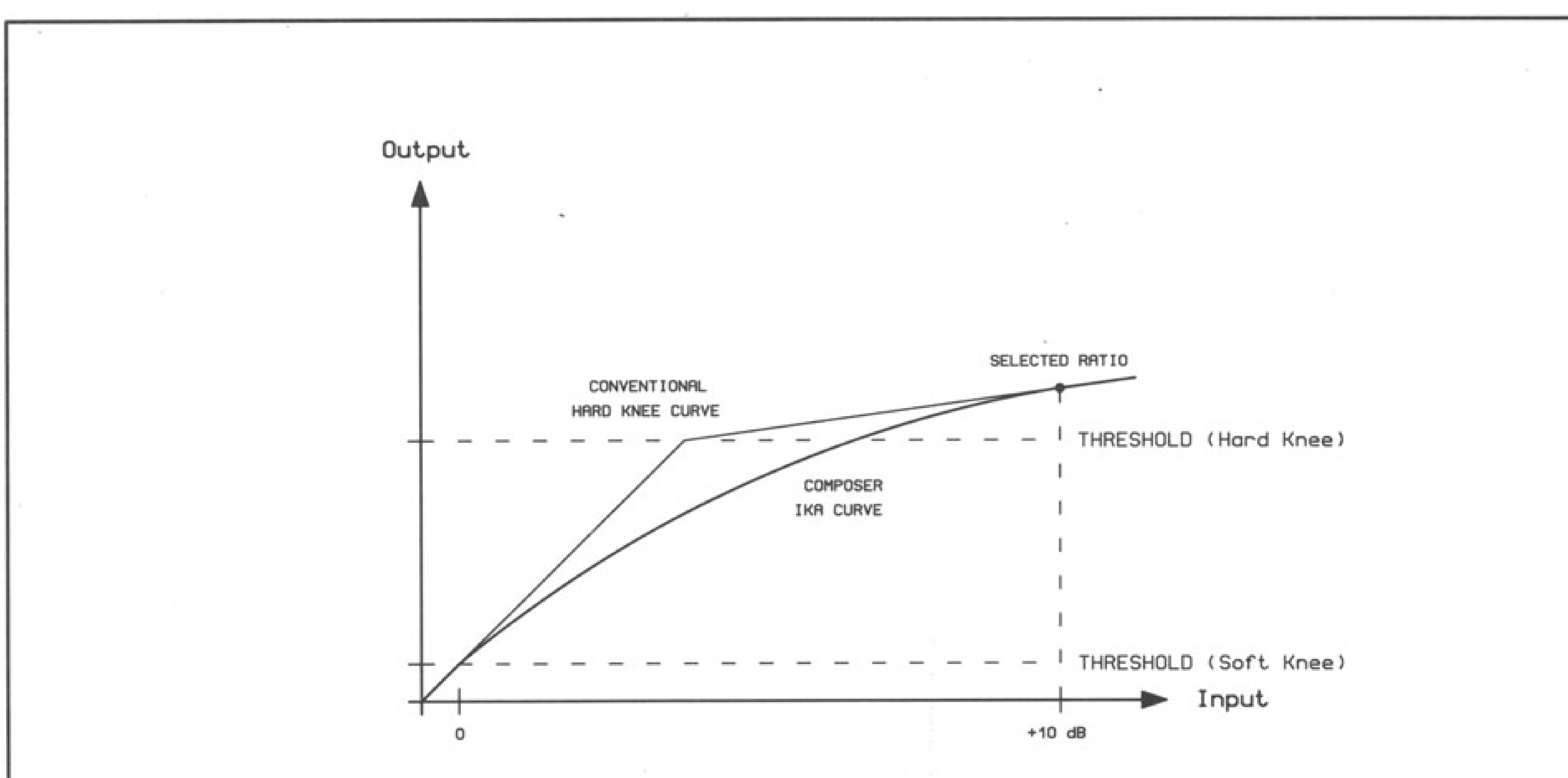


Fig. 19 Curve characteristics of the IKA circuit

ATTACK control

The characteristics of a compressor are determined to a great extent by the attack time. This is the amount of time that elapses before the compressor begins to attenuate the output level, after the threshold point has been exceeded.

A short attack time is required for very fast transients (level peaks) such as handclaps, percussive instruments like snare drums etc, so that the compressor is in a position to regulate these types of peaks. With other kinds of programme material, it can be advantageous to apply longer attack times. It is always recommended however, to begin processing with longer attack times. Whenever it is required, the time should be reduced carefully, as the danger of dynamic distortion usually increases with shorter attack times.

The attack time of the COMPOSER can be set within a range of 0.5 to 100 milliseconds.

RELEASE control

Another feature of the COMPOSER is the release time: this determines the time that the compressor requires to return to its normal gain after the input signal has fallen below the threshold point. The release time is largely dependent on the programme material. If the time is incorrectly set, this can lead to two fundamental problems:

1. If the release time is too short, this causes the overall volume to fluctuate when signals peak above the threshold level and gives the sound an unpleasant pumping effect.
2. If the release time is too long, this causes pumping and breathing side effects when a loud passage is abruptly followed by a quiet passage. The VCA increases the general volume of quiet passages which leads to a disadvantageous tonal effect.

The release time can be set within a range of 0.05 to 5 seconds.

AUTO switch

Attack and release times are important parameters, which significantly influence the quality of the dynamic control process. When the AUTO switch is engaged, the manual ATTACK and RELEASE controls are put out of action. The attack and release times are automatically derived from the programme material by means of intelligent programme recognition, so that setting errors can be effectively avoided.

The auto circuitry combines a programme dependent attack time setting with a new, two-part programme related release time setting.

1. An initially short release time for signals which fall below the threshold, so that the signals can be returned to their original level.
2. A subsequently longer release time, in order to avoid extreme and thus audible compressor action.

The AUTO processor, developed by Behringer, eliminates side effects such as pumping, modulation distortion etc., which are found in conventional compressors.

OUTPUT control

Because compression and limiting are both gain reduction processes, the output of a compressor/limiter is often at a lower level than the normal operating level. This loss of level must be compensated for. A reduction in the output level is required, if the operating level of subsequent equipment is lower (for instance, if the subsequent unit only has a high sensitivity input).

The OUTPUT control has a range of +/- 20 dB, i.e., the output signal can be reduced or increased by a maximum of 20 dB.

IN/OUT switch

The IN/OUT switch is equipped with a "Hard Bypass" function, which means that when the switch is in the OUT position, or when the unit is cut off from the mains, the input connection is directed to the output connector.

The switch is mainly used for direct A/B comparisons, which allows a comparison between the unprocessed and processed signal. When in the OUT position, all processing functions on the Behringer COMPOSER are turned off. If the switch is in the IN position, the unit works as a Compressor, Expander or Limiter, depending on the setting of the controls.

Please note that in the bypass mode, the input signal is still connected to all of the Behringer COMPOSER's circuitry, so that all the required controls can be used "dry", that is without affecting the original signal. This, in conjunction with the GAIN REDUCTION meter, provides a powerful tool for comparing processed and unprocessed signals prior to operating the bypass switch and going "live".

GAIN REDUCTION meter

The GAIN REDUCTION meter consists of eight LED's on the front panel of the Behringer COMPOSER. This meter provides a convenient visual indication of the amount of gain reduction that is taking place at any time. If a signal exceeds the input level of the threshold point, this function of the Compressor comes into play and the GAIN REDUCTION meter shows the actual measurement of gain reduction.

The following diagram shows the relationship between the input and output level of a compressor, dependent on setting the THRESHOLD, RATIO, ATTACK and RELEASE controls. It is clear that during the compression process, the output level is always lower by a certain amount than the input level.

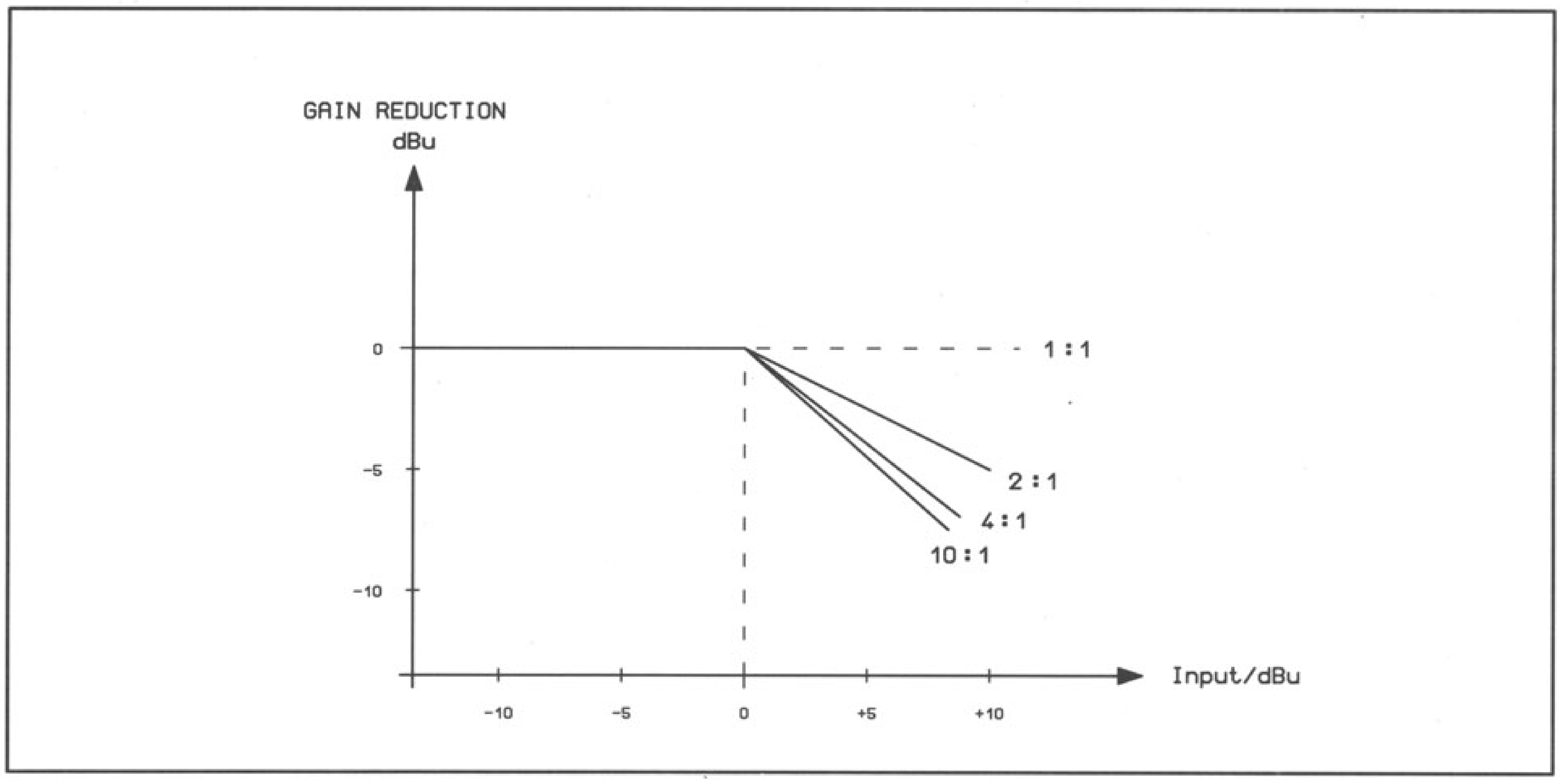


Fig. 20 The effect of a compressor can be expressed as the amount of gain reduction that is taking place for any given input

As an example, consider a signal exceeding the threshold point by 12 dB: with a ratio setting of 2:1, the output level will be increased by only 6 dB (providing the time controls are set accordingly). This means that the signal level has been reduced by 6 dB, which is indicated by the 6 dB LED.

Although the VCA of the Behringer COMPOSER features a control range of 60 dB, it is not useful to display the entire range, as in practice, such a broad control range will hardly be required.

The visual range of the GAIN REDUCTION meter is 30 dB.

INPUT/OUTPUT LEVEL meter

The meter constantly monitors the level of the input or output signal level, depending on the position of the IN/OUT switch. When the switch is in the OUT position the meter monitors the input signal, whereas in the IN position, the output signal is monitored. The meter is referenced to an operating level using the switch provided at the rear of the unit, which is able to select between -10 dBV and +4 dBu.

KEY EXT switch

An external signal can be sent via the KEY RETURN jack, which allows the unit to be controlled externally. By engaging the KEY EXT switch, the Behringer COMPOSER can be used for example as a frequency selective compressor (de-esser etc.).

KEY LISTEN switch

Using this switch will enable you to connect the key control signal to the audio output, whilst at the same time muting the audio input. This function provides you with the ability to monitor the key signal that is returned via inserted equalisers or other external processors. The KEY LISTEN function will assist you with tuning an equaliser's parameters for example.

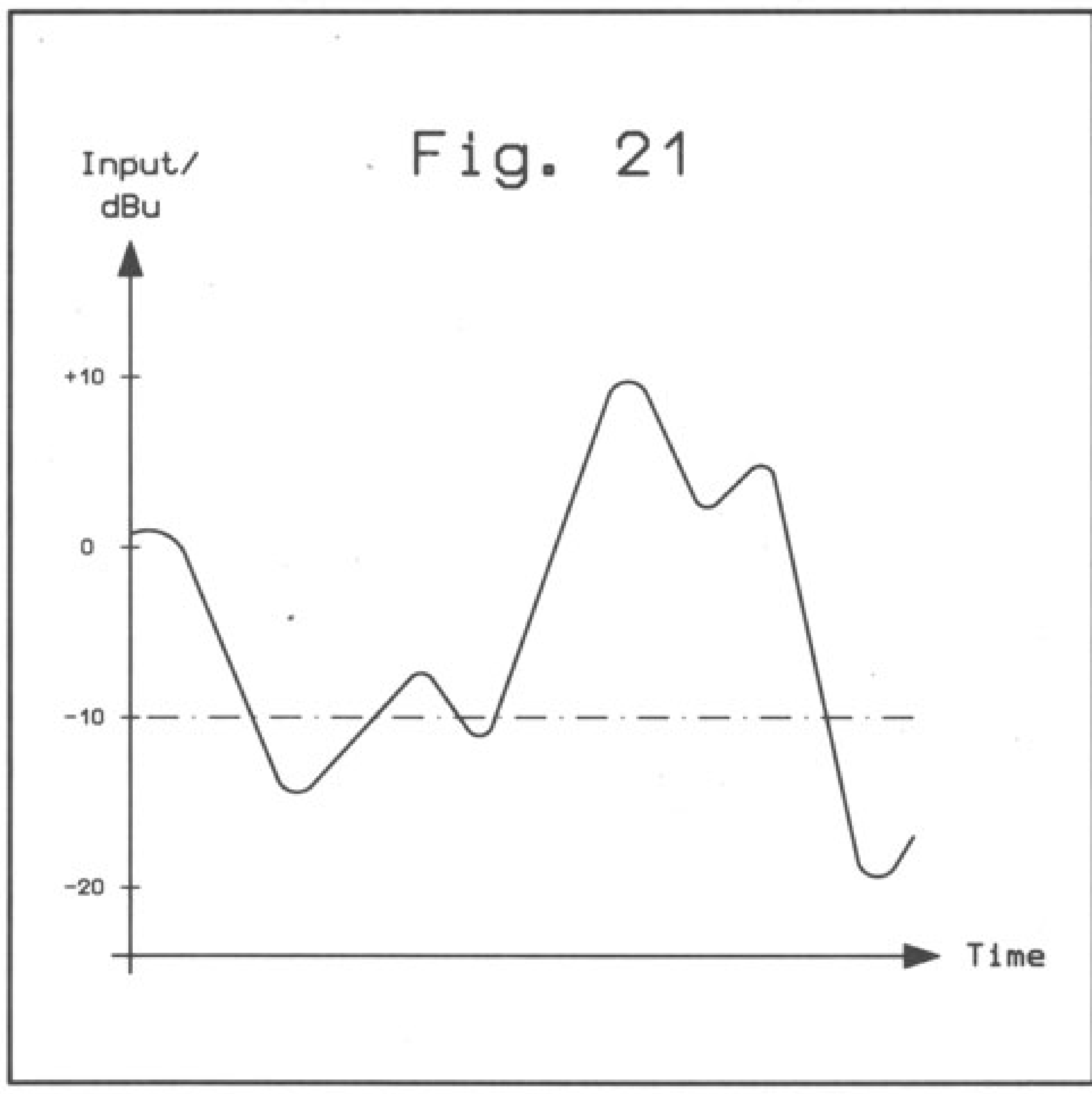
Please note when the KEY LISTEN switch is engaged, the audio processing facility of the respective channel is disabled. When this function is active, a visiual indication will be provided by the switches LED, which will blink.

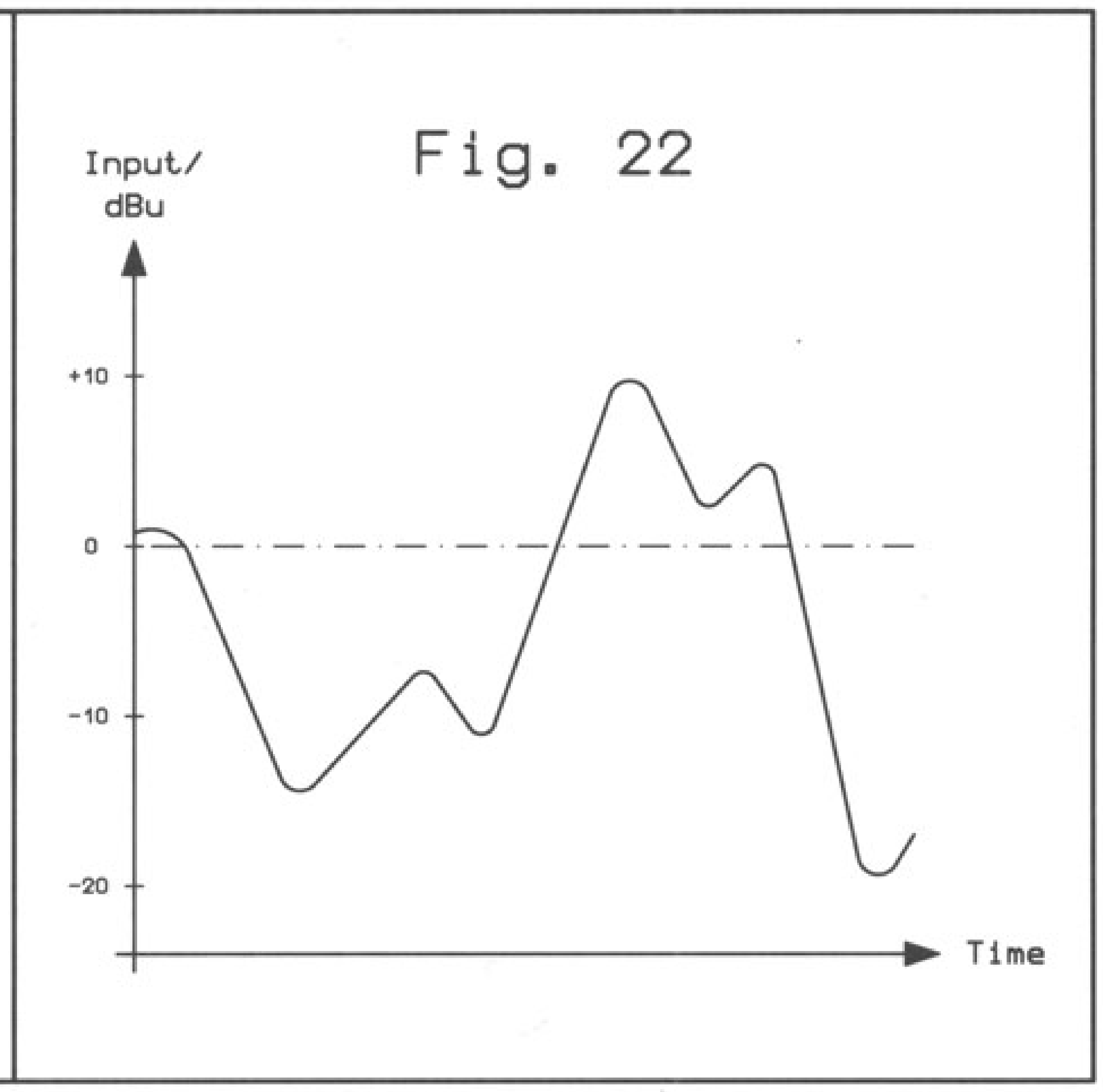
COUPLE switch

When phase coherent stereo signals are to be compressed, it is necessary for the gain controls of both channels of the compressor to be simultaneously controlled, otherwise the stereo image will shift within the sound field, as the relative levels of the left and right signals vary. When the COUPLE switch is engaged, the COMPOSER functions in stereo mode, whereby the left channel takes over the control of both channels, so that the control voltage in channel 2 is replaced by the control voltage in channel 1. When the COUPLE switch is activated, all controls and switches belonging to channel 2 are put out of action, with the exception of the IN/OUT and KEY LISTEN switches (see 5.2 "COMPRESSOR SECTION"). The controls of channel 1 take over the regulation of channel 2. Both channels now work together as in a stereo fader.

6.2.1 The Compression Effect

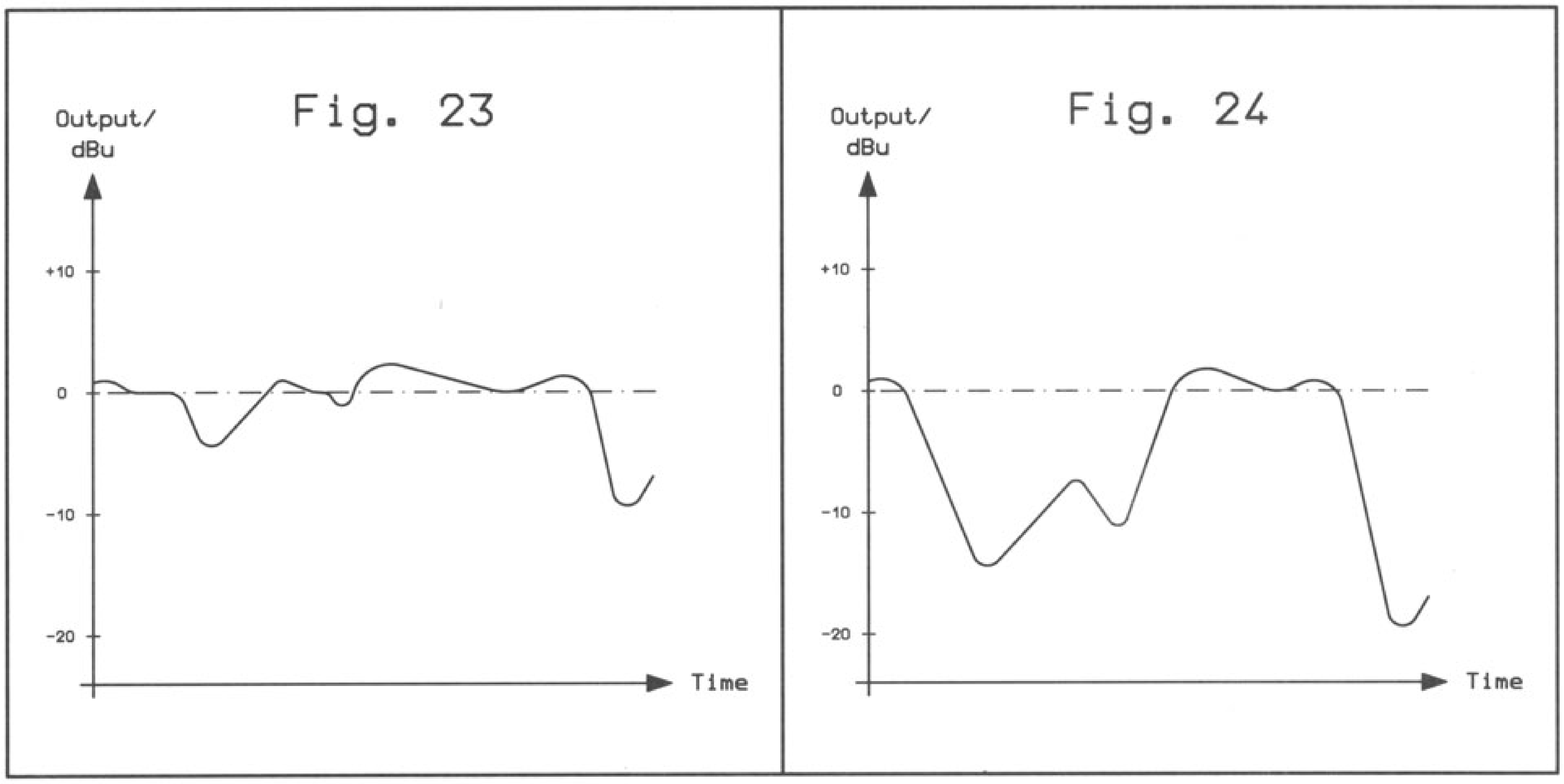
At the beginning of this chapter, we briefly dealt with the basic functions of a compressor/limiter. Now we will discuss this point in more detail, with reference to the threshold setting, input level and compression: consider an input signal which is applied to the inputs of two compressors. The threshold of the second compressor (fig. 22) is set 10 dB higher than the threshold of the first compressor (fig. 21). Since a compressor only affects signals that exceed the threshold level, it is fairly obvious, that the signal of the first unit (fig. 21), will be compressed more, because as the threshold is lower, it is exceeded to a greater extent.





Compressor 1 (fig. 21) and compressor 2 (fig. 22) BEFORE gain control

Assuming that all other controls on both channels are set identically, with gains compensated (OUTPUT control), the processing effect on these signals is shown in the following figures:



Compressor 1 (fig. 23) and compressor 2 (fig. 24) AFTER gain control

It is obvious here, that there is a large difference between these two signals in relation to their dynamic range and the processed signal. In fig. 23 it is shown to have been compressed, whereas the signal in fig. 24 is shown to have been limited.

Furthermore, it is interesting to note that by comparing the input and output waveforms for the compressed mode, the quietest sections of the input signal have been effectively raised in level, whereas the loudest sections have been effectively decreased in level. The overall effect is that both ends of the dynamic range have been pushed towards the middle. The squashing effect of compression is important to remember and highlights the major difference between compressing and limiting.

Compressing and limiting differ in one more aspect: the dynamic settings for attack and release times. For compression purposes, a preferably longer attack and release time is generally the best in order to keep the overall output signal within a specified dynamic range. For limiting applications, considerably shorter times are necessary to control fast transient signals or to increase headroom.

To achieve inaudible compression, it is imperative to work with programme dependent attack and release times. The advantage of programme dependent compression is most apparent, when processing musical material that is varied.

The Behringer COMPOSER is suitable for all applications because of its various time settings and the AUTO processor. With this switch engaged, the unit uses programme dependent attack and release times. When not engaged, the control times can be set manually.

The IKA (Interactive Knee Adaption) circuit enables at low threshold and higher ratio settings (6:1, 10:1, etc.) a limitation for signals which significantly appear above the threshold, whereas signals in the vicinity of the threshold point experience soft and "inaudible" compression. The graphics in fig. 19 show the IKA curve characteristic.

Please note that the limiter function of the Compressor section (RATIO control turned fully clockwise) does not provide a peak limiter function but a programme limiter, which does not simply cut off signal peaks above the threshold point: drastic gain control of this kind usually produces an audible and thus less desirable effect. To provide a more musical effect with this limiter function, the signal peaks are controlled in a more gentle way, with the result that signal peaks may exceed the threshold by a few decibels.

So, if you use the Behringer COMPOSER as an ultimate means of protection against overload, it is recommended that you use the following IGC Peak Limiter for these applications.

6.3 PEAK LIMITER SECTION

Referring back to chapter 6.2, we defined attack time as the time taken for a compressor to respond to programme levels which have exceeded the threshold point.

Because of the physical relationship between the frequency and slew rate, this results in the fact, that for relatively low frequencies a longer attack time was required than for higher frequencies: any unpleasant dynamic distortion would thus be avoided. When compressing a programme mix that includes a wide range of frequencies, a compromise must be made when setting the attack time. This setting would generally suit the lowest frequency components present.

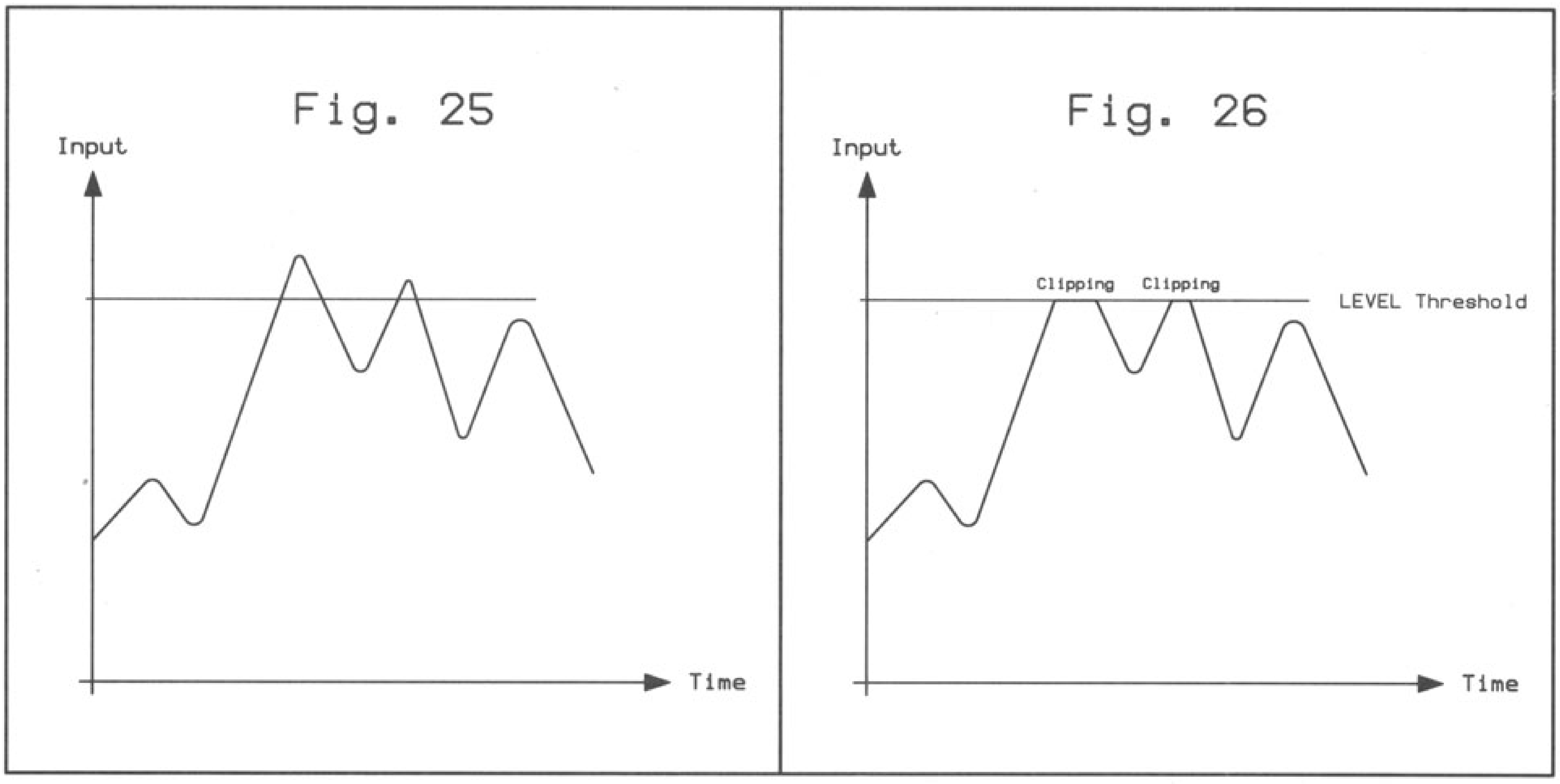
For general dynamic range control using the compressor mode, this is of no serious consequence.

However, in a limiting mode, where we are restricting the peaks of our signals to a maximum operating level to avoid distortion in any subsequent devices. This will result in very fast high frequency signal transients passing through unaffected by gain reduction. These transients could then cause distortion in the following equipment such as tape recorders and radio transmitters. It is therefore necessary to choose the attack time which is as close to "zero" attack as possible, independent of the frequency.

The Peak Limiter section of the Behringer COMPOSER provides an extra stage of gain reduction, with dynamics specifically set for these fast transients.

The COMPOSER's Peak Limiter consists of a new, two-stage IGC *(Interactive Gain Control)* circuit, which intelligently couples the *clipper* and the *programme limiter.*

The *clipper* radically cuts signals above the threshold level. The *"zero" attack* function (instantaneous response) provides absolute protection for a sound system. Overload due to harmful transients is avoided.



The signal BEFORE gain control (fig. 25) and AFTER clipper processing (fig. 26)

However, the clipper has a fundamental problem: even if you do not notice the clipping of certain transients, the limiting of the actual signal amplitude leads to heavy and unpleasant distortion. It is therefore necessary to include another dynamics section, which would additionally reduce the overall level, in order to limit the time the clipper is in effect.

This dynamics control is called a *programme limiter*: if the clipper limits a signal for longer than 20 ms, the programme limiter is engaged and reduces the overall level for 1 second and for the necessary amplitude value to prevent a repeated response by the clipper.

The function of the IGC Peak Limiter is shown in the following figures:

Fig. 27

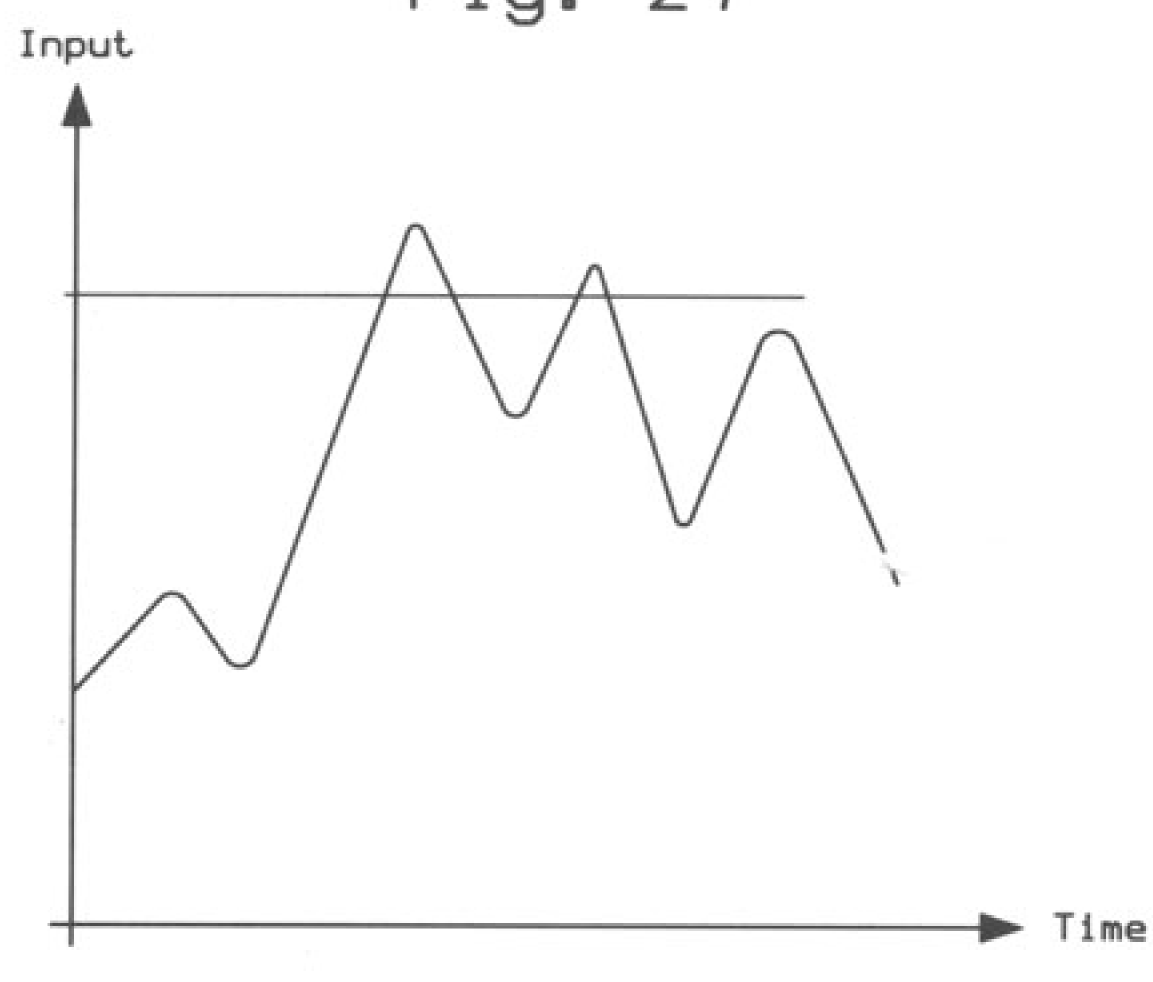
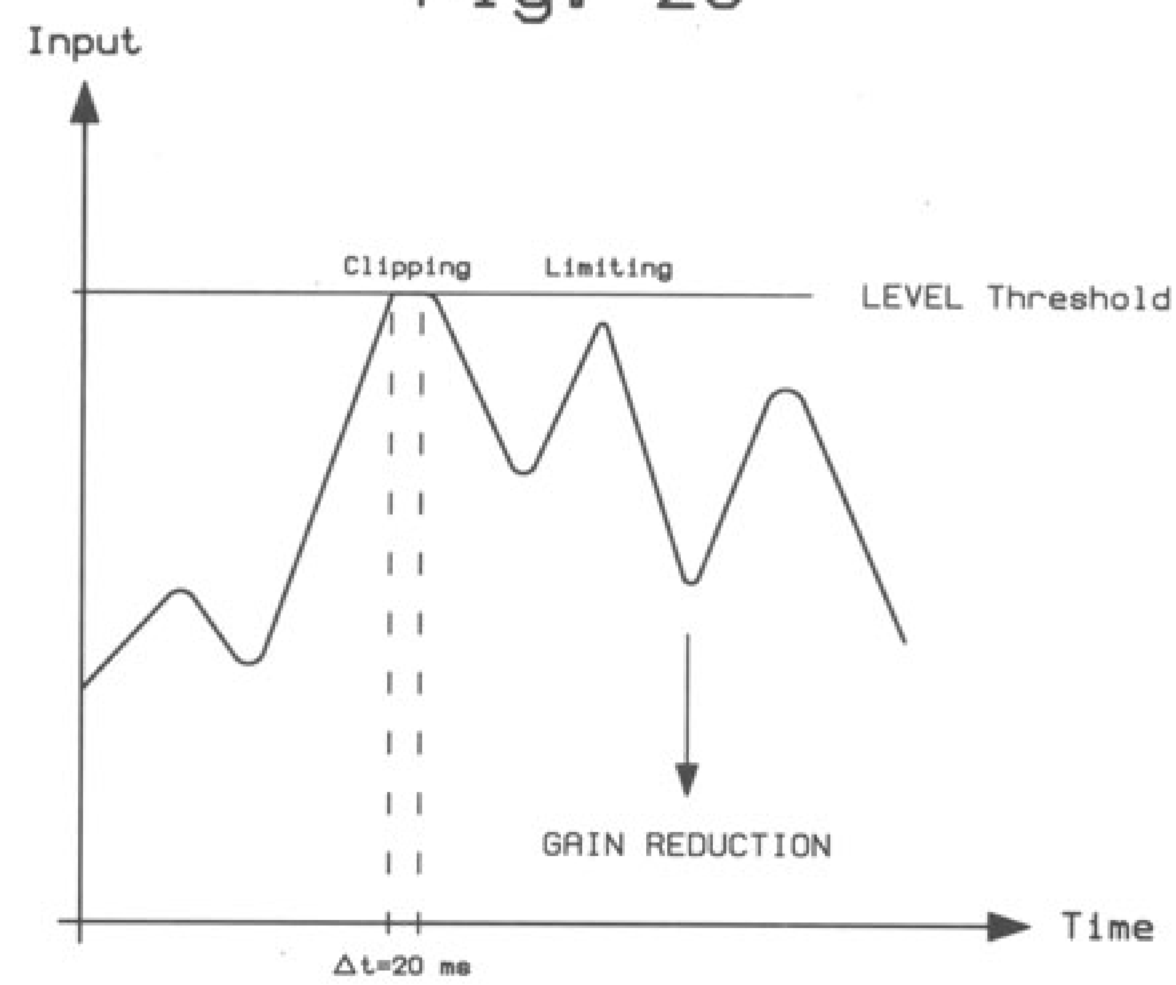


Fig. 28



The signal BEFORE gain control (fig. 27) and AFTER IGC peak limiting (fig. 28)

Clipper and programme limiter function interactively: depending on the programme material, one of the two controls are activated. In this way, optimum limiting is always achieved.

The working level of the Peak Limiter is set by the LEVEL control. If the PEAK LED illuminates the clipper is active. Increased limiting engages the programme limiter and reduces the overall level. This additional level reduction is indicated by the GAIN REDUCTION meter.

Please note that the Peak Limiter is designed to be used in conjunction with the Compressor section. It serves as a Peak Limiter for fast transients, which are unable to be captured by the Compressor section. The sole use of the Peak Limiter section can however, lead to distortion or pumping in programme material which contains very low frequency components.

7.0 APPLICATIONS

7

In this section, several typical applications of the Behringer COMPOSER are discussed. The following basic settings can resolve most dynamic problems. They are the ideal starting point.

Please take the time to study the application examples carefully, in order to be able to make full use of the COMPOSER's capabilities in future.

7.1 MAIN APPLICATIONS AND INITIAL SETTINGS

The main applications of the Behringer COMPOSER can be divided into three categories:

1. The Expander/Gate section is used to eliminate interference and to suppress background noise and leakage on individual tracks in multitrack recording.
2. The Compressor section is used to compress the programme material and to create special effects and unusual sounds, which are used for recording and musical performance.
3. The subsequent Peak Limiter section is designed to protect loudspeakers, tape recorders, transmitters etc. from being overloaded.

7.1.1 Compression/Levelling/Limiting/Clipping

Now that the functions of the individual sections have been clearly explained, we would like to acquaint you with more terms and relationships of the dynamics process.

Compression

A compressor converts a large dynamic level into a restricted range. The extent of the resulting dynamic level is dependent on the threshold, attack, release and ratio settings. As it is the desired effect of a compressor to increase a low level signal, generally the threshold is set low. The "inaudible" compression mode requires fast attack and release times and low ratios. The faster the chosen control times and the higher the compression ratio, the greater the effect on the short term dynamics. This fact is often used to achieve audible and creative sound effects.

Levelling

The levelling mode is used to keep output level constant, i.e. to compensate for long term gain changes, without affecting the short term dynamics. Normally, the threshold is set quite low in order to be able to increase low level signals. Levelling requires slow attack and release time, combined with a high ratio. Because of the very slow response time, levelling has no effect on signal peaks or short term changes in average level.

Limiting

The limiting function requires a fast attack time and a high ratio and release time setting, which is dependent on the specific use and the desired sound effect. As it is usually the task of a limiter to limit only high signal peaks, the threshold is usually set at a high level. The dynamic is reduced dependent on the ratio setting and on the degree by which the threshold point was exceeded. If the attack time is adjusted to control only the average level without affecting signal peaks above the threshold, this is referred to as the *programme limiter.* For this purpose the attack time will be set above 20 ms. If the attack time is further reduced in order to also control signal peaks, this is defined as the *peak limiter.*

Clipping

In contrast to the two previous mentioned limiters, the clipping mode features infinitely fast control times and an infinite compression ratio and creates an unsurpassable barrier ("brickwall") for all signals above a certain level. To be able to control the signal peaks, the clipping function radically cuts signals above the threshold, without affecting the amplitude of the original signal.

If used in normal applications, this function remains inaudible and under certain circumstances it can even lead to an improved sound, because cutting the transients creates artificial harmonics. If misused, clipping can cause very obvious and distasteful distortion, which in an extreme manner, will convert the signal's waveform into a square wave signal. This effect is often produced in guitar distortion devices ("fuzz boxes").

7.2 EXPANDER/GATE SECTION

The main task of the Expander/Gate is to "inaudibly" eliminate undesirable background noise from the usable signal. This assumes that there is a slight level difference between the usable signal and the noise floor, in order to be able to define the operating threshold of the Expander/Gate.

At the same time, the Expander/Gate must respond very quickly (have a very fast attack time) so that the signal's leading edge remains unaltered.

Because the Expander/Gate is self-adapting to the programme material, it will be possible to obtain more satiesfactory results with the new IRC *(Interactive Ratio Control)* circuit than with conventional expanders. When expansion occurs there are no common side effects due to the extremely smooth and unobtrusive action of the circuit.

When the Expander starts to operate, the EXP LED will illuminate. Because the expansion initially starts very smoothly, you may find yourself in a situation where the LED illuminates with little or no perceived gain reduction occuring.

7.2.1 Controlling Leakage In The Studio

Expander/gates are most commonly used to suppress undesirable leakage of sound from one track to another during recording or playback. They are usually used when recording drum kits, where the mics are very close to each other. High volume levels of individual instruments often cause considerable leakage onto all the adjacent mics and results in conflicting frequency and phase coherence problems, as well as unspecified sounds ("comb" filter effects). It is vitally important, that every instrument is recorded into a separate mic and that each mic is individually gated.

Patch the Behringer COMPOSER into a snare drum channel for example and adjust it so that triggering only occurs on snare hits. Each mic should be set to its maximum operating level, monitored (see KEY LISTEN switch) and the THRESHOLD level set so that each snare hit sounds acoustically clean and seperate, as though it was played on its own.

The optimum use of the Expander/Gate depends principally on microphone technique. Be particularly careful, when high frequency instruments are located to the side or rear of a cardioid microphone.

Most cardioids exhibit a sharply rising off-axis response characteristic at higher frequencies. If there is only a 2 or 3 dB difference between the on-axis and off-axis response in the 5 to 10 kHz region, cymbals may leak excessively into the tom mics and you may have hi-hat spilling all over the snare mic.

Please make full use of the directional characteristic of the mics, to acoustically exclude all other instruments as much as possible. Make sure that you do everything possible to achieve source separation with good microphone technique. Otherwise the Expander/Gate is not able to undertake clear acoustic separation.

Sometimes, it is necessary to prevent the Expander/Gate from responding to low frequencies (rumbles etc.), especially if a singer is moving the microphone around on a mic-stand. More information about this topic in chapter 9.2 "THE USE OF AN EQUALISER IN THE SIDECHAIN PATH".

7.2.2 Initial Settings For The Expander/Gate Section

Controls:	Settings:
THRESHOLD control:	-70 dB
RELEASE switch:	SLOW

Begin with very low threshold levels, so that the signal can pass through the unit unaffected. Now turn the control clockwise until all unwanted noise is removed and only the sound of the desired instrument can be heard.

To adapt the unit to the programme material properly, you can additionally choose between a SLOW or FAST release time. In the depressed position, the unit works at a slow release time.

Percussive material with little or no reverb, is processed in fast mode, whereas the slow mode is advantageously used for signals with long durations or signals with heavy ambience. You will find that a fast release time (FAST mode) is more preferable for acoustic separation of most percussive sounds, whilst cymbals and tom toms, normally benefit from the SLOW mode.

If the controls are set correctly, the drum sounds will be "dry", "sharp" and clearly defined. If you do not have enough mics (or COMPOSER channels!) to record each instrument separately, try to create sub-groups: put the snare and mid-toms together, and group the side-toms, bass drum and cymbals together with the help of a mixing console.

The aim is to set up the Expander/Gate and to position the group mics so that each strike on an instrument opens a specific mic and so only that instrument is recorded, whilst the other mics remain "muted".

7.2.3 Reducing Leakage In Stage Mics

The COMPOSER has many uses especially in live-work, on stage and in multi-miking situations: a well set up Expander/Gate can effectively suppress background noise, compressor type pumping noise and microphone leakage etc. without producing any undesirable side effects.

Expander/gates are commonly used for processing vocals. When specifically used with a compressor, the distance and position of the mic in relation to the singer is very critical: the further the distance, the more sensitive the mic is to background noise. Use the Expander/Gate section in the slow release mode to eliminate background noise "inaudibly", that occurs in pauses between the singing.

When used in live situations, leakage of miked instrumentation is substantially reduced, as well as other acoustic contaminants in various recording situations.

7.2.4 Reducing Feedback In Stage Mics

When a singer is using a vocal mic, their voice effectively stops other sounds from entering the mic. But in pauses between the singing, the mic will pick up noise from the house PA and monitors, which can lead to unpleasant feedback problems. If the COMPOSER is inserted into the mic channel, it will shut off the channel when it is not being used, reducing the possibility of feedback. Principally all mics should be included in this application.

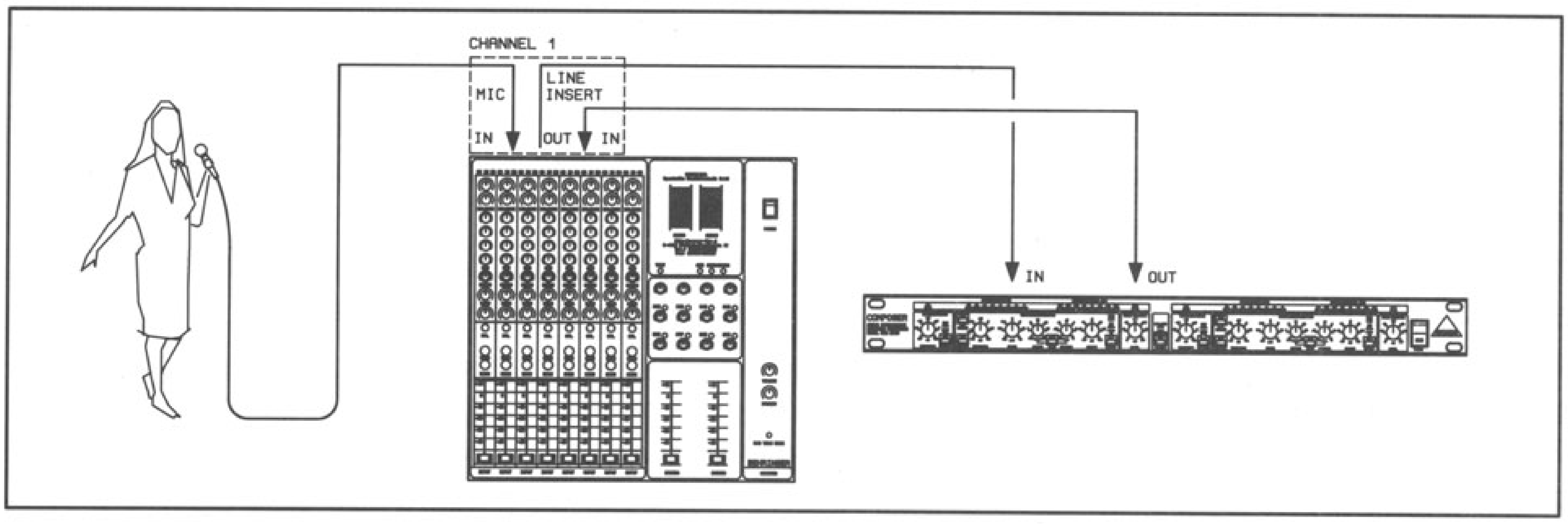


Fig. 29 Gating a stage mic

7.2.5 Noise Reduction On Effects Paths

The effects rack is one of the main overlooked sources of noise in a PA system or recording facility. The prices of reverb and delay units and harmonizers have fallen drastically over the last number of years, which have made these units a common feature of small studios and home recording installations. Installations where there are a number of units however, considerably increase the overall noise level drastically, so that the pleasure in acquiring a new sound effect is diluted in a short period.

It will be useful to use the Behringer COMPOSER as the last component in the chain of effects units and use the noise reduction function of the Expander/Gate section. We recommend that you use a slow release time in order to maintain the natural reverb.

7.2.6 Creative Use Of The Expander/Gate Section

In addition to the previously explained applications, the Expander/Gate can be used to change the sound characteristics. For example, the quality of ambience or reverberation created by an instrument within a room can be modified: when an instrument stops decaying, the reverberation of the instrument falls below the user-defined threshold. The reverberation can be controlled by using the THRESHOLD control and the RELEASE switch. The decay characteristics of the instrument can be controlled using the release switch, so that the natural characteristics of the instrument remains. Experiment with the effect, this control has on the decay of the instrument. In SLOW mode, the signal is gently faded out - in FAST mode, the duration of the reverberation can be removed completely.

7.3 COMPRESSOR SECTION

The task of a compressor is to reduce the dynamic range of programme material and to control the overall level.

The extensive controls of the Compressor section, provide a great range of dynamic effects: from musical and soft compression to limiting signal peaks, right up to extreme and effective compression of the overall dynamics.

For example, a low ratio and very low threshold setting can be used to achieve soft and musical processing of the general dynamics of the programme material.

Higher ratios, together with low threshold settings, create relatively constant volume (levelling) for instruments and vocals. High threshold levels generally limit the overall level of a programme. Ratios greater than 6:1 effectively prevent the output level from significantly exceeding the threshold point (provided that the OUTPUT control is in the 0 dB position).

Please note that the compression of the entire programme material (achieved by low threshold settings) sounds less natural with higher ratio settings. Ratio settings in the range of 4:1 and lower, effect the dynamics of the programme material less and are often used to compress the sound of a bass guitar, a snare drum or a vocal. Sensitive and moderate settings are generally used in mixing and for levelling of programme material in broadcast.

The new IKA (Interactive Knee Adaption) circuit prevents aggressive compression, created by high ratios, from sounding too unnatural. This is achieved with an *interactive* control function, which begins above the threshold level and introduces a "Soft Knee" curve characteristic in the range up to 10 dB above the threshold point. Beyond this range, the signal is subjected to linear ("Hard Knee") compression.

7.3.1 Initial Settings For The Compressor Section

Controls:	Settings:
IN/OUT switch:	IN
THRESHOLD control:	+20 dB
RATIO control:	3:1
AUTO switch:	IN
OUTPUT control:	0 dB
KEY EXT switch:	OUT
KEY LISTEN switch:	OUT

Rotate the THRESHOLD control anticlockwise until an appropriate amount of gain reduction is indicated on the GAIN REDUCTION meter. This operation will be accompanied by an audible drop in output level. The OUTPUT control should now be turned clockwise to reinstate the output level. The level of the unprocessed and the processed signal can be compared by pressing the IN/OUT switch and observing the INPUT/OUTPUT LEVEL meter.

Final adjustments of the controls can then be made to suit your particular requirements, including the RATIO, ATTACK and RELEASE controls. The auto function of the attack and release times, provides programme dependent dynamics processing which suits most standard uses. If a "condensed" or "wider" sound processing technique is required, the attack and release times can also be manually adjusted.

The experienced user will be in a position to specify parameters while in bypass mode and thus realize the effect before the unit is actually switched into operation. This is important during live situations, where a signal needs to be managed efficiently by the engineer, without the convenience of continual A/B comparison.

7.3.2 The COMPOSER As A Sound Effects Unit

In the early 1960's, musicians began looking at the recording process, as a way to create new sounds. The pumping effect which had been avoided by earlier engineers, suddenly became fashionable and was utilised as a creative tool. Laying the groundwork for many of the sounds which are now considered indispensable in contemporary music. In this role, the compressor is used because you can hear it working, and control of the dynamic range is of secondary importance.

The Behringer COMPOSER, with its extensive range of functions, is well suited to this application. Sound effects of this kind can be achieved using "extreme" settings. For this reason, set the threshold control to a fairly low level, the RATIO control to almost maximum and use the ATTACK and RELEASE controls to obtain the desired effect.

Experiment with all the controls in order to get a feel for their function!

7.3.3 The "Muffling" Sound Effect Of A Compressor

Quite often compressors are sometimes accused of "muffling" the sound, whilst at the same time reducing the dynamics. This fact should be investigated further. Bass frequencies contain most of the energy within music and therefore cause the compressor to reduce the overall dynamics. If the music also contains high frequencies along with the bass frequencies, these are also reduced in level. This is the reason why, in an extremely compressed recording of drums, the cymbals and high-hats are acoustically swamped by the sound of the snare or the bass drum. The same effect is experienced when processing reverberated or ambient sounds.

The solution to this basic problem is either to reduce the compression ratio or to slow down the attack time, so that the increasing high frequency transients pass through the compressor unhindered before the compressor takes effect. Although the COMPOSER's new IKA circuit is capable of reducing this side effect we recommend, in certain cases, adding a subtle amount of additional high frequency information to the processed sound. Here, an equaliser or exciter would be the ideal solution.

Please note that we supply a range of high quality equalisers and exciters. Contact us for further information!

7.4 PEAK LIMITER SECTION

Independent of all other control functions, the Peak Limiter section makes it possible to limit the overall output level of the COMPOSER. The peak limiter is designed to be used in conjunction with the Compressor section. Irrespective of their function, you can protect subsequent units from signal peaks, short term overloading and over modulation (transmitters etc.).

7.4.1 Initial Settings For The Peak Limiter Section

Control:	**Setting:**
LEVEL control:	OFF

The LEVEL control of the Peak Limiter sets the threshold level, so that subsequent units are protected from overloading. If the PEAK LED comes on regularly or is on constantly, the OUTPUT control of the Compressor section must be turned down, as this control sets the level of the signal, which is routed to the Peak Limiter section.

If this technique leads to an undesired drop in the overall level, it is recommended that you increase the compression: either, reduce the threshold level, or increase the compression ratio with the RATIO control. The OUTPUT control will compensate for a renewed drop in level.

8.0 SPECIAL APPLICATIONS

8

8.1 USING THE COMPOSER FOR RECORDING AND CASSETTE DUPLICATION

In the recording and duplication field, it should always be the goal to achieve an optimum recording level onto the recording media. Too low or too high recording levels lead to side effects such as noise, distortion etc. In mastering and multitrack recording, as well as in duplication, one should always pay full attention to utilize the full dynamic range of the tape recorder, DAT recorder etc. Principally, it is possible to control the recording level by "riding" faders, which means with low level signals, the gain is increased, whereas the amplitude of high level signal is reduced. It is obvious, that this method is insufficient, because especially in live recordings, the expected signal levels cannot be anticipated correctly. Especially with multitrack recordings, which are run under hectic circumstances, the signal level of all channels cannot be monitored and controlled at the same time. Generally with manual control, it is not possible to achieve satisfying recording results.

An automatic gain control system achieves better and more constant results. Use the COMPOSER by starting with the initial settings, and use its dynamic control functions in order to be able to drive an analogue, as well as a digital recording, noise and distortion free up to the limit of its maximum dynamic range.

8.1.1 The COMPOSER In Digital Recording And Sampling

In an analogue recording, too low recording levels lead to an increased noise level, whereas too high levels will cause a compressed and "squashed" sound. In extreme cases, it will cause distortion due to tape saturation. In contrast to analogue, side effects in the digital field always become extremely audible: with decreasing level, a tape previously recorded with insufficient level, looses resolution: the recording sounds "hard" and looses "atmosphere". With excessive level, the recording sounds harsh and heavy distorted. In order to avoid these effects, the Peak Limiter section of the COMPOSER should be placed before for example a sampler. As a result of this process, a digital recording or a sampling event can be optimally set in level without any problem.

8.1.2 The COMPOSER In Mastering

The mastering process is one of the most critical processing steps in recording. In this production step, it is the goal to achieve a "maximum level" copy of the recording, without any noise or distortion. In many applications it is further required to produce a high average volume. In the field of commercial media, this is apparent especially with records and cassettes, for example which are processed with high average volumes. Quite often in these cases, dynamics suffer dramatically, because the programme material has been compressed and limited too heavily. Using the Compressor and the Peak Limiter section of the COMPOSER allows you to drastically increase the overall volume, without audibly affecting the dynamics.

Proceed as follows:

1. Limit the dynamics of the programme material by 6 dB using the Peak Limiter section. By softly clipping just the transients, the real audio signal will not be limited, resulting in a higher headroom. The overall gain can now be increased by 6 dB, which leads to a higher volume. More than 6 dB should not be limited, otherwise side effects could become audible.
2. As an addition therefore, you should also use compression. It is recommended, that the compression is limited to the "first" 6 dB of the dynamic range only. A high threshold level in addition to the auto mode will give good results.

This effect is particularly noticeable with DAT recorders, whose level indicators achieve a response time of less than 1 ms. Set the DAT recorder at unity and now reduce the LEVEL control of the peak limiter until the PEAK LED starts to illuminate. The "cut" signal peaks cause a reduced recording level of about 6 dB, which is visible on the level indicators of the DAT recorder. Now increase the recording level of the recorder back to unity. The result is a clearly louder recording without any loss of sound.

8.2 THE COMPOSER AS A PROTECTIVE DEVICE

Sound system distortion is usually a result of amplifiers and loudspeakers being driven beyond their limitations by signals clipping. The signal limitations that occur, lead to unpleasant distortion that is dangerous to the speakers.

A speaker diaphragm is required to accelerate, slow down, smoothly change direction and accelerate again in normal operation. Distorted operation (clipping) leads to instant acceleration, instant stop, change of direction and instant acceleration again. Since speaker diaphragms are subject to the laws of physics, they will not take this kind of punishment for long: the diaphragm will either brake up or its voice coil may overheat.

In addition to the damage caused by sustained overload, the speaker may also be damaged by an occasional high level overload. For example, the sound of a microphone falling onto a hard floor. Even if this type of transient does not destroy a speaker outright, it may damage the speaker surround in such a way, as to cause mechanical abrasion and future failure. !t is recommended that you use the Behringer COMPOSER in order to protect the speaker. "Brick Wall" peak limiters are not normally necessary for PA systems, as amplifiers and loudspeakers are tolerant of short signal peaks. Nevertheless, conventional limiters have to be generally driven far beyond the headroom limit of an amplifier, in order to limit the level and length of the transients responsible for overloading the system. The disadvantage of this principle is that the unit's full range cannot be completely used.

If an increase in the average level of up to 3 dB is attained with the COMPOSER's IGC Peak Limiter, this means that you effectively double the power amplification. The COMPOSER can act in this way to convert a PA system of 5,000 Watts into a distortion free 10,000 Watts system.

The following instructions will help you to integrate the unit into your system.

8.2.1 Protection Of A System With A Passive Crossover

If your sound system incorporates a passive crossover network (included in the loudspeaker case), insert the Behringer COMPOSER between your mixing console output and the power amplifier input.

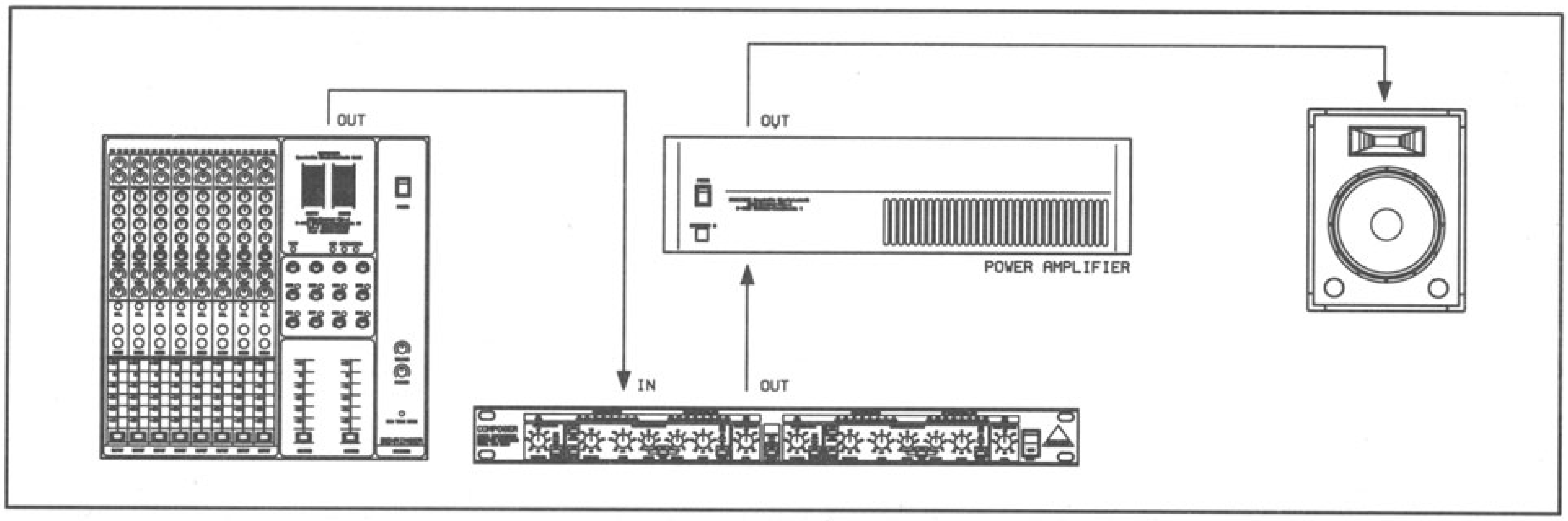


Fig. 30 Integrating the Behringer COMPOSER into a system with a passive crossover network

8.2.2 Protection Of A System With An Active Crossover

For systems using active crossovers, there are two ways to use the Behringer COMPOSER. As shown in fig. 31, the unit may be inserted between the console output and the crossover input. In this application, the Behringer COMPOSER will process the entire audio frequency spectrum.

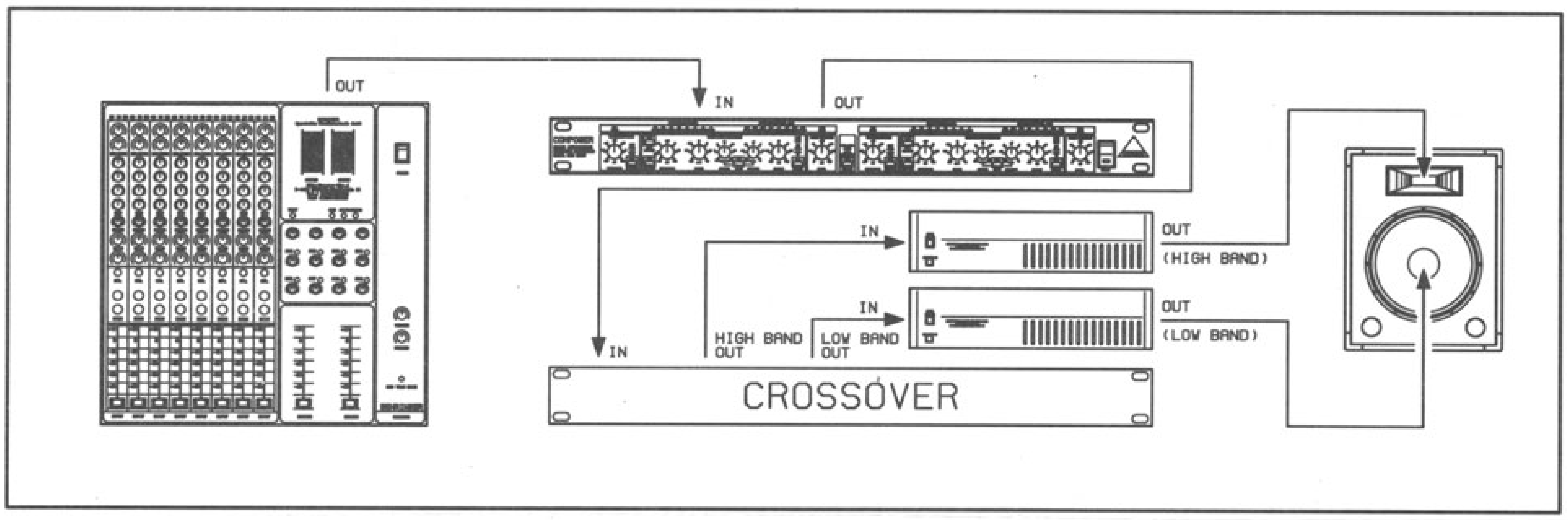


Fig. 31 The Behringer COMPOSER in a two way system

Alternately, the Behringer COMPOSER can be inserted between the output of an active crossover and the input of a power amplifier. In this application it will only affect a specific range of frequencies.

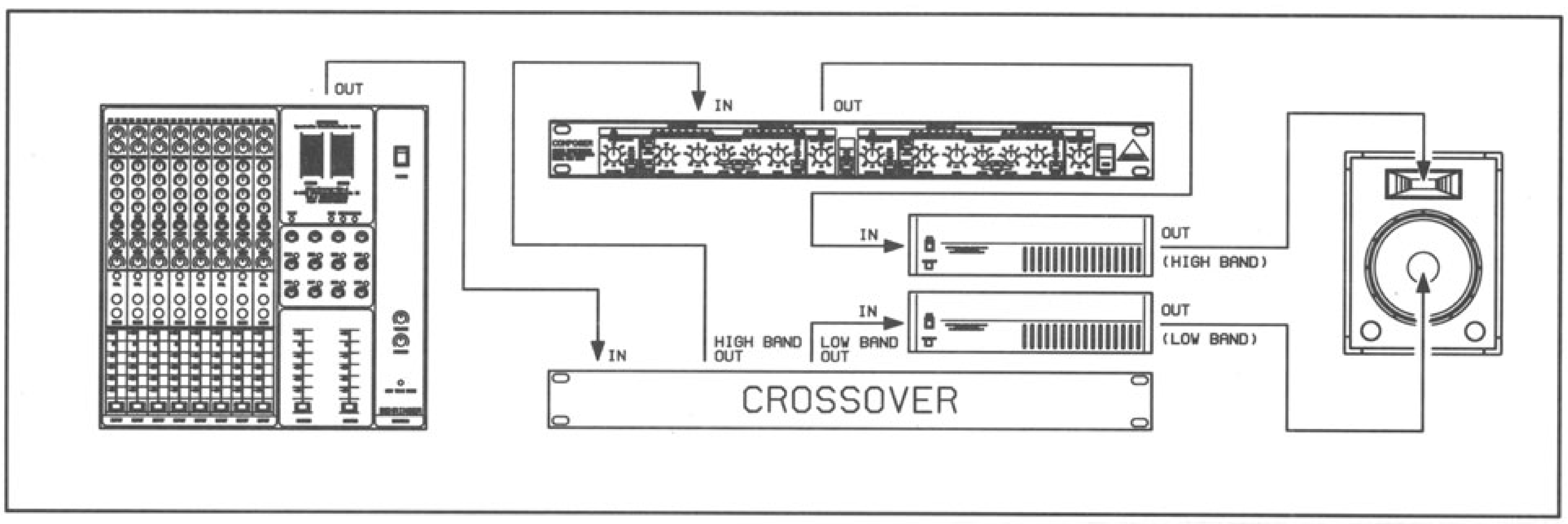


Fig. 32 Compressing the high frequency range with the Behringer COMPOSER

8.2.3 Improving The Sound Of A Processor System

A processor system is understood as a PA system which contains a special active crossover whose outputs are linked via separated power amplifiers to the loud speakers. Each band has its own limiter whose task it is to limit dangerous signal peaks to a certain level. This process avoids overloading the subsequent power amplifier or destruction of the loud speaker.

The crossover frequencies in the crossover unit are further changed during high signal levels to achieve a "loudness contour" suited to the human hearing. But in many cases, this function leads more to a disturbance than to an improvement of the sound quality.

If the COMPOSER is preceding this system, the signal peaks can be eliminated before they reach the limiters of the processing system. The sound quality therefore remains natural and free of side effects caused by the changing frequencies of the crossover.

8.3 USING THE COMPOSER WITH TAPE RECORDERS

The Behringer COMPOSER can be used to prevent saturation of magnetic tape and to improve the signal-to-noise ratio of the tape machine.

In professional recording studios, the saturation level of the tape, system headroom and the output level of the console are all known quantities, making the application of limiting and compressing very easy. Limiting the audio lévels, allows for a higher nominal level of signal to tape, so that the signal-to-noise ratio can be considerably improved.

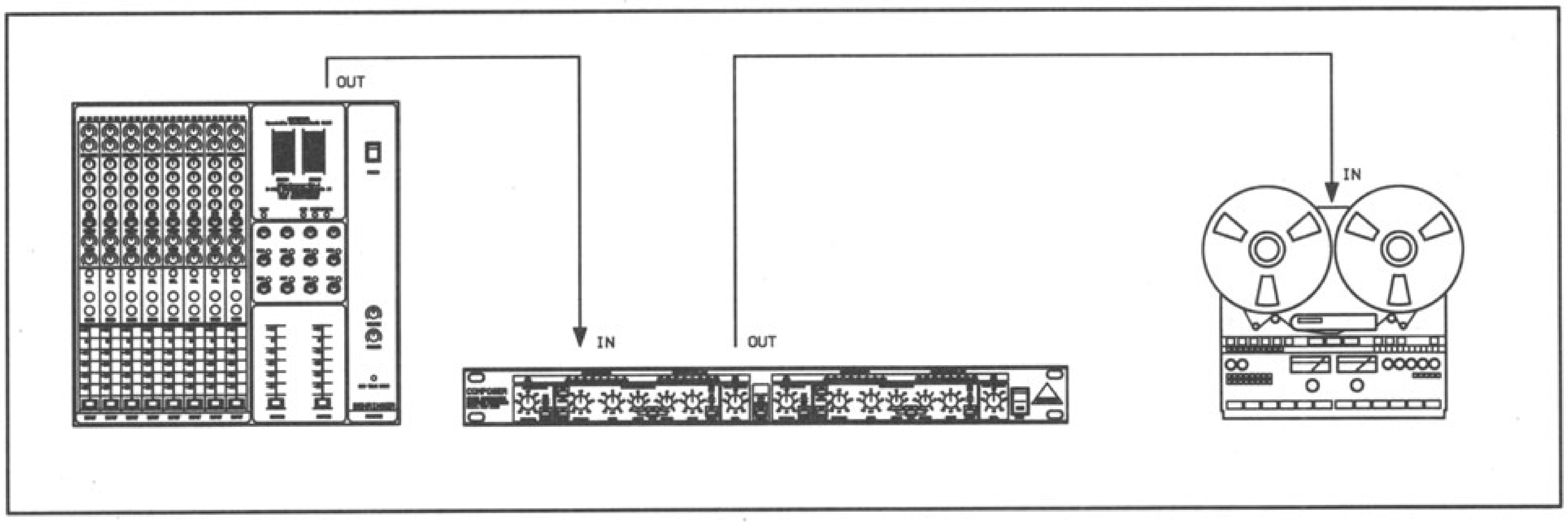


Fig. 33 Using the Behringer COMPOSER to avoid distortion due to tape saturation

8.4 THE COMPOSER IN BROADCAST

The main aim of processing sound recordings for commercial radio and television is to achieve a maximum transmission volume at all costs. Owners of these radio and television stations strive to get bigger audience ratings, because principally, radio programmes are preferred whose reception is louder than the average. By achieving a bigger audience, the broadcast station gains more money from the increasing number of promotion companies.

What is volume?

Volume is defined as the relationship between the average level of programme material to peak-to-peak level, in response to amplitude and duration. The higher the average level and the time it remains at a high level, the louder the programme material will be perceived by the listener.

If you want to run your broadcast station at maximum average volume, proceed as mentioned in chapter 8.1.2 "The COMPOSER In Mastering". Please make sure that the maximum peak level is below the threshold of the transmitter's limiter, otherwise this could lead to very hard and audible use of the transmisson limiters. Keep in mind that a heavy increase in average volume by means of compression always leads to a loss in dynamics and an increased perception of side effects.

The moderate use of the Compressor and the Peak Limiter sections of the COMPOSER result in higher average volumes, free of distortion.

8.5 USING THE COMPOSER TO CHANGE SOUND

8.5.1 Reshaping Sampled Sounds

With the help of the Behringer COMPOSER, existing or new sampled sounds can be brightened up, changed or used to create new sounds. The attack times and the dynamics of the sounds can be changed as desired.

8.5.2 Altering The Texture Of Musical Instruments

It would be impossible to mention here all the ways that compression can be used to create new sounds. However, some typical uses are listed below:

1. Creating a "fatter" snare or kick drum sound
2. "Thickening" acoustic guitars and electric pianos
3. Adding more "punch" to bass guitars
4. Lengthening the sustain of electric guitars etc.

9.0 EXTERNAL SIDECHAIN APPLICATIONS

9

9.1 THE "KEY EXTERNAL" FUNCTION

The Behringer COMPOSER offers an exceptionally usable external facility by using the key external function. By activating the KEY EXT switch, the COMPOSER's control path is disconnected from the audio input and therefore interrupted (see chapter 3.0 "BLOCK DIAGRAM"). The audio input is routed to the KEY SEND output and the KEY RETURN input now receives the new control signal which is derived from an inserted effects processor.

Please note the correct wiring for mains powered units in order to avoid ground loops, as the key inputs and outputs are unbalanced. The working level of external units must be at line level (-20 to +10 dBu) and must be at unity gain.

9.2 USING AN EQUALISER IN THE SIDECHAIN PATH

It is very common to make the response threshold of a compressor frequency-dependent, where a graphic or parametric equaliser is connected to the sidechain path. To retain the threshold setting of the COMPOSER, *unwanted frequencies* should be *reduced* by an equaliser and the *desired frequencies should be kept at the same level.* Should for example, the compressor be controlled by a narrow mid-frequency band, it is advisable to lower the bass and treble controls. The middle frequency control remains at unity gain.

9.2.1 The COMPOSER As A "De-Esser"

"De-essing" is a special application of frequency selective compression. A problem often encountered in recording, is the sibilant (Ssss) sound of the human voice. High frequency, sibilant sounds and pops can produce very high energy levels which can sometimes cause an otherwise normal and undistorted voice to sound very harsh, shrill and sometimes unintelligible. The solution is frequency conscious compression or limiting. The unit responds only to selected frequencies and reduces the level temporarily, as soon as sibilant sounds or pops are detected.

If the detector circuit registers an excessive amount of high frequency information within the programme material, as in a normal compressor, the VCA is activated and the overall level is reduced. As this type of compression affects the whole frequency range, this process is called *broadband* de-essing.

Please note that this type of frequency selective compression is very different from simple, fixed equalisation using notch filters, since de-essing has no effect on the signal except at the instant the sibilant occurs. The general frequency response is principally not affected during this process.

When de-essing, simply insert an equaliser not into the audio path but into the sidechain path of the Behringer COMPOSER. The equaliser is inserted between the KEY SEND output and the KEY RETURN input of the Behringer COMPOSER. While the KEY EXT switch is depressed, the equaliser is inserted into the sidechain loop and controls the unit. With the help of the key listen function, the centre frequencies of the equaliser are then adjusted exactely to match the frequencies of the sibilant sounds. All other frequencies are filtered out, so that with maximum attenuation of these frequency bands, along with a correctly adjusted threshold point, the unit responds solely to the selected signal being produced by the equaliser. The level of the sibilant sounds can therefore be effectively limited.

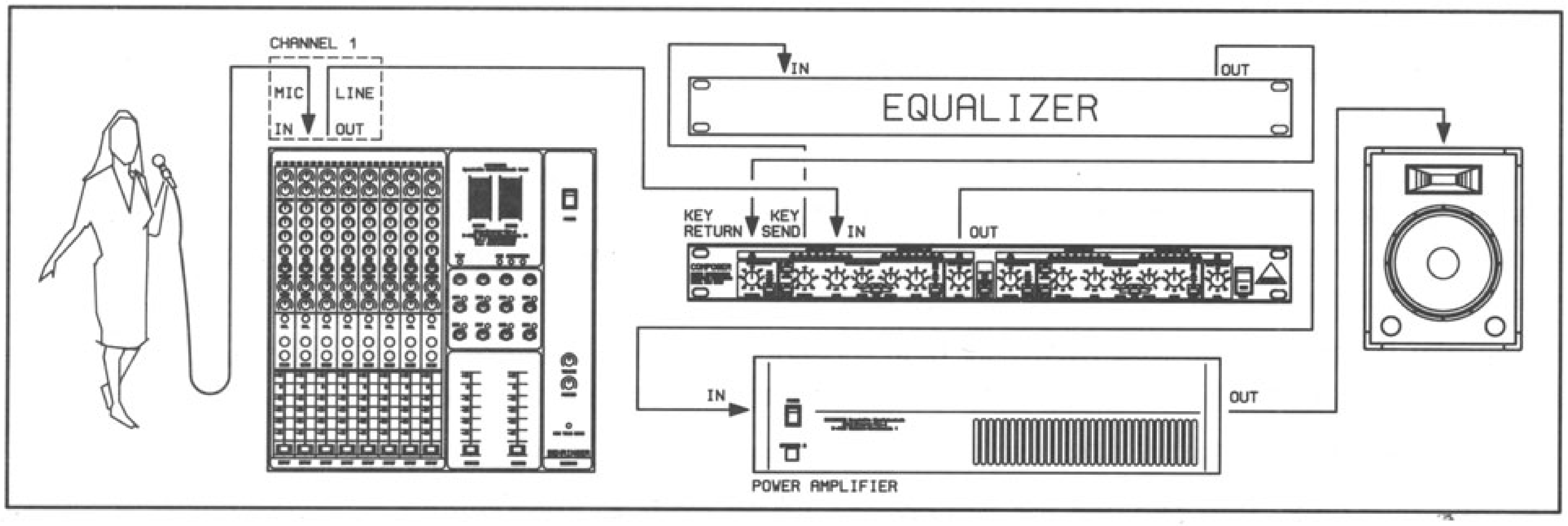


Fig. 34 "De-essing" using the Behringer COMPOSER

Initial Settings For The De-Esser Functions

Controls:	Settings:
KEY EXT switch:	IN
KEY LISTEN switch:	OUT
THRESHOLD control:	+20 dB
RATIO control:	∞
AUTO switch:	OUT
ATTACK control:	0.05 ms
RELEASE control:	100 ms
OUTPUT control:	0 dB
LEVEL control:	OFF

1. Turn the THRESHOLD control anticlockwise until the GAIN REDUCTION meter shows a appropriate drop in level.
2. Now press the KEY LISTEN switch and adjust the equaliser corner frequencies (generally 6 - 10 kHz) by monitoring, until it is within the range of the sibilant.
3. Release the KEY LISTEN switch and recalibrate the THRESHOLD control, so that the unit reacts only when the sibilant sound occurs.

Level compensation using the OUTPUT control is not necessary. Although the recommended attack and release times for this function are proven, the time parameters can be adjusted if necessary to achieve maximum results. The AUTO function should not be used.

9.2.2 Frequency Selective Filtering Of Unwanted Signals

Based on the set-up described in the de-esser section, the unit may also be used to eliminate rumble, hum and equipment noise (air-conditioning systems, camera noise etc.).

Using the KEY LISTEN switch, adjust the frequencies of the equaliser to match the unwanted frequencies and use a peak filter with a high slope. Take care to decrease the amplitudes of the unrequired frequencies. Proceed now as described in the previous chapter 9.2.1 "The COMPOSER As A De-Esser".

This will result in compression of the selected frequencies and thus a decrease in the gain of the programme material.

9.2.3 Suppressing Instruments During Recording

Another function of the Behringer COMPOSER allows helpful correction of previously recorded material.

If for example an excessively loud bass drum needs to be suppressed, reduce all the equalisers frequency bands above 150 Hz. This setting causes frequency specific compression, which reacts as soon as increased energy is detected in this band. By increasing the threshold level, the compression can be made to react to loud hits only.

Generally, it can be said that relatively high threshold settings prevent the overall sound from being impaired and lead to the compression of solo instruments or very loud sounds.

9.2.4 Emphasising Musical Instruments During Recording

On the other hand, you can use the Behringer COMPOSER to bring out an instrument solo or a lead vocal in a cluttered mix.

Using the KEY LISTEN switch, match the frequencies of the equaliser to the frequencies of the instruments to be emphasised and for this, it is best to use a notch filter with a high slope. Please make sure that in this application, you *only reduce the amplitude of the selected frequencies.*

The compression results in a subjective decrease in the volume of the overall programme material. Only the selected frequencies coming from the equaliser remain uncompressed and are therefore perceived as being louder. This inverse type of compression also helps to emphasise instruments during low level passages, so that they become more pronounced.

9.2.5 Reducing Feedback In PA Systems

A common procedure in sound system set-up is equalising the acoustics to remove feedback. This is generally accomplished by turning up the system gain to purposely induce feedback, searching for the centre frequency of the feedback and then equalising at that frequency to remove the feedback.

Once this feedback has been attenuated, the system gain is again increased to induce another feedback point and the whole procedure is repeated until the engineer is satisfied that the significant problem frequencies have been corrected. In spite of this equalising process, feedback remains a difficult problem. Often enough, acoustic changes occur as the audience enters the room, which again leads to feedback problems. In addition, the frequency response of the whole system is modified and thus affected by equaliser operation.

Dynamic feedback control is a better solution. Similar to the previously mentioned de-esser application, an equaliser is not inserted into the audio path but into the sidechain path of the Behringer COMPOSER. To effectively suppress feedback, the centre frequency of the equaliser is correctly adjusted to match the room's resonant frequency. This selected frequency now controls the Behringer COMPOSER.

The signal coming from the equaliser is applied to the sidechain input, while the audio signal is routed through the Behringer COMPOSER.

As soon as feedback occurs, the unit temporarily reduces the system gain and thus effectively suppresses the feedback. In contrast to the technique mentioned above, the frequency response of the PA system is not affected in any way at all.

The use of the Behringer COMPOSER in this application can eliminate the possibility of speaker or ear damage.

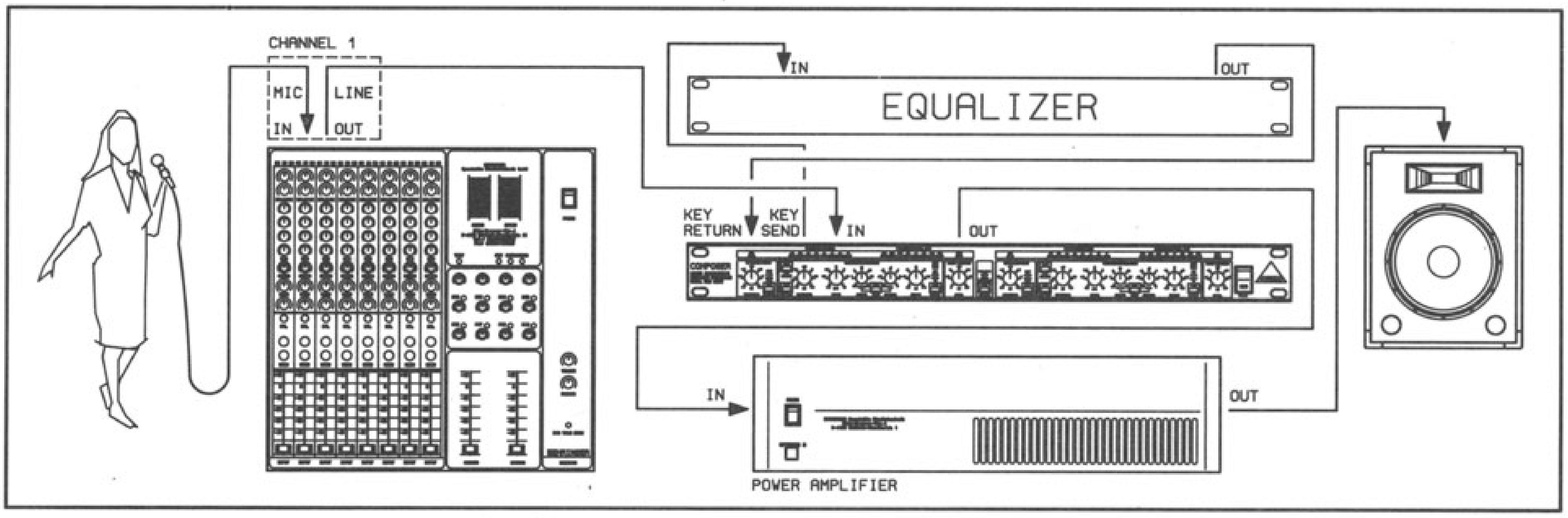


Fig. 35 Reducing feedback in audio systems using the Behringer COMPOSER

9.3 ANTICIPATED COMPRESSION

If you feed the audio signal directly into the KEY RETURN input and send the audio signal through a delay before the audio input, the Behringer COMPOSER can anticipate the need for gain change. With experimentation, the effect can create a "zero" attack time at a given frequency. Additional delay beyond this "zero" attack time will produce a special sound effect, similar to the dynamic envelope inversion you may already be familiar with from reverse tape playback.

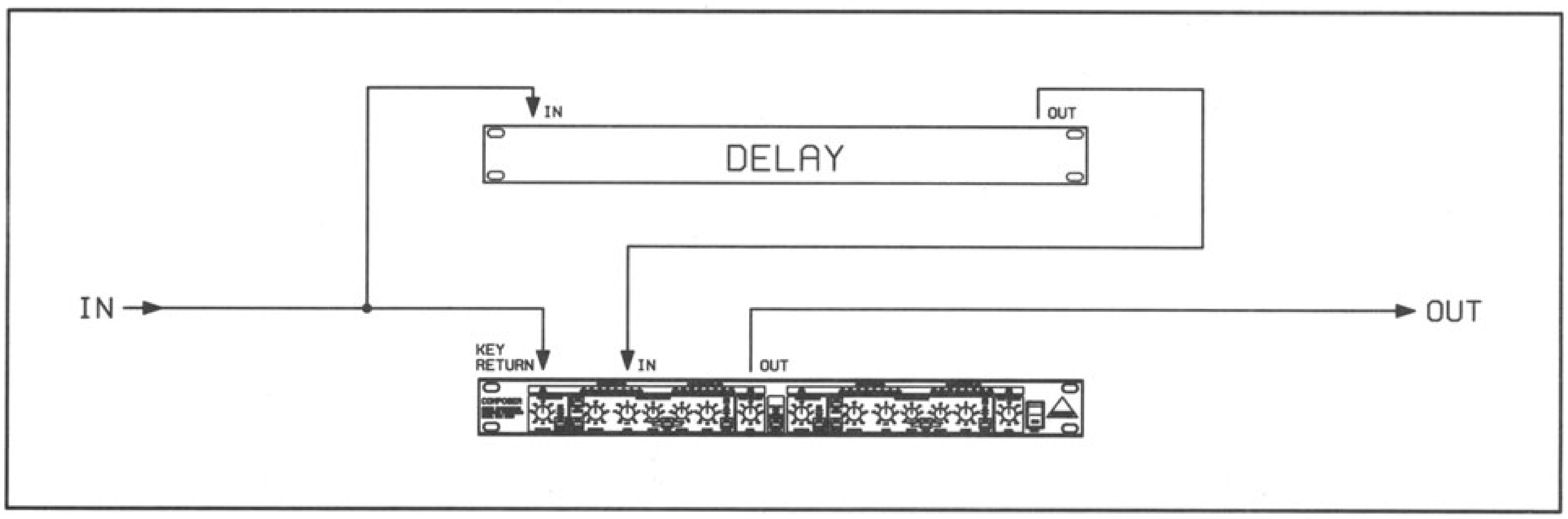


Fig. 36 Anticipated compression using the Behringer COMPOSER

9.4 "VOICE-OVER" COMPRESSION ("DUCKING")

The Behringer COMPOSER can be used to automatically reduce music to a background level, when an announcer is speaking through a microphone. For this purpose, the Behringer COMPOSER is used as an automatic fader and is controlled by the announcer's microphone, which is connected to the KEY RETURN input via a preamplifier. The music output and the announcer's voice, are then mixed. This application is known as "voice-over" compression or "ducking" and is commonly used in discos, radio stations etc.

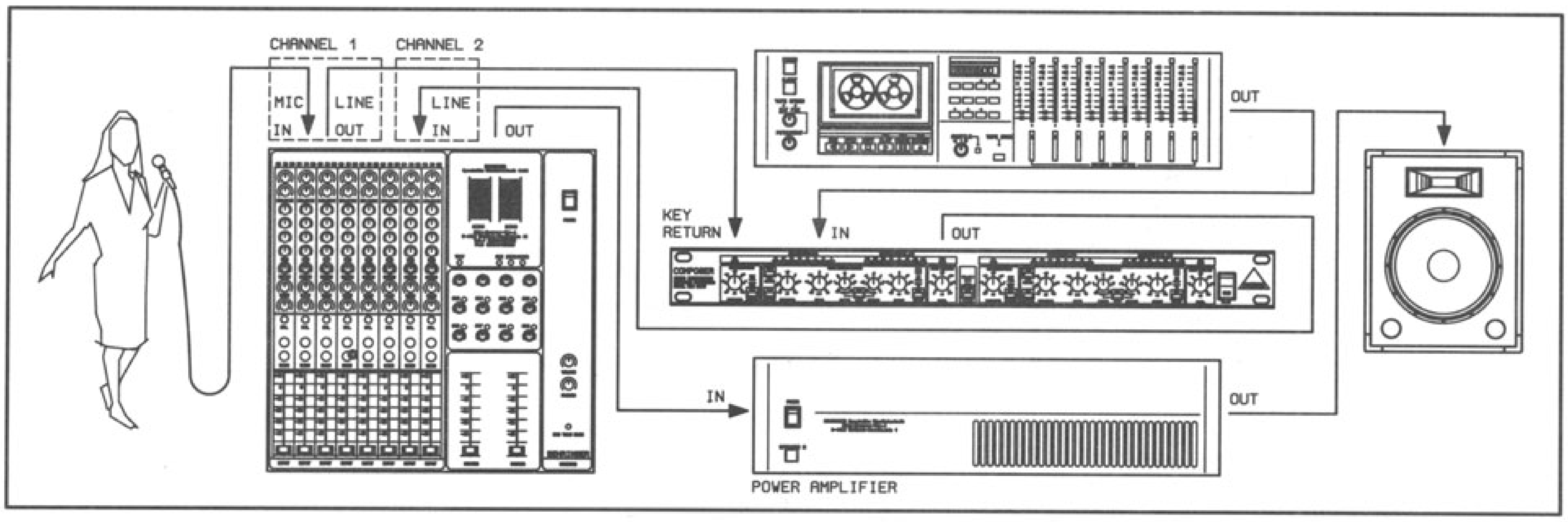


Fig. 37 "Voice-over" compression using the Behringer COMPOSER

9.5 TRIGGERING ADDITIONAL SOUNDS FROM A RHYTHM TRACK

This technique is used to give a rhythm track more "punch". For this purpose, only the Expander/Gate section is required and the Compressor and Peak Limiter sections are switched off. The bass guitar track is connected to the audio chain of the Behringer COMPOSER, whilst the bass drum is connected to the KEY RETURN input. By activating the KEY EXT switch, the bass guitar is now triggered by the bass drum.

Another application allows the sound of the bass drum to be supported or extended by other instruments (synthesizers etc.), where the bass drum is used to trigger a new sound, which is then mixed into the track.

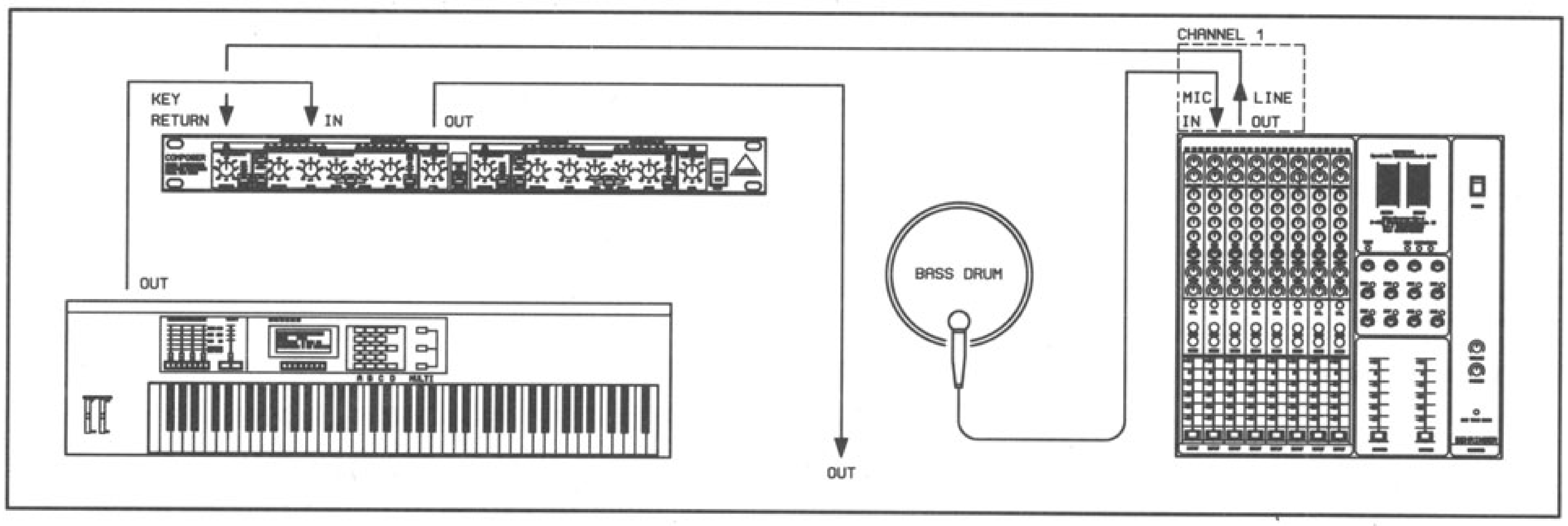


Fig. 38 Triggering a keyboard sound using a bass drum

10.0 SPECIFICATIONS

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AUDIO INPUT

Type	RF filtered, servo-balanced input
Impedance	80 kOhms
Nominal Operating Level	+4 dBu/-10 dBV switchable
Max. Input Level	+20 dBu balanced and unbalanced
CMR @ 1 kHz	›40 dB

KEY INPUT

Type	DC de-coupled, unbalanced input
Impedance	›20 kOhms
Max. Input Level	+20 dBu

AUDIO OUTPUT

Type	Electronically servo-balanced output stage (optional transformer-balanced). Automatic level correction for unbalanced use (correction: 6 dB).
Impedance	‹40 Ohms, balanced and unbalanced
Max. Output Level	+26 dBm balanced, +20 dBm unbalanced
Bandwidth	5 Hz to 50 kHz, +0, -1 dB
THD @ +4 dBu	0.01 % typ.
THD @ +20 dBu	0.1 % typ.
IMD (SMPTE) @ +10 dBu	0.01 % typ.
Noise & Hum, unity gain	›-94 dBu
Noise & Hum, fully off	›-97 dBu
Crosstalk @ 20 kHz	›-85 dBu
CMR @ 1 kHz	›60 dB

KEY OUTPUT

Type	DC de-coupled, unbalanced output
Impedance	‹150 Ohms
Max. Output Level	+20 dBu

EXPANDER/GATE SECTION

Type	IRC (Interactive Ratio Control) Expander
Ratio	programme dependent
Threshold	variable (-70 to +20 dB)
Attack	‹1 ms/50 dB
Release	variable (SLOW: 100 ms/1 dB , FAST: 100 ms/1 dB)

COMPRESSOR SECTION

Type	IKA (Interactive Knee Adaption) Compressor
Threshold	variable (-40 to +20 dB)
Ratio	variable (1.2:1 to ∞ :1)
Attack	variable (0.5 to 100 ms/20 dB)
Release	variable (0.05 to 5 s/20 dB)
Auto	programme dependent attack and release
Output	variable (-20 to +20 dB)

PEAK LIMITER SECTION

Type	IGC (Interactive Gain Control) Peak Limiter
Attack (Clipper)	"zero" attack
Release (Programme Limiter)	approx. 1 s

FUNCTION SWITCHES

In/Out	Relay controlled hard-bypass
Key Extern	Switching to the external key input
Key Listen	Monitoring the external key input
Couple	Linking both channels for stereo operation

INDICATORS

8 element GAIN REDUCTION meter	1/2/4/6/10/15/20/30 dB
8 element INPUT/OUTPUT LEVEL meter	-20/-15/-10/-5/0/+5/+10/+15 dB
LED indicator for each function switch	

POWER SUPPLY

Mains Voltages	100-120/200-240 VAC 50-60 Hz
Power Consumption	9 Watts
Fuse	160 mA (100-120 V); 80 mA (200-240 V) slow-blow
Mains Connection	Standard IEC receptacle

PHYSICAL

Dimension	1 3/4" (44.5 mm)H * 19" (482.6 mm) * 8.5" (217 mm)
Net Weight	3 kg
Shipping Weight	4.3 kg

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11.0 WARRANTY

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§ 1 WARRANTY REGISTRATION

Warranty Registration must be completed and returned to Behringer GmbH within 8 days from the date of purchase.

§ 2 WARRANTY

Behringer GmbH warrants the materials, workmanship and proper operation of this Behringer product for a period of five years from the original date of purchase. If any defects are found in the materials or workmanship, or if the product fails to function properly within the specified warranty period, Behringer GmbH will repair or replace the product, at its discretion.

§ 3 RETURN AUTHORIZATION NUMBER (RA)

1. To obtain factory service call Behringer GmbH for a Return Authorization Number. RA numbers are necessary for proper tracking of your product. Call:

   Tel (0)21 54/42 85 21 (10 am to 5 pm weekdays (MET))

2. The product must be returned in its original shipping carton, freight prepaid to:

   BEHRINGER Spezielle Studiotechnik GmbH
   Otto Brenner Str. 4
   FRG-47877 Willich/Munchheide II
   Federal Republik of Germany

§ 4 WARRANTY AUTHORITY

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Behringer GmbH may, at its discretion, require proof of the original date of purchase (dated copy of original retail dealer`s invoice). Final determination of warranty coverage lies solely with Behringer GmbH. Any Behringer product deemed eligible for repair or replacement under the terms of this warranty will be repaired or replaced within 30 days of receipt of the product at Behringer`s factory. Products which do not meet the terms of this warranty will be repaired and returned C.O.D. with an invoice for labour, materials, return freight and insurance. Products repaired under warranty will be returned freight prepaid by Behringer GmbH, to any location within the boundaries of the Federal Republic of Germany. Outside of Germany, products will be returned freight collect.

§ 5 WARRANTY TRANSFERABILITY

This warranty is extended to the original purchaser, but it is not transferable to anyone who may subsequently purchase this product.

§ 6 LIMITATION OF LIABILITY

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