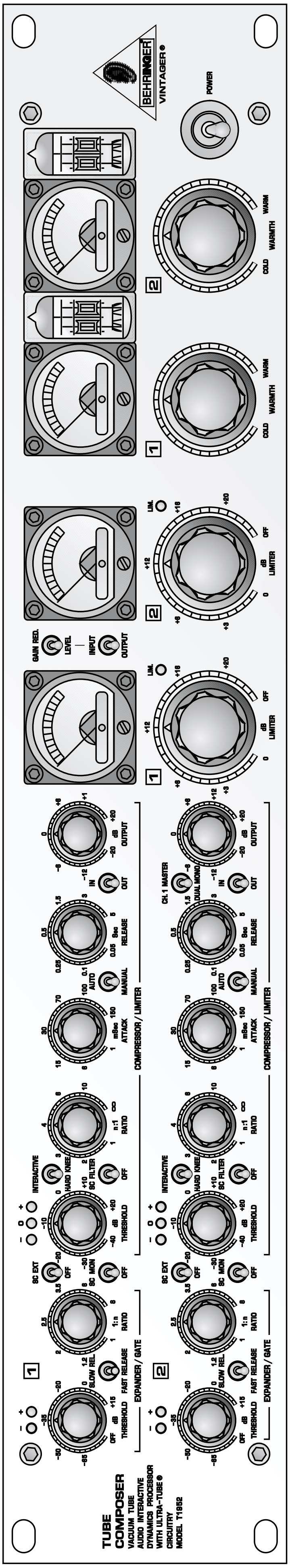
TUBE COMPOSER® T1952



User's Manual

ENGLISH

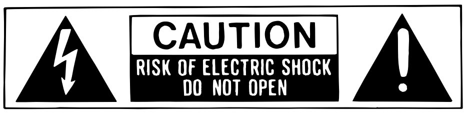
Version 1.1 October 2001

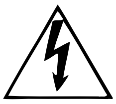
www.behringer.com

SAFETY INSTRUCTIONS

CAUTION: To reduce the risk of electric 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 electric 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 ventilation. 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 appliances (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.
Debris and Liquid Entry:
Care should be taken that debris and/or liquids do not enter 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
- Debris or liquid has entered 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 dropped, or the enclosure damaged.
Servicing:
The user should not attempt to service the appliance beyond that which is described in the operating instructions. All other servicing should be referred to qualified service personnel.

FOREWORD

Dear Customer,

We thank you for expressing your confidence in BEHRINGER products by purchasing the BEHRINGER TUBE COMPOSER. It is one of my most pleasant tasks to write this preface, as our engineering team has made it possible to enhance the traditional tube circuitry design (particularly for our VINTAGER series of products), and adapt it to meet the high sound quality and dynamics requirements of modern, pro-level audio technology. The fact that we are still fascinated by “antique” tube radios and amps as well as the fine and warm tonal character that we usually associate with them, are the reasons why vacuum tubes have kept their ground even in state-of-the-art circuit topologies used especially in professional audio technology or high-end devices. We are particularly proud that we have found an extremely effective symbiosis between solid-state and tube technologies making them affordable to anybody interested in audio technology. As always, our top-priority concern when developing this device was the demanding end user, in other words: you. It was our major goal to meet your demands. Sure, it meant a lot of hard work to develop such a product, but the fun has made it all worthwhile. The shine in the eyes of the many interested musicians at the Music Fair 1997, when they saw our VINTAGER models for the first time, was a lasting incentive driving our development efforts.

It is our philosophy to share our joy with you, because you are the most important member of the BEHRINGER family. With your highly competent suggestions for new products you’ve greatly contributed to shaping our company and making it successful. In return, we guarantee you uncompromising quality (manufactured under the ISO9000 certified management system) as well as excellent technical and audio properties at an extremely favorable price. All of this will enable you to fully unfold your creativity without being hampered by budget constraints.

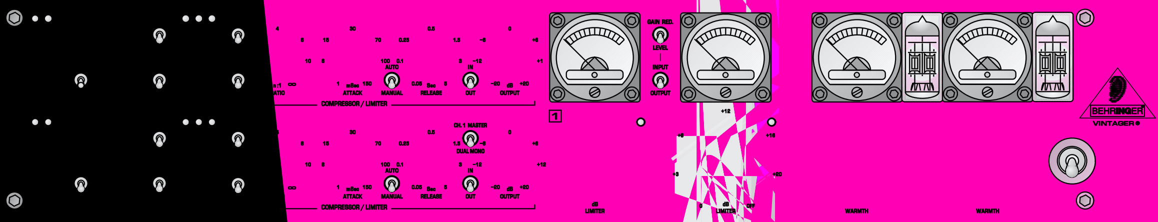
We are often asked how we can make it to produce such high-grade devices at such unbelievably low prices. The answer is quite simple: it’s you, our customers! Many satisfied customers means large sales volumes enabling us to get better conditions of purchase for components, etc. Isn’t it only fair to pass this benefit back to you? Because we know that your success is our success, too!

I would like to thank all people whose help on “Project TUBE COMPOSER” has made it all possible. Everybody has made very personal contributions, starting from the designers of the unit via the many staff members in our company to you, the user of BEHRINGER products.

My friends, it’s been worth the effort!

Thank you very much,

Uli Behringer



...ic processor with selected 12AX7 tubes

...tions: compressor/limiter, gate/expander and peak limiter

...e dosage of desired tube effect

...) program-adaptive compression curcuitry

...Hard Knee” compression modes

...omatic attack and release time adjustment

...ander/gate circuitry for extremely smooth noise reduction

...imiter combines the clipper and the program limiter approach to provide ...ximum allowable level with minimal distortion

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1. INTRODUCTION

In purchasing the new TUBE COMPOSER T1952, you have acquired an extremely efficient and universal dynamics processor, which combines the precision of the solid state technology with the warmth and liveliness of tube technology. The most commonly used dynamic functions are all present. Every channel has its own independent compressor/limiter, an expander/gate and a peak limiter complemented by an extremely low-noise tube stage. The precision and flexibility of the functions are the main outstanding features of this high end unit.

Future proof BEHRINGER technology

The philosophy behind BEHRINGER products guarantees a no-compromise circuit design and employs the best choice of components. The op-amps, type 4580, used in the TUBE COMPOSER are chosen for their superior signal-to-noise ratio, low distortion and linear performance. Additionally, the TUBE COMPOSER uses high quality resistors and capacitors with very tight tolerances, high-grade switches as well other selected components.

Modern manufacturing and design principles

Our engineering team has made it possible to enhance the traditional tube circuitry (particularly for our TUBE COMPOSER) and adapt it to meet the high sound quality and dynamics requirements of modern, pro-level audio technology. The fact that we are still fascinated by “antique” tube radios and amps as well as the fine and warm tonal character that we usually associate with them, are the reasons why vacuum tubes have kept their ground even in state-of-the-art circuit topologies used especially in professional audio technology or high-end devices. We are particularly proud that we have found a highly effective symbiosis between solid-state and tube technologies making them affordable to almost anybody in audio technology.

With the exception of 2 select 12AX7/ECC83 tubes, the TUBE COMPOSER T1952 is based on SMD technology (Surface Mounted Device). These subminiature components known from aerospace applications ensure both extreme packing density and greater reliability. The TUBE COMPOSER is manufactured under ISO9000 certified management system.

☞ **Please keep the manual after reading, in order to use it for future reference.**

1.1 The concept

IKA (Interactive Knee Adaptation) compressor

Our proven IKA (Interactive Knee Adaptation) circuitry successfully combines the concept of a “Hard Knee” compressor with the characteristics of a “soft knee” approach. This program-dependent control characteristic forms the prerequisite both for “inaudible” and musical program compression and for creative and highly effective dynamics processing.

With its IKA circuitry the TUBE COMPOSER is capable of delivering outstanding musical results both in studio and live P.A. applications.

Additionally, the TUBE COMPOSER’s side chain filter allows for limiting the influence low-frequency signal portions usually have on the control logic, so that the compression ratio is mainly determined by those frequencies that are essential to the loudness perceived by the listener—the midrange frequencies.

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 in-appropriate. 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 TUBE COMPOSER the ratio of which, is automatically adjusted, depending on the program 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 TUBE COMPOSERs expander/gate section can be used as an independent unit to eradicate noise offering almost limitless possibilities within this application.

IGC (Interactive Gain Control) peak limiter

A further remarkable feature of the BEHRINGER TUBE COMPOSER is the IGC (Interactive Gain Control) limiter, an intelligent combination of a clipper and a program 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 (program 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 (loudspeaker protection) as for digital situations, where any extreme signal peaks would exceed the maximum headroom and therefore would cause severe problems.

The TUBE COMPOSER uses 12AX7 / ECC83 tubes. These triodes are capable of handling a large dynamic range with little microphony. Add to this their relative ruggedness and above average life span and you can see why it’s one of the most popular and reliable pre-amp tubes on the market. These features also ensure you their availability for many years to come.

Balanced inputs and outputs

As standard, the BEHRINGER TUBE 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).

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.

1.2 Before you get started

Your BEHRINGER TUBE 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 during 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.**

The BEHRINGER TUBE COMPOSER fits into two standard 19" rack units of space (3 1/2" / 89.5 mm). Please allow at least an additional 4" depth for the connectors on the back panel.

☞ **Be sure that there is enough space around the unit for cooling and please do not place the TUBE COMPOSER on high temperature devices such as power amplifiers etc. to avoid overheating.**

The mains connection of the TUBE COMPOSER is made by using the supplied mains cable. It meets all of the international safety certification requirements. Please make sure that all units have a proper ground connection.

☞ **Before you connect your TUBE COMPOSER to the mains, please make sure that your local voltage matches the voltage required by the unit!**

As a standard the audio inputs and outputs on the BEHRINGER TUBE COMPOSER are fully balanced. If possible, connect the unit to other devices in a balanced configuration to allow for maximum interference immunity. The automatic servo function detects unbalanced connections and compensates the level difference automatically (6 dB correction).

1.3 Control elements

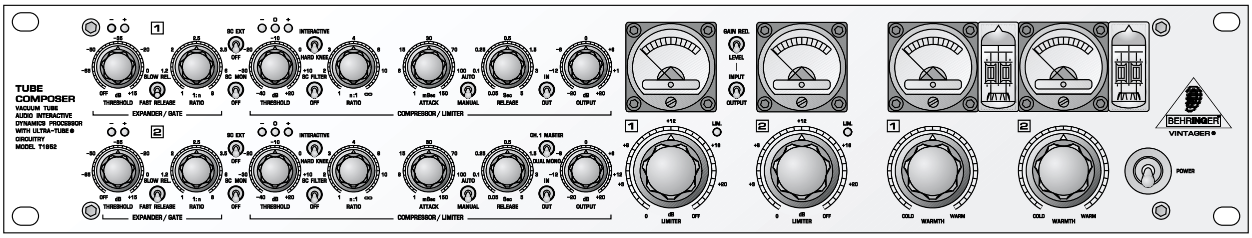

Fig. 1.1: Front of the TUBE COMPOSER

The BEHRINGER TUBE COMPOSER has two identical channels. Each channel is equipped with 7 toggle switches, 9 rotary controls, 6 LEDs and 2 VU meters. The CH 1 MASTER switch is for stereo operation:

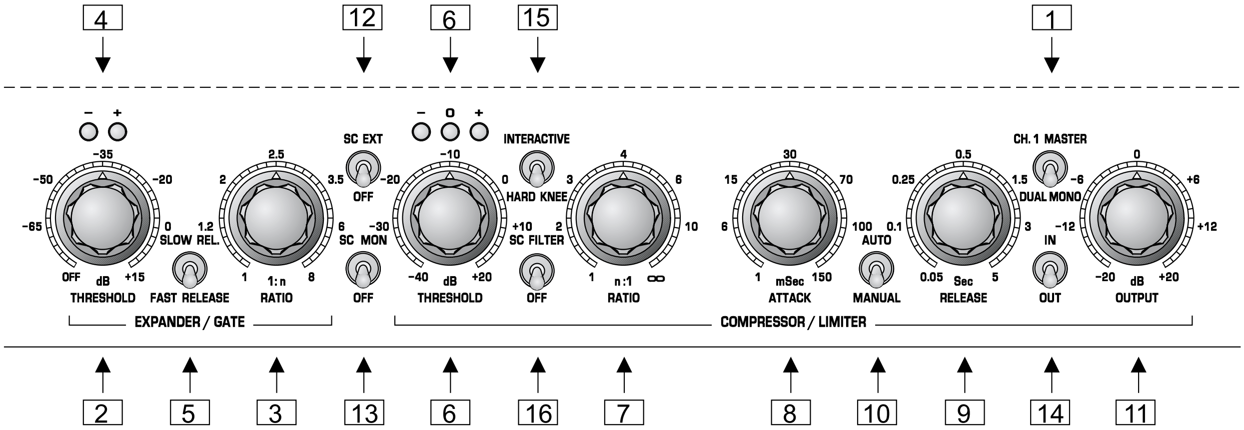

Fig. 1.2: Expander/gate- and compressor section

1 The TUBE COMPOSER converts to stereo mode by engaging the *CH 1 MASTER* switch, where the left channel assumes the control of both audio channels, i.e. the control signal of channel 2 is replaced with that of channel 1. By pressing the CH 1 MASTER switch, you override all the controls and switches of channel 2 with the exception of the IN/OUT, SC MON, SC EXT, SC Filter switches and WARMTH control as well as the PEAK LIMITER control. The controls of channel 1 take over all functions of channel 2.

2 Use the *THRESHOL*D control to determine the threshold point below which expansion occurs. The range of this control is from OFF to +15 dB. Signals above this threshold value pass unaltered, signals below the threshold value are reduced in level according to the adjusted RATIO value. Adjust this control so that music signal can pass but that noise is faded out. The THRESHOLD can be set with the help of the LED above the control. When the signal falls below the threshold "-" lights up, signalling that the expander is attenuating the signal. If the "+" LED lights up, the signal level is above the set THRESHOLD value and the signal is not processed by the expander.

3 The *RATIO* control is used to adjust the expansion ratio when the signal drops below the threshold value. This control determines whether the unit works as an expander (low ratio level) or as a gate (1:8). The expansion ratio can be set from 1:1 to 1:8. Higher values lead to "harsher" suppression while smaller RATIO values lead to a gentler correction:

4 When the signal level is below the set threshold value, the "-" LED lights up. The "+" LED lights up when the signal level is above the set THRESHOLD value.

5 To optimally adapt the expander/gate to the program material, the *RELEASE* switch allows for selecting a SLOW or FAST release time. When you engage this switch, 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.

6 The *THRESHOLD* 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. The THRESHOLD LEDs show the actual state of the input signal relative to the THRESHOLD setting. The IKA-“Soft Knee” range is indicated by the middle yellow LED.

7 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 1:1 to oo:1. A setting of 1:1 leads to no compression, turning the control in the clockwise direction makes the sound increasingly dense. A setting of oo:1 (infinity to one) corresponds to a limiter setting.

8 The *ATTACK* control determines the rate by which the compressor responds to the signal which exceeds the threshold. This control can be adjusted from 1 to 150 milliseconds. Use short ATTACK times for percussion and slower times soft inaudible compression.

9 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. Use short times for fast recovery and maximum output and use longer times to avoid pumping.

☞ **Too long RELEASE time can lead to the fact that the compressor permanently attenuates the signal, without the signal being compressed. Remember when setting effective release times: As short as possible, as long as necessary.**

10 By activating the *AUTO* switch, the ATTACK and RELEASE controls are disabled and the attack and release rates are automatically derived from the program material. This function allows for unobtrusive musical compression of signals or mixes with widely varying dynamics. Only if set to “MANUAL” will the settings of the attack and release controls function.

11 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. By comparing the input and output signal after compression you can set the optimum value. Set the output level so that the processed and unprocessed signals have equally loud signal peaks. The nominal level of the compressed signal will then be louder than the original.

☞ **Please note when using the THRESHOLD control of the peak limiter section, that the OUTPUT control of the compressor section precedes the peak limiter section. If the OUTPUT control is set too high, this can result in continuous peak limiting (see 20 “THRESHOLD control”).**

12 When activated, the *SC EXT* switch severs the connection between the audio input and the side chain path, while at the same time allowing an external signal to be sourced at the SC RETURN jack on the rear panel.

13 Using the *SC MON* switch will enable you to connect the side chain 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 side chain signal that is returned via inserted equalizers or other external processors. The SC MONITOR function will assist you with tuning equalizer parameters for example.

☞ **Please note that when the SC MON switch is engaged, the audio processing facility of the respective channel is disabled.**

14 The *IN/OUT* switch activates the relay and hence the corresponding channel. This switch acts as a so-called “hard-bypass” relay, which means that when the switch is OUT or when the unit is disconnected from the mains, the input jack is directly linked to the output jack. Normally, this switch is used to perform a direct A/B comparison between the unprocessed and the compressed or limited signals.

15 Press the *INTERACTIVE* switch to change from “Hard Knee” to IKA characteristics. IKA provides a very subtle and musical compression of the program material and should therefore be used whenever compression should be more or less inaudible.

16 The *SC FILTER* switch activates a high pass filter in the side chain path and thus limits the influence of low frequencies on the TUBE COMPOSER’s control processes.

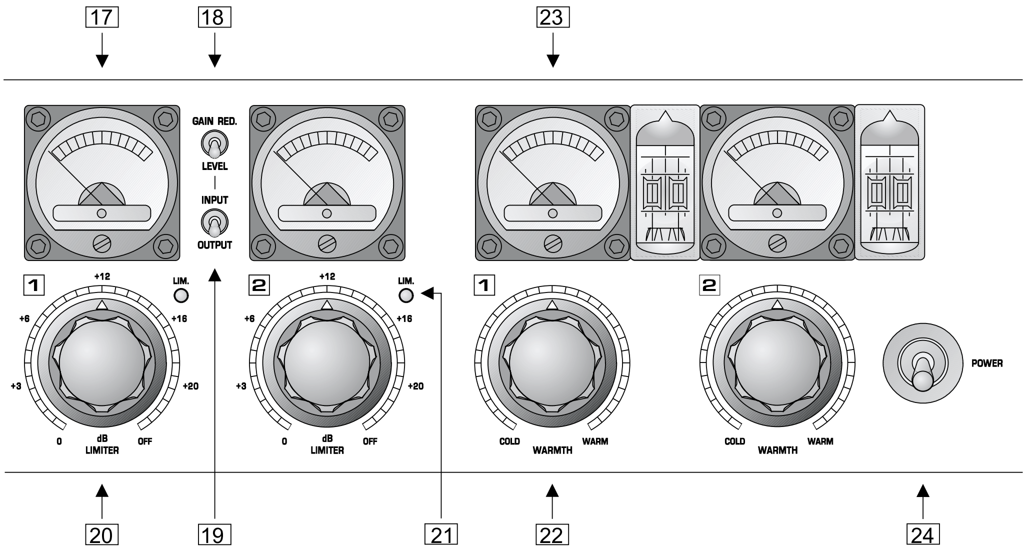

Fig. 1.3: Control elements of the peak limiter and tube sections

17 The *VU METER* can display three levels: the actual gain reduction of the compressor, the input or the output level. Gain reduction is displayed in a range of 1 to 30 dB, the input or output level is displayed in a range of -30 to +10 dB. The nominal level the display is switchable between -10 dBV or +4 dBu with the *OPERATING LEVEL* switch.

18 With the *GAIN RED/LEVEL* switch the above described VU meters can be switched between LEVEL or GAIN REDUCTION mode.

19 If switch 18 is set to *LEVEL*, you can switch further between input and output level.

20 The peak limiter limits the signal to a level adjusted with the *THRESHOLD* control. Owing to its extremely fast response (“Zero” attack), the limiter is capable of limiting signal peaks without audible distortion. Whenever the signal is limited for more than 20 ms, the overall level is reduced for about 1 second to avoid heavy and thus audible signal distortion.

☞ **When you use the peak limiter as a protective device against signal peaks, the THRESHOLD control should be set in combination with the OUTPUT control in the compressor section so that the peak limiter responds rarely or not at all. Thus, only real signal peaks will activate the limiter circuit. However, to produce creative sound effects, the peak limiter can be deliberately set to lower levels.**

21 The *LIM LED* lights up as soon as the limiter function is activated.

22 The *WARMTH* control determines the amount of harmonics the UTC circuit adds to the signal. This is the amount of tube sound (“WARMTH”) that is added.

23 The *WARMTH* meter displays the amount of added harmonics. Thus, the amount of harmonics, or tube sound (“WARMTH”), the UTC circuit adds to the signal is set.

24 Use the *POWER* switch to turn the TUBE COMPOSER on or off. When switched of the TUBE COMPOSER automatically switches to a hard-bypass mode, the signal is then led directly to the outputs.

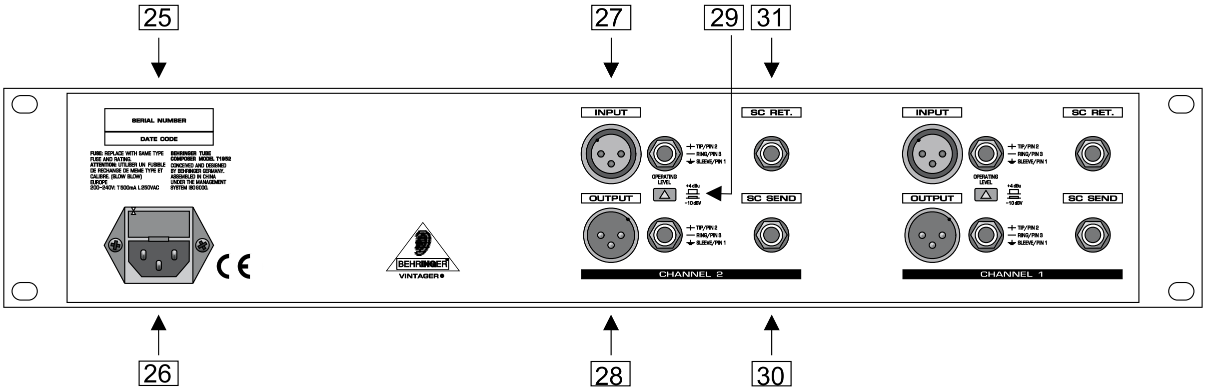

Fig. 1.4: Back panel of the TUBE COMPOSER

25 *SERIAL NUMBER*. Please complete and return the warranty card within 14 days of the date of purchase. Otherwise, you will lose your right to the extended warranty. Alternatively, you can register online at our website under www.behringer.com.

26 *FUSE HOLDER / VOLTAGE SELECTION*. Before connecting the T1952, confirm that the voltage display matches your local mains voltage. When replacing the fuse, you must always use the same type. In many units the fuse holder can be installed in one of two positions, allowing you to switch between 230 V and 115 V. If you wish to operate a unit outside Europe at 115 V, then a stronger fuse must be used (see chapter 6 "SPECIFICATIONS"). The mains connection is made via the IEC receptacle. An appropriate mains cable is included.

27 *AUDIO IN*. These are the audio inputs of your TUBE COMPOSER, available both as balanced 1/4" TRS and XLR connectors. Both the XLR and the jack connector accept unbalanced as well as balanced signals. See chapter 5 "INSTALLATION" when wiring unbalanced.

28 *AUDIO OUT*. These are the audio outputs of your TUBE COMPOSER, which are also designed as 1/4" TRS and XLR sockets. The automatic servo function recognizes balanced or unbalanced connection and automatically compensates for the difference in level (correction 6 dB). These outputs can be transformer-balanced by retrofitting the optional output transformer OT-1.

29 With the *OPERATING LEVEL* switch you can adapt the TUBE COMPOSER to various operating levels, i.e. you can select both the -10 dBV home recording level and the professional studio level of +4 dBu. The level meters are referenced automatically to the selected level, i.e. an optimum operating range of the meters will always be ensured.

30 *SC SEND*. This is the unbalanced side chain output which allows for routing the audio signal to external processing devices.

31 *SC RETURN*. This is the unbalanced side chain input used to return any external or processed control signal. Please note that the side chain signal is only the control signal. The unbalanced connection therefore does not represent any danger for your audio signal.

2. OPERATION

In this section, several typical applications of the BEHRINGER TUBE 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 TUBE COMPOSERs capabilities in the future.

Main applications and initial settings

The main applications of the BEHRINGER TUBE COMPOSER can be divided into three categories: The expander/gate section is used to eliminate interference and to suppress background noise and leakage on individual tracks in multitrack recording. The compressor section is used to compress the program material and to create special effects and unusual sounds, which are used for recording and musical performance. The subsequent peak limiter section is designed to protect loudspeakers, tape recorders, transmitters etc. from being overloaded.

2.1 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 times, 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 dynamics are reduced dependent on the ratio setting and 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 program 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 peak limiter.

Clipping

In contrast to the two previously mentioned limiters, the clipping mode features infinitely fast control times, 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”).

2.2 Expander/gate section

The main task of the expander/gate is to “inaudibly” eliminate undesirable background noise from the usable signal. 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 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.

The BEHRINGER TUBE COMPOSER is equipped with a newly developed IRC (Interactive Ratio Control) expander, the ratio of which is automatically adjusted dependent on the program 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 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.

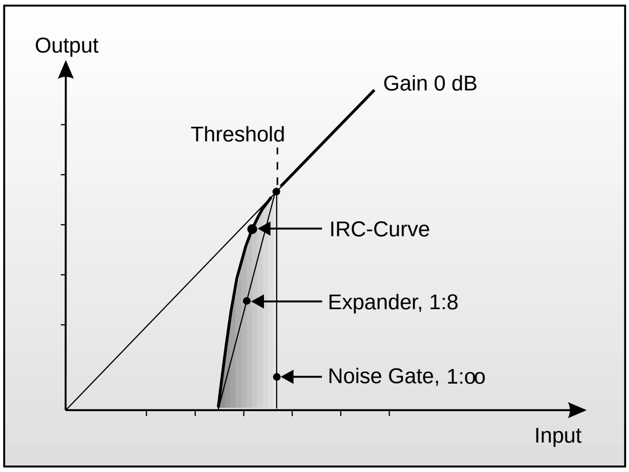

Fig. 2.1: IRC curve characteristic of the expander

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. Expansion therefore occurs extremely “soft” with low ratio settings, while the known negative effects of expansion are inaudible. The attack time of the IRC expander is set automatically and program-dependent, i.e. extremely short for quickly changing signals and slower for a more balanced program material. Since the expander/gate adapts itself automatically to the program material, you will note that the new IRC (Interactive Ratio Control) circuit produces considerably better results than conventional expanders.

Basic settings of the expander/gate section

Adjust the control elements of the TUBE COMPOSER in accordance with the following table. This will switch off all functions except the expander section.

Control element	Setting
THRESHOLD control of the expander section	OFF
RELEASE switch	SLOW REL.
RATIO control of the expander section	1:2
IN/OUT switch	IN
SC EXT switch	OFF
SC MON switch	OFF
THRESHOLD control of the compressor section	+20 dB
AUTO switch	AUTO
OUTPUT control	0 dB
LIMITER control	OFF
WARMTH control	COLD

Tab. 2.1: Basic settings of the expander/gate section

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 program 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 preferable for acoustic separation of most percussive sounds, whilst cymbals and tom toms, normally benefit from the SLOW mode.

The RATIO control determines the ratio between input and output level, for all signals below the selected threshold point. Use this control to determine whether the section works as an expander or as a gate (ratio 1:8).

If the controls are set correctly, the drum sounds will be “dry”, “sharp” and clearly defined. If you do not have enough mics (or TUBE COMPOSER channels!) to record each instrument separately, try to create subgroups: 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”.

2.3 Compressor section

The task of a compressor is to reduce the dynamic range of program 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 program 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 program. 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 program 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 program material less and are often used to compress the sound of a bass guitar, a snare drum or vocals. Sensitive and moderate settings are generally used in mixing and for levelling program material in broadcasting.

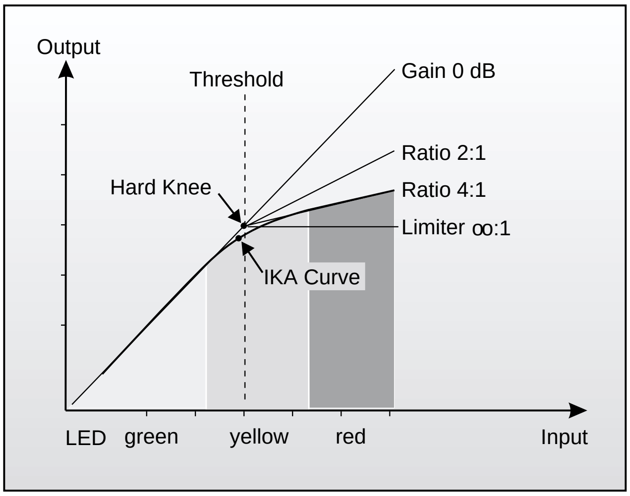

Fig. 2.2: IKA characteristic of the compressor section

The new IKA (Interactive Knee Adaptation) 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.

☞ **With the threshold control completely turned to the right, the threshold value is +20 dB. Since such a value will not be reached in practice, you can use it to disable the compressor section and work exclusively with the expander/gate and limiter circuits.**

Initial settings for the compressor section

In order to acquaint yourself with the possibilities of the compressor section, it is advisable to start with the following settings. The settings for the expander section can remain unaltered.

Control element	Setting
IN/OUT switch	IN
SC EXT switch	OFF
SC MON switch	OFF
INTERACTIVE/HARD KNEE switch	INTERACTIVE
SC FILTER switch	OFF
THRESHOLD control of the compressor section	+20 dB
RATIO control	3:1
AUTO switch	AUTO
OUTPUT control	0 dB
LIMITER control	OFF
GAIN RED./LEVEL switch	GAIN RED.
WARMTH control	COLD

Tab. 2.2: Initial settings for the compressor section

Rotate the THRESHOLD control counterclockwise 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 I/O METER 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 program dependent dynamic 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 in live situations, where a signal needs to be managed efficiently by the engineer, without the convenience of continual A/B comparison.

The SC filter (side chain filter)

Compressors are often blamed for making the sound duller or muffled. This is because most of the energy in a signal is usually in the bass region causing the compressor to react mostly to the bass. When the compressor identifies a strong bass signal the gain must be reduced according to the settings of the compressor. When high frequencies are present they will also be reduced. This is why a highly compressed drum kit sounds muffled, the cymbals are drowned out by the bass and toms. This also happens to ambient noise like reverb.

The solution to this is to reduce the compression ratio or to increase the attack time. Increasing the attack time will cause the compressor to react slower to transients so that high frequency signals will be able to pass through before the gain is reduced.

The TUBE COMPOSER T1952 offers you another more elegant solution. When engaging the SC FILTER, a high pass filter can be switched in the SC signal. This prevents bass signals from dominating the compression process. A bass signal is compressed less than a mid or high frequency signal of the same amplitude. The benefit of this is that the frequency response and attack and release time can remain optimal.

Bass signals are often already processed in pop music. This means that the SC filter is suited to process the entire mix. With the help of the TUBE COMPOSER you can increase the overall loudness without the described side effects usually associated with compression.

The tube stage used in the TUBE COMPOSER is ideally suited to give the compressed signal the finishing touch.

2.4 Peak limiter section

As a section of its own and independent of the remaining control functions, the peak limiter enables you to limit the maximum peak level on the TUBE COMPOSER’s output. It has been designed for use in combination with the compressor section. Independently of all compressor functions, you can protect subsequent devices against signal peaks, short-time overload and excess modulation (radio stations, etc.).

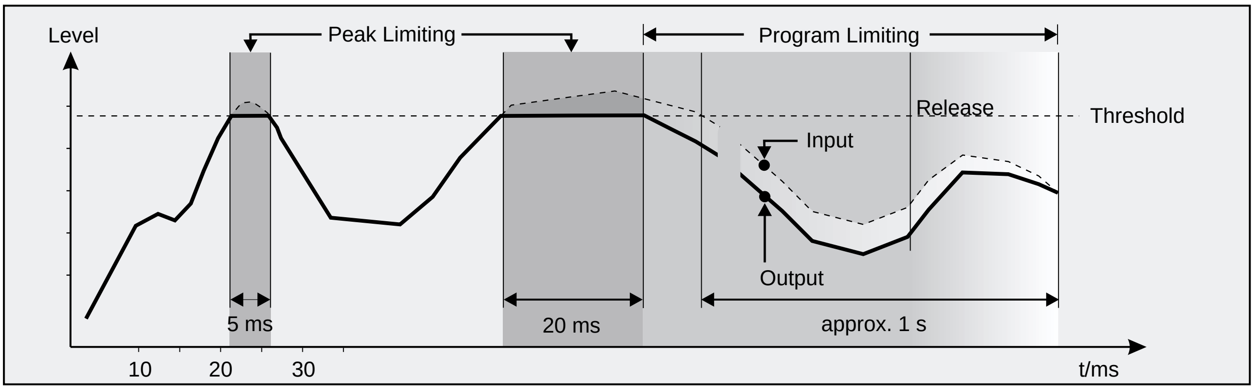

Fig. 2.3: IGC characteristic of the limiter section

The diagram illustrates the functioning of the IGC limiter. The solid graph represents the output signal, while the dashed graph above shows the input signal response. The areas between the graphs represent the amount of gain reduction (bright areas are “clipping areas”, i.e. signal peaks are radically cut off, dark areas show the effect of the program limiter). The limiter is activated when the adjusted threshold is exceeded for more than 20 ms, so as to limit audible clipping to a very short moment. About 1 s after the signals has dropped below the threshold again, the reduction is set to 0 dB, so that input and output signals are identical again (unity gain).

Basic settings of the peak limiter section

This is a good starting point for the peak limiter. The settings for the compressor can remain unaltered.

Control element	Setting
IN/OUT switch	IN
SC MON switch	OFF
LIMITER control	OFF

Tab. 2.3: Basic settings of the peak limiter section

With the LIMITER control you can guard subsequent electronics against overload. The LIM LED shows when the limiter is active. When this LED lights up often, reduce the OUTPUT level in the compressor section.

If this results in an unwanted reduction of the sound pressure level it is also possible to increase the compression. You can do this by either lowering the threshold or increasing the compression ratio. After that, you can set the desired loudness using the OUTPUT control.

2.5 The side chain function

The BEHRINGER TUBE COMPOSER offers an exceptionally usable external facility by using the side chain function. By activating the SC EXT switch, the TUBE COMPOSERs control path is disconnected from the audio input and therefore interrupted. The audio input is routed to the SC SEND output and the SC RETURN input now receives the new control signal which is derived from an inserted effects processor.

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

It is very common to make the response threshold of a compressor frequency-dependent, where a graphic or parametric equalizer is connected to the side chain path. To retain the threshold setting of the TUBE COMPOSER, unwanted frequencies should be reduced by an equalizer 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.

2.6 The vacuum tube of the TUBE COMPOSER

Our engineering team has made it possible to enhance the traditional tube circuitry (particularly for our TUBE COMPOSER) and adapt it to meet the high sound quality and dynamics requirements of modern, pro-level audio technology. The fact that we are still fascinated by “antique” tube radios and amps as well as the fine and warm tonal character that we usually associate with them, are the reasons why vacuum tubes have kept their ground even in state-of-the-art circuit topologies used especially in professional audio technology or high-end devices. We are particularly proud that we have found a highly effective symbiosis between solid-state and tube technologies making them affordable to almost anybody in audio technology.

3. APPLICATIONS

3.1 The TUBE COMPOSER in the studio

3.1.1 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 result 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 TUBE 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 SC MON switch) and the THRESHOLD level set so that each snare hit sounds acoustically clean and separated, 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.

3.1.2 Using the TUBE COMPOSER for recording and cassette duplication

In the recording and duplication field the goal should always be 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 take care 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 signals 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 TUBE COMPOSER by starting with the initial settings, and use its dynamic control functions in order to be able to drive an analog, as well as a digital recording, up to the limit of its maximum dynamic range while remaining noise- and distortion-free.

3.1.3 The TUBE COMPOSER in digital recording and sampling

In an analog 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 analog, side effects in the digital field always become extremely audible: with decreasing level, a tape previously recorded with insufficient level loses resolution: the recording sounds “hard” and loses “atmosphere”. With excessive level, the recording sounds harsh and heavily distorted. In order to avoid these effects, the peak limiter section of the TUBE 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.

3.1.4 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 in small studios and home recording installations. However, multiple effects units considerably increase the overall noise level, so that the pleasure in acquiring a new sound effect is short lived.

It will prove useful to use the BEHRINGER TUBE 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.

3.2 The TUBE COMPOSER in live applications

3.2.1 Reducing leakage in stage mics

The TUBE COMPOSER has many uses 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 “inaudibly” eliminate background noise, that occurs in pauses between singing. When used in live situations, leakage of miked instrumentation is substantially reduced, as well as other acoustic contaminants in various recording situations.

3.2.2 Reducing feedback in stage mics

When a singer is using a vocal mic, the 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 TUBE 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.

3.2.3 The TUBE COMPOSER as a protective device

Sound system distortion is usually a result of amplifiers and loudspeakers being driven beyond their limitations, whereby signals are hard limited by so-called “clipping” of the amplifiers. The signal peaks are thereby “clipped” because the maximum output voltage is reached. This can lead to unpleasant distortions, which is dangerous for loudspeakers.

Apart from the danger of long term overload a loudspeaker can also be damaged by an occasional high level overload, e.g. the sound of a microphone falling onto a hard floor. In order to protect a system or the loudspeakers, the application of the BEHRINGER TUBE COMPOSER is recommended. Conventional limiters must restrict the maximum output level far below the clip-point of the amplifier, in order to limit the height and duration of overloading transients. Thus, the power reserve of the system cannot be fully exploited.

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 TUBE COMPOSER between your mixing console output and the power amplifier input. It is used as the last link in the chain preceding the power amp. Thus, you can effectively avoid damage to the midrange/tweeter range caused by clipping of the high-energy bass signals. This statement, as paradoxical as it may seem at first, can be explained with the fact that especially low-frequency signals with high amplitudes can overload the power supply in the amplifier(s). The resulting clipping (cutting off of signal peaks) produces distortion (upper harmonics), which is abruptly added to the midrange/tweeter signals. For this reason, “weak” power amps, in particular, must be protected by a limiter in their input dynamics.

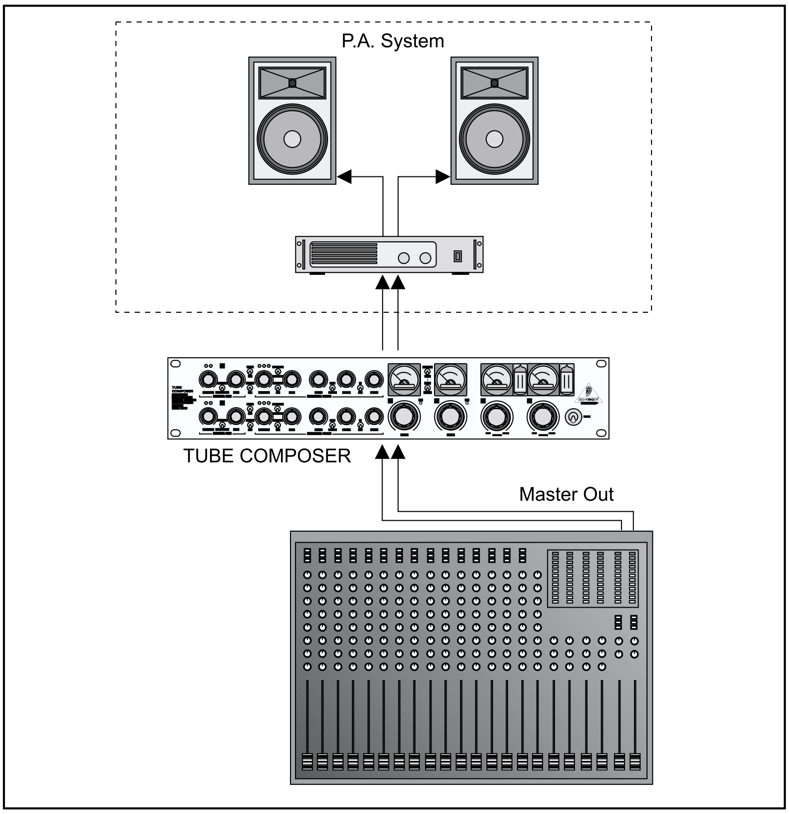

Fig. 3.1: The TUBE COMPOSER as a protective device

Protection of a system with an active crossover

When used with a system with an electronic crossover, connect the TUBE COMPOSER before the crossover. In this application, the BEHRINGER TUBE COMPOSER will process the entire audio frequency spectrum.

☞ **If you want to protect separate units in a multi-way active system you can use a (multi)way compressor/limiter between the electronic crossover and the amplifier(s) like the BEHRINGER MULTICOM PRO MDX4400.**

3.2.4 Improving the sound of a processor system

An electronic crossover divides the total frequency spectrum in separate bands, thus correcting the single units’ frequency responses and time alignments. A processor system is a PA system which contains a special active crossover with additionally dynamic functions that monitor the system performance and optimize the output dependent on the program material. 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 loudspeaker.

In some units a “loudness contour” is applied where, for instance, the bass is boosted at low levels to extend the range of the system at the low end. At higher levels this frequency correction is abandoned in exchange for a higher maximum sound pressure level. In many cases, this function leads to a disturbance rather than to an improvement of the sound quality.

If the TUBE COMPOSER is placed before the processor, 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 dynamic functions of the crossover.

3.3 The TUBE 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 utilized as a creative tool, laying the groundwork for many of the sounds which are now considered indispensable in contemporary music. The compressor is used in this role because you can hear it working, and control of the dynamic range is of secondary importance.

With its extensive range of functions, the BEHRINGER TUBE COMPOSER is well suited to this application. Sound effects of this kind can be achieved using "extreme" settings. To achieve this, 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 of their function!

3.4 The TUBE 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 for example, this is apparent especially with records and cassettes which are processed with high average volumes. Quite often in these cases, dynamics suffer drastically, because the program material has been compressed and limited too heavily. Using the compressor and the peak limiter section of the TUBE COMPOSER allows you to drastically increase the overall volume, without audibly affecting the dynamics.

Proceed as follows:

1. Limit the dynamics of the program 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. Therefore, in addition, 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 gain and now reduce the LEVEL control of the peak limiter until the LIM 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 gain. The result is a clearly louder recording without any loss of sound.

3.5 The TUBE 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 programs whose reception is louder than the average are preferred by the listener. By achieving a bigger audience, the broadcast station gains more money from the increasing number of promotion companies placing adverts.

What is volume?

Volume is defined as the relationship between the average level of program 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 program material will be perceived by the listener.

If you want to run your broadcast station at maximum average volume, proceed as mentioned in chapter 6.1.2. 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 transmission 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 TUBE COMPOSER result in higher average volumes, free of distortion.

3.6 Side chain

3.6.1 The TUBE COMPOSER as "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 selective 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 program 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 equalization 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 equalizer not into the audio path but into the side chain path of the BEHRINGER TUBE COMPOSER. The equalizer is inserted between the SC SEND output and the SC RETURN input of the BEHRINGER TUBE COMPOSER. While the SC EXT switch is active, the equalizer is inserted into the side chain loop and controls the unit. With the help of the side chain monitor function, the centre frequencies of the equalizer are then adjusted exactly 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 equalizer. The level of the sibilant sounds can therefore be effectively limited.

Control element	Setting
SC EXT switch	IN
SC MON switch	OUT
INTERACTIVE switch	OFF
SC FILTER switch	OUT
THRESHOLD control	+20 dB
RATIO control	1:4
AUTO switch	OUT
ATTACK control	1 ms
RELEASE control	150 ms
OUTPUT control	0 dB

Tab. 3.1: Basic setting for the "De-Esser" function

1. Turn the THRESHOLD control counterclockwise until the gain reduction meter reads a clear reduction in level.
2. Now switch on the SC MON and tune the equalizer so that the S sounds are stronger than the rest of the signal (usually between 6 and 10 kHz).
3. Turn of the SC MON function and set the THRESHOLD so that the compressor reacts to the S sounds.

No compensation with the OUTPUT control should be necessary. Set the attack and release times to suit your specific needs. The AUTO function is best not used in this setup.

3.6.2 Frequency selective filtering of unwanted signals

Based on the setup 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 SC MON switch, adjust the frequencies of the equalizer 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 3.6.1. This will result in compression of the selected frequencies and thus a decrease in the gain of the program material.

3.6.3 Suppressing instruments during recording

Another function of the BEHRINGER TUBE COMPOSER allows helpful correction of previously recorded material. If, for example, an excessively loud bass drum needs to be suppressed, reduce all the equalizers 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 only to loud pedal or stick actions. 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.

3.6.4 Emphasizing musical instruments during recording

On the other hand, you can use the BEHRINGER TUBE COMPOSER to bring out an instrument solo or a lead vocal in a cluttered mix. Using the SC MON switch, match the frequencies of the equalizer to the frequencies of the instruments to be emphasized 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 volume of the overall program material. Only the selected frequencies coming from the equalizer remain uncompressed and are therefore perceived as being louder. This inverse type of compression also helps to emphasize instruments during low level passages, so that they become more pronounced.

3.6.5 Anticipated compression

If you feed the audio signal directly into the SC RETURN input and send the audio signal through a delay before the audio input, the BEHRINGER TUBE 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.

3.6.6 “Voice-Over” compression (“ducking”)

The BEHRINGER TUBE COMPOSER can be used to automatically reduce music to a background level, when a presenter is speaking through a microphone. For this purpose, the BEHRINGER TUBE COMPOSER is used as an automatic fader and is controlled by the presenter’s microphone, which is connected to the SC RETURN input via a pre-amplifier. 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.

3.6.7 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, while the compressor and peak limiter sections are switched off. The bass guitar track is connected to the audio chain of the BEHRINGER TUBE COMPOSER, while the bass drum is connected to the SC RETURN input. By activating the SC 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.

4. TECHNICAL BACKGROUND

The steady development of modern sound reinforcement systems has made it possible to produce almost any level of loudness. Yet, the increase in loudness goes in line with a need for optimized audio quality. Today, audiences expect to hear a powerful and transparent sound. To fully understand how the BEHRINGER TUBE COMPOSER works you will need to know the meaning of a decibel and how audio dynamics function.

With the TUBE COMPOSER, as with any other type dynamics processor the amount of boost/attenuation applied is expressed in decibels (dB). What’s a decibel? The abbreviation dB is not a unit (although often used as one), but describes a logarithmic proportion. The entire dynamic range of human hearing (from the threshold of audibility to a jet-airplane, see fig. 4.1) starts with about 0.00002 Pa (threshold of audibility) and goes up to 130 Pa (threshold of pain). This also means that 0 dB is not silence, minus infinity dB will mean absolute silence.

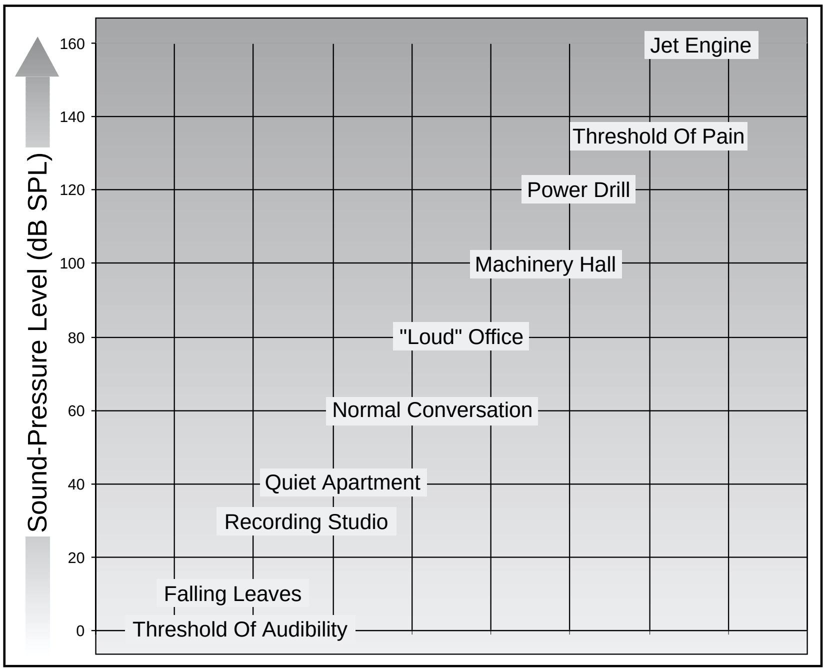

Fig. 4.1: Dynamic range of human hearing

The range of sound pressure levels or the dynamic range of human hearing encompasses a factor of 10,000,000. This enormous range of values is difficult to handle and additionally does not represent the subjective perception of sound, since human hearing tends to use a logarithmic curve. When an increase in loudness by the factor two is perceived as one step, four times the loudness level equals two steps. So, the decibel is a unit of measurement that describes a level in relation to a reference quantity. To make clear which reference quantity is meant, the abbreviation SPL (sound pressure level) is sometimes used together with dB. Starting with a value of 0 dB SPL (= $2*10^{-5}$ Pa) for the threshold of audibility, any dB values can be calculated by means of the following formula:

$$L = 20 \cdot \log \frac{p_2}{p_1}$$

L = e.g. the absolute sound pressure level in dB SPL
p_1 = e.g. a reference sound pressure of 0.00002 Pa
p_2 = the sound pressure (in Pa) produced by the sound source to be calculated
log = decimal logarithm.

As can be seen, human hearing has a very wide dynamic range of about 130 dB, which surpasses the range of a DAT or CD player with an approximate range of 96 dB. From a physical point of view, a 3 dB boost corresponds to an increase in power by the factor 2. However, the human ear perceives a signal to be twice as loud as before only if it is boosted by about 10 dB.

4.1 Audio dynamics

As demonstrated, it is possible to manufacture analog audio equipment with a dynamic range of up to 130 dB. In contrast to analog techniques, the dynamic range of digital equipment is approximately 25 dB less. With conventional record and tape recorder technology, as well as broadcasting, this value is further reduced. Generally, dynamic restrictions are due to noisy storage in transmission media and also the maximum head-room of these systems.

4.1.1 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 non-directional magnetic particles passing the replay head can also cause uncontrolled currents and voltages. The resulting sound of the various frequencies is 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. Due to the laws of physics, improving the design of the magnetic carrier is impossible using conventional means.

4.1.2 What are audio dynamics?

The human ear 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.

The usable dynamic range of electroacoustic 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 program material would be increased considerably.

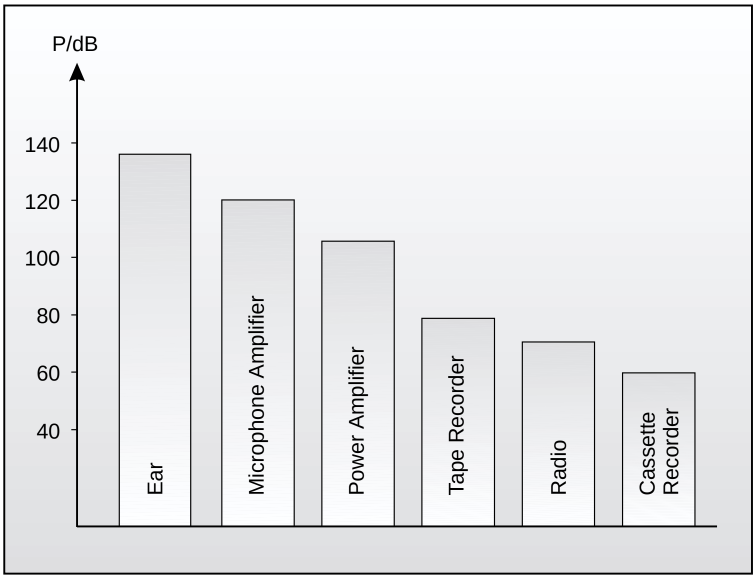

Fig. 4.2: The dynamic range capabilities of various devices

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 program 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.

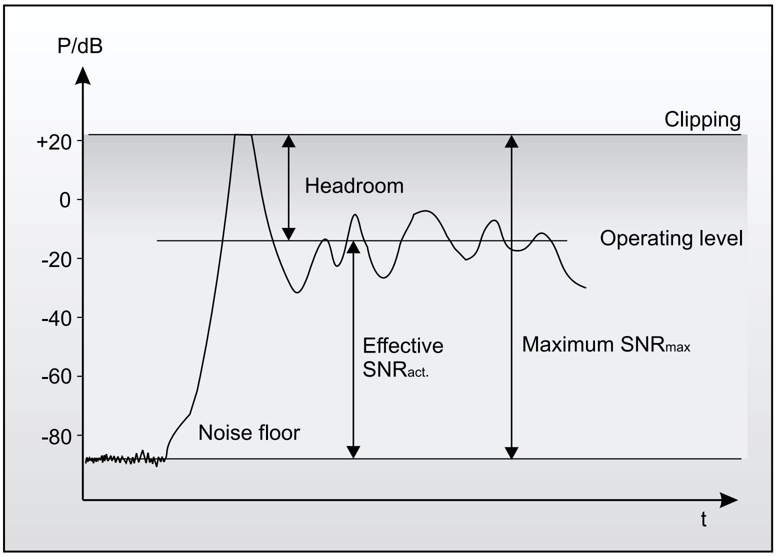

Fig. 4.3: The interactive relationship between the operating level and the headroom

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.

4.1.3 Compressors/limiters

With broadcasting and recording, signal peaks can easily lead to distortion due to the high dynamic range of microphones and other musical equipment. Compressors and limiters reduce the dynamics by means of an automatic gain control. This reduces the amplitude of loud passages and, therefore, restricts the dynamics to a desired range. This application is particularly useful with microphones, to compensate for level changes.

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. Both continuously monitor the signal and intervene as soon as the level exceeds a user-adjustable threshold. Any signal exceeding this threshold will be immediately reduced in level.

Limiters reduce the output level to the adjusted threshold whenever the input signal exceeds this point. With compression, in contrast to the action of a limiter, the signal is reduced in gain relative to the amount the signal exceeds the threshold. The output of a compressor will still rise if the input level is increased, while the maximum output of a limiter will always be equal to the threshold level.

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 so that it only intervenes to protect subsequent equipment from signal overload.

The speed, or rather time settings used can differ greatly depending on use. Although both limiter and compressor use very short attack times, the release time of a compressor is in the 100 ms region whereas a limiter uses release times of seconds. To be exact: The release time is a time constant of an exponential function. It is the time it takes the gain reduction to decrease by 63.2 % (= 8.7 dB).

Because fast level changes are more noticeable than slow changes, long release times are used where unobtrusive signal processing is required. In some cases however, the principal goal is to protect devices as loudspeakers and power amplifiers. In those cases a short release time is more appropriate to ensure that the limiter only intervenes when it is needed and the level returns to normal as soon as possible.

Long release times are better suited when the limiter should remain “inaudible” for instance with broadcasting or club applications or when a signal is transferred to (analog) tape. Please note that when using slow release times you should switch to the level meter menu where the functioning of the limiter can be monitored.

4.1.4 Expanders/noise gates

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

4.2 The tubes used in the TUBE COMPOSER

A closer look at developments and trends in audio technology shows that tubes are currently enjoying a renaissance, in a time when even amateur musicians are free to use digital effects processors and recording media, and ever more affordable digital mixing consoles are becoming a natural part of the equipment of many semiprofessional studios. The manufacturers try with ever new algorithms to get the most out of DSP’s (Digital Signal Processors), the heart of any digital system.

Still, many audio engineers, particularly old hands often prefer using both old and new tube-equipped devices. As they want to use their warm sound character for their productions, they are ready to accept that these “goodies” produce a higher noise floor than modern, transistor-based devices. As a consequence, you can find a variety of tube-based microphones, equalizers, pre-amps and compressors in today’s recording and mastering environments. The combination of semiconductor and tube technologies gives you the additional possibility of using the best of both worlds, while being able to make up for their specific drawbacks.

4.3 Tube history

Due to many patent litigations, it is difficult to determine exactly when the tube was “born”. First developments in tube technology were reported between 1904 and 1906. It was a research task of that time to find a suitable method for receiving and rectifying high frequencies. On April 12, 1905, a certain Mr. Fleming was granted a patent for his “hot-cathode valve” which was based on Edison’s incandescent lamp. This valve was used as a rectifier for high-frequency signals. Robert van Lieben was the first to discover (probably by chance) that the anode current can be controlled by means of a perforated metal plate (grid), one of the milestones in the development of amplification tubes. In 1912, Robert van Lieben finally developed the first tube for the amplification of low-frequency signals. Initially, the biggest problem was to produce sufficient volume levels, which is why resonance step-ups (though impairing the frequency response) were used to maximize the attainable volume. Later, the objective was to optimize the electroacoustic transducers of amplifiers in such a way that a broad frequency band could be transmitted with the least distortion possible. However, a tube-specific problem is its non-linear amplification curve, i.e. it modifies the sound character of the source material. Despite all efforts to ensure a largely linear frequency response, it had to be accepted that tube devices produce a “bad” sound. Additionally, the noise floor generated by the tubes limited the usable dynamics of connected storage media (magnetic tape machines). Thus, a one-to-one reproduction of the audio signal’s dynamics (expressed as the difference between the highest and lowest loudness levels of the program material) proved impossible. To top it all, tube devices required the use of high-quality and often costly transducers and sophisticated voltage supplies.

With the introduction of semiconductor technologies in the field of audio amplification it soon became clear that the tube would have to give way to the transistor, as this device featured an enormously enhanced signal-to-noise ratio, less complex power supply and improved frequency response. Plus, semiconductor-based circuits can be realized much more easily—for less money. Two decades later, the introduction of binary signal processing meant the beginning of a new era of recording media that provided plenty of dynamic response and allowed for loss-free copying of audio signals. As digital media were enhanced, however, many people began to miss the warmth, power and liveliness they knew from analog recordings. This is why purists still today consider digital recordings as “sterile” in sound.

4.4 Design and functional principle of tubes

Tubes can be roughly classified according to the number of electrodes they use. There are tubes with two, three or five electrodes usually referred to as diodes, triodes or pentodes.

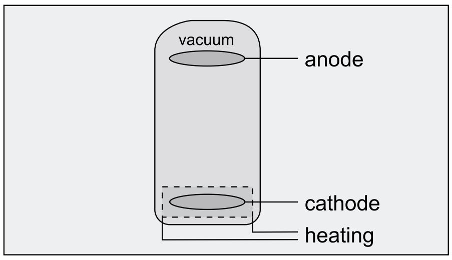

Fig. 4.4: Diode

The diode contains two electrodes in a vacuum glass bulb that have electrical connection to the outside. The vacuum allows for a free movement of electrons. When one of the electrodes is heated up (= thus becoming a cathode), it begins to emit electrons. When a positive dc voltage is applied to the other electrode (= anode), the negative electrons start to wander from the cathode to the anode. With reverse polarity between cathode and anode, a current flow is not possible because the unheated anode emits more or less no electrons. This design was used, for example, as a rectifier in the power supplies of amplifiers. The magnitude and velocity of the flow of electrons depend on the cathode’s temperature, the material it consists of, and the magnitude of the anode voltage. When the electrons hit the anode they produce heat that is dissipated by using large anode plates.

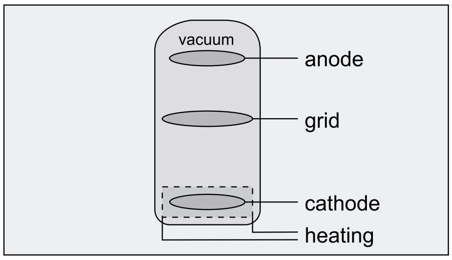

Fig. 4.5: Triode

The triode has an additional metal grid between anode and cathode. By applying a negative voltage, this grid can be used to control the internal resistance of the tube, and hence the anode current. When the grid bias voltage (voltage between cathode and grid) becomes negative, the current flowing to the anode is reduced because the negatively charged grid repels the arriving electrons. As a consequence, there are less electrons to reach the anode. When the bias voltage is raised towards zero, the flow of electrons accelerates. When it finally becomes zero or even positive, the grid current begins to flow which considerably reduces the current flowing to the anode and can possibly destroy the tube. Triodes are most commonly used in pre-amps, often in pairs arranged in one tube (twin triode).

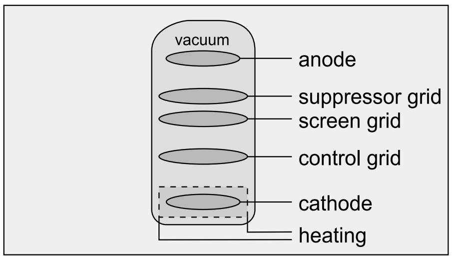

Fig. 4.6: Pentode

In a triode the capacitance between grid and anode is a problem with regard to high frequencies and large amplification factors. For this reason, the pentode has a positively charged screen grid between the control grid and the anode. However, the positive charge of the screen grid attracts electrons emitted from the anode plate when it is hit by arriving electrons. To prevent this electron emission, a decelerating or suppressor grid is placed between anode and screen grid. As it is negatively charged it blocks the electrons, so that they cannot reach the screen grid. Pentodes are most commonly used in power stages.

4.5 Properties of tubes

In general, the saturation (overdriving) of both transistor and tube-based circuits results in various types of distortion. These phenomena are quite complex in the real world, but for the sake of a straightforward mathematical description we are going to classify them as linear and non-linear distortion. Linear distortion is produced by frequency-dependent amplification or attenuation processes such as occurs in all kinds of filters and equalizers. Linear-distortion signals have the same frequency portions both on the input and output sides, but with different phase positions and amplitudes. Non-linear distortions have additional harmonics and distortion components that were not contained in the original input signal.

For example, when the plainest of all oscillations, a sine wave with a fixed frequency f, is overdriven, new oscillations with frequencies of 2*f, 3*f, etc. (integral multiples of the original frequency) are produced. These new frequencies are referred to as upper harmonics grouped as odd and even harmonics. Unlike the transistor, saturated tubes mostly produce even harmonics which are perceived by the human ear as more pleasant in sound than odd harmonics. Another important aspect lies in the fact that tubes produce distortion more gradually than transistors, which is why we speak of the “saturation” of a tube stage. When you overdrive a transistor you get a sudden square deformation of the sine signal applied at the input, which produces an extreme harmonic spectrum at the output.

Non-linear distortions are measured with a distortion factor that consists of the total harmonic distortion [k] and partial harmonic distortions [kn]. The latter are defined as the ratio between the voltage of a single harmonic and the voltage of the distorted overall signal. Thus, the content of even harmonics is expressed as k2, k4, ... and that of odd harmonics as k1, k3, ...

$$k_n = \frac{U_n}{U_{ges}}$$

Formula for calculating partial harmonic distortion

The total harmonic distortion is the root of all squared distortion factors of the second and third degrees. Since the higher harmonics have only little impact on the measured results, they can be neglected.

$$k = \sqrt{k_2^2 + k_3^2}$$

Formula for calculating total harmonic distortion

In tube circuits the distortion factor k2 is used to describe an effect which the human ear classifies as “pleasant”. Also the frequency bands in which distortion occurs play an important role because the human ear differentiates very clearly in the frequency range of human speech.

4.6 The best of both worlds

Despite many efforts neither manufacturers nor developers have succeeded so far in simulating these positive properties of the tube by means of other devices. Additionally, the natural capabilities of the tube to act as a soft limiter can only be mimicked with highly sophisticated circuitry. Today's studio technology requirements are therefore met by a combination of both high-grade semiconductor and tube technologies. In this context, tubes no longer serve their original purpose as amplifiers, but are used for the detailed shaping of sound.

4.7 UTC circuit

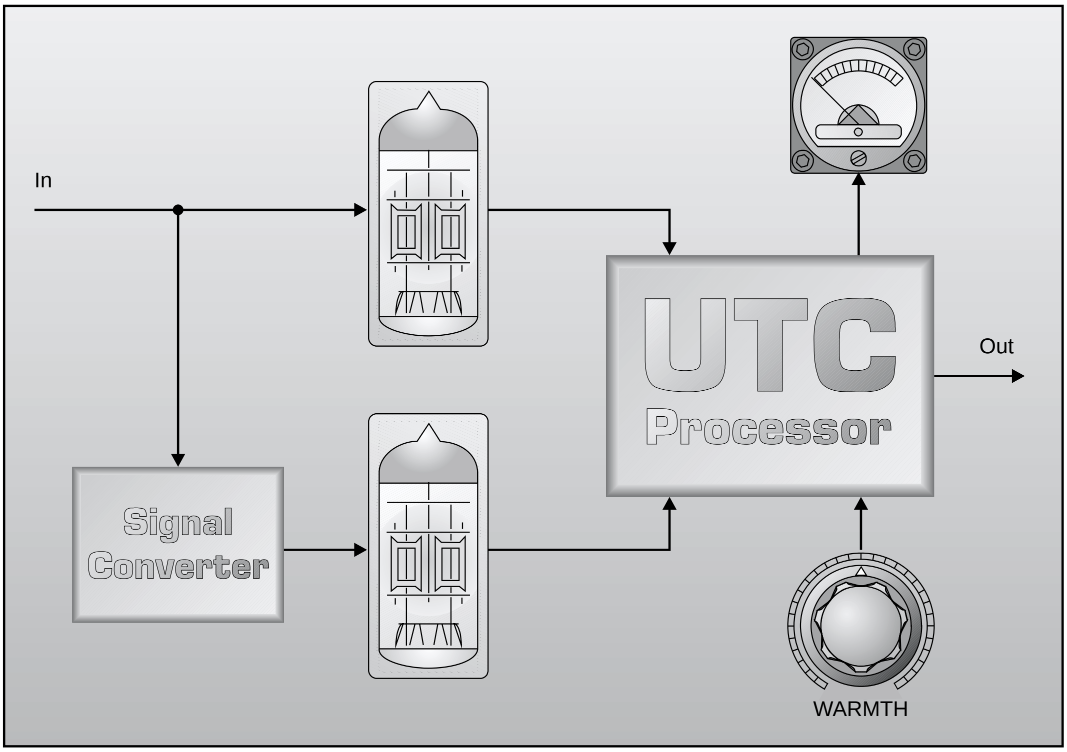

Fig. 4.7: UTC circuit

The TUBE COMPOSER splits up the audio signal applied at the input, and processes it differently for both signal paths. Each of the two tube halves amplifies the original signal and the signal modified in its phase spectrum (twin triode, see chapter 1.1.2). Additional harmonics are produced by slightly overdriving the tube stage. When the two signals are processed by the UTC circuit, the interference noise found in conventional tube circuits can be largely eliminated, and the actual tube effect be added gradually. The more you turn the WARMTH control to the right, the more tube sound will be added to the original signal.

4.8 Studio applications

In a recording studio tubes do not perform the same task as they do in an overdriven guitar amp, where the considerably higher saturation of the tube(s) leads to a full and often deliberate modification of the input signal (in many cases combined with a heavy increase in noise floor levels). In the studio more subtle effects are needed. Here, tube circuits add life to the signal's tonal character and increase its power to make itself heard. Often, tubes also increase the signal's perceived loudness (in relation to the unprocessed signal), i.e. the perceived loudness goes up although the volume level remains the same. This is because the dynamic range of the applied audio signal is limited by the tube circuit, while the amplitude of the signal with the lowest loudness is raised. Thus, increasing tube saturation produces a slight compression effect over the entire dynamic range.

A similar effect can be perceived when analog tape is saturated. This saturation effect also compresses the recorded audio material and produces additional harmonics.

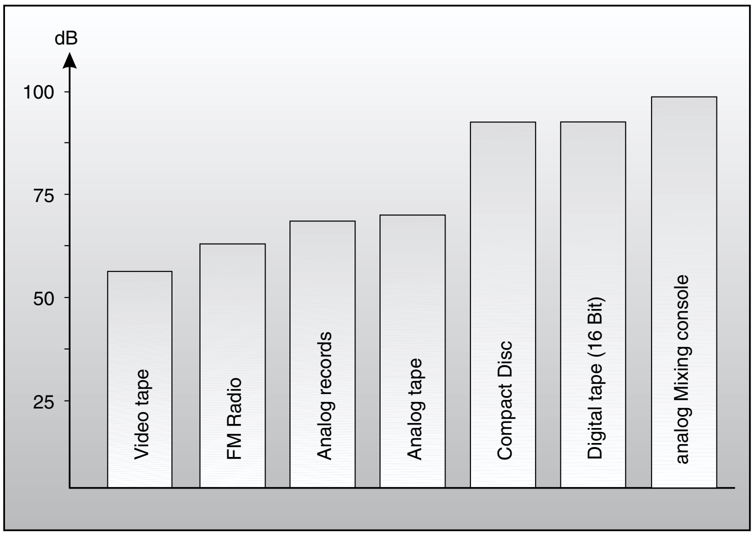

Fig. 4.8: Dynamic range of various media

5. INSTALLATION

5.1 Rack mounting

The BEHRINGER TUBE COMPOSER fits into one standard 19" rack unit of space (3 1/2" / 89.5 mm). 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 TUBE COMPOSER on high temperature devices such as power amplifiers etc. to avoid overheating.

5.2 Mains connection

The mains connection of the TUBE 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.**

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 combined fuse holder/voltage selector at the back, adjacent to the IEC receptacle. **IMPORTANT: This does not apply for general export models which are built for one operating voltage only.**

The AC voltage selection is defined by the position of the fuse holder. If you intend to change the operating voltage, remove the fuse holder and twist it by 180 degrees before you reinsert it. Matching the two markers monitors the selected voltage.

☞ **If the unit is switched to another operating voltage, the fuse rating must be changed. See the technical specifications in the appendix.**

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.

5.3 Audio connections

As standard, the BEHRINGER TUBE COMPOSER is installed with electronically servo-balanced inputs and outputs. This 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).

☞ **Please ensure that only qualified persons install and operate the TUBE COMPOSER. During installation and operation the user must have sufficient electrical contact to earth. Electro-static charges might affect the operation of the TUBE COMPOSER!**

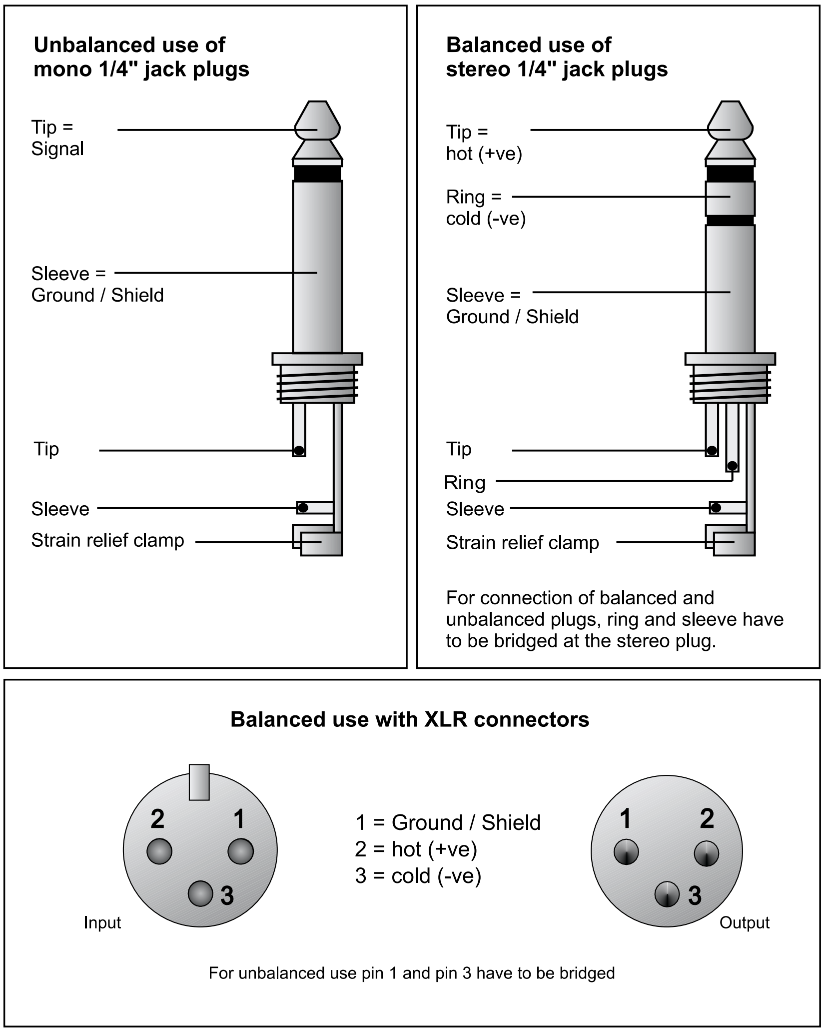

Fig. 5.1: Different plug types

5.4 Selecting the operating level

The BEHRINGER TUBE COMPOSER allows you to switch the internal operating level with the OPERATING LEVEL switches on the rear panel of the unit. Thus, you can choose between the homerecording level (-10 dBV) and the professional studio level (+4 dBu). With this adjustment, the TUBE COMPOSER is adapted to the optimal operating level. Use the input level meter on the front panel to find the appropriate operating level.

5.5 Transformer-balanced output (option)

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.

6. SPECIFICATIONS

Audio input	
Connectors	XLR and 1/4" TRS
Type	RF filtered, servo-balanced input
Impedance	50 kOhm balanced, 25 kOhm unbalanced
Nominal operating level	+4 dBu/-10 dBV switchable
Max. input level	+21 dBu balanced and unbalanced
CMRR	typ. 40 dB, >55 dB @ 1 kHz
Audio output	
Connectors	XLR and 1/4" TRS
Type	electronically servo-balanced output stage (optional transformer-balanced)
Impedance	60 Ohms balanced, 30 Ohm unbalanced
Max. output level	+21 dBu, +20 dBm balanced and unbalanced
SC input	
Connector	1/4" TRS
Type	RF filtered, DC de-coupled, unbalanced input
Impedance	>10 kOhm
Max. input level	+21 dBu
SC output	
Connector	1/4" TRS
Type	RF filtered, DC de-coupled, unbalanced output
Impedance	2 kOhm
Max. output level	+21 dBu
System specifications	
Bandwidth	18 Hz to 30 kHz, +/- 3 dB
Signal-to-noise ratio	>100 dB, unweighted, 22 Hz to 22 kHz
THD	0.008 % typ. @ +4 dBu, 1 kHz, Gain 1 0.04 % typ. @ +20 dBu, 1 kHz, Gain 1
IMD	0.01 % typ. SMPTE
Crosstalk	<-100 dB, 22 Hz to 22 kHz
Stereo coupling	true RMS detection
Expander/gate section	
Type	IRC (Interactive Ratio Control) expander
Threshold	variable (OFF to +15 dB)
Ratio	variable (1:1 to 1:8)
Attack	<1 ms / 50 dB, program dependent
Release	variable (SLOW: 100 ms / 1 dB, FAST: 100 ms / 100 dB)
Compressor section	
Type	IKA (Interactive Knee Adaption) compressor
Threshold	variable (-40 dB to +20 dB)
Ratio	variable (1:1 to oo:1)
Threshold characteristics	variable (Interactive or Hard Knee)
Auto characteristics	wave adaptive compressor
Manual attack time	variable (0.5 ms / 20 dB to 100 ms / 20 dB)
Manual release time	variable (0.05 ms / 20 dB to 5 s / 20 dB)
Auto attack time	typ. 15 ms at 10 dB, 5 ms at 20 dB, 3 ms at 30 dB
Auto release time	program dependent, typ. 125 dB/s
Output	variable (-20 to +20 dB)

Peak Limiter section

Type	IGC (Interactive Gain Control) peak limiter
Threshold	variable (+4 dB to OFF (+22 dBu))
Ratio	oo:1
Stage 1 limiter type	Clipper
Attack	“zero”
Release	“zero”
Stage 2 limiter type	program limiter
Attack	program dependent, typ. < 5 ms
Release	program dependent, typ. 20 dB/s

Function switches

SC extern	switches the detector section to the external SC input
SC mon	monitoring the external SC input, disengaging the normal audio
Interactive	enables the “Interactive Knee Adaptation” characteristics
Auto	enables the automatic and program dependent setting of the attack-/ release times, disengaging the manual attack-/release controls
I/O Meter	switches between input and output for the level meter
In/Out	relay-controlled hard-bypass
Operating level	changes the internal reference level from +4 dBu to -10 dBV
CH 1 master	linking both channels for stereo operation. Channel 1 becomes master
Warmth	variable (+10 dB to +60 dB)

Indicators

Expander/gate threshold	2 LED for under “+” and above “-”
Compressor threshold	3 LEDs for under “+”, Interactive “0” and above “+”
Peak Limiter threshold	1 LED for indication of limiter function
Function switch	LED indicator for each
Warmth	analog meter for the UTC circuitry

Options

Output Transformer	BEHRINGER transformer OT-1 retrofitable

Power supply

Mains Voltages	USA/Canada	120 V ~, 60 Hz
	U.K./Australia	240 V ~, 50 Hz
	Europe	230 V ~, 50 Hz
	General Export Model	100 - 120 V ~, 200 - 240 V ~, 50 - 60 Hz
Fuse	100 - 120 V ~: **T 1 A H**	
	200 - 240 V ~: **T 500 mA H**	
Power Consumption	30 Watts max.	
Mains Connection	standard IEC receptacle	

Physical

Dimensions (H x W x D)	3 1/2" (88.9 mm) x 19" (482.6 mm) x 8 1/2" (217 mm)
Net Weight	approx. 8 kg
Shipping Weight	approx. 10 kg

BEHRINGER is constantly striving to maintain the highest professional standards. As a result of these efforts, modifications may be made from time to time to existing products without prior notice. Specifications and appearance may differ from those listed or shown.

7. WARRANTY

§ 1 WARRANTY CARD/ONLINE REGISTRATION

To be protected by the extended warranty, the buyer must complete and return the enclosed warranty card within 14 days of the date of purchase to BEHRINGER Spezielle Studiotechnik GmbH, in accordance with the conditions stipulated in § 3. Failure to return the card in due time (date as per postmark) will void any extended warranty claims.

Based on the conditions herein, the buyer may also choose to use the online registration option via the Internet (www.behringer.com or www.behringer.de).

§ 2 WARRANTY

1. BEHRINGER (BEHRINGER Spezielle Studiotechnik GmbH including all BEHRINGER subsidiaries listed on the enclosed page, except BEHRINGER Japan) warrants the mechanical and electronic components of this product to be free of defects in material and workmanship for a period of one (1) year from the original date of purchase, in accordance with the warranty regulations described below. If the product shows any defects within the specified warranty period that are not due to normal wear and tear and/or improper handling by the user, BEHRINGER shall, at its sole discretion, either repair or replace the product.

2. If the warranty claim proves to be justified, the product will be returned to the user freight prepaid.

3. Warranty claims other than those indicated above are expressly excluded.

§ 3 RETURN AUTHORIZATION NUMBER

1. To obtain warranty service, the buyer (or his authorized dealer) must call BEHRINGER (see enclosed list) during normal business hours **BEFORE** returning the product. All inquiries must be accompanied by a description of the problem. BEHRINGER will then issue a return authorization number.

2. Subsequently, the product must be returned in its original shipping carton, together with the return authorization number to the address indicated by BEHRINGER.

3. Shipments without freight prepaid will not be accepted.

§ 4 WARRANTY REGULATIONS

1. Warranty services will be furnished only if the product is accompanied by a copy of the original retail dealer's invoice. Any product deemed eligible for repair or replacement by BEHRINGER under the terms of this warranty will be repaired or replaced within 30 days of receipt of the product at BEHRINGER.

2. If the product needs to be modified or adapted in order to comply with applicable technical or safety standards on a national or local level, in any country which is not the country for which the product was originally developed and manufactured, this modification/adaptation shall not be considered a defect in materials or workmanship. The warranty does not cover any such modification/adaptation, irrespective of whether it was carried out properly or not. Under the terms of this warranty, BEHRINGER shall not be held responsible for any cost resulting from such a modification/adaptation.

3. Free inspections and maintenance/repair work are expressly excluded from this warranty, in particular, if caused by improper handling of the product by the user.

This also applies to defects caused by normal wear and tear, in particular, of faders, potentiometers, keys/buttons and similar parts.

4. Damages/defects caused by the following conditions are not covered by this warranty:

▲ misuse, neglect or failure to operate the unit in compliance with the instructions given in BEHRINGER user or service manuals.

▲ connection or operation of the unit in any way that does not comply with the technical or safety regulations applicable in the country where the product is used.

▲ damages/defects caused by force majeure or any other condition that is beyond the control of BEHRINGER.

5. Any repair or opening of the unit carried out by unauthorized personnel (user included) will void the warranty.

6. If an inspection of the product by BEHRINGER shows that the defect in question is not covered by the warranty, the inspection costs are payable by the customer.

7. Products which do not meet the terms of this warranty will be repaired exclusively at the buyer's expense. BEHRINGER will inform the buyer of any such circumstance. If the buyer fails to submit a written repair order within 6 weeks after notification, BEHRINGER will return the unit C.O.D. with a separate invoice for freight and packing. Such costs will also be invoiced separately when the buyer has sent in a written repair order.

§ 5 WARRANTY TRANSFERABILITY

This warranty is extended exclusively to the original buyer (customer of retail dealer) and is not transferable to anyone who may subsequently purchase this product. No other person (retail dealer, etc.) shall be entitled to give any warranty promise on behalf of BEHRINGER.

§ 6 CLAIM FOR DAMAGES

Failure of BEHRINGER to provide proper warranty service shall not entitle the buyer to claim (consequential) damages. In no event shall the liability of BEHRINGER exceed the invoiced value of the product.

§ 7 OTHER WARRANTY RIGHTS AND NATIONAL LAW

1. This warranty does not exclude or limit the buyer's statutory rights provided by national law, in particular, any such rights against the seller that arise from a legally effective purchase contract.

2. The warranty regulations mentioned herein are applicable unless they constitute an infringement of national warranty law.

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