Manual



realpass Single



realpass Duo

Preface by the developer



The design and engineering of the **realbass** originate from Germany. All components, and the equipment and installation, are manufactured exclusively by domestic partners to the highest quality standards. This allows us to implement ideas consistently, sustainably, quickly and efficiently.

All equipment in the **realbass** range are conceived to be purely analogue devices and are governed by this philosophy. Economic factors are of less significance than "undistorted tone" in design and production.

Great attention is paid to the weight in the development of the equipment. Its "Sound" has top priority in this case.

Our own experience gained over many years of playing, in the most varied of configurations, is combined with an equally extensive experience and enthusiasm for electronics

This results in a lot of Headroom and Reserves that allow you all the possibilities of reinforcing and reproducing the natural sound of your instrument without having to enter into any compromises. This applies equally to both electrical and acoustic instruments.

The **realbass** range of equipment offers comprehensive and professional assistance to give full Instrument optimum showcasing of your instrument and its sound on any stage and in all appearance situations.

I wish you lots of enjoyment and relaxed successful performances with **realbass**.

Michael Schäfer

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Michael Schäfer Sommerbergstraße 37 D-66346 Püttlingen.

This manual contains technical descriptions and drawings that are not to be duplicated, spread, exploited or made accessible to third parties neither in whole nor in part without our permission. Any offences are punishable and shall impose obligation to pay damages (lt. UWG BGB).

Validity

The following documentation applies to all product types listed on the cover page. Further variations are possible. The information, details and data are presented without any warranty or liability and is subject to change.

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Translation of Original Document

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1. Important safety instructions



This document must always be read before commissioning for the first time and must be kept available at the location of use. It contains fundamental instructions that need to be observed during use. In addition, and in particular, the applicable directives and regulations issued for the location of use must be observed.

The marks and symbols used, and their significance::



DANGER! WARNING! CAUTION!

This symbol is associated with safety instructions, the non-observance of which can cause severe to fatal health dangers and injuries to persons. Observe the instructions and act particularly carefully in these cases. Pass on all the safety instructions to other users.



Warning of dangerous electrical power.



HINWEIS:

Ratschläge oder Hinweise, welche für die Nutzung und die sichere Verwendung des Produkts hilfreich sind.



Hearing protection

The noise level must not exceed a value of LwA 85 dB. Provide protection with appropriate hearing protection.

2. General instructions before and during commissioning:



Danger of electrocution!

Do not carry out servicing and repairs of any kind yourself inside the housing. Always use our trained service personnel or an authorised expert for servicing and

Faulty connection of the product can lead to life-threatening injuries caused by the electrical power. The product must only be connected to a plug socket that is earthed!

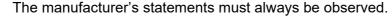


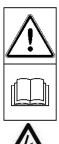
Danger of hearing damage!

The amplifier can get very loud!

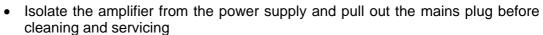
If you are constantly exposed to loud sounds, your hearing can suffer long-term damage.

If you intend to use this amplifier at a high sound level for an extended period, we recommend wearing appropriate hearing protection...





- The product must not be connected in a damp or wet environment. The power cables must not be kinked. Protect the power cables from being stepped on and close to the plug.
- The product must be connected to a socket having an earth connection.
- Never deactivate the earthing of the amplifier or its power cable! With threecore lines, the 3rd. Core must always remain connected to the mains plug socket and earth as an earth connection!
- In the event of problems, and if you have any questions concerning the power cable or mains plug, please consult an experienced electrician.



- Clean the product only using a dry cloth.
- Do not cover or close off the ventilation apertures.
- Do not operate the product close to sources of heat, such as radiators, heaters and ovens.
- Use only genuine manufacturer's accessories / fixing elements. Repairs must only be carried out by trained expert personnel.
- Repairs are only necessary if the product, a power cable, plug or components are damaged. This applies also if liquids or objects have penetrated the product or if the product has been exposed to rain and moisture.





3. General Descriptions

Phantom Power

Microphones and other signal sources often require a separate power supply from a battery, rechargeable battery or power supply. Today, this effort is avoided by supplying connected signal sources with power via the signal cable (microphone cable etc.). Therefore, a supply voltage is coupled to the signal wires of the connection cable in the input of the mixing console. There is always a DC voltage which does not disrupt the transmission of the audio signal as audio signals are always AC voltage.

The main principle of phantom-power is possible for balanced and asymmetrical signalpath. A value of 48V for balanced signal source (microphone) is usual (rare is 24V). By unbalanced source (radio-mikrophone, guitars, bass) establish 9V and 15V more and more. The value available is not stabiliszed and fall with the cable-connected load aground the consumer load (mikrophone, instrument). When there's not cable-connected load, the measured values should be in the declared effective range.

Balanced and unbalanced Signalling

This term requires no definition as it is used rather frequently throughout the description. There are two types of signalling, namely balanced and unbalanced transmission.

Unbalanced Signalling

The benefit of this transmission type is the simpler input and output switching technology of the individual devices. The disadvantage is its sensitivity to humming noises and other interferences, which increases with each metre of cable length. This type is mostly used for short transmission routes such as guitar, bass guitar or keyboard cables etc. The most frequently used ports for this are 6,3mm jack ports.

What is most notable about the unbalanced transmission is the composition of the cables used. They have a signal wire and a shield, which is simultaneously used as signal ground.

Balanced Transmission

Balanced transmission is preferred for the application of very weak signal sources such as dynamic microphones etc. It is also recommended to be used for longer cable lengths.

Its major benefit is its insensitivity to disruptions such as humming noises and interfering signals i.e. line and environment related disruptions. Balanced transmission is less critical with respect to line length.

Its disadvantage is the sophisticated switching technology for in- and output of the separate devices. It used to be applied almost exclusively to sound studio and large stage technology. Due to the drop in prices of electronic bulk sales, it increasingly captured the market and is now commonly used in low-priced devices. The same applies to phantom power being state of the art of current mixing consoles.

Mostly, the used plug connections are 3-pole XLR-connectors. The composition of cables includes two signal wires and a mutual shield also being used as signal ground. The term for this cable type is: Microphone cable.

<u>dB</u>

dB -> Decibel.

Deci -> means a tenth.

Bel(I) -> was a physicist (telephone etc.), after whom this unit was named.

The unit dB has a logarithmic measure and represents the ratio of the input and output voltage, or the input to the output sound level. The most relevant values with their effects in the fields of sound and stagecraft can be seen in the following table

Value (dB)	Ratio	Input Voltage (mV)	Output Voltage (mV)	Volume change
3	1 to 1.414	200	282	Perceptible
6	1 to 2	200	400	easily perceptible
10	1 to 3.162	200	632	Double
12	1 to 4	200	800	Double
20	1 to 10	200	2000	Fourfold
40	1 to 100	200	20.000	Eightfold
-3	1.414 to 1	200	141	perceptible
-6	2 to 1	200	100	easily perceptible
-10	3.162 to 1	200	63	Halved
-12	4 to 1	200	50	Halved
-20	10 to 1	200	20	Quartered
-40	100 to 1	200	2	Eightfold reduction

The table shows that values less than 0 dB signify lowering (reduction, damping), and values in excess of 0 dB signify lifting (amplification, increase). In addition, we can see that volume perception and the voltage values are not in a linear relationship to each other. A change of (+ or - +/-) 3 dB signifies a slight change from a purely acoustic point of view. They are only easily perceptible above +/-6 dB. This is particularly important and must be observed when using tone regulators (EQ), compressors and other effect devices.

The term 0 dB represents a particular situation. This basically means "Input = Output". In certain contexts, it also means an absolute value of 0.775 Veff at 600 Ω .

The principle of phantom power is possible both with balanced and unbalanced signal paths. For balanced signal sources (e.g. microphones) a voltage of 48V is normal (sometimes 24V), and for unbalanced sources (radio microphones, guitars, basses) a voltage of 9V and 15V is gradually becoming established. The voltage available is not stabilised and drops with the connected load by the consumer (microphone, instrument).

However, without a load (without a connected consumer), the values to be measured should be within the stated ranges.

VU- display, Modulation display

V= Volume U=Units

The term VU meter comes from sound studio technology and is used designate a measuring instrument that shows the current level of a signal source.

In order to prevent over-modulation (clipping, distortion) during modulation, the signal level is displayed either on a pointer instrument or by a chain of LEDs. The scale on VU meters of this type is either in % or in dB.

On mixer desks and amplifiers we often use just a single, generally red, LED for the indication. This indicates an impending over-modulation or undesirable distortion of the input signal by lighting up.

High-quality equipment include more more accurate displays. An adequate overview of the incoming signal can be provided by three LEDs. A green one shows an input signal that can be used, a yellow one indicate good modulation and a red LED indicates overmodulation (distortion, clipping) of the input signal.

These input signals can be regulated using appropriate regulators / switches (gain regulators or PAD switches) until just the green or yellow LEDs light up.

In an ideal situation, the indicated value on the VU meter is tapped off after the EQ (tone regulation). Increasing the specific frequencies (treble, bass, mid-frequency) changes the input signal in such a way that distortion (over-modulation) is produced, despite a good setting on the GAIN at the output from the EQ. This should be shown on the VU meter to allow you to reduce the input level accordingly.

Signal path

Your instrument produces a sound that is converted into an electrical signal by a pickup head or microphone. This signal is directed via a cable to a mixer desk or an amplifier where it is processed, modified and finally amplified.

All the locations in the movement of the signal from the instrument to the speaker are refereed to as the signal path. The first stage or an amplifier / mixer desk is called the preamplifier (preliminary stage). Here we process and amplify the input signal. After this we have Low-Cut, compressor, EQ, loop-in paths and so on. The end stage and the speaker or speakers come at the end of the chain

If we then do not use one of the components (for example the compressor or the EQ) then this no longer forms part of the signal path

Ideally, it should be removed from the signal path completely. This occurs only with socalled "True Bypass Switching". The technically more simple variant, the Simple Bypass, just separates off the output from the signal path and the input remains as a load in the signal path.

H/L - changeover

H = High **L** = Low

The designations **H** and **L** stand for the impedance (the input resistance, the apparent resistance Z) of the input circuit of amplifiers. A "high impedance" (**H**) stands for a high input resistance, a "low" (**L**) stands for a low input resistance.

High impedances are normally required for pickup heads having piezo elements without their own pre-amplifier.

They have the benefit of a precise bass, treble and dynamic reproduction. The disadvantage is the sensitivity to electrical interference in the connection line. For this application we should use high-quality cables with good screening, low capacity and, primarily, of short length.

Most instruments offer impedances between 1 M Ω and 2 M Ω . The greater the impedance, the more complicated the input circuit and the more transparent and open the tone. Impedances greater than 2 M Ω present more problems than enjoyment, despite all efforts. Low impedances are used for passive and active electrical instruments. In this case, values between 100 K Ω and 1 M Ω are common. Many guitar and bass amplifiers have either a changeover switch or two different input sockets having different input impedances. However, since these are not level-compensated, there is sometimes a not insignificant difference in volume.

On really high-quality instruments, the **H/L** changeover switch selects a completely separate pre-amplifier for high and low impedances.

Your amplifier also has two separate level-corrected input amplifiers for the jack input that you can select using the toggle switch (H/L). You will not hear any volume differences when you change from \mathbf{H} to \mathbf{L} .

If you use a high-quality cable, you should test how your instrument sounds best. Many "old" passive electrical bases sound fresh and transparent with a higher input impedance that we know only from old valve amplifiers.

<u>Gain</u>

Gain -> Yield, Increase

The term "GAIN" describes the function of the input regulator on mixer desks and instrument amplifiers. This regulator is used to adjust the amplification of the input amplifier. Sometimes it has a scale showing values between (-)50 dB and (+)20 dB. It adapts the input sensitivity of the amplifier / mixer desk to the signal source.

PAD

PAD = Attenuation

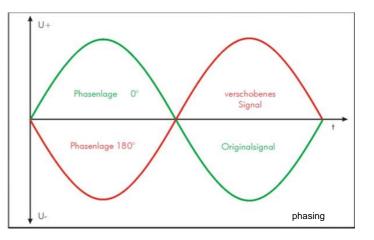
Sometimes the input signal is so high that it cannot be be adjusted with the gain regulator without distortion. The PAD switch helps in this case. It weakens the input signal in front of the gain regulator. This means that the signal gets quieter. Normal values for the attenuation of the input signal are between (-)10 and (-)30 dB.

Phase

Phase = Phasing

The "Phase" or phase reversal switch has an important function. As the word Phase says, it changes the phasing of the signal by 180°. It changes the phasing, so to say.

We need to use a little physics to make this clearer. All sound signals consist of so-called sinusoidal sounds. These have a positive and a negative half-wave (Figure: Phase reversal). The phase reversal switch "transposes" them.



Why do we need this?

A double bass and an amplifier emit the same frequencies. At short distances between the speaker and the instrument they cancel each other out or they add together. Both of these can, in extreme cases, lead to undesirable resonances.

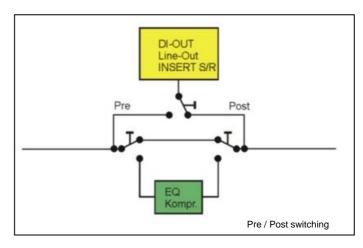
Primarily with low frequencies, the frequencies of a double bass overlap those from the amplifier speaker, for example. If you change the distance to the box or change the phasing using a switch, this phenomenon reduces almost entirely..

pre / post

pre = before, prior
post = after, later

The function of this switch refers to the tapping of a signal before (pre) or after (post) a stage in the signal chain. This mostly refers to tone regulation.

The signal tapping takes place pre (before) or post (after) the tone regulation. This can be tapping for DI Out, Line Out or the Insert (S/R) Path of an amplifier / mixer desk.

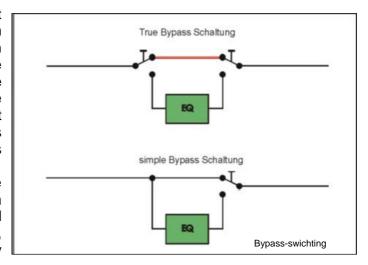


True Bypass

true = real, genuine
bypass = avoidance, bridging

Bypass means avoidance. Here it means the "Non-Use" of a switch section (EQ, compressor etc.). Such Bypass switching normally takes place in the "simplest" way. The output of the circuit section is decoupled from the signal path. However, since the input (EQ / compressor etc.) is always associated with the signal, this continues to be influenced by it.

On the other hand, with a "True Bypass", the input and output of a circuit section are completely released from the circuit section. In this way, with a "Bypass Case", we get a fully uninfluenced signal path.



<u>EQ</u>

EQ -> equalizer, tone regulator

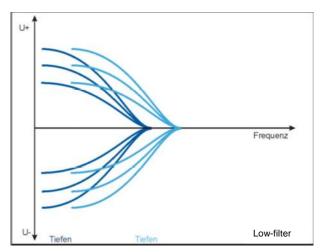
These circuits serve to manipulate tone nuances and frequency responses. In principle, you should use an equalizer as little as possible. Ideally, it is would not be needed at all.

High-Pass - Low-Pass - Band-Pass

A **Low-Pass** is an electrical circuit that reduces or strengthens all frequencies starting from a specific frequency.

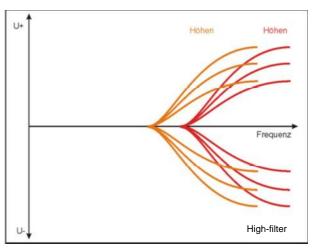
The frequency at which a Low-Pass operates is called the Corner Frequency. The elevation or lowering of the frequencies influenced by the Low-Pass is carried out using the Bass Regulator (Bass).

Some amplifiers / mixing desks have an additional switch or regulator to influence the Corner Frequency.



High-Passes operate in a similar way to the Low-Pass and are electrical circuits that lower or reinforce all frequencies starting from a specific frequency. The frequency at which a High-Pass operates is called the Corner Frequency.

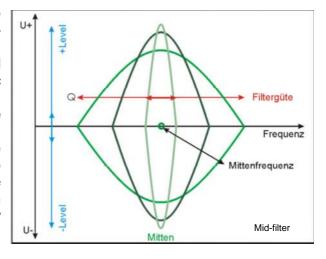
The elevation or lowering of the frequencies influenced by the High-Pass is carried out using the Treble Regulator (Treble). Some amplifiers / mixing desks have an additional switch or regulator to influence the Corner Frequency.



Band-Passes are electrical circuits that are a combination of a High-Pass and a Low-Pass.

This is used to lower or reinforce all frequencies starting from a specific frequency up to another specific frequency. The lifting or lowering is carried out using the Mid-Regulator (Mid).

Some amplifiers / mixer desks have switches or regulators that are used to modify or adjust the mid-frequencies, the reinforcement or lowering and the width (filter quality, Q) of the influenced frequency band.

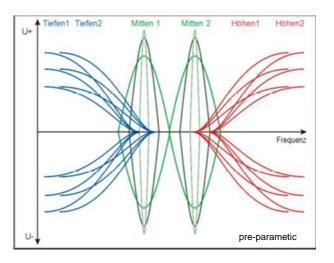


Parametric / Fully parametric

parametric = parametrisable, adjustable

Some EQ circuits are designated as "parametric or fully-parametric EQs".

They have at least one regulator for each range (bass, mid-frequencies, treble) for the reinforcement or lowering of the frequency range, a second one for the point of application (Corner Frequency, Mid-Frequency) and possibly a third one for the width of the frequency range to be influenced. Fully-parametric EQs offer extreme possibilities for influencing sound signals. You should therefore use this tool with great care and economy.



LOW-CUT

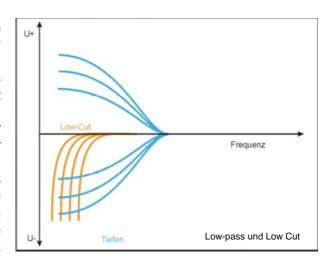
LOW-CUT= ubsonic filter, rumble filter

The LOW CUT refers to a High-Pass which is generally only designed for very low frequencies.

When it is activated, all frequencies (between 10 and 150 Hz) are filtered out. It is generally just a switch (on/off).

Higher quality instruments have a regulator that can be used to adjust the Corner Frequency of the LOW-CUT.

As we can see from the graphic, the bass regulator and LOW-CUT are influenced. The bass regulator can operate (depending on the setting of the LOW-CUT) in the same frequency range and possibly only in a restricted manner.



Compressor / Limiter

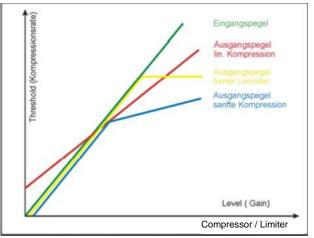
Compressor = acoustic, changes the dynamics (volume) of a signal **Limiter** = limits the maximum level of a signal

The function of a compressor can be very varied and they can be realised in a variety of ways. The principle is quite simple, however. Let's assume a very dynamic signal source. It provides signals with large differences in volume. If these signals are quiet, then they are only slightly reinforced, and if they are very loud, they will be reinforced very loudly.

In order to avoid distortion, the sensitivity of the amplifier / mixer desk must be adapted to the highest possible level (volume) of the signal. This may result in certain parts becoming too quiet under certain circumstances.

This effect can only be counteracted by continuous correction of the input sensitivity. Electrical circuits were developed to avoid this dilemma, and these are called compressors or / and limiters. The difference between the compressor and the limiter is fluid.

A limiter has an adjustable reference level. If the signal level remains below this, then nothing happens. If it gets louder, it dampens down the signal so severely that it always remains less than the set reference level...



A compressor can do a little more but it operates in a similar way to a limiter. It compares the input signal level with its internal reference level. Quiet signals are reinforced and loud ones are weakened.

The whole installation can be influenced and regulated more or less extensively depending on the circuitry. Decisive significance is allocated to the two regulators "Level" and "Threshold".

Level

It determines the overall reinforcement of the compressor. Many possibilities of dynamic influencing are available depending on the setting in combination with the "Threshold Regulator".

Threshold

This regulator is used to set the reference level (deployment threshold). If the signal is louder than this set reference value, then the compressor starts to take effect. The lower the setting of the reference level, the more sections of the signal are captured by the compression, the more the compression.

Since the signal consists of a mixture of frequencies, some of the frequencies are reinforced or reduced and other not by the use of a compressor in the compression range.

This leads to interesting tone variations and makes the sound of an electric bass thick and round without having to use an EQ and without getting louder.

Insert (Send / Return)

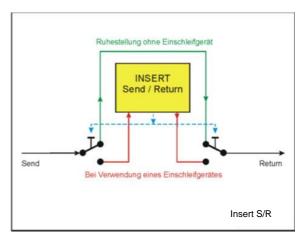
Insert = add (loop-in)
Send = output (transmitter)
Return = input (receiver)

An "Insert Connection" means a connection combination that permits "looping-in" an additional device into the signal path (effect devices such as reverberation, echo, chorus etc.).

This needs an output (send) and an input (return). These are directly connected together in the unused status.

If a device is looped in, the direct connection is separated and both connections are connected to the effect device.

We normally use 6.35 mm jackplugs for connection. Output and input are either executed as one connection (stereo socket) or separately (2 mono sockets)



Ground-Lift

Ground = earthing, earth, signal earth **Lift** = raising, removal (separating))

The "Buzzing" (mains buzzing, humming, motorboating) familiar on all stages occurs when several instruments are connected together both on the mains side (power) and on the signal side (sound).

The actual causes are mostly not easy to find and rectify because they are often associated with the local installation (lights etc.).

In order to provide a solution, many instruments have a switch which separates (switches off) the earthing of the signal (signal earth). This switch is called the "Ground Lift".



The earth line on the mains cable must never be removed or taped over!

Mute

mute = silent, quiet

The mute button / switch / key on an amplifier / mixer desk mutes the instrument. This is normally indicated by a coloured LED. Many instruments allow "Muting" using a footswitch. On multi-channel instruments, alternative "Muting" is helpful for changing input channels.

4. Technical Description

4.1 General points

realbass- amplifiers are designed as combo-amplifiers. This means that both functional units (speaker and amplifier) are housed in a single enclosure. They can also be used separately, however The electronics (head or amplifier) can be removed and mounted in a rack, for example. 19" retaining brackets are available for this purpose (see accessories).

Up to 2 boxes (speakers), each having an impedance of 8Ω , can be connected. Both output sockets are designed as combi-sockets (Speakon / 6.35 mm jack).

4.1 Brief instructions Front



twin-channel amplifier

The following explanations refer to the **single-channel** version of the **realbass**. On the twin-channel amplifier the AUDIO-IN is missing on the second channel.

The input is realised as a **combi-socket** for XLR and jack. Behind this there are three fully separate pre-amplifiers that are automatically activated depending on the plug and switch setting (L/H).I

The XLR section of the input socket is designed as a classic microphone input with and without phantom input (48V) and the jack section is designed as a stereo jack (TIP = signal, RING = phantom input). It contains two separate input pre-amplifiers that can be selected using the toggle switch **L/H**. Here again, a phantom input can be added (15V).

Classic electric bass of **Instr**. Input with jack (Low $Z \rightarrow L$ ") High-quality **FET** input with jack (High Z <> H")

The **VU** display provides information about the signal volume. It evaluates the signal after the EQ, since EQ settings can always lead to large changes to the signal. Correct the signal strength using the gain regulator until the green LED and the yellow LED come on in the **VU** display when playing. If the red LED comes on, the input signal must be reduced (gain regulator).

The **PAD** switch dampens the input signal by 20dB. The phasing of the signal can be turned through 180° using the **PHASE** switch

The **GAIN** regulator adapts the input signal to the amplifier input.

The **LOW-CUT** can be used to limit the low frequencies of the input signal 910-130 Hz). If the regulator is set all the way to the left (7 o'clock), all frequencies below 10 Hz are suppressed, and if the regulator is set all the way to the right (5 o'clock), then all frequencies below 130 Hz are suppressed.



The **compressor** can be looped into the signal path with the associated switch (compressor). The two regulators **Level** (volume) and **Threshold** (compression rate) serve to set the compressor



The Pre/Post switch serves the tapping of the signal for the loop in path (INSERT) before or after the tone regulation. The tone regulation itself is looped into the signal path using the EQ switch.

The operating elements for tone regulation are subdivided into three sections.

The three mid regulators (**MID**):

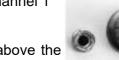
Frequency = used to set the mid frequency of the mid regulator.

Level = regulates the increase or decrease of the volume of the set frequency range.

Q = this regulator selects the quality (breadth) of the selected frequency range.



Audio-IN offers the possibility of reproducing music for practice via a 3.5 mm jack-plug socket. The associated regulator adapts the volume of the input signal to your practice volume. This input is only available on channel 1 with a twin-channel unit (DUO).



You can "mute" each channel separately (silencing). The red LED above the **MUTE** switch comes on in the "muted" status.



The MUTE function can also be operated using a foot-switch. This makes the installed MUTE switch ineffective.

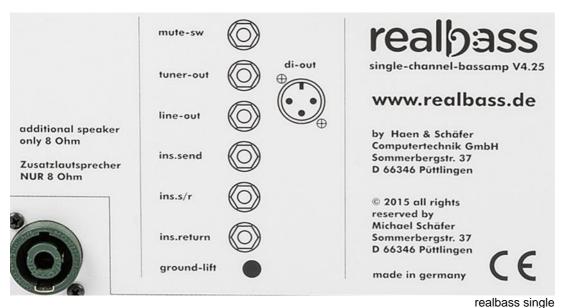
The **SINGLE Version** only has a **Volume Regulator.** This is used to set the maximum volume.

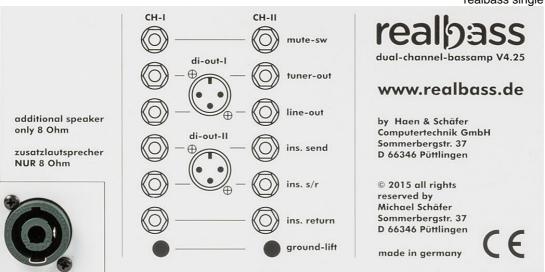


The **DUO Version** has an additional **Volume Regulator** for each of the two channels (adjusting the channel volume) and a **Master Regulator** for setting the overall volume.



4.2 Brief instructions Back





realbass duo

mute-sw tuner-out line-out	Connection for the ANS multi-switch foot-switch for muting Connection for instrument tuners with Line Level (-10dB) Unbalanced output for external units with Line Level (-10dB)
inssend ins. s/r	Output of the loop-in path with Line Level (-10dB) Combi-connection, output and input of the loop-in path with Line Level (-10dB) (stereo jack 6.35 mm) TIP = Send, RING = Return, SHIELD = Ground
ins. return dl-out	Input of the loop-in with Line Level (-10dB) Balanced output per channel for mixer desk FOH (-10dB)
ground-lift	Switch-off for the earth connection of the specific DI output

The mains connection socket is on the back of the **realbass** - amplifier, next to the cooling body.

The mains voltage fuse is installed between the mains switch and the cooling socket.



Warning of electric power

The setting of the mains voltage is marked on your unit. Check this before inserting the mains cable.



Use only 20mm glass fuses having a value of 4AT (slow-response).

The speaker connections are located in the bottom left corner on the back

Speakon/jack combi-socket	8Ω- speaker / back bottom left
Speakon socket	8Ω- speaker / bottom left

Your amplifier is connected to the box by cable when it leaves the works. This cable is inserted in the Speakon socket on the bottom of the amplifier. If you separate the amplifier section from the box you need to remove this cable..



The unit does not have fan cooling.
The hot air must be dissipated by convection.
Do not cover or tape over the ventilation slits.

Your unit is designed for a mains voltage of 230V, 50Hz. On request we can also supply a changeover version for 230V, 50 Hz and 110V, 60 Hz. This can be marked explicitly on the back with the corresponding safety instructions.

All versions operate with 8Ω speakers.

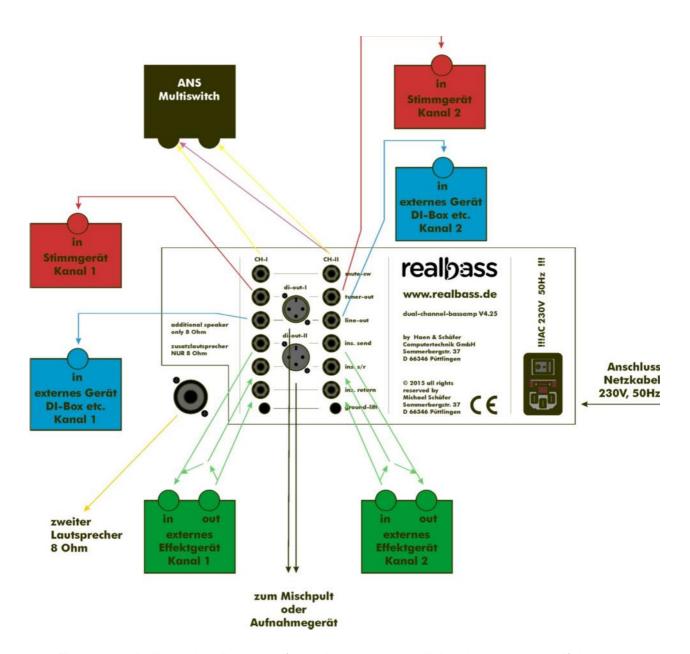
When using two speakers (internal + external or 2x external), a total impedance of at least 4Ω must be maintained (2 x 8Ω speakers).



If this impedance is not reached, then this can lead to damage to the final stage.

4.3 Connection diagramm back side

This overview is intended to provide the experienced user with a simplified general view. More detail is available in the explanations.



The arrows indicate the direction of the signals at the individual connections. If they point away from the specific connection, then it is a pure output.

If they point towards the connection, then it is an input.

5. Commissioning

Take time with initial commissioning and read this chapter carefully. We have taken great care to make all the necessary details understandable even by the layman. However, if you still have any questions then please consult an experienced user or give us a call.

In the following details you will often find the text "switch on or off". Your amplifier mostly has pushbuttons with round black caps. When a switch is off, the cap protrudes further, and if it is switched on this is shown by the cap being depressed. There is generally an LED above the switch to indicate the state of the switch. Please take this into account during operation.

All regulators have a mark in the shape of a white line on the knob. The position of the regulator can be seen using the idea of a clock "dial". The mid position is set when the mark points upwards centrally (12 o'clock), really quiet (closed or off) to the left (7 o'clock) and really loud (fully open) to the right (5 o'clock).

5.1 Initial Commissioning:

- 1. Check the packaging for any transport damage. After unpacking, check the unit for optically perfect condition and check the accessories for completeness
- 2. If everything is OK, check the voltage quoted on the back for which the unit is designed. If you are unsure, ask your dealer or contact our experts.
- **3.** If the mains voltage available is suitable for your unit, plug the mains cable into the mains socket and connect the mains cable to mains plug socket.
- **4.** Push the **MUTE** button(s) downwards (switch down = "mute" on).
- **5.** Turn the overall volume to zero (volume and master regulator all the way to the left).
- **6.** Also turn the Gain regulator to zero (all the way to the left).
 - Now switch the main switch on at the back. The mains check lamp on the mains switch should come on, as well as the mute LED(s).
 - Switch off the compressor and the EQ by actuating the relevant switches.
 Both LEDs (compressor and EQ) should be off (switch pointing upwards = off).
- 7. Then insert the plug on your instrument cable first in the instrument and then into the input socket on the amplifier. For passive (high-ohm) piezo pickup-heads and passive electric bases use the jack input. Switch the toggle switch H/L all the way to the left to position "H" (top). With active pickup-heads and electric basses select the position "L".
 - Use the XLR input for microphones ans similar symmetrical pickup-heads. If your instrument / microphone / pickup-head needs phantom input, then push down the relevant switches. The blue LED indicates whether there is a phantom input.

- **8.** Play a couple of notes and slowly turn the gain regulator to the right until the green LED comes on and the yellow LED flickers (VU display).
 - If the red LED on the **VU** display comes on, turn the **gain** regulator to the left until it goes out.
 - If that is not possible, then push the **PAD** switch down. (Switch down = **PAD** active). No you can proceed with the **Gain** regulator until just the green and the yellow LEDs come on.
- **9.** Push the **MUTE** switch. The Mute LED should go out. (Switch upwards = "mute" off).
- 10. Turn the **Volume** regulator to 100% and the **Master** regulator to 50% (12 o'clock) and play a couple of notes. You should now hear them through the amplifier. Experiment with the **PAD** switch, **Gain** and **Volume** regulators and **Master** regulator and watch the **VU** display while doing so.

The interaction between the Volume and the Master regulator is relatively easy to understand.

The Volume regulator and Master regulator are each set to 50% and this means 50% of 50% = 25% of the possible volume.

50% on the Master regulator gives 50% of the possible volume

11. Turn the **LOW-CUT Regulator fully** to zero (7 o'clock). Play a few low notes and then turn it **fully** to the right (5 o'clock) and play the same notes again. You will then detect a clear difference in the bass reproduction. The foundations of the sound will change.

If the regulator is fully to the left (7 o'clock), the **LOW-CUT** will have an effect on frequencies under 10 Hz. All the deep rumble noise (sub-bass) is thus eliminated. If this regulator is all the way to the right (5 o'clock), then frequencies less than 120 - 130 Hz are damped.

The setting of the **LOW-CUT** will also influence the setting of the **EQ**. The effect of the bass regulator will be less, depending on the frequency at which you set the **LOW-CUT**.

Choose the position of the **LOW-CUT** that sounds best to you at the specific time. This will be different for each space. The bigger or more undamped the space and the louder you play, the more important is the setting of the **LOW-CUT**.

12. A peculiarity of the EQ (tone regulation) of the amplifier is that you can take it out of the signal path completely. Neither the input nor the output are then connected to the signal. The EQ has "disappeared" so-to-say.

Before starting the test, turn the five EQ regulators (bass, frequency, level, Q, treble) to the mid positions (neutral position, 12 o'clock). This setting then approximates to that of the switched off EQ. Play a few notes to get used to the tone. Now switch the EQ on (switch bottom = EQ active) and the EQ LED should come on. Play a little.

You will hear a noticeable difference compared to previously. First turn the bass regulator fully to the left (7 o'clock). The basses are now severely reduced. You will note that this change can no longer be compared with that of the **LOW-CUT**, although the **LOW-CUT** also influences the bass range, depending on the setting. Then turn the bass regulator gradually to the right (5 o'clock). You will be able to hear a strong increase in the lower frequency range.

Observe the **VU** display while doing so. As soon as the red LED comes on, you will need to take back the **Gain** regulator until just the green and the yellow LEDs light up.

Then return the bass regulator to the mid position.

In order to test the function of the treble regulator, carry out the same procedure as with the bass regulator.

Then return the bass and treble regulators to the mid position.

14 For the mid regulation involves three regulators:

This - frequency, level and Q (quality). The basics could already be seen in Chapter 2. Now let's make your first tests with a filter of this type:

- Frequency regulator adjusts the mid frequency that we wish to influence.
- Level regulator determines how much the set frequency is to be lifted or lowered
- **Q** regulator determines the "breadth" of the frequency band of the influence.

For the first test, we leave the **Frequency** regulator in the mid position, the **Q** regulator is turned fully to the right (5 o'clock = narrow frequency band) and the **Level** regulator is turned fully to the left (7 o'clock = maximum damping).

The **Level** regulator now severely lowers the level of the set mid frequency. The **Q** regulator provides an extremely narrow frequency band. Play a few notes whilst turning the **Frequency** regulator in small steps from full left to full right. You will hear a type of WahWah effect. The note will sound thinner and quieter in the middle, depending on the frequency setting.

Now turn the **Level** regulator to the right (5 o'clock) and repeat the procedure with the **Frequency** regulator. You will now hear a reversed WahWah effect. At the same time, the signal will be heard to be thicker and clearly louder in the set range, depending on the setting of the **Frequency** regulator.

Here again you always need to take account of the **VU** display and possibly adapt the level using the **Gain** regulator.

Repeat these tests whilst also varying the setting of the Q regulator (regulator at 5 o'clock = very narrow frequency band, regulator at 7 o'clock = very broad frequency band). With a little practice you will quickly determine how effective and powerful the tone regulation can be.

The optimum situation is when you don't need any tone regulation!

15. Now let's consider the second powerful tone tool, the **compressor**. The basics concerning the functioning of this effect can be taken from Chapter 2.

What ist he effect of the compressor?

- makes the sound more direct,
- makes the sound more direct
- reduces volume differences in various different playing techniques.

The compressor can be completely removed from the signal path completely using a switch (True Bypass Switching). Neither the input nor output are then connected to the signal. The compressor has "disappeared", so-to-say.

When "slapping", high dynamic spikes (volume differences, level differences) occur which often cause annoyance and are undesirable. If we use a compressor in a skilful way these can be reduced considerably without changing the typical sound.

First turn the Level regulator to the mid position (12 o'clock) and turn the **Threshold** regulator fully to the left (7 o'clock). The reinforcement by the compressor should be OK in this way. Switch the Compressor on and push the switch (compressor) down (switch down = compressor on). The green LED above the switch should be on.

Now vary the **Threshold** regulator in both directors and listen for any changes. Play a little and experiment with both regulators until you have understood (heard) the effects.

Some songs need a very even weighty bass sound without volume fluctuations. This can be realised very well using the available Compressor. The regulation behaviour is unobtrusive yet very effective. You may need a certain period of time before you can use it profitably

When you have reached this point in the operating manual, you have become acquainted with all the operating elements. In conclusion, we would ask you to go through these steps several times.

Take the time and trouble to do so. It will help you to become acquainted with all the functions in detail in order to use them safely.

Uncomplicated working and an outstanding sound will be the benefits of this task. Your amplifier will already sound very natural and warm, even without compressor and without EQ. Your instrument and your personal sound will fully come into its own

We wish you lots of enjoyment and success in using your **realbass** amplifier. For suggestions that help us to improve our products we are always grateful. Send us your feedback.

We look forward to it



For questions and professional support about **realbass** we are available by telephone: +49 6806 44449.

6. Technical data

realbass	SINGLE	DUO	
Amplifier (combi)	fanless, single channel fanless, twin-channel		
Speaker	12" High-Power 500W / 900W AES		
End stage	FET, short-circuit protected, analogue, 500 W pea		
Mains unit	conventional with toroidal core transformer 1000 W Peak		
Connection	230V, IEC	C (power) connector	
Speaker connection	2 x Speakon / 6.35 mm jack (combi)		
FET input stage changeover impedance, can be added in Phantom input 15V	yes	per channel	
Microphone amplifier High-End, very low noise, can be added in Phantom input 48V	yes	per channel	
Gain regulator	yes	per channel	
PAD switch 20dB	yes	per channel	
Phase changeover switch	yes	per channel	
Compressor, with Level and Threshold, can be added in (True Bypass)	yes	per channel	
LOW-CUT infinitely variable setting 12 dB, 10 – 130 Hz	yes	per channel	
EQ with fully parametric mids, can be added in, (True Bypass)	yes	per channel	
Audio-IN for recording, can be regulated, stereo jack 3.5 mm	yes	yes (just channel 1)	
INSERT with separate and common Send/Return sockets	yes	per channel	
DI-OUT active (-10dB) (XLR) with Ground Lift switch	yes	per channel	
Line-Out active (-10dB) Jack 6.35 mm	yes	per channel	
Tuner-Out, active (-20dB) Jack 6.35 mm	yes	per channel	
MUTE remote jack 6.35 mm	yes	per channel	
Weight: Carbon fibre housing Wooden housing	from 11,6 Kg from 14,3 Kg	from 13,5 Kg from 14,9 Kg	
Dimensions (HxWxD)	455 x 455 x 385 mm		

7. Optional accessories

We provide a comprehensive range of accessories for our equipment.

- Foot-switch ANS Multiswitch Foot-switch for alternating or complete muting
- 3m remote cable for ANS Multiswitch for alternating or complete muting
- 5m remote cable for ANS Multiswitch for alternating or complete muting
- Protection sleeve (HxWxD) 460 x 460 x 390 mm (included with carbon fibre housing))
- Installation kit for 19" operation of the amplifier section (head) in a 19" rack

8. Guarantee

§ 1 RELATIONSHIP WITH OTHER GUARANTEE RIGHTS AND WITH NATIONAL LAW

This guarantee does not affect the rights of the purchaser with regard to the seller arising from the concluded purchase contract.

These guarantee conditions of the company Haen & Schäfer Computertechnik GmbH are applicable as far as they do not contradict the specific national law with regard to guarantee conditions.

§ 2 ONLINE REGISTRATION

Please register your new unit if possible directly after purchasing at **www.realbass.de** on the internet and please read the guarantee conditions carefully. If your unit is registered with us, including its date of purchase, the procedure in the event of a guarantee claim will be considerably simpler. Many thanks for your co-operation!

§ 3 GUARANTEE

Haen & Schäfer Computertechnik GmbH guarantees the the mechanical and electronic components of the product, in accordance with the conditions quoted here, for a period of one year*, starting from the acquisition of the product by the purchaser. If faults arise during this guarantee period, that are not caused by one of the items in § 5, then Haen & Schäfer Computertechnik GmbH will, according to their own judgement, either replace the unit or repair it using new or renovated replacement parts of the same quality. If this includes the use of replacement parts that improve the unit, then Haen & Schäfer Computertechnik GmbH is entitled to charge the customer the costs of this, according to the own discretion.

In the event of a justified guarantee claim, the product will be returned without charging for the freight.

Any guarantee provisions other than those detailed above will not be covered

§ 4 REPAIR NUMBER

In order to be able to check for justification of the repair under the guarantee, the guarantee provision requires that the purchaser or his authorised dealer call Haen & Schäfer Computertechnik GmbH (see enclosed list) during normal business hours BEFORE sending the unit in, to advise of the fault that has arisen. The purchaser or his authorised dealer will be given a repair number.

The unit must then be returned with the repair number in its original packaging. Haen & Schäfer Computertechnik GmbH will advise where to send the unit.

Freight collect shipments will not be accepted.

§ 5 GUARANTEE CONDITIONS

Guarantee provisions will only be executed if, together with the unit, a copy of the original invoice or cashdesk receipt issued by the dealer is presented. In the event of a guarantee claim, the product will fundamentally be repaired or replaced.

If the product needs to be modified or adapted to comply with the applicable national or local technical or safety requirements of the country, which is not the country for which the product was originally conceived and manufactured, then this does not count as a material or manufacturing fault. The guarantee also does not include the execution of such modifications or adaptions, irrespective of whether they were carried out properly or not. Haen & Schäfer Computertechnik GmbH does not accept any costs for modifications of this type within the framework of this guarantee.

The guarantee does not entitle cost-free inspection or maintenance or repair of the unit, particularly if the defects can be traced back to improper use. Also, the guarantee claim cannot cover defects in wear parts that can be attributed to normal wear. Wear parts are, in particular, potentiometers, switches/buttons, input and output sockets, lights and similar parts.

In addition, damage to the unit is not rectified under the guarantee that is caused by:

- improper use or misuse of the unit for purposes other than its normal purpose with non-observance of the
 operating and maintenance instructions issued by Haen & Schäfer Computertechnik GmbH,
- connection or use of the product in a way that does not comply with the applicable technical or safety requirements in the country where the unit is being operated;
- Damage that is governed by force majeur or by other causes that are not the responsibility of Haen & Schäfer Computertechnik GmbH.

The guarantee right expires if the product has been repaired or opened up by an unauthorised workshop or by the customer themselves.

If a check of the unit by Haen & Schäfer Computertechnik GmbH shows that the damage in evidence does not form part of the guarantee entitlement and that claims cannot be made, the costs of the checking procedure by Haen & Schäfer Computertechnik GmbH will be paid by the customer.

Products without guarantee entitlement will only be repaired if the costs are paid by the purchaser. If there is no guarantee entitlement, Haen & Schäfer Computertechnik GmbH will inform the purchaser of the lack of guarantee coverage. If a written repair order is not issued within 6 weeks of this information for the payment of the costs, then Haen & Schäfer Computertechnik GmbH will return the unit sent in back to the purchaser. The costs of freight and packaging will be invoiced separately and will be charged by cash on delivery. If the repair order is issued with cost acceptance, the additional costs of freight and packaging will also be invoiced separately.

§ 6 TRANSFER OF THE GUARANTEE

The guarantee is provided exclusively for the original purchaser (official dealer's customer) and is not transferable. Except for Haen & Schäfer Computertechnik GmbH, no third party (dealer etc.) is entitled to issue guarantee assurances for Haen & Schäfer Computertechnik GmbH.

§ 7 CLAIMS FOR DAMAGES

As a result of defective performance of the guarantee, the purchaser is not entitled to claim for damages, in particular also not for consequential damages. The liability of Haen & Schäfer Computertechnik GmbH is limited, in all cases, to the goods value of the product.

*More detailed information is available to EU customers from Haen & Schäfer Computertechnik GmbH

Notes		
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