ROSTEC Station 6

Mic/Line Preamplifier and Monitor Controller

Revision 7, September 15, 2017

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Station 6 Preamplifier and Monitor Controller

Features

- 2 microphone/line inputs at the back panel
- 2 auxiliary line inputs at the back panel
- 2 auxiliary cue inputs at the back panel
- 2 high impedance instrument inputs at the front
- Automatic switching between instrument input and mic/line input
- Balanced insert points for microphone channels
- Insert points bypass function at the front panel
- Signal switching by sealed gold contact relays
- True balanced architecture throughout the unit
- Ultra low noise and distortion
- +30 dBu microphone preamp input headroom
- Exceptionally open and transparent sound
- Input circuits have vacuum tube characteristics
- Smooth mic gain adjustment from +10 to +70 dB by potentiometer, no clicks.
- Built-in +48 Volts Phantom Power
- Linear low noise power supply. *No Switchmode!*
- Slot for optional USB/AES/LAN digital interface.
- Inputs and outputs are ESD protected to 23 kV IEC 61000-4-2 and 15 A surge IEC 61000-4-5.
- Sturdy steel metal casing, electrically and magnetically screened.
- Stand-alone desktop or with mountable 19" rack mount flanges

General description

Station 6 is an ultra low noise and ultra low distortion mic/line preamplifier and monitor controller designed with the professional sound engineer in mind.
Station 6 is constructed by using modern day's cutting edge technology, but the basic design philosophy is inspired by the very best sounding audio equipment that have been manufactured over the past 50 years.

Station 6 uses the same highly acclaimed preamplifiers as the LMA8 mic/line preamp, which gives the unit a basic sound quality of extraordinary high class.
The unit is extremely versatile. It can operate in analog mode as a personal audio center for the performing artist, or it can operate in digital mode as a control center for a recording studio or a video dubbing facility.
It has two high performance Class-A microphone/line preamps, two auxiliary line inputs, two monitor line outputs to speakers and a powerful headphone amplifier with talk-back facility.
It also features zero latency feed forward controls from the mic/line inputs to the headphones.
Two cue direct line inputs are available for zero latency monitoring of external signals.

With the **USB module** installed, Station 6 communicates with the DAW via USB full-speed. The USB connection is compatible with USB 2.0, USB 3.0 and USB-C.

With the **AES module** installed, Station 6 communicates with the DAW via a standard AES3 110 ohm transformer balanced interface.

With the **LAN module** installed, Station 6 communicates with the DAW via standard Gigabit Ethernet.

Station 6 features an unusual large input headroom on all inputs, enabling it to handle fast transient and large dynamic level changes. At the same time it reproduces micro-details and environmental depth perspective with a natural openness and an impressive accuracy.
The microphone input circuits are designed so input clipping cannot be experienced. The input circuits have a clipping limit (headroom) at +30 dBu, and because there is always at least +10 dB of gain in the signal path, the output (or the connected equipment) will simply always clip before the input.
The line inputs also have +30 dBu clipping limit and communicates with the DAW at the standard level of +20 dBu as Digital Full Scale.
Again, under normal circumstances, input clipping cannot be experienced.

Station 6 is a purist's dream come true. The design is based on a very stringent philosophy, meaning the shortest possible signal path and the highest possible quality components. The microphone input circuit is a true class A differential gain stage, and it has a transfer characteristic that resembles that of a vacuum tube, giving the unit a natural, relaxed and open sound, yet it maintains high-speed and precise response.

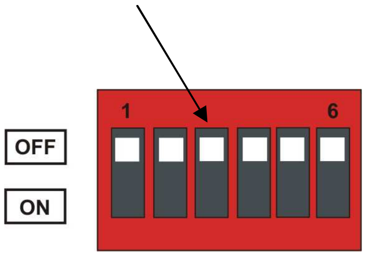
The architecture is fully balanced throughout the unit, which means that the signals between the various circuits are routed as a positive and a negative signal, not using ground to transfer audio signals at all.
This architecture keeps the audio path free from non-linear distortion from currents running in the ground mesh or from "non musical" signals from external electrical fields, power supply noise, crosstalk from other channels etc.

All signal routing and gain regulations are performed by analog components. There are no digital switching and no digital gain regulation to compromise the sound quality. High grade conductive plastic potentiometers and hermetically sealed relays with gold contacts are used throughout the unit. *No compromise!*

Microphone, instrument and line inputs

The 2 ***microphone inputs*** are available at the 25 pin D-SUB female connector on the back panel.
The gain is set by a potentiometer at the front panel from +10 dB to +70 dB in one turn (no clicks!).
The +48 V Phantom Power is switched on/off at the front panel. A Phantom Power block function is provided by setting the ***DIP switch 3*** at the back panel.

In the OFF position, Phantom Power is allowed.
In the ON position phantom power is blocked for both microphone channels. The +48 V LED at the front will flash briefly if switching on phantom power at the front is attempted while the block function is ON.



The microphone inputs also work as line inputs.
The 6 kohm input impedance is perfect for line signals (mind the phantom power though).

The 2 ***instrument inputs*** are available at the 1/4" Jack connectors on the front panel.
They are direct inputs to the microphone preamplifier and follows the gain setting at the front panel.
Plugging in a Jack connector automatically switches off the microphone input at the back panel and changes the input impedance to 1 Mohm, which is perfect for electric guitars and electric bass. The input is equally well suited for line level equipment, such as keyboards and the like.

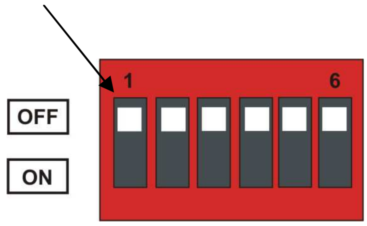
An unusual feature of the instrument inputs is that they are fully balanced. They work equally well with standard unbalanced instrument cables as well as balanced cables. However, using a balanced cable with result in a substantial noise reduction.
This feature is highly advantageous on stages, and in electrically noisy environments. (the cabling is described in detail on page 20)

The 2 ***auxiliary line inputs*** are available at the 25 pin D-SUB female connector on the back panel.
They have fixed gain, and are direct feeds to the DAW in digital mode, and direct feeds to the monitor outputs in analog mode.
The sensitivity is the standard + 20dBu for 0 dBFS and the maximum input level is +30 dBu.

The 2 ***auxiliary cue inputs*** are available at the 25 pin D-SUB female connector on the back panel.
They have fixed gain, and are direct feeds to the cue/headphone outputs. The signal is mixed in hardware with the cue signal from the DAW.
The sensitivity is the standard + 20dBu for 0 dBFS and the max. input level is +30 dBu

The ***talk-back input*** is available at the back panel via a 3.5 mm mini jack connector. The input is intended for dynamic microphones, but a +5 V phantom power for electret or lavaliere microphones can be selected by setting the ***DIP switch 1*** at the back panel.

In the OFF position, no phantom power is provided.
In the ON position +5 V phantom power is provided.



The microphone input gain is set by a trim-potentiometer at the back panel from +26 dB to +70 dB.

Insert points

The 2 microphone channels each have an analog ***insert point*** available at the 25 pin D-SUB female connectors on the back panel.
The direct output of the microphone channels is routed to the ***insert send***, and the ***insert return*** is routed directly to the digital module and subsequently to the DAW.
The insert point can be bypassed by pressing a switch at the front panel.
The direct output from the mic channel is always available at the insert send point, no matter whether the insert point is bypassed or active.

There are two obvious advantages to having these insert points:

It means that an old beloved analog compressor, equalizer or similar can be inserted in the signal chain between the microphone preamp output and the DAW.
It also means that the user has a direct line input to the DAW via the insert return, when the insert point is activated.
The insert points operate at standard level:
Insert send output is + 20dBu for 0 dBFS. Insert return input is + 20dBu for 0 dBFS. Max input level is +30 dBu.

Direct outputs

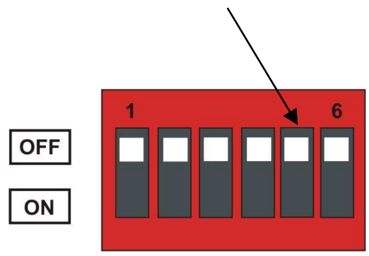
The 2 ***direct outputs*** from the insert points are available at the 25 pin D-SUB female connector on the back panel.
The direct outputs are intended for external monitoring of the inputs and outputs of the insert points.

There are two signal paths:

When the insert point is active, i.e. the inserted analog equipment is inserted in the signal chain, the direct output is sourced from the output of the equipment.
When the insert point is bypassed, i.e. the inserted analog equipment is not inserted in the signal chain, the direct output is sourced from the input of the equipment.

Monitor outputs

The **monitor *outputs (monitor L/R)*** are available at the 25 pin D-SUB female connector on the back panel.
The monitor outputs are intended for use with active control room monitors.
Two different sets of monitors, SPKRS1 or SPKRS2, can be selected by the pushbutton at the front panel.
The monitor outputs feature high performance balanced floating industrial buffers, able to handle any impedance from 600 ohm to infinite.
The nominal output level at 0 dBFS is selected by setting the ***DIP switch 5*** at the back panel.
OFF position is +8 dBu, ON position is +20 dBu.

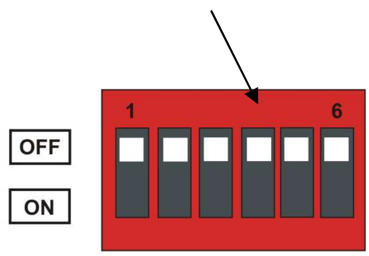


The max output level is +28 dBu, which means that under normal conditions, the buffers always have extra headroom and thus no output clipping will occur.

3 signal paths of the monitor signals are possible:

1. In digital mode, the outputs are sourced from the digital module (from the DAW) as default.
2. In analog mode the outputs are sourced directly from the auxiliary line 3 and 4 inputs as default.
3. In both digital and analog mode, the outputs can be switched to be sourced from the insert return points by setting the ***DIP switch 4*** at the back panel.

In the OFF position, the monitor outputs are sourced as described above in point 1 and 2.
In the ON position, the monitor outputs are <u>always</u> sourced from the insert return point



Cue headphone outputs

The cue outputs (headphone L/R) are available at two 1/4" Jack connectors, one at the front panel and one at the back panel.
The outputs are intended for dynamic headphones and features an unusual high output capability, able to drive any impedance from 4 ohm to infinity. The outputs have a maximum output swing of 45 V peak-peak and a maximum output current of 1.2 Amp.

The headphone amplifier is protected against short circuit and thermal overload. If the maximum current draw is exceeded on either channel, both output signals are muted for approx. 2 seconds after which they are un-muted again.
No limiting or compression of the signals is used. When the signals are present, they are clean and undistorted.

In order to avoid burning your headphones, two power levels are available, selected by a switch at the front panel.
LO is for the smaller in-ear headphones and for semi-pro headphones typically with impedances ranging from 32 to 100 ohms.
HI is for professional headphones with impedances from 100 to 600 ohms.

Observe that these are only guidelines!

The meter can assist you. If you are on LO setting and the RED clipping LEDs light up and you still feel the need for more power, turn down the volume first, then go to the HI setting.
Be careful! you don't want to burn your headphones (!) The headphone amp is unusually powerful.

The cue/headphone outputs have 4 signal sources.

1. In digital mode, the signals are sourced from the digital module (from the DAW). In analog mode, these signals are disabled.
2. Direct feed from the microphone channels, controlled by the volume potentiometers at the front panel.
3. Auxiliary cue inputs, available at the 25 pin D-SUB female connector on the back panel. These input are mixed directly into the signal path.
4. Talk-back signal from the microphone connected to the 3.5 mm mini jack at the back. The signal is activated by a pushbutton at the front and mixed into the cue/headphone signals via its own volume control.

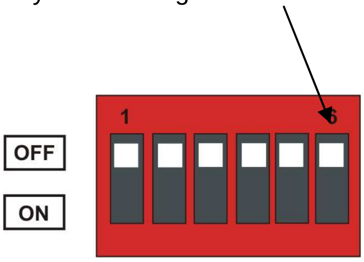
Digital Mode versus Analog Mode

Station 6 can operate in two modes, ***digital mode*** and ***analog mode***. The inputs and outputs are basically the same in the two modes, but there are some notable differences in the internal signal flow.

Digital mode:

Station 6 operates in digital mode as default when a digital module is installed.
This mode is intended for a studio set-up with a DAW with 4 channel recording and 4 channel monitoring.
Analog mode can be selected by setting ***DIP switch 6*** at the back panel. When analog mode is selected, the digital module goes into standby, and all clocks and digital processing is halted.

In the OFF position, the unit operates in digital mode if a digital module is present. If no digital module is present, the unit automatically selects analog mode.
In the ON position, the unit enters analog mode and puts any installed digital module in standby.

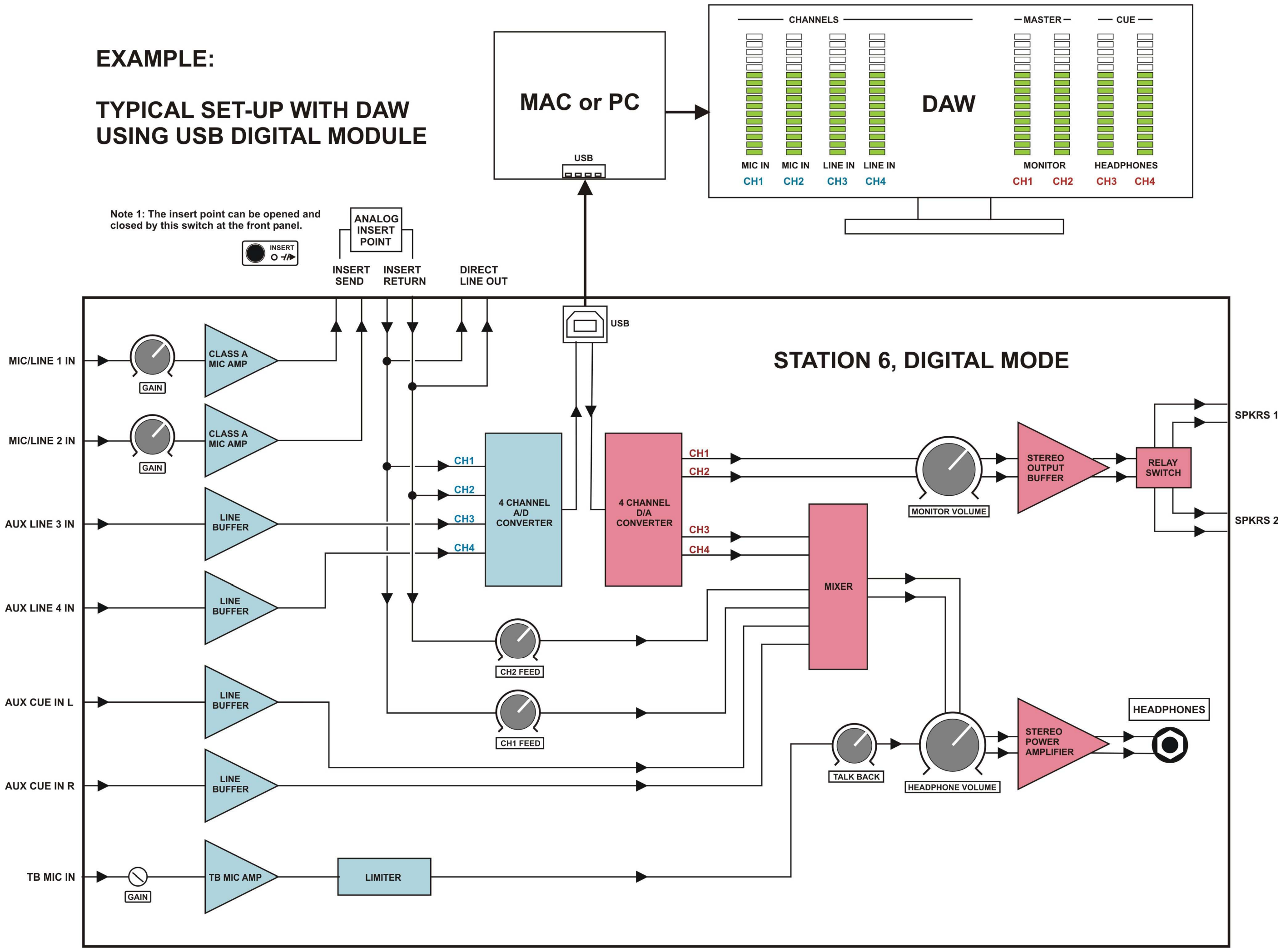


Analog mode:

When no digital module is installed, or if ***DIP switch 6*** at the back panel is set, Station 6 enters analog mode.
This mode is intended for the creative and technically knowledgeable artist, technician or engineer.

In analog mode the digital module is disabled and the A/D and D/A converters are bypassed, but all inputs and outputs are still available.
The signal flow can be configured in lots of various ways. Some examples are shown on the next pages, but only the imagination sets the limits.

Station 6 Preamplifier and Monitor Controller



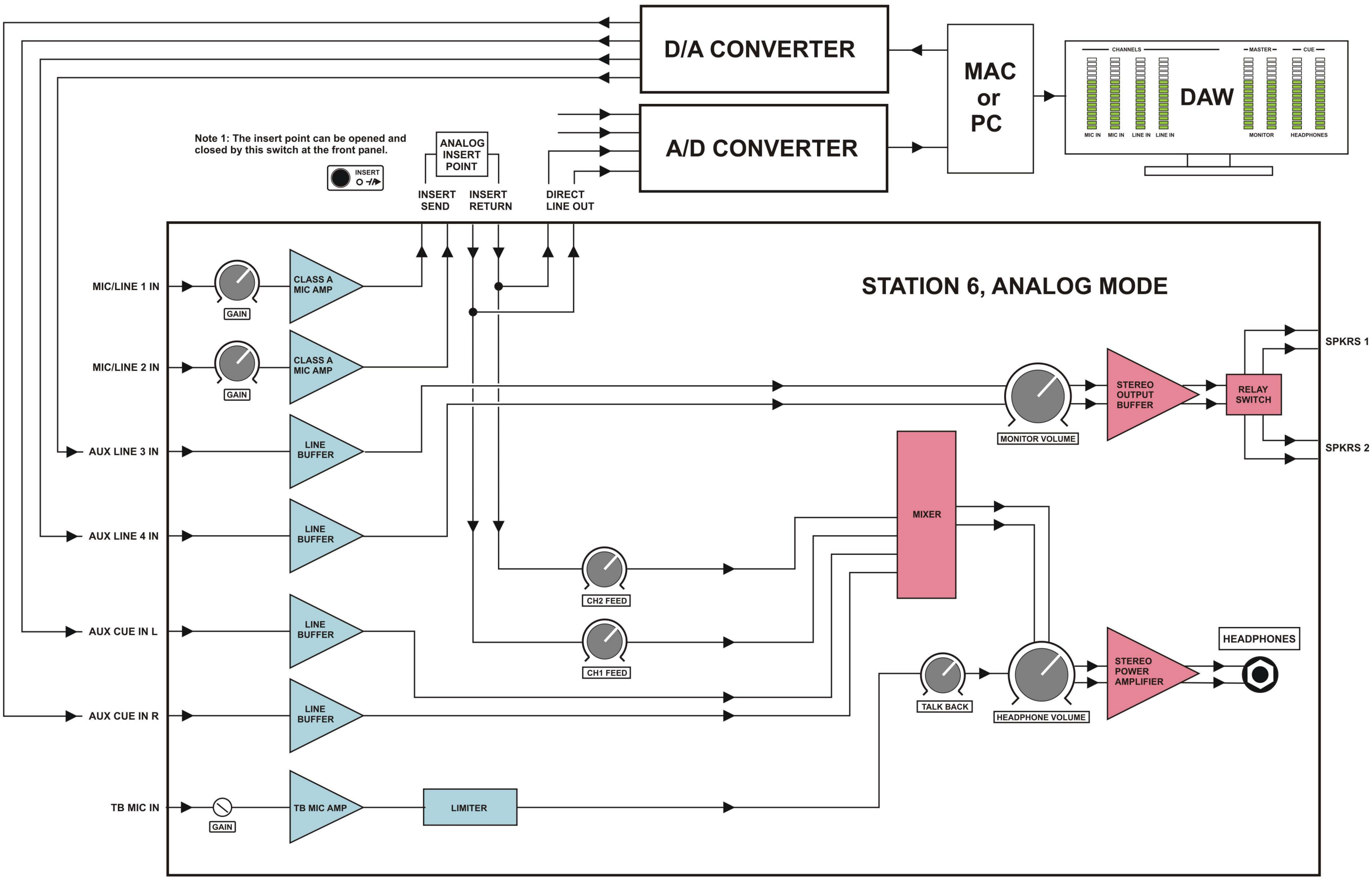
In the typical studio set-up with a DAW, the 4 signals from the inputs Mic1, Mic2, AUX Line3 and AUX Line4, are sent to the A/D converter and subsequently to the DAW via the USB link.
The 4 output signals from the DAW are sent to the D/A converter via the USB link, and subsequently sent to the Monitor and Cue(headphone) outputs. Channel 1 and 2 are sent to Monitor left and right, and channel 3 and 4 are sent to Cue left and right.

Each of the inputs Mic1 and Mic2 have an Insert Point for use with analog equipment like Equalizers, compressors, filters, what not. Note that the inserted analog equipment is placed in the signal chain *before* the A/D and the DAW.
The Line3 and Line3 signals are routed directly to the A/D and the DAW.

The Cue(headphone) outputs receive the signals from a local mixer, which has 4 input sources.

1. The main Cue outputs from the DAW and the D/A
2. The direct feeds from the microphone channels via potentiometers at the front, sourced from the Insert Point Returns.
3. The AUX Cue inputs Left and Right, straight into the mixer and to the Cue(headphones) output.
4. The Talk Back Input from the TB microphone input, controlled by the TB switch and volume potentiometer at the front.

EXAMPLE: TYPICAL SET-UP WITH EXTERNAL A/D AND D/A CONVERTERS



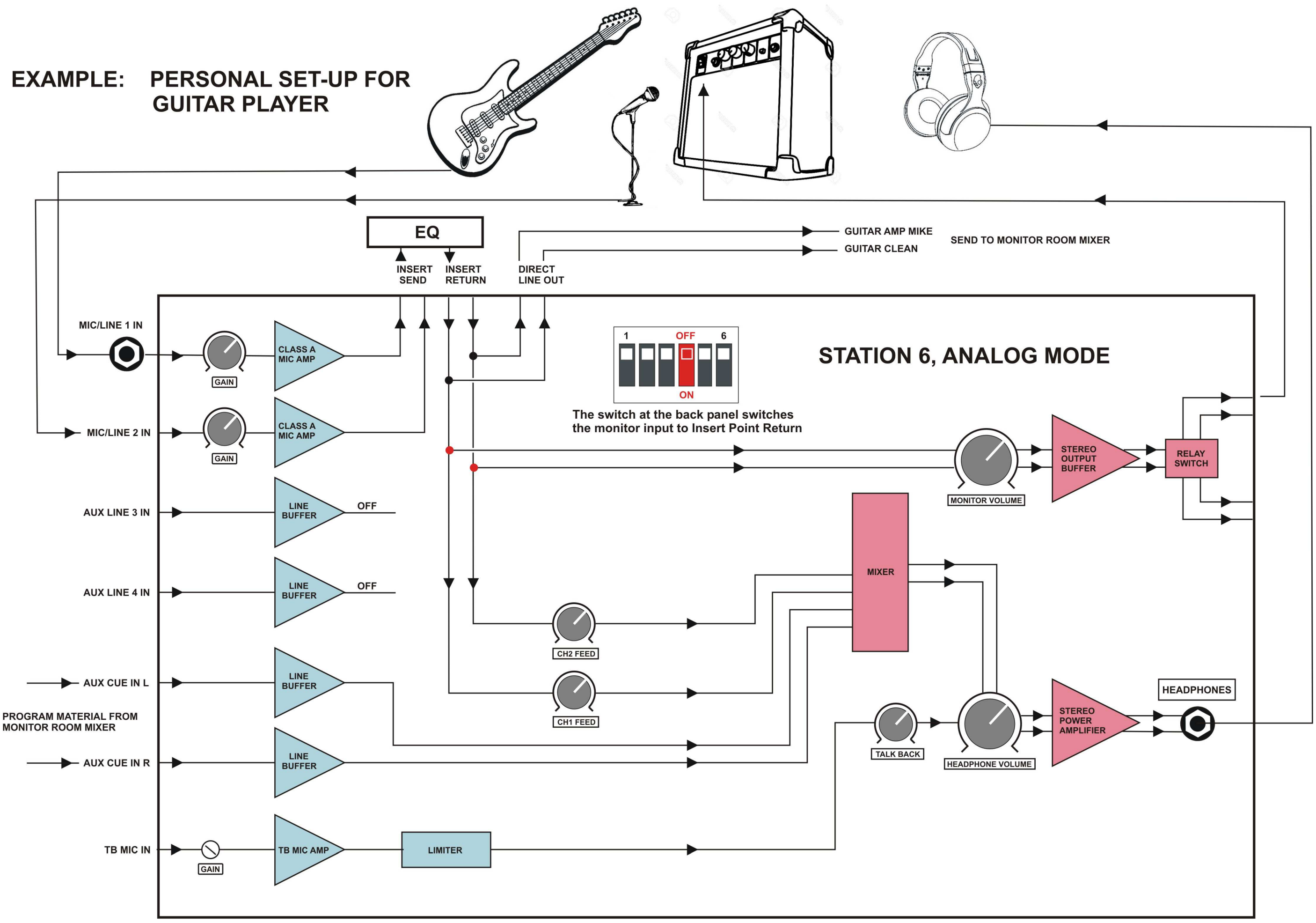
When Station 6 is used with external A/D and D/A converters, it must operate in analog mode. There is no need to install a digital module. With no digital module installed, the unit automatically enters analog mode.
If a digital module is installed, just set the ***DIP switch 6*** to position ON. This will activate analog mode and deactivate the digital module.

Inputs Mic1 and Mic2 are sent to the external A/D converter and to the DAW. Each of the inputs Mic1 and Mic2 have an Insert Point for use with analog equipment like Equalizers, compressors, filters, what not. Note that the analog equipment is placed in the signal chain *before* the external A/D and the DAW.

The monitor (master) outputs from the DAW and the external D/A are connected to the AUX Line 3 and AUX line 4 inputs and directly routed to the Monitor Left and Right outputs.
The Cue outputs from the DAW and the external D/A are connected to the AUX Cue Left and Right inputs and routed, via the mixer, to the Cue(headphones) Left and Right outputs.

The Cue(headphone) outputs receive the signals from a local mixer, which has 3 input sources.
1. The main Cue output from the DAW and the external D/A, via the AUX Cue Left and Right inputs
2. The direct feeds from the microphone channels via potentiometers at the front, sourced from the Insert Point Returns.
3. Talk Back Input from the TB microphone input, controlled by the TB switch and volume potentiometer at the front.

Station 6 Preamplifier and Monitor Controller



Above is an example showing Station 6 used as a personal center for the guitar player. The example is based on analog mode, but it works equally well in digital mode.
If no digital module is installed, the unit automatically selects analog mode. If a digital module is installed, you can choose between analog and digital mode by the ***DIP switch 6*** at the back panel.

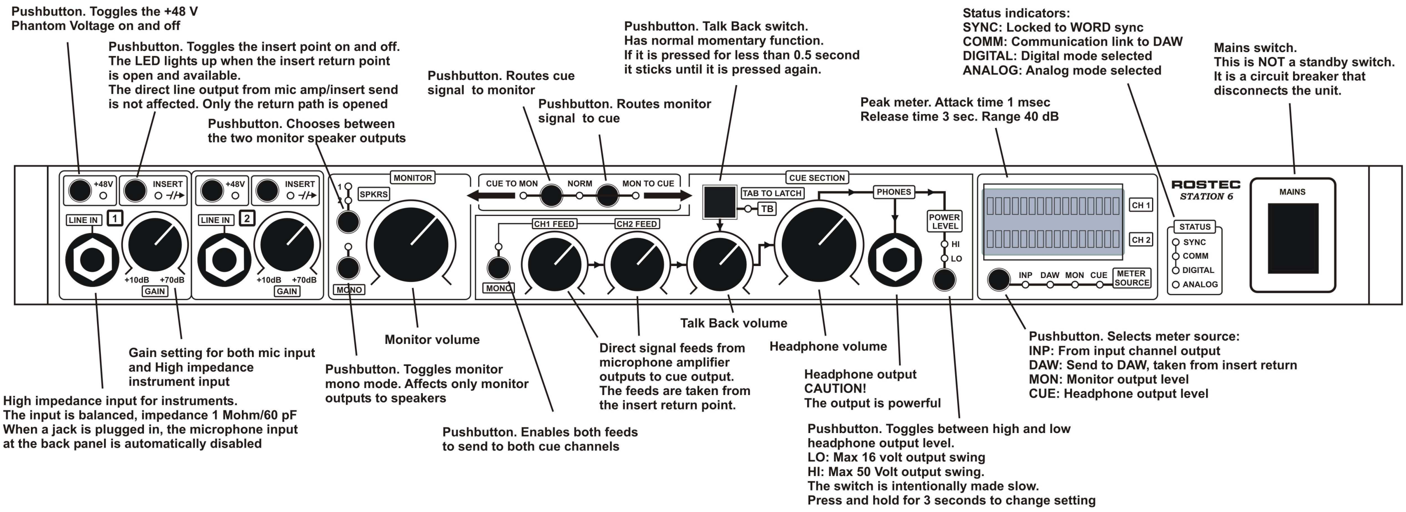
At a glance, the example seems complicated, but it is actually pretty much straight forward.
First, set ***DIP switch 3*** to ON position. This will connect the monitor outputs Left and Right to the Insert Point Return. It will also disconnect the Auxiliary Line 3 and 4 inputs in analog mode. (In digital mode the inputs are still connected to the A/D)

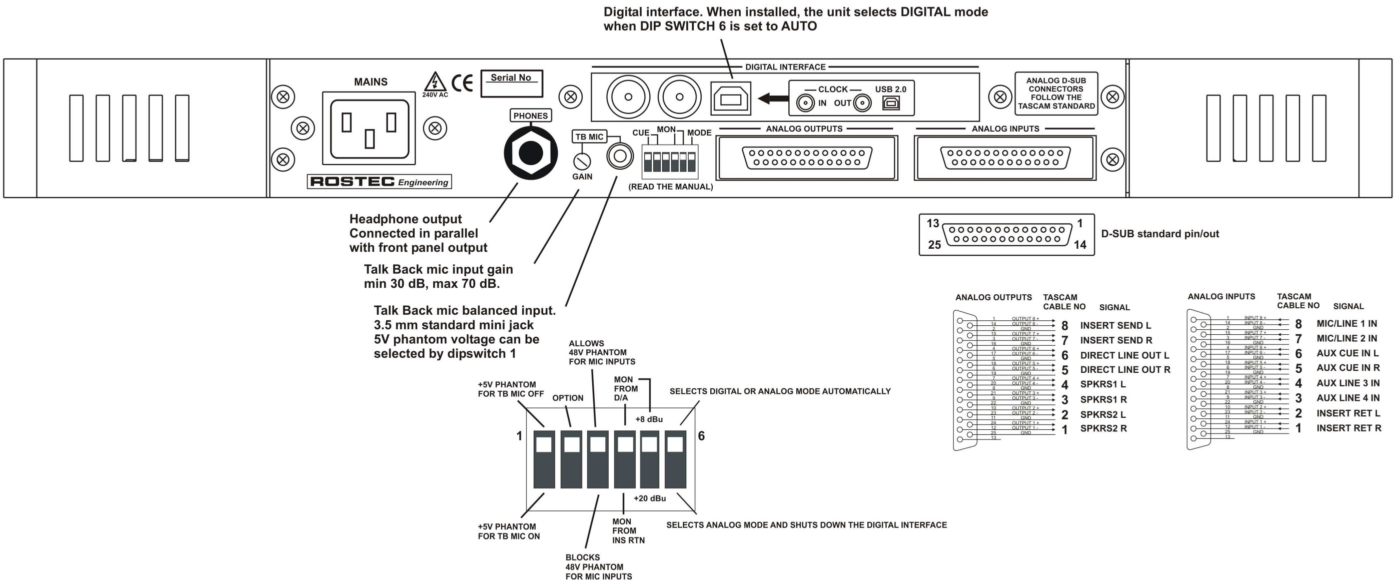
The guitar is plugged into Mic1 Jack input (high-Z) at the front. The signal it then routed through the insert point to the monitor left output, and to the direct output to the control room mixer. The monitor left output is then connected to the guitar amp input.
The microphone in front of the guitar amp is connected to Mic2 input. The signal it then routed through the insert point to the monitor left output, and to the direct line output to the control room mixer.

Program material from the control room mixer is sent to AUX Cue Left and Right input. The guitar player can then listen to the program material from the monitor room, and mix the signals from the guitar direct/guitar-amp by the feed potentiometers at the front.

Operational description

Quick guide, front and back controls and connections

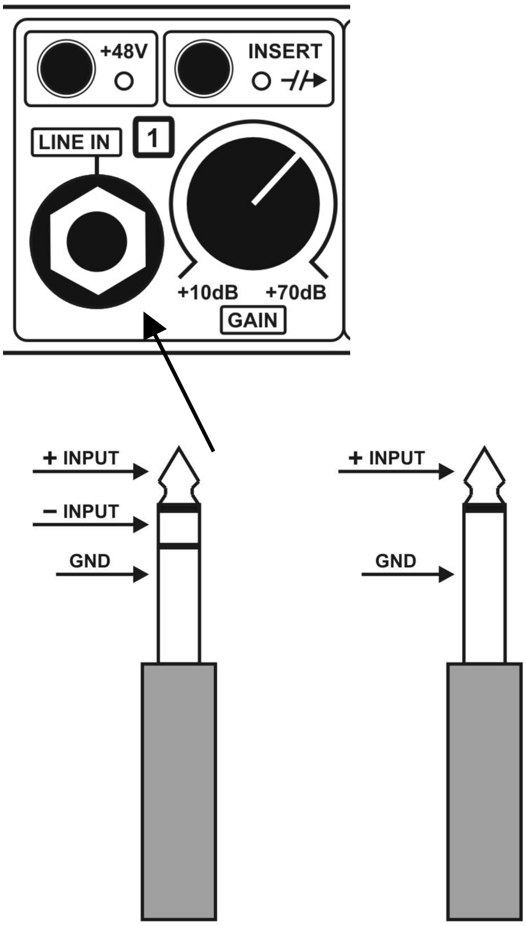




ANALOG OUTPUTS TASCAM CABLE NO	SIGNAL
8	INSERT SEND L
7	INSERT SEND R
6	DIRECT LINE OUT L
5	DIRECT LINE OUT R
4	SPKRS1 L
3	SPKRS1 R
2	SPKRS2 L
1	SPKRS2 R

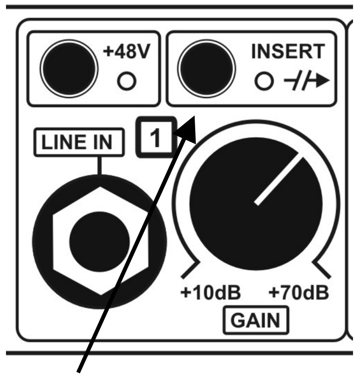
ANALOG INPUTS TASCAM CABLE NO	SIGNAL
8	MIC/LINE 1 IN
7	MIC/LINE 2 IN
6	AUX CUE IN L
5	AUX CUE IN R
4	AUX LINE 3 IN
3	AUX LINE 4 IN
2	INSERT RET L
1	INSERT RET R

Front panel



Instrument/line input

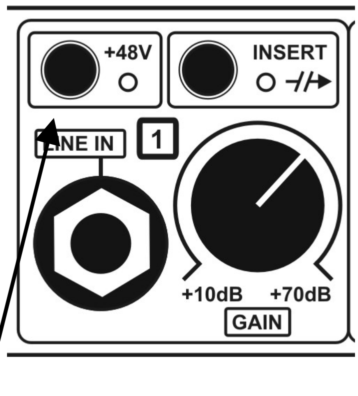
Each channel has a ¼" Jack instrument/line input on the front panel. The input is for use with instruments (like guitar, bass etc) and it can be used in a balanced as well as an unbalanced configuration.
When a Jack is plugged-in, the input circuit automatically de-activates the mic input at the back panel, switches to high impedance mode and activates the line input.
Plugging in a mono Jack enables the normal unbalanced input configuration.
Plugging in a stereo Jack enables the balanced input configuration.
For further information about using the balanced configuration see page 8, *Remote ground connection*



Insert point switch

The toggle switch controls the bypass function of the insert point. The insert point exists electrically between the output of the preamp and the input to the digital module.

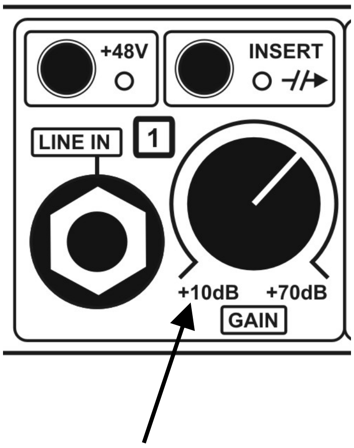
When the LED is ON, the signal chain between the preamp and the digital module is “open” which means that external equipment can be inserted between the insert send and the insert return.
At the same time, the signal at the insert return point is also sent to the direct line output.
When the LED is OFF, the signal is routed directly from the preamp to the digital module. The insert return input is then inactive.
At the same time, the signal from the insert send point (preamp out) is also sent to the direct line output.



+48V Phantom Voltage

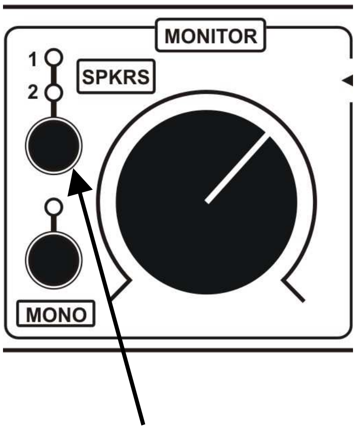
The toggle switch turns the Phantom Voltage at the mic input on and off. The LED indicates the status.
When phantom voltage blocking is selected by setting ***DIP switch 3***, the +48V LED will flash briefly if you push the button. No phantom voltage will be applied.

Station 6 Preamplifier and Monitor Controller



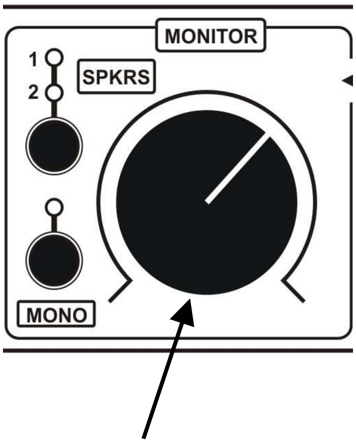
Input gain adjust

Turning the knob controls the input gain from minimum to maximum in one smooth movement. No clicks and no jumps. It works for both instrument/line input and microphone input.
The gain range is +10 to +70 dB from input to the line output on the D-SUB connector at the back.



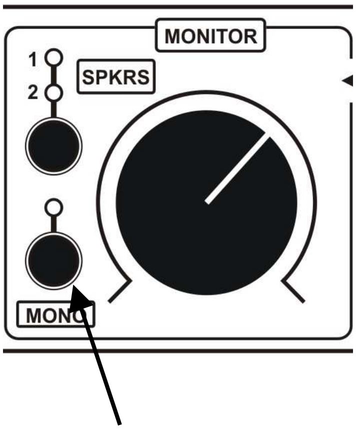
Monitor speaker select

Two different sets of speaker sets can be connected.
The toggle switch controls which speaker set is active.
Only one speaker set at a time can be activated.
The switching is performed by relays with gold contacts, and the levels at both outputs are identical.



Monitor volume control

Turning the knob controls the monitor output level from minimum to maximum in one smooth movement. No clicks and no jumps.
The output level range is from -80 dBu to +8 dBu,
(to + 20 dBu if ***DIP switch 5*** is set)

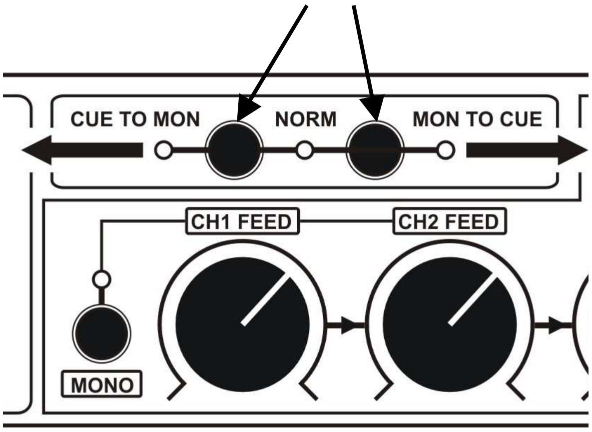


Monitor mono/stereo switch

The toggle switch selects stereo or mono mode of the monitor output.
When the LED lights up, mono mode is selected. No light indicates stereo mode.
When switching between mono and stereo, odd phase relationships in the signal can be revealed. The average output level to the speakers remain unchanged.

Monitor and cue signal routing

A practical way of interchanging the monitor and cue signals is available by two toggle switches.
In a studio set-up, the function comes in handy, when the sound engineer wants to hear in the monitor speakers what the artist hears in the headphones.
Or when the artist want to hear how the monitor mix sounds in the headphones.

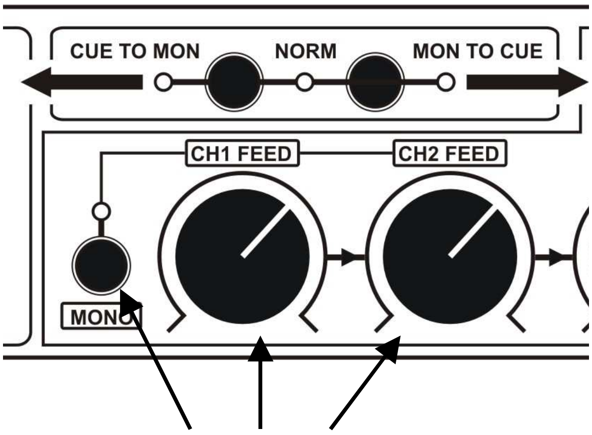


When pressing the Cue To Mon switch, the headphone signal is sent to the monitor speakers. The cue signal in the headphones is not affected. Pressing the switch again brings back the monitor signal into the speakers.

When pressing the Mon To Cue switch, the monitor signal is sent to the headphones. The monitor signal in the speakers is not affected. Pressing the switch again brings back the cue signal into the headphones.

Activating both functions at the same time will simply swap the monitor and the cue signals.

A LED lights up when the corresponding routing functions is activated. The NORM LED lights up when none of the routing functions are activated.

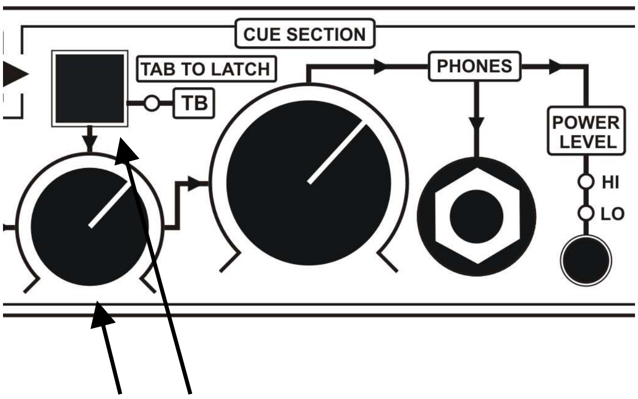


Zero Latency Monitoring

Direct signal feed from the 2 microphone channels to the headphones is provided. The signal are instantaneous analog feed-forward and do not pass any switches or digital circuitry.
The signals are mixed directly into the cue signal, and the volume is controlled by two potentiometers at the front. CH1FEED is sent to cue left and CH2FEED is sent to cue right.

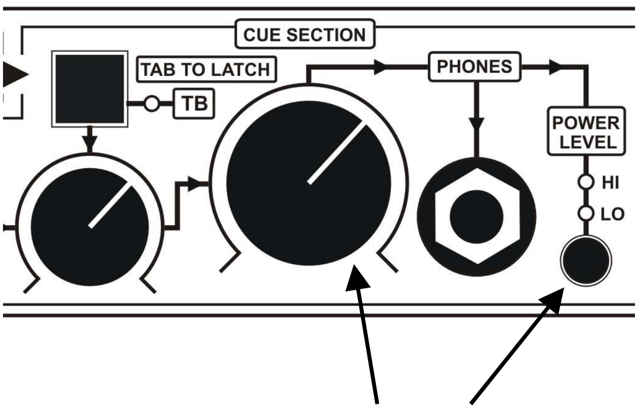
This means that a stereo recording from Mic1 and Mic2 can be sent directly to the headphones with zero latency. In case only one mic channel is used, it would be quite annoying to hear the signal only in one side of the headphones. To get the signal in both sides, mono mode can be selected by a toggle switch at the front.

Note that this switch <u>do not</u> reduce the left and right signals to half level, like a standard mono function would do to keep the sound impression at the same level. Consider this switch as a standard routing function of the signal, with no reduction of signal level. The signal goes to one side or the signal goes to both sides.



Talk Back

The talk back function is controlled by a pushbutton and a volume potentiometer at the front. The talk-back signal is mixed directly into the cue signal.
The pushbutton is a standard momentary switch. Push it and the talk back is active. Release it and the talk back is muted.
However, if you push and release it quickly, within 0.5 second, it "sticks" and the talk-back is kept active. Push it again and the talk-back is muted again.
A red LED lights up as a warning when the talk-back is active.



Headphone output

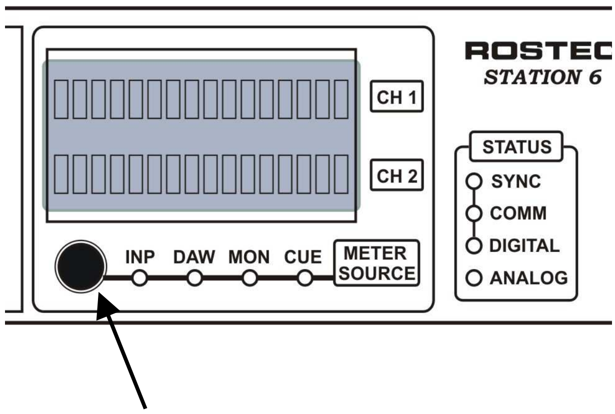
The cue signal is sent to the headphone amplifier and the output to headphones is sent to the 1/4" Jack at the front panel and to the 1/4" Jack at the back panel.
The volume is controlled by the potentiometer and the two level switch. The switch is a toggle switch, that switches between high and low output level. The switch has a 3 seconds delay in order to protect against unintentional use.

The headphone amplifier is powerful. At high setting it is able to give out 45 Volt peak to peak and a maximum current of 1.2 Amp.
The high setting is for professional headphones. It will easily burn semi-pro and budget headphones.
Hence a low setting is provided so you can use your in-ear headphone from you mobile phone or use other low wattage headphones from unknown origins.

The output is quite tolerant of abuse. It is protected against short circuit or excessive current draw, and allows wild an uncontrolled clipping and overdrive.
If the output meets unacceptable conditions, it will simply mute for 2 seconds. When it opens up again, it first tests the conditions before letting the audio through.
It never uses any kind of limiting to degrade the signal quality. If the sound is there, it is pure and clean.

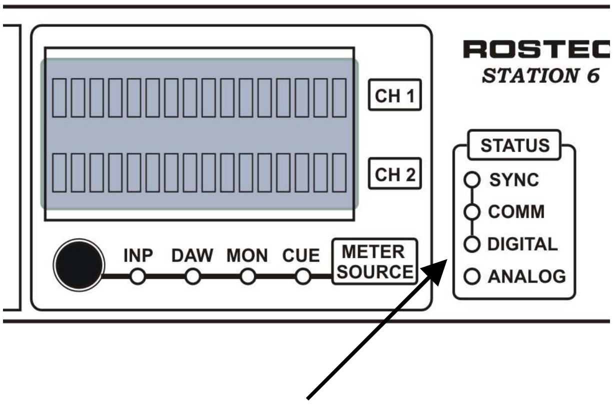
Meter

The meter is a 2x16 LED bar graph stereo peak meter with fast attack and slow release



The pushbutton selects the input source to the meter:
There are 4 stereo sources available:
1: INP. The output of the input channels mic1 and mic2. The signal is sourced at the insert send point.
2. DAW. The signal sent to the DAW. The signal is sourced at the insert return point. When the insert point is bypassed, the signals INP and DAW are identical. When the insert point is active, INP and DAW are the signals at the input and output of the external equipment.
3. MON. The signal sent to the monitor output, measured after the volume control.
3. CUE. The signal sent to the headphones output measured after the volume control.

Status indicators



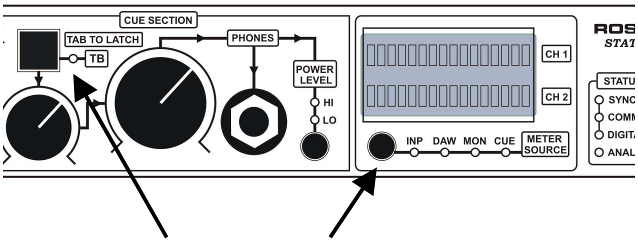
The 4 status LEDs show the internal status of the unit.

1. SYNC. The LED lights up when the digital module is locked and synchronized to an external word clock. If no digital module is installed, or if analog mode is selected, this indicator has no function.
2. COMM. The LED light up when a working handshake link is established between the digital module and the DAW. If no digital module is installed, or if analog mode is selected, this indicator has no function.
3. DIGITAL. The LED lights up when the unit is operating in digital mode.
4. ANALOG. The LED lights up when the unit is operating in analog mode

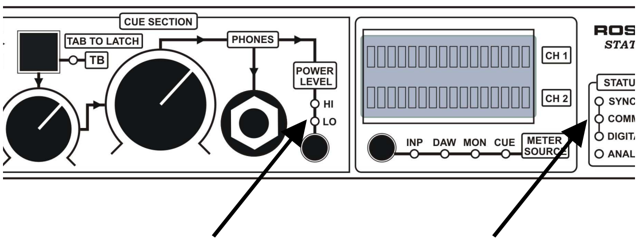
Monitor dim during talk-back

The dim level of the monitor output signal, during talk-back, can be programmed.
4 dim levels are available.0 dB, 6 dB, 12 dB or 18 dB.

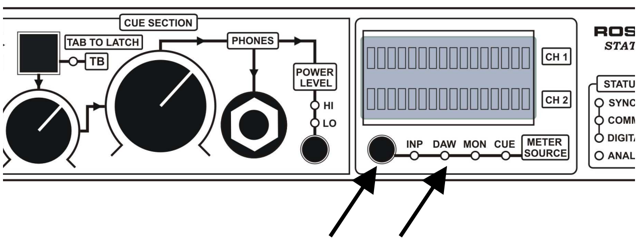
Here is how:



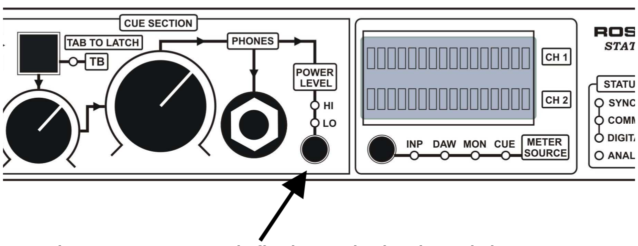
1. Push and hold the Talk-Back button and the Meter button simultaneously.



2. All these LEDs will light up to indicate that the unit is in programming mode: HI, LO, SYNC, COMM, DIGITAL and ANALOG.



3. The Meter source LEDs will now show the dim level. From left to right, 0 dB, 6 dB, 12 dB or 18 dB.
Push the meter switch repeatedly to select the dim level you want. You can freely push the TB button to test while listening to the monitor.

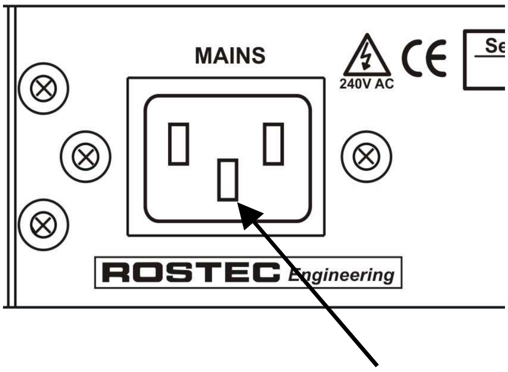


4. When you are satisfied, push the headphone power switch, and the unit returns to normal operation

Back panel

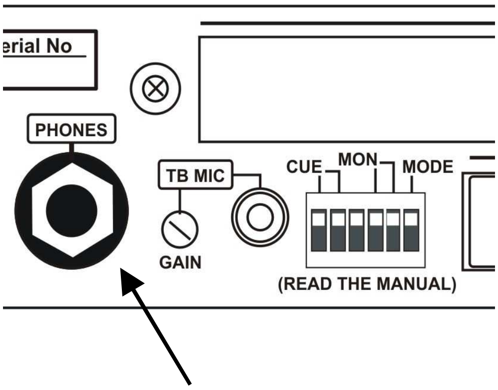
Mains input

The mains power inlet is a IEC 60320 receptacle for use with a standard IEC 60320 C14 connector.
The acceptable voltage is 190 - 264 Volt AC, 50-60 Hz.



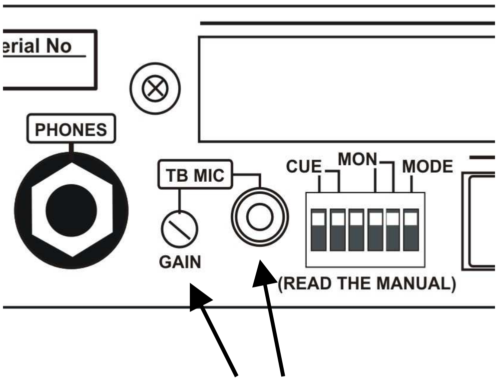
OBS: Always use the unit with ***safety earth***.

Headphone output



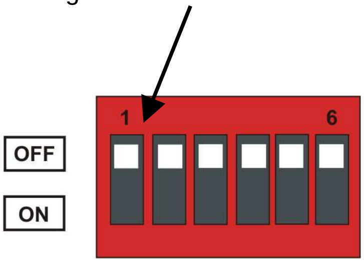
The headphone amplifier output is sent to the 1/4" Jack connector at the back panel and the 1/4" Jack connector at the front panel.
The output signals at the two headphone Jacks, front and back, are identical.

Talk Back input

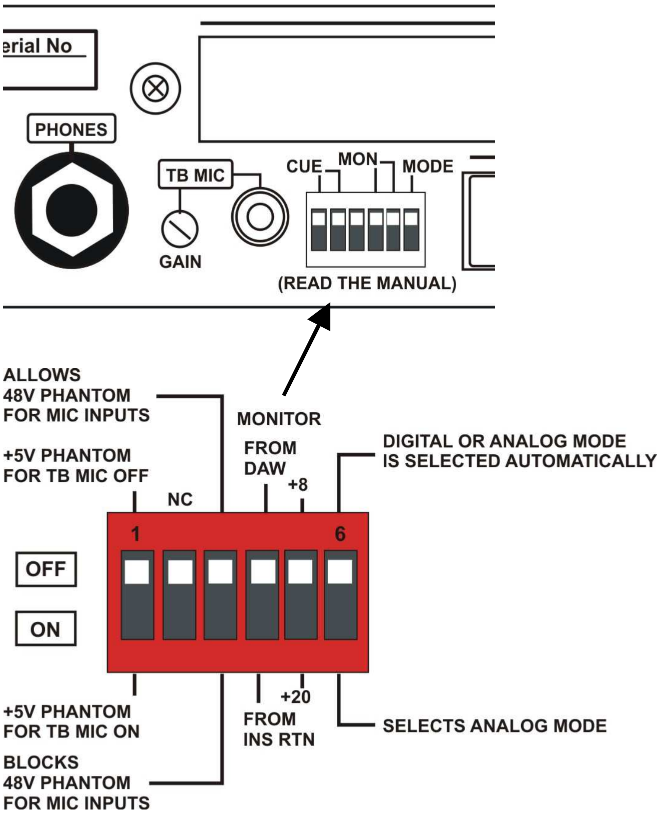


The talk back microphone input can be found at the 3.5 mm stereo mini-jack at the back panel. The input is balanced, and the input impedance is 1.2 kohm.
The input gain can be adjusted from +26 dB to +70 dB by the trimmer potentiometer.

The TB mic input is mainly intended for dynamic microphones, but a +5 V phantom voltage for electret or lavaliere condenser microphones can be applied by setting ***DIP switch 1***.



Function switch summary



Switch 1:
Applies +5V Phantom voltage to the Talk Back mic input.

OFF position: No +5 V phantom voltage is applied. This is for use of dynamic microphones.
ON position: +5 V phantom voltage is applied. This setting is for use of lavaliere condenser microphones.

Switch 2: No function (optional future use)

Switch 3:
Controls +48V phantom voltage blocking. Default is OFF

OFF position: The use of +48 V phantom voltage at the mic/line inputs is allowed.
ON position: The use of +48 V phantom voltage on the mic/line inputs is blocked.
If you push the +48 V button at front anyway, the +48V LED flashes briefly. No phantom voltage will be applied.

This is an essential safety feature to protect line equipment, connected to the mic/line input, from accidentally getting +48 V phantom voltage.

Switch 4
Controls monitor signal source. Default is OFF

OFF position: In digital mode, the signal to the monitor is sourced from the DAW,
In analog mode, the signal to the monitor is sourced from AUX line 3 and 4 inputs.
ON position: The signal to the monitor is always sourced from the Insert Return Point, regardless of whether the unit is in digital mode or in analog mode.

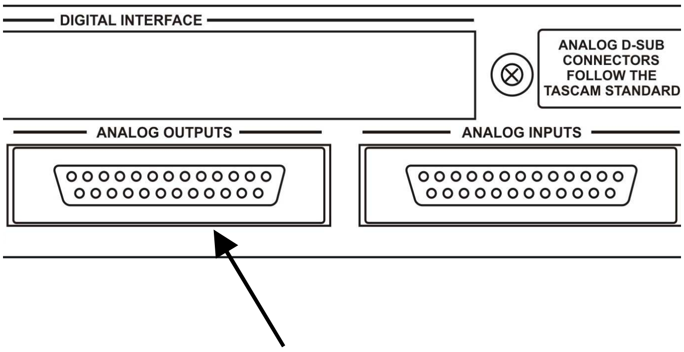
Switch 5
Controls monitor output level. Default is OFF.

OFF position: The signal level at the monitor output is +8 dBu at 0 dBFS (digital full scale)
ON position: The signal level at the monitor output is +20 dBu at 0 dBFS (digital full scale)

Switch 6
Controls digital or analog mode. Default is OFF.

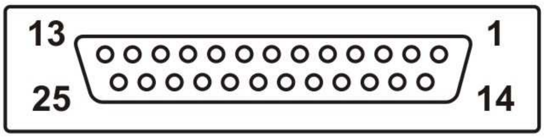
OFF position: Selects digital mode when a digital module is installed. If no digital module is installed, analog mode is automatically selected.
ON position: Selects analog mode. Any installed digital module is disabled and put into standby mode.

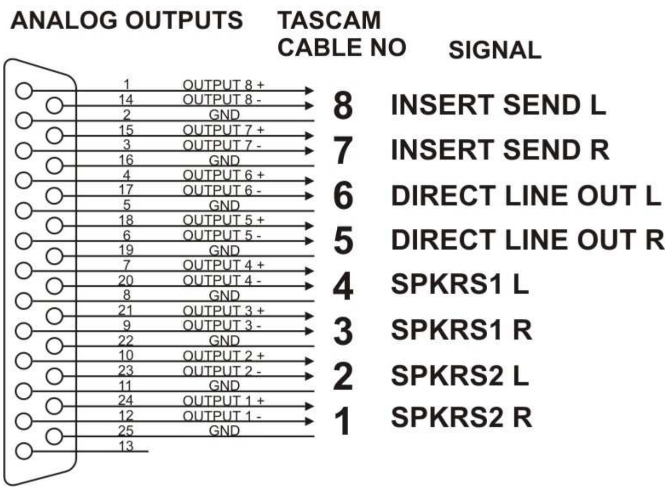
Analog outputs



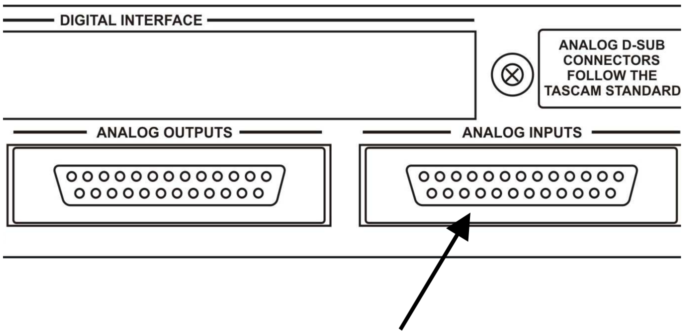
The analog outputs can be found at the back panel at the 25 pin female D-SUB connector.
The pin connections follow the Tascam Standard for analog signals.
The list of signals and their corresponding pin connections is listed below.

D-SUB standard pin-out seen from outside of the box





Analog inputs



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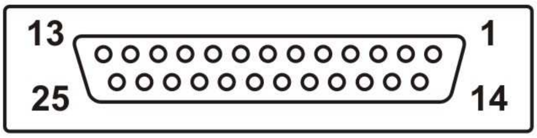
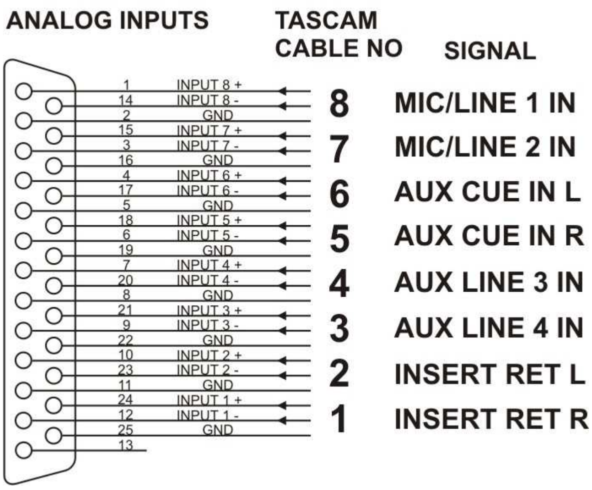
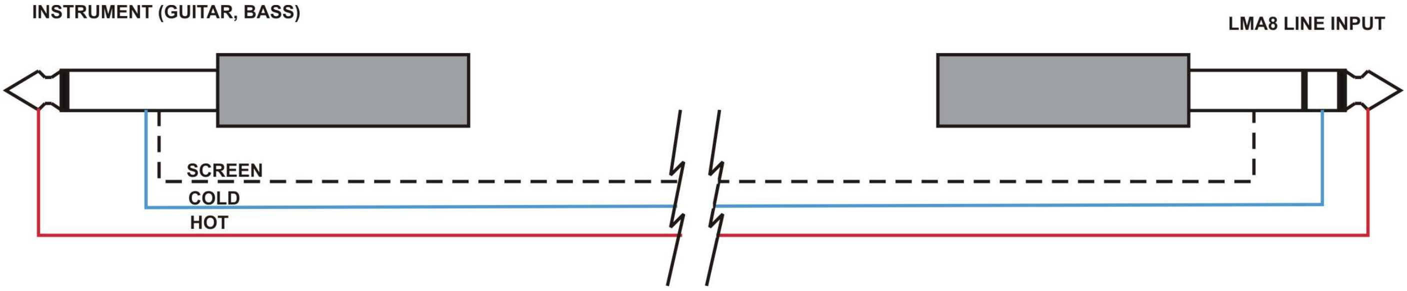
The list of signals and their corresponding pin connections is listed below.

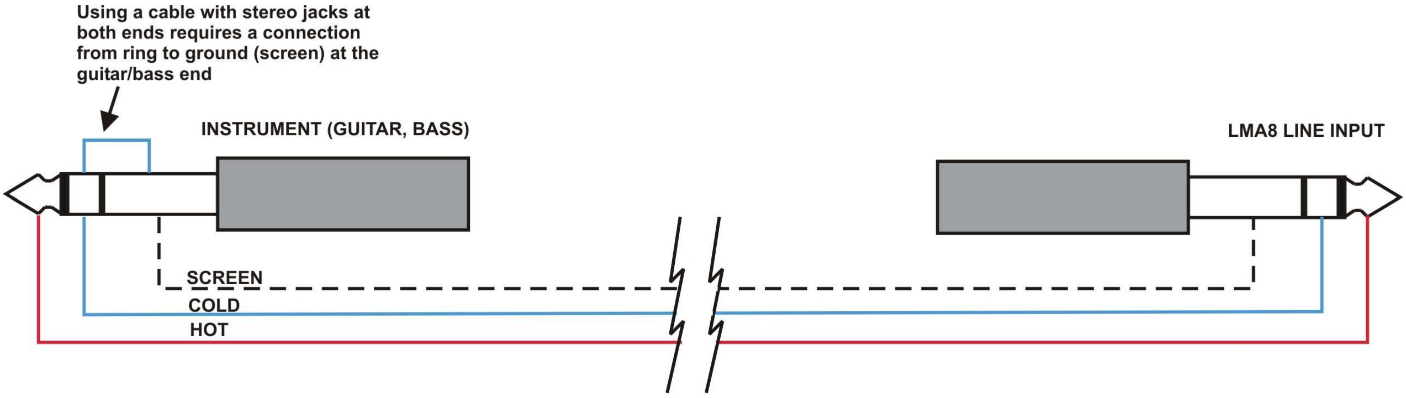
D-SUB standard pin-out seen from outside of the box





Balanced cabling for instrument input





The balanced instrument/line input opens up the unique possibility to use a balanced configuration with instruments like guitar, bass etc. The configuration will cancel out a large portion of the electrical noise induced in the cables. This is quite advantageous when using long cables on stage or in an electrically noisy environment near light dimmers, high power mains installations or nearby radio stations blasting megawatt into the air.
A guitar or bass hook-up is a high impedance system, and as such, it is very sensitive to electrical interference. Using a balanced cable instead of a single core cable, introduces one additional wire to "receive" the environmental noise. This additional noise signal is fed into the balanced input at the negative input terminal, canceling out the noise signal from the positive terminal, all without affecting the sound signal from the instrument.
The configuration will in most cases give a substantial noise reduction, depending on the electrical characteristics of the instrument pick-up. The negative and positive signal lines are terminated by different impedances at the instrument so a total noise cancellation will not take place. The noise reduction is typically in the range of -15 dB to -20 dB.
Note that the noise from the instrument pick-ups is not reduced. Only the noise induced into the cable is reduced.

Station 6 Preamplifier and Monitor Controller

Mechanical and electrical specifications

Dimensions with rack mounts: Width 19 inch, height 1U (44 mm), depth 210 mm
Dimension as desktop: Width 420 mm, height 42 mm, depth 210 mm
Weight: 5.0 kg
Power: 190 - 264 VAC, 50-60 Hz, 15 Watts,
ESD: Protected to 23 kV, IEC 61000-4-2 and 15 A surge, IEC 61000-4-5

Input preamp

Microphone input impedance: 6 kohm
Instrument line input impedance: 1 Mohm/60 pF
Mic input max input level: Balanced +30 dBu, unbalanced +24 dBu
Instrument input max input level: +30 dBu, unbalanced +24 dBu
Insert return max input level: Balanced +30 dBu, unbalanced +24 dBu
Direct output/insert send max output level: Balanced +28 dBu, unbalanced +22 dBu
Insert point nominal level in/out: +20 dBu at 0 dBFS
Preamp gain, input to balanced output: +10 dB to +70 dB
Input noise: -134 dBu (A weighted, 22 Hz - 22 kHz)
Frequency response: 5 Hz - 200 kHz, +/- 0.1 dB
Crosstalk: -120 dB, 20 Hz - 20 kHz, input terminated by 150 ohm

THD+N (classical analysis): 0.00035 % @ 1 kHz, 10 dB gain
THD+N (classical analysis): 0.00075 % @ 1 kHz 30 dB gain
THD (FFT analysis): 0.00013 % @1 kHz, 10 dB gain
THD (FFT analysis): 0.00015 % @1 kHz, 30 dB gain
THD (FFT analysis): 0.00118 % @1 kHz, 60 dB gain

Talk back preamp

Max input level: Balanced +16 dBu, unbalanced +10 dBu
Gain: +26 dB to +70 dB
Input noise: -128 dBu (A weighted, 22 Hz - 22 kHz)
Frequency response: 30 Hz - 20 kHz, @ - 3 dB

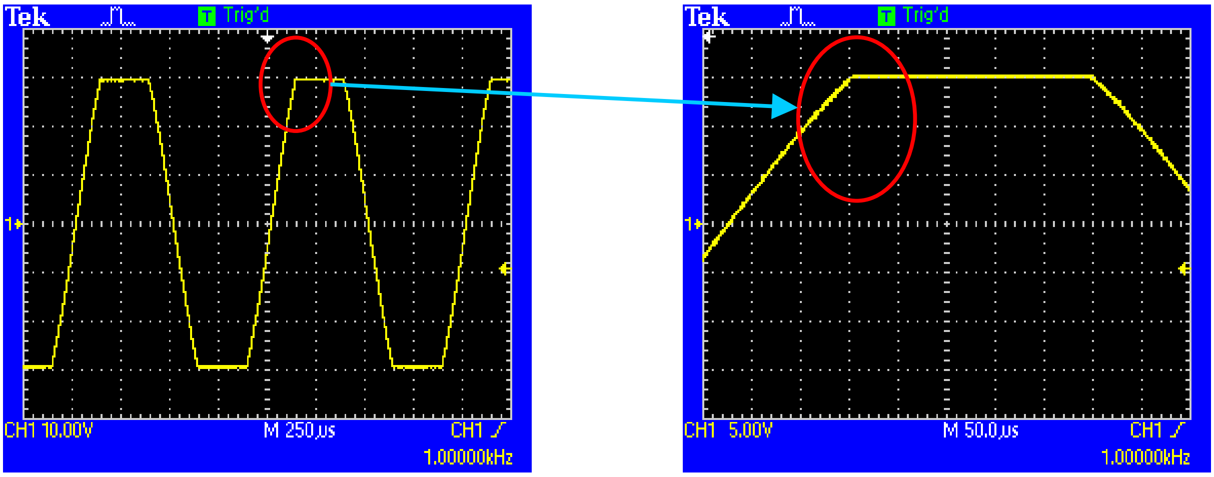
Monitor output

Max output level: Balanced +28 dBu, unbalanced +22 dBu
Nominal level: +8 dBu or +20 dBu at 0 dBFS (depending on ***DIP switch 5*** setting)
Frequency response: 5 Hz - 250 kHz, +/- 0.1 dB
Crosstalk: -120 dB, 20 Hz - 20 kHz
THD+N (classical analysis): 0.0003 % @ 1 kHz
THD (FFT analysis): 0.00008 % @1 kHz,

Headphones output

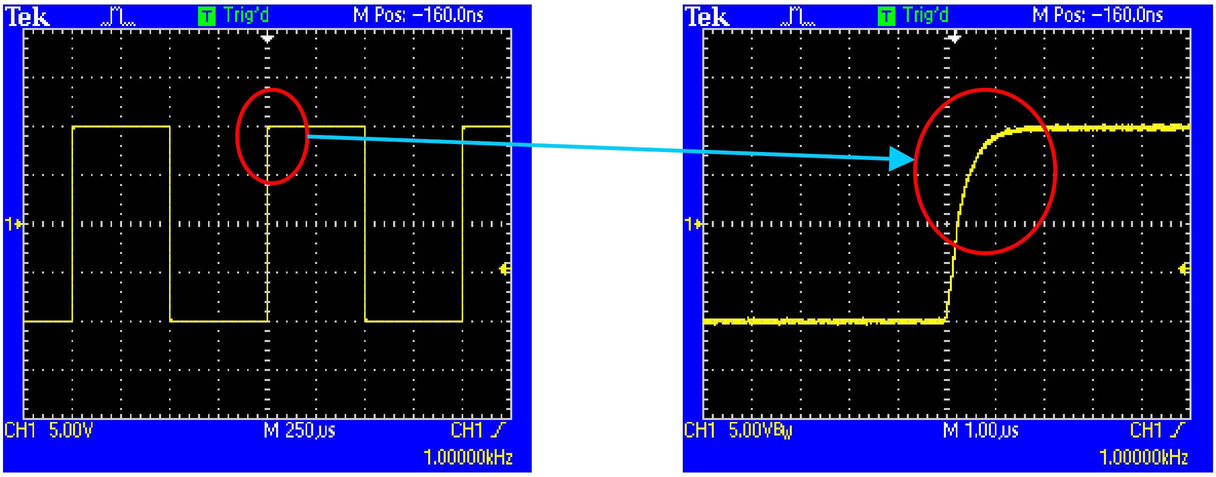
Max output level HI setting: 45 Volt PP
Max output level LO setting: 16 Volt PP
Max current output level; 1.2 Amp
Max power output 2 x 5 watts
Frequency response: 5 Hz - 100 kHz, +/- 0.1 dB
Crosstalk: -94 dB, 20 Hz - 20 kHz
THD+N (classical analysis): 0.0015 % @ 1 kHz
THD (FFT analysis): 0.0004 % @1 kHz,

Clipping characteristics, input preamp



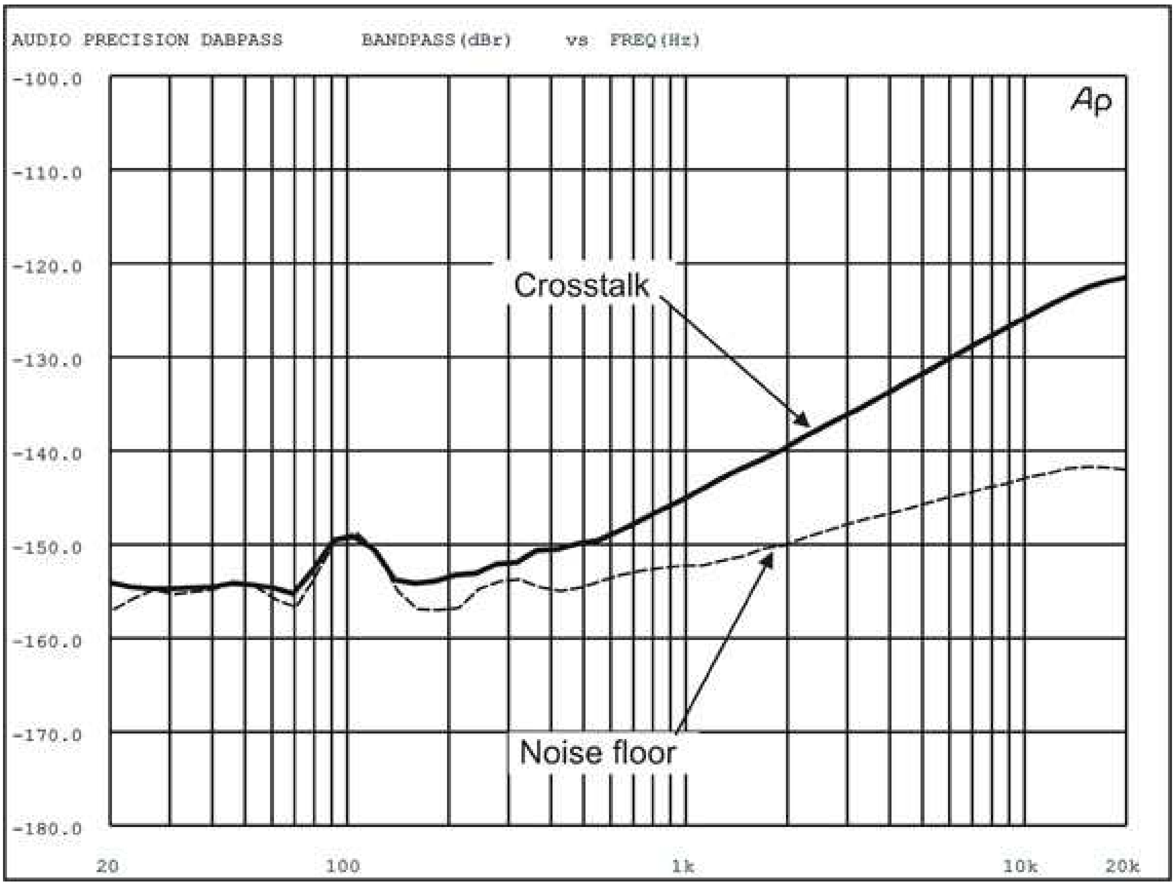
The preamp circuit shows excellent clipping characteristics. When the amp clips, there is no bouncing, ringing, pumping etc. There is just regular and plain clipping. A closer look at the leading edge of the clip shows that there is no overshoot and no recovery delay. There is only instant clipping and instant release.
Observe that the output level is +28 dBu, so most equipment connected to the amp have already gone into clipping at this level. The input clipping always occurs at a 10 dB higher level than the output clipping, thus the input clipping is always masked by the output clipping. *Input clipping cannot be transferred to the output.*

Impulse response, input preamp



The above snapshots show the ideal step impulse response of the preamp. When subjected to a steep transient (in this case a step impulse with 5 nsec rise time), there is no ringing and no overshoot. There is only total control. The amp does NOT produce any signal itself when subjected to transient material. It does not add anything. It stays true to the source!
A closer look at the leading edge of the step impulse shows that there is absolutely no ringing or overshoot. Observe that although the amplitude of the step impulse is +20 dBu, the circuit does NOT go into slew rate mode. The curve remains a true exponential.
The output voltage swings 20 Volts in less than 1 usec. Rise time (10/90 % of the amplitude) is approx 0.6 usec

Noise performance and crosstalk, input preamp



The lower graph shows the spectral density of the input noise by measuring with a sweeping 1/3 octave band pass filter across the audio range. The input was terminated by 150 ohms.
Observe that the noise spectrum exhibits the typical characteristics of white noise in the upper band.
In the lower band, the 1/f corner noise is dominant. A hum component from the power supply can be seen at 100 Hz at a level of approx. -148 dBu. The rms sum of the noise from 22 - 22 kHz is -134 dBu A weighted, or -131 dBu unweighted.

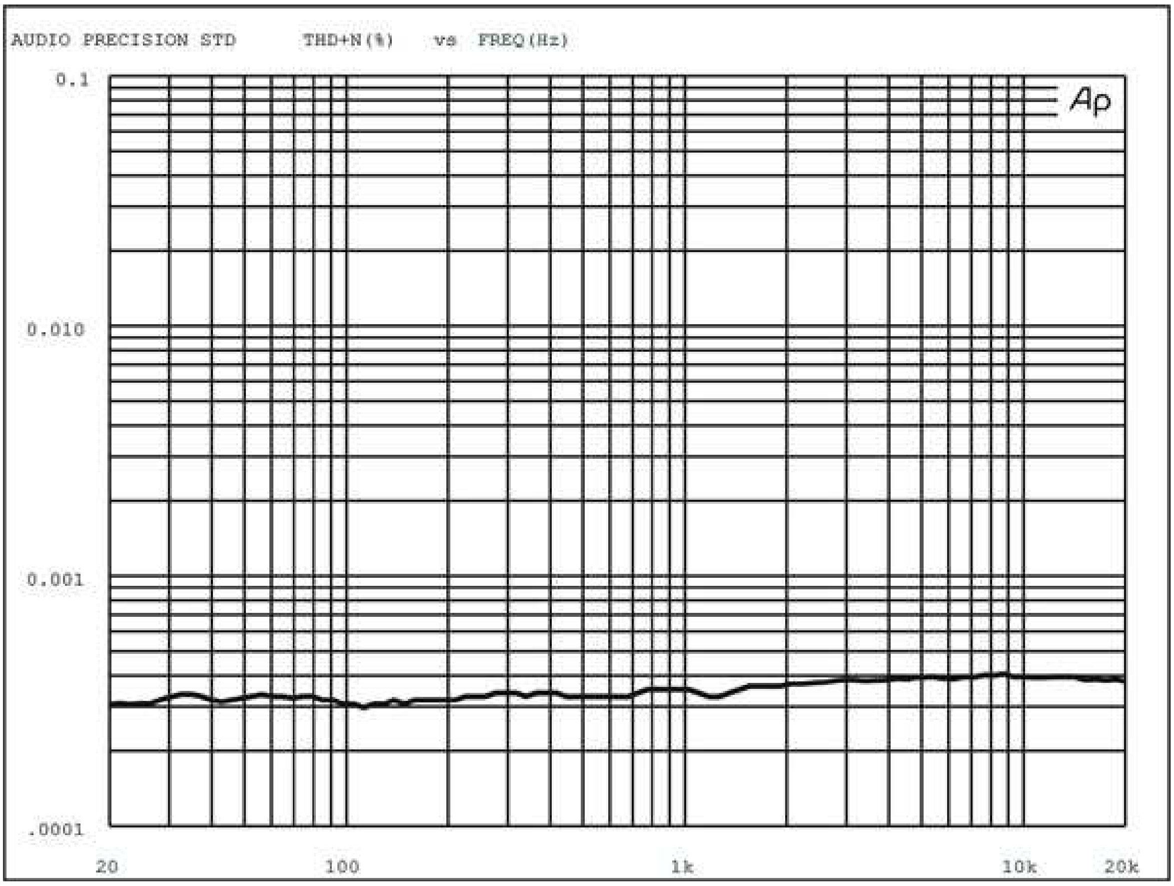
The upper graph shows crosstalk between adjacent channels. A +16 dBu signal was sent to channel 1, gain setting +10 dB. The output level of channel 1 was +26 dBu.
The input of channel 2 was terminated by 150 ohms, and the signal level was measured at the output of channel 2.

The graph shows virtually no crosstalk below 100 Hz, which indicates that there is no transfer of audio signals from signal dependent ground currents.
The crosstalk increases moderately at higher frequencies, but it is still below -120 dBu at 20 kHz. This indicates a small airborne capacitive signal transfer internally between components.

Channel 1 is blasting at +26 dBu and channel 2 sees only -120 dBu at the input. This clearly demonstrates the benefits of a <u>true balanced architecture.</u>

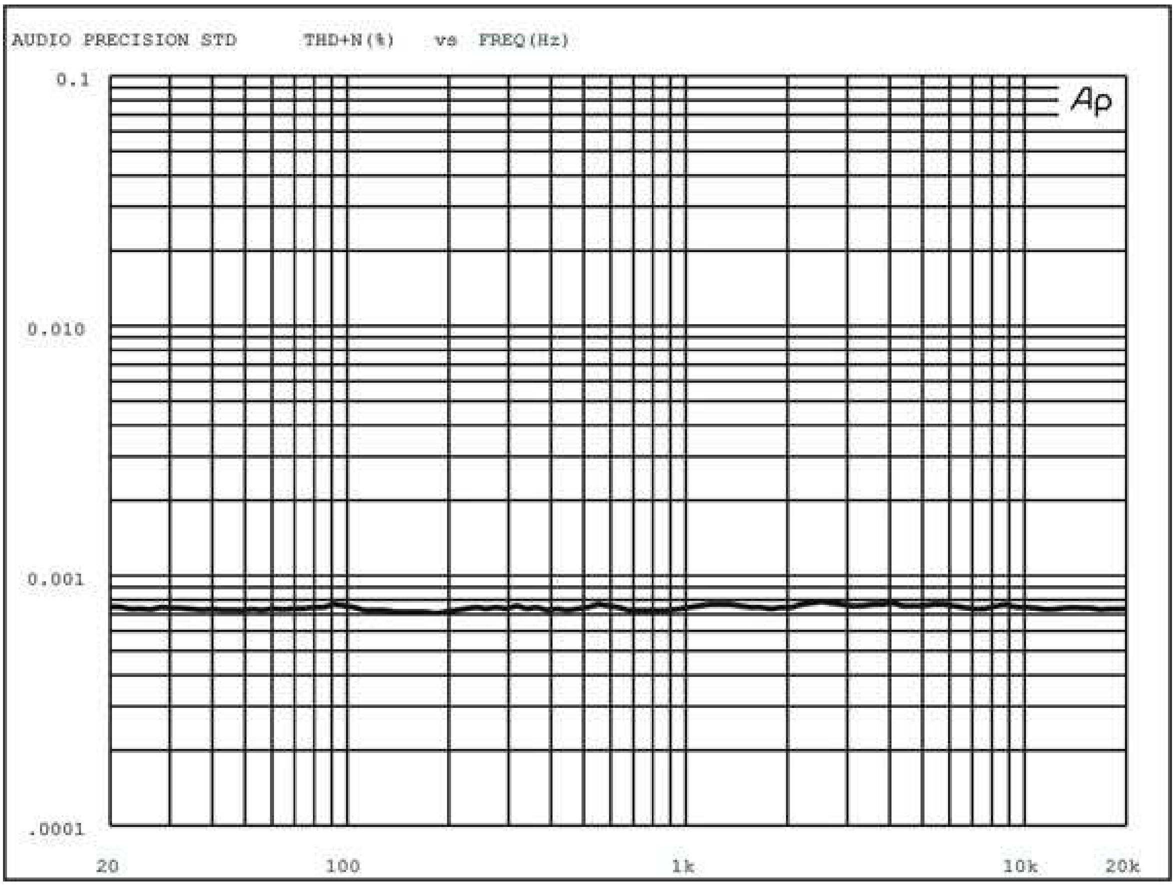
Classical Distortion Analysis

The standard analysis of distortion in audio circuits has traditionally been "Total Harmonic Distortion + Noise".
This method has only limited usefulness in this case, because the distortion of the LMA8 is so low, that it is masked by noise at higher gain settings.
The LMA8's input noise is among the lowest modern technology can offer, and the distortion of the unique input amplifier circuit is so low, that it can only be quantified by classical analysis at lower gain settings. At higher gain settings, a FFT analysis is required.



Input preamp, distortion+noise at +10 dB gain

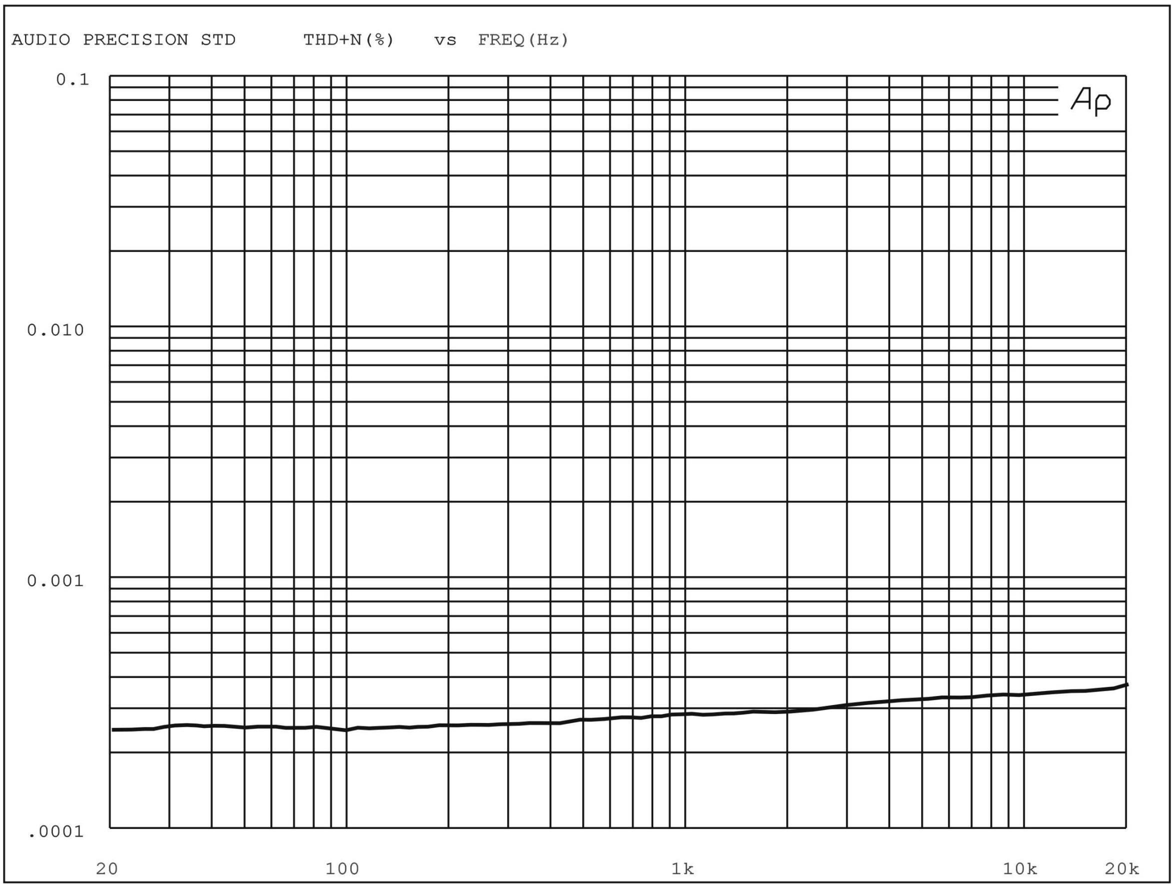
The THD+N is below 0.0004 % in the whole audio range. The measured THD+N result shows good linearity from 20 Hz to 20 kHz; only a slight increase above 2 kHz is seen.
The input noise contribution at +10 dB gain corresponds to a level of approx. 0.00007 %.
Thus it can be seen that the graph **mainly** represents the distortion of the circuit.



Input preamp, distortion+noise at +30 dB gain

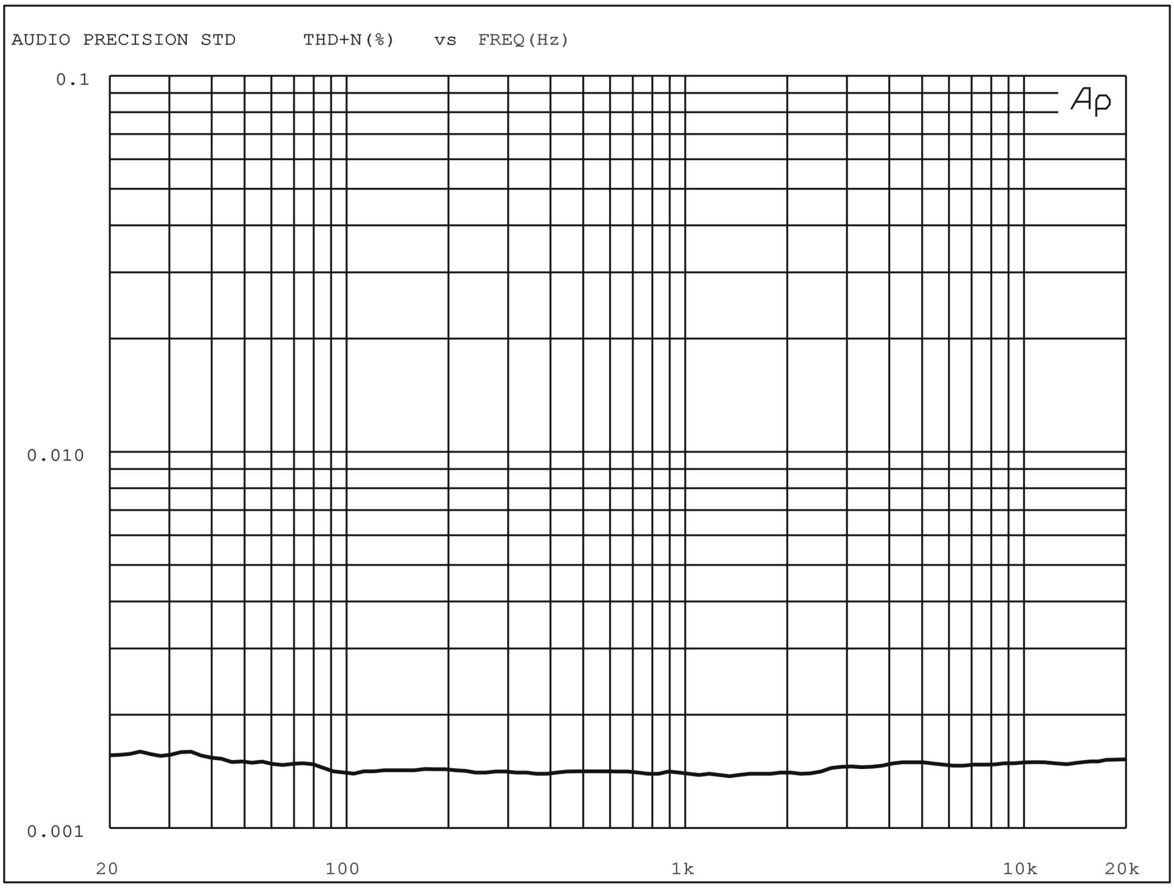
The THD+N is below 0.0008 % in the whole audio range. The measured THD+N result is very close to linear from 20 Hz - 20 kHz.
The input noise contribution at +30 dB gain corresponds to a level of approx. 0.0007 %.
The distortion of the circuit is still below 0.0004 %.
Thus it can be seen that the graph **mainly** represents the noise of the circuit.

Increasing the gain setting further has little effect on the distortion figure, but will only add noise to the measurement.



Monitor output, distortion+noise at 20 dBu output level

The THD+N is below 0.0004 % in the whole audio range. The measured THD+N result is very close to linear from 20 Hz - 20 kHz
The noise contribution of the circuit at +20 dBu output level can be ignored. The circuit noise corresponds to a level of approx. 0.00015 %.Thus it can be seen that the graph **mainly** represents the distortion of the circuit.



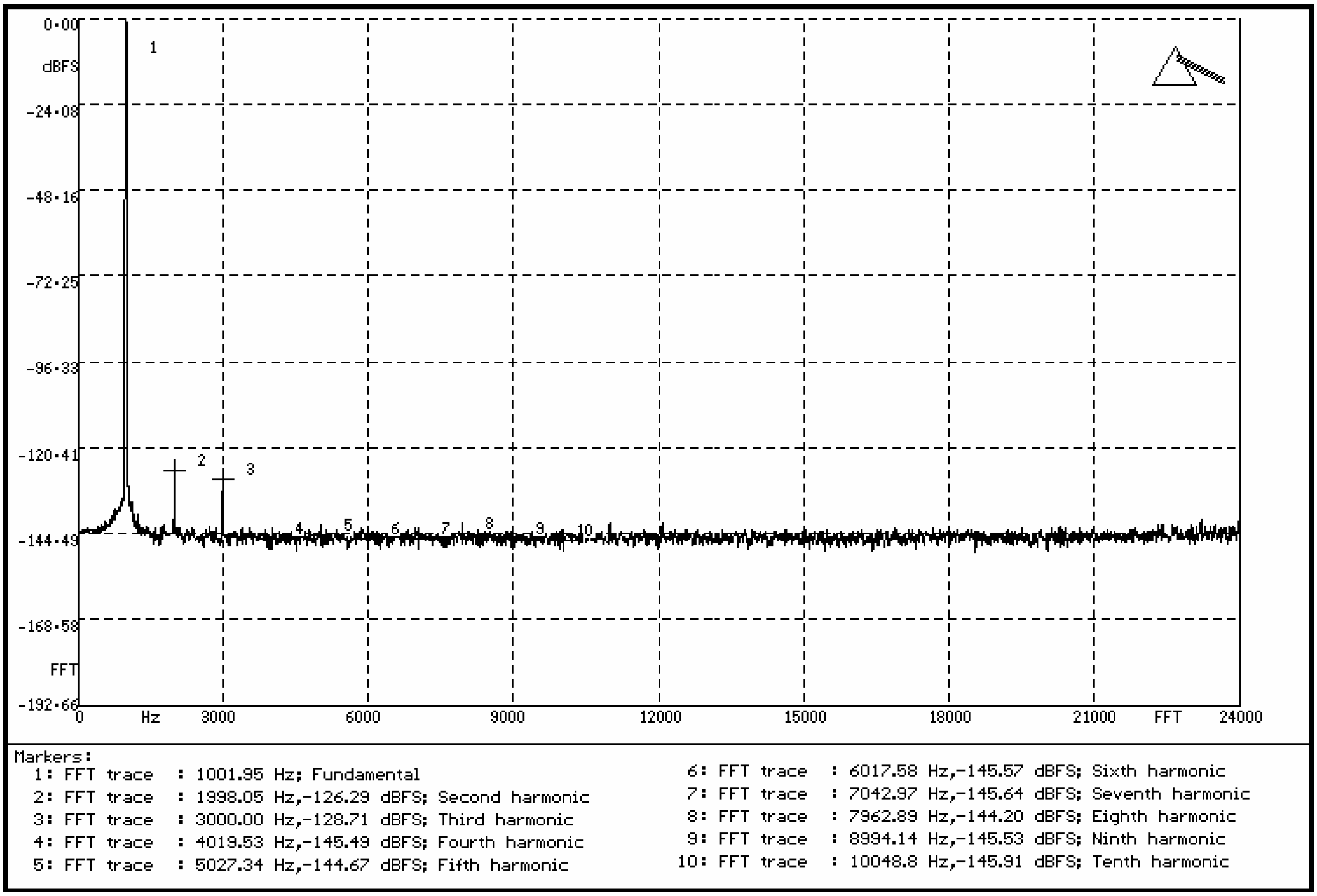
Headphone output, distortion+noise at 45 V PP output level into a 50 ohm load

The THD+N is below 0.0015 % in the whole audio range. The measured THD+N result is very close to linear from 20 Hz - 20 kHz.
The noise contribution of the circuit at 45 V PP output level can be ignored. The circuit noise corresponds to a level of approx. 0.0002 %. Thus it can be seen that the graph **mainly** represents the distortion of the circuit.

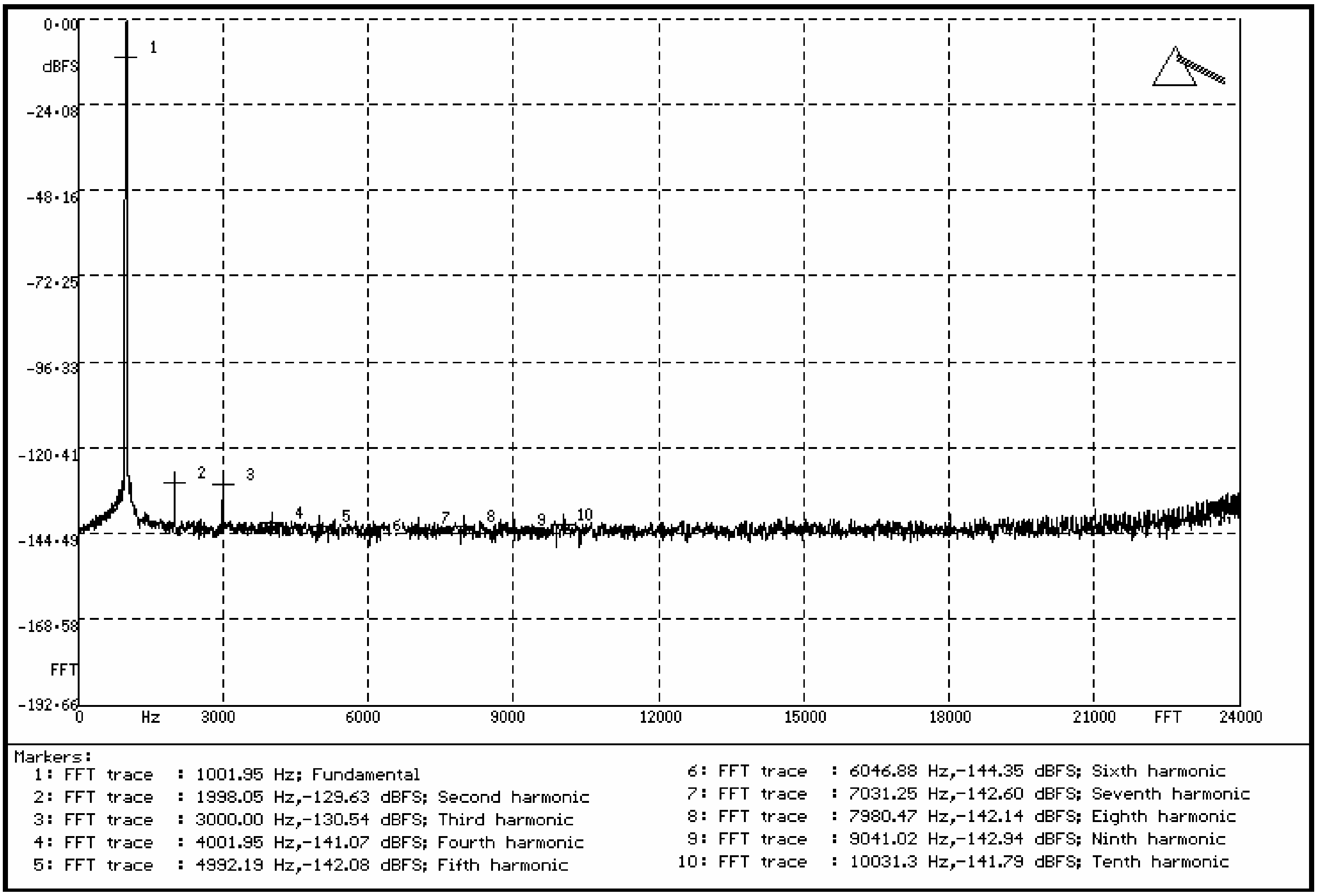
FFT distortion analysis

FFT analysis has the ability to detect distortion that is buried in the noise. Noise is a statistical phenomenon, and by reading several samples and summing the results, the FFT analysis reduces the noise to its average value, thus revealing the distortion components otherwise buried in the noise.

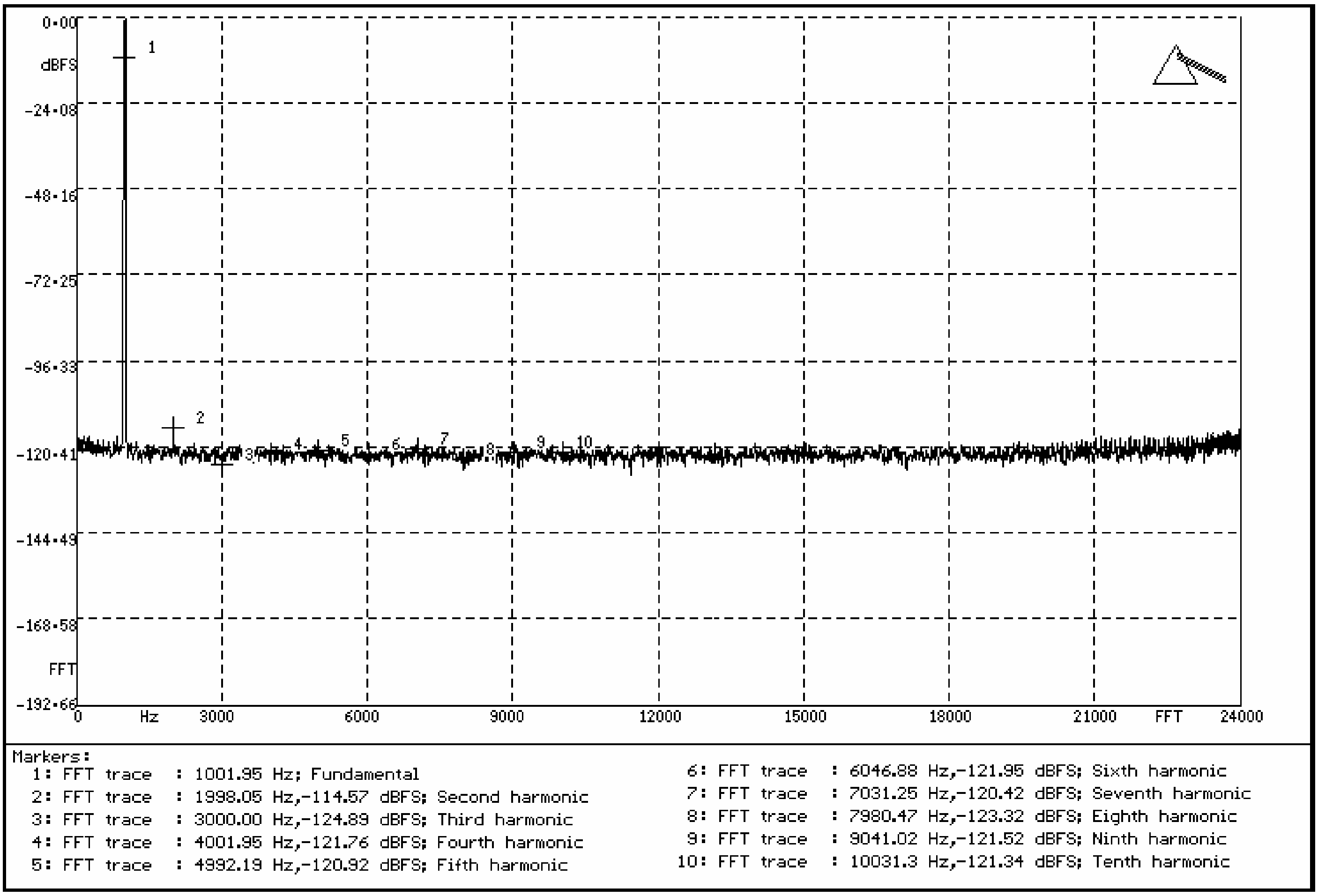
Below are plots of the distortion characteristics at various gain settings. Distortion levels are listed for 2nd to 10th harmonic and are quantified in dBFS. As a guideline, -120 dBFS equals 0.0001 %, -140 dBFS equals 0.00001 %



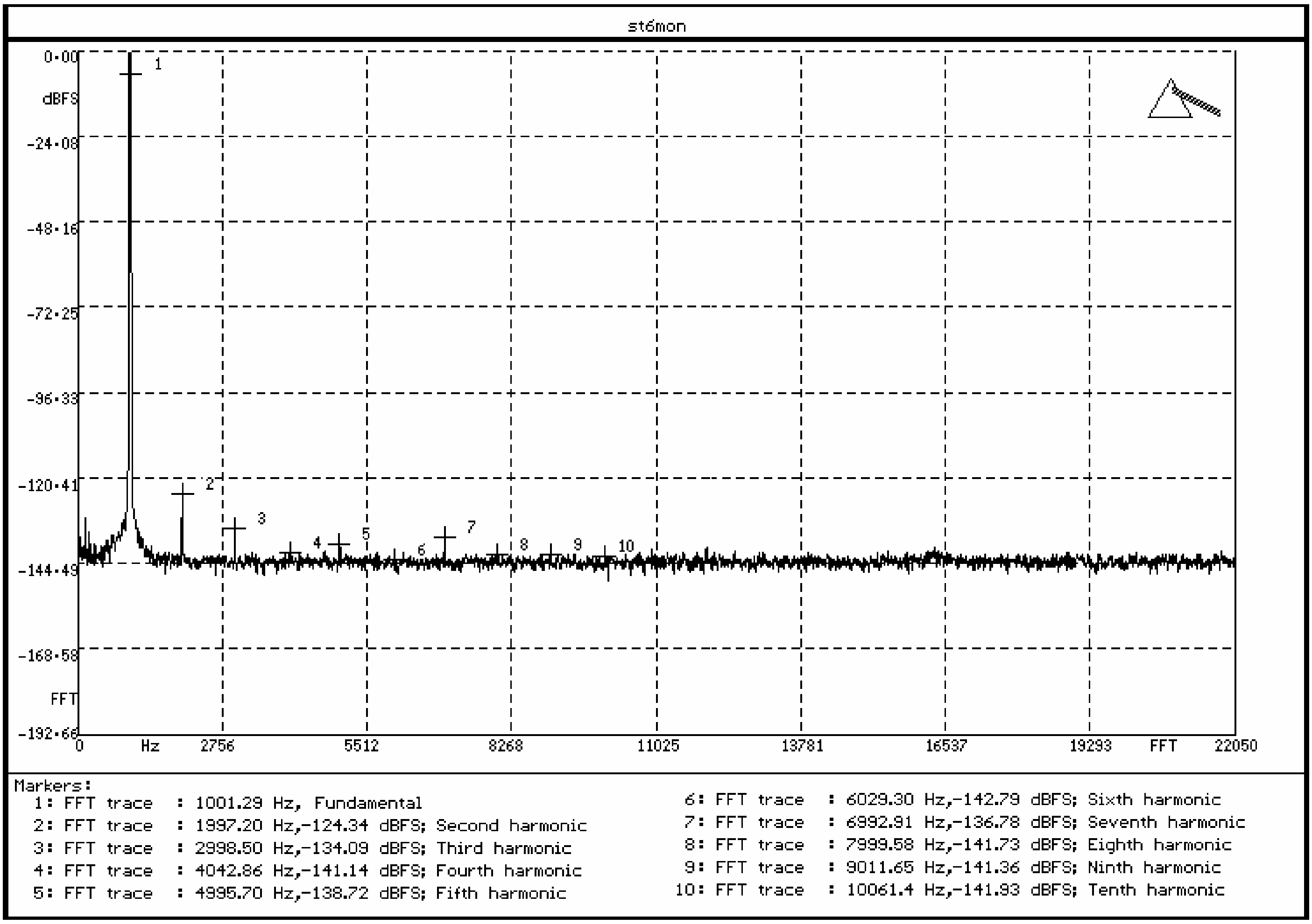
Input preamp, gain +10 dB, frequency 1 kHz. The RMS sum of all distortion components (THD) is 0.00013 %



Input preamp, gain +30 dB, frequency 1 kHz. The RMS sum of all distortion components (THD) is 0.00015 %



Input preamp, gain +60 dB, frequency 1 kHz. The RMS sum of all distortion components (THD) is 0.00118 %



Monitor output, output +20 dBu, frequency 1 kHz. The RMS sum of all distortion components (THD) is 0.00016 %