Learning Targets

I can identify the steps to take that will….

With podcasts and any kind of audio, the raw sound you record is rarely going to be the end product that people hear.

There are a number of post production effects, also known as FX, that can help you craft a more pleasing sound for your podcast. When mixing sound, there are four main tools that you should focus on. Volume, panning, EQ and compression.

Volume and panning are relatively easy to understand. It’s how loud the instruments are in relation to each other, and where they are placed in the stereo field. Sounds simple, but don’t underestimate the importance of these factors. These are the building blocks of any great mix.

Then there's equalization. Equalization, or EQ is one of the most basic types of processing that help to achieve a better and more professional sound for the vocal tracks in your podcast.This vital process is the main tool that we have as mixers that allows us to shape sounds to our liking. Whereas volume balancing allows us to control the overall level of an instrument or voice, EQ allows use to zoom in to a sound and adjust the volume of the individual frequencies.

With EQ alone you can remove nasty elements, exaggerate pleasing elements, make things sound different and create space in your mix.

The final essential tool, compression, which I'll cover in another tutorial.

https://flypaper.soundfly.com/produce/eqing-vocals-whats-happening-in-each-frequency-range-in-the-human-voice/

Every audible sound sits within the frequency range of human hearing.

A high pitched sound, like a drum cymbal or hissing sound, resides in the top end of the spectrum. A low pitched sound, such as a bass guitar or kick drum, resides in the bottom end of the spectrum.

Every instrument has a fundamental frequency but also has higher overtones and harmonics that give it’s character of sound. An organ sounds different to a bass guitar because of the different overtones and harmonics.

With equalization, you can adjust the character and tone of a sound by boosting or cutting these different frequencies.

Frequency Spectrum

Sub bass

Bass

Midrange

High Mids

High freqs

https://music.tutsplus.com/tutorials/eq-for-beginners-part-1-what-you-need-to-know--cms-25827

Here are some examples of different frequency ranges on an electric guitar.

Each example included a narrow boost of just over 10dB for some of the frequency ranges in the table above.

I used a drastic boost to make the differences obvious.

- Clean (no EQ)
- Warmth
- Mud
- Honk
- Tinny
- Definition

It’s important to remember that you can’t completely change a sound with equalization. You can’t create new frequencies. You can only remove or exaggerate what’s already there.

Frequency Ranges

https://www.youtube.com/watch?v=lVTfsbtj8IU

https://podigy.co/podcasters-eq/

https://www.wonderopolis.org/wonder/why-does-my-voice-sound-different-on-a-recording#:~:text=When%20you%20speak%20and%20hear,you%20than%20it%20really%20is.

https://www.youtube.com/watch?v=FUlEd98BaHU

https://www.youtube.com/watch?v=13viGJU8uj8&list=RDFUlEd98BaHU&index=3

XENYX Mic Preamp

Features:

◊ 130 dB dynamic range for an incredible amount of headroom

◊ A bandwidth ranging from below 10 Hz to over 200 kHz for crystal-clear reproduction of even the finest nuances

◊ The extremely low-noise and distortion-free circuitry guarantees absolutely natural and transparent signal reproduction

◊ They are perfectly matched to every conceivable microphone with up to 60 dB gain and +48 volt phantom power supply

◊ They enable you to use the greatly extended dynamic range of your 24-bit/192-kHz HD recorder to the full, thereby maintaining optimal audio quality

Extreme columns may damage your hearing and/or your headphones or loudspeakers.

Turn the MAIN MIX faders and phones control in the main section fully down before you switch [illegible] the unit. Always be careful to set the appr[illegible] volume.

General Mixing Console Functions

A mixing console fulfils three main functions:

- Signal processing
- Signal distribution
- Mix

General Mixing Console Functions
Signal Processing

- Preamplification
- Level-setting
- Frequency response correction
- Effects mixing

Preamplification

Frequency Response Connection

Level-setting

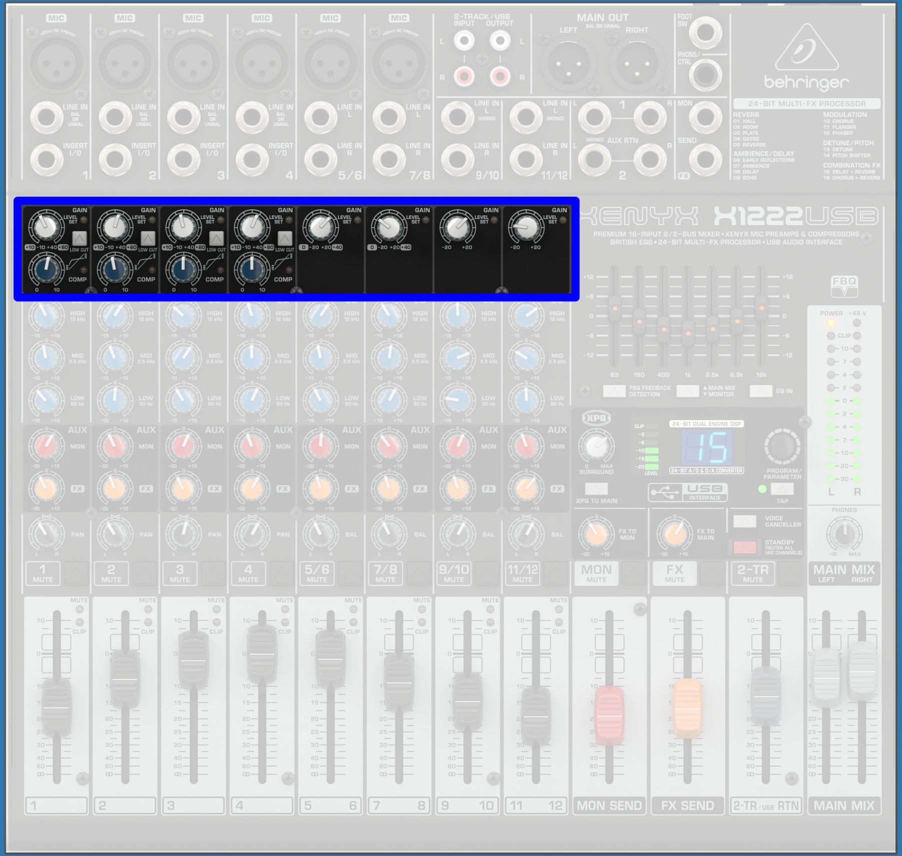
General Mixing Console Functions
Signal Processing

Preamplification
Microphones convert sound waves into voltage that has to be amplified several-fold; then, this voltage is turned into sound that is reproduced in a loudspeaker.

Because microphone capsules are very delicate in their construction, output voltage is very low and therefore susceptible to interference.

Therefore, mic signal voltage is amplified directly at the mixer input to a higher signal level that is less prone to interference. This higher, interference-safe signal level has to be achieved through amplification using an amplifier of the highest quality in order to amplify the signal and add as little noise to it as possible.

The XENYX Mic Preamp performs this role beautifully, leaving no traces of noise or sound coloration. Interference that could take place at the preamplification level could affect signal quality and purity, and would then be passed on to all other devices, resulting in inaccurate sounding program during recording or playback.

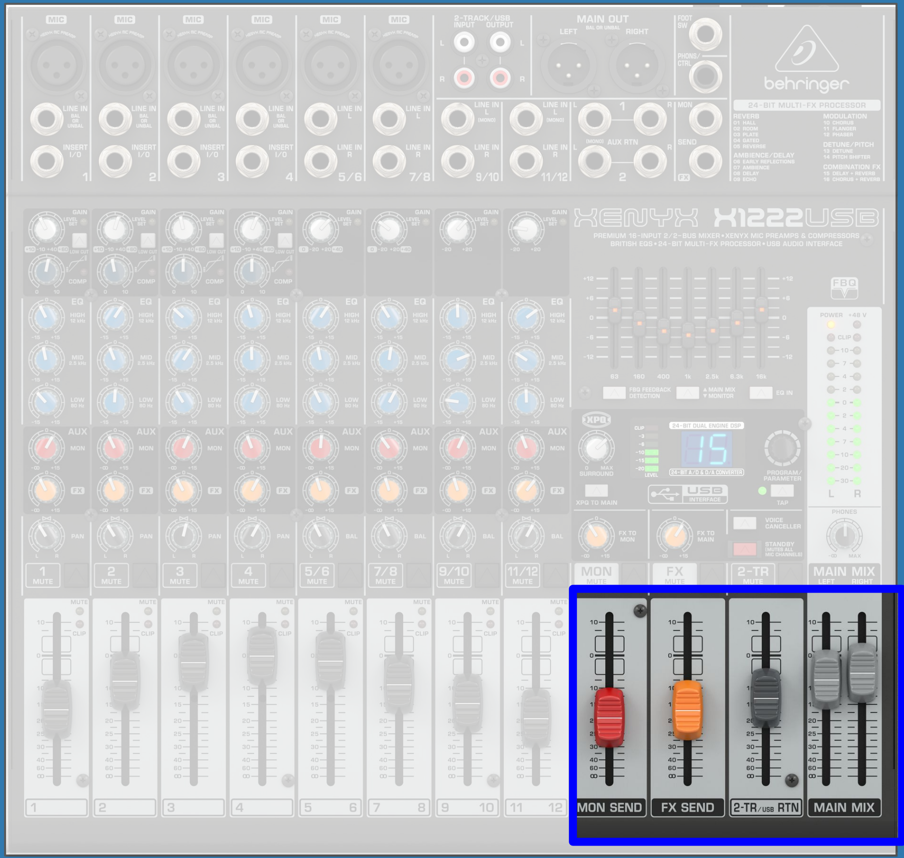


General Mixing Console Functions

Signal Processing

Level-setting

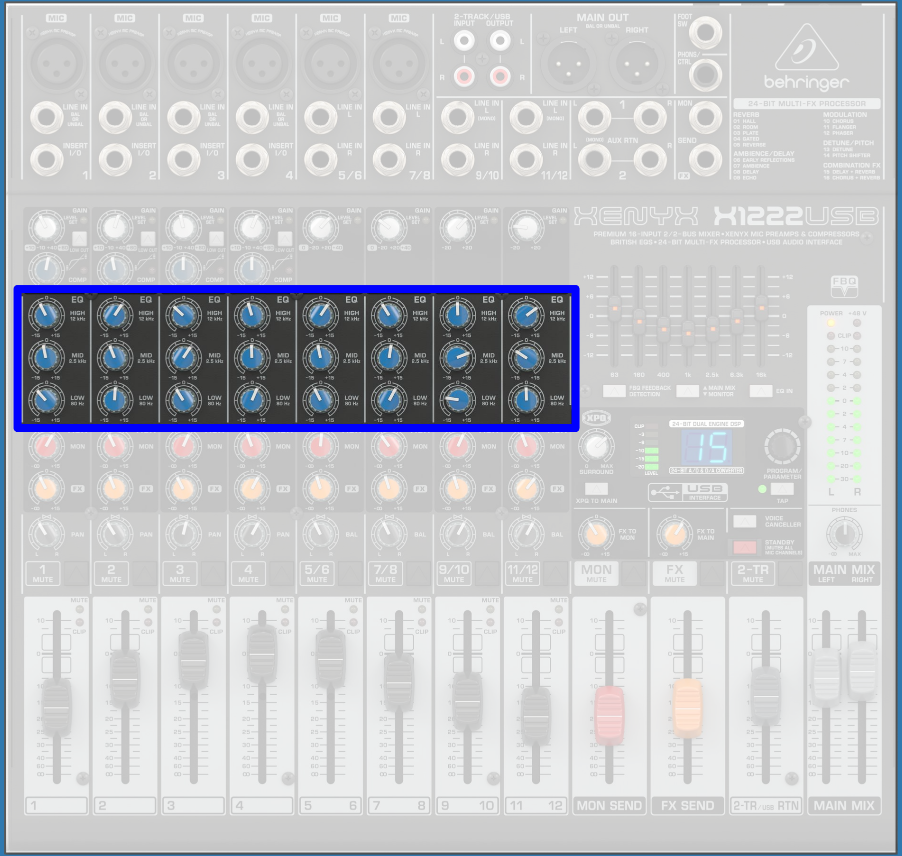
Signals fed into the mixer using a DI-box (Direct Injection) or the output of a sound card or a keyboard, often have to be adjusted to the operating level of your mixing console.



General Mixing Console Functions

Signal Processing

Frequency response correction
Using the equalizers found in each channel strip, you can simply, quickly and effectively adjust the way a signal sounds.



General Mixing Console Functions
Signal Processing

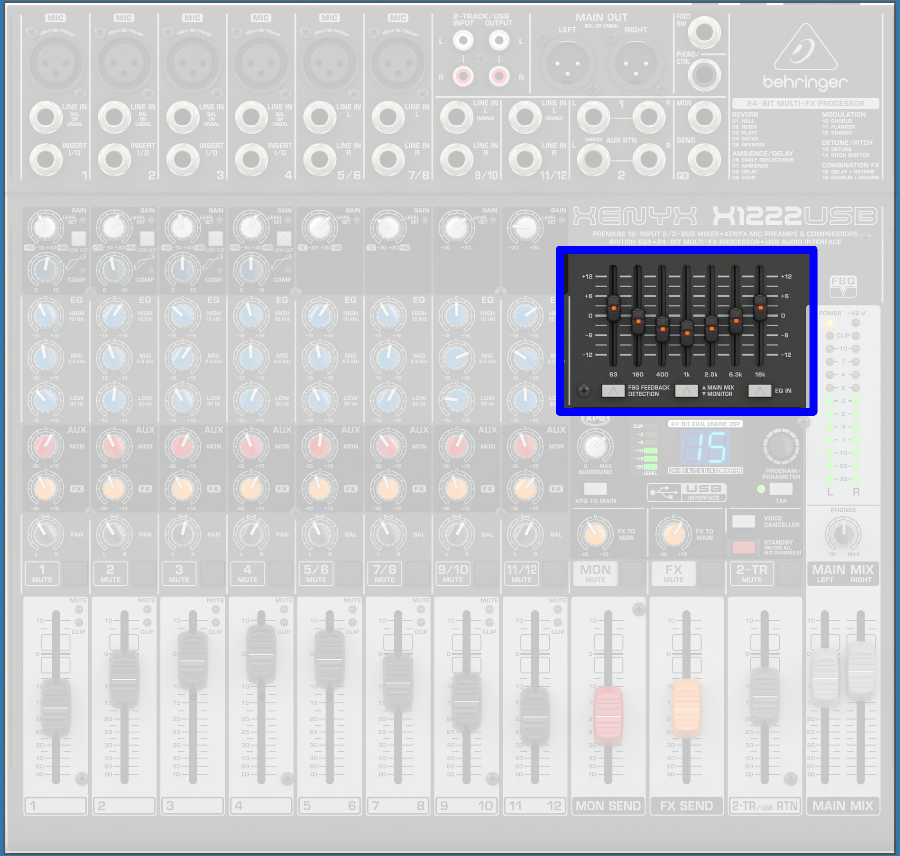
Effects mixing

In addition to the effects processor contained in your mixer, using the insert connectors on the mono channels and both aux busses lets you insert additional signal processors into your signal path.

General Mixing Console Functions

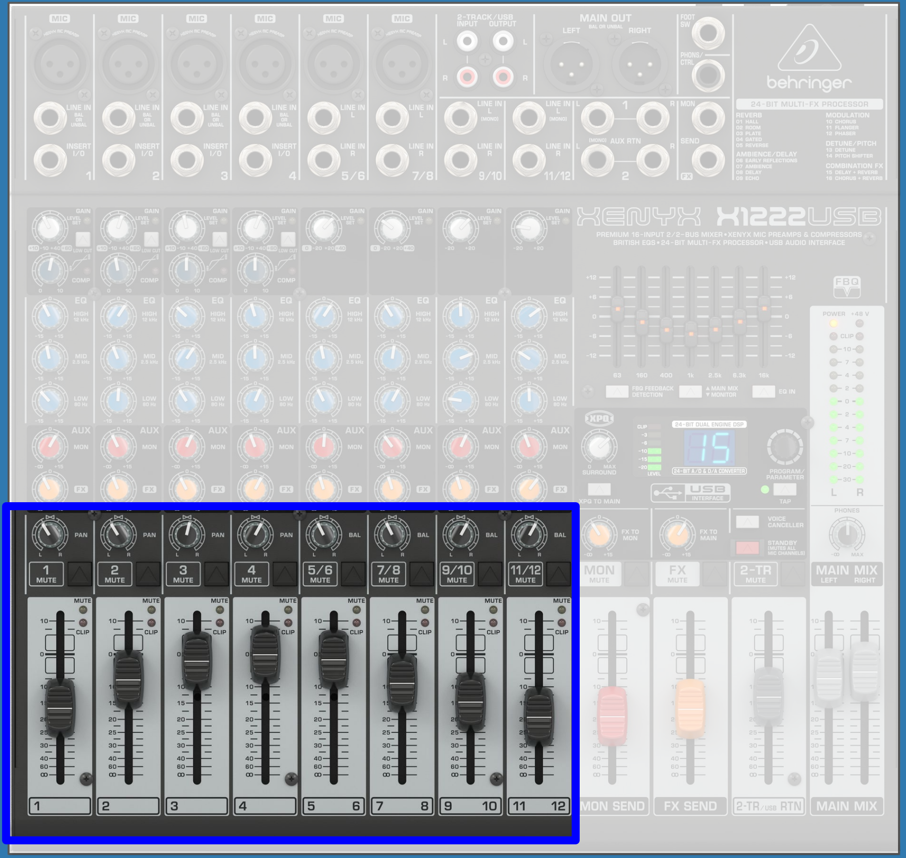
Signal Distribution

Individual signals adjusted at each channel strip are laid out at the aux sends and returns, and are either fed into external effects processors or fed back to the internal effects processor. Then, the signals are brought back into the main mix either via the aux return connectors or via direct internal wiring. The mix for the on-stage musicians is also created using the aux sends (monitor mix). Similarly, for example, signals for recording equipment, power amplifiers, headphones and 2-track outputs can also be taken.



General Mixing Console Functions
Mix

All other mixing console functions fall under this vital category. Creating a mix means primarily adjusting the volume levels of individual instruments and voices to one another as well as giving them the appropriate weight within the overall frequency spectrum. Likewise, you'll have to sensibly spread individual voices across the stereo image of a signal. At the end of this process, adjusting the level of the entire mix to other equipment in the signal path is required (e. g. recorder/crossover/amplifier). The control surface of BEHRINGER mixing consoles is optimized in such a way that these functions become easy to fulfill while the signal path remains simple to follow.



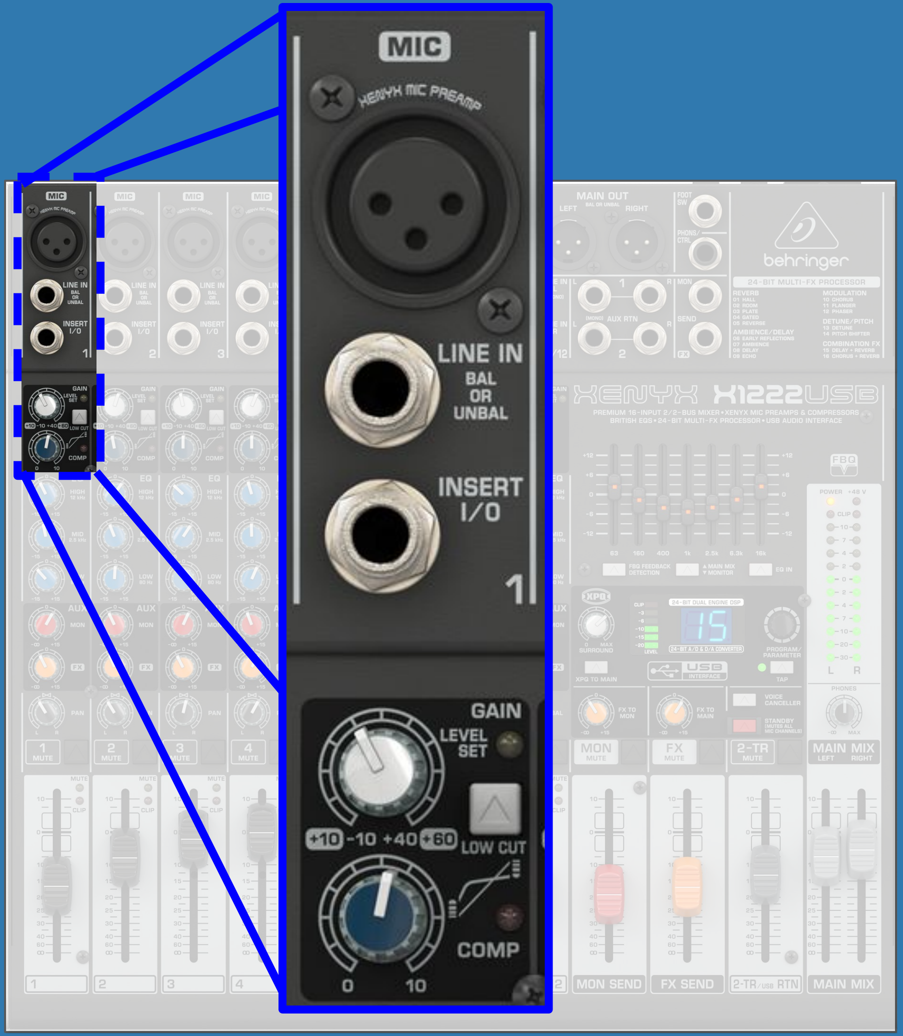
Control Elements and Connectors

- Mono channels
- Stereo channels
- Connector panel and main section

Mono Channels

Microphones and line inputs

- Mic
- Line In
- Insert
- Low cut
- Gain
- Level Set
- Compressor



Mono Channels

Microphones and line inputs

Mic

Each mono input channel has a balanced microphone input via the XLR connector and also features a switchable +48V phantom power supply for condenser microphones.

Mute your playback system before you switch on phantom power supply to prevent potentially damaging thumps being directed to your loudspeakers. (Chapter 2.5 “voltage supply, phantom power and fuse”

???? How to do this?

Mono Channels

Microphones and line inputs

Line In

Each mono input also has a balanced line input on a ¼' jack connector.

Unbalanced devices (such as mono jacks) can also be connected to these inputs.

You can use either the microphone input or the line input of a channel, but not both at the same time!

Mono Channels

Microphones and line inputs

Insert

Insert points enable the processing of a signal with dynamic processors or equalizers. They are sourced pre-fader, pre-EQ and pre-aux send. Unlinke reverb or other effects devices, whose signals are usually added to the dry signal, dynamic processors are most effective on the complete signal. In this case, aux send paths are a less-than-perfect solution. It is better to interrupt the signal path and insert a dynamic processor and/or equalizer. After processing, the signal is routed back to the console ar precisely the same point it left. However, the channel signal path is interrupted only if a phus is inserted into the corresponding jack (stereo phone plug: tip = signal output; ring = return input). All mono input channels are equipped with inserts.

Mono Channels

Microphones and line inputs

Low cut

The mono channels of the mixing consoles have a high slope LOW CUT filter for eliminating unwanted, low-frequency signal components (80 Hz, 18bD/octave).

Mono Channels

Microphones and line inputs

Use the GAIN control to adjust the input gain. This control should always be turned fully counter-clockwise whenever you connect or disconnect a signal source to one of the inputs.
The scale has 2 different value ranges: the first value range (+10 to +60 dB) refers to the MIC input and shows the amplification for the signals fed in there.
The second value range (+10 to -40 dB) refers to the line input and shows its sensitivity.
The settings for equipment with standard line-level signals (-10 dBV or +4 dBu) look like this: While the GAIN control is turned all the way down, connect your equipment. Set the GAIN control to the external devices' standard output level. If that unit has an output signal level display, it should show 0 dB during signal peaks. For +4 dBu, turn up GAIN slightly, for -10 dBV a bit more. Tweaking is done using the LEVEL SET LED

Mono Channels

Microphones and line inputs

Level Set

This LED lights up when the optimum operating signal level is achieved. During normal use, this LED should only light up during single peaks.

Mono Channels

Microphones and line inputs

COMPRESSOR

Each mono channel features a built-in compressor which lowers the dynamic range of the signal and increases its perceived loudness. The loud peaks are squashed down and the quiet sections are boosted. Turn the COMP knob clockwise to add more compression effect. The adjacent LED will light when the effect is engaged.

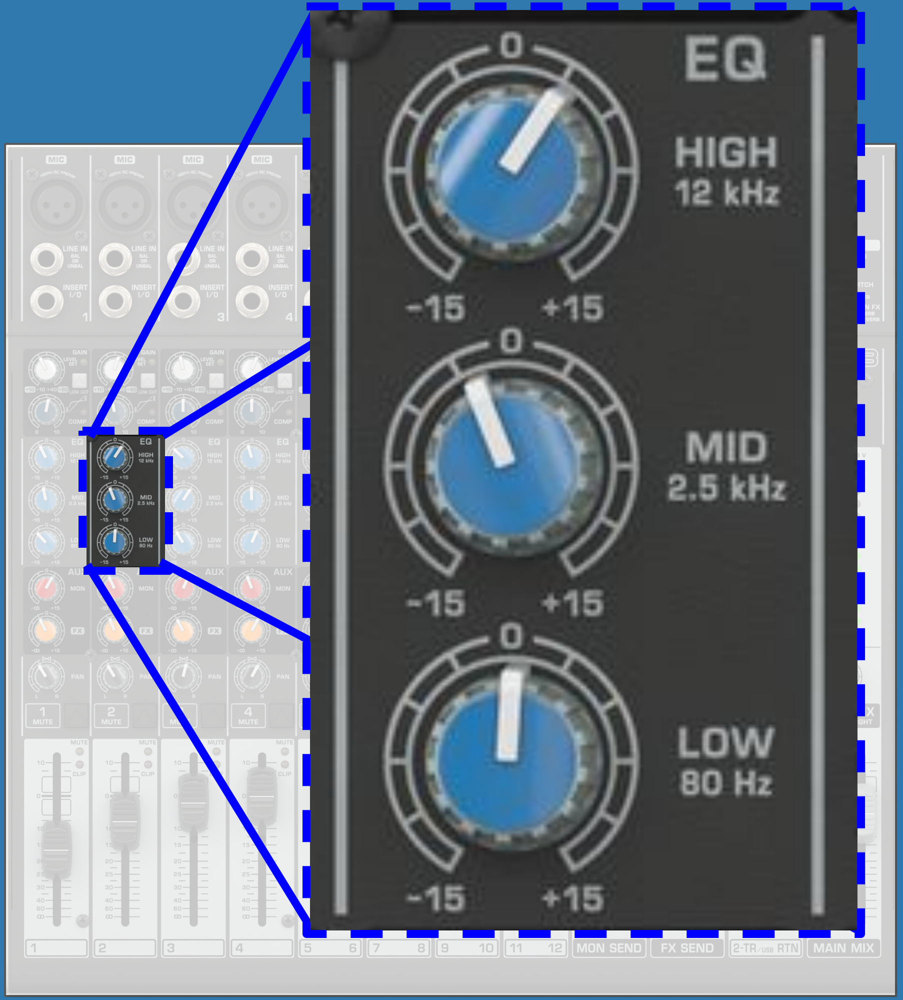
Mono Channels

Equalizer

All mono input channels have a 3-band equalizer.

All bands provide boost or cut of up to 15 dB.

In the central position,the equalizer is inactive (off).



Mono Channels

Equalizer

The circuitry of the British EQs is based on the technology used in the best-known top-of-the-line consoles and providing a warm sound without any unwanted side effects. The result are extremely musical equalizers which, unlike simple equalizers, cause no side effects such as phase shifting or bandwidth limitation, even with extreme gain settings of ±15 dB.

Mono Channels

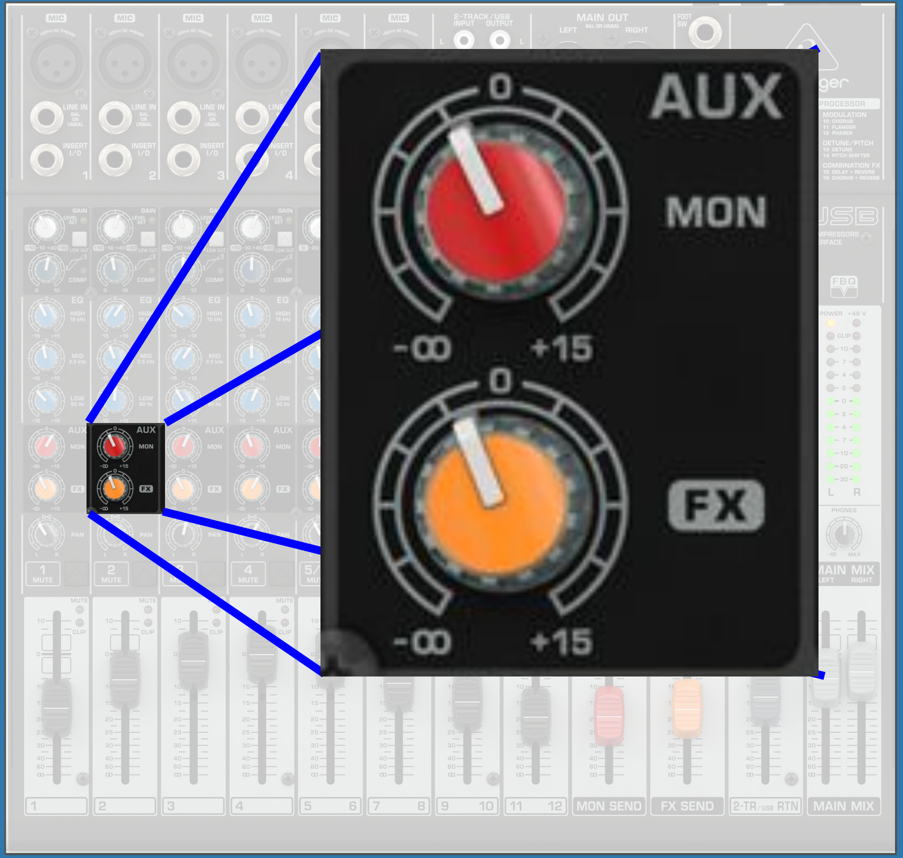
Equalizer

The upper (HIGH) and the lower (LOW) bands are shelving filters that increase or decrease all frequencies above or below their cut-off frequency. The cut-off frequencies of the upper and lower bands are 12 kHz and 80 Hz respectively. For the mid range, the console features a semi-parametric equalizer with a filter quality (Q) of 1 octave, tunable from 100 Hz to 8 kHz. Use the MID control to set the amount of boost or cut, and the FREQ control to determine the central frequency.

Mono Channels

Aux sends (MON and FX)

Aux sends thake signals via a control from one or more channels and sum these signals to a so-called bus. This bus signal is sent to an aux send connector and then routed, for example, to an active monitor speaker or an external effects device. The return from an external effects device can then be brought back into the console via the aux return connectors.



Mono Channels

Aux ends (MON and FX)

For situations that require effects processing, the aux sends are usually switched post-fader so that the effects volume in a channel corresponds to the position of the channel fader. If this were not the case, the effects signal of the channel would remain audible even when the fader is turned to zero.

Mono Channels

Aux ends (MON and FX)

When setting up a monitor mix, the aux sends are generally switched to pre-fader; ie. they operate independently of the position of the channel fader.

Both aux sends are mono, are sourced after the equalizer and offer up to +15 dB gain.

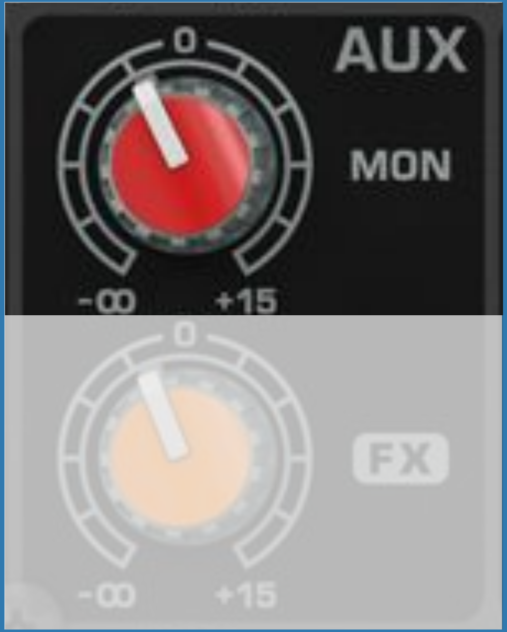
If you press the MUTE switch of the respective channel, aux sends and returns (MON and FX) are not being muted.

??Where is the MUTE switch??

Mono Channels

Aux ends (MON and FX)

Aus send 1 (MON) is wired pre-fader and is thus particularly suitable for setting up monitor mixes.



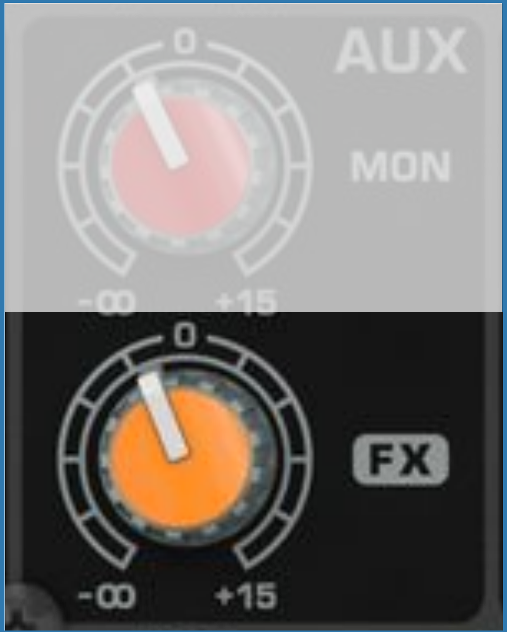
Mono Channels

Aux ends (MON and FX)

FX

The Aus send labeled FX is for feeding external effects devices and is thus set up to be post-fader.

The FX send is routed directly to the built-in effects processor. To make sure that the effects processor receives an input signal, you shouldn’t turn this control all the way to the left (-oo). Don’t have the FX MUTE switch pressed, and you should also not have the FX SEND fader pulled down.

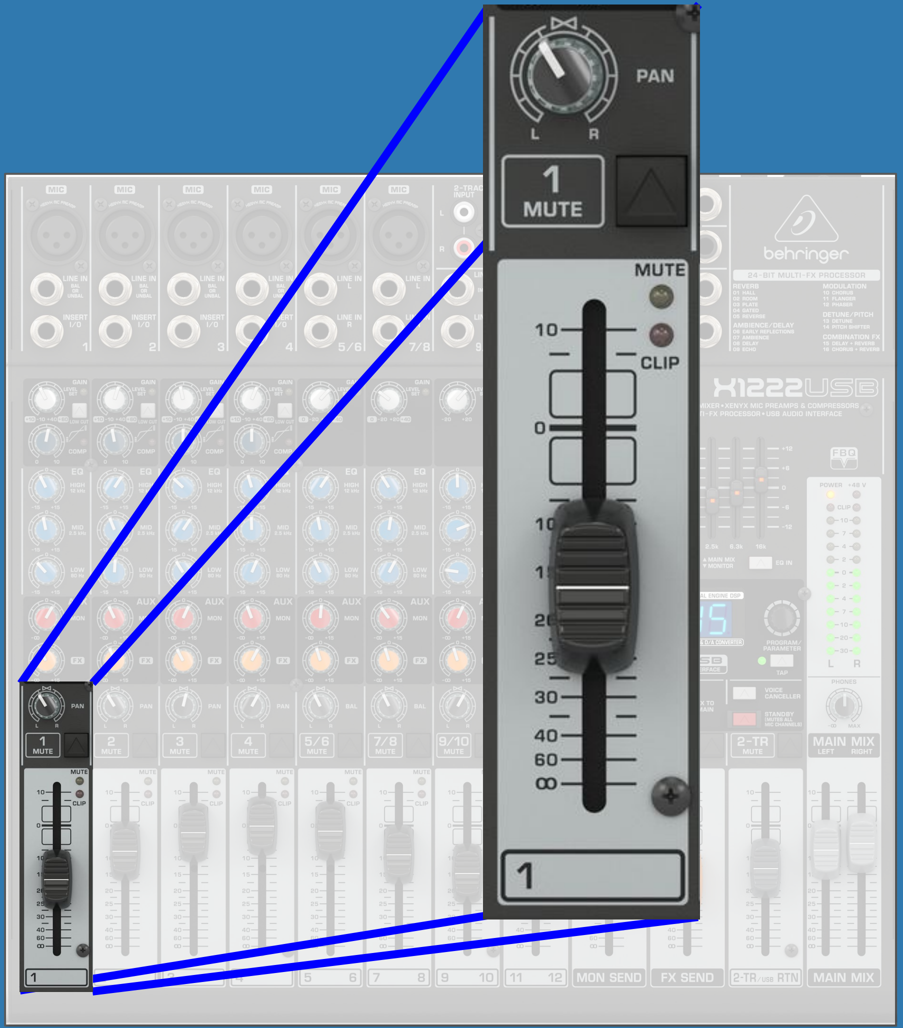


Mono Channels

PAN

mute switch

Channel fader



Mono Channels

Pan, mute switch and channel fader

PAN

The PAN control determines the position of the channel signal within the stereo image. This control features a constant-power characteristic, which means the signal is always maintained at a constant level, irrespective of position in the stereo panorama.

When working with subgroups, you can use the PAN control to assign the signal to just one output, which gives you additional flexibility in recording situations. For example, when routing to subgroups 3 and 4, panning hard left will route the signal to group output 3 only, and panning hard right will route to group output 4 only.

Mono Channels

Pan, mute switch and channel fader

MUTE

The MUTE switch breaks the signal path pre-channel fader, hence muting that channel in the main mix. The aux sends which are set to post-fader are likewise muted for that channel, while the pre-fader monitor paths remain active irrespective of whether the channel is muted or not.

MUTE LED

The MUTE LED indicates a muted channel.

Mono Channels

Pan, mute switch and channel fader

CLIP-LED

The CLIP LED lights up when the input signal is driven too high. In this case, lower apparent frequency increase on the channel EQ to avoid distortion. For example, lower the mids and the highs somewhat to emphasize the bass. If you don't wish to change the EQ settings under any circumstances, try lowering the GAIN control somewhat (counterclockwise). If you inserted an external effects processor via the insert connector (e.g. a dynamic processor), then you should also control its output signal level. It should not be higher than its input signal level (0 dB). The channel fader determines the level of the channel signal in the main mix.

◊ Attention: Since the aux path for the effect processor is connected post-fader, the channel fader has to be turned up in order to get this channel's signal to the effects processor!

Stereo Channels

Channel inputs

Each stereo channel features two line-level inputs on ¼” connectors for left and right channels. Channels 9/10 and 11/12 can also be used in mono if you only use the connector labeled “L”.

Both channels ⅚ and ⅞ feature an additional balanced XLR input for microphones with available +48 V phantom power.

Stereo Channels

Channel inputs

All stereo channel strips have a GAIN control for level setting. In those channels in which a mic input is present in the channel, the GAIN control has two scales: just like in the mono channels, there is a 0 to +40 dB scale that show the preamilification of the mic signal, the +20 to -20 dB scale shows the sensitivity for the corresponding input level that is applied to the line input.

Both inputs can also be used with balanced or unbalanced connectors.

Stereo Channels

Equalizer Stereo Channels

The equalizer of the stereo channels is, or course, stereo. The filter characteristics and crossover frequencies are the same as those of the mono channels, A stereo equalizer is always preferable to two mono equalizers if frequency correction of a stereo signal is needed. There is often a discrepancy between the settings of the left and the right channels when using separate equalizers.

Stereo Channels

Aux sends stereo channels

In principle, the aux sends of the stereo channels function in just the same way as those of the mono channels. As aux sends paths are always mono, the signal on a stereo channel is first summoned to mono before it reaches the aux bus.

Stereo Channels

Balance, mute switch and channel fader

BAL

The function of the BAL(ANCE) control corresponds to the PAN control in the mono channels.

The balance control determines the relative proportion between the left and right input signals before both signals are routed to the main stereo mix bus.

The MUTE switch, MUTE LED, CLIP LED and channel fader function in the same way as the mono channels.

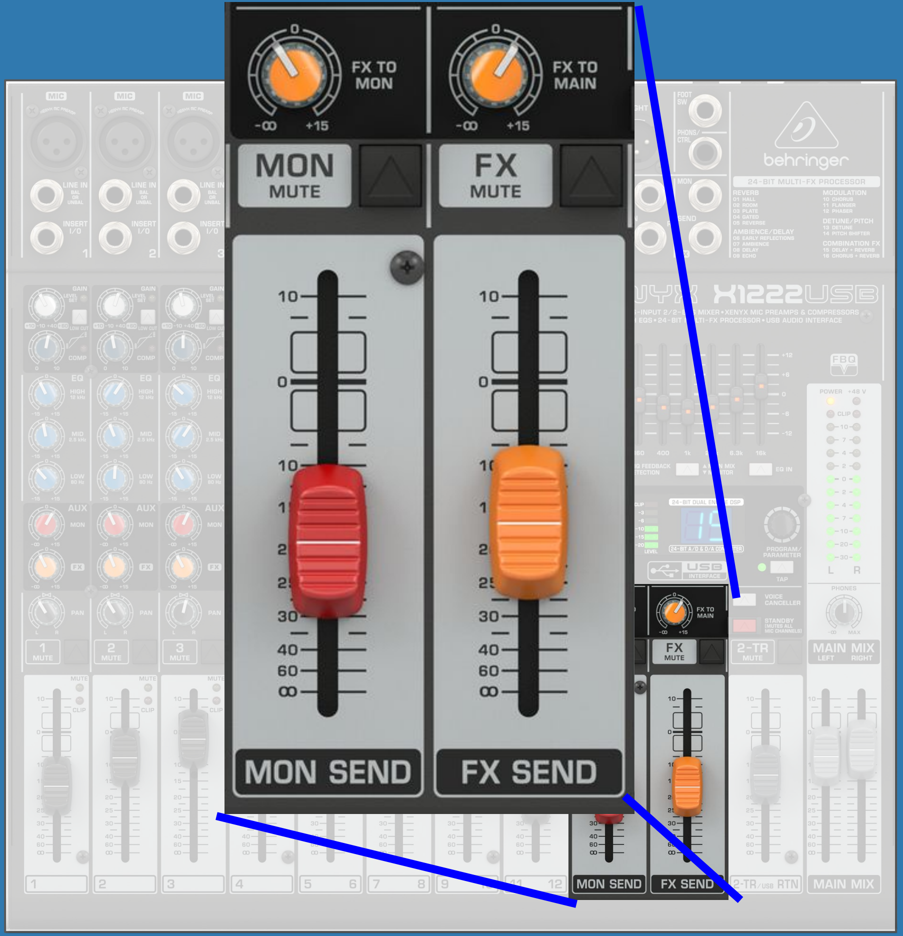
Connector panel and main section

Whereas it was useful to trace the signal flow from top to bottom in order to gain an understanding of the channel strips, we now look at the mixing console from left to right. The signals are, so to speak, collected from one point on each of the channel strips and then routed to the main section all together.

Connector panel and main section

Monitor send and FX send channels

A channel signal is routed to the MON(ITOR) send bus if the MON control is turned up on the corresponding channel.



Connector panel and main section

Monitor send and FX send channels

MON SEND

The aux send control MON SEND acts as master control for the monitor bus and determines the level of the summed signal that is taken from the mixer via the MON SEND connector and that can for example be fed to an amplifier for monitor purposes.

Connector panel and main section
Monitor send and FX send channels

MON SEND

Using the audio signal from this output, you can also feed a subwoofer if you don’t require stage monitors. To this end, you should implement a crossover in your signal path pre-subwoofer and pre-amplifier, so that only low frequencies are fed into the subwoofer. You can achieve the same effect by using the built-in graphical equalizer. Lower all frequencies above 160 Hz and assign the equalizer to “Monitor”.

Connector panel and main section

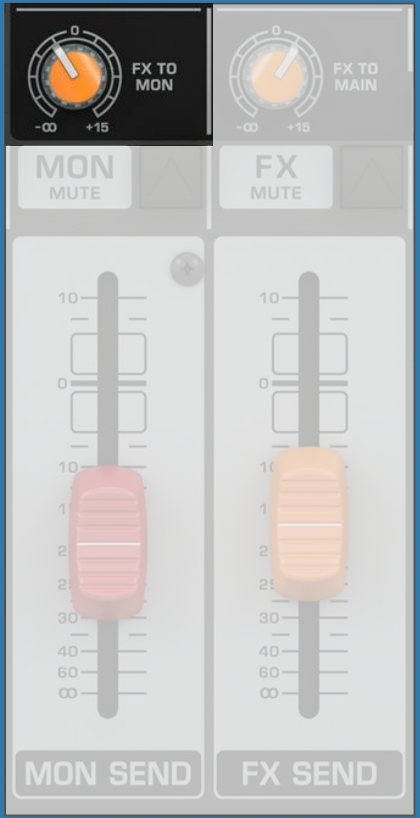
Monitor send and FX send channels

When you use the MAIN MIX fader to reduce the overall volume, keep in mind that the subwoofer is still receiving a signal!

Connector panel and main section
Monitor send and FX send channels

FX TO MON

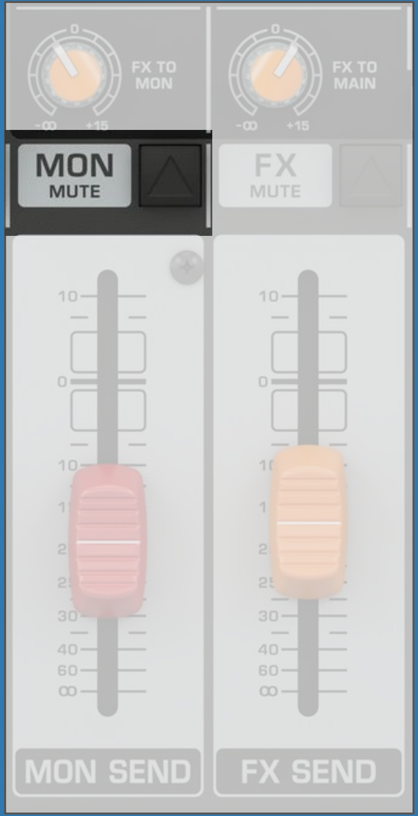
You can use this control to insert an effects signal from the built-in effects processor to your monitor mix. Of course, to do this, your effects processor must first receive a signal, i.e. the FX controls in the channel strips must be turned up, and the FX SEND fader (see fig. 2.6) hast to be open.



Connector panel and main section
Monitor send and FX send channels

MON MUTE

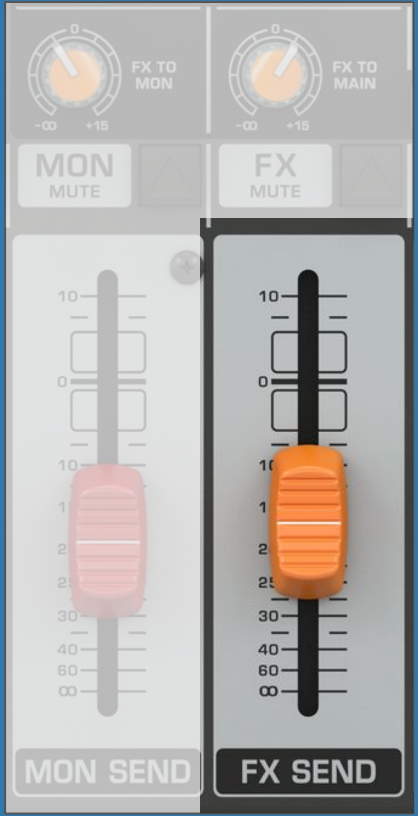
If the MON MUTE switch is pressed, the monitor bus is muted, i.e. there is no signal at the MON SEND connector.



Connector panel and main section
Monitor send and FX send channels

FX SEND

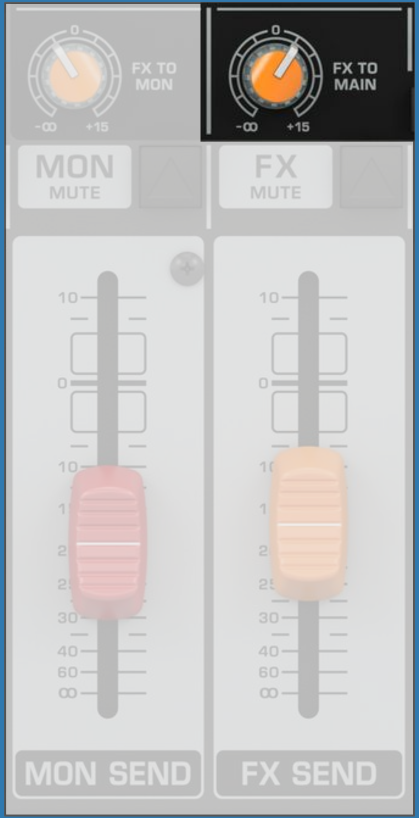
The FX SEND fader determines the overall level of the effects bus. Both external effects processors (via the FX SEND connector) and the built-in processor only receive an input signal if this control is open.



Connector panel and main section
Monitor send and FX send channels

FX TO MAIN

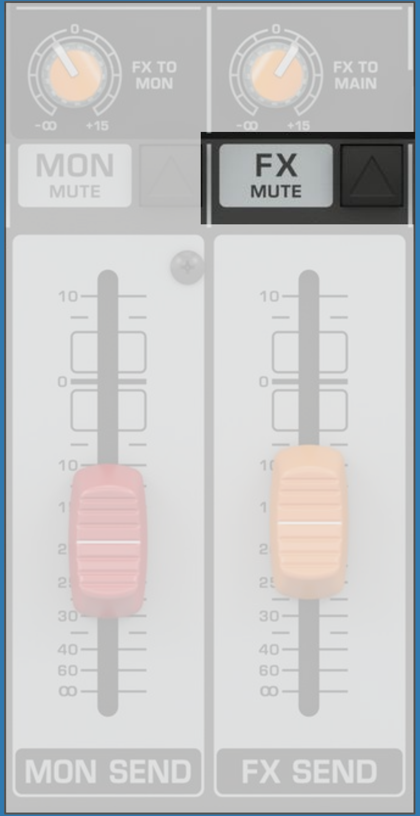
Use the FX TO MAIN control to feed the effects signal into the main mix. If the control is turned all the way to the left, no effects signal can be heard.



Connector panel and main section
Monitor send and FX send channels

FX MUTE

If the FX MUTE switch is pressed, the effects channel is muted, i.e. no signal is present at the FX SEND connector and the effects processor no longer receives an input signal.



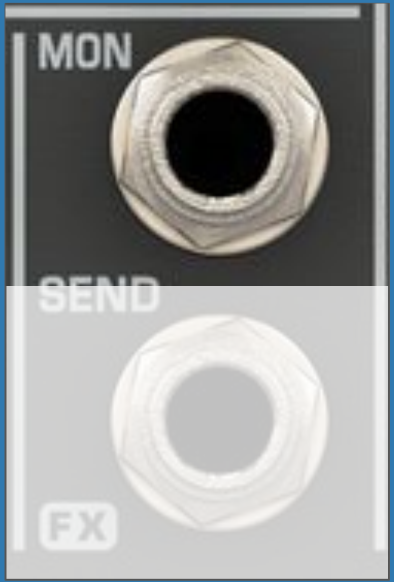
Connector panel and main section

Monitor send and FX send connector

Connector panel and main section
Monitor send and FX send connector

MON SEND

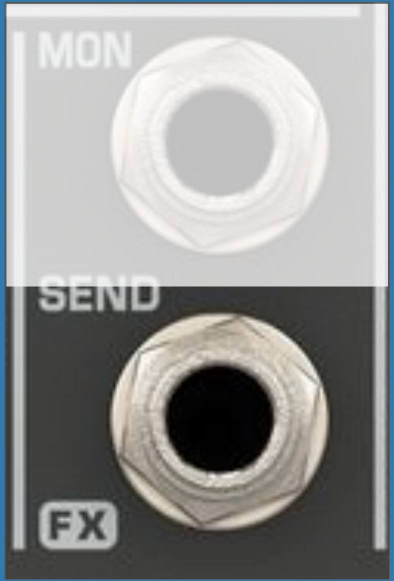
Connect the input of your monitor power amp or an active monitor system here to make the monitor mix audible to the musicians on the stage. The signal mix is created using the channels’ MON controls.



Connector panel and main section
Monitor send and FX send connector

FX SEND

The FX SEND connector outputs the signal you picked up from the individual channels using the FX controls. You can connect this to the input of an external effects device in order to process the FX bus' master signal. Once an effects mix is created, the processed signal can then be routed from the effects device outputs back into the AUX RETURN connectors.



Connector panel and main section

Monitor send and FX send connector

◊ If the connected effects processor receives no input signal,

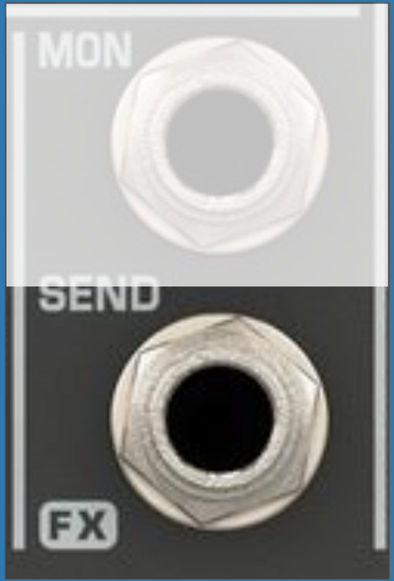
the FX MUTE switch is probably pressed and/or the FX SEND control

is too low. This also goes for the built-in effects processor.

◊ Adjust your external effects processor to 100% wet (effects signal

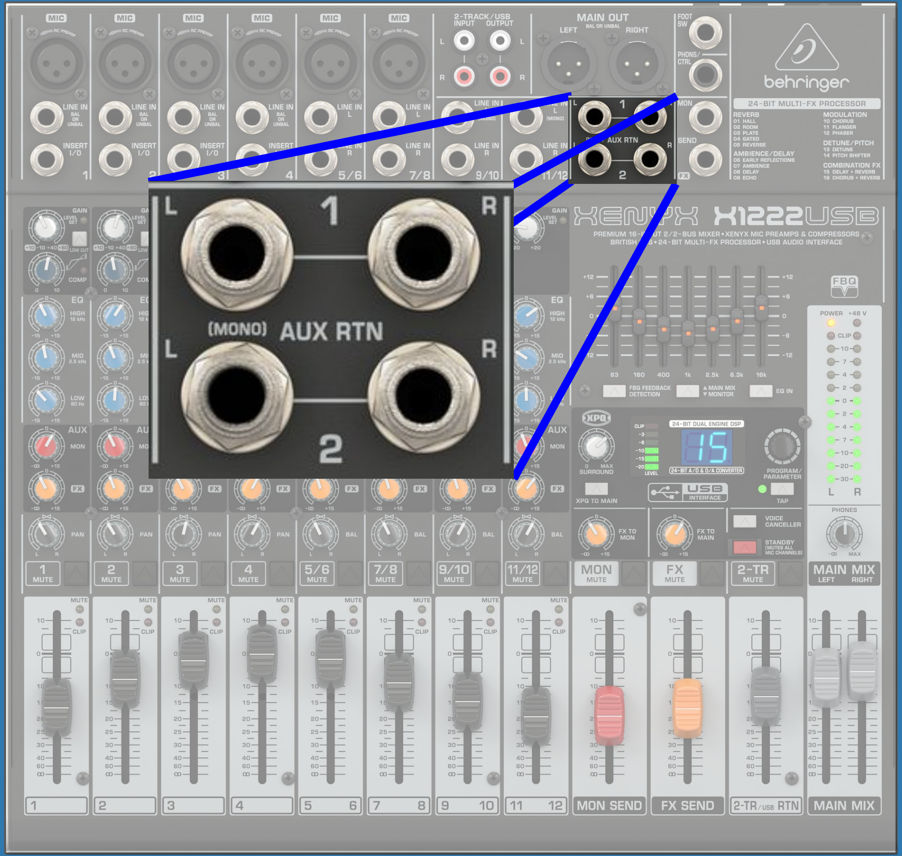
only), because the effects signal is added to the main mix along with

the “dry” channel signals.



Connector panel and main section

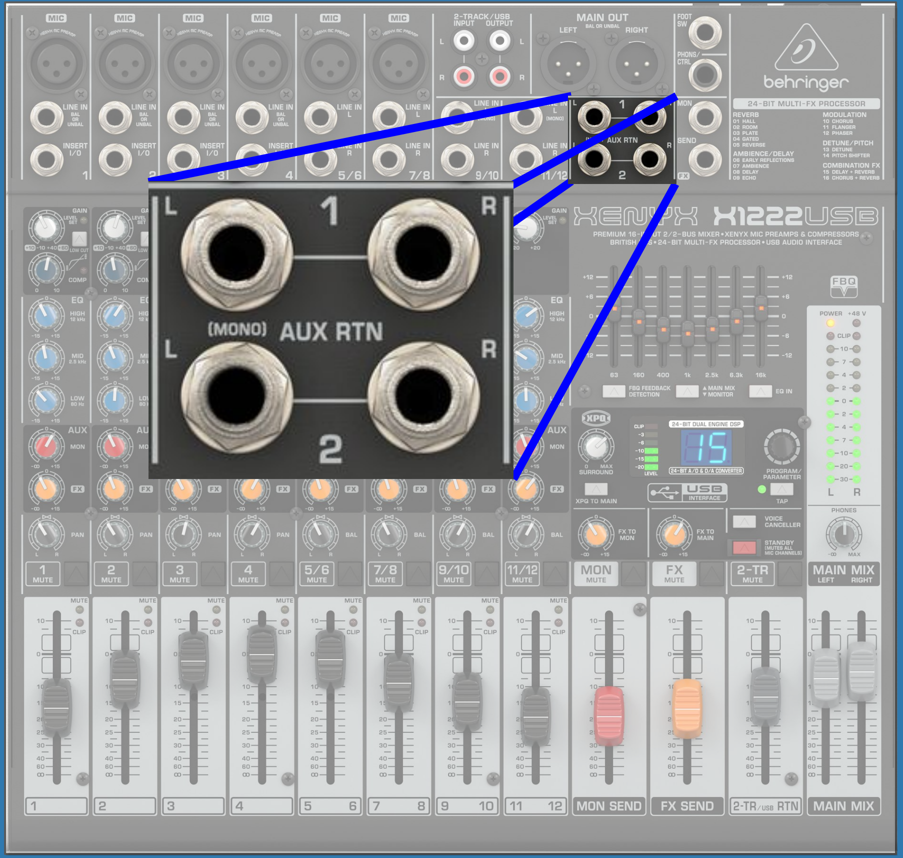
Aux return connectors



Connector panel and main section

Aux return connectors

AUX RETURN 1
The AUX RETURN 1 connectors generally serve as the return path for the effects mix generated using the FX send. This is where you connect the output signal of the external effects device. If only the left connector is used, the aux return 1 automatically operates in mono.
◊ You can also use these connectors as additional line inputs.



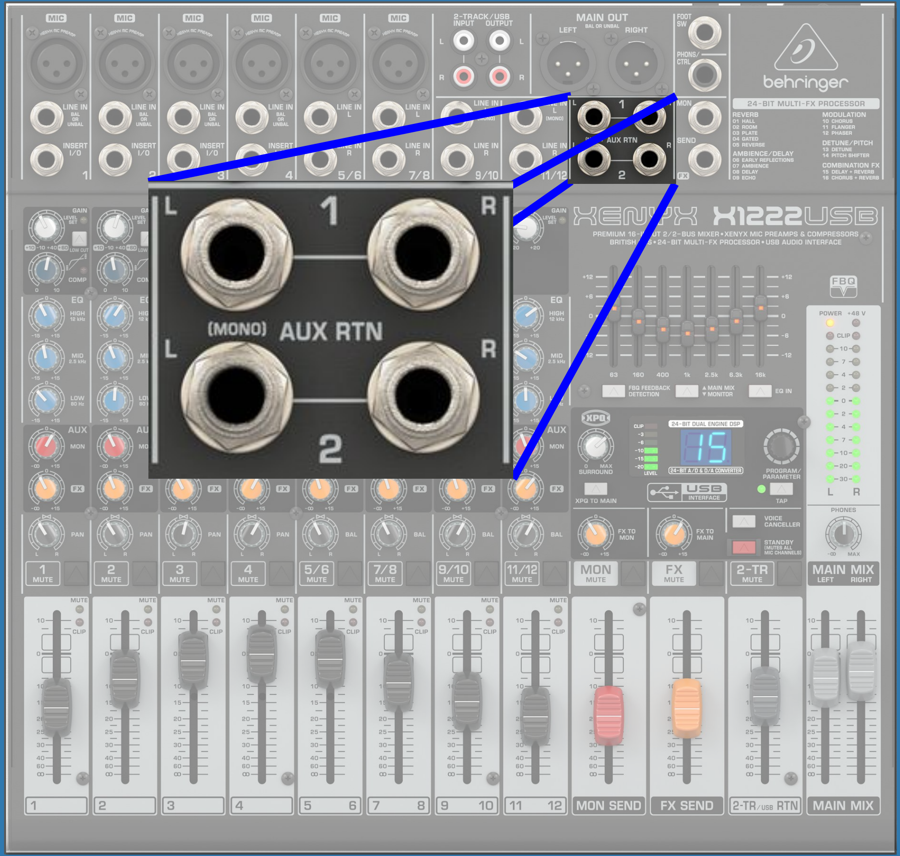
Connector panel and main section

Aux return connectors

AUX RETURN 2

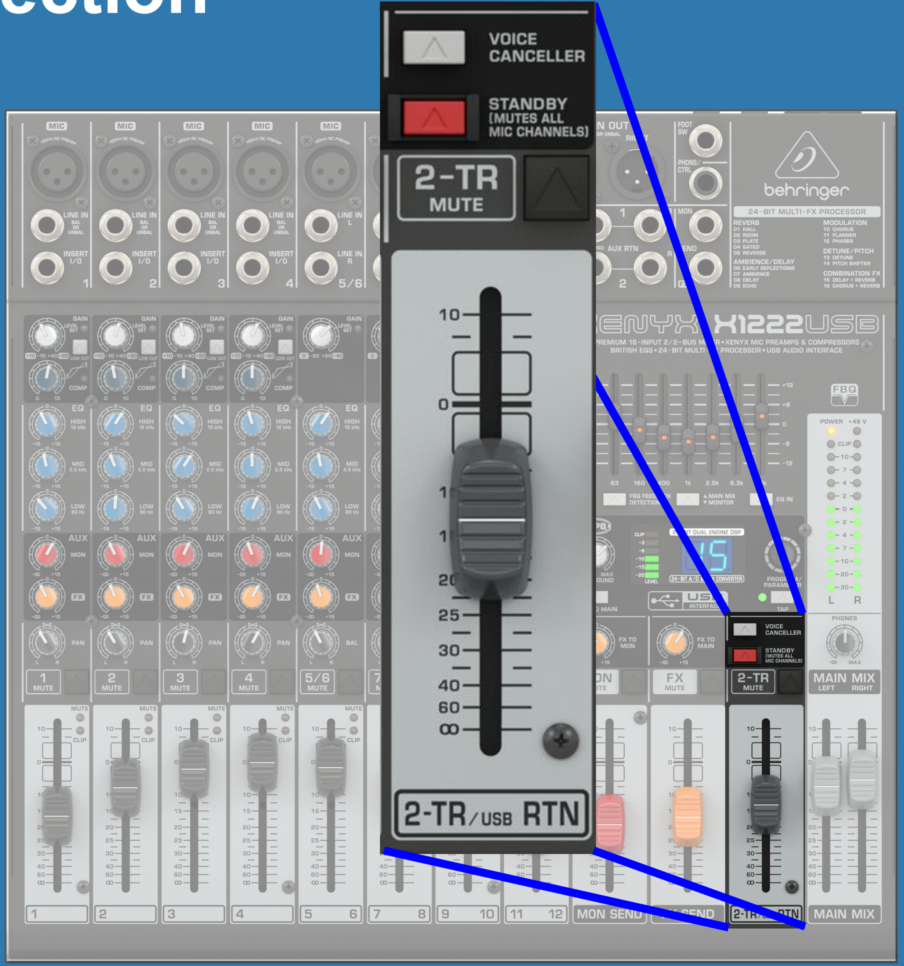
The AUX RETURN 2 connectors are used exactly the same way as the AUX RETURN 1 connectors. If these connectors already function as additional inputs, you can route the effects signal back into the console via a different stereo channel, with the added benefit that the channel EQ can be used to adjust the frequency response of the effects return signal.

◊ In this instance, the FX control of the channel being used as an effects return should be turned fully counter-clockwise, otherwise feedback problems can occur!



Connector panel and main section

CD/tape return channel, voice canceller and connection socket



Connector panel and main section

CD/tape return channel, voice canceller and connection socket

This channel, intended especially for connecting stereo signal sources

(CD players, DAT recorders or even sound cards) features a particularly

practical feature: the VOICE CANCELLER.

Connector panel and main section

CD/tape return channel, voice canceller and connection socket

VOICE CANCELLER

Here, you have a filter circuitry that lets you almost entirely remove the vocal portion of a recording. The filter is constructed in such a way that voice frequencies are targeted without majorly affecting the rest of the signal. Additionally, the filter seizes only the middle of the stereo image, exactly there where the vocals are typically located.

Possible applications for the Voice Canceller are obvious: you can very simply stage background music for Karaoke events. Of course, you can also do this at home or at your rehearsal room before you hit the stage. Singers with their own band can practice singing difficult parts using a complete playback from a tape player or a CD, thus minimizing rehearsal time.

Connector panel and main section

CD/tape return channel, voice canceller and connection socket

STANDBY

If the STANDBY switch is pressed, all input channels with a mic connector (XLR connector) are muted. During breaks or stage conversion, you can prevent noise from entering the sound system via the microphones. Such noise can in the worst-case scenario even irreparably damage loudspeaker membranes. The cool thing about this is that the main mix faders can remain open, so that you can play music from a CD at the same time. Similarly, the faders for the muted channels can also remain in their position. To bring in other sound sources, you can use the CD/tape inputs, stereo input channels 9 to 12 and the aux return inputs.

Connector panel and main section

CD/tape return channel, voice canceller and connection socket

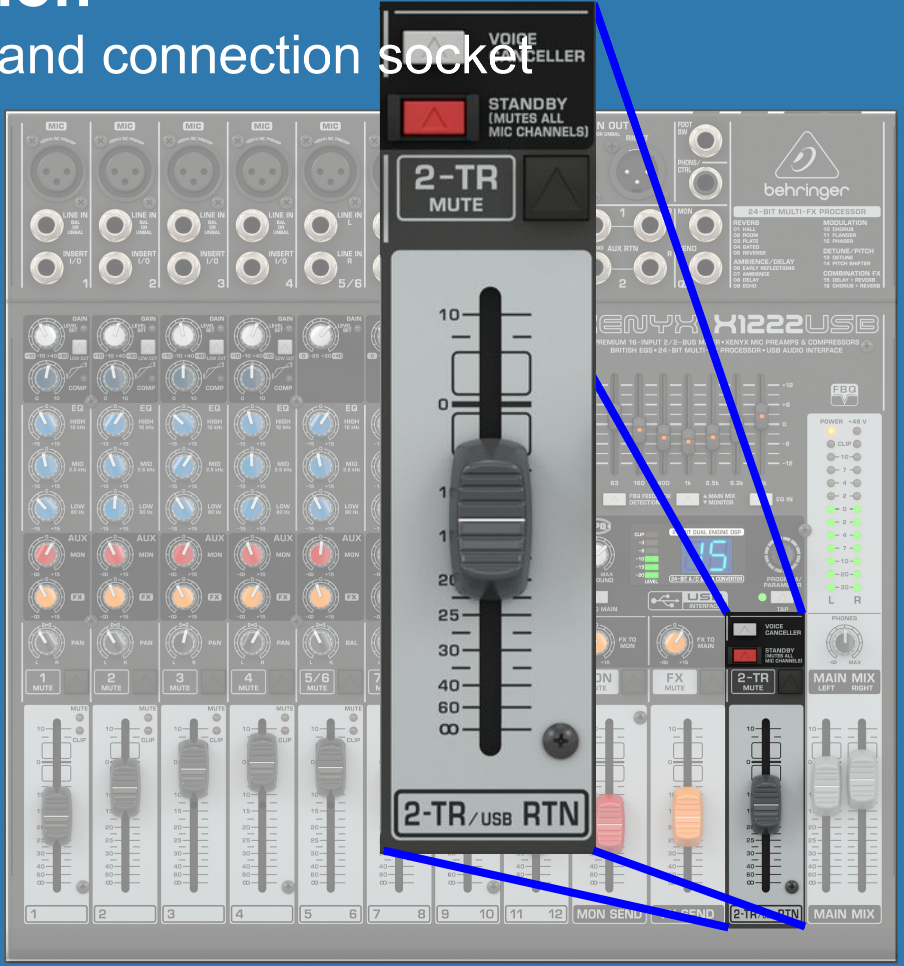
CD/TAPE MUTE

Using this switch, the input signal from the CD/tape inputs is muted.

Connector panel and main section

CD/tape return channel, voice canceller and connection socket

CD/tape return channel, voice canceller and connection socket




behringer
24-BIT MULTI-FX PROCESSOR
XENYX X1222USB
PREMIUM 16-INPUT 2/2-BUS MIXER • XENYX MIC PREAMPS & COMPRESSORS
BRITISH EQS • 24-BIT MULTI-FX PROCESSOR • USB AUDIO INTERFACE
MAIN OUT
2-TRACK/USB
MON SEND
FX SEND
2-TR/USB RTN
MAIN MIX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MAIN OUT
2-TRACK/USB
MON SEND
FX SEND
2-TR/USB RTN
MAIN MIX

HEADING

HEADING

CONTENT


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24-BIT MULTI-FX PROCESSOR
REVERB
01 HALL
02 ROOM
03 PLATE
04 GATED
05 REVERSE
AMBIENCE/DELAY
06 EARLY REFLECTIONS
07 AMBIENCE
08 DELAY
09 ECHO
MODULATION
10 CHORUS
11 FLANGER
12 PHASER
DETUNE/PITCH
13 DETUNE
14 PITCH SHIFTER
COMBINATION FX
15 DELAY + REVERB
16 CHORUS + REVERB
MIC
LINE IN
INSERT I/O
2-TRACK/USB INPUT OUTPUT
MAIN OUT LEFT RIGHT
FOOT SW
PHONES/CTRL
AUX RTN
MON
SEND
GAIN
LOW CUT
COMP
EQ
HIGH 12 kHz
MID 2.5 kHz
LOW 80 Hz
AUX
MON
FX
PAN
BAL
MUTE
CLIP
FBQ
FBQ FEEDBACK DETECTION
MAIN MIX / MONITOR
EQ IN
XPQ
SURROUND
XPQ TO MAIN
24-BIT DUAL ENGINE DSP
24-BIT A/D & D/A CONVERTER
PROGRAM/PARAMETER
USB INTERFACE
TAP
FX TO MON
FX TO MAIN
VOICE CANCELLER
STANDBY (MUTES ALL MIC CHANNELS)
POWER
+48 V
PHONES
MAIN MIX LEFT RIGHT
MON SEND
FX SEND
2-TR/USB RTN
MAIN MIX

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XENYX X1222USB
MAIN OUT
MAIN MIX
FX SEND
MON SEND
2-TR/USB RTN
MUTE


XENYX X1222USB
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MAIN OUT
MON SEND
FX SEND
2-TR/USB RTN
MAIN MIX
MAIN MIX LEFT RIGHT
24-BIT MULTI-FX PROCESSOR
USB INTERFACE
15


XENYX X1222USB
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POWER ON
PHANTOM ON
USB INTERFACE


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24-BIT MULTI-FX PROCESSOR
XENYX X1222USB
MAIN OUT
MON SEND
FX SEND
2-TR/USB RTN
MAIN MIX