MIXER, CABLES AND CONNECTORS REVISION PACK

SHORT QUESTIONS

In bars 2-5 a pre-fade auxiliary send has been used to apply reverb. Explain how you can tell the auxiliary send is pre-fade rather than post-fade. Describe the effect it creates.

(4)

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Question Number	Question	Mark
1(e)	In bars 2-5 a pre-fade auxiliary send has been used to apply reverb. Explain how you can tell the auxiliary send is pre-fade rather than post-fade. Describe the effect it creates.	4
	Acceptable Answers	
	The dry signal fades in / no dry sound at the start (1). With a pre-fade send, the wet signal remains constant (1) because the position of the fader does not affect the aux send amount (1). With a post-fade send, dry and wet signal fade together (1). It sounds like the chords are getting closer (1), (rather than just getting louder).	

(c) (i) Name the audio connector shown in the picture below.

(2)

....

(ii) State **two** uses for this type of connector.

(2)

1 ....

....

2 ....

....

Question number	Answer	Mark
3(c) (i)	Any two from: TRS / Tip Ring Sleeve (1) Jack / Phone / ¼" / 6.3mm (1) Stereo / Balanced (1)	**(2)**

Question number	Answer	Mark
3(c) (ii)	Any two from: Headphones (1) Insert cable / Y-lead / effects loop / side-chain insert (1) Balanced speaker / monitor cables (1) Balanced amplifier input/output (1) Balanced input/output on audio interface (1) Balanced input/output on headphone amp (1) Balanced input/output on mic pre-amp (1) Balanced input/output on mixing desk (1) Stereo effects return (on mixer) (1) Patchbay patch cable (1) Footswitch (1) Not reference to guitar leads which are unbalanced.	**(2)**

(d) (i) Name the audio connector shown in the picture below.

(1)

....

(ii) This type of connector is used for balanced connections. Explain how balanced connections minimise noise.

(3)

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(iii) State **two** possible sources of unwanted noise in an audio cable.

(2)

....

....

Question number	Answer	Mark
3(d) (i)	XLR (1) Cannon (1)	**(1)**

Question number	Answer	Mark
3(d) (ii)	Any three from: Shielded cable / ground (1). Carrying two identical signals (1). Out of phase signals / opposite polarity (1). Differential amplifier (1) signals back in phase (1) cancels noise at end of cable run (1) leaves only original signal (1) Impedance matching (1) Source and Load impedance the same (1)	**(3)**

Question number	Answer	Mark
3(d) (iii)	Fault in the cable or plug / bad connection (1) not 'faulty socket'. Ground loops (1) not just 'hum' Mains power source (1) Mobile phones (1) Dimmer switches / lighting (1) Noise caused by interference (1) and/or electromagnetic induction (1)	**(2)**

Figure 1 shows a selection of leads. Identify and explain features and applications of these leads.

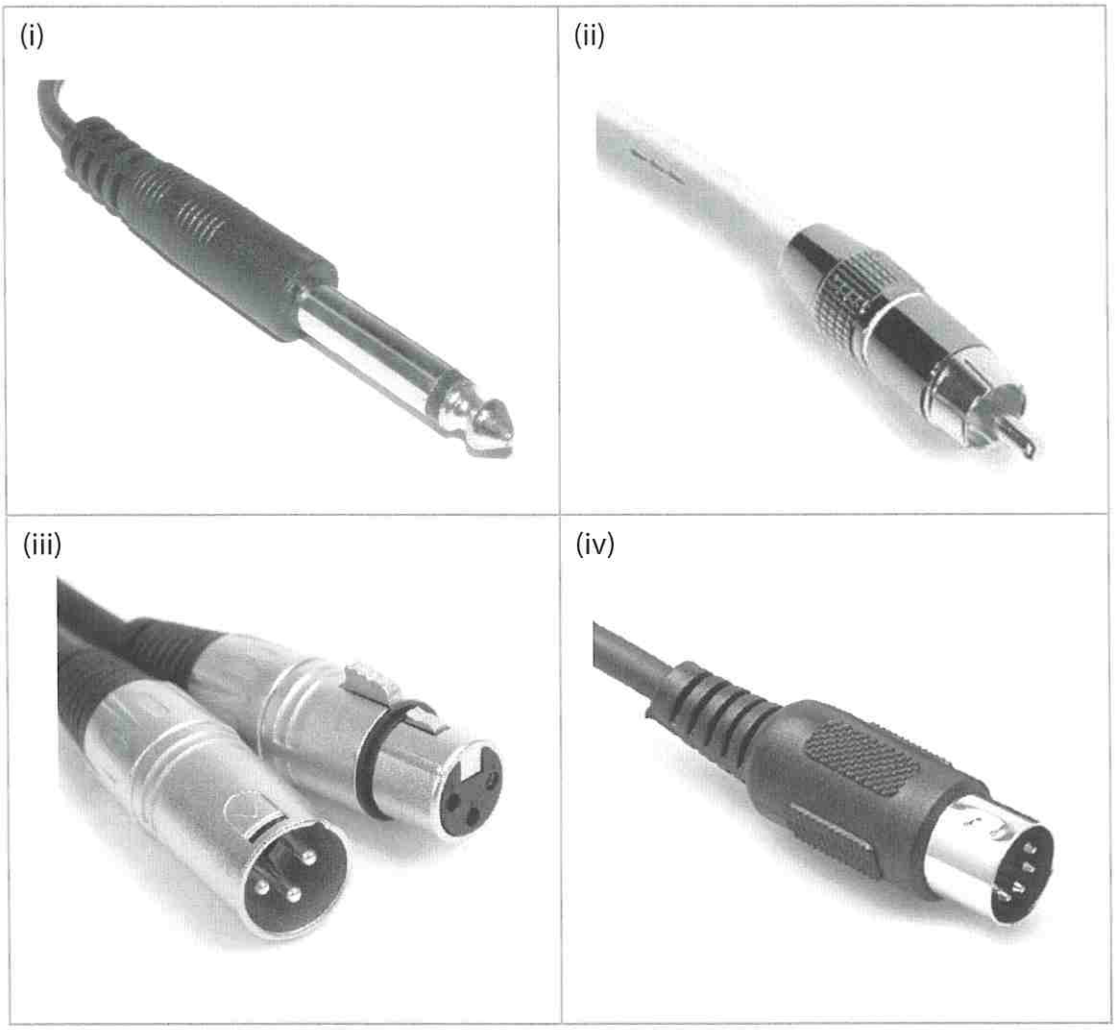
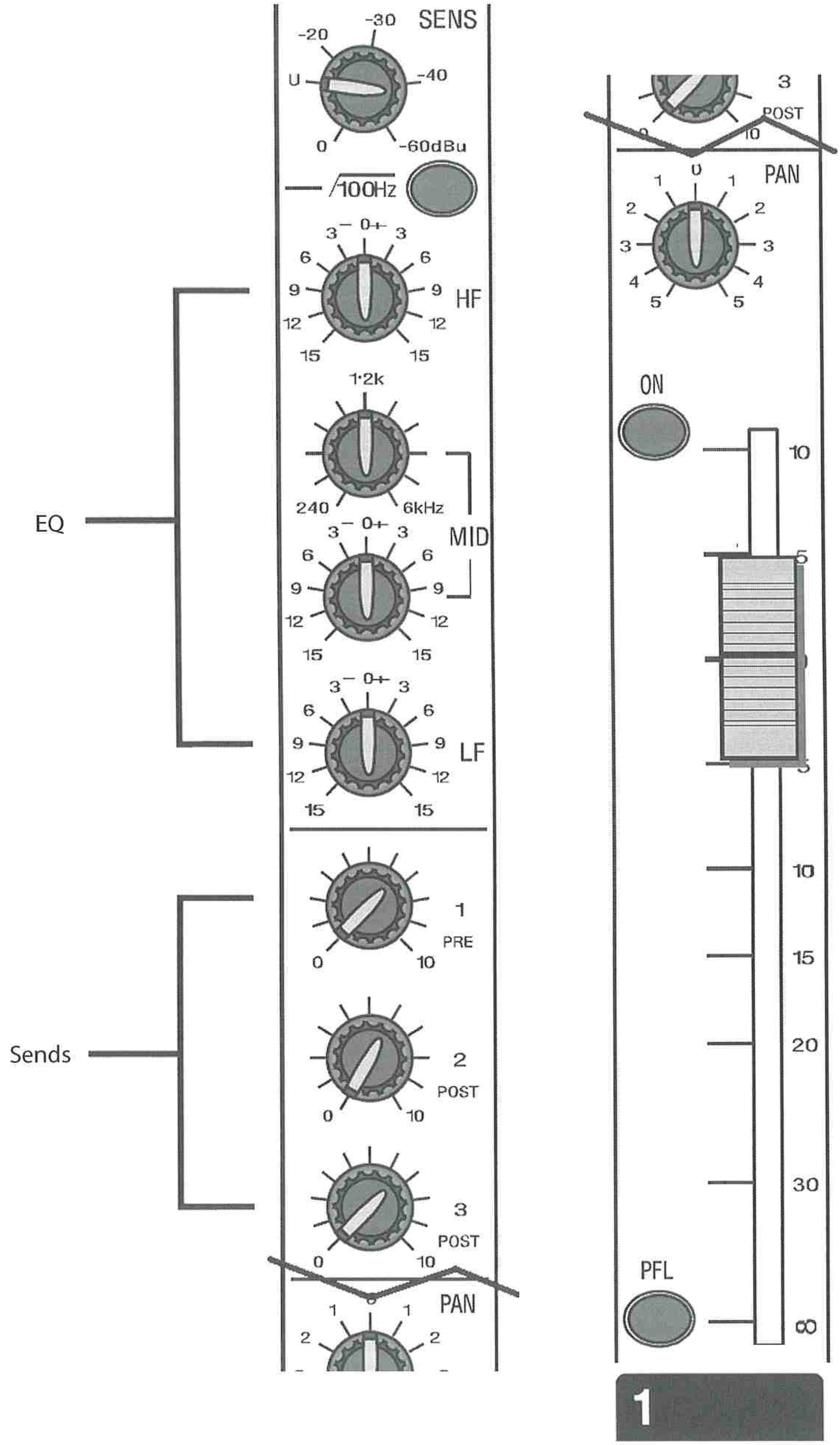


Figure 1

Figure 1 shows a channel on an analogue mixing desk. Many of the controls are similar to those on a digital audio workstation track. Explain the function of the controls and specifications that can be seen in the picture. Give **one** practical use for each control.



SENS
-20
-30
U
-40
0
-60dBu
100Hz
HF
1·2k
240
6kHz
MID
LF
1
PRE
0
10
2
POST
3
POST
PAN
ON
PFL
10
5
0
5
10
15
20
30
∞
1
EQ
Sends

Figure 1

Music Production Terminology

The Mixing Desk

Many of these functions are also available in DAWs or as audio interface hardware, so keep an eye out for these terms as the are likely to function in the same way. Let's work our way down a typical mixing desk channel strip.

Input
This will normally be the first item you will find. When a mixing desk has more than one input per channel, this button selects the source. Most mixing desks will have at least two inputs, one being the 'microphone input' (for connecting microphones) and the other being a 'line input' (for connecting line level devices).

The switch you will see on the channel will often be labelled as 'line'. This indicates that when the switch is up (or off) then the default microphone input is being used and when the switch is down (or on) then the line input is being used.

Some mixing desks may have more than one line input, in which case there may be more than one switch labelled 'line 1', 'line 2' etc.

Pad
Sometimes a signal may be so loud that even the trim control cannot reduce the level enough, therefore the pad switch enables you to reduce the level by a specified amount (e.g. 10 or 12dB) while still leaving you with the fine adjustment provided by the trim or gain control. The pad switch is used purely to attenuate (make quieter) the incoming signal.

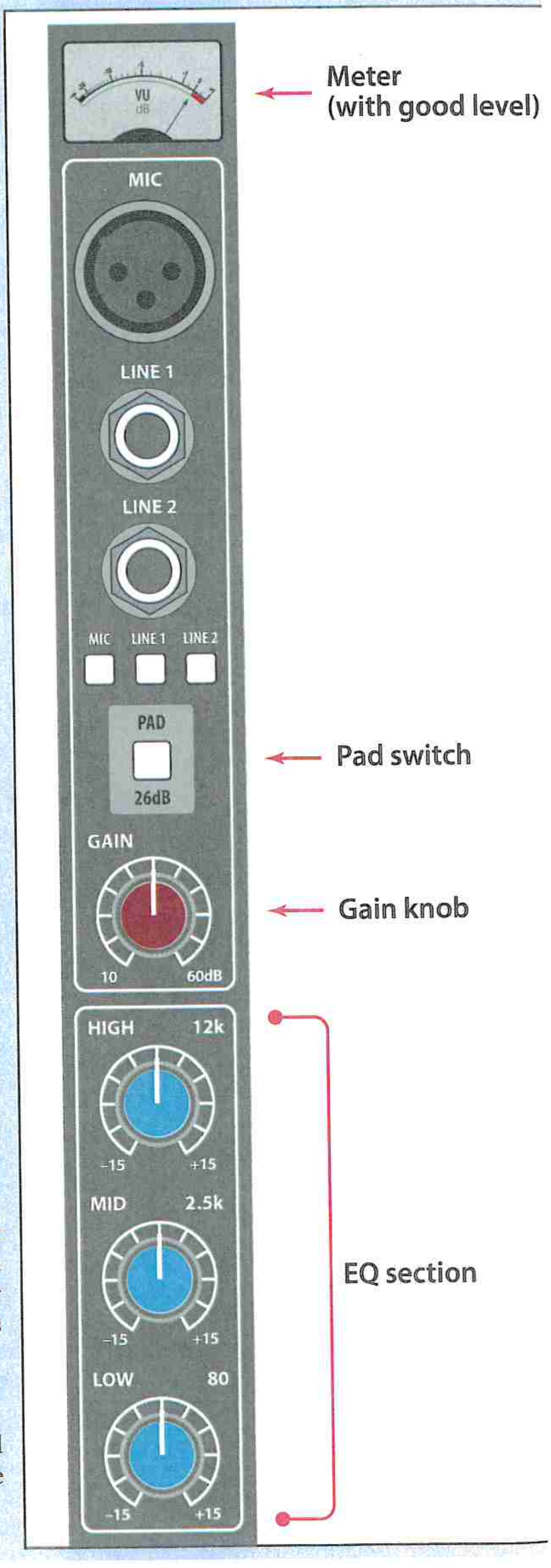
Gain/Trim
The gain pot on a mixing desk or audio interface determines the amount of signal entering the channel.

As a microphone doesn't have any power, a microphone signal will need amplifying so that it is loud enough to be heard, therefore the gain control is used to raise the level of the signal.

As line level equipment is louder than microphones, the gain pot on a line input might have an indentation at the 12 o'clock position. This is the point at which no gain is being applied and the signal entering the channel is the same level as the source. Turning the control to the left reduces the amount of signal and turning it to the right increases the signal. As it can also make the signal quieter, this control is referred to as 'trim' rather than gain.

If you have a meter (a display which shows you how loud the signal is) then as a general rule, you should aim to have enough signal entering the channel so that the meters are in the yellow or orange range. Green is OK but may be a little quiet. Red means that the signal is likely to distort because it's too loud.

Equalisation
The equalisation or EQ is used to balance the tone of the signal in the channel. More information can be found on EQ in the Grade 2 curriculum.



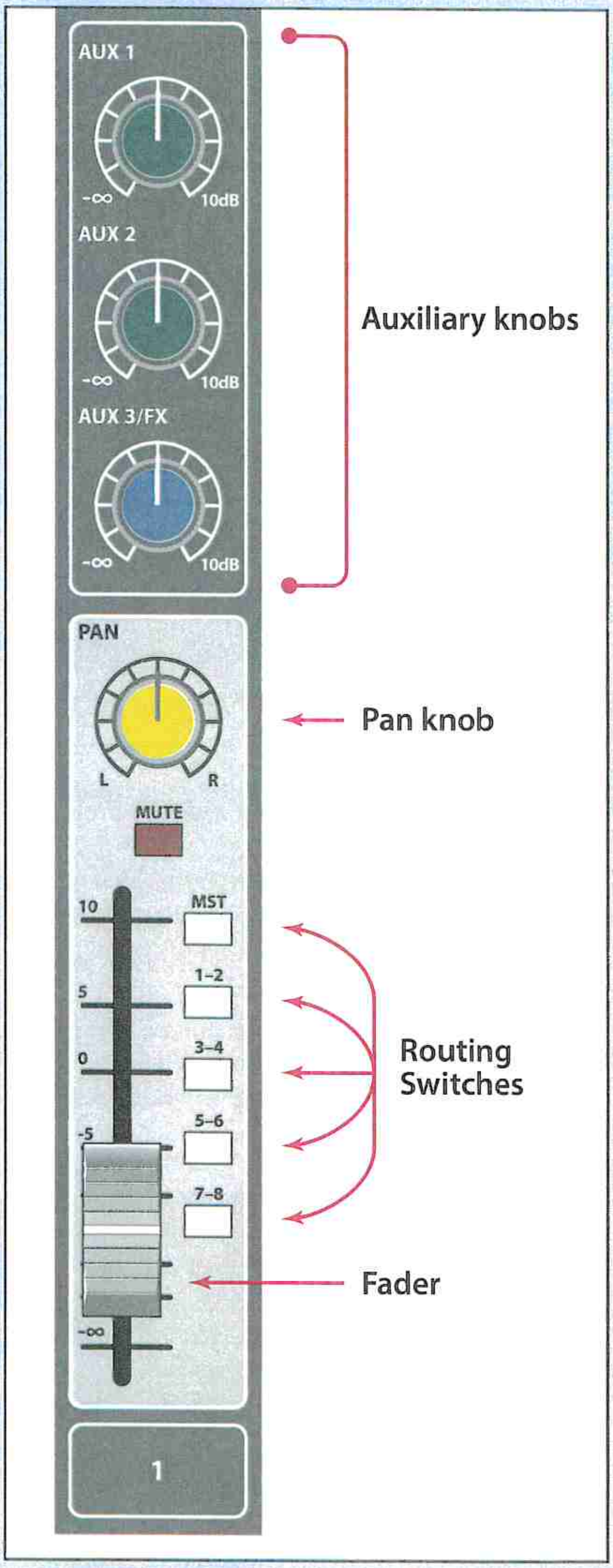
Auxiliaries
Auxiliaries are the part of a mixing desk which enables you to end part of a signal in a channel to an additional destination. Often used for sending some signal to a reverb, delay (echo) or other effect.

Pan
The pan (panorama) control is used to distribute the signal between the left and right speaker. This makes it possible to make a sound appear as if it's coming from the left side of the room, the ight side, or anywhere inbetween. This is very useful when you are mixing a very busy arrangement, such as multiple guitars and drums as more space is available to fit everything in.

Routing
The routing controls will vary in complexity from one mixing desk to another and some desks may not have any routing at all. If there is no routing, then the output of the channel will automatically be sent to the master fader.

Additional routing makes it possible to send the signal to other destinations, such as a recording device.

Fader
The fader is the part of the mixing desk which adjusts the level of the signal leaving the channel. You will use the faders to either adjust the level that is being recorded, or to adjust the balance between instruments when mixing.



Master Fader
This is normally on the right hand side of a mixing desk, or sometimes in the middle of larger mixing desks. This controls the level of the overall mix. For example, you might use this to fade out at the end of a song.

Music Production Terminology

Studio Equipment

Controlling the signal path in the studio is key to being able to get the most from the equipment. The methods of controlling the signal path are collectively known as 'routing'.

The term 'routing' is used because you are finding a route or path for the signal to take through the equipments inputs, outputs and functions. There are many switches and knobs which enable you to make choices as to where the signal goes.

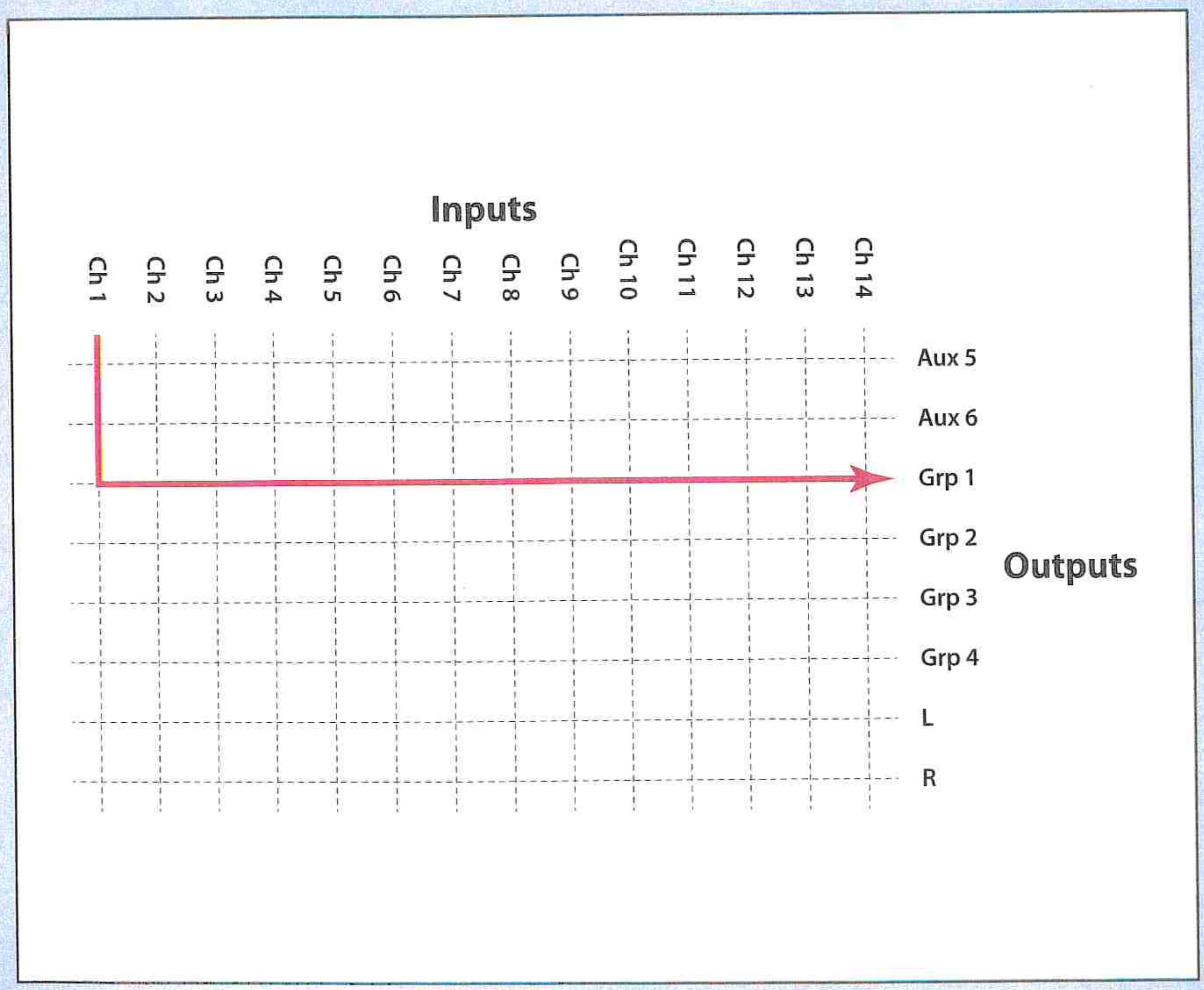
Whether you are using hardware or software for these functions, they will adhere to the same concepts and will generally use the same names.

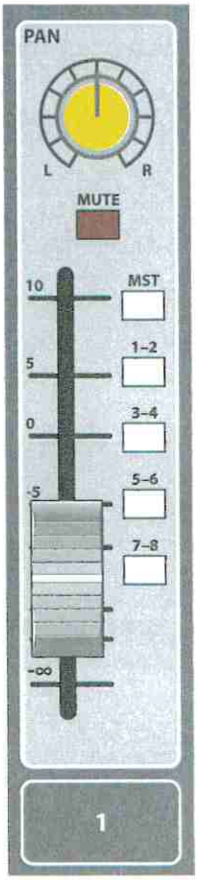
Routing Matrix

The routing matrix is a series of switches you may find on a channel of a mixing desk. These switches control where the signal will go once it leaves the channel.

The number of potential destinations will vary, depending on the size and complexity of the mixing desk. These destinations are called buses.

Buses are the internal cabling that runs through the mixing desk, through all the channels. Think about them like transport buses which take people from which ever stop they get on at to their destination.

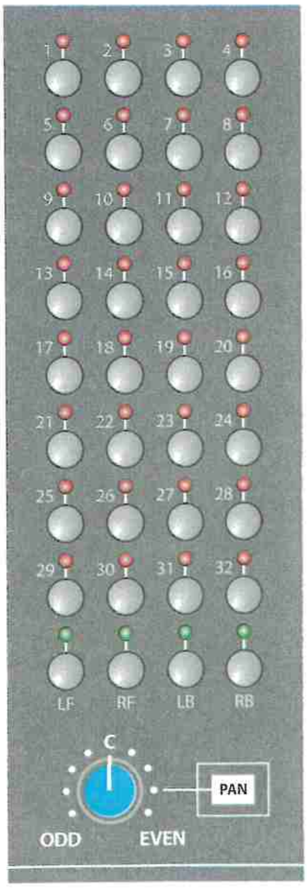




A small mixing desk may not have any routing at all with the default being that the output of the channel goes to the master fader or main output, also known as the 'mix bus'.

A mid size 'project studio' desk may have 8 buses, numbered one to eight. Plus the ninth option called 'mix'. This means there are 8 destinations that the signal can be routed to or the signal can be sent to the master 'mix' fader.

It's quite common on smaller consoles for these buses to be in stereo pairs i.e. '1&2', '3&4' and so on. In order to send to just 'bus 1', you would need to use the routing switches to select '1&2' and then use the channel pan control to pan to the left. To send to 'bus 2', you should pan to the right.



A large professional mixing console may have 24, 32, 48 or even 64 buses.

Each bus will have an associated 'group'. A group is a fader or knob which controls the overall level of everything being routed to that bus. This means that if you are sending signal from multiple channels to bus 1, then you can use group 1 fader to adjust the overall level of everything on that bus.

These buses and groups can be used for any purpose you need them to, that's up to you. When mixing, you might use them to group together sets tracks. For example, if you have a drum kit recorded on channels 1 to 8, then you could route channels 1 through 8 to bus 5 which then makes it possible to raise or lower the level of the whole drum kit with just the group 5 fader.

Another option is to use the groups as a way of controlling the level that you're sending to your recording device by connecting the group output to the input of your DAW or tape machine.

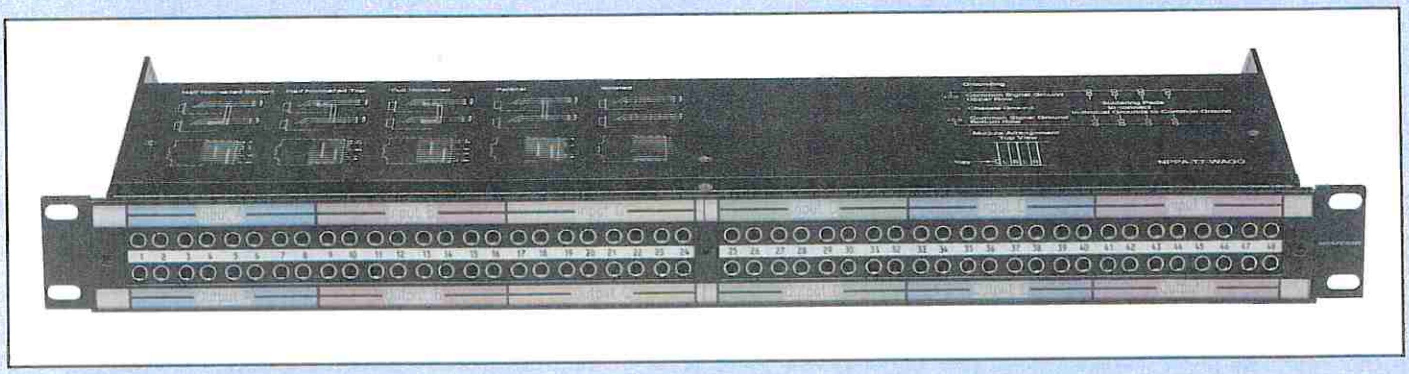
Section A | Music Production Theory

Auxiliaries
The auxiliary section of the channel is used to send a portion of the channel's signal to another location. This doesn't affect the signal in the channel itself as it is effectively taking a copy.

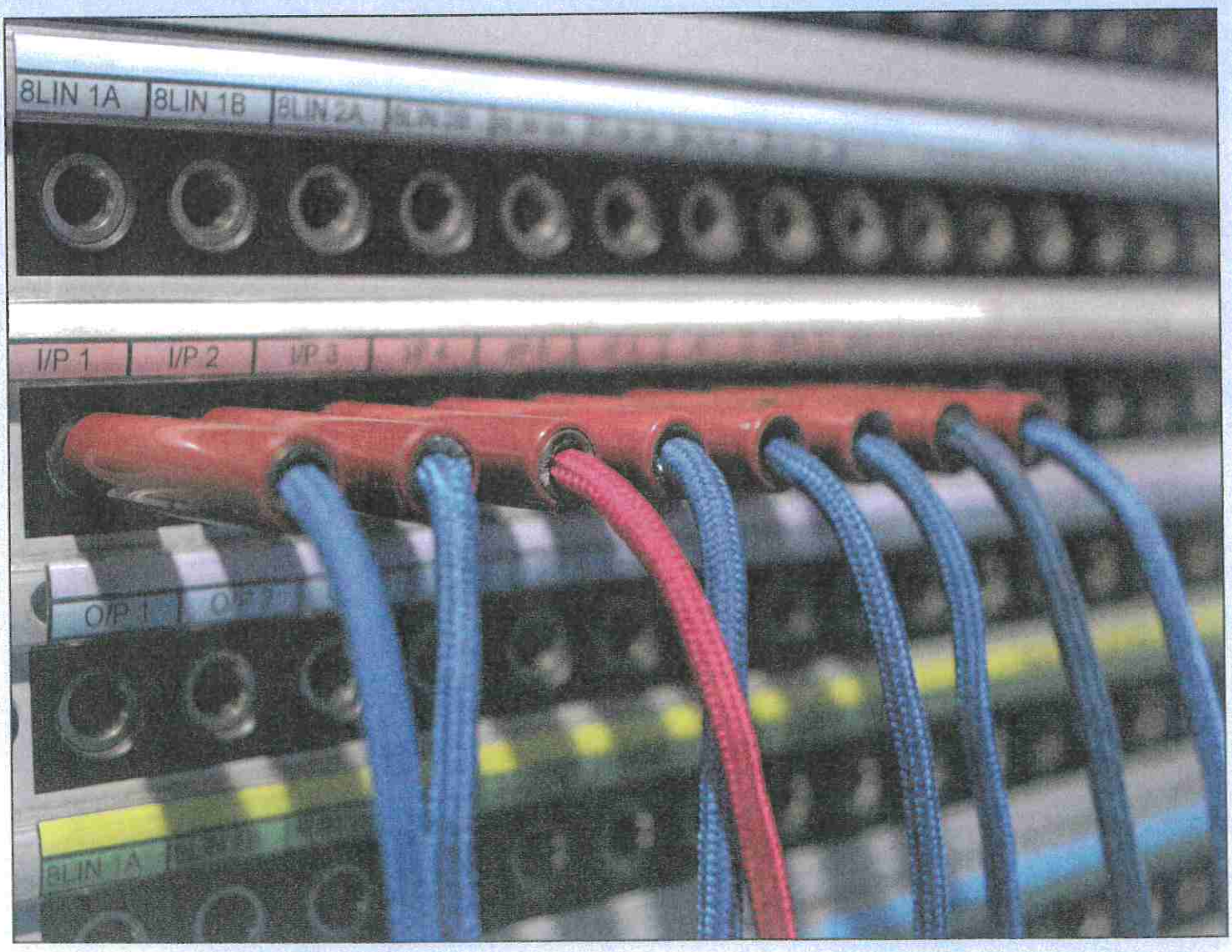
This is most commonly used for effects sends (reverb, delays etc.) or for creating headphone mixes for the musicians.

Patch Bay
The most basic routing within a studio is performed at the patch bay as it enables you to connect one device to another.

The patch bay is normally located in one of the equipment racks and is a large number of jack sockets, each of which relate to an input or output of a studio device.



For example, the inputs of the mixing desk will be on the patch bay as will the inputs and outputs of any compressors, effects or other devices.



This makes it possible to easily connect an output from one device to the input of another using a short patch cable. Without the patch bay you would need to run lengthy cables across the studio, reaching around to the rears of devices. This would be very difficult and would slow the recording process down considerably.

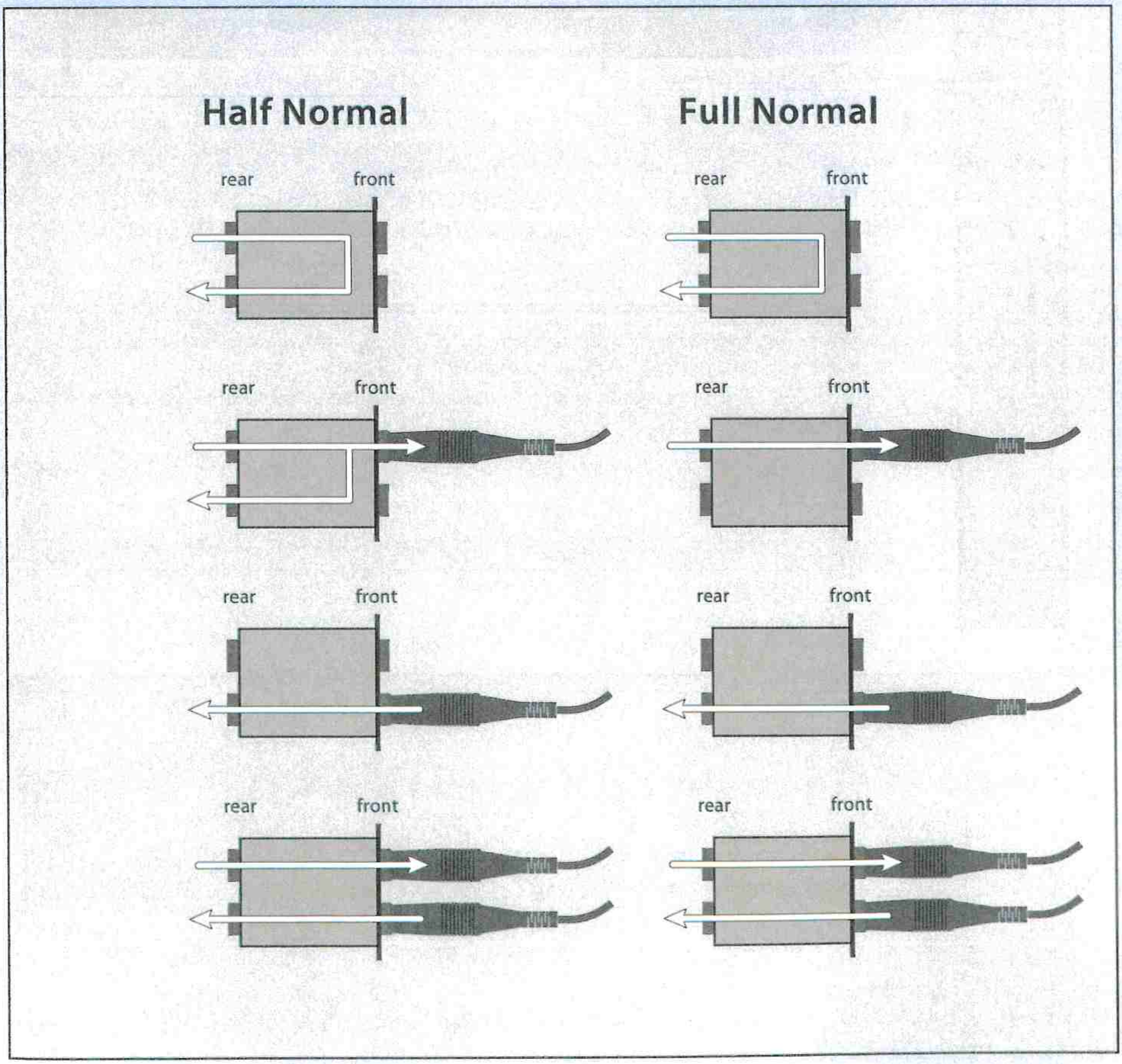
A patch bay may use ¼" jack connectors or the more professional Bantam connectors. Bantam connectors are thinner so it's possible to fit a lot more onto one strip of patch bay.

Normalling / Half Normalling

As there are some devices that are almost always connected together, the patch bay has a function called normalling.

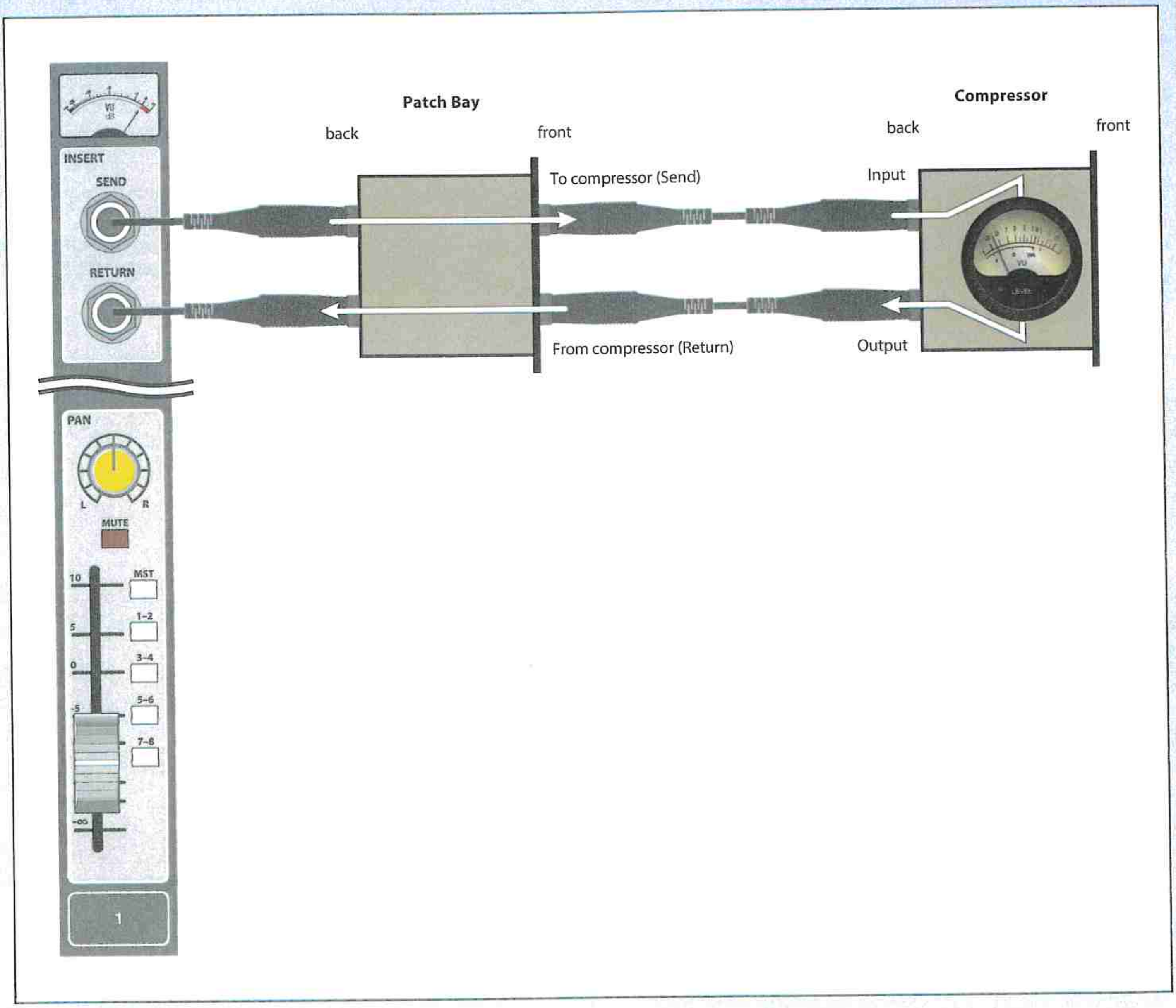
When an output on one row of the patch bay is 'normalled' to the input on the row below it, this means the output and input are automatically connected unless a patch cable is connected to either of those two sockets. When a cable is connected, the automatic connection is broken, allowing the sound engineer to patch a different device into that input.

Half normalling is very similar but the connection is only broken when a patch cable is connected to the input socket. This is useful if you want the signal to continue from output to input and you also want to send the output signal somewhere else as well.



Inserts
Inserts are a pair of patchable connections in the channel of a mixing desk. The pair makes a loop, featuring a send (output) and return (input). This is useful if you want to process the signal on that channel and have it return to the channel such as when you want to use a compressor or gate.

The signal flows through the channel until it gets to the insert send. The insert send socket will appear on the patch bay, enabling you to patch the insert send to the compressors input, then patch the compressor's output to the insert return. The signal then returns to the channel and continues on it's way as normal.



:anced Audio

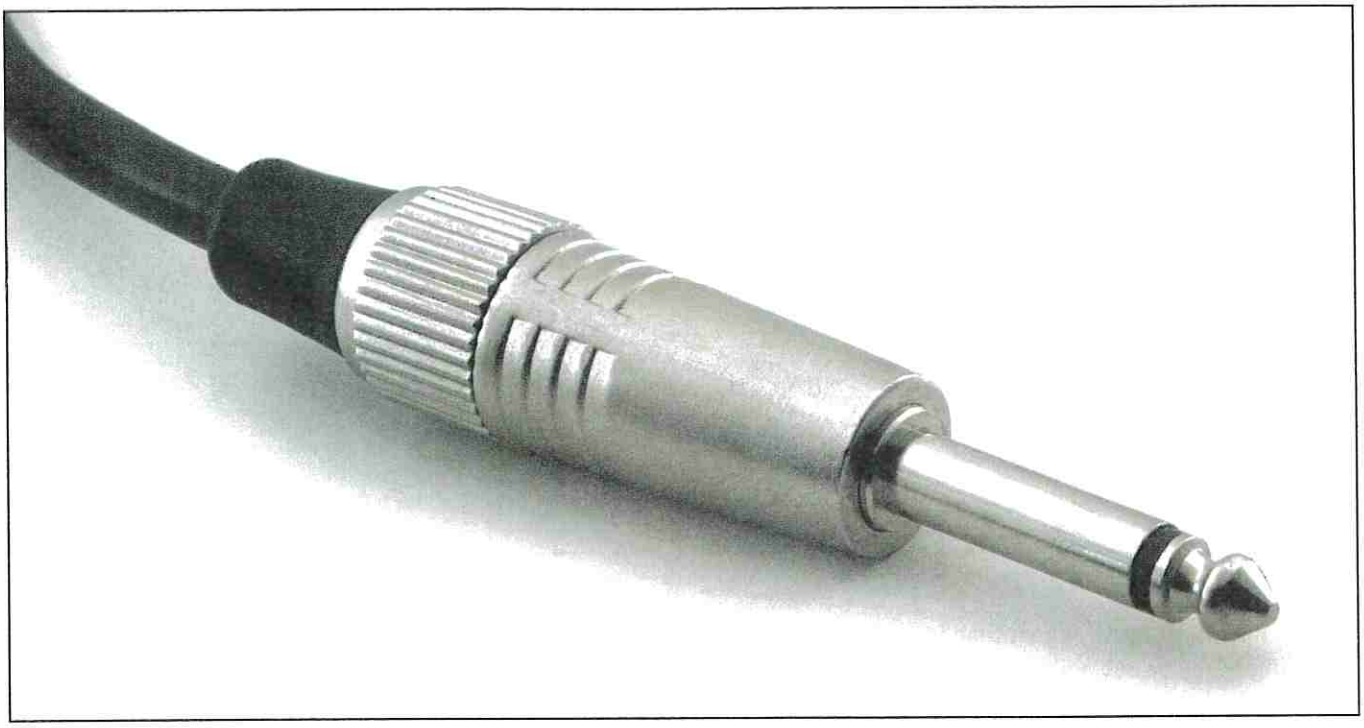
lio connections in a studio can be broken down into two categories:

- Unbalanced
- Balanced

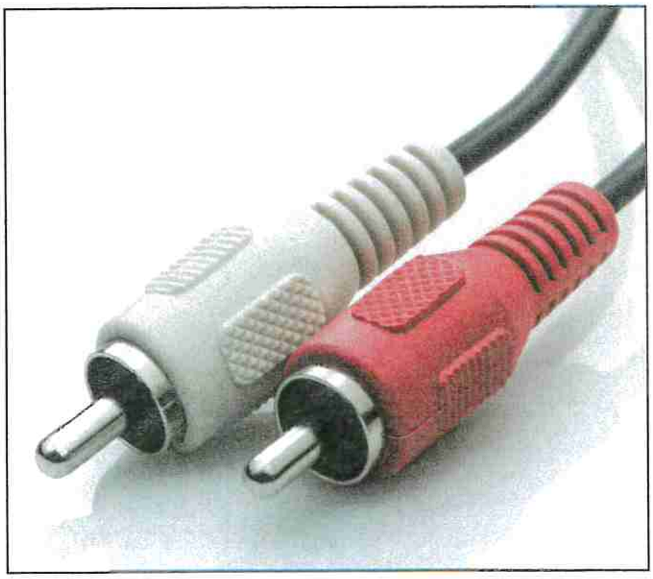
nbalanced

n unbalanced connection uses cabling and a connector with only two points of contact.

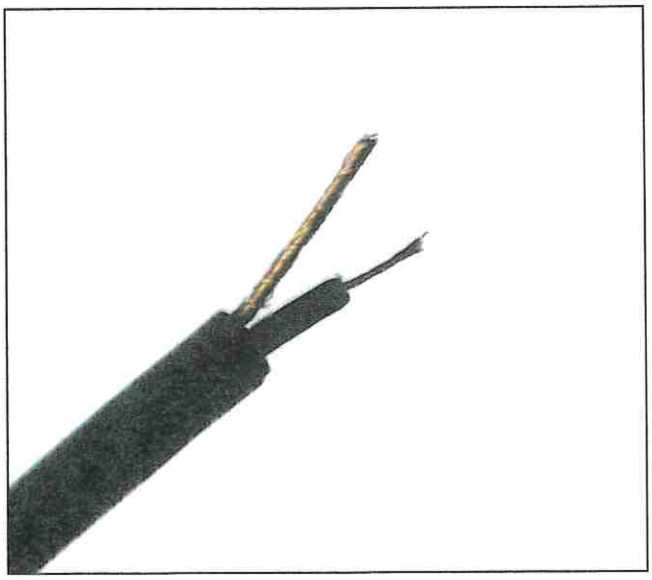
On a jack connector, this would be the tip and the sleeve.



Another type of unbalanced connector is the RCA/ Phono connector.



These are the simplest types of connections as they use one signal cable and a screen.

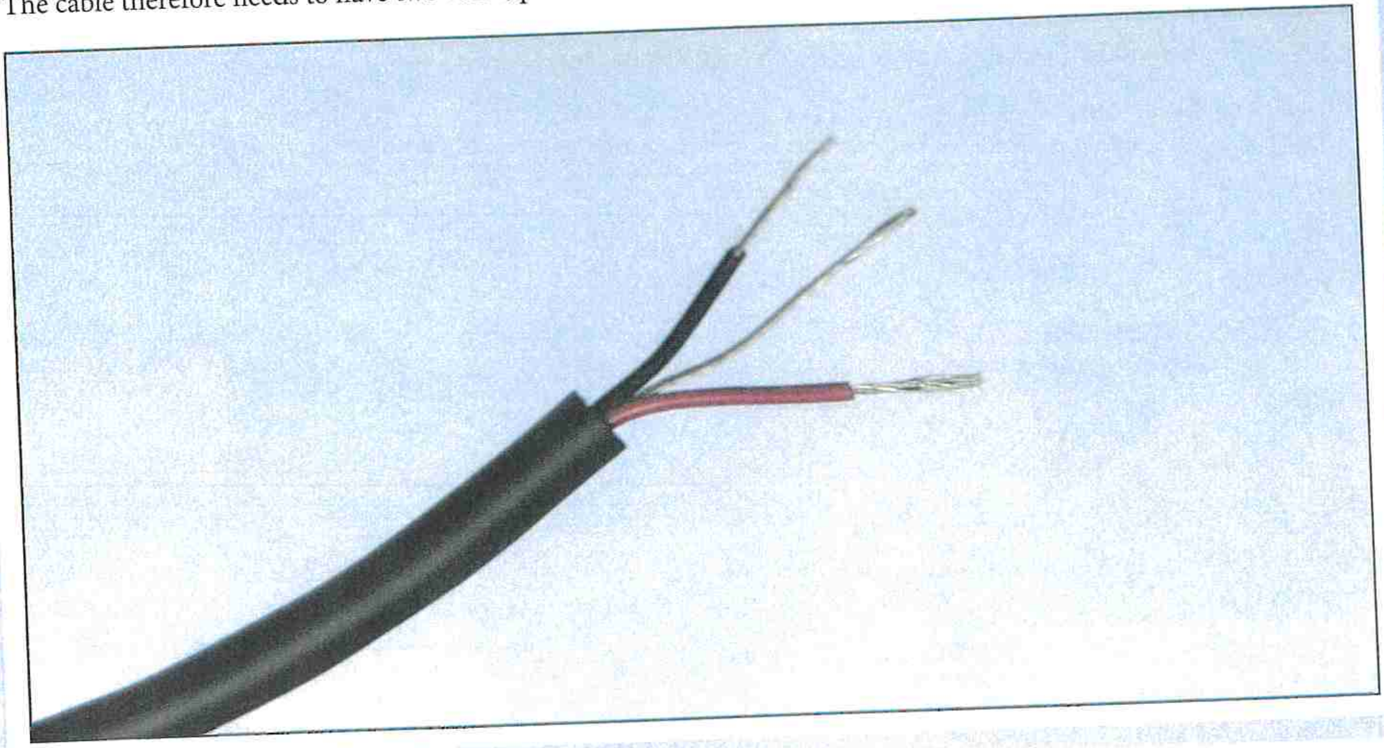


Balanced

Balanced connections use connectors with three points of connection.

These are referred to as the hot (+), cold (-) and screen (or ground/earth).

The cable therefore needs to have two cores, plus the screen.

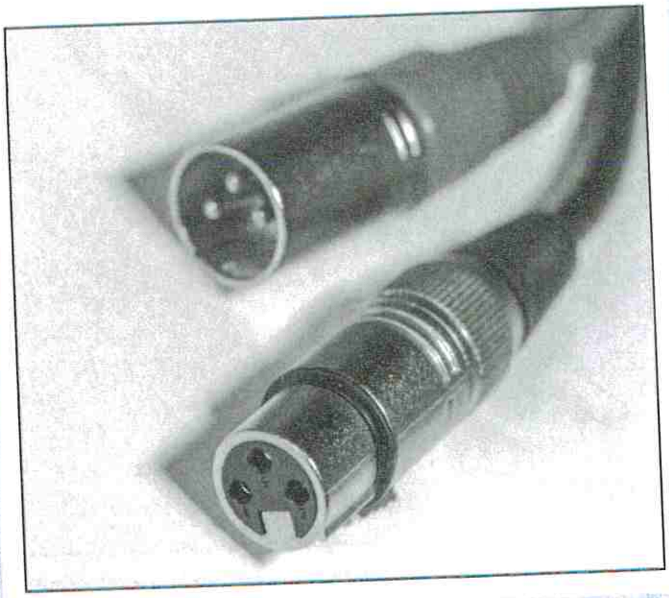


Some connectors may look similar to their unbalanced equivalents, but they will have an additional signal connection. In the case of a balanced jack plug it will have a tip, ring and sleeve.

Tip = Hot
Ring = Cold
Sleeve = Ground

XLR connections can be balanced and use the three pins for the three required connections. In the UK:

Pin 1 = Ground
Pin 2 = Hot
Pin 3 = Cold



Why Use Balanced Audio?

The purpose of balanced audio is to reduce the risk of external interference.

When running cables around a studio, or on location, there are many electrical devices nearby which can have an effect on the audio if the cable runs past them. These might be mains cables causing hum or radio devices causing high frequency crackles.

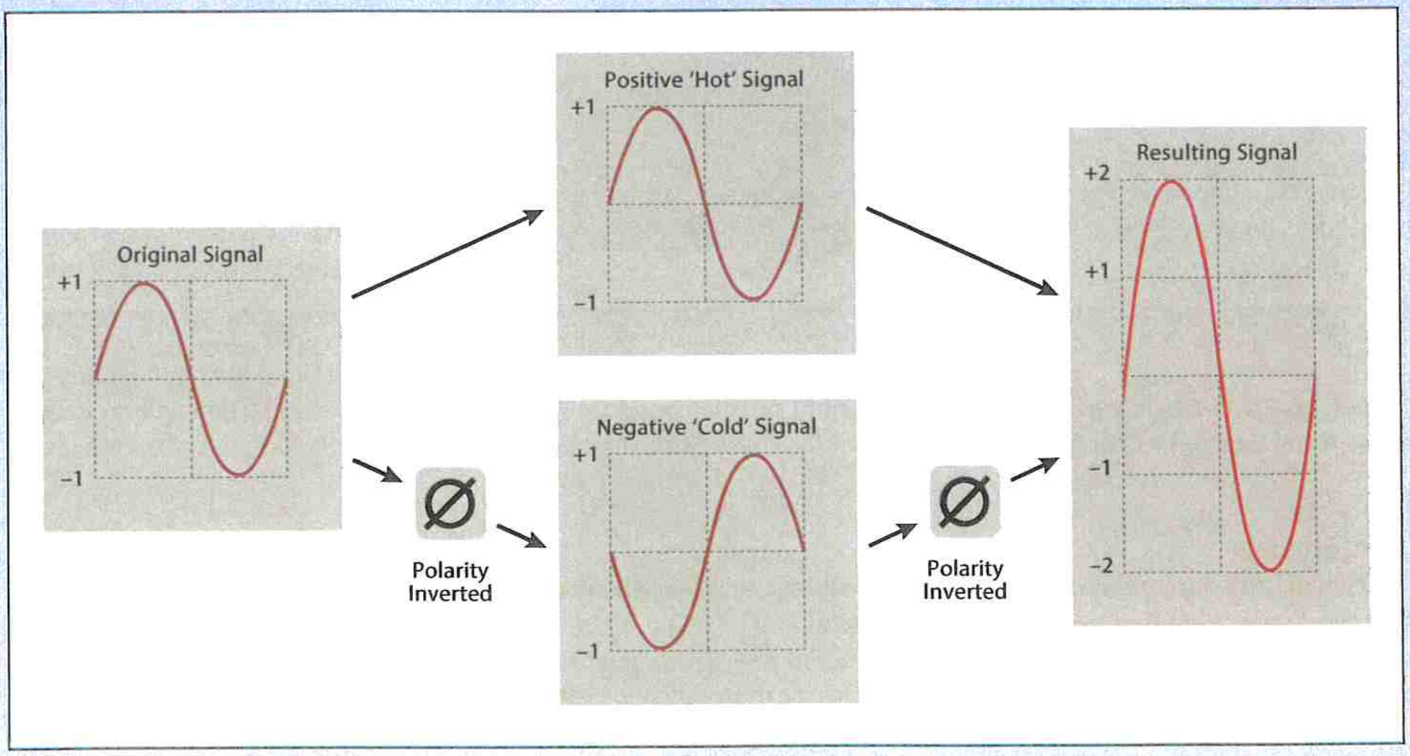
An unbalanced cable uses it's screen to give it some protection but this isn't always enough.

A balanced audio system uses a clever method for eliminating any external interference when it is received at its destination. This means that the cables can be longer and as a side effect the audio signal will be louder.

How Balanced Systems Work

In a balanced audio system the exact same audio signal is sent along two wires (hot and cold), but the polarity of one of the signals is inverted. This happens in the output device, not in the cable itself.

Polarity Inversion

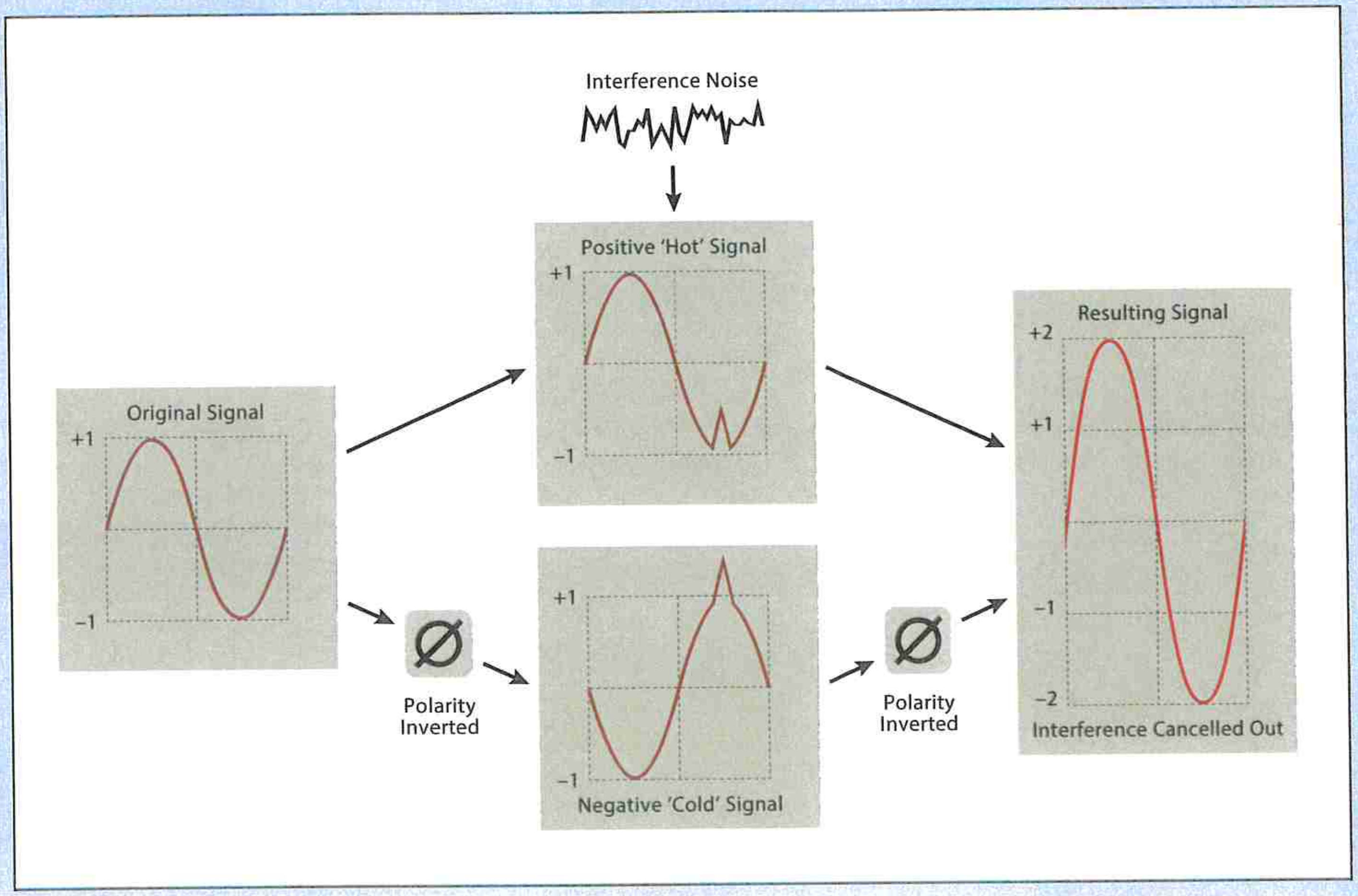


When interference or noise is introduced into the cable, it is introduced equally to both the original and the inverted signal.

When the signal arrives at its destination, the inverted signal is inverted back to it's original orientation and both signals are combined. The wanted signal will now be in phase with itself and will increase in level but the noise will now be out of phase with itself and will therefore cancel out to nothing.

This is called common mode rejection and ensures that any noise in the signal is eradicated.

Canceled Noise



If a fault occurs in a balanced cable and either the hot or cold connection is broken, then the connection will no longer be balanced and the level will drop by around 6dB.

Cloud Storage

'The Cloud' is used to refer to any internet based storage solution. As internet upload and download speeds have increased, this type of service has become more and more useful.

Any kind of file can be stored in the cloud, as long as it can be uploaded via an internet connection.

As the files are accessed via the internet, they can be accessed from anywhere in the world that has a connection. This has revolutionised collaboration in many industries, for example a producer in London can upload a file to their cloud storage and share it with another producer in Los Angeles. Their co-producer can the continue working on the file immediately, and can share it back when ready.

An added bonus is that the storage used by the Cloud service provider will almost certainly be backed up regularly. This means that if you lose a file, they will be able to help you recover it, in some cases almost instantly.

This makes long distance collaboration fast and effective.

However, even a very fast internet connection will not operate quickly enough to stream 24 tracks of high quality audio in real time. Therefore, all files must be downloaded to continue working on them. This can still cause a problem if one of the producer's internet connections is slow.

Digital Audio Data Transfer

AES/EBU
Short for Audio Engineering Society / European Broadcast Union. This is a digital audio data transfer standard, used to transfer a mono or stereo signal. The standard connection used for this purpose is XLR.

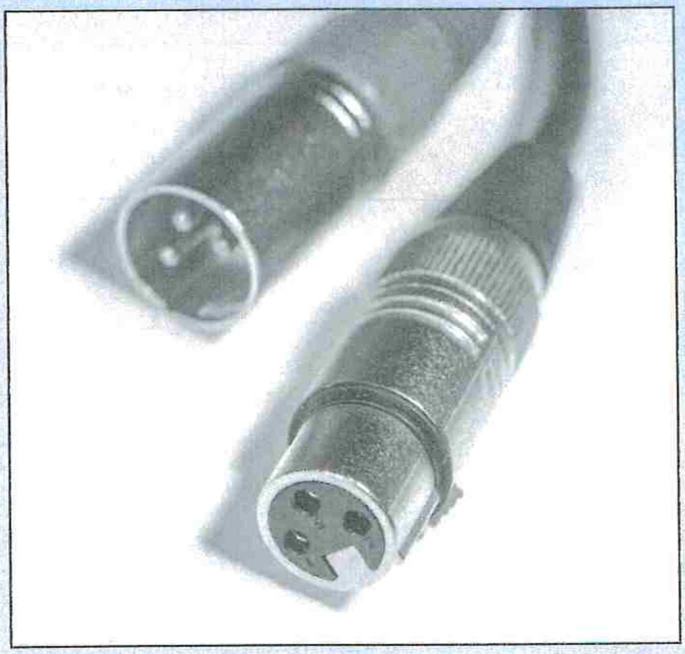
SPDIF
This stands for Sony Philips Digital Interface. It is very similar to AES/EBU but includes a wordclock signal - which reduces the risk of data errors.

ADAT Lightpipe
ADAT is a format created by Alesis in the 1990s and is an abbreviation of 'Alesis digital audio tape'. Its original purpose was to connect digital tape machines but quickly became adopted by other manufacturers for other purposes. ADAT lightpipe is capable of carrying 8 channels of audio along one fibre optic cable.

MADI
MADI is an initialism of 'multichannel audio digital interface' and is a standard agreed by the Audio Engineering Society for the transfer of large numbers of audio channels. It is capable of transferring up to 64 channels of high quality audio using fibre optic cable. It is generally only used in very high specification studios.

Connections

XLR
Common purposes:
Microphones, AES/EBU, professional headphones



Phono/RCA
Common purposes:
Consumer hi-fi connections,
SPDIF (Sony Philips Digital InterFace)

3.5mm Jack / Mini-jack
Common purposes:
Consumer headphones

¼" Jack (Quarter inch jack)
Common purposes:
Guitars, amplifiers, patch cables, consumer headphones

5-Pin DIN
Common purposes:
MIDI (musical instrument digital interface)

D-Sub
Common purposes:
Connecting 8 channels of analogue or digital audio from one device to another. The is a 'D' shaped connector with 24 pins to carry the signal.

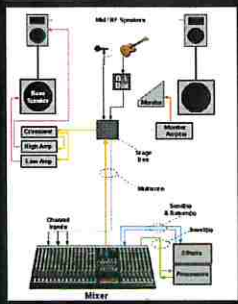
Speakon
Common purposes:
Connecting PA amplifiers and PA speakers.

WHAT DOES A MIXER DO?

A mixer combines(blends) and processes multiple audio signals(INPUT) and sends(or diverts) the signal to a specific location(OUTPUT).

In Live Sound application, all the sound on stage(which is converted by a microphone or DI) is sent to an INPUT CHANNEL on the mixer. The mixer will increase the level(GAIN/PRE-AMP) and process it(EQ and built-in FX). The combined signal is sent(via an OUTPUT BUS) to FLOOR MONITORS for performers and LOUDSPEAKERS for the audience.

In the Studio, the need for a mixer has diminished. AUDIO INTERFACES provide professional quality PRE-AMPS to plug sound sources directly into your DAW without the need of a mixing desk. The advantage of this is that there is less cables and connectivity(which inevitably means less noise). However, having a routing matrix of a good quality mixer will allow you to link various pieces of equipment together(e.g. vintage processors). In addition to this, a digital mixer includes digital connectivity to your Mac/PC giving you added flexibility and convenience.



NB: A SOUND ENGINEER MUST HAVE EXPERT KNOWLEDGE OF MIXERS- MINIMUM REQUIREMENT

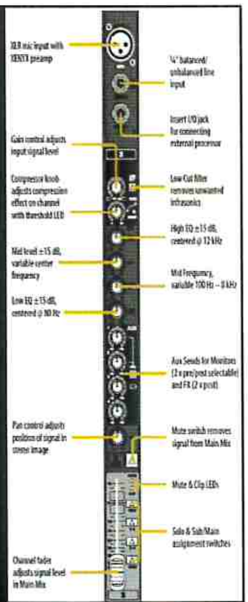
CHANNEL STRIP

Depending on the size on design of the mixer, generally a channel strip can be divided into the following sections:

- INPUT-consist of the physical connectors(XLR and/or TRS Jack), Input Gain(pre-amp), Pad Switch
- EQUALISER-Shelving EQ, Semi-Parametric, Hi-Pass Filter
- AUX SENDS-Sends to Master Aux,Pre/Post fader button
- ROUTING-Fader, Pan, Main Mix and Subgroup buses

Digital Mixers have one set of channel strip controls that is assigned to a selected channel. Analogue mixers have a channel strip for each input channel.

Digital Channel Strip



Analogue Channel Strip

TASK 1: THE CHANNEL STRIP

https://www.youtube.com/watch?v=p_Ch_Cml1-M

Watch the video and complete the channel strip worksheet

MIC AND LINE INPUTS

- These inputs operate and different levels
- MICROPHONE INPUTS accept the low levels of a microphone(-70dBu to 50dBu) and sends it to the pre-amp(gain) which boosts the level(-20dBu to +4dBu). This is usually an XLR input
- LINE INPUTS accept line level equipment(+4dBu). The level is high enough and does not require the extra gain to operate. The line level is a ¼" Jack input and usually bypasses the pre-amp.

Channel Inserts-Unbalanced send and return Jack Socket(Input and Output on 1 connector). These are used to insert extra processor in the signal chain. E.g. compressor. More on that later.....

Instrument/Guitar Level is lower than line level and higher than mic level, therefore, we use a DI(Direct Inject) to much the input impedances so that we operating at the correct level. Often Audio Interfaces have an INSTR or Hi-Z switch so that you can plug your guitar directly into the input without the need of a DI.

Microphones cannot operate on line inputs. The signal is too low. However, line level equipment can operate on a Mic Input if the mixer has a pad switch to reduce the level.

GAIN/ PRE-AMP

Signal LED-Illuminates Green when there's SIGNAL. RED when the channel is overloading i.e. DISTRORTION

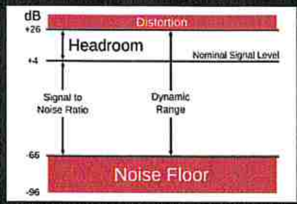
HPF High Pass Filter-Reduces the level of low frequencies below a specific frequency(called the cut-off frequency

Pad Switch
Often mixers have a PAD switch in this section. It reduces the SENSITIVITY of the input channel by a specific level(usually -20dB)

The GAIN increases the level(low-voltage) of the incoming signal and is usually the first pot(potentiometer) on the channel strip. Setting the gain is an integral part of getting a good overall sound. If the gain is too low, you won't hear the signal and quality of the sound will be low as the signal-to-noise ratio is low i.e. less signal more noise. If it's too high distortion occurs. You must get the balance absolutely spot-on. You must spend time in sound checks setting the correct gain level.

Top quality mixers have top quality(noise free) pre-amps.

All audio equipment has a NOISE FLOOR, which means the moment you plug a microphone in the input noise is present in your signal path. The lower your signal when you set your gain, the closer it is to the noise floor(SNR-SIGNAL TO NOISE RATIO) therefore when you increase the signal level you increase the noise too. Hence you have to ensure a good strong signal so that there is more signal than noise present, in addition you have leave sufficient HEADROOM to prevent DISTORTION.



KEY WORD SO FAR….MAKE SURE YOU MADE A NOTE OF THESE

1. PRE-AMP/Gain
2. HPF(HIGH PASS FILTER) or LOW CUT
3. DISTORTION
4. HEADROOM
5. SNR(SIGNAL-TO-NOISE RATIO)
6. NOISE FLOOR
7. PAD SWITCH

Task:

Find the following POTS and SWITCHES on the mixer. What does this tell you about the functionality of the mixer.

GAIN
HPF
PAD

3/13/2020

EQUALISER

EQ processes or shapes the sound of the incoming signal. There are 2 uses for EQ, creative or corrective. Use EQ only when your overall mix is deficient and sound sources require enhancement. Small format mixers have a basic shelf EQ, so use it only if it's necessary.

RULE: CUT before your BOOST. Boosting adds more gain and generally more problems.

Channel EQ is usually SHELVING EQ with a set frequency, divided in high, mid and low. These frequencies(measured in Hz) can be cut(reduced) or boost(increased) by a specified level e.g. -15db cut and 15dB boost.

Bigger format mixers have SWEEPABLE MIDS, which allows you to select a specific mid-range frequency i.e more flexibility to shape sound.

Digital mixers(and music production software) have full PARAMETRIC EQs which allows you to select any frequency to cut and boost i.e. SUPER flexible

AUX SENDS

AUX sends/diverts or duplicates the incoming signal to the Main Aux Sends BUS. An Aux send is effectively another output and has physical connectors(jack or XLR) attached to it.

In Live Sound application, we use an Aux bus to send the signal to the onstage floor monitors or to send the signal to outboard equipment for extra processing. In the Studio, we would use to send a headphone mix to the musicians.

When using outboard equipment the signal that is send to the processor needs to return to the mixer. The processed signal is connected therefore to the AUX RETURN. More about that later...

Small format mixers have 1 or 2 Aux sends. Bigger mixers have 6 or 8 Aux(hence more flexibility).

PRE/POST button

Some Aux sections have a PRE or POST button. PRE stands for PRE fader(or before fader). This means that the signal that is sent to the Main Aux send is not affected by the fader movement i.e your monitor mix is not affected by your FOH mix. POST stands for POST fader, if you turn down the fader, you turn down the send.

3/13/2020

KEY WORD SO FAR….MAKE SURE YOU MADE A NOTE OF THESE

1. AUX SEND
2. EQ
3. PRE/POST
4. SWEEPABLE MIDS
5. SHELVING EQ
6. CUT/BOOST

Task:

Find the following POTS and SWITCHES on the mixer. What does this tell you about the functionality of the mixer.

AUX
PRE
EQ
HI
MID
LOW

3/13/2020

ROUTING SECTION

Once your signal has been processed, you will send the signal to a bus or busses. The controls for these buses are found in the Master Section

Linear Fader-Increases the overall level of the Input channel. Faders are used on mixers as it is easy to see the levels of your various sound sources in relation to each other. This is useful when mixing live.

PAN POT-Sends the input channel to the left or right bus. It places the signal in the stereo field. C position sends the signal in equal amounts to both speakers

MUTE button-Cuts the signal. When the mute button is engaged you cannot hear the audio on that channel.

SOLO button-isolates the channel. When the solo button is engaged only the solo'ed channel can be heard. You can solo several channels.

Routing buttons-sends the signal to an output bus. To send the signal to the FOH Speakers L R(also called STEREO or MAIN MIX) must be engaged. The numbers 1-2/3-4 are used to send the signal to sub-group busses. This is useful if you want to control multiple channels with 1 or 2 faders

PFL-Pre Fade Listen Often mixers have a SOLO/PFL button(or 2 separate buttons). This is useful to solo the track when setting your gains. The fader will not affect the level when it's engaged

Digital mixers have a huge amount of routing flexibility, it therefore can seem complicated. The rule generally is the your want to route(or send) a signal to a specific destination(or output) on your mixer.

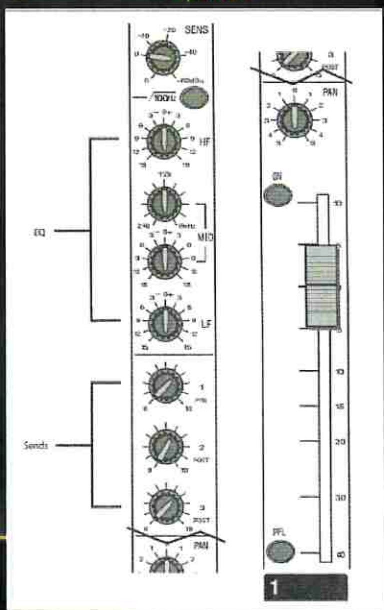
Task: IN PAIRS

DESCRIBE THE FUNCTIONALITY OF THIS MIXER BY EVALUATING THE CHANNEL STRIP

WRITTEN TASK- 16 MARKS

Component 4-style essay question

Figure 1 shows a channel on an analogue mixing desk. Many of the controls are similar to those on a digital audio workstation track. Explain the function of the controls and specifications that can be seen in the picture. Give **one** practical use for each control.



MIXERS SUMMARY

- All mixers have **4, 8, 16 Channels etc**
- **Mixer Inputs** – XLR/J2J inputs for microphones and DI boxes – controlled by input gain
- **Mixer Outputs** – XLR/J2J/Speaker Cable outputs to connect mixer to speakers – controlled by output gain
- **Routing** – process of moving a signal to another location
- **Busses** – group one or more channels together to route to a particular destination i.e auxiliary send or FX
- **Auxiliary Sends** – route a channel/output to a particluar destination i.e monitor or in/ear monitor
- **Group Tracks** – grouping channels either onto a desk or in a logic project to attach identical FX, gains, levels etc – particularly useful on small desks to reduce number of channels needed – all drums to one channel to free space
- **Digital FX** – built in FX i.e Reverb, Compression, Phasing, Delay
- **Talkback** – microphone built in or attached to mixer to link producer to headphones/performer – in ear
- **Foldback** – microphone built in or attached to mixer to link producer to monitors/performer – out loud

End of lesson
Reminders

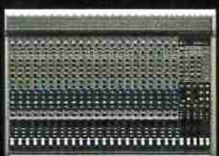
Snap it up

HOMEWORK
DIGITAL VS ANALOGUE MIXERS

HTTPS://WWW.WIREREALM.COM/GUIDES/TOP-10-BEST-AUDIO-MIXERS

Research the spec of the following mixers. Draw up a **comparative spec sheet** and write a **paragraph** on the pros and cons of each mixer.

MACKIE SR24.4 VLZ Pro



MIDAS M32

CONNECTORS

- BALANCED

3 points of contact: hot (+), cold(-) and screen(or ground/earth)

Balance cables reduce the risk of interference and is the connection of choice for all professional equipment. Balance cable can therefore run for longer distances

XLR and TRS Jacks are balanced connectors

- UNBALANCED

Only 2 points of contact and is the simplest type of connection. It consists of a signal cable and a screen.

TS Jacks and phono(RCA) are unbalanced connections

[illegible]

- ANALOGUE

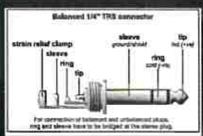
Analogue connections carries an audio signal(voltages) down a balanced or unbalanced cable. The quality of the connector and impedance of the cable impacts on the quality of the signal.

Signal loss occurs when the cable is run over long distances

- DIGITAL

Digital connections carry digital data along a cable. Digital signals provide a cleaner signal with better signal to noise ratio and therefore can be run over long distances. The only downside is compatibility, all equipment must share the same sample rate and clock.

JACK

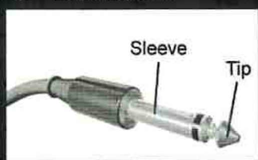


Originally invented for manual telephone switchboards, the standard 1/4-inch plug has been around since the late 1800s and is still the most common type of connectors for musicians. The **TRS** acronym stands for "Tip-Ring-Sleeve," representing the three parts of the connector (Tip = positive, Ring = negative, and Sleeve = ground) and carries a balanced signal. Balanced signals are needed for longer runs of cable, as unbalanced signals can create noise over long distances.

1/8-inch plug or mini-jack are most commonly found on headphones and various commercial audio equipment.

The **TS** (Tip-Sleeve) version is a two-wire, unbalanced version. Due to their similar appearance and physical compatibility, it's easy to confuse a TRS connector with a two-conductor unbalanced quarter-inch plug — and if you're only hearing one side of your stereo field, chances are that's exactly what you've done.

Standard quarter-inch TS plugs are most commonly used for guitar and other electrical instrument cables. The stereo versions are common on headphones, insert cables, and inputs and outputs on audio equipment that don't need long runs of cable to connect — like studio mixers, compressors, and preamps.



XLR

Introduced by Cannon Electric (now part of ITT), **XLR** is sometimes still referred to as a cannon connector, or colloquially as "Extra Long Run," referring to the extra long cables that can be used with this balanced connection.

The connectors are circular, with anywhere from three to seven pins. Most commonly, the balanced three-wire version has been the de facto standard for microphones for many years. Three-pin XLRs are also used for loudspeaker connections, low-voltage power supplies, lighting controllers, and a host of other applications. They also carry digital audio signals through the professional-grade AES/EBU protocol.

AES/EBU DIGITAL XLR

Audio Engineering Society/European Broadcast Union

Digital data transfer standard used to transfer mono or stereo signal.

Audio Engineering Society/European Broadcast Union

RCA

Sometimes referred to as phono or cinch connectors **RCA** is a two-conductor, unbalanced protocol commonly found on "prosumer" gear, as well as on turntables and consumer electronics. The RCA name comes from the electronics giant Radio Corporation of America (now a part of BMG), who introduced the design in the 1940s.

S/PDIF DIGITAL RCA

Sony and Philips Digital interface is similar to AES/EBU but includes a word clock signal reduces the risk of data errors

https://www.youtube.com/watch?v=mR5vD6OFLSs

ADAT LIGHTPIPE

ADAT Lightpipe began on the Alesis ADAT digital multitrack tape machines introduced in 1992. Lightpipe carries eight channels of uncompressed 24-bit digital audio (at a 44.1 kHz or 48 kHz sample rate) over a fiber optic cable, and was originally used to transfer audio between ADAT recorders. While the ADAT recorder itself has found its way into the history books, its Lightpipe format has become a ubiquitous standard, used in digital mixing consoles, converters, audio interfaces, effects devices, and more.

BNC

The **BNC** (British Naval Connection) coaxial protocol is used in the studio to carry **Word Clock**, a signal used to synchronize DAWs, digital tape machines, and other devices connecting via S/PDIF, AES/EBU, ADAT, or other digital audio formats.

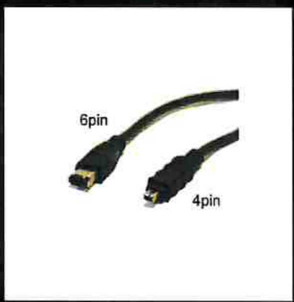
USB

Universal Serial Bus has become an industry standard since its introduction in the late 1990s. It was designed as a connectivity protocol for computers, and is used today for everything from keyboards, printers, smartphones, and hard drives, to audio interfaces, MIDI interfaces, and other audio devices. It's rare these days to find a computer or device that is not equipped with at least one USB port.

FIREWIRE

FireWire is the trade name created by Apple for the **IEEE 1394** high-speed serial bus protocol. Introduced on Apple computers around 1999 (and on some Sony computers as **i.Link**), FireWire is a contender with USB for video and multichannel audio communication, as well as for hard disks and other computer peripherals.

6-pin **FireWire 400** connections can be found on a majority of audio interfaces made over the last decade. A 4-pin FireWire 400 connection used to be common on Windows-based operating systems, but now are not as prevalent



THUNDERBOLT

Developed by Intel in collaboration with Apple, **Thunderbolt** is the latest high-speed cable protocol for connecting computer peripherals such audio interfaces, hard drives, display monitors, video equipment, and more.

MADI

MADI (AES10) is an industry-standard electronic communications protocol that defines the data format and electrical characteristics of an interface carrying multiple channels of digital audio. The main advantages of MADI are that it supports a great number of channels per line, plus transmission of audio signals over 100 meters of coax and up to 2 km of optical fiber with the multi-mode model, or up to 20 km of optical fiber with the single-mode model

D-SUB

Carries digital or audio signals up to 8 channels. It's a 24 pin D shaped connector used by many manufacturers.

SPEAKON

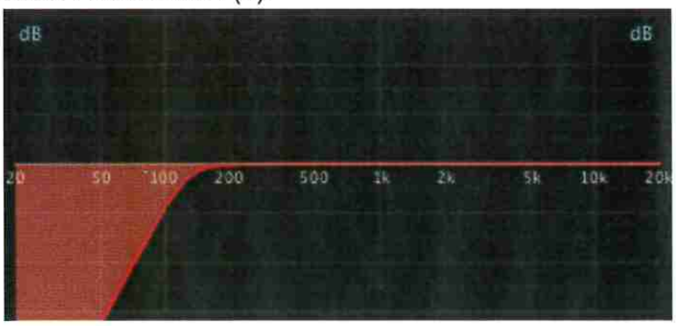
Used in live sound to connect power amplifiers and speakers. Comes in 2 pin and 4 pin varieties.

IDENTIFY AND EXPLAIN THE CONNECTIVITY AND FEATURES OF THIS AUDIO INTERFACE.

How does it connect to a computer? How many inputs and outputs does it have? Types of equipment it can connect to? What is the purpose of the unit? Where might you find this type of equipment?

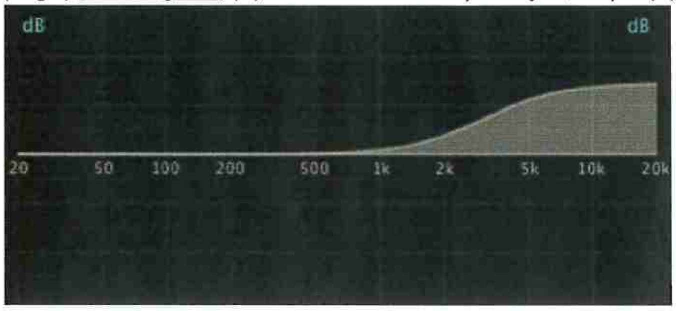
Question Number	Question	Mark
4(b)	Figure 1 shows a selection of leads. Identify and explain features and applications of these leads.	16
	Acceptable Answers	
	In this mark scheme, italics mean that the mark should not be credited multiple times. <u>Underlined technical terms must be spelt correctly</u> All comments must relate to the correct cable in order to gain credit.. The **bold** name of the cable can only be credited if it's clearly linked to a picture. **All cables:** <u>Coaxial</u> (1) <u>shield</u> / <u>sleeve</u> (1) is ground/earth (1) to reduce interference (1) . Signal loss over long runs (1). **(i) ¼ inch / <u>6.3mm</u> / <u>Jack</u> / <u>TS</u> (1)** Guitar / line / DI box input / accept any other valid application (1) *<u>Analogue</u> / <u>analog</u> (1)* *<u>Mono</u> (1)* *<u>Unbalanced</u> (1) so prone to hum (1)* <u>Tip</u> (1) carries the signal (1). **(ii) <u>Phono</u> / <u>RCA</u> (1)** Hi-fi / CD players / DJ mixers / accept any other valid application (1) *<u>Analogue</u> / <u>analog</u> (1)* *<u>Mono</u> (1)* <u>Stereo</u> if there are two cables (1) white/black is left, red is right (1) *<u>Unbalanced</u> (1) so prone to hum/interference (1)* *Also used for digital audio (1)* SPDIF (1) *connecting DAT / Digital Audio Tape / CD audio / PCM / audio interfaces (1) sample rate of 44.1kHz / 48kHz (1).* Also compressed / AC3 / digital audio for DVD / surround (1).	

	(iii) <u>XLR</u> / <u>Cannon</u> (1) <u>Male</u> and <u>female</u> (1). Signal flows from male to female (1). 3 pins: <u>Positive, negative and ground</u> / <u>Hot, cold and neutral</u> (1). Microphones / balanced line level / output of a DI box / accept any other valid application (1) not just 'mixing desk'. *<u>Analogue</u> / <u>analog</u> (1)* *<u>Mono</u> (1)* <u>Balanced</u> (1) *so less noise (1)* over long runs (1). Description of balanced signals: The signal is split into two copies of the signal (1) one positive, one negative / out of phase (1). When noise is introduced in the cable, it's in the same phase in both signals (1). When these are combined the phase is reversed on one signal so that both signals are in phase again (1) so the noise cancels out (1). Carries phantom power / 48V (1) for condenser mics / active DI boxes (1) credit valid description of how this works (1) *Also used for digital audio (1)* AES (1) *connecting DAT / Digital Audio Tape / CD audio / PCM / audio interfaces (1) sample rate of 44.1kHz / 48kHz (1).* Daisy-chained (1). Locking mechanism (1). **(iv) <u>MIDI</u> (1)** Musical Instrument Digital Interface (1) Used to connect synthesisers / accept any other valid application (1) not just 'audio interface'. 5 pin <u>DIN</u> (1) Digital data / not audio (1) Accept any MIDI data command e.g. note on, controller, pitchbend (1). In / out (1). <u>Thru</u> (1) produces a copy of the input (1) so equipment can be <u>daisy-chained</u> (1). Superseded by USB (1). Credit references to 1970s consumer audio leads (1).	

Question Number	Question	Mark
4(b)	Figure 1 shows a channel on an analogue mixing desk. Many of the controls are similar to those on a digital audio workstation track. Explain the function of the controls and specifications that can be seen in the picture. Give **one** practical use for each control.	**16**
	Acceptable Answers	
	In this mark scheme, italics mean that the mark should not be credited multiple times. Underlined technical terms must be spelt correctly **Sens** *Gain* (1). Not 'volume'. Pre-amp (1) *U = unity gain (1)* Set to achieve a good signal to noise ratio (1). Too high will be distorted (1). Too low will introduce noise (1) Credit reference to different impedance/levels of sources (1). **For each EQ/filter graph**: Correct shape (1) Labelled axes should be credited for one graph only. x-axis: Hz/Frequency (1). Appropriate numbers ranging from 20 to 20k (1) y-axis: dB/volume/gain / appropriate numbers: e.g. +-15 (1). **100Hz** HPF / High pass filter / low cut filter / cuts low frequencies (1). Credit any valid example of use: e.g. reduce plosives / reduce proximity effect / rumble filter (1). 	

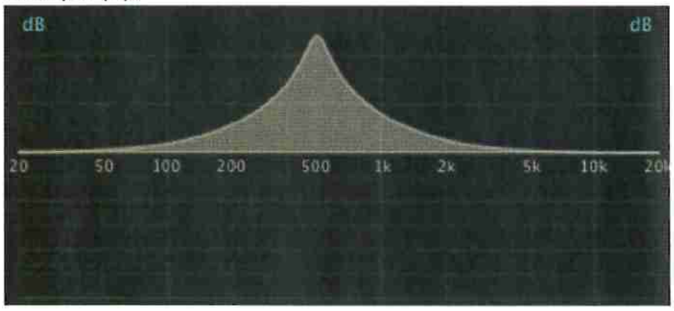
EQ
Equalisation (1) Boosts or cuts (different frequencies) (1)
Gain (1).

High frequencies (1)
(High) *shelving EQ* (1). *Fixed cutoff/frequency/Q/slope (1).*



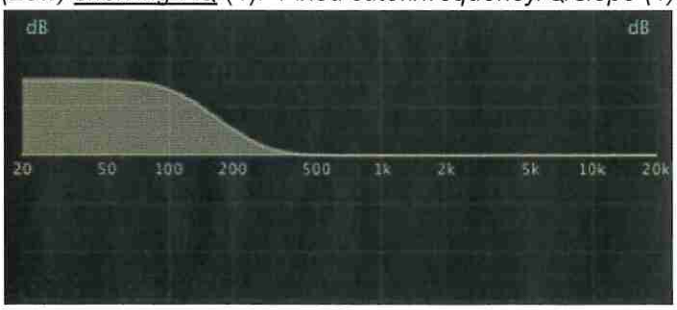
Credit any valid example of use: e.g. bring something (vocal / reverb / acoustic guitar) forward in the mix / clarity on overheads / correct lack of HF response from dynamic mics / gentle high-shelf boost for mastering / loudness curve / increases perceived loudness (1)

Mid (no mark available)
Sweepable/selectable (centre) frequency (1). Bell / peak (1). *Fixed Q/slope* (1).



Credit any valid example of use: e.g. boost at around 2kHz to bring out the beater of a kick drum / mid-band scoop on a distorted electric guitar / reduce LM to make mix less muddy / reduce sibilance (1)

Low frequencies (1) [don't credit if already credited 'high frequencies']
(Low) shelving EQ (1). *Fixed cutoff/frequency/Q/slope (1).*



Credit any valid example of use: e.g. more bassy kick drum / more bassy bass guitar / loudness curve / increases perceived loudness (1)
Accept answers for HPF if not already credited and specific reference to attenuating LF.

	Sends Effects / foldback (1) [general not linked to pre/post] The signal is sent to separate hardware/aux track/bus track / The wet signal will return to a separate channel (1). Bus/group several tracks together (1) <u>*Pre-fade*</u> (1). *The volume of the send is not affected by the fader /before the fader in the signal path/independent of fader* (1). Credit any valid example of use: e.g. monitor mixes / fading the dry signal of the track leaving the reverb behind making parts move backwards and forwards in the mix (1). <u>*Post-fade*</u> (1). *The volume of the send is affected by the fader* (1). Credit any valid example of use: e.g. adding effects like reverb, chorus, delay etc (1). **Pan** Stereo field / left and right (1) by adjusting the amplitude of the two sides of a stereo output (1). Credit any valid example of use: e.g. pan guitar to the left to separate from the lead vocal in the centre / opposite pan a stereo pair of microphones / opposite pan double tracked guitars / pan instruments to resemble where they would appear on a stage (1). **On** Can be used to mute the channel (1), lower noise (1), prevent feedback (1). Routes/assigns (1) the signal to main outputs/L-R (1). **Fader (1)** Volume <u>output</u> (1). Credit any valid example of use: e.g. to balance the tracks / riding the faders to make a fade out (1). Logarithmic scale (1) Greatest sensitivity to movement around 0 (1) *0 = unity gain (1)* **PFL** Pre-fade listen (1). Solo (1). *The volume of the PFL is not affected by the fader /before the fader in the signal path/independent of fader (1). Unity gain (1).* Credit any valid example of use: e.g. check for problems with this track (1).	

(Total for Question 4 = 16 marks)