Notes

9

AUDIO CONSOLE

9.1 INTRODUCTION

Previously, you studied about various sound equipments such as microphones, loudspeakers etc. As understanding these equipments is very important for the sound technician, therefore, here, you will learn about audio mixing console, their different parts or sections, working and their uses etc. You will also learn about different types of audio consoles, i.e., analog and digital, with their respective layout of controls, switches and operation etc.

9.2 OBJECTIVES

In this lesson, you will be able to

- explain the function and signal flow of an audio mixing console
- define and compare different types of mixing consoles
- identify different parts/sections of typical analog/digital console and explain its different controls with their usage

9.3 A SIMPLE ANALOG MIXER/ (STEREO) MIXING CONSOLE

A simple analog audio mixer combines several incoming signals into a single output signal. This cannot be achieved simply by connecting all the incoming signals in parallel and then feeding them into a single input, because they may interfere with each other. The signals need to be isolated from each other providing individual control of at least, the level of each signal.

Practically speaking, mixing consoles not only allow simple mixing but they can also provide phantom power for condenser/capacitor microphone, pan-control (where by each signals can be placed in any desired position in the stereo field), filtering, equalization, routing and monitoring facilities which enable routing any number of sound sources to a desired loud speaker for listening without affecting the mixer's main output.

Let us discuss a simple six channel analog audio mixer. This particular mixer will have six inputs and two outputs. Professionally we called it as a six-into-two (6:2) mixing console. The inputs will usually be XLR, TRS (tip-ring-sleeve) or balanced and TS (tip-sleeve) or unbalanced. According to the input source (microphone or line) the connectors vary (see Fig. 9.1).

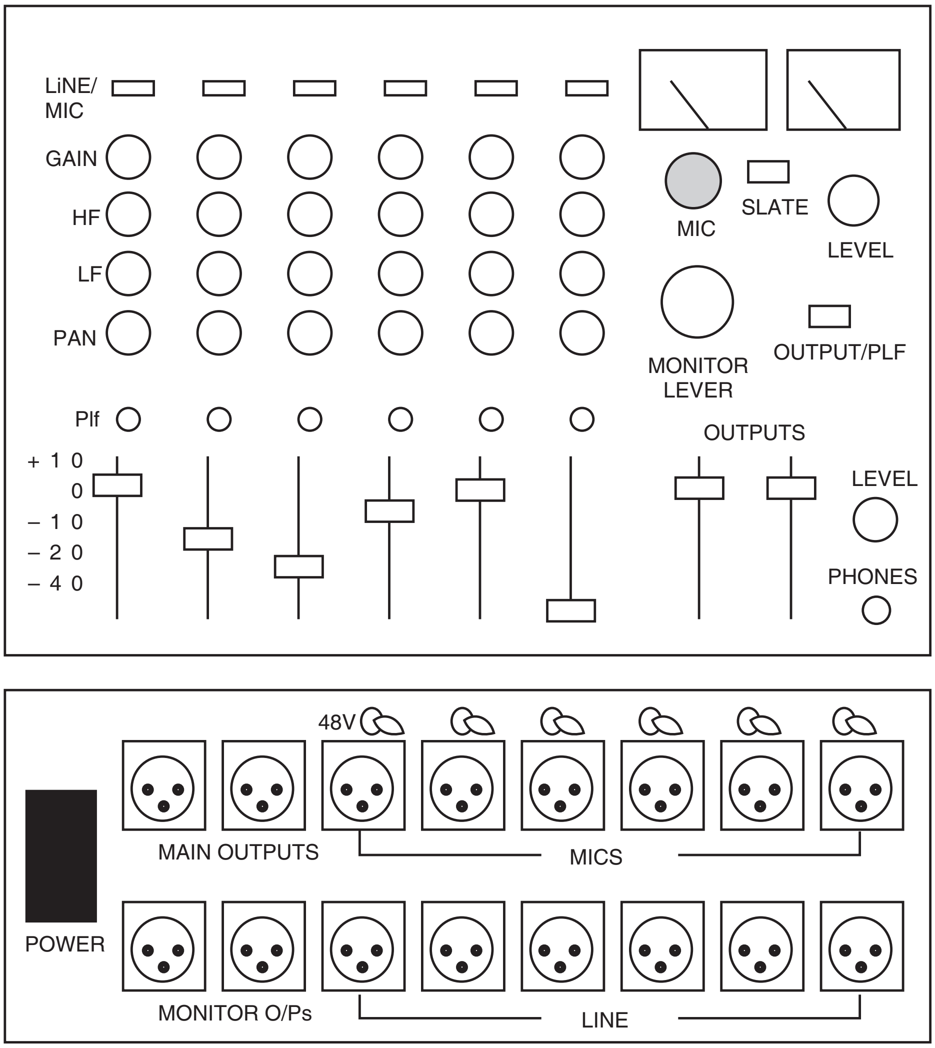

Fig. 9.1

The outputs are also in three pin XLR Type connectors. The outputs are XLR-Male connectors having three pins. The inputs will be having XLR- Female connectors. There is a phantom power of +48V (DC) Switch for Microphone selection, if and when required. If we are using condenser Microphone we need to supply the microphone with a +48V DC current from the Audio Mixer to operate the microphone. The signal received from this microphone will then be routed to a stereo or master bus.

Bus: A bus is a section in the signal path of an audio mixer which is used to combine different audio sources and deliver them as a whole to a specific destination.

Let us take a look at the signal flow in a simple analog audio mixer through its different sections.

9.3.1 Input Section

In the signal flow, the first section is the input gain control (rotary fader type) which is commonly known as Pre – Amp (pre-amplification). This control adjusts the degree of amplification provided by the input amplifier and is labeled in decibels (dB) either increasing or decreasing steps. Continuously variable inputs are normally switchable between Microphone and line position, depending upon the output level of the microphone or line connected to the channel input. We have to choose the 'Microphone' or 'line' input as per requirement.

For a 'microphone', high amplification is required as the microphone input is low and for 'line' inputs, little amplification is used and the gain control normally provides adjustment either side of unity gain (0 dB) perhaps (±20dB). This process of controlling the input levels is called as input gain control.

9.3.2 Equalization

Then the next section is equalization. This section will have controls for two frequency bands (in the case of the figure provided), the high and the low frequency. Boost and cut of around ±12 db over low and high frequency bands is available. In this section we can control the tone of the signal through boosting and cutting of the high and low frequencies.

9.3.3 Channel Fader

The last control of the input section is the channel fader which controls the overall level of the channel. It provides a small amount of gain (up to 12 db) and infinite attenuation (decrease). The fader control is specially designed for the purposes of level control

There are two types of Faders:

1. Rotary Faders. For example – Input gain control, equalization, etc.
2. Straight Movable Faders or Channel Faders. For example – stereo bus, etc.

9.3.4 Pan Control

Pan control on a Mixer is used for placing a signal anywhere between left or right in the stereo field. It works by splitting a single signal from the input into two signals.

These two outputs of the pan-pot, usually feeds the left and right channels of the stereo Mix bus. Also the signal can be placed in the centre which results in equal level in both L and R and hence no change in the perceived level.

Only 18 dB of level difference, is required between left and right channels to give the impression that a source is either fully left or fully right in a loud speaker stereo signal. But most pan-pots are designed to provide full attenuation of one channel when rotated fully towards the other, thereby changing the levels as shown in the Fig. 9.2.

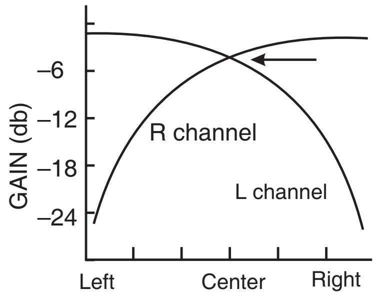

Fig. 9.2

9.3.5 PFL/Pre Fader Listening

This is a facility which provides a signal to be monitored without routing it to the main outputs of the mixer. It also provides a means for listening to a signal in isolation in order to adjust its level or EQ.

A PFL switch on each channel routes the signal before reaching the channel fader to the PFL bus. There is also a master PFL switch which switches the mixer's monitor output to monitor the PFL bus as shown in the Fig. 9.3.

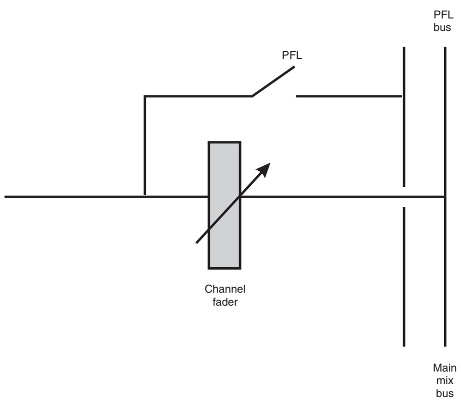

Fig. 9.3

In case of high quality audio mixing and live console a separate small PFL loud speaker is provided on the mixer itself so that the input signal can be checked without affecting the main monitor.

PFL has great advantages, in live work and broadcasting since it allows the engineer to listen to sources before they are sent to the master fader. It can also be used in studio recording to separate any source from all the others, without cutting all the other channels to adjust equalization and other processing with greater ease.

9.3.6 OutPut Section

The two main output faders control the overall level of all the channel signals which have been summed on the left and right mix buses. The outputs of these faders, feed the main output on the back panel of the mixer. The monitor gain control adjusts the loudspeakers output level, without affecting the main line output level but if we make any change in the main Fader gain, it will affect the monitor output.

There is an option of slate, which is used by the sound engineer/ sound operator and comprises of is a small microphone, mounted on the audio mixer, which is routed to the main outputs, so that comments from the engineer or operator (such as take number, announcement) can be recorded on a tape machine connected to the main outputs. There is a rotary level control to adjust the slate level.

Signal path from channel input to main output on a simple mixer is shown in Fig. 9.4.

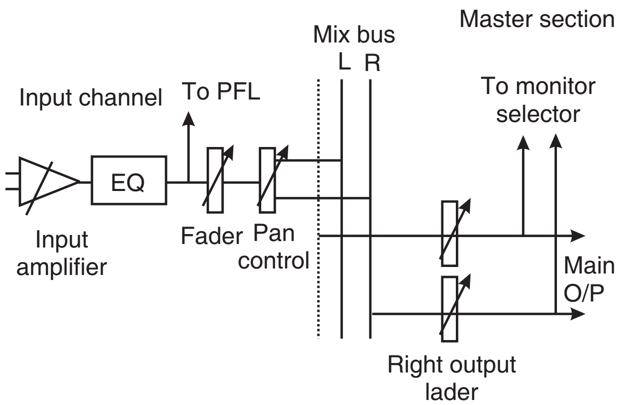

Fig. 9.4

Notes

INTEXT QUESTIONS 9.1

Choose any one of the following:

1. In a mixer, the first section of the signal flow is

   (a) Equalization (b) Input gain control

   (c) Routing (d) buffer

2. Equalization can control the .... of the signal through boost or cut of frequencies

   (a) Volume (b) Level

   (c) Tone (d) amplitude

3. .... is specially designed for the purpose of level control

   (a) Fader (b) Pan

   (c) Pad (d) Slate

4. In a mixer .... is used for placing of an audio signal to left, right or centre

   (a) Fader (b) Pan

   (c) PFL Switch (d) Potentiometer

5. The option used by sound engineer/operator to record any comments on tape is

   (a) Bus (b) Slate

   (c) Pan (d) Mixer

6. PFL or Pre Fade Listening has great advantages in which type of work

   (a) Live work and broadcasting

   (b) Film Production

   (c) Radio broadcasting

   (d) Mixing

7. Phantom power in microphone is .... volt DC

   (a) +45 (b) +48

   (c) +50 (d) –48

9.4 CONCEPT OF MULTI-TRACK MIXING (Multi -track Mixer)

Music recording generally requires two distinct stages.

1. Track laying
2. Mix-down

1. **Track laying**: The Musical tracks are recorded on a Multi-track Recorder separately. These include background tracks, rhythm tracks followed by lead tracks and vocals.
2. **Mix down Stage**: All the recorded tracks (Vocal and Music) are played back through the Mixer and combined into a Stereo or Surround Mix, to make the finished product, which is made into a Commercial Release.

In Multi-track Recording, two signal paths are processed. One from Mic/Line Input and another is from the Multi-track recorder. These two signals are then sent to the Stereo Mix down (Monitor path). Fig. 9.5.

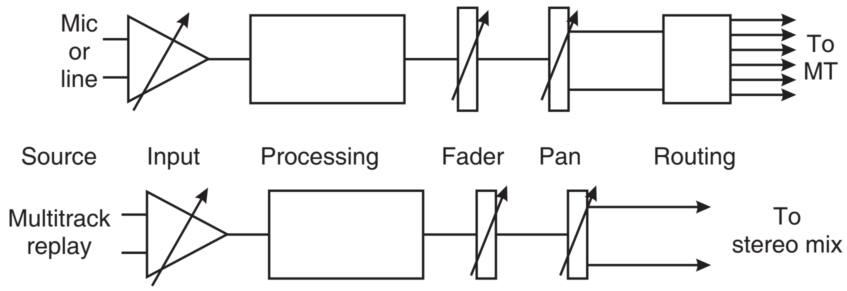

Fig. 9.5

We record the microphone signal into Multi-track, while also mixing the return signal from the multi-track recorder into one stereo signal, so that the Sound Engineer or Operator can hear what the final product will sound like. If any kind of overdubbing is required, then that has to be done at this stage. This stage is called "Track Laying".

Then comes the Mix-down stage which forms the basis of Stereo Mix down.

There will be two signal paths, one from the microphone or line source to the Multi-track Recorder (called as the channel path) and one from the Multi-track recorder back to the Stereo Mix (called as the Monitor Path).

Some basic signal processing such as Equalization, will be required in the channel path to the multi-track recorder. Whereas more signal processing features are usually applied at the time of the Mix down. In some cases like live Broadcasting, basic processing is used during the live work rather than the mix down stage.

9.4.1 Grouping

Grouping means simultaneous control of more than one signal at a time. It usually means that one fader controls the level of a number of slave channels.

It is usually used for reducing the number of Faders that the sound Engineer/ Operator has to handle. This is applied when there are more number of faders to be operated. So we need to make a group of faders and control by one fader.

There are two types of grouping:

1. Audio Grouping

2. Control Grouping

Audio Grouping: Audio Grouping means making a single audio output to take a number of channel inputs. A single fader controls the level of the summed signal & there will be a group output from the Console. Fig. 9.6.

The Stereo mix output from the Console is an effective audio group, one for the left and one for the right, as they constituted a sum of all the signals routed to the stereo output and includes the overall level control. Some older consoles will have routing buttons on top of each channel module.

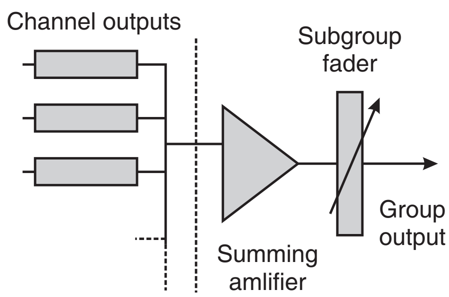

Fig. 9.6

The Master Fader, for audio groups, will be in the form of four or eight faders in the central section of the console (Fig. 9.7). They can be arranged in such a way that we can pan a channel between odd and even groups and it would be common for two of these groups. It is common for eight audio group faders to be used as subgroups themselves having routing to the stereo Mix so that the channel signal can be made more easily manageable by routing them to a subgroup & here to the main – mix via a single level control.

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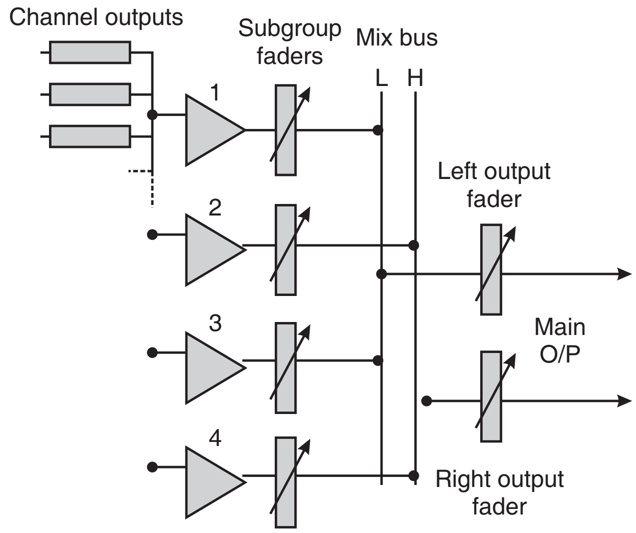

Fig. 9.7

From Fig. 9.7 without pan control the subgroup 1 and 3 goes to the left Mix bus and 2, 4 goes to the Right Mix Bus.

Control Grouping: Control group is different from audio grouping, because it doesn't give rise to a single summed audio output for the group. The levels of the Faders in the group are controlled from one fader. But these outputs remain separate. Generally its effect is to a large hand moving many faders at the same time. Each Fader maintaining its own level in relation to the others.

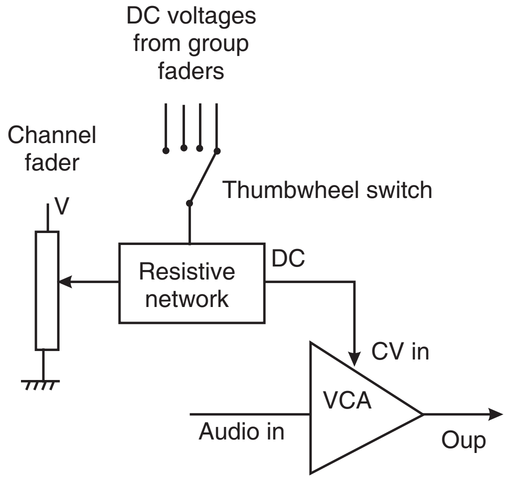

Fig. 9.8

Control group is controlled by voltage controlled amplifier (VCA). Its gain is controlled by a DC voltage applied to a control pin.

In VCA, Fader audio is not passed through the fader itself but is routed through VCA, so the Fader carries DC instead of Audio and the audio is controlled indirectly (Fig. 9.8).

The latest alternative to the VCA Fader is the DCA Fader (Digitally Controlled attenuator) whose gain is controlled by digital values (binary) instead of DC voltage. This is easier to implement in digitally Controlled Mixer.

Normally there are dedicated VCA group master Faders in non automated system, it will control the overall levels of any channel faders assigned to them. The channel audio outputs would normally be routed to the main mix directly and the grouping affecting the levels of the individual channels in their mix.

INTEXT QUESTIONS 9.2

Choose any one of the following:

1. How many signal paths are processing in multi-track recording?

   (a) One (b) Two
   (c) Three (d) Four

2. Signal processing features are usually applied more in which stage?

   (a) Track lying stage (b) Mix down stage
   (c) Multi-track recording (d) Background track recording

3. Control grouping is controlled by

   (a) DC voltage
   (b) AC voltage
   (c) VCA (Voltage Controlled Amplifier)
   (d) None of above

9.5 DIGITAL MIXER

A Digital Mixer would normally comprise of number of similar sections/modules. The main section would comprise of number of Channel strips for control, processing and monitoring of each channel .

The digital audio mixer will have eight sections in its channel strip

1. Input section
2. Routing Section
3. Dynamic section
4. Equalizer section
5. Channel & Mix Controls
6. Auxiliary section
7. Master Controls
8. Metering Section

Channel Strip: Typical layout of a channel strip is shown in Fig. 9.9.

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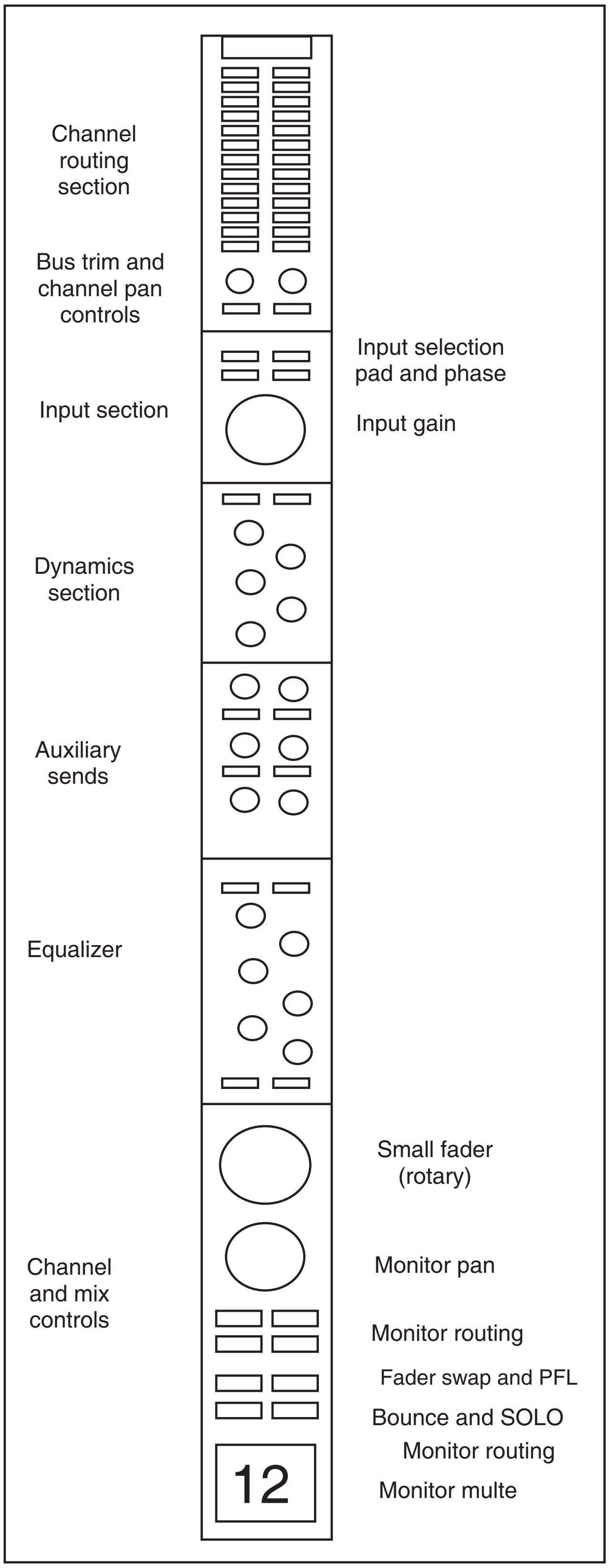

Channel routing section
Bus trim and channel pan controls
Input selection pad and phase
Input section
Input gain
Dynamics section
Auxiliary sends
Equalizer
Small fader (rotary)
Monitor pan
Channel and mix controls
Monitor routing
Fader swap and PFL
Bounce and SOLO
Monitor routing
12
Monitor multe

Fig. 9.9

9.5.1 Input Section

This section includes the following components:

1. **Input Gain Control :** It sets the Mic or line input amplifier gain to match the level of the incoming signal, this control is often a coarse control in 10 db steps, sometimes accompanied by Fine trim, steps of 5dB or 10dB make for easy reset of control to an exact gain setting, and precise gain matching of channels.

2. **Phantom power**: Many professional Mics require 48V Phantom powering and, sometimes switch on the module to turn it on/off . Occasionally this switch is re-attached on the console, may be in a central assignable switch panel.

3. **MIC/Line Switch**: It switches between the channels, MIC input and Line input. The line input could be the playback output from a tape machine. On another line, signal such as synthesizer, guitar (electronic) or sound effect devices.

4. **PAD:** usually used for attenuating(decreasing) the high level mic input signal by something like 20db, for situations when the mic is in from a kick drum or when the mic is in the field of high sound pressure. For example its output may be so high as to cause the mic input to chips, requiring that the pad be used on some occasions.

5. **HPF/LPF** : High Pass Filter/ Low Pas Filter can sometimes be switched in at the input stage to pass only high frequency or low frequency component of the signal. These can be used to filter out unwanted rumble. Filtering rumble at this stage can be an advantage because it saves clipping later in the chain.

9.5.2 Routing Section

This section includes the following components/ Controls:

1. **Track Routing Switches**: The number of Routing Switches depends on the Console; some will have 24, some 32 & some 48. The switches route the channel path signal to the multitrack machine, and it is possible to route a signal to more than one track, the track assignment is often arranged as panes of track, so that odd and even track can be assigned together with a pan-pot used to pan between them as a stereo pair.

2. **Mix Routing Switches**: Sometimes there is a facility for routing the channel path output signals to the main monitor Mix, or to one of perhaps four output groups and these switches will often be located along with the track routing.

3. **Channel pan switch**: used for panning channel signals between odd and even tracks of the multitrack in conjunction with the routing switches.

4. **Odd/Even/Both switch:** This switch will determine whether the signal is sent to the odd channel only, the even channel only or both (In which case the pan control is operative)

5. **Direct switch:** used for routing the channel output directly to the corresponding track on the multi-track machine without going via the summing buses. This can reduce the noise level from the console since the summing procedure used for combining a number of channel output to a track bus can add noise of a channel, is routing directly to a track no other signal can be routed to that track.

9.5.3 Dynamic Section

Some advanced consoles incorporate dynamics control on every module. So that each signal can be treated without resorting to external devices. These normally incorporate compressor and expander sections, which can act as limiters and gates respectively. If required, system allows that EQ to be placed in the side chain of the dynamic section also, providing frequency sensitive limiting among other things. It is usually possible to link the action of one channel's dynamics to the next in order to "gang" stereo channel so that the image doesn't shift when one channel has a sudden change in level while the other doesn't. When dynamics are used on stereo signal it is important that left and right channels have the same setting otherwise the image may be affected.

9.5.4 Equalization Section

The EQ section is usually split into three or four sub sections, each operating on a different frequency band and to have similar functions. These will be described in general here.

(a) HF, MID1, MID2, LF

A high frequency band, high Mid, Low Mid and Low frequency band equilizations are often provided. If the mode is parametric, these bands will allow continuous variation of respective frequency 'Q' and boost/cut. If not parametric, then there may be few switched frequencies for the Mid frequency band and perhaps a fixed frequency for LF and HF bands.

(b) Peaking/Shelving or Bell

Often provided on the upper and lower bands for determining whether the filter will provide boost/cuts over a fixed band (whose band width will be determined by 'Q') or whether it will act as a shelf with the response rising on nothing off above or below a certain frequency.

(c) 'Q'

The 'Q' of a filter is defined as its center frequency divided by its bandwidth (the range between frequencies where the output of the filter is 3db lower than the peak output) in practice this affects "the sharpness" of the filter peak or notch, high 'Q' giving the sharpest response and Low Q giving a very broad response.Fig. 9.10.

Low 'Q' would be used when Boost/Cut over a relatively wide range of frequencies is required.

While high 'Q' is used to Boost/cut one short specific region

(d) Frequency Control

Sets the center frequency of peaking filter or the turn over frequency of a shelf

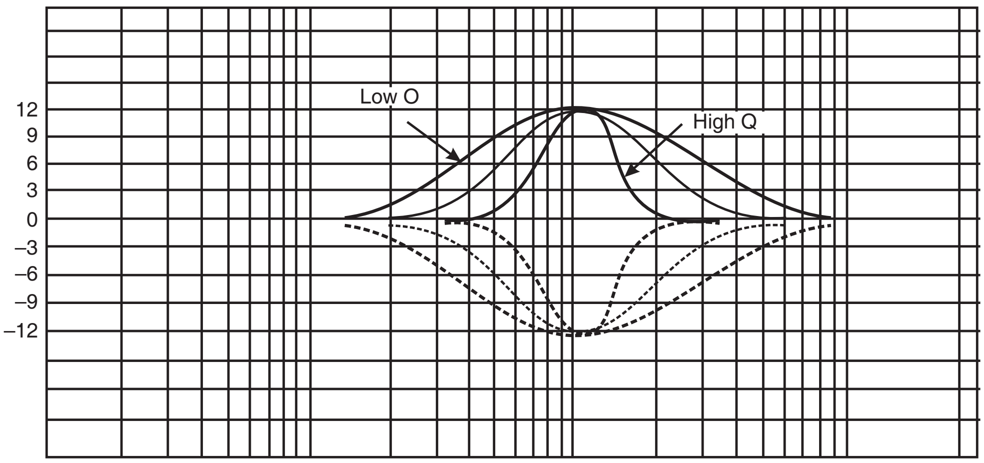


Fig. 9.10: (Equalization Section)

(e) Boost/Cut

Determines the amount of boost or cut applied to the selected band usually up to maximum around ±15db

(f) HPF/LPF

Sometimes the high and low pass filters are located here instead of at the input. In addition they normally have a fixed frequency turnover point and a fixed roll-off of either 12 or 18db per octave. These will often operate even if the EQ is switched off

(g) In/Out

Equalization circuits can introduce noise and phase distortion, so they are best switched off when not required, by input cut off switch.

9.5.5 Channel and Mix Control Section

This section generally has following controls/ components

(a) Pan

This control is a continuous rotary & knob and is used to place the signal of that channel in any desired position in the stereo picture.

(b) Fader Reverse

Swaps the faders between mix and channel paths, so that the large fader can be made to control either the mix level or the channel level.

Some systems defeat any fader automation when the large fader is put in the channel path. Fader reverse can often be switched globally and may occur when the console made is changed from recording to mix down.

(c) Line/Tape or Bus/Tape

In Line or Bus mode the monitor paths are effectively 'Listening to' the line output of the console's track assignment buses while in 'Tape' mode the monitor paths are listening to the off tape signal.

(d) Bus or Monitor Bus

It routes the output of the monitor fader to the input of the channel path (channel fader) so that the channel fader will be used as a post-fader effects send to any one of the multi-track buses.

(e) Mute or cut

There are two types of cut switches one for cutting the channel signal from the multi-track send the other for cutting the Mix signal from the mix.

(f) PFL

Pre fade listen is the signal monitoring without pass through fader, the signal coming from the source without routing to fader.

(g) AFL/Solo

After fade listen is similar to PFL this is sometimes called as solo, which routes a panned signal of the track to the main monitor, cutting all other signals, these functions are useful for isolating signals at the time of setting. In most of consoles, the AFL Bus will be stereo. Solo functions are useful when applying EQ and effects, one may hear the isolated sound and treat individually without hearing the rest of Mix.

Notes

INTEXT QUESTIONS 9.3

Choose any one of the following:

1. Signals coming from tape recorder or CD player will be fed to which input of the mixer?

   (a) Microphone input (b) Line input

   (c) Aux input (d) Metring input

2. .... is used for removing of rumble or hiss from signal.

   (a) Filter (b) PAD

   (c) Phantom power (d) Live swich

3. Compressor and Expander coming under which section of the audio mixer?

   (a) Input (b) Output

   (c) Master control (d) Dynamic

4. Generally equalization explained in to .... or .... section

   (a) One or Two (b) Two or Three

   (c) Three or Four (d) One or four

5. The maximum boost or cut applied to a selected band is

   (a) Plus minus 10 (b) Plus minus 15

   (c) Plus minus 20 (d) Plus minus 25

6. AFL or after fade Listening is also called?

   (a) Pan (b) Pot

   (c) Solo (d) Aux

9.5.6 Auxiliary Sends

Auxiliary sends referred as Aux Sends, are taking the signals from either the channel or Mix paths and appear as outputs from the console which can be used for fold back to the vocal booth musicians, effects ends, cues and so on.

Each aux will be a master gain control, usually in the centre of the console for adjusting the overall gain of the signal, sent from the console.

Aux sends are often a combination of mono and stereo buses Mono sends are usually used as routes to effects, while stereo sends may have one level control and a pan control per channel. The no of Auxiliary sends depends on the console, but there can be up to ten on an ordinary console.

(a) ***Aux Sends****:* It controls for the level of each individual channel in the numbered aux mix.

(b) ***Pre/Post switch****:* It determines that whether the signal send in taken off before or after the fader. If it is before, then the send will still be live, even when the fader is down, effects sends will normally be taken post fade.

(c) ***Mix/Channel****:* Determines whether the send is taken from the Mix or channel paths: it will often be sensible to take the send from the channel path, when effects are to be recorded on to multi-track rather than on to the Mix. This function is labeled 'WET'

(d) ***MUTE****:* It will cut the numbered send from the aux mix.

9.5.7 Master Control Section

This section normally placed on the right hand end of the Audio Console or now, in the digital Mixers, it is on the middle of the Console. It has following facilities:

(a) ***Monitor Selection***: Monitor selection means it will feed the signal to the loud speaker of the control Room/Studio but not the mix output, there are many switches to select the source to be monitored like. Aux sends, the main stereo mix on Tape Machines etc.,

(b) ***DIM****:* It decreases the level of the signal feed to monitor around 40db, for quick silencing of the room.

(c) ***Record/Overdub/Mix Down****:* This Facility, depends the mode of operation, the Mic/Line input switching, large and small faders and auxiliary sends. This will over write the signal on one another or it will dub the signal on an appropriate position (between memories we have given)

(d) ***Auxiliary Level Control****:* This is the master control for setting the overall level of each Aux send output.

(e) ***Talkback or Fold back****:* Talkback is usually placed on the console, having a small built in Microphone, which is usually used for giving instruction to the studio hands (instrumentalist) from the control room and which can be routed to a number of destinations like Aux sends, Mix bus or studio loud speakers etc.,

(f) ***Oscillation****:* For analog tape recording, we need the magnetic tape to be given bias, which is a lining/ signal of frequency of accurate 1khz and 10khz tone. The 10khz tone is the accurate setting the bias of an analog tape machine, in which the tone (1khz, 10khz) is fed to the tape from the Mixers "oscillator" option or sometimes it is given from the Tape Recorder facilities option.

(g) ***Slate:*** The slate would be used for recording or take information on to tape:

(h) ***Master Faders:*** There may be either one stereo fader or left and right faders, to control the overall Mix output level. Often the group master fader will reside in this section.

INTEXT QUESTIONS 9.4

Choose anyone of the following

1. In an ordinary mixing console the number of aux send present is ....
   (a) One (b) Two
   (c) Ten

2. Aux send in the mixer are normally used for ....
   (a) Recording (b) Playback
   (c) Fold back or Talk back

3. Monitor selection is fed the signal to ....
   (a) Channel path (b) Aux send
   (c) Loud speaker

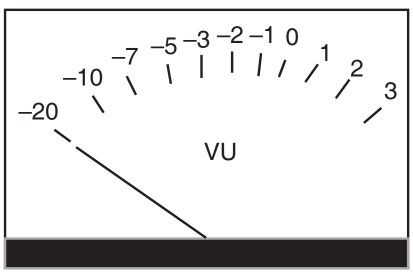
4. .... is used for quick silencing of the room
   (a) Attenuation (b) Pad
   (c) DIM

5. Sound engineer gives instruction to studio through
   (a) Oscillation (b) Talk back/Fold back
   c) Slate

6. The process using for biasing the magnetic tape is
   a) Oscillation b) DIM
   c) Attenuation

9.5.8 Metering System/ Section

Metering system is placed on audio consoles to measure the level of the audio signal feeding in & coming out from the Mixer. This is important for measuring the audio level without noise & distortion and to record the correct sound.



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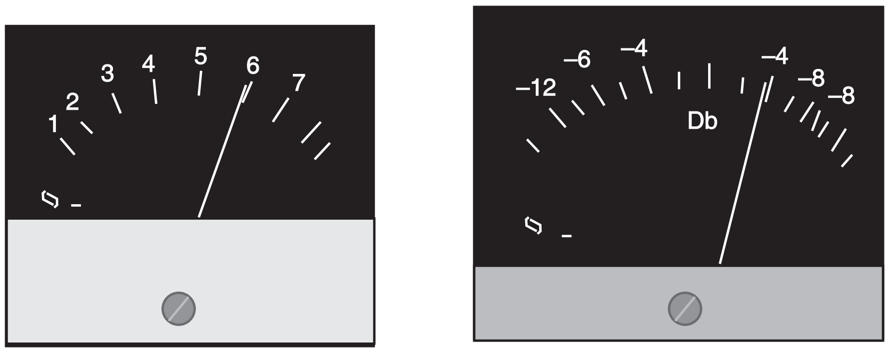

Fig. 9.11

Usually two types of metering are provided in the audio console.

1. Mechanical Meter
2. Electronic Bar – graph meter.

Mechanical Meter: Generally two types of Mechanical meter have been used. Fig. 9.11.

(a) VU (volume unit) Meter
(b) PPM (Peak Program) Meter

In the VU meter, there is associated variable attenuator, which could vary the electrical alignment level for O VU up to +24dBu. Now it is common for this to be fixed these days at OVU = ±4 dBu

Disadvantage in Mechanical Meters

PPMs respond well to signal peaks, that is they have a fast rise time, where as VUs are quite opposite, they have a very slow rise time. This means that VUs donot give a true representation of the peak level going on tape. Many people are used to working with VUs, they are good to measuring continuous signals such as tones, but their value, for monitoring program material, is dubious in the age of digital recording.

VUs have no control over fall time of the needle, which is much the same as the rise time. Whereas PPMs have fast rise time & longer fall time, which is more useful.

Normally PPMs are designed to graduate peaks that would cause audible distortion but doesn't measure the absolute peak level of the signal.

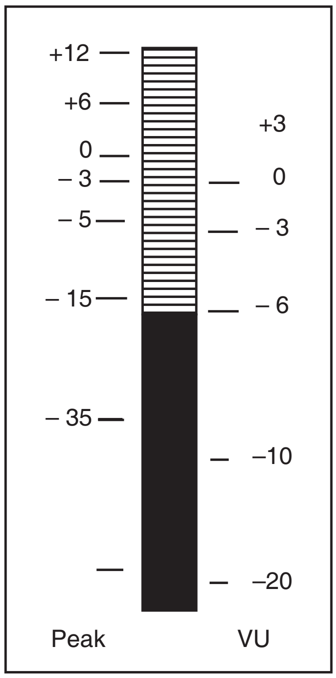

Fig. 9.12

It is important to note that meters can take a lot of space on the console & it may be impossible to find space for one meter per channel, in case of multi-track consoles.

Electronic Bar graph Meter

In Electronic bar graph meter (Fig. 9.12) there can be an infinitely fast rise time, although this may not be ideal in practice but cheaper bar graphs are made out of new of LEDS (Light Emitting Diode) & the resolution accuracy depends on the number of LEDs used.

There is a facility provided to switch the peak response of these meters from peak to VU mode. where they will imitate the scale and ballistic response of a VU meter.

The main advantage of the Bar graph vertical VU meter is that it takes a little space on the audio mixer.

9.6 AUTOMATION

In Mixers, the automation means storing fader positions dynamically against time of reiteration at a later point in timesynchronous with recorded material.

The purpose of automation has been to assist the engineer/operator, in mix down, the no of fader's that need to be handled at once, become too difficult for one person.The fader automation has resulted in engineers being able to concentrate on sub areas of a mix at each pass, gradually building up the finished product and refining it.

Fader Automation: There are two common means of memorizing and controlling the gain of a channel

1. Which stores the positions of the fader and uses this data to control the gain of a VCA or DCA (digital controlled attenuator)
2. Which also stores the fader movements but uses this information actually to drive the fader's position using a motor?

Common Operational Mode's are:

Write: It memorizes the level of the channel corresponding to the fader position

Read: Channel level controlled by data received from previously stored Mix.

Update: Channel level controlled by a combination of previously stored Mix data and current fader position

Group: channel level controlled by combination of channel fader position and that of a group master

INTEXT QUESTIONS 9.5

1. How many types of level meters are there in the audio mixing console?
2. Which type of meter is taking a lot of space on the audio mixing console?
   (a) Mechanical Meter
   (b) Electronic Bar – graph meter.
3. Which type of meter uses LED
   (a) Electronic bar-graph meter
   (b) VU meter
   (c) PPM meter

9.7 WHAT HAVE YOU LEARNT

In this lesson you have learned following aspects of Audio Mixing Consoles

- Different sections of an analog audio mixing console with their functions.
- Different stages of multi-track mixing.
- Grouping of faders during multi-track mixing, their types and how they are operate.
- Digital mixing console and its different sections with their functions.
- Different types of meter, their advantage and disadvantages.

9.8 TERMINAL EXERCISE

1. Briefly explain the sections involved in simple analog mixer.
2. Briefly discuss the two distinct stages during music recording.
3. What is grouping? Briefly describe types of grouping?
4. How many sections are there in a digital mixer?
5. Briefly explain different sections of an input section?
6. Briefly explain different sections of routing section?
7. What is dynamic control in the digital mixer?
8. Briefly explain different sections of equalization section?
9. Briefly explain different sections of channel and mix control section?
10. Discuss the four points in auxiliary send?

11. Briefly explain different sections of master control section?
12. What is the function of metering system in the mixer? How many type of meter generally present in the mixer?
13. What are the disadvantages in mechanical meter?
14. What is automation? Briefly describe the four operational modes are there in the automation process

9.9 ANSWER TO INTEXT QUESTIONS

9.1

1. (c)	2. (c)	3. (a)	4. (b)
5. (b)	6. (a)	7. (b)	

9.2

1. (b)	2. (b)	3. (c)

9.3

1. (b)	2. (a)	3. (d)	4. (c)
5. (b)	6. (c)		

9.4

1. (d)	2. (c)	3. (c)	4. (c)
5. (b)	6. (a)		

9.5

1. (Two)	2. (a)	3. (a)

9.10 GLOSSARY

1. **Phantom power:** It is a +48V DC power supply which is given to the microphone to operate it. (i.e., for condenser or capacitor microphones)
2. **Routing:** Assigning into desired position or track
3. **Monitoring:** Listening the sound signal or the sound track
4. **XLR Connector:** Type of 3-pin connector for connecting audio signals
5. **Module:** a part of a section
6. **Frequency band:** a set/ range of frequencies
7. **Pre/Post:** before/after.