DESIGN OF STAGE AND RECORDING CONSOLE

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Dedicated to my beloved family, my father (Nupe ak Ahmee), my mother (Noiap ak Labut), my brothers (Marthion, George and Canisius), my sisters (Byna, Fowler, Cornelia and Josephine) and my beloved ones who are always there to support and have given encouragements.

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ABSTRACT

The console is where the mixing and amplification of the signal from audio producing media is performed. The console is widely used in the production of quality audio, such as in the recording studio and radio station. The basic operation of a console is amplifying, balancing, combining, monitored, and routing signals. At the console, signals from microphones, disc players, electric and electronic musical instruments, and tape recorders are amplified, balanced, combined, monitored and routed for broadcasting or recording. Other than that, they also perform additional functions, such as equalization, stereo panning, reverb send and return, limiting and compressing, patching and all-digital signal processing.

ABSTRAK

Konsol adalah peralatan mengadun dan menguatkan isyarat bunyi dari media penghasilan bunyi. Konsol digunakan secara meluas dalam penghasilan bunyi berkualiti, seperti dalam studio rakaman, dan stesyen radio. Operasi asas sebuah konsol ialah menguat, mengimbang, mengabung, mengawal, dan menyalurkan isyarat bunyi. Pada peralatan konsol, mikrofon, pemain cakera padat, instrumen elektrik dan elektronik, serta pita rakaman dikuatkan, diseimbangkan, diadun, dan disalurkan untuk disiarkan atau dirakam. Selain itu, konsol juga turut mempersembahkan fungsi tambahan seperti penataan, pembahagi stereo, menghad dan memadat, dan kesemua pemprosesan isyarat digital.

CONTENT

CHAPTER 1

1.0 INTRODUCTION

1.1 What is a stage and recording audio mixing console?

The stage and recording console is where the mixing and amplification of the signals from audio sources or medias are done. The basic operation of a console is amplifying, balancing, combining, monitored, and routing signals. At the console, signals from microphones, disc players, electric and electronic musical instruments, and tape recorders are amplified, balanced, combined, monitored and routed for broadcasting or recording. Other than that, they also perform additional functions, such as equalization, stereo panning, reverb send and return, limiting and compressing, patching and all digital signal processing.

According to John L. Hood (Audio Electronics), most consoles share two design characteristics. First, they take, or input, signals from a sound source; send, or output, signals to broadcast or recording; and enable the signals to be monitored, electronically and acoustically. Second, these systems function according to prescribed signal flow. Although complexity of design and pattern of signal flow vary between consoles, most operate in the same way.

The most common consoles may have 6 to 12 channels of input and 1 to 2 channels of output. The input channels may include microphone, disc players, auxiliary instrument, tape recorder, acoustic guitar and electronic musical instruments.

1.2 What is a Mixer For?

A mixer actually has some important similarities to a simple home stereo receiver. The receiver has control panels that allow switching between different components of audio system, so users can listen to the CD player, or the phonograph, or a cassette deck, as they desire to. A stereo receiver also has controls to set overall volume, the balance between left and right speakers, and some tone controls to shape the overall sound.

According to John L. Hood (Audio Electronics), a mixer does many of the same things, including changing levels and tone. The most important difference between a mixer and a stereo system is that a mixer allows you to control and combine or mix sounds from many different sources at once. Rather than simply choose between two or more sounds, a mixer gives you the option to combine many sounds at the same time.

1.2.1 Three simple reasons for mixers and mixing:

i. Changing the character of an individual soundmaking it louder, softer, brighter, fatter, etc.

ii. Combining many sounds together, and finally;

iii. The artistic processes of blending of all sources in a finished "mix," either recorded to tape or for a live audience.

1.2.1.1 Changing the Character of a Sound

Every sound has its own basic properties sounds can be soft, loud or somewhere in between. Sounds can also be shrill, muffled, smooth, bright, dark, etc.

Within limits, a mixer has the ability to alter the properties of sounds. For example, a sound can be made louder or softer with a simple touch of dial of the control panel. However, when it comes to changing basic character of the sound, there are limits. For example, the bright sound of a steel-string acoustic guitar can be made darker and rounder a little more like nylonstring guitar to record a classical guitar piece. But, however, it is still have a guitar sound, which means the basic sound of an instrument cannot be changed to another by a mixing its sound.

1.2.1.2 Combining Many Sounds

A single recording or live performance can include dozens of individual sounds or instruments. Therefore, a crucial function of a mixing board is to serve as a traffic manager for all the individual audio signals involved in a project. Signal routing is important because you may not want all your signals to go to the same place. For instance, you might want a vocal microphone to go through an echo or other "effects unit."

Mixing consoles offer many ways to route signals. The most familiar is the pan control. This pan control is similar to the balance knob on a hifi; the pan control directs a signal to the left or right side of a stereo sound system, or somewhere in between. In practice, this makes it possible to simulate the original location of each performer in the finished stereo mix.

1.2.1.3 The Final Mix

The finished product of the mixer will be a blend of sounds or musical instruments, which is balanced as desired. This "mix" typically takestwo forms, the produced sound can be recorded to tape to be replayed later and perhaps duplicated and distributed to others, or alternately, mixing a live performance and helping to balance the sound heard by the audience over a sound system.

In either case, the mixer is the tool that will allows the creations these finished pieces of sounds. Once the basic functions of the mixing board have been mastered, the produced sound can be creative and harmonic.

1.3 Benefit of Sound Mixing

1.3.1 Feedback Control

An open microphone located in a room with a loudspeaker system creates a potential source of feedback as the sound from the speaker system reenters the microphone. As someone speaks, the microphone picks up the sound and sends it to the speaker system. The sound comes out of the speaker system and then reenters the microphone.

This recirculation cycle can be repeated many times in reflective rooms or where very little margin exists between the necessary loudness level and the maximum possible loudness level If the sound level in the room is high enough, the recirculation will build up to the point where a loud howling or squealing will occur. This oscillation is referred to as acoustic feedback. As more microphones are added to the system, the problem grows worse. An automatic mixer solves this basic problem by applying a process commonly referred to as NOM attenuation.

NOM stands for: Number of Open Microphones. An automatic mixer differentiates between active and inactive microphones, turns down the inactive ones and then adjusts the overall sound system level according to how many active microphones remain. The result is that all of the microphones in the room act the same as a single microphone with respect to feedback.

There are many different types of mixers available from different manufacturers. As simply known, low cost types offer nothing more than a switch that opens and closes the microphone with voice activity. Better ones use more involved algorithms to differentiate between active and inactive microphones. The best mixers utilize sophisticated algorithms with either analog or digital controls to smoothly turn microphones up and down or distribute gain among the channels to apply NOM attenuation in a seamless, inaudible manner.

1.3.2 Improved Sound Quality

A mixer also minimizes recirculated sound and background noise to dramatically improves the sound quality in public address, teleconferencing and recording systems.

In a public address system, the sound will recirculate through the microphones and speakers and hit the listener's ears numerous times, spaced a few milliseconds apart, making it very difficult to understand what is being heard. Background noise gathered by unused microphones in any type of sound or recording system will also add an unnatural "hollow" character to the sound, reducing intelligibility.

Mixers can also be an important addition to a studio production facility where multiple microphone systems are used to produce talk shows, by reducing background noise and applying NOM attenuation. The lower background noise improves the clarity of the sound and the NOM attenuation automatically limits the overall sound level to avoid overloading the audio signal chain.

CHAPTER 2

2.0 LITERATURE REVIEW

2.1 Basic Console Design

At first glance the larger, more complex consoles, with hundreds of pushbuttons, knobs, switches, and various colored lights, may seem intimidating. Most consoles, however, share two design characteristics. First, they take, or input, signals from a sound source; send, or output, signals to broadcast or recording; and enable the signals to be monitored, electronically or acoustically. Second, these systems function according to a prescribed signal flow. Although complexity of design and pattern of signal flow vary between consoles, most operate in the same way.

2.2 Input/Output specification

Two indications of a console's and capabilities are provided by the number of input sources it can accommodate at the same time and the number of discrete signals it can output. Consoles are often specified according to the number of its input and output. For example a 6X 1 (6in, 1-out) console has 6 inputs and 1 output; a 6X 2 (6-in, 2-out) console has 6 inputs and 2 outputs; a 32 X 32 (32-in, 32-out) console has 32 inputs and 32 outputs. Clearly, a 6 X 1 console is limited to situation requiring few sources to input and onechannel or monophonic, output. A 6 X 2 is still limited to few input sources but is capable of a two-channel or stereophonic output. By contrast, a 32 X 32 console can be used for a large-scale recording assignment requiring multiple inputs and outputs. Some consoles even has a three-number specification, such as 24 X 8 X 2, whereas the 24 indicates the 24 inputs, 8 submaster and 2 master outputs. The sub-master is for sub-mixes, in which selected inputs are combined before they reach the master output, where all inputs are combined.

The design problem is threefold;

1. To get the signals from each sound source to inputs in the console.
2. To combine these at some point into an output that can be conveniently routed to broadcast, and
3. To provide a way to hear the resulting signals.

2.2.1 Input system

The input section takes an incoming signal from a microphone, disc player, or tape recorder, processes it, and routes it to the output and monitor sections. A simple input module may consist of an input connector, an amplifier to boost the level of the input signal, and a volume control. The input module might be capable of additional function, such as phantom power supply for microphones that requires voltage to operate, equalization, cue and reverb send, panning, and channel selection.

2.2.1.1 Microphone input.

The microphone changes sound energy to electric energy. However, it is too weak to go very far. Because it needs a power boost to get anywhere, the first component on the console input system should be preamplifier- a device that boost the signal from the microphone to usable proportions. Although we construct a stereo console, microphone channel are always mono.

2.2.1.2 Loudness control

A preamplifier boosts the signal a fixed amount and has no way to varying the microphone's level of output or loudness. The device that regulates the level coming from a preamplifier is called a potentiometer, or pot. A pot is also known as an attenuator, a gain or volume control, or, most commonly a fader. On the console the fader controls the amount of signal that comes from the preamp by being rotated or moved up and down.

2.2.1.3 Other Inputs

Compact disc (CD) players, like microphones also require pre-amplification. But, unlike microphones, broadcastquality CD players are medium-level signal sources and do not require as much pre-amplification.

Tape recorder- open-reel and cartridges – are high-level sound sources. They still require preamps at the console, but these do not have as much gain as the preamps used for microphones and CD players. Because various pieces of equipment have different output level, each input on the console is electrically matched to take a particular component; microphone to microphone preamp, CD player to CD player preamp, tape recorder to tape recorder preamp and so on. Input channels carrying the left and right stereo signals are designed to feed them in tandem.

2.2.2 Output System

With the equipment described so far it is possible to feed sound from microphones, CD players, and tape recorders to their separate inputs at the console, control their amplitude, and turn their signals on and off. But, there no way to

1. Combine their signals,
2. Send their signals out of the console,
3. Measure the amount of signals passing through each channel that has its level regulated by a fader, and
4. Hear the sound.

The output system routes the signals from the console to a recorder or master control. It includes a network that combines the signals from the input section, and volume controls for the submasters and master that regulate overall output level.

Each signal can be combined and then fed through a single, tandem output from the console. For these signals to be fed into one output line, there must be a mixing network, known as combining amplifier, summing network, active combining network (ACN), or bus, the most common term, to combine them. The level of combined signals is regulated by master fader. Regardless of how many input signals are fed to the output section, usually they are all eventually combined into one (mono), two (stereo), or more (for four-channel and six-channel Dolby Stereo) master outputs. Many consoles with two master outputs combine them into a single, tandem, and volume control. Also, a relatively recent console design combines some of the functions of the input and output section.

2.3 Recording

2.3.1 Basic recording

Recording can involve nothing more than a single microphone and a cassette deck. Recording studios can also have limited equipment. At other end of the scale, professional studios use a variety of soundprocessing devices, producing tapes suitable for submission to record or compact disc manufacturers or radio and TV broadcasting stations. The recording equipments may include tape recorders, magnetic tape writer, or even CD writers.

2.3.2 The Analog Recording Process

Analog recording devices use a plastic tape coated with magnetic particles moving across a magnetic recording head at a constant speed to record and playback. There is always an "erase head", first in the tape path, to erase and re-align the tape particles before they hit the "record head". In the "two-head" machine there is one head for both recording and playback. The "three-head" design features one head dedicated to recording, the "sync head", and another for playback, "the repro head". Professional machines have three heads.

There is a limit to the intensity of the signal that the tape particles can actually absorb and reproduce. The two parameters that interact to maximize the tape's ability to correctly record and playback are "tape speed" and "bias". At a faster speed, there is more tape area for a given signal, i.e. more tape particles to record. Most professional analog multi-track recorders run at 30 ips (inches per second). "Bias" is a process that was discovered by accident. It was found that when a high frequency signal, 100 KHz or so, much higher than human hearing, was recorded along with the normal signal the magnetic particles did a better job of recreating the higher frequencies.

It is a complicated process the tape machine must be mechanically and electronically aligned to very fine specifications. First, to ensure that it physically handles the tape gently during shuttling, rewind and fast forward. Although tape formulations have improved greatly over the years, mechanical problems can damage the tape by stretching or wrinkling it. Other problems include loss of particles of the tape called shedding, speed fluctuations which produce "wow and flutter" and improper tape to head contact.

Furthermore, the electronics have to record the input signal and play it back faithfully. This is where tones on your master tapes become very important. They are required to properly align the electronics in the tape machine so when you work at different studios, your tape sounds like the original. When all these parameters are aligned correctly, you stand a good chance of hearing back a reasonable facsimile of what you recorded previously.

2.3.3 The Digital Recording Process

The digital recording process is far simpler mechanically, but much more involved electronically. The input signal is sampled 1000's of times per second and each acoustic slice is given it's own digital number, consisting of 0's and 1's. Theoretically, the "analog-to-digital converter" (ADC) receives the analog input and converts it into a stream of numbers and conversely, the "digital-toanalog converter" (DAC) reverses the process.

The "sampling rate", or how many times per second the sound is sliced is the main factor in how well the sound will survive its digitization. CD's are sampled at 44.1 K or 44,100 times per second, and that has become an industry standard. Some formats offer 48 K sampling as well. DAC's and ADC's are not created equally however and there are differences in how these machines sound, despite the theoretical consistency of 0's and 1's.

Digital tape machines use mechanical transports and plastic tape as a storage medium for the digital information. The Alesis ADAT and Tascam DA-88 are examples of new inexpensive digital multitracks. Another approach gaining acceptance is hard disc recorders. Some have computers with software as front-end controllers, like the Digi-Design and Soundscape machines, while others are dedicated boxes you plug hard discs into for storage, like the EMu Darwin, Vestax and Akai recorders.

With these random access digital recorders, the size of the hard discs limits the amount of recording time. Locating is a snap, as is editing. When this approach is combined with a computer as the interface, you have a powerful word processor for music. Anyone who has used a Mac or Windows onan IBM knows how to drag and click with a mouse and that's basically how you manipulate the sound files.

2.3.4 Theory of Multi-track Recording

Multi-track recorders are simply tape machines that allow you to record tracks and then overdub additional tracks in any order. For instance, you might record a drummer on four tracks, then go back and record a guitar part, etc. To do this, the tape machine must be able to record one track while it is playing back the others. In an analog machine, it must do thisfrom the same recording head. This is the job of the "sync head".

Digital machines don't rely on sync heads and repro heads they are just reorganizing 0's and 1's. Depending on the device, sometimes the tape based digital machines are not as flexible as the random access machines.

It is possible to "lock up" more than one multi-track tape machine to get more tracks. This is usually done with two identical machines and SMPTE. SMPTE is an acronym for a time-code that was originally developed for the motion picture industry. It sounds like a high-pitched squeal but to devices that can "read" it, it looks like a running clock. For lock up, we would "stripe" two multi-track tapes; for one song we might need five minutes, so we set the SMPTE "writer" to write from 0:00:00:00 to 5:00:00:00 minutes.

SMPTE is displayed as "hours:minutes:seconds:frames:sub frames", although not all devices read sub-frames (there are 80 subframes). There are four types of SMPTE. They are 30 frames per second (fps) drop frame, 30 fps non-drop frame, 25 fps, and 24 fps. In the United States, audio professionals generally use 30 fps non-drop frame and in England and Europe, they use 25 fps. The 30 fps drop frame, sometimes called "29.97", is used for video and film applications in the United States.

As has become the industry practice, we record this SMPTE on the highest edge track on each tape. For example, track 8 on an 8-track, track 24 on a 24-track, etc. So, now we have identical "striped" tapes on their respective multi-tracks. The next step is to use a synchronization device designed to read the SMPTE off each machine and control the motors of both to keep them locked together. One machine becomes the "Master" and the other, or "Slave", chases the master machine. Two of the most popular professional systems that do this are the Lynx and Adam Smith synchronizers. This is the basic concept and it is possible, with the right interfaces and connections, to lock up different types of multi-tracks, VCR's, time-code equipped DAT machines, digital editors, etc. The concept of this recording is simple, but the execution can be complicated.

2.4 Effects Devices

Effect device include these cool stuff; reverbs, phase shifters, delays, chorus', harmonizers, echoes and combinations never been heard before. These devices have come a long way in 20 years. Analog electronics and spring reverbs have given way to very powerful digital multi-effects units with MIDI capabilities, memory for your favorite patches and wide dynamic range.

2.4.1 Reverb

Reverb units attempt to recreate the sound of a particular space. The way a "space" sounds is a product of the size, whether or not the interior surfaces are hard and reflective or soft and absorbent, and how these interior surfaces are arranged. All these factors interact to produce the reverberant sound. Two spaces can have the same interior volume, but it is shaped very differently and that makes all the difference.

The primary types of spaces are rooms, halls and plates but also can include chambers, churches, clubs and any number of wild spaces. Some unit give you a few parameters to tweak others give you pages of possibilities. All start with at least the "size" of the space. Other tweak able options include the volume and intensity of "early reflections", the amount of pre-delay to the reverb, which delays the input send into the reverb and "diffusion" and "depth" settings which have to do with how intensely the reverb spreads out in the stereo field.

Special types of "reverse reverbs", where the sound envelope is turned around and ramps up in volume rather than trailing off are actually inspired by the analog trick of "backwards reverb". The important distinction is that "reverse reverbs" occur after the sound just like regular reverbs. "Backwards reverb" occurs before the sound and seems to ramp up to the sound.

2.4.2 Echoes and Delays

An echo is an acoustic phenomenon where a sound is repeated. The classic example in 'mother nature' is an echo canyon. Delay units recreate this capability and give us control of several different parameters. The two most basic are "delay time", the period between the input and the delayed signal's output, and "feedback" or "regeneration" as it's sometimes called, which is how many times the signal repeats as it fadesaway.

Generally, each successive repeat decreases in volume. When the feedback control is raised past a certain point, the repeats get louder and louder. This is called "Runaway Feedback". Some units have an "Infinite Hold" or "Freeze" feature, which captures the input signal and keeps repeating it until you stop it. To the untrained ear, discrete echoes start to be distinguished at about 20 mili-seconds. Delays lower than this start to sound more like flangers and chorus effects. Most delay units also give you this capability.

Early analog electronic delays began to show up in the 1970's. The fidelity was somewhat limited and delay times only went to 600 or 700 mili-seconds at most. Digital delays began to show up in the 1980's and delay times of several seconds became more common. Modern units offer superb sonic, patch memories and MIDI implementation.

2.4.3 Flanging, Chorus and Phasing

Similar to "reverse reverbs", these effects were inspired by analog recording techniques. Sending the vocal to another tape machine in record and mixing the signal coming off the repro-head back in with the original created the famous “slap-back echo” of the 1950’s and early 60’s. The distance between the record head and repro-head and the speed of the second machine determined the time of the "slap-back".

"Flanging" was a variation on this technique. The process starts similarly, but by rubbing the edge of the flange of the tape reel on the 2nd machine ever so slightly, the characteristic "Flanging" sound was produced. The time delays involved with flanging, chorus and phase shifting are usually well below 15 mili-seconds.

Digital delay units simulate these effects by incorporating an oscillator into their circuitry, which allows you to control the speed and depth of the signal being recombined with the original. Modern devices give you mono and stereo flanging and chorus effects. Phase Shifters introduce slight time delays that also change the phase of the signal being recombined. This gives the distinctive deep sweeping effect that they are known for.

2.4.4 Harmonizers and Exciters

Harmonizers are basically similar to digital delays except that they allow us to tune the pitch of the delay. When slightly detuned and added back in with the original signal, they are a useful tool for electronically thickening vocals, guitars, etc. 'Smart' harmonizers are available now that can add designated pitches according to different musical scales in their internal processors.

The most over-used effect of the 2000-2001 recording season has to be the Antares Auto Tune. Originally developed as a plugin for the ProTools recording format, its now available for most major software recording packages and as a stand-alone hardware unit, the ATR1. It can indeed repair pitch problems in a performance, but it is now being used as an effect on pop, rap, dance and even country songs. When it is overdriven, it resembles a vocorder or vocal recorder but with more of the human voice intact.

2.5 Sound/Signal Amplification

The sound or signal generated by vocal source is very small in amplitude; this kind of signal may become unintelligent if it is to travel over a distance. The amplification of such signal ensure the intelligence of the signal when it reaches a distant location depend on the levelof amplification. While in electronic music instruments or sound producing instruments the signals produced can be controlled by the preamp before it reaches the power amplifier. All these sound sources must be fed to an audio power amplifier to amplify the signal to achieve desired amplification level.

2.5.1 Audio Power Amplifier

In general, the purpose of an amplifier is to take an input signal and make it stronger or in more technically correct terms, increase the amplitude to a desired level Amplifier is found in all kinds of electronic devices designed to perform any number of functions. There are many different types of amplifiers, each with a specific purpose. For example, a radio transmitter uses an RF Amplifier or Radio Frequency amplifier; such amplifiers are designed to amplify signals so that it may drive an antenna.

While audio power amplifiers are those amplifiers, which are designed to drive loudspeakers. After an audio signal has been mixed, equalized and otherwise processed at a standardized line level, it is sent to the power amplifier. The job of audio power amplifier is to increase the power of the signal