Audio Production Course Manual

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Three Pillars of Managing Sound: The Technical, the Method, and the Creative Approach

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Contents

Welcome to CLC Audio Production Course. I began my journey into audio production as a teenager in 1989 when a friend left his 4-Track cassette recorder in a basement rehearsal space. I toyed with the faders, plugged in my guitar, and pressed the red flashing button marked "record." I listened to the playback, quickly wrote a second guitar part, and repeated the steps. That process of writing, playing, recording, and mixing my own music enthralled me. From that moment forward, I was hooked on audio production.

Since then, I've spent much of my adult life working in recording studios, mixing live sound, and living inside my headphones thinking about all the different kinds of production techniques used to create interesting music. My hope is that I can share what I've learned and connect students to educational resources that will build a basic knowledge of audio production.

This course is designed to help beginners get a feel for what audio production is, while also helping more advanced students expand their own production skills. I've organized this class into three units that hopefully lead students towards the final project: designing a home studio.

Unit One: Microphones, Mixing Boards, Cables, and Speakers

Unit Two: History of Audio Recording, Pro Tools Recording Software/Mixing a Session, Types of Audio Files, and Mastering

Unit Three: Session Psychology, Recording on a Budget, Acoustic Room Treatments, and Designing a Home Studio Space

Learning Outcomes: Upon completion, students should have the ability to–

1. **Properly use microphones and p.a. equipment to mix a live audio event**
2. **Operate Pro Tools or other related recording software/hardware to produce a small audio project**
3. **Differentiate between audio formats and have the ability to convert one audio file to another**
4. **Understand acoustic room treatments, recording budgets, and basic designs of home studio spaces**

One of the first things I'd like to share is that when I'm working on a project, I rarely think of myself as an audio producer. To me, *music producer* is a loaded phrase with some negative connotations from my own experiences working in over-priced, industrial studio settings.

What I'm really doing, whether it's recording an acoustic guitar, making electronic beats, mixing a live band, or editing a podcast, is this: I am simply ***managing sound***. I take sounds created and manage them in organized ways for a particular listener. I've learned that not every project needs to be surgically produced on corporate budget. Sometimes a project only requires a clean and honest sound to have the desired result. When presented with a new audio production job, I often look at what the end goal is. Am I helping a friend to make a quality demo? Am I being hired to cost-effectively run a small studio project? Maybe the job calls for better, professional-style gear because it's for a larger broadcast audience. Each scenario requires me to use different tools and devote a certain amount of creativity energy to accomplish it. However, in each case, I am still ***managing sound*** for a particular type of listener.

Keep in mind, a great performance can be magical, but effective sound-management takes some know-how to audibly capture that magic. Welcome to Audio Production!

So, where do we begin when we want to capture those magic sounding moments? For me, I always start by applying a **three pillar approach**. Students should try to incorporate these three pillars into every project to get the best results in audio production.

PILLAR ONE: THE TECHNICAL:

- **What** is being used? What are the components? How does it work? What is it compatible with?
- What do the **specifications,** instructions, and charts mean?
- Is it being **used in a safe and proper way**?

PILLAR TWO: THE METHOD:

- **How** is it being used?
- Are there **different ways** to to apply it?
- **Why** is it being used this way?
- What are the budgetary, practical, and **socio-economic** reasons for using it?

PILLAR THREE: THE CREATIVE APPROACH:

- What **Principals of Artistic Design** are being used (Contrast, Rhythm, Repetition, Flow, Proximity, Alignment, Balance, Emphasis and Proportion)?
- Is it **original**?
- Is the **emotional input** being heard in the output?
- Does it feature **Aesthetic Honesty**?

Every project requires some use of all three pillars. For example, when mixing a band on a digital audio workstation you will need to know **technical** details for assigning microphone inputs; you will need to incorporate different **methods** of microphone placement; and you will need to a **creative approach** when balancing your mix to produce a signature sound. You don't have to be an expert in all three pillars, but the more knowledge and experience you gain concerning all three will give you the biggest toolbox to work with. I will go back to these pillars time and time again throughout this course to point out how they can help improve your audio production projects.

PART I

Units and Chapters

1

Unit One, Part One: Sound Waves and Microphones

SOUND WAVES

Before we can start learning about audio production and microphones, let's get a general idea of how sound waves work. Though the study of sound waves is mostly a **technical** matter, understanding the science behind it can help us manage it. How an audio producer adjusts the settings on microphones **(method)** can affect how sound waves enter the device. And ultimately, how those soundwave frequencies are manipulated within a mix can dramatically change the emotional response a piece of audio has on a listener **(creative approach)**.

HOW SOUND WORKS

Sound is the term to describe what is heard when sound waves pass through a medium to the ear. All sounds are made by vibrations of molecules through which the sound travels. For instance, when a drum or a cymbal is struck, the object vibrates. These vibrations make air molecules move. Sound waves move away from their sound source (where they came from) traveling on the air molecules. When the vibrating air molecules reach our ears, the eardrum vibrates, too. The bones of the ear vibrate in the same way that of the object that started the sound wave.

These vibrations let you hear different sounds. Even music is vibrations. Irregular vibrations are noise. People can make very complex sounds. We use them for speech.

Sound waves are longitudinal waves with two parts: *compression* and *rarefaction*. Compression is the part of the sound waves where the molecules of air are pushed (*compressed*) together. Rarefaction is the part of the waves where the molecules are far away from each other. Sound waves are a sequence of compression and rarefaction.

HERE ARE THREE VIDEO LINKS THAT EXPLAIN THE BASIS OF WHAT WE WILL BE WORKING

WITH: **SOUND WAVES, DECIBELS, AND DIGITAL AUDIO.**

MICROPHONES

A microphone is a form of transducer. That means it converts a sound wave into an electronic signal carried by wire. Generally, when that electric signal is sent through an amplified speaker, it is then converted back into a sound wave. Microphones may also use a processor to convert the sound wave into signal (or code) that can be used by a computer. Understanding the *technical* aspects of microphones is so very important for producing good audio. It also ensures electrical safety to you and your equipment. As you gain more experience with microphones, where you place them in relation to what is being recorded (method) will become equally important. Experimenting with both of those aspects will help you develop your own signature sounds using your favorite microphones and microphone placements *(creative approach*).

For many of us who started out with analog recording, microphones were the gateway into audio production. But whether you use analog or digital audio equipment, the right microphone pointed in the proper direction can help a performance stand out. There are dozens of styles of microphones and a wide variety of prices. To begin however, let's concentrate on the two most common types of microphones: Dynamic and Condenser.

Dynamic Microphone – In a dynamic microphone, a thin diaphragm is connected to a coil of wire, called a voice coil, which is precisely suspended over a powerful magnet.

- As the sound waves strike the diaphragm, it causes it to vibrate moving the voice coil through the magnetic field generated by the magnet generating a small bit of electricity which is sent down the output leads.
- This is the **electromagnetic principle**.
- ADVANTAGE: They are simply constructed and can handle loud sources without much distortion.
- DISADVANTAGE: They are weak when trying to capture soft distant sources because the diaphragm requires a lot of sound energy to move.
- DISADVANTAGE: Dynamic microphones have a heavy diaphragm along with additional weight from the coil of wire.
    - It therefore takes longer for the

diaphragm to react to a sound wave causing a less accurate recording.

Condenser Microphone – Condensers use two charged plates; one fixed and one which can move acting like a diaphragm.

- There's no coil.
- The two charged electric plates create what's called a capacitor. As sound waves strike the electrically charged diaphragm, it moves in relation to the fixed plate changing its capacitance and generating a very small electric charge which is amplified inside the microphone.
- This is the **electrostatic principle**.
- ADVANTAGE: Because you're not moving a coil, condensers are more responsive in the high frequencies.
- ADVANTAGE: Because of the lack of magnets, condenser microphones can be very small.
- Because condensers work with electrically charged plates, they require some sort of outside power.
- Some microphones have the option of an onboard battery while all condensers can utilize something called Phantom Power.

Phantom Power – +48v of energy sent down the microphone cable to a condenser microphone from the audio recording or mixing board.

- This power enables the electrically charged diaphragm to move in response to sound waves.

Directional Response – Directional response is represented by something called a polar pattern.

Polar Pattern – Polar pattern is how well the microphone "hears" sound from different directions.

"On Axis" and "Off Axis" – On axis is directly in front of the sound source. Off axis is not directly in front of the sound source.

Omnidirectional Mics – This mic polar pattern is responsive to sound from all directions, you don't have to be "on axis" to be picked up.

- Lavalier and lapel mics are small condenser microphones with an omnidirectional pickup pattern that can be placed on a person.
- Boundary mics are omnidirectional condenser mics. They are positioned flush with a surface that capture sound as it rolls off the flat surface. Boundary mics are used in stage production and conference tables.
- ADVANTAGE: These mics are useful for picking up sound in a general area.
- ADVANTAGE: Lavalier / lapel mics are small and

can be placed just about anywhere.

- ADVANTAGE: Boundary mics do not draw attention to themselves because they lay flat on the floor or wall.
- DISADVANTAGE: They will pick up all the unwanted sound in the area.
- DISADVANTAGE: Lavalier, lapel, and boundary mics won't have the same richness of sound as a shotgun or studio condenser mic.

Directional Mics: Cardioid Pattern – Most basic pattern.

- Heart-shaped pick up pattern.
- ADVANTAGE: Picks up what's in front but not behind.
- ADVANTAGE: It is suited for a live performance as it picks up the sound on axis but won't pick up what's behind it, like crowd noise or feedback from a speaker.

Directional Mics: Hypercardioid and Supercardioid Patterns – More directional than cardioid.

- Skinnier heart-shaped pick up pattern.
- Picks up the front and sides and rejects 150 degrees to the rear.
- Shotgun mics are supercardioid.

- ADVANTAGE: Great for recording location audio while trying to filter out some of the unwanted ambient sound.
- DISADVANTAGE: Can exhibit strange phasing sound effects when used in small spaces.

Directional Mics: Figure 8 Pickup Pattern / Bi-directional – The polar pattern looks like a figure 8.

- ADVANTAGE: Useful for certain musical applications or interviews with a person on each side of the mic.

Frequency Response of Mics

Understanding Frequency Response

What is Frequency Response

Microphone Response

Polar Patterns Demonstrated

How Does Polar Pattern Work

Most microphones come with a manual. If not, you can find one online by searching the brand and model number of the mic. These manuals can be very helpful in three areas. First, it will show you on a chart the polar pattern of the mic. Second, it may show you what certain switches will do that are located on the microphone and what is considered the front and back of it. Third, it will feature the Frequency Response. This chart will show a line (or lines) going from

20hz all the way up to 20khz. What to pay attention to is where this line is flattest on the chart. That area (range) is where the microphone most accurately picks up sound on the frequency spectrum. If it flattens out in low end areas of the spectrum, you can imagine that mic is best for bass sounds. If it is flat within the range of human vocals (approx. 100 to 120hz), then you can assume that particular microphone will work great on vocals. These are good places to start in your audio production development, but keep in mind that experimenting with different mics in various ways is the best way to discover what is the best mic to use for a given circumstance. If it sounds good, you're probably doing it right! Below is a link that explains the frequency spectrum in detail.

Audio Frequency Spectrum from Teach Me Audio

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Unit One, Part Two: Mixing Boards and Channel Strips

The technology of mixing boards has changed immensely since I started working in audio production. Since the 1980s, these consoles have gone from cumbersome, furniture-sized Rube Goldberg machines, to light-weight control panels with built in USB ports, to wireless, handheld digital mixers with a multitude of effects and memory options. However, all mixing boards share the same basic principle when learning to use them: **To organize your mix, follow the channel strip from top to bottom and the faders from left to right.**

When I first started using mixing boards, I only used them for multi-track volume control. And my mixes suffered because I didn't have knowledge on what each control's function could accomplish (**technical**). My mixes also suffered because I only knew one way to control what signals were going into the board and the volume that was leaving the board (**method**). It was hard to have any **creative approach** because, to me, mixing boards were simply giant volume controls. However, this equipment and its digital counterpart are so much more than that.

As I got better at all three pillars of audio production in regards to mixing consoles, my work improved and ultimately, I could confidently take on a wider variety of jobs mixing audio. I was not intimidated by the size or name brand of the equipment. If I could follow my order of operation for each channel strip, then I could start managing how I wanted things to sound–not just how loud. Students should learn what each of the basic inputs/functions of the traditional channel strip are and how to use the main/monitor outputs along with proper Equalization, AUX inputs, and effects.

Below is an in-depth look at the make-up and functions of mixing consoles. At the end of this material are video links that explain what mixing consoles are made up of (**technical**), how to run a sound check (**method**), and how to mix live sound (**creative approach**).

In class, we will start with a basic four-channel mixer with one p.a. speaker and a microphone. Once we have learned how to set up that simple system, we move up to larger boards with more inputs and multiple speakers including monitors. But no matter what you're working on, use your three pillars and follow the channel strips from top to bottom to get the best results.

MIXING CONSOLES

In sound recording, reproduction, and sound reinforcement systems, a **mixing console** is an electronic device for combining sounds of many different audio signals. Inputs to the console include microphones used by singers, picking up acoustic instruments, signals from electric or electronic instruments, or recorded music. Depending on the type, a mixer is able to control analog or digital signals. The modified signals are summed to produce the combined output signals which can then be amplified through a sound reinforcement system.

Mixing consoles are used in many applications including recording studios, public address systems, sound reinforcement systems, nightclubs, broadcasting, television, and film post-production. A typical, simple application combines signals from microphones on stage into an amplifier that drives one set of loudspeakers for the audience. A DJ mixer may have only two channels for mixing two record players. A coffeehouse's tiny stage might only have a six-channel mixer, enough for two singer-guitarists and a percussionist. A nightclub‘s stage mixer for rock music shows may have 24 channels for mixing the signals from a rhythm section, lead guitar and several vocalists. A mixing console in a professional recording studio may have as many as 96 channels.

In practice, mixers do more than simply mix signals.

- They can provide phantom power for condenser microphones

- Pan control changes a sound's apparent position in the stereo sound field (Left and Right)
- Filtering and equalization enables sound engineers to boost or cut selected frequencies to improve the sound
- Dynamic range compression allows engineers to increase the overall gain of the system or channel without exceeding the dynamic limits of the system
- Routing facilities send the signal from the mixer to another device
- Monitoring; whereby one of a number of sources can be routed to loudspeakers or headphones for listening, often without affecting the mixer's main output
- Onboard electronic effects, such as reverb, intended for small venue live performance applications

CHANNEL STRIPS

The channel input strips are usually a bank of identical monaural or stereo input channels arranged in columns. Typically, each channel's column contains a number of rotary potentiometer knobs, buttons, and faders for controlling the gain of the input preamplifier, adjusting the equalization of the signal on each channel, controlling routing of the input signal to other functional sections, and adjusting the channel's contribution to the overall mix being produced.

The types of inputs that can be plugged into a mixer depend on the intended purpose of the mixer. A mixer intended for a live venue or a recording studio typically has a range of input jacks, such as XLR connectors for microphones, outputs from DI boxes, and 1/4″ jacks for line level sources. A DJ mixer typically has RCA connector inputs for pre-recorded music being played back on turntables or CD players, and a single mic input.

Depending on the mixer, a channel may have buttons which enable the audio engineer to reroute the signal to a different output for monitoring purposes, turn on an attenuator pad (often reducing the signal by 15 or 20 dB to prevent audio clipping), or activate other features, such as a high-pass filter. Some higher-priced mixers have a parametric equalizer or a semi-parametric equalizer for one or more of the equalizer frequency bands.

The channel strips are typically numbered so that the audio engineer can identify the different channels. For each channel input, a mixer provides one or more input jacks. On mid to large sized live venue and sound recording consoles, these input jacks are numbered as well and consolidated in a patch bay. On smaller mixers, the input jacks may be mounted on the top panel of the mixer to facilitate the connection and disconnection of inputs during the use of the mixer.

The input strip is usually separated into sections:

- Input jacks
- Microphone preamplifiers
- Equalization
- Dynamics processing (e.g. dynamic range compression, gating), if supported
- Routing, including direct outs, auxiliary-sends, panning control, and subgroup assignments
- Level-control faders (on small mixers, these may be rotary knobs to save space)

On many consoles, these sections are color-coded for quick identification by the operator. Each signal (e.g., a singer's vocal mic, the signal from an electric bass amp's DI box, etc.) that is plugged into the mixer has its own *channel*. Depending on the specific mixer, each channel is stereo or monaural. On most mixers, each channel has an XLR input, and many have RCA or quarter-inch TRS phone connector line inputs. The smallest, least expensive mixers may only have one XLR input with the other inputs being line inputs. These can be used by a singer-guitarist or other small acts.

INPUTS

The first knob at the top of an input strip is typically a *trim* or *gain* control. The input/preamp conditions the signal from the external device and this controls the amount of amplification or attenuation that is applied to the input signal to bring it to a nominal level for processing. Due to the high gains involved (around +50 dB, for a microphone), this stage is where most noise and interference is picked up. Balanced inputs and connectors, such as XLR or phone connectors that have been specifically wired as balanced lines, reduce interference problems.

A microphone plugged directly into a power amplifier would not produce an adequate signal level to drive loudspeakers, because the microphone's signal is too weak; the microphone signal needs a preamplifier to strengthen the signal so that it is strong enough for the power amplifier. For some very strong line level signals, the signal that is plugged into the mixer may be too strong, and cause audio clipping. For signals that are too strong, a 15 dB or 20 dB pad can be used to attenuate the signal. Both preamplifiers and pads, and the controls associated with them, are available in the input section of most mixing consoles.

Audio engineers typically aim at achieving a good *gain structure* for each channel. To obtain a good gain structure, engineers usually raise the gain as high as they can before audio clipping results; this helps to provide the best signal to noise ratio.

A mixing console may provide insert points after the input gain stage. These provide a send and return connection for external processors that only affect an individual channel's signal. Effects that operate on multiple channels connect to auxiliary sends (below).

AUX SENDS

The *auxiliary send* routes a split of the incoming signal to an auxiliary bus, which can then be routed to external devices. *Auxiliary sends* can either be pre-fader or post-fader, in that the level of a pre-fader send is set by the *auxiliary send* control, whereas post-fade sends depend on the position of the channel fader as well. *Auxiliary sends* can send the signal to an external processor such as a reverb, with the return signal routed through another channel or designated auxiliary return. Post-fader sends are normally used in this case. Pre-fade *auxiliary sends* can provide a monitor mix to musicians on stage (which they hear through monitor speakers pointing at the performers or in-ear monitors); this mix is thus independent of the main mix produced by the faders.

Program channels on a radio sound board

Most live radio broadcasting sound boards send audio through *program* channels. Most boards have 3-4 program channels, though some have more options. When a given channel button is selected, the audio will be sent to that device or transmitter. Program 1 is typically the on-air live feed, or what those listening to the broadcast will hear. Other program channels may feed one or more computers used for editing or sound playback. Another program channel may be used to send audio to the talent's headset if they are broadcasting from a remote area.

EQ

Further channel controls affect the equalization of the signal by separately attenuating or boosting a range of frequencies.

The smallest, least expensive mixers may only have bass and treble controls. Most mid-range and higher-priced mixers have bass, midrange, and treble, or even additional mid-range controls (e.g., low-mid and high-mid). Many high-end mixing consoles have parametric equalization on each channel. Some mixers have a general equalization control (either graphic or parametric) at the output, for controlling the tone of the overall mix.

CUE SYSTEMS

The cue system allows the operator to listen to one or more selected signals without affecting the console's main outputs. A sound engineer can use the cue feature to, for instance, get a sound recording they wish to play soon cued up to the start point of a song, without the listeners hearing these actions. The signal from the cue system is fed to the console's headphone amp and may also be available as a line-level output that is intended to drive a monitor speaker system. The terms AFL (after-fader listen) and PFL (pre-fader listen) are used to describe respectively whether or not the level of the cue signal for an input is controlled by the corresponding fader. Consoles with a cue feature have a dedicated button on each channel, typically labeled *Cue*, *AFL*, *PFL*, *Solo*, or *Listen*. When cue is enabled on multiple channels, a mix of these signals is heard through the cue system.

Solo in place (SIP) is a related feature on advanced consoles.

It typically is controlled by the cue button, but unlike cue, SIP affects the output mix; It mutes everything except the channel or channels being soloed. SIP is useful for setup of a mixing board and troubleshooting, in that it allows the operator to quickly mute everything but the signal being adjusted. For example, if an audio engineer is having problems with clipping on an input, they may use SIP to solely hear that channel, so that the problem can be diagnosed and addressed. SIP is potentially disastrous if engaged accidentally during a performance, as it will mute all the channels except one, so most consoles require the operator to take very deliberate actions to engage SIP.

BUSSES AND SUBMIX

Each channel on a mixer has a volume control (*fader*) that allows adjustment of the level of that channel. These are usually sliders near the front of the mixing board, although some smaller mixers use rotary controls to save space. The signals are summed to create the main *mix*, or combined on a *bus* as a submix, a group of channels that are then added as a whole to the final mix. For instance, many drum mics could be grouped into a bus, and then the proportion of drums in the final mix can be controlled with one bus fader. A bus can often be processed just like an individual input channel, allowing the engineer to process a whole group of signals at once. Once again using the drum kit example, the use of bus-processing can enable the sound engineer to run all of the drum kit through an audio compressor effect to reduce unwanted signal peaks, rather than having to route all of the 10 or more mic signals on the drum kit individually. There may also be insert points for a certain bus, or even the entire mix.

VCA GROUPS

Some higher-end consoles use voltage-controlled amplifier (VCA) VCAs function somewhat like a submix but let the operator control the level of multiple input channels with a single fader. Unlike subgroups, no sub-mix is created. The audio signals from the assigned channels remain routed independently of VCA assignments. Since no sub-mix is created, it is not possible to insert processing such as compressors into a VCA/DCA group. In addition, on most VCA-equipped consoles, post-fader auxiliary send levels are affected by the VCA master. This is usually desirable, as post-fader auxiliary sends are commonly used for effects such as reverb, and sends to these effects should track changes in the channel signal level.

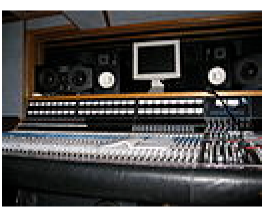
MASTER OUTPUT CONTROLS

The master control section is used to adjust the levels of the overall output of the mixer. The master control section on a large live venue or sound recording mixer typically has sub-group faders, master faders, master auxiliary mixing bus level controls and auxiliary return level controls. On most mixers, the master control is a fader. However, on some small mixers, rotary knobs are used instead to save space.

In a typical live sound mixing context, with a band playing at a venue, consisting of a rhythm section, solo instrumentalists and singers, the master control section allows the audio engineer to control the volume of the entire group with just one fader (for monaural mixers) or a pair of left and right faders (for stereo mixers).

Subgroup and main output fader controls are often found together on the right hand side of the mixer or, on larger consoles, in a center section flanked by banks of input channels. Matrix routing is often contained in this master section, as are headphone and local loudspeaker monitoring controls. Talkback controls allow conversation with the artist through their monitors, headphones or in-ear monitor. A test tone generator might be located in the master output section. Aux returns such as those signals returning from external processors are often in the master section.

METERING



Meter bridge on *ALegacy Plus* console

There are usually one or more VU or peak meters to indicate the levels for each channel, for the master outputs and to indicate whether the console levels are clipping the signal. The sound engineer typically adjusts the gain of the input signals to get the strongest signal that can be obtained without causing clipping. Having the gain set as high as possible improves the signal to noise ratio. Most mixers have at least one additional output besides the main mix. These are either individual bus outputs or *auxiliary outputs*, used, for instance, to output a different mix to onstage monitors.

The meters may be above the input and master sections or they may be integrated into the input and master sections themselves. Meters may have needles or LEDs. On meters using LEDs, there may be different colored LEDs to indicate when there is signal present in the channel's input; the audio level of the channel, typically by lighting up more LEDs; and clipping, which may be indicated using a different colored

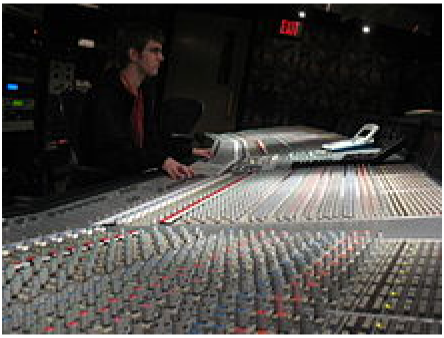
LED. In one popular color-coding system, green LEDs indicate signal presence and the audio level; one or more amber LEDs indicate that the channel is approaching clipping; and one or more red LEDs indicate clipping.

As the human ear experiences audio level in a logarithmic fashion, mixing console controls and displays are almost always labeled in decibels, a logarithmic measurement system. Since the decibel represents a relative measurement, and not a unit itself, the meters must be referenced to a nominal level. Most professional audio equipment is referenced to a nominal level of +4 dBu, while semi-professional and domestic equipment is usually referenced to a nominal level of −10 dBV.

ROUTING AND PATCHING

For convenience, some mixing consoles include inserts or a patch bay or patch panel. Patch bays are more common in recording mixers than live sound mixers. In live sound, the cables from the onstage microphones and instrument outputs are not typically plugged directly into the mixer, because this would require a large number of individual cables to go from the stage to the mixer. Instead, the onstage mic and instrument cables are typically plugged into the stage box of a snake cable which runs from the stage to the mixer. The snake is then plugged into the mixer.

OTHER FEATURES



A sound engineer at the controls of a SSL9000J mixer

Most, but not all, audio mixers can

- use monaural signals to produce simulated stereo sound through panning.
- provide phantom power required by condenser microphones.

Some mixers can

- add onboard external effect units (reverb, echo, delay). Mixers with onboard digital effects typically offer a wide range of these effects.
- create an audible test tone via an oscillator. The test tone can be used to troubleshoot issues before the band arrives and determine if channels are functioning properly.
- read and write console automation.
- be interfaced with computers or other recording

equipment.

- control or be controlled by a digital audio workstation via MIDI, USB or other communication interface.
- be powered by batteries.
- provide amplifier power for external passive speakers

MIRRORING

Some mixing consoles, particularly those designed for broadcast and live sound, include facilities for *mirroring* two consoles, making both consoles exact copies of each other with the same inputs and outputs, the same settings, and the same audio mix. There are two primary reasons for doing this; one, in the event of a hardware failure, a second redundant console is already in place and can be switched to (an important feature for live broadcasts); second, it allows the operators to set up two identical mix positions, one at front of house — where the audio will be mixed during a performance — and the other at some other location within the theater (e.g., with the broadcasting equipment); this way, if the acoustics at front of house are unfavorable, a mix can be programmed at an acoustically better position in the room, and the presets (on the faders and knobs) can be accessed from the front of house console during the performance.

DIGITAL VS. ANALOG

Digidesign's Venue Profile mixer on location at a corporate event. This digital mixer allows audio plug-ins from third-party vendors

See also: Comparison of analog and digital recording

Digital mixing console sales have increased dramatically since their introduction in the 1990s. Yamaha sold more than 1000 PM5D mixers by July, 2005, and other manufacturers are seeing increasing sales of their digital products. Digital mixers are more versatile than analog ones and offer many new features, such as reconfiguration of all signal routing at the touch of a button. In addition, digital consoles often include processing capabilities such as compression, gating, reverb, automatic feedback suppression and delay. Some products are expandable via third-party audio plug-ins that add further reverb, compression, delay and tone-shaping tools. Several digital mixers include spectrograph and real-time analyzer functions. A few incorporate loudspeaker management tools such as crossover filtering and limiting. Digital signal processing can perform automatic mixing for some simple applications, such as courtrooms, conferences and panel discussions.

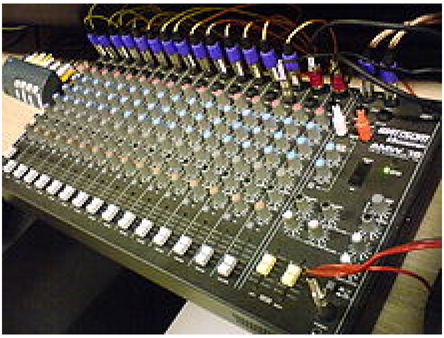
LATENCY

Digital mixers have an unavoidable amount of latency, ranging from less than 1 ms to as much as 10 ms, depending on the model of digital mixer and what functions are engaged. This small amount of latency is not a problem for loudspeakers aimed at the audience an not necessarily a problem for monitor wedges aimed at the artist, but can be disorienting and unpleasant for in-ear monitors where the artist hears their voice acoustically in their head *and* electronically amplified in their ears but delayed by a couple of milliseconds.

Every analog to digital conversion and digital to analog conversion within a digital mixer introduces latency. Audio inserts to favorite external analog processors make for approximately double the usual latency. Further latency can be traced to format conversions such as from ADAT to AES3 and from normal digital signal processing steps.

Within a digital mixer, there can be differing amounts of latency, depending on the routing and on how much DSP is in use. Assigning a signal to two parallel paths with significantly different processing on each path can result in comb filtering when recombined. Some digital mixers incorporate internal methods of latency correction so that such problems are avoided.

EASE OF USE



16-channel mixing console with compact short-throw faders

Analog consoles have a column of dedicated, physical knobs, buttons, and faders for each channel, which is logical and familiar to generations of audio engineers who have been trained on analog mixers. This takes more physical space but can accommodate rapid responses to changing performance conditions.

Most digital mixers use technology to reduce physical space requirements, entailing compromises in user interface such as a single shared channel adjustment area that is selectable for only one channel at a time. Additionally, most digital mixers have virtual pages or layers that change fader banks into separate controls for additional inputs or for adjusting equalization or aux send levels. This layering can be confusing for some operators. Many digital mixers allow internal reassignment of inputs so that convenient groupings of inputs appear near each other in the fader bank, a feature that can be disorienting for persons having to make a hardware patch change.

On the other hand, many digital mixers allow for extremely easy building of a mix from saved data. USB flash drives and other storage methods are employed to bring past performance data to a new venue in a highly portable manner. At the new venue, the traveling mix engineer simply plugs the collected data into the venue's digital mixer and quickly makes small adjustments to the local input and output patch layout, allowing for full show readiness in very short order. Some digital mixers allow offline editing of the mix, a feature that lets the traveling technician use a laptop to make anticipated changes to the show, shortening the time it takes to prepare the sound system for the artist.

SOUND QUALITY

A studio engineer at a Control 24 mixing surface

Both digital and analog mixers rely on analog microphone preamplifiers, a high-gain circuit that increases the low signal level from a microphone to a level that is better matched to the console's internal operating level. In this respect, both formats are on par with each other. In a digital mixer, the microphone preamplifier is followed by an analog-to-digital converter. Ideally, this process is carefully engineered to deal gracefully with overloading and clipping while delivering an accurate digital stream. Further processing and mixing of digital streams within a mixer need to avoid saturation if maximum audio quality is desired.

Analog mixers, too, must deal gracefully with overloading and clipping at the microphone preamplifier and as well as avoiding overloading of mix buses. Very high-frequency background hiss in an analog mixer is always present, though good gain stage management and turning unused channels down to zero minimizes its audibility. Idle subgroups left "up" in a mix add background hiss to the main outputs. Many digital mixers avoid this problem by low-level gating. Digital circuitry is more resistant to outside interference from radio transmitters such as walkie-talkies and cell phones. Hiss can be reduced with electronic noise reduction devices or with an equalizer.

Many electronic design elements combine to affect perceived sound quality, making the global "analog mixer vs. digital mixer" question difficult to answer. Experienced live sound professionals agree that the selection and quality of the microphones and loudspeakers (with their innate higher potential for creating distortion) are a much greater source of coloration of sound than the choice of mixer. The mixing style and experience of the person mixing may be more important than the make and model of audio console. Analog and digital mixers both have been associated with high-quality concert performances and studio recordings.

REMOTE CONTROL

Hip hop producer Chilly Chill behind a large audio console in a recording studio

Analog mixing in live sound has had the option since the 1990s of using wired remote controls for certain digital processes such as monitor wedge equalization and parameter changes in outboard reverb devices. That concept has expanded until wired and wireless remote controls are being seen in relation to entire digital mixing platforms. It is possible to set up a sound system and mix via laptop, touchscreen or tablet. Computer networks can connect digital system elements for expanded monitoring and control, allowing the system technician to make adjustments to distant devices during the performance. The use of remote control technology can be utilized to reduce the amount of venue space used for the front of house mixing console, nicknamed “seat-kills” in the music industry. As such, using remote control technologies in a venue can enable them to fit more paying customers into the venue.

SOFTWARE MIXERS

For recorded sound, the mixing process can be performed on screen, using computer software and associated input, output and recording hardware. The traditional large control surface of the mixing console is not utilized, saving space at the engineer's mix position. In a software studio, there is either no physical mixer fader bank at all or there is a compact group of motorized faders designed to fit into a small space and connected to the computer. Many project studios use such a space-efficient solution, as the mixing room at other times can serve as the business office, media archive, etc. Software mixing is heavily integrated as part of a digital audio workstation.

APPLICATIONS

A small four-channel mixer that could be used for a singer-guitarist's performance at a small coffeehouse.

Public address systems in schools, hospitals and other institutions use a mixing console to set microphones to an appropriate level and can add in recorded sounds such as music into the mix. PA mixers usually have controls that help to minimize audio feedback.

Most rock and pop bands use a mixing console to combine musical instruments and vocals so that the mix can be amplified through a nightclub's PA system. Among the highest quality bootleg recordings of live performances are so-called soundboard recordings sourced directly from the mixing console.

Radio broadcasts use a mixing desk to select audio from different sources, such as CD players, telephones, remote feeds, prerecorded advertisements, and in-studio live bands. These consoles, often referred to as "air-boards" are apt to have many fewer controls than mixers designed for live or studio production mixing, dropping pan/balance, EQ, and multi-bus monitoring/aux feed knobs in favor of cue and output bus selectors, since, in a radio studio, nearly all sources are either prerecorded or preadjusted.

DJs playing music for dancers at a dance club use a small DJ mixer to make smooth transitions between different songs which are played on sound sources that are plugged into the mixer. Compared with other mixers that are used in sound recording and live sound, DJ mixers have far fewer inputs. The most basic DJ mixers have only two inputs, though some have four or more inputs for DJs using a larger number of sound sources. These sound sources could include turntables, CD players, portable media players, or additional electronic instruments such as drum machines or synthesizers. The DJ mixer also allows the DJ to use headphones to cue the next song to the desired starting point before playing it.

Hip hop music DJs and Dub producers and engineers were early users of the mixing board as a musical instrument. In the 1970s, hip hop DJs developed a technique of adjusting the fader and crossfader controls of mixers at the same time as they manipulated records on turntables, creating unique rhythmic "scratching" effects.

Noise music musicians may create feedback loops within mixers, creating an instrument known as a no-input mixer. The tones generated from a no-input mixer are created by connecting an output of the mixer into an input channel and manipulating the pitch with the mixer's dials.

GALLERY

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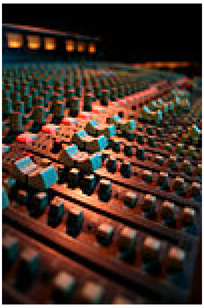
  BBC Local Radio Mark III radio mixing desk

- 

  Allen & Heath Mixing desk for live performance

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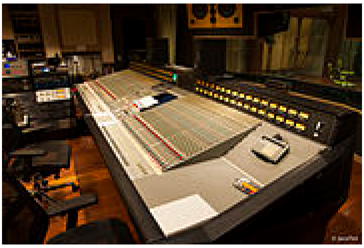
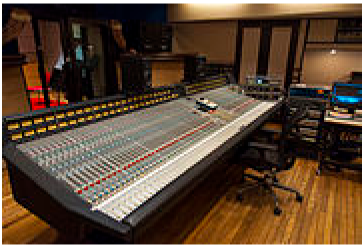
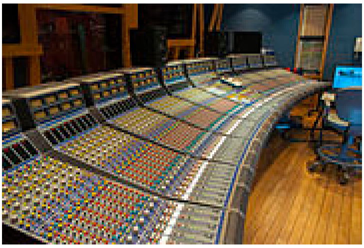
  Mackie CR1604-VLZ mixing console in a home studio

- 

  Solid State Logic SL9064J

- 

  Solid State Logic SL4064G+

- 

  Focusrite Console 72 in 48 out with GML Fader Automation

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  Harrison SeriesTEN

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  Gecko Exodus Odyssey MXR 5204L

USING MIXING CONSOLE AND RUNNING SOUND CHECK LINKS

Using a Mixing Console/Channel Strips

Live Sound Check

Mixing Live Sound

3

Unit One, Part Three: Speakers, Cables, and Microphone Stands

First things first, the **technical** aspects of audio speakers/headphones and cables are matters of personal and equipment safety. Serious attention should be paid to safety issues such as hearing damage, electrical shock, and falls – to name a few. Equipment and software can also be damaged by not using the proper cables or power cords. Please read all manuals and safety instructions before experimenting with this aspect of audio production.

Audio cables come in all sorts of lengths and gauges, each with its own style of connector end and specific use. Read and watch the links below to get your head around the dozens of different cables an audio producer will need to recognize. Become familiar with balanced/unbalanced cables and end connectors.

A lot of audio producers, including myself, are resigned to owning a big box of random cords, cables, and adapters, probably all tangled together and not labeled. However, for the best/safest use of equipment in a studio or around a stage, the most used cables like XLRs, 1/4″ instrument cables, headphone adapters, and USB adapters, should be wrapped up neatly, and placed in an accessible place. Consider labeling your cables clearly and include length. Many audio production spaces use dim lighting during performances, and you may need to grab a specific type and length of cable in the dark. For example, the theater on our campus keeps all their cables separated and organized on eye-level hooks along a labeled storage room wall. A small arena I once worked at as a weekend roadie kept all their miles of cables wrapped in large lettered gig boxes that could be safely wheeled across the building quickly to where they were needed. Even at my small home desk studio, my guitar cords and my two most used XLR cables are always wrapped up and hung from an unused music stand. Though mine is not the best system in terms of organization, it does keep them at arm's length and out of the way.

For live events and for recording, installing the right speakers for the job can really enhance audio production (**method**). I've found that sometimes really expensive or really large/loud monitor speakers do not translate outside the studio. Often a mid-priced or even archaic set of old school stereo speakers give the truest representation of what will be heard on any system.

When choosing what kind of speakers you want to use consider the following:

How loud do they need to be, and do I have the proper power to run them?

Are the speakers in acoustic position (this is covered in a later part of the manual)?

Do the speakers fit the space?

Are they applicable to all the situations needed?

How many speakers are required for the situation?

For example, installing huge 1000 watt speakers for a small studio space might not be the most efficient use of the area or be needed to go that loud. Additionally, plans to use only two powered p.a. speakers for a live event in a gymnasium might not be loud enough for the band to hear themselves.

Try to fit the size, loudness, style, and number of speakers to the situation. Understanding that will help an audio producer make better **technical** and **creative choices**.

One thing I've learned over the years is that if my job requires me to listen, I need to get a set of monitor speakers and headphones I'm not constantly fighting with. My personal workstation is small, so I chose smaller monitor speakers. I also needed speakers that could connect from one piece of recording gear (laptop) to an older reel to reel eight-track. And, though I like loud, I have people living in the house with me, so I chose some characteristics of clarity over volume. The money I saved allowed me to upgrade on headphones. And even then, I didn't have to go over budget. Since I work a lot from this permanent audio workstation, I didn't need the sturdiest cans or wireless earbuds. My home set-up has the proper speakers for the space, a nice set of noise-cancelling headphones, and I stayed under budget (**method**). I've included materials below to understand speakers and cables better.

How to Wrap Your Cables

A Beginners Guide to Hi-Fi Speakers–Video

How to Set Up a PA System (live band) Video

Balanced vs Unbalanced Cables

A Quick DJ Guide to Cables

AUDIO CABLES

Audio Cables

Audio Cables (balanced and unbalanced) Landr

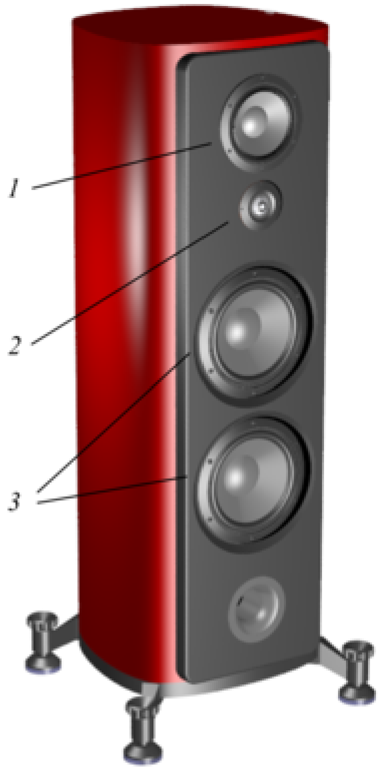
SPEAKERS

A loudspeaker is an electroacoustic transducer; a device which converts an electrical audio signal into a corresponding sound. The most widely used type of speaker is the dynamic speaker. The sound source (e.g., a sound recording or a microphone) must be amplified or strengthened with an audio power amplifier before the signal is sent to the speaker.

The dynamic speaker was invented in 1925 by Edward W. Kellogg and Chester W. The dynamic speaker operates on the same basic principle as a dynamic microphone, but in reverse, to produce sound from an electrical signal. When an alternating current electrical audio signal is applied to its voice coil, a coil of wire suspended in a circular gap between the poles of a permanent magnet, the coil is forced to move rapidly back and forth due to Faraday's law of induction, which causes a diaphragm (usually conically shaped) attached to the coil to move back and forth, pushing on the air to create sound waves. Besides this most common method, there are several alternative technologies that can be used to convert an electrical signal into sound.

Speakers are typically housed in a speaker enclosure or speaker cabinet which is often a rectangular box made of wood or sometimes plastic. The enclosure's materials and design play an important role in the quality of the sound. The enclosure generally must be as stiff and non-resonant as practically possible. Where high fidelity reproduction of sound is required, multiple loudspeaker transducers are often mounted in the same enclosure, each reproducing a part of the audible frequency range (picture at right). In this case, the individual speakers may be referred to as drivers and the entire unit is called a loudspeaker. Drivers made for reproducing high audio frequencies are called tweeters, those for middle frequencies are called mid-range drivers and those for low frequencies are called woofers. Extremely low frequencies (16Hz–~100Hz) may be reproduced by separate subwoofers.

Smaller loudspeakers are found in devices such as radios, televisions, portable audio players, computers, and electronic musical instruments. Larger loudspeaker systems are used for music, sound reinforcement in theatres and concert halls, and in public address systems.

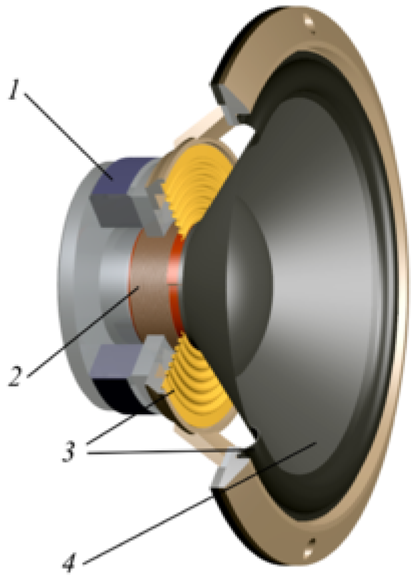


Loudspeaker for home use with three types of dynamic drivers

1. Mid-range driver
2. Tweeter
3. Woofers

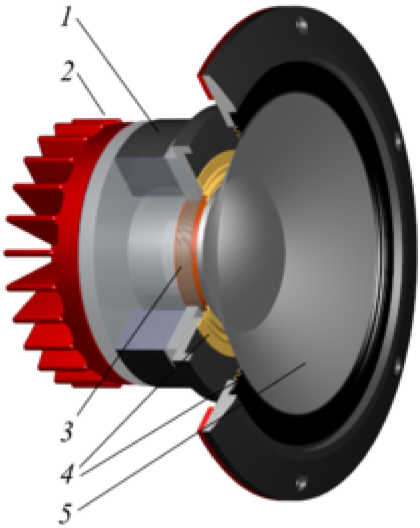
The hole below the lowest woofer is a port for a bass reflex system.

DRIVER DESIGN: DYNAMIC LOUDSPEAKERS



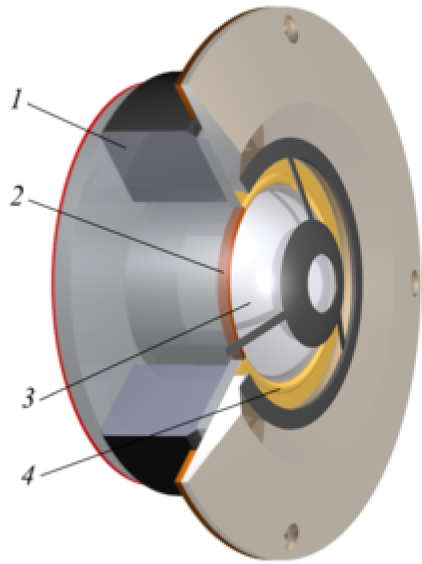
Cutaway view of a dynamic loudspeaker for the bass register.

1. Magnet
2. Voice coil
3. Suspension
4. Diaphragm



Cutaway view of a dynamic midrange speaker.

1. Magnet
2. Cooler (sometimes present)
3. Voice coil
4. Suspension
5. Diaphragm

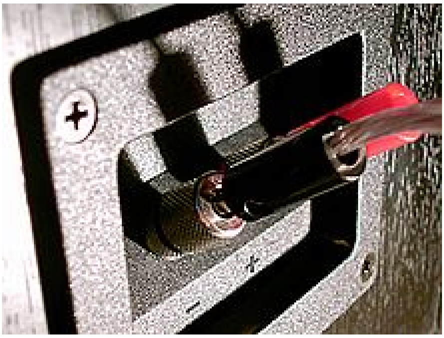


Cutaway view of a dynamic tweeter with acoustic lens and a dome-shaped membrane.

1. Magnet
2. Voice coil
3. Diaphragm
4. Suspension

The most common type of driver, commonly called a dynamic loudspeaker, uses a lightweight diaphragm, or cone, connected to a rigid basket, or frame, via a flexible suspension, commonly called a spider, that constrains a voice coil to move axially through a cylindrical magnetic gap. A protective cap glued in the cone's center prevents dust, especially iron filings, from entering the gap. When an electrical signal is applied to the voice coil, a magnetic field is created by the electric current in the voice coil, making it a variable electromagnet. The coil and the driver's magnetic system interact, generating a mechanical force that causes the coil (and thus, the attached cone) to move back and forth, accelerating and reproducing sound under the control of the applied electrical signal coming from the amplifier. The following is a description of the individual components of this type of loudspeaker.

WIRING CONNECTIONS



Two-way binding posts on a loudspeaker, connected using banana plugs.

A 4-ohm loudspeaker with two pairs of binding posts capable of accepting bi-wiring after the removal of two metal straps.

Most home hi-fi loudspeakers use two wiring points to connect to the source of the signal (for example, to the audio amplifier or receiver). To accept the wire connection, the loudspeaker enclosure may have binding posts, spring clips, or a panel-mount jack. If the wires for a pair of speakers are not connected with respect to the proper electrical polarity (the + and − connections on the speaker and amplifier should be connected + to + and − to −; speaker cable is almost always marked so that one conductor of a pair can be distinguished from the other, even if it has run under or behind things in its run from amplifier to speaker location), the loudspeakers are said to be "out of phase" or more properly "out of polarity." Given identical signals, motion in one cone is in the opposite direction of the other. This typically causes monophonic material in a stereo recording to be canceled out, reduced in level, and made more difficult to localize, all due to destructive interference of the sound waves. The cancellation effect is most noticeable at frequencies where the loudspeakers are separated by a quarter wavelength or less; low frequencies are affected the most. This type of miswiring error does not damage speakers, but is not optimal for listening.

With sound reinforcement system, PA system and instrument amplifier speaker enclosures, cables and some type of jack or connector are typically used. Lower- and mid-priced sound system and instrument speaker cabinets often use 1/4″ speaker cable jacks. Higher-priced and higher powered sound system cabinets and instrument speaker cabinets often use Speakon connectors. Speakon connectors are considered to be safer for high wattage amplifiers, because the connector is designed so that human users cannot touch the connectors.

WIRELESS SPEAKERS

Main article: Wireless speaker

HP Roar Wireless Speaker

Wireless speakers are very similar to traditional (wired) loudspeakers, but they receive audio signals using radio frequency (RF) waves rather than over audio cables. There is normally an amplifier integrated in the speaker's cabinet because the RF waves alone are not enough to drive the speaker. This integration of amplifier and loudspeaker is known as an active loudspeaker. Manufacturers of these loudspeakers design them to be as lightweight as possible while producing the maximum amount of audio output efficiency.

Wireless speakers still need power, so require a nearby AC power outlet, or possibly batteries. Only the wire to the amplifier is eliminated.

SPECIFICATIONS

Specifications label on a loudspeaker
Speaker specifications generally include:

- **Speaker or driver type** (individual units only) – Full-range, woofer, tweeter, or mid-range.

- **Size** of individual drivers. For cone drivers, the quoted size is generally the outside diameter of the basket. However, it may less commonly also be the diameter of the cone surround, measured apex to apex, or the distance from the center of one mounting hole to its opposite. Voice-coil diameter may also be specified. If the loudspeaker has a compression horn driver, the diameter of the horn throat may be given.
- **Rated Power** – Nominal (or even continuous) power, and peak (or maximum short-term) power a loudspeaker can handle (i.e., maximum input power before destroying the loudspeaker; it is never the sound output the loudspeaker produces). A driver may be damaged at much less than its rated power if driven past its mechanical limits at lower frequencies.[43] Tweeters can also be damaged by amplifier clipping (amplifier circuits produce large amounts of energy at high frequencies in such cases) or by music or sine wave input at high frequencies. Each of these situations might pass more energy to a tweeter than it can survive without damage.[44] In some jurisdictions, power handling has a legal meaning allowing comparisons between loudspeakers under consideration. Elsewhere, the variety of meanings

for power handling capacity can be quite confusing.

- **Impedance –** typically 4 Ω (ohms), 8 Ω, etc.
- **Baffle or enclosure type** (enclosed systems only) – Sealed, bass reflex, etc.
- **Number of drivers** (complete speaker systems only) – two-way, three-way, etc.
- **Class** of loudspeaker:[46]
    - Class 1: maximum SPL 110–119 dB, the type of loudspeaker used for reproducing a person speaking in a small space or for background music; mainly used as fill speakers for Class 2 or Class 3 speakers; typically small 4″ or 5″ woofers and dome tweeters
    - Class 2: maximum SPL 120–129 dB, the type of medium power-capable loudspeaker used for reinforcement in small to medium spaces or as fill speakers for Class 3 or Class 4 speakers; typically 5″ to 8″ woofers and dome tweeters
    - Class 3: maximum SPL 130–139 dB, high power-capable loudspeakers used in main systems in small to medium spaces; also used as fill speakers for class 4 speakers; typically 6.5″ to 12″ woofers and 2″ or 3″

compression drivers for high frequencies

    - Class 4: maximum SPL 140 dB and higher, very high power-capable loudspeakers used as mains in medium to large spaces (or for fill speakers for these medium to large spaces); 10″ to 15″ woofers and 3″ compression drivers

and optionally:

- **Crossover frequency(ies)** (multi-driver systems only) – The nominal frequency boundaries of the division between drivers.
- **Frequency response** – The measured, or specified, output over a specified range of frequencies for a constant input level varied across those frequencies. It sometimes includes a variance limit, such as within "± 2.5 dB."
- **Thiele/Small parameters** (individual drivers only) – these include the driver's F_s (resonance frequency), Q_{ts} (a driver's Q; more or less, its damping factor at resonant frequency), V_{as} (the equivalent air compliance volume of the driver), etc.
- **Sensitivity** – The sound pressure level produced by a loudspeaker in a non-reverberant environment, often specified in dB and measured

at 1 meter with an input of 1 watt (2.83 rms volts into 8 Ω), typically at one or more specified frequencies. Manufacturers often use this rating in marketing material.

- **Maximum sound pressure level** – The highest output the loudspeaker can manage, short of damage or not exceeding a particular distortion level. Manufacturers often use this rating in marketing material—commonly without reference to frequency range or distortion level.

Microphone Stands

A **microphone stand** is a free-standing mount for a microphone. It allows the microphone to be positioned in the studio, on stage or on location without requiring a person to hold it.

The most basic microphone stand is a *straight stand*. It uses a dome-shaped round metal base, or a tripod base, into which is threaded a post for mounting the microphone (most commonly a 5/8-27 threaded hole). This post may be made up of two or more telescoping tubes that fit inside each other, allowing for quick height adjustment. The mechanism for adjusting the height is called the clutch.[1]

There are various versions of the straight stand known as the "desk stand" (short version of straight stand) and heavy duty microphone stand (heavier base and larger tubes) to handle heavy microphones.[2] The tubes used on the straight stand usually have a shiny chrome plating to resist scratching, but may also be finished in a matte black.

A very popular updated version of the straight stand uses the "folding tripod base stand", instead of the round, domed metal base. This folding base allows for easier packing of the stand when moving from location to location and reduces the weight of the stand. However, to compensate for the lack of weight at the base while still maintaining stability, the three "feet" of the tripod must extend out beyond the radius of a round base. The trade-off is that these "feet" may become a trip-hazard on a dark stage.

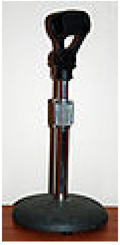
A number of accessories make microphone stands more useful. Most of these are designed to get the microphone closer to the user without placing the upright portion of the stand directly in front of the performer.

A "boom arm" attaches to the top of the stand so the microphone can move in the horizontal plane. A guitar player, for example, might use this to place the microphone directly in front of his mouth without having the upright portion of the stand in the way of the guitar. It also lets musicians have the microphone closer to the sound source when floor space is at a premium. This can be particularly useful when placing microphones on a drum stand when the microphone stands must compete for space with things like cymbal stands. Boom arms are offered both in fixed length and adjustable (telescoping) lengths.

Another handy device for adjusting microphone placement is a flexible *goose neck* tube. Made of a spiral-wound core of steel, goosenecks are made in various lengths and finishes and provide the ability to make minute changes in microphone position.

Microphones typically attach to the stand via a detachable microphone holder screwed to the end of the boom.

Various male/female adapters are available to connect dissimilar sizes. Note: A compatible ¼″ 20 tpi UNC is common in photography tripods

- 

  Desktop microphone stand

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  Shure model S36 desktop microphone stand

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  Microphone stand with folding tripod base, boom arm, and shock mount.

  Shock Mount Microphone Clips

SHOCK MOUNTS

Shock mounts for microphones can provide basic protection from damage, but their prime use is to isolate microphones from mechanically transmitted noise. This can originate as floor vibrations transmitted through a floor stand, or as “finger” and other handling noise on boom poles. All microphones behave to some extent as accelerometers, with the most sensitive axis being perpendicular to the diaphragm. Additionally, some microphones contain internal elements such as vacuum tubes and transformers which can be inherently microphonic. These are often cushioned by resilient internal methods, in addition to the employment of external isolation mounts.

4

Unit One, Part Four: Mixing Live Project and Microphone Placement

At this point of the unit, we should have a basic understanding of microphones, mixing consoles, cables, and speakers. Now we need to put those things together to create a “live mix” for a performer or a small group. Whether we have to set up one microphone for a speaking event or 16 separate channels for a full band on a large stage, *where* we place the microphones and *what* microphones we choose will quite often determine how well we manage sound for that event. These techniques can be applied to recording sessions or rehearsal spaces as well.

When I work with microphones and a mixing console for a live event, I start by following the general rules of microphone placement and channel strip settings (**method**). I also try to use the appropriate equipment to fit the specifications of the event and for safety (**technical**). But I almost always keep an open mind/ear to hear if there are any creative tweaks I can implement to give the overall sound an aesthetic upgrade **(creative approach)**. Often, subtle artistic uses of panning or effects can really heighten the quality of the performance.

GENERAL TECHNIQUE

A microphone should be used whose frequency response will suit the frequency range of the voice or instrument being recorded.

Vary microphone positions and distances until you achieve the monitored sound that you desire.

In the case of poor room acoustics, place the microphone very close to the loudest part of the instrument being recorded or isolate the instrument.

Personal taste is the most important component of microphone technique. Whatever sounds right to you, is right.

WORKING DISTANCE

Close Miking

When miking at a distance of 1 inch to about 3 feet from the sound source, it is considered close miking. This technique generally provides a tight, present sound quality and does an effective job of isolating the signal and excluding other sounds in the acoustic environment.

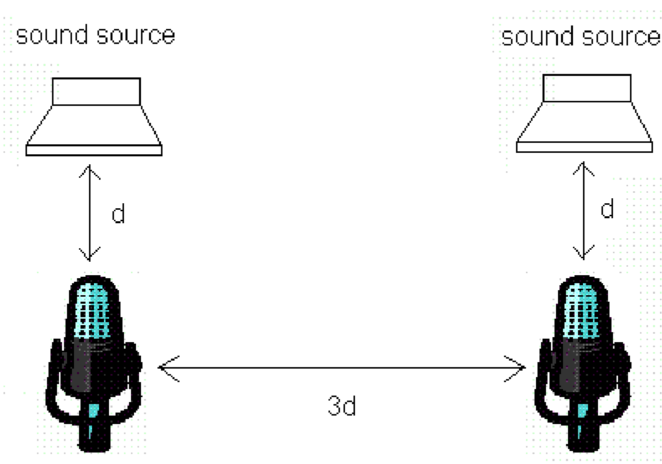
Leakage

Leakage occurs when the signal is not properly isolated and the microphone picks up another nearby instrument. This can make the mixdown process difficult if there are multiple voices on one track. Use the following methods to prevent leakage:

Place the microphones closer to the instruments.
Move the instruments farther apart.
Put some sort of acoustic barrier between the instruments.
Use directional microphones.

3 to 1 Rule

The 3:1 distance rule is a general rule of thumb for close miking. To prevent phase anomalies and leakage, the microphones should be placed at least three times as far from each other as the distance between the instrument and the microphone.



PLACEMENT FOR VARYING INSTRUMENTS

Amplifiers

When miking an amplifier, such as for electric guitars, the mic should be placed 2 to 12 inches from the speaker. Exact placement becomes more critical at a distance of less than 4 inches. A brighter sound is achieved when the mic faces directly into the center of the speaker cone and a more mellow sound is produced when placed slightly off-center. Placing off-center also reduces amplifier noise.

<u>Brass Instruments</u>

High sound-pressure levels are produced by brass instruments due to the directional characteristics of mid to mid-high frequencies. Therefore, for brass instruments such as trumpets, trombones, and tubas, microphones should face slightly off of the bell's center at a distance of one foot or more to prevent overloading from windblasts.

<u>Guitars</u>

Technique for acoustic guitars is dependent on the desired sound. Placing a microphone close to the sound hole will achieve the highest output possible, but the sound may be bottom-heavy because of how the sound hole resonates at low frequencies. Placing the mic slightly off-center at 6 to 12 inches from the hole will provide a more balanced pickup. Placing the mic closer to the bridge with the same working distance will ensure that the full range of the instrument is captured.

Some people prefer to use a contact microphone, attached (usually) by a fairly weak temporary adhesive, however this will give a rather different sound to a conventional microphone. The primary advantage is that the contact microphone performance is unchanged as the guitar is moved around during a performance, whereas with a conventional microphone on a stand, the distance between microphone and guitar would be subject to continual variation. Placement of a contact microphone can be adjusted by trial and error to get a variety of sounds. The same technique works quite well on other stringed instruments such as violins.

Pianos

Ideally, microphones would be placed 4 to 6 feet from the piano to allow the full range of the instrument to develop before it is captured. This isn't always possible due to room noise, so the next best option is to place the microphone just inside the open lid. This applies to both grand and upright pianos.

Percussion

One overhead microphone can be used for a drum set, although two are preferable. If possible, each component of the drum set should be miked individually at a distance of 1 to 2 inches as if they were their own instrument. This also applies to other drums such as congas and bongos. For large, tuned instruments such as xylophones, multiple mics can be used as long as they are spaced according to the 3:1 rule.

Voice

Standard technique is to put the microphone directly in front of the vocalist's mouth, although placing slightly off-center can alleviate harsh consonant sounds (such as "p") and prevent overloading due to excessive dynamic range. Several sources also recommend placing the microphone slightly above the mouth.

Woodwinds

A general rule for woodwinds is to place the microphone around the middle of the instrument at a distance of 6 inches to 2 feet. The microphone should be tilted slightly towards the bell or sound hole, but not directly in front of it.

STEREO (2) MICROPHONE PLACEMENT

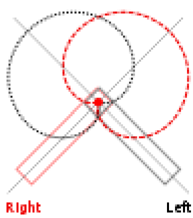
There exist a number of well-developed microphone techniques used for recording musical, film, or voice sources. Choice of technique depends on a number of factors, including:

- The collection of extraneous noise. This can be a concern, especially in amplified performances, where audio feedback can be a significant problem. Alternatively, it can be a desired outcome, in situations where ambient noise is useful (hall reverberation, audience reaction).
- Choice of a signal type: Mono, stereo or multi-channel.
- Type of sound-source: Acoustic instruments produce a sound very different from electric instruments, which are again different from the human voice.
- Situational circumstances: Sometimes a microphone should not be visible, or having a

microphone nearby is not appropriate. In scenes for a movie the microphone may be held above the picture frame, just out of sight. In this way there is always a certain distance between the actor and the microphone.

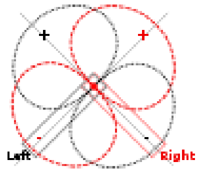
- Processing: If the signal is destined to be heavily processed, or "mixed down", a different type of input may be required.
- The use of a windshield as well as a pop shield, designed to reduce vocal plosives.

X-Y Technique



XY Stereo

Here there are two *directional* microphones at the same place, and typically placed at 90° or more to each other. A stereo effect is achieved through differences in sound pressure level between two microphones. Due to the lack of differences in time-of-arrival and phase ambiguities, the sonic characteristic of X-Y recordings is generally less "spacey" and has less depth compared to recordings employing an AB setup.



Blumlein Stereo

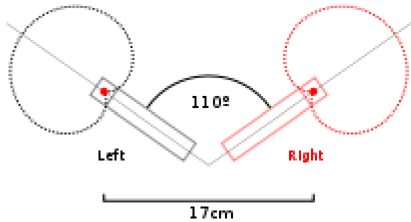
When the microphones are *bidirectional* and placed facing +-45° with respect to the sound source, the X-Y-setup is called a Blumlein Pair. The sonic image produced by this configuration is considered by many authorities to create a realistic, almost holographic soundstage.

A further refinement of the Blumlein Pair was developed by EMI in 1958, who called it "Stereosonic". They added a little in-phase crosstalk above 700 Hz to better align the mid and treble phantom sources with the bass ones.

<u>A-B Technique</u>

This technique uses two parallel microphones, typically omnidirectional, some distance apart, capturing time-of-arrival stereo information as well as some level (amplitude) difference information, especially if employed close to the sound source(s). At a distance of about 50 cm (0.5 m) the time delay for a signal reaching first one and then the other microphone from the side is approximately 1.5 ms (1 to 2 ms). If the distance is increased between the microphones it effectively decreases the pickup angle.

<u>The ORTF Technique</u>



It was devised around 1960 at the Office de Radiodiffusion Télévision Française (ORTF) at Radio France.

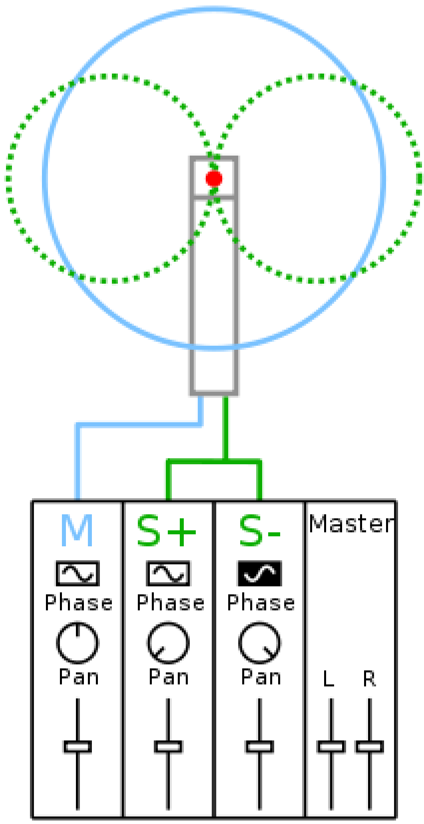
ORTF combines both the volume difference provided as sound arrives on- and off-axis at two cardioid microphones spread to a 110° angle, as well as the timing difference as sound arrives at the two microphones spaced 17 cm apart.

The microphones should be as similar as possible, preferably a frequency-matched pair of an identical type and model.

The result is a realistic stereo field that has reasonable compatibility with mono playback. Since the cardioid polar pattern rejects off-axis sound, less of the ambient room characteristics are picked up. This means that the mics can be placed farther away from the sound sources, resulting in a blend that may be more appealing. Further, the availability of purpose-built microphone mounts makes ORTF easy to achieve.

As with all microphone arrangements, the spacing and angle can be manually adjusted slightly by ear for the best sound, which may vary depending on room acoustics, source characteristics, and other factors. But this arrangement is defined as it is because it was the result of considerable research and experimentation, and its results are predictable and repeatable.

Mid/Side Technique



Mid-Side Stereo

This technique employs a bidirectional microphone (with a Figure of 8 polar pattern) facing sideways and a cardioid (generally a variety of cardioid, although Alan Blumlein described the usage of an omnidirectional transducer in his original patent) facing the sound source. The capsules are stacked vertically and brought together as closely as possible, to minimize comb filtering caused by differences in arrival time.

The left and right channels are produced through a simple matrix: Left = Mid + Side, Right = Mid − Side ("minus" means you add the side signal with the polarity reversed). This configuration produces a completely mono-compatible signal and, if the Mid and Side signals are recorded (rather than the matrixed Left and Right), the stereo width (and with that, the perceived distance of the sound source) can be manipulated after the recording has taken place.

LINKS FOR MICROPHONE PLACEMENTS

Microphone Placement on Instruments

Stereo Mic Techniques

Vocal Mics in the Studio

MIXING A LIVE BAND

Live sound mixing is the blending of multiple sound sources by an audio engineer using a mixing console or software. Sounds that are mixed include those from instruments and voices which are picked up by microphones (for drum kit, lead vocals and acoustic instruments like piano or saxophone and pickups for instruments such as electric bass) and pre-recorded material, such as songs on CD or a digital audio player. Individual sources are typically equalized to adjust the bass and treble response and routed to effect processors to ultimately be amplified and reproduced via a loudspeaker system.[1] The live sound engineer listens and balances the various audio sources in a way that best suits the needs of the event.[2]

Audio equipment is usually connected together in a sequence known as the signal chain. In live sound situations, this consists of input transducers like microphones, pickups, and DI boxes. These devices are connected, often via multicore cable, to individual channels of a mixing console. Each channel on a mixing console typically has a vertical "channel strip", which is a column of knobs and buttons which are used to adjust the level and the bass, middle register and treble of the signal. The audio console also typically allows the engineer to add effects units to each channel (addition of reverb, etc.) before they are electrically summed (blended together).

Audio signal processing may be applied to (inserted on) individual inputs, groups of inputs, or the entire output mix, using processors that are internal to the mixer or external (outboard effects, which are often mounted in 19″ racks). An example of an inserted effect on an individual input is patching in an Autotune rackmount unit onto the lead vocalist's track to correct pitch errors. An example of using an inserted effect on a group of inputs would be to add reverb to all of the vocalists' channels (lead vocalist and backing vocalists). An example of adding effects to the entire output mix would be to use a graphic equalizer to adjust the frequency response of the entire mix.

Front of House Mixing

The front of house (FOH) engineer focuses on mixing audio for the audience, and most often operates from the middle of the audience or at the last few rows of the audience. The output signals from the FOH console connects to a Sound reinforcement system. Other non-audio crew members, such as the lighting console operator, might also work from the FOH position, since they need to be able to see the show from the audience's perspective.

Foldback

The foldback or monitor engineer focuses on mixing the sound that the performers hear on stage via a stage monitor system (also known as the foldback system). The monitor engineer's role is important where the instruments and voices on the stage area is amplified. Usually, individual performers receive personalized feeds either via monitors placed on the stage floor in front of them or via in-ear monitors. The monitor engineer's console is usually placed in the wings just off-stage, to provide easier communication between the performers and the monitor engineer.

For smaller shows, such as bar and smaller club gigs, it is common for the monitors to be mixed from the front of house position, and the number of individual monitor mixes could be limited by the capabilities of the front of house mixing desk. In smaller clubs with lower- to mid-priced audio consoles, the audio engineer may only have a single "auxiliary send" knob on each channel strip. With only one "aux send", an engineer would only be able to make a single monitor mix, which would normally be focused on meeting the needs of the lead singer. Larger, more expensive audio consoles may provide the capabilities to make multiple monitor mixes (e.g., one mix for the lead singer, a second mix for the backing vocalist, and a third for the rhythm section musicians). In a noisy club with high-volume rock music groups, monitor engineers may be asked for just the vocals in the monitors. This is because in a rock band, the guitarist, bassist and keyboardist typically have their own large amplifiers and speakers, and rock drums are loud enough to be heard acoustically. In large venues, such as outdoor festivals, bands may request a mix of the full band through the monitors, including vocals and instruments.

Drummers generally want a blend of all of the onstage instruments and vocals in their monitor mix, with extra volume provided for bass drum, electric bass and guitar. Guitar players typically want to hear the bass drum, other guitars (e.g., rhythm guitar) and the vocals. Bass players typically ask for a good volume of bass drum along with the guitars. Vocalists typically want to hear their own vocals. Vocalists may request other instruments in their monitor mix, as well.

Broadcast

The broadcast mixer is responsible for audio delivered for radio or television broadcast. Broadcast mixing is usually performed in an OB van parked outside the venue.

Sound Checks and Technical Rehearsals

For small events, often a soundcheck is conducted a few hours before the show. The instruments (drum kit, electric bass and bass amplifier, etc.) are set up on stage, and the engineer places microphones near the instruments and amplifiers in the most appropriate location to pick up the sound and some instruments, such as the electric bass, are connected to the audio console via a DI box. Once all the instruments are set up, the engineer asks each instrumentalist to perform alone, so that the levels and equalization for the instruments can be adjusted. Since a drum kit contains a number of drums, cymbals and percussion instruments, the engineer typically adjusts the level and equalization for each mic'd instrument. Once the sound of each individual instrument is set, the engineer asks the band to play a song from their repertoire, so that the levels of one instrument versus another can be adjusted.

Mics are set up for the lead vocalist and backup vocalist and the singers are asked to sing individually and as a group, so that the engineer can adjust the levels and equalization. The final part of the soundcheck is to have the rhythm section and all the vocalists perform a song from their repertoire. During this song, the engineer can adjust the balance of the different instruments and vocalists. This allows the sound of the instruments and vocals to be fine-tuned prior to the audience hearing the first song.

In the 2010s, many professional bands and major venues use digital mixing consoles that have automated controls and digital memory for previous settings. The settings of previous shows can be saved and recalled in the console and a band can start playing with a limited soundcheck. Automated mixing consoles are a great time saver for concerts where the main band is preceded by several support acts. Using an automated console, the engineer can record the settings that each band asks for during their individual soundchecks. Then, during the concert, the engineer can call up the settings from memory, and the faders will automatically move to the position that they were placed in during the soundcheck. On a larger scale, technical rehearsals may be held in the days or weeks leading up to a concert. These rehearsals are used to fine-tune the many technical aspects (such as lighting, sound, video) associated with a live performance.

Training and Background

Audio engineers must have extensive knowledge of audio engineering to produce a good mix, because they have to understand how to mix different instruments and amplifiers, what types of mics to use, where to place the mics, how to attenuate "hot" instrument signals that are overloading and clipping the channel, how to prevent audio feedback, how to set the audio compression for different instruments and vocals, and how to prevent unwanted distortion of the vocal sound. However, to be a professional live sound engineer, an individual needs more than just technical knowledge. They must also understand the types of sounds and tones that various musical acts in different genres expect in their live sound mix. This knowledge of musical styles is typically learned from years of experience listening to and mixing sound in live contexts. A live sound engineer must know, for example, the difference between creating the powerful drum sound that a heavy metal drummer will want for their kit, versus a drummer from a Beatles tribute band.

LINKS FOR MIXING LIVE SITUATIONS

How to Set Up a PA System

Mixing a Live Band

Four Microphone Drum Set-Up

10 Ways to Mic a Guitar Amp

Live Vocal EQ

EQ CHEAT SHEET

ACOUSTIC DRUMS

Be careful when boosting top end on close mic'd tracks. This can accentuate cymbal bleed and make the drums sound harsh. If you're struggling to achieve brightness without bringing up the cymbals, try using gates to reduce the bleed between hits. You can also layer in drum samples (my first choice) and EQ them for brightness without bringing up the bleed.

There are two approaches for EQing overheads. You can either:

- Filter out all the low end and use them as cymbal mics, or…

- Leave them as-is and use them to form the overall sound of the kit

Option #1 will create a separated, sculpted sound that works well in modern genres. Option #2 will lead to a more natural sound that works well for folk and acoustic music.

Try rolling off everything below 40 Hz on the kick. This can often tighten things up.

Since each mic has bleed, you should always EQ drums with all the mics playing together. You can often filter the hi-hat aggressively. Try cutting everything below 500 Hz.

Acoustic drums will often need lots of EQ. Don't be afraid to boost or cut by 10 dB or more.

Listen to a few modern records and notice how bright the kick is. The key to getting your kick to cut isn't more low end, but more top end.

VOCALS

When boosting top end, listen for harshness. Often times, this will occur when your boost extends too far down the frequency spectrum. If this happens, move the boost higher up or tighten the Q. You can also add a small cut to the upper midrange to counteract any harshness.

High-pass filtering will often be necessary, but it's not always needed. If you don't hear a problem, there's no need to fix it.

You can high-pass female vocals much higher than male vocals without affecting the sound of the voice.

Listen for resonances in the lower midrange.

BASS

To add presence, boost higher than you think. The solution is not more low end, but to bring out the harmonics (start around 700 – 1200 Hz).

Distortion will often do a better job at adding presence than EQ.

ACOUSTIC GUITARS

Don't be afraid to roll off the low end. Often times, all you want is the sound of the pick hitting the strings.

Watch for resonances in the lower midrange.

ELECTRIC GUITARS

Listen to them with the bass. Often times, you can remove quite a bit of low end. The guitars may sound thin on their own, but with the bass, they'll sound great.

Watch for harsh resonances in the upper midrange (2 – 4 kHz).

If electric guitars are recorded well, they often need little-to-no EQ. For presence, boost around 4 kHz.

SYNTHS

Your approach should vary widely depending on what you're working with.

Many modern synths are ear-piercingly bright. Don't be afraid to roll off top end. This can help them sink back into a mix.

Often times, synths will fill up the entire frequency spectrum. In a busy mix, you'll often have to whittle them down using high and low-pass filters.

You can be aggressive when EQing synths, because we have no expectations about what they should sound like. This gives you more flexibility than when EQing an organic instrument, where you can't stray too far from what the instrument sounds like in real life.

PIANO

Boosting 5 kHz can bring out the sound of the hammers hitting the strings. This will make the piano sound harder, which can help it cut through a busy mix.

5

Unit Two, Part One: History of Audio Recording

I love learning about the history of audio recording. I find it interesting not only on all three levels of our production pillars, but it's a fascinating way to look at (and listen to) human history. Generally, historians recognize the mid to late 1800s as the dawn of audio recording. Some have even argued that *true* beginning of managed sound didn't start until the digital era (1980s) and consider most early analog recordings as non-interactive representations. They regard binary notation as the true language of produced sound. Though I disagree with that analysis, I do see some interesting points in it. The other side of that argument is the great debate between analog sound versus digital sound. I feel, in some ways, lucky because I got to work in both eras of production and discovered value to all styles of audio production and recording.

The more I read about the history of audio production and recording, the more I find that the time line goes back much further than 1875. In fact, some very old temples, dating back thousands of years, are acoustically treated by their designers and some hand carved musical instruments have been dated back even older than that.

I encourage students to read and listen to and about older formats of audio recordings. Imagine what it was like to hear an Italian opera for the first time on a shellac disc in the 1920s. What methods and technologies did the audio engineers employ before electricity? I firmly believe that the best way to advance our own audio production quality is to combine the old with the new and keep an ear open for unique ways from every era to manage those sounds.

Courtesy: National Archives and Records Administration

[The Architecture of Sound (TED Talk)]
[Greg Milner "Perfecting Sound Forever"]
[Brief History of Audio Recording]

A BRIEF HISTORY OF SOUND RECORDING

Experiments in capturing sound on a recording medium for preservation and reproduction began in earnest during the Industrial Revolution of the 1800s. Many pioneering attempts to record and reproduce sound were made during the latter half of the 19th century – notably Scott's phonautograph of 1857 – and these efforts culminated in the invention of the phonograph by Thomas Edison in 1877. Digital recording emerged in the late 20th century and has since flourished with the popularity of digital music and online streaming services.

The Acoustic Era (1877–1925)

The earliest practical recording technologies were entirely mechanical devices. These recorders typically used a large conical horn to collect and focus the physical air pressure of the sound waves produced by the human voice or musical instruments. A sensitive membrane or diaphragm, located at the apex of the cone, was connected to an articulated scriber or stylus, and as the changing air pressure moved the diaphragm back and forth, the stylus scratched or incised an analogue of the sound waves onto a moving recording medium, such as a roll of coated paper, or a cylinder or disc coated with a soft material such as wax or a soft metal.

These early recordings were necessarily of low fidelity and volume and captured only a narrow segment of the audible sound spectrum — typically only from around 250 Hz up to about 2,500 Hz — so musicians and engineers were forced to adapt to these sonic limitations. Bands of the period often favored louder instruments such as trumpet, cornet, and trombone, lower-register brass instruments (such as the tuba and the euphonium) replaced the string bass, and blocks of wood stood in for bass drums; performers also had to arrange themselves strategically around the horn to balance the sound, and to play as loudly as possible. The reproduction of domestic phonographs was similarly limited in both frequency-range and volume.

By the end of the acoustic era, the disc had become the standard medium for sound recording, and its dominance in the domestic audio market lasted until the end of the 20th century.

The Electrical Era (1925–1945)

The 'second wave' of sound recording history was ushered in by the introduction of Western Electric's integrated system of electrical microphones, electronic signal amplifiers and electromechanical recorders, which was adopted by major US record labels in 1925. Sound recording now became a hybrid process — sound could now be captured, amplified, filtered, and balanced electronically, and the disc-cutting head was now electrically powered, but the actual recording process remained essentially mechanical – the signal was still physically inscribed into a wax 'master' disc, and consumer discs were mass-produced mechanically by stamping a metal electroform made from the wax master into a suitable substance, originally a shellac-based compound and later polyvinyl plastic.

The Western Electric system greatly improved the fidelity of sound recording, increasing the reproducible frequency range to a much wider band (between 60 Hz and 6000 Hz) and allowing a new class of professional – the audio engineer – to capture a fuller, richer, and more detailed and balanced sound on record, using multiple microphones connected to multi-channel electronic amplifiers, compressors, filters and mixers. Electrical microphones led to a dramatic change in the performance style of singers, ushering in the age of the "crooner", while electronic amplification had a wide-ranging impact in many areas, enabling the development of broadcast radio, public address systems, and electronically-amplified home record players.

In addition, the development of electronic amplifiers for musical instruments now enabled quieter instruments such as the guitar and the string bass to compete on equal terms with the naturally louder wind and horn instruments, and musicians and composers also began to experiment with entirely new electronic musical instruments such as the Theremin, the Ondes Martenot, the electronic organ, and the Hammond Novachord, the world's first analogue polyphonic synthesizer.

Contemporaneous with these developments, several inventors were engaged in a race to develop practical methods of providing synchronized sound with films. Some early sound films — such as the landmark 1927 film The Jazz Singer – used large soundtrack records which were played on a turntable mechanically interlocked with the projector. By the early 1930s, the movie industry had almost universally adopted sound-on-film technology, in which the audio signal to be recorded was used to modulate a light source that was imaged onto the moving film through a narrow slit, allowing it to be photographed as variations in the density or width of a "soundtrack" running along a dedicated area of the film. The projector used a steady light and a photoelectric cell to convert the variations back into an electrical signal, which was amplified and sent to loudspeakers behind the screen.

The adoption of sound-on-film also helped movie-industry audio engineers to make rapid advances in the process we now know as "multi-tracking", by which multiple separately-recorded audio sources (such as voices, sound effects and background music) can be replayed simultaneously, mixed together, and synchronized with the action on film to create new 'blended' audio tracks of great sophistication and complexity. One of the best-known examples of a 'constructed' composite sound from that era is the famous "Tarzan yell" created for the series of Tarzan movies starring Johnny Weissmuller.

Among the vast and often rapid changes that have taken place over the last century of audio recording, it is notable that there is one crucial audio device, invented at the start of the "Electrical Era", which has survived virtually unchanged since its introduction in the 1920s: the electro-acoustic transducer, or loudspeaker. The most common form is the dynamic loudspeaker – effectively a dynamic microphone in reverse. This device typically consists of a shallow conical diaphragm, usually of a stiff paper-like material concentrically pleated to make it more flexible, firmly fastened at its perimeter, with the coil of a moving-coil electromagnetic driver attached around its apex. When an audio signal from a recording, a microphone, or an electrified instrument is fed through an amplifier to the loudspeaker, the varying electromagnetic field created in the coil causes it and the attached cone to move backward and forward, and this movement generates the audio-frequency pressure waves that travel through the air to our ears, which hear them as sound.

Although there have been numerous refinements to the technology, and other related technologies have been introduced (e.g. the electrostatic loudspeaker), the basic design and function of the dynamic loudspeaker has not changed substantially in 90 years, and it remains overwhelmingly the most common, sonically accurate and reliable means of converting electronic audio signals back into audible sound.

The Magnetic Era (1945–1975)

The third wave of development in audio recording began in 1945 when the allied nations gained access to a new German invention: magnetic tape recording. The technology was invented in the 1930s but remained restricted to Germany (where it was widely used in broadcasting) until the end of World War II. Magnetic tape provided another dramatic leap in audio fidelity — indeed, Allied observers first became aware of the existence of the new technology because they noticed that the audio quality of obviously pre-recorded programs was practically indistinguishable from live broadcasts.

From 1950 onwards, magnetic tape quickly became the standard medium of audio master recording in the radio and music industries, and led to the development of the first hi-fi stereo recordings for the domestic market, the development of multi-track tape recording for music, and the demise of the disc as the primary mastering medium for sound. Magnetic tape also brought about a radical reshaping of the recording process — it made possible recordings of far longer duration and much higher fidelity than ever before, and it offered recording engineers the same exceptional plasticity that film gave to cinema editors — sounds captured on tape could now easily be manipulated sonically, edited, and combined in ways that were simply impossible with disc recordings.

These experiments reached an early peak in the 1950s with the recordings of Les Paul and Mary Ford, who pioneered the use of tape editing and multi-tracking to create large 'virtual' ensembles of voices and instruments, constructed entirely from multiple taped recordings of their own voices and instruments. Magnetic tape fueled a rapid and radical expansion in the sophistication of popular music and other genres, allowing composers, producers, engineers and performers to realize previously unattainable levels of complexity. Other concurrent advances in audio technology led to the introduction of a range of new consumer audio formats and devices, on both disc and tape, including the development full-frequency-range disc reproduction, the change from shellac to polyvinyl plastic for disc manufacture, the invention of the 33rpm, 12-inch long-playing (LP) disc and the 45rpm 7-inch "single", the introduction of domestic and professional portable tape recorders (which enabled high-fidelity recordings of live performances), the popular 4-track cartridge and compact cassette formats, and even the world's first "sampling keyboards", the pioneering tape-based keyboard instrument the Chamberlin, and its more famous successor, the Mellotron.

The Digital Era (1975–present)

The fourth and current "phase", the "digital" era, has seen the most rapid, dramatic and far-reaching series of changes in the history of audio recording. In a period of fewer than 20 years, all previous recording technologies were rapidly superseded by digital sound encoding, and the Japanese electronics corporation Sony in the 1970s was instrumental with the first consumer (well-heeled) PCM encoder PCM-1 Audio Unit, introduced in 1977. Unlike all previous technologies, which captured a continuous analogue of the sounds being recorded, digital recording captured sound by means of a very dense and rapid series of discrete samples of the sound. When played back through a digital-to-analogue converter, these audio samples are recombined to form a continuous flow of sound. The first all-digitally-recorded popular music album, Ry Cooder's Bop 'Til You Drop, was released in 1979, and from that point, digital sound recording and reproduction quickly became the new standard at every level, from the professional recording studio to the home hi-fi.

Although a number of short-lived "hybrid" studio and consumer technologies appeared in this period (e.g. Digital Audio Tape or DAT, which recorded digital signal samples onto standard magnetic tape), Sony assured the preeminence of its new digital recording system by introducing, together with Philips, the digital compact disc (CD). The compact disc rapidly replaced both the 12″ album and the 7″ single as the new standard consumer format, and ushered in a new era of high-fidelity consumer audio.

CDs are small, portable and durable, and they could reproduce the entire audible sound spectrum, with a large dynamic range (~96dB), perfect clarity and no distortion. Because CDs were encoded and read optically, using a laser beam, there was no physical contact between the disc and the playback mechanism, so a well-cared-for CD could be played over and over, with absolutely no degradation or loss of fidelity. CDs also represented a considerable advance in both the physical size of the medium, and its storage capacity. LPs could only practically hold about 20–25 minutes of audio per side because they were physically limited by the size of the disc itself and the density of the grooves that could be cut into it — the longer the recording, the closer together the grooves and thus the lower the overall fidelity. CDs, on the other hand, were less than half the overall size of the old 12″ LP format, but offered about double the duration of the average LP, with up to 80 minutes of audio.

The compact disc almost totally dominated the consumer audio market by the end of the 20th century, but within another decade, rapid developments in computing technology saw it rendered virtually redundant in just a few years by the most significant new invention in the history of audio recording — the digital audio file (.wav, .mp3 and other formats). When combined with newly developed digital signal compression algorithms, which greatly reduced file sizes, digital audio files came to dominate the domestic market, thanks to commercial innovations such as Apple's iTunes media application, and their popular iPod portable media player.

However, the introduction of digital audio files, in concert with the rapid developments in home computing, soon led to an unforeseen consequence — the widespread unlicensed distribution of audio and other digital media files. The uploading and downloading of large volumes of digital media files at high speed was facilitated by freeware file-sharing technologies such as Napster and Bit Torrent.

Although infringement remains a significant issue for copyright owners, the development of digital audio has had considerable benefits for consumers and labels. In addition to facilitating the high-volume, low-cost transfer and storage of digital audio files, this new technology has also powered an explosion in the availability of so-called “back-catalogue” titles stored in the archives of recording labels, thanks to the fact that labels can now convert old recordings and distribute them digitally at a fraction of the cost of physically reissuing albums on LP or CD. Digital audio has also enabled dramatic improvements in the restoration and remastering of acoustic and pre-digital electric recordings, and even freeware consumer-level digital software can very effectively eliminate scratches, surface noise and other unwanted sonic artefacts from old 78rpm and vinyl recordings and greatly enhance the sound quality of all but the most badly damaged records. In the field of consumer-level digital data storage, the continuing trend towards increasing capacity and falling costs means that consumers can now acquire and store vast quantities of high-quality digital media (audio, video, games and other applications), and build up media libraries consisting of tens or even hundreds of thousands of songs, albums, or videos — collections which, for all but the wealthiest, would have been both physically and financially impossible to amass in such quantities if they were on 78 or LP, yet which can now be contained on storage devices no larger than the average hardcover book.

The digital audio file marked the end of one era in recording and the beginning of another. Digital files effectively eliminated the need to create or use a discrete, purpose-made physical recording medium (a disc, or a reel of tape, etc.) as the primary means of capturing, manufacturing and distributing commercial sound recordings. Concurrent with the development of these digital file formats, dramatic advances in home computing and the rapid expansion of the Internet mean that digital sound recordings can now be captured, processed, reproduced, distributed and stored entirely electronically, on a range of magnetic and optical recording media, and these can be distributed anywhere in the world, with no loss of fidelity, and crucially, without the need to first transfer these files to some form of permanent recording medium for shipment and sale.

Music streaming services have gained popularity since the late 2000s. Streaming audio does not require the listener to own the audio files. Instead, they listen over the internet. Streaming services offer an alternative method of consuming music and some follow a freemium business model. The freemium model many music streaming services use, such as Spotify and Apple Music, provide a limited amount of content for free, and then premium services for payment. There are two categories in which streaming services are categorized, radio or on-demand. Streaming services such as Pandora use the radio model, allowing users to select playlists but not specific songs to listen to, while services such as Apple Music allow users to listen to both individual songs and pre-made playlists.

Acoustical Recording

The earliest method of sound recording and reproduction involved the live recording of a performance directly to a recording medium by an entirely mechanical process, often called “acoustical recording”. In the standard procedure used until the mid-1920s, the sounds generated by the performance vibrated a diaphragm with a recording stylus connected to it while the stylus cut a groove into a soft recording medium rotating beneath it. To make this process as efficient as possible, the diaphragm was located at the apex of a hollow cone that served to collect and focus the acoustical energy, with the performers crowded around the other end. Recording balance was achieved empirically. A performer who recorded too strongly or not strongly enough would be moved away from or nearer to the mouth of the cone. The number and kind of instruments that could be recorded were limited. Brass instruments, which recorded well, often substituted instruments such as cellos and bass fiddles, which did not. In some early jazz recordings, a block of wood was used in place of the snare drum, which could easily overload the recording diaphragm.

Phonautograph

In 1857, Édouard-Léon Scott de Martinville invented the phonautograph, the first device that could record sound waves as they passed through the air. It was intended only for visual study of the recording and could not play back the sound. The recording medium was a sheet of soot-coated paper wrapped around a rotating cylinder carried on a threaded rod. A stylus, attached to a diaphragm through a series of levers, traced a line through the soot, creating a graphic record of the motions of the diaphragm as it was minutely propelled back and forth by the audio-frequency variations in air pressure.

In the spring of 1877 another inventor, Charles Cros, suggested that the process could be reversed by using photoengraving to convert the traced line into a groove that would guide the stylus, causing the original stylus vibrations to be recreated, passed on to the linked diaphragm, and sent back into the air as sound. Edison's invention of the phonograph soon eclipsed this idea, and it was not until 1887 that yet another inventor, Emile Berliner, actually photoengraved a phonautograph recording into metal and played it back.

Scott's early recordings languished in French archives until 2008 when scholars keen to resurrect the sounds captured in these and other types of early experimental recordings tracked them down. Rather than using rough 19th-century technology to create playable versions, they were scanned into a computer and software was used to convert their sound-modulated traces into digital audio files. Brief excerpts from two French songs and a recitation in Italian, all recorded in 1860, are the most substantial results.

Phonograph/Gramophone

The phonograph, invented by Thomas Edison in 1877, could both record sound and play it back. The earliest type of phonograph sold recorded on a thin sheet of tinfoil wrapped around a grooved metal cylinder. A stylus connected to a sound-vibrated diaphragm indented the foil into the groove as the cylinder rotated. The stylus vibration was at a right angle to the recording surface, so the depth of the indentation varied with the audio-frequency changes in air pressure that carried the sound. This arrangement is known as vertical or "hill-and-dale" recording. The sound could be played back by tracing the stylus along the recorded groove and acoustically coupling its resulting vibrations to the surrounding air through the diaphragm and a so-called "amplifying" horn.

The crude tinfoil phonograph proved to be of little use except as a novelty. It was not until the late 1880s that an improved and much more useful form of the phonograph was marketed. The new machines recorded on easily removable hollow wax cylinders and the groove was engraved into the surface rather than indented. The targeted use was business communication, and in that context, the cylinder format had some advantages. When entertainment use proved to be the real source of profits, one seemingly negligible disadvantage became a major problem: the difficulty of making copies of a recorded cylinder in large quantities.

At first, cylinders were copied by acoustically connecting a playback machine to one or more recording machines through flexible tubing, an arrangement that degraded the audio quality of the copies. Later, a pantograph mechanism was used, but it could only produce about 25 fair copies before the original was too worn down. During a recording session, as many as a dozen machines could be arrayed in front of the performers to record multiple originals. Still, a single "take" would ultimately yield only a few hundred copies at best, so performers were booked for marathon recording sessions in which they had to repeat their most popular numbers over and over again. By 1902, successful molding processes for manufacturing prerecorded cylinders had been developed.

The wax cylinder got a competitor with the advent of the Gramophone, which was patented by Emile Berliner in 1887. The vibration of the Gramophone's recording stylus was horizontal, parallel to the recording surface, resulting in a zig-zag groove of constant depth. This is known as lateral recording. Berliner's original patent showed a lateral recording etched around the surface of a cylinder, but in practice, he opted for the disc format. The Gramophones he soon began to market were intended solely for playing prerecorded entertainment discs and could not be used to record. The spiral groove on the flat surface of a disc was relatively easy to replicate: a negative metal electrotype of the original record could be used to stamp out hundreds or thousands of copies before it wore out. Early on, the copies were made of hard rubber, and sometimes of celluloid, but soon a shellac-based compound was adopted.

"Gramophone", Berliner's trademark name, was abandoned in the US in 1900 because of legal complications, with the result that in American English Gramophones and Gramophone records, along with disc records and players made by other manufacturers, were long ago brought under the umbrella term "phonograph", a word which Edison's competitors avoided using but which was never his trademark, simply a generic term he introduced and applied to cylinders, discs, tapes and any other formats capable of carrying a sound-modulated groove. In the UK, proprietary use of the name Gramophone continued for another decade until, in a court case, it was adjudged to have become genericized and so could be used freely by competing disc record makers, with the result that in British English a disc record is called a "gramophone record" and "phonograph record" is traditionally assumed to mean a cylinder.

Not all cylinder records are alike. They were made of various soft or hard waxy formulations or early plastics, sometimes in unusual sizes; did not all use the same groove pitch; and were not all recorded at the same speed. Early brown wax cylinders were usually cut at about 120 rpm, whereas later cylinders ran at 160 rpm for clearer and louder sound at the cost of reduced maximum playing time. As a medium for entertainment, the cylinder was already losing the format war with the disc by 1910, but the production of entertainment cylinders did not entirely cease until 1929 and use of the format for business dictation purposes persisted into the 1950s.

Disc records, too, were sometimes made in unusual sizes, or from unusual materials, or otherwise deviated from the format norms of their eras in some substantial way. The speed at which disc records were rotated was eventually standardized at about 78 rpm, but other speeds were sometimes used. Around 1950, slower speeds became standard: 45, 33⅓, and the rarely used 16⅔ rpm. The standard material for discs changed from shellac to vinyl, although vinyl had been used for some special-purpose records since the early 1930s and some 78 rpm shellac records were still being made in the late 1950s.

Electrical Recording

Until the mid-1920s records were played on purely mechanical record players usually powered by a wind-up spring motor. The sound was "amplified" by an external or internal horn that was coupled to the diaphragm and stylus, although there was no real amplification: the horn simply improved the efficiency with which the diaphragm's vibrations were transmitted into the open air. The recording process was, in essence, the same non-electronic setup operating in reverse, but with a recording, stylus engraving a groove into a soft waxy master disc and carried slowly inward across it by a feed mechanism.

The advent of electrical recording in 1925 made it possible to use sensitive microphones to capture the sound and greatly improved the audio quality of records. A much wider range of frequencies could be recorded, the balance of high and low frequencies could be controlled by elementary electronic filters, and the signal could be amplified to the optimum level for driving the recording stylus. The leading record labels switched to the electrical process in 1925 and the rest soon followed, although one straggler in the US held out until 1929.

There was a period of nearly five years, from 1925 to 1930 when the top “audiophile” technology for home sound reproduction consisted of a combination of electrically recorded records with the specially-developed Victor Orthophonic Victrola, an acoustic phonograph that used waveguide engineering and a folded horn to provide a reasonably flat frequency response. The first electronically amplified record players reached the market only a few months later, around the start of 1926, but at first, they were much more expensive and their audio quality was impaired by their primitive loudspeakers; they did not become common until the late 1930s.

Electrical recording increased the flexibility of the process, but the performance was still cut directly to the recording medium, so if a mistake was made the whole recording was spoiled. Disc-to-disc editing was possible, by using multiple turntables to play parts of different “takes” and recording them to a new master disc, but switching sources with split-second accuracy was difficult and lower sound quality was inevitable, so except for use in editing some early sound films and radio recordings it was rarely done.

Electrical recording made it more feasible to record one part to disc and then play that back while playing another part, recording both parts to a second disc. This and conceptually related techniques, known as overdubbing, enabled studios to create recorded “performances” that feature one or more artists each singing multiple parts or playing multiple instrument parts and that therefore could not be duplicated by the same artist or artists performing live. The first commercially issued records using overdubbing were released by the Victor Talking Machine Company in the late 1920s. However, overdubbing was of limited use until the advent of audio tape. Use of tape overdubbing was pioneered by Les Paul in the 1940s.

Magnetic Recording

Wire recording or magnetic wire recording is an analog type of audio storage in which a magnetic recording is made on thin steel or stainless steel wire.

The wire is pulled rapidly across a recording head, which magnetizes each point along the wire in accordance with the intensity and polarity of the electrical audio signal being supplied to the recording head at that instant. By later drawing the wire across the same or a similar head while the head is not being supplied with an electrical signal, the varying magnetic field presented by the passing wire induces a similarly varying electric current in the head, recreating the original signal at a reduced level.

Magnetic wire recording was replaced by magnetic tape recording, but devices employing one or the other of these media had been more or less simultaneously under development for many years before either came into widespread use. The principles and electronics involved are nearly identical. Wire recording initially had the advantage that the recording medium itself was already fully developed, while tape recording was held back by the need to improve the materials and methods used to manufacture the tape.

Magnetic recording was demonstrated in principle as early as 1898 by Valdemar Poulsen in his telegraphone. Magnetic wire recording, and its successor, magnetic tape recording, involve the use of a magnetized medium which moves with a constant speed past a recording head. An electrical signal, which is analogous to the sound that is to be recorded, is fed to the recording head, inducing a pattern of magnetization similar to the signal. A playback head can then pick up the changes in the magnetic field from the tape and convert it into an electrical signal.

With the addition of electronic amplification developed by Curt Stille in the 1920s, the telegraphone evolved into wire recorders which were popular for voice recording and dictation during the 1940s and into the 1950s. The reproduction quality of wire recorders was significantly lower than that achievable with phonograph disk recording technology. There were also practical difficulties, such as the tendency of the wire to become tangled or snarled. Splicing could be performed by knotting together the cut wire ends, but the results were not very satisfactory.

On Christmas Day, 1932 the British Broadcasting Corporation first used a steel tape recorder for their broadcasts. The device used was a Marconi-Stille recorder,[11] a huge and dangerous machine which used steel tape that had sharp edges. The tape was 0.1 inches (2.5 mm) wide and 0.003 inches (0.076 mm) thick running at 5 feet per second (1.5 m/s) past the recording and reproducing heads. This meant that the length of tape required for a half-hour program was nearly 1.8 miles (2.9 km) and a full reel weighed 55 pounds (25 kg).

Magnetic Tape Sound Recording

7″ reel of ¼” recording tape, typical of audiophile, consumer and educational use in the 1950s–60s

Engineers at AEG, working with the chemical giant IG Farben, created the world’s first practical magnetic tape recorder, the ‘K1’, which was first demonstrated in 1935. During World War II, an engineer at the Reichs-Rundfunk-Gesellschaft discovered the AC biasing technique. With this technique, an inaudible high-frequency signal, typically in the range of 50 to 150 kHz, is added to the audio signal before being applied to the recording head. Biasing radically improved the sound quality of magnetic tape recordings. By 1943 AEG had developed stereo tape recorders.

During the war, the Allies became aware of radio broadcasts that seemed to be transcriptions (much of this due to the work of Richard H. Ranger), but their audio quality was indistinguishable from that of a live broadcast and their duration was far longer than was possible with 78 rpm discs. At the end of the war, the Allies captured a number of German Magnetophon recorders from Radio Luxembourg that aroused great interest. These recorders incorporated all of the key technological features of analogue magnetic recording, particularly the use of high-frequency bias.

Development of magnetic tape recorders in the late 1940s and early 1950s is associated with the Brush Development Company and its licensee, Ampex; the equally important development of magnetic tape media itself was led by Minnesota Mining and Manufacturing corporation (now known as 3M).

American audio engineer John T. Mullin and entertainer Bing Crosby were key players in the commercial development of magnetic tape. Mullin served in the U.S. Army Signal Corps and was posted to Paris in the final months of World War II; his unit was assigned to find out everything they could about German radio and electronics, including the investigation of claims that the Germans had been experimenting with high-energy directed radio beams as a means of disabling the electrical systems of aircraft. Mullin's unit soon amassed a collection of hundreds of low-quality magnetic dictating machines, but it was a chance visit to a studio at Bad Neuheim near Frankfurt while investigating radio beam rumours that yielded the real prize.

Mullin was given two suitcase-sized AEG 'Magnetophon' high-fidelity recorders and fifty reels of recording tape. He had them shipped home and over the next two years, he worked on the machines constantly, modifying them and improving their performance. His major aim was to interest Hollywood studios in using magnetic tape for movie soundtrack recording.

Mullin gave two public demonstrations of his machines, and they caused a sensation among American audio professionals—many listeners could not believe that what they were hearing was not a live performance. By luck, Mullin's second demonstration was held at MGM studios in Hollywood and in the audience that day was Bing Crosby's technical director, Murdo Mackenzie. He arranged for Mullin to meet Crosby and in June 1947 he gave Crosby a private demonstration of his magnetic tape recorders.

Crosby was stunned by the amazing sound quality and instantly saw the huge commercial potential of the new machines. Live music was the standard for American radio at the time and the major radio networks did not permit the use of disc recording in many programs because of their comparatively poor sound quality. But Crosby disliked the regimentation of live broadcasts, preferring the relaxed atmosphere of the recording studio. He had asked NBC to let him pre-record his 1944–45 series on transcription discs, but the network refused, so Crosby had withdrawn from live radio for a year, returning for the 1946–47 season only reluctantly.

Mullin's tape recorder came along at precisely the right moment. Crosby realized that the new technology would enable him to pre-record his radio show with a sound quality that equalled live broadcasts and that these tapes could be replayed many times with no appreciable loss of quality. Mullin was asked to tape one show as a test and was immediately hired as Crosby's chief engineer to pre-record the rest of the series.

Crosby became the first major American music star to use tape to pre-record radio broadcasts and the first to master commercial recordings on tape. The taped Crosby radio shows were painstakingly edited through tape-splicing to give them a pace and flow that was wholly unprecedented in radio. Mullin even claims to have been the first to use “canned laughter”; at the insistence of Crosby’s head writer, Bill Morrow, he inserted a segment of raucous laughter from an earlier show into a joke in a later show that had not worked well.

Keen to make use of the new recorders as soon as possible, Crosby invested $50,000 of his own money into Ampex, and the tiny six-man concern soon became the world leader in the development of tape recording, revolutionizing radio and recording with its famous Ampex Model 200 tape deck, issued in 1948 and developed directly from Mullin’s modified Magnetophones.

Multitrack Recording

The next major development in the magnetic tape was multitrack recording, in which the tape is divided into multiple tracks parallel with each other. Because they are carried on the same medium, the tracks stay in perfect synchronization. The first development in multitracking was stereo sound, which divided the recording head into two tracks. First developed by German audio engineers ca. 1943, two-track recording was rapidly adopted for modern music in the 1950s because it enabled signals from two or more microphones to be recorded separately at the same time (while the use of several microphones to record on the same track had been common since the emergence of the electrical era in the 1920s), enabling stereophonic recordings to be made and edited conveniently. (The first stereo recordings, on disks, had been made in the 1930s, but were never issued commercially.) Stereo (either true, two-microphone stereo or multi mixed) quickly became the norm for commercial classical recordings and radio broadcasts, although many pop music and jazz recordings continued to be issued in monophonic sound until the mid-1960s.

Much of the credit for the development of multitrack recording goes to guitarist, composer and technician Les Paul, who also helped design the famous electric guitar that bears his name. His experiments with tapes and recorders in the early 1950s led him to order the first custom-built eight-track recorder from Ampex, and his pioneering recordings with his then-wife, singer Mary Ford, were the first to make use of the technique of multitracking to record separate elements of a musical piece asynchronously — that is, separate elements could be recorded at different times. Paul's technique enabled him to listen to the tracks he had already taped and record new parts in time alongside them.

Multitrack recording was immediately taken up in a limited way by Ampex, who soon produced a commercial 3-track recorder. These proved extremely useful for popular music since they enabled backing music to be recorded on two tracks (either to allow the overdubbing of separate parts or to create a full stereo backing track) while the third track was reserved for the lead vocalist. Three-track recorders remained in widespread commercial use until the mid-1960s and much famous pop recordings — including many of Phil Spector's so-called "Wall of Sound" productions and early Motown hits— were taped on Ampex 3-track recorders. Engineer Tom Dowd was among the first to use the multitrack recording for popular music production while working for Atlantic Records during the 1950s.

The next important development was 4-track recording. The advent of this improved system gave recording engineers and musicians vastly greater flexibility for recording and overdubbing, and 4-track was the studio standard for most of the later 1960s. Many of the most famous recordings by The Beatles and The Rolling Stones were recorded on 4-track, and the engineers at London's Abbey Road Studios became particularly adept at a technique called "reduction mixes" in the UK and "bouncing down" in the United States, in which several tracks were recorded onto one 4-track machine and then mixed together and transferred (bounced down) to one track of a second 4-track machine. In this way, it was possible to record literally dozens of separate tracks and combine them into finished recordings of great complexity.

All of the Beatles classic mid-1960s recordings, including the albums Revolver and Sgt. Pepper's Lonely Hearts Club Band, were recorded in this way. There were limitations, however, because of the build-up of noise during the bouncing-down process, and the Abbey Road engineers are still famed for their ability to create dense multitrack recordings while keeping background noise to a minimum.

4-track tape also enabled the development of quadraphonic sound, in which each of the four tracks was used to simulate a complete 360-degree surround sound. A number of albums were released both in stereo and quadrophonic format in the 1970s, but 'quad' failed to gain wide commercial acceptance. Although it is now considered a gimmick, it was the direct precursor of the surround sound technology that has become standard in many modern home theatre systems.

In a professional setting today, such as a studio, audio engineers may use 24 tracks or more for their recordings, using one or more tracks for each instrument played.

The combination of the ability to edit via tape splicing and the ability to record multiple tracks revolutionized studio recording. It became common studio recording practice to record on multiple tracks, and bounce down afterward. The convenience of tape editing and multitrack recording led to the rapid adoption of magnetic tape as the primary technology for commercial musical recordings. Although 33⅓ rpm and 45 rpm vinyl records were the dominant consumer format, recordings were customarily made first on tape, then transferred to disc, with Bing Crosby leading the way in the adoption of this method in the United States.

Further Developments

Analog magnetic tape recording introduces noise, usually called "tape hiss", caused by the finite size of the magnetic particles in the tape. There is a direct tradeoff between noise and economics. Signal-to-noise ratio is increased at higher speeds and with wider tracks, and decreased at lower speeds and with narrower tracks.

By the late 1960s, disk reproducing equipment became so good that audiophiles soon became aware that some of the noise audible on recordings was not surface noise or deficiencies in their equipment, but reproduced tape hiss. A few specialist companies started making "direct to disc recordings", made by feeding microphone signals directly to a disk cutter (after amplification and mixing), in essence reverting to the pre-War direct method of recording. These recordings never became popular, but they dramatically demonstrated the magnitude and importance of the tape hiss problem.

Audio Cassette

Before 1963, when Philips introduced the Compact audio cassette, almost all tape recording had used the reel-to-reel (also called "open reel") format. Previous attempts to package the tape in a convenient cassette that required no threading met with limited success; the most successful was 8-track cartridge used primarily in automobiles for playback only. The Philips Compact audio cassette added much-needed convenience to the tape recording format and a decade or so later had begun to dominate the consumer market, although it was to remain lower in quality than open-reel formats.

In the 1970s, advances in solid-state electronics made the design and marketing of more sophisticated analog circuitry economically feasible. This led to a number of attempts to reduce tape hiss through the use of various forms of volume compression and expansion, the most notable and commercially successful being several systems developed by Dolby Laboratories. These systems divided the frequency spectrum into several bands and applied volume compression/ expansion independently to each band (Engineers now often use the term "compansion" to refer to this process). The Dolby systems were very successful at increasing the effective dynamic range and signal-to-noise ratio of analog audio recording; to all intents and purposes, audible tape hiss could be eliminated. The original Dolby A was only used in professional recording. Successors found use in both professional and consumer formats; Dolby B became almost universal for prerecorded music on cassette. Subsequent forms, including Dolby C, (and the short-lived Dolby S) were developed for home use.

In the 1980s, digital recording methods were introduced, and analog tape recording was gradually displaced, although it has not disappeared by any means. (Many professional studios, particularly those catering to big-budget clients, use analog recorders for multitracking and/or mixdown.) The digital audio tape never became important as a consumer recording medium partially due to legal complications arising from "piracy" fears on the part of the record companies. They had opposed magnetic tape recording when it first became available to consumers, but the technical difficulty of juggling recording levels, overload distortion, and residual tape hiss was sufficiently high that unlicensed reproduction of magnetic tape never became an insurmountable commercial problem. With digital methods, copies of recordings could be exact, and copyright infringement might have become a serious commercial problem. Digital tape is still used in professional situations and the DAT variant has found a home in computer data backup applications. Many professional and home recordists now use hard-disk-based systems for recording, burning the final mixes to recordable CDs (CD-R's).

Recording on Film

The first attempts to record sound to an optical medium occurred around 1900. Prior to the use of recorded sound in film, theatres would have live orchestras present during silent films. The musicians would sit in the pit below the screen and would provide the background noise and set the mood for whatever was occurring in the movie. In 1906, Eugene Augustin Lauste applied for a patent to record Sound-on-film, but was ahead of his time. In 1923, Lee de Forest applied for a patent to record to film; he also made a number of short experimental films, mostly of vaudeville performers. William Fox began releasing sound-on-film newsreels in 1926, the same year that Warner Bros. released Don Juan with music and sound effects recorded on discs, as well as a series of short films with fully-synchronized sound on discs. In 1927, the sound film The Jazz Singer was released; while not the first sound film, it made a tremendous hit and made the public and the film industry realize that sound film was more than a mere novelty.

The Jazz Singer used a process called Vitaphone that involved synchronizing the projected film to sound recorded on a disc. It essentially amounted to playing a phonograph record, but one that was recorded with the best electrical technology of the time. Audiences used to acoustic phonographs and recordings would, in the theatre, have heard something resembling 1950s "high fidelity".

However, in the days of analog technology, no process involving a separate disk could hold synchronization precisely or reliably. Vitaphone was quickly supplanted by technologies which recorded an optical soundtrack directly onto the side of the strip of motion picture film. This was the dominant technology from the 1930s through the 1960s and is still in use as of 2013 although the analog soundtrack is being replaced by digital sound on film formats.

There are two types of a synchronized film soundtrack, optical and magnetic. Optical soundtracks are visual renditions of sound wave forms and provide sound through a light beam and optical sensor within the projector. Magnetic soundtracks are essentially the same as used in conventional analog tape recording.

Magnetic soundtracks can be joined with the moving image but it creates an abrupt discontinuity because of the offset of the audio track relative to the picture. Whether optical or magnetic, the audio pickup must be located several inches ahead of the projection lamp, shutter and drive sprockets. There is usually a flywheel as well to smooth out the film moves to eliminate the flutter that would otherwise result from the negative pulldown mechanism. If you have films with a magnetic track, you should keep them away from strong magnetic sources, such as televisions. These can weaken or wipe the magnetic sound signal. Magnetic sound on a cellulose acetate film base is also more prone to vinegar syndrome than a film with just the image.[why?][citation needed]

A variable density soundtrack (left) and a bi-lateral variable area soundtrack (right)

For optical recording on film there are two methods utilized. Variable density recording uses changes in the darkness of the soundtrack side of the film to represent the soundwave. Variable area recording uses changes in the width of a dark strip to represent the soundwave.

In both cases, a light that is sent through the part of the film that corresponds to the soundtrack changes in intensity, proportional to the original sound, and that light is not projected on the screen but converted into an electrical signal by a light-sensitive device.

Optical soundtracks are prone to the same sorts of degradation that affect the picture, such as scratching and copying.

Unlike the film image that creates the illusion of continuity, soundtracks are continuous. This means that if film with a combined soundtrack is cut and spliced, the image will cut cleanly but the soundtrack will most likely produce a cracking sound. Fingerprints on the film may also produce cracking or interference.

In the late 1950s, the cinema industry, desperate to provide a theatre experience that would be overwhelmingly superior to television, introduced widescreen processes such as Cinerama, Todd-AO and CinemaScope. These processes at the same time introduced technical improvements in sound, generally involving the use of multitrack magnetic sound, recorded on an oxide stripe laminated onto the film. In subsequent decades, a gradual evolution occurred with more and more theatres installing various forms of magnetic-sound equipment.

In the 1990s, digital audio systems were introduced and began to prevail. In some of them the sound recording is again recorded on a separate disk, as in Vitaphone; others use a digital, optical sound track on the film itself. Digital processes can now achieve reliable and perfect synchronization.

Digital Recording

The first digital audio recorders were reel-to-reel decks introduced by companies such as Denon (1972), Soundstream (1979) and Mitsubishi. They used a digital technology known as PCM recording. Within a few years, however, many studios were using devices that encoded the digital audio data into a standard video signal, which was then recorded on a U-matic or other videotape recorder, using the rotating-head technology that was standard for video. A similar technology was used for a consumer format, Digital Audio Tape (DAT) which used rotating heads on a narrow tape contained in a cassette. DAT records at sampling rates of 48 kHz or 44.1 kHz, the latter being the same rate used on compact discs. Bit depth is 16 bits, also the same as compact discs. DAT was a failure in the consumer-audio field (too expensive, too finicky, and crippled by anti-copying regulations), but it became popular in studios (particularly home studios) and radio stations. A failed digital tape recording system was the Digital Compact Cassette (DCC).

Within a few years after the introduction of digital recording, multitrack recorders (using stationary heads) were being produced for use in professional studios. In the early 1990s, relatively low-priced multitrack digital recorders were introduced for use in home studios; they returned to recording on videotape. The most notable of this type of recorder is the ADAT. Developed by Alesis and first released in 1991, the ADAT machine is capable of recording 8 tracks of digital audio onto a single S-VHS video cassette. The ADAT machine is still a very common fixture in professional and home studios around the world.

In the consumer market, tapes and gramophones were largely displaced by the compact disc (CD) and a lesser extent the minidisc. These recording media are fully digital and require complex electronics to play back. Digital recording has progressed towards higher fidelity, with formats such as DVD-A offering sampling rates of up to 192 kHz.

Technique

The analog tape recorder made it possible to erase or record over a previous recording so that mistakes could be fixed. Another advantage of recording on tape is the ability to cut the tape and join it back together. This allows the recording to be edited. Pieces of the recording can be removed, or rearranged. See also audio editing, audio mixing, multitrack recording.

The advent of electronic instruments (especially keyboards and synthesizers), effects and other instruments has led to the importance of MIDI in recording. For example, using MIDI timecode, it is possible to have different equipment 'trigger' without direct human intervention at the time of recording.

In more recent times, computers (digital audio workstations) have found an increasing role in the recording studio, as their use eases the tasks of cutting and looping, as well as allowing for instantaneous changes, such as duplication of parts, the addition of effects and the rearranging of parts of the recording.

6

Unit Two, Part Two: Pro Tools Recording Software and Mixing a Session

Pro Tools recording software is a digital audio workstation or DAW. It is the industry standard for modern recording. However, many other DAWs can be installed at prices ranging from free to hundreds of dollars. Pro Tools features some advantages in how expansive it is along with how professional its audio sounds. There are other DAWs that are easier to learn and there are some DAWs that are made for specific audio purposes. For this course, however, we will study Pro Tools exclusively and use it for our recording projects. We generally feel that if a student can learn how to use Pro Tools, that knowledge will translate to all other DAWS.

Again and again, no matter what equipment you have in front of you to record audio, if you use our three pillars of good production, it will result in the best possible managed sound.

Here is a link to the Pro Tools reference manual Avid Pro Tools Reference Manual. It can be intimidating, to say the least, trying to understand these very detailed instructions from start to finish. My suggestion is to first watch some introductory videos and then take the manual one step at a time.

Video Link Introduction to Pro Tools (Everything You Need to Know to Get Started)

Once students become familiar with creating tracks and basic recording procedures, then it is time to use the various *methods* of sound management that have been discussed earlier. Microphone placement, mixing strategies, adding effects or other processors; all those ideas can be employed here.

As a project nears its finish, consider the principals of creative approach such as contrast, balance, flow, etc… This will help give your sound management an overall signature to your work. Pro Tools features endless possibilities on what it can do for your recordings and audio creations. But it also requires the most training to learn how to use it.

DIGITAL AUDIO WORKSTATION

DAW–Digital Audio Workstation

A digital audio workstation (DAW) is an electronic device or application software used for recording, editing and producing audio files. DAWs come in a wide variety of configurations from a single software program on a laptop, to an integrated stand-alone unit, all the way to a highly complex configuration of numerous components controlled by a central computer. Regardless of configuration, modern DAWs have a central interface that allows the user to alter and mix multiple recordings and tracks into a final produced piece.

DAWs are used for producing and recording music, songs, speech, radio, television, soundtracks, podcasts, sound effects and nearly any other situation where complex recorded audio is needed

DAW can simply refer to the software itself, but traditionally, a computer-based DAW has four basic components: a computer, a sound card or other audio interface, audio editing software, and at least one user input device for adding or modifying data. This could be as simple as a mouse and keyboard or as sophisticated as a piano-style MIDI controller keyboard or automated audio control surface for mixing track volumes.

The computer acts as a host for the sound card, while the software provides the interface and functionality for audio editing. The sound card typically converts analog audio signals into digital form, and digital back to analog audio when playing it back; it may also assist in further processing of the audio. The software controls all related hardware components and provides a user interface to allow for recording, editing, and playback.

Computer-based DAWs have extensive recording, editing, and playback capabilities (and some also have video-related features). For example, they can provide a practically limitless number of tracks to record on, polyphony, and virtual synthesizers or sample-based instruments to use for recording music. DAWs can also provide a wide variety of effects, such as reverb, to enhance or change the sounds themselves.

Simple smartphone-based DAWs, called mobile audio workstation (MAWs), are used (for example) by journalists for recording and editing on location.

As software systems, DAWs are designed with many user interfaces, but generally, they are based on a multitrack tape recorder metaphor, making it easier for recording engineers and musicians already familiar with using tape recorders to become familiar with the new systems. Therefore, computer-based DAWs tend to have a standard layout that includes transport controls (play, rewind, record, etc.), track controls and a mixer. A waveform display is another common feature.

Single-track DAWs display only one (mono or stereo form) track at a time. Multitrack DAWs support operations on multiple tracks at once. Like a mixing console, each track typically has controls that allow the user to adjust the gain, equalization and stereo panning of the sound on each track. In a traditional recording studio additional rackmount processing gear is physically plugged into the audio signal path to add reverb, compression, etc. However, a DAW can also route in software or use audio plug-ins (for example, a VST plugin) to process the sound on a track.

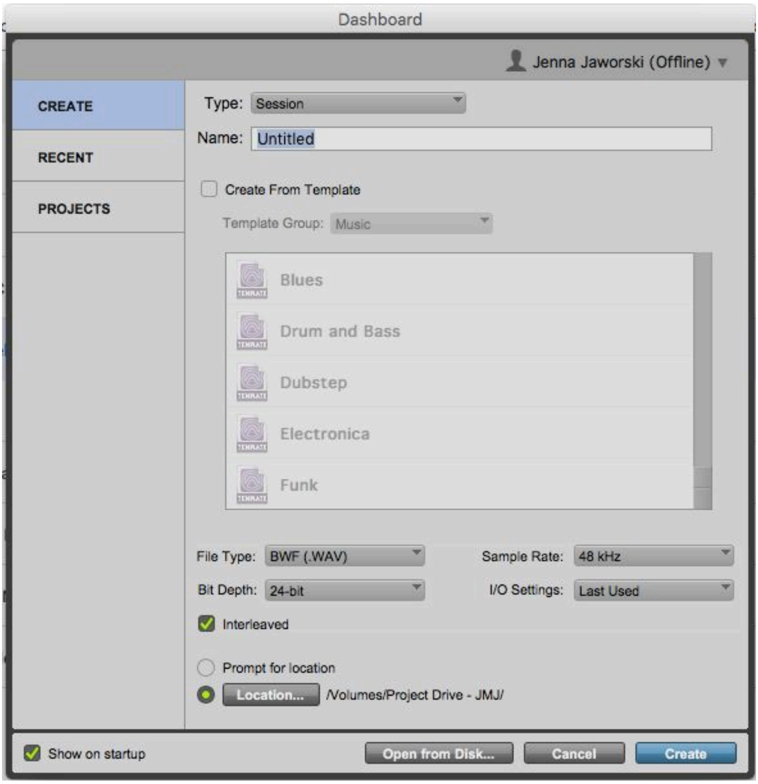
Perhaps the most significant feature available from a DAW that is not available in analog recording is the ability to undo a previous action, using a command similar to that of the undo function in word processing software. Undo makes it much easier to avoid accidentally permanently erasing or recording over a previous recording. If a mistake or unwanted change is made, the undo command is used to conveniently revert the changed data to a previous state. Cut, Copy, Paste, and Undo are familiar and common computer commands and they are usually available in DAWs in some form. More common functions include the modifications of several factors concerning a sound. These include wave shape, pitch, tempo, and filtering.

Commonly DAWs feature some form of mix automation using procedural line segment-based or curve-based interactive graphs. The lines and curves of the automation graph are joined by or comprise adjustable points. By creating and adjusting multiple points along a waveform or control events, the user can specify parameters of the output over time (e.g., volume or pan). Automation data may also be directly derived from human gestures recorded by a control surface or MIDI controller.

MIDI recording, editing, and playback is increasingly incorporated into modern DAWs of all types, as is synchronization with other audio or video tools.

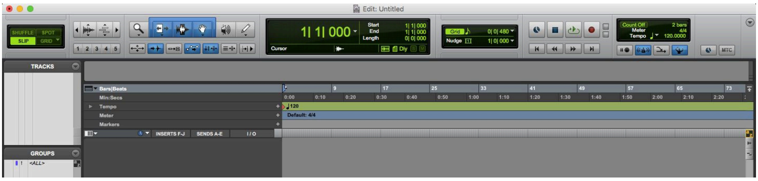
BEHIND THE SPEAKERS PRO TOOLS QUICK START GUIDE

CREATING A NEW TRACK

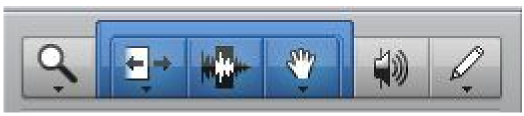


My go-to settings for sample rate and bit depth are **44.1 kHz** and **24-bit.** 'Location' specifies where your session will be saved to.

EDIT WINDOW



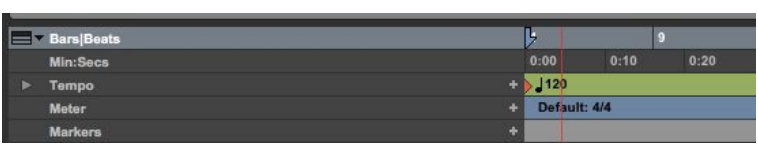
Tools

123456

- The Zoom Tool – lets you zoom in on tracks.
- The Trim Tool – to edit/cut the ends of audio clips.
- The Selector Tool – to highlight specific parts of an audio clip.
- The Grab Tool – to select full audio clips.

Select the bar above 2 – 4 to enable the 'Smart Tool'.

- The Scrubber Tool – emulates scrubbing through tape.
- The Pencil Tool – for drawing automation.



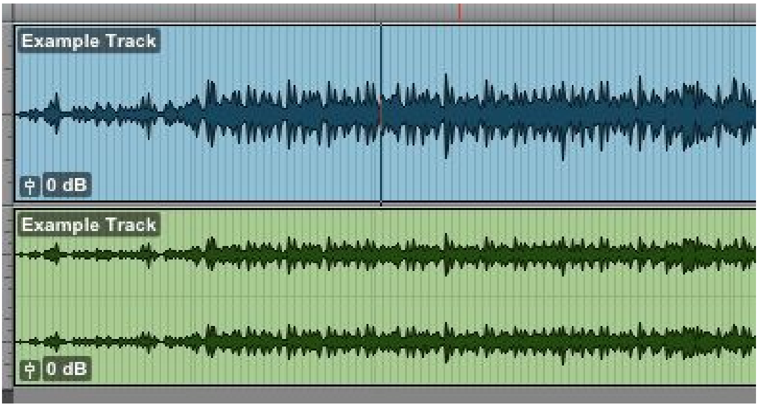
- The box to the left of 'Bars/Beats' provides the option to view the timeline in various measurements – bars and beats, minutes and seconds, samples, or timecode for film.

- To add a marker to the timeline (ex: Intro, Chorus, Verse), click the '+' button.

ADDING A NEW TRACK

Go to 'Track' -> 'New' or use the shortcut Command + Shift + N.

A mono track uses one audio track, a stereo track uses two.



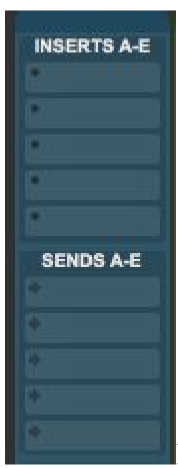
A mono track in blue, a stereo track in green.

- An audio track contains audio clips

- An aux track doesn't contain its own audio, but it can have audio sent to it.
- A master fader controls the volume of the entire session.

MIX WINDOW

Inserts/Sends Section



Inserts – Where you can add plugins on a track.

Sends – Used to send a copy of a track to another place in your DAW. Often used for adding reverb and delay.

I/O Section

The top box is the input selection.

The bottom box is the output selection.

Bottom Section

Pan knob

I = Input monitoring

Re
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Solo

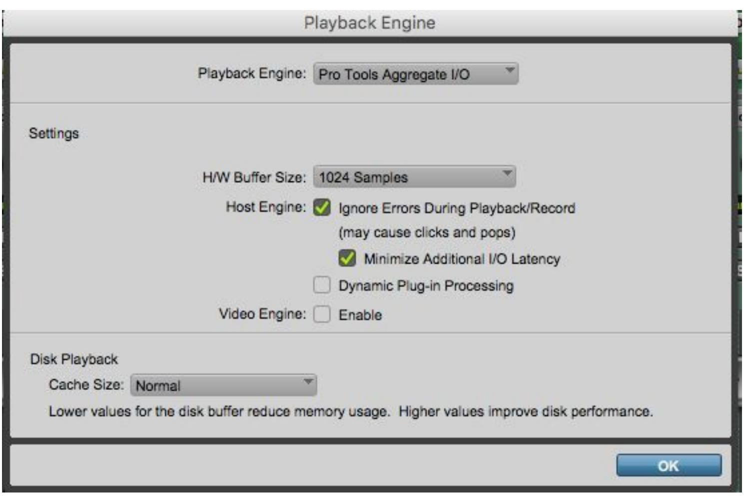
M =

Mute

The section with "Audio 1" is the track name (double-click to rename).

This square is the comments section (you can add notes about a track here).

PLAYBACK ENGINE

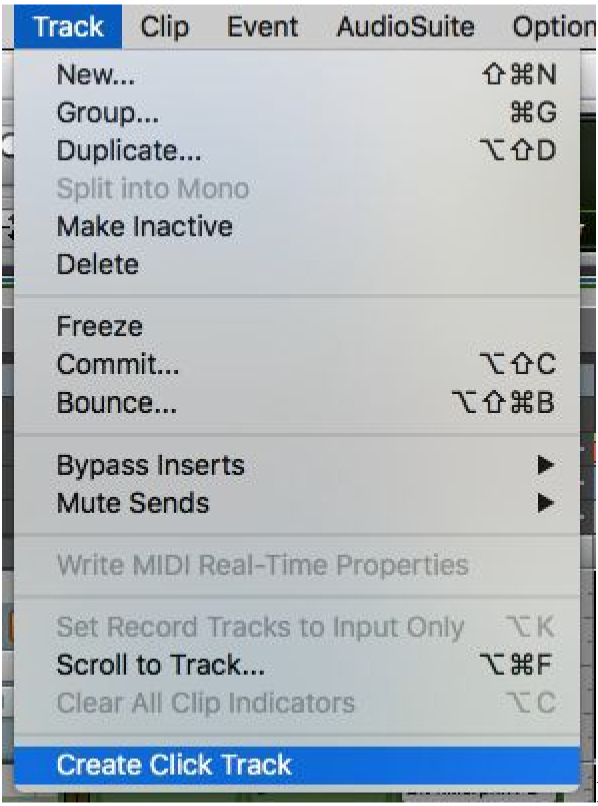


Go to 'Setup' -> 'Playback Engine'.

Under Playback Engine, select your audio interface.

For 'H/W Buffer Size', smaller values are better when recording. When mixing, larger sample sizes are fine.

CREATING A CLICK TRACK



From the top menu, select 'Track' -> 'Create Click Track'.

Open the Transport by selecting Window -> Transport. You can change the tempo within the box on the far right.

Highlight the current tempo, type in your desired tempo, then hit enter. You may need to press the blue conductor button to turn off the conductor track first.

BASIC AUDIO EDITING

Separating Clips – Either click to specify one spot, or highlight to choose two spots at once, then do one of the following:

- Right-click at the desired spot, then choose 'Separate'
- Command + E
- 

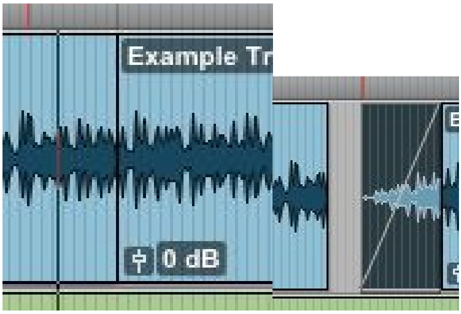
  'Edit' -> 'Separate Clip'

Moving Clips – Click on and hold the desired clip, then drag it across the timeline.

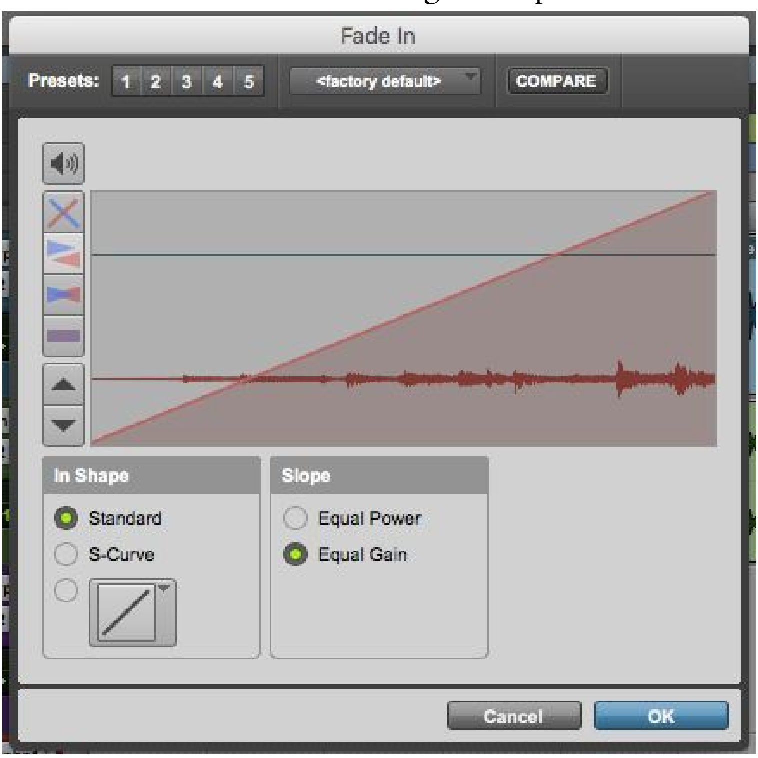
Trimming Clips – To the right or left of the clip, the mouse will change into a [or]

- When this appears, click and drag the track to trim

Fading Clips – Select the box above tools 2 – 4 to enable the 'Smart Tool'. When the mouse changes to a square with a diagonal line through it, you can click and drag a track to add a fade.



Double click on this fade to change its shape.



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e you:

Finalized the arrangement?

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s?

Tuned all vocals?

Time-aligned drums and other tracks?

Crossfaded all edits and checked for clicks and pops?

Deleted unused or muted tracks?

Printed virtual instrument tracks to audio?

Ordered and grouped tracks in a logical way?

Labeled all tracks?

Color-coded tracks and clips?

Added markers for song sections?

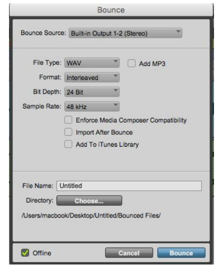
Cleaned up unwanted noise on each track?

Optimized gain staging, so you're not clipping individual tracks or your master bus?

BOUNCING YOUR MIX

Make sure the part of the timeline you want bounced is highlighted. Make sure nothing is muted/solo'ed.

Go to ‘File’ -> ‘Bounce to’ -> Disk.



Recommended Settings

Format: Interleaved

Bit Depth and Sample Rate – typically 44.1 kHz, 24-bit

Uncheck ‘Impor

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SHORTCUTS

- Command + S – Save
- Command + Shift + N – New track
- Command + Shift + I – Import Audio
- Option + Shift + I – Import Session Data
- Command + Z – Undo
- Command + M – Mute selected clip
- Command + G – Create a group for the selected tracks
- Command + = – Quickly flip between the Edit and Mix Windows
- Command + E – Separate clip
- Highlight a section + E – The selected clip will fill

the screen

- Hitting E again will return it to normal view

MIXING AUDIO

In sound recording and reproduction, audio mixing is the process of optimizing and combining multitrack recordings into a final mono, stereo or surround sound product. In the process of combining the separate tracks, their relative levels (i.e. volumes) are adjusted and balanced and various processes such as equalization and compression are commonly applied to individual tracks, groups of tracks, and the overall mix. In stereo and surround sound mixing, the placement of the tracks within the stereo (or surround) field are adjusted and balanced.

Audio mixing techniques largely depend on music genres and the quality of sound recordings involved. The process is generally carried out by a mixing engineer, though sometimes the record producer or recording artist may assist. After mixing, a mastering engineer prepares the final product for production.

Audio mixing may be performed on a mixing console or in a digital audio workstation.

Digital audio workstations (DAW) can perform many mixing features in addition to other processing. An audio control surface gives a DAW the same user interface as a mixing console. The distinction between a large console and a DAW equipped with a control surface is that a digital console will typically consist of dedicated digital signal processors for each channel. DAWs can dynamically assign resources like digital audio signal processing power, but may run out if too many signal processes are in simultaneous use. This overload can often be solved by increasing the capacity of the DAW.

Outboard audio processing units (analog) and software-based audio plug-ins (digital) are used for each track or group to perform various processing techniques. These processes, such as equalization, compression, sidechaining, stereo imaging, and saturation are used to make each element as audible and sonically appealing as possible. The mix engineer also will use such techniques to balance the "space" of the final audio wave; removing unnecessary frequencies and volume spikes to minimize the interference or "clashing" between each element.

Processes that affect signal volume or level

Faders – The process of attenuating (lowering) the level of a signal. This is by far the most basic audio process, appearing on virtually every effect unit and mixer. Utilizing controlled fades is the most basic step of audio mixing, allowing more volume for prominent elements and less for secondary elements.

Boost – The process of adding volume to a signal. Boosting is done using extremely slight amounts of amplification, enough to raise a signal without pushing it over the limit of a pre-amplified signal. Some volume control units may feature the ability to both boost and attenuate a signal.

Panning – The process of altering the balance of an audio signal between the left and right channels of a stereo signal. The pan of a signal may be modified via a simple two-way pan control or an "auto panner" that continuously modulates and changes the pan of a signal. Panning is often used in the mixing process to "arrange" the track elements, simulating the placement of live bands.

*Compressor*s – The process of reducing the dynamic range or difference between loudest and quietest parts of a signal. This is done with automatically controlled fader, which will reduce the signal volume after a user-adjustable threshold is hit. The ratio of reduction to gain above the threshold is often also controllable, as well as the time it takes for reduction to activate (attack) or release. Most compressors will also apply a boost after the gain reduction is replied to compensate for the quieter signal. Compression has many uses in the mixing process, from evening out vocal volume to enhancing drums.

Limiters – Limiting is essentially an extremely harsh form of compression- instead of applying gentle reduction to audio above the threshold, limiters forcibly "flatten" it down, allowing no signal above the threshold. Many limiting units also have built-in compressors that reduce the amount of audio actually passing the threshold. Many limiters also use digital algorithms to "soften" the harsh sound of limited audio, morphing the wave instead of completely decapitating it (by removing part of the waveform entirely, intense distortion and vastly altered tones can occur.) Softer limiters are used with generous amounts of compression to create a more consistently loud track with less volume fluctuation, and harder limiters can be used as distortion effects or emergency safeties to protect large speaker systems from blowing out. Many analog amplifiers are fitted with their own basic limiters to prevent the high-voltage circuitry from overloading and blowing out.

Dynamic expansion — Dynamic expansion is essentially compression with an inverted threshold- any signal below a certain threshold is dynamically reduced while signals above the threshold remain untouched. Expansion is most commonly used to give volume to certain elements of recordings- e.g. the bass and snare drums of a drum recording.

Noise gating – When a signal drops below a set threshold, a gate will reduce gain until the output signal is forced below a certain level, and continue to hold the gain at that level until the input rises above the threshold.

Processes that affect frequencies

The frequency response of a signal represents the amount (volume) of every frequency in the human hearing range, consisting of (on average) frequencies from 20 Hz to 20,000 Hz (20 kHz.) There are a variety of processes commonly used to edit frequency response in various ways.

Equalization – Equalization is a broad term for any device that can alter parts of a signal frequency response. Some EQs use a grid of faders or knobs which can be arranged to shape each frequency, whereas others use bands that can target and subsequently boost or cut selectable series of frequencies.

Filters – Filters are used to eliminate certain frequencies from the output. Filters strip off part of the audio spectrum. There are various types of filters. A high-pass filter (low-cut) is used to remove excessive room noise at low frequencies. A low-pass filter (high-cut) is used to help isolate a low-frequency instrument playing in a studio along with others. And a band-pass filter is a combination of high- and low-pass filters, also known as a telephone filter (because a sound lacking in high and low frequencies resembles the quality of sound over a telephone).

Processes That Affect Time

Reverbs – Reverbs are used to simulate acoustic reflections in a real room, adding a sense of space and depth to otherwise "dry" recordings. Another use is to distinguish among auditory objects; all sound having one reverberant character will be categorized together by human hearing in a process called auditory streaming. This is an important technique in creating the illusion of layered sound from in front of the speaker to behind it. Before the advent of electronic reverb and echo processing, physical means were used to generate the effects. An echo chamber, a large reverberant room, could be equipped with a speaker and microphones. Signals were then sent to the speaker and the reverberation generated in the room was picked up by the two microphones.

7

Unit Two, Part Three: Types of Audio Files

An apparatus that uses binary code such as a computer or an iPhone manages sound in a digital format. There are many different types of playback devices and various ways these files are stored. Additionally, analog sound systems are still used to play and archive sound. An audio engineer can quickly discover there exists a vast expanse of recording formats that are not easily compatible or accessible. Understanding how the older systems work can help with most modern versions. And vice-versa. Take time to play and listen to music on a wide range of equipment from different eras. Learn how to convert one to the other and your expertise in the production field will certainly grow.

People that work in audio production need to know the differences between varying digital and analog files, how to convert one to another, and what each is best used for (**technical**). How we save our files, and how we transmit these files, and why we do so could be considered the **method** in our three pillar approach. The order in which we present these audio files or perhaps the final decision on where to share these files would fall under our **creative approach**.

For example, in the past, I've been hired to convert a client's old 78 vinyl record collection into digital files that could be played and organized on a typical home laptop computer. That required choosing the right equipment and cables to play records at the right speed and turning that analog signal into a digital one. Once transferred to my computer, I could then make some equalization adjustments and bring the overall volume up (or down) to a presentable level. Finally, I had to convert those large files into smaller MP3 files that I could electronically send back to the client.

In addition to the audio, I scanned the liner notes and artwork and included those attachments along with with MP3s. Once again, I used my **technical** knowledge to identify what equipment would be needed, I used different **methods** and orders of operation to make quality conversions, and I topped it off with some **creative approach** in packaging the final product so that it was an artistically unique experience for my client.

Timeline of Audio Formats

Format Guide to Sound Recordings

DIGITAL AUDIO FILE FORMATS

There are two major groups of audio file formats:

Those using lossless compression, e.g. like WAV, FLAC

Those using lossy compression, e.g. MP3, Ogg Vorbis, WMA, AAC

In the lossless compression of a piece of data, nothing is lost during the compression and the original data is restored upon uncompressing. In lossy compression, some data is lost during compression and upon uncompressing the data is not identical to the original but possibly close to it. Lossy compression is used mainly in the compression of multimedia data like audio or video where the loss of some details is tolerable under certain conditions, e.g., the human eye is unable to discern the loss in certain details of an image or video.

WAV

WAVE form audio format (WAV) is a Microsoft and IBM audio file format for storing audio on PCs. It is the main format used on Microsoft Windows systems for raw audio storage. The WAV format is most commonly used with an uncompressed, lossless storage method (pulse-code modulation) resulting in comparatively large audio files. Today, the WAV audio format is no longer popular being superseded by other more efficient means of audio storage.

FLAC

Free Lossless Audio Codec (FLAC) is a popular lossless audio format with compression designed specifically for audio data streams, achieving compression rates of 30–50 percent. The format specification is publicly available and forms part of the FLAC Open Source project. It is supported by a growing list of audio software and devices.

MP3

MPEG-1 audio layer 3 (MP3) is a popular lossy compression audio format. The MP3 specification was set by the Motion Pictures Experts Group (MPEG), a working group of ISO/IEC charged with the development of video and audio encoding standards. The compression scheme and format for MP3 forms part of the MPEG-1 video and audio compression standard specifications and is an ISO standard, ISO/IEC 11172-3.

MP3 is one of the most popular audio file formats in use today. Music files encoded with MP3 are particularly popular on music exchange and download sites on the Internet due, in part, to the relatively small size of such files and the wide availability of free software on PCs that allow easy creation, sharing, collecting and playing of MP3 files.

WMA

Windows Media Audio (WMA) is a lossy compression audio file format developed by Microsoft. It is a proprietary format but is widely used and supported due to the popularity of the MS Windows platform.

AAC

Advanced Audio Coding (AAC) from MPEG is a lossy data compression scheme intended for audio streams. It was designed to provide better quality at the same bit-rate than MP3, or the same quality at lower bitrates (and hence smaller file sizes). The compression scheme and format for AAC forms part of the MPEG2 video and audio compression standard specifications and is an ISO standard, ISO/IEC 13818-7. This MPEG-2 AAC specification makes use of patents from several companies and a patent license is needed for products that make use of this standard.

The newer **MPEG-4** standard also specifies an audio compression technology that incorporates MPEG-2 AAC. This is known as MPEG-4 AAC, and is an ISO standard, ISO/IEC 14496-3.

Apple's popular iTunes service and iPod products have music available in AAC and this has led to an upsurge in the popularity of AAC despite the required patent license royalty payments.

RealAudio

RealAudio is a proprietary audio format developed by RealNetworks for low bandwidth usage. It was first introduced in 1995 and it became popular especially for streaming audio, i.e., the audio is being played in real time as it is downloaded. Many radio stations use RealAudio to stream their programs over the Internet.

Ogg Vorbis

Ogg Vorbis is a compressed audio format that is believed to be free of patents and royalty payments. The format originated from the Xiph.Org Foundation, a non-profit organization dedicated to producing free and open protocols, formats and software for multimedia.

Ogg Vorbis uses the Vorbis lossy audio compression scheme. The audio data is wrapped up in the Ogg container format, the name of Xiph.org's container format for audio, video, and meta-data – hence the name Ogg Vorbis. The Ogg Vorbis specification is in the public domain and is completely free for commercial or non-commercial use. There is growing support for the Ogg Vorbis format from software and hardware devices as well as online audio services.

Audio Formats

Format	Organization	Published	Non-Proprietary	International Standard
WAV	Microsoft	Yes	No	No
FLAC	Xiph.Org	Yes	Yes	No
MP3	MPEG/ISC	Yes	Yes	Yes
WMA	Microsoft	No	No	No
AAC	MPEG/ISO	Yes	Yes	Yes
RealAudio	RealNetworks	Yes	No	No
Ogg Vorbis	Xiph.org	Yes	Yes	No

AUDIO GUIDANCE: IDENTIFYING AUDIO FORMATS

HOW DO I IDENTIFY AUDIO FORMATS?

Sound recordings come in a variety of shapes and sizes and have been around since the late nineteenth century. While there are a number of audio formats you may find at home or in a professional environment, each one presents different requirements for identification, physical handling, and playback equipment used. Some of the more common types of audio include ¼-inch open reel tape, audio cassettes, and grooved disc recordings.

Open Reel Quarter Inch Tape

Magnetic recording tape was invented in 1928. However, it wasn't until the late-1940s that the recording industry fully adopted the format. Compared to grooved recordings, magnetic tape offered higher fidelity, longer record times, and the ability to be edited.

¼-inch open reel audio tape is supplied in reels having a diameter of 5-inches, 7-inches, and 10.5-inches and may vary in length depending on the tape speed. 1-inch and 2-inch open reel tapes (in width) also exist and were primarily used in professional environments.

- There are different types of tape base for ¼-inch width audio tape. Acetate and polyester are most common, although you may encounter PVC or paper base as well.

- An easy way to tell if you have one of the two most common types of ¼-inch tape is to hold the tape up to a light source. If the tape appears opaque it is polyester; if the tape is translucent, it is acetate.
- Tape speeds vary between 30 ips (inches per second) to 15/16 ips, with speeds of 7.5 ips and 3.75 ips being the most common.

Audio Cassette

This tape format was introduced in the United States in the mid-1960s by Phillips. Cassette recordings have been used for a number of purposes, including oral histories, lectures, conferences, and music. Since they were primarily used for convenience and not necessarily quality, cassettes are not considered to be a stable or high fidelity medium.

- Cassettes are made with polyester base tape, have just over a 1/8th inch tape width, and a slow playback speed of 1 7/8 ips, thus contributing to a limited dynamic range and frequency response.

Wire Recording


Wire Recording

An alternate form of magnetic recording, the wire recording was initially introduced in the late 19th century, but further developed in the US during WWII. Spools are about 3 ½-inches in diameter. This format was ultimately succeeded by tape due to its low sound quality. While stainless steel wire recordings (post WWII) are not susceptible to the same types of degradation as open-reel tapes, they may be damaged easily and playback equipment is obsolete. You may find the wire recordings in your collection date from around the 1940s-1960s.

Magnabelts


Magnabelt

These magnetic belts are similar in appearance to dictabelts and were used for dictation purposes, but do not contain grooves.

Digital Audio Tape (DAT)

Digital Audio Tape (DAT)

Digital Audio Tapes were introduced by Sony in 1987, and are magnetic tapes that store audio digitally. A cartridge houses a 4mm magnetic tape and recordings can be at, below, or above the 44.1 kHz sampling rate of CDs.

Grooved Recordings

Grooved Recording

You may encounter grooved discs at home or in professional collections. Discs vary in size and speed and have different coatings and substrates, which may include:aluminum, lacquer/acetate, plastic, and cardboard. Below are several of the more common types of phonograph records:

Wax Cylinders

The earliest recording medium produced commercially, these are grooved wax- cylinders, which are the predecessors to the grooved disc. Wax cylinders can be solid or have a core of cardboard. They are very fragile and should be handled with care, including the grooves which can easily be scratched.

Courtesy: National Archives and Records Administration License Link

8

Unit Two, Part Four: Mastering

The last step in audio recording is Mastering. When people ask, “What is mastering?” the answers can be frustratingly mysterious. And when looking at the liner notes of just about any album recorded the last 50 years, the mastering (and re-mastering!) credits are listed separately from the production and engineering credits. So what does mastering do, and how important is it to the final production?

Mastering a piece of audio requires different equipment and acoustic room treatments than what was used to record and mix it. The person mastering a project will go through the final mix, the overall EQ, and volume levels with a fine tooth comb on precision hardware and software dedicated solely to this operation. He/she may also add digital track information needed for cataloging. The idea is to get the finished audio in perfect order with the best dynamic range and the most uniform volume possible. There are some

technical standards that professional audio must adhere to. That way, it will sound as consistent on a home stereo as it does in a club as it does on the radio or streaming service. All other copies of the result will come from this mastered track. The best analogy I can think of is running your audio through a carwash so it looks a shiny and clean on the outside as the rest of the cars on the lot. Whether or not it runs or not, that depends on what is on the inside and who's driving it.

An equally important question is: Should every project be mastered? Short answer, no. Mastering costs are separate from recording costs *(***method***)*. If you plan to record a demo for your friends to listen to or make an informal piece of audio for a one-off performance, you probably don't need to master your project. However, let's say your new song is getting listeners from outside your group of friends or you want to release the song(s) on your streaming page for public download, in that case, because people will be listening to it on different formats, you might consider putting it through the mastering carwash.

Any professional recording made for repeated public broadcast should probably be mastered. From personal experience, mastering usually does make a noticeable difference in quality, especially for a piece of vinyl. Make sure you trust the person mastering the project. This is not a process whereby you sit next to the person and work intimately like you do in recording or mixing. Mastering is done usually off-site and is generally not a collective process.

The choice of who does the mastering can be a *creative choice* as much as a price-conscious one.

I personally have never mastered a project myself. But I have gotten to know people who do engage in this area of audio production. Mastering will take experience and training that is outside the realm of this particular course, but everything learned up to here can apply towards that sort of education. Most of the people I know in that field started out in small studios or as sound mixers for small clubs/events. After gaining experience and developing finely tuned ears, they decided to venture further in that direction and have done great work in mastering audio ever since.

Below are some expanded definitions of mastering along with some helpful links.

Mandy Parnell: Mastering Audio

Sage Audio: What is Mastering

MASTERING

Mastering, a form of audio post production, is the process of preparing and transferring recorded audio from a source containing the final mix to a data storage device (the master), the source from which all copies will be produced (via methods such as pressing, duplication or replication). In recent years digital masters have become usual, although analog masters—such as audio tapes—are still being used by the manufacturing industry, particularly by a few engineers who specialize in analog mastering.

Mastering requires critical listening; however, software tools exist to facilitate the process. Results depend upon the intent of the engineer, the skills of the engineer, the accuracy of the monitors, and the listening environment. Mastering engineers often apply equalization and dynamic range compression in order to optimize sound translation on all playback systems. It is standard practice to make a copy of a master recording—known as a safety copy—in case the master is lost, damaged or stolen.

PROCESS

The source material, ideally at the original resolution, is processed using equalization, compression, limiting and other processes. Additional operations, such as editing, specifying the gaps between tracks, adjusting level, fading in and out, noise reduction and other signal restoration and enhancement processes can also be applied as part of the mastering stage. The source material is put in the proper order, commonly referred to as assembly (or 'track') sequencing. These operations prepare the music for either digital or analog, e.g. vinyl, replication.

If the material is destined for vinyl release, additional processing, such as dynamic range reduction or frequency-dependent stereo–to–mono fold-down and equalization may be applied to compensate for the limitations of that medium. For compact disc release, *start of track*, *end of track*, and *indexes* are defined for playback navigation along with International Standard Recording Code (ISRC) and other information necessary to replicate a CD. Vinyl LP and cassettes have their own pre-duplication requirements for a finished master. Subsequently, it is rendered either to a physical medium, such as a CD-R or DVD-R, or to computer files, such as a Disc Description Protocol (DDP) file set or an ISO image. Regardless of what delivery method is chosen, the replicator factory will transfer the audio to a glass master that will generate metal stampers for replication.

The process of audio mastering varies depending on the specific needs of the audio to be processed. Mastering engineers need to examine the types of input media, the expectations of the source producer or recipient, the limitations of the end medium and process the subject accordingly. General rules of thumb can rarely be applied.

Steps of the process typically include the following:

1. Transferring the recorded audio tracks into the Digital Audio Workstation (DAW)
2. Sequence the separate songs or tracks as they will appear on the final release
3. Adjust the length of the silence between songs
4. Process or sweeten audio to maximize the sound quality for the intended medium (e.g. applying specific EQ for vinyl)
5. Transfer the audio to the final master format (CD-ROM, half-inch reel tape, PCM 1630 U-matic tape, etc.)

Examples of possible actions taken during mastering:

1. Editing minor flaws
2. Applying noise reduction to eliminate clicks, dropouts, hum and hiss
3. Adjusting stereo width
4. Equalize audio across tracks for the purpose of optimized frequency distribution
5. Adjust volume
6. Dynamic range compression or expansion
7. Peak limit
8. Inserting ISRC codes and CD text
9. Arranging track in their final sequential order
10. Fading out the ending of each song (if required)
11. Dither

PRE-MASTERING CHECKLIST

HAVE YOU:

Created a mix that's musically balanced, tonally balanced, and supports the emotion of the song?

Compared your mix to several references?

Left enough

headroom to avoid clipping?

Checked the entire mix for clicks and pops? (tip: listen on headphones)

Turned off any “analog noise” within your plugins? (see image below)

Removed fade-ins and fade-outs? (unless you’re using an automated service like LANDR, in which case you should leave these in)?

Exported a high-quality WAV or AIFF file? (no MP3s!)

[Written by Jason Moss, Behind the Speakers]

9

Unit Three, Part One: Session Psychology

In previous units, we have looked at how to capture a magical performance in terms of what the audio producer can do on his/her end. But how can we help inspire these moments? What can we do to keep the work moving forward (and on budget) in productive ways whether we are managing sound for others or for our own performances? And at the same time, how can a producer or audio engineer create an atmosphere that encourages good artistic energy that doesn't feel like punching the clock. We call this *session psychology*.

Below are many links and other resources discussing the role and psychological aspects of running a recording session or live sound scenario.

Sylvia Massy-Recording on a Budget

Role of the Music Producer

Tape Op: Studio Etiquette
Breaking Down David Bowie's Heroes
Berklee Online: Studio Etiquette
Recording Session Prep
Working with a Singer and a Songwriter in Studio
Psychology and the Music Producer

PSYCHOLOGY AND THE MUSIC PRODUCER

Psychology and The Music Producer–Jon Marc Weiss and Andre Calihana, Disc Makers Blog, April 15, 2020

To make a great recording, sometimes a music producer has to do it all. From communicating, coaxing the best possible performance, and keeping an artist comfortable, a lot of a producer's skills have little to do with recording techniques.

Jon Marc Weiss is an accomplished recording engineer, studio designer, and musician with 30 years' experience. I sat with Jon to talk about being both an engineer and producer, some of the tricks of recording a good performance, and how to handle things when they start going off the rails.

Is it common to have a person who is solely acting as a music producer on an independent music production?

In higher level projects, there is going to be a record producer, and the producer is going to talk to the artist and act as a liaison with the engineer. The audio engineer is going to do very little communication with the artist. But there's not always a producer, even on some bigger projects. I'd say the minority — maybe 40 percent — of the sessions I've worked on in my career had producers in the room.

A lot of times the producer is the artist, and sometimes they want to make all the decisions, they don't look for any advice from the sound recording engineer. They see the engineer as someone who just pushes buttons. Others see the audio engineer as the person who makes them sound amazing, and they have high expectations.

So that means on 60 percent of the sessions where I work as a recording engineer, I'm actually doing a lot of the producing. I make small suggestions about improving the music, but the majority of the time when I'm not pressing buttons and EQing stuff, I'm working with the artist directly and talking back and forth with them about getting the best take. My job is not only to make the sound recording and handle that side of things, but also to work with the artist and make sure they're happy and that they're getting the most of the experience and their money — whether it's their first time in the studio or their hundredth.

How important is it to set expectations for who is responsible for what before the sessions begin?

It's very important. A lot of times it comes to the point where I have to say to the group, "I have to just talk to one person." I'll accept criticism and suggestions from everyone, but it gets to the point where the guitar player wants something, and the bass player comes in and he wants something different. You have to find a point person. If you've got conflicting stuff coming from all these different people, you're putting the engineer in a difficult situation.

Before the session, you also have to sit with the artist and get a feel for what the style of music is and how much experience they have in the studio. I think that's really important for the recording engineer to know, to know how to coach the person. Someone who's recorded in studios for years and knows how to work the mic and headphones is going to put the headphones on and start singing and you're probably going to get what you need. When it's someone's first time in the studio, or maybe they go in once a year, they're going to be a bit greener and they're going to need some coaching. Not just with their performance, but the way they're working the mic. Or maybe they're crinkling papers and you have to tell them, "Hey, the microphone's really sensitive." Some people don't know that. I've been in sessions where we had to mute the space in between lines so that all the extraneous noise didn't go to tape. Sometimes our job is to help the talent learn how to be a better musician.

I've found that the bigger the artist, the more they tend to let the recording engineer do what they do. Part of it is they're picking you — they have 20 different studios they can afford to go to and they're usually choosing to go to yours because of the engineer's reputation. If you have somebody coming in off the street and it's their first time in a studio and they want to control everything… that can turn into a big mess and setting expectations and roles is even more important.

What's the most crucial thing to focus on as a music producer?

Enthusiasm. Support the artist and be enthusiastic about what they're doing. Even if you're telling them it's not a great take, be enthusiastic about the fact that you think they can do a better job. You have to be good at focusing the artist and getting them to do the best they can. That's obviously different from one performer to the next.

People are very sensitive about their artistry, so you really have to watch what you say. You have to make sure that nothing is condescending and that all of the tips and feedback you give are constructive. Like, if the artist says, "Ah, that take sucked," you might say, "OK, let's roll that again, I think you can get a better one," not, "Yeah, you're right, that really sucked." There's psychology there for sure.

Sometimes you can get a great performance in terms of dynamics and emotion, but the singer is having difficulty singing in tune. As an engineer you have to deal with that. It's common that you'll be working with an artist and there are some bad notes, but the energy and the performance are so good. That's when the producer might decide to come back at a later time and surgically improve the take – I've done it with timing, I've done it with pitch, I've taken a great chorus and copied it and pasted it in other places to keep the energy at a certain level.

Also, if you're an engineer and you're not digging the music at all, you can give some bad vibes off. I don't care what level you're at as an engineer, different levels of talent are going to come through. You may not be as thrilled with today's session as you were with yesterday's, but you have to just deal with what you have, make the best of it, and stay upbeat. The worst thing that can happen, as an artist, is to go into a studio to work with an engineer who is just a dead fish. They're not giving any feedback, they're not into the music — that will kill the whole vibe. You have to bring a certain energy level, and you have to be consistent. You'll always get a better take than if you're negative and don't try to help the artist achieve their best.

Would you say creating a comfortable environment is important, too?

Yes, absolutely. I had this one session, with this vocalist, a young woman, and her dad and her husband were there, and they thought she was going to be the next big thing. But we just couldn't get a good take out of her. Her dad was totally on her — he was like, "When you're in front of your mirror in your bedroom, you do such a good take, and then we come into the studio and you can barely perform." Part of the problem was certainly that her dad and her husband were putting way too much pressure on her. You're not going to get a great performance out of anyone that way.

Another big thing to remember is that this is supposed to be fun. When it starts to become work, it's time to change the focus. So I asked her, "What's different about when you're in your room?" Obviously, she's in a comfortable environment and she's relaxed in her own room, so we brought the mirror, the bedside table, and candles from her room. I suggested she start burning candles in her room when she rehearsed and we replicated that in the studio, and it did the trick. She just needed something familiar to make her feel at home.

Bringing a little bit of home along is a great idea.
Well, sure. When you're rehearsing, it's typically in a room with the whole band, and everyone steps on each other enough to create this blanket of comfort. Then you step into a studio and it can be almost a clinical environment. You've got to be careful as an engineer not to make it too clinical and sterile. You've got to keep the smiles going and keep the vibe going.

I remember you saying that bass players often have a difficult transition from live to studio.
Like anyone else in a band, the bass player tries so hard to get that tone that they hear onstage. But when you record bass guitar, it tends not to sound like what it does when they're standing next to their amp. I've just come to find that bass players tend to be the most difficult to please in the recording environment. Especially when you're working in a mix situation, and the bass player says, "Hey, can you solo my bass? I want my bass to sound like Geddy Lee (Rush)." And you're thinking, "That's not going to work in the context of this music and this recording."

The bass needs to stand out, but not stand out too much. The go-to frequency to pull out of the bass is 250 Hz, because that just muddies up the mix when you've got the bass in there with the guitars and the drums and everything else. You start losing the definition of the individual instruments.

When you're working with the mix, it's a puzzle, you don't want to have all these overlapping frequencies. Now some instruments are going to overlap each other by default, so you have to work to make room for each instrument. You need to scoop out the drums to make room for the bass, and scoop out the bass to make room for the guitar.

So, in many cases, when you solo the track, the bass player's not going to be very happy with the tone on tape. I've had so much trouble with that. I've also had some incredible outcomes, where in the end, the bass player's like "Wow, that sounds so good in the mix! I totally get why you needed to put that top end on the bass."

Got any tricks to settle down a nervous performer? Sometimes they're shredding while they're warming up and then the red light goes on and they totally freeze.

There are lot of ways to settle an artist. A lot of times they're just nervous. Sometimes, for a guitar player or a bass player, it's the fact that they're on the other side of that window that's making them nervous. So you run a cable to the other room, mic the amp, and let the player sit in the studio with the engineer, right in front of the monitors, and that can break the ice.

Sometimes you just have to let them know that there's time, there's no pressure and no hurry, and I'm not going to press record until they're ready. Of course, a lot of times you're still pressing record to see if you can catch something magical. In this one studio, we had a little cap that we could put over the light so you couldn't see that it was recording — that helped with some people.

Jon Marc Weiss is the Director of IT Operations for Disc Makers and also an accomplished recording engineer, studio designer, and musician with 30 years' of industry experience. He owns and operates a private studio called Kiva Productions right outside of Philadelphia in Jenkintown, PA, where he records and produces his own music along with local and national acts. Check out Kiva Productions on Facebook.

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Unit Three, Part Two: Room Acoustics and Sound Treatments

When I set up my first home basement studio, one of my biggest concerns involved keeping the noise out of my neighbors' living spaces. In fact, during some of the "louder" sessions, that might have been *my only concern* since my small place sat smack dab in the middle of a populated neighborhood with houses stacked right next to each other. The couple next door often looked at us suspiciously as vans parked in front of their porches and off-loaded giant amps into my basement door. I made sure I stuffed the window spaces with old blankets. I stapled large pieces of packing foam to the ceiling space above the drum riser. My friends helped me build home-made sound baffles out of wood pallets and carpet pieces.

It worked pretty well. The rest of my house shook anytime we recorded a band in the basement. But the noise didn't carry very far once outside the exterior walls. In the eight years that we recorded in that little basement, we only had one "official" noise complaint.

But, for all that amateurish sound proofing, nothing I stapled to the walls or stuffed under the doors did much to improve the audio *inside* the studio. I did not have the **technical** know-how for managing audio as it bounced off of walls, air vents, and ceilings.

Later, when I worked in bigger and more professional studios, I definitely noticed that whoever designed those spaces took time to consider how sound moved around the different rooms. They weren't just trying to keep the neighbors happy; they were also making the inside of the studio sound better. I saw that a room's reverberation could be controlled with curtains pulled across walls. I noticed that the corners by the ceiling had canvass covered blocks to absorb or deflect unwanted noises. Double paned plexiglass baffles stood around the drums. The distance and direction of the monitor speakers was measured and adjusted to a fixed point. This took not only technical knowledge, but **method** played a big roll as well. How far to pull the sound absorbing curtains across the wall seemed as important as choosing the right microphones.

If I could go back and re-design my old basement studio, I would certainly use those acoustic treatments I learned at the more professional studios. I never considered treating the corners of the ceiling with any sort of sound diffuser or absorber. The ability to control the sound in a recording room allows for more **creativity** in the long run.

Below are various links and other informative material on room acoustics and sound treatments.

How to Set Up Your Home Monitor Speakers

How Sound Works In Rooms

Acoustic Panels–What and Where

Soundproofing is any means of reducing the sound pressure with respect to a specified sound source and receptor. There are several basic approaches to reducing sound: increasing the distance between source and receiver, using noise barriers to reflect or absorb the energy of the sound waves, using damping structures such as sound baffles, or using active anti-noise sound generators. There are 5 elements in sound reduction (Absorption, Damping, Decoupling, Distance, and Adding Mass). The “Absorption” aspect in soundproofing should not be confused with Sound Absorbing Panels used in acoustic treatments. “Absorption” in this sense only refers to reducing a resonating frequency in a cavity by installing insulation between walls, ceilings or floors. Acoustic Panels can play a role in a treatment only after walls or ceilings have been soundproofed, reducing the amplified reflection in the source room.

Two distinct soundproofing problems may need to be considered when designing acoustic treatments—to improve the sound within a room (see reverberation), and reduce sound leakage to/from adjacent rooms or outdoors (see sound transmission class and sound reduction index). Acoustic quieting and noise control can be used to limit unwanted noise. Soundproofing can suppress unwanted indirect sound waves such as reflections that cause echoes and resonances that cause reverberation. Soundproofing can reduce the transmission of unwanted direct sound waves from the source to an involuntary listener through the use of distance and intervening objects in the sound path.

An anechoic chamber, showing acoustic damping tiles used for sound absorption.

Absorption

Sound absorbing material controls reverberant sound pressure levels within a cavity, enclosure or room. Synthetic Absorption materials are porous, referring to open cell foam (acoustic foam, soundproof foam). Fibrous absorption material such as cellulose, mineral wool, fiberglass, sheep's wool, are more commonly used to deaden resonant frequencies within a cavity (wall, floor, or ceiling insulation), serving a dual purpose for their thermal insulation properties.

Both fibrous and porous absorption material are used to create acoustic panels, which absorb sound reflection in a room, improving speech intelligibility.

Porous Absorbers

Porous absorbers, typically open cell rubber foams or melamine sponges, absorb noise by friction within the cell structure. Porous open cell foams are highly effective noise absorbers across a broad range of medium-high frequencies. Performance can be less impressive at lower frequencies.

The exact absorption profile of a porous open-cell foam will be determined by a number of factors including the following:

Cell size

Tortuosity

Porosity

Material thickness

Material density

Resonant Absorbers

Resonant panels, Helmholtz resonators and other resonant absorbers work by damping a sound wave as they reflect it. Unlike porous absorbers, resonant absorbers are most effective at low-medium frequencies and the absorption of resonant absorbers is always matched to a narrow frequency range.

Damping

Damping means to reduce resonance in the room, by absorption or redirection (reflection or diffusion). Absorption will reduce the overall sound level, whereas redirection makes unwanted sound harmless or even beneficial by reducing coherence. Damping can reduce the acoustic resonance in the air, or mechanical resonance in the structure of the room itself or things in the room.

Decoupling

Creating separation between a sound source and any form of adjoining mass, hindering the direct pathway for sound transfer.

Decoupling a wall involves the use of Resilient Isolation Clips or Sound Damping Pads. The clips should be staggered when installed (every other stud) to create fewer pathways for sound to transfer. The Resilient Isolation Channel easily clicks into the Resilient Clips, resulting in a 1 5/8” gap between the stud and drywall. Fine thread screws are used to screw the drywall into the Resilient Channel. Screws should be the correct length in order to not pierce a stud, this will compromise the efficiency of the decoupled wall.

Distance

The energy density of sound waves decreases as they become farther apart, so that increasing the distance between the receiver and source results in a progressively lesser intensity of sound at the receiver. In a normal three-dimensional setting, with a point source and point receptor, the intensity of sound waves will be attenuated according to the inverse square of the distance from the source.

Mass

Adding dense material to a treatment in order to stop sound waves from exiting a source wall, ceiling or floor. Use of Mass

Loaded Vinyl, Drywall, Soundproof Sheetrock, Plywood, MDF, Concrete or Rubber. Different widths and densities in soundproofing material reduces sound within a variable frequency range. Use of multiple layers of material is essential to the success in any treatment.

Reflection

When sound waves hit a medium, the reflection of that sound is dependent on dissimilarity of the surfaces it comes in contact with. Sound hitting a concrete surface will result in a much different reflection than if the sound were to hit a softer medium such as fiberglass. In an outdoor environment such as highway engineering, embankments or paneling are often used to reflect sound upwards into the sky.

Diffusion

If a specular reflection from a hard flat surface is giving a problematic echo then an acoustic diffuser may be applied to the surface. It will scatter sound in all directions. This is effective to eliminate pockets of noise in a room.

Noise Cancellation

Noise cancellation generators for active noise control are a relatively modern innovation. A microphone is used to pick up the sound that is then analyzed by a computer; then, sound waves with opposite polarity (180° phase at all frequencies) are output through a speaker, causing destructive interference and canceling much of the noise.

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Unit Three, Part Three: Home Studio Spaces

When designing your own home audio space, please remember that electrical safety is part of our **technical** pillar and staying within a budget is part of our **method** pillar. It goes without saying that when choosing home recording equipment, safety requirements and cost are both extremely important. Thoroughly research your gear, measure your space, and ask questions.

Another important aspect in putting a home studio together is take time and visualize it, imagine it, and day dream about it. How do you see it in your head, and how do you imagine your recordings sound? What kind of audio recordings will be made in your studio? Start small, but dream big. Read as much as you can about other studios and make notes on what your dream studio will sound and look like.

Creative people and successful people alike are visionaries first and foremost.

My first home studio wasn't much to brag about. It featured a twelve channel mixing board, an early version of a cassette four-track, and about eight cheap microphones. I mixed bands down to DAT tapes that made copies in real time. But I stayed with it and kept making adjustments to my space and equipment as time went on. Making it affordable for other struggling musicians like myself was important to me, so I made it affordable with different pricing to fit an artist's needs. But I also wanted the recordings to sound presentable for college radio broadcast and full of energy. The more I experimented with it, the better the demos started to sound and the more efficiently I could work. Not to mention, it started to help pay the rent. I recorded dozens and dozens of projects over the course of five years and eventually was hired by a bigger studio as my body of work grew.

I tried with each session to expand my technical knowledge along with using solid recording and mixing methods. Since I didn't have top-end equipment, I had to rely on good microphone placement and creative energy more than adding processed effects and expensive pre-amps. In the end, once I got better at those first two pillars, my creative side could take over, and I began to create my own signature sound. I liked the way bands and performers sounded when they felt at ease in my basement late on a Friday night. I understood that music fans can hear when a band is having fun and sounds like they have a secret they are sharing with a listener.

Maybe it didn't sound completely clean and up to professional industry standards, but the more proficient I got at it, the more those demos got passed around town (and sometimes out-of-town). Some of those songs started to receive college radio airplay and positive reviews. One night driving home from a gig, I turned on the radio and heard the late-night college radio deejay say, "Here's the newest single from Shaky Ray Records…" What a great feeling hearing that a song from my home studio was on the radio! And I specifically remember–it did sound like an energetic band with a secret making rock music in my tiny basement studio. Suddenly, it dawned on me that the music sound that my friends and I created was changing the industry standard to something more earnest and energetic. Of course, just like any *sound manager,* I thought, "This is great, but I wish I would have mixed it better!"

Below is an outstanding PDF link from Disc Makers on all the different aspects of putting together a modern home studio.

Disc Makers Home Studio Handbook

MORE USEFUL LINKS AND HOME RECORDING INFORMATION

Home Studio Set-Up

Apartment Music Studio Set-Up

Tinashe TV–Inside Home Studio/Bedroom

Home Basement Studio In Depth

Home recording is the practice of recording sound in a private home instead of a professional recording studio. A studio set up for home recording is called a home studio or project studio. Home recording is widely practiced by voice actors, narrators, singers, musicians, podcast hosts, and documentary makers at all levels of success. The cost of professional audio equipment has dropped steadily as technology advances during the 21st century, while information about recording techniques has become easily available online. These trends have resulted in an increase in the popularity of home recording and a shift in the recording industry toward recording in the home studio

A recording studio is a specialized facility for sound recording, mixing, and audio production of instrumental or vocal musical performances, spoken words, and other sounds. They range in size from a small in-home project studio large enough to record a single singer-guitarist, to a large building with space for a full orchestra of 100 or more musicians. Ideally both the recording and monitoring (listening and mixing) spaces are specially designed by an acoustician or audio engineer to achieve optimum acoustic properties (acoustic isolation or diffusion or absorption of reflected sound echoes that could otherwise interfere with the sound heard by the listener).

Recording studios may be used to record singers, instrumental musicians, voice-over artists for advertisements, dialogue replacement in film, television, and animation, or to record accompanying musical soundtracks. The typical recording studio consists of a room called the "studio" or "live room" equipped with microphones and mic stands, where instrumentalists and vocalists perform; and the "control room", where sound engineers, sometimes with record producers, as well, operate professional audio mixing consoles, effects units, or computers with specialized software suites to mix, manipulate (e.g., by adjusting the equalization and adding effects) and route the sound for analogue recording or digital recording. The engineers and producers listen to the live music and the recorded "tracks" on high-quality monitor speakers or headphones.

Often, there will be smaller rooms called "isolation booths" to accommodate loud instruments such as drums or electric guitar amplifiers and speakers, to keep these sounds from being audible to the microphones that are capturing the sounds from other instruments or voices, or to provide "drier" rooms for recording vocals or quieter acoustic instruments such as an acoustic guitar or a fiddle. Major recording studios typically have a range of large, heavy, and hard-to-transport instruments and music equipment in the studio, such as a grand piano, Hammond organ, electric piano, harp, and drums.

LAYOUT
RECORDING STUDIOS GENERALLY CONSIST OF THREE OR MORE ROOMS:

The live room of the studio where instrumentalists play their instruments, with their playing picked up by microphones and, for electric and electronic instruments, by connecting the instruments' outputs or DI unit outputs to the mixing board (or by miking the speaker cabinets for bass and electric guitar);

Isolation booths are small sound-insulated rooms with doors, designed for instrumentalists (or their loud speaker stacks). Vocal booths are similarly designed rooms for singers. In both types of rooms, there are typically windows so the performers can see other band members and other studio staff, as singers, bandleaders and musicians often give or receive visual cues;

The control room, where the audio engineers and record producers mix the mic and instrument signals with a mixing console, record the singing and playing onto tape (until the 1980s and early 1990s) or hard disc (1990s and following decades) and listen to the recordings and tracks with monitor speakers or headphones and manipulate the tracks by adjusting the mixing console settings and by using effects units; and

The machine room, where noisier equipment, such as racks of fan-cooled computers and power amplifiers, is kept to prevent the noise from interfering with the recording process.

Even though sound isolation is a key goal, the musicians, singers, audio engineers and record producers still need to be able to see each other, to see cue gestures and conducting by a bandleader. As such, the live room, isolation booths, vocal booths and control room typically have windows.

Recording studios are carefully designed around the principles of room acoustics to create a set of spaces with the acoustical properties required for recording sound with accuracy. Architectural acoustics includes acoustical treatment and soundproofing and also the consideration of the physical dimensions of the room itself to make the room respond to sound in the desired way. Acoustical treatment includes and the use of absorption and diffusion materials on the surfaces inside the room. Soundproofing provides sonic isolation between rooms and prevents sound from entering or leaving the property. A Recording studio in an urban environment must be soundproofed on its outer shell to prevent noises from the surrounding streets and roads from being picked up by microphones inside.

EQUIPMENT FOUND IN A RECORDING STUDIO COMMONLY INCLUDES:

A professional-grade mixing console
Additional small mixing consoles for adding more channels (e.g., if a drum kit needs to be miked and all of the channels of the large console are in use, an additional 16 channel mixer would enable the engineers to mix the mics for the kit)
Microphone preamplifiers
Multitrack recorder or digital audio workstation

Computers
A wide selection of microphones typical for different types of instruments
DI unit boxes
Microphone stands to enable engineers to place microphones at the desired locations in front of singers, instrumentalists or ensembles
Studio monitors designed for listening to recorded mixes or tracks
Studio monitoring headphones (typically closed-shell, to prevent sound from "leaking" out into the microphones)
"On Air" or "Recording" lighted signs to remind other studio users to be quiet
Outboard effect units, such as compressors, reverbs, or equalizers
Music stands

ISOLATION BOOTH

An isolation booth is a standard small room in a recording studio, which is both soundproofed to keep out external sounds and keep in the internal sounds, and like all the other recording rooms in sound industry, it is designed for having a lesser amount of diffused reflections from walls to make a good sounding room. A drummer, vocalist, or guitar speaker cabinet, along with microphones, is acoustically isolated in the room. A professional recording studio has a control room, a large live room, and one or more small isolation booths.

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Unit Three, Part Four: Final Project--Design Your Own Studio Space

Here is the scenario for our final project:

Over the summer, you helped renovate your cousin's shop. In return, your cousin has given you the key to the shop's basement for you to build an after-hours recording studio. You have saved up $3,000 and decided to spend that money on the equipment you need to get started.

The basement comes with five approximately 8′ x 10′ rooms and one large closet (Room 6). The rooms are empty except for a nice movable work table. Plus you already own a good computer with the newest version of Pro Tools and a 49 key midi controller.

Other than those items (work table, computer, and Pro Tools with 49 key midi controller), you will need to purchase the equipment required for starting up the kind of studio you've always dreamed of having. You might need microphones, cables, midi keyboards, sound proofing, etc.. All the things we've learned about in this course will apply here. You have to spend at least $2,000 but can't go over your savings of $3,000.

Use the floor plan included to write or draw what each room is being used for and where you plan to place the equipment. **NOTE:** One of the rooms can be used as an already existing bathroom if needed.

Use a separate sheet of paper to list the following:

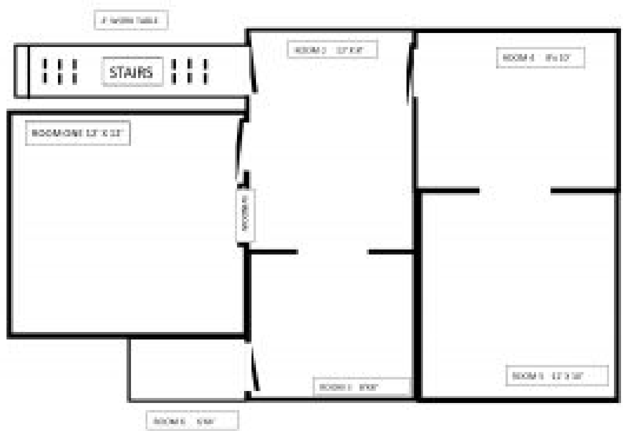
1. **The name of your studio**
2. **What kind of projects is your studio being designed for?**
3. **A brief sentence or two on why people should choose your studio for recording or performing**
4. **What services do you offer?**
5. **What are your rates for these services?**
6. **What is each room used for?**
7. **List each item of audio equipment or other items you plan to purchase, the cost of each item, and which room(s) you plan to use it in**

Remember to reflect on our three pillars of audio production when making choices for your studio.

The instructor will make changes to your rough draft. Upon completing the final drafts, students will present their studio to the class and take questions or address comments.

A blank floor plan is shown and linked below. An example of a final project is also included.

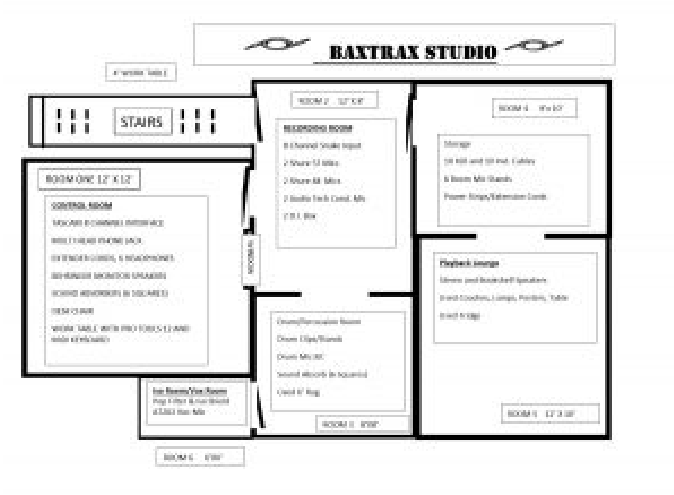
Blank Floor Plan Link



Final Project Completed Example:

BAXTRAX STUDIO

Floor Plan Filled Out



BAXTRAX STUDIO

7355 N. Lakewood St. Baxter, MN

218-555-1643 baxtraxstudio@NTML.com

Rec Room Comfort///Great Live Sound

Pro Tools 12 with essential plug-ins

Studio made for solo performers, small rock and acoustic combos, hip hop, jazz, spoken word

Ideal for high quality demos and 1-3 song based projects

Rates: $35/hr.

Packages: Monday Night Special $125 for four hours (7-11pm)

What you bring: Your own instruments and amps. Original songs (no cover bands).

What Baxtrax Studio provides: Three separate recording areas for performers including designated drum and percussion room, isolation booth and 6'x10' band room. Acoustically treated control room with Pro Tools 12. Listening lounge and easy load-in through large shop doors.

General guideline for booking a session: One song = Two hours of studio time (one hour to record, one hour to mix)

Room 1: Control Room

Pro Tools 12 with Basic Plug-Ins

48 Key Midi Controller

Work Desk Placed in Front of Window

Tascam 8 Channel Interface ……………..………………… $300

Multi Channel Headphone Jack (5)………………… $50

6 Behringer Headphones……………….……………… $250

Behringer Monitor Speakers……..……………………… $125

Wall Mount Sound Absorber (6)……………………… $50

Desk Chair………………………………………………….. $50

Room 2: Recording Room

8 Channel Input Snake………………………………… $100

2 57 Mics.. $200

2 48 Mics... $100

2 Audio Tech Condenser Mic............................. $200

2 DI Box $200

Room 3: Drum Room

Various Drum Mic Clips and Stands....................... $100

Drum Mic Kit... $200

Sound Treatment Squares (6)............................. $50

Rug... ... $25

Room 4: Storage

10 XLR and 10 Various Cables................................ $225

6 Mic Stands.. $150

Power Strips/Extension Cords............................... $50

Room 5: Playback Lounge

Used Couches/Posters/Lamps.......................... $150

Playback Stereo with Bookshelf Speakers............... $200

Used Fridge... $50

Room 6: Iso Booth (Vocals and Solos)

Vocal Iso Shield...................................... $50

Audio Tech AT202 Mic $100

Pop Filter... $25

Total.. $3,000

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Useful Links

What is Sound?

- How Sound Works
- A Quick Guide to Microphones
- Films on Demand–Microphones
- Check Phase on Mics
- Drum Mic Set-Up
- How to Mic a Band
- Cables and Connectors
- Understanding Audio Cables
- How Audio Mixers Work
- Three Rules for EQ
- EQ Explained
- Basic P.A. Set-Up
- Setting Up a P.A. System
- Podcast Audio Gear and Set Up
- Audio Formats

DAW for Beginners
Waves Audio YouTube Playlist
Getting Started with Pro Tools
Help Me Devvon–Audio Tutorials
Behind the Speakers–Mixing Tutorials
What is Mastering and Why It's Important
Disk Makers Blog for Musicians
Interview with Greg Milner, "Perfecting Sound Forever"
History of Audio Recording
Guide to Acoustic Treatment
Do You Need Acoustic Treatment
What Equipment is Needed for a Home Studio
10 Things You Need for a Home Studio
Sylvia Massy-Unconventional Recording
Tape Op Magazine

PART II

About the Producer of this Manual

Mark Lindquist is an award winning audio engineer and musician from Duluth, Minnesota. He currently teaches music and audio production at Central Lakes College in Brainerd, Minnesota. His students write, perform, record, and produce a full length album of original material in a different format each year. Proceeds from these records are donated to the CLC Foundation for students in need of emergency funding. Their work can be found here: CLC Audio Club on Bandcamp


CLC AUDIO CLUB PRESENTS:
A Secret Garden
A compilation album of home, personal, and demo recordings.


CLC Audio Club
MERISTEM

Cover illustration by **Mindy Johnson.** Mindy is an artist and musician from Minneapolis, Minnesota. Her bands The Keepaways and Wolf Blood are both regarded as ground breaking hard core bands from the renowned Duluth, Minnesota music scene.

RAIDER SOUND LABS