Journeyman’s Test Study Guide

I.A.T.S.E. Local 122

Audio

(part 1)

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Basics of Sound

A MODEL OF A SOUND SYSTEM

In its simplest form, a sound system operates by converting sound waves (physical energy propagated by air or in some sort of medium) into electrical energy, increasing the power of the electrical energy using electronic circuitry, and then converting this resultant electrical energy back into physical energy in the form of sound waves.
If you have no idea of what we speak, go pick up a physics textbook and read the parts about sound and waves and air pressure and why it's more difficult to hear when you're in an airplane.

Devices that convert energy from one form into another are called **transducers**. Some examples of common transducers are loudspeakers, microphones, contact pickups, and headphones. Microphones and contact pickups (types of **input transducers**) convert fluctuating sound waves (physical energy in the air or in some other medium) into an electrical current that is an analog representation of the original sound wave. That is to say, if a large mass of air produced by a kick drum is picked up by a microphone, a large electrical current will represent that mass of air. Loudspeakers and headphones (which are mini-loudspeakers, essentially), converts the electrical signal back into physical energy.

Devices that actually change the characteristics of the electrical audio signal are called **signal processors**. In its simplest form, a signal processor increases the power of the electrical signal (coming out of the microphone and going into the loudspeaker). This sort of signal processor is an **amplifier**. Sound systems often include many more types of signal processors, which are used to effect change on any number of different audio signals.

GOES-INTA

Here are some examples of input transducers:

Air pressure or velocity microphones:
convert sound waves traveling in the air into an electrical signal, which travels over, say, a microphone cable. A diaphragm, usually made of some very fine metal, vibrates within a magnetic field, which results in an electrical current.

Contact pickups:
convert sound waves traveling in a medium (wood, metal, skin) into an audio signal. Commonly found on instruments such as acoustic guitars, concert basses, violins, mandolins; sometimes found on pianos; sometimes used on dance floors to pick up tapping sounds...

Magnetic pickups
convert fluctuating waves of induced magnetism into an audio signal, such as found on an electric guitar.

Magnetic Tape heads
convert fluctuating magnetic fields recorded along a band of metal-coated plastic into an audio signal.

Laser pickups
convert patterns of pits and bumps imprinted on a DVD, CD, LD, or MD into a digital data stream that is then translated into an analog audio signal.

Optical pickups
convert variations in density or light on a band of photographic film into an audio signal. This is how your local cinema projector works.

GOES-OUTTA

Here are some examples of output transducers:

Loudspeakers: woofers:
designed to reproduce low frequencies. A woofer loudspeaker works exactly in the same fashion that a **dynamic microphone** works, but backwards. **Subwoofers** are loudspeakers specifically designed to reproduce very low frequencies.

Loudspeakers: midrange:
designed specifically to reproduce middle frequencies. Probably looks a lot like a small woofer.

Loudspeakers: tweeters:
designed to reproduce the highest frequencies. There are several types of tweeters, including **dome tweeters** and **compression-drivers**.

Go to http://en.wikipedia.org/wiki/Loudspeaker#Driver_design for a description of loudspeaker designs and builds.

Headphones:
full-range transducers designed to fit in or on the ears.

While this is by no means a complete list of transducers, it gives you an idea of what we speak. There are advances in loudspeaker technology, and developments such as a planar, or flat, loudspeakers have come into use; however, the basic way in which the loudspeaker works has not changed.

GOES-SOMEWHERE-IN-THE-MIDDLE-A

Here are some examples of signal processors, and what they do. It should be noted, however, that the technical definition of "signal processor" is a device which alters the audio signal in a **non-linear** fashion; thus a simple level control or amplifier is technically not a signal processor:

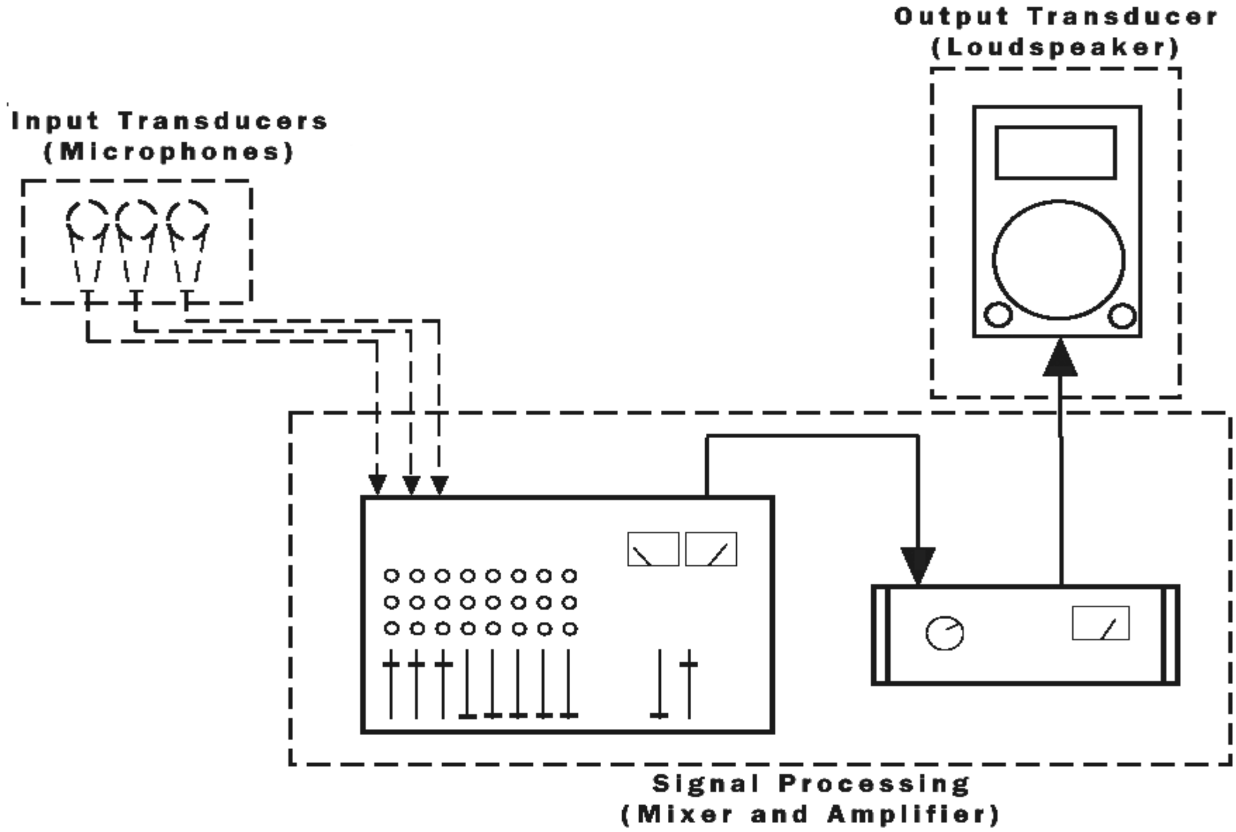
Mixing Console, Mixing Desk, Mixer, Console, whatever:
Whatever nomenclature you use, it's still basically the same thing. In its simplest form, a mixing desk takes more than one input (in its electrical form) and sums the signals together into more than one output (still in electrical form). A more complex example could be a desk used for live performance, with forty-eight inputs (vocal microphones, band microphones, etc.), and sixteen outputs (loudspeaker system on stage, loudspeaker system in the house, more loudspeakers in the rear of the house, a couple of tape-recorders, etc., etc.). The mixing desk is your central control station; this is where you make the vocalists louder than the band, or the violinist softer than the cellist.

Equalisers (or Equalizers, depending on your preferred side of the pond):
Equalizers are units that affect certain portions of the frequency spectrum with a boost (gain) or cut (attenuation). From your electrical engineering knowledge, you know that a circuit that affects a certain portion of a given frequency spectrum is called a **filter**. You will see the word "filter" used quite often when describing bands of equalization. We'll go into more detail later.

Echo/Delay:
Delay and Echo should not be confused with Reverberation. While reverberation is the homogeneous sound image caused by multiple reflections, delay refers to one or more distinct sound images, repeated over time. Delay units allow for special, other-worldly effects when used in a recording situation; in live sound delay units are used to correct time differences between different systems of loudspeakers. (More will be discussed later).

Amplifiers:
Amplifiers are a basic building block of electrical engineering. In its theoretical form, an amplifier takes an electrical signal and increases its power in a linear fashion- that is to say, the output waveform matches the input waveform, only it's more powerful. While the linearity of common power amplifiers is not necessarily perfect, that is essentially what a common power amplifier does. Power amplifiers are units designed to take an audio signal and make it powerful enough to drive a loudspeaker; due to the loudspeaker construction and its inherent inefficiency, a lot of power is required to produce sound. Other types of amplifiers used in audio include preamplifiers, which are small amplifiers that take an audio signal from a microphone and increase its power so that it can be better manipulated.

A BASIC SOUND SYSTEM



The illustration above illustrates a simple, practical sound system. Note the four sections- input transducers (microphones), mixer (pre-amplifiers), amplifier, and output transducers (speakers). The three microphones are connected to separate inputs on the mixing desk. On each input the mixing desk provides pre-amplification, which amplify the **microphone level** signals to **line level**; equalization, which provides the means to contour the tonal balance of each microphone; and level control, which allows the operator to adjust the relative level of each microphone individual. The mixing desk then sums the inputs to a single line-level output. The output of the console is connected to a power amplifier, which boosts the console's line level output to a level suitable to drive the loudspeaker (line levels typically run from 0.1 to 100 mW, whilst loudspeakers require 0.5 to 1 kW, approximately). The loudspeaker converts the power amplifier output signal into sound pressure waves. The level of the sound is much higher than that of the three orators speaking unaided.

Every sound system is merely an extension of this basic model. The principles that apply to this simple model also apply to large-scale concert reinforcement systems.

THINGS TO REMEMBER

The environment in which the sound system is used can alter, both positively and detrimentally, the output of the system. In a free-field environment, such as on a field-hockey field, there are very few objects that will reflect the sound- trees, grass, and girls in field-hockey garb will tend to absorb sound, rather than reflect it. We can eke more level out of the sound system because we do not have to worry about reflected sound waves getting back into the microphone, causing a rather unpleasant ringing sound known as **acoustic feedback**. In a small room with wooden walls, we have to worry about reflections caused by the amplified sound bouncing off the walls and affecting the overall intelligibility of the system; we are concerned with the amount of **gain** we can get out of the system before the unpleasant feedback sound. In a small room with padded walls (heh heh), we are less concerned about intelligibility as the padding will tend to absorb sound rather than reflect it. Rooms that are rather non-reverberant are usually termed "dead", where-as highly reflective rooms are considered to be "live". When using an amplified sound system, less reverberation in a room is normally a better environment because there are fewer reflections in the room, which can confuse proper articulation in the listening environment.

There are other factors to consider when designing and installing a system, including proper speaker selection and positioning, proper microphone selection and positioning, and proper tuning or equalization of a system in a given room. And of course there are issues on ease-of-operation and the system's intended use—whether for music, speech, theatre, or playback.

Input Devices 'n' Stuff Like That...

LINE INPUTS

The term **line level** is defined as a standard voltage for the audio signal outputs of many pieces of sound equipment. -10dBV (or 0.316V rms) is the generally-accepted standard for so-called "semi-professional" equipment, and +4dBm (or 1.23V rms) is the generally accepted standard for "professional" equipment. Technically any voltage over 25mV rms is considered "line-level," but in the modern audio world we use the -10dBV and +4dBm references. Line-level outputs can come from equipment such as cassette decks, CD players, DAT decks, MD players, DAW computers, and the list goes on. Outputs from mixing desks and subsequent pieces of equipment are also line-level.

A **line input** is an input designed to receive line-outputs, at either standard, and probably any signal that exists between the two standards. Most mixing desks will provide either a separate input connector, or a pushbutton switch, by which it knows to accept the line-level signal. In theatre, one would presumably connect some sort of playback device to a line input on a desk and use it for background music or sound effect playback.

As a very basic practical note, remember when plugging things in that line-levels are much "stronger" than **microphone levels**; if your mixing desk is expecting a microphone-level signal, and you feed it a line-level signal, one or more circuits on your desk are going to overload, resulting in distortion. While distortion in and of itself will not necessarily harm the mixing desk, if subsequent pieces of equipment are powered on and active, severe distortion in loudspeakers can be damaging, not to mention annoyingly loud. If you have to patch equipment while the equipment is on (which is not a *wonderful* idea), keep all level controls <u>down</u>.

So, what are some things we might be connecting to an input?

PLAYBACK GEAR

The outputs from popular playback gear, such as DAT decks, MiniDisc machines, Compact-Disc machines, computer-based audio workstations, and the like all have line-level outputs. Depending on the piece of gear, the outputs may be unbalanced -10dBV signals or +4dBm balanced signals. Unbalanced outputs usually terminate in ¼" TS jacks or RCA (Cinch) jacks, whilst balanced signals usually terminate in three-pin XLR-type connectors or ¼" TRS jacks.

INSTRUMENTS

Electronic instrument levels are comparable to line-level signals. Generally, they are unbalanced semi-pro -10dBV signals. If the destination, i.e. your mixing desk, is in close proximity to your keyboards, samplers, tone generators, drum machines, or other electronic devices, then there is very little reason not to take an output directly out of the instrument and connect it directly to a line input on the mixing desk. If, however, it is necessary to place the mixing desk far away from the instruments (i.e. in the case of orchestra pit to front-of-house), one should use a **direct box** in order to rectify impedance differences between the instrument and the mixing desk, and to take the high-impedance -10dBV unbalanced signal and convert into a low-impedance microphone level balanced signal.

DIRECT BOXES

Direct boxes are devices that convert unbalanced -10dBV line level outputs into balanced, microphone-level (or line-level) inputs. They are used heavily in sound reinforcement, when it is necessary to drive long runs of cable between instruments and the front-of-house mixing desk. Its operation is quite simple—take an output from the instrument in question and patch it into the "Input" on the direct box. A "Link" output of the direct box allows daisy-chaining of the instrument output into a local amplifier or mixer arrangement. The "Output" of the direct box, usually an **XLR connector**, is the microphone-level balanced output, which is run to the front-of-house mixer. Direct boxes can also have other options—an internal **attenuator pad** switch, a low-frequency rolloff switch, and/or a **ground lift** switch. Attenuator pads can aid in reducing the level at the mixing desk from a hot signal, an LF rolloff switch can aid in cleaning up low-frequency garbage, and ground lift switches can aid in rectifying potential (ha ha!) **ground loops**—if audio equipment is grounded in too many places, a loop of potential difference will form, and will induce a hum into the audio system. The ground lift switch will disconnect the shield of the microphone cable from the direct box, thereby eliminating one possible loop point.

MICROPHONES

The term **microphone level** is the level, or voltage, of signal generated by a microphone, usually somewhere around 2mV. Compare this number with that of the two line-level specifications (0.316V rms and 1.23V rms), and you see how significant the difference is. The output level from a microphone is this small due to the size of the transducer element—a very small amount of air pressure moves a very small, thin diaphragm through a small magnetic field. The resultant signal is also very small. In order to cleanly process, mix, and divide the microphone-level signal, we must use a **preamplifier** to boost the miniscule audio signal from the microphone to approximately line-level so that the signal may be manipulated at a decent level. Microphone preamplifiers, or "mic pres", are commonly found inline on the input channel of most mixing desks. There are also standalone units whose sole purpose is to amplify microphone-level signals to line-level; the use of external mic preamps is often noticed in studio sound recording, where every bit of quality counts.

There are thousands of different types of microphones available... and every single one has its own electrical, mechanical, and tonal qualities that make it suited for one or another type of application.

ELECTROMECHANICAL THEORY

As we discussed previously, microphones convert air pressure fluctuations into electrical energy. So how do they do that? Well, there are a couple of different methods. One of the most common types of microphones is the **dynamic microphone**. The dynamic microphone works very much in the same fashion as a loudspeaker—except in reverse. A flexible diaphragm is mounted to a coil of fine wire. The coil is mounted in the air gap of a magnet such that the diaphragm-and-coil are free to move within the air gap of the magnet. When a sound pressure wave strikes the diaphragm, the diaphragm-and-coil vibrate in response; the coil moves back and forth within the magnetic field provided by the magnet. As the coil moves back and forth through the lines of magnetic force, a small electrical current is induced in the gap. [This is standard electromagnetic physics.] The magnitude and direction of the induced current is directly related to the motion of the coil; thus the current is an electrical representation of the sound pressure wave, which struck the diaphragm.

Dynamic microphones are very durable. They are extremely common in rock-and-roll sound reinforcement or in other instances in which they may be subject to shock or extreme environmental conditions. They are good, all-purpose microphones that have predictable characteristics and low cost.

Another type of common microphone is the **condenser microphone**. The condenser microphone operates by creating an electrical capacitor whose capacitance varies as a function of the sound pressure wave. A thin diaphragm is stretched in front of a metal disc, called the backplate. The diaphragm and backplate do not touch each other, but are very close, and thus create a capacitor, or condenser. When a sound pressure wave strikes the diaphragm, the distance between the diaphragm and the backplate changes, thus changing the capacitance of the condenser. If the capacitor system is given a fixed electrical charge (called "polarizing voltage"), changing the capacitance of the condenser will proportionally change the backplate voltage as a function of the sound pressure wave. The backplate voltage, then, is the audio signal. In order to create the fixed electrical charge, an external voltage must be applied to the microphone, which is customarily done through the use of a battery pack, or, more commonly, through the use of **phantom power**, voltage sent via the microphone cable from a microphone preamplifier or the mixing desk.

Condenser microphones are slightly less durable than their dynamic counterparts, so they are not used as much in rock-and-roll situations, where they may be susceptible to shock. In orchestral reinforcement, however, their sound quality cannot be beaten by a dynamic microphone. In theatrical sound reinforcement, you may find condenser mics anywhere—in the orchestra pit, on actors themselves as part of a wireless microphone system, as shotgun (rifle) microphones, or as foot (float) microphones.

POPULAR POLAR PATTERNS

Polar patterns, also called **pickup patterns**, describe the way in which a microphone element will respond to sounds at different frequencies coming from different directions. Think of the polar pattern as the targeting range of the microphone in question. It is a polar graph that displays the pickup pattern on all four axes (0°, 90°, 180°, and 270°).

Omnidirectional microphone elements, as their name implies, pick up sound pressure waves more or less equally from all directions. Omnidirectional microphones are used often in recording studios and in other situations in which **gain before feedback** is not of great concern. They are also the preferred type of lavalier microphone used in theatrical sound reinforcement, mainly because the omnidirectional polar pattern does not emphasize the resonant low-frequencies of the chest cavity—the phenomenon of low-frequency emphasis is known as the **proximity effect**.

Probably the most popular type of microphone **capsule**, or element, is the **cardioid** element, also known as a **unidirectional** pattern. While the omnidirectional microphone element is more-or-less equally sensitive to sound coming in from all directions, the cardioid element is most sensitive to sounds coming in along its main axis (0°), and rejects sound waves coming in along its rear axis (180°). The cardioid pattern rejects some sound from the side axes (90° and 270°). The directional characteristics of the cardioid capsule make it an ideal choice for sound reinforcement applications in which system gain and the reduction of feedback are primary concerns. Because of their directionality, one can "aim" the microphone at the sound source in the best possible fashion in order to reject extraneous noise.

A derivative of the cardioid pickup pattern is the **supercardioid** pattern. The supercardioid rejects more sound from the sides than a cardioid microphone, but it has a small rear lobe at 180°. They are used in situations in which greater side rejection is necessary, but some rear pickup is tolerated. The supercardioid microphone also has greater directionality from the front, and are often used as shotgun, or rifle, microphones in video and film sound production.

A slightly stranger but very useful pickup pattern is the **figure-8** or **bidirectional** microphone. As the name implies, bidirectional microphones are most sensitive to sounds coming in along the front or rear axes (0° and 180°), but reject sounds from the sides (90° and 270°). They may be used in reinforcement situations in which two adjacent instruments require microphones, such as in between two tympani drums in an orchestra pit.

GAIN!

Gain in a sound system is the amount of amplification or additional level obtainable in a sound system, quantifiable in decibels or as power or voltage ratios. Proper system structure, proper microphone placement, and proper loudspeaker placement will all help to maximize a sound system's acoustic gain. Briefly, when dealing with microphone-cum-loudspeaker placement, we should keep the distance between the microphone and the loudspeaker as large as is humanly possible, keep the distance between the microphone and the sound source as small as possible, and use directional microphones and loudspeakers, well-placed so the interaction between the two is minimized.

Taken from Kai Harada's Sound Handbook Online