Monitors

One of the most confusing aspects of audio reproduction involves loudspeakers and their effect on how we perceive the sound of our work. Just as microphones each have a characteristic sound, speakers have distinct colorations which they impart to the sound we play through them. The similarities are not coincidental: both microphones and speakers are physical transducers, with mechanical properties which directly affect how they perform the task of converting energy to and from mechanical and electronic representations.

Since we cannot hear electronic signals directly, we need some form of transducer to convert them back to mechanical vibrations we may hear. Like microphones, loudspeakers come in a variety of types; however the main type is similar to a dynamic microphone in reverse: applying an electronic signal to a coil attached to a diaphragm causes the coil and diaphragm to move in a fixed magnetic field. There are also electrostatic loudspeakers which may be thought of as more similar to a condenser microphone and even plasma speakers, which use high-voltage discharges to move the air, but these types are not commonly found in the studio and will not be discussed further here.

![Dynamic loudspeaker diagram](image)

Figure 1: Dynamic loudspeaker

Figure 1 shows a typical dynamic loudspeaker. The voice coil attaches to the speaker cone, which is attached to the frame with flexible material forming a surround. The coil sits within the magnetic field created by the permanent magnet. When current flows in the coil, its magnetic field interacts with the fixed magnetic field and causes the coil and cone to move. This is very similar to a dynamic microphone in construction and loudspeakers can actually be used as microphones if attached to a bass drum, for instance. Like microphones, there are many variations on the basic design that produce quite different sounding reproduction of the driving signal.

The speaker in Figure 1 radiates sound as a dipole, that is it projects sound waves both forward and backward. It is unable to efficiently couple its movement to the surrounding air and therefore produces little sound. In order to best couple its movement to the air, it needs to be placed in an enclosure that can optimize its ability to move the surrounding air. There are several ways to enclose the speakers, each of which has advantages and disadvantages.
Figure 2: Speaker cabinet types.

Figure 2 shows how a speaker can be mounted in an enclosure. The infinite baffle prevents any sound pressure from the front reaching the rear. Half of the air motion would be audible in the front of the speaker and half at the rear, effectively wasting half of the radiated power. This is not a practical way of mounting speakers in most situations. The finite baffle also isolates front from back, though not completely, but is far more practical for moveable speakers. Many combo guitar amplifiers use this system. The acoustic suspension system fully encloses the rear of the loudspeaker. This prevents sound waves from the front and rear from interfering with each other but is inefficient, in part because it wastes the rearward motion but also because the trapped air behind the speaker can present a non-linear load on the motion of the speaker cone. Acoustic suspension speakers are capable of good sound but require high power amplifiers. They may have less low frequency output for a given size, but they have a tight, defined bass sound. Many speakers use the bass reflex cabinet. Here, the energy produced in the rear of the speaker is channeled through a carefully tuned port to the front. Bass reflex speakers are efficient and produce extended bass response.

Unfortunately it is quite difficult to make a single loudspeaker that is equally efficient over the whole range of audible frequencies. The wavelengths involved range from many meters to a few centimeters. Speakers that produce good low frequency coupling are too large to move as quickly as a 15 or 20 kHz sound requires. There is also the issue of Doppler shift when the radiating surface is moving large distances at a slow rate while also vibrating very quickly to produce the high frequency content of the sound. For these reasons, most monitor speakers are assembled from two or three separate drivers. In order to do so, the signal must be split by filters so that each driver receives only the frequency range it is designed to reproduce. This is the cross-over filter. Speakers that are driven from external power amplifiers require built-in cross-over filters that can handle large currents and voltages since the driving power signal is split to each driver passively. Active monitors, those with built-in power amplifiers for each driver, use cross-over filters at signal level, requiring less robust components. The amplifiers can also be carefully matched to the individual driver elements. Self-powered speakers are the default now.

When mounting multiple drivers in a speaker cabinet, their physical arrangement is important. Since our sound signal is split and fed to different drivers, the time alignment of the sections must be carefully preserved. This means the individual speakers must be mounted so that each radiates the sound at the same time in order to recreate the original sound wave. Many times, the speakers seem to be placed at different
depths on the front of the cabinet. This is because their acoustic centers, the plane from which the sound is generated, are at different depths on each driver. The acoustic centers must be aligned to produce the coherent wavefront we expect.

![Figure 3: Radiation patterns for typical multi-driver loudspeaker.](image)

Each radiating source in a multi-driver loudspeaker generates its own directional pattern. Low frequencies are widely dispersed while higher frequencies tend to be beam-like (See Figure 3). This complicates speaker design as we need the listener to receive equally the sound waves from each driver. For a listener directly on-axis, the sound is balanced while one sitting off-axis hears a different ratio of high-to-low frequencies. This directionality also affects how the speaker interacts with the room in which it is positioned.

In addition to the speakers’ design, their placement in the room can have a profound effect on what we hear. Low frequencies tend to “build up” through resonance and reinforcement near walls and especially in corners. Therefore, the placement of the speakers should minimize these problems. They should also be positioned so that reflections will not bounce back and interfere with the direct sound radiation. Many monitors have controls that alter their response so they can be placed against a wall, in a corner, or in free space without significant difference in sound. People often think they can fix bad room sound by equalizing the sound system: this is a mistake. The effects that tend to ruin the sound are time-domain problems, involving time delays associated with sound bouncing off of surfaces and creating interference patterns in the room. No amount of equalization will fix this except possibly for a tiny spot in which the measurement microphone is positioned. Standing waves must be addressed by absorbers and diffusers built into the room to break up reflections and absorb unwanted accumulations of bass frequencies.

Speakers are also classified by the distance at which they are designed to focus their sound output: near-field, mid-field, and far-field. These terms refer to the listener’s relative perception of the direct to reflected sound ratio. Near-field speakers are designed to be heard in the near field, where the direct sound far exceeds the reflected sound intensity. Near-field monitors are desirable in part because their sound is largely independent of the room, hence their popularity for home and project studios where acoustics are often less than perfect. Mid-field monitors are designed to be heard from a longer distance, where the reflected sound and direct sound are about equal. Far-field speakers are designed to “throw” the sound a longer distance from the speaker, where the reverberant field may be stronger than the directly radiated sound. Far-field speakers are often soffit or wall mounted in large control rooms.

Finally, there are the very near-field speakers we call headphones. Headphones are small speakers mounted for direct-to-the-ear delivery of sound. While they are often used for monitoring during recording, for overdubbing, for example, they are generally not designed for use in critical tasks like mixing. Although some headphones are designed as free-field, they produce effects quite unlike speakers since they are directly coupled
to our ears with little air to intervene. Whereas loudspeakers produce sound that appears to be coming from the recorded source in space, headphone playback seems to be coming from inside our heads. Headphones can be useful in determining panning (spatial placement in a mix) and for hearing effects like reverb which may tend to get lost in a room’s acoustic field, but they are not likely to represent the sound any loudspeaker is going to produce very accurately. And a caution: it is quite easy to deliver dangerously loud sounds via headphones and ear fatigue can soon be a problem. It is strongly recommended that prolonged headphone use be tempered by frequent rest periods.

Just as with microphones, we must practice and experience the results over time in order to fully appreciate how the monitors we use affect the sound we produce. Fortunately, CCRMA has several different sets of monitors that can be compared to evaluate the contribution of the loudspeaker to the final sound of our recordings. The main control room speakers are Westlake BBSM-10s, old but reliable passive monitors. These speakers have limited bandwidth by today's standards, but within their range are very flat and have excellent imaging. They are robust and good for tracking. The other main control room speakers, ADAM A77x monitors, are extended range speakers using ribbon tweeters that produce up to 35 kHz output while the woofers go as low as 35 Hz. Within the frequency range both cover, there are distinct similarities with the BBSM-10s. The A77x speakers allow a better idea of the low frequency content than the Westlakes without needing to introduce a subwoofer. There is also a pair of JBL 4208, which have an exaggerated, rather boomy low end that approximate many home stereo speakers. It is possible to create mixes which sound good on all of these systems.

While we usually mix on a single set (or at most a couple of sets) of monitors, we can take our mixes around to the other studios and listen on different speakers in different rooms to get an idea of how the studio mixes translate in other settings. This often reveals some problems we might not have heard on the loudspeakers in the control room. Listening on other audio playback devices such as computers, iPods and car stereos will also reveal information about the mix that might not be obvious on studio monitors. You will find that the more time you devote to evaluating your mixes on the different monitor systems, the less mastering will be required for your final product. Our ultimate goal is to be able to hear a mix on the studio monitors and know how it will sound everywhere it might be played. That takes lots of experience.