Technology Brief 24
Electrical Engineering and the Audiophile

The reproduction of high-quality music with sufficient fidelity to sound like a live performance in one's living room was one of the technological hallmarks of the 20th century. In these days of iPods and online music distribution, good music is increasingly accessible to many people. The price of good quality tuners, amplifiers, and speakers continues to drop, and driven mostly by demand for home entertainment audio/video systems, audio equipment is increasingly "user-friendly." The reproduction of theater-quality or live-performance sound in a confined space is challenging enough to be a profession unto itself. It also can be a very rewarding technical hobby for the well-versed electrical engineer. In this Technology Brief, we will cover some of the basics of audio equipment and relate them directly to the concepts taught in this book. Several good audiophile websites exist with more in-depth treatments of these (and other) topics; beyond the audiophile community, the sub-field of audio engineering has an extensive academic and professional literature to consult.

The Basics

Reproduced sound starts out as an analog (e.g., the vinyl record) or digital (e.g., the mp3 file) recording. How that recording is made from real sound with high fidelity is beyond the scope of this Brief (and is a large component of the audio engineering profession). That recording is converted into an electrical signal that is first amplified and then transmitted via cables to speakers. **Figure TF24-1** shows a schematic of the process.

The ***audible spectrum*** of the human ear extends from about 20 Hz to 20 kHz, although the frequency response may vary among different individuals depending on age and other factors. An audio signal is a superposition of many sinusoids oscillating at different frequencies—each with its own individual amplitude. When we say a sound has a lot of ***bass***, for example, we mean that the low-frequency segment of its spectrum (20 to 100 Hz) has a large amplitude when compared with higher-frequency components. Conversely, very shrill or high-pitched sounds have large-amplitude components in the high-frequency range (10 kHz to 20 kHz). When converting an electrical recording back into the original sound that generated it in the first place, the reproduction fidelity is determined by the degree of distortion that the spectrum undergoes during the playback process. In practice, minimizing ***spectral distortion*** can be quite a challenge!

Each component of the sound-reproduction system shown in **Fig. TF24-1** is characterized by its own transfer function relating its output to its input, and since each of these components is equivalent to a circuit composed of resistive and reactive elements, its transfer function

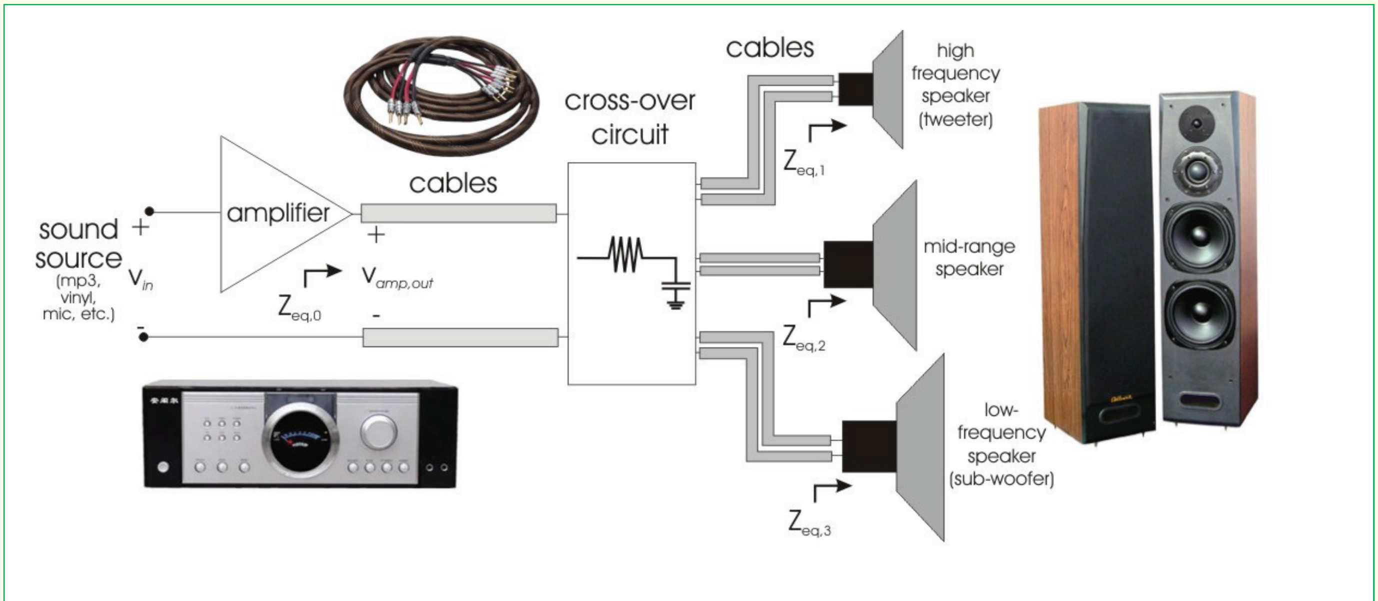

Figure TF24-1: Schematic of a basic audio-reproduction system.

is bound to exhibit a non-uniform spectral response. The amplifier, for example, may act like a filter, favoring parts of the audible spectrum over others. The cables, which behave (electrically) like the RC transmission line of **Fig. 7-35**, will favor low-frequency spectral components over high-frequency components. Thus, unless the components of an audio-reproduction system are well designed in order to generate a transfer function with a nearly flat spectral response over the audible range, the reproduced sound will exhibit a distorted spectrum when compared with the original spectrum. While there are many objective metrics by which to judge the fidelity of audio equipment, every listener processes a given sound differently, introducing a subjective component into the experience. A great way to appreciate the concepts introduced in this Technology Brief is to walk into a high-end audio-systems store with three favorite CDs and then to listen to them on many different amplifier-speaker combinations.

Amplifiers

It takes quite a bit of power to drive speakers to produce sound in a room. The function of an amplifier is to boost the audio signal's power high enough to drive the speakers. In doing so, the amplifier must:

- keep frequency distortion to a minimum, and
- introduce as little noise as possible into the signal.

In order to keep frequency distortion to a minimum, the amplifier's response must be as uniform as possible; in other words, signals of different frequencies and different amplitudes must be amplified with exactly the same gain. To address this, many different transistor–amplifier topologies have been developed over the years. These amplifiers are grouped into classes based on behavior and topology; the principal differences lie in circuit complexity, power consumption, and the degree of fidelity with which the circuit reproduces an input signal. Audio-amplifier circuit topologies are categorized by letter—currently from A to G. Although a description of each class lies beyond the scope of this discussion, very succinct overviews can be found in many places online. In order to reduce the noise during the amplification step, two-amp stages often are used. The first stage is called the pre-amplifier. Pre-amps have very good noise characteristics and amplify the signal partway (this mid-level signal is called the ***line signal***). Often, this is simply an amplification of the voltage level. The power amp then boosts this signal (which does not have much noise) to a level high enough to drive speakers; this usually requires significant current amplification to provide enough overall power to the speakers.

Cables

The cables that transfer a signal between sources, amplifiers, crossovers, and speakers can themselves distort the signal. Cables behave exactly like the transmission line of **Fig. 7-35**; the distributed resistance and capacitance act like a filter with an associated frequency response. In general, cables should be:

- as short as possible,
- properly impedance-matched to both the output of the amplifier and the input of the speakers, and
- properly terminated so the cable connections to the equipment do not introduce capacitances.

All three of these objectives easily are accomplished using industry-standard cables and connectors. In some modern systems, transmission of audio signals between non-speaker components (e.g., from a tuner to an amp or from an amp to a TV) is often performed in digital form so as to eliminate both noise issues and frequency distortion.

Speakers

A speaker is any electro-mechanical device or transducer that converts an electrical signal into sound. Electromagnetic transducers are the most commonly used type for consumer audio applications (**Fig. TF24-2**),

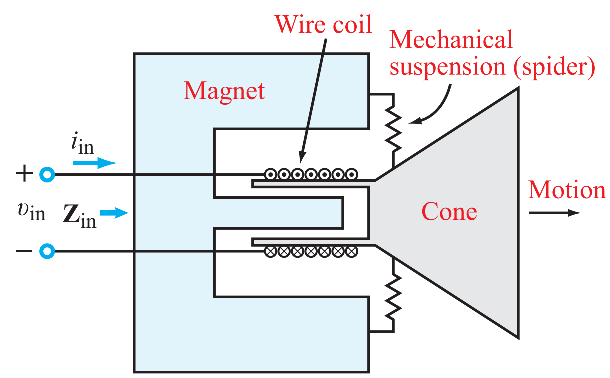

Figure TF24-2: Conceptual illustration of an electromagnetic speaker transducer. The current from the amplifier runs through a coil that induces an electromagnetic force on the cone in proportion to the amplitude of the input signal. The cone motion produces pressure waves and, hence, sound.

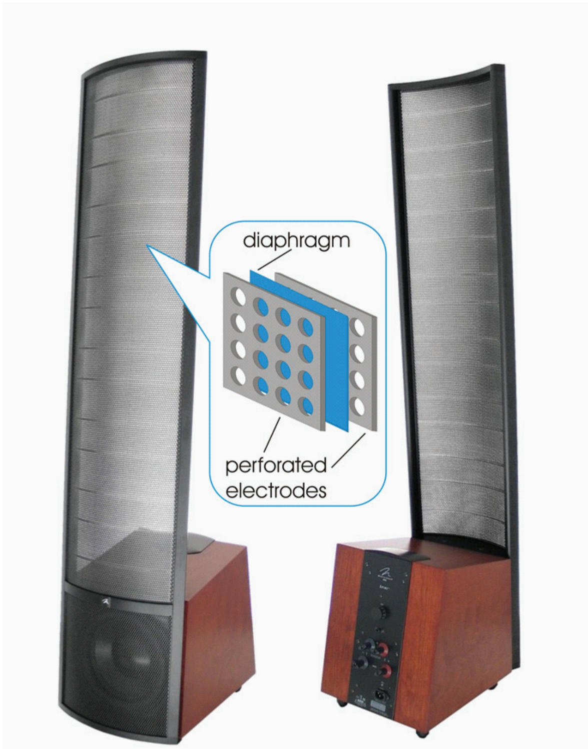

Figure TF24-3: Electrostatic speakers consist of a very thin ($\sim 20\ \mu$m) polymer membrane, called a diaphragm, which is coated with a conductor. This membrane is suspended between two perforated electrodes. The diaphragm is held at a dc potential of several kV. The voltage between the electrodes and the diaphragm is driven by the amplified audio signal so as to displace the diaphragm and move air (which produces sound). The principle behind actuation is very similar to that discussed in Technology Brief 10: Micromechanical Sensors and Actuators. Most electrostatic speakers have poor base response, so they are usually paired with subwoofers (like the one shown here). (Image courtesy of MartinLogan, Ltd.)

although several other technologies, such as electrostatic speakers (**Fig. TF24-3**), exist as well. The principal metric when choosing a speaker is arguably its frequency response (**Fig. TF24-4**). Ideally, a speaker will provide a very flat response. This means that signals at different frequencies recreated into sound all at the same audio level. Generally speaking, very small speakers have difficulty reproducing very low frequencies (i.e., bass); a deep drum or baseline may be lost entirely when listening through a small speaker.

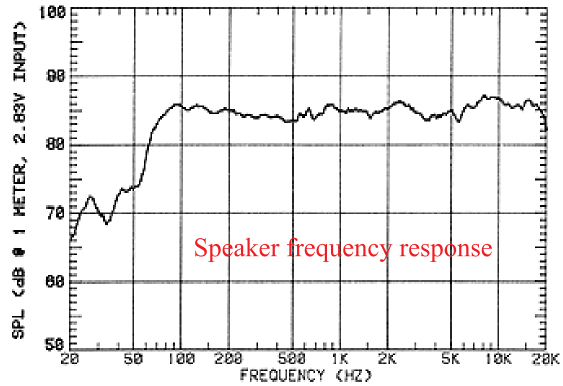

Figure TF24-4: Frequency response of a good consumer-quality speaker consisting of a tweeter and woofer.

The most common method for obtaining a nice flat frequency response is to drive several speakers together—each with a different but complementary frequency response. When listened to as a group, the frequency response is close to flat. For example, ***tweeters*** are small speakers intended for reproducing high-frequency sound, while ***woofers*** only reproduce the lowest frequencies. A common entry-level speaker consists of a tweeter, a mid-range speaker, and a woofer all housed together. With appropriate crossover circuits the ensemble can exhibit a good response.

Crossover circuits

As we noted earlier, most speakers cannot handle the entire range of frequencies in the audio range. In order to split the signal for use by the different speakers (such as a tweeter, a mid-range speaker, and a sub-woofer), passive filters are used. The signal is applied to a set of filters that produce three outputs: one output contains only low frequencies in some range, a second contains mid-range frequencies and a third output contains only high-frequency harmonics. In this way, each speaker receives a dedicated signal that contains only the frequencies it can reproduce properly. Designing crossovers can be an involved process that takes into account many variables, including the amount of current in the input signal, the input impedances of all of the speakers, and the frequency range of each speaker. Without careful design, the crossover circuit can provide too much signal power to one speaker and too little to another, thereby distorting the overall frequency response heard by the listener.