Name____ Period________

(Chapter 1) Sound Reinforcement System Basics

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1. Amplifying sound and delivering it pleasantly and effectively to the ears of an audience is known as _________________________________.
2. The term transducer refers to ____________________________________

____.

3. A Microphone changes sound waves (acoustical energy) into an _________________________________.
4. An amplifier makes an electrical copy of ______________________________

____.

5. The combined increase in signal strength is known as _____________.

6*. In what situation would you want an amplifier that does not increase the signal strength?__________________________________

7. What does a typical amplifier require to produce their output signal? ________

____.

8. Functions beyond reshaping, splitting and basic manipulation of an audio signal is known as ____.

9. When more than one microphone and/or other type of audio source is in use, an ______________________ allows combining of the signals into one or more outputs.

10. Equalizers might consist simply of ________________________________.

11. Speakers aimed to allow talkers or performers to monitor themselves are literally called _____________________.

12. Effective use of _____________ is considered by many audio engineers to be the single most important step in providing high quality sound reinforcement with good reason.

13. Use of ___________________ with the appropriate tonal-quality-related (frequency response) characteristics can greatly simplify the equalizing and mixing process, particularly with a music performance system.

14. A microphone's directional ______________________________can, when intelligently used, help to provide the best possible pickup of intended sound sources.

15. ____________________, used mainly for hallow-bodied acoustic instruments, are another commonly used type of ____.

16. Signals from _________________________(as build into electric bass guitars), and signals from electronic sources such as modern keyboard instruments and guitar processors, ordinarily would enter a sound reinforcing system via _________________________________, also known as direct injection.

17. Sources such as CD players, or audio signal from videotape or film sound track can be adapted to the input of any reinforcement system with the use of simple ________________________________ as described in chapter 16.

18. Draw and label your own basic system layout.

19. Mixers, for all but the most basic applications, normally allow ________________

____

____.

20. Additional outboard devices can also be added to automatically regulate the desired maximum and/or minimum signal levels, called _________________________

____

____.

21. Define/describe Effects loop- ____

____

22. A mixer may be designed to have ___________________________to allow signals to be readily sent to different destinations such as to the inputs of a multitrack tape recorder.

23. The term Equalization was originally _________________________________

____.

24. The term _______is now applied to any intentional alteration of frequency response.

25. An ____________ usually allows for control of from 2 to 4 tonal ranges within the entire hearing range.

26. The two types of EQ most commonly found on a mixer are ______________________ and ________________________ EQ.

27. The most common format for outboard Eqs is the ubiquitous _____________ EQ.

28. Practical applications for this type of finely divided EQ might be to compensate for irregularities in ____

____.

29. An amplifiers basic function, is to produce ______________________.

30. The 3 types of low level amplifiers that are known to exist are _____________

____.

31. Preamplifiers (preamps, for short), serve to ____________________________

____.

32. Power amplifiers serve to provide an output signal ______________________

____.

33. What is the final step in taking an electrical signal back into the acoustical realm? ____

34. Can one speaker element be consistent enough to control the directional pattern of sound in an effective way? ____

____.

35. For all but the most basic applications, the loudspeaker's task is divided among ____.

36. A frequency dividing network which is used to assign the frequencies to a particular component is called a what? ___________________________________

37. The task of a crossover is to ____

____

____.

38. The crossover serves to help protect speaker elements from being ___________

____.

39. The quality of ____

____.

40. The perception of __________________ often varies significantly from one individual to another.

CHAPTER 1

SOUND REINFORCEMENT SYSTEM BASICS

Amplifying sound and delivering it pleasantly and effectively to the ears of an audience—this simple objective known as sound reinforcement is in some ways the most basic task in all of the various audio fields, yet in other respects can be among the most difficult to accomplish in actual practice.

a) Fundamental System Concept.

First, a few of the very basics for the previously unfamiliar reader. Fig. 1.1 shows a diagram of the most basic sound reinforcement system possible, consisting simply of two transducers and an amplifier. The term ***transducer*** refers to any device which changes one kind of energy to another. A microphone will change sound waves (acoustical energy) into an equivalent audio signal (electrical energy); a speaker of-course changes an audio signal into sound waves. The playback head of a tape recorder or sensor of a CD player, among other devices, also fall into the category of transducers, providing an electrical audio signal which can be fairly easily adapted to an input of any sound reinforcement system.

An ***amplifier*** makes an electrical copy of an electrical signal (technically it need not be a stronger copy to be termed an amplifier). Certainly a much stronger signal is ultimately required to drive the speaker. This is accomplished in multiple stages, each of which involves one or more amplifiers at each component stage (in simple systems these may be integrated into one chassis). The combined increase in signal strength, known as ***gain***, can amount to as much as 1,000,000,000-or-more times the strength of the input signal.

In a sense, each component in a basic system can be thought of as making identical or approximate copies of the signals it receives. Transducers accomplish this through their particular method of transfer from one type of energy to another (such as from acoustical to electrical or vice-versa). Amplifiers produce their output signal with energy supplied by a separate power source, such as is drawn from a standard electrical outlet. The input signal is used only as a guide in producing the correct output signal.

Depending on the need, sound reinforcement systems are normally designed to deal with audio signals in a number of other ways beyond the level controls ("volume/gain/fader"). Once a sound is converted into an equivalent electrical signal, there are numerous possible ways in which it can be mixed, reshaped, split apart and otherwise manipulated. The various additional functions generally are known as ***signal processing***.

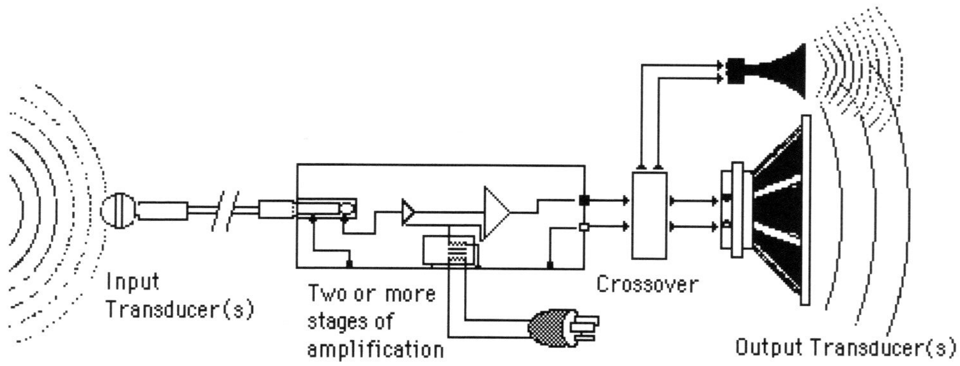


Figure 1.1

When more than one microphone and/or other type of audio source is in use, an audio mixer—as might be expected—serves to allow an input channel for each, and to effectively combine their signals into one or more combined output signals. A mixer normally will also allow for a number of other opportunities to process and reroute the signals it handles. ***Equalizers*** might consist simply of a couple of tone controls, or may be more finely divided to allow control over a number of narrower frequency ranges, from the deepest bass to the highest treble. A typical system will likely involve additional equalization through one or more separate (outboard) units, the use of effects devices to enhance the sound, and/or the creation of extra signal copies to be sent to additional speakers aimed to allow talkers or performers to monitor themselves—literally called ***monitors***.

Commonly more than one—and sometimes all—of the necessary stages of amplification, signal processing, and mixing functions are combined within one chassis; this type of unit is normally intended for relatively basic applications. Beyond this type of basic application, modern systems generally are made up of a series of interacting components which, within certain limits, can be interconnected on an as-needed basis.

b) Microphones and Other Input Transducers.

Microphones, in providing entry for a sound into the electrical realm of the sound system, play a very strategic role in determining the quality of the sound that will finally be delivered by the speakers. Effective use of microphones is considered by many audio engineers to be the single most important step in providing high quality sound reinforcement, with good reason.

Microphones are the first essential step in shaping the tonal quality of sounds as they enter the system. Use of microphones with appropriate tonal-quality-related (frequency response) characteristics can greatly simplify the equalizing and mixing process, particularly with a music performance system. ("Frequency response" is defined in Chapter 4(a). The term "frequency", along with other basic audio terms, is thoroughly explained in Chapter 2. Altering frequency response is, for example, what a simple tone control or equalizer accomplishes.)

A microphone's directional pickup pattern can, when intelligently used, help to provide the best possible pickup of intended sound sources (Fig. 1.3). By the same token, it can also aid in minimizing the pickup of unwanted sounds, including the output sound of the system itself (an excess of which is responsible for what is popularly known as "feedback").

Contact pickups, used mainly for hollow-bodied acoustic instruments, are another commonly used type of input transducer. Signals from ***magnetic pickups*** (as built into electric bass guitars), and signals from electronic sources such as modern keyboard instruments and guitar processors, ordinarily would enter a sound reinforcement system via ***direct input,*** also known as ***direct injection***. This procedure is further described in Chapters 13 and 16. Instruments such as electric guitars might be either miked or sent direct, depending on whether or not the guitar amp itself plays an important role in the desired sound quality.

Other audio sources might of-course include playback systems for recorded media, such as tape decks, CD players, or in special applications perhaps the audio signal from a videotape or film sound track. Normally such sources can be adapted to the input of any reinforcement system with the use of simple adaptors as described in Chapter 16.

c) Mixers and Related Accessory Units.

Beyond their basic function of mixing different input-signals to form combined output signals, most audio mixers are designed to perform a number of other signal processing functions.

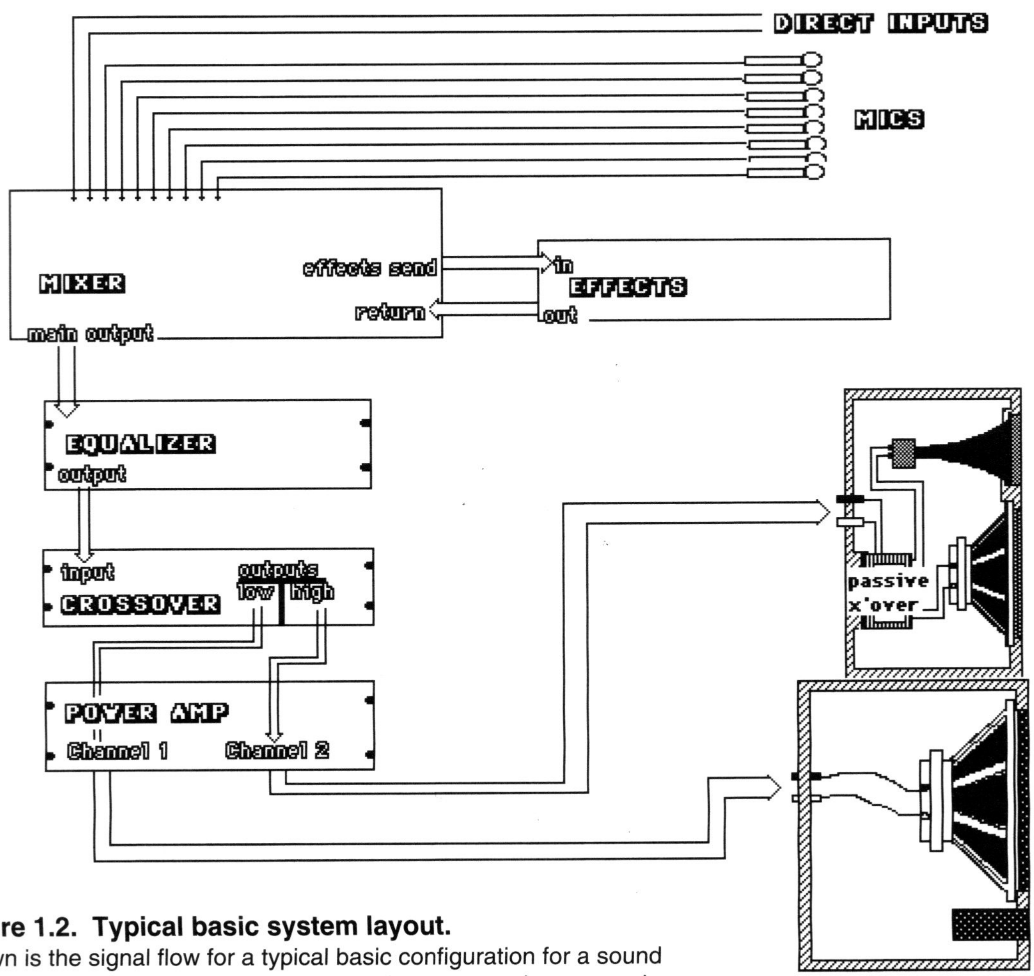


Figure 1.2. Typical basic system layout.
Shown is the signal flow for a typical basic configuration for a sound reinforcement system. Not included here is stage monitor setup. Larger systems essentially tend to be expanded versions of this basic type of layout. A larger system would be likely to include limiters and perhaps other signal processing devices as introduced in Chapter 8.

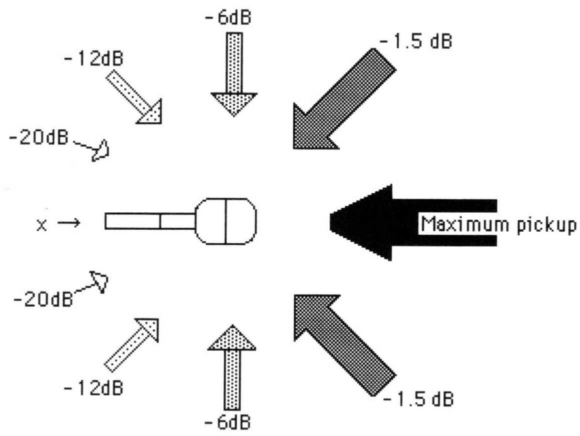


Figure 1.3. Representation of a basic microphone directional pattern as the mic "hears" it. Here, larger thicker arrows represent stronger pickup of sound coming from those directions. The pattern shown is known as a "cardi-oid" pattern, named after the heartlike shape of the pattern on the graphs shown in Chapter 5. A pattern of this type, also known as a "unidirectional" pattern, is useful in reducing pickup of unwanted sounds when properly used. (This pattern is of course three-dimensional, so rotating such a mic without changing its front-to-back orientation would not change this basic scheme. Incidentally, the abbreviation "mic" is used throughout this book—spell it "mike" if you prefer.)

Mixers, for all but the most basic applications, normally allow the signal strength to be adjusted at several stages, to make optimum use of the system's circuitry. Additional outboard devices can also be added to automatically regulate the desired maximum and/or minimum signal levels, called ***compressors***, ***limiters***, and ***gates*** (these are described in Chapter 8).

In addition to providing some kind of tone control (EQ) on each individual input channel (see Fig. 1.4), many mixer designs also provide on-board equalizers for the mixed output signals as well.

The vast majority of mixers allow signals to be split into separate electronic paths to allow the addition of any number of effects designed to enhance or modify the sound as well. Current technology has made an overwhelming selection of such devices available for both practical and creative use. The modified signal is then returned to an additional input to be included as part of the mix. (This type of supplementary signal route is usually called an ***effects loop*** or ***auxiliary loop***).

Separately adjustable signal copies may be sent to additional power amplifiers and in-turn to monitor speakers (traditionally called *foldback,* but today usually referred to simply as "stage monitoring". This is a function which all but the most basic mixers ordinarily provide. In high-level systems, separate mixers are often used strictly for the purpose of mixing the monitor sound.

Among other basic features, a mixer may be designed to have multiple outputs to allow signals to be readily sent to different destinations such as to the inputs of a multitrack tape recorder. ***Submasters***, if included, can allow the operator to divide input signals into categories at-will and control them in groups (these and other mixer functions are described in Chapter 7).

As indicated before, additional functions are commonly included within the same chassis. The extreme case of this is a mixer with graphic EQ and speaker-level outputs (such a unit is usually called a ***mixer/amplifier*** or ***powered mixer***). A number of currently manufactured units also include features such as built-in digital reverb.

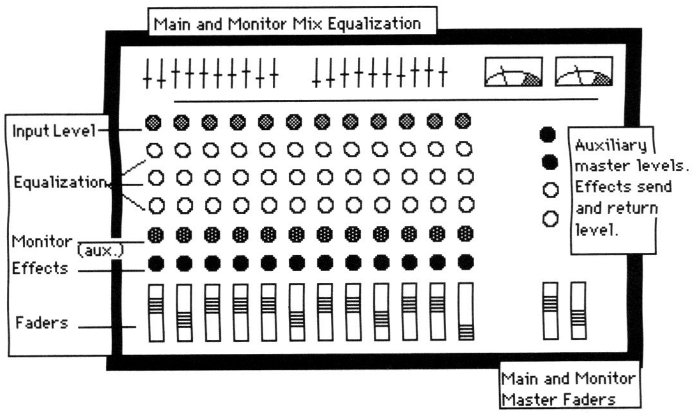
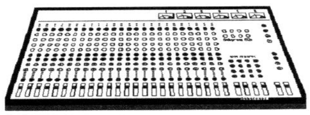


Fig. 1.4. Typical basic mixer layout. Mixers of this size (12 channels and under) and basic design (with graphic EQ) are marketed both with and without internal power amplifiers (speaker level outputs). An all-inclusive design of this kind can of course be quite handy for simple applications, particularly portable ones. Comparatively inexpensive and easy to operate, the compromise is a reduction in flexibility which would be afforded by using separate components for EQ, power amplification, crossover, limiting, etc.

(A mixer with this basic layout commonly uses both output faders and graphic EQ's for left and right in a "stereo" format, somewhat different than shown here. Essential concerns regarding these and larger mixers, as below, are described in Chapter 7 and throughout Part III.)



d) Equalizers.

The term ***equalization*** was originally applied to the process of electronically compensating for deficiencies in the ability of a component or system to accurately reproduce sounds or audio signals. Generally, the term (***EQ*** for short) is now applied to any intentional alteration of frequency response—including tone control—whether for practical or creative reasons.

As mentioned, a mixer normally provides the capability to equalize individual channels (this is called "on-board" EQ). An on-board EQ usually allows for control of from 2 to 4 tonal ranges within the entire hearing range. A three-band EQ, for example, would allow emphasis or de-emphasis of bass, midrange and treble frequencies. Commonly, mixers provide ***switchable*** EQ, which allow the operator to choose between two or more preset frequency ranges for a given knob, or ***sweepable*** to allow the operator to choose the affected frequency range on a much more gradual basis. The most versatile mixers may include fully ***parametric*** EQ, which allow control of the three important parameters (aspects) affecting this process. (These basic forms of EQ design are described in Chapter 6.)

A high-power or high-quality system for a large audience or other critical application normally needs to allow fairly precise control over many finely divided tonal ranges (bands). The most common format for outboard EQs is the ubiquitous ***graphic*** EQ, though other configurations are used as well. Currently manufactured equalizers can provide separate control over as many as 45 separate slices of the human hearing range (in the case of several obscure models many more such slices are provided).

Others allow the operator to "zero in" on the relatively precise audio frequencies necessary for a particular application. The practical applications of this type of finely divided EQ might be to compensate for irregularities in microphones, loudspeakers and room acoustics, to creatively alter a vocal or instrumental sound, or to help eliminate the obnoxious feedback squeals well known to every performer and public speaker.

e) Amplifiers.

An amplifier's basic function, as explained, is to produce a signal copy. Unlike microphones (in which different frequency response curves often are a clearcut advantage) and equalizers (which allow intentional changes of frequency response) the amplifier's task is to make an *accurate* signal copy—one with the least possible alteration of the form of the input signal. The signal's strength may increase, but ideally the essential form (it's "sound") should not.

Amplifiers are normally used in the design of systems and components at every important junction in their often myriad circuitry. Within components such as mixers and other signal processing units, low-level amplifiers serve to isolate circuits from one another, thereby allowing individual circuits to fulfill the various internal signal-processing functions. Amplifiers also serve to compensate for losses of signal strength within the circuitry. These are usually called ***line amplifiers*** or ***line drivers***. Each signal processing component has a line amplifier connected to the ouput jack(s), which generates the output signal that feeds the next component's input via the cable connecting the two. Among other types of low-level amplifiers also utilized are ***combining amplifiers***, which perform the actual mixing function within a mixer (see Fig. 1.5). An audio signal being generated into a component's input might, depending on the component's design, be processed through a ***differential amplifier***. Generally we need not be concerned with these, except to be aware that they exist, that they should be capable of producing a reasonably accurate signal of adequate strength, and that the inputs and outputs should be electronically compatible with other components in a given system. (The electronic compatibility is fairly easily managed in most cases, while the "adequate strength" occasionally will fall short, particularly with low-budget components.)

Preamplifiers ("pre-amps", for short), serve to boost a signal level prior to power amplification. Preamplifiers serve to produce a signal of sufficient strength to accomodate the input requirements of power amplifiers—that is, to "drive" them to the output level of which they are capable. Ordinarily, separate preamplifier components are used only for certain home-stereo applications. Normally, well-designed sound reinforcement components have line-level outputs sufficiently strong to eliminate the need for additional preamplification. We may, though, encounter the term on the access jacks of musical instrument amplifiers, in which case we can assume it refers to a line level. As well, the term can describe a unit designed to boost a very low-level mic output to a higher mic level or to a line level.

Power amplifiers serve to provide an output signal strong enough to drive the speakers. As mentioned before, several or all of the amplification tasks described in this section can be carried out within one chassis. The extreme case is a mixer with graphic equalization and speaker-level outputs, in which all of the basic electronic processes are accomplished by a single unit. In custom designed systems, the normal procedure is to use standard rack-mountable power amplifiers designed to accomplish only this final stage of amplification. (See also Chapter 4, section "o".)

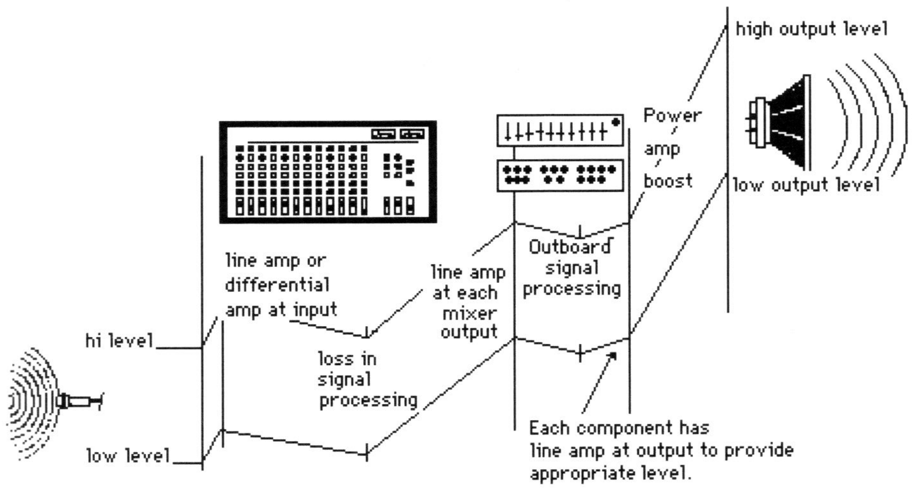
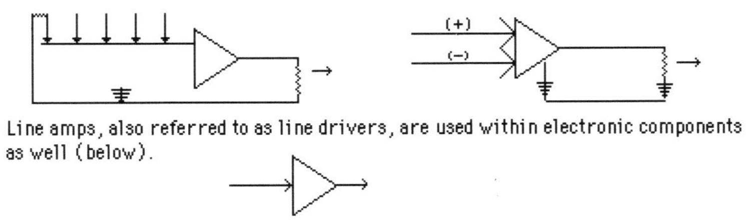


Fig. 1.5. Basic amplification stages in a sound reinforcement system. Energy loss is involved in both input and output transducers, as well as in each of the signal processing stages, which is compensated for by amplifiers—in addition to the expected increase in sound level provided at the speaker output stage.

Amplifiers also serve in other ways in the internal electronics of components. Below left: A combining amplifier provides an effective summing of individual signals. Below right: A differential amplifier is used at the input stage of many component designs. (Other designs use small input transformers followed by a line amp.) *Note: slight liberty has been used in schematic representations.*



f) Speakers.

Speaker components, in implementing the final step from the electrical realm back into the acoustical realm, are responsible not only for creating sound waves out of electrical signals, but also for directing the sound in a consistent manner that is appropriate to the application. Over the years, this has represented an immense challenge to the designers of loudspeakers.

At the most basic level, one full-range transducer may be capable of reproducing more or less the whole human hearing range (as in headphone speakers or many inexpensive home stereo speakers). But the behavior of sound does not allow one speaker-element to control the directional pattern in a consistent enough way to be effective for most sound reinforcement applications. (This type of limitation can be experienced with any low-budget home-stereo speaker simply by moving gradually from one side to the other and noting the changes in tonal quality as you move. Directly in front of such a speaker, the very highest treble is readily heard, while off to the side the lower tonal ranges tend to be much more predominant by comparison.) Also, and at least as importantly, at high sound levels it becomes physically impossible for one speaker element to effectively handle such a wide frequency range, from the very low bass to the very high treble.

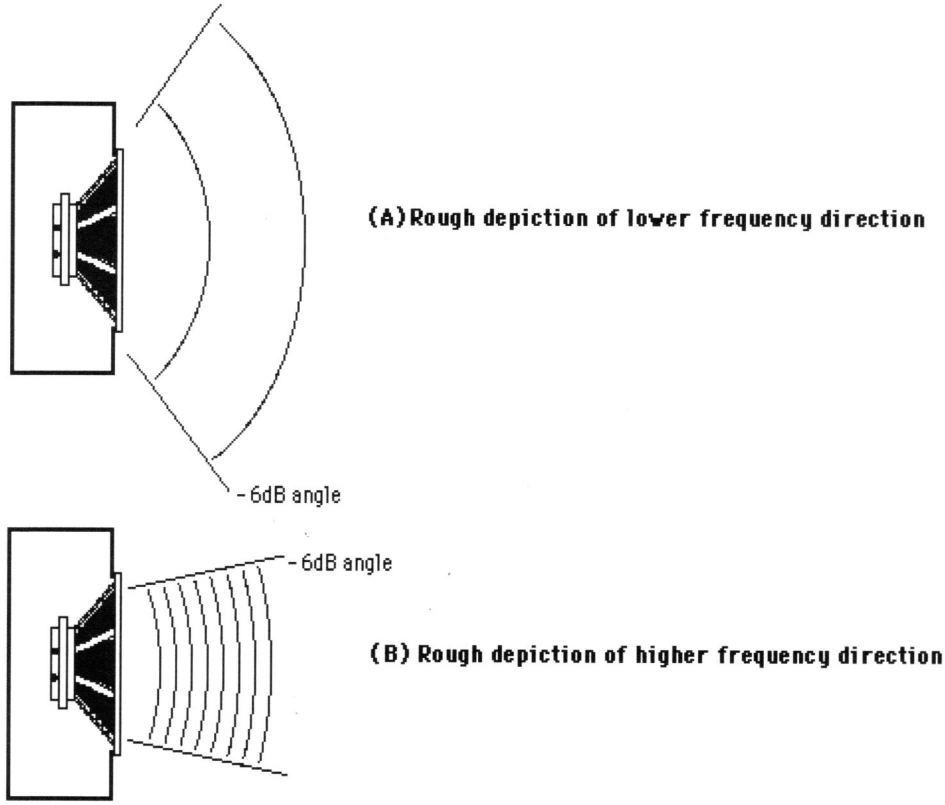


Fig. 1.6. A single speaker component of this type has massive variations of directivity, and is also unable to effectively reproduce wide tonal ranges at high levels without substantial distortion. These are why multiple components are used in modern sound reinforcment systems. The basic concerns involved in the use of multiple components are discussed in Chapter 9, and throughout Part III.

For all but the most basic applications, the loudspeaker's task is divided among two or more components, each of which is (ideally, at least) best suited for reproducing the frequencies in its intended range. Typical systems involve two, three or four—sometimes as many as five—frequency ranges, each handled by a different type of component. (More is possible but not at all necessary. Commonly, as in most home-stereo units, a multiple system is integrated into one cabinet, though for sound reinforcement applications the design of components often differs greatly from those in typical home-stereo-type applications.) Each basic design approach tends to have its own advantages and drawbacks, outlined in Chapter 9.

When two or more components are used in this way, the signal that powers each component needs to be confined to the band of frequencies for which that component is responsible. This is accomplished by a ***frequency dividing network***, commonly referred to as a ***crossover***.

g) Crossovers.

The task of the crossover is to divide its output into separate circuits, each covering a fairly specific ***band***, or frequency range. This allows each speaker-component to reproduce only the frequency range within which it operates best. As well, the crossover serves to help protect speaker elements from being damaged by operating outside the limits of their designated frequency range.

In systems designed for low-to-moderate-level use, crossovers can easily be installed after the power amplifier stage. When higher sound levels or finer operational control are required, a very significant increase in system efficiency can be achieved by dividing the audio spectrum before the

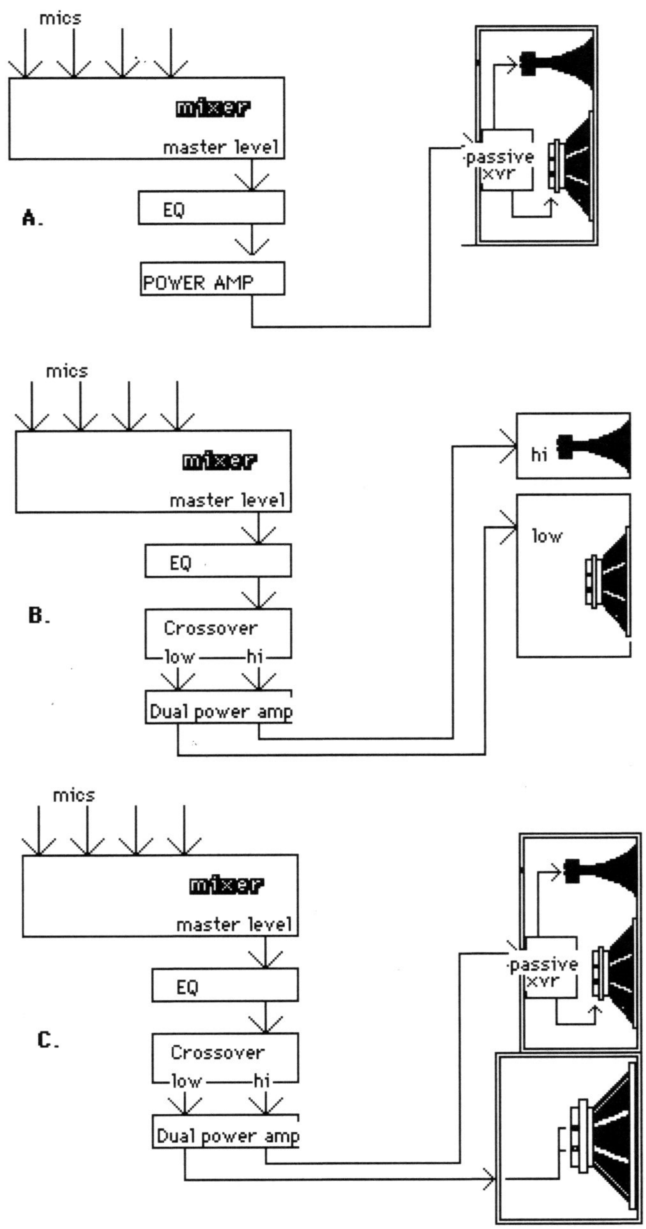


Figure 1.7
Typical crossover applications in a simple system.
(A) internal passive crossover, most often built into a standard multiple-component enclosure;

(B) simple "bi-amped" 2-way system with an active cross-over;

(C) standard two-way enclosure, with "sub-woofer".

The factors involved in this aspect of system use and design are discussed in Chapters 9,10, 11, 13, and 14.

power-amplifier stage. This type of unit is referred to as an ***active crossover***, or ***electronic crossover***; the former is called a ***passive crossover***.

Each has its advantages. Passive crossovers require a smaller number of power amplifiers, less wiring between components, and are generally more convenient for the user. Active crossovers, on the other hand, make more efficient use of total power amplifier output and more readily allow for accurate fine tuning of crossover points, and also can allow control of several other factors involved in dividing frequency ranges. Often, both are used very effectively in the same system, as illustrated in Fig. 1.7.

i) Practical System Concerns.

The overall effectiveness of a system certainly is no better than its weakest link. Beyond the quality of the components themselves, the quality of cables and methods of wiring play fundamental roles in a system's effectiveness. The manner in which the flow of audio signals passes from one stage to another within the system (and/or within a component) also plays a role both in audio quality and in practical aspects of operation. Chapters 5 through 10 overview sound reinforcement components. Chapters 11 through 16 overview system-related aspects of sound reinforcement.

j) The Acoustic Environment.

The acoustical realm is the first and last link in the chain from the sound source(s) to the listeners' ears. While this basic fact is obvious to most, the characteristics of the acoustic environment are often overlooked or misunderstood by many operators and inexperienced designers of sound systems.

Chapter 2 will attempt to introduce an initial perspective on sound in general. Later chapters will attempt to broaden this perspective.

k) Human Factors.

Sound reinforcement systems are oftentimes thought of simply as electronic (or electroacoustic) systems. But there is an often neglected human element involved at both ends of any such system. It is this human element—performers, public speakers and an audience of individuals—for which the system exists, and to which the system is ultimately accountable.

Among other things which are of important concern, the human ear itself behaves very differently than might be expected from looking at electronic measurements and meters. The perception of sound often varies significantly from one individual to another, changes according to the intensity of the sound, and also can change from moment-to-moment and place-to-place in a given arena in some interesting and sometimes almost bizarre ways. Certainly the potential for long term or permanent hearing loss is also of major concern with high-level systems. (Chapter 3 introduces the basics of human hearing and some of the ways it typically affects the process of sound reinforcement.)

In addition, practical considerations involving the various needs of performers and talkers come almost constantly into play. We will attempt to bring these concerns into perspective in later chapters.