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NAB™ BROADCASTERS

A BROADCAST ENGINEERING TUTORIAL for NON ENGINEERS

2nd Edition

Preface

Many times, people without engineering backgrounds need to have a general understanding of broadcast engineering issues. This is true for broadcast managers who come from sales, finance or programming backgrounds, for lawyers who work with broadcast clients, and for members of the financial community who deal with the broadcasting industry. It is also true for engineering trainees who have no engineering experience but who want to develop a knowledge base from which to launch a broadcast engineering career. This book is written for all of these people. It describes the engineering aspects of broadcast facilities in very general terms with the goal of providing non-engineers with enough knowledge about broadcast engineering to enhance the work they are doing in their respective fields.

In this second edition, new material has been added to explain digital television technology, and to further explain the digital audio formats that have rapidly become commonplace in radio broadcast studios.

We hope that the information in these pages will help to further their understanding of our trade, and thus enhance their ability to perform the broadcast-related functions of their jobs.

NAB Science and Technology Department
April, 1999

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Introduction

In its most general form, a broadcast station consists of two basic facilities: the studio complex and the transmitter. The studio complex is the place where the programming that is broadcast by the station originates. The transmitter is the device that actually broadcasts the programming material out over the air. In between the two is a hard-wired, or wireless, connection called the studio-to-transmitter link.

Part I of this book will cover the studio. It will describe the various pieces of equipment that are found in a typical broadcast studio, and it will explain how they work together. Part II will cover the studio-to-transmitter link, or STL. It will explain the different types of STLs and what the advantages and disadvantages are to using each one. Part III will cover the transmitter site — including the transmitter, transmission line and antenna. It will describe the modifications that the transmitter makes to the program material received from the studio in order to transport this material to receivers at distant locations. Part V gives a general overview of the Federal Communications Commission's technical criteria for allocating broadcast channels.

All three types of broadcast facility (AM, FM and TV) are covered in this book. When there is little technical difference between two facilities — as, for example, is the case with AM and FM radio studios — they will be covered together.

And now, on to Part I.

Part I: The Studio

Radio Stations

Many people may find it easiest to understand the operation of a radio station studio if they compare the studio setup to that of their home stereo. Generally speaking, the operation of a radio station studio is very similar to the operation of a typical home stereo — with the primary differences being 1) there is generally a lot more equipment in a studio setup than in a home stereo, and 2) the studio setup allows the program material from multiple inputs to be mixed together and then output as a combined signal, while a home stereo usually only permits a single input source to be sent to the speakers, headphones, recorder, etc. at any particular time.

The following is a list of some of the equipment that one is likely to find in a radio studio:

- √ Cart ("cartridge") players/recorders
- √ Cassette players/recorders
- √ CD players
- √ Computers
- √ Digital audio tape players/recorders
- √ Distribution amplifiers
- √ Headphones
- √ Microphones
- √ Mixing boards

- √ Reel-to-reel players/recorders
- √ Speakers
- √ Telephone hybrids

In many modern radio studios, analog equipment has been replaced by new digital equipment because the digital equipment is more reliable and generally permits more efficient use of a station's resources. As we review the technical characteristics of the various pieces of common studio equipment, we will start with analog equipment -- which was heavily used in the past and is still used in many studios today -- and lead into digital equipment -- which is in many studios today and will be in all studios of the future.

Analog Tape Players/Recorders

Cart, cassette and reel-to-reel players/recorders all have one major thing in common: they all use magnetic tape as the medium on which audio information is stored. Each of these devices has a different aspect that makes it particularly suitable for certain applications. A cart (short for "cartridge") machine, is especially useful for playing short "programs," such as commercials and songs. A portable cassette machine, because of its compact size, is particularly useful for recording audio in the field, such as news interviews. A reel-to-reel machine, because of its long lengths of easily accessible recording tape, is most useful for recording and playing back long programs, and for editing program material.

The type of tape used in a tape player/recorder varies from machine to machine. Cart machines have tape cartridges which contain a single loop of tape that is created by taking a piece of tape and connecting its ends together with adhesive tape. (The adhesive tape used to perform this function is called *splicing tape*, and the act of cutting and taping magnetic tape is called *splicing*.) The advantage to having the single loop of tape is that it never has to be rewound -- it always rotates in the same direction. When a recording is made on a cart, *cue tones* are placed on the tape by the recorder at the exact point on the tape just before the place where the program material is to be recorded. Cue tones are tones that are recorded on a separate part of the tape from the main audio information as illustrated in Figure 1.

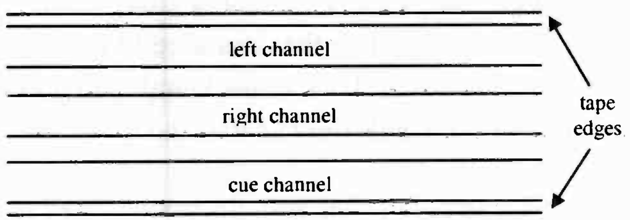


Figure 1: Cue Channel on Stereo Cart Tape

The cue tones are detected by the cart machine when it is playing back the tape, but they are not audible. During playback, when the cart machine hears these cue tones, it stops playing. Playback can then be restarted by pressing the "start" button. The great advantage to this system is that the disc jockey, or board operator, who is playing the commercial, or song, or whatever is on the tape, only has to worry about starting it. A button is pressed to start the tape and, once its audio has finished playing, it recycles itself all the way back to the beginning of the audio again and stops, ready to be played the next time it is needed. No stop or rewind buttons ever need to be pressed.

A slightly more advanced type of cart machine puts another cue tone on the recorded tape at the exact end of the recorded material. When this cue tone is detected during playback, it signals the cart machine to start the tape in another cart machine. In this manner, a group of cart machines can be hooked together and used to play a series of commercials, or songs, back to back with perfect timing. The human operator needs only to start the first cart.

Most people are very familiar with the operation of a cassette deck, so we will not spend too much time describing it here. We will, however, go over some of the details of noise reduction technology, which plays a particularly important role in cassette decks.

A cassette deck basically operates in the same manner as a reel-to-reel tape machine, with two important distinctions. The first distinction is that in a cassette system the two reels (the supply reel and the take-up reel) are encapsulated in a small plastic cassette. The second distinction is that cassette tape is narrower, and plays and records at a single, generally slower speed than reel-to-reel tape. (Reel-to-reel machines used in broadcast facilities usually permit the user to select from multiple tape speeds.)

The narrower tape in a cassette, and its generally slower speed, make cassette recordings generally noisier than reel-to-reel recordings. In order to combat this noise, manufacturers have:

1. developed tape coatings that increase the maximum level of the audio that can be stored on magnetic tape, thus increasing the dynamic range of recorded material;

2. introduced bias signals to the recording/playback process to overcome distortion at low signal levels; and

3. developed noise reduction (equalization) circuits -- the most widely recognized of which are the various Dolby® circuits.

<u>Tape coatings</u>

The type of magnetic coating used on a recording tape is important because, in general, the more magnetic the tape is the higher the maximum signal level that can be stored on it. Increasing the maximum signal level that can be stored on the tape allows audio material with a greater *dynamic range* (difference between the loudest and softest audio levels) to be stored on the tape.

Tapes that use coatings with chromium dioxide (CrO_2) as the magnetic material were the first big coating-related breakthrough in noise reduction technology. CrO_2 tapes have better high-frequency performance and lower noise than tapes with simple ferric-oxide coatings. Later, pure metal particles began being used to produce ground metal powders for coating tapes. This development enabled even greater signal levels to be stored on a tape without distortion, and further improved the dynamic range of recorded material.

Bias

The material on a recording tape is magnetic, and the tape head that transfers the audio material to the tape is a magnet. When the magnet (tape head) first applies its magnetic field to the tape, the magnetic particles on the tape are a little resistant to begin moving. Once they begin moving they move smoothly — but for a small fraction of a second when the magnetic field is first applied, particularly if the magnetic field is not very strong, their movement is a little rough and unpredictable. This poses a significant problem in recordings where the signal level being recorded is soft because the rough, unpredictable movement of the magnetic particles in the weak field results in a recording that sounds distorted to the human ear.

In order to overcome this weak signal distortion problem, a *bias* signal is added to the recorded material. This bias signal is an inaudible tone, typically at a frequency around 100 kHz which is way above the range of human hearing, and its purpose is to increase the strength of the magnetic field created by the recording head in order to insure that the magnetic particles on the tape will move smoothly and predictably, even when the audio being recorded is at a low level.

As one might imagine, the amount of bias required to insure that the particles on the tape will move smoothly and predictably varies from tape type to tape type. Generally speaking, Type I ("normal") tapes require the least amount of bias, Type II ("chrome") tapes require more bias, and Type IV ("metal") tapes require the most bias. (Type III was used to refer to tapes with dual-layer coatings, one chrome and one normal (ferric). These types of coatings are generally not used very much.)

Although the above generalizations regarding tape type and the amount of bias required are true, it is also true that the amount of bias required varies widely among tapes of the same type. For this reason, most tape decks include bias-adjusting circuitry. This circuitry is usually inside the tape deck and not user controllable, though some tape decks do provide external user controls. If too much bias is used, high frequencies (treble) will be somewhat muted and the recording will sound dull. If too little bias is used, high frequencies will be amplified and the recording will sound tinny.

Because the tape head only needs to alter the orientation of the magnetic particles on the tape when recording, selecting a bias setting is only necessary when recording. There is no need to select a bias setting during playback.

Equalization (EQ)

One of the inherent characteristics of the tape recording and playback process is that, when a tape is played back, the audio at the lowest and highest frequencies will not be as loud as it was in the original material. To correct this problem, equalization is employed. In essence, the highest frequencies are amplified during the recording process so that they end up being recorded on the tape at a level that is higher than their "natural" level. Then, during the playback process, these same frequencies are suppressed, but to a lesser degree than they were originally amplified. This way, when the normal reduction in the higher frequencies occurs during the playback process, the end result is an audio signal that sounds like the original material. The lower frequencies are not given any special treatment during the recording process, but they are amplified during the playback process.

To illustrate this concept, let's imagine a hypothetical signal using an arbitrary signal strength scale of 0-5, with 0 being the softest audio and 5 being the loudest. If, in the original material, the level of the lowest and highest frequencies is 3, then without equalization they will be played back at a level of 2 (see Figure 2). In order to compensate for this loss of 1 unit of signal level during the playback process, the highest frequencies are amplified during the recording process to a level of 5. Then, during the playback process, they are suppressed to a level of 4 which, when accompanied by the inherent loss of 1 unit of signal level in the playback process, results in a played back signal level of 3. The level of the lowest frequencies is simply amplified during playback to restore them to their natural level of 3. The recorded and played back signal levels in a system using equalization are illustrated in Figure 3.

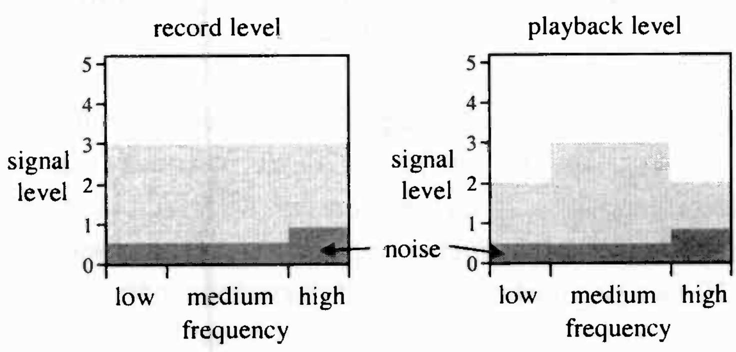


Figure 2: Example of Recording Process with No Equalization

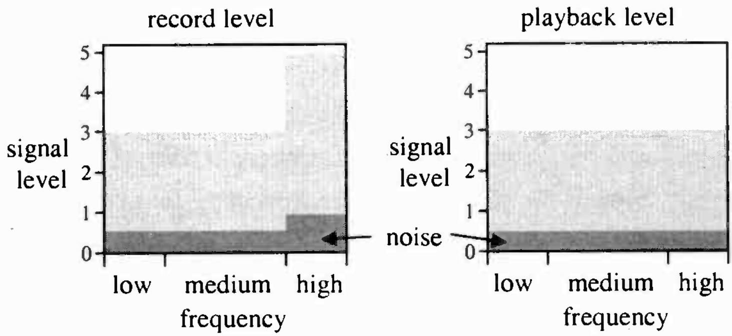


Figure 3: Example of Recording Process with Equalization

In practice, the amount of equalization used is specified by a time constant of either 70 or 120 microseconds (μs). Recording with a 70 μs time constant will result in more amplification of the higher frequencies, and playback using a 70 μs time constant will result in greater suppression of the higher frequencies. In essence, the smaller time constant means that the equalization circuitry reacts to the higher frequencies faster, resulting in a greater degree of equalization.

The reason that the higher frequencies are amplified during recording and the lower frequencies are not has to do with tape hiss. The hiss often heard on a tape recording consists of higher frequencies. By amplifying the higher frequencies in the audio before they are recorded to tape, the difference in signal level between the recorded material and the hiss is increased. This way, when the level of the higher frequencies is reduced to some degree during playback, the level of the hiss will be reduced also. The difference between the audio signal level and the hiss, or other noise on the tape, is called the *signal-to-noise ratio.* The larger the signal-to-noise (S/N) ratio, the better the recording sounds. A low S/N ratio will result in a recording that sound "hissy."

Dolby® noise reduction technology is a sophisticated form of equalization. Dolby A uses amplification during recording and suppression during playback in the manner described above *except* that Dolby A technology operates over the entire audio range — not just the lowest and highest frequencies. Dolby A was originally developed for the professional recording industry. Dolby B circuitry is a less complex — and therefore less expensive — version of Dolby A. It operates primarily at higher frequencies. Dolby C is an enhanced version of Dolby B which covers more frequencies and uses a larger signal boost during recording. The larger signal boost during recording means that there is more signal suppression during playback, resulting in a greater reduction in the level of extraneous noise.

Well, that concludes a rather thorough overview of noise reduction technology and how it relates to analog tape recording. Let's continue on now with the third and final type of analog tape machine found in many broadcast facilities — the reel-to-reel.

As mentioned earlier, a reel-to-reel machine basically operates in the same manner as a cassette player/recorder, except that it uses wider (¼-inch versus ⅛-inch) tape which can move at different user-selectable speeds, and the two reels are not encapsulated in a plastic cassette case as they are in a cassette system. The wider tape, and the ability to move the tape at faster speeds, make reel-to-reel recordings less noisy than cassette recordings. The other major benefit of a reel-to-reel system is that its easily accessible tape enables smooth editing of program material through the use of splicing. If, for example, an interview has been recorded on reel-to-reel tape, and parts of it need to be cut out due to time constraints, or appropriateness of content, then the part of the tape which is to be left out of the final product can simply be cut out, and the remaining portions taped together with splicing tape. This process is not possible (or, at least certainly not practical) when cassettes or carts are being used.

Digital Audio Tape Players/Recorders

Digital audio tape (DAT) players/recorders are sort of a cross between analog tape equipment and compact disc players. DAT equipment offers significant advantages over analog equipment because its underlying digital technology enables it to record and play back audio that is not as noisy as audio recorded on analog equipment. From a playback perspective, DAT equipment is not quite as desirable as compact disc equipment because DAT tapes are subject to wear and tear and will eventually wear out whereas compact discs will never wear out, if they are properly cared for. However, from a recording perspective, DAT equipment has a tremendous advantage over compact disc equipment because it is much less expensive to make a DAT recording than it is to make a CD recording for the one-recording-at-a-time purposes of the typical broadcaster — and many recordable CDs cannot be used for re-recording, whereas DAT tape can be erased and recorded over just like analog tape.

The reason that DAT tape's digital technology makes it less noisy than analog tape is that the digital coding on a DAT tape makes extraneous noise on the DAT tape virtually invisible to the DAT equipment. Figure 4 provides an example of why this happens. Basically, as long as the amount of noise on the tape is not so high that it prevents the DAT player from distinguishing between high and low signal levels (ones and zeros), the signal read off the tape will be a series of ones and zeros. As shown in Figure 4(a) and (b), the digital audio signal read from the tape will be the same series of ones and zeroes even if the amount of noise on the tape increases, as long as the noise level does not increase to the point where the player cannot accurately determine whether a symbol is a one or a zero (Figure 4(c)). With an analog tape, the sequential noise level increases illustrated in Figure 4 would each further degrade the audio that is reproduced by the player — a drawback which is, for the most part, overcome by the digital coding.

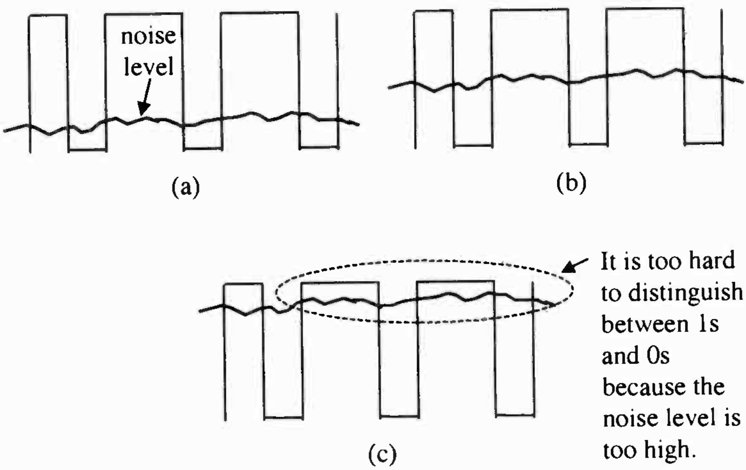


Figure 4: How a Digital Signal Relates to Noise

While digital audio tape offers improved audio performance over analog tape, it is still subject to the same wear and tear that plagues analog tape. This includes such things as having the tape machine "eat" the tape due to malfunctions with the tape turning mechanics in the machine, or problems with the tape cassette itself. It also includes stretching of the tape, which happens over time and generally more often with longer tapes (*i.e.*, ones that have longer playing times)

Compact Disc Players

Compact discs (CDs) are, to date, the most reliable media for storing digital information. The reason is simple — on a compact disc, the digital information is permanently etched, or carved, into the plastic that makes up the CD. It cannot be erased by passing through a magnetic field like the information on a recording tape can, and really the only way to damage the information on a CD is to physically damage the CD itself by breaking it or severely scratching it. Small scratches on a CD are often not a problem for most modern CD players used in broadcast facilities because the players are able miss a few 1s and 0s here and there in the digitally recorded audio and still accurately reconstruct the recorded music. They are able to do this because the digital data on the CD actually contains more digital bits than are necessary to encode the audio information. These additional bits are added, in a specially coded manner, to enable the CD player to accurately determine what the correct value of a missing or damaged piece of digital data is. This system of adding these additional bits is called an *error correction system.* An error correction system is only capable of fixing errors in the data up to a certain point. If there are too many missing or damaged pieces of data, even the error correction system will fail and the CD will skip or stop playing.

The other thing that helps to make a CD so durable is the fact that it is not subject to any wear and tear during the playback process. A CD player reads information off of a CD by shining a light (a laser) on it and analyzing the reflections of this light that are caused by the CD. Because there is never any mechanical contact between the laser and the CD, there is no wear and tear on the CD. A magnetic recording tape, on the other hand, is subject to a lot of wear and tear because during both playback and recording it is being dragged over the tape head.

Some CD players used in broadcast facilities have both analog and digital outputs. If the station's audio system is analog-based, then the analog outputs can be used to feed a signal into the mixing board. If, on the other hand, the station's audio system is digitally-based, then the digital outputs can be used to feed information into the mixing board. In a digitally-based audio system, digital outputs from a CD player are generally more desirable because they allow the station to avoid installing an analog-to-digital (A/D) converter between the CD player output and the mixing board input. This is advantageous because, in general, every time an audio signal has to go through a conversion process it is degraded to some small degree. So, it is desirable to keep the number of conversions to a minimum.

The Mixing Board

The heart of a radio studio — the thing that allows several program sources to be fed simultaneously to the transmitter — is the mixing board, or console. A basic mixing board is simply a device that has multiple signals being fed into it from different program sources (such as a microphone, a CD player, and a tape player). The mixing board allows its operator to combine (mix) the signals from the various inputs to produce a single output signal that is a combination of the various input signals. Figure 5 illustrates the basics of mixing board operation.

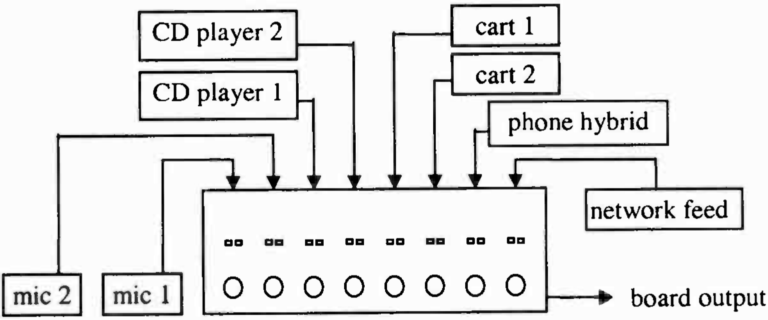


Figure 5: Illustration of a Basic Mixing Board Setup

The mixing board functions in a manner that is very similar to a home stereo system. In a home stereo, various program sources (such as a cassette deck, a CD player, and a turntable) are connected to a single amplifier. The user must then select which one of the sources to amplify at any given time — a selection which is often made by choosing a single button to press from a series of buttons on the front panel of the amplifier. A mixing board also connects several input sources to a single amplifier. The big difference between a mixing board and a home stereo, however, is that the mixing board allows the user to select *multiple* input sources (*simultaneously*) — a selection which is usually made by choosing one, or more, buttons to press from a series of buttons on the front panel of the mixing board.

To illustrate the significance that the ability to mix multiple inputs plays in producing an on-air radio program, consider the sequence of events that occurs when a radio announcer introduces a song. The announcer talks into the microphone to describe the song that is about to be played. While the announcer is talking, the select switch on the mixing board for the microphone input is selected, and the microphone is "potted up." (The phrase "pot up" is derived from the name of the electronic device used to control the level of the selected signal in many mixing boards — a *potentiometer*, or variable resistor.) At the same time, the select switch for the device that will play the song (*e.g.*, a CD player) is also selected and potted up, though no audio is coming from the CD player because it has not yet been started. When the announcer is finished introducing the song, the start button for the CD player is pressed and the music begins playing. At this point, the select switch for the microphone is deselected, or turned off, and the mixing board is no longer mixing any signals — it is simply amplifying a single signal (the CD player).

The above is just one example of how a mixing board is used to produce an on-air broadcast signal. There are many others using all different kinds of input sources, and even other scenarios involving the two input sources described above. For example, many radio stations believe it sounds better to the listener when the music from the CD player is actually started *before* the announcer has finished introducing the song. This helps to insure that there is absolutely no silence, or "dead air," between the announcer's introduction and the actual start of the song.

The importance of the mixing board becomes apparent when one considers what it would be like to introduce a song, and begin playing the song, using a device like a home stereo system that allows only one input to be selected at a time. Using such a device, the announcer would have to select the microphone, introduce the song, then *simultaneously* deselect the microphone, select the CD player and start the CD playing. Such a system would certainly result in an on-air signal that sounds choppy and unprofessional with lots of "clicks" and "pops."

Well, by now you should be comfortable with the image of a mixing board as an extra fancy amplifier like the ones used in many home stereo systems. The outputs of the various audio-generating devices (CD players, microphones, tape players, etc.) are connected to the inputs of the mixing board, and the output of the mixing board is sent to the studio monitors (speakers) and off to the transmitter for broadcast.

Telephone Hybrids

A telephone hybrid is a piece of equipment that converts incoming audio from a telephone line into a "line level" signal that can be fed into a mixing board, tape recorder, etc. It also converts a "line level" signal coming out of a mixing board into an audio signal that can be fed over the phone line.

Telephone hybrids are essential pieces of equipment for stations that do a lot of on-air talking to people who have called in. The hybrid allows the DJ or talk show host to hear the caller through the mixing board without having to pick up a telephone handset, and it allows the caller to hear the DJ or talk show host speaking through the microphone connected to the mixing board. By using the hybrid, the broadcaster ensures that only the caller's voice is of "telephone quality," while the DJ or talk show host's voice remains of "broadcast quality."

Microphones, Headphones and Speakers

Microphones, headphones and speakers will all be covered together because they all perform very similar functions. Microphones convert sound waves created by human voices, instruments, or other things, into electrical signals which can be fed into a mixing board, or another electronic device. Headphones and speakers take electrical signals and convert them into sound waves which can be heard by the human ear.

The electrical signal produced by a microphone is of a very low level, and it needs to be fed into a microphone preamplifier before it is mixed with other studio audio signals. In most cases, the microphone preamplifier is included inside the mixing board, so no additional equipment is needed. Care must be taken to insure that only microphones are connected to the microphone input on a mixing board. Connecting a device with a high output signal, such as a CD player, to the microphone input on a mixing board will overload the mixing board input and might cause damage.

While each different model of broadcast microphone is designed a little bit differently, they all have generally similar design principles. All microphones have a surface that, when impacted by a sound wave, causes a corresponding change in the properties of an electrical circuit. To illustrate the design principles that apply to broadcast microphones, let's consider the designs of three different microphones that are commonly found in broadcast use — the dynamic moving coil, the ribbon and the condenser.

In the dynamic moving coil microphone a drum-like surface called a "diaphragm," is impacted by the incoming sound waves and it moves up and down in a corresponding manner. The back of the diaphragm is connected to a metal coil which slides up and down over a magnet. This sliding of the coil over the magnet causes an electrical signal to be created in the coil. This electrical signal is a reproduction, in electrical form, of the sound waves that hit the diaphragm. The ends of this coil are connected to the plug on the end of the microphone and can be fed from there into a mixing board.

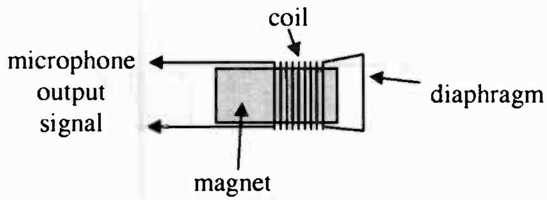


Figure 6: Dynamic Moving Coil Microphone

The ribbon microphone operates using essentially the same principle of the dynamic moving coil microphone — that an electrical signal will be produced in a wire which is moving through a magnetic field. In the ribbon microphone, a very thin piece of metal foil (the ribbon) is suspended in a magnetic field in such a manner that incoming sound waves impact the ribbon and cause it to move back and forth in the magnetic field. This movement of the ribbon within the magnetic field causes an electrical signal to be created in the ribbon which is an electrical reproduction of the sound waves that hit the ribbon. The ends of the ribbon are connected to the plug on the end of the microphone and can be fed from there into a mixing board.

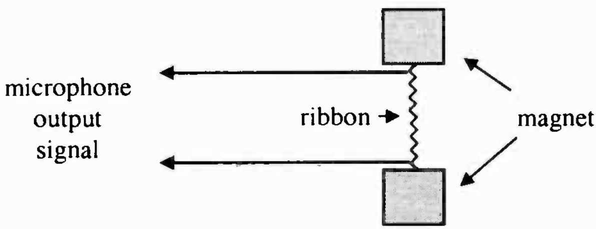


Figure 7: Ribbon Microphone Design

The condenser microphone operates using a different principle than a dynamic moving coil or ribbon microphone. The operation of the condenser microphone is based on the operation of a *capacitor*. A capacitor is an electronic device with two leads which allows electricity to flow from one lead to the other at a varying rate, depending on how easily the material between the two leads allows electricity to pass. In the condenser microphone, incoming sound waves strike a diaphragm which is situated in front of a metal plate called the "back plate." Together, the diaphragm and the back plate form a capacitor. The ability of the material between them (air) to allow electricity to pass is dependent on how far apart they are. So, if electricity is applied to the circuit in a condenser microphone, the flow of this electricity will vary in proportion to the capacitance of the capacitor, which itself will vary in accordance with the sound waves hitting the diaphragm. In this manner, an electrical signal is produced at the microphone output which is an electronic version of the incoming sound waves that are hitting the diaphragm. The main advantage of the condenser microphone is that the capacitor circuit is much smaller and lighter than the magnets used in the dynamic moving coil and ribbon microphones. For this reason, lapel, or clip-on microphones are typically of the condenser type.

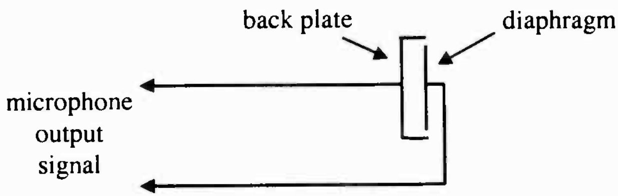


Figure 8: Condenser Microphone Design

A speaker, as one would imagine, operates in basically the exact opposite manner of a microphone. In a speaker, an electrical signal (of a much higher level than the one that comes out of a microphone) is fed into a metal coil located in a magnetic field. This metal coil is attached to a lightweight surface called the — yes, you guessed it — diaphragm. The changing electrical signal in the coil causes it to move back and forth in the magnetic field and, because the coil is attached to the diaphragm, this causes the diaphragm to move back and forth too. It is the diaphragm's movement against the outside air that creates the sound waves which can be heard by the human ear. These soundwaves, of course, correspond to the electrical signal that is fed to the speaker through the speaker wire.

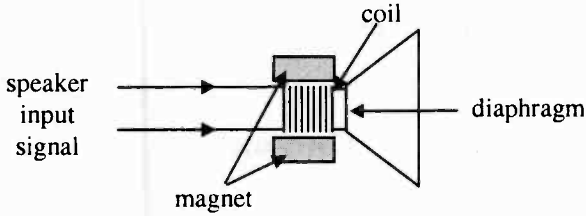


Figure 9: Typical Speaker Design

Headphones operate in a manner that is essentially the same as the manner in which speakers operate, the main difference being that the electrical signal levels fed into headphones are not as strong as those that are fed into speakers and, of course, the physical elements of a headphone speaker are generally smaller than those of a loudspeaker.

The unique thing about headphones, speakers and microphones is that, no matter how many revolutionary changes occur in broadcast equipment design, these devices will always operate in an essentially similar manner. While it may be possible to convert all of the other audio signals running around a broadcast facility to digital pulses — headphones, speakers and microphones will have to retain their analog design because, at least as far as the evolutionary eye can see, human voice and human hearing will continue to be in analog form.

Computers

Well, speaking of converting the audio signals in a broadcast facility to digital pulses, this is certainly the trend in modern radio facilities. Nowadays, many radio stations have most of their prerecorded music, commercials, etc. stored on computer hard drives. The continually increasing size of these hard drives, and their continually decreasing cost (on a per megabyte basis), have made this possible.

There are many advantages to converting a radio station's studio facilities to digital technology. One such advantage is that digital recording material can generally overcome noise in the audio path better than analog recording material. Another advantage is that hard disk-based digital recordings are easier to automate than analog tape recordings because disk-based systems can be run by a single computer program on a single machine, whereas tape-based systems require the coordinated operation of multiple machines through the use of cue tones. Yet another advantage is that a computer system is subject to less mechanical wear and tear than a tape-based system, so it is more likely to have longer periods of time between mechanical malfunctions than a tape-based system. Also, disk-based systems make log keeping much easier because the computer that controls the system knows when it has played a song, or a commercial, or whatever, and it can automatically create and print its own program log.

When a radio station uses a disk-based audio system, there are still two places where audio material must remain analog. The first, as mentioned above, is at all of the microphone inputs and speaker/headphone outputs. Human voice and human hearing are still analog and therefore require analog mics and analog speakers and headphones. The second is at the output of the transmitter. All of the broadcast radio receivers that listeners are using today are designed to receive analog radio (AM or FM) transmissions. So, the final signal that comes out of the transmitter must still be analog. Using equipment that is on the market today, it is possible to have a DJ's voice converted to digital immediately after leaving the microphone and have it remain in digital form until after it has been fed into the transmitter, which then produces an analog output signal based on the digital input.

Some radio stations have converted partly to computer — they may have all of their commercials stored on a computer, but still receive an analog satellite feed. In these situations they will need to use a mixing board that is equipped with both analog and digital inputs. There are several such mixing boards on the market today. The standard format for the digital input signals on these boards is usually the *AES/EBU* digital format. "AES/EBU refers to a standard format of digital bit transmission adopted by the Audio Engineering Society and the European Broadcasting Union.

Digital Audio Basics

There are three basic concepts that one needs to understand in order to have a good basic understanding of digital audio. These are resolution, sampling rate and bit rate.

The *resolution* of digital audio is the precision with which the digital signal, at any particular instant in time, matches the original analog material from which it was created. Resolution, like many aspects of digital systems, is measured in bits. The higher the number of bits (and thus the resolution), the more accurately the digital signal represents the original analog material. For example, 16-bit audio more precisely replicates original analog material than does 8-bit audio.

One of the keys to understanding digital resolution is understanding the relationship between the number of bits of data in each digital sample and the amount of resolution that each sample has. On the surface it might appear that 16-bit digital resolution is twice as good as 8-bit resolution. This is not the case, however. In reality, 16-bit resolution is *256 times* as good as 8-bit resolution.

To understand why this is so, let's consider an example. Let's imagine that we have a thermometer that can read temperatures in the range 0°-127°. If we only have one digital bit to represent the reading from the thermometer – that is, one bit of digital resolution – then a logical way to digitally code the temperature from the thermometer would be to say that the digital bit is a zero whenever the temperature is below 64° and it is one whenever the temperature is at or above 64°. Clearly, this is not a very accurate representation of the actual temperature reading from the thermometer.

If we have two digital bits to represent the reading from the thermometer then we could assign a specific digital bit combination to four different temperature ranges. The bit combination '00' could represent temperatures below 32°. The bit combination '01' could represent temperatures from 32° to 63°. The bit combination '10' could represent temperatures from 64° to 95°. And, the bit combination '11' could represent temperatures above 95°. Note that going from one bit of digital resolution to two bits of digital resolution doubled the number of temperature ranges that could be represented digitally, and thus doubled the accuracy of the digital representation of the temperature reading.

If we were to add yet another bit of digital resolution to this system then temperature ranges could be represented digitally as follows:

Digital Bit Combination	Temperature Range
000	0°-15°
001	16°-31°
010	32°-47°
011	48°-63°
100	64°-79°
101	80°-95°
110	96°-111°
111	112°-127°

Going from two bits of digital resolution to three bits doubled the accuracy of the digital representation of the temperature reading once again.

Clearly, there is a pattern here. Each time a single bit is added to the digital representation of the temperature reading the accuracy with which the digital representation depicts the actual temperature doubles. This makes perfect sense, when you think about it, because each digital bit has only two possible values – 0 and 1. So, when a single bit of digital resolution is added to a system all of the previous digital codes can still be used – let's say they represent the same things they did before the new bit was added but now they represent them when the new bit is '0' – and an entire new set of digital codes becomes available that is equal in size to the one that existed before the new bit was added – in this example all of these new codes would be the ones that existed before the new bit was added but now with the new bit included and set to the value of '1.'

It should now be clear why 16-bit digital audio represents the original analog material with 256 times more accuracy than 8-bit digital audio. Following the pattern we just discussed, 9-bit digital audio would be twice as accurate as 8-bit audio, and 10-bit audio would be twice as accurate as 9-bit audio. Continuing all the way up to 16-bit audio we would find that the accuracy of 16-bit audio is equal to the accuracy of 8-bit audio x 2 x 2 x 2 x 2 x 2 x 2 x 2 x 2, which is another way of saying the accuracy of 16-bit audio is equal to the accuracy of 8-bit audio times 256.

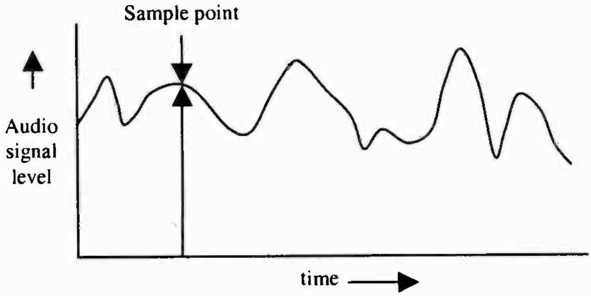


Figure 10: A Digital Sample of an Analog Audio Signal

Whether the resolution of the digital sample is 8-bit, 16-bit or whatever, each individual sample represents the level of the audio signal at a particular instant in time. Sampling an audio signal is a lot like sampling the thermometer in the example we just discussed above. Probably the biggest difference between sampling an audio signal and sampling a temperature reading is that the audio signal changes value much more rapidly. For this reason, the audio signal must be sampled much more frequently than the thermometer in order to provide an accurate digital representation of the original information.

Sampling Rate

The *sampling rate* is the rate at which digital samples are made of the original material. The more often the original material is sampled, the more accurately the digital reproduction represents the original material.

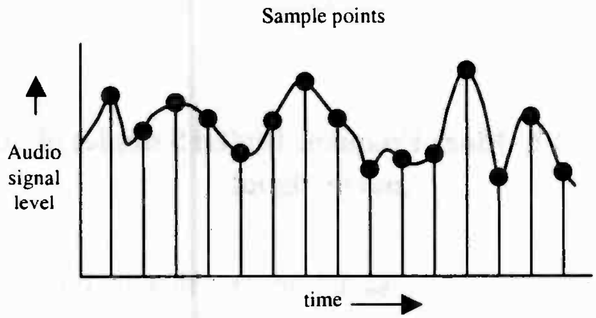


Figure 11: Periodic Digital Samples of an Analog Signal

Figure 11 shows an analog signal being sampled at some regular interval. Figure 12 shows the same analog signal being sampled twice as often. As can be seen by comparing these two figures, the more often a signal is digitally sampled, the closer the series of resulting sample points represents the original signal.

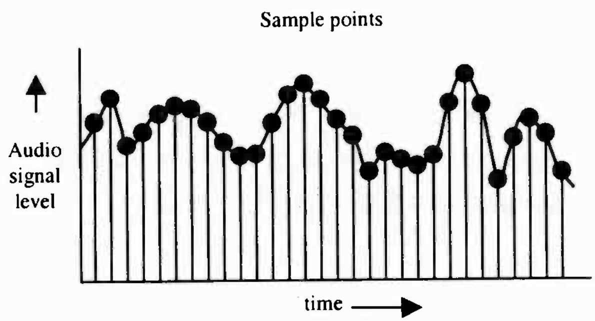


Figure 12: More Frequent Digital Samples of an Analog Signal

There are three common sampling rates that are often used for digital audio. These are 32,000 samples per second, 44,100 samples per second, and 48,000 samples per second. Usually these sampling rates are referred to simply as 32 kHz, 44.1 kHz and 48 kHz, respectively. Digital compact discs have a digital sampling rate of 44.1 kHz.

Bit Rate

The bit rate necessary to transport a digital audio signal is directly related to the digital resolution of the digital audio, and its sampling rate. Using the digital resolution and the sampling rate for compact discs, for example, we can calculate the bit rate necessary to transport CD audio.

	CD digital resolution:	16 bits/sample/channel
x	CD sampling rate:	44,100 samples/second
	CD bit rate per channel:	705,600 bits/second/channel
x	2 stereo channels:	2
	Total CD bit rate:	1,411,200 bits/second

There are eight bits in each byte of data on a computer disk. So, in order to store one second of compact disc stereo audio on a computer disk 1,411,200 ÷ 8 = 176,400 bytes of disk space is required. A typical three minute long song would require 176,400 bytes x 180 seconds = 31.752 megabytes of disk space.

Compression

In order to conserve disk space, and also to make it possible to send digital audio signals through channels that are not capable of carrying all 1,411,200 bits per second from a CD, a technique called *compression* is used. In order to compress a digital audio signal some of the digital bits in the audio signal are discarded, and the remaining bits can be encoded in a manner that reduces the total number of bits needed to transmit the audio.

The reason some bits can be discarded when compressing a digital audio signal is that the audio they represent cannot actually be heard by the typical listener. For example, if a very loud tone is accompanied by a very quiet tone on a slightly different audio frequency, in most cases the human hearing system will not even recognize the existence of the quiet tone. Therefore, the digital bits used to represent the quieter tone can be discarded without perceptibly altering the audio.

After all of the bits representing audio that generally cannot be heard have been discarded, special digital coding techniques can be used to further reduce the bit rate. Because there are just about as many digital audio compression systems as there are companies that make digital audio equipment, there are many different ways that coding techniques are used to reduce the data rate necessary to transmit digital audio. It is beyond the scope of this book to discuss all of these, but one general example of how coding can be used to reduce bit rate will give you an idea of how this is possible.

Let's say that the numerical values associated with individual digital sample points in a segment of audio are:

5, 12, 7, 9, 5, 12, 7, 9, 5, 12, 7, 9, 5, 12, 7, 9, 5, 12, 7, 9

It is possible to represent this series of values by simply transmitting each individual value, and in fact this is how a compact disc system works. It is also possible, however, to simply transmit 5, 12, 7, 9 followed by the instruction "repeat four more times." In this manner, the amount of data necessary to transmit a long series of repetitious digital values can be reduced.

The AES/EBU Digital Format

As mentioned earlier, the AES/EBU format is a standardized format for transporting digital audio information from place to place in a broadcast studio. It is the most common standard used for this purpose in the radio broadcasting industry.

Basically, in order to get digital audio information from one place to another in a radio station studio, a stream of digital bits must be carried – usually through a cable – from the originating point to the receiving point. In order for the device receiving the bits to understand which ones belong where, a standardized format for transporting the bits must be defined. This is what AES/EBU does.

In the AES/EBU format the stream of digital bits is organized into 64-bit long segments called *frames*. Each of these frames is further broken down into two sub-frames. Sub-frame 1 carries the digital audio information for audio channel 1, and sub-frame 2 carries the digital audio information for audio channel 2. In the vast majority of radio stations broadcasting music the two sub-frames correspond to the left and right channel of the stereo audio. The AES/EBU frame structure is illustrated in Figure 13.

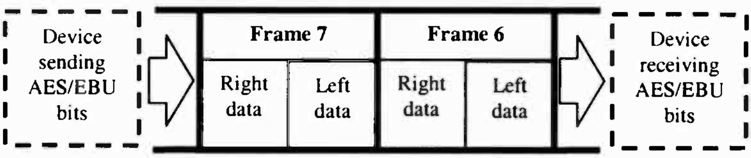


Figure 13: The AES/EBU Format

AES/EBU is *not* a file storage format. It is a standardized format for *transporting* digital audio from one point to another. There are many different digital audio file storage formats, almost as many as there are manufacturers of digital audio storage equipment. Typically, when a digital audio file is retrieved from a hard disk and sent, say, to a mixing board, the playback device (the hard disk-based system) reads the file from the disk, formats it into an AES/EBU data stream, and sends it out through a cable to the mixing board. The mixing board then receives the digital audio through an AES/EBU-compliant input port. Of course, in order to make use of the AES/EBU format, the sending device must be capable of sending data in this format *and* the receiving device must be able to receive this format.

Whether a radio station is using a hard disk-based system, or a completely analog studio system, after the audio program material leaves the mixing board, and before it is delivered to the transmitter, there are several important pieces of equipment that it travels through. This equipment can be broken down into three categories: distribution amplifiers and servers, audio processing equipment and Emergency Alert System (EAS) equipment.

Distribution Amplifiers and Servers

Distribution amplifiers (or DAs, as they are often referred to) are relatively simple pieces of equipment which take an electronic signal and distribute it to several places. They are a necessity in an analog broadcast studio because a particular signal will generally only come out of a mixing board via one particular output connection and this single connection, by itself, cannot be used to feed multiple pieces of equipment without harming the output signal or, worse, damaging the equipment.

As an example of why a DA is needed, consider a typical radio studio setup where the DJ talks on the air to people who have called in over the phone, and records some of these conversations for later rebroadcast. In this situation, the output of the mixing board needs to be fed through a telephone hybrid into the telephone line so that the caller can hear what is being transmitted over the air through the phone line. In addition, the output of the mixing board needs to be fed into a recording device so that the conversation between the DJ and the caller can be recorded. Finally, the output of the mixing board also needs to be fed to the transmitter for broadcast over the air. The function of the DA in this scenario is to take the single output signal from the board as an input and resend it, at its full original strength, to all three locations.

DAs can also be used to feed the output of a mixing board in one studio into the input channel of a mixing board in another studio. Or, they can be used to feed multiple recording devices (such as a cart recorder, a reel-to-reel recorder and a cassette recorder) in a single studio.

A studio that has been completely converted to a hard disk-based digital format will need to have the computer equivalent of a DA — called a *server* — for the same general reasons that a DA is needed in an analog studio.

Servers are the devices that allow, for example, everyone in an office to share a single copy of a word processing program over a computer network. Similarly, they can also allow multiple recording devices to share a single hard disk version of a song in a radio studio.

The major difference between a server and a DA (other than the fact that the server receives and sends digital computer signals and the DA receives and sends analog audio signals) is that the server is also a storage device. Songs, commercials, newscasts and all other types of audio segments used in a broadcast facility can be stored on a server for later recall by whatever playback device wants to use them. In addition, the server can be used for "live" retransmission of a digital signal as it receives the signal. A DA, on the other hand, is only capable of sending out audio that it is receiving.

Audio Processing Equipment

The purpose of audio processing equipment is to create a "signature sound" for the radio station, or at least to take the "plain old audio" that comes from the microphone, CD player, tape machine, etc. and enhance it in order to make it sound better. Audio processing is as much an art as it is an engineering science. Some stations do a lot of it and employ several different pieces of equipment in the process.

Other stations do less and might only have a single piece of processing equipment. Most stations, particularly commercial ones that are competing with other stations for listeners and advertising dollars, do at least a moderate amount of audio processing.

From an engineering standpoint, the purpose of audio processing is to maintain the level of energy in the station's audio to within a specified range. Usually, this is done on a frequency band by frequency band basis. The best way to understand how it works is to imagine an equalizer similar to one you might have with your home stereo or car radio. An equalizer, as those familiar with them know, is designed to amplify, or suppress, the level of signal within particular portions of the audio frequency band. Increasing the level of higher frequency signals, or decreasing the level of lower frequency signals, will make the audio have more "treble." Decreasing the level of higher frequency signals, or increasing the level of lower frequency signals, will make the audio have more "bass." What sets typical broadcast processing equipment apart from a normal equalizer is that the amount of equalization performed by the broadcast processor is dynamic (*i.e.*, it changes with time) and it is usually a function of the program material.

Let's consider an example of how a broadcast audio processing system might work. For this example we will assume that the processing equipment works over three different frequency bands -- low (bass), mid-range, and high (treble). Let's say that the station using this equipment wants the on-air signal to have as high a level (volume) as possible in all three bands. In this situation, the processor will be set to increase the signal level in each band.

In a home stereo system, increasing the signal level across all frequencies is very simple — the level (volume) control for each frequency is turned up. In a broadcast audio processing system, however, things are a bit more complicated. This is due largely to the fact that FCC rules limit the level (volume) of the transmitted audio.

The volume of the transmitted audio is very important to most stations. Although some will do it for other reasons, the primary reason that most radio stations use audio processing is to increase the loudness of their signals. Many broadcasters believe that a signal which sounds louder will be perceived by the listener as being stronger and therefore better! The secret to making a broadcast station sound loud is to increase the level of the softer portions of the program material, and decrease the level of the louder portions of the program material, to the point where the output of the audio processing equipment is kept at as constant a level as possible. The reason that keeping the output level nearly constant is important is because the radio station must remain in compliance with the FCC's modulation limits.

Modulation increases and decreases with the level of a station's program material. The stronger (*i.e.* louder) the program material is when it is fed into the transmitter's exciter, the greater the modulation level of the transmitted signal. (See Part III for a description of the transmitter and the exciter.) In fact, the modulation level of a broadcast signal can basically be thought of as the volume level of the signal.

Generally speaking, the FCC sets a maximum limit on modulation for two reasons. First, it helps to insure that one broadcaster's signal does not interfere with another broadcaster's signal and, second, it helps to insure a reasonably similar level of audio from all stations, providing a generally stable listening environment for the audience.

Let's get back to our example of making a radio station's signal sound as loud as possible. There are several pieces of equipment which are typically used in the processing process — namely equalizers, compressors/expanders, limiters and clippers. These pieces of equipment are generally installed in a station's air chain in the order shown in Figure 14.

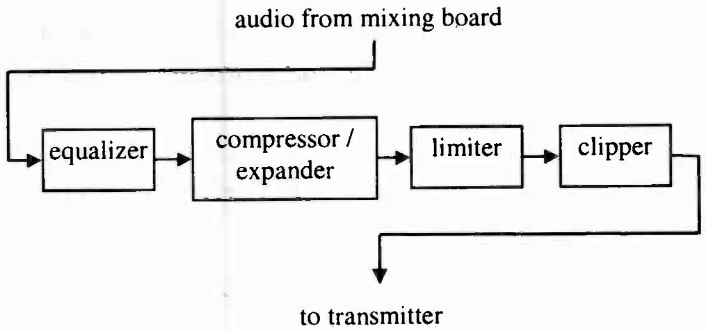


Figure 14: Processing Equipment in a Typical Air Chain

Although shown as separate pieces of equipment in Figure 14, the equalization and compression/expansion functions are often performed by the same piece of equipment. Equalization is needed to perform the actual boosting of the signal level over the appropriate frequency range (in our example, the entire frequency range). Compression is needed to ensure that the boosted signal does not exceed the FCC modulation limit. Expansion is needed to ensure that low-level (quiet) signals, such as background noise and electronic hiss, are suppressed and not amplified to the point that they become annoying. A limiter is needed to further suppress any peaks in the signal that still exceed the

FCC modulation limit after compression, and a clipper can "chop off" any excessive peaks that make it out of the limiter. Let's look at some pictures that illustrate what happens during each step in the audio processing process.

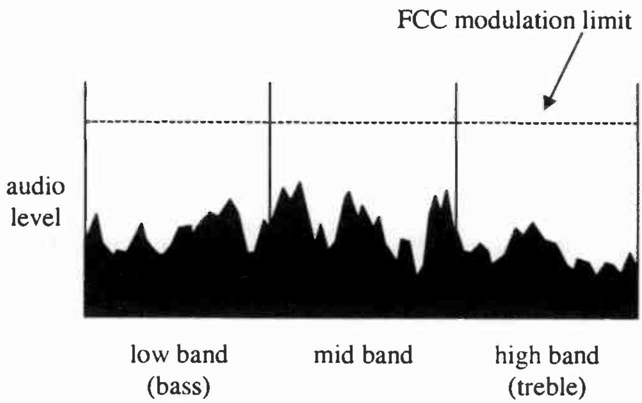


Figure 15: Unprocessed Audio

Figure 15 illustrates the signal level (volume) of an unprocessed audio signal across all audio frequencies. The simplest and most intuitive way to increase the loudness of this signal is simply to increase the signal level (turn up the volume) across all frequency bands using an equalizer. The signal that results from this action is illustrated in Figure 16. (A station that, for example, is interested in having more bass in its signal might increase the lower frequencies to a greater degree than the higher frequencies.)

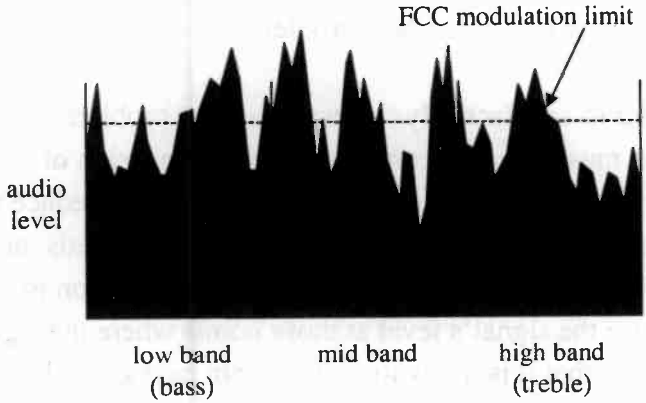


Figure 16: Amplified (Louder) Audio with No Compression or Expansion

By looking at Figure 16 we can see that simply turning up the volume of the audio produces a major problem for a radio station — overmodulation. All of the signal that lies above the dashed "FCC modulation limit" line in Figure 16 will cause the radio station's signal to exceed the modulation (*i.e.*, volume) limit specified by the FCC. This overmodulation might cause the station's signal to interfere with the signals from other broadcast stations. Another, somewhat more subtle problem that is caused by simply turning up the volume of the entire signal is the amplification of lower level (softer) signals which, in many cases, are likely to be just background noise or electronic hiss. The "valleys" in the signal shown in Figure 16 are the areas where this might be a problem.

In order to satisfactorily correct these two problems, the station must do some compression and expansion of its audio. Specifically, it must use compression to reduce the audio signal's level at those points where it exceeds the FCC's modulation limit, and it must use expansion to decrease the signal's level at those points where the signal is so low that it is likely to only contain background noise or electronic hiss. An illustration of where compression and expansion might be used is provided in Figure 17.

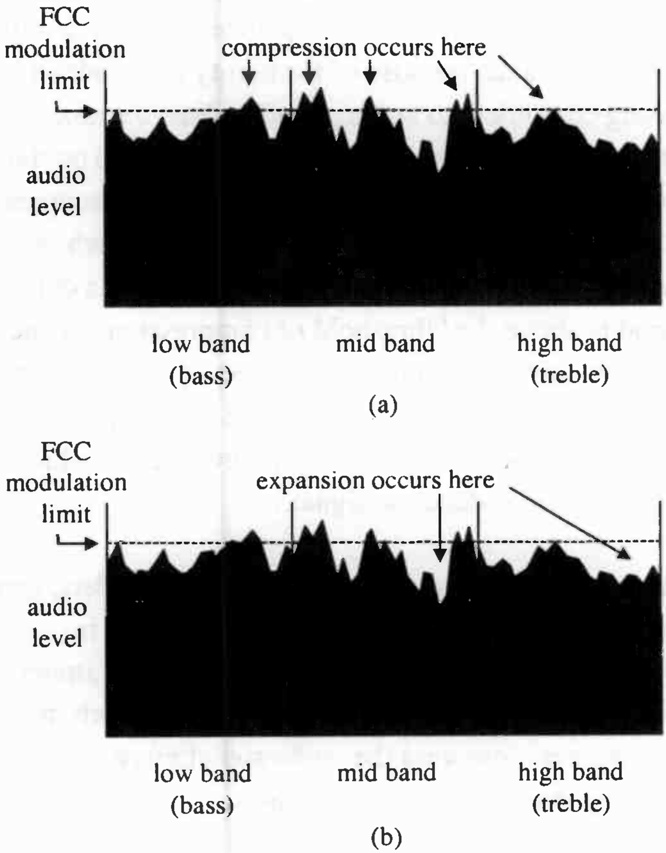


Figure 17: Amplified Audio with Compression and Expansion

It is worth repeating here that broadcast processing equipment differs from a typical equalizer found in many home stereo systems because of the ability of broadcast processing equipment to automatically adjust the amount of compression, expansion, etc. that it is doing based on the program material. A home equalizer will always suppress a signal at, for example, 1 kHz, if it is set to do so, while a broadcast processor will suppress a signal at 1 kHz only if the signal is above the "threshold of compression." (The threshold of compression is the signal level above which the compressor will reduce the signal. Similarly, the "threshold of expansion" is the signal level below which the expander will reduce the signal.)

In addition to equalizers, compressors and expanders, there are two other devices which are commonly found in broadcast audio chains — limiters and clippers. Limiters and clippers are both essentially compressors which, to varying degrees, compress the audio signal more aggressively than a "plain" compressor.

A *limiter* is typically used to take the peaks that still exist in a signal after compression and knock them down further. This is sometimes necessary when, after compression, a signal still has peaks in it that are high enough to result in overmodulation and a violation of FCC rules. A *clipper* is generally used as a last resort to "chop off" any remaining peaks of overmodulation after the signal has passed through both the compressor and the limiter. A clipper, if not used correctly, can cause severe distortion to a signal because it literally clips the peaks off — it does not "softly adjust" the peaks like the compressor and limiter.

Although the configuration of processing equipment described above is a typical one, it should be noted that equalizers, compressors, expanders, limiters and clippers can be used in a variety of configurations. As we said earlier, audio processing is as much an art as it is an engineering science, and some "artists" may prefer to use only certain pieces of processing equipment.

Well, that about covers audio processing. As we mentioned earlier, after a radio station's program material leaves the main mixing board, it will generally travel through three types of equipment before being fed to the transmitter. These are distribution amplifiers or servers, audio processing equipment (all of which we just covered) and EAS equipment. Now, on to EAS equipment.

Emergency Alert System Equipment

The Emergency Alert System, or EAS, was first implemented on January 1, 1997, replacing the old familiar Emergency Broadcast System (EBS). The EAS is the communications network that has been designed by the Federal Government to allow the President to quickly and efficiently speak to the entire nation in the event of a national emergency.

Although the primary function of the EAS is to provide a means for issuing national alerts, it has to date only been used for its secondary purpose -- providing state and local officials with a means of alerting local communities about local emergencies like severe weather, chemical leaks, and fires.

From an engineering standpoint, the way EAS operates is relatively simple. As shown in Figure 18, an EAS encoder/decoder is installed in a station's *air chain* in such a way that it can interrupt the flow of normal programming to the transmitter in order to insert an emergency message. (A station's "air chain" is the path that its on-air program material follows from the program source to the transmitter.)

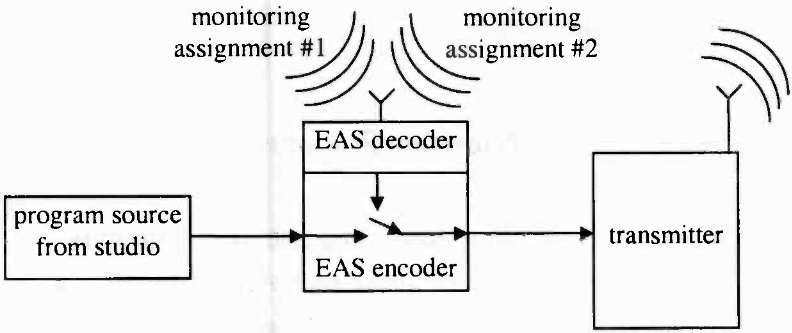


Figure 18: EAS Equipment in a Radio Station Air Chain

The EAS decoder is constantly monitoring the transmissions from the two sources that it has been assigned to monitor. These two sources are usually either other broadcast stations or NOAA Weather Radio. The reason that the decoder is required to monitor two sources is to help insure that it will still receive an alert message if one of its monitoring assignments happens to be off the air.

If an alert is received by the EAS decoder, and it is a type of alert that the station using the equipment has determined should be on the air, the EAS encoder will break into the station's air chain and put the alert on the air. Encoders can be programmed to do this automatically, or they can be programmed to require a person to manually interrupt the station's programming.

Television Stations

The program material produced in a television studio is basically the same as the program material produced in a radio studio — except, of course, for the addition of a video signal to accompany the audio. On the surface, adding a video signal sounds like a relatively simple task — and, in some respects, it is. However, the video signal does significantly increase the complexity of a television studio over that of a radio studio. The main reason for the added complexity is the need to insure that the video and audio signals in a television studio remain in synch, and that all of the video switching equipment is timed correctly. We will cover these two aspects of a television studio in greater detail here, and we will build a solid foundation for understanding video timing issues by learning about how a television picture is created. We will not, however, go over all major *audio* components of a television studio because the audio equipment used in a television studio is generally very similar to the audio equipment used in a radio studio (although the typical TV studio setup usually involves the use of mostly "live" audio sources while the typical radio studio setup usually involves more prerecorded audio sources). For a review of studio audio you may read the preceding sections on radio station studio facilities.

Now, let's begin by learning about the standard video signal used in an analog television studio — NTSC video.

NTSC Video

"NTSC" refers to the National Television Systems Committee - the committee that, decades ago, designed the standard for today's analog television transmissions. A new, completely digital television standard has been developed by the Advanced Television Systems Committee, or ATSC, and it will gradually be implemented in television stations in the years to come.

NTSC video signals are, in reality, a rapid-fire series of still pictures that are projected on a television receiver at a rate of 30 pictures per second. Each of these pictures is called a "frame." This rapid-fire series of still pictures creates the illusion that the picture on the TV is moving.

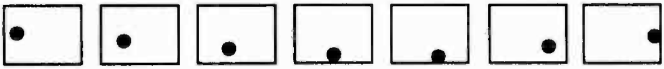
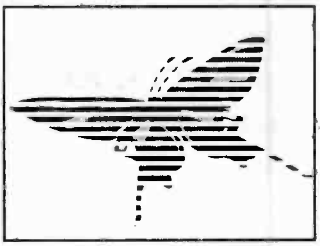


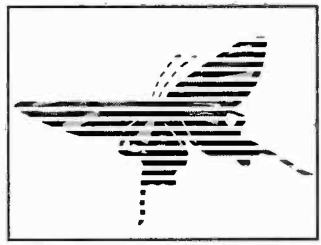
Figure 19: Series of Still Pictures that Create Illusion of Motion

Figure 19 provides a simple example of a series of still pictures which might be used to create the illusion that a ball is bouncing across the screen.

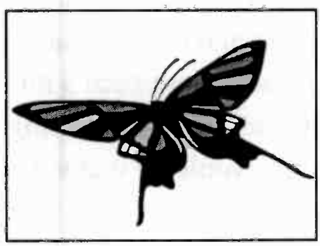
In the NTSC system, each video picture frame is painted on the television screen, from top to bottom, one horizontal line at a time. There are 525 horizontal lines in each frame (483 of which form the actual picture), but they are not painted in successive order (*i.e.*, 1, 2, 3 …, etc.). Instead, all of the odd-numbered lines are painted first, followed by all of the even-numbered lines. This process is called *interlacing*.



field A-1



field A-2



frame A

Figure 20: Two Interlaced Fields for Each NTSC Frame

The two images that are created during the interlacing process (the picture with only odd-numbered lines and the picture with only even-numbered lines) are called "fields." There are two fields for every frame, as illustrated in Figure 20. Since the frame rate is 30 pictures per second, the field rate is 60 fields per second, or one field every 1/60th of a second. The odd- and even-numbered lines are interlaced together to form the complete picture. Interlacing is used because it helps to eliminate flickering of the TV screen. How? Well, the best way to answer this is to look at what would happen if interlacing were *not* used.

As we noted earlier, there are 30 frames of video per second. This means that each individual line on the TV screen is updated, or "refreshed," 30 times per second (this is true in both interlaced and non-interlaced situations — or at least it would be if there were such a thing as non-interlaced NTSC video). If interlacing were *not* used, then each picture frame would be painted on the screen, from top to bottom, in its entirety, and would then be followed by the next picture frame, and the next one, and so on and so on. The effect this would have on the overall picture is best illustrated by isolating two adjacent lines of video on the TV screen. For our example, let's randomly pick lines 137 and 138. On a non-interlaced screen lines 137 and 138 will be refreshed at essentially the same time. (Line 138 will actually be refreshed a very tiny fraction of a second after line 137, but this extremely small time difference is not relevant as far as understanding the difference between interlaced and non-interlaced screens is concerned.) After lines 137 and 138 are refreshed, they are refreshed again 1/30th of a second later. To the human eye, it appears as though the area on the screen encompassed by lines 137 and 138 is being refreshed 30 times per second.

On an interlaced screen line 138 will be refreshed 1/60th of a second *after* line 137 because the first field (the odd-numbered lines) are refreshed during the first half of each 1/30th of a second frame, and the second field (the even-numbered lines) are refreshed in the second half of each frame. Because, at a normal viewing distance, the human eye cannot distinguish between lines 137 and 138, the net effect of the interlacing is to make it *appear* that both lines are being refreshed 60 times per second — when in fact each individual line is only being refreshed 30 times per second.

The refresh rate is very important because, if the rate is too low, each line on the screen will have noticeably started to fade from the screen before it is refreshed, causing the screen to flicker. Increasing the refresh rate (or, in the case of NTSC interlacing, creating the *appearance* of an increased refresh rate) helps to reduce flickering.

To review, in an interlaced NTSC television picture, each of the lines in the picture is still refreshed 30 times every second. However, to the human eye, the interlacing makes it *appear* as though the screen is being updated twice as often, or 60 times every second. What makes this possible is the fact that the human eye generally cannot perceive the fact that two adjacent lines on the video screen are being refreshed at different times when there is only a period of 1/60th of a second between their respective refresh times. Because the eye cannot perceive the difference in refresh times, each half-screen refresh (odd- or even-numbered lines) has nearly the same effect on the viewer as a full screen refresh, the effect is to create the appearance that the full screen is being refreshed twice as often, or 60 times per second. Increasing the *apparent* refresh rate causes screen flicker to be reduced.

Readers who are familiar with computer equipment will know that *non*-interlaced monitors are the viewing screens of choice for many computer users. You may be asking yourself then, if interlaced video is so good, why do so many computers have non-interlaced screens? Well, the answer turns out to be cost. In a computer system, it is less expensive to implement a non-interlaced display than it is to implement an interlaced display. This is because, in a computer system that uses a non-interlaced display, a single block of memory can store the information for the entire screen, and the entire contents of that block of memory can be transferred to the screen 75 times every second (or more, or less, depending on what the video refresh rate is). In an interlaced system, on the other hand, additional memory and/or control circuitry would be necessary because the odd-numbered lines would have to be painted first, followed by the even-numbered lines. There would either have to be two separate blocks of video memory (one for odd-numbered lines and one for even-numbered lines) or a more complex procedure for only transferring half of the lines in a single block of memory to the screen followed by the other half. In either case, the cost of implementing an interlaced system is greater than the cost of implementing a non-interlaced system when it comes to the number of memory chips and circuitry required.

Computer makers have the option of producing non-interlaced systems because the amount of video information that can be transferred from a computer to its monitor is large enough that a full-screen, non-interlaced refresh rate can be achieved which does not result in screen flicker. In other words, an entire computer screen (both odd and even lines) can be updated at a typical rate of about 75 times per second because enough video information, at a high enough rate, can be fed from the computer to the monitor to make this happen. In the NTSC television channel, on the other hand, only 30 frames worth of video information can be fed to the screen every second. This is because the amount of video that can be sent to an NTSC television screen is restricted by the size of the NTSC television channel, which is 6 MHz wide. It is also restricted by the fact that 30 frames of video per second was pretty much the best that available technology could do in a 6 MHz channel when the NTSC standard was defined.

Before we go on to discuss the importance of video timing and audio-video synchronization in a television studio, let's finish our discussion about the basics of NTSC video by learning about methods for including additional, non-video information in an NTSC video signal.

The Horizontal Blanking Interval

As we have already learned, there are 525 horizontal lines of information in an NTSC video signal, 483 of which carry the actual picture information. The lines that carry the picture information are painted on the screen in a top-to-bottom manner with all of the odd-numbered lines being painted first followed by all of the even-numbered lines.

The *electron gun* inside the *cathode ray tube (CRT)* is the device inside the television receiver that actually paints the video picture. It shoots a beam of electrons at the back of the video screen in a left-to-right, top-to-bottom manner (odd-numbered lines first, even-numbered lines second). Each time the electron gun's beam reaches the right edge of the picture screen it must stop and then move back to the left-hand side of the screen in order to start painting the next line. If the electron gun were to remain on during this entire process, it would end up painting a line of video on the screen and then immediately painting a streak right below the line of video while it retraced its path back to the left-hand side of the screen. In order to prevent this from happening, the electron gun is turned off after it reaches the far right-hand side of the screen and it remains off until the beam is positioned back on the left-hand side of the screen and ready to begin painting the next horizontal line. The period when the electron gun is off while it is retracing its route over the screen is called the *horizontal blanking interval.* The horizontal blanking interval is a very short period of time, significantly less than it takes to actually paint one horizontal line of video on the screen.

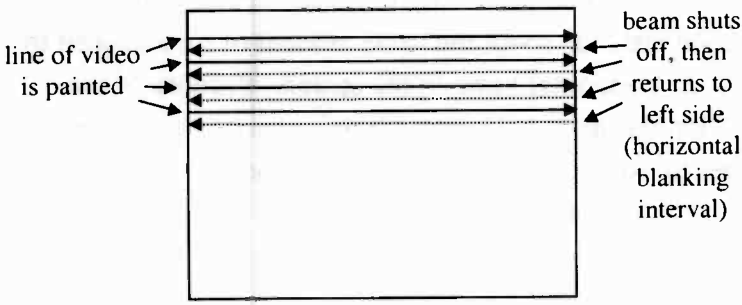


Figure 21: The Horizontal Blanking Interval

The Vertical Blanking Interval

As one would expect, the same concern about putting a streak on the screen must be addressed when the electron beam finishes painting the bottom line of video and needs to be repositioned to the top left-hand corner of the screen. During this operation, the beam is once again turned off while it is retargeted toward the upper left corner. This period when the beam is turned off is called — yes, you guessed it — the *vertical blanking interval.* It takes a lot longer for the electron beam to reset itself from the bottom right-hand corner of the screen to the top left-hand corner than it does to reset itself from the right-hand side of one line to the left-hand side of the next. While the horizontal blanking interval is significantly shorter than the time it takes to paint one horizontal line, the vertical blanking interval lasts for about the same amount of time needed to paint 21 horizontal lines on the screen. Because of the additional time in the vertical blanking interval, it is much easier to use it to transmit auxiliary information than it is to use the horizontal blanking interval.

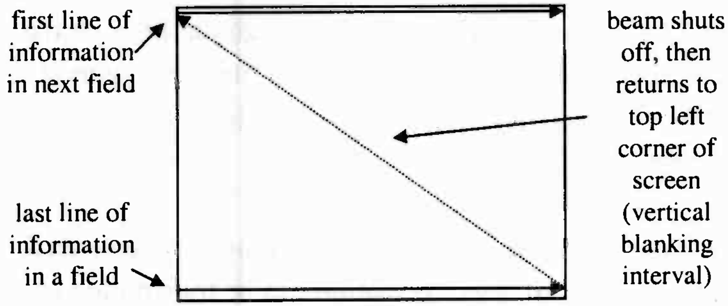


Figure 22: The Vertical Blanking Interval

Because the vertical blanking interval lasts essentially as long as the time it takes to paint 21 horizontal lines on the screen, the time during the vertical blanking interval is referred to as if it were horizontal lines on the screen. It is customary, for example, to refer to "line 21 of the vertical blanking interval" or "line 19 of the vertical blanking interval."

One thing that leads to confusion for some people is the fact that all 21 lines of the vertical blanking interval are transmitted before each *field*. (Remember, a *field* is only half of the lines in a frame — either the odd or even ones.) This makes sense, if you think about it, because it is at the end of each field that the electron beam must reset itself from the bottom right-hand corner of the screen to the top left-hand corner. The reason this leads to confusion is that each field consists of only half the horizontal lines on the screen (odd or even), but each vertical blanking interval (VBI) contains all of its 21 lines — not just odd or even numbered VBI lines.

The Federal Communications Commission has specified what type of auxiliary information can be transmitted on various VBI lines. A summary of the FCC's requirements is as follows:

Lines 1-9	*Vertical synchronization information only* (needed to insure that the TV receiver knows this is the end of one field of video and the start of the next)
Lines 10-18	*Test, cue and control signals; telecommunications (or other applications with prior FCC approval)* — *e.g.*, control signals could be sent from a network to local affiliates to alert the affiliates to the fact that a local commercial break is coming
Line 19	*Ghost canceling signal only* (used by receivers equipped with ghost-canceling circuitry to reduce "ghosts" caused by the simultaneous reception of multiple versions of the same TV signal — usually due to signal reflections)

Line 20	*Test, cue and control signals; telecommunications (or other applications with prior FCC approval)* (see above)
Line 21	*Closed captioning only* (descriptive text used by hearing impaired persons)

ATSC Video

ATSC is the Advanced Television Systems Committee, which adopted a standard for digital television transmission in 1995. This standard was subsequently adopted (with the exception of the video formats portion) by the Federal Communications Commission in 1996.

Like NTSC video signals, ATSC video signals are a rapid-fire series of still pictures that are displayed by a television receiver. That is about as far as the similarity goes, however. ATSC signals can have different aspect ratios than NTSC signals. They also have more, or about the same number of horizontal lines per screen (whether it is more or about the same is a choice available to the broadcaster transmitting the ATSC signal). The rate at which ATSC pictures are painted on the screen can also differ from NTSC. And, ATSC video is transmitted in blocks of compressed digital data, instead of in a continuous stream of analog data like the NTSC signal. Finally, while the audio that accompanies an NTSC video signal is essentially of the same technical format as the audio from an FM radio station, the digital audio that accompanies ATSC video is of a dramatically different format from that used in radio broadcasting.

The ATSC signal is a stream of digital bits that is created by an ATSC encoder and delivered to the television station's transmitter at a rate of 19.39 million bits per second (megabits per second). In order to appreciate how fast this data rate is, consider the fact that a 56K modem for a personal computer – generally considered the state of the art today – has a data rate of only 0.056 megabits per second. Thus, it would take nearly 350 56K modems to handle the same amount of data that a single piece of ATSC equipment can handle.

The signal that is actually routed around the television studio and used for editing, recording, etc., can be one of various video formats – and in fact many stations will likely continue to use their NTSC equipment in the studio through the early stages of digital television implementation. The only requirement is that, in order to broadcast an ATSC digital television signal, the station must have an ATSC encoder that is capable of encoding the signal format used by the station's studio equipment.

The bits in the ATSC data stream are organized into 188-byte blocks. The first byte of each block is used for synchronization. This synchronization byte is needed to clearly identify when a new block of data is beginning so that the device receiving the data stream can properly decode the data. Because one of the 188 bytes in each block of data is dedicated to synchronization, there are only 187-bytes per block available for actual video, audio and/or ancillary data. The data rate of the actual video, audio and/or ancillary data (often referred to as the *payload data rate*) is 187/188 x 19.39 megabits per second, or 19.28 megabits per second.

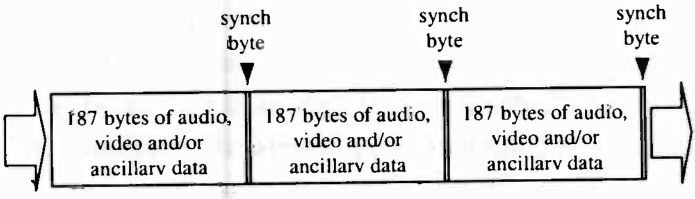


Figure 23: The ATSC Data Stream

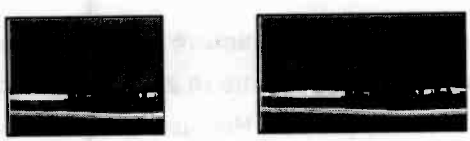
It is worth mentioning here that, although the information data rate of an ATSC signal is 19.39 megabits per second, the data rate of the digital television signal that is broadcast from a DTV transmitter site is much higher. This is because additional data must be added at the transmission point in order to have enough error-correcting information to fix all of the damage caused to the digital signal as it travels from the transmitter to the receiver. We will discuss this more in Part III.

So, we now know how the ATSC data stream is formatted, but what exactly is the information it is carrying? We will explore that next.

Pixels

Chances are that you are already familiar with the concept of a *pixel*. A pixel is a dot on a video screen, the smallest component of a video image. Pixels have become very familiar to most computer users because the resolution of a computer screen – *i.e.* the level of detail on the screen – is usually defined in terms of pixels. Some typical computer screen resolutions defined in terms of pixels (horizontal and vertical) are 640x480, 800x600 and 1024x768. The more pixels there are on the screen, given a fixed monitor size, the more detailed or cleaner the image.

The pixel is also the smallest component of an ATSC video image. One of the high-definition ATSC video formats has, from top to bottom, 1080 horizontal lines across the screen. Each of these lines contains 1920 pixels. You will recall from the section on NTSC video that the analog television system being phased out in the United States has only 525 lines per frame of video, and only 483 of these contain viewable information. It is easy to understand where the term *high-definition television* comes from when you consider that a 1080-line ATSC video frame has more than double the number of horizontal lines of an NTSC video frame.



4:3
NTSC or ATSC
video image

16:9
ATSC
video image

Figure 24: 4:3 and 16:9 Aspect Ratios

There are several reasons that the two aspect ratios, 4:3 and 16:9, were selected for the ATSC standard. The 4:3 aspect ratio was selected because it is the same aspect ratio used in NTSC video, and there are a tremendous number of archived television programs in this format. The 16:9 aspect ratio was selected because it had already gained acceptance in other parts of the world as a compromise between the motion picture industry's desire for as wide a screen as possible and the manufacturing costs of tube-based displays. About 80% of motion pictures are shot at an aspect ratio of 1.85:1, which easily fits into a 16:9 screen with negligible use of letterboxing. (*Letterboxing* is a technique used to fit a video image onto a television screen, without altering the aspect ratio of the original video image, by blacking out the top and bottom portions of the television screen.)

Each line in an NTSC video picture is a continuous stream of video information. Each line in an ATSC video picture is a series of discrete pixels. Because an ATSC video image is a series of discrete pixels, it can be manipulated by computer circuitry at both the transmitter and the receiver more easily. Most importantly, video can be compressed at the transmitter and decompressed at the receiver – this is what makes digital television (DTV) using the ATSC standard possible.

Compression

Compression is an extremely important aspect of the ATSC video signal. In fact, as we just noted above, compression is so important to ATSC video that, without it, high-definition digital television would not be practical. Put quite simply, to compress digital video means to take a complete digital video signal, throw away unneeded parts of it, and encode the remaining parts in a manner that reduces the total amount of data required to store or transmit the video. The process of discarding unneeded components of the video signal, and encoding what remains, is performed by a computer program called a *compression algorithm*. "Algorithm" in this context is just another way of saying "computer program."

The compression algorithm used for ATSC video is very complex. Because it is not necessary to understand exactly how it works in order to have a good understanding of ATSC video we will not try to explain it here. Such an explanation would likely require an entire book by itself. We will, however, briefly discuss two basic principles of compression in order to provide a general understanding of how it is possible to reduce the amount of data in a video signal without noticeably degrading the video.

The first basic principle of compression is the elimination of unneeded data. Basically, data is unneeded if, once discarded, the viewer cannot tell that it is gone. How can data be discarded without the viewer noticing? Well, the secret lies in the fact that digital video cameras and their associated recording equipment are capable of "seeing" more than the human visual system can detect. For example, a digital video recording system is capable of recording the amount of brightness and the amount of color in a picture very accurately. It turns out, however, that the human visual system is generally less sensitive to the amount of color in a picture than the amount of brightness (*i.e.*, certain changes in brightness are easier for the human eye to detect than certain changes in color). For this reason, some of the most detailed color information in a digital video picture can be discarded without changing what human viewers will actually see.

The second basic principle of compression is the efficient coding of the remaining video. The best way to understand how video data can be coded in a more efficient manner so as to require less data overall is to use an example. Let's assume that a particular video image has a red horizontal line 100 pixels in length extending across the screen. The uncompressed video data for this image might include instructions like this:

print pixel 1 in red
print pixel 2 in red
print pixel 3 in red
.
.
.
print pixel 98 in red
print pixel 99 in red
print pixel 100 in red

However, the compressed video data for this image might include instructions like this:

print red pixel
repeat 99 more times

While the type of compression that is used in ATSC video is much more sophisticated than this simplistic example, one can see from this example how it is possible to more efficiently code raw video data in a manner that requires less data overall to transmit the video image but, at the same time, does not degrade the video image.

Frames

As noted earlier, ATSC video signals, like NTSC video signals, are a rapid-fire series of still pictures that are projected on a television receiver. Each of these still pictures is called a frame.

There is a major difference between NTSC video frames and ATSC video frames. In NTSC video each frame is, from a technical design standpoint, identical to the one before it and the one after it. That is to say, although the video data (the picture) carried in the frame changes from frame to frame, the design of each frame itself remains constant. The odd-numbered horizontal lines of video are transmitted first, in successive order, followed by the even-numbered lines. This format remains constant for all frames in every NTSC video stream.

In the digital ATSC video stream there are three distinct frame types used to carry video: *intracoded frames, predictive coded frames* and *bi-directionally predictive coded frames.* These are generally referred to simply as I-frames, P-frames and B-frames. The reason there are three different frame types for ATSC digital video is that using three different types of frame permits further compression of the video signal, thus reducing the amount of disk space needed to store it, and the amount of radio spectrum needed to transmit it.

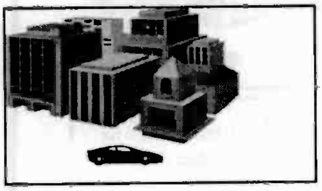


Figure 25: I-Frame

An I-frame can stand by itself, without accompanying P-frames or B-frames. It is *intra*coded, which means that the only data compression coding done to it is within its own borders. Thus, when an ATCS video stream is decoded, an I-frame can be decoded by looking only at the data within itself. *Intra* is a Latin prefix that means "within."

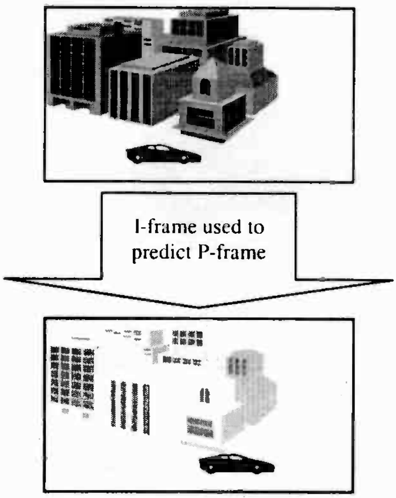


Figure 26: I-Frame Followed by P-Frame

A P-frame cannot stand by itself. At least some, and possibly all, of the picture in a P-frame must be predicted from the most recent I-frame or P-frame. Many video images lend themselves to this kind of prediction because they are either static (a still picture), or they involve some form of predictable motion. The advantage of a P-frame is that it involves more compression than an I-frame (both intra- and *inter*frame coding) and therefore helps to reduce the data rate necessary for transmitting an ATSC signal. The disadvantage of a P-frame is that it cannot be used, by itself, to create a "freeze frame" image of the segment of video that it represents.

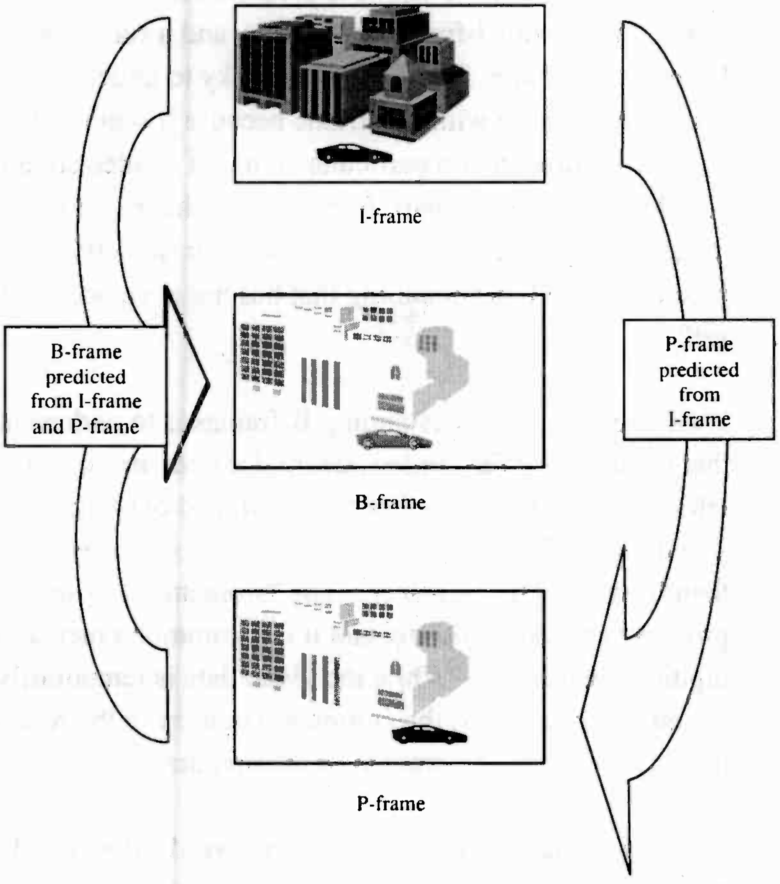


Figure 27: A B-Frame between an I-Frame and a P-Frame

Like a P-frame, a B-frame cannot stand by itself. Unlike a P-frame, however, a B-frame is predicted from *two* other frames, a previous I-frame or P-frame, and a subsequent I-frame or P-frame. It is a little bit tricky to understand what is happening with a B-frame because it seems a bit counter-intuitive that a particular frame in a video stream could be predicted, in part, from a frame that has not even made it to the screen yet. That is, how can something be predicted based on something that has not even occurred yet?

Well, the trick to understanding B-frames is to understand that the digital ATSC video stream does not arrive at a television set and immediately get dumped onto the screen. The digital ATSC data is temporarily stored, processed, and then forwarded to the screen. The "store and forward" process is called *buffering*, and it is a common practice in digital transmission. While the video data is temporarily stored in the receiver, the computer circuitry in the receiver has an opportunity to process or manipulate it.

The reason that the data must be temporarily stored and processed is that it needs to be decompressed. You will recall from the discussion earlier about compression that, during the transmission process, portions of the original video stream are discarded, and the remaining portions encoded, in a manner that reduces the total amount of data needed to transmit the signal. Well, although the compressed television signal that is transmitted over the air contains enough information to recreate a video image on the television screen that appears, to the typical viewer, to be identical to the original video, the compressed signal itself does not carry all of the necessary video data. Instead, it carries instructions that enable the computer circuits inside a DTV receiver to recreate the image.

This is a good place to point out that the buffering which occurs inside a television receiver is the reason that there is a brief delay between the instant that the viewer selects a new channel and the instant that the new channel appears on the screen. Also contributing to this delay is the fact that, in order to begin decoding video from a new channel, the decoder must wait for the next I-frame to be transmitted on that channel. This is because the I-frame is the only one that can stand by itself without any previous frames to use for prediction – so in order to begin decoding a new stream of video a receiver must start with an I-frame.

So, back to the B-frame and its place in this process. As noted above, a B-frame is predicted from a previous I-frame or P-frame *and* a subsequent I-frame or P-frame. For the receiver to be able to construct a B-frame, the two frames from which it is predicted must be transmitted before the B-frame. So, if three consecutive frames of video at the television studio have been encoded in such a manner that the first is an I-frame, the second is a B-frame and the third in a P-frame, the I-frame will be transmitted first, followed by the P-frame and then the B-frame. The receiver will then have the opportunity to receive and store the information for both the I-frame and the P-frame (frames 1 and 3) before it has to decode the B-frame (frame 2).

Frame Rate

Another major difference between ATSC video and NTSC video is that ATSC video may be transmitted at different frame rates. Recall from the discussion about NTSC video that the only frame rate available under that system is 30 frames per second. The limited size of the television channel (*i.e.*, the limited amount of spectrum available in which to transmit the video) basically prevented the NTSC standard from using any frame rates higher than 30 frames per second. Higher frame rates would have required more information to be transmitted over the air, which would have required wider channels.

Because the ATSC standard includes a method for compressing the transmitted video, more video information can be squeezed into the same amount of spectrum. This provides several opportunities that were not available with the NTSC standard. Specifically, it allows bigger pictures to be transmitted (hence the availability of wide-screen pictures); higher frame rates (more pictures per second) to be transmitted; and/or multiple video streams to be transmitted. We discussed the wide-screen aspect of ATSC video earlier, and we will get to its multiple video stream capability shortly. For now, we are focused on the frame rate.

While the ATSC video signal is capable of being transmitted at a higher frame rate than NTSC video, it does not have to be transmitted at a higher frame rate. There are three standard frame rates for ATSC video – 24, 30 and 60 frames per second. Twenty-four frames per second is a rate commonly used for film. Thirty frames per second is, of course, the rate used for NTSC video, and 60 frames per second is a faster frame rate that further reduces screen flicker by refreshing the video screen more often. The more frames per second that are transmitted, the more data that is needed to transmit, or store, the video signal.

At least for the duration of the transition to digital television the vast majority of television studios will be using frame rates of 1000/1001 times the integer frame rates listed above. They will be doing this to facilitate conversion of NTSC pictures to ATSC (the precise frame rate of NTSC pictures is 59.94 frames per second), and in order to avoid problems associated with having multiple timing signals in a single facility.

Interlacing

Each of the three frame rates in ATSC video can be employed in either an interlaced or non-interlaced manner. It should be noted, however, that 60 frame-per-second video (interlaced or non-interlaced) is not available in the1920 x 1080 format, and 30 frame-per-second video is not available in interlaced form for the 1280 x 720 format. The reasons for this are discussed a bit later. Non-interlaced video is often called *progressive* video because each horizontal line is painted on the screen in order, from top to bottom, in a progressive manner.

We discussed what it means for video to be interlaced in the section on NTSC video, so we will not repeat it hear. The choice of whether to use interlaced video or non-interlaced video at a broadcast facility is really one of personal preference. There are many people who will argue that interlaced video is preferable in many applications, and there are many other people who will argue that progressive video is preferable.

Multicasting

Combining several different video programs together and broadcasting them over the same television channel is called *multicasting*. This is something that was not possible with NTSC video in the 6 MHz-wide television channels in the United States. With digital ATSC video, however, it is possible to fit several smaller, lower frame rate video programs into the same television channel that can transmit a single, wide-screen, high definition program.

The number of different video programs that can be squeezed into a television channel is dependent on the data rate necessary for each individual program.

While the interlaced versus progressive aspect of an ATSC video image is an important aspect of the image, it does not have much of an impact on the digital data rate necessary to transmit the image (*i.e.* 30 frame per second video requires essentially the same data rate whether it is transmitted in interlaced or progressive mode). This data rate is mostly dependent on the frame rate and the screen size of the video, as well as the subject matter of the video. All else being equal, larger screen sizes require higher data rates, higher frame rates require higher data rates, and video with lots of motion and/or scene changes in it also requires higher data rates.

The basic video formats (screen sizes and frame rates) that can be used in ATSC video are listed in Table 1.

Table 1: ATSC Video Formats

Screen Size in Pixels	Aspect Ratio	Frame Rate			
1920 x 1080	16:9	24P	30P	30I	
1280 x 720	16:9	24P	30P		60P
704 x 480	16:9	24P	30P	30I	60P
704 x 480	4:3	24P	30P	30I	60P
640 x 480	4:3	24P	30P	30I	60P

The reason that there is no 1920 x 1080 format at 60 frames per second is that the high data rate needed to transmit this format could not be achieved within the limited amount of bandwidth (6 MHz) available for television broadcasting in the United States. Because this format has the largest screen size, and because transmitting 60 frames per second requires essentially double the data rate necessary to transmit 30 frames per second, this combination of screen size and frame rate would require a higher transmission data rate than any of the others.

The reason that the 1280 x 720 format does not include an interlaced version at 30 frames per second is that, when the ATSC standard was developed, this particular format was seen as a "progressive scan only" format.

Because video transmitted in the 1920 x 1080 format at 30 frames per second uses almost all of the capacity of the 6 MHz-wide DTV channel, when this format is transmitted it generally cannot be accompanied by any other video feeds over the same channel. On the other hand, video transmitted in the 640 x 480 format at 24 frames per second uses the least amount of channel capacity of all the ATSC formats. When this format is used it is possible to transmit multiple video feeds over the same 6 MHz wide channel.

There is no specific formula that can be used to specify exactly how many video feeds of various formats can be transmitted over the same channel. This is because the data rate needed to transmit a particular format is not a constant number. The reason it is not a constant number is that the amount of compression that can be performed on a data stream is dependent on the content of the video programming. Video that is generally static, such as a still picture of a station's logo, can be compressed much farther than video that has a lot of scene changes in it. As we noted earlier when talking about frames, the way ATSC video is compressed involves, in part, sending only information about how one frame differs from another rather than complete information for every frame. So, because successive frames of a still picture video image will have no differences between them, very little information needs to be transmitted to explain to the receiver the difference between one frame and the next.

One thing that is very important to keep in mind when considering the multiplexing of several video programs is that the total data rate for all of the mutiplexed video streams added together must not exceed the total data rate available in the DTV channel at *any* time. For example, a single DTV channel might be capable of carrying four different programs most of the time, but if these programs contain commercials that have a lot of motion and scene changes in them it might be the case that the single DTV channel cannot handle a commercial on every single channel at the same time. Clearly, if the commercials will all be run at the same point in time, this could cause a problem. The program might appear fine to the viewer until the commercials come on at which time the video would become distorted due to a lack of available capacity for transmitting all four commercials at once.

It is also worth noting that the ATSC video standard is based on the MPEG-2 video standard. *MPEG* is the *Moving Pictures Experts Group*, an international standards-setting organization that has developed several standards for transmitting digital video. The MPEG-2 system was optimized for transporting then-existing standard broadcast video sources at a data rate of about 4 million bits per second. Therefore, a general rule of thumb could be that an ATSC signal might be capable of carrying between four and five standard NTSC-quality signals (19 million ÷ 4 million = 4.75).

Ancillary Data

When a broadcaster opts to transmit a single standard definition picture essentially equivalent to its NTSC programming then that broadcaster will have some extra data capacity available in its transmitted signal. This capacity can be used to transmit any kind of data for any purpose. It could be used to insert supplemental data about advertised products, to provide stock quote updates, or even to provide a one-way path for high-speed Internet downloading. There is one important catch, however, that has nothing to do with engineering or technology. The government has imposed a five percent tax on the gross revenues that any commercial television broadcaster receives for providing ancillary (not program related) data services using their DTV signal. These revenues are also subject to normal income taxes as well.

DTV broadcasters do not have to limit their programming to a lesser-than-high-definition format in order to transmit ancillary data. The ATSC signal is capable of carrying what is referred to as *opportunistic data.* This is data that is transmitted whenever an opportunity becomes available in the ATSC signal. For example, if a wide-screen high-definition program is being transmitted there will still be times during the program when the data rate necessary to carry the video information will be reduced. This can happen, for example, during a commercial when a still picture of a product is on the screen. Because the picture is not changing, less data in the ATSC signal needs to be allocated to update the video. This means that *more* data capacity is available.

Before concluding our discussion about ancillary and supplemental data it is important to note that the ATSC signal can also be used to carry closed captioning information, which has a fixed data rate per video program.

DTV Audio

In NTSC television broadcasting the audio transmitted along with the video is of essentially the same technical format as the audio transmitted by an FM radio station. In

NTSC television, the video signal and the audio signal are transmitted as two separate entities. In the ATSC system, however, this is no longer the case. As discussed earlier, the ATSC system involves a single signal that is a continuous stream of data packets. Each individual data packet can carry audio, video and/or ancillary data. It is up to the ATSC receiver to sort them all out.

The packets of audio data in an ATSC signal conform to a system developed by Dolby Labs called *AC-3*, a specific version of which is incorporated into the ATSC standard. Most people refer to ATSC audio simply as "AC-3" audio. The AC-3 system offers many improvements over the audio system used with the NTSC television system. One of these is that, because it is a digital system, static is virtually eliminated.

The AC-3 system provides six channels of surround sound. One of these six channels is a "low frequency effects" channel which provides low frequency audio to a subwoofer speaker that enhances certain on-screen events like rockets taking off and trains passing by. Because of the very limited audio frequency range carried over the "low frequency effects" channel this channel requires a lot less data to convey its audio information than the other, "normal" audio channels. For this reason, many people refer to the low frequency effects channel as only a tenth of a channel, and they refer to the overall ATSC audio system as a 5.1 channel system. Other people might refer to it as a 5 + 1 channel system in order to accentuate the difference between the low frequency effects channel and the other channels.

The 6 channels of ATSC audio are intended to be heard through speakers generally positioned as shown in Figure 28.

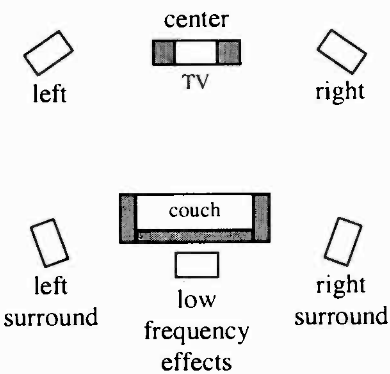


Figure 28: Living Room Layout of Surround Sound System

In addition to the six (or 5.1) channels of surround sound audio, there are a number of other supplemental optional features available with AC-3. These include channels for:

- information for the visually impaired
- information for the hearing impaired
- dialog
- commentary
- emergency information
- voice over information

The *Visually Impaired (VI)* service is a single audio channel used to describe the video scene that it accompanies. Its purpose is to allow visually impaired people to "watch" a television program by receiving periodic audio updates of the on-screen activity, and it normally will accompany the basic (mono, stereo or surround sound) audio service.

The *Hearing Impaired (HI)* service is a single audio channel than contains only dialog, and this dialog may be processed for improved intelligibility by hearing impaired viewers. It is intended to be received simultaneously with the basic (mono, stereo or surround sound) audio service so that the viewer will hear an emphasized version of the dialog for better intelligibility, while still hearing some of the music and effects. Closed-captioning information, which can also be provided for hearing impaired viewers, is transmitted as video data and not as part of the AC-3 audio service.

The *Dialogue (D)* service is intended to carry only the program dialog, without any of the music or sound effects that might be present in the program. This audio service is intended to accompany another audio service, the *Music and Effects (ME)* service, which contains only the music and effects from the program and not any dialog. The reason for having these two separate services available is to enable multiple language versions of the same program to be broadcast. Using this method, a viewer theoretically can receive the video and accompanying music and effects, and then select between the different languages available over the Dialogue service. Because of the complexity involved in creating several separate dialog signals to accompany the same music and effects, this combination is not expected to be used very often.

The *Commentary (C)* service, like the Dialogue service, is intended to accompany the basic audio service (mono, stereo or surround sound, including music, effects and dialog). It allows commentary to be included in a program that "talks over" the main program audio. The digital data stream that makes up the Commentary service includes digital instructions that can lower the level of the main audio when commentary audio is present. In this manner it can be assured that the commentary audio will be intelligible over the main audio.

The *Emergency (E)* service permits the insertion of emergency messages into the audio stream. Whenever an emergency message is present an ATSC receiver will stop playing other audio and play the emergency message. Once the emergency message has finished the other audio will resume.

The *Voice-over (VO)* service serves exactly the same purpose as the Commentary service, except that the Voice-over service takes precedence over the Commentary service. That is, if Voice-over audio and Commentary audio are both present, the volume of the Commentary audio can be reduced by digital instructions in the Voice-over data in order to ensure that the Voice-over audio is heard above the Commentary audio.

Timing

Well, now that we have an understanding of television video, we can begin to understand the importance of timing in a television facility.

You may have heard the expression "timing is everything." Well, this often-used expression may very well have first been uttered by a TV engineer. Timing is a very important aspect of TV engineering, and nowhere is the importance of timing more evident than in a television studio.

The reason that timing is so important is really very simple. If you have two video signals arriving at a switch, and they are not timed correctly, then whenever you switch from one video signal to the other there will likely be a noticeable “bounce” or “jump” in the output video signal. As an example of why this occurs, consider the two video signals pictured in Figure 29. Note that each new field in Signal 2 starts midway through a field in Signal 1 (and vice versa). If the output of the switch is carrying Signal 1, and the switch is changed at the point designated by the arrows to cause the output to carry Signal 2, then only half a field’s worth of video will be available from Signal 2 before another vertical blanking interval is encountered. The result is, at the output of the switch, the entire second field in Signal 1 will be displayed on the TV set followed by only the second half of the second field of Signal 2. This will be followed by the complete third field of Signal 2, and the complete fourth field, and so on. Each field gets displayed correctly except for the half field from Signal 2 that gets displayed immediately after the switch.

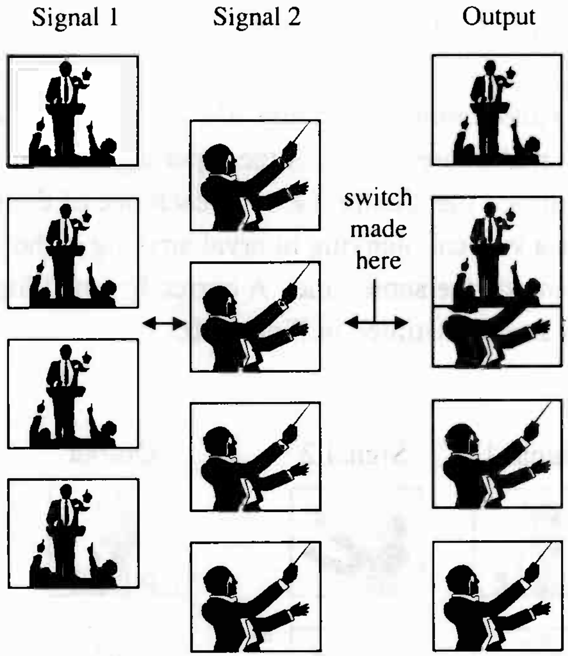


Figure 29: Example of Improperly Timed Video Switch

Because the half-field problem happens so fast (within 1/60th of a second) it does not appear to the naked eye as much more than a "bounce" or "blip" in the output video signal. As video problems go, this is not the worst thing that could happen. (For example, complete loss of video would be much worse.) However, little bounces in the final video product — just like "pops" and "clicks" in audio switching — make a broadcast seem unprofessional to the viewer, so they need to be avoided.

The way to avoid timing problems like the one illustrated above is to make sure that all of the input signals to a video switch are timed identically (*i.e.*, that each one of the input signals has a vertical blanking interval arriving at the switch at *exactly* the same time. A correctly timed Signal 1 and Signal 2 are illustrated in Figure 30.

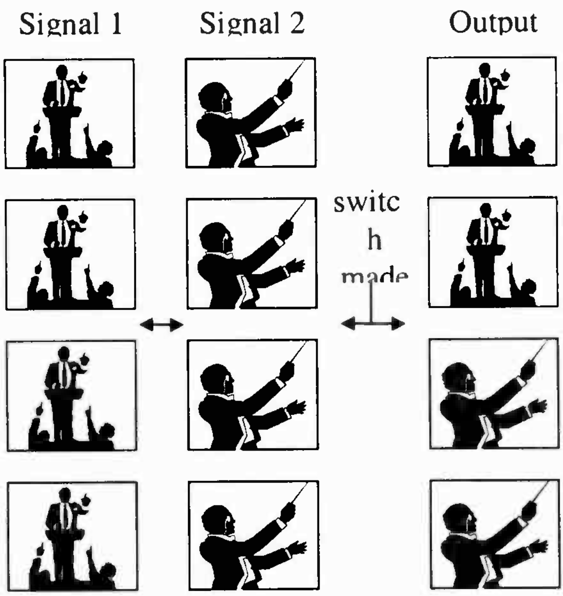


Figure 30: Example of Properly Timed Video Switch

How does one insure that all of the video signals arriving at a switch are timed exactly right? Well, there are basically two methods for achieving this. The first is to insure that all of the video sources that are feeding into the switch (cameras, video tape recorders, character generators, etc.) are sending vertical synchronization information at exactly the same time, and that the cables connecting the switch to these video sources are exactly the same length. The second, and more common method for addressing the timing issue in modern television studios is to use the *genlocking* circuitry which is included in most modern television studio equipment. Genlocking circuitry enables a video device, such as a camera, video tape recorder, etc., to lock its synchronization pulses to (*i.e.*, synchronize them with) synchronization pulses received from another device. Typically, this "other device" is a *master synch generator*, a single synch pulse generator that provides synchronized synch pulses to multiple pieces of equipment. Genlocking each piece of equipment to the master synch generator, ensures that each of the synch pulses coming *from* each piece of equipment will, themselves, be synchronized. Once it is known that the synch pulses coming from each camera, video tape recorder, etc. are synchronized, then ensuring that these signals are synchronized when they arrive at a video switch is simply a matter of ensuring that the cables from the various pieces of equipment to the switch are the same length.

Audio-Video Synchronization

The term *audio-video synchronization* refers to the fact that the audio in a television program (*e.g.*, a spoken voice) must match up — or be synchronized — with the video picture that accompanies it (*e.g.*, a person's mouth movements). This is not a trivial engineering task — a lot of effort goes into maintaining synchronization throughout a television station.

The industry practice for audio/video synchronization stipulates that a television audio signal should never be more than 25 milliseconds (25 thousandths of a second) ahead of, nor more than 40 milliseconds behind, its associated video signal. In order to insure that their facilities meet this requirement, television broadcast engineers will often use a device, like the ones alluded to above, known as a *delay*. Delays come in two forms — *audio delays* and *video delays*. The decision about which one to use is based on which signal (audio or video) is leading the other.

If the video signal is leading (ahead of) the audio signal then a video delay can be used to bring the two signals into synch. If the audio is leading, then an audio delay is needed.

A delay is simply a device that takes a signal (audio or video) as an input and stores it for a very brief moment before sending it out. Even if a television studio is well synchronized without the use of any delays, it is usually very important to have an audio delay unit and a video delay unit on hand and available for use anyway, in case a signal being fed into the studio from outside is out of synchronization and needs to be corrected. Figure 31 illustrates a situation where an audio delay would be needed to correct an incoming signal.

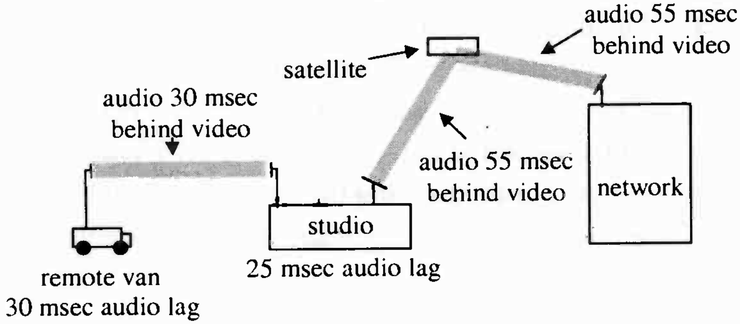


Figure 31: Example of Audio-to-Video Synchronization Problem

In Figure 31, a television signal is being sent from a remote newsgathering van back to a local studio, and then from the local studio through a satellite to a TV network facility where it will ultimately be fed to the entire country. In this example, the audio sent from the remote van is 30 milliseconds behind the video from the van, which is close enough to conform to the industry practice and not be distracting to the viewer. The audio and video signals are then fed into the studio facility which, itself, produces a 25 millisecond delay in the audio with respect to the video. While both the remote van and the studio, by themselves, conform to the industry practice, when added together they produce a signal that is out of compliance with industry practice, and which will be noticeably out of synch to the viewer. The audio that ultimately arrives at the network studio is 30 milliseconds + 25 milliseconds, or 55 milliseconds behind the video. In this case, a video delay would be needed to delay the video by at least 15 milliseconds in order to bring the entire system into compliance with industry practice and avoid noticeable synchronization problems on air.

Throughout our discussion on timing and synchronization we have alluded to various video input devices such as the video tape recorder (VTR), camera and character generator. It seems appropriate, therefore, to wrap up our discussion about the engineering aspects of television studios with an overview of the video recording and playback equipment that is commonly found in today's television studios.

Video Tape Recorders

Video tape recorders, or VTRs, are a lot like their audio counterparts. They all use magnetic tape as the media on which the program material is stored, and different types of machines can use tapes with different types of magnetic coatings on them. The magnetic tape is sometimes stored on open reels, while other times it is housed in plastic cassettes. VTRs, like their audio counterparts, also come in both analog and digital form.

On a typical video tape, there are four or five channels of information recorded. One is the video picture information, two more are the left and right audio information, a fourth is the control codes, and the fifth is the time code (though time code is sometimes recorded in the video signal as part of the vertical blanking interval).

It should be pretty obvious what the audio and video channels are used for, so we will not discuss them in much detail here. It is worth noting, however, that the video information takes up, by far, the most amount of space on the tape. Typically, the video information is recorded in a wide band in the center of the tape and the audio information and control codes are stored towards the outer edges.

The control codes on a video tape are most often used by the VTR playing back the tape to signal other devices that they are done playing. In the same manner that audio cart machines are hooked together in a radio studio, a series of VTRs can be hooked together in a television studio to automatically play back their tapes in a rapid-fire sequence. This is a very important function of these machines because commercial breaks have a specific amount of time allotted to them, and the recorded commercials that are played from the tape machines are designed to fit exactly into these breaks. If each successive playback machine is not started right on cue, it might result in the last commercial in the break not having enough time left to fit in — not to mention the fact that not starting each tape right on time will look sloppy to the home viewer.

The standard time code used in television video tape recording is called *SMPTE time code*. SMPTE time code was developed by the Society of Motion Picture and Television Engineers (SMPTE), and it is a digital code that is recorded on the video tape which identifies how many hours, minutes, seconds, and video frames have passed since the beginning of the recording. SMPTE time code is very useful in editing applications because it allows a person who is producing a single video tape from two or more pieces of recorded material to identify the exact frame where dubbing of the first tape onto the recording device should stop and dubbing of the second tape onto the recording device should begin. SMPTE time code also has many other useful purposes. It can, for example, also be used to synchronize a separately recorded audio recording with a video recording.

Due mostly to the large amount of recording that news crews do in the field, cassette-style VTRs are the most widely used today. Digital VTRs, where the audio and video information are stored on the tape as a series of 1s and 0s, have become very popular in large part because they allow a recording to be copied from tape to tape many times (up to about 30) with very little degradation of the program material. The following paragraphs provide an overview of the most common VTR formats in use today.

1-inch C format

The one-inch, C format VTR is an analog recording and playback device which uses one-inch wide videotape that is stored on open reels. This format permits storage on videotape of three audio channels, one video channel, one control channel, and one synchronization channel in the manner illustrated in Figure 32.

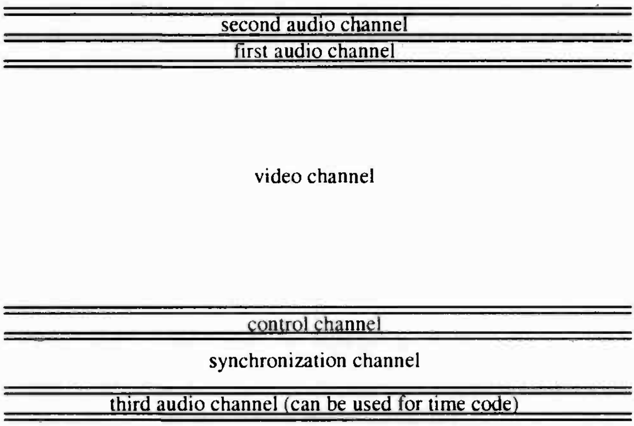


Figure 32: Storage of Information on 1-inch C Format Videotape

D-1 format

The D-1 format VTR is a digital recording and playback system which uses 19-mm wide videotape stored in cassette cartridges. There are three standard cassette sizes for the D-1 format — L, M and S (for "large," "medium" and "small"). L-size cassettes hold up to 76 minutes of information, M-size cassettes hold up to 34 minutes, and

S-size cassettes hold up to 11 minutes. In the D-1 format, both audio and video information are recorded in the wide center stripe on the tape, and the cue, control and time code information is stored on narrower outside tracks. The video information in the D-1 format is stored in component form. (Remember, this means that instead of a single composite video input signal there are three separate component signals — in this case one monochrome signal and two other signals carrying the color information.) An illustration of where information is stored on a D-1 tape is provided in Figure 33. All of the information stored on a D-1 tape is digital.

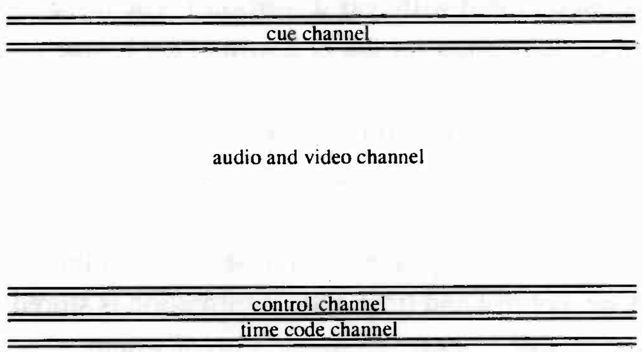


Figure 33: Storage of Information on D-1, D-2 and D-3 Format Videotape

D-2 format

The D-2 format VTR, like the D-1 version, is a digital recording and playback system which uses 19-mm wide videotape stored in cassette cartridges. The three major differences between the D-2 format and the D-1 format are that the D-2 format uses composite video instead of component video, the cassettes used in the D-2 system are longer, and the tape used in the D-2 system is coated with ground metal powder instead of metal oxide. (Recall from the section on audio tape recording earlier in this Part that tapes coated with metal powder are generally better than tapes coated with metal oxides because they enable signals to be stored on them with less distortion.) The three standard cassette sizes for the D-2 format are L-size (208 minutes), M-size (94 minutes), and S-size (32 minutes). In the D-2 format, like in the D-1 format, both audio and video information are recorded in the wide center stripe on the tape. However, as noted above, the video is stored in composite form instead of component form. Cue, control and time code information is stored on narrower outside tracks. An illustration of where information is stored on a D-2 tape is provided in Figure 33. As with the D-1 tape, all of the information stored on a D-2 tape is digital.

D-3 format

The D-3 format is essentially the same as the D-2 format except that it uses ½-inch tape instead of 19 mm tape. One-half inch is equal to 12.7 mm, so the ½-inch tape is narrower than the 19 mm tape. The narrower tape and smaller cassettes of the D-3 format enable the equipment associated with it to be of a smaller size and lighter weight. This makes D-3 equipment particularly useful for electronic news gathering (ENG) and other applications that require the use of portable VTRs.

M, Betacam, M-II and BetacamSP formats

The M, Betacam, M-II and BetacamSP formats use ½-inch videotape to record component video information in analog form. It was the success of these small size VTR formats that led to the development of the D-3 format.

Digital Betacam, DVC Pro formats

These are the digital successors to the small size VTR formats like Betacam.

Character Generators

A *character generator* is basically a simple computer system whose capabilities are limited to having characters typed on its screen from a keyboard, fed to its screen through a communications port, or read to its screen from a disk drive. The user can select both the background color of the character generator's screen and the color of the characters that are displayed on the screen. In additional to allowing alphanumeric characters to be displayed, character generators also allow other digital images to be displayed — such as logos.

An example of how keyboard input to a character generator might be used would be when a news reporter's name is added to the bottom of the screen during a newscast. An example of how an external source might supply data to a character generator through the data port would be when a warning about a weather emergency is received by a television station's Emergency Alert System equipment, and the warning is automatically fed out over the air by using the character generator to scroll a text message across the screen.

A typical personal computer may be used as a character/image generator if it is equipped to output NTSC video signals. The video output on the vast majority of home computer systems does not conform to the NTSC standard, but add-on boards are available for adding this capability.

Television Cameras

The television camera, as one would expect, performs essentially the opposite function of a television receiver. Whereas the receiver takes an incoming video signal and uses it to "paint" a series of rapid-fire still photographs (30 frames per second, 2 interlaced fields per frame) on the phosphor coating that covers the inside of the TV screen, the camera takes an incoming video image, coverts it to a corresponding electric image which has — instead of variations in light level — variations in electric charge-density level, and then scans this electric charge image to produce an electronic version of the video image. The device that did the scanning of the image in older television cameras was called a *camera tube*, or a *pickup tube*. In these tubes, an electron beam actually scanned the electric charge on the inside surface of the tube to create the electronic version of the video image. In modern cameras, the device that does the scanning is called a *charge coupled device*, or *CCD*.

The CCD used in a television camera is a pair of electronic matrices. Both of these matrices have a very large number of individual charge "cells" on them. Each of the cells in the first matrix is charged up to a level that is proportional to the amount of light that is hitting it through the lens of the camera. All of the individual charges from each one of the cells in the first matrix are transferred to corresponding cells in a second matrix which is shielded from light. The circuitry in the camera then scans the second matrix to produce an electronic version of the video image.

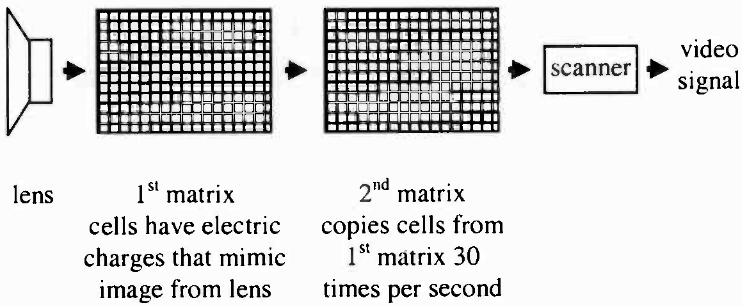


Figure 34: The CCD Imaging Process

The reason that the charges in the first matrix must be transferred to a second matrix before they are scanned by the camera circuitry is that the charges in the first matrix are constantly changing as the image through the lens changes, so a "snapshot" of the first matrix must be created in the second matrix to ensure that the correct fixed image is converted to an electrical signal by the camera.

The best-performing CCD color cameras actually have three CCDs inside of them — one to scan the electric charge image created by the red light coming through the lens, one to scan the image created by the green light, and one to scan the image created by the blue light. A three-CCD color camera will have a prism inside of it which splits the incoming optical image into three different color beams (red, green and blue). Each of these beams is then fed through an additional filter which removes any remaining unwanted colors from each of the signals. The purified individual color signals are then fed into their own individual CCDs. This type of color separation system is called a *three-imager prism type separation system*.

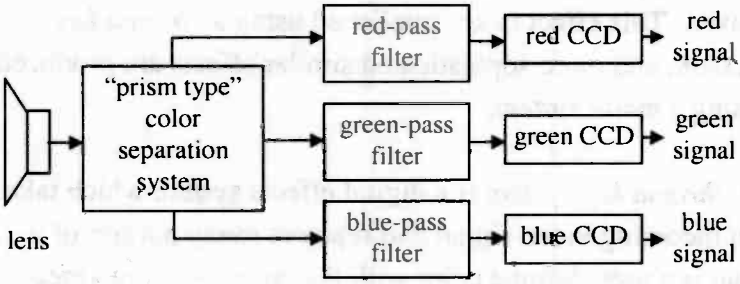


Figure 35: Basic Operation of a Color CCD Camera

The three separate color signals that come from the three separate CCDs are referred to together as a *component video signal.* This is because each of the individual colors from the full-color picture is transmitted as an individual component. The colors are eventually combined together to form a single color video signal at which point the single signal is referred to as a *composite video signal* (because it is a composite picture made up of the three components.)

Chroma Key and Matte Systems

Many times it is desirable to combine two video signals in such a manner that one appears to sit on top of the other. This occurs quite frequently in weather forecasts when the weather forecaster is made to appear standing in front of, for example, a moving satellite picture showing cloud cover. This effect is accomplished using a *chroma key system*, and more sophisticated similar effects are produced using a *matte system.*

A *chroma key system* is a digital effects system which takes an incoming video signal and replaces every portion of it that is a user-defined color with the corresponding video from a second incoming video signal. The result is an output signal which has combined selected portions of the two input signals. Let's look at the various specific aspects of a typical weather forecasting set to get an understanding of how a chroma key system works.

On a typical weather forecasting set, the weather forecaster stands in front of a large, blank wall that is painted one solid color, typically blue or green. It is important that this background color not match the color of any of the forecaster's clothing, hair, skin, etc. A camera is focused on the weather forecaster with the wall in the background, and the video from this camera provides the first input signal for the chroma key system. The second input signal for the chroma key system is simply the satellite video image. The chroma key system is then instructed to take every portion of the first input signal that matches the color of the wall and replace it with the corresponding portion of the second video signal. Since the wall should be the only thing in the first video signal that is the same color as the wall, this replacement creates the illusion in the final product that the weather forecaster is standing in front of the satellite video. That's why it is important that the forecaster's clothing, hair, skin, etc., not match the wall, or those elements would become part of the illusion, as well.

A chroma key system makes the weather forecaster's job a little bit tricky because the weather forecaster wants to appear to be pointing to portions of the satellite video in the final product. In order to do this, however, the forecaster must actually be pointing to places on the blank wall. Typically, there is a monitor showing the composite picture of the forecaster and the satellite video somewhere just out of range of the camera that is focused on the forecaster. The forecaster looks at this monitor while doing the forecast to insure that he/she is pointing to the correct portion of the satellite video.

A matte system operates in essentially the same manner as a chroma key system — except that the matte system can replace *multiple* colors in the first video signal with the video from the second video signal.

Video Mixing Board

A *video mixing board* (sometimes referred to simply as a *master control board*) performs essentially the same function as the audio mixing board that was discussed earlier in the radio section. The main difference between the two, of course, is that the video mixing board is mixing audio *and* video signals instead of just audio signals. Another important difference between audio mixing and video mixing, which we have already covered, is the fact that the timing of the video signals being mixed is much more important than the timing of audio signals being mixed because switching between two video signals whose vertical synch pulses are not precisely synchronized will usually result in a "bounce" in the output video signal.

A good way to think about the mixing and switching of video signals, in comparison with the mixing and switching of audio signals, is to think of a completely black video signal as very quiet (but not completely silent), and a completely white video signal as very noisy. The reason for this is that the darker a video image is, the lower the voltage of its corresponding NTSC video signal and, conversely, the lighter the image the higher the voltage of its corresponding video signal. Although the completely black signal has a low voltage, it does *not* have zero voltage. The zero voltage portions of an NTSC video signal are reserved to the horizontal and vertical blanking intervals.

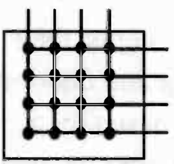
Because the low voltage black signal does not have zero voltage, it cannot be added to another video signal without having *any* effect on the other signal. If a completely black screen and a "normal" video image are combined together, the result is a somewhat dimmer looking version of the "normal" video. Conversely, if a completely white screen and a "normal" video image are combined together, the result is a brighter version of the "normal" video.

Modern video mixing boards allow many different effects to be used while switching between input signals. These include wipes, fades and blinds, among others. They also allow the board operator to create a "picture-in-picture" effect by inserting one or more video images inside another.

Just as is the case in a radio studio, after a television signal leaves the master control board there are several important pieces of equipment that it travels through before being delivered to the transmitter. This equipment can be broken down into three categories: distribution amplifiers and routing switchers, audio processing equipment and Emergency Alert System (EAS) equipment. We are not going to discuss audio processing equipment here since it was covered earlier in the section on radio studios. The video portion of a television signal is not processed (compressed or expanded) to any significant degree because, in an NTSC video signal, the dynamic range (difference between the highest level of signal and the lowest level of signal) is a very important aspect of the signal.

Distribution Amplifiers and Routing Switchers

Usually, the output of the master control board (the station's on-air program) will be fed into a distribution amplifier which, in turn, feeds several "copies" of this signal to other parts of the studio. (Recall that a distribution amplifier allows a single input signal to feed multiple sources with "copies" of the signal that are equal in quality to the original.) These "other parts of the studio" might include recording equipment, or simple monitors for viewing the outgoing program, and they will usually include at least one routing switcher.



Each output of the routing switcher can be connected to any one of the inputs at any given time

Figure 36: A Typical Routing Switcher Configuration

A *routing switcher* is a device that has multiple inputs and multiple outputs, and a user of any particular output channel can select to have it connected to any particular input channel at any time. The great advantage of a routing switcher is that it allows a number of different signals to be fed to a particular place — such as an input channel on a video mixing board — through only one connection. In other words, instead of wiring all of the signals going into the routing switcher directly into the mixing board (which would take up numerous input channels on the mixing board) each of the inputs is routed through the switcher and few output channels from the switcher are connected to the mixing board, giving the board operator access to *all* of the switcher input channels without taking up more mixing board inputs than necessary.

As is the case with audio systems, there are now *video servers* available for storing and distributing digital video signals. A video server is, basically, a powerful computer that is capable of receiving digital video signals from video sources and distributing them to multiple locations. The major difference between a video server and a distribution amplifier is that a video server is capable of storing digital video information on its disk drives for later retrieval while the distribution amplifier is not equipped to store any video.

Emergency Alert System Equipment

As noted in the earlier section on radio studio equipment, Emergency Alert System (EAS) equipment must be installed somewhere in a broadcast station's air chain. This equipment will typically be installed between the output of the master control board and the transmitter. The only significant difference between the EAS equipment used in a TV studio in comparison to the EAS equipment used in a radio studio is that the equipment used in the TV studio must have the ability to interrupt *both* audio *and* video, or at least to interrupt audio and insert a video text message over top of a portion of the normal video.

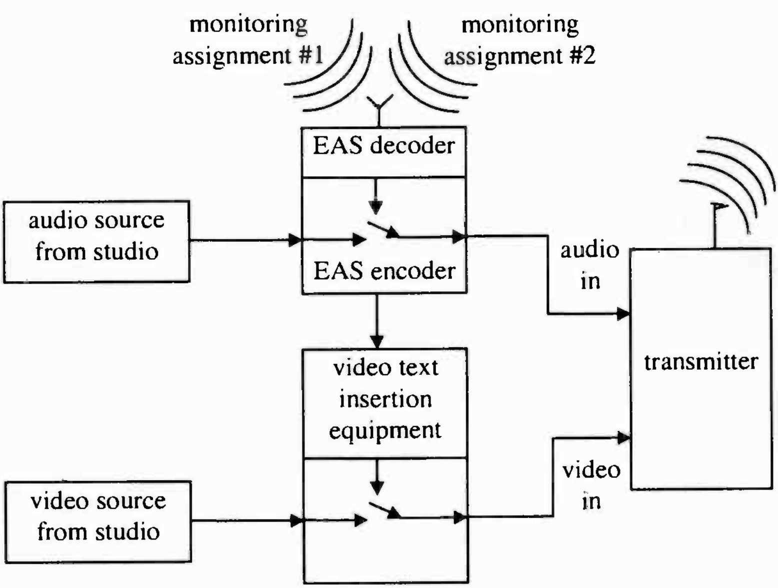


Figure 37: EAS Equipment in a TV Station Air Chain

Part II: The Studio-Transmitter Link

The studio-transmitter link, or STL, is the means by which the program material produced in the studio is transported to the transmitter for broadcast. If the station's transmitter is located adjacent to its studios, then the STL might simply be a cable, or set of cables, stretching from the studios to the transmitter. More often, however, the transmitter is located at a remote location, such as the top of a mountain or tall building. When the transmitter is located at a great distance from the studio, there must be some means of transporting the program material over the property in between. Though nearly any means of transmitting an electronic signal can be used as an STL, most broadcasters use one of two primary methods.

The first of these methods is a microwave link. Microwave STLs involve the installation of a microwave transmitter and an associated antenna at the station's studios, and a microwave receiver and associated antenna at the transmitter site. The antennas used for most microwave STLs are called *parabolic reflector antennas*. They are shaped somewhat like a typical dish antenna but are generally rectangular in shape.

There are several important issues that stations using microwave STLs need to consider. One is that the transmitting antenna at the station's studios needs to have a good "view" of the receiving antenna at the transmitter site.

This means that there cannot be any major obstructions such as mountains, tall buildings, or even heavy foliage, between the two antennas which would block, or partially block, the signal being sent to the transmitter site.

If there are obstructions between the studio site and the transmitter site which will block the microwave STL signal to any significant degree, then the broadcaster generally has three options to consider. The first is to relocate the studio end of the STL, the transmitter end, or both. This can be done by actually moving the studio and/or transmitter site, or by simply moving the STL antennas and making additional arrangements to get the STL signal from the studios to the microwave transmitting antenna and/or from the microwave receive antenna to the transmitter. The second option is to raise the height of the microwave transmitting antenna and/or the microwave receive antenna to that they can each "see" each other over whatever obstructions are in between them. In some cases, it is simply not possible, or financially practical, to do either of these things and the only option left for the broadcaster is to find a means other than the use of microwave signals to get the program material from the studios to the transmitter.

The most common means of getting program material to the transmitter without using microwave transmitting equipment is to use leased telephone lines. The telephone company is capable of providing broadcast quality links between studios and transmitters. Broadcasters who use such links feed their signals into a telephone company-provided box or boxes at the studio facility, and they retrieve them from a telephone company terminal at the transmitter facility. The main advantage of a leased telephone line is that it enables the broadcaster to get program material from the studio to the transmitter without having a line-of-sight path. In most other respects, a leased line is less desirable than a microwave link.

One of the disadvantages of the leased line is cost. Cost varies depending on how much capacity is being leased, and how far it is from the studio to the transmitter, but a typical stereo connection between a radio studio and a radio transmitter can run about $200 per month. A microwave STL system will cost about $10,000 which is about four years' worth of leased line payments. So, from a long-term perspective, the microwave link is financially more attractive than a leased line. (Note that the scales can be tipped back in favor of the leased line, however, if the broadcaster must lease additional tower space on which to install the microwave STL antennas.)

Another disadvantage of the leased line is control in emergency situations. If, for some reason such as a natural disaster or a construction accident, the connection between the studio and the transmitter is broken, the broadcaster using the leased line is at the mercy of the telephone company repair crew. On the other hand, broadcasters with microwave STLs have control over their own destinies when disasters strike because it is up to these broadcasters to maintain their own equipment.

Regardless of what path the signal takes to get to the transmitter (simple cable, microwave link, or leased phone line) there is one other quality of the STL that must be considered — whether it is analog or digital.

If the program material produced at the studio is analog, and the transmitter accepts an analog input, then an analog STL can be used without the need for any additional equipment. If, on the other hand, the program material produced at the studio is analog, and a *digital* STL is employed, then an additional piece of equipment called an *analog-to-digital converter*, or *A/D converter*, is required at the studio end of the STL. The same is true at the transmitter end — if the transmitter only accepts analog inputs, and a digital STL is employed, then a *digital-to-analog* converter (*D/A converter*) is needed at the transmitter site.

The trend in broadcasting, as in most telecommunications-related fields, is toward digital equipment. Broadcast studios that have been converted to digital can connect to a digital STL without the need for an A/D converter. However, if the data rate of the digital signals used in the studio is faster than the data rate of the digital STL, then a device to perform some form of *data compression* will be needed at the studio end of the STL. This compression device takes the incoming digital data stream from the studios and encodes it in a more efficient manner, thus enabling the data stream that it sends out to be at a lower rate than the incoming signal. Some forms of data compression are *lossless*, meaning that the original, uncompressed data stream can be reconstructed in its entirety from the compressed data. Other forms of compression are *lossy*, meaning that some of the data in the original material is lost for good in the compression process. The advantage of lossy compression is that it is generally more efficient (*i.e.*, it permits the compressed data stream to be sent to the transmitter at a lower rate, and therefore at less expense). The disadvantage, of course, is that some of the original data is lost for good. There are forms of lossy compression, however, for which the loss of data is not detectable to the human ear (and eye, in the case of television) — so it is a very attractive option for many broadcasters.

Part III: The Transmitter Site

From an engineering standpoint, the transmitter sites for all types of broadcast facilities (AM, FM and TV) are generally very similar. As shown in Figure 38, each transmitter site must have a transmitter to create the high-powered signal for broadcast, a transmission line to carry the signal from the transmitter to the antenna, and an antenna. Although AM, FM and TV transmission facilities are *generally* very similar, there are some significant differences between them — most notably the different types of antennas that are used. We will cover antennas later, after we talk about the transmitter and the transmission line — but first let's talk about the arrival of the signal to be transmitted at the transmission site.

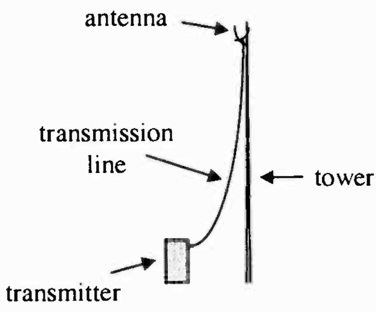


Figure 38: Basic Components of Transmitter Site

A signal arrives at the transmitter site via the studio-transmitter link, or STL, (see Part II for a discussion of the STL). The STL can be analog or digital. If it is digital, and the transmitter inputs are analog, then the signal must be converted to analog using a digital-to-analog (D/A) converter.

Once the signal has arrived at the transmitter site and gone through any necessary converters, it is then in one of two forms — a component signal, or a composite signal. A component audio signal simply consists of the same audio signals that travel around the studio. These signals are essentially the same (except for a slight difference in signal level) as those that come out of the "line out" terminals on home electronics equipment. A component video signal consists of three separate signals — usually one for red picture information, one for green, and one for blue; or, one for monochrome picture information and two used to add color to the monochrome signal. Component video signals are used in a number of studio applications, but it is unusual for a component video signal to be fed from the studio to the transmitter site because of the additional cost and complexity involved. Usually the video information sent to the transmitter site is composite. The NTSC video standard discussed in Part I is a composite signal (because it includes all of the video information in a single signal), and it is in the NTSC composite format that video information is usually sent to an analog television transmitter site from the studio. The ATSC video standard discussed in Part I is also a composite signal, and a digital television signal can be delivered to the transmitter from the studio in this format.

If a component signal arrives at the transmitter site over the STL (as is often the case with audio) then it must first be converted to a composite signal before it can be fed to the transmitter.

So, you ask, what is a composite signal ... and why is it needed? Well, a *composite signal* is, basically, a single signal which contains all of the information from multiple component signals. A simple example of a composite signal is a monophonic audio signal. Most recorded music nowadays has been recorded in stereo. If a radio station is broadcasting in mono, and it is playing a stereo recording, it will want its monophonic signal to include *both* the left and right channels from the recording. Consequently, it needs to take its component signals (the left and right channel audio) and add them together to make a single signal that contains both the right and left channel information — a composite signal.

A monophonic radio signal is a simple example of a composite signal. Even radio stations that broadcast stereo music need to create a composite signal. The reason is simple — even though these stations are broadcasting *both* left and right channel audio information, they only have *one* channel over which to transmit it (their carrier frequency, *e.g.*, 92.7 MHz). So, they need to somehow combine the two audio signals they have into a single signal in such a manner that will permit this single signal to be separated and heard as two distinct left and right channels by a stereo receiver — but at the same time will allow monophonic receivers to hear a single, composite signal that contains *both* left and right channel information.

Stereo sound is very prevalent at FM radio and TV broadcast stations, and it is used to a somewhat lesser degree in AM radio stations. No matter what type of broadcast station is involved (AM, FM or TV), all stereo broadcast systems use the same method of coding and decoding the left and right audio channel in order to insure that *both* stereophonic and monophonic receivers are able to play their audio.

Because monophonic receivers must still be able to receive a signal from a stereophonic transmitter, it is not sufficient to simply transmit a left channel signal and a right channel signal. Instead, a main program channel must be transmitted that can be used by a monophonic receiver (*i.e.*, the main program signal must consist of both the left and right audio channel) and a stereo program channel must be transmitted that can be coupled with the main program channel to produce left and right program material at a stereo receiver. Figure 39 illustrates the method that broadcasters use to achieve this objective.

Note how, in Figure 39, a "main" signal can be transmitted which consists of *both* the left and right audio channels. Then, in addition, a supplementary signal is transmitted which, when added to and subtracted from the main signal, can be used to produce stereo sound at the receiver.

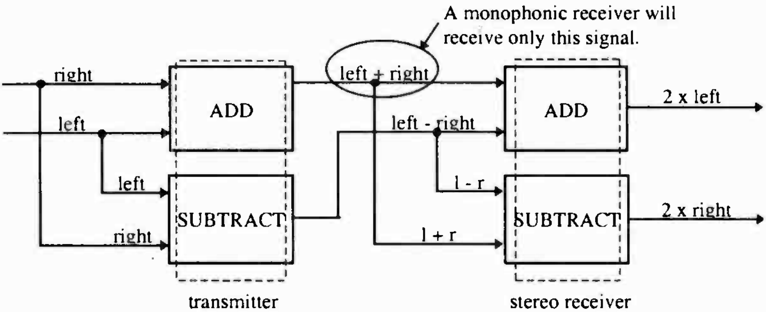


Figure 39: Block Diagram of Stereo Transmission System

Before we lose track of where we are, let's review. Thus far in this Part we have learned that the information to be transmitted by the transmitter arrives at the transmitter site via the studio-to-transmitter link (STL). We have also learned that, if the signal(s) which arrive at the transmitter site are not in composite form, then they must be converted to composite form before being fed to the transmitter. We know that conversion to composite form is necessary because each broadcaster has only one channel over which to transmit programming.

The Stereo Generator

Whether we are talking about an FM or TV station, the *stereo generator* is the piece of equipment that is used to create the composite stereo signal. A stereo generator will generally have two inputs — one for left channel audio and one for right channel audio, and it will have a composite output which is a one-wire connection that carries the composite signal. The composite output of the stereo generator is the left + right signal, the stereo pilot (19 kHz for FM, 15.734 kHz for TV) and the left - right signal centered on a frequency that is two times the stereo pilot frequency.

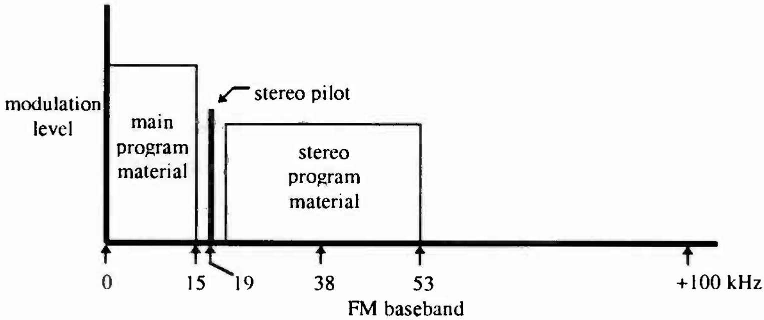


Figure 40: Composite Output of FM Stereo Generator

AM stations do not use stereo generators, *per se.* The actual combining into a "single wire signal" of the left + right and left - right signals in an AM stereo system is done by the exciter. The composite output of the AM stereo exciter is the left + right signal and the left - right signal, the latter of which is slightly shifted in time (delayed behind) the left + right signal to insure that it does not interfere with monophonic AM receivers. (Stereo AM receivers know to look for the time-shifted left - right signal.) The term used to describe this time-shifted signal is *quadrature amplitude modulation.* It will be discussed in a little more depth in the section on subcarrier generators.

If composite audio is fed over the STL, then the stereo generator will be located at the studio site. If component audio is fed over the STL, then the stereo generator will be located at the transmitter site.

Typically, the output of the stereo generator is fed into the exciter (see the section on the exciter below). For stations that do a lot of audio processing, however, there is sometimes some additional audio processing equipment installed between the stereo generator and the exciter, usually to aid in efforts to make the station's audio sound louder.

Subcarrier Generators

Subcarrier generators are devices used to add additional information to a broadcast signal which may or may not be associated with the broadcaster's main programming. For example, it is very common for FM radio stations to transmit paging information via subcarrier. Although less common, it is also possible for an analog television station to use a subcarrier to transmit a second audio signal, in addition to the audio that accompanies its video signal. Digital television stations can also transmit additional audio, video, or data signals – however, the way digital television signals are constructed and transmitted is completely different from the way analog radio and television signals are constructed and transmitted.

Technically speaking, the capacity to send additional audio, video and/or data over a digital television channel is not made available through the use of subcarriers. Instead, the digital television signal, itself, is simply a series of digital bits, some of which can be allocated for supplementary audio, video and/or data. We will talk about the transmission aspect of digital television a bit later.

The stereo generator (see earlier section) is a subcarrier generator which generates the stereo pilot signal and the stereo subcarrier.

A *subcarrier* is a special type of carrier. It has all of the characteristics of a carrier, except that it, itself, must be added to the "host" carrier in order to be delivered to a receiver. Subcarriers can exist in AM, FM and TV transmission systems. However, AM radio channels have very little extra room for subcarriers after the audio is added so subcarriers are used very little — if at all — in AM radio systems. On the other hand, subcarriers are very common in FM radio and TV because there is plenty of extra room for them to be added. Although the limitations of the AM channel generally prevent AM broadcasters from using subcarriers, these broadcasters can still transmit "subcarrier-like" signals using *quadrature amplitude modulation (QAM)*, which will be explained shortly.

One of the most common subcarriers in use today is the stereo pilot. The stereo pilot is an *unmodulated subcarrier* — that is, a subcarrier that does not have any additional information added to it. The main purpose of the stereo pilot is to tell receivers that the host FM or analog TV station is broadcasting in stereo. If an FM or analog TV station does not transmit the standard stereo pilot signal, then the receivers that receive its signal will assume that the station is broadcasting a monaural program — and they will not split the received audio into a left and right channel. Figure 41 shows the 19 kHz stereo pilot in the baseband spectrum of a standard FM broadcast station. (For the rest of this discussion about FM/TV subcarriers, we will use the *FM baseband* signal for illustrative purposes. The analog TV aural baseband signal is essentially the same as the FM baseband signal, so if you understand how subcarriers are added to FM signals you also understand how subcarriers are added to analog TV aural signals.)

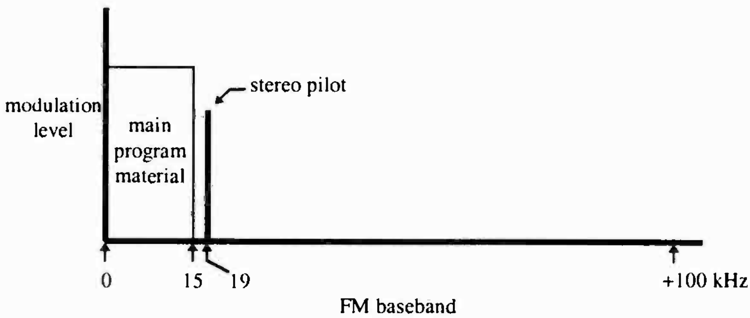


Figure 41: FM Monophonic Signal with Stereo Pilot

To understand the figures being used here to illustrate FM subcarriers, it is necessary to understand what the *FM baseband*, or *FM composite baseband* is. The FM composite baseband signal is the signal that comes out of the stereo generator (and any subcarrier generators, if they are used). From 0 to about 15 kHz, it contains the same range of audio signals that you will find on a typical equalizer used in a home stereo system. What makes the composite baseband signal special is that there are additional signals added to the audible ones, signals that are just above the range of human hearing and which therefore cannot be heard by the human ear. These signals are used to transmit the encoded stereo information, and other additional information, in the manner described below.

The stereo pilot, in addition to being used to alert receivers to the fact that a station is transmitting in stereo, has another very important purpose — its harmonics can be used as other subcarriers, as needed. *Harmonics* are a "side-effect" of radio transmissions. They are extraneous signals, produced on frequencies that are separated from the main frequency by multiples of itself. The second harmonic of 19 kHz is (2 x 19 kHz), or 38 kHz. The third harmonic of 19 kHz is (3 x 19 kHz), or 57 kHz.

In the FM transmission system, the program material necessary for monaural transmission is transmitted on (i.e. modulated onto) the main carrier without the use of a subcarrier. The stereo pilot is the 19 kHz subcarrier, and the program material necessary to deliver stereo programming (the "left minus right" signal) is modulated onto the second harmonic of the 19 kHz stereo pilot (the 38 kHz subcarrier) as shown in Figure 42.)

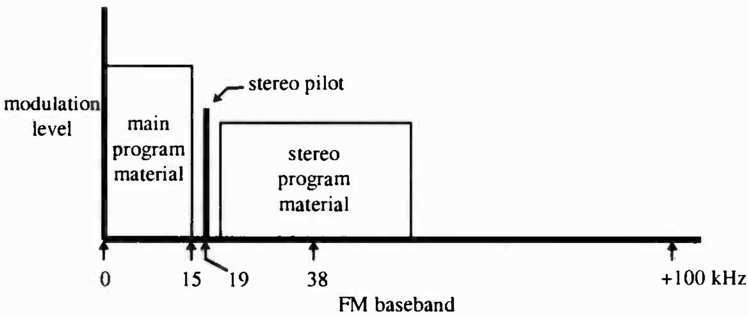


Figure 42: FM Stereophonic Signal

As Figure 42 indicates, the monaural program material occupies the baseband spectrum from 0 to 15 kHz. The stereo pilot appears at 19 kHz, and the stereo program material occupies the baseband spectrum from 15 kHz below, to 15 kHz above, the 38 kHz *sub*carrier, which is the spectrum from 23 kHz to 53 kHz in the FM baseband. (*Note:* As shown in Figure 42, the stereo program material occupies the spectrum from 15 kHz below to 15 kHz above its 38 kHz subcarrier. The *main* program material, on the other hand, only occupies the spectrum from 0 to 15 kHz in the FM baseband — it does not extend down to -15 kHz — because, in the baseband, the main program material is not modulated onto any carrier or subcarrier. It is only when the program material is modulated onto a carrier or subcarrier that the "plus and minus" effect occurs. For a little more discussion on this "plus and minus" effect, see the section on the exciter which appears after this one.)

Figure 42 illustrates the fact that an FM stereo signal actually only occupies a little more than half of the baseband spectrum allocated to it. The excess channel capacity that is left over presents a significant opportunity for FM broadcasters to generate additional income by leasing out some, or all, of their excess channel capacity to subcarrier service providers. The same is true for TV broadcasters, though the subcarrier space available to them is slightly different than the subcarrier space available to FM stations.

The three most common FM-band subcarriers in use today are the 57 kHz, 67 kHz and 92 kHz subcarriers. Widespread use of the 67 kHz subcarrier began back in the days when the FCC did not allow subcarriers to occupy any of the baseband spectrum above 75 kHz. At that time, centering a subcarrier at 67 kHz allowed it to make optimum use of available bandwidth and at the same time remain as far removed from the stereo program material as possible, as illustrated in Figure 43.

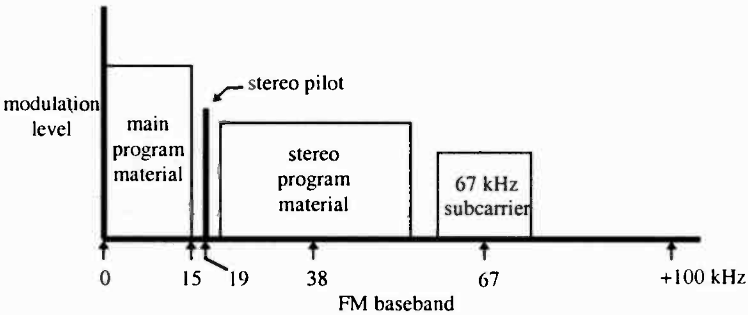


Figure 43: FM Stereophonic Signal with 67 kHz Subcarrier

In the early 1980s, the FCC modified its rules to allow subcarrier use of the spectrum from 75 kHz to 99 kHz in the FM baseband. This has resulted in widespread use of the 92 kHz subcarrier, which has become a *de facto* standard because it allows the subcarrier's operations to stay below 99 kHz while maintaining a safe separation distance from a 67 kHz subcarrier. Figure 44 shows the baseband of an FM station transmitting stereo programming and two subcarriers (one each at 67 kHz and 92 kHz).

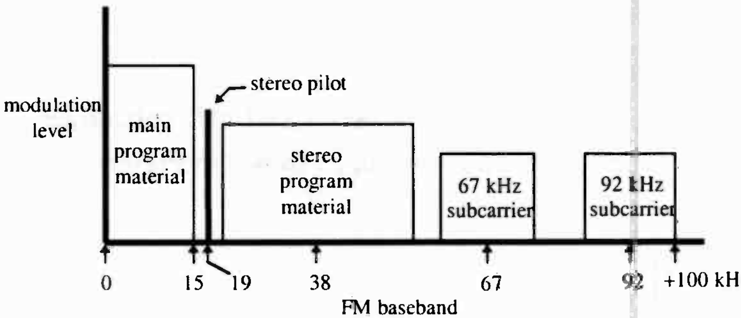


Figure 44: FM Stereophonic Signal with 67 and 92 kHz Subcarriers

In 1993, the *National Radio Systems Committee (NRSC)* adopted a standard for transmitting data at 1187.5 bits per second on a subcarrier at 57 kHz in the FM baseband. This standard is called the *United States Radio Broadcast Data System (RBDS) Standard.* An updated edition of this standard was adopted by the NRSC in 1998.

The recommended bandwidth for the RBDS signal is approximately 4 kHz (*i.e.*, 57 kHz ± 2 kHz). It is centered on 57 kHz because 57 kHz is the third harmonic of 19 kHz, the FM stereo pilot frequency. This aids in both transmitter and receiver design because it eliminates the need for a separate radio frequency *oscillator* for the RBDS signal.

(An oscillator is a circuit that generates a radio frequency signal.)

The RBDS signal can fit in between a 67 kHz subcarrier and the stereo program material, as shown in Figure 45.

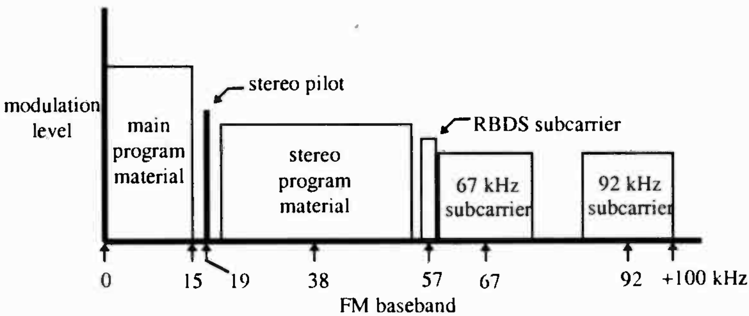


Figure 45: FM Stereophonic Signal with 67 and 92 kHz Subcarriers and RBDS Subcarrier

The composite baseband signal of an analog television station's audio channel looks very much like the composite baseband signal of an FM station. In addition to stereo audio, there are two standardized subcarriers defined for the TV audio channel. These are the second audio program (SAP) channel, and the professional (PRO) channel. The SAP channel is a monophonic audio channel that television broadcasters can use to transmit any type of audio information they desire. To date, there are not many broadcasters making use of this channel. The PRO channel can be used for sending voice or data information. It is a narrower channel than the SAP channel and therefore only permits voice transmissions of a quality similar to a typical telephone line. The PRO channel, like the SAP channel, is not yet in widespread use. Television broadcasters are not required to use the SAP or PRO channels if they choose to transmit subcarriers. They may, instead, transmit subcarriers on different baseband frequencies if they desire. An illustration of the typical analog TV station's aural baseband signal is provided in Figure 46. The frequencies on the horizontal axis have been rounded off to make the illustration easier to read.

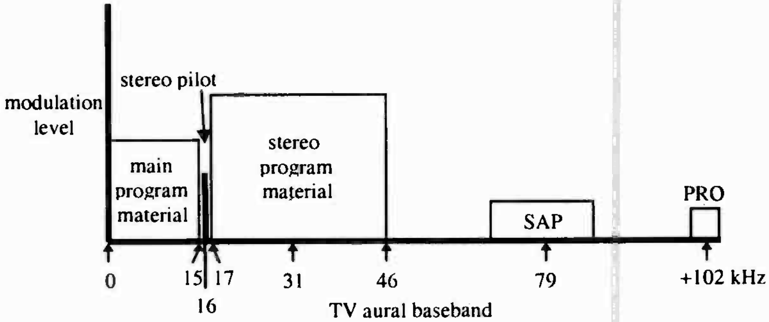


Figure 46: TV Aural Baseband Signal

As was noted earlier, the AM broadcast channel is significantly narrower than the FM broadcast channel and the TV broadcast channel. For this reason, there is an extremely limited amount of channel capacity available for AM subcarriers. To illustrate this point, Figure 47 shows the AM- and FM-band *emissions masks* defined by the FCC. The emissions mask is the limit placed on the signal strength of the broadcast signal, and it is defined over a range of frequencies surrounding the carrier frequency. For AM-band stations, the emissions mask is defined in Section 73.44 of the FCC Rules. For FM-band stations it is defined in Section 73.317. A broadcast station's signal strength at specific frequencies must decrease as the frequencies become farther away from the carrier. This is to protect stations operating on nearby frequencies from interference.

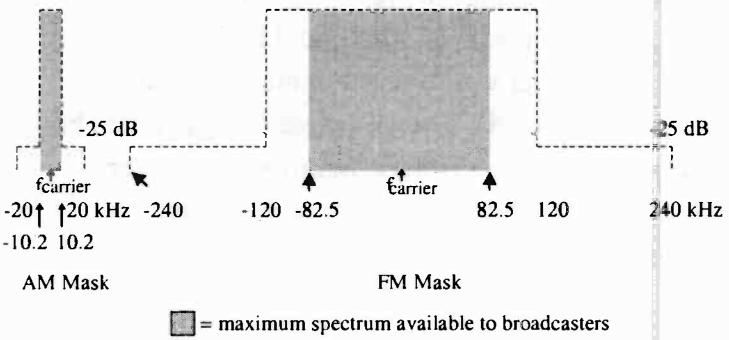


Figure 47: Comparison of AM- and FM-Band Emissions Masks

As Figure 47 illustrates, the radio spectrum available to an AM station is approximately ten percent of that available to an FM station. Furthermore, an AM station's program material (whether the station is broadcasting in monaural or stereo mode) occupies all of the radio spectrum, and all of the baseband spectrum, assigned to the station. This makes it extremely difficult to place a subcarrier in an AM channel without the subcarrier causing interference to AM reception. Despite this difficulty, however, methods have been proposed for installing such subcarriers.

In 1992, the National Radio Systems Committee requested proposals for an AM enhancement to RBDS. In response to this request, a proposal was received which described a data subcarrier system for the AM band that called for the use of data subcarriers at 9.2 kHz and 9.8 kHz above the AM carrier, at a level well below the main program material (see Figure 48). Development of this system has never been completed, however, because it was later determined that the data subcarriers were audible on wide-band AM receivers.

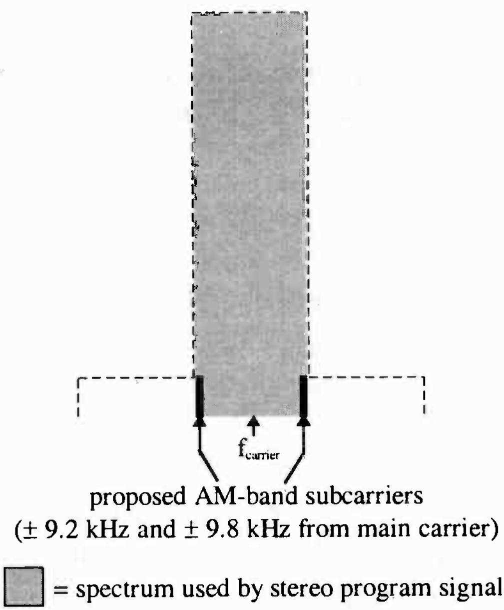


Figure 48: Proposed AM-Band Subcarriers

Despite the difficulty in developing an AM-band subcarrier system that will not interfere with an AM station's main program material, all is not lost for AM data broadcasting. Another method for *multiplexing* (*i.e.*, adding together) signals in an AM broadcast channel is called *quadrature amplitude modulation*, or *QAM*. This method has been successfully used for years as a means of allowing AM broadcasters to transmit auxiliary information.

QAM is a modulation method in which two separate signals (*e.g.*, a main program signal and an auxiliary data signal, or a main program signal and a stereo program signal) are modulated onto two separate carriers that are of the same frequency, but that are 90° out of phase with one another. Because they are on the same frequency, they occupy the same portion of the radio spectrum (*e.g.*, the same AM-band channel). Because they are out of phase, they can be detected separately, by two separate receivers (or by a single AM stereo receiver). Figure 49 shows an example of a QAM signal.

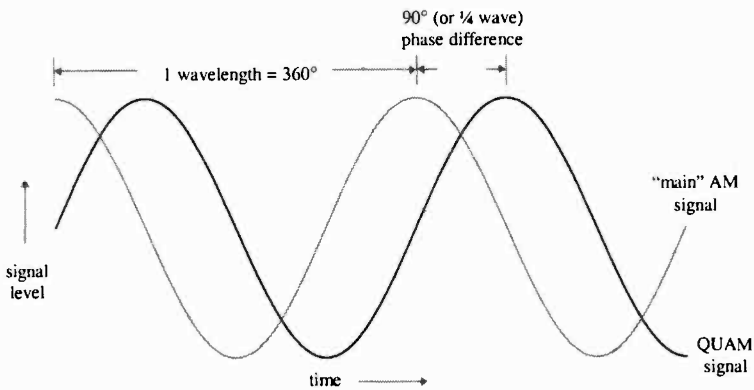


Figure 49: Example of Quadrature Amplitude Modulation (QAM)

It should be noted that the quadrature signal cannot be much more than 90° out-of-phase from the main program signal because, if it were, it would begin interfering with the in-phase signal. That is, if the peaks of the quadrature signal were to correspond too closely with the valleys of the in-phase signal, the two signals would end up canceling each other out.

So, now we know about subcarriers, and the baseband signal. It is the baseband signal that gets fed into the exciter for modulation onto the station's main carrier. In the case of an analog TV station, there are two baseband signals (audio and video) fed into two exciters for modulation onto two carriers. To learn about what the exciter's function is, keep reading.

The Exciter

The *exciter* is the device that takes the composite signal and converts it to a radio frequency signal. In the case of AM radio, the exciter takes the incoming composite signal and amplitude modulates it onto the carrier frequency used by the station in question. In FM radio, the exciter takes the incoming composite signal and frequency modulates it onto the station's carrier. In analog television, the video exciter takes the incoming composite video signal and amplitude modulates it onto the station's video carrier — and the aural exciter takes the incoming audio signal and frequency modulates it onto the station's aural carrier. In digital television, there is a single exciter that takes the digital ATSC data stream and modulates it onto the station's carrier using amplitude modulation.

Let's look a little more closely at carrier frequencies, their relationship to channels, and the different types of modulation used in broadcasting.

Each radio channel contains one *carrier*, while each analog television channel contains two carriers — one for audio information and one for video information, and each digital television signal contains a single carrier. A carrier is, quite simply, a signal of a particular frequency to which additional information is added for delivery to a receiver. The "additional information," as we have just learned, is the composite signal. A carrier is called a "carrier" because it "carries" the program material, in composite form, to the receiver. It is then up to the receiver to take the composite signal and convert it back to component form for listening and/or viewing. In the AM broadcast band, carriers are frequencies between 535 and 1705 kHz. In the FM band they are frequencies between 88 and 108 MHz. In the TV bands they are frequencies between 54 and 72 MHz, 76 and 88 MHz, 174 and 216 MHz and 470 and 806 MHz.

Now that we understand how all broadcast transmitters have to be fed a composite signal for transmission on their single carrier frequency (or, in the case of analog TV stations, their two carrier frequencies), let's take a few moments to learn about how the various types of transmission (AM, FM and analog TV and digital TV) are different. We are going to start with AM and FM transmission because, once you understand these two, understanding analog television is easy. Why? Because an analog television signal is simply composed of one AM and one FM signal. The FM signal that is part of an analog TV signal is the TV station's audio — and there is really very little technical difference between a TV station's audio signal and an FM radio station's signal. An analog TV station's video signal is basically very similar to an AM radio station's signal — except, of course, that the TV station is transmitting a series of codes that are used by a TV receiver to paint a video picture on the screen, while the AM radio station is transmitting audio information. Once you understand the basics of AM and FM transmission you will be able to gain an understanding of analog TV transmission rather easily. Digital TV transmission is considerably different than analog TV transmission. We will discuss digital TV transmission separately, after we cover analog transmission.

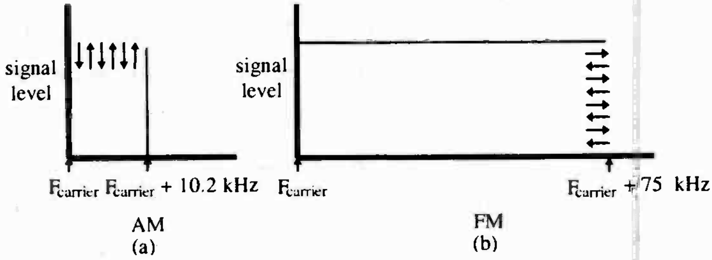


Figure 50: Radio Spectrum Occupied by AM and FM Signals

AM stands for *amplitude modulation.* In AM transmissions, program material is modulated onto the carrier signal in a manner that causes changes in the amplitude of the carrier signal which correspond to changes in the volume of the source program material. In AM transmissions, the frequency of the modulating signal (the program material) is combined with the carrier to form a composite signal which, at any particular instant in time, will have a frequency somewhere between the carrier frequency minus about 10 kHz and the carrier frequency plus about 10 kHz. (Remember, a composite signal is a signal that is formed by adding two or more signals together. In the case of the composite radio frequency signal, one can think of the two signals being added as the carrier and the composite baseband signal.) The reason that the spectrum occupied by the AM signal extends out from the carrier by about ±10 kHz is that the modulating signal (the program material) is within the audio frequency range from 0 to about 10 kHz. The reason that the spectrum occupied by the composite carrier signal extends from about 10 kHz *below* the carrier to 10 kHz above the carrier — instead of simply from the carrier to about 10 kHz above the carrier — is that both the program material and the carrier wave have "upward sloping" and "downward sloping" components, like the waves on the ocean, and the composite carrier signal that results from the addition of a particular piece of program material will vary (between carrier plus modulating program frequency and carrier minus modulating program frequency) depending on whether the carrier and the modulating program frequency happen to be sloping upward or sloping downward at the instant that they are combined.

In an AM transmission, the bandwidth of the composite radio frequency signal remains essentially constant (carrier frequency ± about 10 kHz) while the amplitude of the composite carrier signal increases and decreases with the level (volume) of the modulating signal.

FM stands for *frequency modulation*. In FM transmissions, program material is modulated onto the carrier signal in a manner that causes changes in the frequency of the carrier signal which correspond to changes in the volume of the source program material. In FM transmissions, the frequency of the modulating signal (the program material) is combined with the carrier to form a composite radio frequency signal which, at any particular instant in time, will have a frequency somewhere between the carrier frequency minus about 75 kHz and the carrier frequency plus about 75 kHz. The reason that the spectrum occupied by the FM signal extends out from the carrier by about ±75 kHz is that the Federal Communications Commission has stipulated that this is as much *frequency deviation* as it will allow in the FM band in order to protect signals on adjacent channels from interference. As with the AM signal, the reason that the spectrum occupied by the composite radio frequency signal extends from about 75 kHz *below* the carrier to about 75 kHz above the carrier — instead of simply from the carrier to about 75 kHz above the carrier — is that both the modulating signal and the carrier wave have upward sloping and downward sloping components, and the composite carrier signal varies (between carrier plus modulating program frequency and carrier minus modulating program frequency) depending on whether the carrier happens to be sloping upward or sloping downward at the instant that the program material is added.

In an FM transmission, the amplitude of the composite radio frequency signal remains essentially constant while the bandwidth of the composite carrier signal increases and decreases with the level (volume) of the modulating signal.

An illustrative way to picture what happens in an FM transmission is to imagine a simple tone, say a 1 kHz tone, as the program material being used to FM modulate a carrier. Let's imagine that the volume of this tone is relatively low, so it is not FM modulating the carrier to the maximum allowed ±75 kHz, but instead it is modulating the carrier to ±50 kHz. A good way to picture what is happening in this situation is to imagine a very narrow "spike" of a signal which is bouncing back and forth from 50 kHz above the carrier to 50 kHz below the carrier at a rate of 1000 times per second (1 kHz). Now, if you imagine that the volume of the 1 kHz tone is increased to the maximum level allowed by the FCC, then the "spike" would be bouncing back and forth from 75 kHz above the carrier to 75 kHz below the carrier at the same rate — 1000 times per second.

Of course, normal program material consists of many different frequencies, not just a simple tone. However, the principle is still the same — each of the individual frequency components of the program material causes the composite radio frequency signal to "swing back and forth" at a rate equal to whatever the frequency of the program material is, and the distance that the composite radio frequency signal swings away from the carrier is determined by the amplitude (volume) of the program material.

As we noted earlier, analog (NTSC) television signals use a combination of AM and FM modulation. The video signal in an analog TV transmission is an AM signal, and the audio signal is FM. The two major differences between the AM picture signal in an analog TV channel and an AM radio signal are 1) the AM picture carrier is carrying video information instead of audio; and 2) the modulated AM picture signal occupies roughly 25 times as much of the radio frequency spectrum as a modulated AM radio signal. The FM audio signal in a TV channel is very similar to an FM radio signal — there are only minor differences between the two.

The video signal in an analog TV transmission would actually occupy a lot more than about 25 times the spectrum of an AM radio signal if it were not for the fact that the TV signal uses a special form of AM modulation known as *vestigial sideband.* Basically, all this means is that most of the lower half of the TV video signal is not transmitted (recall that the normal *double sideband* signal would extend from the video carrier minus the modulating frequency to the video carrier plus the modulating frequency), because the information necessary to recreate the video picture on a receiver can still be obtained even with this information missing. The main reason this technique is used is to conserve spectrum. A television channel, as it is, takes up 6 MHz of spectrum — roughly 300 times as much spectrum as an AM radio channel, and 30 times as much as an FM radio channel.

As noted earlier, an analog television station will have two exciters — one for video and one for audio. In their most common configuration, these two exciters will feed their output signals into two separate power amplifiers. Then, the outputs from the two power amplifiers will be combined and fed up a single transmission line to a single antenna.

A digital television station will have a single exciter that modulates the DTV digital bit stream (which includes all audio, video and ancillary data being broadcast) onto the station's carrier frequency. The output of a DTV exciter is a series of pulses that can each have one of eight different amplitude levels (-7, -5, -3, -1, 1, 3, 5 and 7). The DTV exciter will output 10.76 million of these pulses every second. The fact that each pulse can have one of eight distinct amplitude levels enables each pulse to represent three bits of digital data. How is this possible? Table 2 illustrates how it is possible.

Table 2: Data Bits Represented by Different DTV Pulse Levels

Pulse Level	Data Bits Represented
#1	000
#2	001
#3	010
#4	011
#5	100
#6	101
#7	110
#8	111

As can be seen in Table 2, three digital bits can be arranged in eight unique combinations. If each one of these combinations is assigned to a specific DTV pulse level, then the 10.76 million pulse-per-second output of a DTV exciter can represent 10.76 x 3 = 32.28 million bits per second of digital data.

You may recall from the earlier section on studio equipment for digital television that the bit rate for the ATSC data stream is 19.39 million bits per second, and its payload data rate (the data rate for the actual video, audio and ancillary data being carried) is 19.28 million bits per second. The obvious question is, why is the data rate coming out of a DTV exciter so much higher than the data rate of the actual video, audio and ancillary information being broadcast?

The reason that the DTV transmitter equipment adds so much additional data to the broadcast signal is that this additional data is needed to correct errors that occur in the transmission path. For example, a common problem with over-the-air analog television signals is "ghosting," a phenomenon caused when multiple signals from the same transmitter arrive at the receiver at slightly different times. This can happen, for example, when a receiver is receiving a signal directly from the television transmitter and a second signal that is leaving the same transmitter, bouncing off of a mountain or ridge, and then reflecting toward the receiver. Multiple DTV signals arriving at a receiver in this manner can interfere with one another and cause reception to be lost. In order to prevent this from happening error correction codes are added to the broadcast DTV signal which help receivers to fix problems caused by interference.

The Power Amplifier

A broadcast transmitter has two basic components: an exciter and a *power amplifier*. As we just discussed, the exciter is the piece of equipment that takes the audio, video and subcarrier frequency signals received from the studios and encodes them onto a radio frequency signal for transmission through the air. Sometimes the exciter is installed inside the transmitter housing and other times it is installed in an equipment rack next to the transmitter. An exciter, by itself, is actually a low power transmitter. It can be used to broadcast a signal simply by attaching an antenna to it. Exciters are not nearly powerful enough, however, to enable a broadcast station to reach its entire coverage area. So, instead of being fed directly into an antenna, the output of the exciter is fed into a *power amplifier* which greatly increases the power of the radio frequency signal to be transmitted.

The power amplifier, as one would imagine, is the device at the transmitter site which uses the most electricity. It must use a lot of electricity in order to create a very strong signal. The heart of many power amplifiers is an electron tube — or, in some cases, two or more tubes. In these amplifiers it is the tube that actually does the amplifying. The tube gets very hot during transmitter operation and must be cooled by a cooling system. Because of the harsh conditions under which they operate, electron tubes wear out. The speed with which they wear out varies from tube to tube and transmitter to transmitter, but they typically have to be replaced every few years. In order to get away from the tube-replacement routine, some modern transmitters have been designed without tubes using solid-state electronics. Although solid-state transmitters do not require periodic tube replacement, the electronic components used in their amplification circuits will still wear out and require replacement — but typically not for at least a decade or so. Solid-state electronics have been used very little in the highest-powered power amplifiers, such as those found in many UHF TV transmitters, because they are considerably less efficient than tube amplifiers at these higher powers and therefore cost the broadcaster too much extra money in electric bills.

Radio transmitters generally have one signal coming out of the power amplifier which gets fed to the transmission line. Analog television transmitters generally have two signals — one coming out of each power amplifier (audio and video), while digital television signals have only signal coming out of a single power amplifier. In a typical analog television system, the audio and video signals are combined at the output of the power amplifier to form a composite signal which is then fed into a single transmission line. In a typical digital television system no combining is necessary because there is only a single signal coming out of the power amplifier. In some cases, where a television station's analog and digital frequency assignments are on adjacent channels, combining of the analog and digital signals is done after the power amplifiers so that the analog and digital signals can share the same antenna. If the analog and digital signals are far apart in frequency, however, sharing of the same antenna is not a plausible alternative. This is because antennas are designed to operate most efficiently on specific frequencies, and an antenna that is an efficient radiator on one frequency will generally not be an efficient radiator on most other frequencies.

The Transmission Line

The transmission line that connects the transmitter to the antenna is generally either a flexible length of coaxial cable, a rigid piece of coaxial cable, or a waveguide.

The type of coaxial cable that would be used at a broadcast transmitter site can be thought of as an extremely thick piece of cable television wire. This cable is usually thicker than a typical garden hose. It has to be thick in order to handle the high-powered signal from the transmitter without melting.

Rigid pipes are sometimes used to make a non-flexible type of coaxial "cable" for use as a transmission line. Rigid coaxial cables are manufactured by running a piece of metal pipe through the center of a larger piece of metal pipe, and keeping the center piece of pipe exactly centered by installing plastic spacers between it and the outer pipe throughout the length of the "cable." Rigid coaxial lines, as their name implies, cannot be bent.

Waveguide is best described as "duct work for radio frequencies." Waveguide looks a lot like the metal ducts you might find in a home or office air conditioning system. The difference, of course, is that air conditioning ducts are used to transport air from one point to another and waveguides are used to transport radio frequencies from one point to another. Because waveguide does not have a center conductor, nor any of the insulating material associated with a center conductor, it is the most efficient type of broadcast transmission line. Being "most efficient" means that, when waveguide is used, more of the transmitter's energy makes it to the antenna without being lost in the transmission line. Waveguide's main disadvantage is that it is considerably larger than coaxial transmission lines and therefore is more difficult for a broadcast tower structure to support — particularly when the wind is blowing. It is most often used in situations where the transmitter power is very high because its superior efficiency helps the broadcaster save a significant amount of money on the monthly electric bill.

The Antenna

At the beginning of Part III we noted that all broadcast transmitter sites (AM, FM and TV) are generally very similar — they each have a transmitter, a transmission line, and an antenna. While this is certainly true, it is also true that the broadcast antenna is one thing that differs greatly from one broadcast service to the next.

To understand why antennas vary widely from one service to the next, you must first understand two very basic things. The first is that, in order to operate most efficiently, the length of any antenna (transmitting antenna *or* receive antenna) must be a function of the *wavelength* of the transmitted signal. Generally, this means that the antenna will be somewhere between ½-wavelength and ¼-wavelength long.

A wavelength is simply the distance in space that one cycle, or one "wave," of a signal occupies. (Recall our earlier discussion about how a radio/TV signal resembles the waves on the ocean.) It is easy to calculate the wavelength of a signal if you know its frequency. The only other bit of information you need to know is the speed of light, which is the speed at which all radio and television signals travel. The speed of light is 300 million meters per second. Let's calculate the wavelength of some typical broadcast signals:

AM — 1120 kHz

300 million meters/sec ÷ 1120 thousand cycles/sec = <u>268 meters/cycle</u>

FM — 98.1 MHz

300 million meters/sec ÷ 98.1 million cycles/sec = <u>3 meters/cycle</u>

VHF TV — Channel 8 (picture carrier)

300 million meters/sec ÷ 181.25 cycles/sec = <u>1.7 meters/cycle</u>

UHF TV — Channel 40 (picture carrier)

300 million meters/sec ÷ 627.25 million cycles/sec = <u>½ meter/cycle</u>

As the above exercise illustrates, the wavelength of an AM radio signal is much longer than the wavelength of an FM radio signal, and the wavelength of an FM radio signal is significantly longer than the wavelength of a UHF TV signal. Because of these differing wavelengths, there are significant differences between the antennas used to transmit and receive AM, FM and TV signals.

Usually, an AM radio station's transmitting antenna is simply the station's tower. The actual metal structure of the tower is hot (*i.e.* energized with electrical energy from the transmitter) and it can severely shock, and even kill, a person who walks up to it and touches it while standing on the ground. For this reason, the FCC requires this type of tower to have a fence around it that will prevent people from touching the tower. The height of the tower is dependent on the transmitting frequency of the AM radio station. The lower the station's frequency is, the taller the tower.

Sometimes, multiple AM radio towers are used together as part of a *directional antenna system*. Many people have seen AM DAs, as they are called, while driving down the road — though they may not have known what they were looking at. An AM directional antenna involves two or more tall radio towers standing in a straight line, or sometimes standing in a parallelogram ("leaning rectangle") formation. The purpose of the AM DA is to direct the transmitted energy toward the city of license, but to block it from traveling toward other cities where it might cause interference to other licensed stations. Figure 51 and Figure 52 provide a general illustration of the difference between a non-directional AM signal and a directional AM signal.

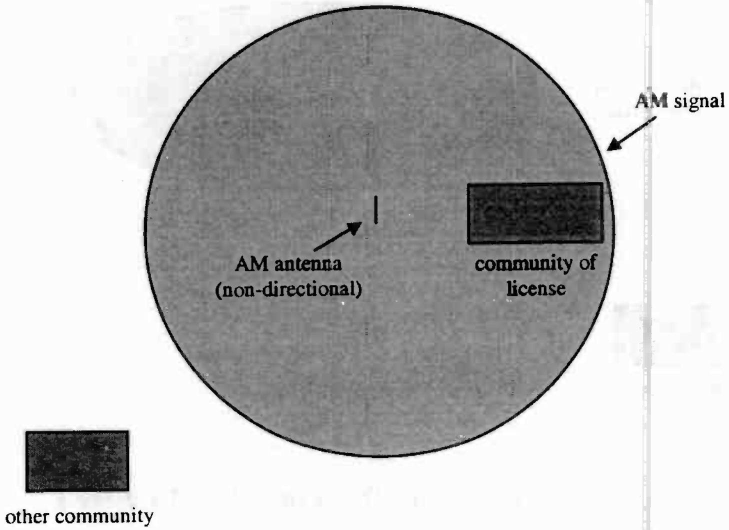


Figure 51: Example of Non-Directional AM Pattern

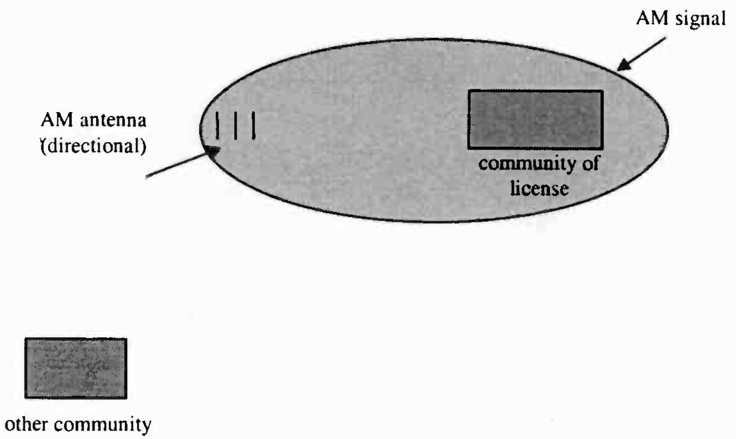


Figure 52: Example of Directional AM Pattern

In many instances, AM stations are authorized by the FCC to operate in a non-directional mode during the day and in a directional mode at night. The reason for this is that an AM signal that does not interfere with signals in other communities during the day may do so at night due to skywave propagation. For more information about skywave propagation, and the FCC's methods for ensuring that it does not result in interference to other broadcast stations, see Part V.

In addition to its directivity, another very important aspect of an AM antenna is its ground radial system. AM radio waves travel across the surface of the earth better and farther when the conductivity of the ground that they are traveling over is greater. (AM *ground conductivity* is a measure of how well the ground in a particular geographical region conducts, or carries, AM radio signals. The higher the conductivity, the better the AM signal travels.) In order to give an AM radio signal a "good start" as it leaves the transmitter, a series of *ground radials* must be installed around the transmitter. These are generally copper wires that are buried in the ground and extend outward from the base of the antenna for about 100 meters (nearly 400 feet) in a pattern similar to the one shown in Figure 53, except that a standard AM ground radial system will have 120 equally spaced radials, many more than shown in Figure 53. The actual length of each ground radial is a function of the transmitting frequency of the station. The lower the transmitting frequency, the longer the radials.

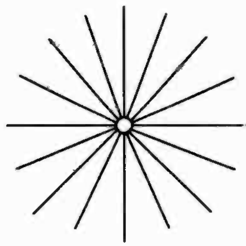


Figure 53: Overhead View of AM Ground Radials

Because of their dependence on ground conductivity and ground radials, AM radio transmitting antennas are usually found near the communities that they are intended to cover at approximately the same ground level. FM and TV transmitting antennas, on the other hand, are usually found near the communities they are intended to cover but very high off of the ground. The reason for this is that radio waves in the FM and TV bands are not as dependent on ground conductivity as AM band radio waves are, but instead are much more dependent on being able to "see" the receivers that they are transmitting to. Another way of saying this is that FM and TV band signals are "line-of-sight" signals which perform best when they have an unobstructed path between the transmitter and receiver. Generally speaking, as radio frequencies get higher and higher they become more and more dependent on having a line-of-sight view between the transmitter and the receiver.

To understand this point think about visible light, which exists at frequencies that are even higher than what are commonly considered "radio frequencies." Visible light has a very hard time getting over and around things such as mountains. FM and TV frequencies, however, have a somewhat easier time getting over and around mountains (though they perform best in an environment with no obstructions), and AM frequencies have an even easier time.

Because of their significantly shorter wavelength, FM and TV antennas are much shorter than AM antennas — typically only a few feet long. They are mounted onto their supporting structure with clamps and/or other mounting hardware and then attached to the transmission line which extends from the transmitter up the tower to the antenna. Because an FM or TV signal will be able to "see" farther, and therefore will propagate farther, if its antenna is raised up higher, the FCC's rules require a broadcaster who has been authorized for a given power at a given location to lower the station's transmitter power if the antenna is to be raised. Conversely, if the antenna is moved to a lower location on the tower from its originally authorized height then the transmitter power may be increased. The reason for this requirement is to help insure that the broadcast transmitter in question does not cause interference to other broadcast signals authorized in other communities.

You may have heard FM or TV antennas referred to as "single bay," "2-bay" or "4-bay" antennas, etc. The word *bay*, in this case, means a single antenna. Another way to describe a 4-bay antenna would be to call it a "4-antenna antenna system." Multiple bay antennas are often used in FM radio and TV transmission systems in order to make the transmission system more efficient or, sometimes, to reduce the amount of signal that is transmitted towards the ground in the immediate vicinity of the tower. The more bays that are added to an antenna system, the more focused the transmitted signal becomes. For example, Figure 54 shows how the transmitted signal from a two bay antenna might differ from the transmitted pattern of an eight bay antenna. (Figure 54 is intended only to illustrate the concept of multi-bay transmission patterns and it is not drawn to any particular scale.)

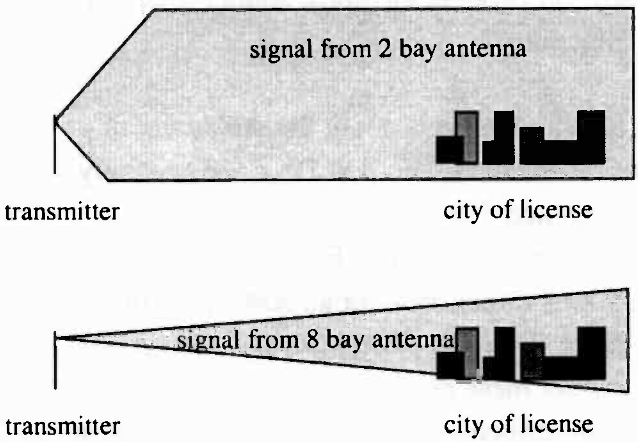


Figure 54: Effect of Additional Antenna Bays on FM/TV Signal

The astute reader will realize that a multi-bay FM or TV antenna operates using the same principles as an AM directional antenna. The major difference between the FM/TV multi-bay antenna and the AM directional antenna (aside from the frequency, and therefore size, difference) is that the multiple antennas in an FM/TV array are stacked vertically on the tower while the multiple antennas in an AM DA are lined up horizontally along the ground. The result is that the FM/TV multi-bay antenna focuses its transmitted energy in the up-down direction whereas the

AM DA focuses its energy in the north-south (or east-west, or whatever) direction.

Although the above statements about the directivity of multi-bay FM/TV antennas, and AM DAs, are generally true, it should be noted that in some instances it is desirable, or necessary, to focus an FM/TV signal in a particular horizontal direction — or an AM signal in a particular up-down direction. There are antenna systems that are designed for these purposes, too, using concepts similar to those described above. Some such adjustments can be made electronically by adjusting the timing relationship between the signals that arrive at each of the antennas in a "multi-antenna antenna system," or antenna array.

The Remote Control

The vast majority of broadcast transmitters in operation today are connected to a remote control unit. The most popular form of remote control device on the market today is the dial-up remote control. This device acts somewhat like a telephone answering machine in that it connects to a phone line and automatically answers incoming calls. The dial-up unit sits next to the transmitter and is basically a series of switches that are connected, by extension cable, to the switches that control the transmitter. The transmitter operator can simply call the remote control unit from any touch tone phone and control the switches in the remote control unit — and, by extension, the switches on the transmitter — by pressing buttons on the touch tone phone.

More sophisticated dial-up units also permit various transmitter parameters to be monitored over the phone. For example, a transmitter operator might call the remote control unit and press a series of buttons that direct the remote control unit to report on the transmitter's current output power.

Part IV: Remote Broadcasting Facilities

In the television broadcasting industry the predominate form of remote broadcasting is electronic news gathering (ENG) operations. In the radio broadcasting industry, remote broadcasting usually takes the form of small remote studio setups used for promotional purposes at retail establishments, county fairs, etc. There really is not a lot to add about this aspect of broadcasting that is not covered elsewhere in this book. The cameras, VTRs and microphones used in ENG operations are essentially the same as those used in studio applications, and the microphones and mixing boards used in radio remote setups are essentially the same as those used in "real" radio studios. The reason we have a separate Part of this book devoted to remote broadcasting is that there is one important aspect of remote broadcasting which is somewhat different from the other things covered in this book — the link between the remote location and the studio.

The link between a remote location where a news story is being covered for television and a television studio is similar to the studio-transmitter link (STL) covered in Part II. The major difference is that the remote end of this link is mobile, and usually involves a microwave transmitter and antenna installed on a van or truck. The operator of the equipment in the vehicle must aim the microwave transmitting antenna in the direction of the studio where the microwave receive antenna is located.

The operator must then feed the audio and video from the camera, microphone, video tape recorder, and/or whatever other equipment is being used at the remote location into the microwave transmitter and send it back to the studio. A lot of coordination must go on between all of the operators of microwave ENG equipment in a given geographical region in order to ensure that each microwave link does not interfere with any of the other microwave links in the region.

For radio stations, the typical link between a remote studio setup and the "real" studio is often times not as robust as the radio station's studio-transmitter link. This is because, in a typical radio remote broadcast, it is common for the music (which requires a more robust signal path than voice because music involves a wider range of audio frequencies) to be played from the main studio so the link between the remote location and the main studio only has to carry voice. Sometimes standard dial-up phone lines are used to connect the remote location to the studio, other times special phone lines that are capable of carrying a wider range of frequencies are employed. Remote pickup units (RPUs) are also very popular. An RPU is a radio version of television ENG equipment. It involves a transmitter with a directional antenna — usually mounted on a van or truck — at the remote site which sends the audio from the remote site back to the main studio.

Part V: FCC Technical Rules

The FCC's technical rules regarding broadcast stations have this primary objective: to insure that a broadcast station's signal does not cause unacceptable amounts of interference to a) another broadcast station's signal; or b) signals in other telecommunications services. Another objective of the FCC's rules is to insure that the signals transmitted by the nation's broadcasters are of good technical quality.

In order to insure that broadcast stations do not cause interference to one another, the FCC has adopted complex rules regarding the allocation of broadcast frequencies to local communities. These rules are based on the propagation characteristics of the broadcast signals involved. These "propagation characteristics" are the particular qualities of a signal that determine how it behaves as it travels through the atmosphere (*i.e.* how far and in what directions it travels). AM radio stations have distinctly different propagation characteristics than FM and TV stations and the FCC has therefore adopted somewhat different allocation procedures for AM stations than for FM and TV.

What makes an AM signal so much different than an FM or TV signal is the need to consider skywave propagation from an AM transmitter. *Skywave propagation* is a phenomenon that occurs after sunset in the AM band when radio signals bounce off of layers of the Earth's atmosphere.

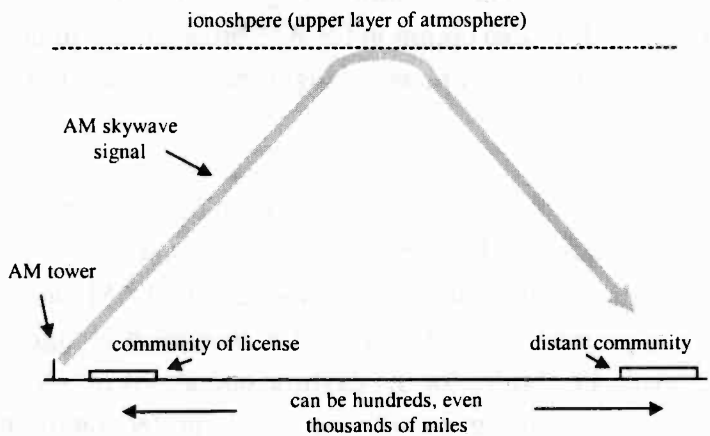


Figure 55: Illustration of AM Skywave Propagation

By bouncing off these layers of the atmosphere, an AM signal is capable of traveling from its transmitter to a far off city, while skipping over many places in between. For this reason, care must be taken when allocating an AM frequency to a particular location to insure that the new AM station will not cause unacceptable levels of interference to distant AM stations because of skywave interference. Due to the need to protect other AM stations from skywave interference, the FCC allocates some AM frequencies to local communities on a "daytime-only" basis. Nighttime operation is prohibited in order to insure that these stations do not cause skywave interference to other AM stations that are in other cites. These stations are generally called "AM daytimers." It is also common for AM broadcasters to be granted permission to operate at night, but at a much lower power.

During the daytime, skywave propagation is not an issue for AM broadcasters. This is because a lower layer of the Earth's atmosphere that absorbs upward bound AM radio waves is present during the day, but it disappears at night. As a result, FCC rules for the daytime operations of AM broadcasters are designed with the goal of protecting these broadcasters only from each other's groundwave signals. An AM station's *groundwave signal*, as the name implies, is a signal that travels from the station's transmitter over the earth at ground level. How well (*i.e.*, how far) the groundwave signal travels, when transmitter power is held constant, is a function of the Earth's *ground conductivity* over the route being traveled by the signal. Stations located in parts of the country where ground conductivity is high will have signals that propagate farther over the Earth's surface than those in areas of the country where ground conductivity is low.

Just as ground conductivity is very important to the propagation of an AM signal, so too is the grounding system of an AM broadcast antenna. It is very important that an AM antenna have a good system of *ground radials* associated with it. These are long lengths of wire that extend outward from the tower for a distance of approximately 100 meters (nearly 400 feet). The actual length of a particular station's ground radials will be dependent on the station's transmitting frequency. These ground radials help to insure a good relationship between the AM antenna system and the Earth's ground. This relationship is very important because, as we just discussed, an AM signal travels better in areas where there is good ground conductivity. Having a poor connection to the Earth's ground is essentially the same as decreasing the ground conductivity in the surrounding area.

While skywave propagation of AM signals occurs only at night, groundwave propagation occurs during all hours of the day. So, while the FCC generally only needs to consider skywave propagation when determining an AM station's nighttime operating parameters, it must consider groundwave propagation for both daytime *and* nighttime operating parameters.

The FCC's rules for allocation of FM radio channels are much different than the rules for AM channels because the propagation characteristics for FM radio signals are much different than those for AM. For FM radio, skywave propagation is not an issue. Furthermore, groundwave propagation is not an issue either. Now, you may be asking yourself how an FM signal gets from an FM transmitter, which is relatively close to the ground, to an FM receiver, which is also relatively close to the ground, without groundwave propagation. Well, the answer is that there is a subtle difference between the way an AM groundwave signal travels and the way an FM signal travels. The AM signal literally follows the Earth's surface, going up and down over hills, etc. How well it propagates, as we just learned, is a function of the conductivity of the Earth's surface over the distance it is traveling. An FM signal, on the other hand, travels in a line-of-sight path. Its propagation is generally not dependent on the conductivity of the land it is traveling over. Further, an FM signal has a tendency to reflect off of the Earth's surface on those occasions when it encounters the Earth's surface. It will also reflect off of large manmade objects, such as building and even some vehicles such as airplanes and buses. An FM transmitter located in an area where there are a lot of mountains or large hills will have major propagation problems for two reasons.

First, the signal from an FM transmitter located on one side of a mountain will have a hard time reaching the other side of the mountain because the mountain will block the FM signal. Second, as the mountain blocks the FM signal it reflects some of the transmitted energy off into a different direction from that in which it was originally traveling. This results in a phenomenon called *multipath interference* which, as the name implies, is a type of interference experienced by FM receivers when they receive two or more signals from the same FM transmitter, one via one path and another via a second path. An illustration of a multipath interference situation is provided in Figure 56. Although these two signals are, in most respects, the same – because they come from the same transmitter – they appear to the receiver as if they are two different signals because they are slightly out of synch (*i.e.*, one of them arrives at the receiver a small fraction of a second before the other).

It is the fact that the two signals are out of synch that causes them to interfere with one another. This type of interference often manifests itself as the *picket fencing* effect that many motorists notice on their car radios as they pull up to a stop light. The picket fencing, or fading in and out of the signal, is caused when the two or more signals involved alternate between causing a lot and a little interference, depending on exactly where the receiver is.

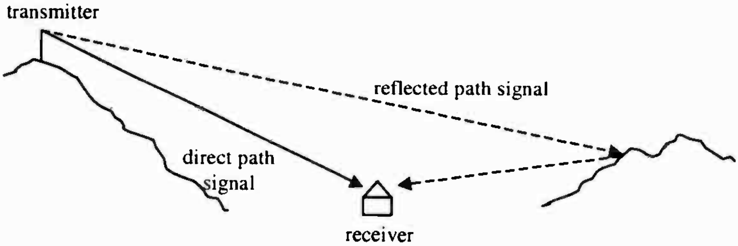


Figure 56: Illustration of Multipath Interference

The FCC's allocation procedure for FM radio stations takes into account the general layout of the land around the proposed antenna site, and the distance between the proposed antenna site and other FM stations on the same and adjacent channels. At the lower end of the FM band, which is very close in frequency to TV channel 6, the distance to the nearest TV channel 6 transmitter is also taken into account. As long as the signal of the proposed FM station is far enough away from other stations on the same and adjacent channels that it will not interfere with these other stations, the FCC will allocate the frequency for the proposed station. The FCC has specific separation distance requirements in its rules that it uses to determine whether or not a proposed station is far enough away from existing stations.

TV signals propagate in basically the same manner as FM signals. The multipath effect referred to above with regard to FM signals also has an impact on NTSC (analog) TV audio and video signals. Multipath interference will cause distortion in the received audio signal from an NTSC TV station, and it will cause the *ghosting* effect that many people have experienced in their received TV video. Ghosting occurs when two or more video signals from the same NTSC TV transmitter arrive at the receiver at the same time, causing multiple images (ghosts) to be painted on the screen. ATSC (digital) TV signals are immune to ghosts caused by reflected signals because digital television signals contain digital codes that enable a receiver to lock on to one specific signal for reception. Because DTV receivers can lock on to one specific signal and ignore all other reflected versions of the same signal, ghosting is not a problem for DTV.

You may recall from our discussion about NTSC video in Part I that one of the signals that can be transmitted in the vertical blanking interval by an analog television station is a *ghost canceling signal.* This special signal is used by analog receivers equipped with ghost canceling circuitry to combat problems caused by multipath reception in TV receivers.

Because the propagation characteristics of TV signals are so similar to those for FM signals, the FCC uses a substantially similar procedure for determining whether or not a proposed TV channel can be allocated to a particular community. It considers the power and height of a proposed TV transmitter to determine if it will interfere with existing signals.

In addition to having standards which keep different broadcast signals on the same or adjacent channels *geographically separated*, the FCC also has rules which are intended to insure that each broadcaster's signal stays *separated in frequency* from its neighbors. These are the FCC's modulation limits, and they help to insure that one station's signal does not bleed over into the channels of other stations located next to it on the dial.

Conclusion

You made it! This tutorial is now complete! To sum everything up nice and neatly, we have learned that the basic engineering components of a broadcast facility are the studio, the studio-to-transmitter link, and the transmitter. We have learned about the major components of broadcast studio and transmitter sites, and we have seen how they connect together to produce the over-the-air signal. We have also seen that AM, FM and TV facilities have a lot in common, but that there are some very important differences among them.

To be sure, there are many specialized pieces of broadcast equipment that we have not covered here (to cover *everything* would require a multi-volume set of books!) but what we have covered gives us a very good understanding of the basic engineering aspects of a broadcast facility.

We hope that you now feel more comfortable discussing broadcast engineering issues — and that the things learned from this book will enable you to do whatever it is you do more effectively.

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