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COMMUNICATIONS - ELECTRONICS

SYSTEMS APPROACH TO WIDEBAND COMMUNICATIONS

15 DECEMBER 1978

DEPARTMENT OF THE AIR FORCE

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HQ Air Force Communications Service
Scott Air Force Base, Illinois 62225

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SYSTEMS APPROACH TO WIDEBAND COMMUNICATIONS

This pamphlet presents a single reference document on the systems approach to multichannel wideband communications for operations and maintenance people. It is intended for use by both on-site operations and maintenance personnel and the commanders and staff personnel at all echelons. It covers a broad range of topics that apply to wideband systems, with emphasis being on technical considerations relating to system performance. It is not a detailed handbook that treats all topics in depth; however, those topics deserving added emphasis are presented on a level that the average on-site or staff person can understand and apply. The information and guidance presented herein can be applied to all multichannel wideband systems run by AFCS. Emphasis has been placed on the DCS and wideband links that employ line-of-sight and tropospheric scatter as the prime transmission means. Comments and suggestions for improving this pamphlet are encouraged and should be sent to AFCS/DOY.

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**D. G. Brennan, "Linear Diversity Combining Techniques," Proceedings of the IRE, pp 1075-1102, June 1959

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Chapter 1
SYSTEMS CONCEPT

1-1. General. The word "system" is a familiar one to most of us in this day and age. We talk about communications systems, transportation systems, supply systems, and others in everyday life; however, most of us do not concern ourselves with the details of the total system but tend to be interested only in that portion which appears visible to us. This pamphlet deals with a specific category of systems, that is, the military wideband communications systems. The purpose, of course, is to urge the communicator to "think systems." This chapter introduces system concepts and applies them to a sample wideband communications system.

1-2. Terms Explained:

a. Systems. Webster defines a system as "a set or arrangement of things so related as to form a whole." The point which must be emphasized is the interrelationship of the working of the parts which influences the outcome or the objective of the whole. No one would argue that a flat tire certainly degrades the automobile's ability to perform its "system objective" of providing transportation. So it is with the communications systems of the military.

b. System Objectives. The basic consideration for any system is its objective or set of objectives. The objectives, often called a mission or function, are the basis for the existence of a system; they describe what the system is supposed to do. For example, the objective of a communications system is to provide suitable communications service to and between its customers or subscribers.

c. Requirements. The preceding statement concerning system objective immediately raises the question: What is suitable service? The *requirements* are the levels of or types of performance the system must meet to accomplish its objective. These requirements may include the number of channels provided, the specifications on frequency response and delay distortion for a channel, or the limits on noise in a channel.

d. System Components. A system is made up of many components or things which are arranged or connected in order to accomplish the objective of the whole. As in the earlier example of the automobile tire, each component has a function to perform within the system. The interaction of components with different functions determines the overall performance of the system; therefore, system requirements must be translated into component requirements in building the system.

(1) Equipment. Often, one of the major classes of system components is equipment or hardware. In a communications system, this class includes radios, multiplex, antennas and waveguides, patch panels, and test equipment. How well the different components interface with each other has a direct effect on system performance.

(2) People. A common fallacy when looking at the things which make up a system is to forget the *people.* In spite of some argument to the contrary, systems work for people and it is the people who make systems work. How well a system does its job in meeting the performance objectives depends solely on the people who design, construct, operate, and maintain it. So, in any discussion of systems, it is necessary to include people. The involvement of people is in the form of two different roles - the managerial role and the functional role. The people involved in the functional role are those that actually install, operate, and maintain the equipment. They are exercising direct control over the system. The managerial role is an equally important activity for its purpose is to assure that all of the system performance objectives are being met with the most efficient use of the system resources. The basic distinction is that people involved in the functional role exercise direct control over the equipment, while those in the managerial role exercise indirect control and, even then, only through functional personnel.

e. Uncontrollable Components. Probably the most significant and, at the same time, most uncontrollable of the components of the system, particularly in the case of troposcatter communications, is the radio path between the transmit and receive antennas. Radio wave propagation will be treated in detail in chapter 7; however, it must be recognized early as one of the more critical components of the entire radio system.

f. Environment. Whereas the components are part of the system and are "inside," the environment consists of the things that affect system performance yet lie "outside" the system. The environment is important because it too determines how well the system performs.

1-3. Wideband Communications Systems. The word "wideband" is generally associated with communications media that carry many channels of voice communications or occupy the frequency bandwidth of many voice channels. A system carrying television or very high speed data would be considered wideband, for example. We will define any system which has an information bandwidth of 20 kHz or greater as a wideband system.

a. Multichannel Systems. In the early days of telephone communications, each instrument required its own pair of wires in order to provide customer service. Those days are past, but many people still visualize each user as having his own radio. Systems today absorb many customers, combine their signals, and send the composite signal over a common transmission medium. Since the composite bandwidth being transmitted is the sum of the individual bandwidths, these multichannel systems are almost always wideband. In fact, some wideband systems are the transmission medium for a larger communications system.

Hence, we see that a system can also be a component in a larger system. The definition of "system" allows us to build a hierarchy of systems with each system being a subsystem of a larger one.

b. Channels and Circuits. Several terms must be defined before we can understand a wideband system well. Generally speaking, they are in one of two categories, either associated with the equipment or the service being provided. For instance, a *channel* is associated with the equipment and is independent of the service being carried. There are certain physical characteristics unique to a specific channel in the equipment and these will be discussed in detail in the sections which cover the radio and the multiplex. A *circuit*, on the other hand, is associated with the service being provided. It is essentially the connection between two customers. The routing of long circuits can be, and often is, very complex. If they travel through many radios, they might be found on any of the channels in the radio depending on which channels are available in that equipment. Furthermore, they might be changed from one channel to another without the customer even knowing the difference.

c. Groups and Trunks. Like channels, groups are associated with the equipment. A group is composed of 12 channels, in the case of analog modulation techniques. These will be discussed in greater detail in specific sections of this publication.

The service-associated counterpart of the group is the *trunk*. Composed of one or more circuits, the trunk may travel through several radios before reaching its destination.

d. Links. Finally, links are the highest level of concern. The *radio link*, as the name implies, is associated with the radio equipment. It is from one radio to another. The service link, most commonly just called a *link*, is from one voice frequency "breakout" to another.

1-4. Systems Concept Applied - Genesis of a System. We have developed a hierarchy of concepts applied to systems. An objective is defined which generates requirements which are filled by components. We can illustrate these concepts by considering the generation of a sample communications system from a set of objectives.

a. Basic Objectives. The geographical setting for our hypothetical system is shown in figure 1-1. Five air bases are supporting strategic air operations. Able Site is a remote air base with command headquarters at Dolly. Gallow is a fighter interceptor base supported by an air defense facility which is located 180 miles away on an off-shore island at Hidden Valley. Central City is close to a large community with access civilian cables and landlines. The communications objectives in this area are:

(1) To support a primary alerting system from a higher headquarters.

(2) Provide high speed data to command and control computers at headquarters.

(3) Compile weather data for strike force operations.

(4) Exercise air traffic control.

(5) Extend aeronautical station operations.

(6) Provide operational and administrative digital data service for personnel and logistic support.

b. System Requirements. When objectives are considered in detail, they lead to requirements for the systems. For example, Able Site is a remote base with a strategic mission. Its requirements encompass every facet of service (voice, telegraph, data terminals, facsimile, etc.). These services can be provided by approximately 100 communications channels to the four other bases. A similar description could be given for the other locations. These channel requirements for the system can be summarized by a diagram as shown in figure 1-2. Direct radio paths for these channels are impractical due to the number of radios that would be involved (Able Site would need four radios, Central City four, Dolly three, etc.). It is far more efficient to group the channels into common links (figure 1-3). This requires only one radio each at Able, Dolly, Gallow, and Hidden Valley, and four at Central City. Central City thus becomes the "hub" of the system.

Channels are organized into *groups* of 12 channels. These groups can be broken down into channels, when needed, or passed through a station to the next station. These alternatives are used to develop an orderly channelization plan.

c. Terrain Considerations. Let us assume we are faced with a variety of terrain features that include open water, mountains, and tropical jungle. Note that Able Site is separated from Central City by a long, high ridge. Going east from Central City, there is a jungle and marsh terrain plus ocean separating it from Hidden Valley. Hidden Valley, in turn, is located on the far side of Remote Mountain. These conditions pose engineering problems for the communications planner. The selection of what type of wideband equipment to use can be greatly affected by these terrain features. For Able Site to communicate with Central City, another station must be added on the ridge to receive Able Site's signal and transmit it to Central City. The composite signal is not broken down at this site. Such a station with no breakdown of the signal is called a repeater. The different types of repeaters are discussed later.

A repeater is also needed on Remote Mountain to connect Hidden Valley with Central City. A troposcatter radio is used to span the jungle, marsh, and ocean between Central City and Remote Mountain. The final geographical arrangement of the system is shown in figure 1-4, along with the size of the antennas and the power of the transmitters. The final channelization plan, showing the groups and through-groups, is given in figure 1-5. By our definition, there are four links in this system: Able - Central City; Dolly - Central City; Hidden Valley-Central City; and Gallow- - Central City. Although the path from a terminal station to a repeater is not classified as a link (there is no voice frequency breakout), we refer to this path as a radio link IAW our definition. Hence, we have six radio links in this system.

This system has some definite drawbacks from a military point of view: for example, there is no backup capability in the event of sabotage at Central City. Our purpose, however, is to present an uncomplicated example of a wideband system.

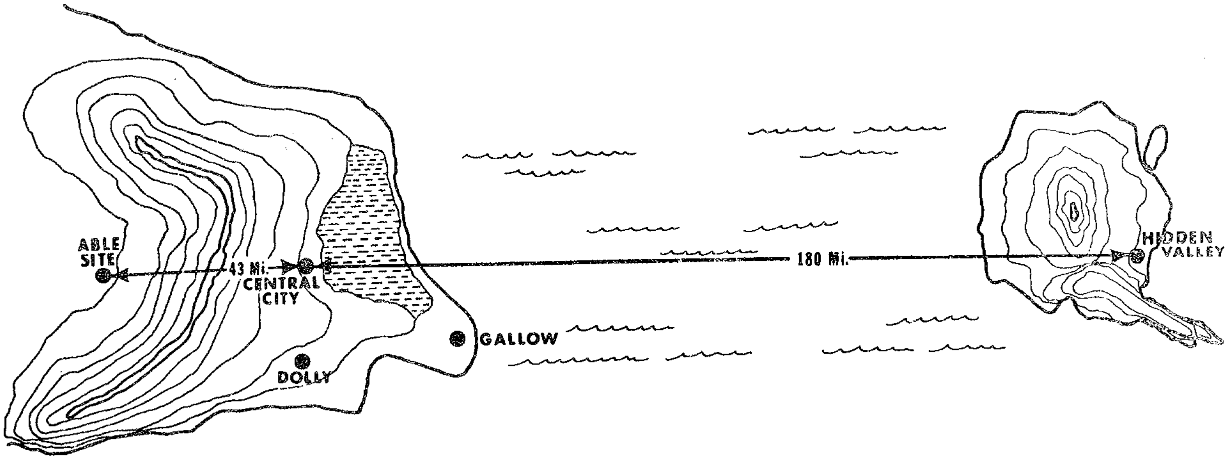


Figure 1-1. Geographical Layout of Bases.

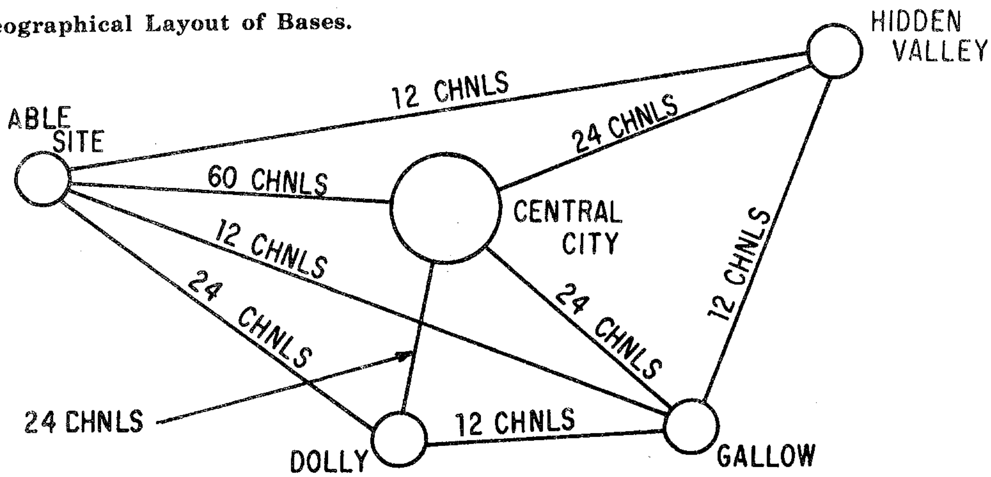


Figure 1-2. Channel Requirements.

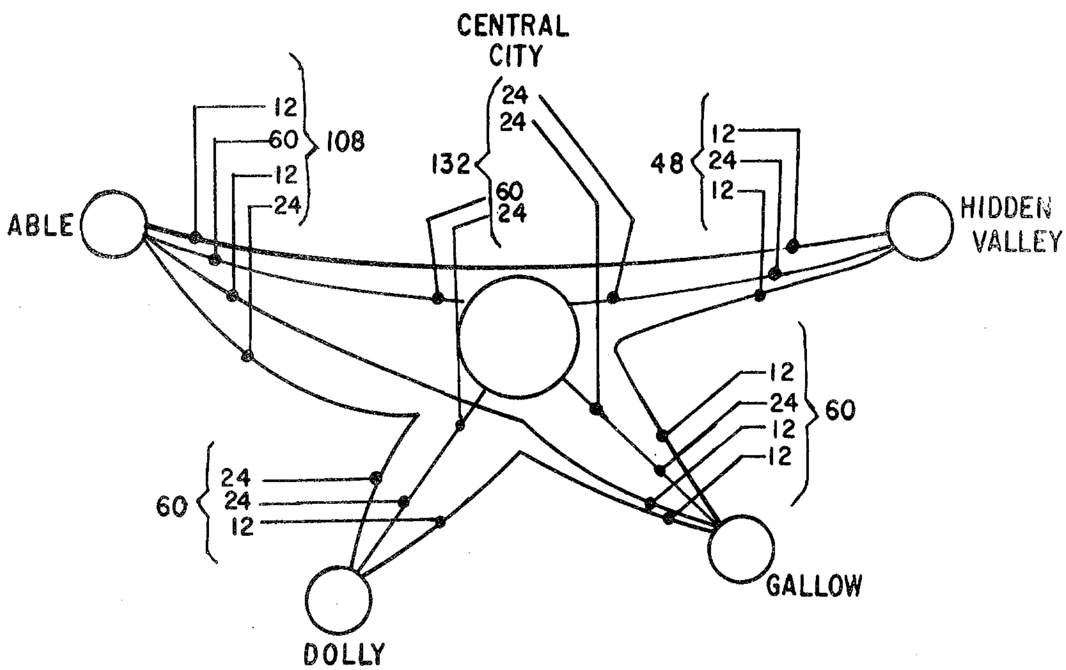


Figure 1-3. Efficient Routing of Channels.

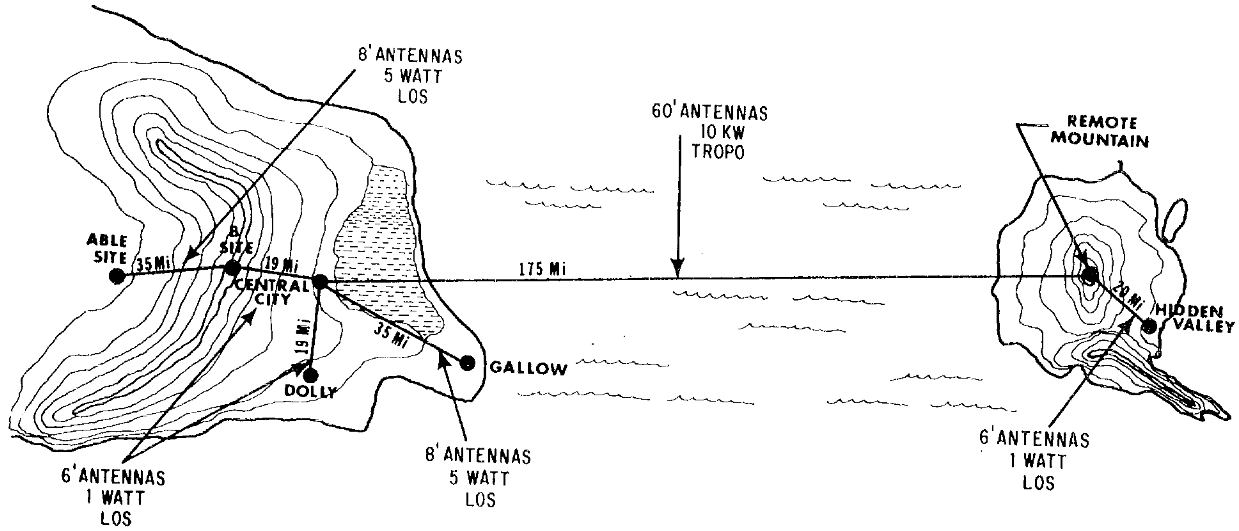


Figure 1-4. Final Geographical Configuration.

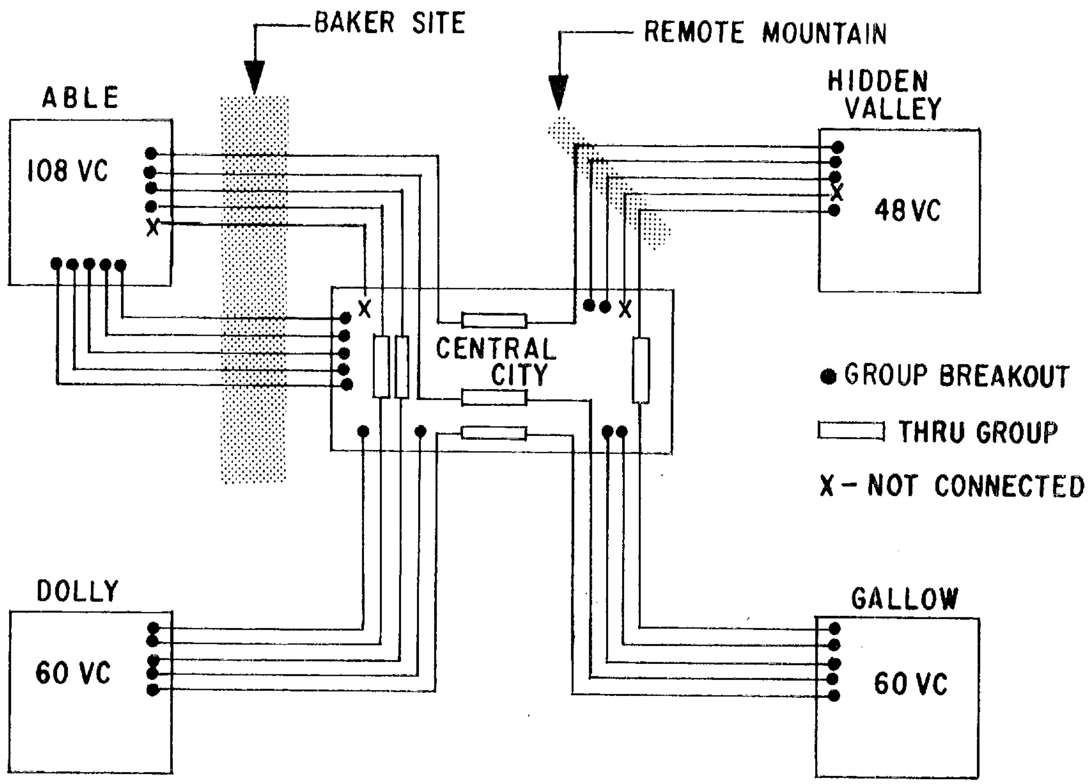


Figure 1-5. Detailed Channelization Plan.

1-5. Systems Approach. Any entity can be analyzed by basically two different methods: the so-called "black-box" approach or the systems approach. The key to the difference lies in the definition we used of a system - "a set or arrangement of things so related so as to form a whole." If we analyze the components of the system, as is the case with the black box approach, we will not be able to understand the effects which that component has on the *whole* system.

Let's suppose for a moment that there are a large number of components in our system and, further, that each of these components, while still within the specifications for that particular component, are operating at the lower limit of their specification. It would not be unreasonable to suspect that, as many of these components are put together in a very large system, the performance of the entire system might be below an acceptable level.

In addition, in using the systems approach in communications analysis, it is desirable to monitor parameters which duplicate, as closely as possible, the service required by the customer. When a problem is detected, an isolation procedure is initiated to identify the source of the problem. This procedure requires coordination between stations in the system. In many cases, the problem, as it appears in a single station, can be solved by an action at that station; however, while the problem may disappear at this single station, it may appear at another location. This is why all corrective actions must be coordinated at the system level. The black box method of troubleshooting should be initiated only after the source of the problem has been isolated to a single component. After resolution, the performance of the system should be checked to make sure that the system is operating in the proper way.

1-6. Performance Measures. Simply stated, the objectives of a communications system are to pass information rapidly, reliably, and with high quality. These attributes can serve as performance measures - indications of how well the system is meeting its objectives. The performance measures can be related directly to the signals carried over the system.

a. Quality. Quality is related to how closely the output signal resembles the input signal. Since no communications system is perfect, the signal is impaired as it traverses the system. It becomes distorted because of non-linearities in the system; additionally, noise, which is not related to the signal, interferes with the signal and tends to mask it. In principle, the quality of the wideband system depends on the relative levels of signal, distortion, and noise. In a high quality system, the signal levels are approximately one million times more powerful than the distortion and noise levels.

b. Availability. Availability is a measure of the percentage of time that the system is available for use. The total system availability of a multichannel system represents a composite of the availabilities of the individual channels. The concept of total system availability, however, has little useful meaning. Total system performance is important, but it is extremely difficult to determine. A very meaningful figure is circuit availability because it is more closely related to service.

c. Reliability. Reliability is a measure of the ability of the system to carry traffic. It is related to both the availability for use and the quality of service provided. Reliability is most meaningful when related to circuits. Factors which affect the reliability are the reliability of the equipment, radio wave propagation, traffic loading, the type of information being passed, etc.

d. Speed. Speed in an overall communications system has two meanings. The first relates to establishing a rapid circuit connection and is mainly the function of the switching speeds, either manual or automatic. Though important from the point of view of the entire communications system, it will not be stressed herein. The second meaning, usually referred to as the transmission rate or signaling speed, relates to how rapidly information can be passed over the system once a connection has been established. This measure applies more to digital signals than to analog signals. It relates very closely to the quality of a digital circuit.

1-7. Performance Indicators:

a. General. It should be apparent at this point that the key to the systems approach is the selection of the performance indicators. They must be selected wisely if a true measure of system performance is to be obtained. Improper selection may result in an inaccurate measure of system performance. Obviously, they must relate very closely to the performance measures of the system. The performance indicators of a system can be divided into three groups:

(1) System Indicators. The first group contains indicators that provide a measure of the entire system performance. By monitoring this group, we will get a quick assessment of how well the system is performing. Usually, the system indicators do not locate the trouble in the system - they merely indicate the system performance levels. The reason for this is that these indicators measure the combined performance of the system components, their interaction with each other, and the environmental factors. It is like measuring the air pollution in a large city and finding that it exceeds an acceptable level. These measurements indicate degraded performance but they do not provide a direct indication of the causes or the sources of the problem. Additional activity is necessary to locate and correct the causes. This immediately suggests the need for another group of indicators that gives the performance measure of smaller portions of the system.

(2) Component Indicators. The second group of performance indicators measures the performance levels of the components of the system. These indicators serve two useful purposes. First, they form the basis for a continual quality control program that measures the performance levels of the components on a repeated basis. Secondly, and just as important, they are used in troubleshooting and fault isolation actions when degraded performance is detected in the system.

(3) Subsystem Indicators. This group of indicators is very similar to the component indicators. The only difference is that the subsystem indicators measure the combined performance levels of a group of components that perform a related function. These indicators, like the component indicators, are very useful in a continual quality control program and in troubleshooting and fault isolation actions.

b. Acceptable Performance Level. For all of the performance indicators, there must be an acceptable level of performance. The system performance levels are established at a point that will ensure that the performance objectives of the system will be met. The component and subsystem performance levels are then established at a point that ensures that the system performance levels can be achieved. Once they are established, the performance levels become a yardstick by which the people who run the system determine whether the system performance is acceptable. They form the basis of the system quality control program.

c. The "Failure" Indicator. Another measure of system performance is the customer complaint. It is the final indicator of system performance; it is an indicator of system failure. Performance indicators are designed to alert us before the system degrades enough to elicit a complaint. If customer service becomes unsatisfactory, we have failed.

1-8. The One and Only Commandment. Above all else, SERVICE TO THE SUBSCRIBER IS OF PRIME IMPORTANCE in all communications systems.

Chapter 2
MANAGERIAL AND FUNCTIONAL PERSONNEL

2-1. General. Before getting into the technical aspects of wideband systems, let's take a brief look at the people involved with the communications systems - managerial and functional personnel. Although a thorough coverage of these two areas would easily fill a book, we can place them in perspective by examining the functions they perform in the wideband communications systems.

2-2. Management's Role. The main function of management in a wideband system is to ensure that the system objectives are being met with the most effective use of resources. In terms of the overall AFCS communications mission, wideband systems fall into two categories. First, there is that portion of the wideband system which is in the Defense Communications System (DCS) for which AFCS has the operations and maintenance responsibility. The second category includes the wideband systems that are a part of AFCS' tactical mission. Since it is not our intent to give a detailed treatise on management, we will consider only the first case. Recognize, however, that the functions AFCS performs in the tactical mission are basically the same as those which we will cover. In addition, many tactical wideband systems interface with the DCS.

a. The Defense Communications Agency's (DCA) Role in the DCS. DCA performs the broad functions of operational direction and management control for the entire DCS. The wideband system within the DCS that is operated and maintained by the military departments, including AFCS, is included in these functions. Under operational direction, DCA specifies how the DCS will be run. It prescribes the system standards, including engineering as well as operational and maintenance standards. They establish the policy and provide the guidance and direction which applies to the operation of the DCS. Under management control, DCA controls the resources of the DCS to ensure that all system objectives are being met. In fulfilling these functions, DCA provides the operational direction through the DCA Operations Control Complex (DOCC) It also receives its near real-time status directly from the operating sites through the DOCC. DCA Circular 310-70-1, volume I, treats DCA's management role and the DOCC in more detail. It is worth noting that DCA does not perform any operational and maintenance functions. That is where AFCS and the military departments come in.

b. AFCS' Role in the DCS. The primary functions of AFCS in the DCS are engineering, installation, operation, and maintenance for portions of the DCS. AFCS is responsible for these functions for approximately 60 percent of the wideband systems within the DCS. Since we are highlighting the operation and maintenance of wideband systems herein, we must recognize that these primary functions are performed at the operating site locations. The managerial functions performed by commanders and staff people at all echelons above the site locations are all aimed at ensuring that the system performance objectives are being met. All of the people above site level are involved in the managerial function. There are some on-site people, however, who also perform management functions, namely the site commander and his key supervisory people.

2-3. Functional Personnel. The primary activity of the functional personnel is to operate and maintain the equipment in the wideband system at the performance levels that satisfy the system performance objectives. Recognize that this function emphasizes system performance and not just the components of the system. The functional personnel consist of the operating site personnel and includes the site commander, technical controllers, and maintenance people. Although these people can directly control only the equipment at their site, it is important to recognize the relationship of the site to the rest of the system. A site is just a small portion of the wideband system and a place where some of the equipment is located. Even the simplest wideband system, with one link, consists of two sites. So the site people, in performing their function, must think in larger terms than just the components at their site or even the site itself. The prime function of on-site personnel is to ensure that their particular site does not contribute to the degradation of the *total system performance.*

a. Site Commander. The site commander is responsible for the total operation of the site. He/She is often described as being the site manager; however, this implies that his/her duties are largely administrative in nature. While this is true to a large extent, it is not sufficient in today's systems. One of the most important abilities a site commander must possess in order to manage today's complex systems is *technical ability.* The mission of a site is basically a technical one - to provide optimum communications. The site commander must provide the necessary leadership in the mission area as well as the administrative area. It is in this sense that he/she is involved in the functional role in the communications system.

b. Site Engineer. At many of the sites where the mission is large enough that the administrative and technical tasks are beyond the capability of one person, a site engineer is assigned. It is the engineer's responsibility to provide a greater depth of technical expertise. This expertise is also shared with other locations because a part of the mission of a site at which an engineer is assigned often has the Facility Control Office (FCO) function. This means that the facility exercises technical and restoral control over other smaller sites.

c. Tech Controllers. Complex systems do not automatically take care of themselves. The system performance indicators must be continuously monitored and corrective actions coordinated if a systems approach is to be maintained. This function is performed by tech controllers. In a very real sense, tech controllers are system managers. They are capable of "taking the pulse" of the system, that is, performance assessment and quality control.

Integral with this responsibility are the functions of fault isolation and control of fault correction. These functions must be approached from a systems viewpoint and, again, the tech controller is responsible for this action. He/She is required to anticipate difficulties in sufficient time to permit corrective action *before* excessive degradation or circuit failure occurs within the system.

In the event of circuit failure, a tech controller must reroute the customer and coordinate the restoral of the circuit. While this function is often emphasized, it should be avoided, if possible. If reroute and restoral action is necessary, the system has *failed* in its objective. The primary objective of tech controllers must be to prevent outages or reroutes. Only by emphasizing the quality control aspect of his/her duties can the tech controller fulfill his/her role as a system manager.

d. Maintenance Technicians. Thus far, we have concentrated on monitoring system performance and isolating faults; however, these actions are useless unless a skilled maintenance capability exists to repair the faults. The system, for all its complexities, is still made up of components which must be kept within operating specifications. The maintenance technician is ultimately responsible for the performance of the system hardware.

The principle of quality control to prevent failure also applies to the maintenance of a system. Degradation should be diagnosed and the fault corrected *before* a failure occurs. In this respect, maintenance technicians and tech controllers must work together. A maintenance quality control program should detect and correct component discrepancies before they cause system degradation. It is the tech controller's responsibility to detect system degradation, isolate the link, and rely on the maintenance technician to correct the fault. Such teamwork is essential to the systems approach.

It is also essential for the maintenance technician to think *systems* when performing his/her duties. Many maintenance actions at a site can affect another site's performance. If proper coordination is not maintained, system degradation results. For example, changing the transmitter deviation at one site requires the distant end to make receiver baseband adjustments.

Another detriment to the systems approach is in station "peaking" of equipment. For example, what if a transmitter is looped back, connected to the receiver at the same site, and "peaked" for best performance? When this transmitter is put on the air, it will not be operating with that receiver; hence, poor system performance may result. All maintenance must be accomplished with the system in mind.

Chapter 3
UNITS AND MEASURES

3-1. General. To maintain and provide a reliable wideband communications system, relatively free of troubles, AFCS personnel have at their disposal the best tools in the world. The proper use of these tools can be taught to the personnel who use them, but the problem usually involved in the operation and maintenance of a wideband communications system encompasses more than the use of tools as such. The word "tools," for instance, includes the many types of test equipment that AFCS personnel must know how to properly use. The use of this test equipment involves many simple and many complicated tests. Invariably, the results of these tests are read or recorded by the use of a complicated language that must be learned, and many times relearned, by the technicians involved. This terminology is unique to units and measures and the only language by which the test equipment can be read. Unfortunately, many technicians who learn this language do not continue to use and practice it and soon it becomes misused, misspelled, and misunderstood. For one thing, the language involves terminologies derived from logarithmic equations; therefore, we can see that, unless some of the rudiments are understood, the language of units and measurements can never really be understood and used to its fullest extent.

3-2. Logarithms as a Tool. Mathematicians found out long ago that addition and subtraction are simpler processes to perform than multiplication and division. They were interested not only in paper work calculations, but in analog representations.

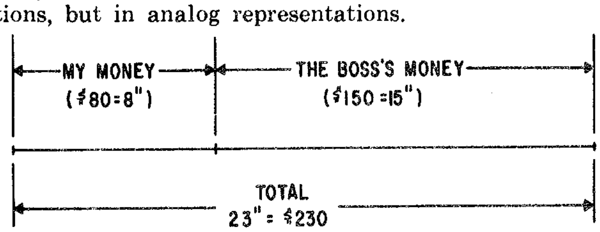


Figure 3-1. Analog Addition.

A simple example is the stick analogy. Assume that you have two sticks. One is proportioned to the amount of money you have (a short stick), the other is proportioned to the amount of money the boss has (a longer stick). In order to compute the total money the two of you own, you could add - mathematically - these two sums. This addition could be done in analog fashion by laying the two sticks end-to-end and measuring the total length (figure 3-1). This is a good theory, but addition is not so very hard. Now consider another mathematical process. Which is faster, adding or multiplying 47.32 and 23.74?

```
 47.32         47.32
+23.74        x23.74
 71.06         18928
              33124
             14196
             9464
           1123.3768
```

It is evident that the multiplication process is considerably more time-consuming and requires more mental processes. It is more advantageous to use a method of multiplying, using an analog device, such as can be used to add two lengths or numbers. In order to do this, logarithms were devised.

A logarithm is an exponent and, IAW laws for exponents, they may be added to multiply two numbers having the same base. This simply states that since $10^2 = 100$ and $10^3 = 1000$, then $10^2 \times 10^3 = 10^5$. In this example, 10 is the base number, while 2 and 3 are exponents. We see that by adding the exponents ($2 + 3 = 5$), we get the exponent of the product. We now have a way of adding numbers to represent multiplication. The only pitfall is that the method requires that any two numbers which are to be multiplied must be expressed as powers of the same base. This can be stated for the general case by writing:

$$B^x \times B^y = B^{x+y} \qquad \text{Eqn 3-1}$$

The base number, B, has been fixed at 10 for most communications work. When the base is fixed, then tables of logarithms (for the base 10) may be used to find the "log" of a number. Every time we want to find a logarithm, we are asking the question, "What exponent of the base 10 is required to get the number N?" In the shorthand of mathematics, this question becomes:

$$\log_{10} N = ? \qquad \text{Eqn 3-2}$$

When using the base 10, it is common to omit the subscript "10" since it is understood to be 10 without mention. For the general case, the following expressions show the relationships we have been discussing:

$$\text{if } \log A = X \qquad \text{Eqn 3-3}$$

$$\text{then } 10^X = A \qquad \text{Eqn 3-4}$$

What significance should be given to a negative logarithm or exponent? If the following numbers are familiar to you, you know the answer:

$$.001 = 10^{-3}$$
$$.01 = 10^{-2}$$
$$.1 = 10^{-1}$$
$$1.0 = 10^{0}$$
$$10 = 10^{1}$$
$$100 = 10^{2}$$
$$1000 = 10^{3}$$

It is apparent that $10^{-2} = \frac{1}{10^2}$ which, when divided out, gives .01 as a decimal. In the general case then:

$$B^{-y} = \frac{1}{B^y} \qquad \text{Eqn 3-5}$$

With some practice, many logs can be approximated. If the log of 20 is desired, the reasoning would start by remembering that log 100 = 2 (why? because $10^2 = 100$). Similarly, log 10 = 1. We are looking for the log of a number between 10 and 100 - log 20 is between log 10 and log 100; therefore, log 20 is between 1 and 2. Later we will add information which will allow us to get a much closer approximation: log 20 = 1.3.

Our purpose was to find a way to multiply by adding; this has been done by adding the logs of the numbers. It follows, then, that to divide two numbers, we subtract their logs. Eqn 3-1 did not specify whether the exponents were positive or negative. Suppose that B^{-y} is used. From Eqn 3-5, this may be interpreted as $1/B^y$, then the following reasoning may be applied to support the statement that division of numbers involves subtraction of their logs:

$$\frac{B^x}{B^y} = B^x \times \frac{1}{B^y} = B^x \times B^{-y} \qquad \text{Eqn 3-6}$$

using Eqn 3-1: $B^x \times B^{-y} = B^{x-y}$ Eqn 3-7

Because it is so important, the following statements should be re-examined and understood:

a. The logarithm of a number is an exponent.

b. To multiply two or more numbers, add their logarithms.

c. To divide by a number, subtract the logarithm of the divisor from the logarithm of the dividend.

A slide rule is an example of an analog device which adds and subtracts physical distances to do multiplication and division. The numbers on the slide rule are placed so that their relative positions are proportional to the logarithms of those numbers. The scale is said to be logarithmic and this type of division occurs quite frequently in the instruments a communicator uses. The scale shown in figure 3-2 was taken from a slide rule; note the relative positions of numbers. The numbers compress, becoming closer together on the right end of the scale, indicating that the difference between log 9 and log 10, for example, is much less than the difference between log 2 and log 3.

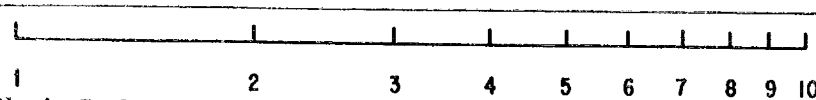


Figure 3-2. **Logarithmic Scale.**

The following examples will show how logs are used in computation and will conclude this brief explanation of logarithms as a tool. In the next paragraph, we will put this knowledge to use in defining decibel (dB). Perform the indicated mathematical operations using logs:

(1) $\frac{100}{10}$:

$$\log 100 - \log 10 = \log \frac{100}{10}$$
$$2 - 1 = 1$$
$$1 = \log \frac{100}{10}$$
$$\text{therefore } \frac{100}{10} = 10^1 = 10$$

(2) 100 x 10:

$$\log 100 + \log 10 = \log (100 \times 10)$$
$$2 + 1 = \log (100 \times 10)$$
$$3 = \log (100 \times 10)$$
$$\text{therefore } 100 \times 10 = 10^3 = 1000$$

(3) $\frac{100}{50}$:

$$\log 100 - \log 50 = \log \frac{100}{50}$$
$$2 - 1.7 = \log \frac{100}{50}$$
$$.3 = \log \frac{100}{50}$$
$$\text{therefore } \frac{100}{50} = 10^{.3} = 2$$

3-3. The dB and Its Meaning. Consider a car with a 4.11 to 1 rear end in it. The 4.11:1 ratio is a measure of the number of turns of the drive shaft (4.11) compared to the number of turns of the rear wheels (1). The same ratio could be found by counting the number of teeth on the crown gear (37) and dividing by the teeth on the pinion (9). This process is shown in figure 3-3.

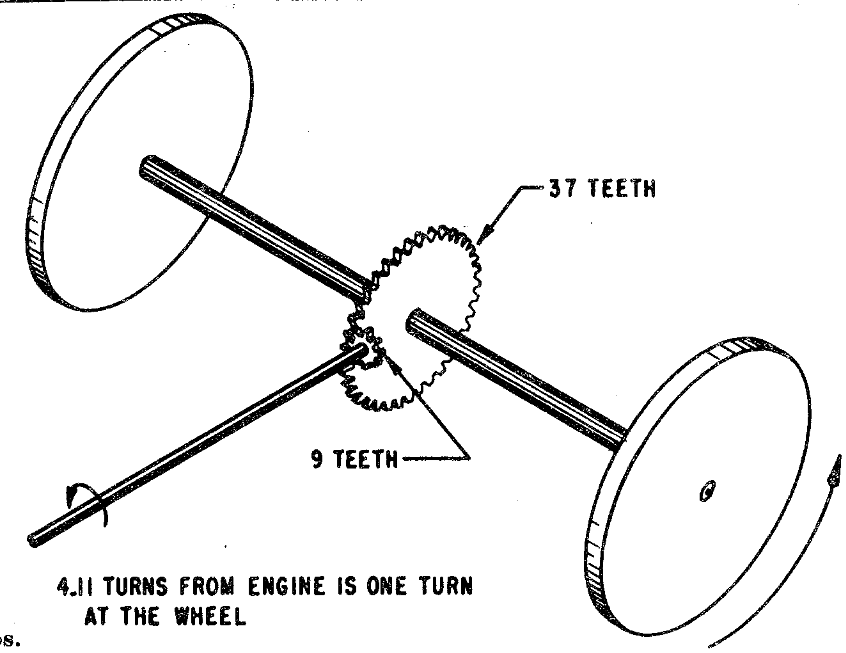


Figure 3-3. Ratios.

$$\frac{37 \text{ teeth}}{9 \text{ teeth}} = 4.11 \text{ ratio} \qquad \text{Eqn 3-8}$$

Now consider an amplifier which will give an output of 1.11 watts of power for an input of 0.27 watts. The output of this amplifier could be compared to its input and the result expressed as a ratio:

$$\frac{1.11 \text{ watts}}{0.27 \text{ watts}} = 4.11 \qquad \text{Eqn 3-9}$$

The results of Eqns 3-8 and 3-9 are identical, even though gear teeth were the subject of Eqn 3-8 and watts were the subject of Eqn 3-9. It would be meaningless to mix gear teeth and watts but, as long as the units of measure are the same, we can express two quantities as a ratio. A ratio of two numbers is without units of measure; it is a pure number and by looking only at a ratio, you can't tell whether the number was derived from automobile rear ends or amplifier calculations.

The ratio of two numbers can be found by using logarithms; this was done in examples (1) and (3) of paragraph 3-2. The ratio of $\frac{100}{10}$ was found to be 10. The ratio of $\frac{100}{50}$ was 2. These ratios could be expressed as logs and then left that way. For example, find the ratio of 100 to 50 and express this as a log:

$$\log 100 - \log 50 = \log \frac{100}{50}$$

$$2 - 1.7 = \log \frac{100}{50}$$

$$.3 = \log \frac{100}{50} \qquad \text{Eqn 3-10}$$

$$.3 = \log 2 \qquad \text{Eqn 3-11}$$

There is a small difference between Eqns 3-10 and 3-11. In Eqn 3-10, the process of division (expressing the ratio between 100 and 50) was indicated; in Eqn 3-11, this division has been done. This shows that the ratio of 100 to 50, 50 to 25, 12 to 6, or 2 to 1 are all the same, namely 2:1 and that this ratio expressed as a log also remains unchanged.

Early in this century, engineers and technical people found themselves dealing with many ratios: ratios of sound power, ratios of input to output power, ratios of power improvement. Many of these items were logarithmically related. Additionally, logs provided certain other advantages, some of which were discussed previously. The term "Bel" was proposed (in honor of Alexander Graham Bell) as a way of indicating that a power ratio has been expressed in log form. In order to be more practical for daily work, one-tenth of a Bel was adopted, using the prefix "deci" (1/10 th) to indicate this smaller quantity. The term "Bel" is not common, decibel (dB) being preferred.

THE DECIBEL IS A POWER RATIO EXPRESSED AS A LOGARITHM AND MULTIPLIED BY 10:

$$\text{dB} = 10 \log \frac{A}{B} \qquad \text{Eqn 3-12}$$

Using this definition and equation 3-11, we see that a ratio of 2:1 is the familiar 3 dB. If we wanted a ratio of 1:2 or 1/2, the ratio in dB becomes -3 dB, which can be derived from the information presented earlier.

The path of a communications signal always flows through devices which will multiply the signal power; sometimes the multiplier is a number less than one (1/2, for example) thus creating a loss instead of a gain. Figure 3-4 shows two amplifiers that double the input power connected to a pad which halves the input power. By multiplying the gain figures, the power out is found to be 2 x 2 x 1/2 = 2 times greater than the input. By adding the dB gains and losses, the output is 3 + 3 - 3 = +3 dB. The 3 dB output does not indicate the level of power out of the system; it does not say what frequency is involved nor what the impedance is; it indicates only that the output is 3 dB higher than the input.

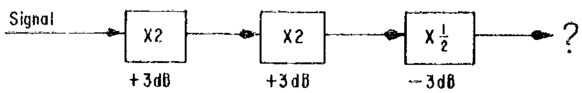


Figure 3-4. Power Ratios Expressed Two Ways.

Table 3-1 will add more meaning to the relative size of the dB and allow more flexibility for other examples. Note that the only difference between ratios greater than 1 and ratios less than 1 is a sign change in the dB ratio. The ratio 80:1 is 19 dB. The ratio 1:80 or .0125:1 is -19 dB. Also note that every time a ratio doubled, 3 dB was added. When power is halved, 3 dB is lost.

Figure 3-5 is a two-stage amplifier followed by two 3 dB pads. The input has been set at .001 watts and the chart shows graphically what is happening to the signal. This figure will be further expanded in the paragraph on dBm.

3-4. The Jump to dBm. The ratio of two numbers expressed as a logarithm X10 is a number identified by the word "decibel." Ratios are comparisons between like quantities; teeth to teeth, volts to volts, milliwatts to milliwatts; therefore, a ratio in dB doesn't say what power level is involved, only how much difference there is between two quantities. A difference of +13 dB may be 1 mW to 20 mW or it may be 1 kW to 20 kW.

By agreeing on a unit of reference, all other levels can be compared to a single point. For the dBm, the reference is 1 mW of power. In order to express any other power in dBm, it must be compared to 1 mW of power. What is the ratio of 100 mW to 1 mW? One hundred, of course! From table 3-1, a ratio of 100:1 is 20 dB; however, since we have compared 100 mW to the reference level for dBm, this is also 20 dBm.

Look at figure 3-5 again and note that the input to the 3 dB amp is 1 mW. When this input is compared to the reference level for dBm, the ratio is 1:1 or 0 dBm. The first stage doubled the power to 2 mW, 0 dBm + 3 dB = 3 dBm, the power level in dBm out of the first stage. The remaining points are +13 dBm, +10 dBm, and +7 dBm respectively. If the input is changed to 1 watt, some changes occur (figure 3-6).

The input is now 30 dBm (1000 mW=1 watt). What was formerly an increase of 1 mW in the first stage is now 1 watt; this is 999 mW more, but the increase in dB from input to output is still 3 dB. The graph is now plotted in dBm instead of mW, which has changed in shape, even though the numbers remained the same. By plotting power levels in dBm instead of watts, the same span of values can be plotted in less space. The "compression" effect of logs allows a new perspective when examining levels by placing equal emphasis on equal ratios, rather than on equal power changes (a 3 dB loss at one point in figure 3-6 is 10 watts, at another point it is only 5 watts).

Numerical Power Ratio	*Power Ratio in dB*	*Numerical Power Ratio*	*Power Ratio in dB*
1:1	0 dB	1:1	0 dB
2:1	3	1:2	-3
4:1	6	1:4	-6
8:1	9	1:8	-9
10:1	10	1:10	-10
20:1	13	1:20	-13
40:1	16	1:40	-16
80:1	19	1:80	-19
100:1	20	1:100	-20
160:1	22	1:160	-22

Table 3-1. Power Ratios Converted to Ratios in dB.

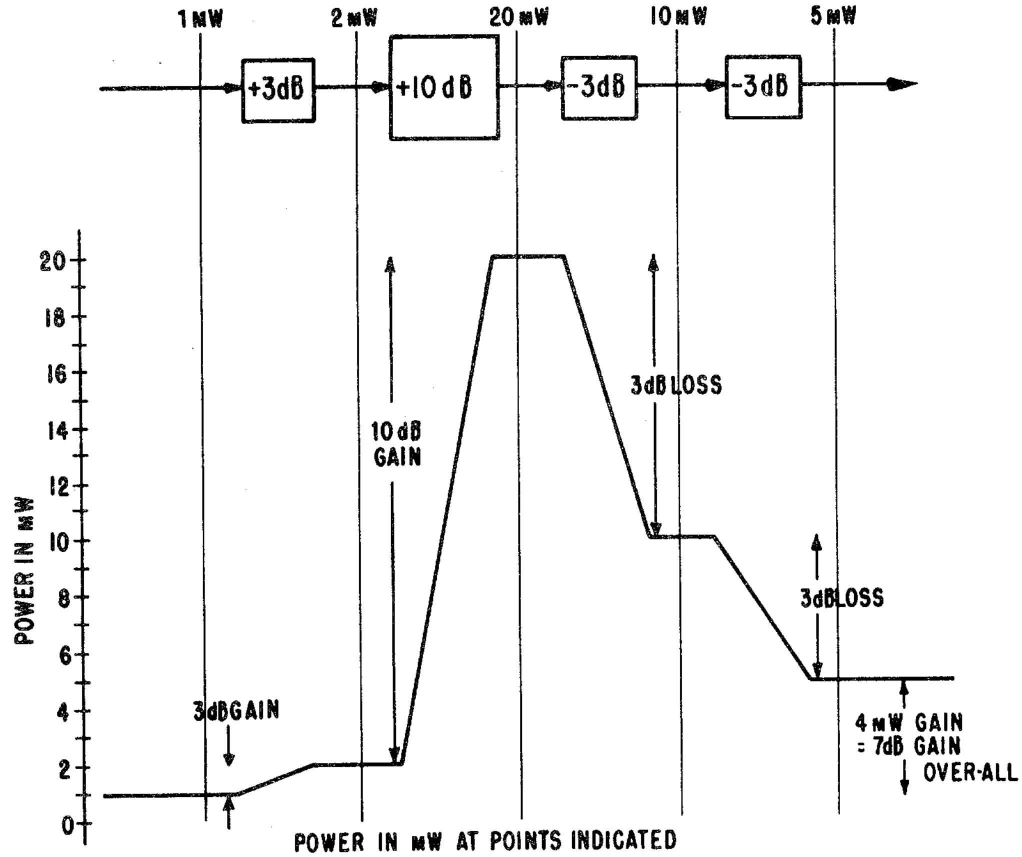


Figure 3-5. Graph of Power Levels. Note that a 3 dB gain or loss is not necessarily the same number of milliwatts at all points.

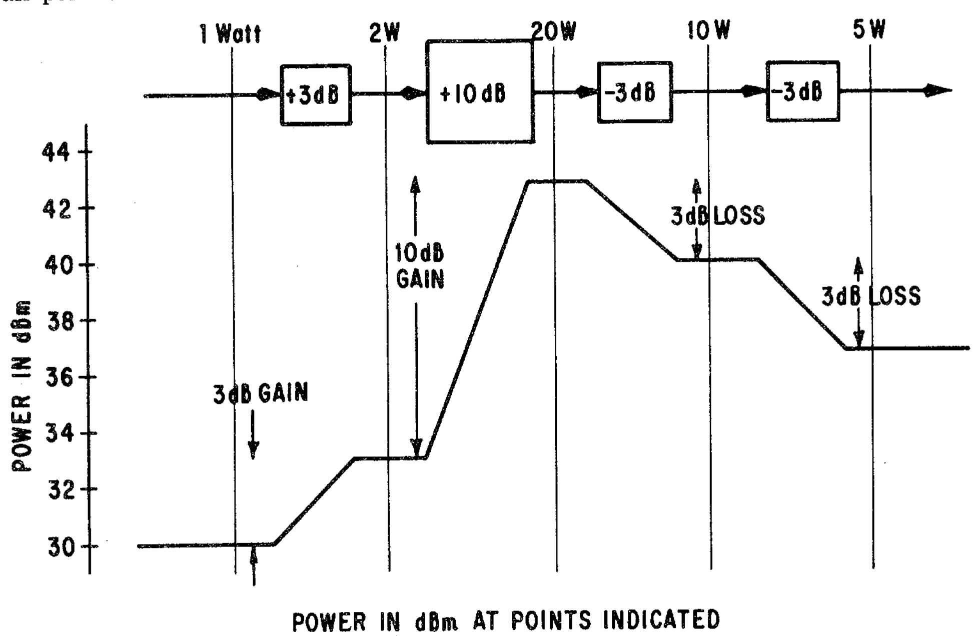


Figure 3-6. Graph of Power Levels in dBm.

One milliwatt is not the only possible reference. One watt could become the reference level; the unit of power would then be the dBW, figure 3-6 would then start out at 0 dBW at the bottom and increase to +13 dBW.

With the exception of final power output stages for transmitters, the dBm is more convenient and more common for use.

THE dBm IS A UNIT OF POWER

$$\text{dBm} = 10 \log \frac{A \text{ mW}}{1 \text{ mW}} \qquad \text{Eqn 3-13}$$

3-5. The Use of dB and dBm. It is sometimes easy to forget that dB and dBm are logarithmic and, therefore, must follow the rules applied to logs. The addition of logs is a process of multiplication; similarly, the subtraction of logs is a process of division.

Two signals, one at +3 dBm and the other at +9 dBm level are being fed into a hybrid transformer. Assuming that the transformer is loss-free, it will neither add nor subtract power from the system. What is the output in figure 3-7 then?

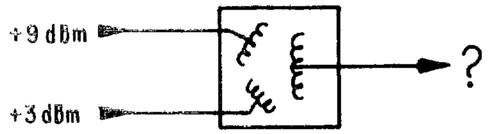


Figure 3-7. Hybrid Transformer.

The first impulse would be to add 9 dBm and 3 dBm, however, this would correspond to a multiplication process. Since the hybrid is adding the two power levels, it is necessary to convert back to actual power levels, add them, then convert back to dB. Using table 1, the output would be computed as:

2 mW +8 mW = 10 mW or 10 dBm

Charts and tables are available to allow the operation to be done without conversion.

If a composite signal with a level of -13 dBm is to be formed from 16 equal inputs, what is the dBm level of one input in figure 3-8? Since all the inputs are equal, the ratio of the output to any particular input is the same as any other. Since our black box adds all the inputs without loss, this ratio must be 16:1. Looking at table 3-1, 8:1 is found to be 9 dB; mentally double this to 16:1 and reason that this adds 3 dB, therefore, 16:1 is 12 dB. Each input must then be 12 dB lower than the output or -25 dBm.

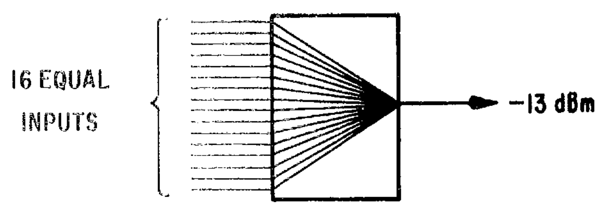


Figure 3-8. Composite Levels.

By expressing gains, losses, and powers in dB and dBm, it becomes a simple matter of addition to determine the levels in a system. As one technician aptly put it, "By using dB, I stay away from using ridiculous numbers." Consider a power of -18 dBm. When converted to milliwatts, this becomes .01585 mW. Now what would the power be when fed through an amplifier with a gain of 316 (25 dB)? The answer is 316 x .01585 mW = 5.009 mW or, more simply: +25 dB + (-18 dBm) which is +7 dBm. The algebraic addition in dB avoids the use of "ridiculous" numbers.

3-6. The Test Level Point (TLP) and dB. Test Level Points or Transmission Level Points are marked in dB in tech control center diagrams, on active equipment jacks, and in the schematics for the equipment. The TLP is specified in dB and establishes a reference level from which test tone levels may be reckoned. A TLP is nothing more than a point where a signal may be tested or measured. The dBm level specified at a TLP is an engineered level and is the important point to understand. This level is the level which would appear at that point if a 0 dBm signal is inserted at a Zero Test Level Point (0 TLP) "up-stream" from the point where the measurement is made. In practice, you don't have to be at a 0 TLP to start the signal going; inserting a signal of -7 dBm at a -7 TLP would produce the same effect down the line as a 0 dBm signal at a 0 TLP.

The concept of TLPs allows the gains and losses over a path to be monitored and adjusted. A known level is inserted at point A; when this level is known and TLPs further down the line are known, corrections can be made. Figure 3-9 shows how this might be done.

At Site A, a signal 10 dB down from the specified TLP value is inserted; therefore, it should appear 10 dB down at all TLPs. In practice, a man at Site D might ask for a "10 down" tone from Site B. He really doesn't care what the TLP at Site B is; just so long as Site B knows and ensures that he's 10 down going in, the check at Site D can be performed.

The dBm0 formalizes the above practice and allows written recording of these levels without need for any other explanation:

A level given in dBm0 is given with reference to the TLP:

$$\text{dBm0} = \text{X (dBm)} - \text{TLP (dB)} \qquad \text{Eqn 3-14}$$

Using Site B in figure 3-9 as an example, "X" in equation 3-14 would be the actual level measured (-26 dBm). The TLP is -16 dB. Using Eqn 3-14:

$$\text{dBm0} = -26\ \text{dBm} - (-16\ \text{dB})$$

$$= -26\ \text{dBm} + 16\ \text{dB}$$

$$= -10\ \text{dBm0}$$

This result states that a signal of -26 dBm measured at a -16 dB TLP is -10 dBm0 or "10 down." Zero dBm0 is always the value of the TLP and "up" and "down" are reckoned (referenced) with respect to that level.

Examples:

1. A -10 dBm0 tone is passing through your equipment. You are at a +7 TLP and are reading the tone at -9 dBm:
   a. Is the tone "hot" or "cold?" How much? 6 dB "cold" or -16 dBm0
   b. What should the level be? -3 dBm or -10 dBm0

2. At a -16 TLP, what level will you use to put a -5 dBm0 tone into the system? -21 dBm

3-7. Impedance and dB Measurements. In order to use the dB or dBm in practical work, it is vitally important to know how to measure these units with the meters available. Figure 3-10 shows four resistors, each with a different voltage being measured across it. All resistors shown above are dissipating 0 dBm; a voltmeter would read the values shown.

The power dissipated in each resistor can be calculated using Eqn 3-15.

$$P_R = \frac{(E_R)^2}{R} \qquad \text{Eqn 3-15}$$

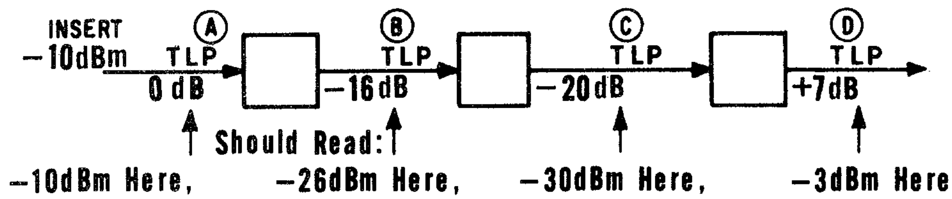


Figure 3-9. Test Level Points (TLP). Points A through D could be in the same equipment chassis or could be different sites in a system. The TLP is an engineered value which establishes a reference signal level at a point.

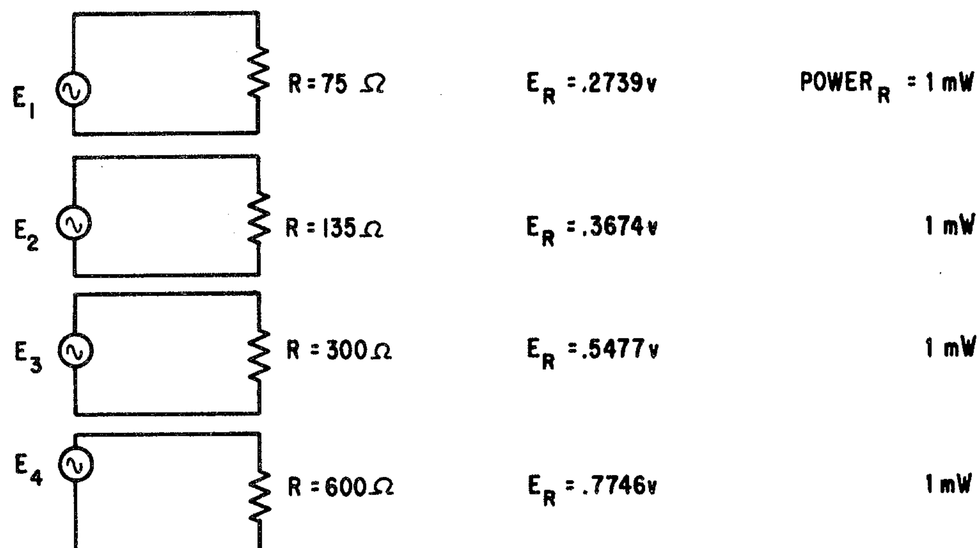


Figure 3-10. Zero Reference Level for dBm.

The power through each resistor is the same, while the voltages vary greatly. Zero dBm can be measured across any value of resistance; the difficulty comes in converting the voltage read from a voltmeter into power without cranking out the mathematics of Eqn 3-15.

Most VOMs and VTVMs have a scale marked in dB; also, there will generally be a note that 0 dB = 1 mW across a specified resistance. Assume that the meter used for the above measurements specified that 0 dB = 1 mW across 600 ohms. This means that the dB scale can be read in dBm directly from the meter (figure 3-11) when a voltage is being measured across 600 ohms. For example, the meter would read .77 volts (0 dBm) (figure 3-10) for the 600 ohms case. As a convenience to the user, the calculation of Eqn 3-15 has been done and the results plotted in dB(m) on the meter face. For the special case where R_L = 600 ohms, the dB scale can be read in dBm; for any other value of resistance, the scale must be read as a dB scale and a fixed correction factor added or subtracted to get dBm.

If .7746 volts is measured across 300 ohms, the dB scale will still read 0 dB, but this is no longer equal to 0 dBm. By holding E_R constant in Eqn 3-15 and halving R_L, the power is seen to double; therefore, the correction factor which must be used is +3 dB. With this factor taken into account, power in dBm can be derived from the dB scale on the meter. For example, across 300 ohms we read .55 volts or -3 dB. By adding the 3 dB correction, this becomes 0 dBm, which agrees with figure 3-10. In the general case, the correction factor to be added can be found from Eqn 3-16 where Z instrument equals the impedance for which the dB scale may be read in dBm. Z load equals the actual load across which the voltage is being measured.

$$\text{dB correction} = 10 \log \frac{Z \text{ inst}}{Z \text{ load}} \qquad \text{Eqn 3-16}$$

By again examining Eqn 3-15, it can be seen that, if the voltage across a resistor is doubled, it will have the effect of multiplying the power by 4 (+6 dB), due to the E^2 term. Looking at figure 3-11, a voltage of .5 volts is

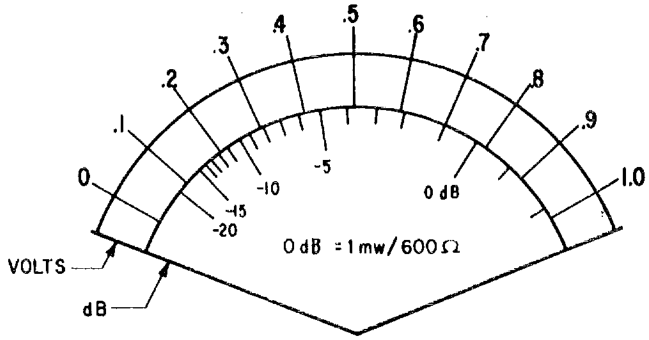


Figure 3-11. Volt/dB Meter.

read as -3.8 dB and when the voltage is doubled to 1 volt, the dB reading is 6 dB higher (2.2 - (-3.8) = 6 dB), as we should expect from analyzing Eqn 3-15.

The level of a signal must always be measured across a specified impedance, to ensure that the circuit is seeing the load it normally feeds. In some cases, it is necessary to bring the signal out of the system, disconnecting it from its normal load. Then, before an accurate level can be measured, the signal must be terminated in this "characteristic" impedance or load. Some meters have a "bridging" and a "terminating" mode with the load impedance switch-selectable and supplied internally.

Due to the complexity of modern communications systems, the number of test points, the variety of measurements, and the possibility of two measurements being made at the same point at the same time (that is, dB level and frequency), there always exists the possibility of an incorrect level measurement. Through carelessness or oversight, it is easy to either double-terminate a test point or measure the level without termination. To avoid such possibilities, the technician *must* know the impedance characteristics of each instument he uses.

Although no single rule can be stated to spot or avoid false measurements, the best indicator that an error in measurement has been made is a ± 3 dB disparity in levels. The reason may be seen by comparing the situations shown in figure 3-12. The part within the dotted box could represent the final output of a voice channel at the multiplex receive patch bay. The tone on the channel is originally measured across the correct impedance of 600 ohms. With the circuit double-terminated as in B, the voltage would be less and the apparent level is 3.5 dB lower. If the level were adjusted in B, then returned to service as in A, the circuit would be about 3 dB hot. A 6 dB error will result if a level is adjusted on an unterminated circuit.

In this paragraph, we have examined the relationship of impedance to both voltage and power (in dBm). The same voltage across different impedance corresponds to different power levels. Power across a known load resistor can be represented on the face of a voltmeter in dBm. The scale is marked in dB, which can then be used regardless of load and, in the special case of the stated resistance, the scale is read in dBm. Finally, double terminations generally cause a 3.5 dB error in reading. Circuits which are not terminated will show a 6 dB error.

3-8. Example Calculations. The following examples are included for the reader who wants additional practice using dB. You will get maximum benefit from the problems by trying them first before referring to the solutions.

a. What is the gain in dB of an amplifier having 600 ohm input and output impedances when the input level is 3 mW and output is 240 mW?

$$\text{dB} = 10 \log \frac{240}{3}$$

$$= 10 \log 80$$

$$= 10 \times 1.9031$$

$$= \underline{19 \text{ dB}} \qquad \text{(answer)}$$

b. What is the amplifier gain when the input is .10 mW and the output is 10 watts and input and output impedances are 600 ohms and 8 ohms, respectively?

$$\text{dB} = 10 \log \frac{10000}{.10} \qquad \text{change to same units}$$

$$= 10 \log 10^5$$

$$= 10 \times 5.00$$

$$= \underline{50 \text{ dB}} \qquad \text{(answer)}$$

(Note that the powers were stated; therefore, forget about the impedances)

c. There are two tones, one at -21 dBm, the other at -9 dBm. What is the dB difference?

$$\text{dB difference} = \text{dBm}_1 - \text{dBm}_2$$

$$= 21 - 9$$

$$= \underline{12 \text{ dB}} \qquad \text{(answer)}$$

d. What is a power level of 5 milliwatts expressed in dBm?

$$dBm = 10 \log P_1/P_2 \qquad (P_2 = 1\ mW)$$
$$= 10 \log 5$$
$$= 10 \times 0.6990$$
$$= 6.99 \text{ dBm or } \underline{7 \text{ dBm}} \quad \text{(answer)}$$

e. What is a power level of .25 milliwatts expressed in dBm?

$$dBm = 10 \log P_1/P_2 \qquad (P_2 = 1\ mW)$$
$$= 10 \log .25$$
$$= 10 \log 1/4$$
$$= 10 \log 1 - 10 \log 4$$
$$= 0 - 6.021$$
$$= -6.021 \text{ or } \underline{-6 \text{ dBm}} \quad \text{(answer)}$$

f. If a dB or VU meter is used to terminate a line and the meter is incorrectly left in the high impedance position, how much will the meter read if the actual level is -21 dBm?

(answer) Because the meter will read about 6 dB high, the level will be indicated as approximately -15 dBm.

g. A dB meter calibrated for 0 dBm = 1 mW in 600 ohms is used to measure a level in a 150 ohm circuit. The meter is used in bridging mode and reads -15 dB. What is the true level?

(answer) The correction factor is

$$dB = 10 \log \frac{R_1}{R2}$$
$$dB = 10 \log \frac{600}{150}$$
$$= 10 \log 4$$
$$= 10 \times 0.601$$
$$= 6 \text{ dB}$$

The correct level is, therefore, $\underline{-9 \text{ dBm.}}$ (-15 + 6)

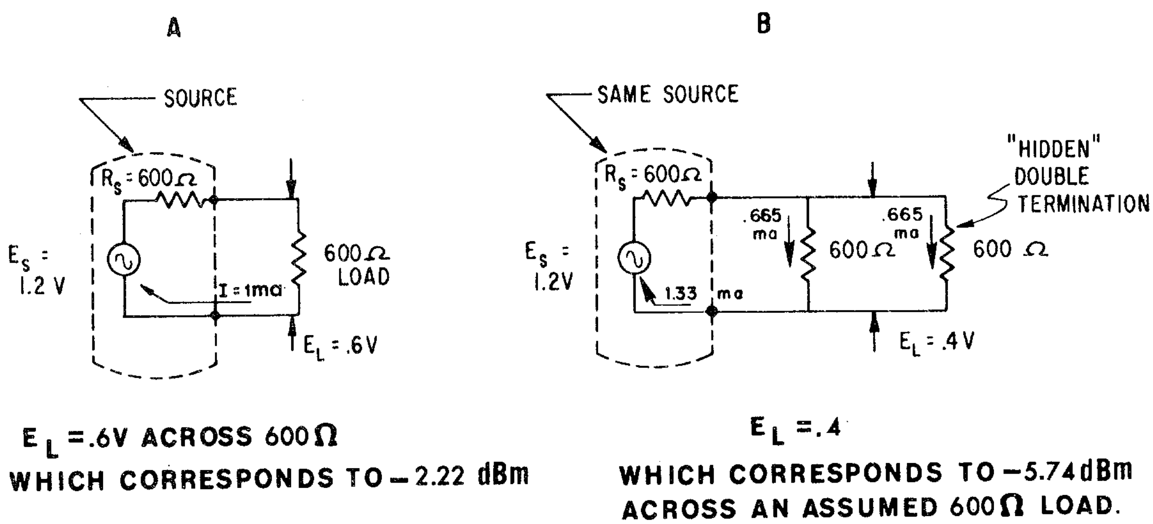


Figure 3-12. Termination Error. Comparison of a proper termination (A) and double termination (B) shows that more than a 3 dB error is introduced at B.

Chapter 4

SIGNALS AND NOISE

4-1. General. Information, in electrical communications, takes the form of amplitude, frequency, and phase varying signals. Reliable transfer depends on the information reaching the distant location in the same form as it was originally transmitted. This chapter deals with the characteristics of signals found in wideband systems and with the distortions and impairments they may suffer in transversing the system.

4-2. Signal Characteristics:

a. The Frequency Spectrum. Sinusoidal waves appear in nature over a tremendous range of frequencies. Household AC current has a frequency of 60 Hz while, at the other end of the spectrum, cosmic rays have a frequency of 1,000,000,000,000,000,000,000 Hz (one billion trillion cycles per second). Frequencies in between, and the phenomena they represent, are shown in figure 4-1. The low end of the frequency spectrum consists of audio, or sound, waves. The human ear can hear sounds from about 20 - 20,000 Hz. Above 10^{12} Hz, waves are commonly called rays. Visible light occurs in this region. Radio waves occur from about 10^4 Hz to 300 GHz (300 X 10^9 Hz). The prefixes used in specifying frequencies are shown in figure 4-1.

For convenience, the radio frequency spectrum has been divided into bands; each band spans a decade of frequencies. The band classification system now in use is shown in figure 4-2. Each band has its own properties and its own use. The microwave region includes the VHF band and above.

b. Basic Parameters of Sinusoidal Signals. All of the signals found in wideband systems contain sums of sine waves, products of sine waves, or both. A square wave, for example, can be constructed from a fundamental sine wave plus all of its odd harmonics; therefore, a knowledge of the simple sine wave is essential to the understanding of signals in wideband systems. *A sine wave can be characterized by three parameters: amplitude, frequency, and phase.* Amplitude and frequency are shown in figure 4-3.

In many cases, the peak value does not define a sine wave in the most usable form. Often it is desirable to express the value of a sine wave in terms of an equivalent DC level which would produce the same heating effect as the AC wave. The term given to this is the effective or Root Mean Square (RMS) value. For a single sine wave, the RMS value is $1/\sqrt{2}$ times the peak value. Waveforms other than sine waves will have different peak-to-RMS ratios. Like any periodic function, a sine wave has a period (T) after which it repeats itself. The number of cycles which the sine wave goes through in one second is the frequency expressed in Hertz (1 Hertz = 1 cycle/second).

Often, sine waves are described in terms of wavelength rather than frequency. The wavelength (λ) of a sine wave is shown in figure 4-4. The frequency and wavelength are related by the velocity of propagation of the wave through a medium. For propagation in free space, this velocity is essentially the speed of light (300,000,000 meters/second). Wavelengths are measured in units of length (meters, centimeters, etc.).

Phase refers to the time interval between one event and a second related event. As we use it, it refers to the time relationship of two or more sine waves. Instead of using time units to specify phase, we can measure the phase difference in fractions of a cycle. Considering one complete cycle as 360^0, any fraction of a cycle can be specified in degrees. Choosing an arbitrary reference as 0^0, phase allows us to determine the phase difference between two signals which is usually the parameter of interest. Figure 4-5 shows a cycle divided into degrees and presents two sine waves which are 90^0 out of phase. Often, fractions of a cycle are also specified in radians. The relationship between degrees and radians is shown below.

360 degrees = 2π radians

57.3 degrees = 1 radian

c. Complex Signals. Most of the signals we deal with in wideband systems are complex waves; that is, they are signals which are combinations of simple sine waves which have definite amplitude and phase relationships with respect to each other. An FM signal, for example, is composed of a carrier frequency plus an infinite number of sideband frequencies. There are two common techniques which allow us to observe and analyze complex signals. The first technique is to look at a complex wave on a time basis or in the time domain. Use of an oscilloscope is an example of time domain analysis. The oscilloscope display provides information about how the parameters of the wave change as a function of time. Time domain analysis can also provide information about the time relationship (that is, phase relationship) of two or more separate signals. Figure 4-6 is a time domain representation of a complex wave.

The second method of complex wave analysis is frequency domain analysis. This type of analysis allows us to examine a complex wave on the basis of its frequency content. It allows us to see the various components which comprise a complex signal. When we use a frequency selective voltmeter (FSV) or a spectrum analyzer, we are using a simple form of frequency domain analysis. Figure 4-7 is a representation of a complex wave in the frequency domain.

The selection of which method of analysis to use is based on the signal to be analyzed and what parameters are to be observed. For example, let's assume it is desired to determine the effects of passing an FM signal through a bandpass filter. For simplicity, we will assume a single tone is used to modulate the carrier. If we were to observe the signal in the time domain with an oscilloscope, the display would appear garbled both at the input and output of the filter and very little could be determined. In the frequency domain, however, the filter effects could be easily determined. Figure 4-8 shows this.

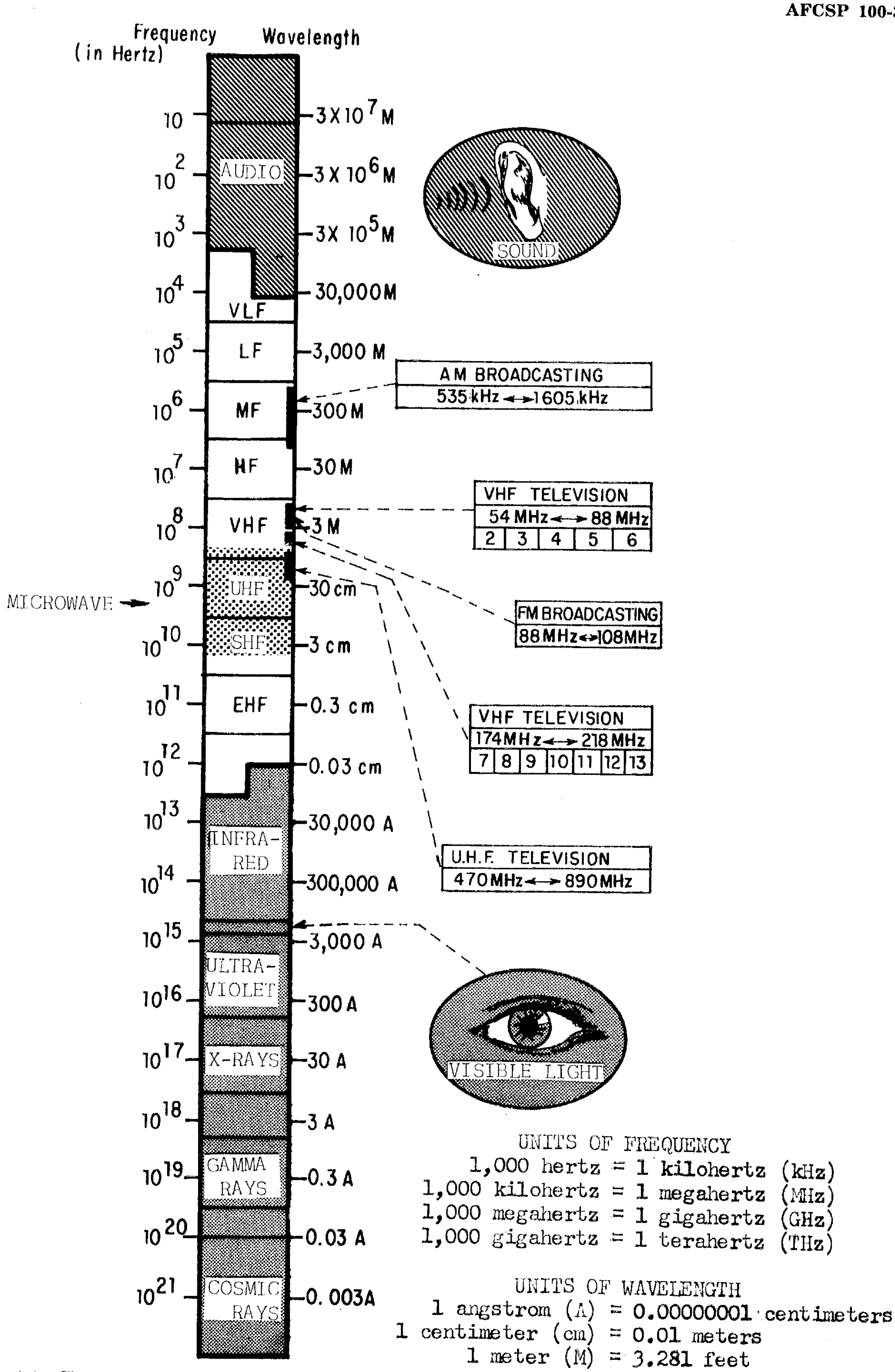


Figure 4-1. Frequency Spectrum.

FREQUENCY	CLASSIFICATION	ABBR
10 kHz to 30 kHz	Very Low Frequencies	VLF
30 kHz to 300 kHz	Low Frequencies	LF
300 kHz to 3000 kHz	Medium Frequencies	MF
3,000 kHz to 30,000 kHz	High Frequencies	HF
30,000 kHz to 300 MHz	Very High Frequencies	VHF
300 MHz to 3,000 MHz	Ultra High Frequencies	UHF
3,000 MHz to 30,000 MHz	Super High Frequencies	SHF
30,000 MHz to 300,000 MHz	Extremely High Frequencies	EHF

Figure 4-2. Frequency Classification.

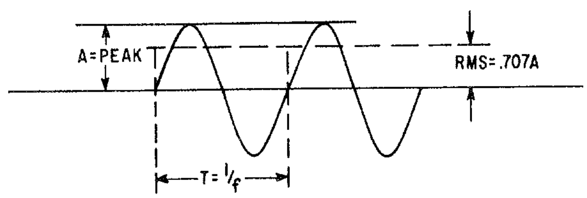


Figure 4-3. Sine Wave.

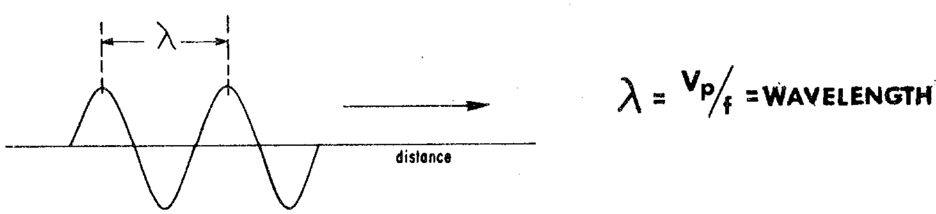


Figure 4-4. Sine Wave in Space.

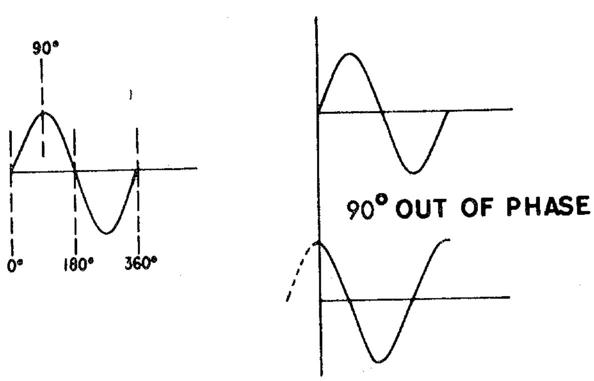


Figure 4-5. Amplitude and Frequency of a Sine Wave.

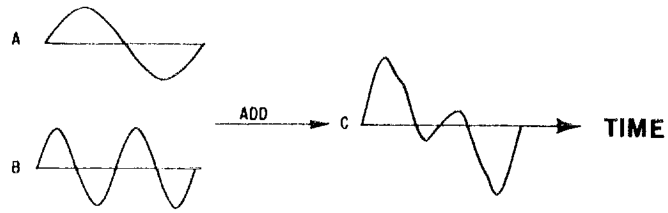


Figure 4-6. Complex Signal. Two sinusoidal waves add to produce a complex signal. Conversely, any complex signal can be represented by a sum of sinusoidal signals.

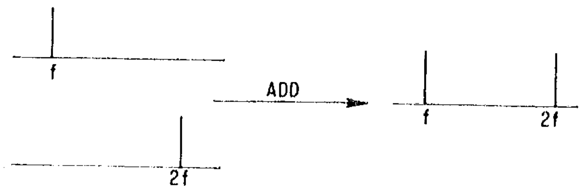


Figure 4-7. Frequency Domain Plot, Amplitude Versus Frequency. Frequency domain analysis of the addition shown in figure 4-6. Note that the frequency and amplitude content are shown in this diagram. Waveshape and phase information is not given.

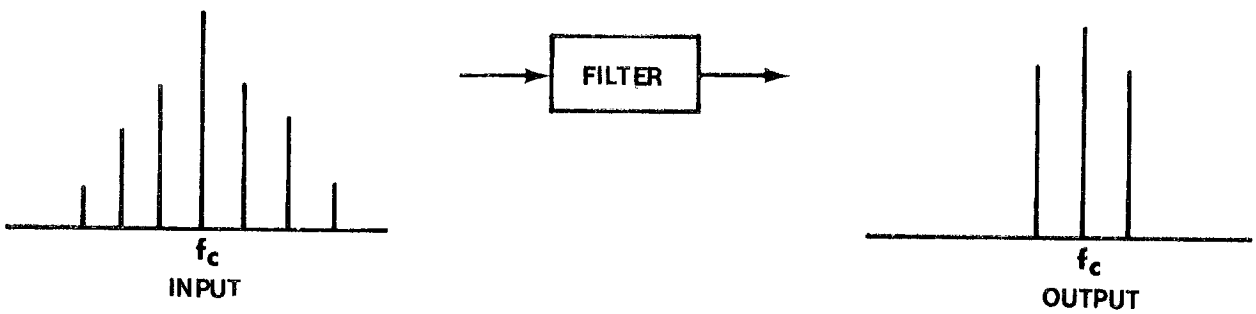


Figure 4-8. Effects of Filter on Frequency Domain.

It is apparent that the filter has altered the signal by attenuating all but the carrier and the first pair of sideband frequencies.

Both time domain and frequency domain techniques are important ways to analyze complex signals. Detailed analysis in the frequency domain can provide information not only about the spectral content of a complex wave but also about the phase relationship of the components. We will see in a later section on nonlinearities the importance of the amplitude and phase relationships in complex signals.

d. Characteristics of Speech. To understand the characteristics of a voice channel, we must first understand the unique characteristics of speech. These characteristics and numerical values have resulted from studying many, many speakers on a statistical basis.

A speech signal is a very irregular, complex signal. If viewed in the time domain, it appears as highly peaked waveform. In order to express the degree of variability of a signal, a figure called the peak factor has been defined. The peak factor is the ratio of the peak power (instantaneous power) of a signal to the average power. The peak power is related to the peak voltage. In communications work, the voltage used is that voltage which is not exceeded more than .001% of the time. If the peak voltage and RMS voltage of a signal are known, then the ratio of peak-to-average power in dB is 20 log(V_{peak}/V_{RMS}). The peak-to-RMS ratio of a single speech signal is 19 dB (figure 4-9).

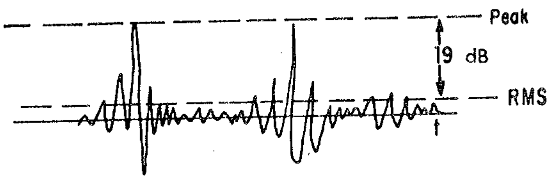


Figure 4-9. Signal Peak-to-RMS Voltage. The peak-to-average power ratio of a typical speech signal is 19 dB. Peak voltage here is that level exceeded less than 0.001% of the time.

The sound intensity of speech also varies widely (figure 4-10) due to the wide diversity of talkers. The dynamic speech range of a single talker is about 40 dB; however, the strongest sounds of a loud talker may be 70 dB above the weakest sounds of a soft talker.

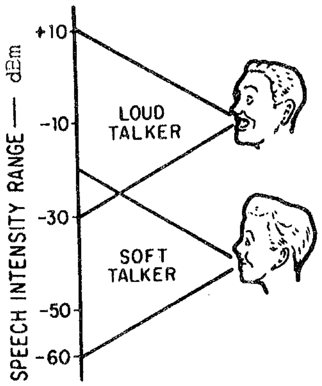


Figure 4-10. Speech Intensities of People. (Courtesy GTE Lenkurt)

In the frequency domain, speech sounds are basically limited to frequencies below 5 kHz. The distribution is not uniform; 80% of the energy is contained in frequencies below 1 kHz.

While the intensity of a speech signal is one indicator of its quality, it is not the only one. Intelligibility is another important consideration. Strangely enough, intensity and intelligibility are independent of each other. While 80% of the energy is carried in frequencies below 1 kHz, 80% of the intelligibility is carried in frequencies above 1 kHz. This means that both low and high frequencies must be included in any voice channel; however, bandwidth must be limited for efficient use of the frequency spectrum so a compromise must be reached. Many experiments have been done to test subjective reactions to bandwidth restrictions. As a result of these tests, the "standard" voice channel bandwidth is 3.1 kHz from 300 Hz to 3.4 kHz.

e. Data on Voice Channel. Since a voice channel is almost universally available, other services using digital data design their equipment to operate over voice channels. A voice channel designed for speech signals may present problems when it is used for transmitting data. The bandwidth required for data transmission depends on the speed of the data. The higher the speed, the more bandwidth required. High speed data requires far greater bandwidth than a voice channel can supply.

In contrast to voice, binary data signals are generally regular in pattern and flow continously in a channel. The peak-to-average power ratio for a frequency shift keyed data signal is only 3 dB (compared to 19 dB for a voice signal).

f. Statistics of Many Voice Channels. When many voice channels are put together, the resulting complex signal can only be described statistically. The equivalent power of a multichannel signal is dependent, not only on speech level distribution, but on circuit use as well.

Due to the nature of speech conversation, a busy channel (one to which a user is connected) does not have a continuous input. One party is listening while the other is talking and there will be punctuation pauses in the conversation. Extensive investigation has been done to determine the number of channels (n) actually transmitting power (which is termed "active") compared to the nominal number of channels that are busy (N). The ratio of n/N is called the activity coefficient or activity factor (k). The activity factor depends on the number of channels in the system. Beyond 240 channels, the average effect between channels is stable and the activity factor is essentially constant. Below 240 channels, however, the factor increases as N decreases; the averaging effect between channels is less. Measured activity factors are shown in table 4-1.

ACTIVITY COEFFICIENT		
Number of Channels		*Activity Coefficient*
Nominal, N	*Active, n*	*k*
12	7	0.58
24	11	0.47
60	23	0.38
120	41	0.34
240	77	0.32
600	180	0.30
960	288	0.30

Table 4-1. Measured Activity Factors.

These statistics were used to establish a mean per channel loading level for speech systems with pilot and signaling tones considered. This level is generally accepted at -15 dBm0 for systems of 240 channels or greater and - 1 -6 log (N) for N less than 240. These loading levels are International Radio Consultative Committee (CCIR) standard. DCA loading levels per channel are -10 dBm0, which anticipates 100% data loading.

Data channels nearly always have activity factors of 1; hence, the mean level in a channel is the mean level of the transmitting data. Depending on the per channel level, data channels can load a system much more severely than voice channels.

g. Multichannel Peak Factor. When we consider the final multichannel signal, there are two properties that are important to us: the RMS level of the composite signal and the multichannel peak factor. Since all signals are considered as random and add on power basis, the RMS level of the composite signal is the sum of the effective per channel levels. The peak factor, however, is a statistical quantity. We define the peak of a complex signal as that level not exceeded more than .001% of the time. We have seen that single speech signal has a peak factor of 19 dB; however, when many speech signals are put together, the peaks do not add together as much as the RMS level adds (due to the random nature of the peaks). The peak factor then tends to decrease as N increases. For systems with more than 60 active channels, the peak factor is essentially a constant at 13 dB (figure 4-11).

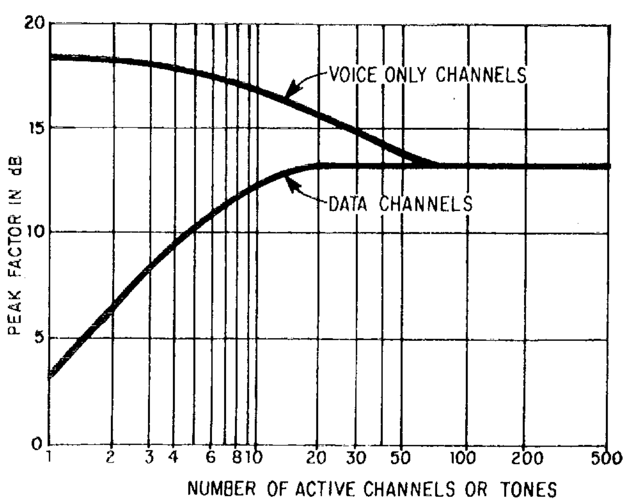


Figure 4-11. Peak Factor. (Courtesy GTE Lenkurt)

A single data signal is a fixed, non-random waveform with a peak factor of 3 dB; however, as many data signals with different frequencies and phases are combined, the composite signal becomes increasingly random. For N greater than 20 active channels, the peak increases to a constant value of 13 dB, the same factor we use for multichannel voice systems.

h. Signaling Tones. In addition to the voice and data information in the channels, other signals are present on a wideband system. Signaling tones are used for establishing connections, announcing incoming calls, etc. In general, signaling tones are contained within the channel they supervise. A keyed signaling tone can be placed at the top edge of a channel, beyond the voice band, or it can be within the band (usually at 1600, 2400, or 2600 Hz). Signaling tones do not operate when a channel is active, so collocation with voice frequencies does not interfere with normal conversation.

i. Pilot Tones. Another class of signals in the baseband is pilot tones. These tones are used mostly for level monitoring, alarm activation, and synchronization purposes. Each group of 12 channels has a pilot tone associated with it. The pilot tone level is monitored to ensure system alignment. Most multiplex equipment generates a pilot at 96 kHz which synchronizes the frequency of all the oscillators in the system. This is extremely important when single sideband suppressed carrier modulation is used.

A final pilot signal is generated by the radio equipment. It is used to monitor the RF signal and activate alarms if there is a loss of signal.

4-3. Distortion:

a. Introduction. The problem of distortionless transmission is a basic one in many fields of communication in which the shape and appearance of the input must be retained after passage through electrical circuits, through wire or cable, through air, etc. A system producing a distortionless output is said to be linear. If a signal is to be passed through a system without distortion, the overall system response must have a constant amplitude response characteristic over the frequency spectrum of the input and its phase shift must be linear over the same frequency range. A signal may be distorted by passing through the different parts of a system, but phase or amplitude equalizing networks may be introduced to correct for distortion. It is the overall characteristics that determine the ultimate output and distortion of the input signal.

b. Amplitude and Phase Shift. The amplitude and phase shift characteristics of transmission systems are of fundamental importance since departure from a "linear system" will introduce distortion in the output signal. Any transmission system can be approximated by a network which will have a specific attenuation (gain) and phase shift as a function of frequency. A typical variation of attenuation and phase shift for a low pass filter is shown in figure 4-12.

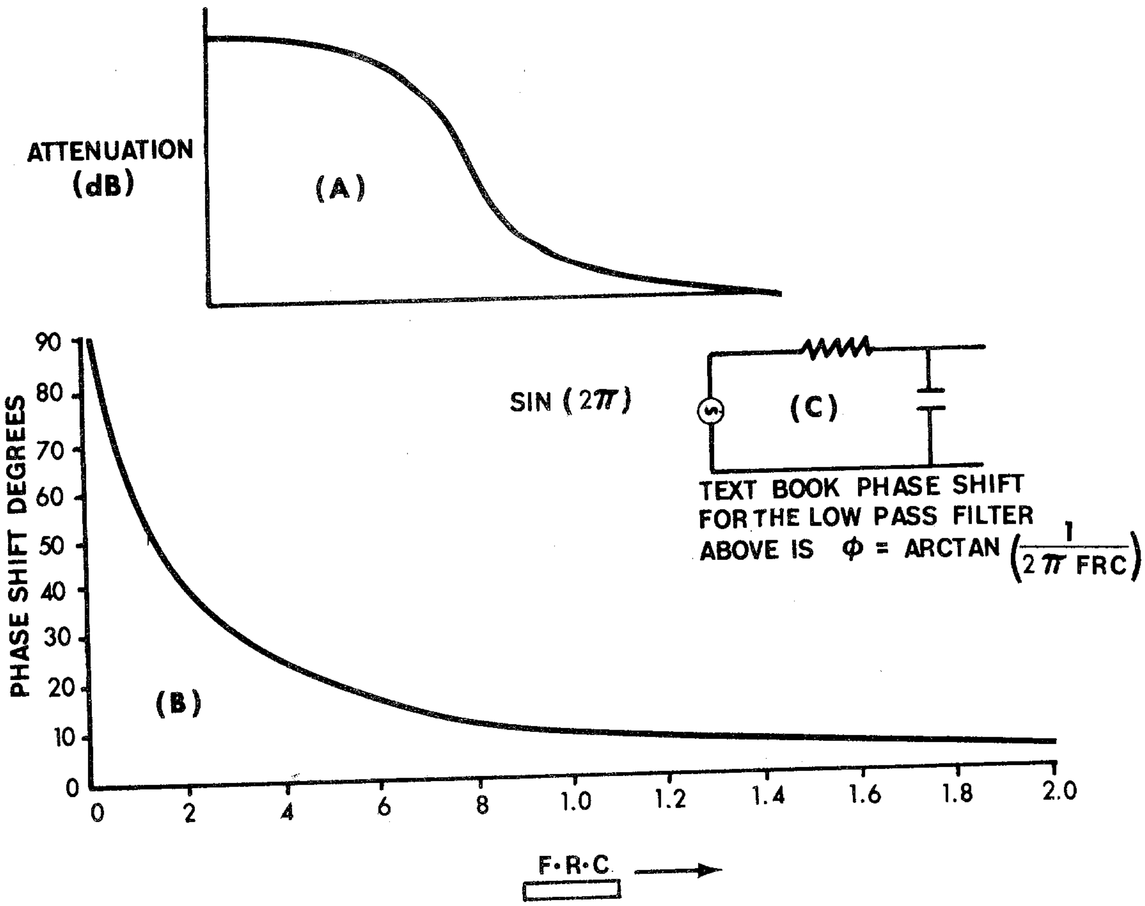


Figure 4-12. Attenuation and Phase Shift.

c. Phase Delay. The time delay between the input and output wave form is called phase delay or propagation time. If the phase shift characteristics are known, the propagation time (T_p) can be computed for any frequency by Eqn 4-1.

$$\text{Propagation Time} = \frac{\text{Phase Shift (Degrees)}}{\text{frequency (Hz) } 360} \quad \text{Eqn 4-1}$$

As an example in the use of this equation, suppose that the phase shift at 1000 Hz is 5^0 and that it is 15^0 at 3000 Hz (linear phase shift); then the propagation time for the two frequencies will be the same and no distortion of the output signal will be produced. If the waveform made up of the two frequencies was transmitted over a system with non-linear phase delay characteristics, the output waveform would be severely distorted because of the differences of the arrival times of each component at the output. This is shown in figure 4-13.

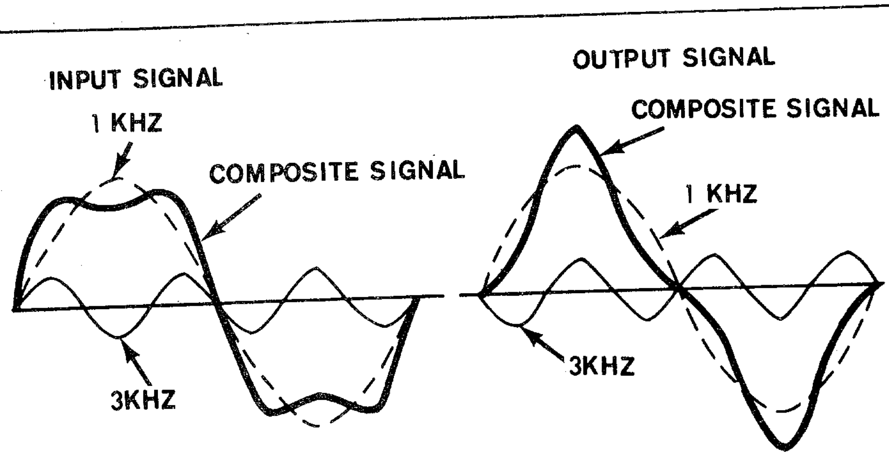


Figure 4-13. Effects of Non-Linear Phase Shift Characteristic on the Composite Signal Made up of a 1 kHz and 3 kHz Sine Wave.

d. Envelope Delay Distortion and Envelope Delay. It is not practical to measure the phase delay of a circuit by sending a single frequency due to the difficulty of establishing a phase reference; however, the derivative of phase with respect to frequency can be measured. This is called envelope delay and is defined by Eqn 4-2. A typical graph of envelope delay is shown in figure 4-14.

Envelope Delay $(T_e) = d\phi/d\omega \approx \Delta\phi / \Delta\omega$ Eqn 4-2

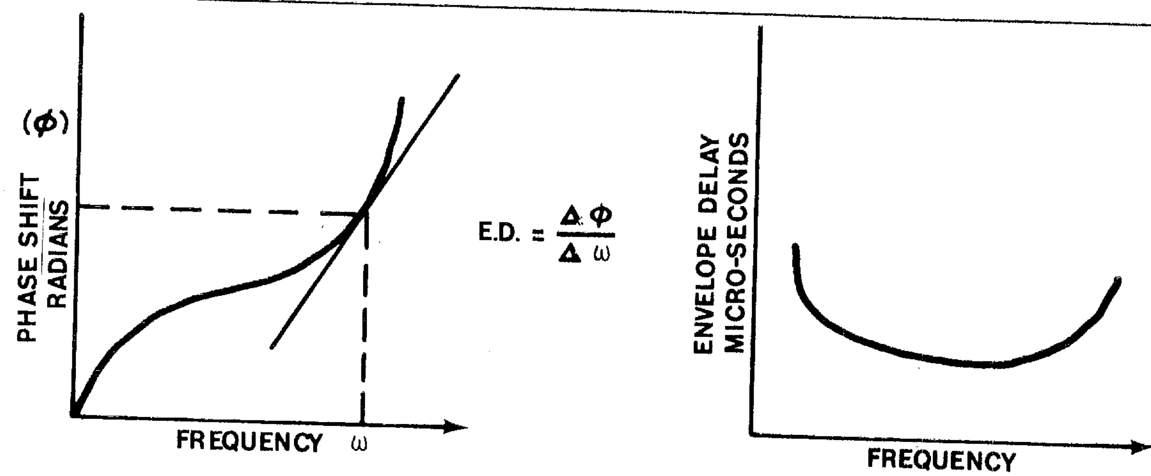


Figure 4-14. Envelope Delay.

Envelope delay is measured by applying a Voice Frequency (VF) carrier amplitude modulated by a low frequency signal to the channel. The VF carrier is then varied in frequency and the variations of the phase of the low frequency signal are measured. This is discussed further in chapter 19.

The most talked about form of distortion in communications is envelope delay distortion. This is defined as the maximum difference of the envelope delay characteristics over a band of frequencies but is not directly related to delay distortion (T_d). Envelope delay has no physical significance except to compare it with other values determined in a similar manner for other facilities. Envelope delay distortion is shown in figure 4-15.

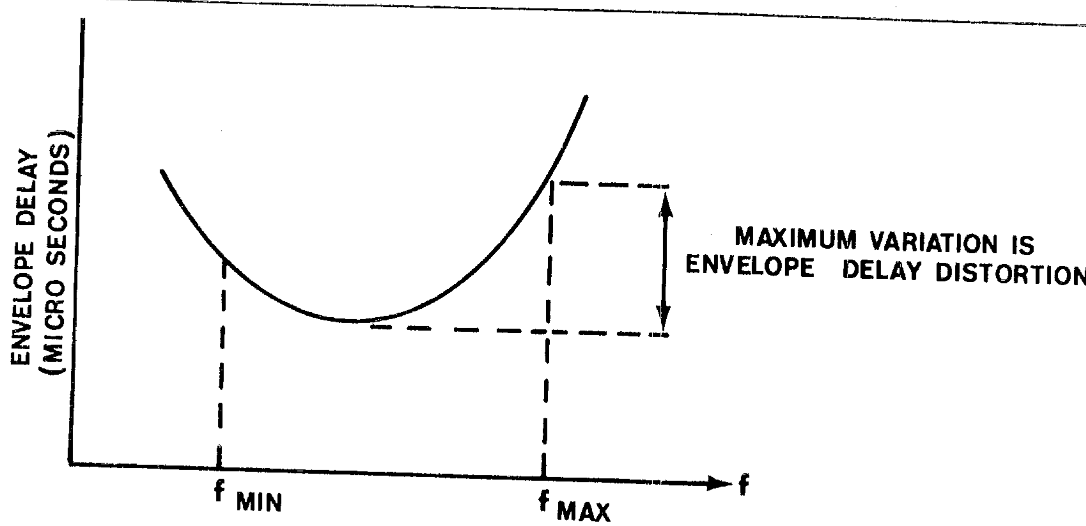


Figure 4-15. Envelope Delay Distortion.

e. Non-Linear Distortion. Non-linearities, as discussed previously, result in the generation of signal components which were not present in the original signal. The addition of these new signal components to the original signal results in a form of distortion termed non-linear distortion. Usually, non-linear distortion is identified and measured by the effect it has on certain signals. For example, if a single frequency is passed through a non-linear device, the results will be the generation of harmonics or harmonic distortion. Figure 4-16 shows this.

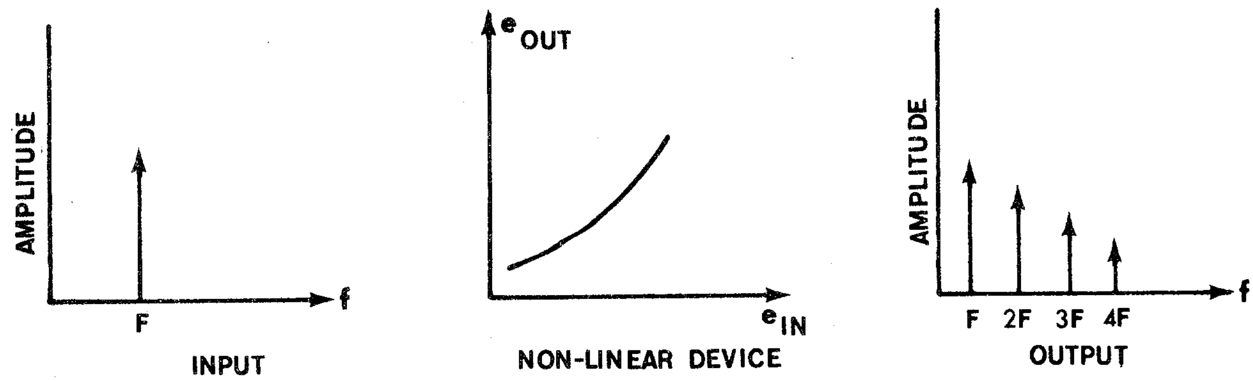


Figure 4-16. Harmonic Distortion.

The voltage transfer function of the non-linear device in figure 4-16 can be approximated by a series expansion as:

$$e_o = (e_i)(a_1) + (e_i)^2(a_2) + (e_i)^3(a_3) + \ldots \qquad \text{Eqn 4-3}$$

where a_n is the peak amplitude of individual sine waves.

Application of a single frequency sine wave ($e_i = A\cos\omega t$) so such a device results in the output being:

$$e_o = 0.5\, a_2 A^2 + [a_1 A + 0.75\, a_3 A^3] \cos(\omega t) + [0.5\, a_2 A^2]\cos(2\omega t) + [0.25\, A^3]\cos(3\omega t)$$

Examination of the output shows the generation of 2d and 3d harmonics of the fundamental. Further examination indicates that the amplitude of the 2d harmonic is proportional to a_2 and the square of the sine wave amplitude. Thus, for a 1 dB increase in the input power, a 2 dB increase in the 2d harmonic can be expected. Similarly, the 1 dB increase in input power results in a 3 dB increase in the 3d harmonic. Harmonic distortion is measured in terms of the power in the individual harmonics (2d, 3d, etc., order distortion) or the power sum of all the harmonics (total harmonic distortion) and is referenced to the power in the fundamental.

If two or more tones are passed through a non-linear device, additional components other than harmonics will be generated. These additional components create intermodulation distortion. This type of distortion appears as tones which are linear combinations of the input frequencies. The frequencies of the intermodulation products are given by the following relationship:

$$\text{Intermod Freqs} = k_1 f_1 \pm k_2 f_2 \pm k_3 f_3 \pm \ldots \pm k_n f_n$$

where the fs are the input frequencies and the ks are non-negative integers. Intermodulation distortion products are termed 2d order if the sum of $k_1 + k_2 + \ldots + k_n$ is equal to two and 3d order if the sum is equal to three. Higher order distortions are determined in the same manner. As an example, the 3d order intermod frequencies for input frequencies f_1 and f_2 would be:

$$f_1 \pm 2f_2 \text{ and } 2f_1 \pm f_2 \text{ when } k_1 + k_2 = 3$$

Intermodulation distortion, like harmonic distortion, is measured in terms of the powers of the various products and is usually referenced to the sum of the powers of the fundamental frequencies. Intermodulation and harmonic distortion are calculated from the following relations:

$$\text{Intermod/Harmonic Dist (dB)} = 10 \text{ Log} \left[\frac{\text{Sum of Powers of Dist Components}}{\text{Sum of Powers of Fundamentals}}\right] \qquad \text{Eqn 4-4}$$

$$\%\ \text{Distortion} = \text{Antilog} \left[\frac{\text{Distortion (dB)}}{20}\right] 100\% \qquad \text{Eqn 4-5}$$

The 20 factor in the % distortion equation is to convert to a voltage factor. The amplitude of the intermod distortion products behaves identically to the harmonic products in that a 1 dB increase in the input power causes a 2 dB increase in 2d order products and a 3 dB increase in 3d order intermod products. Relating this to the equations above shows that there will be a dB for dB increase in total distortion if 2d order distortion products dominate and a 2 dB for 1 dB increase in input if 3d order distortion products are predominate. Initially, in normal operation, 2d and 3d orders have the greatest amplitude with higher order products being negligible; however, if the input is continually increased, a point will be reached where higher order products become significant and total non-linear distortion increases rapidly for small changes in input power (as much as 20 dB for a 1 dB increase in input power). This point is called the system break point or overload point.

It should be remembered that the relative increases in non-linear distortion described above are based on an approximation and that actual measurements may not follow this approximation exactly. In actual circuits, the magnitude of the non-linearity may be dependent on input power as in the case of compandered systems; likewise, the approximation may fall short if terms above 3d order are not initially insignificant. Some of the causes of non-linear distortion are:

(1) Non-linearity in multiplex and baseband equipment.

(2) Non-intelligible crosstalk in multiplex equipment.

(3) FM transmitter non-linearity.

(4) Impedance mismatch and defects in waveguide and antenna.

(5) Multipath effects in propagation medium.

(6) FM receiver non-linearity.

(7) Non-linear phase delay.

f. Attenuation Distortion. Attenuation distortion (also called frequency response or amplitude distortion) is a difference in loss at one frequency with respect to the loss at another frequency. This can occur in conjunction with phase distortion but it is not necessarily part of it. Figure 4-17 shows an example of attenuation distortion. In this example, the frequency response of the device through which the signal is passed is such that it attenuates all frequencies greater than twice the input signal. As a result, the output bears little resemblance to the original signal. This is, of course, an exaggerated example, however, it does illustrate the effect of altering the relative amplitudes of different components of a complex signal.

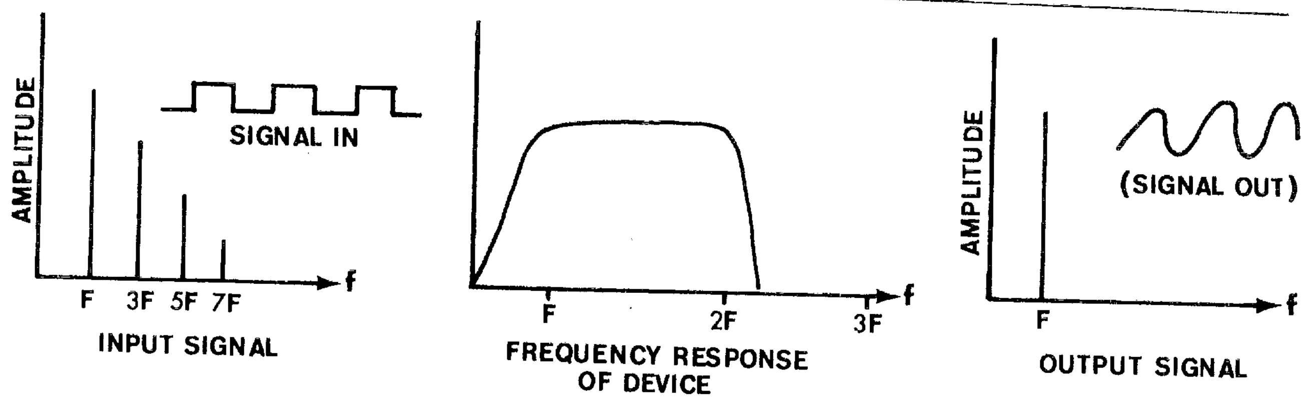


Figure 4-17. Attenuation Distortion.

Attenuation distortion is often caused by filters in the system which attenuate signal components at the edges of the desired spectrum. Figure 4-18 shows this. Ideally, all frequencies between f_1 and f_2 undergo the same attenuation; however, in actual practice, this cannot be achieved. Band edge roll-off is the term used to describe how rapidly the response of a filter falls off as a function of frequency. In low-speed digital traffic where frequency shift keying is being used, band edge distortion can cause the signal amplitude to fall below recognizable levels at the edges of the used spectrum.

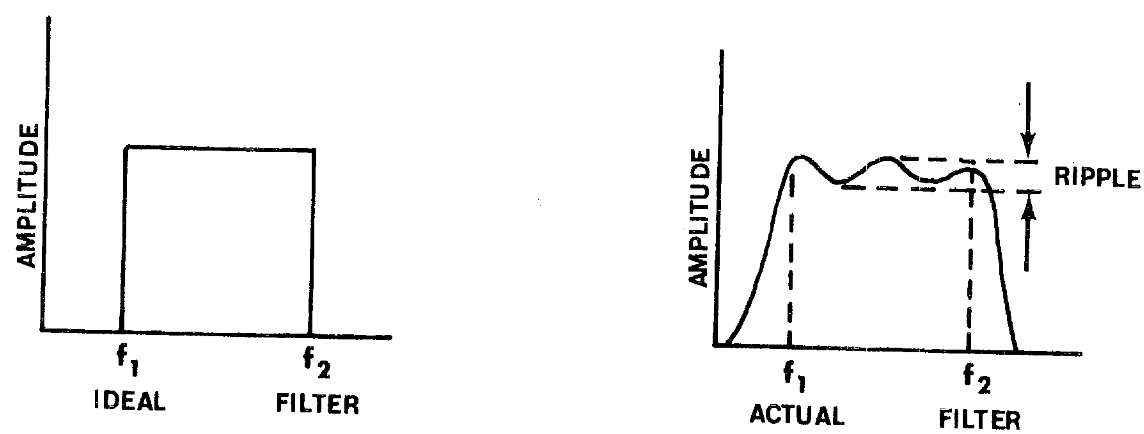


Figure 4-18. Effects of Band Edge Roll-Off.

Attenuation distortion can also result from in-band ripple. Ripple is also shown in figure 4-18. In-band ripple is often due to impedance mismatches in the line. The distortion resulting from ripple is shown in figure 4-19.

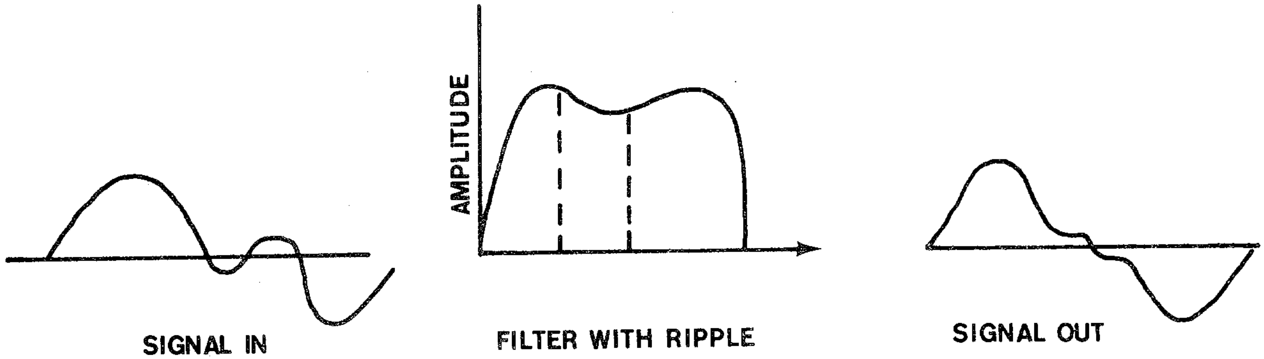


Figure 4-19. Attenuation Distortion (In-Band Ripple).

g. Echo and Reflection. Echo is named after the audible effect of a returned sound interfering with the originally produced sound. It is the return of part of a signal back up the transmission line over which it was propagated. It is especially a problem in voice communications and often results from hybrid circuits. Two-to-four wire hybrids are not perfectly balanced and some of the signal will return down the line.

Echo and reflection are related in that they cause basically the same effect. Reflection is usually referred to in general transmission line discussions as echo in voice transmissions. Echo is measured in terms of Echo Return Loss (ERL), which is a measure of the attenuation of the reflected signal. The larger the echo and the longer the delay between it and the original signal, the more noticeable it is. Obviously, echo is more of a problem on long lines.

Impedance mismatches or faults in a transmission line can bring about reflections of the signal which can interact with the oncoming signal (figure 4-20). Depending on the amplitude and relative phase of the reflected wave, the interaction with the incident wave can cause phase and amplitude distortions. The distortions are directly proportional to the reflection coefficient of the line and the modulating frequency.

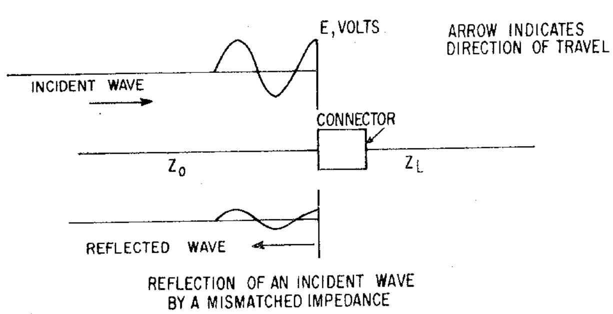


Figure 4-20. Incident Wave Reflection by Impedance Mismatch.

The reflection coefficient is calculated from the characteristic impedance of the line and its load impedance by equation below:

$$\text{Reflection Coefficient} = \frac{Z_L - Z_0}{Z_L Z_0} \qquad \text{Eqn 4-6}$$

where Z_L is the load impedance and Z_0 is the characteristic impedance

Imperfect impedance matches at switch points, transformers, and device input can affect the reflection coefficient to the point that serious phase distortion could result from the reflections. The resultant phase distortion is of the same type as described in the section on delay distortion except that here it is the result of the interaction of the original and reflected signals on the line.

h. Phase Jitter. A problem which has not received much attention before now is that of phase jitter. Its real cause is difficult to identify but its effects are easy to see. Figure 4-21 is an example of what phase jitter of a single sinusoid would look like on an oscilloscope. Jitter is shown by the "smear" of the signal over the horizontal axis and is measured in degrees.

One definition of phase jitter is short time phase instability of a signal. Amplitude jitter is also possible. A long-term phase jitter is much like a frequency shift or frequency instability. The only statement of cause would be non-linearities of the devices, lines, etc., involved, which is really the cause of any form of distortion.

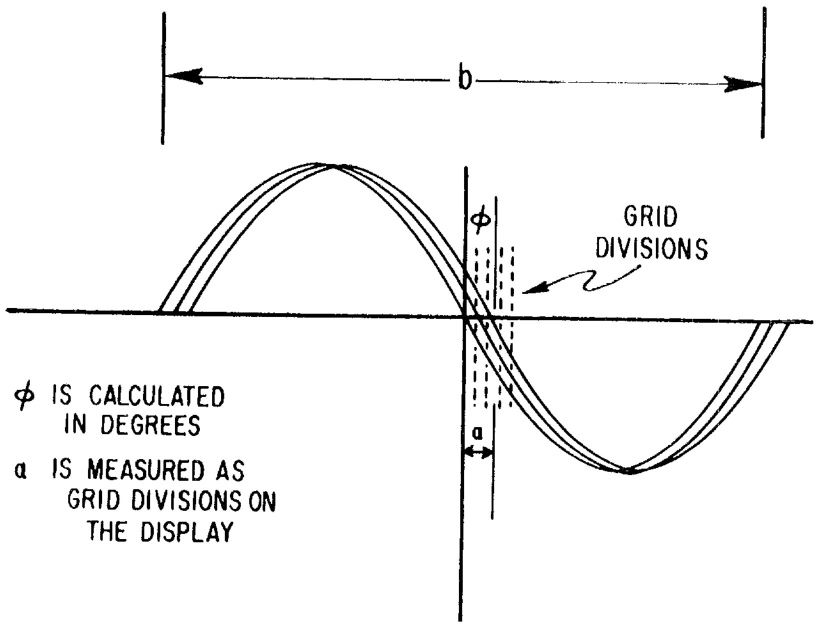
One description of phase jitter is an analogy to phase modulation techniques. Whether the cause of jitter is related to the techniques of phase modulation is not known for sure. Some common properties of jitter on wideband systems that have been observed are:

(1) It often contains multiples of power line frequency.

(2) There is a 75-85 Hz component.

(3) The predominant frequency is always below 300 Hz.

(4) The value of 0 is most often between 1° and 25°.



$$\phi = \frac{a\,(\text{GRID DIVISIONS}) \times 360\,(\text{DEGREES/CYCLE})}{b\,(\text{GRID DIVISIONS/CYCLE})}$$

Figure 4-21. Phase Jitter Display.

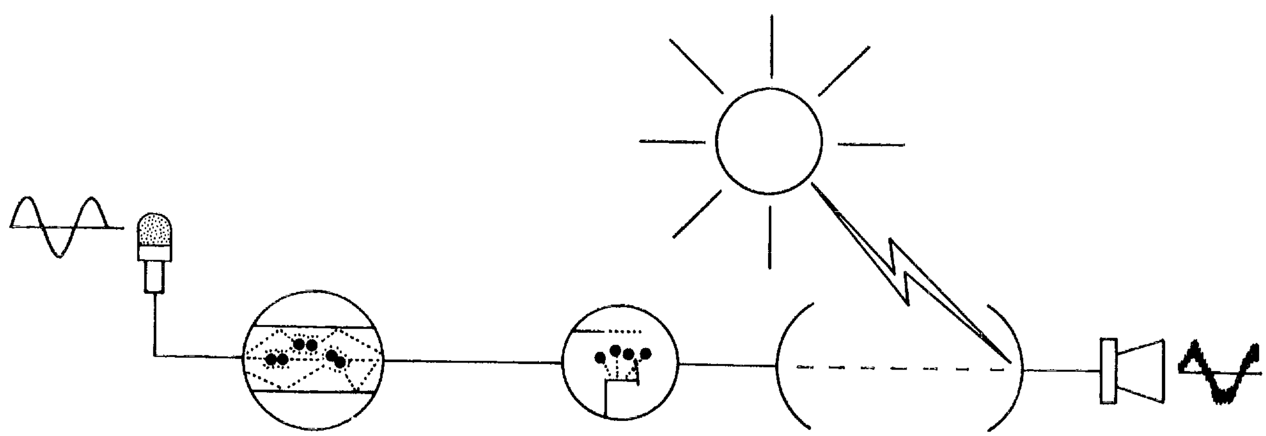


Figure 4-22. Thermal Noise Entry.

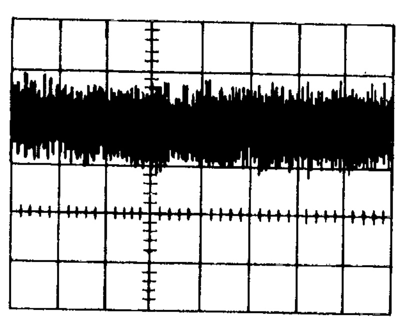


Figure 4-23. White Noise. (Courtesy GTE Lenkurt)

4-4. Noise:

a. Introduction. The major impairment of electrical communications signals is noise. As used here, noise refers to all signals which are random in nature and mask the actual communications signal. In extreme cases, the desired signal can be completely lost when disrupted by excessive noise. Several major types and sources of noise are discussed below.

b. Thermal Noise. Thermal noise is inherent in nature. All objects radiate energy over a wide range of frequencies. This radiation, in turn, appears as noise to a radio antenna. Thermal noise is also generated in electrical circuits by the random motion of free electrons caused by thermal agitation. In vacuum tubes, cathode emission of electrons is a random process. Figure 4-22 shows how thermal noise enters a communications system. These processes add to produce a composite level of thermal noise often called "white" noise (being analogous to white light which is composed of all visible light frequencies). A sample of white noise is shown in figure 4-23.

The amount of thermal noise power emitted by a source is directly proportional to the absolute temperature of the source and the bandwidth of the noise under consideration. For example, a system with a 10 MHz bandwidth admits twice as much thermal noise as a system with a 5 MHz bandwidth. Similarly, an antenna pointed at a satellite in space would see far less thermal noise than one pointed along the earth, due to the low radiating temperature of space. There are two major classes of thermal noise commonly found in communications systems: "front-end" and "idle" noise.

(1) Front End Noise. The more important portion of the thermal noise includes noise generated by the antenna plus the noise generated in the front end circuits of the receiver. This noise undergoes amplification within the receiver along with the RF carrier and, as a result of the FM process, the noise at the output of the receiver will vary inversely with the RF carrier input level. This will be covered in more detail in chapter 6.

(2) Idle Noise. The second class, usually referred to as idle noise, is generated by the transmitter circuitry and later portions of the receiver circuitry. This noise is not affected by RF signal strength; thus, it represents an absolute limit on the noise performance of the system.

c. Impulse Noise (IPN). IPN is defined as any burst of noise which exceeds the RMS noise by a given magnitude. It consists mainly of discrete, high amplitude pulses of short duration and is caused by lightning, aurora, ignition noise, power lines and associated switching equipment, and dialing impulses in telephone systems. Speech signals are virtually immune to disruption by IPN. A speech sound is sustained over a period of time and noise impulses are too brief to have an effect on the intelligence present in the sound. The relative durations of speech and impulses can be seen in figure 4-24. While having little effect on speech, IPN presents a great problem for data. There is no redundance in a data pulse as there is in speech. Impulses can easily be mistaken for data pulses; a short burst of IPN can turn a data stream into a meaningless jumble. In figure 4-25, impulses crossing the decision level are counted as data pulses. Note that white noise below the decision level does not affect data. Even if the impulses are much shorter in duration than the data pulses, the impulses can cause tuned circuits to ring and interfere with the data signal.

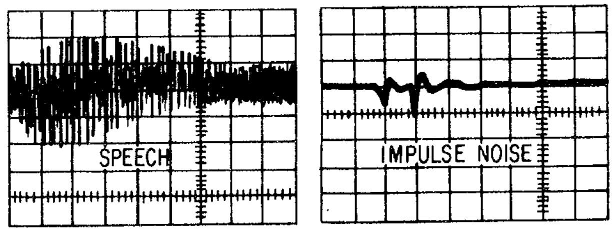


Figure 4-24. Oscilloscope Displays of Speech and Impulse Noise. (Courtesy GTE Lenkurt) Impulse noise is much shorter in duration than a speech sound; however, data is vulnerable to interference. Note ringing following impulses.

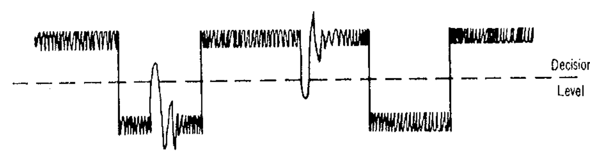


Figure 4-25. Effect of Impulses on Data.

d. Crosstalk. Crosstalk is the "leaking of a signal from its allotted channel into other channels." Not only does this tend to violate one's privacy but, if the crosstalk is intelligible, it has a very disturbing effect (figure 4-26).

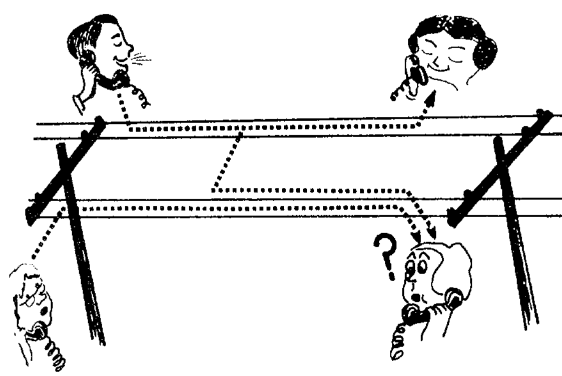


Figure 4-26. Crosstalk. (Courtesy GTE Lenkurt) Crosstalk interferes with the desired conversation and reduces the privacy of communications.

In wideband systems, crosstalk is generated in the multiplex equipment. Adjacent channels in a frequency division multiplex are separated by filters. If signal levels become excessive or if the filters are not selective enough, some signals from an adjacent channel may appear in another channel. Good equipment design and proper operating procedures can control intelligible crosstalk effectively in radio communications systems.

e. Noise Temperature and Noise Figure. There are two common methods of expressing the thermal noise characteristics of two port devices (such as amplifiers and receivers). The concept of noise temperature or equivalent input noise temperature is the most basic approach and is commonly referred to in dealing with satellite systems. Noise figure is somewhat more specific and is a common parameter used in terrestrial systems. The following paragraphs introduce the basics of both noise temperature and noise figure.

(1) Noise Temperature. If we were to measure across a resistance, we would get a voltage produced by the random motion of free electrons caused by thermal agitation. The equivalent circuit of the resistance could be represented as shown in figure 4-27, where e_n is the noise voltage produced by the random movement of electrons and R is a noise-free resistance. In this case, the available noise power which could be transferred to the load (R_L) would be:

$$P_{available} = \frac{e_n^2}{4R} \quad \text{if } R = R_L \qquad \text{Eqn 4-7}$$

The noise voltage (e_n) is random, both with time and amplitude, and is expressed as the average of the square of the noise voltage. In practically all cases, e_n can be written as:

$$e_n^2 = 4kTRB \qquad \text{Eqn 4-8}$$

where, k = Boltzman's constant
(1.38×10^{-23} joules/°Kelvin)
T = Absolute temperature (°Kelvin)
R = Resistance (ohms)
B = Measurement bandwidth (Hz)

The available noise power is then:

$$P_{available} = kTB \text{ watts} \qquad \text{Eqn 4-9}$$

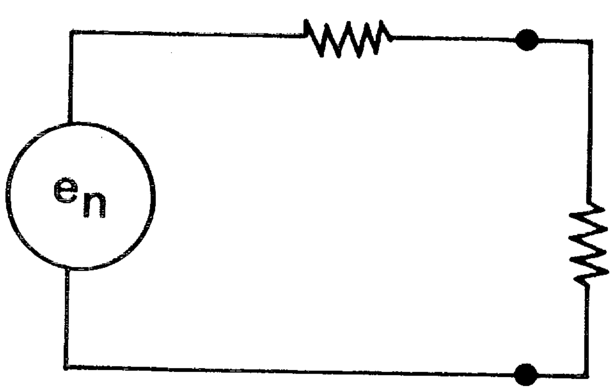


Figure 4-27 Equivalent Circuit of a Resistance.

As indicated above, if the temperature of the resistance were to change, the noise power would also change. This leads us to the procedure of referring to a noise power in terms of a noise temperature, even though the noise might have its origin in something other than a hot resistance; that is, equating noise power to a temperature which would produce the same power available from a resistance. The following example will illustrate the use of the concept of noise temperature.

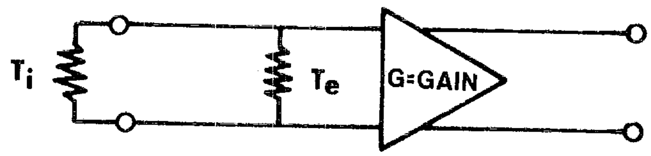


Figure 4-28. A Practical Amplifier.

Suppose that it is necessary to characterize the noise contribution of a practical amplifier. The noise output will consist of the input noise supplied times the gain of the amplifier plus an additional amount of noise due to the amplifier itself. With this in mind, a practical amplifier can be represented as shown in figure 4-28, where the amplifier is ideal (no noise contribution) and T_e equates to the additional noise power measured at the output not attributable to the noise supplied by the input termination. T_e is termed the effective input noise temperature of the amplifier and establishes the noise contribution of the amplifier. Knowing the effective input noise temperature allows us to calculate the total noise output power. If the input termination temperature is T_i, the total noise power out will be:

$$P_{N_{total}} = GkB(T_i+T_e) \qquad \text{Eqn 4-10}$$

Where, G = amplifier gain

B = measurement bandwidth

(2) Noise Figure (Noise Factor). If, in figure 4-28, we were to specify the input termination temperature to be 290°K, we could define the parameter noise figure. The noise figure of a two-port device is the ratio of the total noise output to the noise power attributable solely to the input termination at the standard temperature (290°K); thus, the noise figure is related to the effective input noise temperature as follows:

Noise Figure = Total Noise Out/Available Noise

$= (T_e kG + 290kG)/290kG$

$= (1 + T_e/290)$

Expressed in dB, the noise figure is:

Noise Figure 10 log $(1 + T_e/290)$ dB Eqn 4-11

Both noise figure and noise temperature allow us to characterize thermal noise contributions of two-port devices. The selection of which parameter to use depends on the application. Both concepts have both advantages and disadvantages in particular applications.

f. Noise Unit. The number of noise units used to measure disturbances in a 3 kHz voice channel is material enough for a separate book. The intent in this section is merely to acquaint you with a representative sample or two. It is frequently necessary to refer to a conversion table in order to cross-reference one system of measurement to another.

Why are so many different units needed to measure noise? This need has been caused by: the many types of noise present within a system; the types of handsets or reproducers that have been in common use (reproducing the noise as well as the speech); and the different ways in which noise affects modern-day communications equipment (data, facsimile, voice, etc.). Noise in a voice channel can assume many forms (for example, white noise, IPN, discrete noise, etc.). The disturbing effect of these types of noise on conversation was originally the criteria on which units of measure were based; however, in order to disturb conversation, the noise must be reproduced in the handset. This meant that the frequency response of the handset played a part in determining what electrical noise was even reproduced; therefore, the model numbers of some Western Electric handsets have accompanied the designations of some noise units (144, F1A). Nowadays, with the advent of data-type communications, these older noise units based on speech are inadequate to express how disturbing a particular noise might be.

Let's examine exactly what is meant by "weighting" a reading. Through intensive study, it has been found that sound power around 1000 Hz is very disturbing to a listener, since it interferes with his communication. A sound of equal power at 200 Hz or 5000 Hz, injected into the listener's ear, will be heard, but it will not disturb his ability to communicate nearly as much. These frequencies would have to be almost 25 dB stronger before the effect would be noticed by the listener to be the same as the 1000 Hz tone. This type of comparison has been made by using a Western Electric-type 500 telephone set at all frequencies in the voice channel and each frequency given a weighting. Figure 4-29 shows the result, which is called "C-Message" weighting. This curve might also be thought of as a filter response curve, since a filter is the device used to weight a noise reading. The curve shows that a tone at 200 Hz would have to be 25 dB higher in order to give the same power output from the filter network as compared to a 1000 Hz tone.

With such a filter placed before the meter, the composite noise in the channel can be measured and read out in dB or dBm C-Message-weighted. Other similar noise units include 144 line-weighting, F1A line-weighting, psophometric-weighting, and flat-weighting. The flat-weighting curve is essentially one which allows from 0 to 3 kHz to be weighted evenly (no weighting) with a rapid cutoff beyond 3 kHz. This weighting is most nearly representative of "all" the noise in the voice channel weighted on an equal basis.

Imagine, for a moment, that the curve in figure 4-29 extended left from 1000 Hz as a straight line which follows the 0 dB line. This would be the response curve for the flat-weighting network. Next, imagine that a 1 kHz tone is passed through either the C-Message or the flat-weighting filter. Would there be any difference? No, of course not, since both response curves are the same at these frequencies. But if the same thing were tried at 500 Hz, there would be a difference in output. Carrying this one step further, if uniformly distributed noise (white noise) were passed through both networks, there would be a difference due to the different weightings.

Suppose now that a single tone of 1 mW at 1 kHz is passed through the C Message filter and read - it would read 0 dBm, assuming there was no insertion loss. Now, if the filter were bypassed and the reading taken again, it would still read 0 dBm. If this were repeated using white noise, it would be found that, with the filter bypassed, the reading would be 1 mW, or 0 dBm. When the filter is included, though, the reading would be approximately -1.5 dBm. This simply shows that an unweighted white noise reading can be converted to C-Message weighting by subtracting 1.5 dB (this is sometimes rounded off to 2 dB). It is important to remember that this conversion is applicable only to band limited (300-3400 Hz) white noise measurements as pointed out in the following paragraph.

Particular care should be exercised when measuring data channels traversing our wideband systems. Noise measurements should not be "weighted" as this might suppress interfering noise before it reaches the meter. As an example, the lowest frequency in some Voice-Frequency Carrier Telegraph (VFCT) systems is 282.5 Hz. By examining the C-Message weighting curve, it becomes apparent that any noise or interfering tone near this frequency is attenuated by approximately 15 dB, relative to 1000 Hz. If such an interfering tone were present, it might never be "seen" using the C-Message weighting. On such circuits, 3 kHz flat measurements should be taken.

Psophometric weighting is used primarily in Europe. It employs "psophometric voltage" units established by the International Telegraph and Telephone Consultative Committee (CCITT) which are linear (picowatts) rather than logarithmic (dB). The weighting curve closely resembles that of the C-message and F1A units. One difference is the fact that psophometric weighting is referenced to 800 Hz instead of 1000 Hz. The psophometer (a common piece of test equipment on many of our European sites) in an AC voltmeter with a psophometric weighting network. Under the same conditions as previously mentioned for white noise through a C-Message filter, the loss through a psophometric filter is approximately 2.5 dB. This is often rounded off to 2 dB.

Once the weighting has been determined, the only other consideration is the unit for measurement (watts, dBm, etc.). One system involves measurement in picowatts and another involves dB-related units. It was found that a 1 kHz tone had a negligible disturbing effect on the human ear at a level of -90 dBm; therefore, -90 dBm is sometimes used as a reference level, allowing noise to be expressed in dBrnc (dB, reference, noise, c-msg (weighted)). Because the reference level is so low, all measured values are positive numbers. Figure 4-30 shows how such a noise reading might be taken.

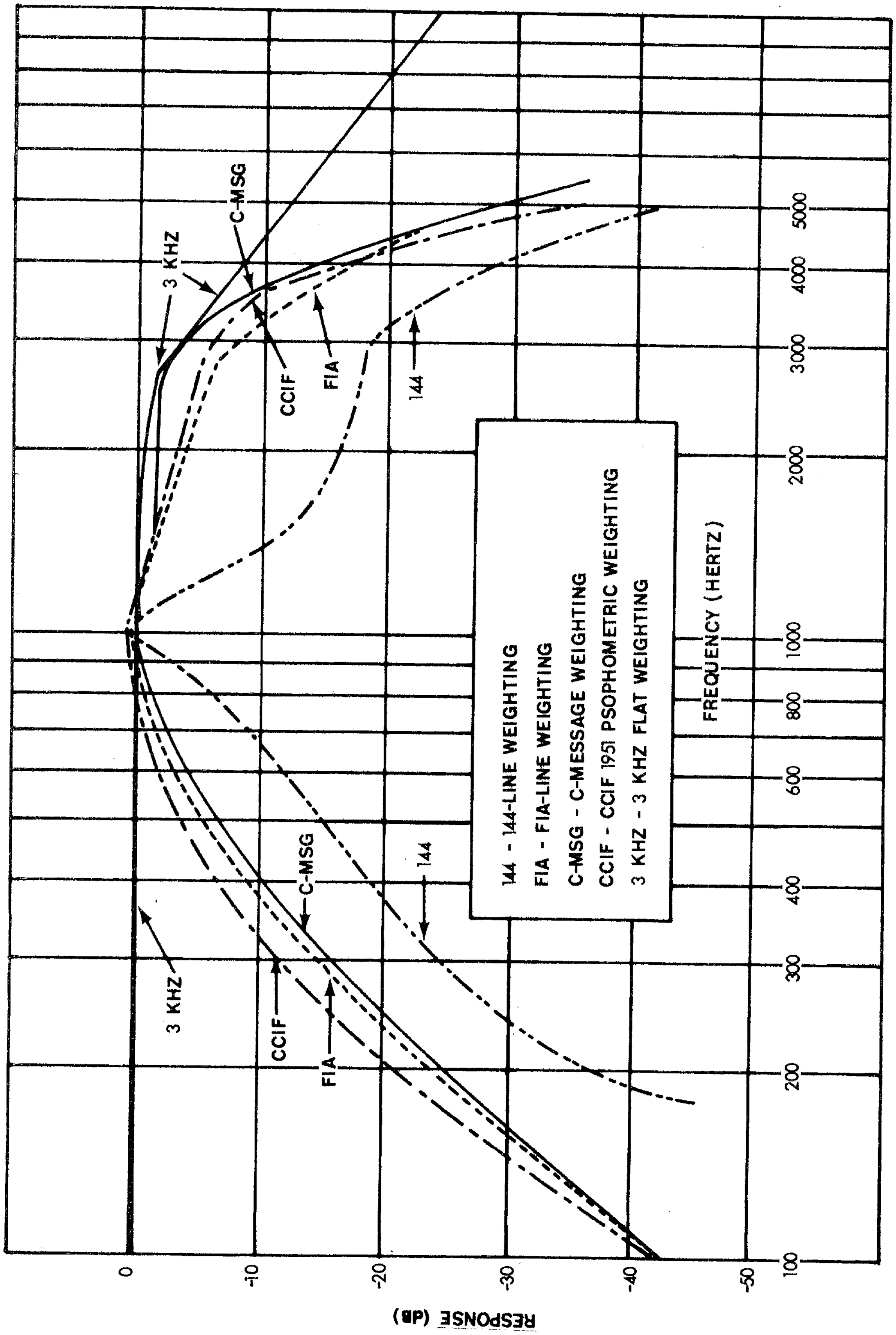


Figure 4-29. Noise Measurement Weighting Characteristics.

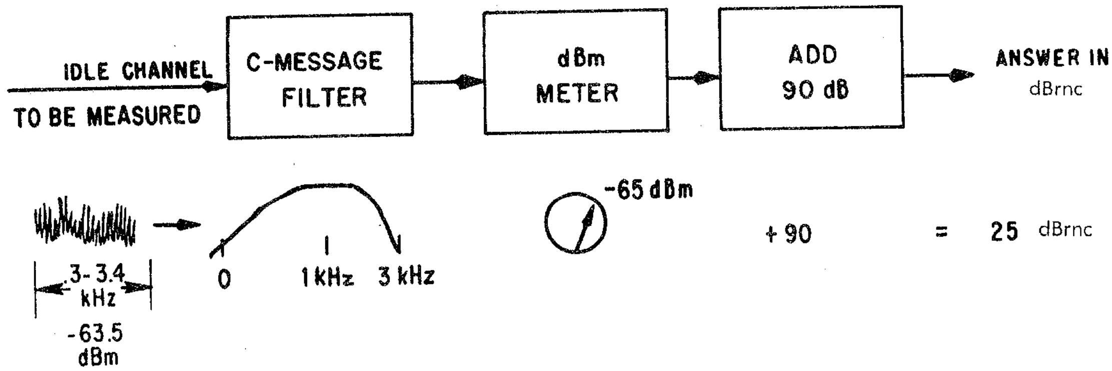


Figure 4-30. Idle Channel Noise Measured with a Weighting Filter, Computed in dBrnc.

To this point, it has been assumed that noise has a flat frequency (white) power distribution. Noise measured on a voice channel seldom has power equally distributed across the voice channel. In addition to white thermal noise, the voice channel generally has additional noise due to crosstalk, multiplex pilot leakage, and power line harmonics. Because of this, noise measurements using 3 kHz flat and C-Message filters often differ by more than a couple of dB.

One of the most important performance parameters is the relationship between the noise and signal levels. Normally, this relationship is expressed as a ratio of the signal-to-noise (S/N) measured over a limited bandwidth. For voice channels, this bandwidth is typically 3 kHz. S/N ratios measured on a voice channel normally assume a signal level of 0 dBm0, hence, the S/N is the reciprocal of the noise measured in dBm0 or:

S/N (dB) = -X (dBm0) Eqn 4-12

where X is the noise referenced to the equipment TLP.

For practical purposes, a power reading made at the mux channel breakout with an RMS voltmeter is flat-weighted. Assuming white noise, if an idle channel reading of -60 dBm is made, this becomes -60 dBm -1.5 = -61.5 dBmc. If the -60 dBm reading was made at a -10 dB TLP, then the noise in dBm0 is -50 dBm0, resulting in a reading of -50 dBm0 -1.5 = -51.5 dBmc0.

Table 4-2 may be helpful in correlating the various types of noise units with one another. The conversions are good only for evenly distributed noise. Example: At a -10 TLP, idle channel noise is measured with an RMS voltmeter. This reading would be a flat-weighted measurement, assuming negligible carrier leak. The idle channel noise reads -68 dBm. This is -58 dBm0 or a S/N = 58 dBm0. From the table, -58 dBm0 = -1.5 - 58 = -59.5 dBmc0. Example: Convert a noise reading of -60 dBm at a + 7 TLP into S/N. To find S/N in dB, the level of the noise in dBm is subtracted from the level of a 0 dBm0 test tone; therefore, S/N = + 7 dBm - (-60 dBm) = 67 dB. Note that the S/N is numerically equal to the noise expressed in dBm0, but of opposite sign.

	TO			
FROM	*dBm0*	*dBmc0*	*dBmp0*	*dBa0*
dBm0 (3 kHz Flat)	—	dBm0 -1.5	dBm0 -2	dBm0 +82
dBmc0 (C-Message)	dBm0 + 1.5	—	dBmc0 -0.5	dBmc0 + 83.5
dBmp0 (Psophometric)	dBmp0 +2	dBmp0 +0.5	—	dBmp0 +84
dBa0 (F1A Weighting)	dBa0 -82	dBa0 -83.5	dBa0 -84	—

Table 4-2. Filter Weighting Conversion Factors for Noise Measurements.
The above equations are accurate only for white noise signals uniformly distributed in the 300-3400 Hz voice band.
Example: From dBmc to dBm— *** dBm = dBmc +1.5

Chapter 5

MODULATION THEORY

5-1. General. When the feasibility of wireless communications was shown, it soon became obvious that the wavelengths of voice frequencies were far too long for practical transmission. Information had to be carried by some higher frequency wave. Time and experimentation found the answer in modulation, the process of conveying low frequency information by varying a basic characteristic of a high frequency carrier wave. Modulation may be defined as that process whereby a signal is transformed from its original form into a signal that is more suitable for transmission over the medium between the transmitter and receiver. A modulated signal can be represented mathematically by the following expression:

$$e_m(t) = A(t)\cos(\omega_c t + \phi(t)) \qquad \text{Eqn 5-1}$$

This expression shows that there are two basic parameters which can be varied: the amplitude of the signal ($A(t)$) or the phase angle of the signal ($\omega_c t + \phi(t)$). Either or both of these parameters may be varied to convey information. Amplitude modulation (AM) is accomplished by varying the amplitude of the signal and holding the phase shift ($\phi(t)$) constant. The terms "frequency modulation (FM)" and "phase modulation (PM)" are special cases of angle modulation and will be discussed in subsequent paragraphs.

5-2. Amplitude Modulation (AM). For an amplitude-modulated signal, Eqn 5-1 may be written as:

$$e_m(t) = A(t)\cos\omega_c t \qquad \text{Eqn 5-2}$$

As shown in this equation, the AM process is one in which the carrier ($\cos w_c t$) is multiplied by the information or modulating source ($A(t)$). If $A(t)$ is a single frequency sinusoid, the resulting amplitude-modulated signal may be written in the following form:

$$e_m(t) = \cos\omega_m t\cos\omega_c t \qquad \text{Eqn 5-3}$$

Using a trigonometric identity to expand this equation will allow us to examine the properties of an AM signal in more detail. In expanded form, Eqn 5-3 becomes:

$$e_m(t) = 1/2\cos(\omega_c - \omega_m)t + 1/2\cos(\omega_c + \omega_m)t \qquad \text{Eqn 5-4}$$

This expansion shows that the effect of multiplying two time-varying signals is to translate the modulating signal frequency symmetrically around the carrier. This process holds true for both simple modulating signals, as shown above, and complex signals, as shown in figure 5-1.

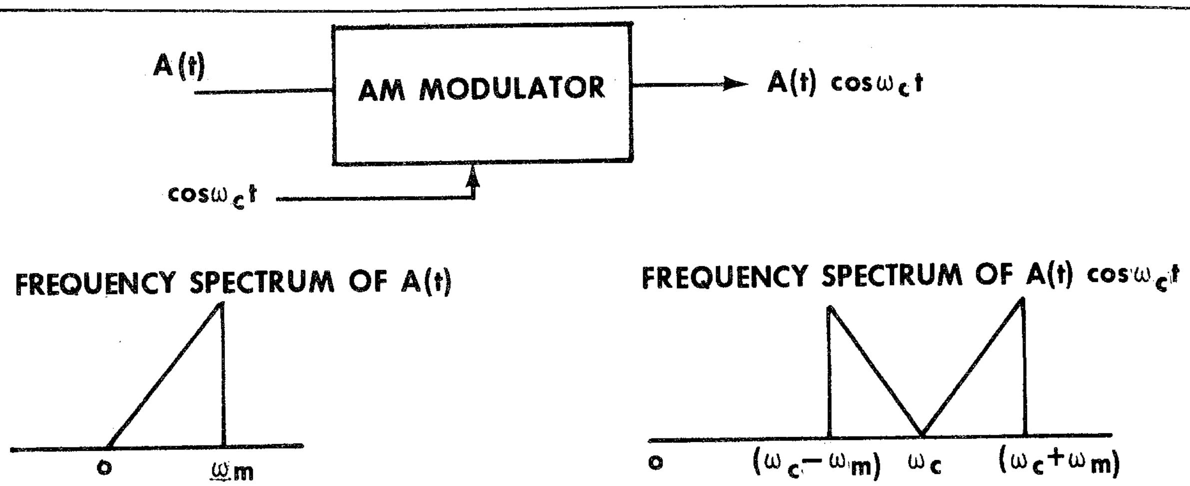


Figure 5-1. Amplitude Modulation Process.

It is important to note that Eqn 5-4 contains no component at the original carrier frequency but only sideband frequencies at $\omega_c \pm \omega_m$. For this reason, this type of AM is called double sideband suppressed carrier (DSB/SC). The time domain representation resulting from DSB/SC modulation, using a single sinusoid as the modulating signal, is shown in figure 5-2.

As shown in this figure, the amplitude of the carrier is varied proportionally with respect to the modulating signal magnitude and, when the modulating signal goes negative, the phase of the carrier is reversed (that is, changed by 180^0). This results in complicating the detection process (that is, recovery of the original modulating signal) by requiring coherent detection. Coherent detection requires multiplying the modulated signal by a carrier that is in phase with the signal used in the modulation process. In DSB/SC systems, a pilot carrier is normally transmitted with the modulated signal to facilitate detection.

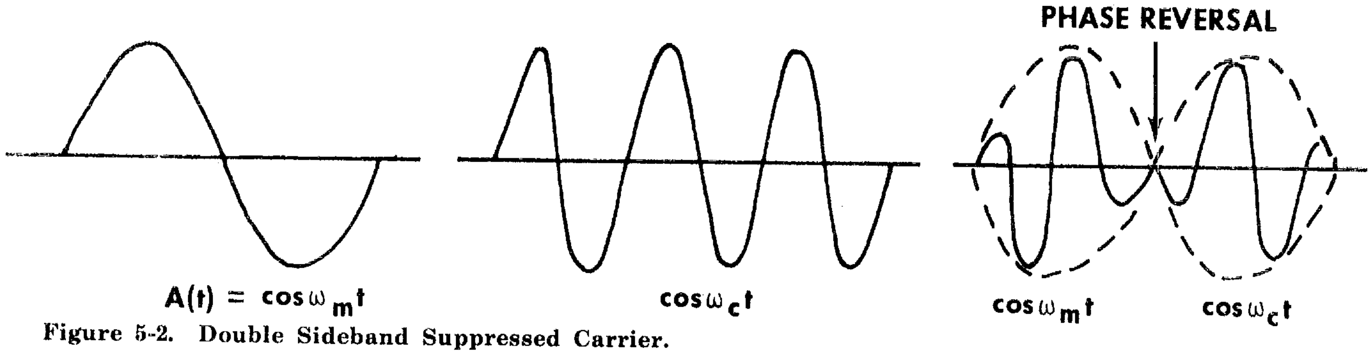


Figure 5-2. Double Sideband Suppressed Carrier.

Referring to Eqn 5-2, if we were to restrict the value of A(t) to positive values, our analysis of the modulated signal would be somewhat different. This restriction can be accomplished by biasing the modulating signal as shown in Eqn 5-5 and in expanded form in Eqn 5-6.

$$e_m(t) = (1 + m\cos\omega_m t)\cos\omega_c t, \qquad 0 < m \leq 1 \qquad \text{Eqn 5-5}$$

$$e_m(t) = \cos\omega_c t + \frac{m}{2}\cos(\omega_c - \omega_m)t + \frac{m}{2}\cos(\omega_c + \omega_m)t \qquad 0 < m \leq 1 \qquad \text{Eqn 5-6}$$

Contrary to before, the modulated signal now contains a component of the original carrier. AM modulation of this type is termed double sideband transmitted carrier (DSB/TC). In Eqns 5-5 and 5-6, m is called the modulation index and is equal to unity for 100% modulation. The effects on the modulated waveform as a result of changing the modulation index are shown in figure 5-3. Unlike DSB/SC detection which requires a carrier with appropriate phase to be reinserted at the receiver, all of the information necessary for detection of DSB/TC is contained in the amplitude-modulated wave. This allows a simple amplitude sensitive device (such as a diode detector) to be used to recover the original modulation signal.

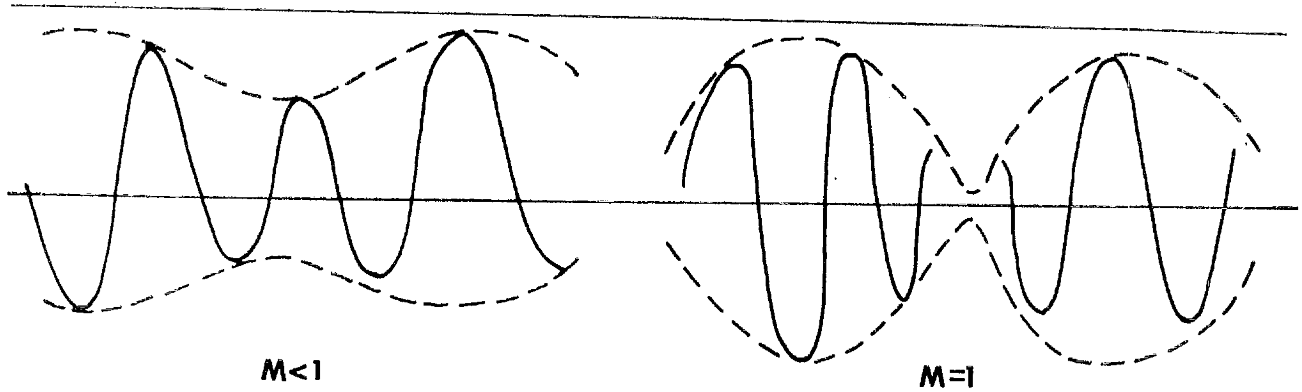


Figure 5-3. Effects of Changing Modulation Index (m).

If we examine the power distribution in a DSB/TC signal, we can see that it is divided between the carrier and the upper and lower sidebands. Referring to Eqn 5-7, we can see that the relationship of the power contained in the information-carrying sidebands to the carrier power is:

$$P_{tran} = P_c\left(1 + \frac{m^2}{2}\right) = P_c + P_{sb} \qquad \text{Eqn 5-7}$$

$$\frac{P_{sb}}{P_c} = \frac{m^2}{2}$$

where, P_{tran} = Total transmitted power
P_c = Carrier power
P_{sb} = Sideband power

The maximum power obtainable in the sidebands occurs with 100% modulation or when m = 1; however, even with this mod index, only 1/3 of the total power is in the sidebands, while the remaining 2/3 is wasted in the carrier. It is also worth noting that the carrier power remains constant and that the power in the sidebands comes from the modulating source, thus making transmitter design more costly. We will see in later paragraphs that this is not true of other forms of modulation (that is, FM and PM).

An important consideration in any transmission system is the transmission bandwidth requirements. We have seen that, in AM, two sidebands are generated, each containing the same information. In order to reduce frequency spectrum usage, all or a portion of one of the sidebands can be removed by filtering the signal prior to transmission. Single sideband is the term given to systems in which only one sideband is transmitted while vestigial sideband is the term used to describe systems which suppress a portion of one of the sidebands.

5-3. Frequency Modulation (FM):

a. General. FM is a special case of angular modulation. It is a process where the modulating wave varies the instantaneous frequency of the carrier wave. In this technique, intelligence is conveyed by variations in the frequency of the carrier. Referring back to Eqn 5-1, a frequency-modulated signal may be expressed as:

$$e_m(t) = A_c \cos\left(\omega_c t + \frac{KA_m}{\omega_m} \sin \omega_m t\right) \qquad \text{Eqn 5-8}$$

where, modulating signal $= A_m \cos \omega t$

instantaneous frequency deviation $= KA_m \cos \omega_m t$

As shown, only the amplitude of the modulating signal (A_m) controls the magnitude of the frequency change. For example, suppose a 1 MHz carrier is modulated by a 1 volt peak sinusoidal signal and the frequency of the carrier varies from 999,500 Hz to 1,000,500 Hz. If we now apply a 2 volt peak sinusoidal signal, the carrier frequency will vary 999,000 Hz to 1,001,000 Hz. The amount of frequency change from the carrier frequency is called the frequency deviation. Our first example had a peak deviation of 500 Hz, while our second had a peak deviation of 1000 Hz.

The frequency of the modulating wave controls the rate of change of the carrier frequency. Consider two modulating frequencies of 1000 Hz and 2000 Hz and of the same amplitude. The total deviation caused by these two tones will be the same, but the changes will occur at different rates. Waveforms showing the FM process are shown in figure 5-4.

b. Modulation Index. In Eqn 5-8, the factor $\frac{KA_m}{\omega_m}$ is called the modulation index. It is the ratio of the peak frequency deviation to the modulating frequency. For a sinusoidally-modulated carrier where the peak deviation and modulating frequency are expressed in Hz, the modulation index is:

$$\text{Mod Index } (\beta) = \frac{\text{Peak Deviation } (\Delta f)}{\text{Modulating Freq (fm)}} \qquad \text{Eqn 5-9}$$

The modulation index is an important factor in determining signal bandwidth and system noise performance (figure 5-6).

c. Frequency Spectrum of an FM Signal. If we analyze a frequency-modulated signal in the frequency domain, we will see marked differences from the AM case. Whereas the AM signal had only two sidebands, the FM signal contains a large number of sidebands, even with a single modulating frequency. Differences in the spectra of the two systems are shown in figure 5-5. While there are a great many sidebands in the FM signal, only a limited number of sidebands contain sufficient energy to be considered significant. We will consider any sideband which contains more than 1% of the power in the unmodulated carrier as significant. The number of significant sidebands and, hence, the signal bandwidth, depends on the modulation index. The index number itself does not directly tell us anything. The calculation of individual sideband amplitudes requires the use of complex Bessel functions; however, estimates of the signal bandwidth can be made from the modulation index. In general:

$$\text{Bandwidth (BW)} = 2fm\,(\beta + 1) \qquad \text{Eqn 5-10}$$

where: fm = modulating frequency
β = modulation index

Again, this is only an estimate. Others often use $Bw = 2fm(\beta + 2)$ for an estimate.

This includes all sidebands having more than 1% of the total power of the unmodulated carrier. The effect of modulation index on bandwidth can be seen in figure 5-6. The modulating frequency was held constant in the figure and the deviation was varied to change the modulation index.

d. Power Distribution. In an AM system, we saw that the power in the carrier was constant and power for the sidebands had to come from an external source. This is not the case in FM. Power is taken from the carrier and put into the sidebands as modulation is applied. The total power in an FM signal is constant, regardless of the degree of modulation present.

The amplitude of the carrier depends on the modulation index and is also determined by a Bessel function. There are values of index where the carrier amplitude is zero. These points can be used to set modulation index to a particular value, as described in chapter 19.

e. Complex Modulating Signals. Thus far, we have considered only a single modulating tone. In general, the modulating signal will contain many frequencies. For the case of an FM modulating signal, the modulation index can be defined as:

$$\beta = \frac{\text{Composite Peak Freq Deviation}}{\text{Highest Multiplexed Channel Freq}} \qquad \text{Eqn 5-11}$$

Equation 5-10 is still valid for bandwidth.

5-4. Phase Modulation (PM). Another special case of angle modulation is PM. This type is defined as angle modulation in which the instantaneous phase deviation is proportional to the modulating signal voltage. Referring to Eqn 5-1, with A(t) held constant, PM can be represented as:

$$e_m(t) = A_c \cos(\omega_c t + KA_m \cos \omega_m t) \qquad \text{Eqn 5-12}$$

where, $A_m \cos \omega_m t$ = modulating signal

and $KA_m \cos \omega_m t$ = instantaneous phase deviation

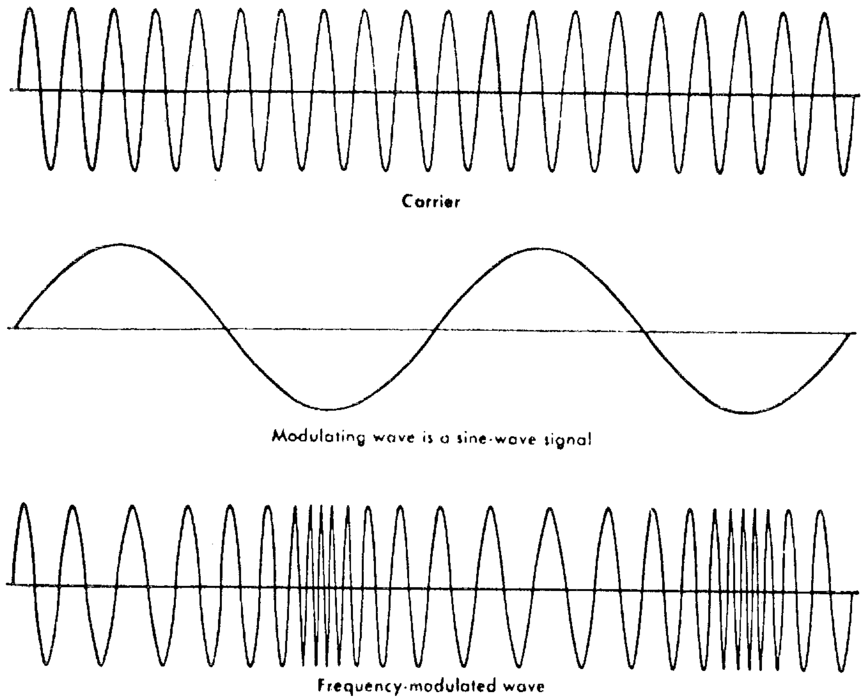


Figure 5-4. FM Waveforms.

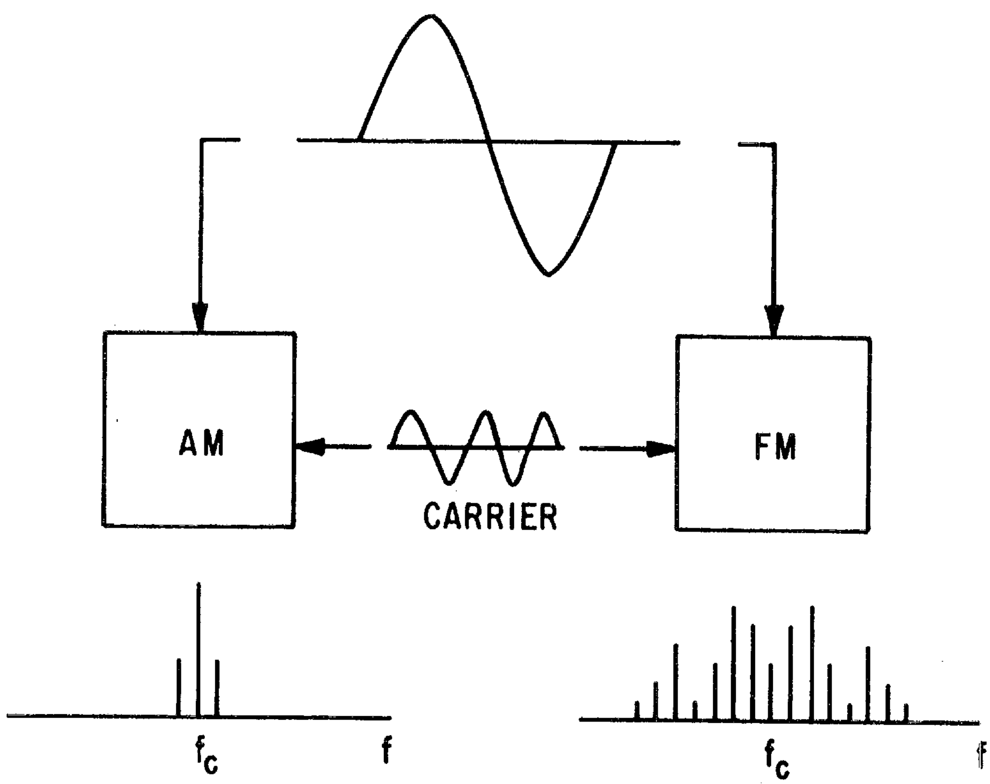


Figure 5-5. Comparison of AM and FM Frequency Spectra.

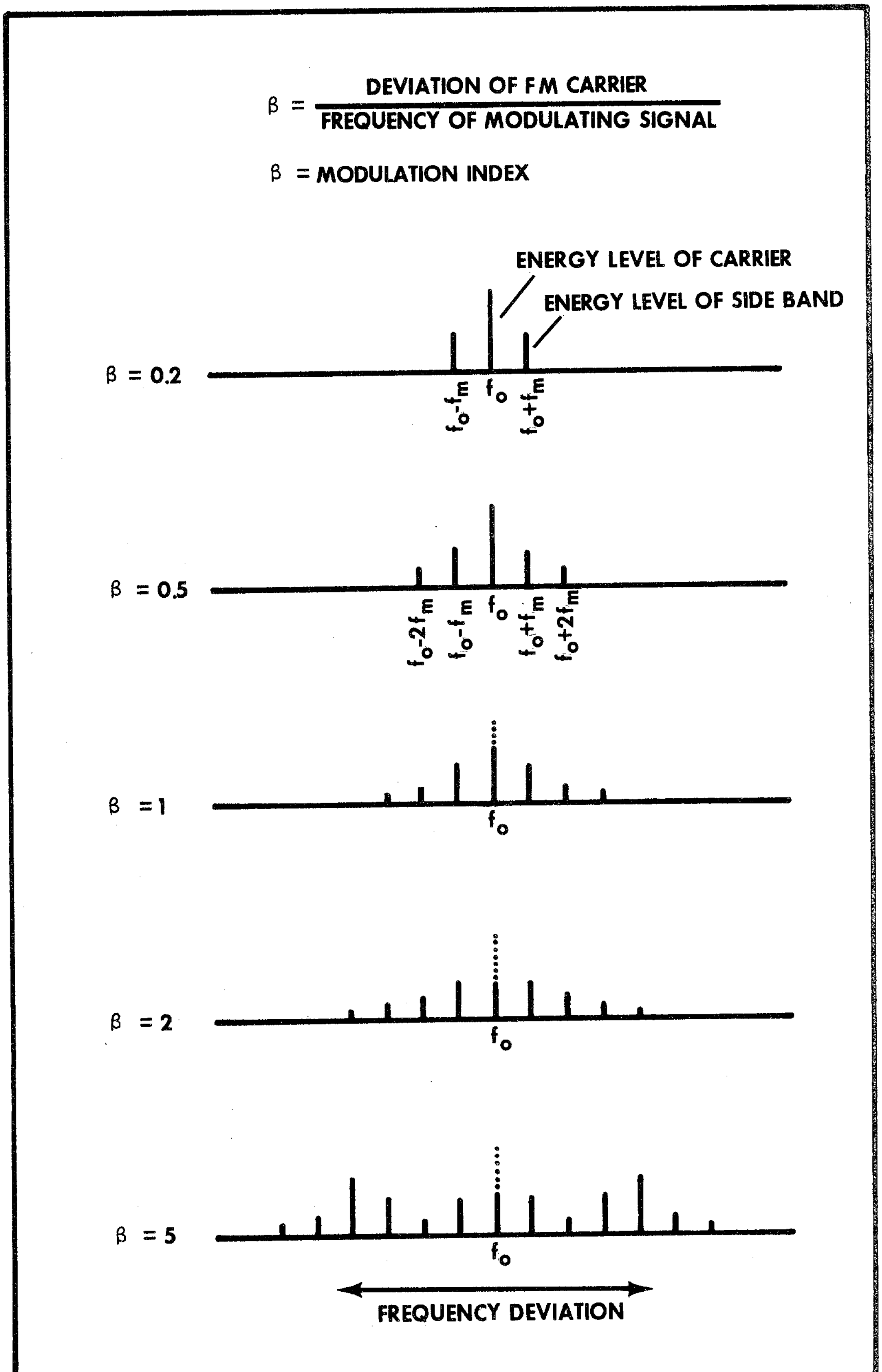


Figure 5-6. Interaction of Modulation Index and Bandwidth.

As shown in this equation, the peak phase deviation is independent of the modulating frequency; however, for a phase-modulated signal, the instantaneous frequency deviation is a function of both the amplitude and the frequency of the modulating signal. The instantaneous frequency deviation is the first derivative of the phase deviation or, from Eqn 5-12:

Instantaneous Frequency Deviation =

$$-KA_m \omega_m \sin \omega_m t$$

If we put the modulating signal through a filter which attenuates the signal linearly with increasing frequency (that is, an integrator), the output of a phase modulator will be an FM wave. This technique is sometimes used to generate an FM signal in wideband systems. FM generation, using a phase modulator, is shown in figure 5-7.

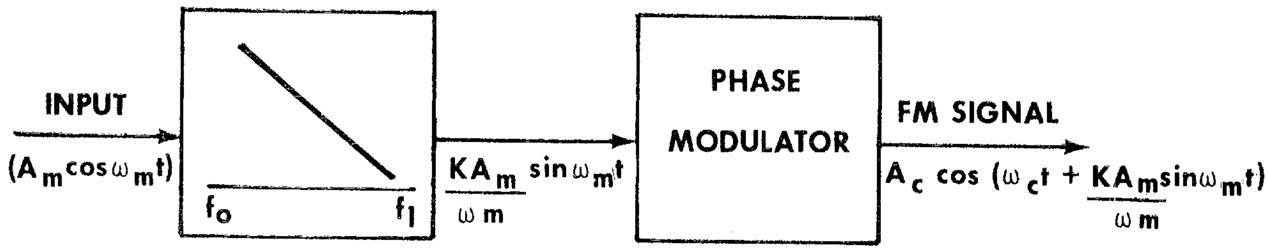


Figure 5-7. Indirect FM, Using a Phase Modulator.

PM and FM are closely related since any variation of the phase of a carrier results in a variation in the frequency and vice versa. In fact, the waveform alone cannot be used to distinguish between the two - a knowledge of the modulation function is necessary to make the correct determination. An important point to note about PM is that the modulation index (KA_m) does not change with frequency as it does with FM. As a result, S/N is constant regardless of frequency.

5-5. Demodulation. Every modulation process must have a corresponding demodulation process if communication is to be established. Demodulation or detection is simply recovering the original signal from the received, modulated signal.

a. AM Detection. In an AM system, intelligence is carried by the "envelope" or RF amplitude. Demodulation is readily accomplished with an envelope detector consisting of a rectifier and a low-pass filter (figure 5-8). The diode rectifies the signal and the filter passes the low frequency signal and rejects the carrier.

Another type of AM demodulator is the synchronous, or product, detector. Here, a carrier is mixed with the AM wave, shifting the sidebands back to voice frequencies. A filter is used to remove the intelligence.

b. Single Sideband Detection. A single sideband signal is more difficult to demodulate. Since there is no carrier, we must provide a carrier of the proper frequency at the receive end. The carrier is mixed with the sidebands in a synchronous detector, which shifts the frequencies in the sidebands back to their original place in the spectrum. If there is any frequency error in the reinserted carrier, the demodulated frequencies will be in error and distortion results.

If the reinserted carrier does not have the same phase as the carrier used in transmitting, another type of distortion (phase distortion) will occur. All frequency components will be shifted in phase by an identical amount. This distortion is not serious in voice communications, but it does give a "Donald Duck" voice effect. In video or wideband data systems, this phase distortion may be intolerable; therefore, modern multiplex systems synchronize their carriers in frequency and phase.

c. FM Detection. Demodulation in wideband FM systems is accomplished with a device called a discriminator. This unit converts the frequency variations of the signal to a wave that represents the modulating intelligence. It is sensitive to the amount of frequency change in the received signal. The output voltage is proportional to the magnitude and direction of the frequency shift. Figure 5-9 shows a typical response curve for a discriminator.

In this example, the output voltage is zero when the input is at center frequency, negative when the input is above center, and positive below center frequency. As long as the response curve of the discriminator matches the modulator curve (ideal curves linear), there will be no distortion in the output. A more complete discussion of discriminators is in chapter 10.

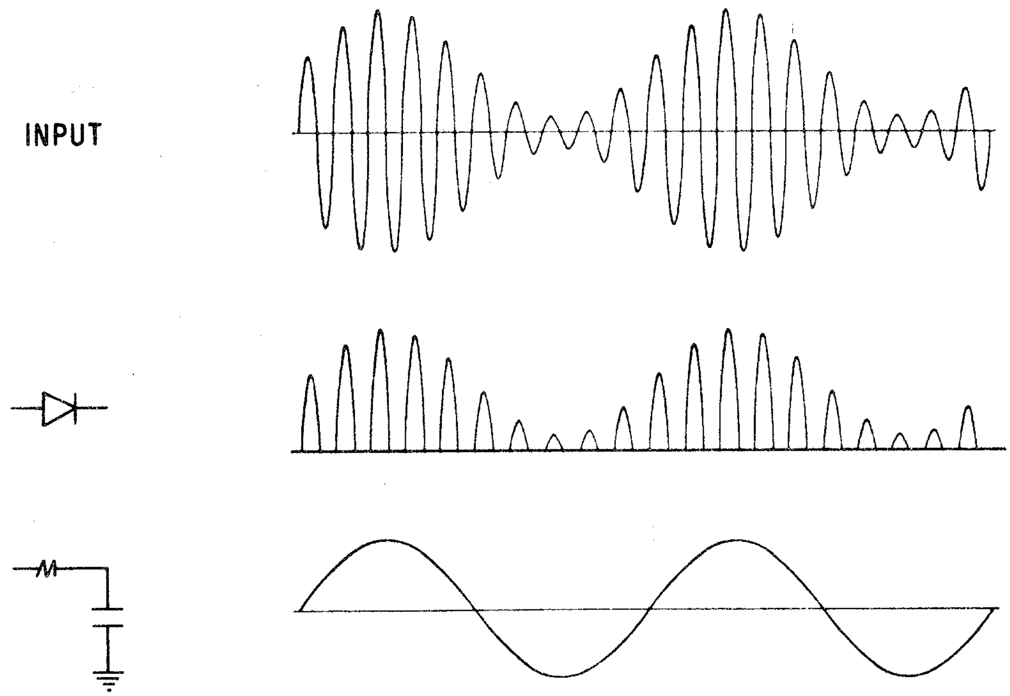


Figure 5-8. AM Detection.

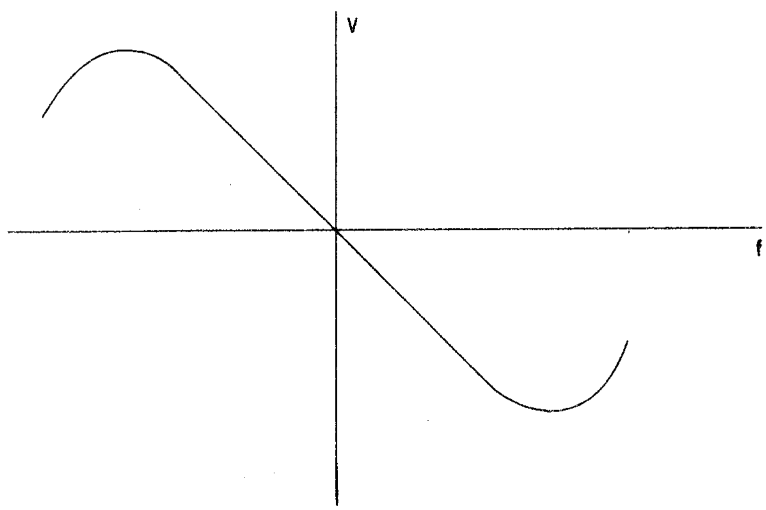


Figure 5-9. Typical FM Discriminator Curve.

Chapter 6
NOISE PERFORMANCE OF FM SYSTEMS

6-1. Introduction. As a radio terminal transmits a baseband signal from one location to another, the signal is easily influenced by non-ideal characteristics of the overall transmission medium. The effect of these imperfections is to introduce noise into the reconstituted (demodulated) baseband at the last microwave (M/W) terminal.

The noise at the baseband of an analog microwave receiver is a mixture of noise from such sources as radio terminal thermal and intermodulation noise, crosstalk, and frequency division multiplexer noise. This noise can generally be divided into two broad categories. The first category is noise which is independent of the signal applied to the terminal baseband. This noise includes the effect of thermal noise generated within the receiver, the phase noise of the transmitter, cable crosstalk from other baseband signals, radio frequency interference, and multiplexer tone leakage, power supply ripple, thermal noise, and miscellaneous sources. Since this noise has no relationship to the baseband signal, it can be measured with the baseband signal removed from the transmission equipment. It is called "idle noise." The other main noise category includes the noise which is directly related to the baseband signal frequency components and power level. Although this noise includes such effects as baseband cable and radio terminal crosstalk, this type of noise is loosely termed "intermodulation noise." This noise can only be measured with signal applied to the transmission equipment baseband.

6-2. Idle Noise. The primary method of characterizing the thermal noise performance of an M/W radio is through the use of what is called an FM slot noise quieting curve or, simply, quieting curve. The quieting curve is a plot of noise in the baseband of a "M/W FM receiver as various levels of unmodulated RF signals are applied to the receiver. The baseband noise is measured with a FSV with a measurement bandwidth of 3.1 kHz, the nominal frequency width of a single telephone circuit. The FSV is tuned to different frequencies in the baseband of the receiver while the received signal level (RSL) applied to the receiver is varied. By plotting these noise measurements as a function of baseband frequency and RSL, it is possible, given an actual operational RSL of the receiver, to predict the thermal noise due to the M/W terminal which will be added to a particular multiplexed telephone circuit. Figure 6-1 shows the noise performance of a typical FM receiver as the RSL is varied. This quieting curve is divided into four regions for analysis.

Region A is the region of the quieting curve where the RF signal has essentially been lost and the slot noise is dependent on the thermal noise generated within the front end of the receiver, the receiver intermediate frequency (IF) response, and the amount of receiver limiting. Region B is the non-linear region of the quieting curve. This region is a complex function of the signal introduced into the receiver, the thermal noise generated, the receiver IF response, and degree of signal limiting. Region C is the linear region of the receiver. Here, the noise decreases in direct proportion to the increase in signal introduced into the receiver. Region D is the so-called saturated region of the receiver. Here, the slot noise at the baseband of the radio is independent of the RSL and receiver-generated noise. Performance in Region D is limited by such things as the spectral purity (phase noise) of the signal generator used to perform receiver quieting noise, receiver local oscillator phase noise, and the baseband circuitry intrinsic noise. A well-engineered line-of-sight (LOS) system will usually operate most of the time in Region D, while a tropo system may operate in all four regions, spending most of the time in Region C.

Due to the computational difficulties involved in obtaining exact results for low carrier-to-noise (C/N) ratios (Region A), equations for the baseband noise will not be presented for C/N ratios less than +5 dB. The baseband noise in Region B, where the C/N ratio is greater than +5 dB, can be calculated by Eqn 6-1:

$$N(\mathrm{dBm0}) = -139.1 - 20\log \Delta f_{/ch\ rms}(\mathrm{kHz}) - RSL(\mathrm{dBm}) + NF(\mathrm{dB}) - P(\mathrm{dB}) + 10\log\left(f^2(\mathrm{kHz}) + (0.38797)\left[\frac{\sqrt{P}\,B^2(\mathrm{kHz})}{e^P}\right]\right) \qquad \text{Eqn 6-1}$$

and

$$C/N(\mathrm{dB}) = RSL(\mathrm{dBm}) + 114.0 - NF(\mathrm{dB}) - 10\log B_{IF}(\mathrm{MHz}) \qquad \text{Eqn 6-2}$$

where

p = antilog $((C/N)/_{10})$
e = base of natural (Naperian) logarithms ≅ 2.7183
RSL = received signal level
NF = receiver overall noise figure in dB measured at the same point at which RSL is to be measured
f = baseband frequency of the slot used to measure receiver baseband noise
$\Delta f_{/ch\ rms}$ = (per channel) rms deviation in kHz for a 0 dBm0 sine wave test tone (at baseband pivot frequency if emphasis is used)
B = receiver 3 dB IF bandwidth
P = baseband pre-emphasis in dB relative to baseband pivot frequency (if emphasis is not used, this factor is 0 dB).

Figures 6-2 and 6-3 yield the pre-emphasis relative to the pivot frequency for the most common emphasis networks.

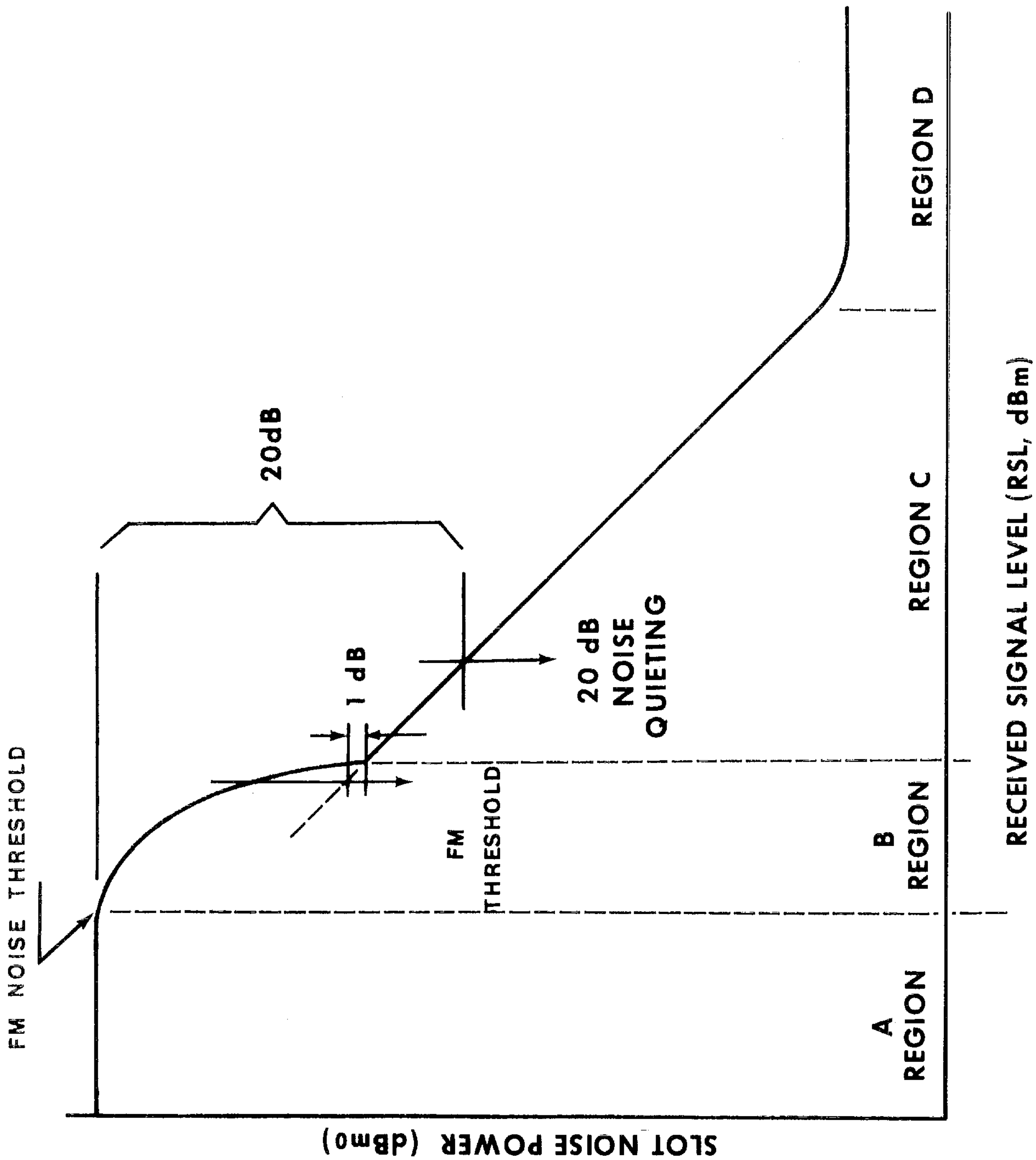


Figure 6-1. Slot Noise Versus Received Signal Level (RSL).

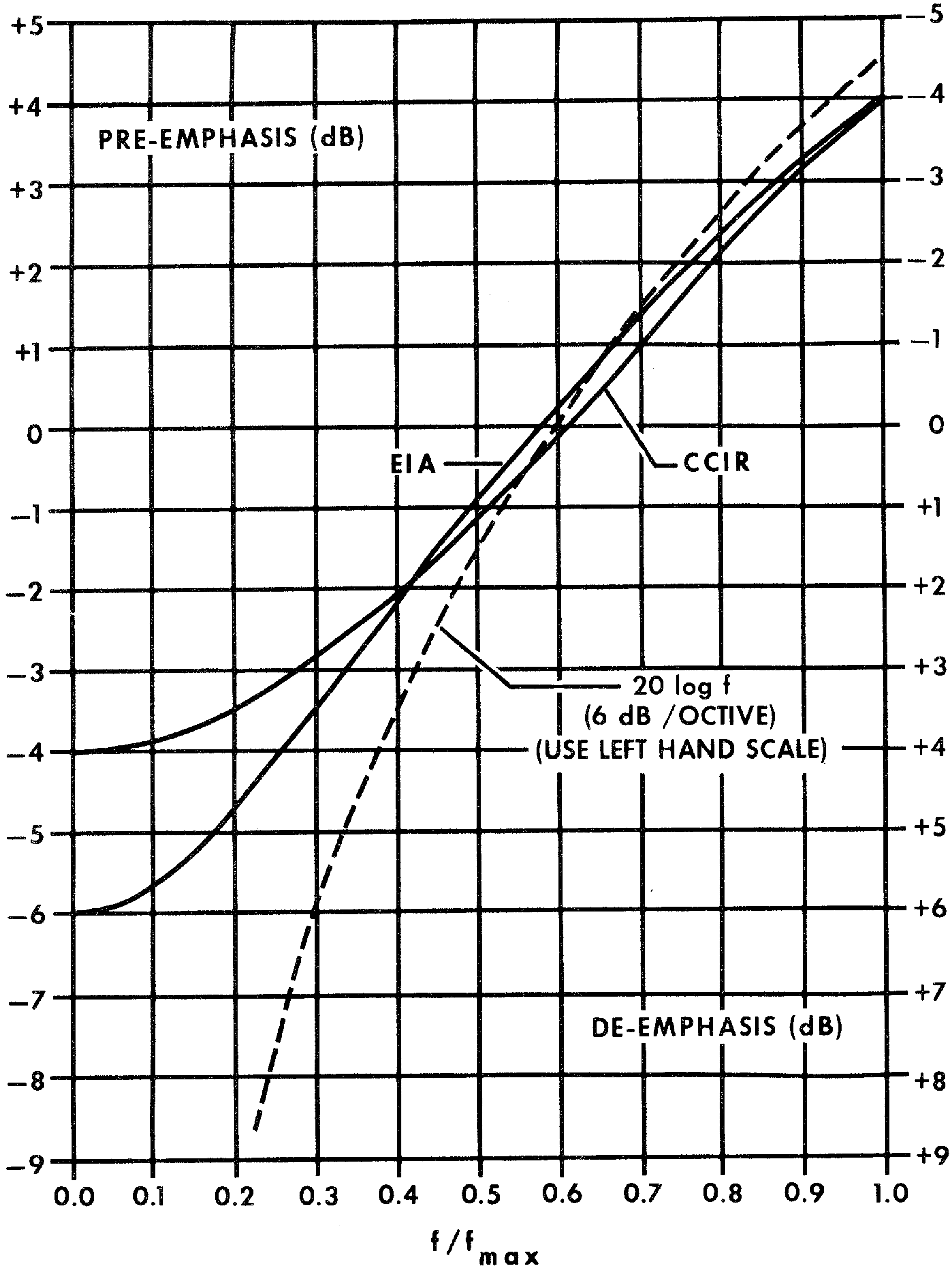


Figure 6-2. CCIR/EIA Emphasis Curves.

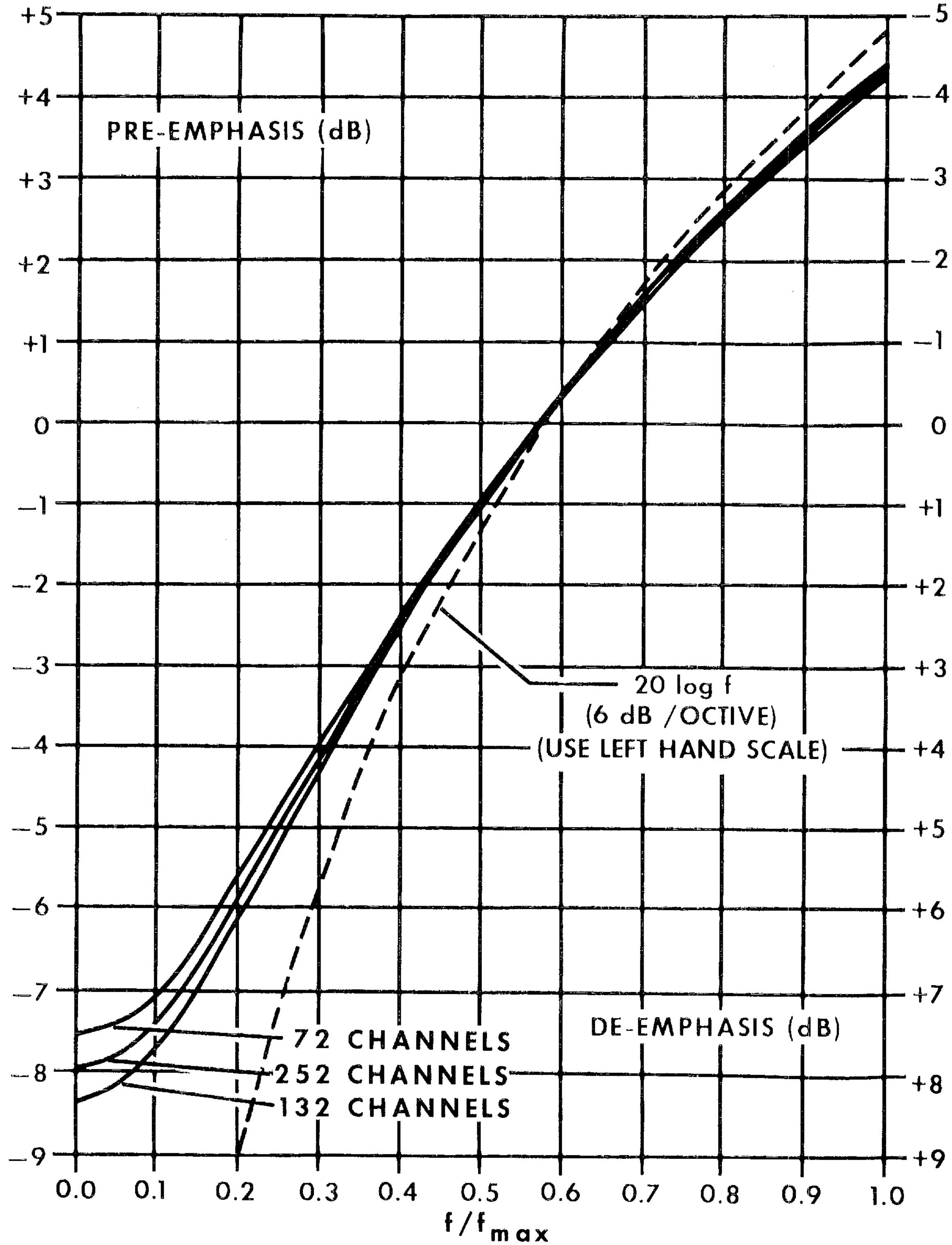


Figure 6-3. REL Emphasis Curves.

Region C can be defined by Eqn 6-3. The symbols used were stated previously.

Eqn 6-3

$$N(dBm0) = 139.1 - 20 \log \Delta f_{/ch\ rms}(kHz) - RSL(dBm) - NF(dB) + 20 \log f(kHz) - P(dB)$$

The last region of interest is D. It can be shown that, in Region D, the noise is made up of the 6 dB per octave thermal noise plus the time derivative of any phase noise associated with the received carrier. The differentiated phase noise is called frequency noise. It is distinctly different noise than the receiver thermal noise. In Region D, the thermal noise is negligible and the dominant noise source is the frequency noise of the received carrier. At a high, stable carrier-to-thermal noise ratio with no multipath and the carrier tuned to the center of a symmetric IF, the only significant frequency noise sources are the various oscillators in the transmitter and receiver.

In general, the phase noise of an oscillator varies in power as $1/f^2$ where f is the frequency of the measurement slot away from the carrier frequency. The noise at the baseband of an FM demodulator, however, is the time derivative of the phase noise. After demodulation, the frequency noise appearing at the baseband of the radio is essentially flat, independent of baseband frequency. This is the case for baseband frequencies greater than roughly 100 kHz. Below this frequency, the noise gradually increases. The frequency noise within the first few kilohertz of the baseband can be several times the noise in the mid region of the baseband. Often the output of an oscillator is multiplied to derive a higher frequency. Multiplying the output of the oscillator increases the effective deviation of the frequency noise. For example, doubling the oscillator frequency causes a 6 dB increase in frequency noise due to doubling the effective deviation of the frequency noise components. In the gigahertz region, the noise figure of the multiplier diodes or up/down converter mixer diodes can be a significant factor. The noise figure of the diodes can cause a significant increase in slot noise above roughly 1 MHz in the radio baseband. Figure 6-4 graphs typical FM receiver baseband slot noise due to a free-running, unmodulated microwave (M/W) oscillator.

There are several oscillators in a M/W system. The primary oscillators of interest are the local oscillators for up and down frequency conversion and the frequency-modulated oscillator at the transmitter (voltage-controlled oscillator (VCO)). The up and down converter local oscillators can be phase-locked to reduce low frequency noise. Certain hybrid configurations allow the noise sidebands of the oscillator to be suppressed. These methods cannot be used on the modulated oscillator; however, since any method used to suppress the inherent noise of the oscillator will also suppress the modulation, the oscillator to be modulated *must* have inherently low noise if the output of this oscillator is to be multiplied.

There are a couple of common FM receiver specifications related to the slot noise performance of the receiver in the A, B, and upper C regions. One of the specifications is 20 dB (and occasionally 30 dB) noise quieting. The quieting specification is the RSL at which the noise in a specified baseband slot is 20 dB (or 30 dB) lower than the noise in the slot with no RSL present at the receiver input. Theoretical, 20 dB and 30 dB C/N ratios are in tables 6-1 and 6-2. The C/N values can be converted to RSL single receiver parameters by the following formula:

$$RSL(dBm) = C/N(dB) - 114.0 + NF(dB) + 10 \log B(MHz) \qquad \text{Eqn 6-4}$$

The other specification is FM slot noise threshold or, simply, FM threshold. The most common definition of FM threshold is the RSL at which the receiver baseband slot noise has increased 2 dB for a 1 dB reduction in RSL. This is often determined graphically with quieting curves by extending the straight line of Region C into Region B and defining FM theshold as the RSL at which slot noise is 1 dB above the straight line extension. Generally, FM threshold is measured in a narrow (nominal 3.1 kHz wide) noise slot. Theoretical narrow slot FM threshold values are in table 6-3. For the FM threshold charts, (f/B) is the slot baseband frequency divided by the IF bandwidth. Normally, the 20 dB noise quieting point (measured on the high slot) is within a few dB of the FM threshold point (figure 6-1).

f/B	*C/N (dB)*
0.2	11.8
0.1	7.2
0.05	6.4
0.025	6.3
0.01	6.2
0.001	6.2
0.0001	6.2
0.0	6.2

Table 6-1. Theoretical 20 dB Noise Quieting (Narrow Slots).

f/B	*C/N (dB)*
0.2	21.8
0.1	15.5
0.05	9.6
0.025	8.2
0.01	8.0
0.001	8.0
0.0001	8.0
0.0	8.0

Table 6-2. Theoretical 30 dB Noise Quieting (Narrow Slots).

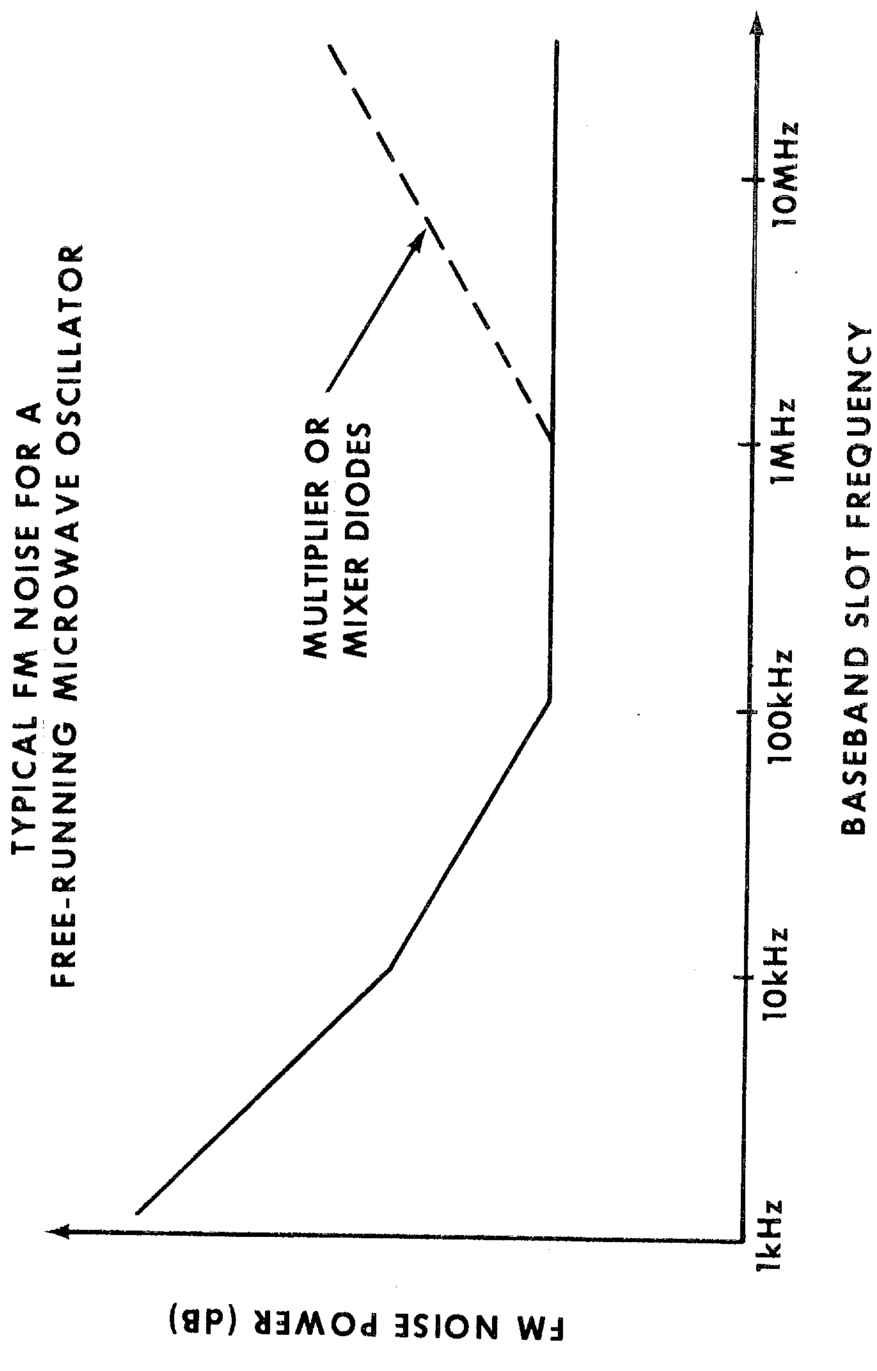


Figure 6-4. Typical Microwave Oscillator Noise.

f/B	*C/N (dB)*
0.2	6.4
0.1	7.7
0.05	8.7
0.025	9.5
0.0158	9.9
0.0147	10.0
0.0134	10.1
0.01	10.3
0.005	10.9
0.0025	11.4
0.001	11.9
0.0005	12.3
0.00025	12.7
0.0001	13.1
0.0	Undefined

Table 6-3. Theoretical FM Noise Thresholds (Narrow Slots).

Having defined the key points on the receiver characteristic curve, we will now discuss the effect of system parameters on the thermal noise performance of the receiver. The receiver noise figure controls not only the point of threshold but also the value of S/N at higher RSLs. Lowering the noise figure will lower the FM threshold of the receiver, allowing satisfactory operation at lower RSLs. The value of the S/N at threshold remains the same. Since S/N then increases dB for dB with RSL, any lowering of the threshold point also causes a corresponding increase in S/N above threshold. For example, if the threshold point is lowered from -85 dBm to -88 dBm, there will be a 3 dB increase in S/N at all points above threshold (in Region C). The effect of noise figure changes on performance is shown graphically in figure 6-5. Higher noise figures result in performance degradation, while lower noise figures improve performance.

The IF bandwidth determines the thermal noise power entering the detector and, hence, the RSL at which the threshold occurs; however, the noise in any one channel will not be increased or decreased above threshold (figure 6-6). Threshold extension panels use this principle of reducing bandwidth to lower the threshold.

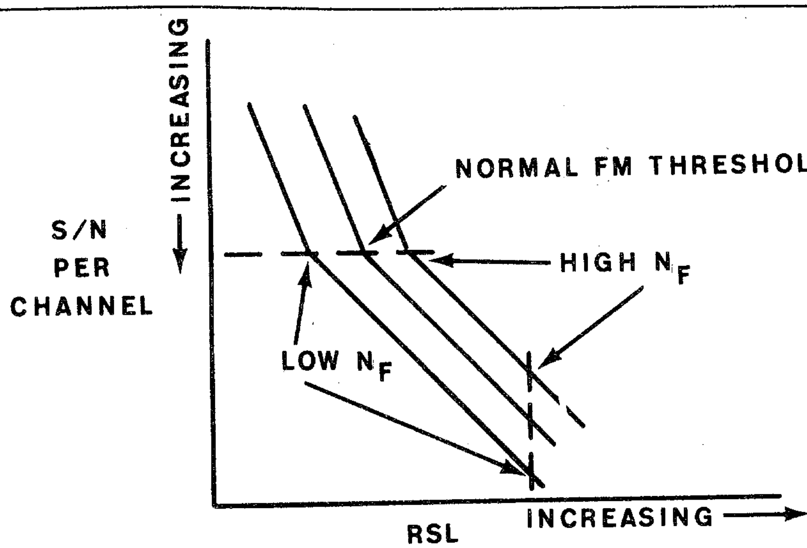


Figure 6-5. Effect of Noise Figure Changes on System Performance.

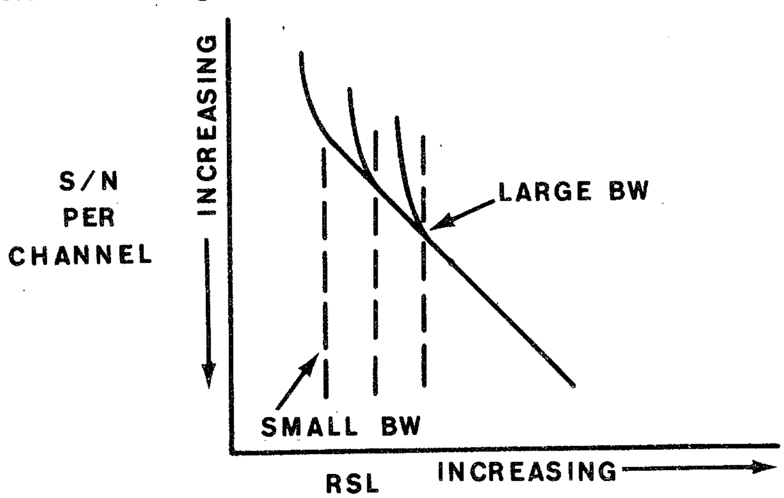


Figure 6-6. Effect of IF Bandwidth Changes on Noise Performance.

Only threshold is affected by bandwidth changes; per-channel noise above threshold is independent of bandwidth.

Baseband noise increases and FM threshold is degraded if the IF frequency response is unsymmetric or if the RF signal is not centered in the IF response of the receiver. The effect of unmodulated carrier RFI relatively close in frequency to the desired RF signal frequency is to produce significant noise at the baseband frequency equal to integer multiples of the frequency difference between the RFI and the desired signal. In addition to these beat product noise spikes, overall slot nose is increased at all baseband frequencies and baseband signal suppression starts early. For unmodulated carrier RFI with relatively large frequency offset, the beat products may fall outside the normal baseband frequency range; however, overall slot noise is increased more than for a carrier closer to the desired signal.

Because FM signals carry no information in the amplitude of the transmitted signal, it is possible to use a limiter in the receiver to suppress any residual AM in the received signal before it is demodulated. The limiter can significantly improve the received signal and thermal noise performance of the receiver near FM noise threshold (Region B).

With prominent or hard-limiting and large RSLs, the baseband signal remains constant; however, as the signal falls below FM improvement threshold, the baseband signal is suppressed rather abruptly and is reduced 2 dB for every dB of reduced signal level. Compare this to the 1 dB for 1 dB reduction in baseband signal level shown in Region C of figure 6-1.

With no limiting or soft-limiting and RSLs reduced to values in Region B, noise in the baseband increases dramatically; however, in Region A, there is less noise. In Region C, the baseband signal level starts to be lost at a much stronger RSL than with hard-limiting; however, the loss of the baseband signal is more gradual.

6-3. Intermodulation Noise. While passing through a wideband radio system, a message is present in either of two basic forms: amplitude-modulated signals or frequency-modulated signals. This message signal is normally transmitted from one station to another in the frequency-modulated form, demodulated and processed in amplitude-modulated form, and retransmitted in frequency-modulated form. The presence of modulation in either form causes instantaneous, non-sinusoidal changes in either voltage or frequency within the wideband system circuits. Because these circuits cannot be designed so that they are completely linear (that is, a linear relationship between input and output) or designed so that they respond instantaneously to rapid changes in voltage and frequency caused by the message signal, spurious signals are produced. One key point here is that intermodulation distortion (that is, the spurious signals which are produced) is caused by the presence of modulation (a signal being present) and is not present when the system is idle.

Although the intermodulation noise produced by each channel in a wideband system may be small, the additive effect produced by a large number of message signals can be large enough to impair communications circuits. Some of the spurious signals which are produced will fall within the passband of adjacent channels while some may fall within the bandwidth of the channel causing the intermodulation. The magnitude of the intermodulation noise is a function of the equipment characteristics, the number of channels in the system, and the modulation level.

There are several ways in which intermodulation noise can be produced. The following paragraphs discuss the three mechanisms which produce intermodulation noise in a wideband system.

6-4. Amplitude Distortion (Linearity or Differential Gain). When the message signal is in the amplitude-modulated form, the signal information is carried by the instantaneous value of the signal voltage; that is, the signal is a voltage that changes amplitude as a function of time. In this case, intermodulation noise is caused by non-linearity of the transfer characteristic of the networks and amplifiers that process the signal. All such circuits - because none are perfect - introduce a degree of non-linear distortion of the waveform. The result is the generation of intermodulation noise products.

Ideally, if the input voltage to a device were doubled, the output would also be doubled - this implies a linear amplitude transfer characteristic. Such a relationship is depicted by the dashed line in figure 6-7 while a typical response is indicated by the solid line.

Notice that, so far, nothing has been said about linearity versus frequency, only linearity versus drive level. In actual circuits, it is quite common for the linearity of the circuit not only to be a function of input voltage level, but also a function of input voltage frequency. Thus, a circuit might be linear over a wide dynamic range of input voltages at one frequency or band of frequencies but be non-linear over the same dynamic range of input voltages at a different frequency or band of frequencies.

This variance of the voltage transfer characteristic is sometimes called linearity and sometimes called differential gain. The units are either in dBs or expressed as a percent. Regardless of the name attached, the concept is the same. It can be measured directly, using a M/W link analyzer, or its effect can be measured using the Noise Power Ratio (NPR) test set.

Be careful here to understand that the discussion has been about amplitude non-linearity and not frequency response. A frequency response problem in a baseband module (either before modulation in the transmitter or after demodulation in the receiver) will not cause intermodulation products to be generated but will simply cause an uneven amplitude response of the baseband signal; however, amplitude non-linearities in the IF and RF sections of a radio, if left unequalized before passing through a non-linear device (such as a limiter or TWT), will cause spurious baseband frequencies to be generated because of a complicated phenomena known as AM to PM (or FM) conversion.

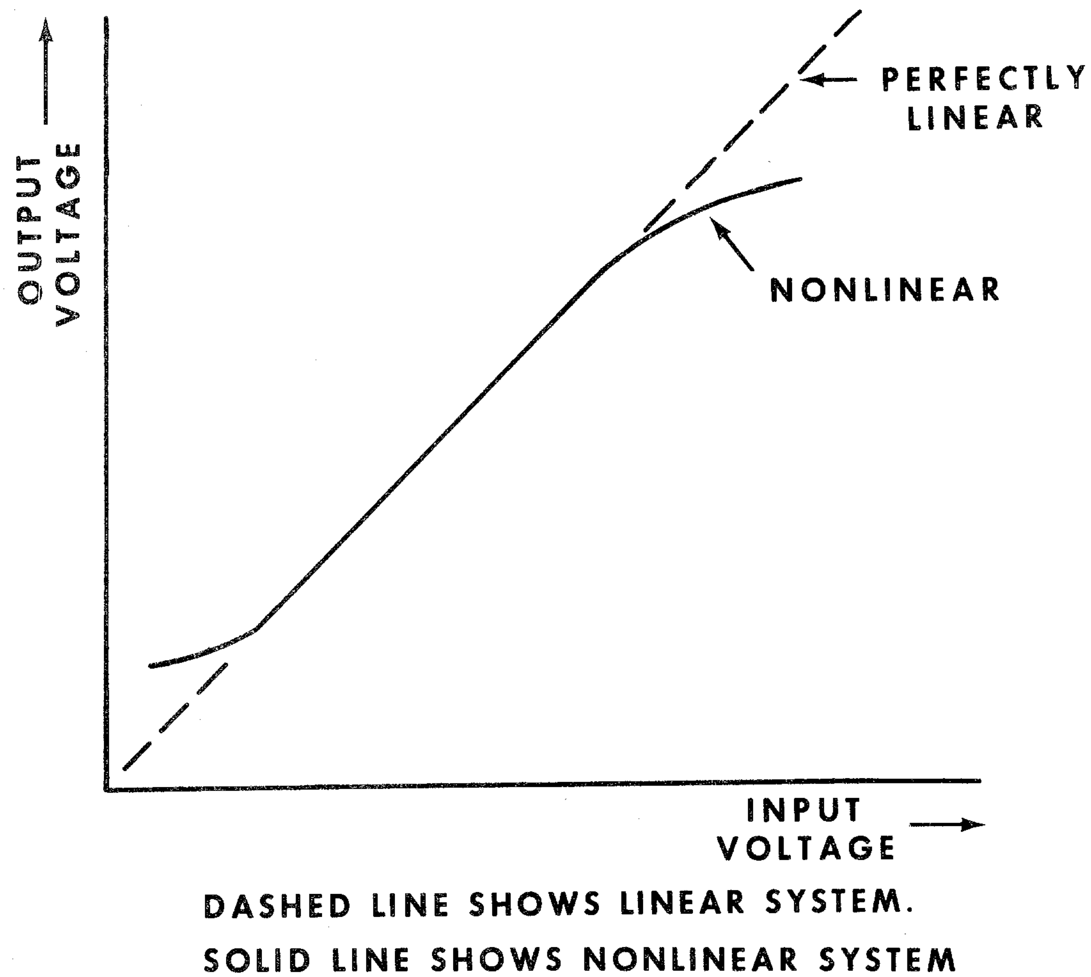


Figure 6-7. Amplitude Characteristic of Transfer Functions.

6-5. Phase Distortion. When the signal is in a frequency-modulated form, the information is carried by the instantaneous frequency of the FM signal. In this case, intermodulation noise is caused by the non-linear frequency/phase transfer characteristics of the networks and amplifiers which process the signal. In an ideal circuit, all frequencies would pass with the same absolute delay, that is, transit time. If this were achievable, the phase response would be as depicted by the dashed line in figure 6-8. The graph shape is intuitive; for example, a delay of x nanoseconds at frequency fl would be a phase shift of Ø° and would cause 2Ø° phase shift at 2 fl.

Unfortunately, in the real world, the phase/frequency transfer characteristic in non-linear, like the solid line in figure 6-8. The result is that all frequencies of interest do not experience the same transit time. This phase non-linearity can be caused at baseband where the result is excessive group delay or can be caused at IF or RF after the modulation process where the result is intermodulation distortion.

Group delay, usually measured in nanoseconds, can be thought of as the slope or rate of change of the phase characteristic in figure 6-8. For the dashed line in figure 6-8, the group delay would be like the dashed line in figure 6-9, while the group delay for the solid line in figure 6-8 would be like the solid line in figure 6-9.

6-6. Echo Distortion. Like the previously mentioned intermodulation distortion sources, echo distortion is also a modulation-related problem - it is only present during modulation. It is caused by the presence of one or more delayed signals (identical to the desired signal, only delayed in time). This delayed signal causes a phase-sensitive group delay and amplitude ripple across the FM spectrum. The result is intermodulation noise created during the demodulation process. It is a very complex function involving the relative magnitude and delay time of the echo signal with respect to the main signal; the amount of traffic on the system (affects peak deviation), per channel deviation, and the relative position of the channel in the baseband frequency spectrum. Echo distortion is much more significant in the higher baseband frequencies and much more dominant for echoes with long delay time. Two of the most likely causes are mismatches in the transmission line system (figure 6-10) and reflection in the path (figure 6-11); however, other sources (such as long IF cables and secondary paths on periscopic antenna systems) are possible (figure 6-12).

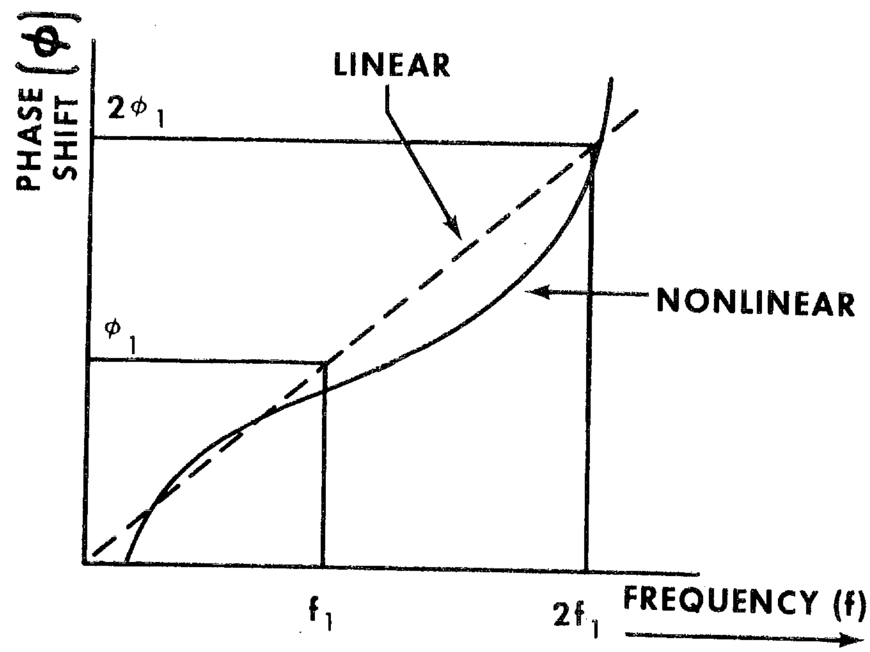


Figure 6-8. Phase/Frequency Transfer Characteristic.

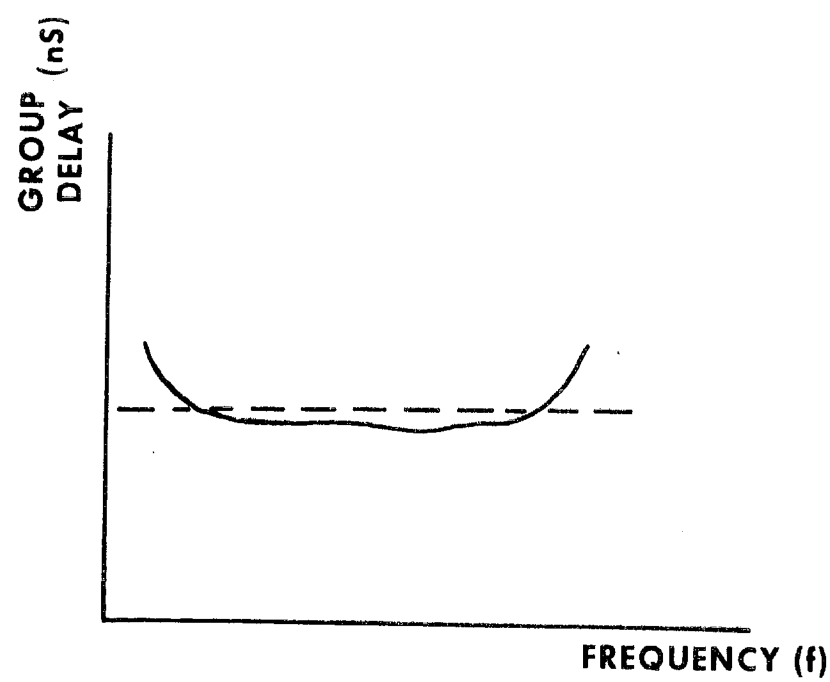


Figure 6-9. Group Delay Versus Frequency.

6-7. Measurement of Amplitude and Phase Non-Linearities and Echo Distortion. The most commonly used method of directly measuring amplitude and phase distortion is to use a M/W link analyzer. This test set measures the relative phase shift and amplitude differences between the upper and lower sidebands of a high frequency test signal (83.333 kHz, for example) which is superimposed on a low frequency (< 100 Hz) sweep signal. The idea is that the sweep signal changes the dynamic operating point of the system as it sinusoidally changes in amplitude. As the instantaneous voltage of the sweep signal is changing, so is the drive level to the system under test.

Some M/W link analyzer manufacturers distinguish between linearity and differential gain and between group delay and differential phase. There is really no difference other than in the frequency of the test tone, with the "differential" name used when the test tone frequency is above 500 kHz or so.

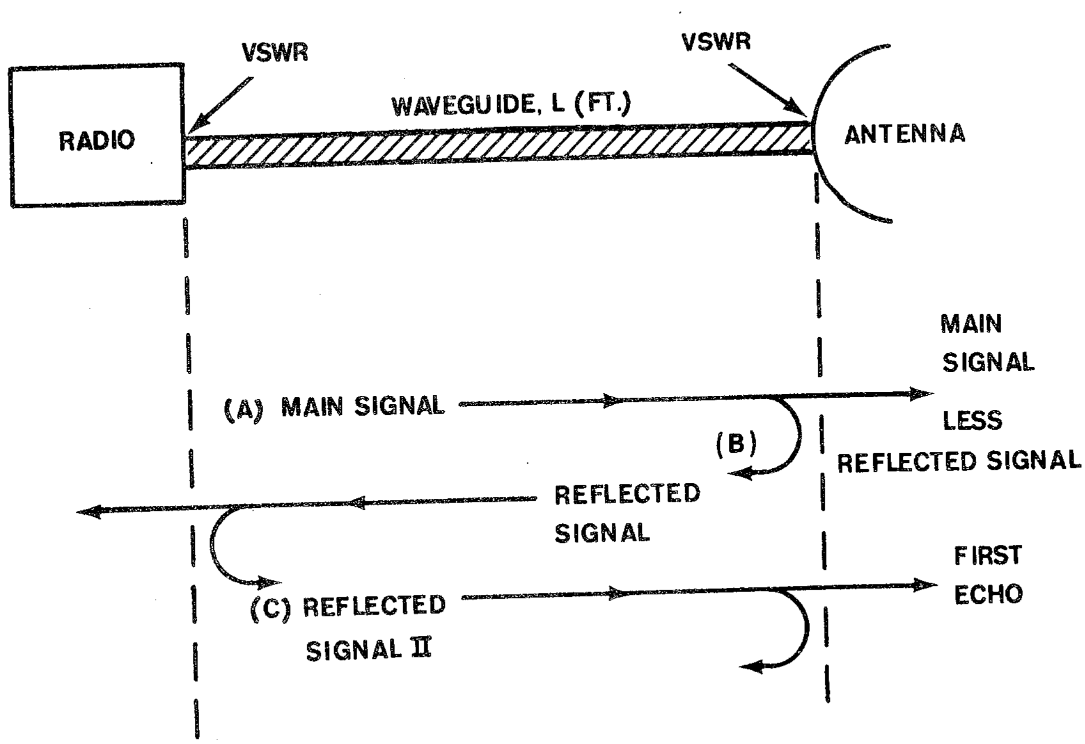


Figure 6-10. Typical Transmission Line Losses.

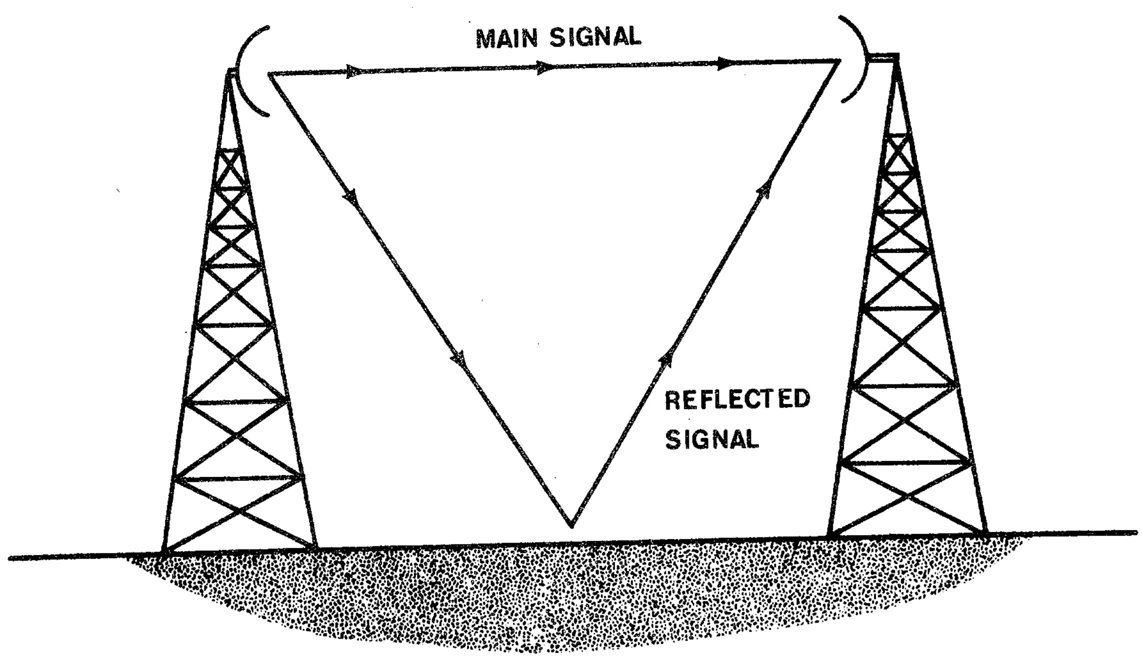


Figure 6-11. Example of Reflections in a Path.

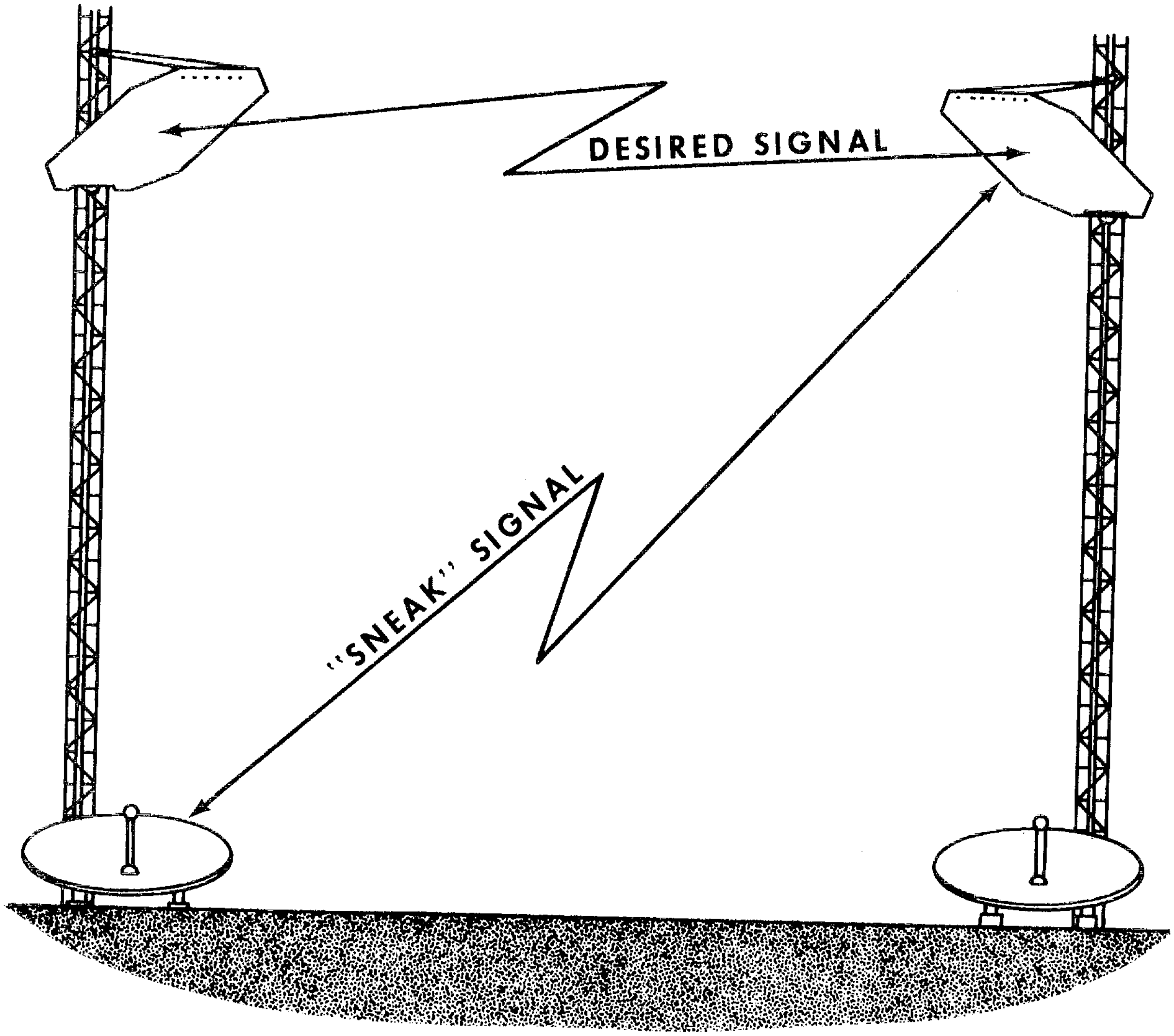


Figure 6-12. Secondary Paths on Periscope Antenna Systems.

The reason for this distinction is that some forms of distortion are a function of test tone frequency; thus, a low frequency test tone would be used to highlight one form of distortion (amplitude non-linearity, for example) while the higher tones would be used to accentuate others (AM to PM conversion, for example). The main point being that if good correlation is desired between the M/W link analyzer displays and NPR measurements, the test tone frequency used on the analyzer should correlate as closely as possible to the slot frequency used on the NPR test set.

6-8. Indirect Measurement of Amplitude and Phase Distortion and Echo Distortion. By indirect measurement, it is meant the measurement of the effects of these distortions rather than measuring the distortions themselves. This measurement (described in chapter 19) is called NPR measurement. Although Chapter 19 discusses the uses of NPR as a trouble-shooting tool, a justification for using the NPR test set to measure system performance is in order.

6-9. White Noise Load Testing. The baseband input to a wideband radio system is the sum of the individual channel loads at the multiplex. Most often, this signal is the sum of many 4 kHz-spaced voice channels. It is interesting to note that during busy-hour periods, this spectrum has so many signals that it takes on the appearance of white noise. Because of this appearance, the busy-hour baseband spectrum can be simulated with white noise of the proper level and bandwidth.

If this white noise test signal were impressed on a quiet, non-distorting wideband system, the test signal would be reproduced at the receiving end with no distortion or degradation; however, as previously discussed, the wideband transmission system is never perfectly quiet or never perfectly linear in amplitude and delay or free from echo distortion. Noise, harmonics, and intermodulation noise products are produced and degrade the reproduced signal at the distant receive end.

The idea then with white noise testing is to place a number of "quiet" slots in the white noise spectrum before it is applied to the baseband of the system under test. The "spillover" into the "quiet" slots which results during transmission is measured at the receive baseband output of the distant end. The noise measured in these "quiet" slots will be due to the distortion caused by delay and amplitude non-linearity, echo distortion, and system-idle or thermal noise contribution.

6-10. Determination of Proper System White Noise Loading Level:

Speech Loading. The total load applied to a broadband transmission system is the sum of the loads of the individual channels. In determining the total load, there are two main factors. These are: What RMS input signal level can be tolerated? What percentage of the busy hour can tolerate some overloading due to peak input signal level? As stated in chapter 4, these are based on probability functions. These essentially become voltage-time functions. In other words, what will the instantaneous magnitude, both peak and RMS, of the input signal be at any given instant? If the total load consists entirely of voice signals, the loading effect and how it changes with time will be determined by:

a. The number and distribution of active channels during the busy hour.

b. The distribution of volumes which the system must handle.

c. The distribution of signal peaks.

Factors a and b cause slow variations in the total load and combine to produce the maximum RMS load on the transmission system. Factor c causes instantaneous variations and is caused by the peaks in the signals of several talkers occurring at the same time. The total load at any given instance of time is determined by the number of active channels during that instance of time.

Many studies have been conducted to determine what percentage of the busy-hour a channel might be active. This is sometimes referred to as the activity coefficient. Systems with fewer channels will have the highest activity coefficient because the averaging effect between channels will be less. It was agreed internationally that, for speech loading of greater than or equal to 240 channels, the mean level assigned to each channel would be -15 dBm0 and the composite level could be determined on a 10 log N basis. For fewer than 240 channels and greater than or equal to 60 channels, the averaging effect between channels is less, so a separate expression was developed. For systems less than 60 channels, no CCIR recommendations were made; however, the Federal Communications Commission (FCC) has made some. The following expressions apply for determining the system load, hereinafter referred to as the noise load ratio (NLR):

$$\text{NLR (dBm0)} = -15 + 10 \log N \text{ for } N \geq 240 \qquad \text{Eqn 6-5}$$

$$\text{NLR (dBm0)} = -1 + 4 \log N \text{ for } 60 \geq N \geq 239 \qquad \text{Eqn 6-6}$$

$$\text{NLR (dBm0)} = 2.6 + 2 \log N \text{ for } 12 \geq N \geq 59 \qquad \text{Eqn 6-7}$$

The volume of data-type transmissions is continually increasing and is expected to continue upward in the future. Data signals impose a much greater load on multichannel transmission systems than do ordinary voice channels. If too many channels of a multichannel transmission system are assigned to carry data-type signals, it can effectively reduce the load-handling capacity of the system. This could mean that some of the channels normally carrying voice traffic would have to be disconnected or a redistribution of channel assignment would have to be made with other groups.

Because of this, DCA has decided that new systems would be specified to operate under continuous data loading. They determined that all new radio equipment would maintain specified performance when loaded with data (tone packs) at -10 dBm0 per channel. Although most data is applied at -13 dBm0 per channel, the -10 dBm0 provides some margin of safety. Thus:

$$\text{NLR (dBm0)} = -10 + 10 \log N \quad \text{for any N} \qquad \text{Eqn 6-8}$$

6-11. System Noise and Deviation Sensitivity.

Before discussing intermodulation noise further, it is important to understand the relationship in a wideband FM system between noise and deviation. In FM radio systems, baseband signal voltage amplitude fluctuations are translated into RF carrier frequency deviations (chapter 10), controlled in width by the deviation sensitivity (kHz/volt) of the transmitter. The idea would be to set this deviation sensitivity at some optimum balance so that maximum system performance can be obtained. As depicted in figure 6-13, as the transmitter deviation is increased, the output at the receiver baseband on the distant end must be reduced to maintain proper system gain. This reduction in gain at the receive end causes a reduction in system idle or thermal noise; however, one cannot keep increasing transmitter deviation forever. A point will be reached where the transmitter is deviated into non-linear or unequalized spectra. The result is increasing intermodulation which becomes rapidly dominant.

As long as thermal or idle noise is the controlling factor in system performance, increasing deviation will improve system performance; however, increased deviation, in turn, causes intermodulation noise which can quickly become the controlling factor. The effects of deviation changes at high and low RSLs are shown in figures 6-14 and 6-15. The controlling of deviation is then a tradeoff between thermal or idle noise and intermodulation noise. An optimum setting of deviation must be taken into account based on the expected RSL and system loading. Figures 6-16 and 6-17 depict this. These optimum deviation settings have been determined by the manufacturer or system engineer and should not be "tailored" in the field.

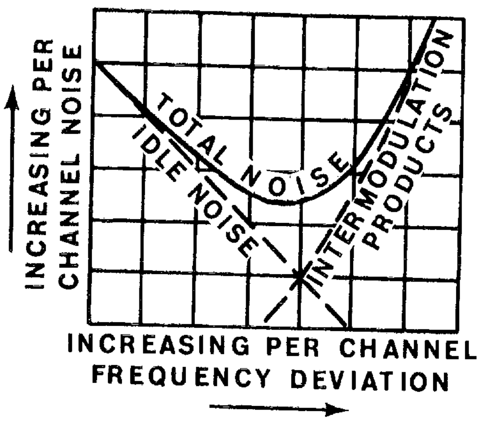


Figure 6-13. NPR Versus Noise Loading Curve (Bucket Curve).

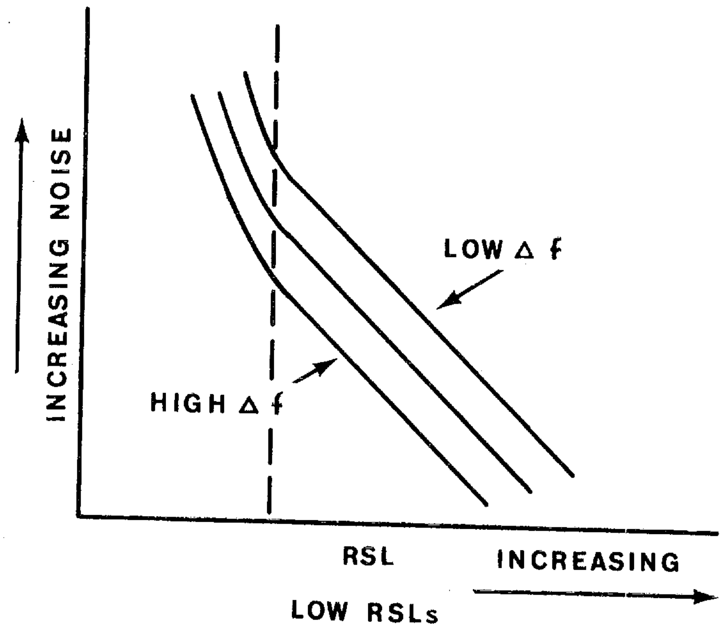


Figure 6-14. Effects of Deviation Changes at Low RSLs.

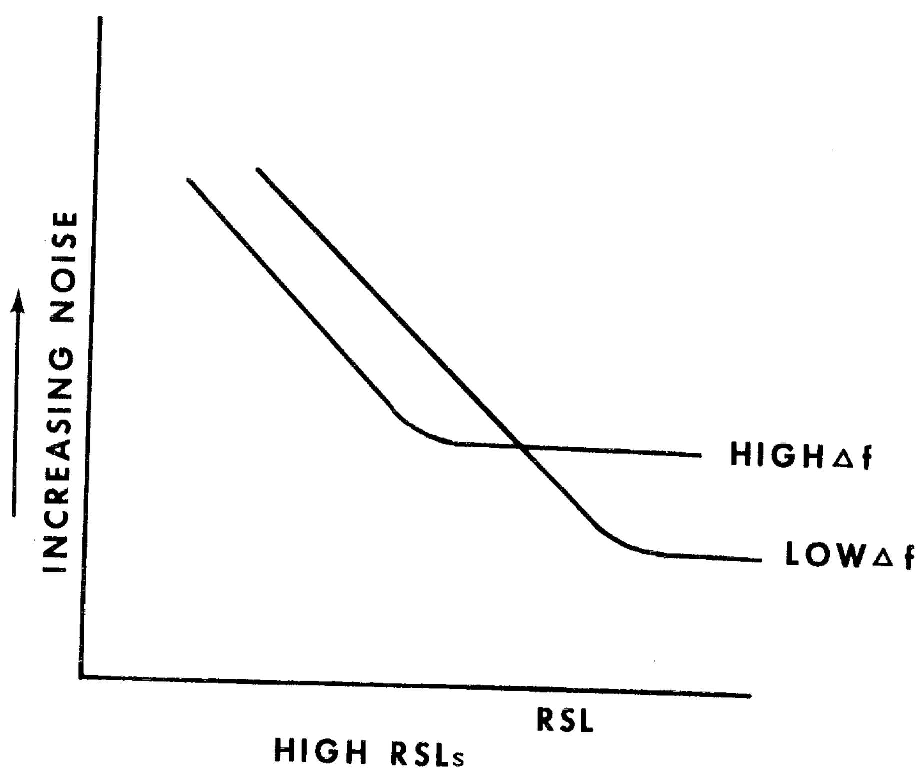


Figure 6-15. Effects of Deviation Changes at High RSLs.

6-12. System Noise and System Loading Level. As the number of channels into the system increases, so does the NLR. This increasing load causes the transmitter to be deviated further - this is the same effect as increasing deviation sensitivity. If one were to measure the noise in an idle channel (3.1 kHz slot) at the receive end while the system load was being increased at the transmitter, a curve similar to figure 6-18 would be produced. Initially, as the noise load is gradually increased from that equivalent to a few channels, there would be no change because the intermodulation noise caused by this light load would be completely dwarfed by the system thermal or idle noise which is not a function of system loading. Eventually, however, a noise load equivalent to many channels would cause the intermodulation contribution to the system noise to become significant. The point at which the idle channel noise (ICN) increased 3 dB (implying equal intermodulation and idle contributions) is called the break point. Further increases in system load beyond this point will cause rapid increases in system noise. Soon the intermodulation noise would be disruptive to traffic.

The most powerful way, from a troubleshooting and analytical point of view, to present this noise versus loading, which has just been discussed, is in the form of NPRs versus loading or "bucket curve." The "bucket curve" is a graphical presentation of noise within a wideband system, displayed in a form that permits separation and identification of the individual intermodulation and idle degradations. The "bucket curve" is simply a plot of NPR at many noise loading levels with some levels being well above the normal level and some well below the normal level. The slot frequencies are usually selected so that one falls near the low end, one near the middle, and one near the top of the baseband frequency spectrum. Chapter 19 discusses NPRs in more detail while the *Lenkurt Demodulator, March/April 1976, "Bucket Curves,"* discusses bucket curves and troubleshooting procedures in great detail.

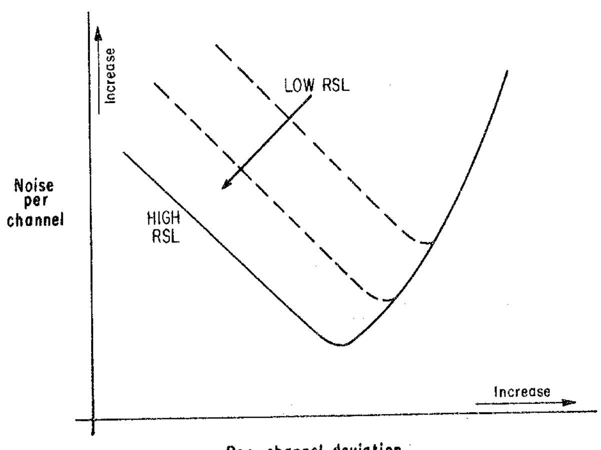


Figure 6-16. Changes in the Deviation Curve with RSL Changes. RSL only affects thermal portion of the curve.

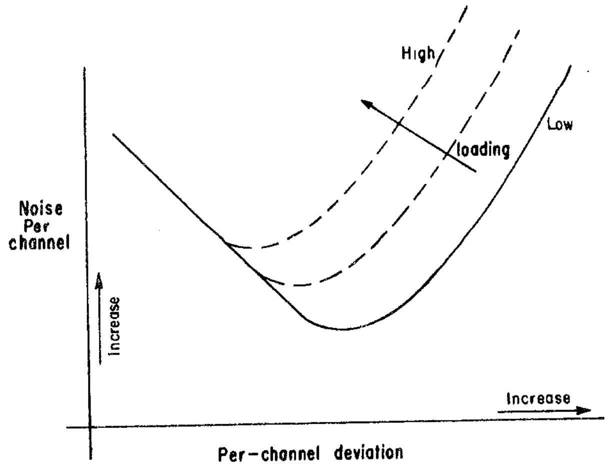


Figure 6-17. Changes in the Deviation Curve with Changes in System Loading. Only the intermodulation region is affected.

NPR as an Analytical Tool. If an NPR loading curve measurement is taken in three slots in the baseband, the data can be used as a maintenance tool. Beyond the optimum drive point, the noise rises due to intermodulation products. The steepness of the curve depends on the degree of non-linearity in the system. We have stated that noise in different slots is caused by different processes. Intermodulation noise in the low slot is caused by "static" distortion; noise in the high slot is caused by "dynamic" distortion.

Static distortion occurs while the signal is in the form of amplitude variations (that is, before the modulation or after demodulation). Dynamic distortion occurs when the signal is carried as FM as in RF amplifiers, waveguide, receiver IF, etc. If an intermodulation problem is seen in one slot and not the other, the particular type of distortion problem is indicated. For example, if the curves of figure 6-19 were obtained, static distortion problems would be indicated. A partial list of causes of static distortion includes:

a. System deviation adjustments.
b. Missing 75 ohm terminations.
c. Microphonic noise.
d. Defective coupling capacitors in limiter.
e. Gassy electron tubes.
f. Linearity adjustments in transmitter and receiver.
g. RF interference.

A partial list of dynamic distortion causes includes:
a. Impedance mismatches.
b. Defects in waveguide (dents, water, etc.).
c. Antenna defects.
d. Propagation time and phase characteristics of RF filters.

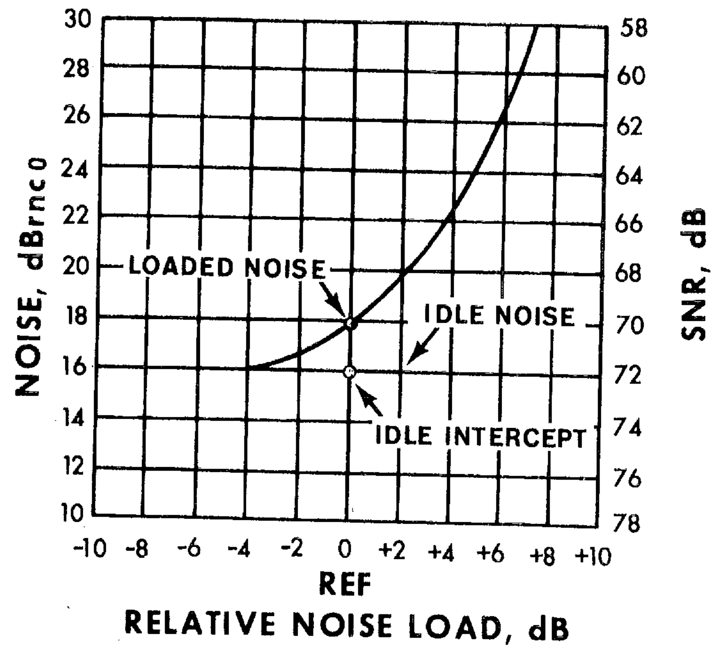


Figure 6-18. System Noise Versus System Loading.

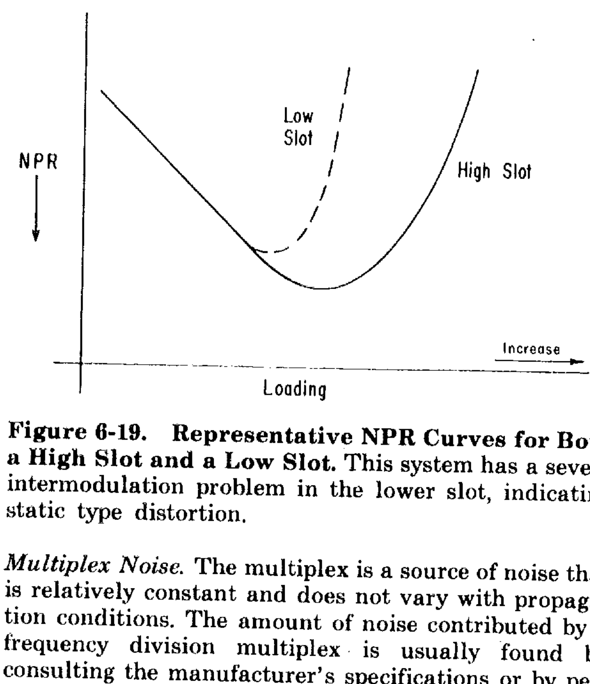


Figure 6-19. Representative NPR Curves for Both a High Slot and a Low Slot. This system has a severe intermodulation problem in the lower slot, indicating static type distortion.

Multiplex Noise. The multiplex is a source of noise that is relatively constant and does not vary with propagation conditions. The amount of noise contributed by a frequency division multiplex is usually found by consulting the manufacturer's specifications or by performing a loopback of the multiplex under a fully-loaded condition (paragraph 20-5). In the pulse code modulation multiplex, the noise contribution is essentially determined by the number of quantizing levels and the level of the applied signal. Use of companders is normally employed to reduce the dependency of the S/N on the applied signal and to provide a relatively constant S/N over a wide range of input signals.

To this point, M/W receiver noise performance has only been considered for a single receiver. An actual M/W terminal normally will have two or more receivers with the capability to contribute to the baseband output of the terminal. The outputs of the various individual receivers are added (combined) to produce a baseband signal which, if the combiner functions properly, will have superior S/N performance when compared to the performance of a single receiver. The received signal from the various receivers can be combined as an IF signal (pre-detection combining) or as a baseband signal (post-detection combining). Pre-detection combiners provide significantly better combiner signal improvement near FM threshold when compared with post-detection combining as long as interference is not a factor. In a radio frequency interference (RFI) or jamming environment, a pre-detection combining technique, based on total signal power, may tend to seriously degrade the overall system noise performance by favoring the receiver with the interfering signal. Post-detection combining, based on out-of-band noise, will generally outperform pre-detection combining in this situation by favoring the receiver with the least baseband noise. Post-detection combining is the most common form of M/W wideband receiver combining.

If the combiners do not function properly, baseband signal level stability and IPN performance can be seriously degraded. In general, three types of combining are used: switching (selection), equal gain, and maximal ratio (also called ratio-squared or variable gain) gain. The effect of combiners is a complicated subject. The relative performance of combiners depends on the type of RSL fading the M/W terminal experiences. There are, however, two cases of special interest. The idealized situation for an LOS terminal is for all receivers to see a constant, unfading RSL which is the same power level at all receivers. For post-detection (baseband) combining, the selection combiner will give no improvement over a single receiver. The equal gain and variable gain combiners will have exactly the same S/N improvement. In either case, the improvement will be ten times the common logarithm of the number of receivers used for combining. Unfortunately, this idealized situation is seldom realized in practice. Diversity engineering sometimes intentionally makes the RSL of one receiver different than that of the other receiver. Also, different waveguide run lengths and waveguide hybrid losses can also invalidate the idealization. On the other end of the scale, the receivers generally experience randomly fading receive signals. If the receivers all have the same average RSL and if the RSL are all Rayleigh amplitude-distributed, then the combiner S/N improvement will be, on the average, the values shown in figure 6-20. Unfortunately, in real life, for various reasons, the average RSLs of the various receivers is often different by a few dB. Average combiner noise performance under practical conditions is difficult to predict accurately without sophisticated parameter measurement of the actual radio path.

Combiners serve to improve the baseband signal to noise performance of a M/W terminal. Since the signal into the combiner with the most noise will be rejected, the most noise at the combiner output will be, on an average, less than the noise out of a single receiver. As a rough estimate of the actual slot noise that will appear at the combined baseband of a M/W terminal, the curves on figure 6-20 can be used. If the terminal is LOS and the RSLs are stable and equal, the following conditions are approximately correct. For selection combining, the noise at the combined baseband will be exactly the same as for a single receiver. The baseband noise can be read directly from the quieting curve of a single receiver. If the terminal uses equal gain or ratio-squared combining, the noise at the combined baseband will be less than for a single receiver by the amount shown on the following chart for variable gain combining.

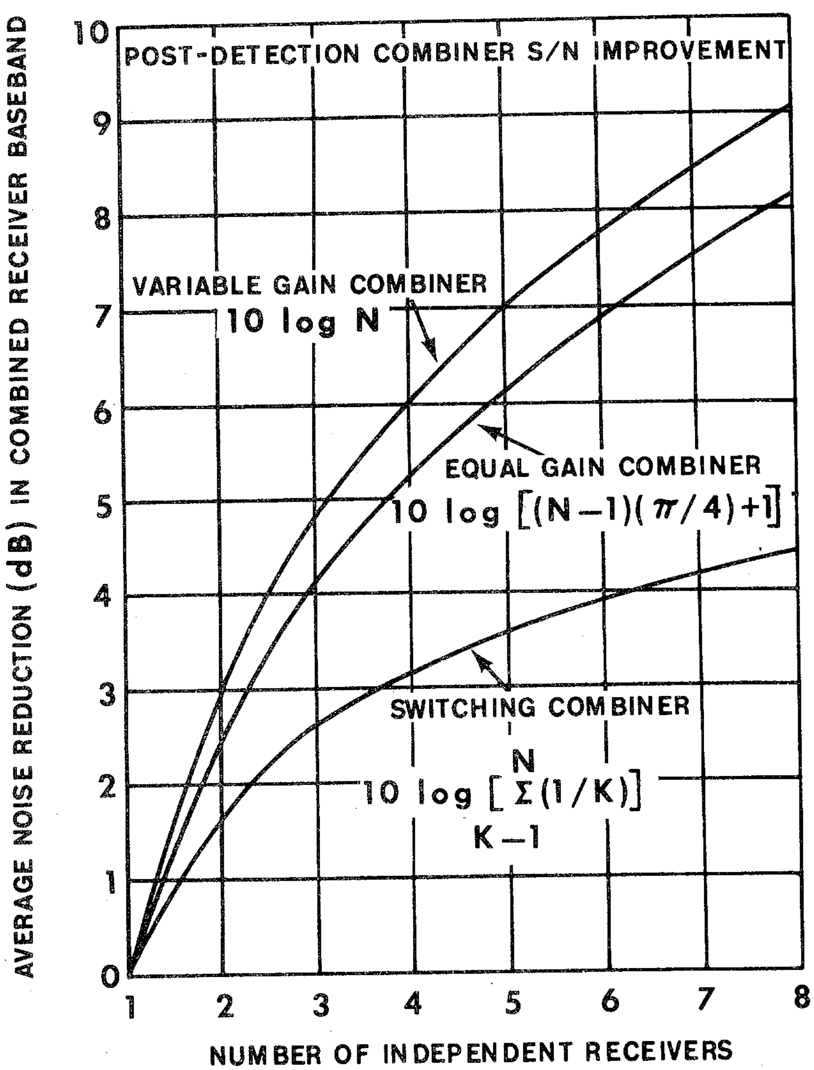


Figure 6-20. Combiner Improvement Curves. (D. G. Brennan, "Linear Diversity Combining Techniques," Proceedings of the IRE, pp 1075-1102, June 1959.)

The slot noise at combined baseband will be the slot noise read from the quieting curve minus the dB valve read from the chart. For a tropo terminal, if the RSLs are the same and all are randomly fading, then the noise improvement will be found by using the curve which applies to the type of combining appropriate. The slot noise at the combined baseband will be the slot noise value read from the quieting curve minus the dB noise improvement factor determined from figure 6-20.

Chapter 7
Radio Wave Propagation

7-1. General. Just as a chain is no stronger than its weakest link, a radio communications system is no stronger than its weakest component. The propagation of radio waves is one of the vital factors in a communications system and it is the link in the chain which the communicator has the least control of. While transmitters, receivers, and other RF equipment can be designed to known standards, radio propagation is controlled by nature alone. Man can only observe the manner in which nature controls radio waves and plan his systems to fit. In this chapter, we will explore the behavior of radio waves between transmitting and receiving antennas and how this behavior relates to the performance of a wideband communications system.

7-2. Propagation:

a. Radio Wave Action in Free Space. In order to understand radio wave propagation, it is first important to understand the behavior of radio waves when they are not influenced by any external factors. This understanding is best obtained by considering the action of the waves in free space, that is, where there is no atmosphere, temperature variances, or the like.

What exactly does a radio wave do in free space? On leaving its source of radiation (antenna), a radio wave travels outward in straight lines from the antenna at the speed of light (approximately 300,000,000 meters per second) and continues without stopping (figure 7-1).

At this point, let us detour for a moment and consider the subject of radio wave direction. The example so far has shown the waves propagating in all directions from the antenna. Such an antenna is known as an omnidirectional antenna. The majority of antennas used in communications are not omnidirectional, but are designed to direct, or focus, the waves in a special direction. For the rest of this paragraph, only examples of directional antennas will be shown and discussed. A directional-type antenna could be portrayed as shown in figure 7-2. Concentrating transmitted power in narrow beams gives an "apparent" increase in power. This increase is expressed as antenna gain in super high frequency (SHF) path loss equations.

It is often expedient to portray the radiating wave as a single line. Now let us proceed to the more practical study of how radio waves behave in real life, as opposed to free space.

b. Factors Which Affect Propagation. Since radio waves are similar to light waves in many respects, certain optical principles are useful in describing the propagation of radio waves in space. The most important are refraction, reflection, and diffraction. In addition, a number of attenuation factors are used to predict the effects of these principles on the intensity of the wave.

(1) Refraction. Refraction, or the continuous bending, of radio waves occurs because of variations of temperature and water vapor pressure in the atmosphere. In the lower atmosphere, both temperature and water vapor pressure normally decrease with increasing altitude, causing a change in the index of refraction. This almost linear decrease in the index of refraction causes the radio wave to travel at a greater velocity at the higher altitudes. The effect is to bend the wave downward. The amount of downward bending under "normal" atmospheric conditions is of the same relative magnitude as the radius of the earth. If we view the earth as a flat plane (figure 7-3A), a radio beam would appear to droop in the middle. Analyzing a radio path, using this technique, would require computing the amount of droop for any changes in equipment location or configuration. If, on the other hand, the radio path were viewed as a straight line with the earth bulging up (figure 7-3B), interference points and radio horizons are easily determined.

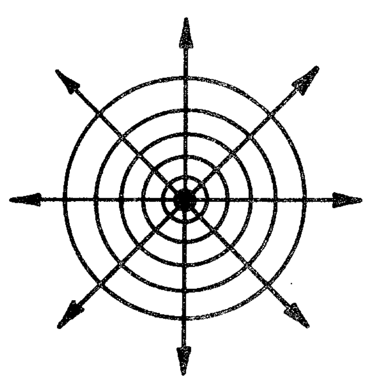


Figure 7-1. Radio Wave Movement from an Antenna in Free Space.

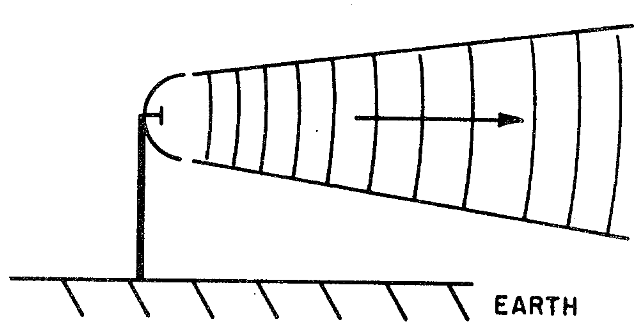


Figure 7-2. Radio Wave Movement from a Directional Antenna.

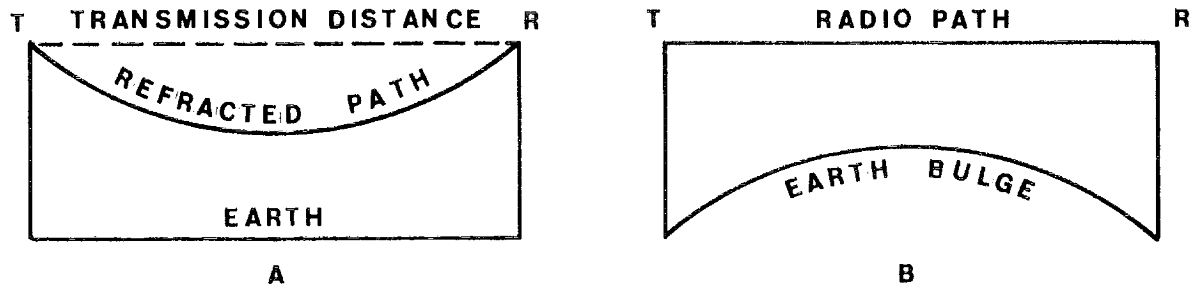


Figure 7-3. Effects of Refraction on the Radio Wave Path.

There is a standard way of expressing the rate of bending. The amount of bending is defined by the refractive index gradient or, more commonly, the effective earth radius factor (K). The K factor, multiplied by the true earth radius, gives the radius of a fictitious earth curve. Any change in the amount of beam-bending or refraction index can be expressed as a change in the K value. The definition for "standard atmosphere" is a 4/3 earth radius (figure 7-4). The earth becomes increasingly flat as the value of K increases until at K= ∞ the earth appears to be flat to the radio wave. Figure 7-5 shows the bending of the radio waves as the value of K changes. The earth curvature for various values of K can be calculated from the relationship:

$$h = \frac{d_1 d_2}{1.5K} \quad \text{where}$$

h = vertical change in feet

d_1, d_2 = distance to the respective ends of the path in miles.

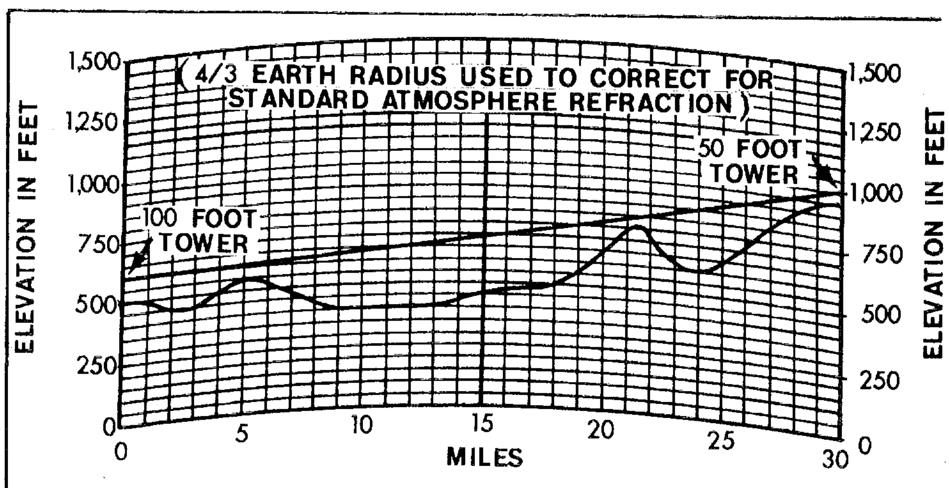


Figure 7-4. Earth Profile Chart at 4/3 Radius. (Courtesy GTE Lenkurt)

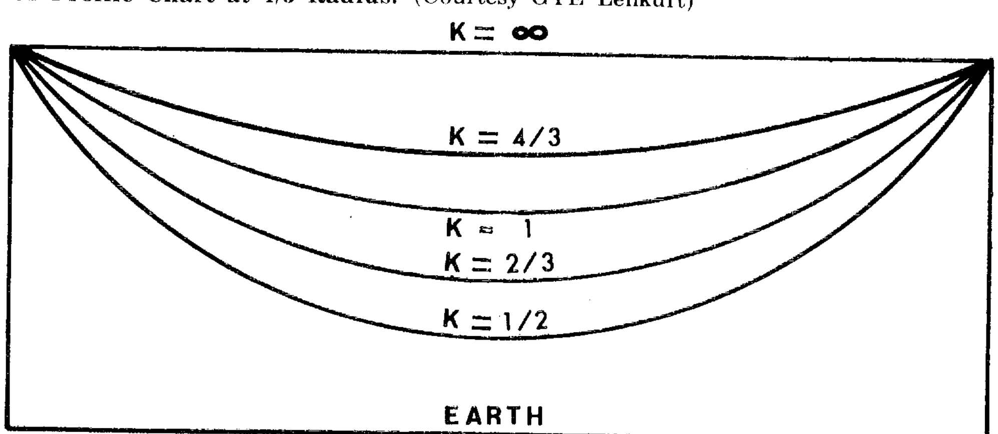


Figure 7-5. Bending of Radio Wave as a Function of K Values.

For tropo, there is little change in RSLs if K remains greater than 1.0, providing ducting is not a factor; however, outside of this normal range, problems do occur. With K less than 1, earth bulge becomes more pronounced and the path could be obstructed and be vulnerable to excessive multipath fading. If K becomes negative, a condition known as ducting exists and the radio beam may be trapped within the duct (figure 7-6).

As the atmosphere becomes more super-refractive (K becomes greater), little change will be seen in RSLs until the refractive index goes beyond infinity and becomes negative. At that point, a duct is formed. Ducting occurs when a temperature or humidity inversion occurs, making the thicker air on top instead of the bottom, as it usually is. The area of inversion will be relatively thin and can be viewed as a duct, or layer, of special air. The wave will tend to bounce back and forth between the edges of this layer and can, as a result of this ducting, be bent away from the intended destination (figure 7-6). The angle of incidence of the signal to the duct is critical. As a general rule, if the signal intersects the duct at less than 0.5°, the signal will be reflected. If the intersect is greater than about 2.5°, the signal will pass through the duct with some attenuation. Between these two intersects, the signal will enter the duct and be trapped.

A duct is a relatively rare occurrence and can be identified by its long duration (usually several hours). Note that if the duct happens to lay between and include the transmitting and receiving antenna, communications may actually be improved markedly; however, just as ducting is rare, the chance of an inter-antenna duct is even rarer.

All of our examples involving ducts so far have depicted a vertical path change. It should be observed that a horizontal duct is also possible (figure 7-7).

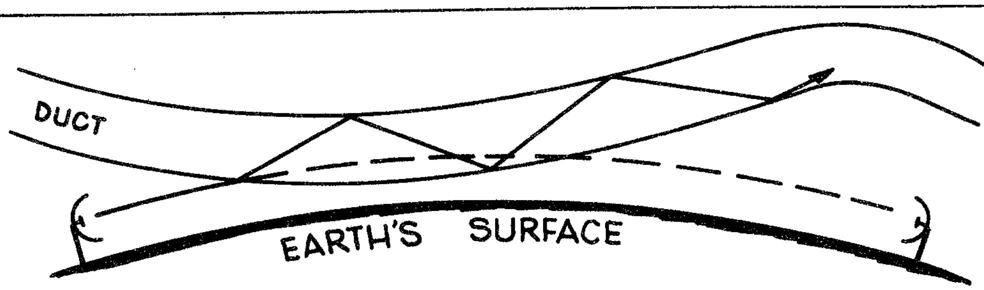


Figure 7-6. Radio Wave Ducting. Radio waves tend to be trapped inside and guided by ducts in the atmosphere.

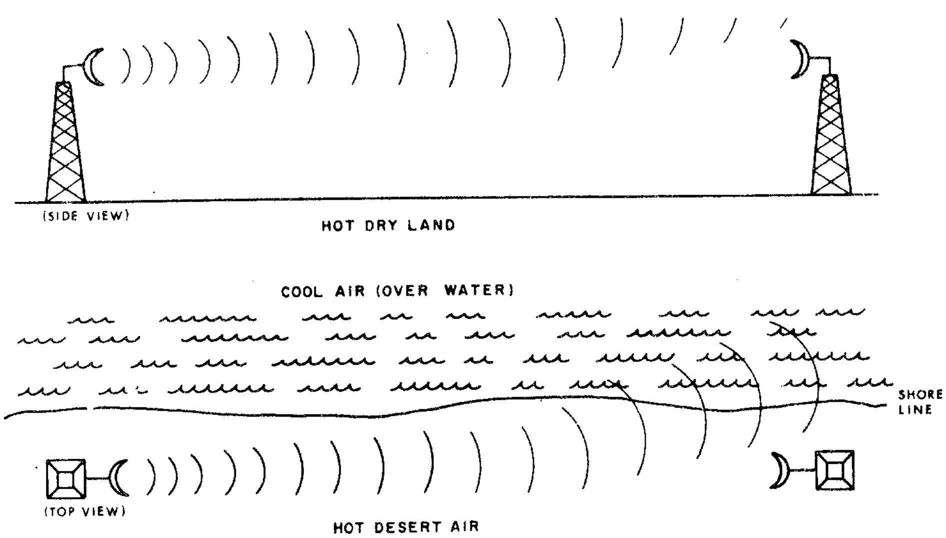


Figure 7-7. Horizontal Duct.

(2) Fresnel Zones. Like the index of refraction (K factors), the concept of Fresnel zones is necessary to the understanding of radio wave propagation. Outside the immediate vicinity of the transmit antenna, the energy radiates outward and may be approximated by a plane wave. The signal reaching the receive antenna is made up of many sources that travel different distances and have different phase relationships. The concept of Fresnel zones allows the prediction of the various phase relationships. Fresnel zones are plots of points that produce $\frac{\lambda}{2}$ increases in path length compared to the beam centerline. The plots produce circular rings around the beam centerline. The odd-numbered Fresnel zones boundary is made up of

all possible points from which a wave could be reflected with a path length difference of a half-wavelength. From the concept of wavelength (as was discussed in chapter 4), it would seem that a half-wavelength difference between the reflected and the direct wave would cause complete cancellation but it is assumed that the reflecting surface imparts a phase reversal (180°) which makes the energy of the direct and reflected wave add rather than cancel. Similarly, the even-numbered zones impart a full wavelength delay between the direct and reflected wave which causes complete cancellation of the received energy. The radii of the Fresnel zones may be calculated for any point along a path by using the following formula:

$$F_n = 72.1\sqrt{\frac{n\, d_1 d_2}{f_{GHz}\, D}}$$ where

F_n = radius of the nth Fresnel zone in feet.

d_1 = distance from a point to one end of the path in miles.

d_2 = distance from the same point to the other end in miles.

f_{GHz} = frequency in gigahertz.

D = total path length in miles.

or $F_n = 17.3\sqrt{\frac{n d_1 d_2}{f_{GHz}\, D}}$ where

D, d_1, d_2 distances in Km

f frequency in GHz

Fn is meter

First Fresnel zone energy boundary along a path is shown in figure 7-8.

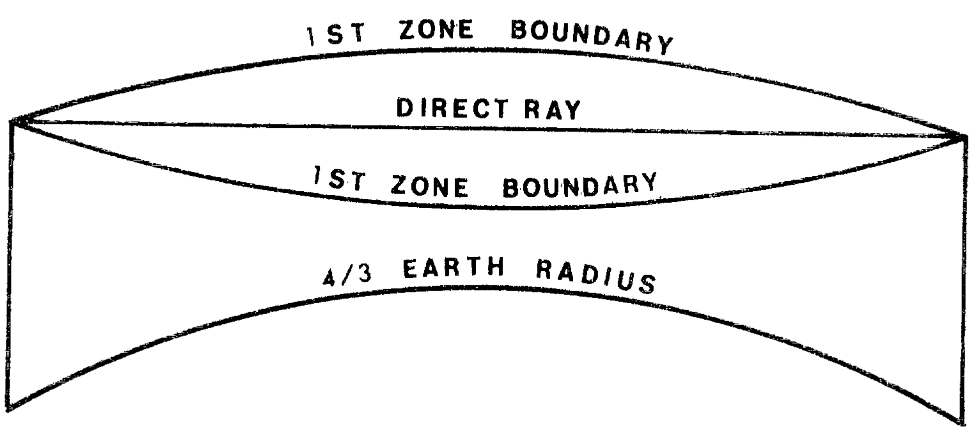


Figure 7-8. First Fresnel Zone Boundary.

(3) Reflection. With the knowledge of Fresnel zones and the fact that radio waves behave like light waves, the variations of receive signal strength can be investigated as a function of reflections. When a wave is reflected from a smooth surface, the angle of incidence and the angle of reflection are the same. There are many possible reflection points of a radio beam (figure 7-9) along a path but there usually is only one point where the reflected energy will reach the receive antenna. Calculation of this reflection point is based on the height of the transmit and receive antennas and the path distance. For flat earth ($K = \infty$), the reflection point satisfies the following relationship:

$$\frac{h_1}{h_1 + h_2} = \frac{d_1}{d_1 + d_2}$$

h_1 = elevation of the lowest antenna in feet.

h_2 = elevation of the higher antenna in feet.

d_1 = distance from h_1 end to reflection point in miles.

d_2 = distance from h_2 end to reflection point in miles.

When K = 4/3, the reflection point can be calculated from the following formula:

$$\frac{h_1}{d_1} - \frac{d_1}{2} = \frac{h_2}{d_2} - \frac{d_2}{2}$$

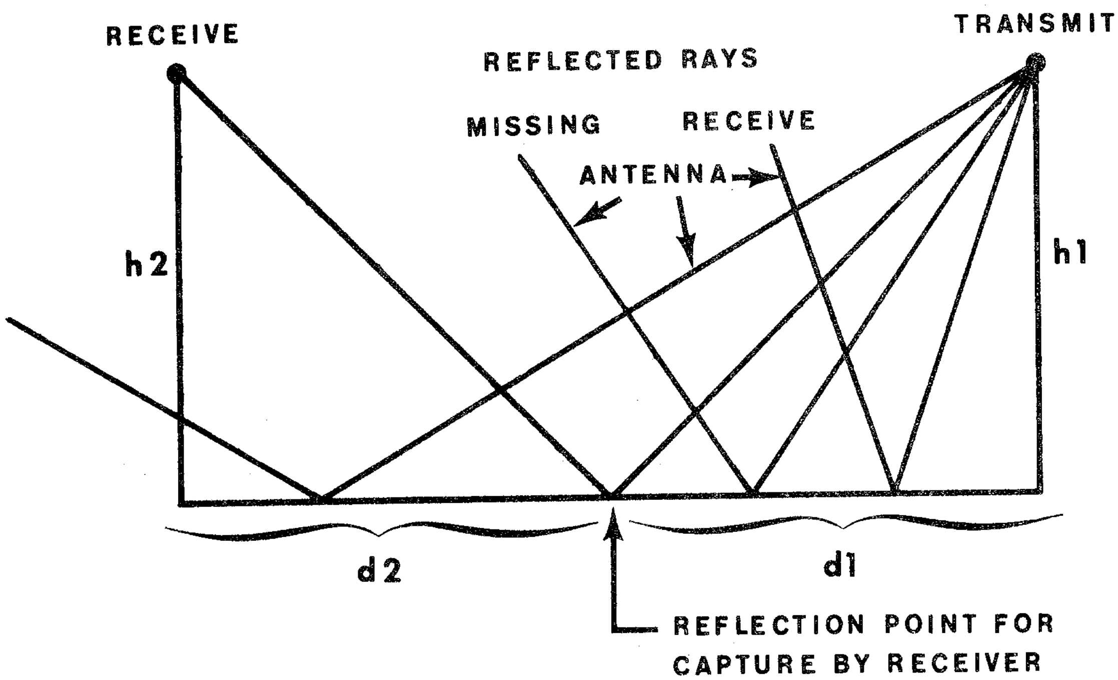


Figure 7-9. Incident and Reflected Beams on Flat Earth (K=∞).

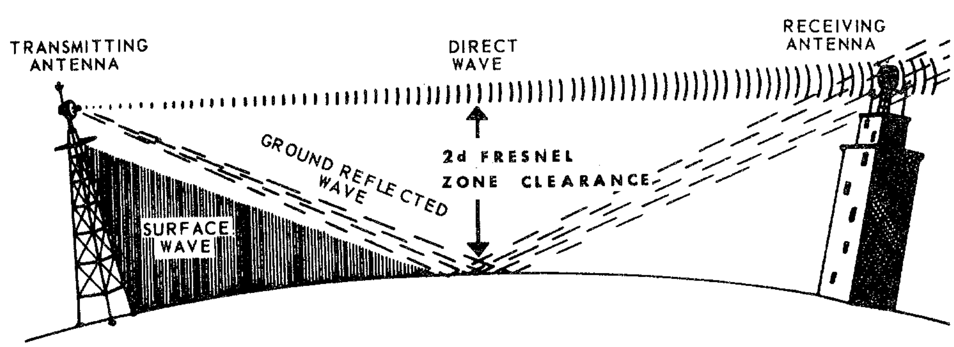


Figure 7-10. Ground Reflected Wave with 2d Fresnel Zone Radius.

The solution is found by trial and error until values are found which satisfy the equality or very nearly so. The symbols used here were defined previously. The effects of reflections on the receive signal strength is found by recalling that the even-numbered Fresnel zone reflections cause partial cancellation of the received signal and that odd-numbered Fresnel zone reflection will reinforce the received energy. Reflections can occur from the upper surfaces of the atmosphere or from the ground. It is the ground-reflected wave which concerns us now.

If there is a good "mirror" surface available for the ground wave, it will be reflected into the receiving antenna. A body of water or smooth earth is a good reflector, while wooded terrain provides weaker reflections. In general, a reflected ground wave is an undesirable situation. When at all possible, systems are designed to have the reflected wave cancelled by the surrounding terrain. Figure 7-10 depicts a ground-reflected wave. If the distance between the reflection point and the direct wave is an even Fresnel zone radius, complete cancellation of the received energy may occur if the surface is a good reflector.

(4) Diffraction. Wideband LOS radio paths (and antenna height) are normally selected to provide LOS clearance between the transmitter and the receiver; however, a direct path does not always guarantee good transmissions. If the wavefront passes too near an obstacle (such as a hilltop or building), partial obstruction of the radiated energy will occur. If the M/W path is subjected to refraction conditions (inverse bending), partial blockage of the transmitted energy occurs but some of the energy still arrives at the receiver by diffraction. The amount of "obstruction" or diffraction loss is dependent on the Fresnel zone radius of the wavefront as compared to the area of the obstruction and the reflectivity of the obstruction. The first Fresnel zone carries the bulk of the radio wave energy. If an obstacle is placed in front of the wave so that all of the wavefront below the LOS is blocked, half of the central area is obstructed and a 6 dB loss of energy occurs (figures 7-11 and 7-12).

Now let's see how an obstruction can help us. If an obstruction is at the height where all of the first Fresnel zone is exposed, the power received at the antenna will be greater than it would be if there were no obstruction at all.

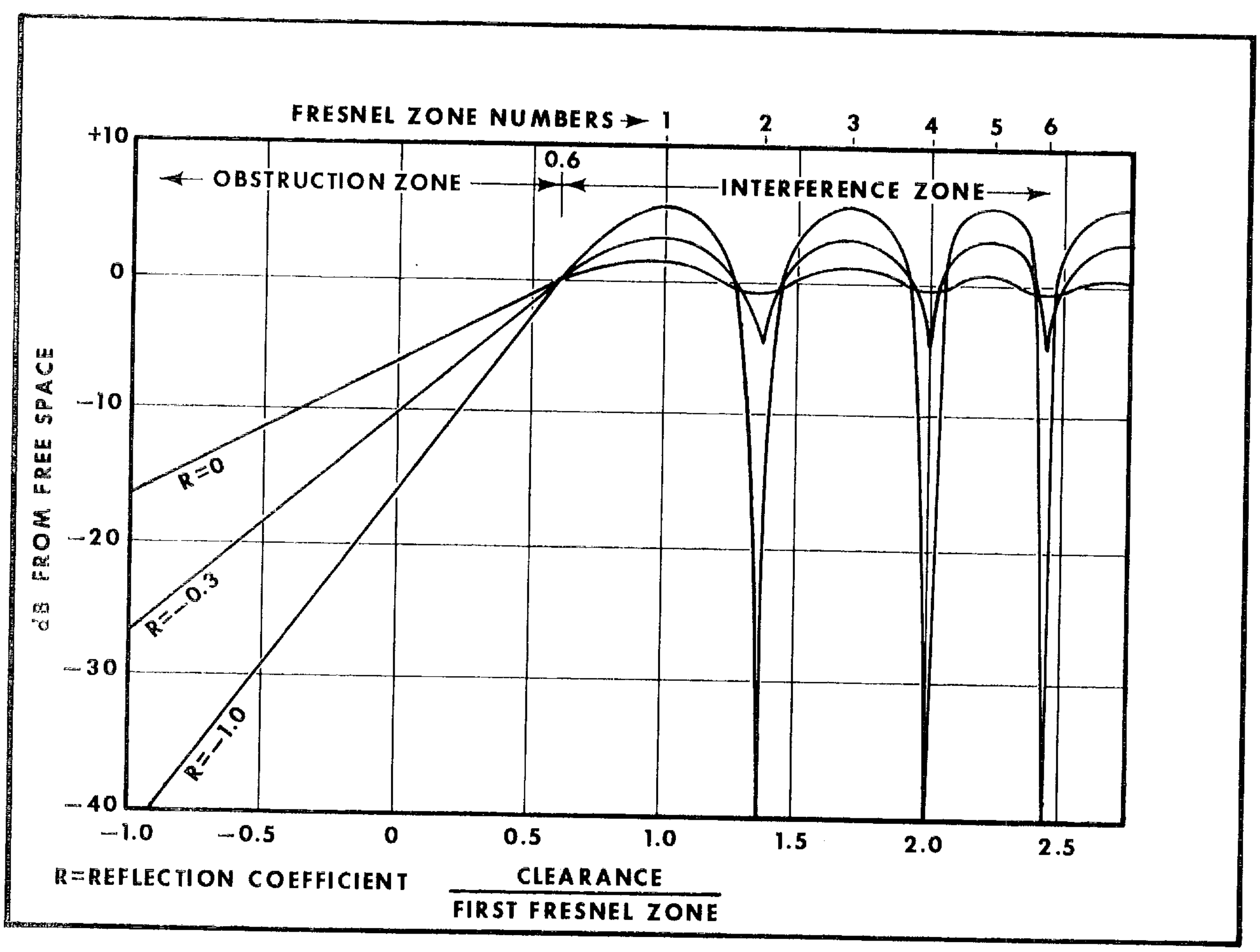


Figure 7-11. Loss Estimations for Fresnel Zone Clearance. (Courtesy GTE Lenkurt)

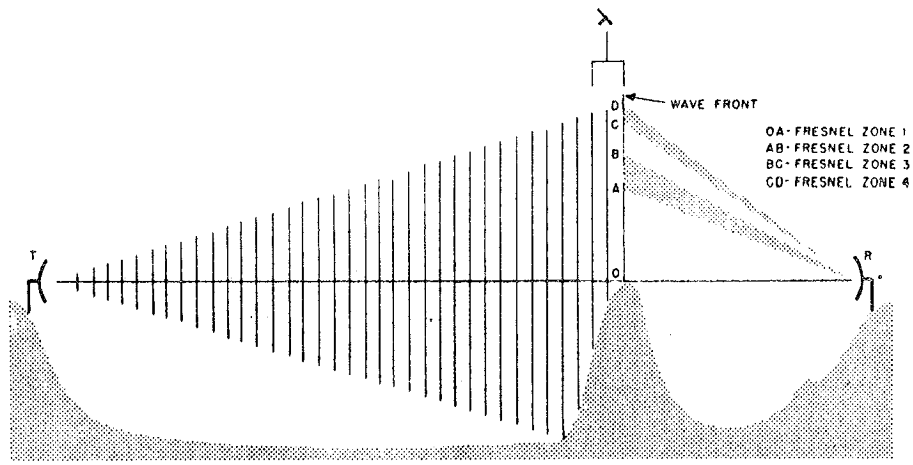


Figure 7-12. Fresnel Zone Destructive Interference. (Courtesy GTE Lenkurt)

This is because the higher-numbered zones are not allowed to reach the antenna and cancel part of the energy (figure 7-13).

What does all this mean? What do we want in a communications system? In general, first Fresnel zone clearance is considered to be very desirable, but clearance of 0.6 of the first Fresnel zone causes essentially no loss above the free space loss. This is the basis for defining an LOS path as one that has a minimum of 0.6 first Fresnel zone clearance. Of course, clearance greater than first Fresnel zone is also adequate, provided no reflection problems exist. Paths without adequate clearance are not desirable because atmospheric refraction may change and cause a "blackout" condition. If a signal just clears an obstacle under normal conditions, a decrease in refraction may cause the path to be obstructed. Remember that a decrease in atmospheric refraction is also known as "earth bulge."

If the earth bulges from the bottom of the first Fresnel zone to the center of the zone, for example, a 6 dB loss will occur in the received signal; therefore, it is important when setting up a new link to consider the average and worst conditions that may occur and locate the antennas at heights which will provide as much signal strength as possible. Although diffraction is an LOS phenomenon, short tropo paths may experience the same effects at the end terminals. The horizon may actually change as the K factor decreases.

The atmospheric changes cause the loss factor in RSL to vary. There are a number of factors that add together to form the path loss, the atmospheric changes determining which factors and their magnitude. So far, we have discussed only losses due to reflective terrain and Fresnel zone blockage (diffraction). Generally speaking, anything that the radio wave travels through, other than a vacuum, will slow the wave down and at the same time absorb some of its power. The property of the medium through which the radio wave passes is called its basic path attenuation.

(5) Basic path loss. Free space is the region that is free of all objects that might reflect or obstruct the radio wave. Under free space conditions, the radio wave spreads out according to the inverse square of the distance, as in optics. The free space loss can be determined from the following formula:

$$\text{Free Space Attenuation} = 22 \text{ dB} + 20 \text{ Log } \frac{D}{\lambda}$$

where D = distance between antennas

λ = transmit wavelength

(D and λ must be in the same units)

Determination of free space loss, of itself, provides little useful information. A number of "path loss" equations exist which account for additional losses imposed by operating in a real world environment. Comparisons of the various methods are contained in Panter's book, *Communications Systems Design.* In addition, allowance for special situation factors, not contained in the basic path loss formulas, must be included when applicable.

(6) Atmospheric Attenuation. The attenuation in the atmosphere, under normal conditions, consists of contributions by the oxygen and water vapor in the air. Selective absorption, due to water vapor and oxygen in the atmosphere, first occurs at 24 GHz and 60 GHz, respectively, and is, hence, of less consequence at 2-10 GHz. Table 7-1 shows the combined atmospheric attenuation of oxygen and water vapor by path length and frequency.

(7) Rain Attenuation. Rain will attenuate a radio signal to varying degrees, depending on many factors, including: frequency of the wave, distance the wave travels in the rain, the size and shape of the raindrops, and density (or heaviness) of the rainfall. The relationship of these factors is quite difficult to determine and much work in this field is still needed. Figure 7-13 shows average attenuation by frequency versus rainfall rate. The problem in figuring rain attenuation is compounded by the fact that areas of rainfall are a mixture of light precipitation and intense cells of heavy rain. Lenkurt Demodulation, Vol II, Oct 69, covers this subject with much additional information.

(8) Fog Attenuation. We know even less about the effects of fog than we do about rain. Generally speaking, fog has less attenuation than rain. Figure 7-14 shows the relative attenuation of both rain and fog. Fog is essentially the same as rain, except that it has smaller droplets. Its primary disruptive effect on

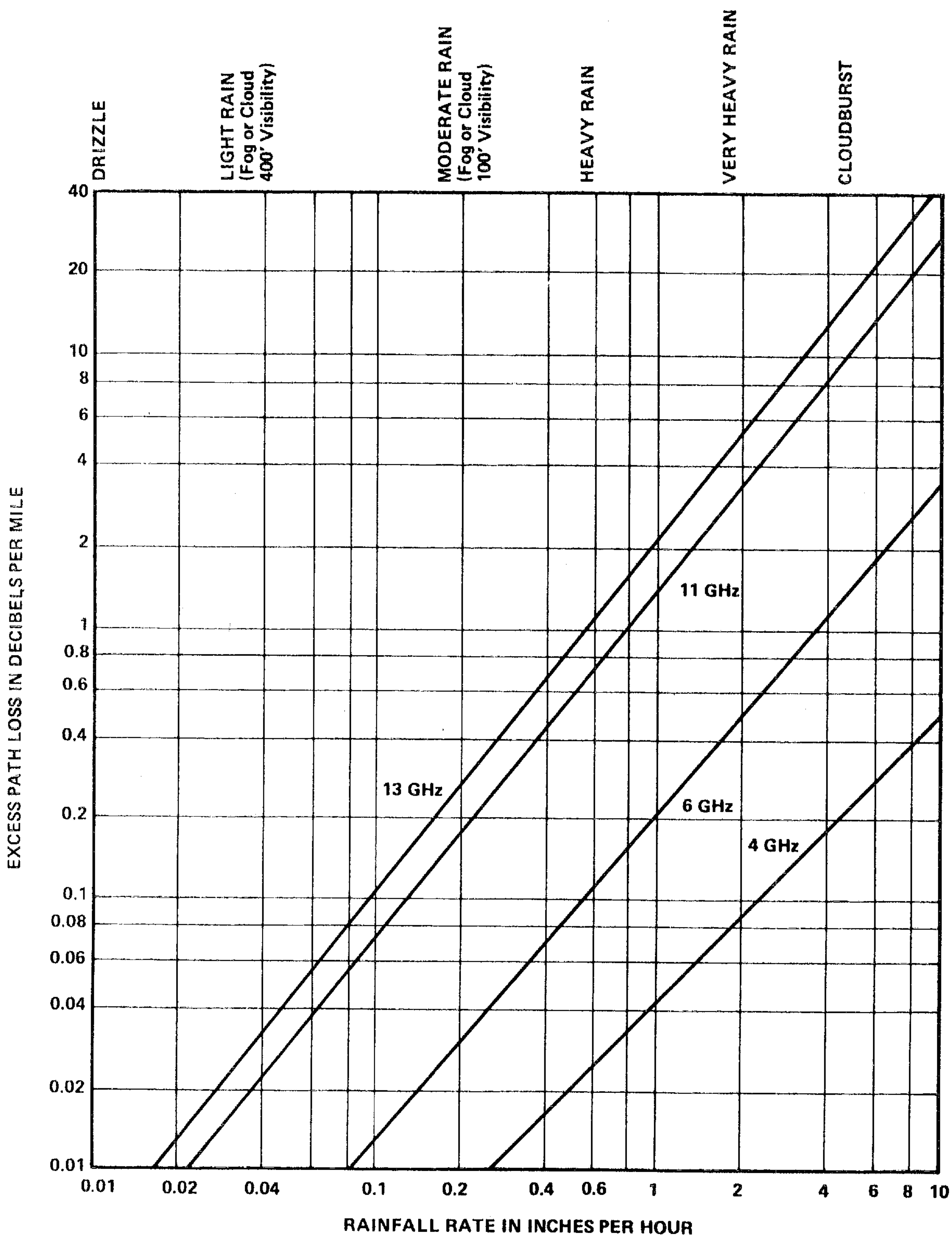


Figure 7-13. Rain Attenuation Versus Rainfall Rate. (Courtesy GTE Lenkurt)

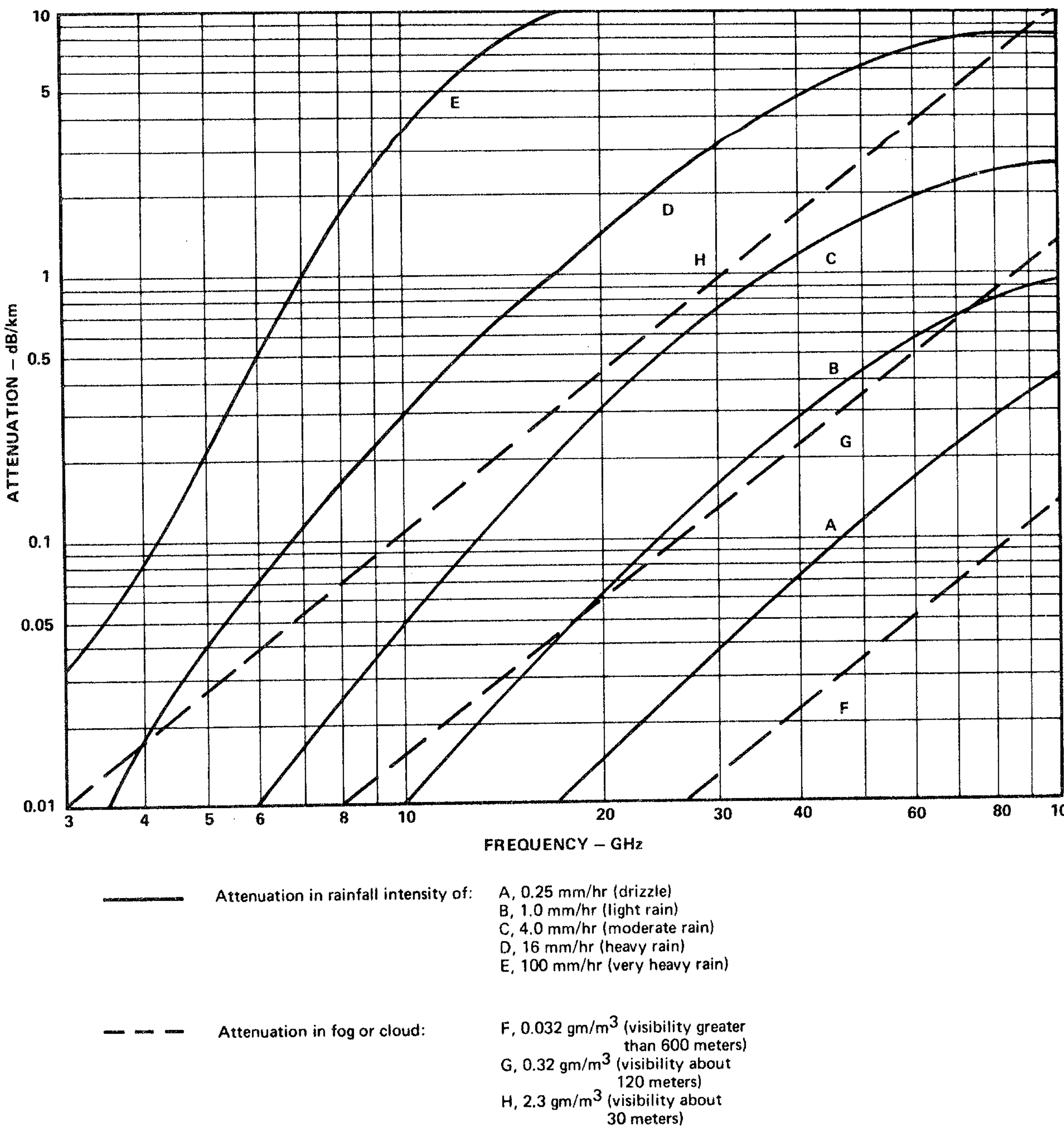


Figure 7-14. Rain and Fog Attenuation. (Courtesy GTE Lenkurt)

ATTENUATION IN dB					
Path Length Miles	*2-6 GHz*	*8 GHz*	*10 GHz*	*12 GHz*	*14 GHz*
20	.20	.26	.32	.38	.48
40	.40	.52	.64	.76	.96
60	.60	.78	.96	1.14	1.44
80	.80	1.04	1.28	1.52	1.92
100	1.00	1.30	1.60	1.90	2.40

Table 7-1. Atmospheric Attenuation in dB.

communications lies in the fact that fog is normally accompanied by still air and temperature inversions which can cause refraction of the waves. Fog causes another problem as it occurs close to the ground and can form a very good source of reflections to waves from antennas located on higher ground.

(9) Snow and Hail Attenuation. All that can be said of snow and hail is that they have less attenuation than rain, due to a lower moisture content. No data exists for the attenuation of these elements. The attenuation of snow depends on its area of coverage, flake size, and moisture content. Hail, as ice, has a reduced tendency to cause refraction, compared to water, and absorbs less energy.

7-3. **Fading of the Received Signal.** There are two general types of fading associated with radio wave propagation: attenuation fading and multipath fading. Attenuation fading is usually caused by variations of the K value with time. Normally, attenuation fading is the result of the K value decreasing (more earth bulge) causing the clearance for the radio wave to be reduced

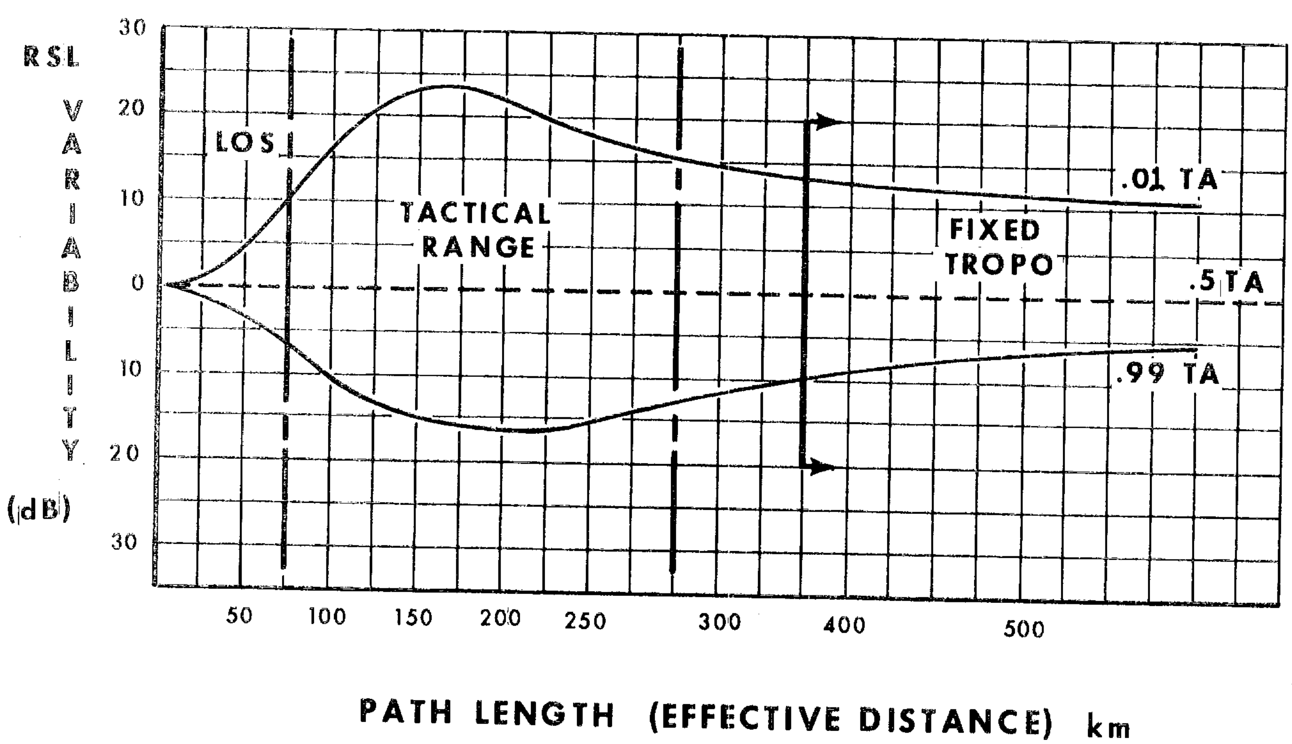


Figure 7-15. Variability of Long-Term Fade.

and introducing obstruction-type losses. Attenuation fading can also be caused by signals being trapped in a "duct." When both the transmitter and receiver are within the duct, the signal strength can be increased. On a path where both the transmitter and receiver are outside the duct, the signal strength will be reduced. Attenuation fading is often called seasonal or long-term fade. The time factor associated with them can be hours or days. Since the phenomenon affects all radio paths and all frequencies essentially the same, diversity techniques discussed earlier do not provide protection against this type of fade. NBS Tech Note 101 provides a probability distribution of the magnitude of this type fade versus an artificial path distance factor that takes into account end configurations. Figure 7-15 was derived from empirical data and approximates the variability of a 2-10 GHz path over terrain similar to the midwestern US. The 99% time availability line (predicted over a yearly distribution) shows maximum variability near 100 miles. LOS and fixed tropo system paths experience relatively low variability, but tactical tropo systems must operate in the most vulnerable range.

a. Multipath Fading. Whenever a transmitted signal travels two or more separate paths and both (or all) of these signals arrive at the receiving antenna, we are experiencing multipath transmission. These signals combine at the receiver producing a RSL which can be significantly different from any single component. The multiple paths can be caused by refraction, reflection, or both.

At one time or another, we have all seen the effects of multipath transmissions or fading on our home TV receiver. The "flutter" of the picture as an aircraft flies between the TV station antenna and our home is caused by receiving two different signals at our TV set, causing rapid variations in the RSL and disturbing the picture quality (figure 7-16).

The same type of thing can occur on wideband links. If two paths occur and they differ in length by one-half wavelength, or odd multiple thereof, the two signals arrive 180° out of phase and almost total cancellation will occur. Conversely, if the paths differ by one wavelength or multiple, the two signals will be in phase and give a total RSL up to 6 dB higher than the normal, direct path. Physically, these paths need not be much different in length since one wavelength at 4000 MHz (a commonly used band) is only three inches.

Multipath fading on M/W (LOS) links is usually a short-term phenomenon, identifiable by short, deep fades occurring rather infrequently.

On troposcatter links, multipath fading is more the rule than the exception, with constantly varying RSLs. Even so, particularly severe multipath fading does occur with more rapid and deeper fading occurring at those times.

The effects of multipath vary depending on the type of system involved. For analog M/W LOS systems, the path length differences are not usually long enough to cause any noticeable distortion of the baseband signal. A typical 600 channel baseband has a spectrum of 60-2660 kHz.

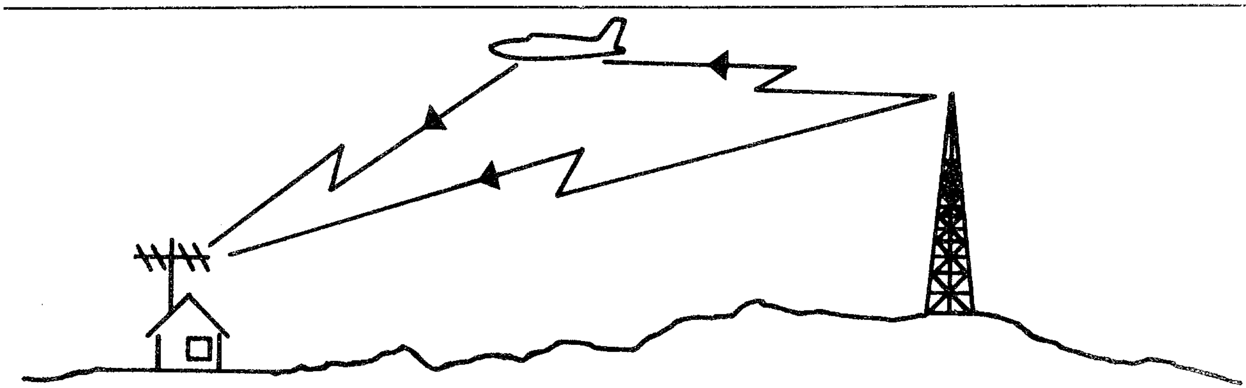


Figure 7-16. Multipath Fading.

The wavelength of a 2660 kHz signal is 370 feet. If the modulated RF signal experienced a path length difference of 6 feet, the resulting phase shift of the delayed signal, when demodulated, is 5.84°. This amount of phase shift equates to 6.1 nanoseconds delay and may or may not be significant, depending on the system application. For this analog system, it is probably negligible. For a digital system, however, this delay may be significant. For a system with a 12.35 megabit per second bit rate, the bit time is 81 nanoseconds. A delay of 6.1 nanoseconds will probably not disturb the signal; however, if the delay was about 20 nanoseconds, significant intersymbol interference may occur so as to cause system degradation. A general rule of thumb for digital systems is that if the delay is 25% or more of a bit time, it will probably degrade the system and should be corrected. Stated another way, the delay should be kept less than 25% of a bit time to minimize intersymbol interference from this source.

In contrast, very long troposcatter systems can experience significant baseband distortion because of the large number of paths involved and the wide variation in path lengths. The resulting distortion is known as "path intermodulation." In addition, the rapidly varying RSL will cause similar variations in receiver thermal and idle noise, as explained in chapter 6.

The remedy for fading is proper path engineering. Excess signal level, above the minimum required for communication, provides a quantitative measure of protection from atmospheric changes. The excess signal is labeled fade margin and the more margin available, the better.

b. Fade Margin. Some multipath fades reduce the received signal strength by only a few dB, but deep fades may cause a drop of 40 dB. In order to assure that enough signal reaches the receiving antenna, enough extra signal strength must be available during "normal" propagation to compensate for most fades. This extra signal strength is expressed in terms of dB and is called the fade margin. Fixed systems are engineered to provide more than 30 dB fade margin in LOS paths and usually more than 40 dB on tropo. Tactical systems are routinely pushed to the limit on distance and so have little fade margin. Predicting successful paths requires significant tactical engineering experience. Signal levels can normally be predicted to within 6 dB on most links. Tactical systems experience more variability and are, therefore, more difficult to accurately define. The difference between the receive signal and minimum usable signal is predicated on the amount of ICN that can be tolerated by subscriber equipment.

7-4. Modes of Propagation. Radio paths in the wideband systems we are addressing can be lumped into two categories: LOS (both antennas can "see" each other) in which the transmit and receive antennas are within radio horizon and tropo, where horizon points can be identified between antennas. There is a no-man's land between the two that is treated as a special case. It's called diffraction and is characterized as a very dynamic area. LOS distances of 20-75 miles are possible with M/W sites, a tropospheric scatter site can communicate over distances up to about 500 miles. Tactical systems normally operate between 30 and 170 miles. Often a clear distinction cannot be made in the tactical world as to which mode of operation is dominant - tropo, LOS, or diffraction. Tactical shots sometimes experience mode changes during operation due to propagation characteristics.

a. LOS. LOS paths are those in which the transmit and receive antenna can see each other with a minimum of 0.6 first Fresnel zone clearance. LOS paths experience free space loss and multipath fading. The other factors are usually so small that they can be ignored; however, LOS calculations are not appropriate if the K factor change results in a path with less than 0.6 first zone clearance.

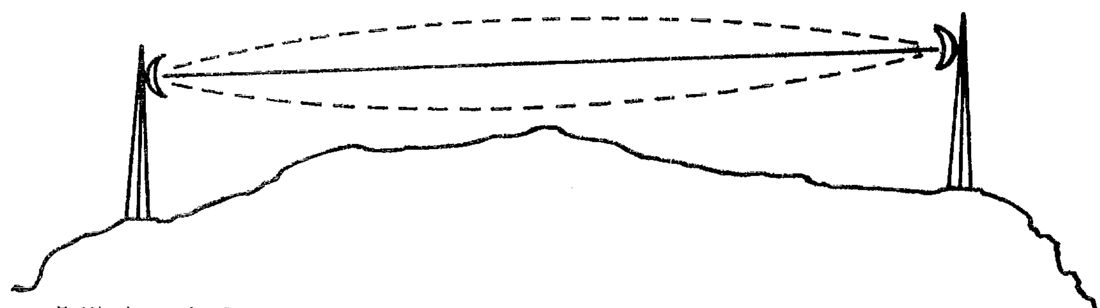


Figure 7-17. A typical LOS Path.

An LOS radio path is usually designed for a direct line between the transmitting and receiving antennas; however, this direct path is not always possible because a hill or other object may be in the way. One of the ways such an obstacle can be overcome is by a phenomenon known as "knife-edge" diffraction. This is the ability of a radio wave to be bent slightly over the edge of a sharp obstacle. The important factors here are sharpness and height. A sharper edge causes less attenuation to be placed on the signal. The obstacle must also be the proper height to allow this effect. Generally speaking, neither terminal should be closer than 2 miles for every 1000 feet of obstacle height.

Another way to deflect a signal is by using "passive" reflectors or back-to-back parabolic antennas to intercept the signal and direct it toward the destination. Examples in the use of "knife-edge" diffraction and passive reflectors are shown in figures 7-18 and 7-19. An analysis of "knife-edge" and rounded obstacle diffraction is presented in NBS Tech Note 101, vol I, chapter 7. It provides the mathematics for estimating loss factors. Lenkurt Demodulator, vol I, page 313, Jul 63, has an excellent discussion on passive reflectors. Special emphasis should be placed on page 321, line 28, where efficiency of such a system is related to distances. Passive reflectors are not very efficient but their usefulness can be improved by placing them near one of the terminals.

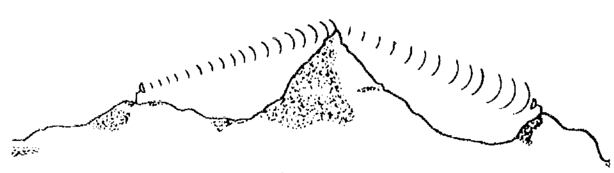


Figure 7-18. Knife-Edge Effect. Radio wave being diffracted by a sharp object.

b. Tropo. Troposcatter paths have definable horizon points and a common volume for transmit and receive antennas (figure 7-20). They have very high losses and, therefore, require high-powered transmitters and sensitive receivers.

There is no completely satisfactory explanation for tropospheric radio communications far beyond the horizon. Philip F. Panter, in his book, *Communication Systems Design,* gives an excellent overview of the three major theories expounded today: blob, turbulent layer, and layer-reflections. Each theory has its proponents and a number of troposcatter path loss equations have been developed. They weigh factors differently but all contain the same basic factors. Using the same set of data, they all agree within acceptable limits. Panter also reviews these methods and their limitations.

All the available predictive methods begin with a median RSL. From this, a total fade margin figure is computed. An adjustment for long-term fade, dependent on desired time availability (TA) factor, is subtracted from the total fade margin and the remainder used to determine relative security from multipath fading. Tactical systems have much less fade margin and, therefore, do not provide usable reliability figures using the above method. Current emphasis is on predicting the TA factor for tactical paths and giving a subjective appraisal of its relative merit.

Emphasis on tropo prediction is stronger in tactical communications than their fixed world counterparts. Experience is the best teacher and it is not unusual for a tactical communicator to engineer a dozen or more paths a year. Interpretation of the availability of communications is not a casual exercise; it requires knowledge of the assumptions used to develop the formulas and ingenuity in applying them to a particular case.

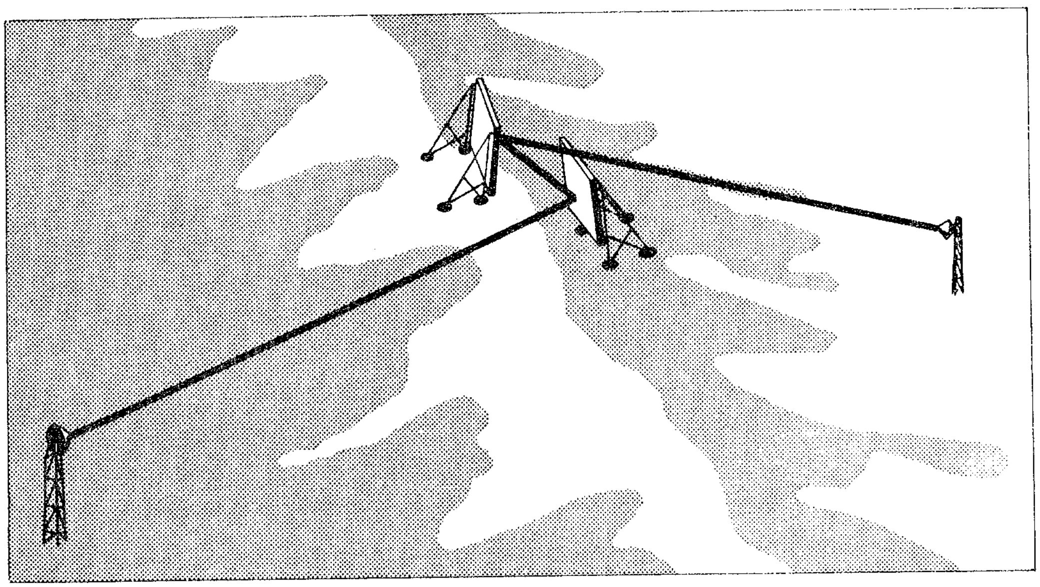


Figure 7-19. Passive Reflections. (Courtesy GTE Lenkurt)

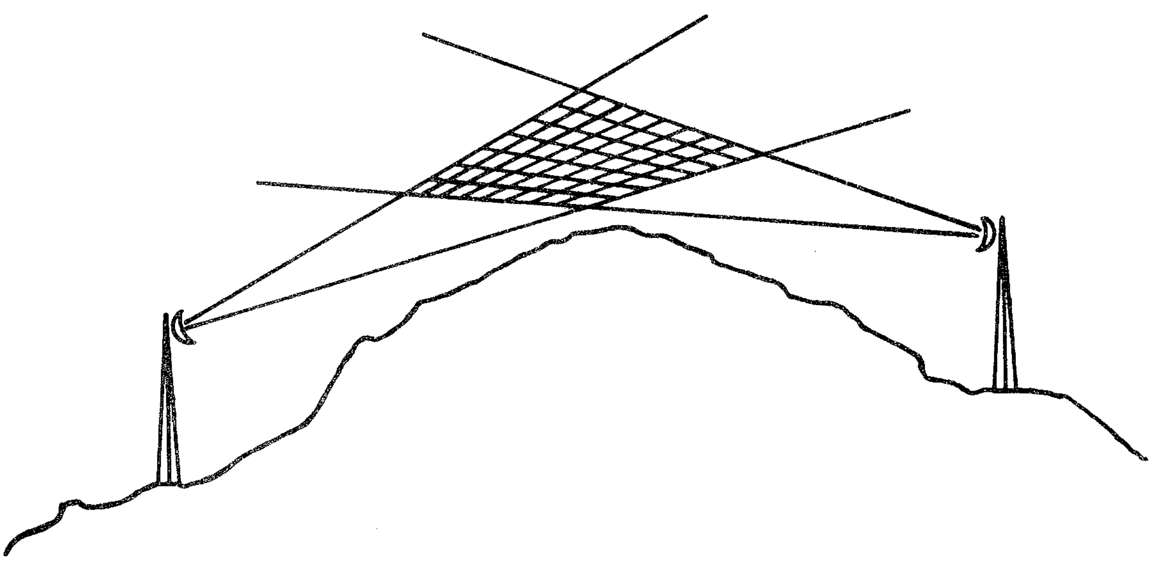


Figure 7-20. A Typical Tropo Path.

Chapter 8

RELIABILITY/PROBABILITY

8-1. General. In this chapter, we will discuss some of the concepts of probability and how it affects communications reliability; however, we will do our best to steer clear of the complex mathematics that is so easy to get mired in when this subject is pursued. Although it is not necessary to understand the math, it is essential that the language be spoken if we are going to understand the concepts; therefore, let's start out by defining some of the more commonly used terms in the trade.

8-2. Probability Terms:

a. Probability. This refers to the likelihood of a specific event occurring during a test or trial. For example, if the test is the flip of a coin, there are two possible events or outcomes: heads or tails. The probability of obtaining a head is 1/2 as is the probability of obtaining a tail. Notice that the sum of these two probabilities is one because they represent all possible outcomes for flipping a coin (assuming the edge is not an end state). This is true for any trial: the sum of the probabilities of all possible events will equal one and the probability of any specific event is between zero and one.

b. Probability Distribution. If we plot the probability of events for a test, we have a histogram, which shows the probability for all of the events which can occur. For example, in our simple case of flipping coins, the probability distribution is as shown in figure 8-1A.

Similarly, if we roll a six-sided die, the probability that any one side will come up is 1/6 or .167. The probability distribution for this is shown in figure 8-1B. If we complicate the situation just a little and roll two dice at a time, the possible results are shown in table 8-1.

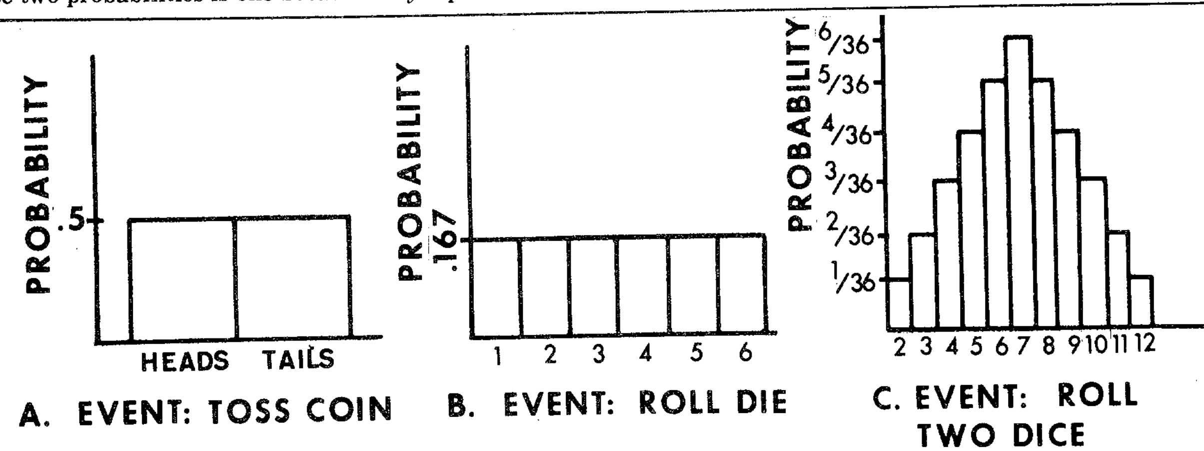


Figure 8-1. Probability Distributions.

NUMBER ON SECOND DIE \ NUMBER ON FIRST DIE	1	2	3	4	5	6
1	2	3	4	5	6	7
2	3	4	5	6	7	8
3	4	5	6	7	8	9
4	5	6	7	8	9	10
5	6	7	8	9	10	11
6	7	8	9	10	11	12

Table 8-1. Possible Results With Roll of Two Dice.

Notice there are 36 possible combinations, with results ranging from 2 to 12. There is only one combination that gives 2 and one that gives 12; there are six combinations that give 7. All other results fall between these two extremes. Figure 8-1C shows the probability distribution for this event.

So far, in all the examples we have looked at, there has been a discrete number of results. We toss a coin and we have two possibilities - heads or tails. We roll a die and there are six possibilities - 1 through 6. We roll two dice and there are eleven possibilities - 2 through 12. But what if the event is measuring the ICN on a tropo system? In this case, instead of being *discrete,* the variable is *continuous.* No matter how close two readings are, there is always room for another between them. One of the most common probability curves is the normal distribution. This is the familiar bell curve (figure 8-2). Many events (such as animal size, noise in communications systems, and number of words printed on a given size page) occur with the normal distribution. Although the normal distribution is the most common, it is by no means the only one. Figure 8-3 shows examples of other distributions. There is really no limit to the number of possible distributions. As a matter of interest, a man named Lord Rayleigh showed that the random phase cancellations that cause multipath fading occur in a predictable manner. The probability distribution he developed to describe this phenomenon now bears his name.

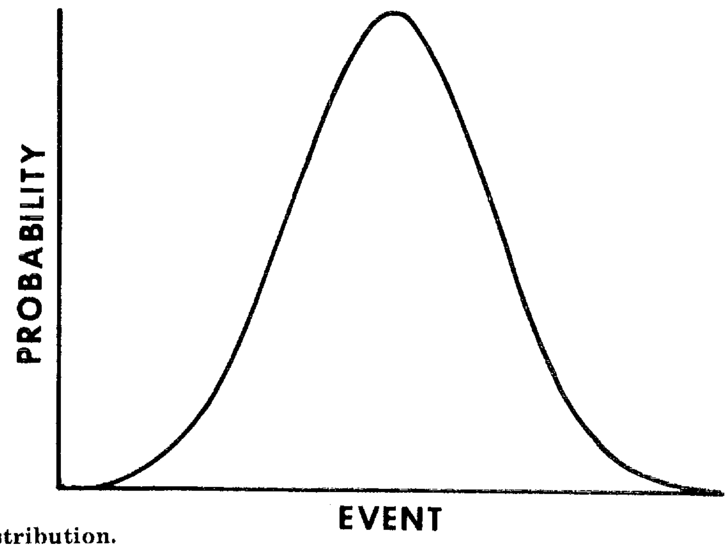


Figure 8-2. Normal Distribution.

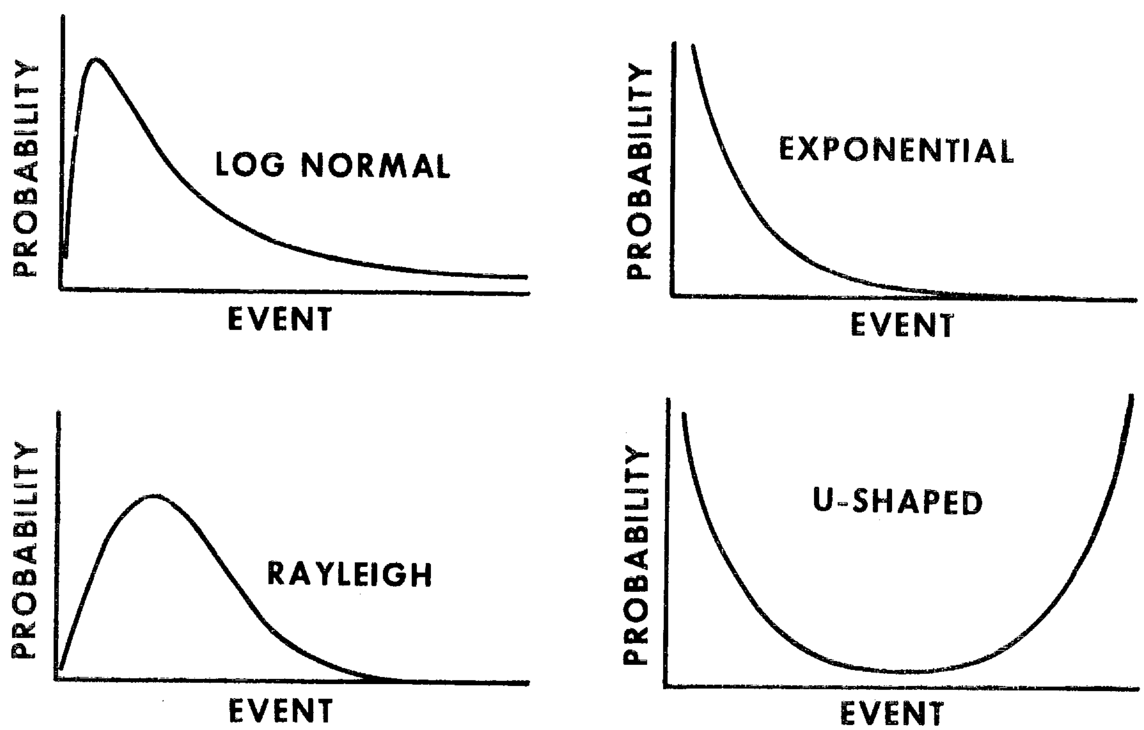


Figure 8-3. Examples of Different Probability Distributions.

c. Mean. This is the arithmetic average of a set of data. For example, if the voltage is measured on five power supplies as -56, -60, -62, -55, and -57 volts, then the mean (m) is:

$$m = \frac{(-56 + (-60) + (-62) + (-55) + (-57)}{5} = -58 \text{ volts}$$

d. Median. This is the point in a set of data at which half of the readings fall above and half fall below. In the above example, if we order the points (-55, -56, -57, -60, -62), we see that -57 volts is the median because -55 and -56 are on one side of it while -60 and -62 are on the other.

e. Mode. This is the event of a probability distribution which has the greatest probability of occurring. In the normal distribution, the mean, the median, and the mode all occur at this peak.

f. Standard Deviation. This is a parameter of a probability distribution that gives a measure of its spread or scatter. The standard deviation for a set of data would be small if all readings were tightly grouped and becomes larger if the data is spread out. Figure 8-4 shows the relationship of normal distributions with different standard deviations. Approximately 68% of the readings will be within $\pm$ 1 standard deviation of the mean, approximately 95% will be within $\pm$ 2 standard deviations, and 99.73% will be within $\pm$ 3 standard deviations for a normal distribution.

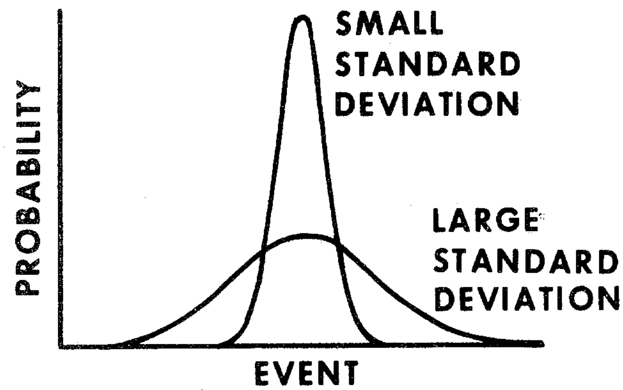


Figure 8-4. Relationship of Normal Curves With Different Standard Deviations.

g. Regression Analysis. This is an analytical process used to show the relationship between numerical variables. ICN, RSL, and baseband loading (BBL) are all numerical variables and can be examined by regression analysis. Regression analysis can also be used to show changes in these variables over time (Time Series Analysis). Because it has a sound mathematical basis, regression analysis can be used both as a measurement and as an estimation process.

To visualize the simple regression process (two variables), we first list out the data as shown below:

Month	*Mean of ICN Readings*
1(Jan)	-55.1
2(Feb)	-57.3
3(Mar)	-55.9
4(Apr)	-58.1
5(May)	-55.7
6(Jun)	-54.5
7(Jul)	-57.2
8(Aug)	-60.1
9(Sep)	-58.5
10(Oct)	-58.7
11(Nov)	-57.4
12(Dec)	-57.6

In examining the change in the variable of interest over time - in this case, the ICN readings - we can prepare a scatter diagram to visualize the rest of the process. A typical scatter diagram is shown in figure 8-5, with the points connected for clarity only.

A straight regression line is computed for the points to give the best fit possible. This line is computed using what is called the method of least squares. The mathematics of this computation are simple and straightforward, but it is very tedious with large amounts of data unless a computer is used. The procedures and formulas for doing this can be found in any good book on statistics.

Confidence intervals can be mathematically established for the regression line to make statements about predicting individual values of ICN or about comparing the regression line with some standard. Figure 8-6 shows the regression line and the confidence intervals for the data. The regression line is solid; the broken lines show the confidence interval and include all of the data points. Any points outside the confidence intervals would indicate a significant departure from the normal data dispersion.

A systems analyst can draw these lines and say "I have XX percent confidence that the data points between these lines are valid." The size of the "XX" depends on the wishes of the analyst and the number he/she plugs into his/her formulas.

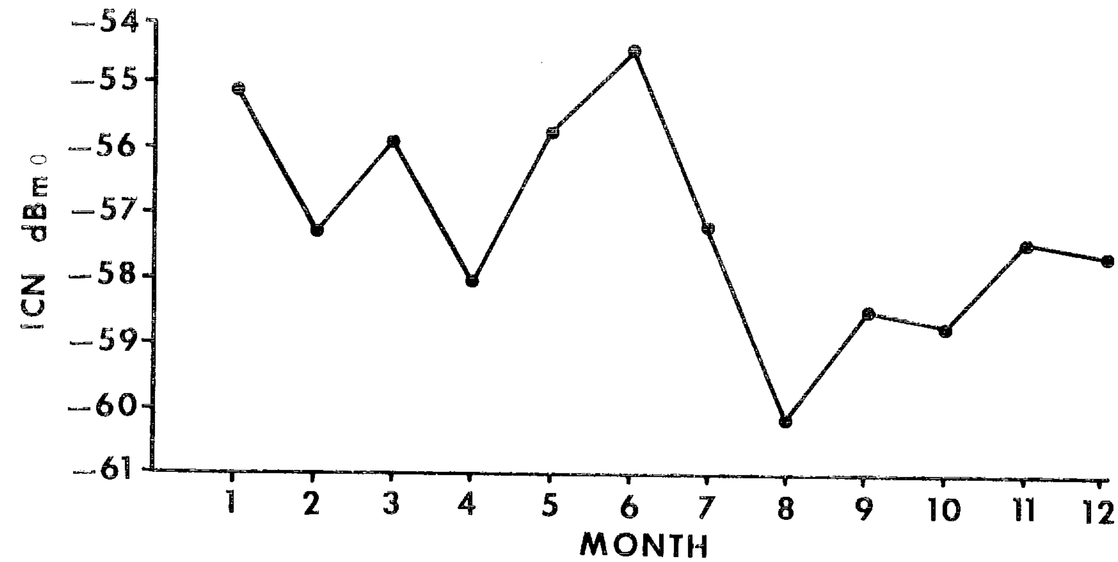


Figure 8-5. Scatter Diagram.

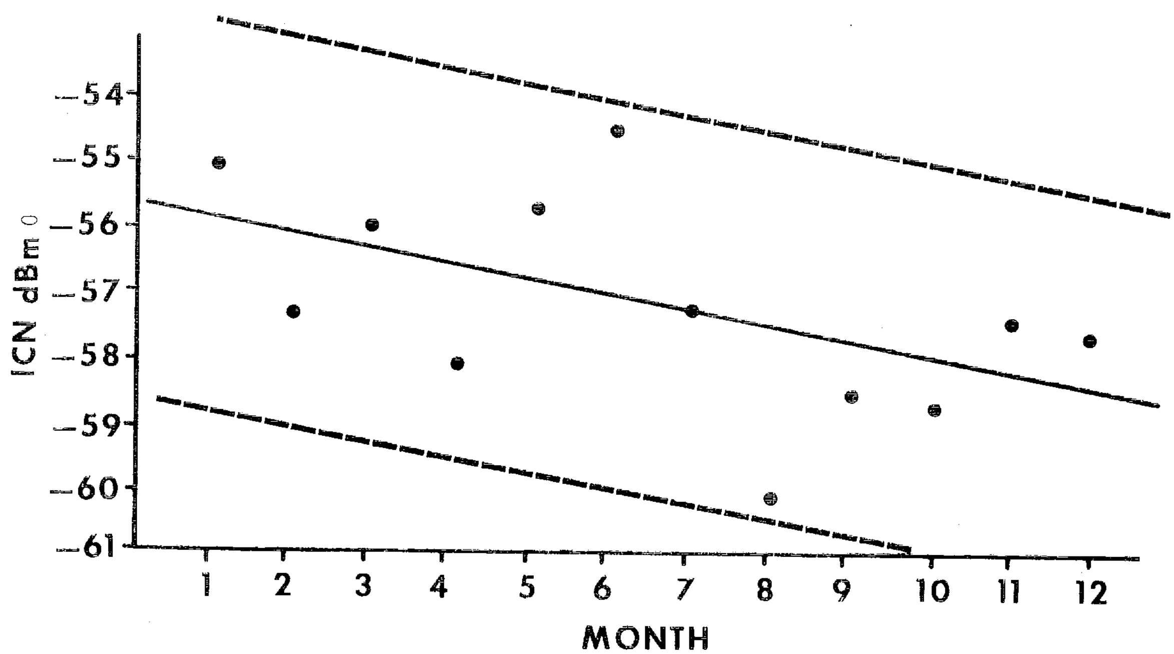


Figure 8-6. Confidence Intervals.

The real value of regression analysis is to be able to detect a trend. The trend that is not readily apparent in figure 8-5 becomes easy to see in figure 8-6.

8-3. Probability in Communications. Probability theory is used in communications engineering to help assure the required reliability is provided. To determine reliability, we need to look at two different areas. First, we need to know the probability that the RF energy will be propagated through the atmosphere. Second, we need to know the probability of a hardware failure. Then we need to know how to combine all these factors to estimate the reliability.

As we mentioned earlier, the probability distribution developed by Lord Rayleigh can be used to predict multipath fading. This distribution can be plotted to show "Period of time ordinate level is exceeded" vs "Relative signal level in dB" (figure 8-7).

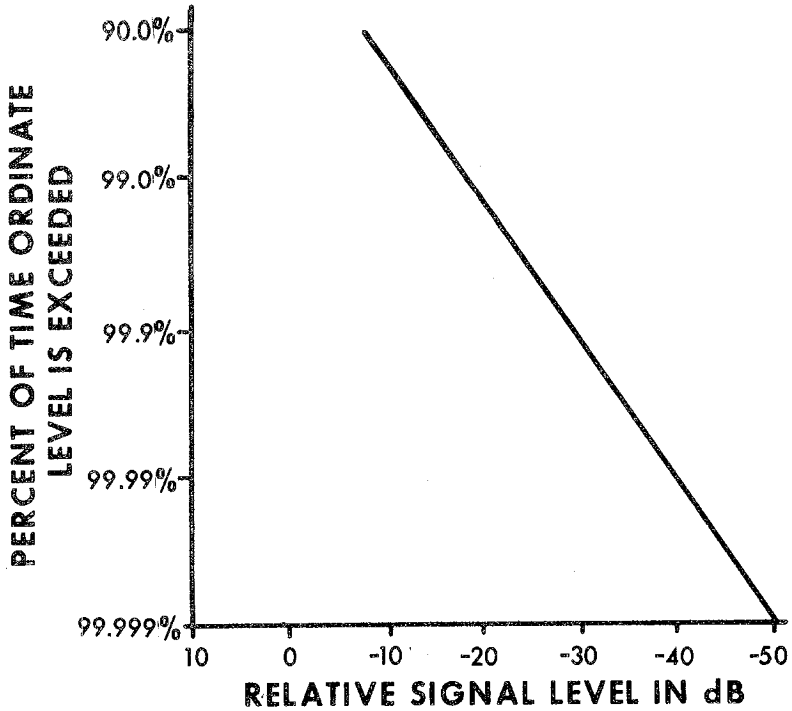


Figure 8-7. Rayleigh Distribution.

This curve shows the probability of M/W fades of various levels using no diversity techniques. For example, the received signal will stay within 35 dB of normal about 99.98% of the time. Or to put it another way, fades of 35 dB or more from the median signal occur only .02% of the time (in a year, this would equal 105 minutes of transmission interruption). The curve can also be used to determine the approximate fade margin required for a prescribed reliability level. For example, for 99.99% reliability, a fade margin of 38.5 dB is needed. (Note that this distribution holds true only for multipath fading.) Further, this is not a precise prediction, only an estimate. We speak of the fade margin as the dB difference between the receiver FM improvement threshold and the median or mean receiver signal input level. Propagation reliability is, then, the percentage of time that the RSL is above FM improvement threshold.

The probability of hardware failure is usually estimated from historical data. When the characteristics of each component in a system are known, then the probability of successful operation of the complete system can be estimated. This is done using laws of series and parallel probability theory.

a. Series Probability. The probability of reliable operation decreases whenever more components are added in series to the signal flow. For example, reliability is reduced when more LOS links are added or more amplifier stages are inserted. The more components in series with a signal, the more components there are that can go wrong. If all components of a system are considered to be equally reliable and are in series (the failure of any component results in system failure), the overall system reliability (R) can be stated as:

$$R = r^n \qquad \text{Eqn 8-1}$$

Where r = reliability of each component
n = number of components

Let us see how this works with some sample values of "n" and "r" in table 8-2.

At this point, it is interesting to note that, even though a system may have ten components, each with 99% reliability, the overall system reliability is only 90.4%. This is true because each of the possible component failures may occur at different times and, as stated earlier, any one outage can disable the entire system.

Eqn 8-1 holds true when all elements of a system have the same reliability. Actually, this rarely occurs, but we are fortunate in that the formula can be modified to fit real-life situations, as follows:

$$R = (r \times r_2 \times r_3 \times r_4 \ldots . r_n) \qquad \text{Eqn 8-2}$$

Simply stated, the formula shows that the overall reliability is the product of all the individual reliabilities. Consider the examples shown in figure 8-8.

n (Number of Components)	*r (Reliability of Components)*	*R (Total Reliability)*
100	.99	.366
500	.99	.0066
1000	.99	4.3 X 10^{-5}
100	.99.9	.904
500	.99.9	.606
1000	.99.9	.368
100	.99.99	.990
500	.99.99	.951
1000	.99.99	.905

Table 8-2. Component Reliability.

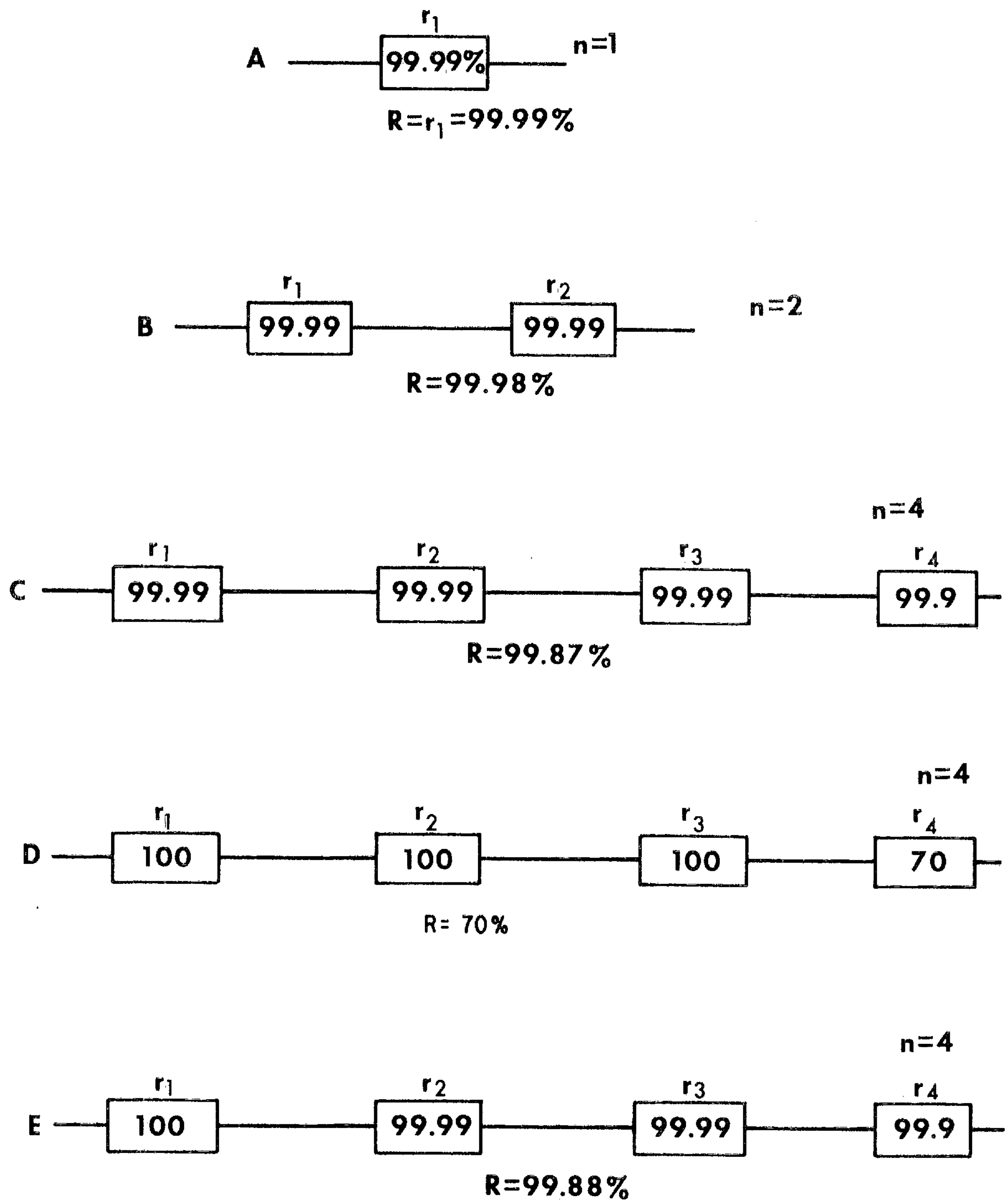


Figure 8-8. Series Reliability. Note that it takes only one bad link, as in D to ruin the whole system's reliability.

As we have shown, a chain is no stronger than its weakest link but, in this case, the chain itself can be weaker than the weakest link. The implications are clear. If, in the last example, r_1 and r_4 are terminal stations with r_2 and r_3 as repeaters, the overall reliability is actually less than the lowest individual reliability. In this case, r_4, by having less than the goal of 99.99% (99.90 to be specific) actually gives the system 99.88% reliability. This example illustrates why a system cannot provide service as expected even when the individual stations meet or exceed the requirement. For this reason, a site commander must not be concerned only with meeting or exceeding his/her own site's reliability, but must also be aware of every other site's performance if the system is to perform adequately.

b. Parallel Probability. Now that we have learned to deal with a series of individual reliabilities, let's consider parallel reliability. This involves the principle of redundancy, which simply means that an alternate path is provided in case the primary path fails. The communicator sees redundancy every day, in operational/standby equipment, diversity transmitters and receivers, and alternate path routings. A parallel redundant circuit will always improve the overall circuit reliability and it is this aspect that distinguishes it from the series circuit. Consider the simple parallel circuit in figure 8-9. The overall reliability (R) of such a circuit is given by the formula:

$$R = r_1 + r_2 - (r_1 r_2) \qquad \text{Eqn 8-3}$$

Some sample values of r_1 and r_2 with the resultant (R) are shown in table 8-3.

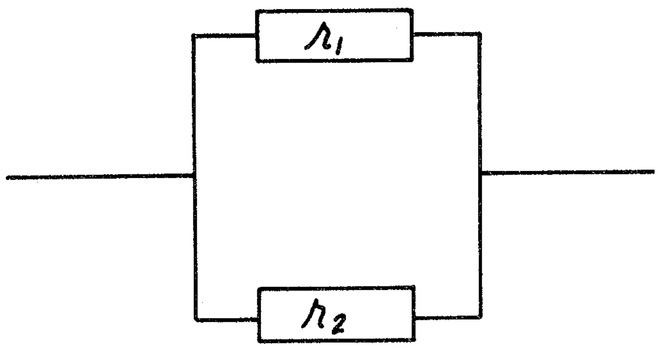


Figure 8-9. Simple Parallel Circuit.

Note in the last case that, even though each half of the circuit achieved only 99%, together they could achieve 99.99%. Now it would seem that all we have to do to achieve our goal of 99.99% is to put a lot of lower quality components in parallel. If we ignore the obvious economic impossibility, we might think this is true; however, the law of diminishing returns sets in when we use parallel circuits. Consider the simple triple-parallel circuit in figure 8-10.

$$R_T = r_1 (1-r_2) + r_2 (1-r_3) + r_3 (1-r_1) + r_1 r_2 r_3 \qquad \text{Eqn 8-4}$$

Now let us show some typical values in table 8-4.

r_1 (%)	r_2 (%)	R (%)
50	50	75.00
50	75	87.50
75	75	93.75
90	75	97.50
90	90	99.00
95	90	99.50
95	95	99.75
99	95	99.95
99	99	99.99

Table 8-3. Reliability for Two Systems in Parallel.

r_1 (%)	r_2 (%)	r_3 (%)	R_T (%)
90	90	90	99.90
95	95	95	99.99
99	99	99	99.9999

Table 8-4. **Reliability for Three Systems in Parallel.**

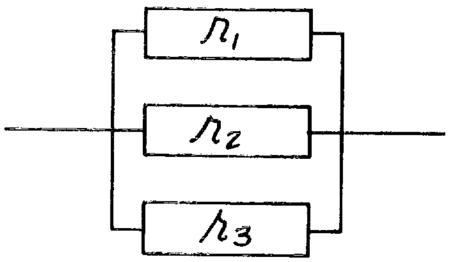


Figure 8-10. **Triple Parallel Circuit.**

While the values in table 8-4 make it look very easy to achieve 99.99% in circuit reliability, bear in mind that this same goal can be achieved with a 33% reduction in equipment by a 4% increase in station reliability from 95% to 99%, using a dual-parallel circuit.

c. Series - Parallel Probability. In real life, the problem is one of analyzing a complex series - parallel reliability. As we have already mentioned, the same circuit analysis technique used for electronic circuit analysis can be used here. Series circuits are first combined into an equivalent and then parallel circuits are combined one by one and replaced by successive equivalents. Let us consider one practical example. This system will consist of four parallel circuits with each alternate having its own series circuit (figure 8-11). The detailed math will not be shown here. Suffice it to say the system can be simplified, in turn (figure 8-12).

8-4. **Summary.** As we stated earlier, the application of the concepts of mathematical probability theory in analyzing probable circuit and component reliability can be very useful. The foregoing text has illustrated this application through examples applied to specific phases of communications.

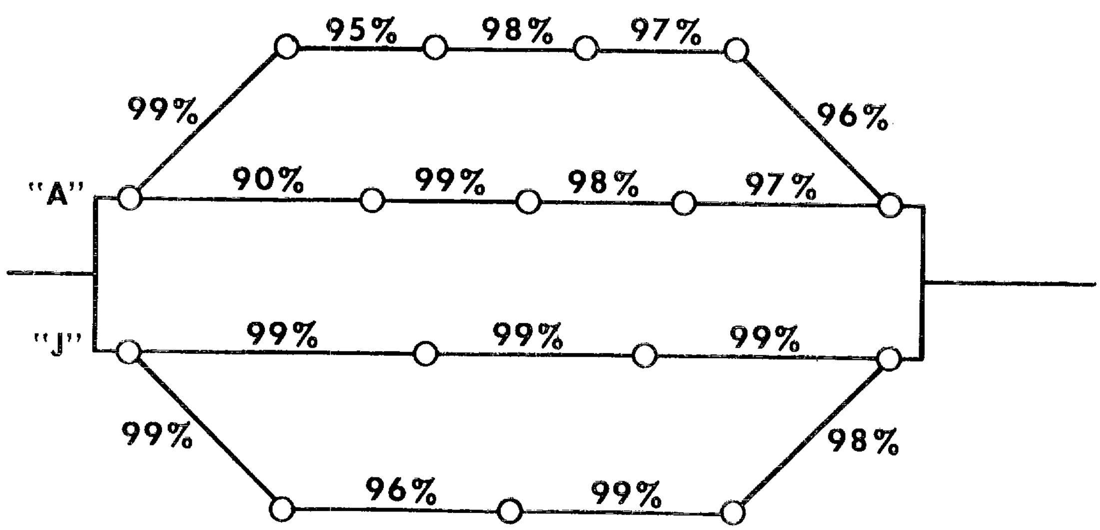


Figure 8-11. **Complex Series Parallel Reliability.**

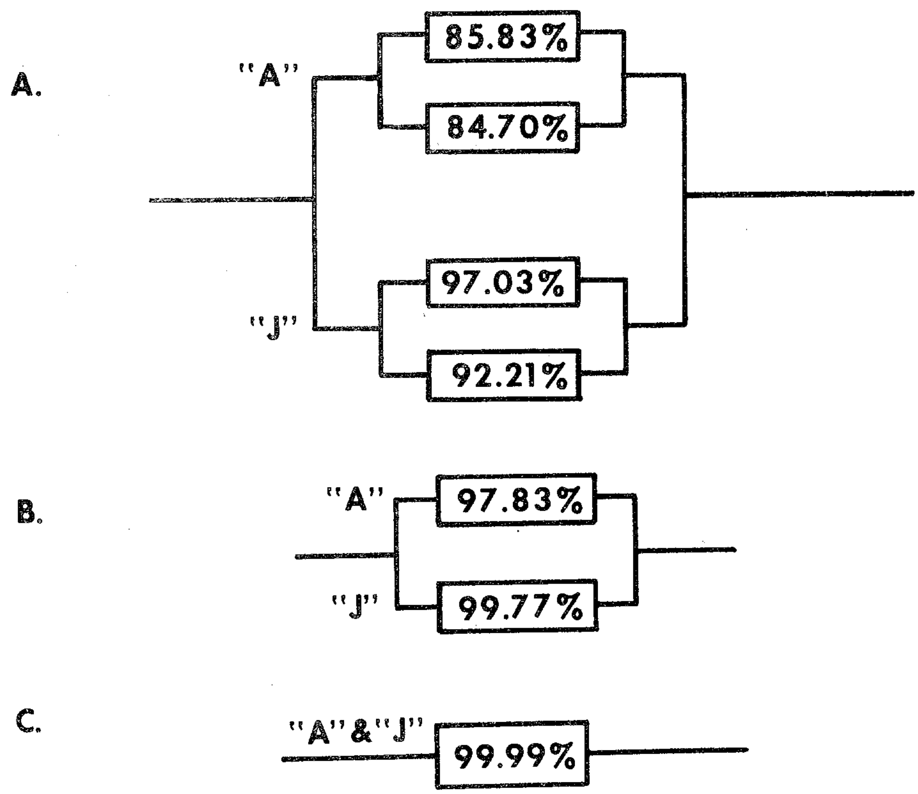


Figure 8-12. Simplification of Complex Reliability. The overall reliability of figure 8-11 is found by making the intermediate solutions shown in A and then B. The result shown in C is the reliability of figure 8-11.

Chapter 9

MULTIPLEX COMPONENTS

9-1. General. Early in the development stages of telephone communications, a search was made for a means of transmitting more than one telephone conversation simultaneously over a single pair of wires. It was recognized that telephone service would be limited to the number of lines in service and that the requests for service were far in excess of the number of lines available. This search for a more efficient way to transmit multiple signals led to the process of multiplexing. We have since progressed from early open-wire systems, carrying a few telephone channels, to coaxial cable and wideband radio systems capable of hundreds, or even thousands, of voice frequency channels. This chapter will explain multiplexing methods used in our wideband systems.

9-2. Multiplexing. This is the means by which a number of circuits are combined to form a composite signal suitable for transmission over a broadband medium. To transmit a number of signals simultaneously over the same medium, the signals must be sufficiently separated so that they will not interfere with each other. These signals can be separated in time or frequency. The first method of separation is called Time Division - a form of multiplexing in which each signal occupies the entire bandwidth but only for small portions of time. This type of multiplexing is often used in data circuits.

The second method of separation is called Frequency Division - a form of multiplexing in which a small portion of the total bandwidth (typically 4 kHz) is preassigned on a full-time basis to a particular message channel.

9-3. Time Division Multiplexing (TDM). This is a direct and basically simple multiplexing method. In time division, the information is transmitted intermittently instead of continuously as in frequency division. Continuous transmission is unnecessary providing the conditions of the sampling theorem are met. Simply stated, the sampling theorem says: "If a message that is a function of time (f(t)) is sampled instantaneously at regular intervals and at a rate at least twice the highest significant message frequency, then the sample contains all of the information of the original message." With this method, the transmitting circuits, using a common medium, need not be connected together as they would be in frequency division multiplexing; instead, they are arranged so that each transmitting circuit is successively connected, in turn, to the common transmission medium for short periods of time (figure 9-1).

In TDM, signals are separated in time by briefly sampling each channel in regular sequence. A signal distributor "at the receive" is necessary to differentiate the sampled pulses as they arrive in sequence and to route them to the correct lines at the proper time. This requires synchronization between the transmitter and the receiver. Loss of synchronization causes all channels to garble and the information transmitted to be lost.

A commutator, or other type of switching device, can be used to successively connect each channel to the common transmission medium for short intervals; however, in most systems today, electronic switching and gating techniques are used. Regardless of the switching method used, it can be seen that each input signal to the transmitter will be broken up into a series of pulses called samples. These samples represent a portion of the entire input signal. The length of the sample is not critical, since it has been found that time duration can be reduced as much as required without downgrading the information being transmitted. The only critical consideration is the rate of sampling, as was discussed earlier.

9-4. TDM Scheme. The use of TDM of signals in digital form allows interconnection of several digital signals to form higher signaling rates and, consequently, more channels onto one line. Figure 9-2 shows a plan for TDM of 192 Pulse Code Modulation (PCM) channels into a common line at a 12.6 megabit per second rate (Mb/s).

a. Channel Modulation. The interface between the analog world and the digital world is made by the PCM channel bank. In this bank, the incoming signal is passed through a low pass filter which limits the signal to a bandwidth of less than one-half of the sampling frequency. The signal is sampled at a rate of 8000 times per second. Samples from 24 channels are applied to a common line. Since the sampler operates sequentially, the voltage on the common line is a time division multiplexed version of all the pulse amplitude modulation (PAM) samples (figure 9-3). The voltage on the common line is processed by the coder which is shared by all the channels to produce the PCM words. Before this signal can be applied to a transmission facility, additional processing is necessary. First, signaling information is inserted to aid the receiver in locating a block of 24 PCM channels. A typical channel bank format is shown in figure 9-4. In the channel bank, each channel is coded into an eight-bit binary word. Every sixth frame, one bit (the least significant) of the digital word from each channel is used to transmit signaling; also, a framing bit is added which results in 193 bits per sampling frame. Calculations below show make-up of a frame:

8 bits/channel X (24 channels) + 1 framing bit = 193 bits/frame

Since there are 8000 frames per second, the required bit rate of the T1 line is 1.544 Mb/s. The receiver performs the inverse of the transmit section. It also performs a search for the framing bit. From knowledge of the framing bit, the signaling bits are identified with individual channels and the PCM code word is converted to quantized PAM samples. The PAM samples are then converted to an analog voltage that is proportional to the transmitted message.

b. Higher Level Time Division. The next level in the PCM hierarchy combines eight T1 pulse streams where they are time division multiplexed to a 12.6 Mb/s

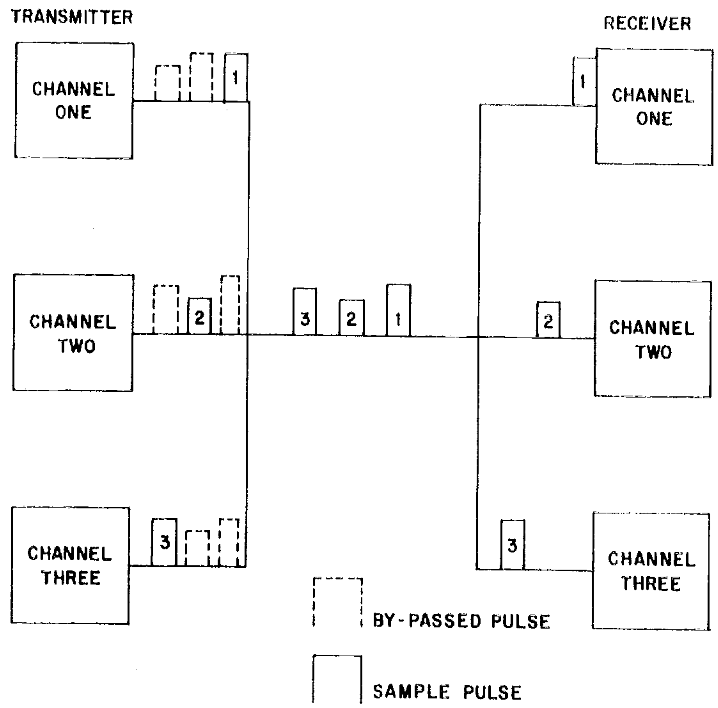


Figure 9-1. Time Division Multiplexing.

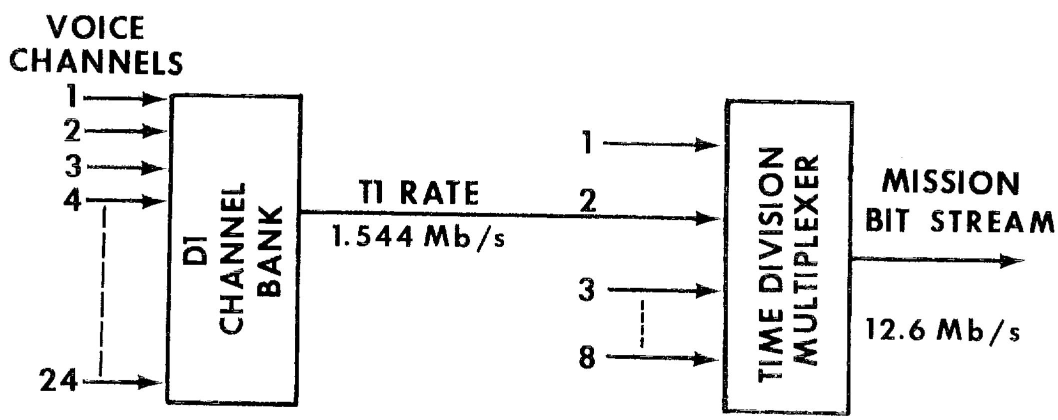


Figure 9-2. Time Division Multiplexing of 192 PCM Signals.

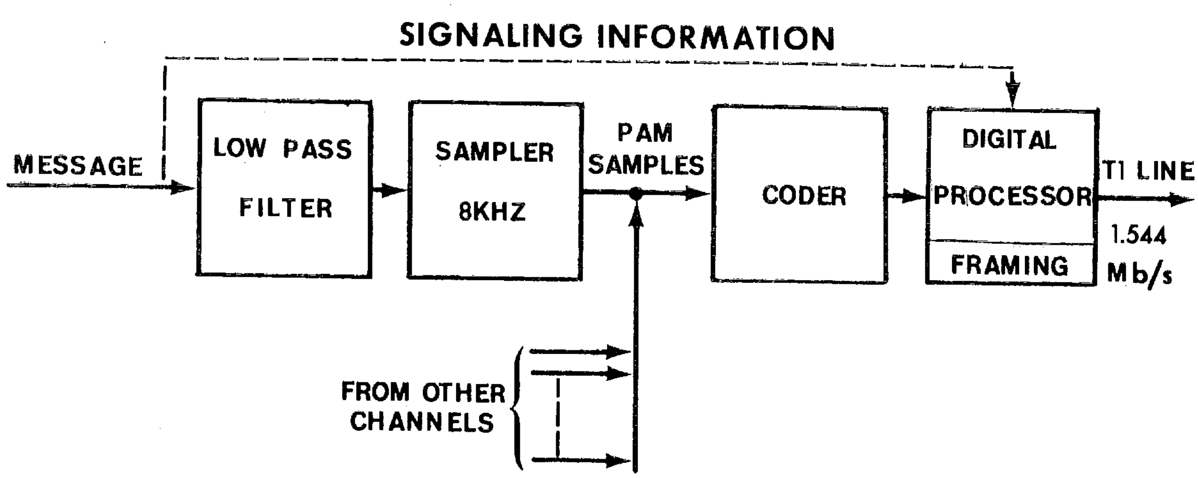


Figure 9-3. Signal Samples Applied to Common Line.

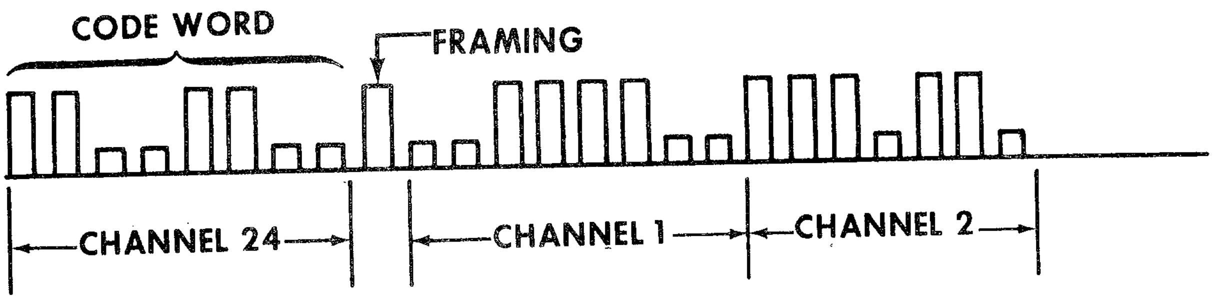


Figure 9-4. Example, PCM Channel Bank.

rate. This is slightly higher than eight times the T1 rate which allows for adding the necessary framing and stuffing bits. This gives a maximum capacity of 192, 4 kHz channels per 12.6 Mb/s (mission bit stream). These mission bit streams could be combined to form larger systems.

9-5. Frequency Division Multiplexing (FDM). In frequency division, or carrier systems as it is sometimes called, each multiplex circuit is preassigned a specific channel frequency for transmission on a full-time basis. This is accomplished by modulating the carrier frequency with the voice channel frequency. In effect, the VF channel is translated to a higher frequency spectrum by the modulated carrier frequency. A simple illustration of FDM is shown in figure 9-5. A large number of VF channels can be translated by this method and all channels can be transmitted simultaneously.

a. Frequency Allocation. Three modulation steps are commonly employed to obtain 600 voice channel capability. In the first step, 12 channels are combined in one base group. In the second step, five base groups are translated to a supergroup. The last step combines ten supergroups into a composite baseband signal containing 600 channels for transmission. The baseband frequency allocation for a 600-channel system is from 60 to 2540 kHz. This system is shown in figure 9-6.

b. Carrier/Multiplex System Translation. When a carrier frequency is used to shift the frequency of incoming signals to preassigned positions in the frequency band, we say that they are translated. The multiplexing of translated signals is accomplished by adding all frequencies together to form a composite signal. A common carrier modulation plan for 600 VF channels will now be explained to show the principles of the translation process (figure 9-6).

In order to spread the 600 channels across the baseband or line frequency spectrum, three stages of modulation are required. In the first stage, each VF input signal modulates one of 12 channel carriers spaced 4 kHz apart. The lower sidebands are selected to provide a 60 to 108 kHz 12-channel group. In the second stage, five 12-channel groups each modulate a separate group carrier to produce a 60-channel supergroup with a frequency spectrum of 312-552 kHz. Ten of these supergroups are needed to form one 600-channel mastergroup. In the final stage of modulation, nine of the ten supergroups each modulate a separate supergroup carrier, resulting in line frequencies ranging from 60 to 2540 kHz. One of the supergroups (supergroup 2) is applied directly to the line at the 312-552 kHz frequency level.

c. Synchronization. Synchronization is needed to keep the carrier frequencies at the local and distant terminals the same. This is accomplished in various ways, but the most common method is a master/slave technique in which the slave terminal's oscillators are phase-locked to a pilot frequency, which is derived from the frequency generator at the master terminal. Any changes in frequency at the master terminal are followed, or tracked, by the slave terminal. Tracking is accomplished by sending an "error" voltage, which is derived from a comparison of the two frequencies to the slave oscillator. This error voltage then corrects the slave oscillator to make it match the frequency transmitted from the master terminal.

d. Multiplex Terminal Equipment. Previously, the three stages of modulation required to make translations for 600 channels of operation were discussed. In the next section, a description of the equipment used in the translation process will be presented. There are six basic units used - three for the modulation process and three for the demodulation process (figure 9-7). A simplified block diagram of the entire multiplexer is shown in figure 9-8.

9-6. Frequency Division Modulation. Most modulators used in commercial systems are balanced suppressed carrier types. The channel units provide the amplification and modulation which is required for the incoming voice signals. Base and supergroup units are designed to provide a linear response over the required bandwidth.

a. Channel Modulation. Each channel needs a channel modem - one for each end of the system. The modem contains both the modulator and the demodulator. When a channel carrier is modulated by the incoming audio signal, several signals are produced. A channel filter is used to select the lower sideband, while the upper sideband and carrier leakage are suppressed. The output of the filter is series-tuned and tied to a combiner network. In the combining network, all 12 channel outputs are combined into a composite signal, which is then applied to the base group modulator.

b. Base Group Modulator. This modulator operates in the same manner as the channel modulator; however, it translates an entire group to a higher position in the frequency spectrum. There are five types of base group modulators, but they differ only in operating frequency. The input signal to the base group modulator consists of the 12 channel outputs of the channel modems. The composite signal is amplified and modulated in a balanced modulator. The lower sideband is selected by an LC network, further amplified, then combined with other base groups to form a 60-channel supergroup.

c. Supergroup Modulator. Five base groups are then combined to form a single supergroup. In an identical manner, the individual supergroup signals are translated by the supergroup modulators to their position in the master group spectrum. This composite signal is amplified and transmitted as a unit. The receiver terminal reverses the order of modulation and repeats the same steps the transmitter accomplished. In the receiver, the incoming signal is demodulated down to supergroups by the supergroup demodulator. The incoming signal is then reduced to groups by the group demodulators.

d. Supergroup Demodulator. Ten supergroup demodulators are needed to translate the incoming wideband signal. Since supergroup 2 requires no translation, this demodulator is only used to provide isolation and amplification for the base group demodulators. When the entire baseband signal is applied to the input of the supergroup demodulator, it is divided by filters into two groups, odd and even. Each filter selects a specific supergroup and this selected group is translated down the frequency spectrum to the 312-552 kHz basic supergroup output.

e. Base Group Demodulator. The basic supergroup output contains 60 channels within the five base groups. It is applied to the inputs of each base group demodulator. In each demodulator, there is a filter which selects a particular base group frequency band. When the selected signal is mixed with a carrier frequency, a 12-channel, 60-108 kHz base group is formed. The demodulated signal is filtered, amplified, and then applied to the inputs of the 12 channel demodulators.

f. Channel Demodulation. In the previous discussion of channel modulation, it was stated that both the modulator and demodulator may be contained in the same channel modem. These 12 channel modems differ only in filter and carrier frequency. Each channel modem contains a filter which selects one 4 kHz channel from the output of the base group demodulator and applies it to a balanced demodulator. The output of the demodulator is the same as the transmitted audio signal.

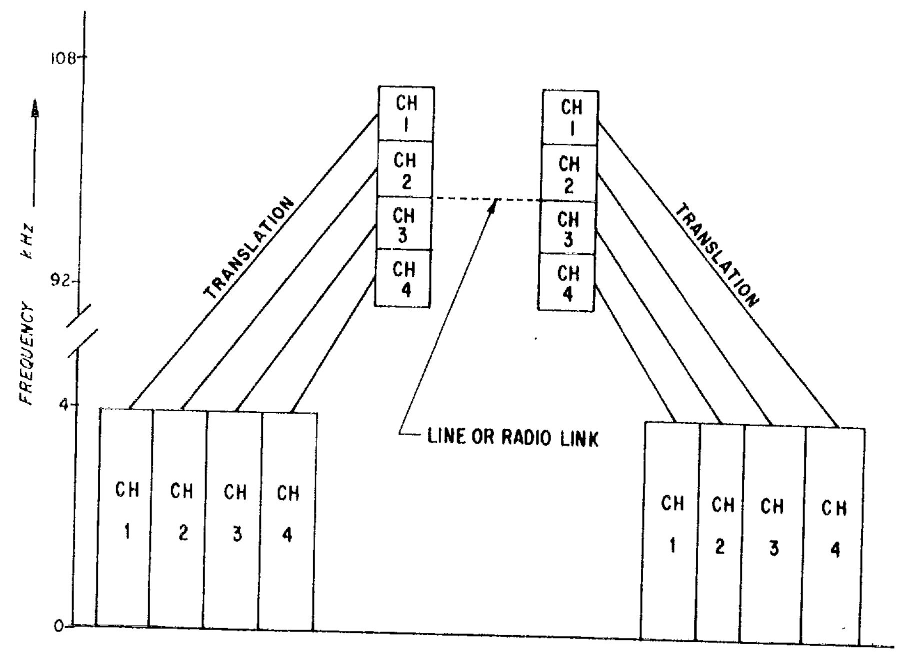


Figure 9-5. Frequency Division Multiplex Translation.

FREQUENCY ALLOCATION and MODULATION PLAN (CCIR)

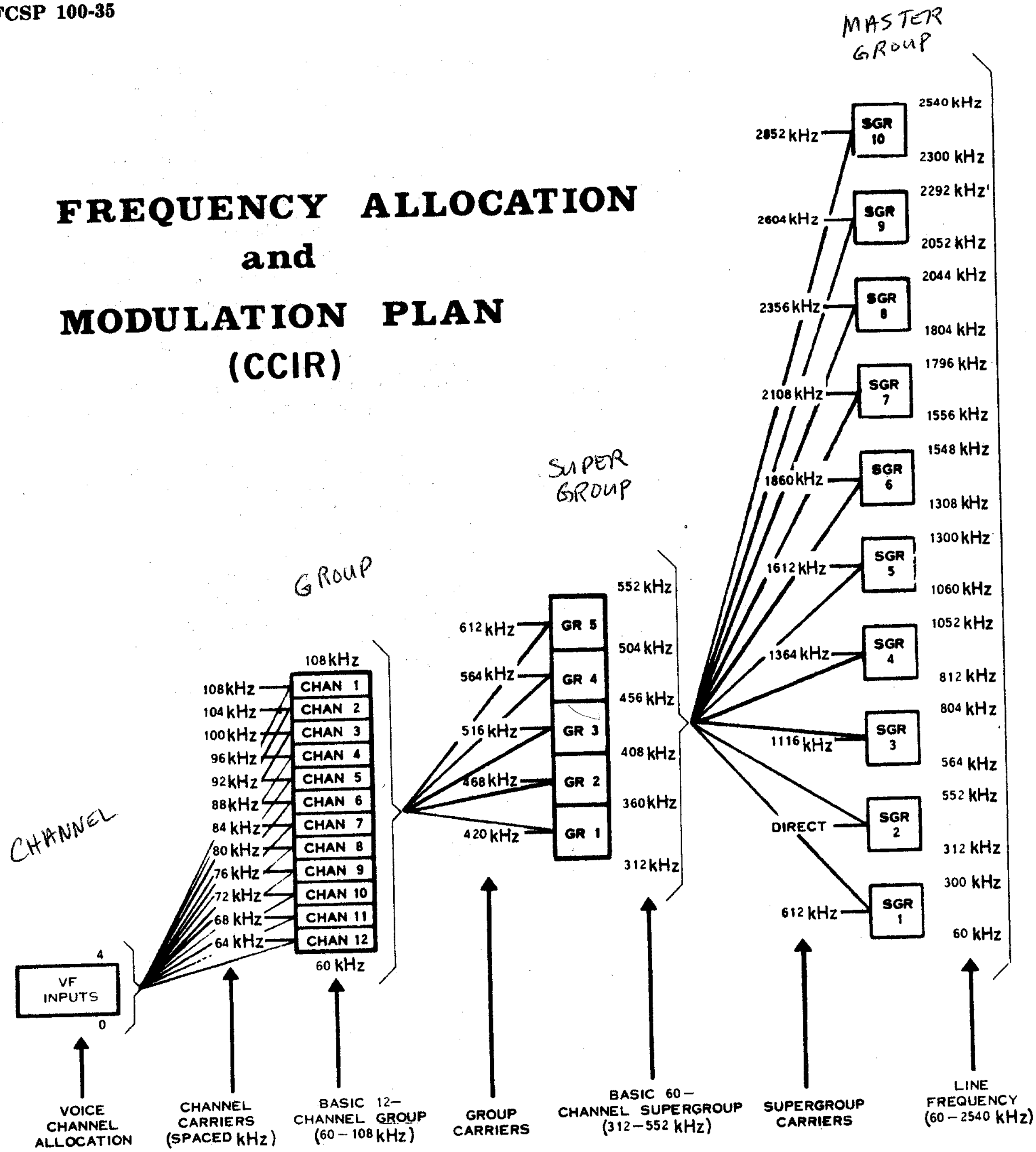


Figure 9-6. Frequency Allocation and Modulation Plan.

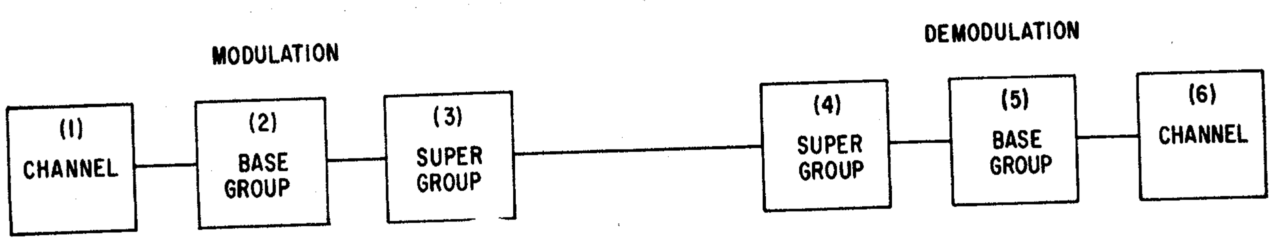


Figure 9-7. Multiplex Basic Units.

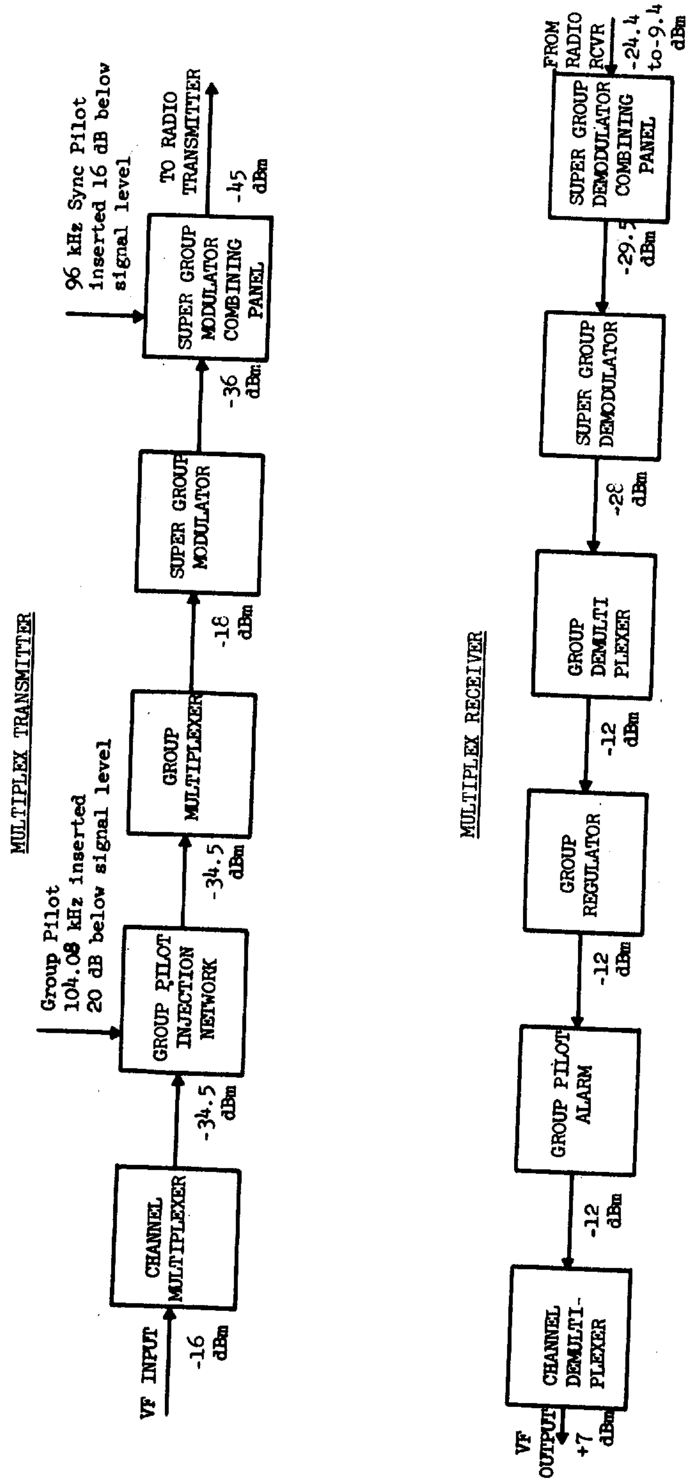


Figure 9-8. Simplified Transmit/Receive Signal Path.

9-7. Signal Characteristics:

a. Signal Input. Our attention is now turned to signal characteristics (for example, what the input signal is, the methods of signaling, the transmit signal path, and load capacity). To begin, consider a single-channel multiplexer. Figure 9-9 shows that the audio input signals are coupled through an input transformer (T_1) and pad to a balanced modulator. The carrier input is applied to the base of Q1 and Q2.

A balanced modulator must have both these inputs to have an output. In addition, the carrier signal is higher in amplitude and higher in frequency than the audio input signal. The modulator is designed to provide amplification as well as modulation and carrier suppression. Each channel appears in the group as the lower sideband of a suppressed carrier. The 12 channel carriers in the group are spaced at 4 kHz intervals from 60-108 kHz. The guard space between channels is used to prevent signal overlapping. Since not all of the 4 kHz band is used as guard space, this usually amounts to about 800 Hz (figure 9-10).

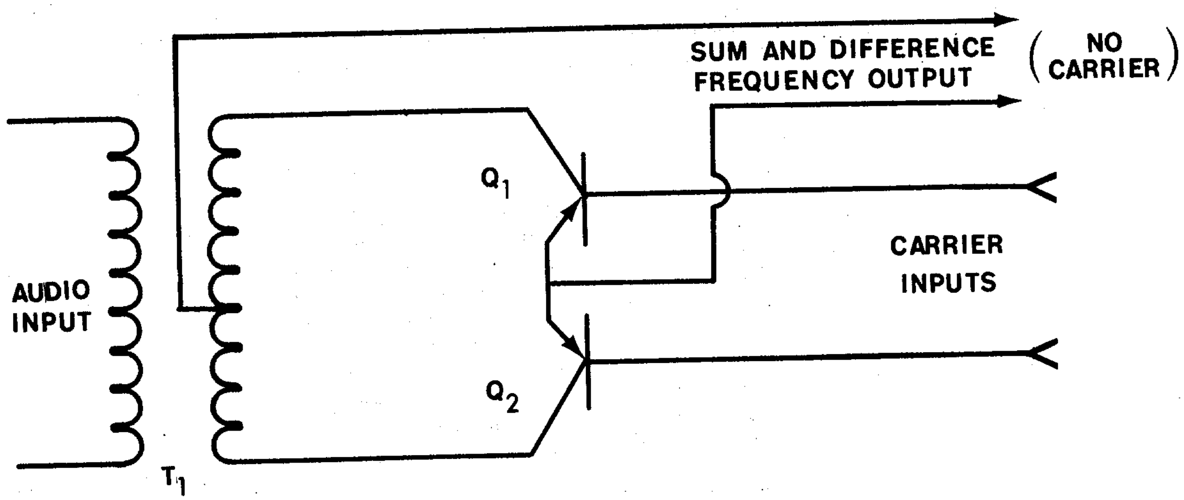


Figure 9-9. Typical Multiplex Balanced Modulator.

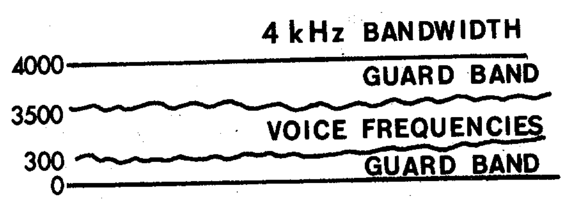


Figure 9-10. Voice Frequency Channel.

b. Methods of Signaling. In some cases, this 4 kHz channel bandwidth is used not only for voice, but also for signaling. In these cases, channel filters must be designed to prevent mutual interference between speech and supervisory signals which share the channel. In most carrier systems, one of three signaling methods is used: in-band, out-of-band, or separate channel.

For the most part, separate channel signaling is used for high density dedicated routes or in those special circumstances where signaling cannot be conveniently handled in the communications channel. This method is convenient, but uneconomical. The repair and maintenance required is also more complicated and reliability may be less, since both channels are subject to failure.

The remaining two methods of signaling are more commonly used. With out-of-band signaling, channel filters are designed with an upper cutoff frequency well below the ceiling of the channel. This filter leaves part of the spectrum available for signaling. Usually, a single tone is used and it is keyed to transmit signaling information. Besides being more flexible, out-of-band signaling can be easier and more economical to use. This is particularly true if some channel bandwidth can be sacrificed.

The in-band signaling method is being used more frequently than the other two methods. Its primary advantage is the extreme flexibility it provides. Signaling tones are transmitted at 1600, 2400, or 2600 within the speech band. The speech and supervisory signals share the same transmission media, but at different times. In-band signaling is arranged so that it either precedes or follows the speech signal, allowing the signaling and speech signals to share the same channels but at different times. Since the supervisory signal becomes part of the transmission, it is not necessary to use DC repeaters when going from one link to another. The lack of DC repeaters eliminates delay and pulse distortion found in out-of-band signals when they traverse many links.

The main objection to in-band signaling is that the signaling tones lie in the speech band. Under this circumstance, speech energy at the signaling frequency may be able to degrade the supervisory signal by a phenomenon called talkdown. Talkdown can cause two undesirable things to happen. First, it can cause false signals with voice energy to interfere with the conversation or it can cause signaling tones to become audible, which also interferes with speech conversation.

Talkdown can be prevented by adding a guard circuit, which is used to distinguish between speech and signaling tones. More protection can be provided by selecting the correct frequency for the in-band signaling tone. It is better to use a higher signal frequency since speech energy declines at higher frequencies; therefore, the possibility of talkdown is reduced.

c. Load Capacity in FDM Systems. Load capacity is defined as the volume of traffic a system can handle without undue distortion and noise. "Undue" is meant to describe the point at which interference seriously affects either the accuracy or the intelligibility of the transmitted information. The number of channels and the type of load determines the loading of a multiplex system.

It is commonly assumed that, if a system has 600 channels, all channels may be used simultaneously. If they were, the system would be severely overloaded; however, this is rarely the case, as systems are designed for maximum probability of use rather than maximum possible use. As a result, the speech systems are designed with characteristics of speech signals and statistical speech distribution in mind.

Allowances are made for signaling tones, pilot signals, carrier leak, and small amounts of teletype traffic. Data and teletype signals present a continuous load, whereas speech signals present a sporadic load. If the number of channels used for data and teletype exceeds design limitation, overloading occurs. Voice channels may have to be disconnected to relieve the overload, reducing system capability.

Transmission requirements of a multiplex system also affect load capacity. Certain requirements may establish the levels of the input signals to the multiplex terminal and, at the same time, establish the maximum permissible output noise level. When signals are applied at high levels or if noise requirements are unrealistically high, the load capacity of the multiplex equipment will be reduced. In the following section, a discussion of signal impairments will further describe the effects of overloading.

We have seen what the signal characteristics are for voice and data signals. Essentially, these characteristics decide the input to the multiplex. Through several discussions, we have taken these channel inputs, translated them in frequency, each to its "own" spot in the baseband, and we have added extra signals to provide level control (group pilots) frequency and level control (96 kHz sync pilot) and signaling or supervisory control by use of in-band and out-of-band tones. When these complex waveforms are added to form the AM signal at the baseband output, they present a signal which would appear as "hash" on an oscilloscope.

By using a FSV, it is easy to dial any single frequency and correlate that frequency, or narrowband of frequencies, with the voice channel from which it came. By sweeping over a range of baseband frequencies, it is possible to identify baseband components as to power level, separation, and duration, as well as identifying the differences between voice and data. This information can be displayed by using a FSV and its associated spectrum display unit. Such a display is shown in figure 9-11, with some of the important baseband components identified. This baseband is for a 120-channel system and the drawing overlaps five separate pictures in order to present the entire baseband in one diagram. Operation of the spectrum display unit is outlined in chapter 17.

9-8. Signal Impairments. In our previous discussions, we have seen how several signals may be transmitted over a single transmission medium. We also noted that the majority of high density systems (more than 24 channels) use FDM and, as such, individual channels are translated to their separate positions in the frequency spectrum using AM. These systems are subject to impairment from overload, distortion, noise, crosstalk, and carrier leak. With the exception of carrier leak, these impairments are discussed in chapter 4. Carrier leak is another signal impairment that causes degradation of the multiplex output signal. Carrier leak is the result of the unbalanced condition of a suppressed carrier modulator. These modulators are balanced, but they are not balanced perfectly; hence, some carrier leakage will be present. Ideally, a balanced condition exists when the input audio signal and the carrier frequency are balanced out, leaving only the sum-and-difference frequencies in the output. With carrier leak present, the carrier frequency is also present in the output.

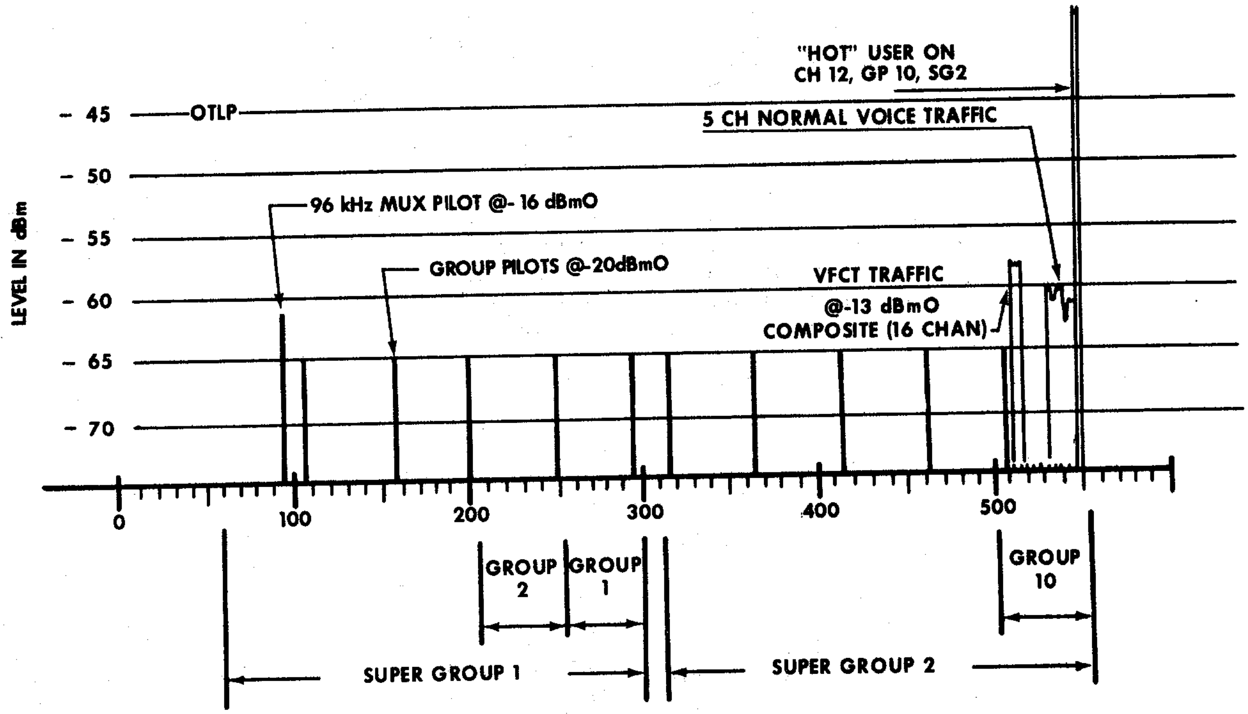


Figure 9-11. Baseband Spectrum Display.

Chapter 10
RADIO COMPONENTS

10-1. General. This chapter discusses the principles and characteristics of the radio component of the wideband communications system. The operation of the transmitter and receiver component is also described. This discussion is applicable to both types of wideband radio systems used in the DCS (LOS and tropo systems). Differences in frequency, power considerations, antenna positioning, and path lengths for these systems require some difference in equipment used. The descriptions which follow apply to both systems generally and contain discussions dealing with specific aspects of these differences.

10-2. Function. The function of wideband radio systems is divided into two parts. First, the transmitter must convert the input baseband to a modulating signal, then superimpose this modulation on a carrier wave and, finally, deliver the modulated wave to the antenna. The second part of the radio is the receiver. Its function is to receive the modulated wave and recover the amplitude and frequency of the modulating wave that was impressed on the carrier at the transmitter.

10-3. Description:

a. FM Principles. The majority of wideband systems use FM because it has an inherent low noise level. In addition to the noise advantage, FM has another characteristic which is often called the "capture" effect. An example of this effect is where two signals in the same frequency band are available at the receiver; the one appearing at the higher level can be accepted to the near exclusion of the other.

Two other features of FM make its application to FDM desirable when bandwidth is not a critical limiting factor. First, separate regulation is not necessary because, in FM receivers, the output signal level is insensitive to input signal level variations above the receiver threshold level. Secondly, synchronization is not a problem due to the detection method used in FM.

b. FM Radio Characteristics. To provide high quality communications, the radio equipment must possess certain characteristics. These include frequency stability, bandwidth, fidelity, gain, sensitivity, and selectivity. All of these (except sensitivity and selectivity) will be discussed in the following paragraphs. Sensitivity and selectivity will be discussed in conjunction with the receiver.

(1) Frequency Stability. The frequency stability of radio equipment depends on the frequency stability of the oscillators used in the equipment. These oscillators generate basic radio frequencies with as much stability as can be provided. Three factors affect the operating frequency of an oscillator. First, geometric factors in which the effective inductance and capacitance are changed directly through mechanical motion; second, pulling factors in which reactance is coupled into the oscillator circuit from the load; and third, pushing factors in which reactance is introduced by changes in input conditions (such as voltage, current, or magnetic fields).

The oscillators generally used in FM applications are variable frequency oscillators which may be modulated for FM purposes. Although they are adjustable to any frequency in their frequency range, the advantage of frequency adjustment does present a minor problem of frequency instability.

At lower frequencies, it is not too difficult to maintain the desired stability; however, at higher frequencies it becomes increasingly difficult to obtain stability. Even when the same percentage of stability is attained, serious shifts of frequency may occur at M/W frequencies. For example, a frequency shift of .01 percent at 1 MHz is only 100 Hz and presents no problem. But this same percentage shift at 10,000 MHz is equal to 1 MHz, which is enough to interfere with satisfactory operation. When operating at higher frequencies, a means must be provided to maintain frequency stability.

There are two means of ensuring stable operating frequencies. One is to use a synthesizing circuit with crystal control to maintain the required frequency stability. The second means employs mechanical or electronic automatic frequency control (AFC) systems that correct the oscillator when it drifts from the desired frequency.

In the crystal control method, a quartz crystal is used as the frequency-determining component of the oscillator. Optimum stability from this type of oscillator can be achieved only if the crystal is protected from temperature changes. Because of this, the crystal is placed in a device called a crystal oven which holds the crystal temperature constant. The interior of the oven is maintained at a constant temperature by a thermostatically-controlled heater. With temperature stabilization, crystals provide very precise frequency stability.

The maximum frequency that can be generated by a crystal is somewhat limited. To attain higher output frequencies, frequency multipliers are required. The multiplication factor is determined by the actual oscillator frequency needed.

To overcome the basic frequency limitation of the quartz crystal, a klystron oscillator is also used. Basically, the klystron oscillator is not as stable in frequency as the quartz crystal oscillator. To achieve the required stability, AFC is used. In this method, a discriminator monitors the average frequency of the oscillator or another frequency which is a function of the oscillator frequency. The discriminator detects errors in oscillator frequency and generates a correction signal which is then applied to the klystron oscillator, returning it to the proper operating frequency. Either of these oscillators may be used, but the klystron oscillator is used in higher frequency applications.

(2) Bandwidth. The bandwidth of a device is defined as the difference between the lower limit and the upper limit of its frequency response. The limit in frequency response is the point where the gain of the device falls a preselected amount below the maximum gain. This preselected amount may be a portion of a dB or as large as 3 dB.

Figure 10-1 shows a response curve of an amplifier. Points A and B represent the points where gain falls below a preselected level. These points then identify the upper and lower limit of frequency response. Bandwidth, for this example, is equal to the frequency at B minus the frequency at A. Frequencies between A and B are within the bandpass of the amplifier.

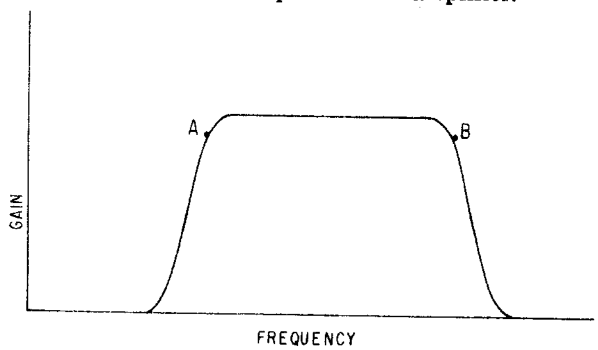


Figure 10-1. Frequency Response.

For communications purposes, the bandwidth of the radio equipment must be just sufficiently broad to pass the carrier and the important sidebands. If the radio does not have sufficient bandwidth to pass all important sidebands, then a portion of the original intelligence is lost. If the bandwidth is wider than required to pass the important sidebands, unnecessary noise is passed in addition to the desired signals. This noise degrades the recovered intelligence.

(3) Fidelity. Fidelity in communications is the faithful reproduction of a signal. This includes the entire waveshape of the signal. Any alteration of frequency content or the phase and amplitude relationships constitutes distortion and, therefore, a reduction of fidelity. Some of the factors which affect fidelity are: linear operation, bandwidth, overall frequency-amplitude response, and delay.

(a) Linear operation refers to the amplitude characteristics of a signal. To operate linearly, a device must present the same response to all amplitude variations. In an amplifier, this means amplifying a change of one unit the same amount wherever it occurs in the signal. In an FM modulator, it means that the same amount of frequency change will be produced by a one unit amplitude change in the modulating signal wherever in the baseband signal this change occurs. In a discriminator, it means that a given amount of frequency change will cause the same amount of change in output wherever in the bandpass of the discriminator this change occurs.

Figure 10-2 is an illustration of nonlinear amplification. The positive portion of the signal was amplified more than the negative portion. Distortion of this type results in the generation of new frequencies in the form of harmonics of the basic signal. For practical purposes, we can consider these new frequencies as an interfering signal or noise. The discriminator in the receiver also presents a number of possibilities for non-linear operation. These will be examined later when we discuss the demodulator of the receiver.

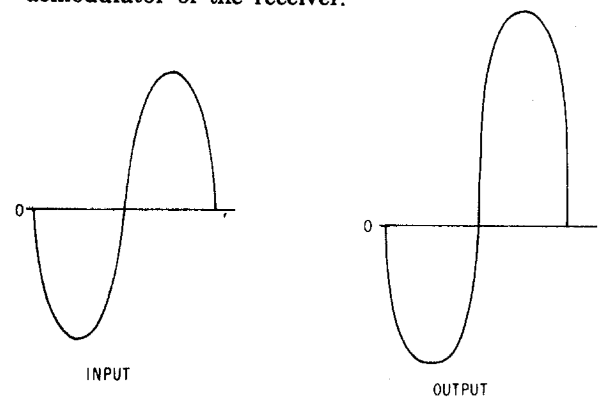


Figure 10-2. Non-Linear Amplification.

(b) A second threat to fidelity in the radio is bandwidth. Previously, we stated that all intelligence in a modulated wave is carried in the sidebands as a result of the modulation process. If any portion of the radio does not have sufficient bandpass to accommodate the carrier and all of the important sidebands, then a portion of the sidebands is either reduced in amplitude or removed from the signal entirely. Reduction or elimination of any of the important sidebands means that the intelligence in that portion has been changed. In this respect, the fidelity of the system suffers.

(c) Closely related to bandwidth is the total frequency-amplitude response of a device. Bandwidth generally applies to the upper and lower frequency limits that an amplifier will respond to. Overall frequency-amplitude response considers the entire bandpass of a device. The degree of flatness of the response is of major interest in this consideration. The word "flatness" refers to how much the amplification or gain of a device varies as input frequency is varied from minimum to maximum frequency. Figure 10-3 shows one frequency-amplitude response curve with poor flatness and one with an improved flatness characteristic. In the example of poor flatness, amplification changes occur as frequency is changed.

Amplification of frequency A has 3 dB more gain than frequency B. The response curve is tipped or tilted down from point A to point C. This is the result of a gradual decline in gain as frequency increases. The improved response curve indicates that all frequencies within the bandpass receive the same gain. Because of this fact, the amplitude relationship of the frequencies present at the input is preserved and fidelity in this characteristic is the result.

(d) Delay of one portion of a signal more than another will cause loss of fidelity. Many signals carried by the wideband communications system are composed of not one frequency, but many; however, each frequency of a particular signal has a definite time relationship to the other frequencies of that signal which must be maintained. A delay of one or more of the component frequencies in relation to others will alter the original signal. This again results in a reduction of fidelity.

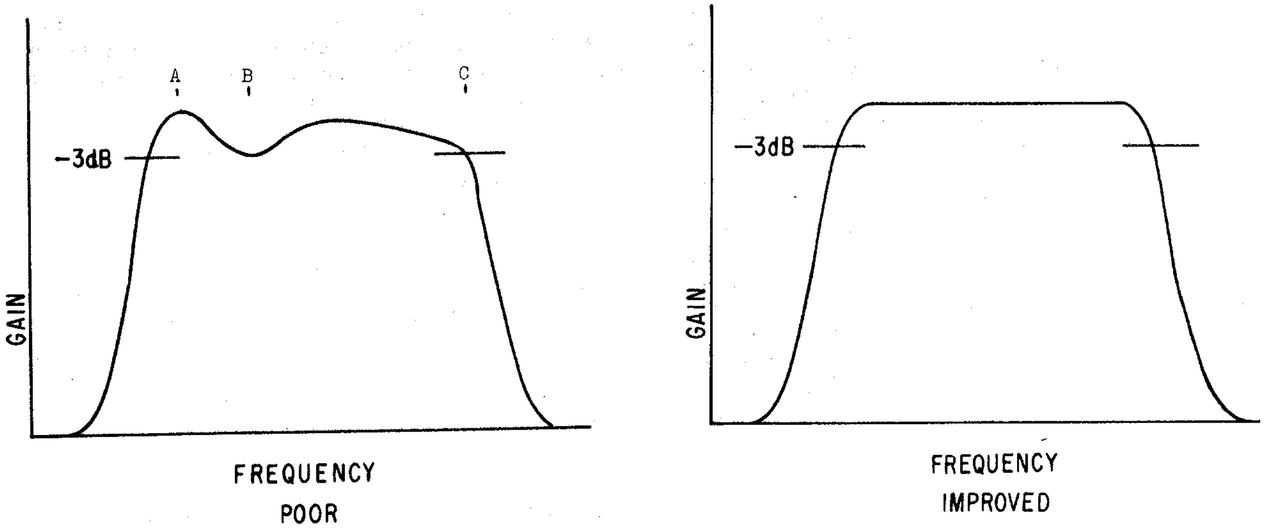


Figure 10-3. Frequency-Amplitude Response Curves.

For communications to be effective, all information entering the system must be delivered to the recipient in an understandable condition. Fidelity plays an important part in assuring that these conditions are met.

(4) Gain. Electronic communications devices require a certain amount of energy to be present in the signal they process. For the output of radio equipment to be useful, it must provide a signal with sufficient energy to actuate the using device. The devices that use this energy vary widely in their input requirements. For instance, the transmitter must provide enough gain after the modulation process so that the signal reaching the receiver is at an adequate power level. This input power to the receiver usually is extremely small, perhaps on the order of a few hundred picowatts. The receiver, therefore, must provide sufficient amplification or gain to raise the input signal to a level where the intelligence can be recovered. The recovered intelligence must then be amplified to a point where it will drive the using device.

10-4. Transmitter:

a. Operation of a Basic FM Transmitter. To transmit information as electromagnetic waves, it is necessary to convert intelligence to a modulating signal, superimpose this signal on a carrier, and deliver the RF power to the antenna system. FM transmitters are largely conventional and consist of three basic sections: the oscillator-modulator, the frequency multiplier, and the power amplifier (figure 10-4).

In the first section, the carrier frequency is generated and frequency-modulated. In the multiplier section, the carrier is increased in frequency to that of the operating frequency and, in the amplifier section, the power is increased to the desired operating level. Let's consider each of these sections now to see how these functions are performed.

(1) Oscillator/Modulator. Frequency-modulated waves are produced by varying the carrier frequency at the modulating signal rate above and below the unmodulated carrier frequency. This can be achieved by using a reactance tube or a voltage variable capacitor to alter the frequency of the transmitter master oscillator. A reactance tube is a conventional vacuum tube connected so as to appear as a capacitive or inductive reactance that varies proportionally to the applied grid potential.

PM is produced by allowing the modulating signal to shift the phase of the carrier, the change of the phase angle being proportional to the instantaneous amplitude of the modulating signal. PM, as such, is seldom employed since it does not use a given frequency channel as effectively as does FM transmission; however, PM is important as a means of obtaining FM. The phase modulator requires an integration circuit or a corrector network to compensate for the effect that the frequency of the modulating signal has on the total frequency deviation.

A direct method of generating FM is to control the frequency of an ordinary oscillator (such as the Hartley oscillator) by a modulating voltage. The most common method of doing this is by means of a reactance tube. If a reactance tube is used with a crystal-controlled oscillator, the resulting modulation is closer to PM than to FM because the frequency deviation that can be secured by varying the tuning of a crystal oscillator is quite small. Although a reactance tube operating on a free-running oscillator is capable of producing wide deviation, the frequency of such a transmitter is susceptible both to variations in power supply voltages and to changes in parameters (due to aging and temperature variations). Such susceptibility results in very poor frequency stability; however, this can be improved through the use of push-pull reactance modulators but, in the end, the desired stability must be achieved through the use of AFC. The sensitivity of the reactance tube modulator depends on the transconductance of the modulator tube and certain circuit factors. Usually this type modulater is capable of causing a frequency deviation of several hundred hertz, with a distortion of a few hundredths of a percent at the oscillator frequencies commonly used.

The Serrasoid modulator is widely used today. It can provide phase shifts of up to about ± 150° or a corresponding frequency excursion of ± 130 Hz, with a distortion of a few tenths of a percent. The noise originating in the modulator can be kept about 80 dB below the signal level obtained with 100% modulation. The Serrasoid modulator (figure 10-5) consists of three tubes which generate a stable, very linear sawtooth wave and a modulated gate. These signals are used to generate a pulse whose time phase is proportional to the instantaneous audio signal. The RF oscillator draws plate current during only a small part of the cycle. This signal is then differentiated and clipped by a cathode follower. These pulses are used to time the sawtooth generator. Because the linearity of the modulator depends on the straightness of the sawtooth wave, great effort is made to achieve the desired waveform. The sawtooth generator is directly coupled to the modulators or gate tube. This tube is normally cathode-biased so that conduction begins when the sawtooth wave reaches about half its peak value. Once conduction starts, the sawtooth is clipped. This causes a negative pulse in the modulator output. The bias of the modulator is varied by the modulating signal, thus varying the clipping level of the sawtooth wave and time of conduction of the modulator tube. The negative-going portion of the wave from the modulator is differentiated and amplified. The time of the resulting pulse can thus be advanced or retarded because the audio signal alters the bias on the modulator tube. These pulses are then applied to the frequency multipliers and reshaped to sinusoidal waves.

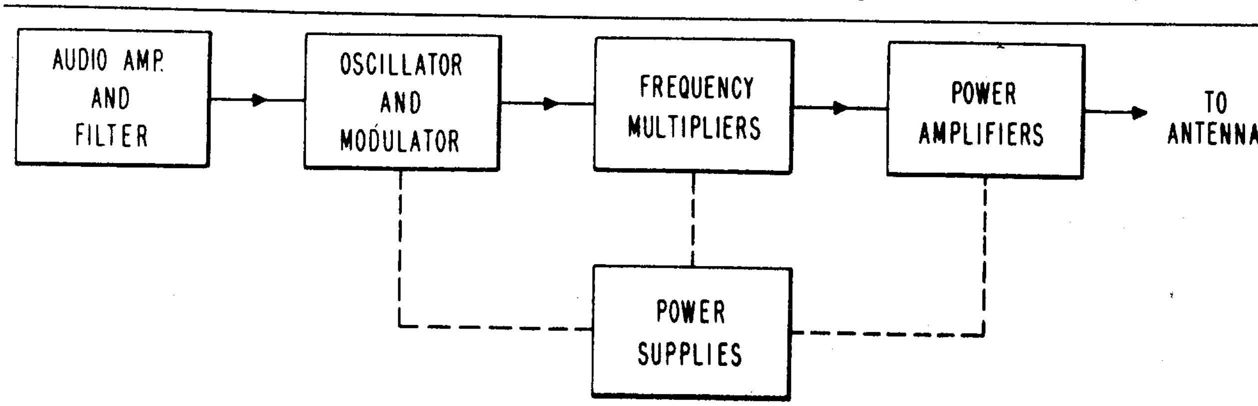


Figure 10-4. Basic FM Transmitter.

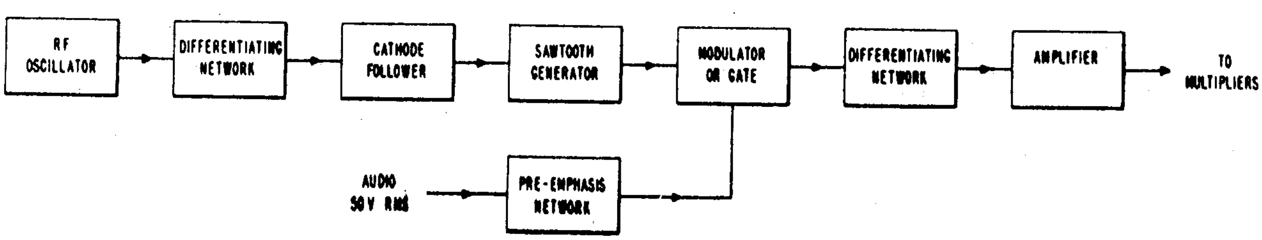


Figure 10-5. Serrasoid Modulator.

(2) Solid State Modulator. The solid state modulator is a state-of-the-art varactance device operating directly at 70 MHz. It has vast overload capability minimizing modulator overload distortion.

(a) Modulator Assembly. The main components of the modulator assembly (figure 10-6) are the transformer and filter assembly, modulator, AFC, exciter power supply, channel capacity selector, and fault monitoring circuits. The modulator assembly receives baseband and orderwire signals from the multiplex equipment. These inputs are routed through the channel capacity selector assembly to the modulator. The orderwire signal is also applied to the modulator after attenuation by the orderwire attenuator.

In the modulator, the baseband signal is amplified and a radio pilot tone is added by a radio pilot generator. The orderwire and baseband signals are then combined and used to frequency-modulate the output of a VCO which is operating at a center frequency of 70 MHz. This center frequency is maintained by the circuits of the AFC section. The FM output of the VCO is amplified and applied to the power divider which provides two identical 70 MHz output signals. One output is applied to the transfer relay in the associated exciter and the second is routed to the transfer relay in a second exciter. This permits two exciters to operate using the modulator assembly from one exciter.

1. Transformer and Filter Assembly. Consists of two transformers and a high pass filter. The assembly's function is to accept a baseband signal which is split into two frequency bands: low frequency (12 kHz to 70 kHz) and high frequency (60 kHz to 1300 kHz) and to provide a filtered, combined baseband output. The low and high frequncy bands of the baseband signal are combined in the transformers and a single baseband output signal is provided. The output is applied to the high pass filter which prevents passage of

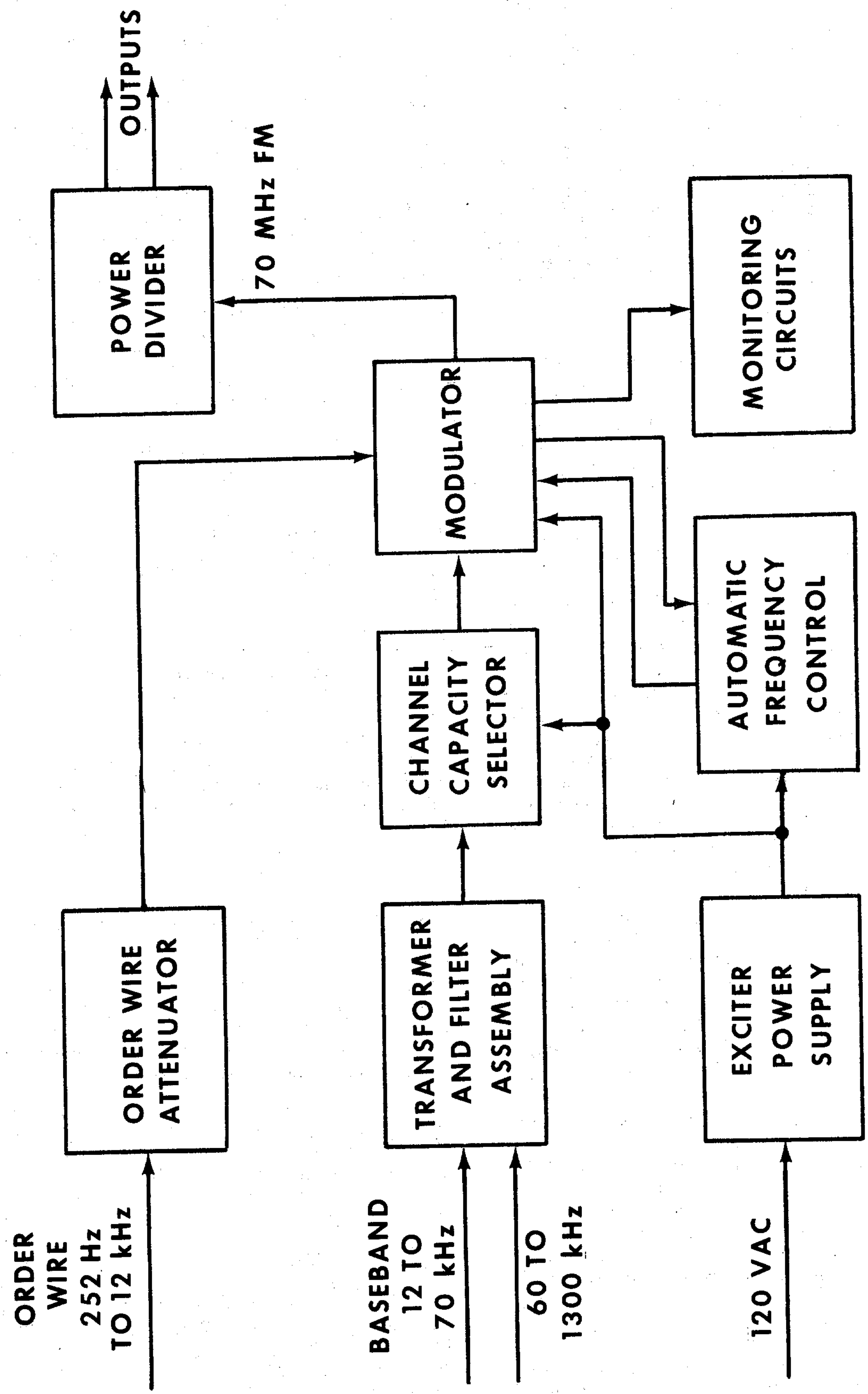


Figure 10-6. Modulator Assembly Block Diagram.

spurious noise and ground currents below the baseband spectrum into the modulator. The output of the high pass filter is applied to the channel capacity selector.

2. Orderwire Attenuator. This is a resistance pad which provides a 600 ohm input for the orderwire signals and attenuates these signals to a suitable level for application to the orderwire input of the modulator.

3. Power Divider Assembly. The power divider contains a hybrid transformer, the primary of which receives the 70 MHz FM signal from the modulator. The two secondaries of the transformer provide RF outputs (two identical 70 MHz signals). These RF signals are the outputs of the modulator assembly.

(b) Modulator (figure 10-7). Consists of baseband amplifier I, radio pilot generator, baseband amplifier II, VCO and diode-function generator oven assembly, IF amplifier, orderwire amplifier and VCO bias, channel selector, deviation control, and crystal oven. The baseband amplifier and radio pilot generator circuits combine the baseband signal and 60 Hz pilot tone. The combined baseband and 60 Hz pilot tone signals are routed through the channel capacity selector which provides the needed pre-emphasis for operation at the selected channel capacity. After pre-emphasis and equalization, the combined baseband is further amplified by baseband amplifier II. The gain of baseband amplifier II is partially offset by the attenuation in the deviation control circuit. This circuit adjusts the combined baseband and pilot tone signal to provide the required modulation index. The combined signal is amplified and then pre-distorted by the action of the diode function generator and passed through the output stage of baseband amplifier II. Pre-distortion of the modulating signal is accomplished by the diode function generator which is located in a temperature-controlled oven for stability. The pre-distortion characteristics are determined by the diode function generator's bias, taken from baseband amplifier II, and by the diode. The pre-distortion is controlled by the bias applied. The amplitude of the bias is adjusted to create the proper amount of pre-distortion to compensate for the non-linearity of the bias-capacitance curve of the VCO varactor diodes. The VCO produces a 70 MHz IF signal which is frequency-modulated by the modulating signal and is maintained at a 70 MHz center frequency by the AFC voltage and bias. The IF amplifier amplifies the 70 MHz FM signal and provides an output to the power divider and an output to the AFC.

(3) Frequency Multiplier. The efficiency and stability of an oscillator decreases as the operating frequency is increased. To satisfy the requirement for stability and efficiency and provide higher transmitter operating frequencies, multiplier stages are added. In this mode, the oscillator signal is fed through a buffer stage to the multipliers. This buffer stage prevents oscillator loading and thus improves frequency stability. The multiplier is operated in a non-linear mode so that the output has a large harmonic content. By tuning the plate circuits of the multiplier to one of these harmonics, a frequency-multiplying action takes place. The multiplication factor depends on which harmonic is chosen, that is, second harmonic selection results in a multiplication factor of two. The final stages of the multiplier are also used to increase the power level to that necessary to drive the power amplifier.

The frequency multiplier performs one additional important service. The modulators produce frequency deviations of relatively small magnitude. The multiplier section multiplies this characteristic of the signal also. By this means, the desired frequency deviation of the transmitter signal is achieved. Final frequency deviation can be computed by the formula:

$$F_d = Mf_d \qquad \text{Eqn 10-1}$$

where: F_d = the frequency deviation of transmitted signal

M = the frequency multiplication factor of the multiplier chain

f_d = the frequency deviation produced by the modulator

The use of frequency multipliers results in several advantages; however, one disadvantage is inherent in this process. Just as desired deviations are multiplied, all oscillator frequency instabilities are multiplied by the same factor.

(4) Power Amplification. After the RF signal has been multiplied to the desired operating frequency, it must be amplified to the required output power level. Without sufficient power, the signal sent to the intended receiver would be unusable. The operating load impedance in RF power amplifiers is usually adjusted to the value that permits the amplifier to operate at high efficiency. With some equipment, an additional power amplifier (called an intermediate PA) is needed because required output power is so high. It is placed between the last frequency multiplier and the output power amplifier.

Power gain is an important factor in amplifiers. It is the ratio of power output of a stage to the driving power input. Power tubes give a wide range of power gain, ranging from triodes with 7 dB to 17 dB to pentodes with a theoretical gain up to 30 dB. The important thing to remember is that regardless of the type of tube, the gain will fall off at an increasing rate as operating frequency is increased.

(5) Power Amplifiers. The power amplifiers used in FM systems are either conventional tubes, klystrons, or traveling wave tubes. The power output of an FM transmitter may usually be increased at any time by the addition of larger power amplifiers to the existing equipment.

b. Wideband Communication Transmitters. The wideband transmitter consists basically of an oscillator, modulator, and amplifier, with the complexity of these units depending on frequency, power output required, and type of modulation used. This section discusses oscillators as they apply to wideband transmitters, wideband modulators, power amplifiers, and M/W amplifiers.

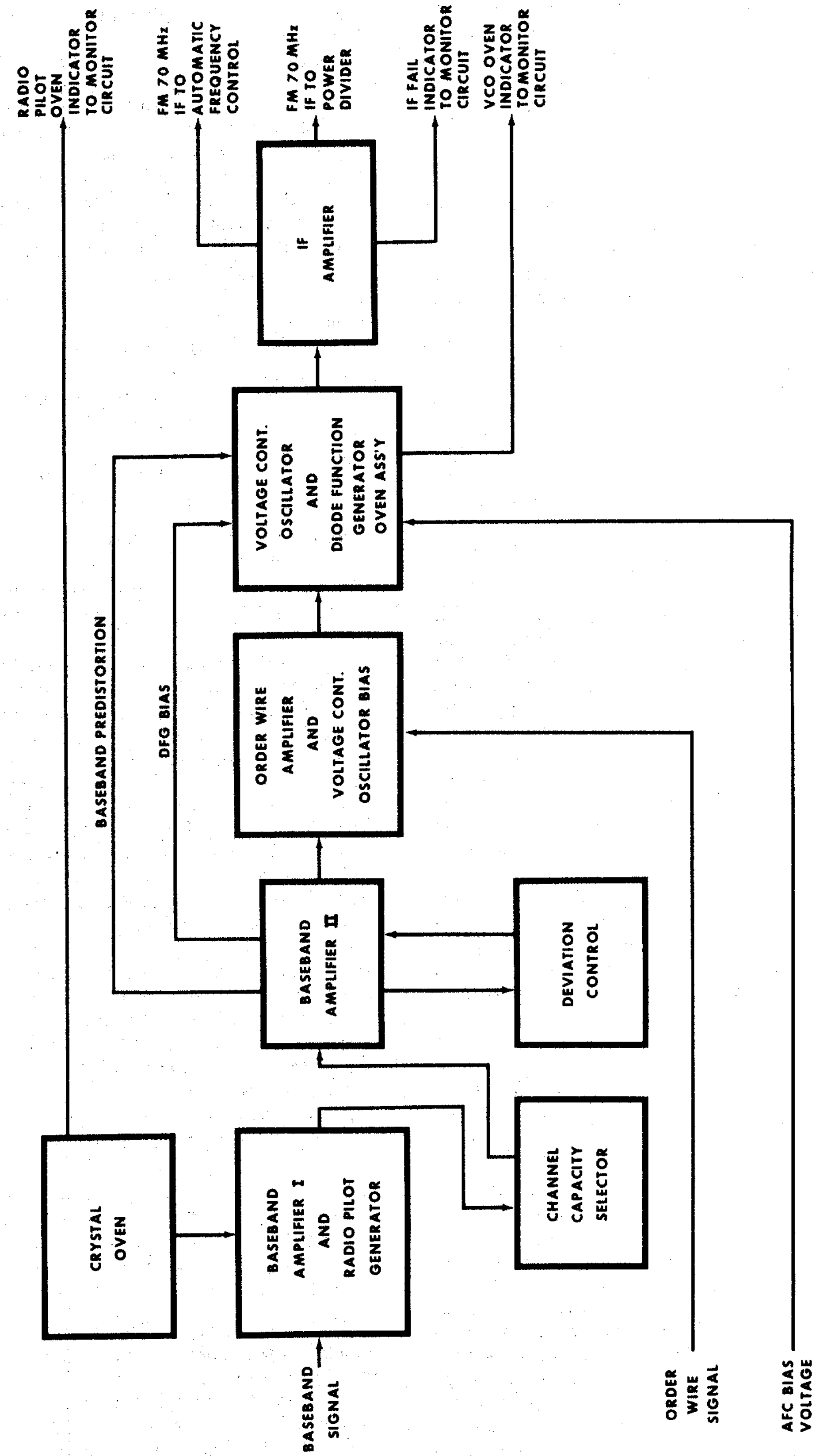


Figure 10-7. Modulator Block Diagram.

(1) Wideband Oscillators. As M/W frequencies were approached, conventional oscillators began to fall off in efficiency, so new systems were developed to produce oscillations at these higher frequencies. The M/W oscillator most commonly used at the present time is the klystron. This type of tube takes advantage of the finite speed of the electron. Two categories of klystrons are used in the wideband communications system:: the reflex klystron and the multiple cavity klystron. The multiple cavity klystron is primarily used for amplification and will be discussed in the M/W amplifier section of this chapter.

The reflex klystron is a low power tube which has a power efficiency of about one percent. Used primarily as an oscillator, it consists of an electron gun, a cavity resonator, a repeller plate, and a collector. The electron gun emits a beam of electrons through the cavity resonator toward the repeller plate. The repeller, because of its potential, repels the electrons back through the resonant cavity to the collector.

The klystron is a velocity-modulated tube. The speed or velocity of electrons in the beam is altered by the reasonant cavity signal the first time the electrons pass through the cavity. The result of this speed change is a bunching of the electrons in the beam. After being repelled by the repeller, the electron bunches return through the cavity and deliver energy to the cavity on this trip. If the time interval prior to the return of the electron bunch is correct, this energy adds to the cavity signal and oscillations occur. The frequency of oscillation is primarily determined by the resonant cavity, but the frequency can also be varied over a small range by changing the repeller plate voltage.

(2) Modulators. FM is used in wideband transmitters to impress the characteristics of the intelligence signal on the carrier. This FM may be accomplished by the direct or indirect method. Two devices are used to produce direct FM. One of these is the reactance tube which we discussed earlier in this chapter. The other is the reflex klystron. In this device, the modulating voltage is applied to the repeller plate. Variations in this voltage alter the electron paths and distance electrons travel in the tube. This changes the time consumed by electrons in travel. Since frequency in this type oscillator is a function of electron travel time, this changes the frequency of the oscillator. Negligible power is required to modulate the oscillator and a large frequency deviation with very little AM can be obtained.

The second method used - indirect FM - is produced by PM. The most common system used to produce indirect FM is the Serrasoid modulator.

(3) Power Amplifiers. The modulated signal may be passed to an amplifier to increase the amplitude of the outgoing signal. The same limitations of conventional circuits at M/W frequencies that applied to oscillators apply as well to amplifiers. In the M/W region, no amplification is possible with conventional vacuum tubes and circuits - either the oscillator itself must supply enough power or specially designed amplifiers must be used; therefore, as frequencies are increased beyond the point where triode amplifiers will operate efficiently, klystron amplifiers and traveling wave tube amplifiers are used.

(4) M/W Amplifiers. The klystron may consist of a single cavity, such as that used for an oscillator, or it may be a special amplifier klystron used for high power and known as a cascade or multiple cavity klystron. This tube is, in effect, two klystrons connected in cascade within the same envelope, with the resonant cavity for the first section functioning as the buncher grid for the second section. The signal to be amplified is fed to the first cavity and the power output is taken from the last cavity. The intermediate cavities are energized by the bunched electron beam and are not supplied with external RF driving power. These tubes are capable of power gains of up to 30 dB, with efficiencies of 30 to 40%.

Traveling wave tube amplifiers are the second type that may be used at M/W frequencies. The traveling wave tube is an amplifier which uses an electron beam and an RF wave traveling together in such a way that the wave accepts energy from the electron beam. The tube consists of an electron gun, a helix, and a collector. The electron gun produces a focused beam of electrons which is directed through the center of the helix to the collector. The helix is a wire that has been formed into a long uniform spiral. Input signals are applied to the helix and cause velocity modulation of the electron stream. In the klystron, this modulation occurs only at the resonant cavities. In the traveling wave tube, the modulation occurs along the entire length of the helix. Bunching of the electron stream is the result of this modulation. As the beam continues the trip from the cathode to the collector, bunching becomes more complete and energy from these bunches is transferred to the helix. This energy is added to the input signals traveling along the helix. Gains from 10 to 60 dB are achieved by this type amplifier.

The output signal is coupled from the power amplifier. By means of the transmission line, this energy is transported to the antenna where it is radiated to the distant receiving station.

10-5. Receiver:

a. Function. The basic radio receiver accepts the modulated radio frequency voltages from the antenna system. Its purpose is to recover the intelligence from this modulated signal and provide this intelligence in a form usable to other components in the communications system. In a simple system, this form may be nothing more than a current, varying at an audio rate, which drives a loudspeaker. As the system becomes more sophisticated, the output becomes more complex. In the wideband communications system, the output is a very complex signal that contains many channels of intelligence. This signal must be treated by other equipment before communications between system customers may be accomplished.

b. Receiver Characteristics. To perform the function of recovering intelligence from a modulated radio frequency wave, the receiver must have certain basic characteristics, including sensitivity, selectivity, bandwidth, fidelity, and gain. Before we look at a basic receiver, let's consider the characteristics of sensitivity and fidelity.

(1) Sensitivity. Receiver sensitivity is defined as the receiver's ability to respond to weak signals. In communications, this becomes a problem of discriminating between the weakest signals to be received and noise.

In chapter 6 we found that current flow in circuit components and heating of components generate noise in electronics devices. In a receiver, these noises and others generated within the circuits are amplified the same as the signals from which we desire to recover intelligence; therefore, the ratio between them remains constant, even though we amplify the signals by a large factor. The noise generated by the circuits then becomes the limiting factor in how small the input signal to a receiver may become before it is unusable.

As input signal decreases, a point is reached where the S/N ratio is no longer acceptable. This limit is primarily determined by the noise level generated in the receiver input. This noise generated in the input circuits, up to and including the input of the first amplifier, is amplified more than any other noise in the receiver. If the gain of this first amplifier is high, noise contributions by the later stages are comparatively small in terms of total noise output.

The noise figure of a receiver, then, is a good indicator of how well the receiver can respond to weak signals - the lower the noise figure, the more sensitive the receiver is.

(2) Selectivity. Selectivity is closely related to bandwidth and can be defined as the receiver's ability to accept one signal and to reject adjacent signals. Selectivity is important in the communications receiver because it enables the receiver to respond to a desired signal and simultaneously reject undesirable adjacent signals and noise.

Figure 10-8, example A, shows a response curve with poor selectivity. Signals at frequency F are technically outside the bandpass of the amplifier; however, the amplifier does respond to them and a considerable amount of frequency F would appear in the amplifier output.

Figure 10-8, example B, shows a curve with the same bandwidth as example A, but with improved selectivity. Frequency F is also outside the bandpass of the amplifier. The number of Hertz from the center of the bandpass to frequency F is the same in both examples. The amplifier's response in example B drops off much more sharply as we move out of its bandpass; consequently, the gain to frequency F is much lower. With this lower gain, the amount of frequency F appearing in this amplifier's output is much less than that in the previous example. The rejection of frequency F, an out-of-band signal, is increased. As a result, example B represents a more selective amplifier.

All receivers possess important characteristics in lesser or greater degrees. To summarize these characteristics, they are sensitivity, bandwidth, fidelity, and gain. Let's look at a basic receiver and see how it functions and how these characteristics play a part in this operation.

c. Basic Receiver Operation. The most common type of receiver in use is the superheterodyne variety. This receiver acquires its name from the heterodyning, or mixing, action that takes place in the first portions of the receiver. The reasons for this operation will be discussed later. The sections that comprise the superheterodyne FM receiver are the front end, mixer/oscillator, IF amplifier, discriminator, and output section. Figure 10-9 assembles these sections into a block diagram of a receiver.

(1) Front End. In previous discussions, we saw how the front-end circuitry determined the basic sensitivity of the receiver. The noise generated within this device was shown to be the limiting factor in sensitivity. In a properly operating receiver, the noise generated in the front end is the major contribution to the noise added to the signal in the receiver. Because of these factors, the first requirement for the front end is that it be a low noise device.

The front end also performs the initial selection of the signal to be received. This is accomplished by a circuit that is tuned to accept the desired signal and reject undesired signals. Because of this tuning action, the front end has a frequency-amplitude response and a bandwidth characteristic. The frequency-amplitude response must be flat to prevent distortion of the signal. The bandwidth must be wide enough to pass the carrier and important sidebands of the desired signal.

The front end of some receivers contains an RF amplifier. The purpose of this device is to provide larger signals for the mixer. Several factors determine whether the front end will contain an RF amplifier or not. In some simple receivers or those receiving strong signals, this amplifier may not be required. In others, the RF amplifier may be deleted to reduce manufacturing cost. A more serious reason for deletion is that, as frequency increases, it becomes more difficult to design and construct an RF amplifier. If vacuum tubes are used as the amplifier, the noise generated by this device, because of its hot cathode, is a limiting factor. Another limitation of the vacuum tube is transit time or the time it takes an electron to move across the space between cathode and anode of the tube. As this time approaches the time of one cycle of the input signal, the tube becomes very inefficient. To combat these shortcomings, other type designs may be used. These will be discussed later, in the parametric amplifier section.

(2) Mixer/Oscillator. The front end passes the selected input signal to the mixer. The purpose of the mixer is to translate the input signal from the radio frequency to a different one, which is usually a lower frequency. This new frequency must contain the same intelligence as was contained in the original receiver input. The intelligence will be carried in a series of sidebands whose structure is essentially the same as the input signal. This new signal is called the IF signal.

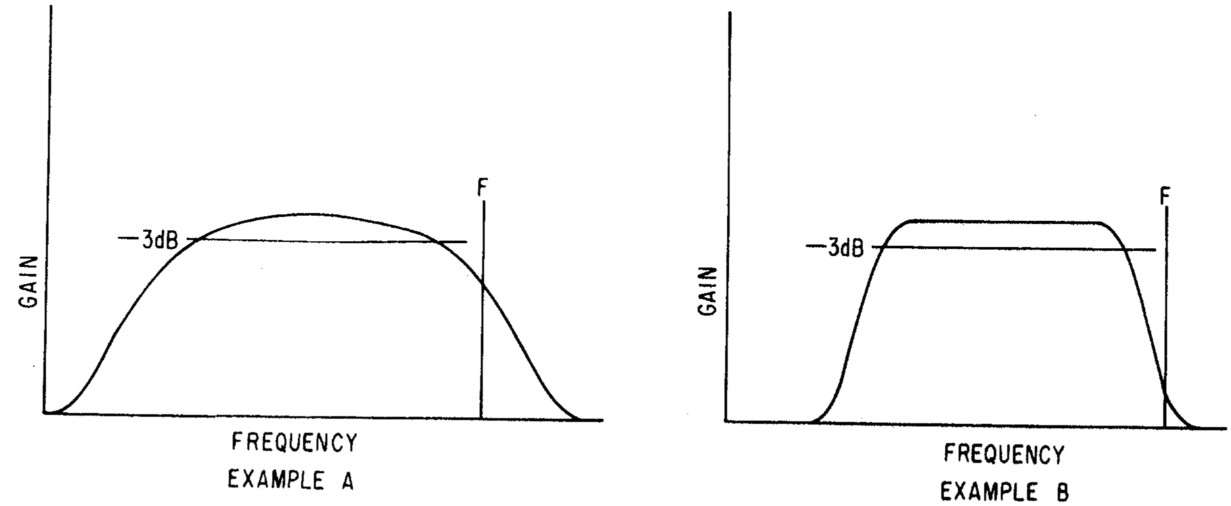


Figure 10-8. Amplifier Response.

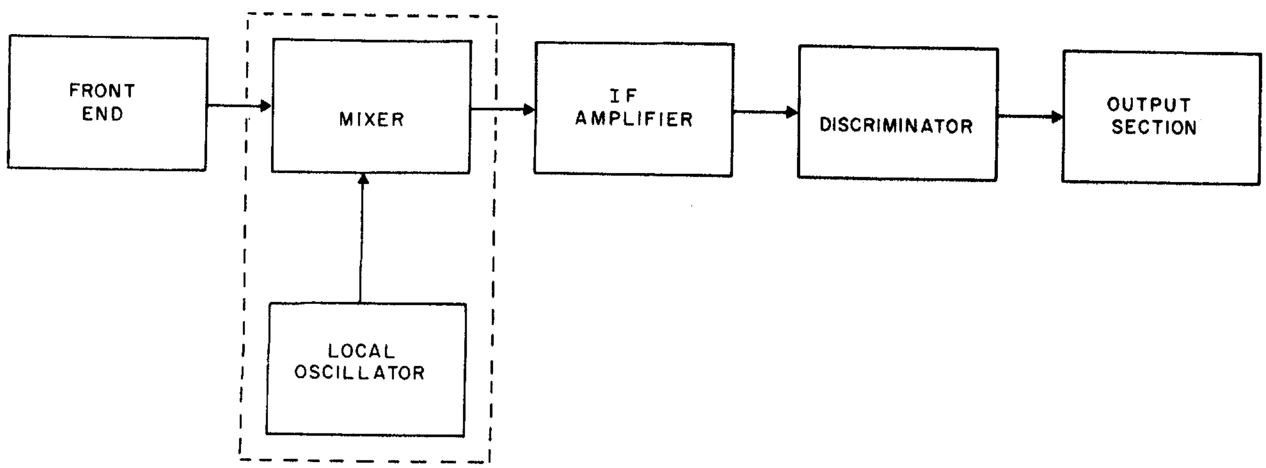


Figure 10-9. Superheterodyne Receiver.

The frequency translation is accomplished by amplitude-modulating the local oscillator signal with the receiver input signal. This generates an upper and lower sideband that both contain the same information as the input signal. The frequency of the local oscillator is selected so that (1) the sum of it and carrier frequency or (2) the difference between it and carrier frequency is equal to the IF. This means that the upper sideband in example 1 or the lower sideband in example 2 is centered at the IF.

An example will perhaps clarify this process. Let's suppose an IF of 30 MHz is selected for the receiver and that the desired receive signal is centered at 200 MHz. The local oscillator frequency required could be 170 MHz or 230 MHz. The difference between 200 MHz and 170 MHz is the desired 30 MHz IF. The difference between 230 MHz and 200 MHz is also the desired 30 MHz IF. Either of these local oscillator frequencies would provide the 30 MHz IF. Mixing products of a 200 MHz receiver input and local oscillator frequencies of 230 MHz and 170 MHz are shown in figure 10-10.

By the same method, after a local oscillator frequency is selected, either of two receiver input signals will result in the IF. If we selected 170 MHz from our previous example as the local oscillator frequency, 200 MHz at the receiver input produced the 30 MHz IF. Another receiver input signal at 140 MHz would also result in the IF. This other frequency that will produce the IF is called the image frequency (figure 10-11). The receiver front end and the mixer input circuit must be selective enough to pass the desired signal (200 MHz) and to reject the image (140 MHz) to prevent interference.

Let's consider the signal requirements for the local oscillator. They are signal amplitude, noise-free output, amplitude stability, and frequency stability.

The amplitude of the signal from the local oscillator must be large in comparison to the input signal, so that overmodulation (greater than 100%) does not occur. Overmodulation would result in generation of new frequency components that would appear as noise in the

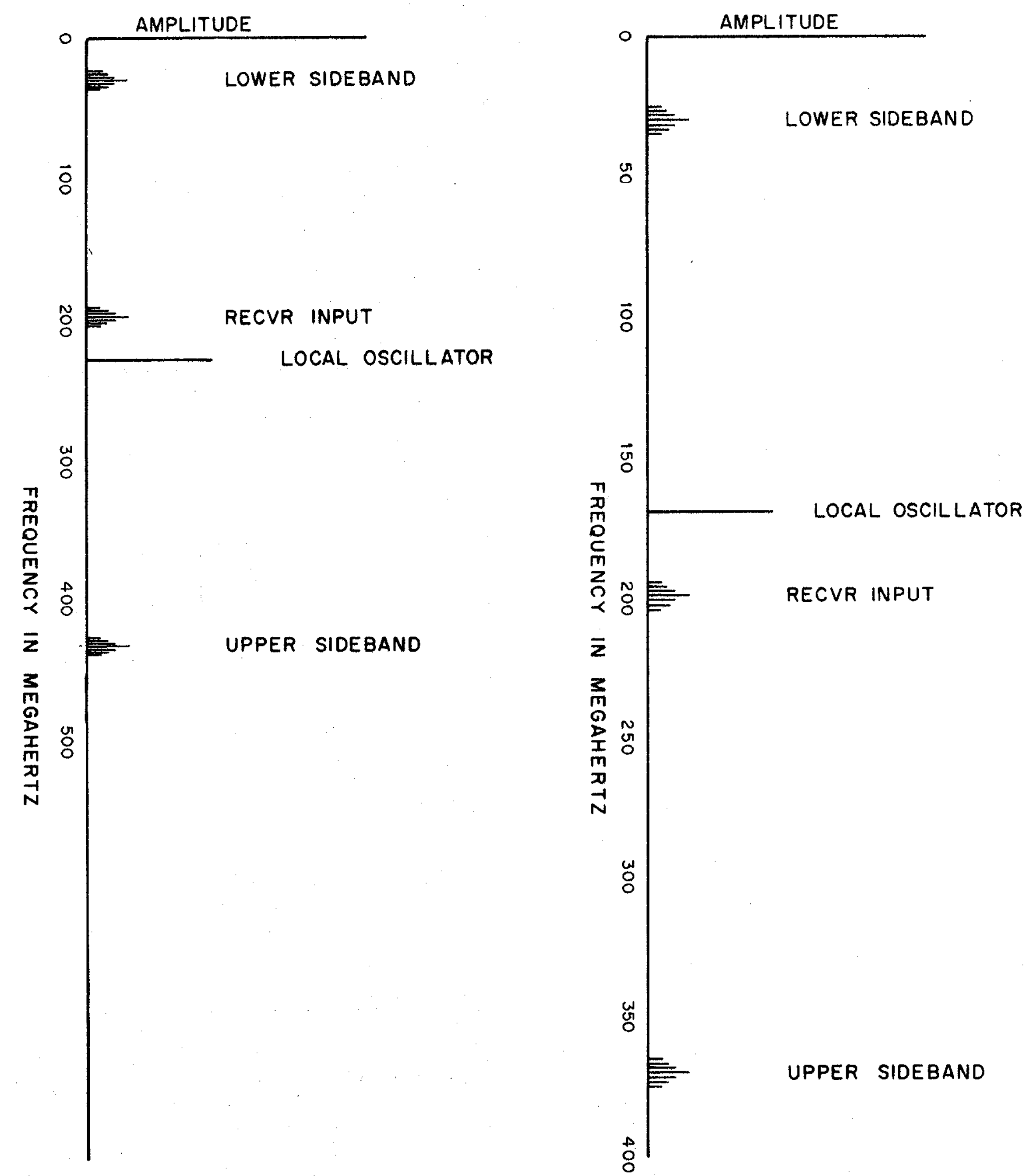


Figure 10-10. Mixing Products.

IF signal. Noise content of the local oscillator signal will also contribute noise to the IF signal. These contributions in noise are especially important if the front end contains no amplifier and the first stage of amplification is the IF amplifier.

The amplitude of the local oscillator must remain constant. If it contains variations in amplitude, these variations will also appear in the sidebands generated and could contribute noise in the output of the receiver.

Frequency stability of the oscillator is an important requirement. Receiver frequency stability depends on it. If the local oscillator changes frequency, then the sideband products of the mixing action will no longer be centered at the IF. This can cause portions of the signal to be outside the passband of succeeding tuned circuits. This would cause the loss of a portion of the intelligence or distortion of the recovered signal. Short term (rapid) frequency changes in oscillator frequency would result in new FM products in the IF signal and would be detected as noise or distortion in the recovered signal.

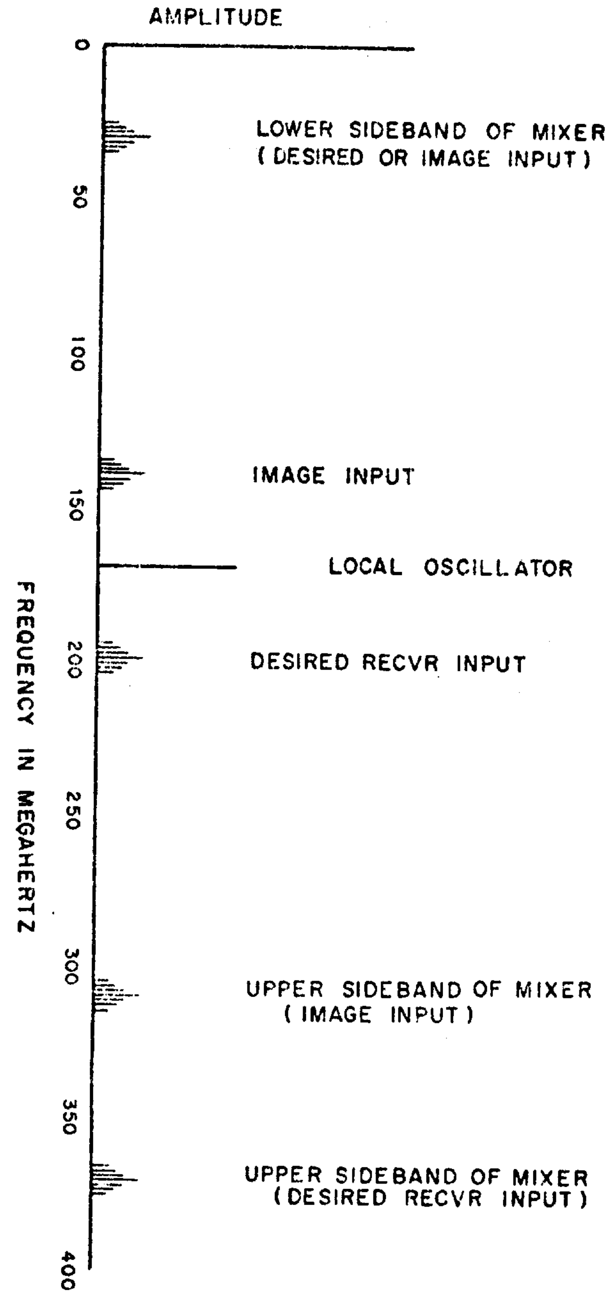


Figure 10-11. Image Frequency.

The mixer is required to present acceptable frequency-amplitude characteristics to three different signals: the input signal, the local oscillator signal, and the IF signal. Let's consider these in the order in which they are listed.

(a) The input to the mixer must have a flat frequency-amplitude response to the receiver input signal. It also must have sufficient bandwidth to pass the carrier and the important sidebands; however, to reject image frequencies, the bandpass must be sufficiently limited to prevent these undesired signals from reaching the mixer.

(b) A second input to the mixer is the local oscillator signal. The input should be tuned to the desired local oscillator frequency so that undesirable harmonics and other extraneous noise from the oscillator are not allowed to enter the mixer.

(c) The output of the mixer will be tuned to the IF. The frequency-amplitude response should be flat and the bandwidth must be just sufficiently wide to pass the IF signal. By limiting the bandwidth of the output in this manner, undesirable noise products from the mixer are prevented from entering the IF amplifier section.

(3) IF Amplifier. Now let's turn our attention to the need for an IF amplifier section to receive amplitude modulations. Usually, receivers are required to tune to signals over a portion of the frequency spectrum rather than to receive only one frequency. Since one of the requirements for a receiver is gain, this creates a problem. It is difficult to design an amplifier that is tunable and that works equally well over a broad portion of the frequency spectrum. Depending on gain requirements, the amplifier will probably use several stages of amplification. If the amplifier does require several stages of amplification, a second problem arises. To obtain the best gain from an amplifier, its tuned circuits must be resonant to the frequency to be passed. By the same token, in a multistage amplifier, best gain is achieved when the tuned circuits of each amplifier are resonant to the frequency to be amplified. Unless one control is used to tune all stages, the tuning of a multistage amplifier is a time-consuming job. If one control is used, then the tuning of all stages must track; that is, the change in resonance frequency, caused by changing the common tuning control, must be the same for every tuned circuit in the amplifier. As the number of circuits tuned increases, the problem of tracking becomes more difficult.

An easier method of amplifying a broad range of individual frequencies is available. By converting these individual frequencies to one frequency and designing a "specialized" amplifier for that frequency, extremely high gains for each frequency can be attained. Designing a stable high-gain amplifier with the desirable bandwidth for one single frequency is vastly easier than designing one for multiple frequency use. This method of frequency amplification is the IF method.

The function of the IF amplifier is to raise the level of the IF signal to a point where it will drive the demodulator. The overall receiver bandwidth is determined by the IF bandwidth. The prime factors of gain frequency and bandwidth will be discussed.

Amplification becomes increasingly difficult as frequency increases. For this reason, the IF selected usually will be lower than the receiver input frequency. The exact value varies widely; however, 70 MHz is a common value used in the wideband receiver.

The IF amplifier can be designed for a very high gain. For this reason, this portion of the receiver is selected to provide the majority of receiver gain. It must provide sufficient gain so that adequate signal levels are provided to the demodulator. The amplifier usually consists of several stages. By using this method, the gain requirement of any one stage can be decreased. This limiting of gain in individual stages makes it easier to achieve the required bandwidth.

Some means of controlling the gain of the IF amplifier is usually incorporated in the receiver. This is necessary because of changes that occur in RSL. These changes in level are caused by changes in the transmitting medium that result in more or less of the transmitter output reaching the receiver input. This control is required so that a relatively stable receiver output may be maintained for a wide variety of input signal levels. As RSL increases, IF amplifier gain is decreased. As RSL decreases, IF amplifier gain is increased. These changes in gain are accomplished automatically; hence, this action is AGC. An additional benefit from AGC is that it can prevent amplifier overloading by extremely strong signals and thus prevent distortion.

AGC also results in less noise in the receiver output. The presence of a signal in the receiver results in some AGC action. This action is to reduce the gain of the IF amplifier.

The IF amplifier can also be designed for a specific bandwidth. The bandwidth that the IF amplifier must possess is determined by the bandwidth of the signals we desire to receive.

For optimum reception, the amplifier will have sufficient bandwidth to accommodate the carrier and important sidebands. Beyond this width, the gain of the amplifier should rapidly decrease so that out-of-band noise and signals are rejected. Other portions of the receiver will be somewhat wider in bandwidth than this; therefore, the IF amplifier usually determines receiver bandwidth and selectivity.

This amplifier will also have a frequency-amplitude response. This response must be flat as in other receiver portions and will prevent distortion of the amplitude relationships of the sidebands in the IF signal.

The IF amplifier performs several important functions in the receiver. This unit:

(a) Contributes the majority of receiver gain.

(b) Determines the receiver bandwidth.

(c) Provides signals of proper level to the demodulator.

(d) Provides AGC for receiver.

(4) Demodulator. The next portion of the receiver is the demodulator. This unit is the device that actually recovers the intelligence from the frequency-modulated input signal. It translates the frequency variations in the IF signal to the original modulating signal. It consists of an amplitude-limiting device and a frequency discriminator. Let's consider their operation in that order. The purpose of the amplitude-limiting device is to remove amplitude variations from the IF signal. During the period that the signal is in space between the transmit and receive antenna, it can pick up amplitude variations because of noise, fading, interfering signals, etc. If these variations in amplitude are allowed to enter the discriminator, they will be detected and added to the recovered signal as interfering noise. Since the original modulating intelligence is carried as frequency variations in the IF signal, these amplitude variations can be limited or clipped from the signal. Using this technique reduces the noise that would otherwise appear in the output of the receiver. Figure 10-12 shows this clipping action. Notice that the frequency variations representing the original modulation are still present after the amplitude variations have been removed.

After the signal has been amplitude-limited, it is applied to a discriminator. This unit converts the frequency variations of the signal to a wave that represents the modulating intelligence. If the radio system is very efficient, this wave will be essentially identical to the original modulating wave.

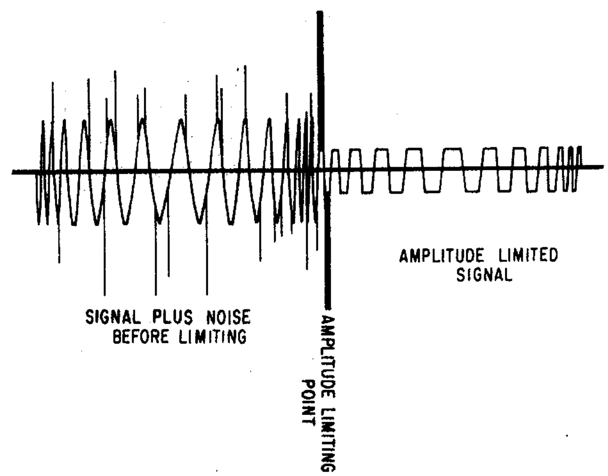


Figure 10-12. Clipping Action.

In the original modulation step, one-half of the modulating intelligence cycle caused the transmitter output frequency to increase above the unmodulated or rest frequency. The amount of frequency change was proportional to the amplitude of the modulating signal.

When it arrived at the modulator, the opposite half-cycle of intelligence caused the output frequency of the transmitter to decrease below rest frequency. The discriminator must then produce one polarity of signal output for signals above rest frequency. Conversely, the opposite polarity of signal output must be produced for signal below rest frequency. The amplitude of these output signals will be in proportion to the amount of frequency change in the received signal.

Since the discriminator consists of tuned circuits which will have a certain bandwidth characteristic, there are limits to the frequency changes it can demodulate linearly. Figure 10-13 shows the response curve for a discriminator. In this example, output voltage is zero when input is at center frequency. With an input signal above center frequency, the output goes positive. When the input signal is below center frequency, the output is negative. This arrangement is called a positive slope discriminator. If the polarity of the outputs for the frequency changes listed were reversed, it is referred to as a negative slope discriminator. The output between points A and B is the linear portion of the discriminator output.

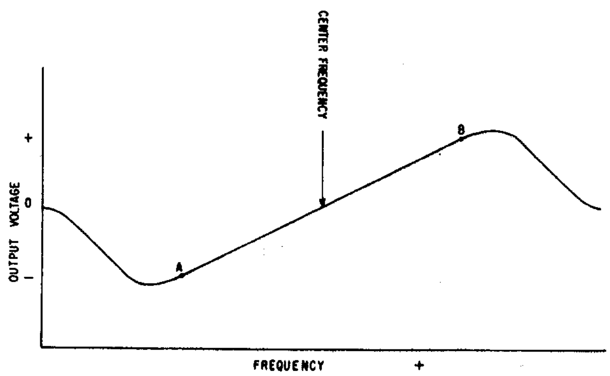


Figure 10-13. Discriminator Output Curve.

Although a discriminator is recovering an intelligence signal from a frequency-modulated carrier, the results of carrier swings outside the linear portion of the discriminator's response curve will be amplitude distortion. Figure 10-14 shows another discriminator output curve.

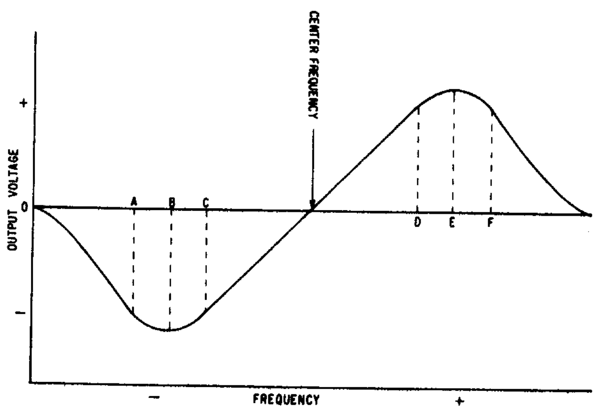


Figure 10-14. Discriminator Output Curve.

As long as the input frequency changes are between points C and D, a given amount of frequency change gives a corresponding change in output; however, between B-C and D-E, a larger frequency change is required to obtain the same given amount of change in the output. In this portion of the curve, the discriminator becomes non-linear. Changes in areas A-B and E-F actually give an inverted output. Change from center frequency to point E causes a continuous increase in the positive output. Change from point E to point F, on the other hand, causes a decrease in the positive output.

Figure 10-15 shows the output wave of the example discriminator and how this wave shape changes as the bandwidth of the incoming signal is increased. A represents the output for a signal with bandwidth limits that do not exceed points C and D on figure 10-14. No distortion is present in this output wave. B represents the output for a signal with a bandwidth that exceeds points C and D.

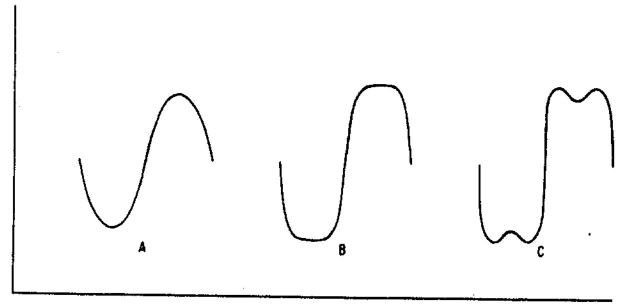


Figure 10-15. Discriminator Output.

Notice the flattening caused in the peaks of the wave. This distortion occurs because of discriminator non-linearity between points B-C and D-E. C represents the output for a signal with a bandwidth that exceeds points B and E. The same flattening of the wave shape occurs, plus the actual depression of the center of the peak. This is caused by the reversal of discriminator response between points A-B and E-F, compared to other portions of the curve.

The distortion of the signal was the greatest in example C. The introduction of a new frequency component is very evident in the peaks of this wave. The cause of increasing distortion was operation of the discriminator in areas of increasing non-linearity. This, in turn, was caused by the bandwidth of the received signal being too great for the discriminator.

Another distortion of output of the discriminator occurs if the rest frequency of the incoming signal does not coincide with the center frequency of the discriminator. Even if the discriminator has the proper bandwidth for the incoming signal, improper centering can cause one extreme of the input to operate in a non-linear portion of the discriminator (figure 10-16).

Points B and D indicate the limits of linear discriminator response. The input signal is between points A and C. The portion of input signal between B and C gives an undistorted output. The portion between A and B, however, is in the non-linear portion of the discriminator output and produces a distorted output. In this case, the negative peaks only are distored (figure 10-17).

If the input were moved up in frequency or to the right on the chart beyond the linearity limit at point D, then the opposite portions of the output - the positive peaks - would be distorted (figure 10-18). Generally, non-linearities result in amplitude distortion of the signal. The result of these amplitude distortions also can be the addition of spurious frequency components to the signal. The spurious frequencies may be considered as noise which tends to mask the intelligence signal. The sum of all these changes is degradation of the reproduced signal or a reduction in the fidelity of the system.

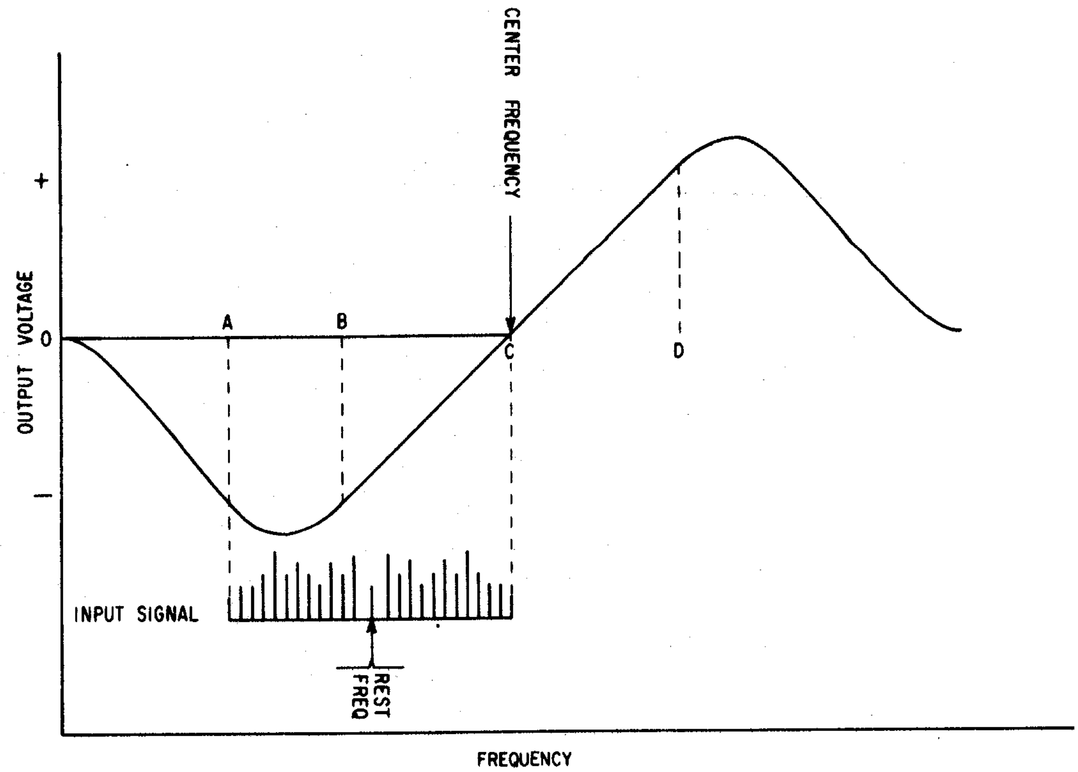


Figure 10-16. Improper Discriminator Centering.

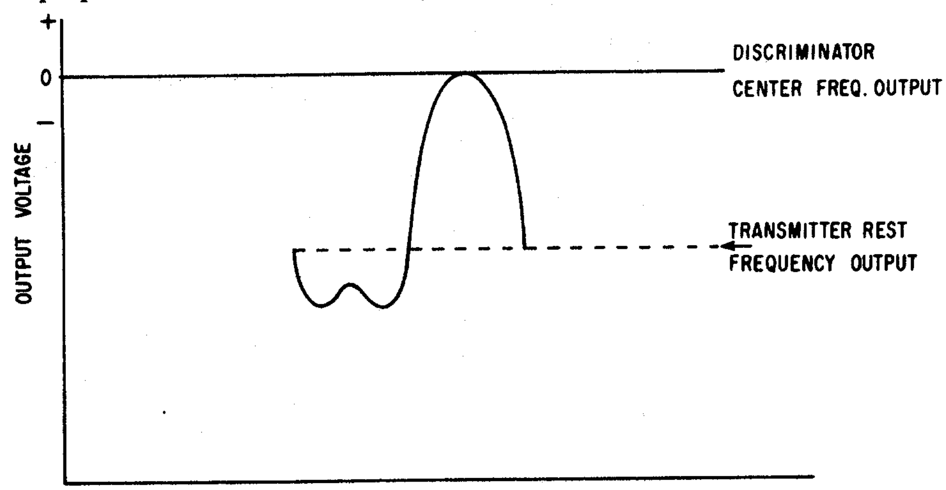


Figure 10-17. Frequency Below Discriminator Center Frequency.

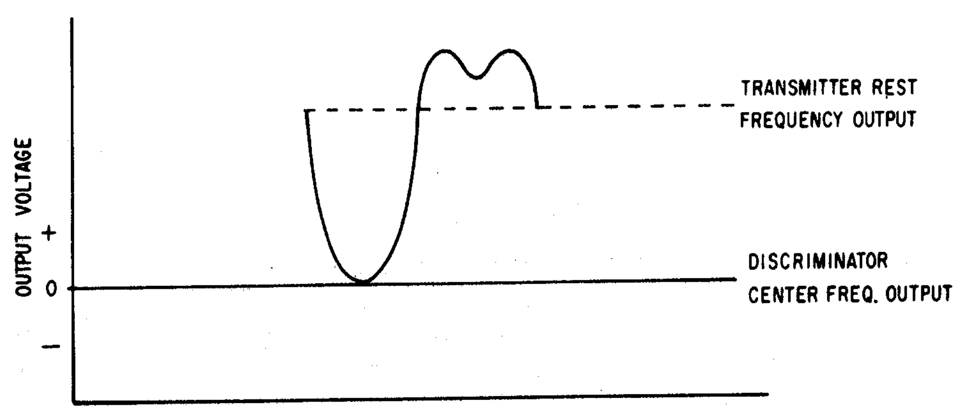


Figure 10-18. Frequency Above Discriminator Center Frequency.

(5) Output Section. The discriminator output, then, is the recovered intelligence. This intelligence signal is fed to the output section of the receiver. The purpose of this section is to increase the level required by the using device. To accomplish this task, the output section must possess the following familiar characteristics. It must have sufficient gain to accomplish the required signal level changes. The frequency-amplitude response must be flat and have sufficient bandwidth to pass the entire signal. Since this is an output device, for optimum transfer of power, it must be able to match the impedance of the using device. The output section will be designed to provide these characteristics so that a signal of acceptable quality and amplitude is provided to the using device.

d. Wideband Communications Receivers. Let's turn our attention now to the wideband communications receiver. Comparing this type of receiver to a basic receiver reveals some additional functions. For purposes of explanation, we will consider that the wideband communications receiver contains the same basic building blocks as the basic receiver. We will find some new functions peculiar to the wideband communications receiver performed as additional functions by the various blocks of the wideband receiver. Figure 10-19 is the block diagram of the wideband receiver with the new functions indicated.

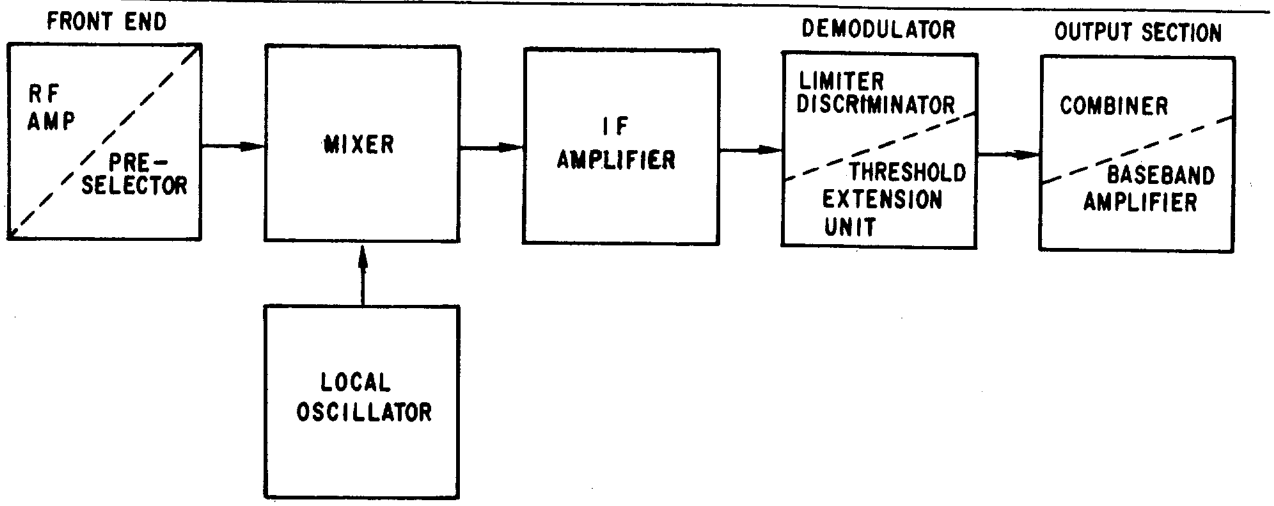


Figure 10-19. Wideband Communications Receiver.

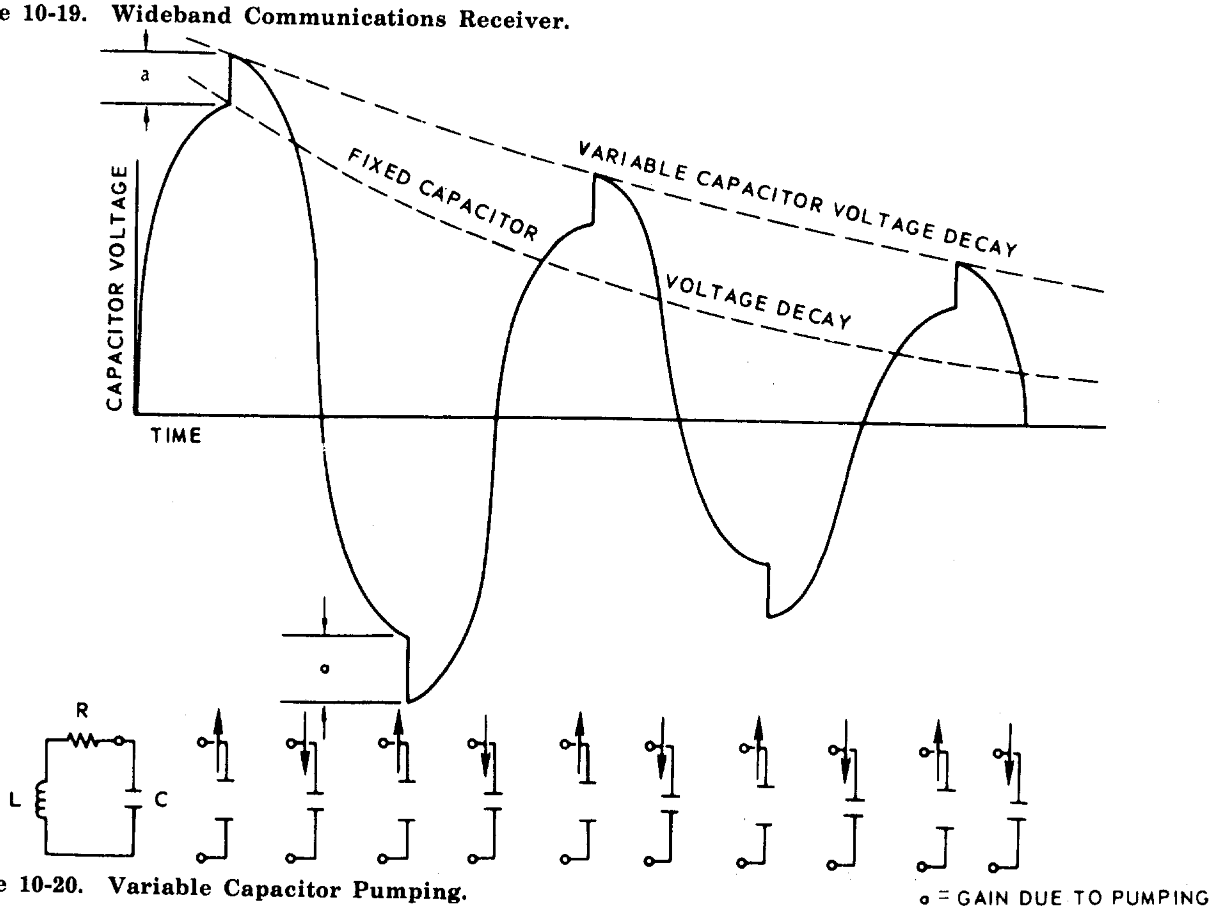


Figure 10-20. Variable Capacitor Pumping.

(1) Low Noise RF Amplifiers. In the wideband communications receiver, the front end performs the same function that it performed in the basic receiver. In many of the communications links, however, very difficult radio paths are encountered. The results of these paths are very weak signals at the receiver. You will recall that the noise generated in the front end of the receiver was the limiting factor in receiving very weak signals. For this reason, some wideband communications receivers are equipped with an extremely low noise front end which also contains an RF amplifier. This amplifier will provide adequate signal levels for the heterodyning process that is to follow. The amplifier used in these front ends usually is one of two different types: the parametric amplifier and the tunnel diode amplifier. Both of these amplifiers exhibit very low noise characteristics and are, therefore, suitable for this application. Let's take a look at the operation of a parametric amplifier first.

Parametric amplifiers produce signal gain by varying a circuit parameter (inductance or capacitance) with a source (pump) frequency. Instead of using DC power as does a conventional amplifier, the parametric amplifier uses AC power to build up signal power. The parameter we will discuss is a capacitance which is provided by a reverse biased diode. Let's see now how a capacitance works and how we can obtain signal gain from this component.

Consider the RCL resonant circuit shown in figure 10-20. If the values of the circuit parameters are fixed and a single input pulse applied, the circuit will oscillate. Suppose now that when the capacitor is fully charged, the plates are instantaneously pulled apart a slight amount. Since the capacitor plates have opposite charges, work is required to separate them. The work done increases the energy in the electric field between the plates. When a charge is placed on a capacitor, the potential across the capacitor can be determined by the equation:

$$E = \frac{Q}{C} \qquad \text{Eqn 10-2}$$

where

E = the potential across the capacitor in volts
Q = the number of coulombs of charge
C = the capacitance in farads

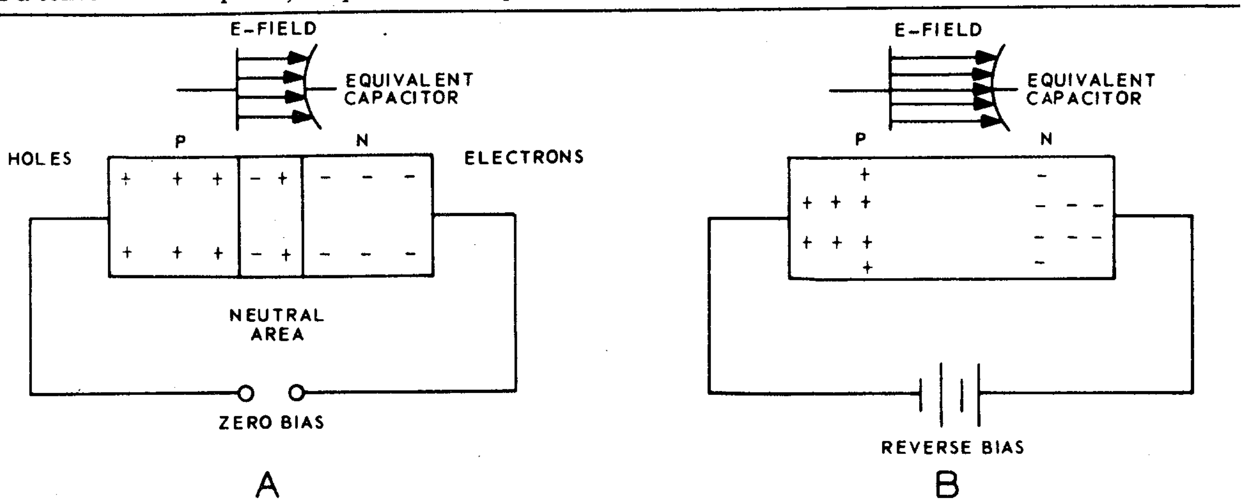


Figure 10-21. PN Junction Diode Varactor.

Since the charge on the capacitor cannot change instantaneously, the sudden decrease in capacitance causes a corresponding increase in the capacitor voltage. A quarter of a cycle later, when the voltage is zero, the plates of the capacitor are not charged; therefore, the plates can be pushed together without taking energy from the circuit. This alternate pumping action is repeated at each quarter-cycle. Regarding the capacitor voltage decay, note that the oscillations die out less rapidly when the circuit is pumped; thus, by pulling capacitor plates apart and pushing together at twice the resonant frequency, energy is fed into the resonant circuit. The net result of this pumping is equivalent to reducing the resistance of the circuit. In effect, then, we have added a negative resistance to the circuit.

Note that the phase of the pump relative to the charge on the capacitor plates must be chosen properly; otherwise, if the pump phase is changed so that the plates are drawn together when they are fully charged, the circuit will give up energy and the oscillations will decrease at a faster rate than when the capacitor is fixed; thus, pumping must be correctly phased with the incoming signal if gain is to be obtained.

Since mechanically pulsating capacitors are impractical at radio frequencies, a device called a varactor is used. The varactor is a PN junction diode which acts like a variable capacitor when a variable bias is applied to it. As shown in figure 10-21, the varactor under the condition of zero bias has a neutral area between the N-type and P-type material. This area can be considered as a dielectric and the region on each side of the neutral area can be considered as the plates of a capacitor.

When reverse bias is applied, as shown in detail B, the free charges are concentrated by the attraction of the holes to the negative terminal and the electrons to the positive terminal. Additional electrons and holes from the neutral area are also added to the equivalent capacitor plates; thus, the density of the free charges is increased, which in turn increases the neutral area and the potential difference between the plates. Capacitance decreases due to the increased separation of the plates.

Figure 10-22 shows how the varactor capacitance varies with changing bias voltage. Note that the change in capacitance is non-linear; this is a very important factor in parametric amplifier operation. Notice also the similarity between the varactor characteristic curve and the familiar Eg-Ip curve of vacuum tube amplifiers.

The graph in figure 10-23 shows the same capacitance curve plus a current curve. The dotted curve shows how current varies with voltage at low frequency. The back voltage at which the current rises sharply is called breakdown voltage (V_B). This is an important parameter because we must operate between zero voltage and V_B in order to avoid drawing current and to keep the capacitance essentially noiseless. If we bias the diode approximately halfway between zero and V_B and apply a sinusiodal voltage of amplitude $V_B/2$, we get the largest possible variation of capacitance, as shown in the illustration.

We have seen that we can increase energy in a resonant circuit by changing the capacitance in the proper way at the proper time. We have also seen that the semiconductor junction diode provides a convenient means of varying the circuit capacitance. There is just one hitch - the pump signal (which varies the capacitance) must be in exactly the right phase relationship with the incoming signal; otherwise, we may get a loss instead of a gain.

This is a serious problem, because even if we could assure a stable pump signal of just twice the input signal frequency, we would have difficulty getting and maintaining the critical phase relationship between the pump signal and the incoming signal. The problem is not an unsolvable one, however. The phase sensitivity of the pump signal in a parametric amplifier can be overcome by introducing a third frequency into the circuit by heterodyning the input signal with the pump signal. The new frequency (the difference between the pump frequency and signal frequency is normally used) is called the idler frequency.

A simplified diagram of a parametric amplifier is shown in figure 10-24. One resonant circuit is at the signal frequency (F_S) and the other is at the idler frequency (F_I). The two resonant circuits are coupled by a varactor diode (note symbol). When pump power is applied, the diode capacitance varies at the pump frequency (F_P). The pump frequency mixes with the signal frequency, producing the idler frequency, which is the difference between the signal and pump frequencies ($F_I = F_P - F_S$). The idler signal functions as a phase lock, so that the addition of idler and signal voltages ($V_I + V_S$) results in a complex waveform which is at zero potential when the pump signal forces the capacitor plates to close. Consequently, energy is put into the circuit regardless of the phase relationship between the signal frequency and the pump frequency.

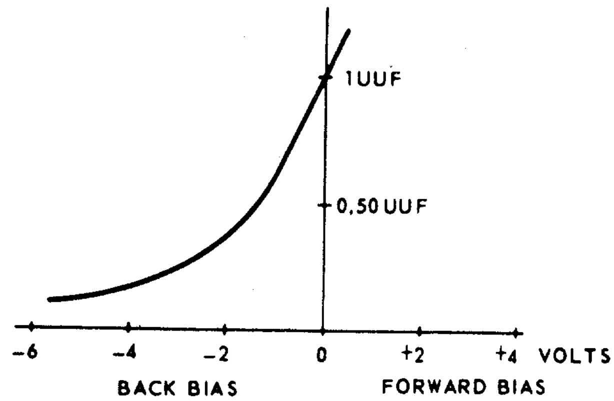


Figure 10-22. Capacities Versus Voltage of a Typical Varactor Diode.

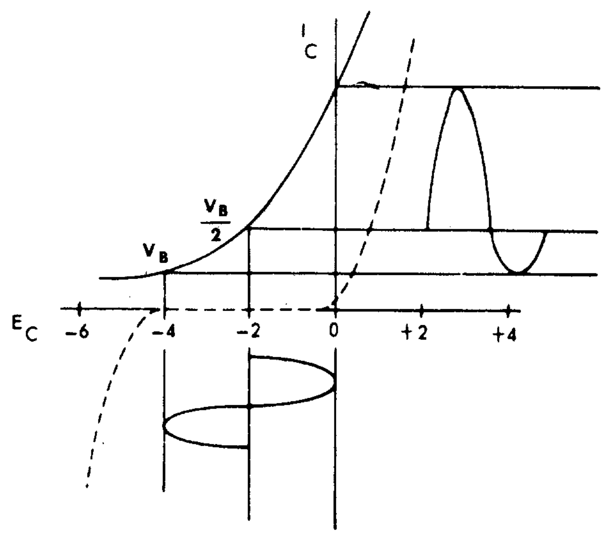


Figure 10-23. Capacitance and Current Versus Voltage Curve.

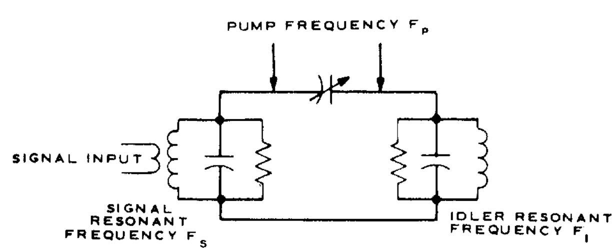


Figure 10-24. Equivalent Circuit for a Parametric Amplifier.

In the circuit just described, the pump frequency could be changed without upsetting the circuit action. If, for example, the pump frequency were increased, the idler frequency would also increase since it is the difference between the pump and the signal frequencies (the pump frequency is always higher than the signal frequency). The new idler frequency would still function as a phase lock. The addition of the idler and signal voltages would produce a resultant voltage which would be at or near zero when the diode capacitance increased and at or near maximum when the diode capacitance decreased; hence, a net energy gain is pumped into the circuit, regardless of the phase relationship between the signal frequency and the pump frequency. It follows, then, that the signal frequency can also be changed without upsetting the circuit action.

Another low noise RF amplifier used is the tunnel diode. The parametric amplifier used a variable capacitance to obtain low noise signal gain. The tunnel diode employs a different quality to achieve the same result. Let's see how the tunnel diode accomplishes its electronic feats. The essential difference between tunnel diodes and conventional diodes is the amount of doping (amount of impurities added to the semiconductor). The tunnel diode is very heavily doped, compared to the conventional diode; therefore, the conductivity of the tunnel diode is much greater (over 1000 times more conductive) and it is not appreciably affected by environment or temperature.

Because the tunnel diode is heavily doped, the barrier region is very narrow (about 10^{-6} inches) and, therefore, has a high barrier potential; electrons tunnel through the junction even though they do not have enough energy to surmount the potential barrier of the junction. Tunneling is inexplicable in terms of the usual electron theory; however, it can be explained in terms of quantum mechanics.

Although it is far beyond the scope of this text to go into the exact theory of quantum mechanics, we can draw certain conclusions that will allow us to get a basic understanding of how the tunnel diode functions. A basic principle is: *Electrons will penetrate an ultra-thin barrier - they cannot otherwise penetrate - if there are vacancies (holes) on the opposite side of the barrier at the same electron energy level as the penetrating carriers.*

The tunnel diode will have maximum tunneling taking place with no bias applied. This amounts to being in the breakdown region with zero bias. If a reverse bias is applied, reverse current will increase with voltage (as shown by the characteristic curve shown in figure 10-25A) because of tunneling. The reverse bias causes the energy levels of the majority carriers to be lower; thus, the increase in reverse current with increasing reverse bias is due to the minority energy levels being increased and the breaking away of minority electrons from their bonds. An equal number of holes is also formed and these join the original holes that were present in the P-material. This production of holes and excess electrons occurs only because the semiconductor is heavily doped; therefore, a heavily-doped semiconductor with reverse bias acts only like a good conductor.

Notice what happens if forward bias is applied; the characteristic curve shown in figure 10-25A shows how the current rises to a peak point and then declines to a valley point as forward bias is increased. The decrease in current with an increase in forward bias is a result of the change in the amount of tunneling taking place. Keeping in mind that electrons will penetrate an ultra-thin barrier if there are vacancies on the opposite side at the same electron energy level, you will see how changes in forward bias affect the amount of tunneling and how negative resistance occurs (the bias of tunnel diode amplification).

Let's analyze the tunneling action as shown by the typical characteristic curve in figure 10-25A. Figure 10-25B through 10-25F shows the energy levels within the diode. Notice that electrons are shown by the symbol (-) and holes are indicated by the symbol (o). Although only a few holes and free electrons are shown in the P and N materials, keep in mind that there are millions. Notice that, for the points along the curve, there is a graphic representation below showing the quantum energy levels. The relative energy levels are indicated by the location of the holes (o) and free electrons (-) within the individual drawings. (High-energy level at the top and low-energy level at the bottom.)

The first point we will consider is figure 10-25A is point A. You will note that this point has zero bias and no external current flow (refer to figure 10-25B). Even with zero bias, tunneling is fully effective because of the very heavy doping; thus, the diode is essentially in a breakdown condition due to the large number of carriers (electrons and holes) tunneling through the barrier and combining. Since these carriers are combining with each other, the action is neutralizing, causing no external current to flow. In other words, the holes and the electrons cancel each other out and no external current can flow.

Figure 10-25C shows a tunnel diode junction with a small amount of forward bias (approximately 50 mv). The negative potential applied to the N-type material raises the electron energy level of the electron carriers. This would have no effect in an ordinary diode because .5 to 1-volt forward bias is required to overcome the barrier allowing current to flow; however, because tunneling is fully effective in this case, substantial current flow is measured externally in the direction shown by the arrow. This point is the peak current point. Beyond this point, current decreases along with tunneling, as explained below.

Figure 10-25D shows a tunnel diode with the forward bias increased further - to point B on the characteristic curve. Let's discuss what is occurring at this point. Notice that, from the peak point to the valley point, current decreases as forward bias increases; therefore, point B lies within the negative resistance region. Why did the current decrease with an increase in forward bias? The decrease in current, even though the bias increased, is a result of tunneling being reduced. As the forward bias increased, the negative potential applied to the N-type material caused the electrons to be at a higher energy level; therefore, there are fewer electrons at the same energy level as the holes in the P-type material. Consequently, electron tunneling is reduced

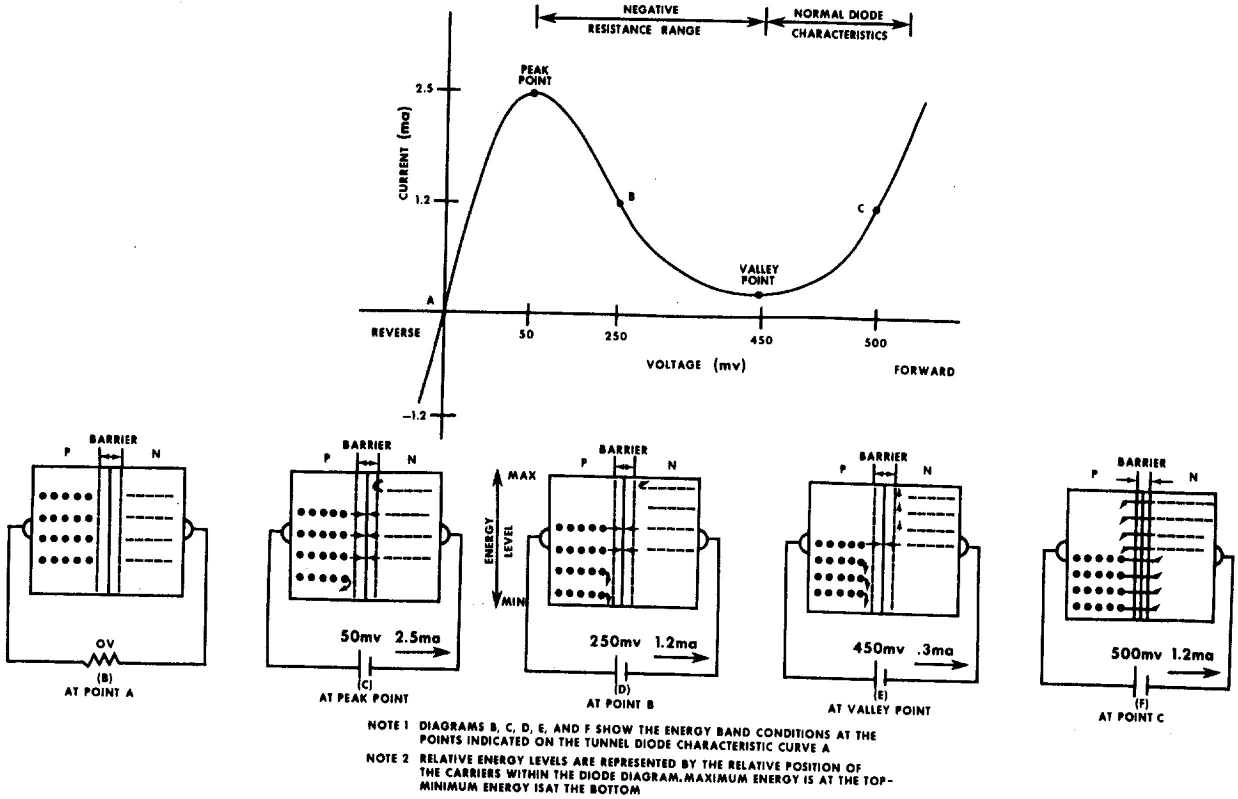


Figure 10-25. Tunnel Diode Operation.

causing the external current flow to reduce to point B. It is within this general range on the curve that the tunnel diode is biased when used as an amplifier or oscillator.

Figure 10-25E illustrates a tunnel diode with the bias increased until the valley point is reached. This point on the curve represents minimum current flow because electron tunneling is minimum. As the forward bias increased from point B to the valley point, the electron energy level of the electrons in the N-material was increased; therefore, tunneling decreased further because there were fewer electrons with the same energy level as the holes on the opposite side of the barrier. Beyond the valley point, current turns upward again and resistance becomes positive. The tunnel diode now behaves in a conventional manner because the electrons and holes have surmounted the barrier and tunneling has ceased.

Figure 10-25F indicates the electron energy level of the tunnel diode at point C on the characteristic curve. Notice that the electron energy has been raised to the point that tunneling is no longer possible because no electrons and holes are at the same electron energy level; however, there is current flow in the external circuit because the electron energy level of the carriers has been raised to a point where the barrier is surmounted and tunnel diode acts like an ordinary PN junction diode.

The tunnel diode, in the negative resistance region, provides a large change in current for a small change in voltage. Two circuit arrangements can be used to capitalize on this characteristic: the series and the parallel configuration. By inserting the tunnel diode in series with the load device, a voltage gain is produced. Inserting the tunnel diode in parallel with the load results in a current or power gain. The parallel configuration is used in some wideband receivers.

The advantage is the use of a parametric amplifier or a tunnel diode amplifier is that these devices generate very little noise and provide signal amplification. These characteristics make both amplifiers well suited for communications purposes.

(2) Mixer/Oscillator. The mixer of the wideband communications receiver provides the same function as the mixer in the basic receiver. Usually a solid state diode is used as the mixing element. This type device is used because of its simplicity and low operating temperature. This low operating temperature results in better noise characteristics and, therefore, enhances receiver operation.

The local oscillator in the communications receiver also performs the same function as the local oscillator in the general receiver. The frequency stability requirements placed on this oscillator are very stringent. AFC and crystal control are the methods used to obtain the required stability. These methods are discussed in paragraph 10-3b.

(3) IF Amplifier. The IF amplifiers in the communications receiver perform the same function that the IF amplifier performed in the basic receiver. The bandwidth requirement for this section is determined by the input signal, The input signal width, in turn, depends on the number of channels assigned to the system. So that the optimum bandwidth of the receiver can be matched to the channel requirement,

several IF amplifier modules with different overall bandwidths may be produced for a particular receiver. The IF amplifier that will satisfy the particular bandwidth requirements of a link will be the one installed.

(4) Demodulator. The demodulator in the wideband communications receiver provides the same functions that the demodulator in the basic receiver provides. In addition to amplitude-limiting and frequency discrimination to recover the baseband signal, a third function is performed in some receivers. This function is threshold extension. Threshold in an FM receiver has been defined as the point where receiver input signal is sufficient to provide a usuable output signal from the receiver. Extending a receiver's threshold means lowering the amount of input signal that the receiver requires to provide a usuable output.

When RSLs are well above threshold and signal fades do not exceed the FM threshold of the receiver, the receiver is able to provide a constant usable output. If, on the other hand, the received signal fades below the FM threshold of the receiver, the output of the receiver becomes unusable. On difficult radio paths, this fading condition is common. To overcome this difficulty and provide a usable receiver output a greater percentage of the time, a method to extend the receiver threshold is incorporated in the receivers used on such paths. The amount of extension varies from 5 to 10 dB.

One of the factors that determines the FM threshold of a receiver is its overall bandwidth. There is a direct relationship between this factor and threshold signal level, that is, if receiver bandwidth is decreased, the level of input signal required to produce FM threshold is also reduced.

Threshold extension can be accomplished by incorporating a device in the receiver that automatically reduces receiver bandwidth just before the input signals drop below FM threshold. The degree of extension depends on the bandwidth reduction imposed.

It was stated earlier that the receiver must have sufficient bandwidth to pass the carrier and the important sidebands and that the bandwidth was determined by the IF amplifier. This amplifier was just sufficiently broad enough to pass the carrier and the important sidebands. It would appear, then, that to reduce receiver bandwidth would mean to discard part of the important sidebands and to lose part of the intelligence.

For our system to retain its quality, something must be done to prevent this loss of information. If a means to reduce the input signal bandwidth is provided at the same time the receiver bandwidth is reduced, then loss of intelligence can be minimized.

With the facts of the last few paragraphs in mind, let's turn our attention to how threshold extension is accomplished. Figure 10-26 is the simplified block diagram of the threshold extension unit.

One input to the threshold extension unit is the IF amplifier output. This signal is applied to the mixer stage. The function of this mixer is the same as previous mixers we have considered. The second input to the mixer, however, is different from our previous discussions. In those applications, the local oscillator generated a fixed frequency. The oscillator in this unit is frequency-modulated by the extended baseband. The frequency changes produced by this modulation will be in the same direction that the incoming frequency is changing. For instance, as the IF input increases in frequency, the oscillator also increases in frequency. The amount of change in the oscillator, however, is less than the amount of change in the IF input.

These changes in oscillator frequency have an effect on mixer output. This effect is to reduce the amount of frequency change that occurs in the sidebands generated in the mixer. This can be seen more readily if we use some mathematical expressions. In an equation form, the input frequency (X) minus the oscillator frequency (Y) is equal to the lower sideband, or $X - Y =$ lower sideband. Now, with a 100 kHz increase in X and Y remaining constant, the lower sideband also increases 100 kHz. If, however, we increase X by 100 kHz and Y by 50 kHz, the lower sideband increases only by 50 kHz. The change in oscillator frequency effectively cancelled 50 kHz of the change in input frequency. A convenient way to express this relationship is that the change in sideband product frequencies is equal to the change in mixer input frequency, minus the change in oscillator frequency.

In our example, the change in oscillator frequency equalled one-half of input frequency change and resulted in decreasing the change in sideband product frequencies to one-half. This was an instantaneous case. If the oscillator continued to follow frequency changes in this ratio, the change in sideband product frequencies would also be only half of the input fre-

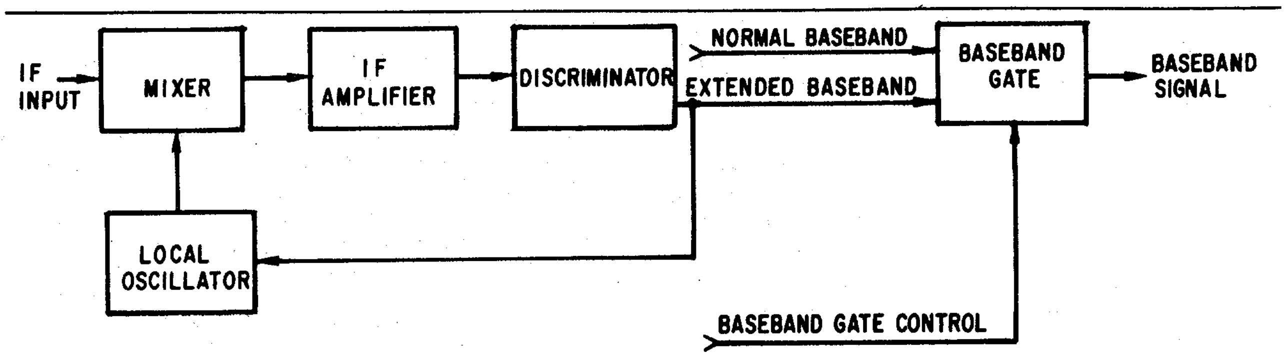


Figure 10-26. Simplified Threshold Extension Unit.

quency change. At the output of the mixer, we select the lower sideband of the mixing action. The bandwidth of this signal is one-half the bandwidth of the input signal. If we desire the compression effect of this circuit to be greater, we can achieve this by merely causing the oscillator to follow more closely the input frequency changes. For example, with the oscillator changing three-fourths as much as the input signal changes, the bandwidth of the signal selected from the mixer is one-fourth of the input signal bandwidth.

The process of making the oscillator follow input frequency changes solves one of our problems of threshold extension, that is, to reduce the input signal bandwidth. Now we can proceed to reduce the receiver bandwidth. This is accomplished in the IF amplifier of the threshold extension unit. The amount of reduction in IF bandwidth and, consequently, to the whole system, is dependent on the amount of extension required. Cutting receiver bandwidth in half improves FM threshold by 3 dB. With this figure in mind, we can compute the bandwidth reductions required for a desired threshold extension. Suppose we require a threshold extension of 6 dB. To achieve this, we must reduce input signal bandwidth and receiver bandwidth to one-fourth of their original values.

The IF section also performs an amplitude-limiting function and provides the discriminator with a bandwidth-compressed amplitude-limited signal. The discriminator, in turn, recovers the intelligence (baseband signal) from this signal; however, the receiver input can now decrease to the extended FM threshold before the discriminator output becomes unusable.

The normal baseband signal and the extended baseband signal are applied as inputs to the baseband gate. The function of this gate is to select one of the inputs as the receiver baseband signal. Control for this gate is derived from receiver noise output. The noise output is rectified and applied to the gate as a control voltage. At a preselected RSL, this control voltage actuates the gate. The function of gate actuation is to pass one of the signal inputs and block the other input. When RSL is above the preselected level, the normal baseband signal is allowed to pass through the gate and the extended baseband signal is blocked. When RSL is below the preselected level, the gate condition is reversed. The extended baseband signal is passed and the normal baseband signal is blocked. The preselected RSL will usually be several decibels above the non-extended FM threshold of the receiver. Choosing this point allows switching to extended threshold before receiver output becomes unusable.

At this point, we might ask why the receiver threshold is not extended at all times. The reason for not operating the receiver in this manner is that, theoretically, threshold extension causes no degradation of recovered baseband; however, in reality, small degradations occur. For this reason, the threshold extension unit is switched out of the signal path when RSL is above normal FM threshold value.

(5) Output Section. Signals from the baseband gate are passed on to the output section. The output section must provide a baseband signal of the proper power level and of the best possible quality to the device using this signal (a multiplex set or another radio). Providing proper impedance match is also a function of the output section.

Providing signals at the proper level and impedance matching are among the functions performed in the basic receiver's output section. Providing the best quality signals is also a function of the receiver. This is accomplished by assuring that the output section has a flat frequency-amplitude response and sufficient bandwidth. These factors must also be satisfied in the wideband communications receiver.

In wideband communications, we normally go one step further by incorporating path diversity to improve the reliability of the system. For this method to be effective, a means to combine the outputs of the several paths into one baseband signal is required. To perform this job, a combiner is usually incorporated in the output section of the wideband comunications receiver.

Combining may be accomplished by a selection action or by an addition action. Selection is the simplest form of combining and is performed by picking the best output from the receiver outputs presented to the combiner. A combining action of this type was used previously in the baseband gate of the threshold extension unit, with one or the other of two inputs being gated through. This is selection-combining in its simplest form.

A more complex form of combining is the maximal ratio or ratio-squared method. In this method, simultaneous addition of all receiver outputs is accomplished. Each receiver contributes a portion of its output to this addition. The amount any one receiver contributes to the combined total is determined by that receiver's RSL.

To see how and why the receiver output is controlled in this manner, let's review for a moment. For a given RSL into a receiver, a corresponding AGC level is produced. The function of this AGC level is to limit the receiver gain. With this level of AGC, then, the noise generated in the receiver is amplified by some factor. The result of these factors is a level of noise in the receiver output which is dependent on RSL. Suppose we decrease this RSL. AGC will also decrease, so receiver gain increases. Since gain increases, so will noise in the receiver output. From this example, we can see that RSL and output noise are inversely related, but the relation will be in porportion, that is, for a given change in RSL, an inversely proportional change in receiver output noise will occur.

Now, let's go back to our combiner. Since the noise in the receiver increases as RSL decreases, the output S/N ratio also decreases as RSL decreases. Another way to express this ratio is to say that the quality of the receiver output signal decreases. If output signal quality decreases, then the amount of that signal added to the combined output should also be decreased. If, on the other hand, the RSL increases, the opposite of the foregoing conditions would be true and more of the signal should be added.

By using the noise output to control the combining action, we can achieve the desired addition of the receivers' outputs. This noise is rectified to produce a negative voltage which is used to control combiner action. The negative voltage is applied to the combiner in a way to control the gain of one combiner amplifier. Figure 10-27 is a block diagram of the maximal ratio-combining method using two inputs.

Each amplifier receives a baseband input signal and a control voltage from a particular receiver. As the RSL for that receiver changes, the control voltage changes. This change causes the combiner amplifier gain to change and, thus, to provide more or less signal to the combined baseband output. A decrease in RSL causes a decrease in the combiner amplifier gain for that receiver. As a result, the receiver contributes a smaller portion of the combined baseband signal. Increasing RSL reverses this process and results in a larger contribution by the receiver to the combined baseband signal. The outputs of the combiner amplifiers all add to become the combined baseband signal.

The combined baseband signal is fed to the baseband amplifier. This unit provides the gain required to bring the combined baseband signal to the required level. It also presents the impedance-matching characteristics that are required for optimum power transfer. The baseband signal is now ready to be applied to the equipment that will use this signal.

(6) Pre-Detection Combiners. Pre-detection combining has inherent advantages over post-detection combining. First, the complexity of a diversity receiver is greatly reduced, resulting in increased reliability and lower costs. Instead of the usual four separate limiters, demodulators, and baseband amplifiers, pre-detection combining requires only one. Second, most pre-detection combiners require no operator adjustments or maintenance to optimize system performance or to offset any drift or aging effects. Third, pre-detection combining improves overall link performance, resulting in improved bit error rates and increased S/N ratios. Combining the signals prior to demodulation increases the average pre-detection S/N ratio, thus keeping the demodulator operating above FM threshold a higher percentage of the time under marginal signal conditions than will four separate demodulators, as in post-detection combining. Fourth, pre-detection combining accepts both FDM and TDM without modification or adjustment. A pre-detection combiner would not need updating for a system change from FDM to TDM except for minor filter changes to accommodate the new spectrum.

The purpose of the pre-detection combiner is to add in phase independent signals from four separate receivers in a multiple diversity system prior to demodulation. This type of combiner has an advantage over a post-detection combiner in that it has a higher average pre-detection S/N ratio than post-detection combining.

The combiner accepts input signals at the IF from each receiver. Due to the propagation medium, the signals arrive at each receiver with random phase and amplitude variations and cannot be summed directly. The pre-detection combiner removes the phase variation from each signal input, weights the amplitude in an optimum manner, and sums the signals together.

The pre-detection combiner works on a double mixing principle and a controlled positive feedback loop common to all channels. The first mixer process isolates the phase difference between channels and the second mixer removes the phase difference and provides the ratio squaring (maximal ratio) function. An AGC maintains the loop gain required for the regenerating loop. A simplified block diagram is shown in figure 10-28. The system functions as follows:

Each of the four inputs are designated as u, v, x, and w, which, for an FM signal, can be written as

$$x_1 = A_x(t)\ \mathrm{Cos}\ (\omega_o t + m(t) + \phi_x(t))$$
$$u_1 = A_u(t)\ \mathrm{Cos}\ (\omega_o t + m(t) + \phi_u(t))$$
$$y_1 = A_y(t)\ \mathrm{Cos}\ (\omega_o t + m(t) + \phi_y(t))$$
$$w_1 = A_\omega(t)\ \mathrm{Cos}\ (\omega_o t + m(t) + \phi_w(t))$$

where

$A_a(t)$ are time variant amplitudes

$\phi_a(t)$ are time variant phase angles

$m(t)$ is the modulation function.

The combined signal is Z, given by

$$Z = B\ \mathrm{Cos}\ (\omega_1 t + m(t) + a)$$

At the first mixing process of channel x, x_1 is mixed with Z which has been AGC-leveled to constant amplitude. The mixer forms the product x_1Z

$$x_2 = x_1 Z = \mathrm{Cos}\ (\omega_1 t + m(t) + a)$$
$$A_x\ \mathrm{Cos}\ (\omega_o t + m(t) + \phi_x)$$

with zonal filtering to accept only the difference frequency.

$$x_2 = A_x\ \mathrm{Cos}\ (\omega_2 t + a - \phi_x)\ ;$$
$$\omega_2 = \omega_o - \omega_1$$

Similarly, for the other channels

$$y_2 = A_y\ \mathrm{Cos}\ (\omega_2 t + a - \phi_y)$$
$$u_2 = A_u\ \mathrm{Cos}\ (\omega_2 t + a - \phi_u)$$

and

$$w_2 = A_w\ \mathrm{Cos}\ (\omega_2 t + a - \phi_w)$$

This first mixing process, it can be seen, has removed the modulation, isolated the phase difference ($a - \phi_A$), and carries the amplitude A_a. Most of the noise is stripped from this signal by narrowband crystal filters providing almost noise-free signals x_2, y_2, u_2, and w_2.

The second mixing process is used to remove the phase differences from the received signals and is given for channel 1 by

$$x_1 x_2 = x_3 = A_x\ \mathrm{Cos}\ (\omega_o t + m(t) + \phi_x)$$
$$A_x\ \mathrm{Cos}\ (\omega_2 t + a - \phi_x)$$

Although either sum or difference frequency may be used, for this example we select lower side where

$$\omega_1 = \omega_o - \omega_2$$

$$x_3 = A_x^2 \text{ Cos } (\omega_1 t + m(t) + a)$$

Similarly

$$y_3 = A_y^2 \text{ Cos } (\omega_1 t + m(t) + a)$$

$$u_3 = A_u^2 \text{ Cos } (\omega_1 t + m(t) + a)$$

$$w_3 = A_w^2 \text{ Cos } (\omega_1 t + m(t) + a)$$

Thus the phase difference terms, ϕ_a, have been removed, the signals amplitudes have been squared to provide the ratio-squaring function, and the signals may now be added to form the combined output, Z.

$$Z = [A_x^2 + A_y^2 + A_w^2 + A_u^2]$$

$$(\text{Cos } (\omega_1 t + m(t) + a))$$

In practice, each of the individual channels is commonly gain-controlled, rather than the sum of channels, to increase the dynamic range of the combiner.

Figure 10-29 is a complete block diagram of the pre-detection combiner.

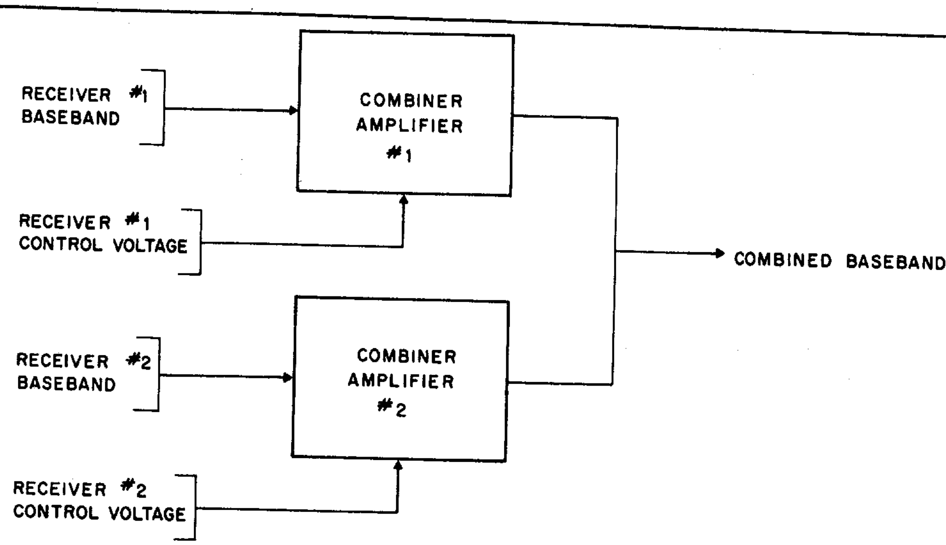


Figure 10-27. Simplified Two-Input Combiner.

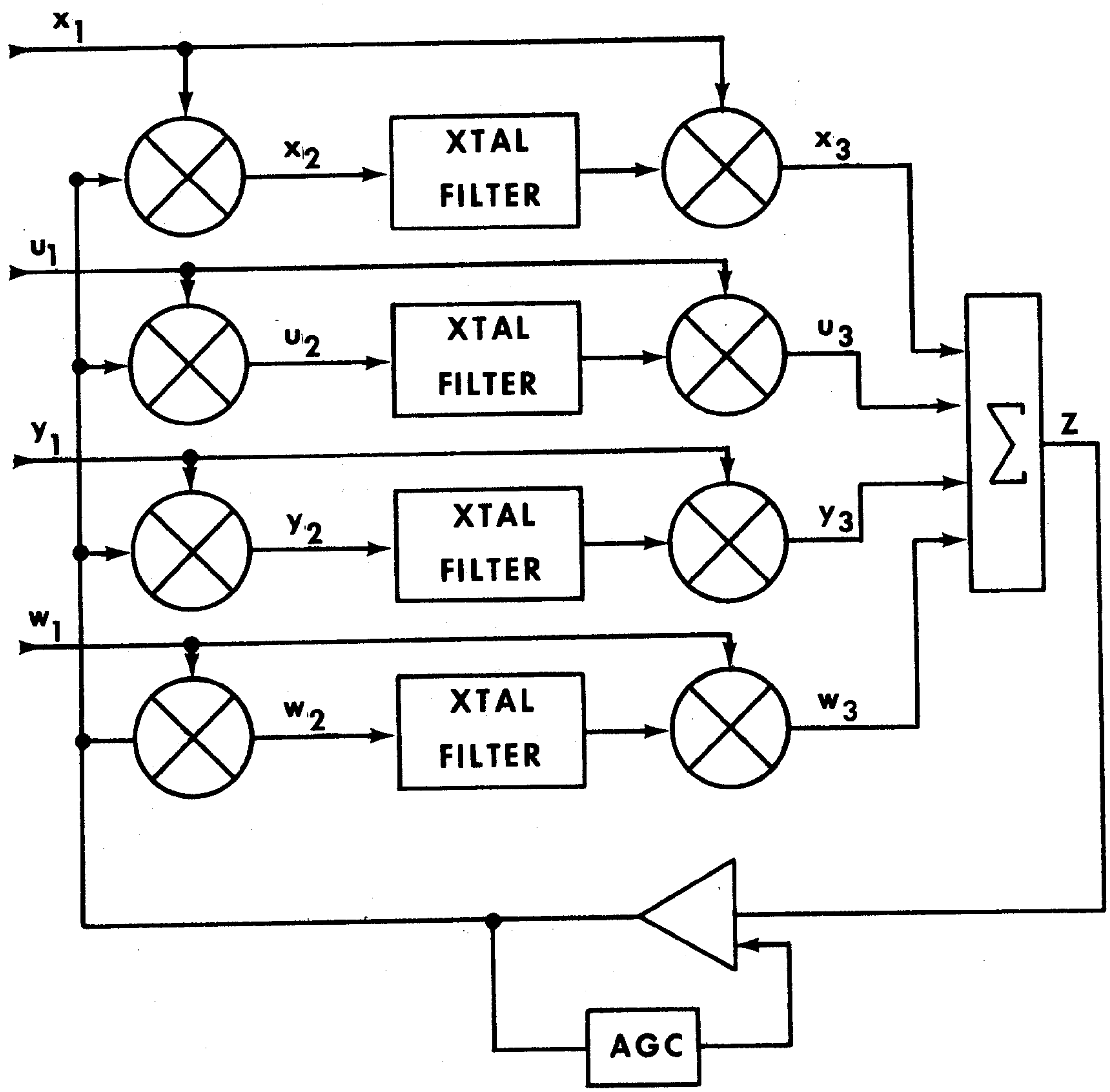


Figure 10-28. Pre-Detector Combiner.

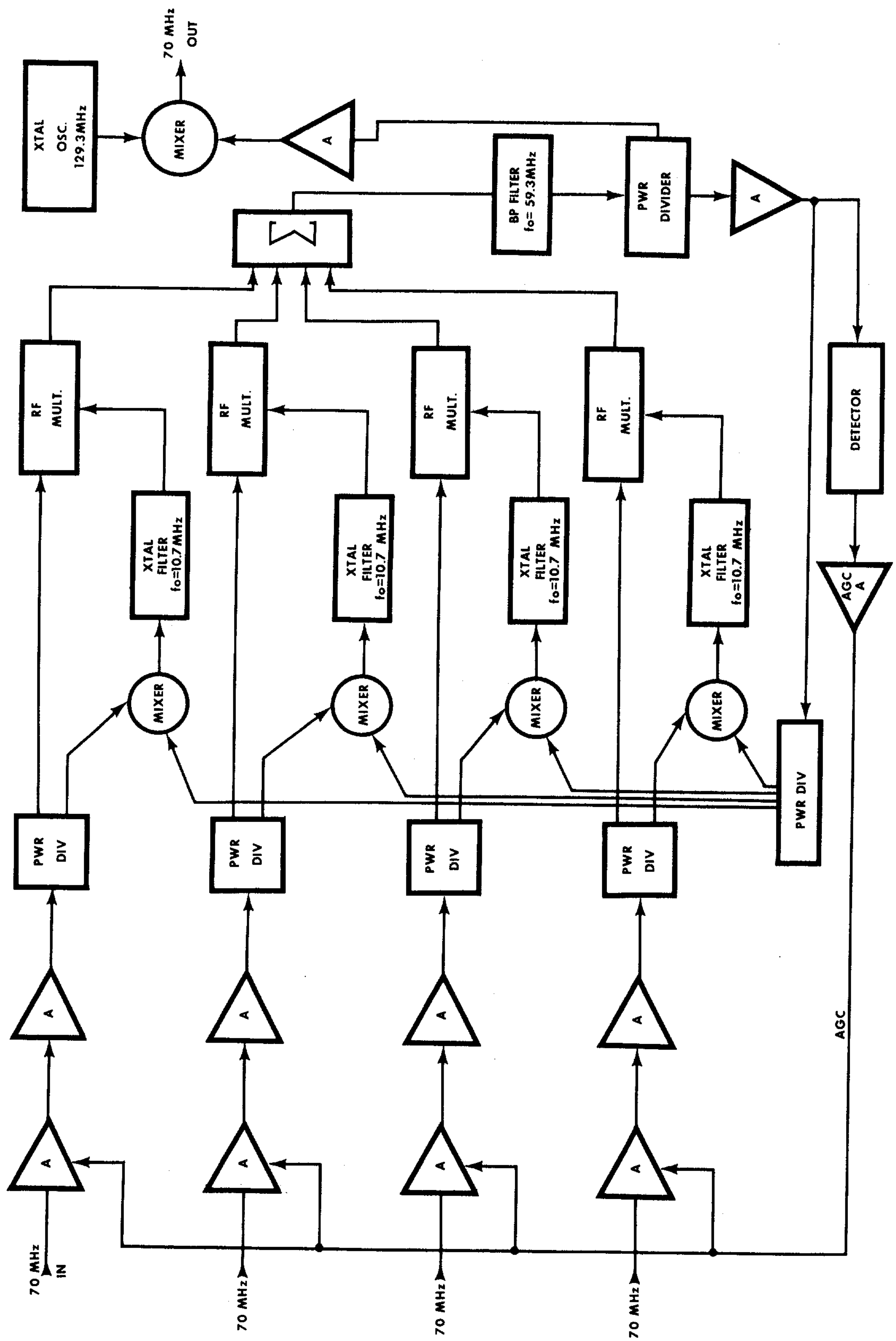


Figure 10-29. Pre-Detection Combiner Block Diagram.

Chapter 11

ANTENNA AND TRANSMISSION LINE SUBSYSTEM

11-1. General. The antenna and transmission line subsystem is the transformer between the medium carrying the radio energy and the transmitting and receiving equipment. The efficiency of the transmission subsystem is dependent on how well the transformation between the medium and the terminals is performed. The antenna itself acts as a converter between the energy in the transmission line and the energy in a free space wave. The transmission line carries the electrical energy between the antenna and the transmitting and receiving equipment.

11-2. Basic Antenna Theory. Before looking deeper into antennas used in wideband radio communications systems, some of their characteristics will be discussed.

Characteristics considered by a design engineer when installing an antenna and transmission line subsystem are: gain, radiation pattern, VSWR, power handling capability, and attenuation. All of these factors must work together in order for this subsystem to offer an efficient means of transforming energy into electrical signals. While the attenuation associated with a transmission line is an important consideration, the antenna losses are usually included in its overall gain value. The attenuation of a transmission line depends on its type, length, and frequency of operation.

When any two transmission line antenna components are attached together, a discontinuity is present at their junction. Any discontinuity causes some of the incident power to be reflected back to the source. The resulting reflections are a cause of distortion as well as power loss. The power loss can be calculated if the VSWR is known.

Power loss can also occur when otherwise good components of wrong impedance are attached together. Maximum power transfer occurs when all impedances are matched.

Just as a loudspeaker unit acts to match the high acoustic impedance of its throat to the low acoustic impedance of the auditorium, so one of the important functions of the antenna is to match the impedance of the electrical energy in the air to that of the transmission line. The transmission line, in turn, is impedance-matched to the receiver.

11-3. Radiation Patterns. The simplest antenna, or radiator, is a theoretical point source commonly called an isotropic radiator. In unobstructed space, such a source radiates energy equally well in all directions. This energy propagates as electromagnetic waves at a rate of approximately 160,000 miles (300,000,000 meters) per second. One millisecond after being radiated, the field of energy is, therefore, present at a distance of 186 miles from the source. This field would represent the surface of a sphere, be uniform in intensity, and have as its center the isotropic radiator source. Figure 11-1 represents the polar plot of directional response of an isotropic radiator.

In the polar plot, response is graphed as seen from above the antenna. Point A represents antenna position and the closed loop around it represents the field energy. Increasing radial distance from point A results in decreased field energy per unit area. This dispersion of energy accounts for the free space loss between antennas. Mathematically, the energy per unit area varies inversely with the square of the radius of the sphere or as 20 log r in dBs. Simply stated, doubling the radius (or distance from point A) reduces the field energy per unit area by 3/4.

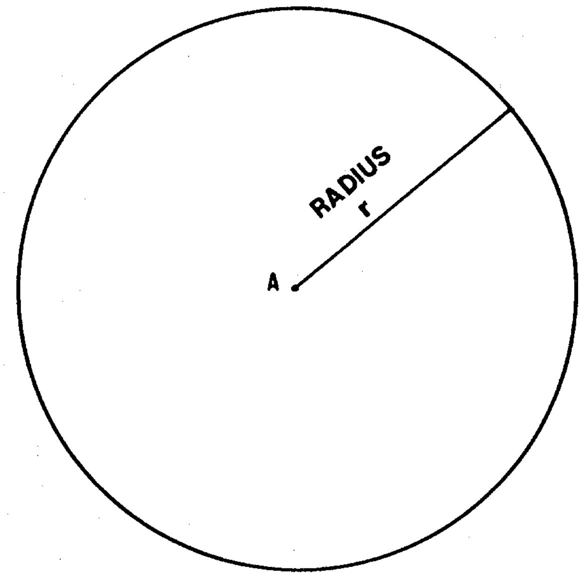


Figure 11-1. Isotropic Radiator Directional Response.

The directional response of an antenna is defined as the antenna's ability to direct energy to or receive energy from a preselected point. This directional response is known as the antenna's radiation pattern. In this sense, the isotropic radiator has a uniform response in all directions.

The directional response of an ideal point-to-point communications antenna would be completely unidirectional. In practical antennas, the ability to direct energy is not so completely defined. Figure 11-2 shows a polar plot of the radiation pattern of a typical directional antenna. As in figure 11-1, point A represents the antenna. Point B indicates point of maximum directional response. Note, however, the responses in other directions.

In extremely directional antennas, the greatest portion of the radiated energy is focused and radiated as a narrow beam. This beam is normally referred to as the main, or major, lobe. The remainder of the energy is dispersed in other directions with varying degrees of focusing. Depending on their position relative to the major lobe, these minor lobes are referred to as side lobes or back lobes of the antenna (figure 11-3).

The major lobe beamwidth of an antenna is an important characteristic. It partially indicates the degree of concentration of the directional response of the antenna. Beamwidth is the angular width of that portion of the major lobe whose response is within 3 dB of the maximum response. Figure 11-4 shows a beamwidth of 5°.

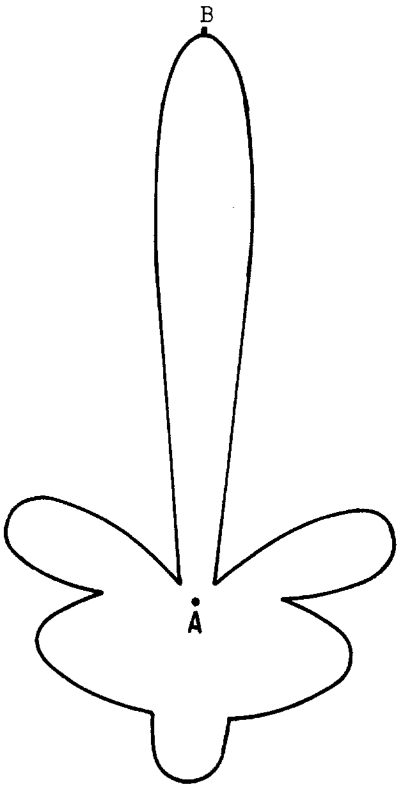


Figure 11-2. Polar Plot of a Directional Antenna.

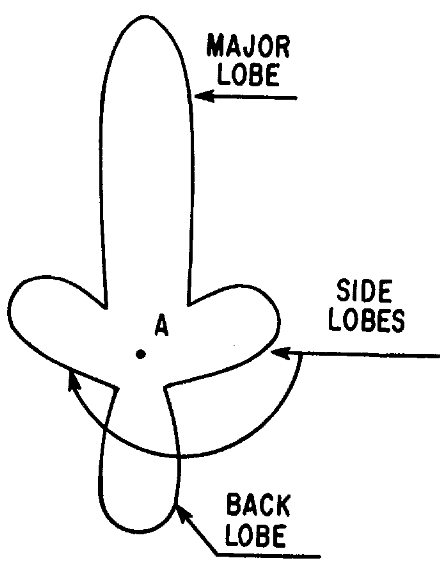


Figure 11-3. Antenna Lobe Structure.

An additional factor indicating concentration of directional response is how well the minor lobes are suppressed or reduced. In figure 11-5, diagram B shows an improvement in minor lobe suppression over diagram A, with a resultant increase of directional response in the major lobe.

Antenna gain is the result of concentrating the directional response of an antenna in a single direction. The free space gain of an actual antenna is the ratio (expressed in dB) of the power in its field to the theoretical power in the field of an isotropic radiator under the following conditions:

a. Amount of power radiated by the two antennas is the same.

b. The points of measurement are:

(1) The same distance from the radiating antenna.

(2) At the point of the maximum intensity of the radiated fields; consequently, the greater the concentration of the directional response of an antenna, the greater the gain. In fact, directivity is the basis of antenna gain.

11-4. Subsystem Degradation. The two largest impairments to communications are human error and noise. Experience has shown that the inexperienced, unknowledeable person who deals with the antenna or transmission line of a microwave wideband system far too often does more harm than good. Respect these components. A nick or dent in a high quality waveguide transmission line or warped antenna line can seriously jeopardize its performance. Follow these rules: don't force it; don't exceed bending radiuses; and, connect components together with the precision they are intended to have. The antenna and transmission line subsystem can give precise, noticeable results when properly installed.

The second largest impairment is noise. To overcome the effects of received noise, certain ratios of desired S/N must be obtained. The ratio of received S/N can be improved by increased transmitter power output and/or higher gain antennas. Since increasing power output without directing the energy toward the desired terminal is very wasteful, antenna gain is used to make most of this ratio improvement. An example of this increase in efficiency is that with a 5 dB increase in antenna gain at both the transmit terminals and the receive terminals, transmitter power output can be reduced to one-tenth of its original value: 5 dB (TX) + 5 dB (RX) = 10 dB. Table 11-1 compares the radiated power required by using isotropic antennas compared to using directional antennas to obtain the same RSL. Gains listed are the sum of transmit and receive antenna gains.

11-5. The Parabolic Antenna. There are many types of antennas and variations of each. The type to be used depends on the type of radio path considered. In point-to-point communications, directional transmitting antennas are used, since energy directed other than to the intended terminal is wasted. Similarly, the receiving antenna must be directional, since response to signals from any direction other than the transmitting point introduces undesirable signals into the receiver.

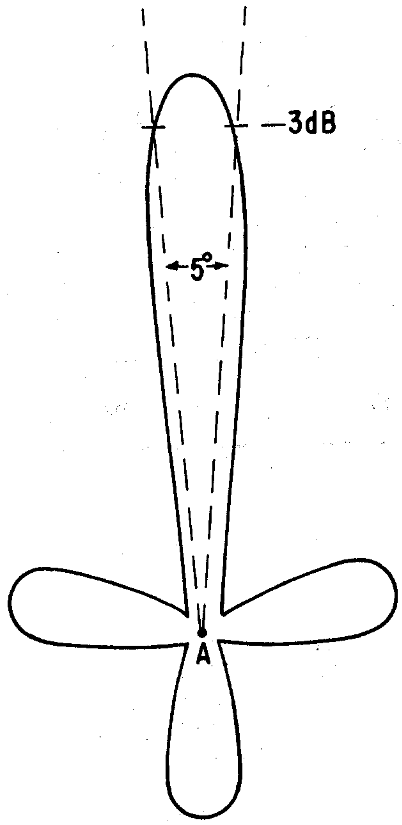


Figure 11-4. Major Lobe Beamwidth.

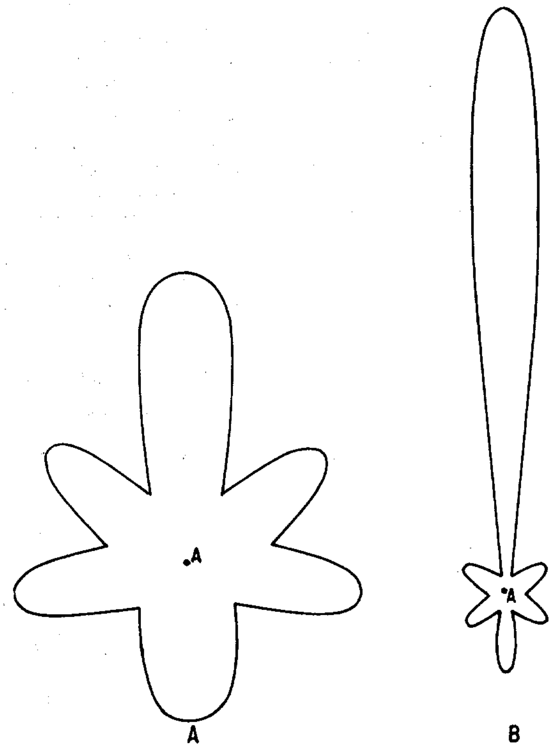


Figure 11-5. Minor Lobe Suppression.

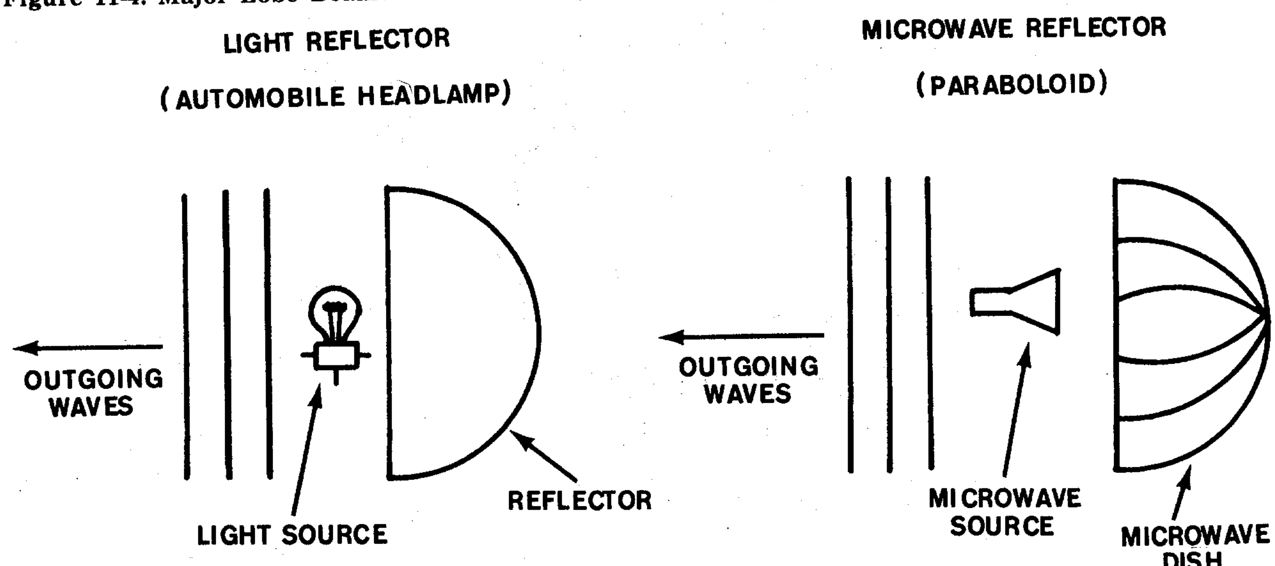


Figure 11-6. Parabolic Reflector Antennas.

The M/W source of figure 11-6 is usually a waveguide horn antenna. By itself, the horn does not have good gain nor is it very directional in its response; but, together with the large capture area of the M/W dish, radio energy is concentrated so that gain and directivity are improved.

The M/W dish has the shape of the curve described in mathematics as the parabola. The parabola reflects the radio energy to the focus where the waveguide horn is located; thus, it acts to increase the radio energy in much the same way a telescope concentrates light.

Depending on the frequency involved, this antenna may vary in size from two feet to 120 feet in diameter. Because of their size and shape, these antennas are known as "dishes" or "billboards." Figure 11-7 shows a "dish" and figure 11-8 shows a "billboard."

During transmission, the reflector focuses the energy incident on it (illuminating it) into a narrow beam of high intensity. This focusing is accomplished in much the same way that the light from a headlight is focused (figure 11-6).

The energy illuminating the reflector may be from the primary feed antenna for transmitting purposes or from the distant station in a receiving situation. Although the energy is moving in opposite directions in these two examples, the reflector's focusing action takes place in both. Figure 11-9A shows focusing of transmitted energy and B shows focusing of received energy.

The ability of the parabolic reflector to concentrate radio energy is a function of reflector size and of the frequency of the radio wave. An increase in either reflector size or frequency results in an increase in the focusing effect and a consequent increase in antenna gain. Figure 11-10 plots gain versus frequency for several common reflector sizes.

	System Using Isotrophic Antenna	*System Using Directional Antennas With Total Gains Of:*			
		70dB	80 dB	90dB	100dB
POWER REQUIRED	10,000,000,000 WATTS	1000 WATTS	100 WATTS	10 WATTS	1 WATT

Table 11-1. Gain of Directional Antenna.

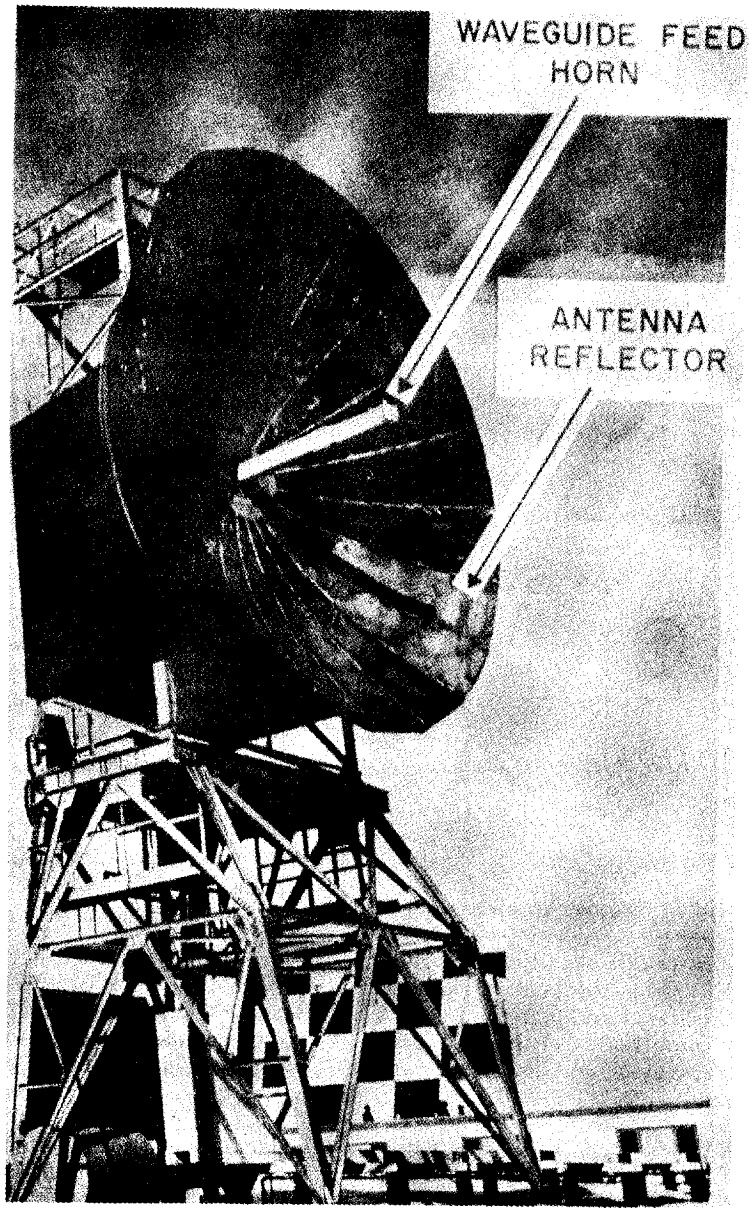


Figure 11-7. "Dish" Antenna.

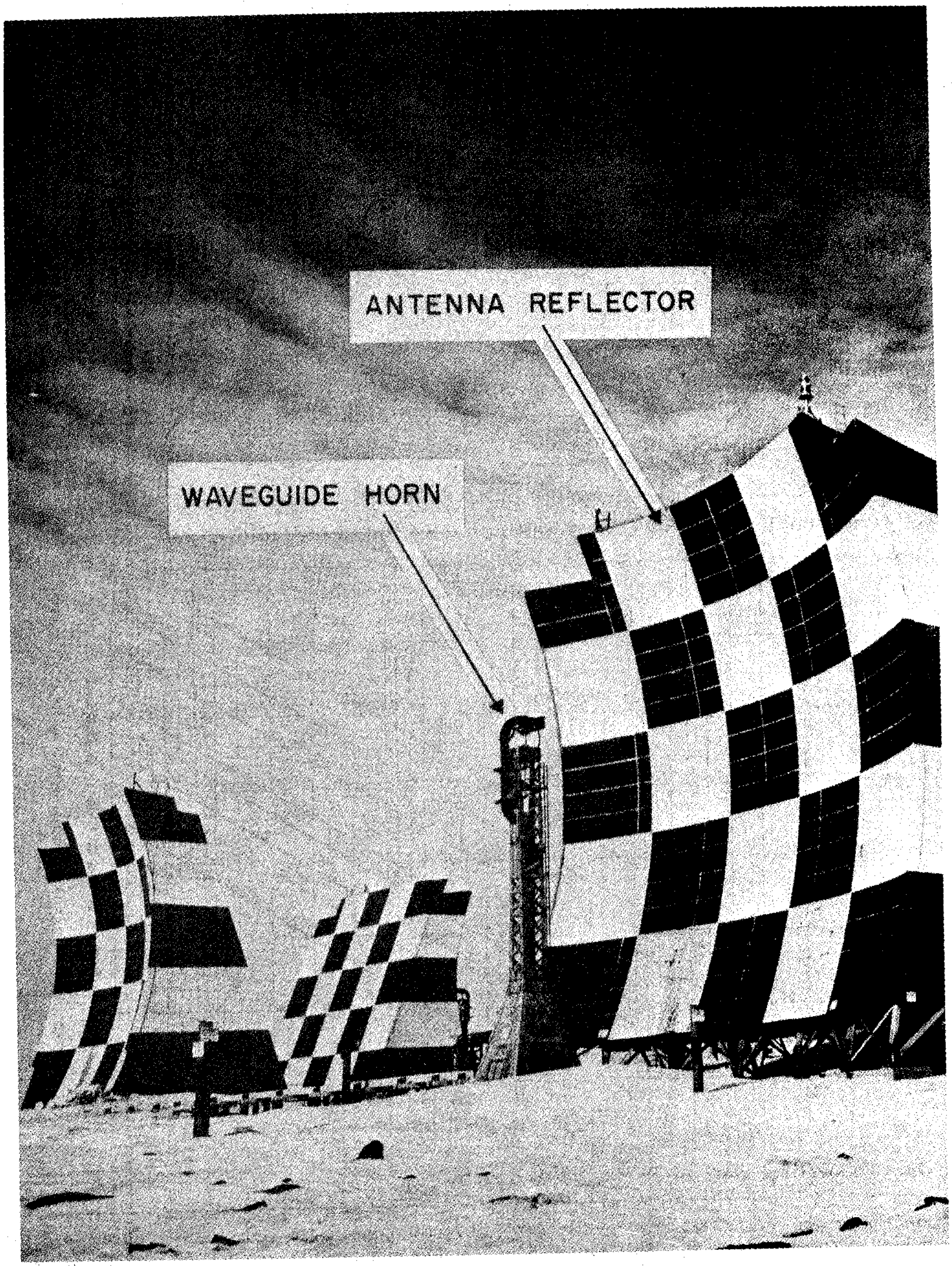


Figure 11-8. "Billboard" Antenna.

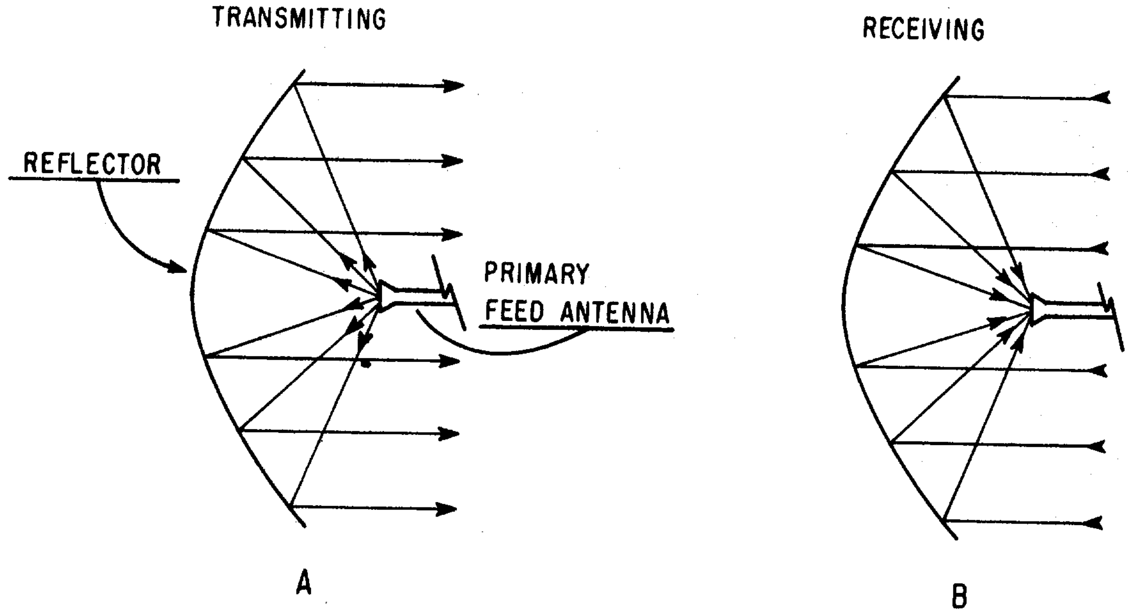


Figure 11-9. Reflector Focusing in Transmission and Reception of Radio Waves.

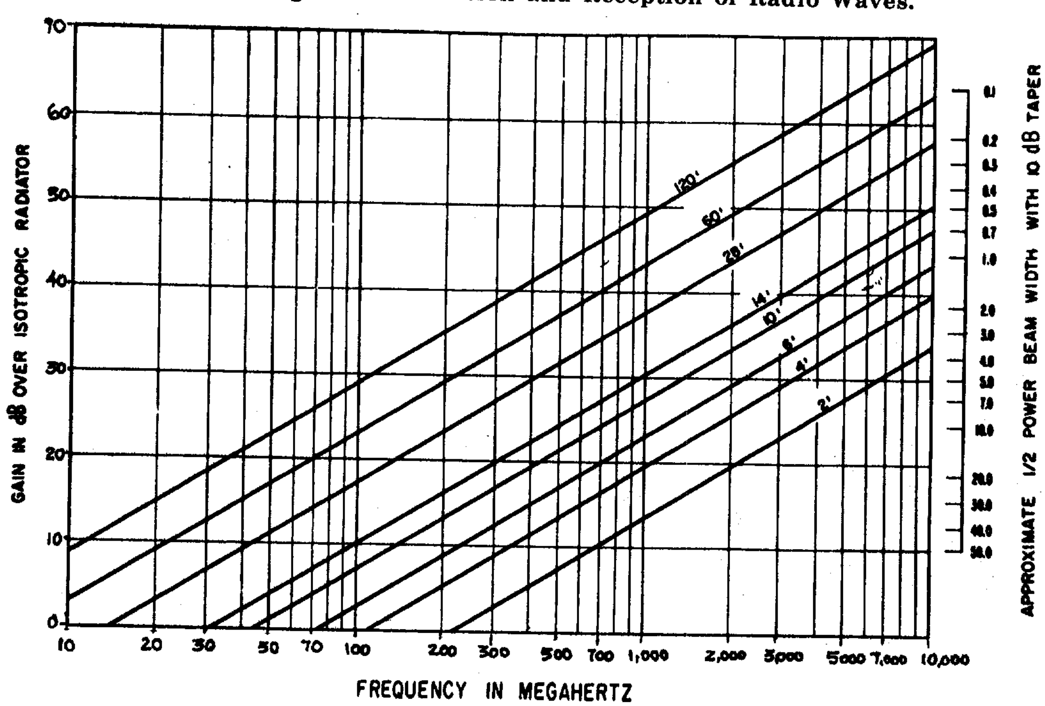


Figure 11-10. Theoretical Gain of Parabolic Antennas.

The parabolic with the narrowest beamwidth and highest gain is the Cassegrain antenna (figure 11-11A). The hyperbolic subreflector acts to more fully and more uniformally illuminate the parabolic dish than the feed horn could by itself. This uniform distribution of field energy is most desirable where a sharp, well-defined directional response is desired; but the uniform distribution also causes an increase in side lobes. To suppress side lobes in the parabolic reflector antenna, tapered reflector illumination is used (figure 11-11B).

In this form of illumination, the energy level incident at the edge of the reflector is reduced in a particular manner below the energy level incident at the center of the reflector. A common value of tapering is 10 dB. This means that the energy level at the edge of the reflector is one-tenth of the energy level at the center of the reflector.

A secondary effect of tapered illumination is a slight widening of beamwidth. Beamwidths shown in figure 11-10 are based on a 10 dB taper and are, therefore, somewhat wider than the mathematically-computed beamwidth for the sizes of reflectors shown.

Shrouds made of absorbent material are used to further reduce side and back radiation (figure 11-11C).

Table 11-2 summarizes the characteristics of parabolic reflector antennas commonly used in wideband radio communications systems.

ANTENNA GAIN	35dB—50dB
ANTENNA BEAMWIDTH	.5 Degrees—3 Degrees

Table 11-2. Antenna Characteristics Summary.

11-6. The Cornucopia Horn. This type of antenna is a unique combination of a parabola and a horn antenna (figure 11-12). It is especially suited for multiple wideband signals, such as used by the telephone industry. Since the DOD is not presently using multiple wideband signals and because of the Cornucopia's weight and cost, the parabola or shrouded parabola is more often used; however, it is interesting to note that the Cornucopia has better front-to-back ratio and is comparable in gain and side lobes (table 11-3).

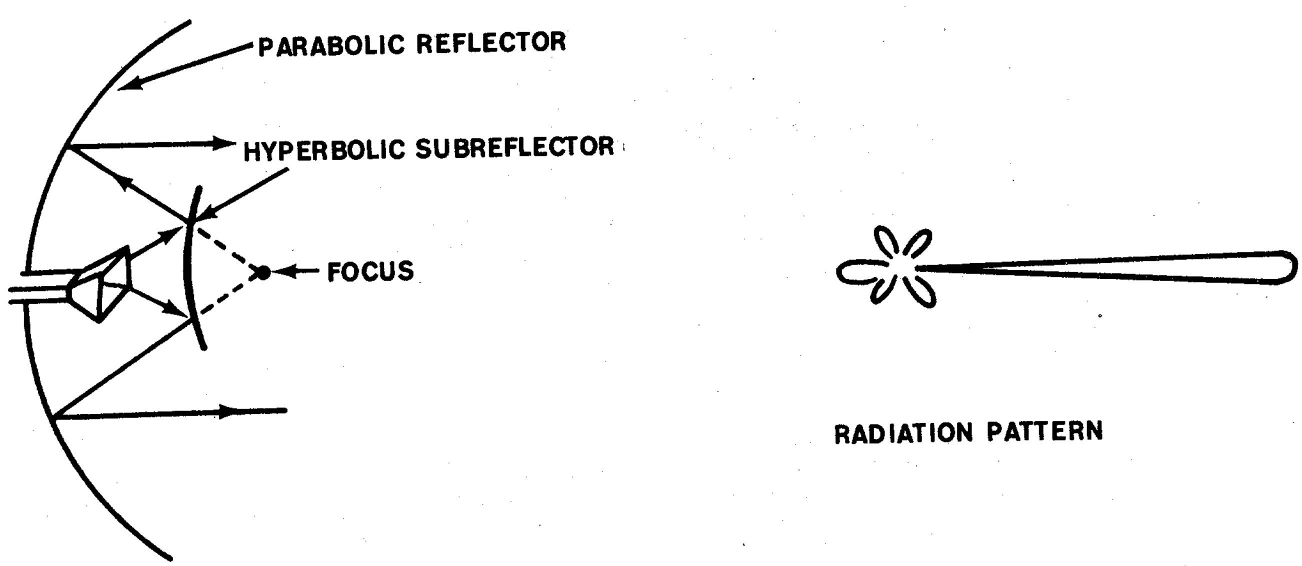
11-7. The Plane Reflector. In some cases, another element is used in the antenna subsystem or in the radio path. This element is the plane or passive reflector.

In the "periscope shot" (figure 11-13), the antenna is mounted near ground level placed to radiate vertically. The reflector is then placed at an angle to reflect the radio energy, acting like a mirror toward the distant station. Reception also occurs in like manner with the radio energy having reverse direction. The size of the periscope reflector is about 4′ x 6′.

The periscope antenna has the following advantages: long runs of expensive waveguide are eliminated, waveguide or coaxial cable pressurizing systems are not required, and maintenance is simplifid; but, because of inherent increase in side and back lobes, interference (especially satellite interference) is a problem. For this reason, as well as increased alighment difficulties, the periscope antenna is rarely used.

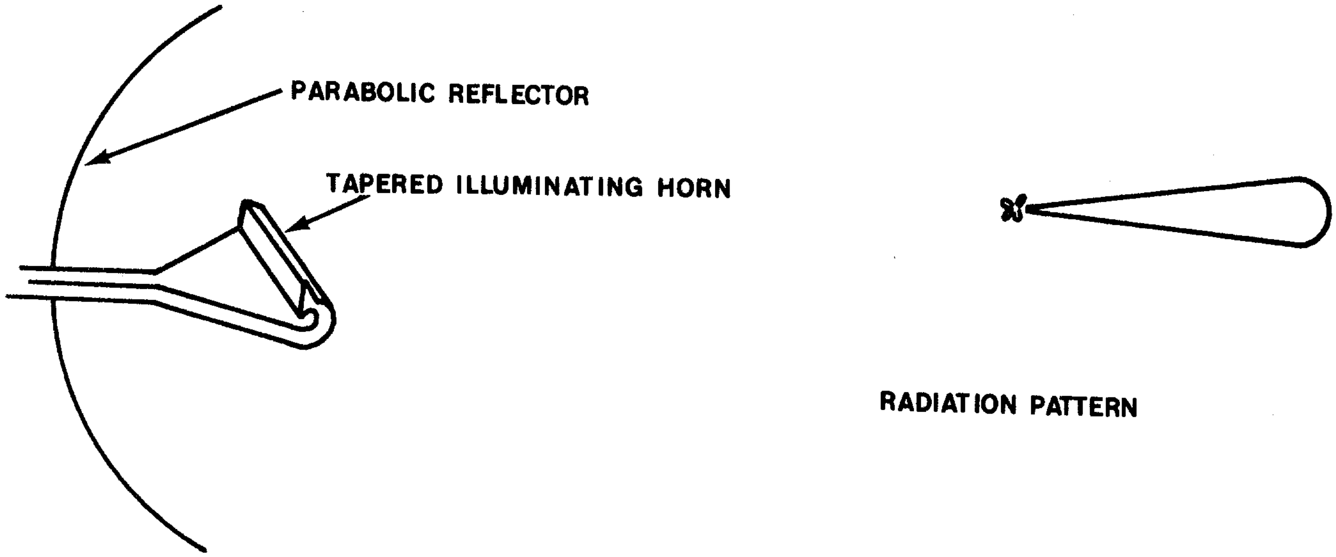
Some paths have severe terrain difficulties where a LOS path is difficult to obtain. In these cases, the shortest distance between two points may not be a straight line. By bending the path of the radio energy horizontally, the required path can sometimes be achieved. Figure 11-14 shows a case of this nature.

In this hypothetical case, a radio link between A and B is required. Placement of either station at the peak of the mountain is impractical because of terrain difficulty. By bending the radio beam through point C, a path over relatively the same elevation is possible. At this point, enter the plane reflector. Plane reflectors used in this manner are known as passive repeaters and perform the same function here as in the periscope shot. The angle of its reflecting surface is such that energy from point A is reflected toward point B and vice versa. The size of this reflector will vary from installation to installation, but a common size used is 24′ x 30′. Figure 11-15 is a 16′ x 24′ plane reflector.

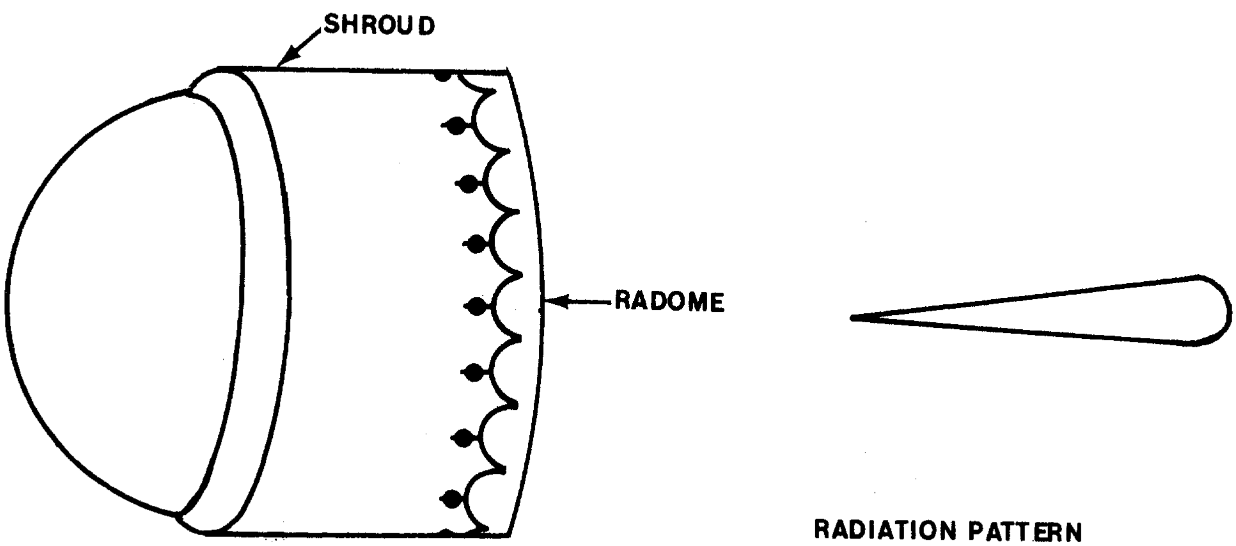


A - Cassegrain Antenna

Figure 11-11. Parabolic Antennas, Reflectors, and Their Radiation Patterns.



B - Parabolic Reflector with Tapered Reflector Illumination



C - Shrouded Parabolic Antenna

Figure 11-11. Parabolic Antennas, Reflectors, and Their Radiation Patterns.

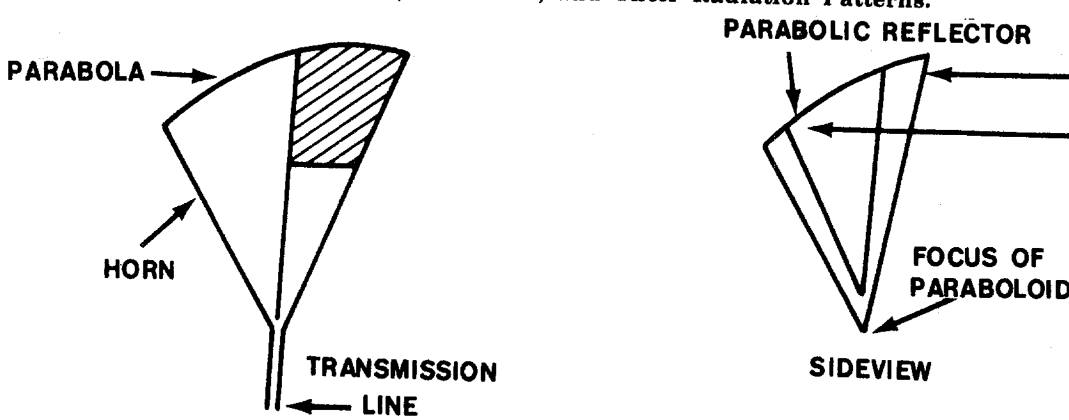


Figure 11-12. The Cornucopia Horn Antenna.

Antennas	*Gain*	*Back Lobe to Front Lobe Ratio*	*Side Lobe Ratio*
The Shrouded Parabola (high performance)	35-50 dB	60-70 dB down	25 dB down
The Cornucopia Horn	40-50 dB	70 dB or better	25 dB or better

Table 11-3. Comparison of Antenna Characteristics.

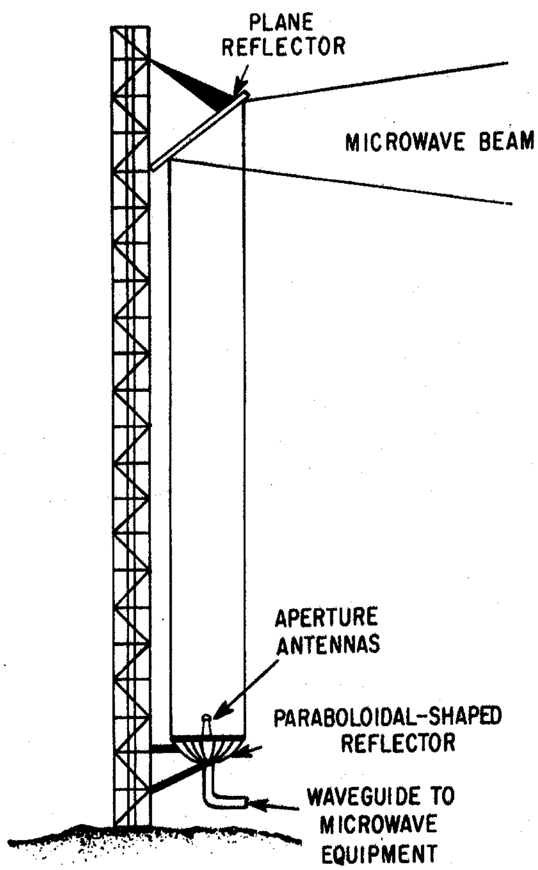


Figure 11-13. Periscope Shot Installation.

Plane reflectors are very unpredictable. Ground reflections near the reflector as well as vertical reflections from path objects (such as buildings and mountains) add in a complex manner with the incident beam energy. This complex wave is usually lower in energy than the original beam. Then the reflected beam again goes through a complex beam addition process on its way to the receiver. Satisfactory field energy may or may not be present at the receiving antenna. Because of these complexities and alignment difficulties, the plane reflector is also rarely used.

11-8. Transmission Lines. The purpose of the transmission line is to interconnect the radio and antenna. The simplest transmission line is two wires: however, as frequencies are increased, losses start to be evident. At 1 MHz, specific wire spacings are used to reduce these losses. Above several hundred MHz, the losses of a two-wire line are too great and another type of transmission line, called the coaxial cable, is used. But even coaxial cable has great losses at GHz frequencies. To reduce these losses, waveguide is used.

11-9. Transmission Line Losses. Transmisssion lines are an extremely important part of the overall communications system, since the radio can't be mounted next to the antenna. Transmission lines must, therefore, be relied on to transmit the radio energy. But they have losses: conductor losses, dielectric losses, hysteresis losses, mismatch losses, and losses due to radiation.

a. The first three - conductor, dielectric, and hysteresis - are basic to the type of transmission line used. These three types are absorptive losses by nature and are dissipated as heat.

b. Mismatch loss is caused by a discontinuity or change in impedance. When any two transmission line components are attached together, a discontinuity is present at their junction. This discontinuity can be reduced to an acceptable level if connected properly; therefore, mismatch loss is not basic or inherent, but can be controlled. Discontinuities cause several problems. One is that maximum power transfer cannot occur. Another, and the worst one, is that discontinuities cause reflections. Reflections, in turn, cause distortion and noise.

c. The last loss is not much of a problem unless the transmission line is split or otherwise open at some undesired point. Some of the power is then radiated through the open portion. This loss causes an undesired side lobe in the radiation or reception pattern.

Again, a word of caution is necessary here. An otherwise good transmission line, if improperly installed, can be the cause of significant noise and system degradation, significant enough to limit system performance (figure 11-16).

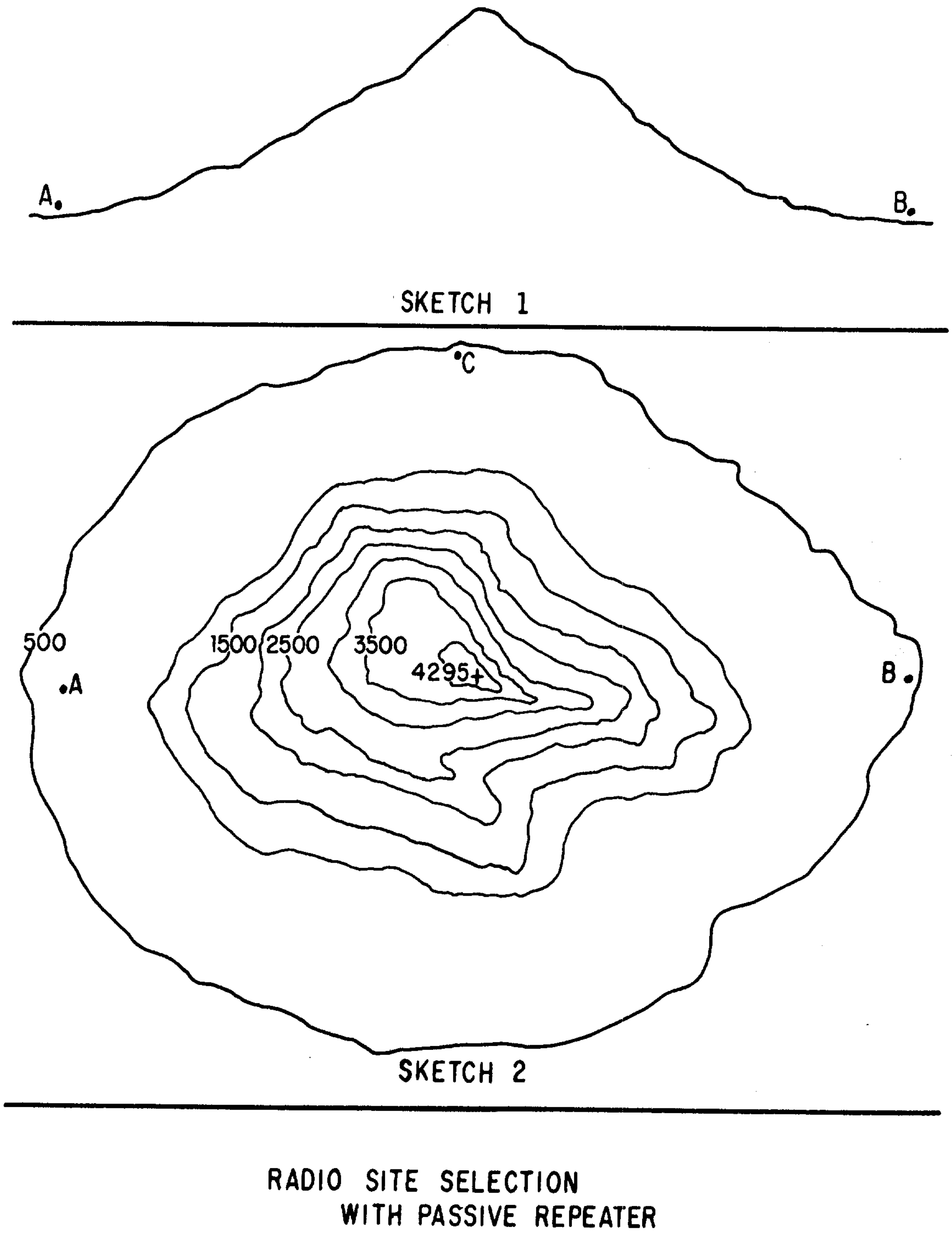


Figure 11-14. Radio Site Selection with Passive Repeater.

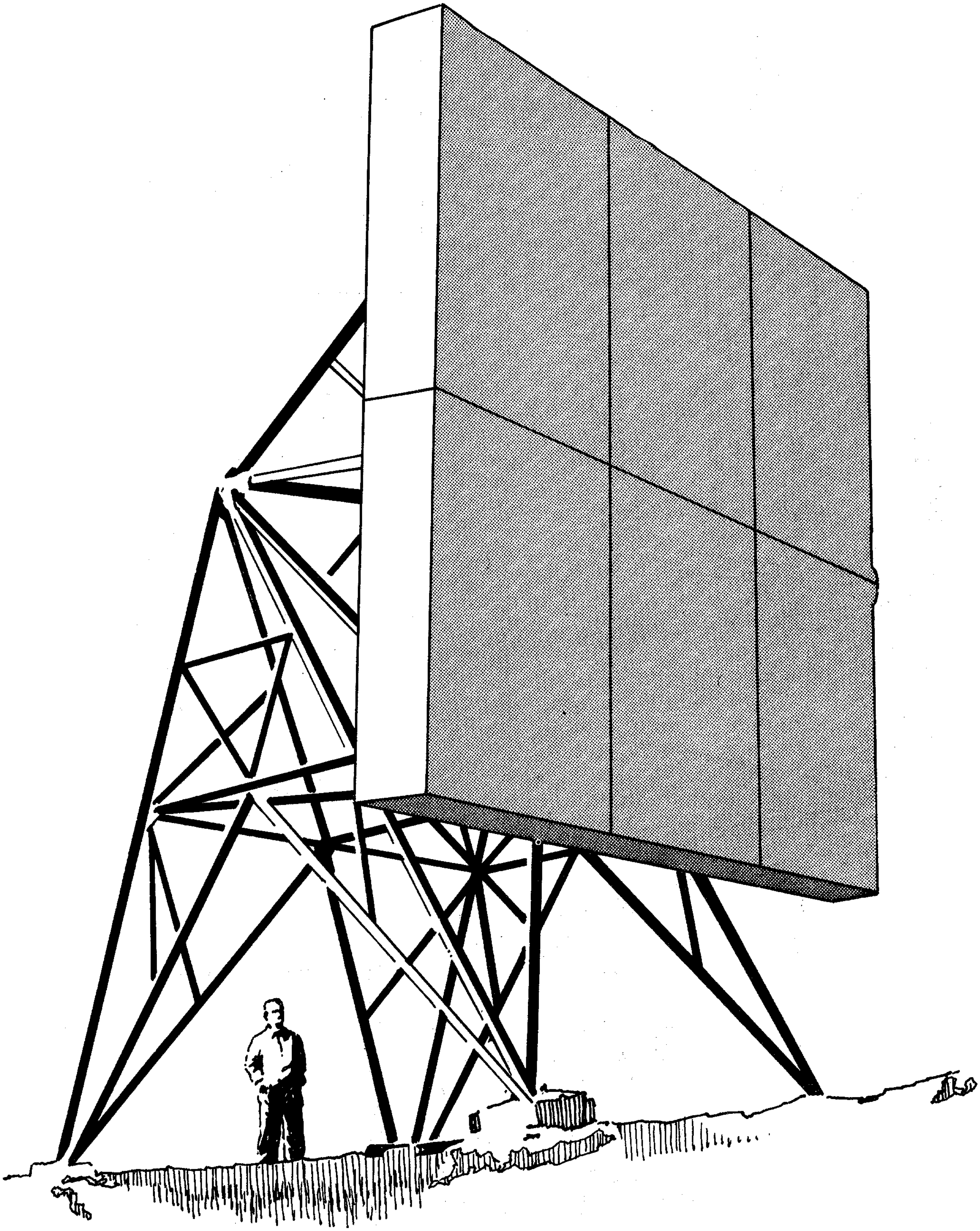


Figure 11-15. Plane Reflector.

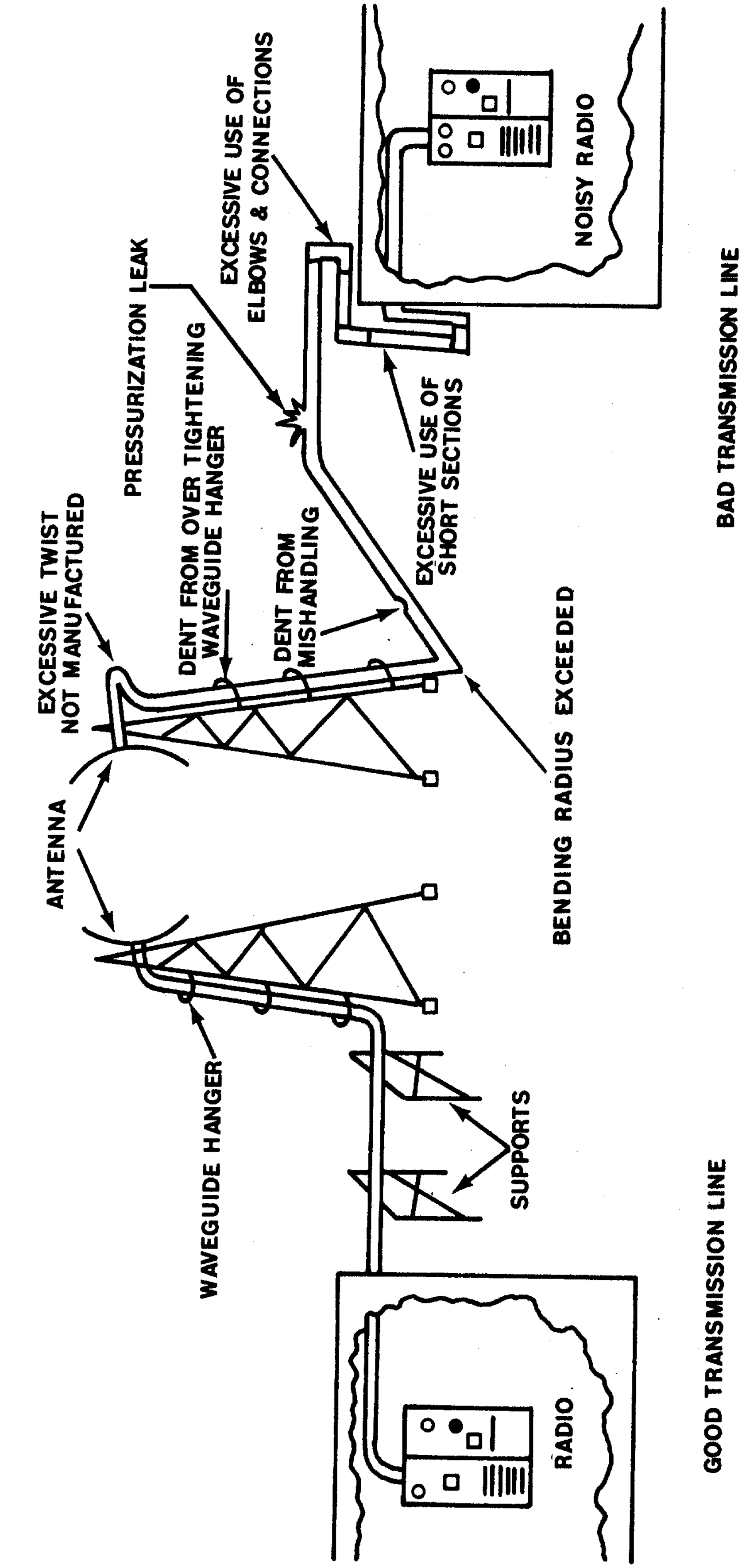


Figure 11-16. Transmission Line Limiting System Performance.

11-10. Mismatch Loss. Figure 11-17 is a graph showing the effect of varying the impedances of the load. If the load impedance doesn't equal the source impedance, power output is reduced. The same is true for reception. Power received with mismatched impedances is reduced.

What happens, then, to the extra power? It is reflected back and forth until it is dissipated. Figures 11-18 and 11-19 show the effect of worst mismatch (open or short) using a DC source. The voltages and currents start down the transmission line with no knowledge that the other end is open or shorted but, when they reach the end, the wave is either reinforced or diminished and the result is reflected back to the source.

A transmission line, however, usually has alternating voltage applied rather than DC. The alternating voltage will create a constant reflection of alternating voltage (figure 11-20 and 11-21). The resultant wave is called a standing wave. If an AC voltmeter were used to measure the voltage at varying points along the line, there would be a definite point for maximum and a definite point for minimum meter voltage. These points are located at integral multiples of 1/4 wavelength. Their ratio is called voltage standing wave ratio (VSWR). If the maximum to minimum voltage were a ratio of 1.5, the VSWR would be 1.5:1. A perfectly matched line where all the impedances are uniform causes no reflections. The VSWR is then 1:1. Although figures for alternating current are not shown, standing waves of current also exist. A current maximum or current loop will be at a voltage minimum or voltage node.

11-11. Transmission Line Types. Reflections and standing waves also exist in a waveguide; but, in a waveguide, the magnetic and electric fields carry the energy. Figure 11-22 shows how field energy in a waveguide may be thought of as reflecting off the walls as it travels down the guide.

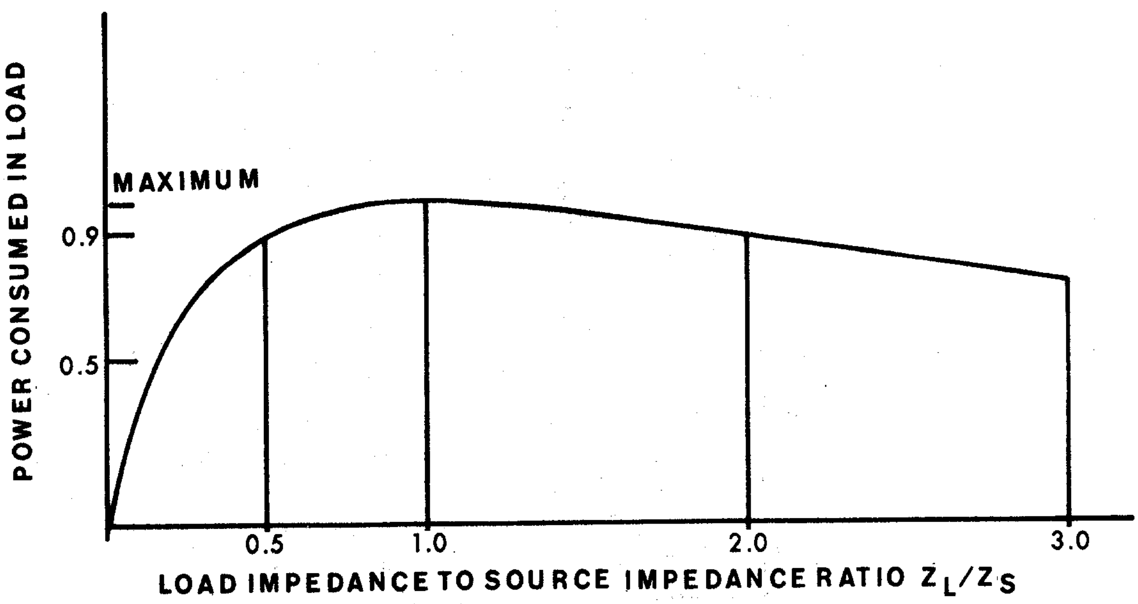


Figure 11-17. Load Power Consumption Compared to Load Impedance.

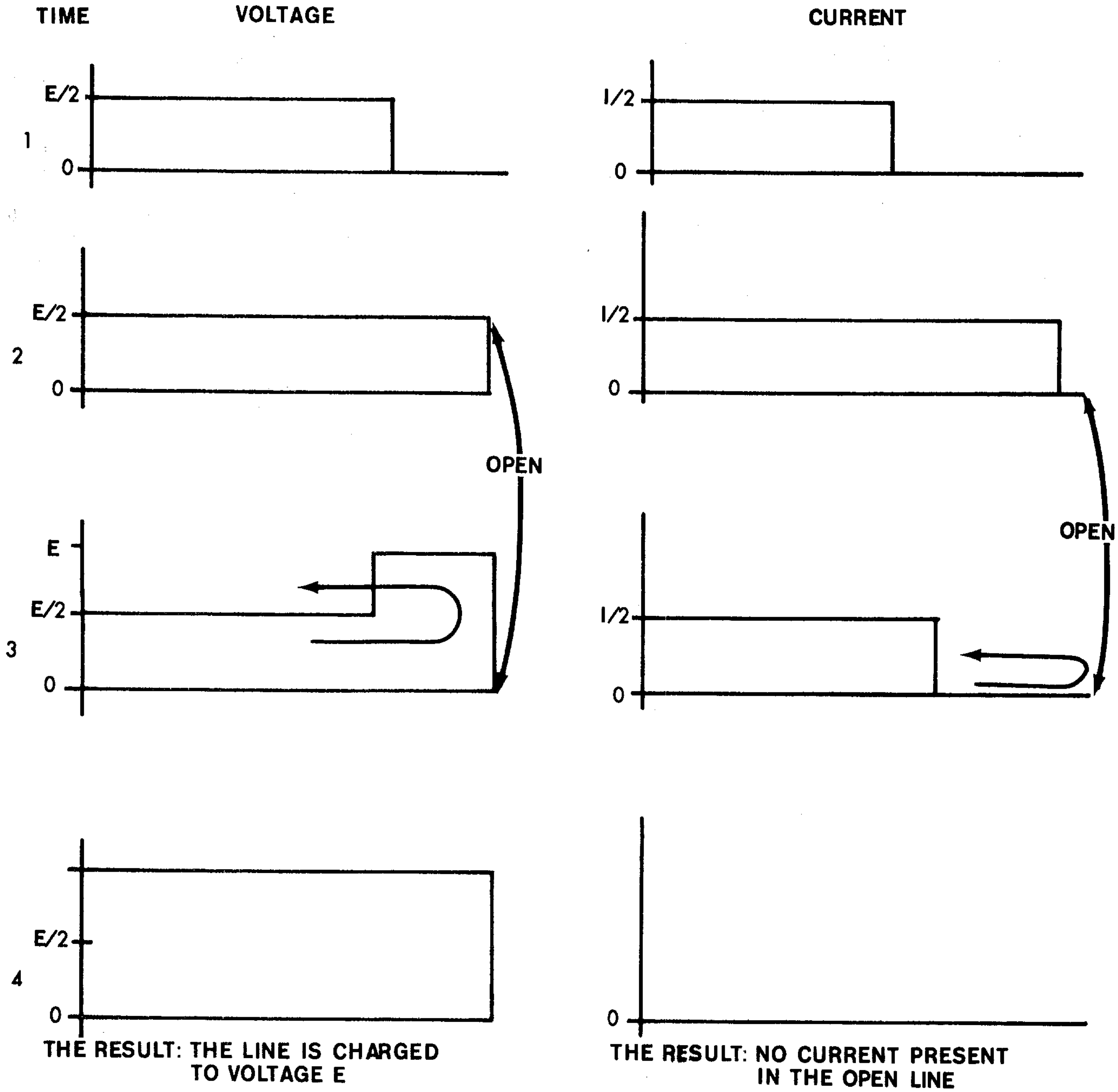


Figure 11-18. Voltage and Currents Traveling Down an Open Transmission Line.

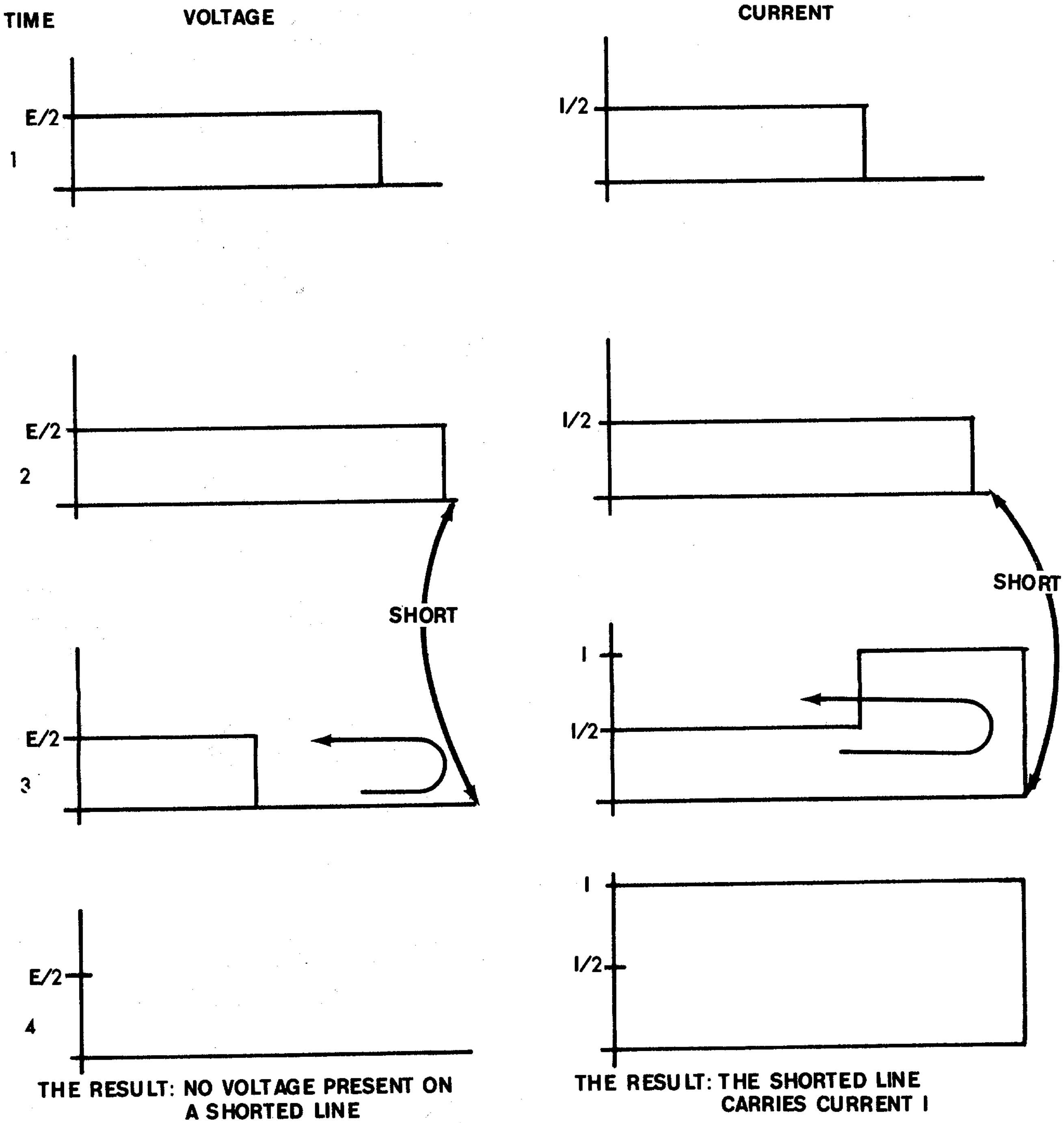


Figure 11-19. Voltage and Currents Traveling Down a Shorted Transmission Line.

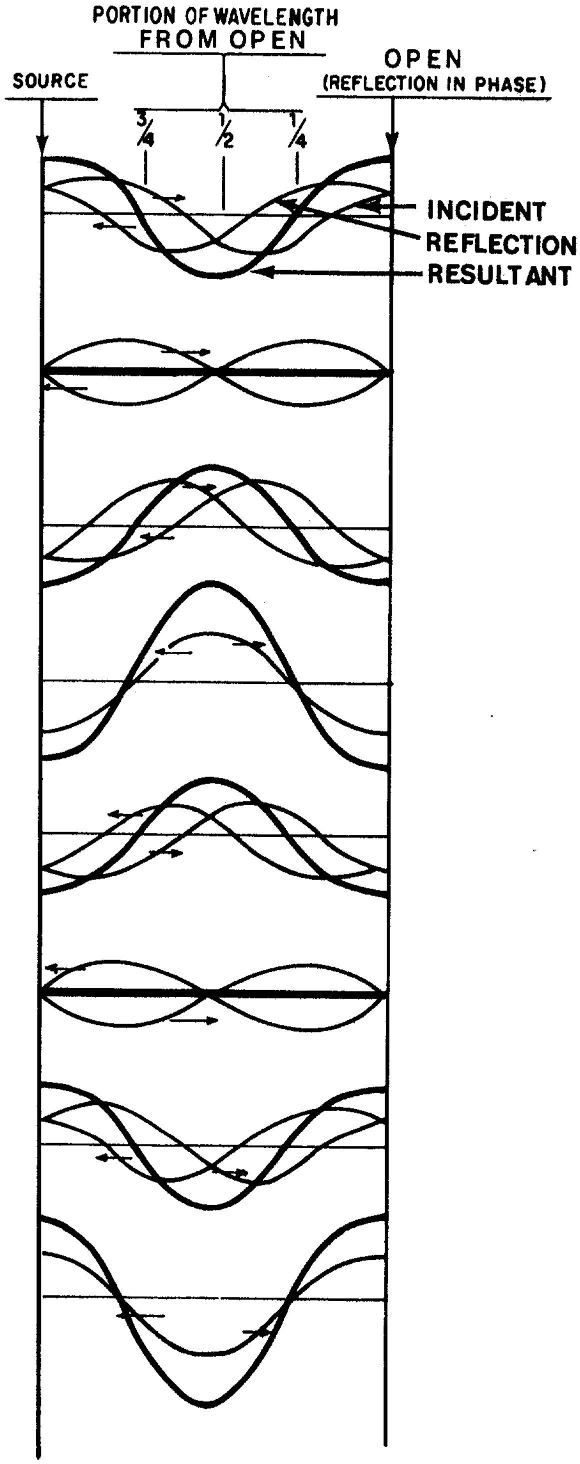


Figure 11-20. Incident and Reflected Voltage on an Open Transmission Line.

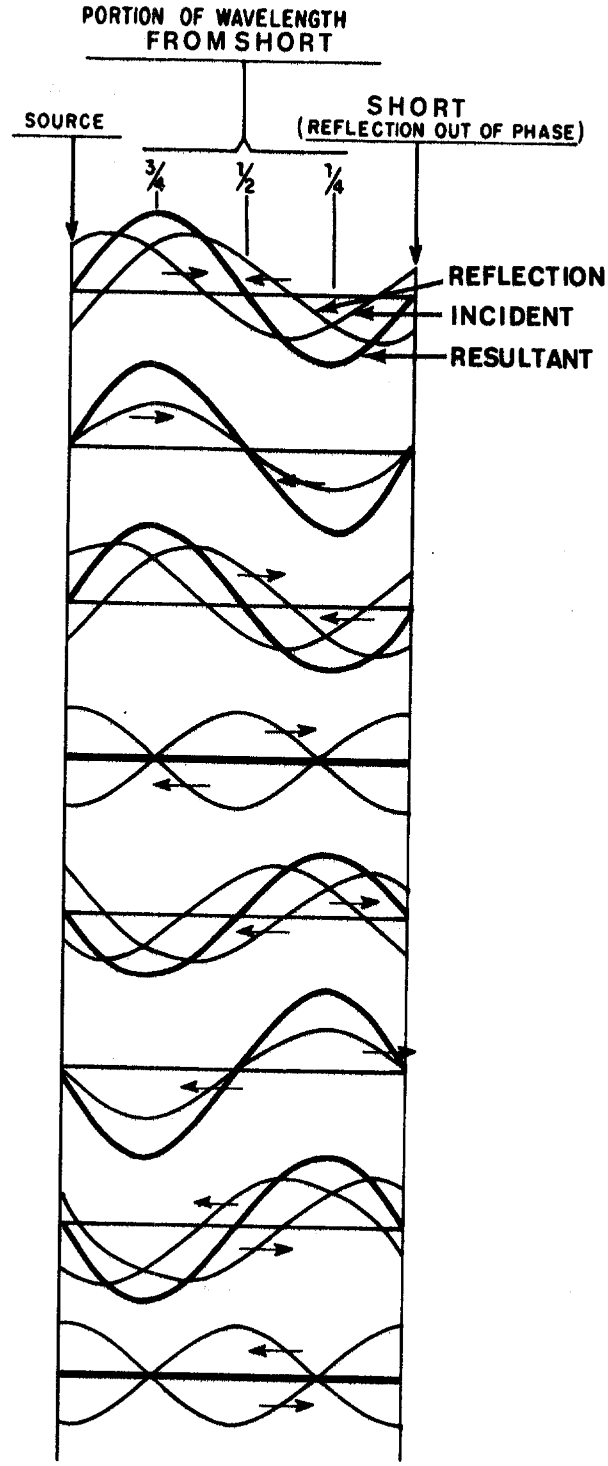


Figure 11-21. Incident and Reflected Voltage on a Shorted Transmission Line.

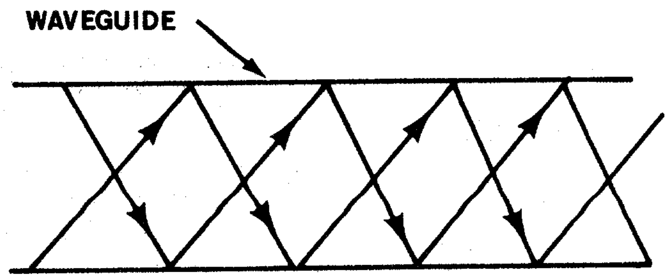
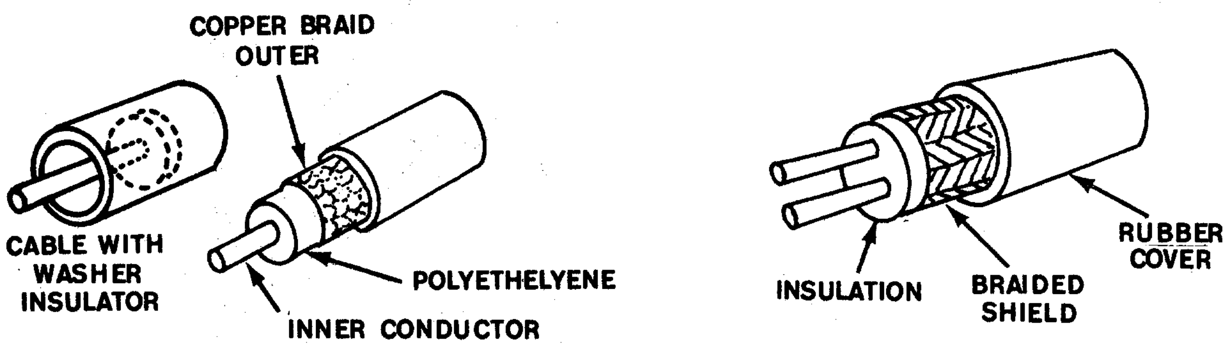
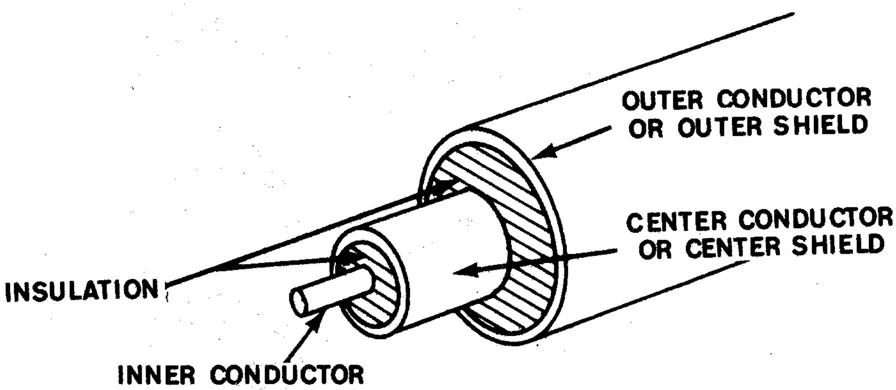


Figure 11-22. Field Energy Propagating Down a Waveguide.



A - Coaxial Cable

B - Two-Conductor Shielded Pair



C - Triaxial Double Shielded Cable

Figure 11-23. Cable Types.

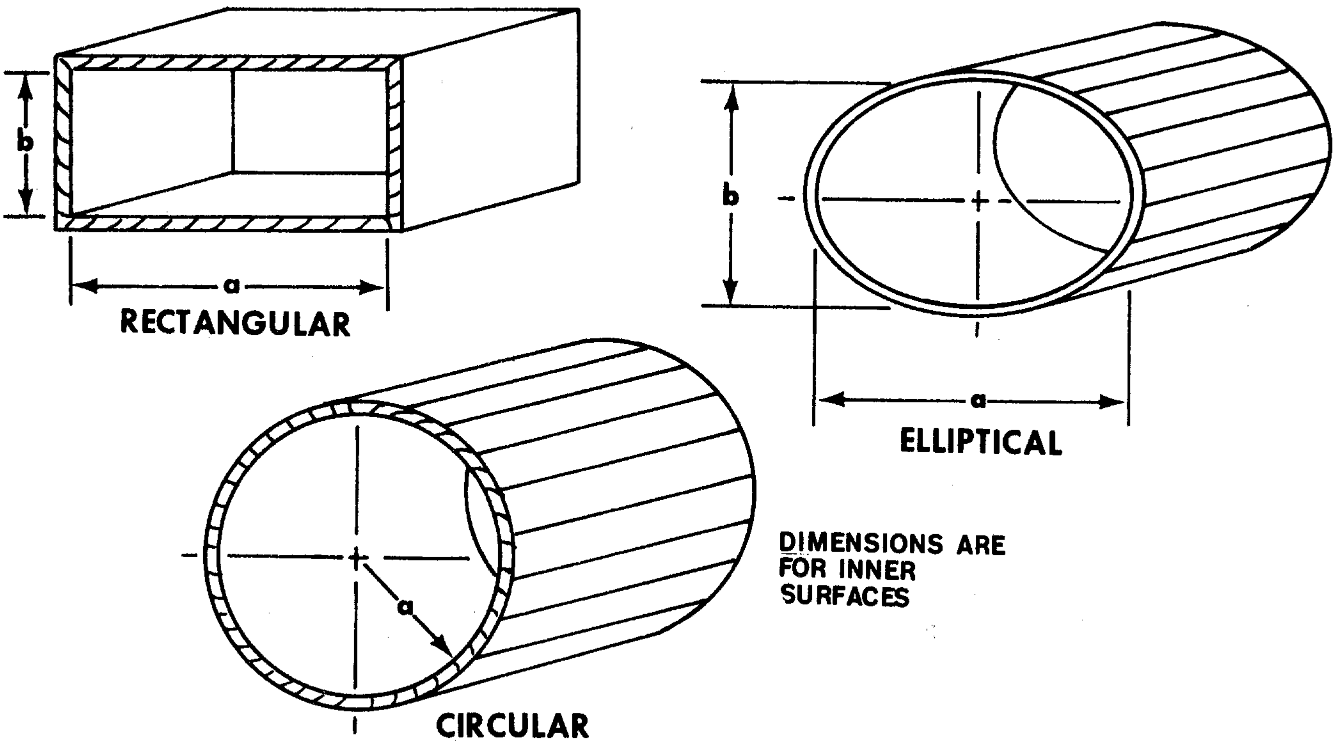


Figure 11-24. Types of Waveguides.

Manufacturing transmission lines is an art. At low frequencies, waveguides would be too large and expensive, so coaxial cable can be used. Figure 11-23a shows the makeup of a coaxial cable. Naturally, the lower the loss of the cable the better its ability to transmit noise-free signal, but the higher its cost. The manufacturer carefully selects the type of conductor material, its plating to lower resistance (silver, for example), type of insulator (styreine, teflon), and size for power handling capability, and impedance (which varies with size). Other types of coaxial cables are used in harsh environments to eliminate noise (figure 11-23b and c).

11-12. Waveguides. Manufacturing waveguides is also an art. Figure 11-24 shows several different types of waveguides. Rectangular waveguide has the advantage of having the widest bandwidth for single mode propagation. A mode is the configuration in which energy is propagated through the waveguide. One mode is characterized by having its electric field perpendicular to the length of the guide. This is the TE (or transverse electric) mode. TM (or transverse magnetic) mode has its magnetic field perpendicular to the length of the guide. Rectangular waveguide is rigid. It comes in lengths up to 20 feet; hence, many connections have to be made along its run up the tower to the antenna. Each time a connection is made, some loss and mismatch is introduced; therefore, the disadvantage of rectangular waveguide is the loss it has due to many connections.

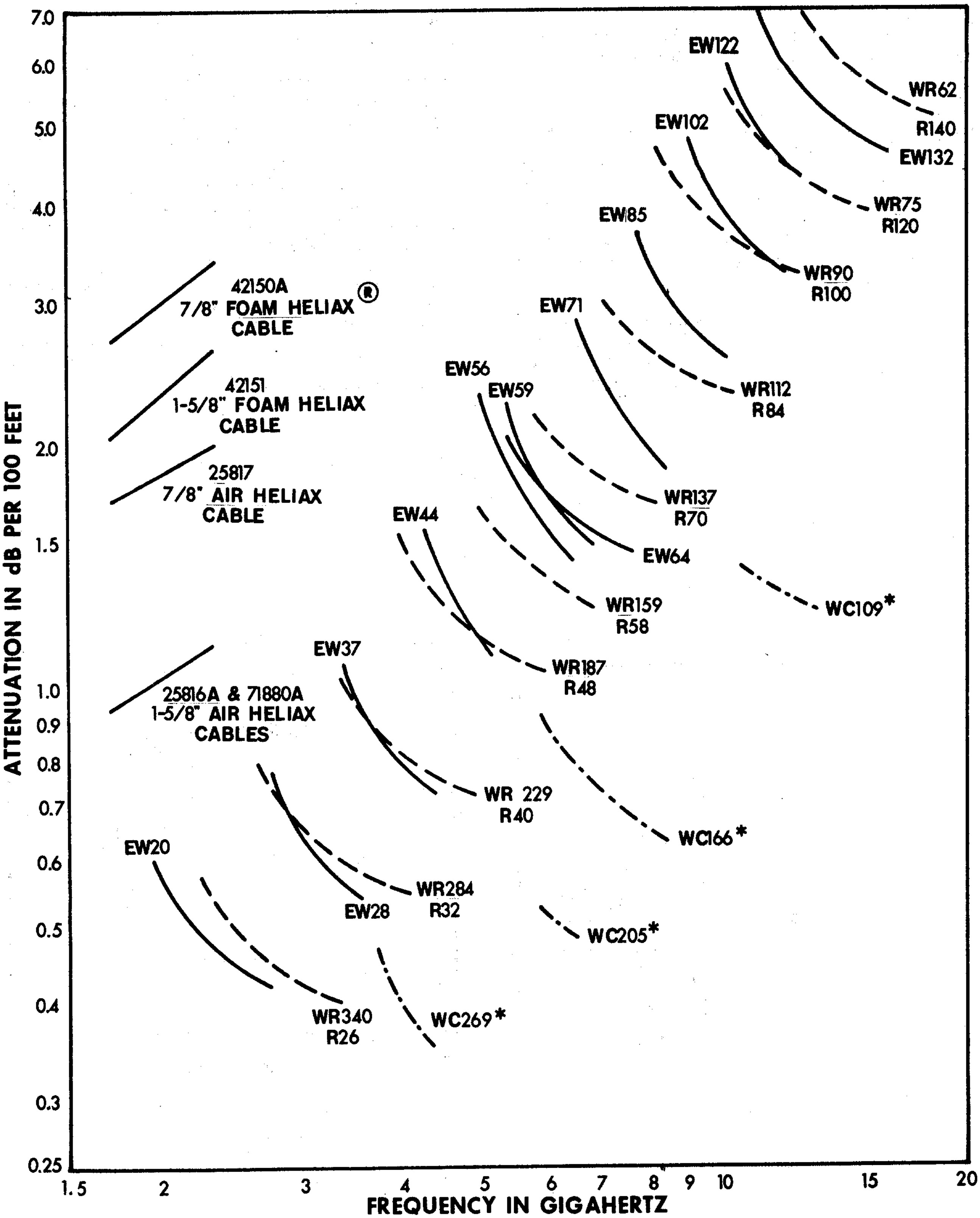
Elliptical waveguide is semi-flexible. It can be bent if proper bending radius procedures are followed; therefore, it can be shipped in one piece on a large reel. It can be procured in any length required for a particular antenna height; therefore, connectors are reduced to a minimum. But for very long runs, elliptical waveguide has enough loss that sometimes circular waveguide is used.

Circular waveguide has the lowest loss of any waveguide. In circular waveguide, the electric and magnetic fields cause fairly uniform current distributions in the wall of the guide. This is not true with rectangular or elliptical. With non-uniform currents, certain portions have greater current densities and thus more loss.

For a 300′ run of waveguide at 13 GHz, typical figures for attenuation are 25 dB loss for rectangular, 24 dB for elliptical, and only 8 dB for circular.

Circular guide will support many modes of energy. It, too, is rigid and requires connections for its run up to the antenna. Connectors in circular guide tend to introduce other modes of propagation. These modes are reflected from the feeds (antenna and radio connections) and represent wasted energy. But as they reflect in the waveguide, they cause a distorted output in the form of delay distortion (similar to echo distortion). For these reasons, the connectors are very accurately manufactured and, for this reason, they are more expensive. Because of the precision required to install circular waveguide, it is not used that much.

Figure 11-25 compares various waveguides to coaxial cable. Notice the much lower loss associated with wavegudes, particularly circular. Elliptical waveguide designation lacks standardization; therefore, while the



ATTENUATION CURVES BASED ON:
VSWR 1.0
AMBIENT TEMPERATURE 24°C (75° F)
HIGH CONDUCTIVITY COPPER

LEGEND:
ELLIPTICAL ————
RECTANGULAR — — — —
CIRCULAR —·—·—·—

* ADD 0.3 dB TO ALLOW FOR TOP & BOTTOM TRANSITIONS

Figure 11-25. Microwave Waveguide and Cable Attenuation. (Courtesy Andrew Corp) numeric portion is standard, each manufacturer may assign his own alpha designator. For example, EW71 and PE71 are similar, but manufactured by different companies.

11-13. Waveguide Flanges. The connectors used to attach waveguides are called flanges. There are many types of flanges. Figure 11-26 shows a choke cover flange in rectangular waveguide. The L groove in the choke section is 1/2 wavelength long. Since it is shorted in the choke, the space or groove looks like a short to the field energy, so the wave travels along the guide, through the flange, and it appears to the wave as a continuous run.

A contact pressurizable rectangular (CPR) flange is shown in figure 11-27. It has a highly machined lip that butts up against the mating flange. This ensures a tight fit. Because the lip surfaces accurately contact each other, the CPR flange has lower VSWR and less radiation loss than the choke cover flange.

Another good flange is the CPR through flange. This flange is similar to the butt flange except the waveguide is continuous through the flange instead of contacting the lip back in the flange (figure 11-28).

The contact unpressurizable rectangular (CMR) flange has no groove for a seal, hence it cannot be pressurized and has less mechanical strength. It is usually used internal to the radios.

11-14. Pressurization. Waveguides and certain coaxial cables are pressurized with dry air or nitrogen so that moisture is kept from entering the guide. Moisture causes corrosion which creates more attenuation and noise. A mylar plastic window keeps positive pressure in the guide.

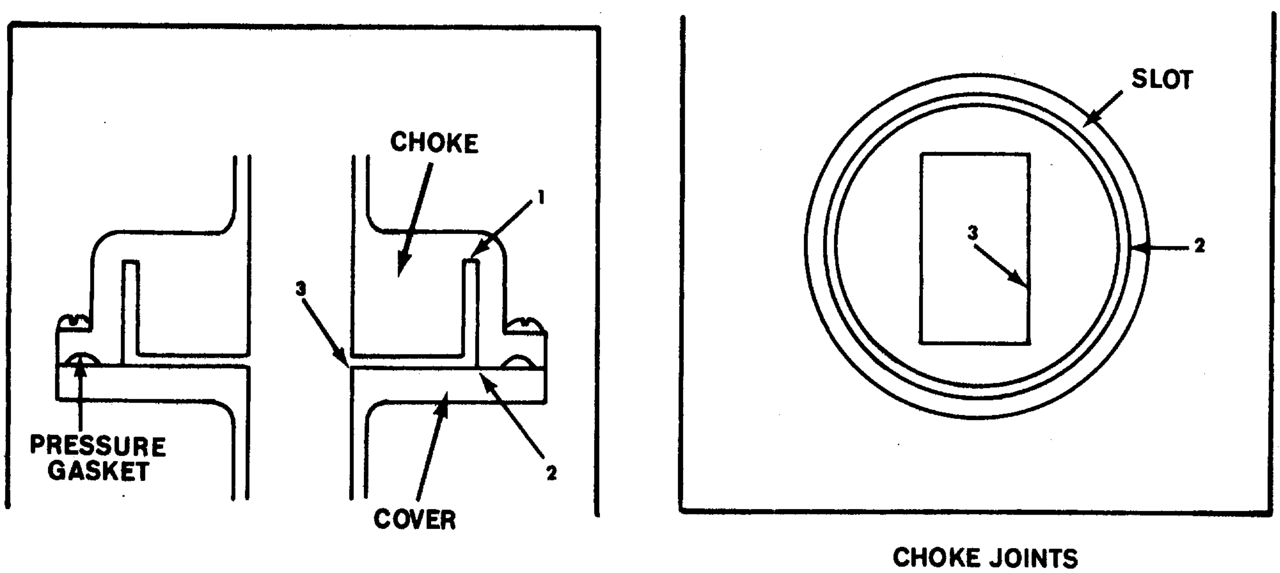


Figure 11-26. Choke Cover Flange.

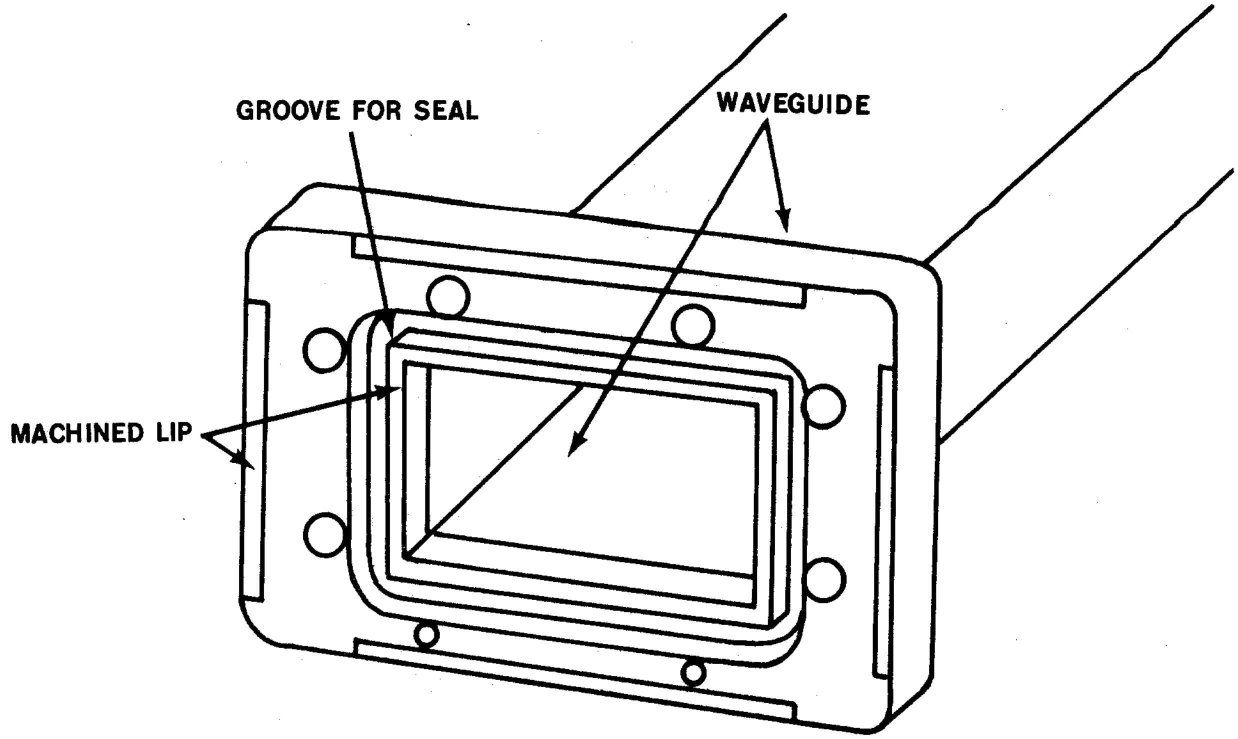


Figure 11-27. CPR Butt Flange.

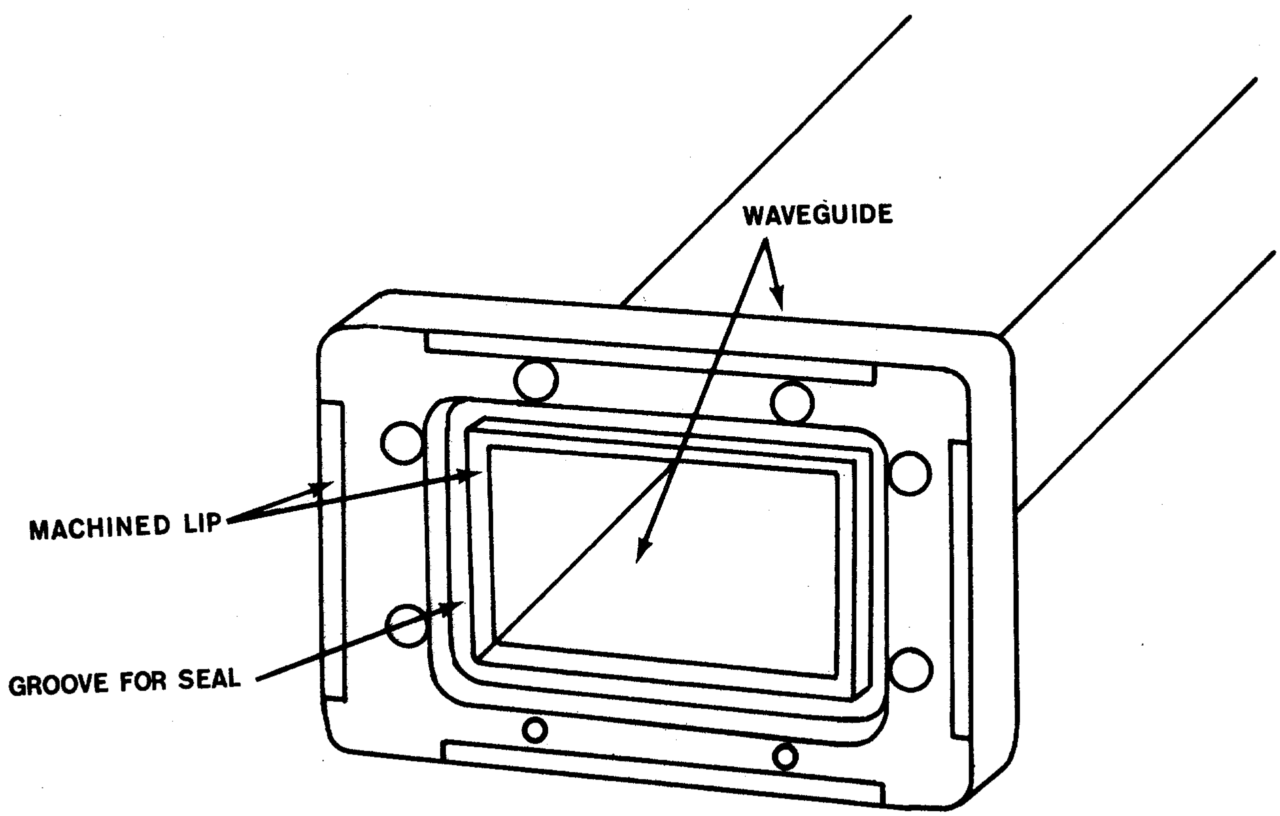


Figure 11-28. CPR Through Flange.

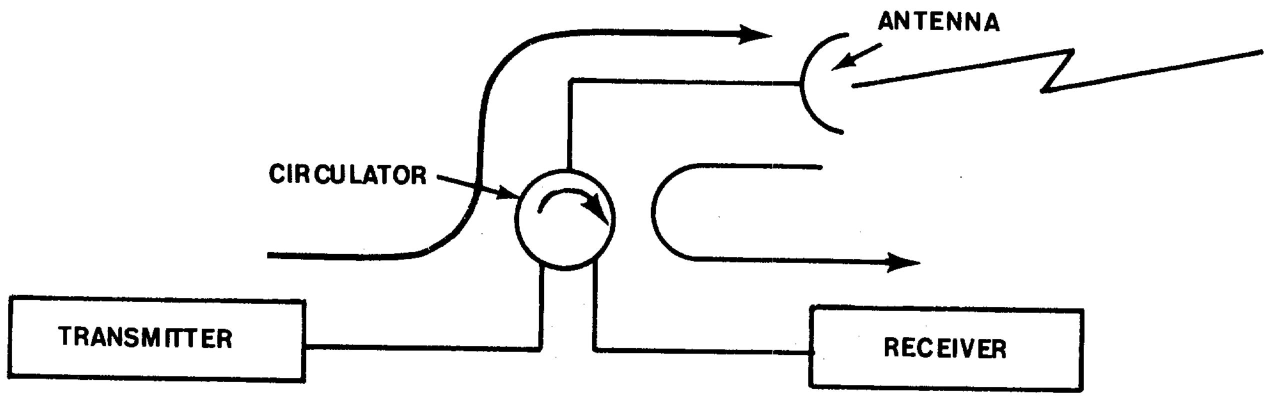


Figure 11-29. Circulator Action Allowing Transmission and Reception on a Single Waveguide.

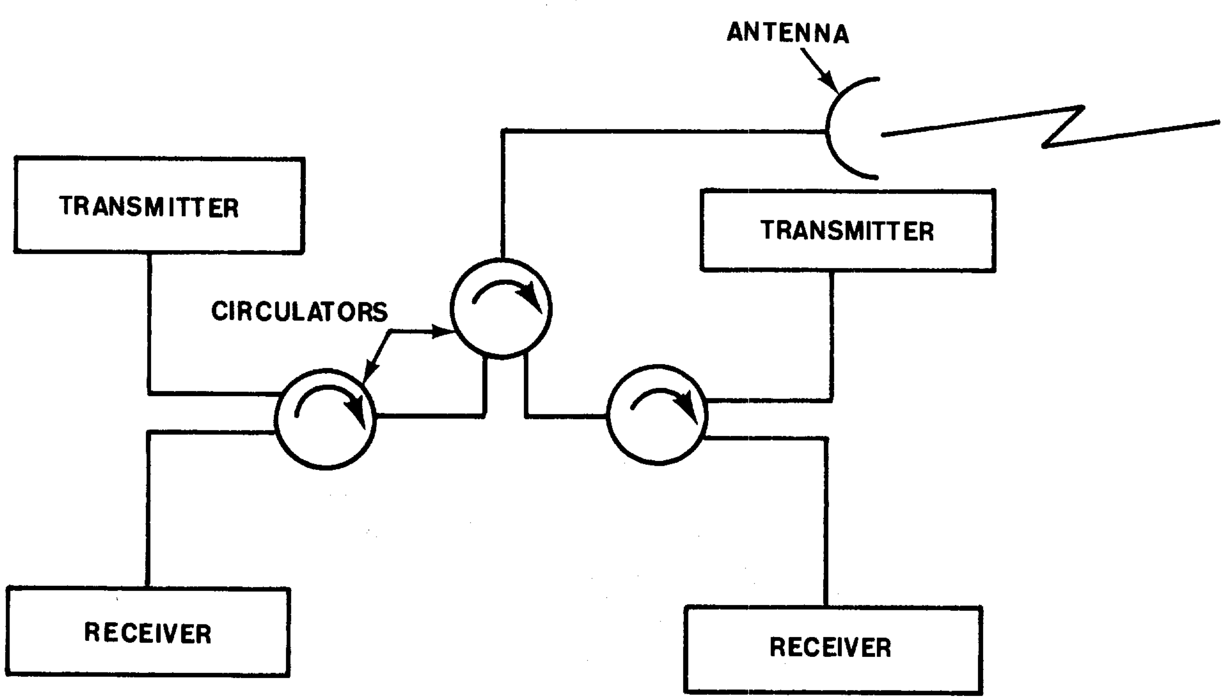


Figure 11-30. Circulator Action Allowing Standby Operation on a Single Waveguide.

11-15. Other Antenna and Transmission Line Components. Since waveguide is expensive, it is desirable to use it for both transmission and reception. Since a magnetic field as well as an electric field is associated with the field energy propagating down the waveguide, magnets can be used to alter the direction of a wave "coming" or alter its direction differently if it is "going." The device that does this is called a circulator. The action of a circulator is shown in figure 11-29. Standby generation can also operate on the same waveguide to the antenna by the addition of two more circulators (figure 11-30). Of course, there is a maintenance problem with one waveguide. The problem being, the system would have to be shut down in order to test the waveguide.

Isolators allow energy to flow in one direction and absorb energy traveling in the reverse direction. They too work with the magnetic fields of the wave as it travels down the guide, for its operation.

They are especially useful in reducing the reflective effects of certain junctions so that delay and echo distortions may be reduced to acceptable levels.

A method of splitting the signal for measurement, etc., is sometimes required. A device to do this with little reflections, if any, is called the magic "T" (figure 11-31). It can also be used to introduce a signal for balanced mixing.

A device closely related to the magic "T" is the rat race hybrid. The rat race can be used to combine two signals as well as divide one signal in half. Two different signals can be combined to give an additive output at two ports and a difference signal at the isolation port.

In addition, many other types of specialized devices exist - couplers, slots, attenuators, and one-way devices just to mention a few.

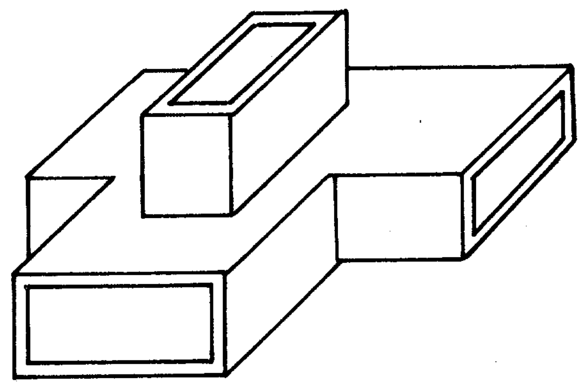


Figure 11-31. The Magic "T."

Chapter 12
TECHNICAL CONTROL FACILITIES

12-1. General. The concept for use of strategically located system management centers, or technical control facilities (TCFs), for military communications systems was implemented prior to the advent of the DCS as known today. Subsequent to the creation of the DCS and the rapid growth of the military system to include multi-channeled wideband links, the need for TCFs became more apparent. These facilities provide the interconnect between the links and, simultaneously, ensure that the customers are provided service by continuously monitoring the circuitry which transits the TCF.

12-2. Function. The major functions of a TCF are technical direction, quality assurance, control, and reporting:

a. Technical Direction. This relates to management of the TCF environment and is further defined as:

(1) Supervision of all transmission links, supergroups, groups, channels, circuits, interfacing equipment, remote transmitter and receiver site, radio relay sites, switching sites, and any other entities integral to the TCF.

(2) Coordination with other connected TCFs and directly connected customers.

b. Quality Assurance. This is the function that actually determines how well the communications system is performing.

(1) Quality Control. Scheduling and performing in-service and out-of-service tests and measurements on all channels, circuits, and equipment in the TCF.

(2) Performance Monitoring.

c. Control:

(1) Determining the reason for outages on circuits, channels, trunks, and groups through fault isolation.

(2) Coordination between customer, maintenance activities, and other TCFs.

(3) Rerouting of circuits or groups and substitution of equipment when tests indicate degradation or failures.

d. Reporting. Each TCF has a responsibility to provide reports to management agencies.

(1) The primary operational manager is the DCA. The DCA management structure is defined by DCA Circular 310-50-5. Reporting requirements are established by DCAC 310-55-1 and serve to ensure that DCA elements are aware of system status.

(2) As the operating and maintenance (O&M) agency, the AFCS management structure also must be aware of system performance. This awareness is accomplished by means of the Navigational/Communications Management Office (NCMO) facility at each operating unit, group, Area, and HQ AFCS. Reporting requirements are established by AFCSRs 100-17 and 100-29, vols I, II, and III.

12-3. Description:

a. TCF. The DCS station TCF is an organic element which functions as the point of interface between the transmission elements of the system, interfaces users with the system, and has the physical, electrical, and manpower capabilities to perform the required functions of a tech control. The size of a TCF will vary in direct proportion to the number and type of circuits, groups, and supergroups for which it is responsible. The manning factor is also based on the size of the facility. All TCF functions and procedures are identified by DCAC 310-70-1, vols I, II, and III. A simplified block diagram is shown in DCAC 310-70-1, vol I.

b. Patch and Test Facility (PTF). The PTF is an organic element of a station or user terminal facility that functions as a supporting activity under the technical direction of a designated TCF. It has the physical, electrical, and manpower capabilities to perform a limited number of the functions of a TCF.

12-4. Technical Control Equipment. Effecting changes in equipment or upgrading of equipment currently installed in AFCS-operated TCFs is a time-consuming task. Many TCFs are using equipment which dates back to the 1958-1963 time frame. Others have more modern equipment, due to the rapidly changing requirements of a specific facility serving a unique mission. The actual function of the TCF remains essentially the same, regardless of equipment age. The equipment within a TCF is designed to permit maximum system visibility and flexibility. Visibility of the system is accomplished by the TCF having the ability to monitor and test every circuit, channel, group, and supergroup entering the facility and all equipment within the facility. Flexibility is gained through conditioning all circuits to bring them up to a common level and having provisions for rerouting any circuit via any equally conditioned channel of any media interfacing at the TCF. Further flexibility is gained through the ability to equalize circuits and substitute various equipments.

a. Patch Bays. In order to properly monitor and manage systems and circuits, the tech controller requires fast and easy access to various points in every circuit and to every item or set of equipment interfaced with the TCF. He/She must be able to measure the parameters of the circuits and equipment at any time and to quickly make temporary equipment substitutions or circuits reroutes, if necessary. The only practical means of doing this is to connect the circuits and equipment to jacks at every point where access may be required. Access is then gained through the use of plug-in cables called "patch cords."

Terminating each circuit and item of equipment into one jack was not completely satisfactory, so a group of jacks (called a "jack set") was devised for each point where circuits or equipment were to appear. Two examples of this are in figure 12-1. As you can see, the signal from the link is routed through the normally closed contacts of the MOD OUT (modulator out) jack to the EQ IN (equipment in) jack. Again, it passes through the normally closed contacts and on to the next point in the TCF. This procedure of using the closed contacts within the jack to avoid the requirement for a patch cord (when the circuit is in its normal configuration) is referred to as "normal through" wiring and is used extensively in audio and DC patch bays. When the plug of a patch cord is inserted into one of the "patch" jacks, the contacts open and the circuit is extended through the patch cord. This permits either termination of the circuits or equipment or rerouting of the circuit on any other channel or spare equipment via a similar jack appearance.

Two "monitor" jacks are also provided. Where this particular jack set is designed for audio use, the monitor jacks are connected in parallel to permit bridging of the circuit with a high impedance measuring instrument. DC patch bay jack sets may have the monitor jacks wired in series with the circuit to permit measurements with low impedance measuring instruments (such as milliammeters). Some DC patch bays employ parallel monitor jacks, thus high impedence measuring equipment must be employed. The jack sets found in the different patch bays in different sections of the TCF may be labeled differently or even have a different number of jacks, but the concept of the "jack set" will remain the same.

The jack sets we are discussing are designed for 4-wire audio circuits. Many patch bays have two additional jacks in each set designated SIG (signaling). These jacks are electrically isolated from the 4-wire audio circuits and provide a normal through path for the DC signaling circuit associated with the audio circuit. When these signaling jacks are installed, the jack sets are then referred to as "6-wire" sets.

Jack sets are usually installed in 19" racks in strips, referred to as a patch panel, with 24 jacks across. Two or more of these patch panels mounted in a rack are referred to as a patch bay.

All patch bay configurations, functional capabilities, and the labeling of appearances are becoming standardized to permit common operational procedures. The following discussion deals with various patch bays found in a TCF.

(1) Supergroup-Group Patch Bay. This patch bay will only be found in a TCF which is collocated with wideband facilities, either radio, cable, or satellite. All VF multiplex equipment is cabled to this patch bay. The bay provides jack appearances for baseband, supergroup, and group connections to the multiplex equipment, as well as baseband inputs and outputs from the radio or other wideband transmission equipment. Bandpass and bandblocking filters and radio order wire inputs and outputs also appear on this patch bay. Because of the unusual jacks employed on this patch bay, through wiring is often not possible and patch cords and "U links" are employed to provide interconnection. This bay provides the capability to perform measurements at the various multiplexed levels and to substitute equipment quickly. It also provides access to measure group and supergroup pilots and access to the radio order wires. The cabling between the supergroup/group patch bay and the multiplex equipment is special grade, or coaxial-type, cable. Because of the wide frequency range passed, the cabling is kept as short as possible to prevent unnecessary loss and stray noise pickup.

Only in recent years has the requirement for this patch bay in a TCF been realized. As a result, newer facilities have incorporated it in the TCF and it is being added to many of the older facilities. In the past, common practice was to collocate it with the multiplex equipment.

(2) VF Patch Bays. The VF patch bay is an audio patch bay employed in most older facilities to provide access to the VF channels immediately after demodulation and before modulation by the VF multiplex equipment. They permit bridging or terminating measurements to be performed without any conditioning equipment connected. The levels employed by the multiplex equipment are maintained at this patch bay. A +7 dB TLP for demodulator "Out" and a -16 dB TLP for modulator "In" are commonly used. The channels are grouped on the VF patch bay as follows:

(a) By type of transmission media (tropo, M/W, etc.).

(b) By transmission link (Westover to Andrews) where the baseband is broken out to group level.

(c) By supergroup and group.

(d) By multiplex channels within each group in ascending order.

Whereas the patch bay serves to interface the TCF with wideband facilities, measurements concerning the VF channels may be made from this point.

As shown in figure 12-2, the VF patch bay may be collocated with the VF multiplex equipment in some older facilities. Due to the similarity in function of the VF patch bays and the circuit patch bays, the VF patch bays may not be installed in future tech control facilities.

(3) VF Circuit Patch Bay (Equal Level). The circuit patch bay is the focal point of a TCF. All audio circuits transiting or terminating at a TCF appear on this patch bay. It provides access for parameter measurements on all circuits as well as facilities for alternate routing of circuits or substituting equipment. The main advantage of this patch bay is that all circuits appear on it at a common reference level. Circuits often enter the TCF at a variety of reference levels from the various transmission media, but they are conditioned to a common level through the use of the conditioning equipment which will be discussed later in this chapter. The TLP most commonly employed at circuit patch bays is 0 dB, although some facilities use a --2 dB TLP. The bays themselves are of standard audio

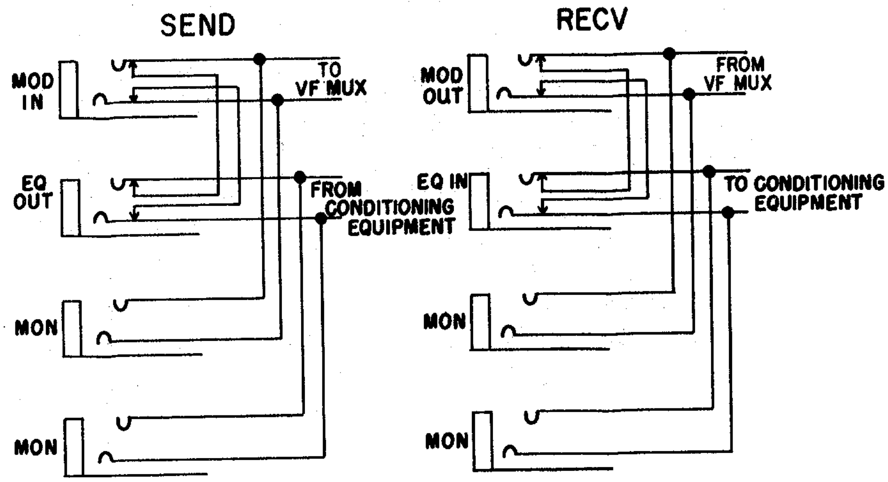


Figure 12-1. Typical VF Patch Jack Sets.

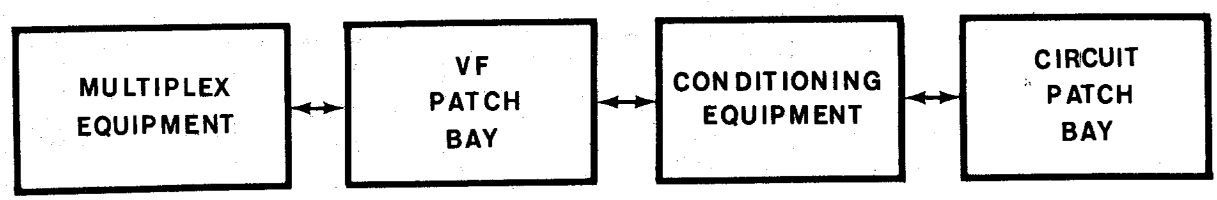


Figure 12-2. Location of VF Patch Bay.

configuration and similar to the VF patch bay. The jacks are designated "Line" and "Drop" instead of "Mod" and "Equipment" but perform identical functions. The circuits appearing on this patch bay may be arranged by:

(a) Multiplex groups having that location as the next VF breakout point, with individual channels in each group arranged in ascending order.

(b) Government-owned landlines to that location.

(c) Commercial landlines to that location.

This patch bay is electrically located in the center of the TCF. It is connected through the conditioning equipment to the VF and cable patch bays and through the VFCT terminal equipment to the DC patch bays.

(4) VF Primary Patch Bay. All cables entering a TCF appear on the primary patch bays—the audio cables on a VF primary patch bay and the DC on a DC primary patch bay. These patch bays interface the TCF with all circuits entering the facility by any means other than VF multiplex. The circuits appearing on these patch bays may be grouped with all cable pairs within the same sheath grouped together or subgrouped according to the end destination of the cable pairs (such as AUTOVON switch, HF receiver site, etc.). The DC patch bay and cables are kept physically and electrically separate from the audio bay and cables to avoid introducing noise (cross-fire) into the audio circuits.

The electrical location of these patch bays is shown in figure 12-3. All circuits from the audio patch bay are connected to the circuit patch bay through the conditioning equipment. The circuits from the DC primary patch bay are connected to the DC circuit patch bays.

(5) DC Patch Bays. DC patch bays are specifically designated to facilitate monitoring, testing, and restoration of DC circuits in a similar manner to the audio patch bays. This patch bay interfaces local DC users and the incoming circuits. As shown in figure 12-4, DC circuits may be derived from cable facilities through the cable patch bay, from VF telegraph multiplex systems (VFCT), and from local DC subscribers, including some TCF teletype order wires.

In some facilities, a RED area PTF may be required to handle clear-text classified traffic before encryption. The DC patch bays in this area are physically and electrically isolated from the BLACK encrypted and unclassified areas.

b. Distribution Frames. A circuit transiting a TCF will pass through at least two patch bays and a certain amount of conditioning equipment. It would present a moderate wiring task to simply interconnect all of the associated equipment for one circuit. Multiply this task by the number of circuits in a TCF (often-

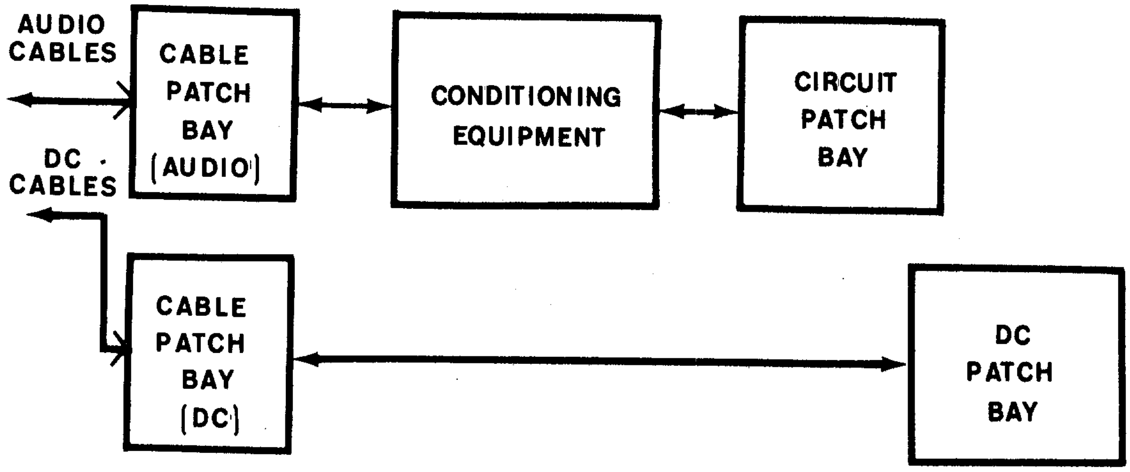


Figure 12-3. Location of the Cable Patch Bay.

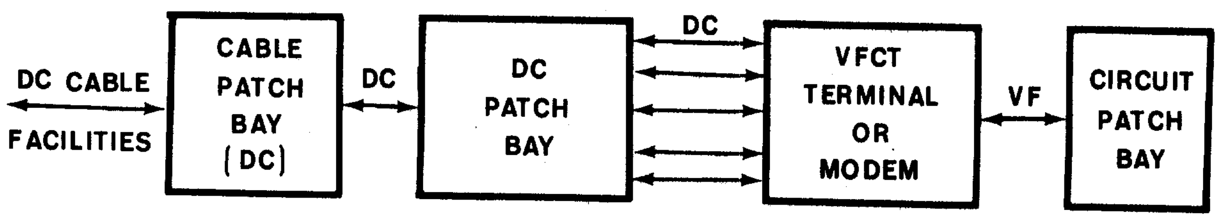


Figure 12-4. Location of the DC Patch Bay.

hundreds) and it becomes a formidable wiring task. Add to it the often daily requirement of changing the routing of some circuits, activating new circuits, and deactivating others, plus occasional changes in conditioning equipment requirements for each circuit and standard wiring techniques are no longer a task—they are an impossibility. For these reasons, distribution frames are used.

Distribution frames are often an open metal frame with rectangular terminal block mounted on both sides of the frame. On one side, they are mounted horizontally and called "horizontal blocks." On the other side, they are mounted vertically and called "vertical blocks." There are rows of metal "pins" passing through each block. An example of a distribution frame is shown in figure 12-5.

All analog and digital patch jacks are permanently connected by cable to the bottom pins of the horizontal blocks. Loop current power supply sources (battery taps) are also provided here.

All audio and digital cable pairs entering the facility, including all channels derived from wideband equipment and all equipment, except patch bays are connected to the pins on the left side of the vertical blocks. Interconnection of equipment is then accomplished by connecting single wires from a pin on the top of the horizontal block to a pin on the right side of a vertical block. This process is referred to as cross-connecting. All cables and cross-connects may be shielded, with one end of the shield connected to a grounding system wired across the blocks. Accurate records are maintained to identify exactly what is connected to each pin on the distribution frame.

There are many variations of distribution frames. Advances in the state of the art have led away from individual blocks and open metal frames and toward enclosed panels with miniature pins designed for mechanical, rather than solder, connections. Distribution frames are subdivided into four classes which indicate only their function within a facility. There are no physical differences between them except perhaps their size.

(1) Main Distribution Frame (MDF). This is the paimary distribution frame for a communications complex and is located adjacent to the TCF. The term "MDF" usually indicates that there are other distribution frames in the complex.

(2) Intermediate Distribution Frame (IDF). The IDF is a smaller version of the MDF and is usually found at PTFs within the same complex as the TCF. It may also be found in larger equipment areas to interconnect the various equipments within that area. The

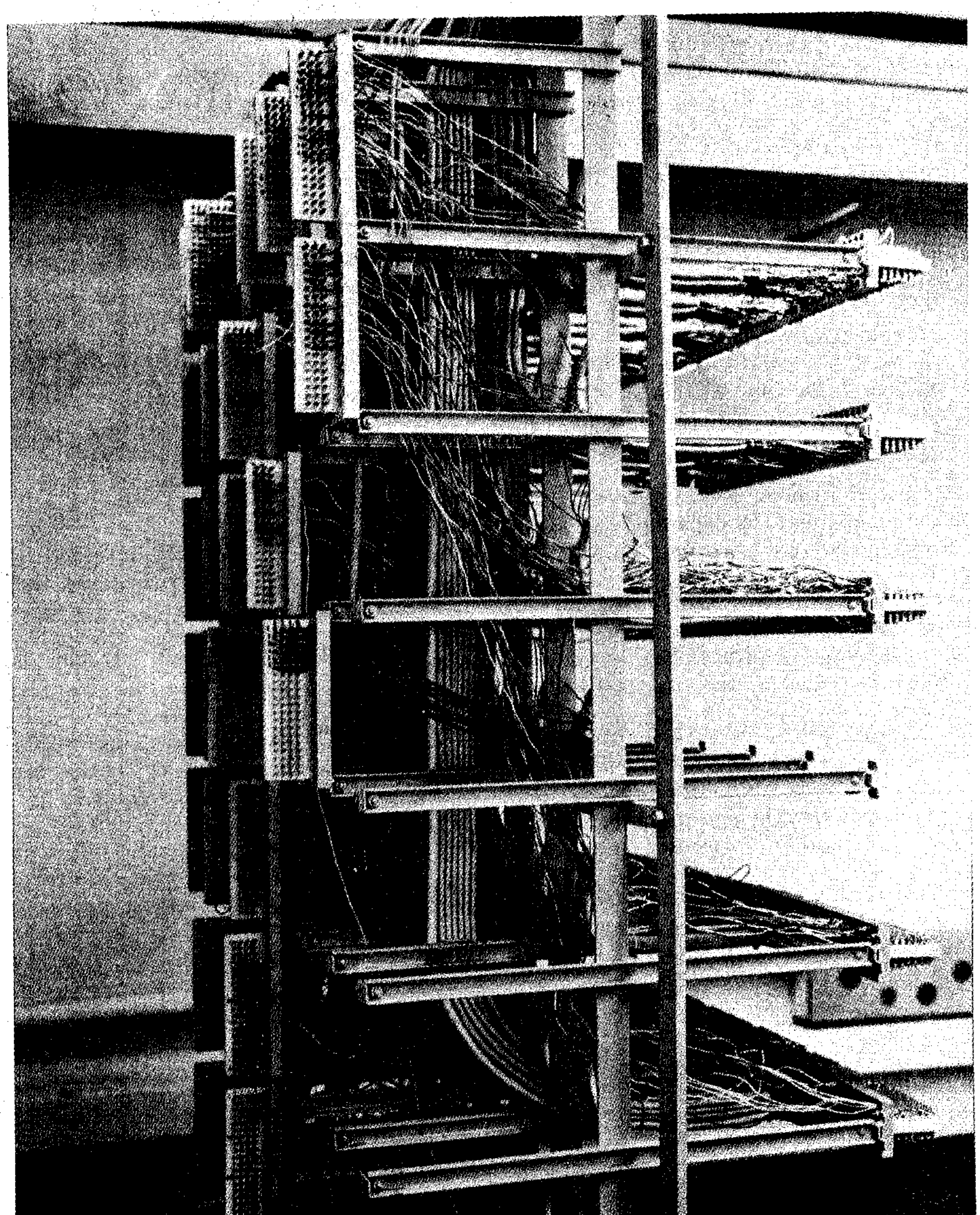


Figure 12-5. Sample Distribution Frame.

IDF is cabled to the MDF for access to the TCF.

(3) Combined Distribution Frame (CDF). The CDF fulfills the function of MDF and IDF at a complex. The term indicates that it is probably the only distribution frame within the complex.

(4) Red Distribution Frames. These frames are installed at facilities passing classified data in clear text. All circuits passed through this frame are encrypted prior to being routed to the MDF.

c. Circuit Conditioning. Though this is a term which has only recently come into use, it is one with which technical controllers are becoming increasingly concerned. "Condition" as a verb, means to "bring about the desired state or situation." "Circuit conditioning," then, can be interpreted as "the act and art of bringing circuits to the desired state or level of performance." The TCF has always been concerned with circuit conditioning; it simply hasn't always been referred to as such in the past. There are many differences between what is now known as circuit conditioning and what has been done at the TCF in years past to establish and maintain the required degree of circuit quality.

Much has been accomplished toward increasing the speed of communications and the capacity of our communications system. The trend is steadily going upward. Some of our newer and more sophisticated systems (AUTODIN and AUTOSEVOCOM, for example) have made possible tremendous increases in speed and volume of communications. But, by the same token that the transmitting and receiving devices have been engineered to accomodate the required volume and speed, the channels which link them must also be engineered to accept increased "bit" rates and reduce distortion to amounts compatible with the increased speed and volume capabilities of the transmitting and receiving devices. As might be expected, this has resulted in a need to place ever-increasing emphasis on circuit quality. Circuit quality, which was entirely adequate for older and slower methods of communications, is no more adequate for the newer methods than the horse-and-buggy dirt roads are for our modern automobiles.

Circuit conditioning, in the context in which it is used here, relates not only to the improvement of the quality of a circuit but also to any intentional change brought about in the parameters of a channel in the TCF. The most common form of conditioning is to insert devices to change the level parameters to permit interfacing at a common level. Signaling mode conversion is also a form of circuit conditioning.

Systems are designed to enable most circuit conditioning elements to be of a passive nature (that is, not requiring a power supply). Pads and amplitude equalizers are passive. The more complex conditioning equipment, (such as signaling conversion units and envelope delay equalizers) are active elements and require power. The obvious advantage of the passive elements is their much lower failure rate.

There are many factors which contribute to the degradation of a signal. The TCF is always concerned with techniques for overcoming the conditions which contribute to the distortion of signals. High-speed methods have introduced problems of their own. A consideration of some of the techniques which such methods involve will perhaps give an additional insight to the nature of the problems.

(1) Frequency-Shift Keying (FSK). This is the most common method of transmitting digital information in audio form. In FSK, two frequencies are used for passing intelligence. At any given point in time, only one frequency would be present. The frequency used would be determined by the digital input, the Mark (1) being one frequency and the Space (0) being the other.

(2) Phase-Shift Keying (PSK). This is another method of transmitting digital information in the analog form. In PSK, digital information is encoded as different phases of a single frequency that is constant in amplitude. Through the use of this method, bits of information can be impressed on the frequency at extremely high speeds.

(3) Parallel Transmission. Another method for increasing the speed of digital communications involves the use of serial-to-parallel and parallel-to-serial converters which accept a serial DC input. More advanced systems store the bits until all bits of a digital character are received and then transmit a parallel DC output; that is, all bits of a character are released simultaniously, rather than one behind the other. Many existing systems divide the serial stream into only two parallel bit streams. The parallel DC outputs key a converter, which simultaneously transmits different frequencies to the VF channel by either FSK or PSK. Each frequency represents one bit of a data character. At the receive terminal, the frequencies are translated back into their original bits of DC information.

When the channel passes the information exactly as it enters the transmit end of the system, there is no problem in extracting the information at the receive end. But this is not always the case. Because of reactive components of bandpass filter networks associated with VF multiplexing equipment, some frequencies are delayed, phase-shifted, or attenuated more than others. This, together with noise impulses and various types of distortion, causes errors in the transmission of the high-speed digital information. Thus, for error-free transmission, some means for correcting or compensating for the circuit-degrading factors must be provided.

d. Conditioning Equipment. In order to provide the correcting and compensating adjustments made necessary by the high-speed systems, numerous items of test and interface equipment have been developed. Such equipment has come to be known as "circuit conditioning equipment." In many stations, it is contained in special racks designated as VF channel or circuit conditioning terminals. Such terminals are made up of the various components required to compensate for amplitude-versus-frequency distortion, envelope delay distortion, echos, and level gains or losses. Unfortunately, there is no means for eliminating noise, but compensating for the other signal-degrading factors minimizes the noise problem.

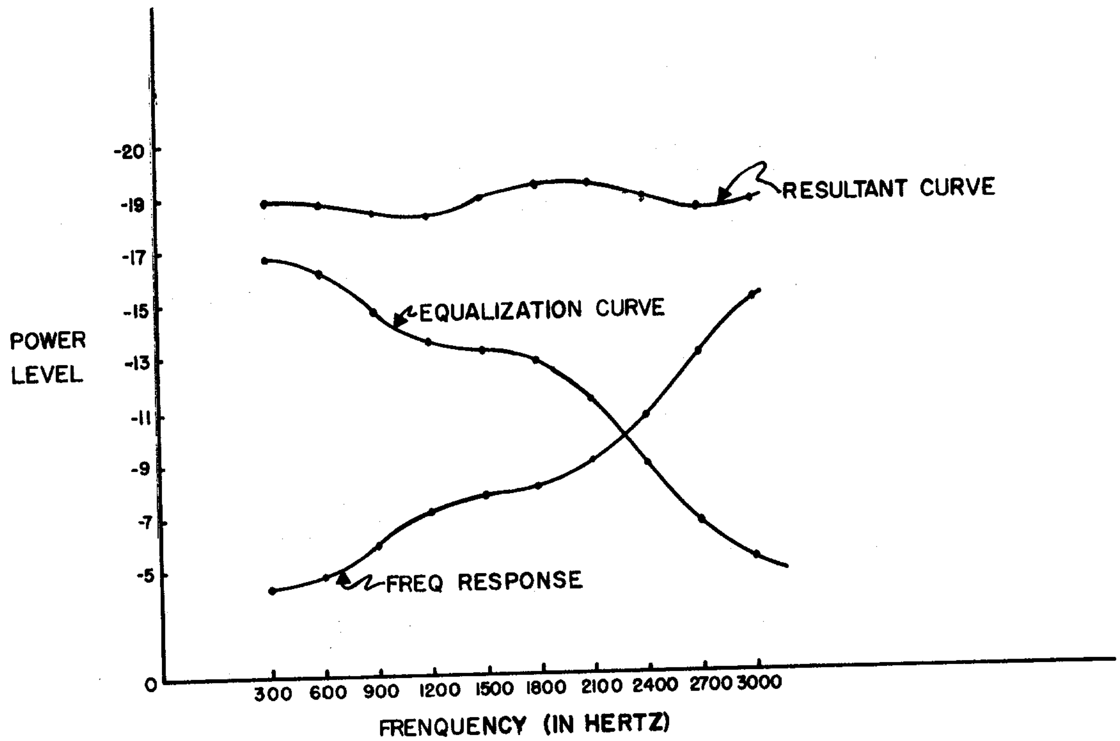


Figure 12-6. Cable Pair Frequency Response Curve.

(1) Amplitude Equalizer. The purpose of the amplitude aqualizer is to compensate for frequency-related amplitude variations in VF communications channels. The theory of amplitude-versus-frequency distortion was discussed in chapter 4 and it was pointed out that VF multiplex equipment seldom causes this type of distortion. Cable facilities may, in certain instances, require amplitude equalization. When required, it is accomplished at the TCF before the cable circuit is interfaced with other transmission facilities. In figure 12-6, a frequency response curve of a cable pair is shown. As can be seen in the graph, as the frequency is increased, the attenuation is increased. This is caused by the distributed constants (R, C, and L) of the cable reacting to the change in frequency. An amplitude equalizer compensates for such unequal losses by inserting additional losses. This additional loss can be adjusted to be greater at lower frequencies than at the higher frequencies. The resultant curve is a frequency attenuation characteristic that is relatively flat from 100 Hz to 5000 Hz.

To enable it to accurately compensate over the operating bandwidth, the applitude equalizer is normally provided with both a high and a low frequency correction network, each having a separate control. Because equalization is accomplished by properly inserting additional losses into the circuit, the resultant output from the amplitude equalizer may be too low to be used directly. A line amplifier may be used to raise the output to the established operating level.

(2) Delay Equalizer. The purpose of a delay equalizer is to compensate for delay distortion resulting from unequal phase-shift characteristics of a VF channel passing signal such as facsimile or data. (Envelope delay distortion is discussed in chapter 4.) A typical delay equalizer is shown in block diagram form in figure 12-7. This delay equalizer is a solid state device consisting of six printed circuit cards. Notice that it contains an input card, an output card, and four cards that are identical to each other with the exception of the delay networks. There are 12 networks that offer delay corrections for frequencies between approximately 500 Hz and 2900 Hz. Each card has two tuned tank circuits, each for a different resonant frequency. Each tank circuit will provide up to 500 microseconds delay to 100 Hz above and below the resonant frequency. Of course, all frequencies would be affected as they pass through every tank circuit.

With all 12 delay networks in use, a channel with 3 milliseconds relative delay can be equalized to within 80 microseconds relative delay. Depending on the channel being conditioned, any or all of the delay networks may be used. As can be readily seen, to delay-equalize a channel may require many hours of exacting work to achieve the best possible results.

The delay equalizer compensates for delay distortion by phase-shifting all frequencies by an amount which, when combined with the delay introduced by the transmission facility, equals the most delay introduced by the transmission facility to any one frequency. Figure 12-8 shows the delay response curve of a typical VF channel, along with the delay (reciprocal) curve of the

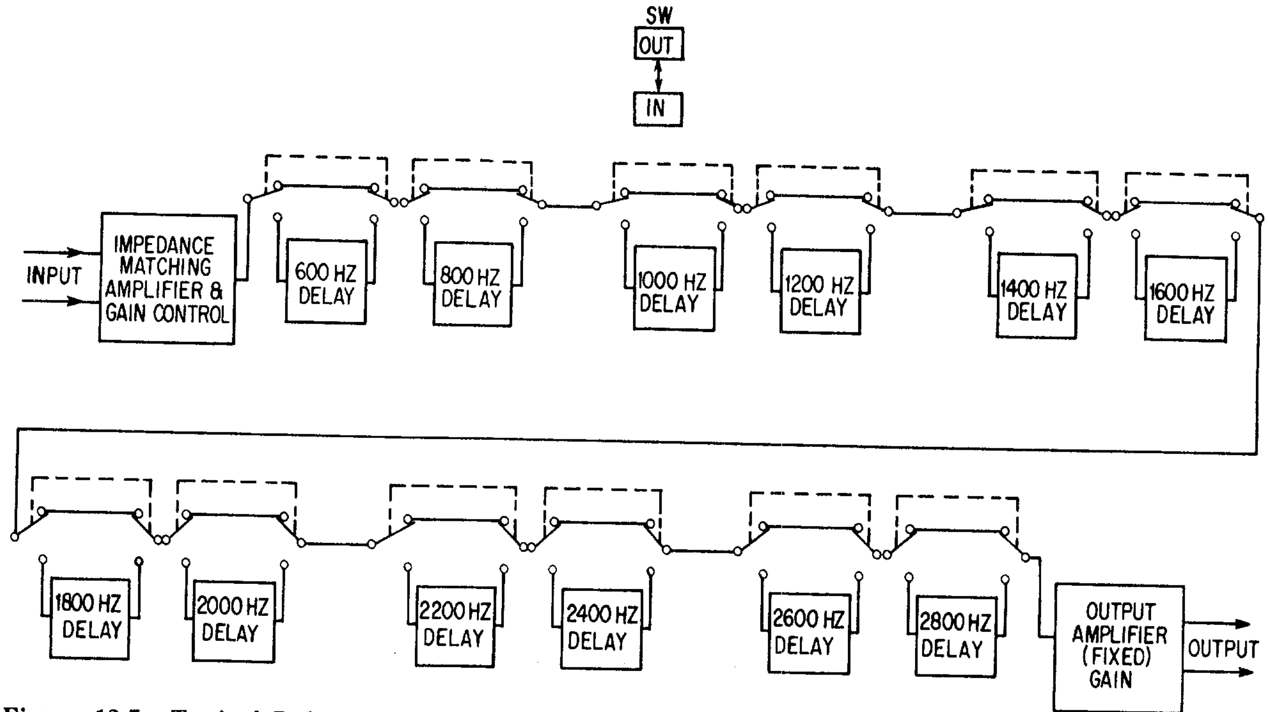


Figure 12-7. Typical Delay Equalizer.

equalizer needed to achieve the resultant delay curve.

The input and output cards also have the amplifiers used to compensate for the signal amplitude loss caused by inserted delay corrections. The input amplifier has an adjustable gain control that allows adjustment for unity gain at the output. The output amplifier has a fixed gain and is not adjustable.

(3) Echo Suppressor. Long-haul point-to-point circuits are especially subject to echoes. Perhaps you have made a long-distance telephone call on the commercial telephone system and were able to hear your own voice return after each spoken word. This is an example of "echo." One type of circuit conditioning equipment is designed to suppress echoes or to prevent them from occurring. This device is aptly named the "Echo Suppressor."

Echo is the effect of a reflected wave which can be noticed by a listener, a talker, or both. If an echo has sufficient magnitude and delay, it may be particularly troublesome to users of the circuit. The effect of echo signals is determined by overall circuit losses. The most common cause of echoes is transhybrid losses in a hybrid transformer. If the impedance of the 2-wire telephone line is not perfectly matched to the 4-wire multiplex equipments, there will be transhybrid loss from the 4-wire receive side to the 4-wire send side.

An echo suppressor is a VF send-receive device which functions primarily on the 4-wire side of a 2-wire/4-wire voice circuit. The function of the echo suppressor is, as its name implies, to suppress echoes or, more technically, to suppress retransmitted time-delayed signals that are inherent in long-haul cable of VF multiplex type communications circuits. It operates in three modes: unidirectional (send or receive), bidirectional (simultaneous send and receive), and data bypass. It is normally used at both ends of a transmission facility, as shown in figure 12-9.

(a) Unidirectional Operation. This is in effect when transmission is attempted in only one direction at a time. The unidirectional mode can be broken down into a send mode and a receive mode:

1. Unidirectional Send Mode. This mode provides no attenuation in the send path to the distant user and 50 dB attenuation in the receive path to the local user. In figure 12-9A, the echo suppressor on the left is in the send mode. The 50 dB attenuation in the receive path will prevent a returning echo from being picked up by the send VF multiplexing equipment and retransmitting it to the distant station.

2. Unidirectional Receive Mode. A look at the echo suppressor on the right (figure 12-9A) will show no attenuation in the receive direction to the user and 5 dB attenuation in the send direction from the user. This condition of the suppressor allows the voice signal from the distant user to pass through it and into the VF multiplexer equipment without affecting it. The 5 dB attenuation on the send path from the local user is a nominal amount used to help reduce the noise received by the user when his receive path is idle.

(b) Bidirectional Operation. Figure 12-9B shows the bidirectional mode of operation. When both echo suppressors are sending and receiving signals simultaneously, both units will assume the bidirectional mode of operation. In this mode, 5 dB attenuation is inserted in both the send and receive paths with each user. Cessation of the send signal from the right-hand unit, for example, would cause the unit on the

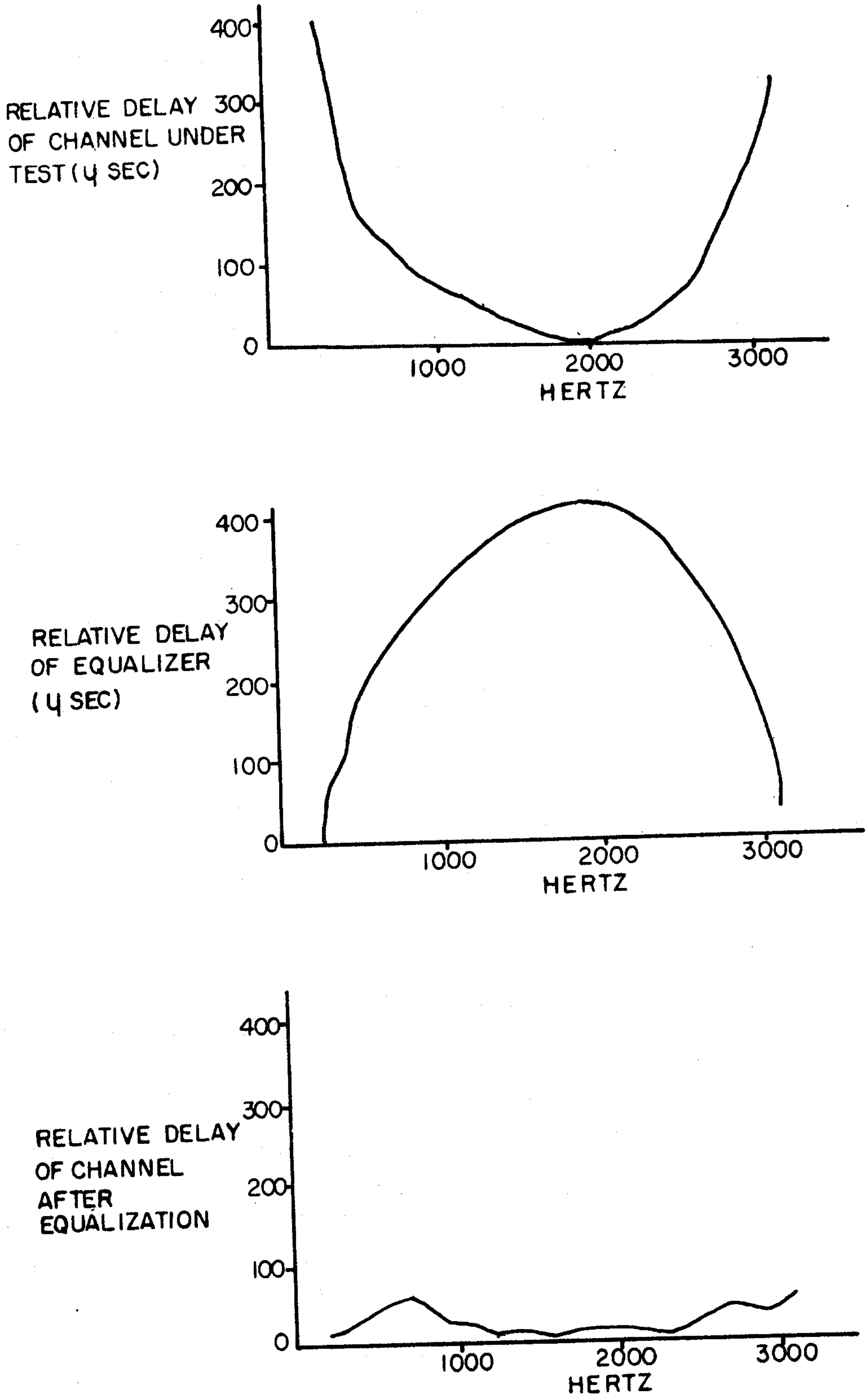


Figure 12-8. Typical VF Channel Delay Response Curve.

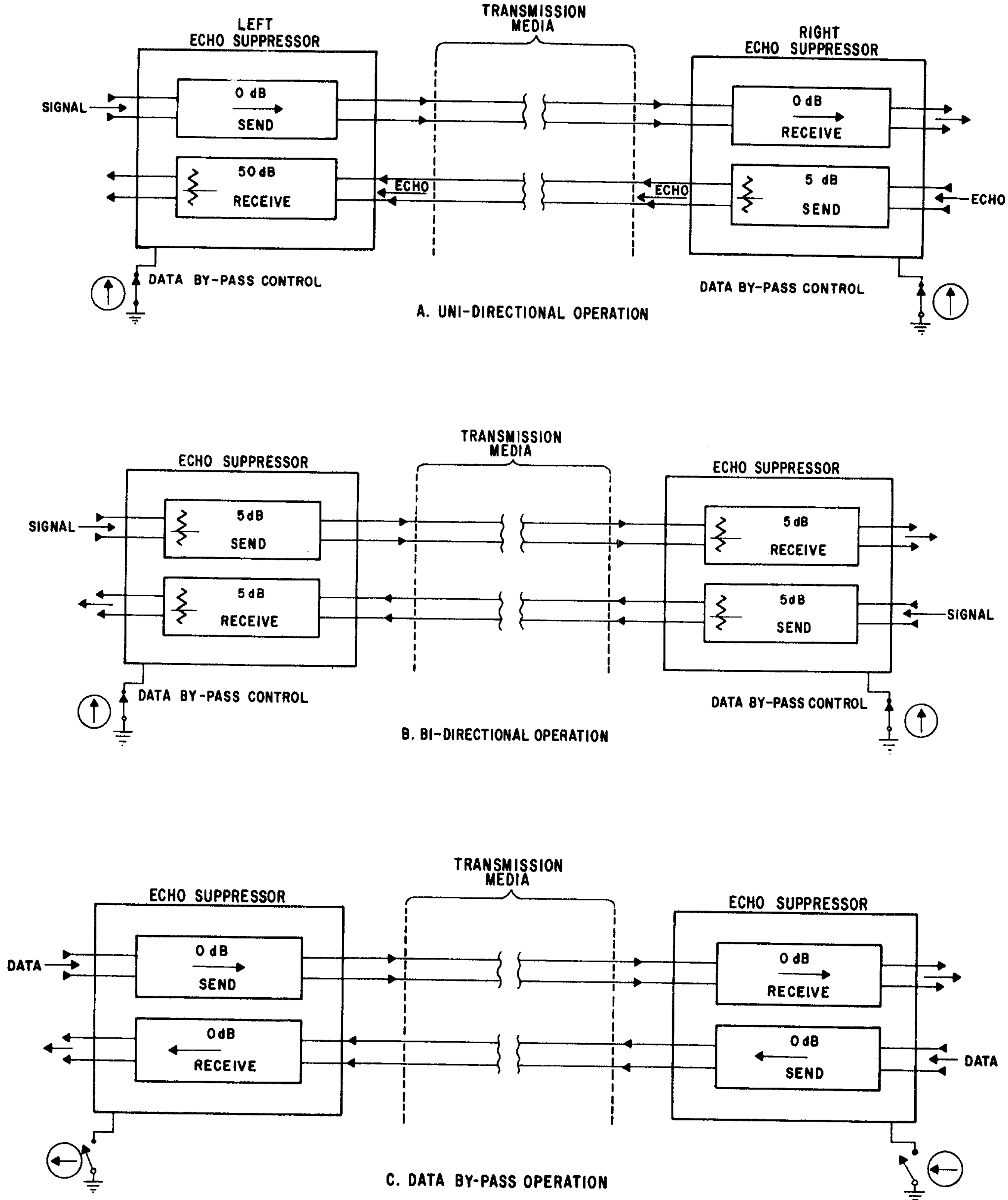


Figure 12-9. Echo Suppressor Application.

right (figure 12-9B) to assume the unidirectional receive mode and the unit at the distant station (left side) to assume the unidirectional send mode. In this way, each unit may switch between the three modes of operation. This switching from one mode to another is controlled by the input sensitivity sections of the echo suppressor. The recommended threshold settings of the sensitivity sections are -26 dBm send and -5 dBm receive.

(c) Data Bypass Operation. In the data bypass mode (figure 12-9C), the ground is removed from the sensitivity section circuitry and the echo suppressor is effectively removed from operation. In this mode, no attenuation is offered in either the send or the receive direction. The data signal is passed through the echo suppressor with no attenuation.

(4) Line Amplifier. The purpose of the line amplifier is to raise signal levels to the common level required for interface. Most wideband systems are designed to provide a 23 dB overall gain on the transmission path; therefore, amplifiers are not normally required to permit interface. In some instances, however, other conditioning equipment employed on a circuit (such as an amplitude-equalizer) may introduce sufficient attenuation to require a follow-up amplifier.

Other circuits entering the TCF through the cable patch bay may be operating at a lower standard level and require an amplifier to raise the level for interface. Amplifiers and conditioning equipment are wired into circuits only if specified by the engineering order for that circuit and are never added to compensate for a fault on a circuit. Line amplifiers are standard audio amplifiers, usually a plug-in module with a variable gain control and a frequency response of ±.3 dB over a 4 kHz bandwidth.

(5) Attenuators (Pads). The purpose of a pad is to permit the insertion of a fixed loss into a balanced 600 ohm line to establish the proper signal level for interface. Pads are the most common item of conditioning equipment and are employed in almost every circuit. Two types of pads are in common use: fixed and strappable:

(a) Fixed Pads. The fixed pad is normally a plug-in module with a fixed amount of attenuation. It is smaller and less expensive than the strappable type and is often used on circuits not requiring other types of conditioning. Because of the standard levels employed in multiplex equipment, most of the pads associated with wideband multiplex at any station will be of two values: one for receive channels and one for send channels.

(b) Strappable Pads. With the increasing requirement for additional conditioning equipment on circuits, different values of pads are required. To meet this increasing requirement, pads adjustable in .5 dB steps from 0 to 31.5 dB are being installed. These pads are adjusted by changing internal strapping and are, therefore, called strappable pads. Figure 12-10 shows a schematic diagram of a typical pad strapped for 16.5 dB loss. Examination of the figure will show six separate resistive networks. These networks can be strapped individually or in combination to obtain the desired amount of attenuation.

e. Signaling Units. Despite the growing trend toward high-speed digital communications, the requirements for voice circuits have not diminished. Many advances have been made in the past few years, such as the advent of the AUTOmatic VOice Network (AUTOVON) and the increasing use of AUTOmatic SEcure VOice COMmunications (AUTOSEVOCOM). These, together with the large number of dedicated voice circuits, comprise a major portion of the DCS. Regardless of the type of voice circuit or the user, one thing is common: they all require a form of signaling. Many forms of signaling are employed by the various types of voice circuits. The only types we will discuss in this chapter are those which will apply to wideband transmission facilities. Two types of signaling are commonly employed with user voice equipment: E&M and 20 Hz.

The earlier technique involved placing a 90 volt low frequency (20 or 50 Hz) signal directly on the talk circuit. The difference in the potential of the ring and the speech made discrimination between the two quite simple for the receive instrument. This type of signaling is still in use on many dedicated DCS voice circuits and for many tactical circuits. This type of ringing proved satisfactory for signaling between two users or with common user switchboard systems, but more sophisticated systems (such as dial) required a more sophisticated signaling system.

The newer system involves the use of a separate pair of wires on which the signaling can be passed. To identify the direction of signaling on this signaling circuit, the letter "E" was extracted from the word "receive" to identify the receive signaling level. The letter "M" was extracted from the word "transmit" to identify the transmit signaling lead. These leads are also sometimes called "Ear and Mouth" to relate to the direction of the signaling. Each of these leads will be in one of two states, one state representing an "on-hook" or idle condition and the other an "off-hook" or in-use condition.

The "E" (receive signaling to the instrument) lead will be open in the on-hook condition and grounded in the off-hook condition. The open or ground is provided by the SF signaling unit if the receive is derived from a transmission media or from the switching center if it is local.

The "M" (transmit signaling from the instrument) lead will be grounded in the on-hook condition and has a -48 VDC potential in the off-hook condition. The ground or -48 VDC is provided by the instrument if it is local or by an interface unit (such as a pulse link repeater) in the case of interfacing two communications media.

Unique signaling functions (such as dialing) are transmitted by pulsing between the two states on the "E" or "M" lead. It is possible to pass the E&M signaling over the same cable pairs as the 4-wire voice circuit through the use of DX1 and DX2 signal lead extension circuits. This equipment uses the neutral (phantom) circuit concept to accomplish this.

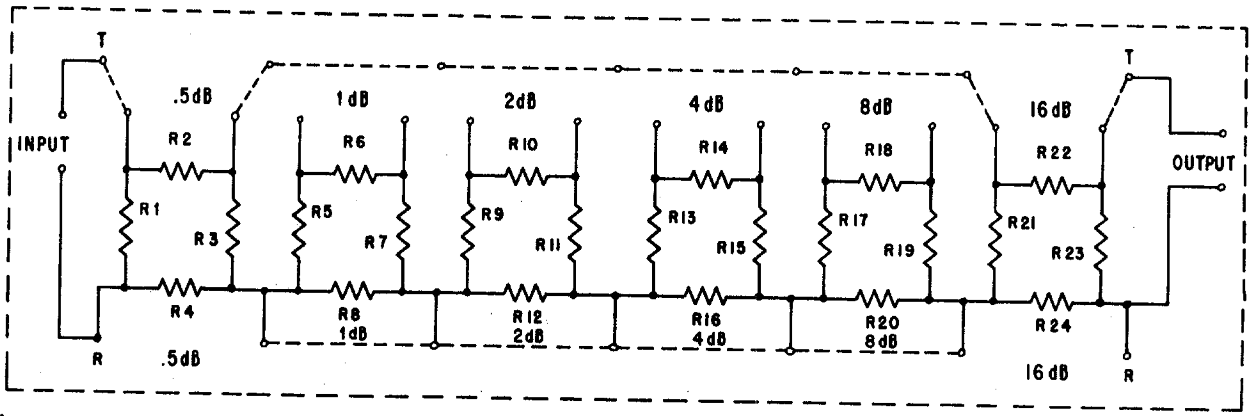


Figure 12-10. Typical Pad Strapped for 16.5 dB Loss.

Because neither E&M nor 20 or 50 Hz signaling are within the 300 to 3000 Hz VF spectrum, they cannot be transmitted directly over VF multiplex channels. They must first be converted to another signaling mode. The two basic modes employed with VF multiplex systems are in-band and out-of-band signaling.

In-Band Signaling. Signaling is accomplished with this mode of inserting a tone within the 300-3000 Hz audio spectrum. Frequencies commonly employed are 1600 Hz on older systems and 2600 Hz on newer systems (such as AUTOVON). The actual signaling with the tone differs with different equipment and will be described in the following paragraphs. In-band signaling may be employed on any VF transmission media. It should be noted that most in-band signaling equipment employs a form of band-block filtering at the signaling frequency; this precludes the use of the circuits with this equipment installed for digital data transmission.

Out-of-Band Signaling. This mode is employed exclusively with VF multiplex systems. It is accomplished by inserting a tone in the guard band between the VF channels. A frequency of 3850 Hz is sometimes used. With this type of ringing equipment installed, the channel may be designated for alternate voice/data operation. The out-of-band signaling equipment is usually either an integral component of the VF multiplex equipment or is collocated with it. In either case, the equipment is designed to generate an out-of-band tone when it receives a signal from the TCF. That signal may be either the separate DC signaling lead ("M" lead) or a 20 or 50 Hz ring on the VF circuit. Conversely, when the out-of-band tone is received from the distant end, it causes the out-of-band signaling unit to generate either a DC voltage on the separate DC signaling lead ("E" lead) or 20 or 50 Hz on the VF channel.

In-band signaling units operate on the same principle as the out-of-band units. The main difference is that the signaling frequency applied to the line is within the 300-3000 Hz spectrum.

(1) SF Signaling Unit. This is the unit employed for AUTOVON circuits. It accepts DC supervisory signals or DC signaling on separate signaling leads (E&M) and generates a 2600 Hz tone which is introduced into the voice circuit. For AUTOVON (and primary alert system (PAS) circuits which use similar equipment), the 2600 Hz tone remains on the circuit when it is idle (not in use). Circuits employing these units are sometimes described as "tone-on-while-idle" circuits. When the correct transmission level for these tones is known, they provide a valuable in-service quality assessment standard for both tech controllers and maintainers. Tone-on-while-idle serves another valuable function in that the user is immediately alerted when the transmission path fails.

(2) TA-182 Ringer. This is the most popular of the earlier generation of signaling units. It accepts a 20 or 50 Hz signaling in the voice channel and generates a 1600 Hz tone in the voice channel. With this type ringing, the 1600 Hz tone is applied to the circuit only during the actual ring. This equipment is still in wide use in common-user switch-board circuits and many dedicated voice circuits; however, it is being phased out in favor of the SF units.

(3) Signaling Extension Units. These units are also incorporated in the conditioning equipment to interconnect two circuits using E&M signaling when the physical distance exceeds the operational limits of the SF units. Signal lead extension circuits (coded DX1 and DX2) are required when the connecting path exceeds 25 ohms. This could occur when interfacing with an on-base AUTOVON subscriber connected to the TCF by intersite cable. These units are not normally required because the SF unit would normally be located at the user's location specifically to avoid this requirement.

(4) Pulse Link Repeater. To understand this device, we must first clarify the "send and receive" concept in a TCF. Figure 12-11 shows a TCF with one East-West circuit transiting it. As you can see, the signal called "receive" from the East becomes "send" to the West and vice versa.

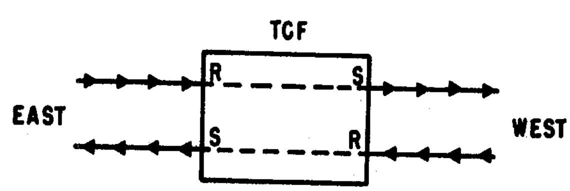


Figure 12-11. TCF Send-Receive Concept.

Because of this configuration, when a circuit employing E&M signaling (perhaps converted from out-of-band signaling) transits a TCF, the discrete functions of the "E" and "M" leads must be interchanged. This is accomplished by inserting a pulse link repeater in the E&M signaling circuit.

f. Alarm Systems. The tech controller is the technical manager of the system and is responsible for monitoring the circuits traversing the system. If equipment failures occur and disrupt portions of the system, it is essential that the tech controller learn of these failures immediately to enable him/her to assess the situation and initiate action to alleviate the impaired service to the users by rerouting or substituting equipment. For this reason, most alarm systems on the communications equipment are extended or remoted to a master alarm panel in the TCF. The number and type of alarms appearing in a TCF will vary, but they must be sufficient to provide the tech controller with an adequate indication of the status of the station and communications equipment employed there. The alarms are divided into two levels which indicate the degree of impairment of the total capability of the facility. Major alarms indicate a loss of more than 10% of the capacity of the cross section, while minor alarms will indicate the loss of less than 10%.

(1) Station Alarms. These monitor power and building conditions in the technical control element. They provide both visual and audible alarms and are capable of being paralleled with remote alarm panels installed in locations within the building complex.

(2) Radio Alarms. These provide status information from unattended intermediate repeater stations. They provide both major and minor conditions. Typical alarms that are transmitted to the alarm center include:

(a) Low transmitter output.

(b) Low receiver input.

(c) High received noise on the radio channel.

(d) Failure of standby equipment.

(e) Failure of power generating equipment (high or low voltage).

(f) Fuse operation.

(g) Low fuel supply for generators.

(h) Open door or window (at unattended stations).

(i) Failure of obstruction or warning light.

(j) Change in waveguide air supply.

(3) Multiplex Alarms. Multiplex terminal equipment alarms include:

(a) High or low-level pilot frequencies.

(b) Failure of standby equipment.

(c) Failure to transfer between operational and standby equipment due to lockout or equipment malfunction.

(d) Fuse operation.

(e) Failure of carrier supply.

(4) Telegraph and Digital Equipment Alarms. Telegraph and digital terminal equipment, including telegraph multiplex and modem alarms, include:

(a) Telegraph distortion exceeding allowable limits on the order wire.

(b) Failure of telegraph carrier supplies.

(c) No transition alarms.

(d) Fuse operation.

(5) VF Terminal Equipment Alarms. These alarms should include failure of signaling supply and fuse operation.

(6) Other Alarms. These include:

(a) Power supply failure/malfunctions.

(b) Antenna tower light failure.

(c) Station intrusion and environmental alarms (as required).

All these alarms should be connected to the station alarm panel, as needed.

g. Performance Monitoring Equipment. Since its inception, the TCF has had the responsibility of maintaining the quality of all signals traversing the systems and directing the localization of failures or faults. To perform these functions, it is obvious that a certain amount of test equipment is required. The earlier concepts of quality control and fault isolation considered an audio oscillator, a VU meter, and a noise meter a complete complement of audio test equipment. A digital distortion-measuring set, a zero center milliammeter, and an oscilloscope were considered a complete test complement for DC work. This equipment was usually installed in the TCF when it was built and, unless an over-build system was installed, no further test equipment could be procured. The basic maintenance concept under which it was installed was that test equipment could be justified only to test end items of equipment. As the state of the art advanced, it was realized that a TCF monitors circuit and system performance more than it tests equipment. The implementation of more stringent requirements for adherence to specific circuit parameters brought with it a new generation of measuring equipment. A table of allowances for "Performance Monitoring Equipment" was established for all AFCS TCFs and PTFs to enable them to procure all needed measuring equipment to augment the installed equipment.

(1) Oscilloscopes. Oscilloscopes are used to permit visual analysis of analog and digital waveforms as well as level measurements, time, phase, and frequency comparsion, and many specialized functions. The oscilloscope is an excellent all-around tool in a TCF and is becoming more popular as tech controllers become more familiar with its capabilities.

(2) Level Measuring Sets. Many different types of level measuring sets are in use in TCFs in the field. Most TCFs use VU meters for signal level measurements as most measurements are made on complex VF signals. DB meters are often available within TCFs.

(3) Noise Measuring Sets. Noise measurements are applicable only when performed using the appropriate noise weighting network. Because of this, "3 kHz Flat" and "C-Message" noise weighting networks are incorporated in noise measuring sets now employed in TCFs. In addition, the noise measuring sets are calibrated in both "dBrn" and "dBm" to permit their use for both noise and signal level measurement. The audio distortion measuring set is not suitable for measurement of harmonic distortion or single tone interference but is valuable for unique troubleshooting applications.

(4) Selective Level Measuring Set. This instrument, also known as a "low frequency selective voltmeter" or "wave analyzer," is used to measure a very narrow bandwidth signal, usually 10 or 100 Hz wide. A detailed discussion of this instrument is in chapter 17. This instrument is suitable for harmonic distortion, single tone interference, longitudinal balance, and unique troubleshooting measurements.

(5) IPN Counter. IPN is one of the major problems on many wideband systems affecting digital transmission. The IPN counter counts the number of impulses which pass certain peak values over a specific period of time. The readings are displayed on three digital counters which can be adjusted for any desired threshold value.

(6) Strip Chart Recorder. This instrument consists of an ink pen which traces on moving graph paper to give a permanent record of the level variations on any audio channel over a period of time. The instrument usually incorporates a log/linear converter so the level variations can be calibrated in dB on the horizontal of the linear graph paper. The vertical direction of the chart represents time. The chart is driven at a fixed rate which can be varied, usually in preset steps. It permits the controller to "watch" a channel or circuit for unusual level variations without having to keep his eye on a meter every second. A number of recordings made on the same circuit at the same time at various breakout points and then later compared permit detailed analysis of tandem-link subsystems or circuits not otherwise possible. Noise can also be recorded in a similar manner.

(7) Impedance Bridge. This instrument can be used to obtain a direct reading, in ohms, of the characteristic impedance of the channel or equipment under test. This saves time and trouble as compared to the other techniques of impedance measurement. The impedance bridge is not yet standard throughout the field but many TCFs, with this instrument or the other technique, are discovering many violations of the terminal impedance standard.

(8) Envelope Delay Measuring Set. This set is one of the more recent additions to the family of audio performance monitoring equipment. It is used to measure envelope delay on audio circuits through the use of a modulated variable audio tone. The measurement can be performed using only the path under test, but more effective use requires a second path on which to pass a reference frequency. This unit may use separate transmitter and receiver units or may be housed in a single case. A frequency counter is incorporated in the unit and the readout is in milli or microseconds at each discrete frequency.

(9) Oscillator. This is another of the basic tools of the tech controller. It is used for measuring net loss, net loss variation, frequency variation, frequency response, change in audio frequency, phase jitter, and for specialized equipment checks.

(10) FSV with Spectrum Display Unit. This is one of the most useful instruments in a TCF. It provides capability for a full range of tests on groups, supergroups, and basebands from the group/supergroup patch bay. In addition, audio channels monitored at the VF, circuit, or cable patch bays can be displayed on the spectrum display unit for detailed analysis of the channel. With this instrument, an experienced tech controller can detect problems in seconds that normally take hours to detect with other standard equipment.

(11) Digital Data Measuring Equipment. TCFs which have digital data (DC) facilities also have additional performance—monitoring equipment for analysis of DC signals, circuitry, and equipment.

(12) Digital Distortion Analyzer. This unit provides direct meter readout in percent of distortion on the DC circuit under test. It also provides an indication of the type of distortion (bias, fortuitous, etc.).

(13) Test Pattern Generator. This unit provides a continous digital DC output or a test message (the quick red fox jumps, etc.), a selected character, or "ACs" (alternating marks and spaces). A selected type of distortion at any percentage value up to 49% may be impressed on the output signal.

(14) Data Analysis Center. An oscilloscope, time base generator, and power supply are usually grouped together with the above two items to form a "Data Analysis Center."

(15) DC Troubleshooting Instruments. DC zero-center milliammeters and voltmeters are often installed in DC patch bays. Formerly, these were the basic DC troubleshooting instruments but, as the full potential of the data analysis center became known, their usefulness diminished.

In conclusion, the amount and type of performance monitoring equipment employed by different TCFs will vary, depending on the size (number of circuits) and mission of the TCF. While the tech controller is tasked with the responsibility of performing various tests with this equipment, it is equally important that every maintainer on the links evaluated by the TCFs be familiar with the test procedures and equipment, for he/she must correct any faults isolated. It is necessary for him/her to be equally familiar with the way in which faults are isolated.

12-5. Order Wire Systems. An adequate order wire network is required for efficient technical control procedures. Information regarding circuit and facility status, circuit order work, and maintenance and trouble conditions must be transmitted over this network. Voice and teletype circuits make up the order wire network and provide the necessary means of coordination between two or more TCFs and between TCFs and special communications users. The number and type of order wires will vary from site to site as dictated by station size, mission, and location. IAW DCAC 310-50-6, three different categories of order wires are to be found in TCFs, PTFs, and users. These categories are explained below:

a. Express Order Wires. Nodal site TCFs serving as Facility Control Office, Circuit Control Office, Special Circuit Control Office, or an Intermediate Control Office for a large number of links, groups, trunks, and/or circuits have at least one express order wire to each adjacent TCF having similar responsibilities.

b. Link Order Wires. TCFs and PTFs connected by a DCS transmission link have at least one link order wire.

c. Local Order Wires. TCFs having subordinate PTFs (and in some cases directly connected users) within the DCS complex under its cognizance will be found to have a local order wire to each facility. A second type of local order wire is noted when a PTF serving as an interface facility between a TCF and users is provided local order wire circuits to the user locations as required.

12-6. Circuit Routing Within the TCF. Many circuit configurations exist for the various types of circuits which are routed through TCF facilities. A standard TCF block diagram as axle standard circuit descriptions and flow diagrams are in MIL STD 188-310, Subsystem Design and Engineering for TCFs, paragraphs 4 and 5. DCAC 300-175-9, enclosure 7, depicts examples of standard circuit power levels.

12-7. Quality Control Documentation, Analysis, and Reports. DCA circulars establish specific requirements for in-service and out-of-service quality control tests and for analysis. Certain tests are required on a scheduled basis, while others are effected when system or circuit failures occur.

Chapter 13
POWER

13-1. General. One of the key contributing factors to the successful operation and maintenance effort of a wideband communications system is the associated power system. A system within a system, it must provide consistent, reliable electrical energy for communications, lighting, maintenance, and environmental control equipment - air conditioners, heaters, etc. Included in our discussion of power systems will be:

a. Voltage.
b. Frequency.
c. Power variation.
d. Load categories.
e. Equipment considerations.
f. Power reliability.
g. Classes of power.

13-2. Function. Simply stated, a power system furnishes power at a voltage and frequency that can be used by the equipment it supports. Primarily, we are concerned with supporting the electronics equipment contained in a facility, subsystem, and system, but power requirements for associated ancillary and auxiliary support equipments must also be included in total system power requirements. In the following paragraphs, we'll attempt to familiarize you with many of the aspects of applied power systems, their peculiarities, applications, and configurations as applied to wideband systems. To start with, a typical example of a power distribution system is shown in figure 13-1.

13-3. Requirements:

a. Voltage. We have a rather peculiar set-up in this day and age in that most electronics equipment currently being built is designed to operate from a 120/208 volt, 3-phase power system. The power plants available, however, have power ratings ranging from 120/240 volts, single phase, to 265/460 volts, 3-phase, with many variations in between. Although much has been done to correct this situation through standardization programs, situations involving installation and interface of equipments in oversea areas have given rise to many unique engineering problems. Peculiar equipments procured for special applications (secure voice systems, crypto, etc.) also contribute to the need for a wide range of power equipment.

b. Frequency. The standard frequency requirement of today's electronics equipment is 60 Hz, although many have the built-in capability of operating also on 50 Hz. Very few, if any, frequency problems are encountered in the United States and Canada, but oversea locations are another story. Since the power furnished by local utility sources overseas is usually 25 Hz or 50 Hz, it becomes necessary to use converters to convert the power to 60 Hz.

c. Power Variation. There are two types of power variations: steady state and transient:

(1) The steady state (long term) variation of the primary power source should not exceed $\pm 5\%$ of the system design voltage and $\pm 0.5\%$ of the design frequency at any load from the minimum to maximum steady state station load. Where commercial power is being used as a primary power source, measurements are made at the point of delivery; thus, for a 120/208 volt, 60 Hz system, acceptable measurements would read 114-125/192-216 volts, 59.7 to 60.3 Hz. Regulation should be provided whenever steady state voltage variation exceeds $\pm 5\%$.

(2) Transient variations are short-term in nature, lasting for 0.5 seconds or less. They are caused by severe electrical storms, lightning, and maintenance. Voltage, when monitored at the point of delivery, should read within -20% and +10% of the nominal value. Measured frequency should read within $\pm 3.3\%$ of the nominal value. Acceptable measurements would therefore read 96-132/166-229 volts with a frequency reading of 58.02/61.98 Hz. These limits should not be exceeded more than five times a month under average operating conditions.

d. Load Categories. Some of the most misunderstood terms concerning power systems are those dealing with the various categories of loads served by the power system. To ensure the use of a common power language, the following terms and definitions are included and are also shown graphically in figure 13-2.

(1) Station Load. The total power requirements of the integrated station facilities expressed in kilowatts (kW).

(2) Nonoperational or Utility Load. Administrative, support, and housing power requirements.

(3) Operational Load. The total power requirements for communications facilities.

(4) Technical Load. The portion of the operational load required for communications, tactical operations, and auxiliary equipment including necessary lighting, air conditioning, or ventilation required for full continuity of communications.

(5) Critical Technical Load. That part of the total technical load which is required for synchronous communications and automatic switching equipment.

(6) Noncritical Technical Load. Remainder of technical load excluding the tactical load.

(7) Tactical Load. The part of the technical load required by the host service consisting of weapons, detection, command control systems, and related functions.

(8) Nontechnical Load. That part of the total operational load used for general lighting, air conditioning, ventilating equipment, etc., for normal operation.

(9) Demand Load. The demand load of a facility is the sum of the operational, including any tactical load and nonoperational demand loads, determined by applying the proper demand factor of the connected loads and a diversity factor to the sum total (figure 13-3).

(10) Operational Demand Load. The sum of

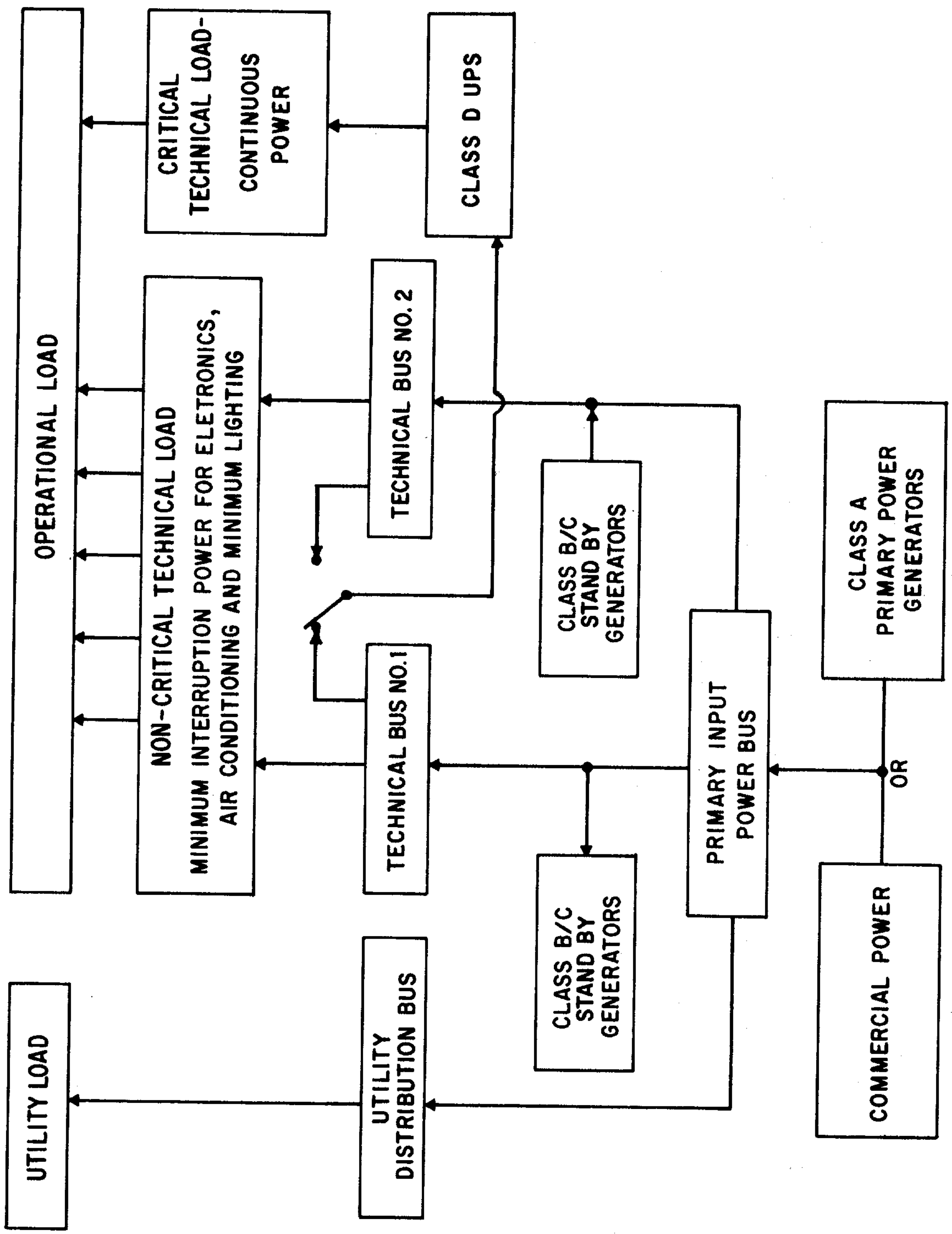


Figure 13-1. Typical Power Distribution System.

the technical demand and nontechnical demand loads of an operating facility.

(11) Technical Demand Load. The technical demand load for each type of facility is determined IAW the following definitions:

(a) Transmitter Facility Demand Load. The load of all transmitters and auxiliary equipment which may be operated on prime or spare antennas or dummy load simultaneously, the load of those in standby condition, control, and keying equipment load, plus lighting, ventilation, and air conditioning load required for full continuity of communications.

(b) Receiver Facility Demand Load. The load of all receivers and auxiliary equipment which may be operated on prime or spare antennas simultaneously, the load of those in standby condition, multicouplers, control and keying equipment, plus lighting, ventilation, and air conditioning load required for full continuity of communications.

(12) Nontechnical Demand Load. The load of ventilating equipment, shop lighting, and other support items which may be operated simultaneously with the technical load.

(13) Demand Factor. The ratio of the maximum demand on a system to the total connected load of the system. (The maximum demand is usually the integrated maximum kilowatt demand over a 15 or 30-minute interval, rather than the instantaneous or peak demand.)

13-4. Equipment Considerations. A power system is comprised basically of three types of equipment: generating, transforming, and switching. The general relationship of these equipments to a load is shown in simplified form in figure 13-4.

In this arrangement, the load is normally furnished power from an off-facility high voltage source. This voltage is then reduced to a value required by the equipment through step-down transformers and is then fed to the load through switching equipment (breakers, buses, etc.). In the event of a power failure of the off-facility source, the load can be furnished power from a standby generator through switching equipment.

a. Generating. The generating equipment most commonly used consists of diesel engines driving AC alternators. The size and type of the diesel engine is dependent on the size of load served and the operational requirements (table 13-1).

b. Transforming. Figure 13-5 shows typical transformer arrangements and some of the variety of voltages presently used in support of wideband facilities. Figure 13-5(a) shows a typical 3-wire, single-phase arrangement. This type of arrangement is used when the motor loads and equipment loads are small. Figure 13-5(b) shows a 3-phase Y or star connection. Systems connected in this manner may carry a Y designation (that is, 120/208Y). This is the most common connection in use. It is capable of serving 3-phase loads by connecting to the 3-phase wires and large, single-phase loads by connecting between any phase wire and neutral. Figure 13-5(c) shows a delta connection. It is used when there are large 3-phase loads (large motors) to be served and a very small proportion of single-phase loads. The single-phase loads are normally served by an additional single-phase step-down transformer connected between any two of the three output wires. Figure 13-5(d) shows a grounded delta connection. Its use is similar to the delta connection, except that by grounding the center of one phase of the transformer, the single-phase loads can be accommodated without the need for an additional single-phase step-down transformer.

Because of basic design considerations, transformers operated at lower frequencies are generally larger in mass than those operating at higher frequencies. Transformers designed for 50 Hz or 25 Hz circuits will operate successfully on 60 Hz systems; however, if transformers designed for 60 Hz systems are used on 25 Hz or 50 Hz systems, excessive heating and burnout will result unless the transformer is grossly oversized in comparison to the load it is serving.

c. Switching. Switching equipment consists of breakers, buses, protective relays, and other items by which the power is distributed and controlled. The breakers serve the two main functions of switching a load on or off and isolating faults and overloads from the system. In performing the isolation function, most breakers have an internal sensing element; however, on some of the larger breakers, the fault-sensing elements are external to the breaker. These external devices are protective relays. Some of the typical relays found in wideband power plants are overcurrent, groundfault, and reverse power. While their names are reasonably self-explanatory, their importance and the importance of the breakers they serve cannot be overemphasized when it is realized that these devices must be capable of isolating or interrupting faults in the 25,000 Amp to 100,000 Amp range under conditions measuring as low as 2-10 cycles (30 to 150 milliseconds).

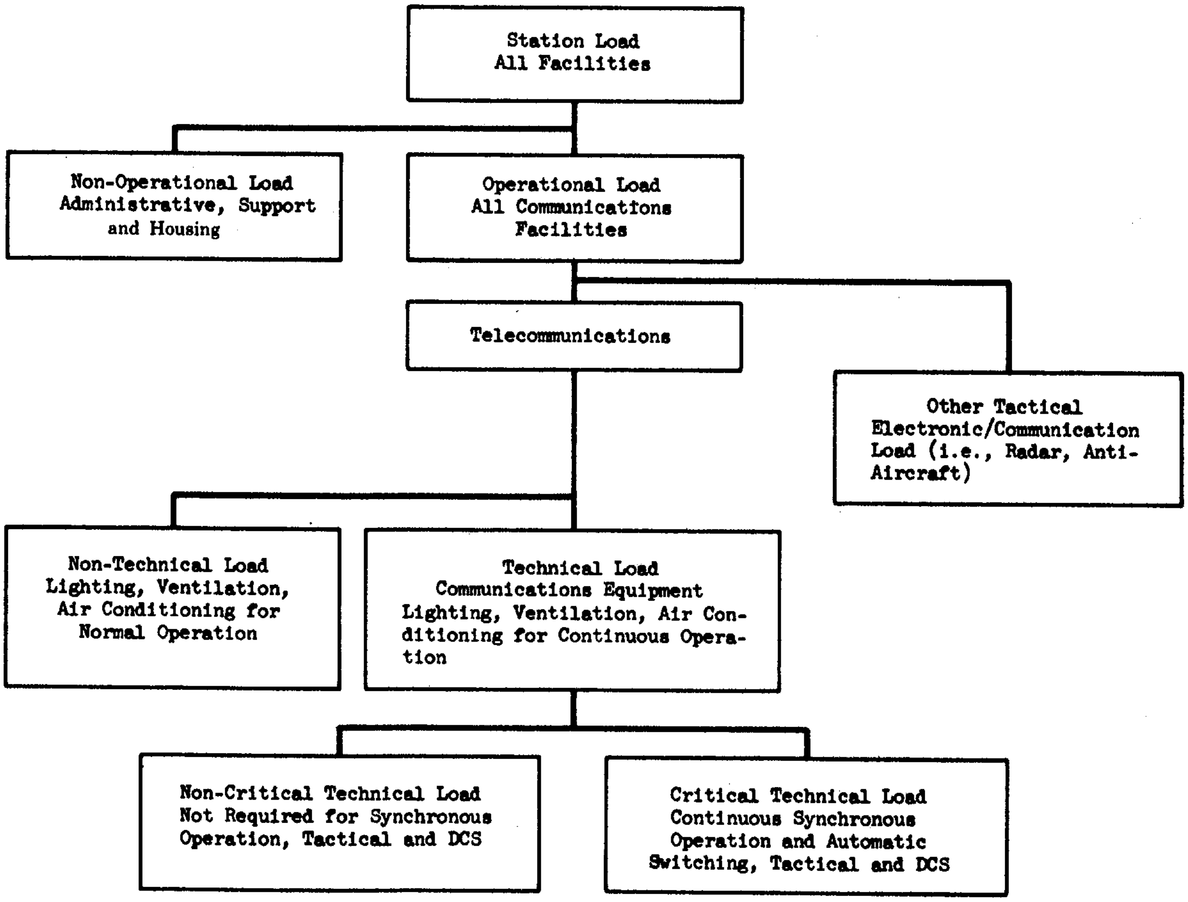


Figure 13-2. Load Categories.

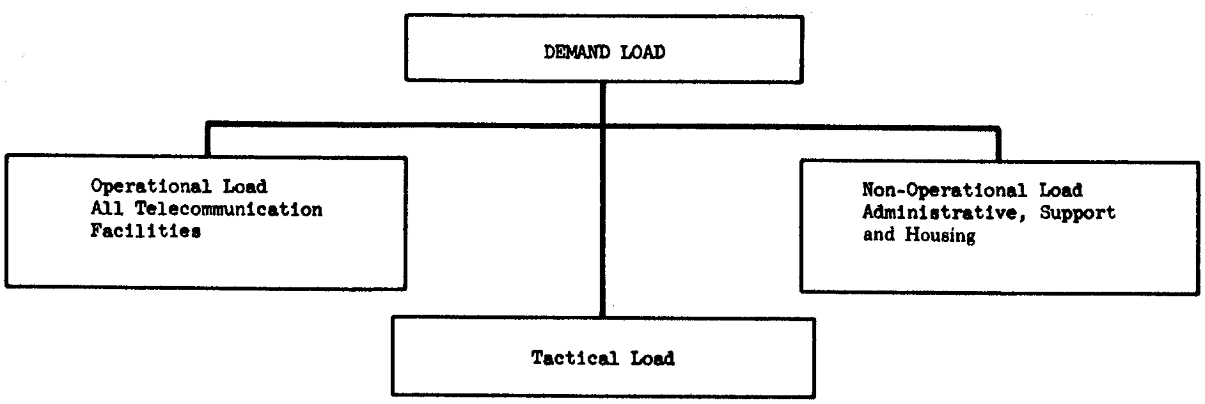


Figure 13-3. Demand Load Categories.

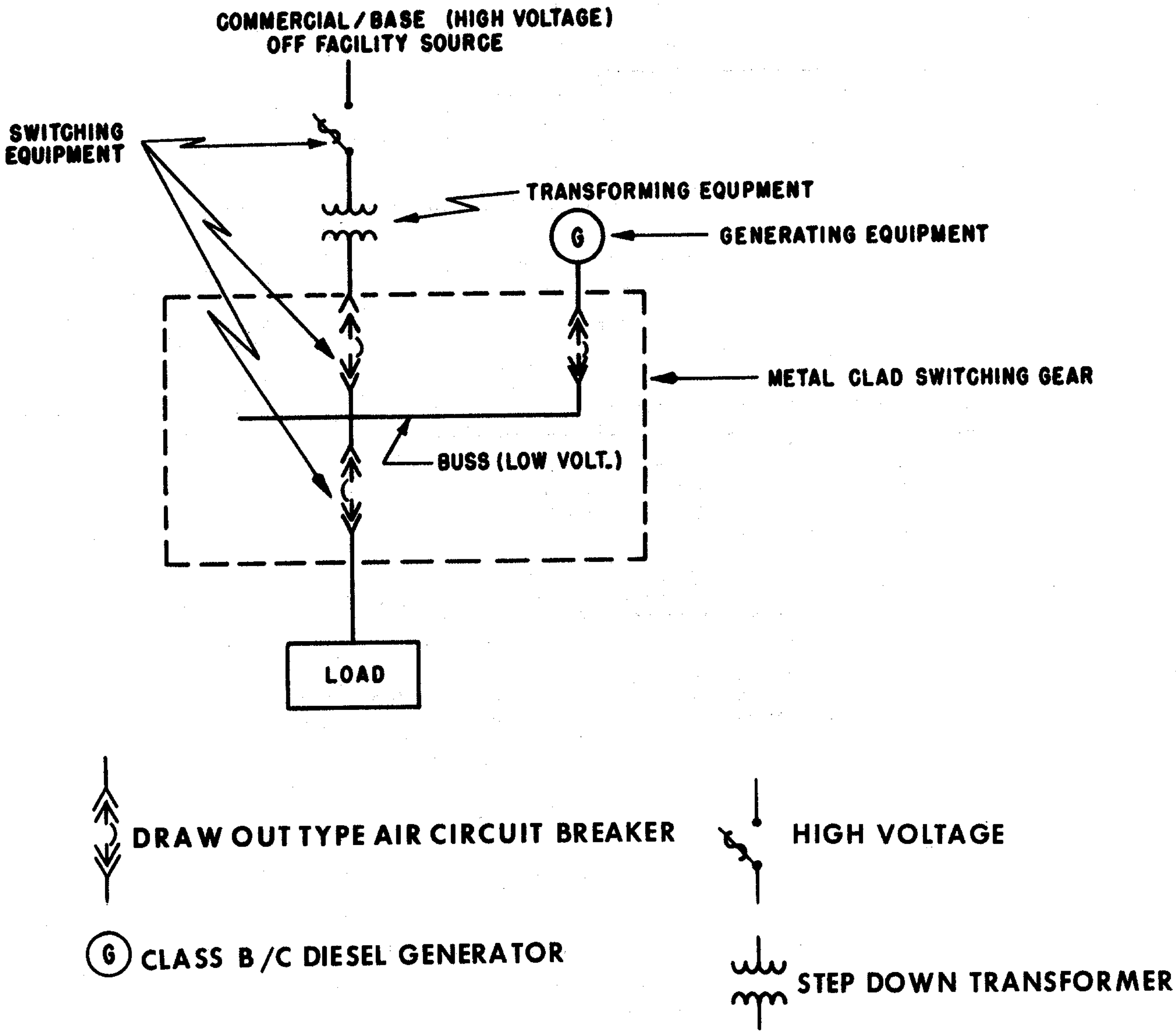


Figure 13-4. Load Configuration.

GENERATING SET SIZE (kW)			
Maximum RPM	*Class A Continuous Operation*	*Class B Long Time Operation (Days)*	*Class C Short Time Operation (Hours)*
1,800	- - - - - - - - -	- - - - - - - - -	Up to 300
1,200	- - - - - - - - -	Up to 300	301 - 600
900	Up to 200	301 - 600	601 Up
720	201 - 500	601 - 1,000	- - - - - - - - -
600	501 - 1,000	1,001 - 1,500	- - - - - - - - -
514	1,000 - 1,500	- - - - - - - - -	- - - - - - - - -
450	1,501 Up	1,501 Up	- - - - - - - - -

Table 13-1. Diesel Engine Characteristics.

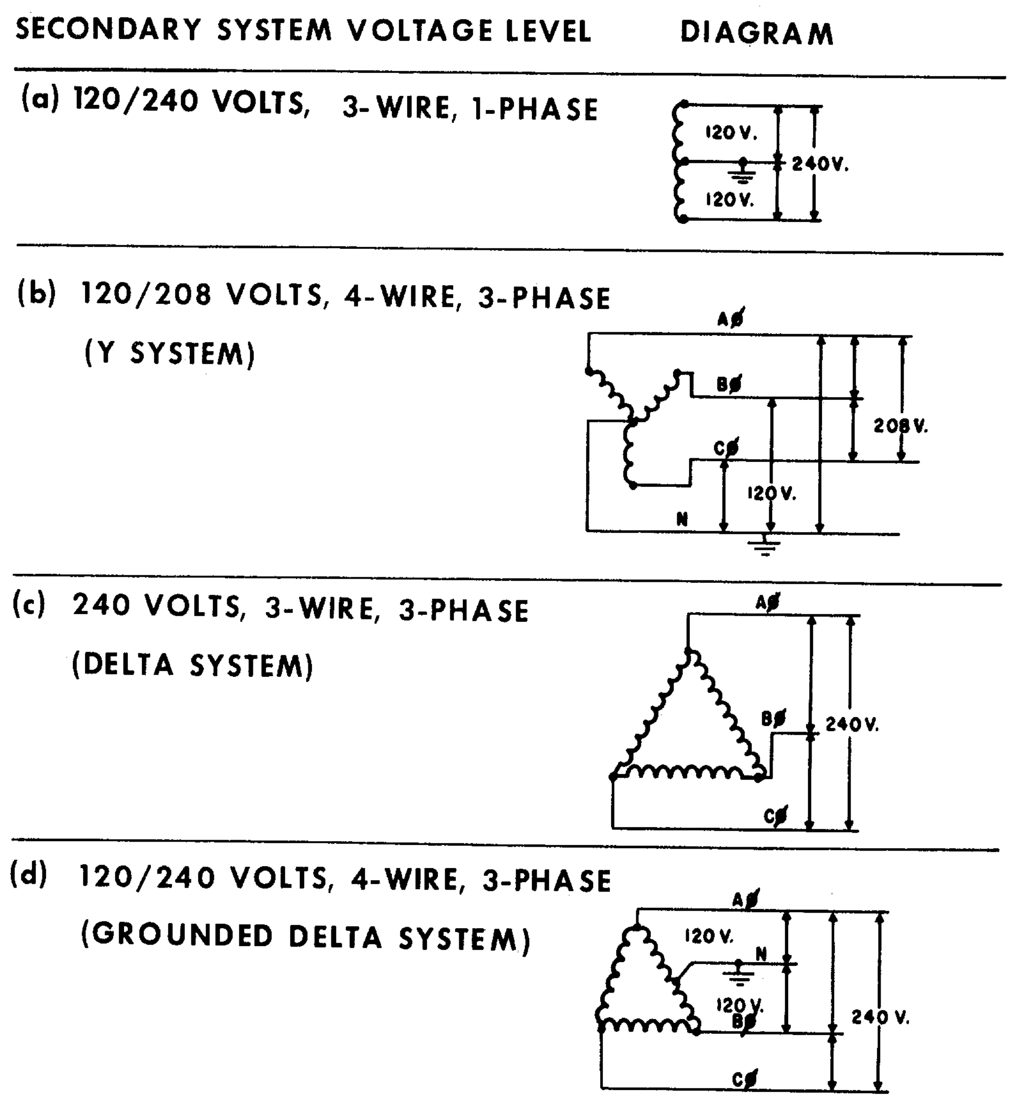


Figure 13-5. Typical Transformer Arrangements.

(e) 480 VOLTS, 3-WIRE, 3-PHASE (SAME AS (c) EXCEPT 480 VOLTS)

(f) 240/416 VOLTS, 4-WIRE, 3-PHASE (SAME AS (b) EXCEPT 240 OR 265 VOLTS TO GROUND)

265/460 VOLTS, 4-WIRE, 3-PHASE

(g) 277/ 480 VOLTS, 4-WIRE, 3-PHASE (SAME AS (b) EXCEPT 277 VOLTS TO GROUND)

Figure 13-5. Typical Transformer Arrangements.

13-5. Power Reliability. Power system reliability has a direct effect on the reliability of a communication system. Simply stated, "if your power system fails, you're off the air." The present criteria for power system reliability, concerning DCS power systems, is that the primary power supply, auxiliary power supply, and distribution system shall provide 99.99% reliability to the techical load bus(es) not to exceed 53 minutes total outage during any one year. This permissible outage time is exclusive of scheduled outages, that is, outages for preventive maintenance inspection or scheduled demand maintenance.

a. Theoretical Considerations. Power reliability is the ratio of the time power of acceptable quality is available to the time it is required. Thus a one minute power outage degrades the power system reliability by one minute or 1/53d of the allowable yearly outage time; however, this one minute power outage may easily cause a 30 minute communications outage. Since the communications system is also required to be 99.99% reliable (53 minutes down time per year) this single one minute power outage has caused a station/system degradation in excess of half (30/53) the allowable yearly system down time. Based on present criteria, if this type outage were to occur 53 times within the year, the power system would have experienced 53 minutes of outages with a reliability of 99.99% while the communications station/system would have experienced 26.5 hours of outages with an unacceptable reliability of 99%.

From the previous discussion, we see that it is possible that an individual power plant, while operating within the prescribed reliability limits (less than 53 minutes outage per year) and by experiencing a number of short duration power outages, can cause a communications station or an entire communications system to experience a reliability factor far below the required 99.99%.

So far we've covered the possible impact that one power system at one site can have on the total communications system. If we now consider the potential impact of, say, four such power plants at four sites, the effect can be disastrous. Assuming the same conditions as in the case of the single plant (99.99% reliability) and that there are no coinciding outages, the total outage time for the communications system would be 1060 hours (4 x 26.5) of outage. This gives a communications system reliability of 98.7%.

In short, the foregoing examples of power systems meet the criteria, but the individual communications stations and the total communications system are in shambles (at least from a reliability standpoint).

It is interesting to note that this discussion, so far, has assumed that the communications equipment has been 100% reliable while all of the "acceptable" power failures were occurring. The obvious fallacy of this assumption only serves to emphasize the almost insurmountable problems encountered when trying to operate a communications system at 99.99% reliability.

Fortunately, power plants are reliable items. It is not uncommon for a Class A prime power plant to exceed 99.99% reliability. For this reason, Uninterruptible Power Systems (UPS) are not used with many Class A plants. One reason Class A plants can operate with such high reliabilities is that they are less subject to such external influences as lightning, starting and stopping of extraneous loads, etc. This is because the plants are usually installed in close proximity to the facility serviced and are designed for the anticipated technical loads, having a relatively steady demand and a high ratio of operational to nonoperational load. Another reason is that, in a Class A plant, it is very easy to put an additional unit on line to cover any potential outage situation. By furnishing more than one source to supply a load, the 99.99% reliability can be met without any one of the sources being this reliable. With two sources, each of the sources need be only 99% reliable.

Since off-station power sources are not this reliable (normally operating in the 95-99% reliability range) the additional required reliability is achieved by using a UPS and a backup power system in conjunction with the off-station source; therefore, based on parallel probability using three sources, 99.99% reliability can be achieved with individual source reliability as low as 95%.

Having identified the potential impact that power system reliability plays in the total station/systems reliability and how the use of multiple sources helps achieve this reliability, let's take a brief look at some of the other means of enhancing or improving the power system reliability.

b. Operational Considerations. Probably the most significant single operational means to enhance the reliability of a power system is through adequate training. Personnel must be trained in all aspects of operating the power plant, such as the development of detailed step-by-step procedures for the performance of each major operation, including the completion of checklists to verify each step.

There are other operational means through which to avoid problems and outages by protecting the communications load from possible momentary power loss. One example is to place sites normally operated on commercial or base power on back-up power when adverse weather is forecast for the vicinity. Another example is to place the critical technical load on back-up power whenever the UPS (no break) is out of service for any reason. Both of these operations will avoid the possibility of commercial power fluctuations and momentary outages causing site or system degradation and possible communications equipment damage. Significant improvement in system reliability can be achieved by careful selection of the method of power plant operation, maintenance, and configuration which offers the best protection to the communications load.

c. Maintenance Considerations. One of the most rewarding areas for improving systems reliability is in the area of power system maintenance. The greatest single cause of power outages is equipment failure. The chief cause of equipment failure is poor maintenance or no maintenance. This is caused by a reluctance to furnish needed communication outage time for maintenance of equipment within the power system. As a result, the maintenance time on most power systems supporting communications systems is reduced to symptomatic maintenance. Under this concept, maintenance is performed when and if the equipment fails. While this method of performing electrical system maintenance may be acceptable for some type facilities (barracks facilities, administrative facilities, etc.), it is not acceptable for long-haul communications facilities. In order to furnish 99.99% reliable power, scheduled preventive maintenance is required and can reduce equipment failures and unexplained outages by 95%.

13-6. Classes of Power. Power sources may be classified in two categories: primary power and auxiliary (back-up) (figure 13-6).

a. Primary. This type of power is used as a reliable source to serve the station main bus. It may originate from a government-owned generating plant or a commercial off-base source.

A primary power plant is one in which prime power is generated locally at the communications facility. The source of power is normally a Class A diesel engine (table 13-1) driving 60 Hz alternators. The alternators usually furnish power at utilization voltages (227/480 or 120/208) and are capable of providing a minimum capacity of 125% of the station demand load. The individual units are sized so that they normally operate in excess of 50% of the rated capacity under any station demand load. A minimum of two spare generator units is recommended, one unit being available for replacement of a failed unit and the other for scheduled maintenance.

Off-base power is furnished to the facility by a commercial source. It is normally furnished to the site at 2.4/4.16 kV (3-phase) or at a higher voltage and is reduced to utilization voltage (120/208) at the site through step-down transformers. Where the off-site power furnished is 50 Hz, a 50 Hz motor driving a 60 Hz generator (converter-MG) is normally used to furnish 60 Hz power to those pieces of comunications equipment which can operate only from a 60 Hz source.

The selection of the type of prime power to be used, either Class A or commercial, depends on the availability and reliability of the commercial power. If the commercial power is not available or is not sufficiently reliable and cannot be made sufficiently reliable with a Class B back-up plant, then use of a Class A is justified.

b. Auxiliary. The function of the auxiliary or back-up power system is to provide for continued operation during periods of failure or degradation of the primary power. There are three types of auxiliary power systems: Class B, Class C, and Class D.

A Class B auxiliary power system consists of a Class B diesel engine (table 13-1) driving a 60 Hz alternator which, in turn, furnishes power at a usable voltage. It is capable of operating for extended periods of time (days). A Class B back-up power system should be placed in service to assume the load during periods of anticipated degradation or outage of primary power (severe weather). Normally, the back-up unit should have the capacity to provide 125% of the operational load. A standby power plant should also be used to provide a minimum of 200% back-up to the operational load. This normally consists of two units: one to carry the operational load and one to be used during scheduled maintenance. The standby power plant must have the capability of synchronizing with the primary power source to permit assumption of the technical load during takeover without interruption of power to the operational load. It must also permit the restoral from back-up power to prime power without interrupting the operational load.

A Class C auxiliary power system consists of a Class C diesel engine (table 13-1) driving a 60 Hz alternator which furnishes power at a usable voltage. It is designed for operating for short periods of time (hours) and, as such, is not an acceptable source of back-up power where extended periods of prime power outage can be expected.

UPS provides continuous power and prevents the occurrence of off-base prime power transients and surges from affecting the technical bus. Its use is restricted only to critical technical loads. While several

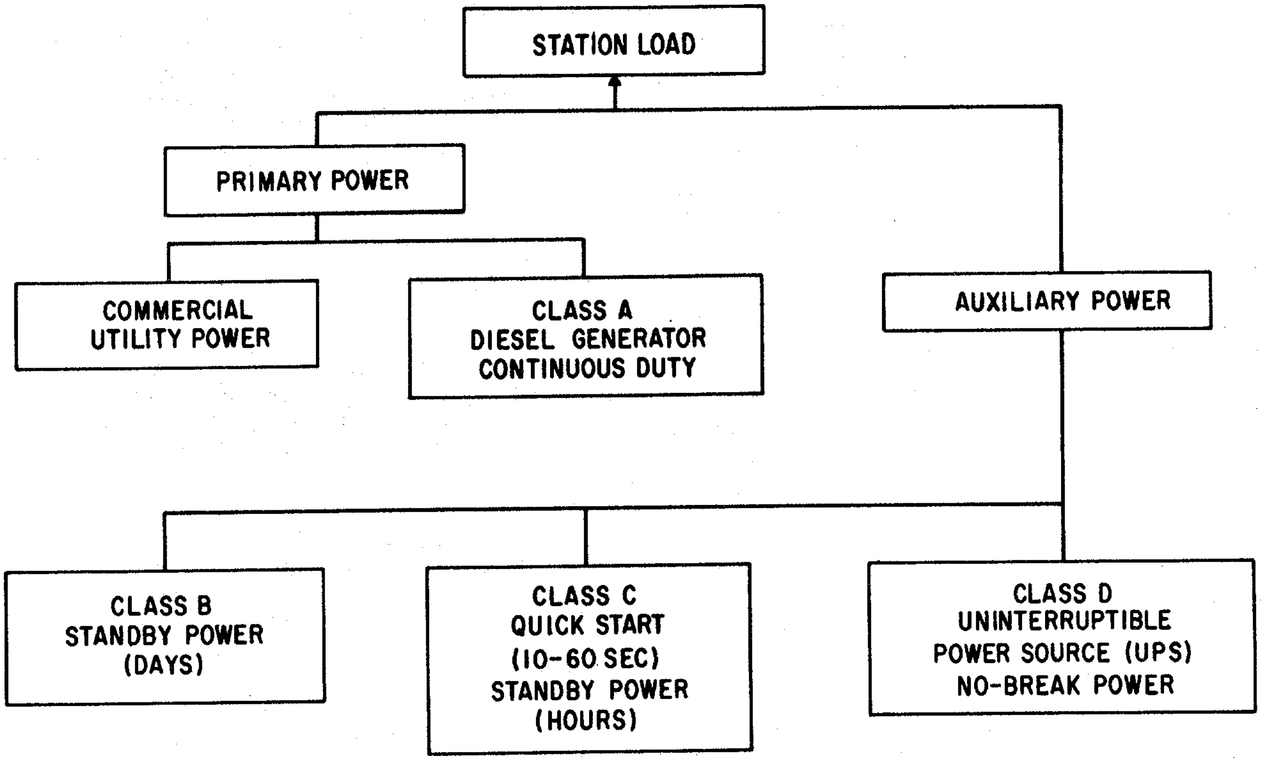


Figure 13-6. Classification of Power Sources.

types of units are available, they fall generally into two classes: rotating mass stored energy system using flywheels for energy storage and rectifier-battery-inverter system in which the energy is stored in fully-charged batteries.

(1) The rotating mass system (figure 13-7) consists basically of an AC motor, AC generator, and flywheel on a common rotating shaft coupled to a diesel engine through an automatic clutch. The AC generator continuously supplies the critical load which requires uninterrupted power. During normal operation, the primary source supplies the AC motor which, in turn, drives the generator and the flywheel. Quality sensing devices within the control system continually monitor the frequency and voltage of the incoming primary power to the motor. In the event the incoming primary power deviates from certain preset voltage and/or frequency tolerance limits, the primary power source is automatically disconnected and the emergency prime mover (diesel engine) is started and brought up to system speed and coupled to the system through the clutch. The kinetic energy stored in the rotating flywheel drives the system during the transition from primary power/AC motor drive to diesel engine drive. During the emergency mode of operation, the control system continually monitors the status and quality of the primary power source. When primary power has been restored and stabilized for a predetermined period, the clutch disengages and the diesel engine is stopped.

(2) The rectifier-battery-inverter system (figure 13-8) consists of a static inverter to supply the AC for the load, a rectifier to provide the DC input to the inverter and to charge the storage battery and a bank of storage batteries to provide the DC input to the inverter on the failure of the primary AC power. The rectifier, inverter, and battery bank are connected at all times, regardless of mode of operation. In the normal mode of operation, the primary AC power source provides the AC power input to the rectifier. The DC output of the rectifier supplies the input to the inverter and, at the same time, supplies current to maintain the battery bank at the float charge. The input AC power requirements are extremely liberal. The system will operate very satisfactorily at varying input AC power frequencies and from AC power of such poor quality as to be unusable for directly supplying power to a given technical load.

During the emergency mode of operation, when the AC primary power source fails, the battery assumes the full load and supplies the DC power for the input to the inverter. Transfer from normal to emergency operation is instantaneous and little effect can be detected in the AC output of the inverter to the critical load.

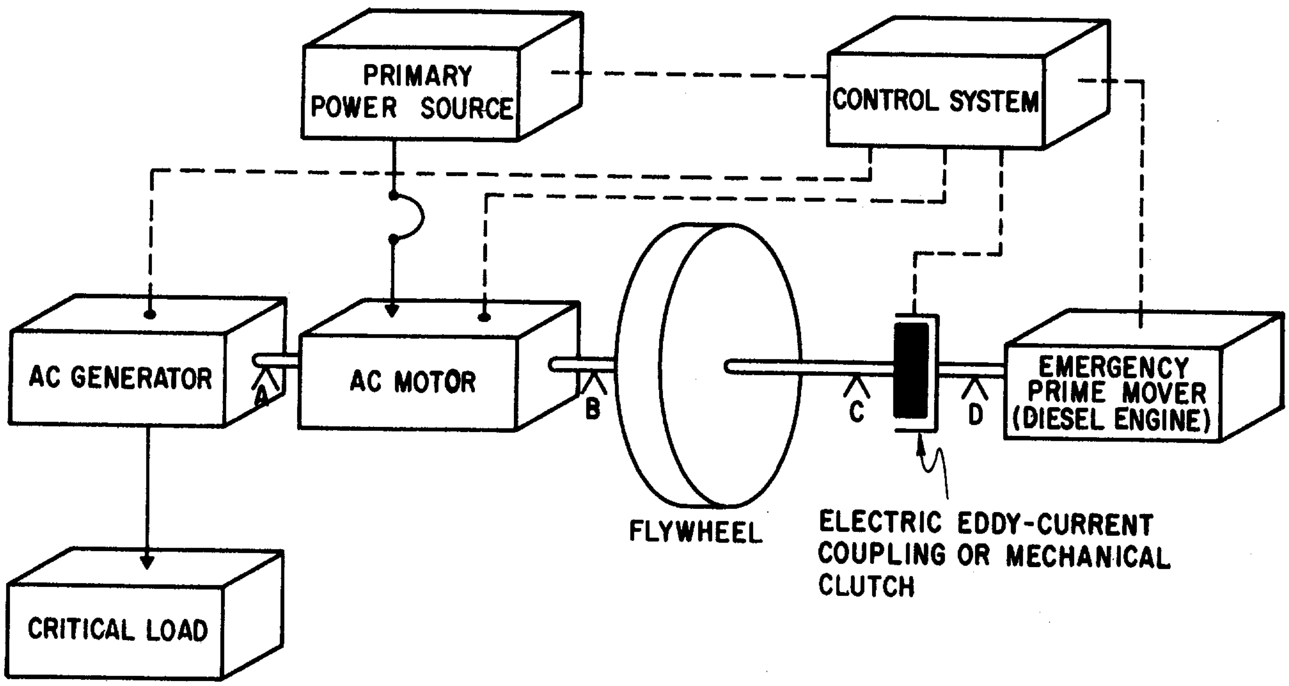


Figure 13-7. Standard Flywheel Motor - Generator UPS Unit - Diesel Driver.

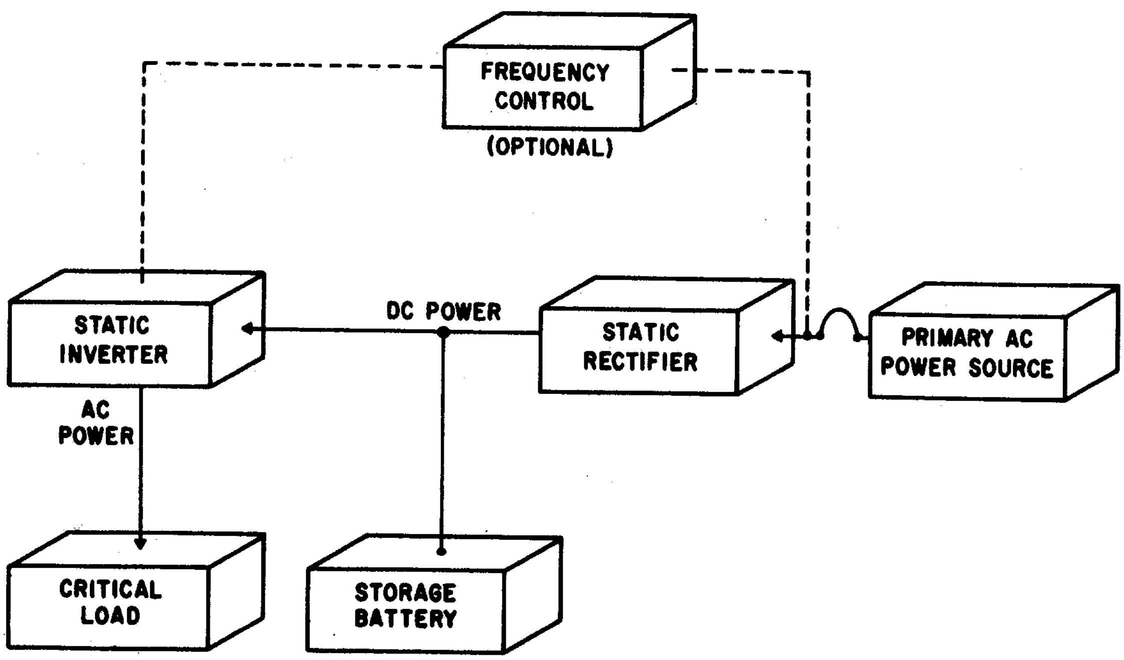


Figure 13-8. Floating Battery, Static UPS Unit.

The system is intended to operate from the battery storage bank during the emergency mode until such time as the primary AC power source is reinstated or until a suitable standby power source can be made available, at which time retransfer to normal operation is automatically effected. This assumes that return to normal operation occurs before the batteries are discharged to a point where they no longer can supply the load. The design time for battery operation is usually 15 minutes, but may be several hours for small power systems. After normal operation is established, the batteries are recharged by the rectifier and maintained on float until the next primary power failure or disruption.

The design of the system is such that it will automatically float over any transient or minor disruption in the primary AC power, due to the inherent smoothing action of the static rectifier and the continuous connection of the battery bank to the input of the static inverter. One of the most significant advantages of this type system over the whole family of rotating mass systems is that it completely eliminates all the maintenance problems associated with rotating units (vibration, bearing, etc.).

13-7. Standard Power Configurations:

a. General. The configuration of a power plant can have a great impact on its potential reliability. For example, some plants, while having the required number of sources to theoretically furnish 99.99% reliability as previously discussed, are connected in such a manner as to make maintenance on certain portions of the power system impossible without a communications system outage. Since even a scheduled outage is charged against system reliability, we have a situation where the system reliability is in jeopardy because the power plant configuration does not permit maintenance without a communications system outage. There are certain basic power plant configurations which permit the required maintenance and also allow for additional operating flexibility (figures 13-9 thru 13-12). In the interest of both power plant and communications system reliability, power plants which do not have this flexibility should be reconfigured to conform to one of the standard AFCS-approved power plant configurations discussed below.

b. Standby Plant - No Critical Technical Load (Configuration # 1):

(1) Normal Operation. The normal mode of operation for this plant configuration (figure 13-9) would be to furnish commercial/base power to the two buses through separate transformer banks. The utility and air conditioning loads would normally be served from one bus and the technical load from the other bus with the tie breaker open. While each load can be served from either bus for operational flexibility and maintenance purposes, it will normally be served from only one side (that is, one of the two available breakers to each load will be open). For example, the breaker which connects the utility load to the utility bus would be considered the "main" breaker for this load and would normally be closed. The breaker which permits this load to be connected to the other (technical) bus would be considered the "alternate" breaker and would normally be open. This arrangement would be similar for the other loads. Loads can be transferred from one bus to another by closing the tie breaker, closing the alternate breaker(s), opening the main breaker(s), and then opening the tie breaker. Thus, either bus can be the technical bus, depending on which combination of breakers are closed. In short, whichever bus is serving the technical load is the technical bus.

(2) Emergency Operation. In the event of a commercial or base power failure, the back-up generators would be placed on line to carry the load with the commercial breakers open. When the commercial or base source has been restored, the generators would be synchronized with the base or commercial source, the load would be transferred to the base or commercial source, and the generators would be shut down. In this way, the transfer of the load is accomplished without a communications interruption. Some advantages of this system are:

(a) The technical load can be and is isolated from the voltage drop (power fluctuation) caused by the starting of air conditioning and other motor load.

(b) Any load can be fed from either of the two available buses. All transformers, buses, and generators are sized to carry the entire station load; thus, the loss of one bus, transformer, or generator due to an electrical fault, sabotage, or any other reason will not cause an extended communications outage, since all of the station loads can be placed on the remaining bus, transformer, or generator.

(c) This flexibility allows any item of the distribution system (generator, transformer, bus, or breaker) to be taken out of service for maintenance without causing a communications outage.

(d) The two buses are normally isolated from each other, one (Bus A) carrying the technical load and the other (Bus B) carrying the utility and air conditioning load. Whenever it is necessary to test an engine generator under load (after minor maintenance or for weekly testing), the engine generator can be used to pick up the utility and air conditioning load as the required test load, while still furnishing normal primary power to the technical load. Since the utility and air conditioning loads are normally unaffected by power transients, short duration outages on these systems will not normally cause a communications outage. Consequently, this test load requirement can be met without installing load banks.

c. Standby Plant with Critical Technical Load Configuration # 2):

(1) Normal Operation. The normal mode of operation for this configuration (figure 13-10) is the same as for Configuration # 1. The utility and air conditioning loads would normally be served from one bus and the technical load from the other bus with the tie breaker open. The critical technical load would be served from the technical bus through the UPS. While each load can be served from either bus, for operational flexibility and maintenance purposes, it would normally be served from only one side (that is, one of the two available breakers to each load will be open). Breaker operation and transfer of loads from one bus to another is the same as that described for Configuration # 1.

(2) Emergency Operation. In the event of a commercial or base power failure, the back-up generators would be placed on line to carry the load

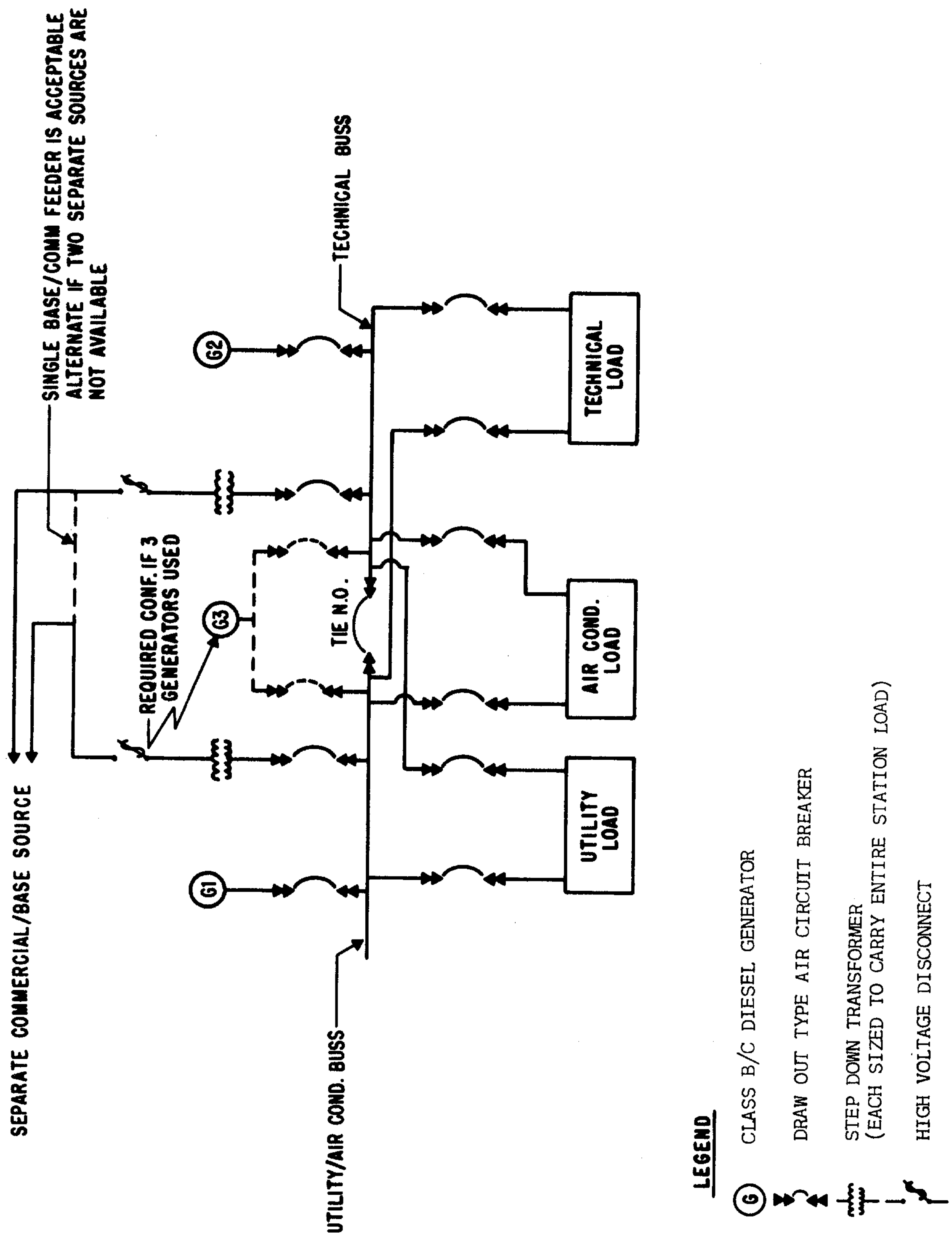


Figure 13-9. Configuration # 1 - Standby Plant with No Critical Technical Load.

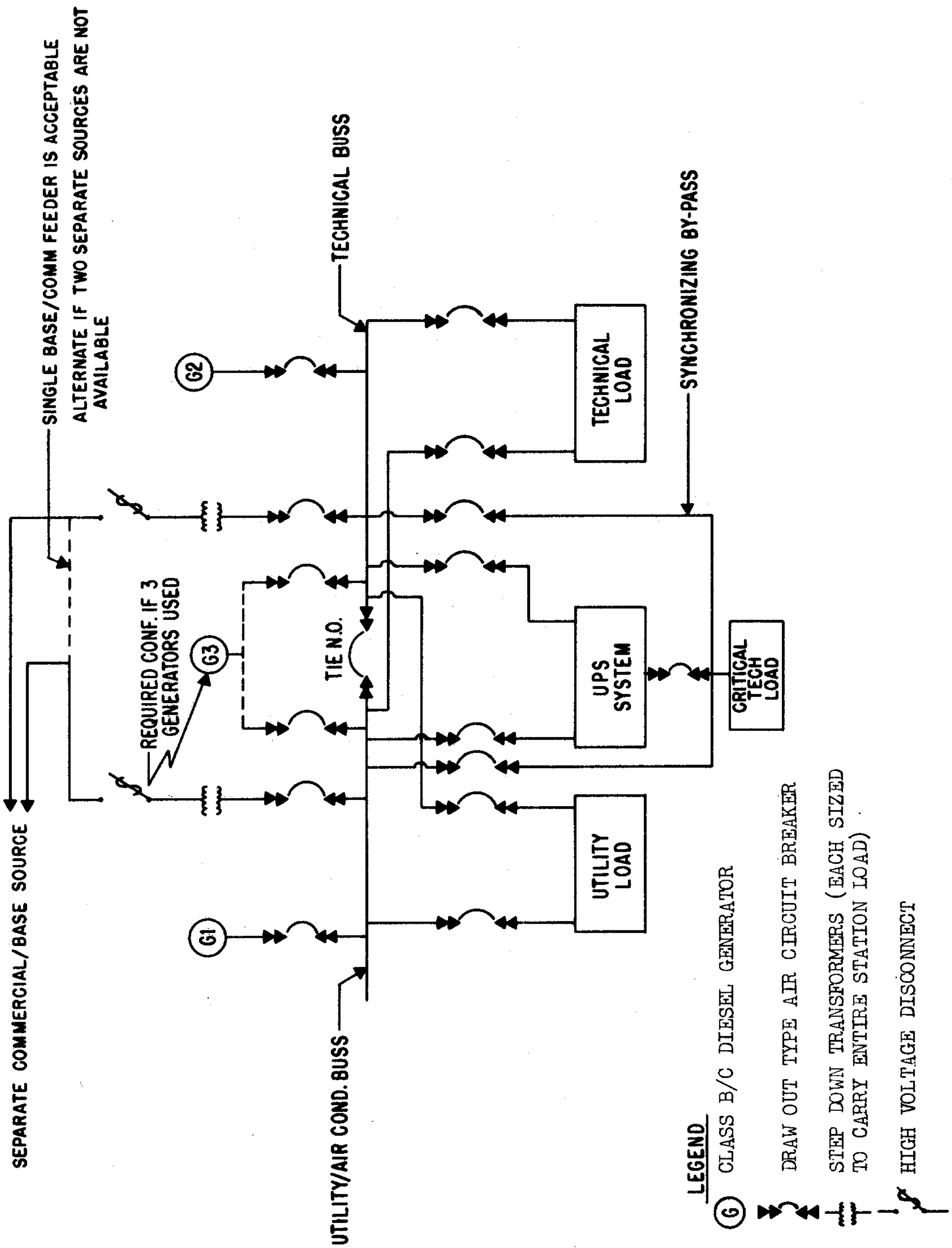


Figure 13-10. Configuration # 2 - Standby Plant with Critical Technical Load.

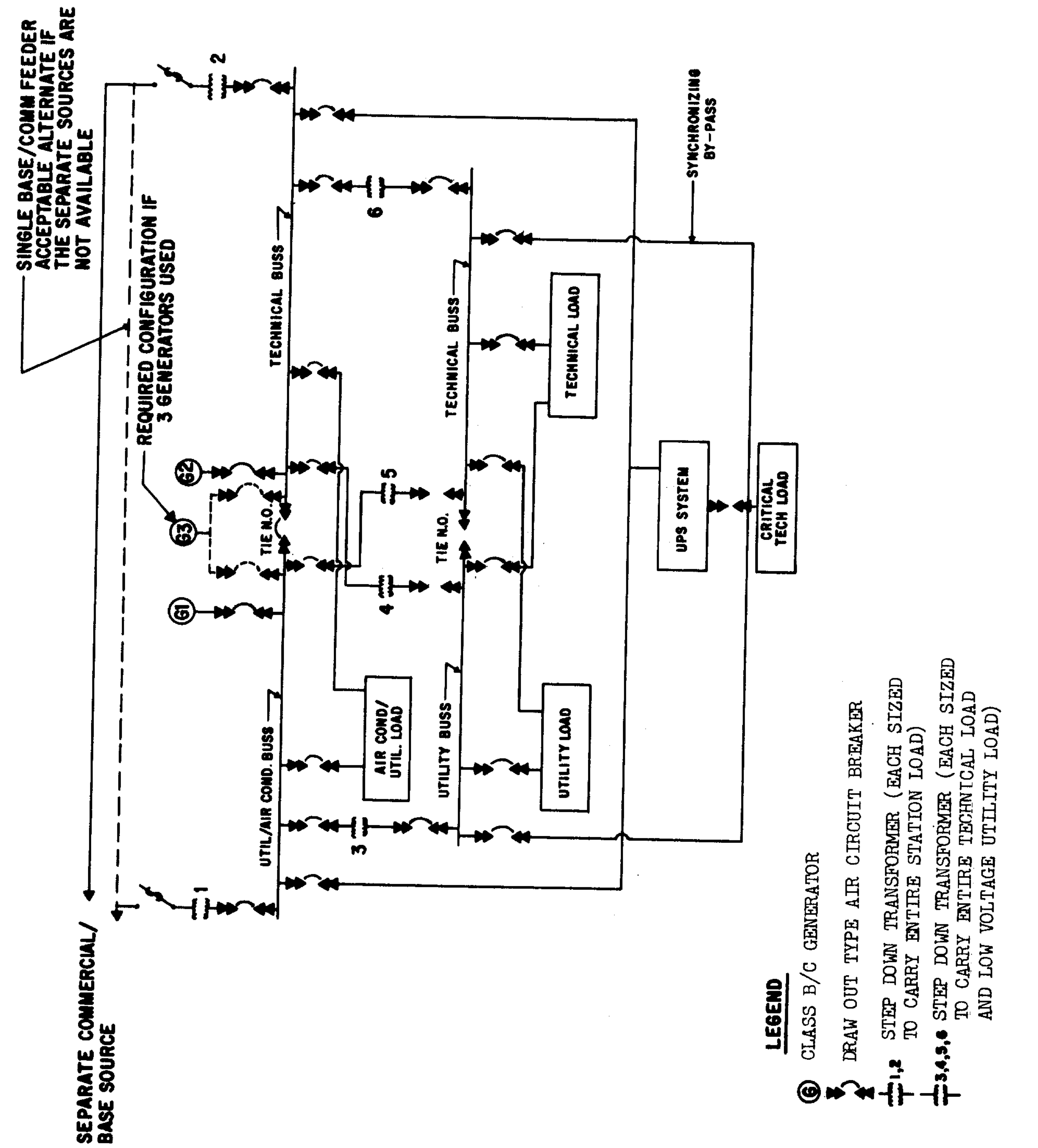


Figure 13-11. Configuration # 3 - Standby Plant with Critical Technical Load and Large Air Conditioning Utility Load.

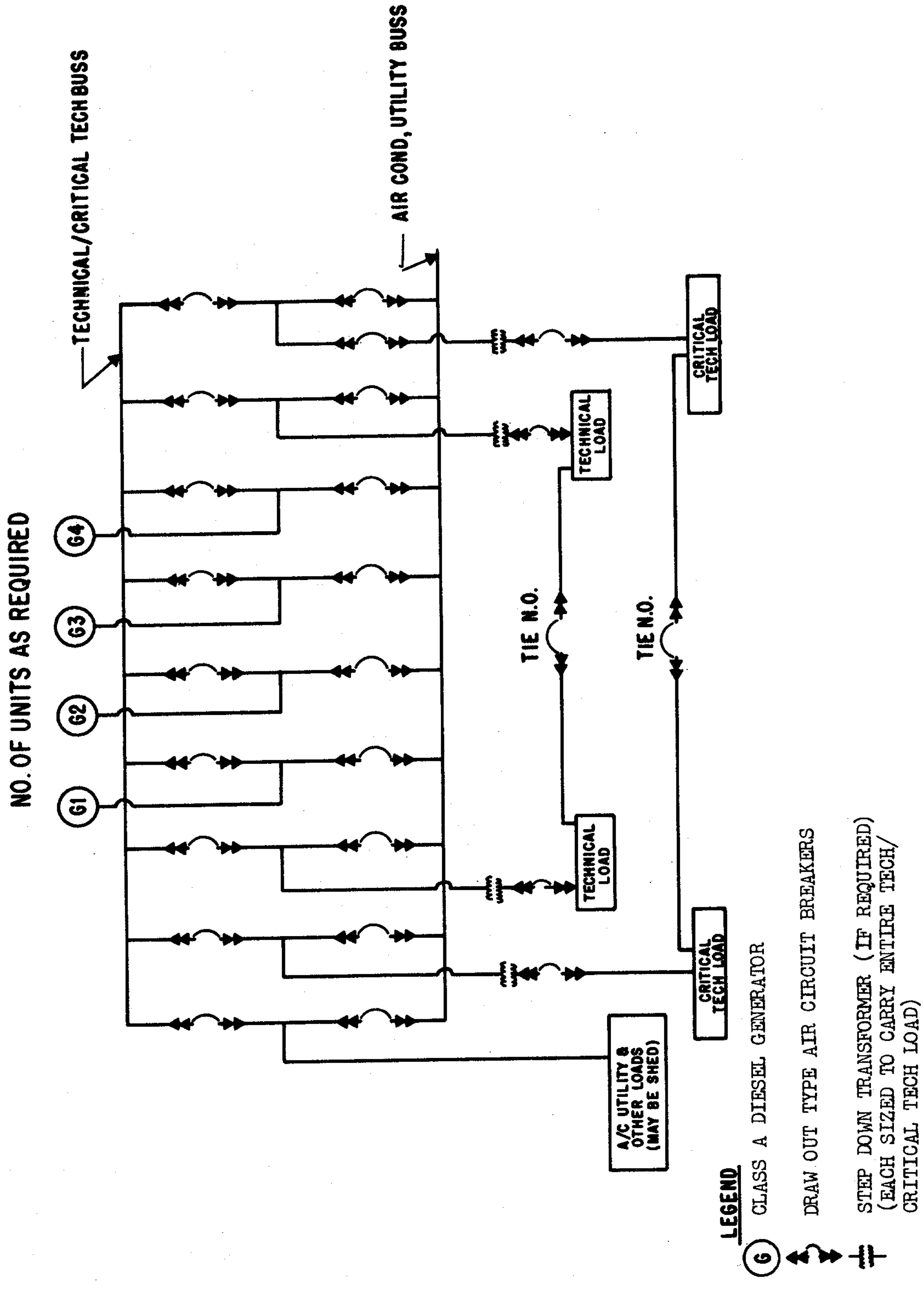


Figure 13-12. Configuration # 4 - Class "A" Plant.

with the commercial breakers open. Until the back-up generator is placed on line, the critical technical load is protected by the stored energy of the UPS. When the commercial or base source has been restored, the generators are synchronized with the base or commercial source, the load is transferred to the base or commercial source, and the generators are shut down. This transfer of the load is accomplished without a communications interruption. Some advantages of this system are:

(a) The technical load and critical technical load can be, and are, isolated from the voltage drop (power fluctuation) caused by the starting of air conditioning and other motor loads.

(b) Any load can be fed from either of the two available buses. All transformers, buses, and generators are sized to carry the entire station load; thus, the loss of one bus due to electrical fault, sabotage, or any other reason will not cause an extended communications outage since all the station loads can be placed on the remaining bus.

(c) This flexibility allows any item of the distribution system (generator, transformer, bus, or breaker (including the entire UPS)) to be taken out of service for maintenance without causing a communications system outage.

NOTE: During the time the UPS is down for maintenance (with the critical technical load being supplied power directly from the technical bus), one of the back-up generators would be used to furnish power to the technical bus to protect the critical technical load from any adverse effects of possible transients eminating from the off-station power source (that is, caused by lightning, high winds, etc.).

(d) The two buses are normally isolated from each other, one (Bus A) carrying the technical load and the other (Bus B) carrying the utility and air conditioning load. Whenever it is necessary to test an engine generator under load (after minor maintenance or for weekly testing), the engine generator can be used to pick up the utility and air conditioning load as the required test load, while it is still furnishing normal primary power to the technical load. Since the utility and air conditioning loads are normally unaffected by power transients, short duration outages on these systems will not normally cause a communications outage. Consequently, this test load requirement can be met without installing load banks.

d. Standby Plant with Critical Technical Load and Large Air Conditioning/Utility Load (Configuration # 3).

(1) Normal Operation. The normal mode of operation for this configuration (figure 13-11) is the same as described for Configuration #1 except that two additional medium voltage (277/480) buses are used to accommodate large air conditioning and utility loads and the critical technical load is served from one of these buses through the UPS. While each load can be served from either bus, for operational flexibility and maintenance purposes, it will normally be served from only one side (that is, one of the two available breakers to each load will be open).

With the exception of the foregoing paragraph, the normal operation of Configuration # 3 is the same as that for Configuration # 2.

(2) Emergency Operation. In the event of a commercial or base power failure, the emergency operation would be the same as described for Configuration #2.

e. Class A Plant (Configuration #4) (figure 13-12):

(1) Normal Operation. The normal mode of operation would be to furnish Class A power to the two buses. The utility and air conditioning loads would normally be served from one bus and the technical and critical technical loads from the other bus. Each bus would be served individually by a separate generator(s). The number of generators connected to each individual bus and the total required for the station is determined on an individual basis depending on size of the technical load, size of the station load, size of generators, etc. While each load can be served from either bus, for operational flexibility and maintenance purposes, it will normally be served from only one side (that is, one of the two available breakers to each load will be open). (Breaker operation is the same as that described for Configuration # 1.) Loads can be transferred from one bus to another by synchronizing the two buses, closing the generator(s) alternate breaker(s), closing the alternate breaker(s) for the loads, opening the main breaker(s) for the loads, and then opening the generator(s) main breakers. Thus, either bus can be the technical bus depending on which combination of breakers is closed. Again, whichever bus is serving the technical load is the technical bus.

(2) Emergency Operation. In the event of a failure of one of the on-line generators, a standby Class A generator would be placed on line to carry the load. Some of the advantages of this system are:

(a) The technical and critical technical loads can be, and are, isolated from the voltage drop (power fluctuation) caused by the starting of air conditioning and other motor loads.

(b) Any load can be fed from either of the two available buses. All transformers and buses are sized to carry the entire station load and spare generators are provided; thus, the loss of one bus, transformer, or generator, due to electrical fault, sabotage, or any other reason, will not cause an extended communications outage since all of the station load can be placed on the remaining bus, transformer, or generator(s).

(c) This flexibility allow any item of the distribution system (generator, transformer, bus, or breaker) to be taken out of service for maintenance without causing a communications outage.

(d) The two buses are normally isolated from each other, one (Bus A) carrying the technical and critical technical load and the other (Bus B) carrying the utility and air conditioning load. Whenever it is necessary to test an engine generator under load (after minor maintenance or for weekly testing) the engine generator can be used to pick up the utility and air conditioning load as the required test load while it is furnishing normal primary power to the technical and critical technical loads. Since the utility and air conditioning loads are normally unaffected by power transients, short duration outages on these systems will not normally cause a communications outage; hence, the test load requirement can be met without installing load banks.

(e) Since this type of power plant is located at the communications facility and serves only the communications facility, the communications equipment is protected from any of the adverse power surges experienced when using commercial or off-base power sources.

Chapter 14
DATA SYSTEMS

14-1. General. Communications can be defined as the transfer of data from one point to another. Whether we use semaphore flags, smoke signals, computers, or a string with two tin cans, we have fulfilled the basic requirement for communications, that is, the transmission of data from one point to another. There are basic elements common to all data systems, regardless of the complexity of the system. In this chapter, we will examine these elements to determine how they affect the efficiency of data communications system and their relation to the wideband communications media.

14-2. Basic Elements. "Data," or more specifically, "digital data" as used in this chapter refers to any form of communications which has been processed into "mark" and "space" coded information. In today's digital data systems, the letters, numbers, and other symbols used in the transfer of information are represented by the sequential transmission of a specific number of electrical pulses. Each of the pulses is binary in nature; that is, each pulse can exist in either of two states. The two states can be referred to as on/off, 0/1, mark/space, positive/negative, etc., and as the significant conditions for binary modulation. These significant conditions, as defined by DCA, are shown in table 14-1.

Each individual pulse is referred to as a binary digit, from which the term "bit" is derived. A code, therefore, consists of the sequential transmission of a specific number of bits. The Baudot Code, which is a code in common use today, consists of five intelligence bits. Since each bit can exist in either of two states, there are 32 (2^5) different five-bit sequences which can be used to represent letters, functions, numbers, etc.

Other codes use a six-bit sequence, permitting 64 (2^6) different bit sequences; others use a seven-bit sequence, permitting 128 (2^7) different bit sequences. A series of 5, 6, 7, etc., bits, depending on the code used, forms a data character. The data character includes any start, stop, or parity bits needed to effect proper equipment operation.

For example, assume that it is desired to transfer 390 characters-per-minute over a data system and that each character interval consists of a "start" bit, five "intelligence" bits, and one "stop" bit (a seven-unit character-interval). It follows that 390 x 7, or 2730 bits, must be passed through the data system in one minute. The duration of each bit is, therefore, 22 milliseconds in length (60 seconds ÷ 2730). There are two methods of specifying modulation rate: in bits-per-second (bps) and in baud. A baud is the signaling rate of the modulated carrier in the channel. The bit rate is the reciprocal of the bit length in seconds.

MARK	SPACE
1	0
+	–
CURRENT-ON	CURRENT – OFF
CKT ACTIVE	CKT PASSIVE
TONE ON	TONE OFF
FREQUENCY LOW	FREQUENCY HIGH
HOLE (PAPER TAPE)	NO HOLE (PAPER TAPE)

Table 14-1. Significant Conditions for Digital Data Systems.

In data systems where all bits are of equal length or where the character interval consists of an integral multiple of bits, the modulation rate, whether specified in bps or in baud, will always be numerically identical. This will be true in all synchronous systems and will be true for all asynchronous systems except those where the duration of the "stop" bit is not an integral multiple of the "intelligence" bits. An example of the latter is the 7.42-unit character-interval consisting of a unit "start" bit, five-unit "intelligence" bits, and 1.42-unit "stop" bit. In the latter case, it is not entirely satisfactory to specify modulation rate in either bps or baud. It is customary and preferable, however, to specify the "maximum" modulation rate in baud for the "shortest" or "unit" pulse duration.

It should be apparent from the foregoing examples that if we keep the modulation rate constant at 45.5 baud in the transmission of the 7.0 unit character-interval and the 7.42 unit character-interval, the data thruput rate will be higher for the lower character-interval; that is, the data thruput rate for the 7.42 unit character-interval is 368 characters per minute, while that for the 7.0 unit character-interval is 390 characters per minute. Table 14-2 summarizes this information.

The basic elements of a data communications system are shown in figure 14-1. Note that in the transfer of data from one point to another, various elements are interrelated, as shown in this figure.

a. Data Sources. In digital systems, the source of data may be a teletypewriter, a telegraph key, a computer, a card reader, or any one of many data source machines which convert electrical or mechanical sensing actions into a series of electrical pulses whose characteristics vary in discrete steps. In analog systems, the source may be a telephone, a tape recorder, a television camera, or other device which converts sensing actions into an electrical signal. The characteristics of this signal vary directly as a function of the signal impressed on the sensing transducer.

b. Transmit Data Interface. Transmit data interfacing equipment includes a broad variety of equipment used for the adaptation of the data source signal to effect compatibility with other interfacing equipment, the transmission media, or the data sink.

c. Transmission Media. This may include transmitters, antennas, the radio propagation medium, radio receivers, land cables, submarine cables, or the air itself. Primarily, it is in this area that the effects of noise and distortion limit the efficient transmission of data.

d. Receive Data Interface. Except for the noise and distortion which has been added to the data, the receive interface equipment merely accomplishes the inverse function of the transmit interface, but not always. Using a PA system in a stadium as an analogy, the spectator is capable of selecting the information he wants to hear (data) while rejecting the poorer quality data and decoding the language. These functions may be considered as the equivalent of the receive data interface block.

e. Data Sink. In digital systems, the data sink will generally perform the inverse function of the data source (that is, a teletypewriter, a computer, or a card punch). But it is sufficient that the data sink serve only as the data receptor or the terminus for the information transmitted.

CHARACTER INTERVAL (NUMBER OF UNIT INTERVALS PER CHARACTER)	CODE LEVEL (NUMBER OF INTELLIGENCE BITS PER CHARACTER)	UNIT INTERVALS IN STOP PULSE	DATA THRUPUT PLATE		PULSE LENGTH (IN MILLISECONDS)		MAXIMUM MODULATION RATE (IN BAUD)	PARITY BIT PRESENT
			CHARACTERS PER MINUTE	WORDS * PER MINUTE	START AND INTELLIGENCE BITS	STOP PULSE		
7.0	5	1	390	65	22	22	45.45	NO
7.0	5	1	636	106	13.47	13.47	74.2	NO
7.42	5	1.42	368	61.33	22	31	45.45	NO
7.42	5	1.42	600	100	13.47	19.18	74.2	NO
10.0	7	1	600	100	10	10	100	YES
11.0	7	2	600	100	9.09	18.18	110	YES
11.0	7	2	818	136.3	6.66	13.32	150	YES

* IN TELEGRAPHY, SIX CHARACTERS ARE CALLED A "WORD".

Table 14-2. Digital Data Systems Summary Characteristics.

14-3. Data Sources and Sinks. These constitute the terminals at which data is originated (source) and terminated (sink). They may be broadly categorized as serving voice, graphics, and printed matter. In general, voice communications are used in those cases where the urgency of the situation precludes the use of the other two categories. Graphics (facsimile and television) are used when voice or narrative communications are not expressive of the data required to be transmitted. Printed matter is the form in which the large majority of data is transmitted by electrical means. The various types of sources (transmitters) and sinks (receivers) used in data systems are discussed below.

a. Teletypewriters. The equipment most commonly used in the transfer of narrative information in digital data systems is the teletypewriter. It provides for the transmission (and reception) of printed page copy which can be read, analyzed, stored, and is immediately available for recall, as necessary. It also provides for the manual entry of information into the data system through the use of a standard typewriter-type keyboard or for the machine entry of data using a tape reader. It provides for the reception of data in printed page copy or in the form of a punched tape.

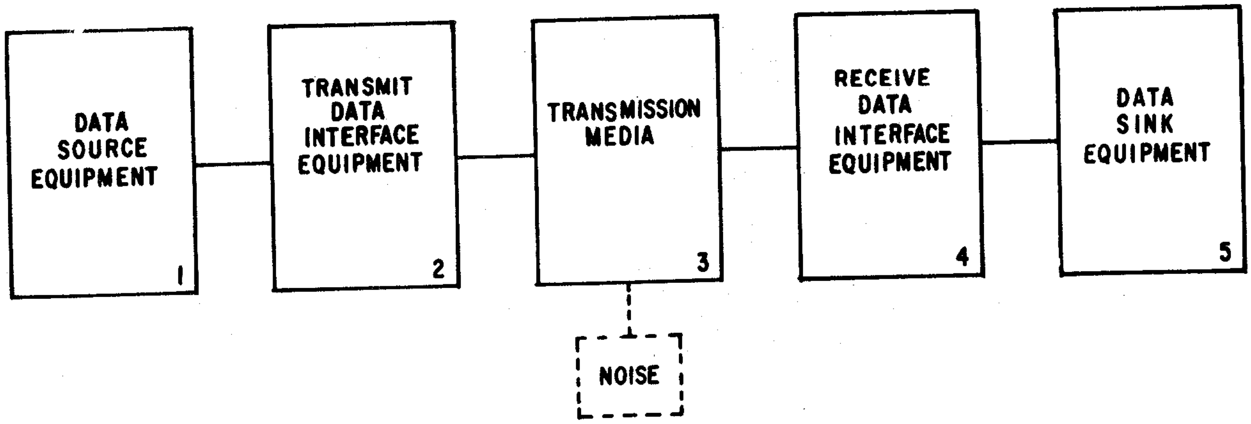


Figure 14-1. Basic Elements of a Data Communications System.

The teletypewriter transmitter (keyboard or tape reader) converts mechanical keyboard typing or tape-sensing actions into a normally serial sequence of binary pulses, each of which is recognizable as an alphanumeric (letter/number/symbol) or a control function character. Since teletypewriters use the asynchronous transmission system, each character requires that start/stop (and in some cases, parity) bits be transmitted as a part of the character sequence. The character-interval bit-sequences most commonly used in teletypewriter systems are shown in figure 14-2.

b. Punched Card Terminals. While the teletypewriter is most commonly used in the communication of narrative, the punched card readers and punches are by far the most common input and output media to the digital computer. The card (figure 14-3) measures 7 3/8" by 3 1/4". The format is a 12 x 80 matrix. The rows, which run horizontally across the card, have the values 12, 11, 0, 1, 2, 3, 4, 5, 6, 7, 8, and 9, counting from the top to the bottom. Each vertical column represents an alphanumeric which is determined by one or more holes punched in the desired column. Punches in rows 12, 11, and 0 are called "zone" punches and punches in rows 1 through 9 are called "digit" punches.

The Hollerith Code (figure 14-3) represents the alphanumerics by a combination of zone and digit punches. The cards can be manually prepared on a keypunch machine or may be prepared IAW on-line electrical signals received from a computer or directly from another card terminal. The presence or absence of punched holes may be either mechanically or optically detected, with the presence of a hole normally read as a "mark" or binary "1." The information contained on the card may be translated into any of the standard binary codes previously discussed or into a specialized computer code. Low-speed card punches and readers operate in the speed range of 4 to 12 cards-per-minute (CPM). High-speed card terminals may operate in the area of 100 to 200 CPM.

c. Facsimile Terminals. The transmission of graphics data (maps, pictures, drawings, etc.) at relatively slow speeds (compared with television) is called facsimile. It is similar in many respects to television, because many of the terms used to describe the methods and quality of transmission (such as scan rate, resolution, synchronization between transmitter (master) and receiver (slave)) are the same. All facsimile systems involve the optical scanning of the printed copy which is attached to a rotating drum. Using a precision optical system, the copy is scanned line-by-line, from left to right, and from the top to the bottom, much in the manner in which a person reads the printed page. The received optical patterns are converted into an analog signal, which varies directly IAW the graphical information appearing on the copy.

Two types of facsimile systems are currently in use: the "gray-scale" and the "black-and-white." The gray-scale type provides an electrical output which is directly proportional to the light reflected from the copy in the various shadings from light to dark. The black-and-white type provides a binary output such that when the reflected light exceeds a preset value, the binary output assumes one value and when the reflected light falls below the preset value, the binary output assumes the other value. The gray-scale system is applicable to the transmission of picture-type data while the black-and-white system is appropriate for the transmission of printed matter. Facsimile data may be transmitted in either analog or binary form. For the purpose of this chapter, we are concerned with its transmission in binary form which permits its encryption and transmission via digital data systems.

d. Computer Terminals. In terms of the total volume of digital data transmitted, the data systems which link computer-to-computer provide for the greatest bulk of data flow at the highest speeds. In general, these circuits process information in a synchronous mode at modulation rates of 2400 baud or higher. In large data switching networks (such as AUTODIN), the high-speed computer-to-computer links provide the means to assure interarea message delivery.

e. Voice Terminal. The telephone or microphone is the device which transforms aural data into electrical signals. These analog signals vary in amplitude and frequency directly with the aural information received. While the analog signals can be transmitted directly, they are included here because of the requirement in some systems to convert the signals

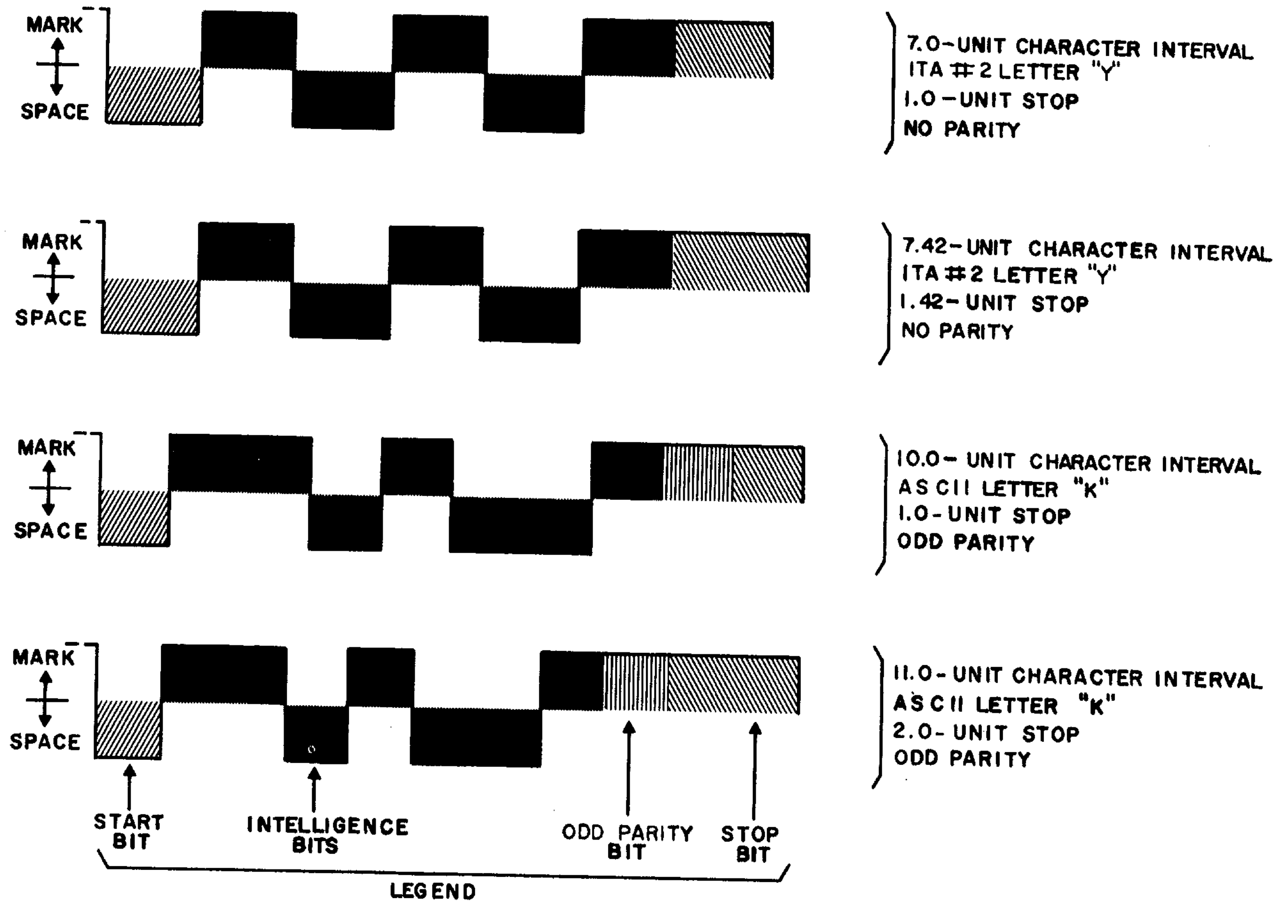


Figure 14-2. Typical Data Character-Internal Bit-Sequences.

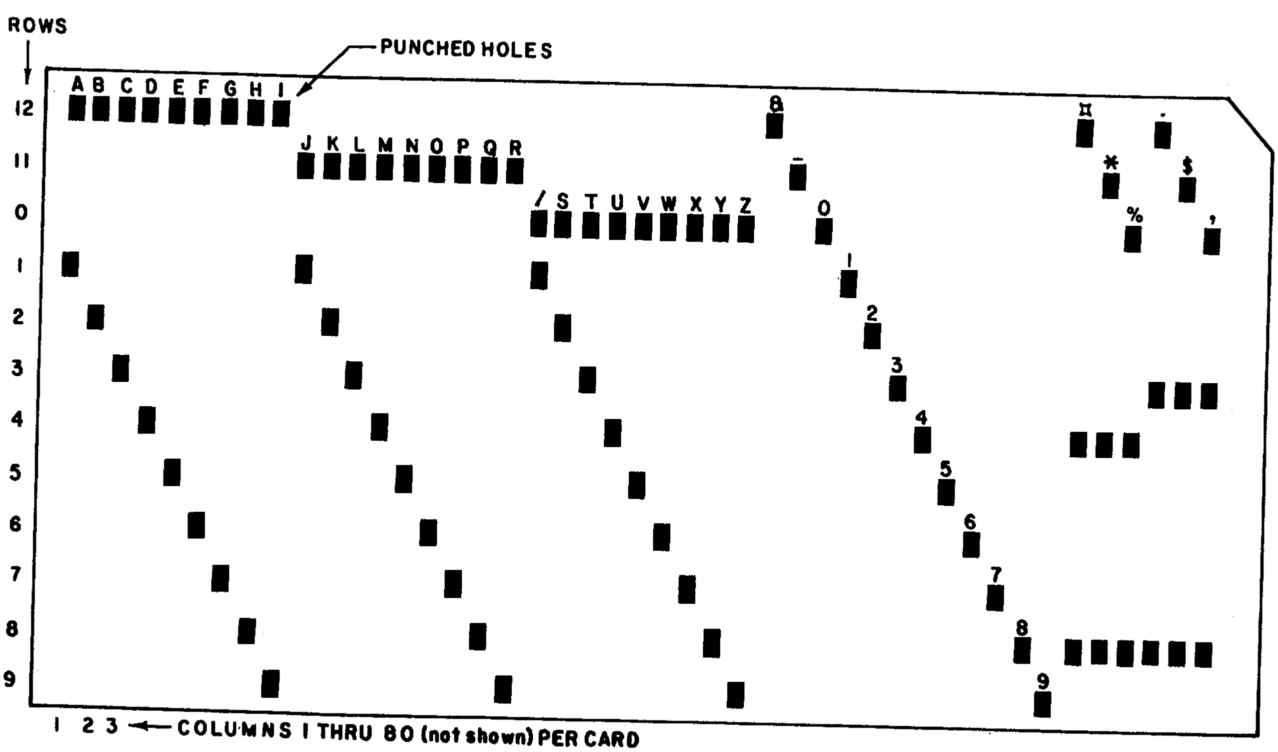


Figure 14-3. Partial Representation of the "Hollerith" Punched-Card Code.

to binary form for encryption and transmission via digital data systems.

f. High-Speed Paper Tape and Magnetic Tape Terminals. This equipment is almost always used in data systems related to computer systems. As such, code, format, and language, common to the computer being served, is used. Modulation rates from 1200 to over 2400 baud are used.

14-4. Data Interfacing Equipment. Referring back to figure 14-1, blocks 2 and 4, we see that there is a category of equipment which we have referred to as "interfacing." While this may not be a good term and certainly not a definitive one, there simply is no other way of effectively categorizing that broad range of equipment — equipment which adapts, modifies, converts, transforms, encodes, translates, or otherwise changes the signals which originate or terminate at the data sources and sinks, as well as the signals which access the transmission media. Let us consider the various functions of this category of equipment.

a. DC Signaling Level Interface Equipment. This equipment provides one of the simplest types of interface functions. Its purpose is to adapt the digital signaling *level* characteristics of one equipment to meet the different digital signaling level characteristics of another equipment in the data system. This interfacing function is shown in figure 14-4a.

b. DC Signaling Mode Interface Equipment. Using a simple example, assume that the binary output is in the neutral mode (for example, the spacing pulse is zero voltage/zero loop current), but the signaling level characteristics are compatible with the modulation rate converter. In this case, the interface equipment must provide for neutral-to-polar mode conversion to effect compatibility. In some cases, one type equipment will provide for both signaling level as well as signaling mode conversion. In addition, the interface equipment will provide for signaling sense conversion. Typically, one equipment uses a positive current-flow for the marking pulse (binary 1) sense and another equipment uses a negative current-flow for the marking pulse (binary 1) sense. The signaling mode and signaling sense interfacing functions are shown in figure 14-4b and c, respectively.

c. Modulation-Rate Conversion. This is a process whereby the rate of data transmission is speeded up or slowed down. For example, certain teletypewriter equipments use the 11-unit American Standard Code for Information Interchange (ASCII code) at a modulation rate of 110 baud (100 WPM). This speed is not compatible with certain encryption and modem clocking equipments, so the modulation rate must be increased to 150 baud (data thruput remains constant at 100 WPM). The output signal resulting from such a conversion then becomes a succession of character intervals separated by a no-traffic or steady-marking condition equal to 26.7 milliseconds. Conversion of a data signal from a lower to a higher modulation rate does not require control of the signal source read-out and only a small amount of converter storage is required. Conversion of a data signal from a higher to a lower modulation rate requires that either the data be clocked out of the source character-by-character or that adequate storage be provided to satisfy the time-averaged message thruput rate which may not exceed the thruput rate at the output of the converter. Modulation-rate converters must contain clocking (either internal or external) to clock the data out of the converter and, where necessary, to control the data read-out from the source equipment. Figures 14-5 and 14-6 show the two directions of modulation-rate conversion.

d. Analog/Digital Converter Equipment. In those cases where it is necessary to protect data transmission from unauthorized disclosure or interception, the data should be encrypted prior to entering the transmission media. This requires that the data be in binary form. Analog-to-digital conversion equipment is, therefore, used to convert facsimile, voice, and television data to binary format. The conversion process consists of pulse amplitude sampling of the analog signal, assigning values to (quantizing) the samples, and the binary coding of the samples. The process requires precise timing (clocking), both at the transmitter terminal and at the receiver terminal (usually slaved). In the usual configuration, the routing of the data timing or clocking pulses is as shown in figure 14-7.

e. Encryption Equipment. Encryption (cryptographic) equipment is required whenever classified data is to be transmitted over unsecured transmission media. Transmissions between terminals are synchronous and require highly accurate timing which may be provided either internally or externally.

f. Digital Data Multiplexing Equipment. There are two common types of digital data multiplexer equipment: TDM and FDM.

(1) Time-division digital multiplexing equipment is used where it is desired to combine several binary data input signals into a single serial binary data output signal. Although the techniques of TDM differ, each method relies on the sequential sampling of each input binary data stream and the placement of each sample into a specific time-slot in the output binary data stream.

The recovery of the data at the receive (demultiplexing) terminal requires that the desampling process be in synchronism with the send (multiplexing) terminal. The modulation rate of the output binary stream will, in general, be equal to or greater than the arithmetic sum of the modulation rates of each of the input data signals. The composite data signal is normally accessed to the communications media via suitable modem equipment. A simplified method of accomplishing TDM is shown in figure 14-8.

(2) FDM equipment performs both the function of combining multiple input binary data signals as well as the function of modulation. In this type of multiplexing, each input binary signal is converted so that the "mark" state is represented by one audio frequency and the "space" state is represented by another audio frequency. The separation of the "mark" and "space" frequencies varies as a function of the modulation rate. In a typical system designed to accommodate 16 input circuits, each with a maximum modulation rate of 75 baud, the "mark" and "space" frequency separation is 85 Hz.

In this system, the "mark" and "space" frequencies are 382.5 Hz and 467.5 Hz, respectively, with the channel #1 center-frequency at 425 Hz. This system can

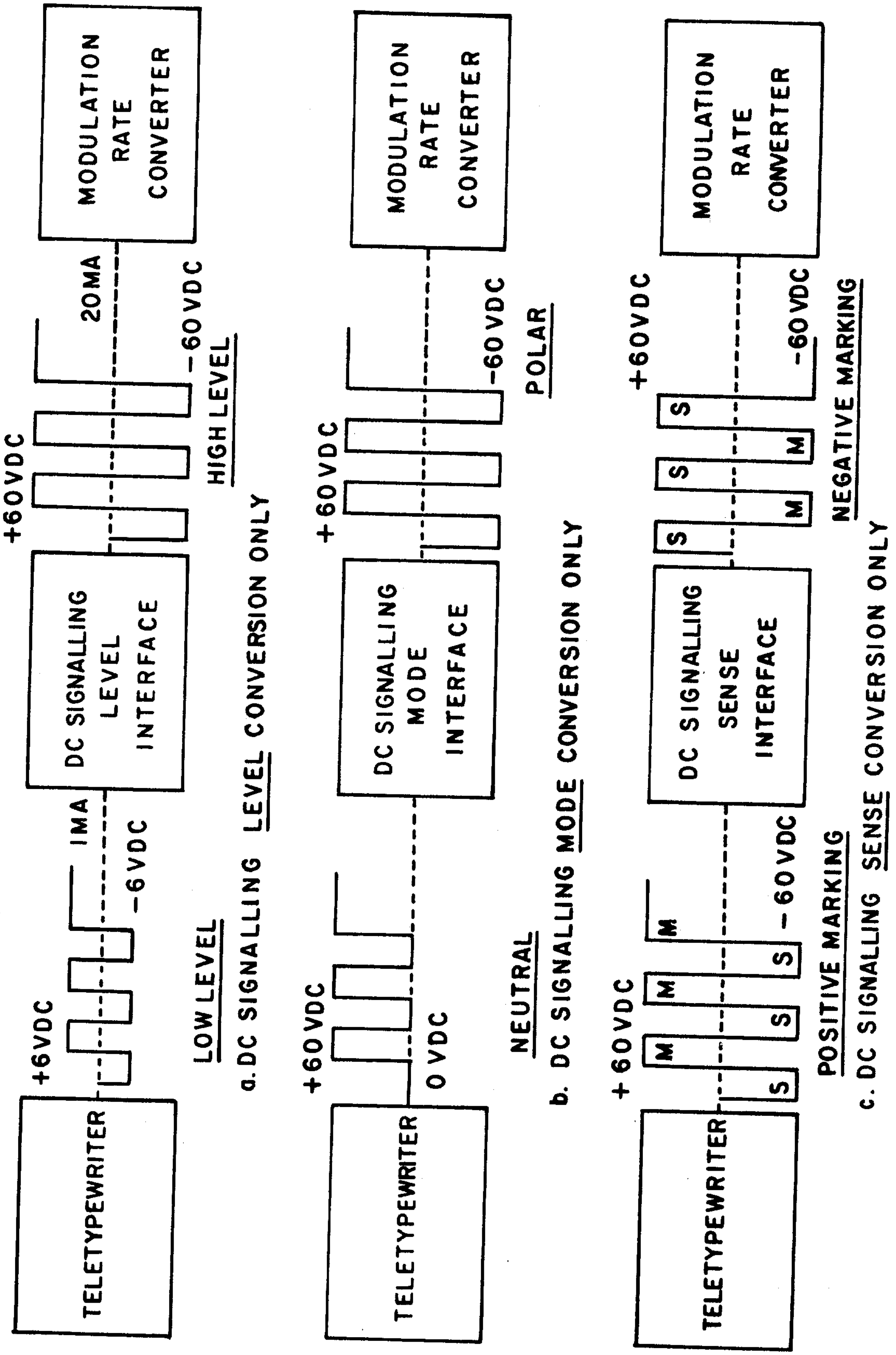


Figure 14-4. DC Signaling Interface Functions.

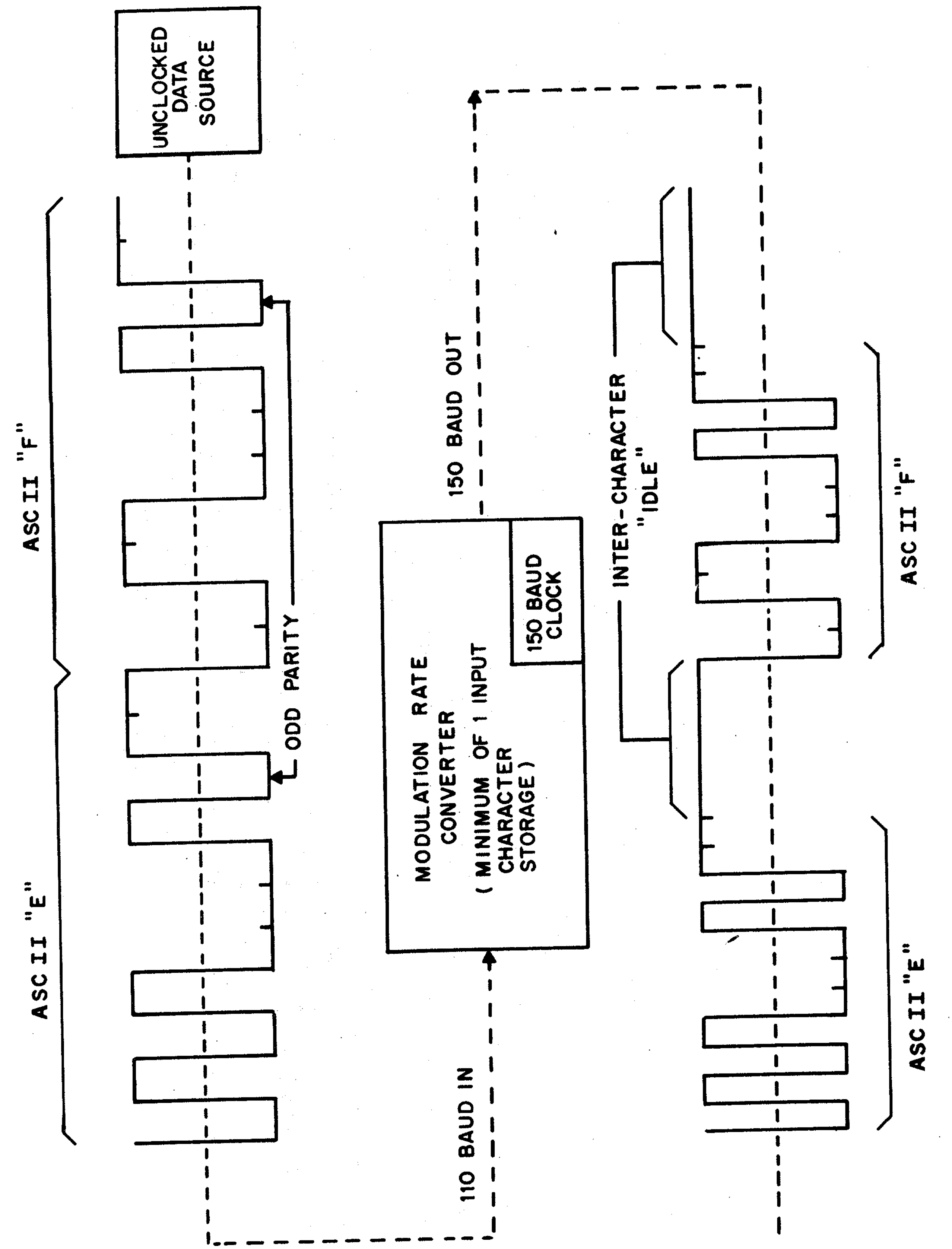


Figure 14-5. Modulation Rate Conversion (Low to High).

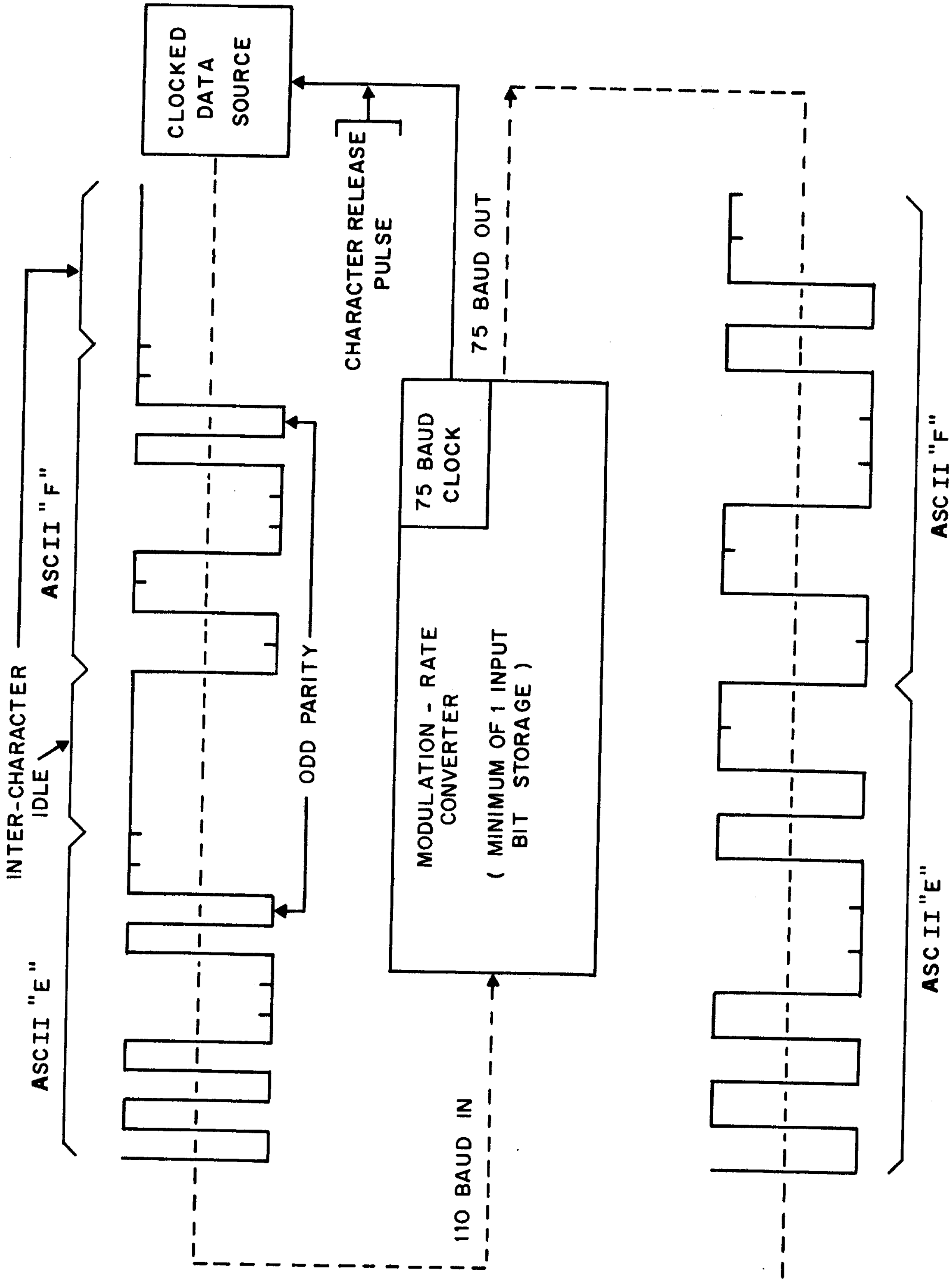


Figure 14-6. Modulation Rate Conversion (High to Low).

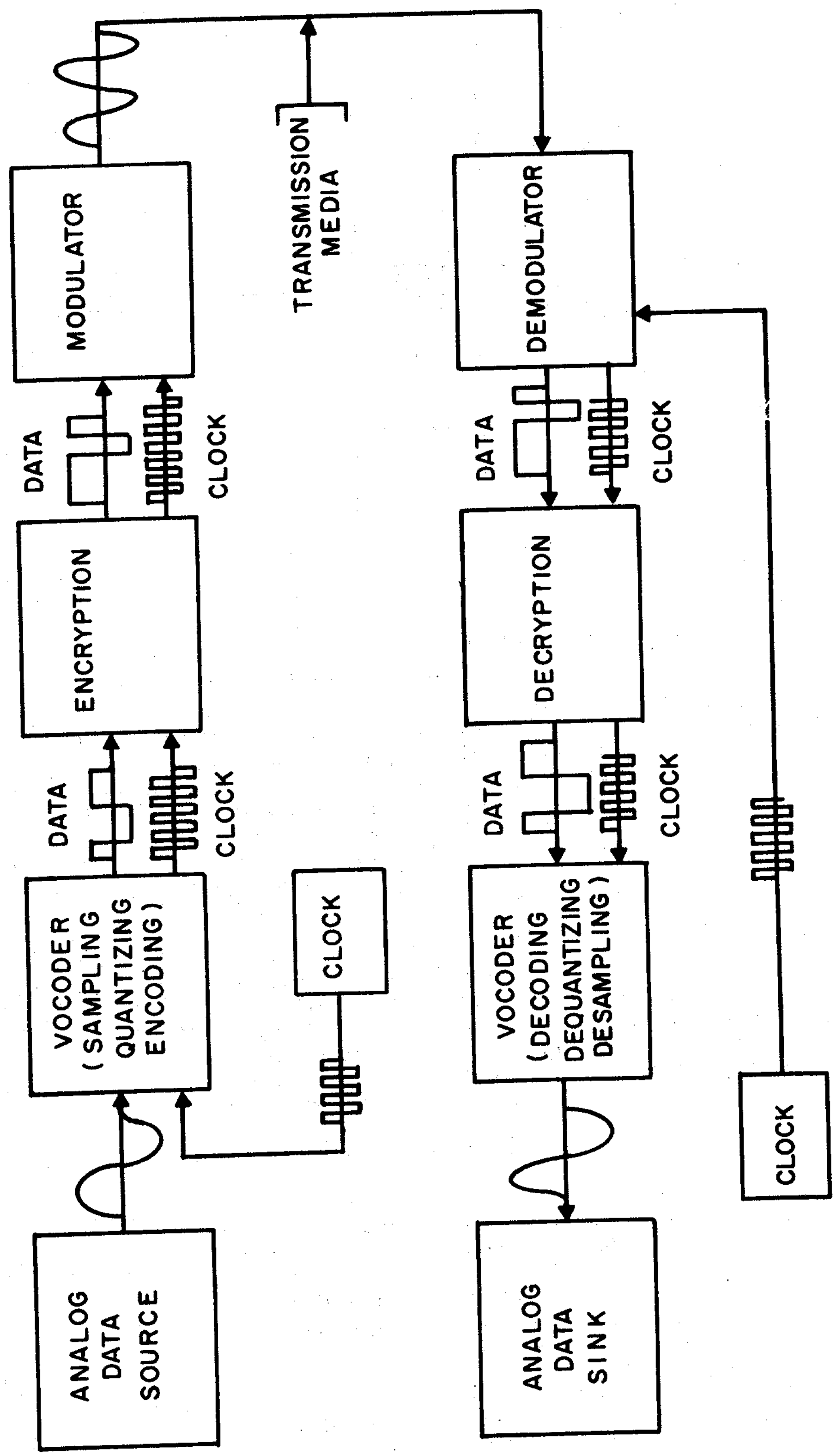


Figure 14-7. Analog/Digital Conversions.

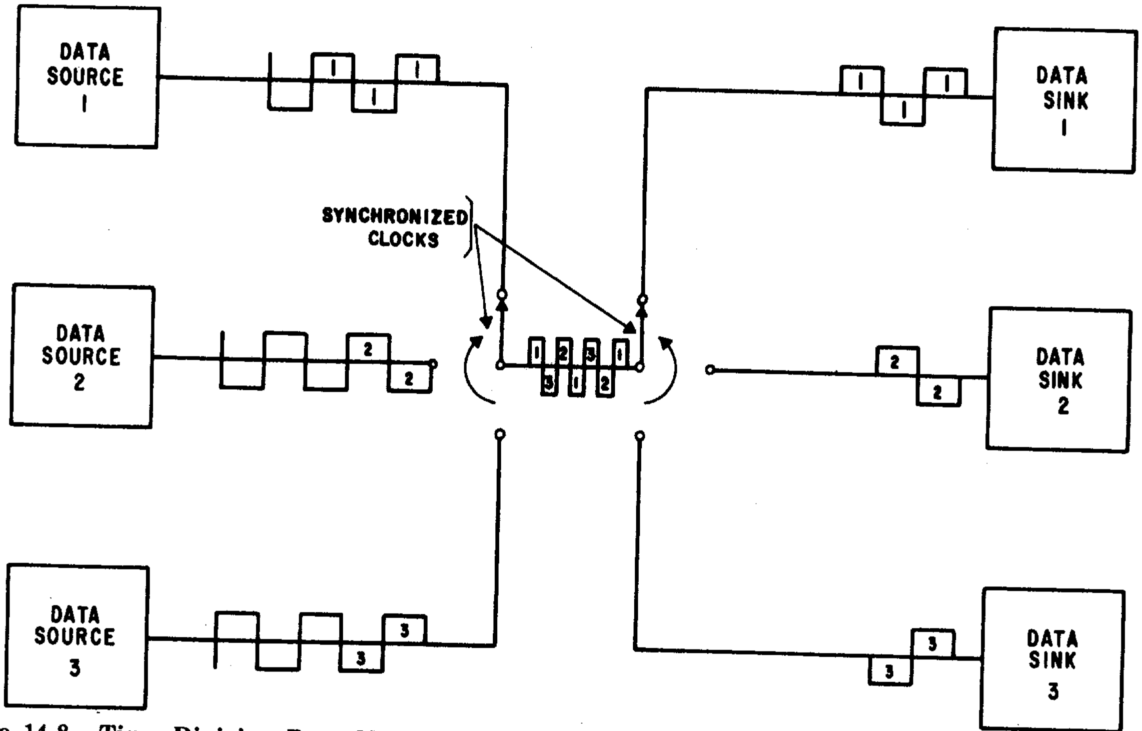


Figure 14-8. Time Division Data Multiplexing.

accommodate 15 additional binary data circuits with channel center-frequencies assigned at 595, 765, 935, 1105, 1275, 1445, 1615, 1785, 1955, 2125, 2295, 2465, 2635, 2805, and 2975 Hz. The adjacent-channel center-frequency separation is 170 Hz and spacing between the "mark" frequency of one channel and the "space" frequency of the next lower adjacent channel is 85 Hz.

The "mark" and "space" frequencies of all 16 channels are thus combined into one voice (3 kHz) band and are ready for entry into the transmission media. At the receive end, the channels are bandpass-filtered into the 16 audio bands and the individual "mark" and "space" frequencies are phase-detected, resulting in the recovery of the binary data for each of 16 output channels. FDM has, therefore, permitted us to transmit 16 each, 75 baud circuits (1200 baud) over one 3-kHz voice channel. This system of multiplexing does not require end-to-end synchronization. Figure 14-9 shows the whole operation of the system described above.

g. Modem Equipment. Up until now, our discussion of data signals in the source, sink, and interface elements of the data system has dealt with data which exists in the binary DC form; however, the transmission of DC pulses through the various transmission media is not practical, except over relatively short distances of cable. Even in these cases (usually intra-area or intra-base systems), interpair cable interference (or crosstalk) makes such use inadvisable. Binary data signals have very low frequency components which extend down to and include DC, while typical communications media have low frequency cutoffs at 200 to 300 Hz.

At the high frequency end, the data signals generally contain significantly higher frequency components than the communications media can accommodate. The modem is the device which adapts the binary DC data signal to an AC signal suitable for transmission over the various communications media. As the requirements for data communications have grown, so too have the efforts to maximize the use of available communications media. These efforts have led to the development of various modulation techniques including amplitude, vestigial sideband, frequency shift, and phase shift modulation.

Although no longer widely used, AM techniques were developed for early, low-speed telegraph systems. In this scheme, an audio or radio frequency is keyed "on" or "off" to represent the "mark" and "space" states of the binary signal. AM is relatively inefficient in its use of bandwidth and is sensitive to the effects of changing transmission levels and noise.

Vestigial sideband modulation (VSB) results from the partial suppression of the carrier and the sideband components in the vicinity of the carrier and the elimination of the remainder of one sideband (only a vestige of the deleted sideband is transmitted). VSB, therefore, has the advantage of approximately halving the bandwidth required for AM for a prescribed data modulation rate.

The FSK type of modulation is relatively insensitive to changes in amplitude, IPN, and other disturbances which affect signal levels and it is, therefore, superior to AM.

PSK modulation is accomplished by shifting the phase of a carrier. The binary "1" and "0" states might then be represented by the 0° and 180° phases of a carrier. PSK modulation requires that an accurate, stable, reference phase be available at the receiver to

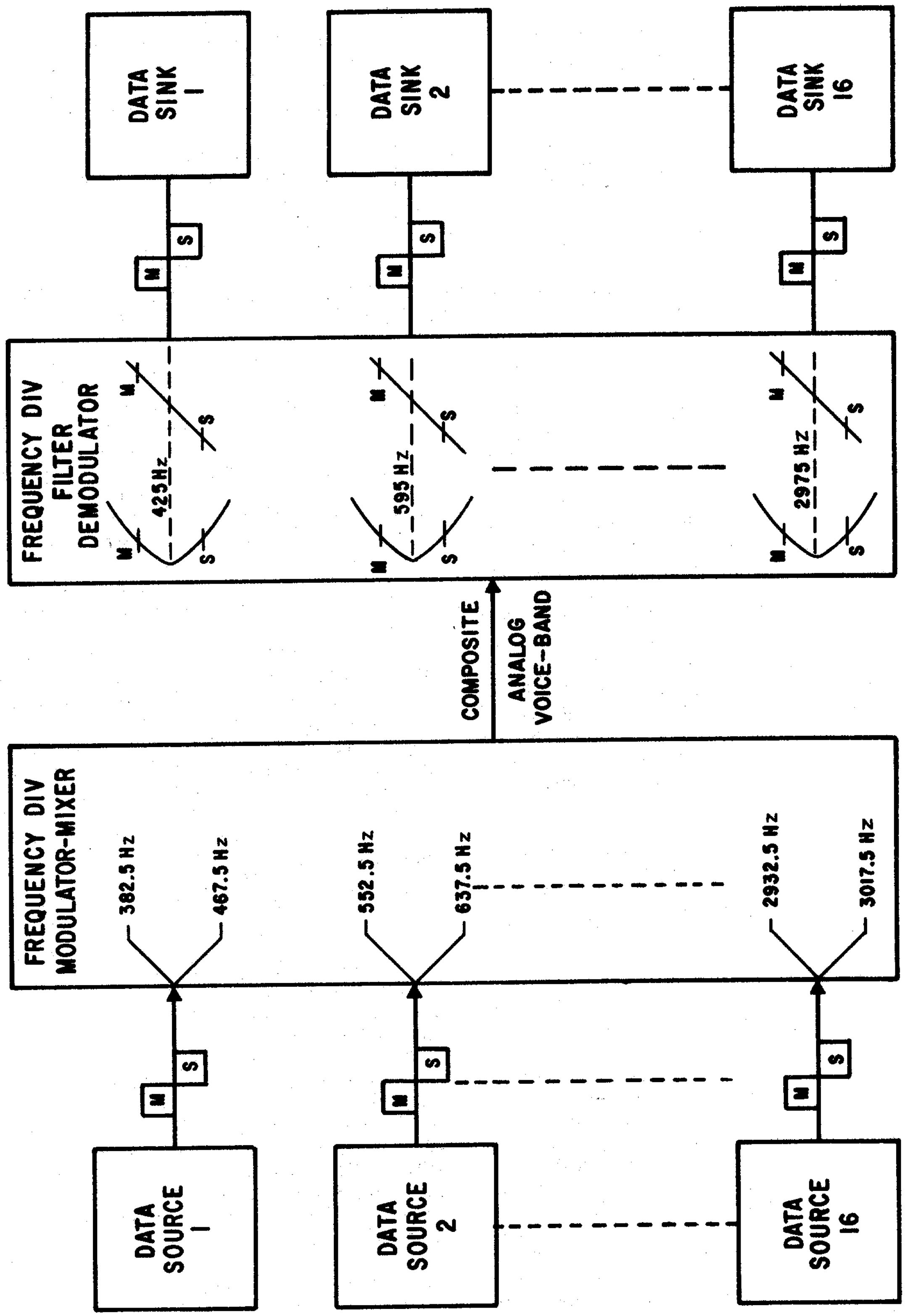


Figure 14-9. Sixteen Channel Frequency Division Multiplex. This equipment, normally referred to as VFCT (Voice Frequency Carrier Telegraph), puts 16 data signals on one voice-band channel.

distinguish between the two phases. This reference problem can be avoided by using differentially-coherent PSK (DC PSK) in which method the information is encoded in terms of phase changes, rather than absolute phases, and detected by comparing phases of adjacent bits. Increasing the number of phase states to three (ternary), 0° - 120° - 240°, or to four (quarternary), 0° - 90° - 180° - 270°, increases the number of data bits that can be transmitted. But the penalty which must be paid for the higher data rate results in increased vulnerability to noise and their media impairments since individual phase states are more difficult to distinguish as the number of states increases.

Most high-speed data modems operate within synchronous systems where precise timing must be provided at both the receive and transmit terminals. These timing sources (clocks) may be provided by the modem or by external terminal equipment. In either case, the timing of receive terminal equipment is normally phase-locked to the timing of the received data bits. In addition, there are applications where system synchronism must be maintained during periods of time when a circuit drop-out occurs. Several data modems are equipped with internal clocks (with typical accuracies of 1 part in 10^7) which permit synchronism to be maintained for circuit drop-out periods ranging from several minutes to a day or more, depending on the modulation rate. With the development of new encoding and modulation techniques, the transmission of data with modulation rates of up to 9600 baud over highly conditioned voice channels has become feasible.

h. Miscellaneous Interface Equipment. Other interface equipment, including format generators, regenerative repeaters, and series/parallel converters whose functions are self-evident, will not be discussed here.

14-5. Characteristics of the Communications Media. The performance of the transmission media is the single, most important factor in the effective transmission of data. While the susceptibility of the various media to the effects of weather, atmosphere, and distance differs, all media can be characterized in terms of bandwidth and S/N ratio. In fact, the classical treatment (by Shannon, Nyquist, et al) relates the theoretical digital capacity of a communications channel directly to the factors of bandwidth and S/N ratio. But this classical relationship is based on two assumptions: that the transfer characteristic of the channel is linear throughout the band with respect to amplitude and phase; and that the noise is truly "Gaussian" (or wideband) in nature. In fact, neither assumption is entirely valid for any communications channel. Nyquist specified that the maximum amount of data (C) which could be transmitted through his "ideal" channel of bandwidth (B) was proportional to the logarithm (to the base 2) of the number of levels (m) that the signal can assume; or, more precisely, C (in bps) = 2B (in Hz) $\log_2 m$.

One might conclude, therefore, that an "easy" method of increasing the flow of data would be to simply increase the number of significant modulation levels (or states) from binary, to ternary, to quaternary, or even higher; that is, that the modulator would recode the input binary stream into an output which could, for example, represent any of eight different phase states (0°, 45°, 90°, etc.). According to the Nyquist formula then, a signal which can be represented in any of eight levels could be used to transmit data at 9000 bps (or baud) through a channel of 3000 Hz. As usual, however, the easy approach falls short in achieving the desired result. As we increase the number of significant modulation states, we make it more and more difficult for the demodulator to separate or distinguish between the various modulation states in the presence of noise or distortion. The S/N ratio, therefore, imposes a practical limit on the number of modulation levels which may be used.

a. Noise. Noise is any unwanted signal introduced into the transmission channel and ranges from predictable noise sources (such as power supply ripple) to completely unpredictable noise (such as atmospheric noise). Noise is introduced into all communications channels. To assure the effective transmission of data, it is necessary to maintain a ratio of signal power to noise power which is sufficient to assure a low data-bit error rate. Noise may be categorized as "Gaussian" (wideband "white" noise) or as IPN. Either type may exist for varying periods or "bursts" and may result in single data-bit errors or in successive data-bit errors depending on the length of the noise "burst." The various transmission media are subject to different kinds of noise. HF radio is subject to atmospheric, adjacent channel, and "Gaussian" noise (during periods of fade). Wirelines are especially subject to switching and "crosstalk" noise. Tropospheric scatter, M/W, and satellite communications media are primarily subject to "Gaussian" noise. The subject of noise and its effect on the transmission media was discussed in more depth in chapters 6 and 8.

The additional major factors which affect the transfer of data over transmission channels are the frequency response and delay distortion characteristics of the channel.

b. Other Media Impairments. Other characteristics of the transmission media may also adversely affect our ability to transmit data. Frequency translation within a data channel may result due to differences between the transmit and receive oscillator frequencies in a single-sideband (or vestigial sideband) suppressed-carrier voice multiplexing system. The effect is to move the data channel modulation frequencies outside, or to the edge of, the receive bandpass filters. Intermodulation and harmonic distortion characteristics of the media may cause the introduction of extraneous frequency products into the data channel. Crosstalk, phase jumps, phase jitter, amplitude jumps, and echoes are other factors which affect data transmission systems.

14-6. Detection and/or Correction of Errors in Data Systems. In the previous paragraph, we discussed the classical theory that the data transfer capacity of a communications channel is directly related to bandwidth and S/N. We saw further that the idealized conditions of linear bandwidth (amplitude and phase) and purely Gaussian noise could not be realistically achieved. The classical theory does tell us, however, that we can increase the capacity of a data channel either by increasing the channel bandwidth, by increasing the signal power or by decreasing the noise power.

We are limited in our efforts to increase channel bandwidth because of the limited spectra available and the tremendous growth of data to be handled. The amount of power we can use is also limited because of the constraints of cross-channel interference and the consequent introduction of noise into other communications channels. Though our efforts to reduce noise power continues, many of the sources of noise remain outside our control. It is apparent, then, that in our efforts to accommodate the requirements for increased data flow, we must tolerate some degree of error in the transfer of data.

Although Gaussian-type noise constitutes a source of data errors, primarily in media, subject to deep signal fades, the primary source of data errors is caused by sharp impulse-type noise bursts whose power is concentrated within a narrow frequency band. In the transfer of data by words or by telephone, sufficient redundance exists for individual letters or even entire words to be in error or lost without significant effect on the overall meaning. Data, however, often consists of a stream of numbers and letters which are seemingly unrelated; therefore, the loss of a single bit results in the loss of the meaning of the character, since there may be no clue in the remaining bits or characters which can lend meaning to the mutilated character. It is easy to see that a single bit-error in the ITA # 2 Code (a change in the fifth intelligence bit) can change the number "1" to the number "7" and vice versa. Consider the consequences of such an error in dispatching troops, citing inventory levels, or ordering equipment. It is important, therefore, that some types of data be delivered error-free and this process involves error detection and error correction.

a. Error Detection Techniques. The first step in the control of errors in data transmission systems is the detection of errors. Most error-detection systems either add additional (redundant) bits in the data transmission or use a code structure which is inefficient. An example of the latter is the IBM 4 of 8 code. This type of code accomplishes error detection by using alphanumeric code combinations in which the "mark" and "space" pulses always occur in the ratio of 4 to 4. This method is effective except when bit errors occur in pairs within the character. It is an inefficient code, in that 8 bits must be transmitted for each of the 70 available code combinations. In the character parity-checking error-detection system, an additional bit (called the "parity" bit) is added to the regular binary code intelligence-bit sequence so that there will always be an odd (or even, if desired) number of "marks" (1s) in each code group.

For example, in the ASCII Code, an eighth bit is added to each group of seven intelligence-pulses to assure that the number of "1s" is odd. The letter "a," which is represented by the sequence 1000001, consists of an even number of "1s." To provide odd parity, the parity or eighth bit would be a "1" and the new parity sequence would become 10000011. For the letter "q," the parity bit would be an "o," since the code for "q" already contains an odd number of "1s."

The parity error-detection system fails whenever the parity pulse itself is in error or when there is an even number of errored bits. For these reasons, it is approximately only 75% effective in detecting errors. As in all redundant systems, the transmission of parity bits slows down the data thruput rate. In addition to character parity checks (often referred to as vertical parity), block parity (horizontal parity) is sometimes also provided, as in AUTODIN. In the AUTODIN system, block (horizontal) parity is provided for each block of 80 characters in addition to character (vertical) parity. An example of this combination of odd parity-error detection for an abbreviated (for space reasons) block of 14 characters is shown in figure 14-10. Although there are other bit-error detection techniques, those described above are the most commonly used today.

b. Error Correction Through Retransmission. Once an error is detected, there remains the problem of error correction. In some systems (once the error is detected at the receive terminal), a printout of a symbol (indicating that the character is in error) is made and subsequent correction is left to the operator. In other systems, the data is checked character-by-character (or, more commonly, block-by-block) and the character or block is retransmitted (on request from the receiver) until the character or block is received error-free. This system is commonly referred to as Automatic Retransmission Query (ARQ).

In the AUTODIN system, the technique of operation is as follows: a block of 80 characters is transmitted and an end-of-block character is sent. At the receiver, both vertical and horizontal block parity checks of the received data are made. If no error is detected, the "ACK" (acknowledge) character is transmitted back to the transmitter and the transmission of the next 80-character block begins. If an error is detected, the "NAK" (negative acknowledge) is transmitted back to the transmitter which proceeds to retransmit the same 80-character block again.

Transmission efficiency, using the ARQ system, is usually not greater than approximately 75% due to the redundant parity bits required, the time waiting for receipt of the "ACK" or "NAK" character, and, on poor transmission channels, the requirement to retransmit some errored blocks several times. In the case of the AUTODIN block length, for example, it takes only one bit error in 648 bits to cause the retransmission of 647 good bits. On poor transmission channels, the reduction of the block-length can result in an overall improvement in the data thruput rate.

c. Forward Error Detection and Correction. The ultimate goal of error detection and correction is the accomplishment of both without the requirement for retransmission. This system is referred to as forward error correction. The main requirement of such a system is that it be capable of identifying exactly what bits are in error and of correcting the errors by merely inverting the bits. A detailed analysis of the method used to effect forward error correction will not be presented here. It is sufficient to specify that all such

ASCII BIT NO.	C	O	M	M	U	N	I	C	A	T	I	O	N	S	BLOCK (HORIZONTAL) ODD PARITY
	(14 CHARACTER BLOCKS)														
1	1	1	1	1	1	0	1	1	1	0	1	1	0	1	→ 0
2	1	1	0	0	0	1	0	1	0	0	0	1	1	1	→ 0
3	0	1	1	1	1	1	0	0	0	1	0	1	1	0	→ 1
4	0	1	1	1	0	1	1	0	0	0	1	1	1	0	→ 1
5	0	0	0	0	1	0	0	0	0	1	0	0	0	1	→ 0
6	0	0	0	0	0	0	0	0	0	0	0	0	0	0	→ 1
7	1	1	1	1	1	1	1	1	1	1	1	1	1	1	→ 1
CHARACTER (VERTICAL) ODD PARITY	↓ 0	↓ 0	↓ 1	↓ 1	↓ 1	↓ 1	↓ 0	↓ 0	↓ 1	↓ 0	↓ 0	↓ 0	↓ 1	↓ 1	↓ 1

Figure 14-10. Character (Vertical) and Block (Horizontal) Combination Parity Check. Parity checks improve accuracy at the expense of data thruput.

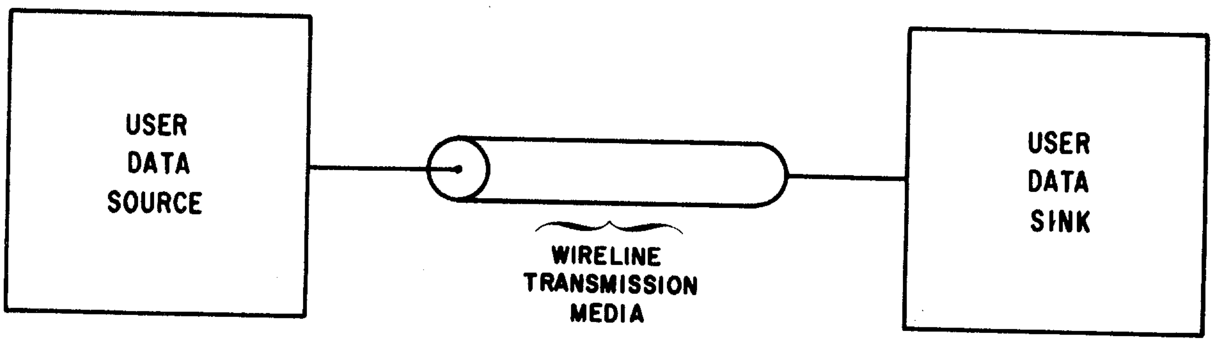


Figure 14-11. Simple User-to-User Data Network.

systems involve the recoding of the data through the addition of additional bits at specific points in the data stream. The coding techniques are varied and, in general, are selected on the basis of the duration of the noise-bursts on a particular communications channel. The extent of redundance and, therefore, the data thruput rate, varies as a function of the degree of error correction provided.

14-7. Typical Data Networks and Systems. The complexity of data networks and systems ranges from the simple cross-base teletype circuit to the more complex computer-operated message and circuit switching systems (such as AUTODIN). Let us consider some of the data systems and networks in use today.

a. User-to-User Data Network. Figure 14-11 shows a simple user-to-user data network, using wireline as the transmission media. In this system, like terminals "talk" to each other and there are no requirements for encryption or other interface equipment. Since it is a simple network, its operation is both uncomplicated and efficient.

b. User-to-User Data Network. In figure 14-12, we show additional requirements for the use of interface equipment and the requirement for an intermediate relay point in the transmission media. This system is more complex and, with the same data thruput rate as shown in figure 14-11, is less efficient.

c. User-to-Message Relay-to-User Data System. In figure 14-13, we show user terminals which require the capability to transmit data to any of several other user terminals. In this case, the data or message arrives at the intermediate message switching system where it is either manually or automatically routed to the desired receive terminal.

In the manual teletypewriter relay , this process consists of an operator transferring a paper-tape from a tape-punch (where it has been received from the sender) to a tape reader (where it will be transmitted to the addressee). Since the process involves the "tearing" of the paper tape at the tape punch, this manual message-relay system is referred to as a "torn-tape" system. In the semiautomatic or the automatic message relay system, various portions or all of the torn-tape functions are accomplished automatically and, in all cases, at a much faster processing rate.

The AUTODIN message switching system (the largest in the world) processes billions of words-per-day

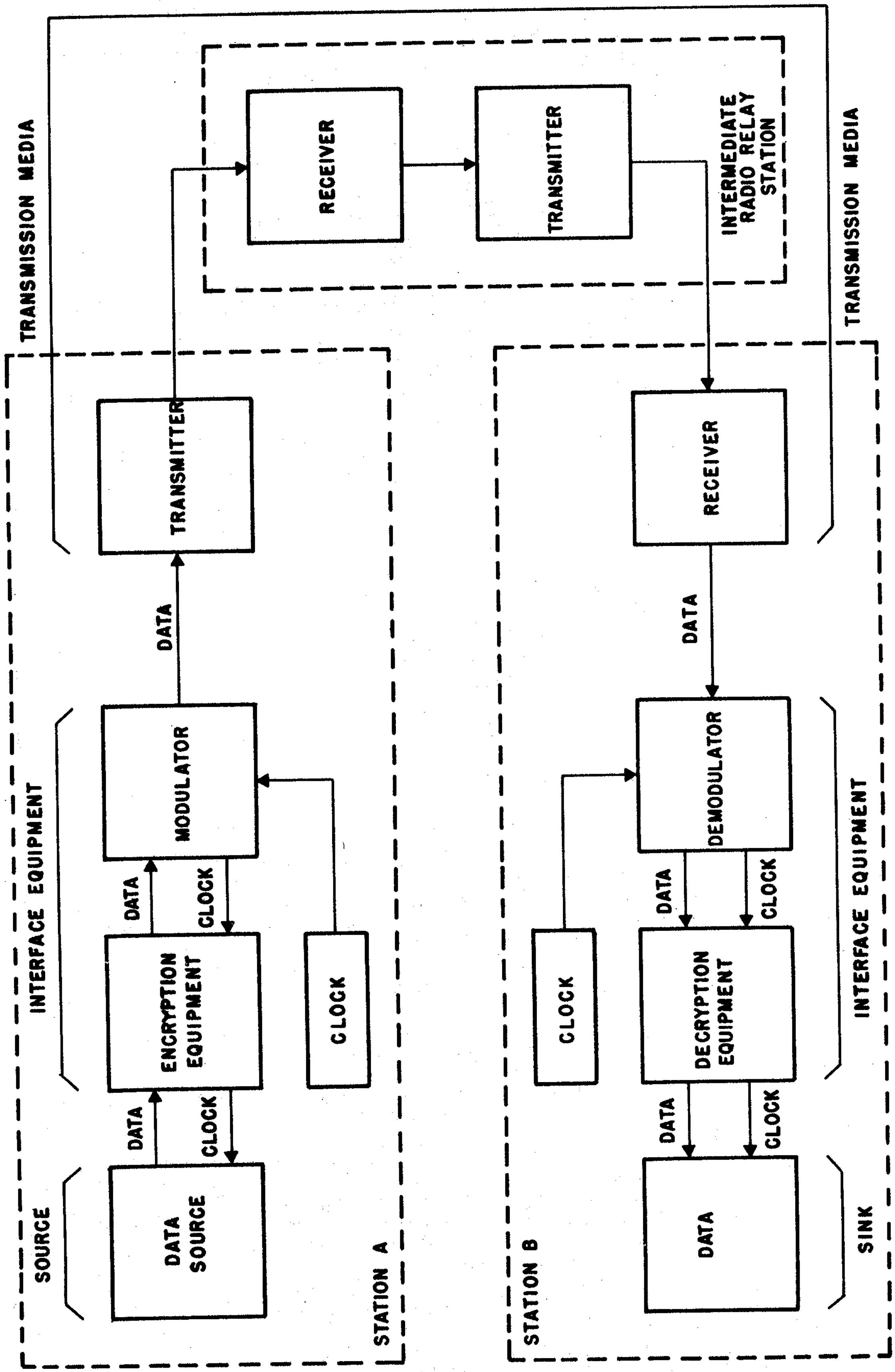


Figure 14-12. Interstation Data Network With Intermediate Relay.

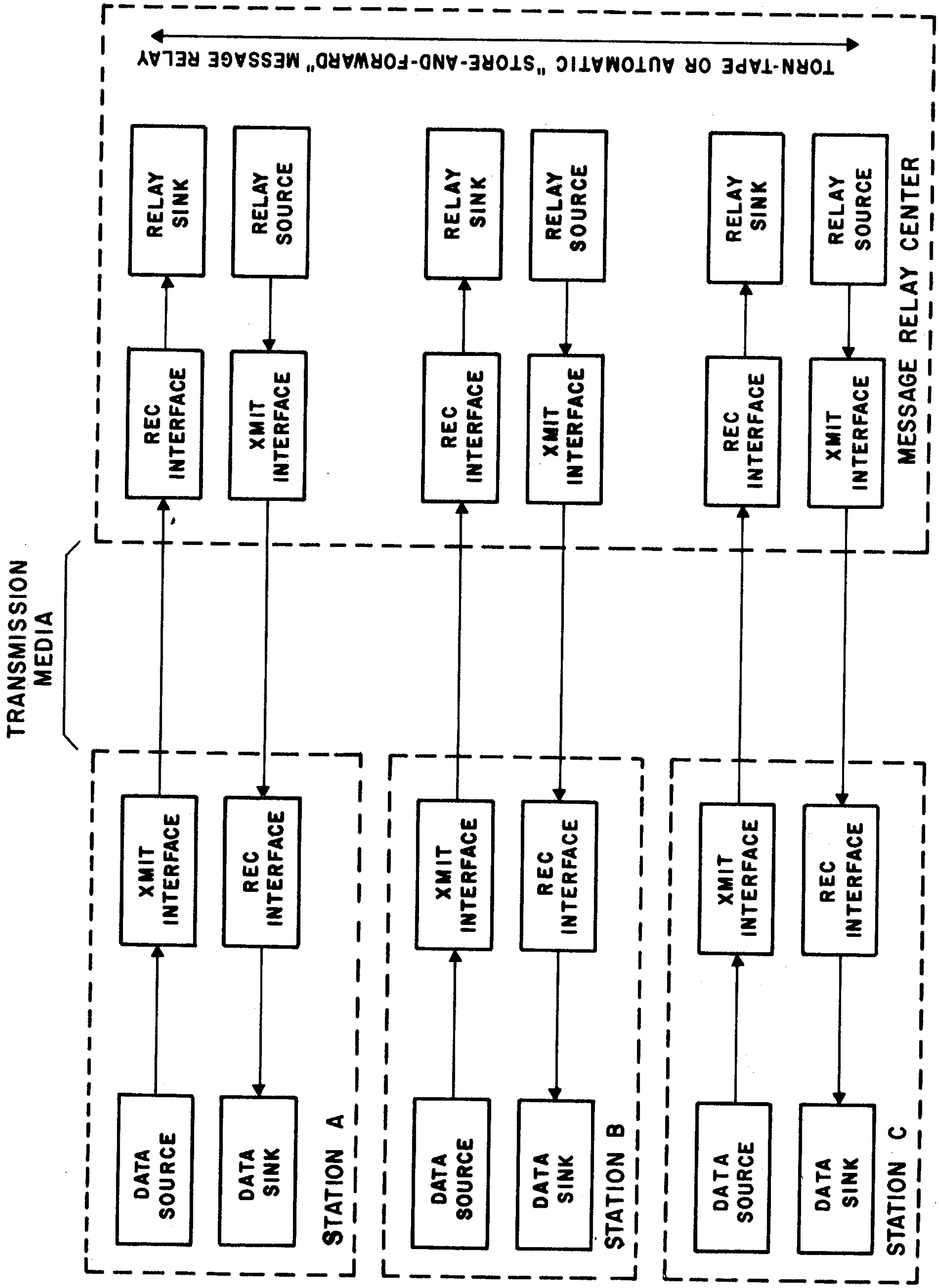


Figure 14-13. Typical Message Relay Data System.

between user terminals throughout the world and, at the same time, does it on a priority basis, accomplishing intermediate error detection/correction, providing protection for classified information, and carrying on the most complex kind of simultaneous translations, bridging an incredibly intricate mixture of equipment, speeds, codes, and languages.

Message relay or message switching systems all involve a period of storage prior to forwarding the data on to the addressee and are generally referred to as "store-and-forward" message relays. Certain AUTODIN switches within the Automatic Electronic Switching Center (AESC) provide circuit switching as well as message switching. Unlike message switching which has an indeterminate "store-and-forward" time, circuit switching provides direct point-to-point service between compatible user terminals.

d. Other Data Systems. In addition to the network configurations described above and which are generally referred to as common-user systems, there are specialized systems which process data related to a specific mission. In the collection and dissemination of weather information, for example, extensive networks have been established in which weather reporting stations transmit data automatically and in a sequential order determined by the Net Control Station (NCS). Other data networks (including Strategic Air Command Control System (SACCS), Ballistic Missile Early Warning System (BMEWS), Defense Special Security Communications System (DSSCS), Space Detection and Tracking System (SPADATS), and Semi-automatic Ground Environment (SAGE)) dedicated to the accomplishment of a specific mission where the requirements for reliability, survivability, error rate, and data thruput are specifically defined.

14-8. Summary. In this chapter, it has not been our purpose to provide an in-depth treatment of all the elements and factors which bear on the transmission of data. It has been our purpose to broadly describe the workings of data systems and to introduce the terminology which is normally associated therewith.

Chapter 15
SYSTEM INTEGRATION

15-1. **General.** Chapter 1 described a sample wideband system incorporating a variety of equipment configurations. System components were subsequently discussed in detail in chapters 9 through 11. In this chapter, we will show how these components are integrated at the site level to form a working link in a system. Three sites will be discussed in detail in order to trace the signal through the system.

15-2. **Able Site.** This site is a terminal station providing duplex terminal service for 60 voice channels to Central City, 24 channels to Dolly, 12 channels to Hidden Valley, and 12 channels to Gallow (figures 1-4 and 1-5). All channels go as far as Central City with 60 terminating and 48 through-grouped to the other sites. Due to distance and terrain between Able Site and Central City, an IF repeater is provided at Baker Site. Certain equipment, depending on the number of channels required, is necessary at each station. Able Site requires proper equipment to provide voice terminal service to 108 customers.

a. **Transmit Direction. The communications traffic at Able Site is VF and may be telephone, data, or facsimile. Our concern at this location is to place the VF information on an RF carrier and transmit it to Central City through Baker Site. In the receive equipment, we take the RF carrier, detect the VF intelligence, and transmit it to our user or customer.**

Figure 15-1 is a block diagram showing major component parts of the LOS terminal at Able Site. The 108 channels of audio information are formed into nine 12-channel groups which make up two supergroups. The supergroups are placed in the proper location on the baseband by the supergroup modulator. After insertion of a multiplex pilot, the complete baseband is transmitted to the radio equipment where it is amplified, combined with the radio order wire and pilot, and used to modulate the radio carrier. The modulated carrier is then transmitted to Baker Site via waveguide, antenna, and LOS radio path.

b. Receive Direction. The receive function is almost the exact reverse of the transmit function. The receive radio frequency signal is received on the same parabolic antenna which was used in the transmit direction. This low level signal is routed through the circulator and into the mixer section. Here it is mixed with a local oscillator (which is ± 70 MHz from the incoming signal) to develop the 70 MHz IF. After amplification, the IF frequency is demodulated and, once again, we have the baseband containing two supergroups of 60-300 kHz and 312-552 kHz.

One group (60-108 kHz) is not used, but may be added later by including extra multiplex equipment. The receive baseband enters the supergroup demodulator **which places each supergroup in the frequency range of 312-552 kHz for entry into the group demultiplexer. Here the five groups forming each supergroup are broken out and sent into the channel demultiplexers.** From the channel demultiplexer, each voice channel goes to the TCF.

c. Customer Connections. Figure 15-2 shows, in simplified form, typical connections between the user and the wideband equipment. The lower portion shows a typical teletype function. The DC output from the teletype unit is routed through the DC patch facility in **technical control and into the keyer of one channel in** the 16-channel VFCT. The keyer then converts the DC to frequency shift audio and combines it with 15 other channels to form a composite VFCT group of 16 channels to be transmitted on one voice channel of the wideband system. Each VFCT channel is assigned a portion of the VF channel with mark and space frequencies separated by 90 Hz. The centers of the channels are separated by 170 Hz.

In the receive direction, this action is taken in reverse, with the composite VFCT group received from a channel in the wideband system being reduced to 16 pairs of frequency shift tones. These tones are converted to DC which serves as the signal to operate the teletype machine. The top part of figure 15-2 shows a typical 2-wire telephone circuit. Only two wires are connected to the telephone instrument. They must provide transmit/receive audio signaling.

Wideband systems are 4-wire with separated transmit and receive pairs. In reality, the transmit could travel on channel 1 and possibly receive on channel 12. Transmit and receive functions, as well as signaling, are interfaced with the wideband system by a 2W/4W **termination unit. This unit incorporates a hybrid to provide isolation between the transmit and receive functions. Signaling, usually 90V 20 Hz/AC, is converted to a frequency and level capable of transmission over the wideband system.**

The two circuits shown are simplified to give some understanding of an operating system from user to user. Operating circuits may operate this way or may be somewhat more complicated. Both DC and audio circuits may be encrypted. The portion herein pertaining to technical control gives more detailed information about circuit connections and should be consulted for typical circuit routing.

15-3. **Remote Mountain.** Remote Mountain is a baseband repeater station which forwards baseband information between Hidden Valley and Central City. It connects with Central via tropospheric scatter and with Hidden Valley via LOS equipment (figures 1-4 and 1-5). No multiplex equipment is installed at this facility; therefore, all measurements of traffic are made using a FSV. Specific functions at this site will not be explained here.

Figures 15-3a and b detail LOS and tropospheric scatter equipment functions pertaining to Central City. Figure 15-4 shows the major equipment located at

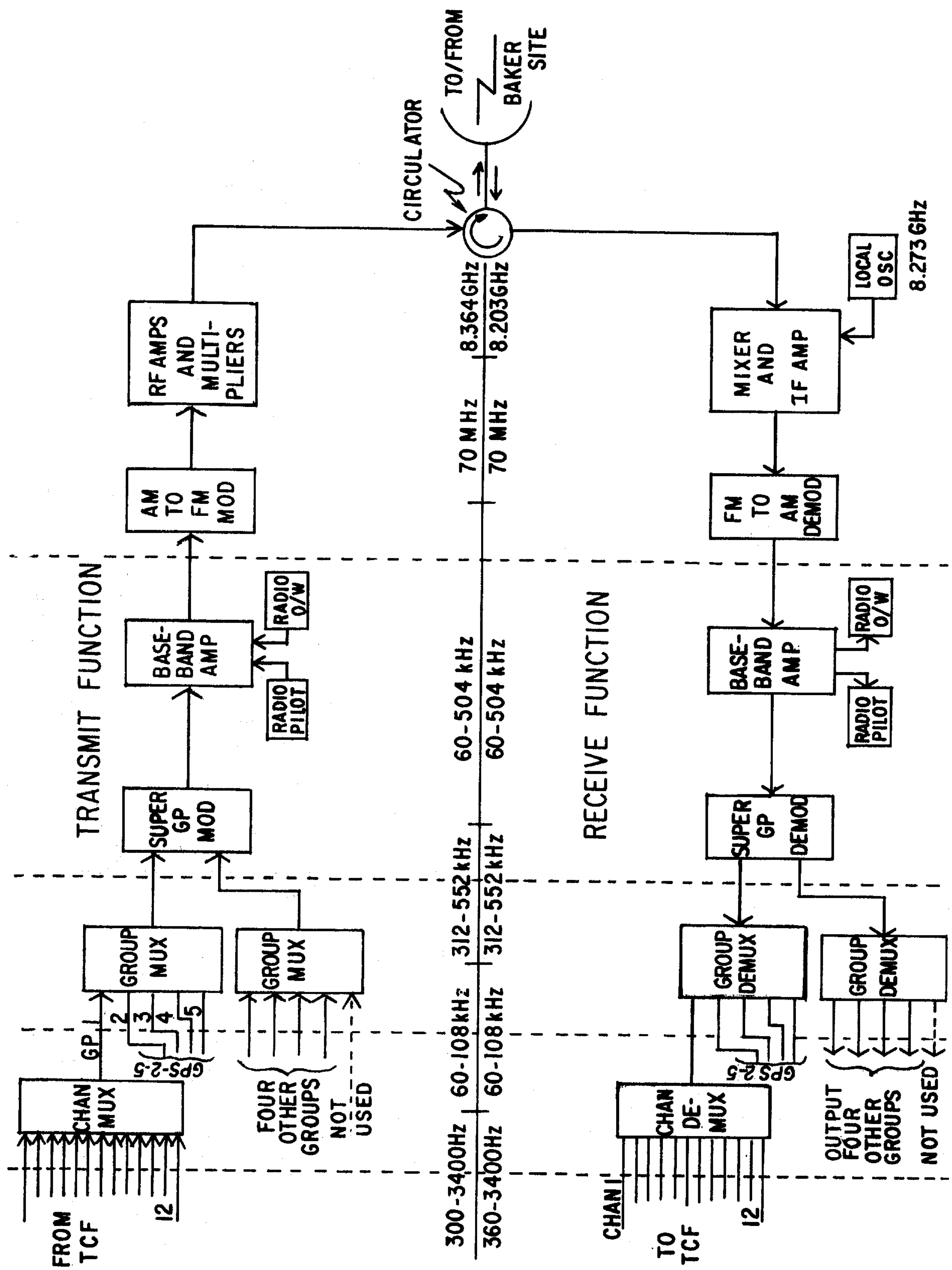


Figure 15-1. Able Site LOS Terminal.

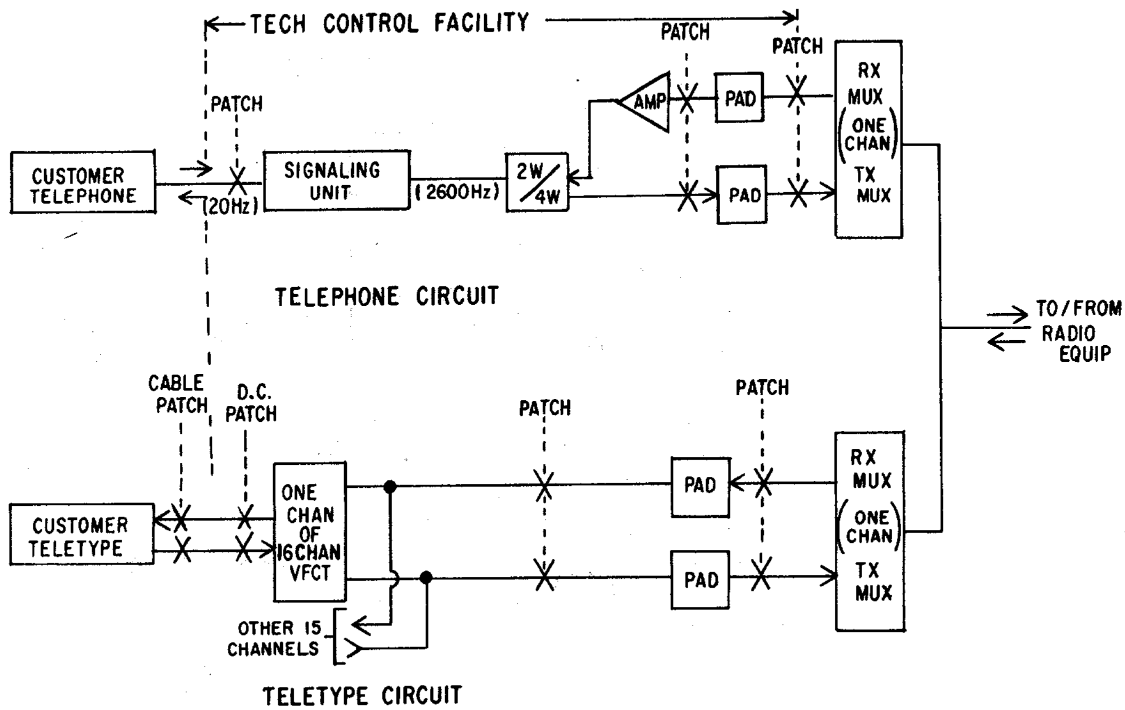


Figure 15-2. Connections Between User and Wideband Equipment.

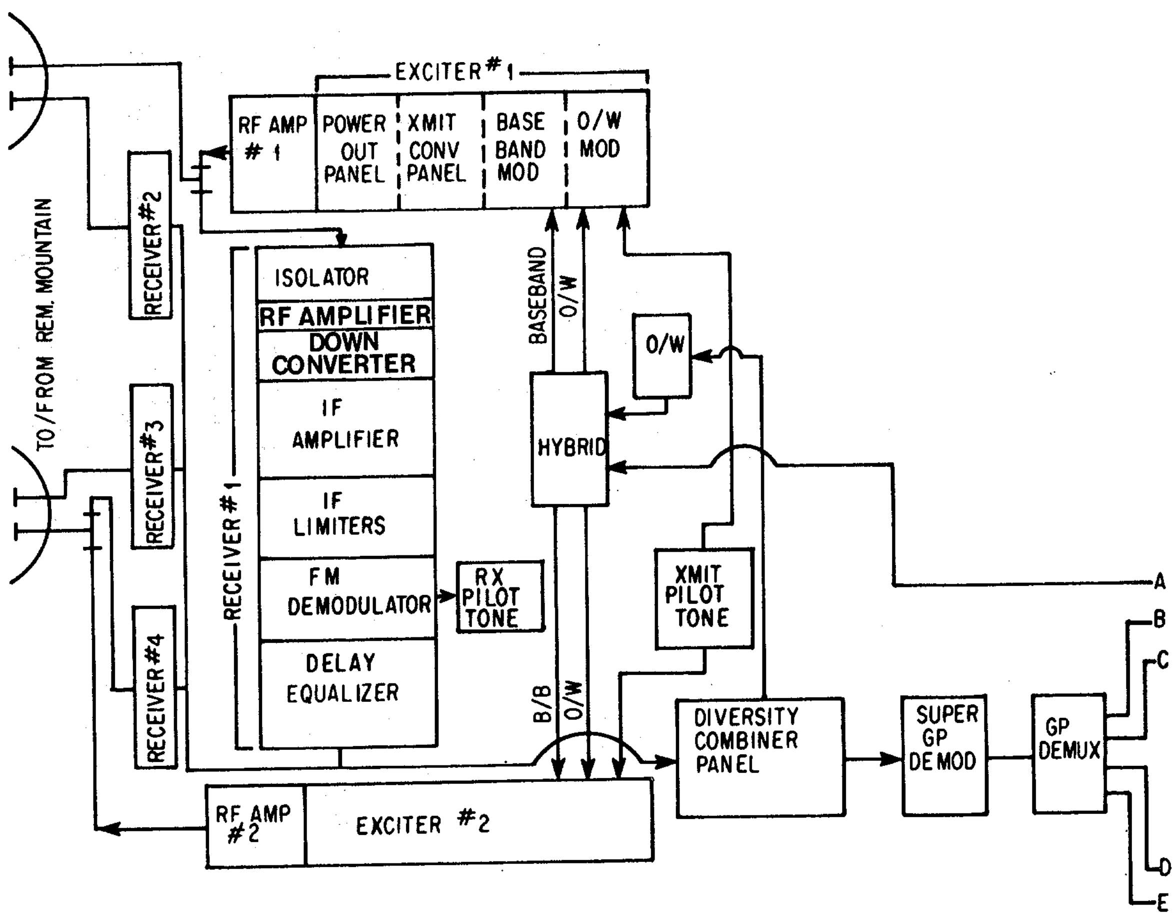


Figure 15-3a. Central City Equipment.

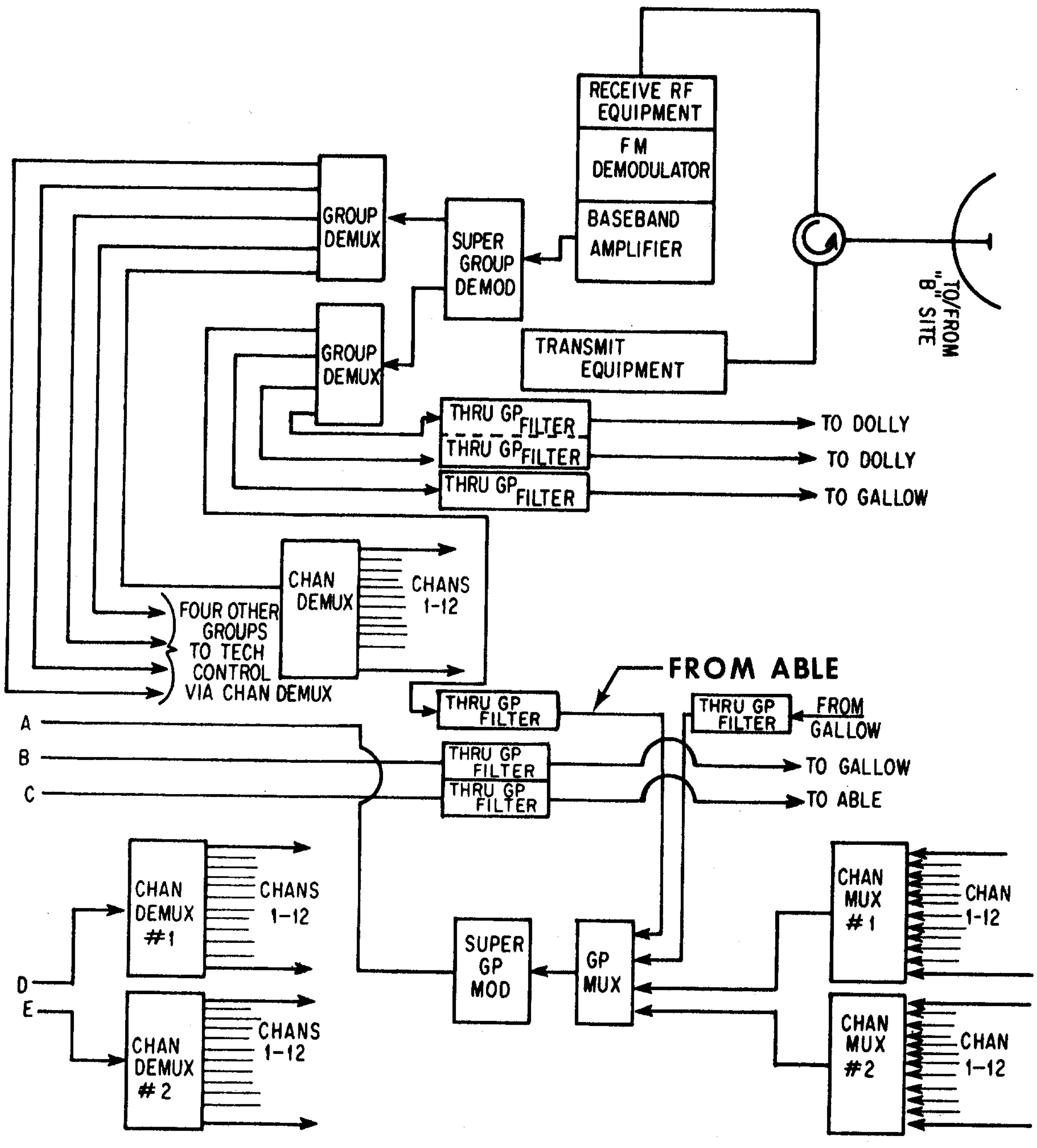


Figure 15-3b. Central City Equipment.

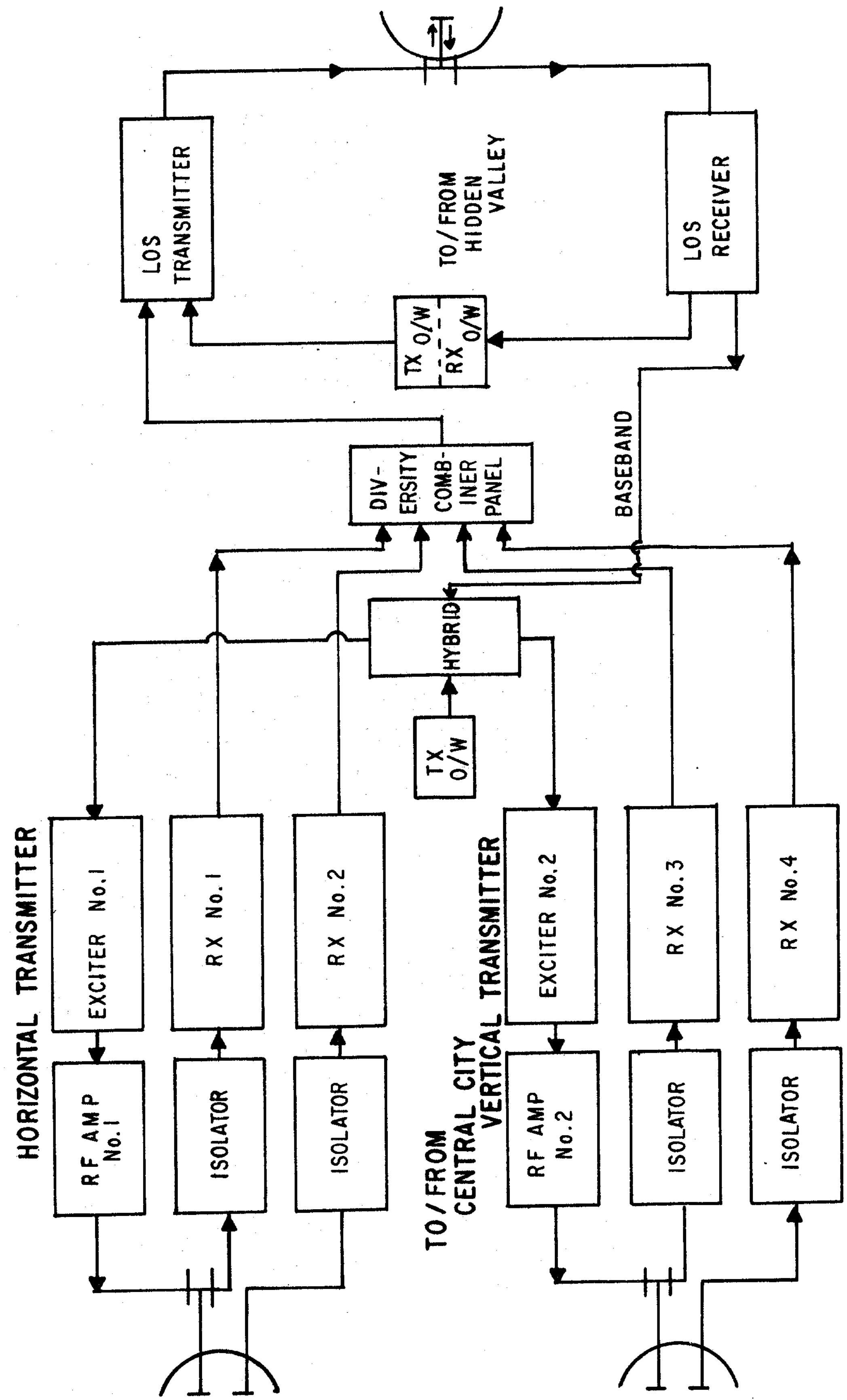


Figure 15-4. Remote Mountain Equipment.

Remote Mountain. No customers are served directly from this location. Patching and monitoring facilities are provided for monitoring switching and substituting receivers in the tropospheric scatter equipment. The baseband from the LOS receiver may also be used to modulate only one exciter while maintenance is performed on the other.

15-4. **Central City.** This station is the hub of our sample communications system and contains a large amount of equipment. Radio links connect Central City with Baker Site, Dolly, Gallow, and Remote Mountain. All links are LOS except for the tropospheric scatter link to Remote Mountain. Our sample system will follow the traffic from Able Site through Central City and out on the link to Remote Mountain. We will also follow the two groups (24 channels) which originate at Central City and pass through Remote Mountain to Hidden Valley.

Figure 15-3b is a simplified block diagram showing the receive LOS function from Baker Site and figure 15-3a shows both transmit and receive functions toward Remote Mountain. The transmit function toward Baker Site will not be considered here since figure 15-1 (pertaining to Able Site) depicts a LOS transmit function.

a. Receive Function from Baker Site. Our intelligence from Baker Site arrives at the parabolic antenna and travels through the waveguide to the mixer assembly where it is mixed with the local oscillator to obtain the 70 MHz IF. The IF is amplified, passed to the FM-to-AM demodulator where the baseband is detected, and sent to the baseband amplifier for amplification. From the baseband amplifier, the baseband is applied to the supergroup demodulator for separation of the supergroups. One complete supergroup is separated into five groups of 12 channels each and terminated in Central City technical control. In the second supergroup, only four groups are used, none of which terminate in Central City technical control. All four groups are "through-grouped" to other stations. Monitoring traffic in these groups at Central City is limited to frequency selective measurements at baseband, supergroup, and group frequencies. Our interest here is to follow the group which originated at Able Site and came to Central City via the IF repeater at Baker Site.

b. Transmit Function to Remote Mountain. As shown in figure 15-3a, the group destined for Hidden Valley, via the baseband repeater station at Remote Mountain, is passed from the "through-group filter" to the group multiplex equipment. The filter ensures that only group frequencies (60-108 kHz) and the proper level are applied to the group multiplex equipment. Amplification or attenuation may take place depending on system design. The group from Able Site is placed on the baseband, along with the two groups originating at Central City and the group from Gallow, by the group and supergroup equipment.

From this point on, a basic difference is noted in the tropospheric scatter equipment when compared to the LOS configuration. To ensure comparable quality to LOS links, the tropo links usually require more diversity. Our mock system employs quadruple diversity (that is, two transmitters, using different frequencies and antenna polarization and four receivers, using frequency, space, and polarization diversity). At least one receiver must always be receiving a signal well above threshold to ensure that high quality circuits are furnished. Now, back to the baseband, which is being transmitted to Remote Mountain.

The output of the supergroup modulator is applied to a hybrid circuit to obtain two outputs for the necessary inputs to each exciter. Since the two paths are identical, only one will be considered here. The baseband is applied through the terminal blocks to the baseband modulator where conversion to an IF takes place (usually 70 MHz). A submultiple frequency from an oscillator is multiplied to a frequency 70 MHz below the frequency of transmission. The 70 MHz modulated signal is then mixed with this carrier frequency using additive mixing to obtain the desired operating frequency. The low level RF passes through the transfer panel to the power output panel for amplification to a value sufficient to drive the 10 KW RF amplifier. This is a linear amplifier providing a nominal 30 dB gain. The output from the amplifier is transmitted through the waveguide and duplexer to the vertical feedhorn on one antenna. The other transmit path is identical except a different frequency is employed and the output **is connected to the horizontal feedhorn of the second antenna.**

c. Receive Function from Remote Mountain. Each receive antenna is connected to two receivers: one for vertical and one for horizontal polarization. One receiver is connected directly to the feedhorn while the other is connected through the duplexer to the feedhorn. Orders of diversity are frequency, space, and polarization. During periods when RSL is sufficient, any single receiver is capable of carrying all traffic. Receiver 1 in figure 15-3a will be used as an example.

The low level RF signal arrives at the antenna and travels through the duplexer to the receiver preselector. The preselector allows only a narrow band of frequencies to pass to the RF amplifier and prevents RF leakage from the duplexer from damaging the RF amplifier. The RF signal is then applied to the down converter for conversion to a 70 MHz IF. After amplification and limiting in the IF assembly, the IF enters the delay equalizer which provides adequate delay to ensure that inputs to all four receiver combiners arrive at the same time. The baseband is then detected in the demodulator panel and provides one of four inputs to the diversity combiner.

The diversity combiner panel provides the best possible output to the multiplex equipment. The input from one receiver, or a combination of receivers, depends on RSL out-of-band noise and pilot tone reception which determines selection of the output. After the supergroup is placed in the proper frequency spectrum by the supergroup demodulator, it enters the group demultiplex equipment and is reduced to four groups. One group **is** through-group filtered to the transmit group multiplex equipment of Baker Site, the other is through-grouped to Gallow. The other two groups enter 12-channel demultiplexers where they are changed to audio and **forwarded to technical control.**

d. Tech Control. We have thus far built an example system with a number of links having the required equipment and personnel to provide the necessary communications capacity. Next, we must provide for the operation and management of the system. This has been accomplished by providing two TCFs and three PTFs. A TCF was installed at Able to interface the large number of subscribers with the system. Another larger TCF was installed at Central City to interface the various links in the system with one another and with the outside world. PTFs are installed at Gallow, Dolly, and Hidden Valley to provide interface with the subscribers at each location. These PTFs operate under the technical direction of the Central City TCF. The configuration of the Central City TCF is shown in figure 15-5. The number of active jack sets on each patch bay is displayed, indicating the interface of 64-wire circuits to the outside world. The TCF itself is among the "in-house" subscribers having a voice order wire on one channel of each of the links as well as voice order wires to the outside world. These voice order wires are used to coordinate quality control, circuit reroute and restoral, and management and reporting functions.

15-5. Conclusion. This chapter has presented a detailed view of system operation. With this overview in mind, the following chapters introduce the concept of system evaluation, as well as the philosophy and procedures for a system quality control program.

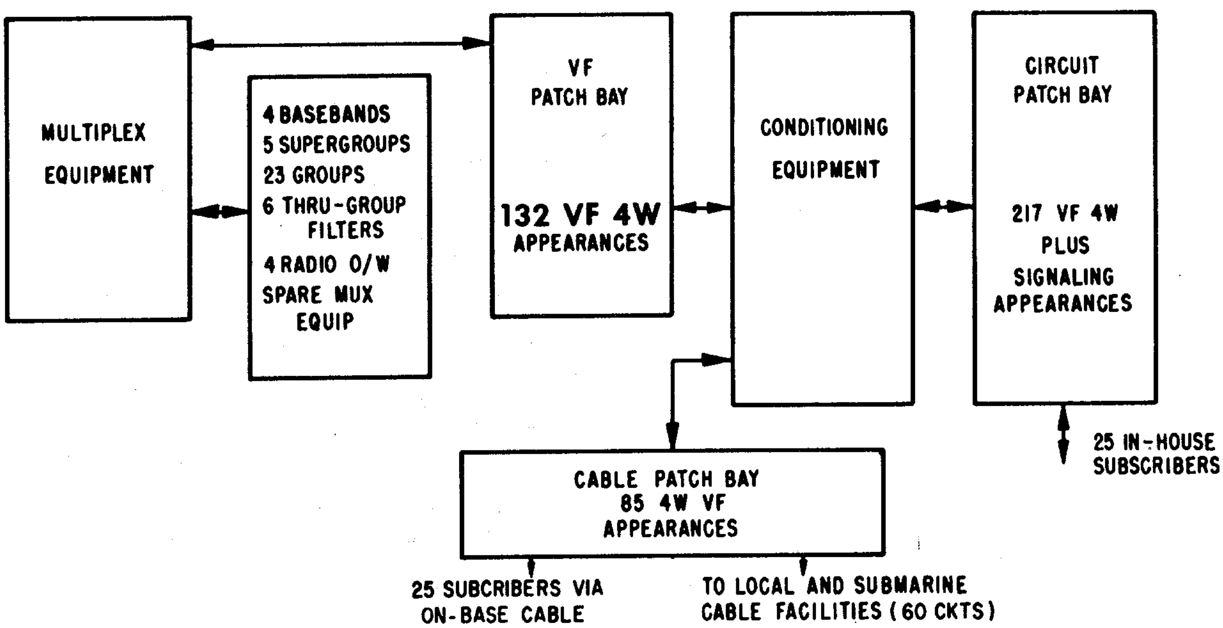


Figure 15-5. Central City Technical Control Facility.

Chapter 16
SYSTEM PERFORMANCE MONITORING

16-1. General. In the communications field, DCA, HQ AFCS, Area, group, and squadron inspectors and evaluators are working at the job of assessing some part of our performance. The purpose of any test is to evaluate performance and to answer the question, "How good is it?" The DCS wideband system is certainly no exception. With switched systems and more critical data performance standards, it becomes increasingly important to know how the wideband system and its component parts are performing.

16-2. Concept of System Evaluation. At one time, the system and subsystem maintenance concept erroneously theorized that if each individual equipment or "black boxes" satisfied their performance requirements, then the system performance requirements would also be automatically satisfied. As more complex systems are introduced into the military environment, the fallacy of this concept becomes increasingly evident. The objectives of a system evaluation are two-fold. First, it must provide on-site measurement and analysis of specified parameters which will enable local detection and correction of degraded conditions before customer service is interrupted. Second, it must provide a data base for system performance which can be used in developing system standards. This data will provide a means for trend analysis to detect gradual system degradation. The value of system evaluation, as a management tool and trend-detection device, rests entirely on the competence and integrity of on-site personnel who perform day-to-day measurements and report the information up-channel. There are two programs that provide the technical understanding to accomplish these objectives. The first program is through the DCA Technical Evaluation Program (TEP). In support of DOD, DCA, and USAF directives, an AFCS technical visits program has evolved. It is referred to by its code name, "Scope Creek." This program has fostered a true technical capability to arrive on a station and measure, on a link-by-link basis, how the system is performing, where it is weak, and what type of quality it contributes to the total system. The data from these tests are distributed to site personnel and to appropriate headquarters agencies, including DCA, with the intent of characterizing the link, providing a basis for predicting performance over tandem links, and indicating areas where improvements can be most readily realized. All parameters of operation are evaluated by the Scope Creek team and nothing is derived from secondary sources if it is possible to measure directly. The findings show how a link performs while the team is on-site. The majority of the tests can be duplicated by station personnel which provides the second program we are interested in - the Performance Monitoring Program (PMP). The PMP is the vehicle for the actual accomplishment of the objectives as stated above. AFCSR 100-67 sets forth policy and assigns responsibilities for administering this program.

The TEP and PMP are closely related in that the TEP determines optimum system performance while the PMP monitors actual performance to determine if a system is performing as expected. The concept of system evaluation, then, is to measure parameters which indicate how well the link(s) measured is performing as part of the system seen by the subscriber.

16-3. Technical Control, The Key to System Monitoring. The TCF and PTF have been discussed in detail. The TCF is the focal point for system performance monitoring. When day-to-day system evaluation is required, the TCF must supply the information. Detection of degrading performance and direction of fault isolation are easiest at this point; the alternative is to wait for subscriber complaint which is a very poor management approach for a system as sophisticated as ours. This recognition of the TCF in no way lessens the requirements for maintenance. Once degradation is identified, the effort to restore acceptable performance (hopefully before the subscriber is aware of trouble) will require the maintainer to know and understand thoroughly the principles outlined herein, along with the peculiarities of his/her equipment.

16-4. Performance Standards. These apply to everything from small components of the DCS to major systems. Standards reached at national and international levels address some of the most basic requirements in communications. In order to work together, frequencies, impedances, power levels, multiplexing schemes, and signaling conventions must be common throughout the system. Standards of this type aid design efforts and may or may not affect performance. For example, an outstanding multiplex might be designed with 432 ohm input impedance and 5 kHz VF channels. Although it would work well, it would be nonstandard.

Another area of standards is based on current technology. These standards state how well the item performs. Things like multiplex ICN in loop-back, frequency response, delay distortion limits, and crosstalk would fall in this area. This type of standard will cause the equipment manufacturers to design their components to a given level of performance. In turn, this will be reflected in the maintenance technical orders (TOs) used by the field to check equipment operation.

a. Established Standards. For our purposes in wideband systems, several military standards can be cited. Their relationship to each other is complex. The source of criteria, or the precedence which sets the basis for new standards, may be generated by industrial, military, or foreign practices; from here, the practice is reflected in one or more of the sources of standards. Recognizing that the TOs reflect the accepted standards at the time of the equipment design or modification, sources of standards are outlined in MIL

STD 188(C); DCAC 300-175-9; DCAC 310-70-1, vol II; and AFCSR 100-29, vols I, II, and III. These sources can be summarized as follows:

(1) MIL STD 188(C), published by the Department of Defense, prescribes technical design standards for tactical military communications. It sets forth presently attainable standards and design objectives which will be attainable in the future. It applies to new systems being designed for military procurement and to major modifications of existing equipment. It is relatively small and contains tables, graphs, and **general information which is useful to the technician.**

(2) DCAC 300-175-9 is a distillation of standards shown in DCAC 310-70-1, vol II. Interpretation of these standards in light of "Scope Creek" testing is provided. This small publication addresses the standards problem which is closest to the personnel who work at the wideband sites, namely, "How good should the individual circuits which comprise a particular link be?" Standards are given in the TO for the radio, multiplex, and antenna, but there isn't a TO for the link between sites A and B; also, some components may be newer than others, which makes a flat statement about performance more difficult.

(3) DCAC 310-70-1, vol II, translates the information contained in other publications into working standards which govern end-to-end circuit performance. The tech controller refers to it to determine whether the measured performance is within tolerance. This volume, rather large in itself, is part of four volumes which comprise the governing regulation on the operation of the DCS.

AFCSR 100-29, vols I, II, and III, outline performance assessment standards that apply to tactical communications systems. They incorporate the concepts of Scope Creek and the PMP program but are specifically geared to the operational life cycle of tactical deployments (planning, engineering, equipment readiness, deployment/employment, post-deployment).

b. Other Sources of Standards. Guidelines are given for "6000 nautical mile circuits." New end items of equipment require certain minimums in order to function properly, but this doesn't help to optimize specific links; therefore, it is necessary to establish standards which can be related directly to the operation of the equipment on links A to B. This performance can then be interpolated to provide an estimate of the performance on a tandem series of links A through D. In order to derive this type of information, it is necessary to ensure that all components meet TO specifications; then key performance indicators (such as VF level, ICN, IPN, and peak digital distortion) must be measured. This must then be correlated with RSL, system loading, and any other uncontrolled variables. The data contained in the original test and acceptance documents is one source of **site standards. Results of "Scope Creek" or** Programmed Depot Maintenance (PDM) visits are another source. By coupling these known attainable values with readings observed by site personnel, definite standards can be established. It should be emphasized that when local standards are established, care must be taken to ensure proper maintenance is performed.

16-5. Tests for System Evaluation. In the following chapters of this publication we will address the tests which permit complete system evaluation. These tests are outlined in three groups: Key Performance Indicators, Subsystem Performance Tests, and Special Evaluation Tests. Although no distinction is made as to who performs each test, the tech control is most likely to monitor key indicators under provisions of the quality control program. Maintenance personnel must understand the key tests and subsystem tests. The discussion of special tests is aimed at understanding performance measurements not mormally performed on a routine basis. Each test explains the general scope, specific procedure, and analysis of results. A discussion of measurement equipment, techniques, and errors precedes the test chapters and a logic system for isolating wideband troubles is presented. The tests are not new; many of the measurements are routinely described in TOs and operating procedures. The descriptions avoid the detailed approach required in TOs while analyzing more completely the meaning of the results. Representative values of measured parameters are quoted to give a feel for good and bad readings.

Chapter 17
BASIC MEASUREMENT TECHNIQUES

17-1. General. This chapter describes basic measurement techniques which can be accomplished through the use of common types of test equipment. An attempt is made to answer some of the questions concerning readings obtained, their accuracy, and values indicated on certain test instruments. Before using any test equipment, an **understanding of its function is most** important. In Air Force management, the phrase "The Right Man for the Right Job" is a commandent. In electronics maintenance or control, the phrase might be "The Right Instrument for the Right Job." The intent of this chapter is to help you choose the right instrument and apply it skillfully on the job.

17-2. Meaning of Measured Values. The efficient operation and maintenance of a communications system requires frequent AC voltage and signal level measurements. Usually, a specified signal is measured in specified units with a specified instrument and compared to a specified criteria. The reason that a specific test instrument is often required to measure a common parameter such as "RMS volts" is because different types of instruments employ different circuitry to measure the same parameter.

a. AC Voltmeters. The three basic types of AC voltmeters which are used in the field today are average responding, peak responding, and true RMS. All three are usually calibrated in peak and RMS volts and often in dB.

(1) Average Responding (AR). This type of AC voltmeter is one of the more common types used. Its basic operating principles are shown in figure 17-1. The AR voltmeter responds to the average value of the input signal, which is seldom the required quantity. Since sinusoidal waves are the most frequently measured waveform, it is a simple matter to calibrate the face of the meter in RMS volts. The average value of a sine wave is 0.636 x peak voltage and the RMS value is 0.707 x peak voltage; therefore, the average value of a sine wave x $\frac{.707}{.636}$ is the RMS value (1.11 x average). With this mathematical relationship known, the meter may be calibrated in both peak and RMS volts, although in reality it measures the average voltage. The frequency response of this type of instrument is often limited to 1 or 2 MHz. Error in the reading, due to harmonic distortion, is low. The Hewlett-Packard 400 series AC voltmeter, the Model 403B voltmeter (used in the HP3550 Transmission Measuring Set), and the 330 series Distortion Analyzer all employ average responding voltmeter circuitry.

(2) Peak Responding (PR). The peak **responding type AC voltmeter is also commonly used.** Figure 17-2 shows the basic circuitry employed.

The capacitor in the input circuit is charged to the peak value of the input signal and this peak value is passed first to a directly-coupled amplifier, then to the meter. The instrument measures the actual peaks of the signal and this is presented on a peak scale. The reading is factored by .707 to provide an RMS equivalent. While the average responding meter will not indicate any impulses, transients, or harmonics, the peak responding type will indicate impulse, transients, and out-of-phase harmonic products. Neither type is completely accurate in its RMS indication of anything other than a pure sine wave. The peak responding type has a very wide frequency response, often to several hundred megahertz. Care must be exercised when using an instrument of this type to measure in RMS units since noise, transients, or harmonics may cause an appreciable error. The Hewlett-Packard Model 410C multimeter is one example of a peak responding AC voltmeter.

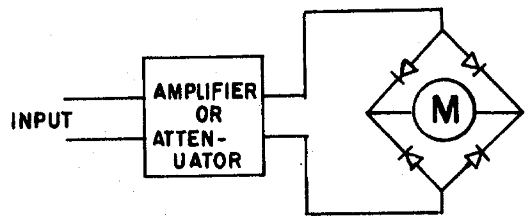


Figure 17-1. Average Responding AC Voltmeter.

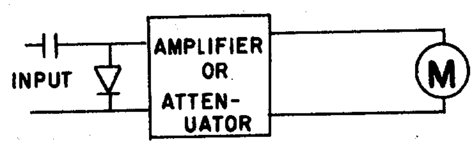


Figure 17-2. Peak Responding AC Voltmeter.

(3) True RMS. The true RMS voltmeter provides a genuine measurement of the equivalent heating power of a waveform, the actual definition of RMS. Rather than using a mathematical technique (the RMS value is obtained by summing the squares of the individual components of the waveform and taking the square root), it actually measures the heating effect. The true RMS voltmeter employs a thermocouple which heats in proportion to the incoming waveform and produces a DC output proportional to the amount of heat generated. A thermocouple is inherently non-linear but this has been overcome by using a second, matched, inversely-coupled thermocouple to compensate for non-linearities of the first. Figure 17-3 shows the basic circuitry. This type of instrument will not show any indication (other than a small change in heating effect) of peaks, transients, or spikes. For this reason, the meter is usually not provided with a peak voltage scale. The response time of a true RMS meter will seem sluggish and the bandwidth is moderate, often 10 MHz. The Hewlett-Packard Model 3400A is one example of a true RMS voltmeter which is necessary for accurate measurement of complex waveforms (such as white noise).

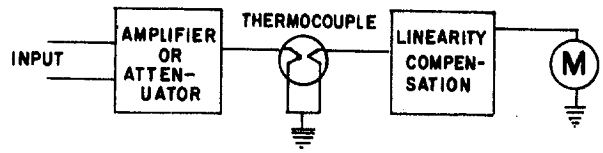


Figure 17-3. True RMS AC Voltmeter.

b. dB Meters. Level measuring sets, or dB meters, are actually AC voltmeters. These meters have a dB scale printed on the face to represent the ratio of the voltages squared. To perform level measurements in dBm with these instruments, we must first consider what we are attempting to do. Level measurements in dBm are actually power measurements. Power is a product of voltage and current. We cannot simply measure voltage and call it power without some qualifications. Ohm's law tells us that power is equal to the voltage squared divided by the resistance. If we *know* the resistance and measure the voltage, we can calculate the power. To avoid this calculation, most AC voltmeters are calibrated in dBms, with the calculations already having been performed using an *assumed* impedance (resistance). As long as the terminal impedance of the circuit under test is equal to the one with which the meter was calibrated and the circuit is properly terminated into that impedance, the meter may be used to measure power in dBm. Level measurements should not be attempted until the standard for the meter (usually printed on the face) and the impedance at the test point have been found and compared.

When a meter with a dB scale calibrated against the desired impedance is not available, a conversion chart (tables 17-1 and 17-2) may be employed to convert voltage into dBm for common impedances. When using a chart of this type, it must be remembered that the circuit must be terminated into its characteristic im**pedance (either its normal load or a resistive termination). The voltage measurements must then be performed with a high input impedance instrument bridging the load, as presented by the normal circuit or a termination.**

Level readings are based on RMS values of the signal, so the inherent advantages and disadvantages of the three types of voltmeters discussed earlier apply equally to level measurements.

c. Noise Weighting Networks. Noise weighting networks are another important consideration when performing measurements of noise. If at all possible, the measurement should be performed with the weighting network specified in the estabished criteria. Use of the wrong network can cause readings to be in error by as much as 15 or 20 dB and a serious fault could be masked. For example, when using a "C-Message" weighting network to perform a measurement, then converting to "Flat-3 kHz" using a table, **the technician might overlook a serious carrier leak or power supply ripple problem in the channel.**

d. DC Measurements. Measurement of DC voltage and current is straightforward and, if established practices are observed, no problems should be encountered in applying the measurement to the published standards; however, care should be exercised when measuring DC voltage that the meter's sensitivity, in ohms per volt, is at least that specified in order to avoid circuit loading.

e. Scale Selection. Accuracy of measurements is a **point of pride among technical people. Our instruments are among the most accurate and are calibrated regularly. Mirrored scales allow for precise correction of parallax by centering the observer in front of the in**strument. But is it on the right range? Just because the needle is pointing to a number and is not "pegged" does not necessarily mean that the instrument is on the proper range. Most voltmeters have ranges that will permit many voltages to be measured on either of two ranges. On one, the needle will indicate on the upper third of the scale and, on the other range, it will appear on the lower third of the scale. If it indicates on the center third of the scale, then only one range can be used. The accuracy of the reading is affected directly, on most voltmeters, by the position of the needle on the scale. Figure 17-4 depicts the percentage of uncertainty of your reading with respect to the position of the needle. The closer to full scale, the more accurate the reading. The grey area indicates the part of the scale on which a different range should be used to perform the same measurement so that it will appear on the upper third of the scale.

17-3. Errors in Measurement. The purpose of this paragraph is to point out measurement errors often made by technicians while performing tests and measurements on actual circuits and equipments. Experience gained through error can be an extremely expensive teacher and perhaps, through the sharing of some of this experience, the cost of our systems can be reduced. Many of the measurement errors being committed are often the result of misconceptions, misinterpretations, and carelessness and may apply to experienced as well as inexperienced technicians and engineers. The earlier sections of this text have stressed many points around which misunderstandings have caused errors.

Carelessness cannot be cured with a book. The following topics are presented in hopes that they may be of some value in pointing out specific areas where many of the errors occur.

a. Test Level Points (TLP). This term, preceded by a number, expresses in dB the net amount of gain or attenuation between that point and a zero reference level. Errors often occur through considering a TLP as **an actual signal level. They are the same only if the** signal is inserted into the system at 0 dBm0. Other errors occur in subtracting the TLP from the signal level rather than adding. A -10 dBm0 signal measured at a -2 dBm TLP is -12 dBm. As another example, a +5 dBm0 signal measured at a +7 dBm is +12 dBm. In both examples, algebraic addition was used to obtain the actual measured level (the level that would be measured by the level measuring set). Other errors **occur through stating the measured signal level in dBm (actual measured level) rather than in dBm0 (level referenced to TLP).**

b. Measuring Impedance With Audio Oscillators. Many test oscillators (such as the HP-200 series) have

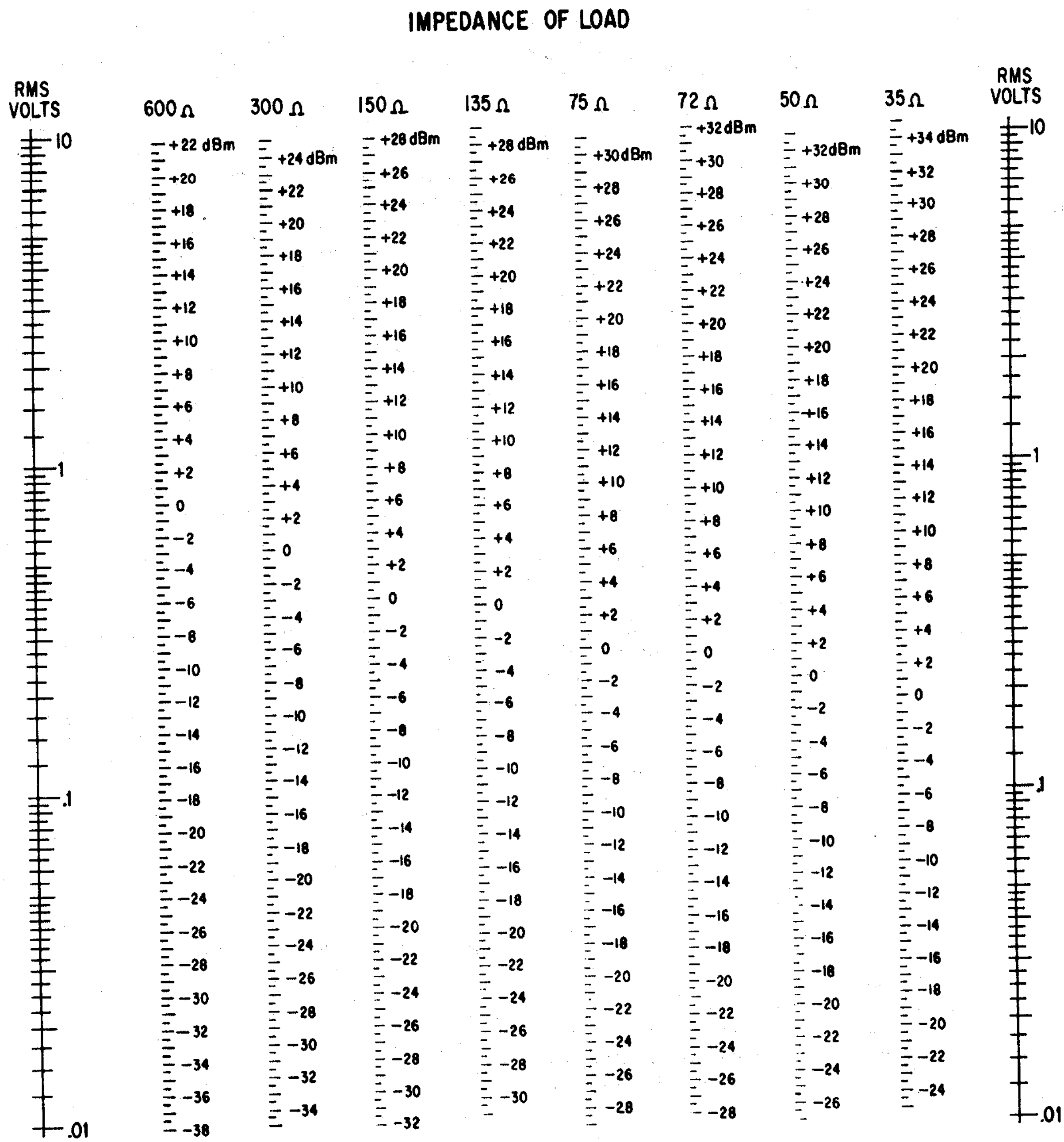


Table 17-1. Voltage to dBm Conversion Scale.

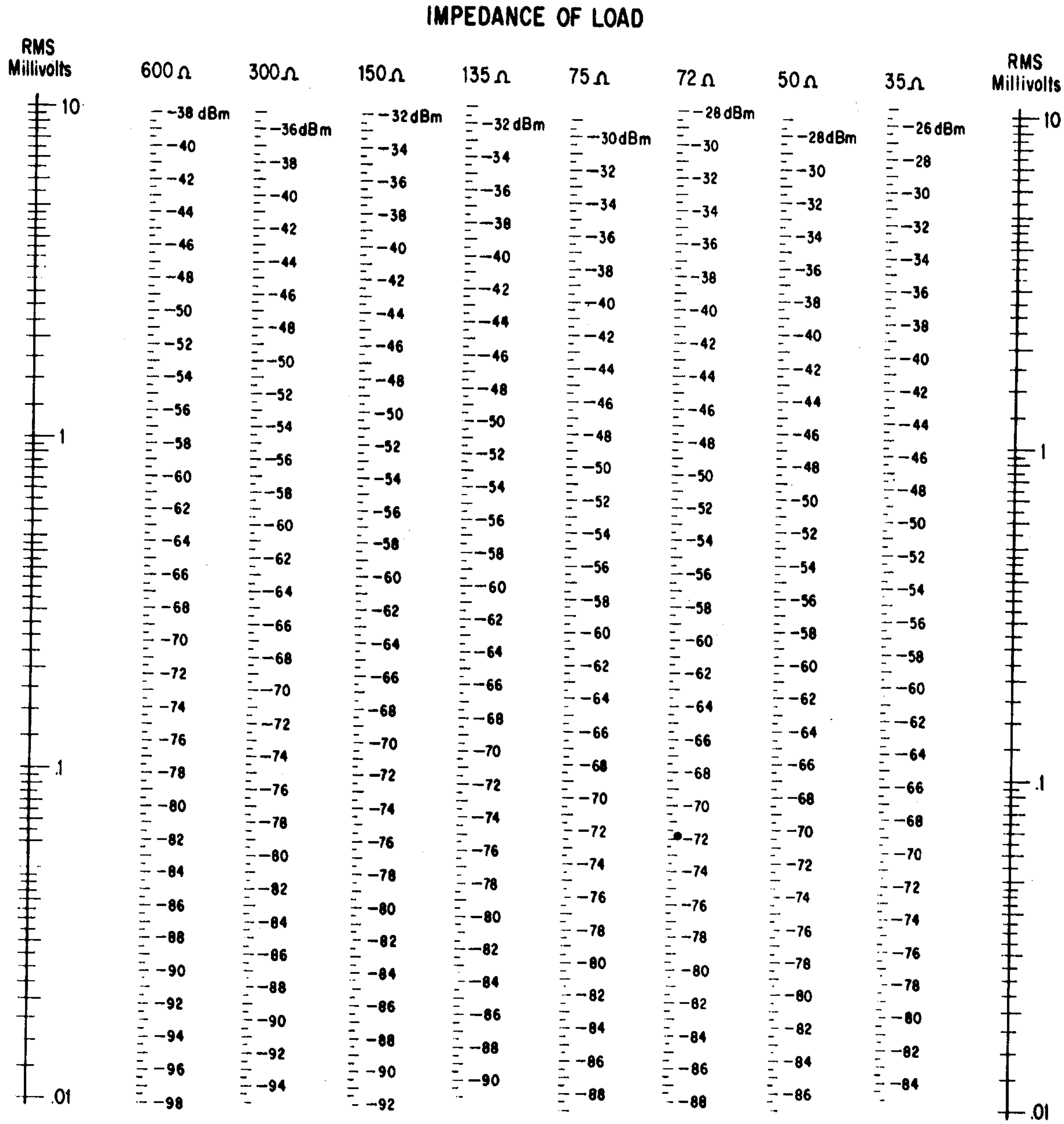


Table 17-2. Millivolts to dBm Conversion Scale.

internal compensation networks to provide a constant output level regardless of load changes. Using an instrument of this type to perform terminal impedance measurements will completely invalidate the measurement. To determine whether an oscillator has this output compensating feature, first measure the output voltage; then, place a resistor of any value across the output and again measure the voltage output. If the reading is the same, the feature is incorporated in the oscillator.

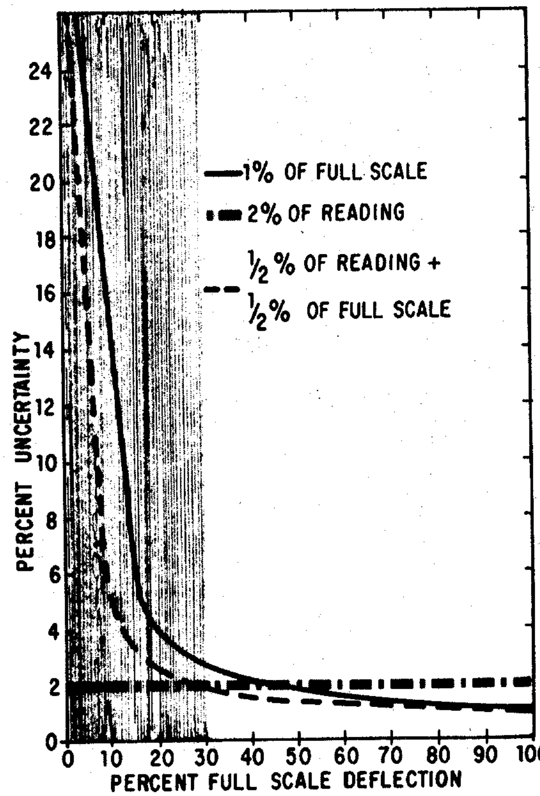


Figure 17-4. Percent Uncertainty for Three Methods of Specifying Accuracy.

c. Using Voltage Scale to Obtain dB Reading. Voltmeters with dB scales calibrated against a standard impedance (such as 600 ohms) are sometimes employed to measure levels on circuits with a different characteristic impedance (such as 150 ohms). The measurement will be in error unless corrections are made. If the voltmeter does not have a dB scale calibrated against the impedance of the circuit you are measuring, measure the voltage and convert to dBm.

d. Providing Proper Termination Prior to Measuring. Circuits are often not terminated into the proper impedance when measurements are being performed. This is sometimes caused through carelessness and sometimes because the front panel markings of an instrument mislead the technician. For example, when you depress the "600 ohm" button on the 1932A distortion analyzer, the input is high impedance, not 600 ohms.

e. Grounding Properly. Many test instruments must be provided with an "earth" ground to provide accurate measurements. Simply rack-mounting an instrument or using a three-conductor AC power cord may not be adequate to prevent introducing stray potentials in the reading.

f. Balanced Versus Unbalanced. Measurements are often performed at both balanced and unbalanced points on equipment and circuits. Test instruments are often provided with both balanced and unbalanced inputs. Failure to match the test instrument to the test point can cause erroneous measurements and may disrupt an entire system.

g. Impedance Matching, Isolation, and Balun Transformers. These are frequently misused. Technicians often do not know the difference between them and use them interchangeably.

(1) Impedance Matching Transformers. These are designed to interconnect devices of different impedance. They ensure proper impedance reflection in both directions. A typical application is the matching of a 75 ohm generator to a 150 ohm line. Some loss may occur in the transformer; however, linear transfer of power may be expected over the design frequency range of the unit. Impedance matching transformers may also provide balanced to unbalanced terminations. As an example, a 50 ohm unbalanced generator may be matched to a 300 ohm balanced line by an impedance matching transformer.

(2) Isolation Transformers. These are designed to isolate one unit or stage from another. They are normally one-to-one ratio (1:1) and match like impedances while providing decoupling or DC isolation between units or stages.

(3) Balun Transformers. The balun transformer is designed specifically to provide interconnection between balanced and unbalanced equipment. It employs a coil which is isolated from ground on one side and a coil which has one terminal connected to ground on the other side. On the balanced side, signal current travels in a closed loop, ideally with no current to ground. On the unbalanced line, intelligence is carried on one wire with the ground providing the return path. The HP-11004A is a balun transformer which converts a single (unbalanced) input to a balanced 135 ohm or 600 ohm output. It passes frequencies from 5 kHz to 600 kHz. The HP-11005A is also a 1:1 ratio balun transformer differing from the HP-11004A in that it provides a 600 ohm resistor which may be switched across the unbalanced side. This permits a 600 ohm line to be terminated in its characteristic impedance while feeding a high impedance instrument. With the 600 ohm resistor switched out, the line side will reflect the impedance of the test instrument. A 1:1 ratio transformer (such as the HP-11005A) *will not* provide a high impedance on the secondary (line) side if the test instrument has a low impedance.

h. Harmonic Distortion in Audio Oscillators. Audio oscillators should be checked periodically for harmonic content in the output signal. Harmonic distortion tests on circuits can become frustrating if the send oscillator has a high harmonic content on the output. Oscillators commonly found in the HP-3550 transmission measuring set (types HP-209A, 204C, 204D) should be operated in the low distortion mode when used for harmonic distortion measurements.

i. Assuming Termination is 600 Ohms Impedance. The terminal impedance of both test instruments and conditioning equipment (amplifiers, pads, hybrids, etc.) should be known. The assumption that they are exactly 600 ohms is not always correct.

j. Measuring an Open Line. In some instances, customers complain about their circuit and promptly disconnect the receive terminal equipment from the line. Then the trouble report reaches the technical control, the controller bridges the traffic channel with a high impedance instrument and notices very high levels. Much time may be lost before discovering that the receive terminal equipment is off-line and the voice channel is not terminated. If the circuit is logged out and levels are checked, the channel should be terminated in the technical control.

k. Proper Termination When Adjusting Frequency. When the frequency of an oscillator is adjusted for a frequency translation test, the oscillator is sometimes patched to the frequency counter, adjusted, and then patched into the line. This will often cause the oscillator to change frequency if the impedance of the frequency counter is different from that of the line. For consistently accurate measurements, always terminate the oscillator into the line and bridge it with a frequency counter.

l. Impedance of Patch Monitor Jacks. Monitor jacks on audio patch bays are normally used for high impedance monitoring. They are not, however, a high impedance source. The monitor jacks are in parallel with the line or equipment jacks and can, if necessary, be employed for circuit patching. Any low impedance (terminating or 600 ohm) devices patched into the monitor jacks will load the circuit and decrease the level. Terminating instruments should never be patched into patch monitor jacks.

m. Bridging Balanced Lines. Bridging a balance circuit with a high impedance unbalanced test instrument will cause a decrease in signal level on the circuit. This is not the only problem! Grounding one side of a balanced circuit also introduces a high degree of noise into the circuit, usually rendering it useless for communications. If you are not positive that your instrument is balanced, use an isolation or balun transformer.

n. Interpreting Frequency Response Criteria. DCA frequency response criteria is usually stated as a range of losses, for example, +3 to -1. This indicates 3 dB more loss allowable to the relative 1 kHz reference and 1 dB less loss than at the 1 kHz reference. It does not indicate circuit gains and losses. As an example, a level is measured at 1 kHz and found to be -10 dBm. The lowest level was found to be -12.5 dBm at 400 Hz and the highest level -9.2 dBm at 2800 Hz. Circuit loss is then determined to be +2.5 dB at 400 Hz and -.8 dB at 2800 Hz, relative to the 1 kHz loss. This concept is quite simple once the techician is familiar with how the information is obtained. Criteria for some equipment may be stated in gains and losses, while others may be in losses only. Determine which basis is being used before applying the criteria.

o. Monitoring Speech Channels. Many quality control programs ensure close level tolerance on all digital data and tone-on-while-idle circuits. Speech circuits are often neglected or treated on a catch-as-catch-can basis. There are two methods which may be employed to measure user speech levels. One is for the TCF to bridge the circuit with compatible ringing equipment, ring-down one user, and request that he/she conduct a circuit check with the other user. Levels of speech and signaling may be measured in both directions. An alternate method is to bridge the circuit with a strip chart recorder and record the levels on the circuit (at a low chart speed) until a representative sample of levels on the circuit has been logged.

17-4. Frequency Selective Measurements:

a. General. It is necessary, on many occasions, to measure the level of a pilot or channel within a baseband. A standard level measuring set would be of no help, for it would measure only the composite level of the entire baseband (and then only if it had a sufficiently broad bandpass). To measure levels within a multiplexed system, it is necessary to employ an instrument which will select one specific segment of the frequency spectrum. An instrument of this type is called a FSV or selective level measuring set. There are two basic types of FSVs, one designed for wideband systems measurements and the second type for measurements within the audio frequency spectrum.

b. FSV. Figure 17-5, a simplified block diagram of a FSV, shows clearly that this instrument is actually a high quality radio receiver. This type of instrument may be bridged across a baseband and the level on any 3.1 kHz segment (VF channel) can be measured. It can also be bridged across a supergroup or group and select any channel. RF receiver quieting curves can also be plotted with this instrument. The FSV also has a narrow passband of 240 Hz which is employed for measuring pilot frequency levels or other discrete tone levels. The FSV offers a variety of input impedances and a choice of terminating or bridging. Together with using great care in the selection of the proper impedance, it must be remembered that all input selections may be unbalanced on some instruments. To perform measurements on a balanced circuit with an unbalanced instrument, a balun transformer must be used to prevent disrupting the signal under test and getting a false reading.

c. Spectrum Display Unit. A companion spectrum display unit is available for operation with FSVs such as the Sierra 128A. The spectrum display unit provides a visual representation of a segment of the frequency spectrum. This instrument is similar in function to an oscilloscope with one major difference: while an oscilloscope presents amplitude (vertical) versus *time* (horizontal), the spectrum display unit presents amplitude (vertical) versus *frequency* (horizontal). Figure 17-6 shows its basic operating principles. Instead of the entire input spectrum being fed to the vertical amplifier as in an oscilloscope, a segment of the input spectrum is swept across a very narrow filter (35 Hz). The output of the filter represents the instantaneous amplitude at the frequency segment being swept past the filter at that instant. The frequency of the segment of the entire spectrum (from the first IF of the FSV) to be swept is determined by another input from the incremental oscillator of the FSV. The bandwidth of the segment to be swept is determined by the voltage-controlled oscillator. With the Sierra 360, options of 0.3 kHz/Div (3.6 kHz displayed), 1 kHz/Div (12 kHz displayed), and 10 kHz/Div (120 kHz displayed) may be selected. The sweep generator simultaneously controls the sweep rate and the rate at which the voltage-controlled oscillator sweeps.

The spectrum display unit is a valuable addition to a FSV. You can literally "see", the segment of the spectrum you have selected with the FSV. The primary value of being able to see a channel is being able to identify the type of signal on the channel and to immediately determine the distribution of the signal or noise levels.

d. Spectrum Analyzer. Another instrument coming into more popular use is the spectrum analyzer. Many operate on the exact same principle as the FSV-spectrum display unit just discussed. Some operate in the VF spectrum and some newer models operate in the RF spectrum up to many gigahertz. This type of instrument should not be confused with early model spectrum analyzers providing a meter readout, as the early models could be identical in function to the distortion analyzer, or wave analyzer, which is discussed below.

e. Wave Analyzer. Audio or low FSVs or wave analyzers are another very useful type of test instrument. They are very similar in function to the FSV discussed earlier. This unit, however, permits analysis of a 10 Hz or 100 Hz segment in the typical 20 Hz to 100 kHz range. Figure 17-7, a simplified block diagram of a low FSV, shows the basic operating principles of this instrument. The purpose of modulating to a higher frequency (205 kHz in figure 17-7) is to permit the use of a small, fixed, very narrow bandwidth bandpass filter. An instrument using a tunable filter and no modulator would not be practical because of the low frequencies involved. This instrument is used for measurement of individual harmonics or intermodulation products on **an audio channel, for determining noise distribution, and for any other measurement of a discrete audio frequency level when other components are also present in the spectrum at other frequencies. It presents approximately the same information as that determined by audio frequency analysis with the spectrum display unit discussed earlier. The wave analyzer will, however, provide somewhat more accurate measurements.**

There are a number of different models of low frequency FSVs in the field at this time. In addition to the above functions, most units also provide an audio output of the selected frequency which can be used to monitor or record the signal. Many units also provide an oscillator output at the measurement frequency. This option is useful for performing frequency response measurements on filters, amplifiers, or any other equipment within the station.

f. Distortion Analyzer. The distortion analyzer is another common instrument employing a frequency selective device; however, this unit has only limited application and its function is opposite to that of FSVs. The distortion analyzer is basically a wideband level measuring set. In this capacity and as a voltmeter, it performs well. Figure 17-8 shows a filter in series with the incoming signal. This device is tunable and slot-rejects the fundamental frequency to which it is tuned. By inserting this device and tuning it to the frequency of a tone on the circuit, the tone frequency is completely

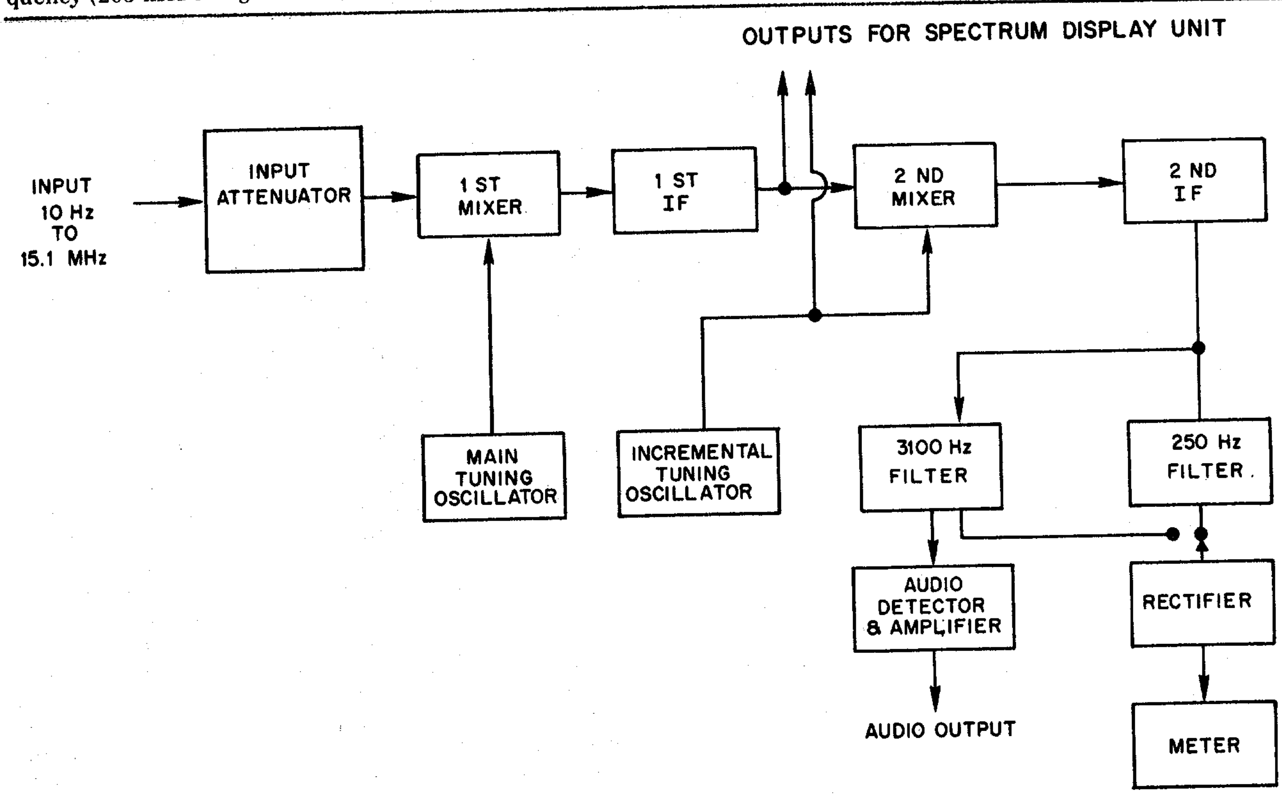


Figure 17-5. Frequency Selective Voltmeter.

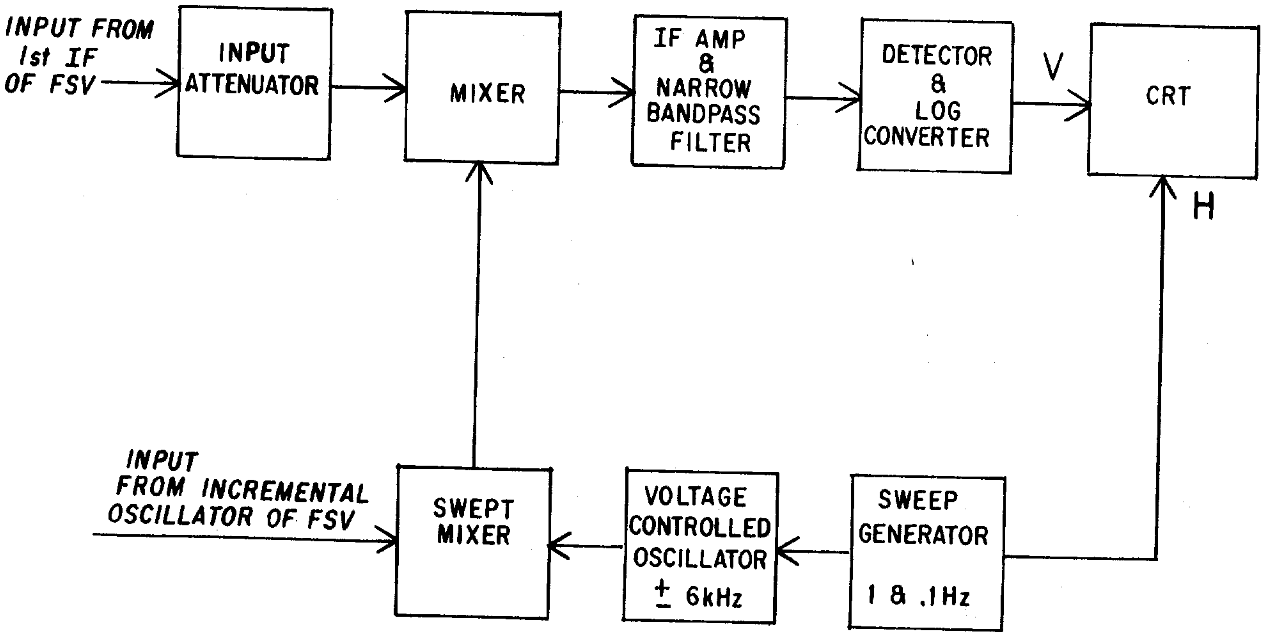


Figure 17-6. Simplified Block Diagram of a Spectrum Display Unit.

blocked and the meter then reads what remains on the circuit. In this way, a fast and easy measurement of the cumulative total level of the harmonics, noise, and crosstalk or other spurious products in a channel can be performed. This type of reading is useful as a quick check to determine the condition of a channel and the measurement is applicable to some equipment specifications. This measurement is not valid to apply to circuit parameters established by the DCA. Harmonic distortion and crosstalk must be measured with a low frequency FSV in order to be considered valid for that purpose.

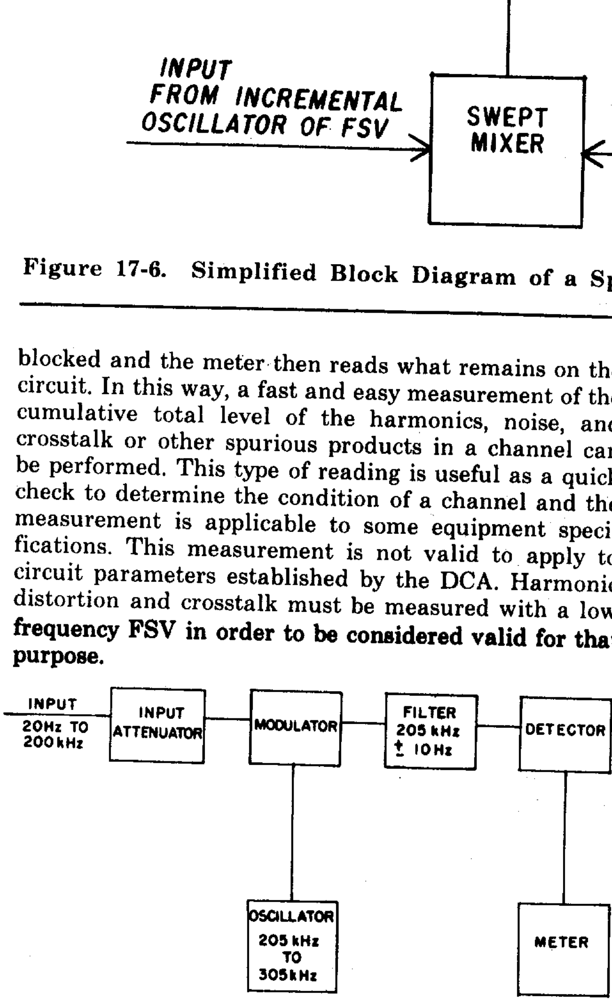


Figure 17-7. Simplified Block Diagram of a Low Frequency Selective Voltmeter.

17-5. Swept Frequency Measurements. To realize the requirement for, and advantages of, swept frequency measurements, visualize an IF alignment using manual methods. After an hour of measuring, plotting, and studying, one coil is turned one-quarter turn. Where do we stand now? Right back at the starting point and ready for another hour of measuring, plotting, and studying before we turn it another quarter turn. The solution is a sweep generator with accompanying oscilloscope display. The frequency is varied electrically between two preset frequencies, with constant level from the generator maintained by automatic leveling controls provided in the sweep generator and controlled by a feedback loop from the directional coupler employed. The RF output from the device under test is rectified by a crystal diode to obtain a DC voltage for application to the vertical deflection plates of the oscilloscope.

Basically, sweep generators provide dynamic displays, whereas single frequency generators provide static displays. Present sweep generators are built so that they can be employed for single frequency, single sweep, manual sweep, and selection of some preset frequencies merely by changing a switch position. In most units, calibrated harmonic markers are available at various increments (such as 1 MHz, 5 MHz, 10 MHz, etc.). They also have AM and FM modulation capabilities. Wide frequency ranges are permitted by changing plug-in oscillator units to produce different frequency bands. Sweep generators are employed primarily for RF, IF, and discriminator response, along with many VSWR checks. The number and type of tests which can be performed are limited only by the capability of personnel operating the test equipment. A leveled output over the entire frequency range of the unit under test is provided by the sweep generator. The swept frequency is produced by a voltage-tuned oscillator which changes frequency in step with a ramp (sawtooth) voltage. Synchronization with the display oscilloscope is accomplished by applying this sawtooth voltage to the horizontal deflection circuits of the oscilloscope. When viewing the display, amplitude is on the vertical axis while frequency is on the horizontal axis. If two vertical inputs are available, the markers from the generator may be displayed separately or

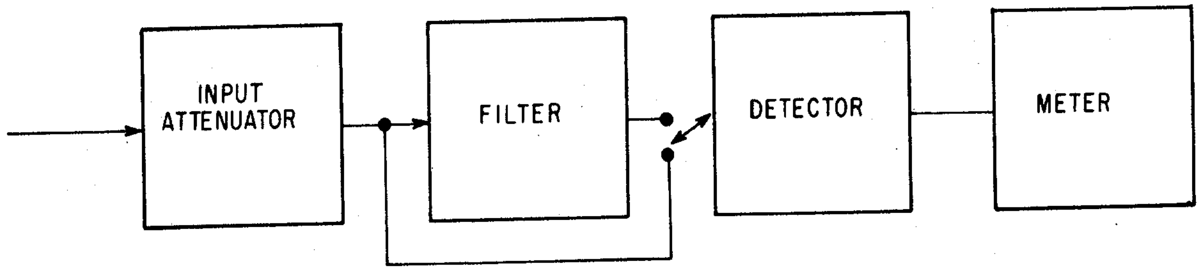


Figure 17-8. Simplified Block Diagram of a Distortion Analyzer.

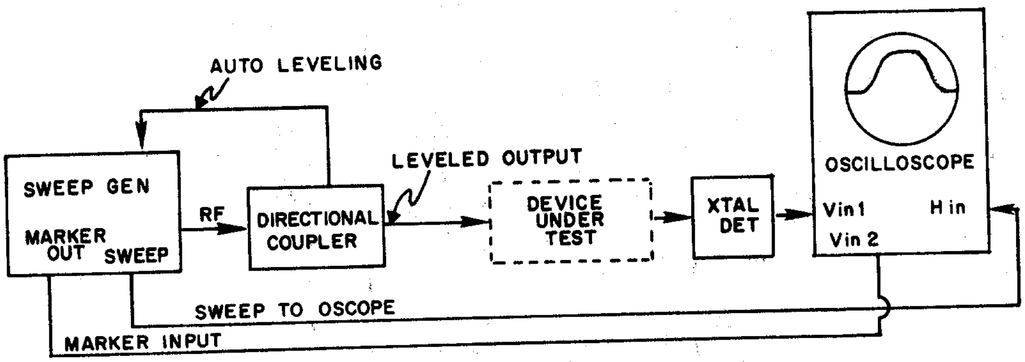


Figure 17-9. Sweep Frequency Measurement.

another spectrum may be displayed, such as the discriminator curve superimposed on the IF amplifier response.

A typical test setup is shown in figure 17-9. Equipment TOs and manuals, along with manufacturer test equipment application notes, provide much information on capability and use of swept frequency measurements.

17-6. RF Transmitter Output Power. The purpose of this test is to measure the power output of the radio-frequency transmitters and power amplifiers (PA) at the transmitter or PA output terminals, that is, at the antenna system input terminals.

Test Procedures - Transmitter/PA Output Power

a. TURN OFF the high voltage to the PA/transmitter.

b. Connect equipment (as shown in figure 17-10) with the dual directional coupler and the bolometer mount-power meter connected to the "forward power" point. Record the directional coupler attenuation value C (available on the side of the coupler or in technical information available in-station). Select attenuators A and B such that their power ratings will not be exceeded and they attenuate the incident and reflected power to levels that will not damage the bolometer or power meter. Record the value of the attenuators. Set **attenuator B, used in the reflected power circuit, at a high enough value to act as a termination. For accurate power measurements, the power loss (L) of the connecting lines, from the directional coupler to the** attenuators and from the attenuator to the bolometer mount input terminals, must be known and recorded.

c. TURN ON the high voltage to the PA/transmitter and apply full modulation to the frequency modulator; use a sinusoidal modulating signal having a frequency approximately equal to the center frequency of the baseband signal. For full modulation, the power level of the modulating (sinusoidal) signal should be:

Mod Sig Power = Loading Factor (dBm0) + Ref Test Tone Sig (dBm)+ **10 dB**

d. Record the power meter reading (R) in dBm.

e. Calculate the output power from this formula:

$$P_f = A + L + C + R - 30, \text{ dBW}$$
$$= \text{Antilog}_{10}\left(\frac{A + L + C + R - 30}{10}\right), \quad \textbf{Watts}$$

Where

P_f = Transmitter output power

A = Attenuator value in dB

C = Directional coupler attenuation value in dB

R = Power meter reading in dBm

L = Power loss of connecting lines from directional coupler to the attenuator and from the attenuator to the bolometer mount input terminals

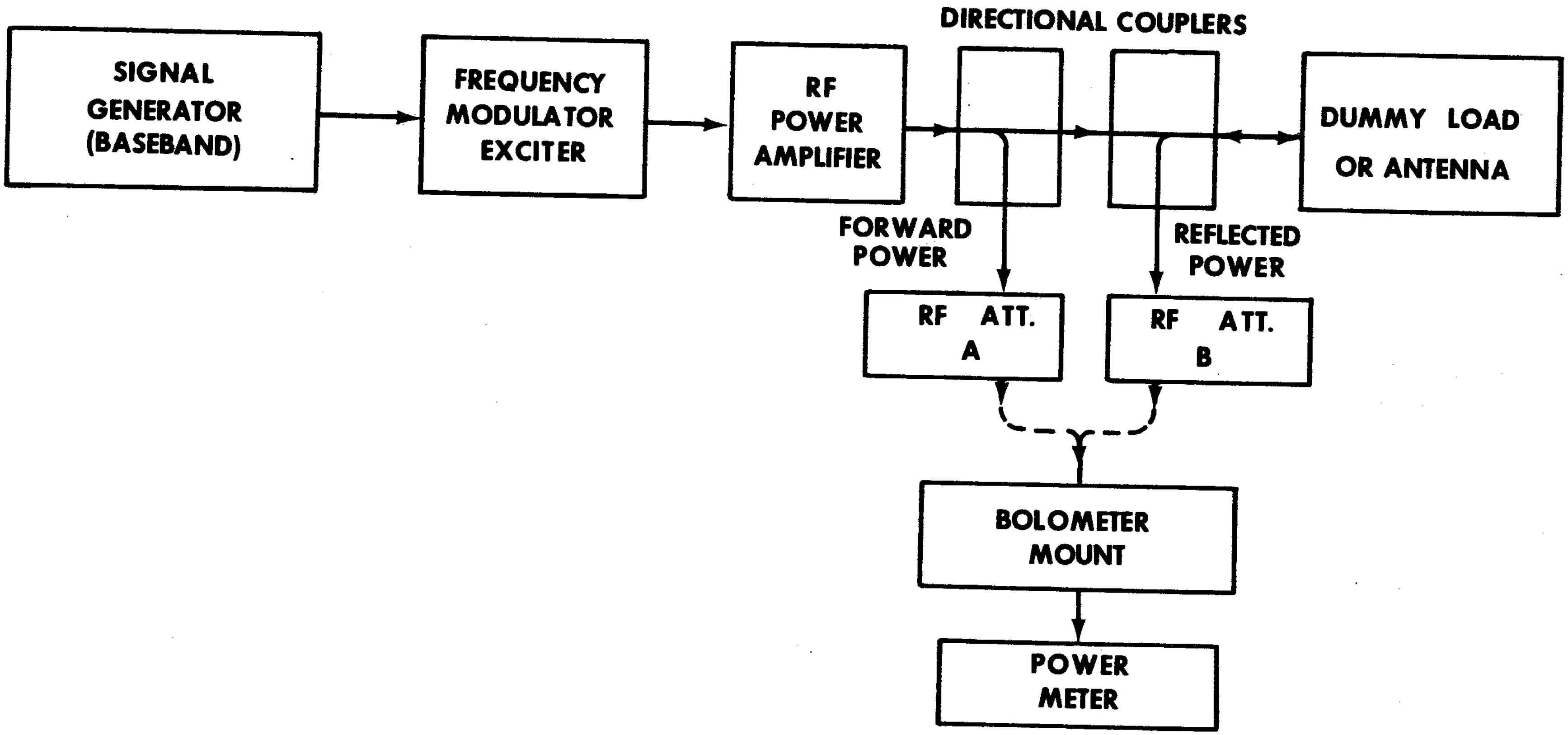


Figure 17-10. RF Transmitter Forward and Reflected Power Measurement Setup.

17-7. RF Transmitter Carrier Frequency Accuracy:

a. The purpose of this test is to determine if the transmitter carrier frequency accuracy is within manufacturer or TO specifications. The transitter under test must be taken out of operation.

b. In some cases, a "low frequency" exciter master oscillator signal is frequency-multiplied to produce the radio transmitter output frequency; however, it is preferred to measure the transmitter frequency directly instead of measuring the master oscillator frequency and then using the multiplying factor to *calculate* the radio transmitter frequency. If AFC of the radio transmitter is used, this AFC *must not* be disconnected during the test.

Test Procedures

(1) Remove the transmitter from operation but *do not* remove the heater power. Disconnect the baseband signal at the frequency modulator input and connect the test equipment (figure 17-11). In some types of transmitter - equipment configurations, this will remove the "drive" signal from the transmitter and, in that case, it will be necessary to substitute an unmodulated signal at the correct frequency, power level, and at the proper point (figure 17-11).

(2) Turn on the RF PA. Note that the trans-mitter may remain connected to the antenna during the test, with reduced RF power. If a dummy RF load is used, however, the transmitter may be tested under full-power conditions.

(3) Record the assigned transmitter carrier.

(4) Couple the RF frequency counter input to a suitable point in the transmitter by means of either a pick-up loop or a directional coupler and through appropriate attenuators. The RF frequency counter may also be connected to the "RF output test point," if available. If neither of these two RF signal monitoring points are available, it may be necessary to measure the master oscillator frequency. The actual transmitter output radio frequency must then be computed, using the frequency multiplying factor to translate the master oscillator frequency to the transmitter-output frequency. Assure that the method used to obtain the signal sample does not introduce frequency instabilities in the equipment under test.

(5) Adjust the RF frequency counter (using the method specified in the manufacturer's operation manual) until the transmitter carrier frequency or the master oscillator frequency is indicated.

(6) Manually record the time-varying transmitter frequency at 5 to 10-second intervals. This test should be conducted for at least five minutes. If the transmitter frequency drift is more than $\pm$150 kHz, a longer test period may be necessary. Record the measured RF transmitter average carrier frequency.

(7) Calculate the RF transmitter carrier frequency accuracy using the formula:

$$A = \frac{F_c - F_{md}}{F_c} \times 100, \text{ percent}$$

where,

A = transmitter carrier-frequency accuracy

F_c = assigned carrier frequency

F_{md} = measured average carrier frequency

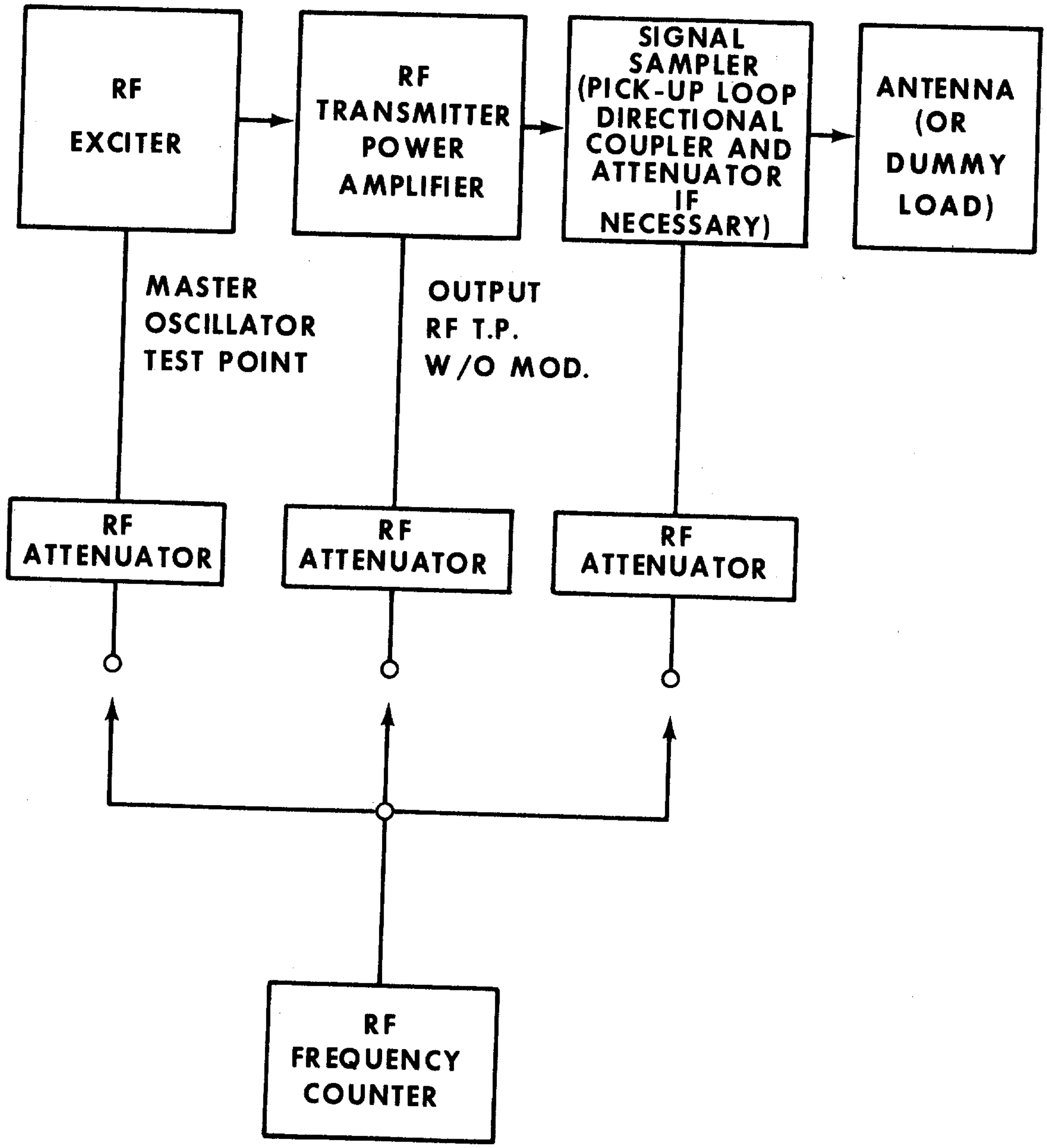


Figure 17-11. RF Transmitter Carrier Frequency Accuracy.

Chapter 18
KEY SYSTEMS PERFORMANCE INDICATORS

18-1. General. There are many different symptoms of trouble that may occur in communications circuits. Many of these symptoms can be considered "key" indicators since they indicate the performance of the system as well as the performance of the individual circuits. These key indicators are levels, noise, and distortion. The indicators all interact with one another and with other performance parameters. Examination of these three indicators can offer an insight into the overall performance of the system.

18-2. Level Management:

a. General. Level is an expression of relative signal strength at various points in a communications circuit. In communications terms, the level is given in dB relative to a reference level (usually 1 milliwatt is the reference used). The unit then becomes dBm. Chapter 3 discusses levels through a communications system and stresses the method of TLP and dBm0. This allows us to place a tone of, say, -10 dBm0 into the system and measure the level at any other point in the system as -10 dBm0 (for a perfectly aligned system). Chapter 6 discussed the effect of loading on system noise performance. The radio and multiplex equipment are usually designed to operate near the system overload point at peak traffic periods. This ensures optimum overall performance; however, this assumes that each user's input is at a certain level and that the system is aligned properly. Any high levels in the baseband raise the loading level to the modulator and can cause severe intermodulation noise in all channels. This situation is shown in figure 18-1.

b. Procedures. Level measurements may be performed in-service or out-of-service. In-service measurements involve the measurement of the level of the traffic on a circuit without disrupting customer service. This offers a valuable performance indicator in that, if the level of the signal is not at the standard transmission level for the type of traffic, a problem is suspected to be present. The problem may be within the system or at a user location. Alignment of channel level should not be attempted with the subscriber signal as the source, since a complex signal (such as a VFCT system) may cause an error in measurement. Whenever possible, troubleshooting should be conducted out-of-service, with the customer's traffic on an alternate route, using a sinusiodal test tone on the affected channel.

It is important that levels be corrected at all points in the system. It is not sufficient to correct a level problem at a point other than at the problem source. Consider the following example. A -10 dBm0 tone is inserted into a system at channel level. Assume that the gain of the group equipment at station A has been adjusted 10 dB too high. The high level will be present until it is detected at the tech control at the next link (station B). If station B "fixes" the symptom by lowering the gain of the channel equipment 10 dB at his/her station, the system looks aligned from tech control to tech control (figure 18-2).

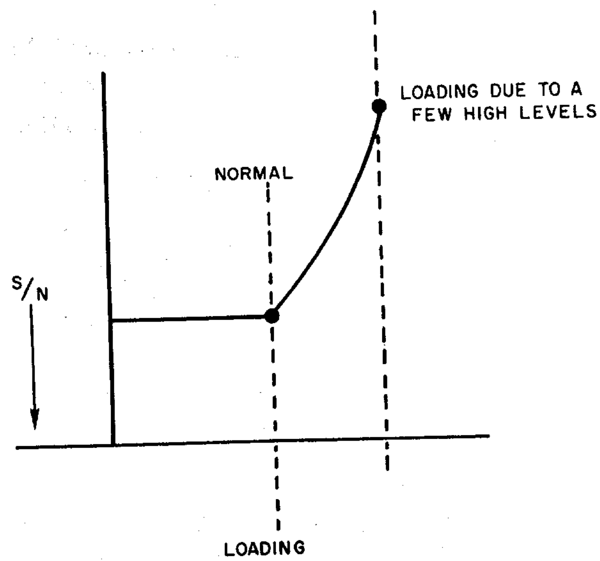


Figure 18-1. Effect of High Levels on Channel S/N.

Although each tech controller sees a -10 dBm0 tone at his facility, the supergroup and baseband circuits are being overloaded by a 0 dBm0 tone. This will cause severe intermodulation noise throughout the baseband (noise that cannot be eliminated by turning down a gain control somewhere else in the system). The proper procedure is to isolate the level problem and correct it at its source. Figure 18-3 shows a circuit which will serve as an example of system level control.

Suppose Station E measures a high level coming from D. E should now contact the first station back along the circuit to which he/she has orderwire contact. In this example, this would be station D. If a level difference exists between D and E, link level adjustments should be initiated. If a high level still exists at D, he/she also should contact the next station back along the circuit. In this case, that is back to station C. This procedure continues until the link(s) or subscriber causing the high level is isolated.

If tech controllers at opposite ends of a link measure a difference on a channel, the channel should be altrouted and a test tone put on the channel between the two stations. As a preliminary check, other channels should be checked in service, preferably one in the same pregroup, one in the same group, one in another group, and one in another supergroup. This can give a clue as to the extent of the problem and, thus, an indication of the source.

The first essential step in correcting the link problem is to measure the test tone level at the receive baseband.

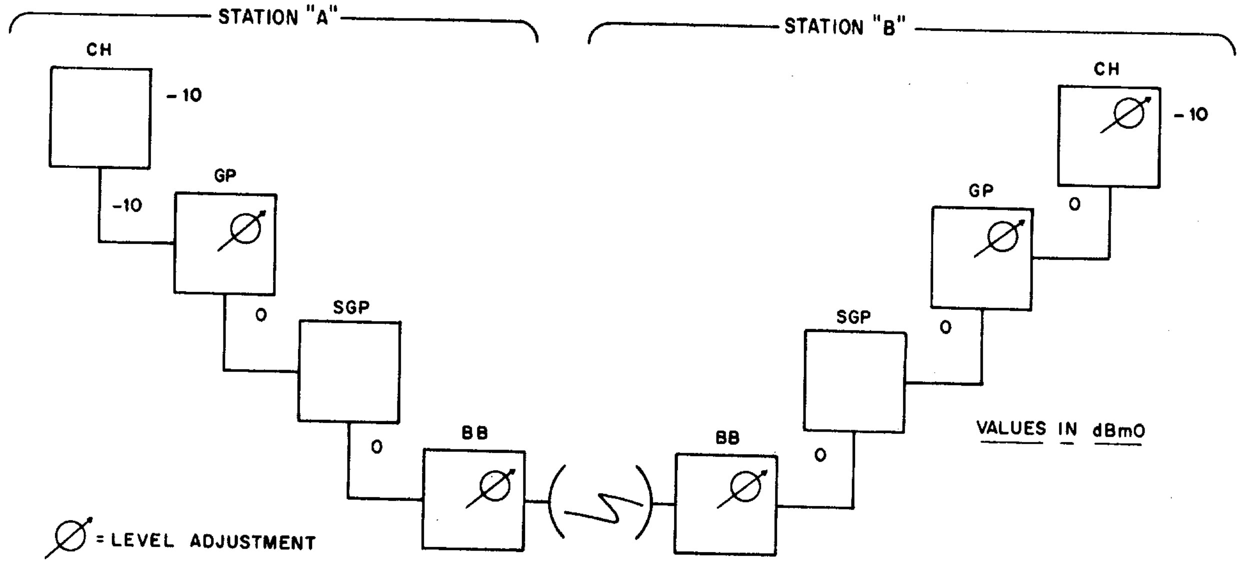


Figure 18-2. Values in dBm0.

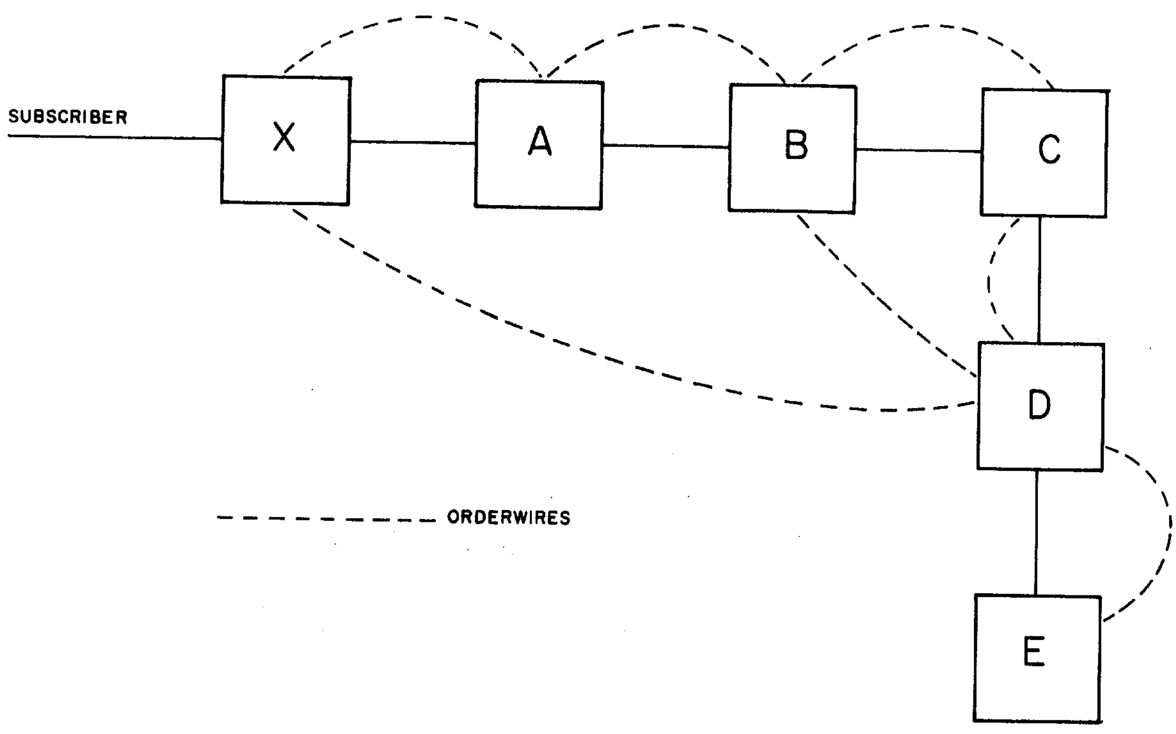


Figure 18-3. Sample Circuit for Level Control.

If the level is correct, the variation must be corrected in the multiplex equipment. If the receive baseband level is incorrect, the transmit baseband at the other station should be checked. (In diversity systems, all combinations of equipment should be tested separately.) Any variation between the two baseband levels should be corrected. Any level problem remaining must occur in the transmit multiplex. TO procedures should then be followed for multiplex alignment. From this procedure, a quality control program for level management of all channels can be derived. The first step is for each station to level the groups and supergroups in the multiplex. This should be performed with the group/supergroup outputs terminated in their proper impedances. A channel near the center of the group/supergroup should be used for leveling purposes. Each station should then transmit test tones to the adjacent station. The baseband levels should be verified and the supergroup/group levels should be set, using channels near the center of the unit being leveled. As a final step, each channel should be leveled, using the level adjustment on the receive channel equipment. (If pregroup level adjustments are provided, they should be set in the same manner as groups.)

In some instances, the level response across a group or supergroup will not be uniform; hence, a high channel in a group may show an incorrect level when measured at a baseband point. This variation should be considered when measuring level at a baseband. The group level adjustment should not be made each time a channel is measured at baseband. Only one channel should be used for group (or supergroup) level adjustments. As an alternative, the group or supergroup pilots may be used for leveling. The important thing is to adjust all multiplex in a station in the same way, allowing interchangeability with a minimum of level error.

18-3. Channel Noise:

a. General. Any extraneous distrubance in a channel which interfere with the desired signal can be classified as noise. The most common form of noise encountered in a channel is white noise. White noise can arise in a channel from thermal sources or from intermodulation products of complex signals. The noise may be added in the multiplex equipment, the radio equipment, or may be due to propagation effects (fading, multipath distortion, etc.). The key parameter for assessing system noise performance is the ICN. This can be measured over any number of links and is the first indication of a system noise problem. The amount of noise in an idle channel is measured directly with a noise meter. Appropriate weighting filters should be used to conform with current standards.

b. Test Procedures. The measurement is made between two specific stations. One station terminates the transmit side of the channel and the second station reads the ICN directly, using this meter to terminate the receive portion of the channel. The two stations may be on opposite ends of a link or separated by many links. A typical measurement is shown in figure 18-4. The figure read on the meter is the ICN from station A to station E. The value of noise taken must be corrected for any variation in level over the channel. If there is any net loss between station A and station E, the ICN must be increased by the amount of net loss. For example:

ICN measured:	-60 dBm0	Net loss: 3 dB
Test tone sent:	-10 dBm0	Corrected ICN:
Test tone received:	-13 dBm0	-57 dBm0

NOTE: Only these corrected figures should be used for analysis.

c. Analysis of Results. DCA has established standards for allowable ICN, based on technical evaluations, Test and Acceptance (T&A) data, statistical data, and theoretical path calculations. These standards are used extensively throughout the DCA Performance Monitoring Program (PMP) as implemented by DCAC 310-70-57, sup 4. Since ICN is a function of both BBL and RSL, all three of these readings must be correlated before any conclusions can be drawn. This is because high channel may be caused by defective radio or multiplex equipment but it may also be caused by overloading the baseband or by low RSL.

The first indication of the problem will be a high ICN reading over a circuit. An attempt should be made to identify the noise source before proceeding. This can often be done by using a monitor speaker and listening to the noise. Any impulses or single tones caused by carrier leak or crosstalk should be evident. As a final check, the channel could be swept with a FSV.

The noise source must now be isolated to the next lower level, the link. Each link, as well as the circuit, should have an operating standard for ICN. Measurements made by tech controls according to established test procedures should isolate the link or links causing the excessive noise.

If excessive noise is indicated on a link, the operating conditions must be checked before the noise measurement can be analyzed. On LOS M/W links, the RSL is normally stable and extended fades are rare; however, on troposcatter links, RSL is extremely variable and is an important factor in analyzing noise readings. If the excess noise on the link is being caused by a low RSL, no action can be taken if propagation is at fault. In diversity systems, combiner action should be checked to verify that all the receivers are experiencing the fade. If the combiner is malfunctioning (that is, not choosing optimum combinations of receivers), this condition should be rectified before further testing. Assuming the RSL to be satisfactory, the possibility of excessive levels on the baseband must be investigated. High levels anywhere in the baseband can overload the radio modulators and demodulators and cause intermodulation noise in all channels. This condition can most easily be detected be sweeping the baseband with a FSV (and spectrum display unit, if available). If a high level or levels are detected in the baseband, the location should be reported to tech control where the channel or channels responsible can be determined. Any level problem should be resolved before proceeding.

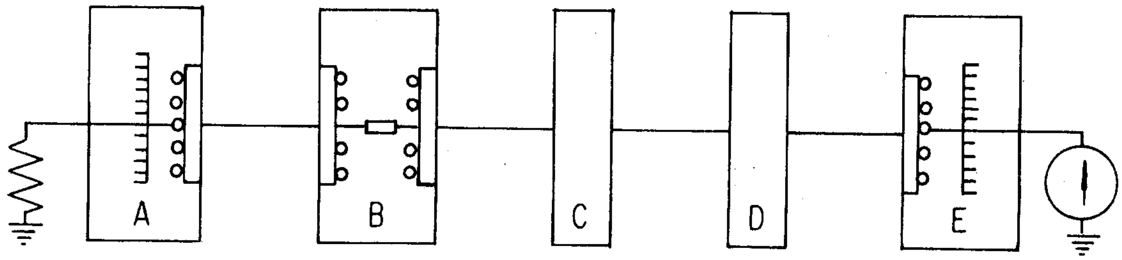


Figure 18-4. Idle Channel Noise Measurement.

If loading and RSL have been eliminated as causes, the problem must be localized to a piece of equipment or attributed to path intermodulation. If diversity is employed, the radio equipment NPR may be measured on a loop-back basis without a system outage. An RF loop NPR/Basic Intrinsic Noise Ratio (BINR) test should be performed according to the procedures given in chapter 19. The NPR should be compared with TO specifications for the equipment. T&A and recent Scope Creek data should verify what the equipment is capable of producing. Again, loading level and RSL must be the same as specified in the standard.

If the loop-back NPR is out of specification in any slot, a problem is indicated in the radio equipment. The BINR in the bad slot(s) should be compared with the NPR. If they are both bad, a thermal noise problem exists in the radio. The receiver noise figure should be checked; this is the most likely cause of excessive thermal noise. If the noise figure is within manufacturer's specifications, the deviation setting on the modulator should be checked against the equipment TO. This must be traced through each stage of the radio. If the BINR is satisfactory, an intermodulation problem exists. An IF loop NPR should be done to further isolate the source. If the IF loop NPR is okay, the problem is in the RF circuitry, exciters, power amps, and RF amplifiers and mixers. If the IF loop is still bad, the bad slot should be noted. A bad NPR in the high slot indicates linearity group delay or mismatch problems in IF circuitry. A bad reading in the low slot indicates linearity problems in the modulator, discriminator, or baseband circuits.

Going back to the initial RF loop-back NPR test, if the NPR is good at both ends of the link, then the problem must lie in the multiplex, the waveguide/antenna system, or the path. A VF two-tone intermodulation test (or a VF harmonic distortion test) should now be done by tech control on the channel under test. If no non-linear distortion is present, the problem must be idle noise in the multiplex equipment. Further tests on multiplex equipment cannot usually be done without taking the equipment out of service. If the VF test reveals an intermodulation problem, there are three possible sources: the multiplex equipment, the waveguide/antenna system, or the path. Since none of these systems are redundant, the tests would have to be done out-of-service. The first test would be a link NPR. There is no set standard for link NPR. The initial T&A and Scope Creek data may be used for comparison if the conditions are comparable. In general, this test will be most useful if taken when the RSL is relatively high. If the link NPR is acceptable, the problem is multiplex intermodulation and could be traced further only with mux-loaded noise tests. If the link NPR is not acceptable (and the RF loop NPR was), waveguide defect or path intermodulation is assumed. Complete VSWR tests, as outlined in the succeeding chapter, will reveal any defects. If the return loss is low at both ends of a section, echo distortion can be a problem. This will cause noise in the high slot. If the VSWR is okay, the noise must be attributed to path intermodulation. This is rarely a controlling factor except in long troposcatter links.

The foregoing procedure is not the only way to go about troubleshooting a problem. The experience and ingenuity of the technician enters into any situation of this kind; however, this section demonstrates the procedure of system troubleshooting and how close coordination between tech control and maintenance can simplify fault isolation.

d. Common Noise Problems. While all of the problems covered in the troubleshooting guide are possible, there are some which are common enough to mention separately.

Experience has shown that poor propagation is not a major problem in our systems; however, it is commonplace to blame any failure or outage on propagation. This should not be done until all other possibilities have been checked (for example, receivers, transmitters, antennas, etc.). A common malfunction, often blamed on propagation, is combiner failure. Any individual receiver in a troposcatter system will experience frequent fading. If the combiner is not functioning properly, the effects of these fades will be seen in the channels.

Also, propagation cannot be poor in one direction only over the same path. If such a "one-way fade" occurs, an equipment problem is indicated.

A high thermal noise problem, which might be mistaken for a propagation fault, is often caused by a bad receiver front-end. Parametric amplifiers are a difficult piece of equipment to maintain. While a "paramet" is designed to have a noise figure of 2 or 3 dB, units in the field often have noise figures of 8-10 dB. Paramets should be suspected and checked whenever high thermal noise is present.

If NPR data indicates intermodulation noise in the radio, the most common sources of non-linear noise should be checked first. The receiver discriminator is the main contributer of intermodulation noise. This noise usually appears in the low slot of an NPR test. In general, precise discriminator adjustments are difficult without proper equipment; however, discriminators must not be tuned for best NPR while looped-back in-station. If any "peaking" is done, it must be with a link test setup. The modulator is also a prime source of intermodulation, especially in the troposcatter Serrasoid modulators. This causes noise mainly in the low-slot in an NPR test.

18-4. Impulse Noise (IPN):

a. General. IPN can enter a channel at any point and be carried throughout the system. Impulses can be generated in the equipment itself, be picked up from power lines, or be generated in the environment and radiated into the station. Excessive IPN can severely impair data transmission and must be traced to its source when it is detected. The IPN test can measure IPN at the system, link, or component level.

b. Test Procedures. Normal meter movements will not respond to rapid voltage impulses due to the inertia and damping of the needle movement; therefore, it is not practical to attempt to measure the amplitude of every impulse that occurs. Instead, we will treat the measurement on a statistical basis. A test set can be connected to the channel to measure the number of impulses above a preset level and below the next higher threshold level. If more than one level is used, a profile of the impulse amplitude distribution can be derived. DCAC 310-70-57, sup 4, establishes thresholds for IPN, with voice band weighting.

c. Analysis of Results. When an IPN counter shows there are 15 or more impulses greater than 72 dBrn0 during a 15 minute period, the problem should be further investigated. If excessive IPN is observed, isolation procedures should be initiated. The IPN test should then be performed on successive links until the link(s) where the noise enters the system can be isolated. On a problem link, the channel should be terminated at the transmit end. An IPN count should be recorded at the receive end of the link at channel level to provide reference data, IPN counts should also be performed on other channels to try to isolate the problem to group, supergroup, or baseband equipment. The most these measurements can do is indicate where the IPN is entering the system. The source of the noise is more difficult to trace. Much may be learned from listening to the time distribution of the impulses. These can sometimes be correlated with ringing tones, telephone dialing impulses, radar sweeps, etc. If the problem has been isolated to the path, possible sources of electrical disturbance should be investigated.

18-5. Peak Data Distortion:

a. General. Digital data is transmitted over most wideband systems. This data may be in the form of VFCT packages or medium speed (2400 baud) AUTODIN circuits. These trunks were designed and installed to operate over wideband systems with very little digital distortion. If the path over which a digital data trunk is transmitted becomes degraded in any way, the digital distortion will increase commensurately. This fact alone indicates that periodic measurement of digital distortion can serve as a key indicator to the performance of the system over which the digital data is transmitted. To further enhance the value of this indicator, analysis of the specific digital distortion measurements often indicates the type of degradation (noise, frequency translation, etc.) which is causing the distortion.

b. Modems and VFCT Systems. To understand the relationship of DC digital signals and VF channels, we must first examine the way in which these DC signals are transmitted. Digital data is generated as a stream of DC pulses which form a specific code. The DC pulses may be transmitted for short distances over metallic circuits but, in order to transmit them over a VF channel, it is necessary to convert the coded intelligence from DC to audio.

The conversion is normally accomplished through the use of a frequency shift "tone keyer." A tone keyer usually contains two oscillators, each operating at a different audio frequency. The actual frequencies selected depend on the bit rate to be employed and the location desired in the spectrum. A unit designed for 1200 baud operation may use 1200 and 2400 Hz and a unit designed for 90 baud operation may employ 372.5 and 467.5 Hz. The keyer is arranged so that only one of the two frequencies is transmitted at any one time. The DC data stream is fed to the input of the keyer and, when the input data bit is a mark (1), the tone keyer transmits one frequency and, when the input bit is a space (0), the first frequency is removed from the line and the second frequency is transmitted. The switching between the two frequencies is a form of modulation and is done electronically. The unit is capable of switching the frequencies at a rate equal to the rate of the input data bit stream so the intelligence may now be transmitted on a VF channel.

The receive station must have a converter or demodulator unit which will produce a mark bit when it receives one frequency and a space bit when it receives the other. The tone keyer and the converter are usually colocated or built as one unit and called a "modem" (MOdulator - DEModulator). Since the modems designed for low bit rate circuits can use narrow frequency shifts (that is, 85 Hz), a number of these modems operating at different frequencies can be connected onto one VF channel. This system of stacking channels is also referred to as FDM and the multichannel terminal units are called VFCT systems. The more common types employ 16 channels, each separated by 170 Hz (from the hypothetical center of one channel to the next) and operate in the spectrum from 372.5 to 3017.5 Hz. Other modulating techniques (such as phase-shifting a single frequency at the rate of the input DC bit stream) are also employed in similar equipment.

c. Digital Distortion. A general definition of distortion, that is, an unintentional change in the electrical signal, applies to DC digital distortion as well as VF path distortion. There are two types of digital distortion which are of primary concern: fortuitous and bias.

(1) Figure 18-5 shows a typical, undistorted serial bit stream. Figure 18-5B shows this same bit stream as it may appear when affected by fortuitous distortion. Fortuitous distortion is random by nature. It does not follow any specific pattern and can easily be identified by a random misshaping of the pulses.

(2) Bias distortion is characterized by its uniformity. It can be defined as a uniform lengthening or shortening of the mark or space pulses, one at the expense of the others. Figure 18-5C shows the serial bit stream with the mark bits elongated at the expense of the space bits. This condition is referred to as "marking bias." If the space bits are elongated at the expense of the mark bits (figure 18-5D), the condition is referred to as "spacing bias." There are many other types of digital distortion (such as speed distortion caused by a difference in operating rates between the send and receive devices, carrier distortion which appears as "jitter" and is caused by the modulation process of the lowest frequency channels of a VFCT system, and cyclic distortion which appears as "hits" occurring at a specific rate). These types of distortion are less prevalent than fortuitous or bias distortion and usually relate to digital transmission equipment faults rather than path faults.

d. Digital Distortion Measurement. Periodic measurement of digital distortion is performed at all DC breakout points. This measurement is usually accomplished at least daily on the actual traffic on the digital circuit (in service). If an excessive amount of distortion is noted on a digital data circuit or if a degrading trend is noted, the traffic is altrouted by the tech controllers and a test signal is inserted into the VFCT channel. The test signal is usually a "test message" (THE QUICK BROWN FOX, etc.) generated at the maximum bit rate for which the channel was engineered (usually 75 baud for a VFCT system). Standard test pattern generators are installed in all TCFs which may be required to perform digital distortion tests. A digital data distortion measuring set is connected to the receive of the channel (figure 18-6).

The newer measuring sets compare the incoming data bit stream to an internally generated time base and display the difference between the two on a meter calibrated in percent of distortion. Switch selections permit the meter to display total distortion, or bias distortion, or other more detailed characteristics of the signal. Another switch selection permits the set to measure the peak amount of distortion over any period of time from seconds to many minutes. The input bit stream may be simultaneously displayed on an oscilloscope to permit visual analysis of the signal while performing a distortion measurement.

e. Comparison with Accepted Standards. A measurement of total peak distortion with the measuring set encompasses all types of distortion and provides a readout of the cumulative total. This number may be compared directly with the criteria for a circuit. A measurement of bias distortion may also be performed and the results compared with the existing standards. These readings should be recorded and analyzed.

f. Analysis of Results. The recorded results should be analyzed as soon as the measurement is completed. If a VFCT system is involved, the channels should be analyzed both individually and as a system. The obvious analysis points of excessive distortion on an isolated VFCT channel or a daily trend increase requires immediate troubleshooting to isolate any digital data transmission equipment faults; however, when analyzing the results of measurements on a complete VFCT system, it is possible to relate these problems to a common cause.

When the audio signals from the modems are transmitted over a VF channel, the various characteristics of that channel affect the audio tones which represent the coded information. When the tones are demodulated, the resultant DC signals may be slightly different from the original signals (distorted) in an amount proportional to the degrading effects of the VF channel. The various characteristics of the channel all interact in causing digital distortion; however, a detailed analysis of the recorded measurements of the DC signal may relate back to a VF channel problem.

Fortuitous distortion on all channels is usually caused by a poor S/N ratio on the path. The poor S/N ratio may be the result of high ICN or low RSLs. This symptom can also be caused by high RSLs, intermodulation distortion, IPN, or phase jitter. High signal levels may be detected in-service, intermodulation distortion is usually accompanied by high ICN, and phase jitter is a rare problem. The fortuitous distortion measurement may be characterized by random hits to 50% distortion. These hits are often indicative of IPN on the path.

Bias distortion of the same type (marking or spacing) may be found in similar amounts on every channel. This often indicates a small amount of frequency translation on the path. Some VFCT systems will show as much as 5% bias for a 1 Hz translation; however, the amount of bias is not proportional to the amount of translation. The increase in bias for subsequent increases in translation are progressively larger.

Another sympton which sometimes occurs involves bias on the "outside" one or two channels of a VFCT system (the channels with the highest and lowest operating frequencies). This may be caused by envelope delay distortion on the VF channel. Fortuitous distortion on the outside channels may be caused by either envelope delay distortion or poor frequency response on the VF channel. Most VFCT systems have some inherent carrier distortion on the low frequency channels and this should be noted on an in-station loopback test.

Many digital distortion problems on circuits employing a single modem (for example, 2400 baud AUTODIN trunks) can be related to VF channel problems; however, it is usually necessary to compare the incoming audio waveshape to a perfect product. Experience on local circuitry is prerequisite to this type of analysis.

Problems such as high ICN or IPN may be observed "riding" the incoming signal on such circuits.

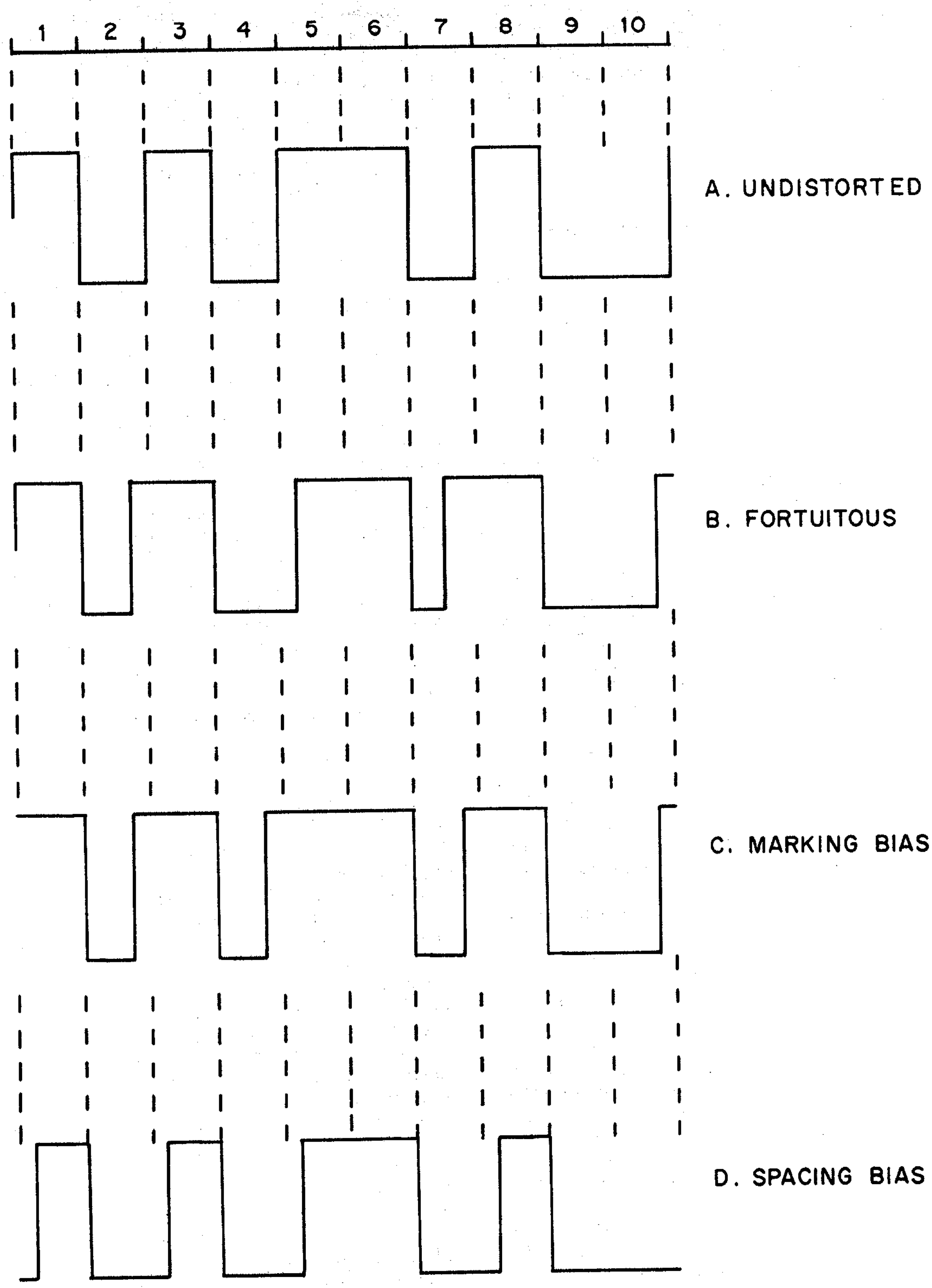


Figure 18-5. Distortion on Digital Signals.

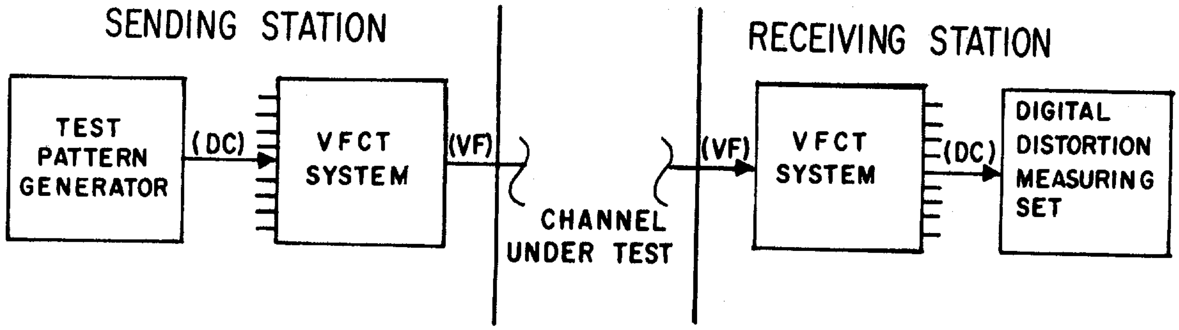


Figure 18-6. Digital Data Distortion Measurement.

Chapter 19
SUBSYSTEM PERFORMANCE TESTS

19-1. General. This chapter describes the tests which have been used by the DCA Technical Evaluation Teams (Scope Creek) to evaluate wideband systems and their individual components. Detailed test procedures will not be presented here. DCAC 310-70-57, sup 1, may be consulted for specific test configurations. The individual tests are useful in identifying faulty or substandard components and a general procedure is outlined for each test but must, of course, be tailored to the particular equipment. The main purpose for presenting these tests is to encourage an understanding of the principles involved - not to present rote instructions to follow. To that end, an analysis of the test results is also presented using representative values which would be measured and the probable meaning of any unusual readings obtained.

19-2. Noise Power Ratio (NPR):

a. General. As stated in chapter 6, NPR is a comprehensive indicator of noise performance in a component or system. The test may be run from any baseband point in the system to any other point where that baseband appears. Normally, the test is done on a link basis and on an in-station/loop-back basis. A link NPR configuration is shown in figure 19-1. Voice communications must be established over the link to coordinate the measurement.

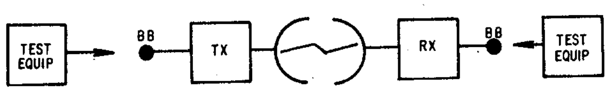


Figure 19-1. Link NPR Measurement.

For an in-station test, the transmitter is looped back to the receiver and the test is run from the transmit baseband to the receive baseband. The transmitter will not be tuned to the same frequency as the receiver, so a frequency translating device (frequency translator, monitor converter, and turn-around mixer) will be necessary to perform an RF loop. If a frequency translator is not available, the equipment should be looped back at the IF level. These configurations are shown in figure 19-2.

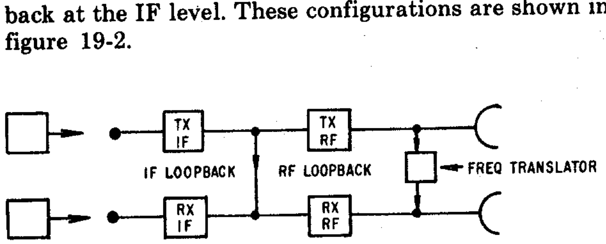


Figure 19-2. Loop-Back NPR Measurements.

An NPR measurement directly measures the total noise in a baseband slot due to thermal sources and intermodulation. The slot bandwidth is usually chosen to correspond to a voice channel, making NPR relatable to per-channel noise. In addition to the NPR, another ratio (BINR) can be directly measured with this test setup. It gives a measure of only the thermal noise in the slot. These two ratios combined allow us to measure the relative contribution of thermal and intermodulation noise to the total channel noise. These tests can be used as periodic checks of radio equipment/link performance and also as a valuable troubleshooting aid.

b. NPR Measurement Technique. The heart of the NPR measurement technique is the use of white noise to simulate a multichannel baseband signal. This procedure was discussed in chapter 6. To accurately simulate the multichannel signal, the noise spectrum must have the proper bandwidth and level. The noise signal is bandlimited by passing it through a high-pass filter and a low-pass filter. The lower and upper cutoff frequencies are chosen to match the actual baseband of the system (figure 19-3).

Figure 19-3. Generation of Simulated Baseband Signal.

The noise-loaded baseband is then adjusted to the proper level. Since noise performance will be worse during the busy hours, the system will be evaluated with simulated busy-hour loading. The theory behind the calculation of loading level was covered in chapter 6. Figure 19-4 gives the loading to be used for a given number of channels. The CCIR curves are for voice channels and much of our present equipment was designed to handle this loading. The other curve assumes 100% data loading and must be used in evaluating new LOS equipment.

For example, suppose we have a 300-channel system designed for voice channels. Using CCIR loading curves, we would set our noise level at +9.8 dBm0. If the system was a 300-channel LOS link subject to data loading, we would set our noise level at +14.8 dBm0. Measurements should be performed at the full channel capacity of the radio equipment. For example, if a 300-channel M/W radio is being used to carry 24 channels, the radio should be tested for 300 channels, its full capability. When making link NPR measurements, loading for full capability may exceed the link authorized RF bandwidth. In these cases, care should

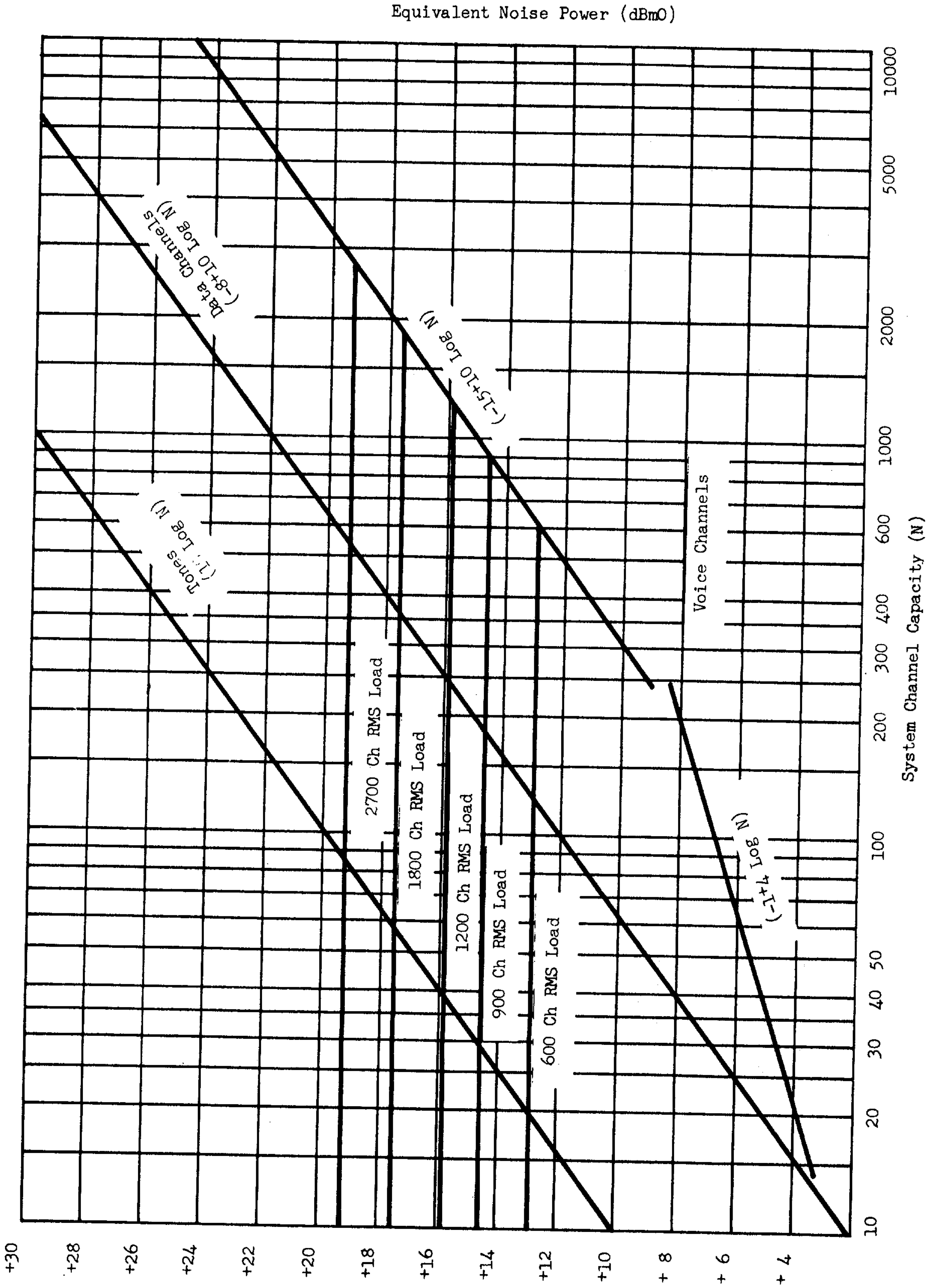


Figure 19-4. Loading Factors for Voice Channels (CCIR), Data Channels at -8 dBm0 Input Level per Channel, and Channels Carrying 0 dBm0 Tones. For data channels at -10 or -3 dBm0 per channel, subtract 2 or 5 dB, respectively, from final loading figure.

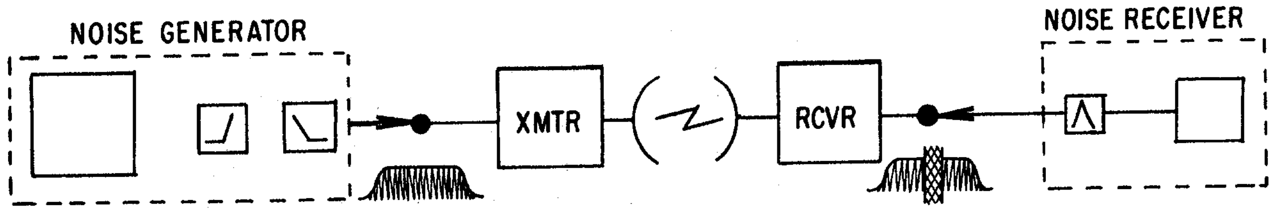


Figure 19-5. Setting Reference Level for NPR Test.

be exercised in performing the tests and actual radiating time should be held to a minimum. This baseband signal is sent through the system under test and measured at the receive baseband. The noise power in a 3 kHz slot somewhere in the baseband is measured with a special noise receiver. This value is used as a reference level. The test procedure so far is shown in figure 19-5.

Now a band stop filter is inserted at the transmit end to create a slot in the transmitted baseband. This slot must coincide with the slot in the noise receiver. Ideally, then, no noise would be measured at the receiver; however, thermal noise will be measured in the slot as will intermodulation products. The amount of noise in the slot is a measure of the system under test. The results are usually expressed as a ratio of the noise measured with the transmit slot out to the noise with the slot in. This ratio is called the NPR. It is expressed in dB and is the ratio of two powers.

The usual technique for direct measurement of NPR is to put an attenuator after the receiver and set a reference on the noise receiver with the transmit slot out. Then, with the transmit slot in, attenuation is taken out until the meter or the noise receiver returns to the reference. The amount of attenuation removed is equal to the NPR in dB. Special test equipment has been developed to measure NPR by the above method. This test consists of a noise generator (with built-in filters which are selected from the front panel), a noise receiver with a choice of receive slots, and a built-in attenuator for direct readout of NPR. A typical measurement using the NPR test set is shown in figure 19-6.

The slots to be used in measuring any baseband are specified by DCA. Table 19-1 lists the high pass, low pass, and slot filters to be used with various channel capacities. NPR/BINR should be recorded for all slots with all test conditions specified.

c. Analysis of NPR. The results of equipment NPR tests should be compared to the manufacturer's specifications for the equipment. Depending on the specific equipment, this is usually 50 or 55 dB. DCA also has standards for NPR, depending on the type of equipment. Link NPR readings are difficult to analyze if propagation conditions are varying. A comparison may be made with the loop test and acceptance data if the conditions (especially RSL) are similar.

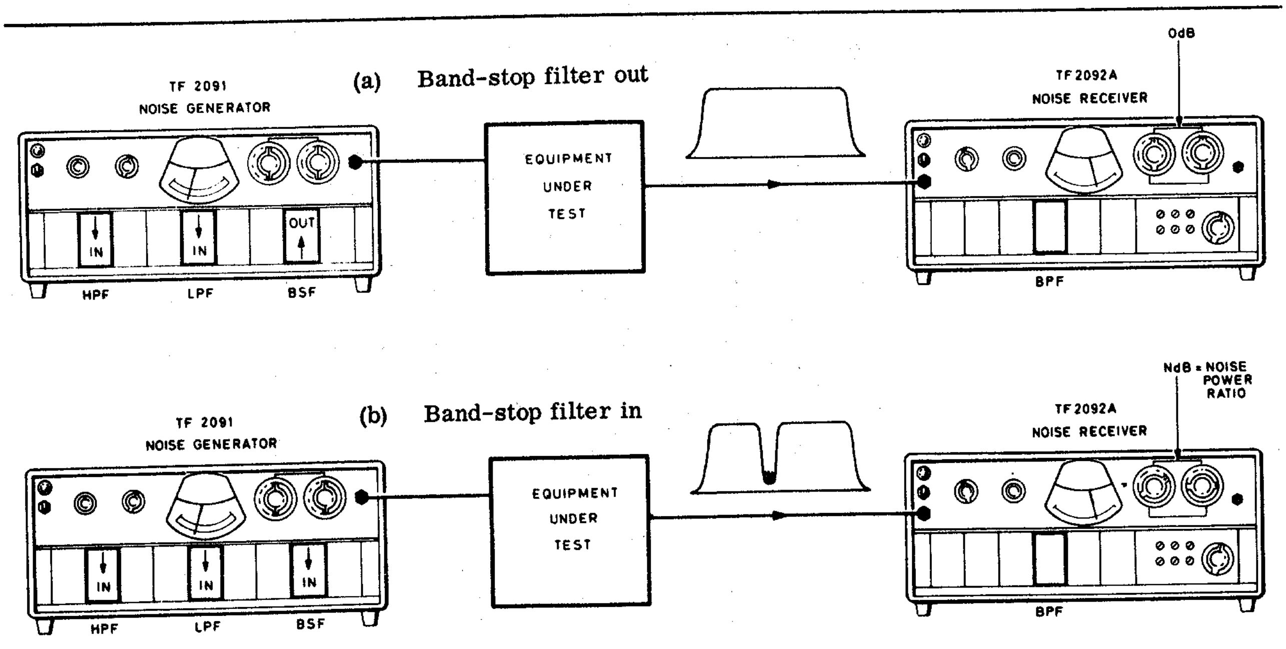


Figure 19-6. NPR Test Set. (Courtesy Marconi Instrument)

If the link NPR is significantly different from either station's loop-back NPR and the link RSL is normal, equipment alignment should be rechecked. Specifically, a transitter must not be aligned with its own receiver for best NPR. It may not match the distant station's receiver as well. If the equipment is all right, a problem may exist in the waveguide or antennas which are causing intermodulation. A check of VSWR would identify any problems in this area. In rare cases, usually on long troposcatter paths, path intermodulation caused by propagation may be the contributing factor. If the equipment NPR is bad, analysis of the data can aid in troubleshooting. If the BINR is also bad, a thermal noise problem is indicated and a check to find the contributing component(s) is in order. More likely, the BINR will be acceptable and a large difference will exist between the NPR and the BINR. This indicates an excess of intermodulation noise and points to a non-linearity in the system as the problem. A clue as to the location of the problem is given by looking at the slots. Problems in the RF and IF circuitry dealing with reflections, RF interference, propagation delay, and non-linear phase characteristics cause excessive noise in the high slot. Amplitude non-linearities in baseband amplifiers, modulators, and demodulators cause noise in the low slot. Since the end product of noise generated anywhere in a system is noise in a channel, it is desirable to convert NPR to per-channel noise. This can be done by applying some correction factors to the NPR. Specifically,

$$\text{S/N Channel} = \text{NPR} + 10 \text{ Log } \frac{(\text{U-L})}{\text{W}} - \text{NLR}$$

U = Low Pass Filter Freq (kHz)

L = High Pass Filter Freq (kHz)

NLR = Loading Level in dBm0

W = Channel Bandwidth in kHz

Channel noise in dBrn0 is equal to 90-S/N

No Of Channels	*Band Limits*		*In-Band Slots*		
	High Pass	*Low Pass*	*Lower*	*Center*	*Upper*
12	12 kHz	60 kHz	27 kHz	40 kHz	50 kHz
24	12 kHz	108 kHz	40 kHz	70 kHz	105 kHz
36	12 kHz	156 kHz	40 kHz	70 kHz	105 kHz
48	12 kHz	204 kHz	40 kHz	105 kHz	185 kHz
60	12 kHz	252 kHz	40 kHz	185 kHz	245 kHz
60	60 kHz	300 kHz	70 kHz	185 kHz	270 kHz
120	60 kHz	552 kHz	70 kHz	270 kHz	534 kHz
240	60 kHz	1052 kHz	70 kHz	534 kHz	1002 kHz
300	60 kHz	1300 kHz	70 kHz	534 kHz	1248 kHz
600	60 kHz	2540 kHz	70 kHz	534 kHz	1248/2438 kHz

Table 19-1. Typical Filter Frequencies for Various Channel Capacities.

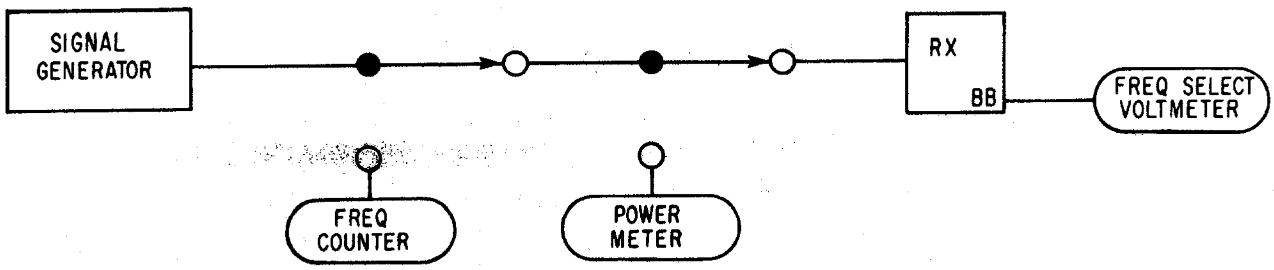


Figure 19-7. FM Quieting Test.

19-3. FM Quieting Curve:

a. General. "Quieting," or reduction of thermal noise level with increasing RSL, occurs in an FM system as a result of the FM demodulation process. The thermal noise response of an FM receiver with varying RSL was discussed in detail in chapter 6. This test allows us to plot the receiver characteristic curve for thermal noise. From this curve, receiver threshold can be determined. The curve can also be used to determine the 20 dB quieting point for the setting of combiners.

b. Test Procedures. The parameter of interest in this test is the per-channel thermal noise. As in the NPR test, we can use 3 kHz measuring slots in the receive baseband to simulate channels. Three slots should be measured, with frequencies as specified in table 19-1. The measurement may be taken with a FSV using a 3 kHz bandwidth. RSL is varied by feeding an unmodulated carrier into the receiver with a signal generator. The noise in the slots is then plotted as a function of RSL (figure 19-8). A typical test setup is shown in figure 19-7.

c. Analysis of Results. A detailed discussion of the receiver characteristics curve can be found in chapter 6. The quieting curve is actually the lower portion of this characteristic curve. The most important parameter that can be derived from this curve is the receiver threshold; therefore, the curve for the highest frequency slot should be used to determine threshold. Extend the linear portion of the curve on the lower end (figure 19-9).

FM threshold is defined as the point where the actual curve is 1 dB different from the asymptote drawn. This value should be recorded on the data sheet. If threshold extension panels are used at the station, quieting curves should be run in all three slots with the extension panels disabled and then a curve should be taken in the high slot with the panel active. The thresholds with and without the extension panel should be recorded and compared.

The receiver 20 dB quieting point can also be derived from the curve for the high slot. The noise in the slot at minimum RSL is used as the reference. The point on the curve which has 20 dB less noise than the reference is the required point. Figure 19-10 applies. The RSL at which 20 dB quieting occurs should be recorded on the data sheet.

19-4. Transmitter RF Frequency Response:

a. General. Response is measured on tropospheric scatter transmitters to ensure that RF bandwidth is sufficient for system loading employed. RF bandwidth on LOS transmitters using IF (70 MHz) modulators may also be measured with this test. Frequency response is measured by using a constant drive level, varying the output RF frequency over the necessary bandwidth, and noting power output versus frequency.

b. Analysis of Results. Recommended RF bandwidths of FM transmitters are given in table 19-2. This table should be used for general information purposes only and not as a standard for all equipment in the Air Force. Consult the appropriate TO or manual for specific equipment bandwidth. Note that required bandwidth is determined by channel capacity, modulation index, and baseband frequency spectrum. An increase in either requires added bandwidth. Narrow bandwidth will result in higher baseband frequencies being penalized with poor S/N ratios. From this, we can see that poor RF transmitter frequency response may result in improper baseband frequency response, high noise, and poor linearity. The transmitter should be tuned such that rated power is obtained with flat response over the usable portion of the response curve. If the transmitter is tuned for maximum power, the response curve may be peaked and narrow with resultant high intermodulation. Also, significant sidebands from higher modulating frequencies will be lost and not available for use in the receiver. When bandpass is allowed to become too wide, the authorized radio frequency spectrum of the broadcast may be exceeded, violating FCC or other regulations. Figure 19-11 shows examples of good and poor frequency response, respectively.

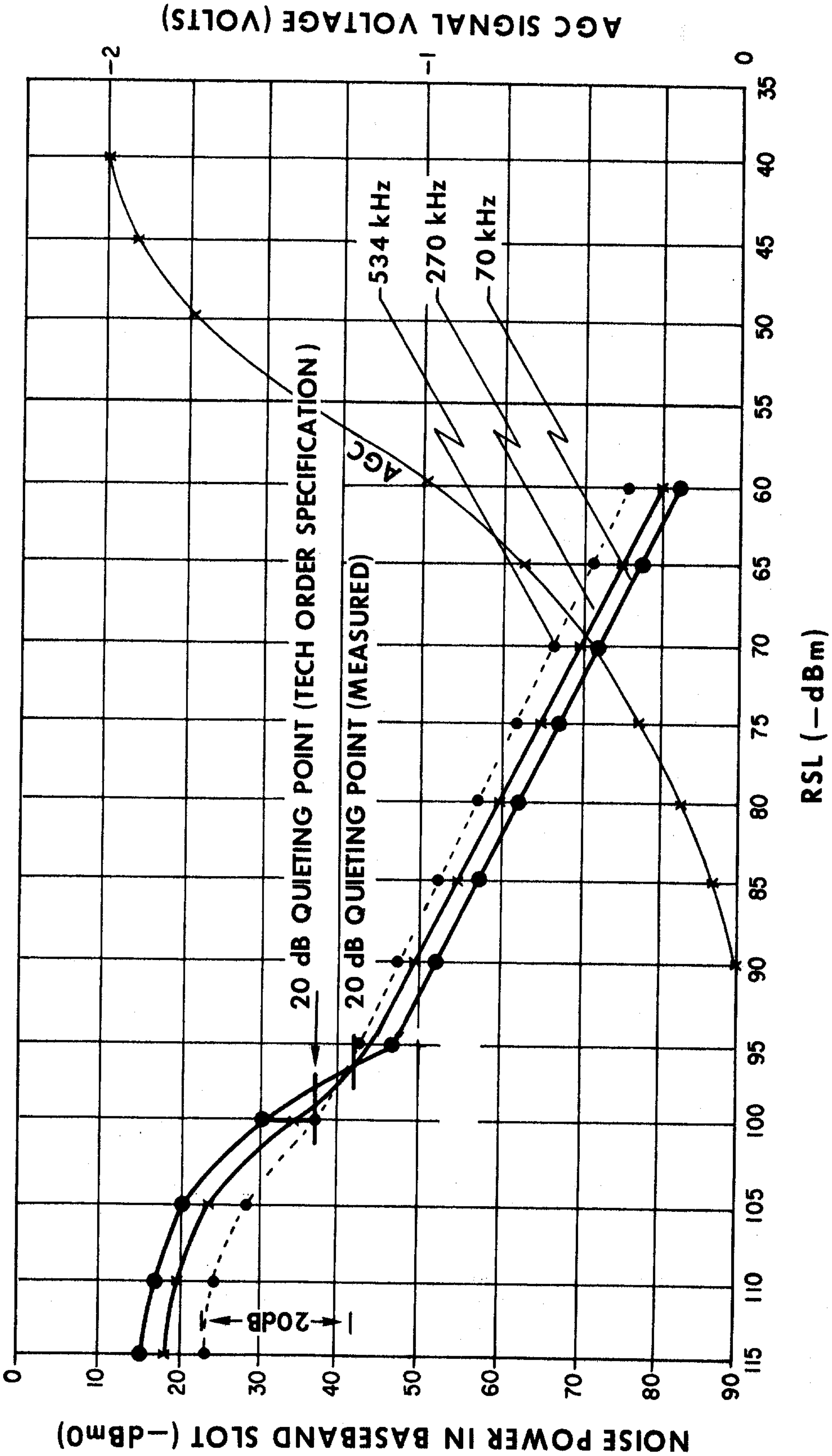


Figure 19-8. Receiver Input RF Signal Level (RSL) (-dBm), AGC, and FM Quieting Curves.

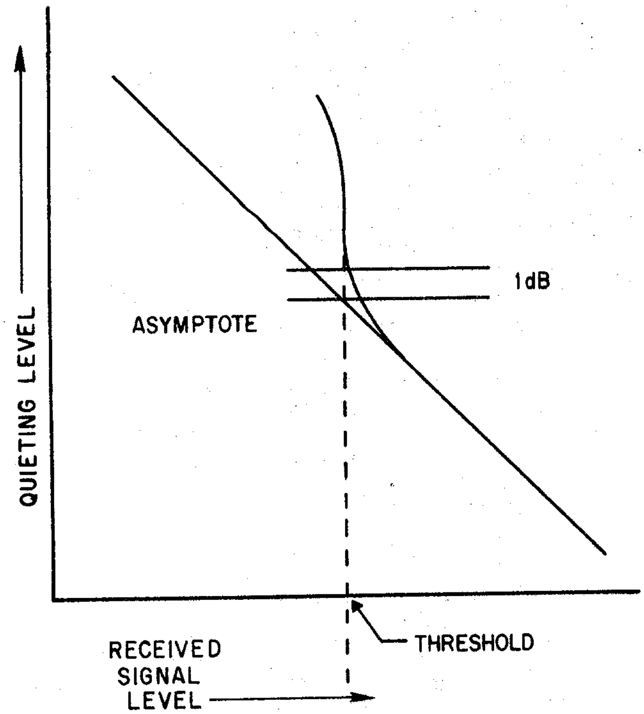


Figure 19-9. Determination of Threshold from Quieting Curve.

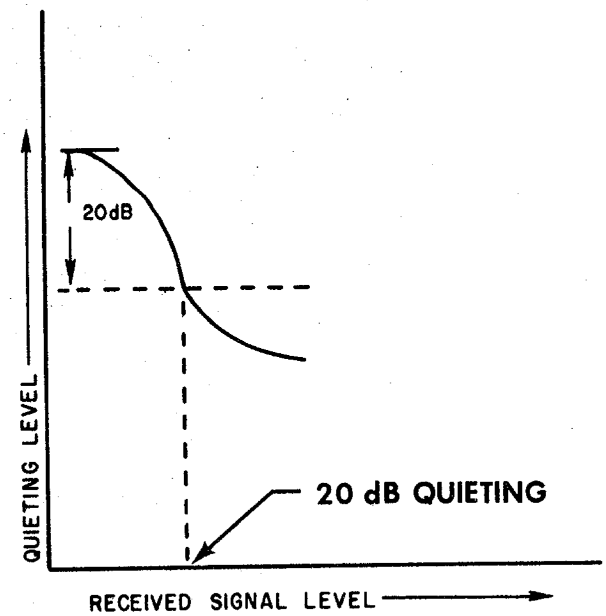


Figure 19-10. Determination of 20 dB Quieting Point.

Channel Capacity	*Modulation Index*	*Baseband Spectrum kHz*	*Recommended Bandwidth*
72	3	12-308	3 MHz
72	6	12-308	6 MHz
72	3	60-360	3 MHz
72	6	60-360	6 MHz
132	3	12-552	6 MHz
252	3	12-1052	10 MHz

Table 19-2. Transmitter RF Bandwidth Reference Chart.

19-5. Receiver IF and Discriminator:

a. General. This test is performed, out of service, on all individual receivers. It ensures proper gain and frequency response through IF amplifiers and proper voltage output from the discriminator. Table 19-2, depicting transmitter RF bandwidth requirements, is also valid for the IF strip of the receiver.

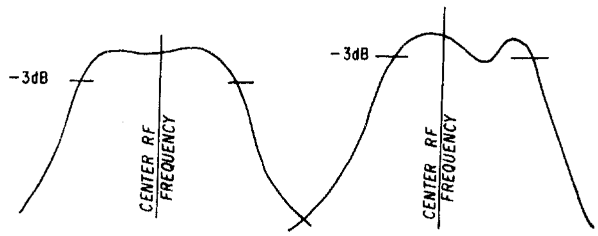


Figure 19-11. Correct and Incorrect RF Frequency Response.

In newer equipment, IFs may be somewhat wider than the examples used. To avoid group delay distortion of the FM signal components, it has become necessary to compensate or remove delay due to phase response in the IF. One method is to widen the IF, moving the phase-shifting problem beyond the needed sidebands. Similar test methods are employed in measuring bandwidth in both transmitters and receivers. When adjustments are required, a sweep generator is used to provide a visual display of total response. This technique speeds up the adjustment process. To plot response curves, the output of the IF may be measured with a spectrum analyzer, RF power meter, or a calibrated crystal detector and oscilloscope. The discriminator data may be plotted using single frequency methods and a DC voltmeter or by sweeping the IF and displaying the curve on an oscilloscope. Figure 19-12 shows a properly tuned IF response and discriminator curve.

b. Analysis of Results. Compare the plotted IF and discriminator response curves with the equipment technical specifications. The 3 dB points on the IF curves should fall at equal distances from center frequency. No spurious oscillations should be present. The discriminator curve should appear as a straight line over the usable IF bandwidth. Discriminator output voltages should normaly be of equal amplitude and opposite polarity at the IF 3 dB points. IF design center frequency should produce zero voltage output, especially if the system is operated at full design capability. Many things can be observed from plotted response curves that are difficult to detect from raw data. Some of the information which can be derived from the IF response curve is IF gain, 3 dB bandpass, response ripple, center frequency, apparent center frequency, and gain at different points of interest. Tune the IF strip to provide maximum gain within the specified bandwidth with the usable area as flat as possible. Response of the IF strip should be checked at periodic intervals and at any time a tube or other component requires changing. After tuning the IF strip, the discriminator curve should be plotted. No other single component in the receiver affects baseband response and intermodulation more than the discriminator.

19-6. FM Modulator Frequency Deviation:

a. General. The amount of frequency deviation of the RF carrier determines the amount of amplitude detected by the discriminator of the distant receiver. If deviation is set too high, the receiver baseband levels will be high; if deviation is set low, baseband levels will be low. The amount of deviation will be determined by equipment design, bandwidth, and channel loading, which are geared to provide the best possible S/N ratio and minimum intermodulation in the receive baseband. The modulated FM carrier will contain numerous sidebands as amplitude of the modulating signal is increased. Higher amplitudes result in additional significant sidebands. Frequency deviation is tested by applying a single sinusoidal wave to the modulator and observing the significant sidebands, which appear symmetrically displayed from the carrier by integral multiples of the modulating frequency on a spectrum analyzer. Power in the carrier will decrease to a minimum (carrier dropout) as amplitude of the modulating frequency is increased and additional sidebands will appear. Other methods are used for determining carrier dropout. Any method is adequate which can detect and measure level in the carrier while disregarding power in the sidebands. Another method employs a crystal-controlled oscillator which is mixed with the carrier; the difference frequency is then routed through a very narrow bandpass filter and the output voltage measured. Output voltage is proportional to power in the carrier frequency. Care must be exercised to ensure that the first dropout is observed and not a subsequent dropout. In either test method, amplitude of the modulating frequency must be set at minimum and increased slowly so that the point where minimum power appears in the carrier can be easily observed.

b. Analysis of Results. Compare measured data with TO or manufacturer's specifications. If minimum carrier power does not occur with the proper input level, modulation sensitivity must be adjusted. The carrier should disappear completely from the display. Power remaining in the carrier at all times indicates distortion or non-linearity in the modulator. This test ensures that power is distributed correctly into the proper number of sidebands. Most of the time, deviation problems appear from faults in the modulator section; however, amplifier, multiplier, and power output stages should not be disregarded when problems persist after troubleshooting the modulator. Tuning in these stages may be varied slightly to provide the most linear output spectrum. Faulty tubes or circuit components may produce noise that causes a distorted output with modulation appearing on the display at points other than those desired. Chapter 5 provides detailed information on FM deviation, the Bessel function, and modulation in FM systems. Close study of that information will prove most helpful in understanding this test and evaluating the results.

19-7. Baseband Frequency and Level Response:

a. General. This test is used to determine the frequency response levels from the transmitter modulator through the receiver demodulator. This test may be performed by looping the modulator to the demodulator in IF or RF loop; it may also be performed in link configuration. Response is measured by inserting a single sinusoidal wave into the modulator and varying

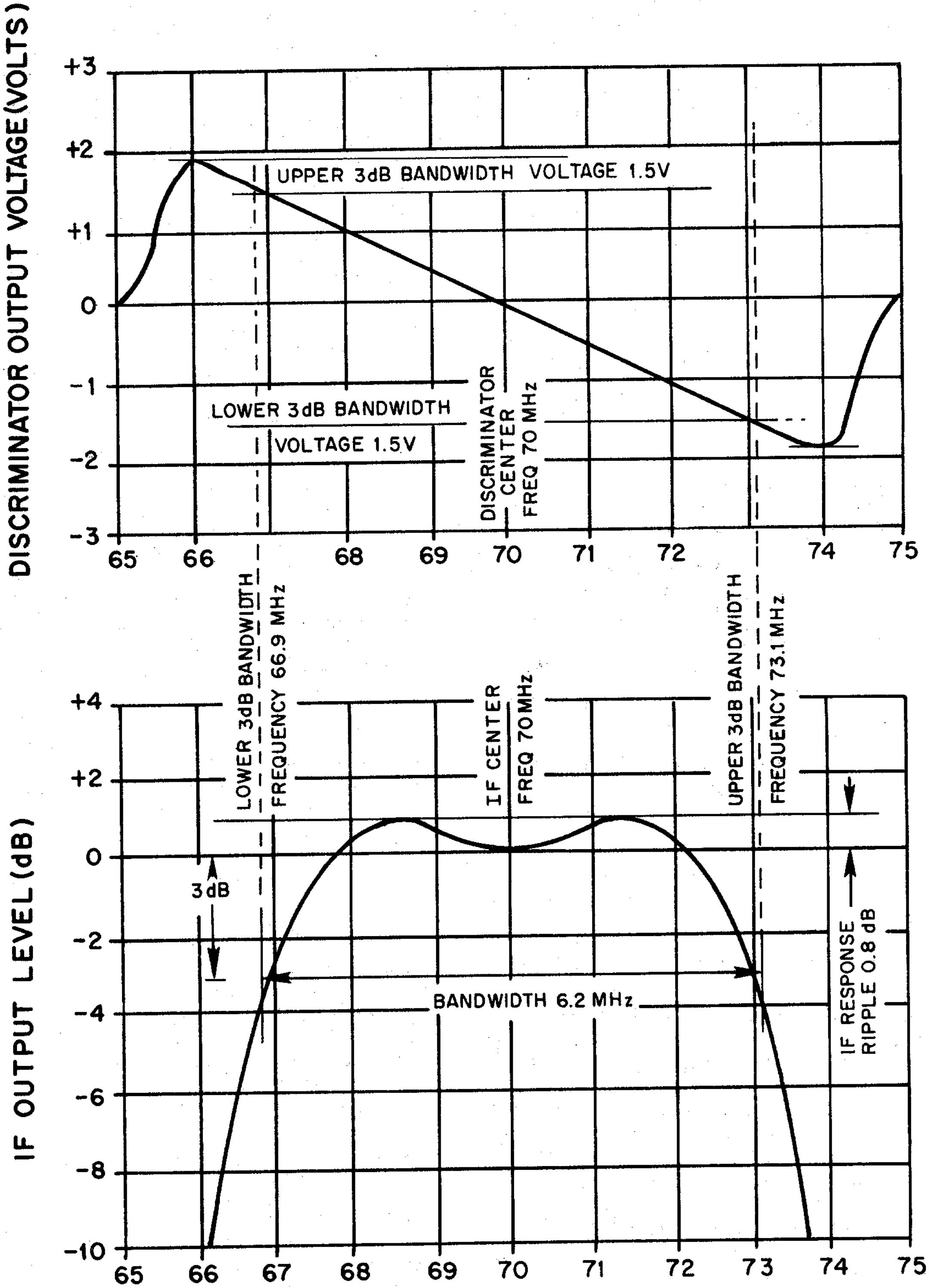


Figure 19-12. Receiver IF Bandpass and Discriminator Characteristics.

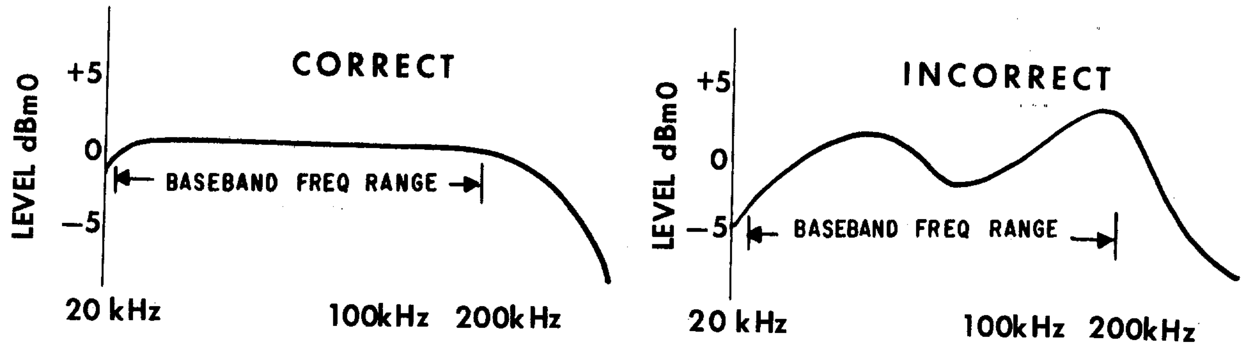


Figure 19-13. Baseband Frequency Range.

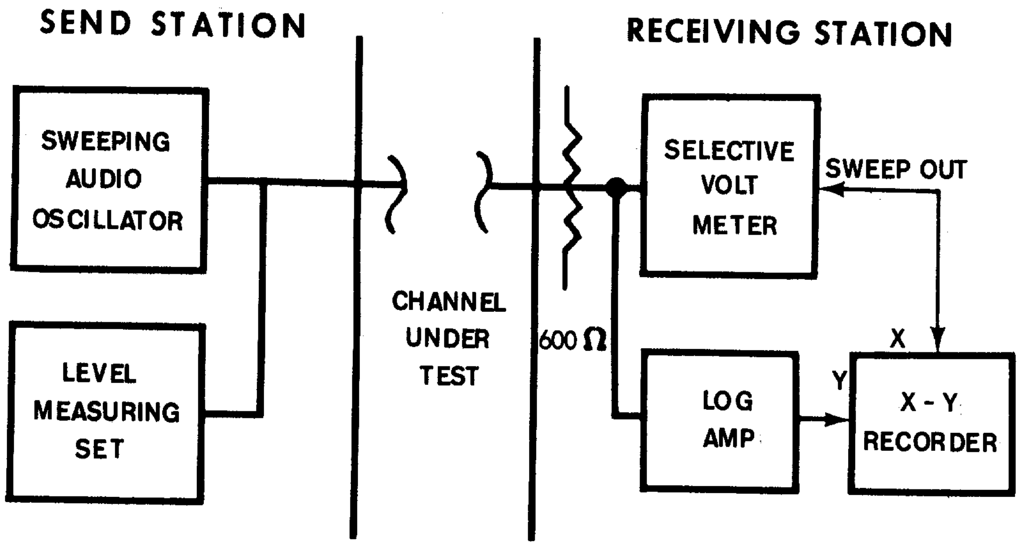


Figure 19-14. VF Frequency Response Test Equipment Configuration.

the frequency (manually or with sweep generator) through the entire baseband while monitoring the level from the demodulator. Ideally, response will remain flat through the entire baseband frequency range.

b. Analysis of Results. Response curves obtained from data recorded should be compared to determine whether a non-linearity exists in either of the transmitters or receivers tested. Comparison checks can usually isolate a level or response problem to the faulty unit. Substitution of the modulator or demodulator can be used for positive fault identification in most cases. After a response problem is isolated to a particular receiver or transmitter, other tests should be performed to identify the unit if spare modules cannot be substituted. Deviation and RF bandwidth may provide useful information in the transmitter; in the receiver, the discriminator of IF response curves may be helpful. Improper input or voltage from the discriminator is very common in baseband response problems. Remember that if adjustments are performed on the modulator or demodulator, not only response but noisepower (NPR) should be checked before returning the unit to service. Figure 19-13 shows examples of good and poor frequency response. A good baseband response will be flat ±0.5 dB over the useful range.

19-8. Frequency Response (VF):

a. General. The VF response test may be performed on any VF transmission path. Its purpose is to measure the amount of attenuation offered by the path at different frequencies within the audio spectrum. The VF response test is sometimes referred to as an "insertion loss test" (test equipment configuration shown in figure 19-14). The test consists of the measurement of the level of a received test signal at each of many different frequencies. The loss at each of these frequencies is then compared to the loss at 1 kHz to determine the frequency response of the channel under test. The results are used to determine whether the multiplex equipment on the VF channel is operating properly. If the circuit has an amplitude equalizer installed, this test will be performed before and after the equalizer to determine whether the equalizer is properly adjusted. The results may also be evaluated by DCA to determine the requirement for amplitude equalization on the circuit.

b. Analysis of Results. The accepted standards are listed as ranges of permissible losses over specified frequency ranges. One example could be a permitted deviation of -1 to +3 dB over a frequency range of 500 to 2800 Hz. This means that within this frequency range, the level must not exceed the limits of 1 dB higher or 3 dB lower than the level at 1000 Hz. A standard of this type is usually accompanied by a second (and possibly third) standard covering a broader frequency range. The second criteria could be -2 to +6 from 300 to 3400 hz. The graph in figure 19-15 shows these example criteria together with a plot of a frequency response curve. The solid straight lines within the graph represent the sample criteria listed above.

19-9. Envelope Delay Distortion:

a. General. As discussed in chapter 4, the phase shift versus frequency characteristics of a voice channel is a primary factor in determining signal distortion. Ideally, one would like to measure this characteristic directly; however, since the phase reference (at the receiving end) required for such a measurement cannot be easily established, another technique has been developed. This technique is to measure the phase shift difference experienced by the two sidebands of an AM signal. The ratio of this difference and the frequency difference of the sidebands $(\frac{\Delta\phi}{\Delta f})$ approximates the slope of the phase versus frequency curve. This slope is termed the envelope delay and the difference in the slope at one carrier frequency relative to another is termed envelope delay distortion.

Before proceeding, it is important to emphasize that envelope delay distortion measurements in themselves have no significance (for example, a measurement of 300 μsec at 1000 Hz and 500 μsec at 3000 Hz does not mean that the 3000 Hz tone would be delayed by 200 μsec relative to the 1000 Hz tone). The value of envelope delay measurements is that they can provide a basis for comparison with other values determined in a similar manner. The results of this test are used to determine whether multiplex equipment on the path is operating properly and may be used by the DCA to determine whether delay equalization is required. When delay equalizers are installed, this test must be performed before and after the equalizer to determine whether the equalizer is properly adjusted.

When a carrier frequency is amplitude-modulated and the resulting signal is passed through a frequency sensitive device (such as a VF channel), the envelope of the signal will be delayed differently than the carrier. If the modulating frequency is low, the sideband frequencies are close enough to assume a linear phase-frequency relationship between the sideband frequencies. By sweeping the carrier frequency over the desired band and measuring the sideband phase difference, the slope of the phase versus frequency curve can be approximated. The envelope delay distortion can be found by subtracting the lowest envelope delay reading from the other delay readings obtained. The lowest reading is considered to be the point of zero relative envelope delay. Zero relative envelope delay normally occurs near 2000 Hz on most VF multiplexed channels.

To understand the reasons for the various equipment configurations which may be used for this test, it is necessary to understand the operating principles of the measuring set employed to make these measurements. The delay measuring set shown in figure 19-16 comprises two sections - a transmit section and a receive section. These sections can be manufactured separately or as one unit.

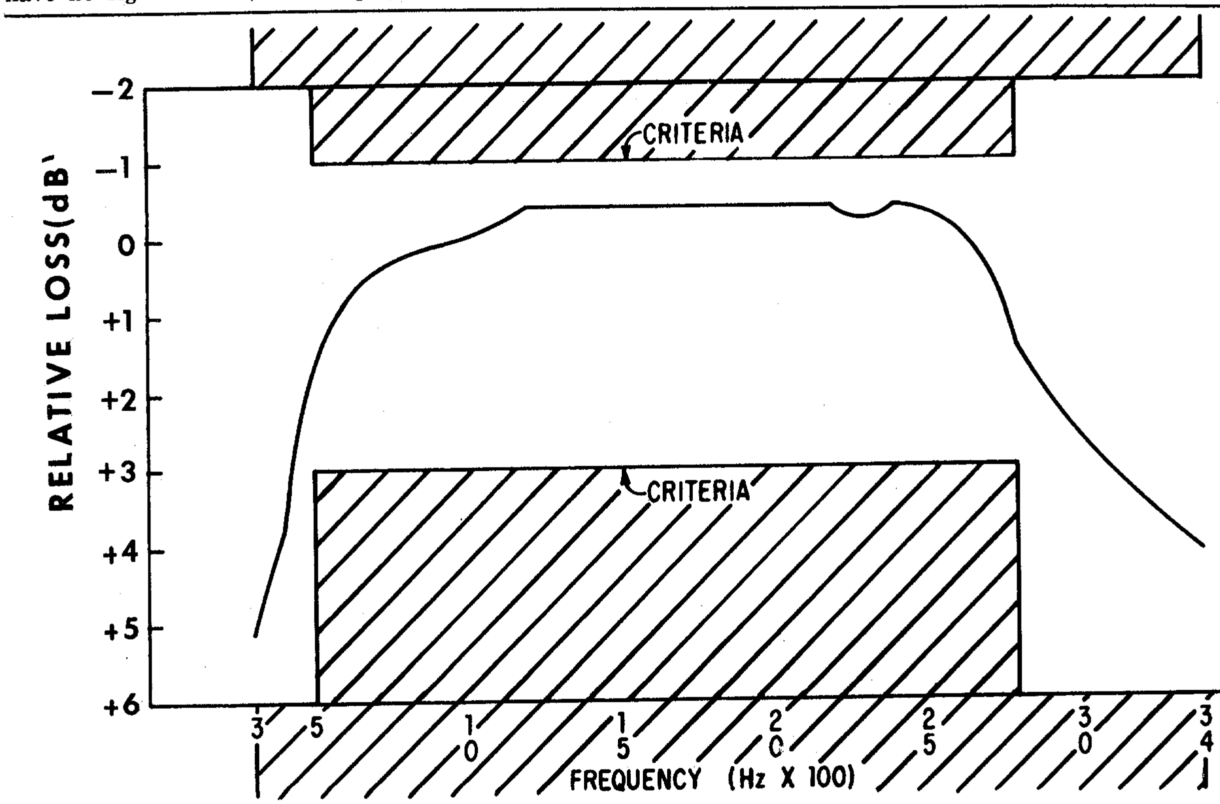


Figure 19-15. Frequency Response Graph.

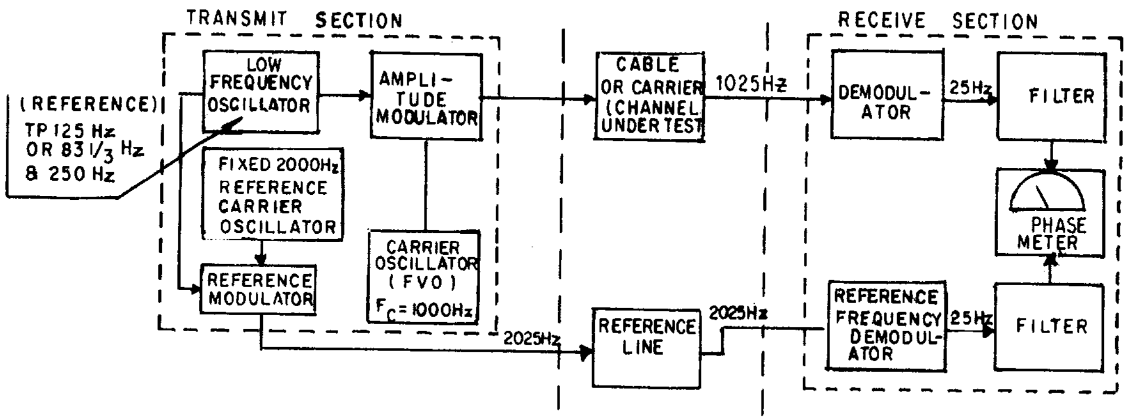


Figure 19-16. Block Diagram of a Delay Measuring Set.

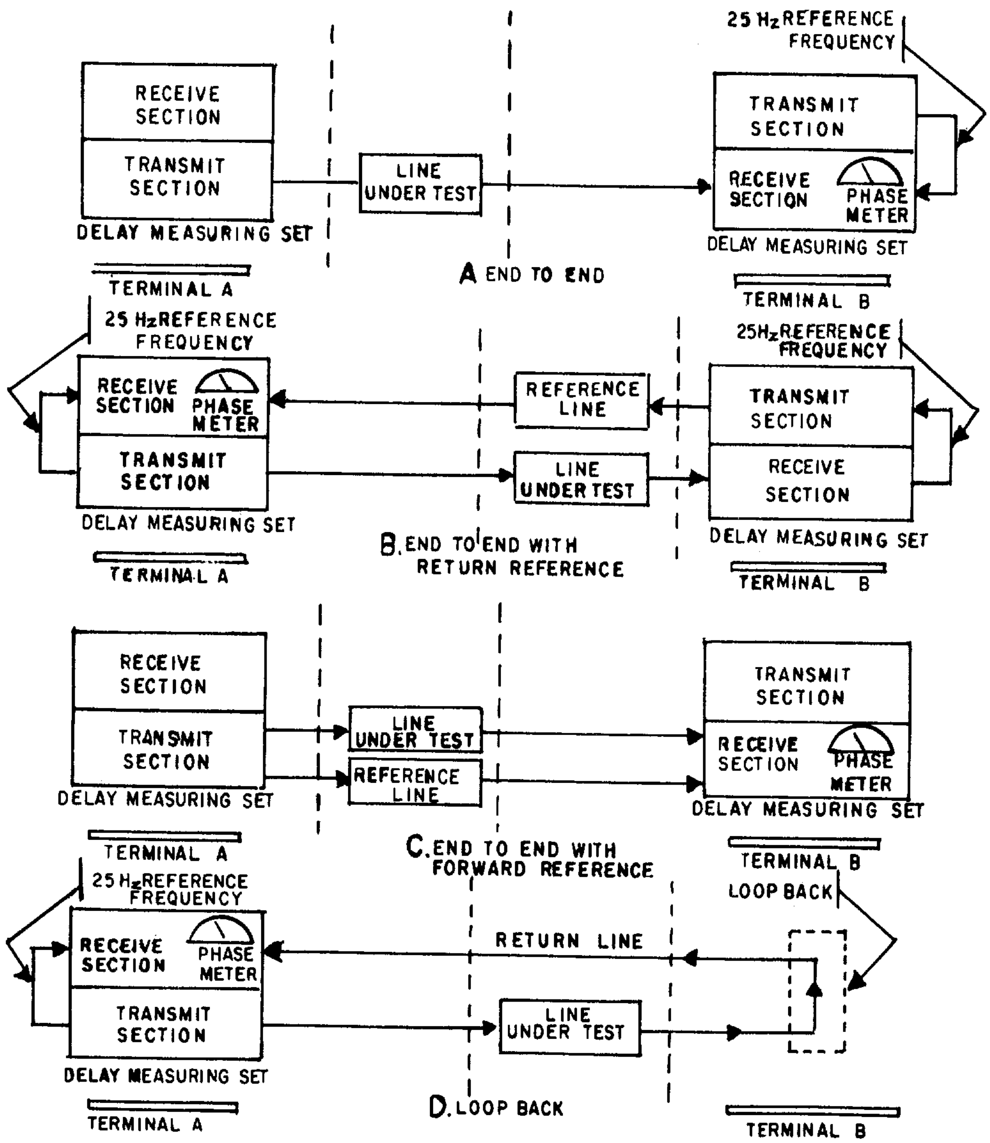


Figure 19-17. Delay Measurement Test Methods.

b. Transmit. The transmit section is an oscillator (carrier generator) with a variable frequency output in the 200 Hz to 500 kHz range. This generator provides the carrier frequencies for the channel under test. Relatively low modulating frequencies are provided separately to amplitude-modulate the carrier frequencies. These modulating frequencies, normally 25 Hz, 83 1/3 Hz, or 250 Hz, also modulate a 2000 Hz carrier reference frequency which, in the particular mode of operation shown, is sent over the reference line to the distant terminal. The demodulated output from the reference line is used by the receive section to obtain the delay on the carrier frequency of the cable or carrier channel under test.

c. Receive. The carrier and reference frequency inputs are passed through separate demodulators. Filters are also provided on the output of the demodulators. The recovered modulating frequencies are fed to the phasemeter circuit, where they are compared. An indication of delay in milliseconds or microseconds is presented on the meter. The meter circuit is usually equipped with controls that permit the meter to be set for a broad scale reading (milliseconds) or a narrow scale reading (microseconds). In addition to the phasemeter, most delay sets also provide a frequency counter to indicate the transmit and receive carrier frequencies.

d. Test Procedures. There are four standard methods of connecting the delay measuring set to measure delay distortion: end-to-end, end-to-end with return reference, end-to-end with forward reference, and loop-back. It may be necessary to have an additional full-duplex line to allow coordination between operators while making these measurements.

(1) End-to-End. Figure 19-17A shows the use of the delay measuring set for end-to-end application. This mode is one of the most commonly used as it requires only one line. The actual measurement is made at the receiving end. It is made by synchronizing the modulation oscillator (25 Hz, 83 1/3 Hz, or 250 Hz) in the receive test set to precisely the same phase relationship as the modulated frequency of a reference carrier (2000 Hz) transmitted from the distant station for that purpose. Once the two oscillators are synchronized, delay measurements can be made by sending the modulation frequency on different carrier frequencies in the bandpass. Some delay measuring sets offer a "check" position which can be used after each measurement to determine whether any drift in modulation oscillator phase relationship has occurred while making the measurement.

(2) End-to-End with Return Reference. Figure 19-17B shows the delay test set connected for end-to-end with return reference measurement. This configuration allows measurement to be made at the transmitting end. A full-duplex channel is needed so that one side can be used as the return reference line.

The modulated variable carrier frequency is transmitted through the VF channel under test. The phase-shifted modulating frequency is removed from the variable carrier frequency and remodulated on a fixed reference frequency carrier and sent back to the transmitting station on the return channel. This fixed carrier is usually the 2000 Hz fixed reference carrier frequency. The retransmitted, phase-shifted, modulating frequency is again demodulated at the original transmitting end, fed to the phasemeter, and compared with the original modulating frequency. Since the delay encountered on the reference line is constant, all time differences noted on the phasemeter would be caused by the line under test reacting to the variable carrier frequency.

(3) End-to-End with Forward Reference. Figure 19-17C shows the test set configuration for end-to-end with forward reference measurement. This mode requires the delay test set transmitter to have two simultaneous-modulated outputs (carrier and reference frequency) and the test set receiver to have two inputs. The primary difference between this mode and the end-to-end mode is that instead of the receive station synchronizing its modulation oscillator to the transmitting station's oscillator, it compares the modulation frequency of the line under test with the modulation frequency on the fixed reference frequency from the reference line. This mode is possible the most accurate since it does not require synchronization and the measured signal uses only one line.

(4) Loop-Back. Figure 19-17D shows the test set configuration for loop-back application. Note that only one test set is used. This is a very doubtful method of measurement because you must assume that the line characteristics are evenly divided between the send and receive legs of the circuit under test. This is normally true, but not always! If a serious fault is encountered, there is no way of determining which line, transmit or receive, is causing it. In theory, the function of the delay measurement set at the transmit station is the same as "end-to-end with return reference." The only difference is that the receive station does not have a test set and the line is looped back to the transmit station. It is recommended that this configuration be used only when the lack of test equipment or facilities prevents one of the three previously discussed methods from being followed.

When the test is being conducted, the 2000 Hz frequency will be transmitted and the delay reading at 2000 Hz will be used as the reference delay. Normally, an automatic sweep will be performed to give a "relative delay of all frequency in the channel to the 2000 Hz reference frequency."

e. Analysis of Results. The standards for envelope delay distortion are given as a maximum amount of relative delay over a specified bandwidth. For example, the standards for one circuit could be:

500-3000 Hz - 3000 microseconds
600-2600 Hz - 1500 microseconds
1000-2600 Hz - 500 microseconds

Comparison of the measurements to the accepted standards is easily seen when plotted on a graph (figure 19-18). On this graph, the envelope delay has been plotted and the straight lines represent the standards shown above. In this instance, the circuit shown is out-of-tolerance at certain frequencies (such as 600 to 775 Hz).

19-10. Frequency Translation:

a. General. The frequency translation test may be performed on any VF path which uses VF multiplex equipment of any type or traverses any radio system. It is not necessary to perform this test on a metallic path (such as a cable pair). The test will determine whether

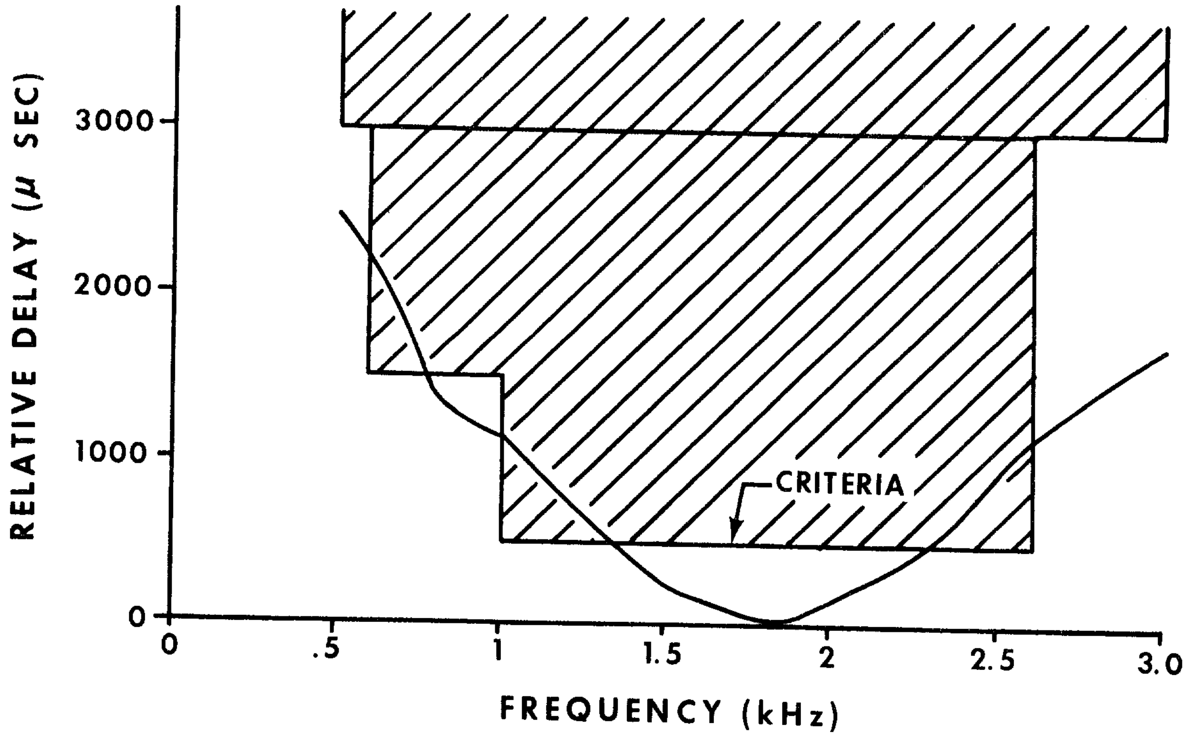


Figure 19-18. Envelope Delay Graph.

there is a frequency error present in any of the carrier frequency oscillators in the multiplex. If a carrier oscillator is off frequency, the received signal will also be shifted in frequency. Small errors of a few Hertz are not noticeable on speech channels but digital data transmissions are extremely sensitive to any frequency error. A 16-channel VFCT system, for example, would be quite distorted by a frequency error of just 10 Hz on the path between the send and receive stations.

Measurements of this unwanted change in the frequency of the signal by the path is called a frequency translation test. It is accomplished by placing an audio tone on the circuit to be tested, then measuring the frequency of the tone at each end of the circuit. Any difference between the two readings is the frequency translation caused by the path. (It must be remembered that if a channel is looped-back for this test, chances are that the oscillator causing an error on a send channel will cause an equal-but-opposite error on the receive channel and thus the error will not be revealed. For this reason, this test should never be attempted on a loop-back over the same system.)

b. Test Procedures. Figure 19-19 shows the test equipment configuration normally employed for this test. The oscillator is terminated by the send line and bridged with a frequency oscillator at the send station. The oscillator is adjusted to exactly 1000 Hz. The receive channel is terminated into a frequency counter. Both stations simultaneously read the frequency on the tone. The frequency counters must be watched for three consecutive counts to ensure a stable reading. The sending station will pass its reading via an order wire to the receive station where the operator will record both readings. Many frequency counters are sensitive to IPN and will give inaccurate or unstable readings when connected directly to a receive channel. This is especially true on long-haul tandem-link circuits. When this problem occurs, an alternate equipment configuration must be employed to isolate the frequency counter from the receive channel. One example is shown in figure 19-20.

The send configuration remains unchanged while the receive channel is terminated into the "X" input of an oscilloscope. The output of an audio oscillator is terminated into a frequency counter and bridged into the "Y" input of the oscilloscope. The frequency of the oscillator is varied slowly until a stationary line (at 45° or 135°) appears on the oscilloscope screen. A stationary line indicates that the frequency of the oscillator is exactly the same as that of the incoming signal. This frequency may be read on the frequency counter.

c. Analysis of Results. The send frequency and receive frequency are compared and the difference is stated as the frequency translation or change in audio frequency. This number of Hertz is compared to the "maximum allowable change in audio frequency" criteria for that circuit. For example, if the send station measures 997.6 Hz and the receive station measures 996.4 Hz, the frequency translation for that path would be 1.2 Hz. If the tolerance for this circuit was ± 5 Hz, it would be satisfactory. If the circuit is out-of-tolerance, the test must be repeated over shorter segments of the circuit, if possible, and, when the faulty link is isolated, the master oscillator frequencies on all multiplex equipment on that link must be measured.

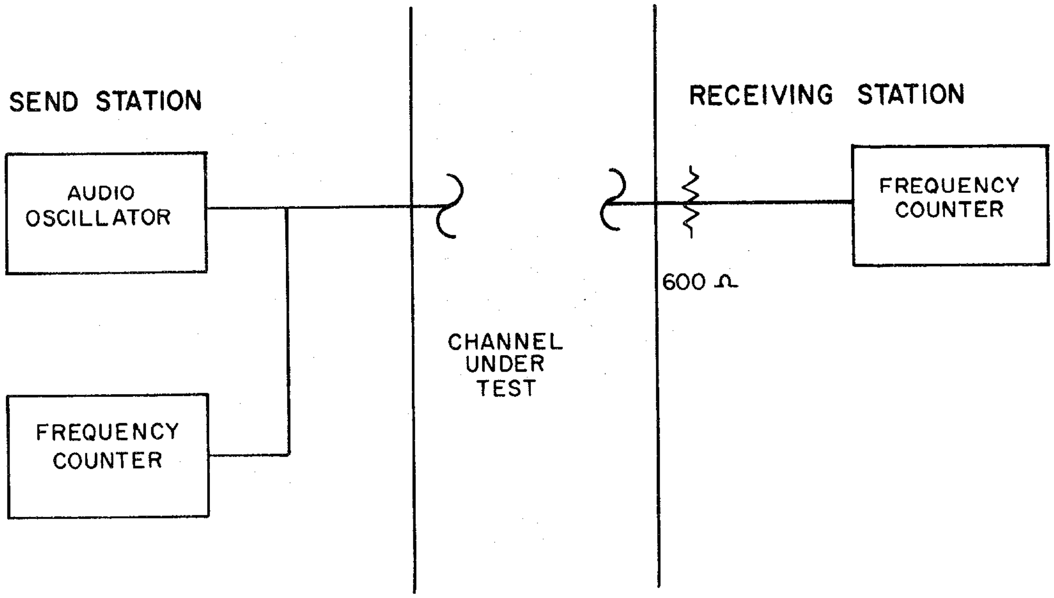


Figure 19-19. Frequency Translation Test Equipment Configuration.

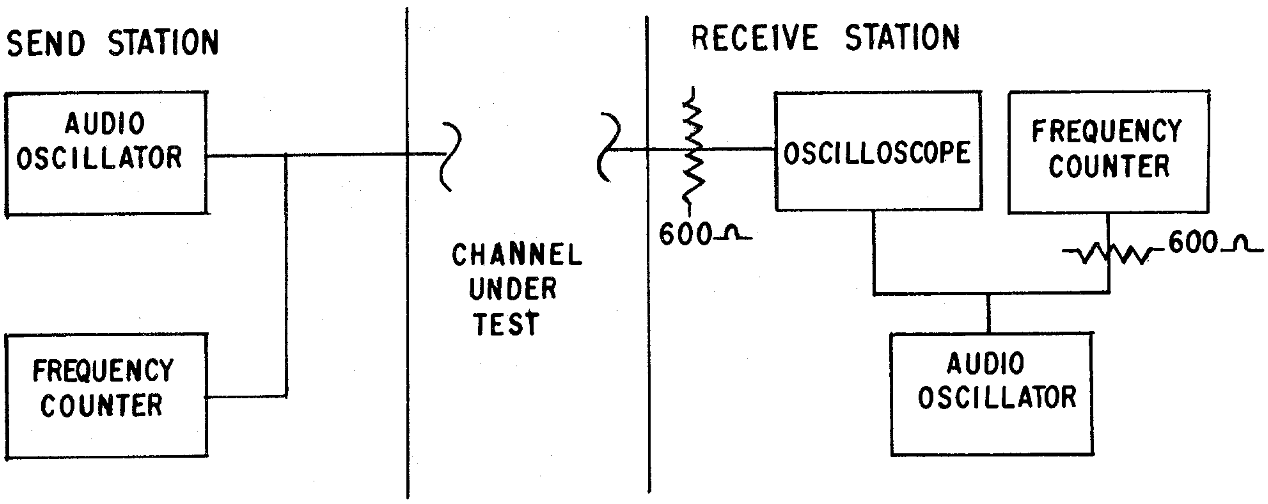


Figure 19-20. Frequency Translation Alternate Equipment Configuration.

The "maximum change in audio frequency" criteria may not apply in certain instances. When there is more than 1 Hz translation on a channel, FSK digital data systems often exceed the maximum permitted amount of bias distortion. In this instance, the frequency translation must be corrected even though it may be within published standards.

19-11. Phase Jitters:

a. General. The phase jitter test may be performed on any VF path which employs VF multiplex or traverses a radio system. It is not necessary to perform this test on a metallic path. The purpose of the test is to measure any rapid incremental changes in the phase of a single frequency transmitted on a VF path. This is accomplished by transmitting a test tone over a VF path and, at the receive station, comparing the phase stability of this tone to that of a locally generated tone. While phase instability on a path will have very little effect on speech circuits, it does degrade the performance of phase-shifted keyed (PSK) VFCT equipment as well as high-baud-rate digital data circuits. While specific techniques have been developed to measure this parameter, little is known about the cause. Whenever phase jitter on a circuit becomes excessive, it should be thoroughly investigated and all conclusions carefully recorded.

b. Test Procedures. Figure 19-21 shows the equipment configuration required for this test.

(1) The oscillator is terminated by the send line and adjusted for 1000 Hz at the proper output level at the send station.

(2) At the receive station, the local oscillator output is terminated into the sync input of the oscilloscope and bridged into the vertical input.

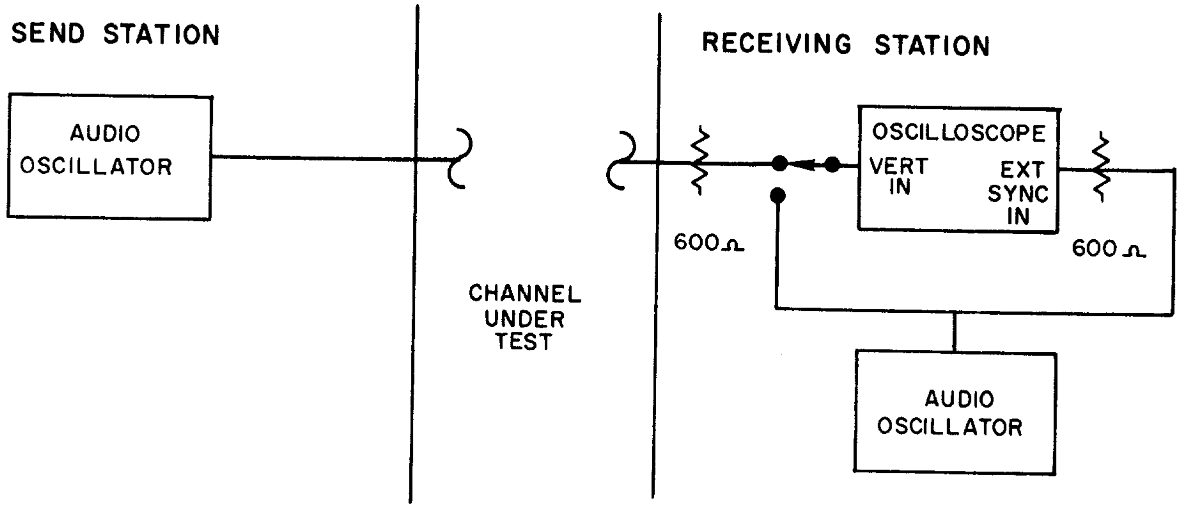


Figure 19-21. Phase Jitter Test Equipment Configuration.

The oscilloscope is then adjusted to display one cycle spanning at least 36 grid divisions. If the oscilloscope has a calibrated time base, .1 ms/cm may be used.

(3) The local oscillator is then disconnected from the vertical input of the oscilloscope and the receive channel is terminated there in its place. The local oscillator (still connected to the oscilloscope sync input) is adjusted slightly until the pattern of the received tone remains stationary. This adustment is sometimes difficult.

(4) Phase jitter will appear as a horizontal "smearing" of the waveform (figure 19-22).

c. Analysis of Results. The amount of phase jitter is computed by measuring the width of the smear at the zero cross-over (horizontal axis) and applying this in this formula:

$$\text{Peak-to-peak phase jitter} = \frac{\text{width of smear (in grid divisions)}}{\text{cycle length (in grid divisions)}} \times 360^\circ$$

For example, if the smear is two grid divisions wide and the cycle covers 72 grid divisions, the phase jitter would be 10°. More accurate measurements can be accomplished using the "X5 magnifier," if available on the oscilloscope, or by initially adjusting the oscilloscope so that one cycle covers more than 36 grid divisions. The formula will still apply as long as the X5 magnifier is linear. The computed phase jitter in degrees may be directly compared to the published standard for that channel. In the example, the 10° phase jitter would be within tolerance if the standard is 15°. This standard of 15° is interim, as the effects of phase jitter on communications systems have not been fully determined.

19-12. Harmonic Distortion. This test may be performed on any VF transmission path. Its purpose is to determine whether any appreciable non-linearities exist on the path. The non-linear operation of devices (such as amplifiers) will cause harmonic distortion. This test is performed by transmitting a 700 Hz tone at -10 dBm0. On the receive side, the 700 Hz tone is notched out and the total harmonic distortion is measured directly in dB or in %. If the total distortion is greater than 1% (40 dB), then further investigation with a FSV is required to determine if spurious or high level harmonics are present. Equipment configuration is shown in figure 19-23.

19-13. Intermodulation Distortion:

a. General. The intermodulation distortion test can be performed on any VF transmission path. Its purpose, as with the harmonic distortion test, is to measure the degree of non-linearity on a transmission path. The test involves placing two discrete tones on the path simultaneously. Any non-linearity on the transmission path will cause these two tone frequencies to beat together and create sum and difference frequencies and harmonics. The receive station then measures the level of the intermodulation products and harmonics of the two frequencies.

The first order intermodulation products are the fundamental frequencies (F1 and F2). The second order products are (F1+F2) and (F1-F2). Third order products are created when the first and second order products intermodulate. Some of the further order products, being sum and difference frequencies, will appear in the audio spectrum and will be measured. This differs from harmonic distortion in that harmonics are multiples of the fundamental frequency.

The test equipment normally employed for this test is a two-tone generator that is adjusted to produce frequencies (that is, 1000 Hz and 1400 Hz) at the same level. At the receive station, the line is terminated into a selective level measuring set. The level of the fundamental frequencies is measured and recorded (they should be of identical level); then measure the level of the intermodulation products with a FSV at each of the following frequencies: 400, 600, 800, 1800, 2400, and 3400 Hz.

$$IM = \sqrt{\frac{E^2_{400} + E^2_{600} + \cdots}{E^2_{1000} + E^2_{1400}}} \times 100\% \text{ or } = \sqrt{\frac{mW_{400} + mW_{600} + \cdots}{mW_{1000} + mW_{1400}}} \times 100\%$$

$$ID(\%) = 100 \text{ Antilog} \left[\frac{ID(dB)}{20}\right]$$

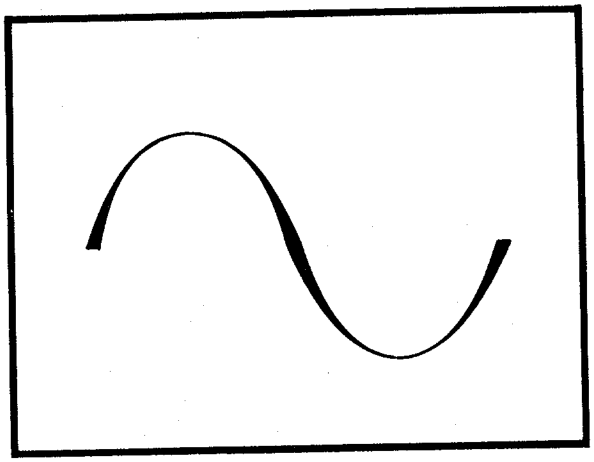


Figure 19-22. Oscilloscope Presentation Showing Phase Jitter.

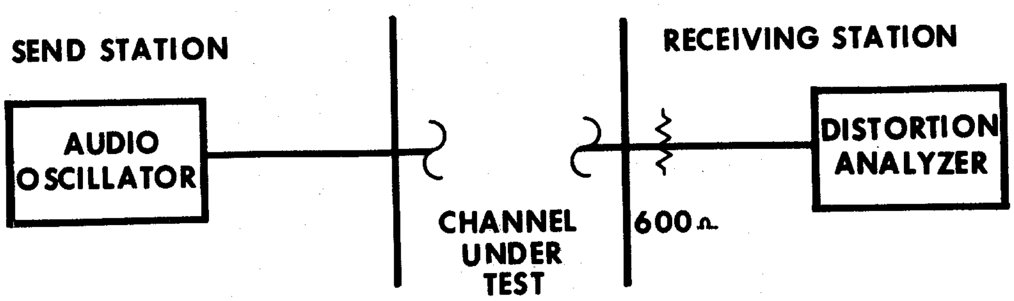


Figure 19-23. Harmonic Distortion Test Equipment Configuration.

b. Analysis of Results. If the distortion is greater than 2%, attempt to isolate the cause of distortion by taking a reading of the distortion products at the group, supergroup, and baseband point in both the transmit and receive directions.

19-14. Voice Channel Intelligible Crosstalk. This test measures the total amount of intelligible crosstalk within a VF channel from all causes. Intelligible crosstalk may be caused by an impedance irregularity in the circuit or capacitive and inductive coupling between disturbing and disturbed channels. These conditions can occur at any point in a communications system, particularly in the distribution frame, wiring, and cabling.

a. Radio Link Far End Crosstalk (Figure 19-24):

(1) At the near end, tune the audio oscillator to 1000 Hz and adjust outut level to obtain 0 dBm0 at the VF patch and test board.

(2) At the far end, connect the noise measuring set, adjusted to 3 kHz flat weighting, to channel 1. Adjust sensitivity controls for a convenient reading on the instrument meter.

(3) At the near end, intermittently disconnect and reconnect the 1000 Hz tone from channel 6 or 7 (disturbing channel). At the far end, record the noise levels and difference and note if the measured noise level of channel 1 (disturbed channel) changed by 1 dB or more.

(4) Maintaining channel 6 or 7 as the disturbing channel, repeat step 3 for channel 2 or 3 and/or 10 or 11 in each group.

(5) At the far end, adjust the selective level measuring set for narrow bandpass and measure the level of the 1000, 2000, and 3000 Hz crosstalk signal in all channels that exhibited a difference of 1 dB or more in steps 3 and 4, above. Record these levels in dBm

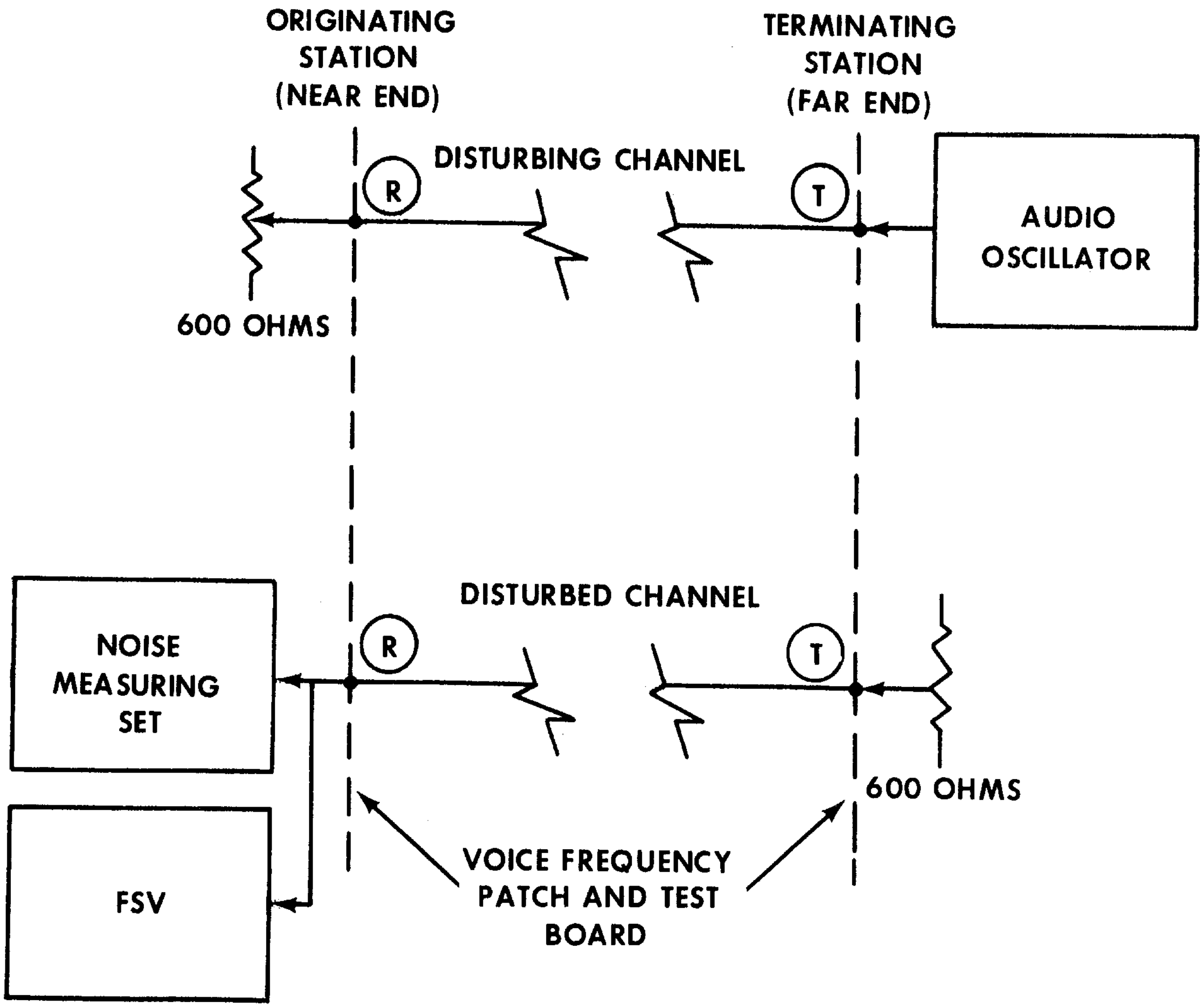


Figure 19-24. Voice Channel Far End Crosstalk.

absolute. If possible, photographically record any significant VF channel crosstalk frequency spectrums displayed on the spectrum display unit.

b. Radio Link Near End Crosstalk (Figure 19-25):

(1) Connect test equipment to channel 6 or 7 of the lowest group as indicated on figure 19-24. All tests are made at one terminal by sending a 1000 Hz tone into the modulator input of channel 6 or 7 and receiving with the noise measuring set (3 kHz flat weighting) and the selective measuring set at the demodulator output of the various channels, as directed. In every case, channel 6 or 7 and the other channel being measured shall be terminated in 600 ohms.

(2) At the near end, tune the audio oscillator to 1000 Hz and adjust output level to obtain 0 dBm0 at the VF patch and test board.

(3) At the near end, connect the noise measuring set, adjusted for 3 kHz flat weighting, to channel 1 (disturbed channel). Adjust sensitivity controls of the selective level measuring set to obtain a convenient reading on the instrument meter.

(4) At the near end, intermittently disconnect and reconnect the audio oscillator from channel 6 or 7 (disturbing channel). Record the noise levels and difference and note if the reading on the meter of the noise measuring set changed by 1 dBm or more.

(5) Maintaining channel 6 or 7 as the disturbing channel, repeat steps 3 and 4 for channel 2 or 3 and 10 or 11 of each group.

(6) At the near end, adjust the selective level measuring set for narrow bandpass and measure the level of the 1000, 2000, and 3000 Hz crosstalk signals in all channels that exhibited a difference of 1 dB or more in steps 3, 4, and 5. Record these levels in dBm absolute. If possible, photographically record any significant VF channel crosstalk frequency spectrums displayed on the spectrum display unit.

c. Analysis of Results. The crosstalk data should be analyzed and evaluated for compliance with specifications. Crosstalk is usually traceable to improper filtering in the multiplex or high levels on channels.

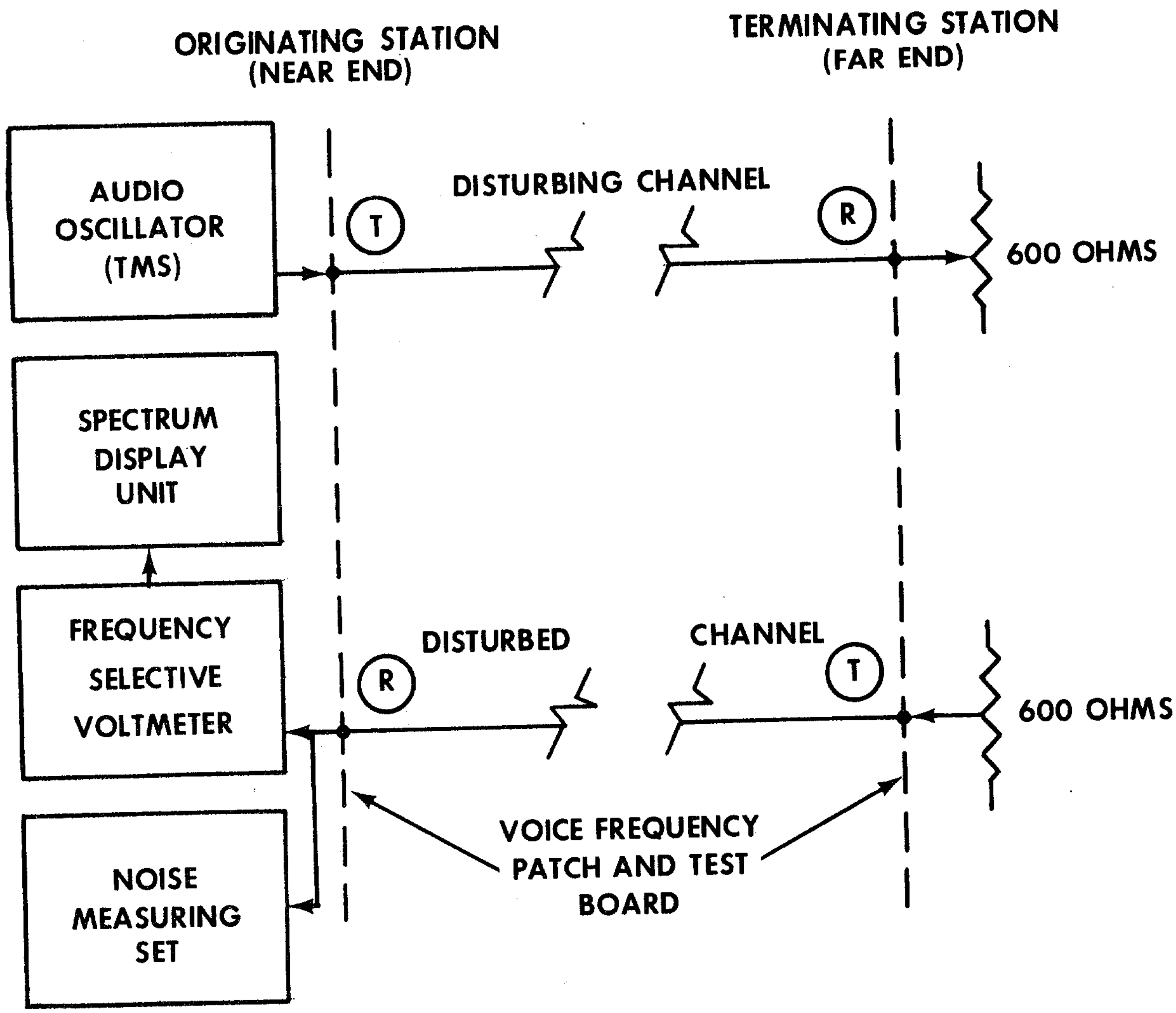


Figure 19-25. Voice Channel Near End Crosstalk.

Chapter 20

SPECIAL EVALUATION TESTS

20-1. General. The special evaluation tests covered in this chapter are those which require unusual equipment or techniques. Particular sites will be able to perform the VSWR tests routinely by virtue of waveguide directional couplers already installed. The other tests are normally performed by Scope Creek or wideband teams on a one-time basis. Data and specifications from these tests are frequently referenced in the text and an understanding of their derivation is desirable.

20-2. Noise Figure:

a. General. In chapter 4, noise figure was discussed as an indicator of the noise added by a device. In wideband systems, the noise figure of the receiver is the limiting factor in noise performance. The noise figure of any multistage amplifier is essentially determined by the first stage (providing the gain of the stage is at least 10 dB); hence, a noise figure measurement performed on the RF amplifier in a receiver determines the receiver's noise figure. (This is why low-noise RF amplifiers are important.)

b. Test Procedures. There are several methods used to measure the noise figure of FM systems; most of them employ the same basic technique. The device under test is terminated and the output level is measured (in receivers, the output must be measured before limiting). A calibrated noise source is then fed into the receiver and the output level is remeasured. The ratio of the second level to the first defines a "Y-factor." The test procedure is shown in figure 20-1.

The Y-factor can be converted to noise figure through graphs supplied with the noise source. The graphs correct for the "noise temperature" of the source, since the measured noise figure depends on the amount of noise injected for the test. Noise figures of front-end amplifiers are usually corrected to a noise temperature of 290°K. The noise figure may be conveniently measured with an automatic noise figure meter (such as the Hewlett-Packard 340B). The method is essentially the same as described above, except that the operation is automatic and the noise figure is displayed directly on the meter.

c. Analysis of Results. The meaning of the noise figure depends on the point in the system at which it is measured. The basic noise figure usually quoted is the value measured into the first receiver stage. The noise figure at any point before that stage is the basic figure, plus the loss in decibels between the two points. For example, consider a system with a 9 dB noise figure and a waveguide run with a 3 dB loss. The noise figure at the antenna would be $9+3=12$ dB (figure 20-2).

One other point must be considered in interpreting a noise figure measurement in a receiver. Usually, the excess noise is injected into a preselector or tuned RF amplifier which will reject the image frequency; however, there are some noise figures measured directly into the receiver mixer. (In this case, noise at the image frequency will be heterodyned to the IF and twice as much noise would be effectively put in as would be calculated.) This makes the noise figure appear 3 dB better than it actually is; hence, any noise figure measurement made directly into the mixer stage must have 3 dB added to it to be correct. These noise figures are sometimes referred to as double sideband noise figure (into mixer) and single sideband noise figure (into preselector). A typical noise figure for a conventional receiver is 11 dB, measured at the preselector. If a state-of-the-art transistor RF amplifier is used, an 8 dB figure can usually be obtained. Parametric and tunnel diode amplifiers have noise figures of 2 dB and below.

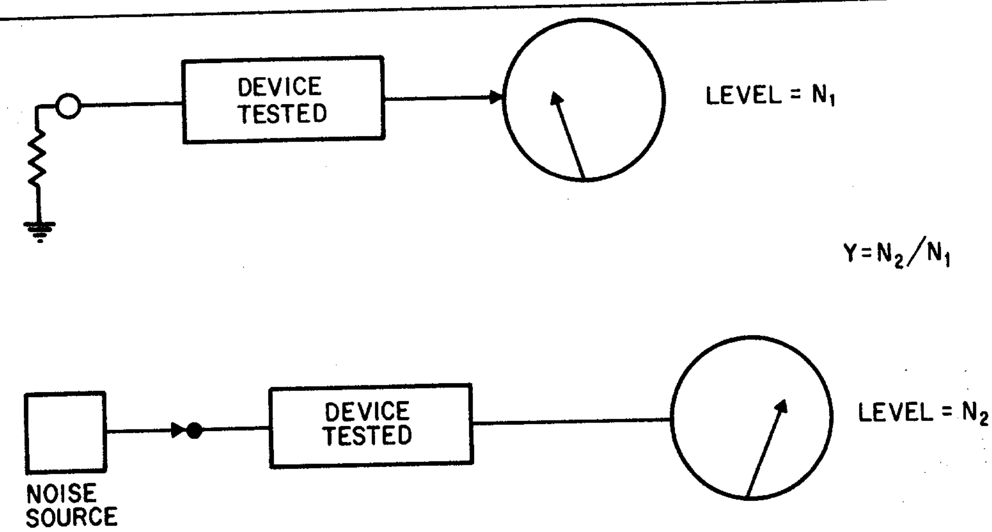


Figure 20-1. Noise Measurement.

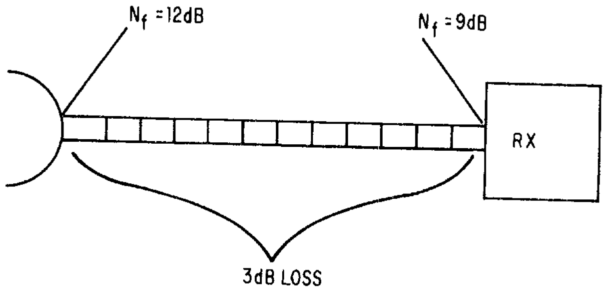


Figure 20-2. Calculation of Noise Figure at Antenna.

As discussed in chapter 6, noise figure has a primary effect on the receiver's noise performance curve. The receiver's threshold is raised if the noise figure increases; also, noise in a channel increases at all RSLs (not just near threshold) with an increase in noise figure. Figure 20-3 shows the effect of a poor noise figure.

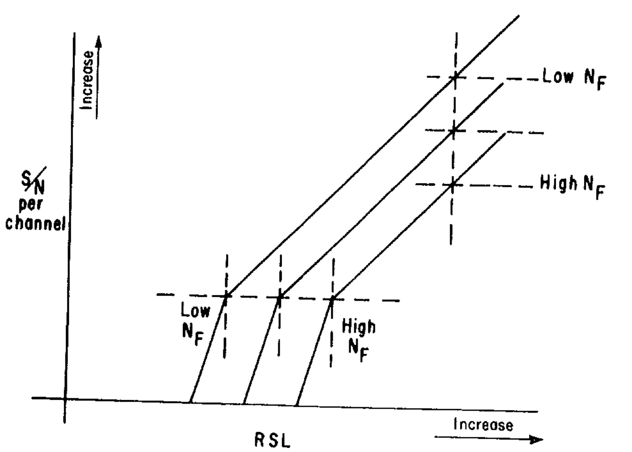


Figure 20-3. Effect of Noise Figure on S/N Ratio.

20-3. Voltage Standing Wave Ratio (VSWR):

a. General. Waveguide or coaxial cable is used in a communications system to connect the transmitter or receiver to the antenna. For proper operation, the impedance of these terminating devices must match the impedance of the waveguide/coax. If a mismatch occurs, some power is reflected at the termination and standing waves are formed. A detailed discussion of standing wave formation is in chapter 11. The amplitude of the reflected wave divided by the amplitude of the incident wave is called the reflection coefficient. Expressed in decibels, this is called the return loss and is always a positive number.

$$\text{Return loss} = -20 \operatorname{Log} \frac{E_r}{E_i} = -10 \operatorname{Log} \frac{P_r}{P_i} \qquad \text{Eqn 20-1}$$

Since E_r/E_i is always less than 1, higher values of return loss represent less reflection and better performance.

Another unit for stating the degree of mismatch is the VSWR. This is defined as the ratio of maximum voltage in the standing wave to the minimum voltage in the wave. VSWR is defined in figure 20-4.

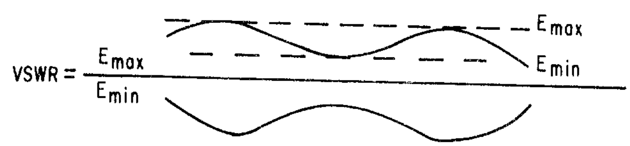


Figure 20-4. Calculation of VSWR From Standing Waves.

The sum of the maximum incident wave and maximum reflected wave is expressed as E_{max}. Similarly, E_{min} is the maximum incident wave minus the maximum reflected wave; hence, if we can measure the forward and reflected waves separately, we can calculate the VSWR. Specifically, if we know the ratio of reverse voltage to forward voltage (or reverse power to forward power), we can obtain VSWR.

$$\text{VSWR} = \frac{1 + E_r/E_i}{1 - E_r/E_i} \qquad \text{Eqn 20-2}$$

or

$$\text{VSWR} = \frac{1 + P_r/P_i}{1 - P_r/P_i} \qquad \text{Eqn 20-3}$$

It can be seen that VSWR and return loss are directly related.

Although there are many ways of measuring VSWR, only the reflectometer method will be covered here. This method uses a directional coupler to sample reflected power.

b. Test Procedures. The equipment necessary to measure VSWR by the reflectometry method is shown in figure 20-5. The generator provides a constant level into the device at the frequency of interest. The reverse power directional coupler samples only the reflected power coming from the device under test. A crystal detector provides a voltage proportional to this power to the voltmeter. Since the incident power is constant, the reflected power becomes an indicator of VSWR. The parameter measured directly by this method is return loss. The calibration of the meter to read directly in return loss requires a precision attenuator and a circuit for the directional coupler. Procedures for using this method are:

(1) Set the generator to the frequency of interest.

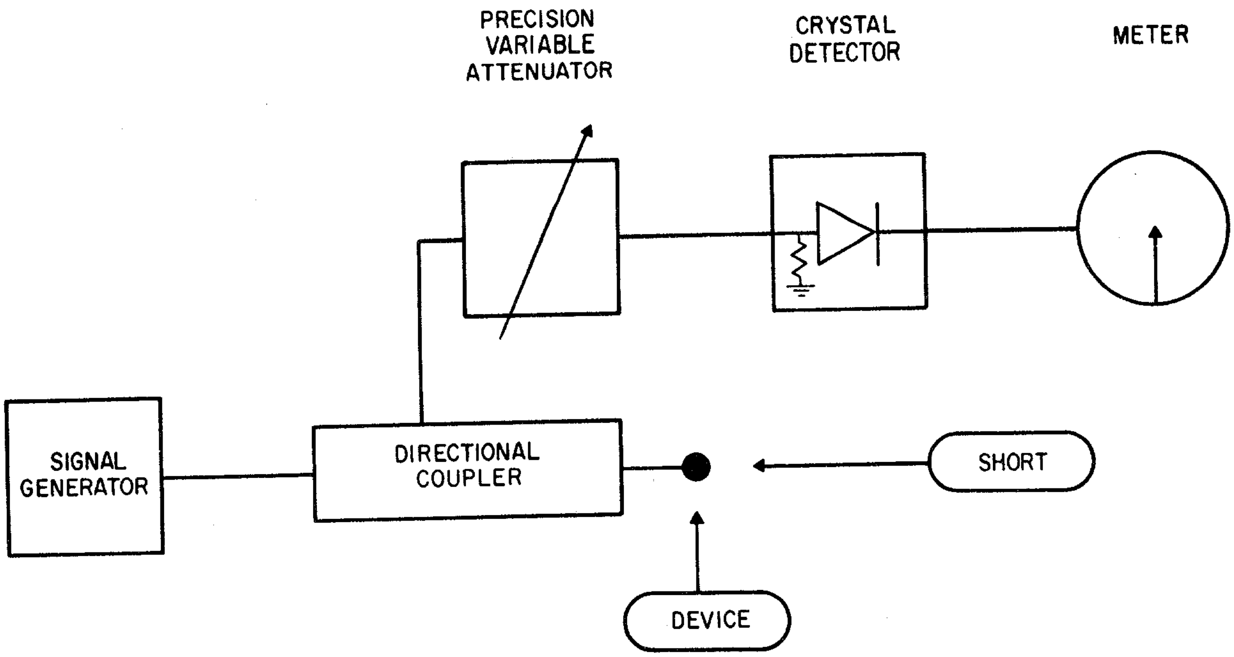


Figure 20-5. VSWR Test Configuration.

(2) Connect the RF output (through an adapter, if necessary) to the directional coupler. Attenuators may be added at the generator output if needed to protect the crystal detector.

(3) Attach a precision variable attenuator to the sampling arm of the reverse directional coupler. Initially, set the attenuator to 0 dB.

(4) Connect the attenuator output through a crystal detector to the voltmeter.

(5) The setup is now ready for calibration. Place a short circuit on the directional coupler in place of the device to be tested. The indication on the voltmeter now represents a reflection coefficient of 1 or a return loss of 0 dB. Mark this voltage on a data sheet and label it 0 dB.

(6) Now place 5 dB of attenuation in the path with the variable attenuator. This simulates a condition of 5 dB return loss. Note this voltage and label it 5 dB on the data sheet.

(7) Repeat step 8 every 5 dB up to 40 dB attenuation.

(8) Remove the short and connect the device to be tested to the directional coupler. Return loss may then be read using the calibrations.

c. Analysis of Results. Reflected power at a mismatch is wasted power. It subtracts from the useful power that can be transmitted; also, if the reflected power is large enough, it can damage equipment (such as radio transmitters). VSWR and return loss are indicators of reflected power and can tell us when corrective action is necessary. Some common VSWR checks are discussed below.

The VSWR looking toward the antenna from the transmitter RF interface point is of two-fold interest. Any reflections at the antenna reduce the radiated power, thus directly affecting system performance; also, reflected power from the antenna is fed back into the transmitter, creating the possibility of damage to the transmitter. DCA standards specify that the return loss measured toward the antenna from the RF interface point shall not be less than 32 dB (VSWR less than 1.05:1). Similarly, the VSWR looking into the receiver form the RF interface point is important. The RF signal here is usually weak. Reflections here would only make it weaker and degrade performance. DCA also specifies a minimum 32 dB return loss for this interface.

The previous VSWR checks were concerned with loss of signal strength due to reflections; however, mismatches can degrade performance in another way. If a reflected wave encounters a mismatch at the power source, a portion of it will be re-reflected and travel out as an addition to the transmitted wave. This extraneous wave will be delayed with respect to the main wave. In FM systems, the effect of this wave is to introduce intermodulation noise into the signal. This form of intermodulation is commonly known as echo distortion and usually occurs in the transmission lines between the transmitter and the antenna and between the receiver and the antenna. In addition to the VSWRs already mentioned, two other VSWR figures become important, the VSWR looking at the antenna from the receiver RF interface point. The standards for these interfaces are identical with the others: 32 dB return loss (VSWR of 1.05:1).

NOTE: Be aware of cabling and connector VSWR when making these tests.

If any VSWR measurement is bad, a mismatch may be indicated or a defect may be present in the waveguide or coaxial cable. VSWR measurements on sections of the system under test will help to isolate the problem. Dents, deformed waveguide, foreign objects in the waveguide, loose connections, and maladjusted filters can cause VSWR problems.

20-4. VSWR, Swept Frequency Method:

a. General. The previous method of determining VSWR is useful for commonly found test equipment and only for single or discrete frequencies; however, a more realistic representation of VSWR is that measurement obtained across a frequency segment. Since an FM radio actually deviates around a center frequency with a band of frequencies, it is important to know the VSWR characteristics at every frequency within this band. The swept frequency method for determining return loss is best suited for this job. The equipment necessary to measure return loss or VSWR by the swept frequency method is shown in figure 20-6. The equipment requires more sophistication than the single frequency method in that a swept frequency generated with constant level output across the band of interest is needed. A spectrum analyzer is used as a tuned receiver and indicator. Calibration is accomplished using precision calibrated loads or terminations or by using the calibration mark on the spectrum analyzer if accuracy has been demonstrated. Full and complete test procedures for this method of testing are found in AFCSP 66-17.

b. Analysis of Results. Once set up and calibrated, this method will give VSWR characteristics across the band of interest. If the spectrum analyzer has storage features, a complete frequency versus return loss picture with calibration lines can be held and photographed for record purpose. Normally, an antenna system for troposcatter or LOS M/W will not be found to meet 1.05:1 VSWR across the band of interest. Valves as high as 1.2:1 at points within the range are commonly found. Valves much higher than this will usually indicate use of flexible waveguide sections, bends, and excess of short waveguide sections. Values at points within the swept frequency of interest that exceed 1.5:1 usually indicates a problem exists within the waveguide assembly. At this point, suspected trouble areas should be examined by a fault location procedure as outlined in AFCSP 66-17.

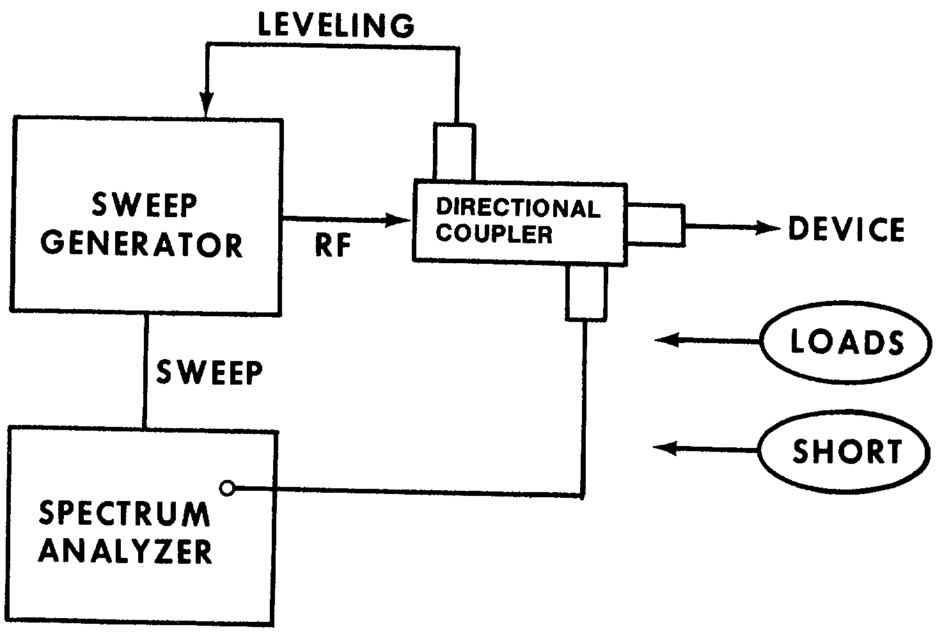


Figure 20-6. Swept Frequency USWR Test Configuration.

20-5. Multiplex Loaded Noise Test:

a. General. Traffic over communications wideband links is ever-increasing. Systems are being designed to carry more VF channels and a larger percentage of these available channels are being used; therefore, load capacity (the volume of high quality traffic a system is capable of carrying) is becoming a more important factor. Many communicators tend to think that maximum load capability is dependent only on the RF equipment and that the multiplex equipment is capable of simultaneously passing quality traffic on all channels. This is seldom true. Many factors enter into multiplex loading. Most important are the number of channels used, channel loading levels, and the type of traffic (voice or data) carried. Data systems usually transmit signals continously and present a constant load while voice-loaded channels are carrying sporadic signals. From this, we can see that data channels present much heavier loading per channel than telephone voice-loaded channels.

This test establishes the overload point of the multiplex equipment and determines how much intermodulation noise is being contributed by the multiplex to the system under various load levels. Noise is measured with a VF noise measuring set using both 3 kHz flat and C-Message weighting. A comparison between C-Message and 3 kHz flat readings helps to determine the frequency of noise present and isolate interference from pilots and carriers present in the baseband.

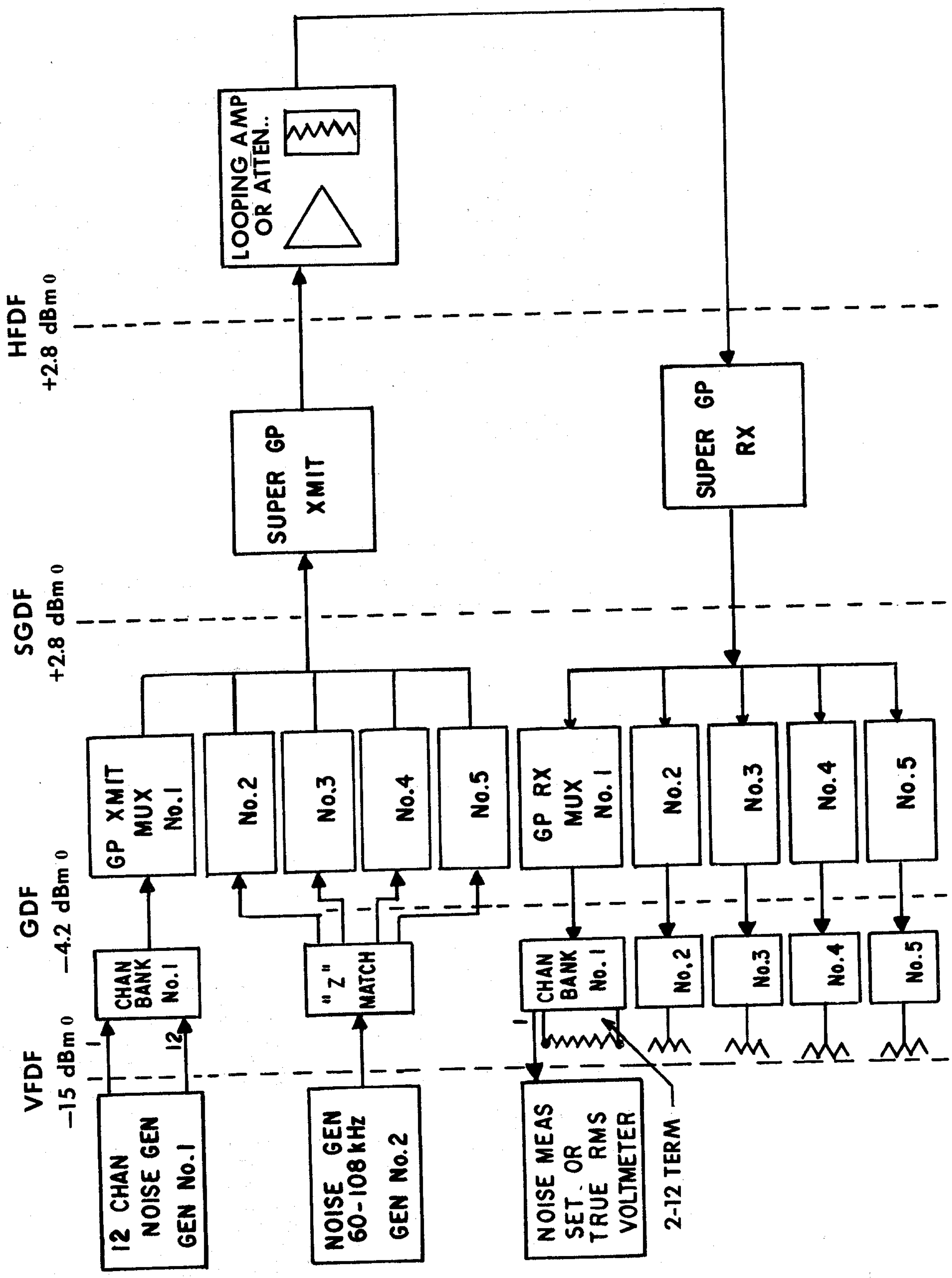


Figure 20-7. Multiplex Loaded Noise Test.

To determine multiplex loading capability, the equipment is loaded with simulated traffic in the form of noise. A generator supplying 12 channels of VF noise is connected to one group while another generator is connected to supply noise input to the remaining four groups at the group distribution frame or group patch bay. This allows loading of one complete supergroup which is looped at the baseband radio modulator/demodulator point. A more nearly ideal situation would be to connect single channel generators to each channel in the supergroup but, in practice, this is impractical due to the amount of equipment required. In the great majority of multiplex systems, no active devices are employed which are common to more than one supergroup; therefore, testing only one supergroup at a time is considered acceptable. The loaded S/N ratio is determined by measuring the noise level with 3 kHz flat weighting at the channel output, then removing the noise input to only that channel and comparing the levels measured with the noise on and with the noise off. Each channel is checked individually.

b. Test Procedures:

(1) Establish test connections as shown in figure 20-7. Determine and set noise levels for -10 dBm0 on all channels as follows:

Power (gen #1) = -10 dBm0 per channel

Power (gen #2) = -10 + 10 logN dBm0
(N = number of channels)

(2) With the noise measuring set, check the level on channels 1, 7, and 12 of each group at the channel output. The level should be -10 dBm0 ±1 dB on all channels. This check will verify that all connections and levels are set to give the proper noise over the complete supergroup range. Only when correct receive levels are measured throughout the supergroup can the test be completed.

(3) Beginning with group 1, channel 1, measure and record the noise level, using both C-Message and 3 kHz flat weighting with the noise on, then with the noise off. Turn the noise on channel 1 back on and repeat the same measurement for each channel in group 1. If time does not permit, only a representative number of channels need be measured in order to obtain the required data.

(4) Move the connections from the 12-channel noise generator to group 2 and connect generator # 2 to groups 1, 3, 4, and 5. Measure channels in group 2 and continue with all other groups in the same manner, loading the group which is being tested with the 12-channel generator and all other groups with generator # 2. When more than one supergroup is used, continue until all supergroups are measured.

(5) Reduce the output of each channel on the 12-channel noise generator by 2 dB. Reduce the output of generator #2 by 2 dB and repeat steps 3 and 4.

(6) Repeat step 5 until the noise generator outputs are reduced 10 dB below the level determined in step 1.

(7) Adjust the noise generator outputs to a level 2 dB above the levels determined in step 1. Repeat steps 3 through 6, except this time increase the noise in 2 dB increments until the levels are 10 dB above the level calculated in step 1. Sufficient data will then be recorded to plot a noise loading curve.

(8) To provide actual loading and performance capability, plot representative information to show a S/N versus noise loading level curve on each group or supergroup as desired. S/N is calculated from the measured data as shown below:

$$S/N = (N_{on} - N_{off}) - N_L$$

where;

N_{on} = level measured with noise on (dBm)

N_{off} = level measured with noise off (dBm)

N_L = Per channel noise loading (dBm0)

Figure 20-8 is representative of this type of curve.

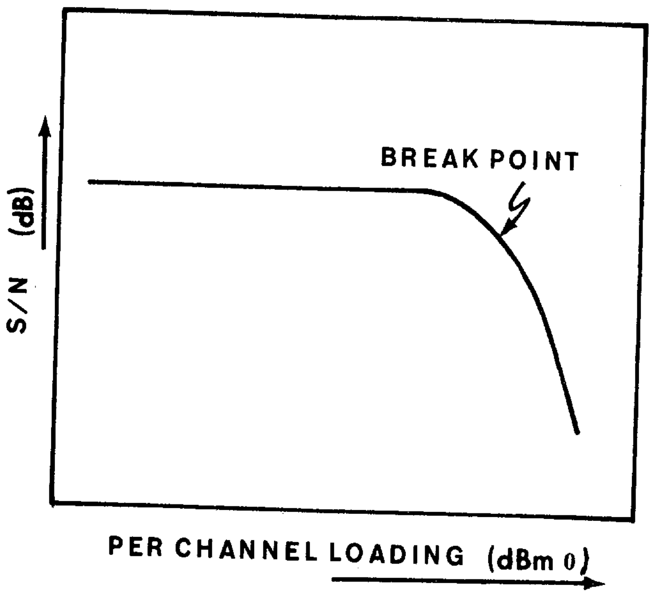


Figure 20-8. S/N Versus Channel Loading.

c. Analysis of Results. For our purposes at this time, we are interested chiefly in the system overload or break point. Different methods may be employed to determine this loading level; however, a sample and usually adequate method is to determine the point where a 2 dB increase in channel loading level causes a drop in S/N of more than 2 dB, indicating excessive intermodulation within the multiplex equipment. Multiplex loading should not be allowed to exceed this point more than a very small percentage of time. The important thing to remember is that very few present-day systems are capable of 100% data loading; therefore, total channel capability and total load capability are not the same. In such cases, all channels in a group or supergroup should not be loaded with data signals at the same time. TOs and manuals on equipment capable of full data loading are almost certain to contain this information. Other multiplex systems are likely to be designed for approximately 20% data and 80% voice loading.

Optimum per-channel loading can be determined from the plotted curve. The best S/N ratio will appear at the bottom of the plotted curve. Significant increases or decreases in level will cause deterioration of the S/N ratio. When levels are decreased, thermal noise will be closer to the intelligence causing degradation in S/N ratio. When levels are too high, intermodulation within active components of the multiplex will result in a lower S/N ratio. As a summation, channel levels should be high enough, but not too high, to provide optimum operation.

20-6. Data Bit Error Rate Test:

a. General. A data stream sent over a wideband system is subject to many types of destructive influences. Propagation anomalies such as deep fades (sometimes below threshold) can cause bursts of errors and IPN causes errors when the impulses are mistaken for data. If the white noise level is high enough, it can totally destroy the information in the data stream. This test is a system test that measures the result of all these influences. The measurement is made from a DC subscriber over a circuit to another DC subscriber. The result is simply the fraction of bits sent over the system that appear as errors.

b. Test Procedures. The error rate test can be performed over a circuit in two ways. Two test sets can be used and the error rate determined in each direction separately. This test setup is shown in figure 20-9. Alternatively, the system can be looped back at one end and a composite error rate determined for both directions. Only one test set is needed for this (figure 20-10). The test set consists of a data code generator and a code detector. The transmitter section generates a predetermined pseudo-random sequence which is applied to the binary input of the system under test. The receiver section receives the code at the other end of the circuit. The received code is compared with an identical locally-generated code and the errors counted and displayed. A synchronizer circuit automatically synchronizes the locally-generated code to the received code. A wide range of bit rates is usually available for testing.

c. Analysis of Results. Standards for error rate can be put in many terms. Error rate is usually expressed as a fraction of bits sent (that is, 1 error in 10^5 bits); however, errors do not occur randomly-they usually come in bursts. DCA standards are written in terms of bursts/time and errors/burst. The error rate test gives a true indication of the quality of a data circuit. Some modems regenerate data pulses and ensure that the length of pulses, etc., are perfect. A peak data distortion test would give zero distortion as a result; however, although the pulses are perfect, errors could be made in regeneration due to distortion in the audio signal. These errors would show up in a bit error rate test. Since errors are caused by a wide variety of system impairments, little analysis can be done on the test data itself; however, it is valuable when correlated with results of other tests. A common test is to correlate error rate with RSL on troposcatter links. Correlations could also be made with IPN tests and ICN results to determine their effect on data transmission.

20-7. Microwave Link Analyzer (MLA) Measurements. The MLA allows detection and optimization of transmission distortions in high density systems. Before the MLA introduction, signal impairments were measured by the use of the white noise loading technique. The noise loading method was an acceptable analysis tool for optimizing low density systems but, with increasing traffic densities, new analysis tools became necessary for reducing circuit distortions. Among the many measurements the MLA performs to analyze a system's characteristics are:

a. Modulator Linearity.
b. Demodulator Linearity.
c. Group Delay.
d. IF Amplitude Response.
e. Gain, Attenuation, and Power Measurements.
f. Modulator-Demodulator Sensitivity.
g. Radio Link Linearity and Delay from Baseband to Baseband.

The MLA is capable of displaying the IF section, delay or group delay, versus frequency response on a CRT. The group delay is the derivative of the phase characteristic with respect to frequency as was discussed in chapter 4. In an ideal system (linear phase), the carrier and sidebands all experience the same transmission time which means that the ideal group delay would be flat across all frequencies. The MLA simplifies the equalization process (flat delay versus frequency) since it provides a dynamic display of the group delay versus frequency. It is important to have the group delay characteristics as flat as possible to reduce distortions that are phase (transmission delay) related. The MLA also measures the modulator linearity by displaying the derivative of the modulator characteristics as a function of frequency on the CRT. This is a more accurate and time-saving method than measuring and plotting a number of carrier dropout levels to determine the modulator linearity. Many other capabilities of the MLA can be found by consulting the manufacturer's user manual. When used effectively, it aids in localizing the sources of noise or distortion to a specific part of the radio system.

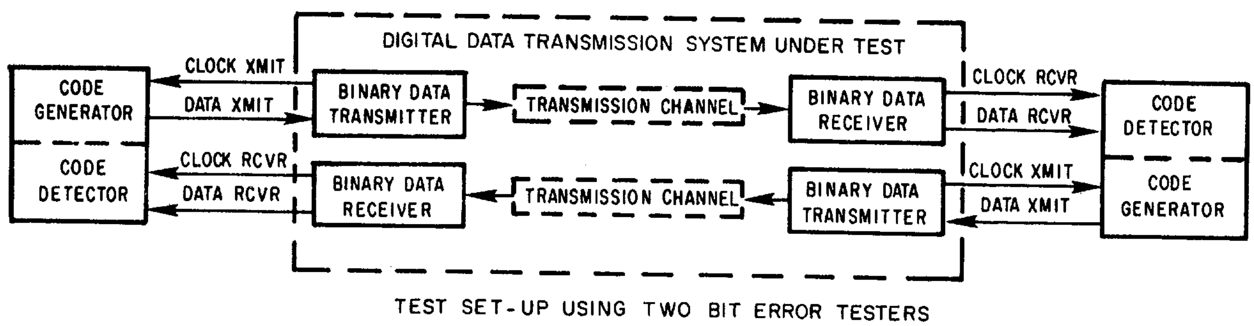


Figure 20-9. Test Setup Using Two Bit Error Testers.

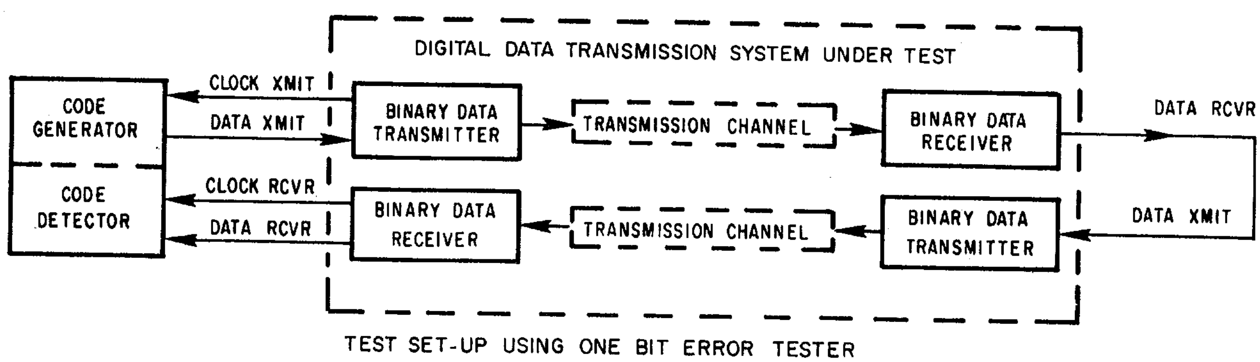


Figure 20-10. Test Setup Using One Bit Error Tester.

Chapter 21

FACILITY GROUND SYSTEM

21-1. General. All electric and electronic equipments that are connected together by conductors (other than radiated electromagnetic waves) require a common reference point. On land, the earth is usually considered a zero level reference point - ground. On satellites or aircraft, the exterior shell is normally used as an artificial ground. In any event, it is a reference point, not a return path for current.

There are three types of ground subsystems or three reasons for having ground subsystems in equipment: lightning protection, safety, and signal quieting. This chapter deals with the earth electrode subsystem (how the various ground subsystems are attached to earth), the three subsystems, and some considerations for grounding mobile systems.

Many of today's facilities violate the principle of independent ground subsystems. Interconnecting them at other than the earth electrode subsystem causes limitations in their designed effectiveness. In the same vein, many communicators view the AC neutral return as a ground conductor. That is not true. A ground subsystem is one that has NO current flowing in it. Current in a ground conductor indicates a problem.

21-2. Earth Electrode Subsystem:

a. The earth is used to dissipate excessive electrical charges caused by man-made and natural sources that might injure personnel operating equipment and damage the equipment involved. Connections to the earth (ground connections) are made to the earth electrode subsystem, which provides the means of obtaining the lowest possible impedance contact with the earth. The ideal earth electrode subsystem consists of a single, large conducting plate that covers the entire area of the installation and is connected to the earth at an infinite number of points. An ideal earth electrode subsystem is not practical; however, it can be effectively approached by installing a suitable ground network that surrounds the area of the installation. This network would consist of continuous copper cables or other suitable conductors buried beneath the earth's surface and connected to driven ground rods, plate electrodes, a water pipe system, etc. The ideal size for the ground bus will depend on the magnitude of the available ground fault currents and operating time of protective equipment. For practical purposes, it should have the equivalent conductivity of not less than #2 AWG wire for most installations. If this system is then developed into a completely interconnecting network, it will approach the ideal.

b. There are two basic types of electrodes used in the earth electrode subsystem: those specifically placed in the earth as electrodes (such as driven rods, buried wire, mats, strips, plates, and other objects as required to establish an electrode of the desired resistance or impedance to earth); and those serving another function (such as water pipes, well casings, reinforcing bars in building foundations, buried metal tanks, and other miscellaneous metal objects buried in the earth). Regardless of what type of electrodes are used to establish an earth electrode subsystem, all must be interconnected by appropriate conductors to form one continuous network. This will ensure that all grounded facilities within the site are at the same ground potential and that connections to the subsystem provide the lowest impedance to earth feasible at that site.

21-3. Lightning Protection Subsystem:

a. Since a telecommunications facility is often located in an area where it is susceptible to numerous lightning strikes, adequate protection must be provided. This paragraph provides a brief description of lightning protection methods and devices used to protect antennas, waveguides, towers, personnel, equipment, and power facilities.

b. Lightning discharges may cause voltages in conductors by induction or by direct strike. On high voltage lines, the insulation may prevent outages by induction, as well as most of the outages caused by direct strike. Because of the rapid attenuation of a surge propagated along a conductor, "sparkover" generally occurs near the point of origin; however, sparkover may occur at dead ends or at other voltage reflection points located a considerable distance from the point of origin.

c. When lightning strikes, the object struck suffers damage commensurate with its relative conductivity. Metal, for example, receives a lightning discharge with little damage to itself. In most cases, conductors (such as telegraph, telephone, and electric light wires) carry a discharge without fusing, except where the discharge enters the metal. Severe damage may occur at the point where the discharge enters or leaves the metal, which results in the metal being fused or melted. Since most of the charge is carried by relatively low current that flows for a fairly long period between component strikes, such discharges are most likely to do damage to nonconducting materials. When insulating or semi-insulating material receives a discharge, the damage is usually severe and may assume an explosive character.

d. When personnel are subjected to direct lightning strikes, the results are nearly always fatal. Although extraordinary escapes from direct strikes have been recorded, it is considered extremely unlikely that this was the actual circumstance. The shock from direct strikes is so great that survival is rare. The major portion of lightning casualties arise from secondary conditions (such as side flashes and induced charges).

e. Fundamentally, it is preferable to "dump" as much lightning strike current directly to earth as is economically feasible. This reduces the proportion of the strike current seeking remote earth over connecting lines and thus simplifies the task of protecting communications facilities. It is, therefore, important to provide as many direct, low impedance paths as possible to the earth electrode subsystem.

f. Protection from excessive voltages on communications circuits is normally provided by open-circuit devices that pass no significant current at the normal operating potentials of the circuit to which they are connected; however, they are capable of arcing over on extraneous voltages at predetermined values and discharge current, usually from an energized conductor to ground. During the period of discharge, such a device will limit voltage across its terminals to values sometimes less, but rarely greater, than the initial sparkover value. Leads to these devices must be kept short so that their ability to limit the anticipated potentials is not impaired. Communications protectors typically consist of gas-tube or solid-state devices or closely spaced, flat carbon electrodes (carbon blocks) discharging in air. The gas tubes use metal electrodes in an enclosure with inert gas at reduced pressure. The gas-tube protector provides essentially the same level of protection as the carbon block device, but the wider gap spacing gives a much longer service life with a commensurate reduction in maintenance. There are a variety of solid-state devices that limit voltage. Generally, their major protection application is on low voltage circuits in communications apparatus as "second-order" protection. These are usually used in conjunction with heavier duty devices (protectors) that provide "first-order" protection against the large extraneous voltages and currents on the outside transmission facilities.

21-4. Equipment Protective Subsystem:

a. The equipment protective subsystem is a network designed for protecting personnel and equipment in the event of a malfunction through an electrical short to a metal equipment enclosure. To accomplish this end, ground connections must be adequate for both normal and fault currents. This includes all exposed noncurrent-carrying metal parts of fixed equipment that are likely to become energized; therefore, these parts must be grounded. In general, the equipment protective subsystem will conform to the requirements established in the National Electrical Code (NEC).

b. Experience gained through quality assurance surveys of grounding systems has shown that one of the major shortcomings in grounding systems is the interconnection and reversal of AC neutral and protective wires of the AC power distribution at various power distribution panels throughout a facility. These installation errors result in additional electrical noise and AC currents in the ground system. The equipment protective subsystem will not normally have any intentional interconnections with the signal reference subsystem except at equipment cabinets which are AC-powered and at the earth electrode subsystem. The equipment protective subsystem should generally follow a "tree" or "crow's foot" configuration from a central or main ground point which, ideally, should be at the power house or primary station transformer ground point or will be bonded directly to the earth electrode subsystem at the communications building, if a protective wire is not available to the main ground point. From this main ground point, the protective wire is carried along with the phase and neutral wires to the main circuit breaker panel, from there to intermediate circuit breaker panels, to the equipment panel, and finally to the equipment itself.

c. To protect personnel from exposure to hazardous voltages, all exposed metallic elements of electrical and electronic equipment are connected to ground with a protective (normally green) wire. Then, in the event of inadvertent contact between the "hot" lead and chassis, frame, or cabinet through human error, insulation failure, or component failure, a good, direct, known fault current path is established to quickly remove the hazard. Grounding of a 3-phase wye power distribution system is done similarly to the single-phase system. As in single-phase systems, the neutral lead is grounded for fault protection at the main ground point. It should never be grounded at the equipment.

d. A separate equipment protective conductor will be included with the AC power distribution if not provided for in the power cable; when run as a separate or independent conductor, it should be placed in the same conduit or duct with the phase and neutral conductors for best results. If this is not possible, it should be installed alongside the AC conduit or power duct and connected to the power panel protective bus bar. It should not be connected to the signal reference ground conductors. A minimum 2-inch separation should be maintained between equipment protective conductors and communications or signal reference conductors if not encased in conduits or ducts.

21-5. Signal Reference Subsystem:

a. The signal reference subsystem is a network designed to control static charges and establish a common reference for signals between sources and loads to minimize interference. This paragraph identifies some basic considerations applicable to installing a signal reference subsystem. The effective signal reference subsystem will be configured to serve a facility operating in the audio or low frequency range; a facility operating primarily at the higher frequency of radio/multiplex equipment; or a facility where both audio and high frequency equipments operate.

b. The signal reference subsystem is a composite of various networks. The configuration of these networks is dependent on the frequencies involved, the functions being performed by the equipment involved, and the amplitude of the signals on the communications wires or in cables. Modern electronic systems seldom have a single ground network that can effectively satisfy grounding requirements while preventing induction or conduction of undesirable currents and voltage in the signal reference subsystem. To minimize interference from the many potential sources, separate networks are required. These networks consist of the VF signal and shield ground, teletypewriter (TTY) signal and shield ground, and the power supply (station and loop) reference grounds. The VF signal and shield ground conductors may be kept separate. The same can be done with the TTY signal and shield ground conductors. High-level TTY shields are especially noisy and should be bused to the earth ground electrode subsystems separately. Other facilities might consist of a single interconnected subsystem where all components are interconnected to form one complex ring grid network. Such a network spreads whatever electrical noise is present over many parallel leads and reduces its overall effect on the communications system.

c. The signal reference subsystem must be separate from the power distribution system to the extent possible with current communications equipment. The AC neutral should never be grounded to the signal reference ground. The AC protective connection at the equipment cabinet is a limitation resulting from present equipment design. Future equipment may be designed without this interconnection, providing separate protective and signal ground points. DC powered equipment requires no AC protective grounding.

21-6. Protection of Transportable Facilities:

a. In transportable installations, stranded conductors and flexible cables are used predominantly. The use of stranded conductors for power wiring is mandatory . Use of solid conductors often results in wiring and lead breakage, which creates major operational problems. Stranded wire should be properly laced or wrapped, with adequately spaced clamps for cable and wire bundling support and sufficient slack at terminal connections. In all cases, installation must be IAW the NEC. In those nonpower instances where solid conductors must be used, it becomes doubly important that they be securely anchored along the entire run to prevent unnecessary flexing; also, there must be enough slack at the termination points to compensate for in-transit motion.

b. Because a transportable shelter is a metal structure, transient voltages will tend to be conducted throughout the shelter if adequate insulation is not provided between noisy equipments (such as TTY and relay-operated switches) and the structural metal itself.

21-7. Grounding of Transportable Equipment at a Fixed Facility:

a. The integration of a transportable C-E assemblage into a fixed site - whether as a semi-permanent addition to the site or to satisfy a temporary contingency - requires proper grounding and power distribution. The grounding and power distribution provisions discussed in the preceding paragraphs are also applicable to transportable configurations when employed as part of a fixed installation. The transportable equipment must be connected to the earth electrode subsystem by a direct route, preferably by two separate routes if economically feasible. Driven ground rods and interconnected conductors will be used, if necessary.

b. Electrical power will normally be provided from the site's primary power source. The connecting power cables will provide the neutral and phase leads and a protective ground (green) conductor. Any support vans

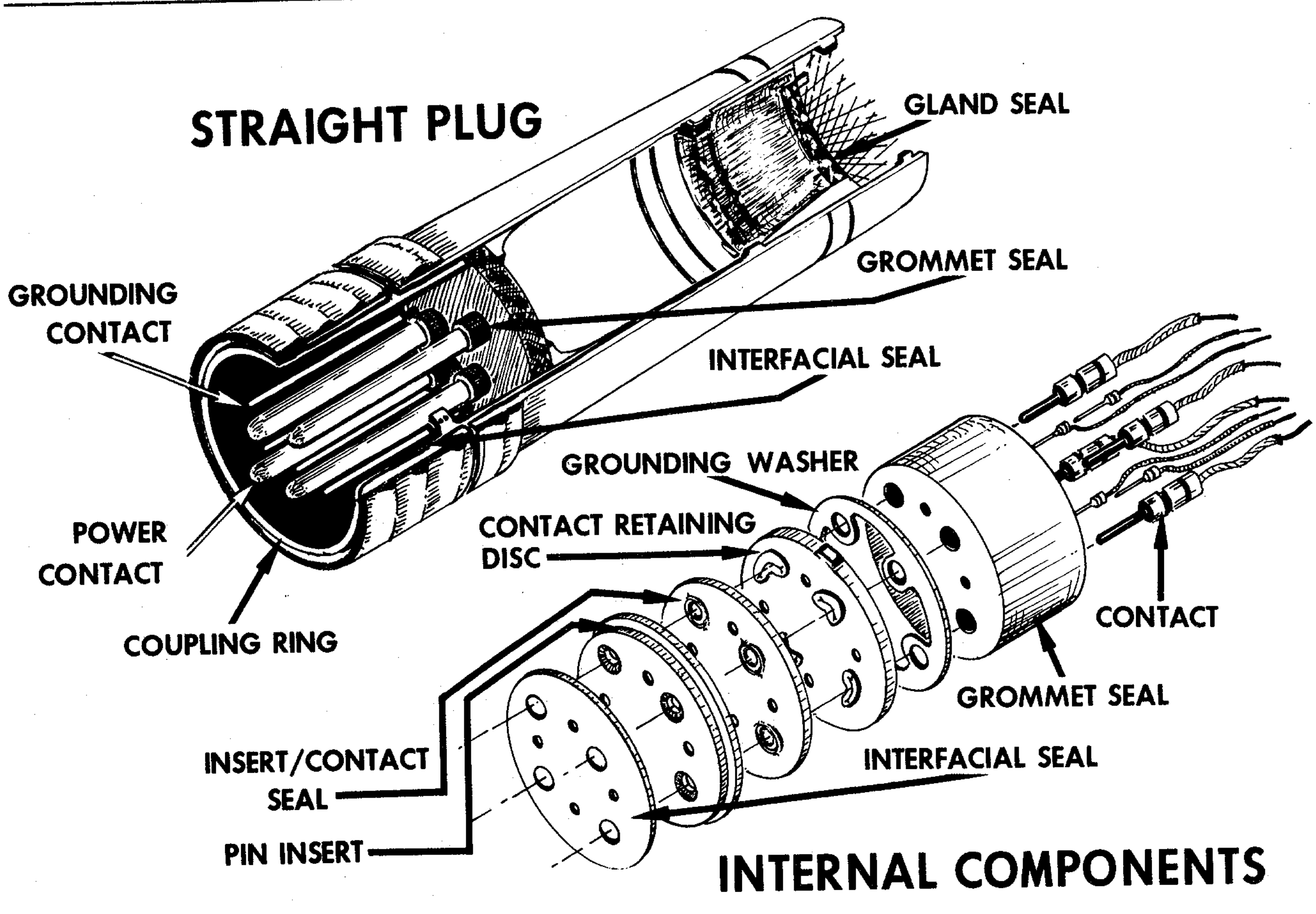


Figure 21-1. Class L Power Cable Connector.

or structures (such as air-conditioners or power supplies) shall also be interconnected by means of the protective ground conductor. Figure 21-1 shows a class L power cable connector for terminating power cables containing five conductors, including a protective ground. A class L or similar type connector which is designed for power distribution using portable power cables should be used for interconnecting transportable facilities at fixed sites. Power cable connectors used with transportable equipment must be durable and capable of withstanding severe impact and extreme field service use. They must be self-quenching when separated under load, waterproof, and meet NEC requirements for grounding.

21-8. Protection Of Transportable Equipment In Remote Locations:

a. The grounding methods required by the NEC and those described herein are applicable to both transportable installations and fixed facilities. The specific method used will result from an evaluation of the operational characteristics of the system, power requirements, local environmental conditions, and other applicable considerations. The driven type of ground electrode offers the most convenient means of providing electrical ground connections and will satisfy the grounding requirements of most transportable installations. It has the advantages of economy of installation and satisfactory ground terminations.

b. Transportable equipment must be grounded IAW the NEC to prevent damage to the equipment from lightning strikes or internal power wiring shorts. Lightning and static charges frequently follow the transmission line to the container and then to ground. A ground with excessive resistance or inductance could place the container at a very high potential with respect to the earth's surface. When the van is ineffectively grounded, the possibility of interference pickup is greatly increased. Resistance or inductance in the grounding system makes the van electrically more susceptible to interference from external sources. Transmitting antenna radiation efficiency is also directly proportional to grounding efficiency at both the antenna and the transmitter equipment. Excessive inductance in the ground lead could place the van above ground potential at radio frequencies, creating personnel and fueling hazards.

c. The equipment cabinets in the van are bonded to a common bus or strap which, in turn, is bonded to the van at an external ground plate. Ground plates should be constructed with sufficient surface area to carry the required current and retain a low resistance connection for long periods of time. All grounding circuits generally terminate at one ground plate. Power circuits require a separate ground. Common ground buses should have a large surface area to reduce losses at radio frequencies. The conductor between the van and the ground rod should be as short as possible. When excessive lengths of ground lead are used and are accidentally coiled, the resulting additional inductance could produce standing waves. The conductor should be capable of carrying at least 100 A. Standard steel or copper-clad steel ground rods approximately 6′ to 8′ in. length and 5/8″ to 3/4″ in diameter are most commonly used. Multiple ground rods, connected in parallel, are sometimes required to obtain a good, low-resistance ground. All shelters and power equipment should be interconnected to form a common earth ground.

d. The probability of antenna damage from lightning is small. A strike contacting the antenna may cause some surface fusing. but it would take a very heavy strike to fuse antenna elements or create holes in the reflector. The metal base plate of the mast and metal guys and anchors will provide some grounding; however, placing a buried counterpoise wire between the mast and the radio trailer ground (a reasonably simple operation) is recommended. Such a buried conductor will provide additional grounding for the antenna and radio trailer; also, the interconnection will provide potential equalization between the two units and thereby reduce the stress on the lead-in line and radio components in the trailer.

e. For safety reasons, the radio cabinets and other metallic objects in the trailer should be connected to the grounded metallic body of the trailer. The bonding previously recommended between the mast and trailer will reduce the buildup of potential between radio components associated with the lead-in line and the grounded cases. Although these recommended measures will prevent the appearance of high potential differences within the area occupied by the mast and trailer, a strike to the antenna can raise the potential of this entire area with respect to remote ground. Consequently, facilities connected to the radio trailer and associated with remote ground (such as the power cables and the cable to the multiplex trailer) will be subjected to this rise in ground potential and should be provided with protective devices (such as gas tubes). In line with this idea, protectors should be placed on all field wire and communications cable conductors entering the trailer. The cable shields should also be grounded to the trailer. The power cable has a grounding conductor that should be connected to the trailer and frame of the engine-generator; this will reduce potential differences between these two units. Driven rods or pipes will be used for establishing a ground. If the use of rods is impossible because of rocky soil, a buried counterpoise wire may be used as a substitute.

f. The cable sheaths entering the multiplex trailer should be grounded to the trailer and all cable conductors should have gas tube protection. Equipment cabinets and other metal objects should be grounded by bonding them to the metallic trailer body. The solid-state multiplex equipment will require special considerations. Grounding of the trailer and engine-generators should be the same as that described in the preceding paragraphs.

g. Since the switchboard trailer contains only conventional telephone equipment, gas tube protectors on all conductors entering the trailer should provide adequate protection. The metallic shields or sheaths of all telephone cables entering the trailer should be commonly grounded to the trailer. Grounding of the trailer, engine-generator, and power cable should be the same as that employed at the other trailers.

h. Where a cable run will ordinarily not exceed a few hundred feet (as in the case of the cable between the multiplex and the switchboard trailers), the protectors employed to protect the equipment at the terminal points will also limit the buildup of core-sheath potential to a considerable degree. Longer cable runs will be more exposed to lightning and higher core-sheath voltages can develop. If plastic-insulated conductor cable is used between these points and the run does not exceed about one-half mile, the probability of cable damage from lightning is small. In extreme cases where it appears that the cable to be used will not withstand stroke currents of as large a magnitude as desired, consider using a paralleling bare shielding conductor connected to the sheath at each terminal point and at intermediate points about 1,000 feet apart. The parallel conductivity provided will reduce the surge current flowing in the cable sheath and thereby lower the core-sheath voltage for a given magnitude of strike current. This shield wire should be in contact with the earth. Its effectiveness would be increased if (in the case of a buried cable) the shield wire is buried in the same trench with the cable and placed several inches above it. When so placed, the shield wire is in a favorable position to intercept a strike before it could arc directly to the cable.

21-9. References.
Recommended documents providing in-depth coverage of the complexities of this subject are:

a. AFCSRP 100-2, Tactical Communicator (a number of articles begining with the Oct-Dec 76 issue).

b. TO 31-10-24.

c. FAA Report No RD-75-215, vol II, DDC No AD A022608.

d. MIL HDBK 419.

e. MIL STD 188-124.

f. DCAN 310-70-1.

Chapter 22
FUTURE PROGRAMS AND TECHNIQUES

22-1. General. It would be impossible to accurately predict and describe all future communications techniques. That is not the purpose of this chapter. Rather, we will discuss near-term projects that are in the process of being implemented or that may soon be used in the Air Force. Some of you will be very familiar with the items discussed but most of you will not. Our purpose here will be to provide general information on programs and capabilities rather than discuss specific details.

22-2. Automatic Technical Control (ATEC). As the name implies, ATEC equipment is designed to automatically do many of the technical control functions that are now accomplished manually. The work done by ATEC can be divided into two general areas. First is the in-service monitoring function. This equipment will automatically measure and print-out such parameters as peak and average power, circuit noise characteristics, frequency offset, and phase jitter. It will do this without interruption to customer service. As presently designed, a single In-service Monitor Function unit is capable of scanning 200 monitor points every 15 minutes. When predetermined limits are exceeded, the equipment will give an alarm. The second functional area of ATEC is fault isolation. Not only does the alarm sound, but by the use of fault isolation algorithms, the equipment determines what and where the problem is. This fault isolation is accomplished automatically, almost instantaneously, and without manual intervention.

a. ATEC will be employed at three levels. The first level will be the station. This is the only place measurements are taken. Equipment here will include the data acquisition and information transfer capabilities to higher level control functions. Other equipment at the station level is shown in figure 22-1. The controller terminal function (CTF) will consist of a cathode ray tube display, a keyboard, a printer for hard copies, and storage capability. It provides the capability for the station tech controller to communicate with measurement acquisition subsystems on-station and with higher levels of the control hierarchy.

b. The second level of control is called the Nodal Control Subsystem (NCS). While the station will have a stand-alone capability, the key element to ATEC is the NCS. There will be several stations under one node. If a fault is found in one station, a fault isolation algorithm, stored at the node, will be implemented, measurements to be taken by station level equipment will be directed, and the station with the actual fault will be notified - all automatically. This capability is the key to the program. In addition, the present quality-of-service measurements that presently are accomplished manually under the PMP will be accumulated at the NCS. Since workload and time constraints will not be a factor, the frequency and number of parameters measured can be greatly expanded for this program. This will require a complete change in concept in order to efficiently use available information. At this time, the new concept has not been completely developed.

c. The third and highest level of system control is the Sector Control Subsystem (SCS). The SCS is actually very similar to the NCS and differs primarily in scope and detail. Two key responsibilities of the SCS are maintaining the data base and directing system alternate routing actions. In addition, the SCS will maintain system status and will act as the interface between adjacent NCSs. The SCS will be the point of contact for theater level O&M agencies.

Let's look at an example of a hypothetical problem to see how it would be processed in the ATEC environment and compare this to the manual system. In this example, let's say low levels are received at station A due to a channel alignment problem at station C (figure 22-2). Under the manual system, the fault isolation would probably proceed somewhat as follows:

Station A

a. Measure voice frequency and baseband levels. Analyze, record, and report findings.

b. Coordinate with station B.

Station B

a. Measure voice frequency and baseband levels at two points. Analyze, record, and report findings.

b. Coordinate with station C.

Station C

a. Measure voice frequency and baseband levels. Analyze, record, and report findings.

b. Coordinate with station B and with local maintenance.

Station B

Coordinate with station A.

This would probably take somewhere in the neighborhood of 25-30 minutes to isolate the fault to station C and another 15 minutes to restore. Under the ATEC environment, stations A and B would not be involved in the problem. All coordination, measurements, analysis, and record-keeping would be automatically accomplished at station C. In addition, station C is automatically notified what and where the fault is, only station C has to report, and there is automatic verification of fix action. In this case, it would probably take 2-5 minutes to isolate the fault and another 2-5 minutes to correct.

A field test for ATEC was started in central Germany in 1975. This is a joint Army/Navy/Air Force project and will be gradually expanded as the system is tested and problems resolved. When completed, the initial procurement will include a sector at Langerkopf, nodes at Langerkopf and Stuttgart, and stations at Langerkopf, Stuttgart, Friolzheim, Hoenstadt, Pirmasens, and Zugspitze.

In-service measurement function

DC measurement function

Baseband measurement function

Out-of-service measurement function

Alarm reporting function

Parameter converter function

Communication interface function

Controller terminal function

Wideband digital measurement

Figure 22-1. Station Level Functions and Equipment.

22-3. Digital Radio Systems. Communications systems to date, especially long-haul systems (such as the DCS), have been largely analog in nature and have been used predominantly for voice transmission. Digital transmission has been mostly teletype, transmitted in analog bearer channels by voice frequency telegraph equipment. Operation of transmission facilities of such systems has become a well-developed art.

Today, communications systems generally, and military systems in particular, are being changed to digital operation with encryption available at either the VF or baseband levels. For example, the Army is working on what they call their FKV (Frankfurt/Koenigstuhl/Vaihingen) digital radio system. The Air Force is in the Process of procuring their first major operational system which they call the Digital European Backbone (DEB). The parameters for digital systems are somewhat different than the parameters measured for analog systems; also, the automation potential is different. Digital radios offer advantages in encryption and regeneration.

There are two methods commonly employed today to convert analog voice waveforms into digital bit streams. These two methods are PCM and delta modulation.

Notice that, although they are called modulation, they are not used to modulate a baseband data sequence onto some carrier. That type of modulation is accomplished either by varying the amplitude, frequency, or phase of a sinusoidal carrier (the classical AM, FM, and PM) or by modulating the duration or timing of pulses in a carrier pulse train (PDM, PPM). PCM and delta modulation are simply analog to digital (A/D) conversion techniques, despite the term "modulation" in their names.

Historically, PCM was the first method used for A/D conversion and it is still the most widely used. It is presently used extensively by commercial telephone companies and is the standard A/D technique used by the US. The first operation in a PCM system is to sample the signal amplitude. The absolute value of the signal amplitude is then expressed as a binary number (code word) for transmission. Sampling theory tells us that if sampling of the signal is accomplished at a rate which is no less than twice the highest frequency contained in the analog signal, the signal can be reconstructed without distortion from the amplitude samples. For example, a nominal 4 kHz voice channel requires sampling at 8,000 times a second in order to reconstruct the original waveform without distortion. So, voice quality PCM A/D conversion entails the formation of 8,000 binary code words per second. These code words are typically 5 to 10 bits long. A higher number of bits gives better quality, but steps up the overall bit rate and, consequently, calls for more bandwidth in transmission. Because of the fixed sampling rate of 8 kHz, only discrete steps of 8 kb/s can be chosen for the ultimate bit rate.

Six-bit code words imply a 48 kb/s data rate; 8-bit words imply 64 kb/s. Because they are discrete code words with a definite beginning and end, synchronization signals have to be inserted between code words. Even with a single channel, word synchronization cannot be avoided.

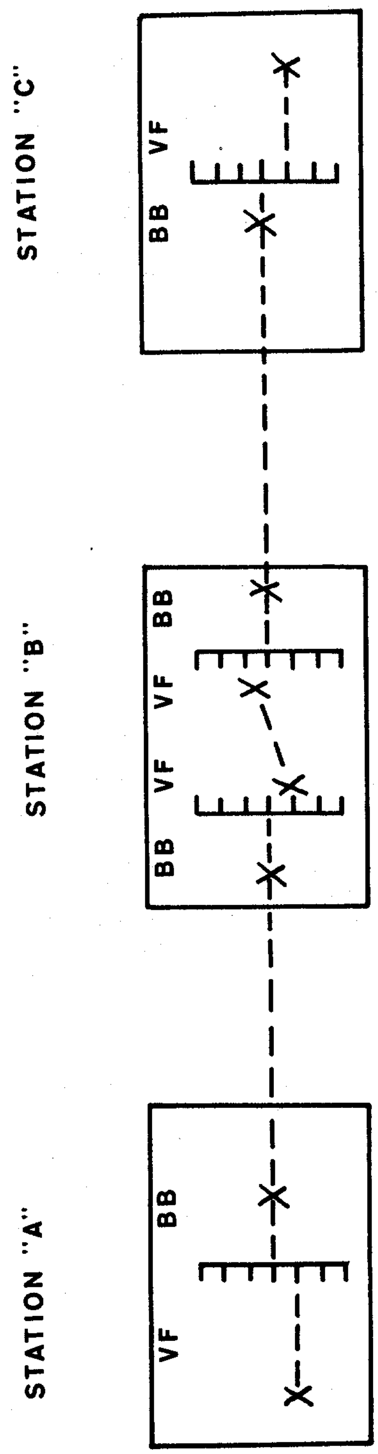


Figure 22-2. Fault Isolation. Measurements made at points X.

In delta modulation, each sampling produces only one binary digit, not a multi-bit code word. This single bit gives an indication of the relative amplitude of the current sample to the previous. A "1" bit indicates that the amplitude sampled is higher than the signal reconstructed from the digital bit stream generated by the previous samplings - a "0" bit indicates a relative amplitude decrease. Figure 22-3 shows the functions of a conventional single channel delta modulator. Input filtering is used to attenuate undesired frequencies present at the audio input (low pass filter). At each sampling time, the comparison function compares the filtered input with the output of the digital decoding function which has reconstructed the signal present during the last sample time. The digital encoding function provides a "0" or "1" signal dependent on the output of the comparator.

If the slope (magnitude) of the analog signal is too great, however, the reconstructed signal will lag the analog input. This condition is called slope overload. Slope overload can be corrected by either taking more samples or increasing the magnitude of the step error signal. Slope overload occurs when signal levels are too large. When they are too small, delta modulation systems exhibit a second problem known as idle noise. As with slope overload, the most satisfactory approach to solving idle noise problems is through gain control techniques.

Continuously Variable Slope Delta (CVSD) modulation is a form of delta modulation which attempts to alleviate the problems of slope overload and idle noise by controlling the size or gain of the step, the error signal, over a continuous range. All NATO countries have agreed to incorporate CVSD as their standard military A/D conversion technique. The TRI-TAC program will use this method. A function block diagram for a CVSD converter is shown in figure 22-4.

The gain of the error signal is varied by monitoring three consecutive bits in the digital bit stream. The gain is adjusted when the three bits are alike. The method depends on the fact that, when the input level to the encoder is increased, more runs of three consecutive digits of the same polarity are produced at the digital output. The CVSD encoder operates by comparing the filtered input speech waveform with the feedback approximation from the loop integrator. The output of the analog comparator is a logic ONE if the speech input voltage is more positive than the feedback voltage and a logic ZERO if it is less. The output of the comparator is shifted through a three-bit shift register. The decoder circuit detects whether these three bits are either all ZEROs or all ONEs. If they are either, a logic ONE is produced at the decoder output. The syllabic smoothing integrater is a filter which smooths the decoder output pulses and produces a control voltage Vc. The pulse height modulator produces a pulse whose amplitude is a linear function of Vc and whose polarity is a function of the bit stored in the first stage of the shift register. A stored "1" produces a positive pulse - a stored "0" produces a negative pulse. The loop integrator produces the approximated speech input signal by integrating these height-modulated pulses.

As the amplitude and, hence, the slope of the speech input signal increases, a greater number of runs of three consecutive like digits is produced. This produces more ONEs at the decoder output and, therefore, the control voltage Vc increases. This increases the peak-to-peak output voltage of the pulse height modulator and enables the feedback approximation to more accurately follow the input speech signal. When the approximation signal exceeds the input waveform, the overload string of ONEs is broken. When this occurs, a "0" level potential is fed to the syllabic filter by the logic, forcing a decrease in Vc and the feedback pulse amplitude. For a sampling rate of 32 kb/s, a variable

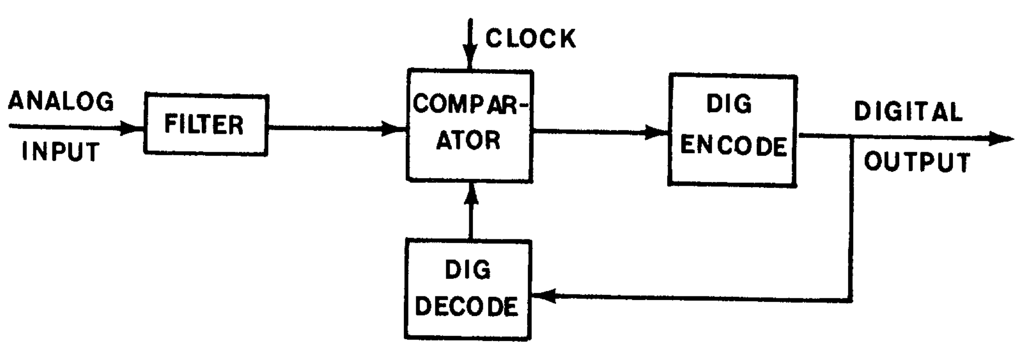


Figure 22-3. Delta Mod Encoder.

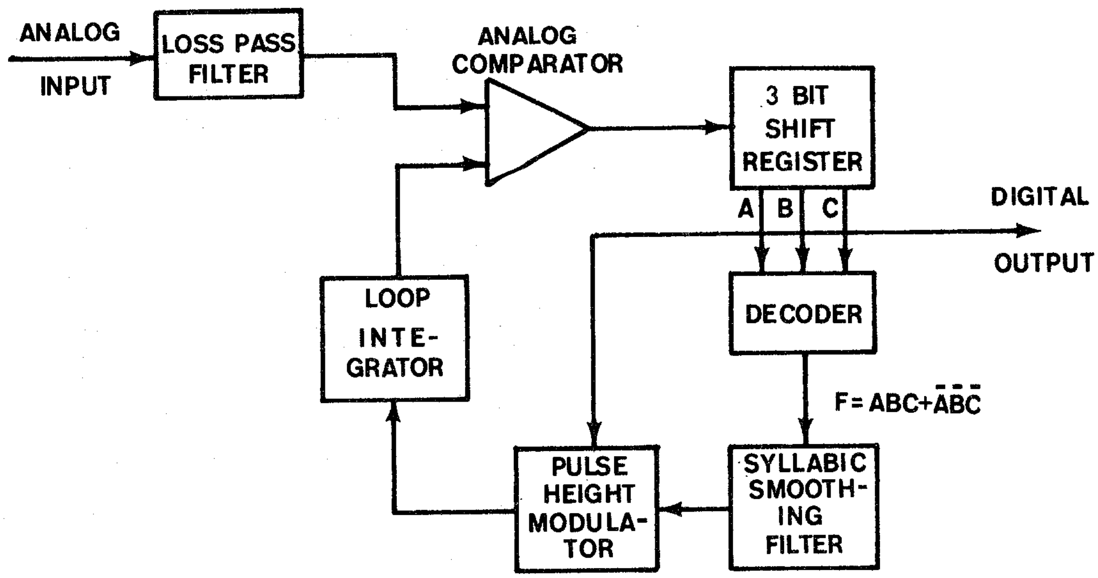


Figure 22-4. CVSD Encoder.

gain of at least 35 dB is required of the pulse height modulator to enable CVSD to decode a wide range of input speech levels.

Figure 22-5 shows a typical CVSD decoder. The decoder works identically to the encoder with identical characteristics except that it does not have a feedback loop and does not require the analog comparator. A low pass filter is included at the output of the decoder to remove quantizing noise and high frequency components above 4 kHz.

Present CVSD systems operate at a 32 kb/s clock rate. It is hoped that advances in the state-of-the-art will allow 16 kb/s conversion within ten years. CVSD has become increasingly popular for several reasons. Because it utilizes only a single bit to indicate relative amplitude instead of a 6 or 8-bit code word, indicating absolute amplitude as in PCM, CVSD is inherently less vulnerable than PCM to bit errors introduced in the digital transmission path. With PCM, a single bit error gives an absolute amplitude error which can vary nearly the full dynamic range of the converter; this value does not depend on the amplitude of the original signal, but only on the position of the error in the code word. In CVSD, however, a bit error causes only a small amplitude deviation proportional to the magnitude of the original analog input signal. An error rate of 10^{-3} is hardly perceptible to the user. Because of this immunity to bit errors, CVSD systems can allow some degree of bit slip. Bit slip occurs when the clocks for the decoder and encoder are not perfectly synchronous. Differing clocks have the effect of adding and deleting an occasional bit and give the effects of a bit error in the transmission path. This property, not obtainable in a PCM system, allows the two ends of a transmission path to operate off two different clocks so long as they are reasonably accurate.

CVSD systems also do not need word synchronization. As mentioned above, PCM systems must contain synchronization bits in the code words. This has the effect of increasing the bit rate, as well as increasing the need for logic in the decoding circuitry.

22-4. **Spread Spectrum.** This term refers to a group of modulating techniques whereby the transmitted bandwidth is deliberately greater than the bandwidth of the information, often by a factor of 10,000 or more. With conventional techniques, the transmitted bandwidth is a direct result of modulation by the information to be sent and is narrow by spread spectrum standards. The signal energy is, therefore, concentrated in a relatively narrow frequency band. With spread spectrum, this signal energy is distributed over a very wide frequency range and the energy in any one small portion of the spectrum is extremely low (often below the noise level). By employing the same modulating function used to generate the spread spectrum signal, a receiver can "collapse" this signal back into a narrow (IF) bandwidth for processing. There are a number of ways to generate and receive spread spectrum signals. Two methods are:

a. In direct sequence systems, DSB/SC modulation of a single carrier frequency is employed. A common way of accomplishing this is by shifting the phase of the carrier in time with a binary sequence (PSK). For example, a shift from a "0" to "1" (or vice versa) in the binary code sequence could reverse the phase of the carrier, that is, shift it by 180°. The bit rate of such binary codes may range from one to several hundred megahertz. The resultant signal spectrum has a main lobe centered at the carrier frequency having a bandwidth of twice the binary code clock rate and side lobes whose null-to-null bandwidth is equal to the clock rate (figure 22-6). The envelope of this spectrum is identical to the mathematical function $[(\sin X)/X]^2$. The actual shape of the spectrum for a direct

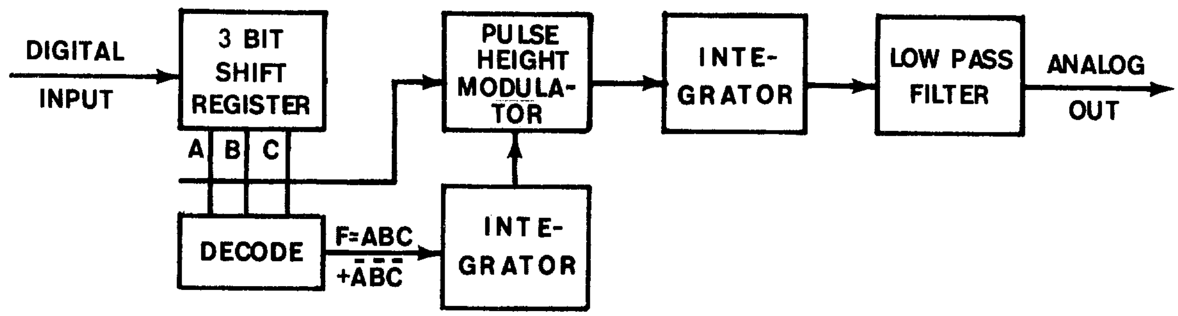


Figure 22-5. CVSD Decoder.

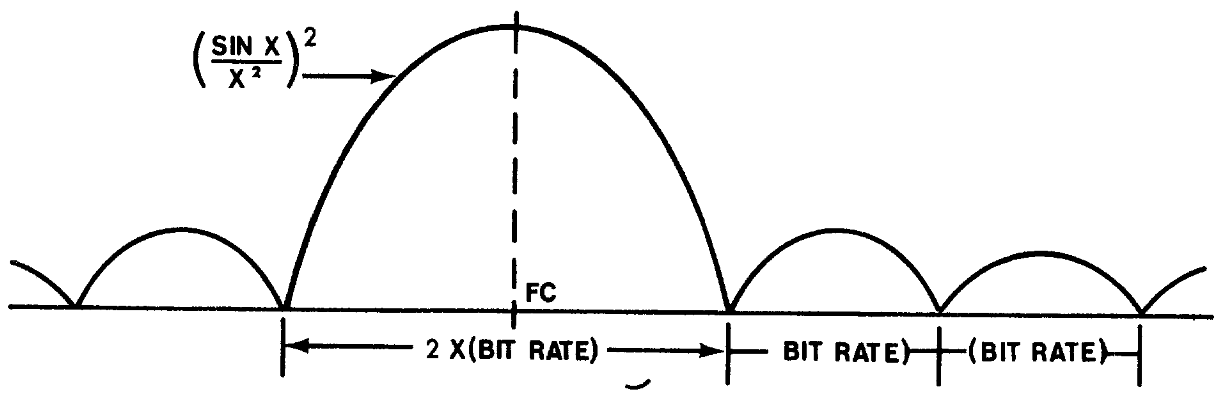


Figure 22-6. Direct Sequence Signal Spectrum.

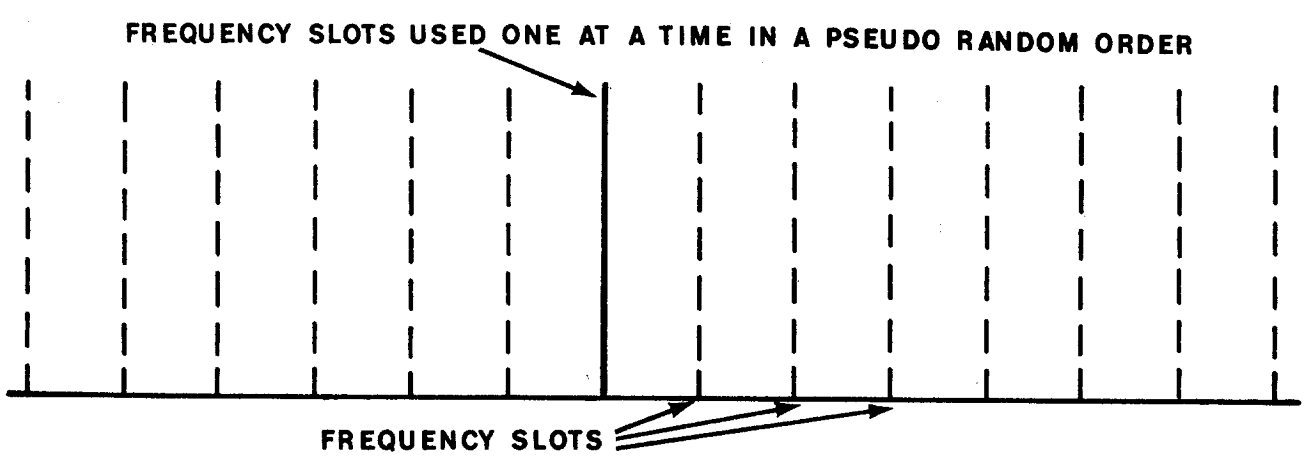


Figure 22-7. Frequency-Hopping Signal Spectrum.

sequence system depends on the type of modulation employed. In addition to PSK, PSK and pulse modulation may be used. To demodulate a PSK direct sequence signal, the receiver must generate a duplicate pseudo-random code to phase-reverse the local oscillator precisely in step with the desired arriving signal. The IF portion of the receiver thus "sees" a steady continuous wave (CW) signal modulated by the information signal. All other arriving signals not in step with the local oscillator appear as harmless wideband noise, only a small portion of which passes through the relatively narrow IF bandwidth. The level of this "noise" is generally far below the level of the desired signal.

b. In a frequency-hopping system, the transmitted signal "hops" from one assigned frequency slot to another in a specified order and at a specified rate determined by a pseudo-random binary code sequence. A duplicate pseudo-random code is generated in the receiver to shift the local oscillator precisely in step with the desired arriving signal. The IF portion of the receiver thus "sees" a steady CW signal modulated by the information signal. As in the case of the direct sequence system, unrelated signals appear as harmless wideband noise. A typical frequency-hopping spectrum is shown in figure 22-7. The width of the spectrum is essentially determined by the highest and lowest frequency slots employed.

Spread spectrum techniques offer significant advantages over conventional techniques. Since the signals in any one small portion of the spectrum may be so small as to be virtually undetectable in the presence of noise (such signals often appear as noise themselves), the transmission of a spread spectrum signal as a whole is undetectable. Even if the presence of these signals were known, the information could not be extracted unless the exact modulation code were known. Thus, security and privacy are inherent features with spread spectrum. Resistance to interference and jamming are also major advantages. Interference tends to be relatively narrowbanded and, as such, destroying a piece of the spread spectrum signal has negligible effect on the whole. The same argument applies to narrowband jamming. Wideband jamming signals would appear to be noise-like to a spread spectrum receiver and the desired signals could still be extracted with little degradation. The main disadvantages of spread spectrum systems are the required complex circuitry and wide bandwidth.

The question now arises, "Why do we need a digital system?" Why not continue to use the existing analog radios? Digital systems offer several advantages. Perhaps the most important one is the ability to bulk-encrypt the traffic and thus deny its access to those that might use it for ulterior motives. The only practical way to encrypt a voice circuit is to digitize it first. A second reason is the ability to regenerate digital signals. As an analog signal is processed through a series of relay sites, each radio link and each piece of equipment it passes through adds additional noise. It is impossible to remove the noise from the intelligence because there is no way to tell one from the other. If small amounts of noise are continually added, a point will be reached where the signal is no longer usable. In contrast to this, as small amounts of noise are added to a digital signal, the corners of each pulse may be rounded or otherwise misshaped, but it is still possible to tell if the signal level is a "mark" or a "space;" therefore, it is possible to reshape the pulses (this is called regeneration) at intermediate points along the circuit and have a signal arrive halfway around the world as clean and sharp as it started out. A third advantage of digital over analog systems can be seen by the effect that RSL has on channel noise (figure 22-8). Note that at high RSL, the analog system is better. This is because the quantized signal is not a true replica of the original, as shown in figure 22-3; however, as the receive signal deteriorates, channel noise on the analog system immediately starts to degrade the system. In contrast to this, on the digital system, there is very little degradation as long as the radio is able to tell the difference between "mark" and "space" pulses. Once this point is reached, of course, the circuit deteriorates rapidly. This feature of digital systems makes it very valuable on marginal links. The last advantage of the digital system that we will mention is economy. Because of the reduced filtering requirements in the multiplexing equipment, there is a potential for considerable savings in installation costs.

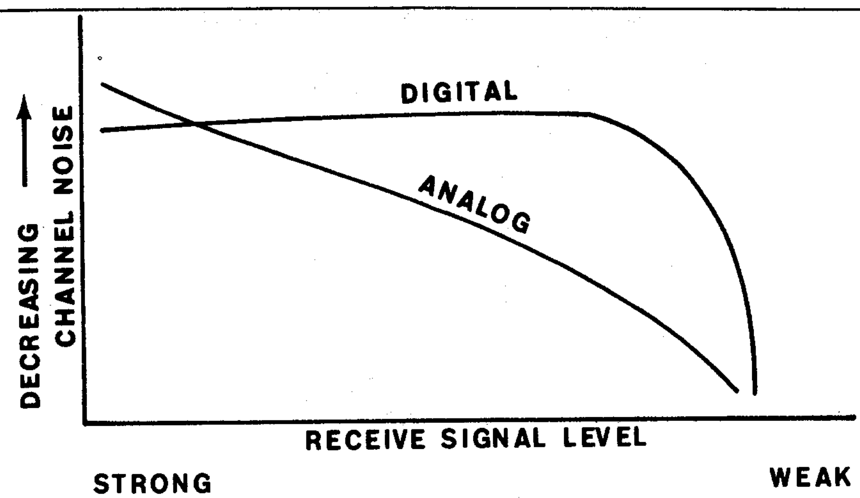


Figure 22-8. A Comparison of Channel Noise and RSL for Digital and Analog Radios.

22-5. Satellite Communications Systems. In 1957, the Soviet Union successfully launched the first man-made earth satellite. This demonstrated man's ability to place objects into an orbit around the earth and marked the beginning of the realization of the tremendous potential of satellite communications. The following paragraphs deal with some of the capabilities, limitations, and considerations involved in satellite communications systems.

a. Basics:

(1) Satellite Communications Link (Figure 22-9). The satellite serves as a relay station between ground stations and is analogous to an intermediate repeater in terrestrial LOS M/W links. Early communications satellites were passive devices in that they only provided a reflective surface to redirect radio waves. The satellites used today, however, are active transponders. They have the capability to receive, amplify, and retransmit signals from ground stations. The major advantage provided by a satellite link is that it is particularly well-suited for long distance communications. A single satellite can provide the link between ground stations separated by thousands of miles which may have otherwise required numerous terrestrial repeaters.

Indeed, because the satellite is visible over large portions of the earth's surface, it may be the only practical means of communication between remote locations.

(2) Satellite Communications Networks. Unlike terrestrial M/W repeaters, communications satellites are designed to serve multiple ground stations. Techniques which make this possible will be discussed in subsequent sections on multiple access techniques. Figure 22-10 is a block diagram of a typical satellite communications network. In this simple illustration, earth terminal 1 transmits one uplink carrier which is received by both terminals 2 and 3. Terminal 2 transmits two carriers of which one is received by terminal 1 and the other by terminal 3. Terminal 3 transmits three carriers, one to terminal 2 and the other two to terminal 1. It is apparent from this illustration that satellite networks offer tremendous potentials and flexibility. By reconfiguring the ground stations, it is readily possible to change the network configuration. Normally, all that is required to establish a new link is to retune the ground station receiver to the distant terminal's corresponding down link frequency. In systems where more than one satellite is used, it is only necessary to redirect the antenna toward the other satellite and adjust the ground station for the appropriate transmit and receive frequencies.

There are, however, many problems associated with satellite networks which must be considered. These problems stem mostly from the satellite transponder. Power limitations of the transponder, coupled with long transmission paths, result in low RSLs at the earth terminals. This, in turn, complicates the design of the earth subsystem and increases the cost. Low noise receiver front ends and large high gain antennas are required to provide high quality communications. Power limitations of the transponder also limit the number of users of the satellite. As more and more users access the satellite, the amount of transmitter power available to each user is decreased, thus reducing the quality of the communications provided. Ground station design is further complicated by the fact that the satellite is not stationary. This is true even of synchronous satellites although their movement is minimal. The earth terminal must be capable of tracking or constantly keeping its antenna directed toward the satellite. With synchronous satellites, this capability is required due to the narrow beamwidth of the terminal's antenna. For large antennas, the mechanics and electronics involved with a tracking antenna greatly increase the cost; also, since the satellite is inaccessible once it is placed in orbit, the transponder must be designed to provide maintenance-free reliable operation. Finally, the aspect which makes a satellite system desirable can also create problems. The visibility of the satellite over large portions of the earth's surface makes it susceptible to jamming by hostile forces. In subsequent paragraphs, we will see some of the techniques used to overcome these problems.

b. Space Subsystem:

(1) Orbital Mechanics. Before discussing the satellite transponder, it is worthwhile to briefly review some of the principles associated with orbiting bodies. Figure 22-11a represents a satellite orbiting the earth in an elliptical path. As the satellite moves through its orbit, its instantaneous velocity changes with respect to its position. At the apogee, or maximum distance from earth, the satellite velocity is minimum, while at the perigee, or minimum distance from earth, the velocity is maximum. The rotation of the earth, the apogee and the perigee, and the inclination of the orbit all affect the percent of time and the locations on the earth where the satellite can be used for communications. (The inclination is the angle that the orbit makes with respect to the equatorial plane. It is measured at the ascending node of the orbit, where the ascending node is defined as the point where the satellite crosses the equatorial plane from the south.)

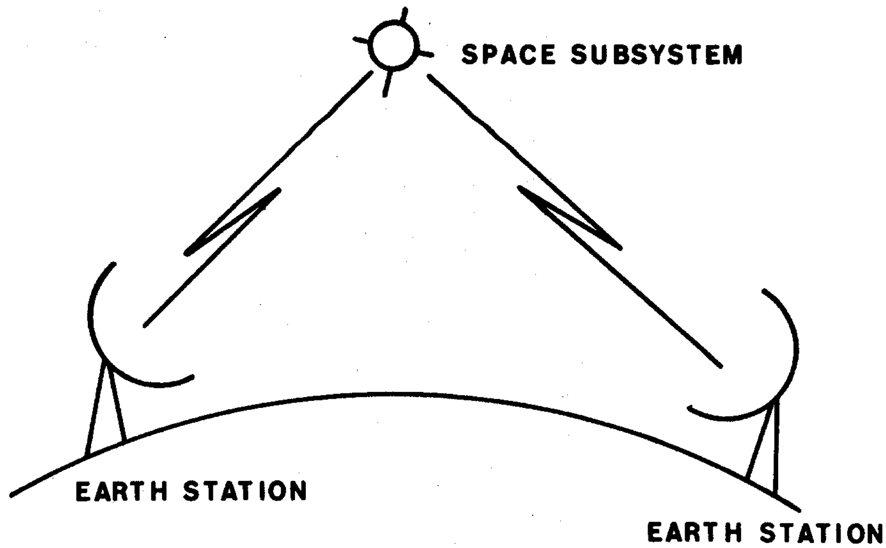


Figure 22-9. Basic Satellite Communications Link.

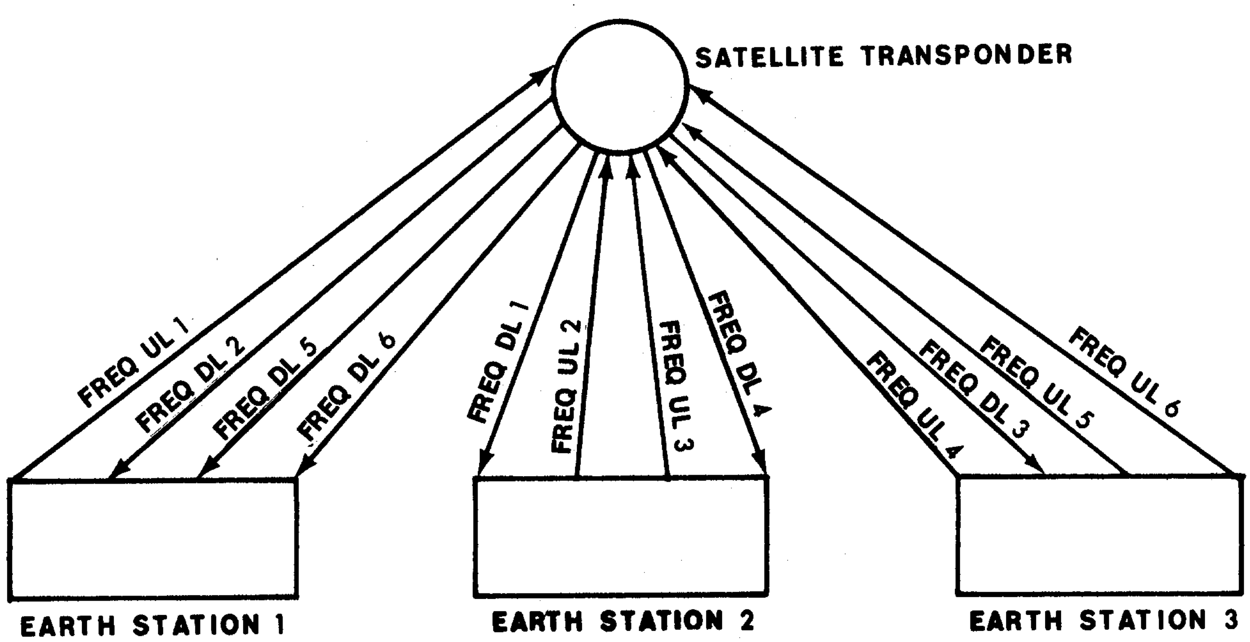


Figure 22-10. Satellite Communications Network.

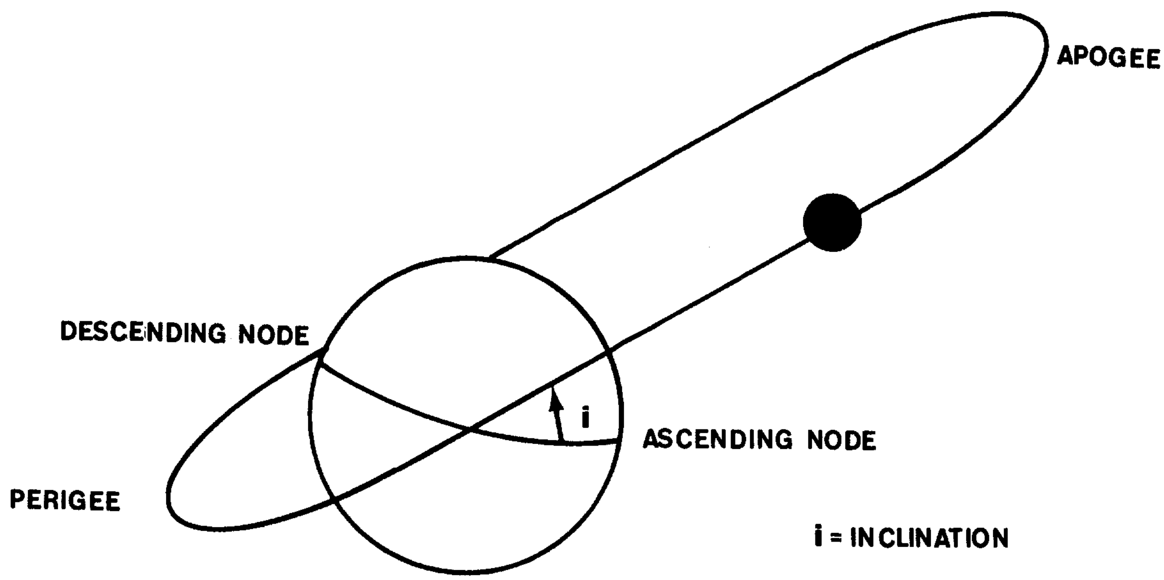


Figure 22-11a. An Elliptical Orbit.

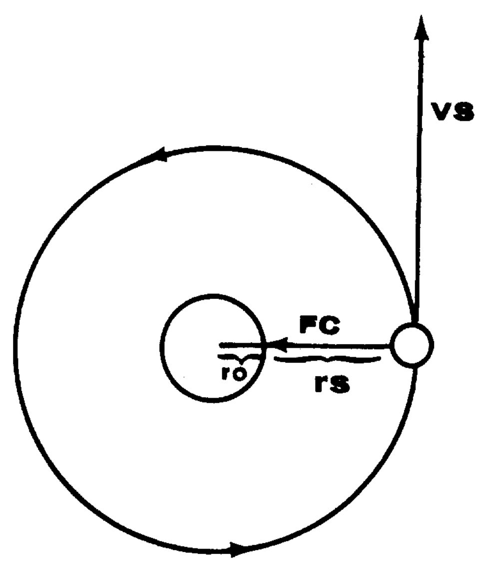


Figure 22-11b. A Circular Orbit.

If we assume a circular orbit (figure 22-11b) (that is, a circle is a special case of an elipsoid), it is possible to show some of the principles applicable to orbiting satellites. With no external forces acting on a satellite, it would travel in a straight line; however, if a force was applied, the satellite direction would change. It can be shown that a continuous force applied perpendicular to the velocity direction will cause the satellite to move in a curved path. The degree of the curvature is dependent on the magnitude of the force applied and the speed and mass of the satellite. For a circular path, the magnitude of the force required is given by Eqn 22-1.

$$F_c = \frac{m_s V_s^2}{r_s} \qquad \text{Eqn 22-1}$$

r_s = radius of circular path

m_s = mass of satellite

V_s = velocity of satellite

F_c = centripetal force

This force is called the centripetal force and must always be directed toward the center of the circle.

Assume now that it is desired to place a satellite in a circular orbit about the earth at some altitude r_s. For simplicity, we will assume that the only force acting on the satellite is the gravitational force exerted by the earth. (In reality, there are other forces which must be considered such as atmospheric drag, fluctuations in the earth's density, etc.) The gravitational force between the earth and the satellite is proportional to the masses of the satellite and the earth and inversely proportional to the square of the distance between them. Eqn 22-2 states this relationship mathematically.

$$F_g = \frac{G m_e m_s}{(r_s + r_o)^2} \qquad \text{Eqn 22-2}$$

G = gravitational constant (6.673 X 10^{-8} $cm^3/gm\text{-}sec^2$)

r_o = earth radius (6,356,912 mi)

r_s = satellite altitude

m_e = mass of earth (5.983 X 10^{24} kg)

m_s = satellite mass

By properly selecting the satellite velocity, the centripetal force required to produce a circular orbit at an altitude r_s can be made equal to the gravitational force between the earth and the satellite. This velocity can be determined from Eqn 22-3 which was derived from Eqns 22-1 and 22-2.

The perios (T), or the time required to complete one revolution, can then be computed from the circumference of the orbit and the velocity of the satellite. Eqn 22-4 may be used to calculate the period of a satellite with a circular orbit.

$$V_s = \sqrt{\frac{G m_e}{r_s + r_o}} \qquad \text{Eqn 22-3}$$

$$T = 2\pi \sqrt{\frac{(r_s + r_o)^3}{G m_e}} \qquad \text{Eqn 22-4}$$

For a satellite to have a synchronous orbit (that is, an orbit in which the satellite would appear stationary as viewed from the earth), two conditions have to be met. First, the orbit must be circular and in the same plane as the earth's equator. Second, the period of the orbit must equal the time required for one earth revolution. With these constraints and using the previous equations, the altitude required for a synchronous satellite is approximately 22,250 miles and the velocity would be about 6,865 miles per hour. If the satellite were moved out of the equatorial plane but still given a circular orbit with a period of 24 hours, it would appear to move in a figure 8 as viewed from the earth.

(2) Satellite Coverage. Transponders which do not have a storage capability require earth terminals to have a simultaneous view of the satellite. The coverage, or portion of the earth's surface where simultaneous view of the satellite is possible, is a function of not only the orbital characteristics, but also of the satellite antenna beamwidth. The area of coverage for a satellite at an altitude r_s with a beamwidth of 2x can be calculated by the following relation (figure 22-12a):

$$A_s = 2\pi r_o^2(1 - \cos\theta) \qquad \text{Eqn 22-5}$$

where $\theta = \sin^{-1}((1 + \frac{r_s}{r_e})\sin \quad) -$

$= \text{beamwidth}/2$

$r_e =$ earth radius

$r_s =$ satellite altitude

For synchronous satellites, the coverage area remains stationary. For nonsynchronous orbits, the area of coverage moves about the earth's surface. Figure 22-12b shows the area over which a satellite path may cross as a function of the orbit inclination. As the inclination (i) increases, this area increases. For polar orbits (that is, inclination =90°), the path may cross over the entire earth's surface.

(3) Satellite Transponders. The satellite transponder is analogous to a repeater in a terrestial communications link; it must receive, amplify, and retransmit signals from earth terminals. Figure 22-13 is a block diagram showing three typical types of satellite transponders. The particular type of transponder used is based on bandwidth and gain requirements. Usually, narrow bandwidth requirements are implemented using double frequency conversion or processing type transponders, with the latter providing some improvement against jamming.

Satellite transponders provide one or more RF channels with each channel normally required to receive and relay several simultaneous signals. With few exceptions, communications transponders have a saturating non-linear input-output characteristic which results in some intermodulation distortion when two or more signals enter the satellite. The magnitude of the intermodulation products are held to acceptable levels by various means. Bandpass filters are employed to channelize the transponder in order to reduce intermodulation distortion. Judicious selection of up-link frequencies also helps reduce distortion. By properly selecting frequencies, it is possible to cause a large portion of the intermodulation products to lie outside of the transponder's operating bandwidth.

Another method used to reduce intermodulation distortion is to operate the transponder at a back-off point below saturation. This reduces the intermodulation distortion; however, it also reduces the effective isotropic radiated power (EIRP). The EIRP is a function of the transponder's amplifier output and the gain of the antenna. We shall see later that the EIRP of a satellite bears significantly on the quality of the communications provided. Only a few dB less radiated power in a system which is already power-constrained can severly reduce system capabilities.

In addition to relaying signals between earth stations, satellite transponders usually provide a beacon signal which is used for acquisition and tracking by the earth terminals. The beacon signal may be generated internally or supplied to the satellite by an earth station. Many times, the beacon is modulated with telementry information necessary to perform stationkeeping and maintenance functions (that is, switching of redundant components in the transponder, orbital correction, repositioning, etc.).

The antenna component of the space subsystem is a key factor in performance and capability. If an antenna is to provide gain, it must focus and direct RF energy. This requires the satellite subsystem to have the ability to always orient its antenna pattern toward the earth. Depending on the method used to stabilize the satellite, there are several techniques employed to accomplish this. Where spin stabilization is used, a toriodal pattern or an electronically/mechanically despun antenna is usually employed. Where gravity gradient stabilization is used, the satellite does not rotate; therefore, the antenna can be mounted to point in the desired direction. Typically, the gains realized with satellite earth coverage antennas range between 2 to 20 dB with beamwidths of about 18°.

(4) Stationkeeping. This is the occasional readjustment of a satellite's position to maintain the desired orbital characteristics. It is necessary due to perturbing forces resulting from gravity irregularities, atmospheric drag, solar radiation, etc. Stationkeeping allows the selection of predictable links between earth terminals and facilities acquisition and tracking by narrow beam terrestrial subsystems. Normally, positioning is achieved by a system of gas jets which can be controlled via ground station commands.

c. Earth Terminal Subsystems:

(1) Antenna and Tracking Components. In many respects, an earth terminal is similar to a terrestrial M/W station. There are, however, some notable differences. The earth terminal antenna component must be capable of tracking a moving satellite. Where high gain requirements dictate large antennas and narrow beamwidths, the mechanics and electronics necessary for accurate antenna positioning can become very complex. Tracking errors of less than a tenth of a

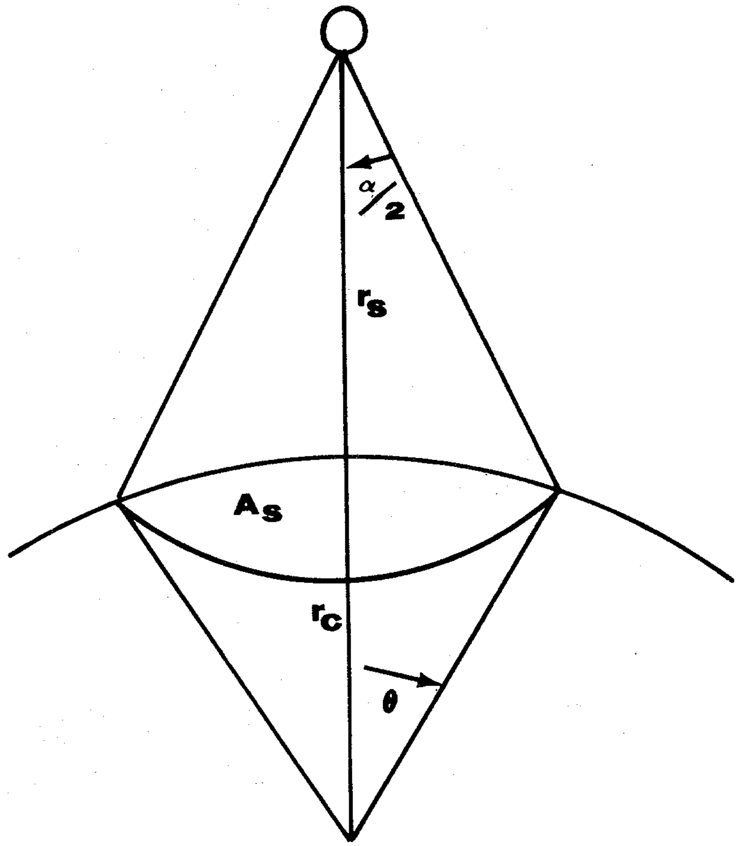


Figure 22-12a. Satellite Coverage. The coverage is a function of both the orbital characteristics and the antenna beamwidth.

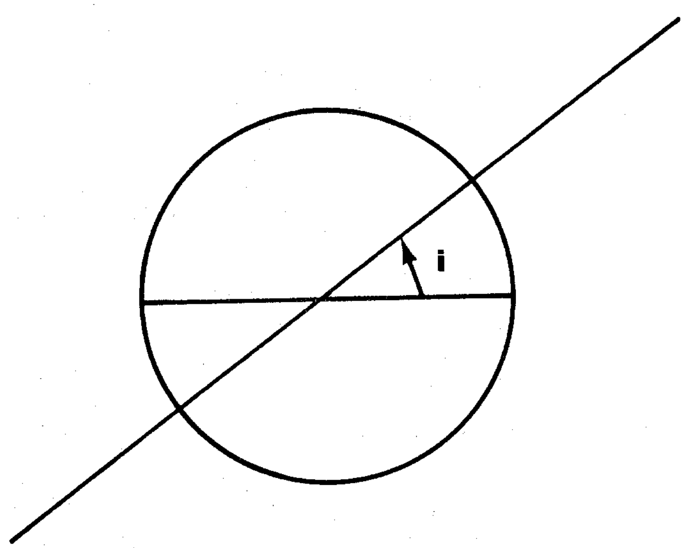


Figure 22-12b. Satellite Coverage.

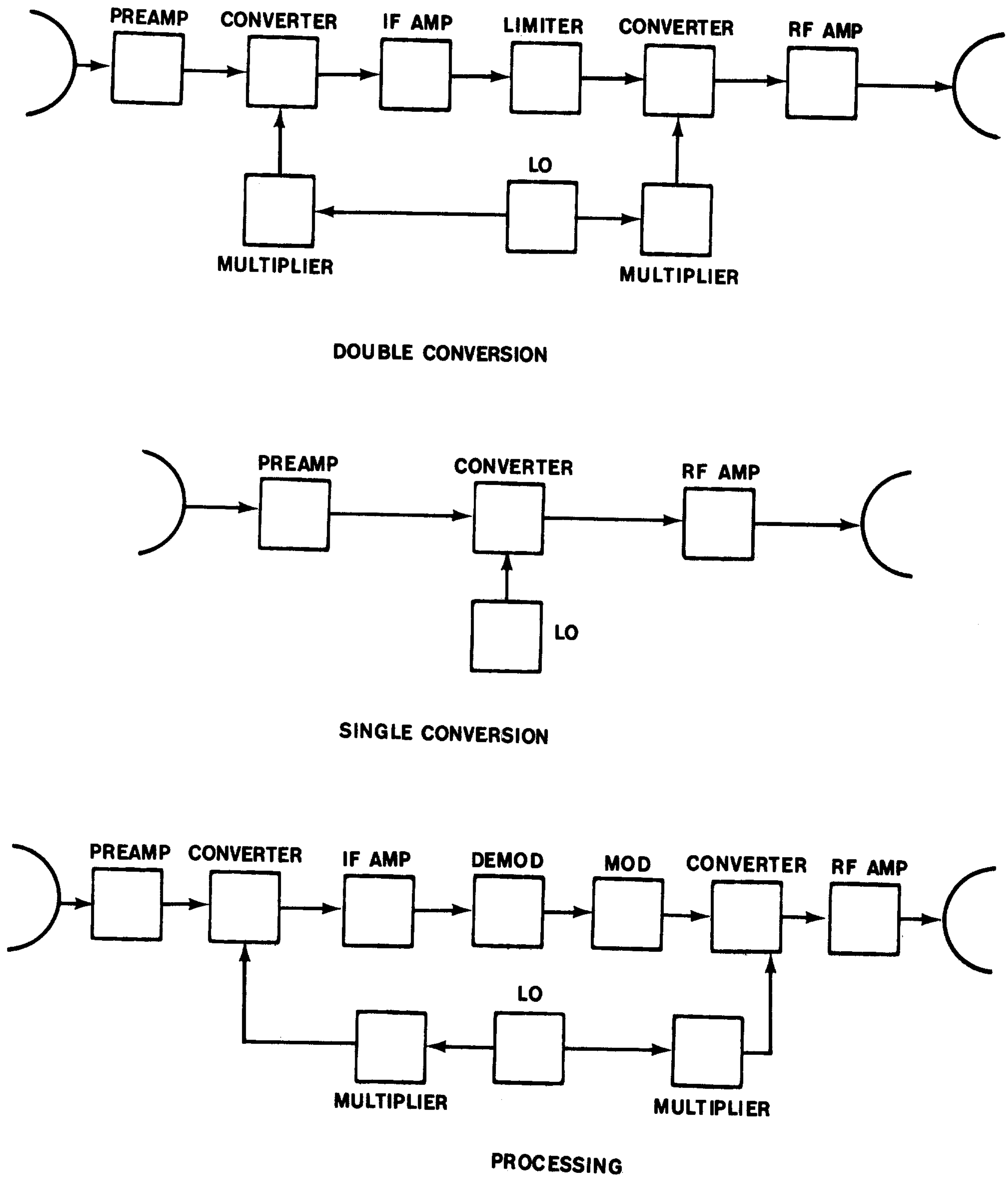


Figure 22-13. Basic Satellite Transponders.

degree can be disastrous to the quality of the communications the terminal provides. The signals received at an earth terminal are low level due to the long transmission path and relatively low transmitter power at the satellite. This means that the antenna must provide high signal gain while contributing low noise. The measure of the antenna's noise contribution is normally expressed in terms of antenna noise temperature. For large antennas, noise temperatures of about 30°K to 80°K are common, depending on the antenna elevation angle.

Antenna polarization is also an important consideration. Because satellites are normally used by terminals which may be geographically widespread, it is necessary to use a type of polarization in which polarization loss is independent of the look angle from the ground station to the satellite. Circular polarization has this property and is used in all satellite systems. Losses due to polarization mismatches, if vertical or horizontal polarization were used, could reach as much as 30 dB.

The most commonly used technique for tracking is monopulse tracking. Figures 22-14 and 22-15 show a basic monopulse system. Referring to figure 22-14, down link signals are received by a four-horn feed system. The signals from feeds X_1 and X_3 are summed by hybrid 1 and applied to hybrid 3. Similarly, signals at feeds X_2 and X_4 are summed and applied to hybrid 3. At hybrid 3, the $X_2 + X_4$ sum is shifted 180° with respect to $X_1 + X_3$ and summed. This results in a difference signal whose magnitude and phase is indicative of pointing errors in azimuth. Difference signals which are in-phase with the incoming signal indicate a tracking error to the right of the satellite while difference signals which are shifted 180° indicate an error in the opposite direction. The difference signal is zero when there is no azimuth error. In a similar manner, hybrids 1, 2, and 3 provide a signal whose magnitude and phase with respect to the received signal is indicative of elevation errors. The received signal is the sum of all four feed horns.

The sum signal and the elevation and azimuth difference signals derived from the feed system follow the path shown in figure 22-15. Difference signals are added sequentially to the sum signal through a ferrite switch and directional coupler. Since the difference signals are phase-coherent, the effect is to amplitude-modulate the sum signal. This amplitude-modulated signal is then amplified and down-converted to an IF which is then used to feed a tracking demodulator. The demodulator recovers the original azimuth and elevation errors signals which are then used to produce servo amplifier signals to position the antenna for minimum tracking error.

Other types of tracking systems include conical scanning (sequential lobing), step track, and programmed tracking. Step track systems move the antenna in selected increments to maximize the received signal strength. For example, if the antenna is incremented one step to the right and the signal level increases, then the next azimuth step will also be to the right. If the elevation were incremented up and the signal level dropped, the next step would be down. By alternating azimuth and elevation steps, the effect will be to maximize the signal strength. Programmed tracking is based on calculations of the satellite's path. Calculated position data is stored in a computer and used to update the antenna pointing angle on a periodic basis.

Conical scan tracking is shown in figure 22-16. In this technique, the antenna beam is continuously rotated using mechanical or electronic means. If the satellite is in the middle of the circle defined by the rotating beam, the RSL will remain constant. If, however, the satellite is at some point other than the center, the received signal's amplitude will vary. By correlating the amplitude variations with the instantaneous beam position, error signals can be derived and used to move the antenna in the direction necessary to minimize tracking error.

(2) Transmitter/Receiver Subsystem. The transmit subsystem of a typical earth terminal is similar to a terrestrial M/W transmitter in that it consists of frequency conversion and amplifier sections. It is different, however, in some respects. Most DCS terminals have multiple transmit carrier capability. The AN/FSC-78 earth terminal has, for example, the capability to transmit nine separate up-link carriers. The number of transmit carriers possible is dependent on the RF bandwidth capability of the amplifier sections, the number of frequency converters, and the intermodulation distortion products generated as a result of multiple input frequencies.

Figure 22-17 represents a typical DCS earth terminal receive subsystem. Again, the receive subsystem is similar to a terrestrial M/W receiver. There are, however, some differences which bear discussion. The RF amplifier in high quality earth terminal receivers is normally a low noise device. Typical noise figures range from .5 to 2.0 dB versus the 8 to 11 dB noise figures for normal terrestrial M/W receivers. Table 22-1 shows some of the amplifiers used in earth terminals along with representative noise contributions of each. The low noise front end amplifier is located as close as possible to the antenna. Normally, the receiver front end is mounted on the antenna pedestal. The purpose of this is to reduce the C/N degradation which results from losses between the antenna and the amplifier input. The necessity to reduce these losses can be shown with a few simple calculations.

The antenna noise temperature of a terrestrial M/W is normally about 290°K. If the loss between the antenna and the receiver input is 5 dB, the equivalent noise temperature that the receiver would see can be determined from Eqn 22-6.

$$T'_a = LT_a + (1-L)T_L \qquad \text{Eqn 22-6}$$

$$= (10^{\frac{-5}{10}})(290) + (1-10^{\frac{-5}{10}})(290)$$

$$= 290°K$$

Where, T'_a = antenna noise temp corrected for losses in °K

T_a = antenna noise temp in °K

L = waveguide loss between antenna and receiver in dB

T_L = temperature of waveguide in °K

Since a satellite terminal antenna is pointed toward the sky, the antenna noise temperature is considerably lower than terrestrial-based M/W. The actual noise temperature of the antenna varies with the elevation angle and frequency. If we use an antenna noise temperature of 35°K (this is typical of an FSC-78 terminal at look angles above 30° and operating in the 8 GHz range) and perform the same calculations as for the terrestrial system, the effect of losses is apparent.

$$T'_a = (10^{\frac{-5}{10}})(35) + (1-10^{\frac{-5}{10}})(290)$$

$$= 209°K$$

There is no degradation in the M/W system whereas in the satellite system the noise temperature of the antenna has been effectively increased by 174°K.

Another notable difference is satellite earth stations is that they are often equipped with multiple frequency converters enabling them to receive several down link carriers simultaneously. Some of the terminals used today in the DCS are capable of operation with as many as 16 separate down link frequencies. This capability is, of course, coupled with an appropriate large receiver RF bandwidth. Practically all of the earth terminals used in the DCS have RF bandwidths of 500 MHz.

d. Link Considerations:

(1) Link Parameters. The primary determining factor of a satellite link performance is the C/N ratio presented to the receiver. This ratio is a function of the transponder effective radiated power, path and transmission losses, receiving antenna gain, and the receiving system noise temperature. Eqn 22-7 shows the relationship of these parameters.

$$C/KT = EIRP - L_{fs} + \left(G/T_s\right) - k - L_o \qquad \text{dB/Hz} \qquad \text{Eqn 22-7}$$

where, C/KT = carrier-to-noise ratio normalized to a 1 Hz bandwidth. (Also termed carrier-to-noise density.)

EIRP = effective isotropic radiated power; the product of the transmitting antenna gain and the power amplifier output. Expressed in dBm it is $P_t + G_a$ where P_t is the transmitter output power and G_a is the gain of the antenna in dB.

L_{fs} = free space loss in dB.

G = receiving system antenna gain in dB.

T_s = receiving system noise temperature in °K.

L_o = transmission losses (that is, polarization loss, tracking error loss, etc.) in dB.

k = Boltzman's constant (1.38×10^{-20} mW-sec/°K)

Using the data shown in figure 22-18, the calculated C/KT would be:

$$C/KT = 55 - 201.7 + 40 + 198.6 - (.25 + .15 + .25)$$

$$= 91.25 \text{ dB/Hz}$$

As shown in Eqn 22-7, the term "G/Ts" can be extremely useful in describing the capabilities of the earth terminal. This ratio is called the terminal's "Figure of Merit" and determines the C/N ratio supplied to the receiver under a given set of conditions. In terms of satellite power usage, Eqn 22-7 means that, for higher G/T_S ratios, less transponder power is required to provide the same quality of communications. Where only a limited amount of power is available, the G/T_S of the ground station is a primary factor in determining the number of users that can effectively use the satellite.

Since the figure of merit is of such importance, it is worthwhile to look at one of the techniques used to measure this parameter. The system noise temperature term "T_S" is a measure of the noise contribution of the sky and ground system and, therefore, can most easily be measured using a Y-factor technique (chapter 20). This technique requires noise to be introduced into the system from an external source whose noise power is accurately known. There are several stars whose flux density is well-documented and predictable and, therefore, applicable to use as the source in Y-factor measurements. Some of the stars normally used are Cassiopeia A, Cynus A, and Taurus A. It can be shown that the terminal G/T_S is given by Eqn 22-8 where the Y-factor is the power ratio of the noise measured at the receiver output with the antenna pointed at the source and away from the source.

$$G/T_s = 10 \log \frac{[(Y-1)4\pi k]}{[(\lambda^2)(s)]} \text{ dB} \qquad \text{Eqn 22-8}$$

where,

k = Boltzman's constant

λ = wavelength at frequency of interest

s = flux density of celestial noise source corrected for frequency, losses, and source diameter to beamwidth ratio

Type	Noise Temperature (°K)
Maser	10
Cooled Parametric	35
Uncooled Parametric	120
Tunnel Diode	530
Schottky Mixer	1000

Table 22-1. Low Noise Amplifiers.

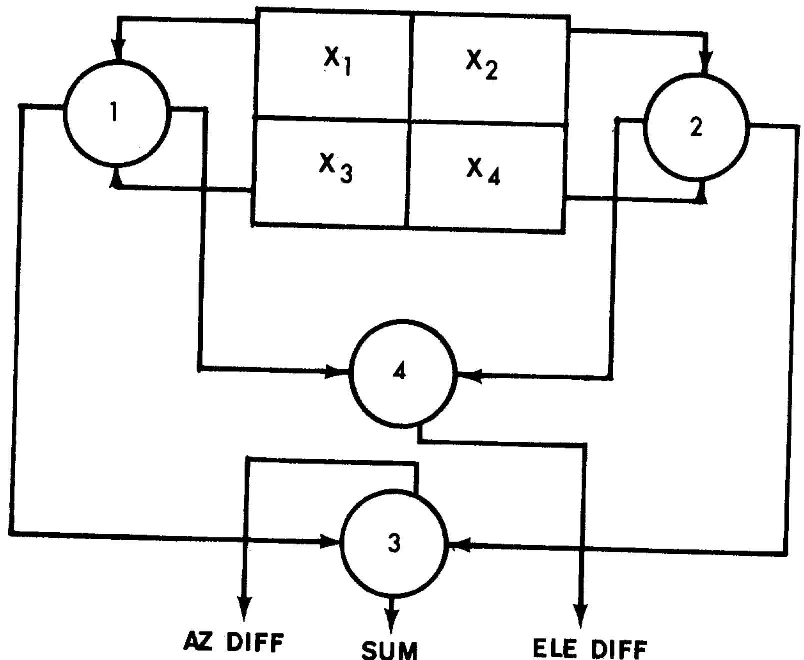


Figure 22-14. Monopulse Tracking Feed System.

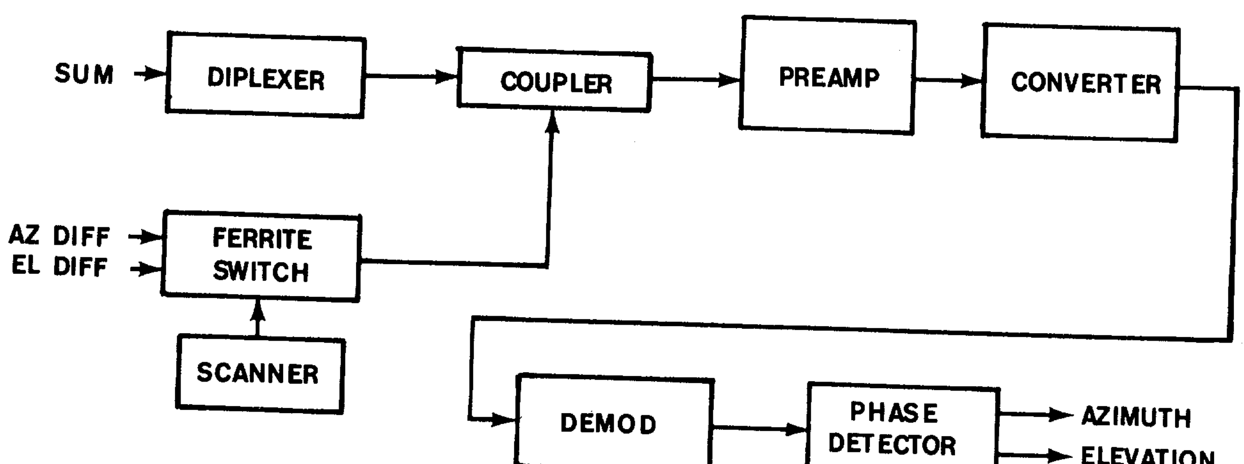


Figure 22-15. Monopulse Tracking System.

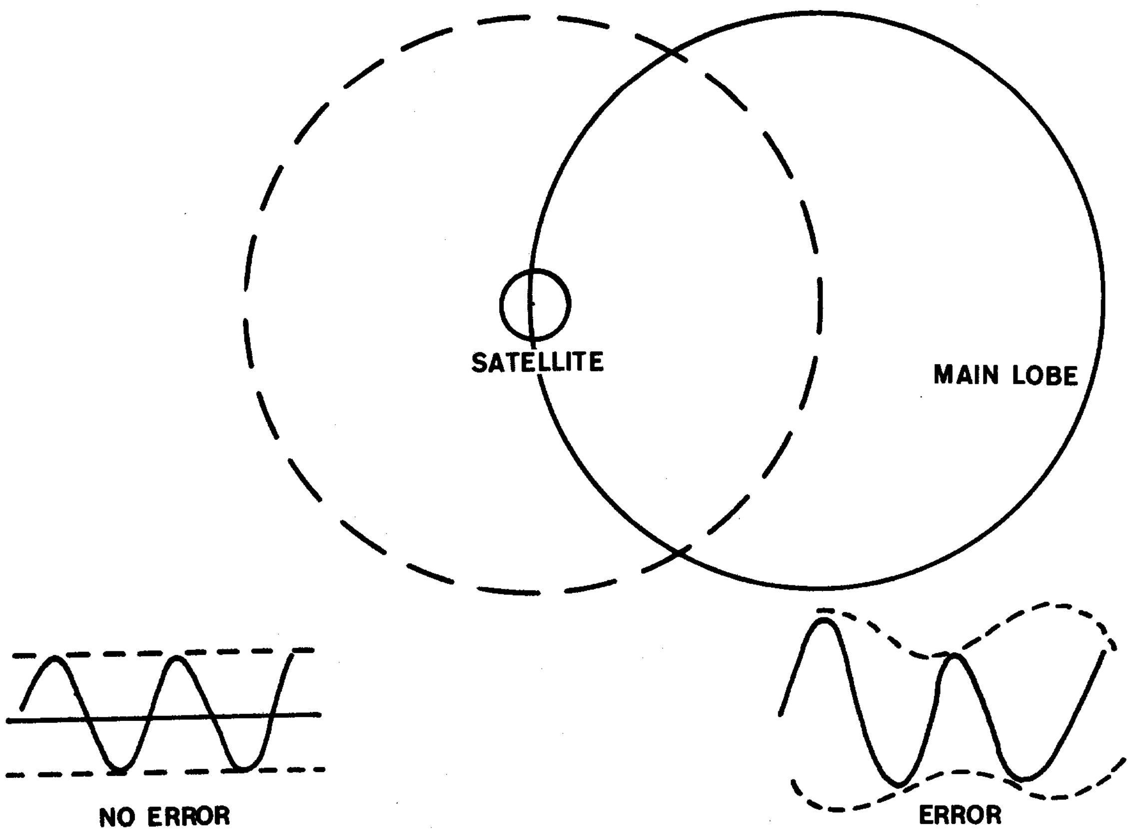


Figure 22-16. Conical Scan Principles.

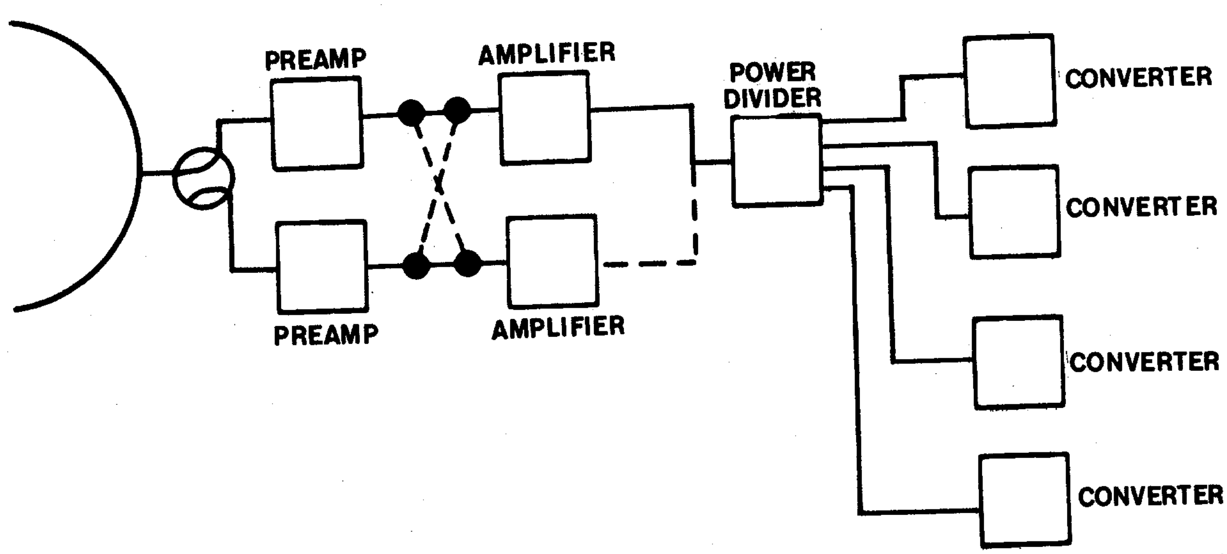


Figure 22-17. Basic Earth Terminal Receive Subsystem.

Thus far, we have only talked about one-way paths. Eqn 22-7 holds true for either the up link path or the down link path. The C/KT for the total path (that is, terminal to transponder to terminal) is affected to some extent by the C/KT at the satellite. For a linear transponder, the total path C/KT is given by Eqn 22-9.

$$(C/KT)_T = \frac{1}{\left(\frac{1}{C/KT_{UL}}\right) + \left(\frac{1}{C/KT_{DL}}\right)} \qquad \text{Eqn 22-9}$$

Figure 22-19 shows the effects of the C/KT at the satellite on the total path.

(2) Transmission Requirements. As mentioned before, the quality of the communications provided depends on the C/KT at the ground station receiver input. For an analog channel, this is shown in Eqn 22-10, where C/KT is some value within the linear region of the receiver's quieting characteristics.

$$S/N = C/KT + 20\log\frac{\Delta f_{rms}}{f} + P_e - 10\log CB_w \qquad \text{Eqn 22-10}$$

Where, Δf_{rms} = per channel RMS deviation

f = channel frequency

P_e = effective pre-emphasis at f

CB_w = channel bandwidth

S/N = channel signal-to-noise ratio

Unlike terrestrial M/W systems, satellite links are engineered to operate near threshold. Typically, the margins in a satellite link are about 6 dB vice the 30 to 40 dB fade margins normally found in M/W systems. The reason for this is again due to the limited power available from the satellite and the economic need to share the satellite transponder with the maximum number of users possible. FM threshold is a much more critical parameter in satellite links than in terrestrial systems. Eqn 22-11 may be used to calculate FM threshold in terms of C/N density.

$$C/KT_{threshold} = CNRTH + 10\log IF_{Bw} \qquad \text{Eqn 22-11}$$

where, CNRTH = carrier-to-noise ratio at threshold

IF_{Bw} = receiver IF bandwidth

Digital channel quality is usually assessed in terms of bit error rate. The performance of a digital system is, therefore, based on the ratio of the energy per bit to noise power density at the receiver required to obtain a given bit error rate. As shown in Eqn 22-12, if the characteristics of the receiver are known (that is, bit error rate versus energy per bit to noise power density), the C/KT required for a given bit error rate can be calculated.

$$E_b/N = C/KT\ (1/R) \qquad R = \text{bit rate} \qquad \text{Eqn 22-12}$$

One of the chief concerns in satellite link engineering is the percent of satellite power required to achieve a given circuit quality. Eqns 22-13 and 22-14 show the relationship of the required EIRP to a given circuit quality for both analog and digital channels.

$$S/N = EIRP - L_{fs} + G/T_s - k - L_o + 20\log(f_{rms}/f) + P_e - 10\log CB_w \qquad \text{Eqn 22-13}$$

$$E_b/N = EIRP - L_{fs} + G/T_s - k - L_o - 10\log R \quad \text{(dB)} \qquad \text{Eqn 22-14}$$

e. System Considerations:

(1) Multiple Access Techniques. As mentioned before, one of the advantages of satellite communications is that one satellite may relay information to several ground stations. There are several ways in which this multi-service capability may be achieved. Some of these are discussed below.

Frequency division multiple access (FDMA) is the most common technique used today and the most easy to implement. In this technique, each uplink carrier is assigned a separate frequency within the transponder's frequency band. The transponder acts like a common amplifier and frequency translator to relay signals back to the earth. One of the shortcomings of this method, however, is that the power available to retransmit signals must be shared by all users of the satellite. This reduces the power available for each down link signal. As the number of users accessing the satellite increases, the quality of the communications decreases.

Problems also arise from the fact that the traveling wave tube used in most satellite transponders is a hard-limiting device, that is, saturation occurs abruptly. When a hard-limiting device is driven into saturation by multiple input signals, the result is the generation of large amplitude intermodulation products. For efficient use, however, a TWT should be operated near saturation. In an FDMA satellite system, this necessitates strict power control over all users of

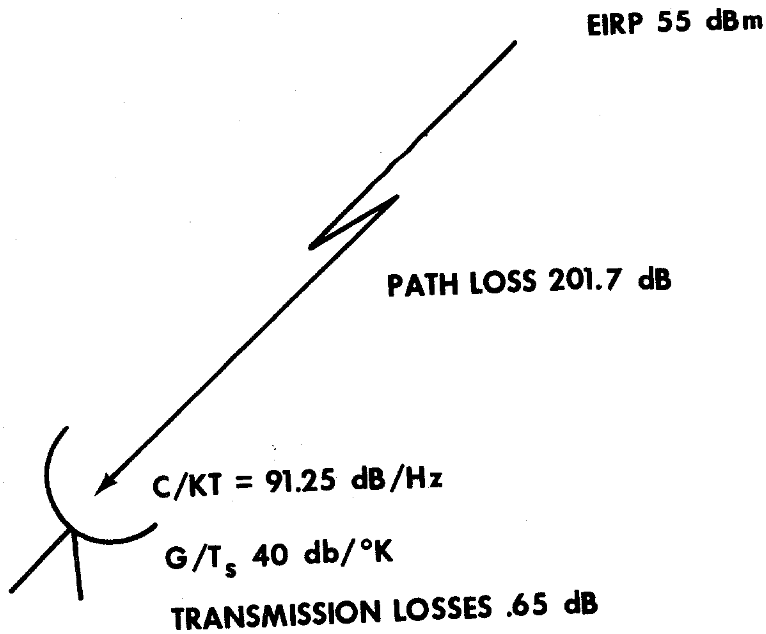


Figure 22-18. Down Link Parameters.

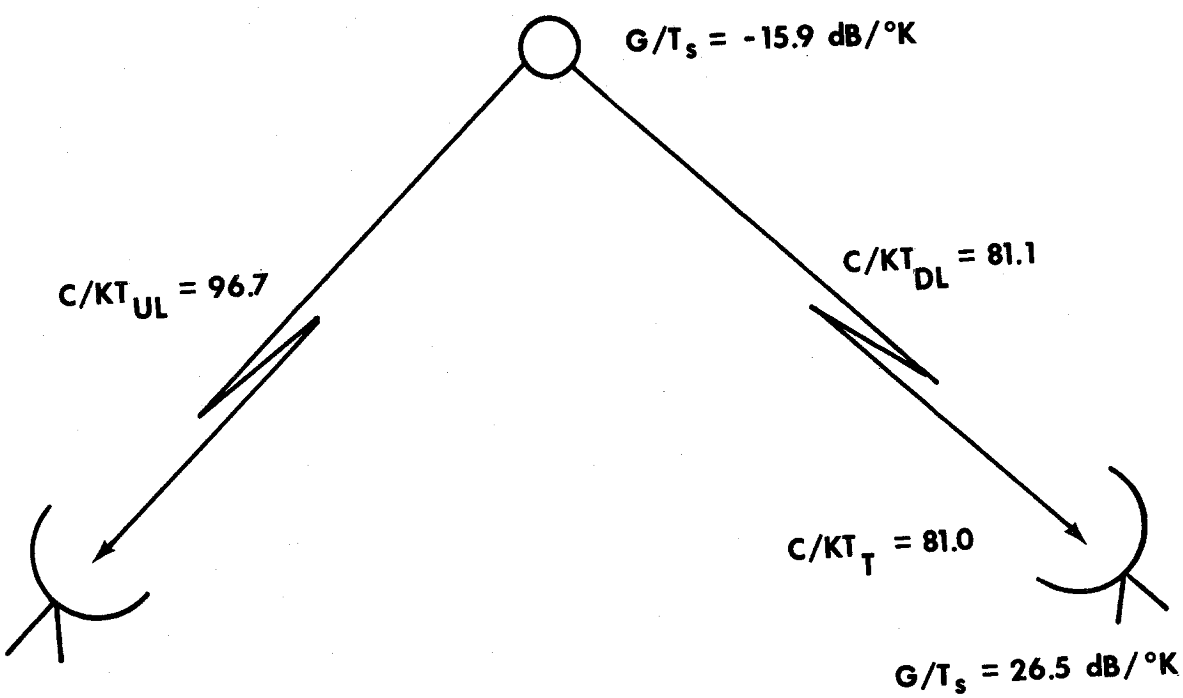


Figure 22-19. Total Path Carrier-to-Noise Density.

the satellite. Normally, in order to exercise adequate control and keep intermodulation products at acceptable levels, the transmit power from ground stations is monitored and maintained at an equivalent transponder input level which results in an output 1-3 dB below TWT saturation. This necessary back-off reduces the EIRP of the transponder and thus reduces the system capabilities. Another detrimental characteristic of the transponder, when used with FDMA techniques, is signal capture. Signal capture is the tendency of larger signals to suppress weaker ones. Although this does have some limited advantages where jamming protection is required, it can create problems in normal use. FDMA techniques also require careful selection of uplink frequencies to void the generation of unnecessary intermodulation products. By judiciously selecting frequencies, it is possible to cause many of the intermodulation products to fall outside the bandpass of the transponder.

The most efficient form of multiple access is time division multiple access (TDMA). This is a technique in which each earth terminal is given exclusive use of the satellite transponder for a specified interval of time. This principle is shown in figure 22-20. Assuming there are four ground stations utilizing the satellite, this figure shows that each one is allowed a specific time interval in which to transmit information. All of the terminals have access to all of the down link transmissions but only process information during the interval in which the desired distant terminal is transmitting.

One of the problems with TDMA is that it requires accurate network timing and all terminals must be in sync with each other. Another problem is that all information must be in digital form. This would require analog information to first be digitized, then transmitted and received, and finally converted back to analog. Since there is not a continuous information path between terminals, buffering or storage capability is also required at the earth stations. Even considering these problems, TDMA is the approach that most future systems will take.

Recalling Eqn 22-14, it can be seen that the energy per bit noise power density (E_b/N) is dependent on the terminal figure of merit. For terminals with low G/T_s s, the bit rate must be reduced in order to maintain the required E_b/N; thus, in a TDMA system, the time interval that different terminals have access to the transponder may vary, depending on the terminal figure of merit, the bit rate, and the through-put requirement. Figure 22-21 is an illustration of this.

Spread spectrum multiple access (SSMA) is similar to FDMA in that the satellite power is shared between users. The difference in this technique is that several users occupy the same frequency spectrum in the transponder simultaneously. Each user's carrier is modulated twice at the ground station; first by the information and then by a high frequency bandspreading signal. The bandspreading signal (usually digital) has the effect of dispersing the power in the carrier over a wide band of frequencies resulting in a spectrum that appears noise-like. The pattern of the spreading signal (the code) is a pseudo-random sequence. An identical code generated at the receiving terminal and synchronized with the transmitting end allows the information signal to be recovered. If different codes which show a low cross-correlation are assigned to different users, then transmissions by several users are possible with only a small amount of interference. This interference does, however, increase with the number of simultaneous users and, therefore, limits the number of ground stations sharing the same transponder. As with FDMA methods, the total power available also restricts the number of users.

(2) System Control. Any communications system requires some sort of system control for optimum performance. This is especially true in satellite systems where several users share a common transponder. The operation of each terminal directly affects all other users in the system. Misaligned or defective equipment at one ground station which results in the requirement to use more satellite power can result in degraded performance for the whole system.

Within a satellite system, the type of control required can be divided into two categories: satellite control and communications control.

(a) Satellite control consists of the actions necessary to maintain correct orbital characteristics and optimum transponder performance. Without periodic adjustments of the satellite's position, orbital characteristics would be unpredictable and system planning impossible. Likewise, it is just as important to be able to switch to redundant equipment in case of component failure or change amplifier gains within the transponder to meet customer demands.

(b) Communications control involves monitoring system performance and future planning. The input signal level to the satellite must be closely monitored to ensure that the transponder is not driven into non-linear operation. Likewise, frequency assignments within the system must be carefully planned to avoid unnecessary interference between users. In TDMA systems, synchronization must be monitored and maintained. Satellite loading and capacity must be judiciously monitored and planning for new communications requirements constantly considered.

22-6. Fiber Optics. The possibility of using fiber optics in modern communications systems shows great promise. Fiber optics provide a method of replacing electromagnetic waves and copper conductors with light waves and hair-thin glass fibers. Light can be transmitted through these fibers in much the same manner as a M/W signal is transmitted through a waveguide. By modulating the light, a single strand is capable of bandwidth capacities of 50,000 voice channels. Silica glass fibers offer several advantages over copper conductors. As just mentioned, the bandwidth of optical fiber is vastly superior to wire. This makes it possible to replace huge bundles of multiconductor wire with a small bundle of fibers. Since silica is one of the most abundant materials on earth and copper is much more scarce, we have great opportunities for considerable cost savings. Additionally, in applications

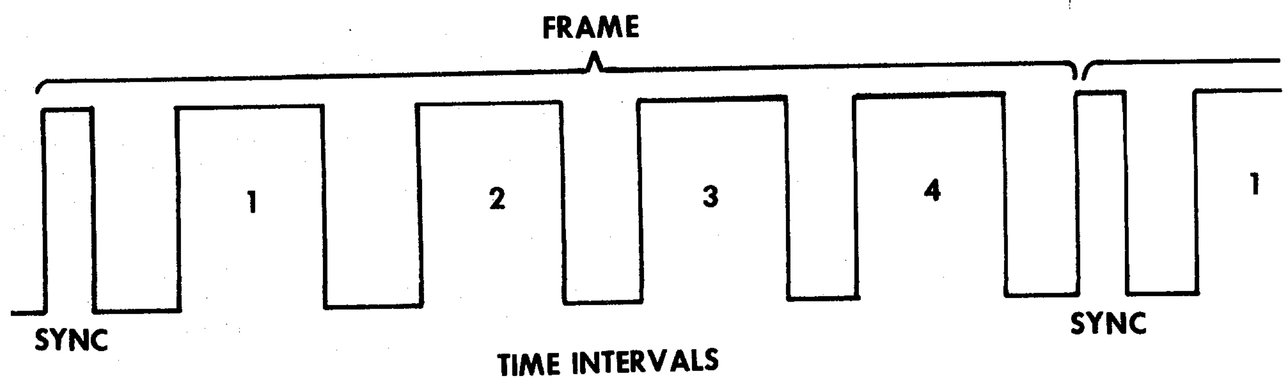


Figure 22-20. Time Division Multiple Access.

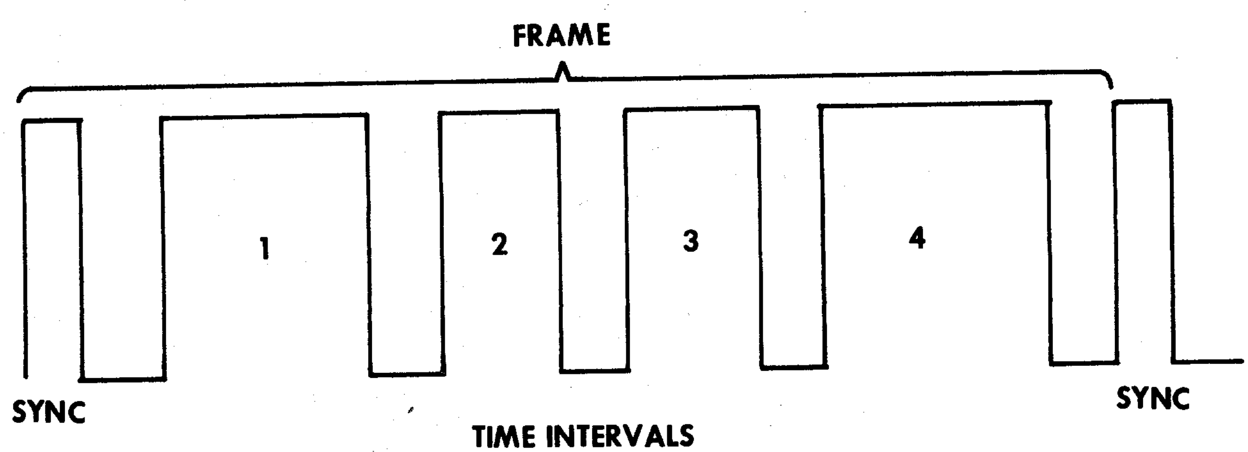


Figure 22-21. Unequal Time Intervals in TDMA System.

where it is important (such as in aircraft), fiber optics offer considerable advantages in weight savings. Another area where fiber optics excel is in freedom from crosstalk. Not only is this a valuable characteristic where circuits are required to traverse long distances within the same bundle but fiber optics have also proven to be of great benefit where it is necessary to suppress spurious electromagnetic radiation for security purposes.

There are several obstacles that must be overcome before use of fiber optics becomes widespread. One of these is overcoming the distance limitations in transmitting the light through the fiber. In the last few years, the loss has been improved from 20 dB per kilometer to two dB per kilometer. Additional advances are made in this area almost daily. If progress continues at its present rate, splicing techniques will be refined, additional hardware (possibly integrated optical circuits) will be developed and loss characteristics will be improved to the point where fiber optics have widespread use throughout the communications field.

Let's take a brief look at how waves are propagated through fibers. An optical communications system consists of a modulated light source, a path for the light signal, and a light detector/demodulator. Just as there are different modes for transmitting electromagnetic waves through a waveguide, there are also different modes for transmitting light through a fiber. Fibers are usually divided into three classes: the single mode/step index fiber, the multimode/step index fiber, and the multimode/graded index fiber. Figure 22-22 shows the construction of each type of fiber and how the index of refraction varies through the cross-section of each one. We can construct a simplified drawing of these waves traversing down a fiber (figure 22-23).

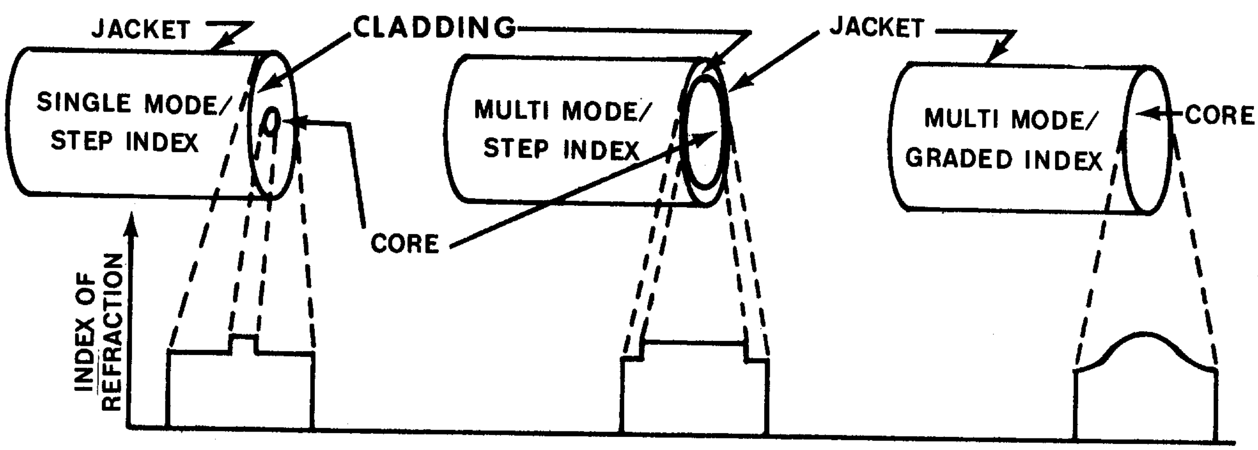


Figure 22-22. Three Types of Optical Fibers and Cross-Sectional Index of Refraction.

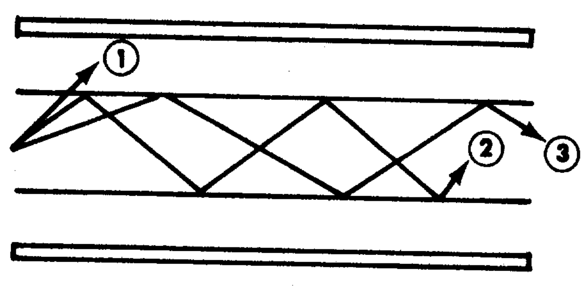


Figure 22-23. Representation of Light Traversing a Multimode/Step Index Fiber.

Since the core and the cladding have slightly different indexes of refraction, the junction between them acts as a reflective surface for rays 2 and 3. We can associate each of these rays with a mode. The higher angle ray (2) represents a higher order mode and travels further than ray 3. Over long distances, this results in signal distortion. When the angle of incident is too high, as in ray 1, the light penetrates the cladding, rather than being reflected at the interface, and is then absorbed by the jacket. In order to eliminate the distortion caused by multimode transmission, the single mode fiber was developed. The core in this fiber is so small that only a single mode is capable of propagating; however, in order to get enough light into this small fiber, a laser must be used as a light source. The multimode/graded index fiber was developed to overcome the disadvantages of the two previous types. In this case, the index of refraction is highest in the center of the core and gradually decreases towards the edges. As a result, waves propagate as shown in figure 22-24. Again, if the angle of incident is too large, the ray is absorbed in the jacket, as with ray 1; however, if the angle is small enough, it is propagated down the fiber, as shown with ray 2, in a sinusoidal manner. This is because changing index of refraction causes the ray to bend as it approaches the edge of the fiber. Also note that, even though ray 2 travels further than ray 3, they propagate at the same speed. This is because ray 3 is confined near the center of the core where the index of refraction is highest; therefore, it does not travel as far as ray 2 but it moves slower. As a result, all rays are propagated at the same rate.

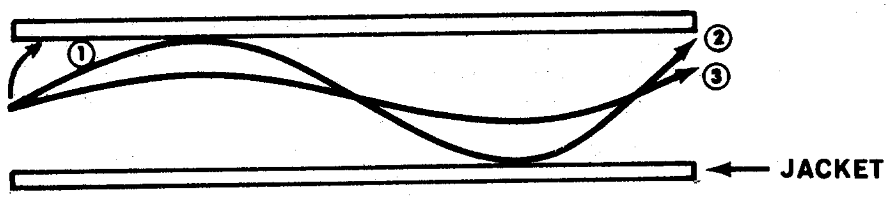


Figure 22-24. Light Propagation in Multimode/Graded Index Fiber.

Figure 22-25 is a block diagram of an optical system. The most common light source is either a light emitting diode (LED) or a solid state injection laser. The detector is usually a silicon photo diode.

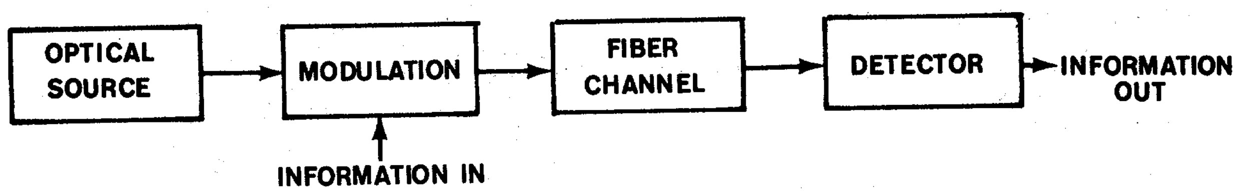


Figure 22-25. Optical Communications System Block Diagram.

22-7. Summary. We have very briefly discussed several of the future programs and techniques that we can expect to see in the Air Force in the not too distant future. Undoubtedly, there are many others that could have been included that were not. Only history will determine how successful these approaches will be in improving communications.

OFFICIAL

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Director of Administration

SUMMARY OF REVISED, DELETED, OR ADDED MATERIAL

This revision includes corrections of many terms and expressions which have changed with the times. It also covers new equipment as presently used and some future programs and techniques (examples: Automated Technical Control (ATEC); Digital Communications Concept; Satellite Communications; and fiber optics).

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GLOSSARY

Abscissa. In a cartesian (rectangular) plotting system, the horizontal (x) axis. Usually used to plot the independent variable.

Activity Factor. The percentage of time during the busiest traffic hour that a channel might be active. It may be computed as:

$$\text{Activity Factor} = \frac{n}{N}$$

n = active channels (speech present)
N = busy channels (off hook, circuit is in use whether there is speech or not).

Amplitude Modulation (AM). Modulation of a carrier wave by varying its amplitude IAW the frequency of the intelligence signal to be transmitted.

Analog. The representation of numerical quantities by physical variables (such as voltage, resistance, etc.). A comparison.

Angle Modulation. Modulation by varying the angle of the sine-wave carrier (such as in phase and frequency modulation).

Antenna Beamwidth. The angle between the directions, on either side of the major lobe of the antenna radiation pattern, where the field intensity is 3 dB less than the maximum value.

Asymptote. A straight line which would be tangent to a curve, if the curve were extended infinitely far.

Asynchronous (Data). The opposite of synchronous. In data, the use of individual start and stop pulses for each character sequence. The process is lengthy but provides a capability of slight send/receive speech differences and is most useful at low modulation rates.

Atmospheric Refraction. Condition of the atmosphere that causes the bending of radio waves due to variations in the refractive index with height in the lower atmosphere.

Attenuation. Decrease in electrical current, voltage, or power encountered in transmission from one point to another. Attenuation is commonly expressed as a ratio of powers expressed in dB.

Audio Frequency. The frequencies that the human ear can hear, usually stated as the range from 20 to 20,000 Hz. Sometimes stated as the bandwidth of a telephone voice channel (300 to 3400 Hz).

Aural. Audio intelligence transmitted in the range of human hearing.

Automatic Digital Network (AUTODIN). Worldwide automatic communications network for DOD end-to-end message-switched digital data communications.

Automatic Frequency Control (AFC). Provision for detection of a carrier frequency and the correction of any deviations by the use of feedback signals.

Automatic Gain Control (AGC). The circuit used to automatically sense variation in received signal levels and adjust the gain of receiver amplification stages to provide a constant output.

Automatic Secure Voice Communications (AUTOSEVOCOM). A network which provides cryptographically secure voice communication using wideband and narrowband voice digitizing techniques.

Automatic Technical Control (ATEC). A technical control system that will automate many of the functions that were previously accomplished manually. This includes automatic measurement of circuit and system parameters and automatic isolation of problem areas by the use of computer alogarithms.

Automatic Voice Network (AUTOVON). Worldwide automatic communications system for end-to-end circuit-switched voice communications for DOD and specific non-DOD subscribers.

Balanced. Electrically symmetrical. The two terminals of a balanced network have the same impedance to ground and have voltages to ground 180° out-of-phase with each other.

Balun Transformer. A transformer used to convert balanced to unbalanced (or vice-versa). Can be used to match impedances and provide symmetrical output from unsymmetrical input. Impedance match can include other than 1/1 ratio, such as 4/1 (75 ohm coaxial cable to 300 ohm antenna).

Band Reject Filter. A filter built to reject a specified band of frequencies while passing all others with little attenuation. Also called band elimination filter or bandstop filter.

Bandpass. The frequency bandwidth between half-power (-3 dB) points of a frequency response, relative to a maximum. See bandwidth.

Bandpass Filter. The reverse of a band reject filter. Allows only a given bandwidth to pass while rejecting all other frequencies.

Bandwidth. The range of frequencies between half-power (-3 dB) points on either side of a maximum. See bandpass.

Base:

1. A number on which a numerical system depends. The base of the decimal system is 10.
2. In the expression "a^x," "a" is the base number and is used as a multiplying factor "x" times.

Baseband. The band of frequencies occupied by the aggregate of the transmitted signals when first used to modulate a carrier.

Basic Intrinsic Noise Ratio (BINR). In loaded noise measurements, the ratio of noise in a slot with loading applied to that with all loading removed. It is an indicator of idle noise in the system under test.

Baud. The modulation rate term indicating the time duration of the shortest signal element. The baud rate and the bit rate are not necessarily equal.

Bessel Function. A complex function used to calculate the sideband amplitudes in an FM signal.

Bias Distortion:
1. Distortion resulting from operation in the non-linear portion of a characteristic curve due to improper bias.
2. Distortion causing teletype mark and space pulses to be lengthened or shortened.

Binary. Consisting of two states or conditions (such as on/off, up/down, 0/1, good/bad, etc.). The base of a binary system is 2.

Bit (Binary Digit):
1. The unit of information equal to one binary state.
2. The unit of storage capacity. Capacity equals the logarithm to base 2 of the number of possible states.

Bridge, Bridging:
1. A device used for measurement by comparison against known standards.
2. To connect a measuring device across a circuit in parallel. A high impedance device is normally used to minimize effect on circuit operation.

Busbar. A heavy strip of wire or bar used to carry high current and/or voltage levels in power plants and power distribution centers.

Capture Effect. The ability of a strong FM signal, on the same frequency as a weaker signal, to control the output of a demodulator. The weaker signal is rejected.

Carrier. The radio frequency wave whose frequency is independent of any modulation; the output when no modulation is present; a signal generated and subsequently modulated; a wave generated in a receiver and then combined with the sidebands for detection.

Carrier to Noise Density (C/kT). Ratio of carrier level to the receiving system noise normalized to a 1 Hz bandwidth. This parameter is one of the basic factors which determine the system channel capacity.

Channel. The portion of spectrum assigned for the transmission of intelligence (such as voice channel, teletype channel, facsimile channel, television channel, etc.).

Character. One symbol of a set of elementary marks or events that may be combined to express information.

NOTE: **A group of characters in one context may be considered as a single character in another, as in the binary-coded decimal system.)*

Characteristic Distortion. A displacement of signal transitions resulting from the persistence of transients caused by preceding transitions.

Characteristic Impedance. The ratio of voltage to current at every point along a transmission line on which there are no standing waves. Also called surge impedance.

Circulator. Microwave coupling device having a number of terminals so arranged that energy entering one terminal is transmitted to the next adjacent terminal in a particular direction.

Class "A" Power. A primary source of power capable of continuous supply, either from local generator or commercial sources.

Class "B" Power. An auxiliary power capable of supply for extended periods (days) of power outage. Usually slow starting.

Class "C" Power. An auxiliary power capable of supply for short periods (hours) of power outage. Quick starting (10-60 seconds) in nature.

Class "D" Power. An auxiliary power capable of supply immediately on failure of the normal power source. Power supplied is thus called uninterruptible or no-break.

Clipping. Limiting the amplitude of a signal to a predetermined maximum value.

Combiner. A device which accepts several inputs and presents the best possible output. This can be done by simply switching the optimum input to the output or by electronically adding all inputs in proportion to their relative quality.

Complex Signal. A signal which cannot be expressed as a single sinusoidal function.

Crosstalk. Undesired signal power injected into a communications circuit from other communications circuits.

dBa (dBrn Adjusted). Weighted circuit noise power in dB referred to 3.16 pW (-85 dBm) which is 0 dBa.

dBa (FIA). Weighted circuit noise in dBa measured on a line with FIA weighting.

NOTE: *1 mW of a 1000 Hz-modulated tone reads -85 dBa, but the same white noise power (300-3400 Hz) will read -82 dBa, due to frequency weighting.*

dBa0. Circuit noise power in dBa referred to zero relative transmission level (0dBr or 0TLP).

dBm. Power in dB relative to 1 mW.

dBmp (dBm Psophometrically Weighted). Unit of noise power in dBm measured with psophometric weighting. Conversion as follows:

dBm (psoph) = 10 log pWp - 90 = dBa-84 = dBm - 2.5 (for flat noise 300-3400 Hz).

dBm0. dBm referred to zero relative transmission level.

dBm0p. Circuit noise in dBm0 measured on a line with psophometric weighting.

dBr. dB relative to transmission level.

dBrn (Decibels Above Reference Noise). Weighted circuit noise power in dB referred to 1 pW (-90 dBm) which is 0 dBrn.

NOTE: With C-Message weighting, a 1 mW 1000 Hz tone will read +90 dBrn, but the same white noise power (300-3400 Hz) will read +88 dBrn; with 144 weighting, the same respective readings will be +90 dBrn and +82 dBrn, due to the different frequency weighting.

dBrn (f1-f2). Flat noise power in dBrn, measured over a frequency band between frequencies f1 and f2 (*for example,* 3 kHz flat).

dBrnc (C-Message). Weighted circuit noise power in dBrn, measured on a line with C-Message weighting.

Decade. In communications, the interval between two frequencies having a ratio of 10:1.

Decibel (dB). A dimensionless unit for expressing the ratio of two powers, the number of dB being 10 times the logarithm to the base 10 of the power ratio or 20 times the logarithm to the base 10 of a voltage or current ratio.

Decibels Above or Below 1 Milliwatt (dBm). Unit of power used to describe the ratio of the power at any point in a transmission system to a reference level of 1 milliwatt.

Decibels Above or Below 1 Watt (dBW). Measure of power expressed in decibels to a reference level of 1 watt.

De-Emphasis. Restoration of a pre-emphasized signal wave to its original form.

Defense Communications Agency (DCA). Agency which reports to the Secretary of Defense, through the JCS, and is responsible for operational control and supervision of the Defense Communications System.

Defense Communications System (DCS). Includes all worldwide, long-haul, Government-owned and leased, point-to-point circuits, terminals, control facilities, and tributaries required to provide communications from the President to and between the Secretary of Defense, the JCS, and other Government agencies; from the Secretary of Defense and the JCS to and between the military departments and the unified and specified commands; from the military departments to and between their major commands and subordinate fixed headquarters; and from the unified and specified commands to and between their component and subordinate commands.

Delta Connection. In power, a three-phase circuit where the windings are connected in a closed ring with the instantaneous voltages around the ring equal to zero. May be used in generating or transforming equipment.

Demodulation. The process of acting on a transmitted wave to recover the intelligence content.

Demodulator. Device which operates on a carrier wave to recover the wave with which the carrier was originally modulated.

Detector. Rectifier tube, crystal, or dry disc by which a modulation envelope on a carrier, or the simple on-off state of a carrier, may be made to drive a lower frequency device.

Dielectric. Nonconducting material through which magnetic lines of force, or electrostatic lines of force, may pass.

Dielectric Constant. Property of a dielectric which determines the electrostatic energy stored per unit volume for unit potential gradient.

Diffraction. Process which produces a diffracted wave when a wave is incident on a sharp boundary. The diffracted wave is projected into the geometrical shadow region.

Diplexer. A device which allows two transmitters or two receivers to operate on the same antenna.

Discriminator. Device in which amplitude variations are derived in response to frequency or phase variations.

Distortion. Undesired change in waveform.

Diversity. The term applied to the simultaneous combining of signals and their detection through the use of space, frequency, or polarization characteristics or combinations thereof.

Double Sideband-Suppressed Carrier (DSB/SC) Modulation. Modulation scheme where the intelligence is carried in two sidebands, one above and one below a suppressed carrier.

Ducting. The trapping of a radio wave between two atmospheric layers into a narrow atmospheric layer.

Duplex. Method of operation of a communications circuit when each end can simultaneously transmit and receive.

Duplexer. Device which permits the use of the same antenna for both transmitting and receiving.

Earth Bulge. A method of depicting the variations in the refraction index. It allows portrayal of the radio beam as a straight line between the transmitter and the receiver.

Echo Distortion. Distortion caused by the double reflection of waves which results in a delayed wave traveling in the same direction as the original wave.

Envelope Delay. Time of propagation, between two points, of the envelope of a wave.

Envelope Detector. A conventional AM rectifier detector with a capacitor across the output terminals. The output voltage is pi times the rectifier output voltage. Most commercial AM receivers employ envelope detectors.

Error Rate (1:10^6, etc.). The number of erroneous bits or characters in a sample, frequently taken per 100,000 characters.

Exponent. In the expression "a^x" "x" is an exponent. It expresses the number of times that "a" is to be used as a multiplying factor. An exponent may be positive, negative, a whole number, or a fraction.

Facsimile. Process by which pictures, images, or other fixed graphic material are scanned and the information converted into electrical signals for local use or transmission remotely to reproduce a likeness of the subject copy in record form.

Fade Margin:

1. Number of dBs of attenuation which may be added to a specified radio frequency propagation path before the signal-to-noise ratio of a specified channel falls below a specified minimum.
2. In this pamphlet, the fade margin in FM systems is the dB difference between the FM improvement threshold and the median RSL.

Figure of Merit (G/T_s). The ratio of the receiving antenna gain to the receiving system noise temperature. The G/T_s ratio describes how well the antenna and receiver front end combination acts to achieve a high C/kT at the receiver.

Fortuitous Distortion. Distortion in a telegraph system which includes effects that cannot be classified as bias or characteristic distortion.

Free Space Loss. The loss encountered by a radio wave in traveling from transmitter to receiver without encountering reflection or obstruction. It is a function of the wavelength of the signal and the distance the wave travels.

Frequency Deviation. In FM, the peak difference between the instantaneous frequency of the modulated wave and the carrier frequency.

Frequency Diversity. The use of different radio frequencies in the simultaneous transmission of intelligence. This system takes advantage of the different wavelengths of the two signals and the tendency of two such different wavelengths to be affected differently under multipath fading conditions. The two or more frequencies can then be processed in a combiner.

Frequency Division Multiplex (FDM). A method of deriving two or more simultaneous, continuous channels from a medium connecting two points by assigning separate portions of the available frequency spectrum to the several channels.

Frequency Domain Analysis. Analysis of complex signals with reference to their frequency content as opposed to time variations.

Frequency Modulation (FM). Angle modulation of a sinusoidal carrier in which the instantaneous frequency of the modulated wave differs from the carrier frequency by an amount proportional to the instantaneous value of the modulating wave.

Frequency Modulation Improvement Threshold. In FM systems, the point where received RF carrier peaks equal or exceed noise peaks 99.999% of the time. Beyond this point, thermal noise is suppressed in direct proportion to received carrier level.

Frequency Response. Measure of effectiveness with which a circuit or device transmits the different frequencies applied to it.

Frequency Spectrum. The total range of frequencies contained in a signal.

Fresnel Zone. Cigar-shaped region surrounding the axis of symmetrical beam antenna. Sum of the distances from any point on the boundary of the first Fresnel zone to each antenna is one-half wavelength longer than the direct path between antennas.

Gaussian Noise. Noise whose amplitude follows Gaussian Probability Distribution. Noise signals which are the result of numerous independent perturbations tend to be Gaussian.

Group. In carrier telephony, a number of voice channels multiplexed together and treated as a unit. Commonly, it is composed of 12 4-kHz bandwith channels, frequency-multiplexed and occupying the band from 60-108 kHz.

Harmonics. Integral multiples or submultiples of a fundamental frequency.

Helix. A wire curved as if wound in a single layer around a cylinder, used as the beam-modulating element in a traveling wave tube.

Hertz. By definition, the unit of frequency equal to one cycle per second.

Heterodyning. Process of combining two signal frequencies in a non-linear device. The result is production of new frequencies (such as the sum and difference of the combining frequencies). Also called mixing or beating frequencies together.

High Speed Data. Defined by DCA as data requiring a bandwidth of 48 kHz or greater.

Hybrid. A transformer or combination of transformers or resistors affording paths to three branches, circuits A, B, and C. This network is so arranged that A can send to C, B can receive from C, but A and B are effectively isolated. In telephone circuits, it is called a two-wire-four-wire terminating set.

Idle Channel Noise (ICN). The residual noise in a communications channel when no communications traffic is being passed. Consists of thermal noise, impulse noise, intermodulation noise, crosstalk, and any other source of extraneous signals or noise in a channel.

Idler Frequency. In a parametric amplifier, a sum or difference frequency generated within the amplifier other than the input, output, or pump frequencies which requires specific circuit consideration for proper amplifier operation.

Image Frequency. In a heterodyne frequency converter, the image frequency is an undesired radio frequency which is as far removed from the local oscillator frequency as the desired radio frequency but on the opposite side (*that is,* if the local oscillator tracks above the desired frequency, image frequency is above local oscillator frequency and vice versa).

Impedance. Total opposition offered to the flow of an alternating current. It may consist of any combination of resistance, inductive reactance, and capacitive reactance. It equals the square root of the sum of the squares of the resistance and the reactance.

Impulse Noise (IPN). Noise due to disturbances having abrupt changes and short duration.

Incident Wave. In a medium of certain propagation characteristics, a wave which impinges on a discontinuity or medium of different propagation characteristics.

Insertion Loss. Loss in energy due to the insertion of a component or device at some point in a system. Usually expressed as the ratio of output to input in decibels.

Intermediate Frequency (IF):

1. Fixed frequency to which carrier waves are converted in a super-heterodyne receiver.
2. Frequency to which a signaling wave is shifted locally as an intermediate step during transmission or reception.

Intermodulation. Modulation of the components of a complex wave by each other, producing waves having frequencies equal to the sums and differences of integral multiples of the component frequencies of the complex wave.

International Radio Consultative Committee (CCIR). A committee of the International Telecommunications Union which studies technical and operating questions relating to radio communications and issues appropriate recommendations.

Inverter. A device for converting direct current to alternating current.

Isolator. A ferromagnetic device used to permit one-way propagation of radio frequency energy. It is used to prevent reflected RF energy from returning to the source device.

Isotropic. Having physical properties that are the same regardless of the direction of measurement. In an isotropic antenna, energy is radiated equally well in all directions.

Jitter. Short-time instability of a signal. The instability may be in either amplitude or phase or both. The term is especially applied to signals reproduced on the screen of a cathode-ray tube.

Kelvin Temperature Scale. The Kelvin, or absolute, temperature scale is based on the molecular theory of temperatures. Zero degrees Kelvin is the point of no molecular motion, the lowest possible temperature. The size of the Kelvin degree is the same as the Centigrade degree. Formulas for conversion to Centigrade and Fahrenheit scales are:

Temp in C = Temp in Kelvin - 273
Temp in F = (9/5) Temp in Kelvin - 241

Klystron. A specialized electron tube that velocity-modulates its electron beam to cause bunching of the electrons. Velocity modulation is introduced by a resonant cavity. Energy from the resultant electron bunching is taken as an output by another resonant cavity.

Knife-Edge Effect. Radio propagation effect in which a radio wave is diffracted by a sharp obstacle and received by a station in the geometrical shadow region.

Limiting. Removal of amplitude variations of a frequency-modulated signal by clipping the peaks of the signal at a preselected level.

Line-of-Sight (LOS):

1. Distance of the horizon from an elevated point including the effects of atmospheric refraction.
2. Loosely, therefore, a microwave radio link which employs a line-of-sight propagation path.

Linear. Having an output which varies in direct proportion to the input.

Link. A communications medium between two points with breakout capability.

Load (Power):

1. The power consumed by a machine or circuit in performing its function.
2. A power-consuming device connected to a circuit.
3. Resistor or other impedance that can replace

some circuit element that is to be temporarily or permanently removed.

Load Capacity. The volume of traffic a system can handle without undue distortion or noise.

Loading. Application of traffic or test signals to the communications system.

Lobe (Antenna). One of the three-dimensional sections of the radiation pattern of a directional antenna bounded by nulls.

Logarithm. A logarithm is an exponent. It is the power to which a fixed number, the base, must be raised in order to produce a given number. Common logarithms use the base 10.

Loop. A point of maximum voltage or current on a transmission line or antenna. Also called an antinode, because it is the opposite of a node. A current node occurs at the same point as a voltage loop and vice versa. Loops and nodes occur at half-wavelength intervals.

Low Speed Data. Defined by DCA as data with a digital rate of 1200 bauds or lower.

Marking and Spacing Intervals. Commonly used teletype term used to designate one of two binary states in the transmission or coding of data. The "space" condition may be the low, off, or negative state, as opposed to "mark," which may be high or positive.

Matrix. Any logical network whose configuration is a rectangular array of intersections of its input-output leads with elements connected at some of these intersections. The network usually functions as an encoder or decoder; loosely, any encoder, decoder, or translator.

Median. In a series of events (or numbers), that event or number which will be exceeded 50% of the time.

Medium Speed Data. Defined by DCA as data that can be transmitted over a nominal 4 kHz bandwidth, with a preferred modulation rate of 2400 baud.

Microwave (M/W). Radio transmission using wavelength of 30 centimeters or less.

Microwave Amplification by Stimulated Emission of Radiation (MASER). A low-noise microwave amplifier utilizing a change in energy level of a material to obtain signal amplification. Common materials are gases (ammonia) and crystal (ruby).

Mismatch Loss (Reflection Loss). The ratio, expressed in dB, of the incident power to the transmitted power at a discontinuity. A measure of the loss caused by reflection.

Modem (Modulator, Demodulator). A terminal panel containing a modulator and demodulator, some circuits of which may be in common.

Modulation. The process in which either the amplitude, frequency, or phase of the radio frequency carrier wave is varied with time IAW the wave-form of superimposed intelligence.

Modulation Index. In angle modulation with a sinusoidal modulating wave, the modulation index is the ratio of the frequency deviation to the frequency of the modulating wave.

Multipath. Propagation phenomenon which results in signals reaching the receiving radio antenna by two or more paths. Usually, these signals will have both amplitude and phase difference.

Nautical Mile. One-sixtieth of a degree of the earth's equator, or 6,080.2 feet.

Nodal Point. (Also Called Junction Point, Branch Point, or Vertex). A terminal of any branch of a network or a terminal common to two or more branches of a network.

Node. A point of minimum voltage or current on a transmission line or antenna. Opposite of a loop.

Noise. Undesired sound or energy disturbance within the frequency band of any transmission channel or device.

Noise Figure. Term used to rate the noise qualities of radio receivers. It is equal to the ratio of the signal-to-noise (S/N) of an ideal receiver to the receiver under test: Nf = S/N ideal ÷ S/N actual. The noise figure is usually expressed in dB; therefore, a receiver with an 8 db Nf adds 8 dB of noise to the signal.

Noise Loading Ratio (NLR). In loaded noise measurements, the ratio of the noise power used to the test tone level. Used to convert noise power ratio to signal-to-noise ratio.

Noise Power Ratio (NPR). The noise power ratio for multichannel equipment is the ratio of the mean noise power measured in any channel, with all channels loaded with noise, to the mean noise power measured in the same channel, with all channels but the measured channel loaded with noise.

Noise Temperature. Term used to rate the noise qualities of communications equipment. It relates noise power to a temperature which would produce the same noise power that would be available from a resistance at that temperature.

Order Wire. A circuit for use by technical control or maintenance personnel for communications incident to line-up and maintenance of communications facilities.

Ordinate. The distance of any point from the X-axis, measured on a line parallel to the Y-axis in a coordinate system.

Overload Point. Power output of a device at which a 1 dB increase in fundamental power causes a 20 dB increase in power of the third harmonic.

Pad. A resistance network which results in attenuation of an electrical signal. Usually expressed in dB.

Parabolic Reflector. Metallic sheet, formed so that its cross-section is in the shape of a parabola. The antenna elements are placed along the line that runs through the focal point of the parabola, parallel to the leading edge of the reflecting surface. As referred to herein, it is a paraboloid so shaped that all rays emanating from a single point (called the focus) on the axis will be reflected in a direction parallel to the axis.

Parallax. Apparent displacement of the position of an object caused by a shift in the point of observation.

Parameter:
1. One of the constants entering into a functional equation and corresponding to some characteristic property, dimension, or degree of freedom.
2. One of the resistance, inductance, mutual inductance, or capacitance values involved in a circuit or network.

Parametric Amplifier. A solid state amplifier used for amplification of radio frequencies. A parametric amplifier uses specially constructed semi-conductor diodes with a source of power supplied by a high-frequency (pump) oscillator, rather than the normal DC power source. The power source must be extremely stable. The output is usually fed directly to the mixer stage, then to a pre-IF amplifier which may employ vacuum tubes. The parametric amplifier is a low noise device and operates at a lower noise figure than any other device except the MASER.

Parity. A bit associated with a character or block, for the purpose of checking the absence of error within the character or block.

Patch and Test Facility (PTF). That maintenance facility which is equipped with jack strips, test points or patch panels, and test equipment required to isolate troubles, which substitutes equipment or lines for maintenance purposes and which does not normally reroute channels.

Path. The principal route over which the radio energy is propagated between the transmit and receive antennas.

Peak Factor. The ratio of the peak voltage of a signal to its RMS voltage. For statistically-distributed signals, peak is defined as that level exceeded only .001% of the time.

Phase Modulation (PM). The form of modulation in which the angle relative to the unmodulated carrier angle is varied IAW the instantaneous value of the amplitude of the modulating signal.

Pilot Tone. A signal wave, usually a single frequency transmitted over the system, for supervisory, control, synchronization, or reference purposes. Sometimes it is necessary to employ several independent pilot frequencies. Most radio relay systems use radio or continuity pilots of their own but also transmit the pilot frequencies belonging to the carrier frequency multiplex system.

Plane Reflector. Device which has no curvature. Used to redirect radiation in a desired direction or directions.

Polarization Diversity. A method of transmission and/or reception of information accomplished by the use of separate vertically and horizontally polarized antennas.

Pre-Emphasis. Intentional alternation of the frequency-amplitude characteristic of a signal wave to reduce adverse effects (such as noise) in subsequent parts of the system, after which de-emphasis is employed to obtain the original transmitted signal.

Preselector. Device, placed ahead of a frequency converter or other device, which passes signals of desired frequencies and reduces others.

Probability. A statistical expression of the likelihood that an event will occur. In the throwing of a single die, the likelihood of throwing a 5 is 1 chance in 6, or a probability of 1/6th. In communications, for example, the likelihood of equipment failure during a given time period can be expressed as a probability.

Probability Distribution. A graph which shows the distribution of the outcomes of a random experiment (y-axis) versus the percentage of outcomes (x-axis) which will be equal to or less than the y-axis value at that point. The normal, or Gaussian, distribution is frequently duplicated when plotting such diverse items as test scores, noise voltages, failure rates, etc.

Propagation:
1. In electrical practice, the travel of waves through or along a medium.
2. Traveling of a wave along a transmission path.
3. Travel of electromagnetic waves or sound waves through a medium. Propagation does not refer to the flow of current in the ordinary sense.

Pulse Amplitude Modulation (PAM). Process in which a band-limited signal is sampled and the value of the instantaneous samples are used to control the amplitude of a pulse carrier.

Pulse Code Modulation (PCM). The process of sampling a band-limited signal and converting the amplitude into a binary-coded word for transmission over a communications channel.

Pump Frequency. A term used in connection with master oscillators indicating the frequency that is higher than the frequency of the signal that is to be amplified. In this publication, normally referring to the pump oscillator that supplies pumping energy for MASER and parametric amplifiers. Operates at twice or some higher multiple of the signal frequency.

pW (picowatt Equal to 10^{-12} Watt or -90 dBm). A unit of absolute power used for both weighted and unweighted noise. 10 log pW = dBm -90.

pWp (pW Psophometrically Weighted). Noise power measured in picowatts with a psophometric filter. pWp = 0.56 pW (for flat noise 300-3400 Hz).

Radiation Pattern. Diagram indicating the intensity of the radiation field of a transmitting antenna at a given distance away from the antenna in all directions. In the case of a receiving antenna, it indicates the response of the antenna to a signal having unit field intensity and arriving from different directions.

Radio Link. A communications medium between two points, not necessarily with breakout capability.

Reactance Tube. A vacuum tube operated in such a manner that it presents almost a pure reactance to the circuit. In this publication, normally referring to the reactance tube modulator, which is a device, the reactance of which may be varied IAW the instantaneous amplitude of the modulating wave applied. Electron tubes are widely used in this manner to effect phase or frequency modulation.

Received Signal Level (RSL). Measurement of power in dBm at the input of a receiver, before any amplification of the desired radio frequency.

Reflected Wave:

1. Sky wave reflected from the ionosphere layer back to earth.
2. Wave reflected from the junction of two media with different indices of refraction.

Reflection. Phenomenon which causes a wave which strikes a medium of different characteristics to be returned into the original medium with the angle of incidence and of reflection equal and lying in the same plane.

Reflectometer. A microwave measuring system arranged to measure the incident and reflected power and determine their ratio.

Refraction. Phenomenon which causes a wave obliquely entering another medium to undergo a change in direction when the velocity of the wave in the second medium is different from that in the first.

Regression Analysis. An analytical method used to determine trending from a set of data points. Regression analysis will find the best possible fit of a straight line to a set of data. By observing the slope of the line, the trend can be determined.

Reliability. Quality or property built into, or inherent in, a device that indicates that the device will probably perform its specified function without failure under given conditions for a specified period of time.

Repeater. Combination of apparatus for receiving either one-way or two-way communications signals and delivering corresponding signals which are either amplified or reshaped or both.

Repeller. Element in a reflex klystron tube which reflects the electrons back toward the grid.

Resonant Cavity:

1. Region enclosed by conducting walls within which resonant fields may be excited.
2. Space normally bounded by an electrically-conducting surface in which oscillating electromagnetic energy is stored and whose resonant frequency is determined by the geometry of the enclosure.

Return Loss. The ratio, expressed in dB, between the power incident on a discontinuity and the power reflected from the discontinuity. (The number of dB reflected power is down from incident power.)

Root Mean Square (RMS). Of alternating currents and voltages, the root mean square value is the effective current or voltage applied.

NOTE: It is that alternating current or voltage that produces the heating effect the same value of direct current or voltage of an equal value would produce. It is equal to 0.707 times the maximum alternating current value for a sinusoidal wave.

SCOPE CREEK. The Department of Defense code name for the Operating and Maintaining Agency Measurement Program. It is in direct support of the DCA Technical Visits Program.

Selectivity. Degree to which a radio receiver can accept the signals of one station while rejecting those of all other stations on adjacent channels.

Sensitivity. The minimum input signal required to produce a specified output signal having a specified signal-to-noise ratio.

Serrasoid Modulator. A type of phase modulator employing a sawtooth wave. It is used in many tropospheric scatter systems to generate an FM signal.

Sideband. The spectral energy distributed above and below a carrier resulting from a modulation process. In transmission, the band of frequencies produced on either side of the carrier frequency including components whose frequencies are the sum or difference of the carrier and the modulating frequencies.

Signal-to-Noise (S/N) Ratio. Ratio of the magnitude of the signal to that of the noise, often expressed in decibels. The signal is defined as the value of the TLP. Note that this is then different from a signal-plus-noise-to-noise ratio.

Single Sideband-Suppressed Carrier Modulation. Modulation whereby the spectrum of the modulating wave is translated in frequency by a specified amount either with or without inversion and the carrier frequency is suppressed. One sideband is removed by filtering so that only one sideband is transmitted, occupying the same amount of frequency spectrum as before the modulation process.

Sinusoidal. Varying in proportion to the sine of an angle or time function.

NOTE: Ordinary alternating current is sinusoidal.

Skin Effect. Tendency of currents to flow near the surface of a conductor thus being restricted to a small part of the total sectional area and producing the effect of increasing the resistance. This effect increases as the frequency of the current increases.

Space Diversity. Form of diversity transmission or reception which uses antennas placed in different locations.

Stagger Tuning. Method of aligning the IF stages of a superheterodyne receiver to produce wide bandwidth. This is accomplished by peaking alternate IF transformers at slightly different frequencies.

Standing Wave. Sinusoidal distribution of current and voltage amplitudes along a transmission line as a result of the reflection of energy from a point of discontinuity (a dent in a waveguide, impedance mismatch, etc.). The distribution of voltage and current are stationary; thus, standing waves are observed. Standing waves indicate that power is being lost in transmission.

Strata. Layers having identifiable common properties (such as layer of atmosphere).

Supergroup. In carrier telephony, five groups (for a total of 60 voice channels) multiplexed together and treated as a unit. A basic supergroup occupies the band between 312 and 552 kHz.

Superheterodyne Reception. Method of receiving radio waves in which a locally generated RF signal is mixed in a non-linear system with the received wave to derive a sum or difference IF, which is then amplified and detected.

Surge Impedance. Another term for characteristic impedance of a transmission line. When a transmission line is terminated in its surge impedance, no reflection will occur and no standing waves will appear.

Synchronize. To adjust the rate (frequency) of two or more time-varying quantities such that all have the same frequency. AC generators must be synchronized before they can be electrically tied to a common bus.

Synchronous. In step or in phase. Running at the same speed as some associated machine.

Synchronous Detector. Detector that inserts a missing carrier signal in exact synchronism with the original carrier at the transmitter.

Synthesizer. Unit capable of generating many crystal-controlled frequencies for multiple-channel communications equipment.

System. A set of components coordinated to accomplish a set of objectives.

System Environment. Those things that affect system performance, but are beyond control of the people who run the system. *For example,* the radio path of a communications system employing radios.

System Noise Temperature. Equivalent noise temperature of a receiver system. It includes the noise contribution of all system components including the antenna and waveguide.

Technical Visits Program (TVP). The DCA Technical Visits Program, established by DCA Circular 300-195-4, is a program to measure and record the performance of the various DCS operational transmission links. Air Force "Project Scope Creek" is a part of this program.

Temperature Inversion. A layer in the atmosphere where normal temperature changes are reversed. As elevation is increased, the temperature rises instead of falling.

Terminal:

1. Fitting to which electrical connections are made.
2. Final station in a radio relay system.

Terminate:

1. To form the conclusion of; finish.
2. Electrically, to close a circuit at either end of a line or transducer by connecting some device thereto. Terminating does not imply any special condition (such as the elimination of reflection).

Test Level Point (TLP). A point in an electrical circuit where the signal level may be measured and compared to a specified value. The power level (in dBm) specified at a TLP is an engineered reference value. Actual test tones will usually be measured at a level below the TLP value.

Thermal Noise. Noise voltage generated in resistors, tubes, and other electronic devices which can be attributed to the thermal agitation of electrons. A state of zero agitation exists only at the very coldest temperature: 0° Kelvin.

Thermocouple. Device consisting of two dissimilar metals in physical contact, thereby forming a junction which is sensitive to heat. A voltage proportional to temperature is generated across the junction when heated.

Threshold:

1. Point at which an effect is first produced, observable, or otherwise indicated.
2. In a radio receiver, it is the point where the signal power presented to the receiver just equals the thermal noise presented to the receiver plus its own internally generated noise. Threshold is governed, in part, by the bandwidth of the receiver.

Threshold Extension. The process by which threshold in a receiver is lowered to detect a weak signal.

Time Division Multiplex (TDM). A method of deriving several channels from a given frequency spectrum by assigning discrete time intervals in sequence to the different channels. During a given time interval, the entire available frequency spectrum can be used by the channel to which it is assigned. In general, TDM systems use pulse transmission. The multiplex pulse train may be considered to be the interleaved pulse trains of the individual channels. The individual channel pulses may be modulated either in an analog or a digital manner.

Transient. Instantaneous surge of voltage or current which occurs in a system owing to a sudden change in conditions and which persists for a relatively short time after the change has occurred.

Transmission Line. Conductor or series of conductors used to carry electrical energy from a source to a load.

Traveling Wave Tube (TWT). An electron tube in which a beam of electrons interacts continuously with a guided electromagnetic wave to produce amplification at ultra-high frequencies.

Troposphere. Part of the earth's atmosphere in which temperature generally decreases with altitude, clouds form, and convection is active. Experiments indicate that the troposphere occupies the space above the earth's surface up to a height ranging from about 6 kilometers at the poles to about 18 kilometers at the equator (3.7-11 miles).

Tropospheric Scatter. Propagation of radio waves by scattering as a result of irregularities or discontinuities in the physical properties of the troposphere.

Tunnel Diode. A semiconductor consisting of a PN junction which has been specially constructed to optimize the characteristics desired. In operation, the tunnel diode differs from other types of PN junctions in that it exhibits negative resistance and the current is carried by electrons that travel across the PN barrier by quantum mechanical tunneling.

Uninterruptible Power System (UPS). See Class "D" power.

Varactor. Two-terminal semiconductor device in which the electrical characteristic of primary interest is a voltage-dependent capacitance. Also called a Step Recovery Diode.

Voice Frequency Carrier Telegraphy (VFCT). Carrier telegraphy in which many telegraph signals are multiplexed together to be transmitted over a single voice bandwidth circuit. Use of the abbreviation "VFCT" usually refers to the equipment which multiplexes the signals.

Voltage Standing Wave Ratio (VSWR). The ratio of the maximum voltage to the minimum on a line with standing waves.

Waveguide. A transmission line comprising a hollow conducting tube within which electromagnetic waves may be propagated; or, a solid dielectric or dielectric-filled conductor for the same purpose.

Wavelength. The distance traveled in one period or cycle by a periodic disturbance. The wavelength of a signal is the frequency divided by the velocity of propagation.

Weighting. The process of assigning different importance, IAW second parameter, to quantities which were originally of equal value. *For example,* in frequency weighting, the amplitude of a signal is weighted according to the second parameter, frequency; thus, a uniform response may remain constant at 1 kHz and be decreased 5 dB at 10 kHz after weighting has been imposed.

Weighting Network, C-Message; Weighting Network, Flat. Two types of weighting, or filter, networks designed to de-emphasize noise values at certain frequencies, based on the relative disturbing effects of noise at the various frequencies.

White Noise. Random acoustic or electric noise having equal energy per cycle over a wide total frequency band.

Wideband. An information bandwidth equal to, or greater than, 20 kHz.

"Y" Connection. In power, a "star" connection whereby all elements have a common point, which may or may not be grounded.

"Y" Factor. In the measurement of a receiver's noise figure, one method involves measuring the noise out with no input (N_1), then a known amount of noise is introduced at the input and the output noise is again measured (N_2). The ratio of the noise powers, N_2/N_1 is the "Y" factor, which, when inserted into the proper formula, will give the noise figure.

Zero Relative Transmission Level (0dBr).

Zero Test Level Point (0TLP).

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