Multiplexing Techniques

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MULTIPLEXING is the process of combining two or more independent channels of intelligence into a composite signal, which is then transmitted through the selected transmission medium to a like terminal, where the process is reversed to restore the channels to their original state.
Frequency-division multiplexing (FDM) does as its name implies—the *frequency* spectrum is *divided* into 4-kHz increments. By means of *multiplexing*, many voice frequency channels can be placed within the frequency spectrum. Time-division multiplexing (TDM) is the process of combining two or more independent channels of intelligence into a composite signal where each channel is separated by time. TDM has become more common since the refinement of solid-state devices, which led to much more stable frequency generation and timing circuits. Both multiplexing techniques provide a means for combining or multiplexing a number of voice-band signals into a composite signal for subsequent transmission over a common line.

Characteristics of Speech

Originally, multiplex systems were designed primarily for the transmission of voice signals from one telephone to another. Telephones convert the human voice to electrical energy, which in turn is connected to a voice channel of a multiplexer. Voice channels are basically designed around the characteristics of the human voice

and ear. The receiving end reconstructs the signal to correspond in both waveform and frequency to the original sound waves so the listener can recognize and understand the words of the speaker.

Characteristics of speech and the effect of noise on speech transmissions

Today, not only speech frequencies, but data and facsimile information are multiplexed and transmitted over the system. Since the characteristics and frequency range of the voice-band signals play such an important part in the design of a multiplex system, let's examine some of the basic characteristics of oral and transmitted speech.

Speech characteristics. From the listener's point of view, the quality of a voice channel can be measured in terms of two parameters—*intelligibility* and *intensity*. Together, these parameters determine the quality of the reception of sounds transmitted over a channel. An interesting point is that they are virtually independent of each other over a broad range of frequencies. Most of the intensity, or speech energy, is concentrated in the lower frequency range, while the high frequencies contribute most to the intelligibility. Figure 2–1 shows that, if frequencies above 1 kHz were eliminated, approximately 86 percent of the voice signal *intelligibility* would be lost. Notice also that if frequencies below 1 kHz were eliminated, approximately 83 percent of the total *energy* in the voice signal would be lost. This indicates that a voice channel must include both the low and the high frequencies.

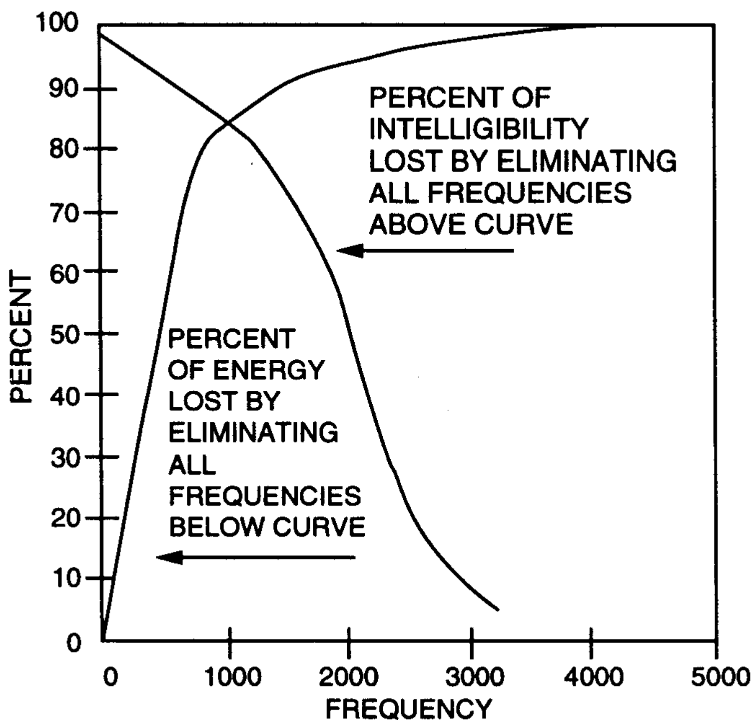


Figure 2–1. Voice frequency characteristics.

Transmitted speech characteristics. The distribution of speech energy over the frequency band varies from instant to instant as different sounds are spoken, because the vowels and consonants that make up the words contain their own distribution of energy over the frequency range. The listener identifies the vowel and consonant sounds by the amount of energy in the speech from moment to moment. How well the ear can do this determines the intelligibility of speech.

Figure 2–2 shows that, on the average, most of the energy is concentrated at the lower frequencies—near 350 Hz. The volume or loudness of speech is therefore determined largely by the lower frequencies. The figure also shows that frequencies below 1 kHz contain 80 percent of the speech energy. The speech intelligence, however, is conveyed principally by the weaker components—the higher frequencies.

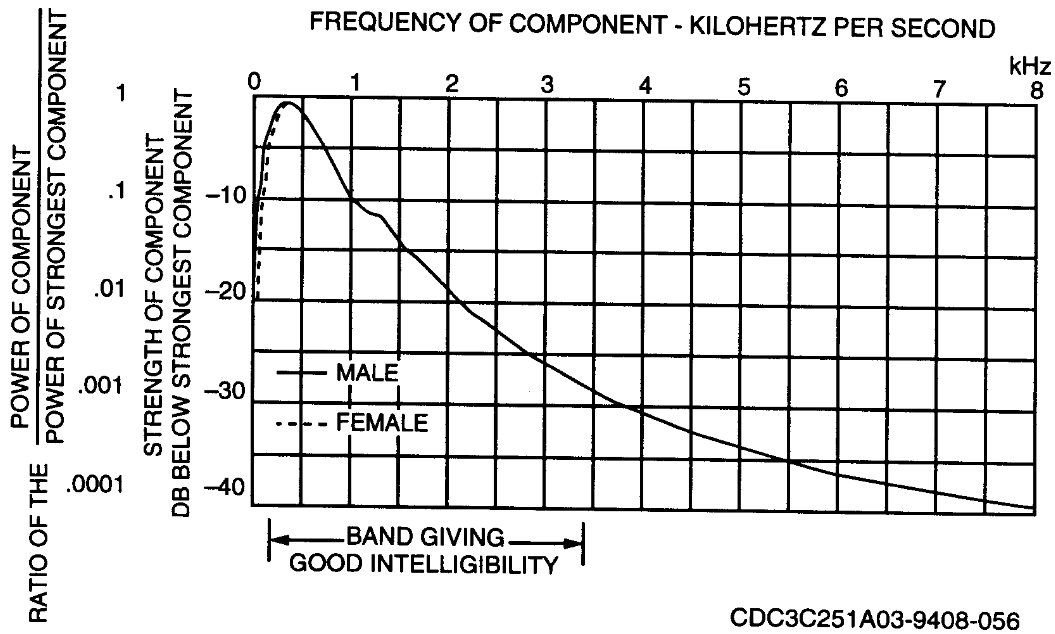


Figure 2–2. Average frequency spectrum of speech.

The sound intensity of speech also varies widely because of the wide diversity of talkers. The dynamic speech range of an average single talker is about 40 dB, which means that the energy content of the loudest speech components is 10,000 times greater than the weakest speech components (fig. 2–3). However, the strongest sounds of a loud talker may be 70 dB above the weakest sounds of a soft talker. Individual telephone conversation signal levels within a multiplex system are normally kept between 0 dBm0 and –31 dBm0.

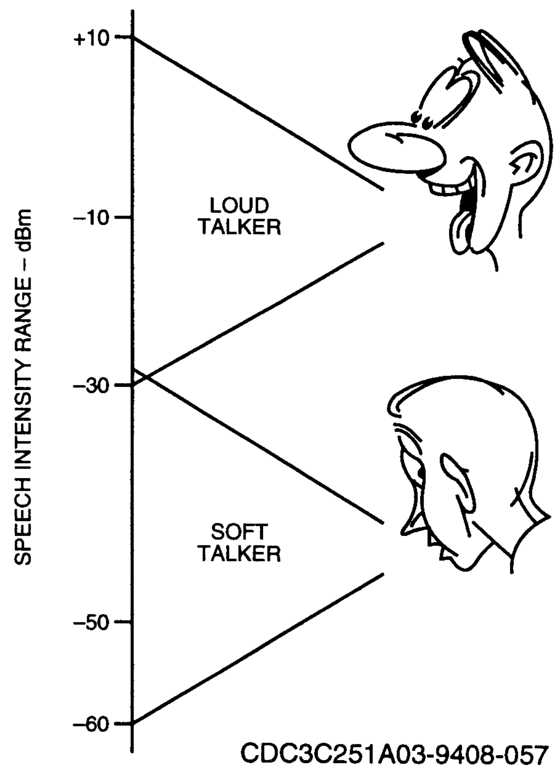


Figure 2–3. Speech intensities of people.

Another significant characteristic of speech is its redundancy. As much as 75 percent of the information content in normal speech is redundant. If a syllable, or even a word, is lost the listener automatically fills in the gap from the context. The result is that the requirements for speech transmission are often less stringent than those for other types of transmission, such as data and facsimile.

After numerous experiments and tests, the standard voice frequency bandwidth is accepted as 3.1 kHz with an actual band of frequencies from 300 to 3400 Hz, referred to as the voice frequency (VF) band. Even though the audio range of the ear extends on each side of the VF band, this band of frequencies provides a bandwidth sufficient to transmit speech without serious degradation.

Noise. Picture yourself in a crowded room talking to a friend. You can understand each other even though there are other people talking at the same time. Imagine that everyone stops talking except you and your friend. You and your friend now seem to be talking too loudly but, in both cases, the same amount of speech power was available to carry the intelligence. The other people's conversations were interfering with your conversation, causing you and your friend to talk more loudly than usual.

In communication equipment there is also interference, which we call *noise*. A signal represents a certain degree of order and pattern, but noise is random and encompasses the entire frequency spectrum. As a signal is being transmitted, noise is constantly trying to mask the signal and, if it is allowed to go too far, destroys the intelligibility of the signal. The signal is always at a disadvantage because it is always undergoing attenuation, while noise is constantly being reinforced at almost every point in the transmission path.

Sources of noise. The speech signal and the noise share the same conductor throughout the transmission path, whether it is wire, electronic components, or free space. Noise is constantly being generated within a conductor as electrons collide with some of the molecules of the conducting material. As the temperature increases, noise also increases because more electrons collide with the agitated molecules (thermal agitation). The amount of noise generated is directly proportional to the temperature of the conductor.

Another fundamental source of noise is called *black body radiation.* The most perfect radiator of energy would also be the most perfect absorber; therefore, a perfect black body would be an ideal radiator. The hotter the object, the more energy it radiates. This energy can be detected by a radio receiver along with its normal signal if the radiating energy is at the same wavelength as the received signal. The noises based on thermal agitation or radiation are sometimes referred to as "white noise," which has a uniform distribution of energy across the visible spectrum.

In most military applications, the output signal of the frequency-division multiplexer is usually applied to a microwave or tropospheric scatter radio set for transmission. The radio equipment is affected by white noise and by noise introduced by intermediate frequency (IF) amplifiers, mixer diodes, and possibly by mismatched antenna and waveguide components that add noise to the multiplexed signal. Although little can be done about noise originated outside the communication system, much progress has been made in reducing the noise that originates within the system.

Frequency-Division Multiplexing (FDM) Techniques

Now that you have studied some of the characteristics of speech and have established some parameters for the audio signal, let us look at the way numerous voice frequency (VF) signals can be modulated individually and then multiplexed into one composite signal. Each multiplex system employs some type of modulation scheme to shift the VF signals received from the subscriber to some suitable baseband frequency range. These schemes are called frequency allocation and modulation plans.

Before we discuss frequency allocation and modulation plans, you need a better understanding of the principle required for multiplexing a number of VF bands on a common line. Basically, each individual VF band can be changed to a different frequency band by amplitude modulation. Different frequencies applied to a common line do not interfere with each other and, therefore, can be transmitted as a composite signal.

Portions of this section were developed with the aid of material from the *GTE Lenkurt Demodulator*. Permission to use this material is gratefully acknowledged.

Principles of FDM

A frequency-division multiplex system, regardless of its channel capacity, is constructed of a number of channel groupings and arranged like *building blocks.* A 12-channel group is one of these building blocks and is a standard that most multiplex manufacturers use in their VF channel buildup. An organization known as the Consultative Committee on International Telegraphy and Telephony (CCITT), located in Geneva, Switzerland, issues the recommendations for standardizing those circuits used by equipment manufacturers throughout the world to ensure system compatibility. This standard 12-channel group has also been adopted by the Defense Information Systems Agency (DISA). Let's see how these building blocks are put together to form FDM systems.

Lower sideband 12-channel group. Refer to figure 2–4,A, and note that each VF channel has a different carrier, ranging from 64 to 108 kHz, assigned. Each carrier frequency has a 4-kHz separation and is harmonically related to 4 kHz. The significance of this is explained later in the text. The lower sideband is selected from each of the channel modulators, giving a total frequency range of 60 to 108 kHz for a 12-channel group.

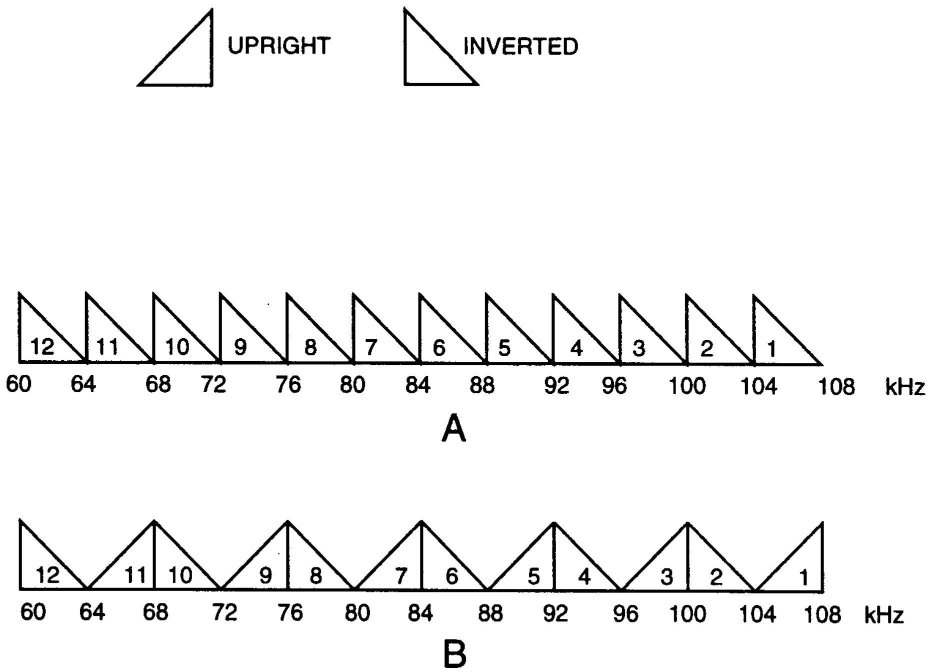


Figure 2–4. Channel modulation and demodulation plan.

As a systems controller, you must be able to locate a specific channel frequency within the multiplexed signal of 60 to 108 kHz. To illustrate this procedure, let us take the entire voice frequency band of channel 1, which is 300 to 3400 Hz, and calculate the translated frequency band. Note that the carrier frequency for channel 1 is 108 kHz. Remembering that the lower sideband is the carrier frequency minus the modulating frequency, proceed as follows: 108 kHz minus 0.3 kHz equals 107.7 kHz; 108 kHz minus 3.4 kHz equals 104.6 kHz, which translates to a frequency band of 104.6 to 107.7 kHz. To simplify our explanation, we round these frequencies off to 104 through 108 kHz. The same procedure holds true for the remaining 11 VF channels with the "stacking" of all 12 channels within a continuous frequency band of 48 kHz.

Using a standard test tone (1004 Hz), let's calculate the translated sideband frequency for channel 4. Using figure 2–4,A, again, you can easily find the carrier frequency for channel 4 (96 kHz). Simply subtract 1004 Hz (1.004 kHz) from 96 kHz to get 94.996 kHz. That's all there is to it!

As shown earlier, channel 1 has an output frequency band of 104.6 to 107.7 kHz. After modulation, the translated frequency band for channel 2 is 103.7 to 100.6 kHz, which leaves a *guard band* of 0.9 kHz between the two channels. The same 0.9-kHz guard band is also between the remaining 10 channels. The purpose of this guard band is to prevent interaction between adjacent channel filters. The total usable frequency bandpass for any channel in an FDM group is determined by the quality of the filter for that channel. The guard band also provides an unused portion of the frequency spectrum that can be used for the insertion of signaling tones or pilot tones. More on this subject is presented later.

Twin-channel 12-channel group. There is another method of obtaining a multiplexed 60- to 108-kHz 12-channel group used in some multiplex equipment. It is called *twin-channel modulation.* The main difference between the two methods is that the twin-channel process uses 6 carrier frequencies for translation rather than 12, as in the lower sideband method. Refer to figure 2–4,B, and note that both channel 1 and channel 2 use the same 104-kHz carrier frequency. An upper sideband of 104 to 108 kHz is selected by channel 1, while the lower sideband of 100 to 104 kHz is selected by channel 2. Note that the upper sideband is selected by all odd-numbered channels, while the lower sideband is selected by all even-numbered channels. Compare the channel modulation plan of the two methods and note that the sidebands of all odd-numbered channels are upright and, therefore, not compatible to "like" channels in the lower sideband method.

Further examination of the modulation process shows that the original low audio frequency (300 Hz) is now the high end of the sideband frequency; conversely, the high audio frequency is now the low sideband frequency. A triangle symbol is used to show the frequency relationship within the sideband in reference to the modulating frequency. Note in figure 2–4 that the triangle is shown in one of two positions, upright or inverted. When the audio signal is applied to a channel, the frequencies are shown by a triangle in the upright position or from the lowest frequency to the highest. After being modulated and with the lower sideband selected, the frequencies become inverted. When this sideband is applied to the next stage of modulation, the lower sideband is selected again and the frequencies revert back to an upright position. In the technical manuals for most multiplex equipment, a diagram shows the sideband relationship throughout the signal path.

If you are working in a large communications center with a number of different types of multiplex equipment, you may need to patch the signals of a 12-channel group from one piece of equipment to another. Before making the patch, make sure that the sideband frequency placement of both pieces of equipment is compatible. Refer to the frequency allocation and modulation plans for both equipment items to make this determination.

If the modulation plan of one piece of equipment shows that the upper sideband was selected, while the other equipment selected the lower sideband, the equipment is not compatible. If the two pieces were improperly patched and the multiplexed signals were demodulated, the low audio frequencies would become the high, while the high audio frequencies would become the low frequencies. If binary information were being transmitted, a binary "1" would become a binary "0" and vice versa. Most present-day equipment selects the lower sideband throughout the modulation process for both transmit and receive circuitry.

Sixty-channel supergroup. The 60-channel supergroup is the next building block in our multiplex system. Refer to figure 2–5 and note that a 60-channel supergroup is made up of five 12-channel groups with an output frequency band of 312 to 552 kHz. The carrier frequencies used for the modulation process are 420, 468, 516, 564, and 612 kHz. The lower sideband is selected from each of the five groups. This results in the output frequency band for group 1 being 312 to 360 kHz, for group 2 being 360 to 408 kHz, for group 3 being 408 to 456 kHz, for group 4 being 456 to 504 kHz, and for group 5 being 504 to 552 kHz. The outputs from the five groups are then combined and applied to a common line.

Six-hundred-channel line frequency or master group. The third and final modulation step is the line frequency or master group. It is made up of ten 60-channel supergroups and uses nine carrier frequencies to translate the supergroup frequency band of 312 to 552 kHz to a master group frequency band. Note in figure 2–5 that supergroup 1 is modulated with a 612-kHz carrier, resulting in a 60- to 300-kHz line frequency (lower sideband). Supergroups 3 through 10 are translated in a similar manner. The exception is the frequency band of supergroup 2, which is applied to the line with no further modulation. Note also that there are frequency bands of no modulation between each supergroup line frequency band. Supergroups 1 and 2 have a 12-kHz guard band between them, while the remaining 8 supergroups have similar 8-kHz guard bands. When all 10 supergroups have been modulated and multiplexed to form a 600-channel master group, the total frequency bandwidth is 2480 kHz wide, encompassing a frequency band from 60 to 2540 kHz.

We have “built” a 600-channel frequency-division multiplex system, starting with a basic building block of a 12-channel group. Depending upon the channel requirements of a particular system, a multiplex system can be designed in increments of 12 channels. In a 600-channel system, there are fifty 12-channel groups, ten 60-channel supergroups, and one 600-channel master group.

Now we are ready to trace a 1-kHz tone through the group, supergroup, and mastergroup levels. Let’s trace channel 7, group 1, supergroup 1, using these steps:

(1) Using figure 2–5,A, find the carrier frequency for channel 7 (84 kHz).

(2) Subtract the 1-kHz tone from the carrier frequency (83 kHz).

(3) Using figure 2–5,B, find the carrier frequency for group 1 (420 kHz).

(4) Subtract the translated frequency for channel 7 (83 kHz) from the group 1 carrier frequency (420 kHz - 83 kHz = 337 kHz). This is the frequency at the supergroup level.

(5) To calculate the frequency at the mastergroup level, simply subtract the 337 kHz from the supergroup 1 carrier frequency found in figure 2–5,C (612 kHz - 337 kHz = 275 kHz).

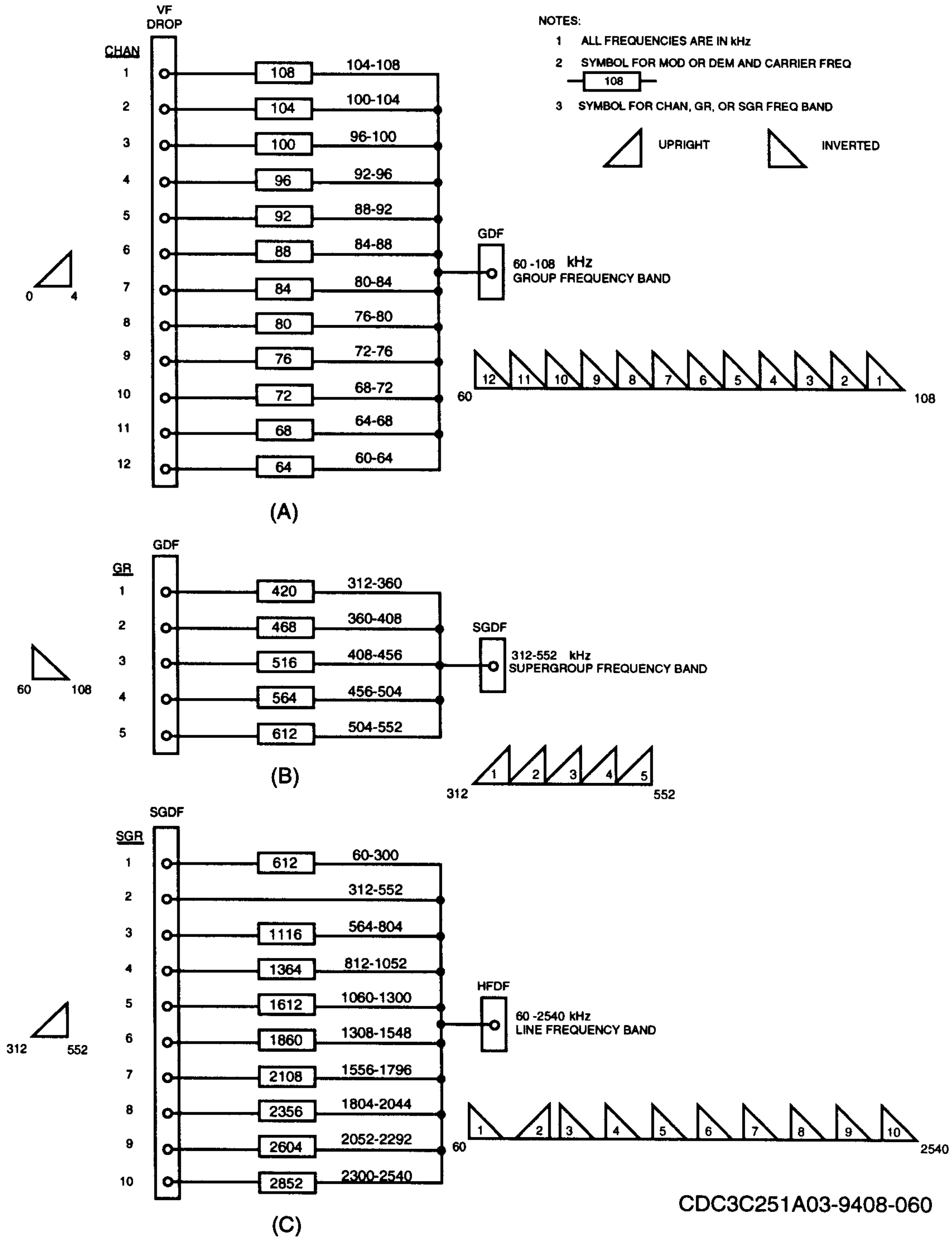


Figure 2–5. Frequency translation plan.

Frequency translation. In the preceding text, we discussed the process of multiplexing a number of VF channels and placing them on a common line. As a systems controller, you must be thoroughly familiar with frequency translation, impedance values, and how the signals flow through FDM equipment. Whether you work with a 4-channel or a 600-

channel system, you must have a working knowledge of each active circuit that is connected or interfaced with your FDM equipment.

Figure 2–6 is a simplified block and level diagram of an FDM terminal that illustrates the interfacing of three different signals to the FDM. Example 1 shows an individual talking on a telephone using the entire channel bandwidth. Example 2 shows an individual dialing a telephone and causing a pulsed signaling tone to be transmitted over the system. Example 3 shows a test tone of 1004 Hz being applied to a channel. The block representing the FDM shows the three steps of modulation; each step represents the number of components needed for a 600-channel system. Under the blocked circuits, the frequency bands associated for each translation step are given. Immediately below the frequency bands are the test level points (TLP) in dBm.

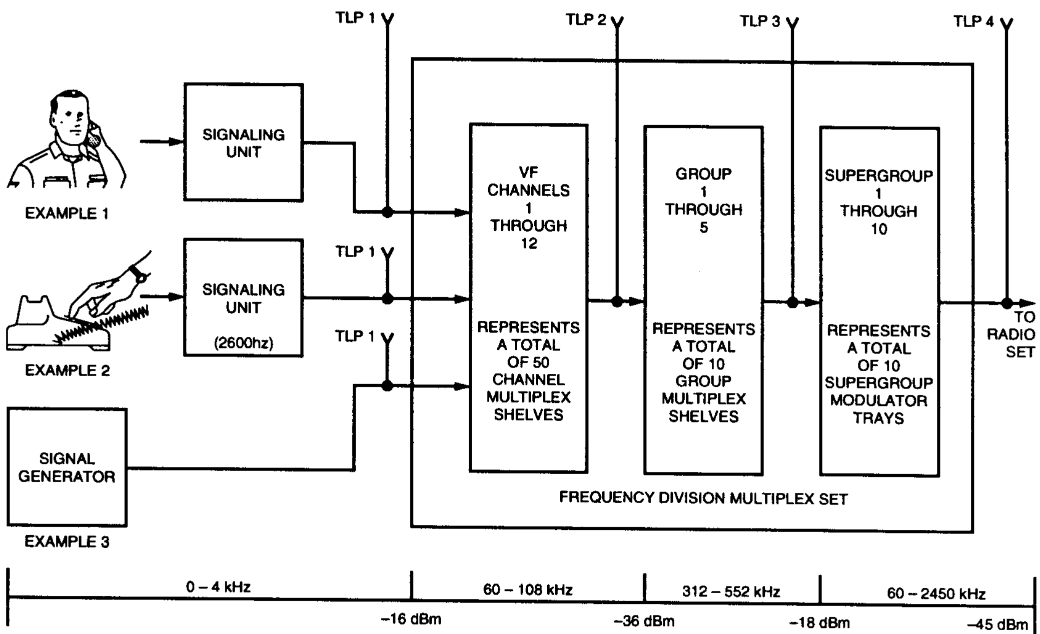


Figure 2–6. Block and level diagrams for a 600-channel FDM terminal.

Calculating power level in dBm at each TLP of an FDM terminal. Earlier in this CDC, you briefly studied basic measurements, levels, and decibels and how very important it is for you to have a complete understanding of this subject. In any system, there must be test points where measurements can be taken to ensure that all equipment in the system is operating within established parameters. Each point of measurement must have an established power reference so that all alignments, adjustments, and tests can be properly accomplished.

Refer again to figure 2–6. Let's suppose that a level of –26 dBm is measured at TLP1. The signal measured was a 1004-Hz test tone that was inserted for testing purposes. As we have mentioned before, the levels of the signals in a system must be correct, especially the test tones that are used for alignment and adjustment. A 1004-Hz test tone is inserted at a level of –10 dBm0 for testing purposes. Using the formula dBm0 = power level measured minus the TLP reference, we find that –26 dBm - (–16 dBm) = –10 dBm0, so the level measured at TLP1 was correct. Sometimes, as in the case of our –10 dBm0 test tone, the level is referred to as a *10 “down” tone.* In other words, the tone was 10 dB “down” from the reference TLP. Systems controllers use this slang terminology as a normal procedure, but you should be thoroughly familiar with the correct technical term, since it is used in testing procedures and technical literature.

Carrier and pilot generation

The circuits that produce the carrier frequencies are some of the most important in FDM equipment. Without these frequencies, the equipment could not operate. As in most FDM equipment, the carrier frequencies are directly or indirectly derived from a common-pulsed waveform. This waveform is rich in the harmonics of the fundamental frequency. By applying it to a filter, a specific frequency can be selected.

Carrier frequency generating equipment. The primary source for all the carrier frequencies in the multiplexer set (fig. 2–7) is the 128-kHz master oscillator. A highly stable 8-kHz pulsed waveform is generated from the output of the 128-kHz oscillator and is used as a source from which all carrier frequencies are produced.

a. *Master frequency generator*. The master oscillator of a multiplex set at one end of a system can be synchronized with its counterpart by designating one terminal as *master* and the other terminal as *slave*. The master terminal, the AN/UCC–4, transmits a 96-kHz synchronizing pilot tone, and the slave terminal compares the phase of the received tone with its own frequency. Any error detected is used to correct the master frequency oscillator at the slave terminal. Two complete master frequency generator trays are included in each multiplexer set with a provision for switching from one unit to the other automatically in case of failure. The output 8-kHz pulsed waveform is applied to the channel carrier generator and the group carrier generator.

b. *Channel carrier generator.* The channel carrier generator uses the 8-kHz pulse waveform from the master frequency generator to produce 12-channel carrier frequencies. The 8-kHz waveform is applied to a pulse-forming network where both an 8-kHz and a 4-kHz pulse are generated and applied to filters. The odd-numbered channels use the 4-kHz pulse, while the even-numbered channels use the 8-kHz pulse. All the channel carrier frequencies are related harmonically to the original pulsed signal. For example, the 108-kHz carrier is the 27th harmonic of 4 kHz. The maximum number of VF channels that a channel carrier generator can supply is 60. A system that requires an additional 60 VF channels must have a pair of channel carrier amplifiers.

c. *Channel carrier amplifier shelf.* Note in figure 2–7 that a system of over 60 VF channels requires one pair of channel carrier amplifiers consisting of one odd channel amplifier and one even-numbered channel carrier amplifier. These amplifiers are installed in pairs to supply 120 VF channels with carriers. Five channel carrier amplifier pairs are required for a 600-channel system.

d. *Group carrier generator shelf.* The 8-kHz pulse from the master frequency generator is used by the group carrier generator shelf to derive five group carriers plus a 124-kHz frequency from which eight supergroup carriers are produced. One output from the group carrier generator is a 516-kHz carrier and is applied to the group carrier supply; a second output contains four group carrier frequencies—420, 468, 564, and 612 kHz—and is also applied to the group carrier supply. A third output is a frequency of 124 kHz, which is applied to the supergroup carrier generator.

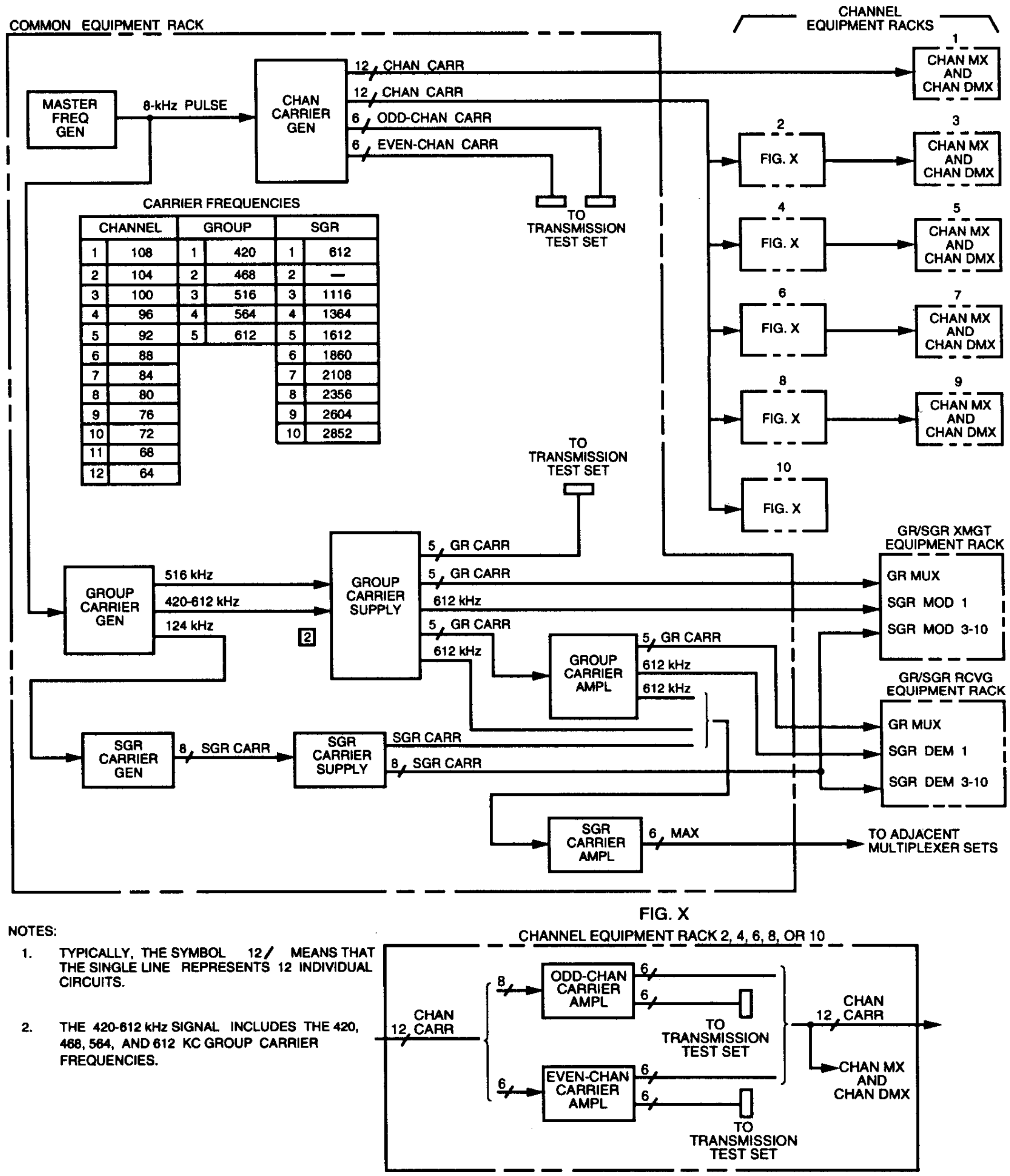
e. *Group carrier supply*. The group carrier supply accepts the five carrier frequencies provided by the group carrier generator, filters the individual frequencies, and amplifies them to the proper level for use by the group modulators and demodulators. The group carrier supply also provides the carrier frequency for supergroup 1, which can be either 612 kHz or 564 kHz, depending upon the line frequency requirement of the system.

f. *Group carrier amplifier shelf.* A group carrier amplifier shelf is needed when the total VF channel capacity exceeds 300 VF duplex channels. It can also be used to supply an adjacent multiplex terminal with group carrier frequencies.

g. *Supergroup carrier generator shelf.* The supergroup carrier generator uses the 124-kHz signal supplied from the group carrier generator to generate the eight carrier frequencies required by supergroups 3 through 10. The frequencies are 1116, 1364, 1612, 1860, 2108, 2356, 2604, and 2852 kHz. These frequencies are filtered before they are applied to the supergroup carrier supply shelf.

h. *Supergroup carrier supply shelf.* The supergroup carrier supply shelf receives the eight carrier frequencies from the supergroup carrier generator shelf and amplifies them to the proper level for use by the supergroup modulators and demodulators.

i. *Supergroup carrier amplifier shelf.* An optional output from the supergroup carrier supply shelf is applied to the supergroup carrier amplifier shelf. These supergroup carriers are amplified and can be applied to an additional multiplex set.



CARRIER FREQUENCIES

CHANNEL		GROUP		SGR	
1	108	1	420	1	612
2	104	2	468	2	—
3	100	3	516	3	1116
4	96	4	564	4	1364
5	92	5	612	5	1612
6	88			6	1860
7	84			7	2108
8	80			8	2356
9	76			9	2604
10	72			10	2852
11	68				
12	64				

NOTES:

1. TYPICALLY, THE SYMBOL 12/ MEANS THAT THE SINGLE LINE REPRESENTS 12 INDIVIDUAL CIRCUITS.
2. THE 420-612 kHz SIGNAL INCLUDES THE 420, 468, 564, AND 612 KC GROUP CARRIER FREQUENCIES.

Figure 2–7. Carrier generation for a 600-channel FDM system.

Functions of the pilot frequency and synchronizing pilot. A pilot is defined as a single frequency that is transmitted over a system to indicate or control its characteristics. It can perform the following functions:

- Be the reference frequency for the alignment of the signal path circuits.
- Provide alarm indications by monitoring the receive circuits.
- Synchronize the carrier frequencies between FDM terminals.
- Regulate the received multiplexed signal level.

The pilot frequency is usually referred to as either a *group pilot* or a *synchronizing pilot* and is inserted along with the multiplex VF channel frequencies.

The group pilot is usually inserted into the signal path along with the VF channel frequencies of each 12-channel group, while the synchronizing pilot is inserted at the line frequency band. The actual frequency of the pilot varies from equipment to equipment, with the most common group pilot frequencies being 64 kHz and 104.08 kHz. CCITT recommends that the group pilot be 104.08 kHz. In some equipment, the group pilot also performs the function of the synchronizing pilot.

Alignment. Built-in test equipment (BITE) is used to perform alignments and adjustments within some FDM multiplexers. The majority of the alignments use the group pilot as the signal level reference. Refer to foldout 1 in the back of this volume and note that the group pilot is inserted at the input of the group multiplex shelf for each of the five 60- to 108-kHz frequency bands. The pilot frequency goes through the same group and supergroup frequency translation and amplification as the VF channels. Whatever happens to one happens to the other. You can say that the pilot frequency represents the multiplexed VF channel frequencies as the composite signal proceeds through the transmit and receive circuits.

Alarm circuits. The group pilot that is inserted into the transmit circuits of an FDM at one end of a system is received by an identical FDM at the other end of the system. Alarm circuits monitor the incoming pilot and provide alarm indications if the pilot is not present or varies above or below a nominal level. Looking again at foldout 1, note that the pilot is monitored at the first step of demodulation in the supergroup demodulator tray. If an amplifier in the parallel amplifier circuit fails or the pilot signal level drops below the nominal level (–20 dBm0), an alarm is indicated.

The group pilot is monitored again at the group pilot alarm shelf. If the pilot varies above or below the nominal level, an alarm is indicated. Therefore, each 60-channel supergroup is monitored and demultiplexed into its five separate 12-channel groups, and each of those in turn is monitored.

Synchronization. In an FDM system, one of the FDM terminals is designated the “master” while the other terminal is designated the “slave” terminal. The master terminal transmits a synchronizing pilot that is received by the slave terminal. The pilot is transmitted only in one direction—master to slave; it is not transmitted back in the other direction. The pilot received at the slave terminal is routed to the carrier generation circuits, where it is used to control the frequency of the master frequency oscillator. This arrangement ensures that the carrier frequencies generated by both terminals are exactly the same.

Level regulation. Many FDMs do not provide signal regulation since the signal level variations of most systems are not serious enough to require it. Where there is an excessive amount of signal variation, it must be regulated. The group pilot frequency is normally monitored for signal level and the circuit gain is controlled automatically. The incoming pilot can vary as much as ±4 dB and the regulator circuit still supplies a stable output of not greater than ±.25 dB.

Types of signaling

Without a means of signaling, it would be impossible for a telephone system to operate. Even the simplest system requires some way for the users to attract each other’s attention when they want to talk. Early telephones used a hand-cranked magneto to generate a signaling voltage. This worked fine (and is still being used in some theater applications), but its transmission range was limited. As the number of telephones increased, the signaling methods changed from a DC signal to a 20-Hz AC signal that helped to extend the range of the system. Soon multiplexing became the major method of transmitting numerous telephone circuits from one point to another. This caused the signaling methods to change again.

The 20-Hz signal was converted to a frequency of 1600 Hz, which then could be passed by a VF channel. One drawback was that the 1600-Hz signaling tone could be duplicated by the human voice or by noise, resulting in false rings. This duplication of the signaling frequency is called talkdown.

Because the majority of speech energy is contained within the lower portion of the VF band, a higher signaling frequency of 2600 Hz is now used. This has helped to eliminate most of the talkdown problems.

Three basic methods of signaling are discussed here—out-of-band and in-band, common channel, and E&M signaling. Regardless of which method is used, the major functions of signaling can be broken down into three areas: information, supervisory, and control. Information signaling is an alert (audible or visible) that announces an incoming call. Supervisory signaling is the conveying of information regarding the switch hook conditions (on-hook or off-hook) at either end of the telephone circuit. Control signaling is the conveying of the dialing or digital information that is necessary to establish the desired connections.

Out-of-band signaling. Out-of-band signaling is defined as using a signaling frequency within the guard band between the VF channels. The frequency normally used and recommended by CCITT is 3825 Hz. The standard frequency used by the Bell Telephone System is 3700 Hz. Out-of-band signaling is used very little in the wide-band communications complex. Some of its disadvantages are that the signaling circuits have to be built into the multiplex channel equipment and cannot be separated, as in the case of in-band signaling. This type of signaling requires sharper frequency cutoff characteristics for the channel filters so as not to cause channel interference and a degradation of the channel frequency response. Another disadvantage is that out-of-band signaling requires a DC repeater at the end of each link of a multiple linked (multilink) system. As the signal passes from one link to another, the signaling tones must be detected and converted to DC and must “key” the signaling equipment in the next link of the system. The disadvantages of high cost, complexity, and signal distortion are the main reasons for not using this type of signaling.

In-band signaling. In-band signaling uses a signaling frequency within the frequency band of the VF channel. The frequencies normally used are 1600 and 2600 Hz. The principal objection to in-band signaling is that the signaling tones are within the channel speech band, leading to possible talkdown. Also, the signaling tones are audible and cannot be used during conversation. The biggest advantage is that both the speech and signaling tones share the same transmission media, but at different times. Since the signaling becomes a part of the modulated channel signal, DC repeaters are not needed as they are in out-of-band signaling.

Common channel signaling. Common channel signaling (CCS) is a third type of signaling technique and is being used more often in communication systems. As communication systems are being updated and modernized, common channel signaling is being used more extensively throughout the world to make use of its advantages. Common channel signaling is a signaling method using a link that is common to a number of channels and necessary for the control, accounting, and management of traffic on these channels.

To put it more simply, the basic concept of common channel signaling is that all signaling for a number of voice channels is carried over one common channel, instead of within each individual channel. As you probably realize, this has a distinct advantage over in-band and out-of-band signaling because it eliminates some of the circuitry and special filtering required in the channels. By preventing the interference of the signaling tones with the intelligence (data or voice) being transmitted over the channel, this type of signaling gives the communications user the best possible service.

Usually used with 24-channel time-division multiplexing (TDM) data streams, common channel signaling reduces the amount of overall noise users hear on their circuits. This is accomplished by eliminating the need for the gain control of the signaling tones within the channel bandwidth. For example, recall from your Keesler training that single frequency (SF) signaling requires stringent level discipline to operate correctly. SF is a telephone signalling method whereby individual frequencies are transmitted at specific power levels to convey dialing and supervisory information. For example, one method uses a steady 2600 Hz tone transmitted at a –20 dBm level to indicate an idle circuit condition. Now, most importantly, remember from earlier in this CDC that harmonics are generated from all signals and these harmonics retain a certain amount of the original signal power.

Figure 2–8 shows a block diagram of a typical common channel signaling (CCS) system. The voice paths come out of the central office switch termination and go directly to the carrier channel banks. The signaling for these voice paths enters the common control signaling interface units by loop signals or *E and M leads.* The M lead transMits "on-hook" or "off-hook" signals to the CCS equipment, and the E lead recEives "on-hook" or "off-hook" signals, from the CCS equipment. We cover E&M signaling in depth later.

Each CCS terminal contains four gate units, each of which is capable of handling six signaling channels.

The function of the interface units is to accept various signaling schemes and make them compatible with the CCS equipment. Signaling between telephone systems is done on an on-hook or off-hook basis, which is a one or zero binary situation. As such, it is well suited to the sampling unit in TDM techniques.

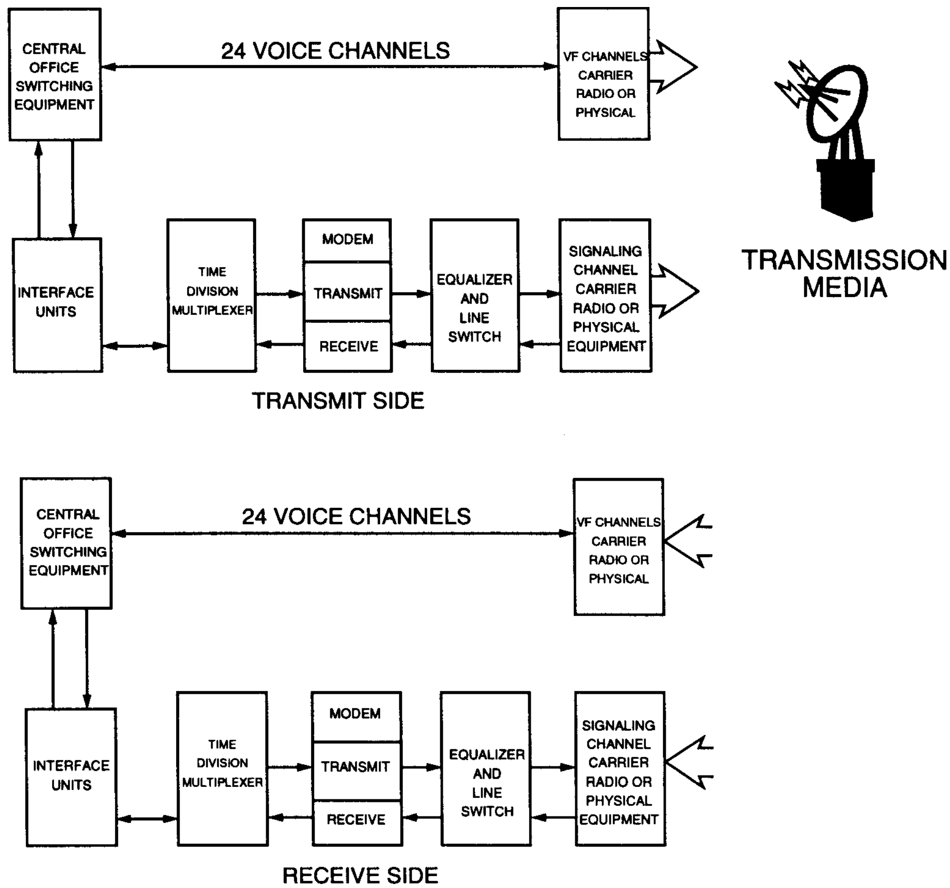


Figure 2–8. Block diagram of common channel signaling.
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A signal gate unit in the time-division multiplexer connects to the signaling portions of the interface units and samples the transmit input for on-hook or off-hook conditions at approximately 2,400 times a second. Each sample is sequentially fed to a 2.4-kilobit modem. In this way, 24 parallel signals are serialized into one TDM stream. At the modem, the TDM stream is transformed into an FSK (frequency shift keying) signal within the voice band which, in turn, can be put over any standard voice channel. The reverse of this procedure takes place at the receiving station.

E&M signaling. As we mentioned previously, transmitting a 20-Hz frequency was an early method of signaling, but this method limited the distance of transmission to approximately 25 miles. To help overcome this problem and extend the range, another type of signaling was developed. This type of signaling uses two leads, called the *E and M leads,* to connect the signaling equipment with the loop or the subscriber equipment. Foldout 2 in the back of this volume shows a simplified in-band signaling unit employing E&M signaling. The M lead is the transmit lead. Another way you can remember this, in addition to the example given above, is by associating the letter M with the word “mouth.” The E lead is the receive lead and can be associated with the word “ear.” A detailed explanation of how the signaling unit functions is presented in the following text.

Audio circuits. Normally, this particular in-band signaling unit is connected between the VF channel and the telephone exchange equipment. A typical signaling unit is diagrammed in foldout 2 at the end of this volume. The transmitted voice signal is connected straight through the drop side (TX DROP 600 ohms) to the line side (TX LINE 600 ohms). A line-cutting relay opens the drop side during the dialing sequence to keep the 2600-Hz signaling tone out of the exchange equipment. The signaling tone is bridged across the transmit voice path from the signal switch circuit.

The receive voice signal enters the signaling circuit from the line side (REC LINE 600 ohms). A transformer matches the impedance values of the line and signaling equipment. A portion of the receive signal is routed to the receive signaling path at the secondary of transformer T2. The receive signal continues past the first 2600-Hz band elimination filter, which prevents the 2600-Hz tone from reaching the receive drop side. A single-stage amplifier and its associated circuitry is used to adjust the receive VF signal to the proper level. A second 2600-Hz band elimination filter ensures that the signaling tone is completely suppressed. Output transformer T1 matches the drop equipment and signaling unit output impedance values and couples the receive VF signal to the drop side (REC DROP 600 ohms).

Transmit signaling. In the transmit direction, signaling is controlled by the application of ground (on-hook) or –48 VDC (off-hook) to the M lead. An on-hook condition is an idle state in which the telephone is not being used. The M lead applies either condition to the line-cutting network, time delay network, signal switch, and high-level switch circuits.

Each time the M lead changes condition, the line-cutting relay is momentarily energized. The line-cutting relay opens the transmit drop line and terminates the transmit line in its characteristic impedance. This is done to reduce the transmission of voltage spikes (transients), which may be generated by the drop equipment during dialing and operations. During the dialing sequence, the M lead condition alternates between a ground condition and –48 VDC at the dialing rate. The time constant of the line-cutting network is such that the drop side (TX DROP 600 ohms) remains cut during the entire dialing sequence and the line-cutting relay does not follow the M lead pulsing.

The E&M lead cross-control network eliminates the effect of transients generated when the E lead is being pulsed by incoming signaling from the line (REC LINE 600 ohms). When both E&M leads are in the on-hook condition, the cross-control network keeps the line-cutting relay energized. The line-cutting network components keep the relay energized when the E lead is being pulsed. These transients would interfere with returning the on-hook signal to the originating office and cause a false off-hook condition on the E lead.

The M lead connects with the 2600-Hz switching circuits through a 20 millisecond delay circuit. This delay circuit provides a delay between the line-cutting operation and the application of the 2600-Hz signal to the line (TX LINE 600 ohms). This delay allows any transients to decay before a signaling tone is transmitted, thus cleaning up the dial pulse on the signal switch circuit.

The signal switch circuit allows the 2600-Hz signaling tone to be applied to the line at a low level during the on-hook (M lead grounded) condition. This condition is known as tone-on idle, which means that a continuous signaling tone is being transmitted even though the circuit is not being used. The power level of the tone during this condition is much lower than when an actual signaling condition occurs (off-hook). There is also a tone-off idle condition option. The signal switch circuit also cuts off the tone during an off-hook (M lead at –48 VDC) condition.

During the dialing sequence, the high-level switch shunts a 12-dB pad. This causes the signaling tone bursts to be 12 dB higher than the normal tone level. This increased level ensures a positive signaling action. When the circuit goes into idle condition and the M lead is again grounded, the signaling tone remains at a high level for approximately 400 milliseconds. When the circuit components of the high-level switch are discharged, the tone-on idle signal drops 12 dB to the normal level.

Receive signaling. In the receive direction, the composite signal at the output of T2 is applied to the limiting amplifier network. The limiting amplifier network removes noise peaks from the signal that could cause false E lead off-hook conditions. The output is applied to a pair of 2600-Hz filters—the signal network and the guard network.

The signal network offers a high impedance to the 2600-Hz signaling tone and passes maximum voltage to the signal detector when this frequency is present. The guard network offers a high impedance to any frequency other than the 2600-Hz signal and minimum impedance to the signaling tone. This network passes maximum voltage when frequencies other than 2600 Hz are present, such as speech and noise.

The signal detector rectifies and doubles the voltage received from the signal network. A positive voltage is applied to the signal switch when a 2600-Hz tone is received. The guard detector rectifies and doubles the voltage received from the guard network, and a negative voltage is applied to the signal switch. The function of the guard circuit is to prevent speech signals from operating the signal switch circuit and pulsing relay, which would provide false E lead operation.

The positive and negative voltages from the outputs of the detector circuits control the signal switch. When the positive voltage from the signal detector is greater than the negative voltage from the guard detector, the signal switch conducts, causing the pulsing relay to energize. If the negative voltage is greater than the positive voltage, the pulsing relay does not energize and the signal switch remains cut off.

The function of the receive signal switch is to pulse the E lead at the incoming signaling rate. The absence of 2600 Hz applies a ground to the E lead. The presence of 2600 Hz places an open on the E lead.

The heart of the pulsing relay circuit is the delay reform circuit. A two-winding saturable-core reactor is used to delay the pulse, reform the pulse duration, and nullify the effect of any pulse less than 20 milliseconds in duration. This action provides an increased protection from false E lead operation and a more positive E lead operation during receive signaling.

The guard relay energizes when the signaling tone is absent from the receive line, as during an off-hook condition at the originating office. This relay closes a set of contacts to remove any guard circuit voltage to the signal switch. The slow release action of the guard relay prevents a false on-hook operation of the E lead during the initiation of the call and ensures a positive signaling action in the receive circuitry.

The pulsing relay is driven by the signal switch circuit through the pulse-delay reform circuit. A ground condition is placed on the E lead during an off-hook condition or the absence of the 2600-Hz transmitted from the distant terminal. An open condition on the E lead indicates an on-hook or tone-on condition from the receive line. During the dialing sequence, the E lead alternates between open and ground conditions in response to the rate of the incoming 2600-Hz tone bursts (dial pulses).

The signaling supply shelf contains two identical oscillator units. The oscillator can supply a total of 144 separate VF channels. If one of the oscillators should fail, the other automatically takes over. The output level is adjustable; if the level drops 4 dB, an alarm activates and the standby unit comes on line.

Specialized components

As a systems controller, you work with many components that are designed for a specific purpose. These specialized components, such as filters, pads, attenuators and hybrids, are used extensively in FDM systems. For instance, in FDM equipment, the frequency spectrum is divided into many discrete frequency bands. Each VF channel occupies an approximate 4-kHz frequency slot within the band. Normally, these frequency bands are separated by electrical filters. Although design of the filters is above the level of this course, you can readily understand filter applications and operation if you have a knowledge of the basic characteristics of the various filter types.

Basic filter configurations. Although we touched on filters earlier, you need a better understanding of how and when they're used. A filter is a device that has a particular resonance characteristic. It commonly consists of capacitors and inductors. In the most basic case, a simple series circuit consisting of a capacitor and an inductor is a filter. Because of the specific requirements of an FDM system, filters must be designed to pass or reject a certain frequency or band of frequencies. Most filters in an FDM system contain a number of reactive components arranged in series or parallel, or a combination of series and parallel depending on the filter requirements.

Design parameters for a filter are obtained from such factors as:

- The impedance, both input and output, into which the filter must be connected.
- The frequency range to be passed or rejected.
- The attenuation or sharpness of the cutoff frequencies to be passed.
- The loss that may be tolerated in the pass band.
- Whether similar filters are operating in parallel (flanking).

Each of these factors should be considered in the application of filters. A change in circuit impedance, such as the removal or addition of a parallel circuit connected to the filter, definitely changes the filter's characteristics.

The basic configurations, shown in figure 2–9, into which the capacitors and inductors can be assembled to form filters are the (1) L-section or half-section, consisting of one series and one parallel arm; (2) the full T-section, consisting of two series arms and one parallel arm (note the resemblance to the letter T); and (3) the full TT-section, consisting of one series arm and two parallel arms (resembling the Greek letter *PI* (π). Several sections of the same type can be joined to improve the attenuation or pass band characteristic.

Filters are usually terminated by resistances of the same value at both the input and output terminals. Normally, the value of this terminating resistance is determined by the circuit with which the filter is used. The input and output impedance values, bandpass, attenuation characteristics, and cutoff frequency of the filter are determined by the values of the capacitors and inductors. The determination of actual component values is a design function and falls outside the scope of this course.

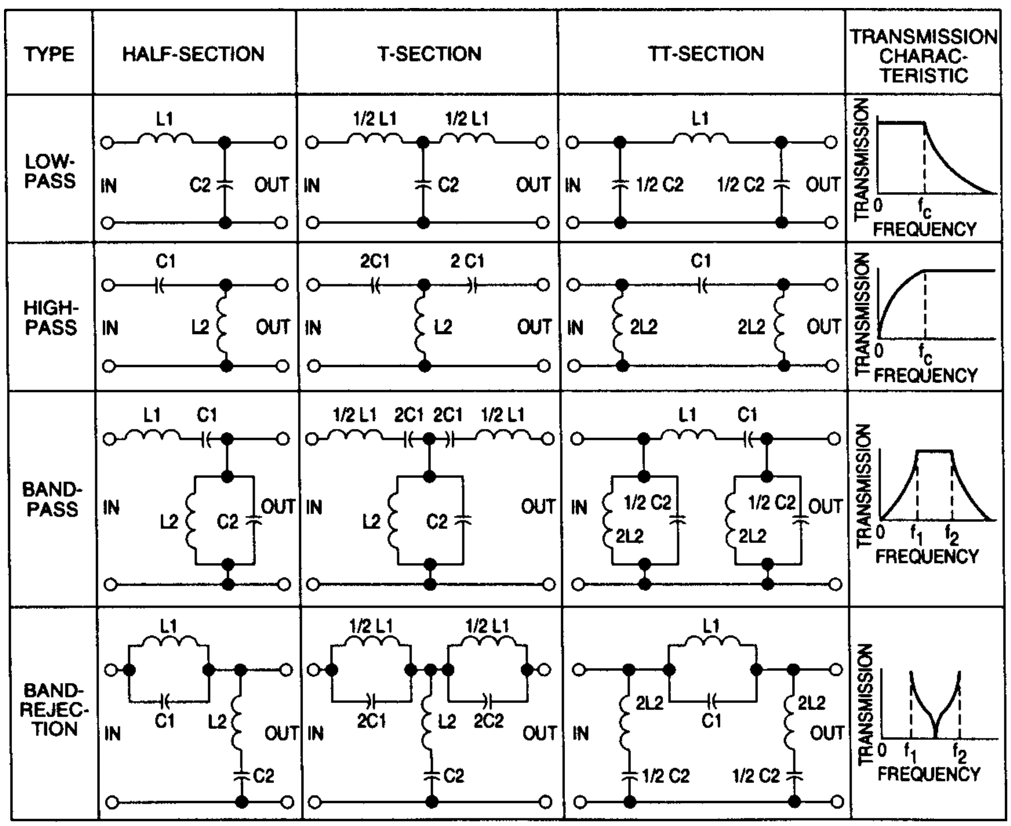

Figure 2–9. Basic filter configurations.

The filters that are used in carrier equipment are classified into four basic types according to their frequency response characteristic. The four types are low-pass, high-pass, bandpass, and band-rejection, or band-elimination. All four types are made up of series-resonant and/or parallel-resonant circuits that determine the frequency characteristics of the filter. Figure 2–10 shows some examples of the schematic symbols used to designate the four types of filters. As you will see in the following paragraphs, these schematic symbols indicate the attenuation characteristic of the various filter types.

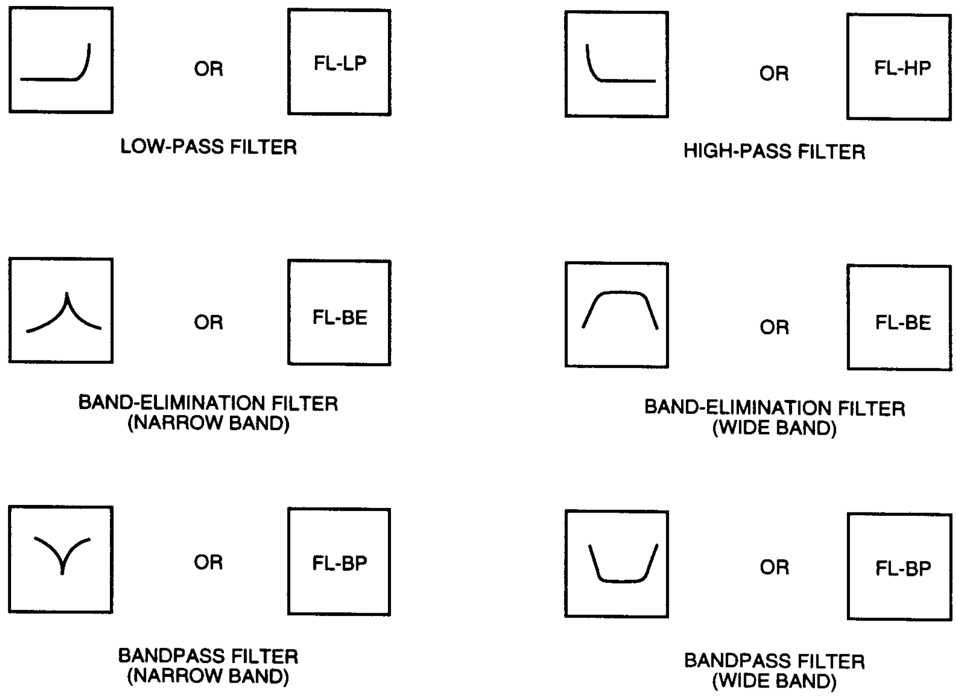

Figure 2–10. Schematic symbols for filters.

Low-pass filters. Low-pass filters pass frequencies below the cutoff frequency (f_c) and effectively block all frequencies above the cutoff frequency. You can understand this action by considering the low-pass T-section filter in figure 2–9. At high frequencies, the inductive reactance of the two coils in series is large; hence, they offer large opposition to the high-frequency current. Therefore, very little high-frequency current flows through the termination (load). In addition, any high-frequency current that does get through the first coil passes through the low reactance of the capacitor and thus does not reach the output.

For low frequencies, however, the inductive reactance is small and the capacitive reactance is large. Thus, these low-frequencies pass readily through both coils to the load. This is shown by the transmission characteristic curve in the figure.

The half-section and TT-section filters can be analyzed in the same manner as the T-section above. Since this is true in all four filter types, we discuss only one configuration for each type of filter.

High-pass filters. High-pass filters transmit frequencies above the cutoff frequency (f_c) and attenuate all frequencies below the cutoff frequency. In the T-section high-pass filter (fig. 2–9), all low frequencies that do get through the first capacitor pass through the low reactance of the coil and do not reach the output. As you recall, at high frequencies the capacitive reactance is small and the inductive reactance is large. Consequently, the high frequencies pass easily through both capacitors to the output and not through the coil.

Bandpass filters. Bandpass filters have both an upper and a lower cutoff point (f_1 and f_2 in fig. 2–9) and permit the transmission of only those frequencies between the two cutoff frequencies. This type of filter is considered to be made up of a low-pass filter and a high-pass filter whose bandpass characteristics overlap.

Now, let's examine the T-section bandpass filter. The series-resonant and parallel-resonant circuits are tuned to the center frequency band desired to be passed by the filter. As in all cases, the series-resonant circuit offers a low impedance within this band, while the parallel-resonant circuit offers a high impedance. Hence, the desired frequencies within the band are applied to the load with negligible attenuation; the unwanted frequencies (frequencies outside the desired band) are applied to a high-series impedance and then shunted through the low impedance of the parallel-resonant circuit to a common connection. You can see that the unwanted frequencies are severely attenuated and do not reach the output.

Band-rejection filters. Band-rejection filters (also called *bandstop* or *band-elimination filters)* suppress a range of frequencies between two cutoff frequencies—f_1 and f_2—and pass freely all signals above and below this range, as shown by the transmission characteristic in figure 2–9. This type of filter is effectively the opposite of the bandpass filter; i.e., frequencies within the band are greatly attenuated because the series-tuned and parallel-tuned circuit in series with the input offers a high impedance to this band of frequencies. The series-tuned circuit in parallel with the output terminals offers very low impedance. Thus, the signals within the band to be eliminated are blocked and shunted from the load through the series-resonant circuit. At the same time, the frequencies outside the pass band are applied through the parallel circuit with very little opposition and are not affected by the high-impedance (series-tuned) circuit.

Figure 2–11,A, shows the application of a low-pass filter and a bandpass filter in a typical VF channel. The frequencies applied to the channel can be anything within the audio-frequency spectrum. The low-pass filter allows only frequencies below 3400 Hz to pass. The output of the modulator is applied to a bandpass filter, which normally selects the lower sideband frequencies. Figure 2–11,A, also shows the characteristics for the filters.

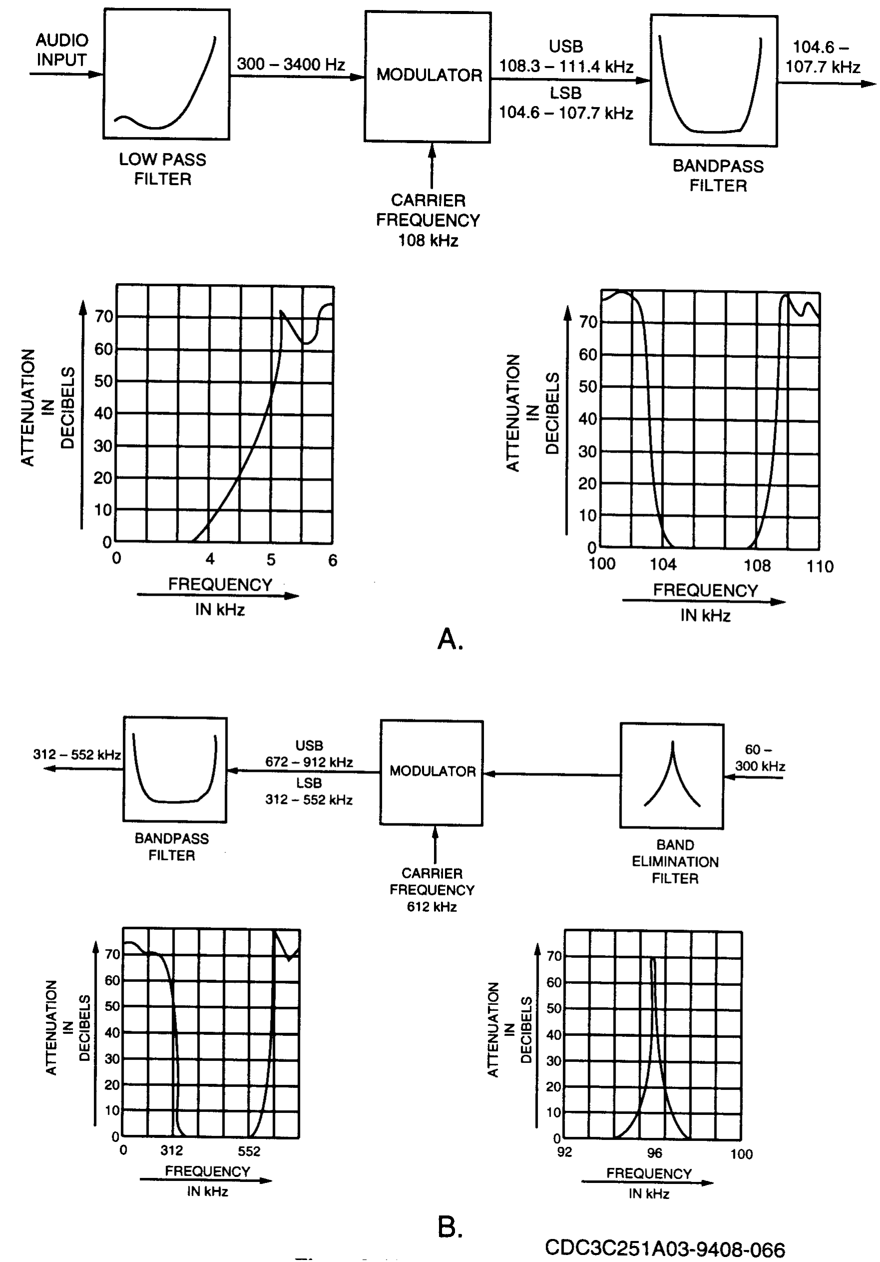

Figure 2–11. Filter applications.

It is sometimes necessary to suppress or severely attenuate a pilot frequency to avoid interference. Use a band-elimination filter to suppress this frequency (fig. 2–11,B).

Applications for resistive pads. In multiplex equipment, it is often necessary to control the attenuation of signals and to attenuate their different frequency components equally. The resistive pad and attenuator networks accomplish these actions.

A pad is defined as a device for reducing the amplitude of a wave without introducing appreciable distortion. In other words, a pad simply reduces the strength of a signal. In general, a symmetrical pad is inserted between matched networks (the source impedance equals the load impedance) for attenuation purposes only. A nonsymmetrical pad is inserted between mismatched networks both for attenuation purposes and for impedance matching. Specific uses of pads are as follows:

a. They are used to reduce the strength of: voice-band signals before modulation, sidebands before modulation, and incoming sidebands before demodulation.

b. They are placed at the input of fixed-gain amplifiers to obtain overall gain variations.

c. They are placed at the input of amplifiers to prevent amplifier overloading.

d. In conjunction with a voltmeter, they can be used to measure the gain of an amplifier.

Four symmetrical pads are the T, the pi, the H, and the square. A fifth pad is nonsymmetrical and is called the *L-pad.* The symmetrical pads are shown in figure 2–12. The T-pad and the pi-pad are basically the same and are used in unbalanced circuits; that is, where one side of the line is at ground potential. In contrast, the H-pad and the square-pad are basically the same and are used in balanced circuits, where neither side of the line is at ground potential.

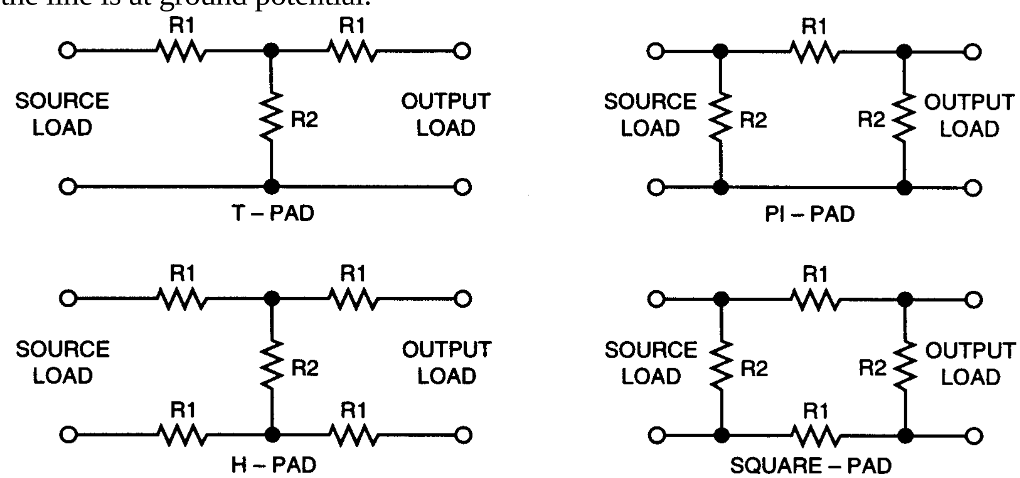

Figure 2–12. Symmetrical pads.

A symmetrical T-pad is represented in figure 2–13. It is inserted between two equal resistances, R_g and R_L. It is symmetrical because it consists of two equal series resistances, R1, one on each side of the junction with the shunt resistance, R2. This pad is not balanced to ground. It is employed when one side (the side not containing series resistances) of the attached network is grounded or when it is not necessary to have the attached network balanced to ground.

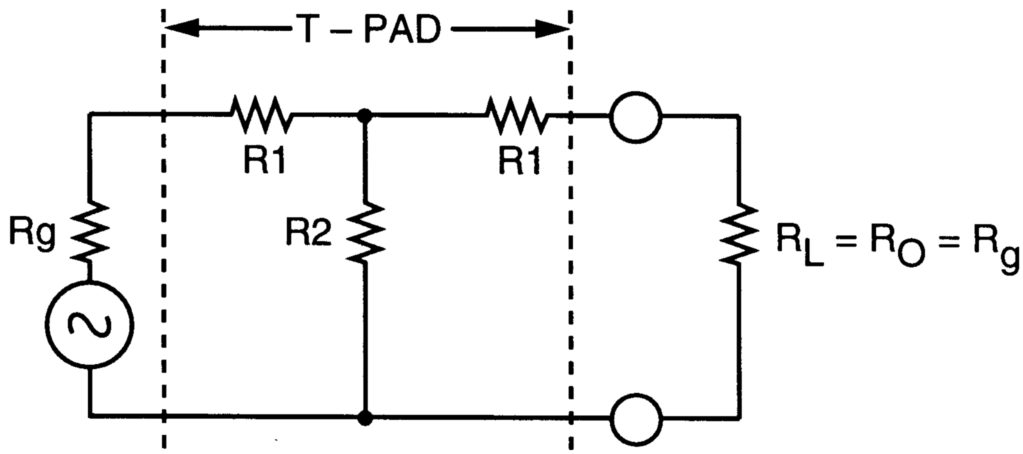

Figure 2–13. Symmetrical T-pad.

Practical uses for hybrids. The basic use of hybrid circuits is to convert from a four-wire to a two-wire condition or vice versa. Any device that provides impedance matching between certain circuits and isolation between other circuits may be referred to as “hybrid junction,” or more commonly as a “hybrid.” It may be a 3-winding transformer, a resistive bridge, or a waveguide device for microwave frequencies. In common telephone or VF circuits, the term means a junction between a balanced four-wire circuit and a balanced two-wire circuit.

Transformer hybrids. The transformer hybrid is the oldest and most widely used hybrid type. Figure 2–14 depicts a transformer hybrid arrangement to connect a signal between a four-wire and a two-wire circuit. A signal coming into the hybrid at the four-wire receive terminals produces a current flow through the primary winding of transformer T1. The current through the two halves of this winding induces equal current in the secondary windings 2 and 3, which are connected to the balancing network and the two-wire line respectively. Thus, half the applied power goes to the two-wire line and the other half is dissipated in the balancing network. Since half the power is wasted, the minimum theoretical loss as the signal goes through the hybrid is 3 decibels. A line amplifier is normally used to raise the signal back to its proper level.

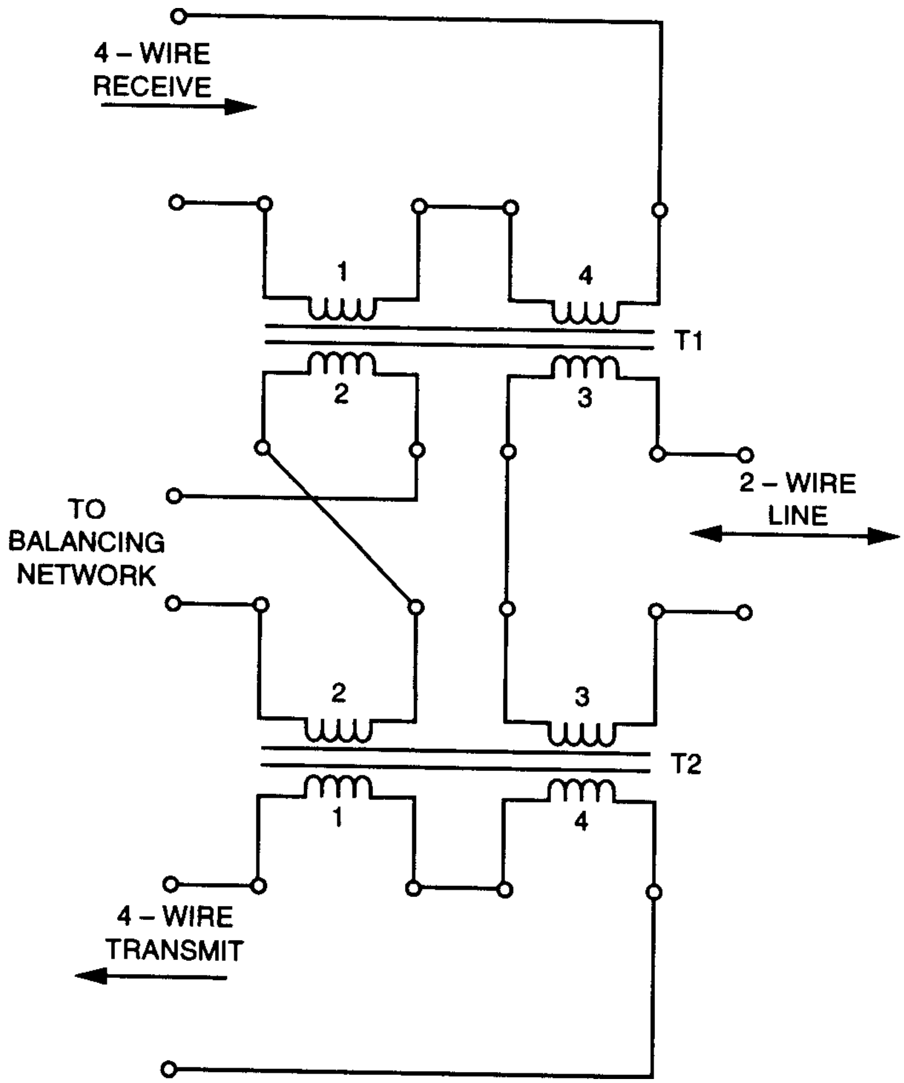

Figure 2–14. Two transformers connected as a hybrid transformer.

Windings 2 and 3 are connected in series to two corresponding secondary windings of transformer T2. The current to the balancing network also flows through winding 2 of T2 and the two-wire line current flows through winding 3. Both these currents induce potentials in the halves of transformer T2 primary, but these induced voltages are equal and of opposite polarity, effectively canceling each other. Thus, no current flows in the transmit branch.

This, of course, is the ideal case, providing complete isolation between the two sides of the four-wire circuit. In a practical case, several factors make the ideal unattainable. Both transformers must be carefully constructed for the voltages involved in the final canceling operation to be actually equal values and of opposite polarity. A slight difference in the windings, for example, would result in a net difference between the voltages and would produce a current through the transmit circuitry.

Such a current could result, even if the transformer windings were perfectly matched, because the impedance of the balancing network must also match the two-wire line. If these two impedance values are not the same, different currents flow and different voltages are induced in winding 1 of T2. These factors prevent full cancellation. The result is the same as though the transformer windings were not matched. By carefully constructing the transformers and matching the balancing network to the two-wire line, the manufacturer can obtain satisfactory isolation between the sides of the four-wire circuit.

For transmission in the other direction, the signal enters the hybrid from the two-wire line and is split equally between the transmit and receive branches. This current flow in the primary windings of both transformers induces voltages in the number 2 secondary windings of both transformers. These induced voltages are equal and of opposite polarity, so no power goes to the balancing network. Again, the minimum loss is 3 decibels because half the power is wasted in the receive branch. In practice, the loss in either direction of transmission is likely to be about 3.5 decibels because of transformer core losses and winding resistance.

Resistive hybrids. A hybrid can also be made of resistors connected in the form of a Wheatstone bridge. Figure 2–15, A, shows a resistive hybrid as usually drawn, while figure 2–15,B, is the same hybrid redrawn to illustrate the bridge configuration more clearly.

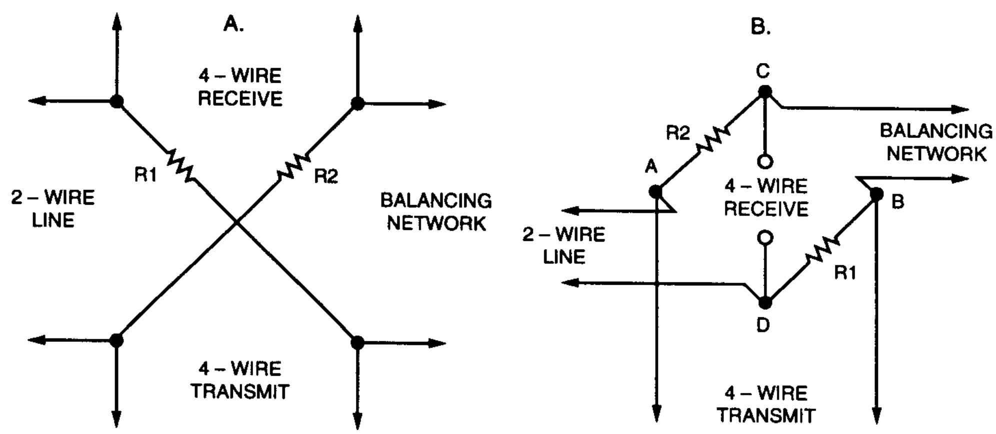

Figure 2–15. Resistive hybrid arrangement.

Assume that a signal is applied to the receive terminals and that both R1 and R2 are equal to the resistance of the two-wire line and the balancing network, which is normally 600 ohms. The signal power is split, with half going to R1 and the balancing network, and the other half going to the two-wire line and R2. Points A and B are at the same potential, so no power is transferred to the transmit branch. Furthermore, since the power dissipated in the resistors and the balancing networks is wasted, the two-wire line receives only one-fourth of the power. In other words, the resistive hybrid has a minimum loss of 6 decibels when all arms of the bridge are equal.

For transmission in the other direction, assume that a signal is applied at the terminals of the two-wire line. This signal is divided equally among the transmit and receive terminals and their associated resistors—R1 and R2. No power goes to the balancing network because points B and C are at the same potential. Since that portion of the signal applied to the resistors and to the transmit branch is wasted, the receive branch gets only one-fourth of the power and again the minimum possible loss is 6 decibels.

In the foregoing discussion, we have assumed that all arms of the bridge have equal impedance, but this is not a necessary condition for hybrid balance. The requirement for balance is only that the product of resistances R1 and R2 must equal the square of the nominal hybrid impedance.

Comparison of hybrid types. Whether to use a transformer or a resistive hybrid may depend upon several factors, but in some cases there may be literally no choice. For example, if the circuit is to be used at frequencies above 1 MHz, a transformer hybrid is not likely to be seriously considered. Such factors as iron loss in the transformer core and interwinding capacitance severely limit the high-frequency performance of transformer hybrids, while resistive hybrids are nearly independent of frequency—at least until the microwave region is reached. On the other hand, if the proposal is a VF application, the choice will probably be a transformer hybrid, simply because its loss is so much less than that of the resistive hybrid.

If physical size and weight enter into the problem, the resistive hybrid has an advantage over the transformer type. Iron-core transformers are heavy and bulky in comparison with resistive hybrids.

Cost is nearly always a factor in the choice of a hybrid. A resistive hybrid is less expensive than one that uses transformers, but the resistance type also has inherently more loss. In many cases, the cost of the hybrid itself must be balanced against the cost of additional gain in the system.

Time-Division Multiplexing (TDM) Techniques

Because time-division multiplexing (TDM) is the most direct method and is basically the simplest, it is being used more extensively now than ever before. In this method, the circuits to be transmitted over a common transmission medium need not be connected together, as in the frequency-division method, but are so arranged that each circuit is successively connected to the transmission medium for a short time interval.

All TDM signal channels "time-share" a common transmission channel and, in effect, are transmitted serially. The direct relationship of the composite signal to the original input signal that you find in the frequency-division multiplex (FDM) subsystems is not in TDM subsystems, because of the nature of the basic TDM process. The TDM method employs sampling techniques. If a signal is sampled at a rate twice its highest frequency component, an adequate representation of the signal may be obtained. If there are a number of channels to be sent over a common transmission path, the first channel is sampled briefly, then the second, etc., until the last channel has been sampled. After the last channel is sampled, the process is repeated. Before each sample is applied to the common transmission path, some form of pulse-modulation technique is used to form the composite TDM signal.

Portions of this section were developed through the use of materials provided by Van Nostrand Reinhold Company, Incorporated. Permission to use this material is gratefully acknowledged.

TDM sampling, bandwidth, and synchronization

Since in time-division multiplexing the modulating wave is repetitively sampled, the size and frequency of samples are prime considerations in retaining the intelligence to be transmitted.

Sampling rate. It has been found that, because the time duration of the sample is not critical, it can be reduced as needed without degrading the information. The samples must be at a rate that is at least twice the highest frequency appearing in the intelligence signal, as in any pulse transmission system (Nyquist sampling rate). The sampling rate commonly used is 8000 hertz. At this rate, each circuit is sampled once every 125 microseconds. The 125-microsecond interval between successive samples of one channel may be used by other channels of the multiplex system. Theoretically, the number of channels that can be obtained is very high.

Bandwidth. As the number of channels is increased, the bandwidth required for transmission increases rapidly because the pulse train spectrum is made up of a fundamental frequency that is equal to the sampling rate (8 kHz) and its harmonics, all of which have an upper and lower sideband produced by the modulation process. As an example of the bandwidth required, consider a PAM system of 30 channels. This system requires a bandwidth of approximately 1 MHz. This is about 10 times the bandwidth of the intelligence (4 kHz times 30 equals 120 kHz), which severely restricts the use of this type of multiplexing.

Research has shown that the bandwidth requirements in time-division systems can be reduced by the use of pulse-coding techniques and the filtering of the transmitted signal. The bandwidth can be reduced to a point that approaches the point required for the intelligence alone. However, the equipment necessary to accomplish this is very complicated, and some of the simplicity of time-division multiplexing is lost.

Synchronization. It is apparent from the basic diagram in figure 2–16,A, and B, that synchronization is critical in time-division multiplexing and a means of maintaining synchronization must be provided. A common method of doing this is transmitting pulses—called *sync* or *marker pulses*—at the beginning of each frame. A frame is the interval occupied by one complete set of pulses and is made up of a marker pulse and one pulse from each channel. In figure 2–16,A, and B, the marker would occur after the channel 4 sample. In a 30-channel system, the marker would appear as the next pulse following the channel 30 pulse.

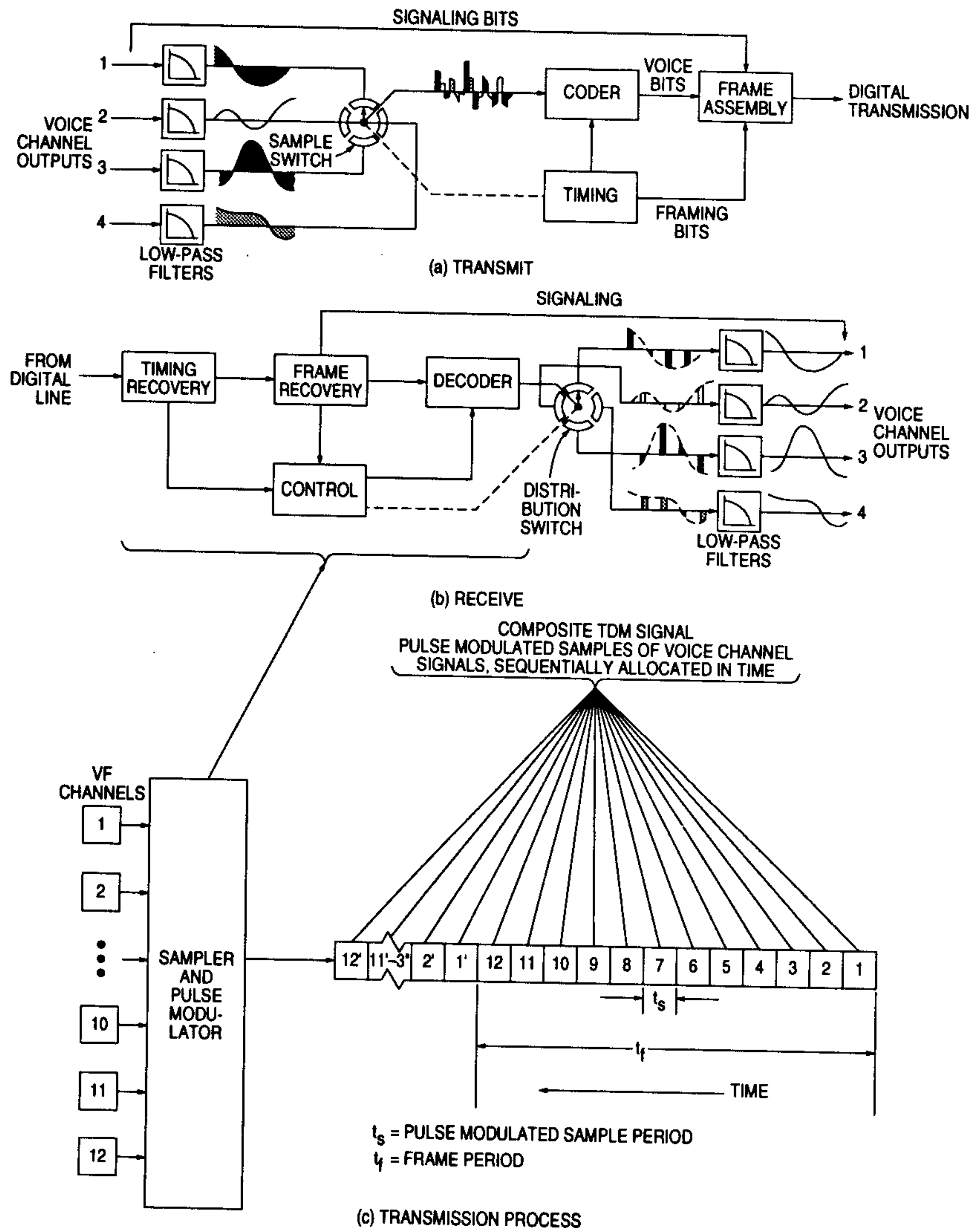

Figure 2–16. Time-division multiplexing process.
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Solving problems concerning frame period or sample rate of TDM subsystems

TDM transmission is based on a serial-type data stream. The telephone or analog signal must be converted to a data bit stream and placed into a time slot, like a column of ducks. Data-type input signals are converted into some type of pulse code and placed into a time slot, called a *frame.* In the following paragraphs, we explain how this is done.

Voice multiplex. In telephone TDM subsystems, all VF channels to be multiplexed are *divided* in the time domain. The information contained in each channel within the composite TDM signal is transmitted during a different instant of time, but overlaps in a common frequency spectrum. Figure 2–16,C, illustrates the basic TDM transmission process for a 12-channel telephone system.

Figure 2–16 illustrates that each voice channel occupies a 4-kHz bandwidth. A sampler must scan and sample each of the voice channels at twice this figure; i.e., at 8 kHz. (Sample rate = 2 times the highest frequency). One complete set of samples (in this case, 12 samples) is generally called a frame. The time it takes to sample all 12 channels is called the *frame period.* In our example, the frame period, t_f, is equal to 125 microseconds:

$$t_f = \frac{1}{\text{sample rate}} = \frac{1}{8\ \text{kHz}} = \frac{1}{8x10^3}\ \text{seconds} = 125\ \text{microseconds}$$

Each sample within a frame period is pulse modulated before it appears sequentially on the composite TDM signal channel. For this example, the time available to represent the sample in a pulse-modulated form is equal to one-twelfth of the frame period. Therefore, the pulse-modulated sample period, t_S, is equal to approximately 10.4 microseconds.

$$\begin{aligned} t_s &= \frac{t_f}{\text{no. channels}} \\ &= \frac{t_f}{12} \\ &= \frac{125\ \text{microseconds}}{12} \\ &= 10.4\ \text{microseconds} \end{aligned}$$

The pulse-modulated sample period can be broken down into smaller sub-periods, depending upon the type of pulse modulation used in the TDM subsystem.

At the receiving end, the reverse process takes place. Each pulse-modulated sample is demodulated to obtain the original voice channel sample. The voice channel samples are then filtered and sequentially applied to the corresponding voice channel to yield a restored voice channel.

Telegraph multiplex. The basic operating principle of TDM telegraph equipment is similar to that of TDM telephone equipment. DC signals from a number of telegraph loops are assembled (interwoven) in sequential order for transmission over a single circuit. Since the DC telegraph signals are already in pulse form, the sampling and pulse modulation required in TDM telephone equipment is not needed. The process is one of simply interweaving telegraph channels into a composite TDM signal that is compatible with the particular transmission subsystem.

In its simplest form, the operation of telegraph TDM equipment is similar to that of parallel-to-serial and serial-to-parallel converters. That is, when the equipment is sending, parallel inputs are converted to a single serial output. Conversely, when the equipment is receiving, a single serial input is converted to a number of parallel outputs.

The modulation rate of the single serial stream depends on the number of associated telegraph loops and the modulation rate on the loops. For example, if 16 unit interval signals of 75 baud are time-division multiplexed, the modulation rate of the serial stream would be 16 x 75 = 1200 baud. Typical telegraph TDM equipment used in the DCS can handle up to sixteen 60-, 75-, or 100-word-per-minute DC telegraph loops.

System characteristics of FDM and TDM

A brief comparison of FDM and TDM is presented in the following text. We are making the comparison with regard to their primary communication requirements concerning bandwidth and noise.

Bandwidth requirements. In single-sideband FDM subsystems, the bandwidth of the composite FDM signal is equal to the number of channels times the bandwidth of each single channel. For example, in a 12-channel system with a 4-kHz bandwidth per channel, the composite signal has a bandwidth of 48 kHz. In a double-sideband FDM system, this bandwidth is doubled; i.e., 96 kHz. These bandwidths are relatively narrow when they are compared with those required in practical TDM subsystems. In a 64-step PCM/TDM system handling twelve 4-kHz channels, the composite TDM signal would have frequency components exceeding 576 kHz, or 48 kHz per channel. This is 12 times greater than the bandwidth required for single-sideband FDM and 6 times greater than the bandwidth required for double-sideband FDM.

If the available transmission subsystem bandwidth is restricted because of technical or economic reasons, FDM is the proper choice. This would be true where the available bandwidth of the transmission subsystem is derived at substantial cost, such as the present-day tropospheric scatter radio and submarine cable systems. If the available bandwidth is not restricted, TDM may be a better choice, since it provides better performance with regard to overall circuit noise and is easier to encrypt.

Noise. From a performance standpoint, the noise in FDM systems increases as the system's length is increased. Noise is cumulative because as it is introduced into the system, it adds to and is amplified with the signal at all repeater stations and terminals. As the system's length is increased, more repeaters and terminals are required, resulting in high overall circuit noise between the originating and terminating user terminals.

In TDM systems, the elements (bits) in the composite TDM signal are usually regenerated at each repeater station or terminal. As we use it here, "regeneration" refers to the process of generating a "clean" pulse upon the receipt of a "noisy" pulse. Noise in TDM systems stays relatively constant between terminals regardless of distance. This is true as long as the noise in the transmission subsystem links does not exceed the threshold of recognition for a proper regeneration of the TDM pulse.