1

Loop Interfaces

The subscriber loop and the equipment inexorably connected to it at each end are functionally defined in this chapter. The most familiar terminal equipment — the ubiquitous telephone instrument — is considered in some detail. Other types of terminal equipment, such as private branch exchanges (PBXs) and key telephone systems, are also considered. Loops used in switched service account for perhaps 90 percent of all loop applications. The next chapter establishes the signaling criteria associated with these loops.

1.1 Loop Function

What is a loop and what does it do? The loop is a transmission and signaling channel or path between a telephone subscriber's terminal equipment and the serving central office or another piece of terminal equipment. "Transmission" implies transfer of information while "signaling" implies the actions required to control the transmissions. Terminal equipment includes:

- Telephone instruments
- Facsimile machines
- Private branch exchanges
- Key telephone systems
- Voice mail systems
- Modems

- Computers
- Alarm systems
- Radio control systems
- Telephone answering machines

as well as many other devices. These are generally referred to as *customer premises equipment* (CPE), which can be defined as any equipment connected by premises wiring to the subscriber side of the *public switched telephone network* (PSTN) interface. Such devices interface voice as well as data signals to the subscriber loop.

The *network interface device* (NID), which serves as the PSTN interface, is described in Chapter 4. The subscriber (that is, telephone customer) can be an individual with one loop or a very large business with thousands of loops.

Put in a more rigorous way, the loop:

- Normally carries power from the central office (switching system or other network equipment) to the terminal equipment
- Transmits bidirectional speech or data signals between the calling and called parties
- Carries bidirectional control signals (supervision) between the calling and called parties and the central office
- Carries call progress tones to the calling party (dial tone, busy tone, ringback tone)
- Carries ringing current to the called party

The foregoing lists the normal signal voltages and currents carried on the loop. It also carries voltages and currents due to maintenance activities as well as those due to the environment, such as induced voltages from nearby power lines, lightning, and other phenomenon. These are discussed in detail in later chapters.

Most loops, shown pictorially in Fig. 1-1, simply consist of a telephone line or outside plant cable pair. Some applications may require a much more complex set of systems, including the cable pair, to make up the overall loop. For example, a foreign exchange circuit, where dial tone from an exchange in one city is provided to a subscriber in another exchange in another city, may include a cable pair, carrier equipment and optical fiber and satellite transmission systems as shown in Fig. 1-2. The loops connected to the terminal equipment at each end are called "tail circuits" or "end links."

The loop can be considered the exchange access portion of an overall telecommunication channel or circuit. This is illustrated in Fig. 1-3 where the exchange access is shown connecting subscriber terminal equipment to a central office. When the subscriber loop is connected to a central office switching system, it is called an access line. Between central offices is the inter-exchange access portion.

Although this book emphasizes the exchange access portion, it is important to realize the loop is designed, built and used to work well within any given exchange and also to work well outside of that exchange through the inter-exchange access portion. This is not necessarily an easy task, as will be seen.

As previously noted, the terminal equipment is connected at one end of the loop, and the central office (CO) is connected at the other end. Of course, the loop does not have to be connected to a central office line circuit and switched as implied here; it can be hard wired to another loop to provide a point-to-point transmission path, it may be bridged to

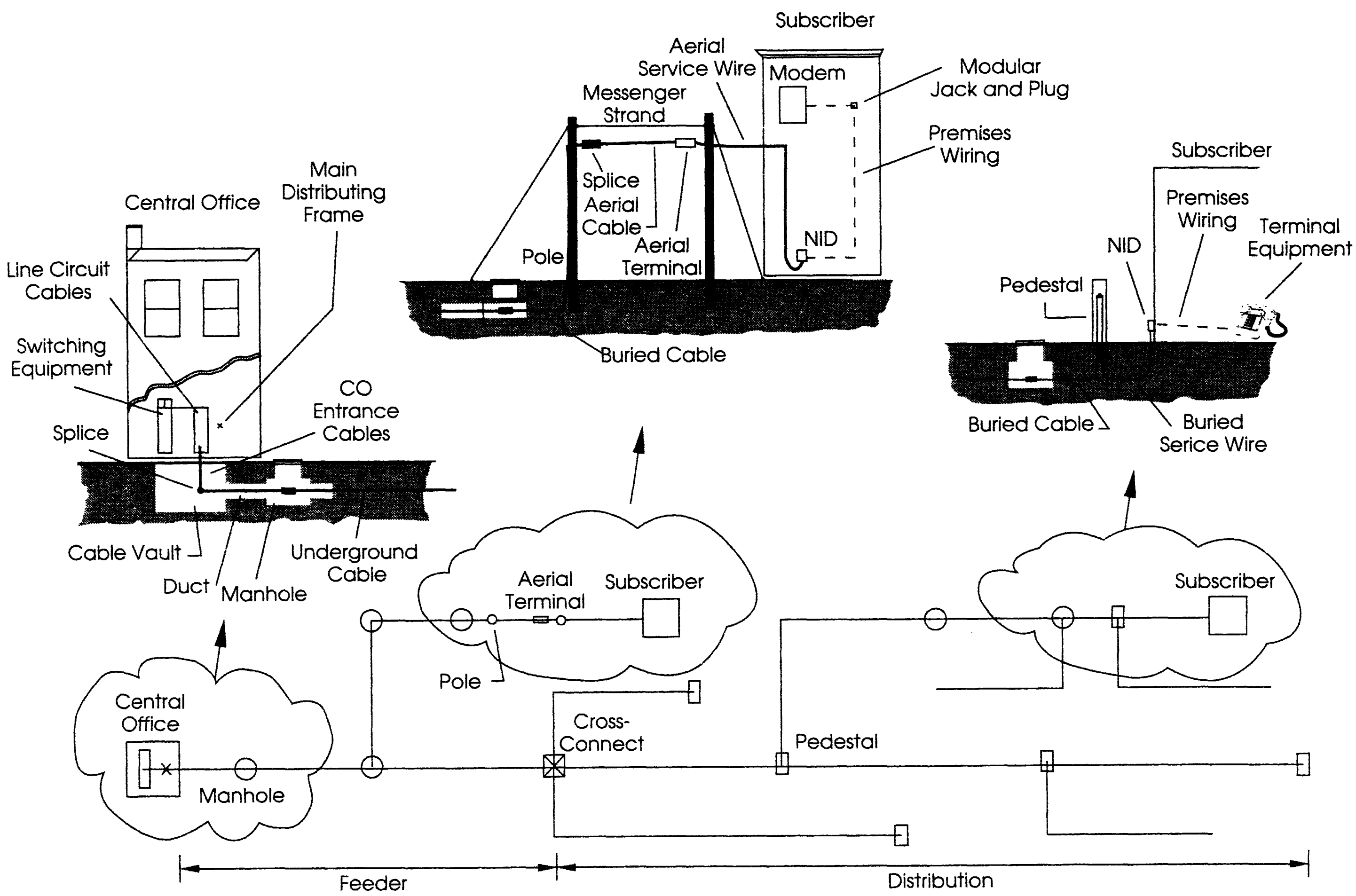


Figure 1-1 Subscriber's Loop

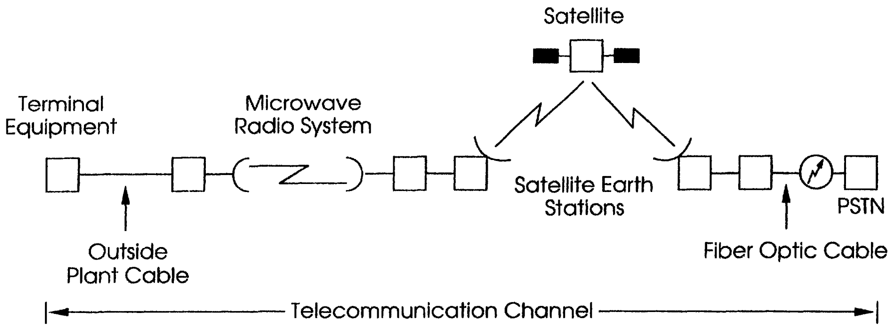


Figure 1-2 A more complex loop

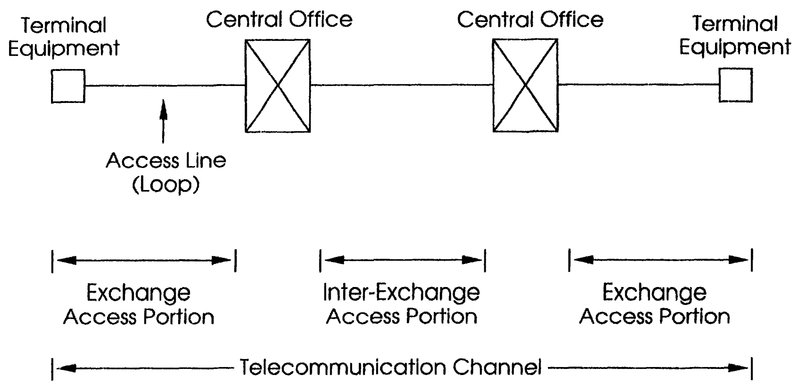


Figure 1-3 Exchange access

other loops (generally through special transmission equipment) to provide a point-to-multipoint transmission path, or it may terminate in some special equipment, such as a computer or specialized signaling system.*

For the very large proportion of cases, the loop and the interface connections are as shown in Fig. 1-4. Each part will be discussed in detail in this and later chapters.

1.2 The Central Office Line Circuit

For purposes of discussion in this section, the central office is considered in its most common way — that of terminating loops on a line circuit and providing a switching function. The central office is a node where loops converge as shown in Fig. 1-5. In this context, the central office is part of a wire center.† A wire center can contain more than one central office switching system.

* The term "bridged" is telephone terminology to indicate a parallel electrical connection.

† A wire center can be said to serve an exchange area, which is a geographical area determined either by the dialing prefix or political boundary.

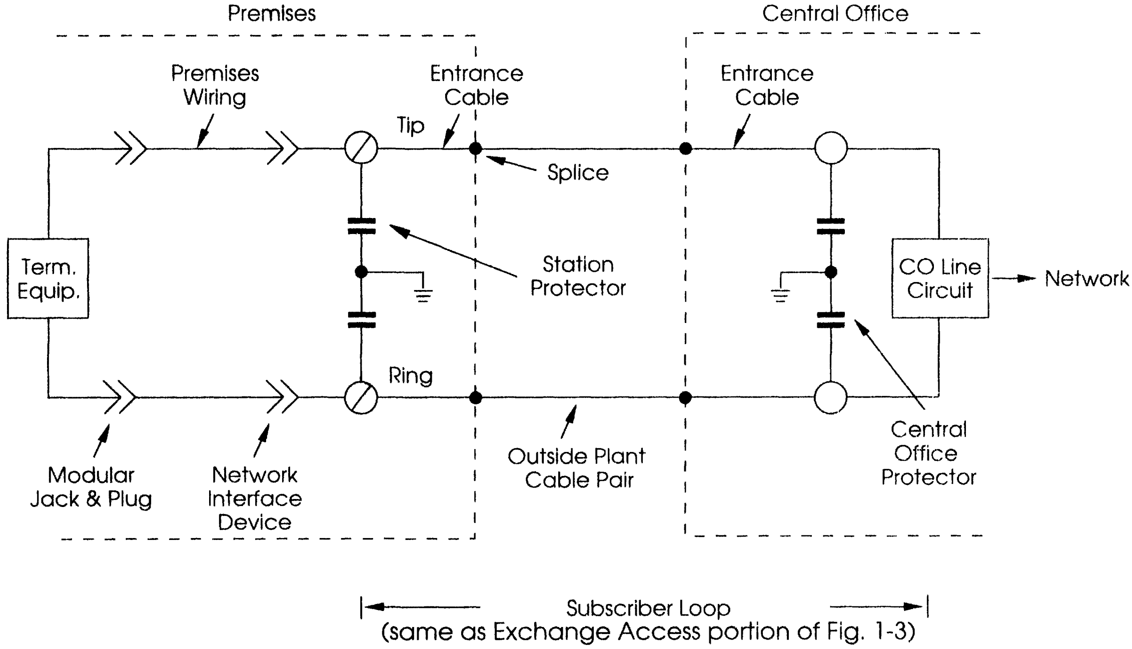


Figure 1-4 Loop and interface connections

The line circuit can be considered a physical part of the subscriber loop when the loop is an access line. The line circuit is the loop's gateway to the PSTN, and it is always located at the end of the loop opposite the terminal equipment. The line circuit:

- Terminates the loop with a balanced 900 Ω (or otherwise standard) termination
- Supplies direct current to the loop from central office battery and ground through balanced feed resistances having a total resistance around 400 Ω
- Supervises the loop currents to determine on-hook and off-hook conditions
- Receives dialed address information from the terminal equipment
- Isolates foreign potentials on the loop from the central office transmission and signaling circuitry.*
- Applies ringing voltage to the loop as well as other call progress tones
- Converts the two-wire loop transmission path to a four-wire (or equivalent) path used internally by the switching system (if the switching system is digital or four-wire — most analog central office switching systems are two-wire and do not make this conversion)
- Provides analog-to-digital and digital-to-analog conversion of the transmission signals (digital central offices only)
- Provides test access for loop and line circuit testing

* A foreign potential is any undesired voltage on a circuit.

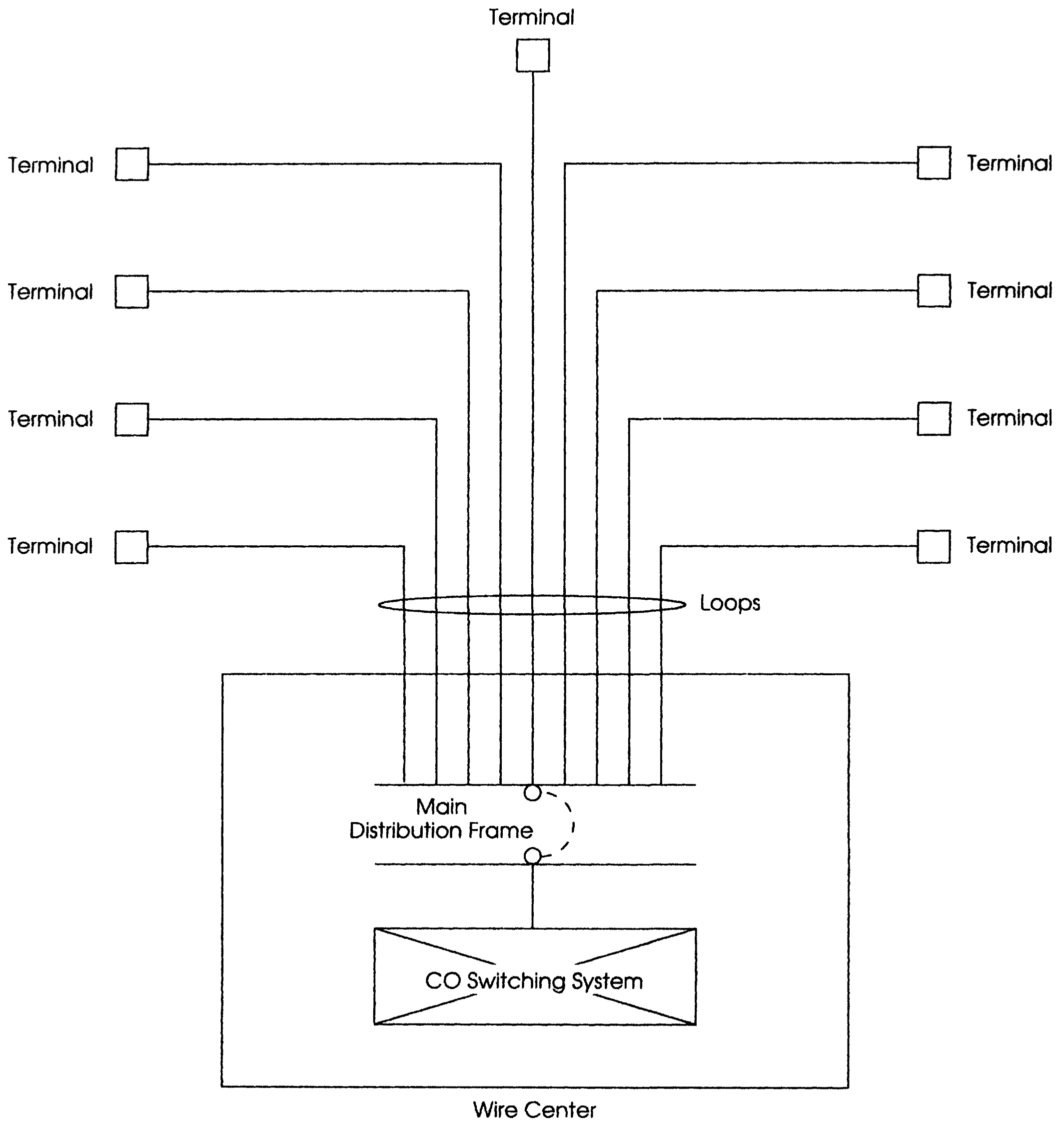


Figure 1-5 Loops converge at a central office

The central office line circuit, as is the subscriber loop, is inherently bidirectional. Even so, the line can be arranged to work in one direction only, if necessary. This means it can be arranged to only terminate calls or only originate calls, but not both. The line circuit functions are colloquially summarized with the acronym BORSCHT (Battery, Overvoltage protection, Ringing, Signaling, Coding, Hybrid, and Test).

1.3 The Telephone Instrument

This and later sections discuss the functions and characteristics of a typical telephone instrument, but not all transmission characteristics are included. Later chapters discuss the transmission requirements under which all terminal equipment must operate.

The largest proportion of loops are permanently associated with the telephone instrument. The transmission properties of both must be designed such that the combination will meet telephone industry objectives. These objectives are, in turn, set to provide satisfactory telephone service to the subscriber.

At one end of the typical loop is the most familiar type of terminal equipment: the telephone instrument. It is the most common means by which the subscriber transmits to or receives signals from the central office. Figure 1-6 shows a simplified schematic of a typical instrument with a tone dial. Telephone instruments are sometimes generically referred to as "500 Sets".*

Virtually all new terminal equipment connected to the PSTN since the early 1980s must be registered with the Federal Communications Commission under Part 68 — "Connection of Terminal Equipment to the Telephone Network" — of its Rules and Regulations [2]. These rules "provide for uniform standards for the protection of the telephone network from harms caused by the connection of terminal equipment and associated wiring thereto . . . ."†

Part 68 does not guarantee the terminal equipment will function adequately, only that it will not harm the network. The difference is worth noting. Some of the regulatory aspects of loop interface and signaling are covered at the end of this chapter.

The basic signals from a telephone instrument may be classified as:

- Forward (from the calling party), consisting of:
  Off-hook (seizure)
  Dialing, and
  On-hook (or clear forward)

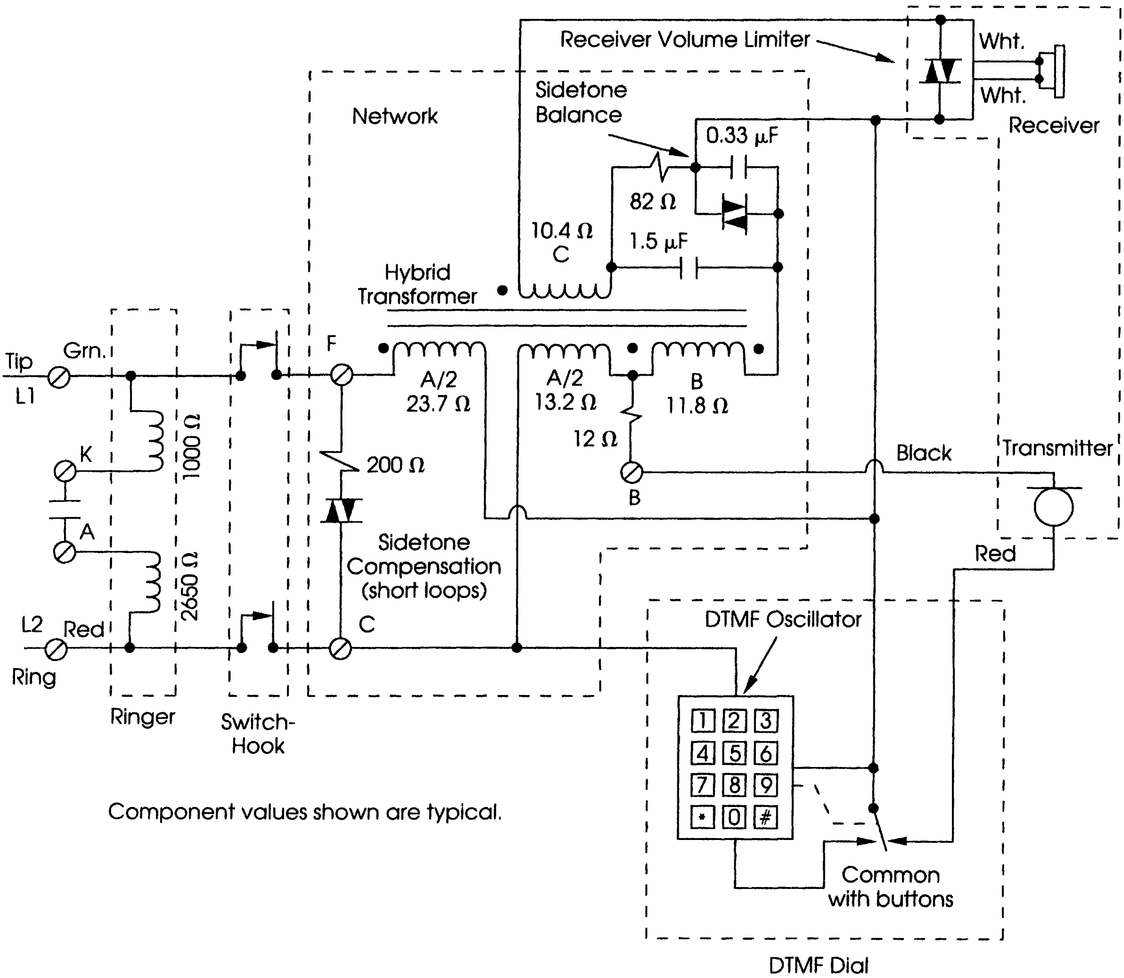
and,

- Backward (from the called party), consisting of:
  Off-hook (answer) and
  On-hook (or clear backward)

Dialing may be in the form of dial pulses (sequential on-hook/off-hook) or tones. The central office interprets the forward and backward signals through the switching system line circuit to appropriately set up or drop a connection.

* The 500 set was first produced in 1950 by the Western Electric Company [1]. Initially, it was only produced with a rotary dial but was later produced with a tone dial. The set with a tone dial is generally called a "2500 set."

† Par. 68.1 of FCC Part 68.



Component values shown are typical.

Figure 1-6 Typical telephone instrument schematic

The telephone instrument itself consists of a:

- Dial
- Network
- Transmitter
- Receiver
- Switch-hook
- Ringer*

* The word "ringer" is used to denote any electro-acoustical device to alert the called party. It does not have to be one with metallic gongs.

The switch-hook provides on-hook supervision by opening the loop when the handset is hung up. Conversely, it provides off-hook supervision by closing the loop when the handset is lifted. The transmitter converts voice or other forms of acoustical energy to a varying electrical current while the receiver converts varying electrical signals back to acoustical energy.

The network consists of several components as shown in Fig. 1-6. Its main function is to couple the transmitter and receiver to the loop and supply sidetone to the receiver.* In electromechanical instruments, it is a hybrid transformer and a balance network that converts the two-wire loop to a four-wire transmit and receive circuit. Many modern telephone instruments use an electronic equivalent of the transformer. The network serves other functions, such as limiting the signal level at the receiver and adjusting the sidetone level to an almost constant and acceptable level while working with a wide variety of loops.

The receiver level is limited to protect the listener's ears during high-level signals or noise impulses. Adequate sidetone is necessary so the talker does not shout or talk too softly because of low or high sidetone. If there is too much sidetone coupling, background noise entering the transmitter will interfere with weak signals at the receiver.

The network and transmitter combination is considered to have about 200 Ω dc resistance in an electromechanical type rotary dial instrument, but this can be quite variable. The actual dc resistance of any instrument is equal to the voltage across the instrument divided by the current through it. The relationship is nonlinear because of the nonlinear elements used in the network.

Instruments (including tone dial types) can have an off-hook resistance up to 400 Ω if they are designed according to Electronic Industries Association Standard EIA-470 [3]. According to the EIA, the dc resistance characteristic of an instrument may vary with several conditions. These are:

1. DC resistance must be less than 300 Ω during the on-hook to off-hook transition, during the make interval of rotary dial pulsing (described later), and for at least 1 s after answer,
2. DC resistance may increase to 400 Ω during *dual-tone multifrequency* (DTMF) signaling (described later), after called party answer, and after the talking connection has been made for at least 1 s,
3. DC resistance may be greater than 400 Ω at loop currents less than 20 mA (adequate operation of an instrument is not specified below 20 mA loop current).

The instrument's off-hook resistance must meet certain minimum requirements, too, to ensure proper operation of instruments intended for parallel use (for example, extensions). This is to prevent "current hogging" by an instrument with a relatively smaller resistance. For these instruments,

1. DC resistance must be greater than 100 Ω from 0 to 30 mA loop current.
2. DC resistance can linearly decrease to 80 Ω between 30 and 50 mA loop current.

* Sidetone is the acoustical signal resulting from a portion of the transmitted signal being coupled back to the receiver of the same handset. A sidetone circuit is sometimes called a "talkback" circuit.

3. DC resistance can linearly decrease from 80 Ω at 50 mA with a negative slope such that the resistance at 120 mA is greater than 33 Ω.

The above requirements are summarized in terms of a voltage-current relationship in Fig. 1-7. Also shown in this figure are the measured voltage-current curves for three typical instruments. One has an electronic network, and two have a conventional transformer-type network. In certain regions, the instruments are indistinguishable.

The ringer, of course, rings when the called party is to receive a call from the calling party. The ringing may be a warbling tone, tweet, bell sound, or it might be a flashing light. The ringer shown in Fig. 1-6 is connected as a bridged ringer. It is not always connected this way. Other wiring variations, such as connection from tip or ring to ground, which is called "divided ringing," are commonly used.* The actual connection depends on the telephone company's policy, its engineering philosophy or party-line requirements.† Ringing methods are discussed in detail later.

The rotary or tone dial provides address signals to the central office, where the address is that of the called party. Additionally, the push-button tone dial may provide control signals to more sophisticated switching systems or terminal equipment at the other end of the connection. For example, most large stock brokerage firms have toll-free numbers their customers can call to get stock or mutual fund information. Once connected, the customer uses pushbutton tones to control the call's routing to a recording that describes the fund or stock of particular interest. Similarly, many companies use "automated attendants" that instruct the caller, through prerecorded announcements, to dial the wanted department or "hold the line" for a live operator.

The typical electromechanical "500" set is not polarity sensitive. That is, it can be connected to the tip and ring leads without concern for possible tip-ring reversal. Tip-ring reversal can be caused by mistakes during outside splicing operations or premises wiring installations. This condition is easily detected by testing the polarity of the loop at the telephone set connections; the tip lead should be positive with respect to the ring lead when the tip and ring leads have the proper polarity.

Telephone sets that use a DTMF dial or have other line powered functions (such as an electronic telephone set) require proper polarity to function. To preclude possible malfunction on initial installation, or at some later time, the set is equipped with a polarity guard, which is nothing more than a common diode bridge. The polarity guard, shown in Fig. 1-8, allows instruments equipped with them to be connected to the network without regard for tip-ring polarity.

1.4 Private Branch Exchanges and Key Telephone Systems

Private branch exchanges (PBXs) and key telephone systems (key systems) provide additional telecommunication services at the customer premises that normally would

* Tip and ring are the two wires of a twisted pair or other telecommunication circuit. They are labeled as such to indicate their sense or polarity. The terms are historical.

† A party line is a given loop or central office line circuit that serves more than one subscriber.

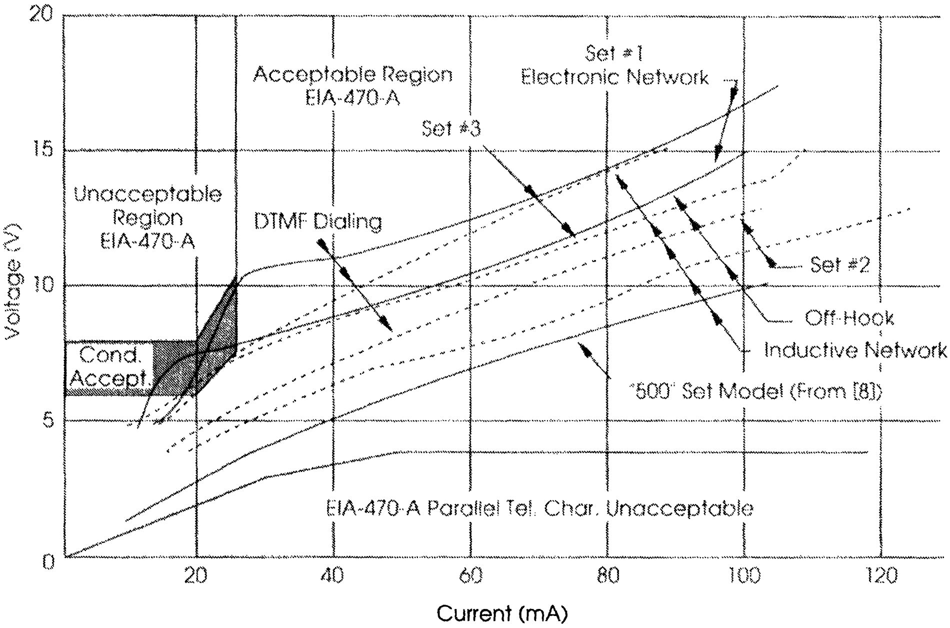


Figure 1-7 Voltage-current relationships for telephone instruments

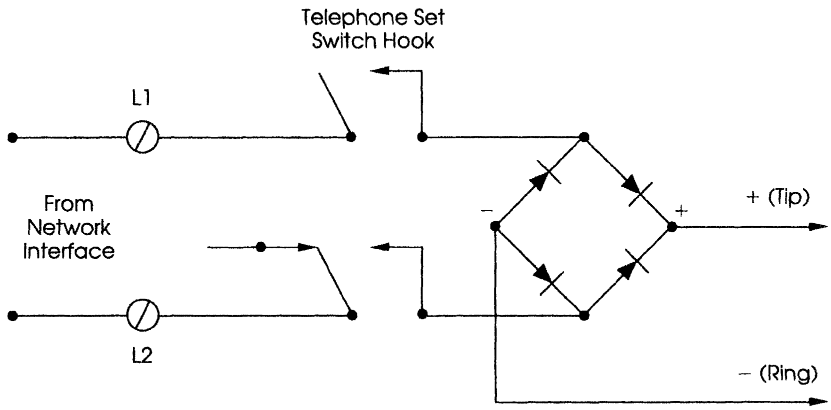


Figure 1-8 Telephone instrument polarity guard

not be available from the single-line telephone instrument previously described. The differences between PBXs and key telephone system applications at one time were quite apparent. The PBX was used in applications requiring approximately 25 or more stations, and key systems were used in smaller applications. The PBX provided a clear switching function requiring station addresses. Also, the PBX concentrated switched traffic into trunk groups.* A key system provided neither of these features.

* A trunk is a shared communication channel between switching systems.

The differences, however, have become blurred due to technological advances and associated economies in the design of each system. Now, a PBX may find an application in a smaller environment, and key systems are available for larger applications. In these situations, the two may have indistinguishable features. Figure 1-9 shows how PBXs and key systems fit into the overall network.

At this point, it should be noted that a group of subscriber loops are used to provide service from the central office switching system to the PBX. In this context, the loops are called PBX trunks or direct outward dialing (DOD) trunks. DOD trunks can be used for two-way service by providing central office dial tone and direct access to the PSTN by the user on outgoing calls and ringing to an operator or attendant on incoming calls.

Another application of a group of loops is for direct inward dialing (DID) trunks. Here, the trunks are used for one-way (incoming) service only by allowing a caller in the PSTN to directly dial a particular PBX extension without operator or attendant intervention.

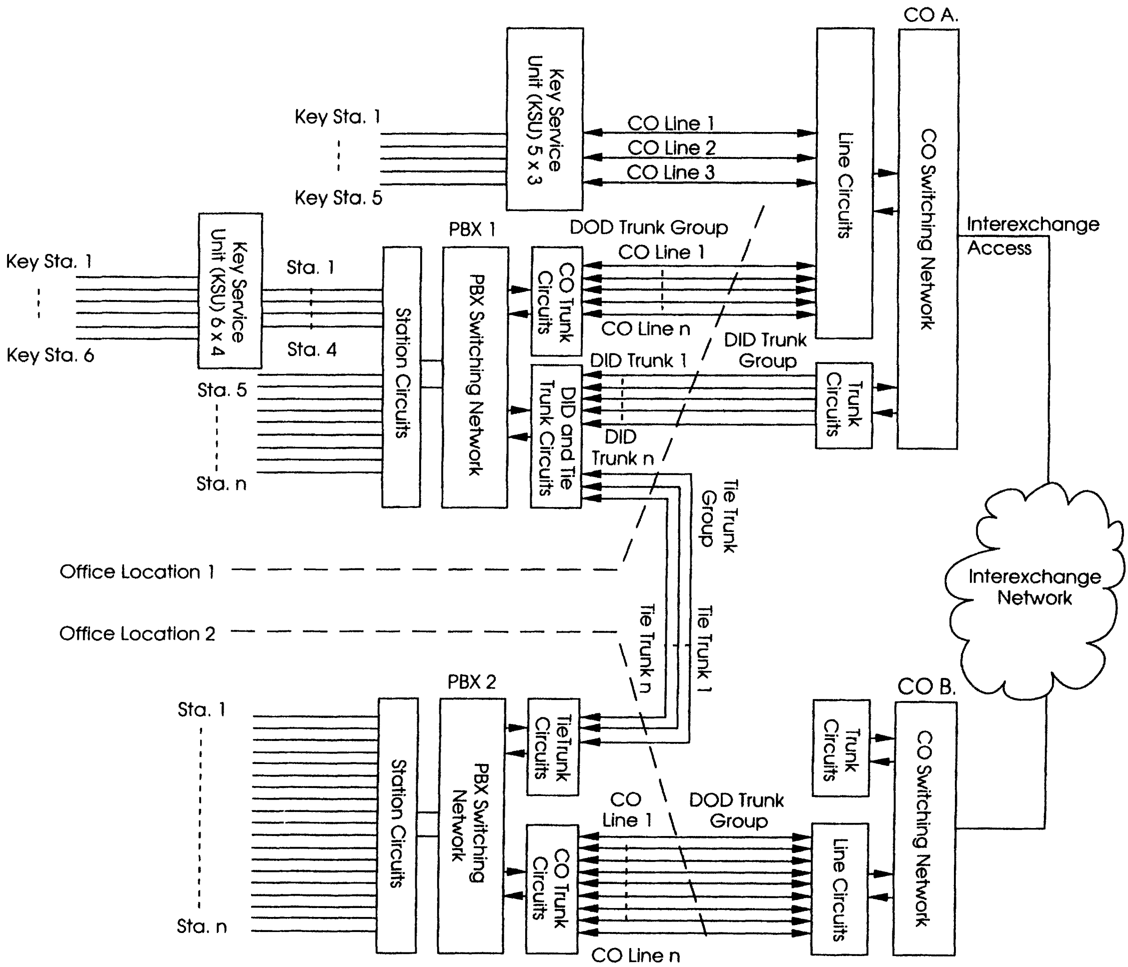


Figure 1-9 PBX and key telephone system network connections

It should be apparent from the previous discussion that a trunk is a common channel or line that is shared by a number of end users. This means trunks are a number of similar lines grouped together from a common functional standpoint. An end user may use any of the lines on any particular call, and the end user has no direct control over which line in the group is used on that call.

The term trunk is also used outside of the context of subscriber loops. In this case, a trunk interconnects switching systems (central offices). Even though the interoffice trunks may use the same facilities as loops, the signaling and transmission requirements for interoffice trunks vary considerably from that used with subscriber loops.

Although the distinction between PBXs and key systems has become blurred in larger systems, key systems can still be categorized as such for most common applications. The key system provides [4]:

- Appearances for multiple directory numbers on a single instrument
- Appearance of a single directory number on multiple instruments
- Status indication on each instrument of each directory number appearing on that instrument
- Ability to bridge multiple instruments onto a busy line
- Busy line hold capability
- Intercom capability

The modern PBX essentially provides the functions of a key system on a single instrument, but it also provides a switching function through [5]:

- Interconnection of instruments at two different locations
- Interconnection of instruments and an attendant (operator) position
- Interconnection of instruments or an attendant position with the PSTN via PBX trunks
- Interconnection of instruments or an attendant position with other PBXs and private facilities via PBX tie trunks

In addition to the blurred distinction between some PBXs and key systems, there may be little difference between PBXs and central office switching systems. Indeed, several PBXs and central office switching systems have been adapted quite successfully to the other role. The major differences have always been, however, the protocol and numbering plans used for on-network and off-network access. Also, the central office line circuit is designed to work in a much more severe environment than the PBX station circuit. The required electrical characteristics of key systems and PBXs are specified in Electronic Industries Association publications [6,7].

1.5 Tariff Aspects

The services and features available on the loop from the PSTN depend on tariffs established by state public service commissions, marketing methods, as well as the technical capabilities of the central office switching systems. The three service categories are illustrated in Fig. 1-10.

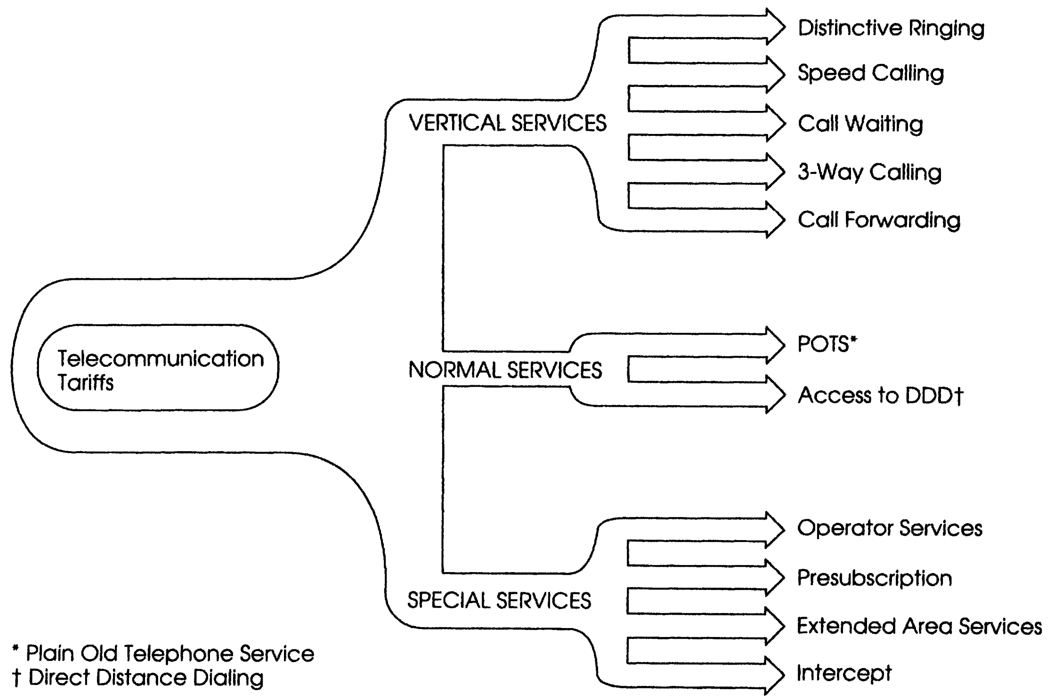


Figure 1-10 Service categories

The normal services are those available upon application for regular telephone service — popularly called "POTS" for "plain old telephone service." These are "basic" services meaning the ability to use the PSTN without significant restriction or engineering but with no added features. Vertical services, on the other hand, are enhanced capabilities that are marketable within the regulatory environment. These usually make the PSTN easier to use and access and provide additional conveniences, such as the ability to conference more than two callers together.

Special handling are those services requiring operator assistance, access to alternate inter-exchange carriers or any other call handling outside the scope of POTS.

1.6 Federal Regulatory Requirements

Part 68 of the FCC rules and regulations governs the connection of any terminal equipment to the telephone network, which consists of the PSTN and certain private lines. Knowledge of certain aspects of part 68 is necessary to the understanding of any such connection. The stated purpose of the FCC rules is to prevent harm to the network. To do this, the rules spell out in detail a number of tests the terminal equipment must pass before it can become certified as suitable for connection to the network. These tests cover external environmental conditions such as temperature, humidity, shock and vibration and voltage surges as well as the failure modes of the equipment under these conditions.

There are a number of other required tests concerning the voltages, currents, signal power levels and load applied by the terminal equipment to the loop (as opposed to the previously described conditions, which are applied to the terminal equipment).

Generally, the terminal equipment must be designed such that the open circuit (no load) voltage at any of its connection points does not exceed 70 V for more than 1 s unless the voltages are used for network control or signaling and are consistent with the telephone company's standards (for example, ringing). When the terminal equipment consists of a PBX, key system or similar system, it cannot apply more than 56.5 V dc or supply more than 140 mA of dc current to the loop for off-premises stations. Other voltage and current limitations are discussed in Chapter 3.

In addition to operational-type voltages mentioned in the previous paragraph, the FCC limits low-level signal type voltages, also. These limitations cover a surprisingly large bandwidth: from approximately 200 hertz (Hz) to 6 megahertz (MHz). All tests for signal power specify the bandwidth over which it must be measured, and all tests are made with loop simulator circuits, shown in Fig. 1-11, or a standard 600 Ω termination.

The loop simulator and standard termination are the only practical way to make such tests. As will be shown in Chapter 5, there is no such thing as a standard subscriber loop, and the simulator promotes standardized evaluation of equipment. It is necessary to limit the signal power over such a wide bandwidth because of multiplexing and signaling equipment used in the network. This equipment makes use of the spectrum up to around 6 MHz, and extraneous signals can disrupt it.

Generally, the signal power applied to the loop simulator by the terminal equipment in the band from 200 Hertz to 4 kiloHertz (kHz) must be limited to –9 dBm under all conditions when averaged over a period of 3 s.* This limitation does not apply to live voice signals. The level of voice signals is discussed in Chapter 5. Signals from certain data communication equipment (DCE), such as most modems, must not exceed –13 dBm over any 3 s interval.

From 4 kHz to 6 MHz, signal power is no longer specified in dBm. Instead, "metallic" voltage, which includes signal voltage plus any other voltages across tip and ring, is specified using a particular impedance at the loop simulator. All voltage levels are specified over an 8 kHz bandwidth.

The metallic (or differential) voltage applied by the terminal equipment to the termination tip and ring conductors must be no greater than 200 mV (–14 dBV) at 4 kHz, decreasing to no more than 100 mV (–20 dBV) at 12 kHz for more than 0.1 s.

The allowable voltage levels continue to decrease at a constant rate up to 90 kHz where the level must be no more than 1.78 mV (–55 dBV) over any 0.1 s interval, and so on as shown in Fig. 1-12. Other specified voltage limitations are shown in this figure, too. These concern longitudinal (common mode) voltages, which must be lower in magnitude than the metallic voltages by approximately 7–16 dBV.

There are certain narrow frequency bands that require special consideration when terminal equipment is being tested. These are 2,450 to 2,750 Hz and 3,995 to 4,005 Hz. The 2,450 to 2,750 Hz band provides signaling functions in the network. These are called

* See Appendix A for a discussion of the decibel (dB).

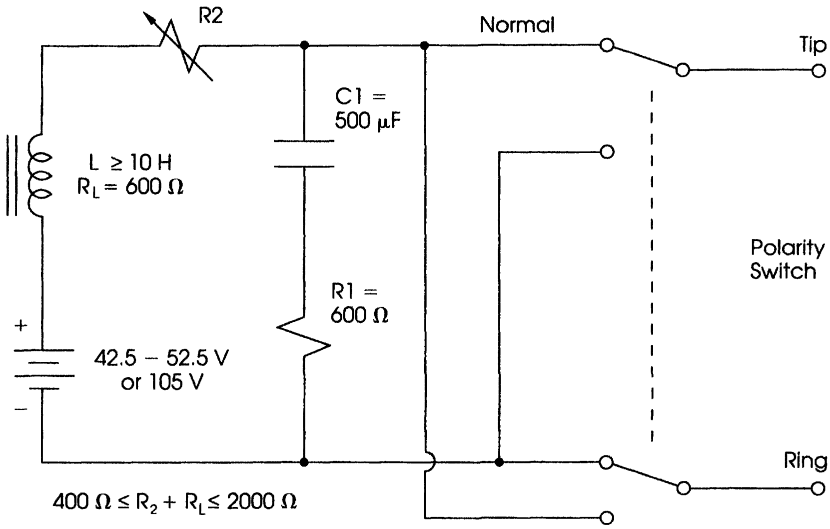
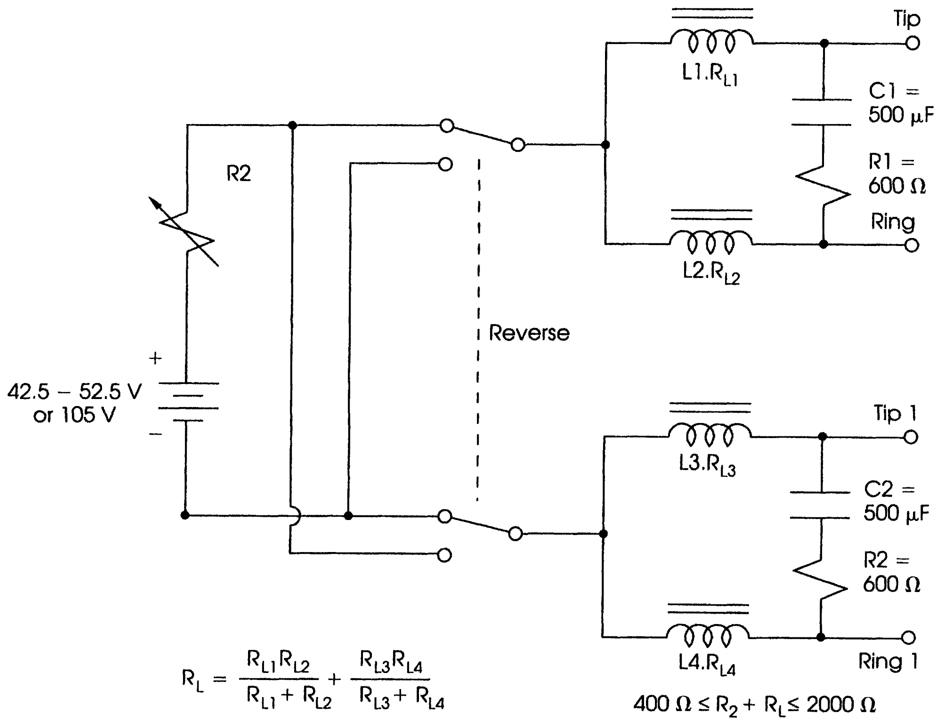


Figure 1-11a FCC two-wire loop simulator circuit



$$R_L = \frac{R_{L1}R_{L2}}{R_{L1}+R_{L2}} + \frac{R_{L3}R_{L4}}{R_{L3}+R_{L4}}$$

Figure 1-11b FCC four-wire loop simulator circuit

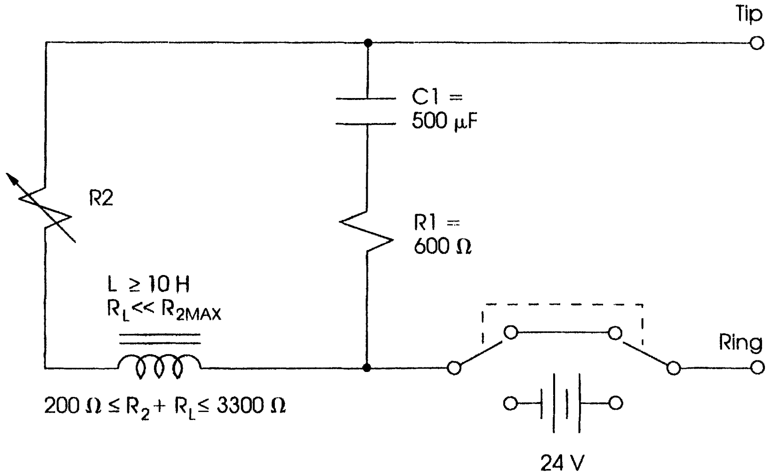


Figure 1-11c FCC Off-premises loop simulator circuit

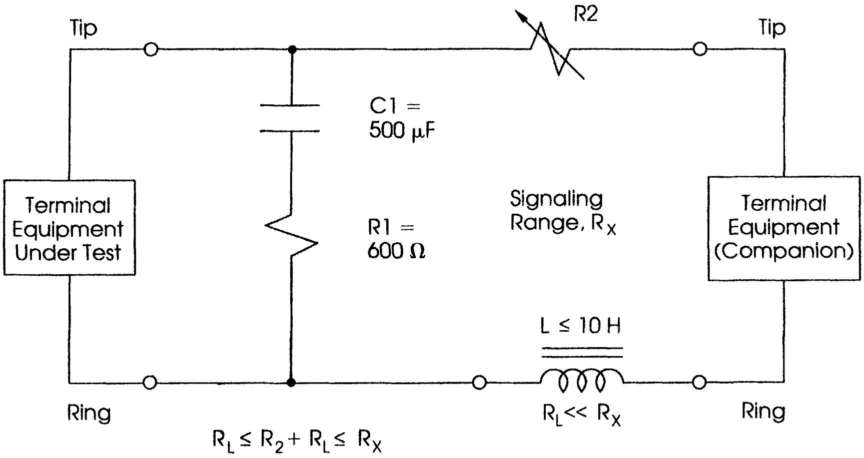


Figure 1-11d FCC private line loop simulator circuit

SF or single-frequency signaling and are used for address transmission and supervision as well as loopback.* Signal levels must be limited to –8 dBm in the SF signaling state and –20 dBm in the idle (on-hook) state. If the terminal equipment does not use SF signaling, the energy in this band must be accompanied by an equal amount of energy in the 800 to 2,450 Hz band. This allows network signaling equipment to differentiate between real and extraneous signals.

The band from 3,995 to 4,005 Hz is one-half the sampling rate for digital loop carrier systems (which is 8 kHz). Unwanted signals in this band can cause serious disruption of digital networks and must be at least 18 dB below the signals in the 200 Hz to 4 kHz band.

* SF and loopback usually apply only to four-wire private line circuits. Loopback is a test function that connects the transmit side of a circuit to its receive side at the far (or loopback) end.

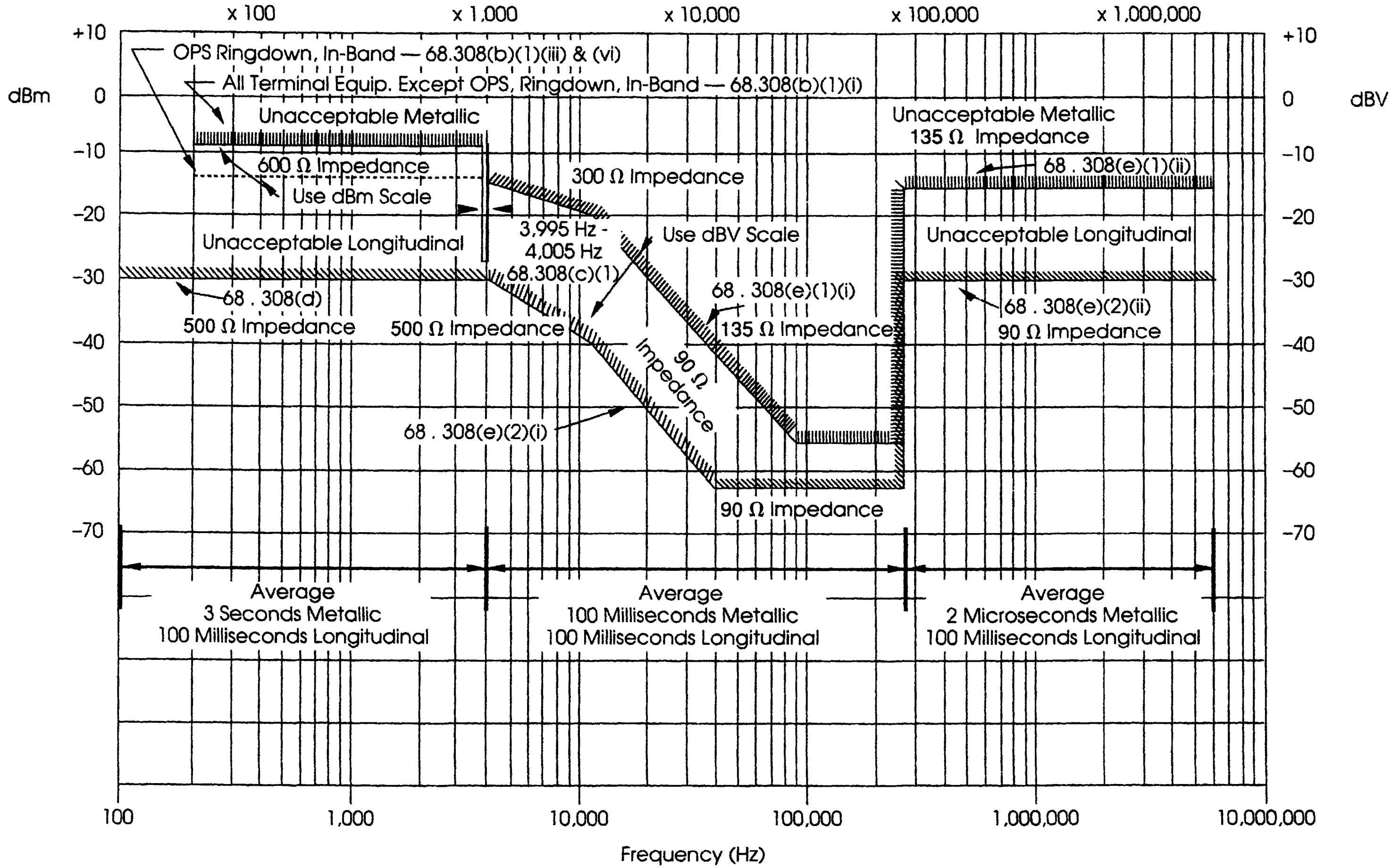


Figure 1-12 FCC signal power limitations, Par. 68.308

The load the terminal equipment places on the idle loop is determined by its *ringer equivalence number* (REN), which is defined by the FCC in Part 68 and described in detail in Chapter 2. A "standard" ringer has a ringer equivalence of 1.0. The maximum ringer equivalence that can be placed on a loop is 5.0. High ringer equivalence may prevent terminal equipment from functioning properly on long loops. The FCC defines the frequency characteristics of ringers by an alphabetical code.

Terminal equipment must provide at least some on-hook load to the loop. This allows the telephone company to test the loop for connected equipment. The minimum load is 40 kΩ or approximately 0.2 ringer equivalence.

The FCC specifies a minimum insulation breakdown voltage for premises wiring of 1,500 Vrms. Of equal importance is the requirement by the National Electrical Code, Article 800, which requires the premises wiring have a voltage rating of 300 V.

Until mid-1990, the actual connection of terminal equipment to the network was required by the FCC to be through standard plugs and standard telephone-company jacks. These are the familiar "modular" plugs and jacks, which are described in Chapter 4. The telecommunication industry interpretation of the previous FCC requirements was controversial and confusing.

In CC Docket No. 88-57 in orders adopted June 8, 1990 and August 10, 1990, the FCC modified, and in some respects clarified, the demarcation point requirements. The resulting changes to Part 68 relax the requirement for a plug and jack demarcation point. Now, a simple splice at least 12 inches from the premises wiring side of the telephone company protector is acceptable as a demarcation point. Jack and plug arrangements are still acceptable. With the FCC ruling, the subscriber has more interconnection device choices and presumably can make better use of premises wiring.

The matter of demarcation points is further confused by the fact that, in many states, the public service commissions (PSCs) have specified the demarcation point as the standard plug and jack arrangement that is integral to the telephone-company provided network interface device. Where the network interface device does not have a plug and jack or does not exist, the demarcation point has been considered by PSCs to be the telephone-company provided protector of some other jack located on the premises. The PSC requirements sometimes conflict with FCC requirements. In almost all cases, however, FCC requirements preempt state PSC requirements.

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