TELEPHONE INSTRUMENTS AND SIGNALS

The Subscriber Loop, Standard Telephone Set, Basic Telephone Call Procedures, Call Progress Tones and Signals, Cordless Telephones, Caller ID, Electronic Telephones, Paging systems.

Telecommunication is the transmission of messages, over significant distances, for the purpose of communication. Telecommunications has typically involved the use of electric means such as the telegraph, the telephone, and the teletype, the use of microwave communications, the use of fiber optics and their associated electronics, and/or the use of the Internet.

The Subscriber Loop

The simplest and most straight forward form of telephone service is called *plain old telephone services*(POTS), which involves subscribers accessing the public telephone network through pair of wires called the *local subscriber loop* (or simply *local loop*). A local loop is simply an unshielded twisted - pair transmission line (cable pair), consisting of two insulated conductors twisted together.



The subscriber loop provides the means to connect a telephone set at a subscriber location to the closest telephone office, which is commonly called an *end office, locale change office,* or *central office.* Once in the central office, the subscriber loop is connected to an *electronic switching system (ESS),* which enables the subscriber to access the telephone network.

Standard Telephone Set

A telephone is defined as "an apparatus for reproducing sound, especially that of the human voice (speech) at a great distance, by means of electricity, consisting of transmitting and receiving instruments connected by a line or wire which conveys the electric current".

Functions of the telephone set

1. Notify the subscriber when there is an incoming call with an audible signal, such as a bell, or with a visible signal, such as a flashing light. This signal is analogous to an interrupt signal on a microprocessor, as its intent is to interrupt what you are doing. These signals are purposely made annoying enough to make people want to answer the telephone as soon as possible.

- 2. Provide a signal to the telephone network verifying when the incoming call has been acknowledged and answered (i.e., the receiver is lifted off hook).
- 3. Convert speech (acoustical) energy to electrical energy in the transmitter and vice versa in the receiver. Actually, the microphone converts the acoustical energy to mechanical energy, which is then converted to electrical energy. The speaker performs the opposite conversions.
- 4. Incorporate some method of inputting and sending destination telephone numbers (either mechanically or electrically) from the telephone set to the central office switch over the local loop. This is accomplished using either rotary diallers (pulses) or Touch-Tone pads (frequency tones).
- 5. Regulate the amplitude of the speech signal the calling person outputs onto the telephone Line. This prevents speakers from producing signals high enough in amplitude to interfere with other people's conversations taking place on nearby cable pairs (crosstalk).
- 6. Incorporate some means of notifying the telephone office when a subscriber wishes to place an outgoing call (i.e., handset lifted off hook). Subscribers cannot dial out until they receive a dial tone from the switching machine.
- 7. Ensure that a small amount of the transmit signal is fed back to the speaker enabling talkers to hear themselves speaking. This feedback signal is sometimes called *side tone* or *talkback*. Side tone helps prevent the speaker from talking too loudly.
- 8. Provide an open circuit (idle condition) to the local loop when the telephone is not in use (i.e., on hook) and a closed circuit (busy condition) to the local loop when the telephone is in use (off hook).
- 9. Provide a means of transmitting and receiving call progress signals between the central office switch and the subscriber, such as on and off hook, busy, ringing, dial pulses. Touch-Tone signals, and dial tone.

Telephone Set, Local Loop and Central Office Switching machines

A basic telephone set requires only two wires and these pair of wires connecting a subscriber to the closest telephone office is called the local loop. One wire is called the **tip**, and the other is called the **ring**. In case, a third wire is used, it's called the sleeve. Since the 1960s, phone plugs and jacks have gradually been replaced in the home with miniaturized plastic plug known as RJ-11 and a matching plastic receptacle shown in figure.



RJ stands for *registered jacks* and is sometimes described as RJ-XX. RJ is a series of telephone connection interfaces (receptacle and plug) that are registered with the U.S. Federal Communications Commission (FCC). The term *jack* sometimes describes both the receptacle and the plug and sometimes specifies only the receptacle. RJ-11 is the most common telephone jack in use today and can have up to six conductors.

Block Diagram of a Telephone Set

A telephone set is an apparatus that creates an exact likeness of sound waves with an electric current. The essential components of a telephone set are the ringer circuit, on/off hook circuit, equalizer circuit, hybrid circuit, speaker, microphone, and a dialling circuit.



Ringer circuit: The ringer circuit, which was originally an electromagnetic bell, is placed directly across the tip and ring of the local loop and its sole purpose is to alert the destination party of incoming calls. The tone of the ringer should be loud enough to be heard from a distance. In modern telephones, the bell has been replaced with and electronic oscillator connected to the speaker. Today, ringing signals can be of any imaginary sound.

On/off hook circuit: The on/off hook circuit (sometimes called a *switch hook*) is nothing more than a simple single-throw double-pole (STDP) switch placed across the tip and ring. The switch is mechanically connected to the telephone handset so that when the telephone is idle (on hook), the switch is open. When the telephone is in use (off hook), the switch is closed, completing an electrical path through the microphone between the tip and ring of the local loop.

Equalizer circuit: Equalizers are combinations of passive components (resistors, capacitors, and so on) that are used to regulate the amplitude and frequency response of the voice signals. The equalizer helps solve an important transmission problem in telephone set design, namely, the interdependence of the transmitting and receiving efficiencies and the wide range of transmitter currents caused by a variety of local loop cables with different dc resistances.

Speaker: The speaker is the receiver for the telephone. The speaker converts electrical signals received from the local loop to acoustical signals (sound waves) that can be heard and understood by a human being. The speaker is connected to the local loop through the hybrid network. The speaker is typically enclosed in the *handset* of the telephone along with the microphone.

Microphone: The microphone is the transmitter for the telephone and it converts acoustical signals in the form of sound pressure waves from the caller to electrical signals that are transmitted into the telephone network through the local subscriber loop. The microphone is also connected to the local loop through the hybrid network. Both the microphone and the speaker are transducers, as they convert one form of energy into another form of energy. A microphone converts acoustical F energy first to mechanical energy and then to electrical energy, while the speaker performs the exact opposite sequence of conversions

Hybrid network: The hybrid network (sometimes called a hybrid coil or duplex coil) in a telephone set is a special balanced transformer used to convert a two-wire circuit (the local loop) into a four-wire circuit (the telephone set) and vice versa, thus enabling full-duplex operation over a two-wire circuit. In essence, the hybrid network separates the transmitted signals from the received signals. Another function of the hybrid network is to allow a small portion of the transmit signal to be returned to the receiver in the form of a *sidetone*.

Dialing circuit: The dialling circuit enables the subscriber to output signals representing digits, and this enables the caller to enter the destination telephone number. The dialling circuit could be a rotary dialer, or most likely an electronic dial-pulsing circuit (or a Touch-Tone keypad) which sends various combinations of tones representing the called digits.

Basic Telephone Call Procedures

When the calling party takes the telephone set off the hook, the switch hook in the set is released, completing a dc path between the tip and ring of the loop through the microphone. The ESS (electronic switching system) senses a dc current in the loop and recognizes this as an off-hook condition and this procedure is referred to as loop start operation. Completing a local telephone call between two subscribers connected to the same telephone switch is accomplished through a standard set of procedures including the following 10 steps. This manner of accessing the telephone system is known as POTS.

- Step 1:- Calling station goes off hook.
- Step 2:- After detecting a dc current flow on the loop, the switching machine returns an audible dial tone to the calling station, acknowledging that the caller has access to the switching machine.
- Step 3:- The caller dials the destination telephone number using one of two methods: mechanical dial pulsing or, more likely, electronic dual-tone multifrequency (Touch-Tone) signals.
- Step 4:-When the switching machine detects the first dialed number, it removes the dial tone from the loop.
- Step 5:-The switch interprets the telephone number and then locates the local loop for the destination telephone number.

- Step 6:- Before ringing the destination telephone, the switching machine tests the destination loop for dc current to see if it is idle (on hook) or in use (off hook). At the same time, the switching machine locates a signal path through the switch between the two local loops.
- Step 7a:- if the destination telephone is off hook, the switching machine sends a station busy signal back to the calling station.
- Step7b:- If the destination telephone is on hook, the switching machine sends a ringing signal to the destination telephone on the local loop and at the same time sends a ringback signal to the calling station to give the caller some assurance that something is happening.
- Step 8:- When the destination answers the telephone, it completes the loop, causing dc current to flow.
- Step 9:- The switch recognizes the dc current as the station answering the telephone. At this time, the switch removes the ringing and ring-back signals and completes the path through the switch, allowing the calling and called parties begin their conversation.
- Step 10:- When either end goes on hook, the switching machine detects an open circuit on that loop and then drops the connections through the switch.

Call Progress Tones and Signals

Call progress tones and *call progress signals* are acknowledgment and status signals that ensure the processes necessary to set up and terminate a telephone call are completed in an orderly and timely manner. Call progress tones and signals can be sent from machines to machines, machines to people, and people to machines.

Signalling can be broadly divided into two major categories: *stat ion signalling* and *interoffice signalling*. Station signalling is the exchange of signalling messages over local loops between stations (telephones) and telephone company switching machines. On the other hand, interoffice signalling is the exchange of signalling messages between switching machines. Signaling messages can be subdivided further into one of four categories: *alerting, supervising, controlling,* and *addressing.* Alerting signals indicate a request for service, such as going off hook or ringing the destination telephone. Supervising signals provide call status information, such as busy or ring-back signals. Controlling signals provide information in the form of announcements, such as number changed to another number, a number no longer in service, and so on. Addressing signals provide the routing information, such as calling and called numbers.

Examples of essential call progress signals are dial tone, dual-tone multifrequency tones, multifrequency tones, dial pulses, station busy, equipment busy, ringing, ring-back receiver on hook, and receiver off hook.

DCS Unit-5		Telephone Instruments and Signal The Telephone Circuit
Tone or Signal	Frequency	Duration /Range
Dial tone	350 Hz plus 440 Hz	Two of eight tones
DTMF	697 Hz, 770 Hz, 852 Hz, 941 Hz,	On. 50-ms minimum
	1209 Hz, 1330 Hz, 1477 Hz, 1033 Hz	Off. 45-ms minimum,
		3-s maximum
MF	700 H- 000 H- 1100 Hz	Two of six tones
	1200 Hz, 500 Hz, 1700 Hz	On, 90-ms minimum,
	1300 Hz, 1300 Hz, 1700 Hz	120-ms maximum
Dial pulses	Open/closed switch	On, 39 ms
		Off, 61 ms
Station busy	480 Hz plus 620 Hz	On, 0.5 s
		Off, 0.5 s
Equipment busy	480 Hz plus 620 Hz	On, 0.2 s
		Off, 0.3 s
Ringing	20 Hz, 90 vrms (nominal)	On, 2 s
	E da vitanigno staw ats	Off, 4 s
Ring-back	440 Hz plus 480 Hz	On, 2 s
		Off, 4 s
Receiver on hook	Open loop	Indefinite
Receiver off hook	dc current	20-mA minimum,
	the stand to the end of the second	So-mA maximum,
Receiver-left-off-	1440 Hz, 2060 Hz, 2450 Hz, 2600 Hz	06.015
hook alert		01, 0.1 \$

Call Progress Tone Summary

<u>**Dial Tone</u>**: Dial tone is an audible signal comprised of two frequencies: 350 Hz and 440 Hz. Dial tone informs subscribers that they have acquired access to the electronic switching machine and can now dial or use Touch-Tone in a destination telephone number. After the subscriber hears the dial tone and starts dialling, it is removed and this condition is called breaking dial tone. Sometimes, dial tone may not be heard even in off hook condition and this condition is called no dial tone.</u>

<u>**Dual-Tone Multifrequency (DTMF)</u></u>: DTMF was originally called** *Touch-Tone.* **DTMF is a more efficient means than dial pulsing for transferring telephone numbers from a subscriber's location to the central office switching machine. DTMF is a simple two-of-eight encoding scheme where each digit IS represented by the linear addition of two frequencies. DTMF is strictly for signaling between a Subscriber's location and the nearest telephone office or message switching center. The following figure shows the four-row-by-four column keypad matrix used with DTMF keypad.</u>**



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The keypad is comprised of 16 keys and eight frequencies. The four vertical frequencies (low group frequencies) are 697 Hz, 770 Hz, 852 Hz and 941 Hz, and the four horizontal frequencies (high group frequencies) are 1209 Hz, 1336 Hz, 1447 Hz and 1633 Hz. The digits 2 through 9 can also be used to represent 24 of the 26 letters. When a digit is pressed, two of the eight frequencies (one from either group) are transmitted. The major advantage of using Touch-Tone signalling over dial pulsing is speed and control. Here, all digits take the same length of time to produce and transmit and also it eliminates the impulse noise produced by mechanical switches used in dial pulses.

<u>Multifrequency</u> (<u>MF</u>): MF tones (codes) are similar to DTMF signals in that they involve the simultaneous transmission of two tones. MF tones are used to transfer digits and control signals between switching machines. MF tones are combinations of two frequencies that fall within the normal speech bandwidth so that they can be propagated over the same circuits as voice, which is called in-band signalling. The two-tone MF combinations and the digits or control information they represent is shown below.

Frequencies (Hz)	Digit or Control	
700 + 900	1	
700 + 1100	2	
700 + 1300	4	
700 + 1500	7	
900 + 1100	3	
900 + 1300	5	
900 + 1500	8	
1100 + 1300	6	
1100 + 1500	9	
1100 + 1700	Key pulse (KP)	
1300 + 1500	0	
1500 + 1700	Start (ST)	
2600 Hz	IDLE	
NA 11-6		

Multifrequency Codes

MF codes are used to transmit the calling and called numbers from the originating telephone office to the destination telephone office. MF tones involve transmission of two of the six possible frequencies representing the 10 digits plus two control signals. The key pulse (KP) signal is used to indicate the beginning of a sequence of dialled digits. The start (ST) signal is used to indicate the end of a sequence of dialled digits.

<u>D</u>íal <u>Pulses</u>: Dial pulsing (sometimes called *rotary dial Pulsing*) is the method originally used to transfer digits from a telephone set to the local switch.



The process begins when the telephone set is lifted off hook, completing a path for current through the local loop. When the switching machine detects the Off-hook Condition, it responds with dial tone. After hearing the dial tone, the subscriber begins dial pulsing digits by rotating a mechanical dialling mechanism and then letting it return to its rest position. As the rotary switch returns to its rest position, it outputs a series of dial pulses corresponding to the digit dialled. When a digit is dialled, the loop circuit alternately opens (breaks) and closes (makes) a prescribed number of times. The number of switch make/break sequences corresponds to the digit dialled (i.e., the digit 3 produces three switch openings and three switch closures). Dial pulses occur at 10 make/break cycles per second (i.e., a period of 100 ms per pulse cycle). All digits do not take the same length of time to dial.

<u>Station Busy</u>: A station-busy signal is sent from the switching machine back to the calling station whenever the called telephone number is off hook (i.e., the station is in use). The station-busy signal is a two-tone signal comprised of 480 Hz and 620 Hz. The two tones are on for 0.5 seconds, then off for 0.5 seconds. Thus, a busy signal repeats at a 60-pulse-per minute (ppm) rate.

<u>Equipment Busy</u>: The equipment-busy signal is sometimes called a *congestion tone* or a *no-circuits-available tone*. The equipment-busy signal is sent from the switching machine back to the calling station whenever the system cannot complete the call because of equipment unavailability. This condition is called blocking and occurs whenever the system is overloaded and more calls are being placed than can be completed. The equipment-busy signal uses the same two frequencies as the station-busy signal, signal except the equipment-busy signal is on for 0.2 seconds and off for 0.3 seconds (120 ppm).

<u>Ringing</u>: The ringing signal is sent from a central office to a subscriber whenever there is an incoming call and its main purpose is to alert the subscriber that there is an incoming call. The ringing signal is nominally a 20-Hz, 90-vrms signal that is on for 2 seconds and then off for 4 seconds.

<u>**Ring-Back</u>**: It is sent to the calling party at the same time the ringing signal is sent to the caller party. The purpose of the ring-back signal is to assure the calling party that the destination telephone number has been accepted and processed and is being rung. It is an audible combination of two tones at 440 Hz and 480 Hz that are on 2 seconds and then off for 4 seconds.</u>

<u>**Receiver On/Off Hook</u></u>:** When the telephone is on hook, the circuit is in idle state and there is no current flowing on the loop. An on-hook signal is also used to terminate a call and initiate a disconnect. When the telephone set is off hook, switch closure occurs causing a dc current to flow on the loop. The receiver off-hook condition is the first step in completing a telephone call. It is also used at the destination end as an answer signal to indicate that the called party has answered the telephone.</u>

<u>Other Nonessential Signals and Tones</u>: Some of the others are call waiting tones, caller waiting tones, calling card service tones, comfort tones, hold tones, intrusion tones, stutter dial tones etc..

Cordless Telephones

Cordless telephones are simply telephones that operate without cords attached to the handset. It is a full duplex, battery operated, portable radio transceiver that communicates directly with a stationary transceiver located somewhere in the subscribers home or office.



The base station is an ac-powered stationary radio transceiver capable of transmitting and receiving both supervisory and voice signals over the subscriber loop in the same manner as a standard telephone. It also must be capable of relaying voice and control signals to and from the portable telephone set through the wireless transceiver. The portable telephone set is a battery-powered, two-way radio capable of operating in the full duplex mode. As it uses a full duplex mode, it must transmit and receive at different frequencies. Base stations transmit on high-band frequencies and receives on low-band frequencies, while the portable unit transmits on low-band frequencies and receives on high-band frequencies.

Cordless telephones using the 2.4 GHz band offer excellent sound quality utilizing digital modulation and twin-band transmission to extend their range. With twin-band transmission, base stations transmit in the 2.4 GHz band, while portable units transmit in the 902 MHz to 928 MHz band.

Electronic Telephones



A typical electronic telephone comprised of one multifunctional integrated- circuit chip, a microprocessor chip, a Touch-Tone keypad, a speaker, a microphone, and several discrete devices is shown above. The major components included in the multifunctional integrated circuit chip are DTMF tone generator, MPU (microprocessor unit) interface circuitry, random access memory (RAM), tone ringer circuit, speech network and a line voltage regulator.

The Touch-Tone keyboard provides a means for the operator of the telephone to access the DTMF tone generator inside the multifunction integrated-circuit chip. The external crystal provides a stable and accurate frequency reference for producing the dual-tone multifrequency signalling tones. Once a ringing signal occurs, the tone ringer circuit activates and drives a piezoelectric sound element that produces an electronic ring. The voltage regulator converts the dc voltage received from the local loop to a constant- level dc supply voltage to operate the electronic components in the phone. The internal speech network contains several amplifiers and associated components as in standard telephone.

The microprocessor interface circuit interfaces the MPU to the multifunction chip. The MPU, with its internal RAM, controls many of the functions of the telephone, such as number storage, speed dialling, redialling and autodialing. The bridge rectifier protects the telephone from the relatively high-voltage ac ringing signal, and the switch hook is a mechanical switch that performs the same functions as the switch hook on a standard telephone set.

Paging Systems

Paging transmitters relay radio signals and messages from wire-line and cellular telephones to subscribers carrying portable receivers.



Standard Simplex Paging System

Standard paging systems are one-way, with signals transmitted from the paging system to portable pager and never in reverse direction. There are narrow-, mid- and widearea pagers. To contact a person carrying a pager, the telephone number assigned to that pager has to be dialed. The paging company receives the call and requests the number the pager person has to call. After the number is entered a terminating signal is appended to the number (#). Then paging system converts it to a digital code and transmits it in the form of a digitally encoded signal over a wireless communications system. If the paged person is within the range of a broadcast transmitter, the targeted pager will receive the message and the number to be called will be shown on an alphanumeric display.

Early paging systems used FM, but modern systems are using FSK or PSK. Each portable pager is assigned a special code called a cap code, which is a sequence of digits or a combination of digits and letters. It is broadcast along with the paging party's telephone number. Upon receiving the signal, the paging unit with demodulate and recognize the cap code. Once its been recognized, then the call-back number and may be a message will be displayed on the unit.

The most recent paging protocol developed is FLEX, which has been designed to minimize power consumption in the portable pager by using a synchronous time-slotted protocol to transmit messages in precise time slots. Each frame consists of 128 data frames, transmitted only once during a 4-minute period. Each frame lasts for 1.875 seconds and

includes two synchronizing sequences, a header containing frame information and pager identification addresses, and 11 discrete data blocks.

Caller ID

Caller ID (identification) enables the destination stations of a telephone call to display the name and telephone number of the calling party before the telephone is answered. This allows subscribers to screen incoming calls and decide whether they want to answer the telephone. The caller ID message is a simplex transmission sent from the central office switch over the local loop to a caller ID display unit at the destination station.

To ensure detection of caller ID signal, the telephone must ring at least twice before being answered as a 3-second window is present in between in which the caller ID signal must be transmitted

Telephone Instruments and Signals The Telephone Circuit

THE TELEPHONE CIRCUIT

The Local Subscriber Loop, Telephone Message- Channel Noise and Noise Weighting, Units of Powers Measurement, Transmission Parameters and Private-Line Circuits, Voice-Frequency Circuit Arrangements, Crosstalk

A telephone circuit is comprised of two or more facilities, interconnected in tandem, to provide a transmission path between a source and a destination. The facilities may be metallic cable pairs, optical fibers, or wireless carrier systems. The information transferred is called the message, and the circuit used is called the message channel.

The Local Subscriber Loop

The local subscriber loop is the only facility required by all voice-band circuits, as it is the means by which subscriber locations are connected to the local telephone company. The sole purpose of a local loop is to provide subscribers access to the public telephone network. The local loop is the primary cause of attenuation and phase distortion occurs when two or more frequencies undergo different amount of phase shift. There are seven main component parts that make up a traditional local loop: Feeder cable (F1), Serving area interface (SAI), Distribution cable (F2), Subscriber or standard network interface (SNI), Drop wire, Aerial and Distribution cable and drop-wire cross-connect point. Two components often found on local loops are loading coils and bridge taps.

Loading Coils: Loading coils placed in a cable decrease the attenuation, increase the line impedance and improve transmission levels for circuits longer than 18,000 feet. Loading coils allowed local loops to extend three to four times their previous length. A loading coil is simply a passive conductor wrapped around a core and placed in series with a cable creating a small electromagnet. Loading coils can be placed on telephone poles, in manholes, or on cross-connect boxes. Loading coils increase the effective distance that a signal must travel between two locations and cancels the capacitance that inherently builds up between wires with distance. Loading coils cause a sharp drop in frequency response at approximately 3400 Hz, this is undesirable for high-speed data transmission. Therefore, for high-performance data transmission, loading coils should be removed from the cables.

Bridge Taps: A *bridge tap* is an irregularity frequently found in cables serving subscriber locations. Bridge taps are unused sections of cable that are connected in shunt to a working cable pair, such as a local loop. Bridge taps can be placed at any point along a cable's length. Bridge taps increase the flexibility of a cable by making it easier to reassign a cable to a different subscriber without requiring a person working in the field to cross connect sections of cable. Bridge taps introduce a loss called *bridging loss*. They also allow signals to split and propagate down more than one wire. Bridge taps that are short and closer to the originating or terminating ends often produce the most interference. Bridge taps and loading coils are not generally harmful to voice transmissions, but if improperly used, they can literally destroy the integrity of a data signal.

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Loop Resistance: The dc resistance of a subscriber local loop is called loop resistance. It depends primarily on the type of wire and wire size. Most local loops use 18- to 26-gauge, twisted-pair copper wire. The dc loop resistance for copper conductors is approximated by

$$R_d = 0.1095/d^2$$

Where, *Rd* = dc loop resistance (ohms per mile) *d* = wire diameter (inches)

Units of Power Measurement

The decibel (dB) is the basic yardstick used for making power measurements in communications. The unit dB is simply a logarithmic expression representing the ratio of one power level to another and expressed as:

 $dB = 10 \log (P_1 / P_2)$, where P1 and P2 are power levels at two different points in a transmission system.

The unit dBm is often used to reference the power level at a given point to 1 milliwatt. One milliwatt is the level from which all measurements are referenced. It is expressed mathematically as:

 $dBm = 10 \log (P / 1 mW)$, where P is the power at any point in a transmission system.

<u>Transmission level point (TLP)</u> is defined as the optimum level of a test tone on a channel at some point in a communications system. The <u>transmission level (TL)</u> at any point in a transmission system is the ratio in dB of the power of a signal at that point to the power the same signal would be at a 0-dBm transmission level point. <u>Data level point (DLP)</u> is a parameter equivalent to TLP except TLP is used for voice circuits, whereas DLP is used as a reference for data transmission. The DLP is always 13 dB below the voice level for the same point.

Transmission Parameters and Private-Line Circuits

Transmission parameters apply to dedicated private-line data circuits that utilize the private sector of the public telephone network. Private-line data circuits have several advantages over using the switched public telephone network:

- Transmission characters are more consistent because the same facilities are used with every transmission.
- > The facilities are less prone to noise produced in the telephone company switches.
- Line conditioning is available only on private-line facilities
- Higher transmission bit rates and better performance is appreciated with private-line data circuits.
- Private-line data circuits are more economical for high-volume circuits.

Transmission parameters are divided into three broad categories:

- Bandwidth parameters:- attenuation distortion and envelope delay distortion
- Interface parameters:- terminal impedance, in-band and out-of-band signal power, test signal power, and ground isolation
- Facility parameters:- noise measurements, frequency distortion, phase distortion, amplitude distortion, and nonlinear distortion.

Bandwidth Parameters

<u>Attenuation distortion</u> is the difference in circuit gain experienced at a particular frequency with respect to the circuit gain of a reference frequency. This characteristic is sometimes referred to as *frequency response, differential gain,* and *1004-Hz deviation*. <u>Envelope delay distortion</u> is an indirect method of evaluating the phase delay characteristics of a circuit. To reduce attenuation and envelope delay distortion and improve the performance of data modems operating over standard message channels, it is often necessary to improve the quality of the channel. The process used to improve a basic telephone channel is called *line conditioning*. Line conditioning improves the high frequency response of a message channel and reduces power losses. Telephone companies offer two types of special line conditioning for subscriber loops: C-type and D-type.

C-type Line conditioning:- *C-type conditioning* specifies the maximum limits for attenuation distortion and envelope delay distortion. C-type conditioning pertains to line impairments for which compensation can be made with filters and equalizers. There are five classifications or levels of C-type conditioning available. The grade of conditioning a subscriber selects depends on the bit rate, modulation technique and desired performance of the data modems used on the line. The five classifications of C-type conditioning are the following:

- > C1 and C2 conditioning pertain to two-point and multipoint circuits.
- C3 conditioning is for access lines and trunk circuits associated with private switched networks.
- C4 conditioning pertains to two-point and multipoint circuits with a maximum of four stations.
- > C5 conditioning pertains only to two-point circuits.

D-type Line conditioning: - *D-type conditioning* neither reduces the noise on a circuit nor improves the signal-to-noise ratio. It simply sets the minimum requirements for *signal-to-noise* (S/N) *ratio* and *nonlinear distortion*. D-type conditioning is sometimes referred to as *high performance conditioning* and can be applied to private-line data circuits in addition to either basic or C-conditioned requirements. There are two categories for D-type conditioning: DI and D2. Limits imposed by DI and D2 are virtually identical. The only difference between the two categories is the circuit arrangement to which they apply. DI conditioning specifies requirements for two-point circuits and D2 conditioning specifies requirements for two-point circuits and D2 conditioning specifies requirements is 9600 bps because without D-type conditioning, it is highly unlikely that the circuit can meet the minimum performance requirements guaranteed by the telephone company.

D-type conditioned circuits must meet the following specifications:

- Signal-to-C-notched noise ratio: ≥28 dB
- Nonlinear distortion
- ➤ Signal-to-second-order distortion: ≥35 dB
- ➤ Signal-to-third-order distortion: ≥40 dB

The signal-to-notched noise ratio requirement for standard circuits is only 24 dB, and they have no requirements for nonlinear distortion.

<u>Interface Parameters</u>

The two primary considerations of the interface parameters are electrical protection of the telephone network and its personnel and standardization of design arrangements. The interface parameters include the following:

- Station equipment impedances should be 600 Ω resistive over the usable voice band.
- Station equipment should be isolated from ground by a minimum of 20MΩ dc and 50 kΩ ac
- The basic voice-grade telephone circuit is a 3002 channel; it has an ideal bandwidth of 0 Hz to 4 kHz and a usable bandwidth of 300 Hz to 3000 Hz
- > The circuit gain at 3000 Hz is 3 dB below the specified in-band signal power.
- The gain at 4 kHz must be at least 15 dB below the gain at 3 kHz
- > The maximum transmitted signal power for a private-line circuit is 0 dBm

<u> Facílíty Parameters</u>

Facility parameters represent potential impairments to a data signal. They include the following:

<u>**10O4-Hz variation**</u>: - The telephone industry has established 1004 Hz as the standard testtone frequency. The purpose of the 1004-Hz test tone is to simulate the combined signal power of a standard voiceband data transmission. The 1004-Hz channel loss for a privateline data circuit is typically 16 dB. A 1004-Hz test tone applied at the transmit end of a circuit should be received at the output of the circuit at -16 dBm. Long-term variations in the gain of the transmission facility are called *1004-Hz variation* and should not exceed ±4 dB. Thus, the received signal power must be within the limits of -12 dBm to -20 dBm.

<u>C-message noise</u>: - C-message noise measurements determine the average weighted rms noise power. Unwanted electrical signals are produced from the random movement of electrons in conductors. This type of noise is commonly called *thermal noise* because its magnitude is directly proportional to temperature. Because the electron movement is completely random and travels in all directions, thermal noise is also called *random noise*, and because it contains all frequencies, it is sometimes referred to as *white noise*. Thermal noise is additive, its magnitude is dependent, in part, on the electrical length of the circuit. C-message noise measurements are the terminated rms power readings at the receive end of a circuit with the transmit end terminated in the characteristic impedance of the telephone line.

Impulse noise: *Impulse noise* is characterized by high-amplitude peaks (impulses) of short duration having an approximately flat frequency spectrum. Impulse noise can saturate a message channel. Impulse noise is the primary source of transmission errors in data circuits. There are numerous sources of impulse noise—some are controllable, but most are not. The primary cause of impulse noise is man-made sources, such as interference from ac power lines, transients from switching machines, motors, solenoids, relays, electric trains, and so on. Impulse noise can also result from lightning and other adverse atmospheric conditions.

<u>Gain hits and dropouts</u>: A *gain hit* is a sudden, random change in the gain of a circuit resulting in a temporary change in the signal level. The primary cause of gain hits is noise transients (impulses) on transmission facilities during the normal course of a day. A *dropout* is a decrease in circuit gain (i.e., signal level) of more than 12 dB lasting longer than 4 ms. Dropouts are characteristics of temporary open-circuit conditions and are generally caused by deep fades on radio facilities or by switching delays.

<u>Phase hits</u>: *Phase hits* (slips) are sudden, random changes in the phase of a signal. Phase hits are classified as temporary variations in the phase of a signal lasting longer than 4 ms. Phase hits, like gain hits are caused by transients when transmission facilities are switched.



Phase hits and phase jitter

<u>Phase jitter</u>: *Phase jitter* is a form of incidental phase modulation-a continuous, uncontrolled variation in the zero crossings of a signal. Generally, phase jitter occurs at a 300-Hz rate or lower, and its primary cause is low-frequency ac ripple in power supplies.

<u>Single frequency interference</u>: *Single-frequency interference* is the presence of one or more continuous, unwanted tones within a message channel. The tones are called *spurious tones* and are often caused by crosstalk or cross modulation between adjacent channels in a transmission system due to system nonlinearities. Spurious tones are measured by terminating the transmit end of a circuit and then observing the channel frequency band. Spurious tones can cause the same undesired circuit behaviour as thermal noise. Single-frequency interference is shown below:

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Single-frequency interference (spurious tone)

<u>Frequency Shift</u>: Frequency shift is when the frequency of a signal changes during transmission. Analog transmission systems used by telephone companies require coherent demodulation and for this the receiver should be synchronous i.e. the frequency must be reproduced exactly in the receiver. The longer a circuit, the more analog transmission systems and the more likely frequency shift will occur.



<u>Phase intercept distortion</u>: *Phase intercept distortion* occurs in coherent SSBSC systems, such as those using frequency division multiplexing when the received carrier is not reinserted with the exact phase relationship to the received signal as the transmit carrier possessed.

<u>**Peak-to-average ratio**</u>: - The difficulties encountered in measuring true phase distortion or envelope delay distortion led to the development *of peak-to-average ratio* (PAR) tests. Low peak-to-average ratios indicate the presence of differential delay distortion. PAR measurements are less sensitive to attenuation distortion than EDD tests and are easier to perform.

Crosstalk

Crosstalk can be defined as any disturbance created in a communications channel by signals in other communications channels (i.e., unwanted coupling from one signal path into another). Crosstalk can originate in telephone offices, at a subscriber's location, or on the facilities used to interconnect subscriber locations to telephone offices. The nature of crosstalk is described as *intelligible or unintelligible*. Intelligible crosstalk includes real loss of privacy. Unintelligible crosstalk usually involves crosstalk between unlike channels, such as different types of carrier facilities, frequency inversion and digital encoding.

There are three primary types of crosstalk: Nonlinear crosstalk, Transmittance crosstalk and Coupling crosstalk.

Nonlinear crosstalk: - It is a direct result of nonlinear amplification in analog communications systems. Nonlinear amplification produces harmonics and cross products (sum and difference frequencies). If the nonlinear frequency components fall into the passband of another channel, they are considered crosstalk. Nonlinear crosstalk can be distinguished from other types of crosstalk because the ratio of the signal power in the disturbing channel to the interference power in the disturbed channel is a function of the signal level in the disturbing channel.

Transmittance Crosstalk: - Crosstalk can also be caused by inadequate control of the frequency response of a transmission system, poor filter design, or poor filter performance. This type of crosstalk is most the prevalent when filters do not adequately reject undesired products from other channels. As this type of interference is caused by inadequate control of the transfer characteristics or transmittance of networks, it is called transmittance crosstalk.

Coupling Crosstalk: - Electromagnetic coupling between two or more physically isolated transmission media is called *coupling crosstalk*. The most common coupling is due to the effects of near-field mutual induction between cables from physically isolated circuits (i.e., when energy radiates from a wire in one circuit to a wire in a different circuit). To reduce coupling crosstalk due to mutual induction, wires are twisted together (hence the name *twisted pair*). Twisting the wires causes a cancelling effect that helps eliminate crosstalk. The probability of coupling crosstalk occurrence increases with cable length, signal power, and frequency.

There are two types of coupling crosstalk: near and far end. *Near-end crosstalk* (*NEXT*) is crosstalk that occurs at the transmit end of a circuit and travels in the opposite direction as the signal in the disturbing channel. *Far-end crosstalk* (*FEXT*) occurs at the far end receiver and is the energy that travels in the same direction as the signal in the disturbing channel.

Voice-Frequency Circuit Arrangements

Two-Wire Voice-Frequency Circuits

Two-wire transmission involves two Wires (one for the signal and one for a reference or ground) or a circuit configuration that is equivalent to using only two wires. Two-wire circuits are ideally suited to simplex transmission, although they are often used for half- and full-duplex transmission.



The above figure shows the simplest two-wire configuration, which is a passive circuit of two copper wires connecting a telephone or voice-band modem at one station through a telephone company interface to a telephone or voice-band modem at the destination station.



The above figure shows an active two-wire transmission system (which provides gain). The one difference between the above two circuits is the addition of an amplifier to compensate for transmission line losses. The amplifier is unidirectional and thus, limits transmission to one direction only (simplex).



The above circuit shows a two-wire circuit using a digital T carrier for the transmission medium. This circuit requires a T carrier transmitter at one end and a T carrier receiver at the other end. The digital T carrier transmission line is capable of two-way transmission but, the transmitter and receiver are not. The digital transmission medium is a pair of copper wires.



The above circuit is an equivalent two-wire circuit as the transmission medium is earth's atmosphere and there are no copper wires between the two stations.

Four-Wire Voice-Frequency Circuits

Four-wire transmission involves four wires (two for each direction-a signal and a reference) or a circuit configuration that is equivalent to using four wires. Four-wire circuits are ideally suited to full-duplex transmission, although they can (and very often do) operate in the half-duplex mode. As with two-wire transmission, there are two forms of four-wire transmission systems: *physical four wires* and *equivalent four wire*.

Four wire circuits have several inherent advantages like they are less noisy, have less crosstalk, and provide more isolation between the two directions of transmission when operating in either half-duplex or full-duplex mode. But, two-wire circuits require less wire, less circuitry, and thus less money than their four-wire counterparts. Four-wire operation has a disadvantage of providing amplification.

The following figures show the block diagrams for four possible four-wire circuit configurations.



Hybrids, Echo Suppressors and Echo Cancelers

When a two-wire circuit is connected to a four-wire circuit, an interface circuit is called a hybrid, or terminating, set. The hybrid circuit used to convert two-wire circuits to four-wire circuits is similar to the hybrid coil found in standard telephone sets.



Hybrid (terminating) sets

The above figure shows the block diagram for a two-wire to *four-wire* hybrid network. The hybrid coil compensates for impedance variations in the two-wire portion of the circuit. The amplifiers and attenuators adjust the signal power to required levels, and the equalizers compensate for impairments in the transmission line that affect the frequency response of the transmitted signal, such as line inductance, capacitance, and resistance. Signals travelling west to east (W-E) enter the terminating set from the two-wire line, where they are inductively coupled into the west-to-east transmitter section of the four-wire circuit. Signals received from the four-wire side of the hybrid propagate through the receiver in the east-to-west (E-W) section of the four- wire circuit, where they are applied to the center taps of the hybrid coils. If the impedances of the two-wire line and the balancing network are properly matched, all currents produced in the upper half of the hybrid by the E-W signals will be equal in magnitude but opposite in polarity. Therefore, the voltages induced in the secondary's will be 1800 out of phase with each other and, thus, cancel. This prevents any of the signals from being retransmitted to the sender as an echo. If the impedances of the two-wire line and the balancing network are not matched, voltages induced in the secondary's of the hybrid coil will not completely cancel. This imbalance causes a portion of the received signal to be returned to the sender on the W-E port ion of the four-wire circuit.

Telephone Instruments and Signals The Telephone Circuit

Balancing networks can never completely match a hybrid to the subscriber Loop because of long-term temperature variations and degradation of transmission lines. The talker hears the returned portion of the signal as an echo, and if the round-trip delay exceeds approximately 45 ms, the echo can become quite annoying. To eliminate this echo, devices called *echo suppressors* are inserted at one end of the four-wire circuit.



Echo Suppressor

The above figure shows a simplified block diagram of an echo suppressor. The speech detector senses the presence and direction of the signal. It then enables the amplifier in the appropriate direction and disables the amplifier in the opposite direction, thus preventing the echo from returning to the speaker. A typical echo suppressor suppresses the returned echo by as much as 60 dB. If the conversation is changing direction rapidly, the people listening may be able to hear the echo suppressors turning on and off (every time an echo suppressor detects speech and is activated, the first instant of sound is removed from the message, giving the speech a choppy sound). If both parties talk at the same time, neither person is heard by the other. With an echo suppressor in the circuit, transmissions cannot occur in both directions at the same time, thus limiting the circuit to half-duplex operation. Long-distance carriers, such as AT&T, generally place echo suppressors in four-wire circuits that exceed 1500 electrical miles in length (the longer the circuit, the longer the round-trip delay time). Echo suppressors are automatically disabled when they receive a tone between 2020 Hz and 2240 Hz, thus allowing full-duplex data transmission over a circuit with an echo suppressor. Full- duplex operation can also be achieved by replacing the echo suppressors with echo cancelers. Echo cancelers eliminate the echo by electrically subtracting it from the original signal rather than disabling the amplifier in the return circuit.

Assignment Questions

- 1. What is a telephone set? Describe in detail the various functional components of a standard telephone set/
- 2. Expalin briefly how caller ID operates and when it is used.
- 3. What is crosstalk and what are the three types of crosstalk in telephone systems? Compare near-end crosstalk and far-end crosstalk
- 4. What considerations are addressed by facility parameters? Compare phase hits and phase jitter
- 5. Compare the operation of a cordless telephone and a standard telephone
- 6. What is meant by transmission line encoding? Compare C-type and D-type line conditioning.
- 7. What is a paging system? Describe in detail with a neat block diagram how a paging system work
- 8. Discuss about the call progress tones and signals.
- 9. What are the steps involved in completing a local telephone call?