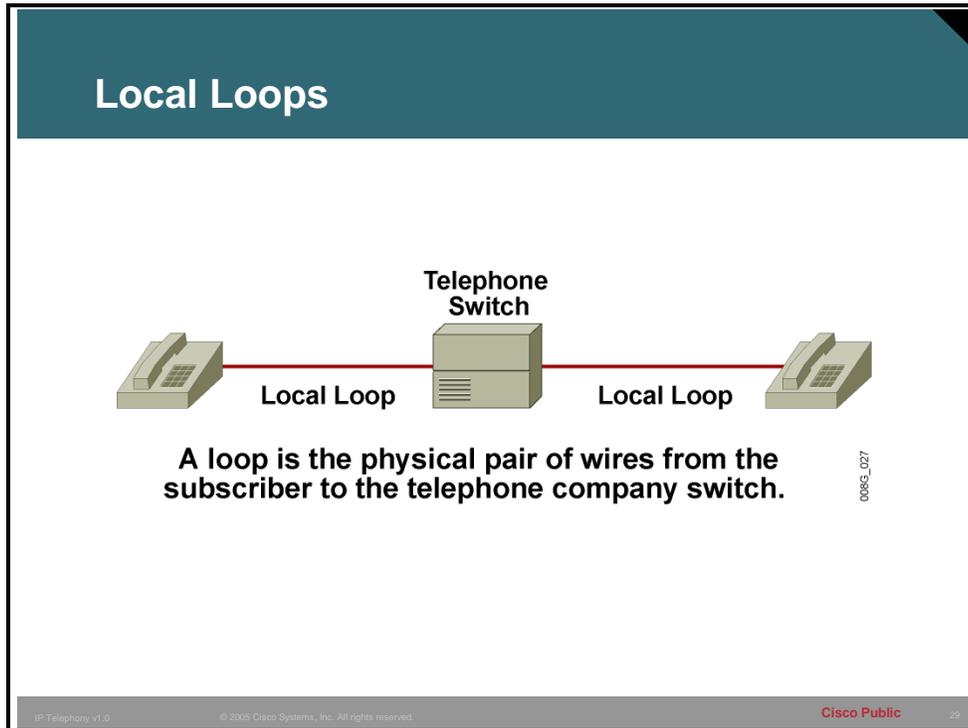


Analog Voice Basics

Local-Loop Connections

This topic describes the parts of a traditional telephony local-loop connection between a telephone subscriber and the telephone company.



A subscriber home telephone connects to the telephone company central office (CO) via an electrical communication path called a local loop, as depicted in the figure. The loop consists of a pair of twisted wires—one is called *tip*, the other is called *ring*.

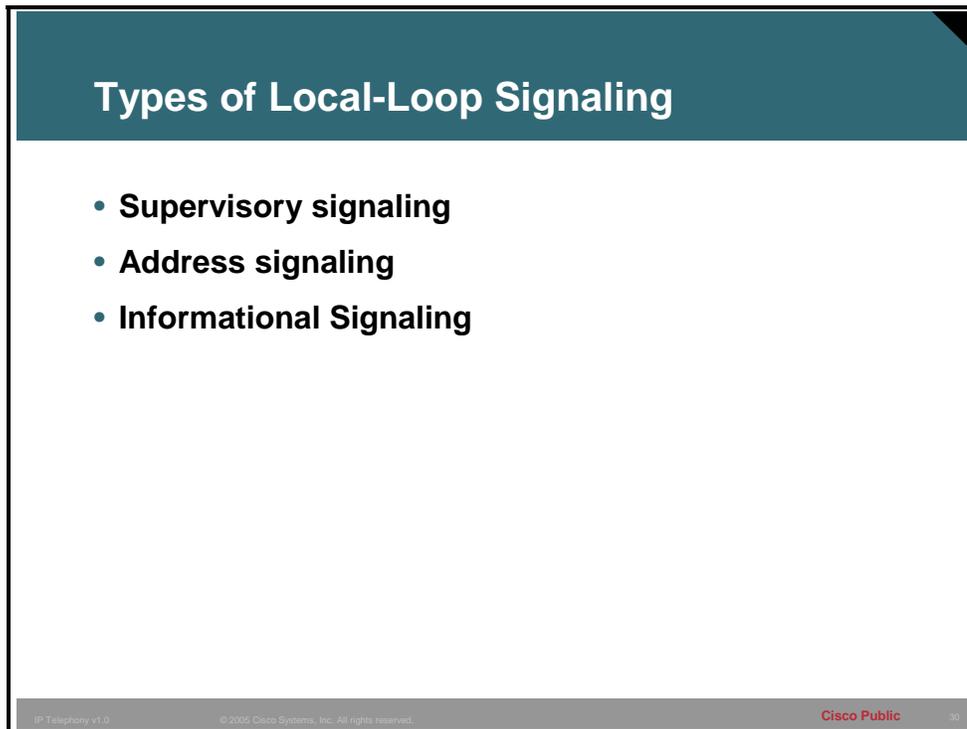
In most arrangements, the ring wire ties to the negative side of a power source, called the *battery*, while the tip wire connects to the ground. When you take your telephone off hook, current flows around the loop, allowing dial tone to reach your handset. Your local loop, along with all others in your neighborhood, connects to the CO in a cable bundle, either buried underground or strung on poles.

Example: Residential Telephone Service

Your home telephone service is provided to you from your service provider by way of two wires. Your home telephone controls whether or not the service on these wires is activated via the switch hook inside the telephone.

Types of Local-Loop Signaling

This topic explains local-loop signaling and lists some of the signaling types.

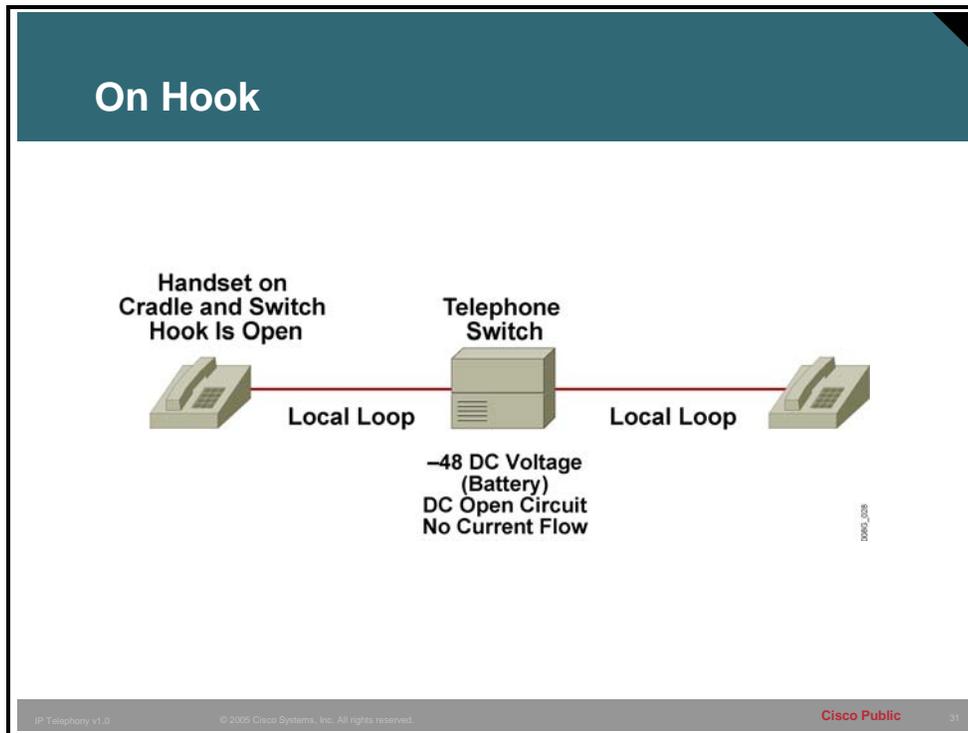


The slide features a dark teal header with the title 'Types of Local-Loop Signaling' in white. Below the header, a bulleted list contains three items: 'Supervisory signaling', 'Address signaling', and 'Informational Signaling'. At the bottom of the slide, there is a footer with the text '© Telephony v1.0', '© 2005 Cisco Systems, Inc. All rights reserved.', 'Cisco Public', and the number '30'.

A subscriber and telephone company notify each other of the call status through audible tones and an exchange of electrical current. This exchange of information is called local-loop signaling. Local-loop signaling consists of supervisory signaling, address signaling, and informational signaling, each of which has their own characteristics and purpose. The three types of local-loop signaling appear on the local loop and serve to prompt the subscriber and the switch into a certain action.

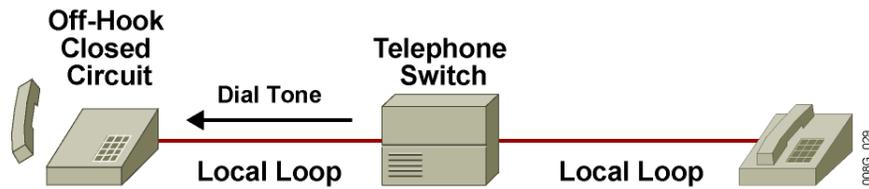
Supervisory Signaling

This topic describes on-hook, off-hook, and ringing supervisory signaling. Supervisory signaling serves to initiate the interaction between the subscriber and the attached switch.



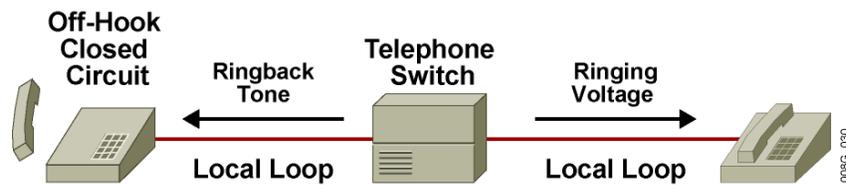
Resting the handset on the telephone cradle opens the switch hook and prevents the circuit current from flowing through the telephone. Regardless of the signaling type, a circuit goes on hook when the handset is placed on the telephone cradle and the switch hook is toggled to an open state. When the telephone is in this position, only the ringer is active.

Off Hook



To place a call, a subscriber must lift the handset from the telephone cradle. Removing the handset from the cradle places the circuit off hook. The switch hook is then toggled to a closed state, causing circuit current to flow through the electrical loop. The current notifies the telephone company that someone is requesting to place a telephone call. When the telephone network senses the off-hook connection by the flow of current, it provides a signal in the form of the dial tone to indicate that it is ready.

Ringling



IP Telephony 101

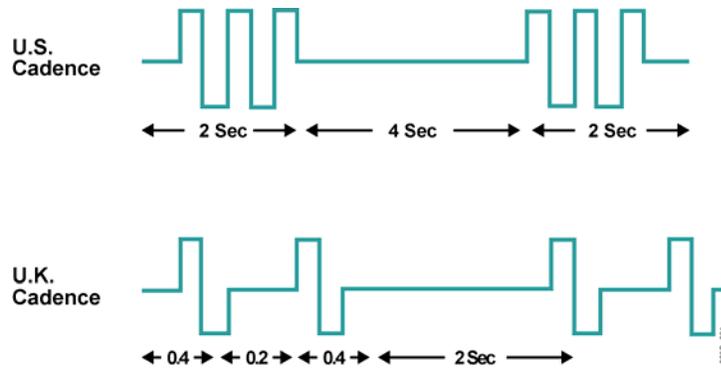
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When a subscriber makes a call, the telephone sends voltage to the ringer to notify the other subscriber of an inbound call. The telephone company also sends a ringback tone to the caller, alerting the caller that it is sending ringing voltage to the recipient telephone. Although ringback tone is not the same as ringing voltage, it sounds similar.

Ringling (Cont.)



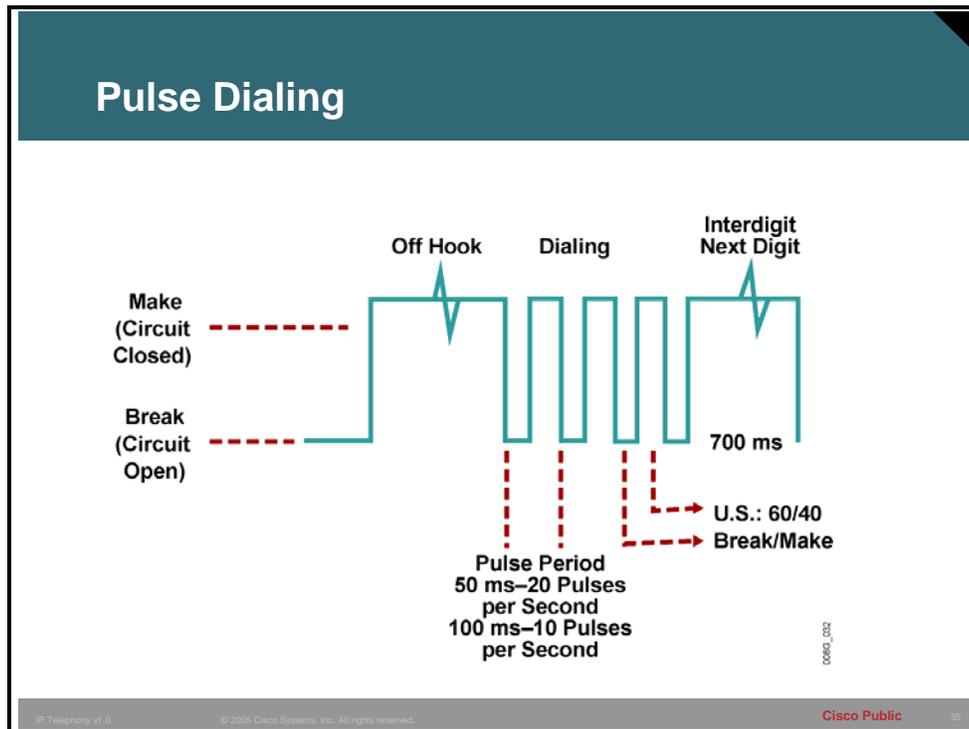
As depicted in the figure, the ringing supervisory tone in the United States is 2 seconds of tone followed by 4 seconds of silence. The United Kingdom uses a double ring of 0.4 seconds separated by 0.2 seconds of silence, followed by 2 seconds of silence.

Example: Ringling Cadences

The pattern of the ring signal, or ring cadence, varies around the world. In the United States, the ring signal, sent by the local service provider, is 2 seconds of ring followed by 4 seconds of silence. Your home telephone rings with this cadence when you have an incoming call.

Address Signaling

This topic describes pulse dialing and dual-tone multifrequency (DTMF) signaling.



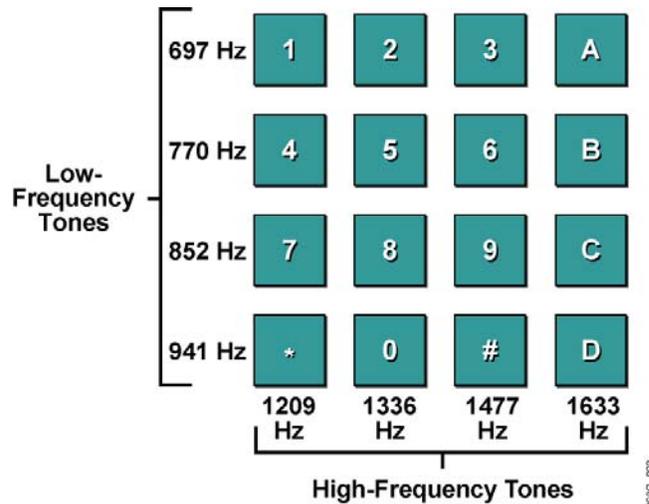
Although somewhat outdated, rotary-dial telephones are still in use and easily recognized by their large numeric dial-wheel. When placing a call, the subscriber spins the large numeric dial-wheel to send digits. These digits must be produced at a specific rate and within a certain level of tolerance. Each pulse consists of a “break” and a “make.” The break segment is the time that the circuit is open. The make segment is the time during which the circuit is closed. In the United States, the break-and-make cycle must correspond to a ratio of 60 percent break to 40 percent make.

A governor inside the dial controls the rate at which the digits are pulsed.

The dial pulse signaling process occurs as follows:

1. When a subscriber calls someone by dialing a digit on the rotary dial, a spring winds.
2. When the dial is released, the spring rotates the dial back to its original position.
3. While the spring rotates the dial back to its original position, a cam-driven switch opens and closes the connection to the telephone company. The number of consecutive opens and closes—or breaks and makes—represents the dialed digit.

Dual Tone Multifrequency



Users who have a touch-tone pad or a push-button telephone must push the keypad buttons to place a call. Each button on the keypad is associated with a set of high and low frequencies. Each row of keys on the keypad is identified by a low-frequency tone; each column of keys on the keypad is identified by a high-frequency tone. The combination of both tones notifies the telephone company of the number being called, hence the term dual tone multifrequency (DTMF).

The figure illustrates the combination of tones generated for each button on the keypad.

Informational Signaling

This topic lists the call-progress indicators and describes their functions.

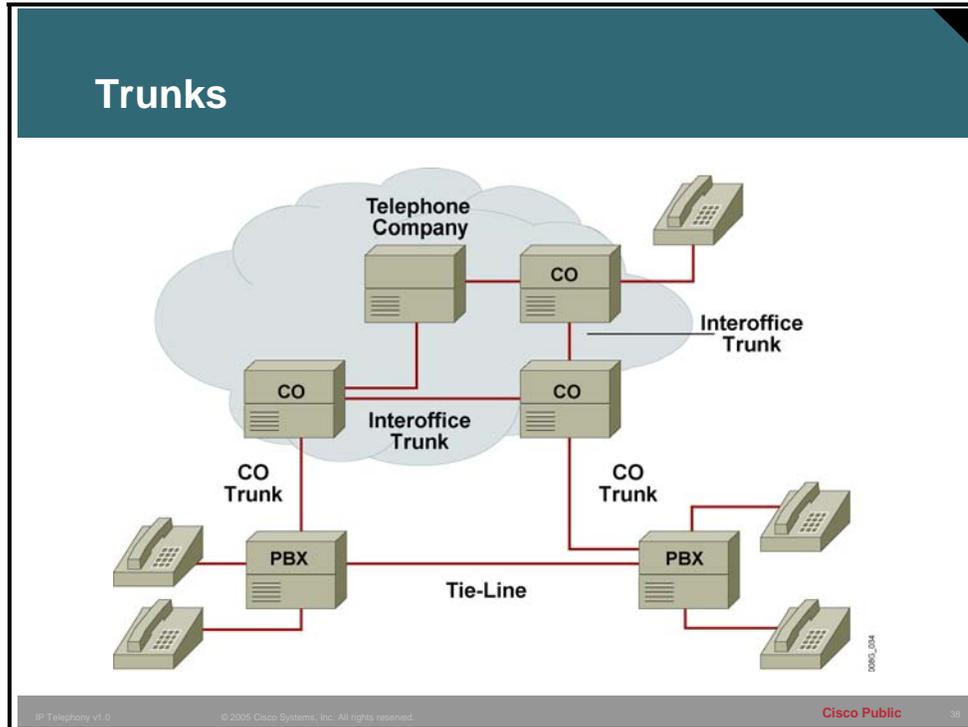
Informational Signaling with Call-Progress Indicators			
Tone	Frequency (Hz)	On Time (Sec) - Off Time (Sec)	
Dial	350 + 440	Continuous	
Busy	480 + 620	0.5	0.5
Ringback, line	440 + 480	2	4
Ringback, PBX	440 + 480	1	3
Congestion (toll)	480 + 620	0.2	0.3
Reorder (local)	480 + 620	0.3	0.2
Receiver off hook	1400 + 2060 + 2450 + 2600	0.1	0.1
No such number	200 + 400	Continuous	
Confirmation tone		Freq. Mod 1 kHz	

Call-progress indicators in the form of tone combinations are used to notify subscribers of call status. Each combination of tones represents a different event in the call process, as follows:

- **Dial tone:** Indicates that the telephone company is ready to receive digits from the user telephone. The Cisco routers provide dial tone as a method of showing that the hardware is installed. In a PBX or key telephone system, the dial tone indicates that the system is ready to receive digits.
- **Busy tone:** Indicates that a call cannot be completed because the telephone at the remote end is already in use.
- **Ringback (normal or PBX):** Indicates that the telephone company is attempting to complete a call on behalf of a subscriber.
- **Congestion:** Indicates that congestion in the long-distance telephone network is preventing a telephone call from being processed. The congestion tone is sometimes known as the all-circuits-busy tone.
- **Reorder tone:** Indicates that all of the local telephone circuits are busy, thus preventing a telephone call from being processed. The reorder tone is known to the user as fast-busy, and is familiar to anyone who operates a telephone from a PBX.
- **Receiver off hook:** Indicates that the receiver has been off hook for an extended period without placing a call.
- **No such number:** Indicates that a subscriber placed a call to a nonexistent number.
- **Confirmation tone:** Indicates that the telephone company is working on completing the call.

Trunk Connections

This topic describes the different types of trunks on a voice network and how they operate.



Before a telephone call terminates at its final destination, it is routed through multiple switches. When a switch receives a call, it determines whether the destination telephone number is within a local switch or if the call needs to go through another switch to a remote destination. Trunks connect the telephone company and PBX switches.

The primary function of the trunk is to provide the path between switches. The switch must route the call to the correct trunk or telephone line. Although many different subscribers share a trunk, only one subscriber uses it at any given time. As telephone calls end, they release trunks and make them available to the switch for subsequent calls. There can be several trunks between two switches.

The following are examples of the more common trunk types:

- **Private trunk lines:** Companies with multiple PBXs often connect them with tie trunk lines. Generally, tie trunk lines serve as dedicated circuits that connect PBXs. On a monthly basis, subscribers lease trunks from the telephone company to avoid the expense of using telephone lines on a per-call basis. These types of connections, known as tie-lines, typically use special interfaces called recEive and transMit, or ear and mouth (E&M), interfaces.
- **CO trunks:** A CO trunk is a direct connection between a PBX and the local CO that routes calls; for example, the connection from a private office network to the public switched telephone network (PSTN). When users dial **9**, they are connecting through their PBX to the CO trunk to access the PSTN. CO trunks typically use Foreign Exchange Office (FXO) interfaces. Certain specialized CO trunks are frequently used on the telephony network. A direct inward dialing (DID) trunk, for example, allows outside callers to reach specific internal destinations without having to be connected via an operator.

- **Interoffice trunks:** An interoffice trunk is a circuit that connects two local telephone company COs.

Foreign Exchange Trunks

- **Foreign Exchange Office**
 - Connects directly to office equipment
 - Used to extend connections to another location
- **Foreign Exchange Station**
 - Connects directly to station equipment
 - Used to provision local service

Foreign exchange (FX) trunks are interfaces that are connected to switches that support connection to either office equipment or station equipment. Office equipment includes other switches (to extend the connection) and Cisco devices. Station equipment includes telephones, fax machines, and modems.

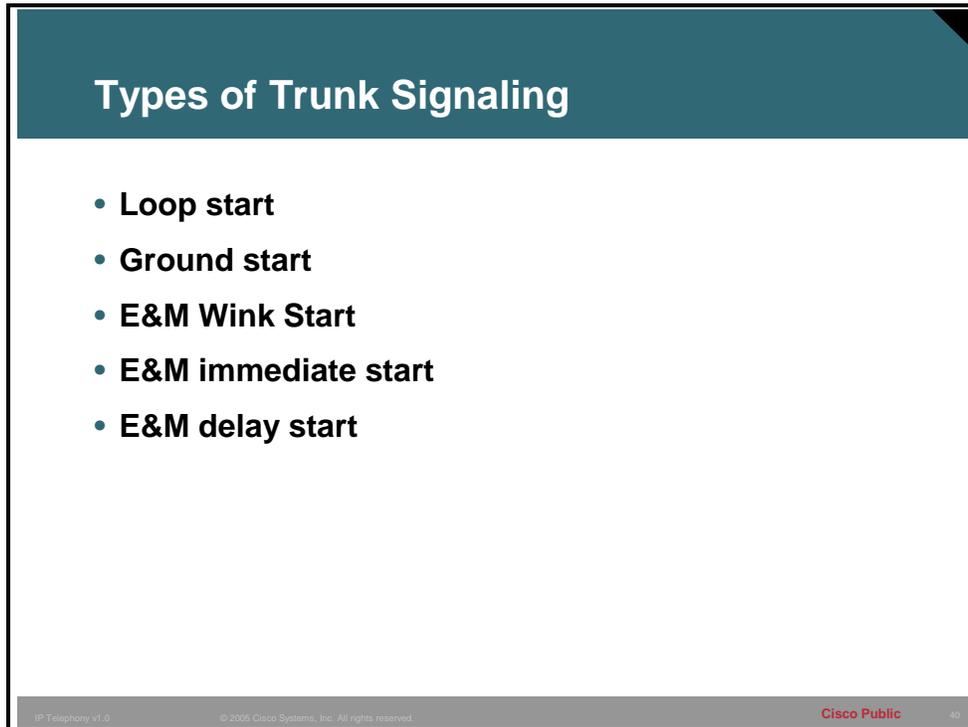
- **Foreign Exchange Office (FXO) interfaces:** An FXO interface connects a PBX to another switch or Cisco device. The purpose of an FXO interface is to extend the telephony connection to a remote site; for example, if a user on a corporate PBX wanted a telephone installed at home instead of in the local office where the PBX is located, an FXO interface would be used. The FXO interface would connect to a Cisco voice router, which would serve to extend the connection to the user home. This connection is an Off-Premises eXtension (OPX).
- **Foreign Exchange Station (FXS) interfaces:** An FXS interface connects station equipment: telephones, fax machines, and modems. A telephone connected directly to a switch or Cisco device requires an FXS interface. Because a home telephone connects directly to the telephone company CO switch, an FXS interface is used.

Example: Foreign Exchange Interfaces

The service provided by local telephone companies for residential phones uses a foreign exchange interface—specifically FXS. This service is provided on two wires. The service is considered a station-side connection because the interface terminates with a telephone.

Types of Trunk Signaling

This topic describes the trunk and line-seizure signaling types.



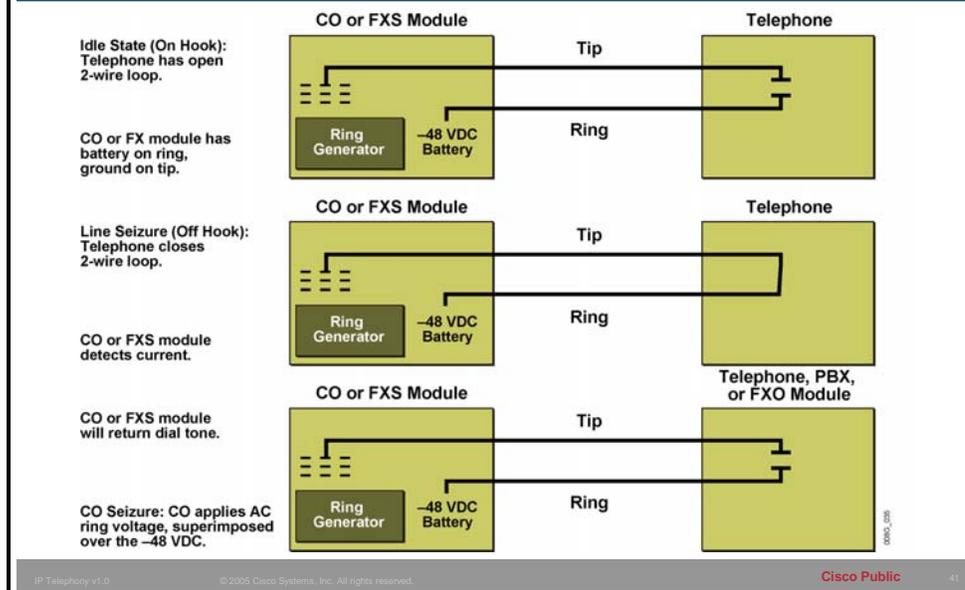
Types of Trunk Signaling

- **Loop start**
- **Ground start**
- **E&M Wink Start**
- **E&M immediate start**
- **E&M delay start**

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There must be signaling standards between the lines and trunks of a telephone network, just as there are signaling standards between a telephone and the telephone company. Trunk signaling serves to initiate the connection between the switch and the network. There are five different types of trunk signaling and each applies to different kinds of interfaces, such as FXS, FXO, and E&M.

Loop-Start Signaling



Loop-start signaling allows a user or the telephone company to seize a line or trunk when a subscriber is initiating a call. It is primarily used on local loops rather than on trunks.

A telephone connection exists in one of the following states:

- Idle (on hook)
- Telephone seizure (off hook)
- CO seizure (ringing)

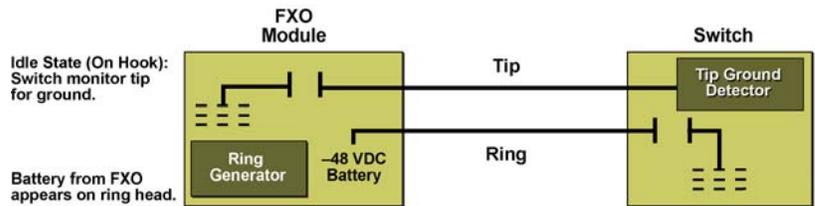
A summary of the loop-start signaling process is as follows:

1. When the line is in the idle state, or on hook, the telephone or PBX opens the two-wire loop. The CO or FXS has battery on ring and ground on tip.
2. If a user lifts the handset off the cradle to place a call, the switch hook goes off hook and closes the loop (line seizure). The current can now flow through the telephone circuit. The CO or FXS module detects the current and returns a dial tone.
3. When the CO or FXS module detects an incoming call, it applies AC ring voltage superimposed over the -48 VDC battery, causing the ring generator to notify the recipient of a telephone call. When the telephone or PBX answers the call, thus closing the loop, the CO or FXS module removes the ring voltage.

Loop-start signaling is a poor solution for high-volume trunks because it leads to glare incidents, or the simultaneous seizure of the trunk from both ends. Glare occurs, for example, when you pick up your home telephone and find that someone is already at the other end.

Glare is not a significant problem at home. It is, however, a major problem when it occurs between switches at high-volume switching centers, such as long-distance carriers.

Ground-Start Signaling



Ground-start signaling is a modification of loop-start signaling that corrects for the probability of glare. It solves the problem by providing current detection at both ends.

Although loop-start signaling works when you use your telephone at home, ground-start signaling is preferable when there are high-volume trunks involved at telephone switching centers. Because ground-start signaling uses a request or confirm switch at both ends of the interface, it is preferable over other signaling methods on high-usage trunks, such as FXOs.

FXOs require implementation of answer supervision (reversal or absence of current) on the interface for the confirmation of on hook or off hook.

Ground-start signaling is not common in Voice over IP (VoIP) networks.

E&M Signaling

- Separate signaling leads for each direction
- E-lead (inbound direction)
- M-lead (outbound direction)
- Allows independent signaling

PBX to Intermediate Device

Type	Lead	On Hook	Off Hook
I	M	Ground	Battery (-48 VDC)
II	M	Open	Battery (-48 VDC)
III	M	Ground	Battery (-48 VDC)
IV	M	Open	Ground
V	M	Open	Ground

Intermediate Device to PBX

Type	Lead	On Hook	Off Hook
	E	Open	Ground
	E	Open	Ground
	E	Open	Ground
IV	E	Open	Ground
	E	Open	Ground

E&M signaling supports tie-line type facilities or signals between voice switches. Instead of superimposing both voice and signaling on the same wire, E&M uses separate paths, or leads, for each.

Example: E&M Signaling

To call a remote office, your PBX must route a request for use of the trunk over its signal leads between the two sites. Your PBX makes the request by activating its M-lead. The other PBX detects the request when it detects current flowing on its E-lead. It then attaches a dial register to the trunk and your PBX, which sends the dialed digits. The remote PBX activates its M-lead to notify the local PBX that the call has been answered.

Types of E&M Signaling

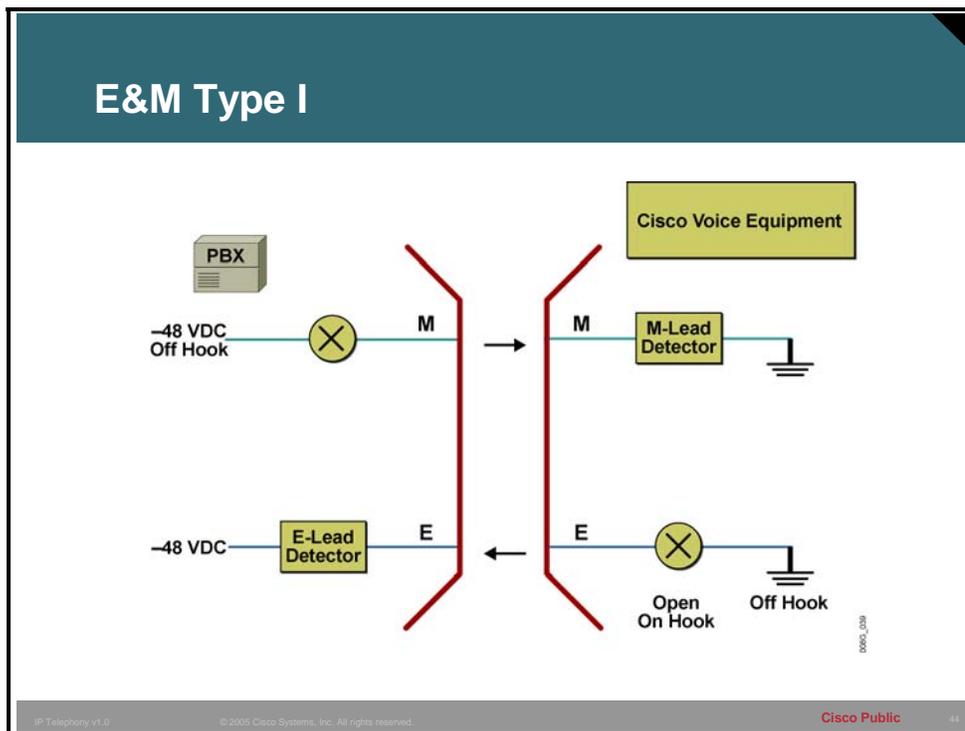
There are five types of E&M signaling: Type I, Type II, Type III, Type IV, and Type V. The E&M leads operate differently with each wiring scheme, as shown in the table.

Types of E&M Signaling

E&M Signaling Type	PBX to Intermediate Device			Intermediate Device to PBX		
	Lead	On Hook	Off Hook	Lead	On Hook	Off Hook
Type I	M	Ground	Battery (-48 VDC)	E	Open	Ground
Type II	M	Open	Battery (-48 VDC)	E	Open	Ground
Type III	M	Ground	Battery (-48 VDC)	E	Open	Ground
Type IV	M	Open	Ground	E	Open	Ground
Type V	M	Open	Ground	E	Open	Ground

E&M Signaling Types

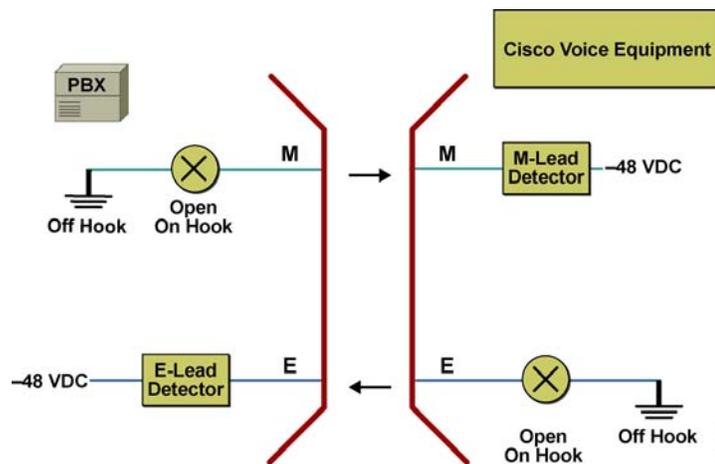
This topic identifies the five E&M signaling types and provides a description of each.



Four-wire E&M Type I signaling is actually a six-wire E&M signaling interface common in North America. One wire is the E-lead; the second wire is the M-lead, and the remaining two pairs of wires serve as the audio path. In this arrangement, the PBX supplies power, or *battery*, for both the M-leads and E-leads. This arrangement also requires that a common ground be connected between the PBX and the Cisco voice equipment.

With the Type I interface, the Cisco voice equipment (tie-line equipment) generates the E signal to the PBX by grounding the E-lead. The PBX detects the E signal by sensing the increase in current through a resistive load. Similarly, the PBX generates the M signal by sourcing a current to the Cisco voice equipment (tie-line equipment), which detects it via a resistive load.

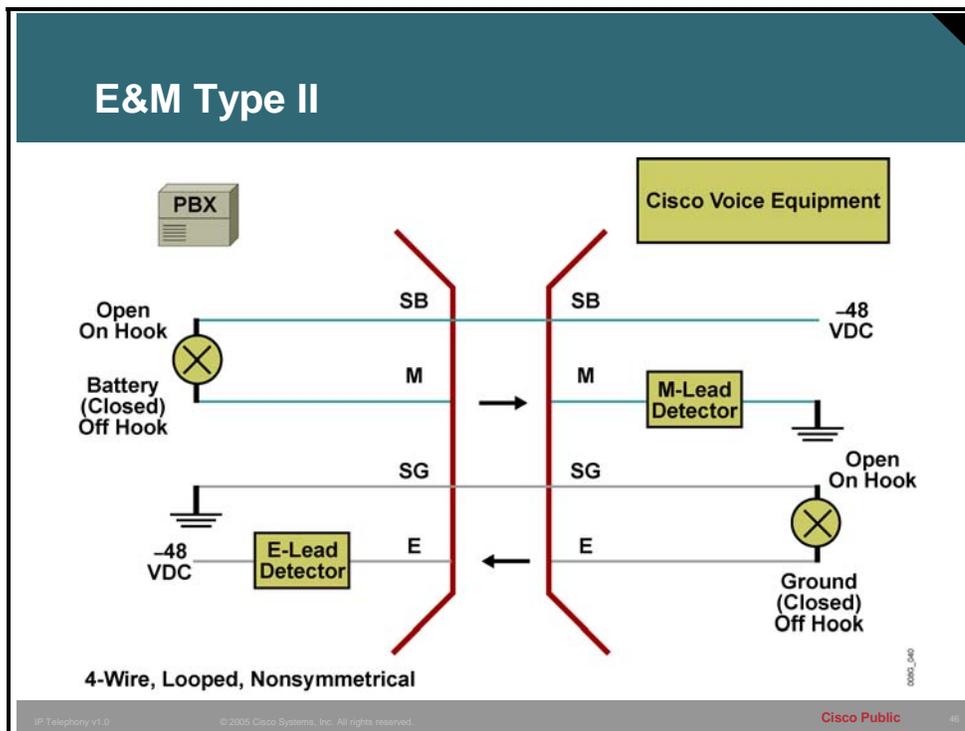
E&M Type V



Type V is another six-wire E&M signaling type and the most common E&M signaling form outside of North America. In Type V, one wire is the E-lead and the other wire is the M-lead.

Type V is a modified version of the Type I interface. In the Type V interface, the Cisco voice equipment (tie-line equipment) supplies battery for the M-lead while the PBX supplies battery for the E-lead. As in Type I, Type V requires that a common ground be connected between the PBX and the Cisco voice equipment.

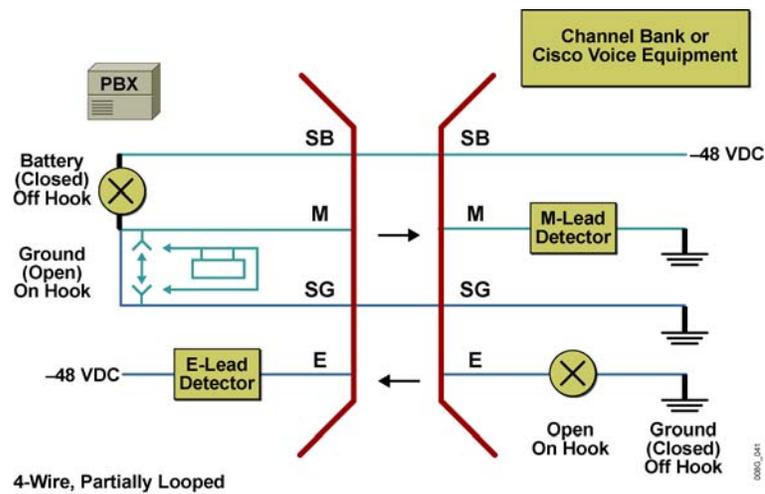
E&M Type II



Types II, III, and IV are eight-wire interfaces. One wire is the E-lead, the other wire is the M-lead. Two other wires are signal ground (SG) and signal battery (SB). In Type II, SG and SB are the return paths for the E-lead and M-lead, respectively.

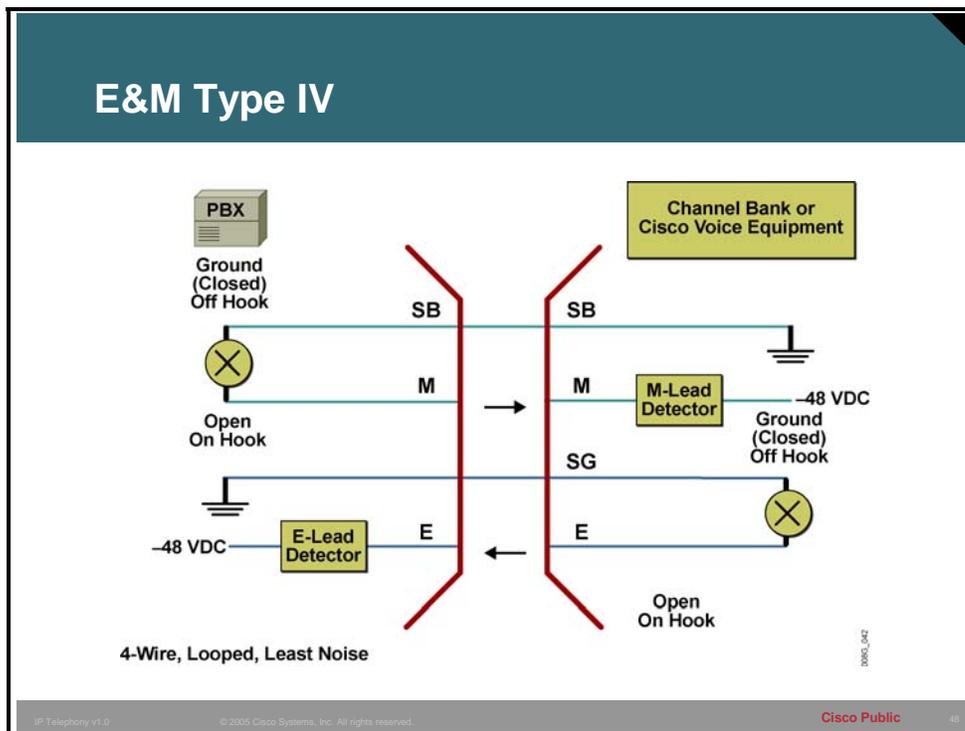
The Type II interface exists for applications where a common ground between the PBX and the Cisco voice equipment (tie-line equipment) is not possible or practical; for example, the PBX is in one building on a campus and the Cisco equipment is in another. Because there is no common ground, each of the signals has its own return. For the E signal, the tie-line equipment permits the current to flow from the PBX; the current returns to the PBX SG lead or reference. Similarly, the PBX closes a path for the current to generate the M signal to the Cisco voice equipment (tie-line equipment) on the SB lead.

E&M Type III



Type III is useful for environments where the M-lead is likely to experience electrical interference and falsely signal its attached equipment. When idle, Type III latches the M-lead via an electrical relay to the SG lead. When the PBX activates the M-lead, it first delatches the SG lead via the relay and signals normally, as in Type II. Type III is not a common implementation.

E&M Type IV

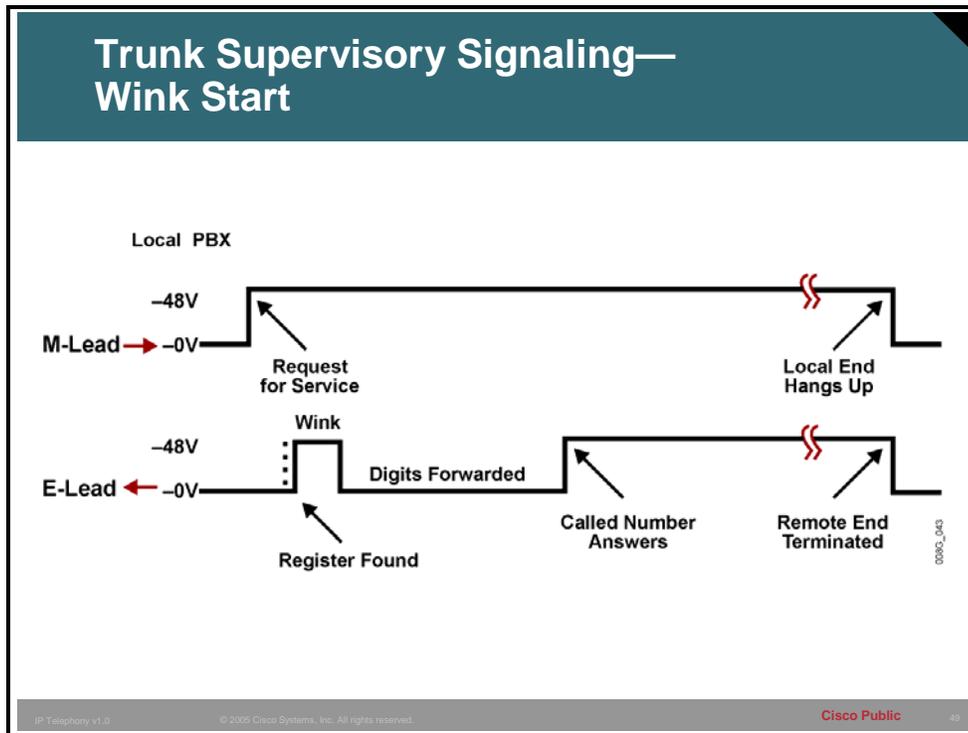


Type IV is a variation of Type II. In this arrangement, the battery source and ground are reversed on the SB and M wires (as compared to Type II). This means that both the SB and SG wires are grounded. Type IV signaling is symmetric and requires no common ground. Each side closes a current loop to signal, which detects the flow of current through a resistive load to indicate the presence of the signal. Cisco voice equipment does not support Type IV.

The most common E&M signaling interface in the United States is Type I, and the most common in European countries is Type V. Other variations exist for special applications and purposes. Cisco does not support Type IV.

Trunk Signal Types Used by E&M

This topic describes Wink-Start, immediate-start, and delay-start signaling as used by E&M signaling.

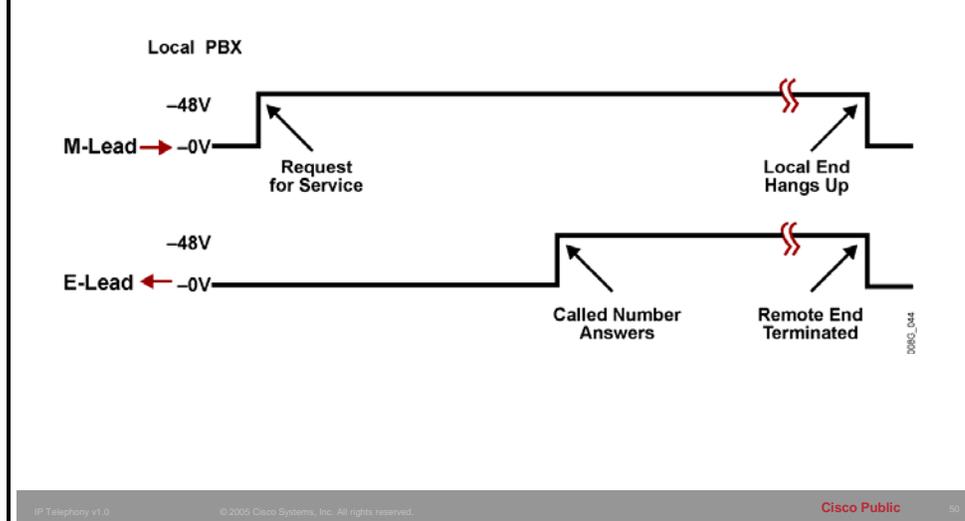


Tie trunks have bidirectional supervisory signaling that allows either end to initiate a trunk seizure. In this way, one PBX seizes the trunk, which then waits for an acknowledgment reply from the remote end. The local end must differentiate between a return acknowledgment and a remote-end request for service. Wink-Start signaling is the most common E&M trunk seizure signal type.

The following scenario summarizes the Wink-Start protocol event sequence:

1. The calling office seizes the line by activating its M-lead.
2. Instead of returning an off-hook acknowledgment immediately, the called switch allocates memory for use as a dial register, in the area of memory it uses to store incoming digits.
3. The called switch toggles its M-lead on and off for a specific time (usually 170 to 340 ms). (This on-hook/off-hook/on-hook sequence constitutes the wink.)
4. The calling switch receives the wink on its E-lead and forwards the digits to the remote end. DTMF tones are forwarded across the E&M link in the audio path, not on the M-lead.
5. The called party answers the telephone, and the called PBX raises its M-lead for the duration of the call.

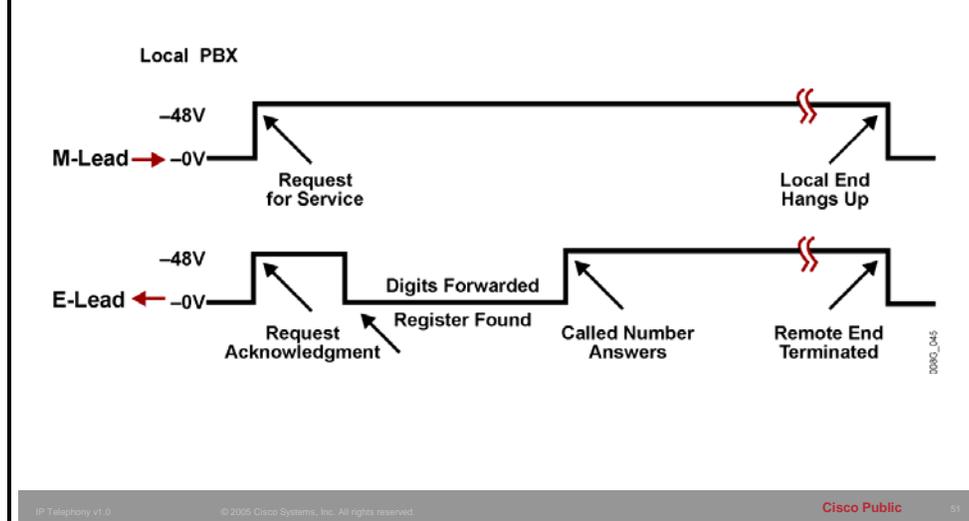
Trunk Supervisory Signaling— Immediate Start



If the timing of the returned wink is too short or impossible to detect, the trunk uses immediate start. This occurs occasionally if a PBX vendor implements Wink Start but does not conform to the standards. The following scenario summarizes the sequence of events for the immediate-start protocol:

1. The calling PBX seizes the line by activating its M-lead.
2. Instead of receiving an acknowledgment, the calling PBX waits a predetermined period (a minimum of 150 ms) and forwards the digits blindly. DTMF tones are forwarded across the E&M link in the audio path, not on the M-lead.
3. The called PBX acknowledges the calling PBX only after the called party answers the call by raising its M-lead.

Trunk Supervisory Signaling— Delay Start

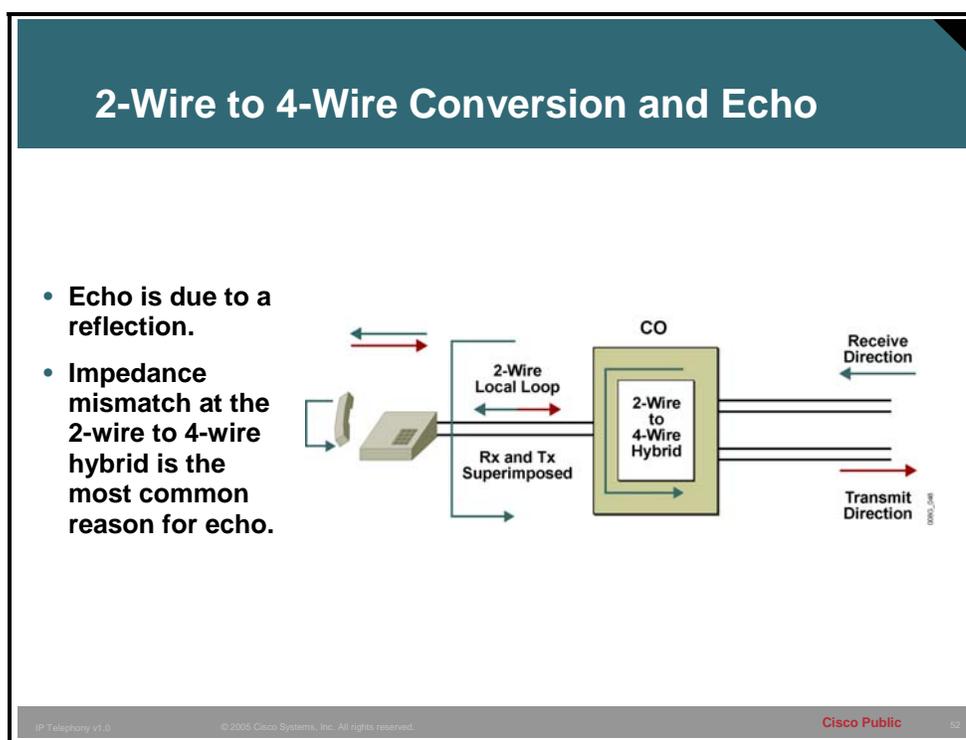


Delay start is the original start protocol for E&M. It is used when all of the equipment is mechanical and requires time to process requests. The following scenario summarizes delay-start signaling:

1. When you place a call, your calling switch goes off hook by activating its M-lead.
2. The called switch acknowledges the request by activating its M-lead, and then rotates armatures and gears to reset its dial register to zero.
3. When the dial register at the called switch is in the ready state, the called switch deactivates its M-lead.
4. The calling switch then sends dialed digits. DTMF tones are forwarded across the E&M link in the audio path, not on the M-lead.
5. When the called party answers, the called switch again activates its M-lead.

Line Quality

This topic describes impairments commonly found in analog telephone circuits and offers solutions to the problem of echo.



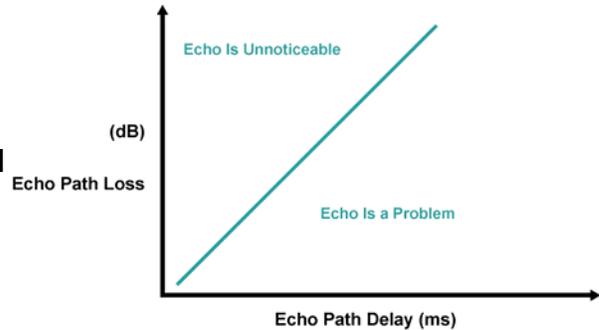
Although a local loop consists of two wires, when it reaches the switch, the connection changes to four wires with a two- to four-wire hybrid converter. Trunks then transport the signal across the network.

Telephone networks can experience two types of echo: acoustic echo and electrical echo. Acoustic echo frequently occurs with speakerphones, when the received voice on the speaker excites the microphone and travels back to the speaker. Electrical echo occurs when there is an electrical inconsistency in the telephony circuits. This electrical inconsistency is called impedance mismatch.

If the lines have a good impedance match, the hybrid is considered balanced, with little or no reflected energy. However, if the hybrid is inadequately balanced, and a portion of the transmit voice is reflected back toward the receive side, echo results.

Echo Is Always Present

- **Echo as a problem is a function of the echo delay and the loudness of the echo.**



Some form of echo is always present. However, echo can become a problem under the following conditions:

- The magnitude or loudness of the echo is high.
- The delay time between when you speak and when you hear your voice reflected is significant.
- The listener hears the speaker twice.

The two components of echo are loudness and delay. Reducing either component reduces overall echo. When a user experiences delay, the conversation can get choppy, and the words of the participants sometimes overlap.

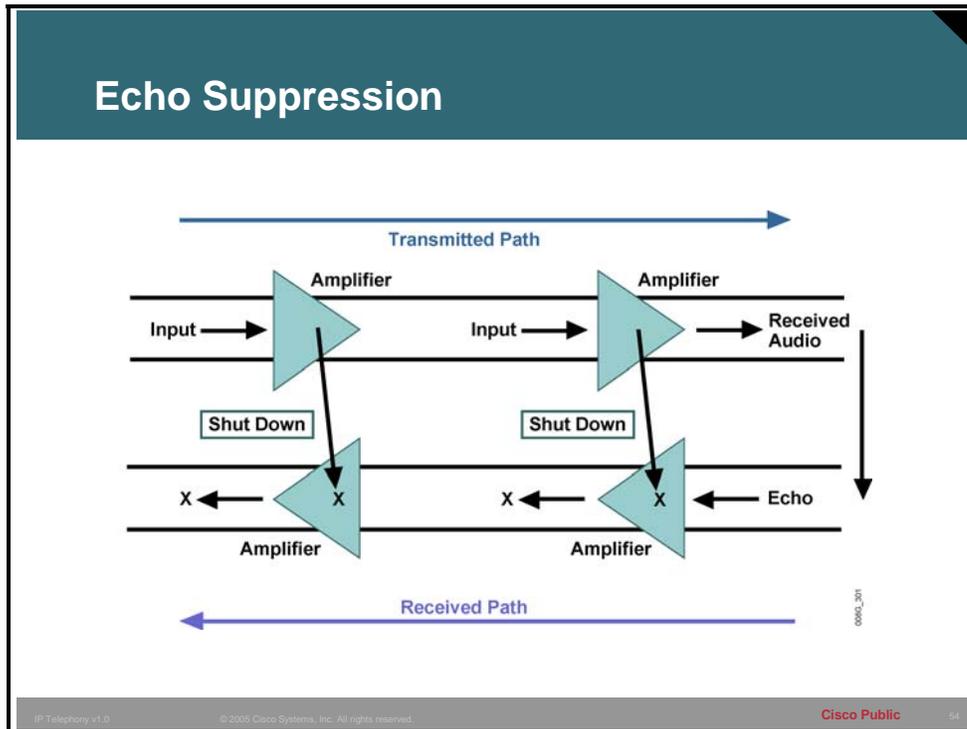
Note Echo tolerance varies. For most users, however, echo delay over 50 ms is generally problematic.

Management of Echo

There are two ways to solve an echo problem in your telephone network:

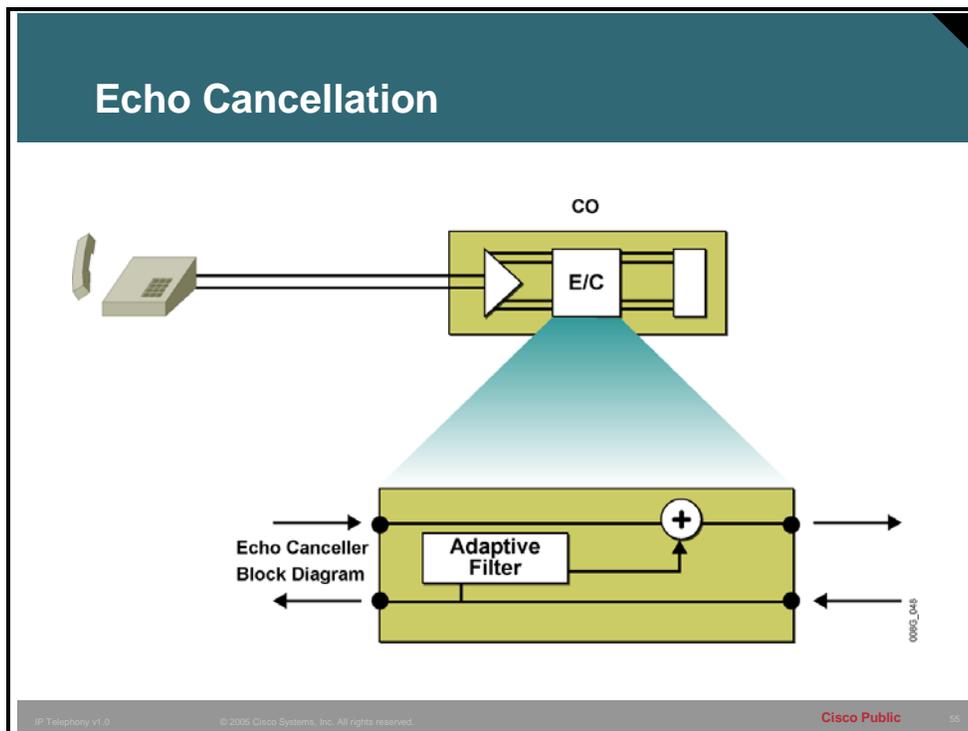
- Echo suppression
- Echo cancellation

This topic explains and compares the two approaches to echo management.



The echo suppressor works by transmitting speech in the forward direction and prohibiting audio in the return direction. The echo suppressor essentially breaks the return transmission path. This solution works sufficiently for voice transmission. However, for full-duplex modem connections, the action of the echo suppressor prevents communication. Therefore, when modems handshake, the answering modem returns a tone of 2025 Hz to the calling modem, which serves to disable the echo suppressors along the transmission path.

Echo Cancellation



Echo suppression has shortcomings in addressing certain echo conflict situations. Echo cancellation is a more sophisticated method of eliminating echo.

Rather than breaking or attenuating the return path (as in echo suppression), echo cancellation uses a special circuit to build a mathematical model of the transmitted speech pattern and subtract it from the return path. This echo elimination method is depicted in the figure.

Note Echo cancellation applies the same technology that is used in audio headphones to cancel ambient noise.

Echo cancellation is the most common method of removing echo in the telephone network today, and is used when it is necessary to adjust for echo on a Cisco device.

Note The echo canceller removes the echo from one end of the circuit only. If echo is an issue at both ends of the circuit, you must apply another echo canceller at the other end.

Example: Echo Cancellation

The headsets used by airline pilots feature a suppression circuit, which cancels ambient noise so that the pilot hears only the audio from the headset. Any ambient noise from the cockpit is cancelled. This is the same technology used in echo cancellers.