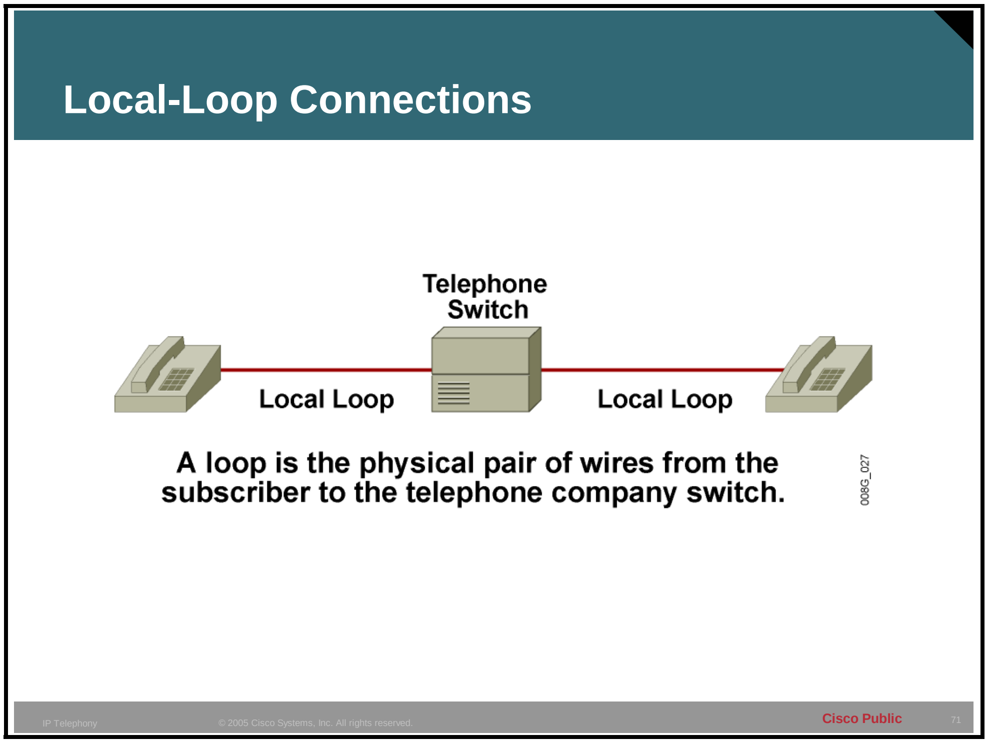
Analog and Digital Voice Interfaces

Local-Loop Connections

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

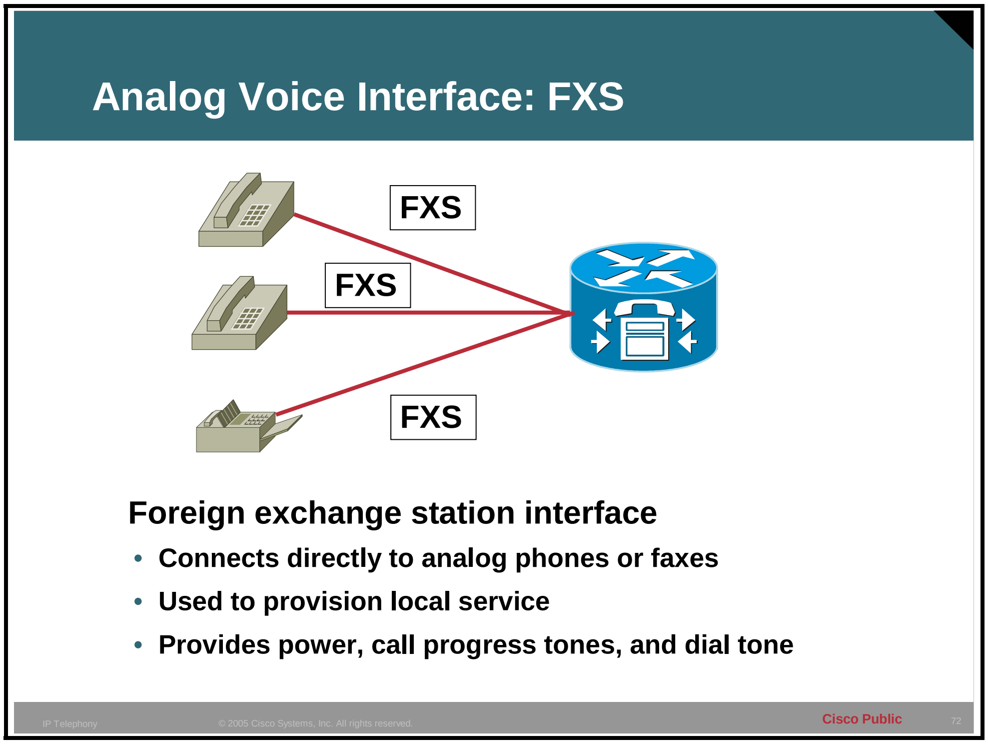


A subscriber home telephone connects to the telephone company central office (CO) via an electrical communication path called a local loop as depicted in the figure. The loop consists of a pair of twisted wires—one is called tip, the other is called ring.

In most arrangements, the ring wire ties to the negative side of a power source, called battery, while the tip wire connects to the ground. When the analog phone or fax goes to the off hook state, an electrical circuit is completed and current flows around the loop. This signals the switch that the analog phone or fax is off hook. The switch will then use a dial tone generator to send a signal that the switch is ready to receive digits. This signal is called dial tone. This wire pair which represents the local loop, along with all the others in your neighborhood, connects to the CO in a cable bundle, either buried underground or strung on poles.

Analog Voice Interfaces

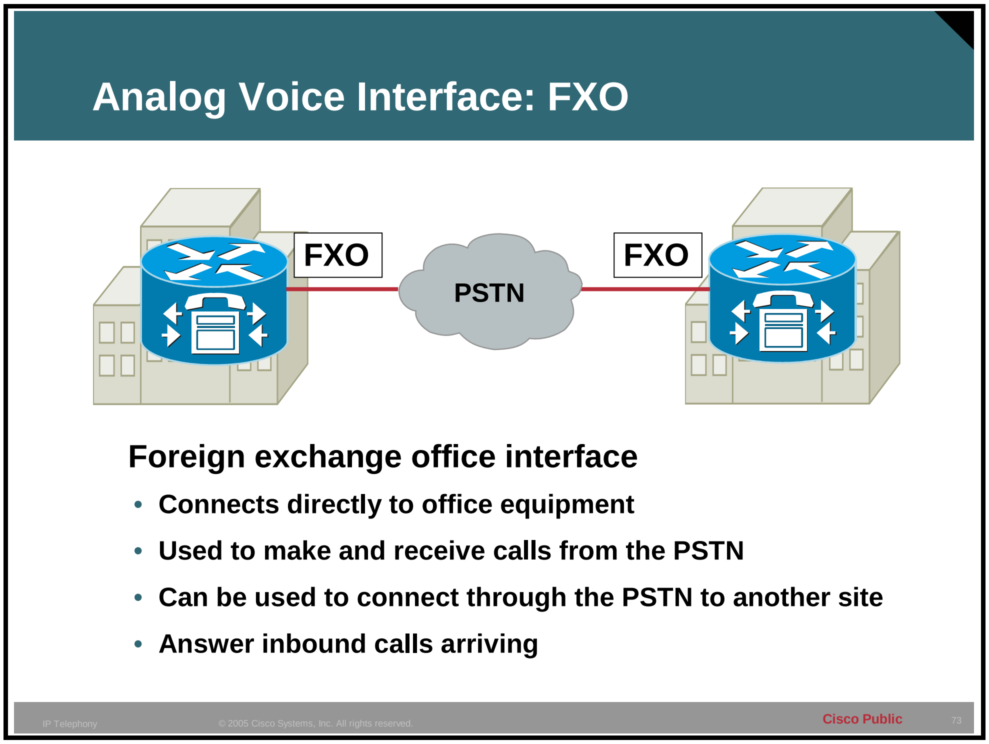
This topic defines the three analog interfaces that can be installed in a voice gateway: Foreign Exchange Station (FXS), Foreign Exchange Office (FXO), and ear and mouth (E&M). It also discusses how each of these interfaces is used.



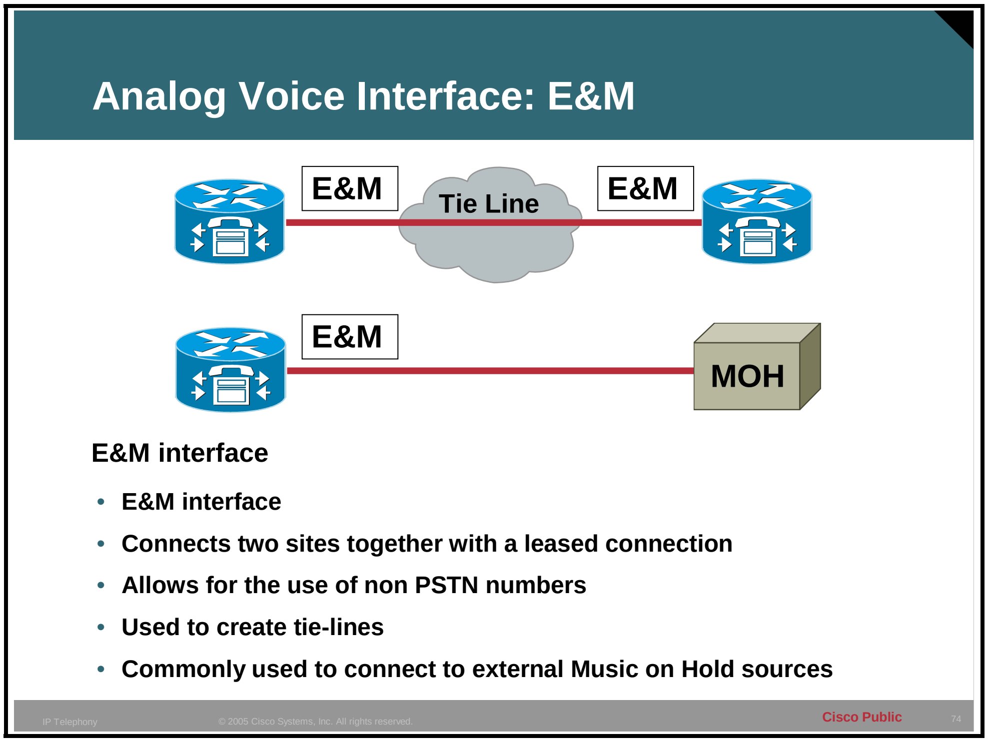
When analog phones or fax machines are used in an IP-based environment, they will need to have a connection into this IP network. This connection takes the form of an FXS interface. The FXS interface provides a direct connection to an analog telephone, a fax machine, or a similar device. From the analog device's perspective, the FXS interface functions like a switch. Therefore, it must supply line power, ring voltage, and dial tone.

The FXS interface contains the coder-decoder (codec), which converts the spoken analog voice wave into a digital format for processing by the voice-enabled device.

Note Analog phones plugged into an FXS port on the CallManager Express router cannot be forwarded to Unity Express voice mail. If voice mail is needed on the analog phones, use the ATA 186 or ATA 188 to connect the analog phone to the network.



For standard analog connections from the CO to enter the IP network, they will need to be terminated on an interface on a voice gateway. A Foreign Exchange Office (FXO) interface can be used for this termination of an analog line. When a call arrives, the FXO interface will answer the call and either present a second dial tone or be configured with a Private Line Auto Ringdown (PLAR). For outbound calls, the FXO interface provides either pulse or dual tone multifrequency (DTMF) digits for outbound dialing.



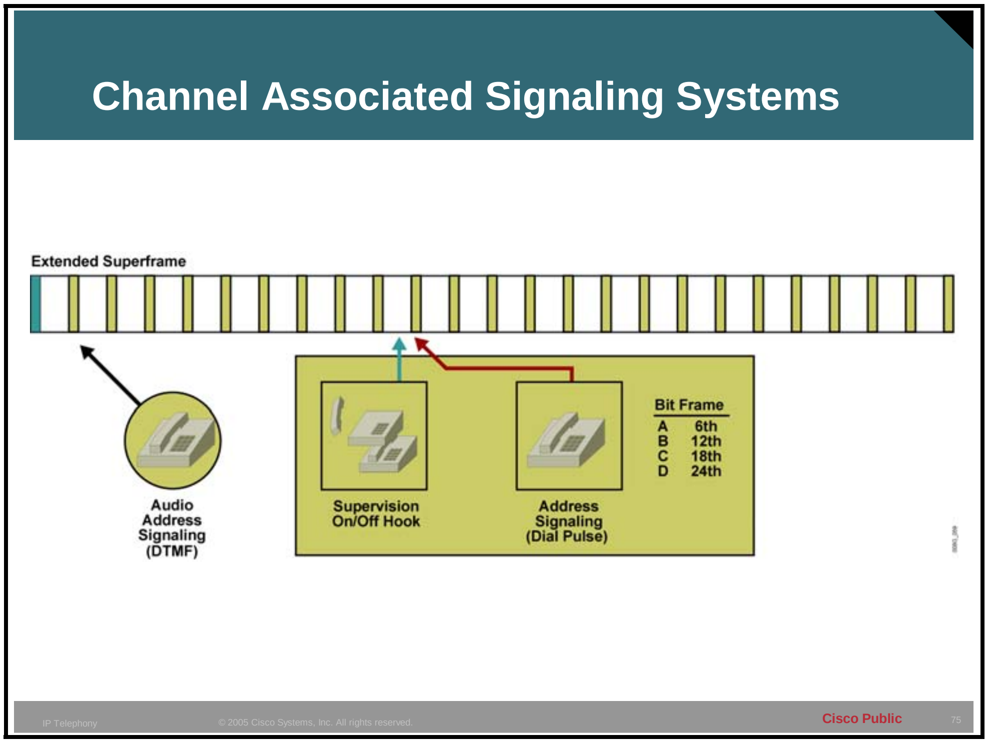
Special analog connections called tie lines can be leased from the carrier. These are typically use to tie two or mores sites together with analog connections. This tie line will terminate in an analog interface on the router so the analog communication can enter the IP network. The E&M interface on the router is where these tie lines can be terminated. E&M signaling is also referred to as “recEive and transMit,” but its origin comes from the term “earth and magneto.” Earth represents the electrical ground and magneto represents the electromagnet used to generate tone.

E&M signaling defines a trunk-circuit side and a signaling-unit side for each connection, similar to the DCE and DTE reference type. The router is usually the trunk-circuit side and the telco, CO, channel bank, or Cisco voice-enabled platform is the signaling-unit side.

Note	Many music on hold services provide an analog E&M interface that may be used to connect to the CallManager Express router.

Channel Associated Signaling Systems: T1

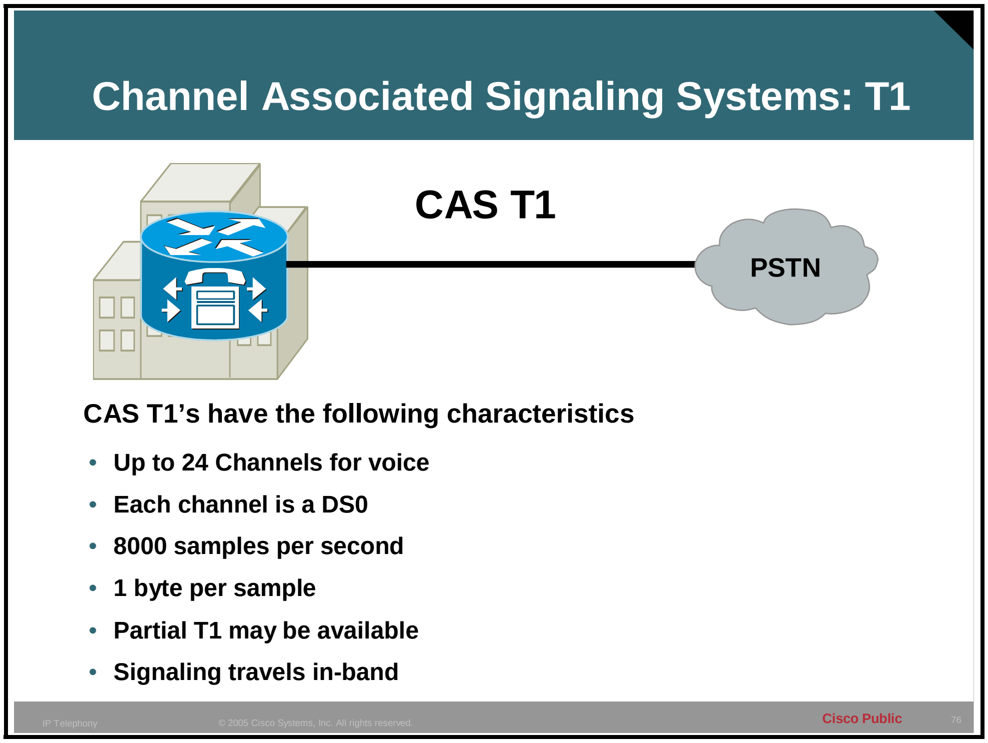
This topic describes CAS and its uses with T1 transmission.



Because the signaling occurs within each DS0, it is referred to as in band. Also, because the use of these bits is exclusively reserved for signaling each respective voice channel, it is referred to as CAS.

SF has a 12-frame structure and provides AB bits for signaling. ESF has a 24-frame structure and provides ABCD bits for signaling.

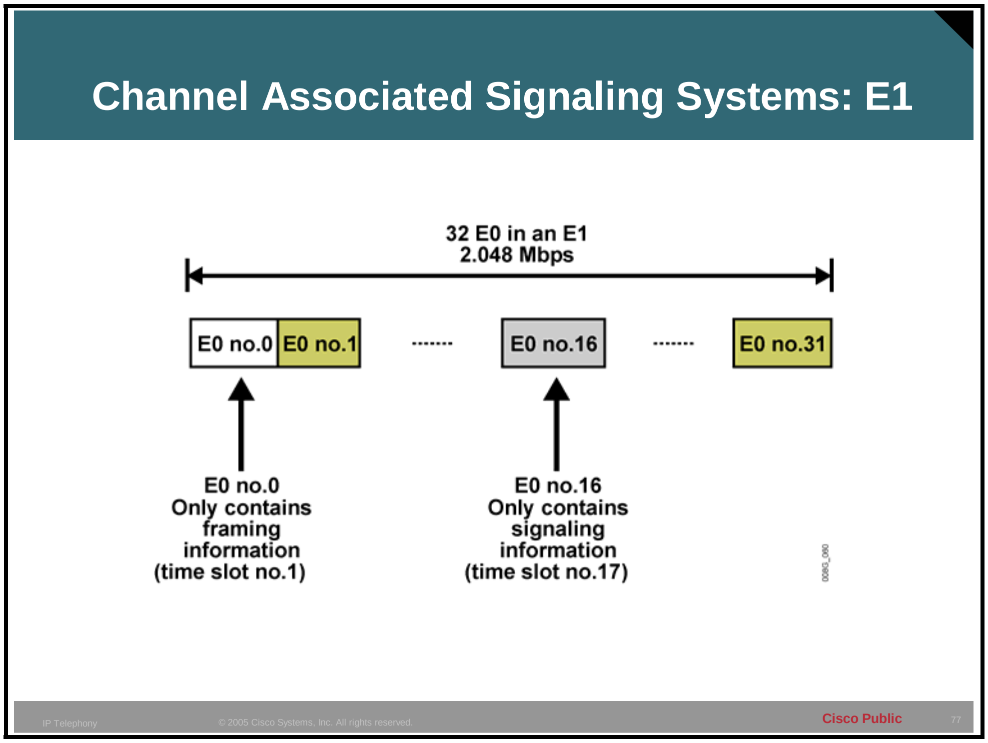
Tones, such as dual tone multifrequency (DTMF) addressing or call progress, can be carried in the audio path. However, other CAS signals must be carried via the robbed bits.



CallManager Express can be connected to the PSTN through a CAS T1 connection. This will provide up to 24 channels for voice. Each channel is a 64 kb DS0.

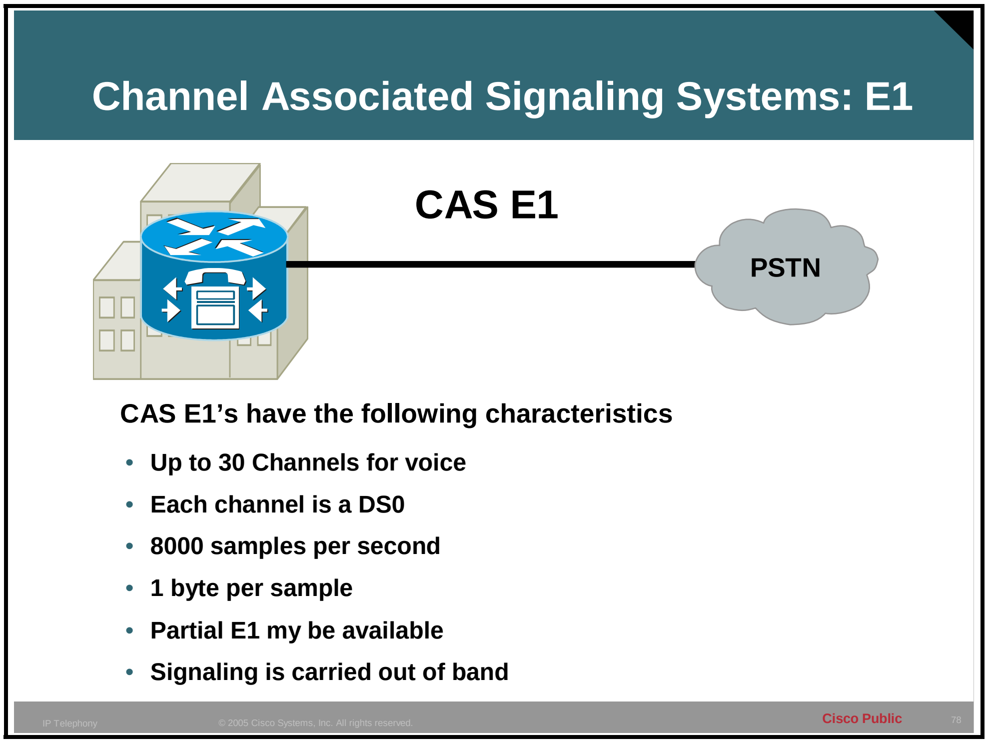
Channel Associated Signaling Systems: E1

This topic describes CAS and its uses with E1 transmission.



In E1 framing and signaling, 30 of the 32 available channels, or time slots, are used for voice or data. Framing information uses time slot 1 (channel 0), while time slot 17 (channel 16) is used for signaling by all the other time slots. This signaling format is also known as CAS because each bearer channel has specific bits in the 17th timeslot assigned for signaling. However, this implementation of CAS is considered out of band because the signaling bits are not carried within the voice channel, as is the case with T1.

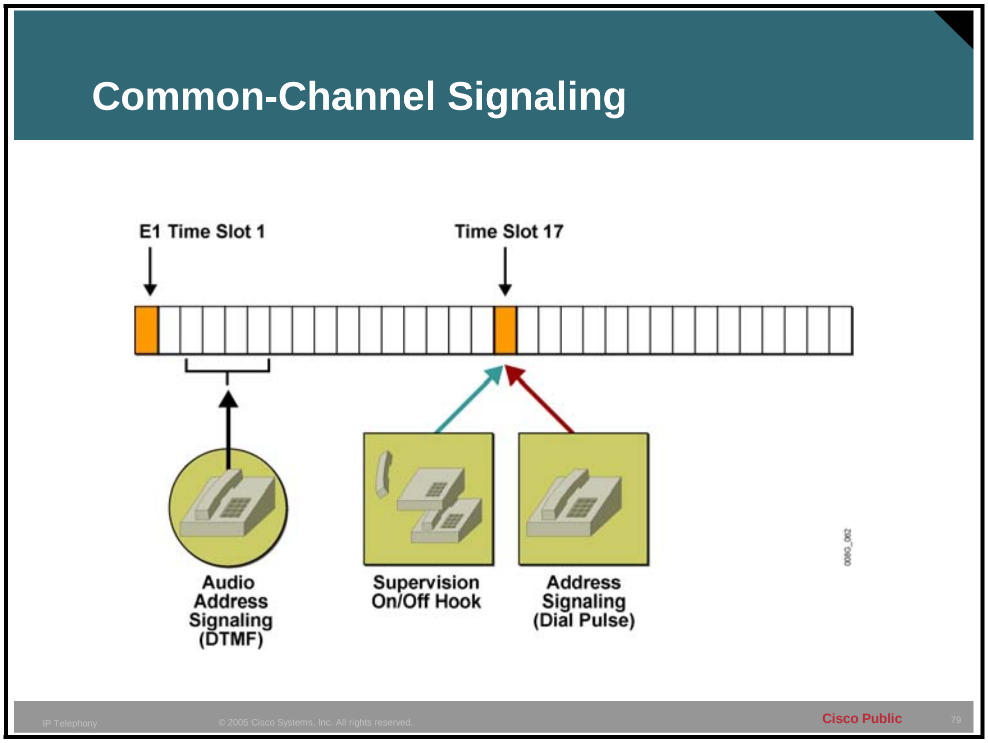
Note	Robbed bit signaling is not used in E1 circuits.



CallManager Express can be connected to the PSTN and can provide up to 30 channels for voice.

Common Channel Signaling Systems

This topic describes common channel signaling (CCS) systems.



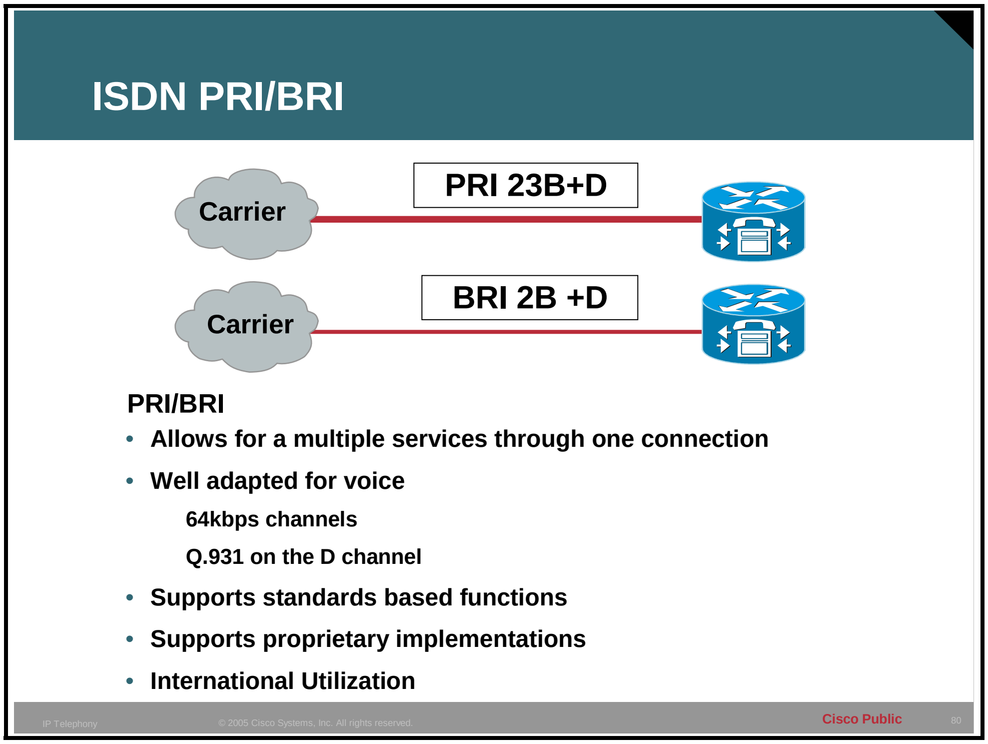
Whereas CAS uses bit time slots assigned to each specific channel, CCS uses a common channel and protocol to setup calls for all the bearer channels. Using ISDN over E1 as an example, the signaling protocol Q931 would use timeslot 17 to exchange call-setup messages for any of the 30 bearer (B) channels.

Examples of CCS signaling are as follows:

- **Proprietary implementations:** Some PBX vendors choose to implement a proprietary CCS protocol between their PBXs for T1 and E1. In this implementation, Cisco devices are configured for Transparent-Common Channel Signaling (T-CCS) because they do not understand proprietary signaling information and must simply transport the signaling without modification or interpretation.
- **ISDN:** Uses Q.931 in a common channel to signal all other channels
- **Digital Private Network Signaling System (DPNSS):** An open standard developed by British Telecom for implementation by any vendor who chooses to use it. DPNSS also uses a common channel to signal all other channels.
- **Q Signaling (QSIG):** Like ISDN, uses a common channel to signal all other channels.

PRI/BRI

This topic describes primary rate interface and basic rate interface and how they may be used to support voice.



One form of Common Channel Signaling is Integrated Services Digital Network (ISDN). Primary rate interface (PRI) and basic rate interface (BRI) are the two ways of implementing ISDN.

Note	Because ISDN is a digital service, the time required to setup a call is significantly less than that of an analog call.

PRI supports 23 (for T1) or 30 (for E1) bearer (B) channels while BRI features two B channels. Each implementation also supports a single data (D) channel used to carry signaling information.

The following are characteristics of ISDN PRI/BRI:

- ISDN channels may carry data, voice, or video.
- Each B channel is 64 kbps, and G.711 pulse code modulation (PCM) requires 64 kbps so this is a perfect match for voice applications.
- The D channel in BRI is 16kbps and in PRI is 64kbps
- ISDN has a built-in call-control protocol known as International Telecommunication Union Telecommunication Standardization Sector (ITU-T) Q.931 that runs on the D channel
- ISDN can support standards-based voice features, such as call forwarding and standards-based enhanced dialup capabilities, such as Group IV fax and audio channels
- ISDN may carry vendor specific PBX features

- ISDN BRI voice is commonly used in Europe; ISDN PRI voice is used worldwide.

