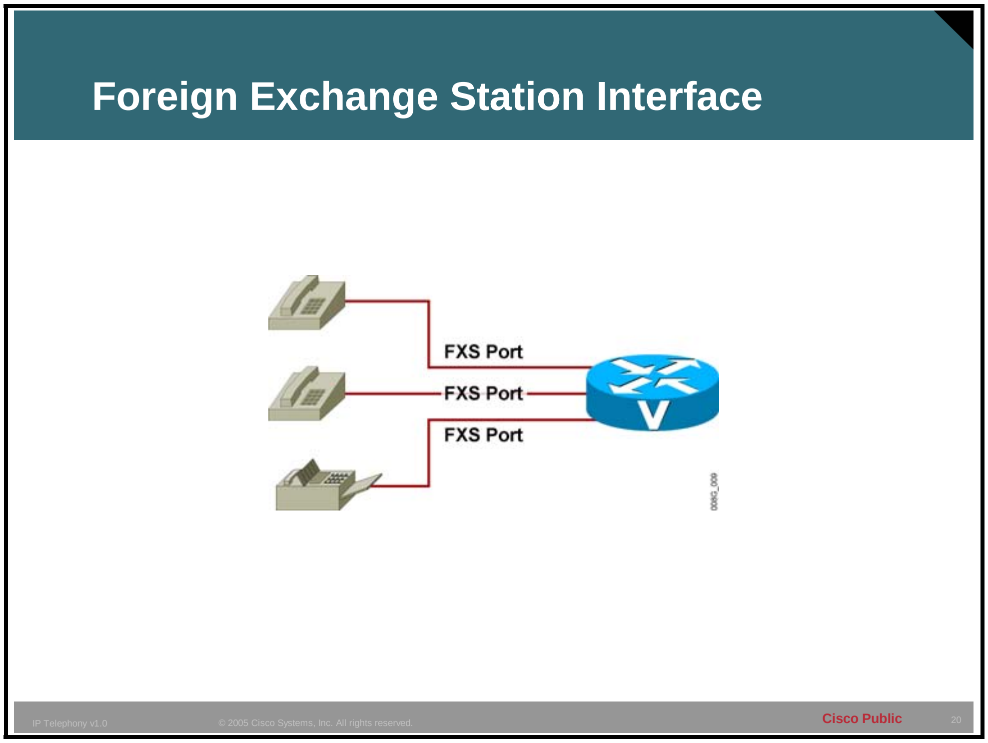
IP Telephony Applications

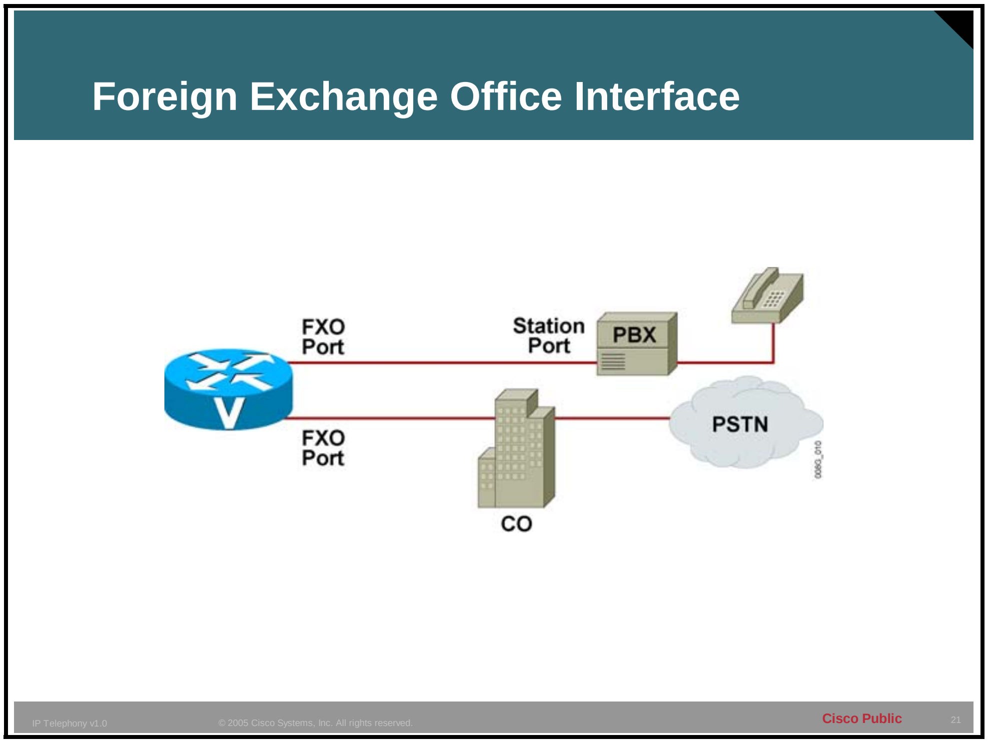
Analog Interfaces

This topic defines three analog interfaces: Foreign Exchange Station (FXS), Foreign Exchange Office (FXO), and ear and mouth (E&M). It also discusses how each of these interfaces is used.



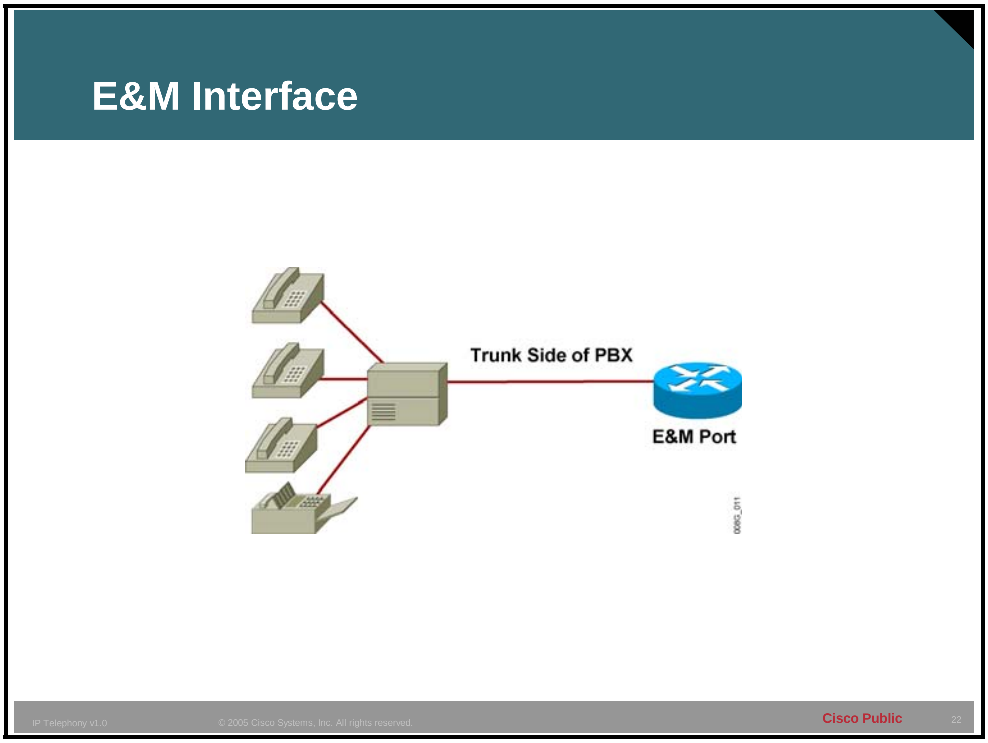
This figure depicts an FXS interface. The FXS interface provides a direct connection to an analog telephone, a fax machine, or a similar device. From a telephone perspective, the FXS interface functions like a switch; therefore, it must supply line power, ring voltage, and dial tone.

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.



This figure depicts an FXO interface. The FXO interface allows an analog connection to be directed at the CO of a PSTN or to a station interface on a PBX. The switch recognizes the FXO interface as a telephone because the interface plugs directly into the line side of the switch. The FXO interface provides either pulse or DTMF digits for outbound dialing.

In PSTN terminology, an FXO-to-FXS connection is also referred to as a foreign exchange (FX) trunk. An FX trunk is a CO trunk that has access to a distant CO. Because this connection is FXS at one end and FXO at the other end, it acts as a long-distance extension of a local telephone line. In this instance, a local user can pick up the telephone and get a dial tone from a foreign city. Users in the foreign city can dial a local number and have the call connect to the user in the local city.

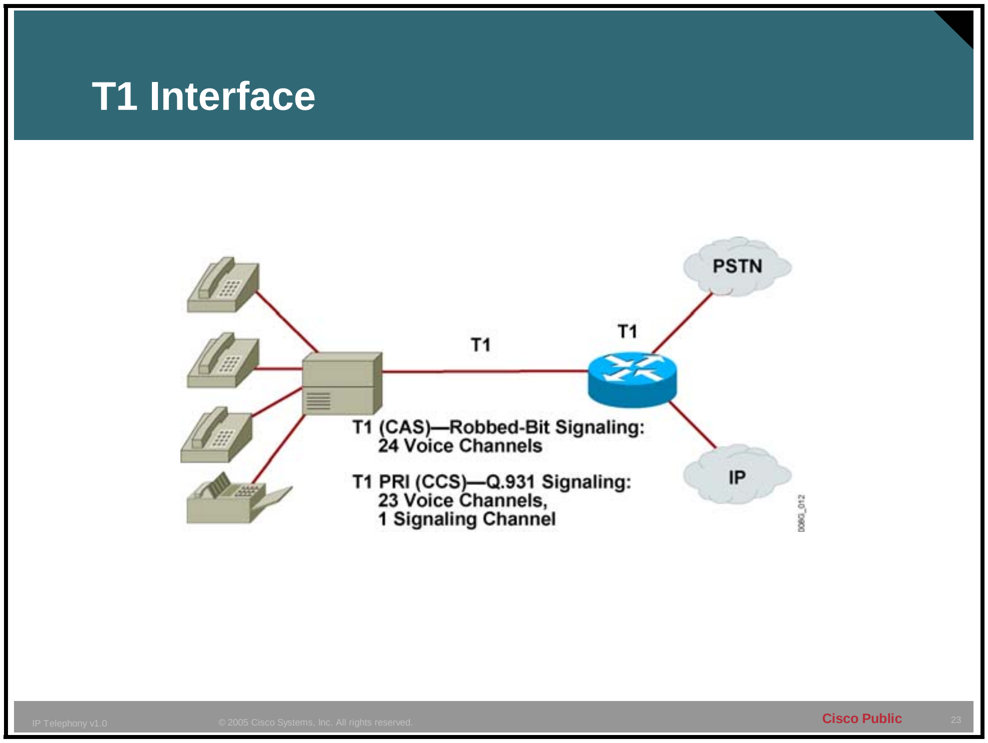


This figure depicts an E&M interface. The E&M interface provides signaling for analog trunking. Analog trunk circuits connect automated systems (PBXs) and networks (COs). E&M signaling is also referred to as “ear and mouth,” but its origin comes from the term “Earth and Magneto.” Earth represents the 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 types. The PBX is usually the trunk-circuit side and the telco, CO, channel bank, or Cisco voice-enabled platform is the signaling-unit side.

Digital Interfaces

This topic describes the three basic digital voice interfaces: T1, E1, and BRI.



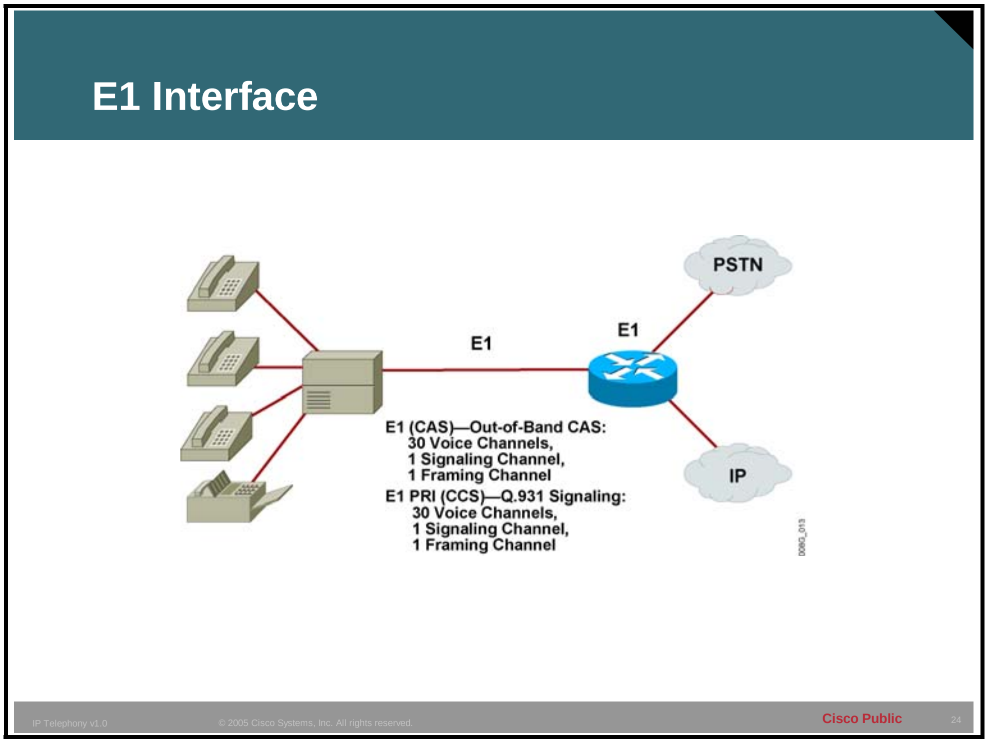
This figure depicts a T1 interface. In a corporate environment with a large volume of voice traffic, connections to the PSTN and to PBXs are primarily digital.

A T1 interface is a form of digital connection that can simultaneously carry up to 24 conversations using two-wire pairs. When a T1 link operates in full-duplex mode, one wire pair sends and the other wire pair receives. The 24 channels are grouped together to form a frame. The frames are then grouped together into Super Frames (groups of 12 frames) or into Extended Superframes (groups of 24 frames).

The T1 interface carries either CAS or CCS. When a T1 interface uses CAS, the signaling robs a sampling bit for each channel to convey in band. When a T1 interface uses CCS, Q.931 signaling is used on a single channel, typically the last channel.

To configure CAS you must:

- Specify the type of signaling that the robbed bits carry; for example, E&M Wink Start. This signaling must match the PSTN requirements or the PBX configuration. This is considered in-band signaling because the signal shares the same channel as the voice.
- Configure the interface for PRI signaling. This level of configuration makes it possible to use channels 1 to 23 (called B channels) for voice traffic. Channel 24 (called the D channel) carries the Q.931 call control signaling for call setup, maintenance, and teardown. This type of signaling is considered out-of-band signaling because the Q.931 messages are sent in the D channel only.



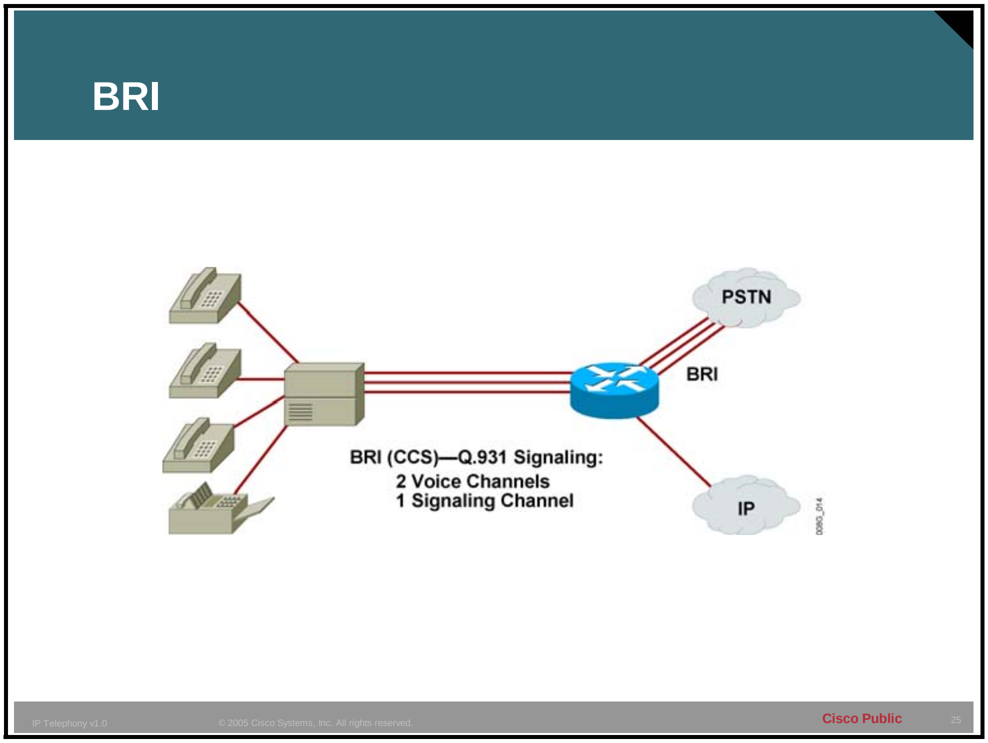
This figure depicts an E1 interface. An E1 interface has 32 channels and simultaneously carries up to 30 conversations. The other two channels are used for framing and signaling. The 32 channels are grouped to form a frame. The frames are then grouped together into multiframes (groups of 16 frames). Europe and Mexico use the E1 interface.

Although you can configure the E1 interface for either CAS or CCS, the most common usage is CCS.

When an E1 interface uses CAS, signaling travels out of band in the signaling channel but follows a strict association between the signal carried in the signaling channel and the channel to which the signaling is being applied. The signaling channel is channel 16.

In the first frame, channel 16 carries 4 bits of signaling for channel 1 and 4 bits of signaling for channel 17. In the second frame, channel 16 carries 4 bits of signaling for channel 2 and 4 bits for channel 18, and so on. This process makes it out-of-band CAS.

When an E1 interface uses CCS, Q.931 signaling is used on a single channel, typically channel 17. When configuring for CCS, configure the interface for PRI signaling. When E1 is configured for CCS, channel 16 carries Q.931 signaling messages only.

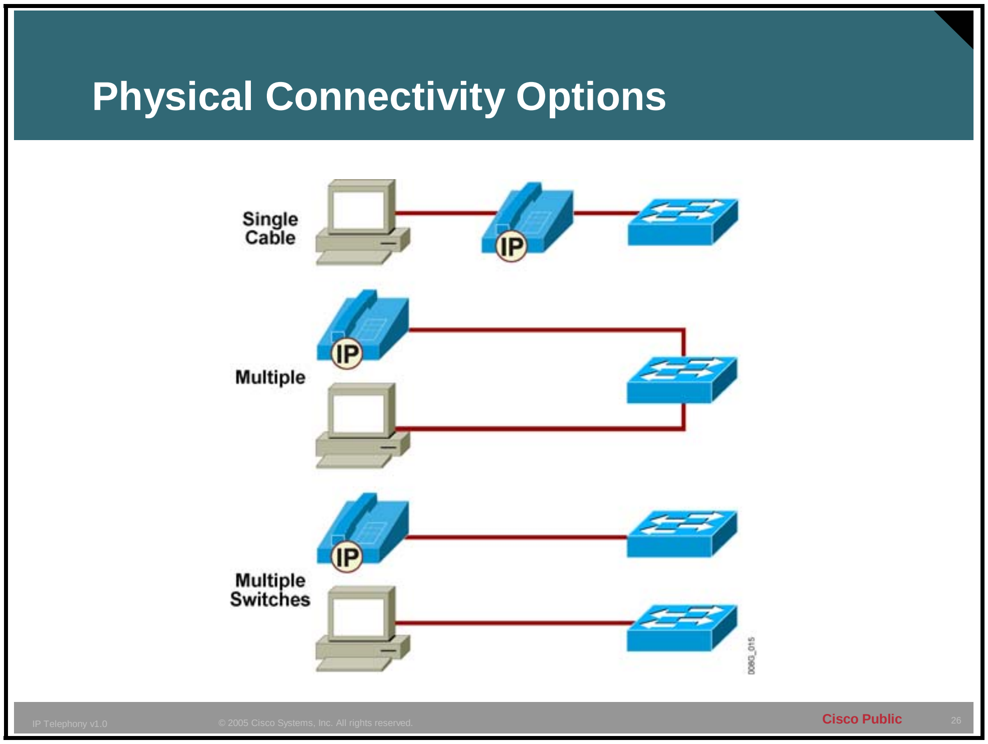


This figure depicts a Basic Rate Interface (BRI). You can use a BRI to connect the PBX voice into the network. Used primarily in Europe for PBX connectivity, BRI provides a 16-kbps D channel for signaling and two 64-kbps B channels for voice. BRI uses Q.931 signaling in the D channel for call signaling.

Note	Cisco Systems does not officially support ISDN telephones.

IP Phones

This topic describes scenarios for desktop telephone connections and the IP Phone built-in switch ports.



This figure depicts physical connection options for IP Phones. The IP Phone connects to the network through a Category 5 or better cable that has RJ-45 connectors. The power-enabled switch port or an external power supply provides power to an IP Phone. The IP Phone functions like other IP-capable devices sending IP packets to the IP network. Because these packets are carrying voice, you must consider both logical and physical configuration issues.

At the physical connection level, there are three options for connecting the IP Phone:

- **Single cable:** A single cable connects the telephone and the PC to the switch. Most enterprises install IP Phones on their networks using a single cable for both the telephone and a PC. Reasons for using a single cable include ease of installation and cost savings on cabling infrastructure and wiring-closet switch ports.
- **Multiple cables:** Separate cables connect the telephone and the PC to the switch. Users often connect the IP Phone and PC using separate cables. This connection creates a physical separation between the voice and data networks.
- **Multiple switches:** Separate cables connect the telephone and the PC to separate switches. With this option, IP Phones are connected to separate switches in the wiring closet. By using this approach, you can avoid the cost of upgrading the current data switches and keep the voice and data networks completely separate.

Multiple switches are used to do the following:

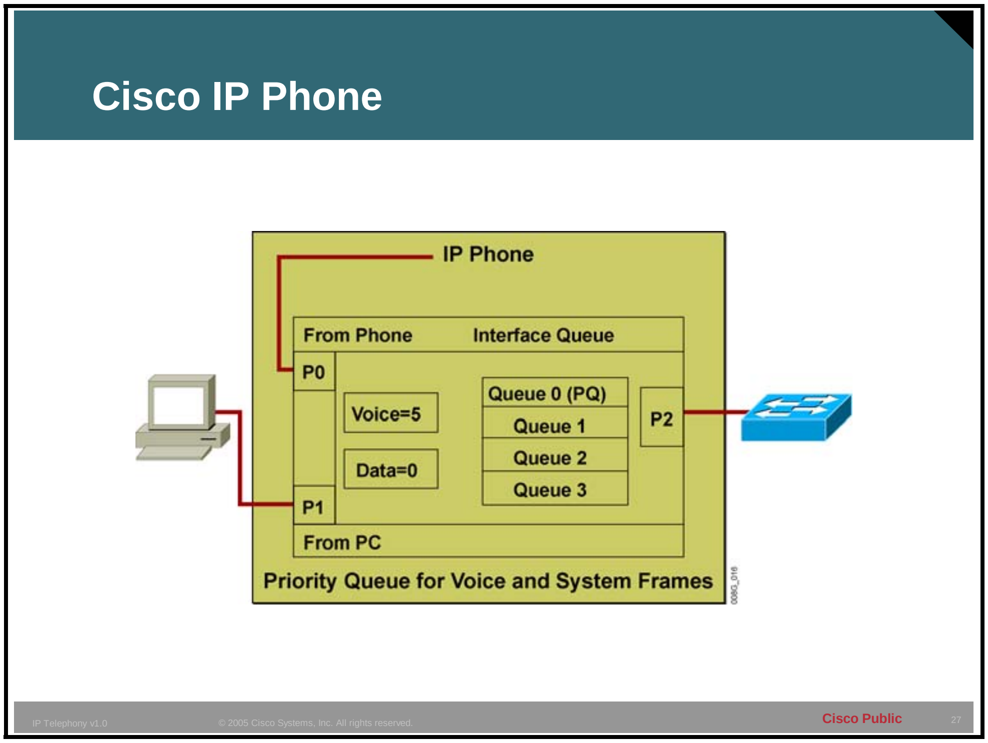
- Provide inline power to IP Phones without having to upgrade the data infrastructure
- Limit the number of switches that need an uninterruptible power supply (UPS)
- Reduce the amount of Cisco IOS Catalyst software upgrades needed in the network
- Limit the spanning-tree configuration in the wiring-closet switches

The physical configuration for connecting an IP Phone must address the following issues:

- Speed and duplex settings
- Inline power settings

The logical configuration for connecting an IP Phone must address the following issues:

- IP addressing
- VLAN assignment
- Spanning tree
- Classification and queuing



The basic function of a Cisco IP Phone depends on a three-port 10/100 switch. Port P0 is an internal port that connects the voice electronics in the telephone. Port P1 connects a daisy-chained PC. Port P2 uplinks to the Ethernet switch in the wiring closet.

Each port contains four queues with a single threshold. One of these queues is a high-priority queue used for system frames. By default, voice frames are classified for processing in the high-priority queue, and data frames are classified for processing in the low-priority queue.

The internal Ethernet switch on the Cisco IP Phone switches incoming traffic to either the access port or the network port.

If a computer is connected to the port P1, data packets traveling to and from the computer, and to and from the phone, share the same physical link to the access layer switch connected to port P2, and to the same port on the access layer switch. This shared physical link has the following implications for the VLAN network configuration:

- Current VLANs may be configured on an IP subnet basis. However, additional IP addresses may not be available for assigning the telephone to the same subnet as the other devices that are connected to the same port.
- Data traffic that is supporting phones on the VLAN may reduce the quality of VoIP traffic.

You can resolve these issues by isolating the voice traffic on a separate VLAN for each of the ports connected to a telephone. The switch port configured for connecting a telephone would have separate VLANs configured to carry the following types of traffic:

- Voice traffic to and from the IP Phone (auxiliary VLAN)
- Data traffic to and from the PC connected to the switch through the IP Phone access port (native VLAN)

Isolating the telephones on a separate auxiliary VLAN increases voice-traffic quality and allows a large number of telephones to be added to an existing network that has a shortage of IP addresses.

Note	For more information, refer to the documentation included with the Cisco Catalyst switch.

Example: IP Phone Installations

Cisco IP Phones deployed in an office environment attach to Ethernet switches. The IP Phone uses the existing cable infrastructure, or the infrastructure is updated to allow one connection for the phone and one for the desktop PC. The connections from the phone and the PC may lead to the same switch or to different switches. In either case, the IP Phone has the capability to prioritize voice frames.

