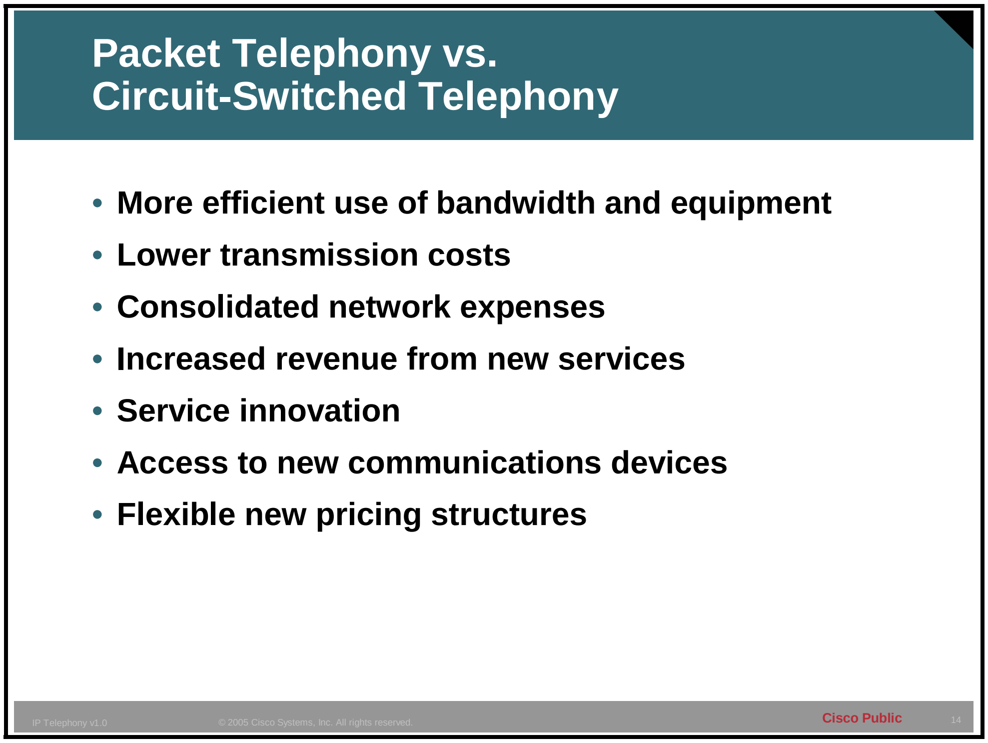
Packetized Telephony Networks

Benefits of Packet Telephony Networks

Traditionally, the potential savings on long-distance costs was the driving force behind the migration to converged voice and data networks. The cost of long-distance calls has dropped in recent years, and other factors have come to the forefront as benefits of converged networks. This topic describes some of these benefits.



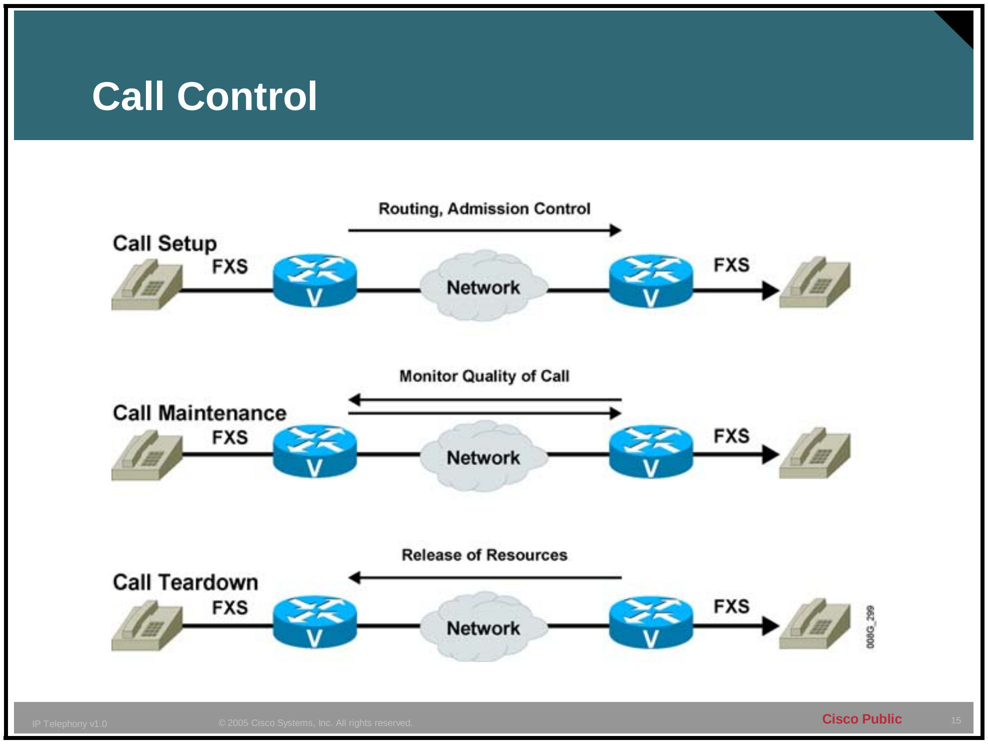
The benefits of packet telephony versus circuit-switched telephony are as follows:

- **More efficient use of bandwidth and equipment:** Traditional telephony networks use a 64-kbps channel for every voice call. Packet telephony shares bandwidth among multiple logical connections and offloads traffic volume from existing voice switches.
- **Lower costs for telephony network transmission:** A substantial amount of equipment is needed to combine 64-kbps channels into high-speed links for transport across the network. Packet telephony statistically multiplexes voice traffic alongside data traffic. This consolidation represents substantial savings on capital equipment and operations costs.
- **Consolidated voice and data network expenses:** Data networks that function as separate networks to voice networks become major traffic carriers. The underlying voice networks are converted to utilize the packet-switched architecture to create a single integrated communications network with a common switching and transmission system. The benefit is significant cost savings on network equipment and operations.

- **Increased revenues from new services:** Packet telephony enables new integrated services, such as broadcast-quality audio, unified messaging, and real-time voice and data collaboration. These services increase employee productivity and profit margins well above those of basic voice services. In addition, these services enable companies and service providers to differentiate themselves and improve their market position.
- **Greater innovation in services:** Unified communications use the IP infrastructure to consolidate communication methods that were previously independent; for example, fax, voice mail, e-mail, wireline telephones, cellular telephones, and the web. The IP infrastructure provides users with a common method to access messages and initiate real-time communications—independent of time, location, or device.
- **Access to new communications devices:** Packet technology can reach devices that are largely inaccessible to the TDM infrastructures of today. Examples of such devices are computers, wireless devices, household appliances, personal digital assistants, and cable set-top boxes. Intelligent access to such devices enables companies and service providers to increase the volume of communications they deliver, the breadth of services they offer, and the number of subscribers they serve. Packet technology, therefore, enables companies to market new devices, including videophones, multimedia terminals, and advanced IP Phones.
- **Flexible new pricing structures:** Companies and service providers with packet-switched networks can transform their service and pricing models. Because network bandwidth can be dynamically allocated, network usage no longer needs to be measured in minutes or distance. Dynamic allocation gives service providers the flexibility to meet the needs of their customers in ways that bring them the greatest benefits.

Call Control

Call control allows users to establish, maintain, and disconnect a voice flow across a network. This topic describes call control services.

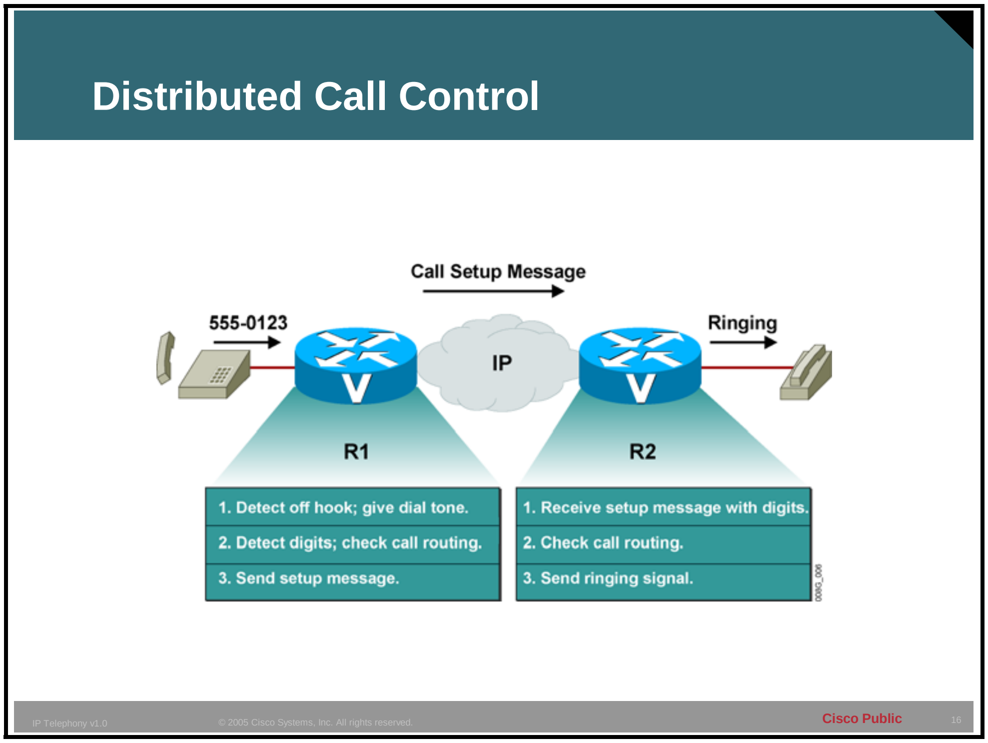


Although different protocols address call control in different ways, they all provide a common set of services. The following are basic components of call control:

- **Call setup:** Checks call-routing configuration to determine the destination of a call. The configuration specifies the bandwidth requirements for the call. When the bandwidth requirements are known, Call Admission Control (CAC) determines if sufficient bandwidth is available to support the call. If bandwidth is available, call setup generates a setup message and sends it to the destination. If bandwidth is not available, call setup notifies the initiator by presenting a busy signal. Different call control protocols, such as H.323, Media Gateway Control Protocol (MGCP), and session initiation protocol (SIP), define different sets of messages to be exchanged during setup.
- **Call maintenance:** Tracks packet count, packet loss, and interarrival jitter or delay when the call is set up. Information passes to the voice-enabled devices to determine if connection quality is good or if it has deteriorated to the point where the call should be dropped.
- **Call teardown:** Notifies voice-enabled devices to free resources and make them available for the next call when either side terminates a call.

Distributed vs. Centralized Call Control

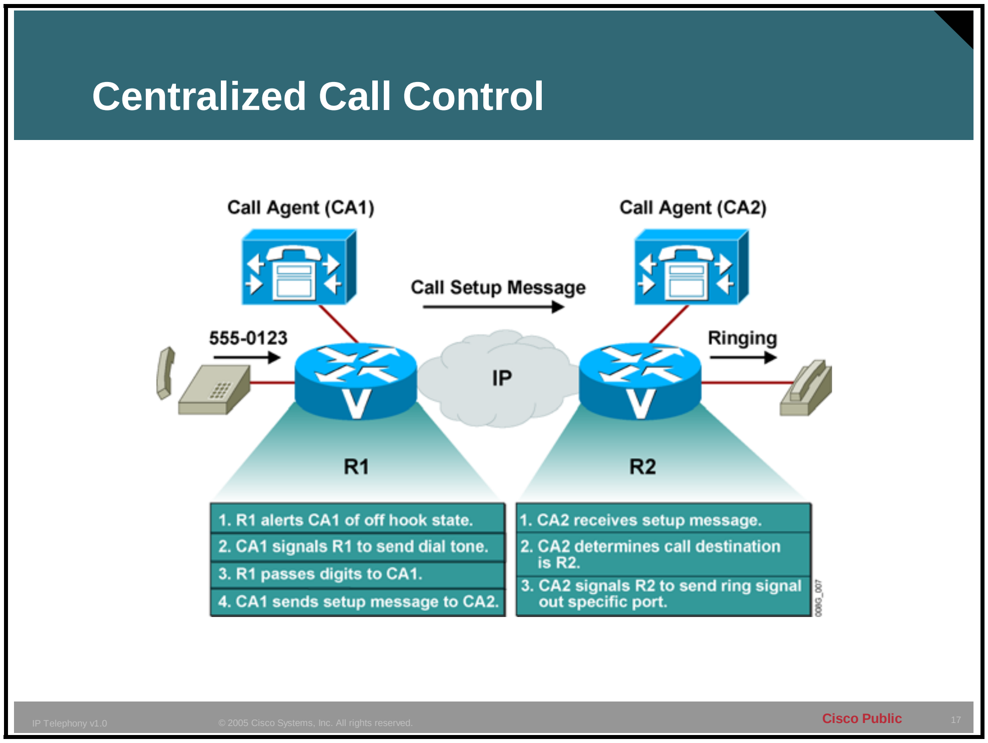
This topic compares distributed and centralized call control.



This figure shows an environment where call control is handled by multiple components in the network. Distributed call control is possible where the voice-capable device is configured to support call control directly. This is the case with a voice gateway when protocols, such as H.323 or SIP, are enabled on the device.

Distributed call control enables the gateway to perform the following procedure:

1. Recognize the request for service
2. Process dialed digits
3. Route the call
4. Supervise the call
5. Terminate the call



Centralized call control allows an external device (call agent) to handle the signaling and call processing, leaving the gateway to translate audio signals into voice packets after call setup. The call agent is responsible for all aspects of signaling, thus instructing the gateways to send specific signals at specific times.

When the call is set up:

- The voice path runs directly between the two gateways and does not involve the call agent.
- When either side terminates the call, the call agent signals the gateways to release resources and wait for another call.

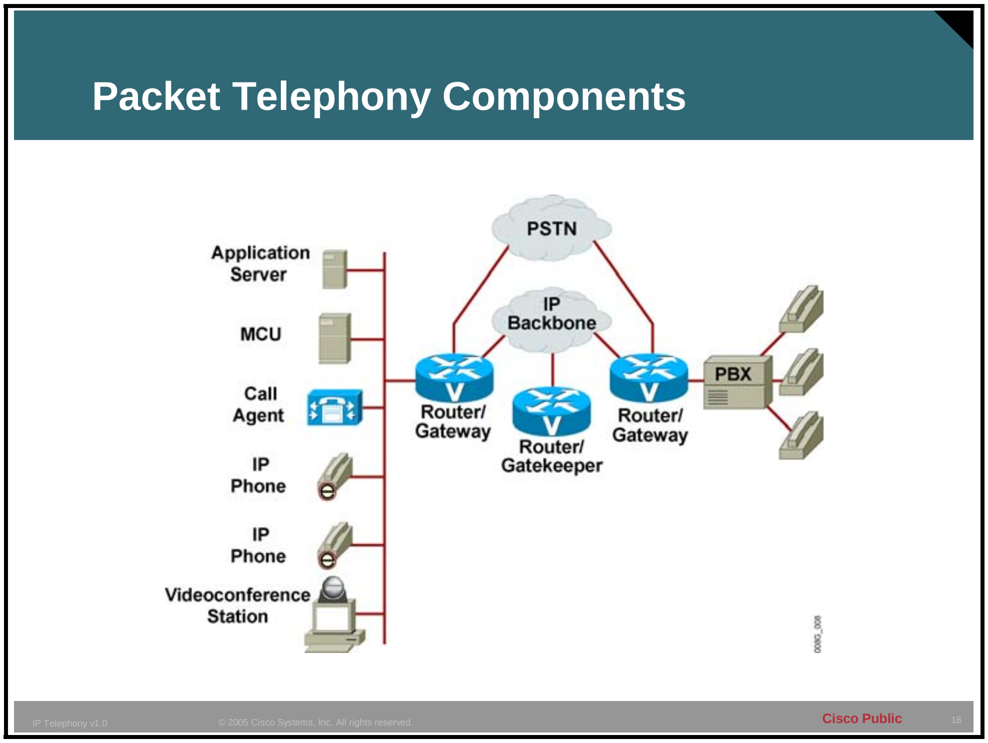
The use of centralized call control devices is beneficial in several ways:

- It centralizes the configuration for call routing and CAC. In a large voice environment, centralization can be extremely beneficial.
- The call agent is the only device that needs the intelligence to understand and participate in call control functions. These call control functions enable the customer to purchase less expensive voice-gateway devices and point to a single device to handle call control.

MGCP is one example of a centralized call control model.

Packet Telephony Components

This topic introduces the basic components of a packet voice network.



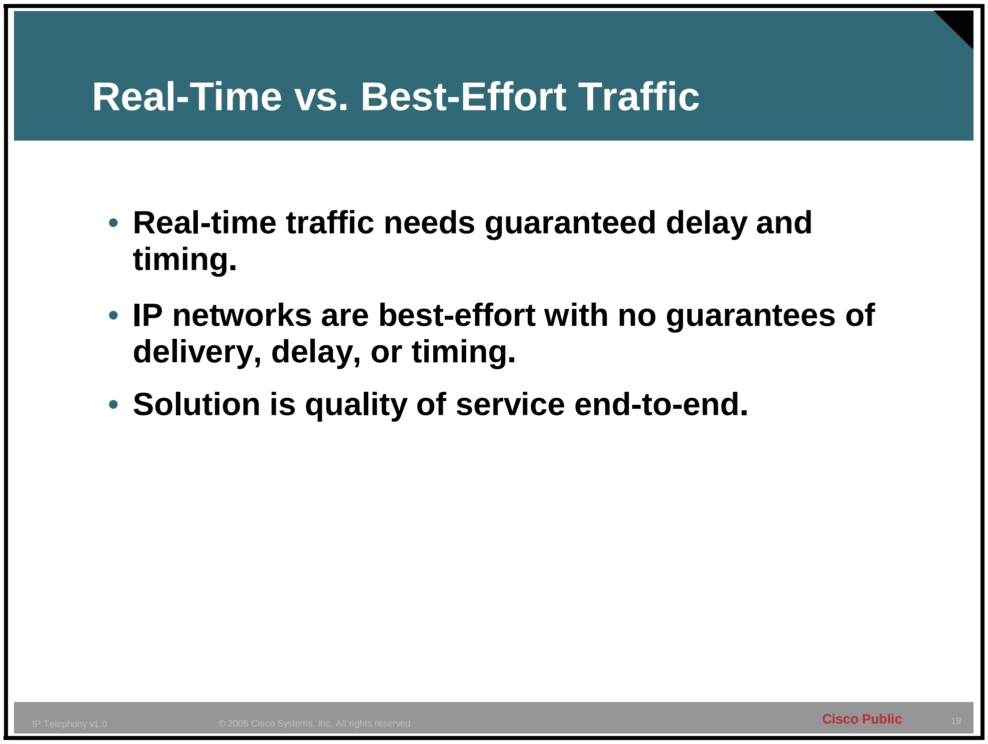
The basic components of a packet voice network include the following:

- **IP Phones:** Provide IP voice to the desktop.
- **Gatekeeper:** Provides CAC, bandwidth control and management, and address translation.
- **Gateway:** Provides translation between VoIP and non-VoIP networks, such as the PSTN. It also provides physical access for local analog and digital voice devices, such as telephones, fax machines, key sets, and PBXs.
- **Multipoint control unit (MCU):** Provides real-time connectivity for participants in multiple locations to attend the same videoconference or meeting.
- **Call agent:** Provides call control for IP Phones, CAC, bandwidth control and management, and address translation.
- **Application servers:** Provide services such as voice mail, unified messaging, and Cisco CallManager Attendant Console.
- **Videoconference station:** Provides access for end-user participation in videoconferencing. The videoconference station contains a video capture device for video input and a microphone for audio input. The user can view video streams and hear the audio that originates at a remote user station.

Other components, such as software voice applications, interactive voice response (IVR) systems, and softphones, provide additional services to meet the needs of enterprise sites.

Best-Effort Delivery of Real-Time Traffic

Voice and data can share the same medium; however, their traffic characteristics differ widely: voice is real-time traffic and data is typically sent as best-effort traffic. This topic compares real-time requirements versus best-effort delivery.



Traditional telephony networks were designed for real-time voice transmission, and therefore cater to the need for a constant voice flow over the connection. Resources are reserved end to end on a per-call basis and are not released until the call is terminated. These resources guarantee that voice flows in an orderly manner. Good voice quality depends on the capacity of the network to deliver voice with guaranteed delay and timing—the requirement for delivery of real-time traffic.

Traditional data networks were designed for best-effort packet transmission. Packet Telephony Networks transmit with no guarantee of delivery, delay, or timing. Data handling is effective in this scenario because upper-layer protocols, such as TCP, provide for reliable—although untimely—packet transmission. TCP trades delay for reliability. Data can typically tolerate a certain amount of delay and is not affected by interpacket jitter.

A well-engineered, end-to-end network is required when converging delay-sensitive traffic, such as VoIP, with best-effort data traffic. Fine-tuning the network to adequately support VoIP involves a series of protocols and features to improve quality of service (QoS). Because the IP network is, by default, best-effort, steps must be taken to ensure proper behavior of both the real-time and best-effort traffic. Packet Telephony Networks succeed, in large part, based on the QoS parameters that are implemented networkwide.

