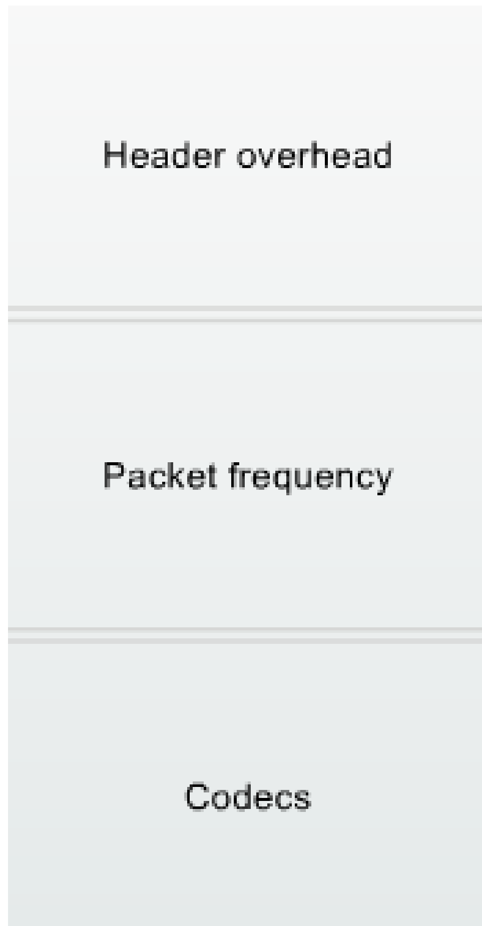
Voice Over IP

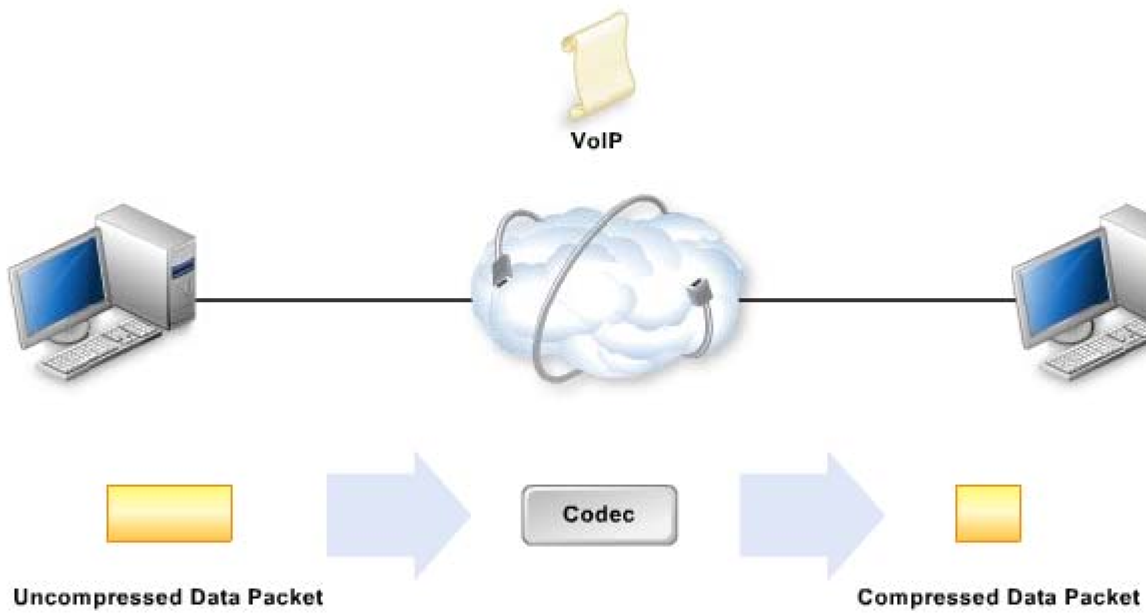
VoIP Setup

Bandwidth



VoIP Setup

Bandwidth Compression

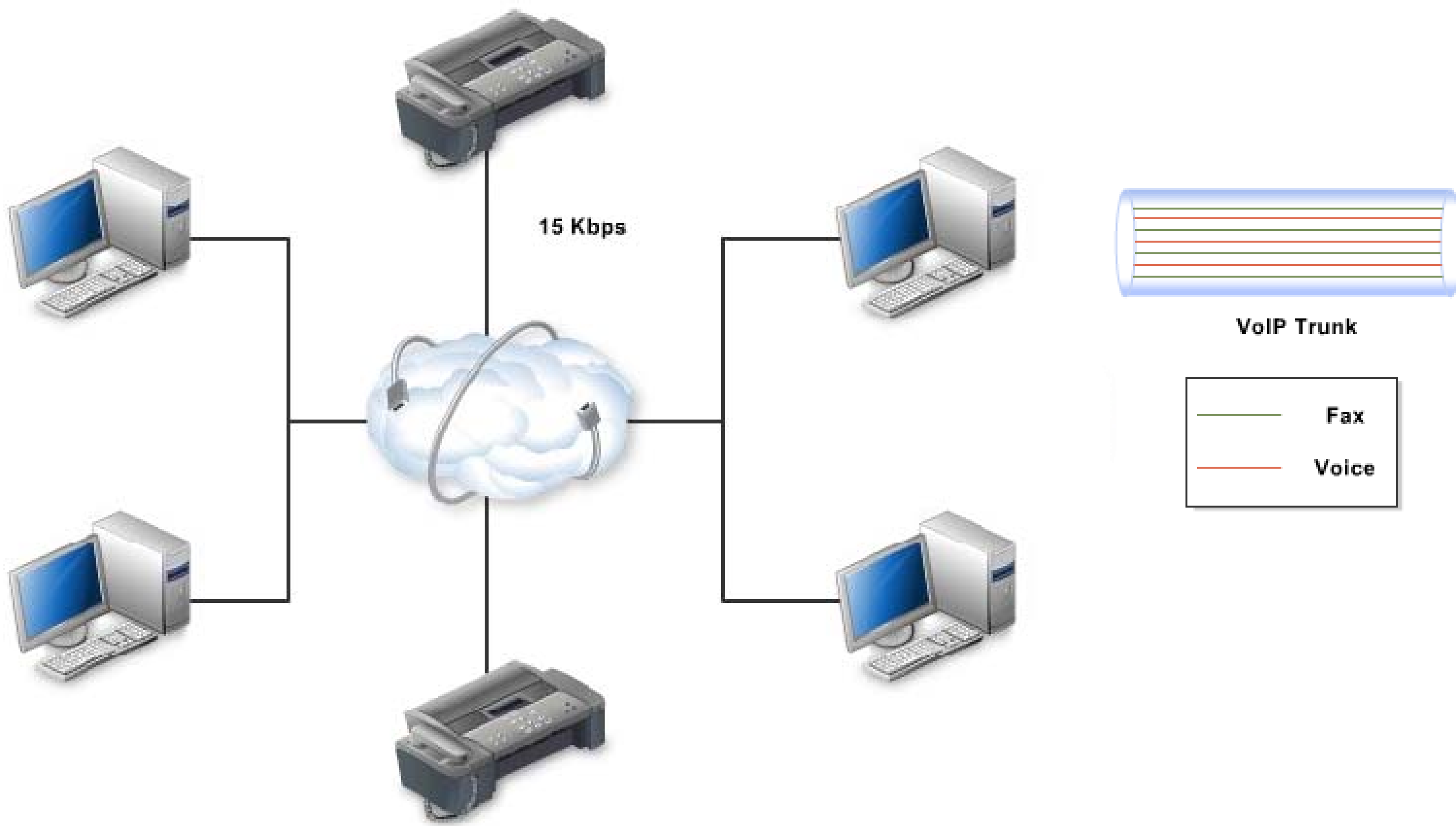


Benefits of Compression:

- Increases speed
- Optimizes available bandwidth
- Less delay on the network

VoIP Setup: Carrying Capacity

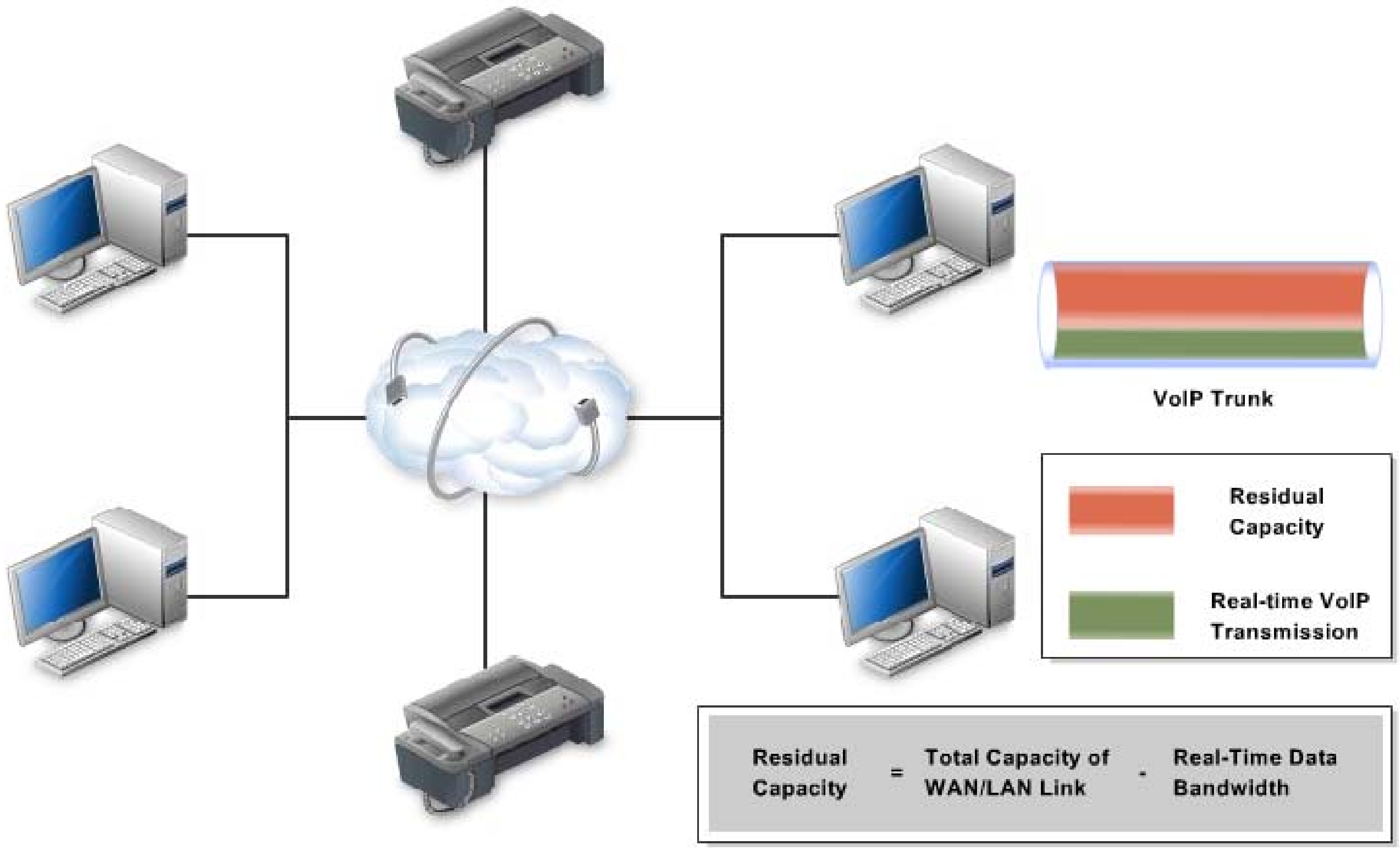
The total carrying capacity of a VoIP trunk is the **number of real-time VoIP calls and faxes** that can be accommodated within the bandwidth provided.



With Silence suppression can lower 2-4Kbps

VoIP Setup: Residual Capacity

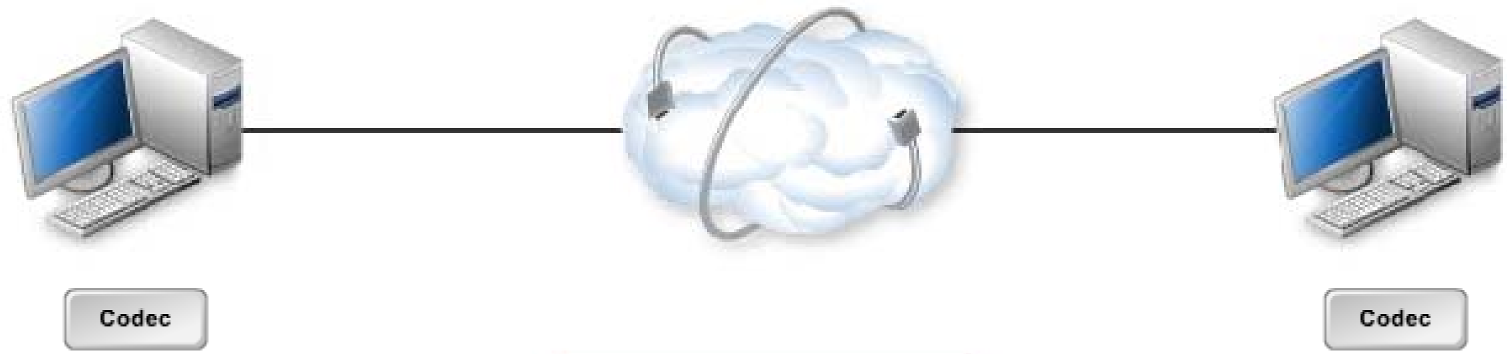
The Residual capacity/bandwidth, of a VoIP trunk is the bandwidth not used by real-time VoIP transmission.



VoIP Setup

Silence Suppression

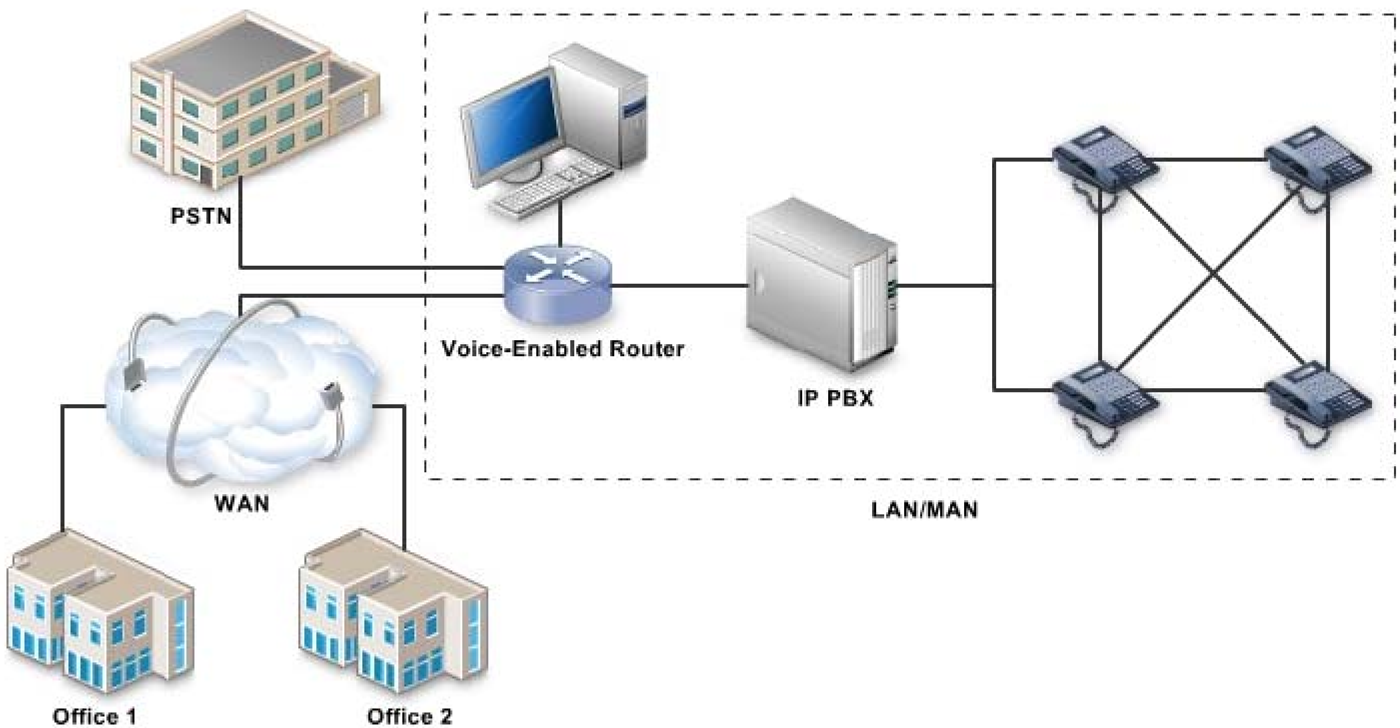
The VAD function is done by a codec in the packet-sending phone or by the gateway.



Can save bandwidth on IP trunks by 40% to 50%

VoIP Network Architecture

Centralized Network Architecture



VoIP Network Architecture

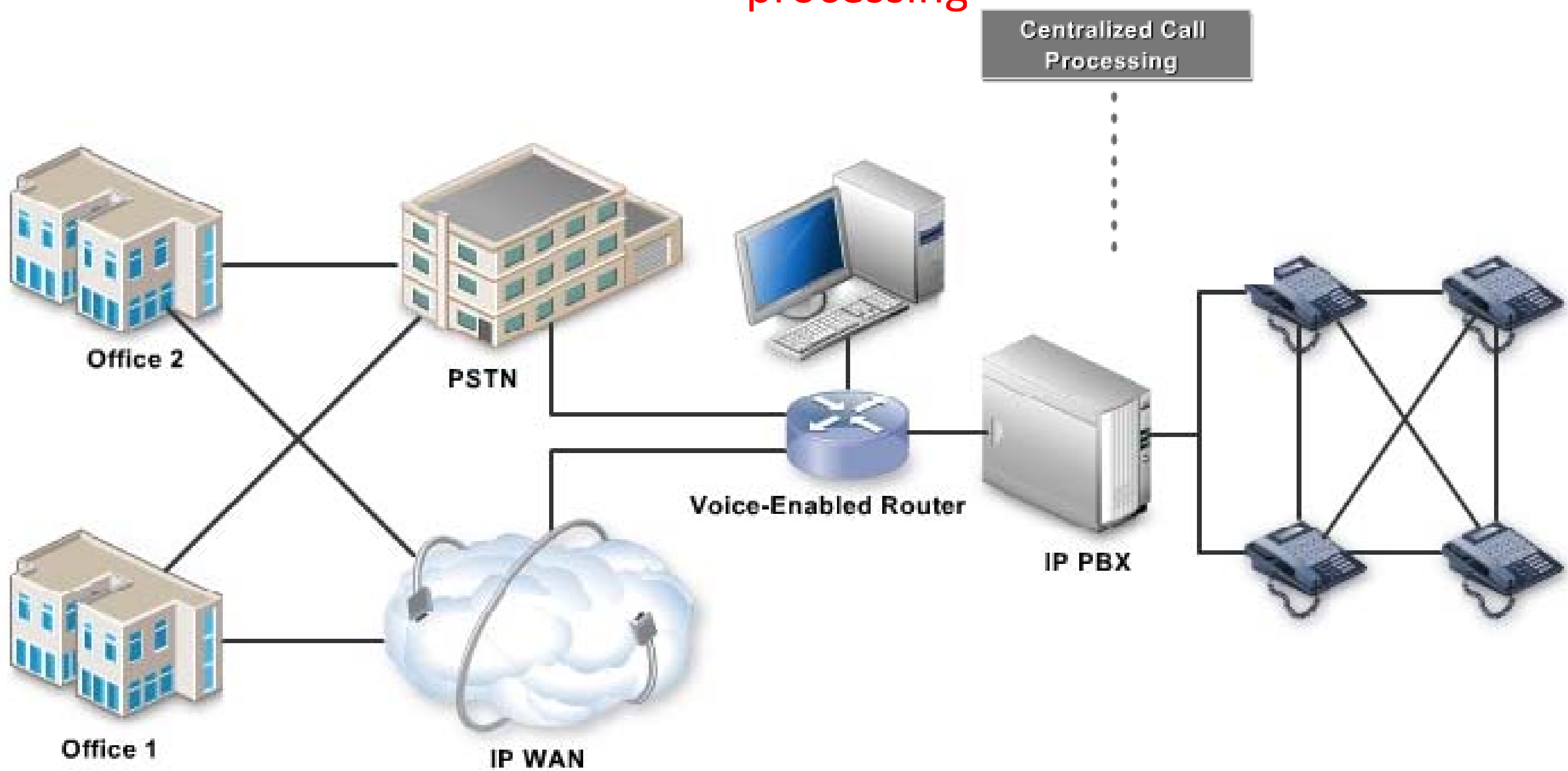
Centralized Network Architecture

Single-site architecture:

- Is cost-effective
- Allows each site to be self-contained
- Has simple dial plans
- Network deployment is easy

VoIP Network Architecture

Distributed Network Architecture with centralized Call processing



Multi-site architecture

VoIP Network Architecture

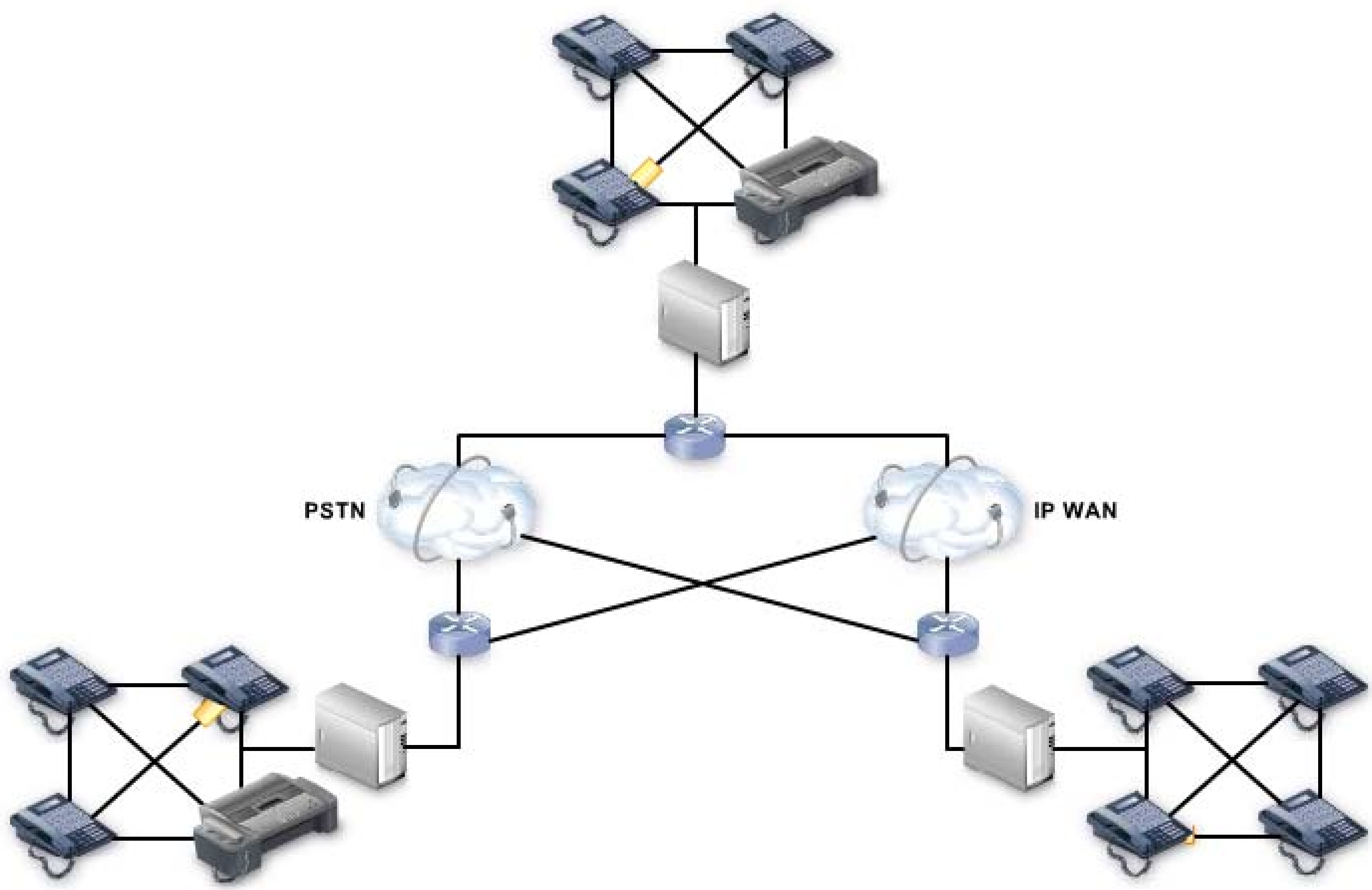
Distributed Network Architecture with centralized Call processing

WAN connectivity between sites on the IP WAN can be established using:

- Leased lines
- Frame Relay
- ATM
- MPLS

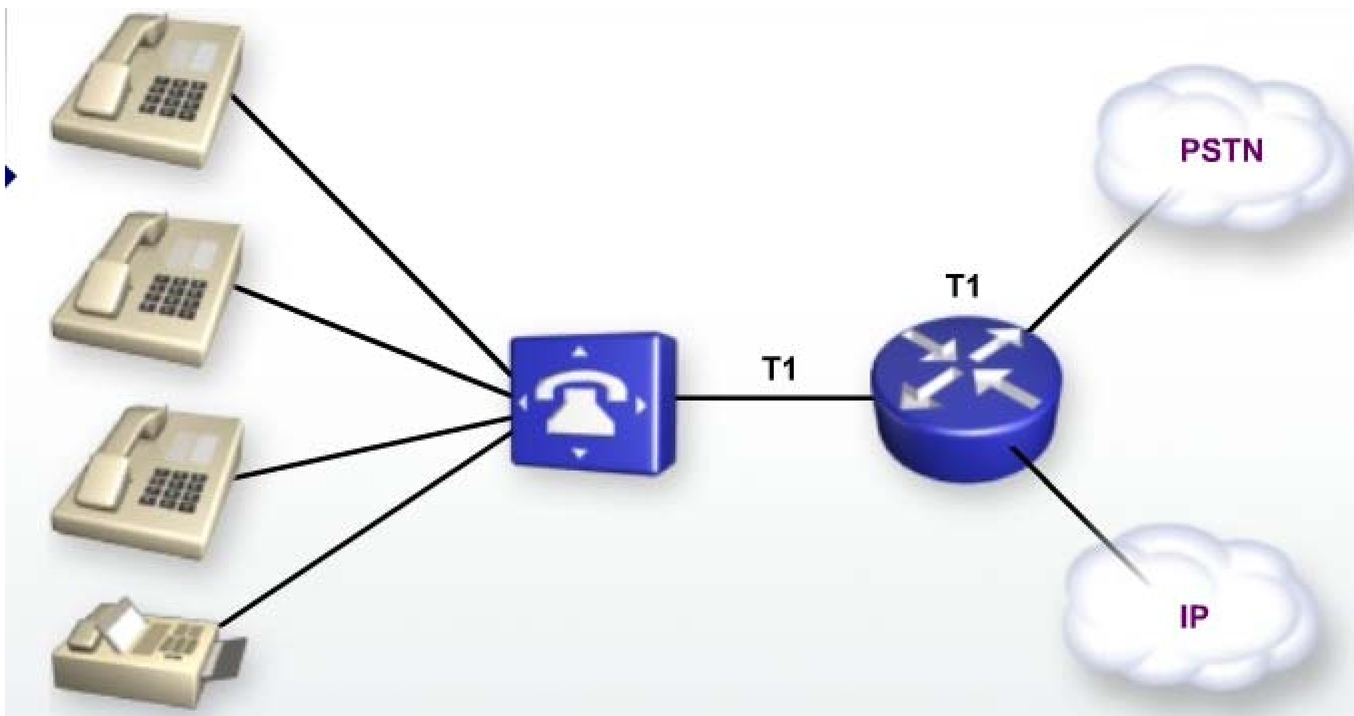
VoIP Network Architecture

Distributed Network Architecture with Distributed
Call processing



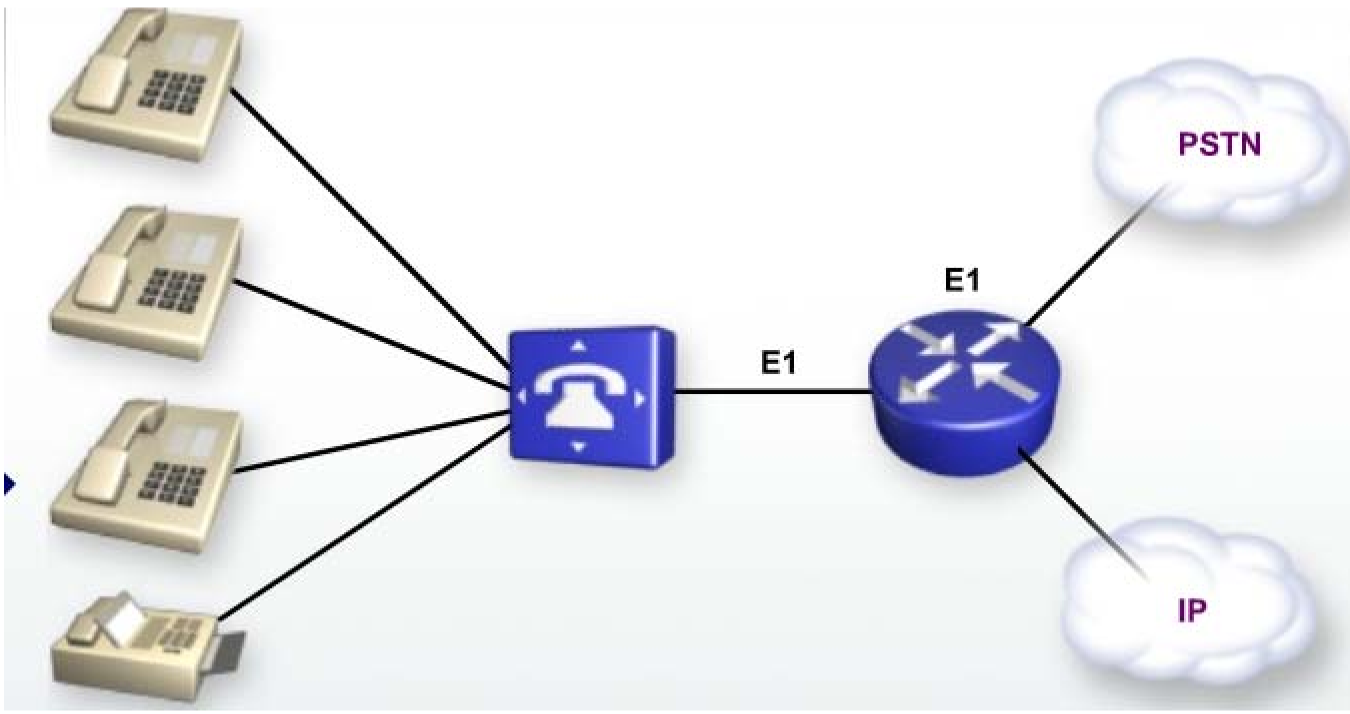
Digital Interfaces

- T1 Interface
  - is a form of digital connection that can simultaneously carry up to 24 conversations using two wire pairs.



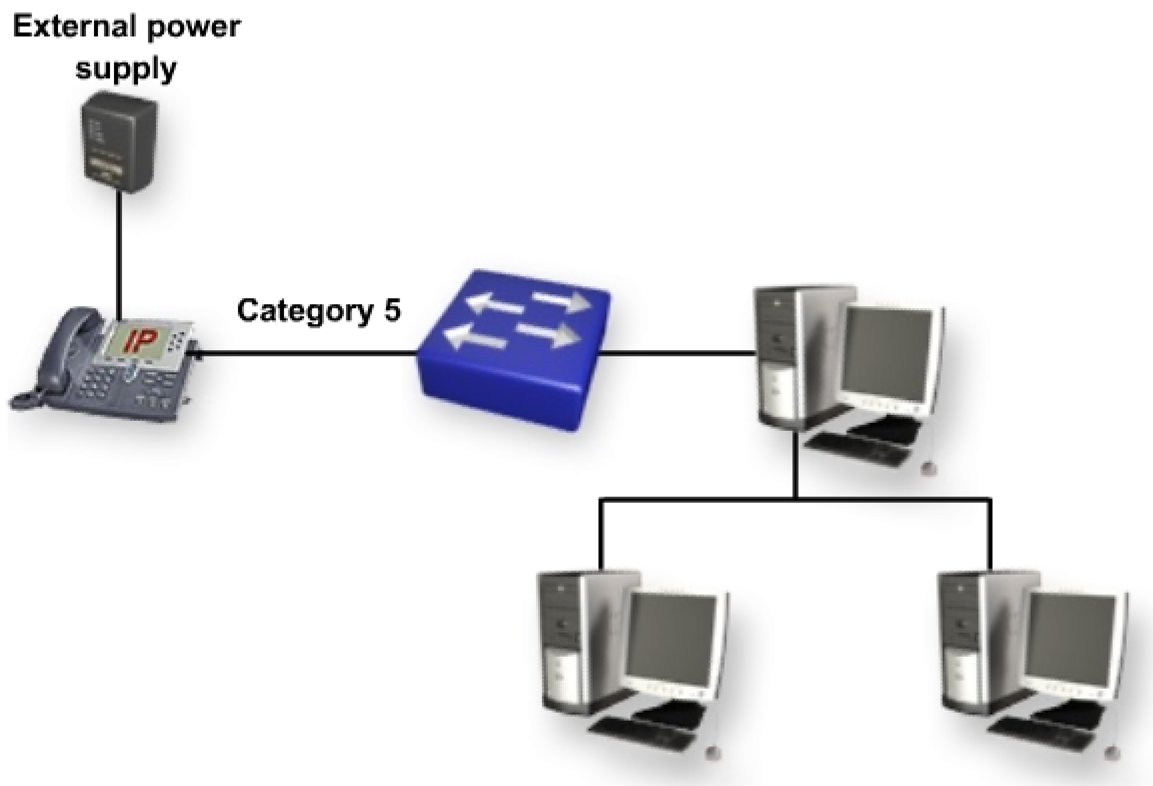
Digital Interfaces

- E1 Interface
  - An E1 interface has 32 channels and simultaneously carries up to 30 conversations.



IP Phone

- IP Phone
  - Use category 5 or better cable to be connected on the network devices.

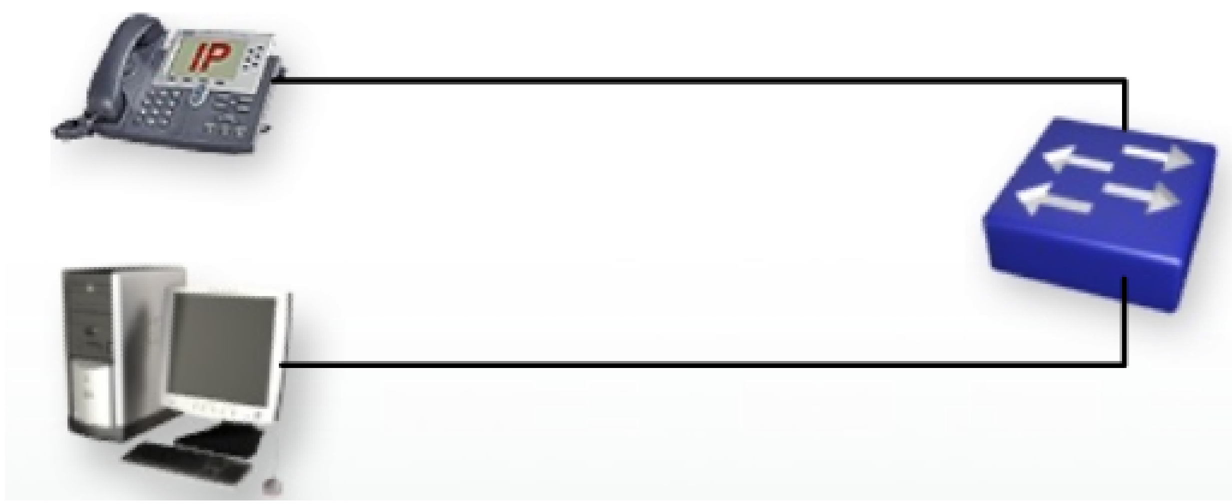


IP Phone

- 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 telephones and a PC. Reasons for using a single cable include ease of installation and cost savings infrastructure and wiring switch ports.

IP Phone

- IP Phone
  - 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.

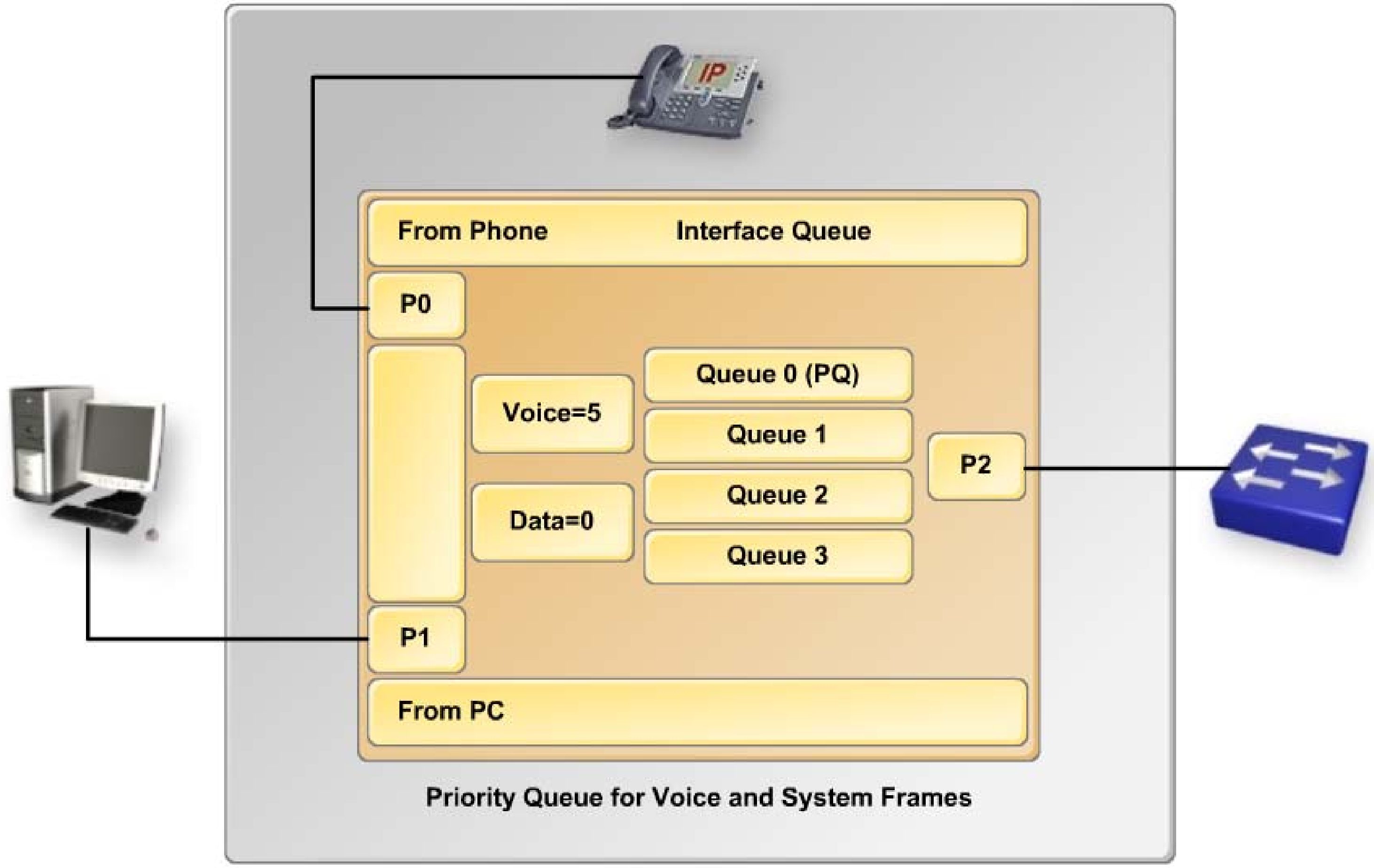


IP Phone

- IP Phone
  - 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 vice and data networks completely separate.

IP Phone

- IP Phone internally



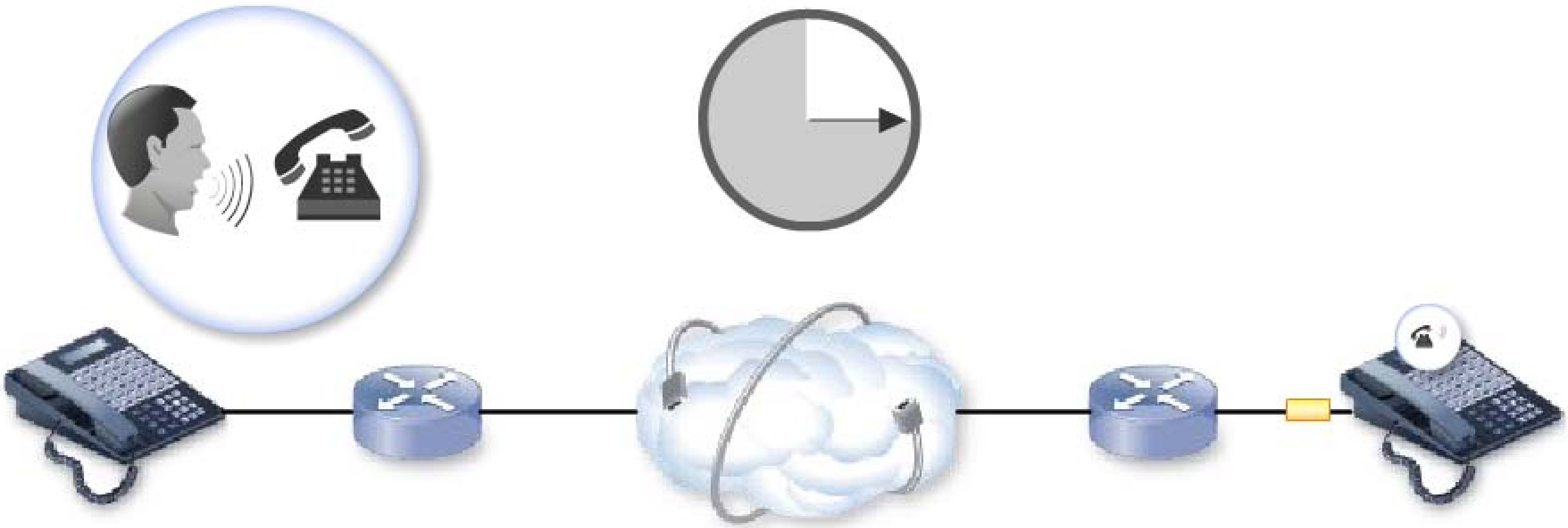
Q/A

If you have an enterprise with a couple of buildings in the same location, which network architecture will you deploy for IP telephony?

- A. A centralized architecture.
- B. A multi-site architecture with centralized call processing.
- C. A multi-site architecture with distributed call processing.
- D. A mix of centralized and distributed multi-site architecture.

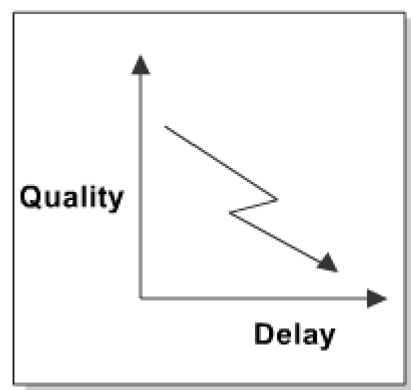
VoIP Issues

Delay



Factors affect Delay

propagation delay
coding
serialization delay



VoIP Issues

Fixed Delay

This accounts for the predictable delay on a voice network, which contributes to the overall delay. The components that contribute to fixed delay include:

- Coding, which is the time taken for an analog voice signal to be converted into a digital signal.
- Packetization time, which is the time taken to convert the digital signals into packets, and reconvert the packets into digital signals at the other end.
- Serialization, which is the sequencing of packets.
- Propagation, which is the time taken for the transmission of packets.

VoIP Issues

Variable Delay

Variable delays arise from queuing, processing of packets, congestion, or path variation. One or more of these factors can possibly contribute to the overall delay on a network.

VoIP Issues

Types of Delays

- Call setup delay
- Propagation delay
- Handling delay
- Queuing delay

VoIP Issues

Call Setup Delay

Call setup delay, also called post dial delay, is the time interval between dialing the last digit and receiving a ringback.

In the traditional phone system, there is no acoustic feedback between dialing and ringback. So, if there is an excessive delay before ringback, the caller may abandon the call thinking that something is wrong.

With Internet telephony, however, additional feedback can be provided during the setting up of the call.

VoIP Issues

Propagation Delay

The length a signal must travel as light in fiber or as an electrical impulse in copper-based networks causes propagation delay.

A fiber network stretching halfway around the world (13,000 miles) induces a one-way delay of about 70 milliseconds (ms). Although this delay is almost imperceptible to the human ear, propagation delays in conjunction with handling delays can cause noticeable speech degradation.

VoIP Issues

Handling Delay

Handling delay, also called serialization delay, is caused by devices that handle voice information. Handling delays significantly degrade voice quality in a packet network. Devices that forward a packet through the network cause handling delay. Handling delays are a larger issue in packetized environments.

VoIP Issues

Queuing Delay

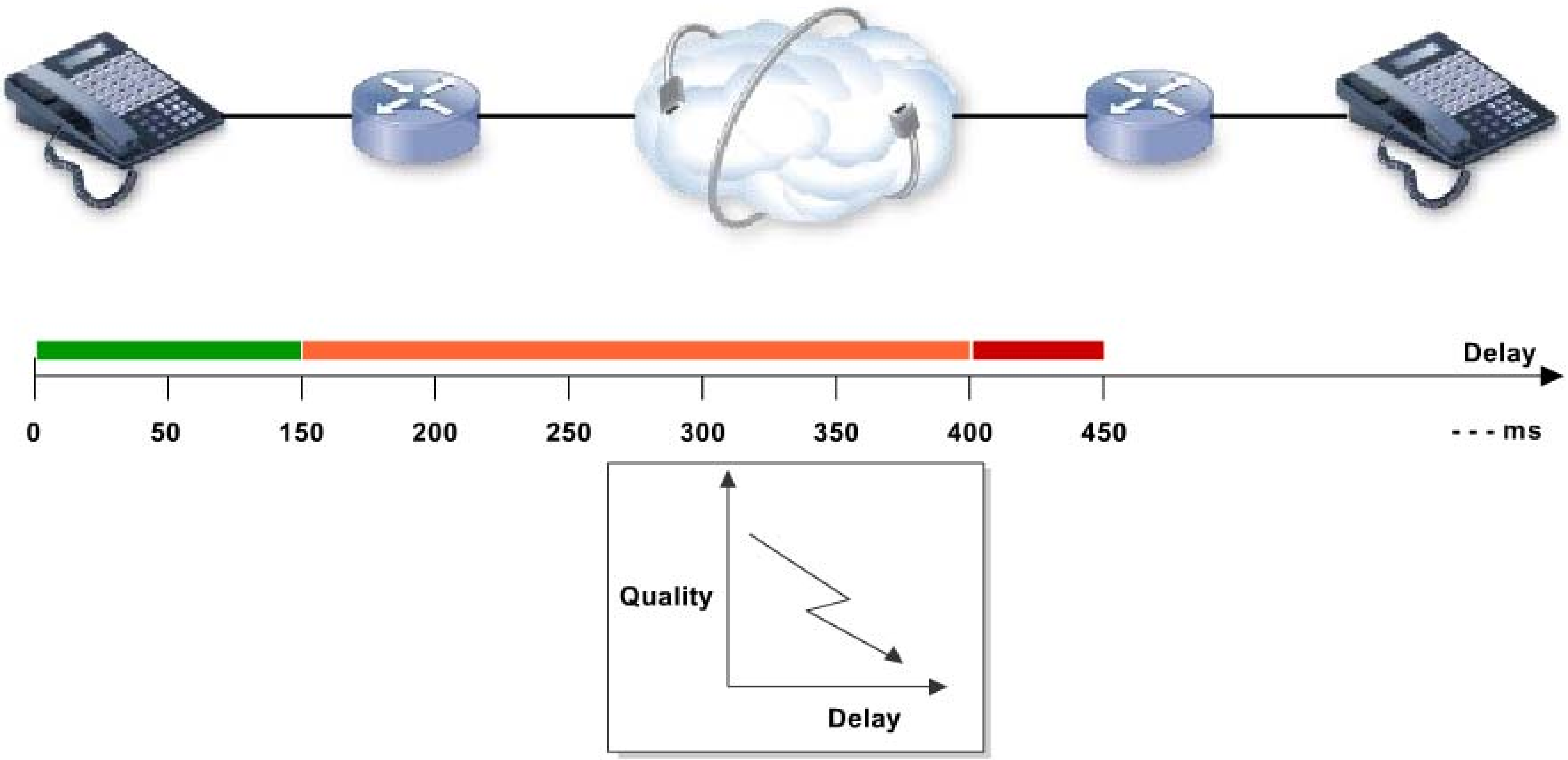
A packet-based network experiences delay as time is necessary to move the actual packet to the output queue. Queuing delay occurs when packets are held in a queue because of congestion on an outbound interface. This happens when more packets are sent out than the interface can handle at a given interval.

The actual queuing delay of the output queue is another cause of delay. You should keep this factor to less than 10 ms.

In an unmanaged, congested network, queuing delay can add up to two seconds of delay (or result in the packet being dropped). This lengthy period of delay is unacceptable in almost any voice network.

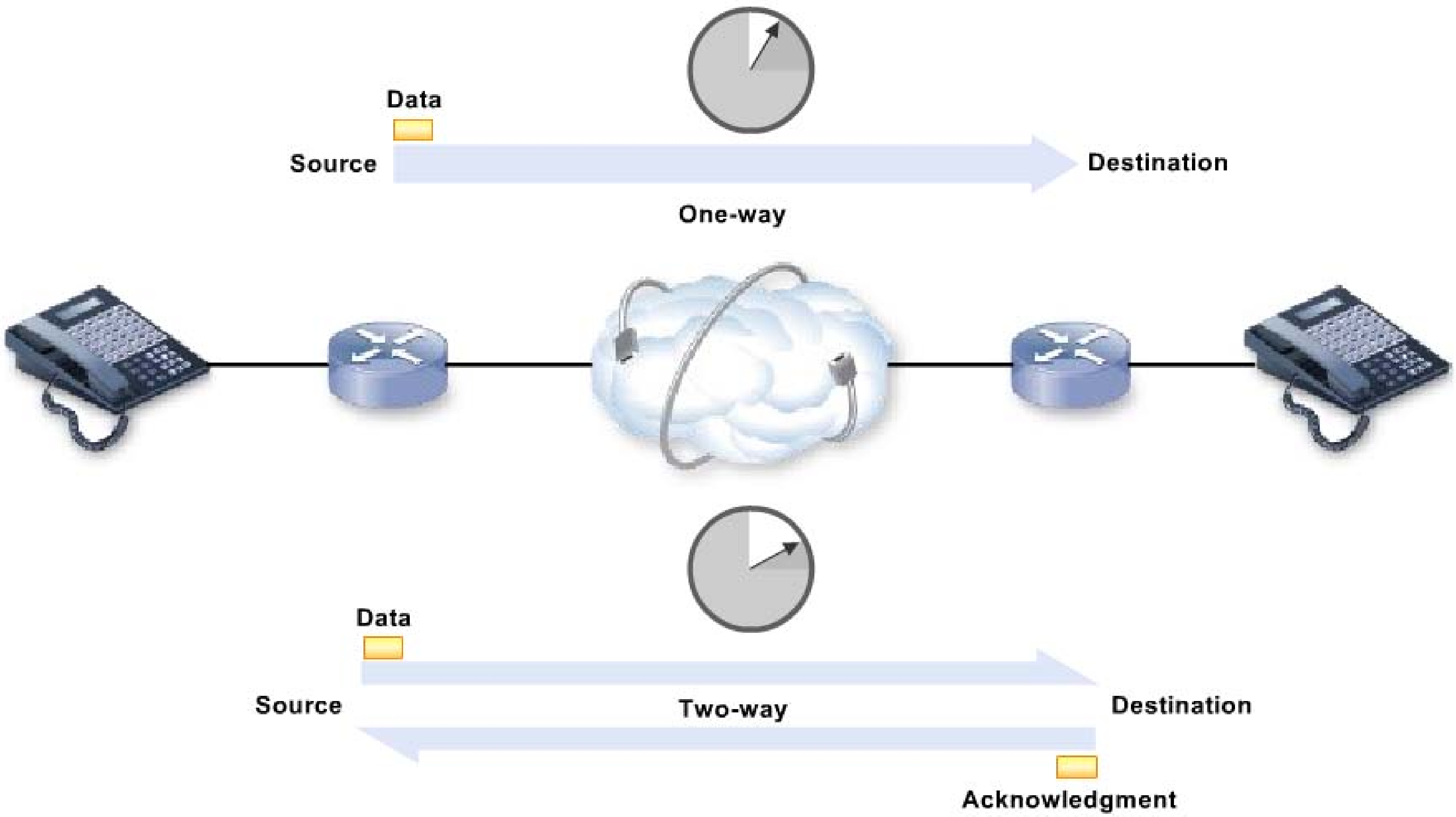
VoIP Issues

Networks Delay Considerations



VoIP Issues

Latency



VoIP Issues

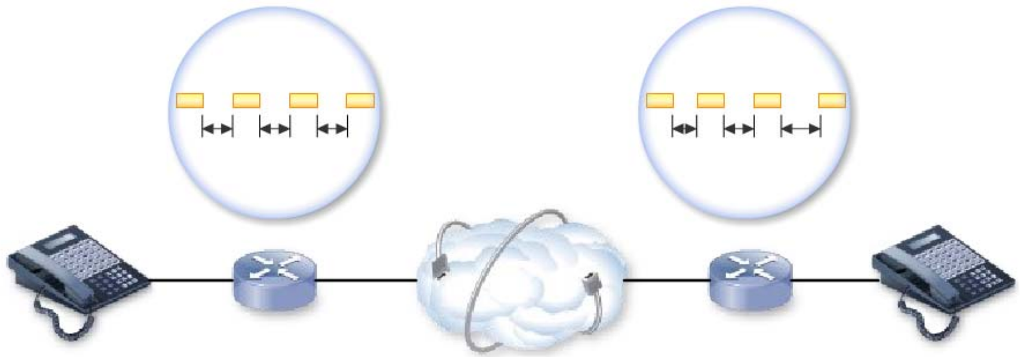
Latency

Latency can be minimized by:

- Increasing the network bandwidth
- Choosing a different codec type
- Fragmenting data packets
- Prioritizing voice packets on the network

VoIP Issues

Jitter

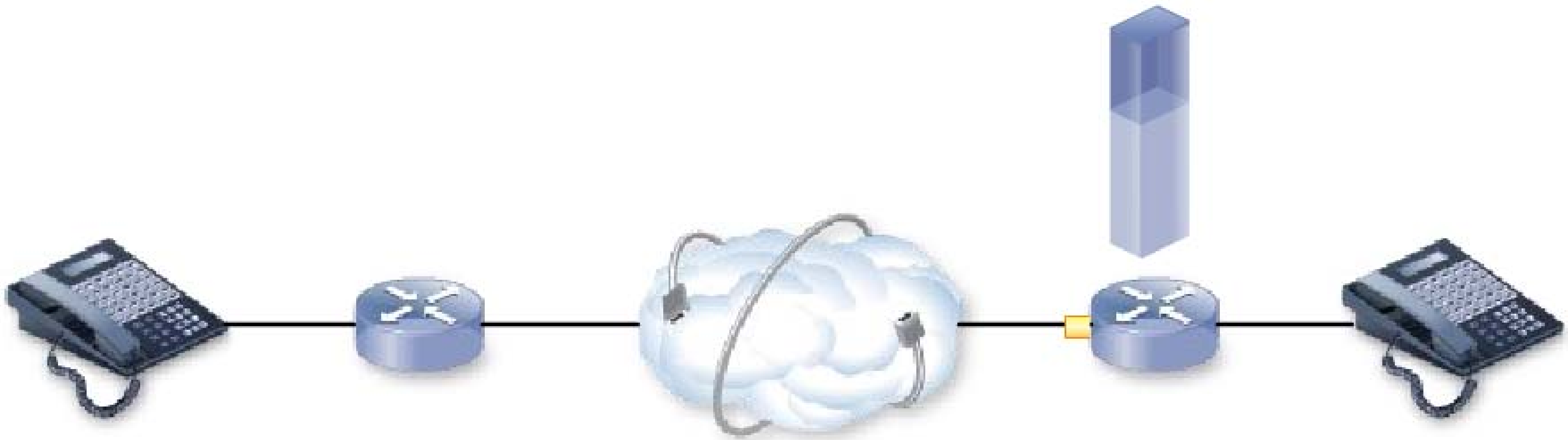


Due to:
- network congestion
- improper queuing
- errors in configuration

VoIP Issues

Solving Jitter problems

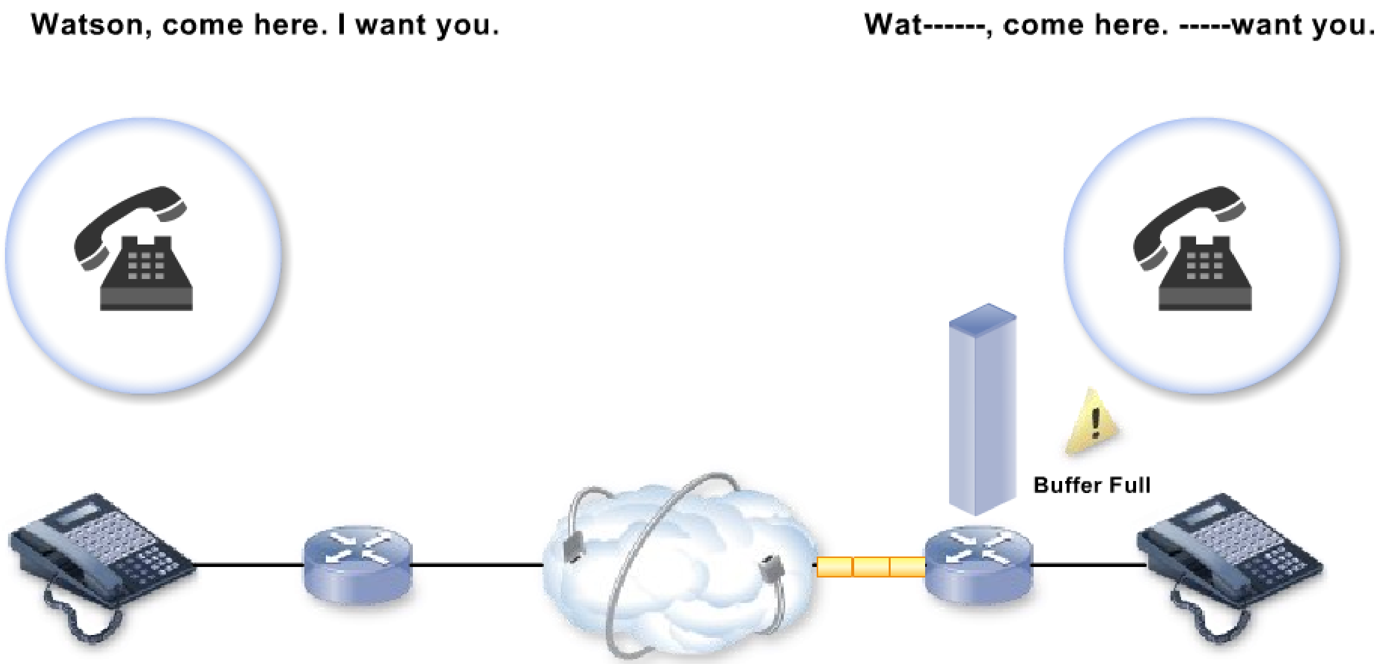
Dejitter buffer
Play-out delay buffer



Dejitter buffers increase latency on the network

VoIP Issues

Packet Loss



Packet loss can be minimized by:

- Designing the network so as to reduce packet loss
- Prioritizing voice packets on the network
- Using appropriate codecs that minimize packet loss

Q/A

Which type of delay occurs due to congestion on a VoIP network?

- A. Propagation delay
- B. Handling delay
- C. Queuing delay
- D. Call setup delay

What amount of delay is acceptable for voice applications?

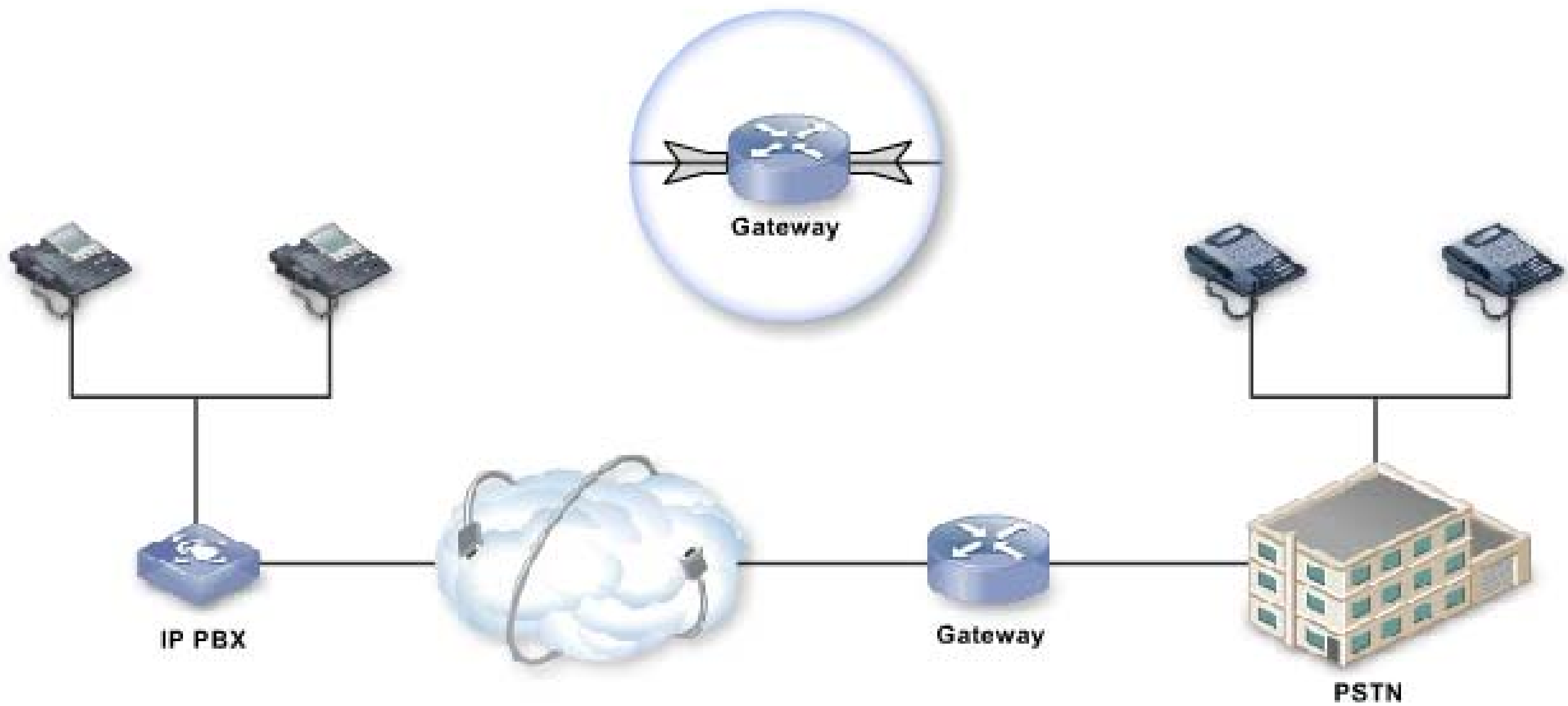
- A. 0 to 150 ms
- B. Above 400 ms
- C. 150 to 400 ms
- D. 200 to 250 ms

Q/A

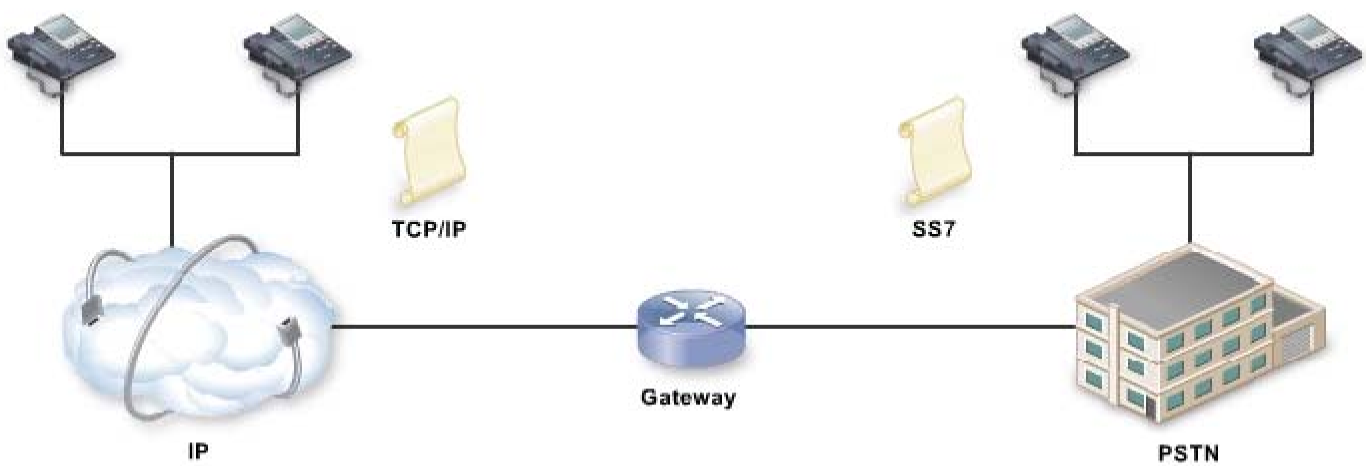
What are the factors that contribute to predictable delay on a voice network?

- ☐ A. Packetization time
- ☐ B. Queuing
- ☐ C. Coding
- ☐ D. Congestion

Gateways



VoIP: VoIP Gateways

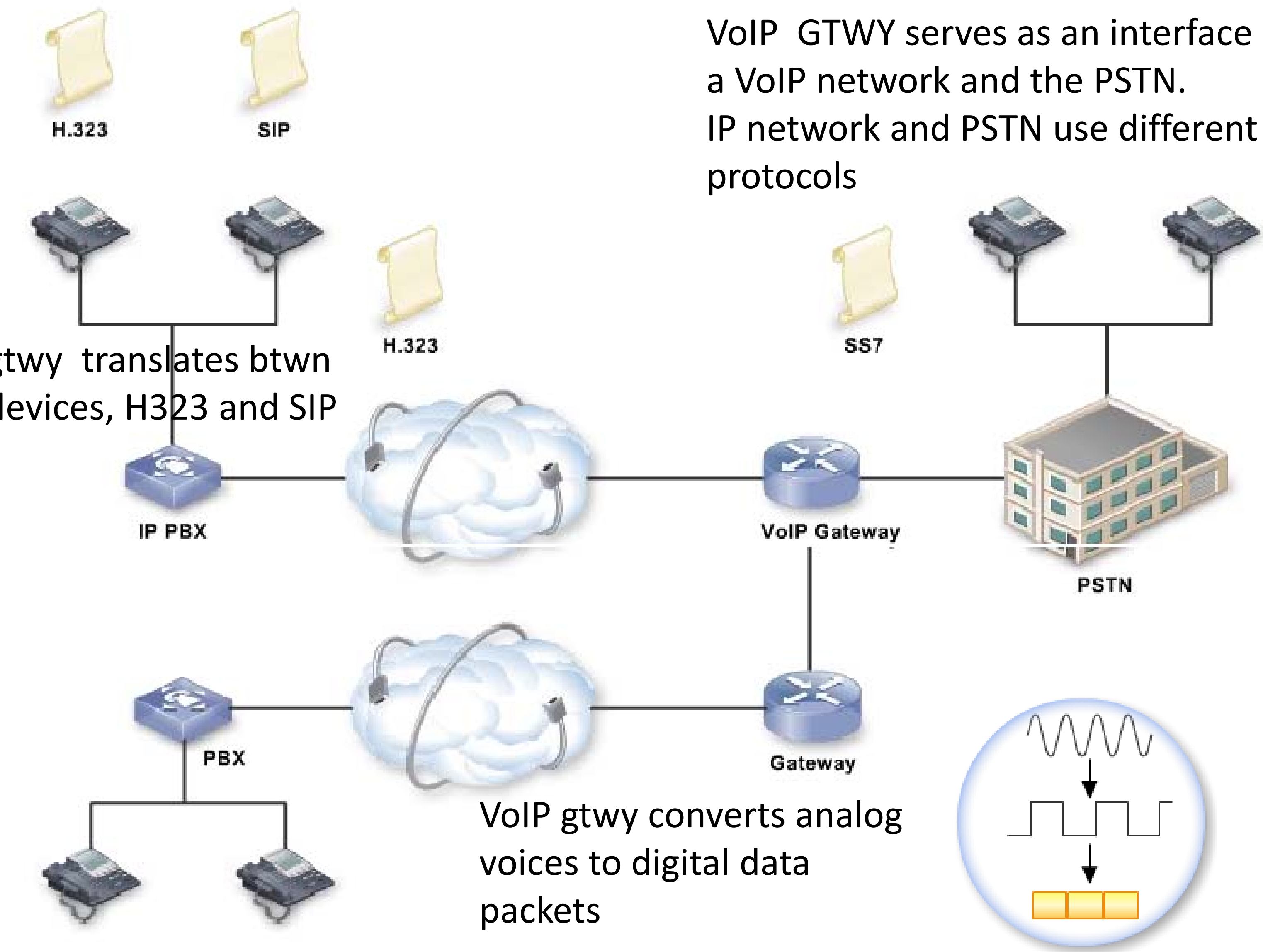


Can accept a packet formatted for one protocol and translate it into a format suitable for another

VoIP: VoIP Gateways

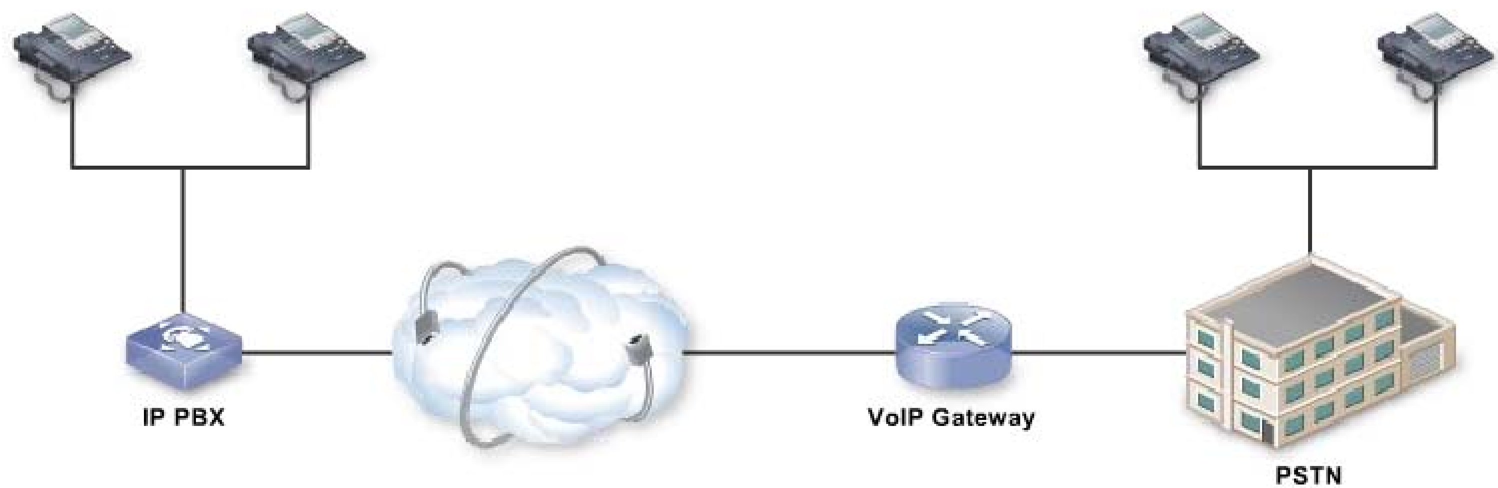
VoIP GTWY serves as an interface between a VoIP network and the PSTN.
IP network and PSTN use different signaling protocols

VoIP gtwy translates btwn VoIP devices, H323 and SIP



VoIP gtwy converts analog voices to digital data packets

VoIP: VoIP Gateways



Hardware-based VoIP gateways:

- Can be standalone units
- Are more dedicated and reliable
- Are preferred by enterprises
- Are available in different configurations

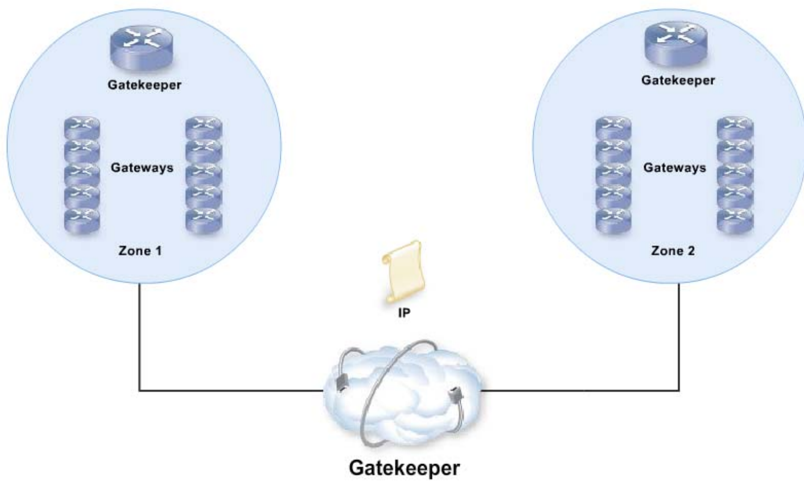
A software-based VoIP gateway:

- Is installed within a router
- Can run on most standard network operating systems
- Is easier to monitor and run
- Is less expensive
- Can be easily upgraded

VoIP: Gatekeeper

A gatekeeper:

- Manages traffic
- Provides address translation for a specific zone
- Eliminates bottlenecks
- Contributes to enhancing the quality of service provided
- Simplifies addressing between endpoints
- Facilitates load balancing
- Allows multiple gateways to use a single IP address for routing calls

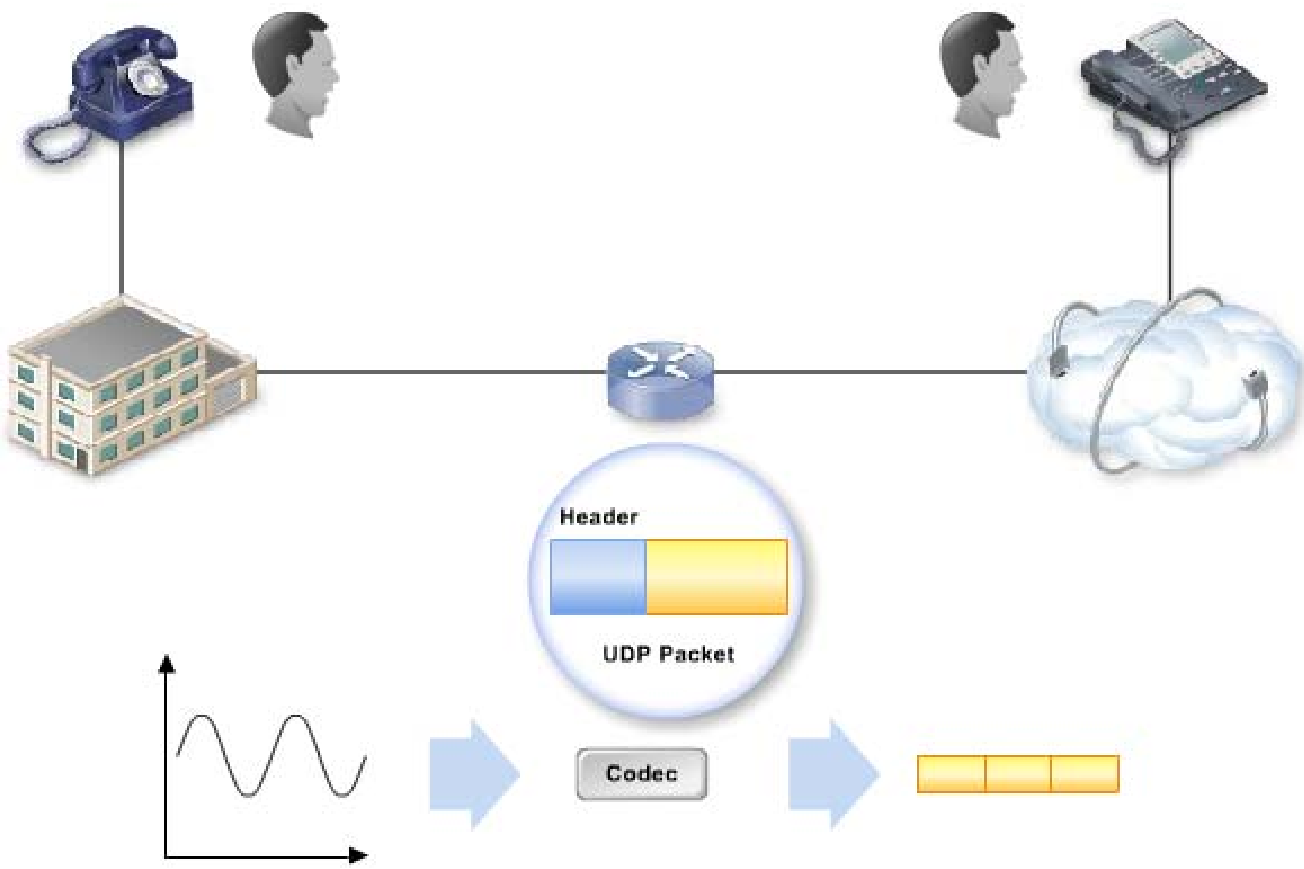


Q/A

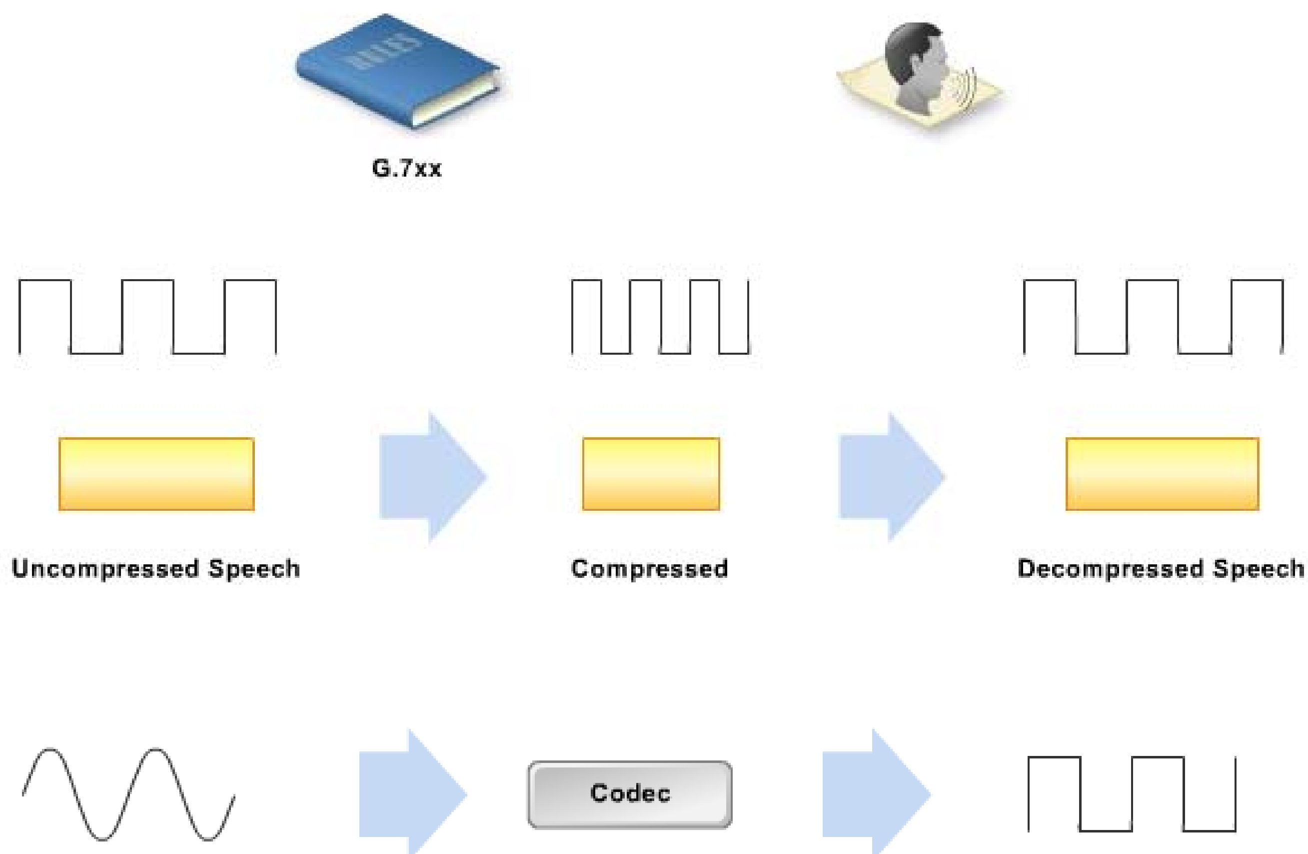
What is the primary function of a gatekeeper?

- A. Connects one phone line to another through a software interface.
- B. Divides a voice network into distinct segments.
- C. Authenticates and authorizes calls between endpoints.
- D. Serves as an interface between a VoIP network and the PSTN.

VoIP: Talk Spurts

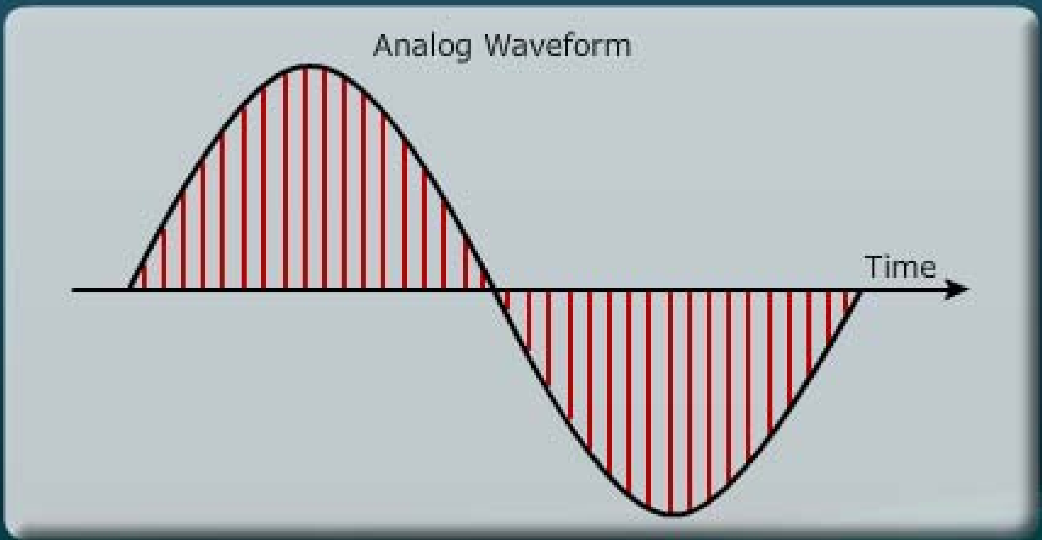


G.7xx Standards

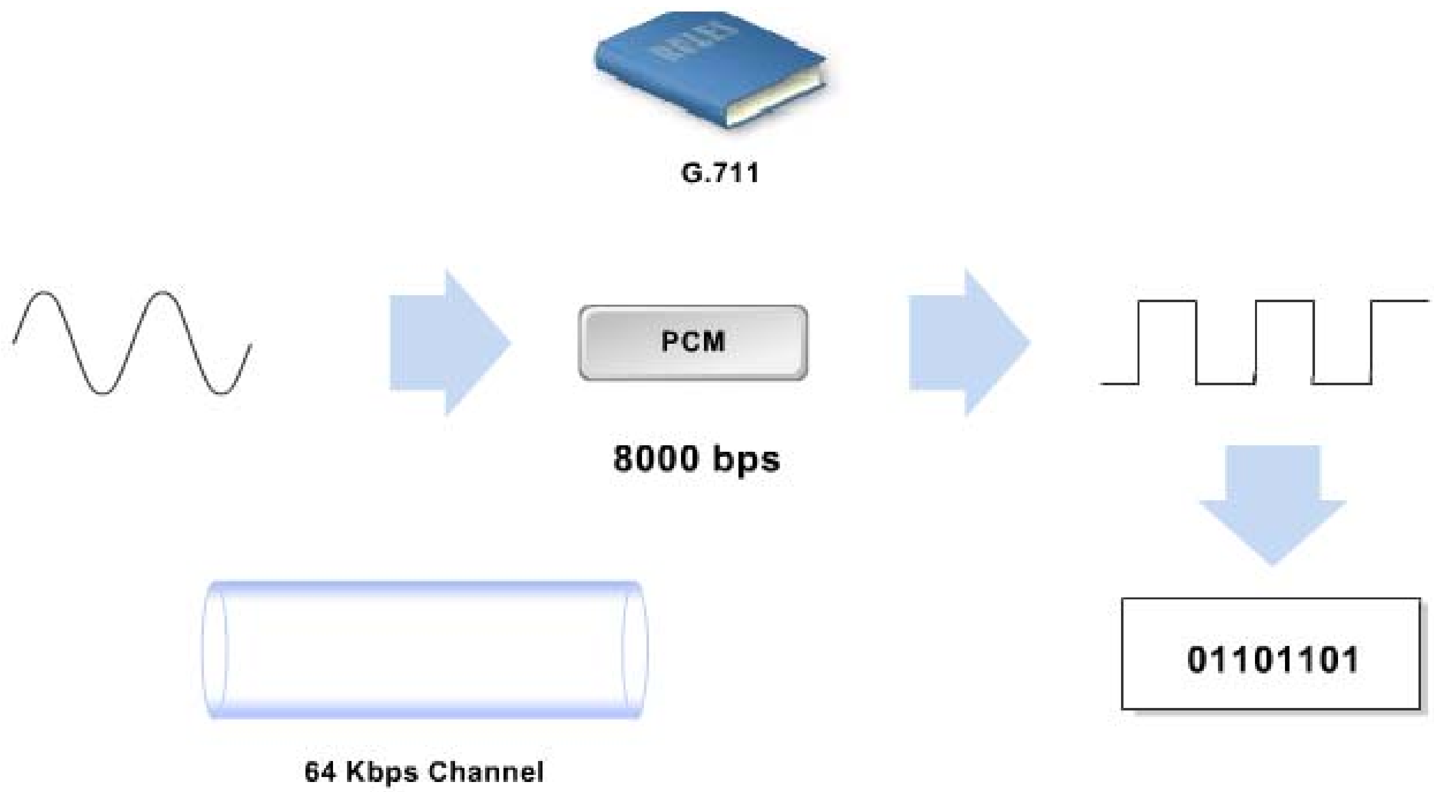


Sampling

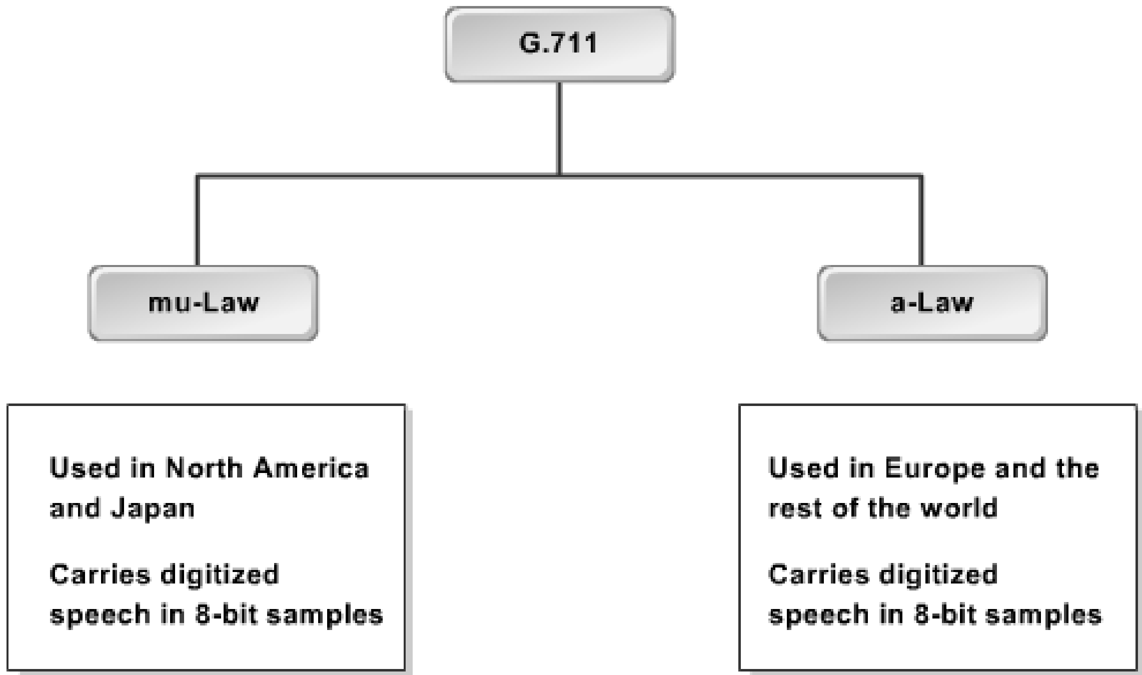
- Significant human articulation range:
  - 300 Hz to 4 kHz
- Nyquist theorem: sampling rate = 2 x maximum articulation frequency
  - 2 x 4 kHz = 8 kHz = 8000/sec
  - Each sample is 1/8000 of a second apart



G.711



G.711

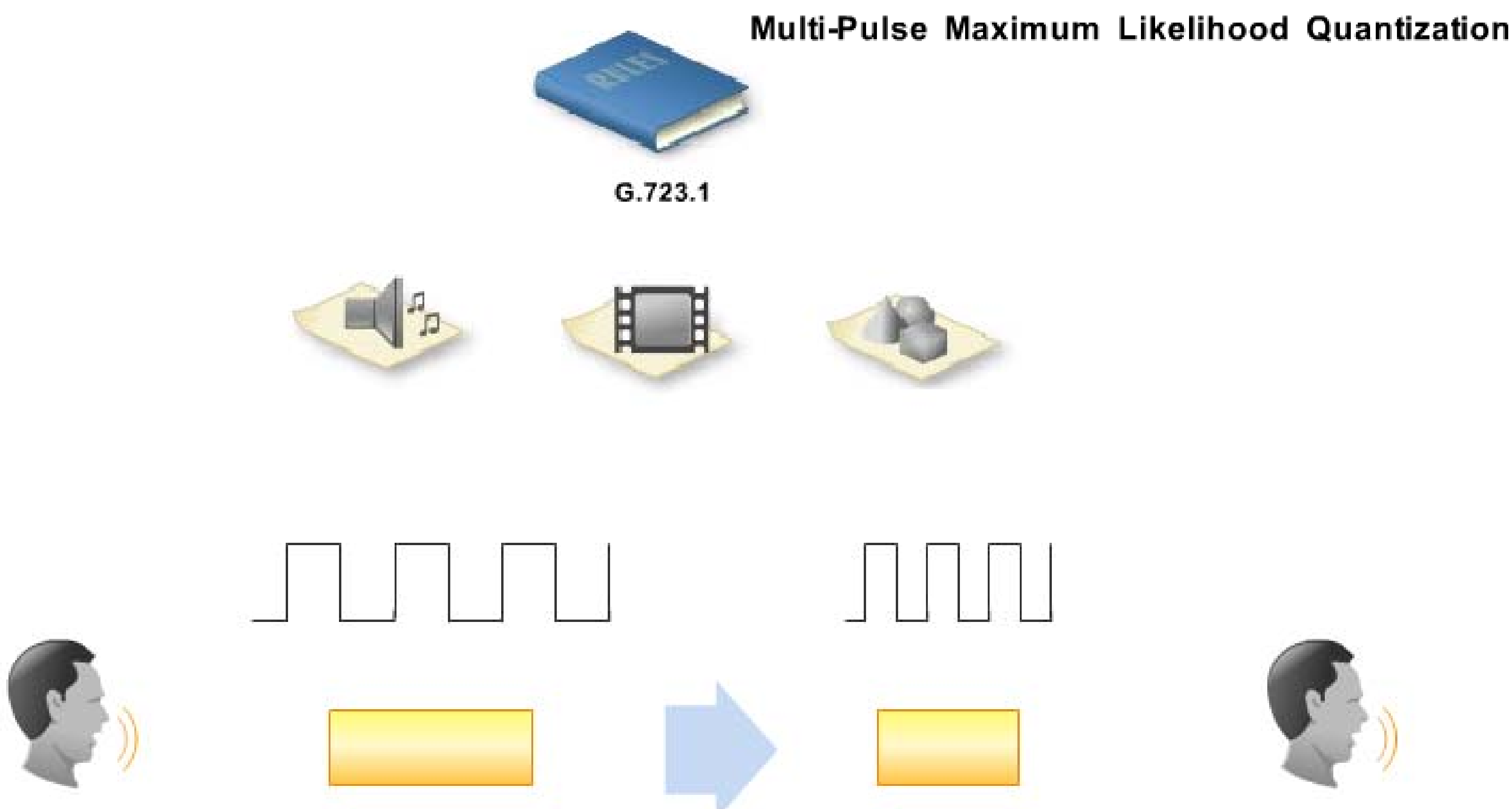


Advantages of G.711:

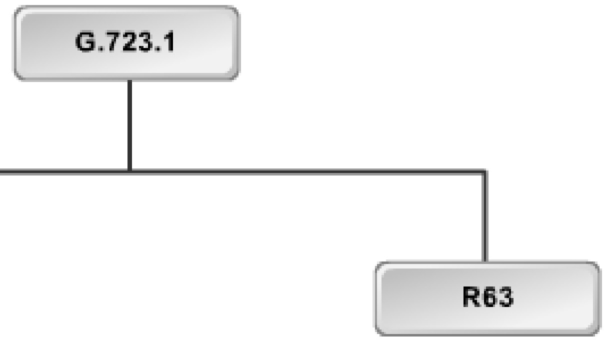
- Encoded voice is already in the appropriate format
- Provides better voice quality for VoIP transmission

G.723.1

G.723.1



G.723.1



Encodes at 5.3 Kbps using 20–byte frames

Uses ACELP algorithm

Provides for additional flexibility

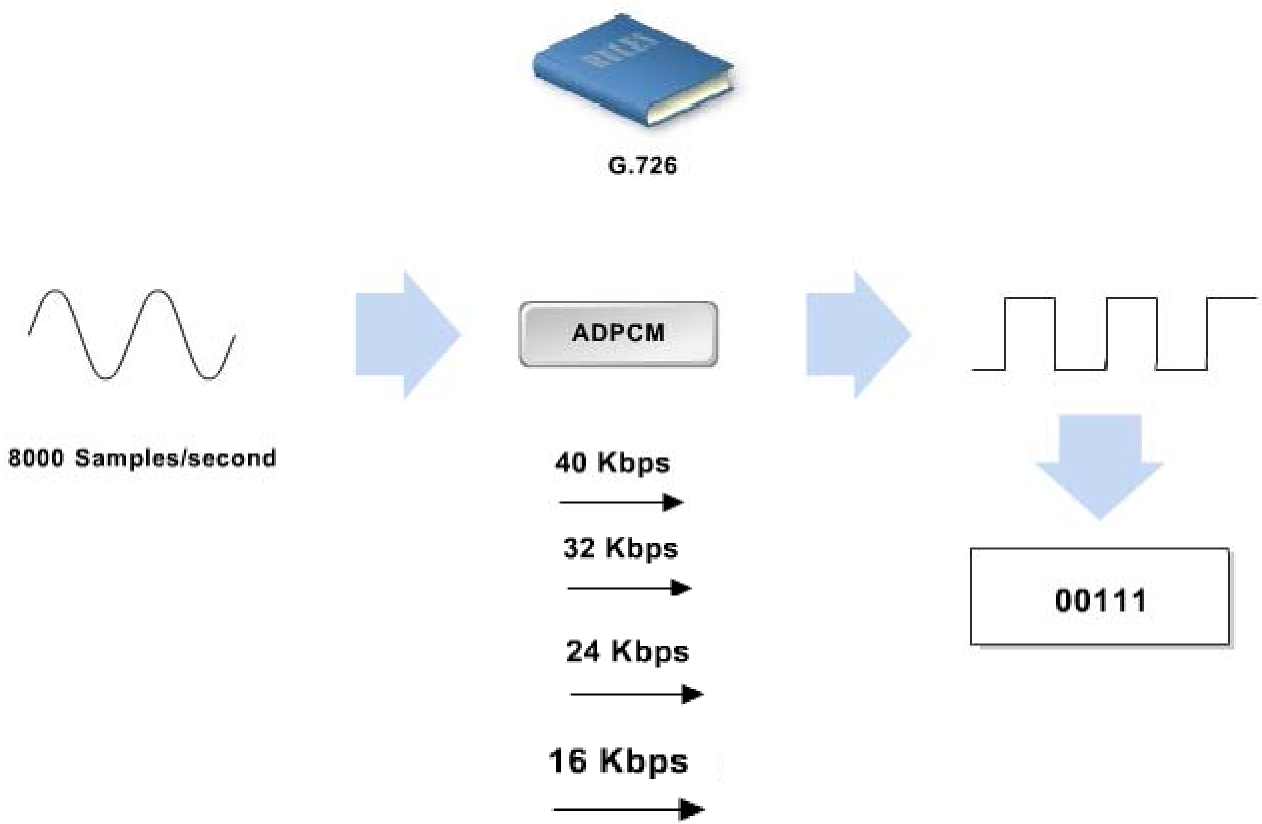
Suited for VoIP applications such as videoconferencing

Encodes at 6.3 Kbps using 24-byte frames

Uses ML-MLQ algorithm

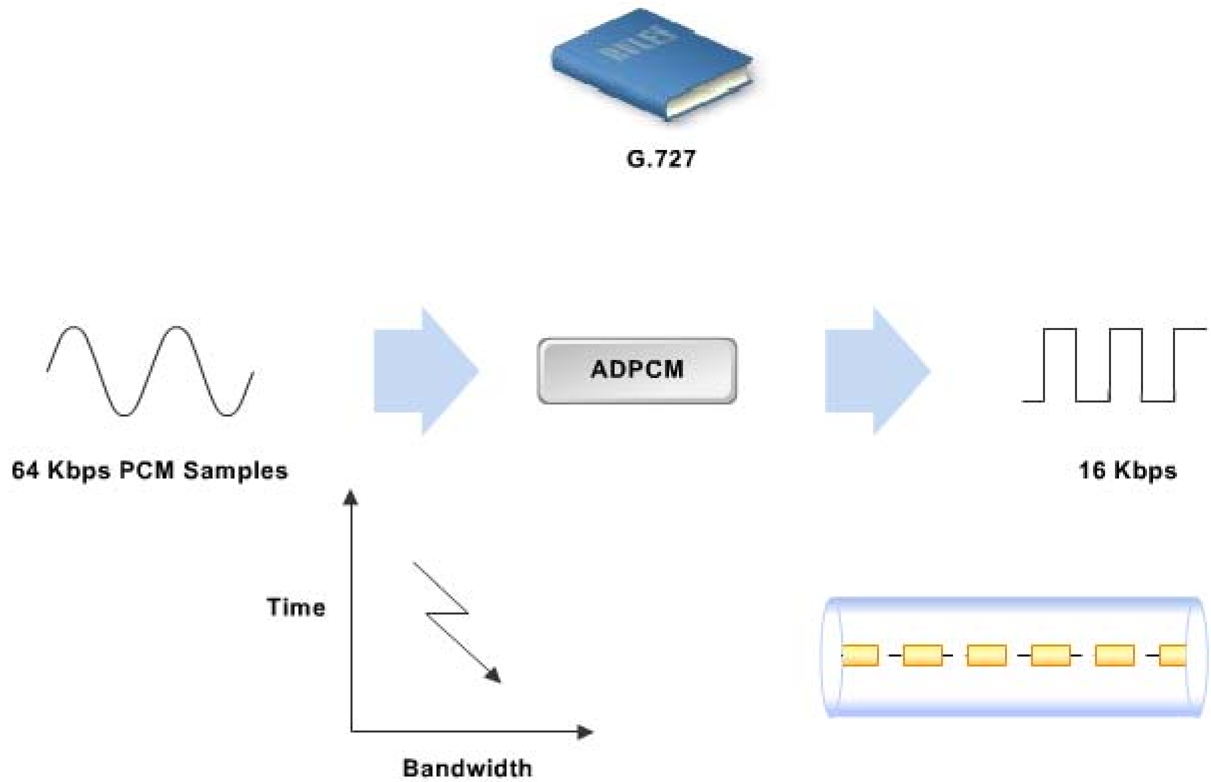
Provides better quality of encoded audio

G.726



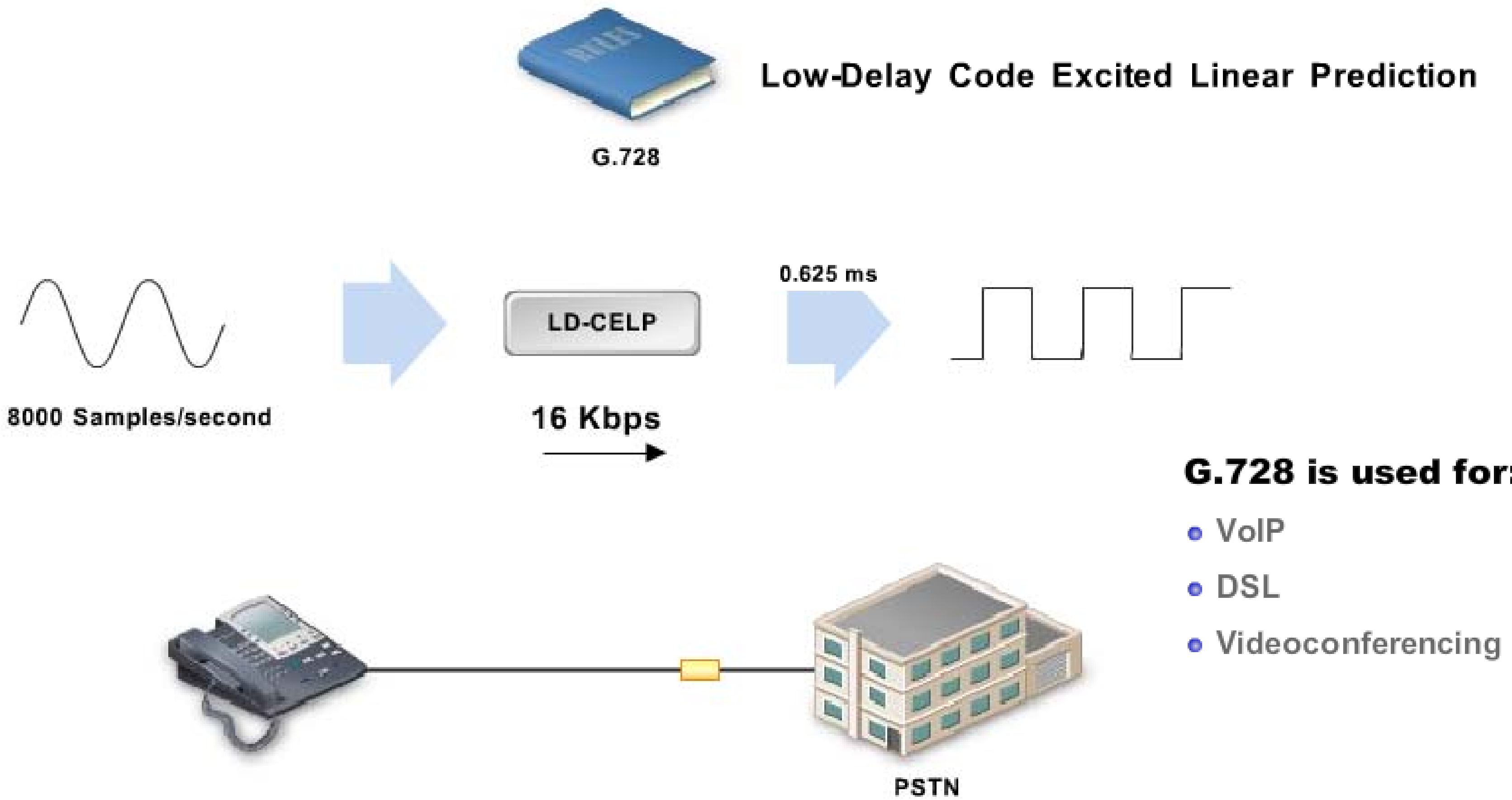
G.726 increases network capacity and consumes less bandwidth

G.727

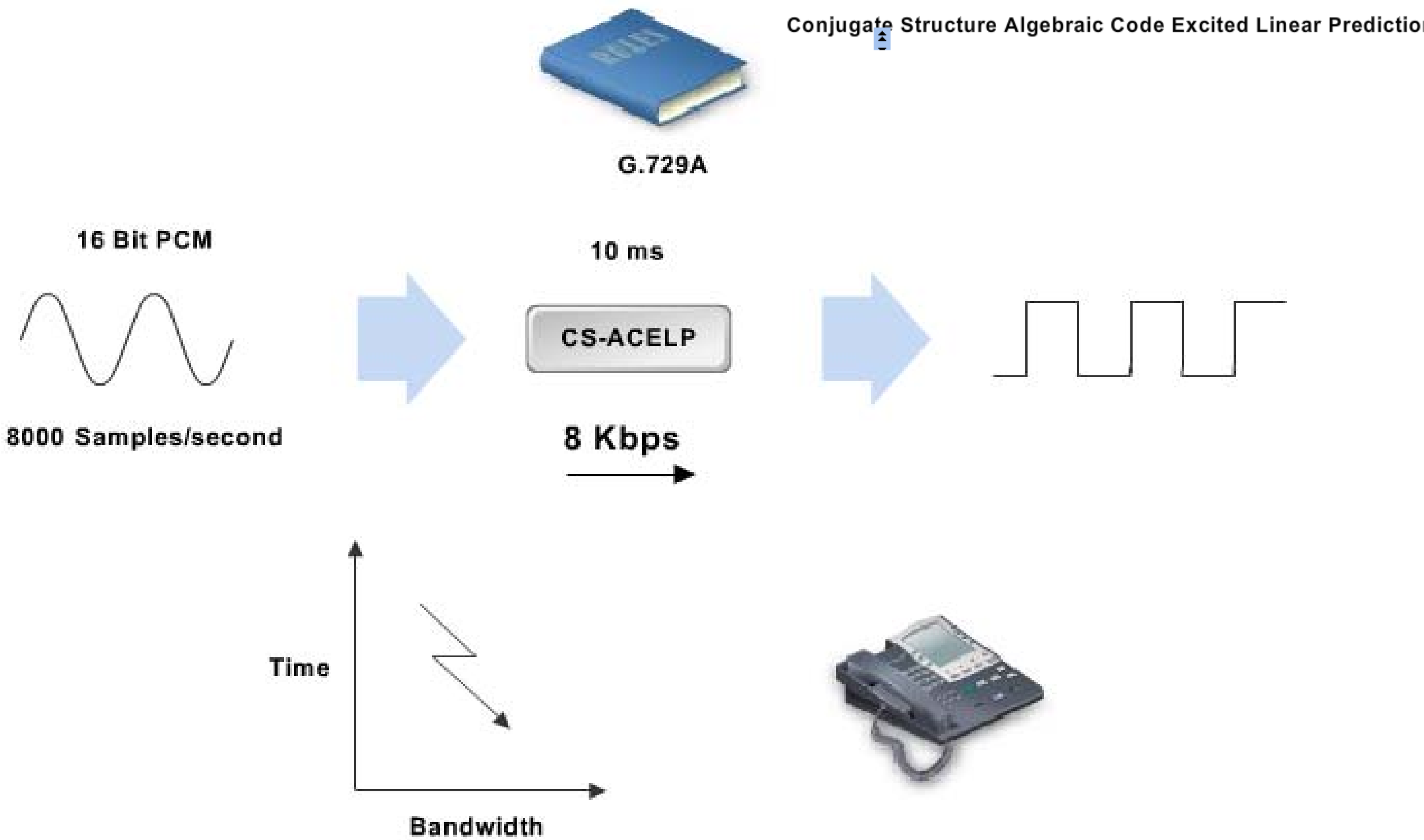


A specialized version of G.726 that is suitable for packet-based technologies such as VoIP

G.728


Low-Delay Code Excited Linear Prediction
G.728
0.625 ms
LD-CELP
8000 Samples/second
16 Kbps
G.728 is used for:
VoIP
DSL
Videoconferencing
PSTN

G.729A



Conjugate Structure Algebraic Code Excited Linear Prediction
G.729A
16 Bit PCM
8000 Samples/second
10 ms
CS-ACELP
8 Kbps
Time
Bandwidth

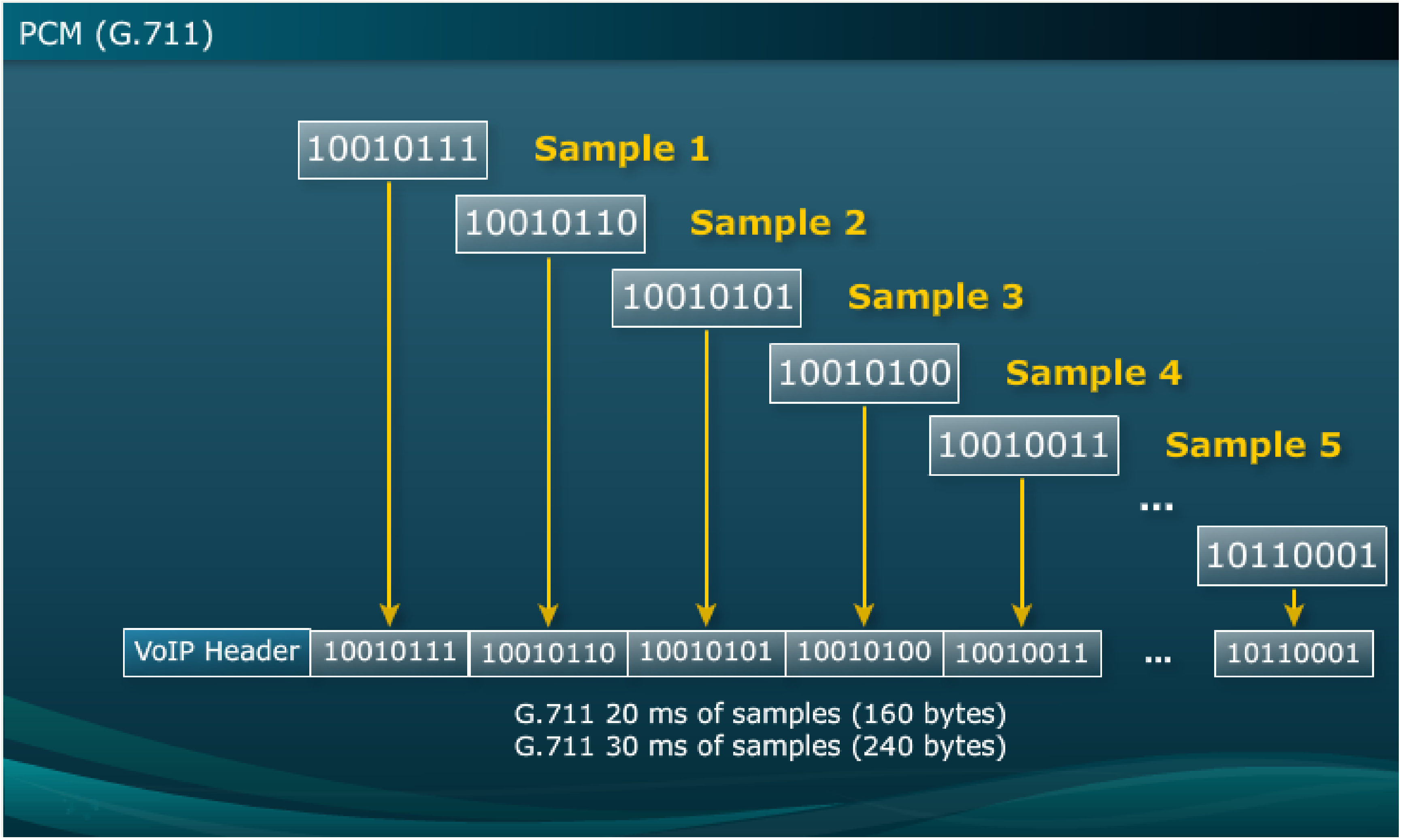
G.729 is going to be the one that we choose in a lot of instances to send across our WAN links. G.729 comes in a couple of flavors. It has an "a", a "b", and an "ab" after it.

Compression

Optional

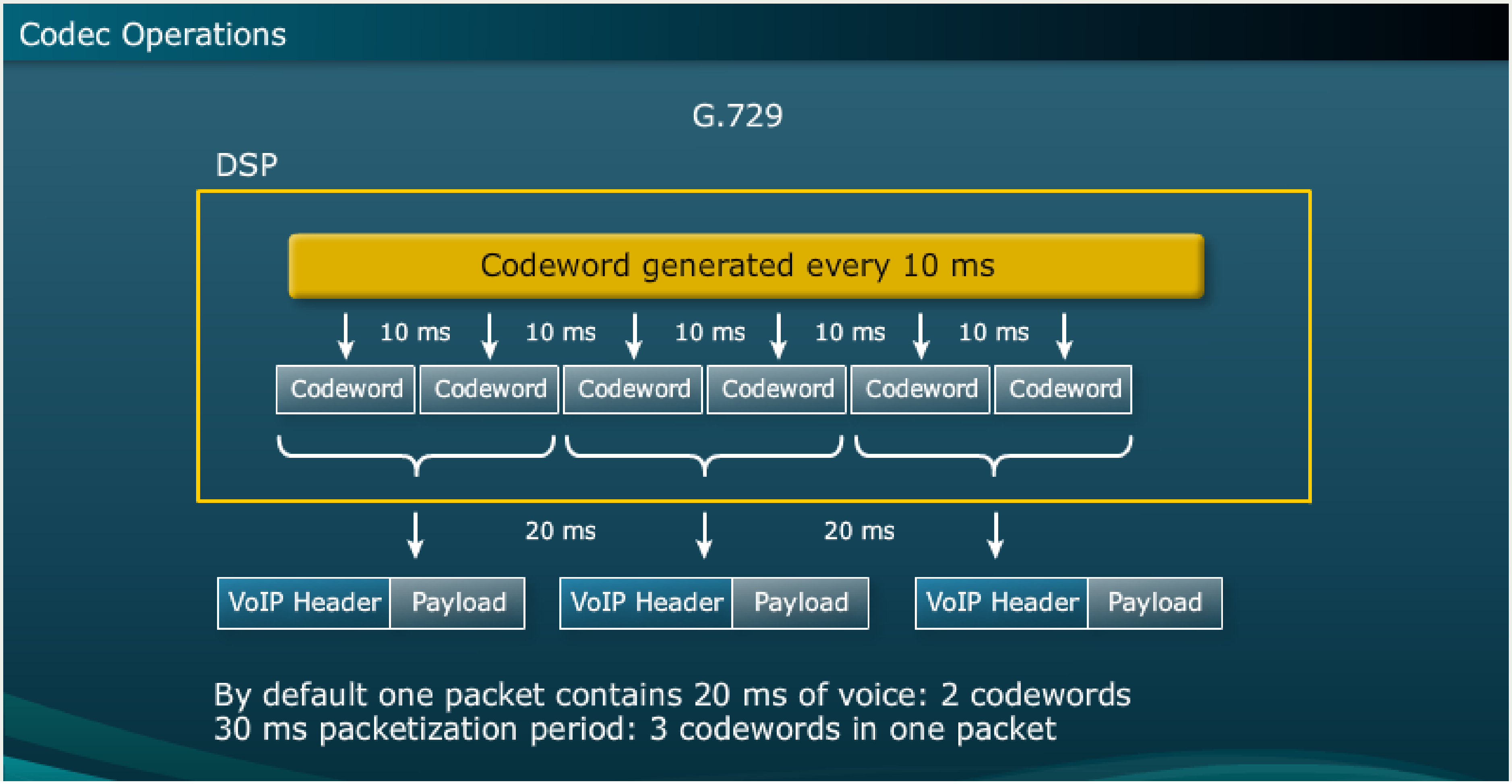
*iLBC = Internet Low Bitrate Codec

Codec	Bandwidth [kb/s]
G.711	64
G.726r32	32
G.726r24	24
G.726r16	16
G.728	16
iLBC*	15.2, 13.3
GSM Full Rate (GSM-FR)	13
G.729 (A/B/AB)	8
G.723r63	6.3
G.723r53	5.3


PCM (G.711)
10010111
Sample 1
10010110
Sample 2
10010101
Sample 3
10010100
Sample 4
10010011
Sample 5
...
10110001
VoIP Header
10010111
10010110
10010101
10010100
10010011
...
10110001
G.711 20 ms of samples (160 bytes)
G.711 30 ms of samples (240 bytes)

Packetization Rate

	20-ms voice length in a packet	30-ms voice length in a packet	40-ms voice length in a packet	60-ms voice length in a packet	80-ms voice length in a packet
Packetization rate	50 p/s	33.3 p/s	25 p/s	16.7 p/s	12.5 p/s
Size of collected G.711 samples for a single packet	160 B	240 B	320 B	480 B	640 B
Uncompressed raw voice bandwidth	64 kb/s	64 kb/s	64 kb/s	64 kb/s	64 kb/s
Layer 3+ uncompressed VoIP bandwidth	80 kb/s	74.7 kb/s	72 kb/s	69.3 kb/s	68 kb/s


Codec Operations
G.729
DSP
Codeword generated every 10 ms
10 ms
10 ms
10 ms
10 ms
10 ms
Codeword
Codeword
Codeword
Codeword
Codeword
Codeword
20 ms
20 ms
VoIP Header
Payload
VoIP Header
Payload
VoIP Header
Payload
By default one packet contains 20 ms of voice: 2 codewords
30 ms packetization period: 3 codewords in one packet

Packetization and Compression Example

G.729

- Layer 3 + bandwidth per call =
  (Voice Payload + Layer 3 Overhead [40B]) x Packets per Second x 8 bits/Byte

	20-ms voice length in a packet	30-ms voice length in a packet
Packetization rate	50 p/s	33.3 p/s
Size of collected, compressed G.729 samples for a single packet	160 B	240 B
Compressed raw voice bandwidth	64 kb/s	64 kb/s
Layer 3+ G.729 VoIP bandwidth	80 kb/s	74.7 kb/s

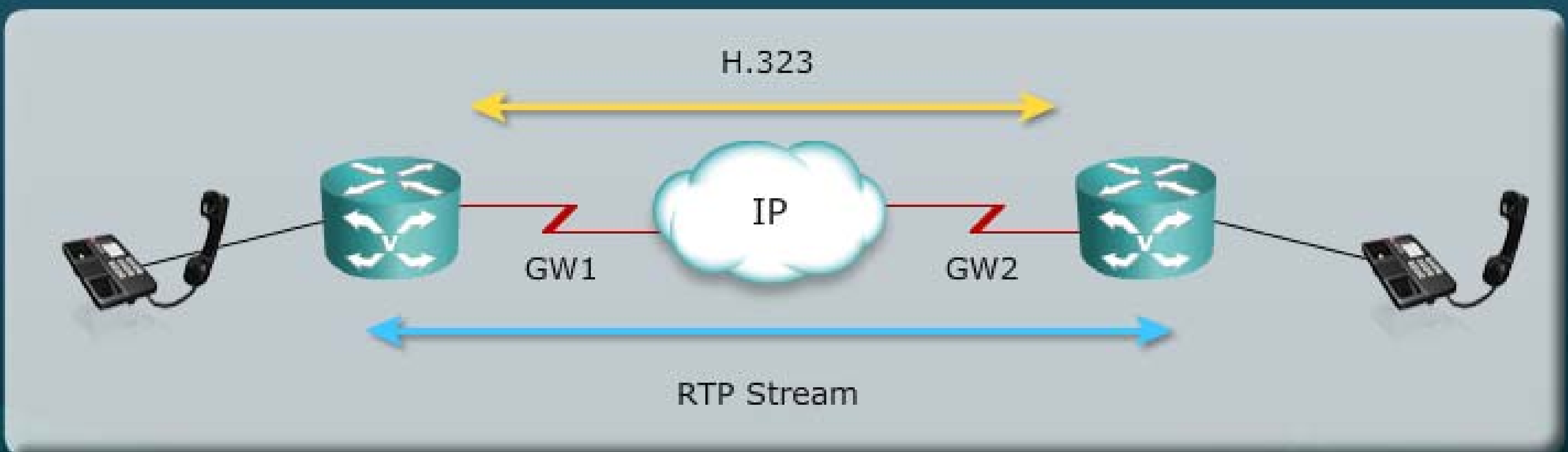
Q/A

Which codec compresses voice at both 6.3 Kbps and 5.3 Kbps?

- A. G.711
- B. G.728
- C. G.723.1
- D. G.729A

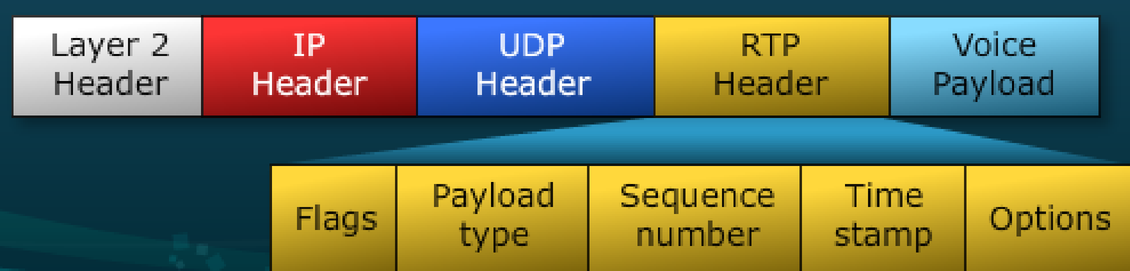
VoIP Gateway

- Real-Time Transport Protocol: Delivers the actual audio and video streams over networks.
- Real-Time Transport Control Protocol: Provides out-of-band control information for an RTP flow.
- cRTP compresses IP/UDP/RTP headers on low-speed serial links.
- SRTP provides encryption, message authentication and integrity, and replay protection to the RTP.



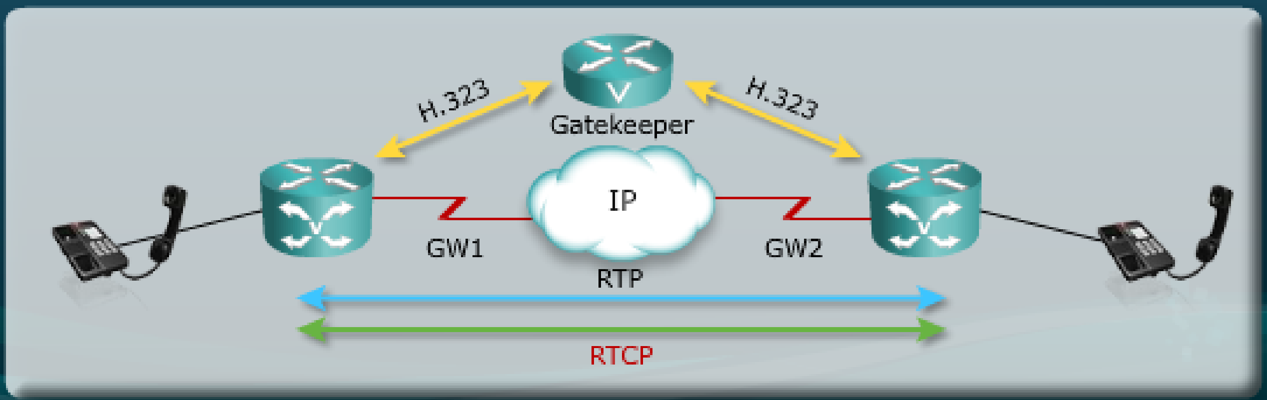
Real-Time Transport Protocol

- Provides end-to-end delivery for real-time data, such as voice and video
- Randomly picks even ports from UDP port range 16384–32767
- Includes the following services:
  - Payload type identification (codec type and media format)
    - Allows the codec to change during transmission, as with fax/modem pass-through
  - Sequence numbering
    - Primarily to detect packet loss
  - Measuring delay/jitter
    - To place packets in the correct timing order (playout delay compensation)



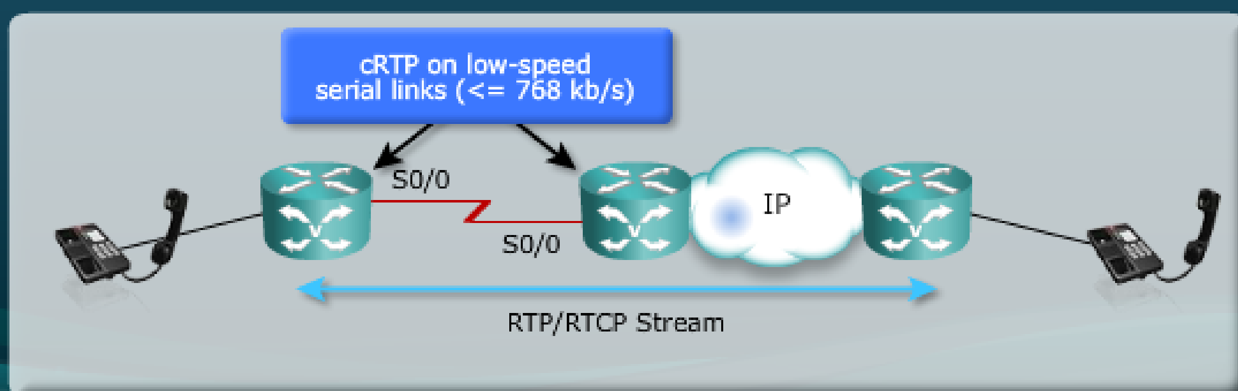
Real-Time Transport Control Protocol

- Monitors media quality and provides control information
- Provides feedback on the RTP session:
  - Packet count
  - Packet delay
  - Octet count
  - Packet loss
  - Jitter (variation in delay)
- Provides a separate flow from RTP for UDP transport use:
  - Is paired with its RTP stream
  - RTP stream UDP port plus 1 (odd-numbered port)



Compressed RTP

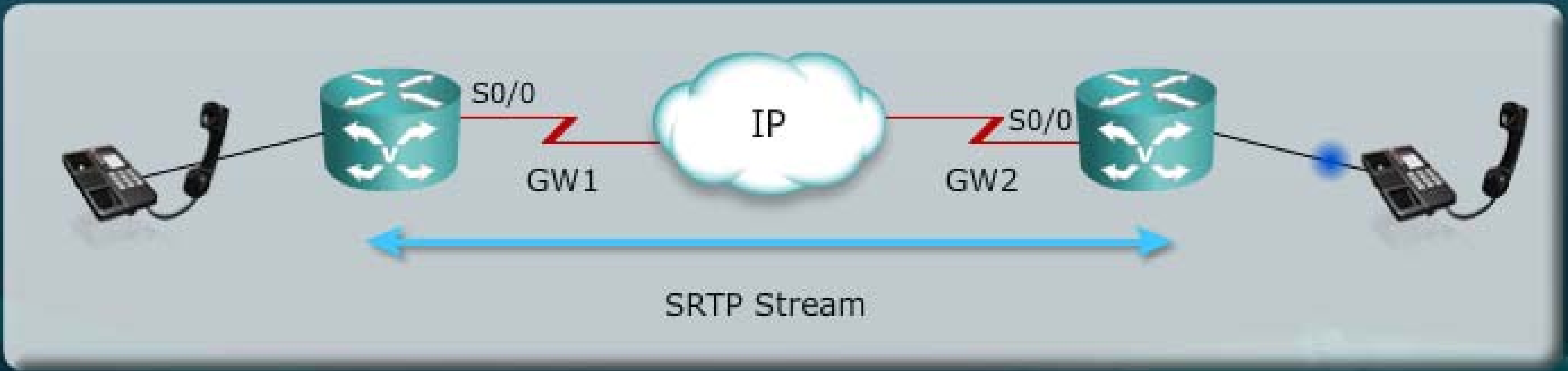
- Maps 40-byte header to 2 (without checksum) or 4 (with checksum) bytes most of the time
- Works point-to-point, must be configured on both ends of the link
- On high-speed links, processing overhead does not justify the bandwidth savings
- Algorithm:
  - Establishes session context (full IP/UDP/RTP headers, few first-order differential values, link sequence number, generation number, and a delta encoding table)
  - Session state shared between the compressor and the decompressor
  - After the context state is established, compressed packets may be sent
  - Only change (delta) indicators are transmitted



Use it on links < 768kbps

Secure RTP

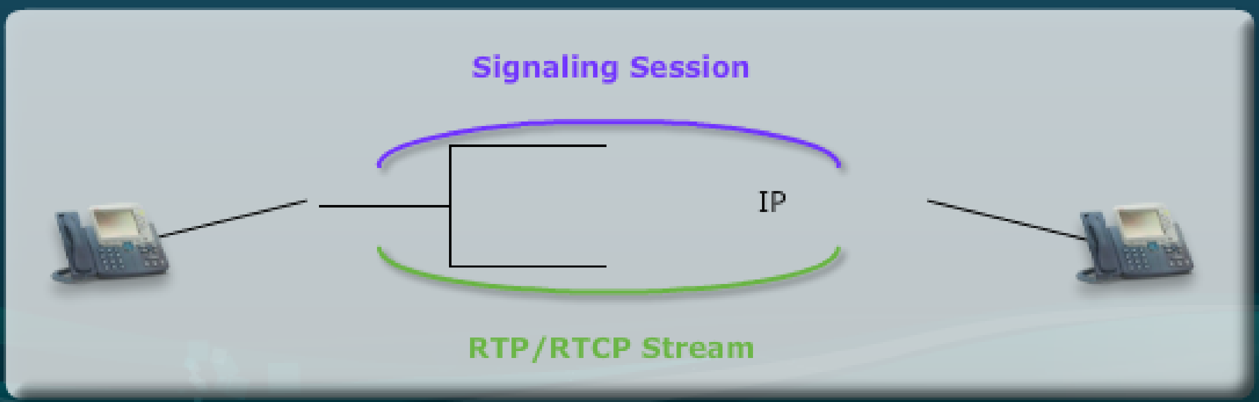
- Encryption
  - Makes the content undecipherable for transit
- Message integrity
  - Adds a fingerprint to detect tampering during transit
- Message authentication
  - Protects the fingerprint with key to guarantee the authenticity of the source
- Replay protection
  - Sequencing prevents the injecting of outdated information



VoIP Media Considerations

Firewalling

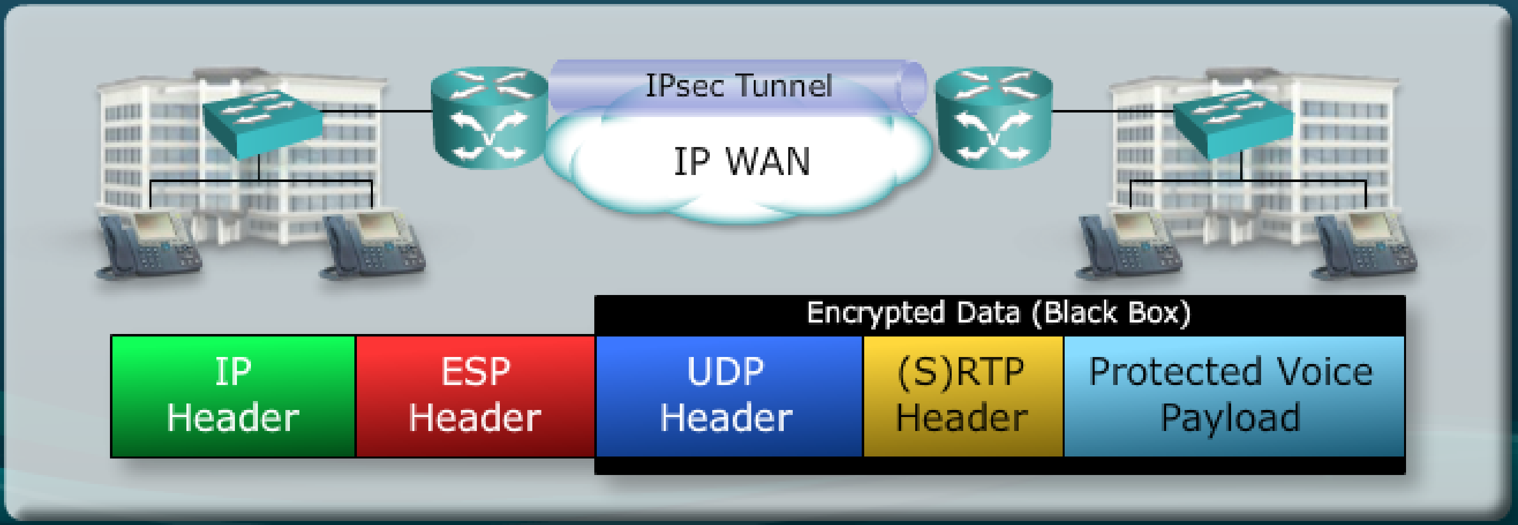
- Signaling sessions use static, well-known ports
  - H.323 (TCP and UDP port 1720), SIP (UDP and TCP port 5060), MGCP (UDP port 2427), SCCP (TCP port 2000)
  - Can be easily allowed through firewalls using static ACLs
- RTP/RTCP use dynamically negotiated UDP ports
  - Difficult to allow through firewalls using static ACLs
  - Stateful firewalls open the RTP/RTCP ports on demand:
    - Works well if RTP/RTCP streams follow the signaling path
    - Transmission fails when RTP and RTCP flows take different paths from signaling



VoIP Media Considerations

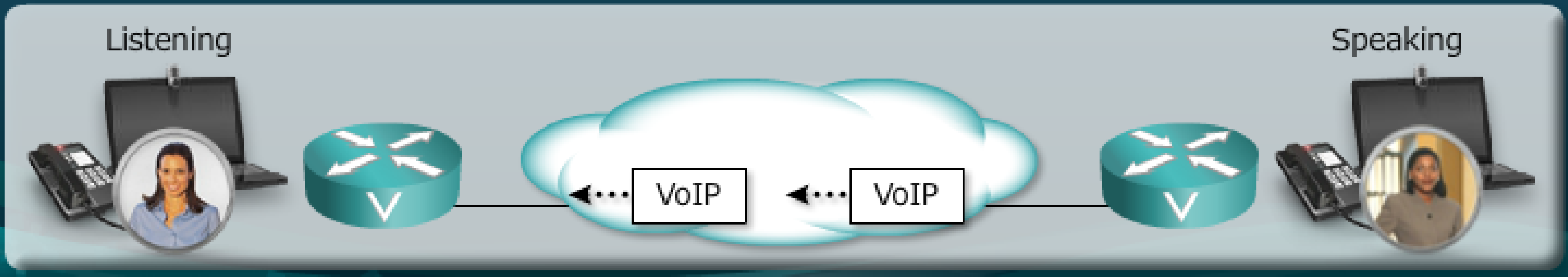
Privacy

- IPsec protection of SRTP packets encrypts already-encrypted data
- Exclude SRTP packets from IPsec protection:
  - To save bandwidth and computational resources
- Prefer SRTP over IPsec:
  - Less overhead
  - More uniform approach (covers other calls, such as from roaming users)



Voice Activity Detection Overview

- Builds on the nature of human conversation
  - One speaks, one listens
- Classifies packets into: speech, silence, and unknown
  - Speech and unknown packets are sent over the network
  - Packets that would carry silence are discarded
- Up to 35 percent bandwidth savings
  - Based on average volume of more than 24 calls
- The sound quality could be slightly degraded by VAD
  - Initial after-silence sounds chopped off



Bandwidth Savings

Codec	Codec speed	Sample size	Frame Relay, no VAD	Frame Relay with VAD
G.711	64 kb/s	240 B	76.3 kb/s	49.6 kb/s
G.711	64 kb/s	160 B	82.4 kb/s	53.6 kb/s
iLBC	13.3 kb/s	30 B	26.1 kb/s	17.0 kb/s
iLBC	15.2 kb/s	20 B	34.4 kb/s	22.4 kb/s
G.729	8 kb/s	30 B	20.3 kb/s	13.2 kb/s
G.729	8 kb/s	20 B	26.4 kb/s	17.2 kb/s

Voice Port Settings for VAD

Music Threshold and Comfort Noise

- Voice Activity Detection active when:
  - Not disabled in the matched VoIP dial peer
  - The negotiated codec supports it
- Tunable voice port parameters:
  - Minimal decibel level of music-on-hold
    - Defines loudness threshold to correctly interpret and transmit MOH
  - Local generation of comfort noise
    - Local telephone hears comfort noise during silence of the other end