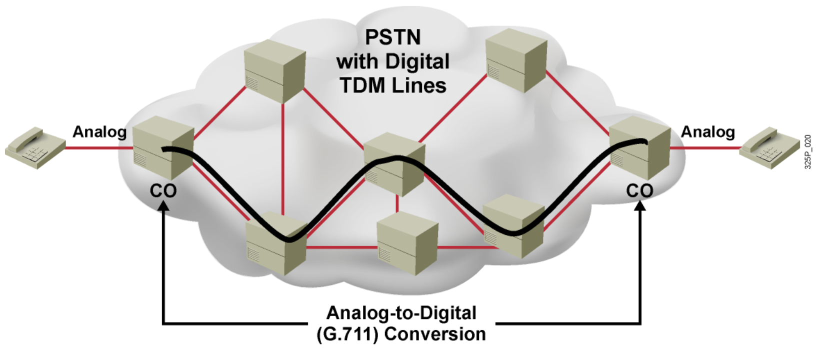
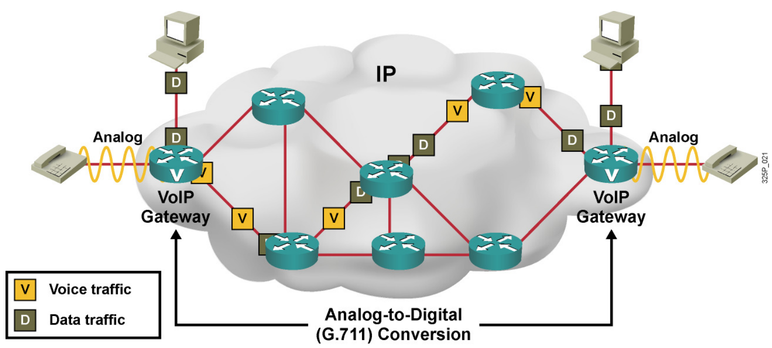
Voice Transport in Circuit-Switched Networks



- Analog phones connect to CO switches.
- CO switches convert between analog and digital.
- After call is set up, PSTN provides:
  - End-to-end dedicated circuit for this call (DS-0)
  - Synchronous transmission with fixed bandwidth and very low, constant delay

Voice Transport in VoIP Networks



- Analog phones connect to voice gateways.
- Voice gateways convert between analog and digital.
- After call is set up, IP network provides:
  - Packet-by-packet delivery through the network
  - Shared bandwidth, higher and variable delays

Jitter

- Voice packets enter the network at a constant rate.
- Voice packets may arrive at the destination at a different rate or in the wrong order.
- Jitter occurs when packets arrive at varying rates.
- Since voice is dependent on timing and order, a process must exist so that delays and queuing issues can be fixed at the receiving end.
- The receiving router must:
  - Ensure steady delivery (delay)
  - Ensure that the packets are in the right order

VoIP Protocol Issues

- IP does not guarantee reliability, flow control, error detection or error correction.
- IP can use the help of transport layer protocols TCP or UDP.
- TCP offers reliability, but voice doesn't need it…do not retransmit lost voice packets.
- TCP overhead for reliability consumes bandwidth.
- UDP does not offer reliability. But it also doesn't offer sequencing…voice packets need to be in the right order.
- RTP, which is built on UDP, offers all of the functionality required by voice packets.

Protocols Used for VoIP

Feature	Voice Needs	TCP	UDP	RTP
Reliability	No	Yes	No ✓	No ✓
Reordering	Yes	Yes ✓	No	Yes ✓
Time-stamping	Yes	No	No	Yes ✓
Overhead	As little as possible	Contains unnecessary information	Low ✓	Low ✓
Multiplexing	Yes	Yes ✓	Yes ✓	No

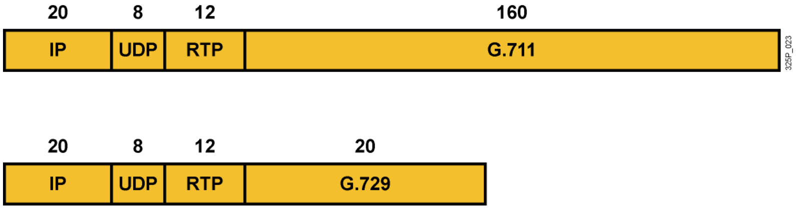
Voice Encapsulation

20	8	12	160
IP	UDP	RTP	G.711 (20 ms of Voice)

20	8	12	20
IP	UDP	RTP	G.729 (20 ms of Voice)

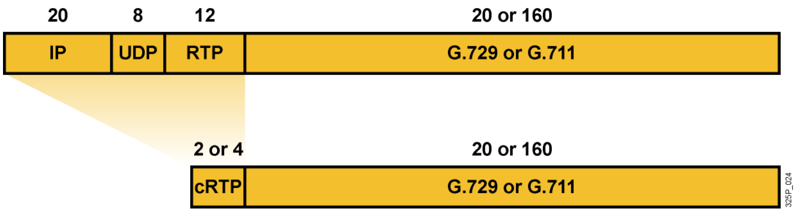
- Digitized voice is encapsulated into RTP, UDP, and IP.
- By default, 20 ms of voice is packetized into a single IP packet.

Voice Encapsulation Overhead



- Voice is sent in small packets at high packet rates.
- IP, UDP, and RTP header overheads are enormous:
  - For G.729, the headers are twice the size of the payload.
  - For G.711, the headers are one-quarter the size of the payload.
- Bandwidth is 24 kbps for G.729 and 80 kbps for G.711, ignoring Layer 2 overhead.

RTP Header Compression



- Compresses the IP, UDP, and RTP headers
- Is configured on a link-by-link basis
- Reduces the size of the headers substantially (from 40 bytes to 2 or 4 bytes):
  - 4 bytes if the UDP checksum is preserved
  - 2 bytes if the UDP checksum is not sent
- Saves a considerable amount of bandwidth

cRTP Operation

Condition	Action
The change is predictable.	The sending side tracks the predicted change.
The predicted change is tracked.	The sending side sends a hash of the header.
The receiving side predicts what the constant change is.	The receiving side substitutes the original stored header and calculates the changed fields.
There is an unexpected change.	The sending side sends the entire header without compression.

When to Use RTP Header Compression

- Use cRTP:
  - Only on slow links (less than 2 Mbps)
  - If bandwidth needs to be conserved
- Consider the disadvantages of cRTP:
  - Adds to processing overhead
  - Introduces additional delays
- Tune cRTP—set the number of sessions to be compressed (default is 16).

© 2006 Cisco Systems, Inc. All rights reserved.