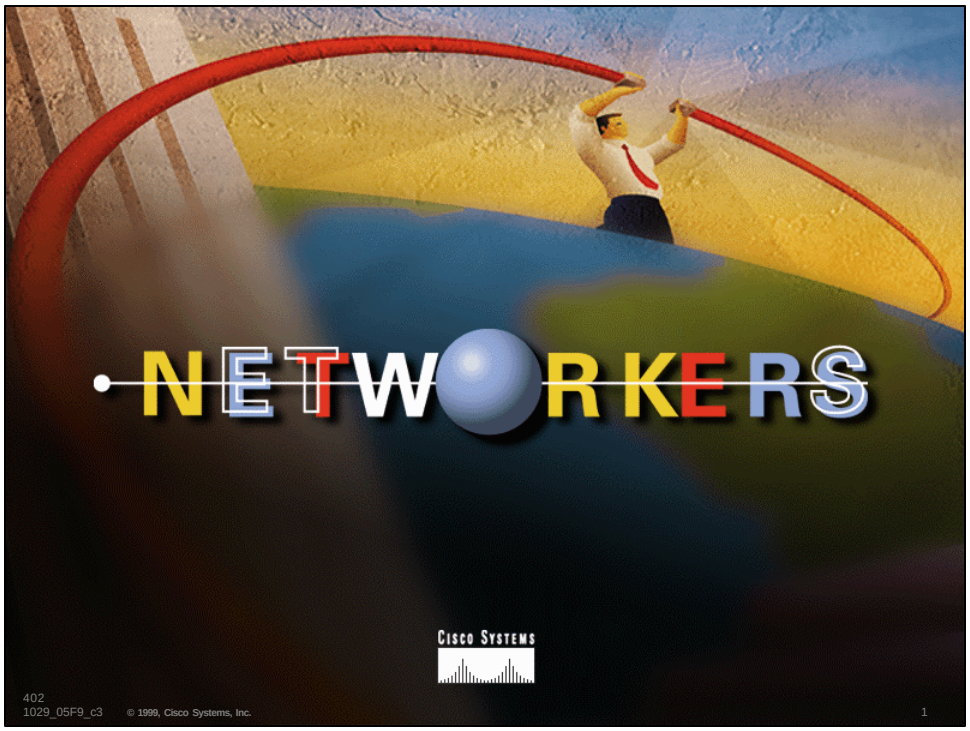

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Introduction

"

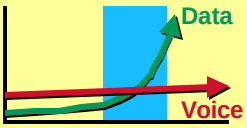
Voice Is "Just" Another Latency-Sensitive IP Application

"

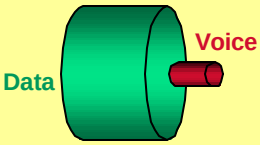
Agenda

- **Why Multiservice Networks?**
- **What Is Multiservice?**
  - **Voice Networking Overview**
  - **Voice Over Data Network Transport Mechanisms**
  - **Multiservice Network Architectures**
- **How Does Voice Over IP Transport Work?**
  - **Applications**
  - **Challenges and Solutions**
- **How Does an IP Phone System Work?**
- **When Can I Implement IP Multiservice?**

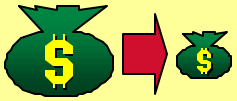
Multiservice—Why Now?



Data Exceeds Voice Over Voice Net

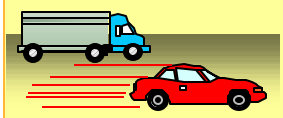


Bigger, Cheaper Bandwidth



Cost Savings
Network Consolidation
Usage Charges
Toll Charges

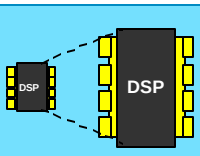
QoS Advances



IP Precedence, RSVP
802.1p, CiscoAssure

Open Standards

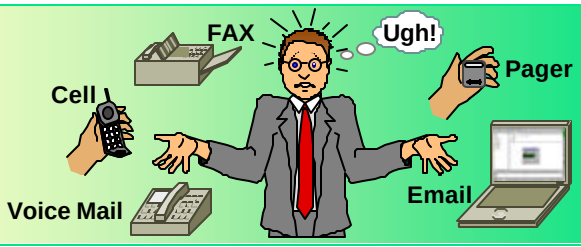
H.323/MGCP
FRF.11/12
AAL1/2/5



Digital Signal Processor Advances

New Integrated Applications (The Big Ones!)

Unified Messaging



Network Enabled Call Centers

- Click-to-Talk Web
- Click-to-Call Back
- Skills-Based Routing
- Assisted Web Navigation
- Computer Telephony Integration (CTI)
- Work From Home

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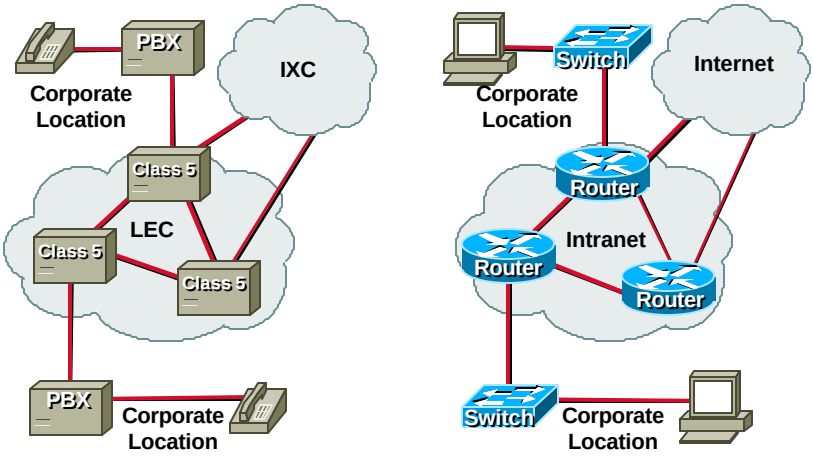
Telephony and Data Architecture Fundamentals Comparison

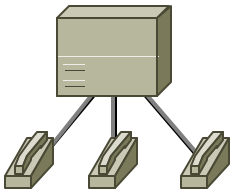


Corporate Voice Network with PBXs (Private Branch Exchange)

Corporate Data Network with Switches and Routers

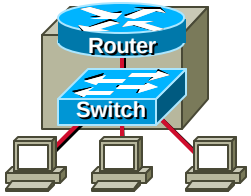
Voice and Data Switching Comparison

Class 5 Switch



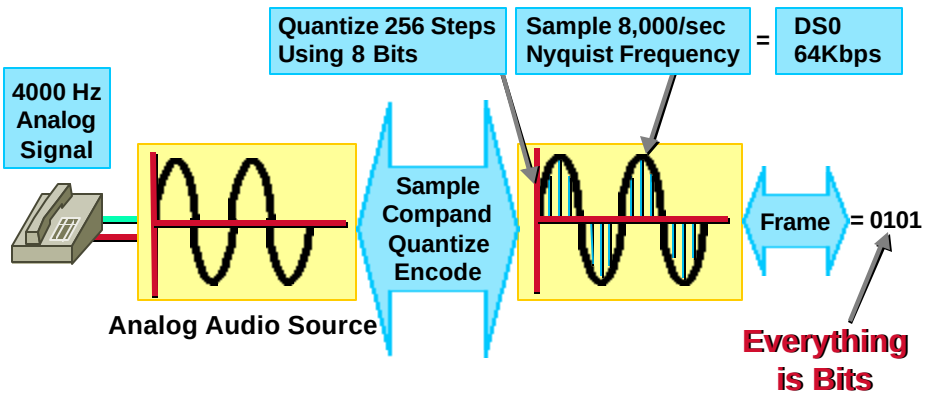
- Handset aggregator
- All telephones get a single analog/digital line (DS0)
- All devices have a phone number defined on the switch
- All devices can simultaneously make a call (calls < trunk DS0s)
- Path selection based on static least cost routing or ARS

Multilayer Switch



- Computer aggregator
- All devices get dedicated bandwidth 10/100/1000 Mbps (autonegotiation)
- All devices have an IP address defined on the host
- All devices run at full line rate (bandwidth < uplink)
- Path selection based on dynamic routing protocol lowest cost

Voice Network Least Common Denominator

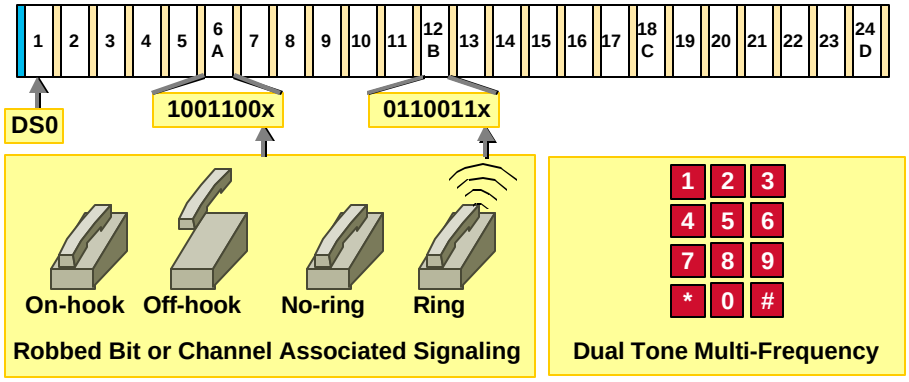


Pulse Code Modulation (PCM) DS0

Digital Signaling Scheme T1/DS1

DS1—Extended Super Frame (8 X 24 Bytes = 192 Bit Frame)

T1—Coding (Ones Density)—AMI, ZCS, B8ZS



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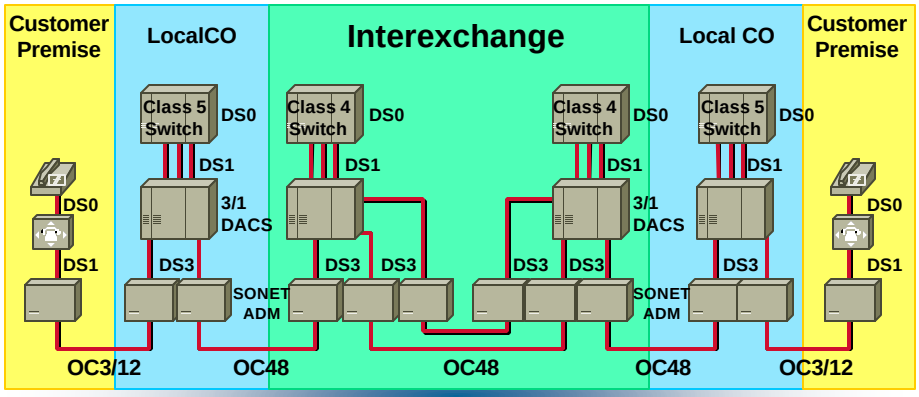
Today's Carrier Voice Infrastructure



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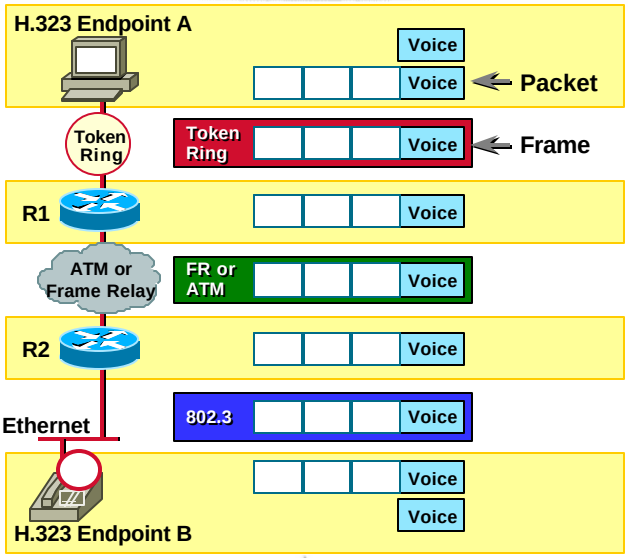
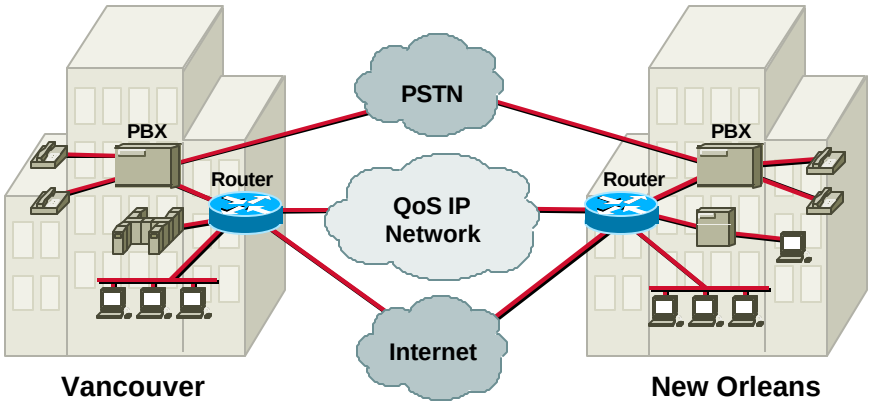
Voice Transport Mechanisms

Layer 3—VoIP	Layer 2—VoFR, VoATM
• Operates in heterogeneous network (ubiquitous)	• Requires rigid homogenous network or L2 gateways
• Connectionless (requires sequence numbers)	• Connection oriented (frames arrive in order)
• "Soft" QoS	• "Hard" QoS
• Layer 2 and 3 overhead	• Layer 2 overhead
• Standards-based H.323 (MGCP coming)	• Standards based (FRF.11/12, ATM AAL1/2/5)

IP Ubiquity



Enterprise Multiservice IP Core Backbone



Reduced Toll/Circuit Costs
Infrastructure Consolidation
Efficient Bandwidth Consumption

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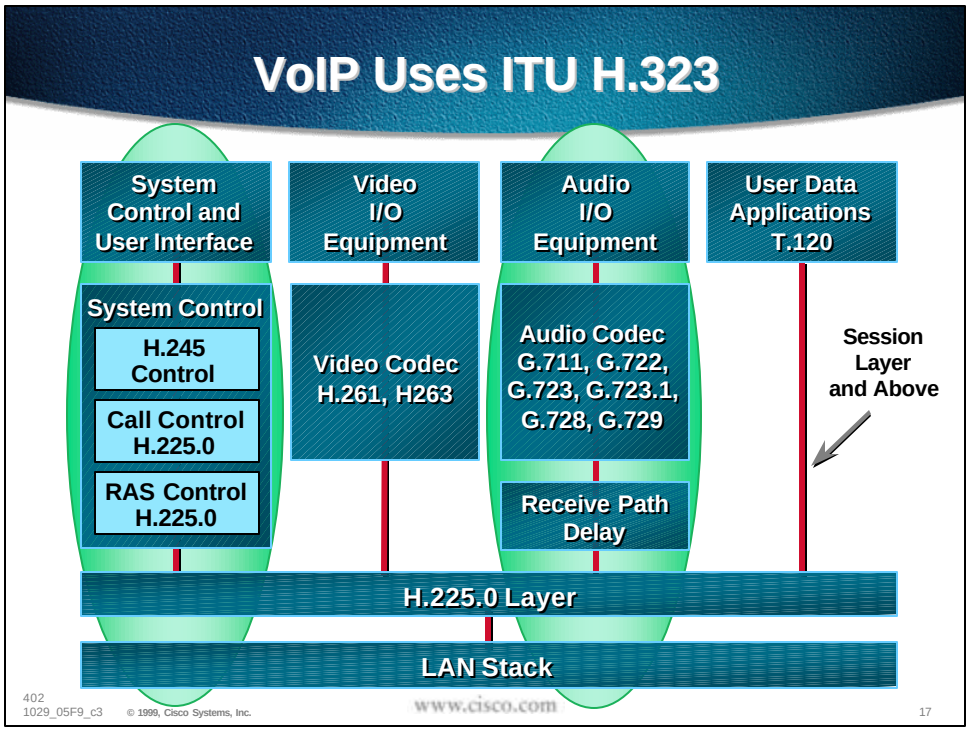
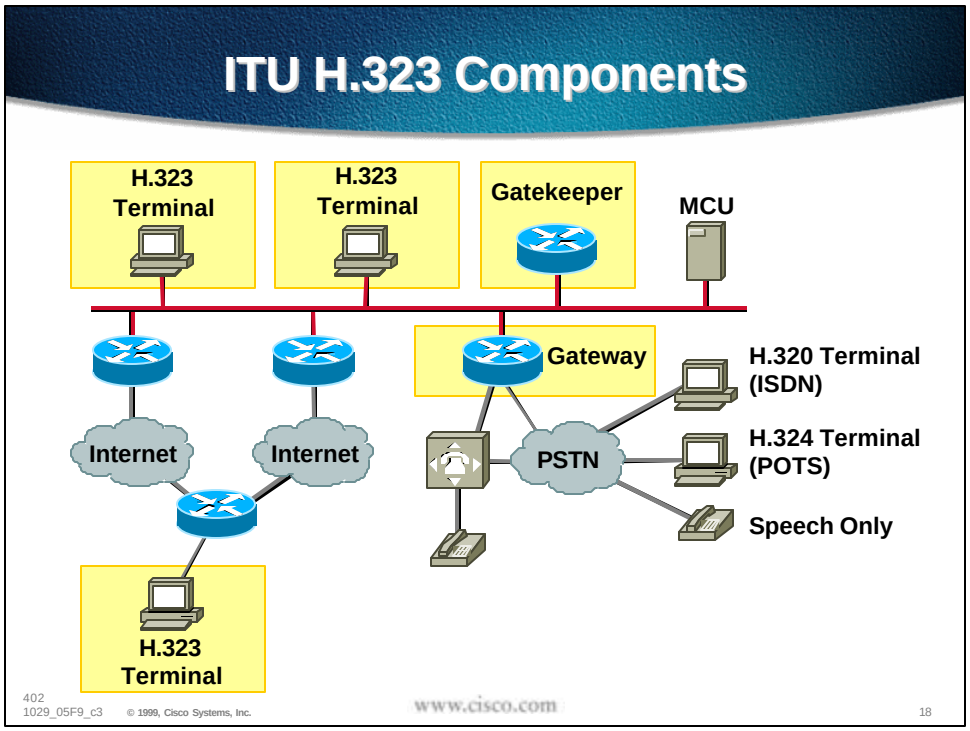
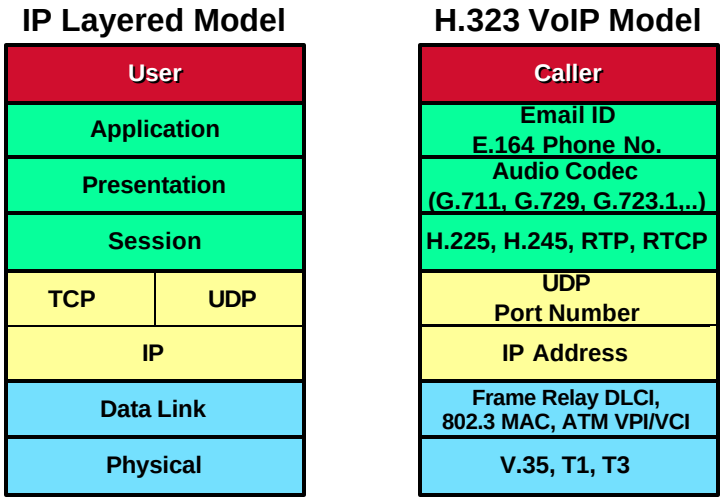




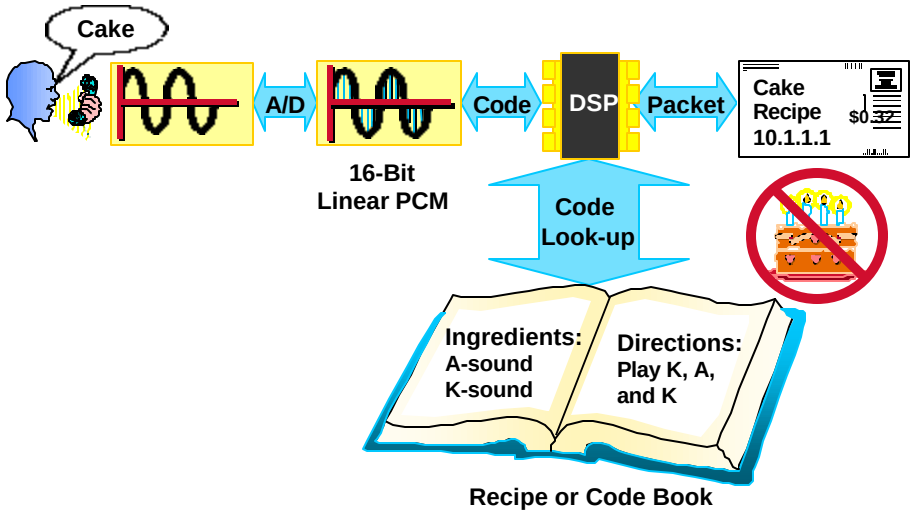
H.323 VoIP Layers



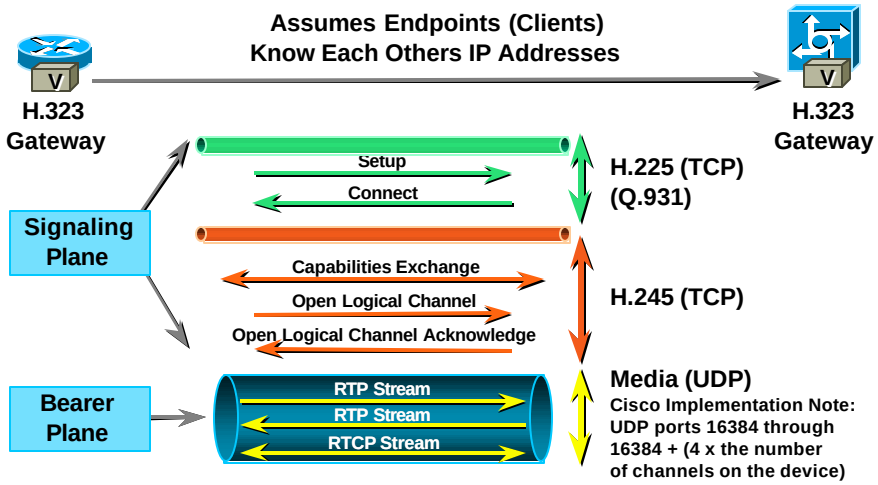
Voice CODEC Cheat Sheet

Encoding Compression	Mean Opinion Score	Native Bit Rate Kbps	Voice Quality	BW	DTMF	Dual Comp	CPU	Music on Hold
G.711 PCM	4.1	64	A	D	A	A	A	A
G.726 ADPCM	3.85	32	B	C	B	B	B	B
G.728 LD-CELP	3.61	16	C	B	B	C	C	C
G.729 CS-ACELP	3.92	8	A	A	B	B	C	C
G.729a CS-ACELP	3.7	8	B	A	C	C	B	D
G.723.1 ACELP	3.65	5.3	C	A	C	D	C	D

Code Excited Linear Prediction (CELP)

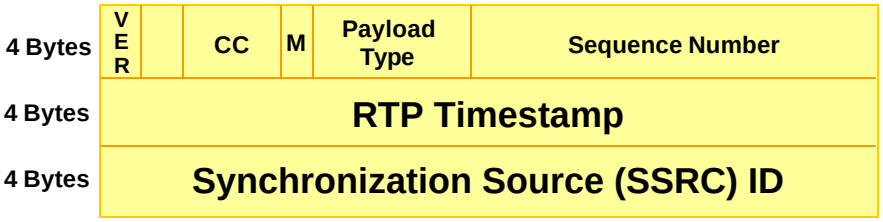


H.323 End Point to End Point Signaling

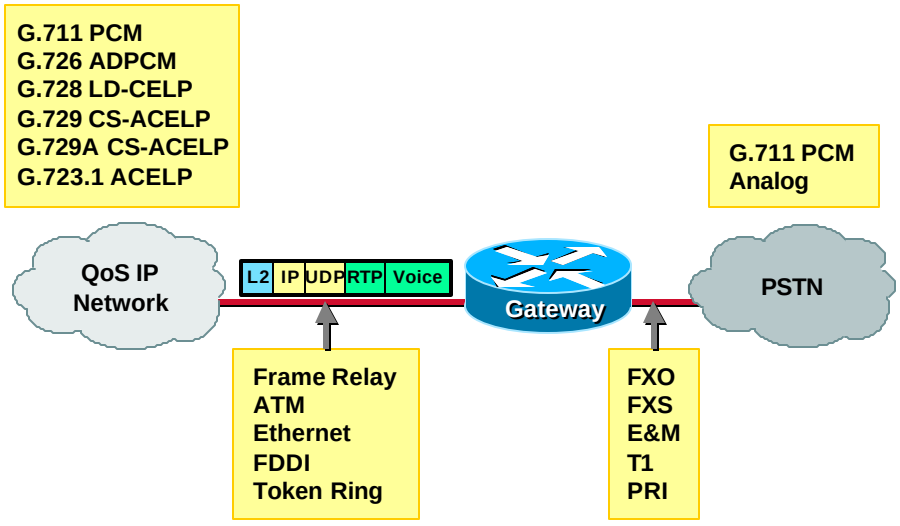


RTP/RTCP—RFCs 1889/1890

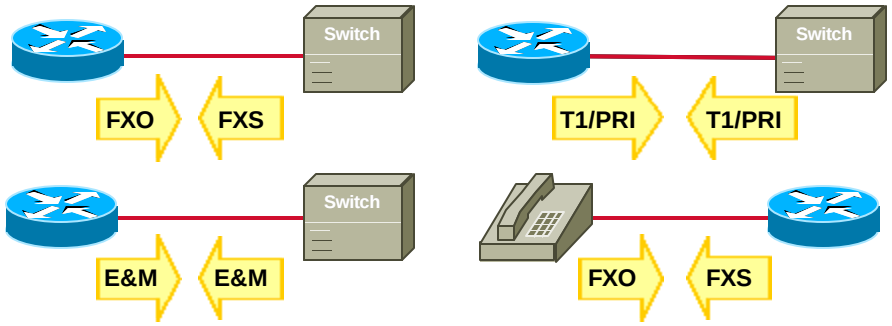
- End-to-end network transport function
  - Payload type identification—voice, video, compression type
  - Sequence numbering
  - Time stamping
  - Delivery monitoring
- RTCP (Real-Time Control Protocol)



H.323 Gateway



Router Voice Interfaces



- FXO—Foreign Exchange Office
- FXS—Foreign Exchange Station
- E&M—Ear and Mouth
- PRI—Primary Rate Interface

Dial-Peer Configuration (Static Routing)


```
r1#
dial-peer voice 1234 pots
 destination-pattern 1234
 port 1/0/0
!
dial-peer voice 4000 voip
 destination-pattern 4...
 session target ipv4:10.1.1.2
dial-peer voice 4000 vofr
 destination-pattern 4...
 session target serial0 122
dial-peer voice 4000 voatm
 destination-pattern 4...
 session target serial0 1
```

Local

VoIP

VoFR

VoATM

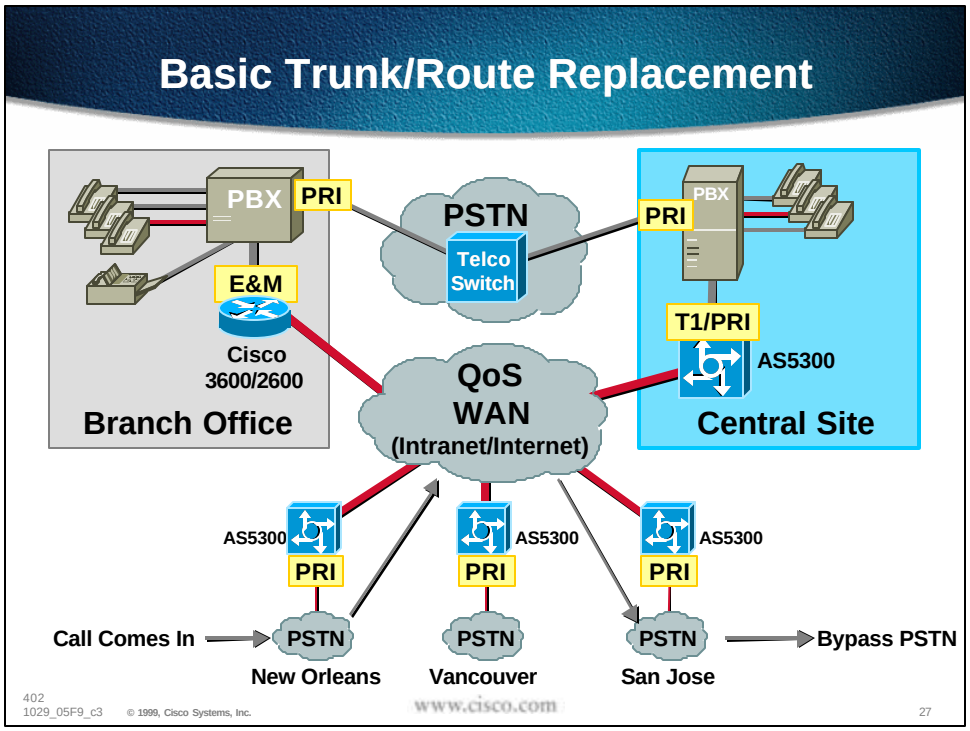
```
r2#
dial-peer voice 4321 pots
 destination-pattern 4321
 port 1/0/0
!
dial-peer voice 1000 voip
 destination-pattern 1...
 session target ipv4:10.1.1.1
dial-peer voice 1000 vofr
 destination-pattern 1...
 session target serial0 221
dial-peer voice 1000 voatm
 destination-pattern 1...
 session target serial0 1
```

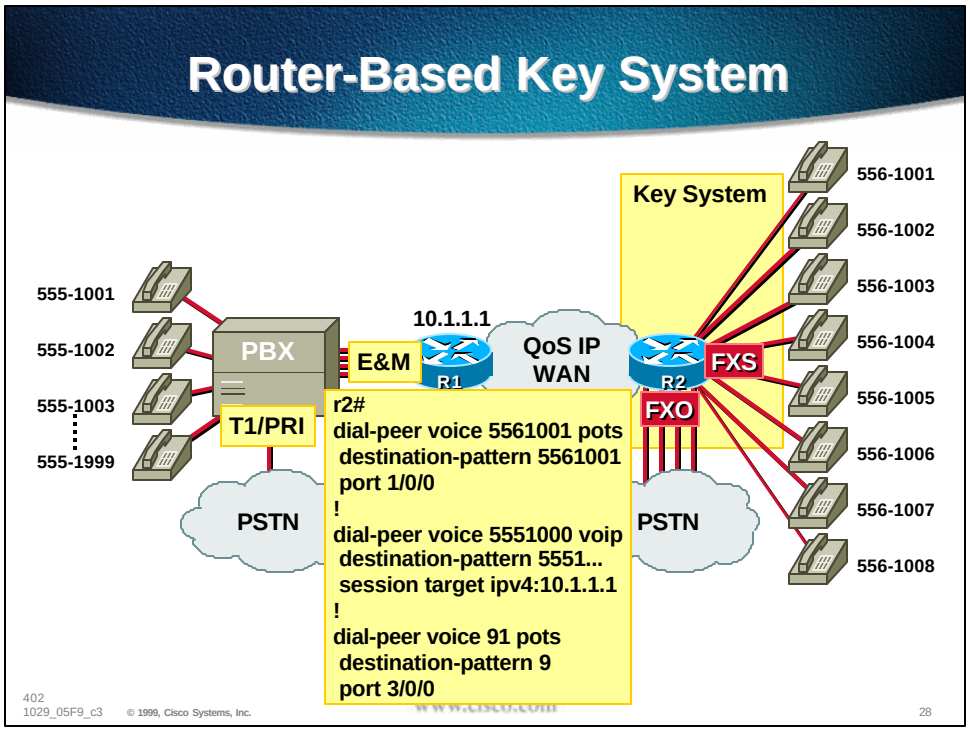
Local Port

IP Address

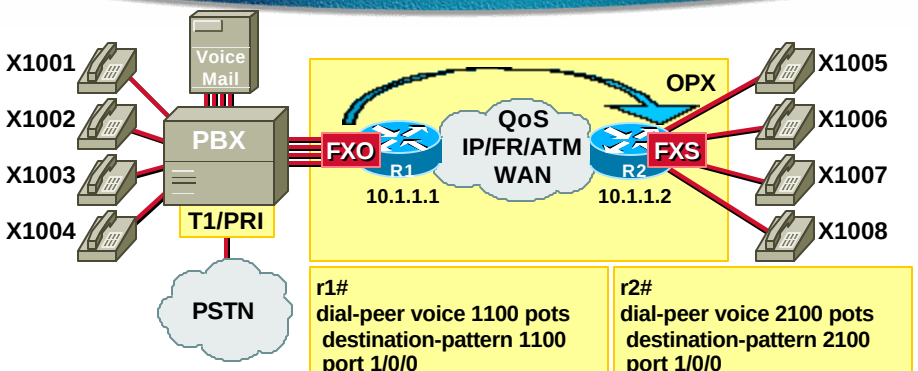
Interface DLCI

ATM VCD





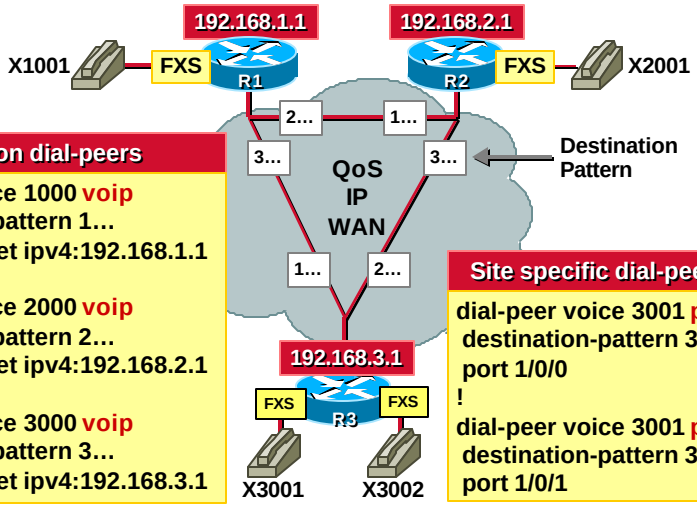
Off Premise Extension (OPX)




```
r1#
dial-peer voice 1100 pots
 destination-pattern 1100
 port 1/0/0
!
dial-peer voice 2000 voip
 destination-pattern 2...
 session target ipv4:10.1.1.2
!
voice-port 1/0/0
 connection trunk 2100
```

```
r2#
dial-peer voice 2100 pots
 destination-pattern 2100
 port 1/0/0
!
dial-peer voice 1000 voip
 destination-pattern 1...
 session target ipv4:10.1.1.1
!
voice-port 1/0/0
 connection trunk 1100
```


VoIP Full Mesh Dial-Peers



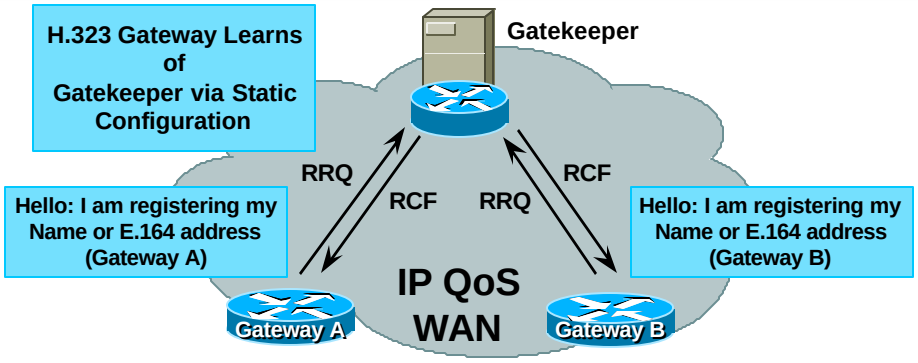
Common dial-peers

```
dial-peer voice 1000 voip
 destination-pattern 1...
 session target ipv4:192.168.1.1
!
dial-peer voice 2000 voip
 destination-pattern 2...
 session target ipv4:192.168.2.1
!
dial-peer voice 3000 voip
 destination-pattern 3...
 session target ipv4:192.168.3.1
```

Site specific dial-peers

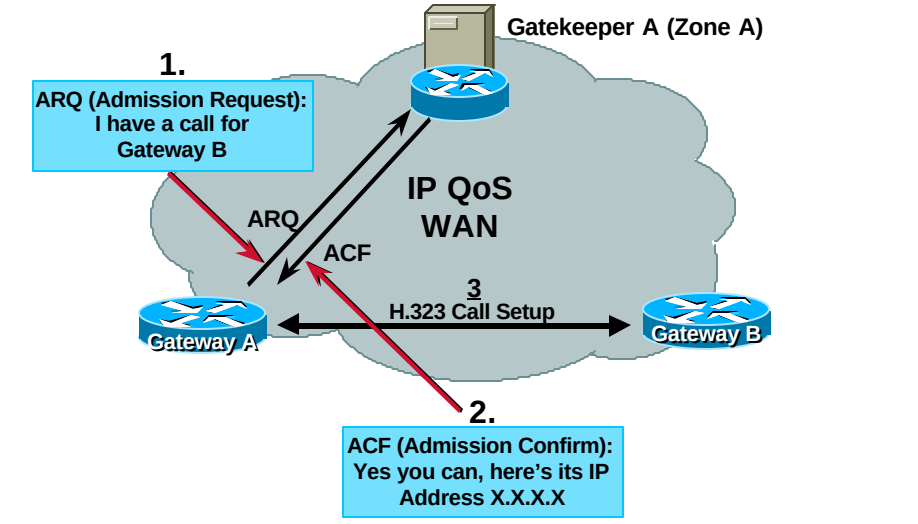
```
dial-peer voice 3001 pots
 destination-pattern 3001
 port 1/0/0
!
dial-peer voice 3001 pots
 destination-pattern 3002
 port 1/0/1
```

H.323 Gatekeeper Call Control/Signaling Gatekeeper Registration



RAS—Registration Admission and Status
UDP Transport Port 1719
RRQ—Registration Request
RRJ—Registration Reject
RCF—Registration Confirm

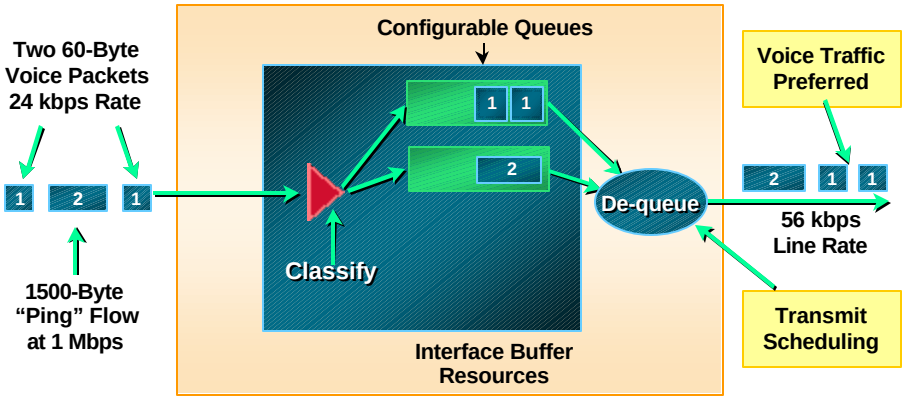
H.323 Gatekeeper Dial-Peer Scaleability



VoIP Low Speed Link (<768 Kbps) Challenges and Solutions

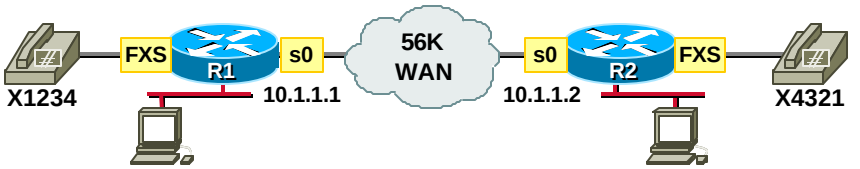
Challenge	Cisco Solutions
Congestion Delay and Delay Jitter	**Intelligent Queuing** WFQ, IP Precedence, RSVP, Priority Queuing
Packet Residency Slow Link Freeze-out by Large Packets	**Interleaving** FRF.12, MLPPP, IP MTU Size Reduction, Faster Link
Bandwidth Consumption Header Size on Low Bandwidth Links	**Compression** Codecs, RTP Header Compression, Voice Activity Detection
WAN Oversubscription, Bursting	**Traffic Management** Router Traffic Shaping to CIR, High Priority PVC, Data Discard Eligibility

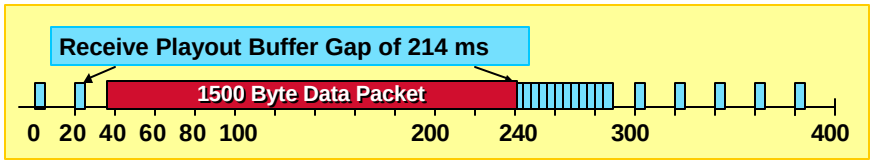
Congestion Avoidance Solutions Intelligent Queuing

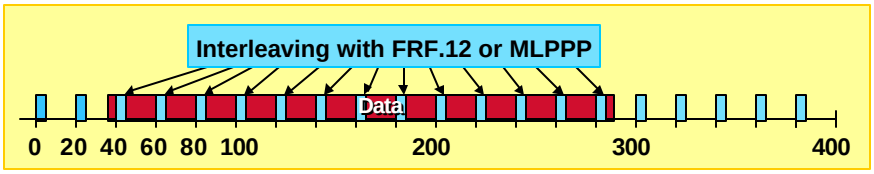


Weighted Fair Queuing (WFQ)
Resource Reservation Protocol (RSVP)
Priority Queuing

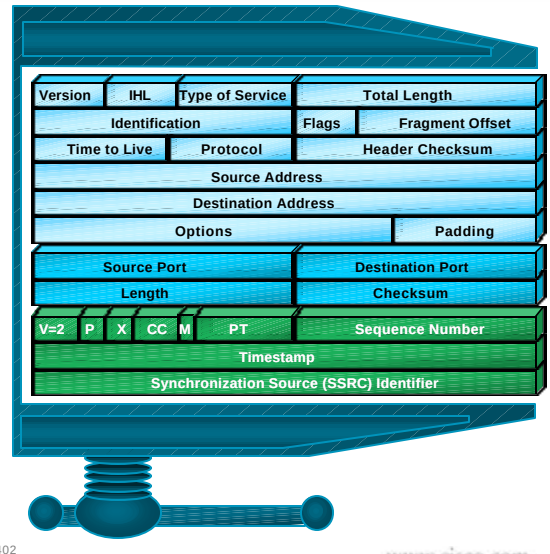
Packet Residency Solutions







VoIP Bandwidth Solution



RTP Header Compression

- 20 ms @ 8 kbps yields 20-byte payload
- IP header 20;
  UDP header 8;
  RTP header 12

  2X payload!
- Header compression 40 bytes to 2 or 4 bytes
- Hop-by-Hop on slow links <512 kbps
- CRTP—Compressed Real-time Protocol

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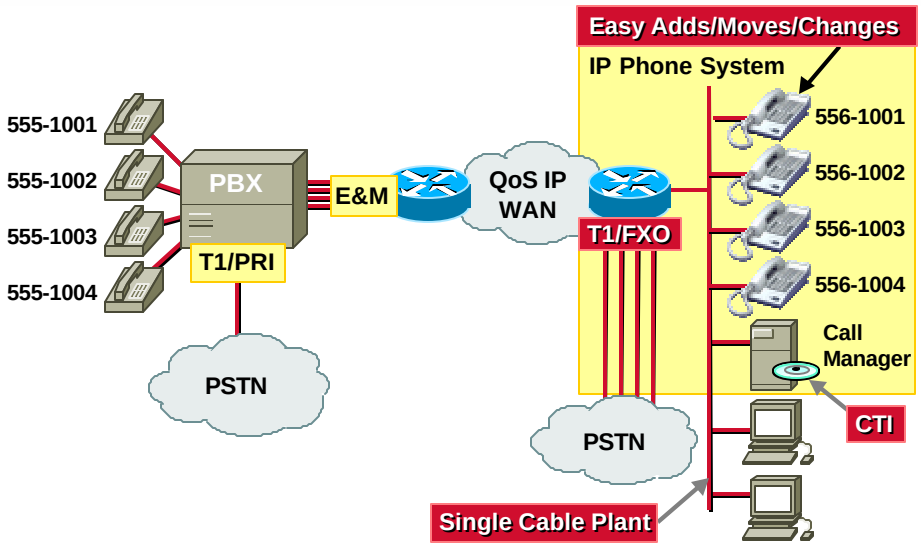
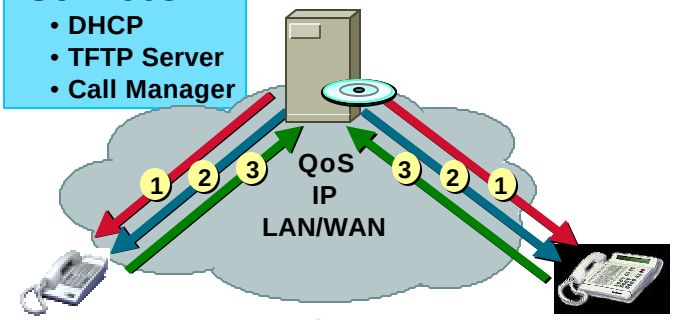
IP Phone System



Phone Initialization

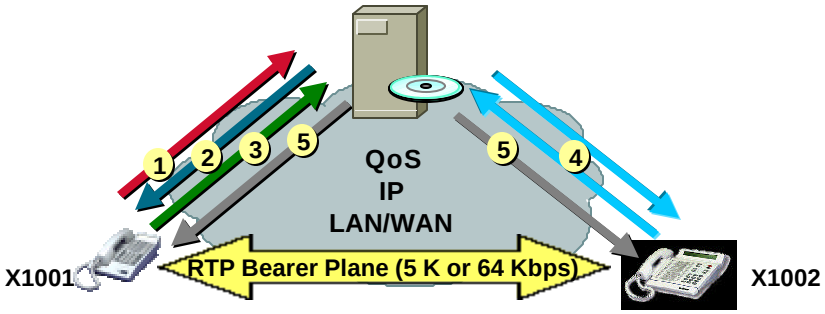
1. Phones make DHCP request to get an IP address, gateway, boot server, etc.
2. Phones make TFTP boot file request to get CM IP addresses and ports
3. Phones register with CM and get templates
4. Phones display CM time and date and are ready to receive/place calls

Services:
- DHCP
- TFTP Server
- Call Manager



Phone Call Control

1. Calling phone sends off-hook message to CM
2. CM directs phone to play dial-tone
3. Phone sends dialed digits to CM as they are collected
4. CM rings called party phone and accepts off-hook message
5. Calling phone initiates bearer VoIP RTP session with called phone
6. Call Manager is notified of disconnect and records call details



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Planning and Implementation

- Today
  - Tie-line replacement
  - Toll-bypass
  - Off Premise Extension (OPX)
  - Router key system replacement
  - Small office IP phone system (< 100 users)
- Tomorrow
  - Virtual call centers
  - Campus IP phone system (> 1000 users)
  - Enhanced integrated data/voice applications
  - Unified messaging

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