



## Introduction

“

**Voice Is “Just” Another  
Latency-Sensitive  
IP Application**

”

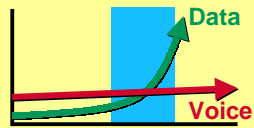
[www.cisco.com](http://www.cisco.com)

## Agenda

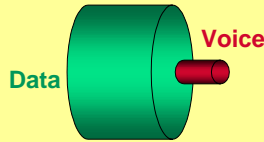
- **Why Multiservice Networks?**
- **What Is Multiservice?**
  - Voice Networking Overview
  - Voice Over Data Network Transport Mechanisms
  - Multiservice Network Architectures
- **How Does Voice Over IP Transport Work?**
  - Applications
  - Challenges and Solutions
- **How Does an IP Phone System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

[www.cisco.com](http://www.cisco.com)

## Multiservice—Why Now?



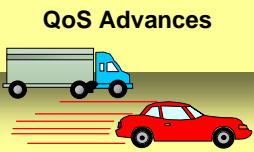
Data Exceeds Voice  
Over Voice Net



Bigger,  
Cheaper Bandwidth



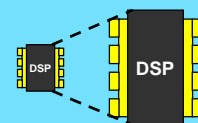
Cost Savings  
Network Consolidation  
Usage Charges  
Toll Charges



QoS Advances  
IP Precedence, RSVP  
802.1p, CiscoAssure

Open Standards

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

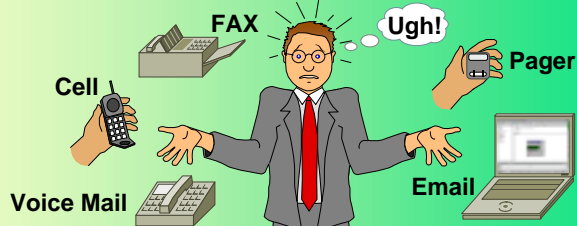


Digital Signal  
Processor Advances

www.cisco.com

## 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

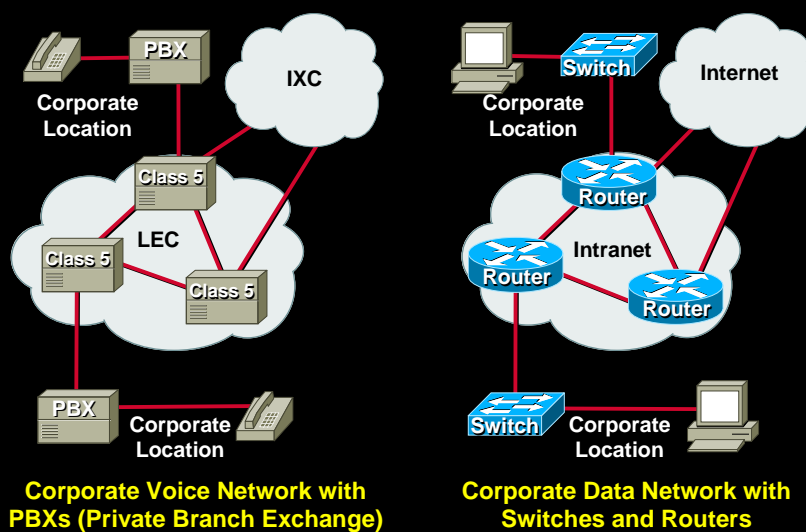
www.cisco.com

## 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?

www.cisco.com

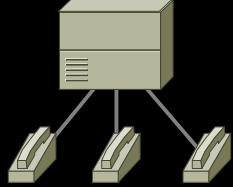
## Telephony and Data Architecture Fundamentals Comparison



www.cisco.com

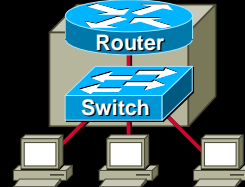
# Voice and Data Switching Comparison

## 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

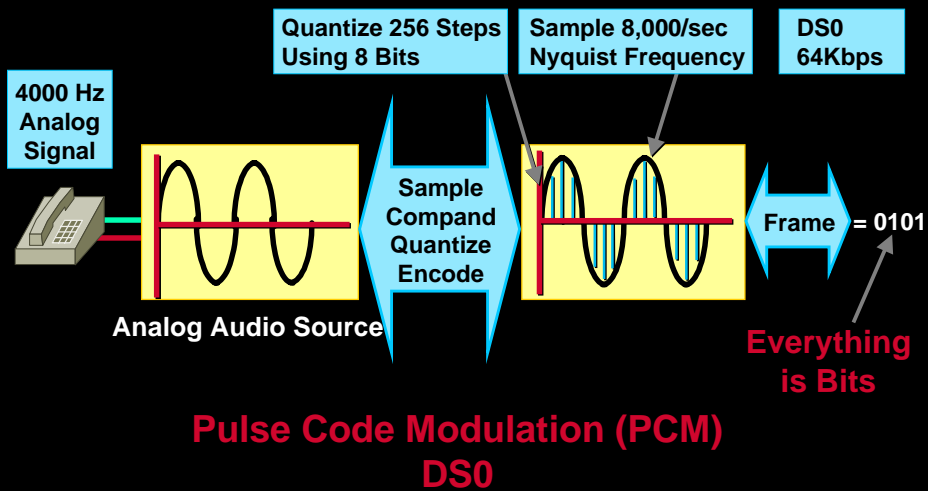
## Multilayer Switch



- 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

www.cisco.com

# Voice Network Least Common Denominator

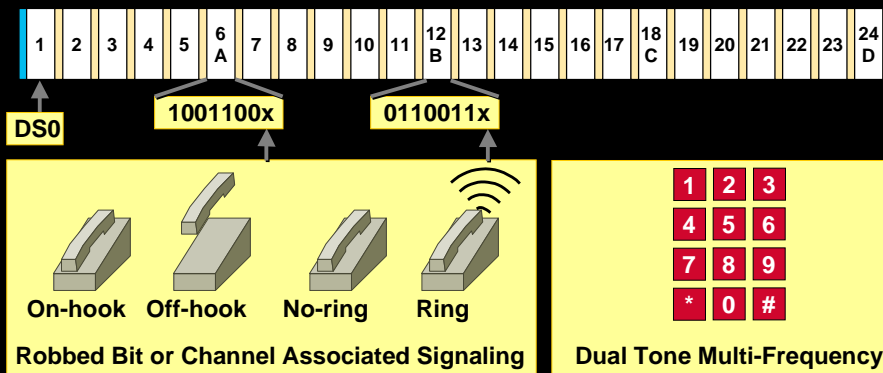


www.cisco.com

# Digital Signaling Scheme T1/DS1

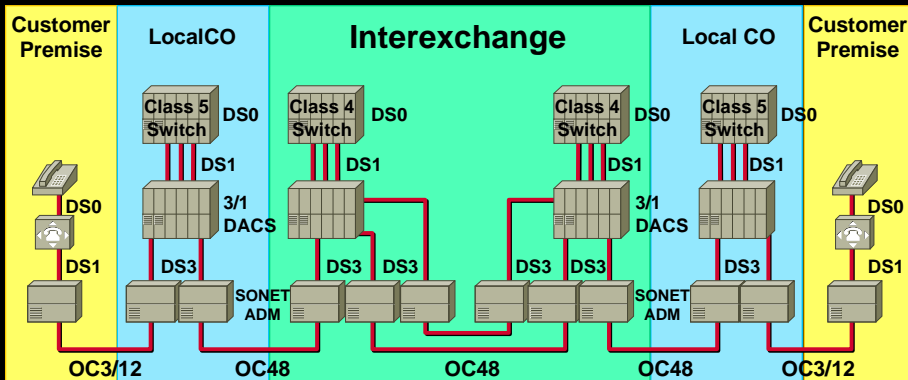
DS1—Extended Super Frame (8 X 24 Bytes = 192 Bit Frame)

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



www.cisco.com

# Today's Carrier Voice Infrastructure



www.cisco.com

## Voice Transport Mechanisms

### Layer 3—VoIP

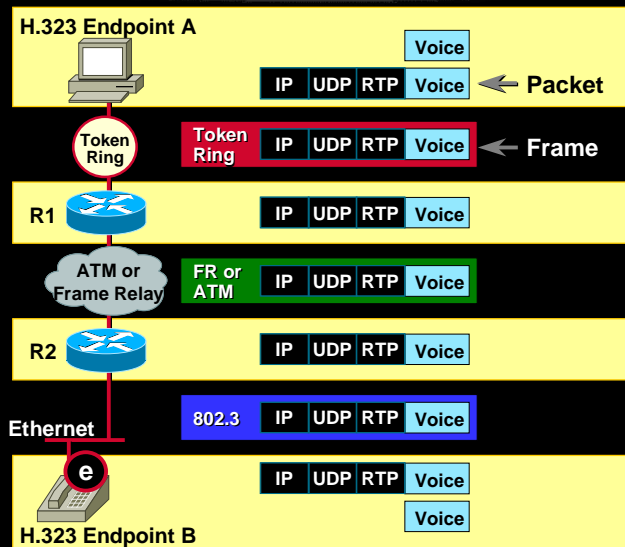
- Operates in heterogeneous network (ubiquitous)
- Connectionless (requires sequence numbers)
- “Soft” QoS
- Layer 2 and 3 overhead
- Standards-based H.323 (MGCP coming)

### Layer 2—VoFR, VoATM

- Requires rigid homogenous network or L2 gateways
- Connection oriented (frames arrive in order)
- “Hard” QoS
- Layer 2 overhead
- Standards based (FRF.11/12, ATM AAL1/2/5)

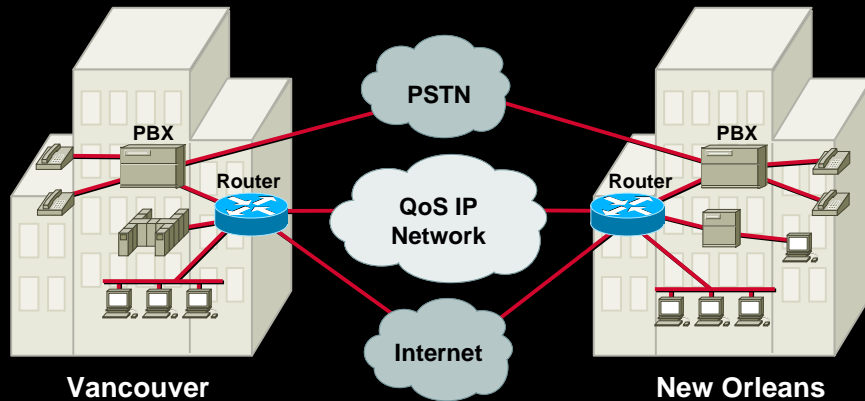
www.cisco.com

## IP Ubiquity



www.cisco.com

## Enterprise Multiservice IP Core Backbone



**Reduced Toll/Circuit Costs**  
**Infrastructure Consolidation**  
**Efficient Bandwidth Consumption**

[www.cisco.com](http://www.cisco.com)

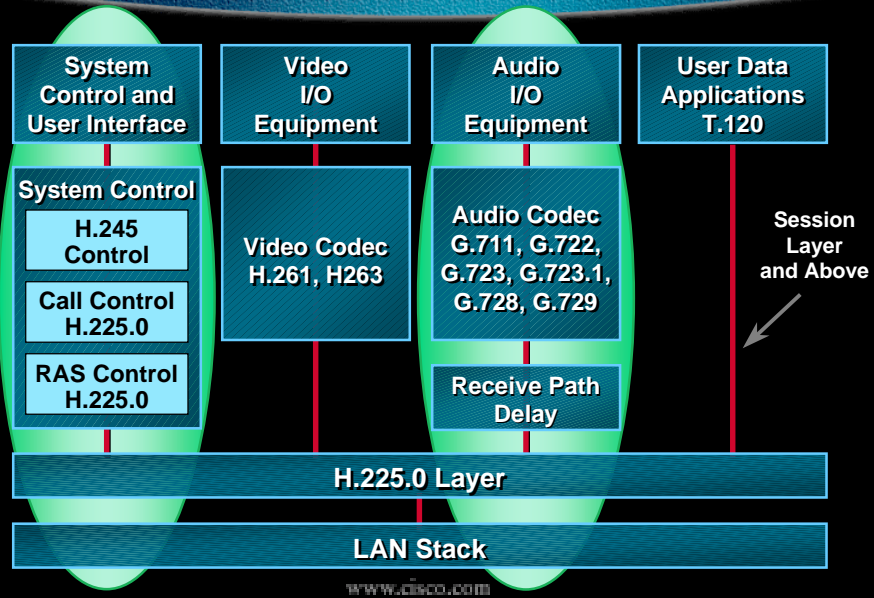
## Agenda

- 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?

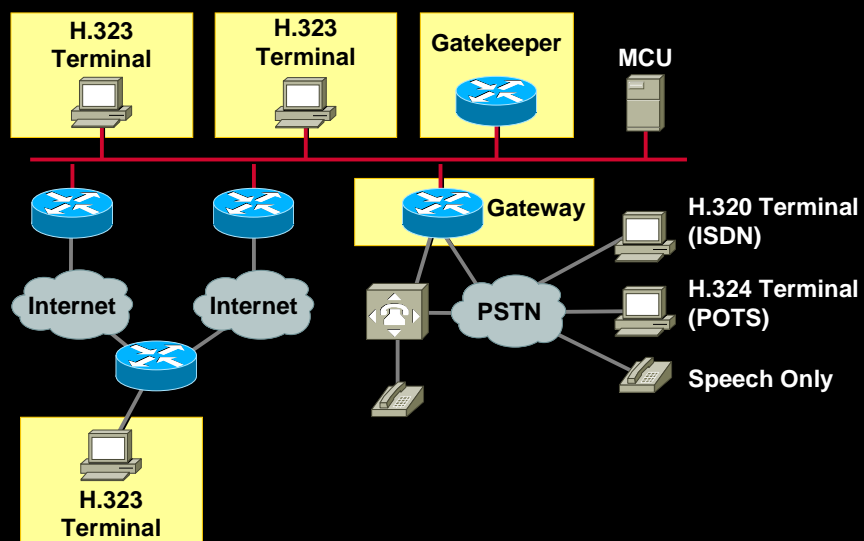
[www.cisco.com](http://www.cisco.com)



## VoIP Uses ITU H.323



## ITU H.323 Components



## H.323 VoIP Layers

IP Layered Model		H.323 VoIP Model
User		Caller
Application		Email ID E.164 Phone No.
Presentation		Audio Codec (G.711, G.729, G.723.1,..)
Session		H.225, H.245, RTP, RTCP
TCP	UDP	UDP Port Number
IP		IP Address
Data Link		Frame Relay DLCI, 802.3 MAC, ATM VPI/VCI
Physical		V.35, T1, T3

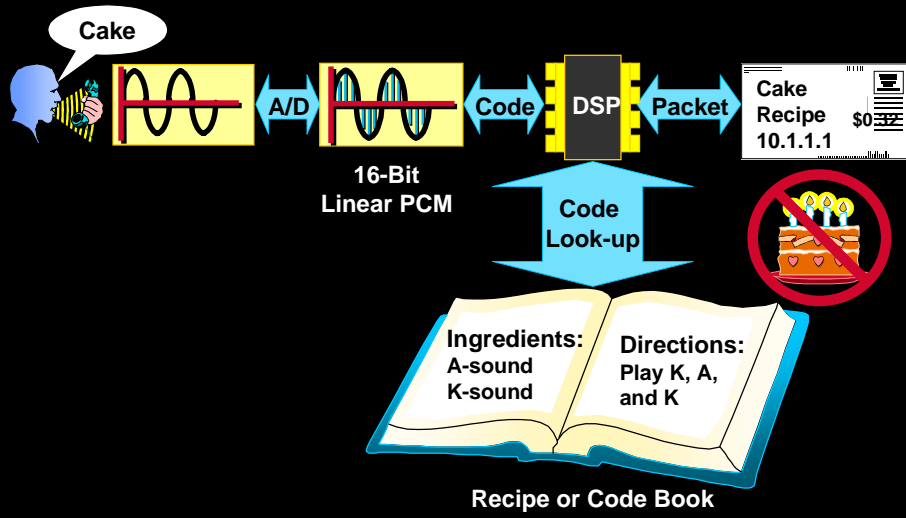
www.cisco.com

## 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

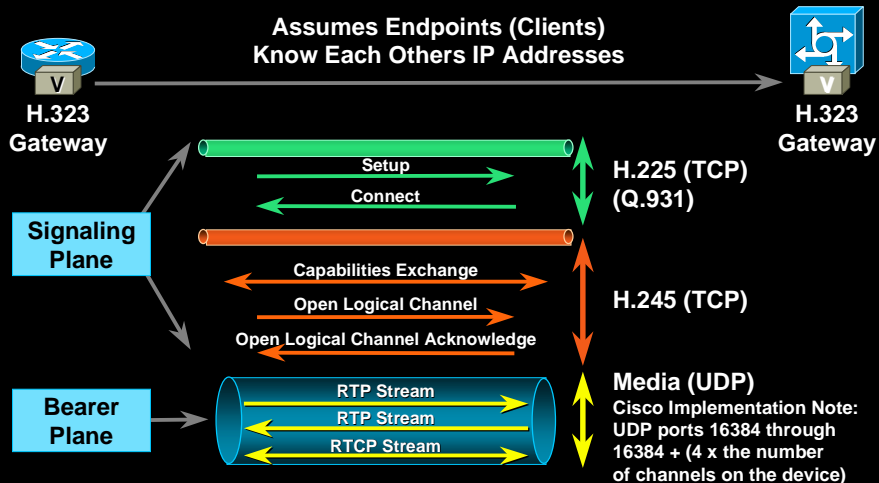
www.cisco.com

# Code Excited Linear Prediction (CELP)



www.cisco.com

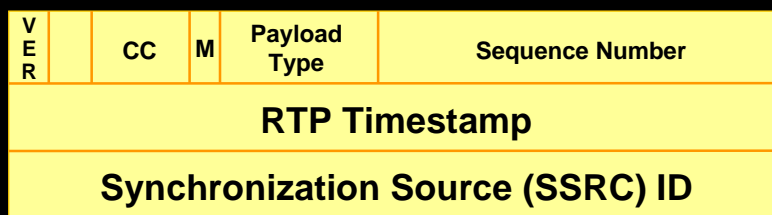
# H.323 End Point to End Point Signaling



www.cisco.com

## 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)

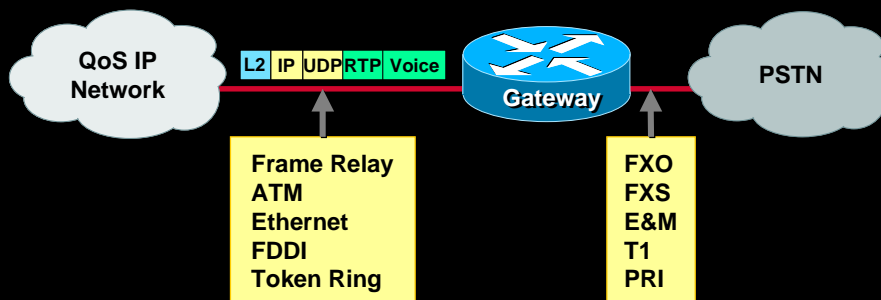


www.cisco.com

## H.323 Gateway

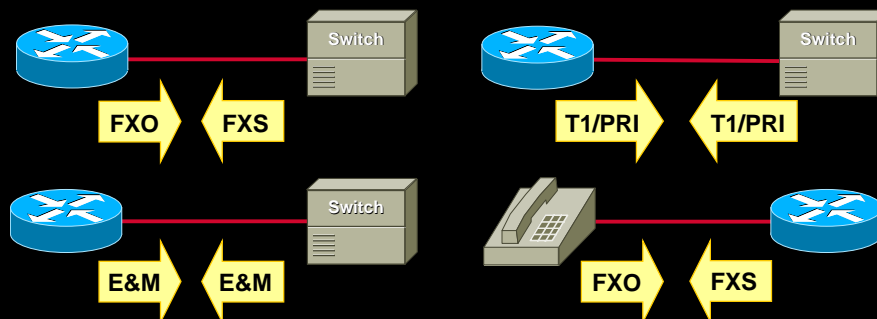
G.711 PCM  
G.726 ADPCM  
G.728 LD-CELP  
G.729 CS-ACELP  
G.729A CS-ACELP  
G.723.1 ACELP

G.711 PCM  
Analog



www.cisco.com

## Router Voice Interfaces



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

www.cisco.com

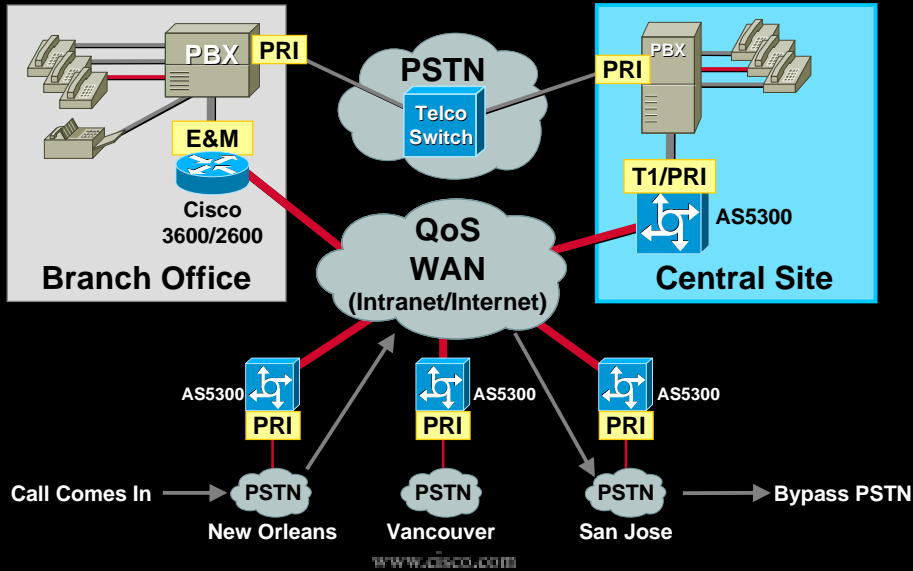
## Dial-Peer Configuration (Static Routing)



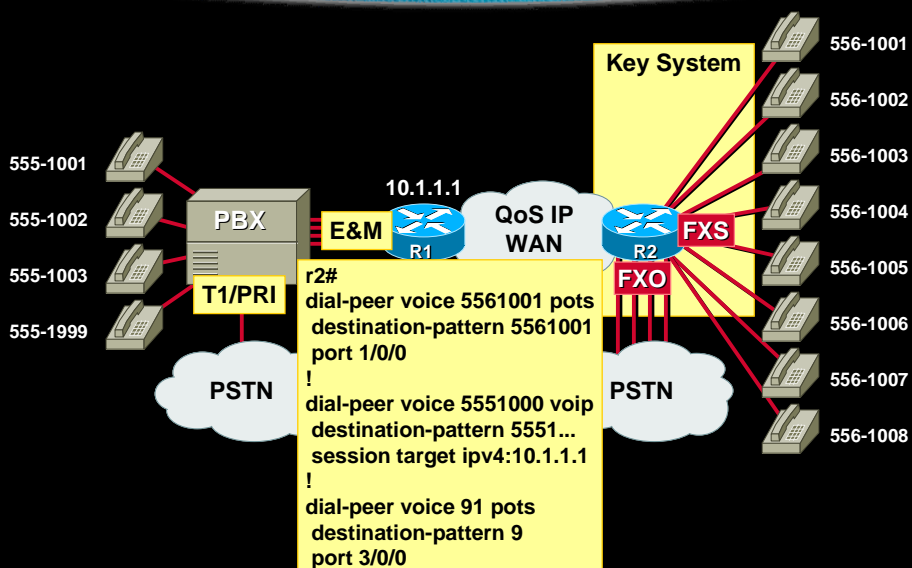
r1#		r2#
dial-peer voice 1234 pots	Local	dial-peer voice 4321 pots
destination-pattern 1234		destination-pattern 4321
port 1/0/0		port 1/0/0
!		!
dial-peer voice 4000 voip	VoIP	dial-peer voice 1000 voip
destination-pattern 4...		destination-pattern 1...
session target ipv4:10.1.1.2		session target ipv4:10.1.1.1
!		!
dial-peer voice 4000 vofr	VoFR	dial-peer voice 1000 vofr
destination-pattern 4...		destination-pattern 1...
session target serial0 122		session target serial0 221
!		!
dial-peer voice 4000 voatm	VoATM	dial-peer voice 1000 voatm
destination-pattern 4...		destination-pattern 1...
session target serial0 1		session target serial0 1

www.cisco.com

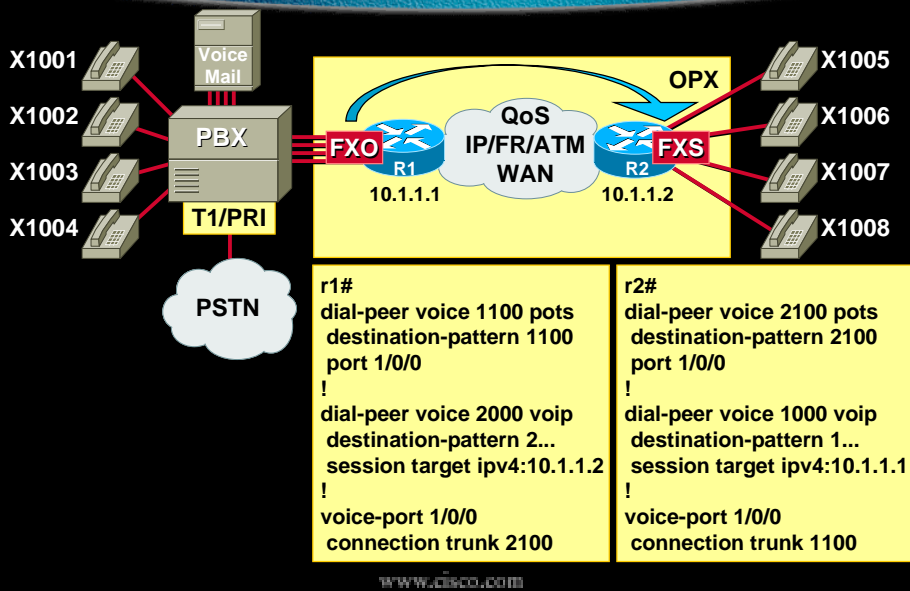
## Basic Trunk/Route Replacement



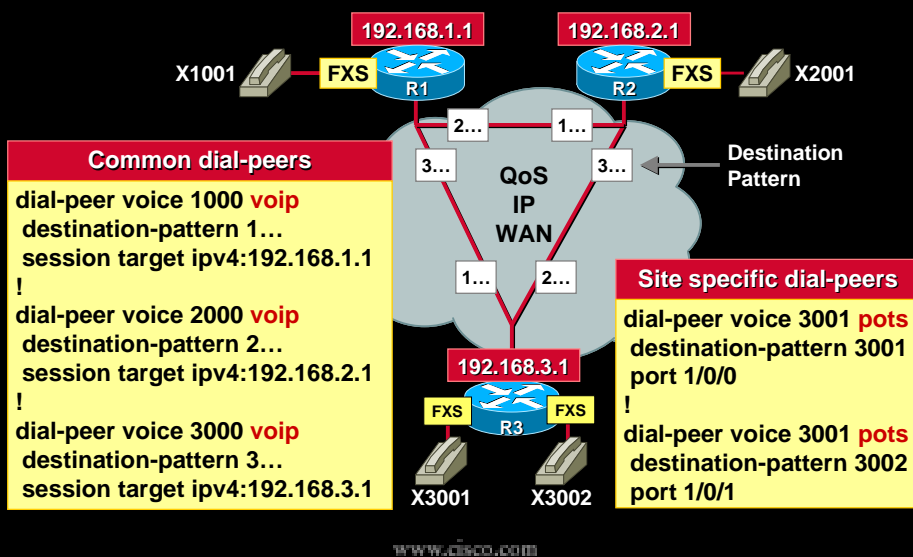
## Router-Based Key System



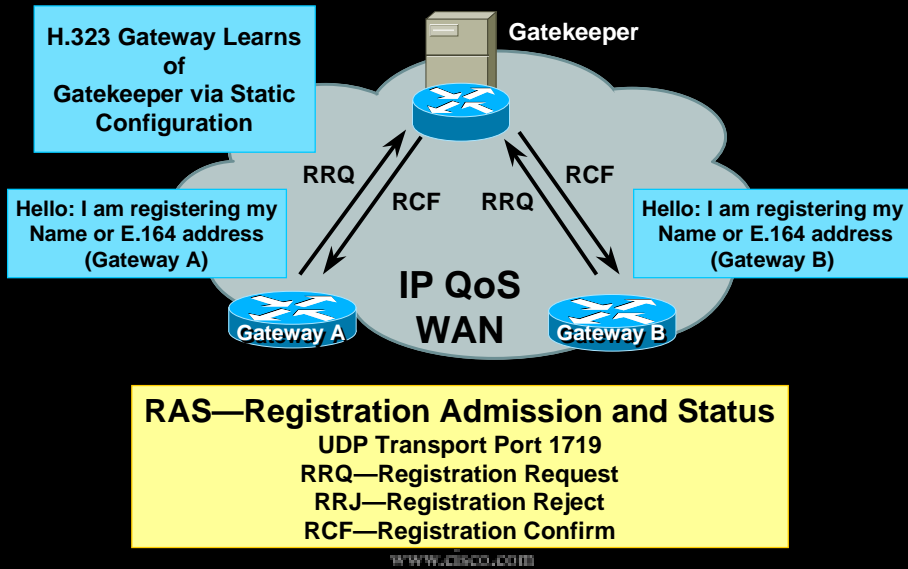
## Off Premise Extension (OPX)



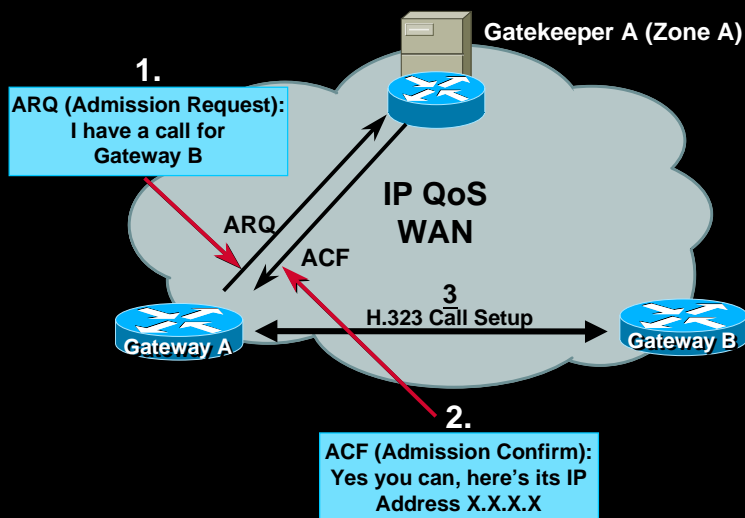
## VoIP Full Mesh Dial-Peers



## H.323 Gatekeeper Call Control/Signaling Gatekeeper Registration



## H.323 Gatekeeper Dial-peer Scalability



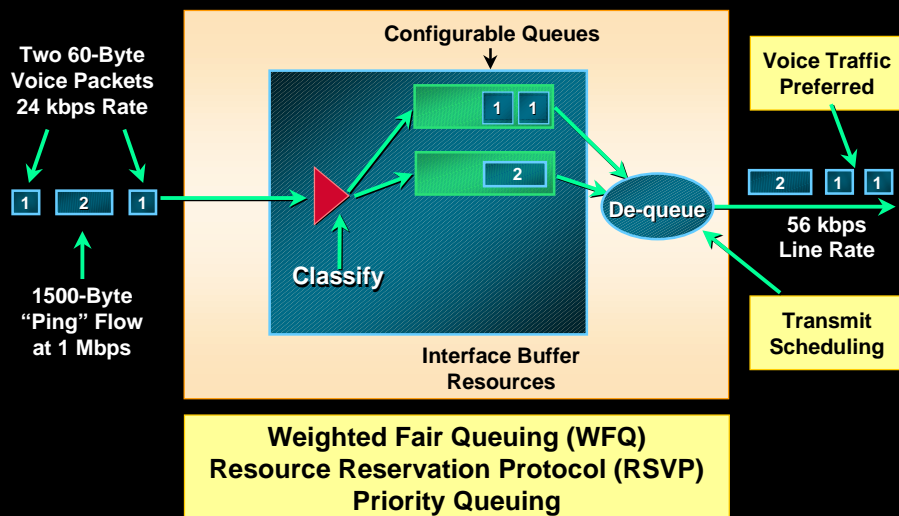


## VoIP Low Speed Link (<768 Kbps) Challenges and Solutions

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

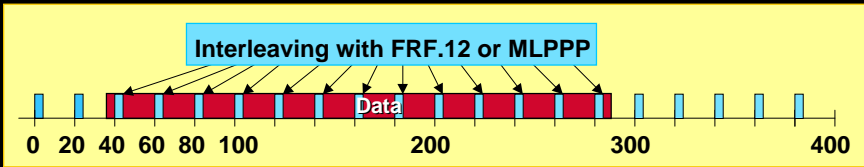
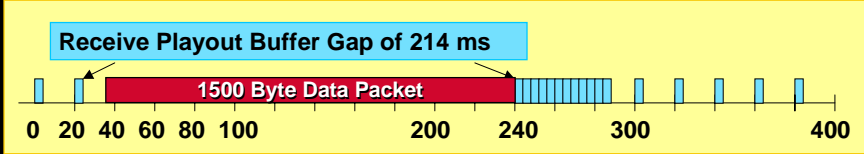
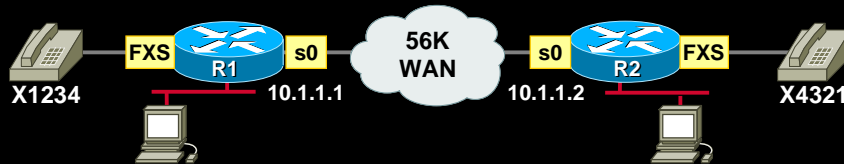
www.cisco.com

## Congestion Avoidance Solutions Intelligent Queuing



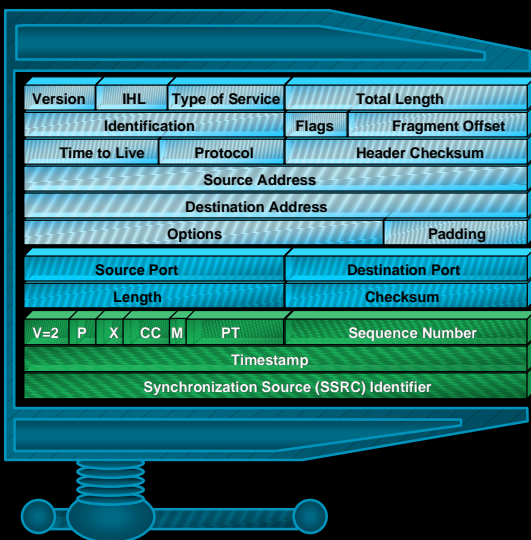
www.cisco.com

# Packet Residency Solutions



www.cisco.com

# 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

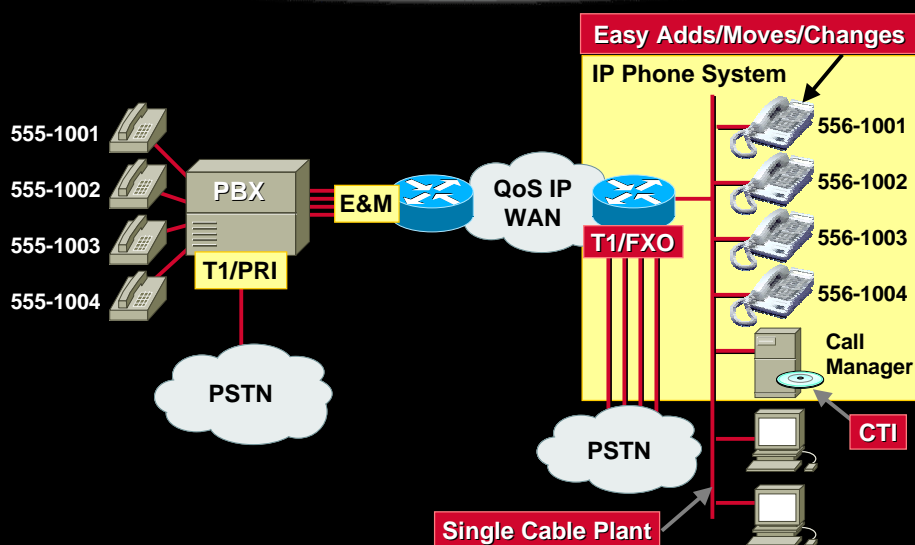
www.cisco.com

## 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?

www.cisco.com

## IP Phone System



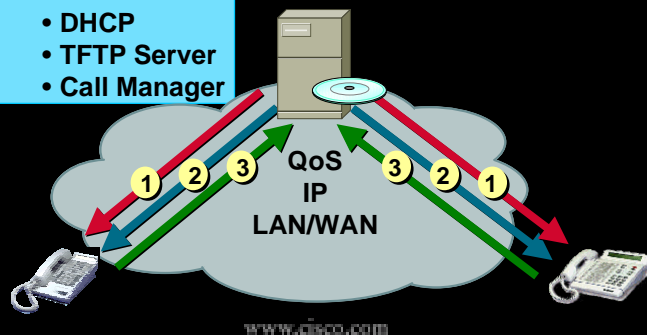
www.cisco.com

## 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

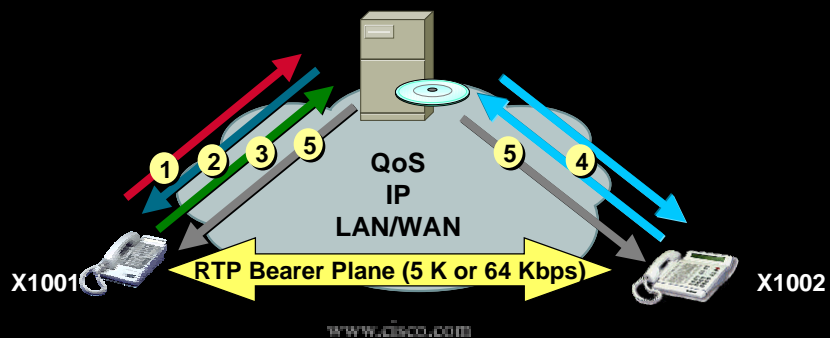
### Services:

- 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



## 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?**

[www.cisco.com](http://www.cisco.com)

## 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

[www.cisco.com](http://www.cisco.com)



**Please Complete Your  
Evaluation Form**

**Session 402**

[www.cisco.com](http://www.cisco.com)

**CISCO SYSTEMS**



**EMPOWERING THE  
INTERNET GENERATION<sup>SM</sup>**

[www.cisco.com](http://www.cisco.com)