Introduction to Packet Voice Technologies

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Why IP Telephony?

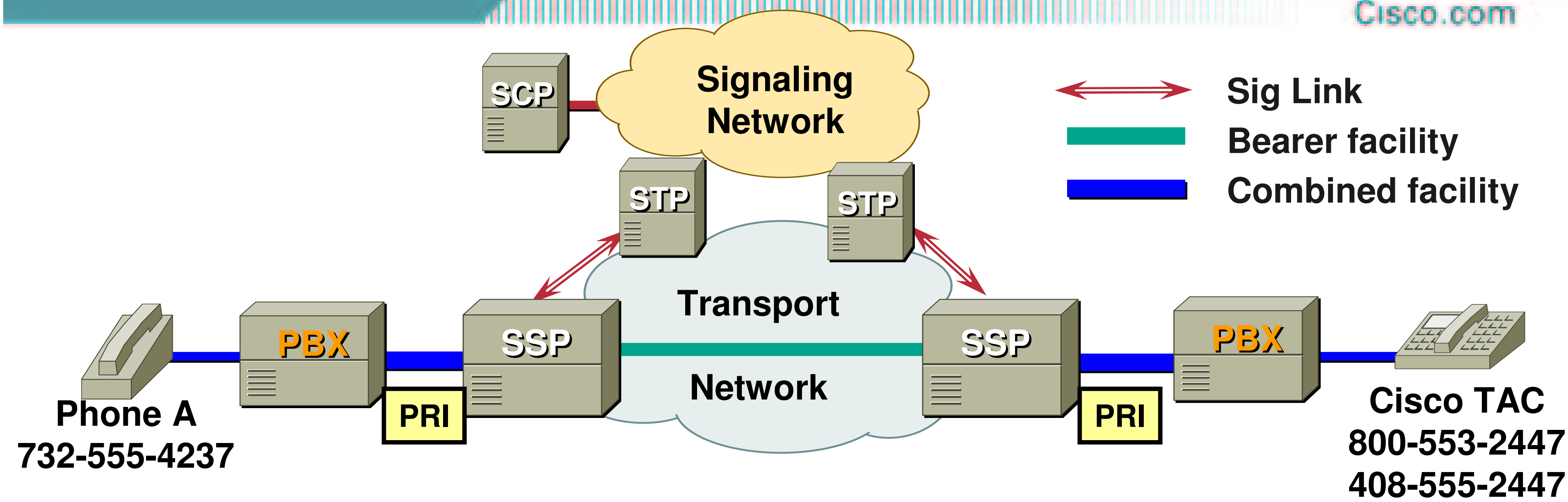
- Messaging
- Directory
- Calendaring
- Find me/Follow me
- 3rd Party Call Control
- Hoteling
- Collaboration
- Mobility
- Internet Contact Center
- Internet telephony

IP Enable Voice for New World Applications

Agenda

- Traditional Voice Review
- Market Maturity
- Packet Voice Basics
- Bearer Technologies
- Signaling Technologies
- Summary

Public ISDN and Signaling System 7



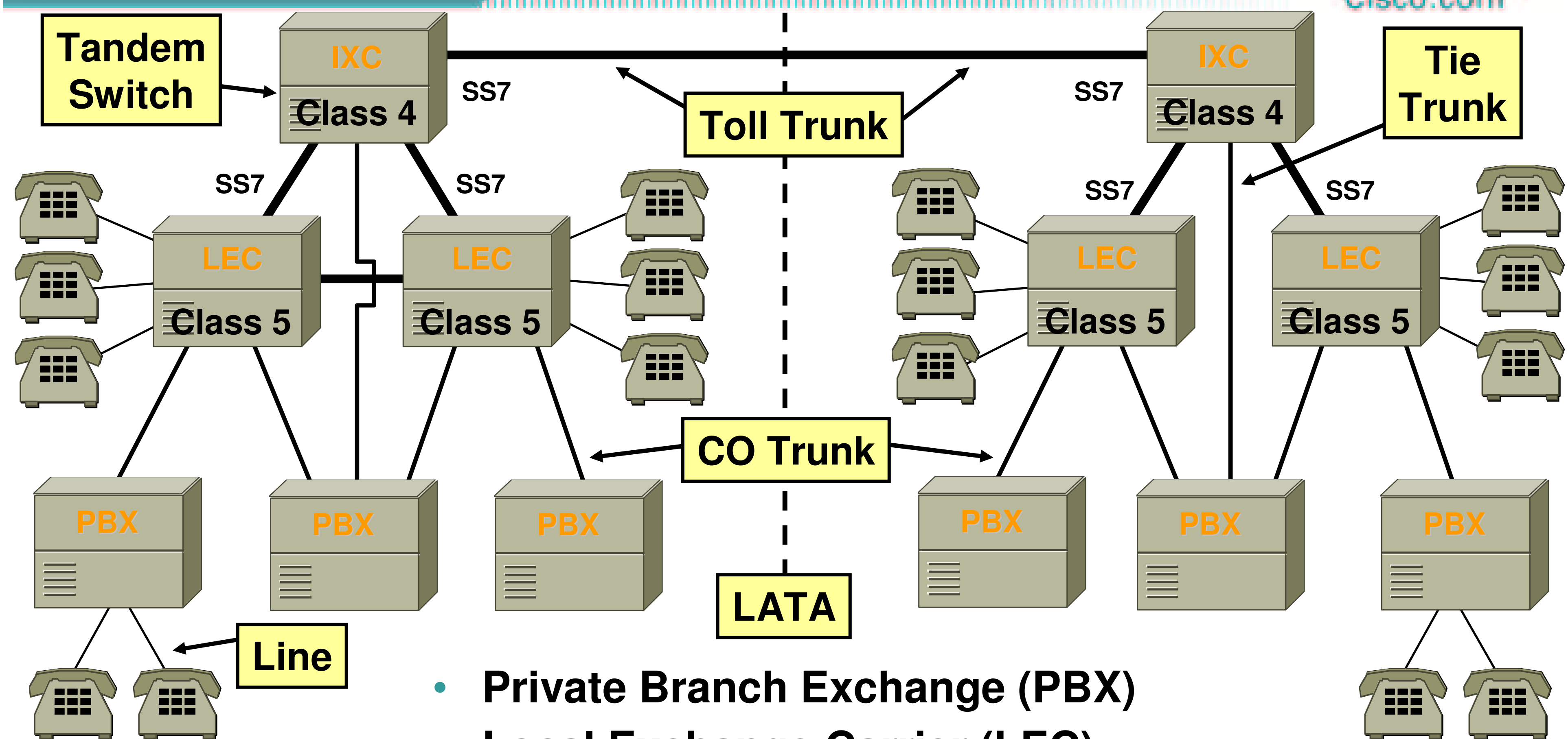
- **SS7 for Advanced Intelligent Network (AIN)**

  Service Control Point (SCP)

  Signal Transfer Point (STP)

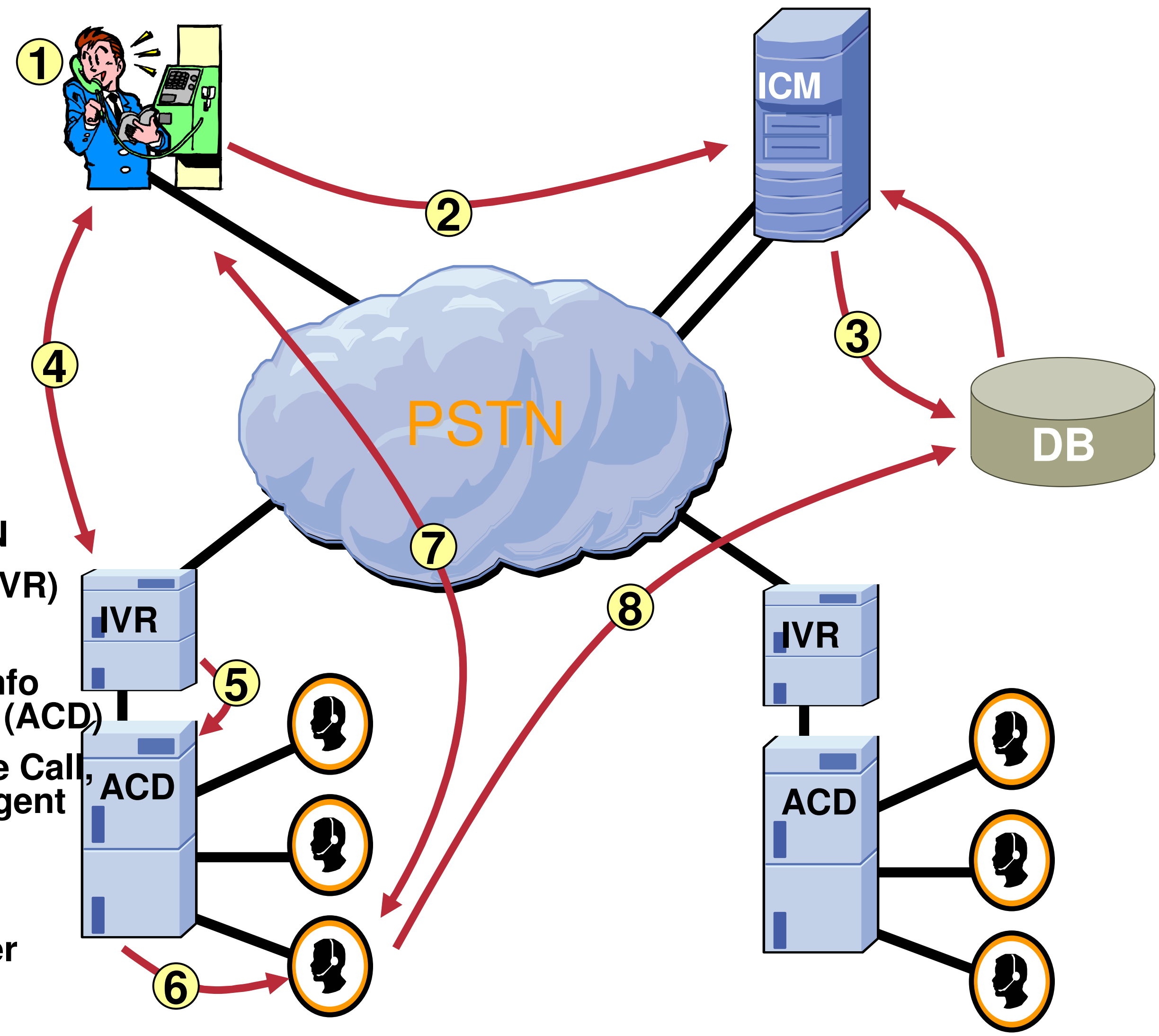
  Service Switching Point (SSP)

Telecommunications Components



- Private Branch Exchange (PBX)
- Local Exchange Carrier (LEC)
- InterExchange Carrier (IXC)
- Central Office (CO)

Call Center Components



1. Customer Dials 800-553-2447 for TAC
2. 800 Number is Directed to Intelligent Contact Manager (ICM)
3. ICM Queries DB for Best Local Number & provides to PSTN
4. Site Interactive Voice Response (IVR) Offloads Skilled Agents
5. IVR Passes the Call & Collected Info to the Automatic Call Distribution (ACD)
6. ACD Monitors Agents, Queues the Call, & Routes Call to Best Available Agent
7. Customer Speaks Directly with Selected Agent
8. Agent Updates DB when Customer Purchases

Types of Signaling

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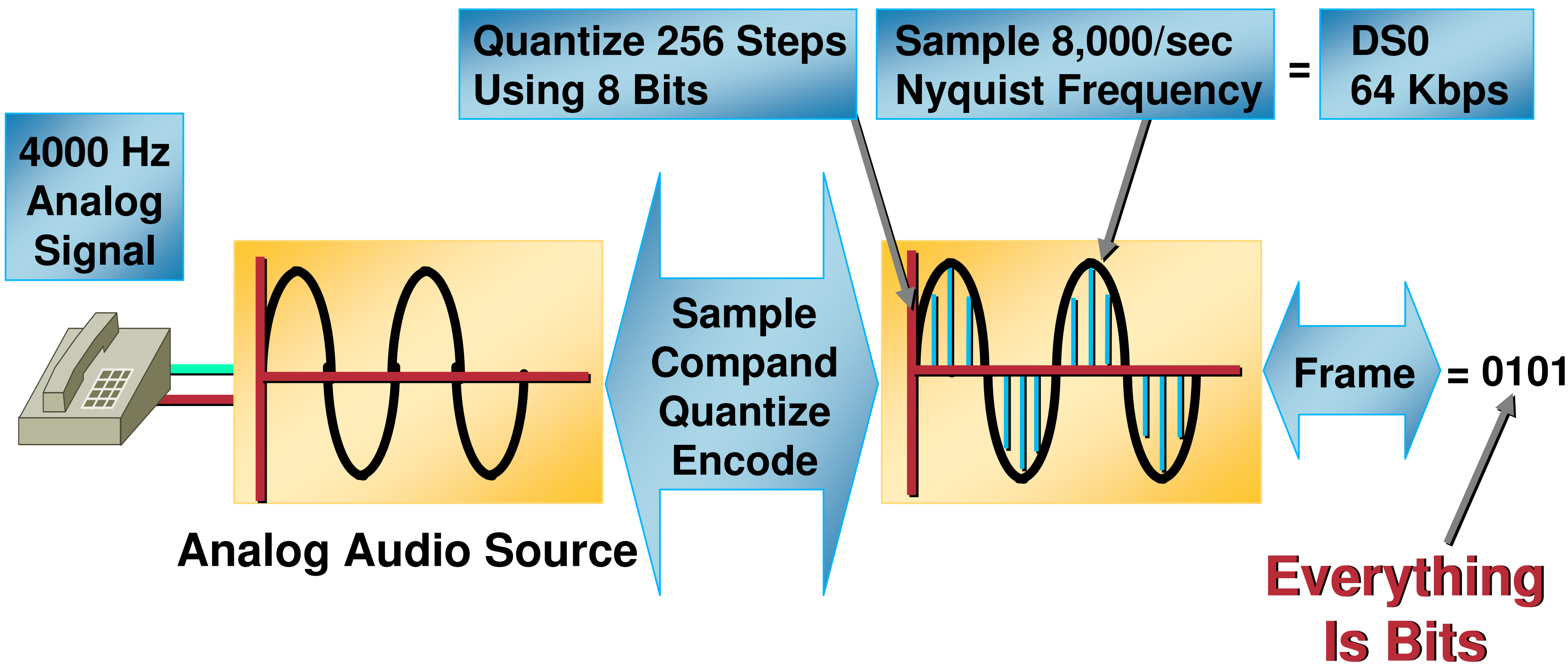
Method of communicating telephony events: off-hook, busy, on-hook, etc.

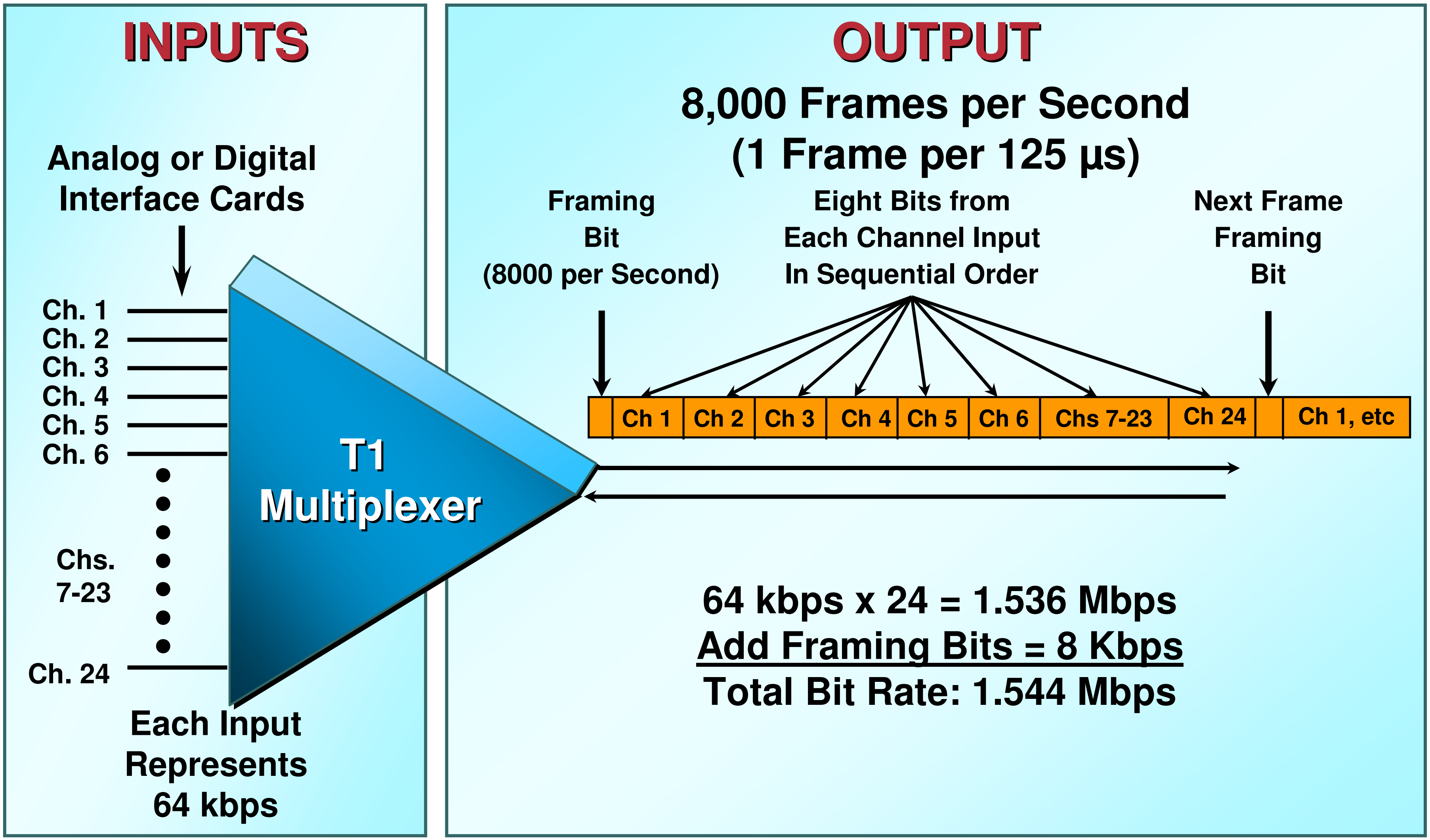
Analog	Digital
•2-wire •Loop start •Ground start	• Digital subscriber lines: 2-wire, 4-wire • Digital trunks: 4-wire
•E&M •2-wire, 4-wire •Five types I-V	• CAS—Channel associated signaling • In-band signaling
•CAMA	• CCS—Common channel signaling • Out-of-band signaling

Pulse Code Modulation



G.711 Pulse Code Modulation (PCM) is the DS0

Time Division Multiplexer
T1 Channel Bank



Digital Channel Associated Signaling

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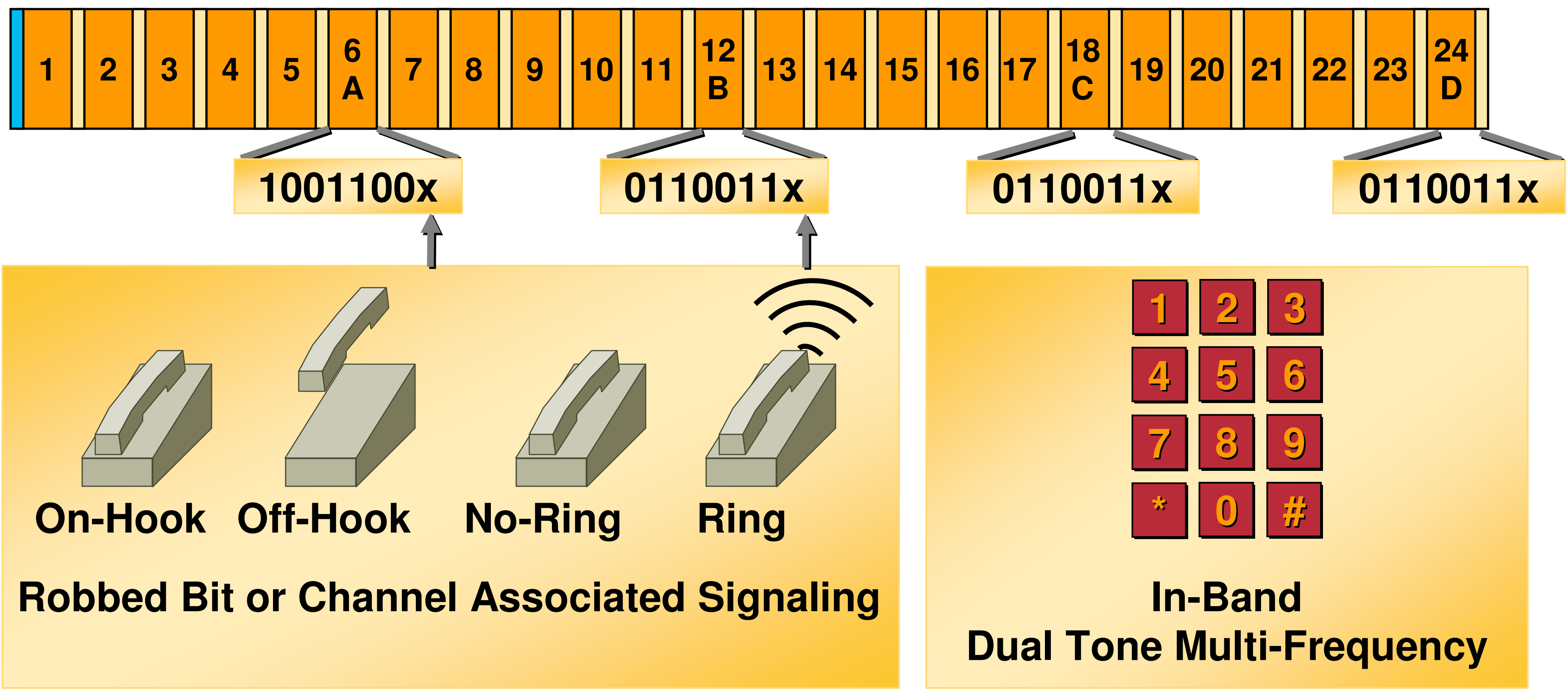
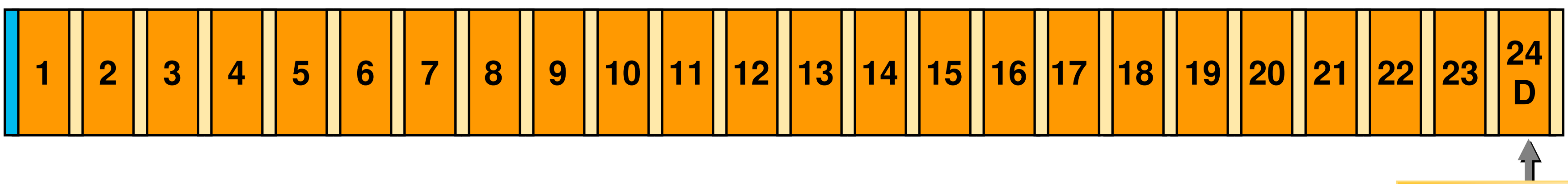
DS1—Extended Super Frame (24 T1 Frames)
T1—Coding (Ones Density)—AMI, ZCS, B8ZS



Digital ISDN
Common Channel Signaling



Q.931
Messaging

- **Integrated Services Digital Network**
- **Primary Rate Interface (PRI), 1.544Mbps**

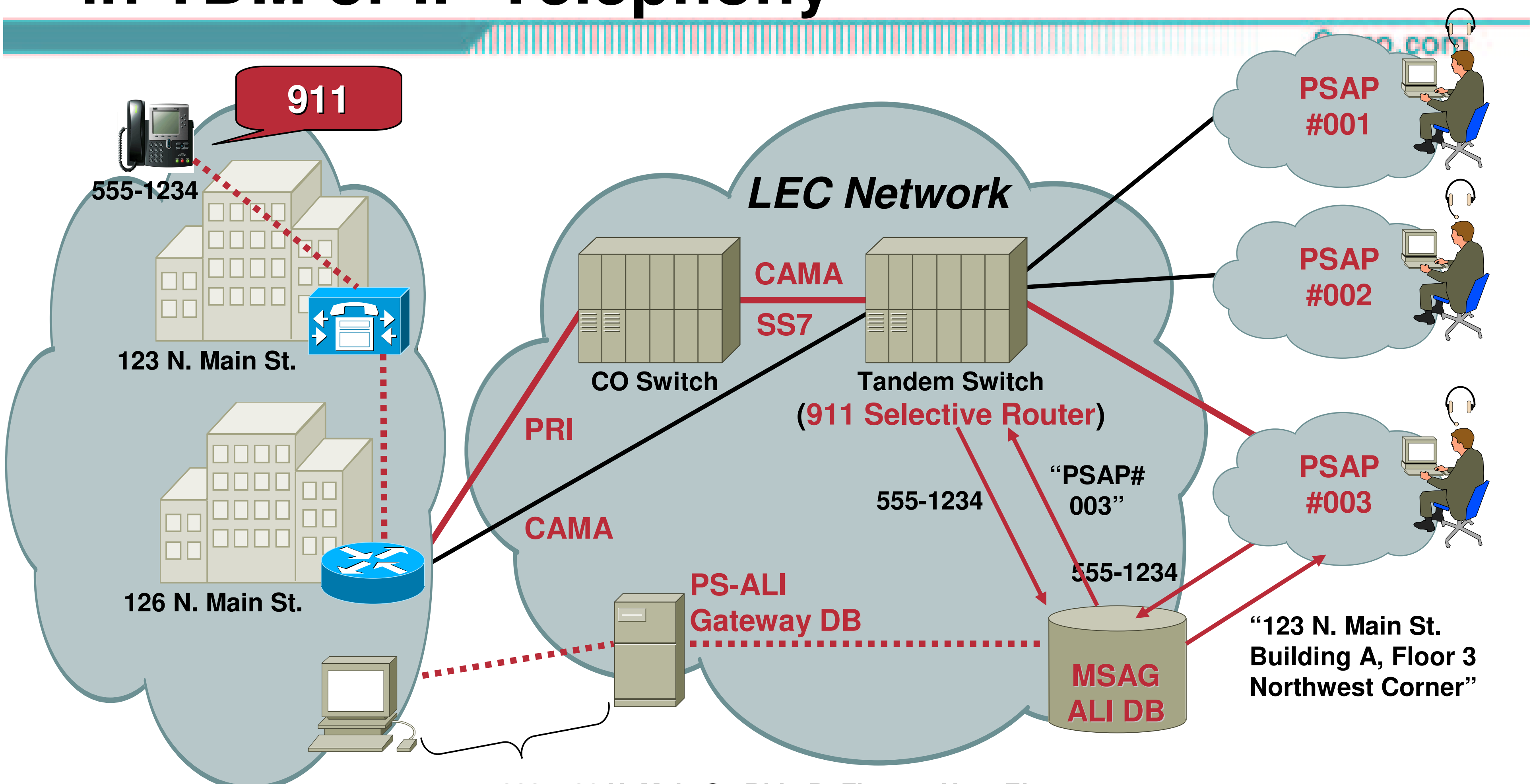
  23B+D channels

- **30B+D ISDN, 2.048Mbps**

  TS #16 for signaling (CCS)

- **QSIG**

Overview: E9-1-1 Call-Flow in TDM or IP Telephony



555-1232: 126 N. Main St, Bldg D, Floor 4, Near Elevator
555-1233: 123 N. Main St, Bldg A, Floor 1, Rear Wall
555-1234: 123 N. Main St, Bldg A, Floor 3, Northwest Corner
555-1235: 123 N. Main St, Bldg B, Floor 2, Southwest Corner

Voice vs. Data Switching

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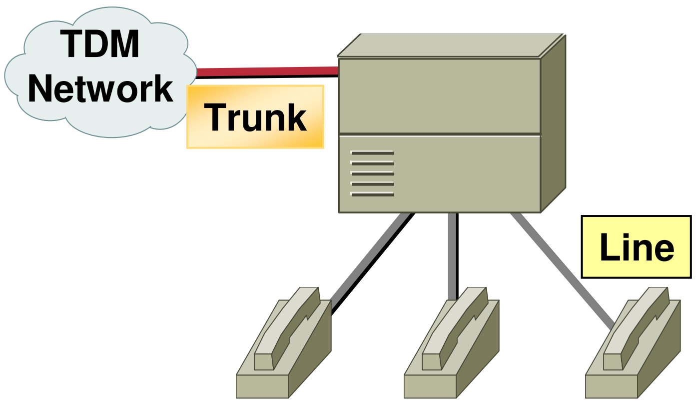
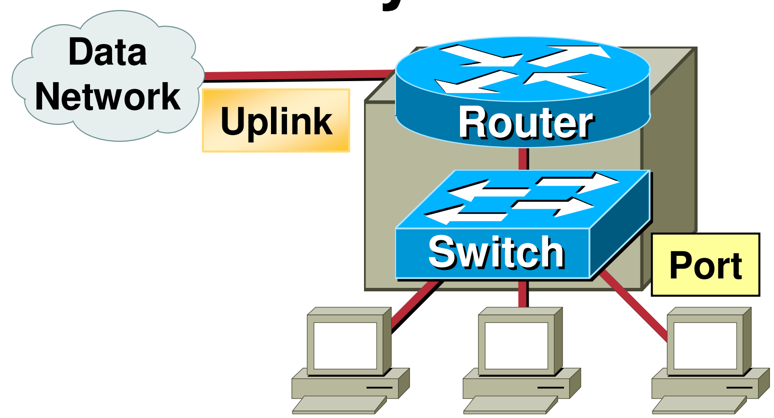
Class 5 Voice Switch



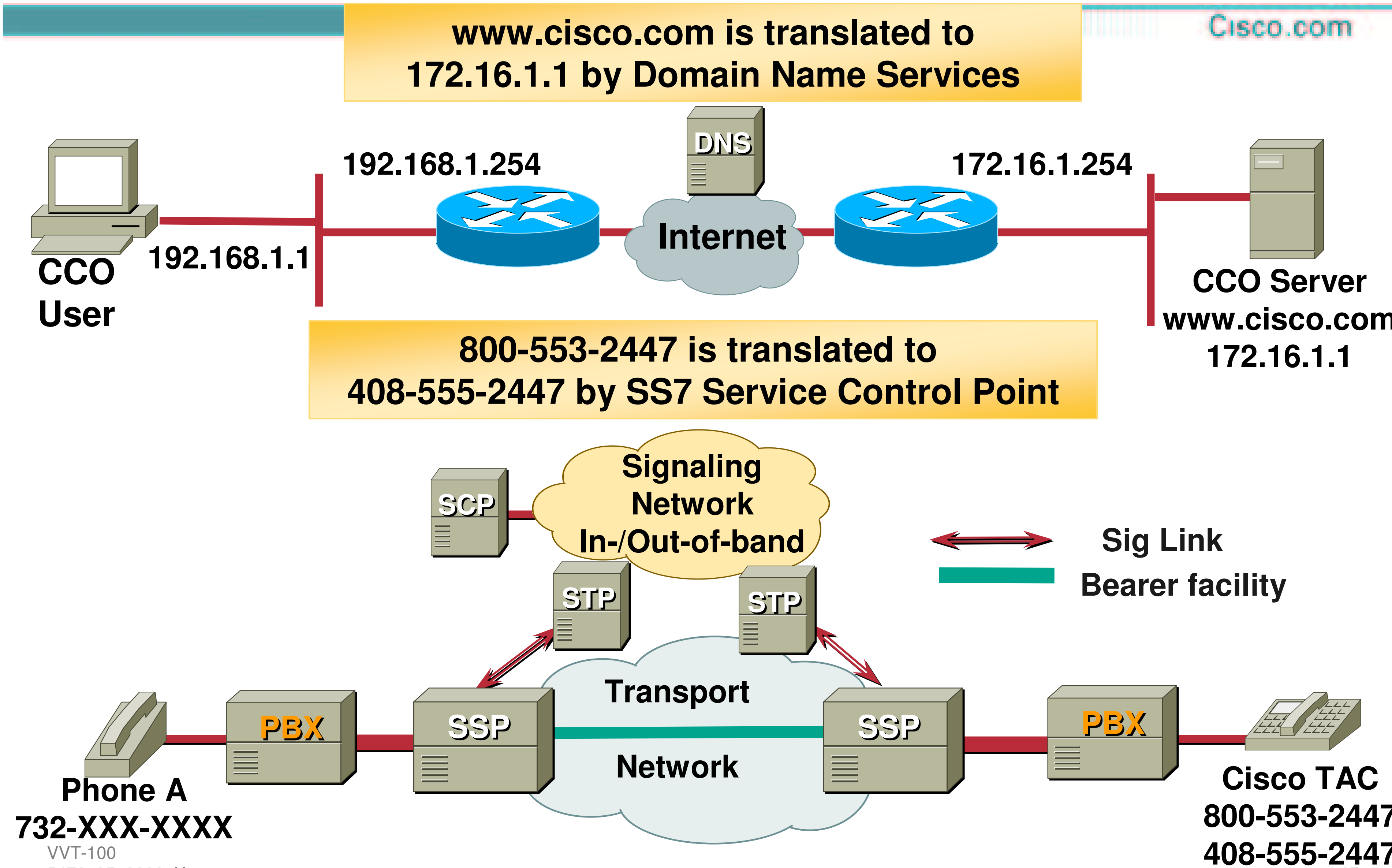
- Handset aggregator
- All telephones get a single analog/digital line (DS0)
- All devices have a phone number defined on the switch
- All devices have access but trunks are oversubscribed (Lines > Trunks)
- Path selection based on static least cost routing or ARS

Multilayer Data Switch



- Computer aggregator
- All devices get dedicated bandwidth (BW) 10/100/1000 Mbps
- All devices have an IP address defined on the host
- All devices have access but uplinks are oversubscribed (Station BW > Uplink BW)
- Path selection based on dynamic least cost route

Address Resolution



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- **Market Evolution**
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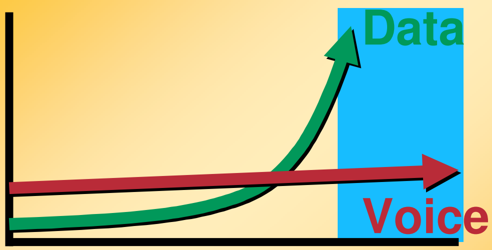
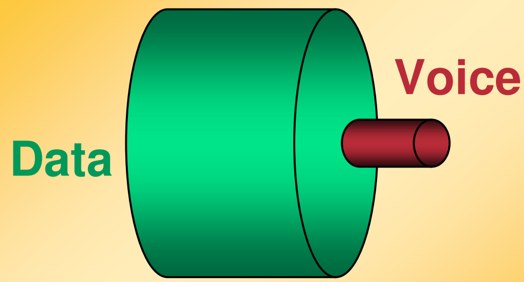
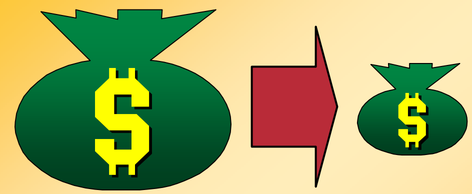
Market Drivers



Data Exceeded Voice
Over Voice Net

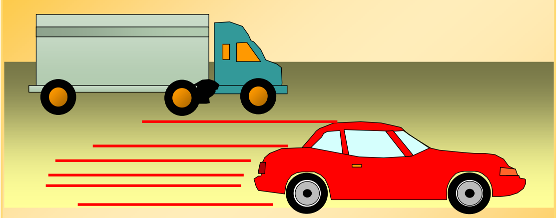


Bigger,
Cheaper Bandwidth



Network Consolidation
Usage Costs
Trunk Charges

QoS Advances



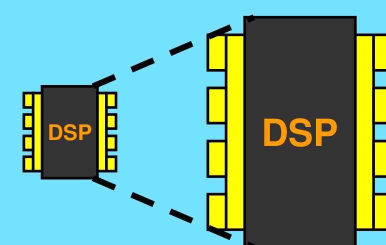
IP Precedence, IntServ,
DiffServ,
CBWFQ, 802.1p

Open Telephony
Standards

H.323/SIP/SCCP/MGCP

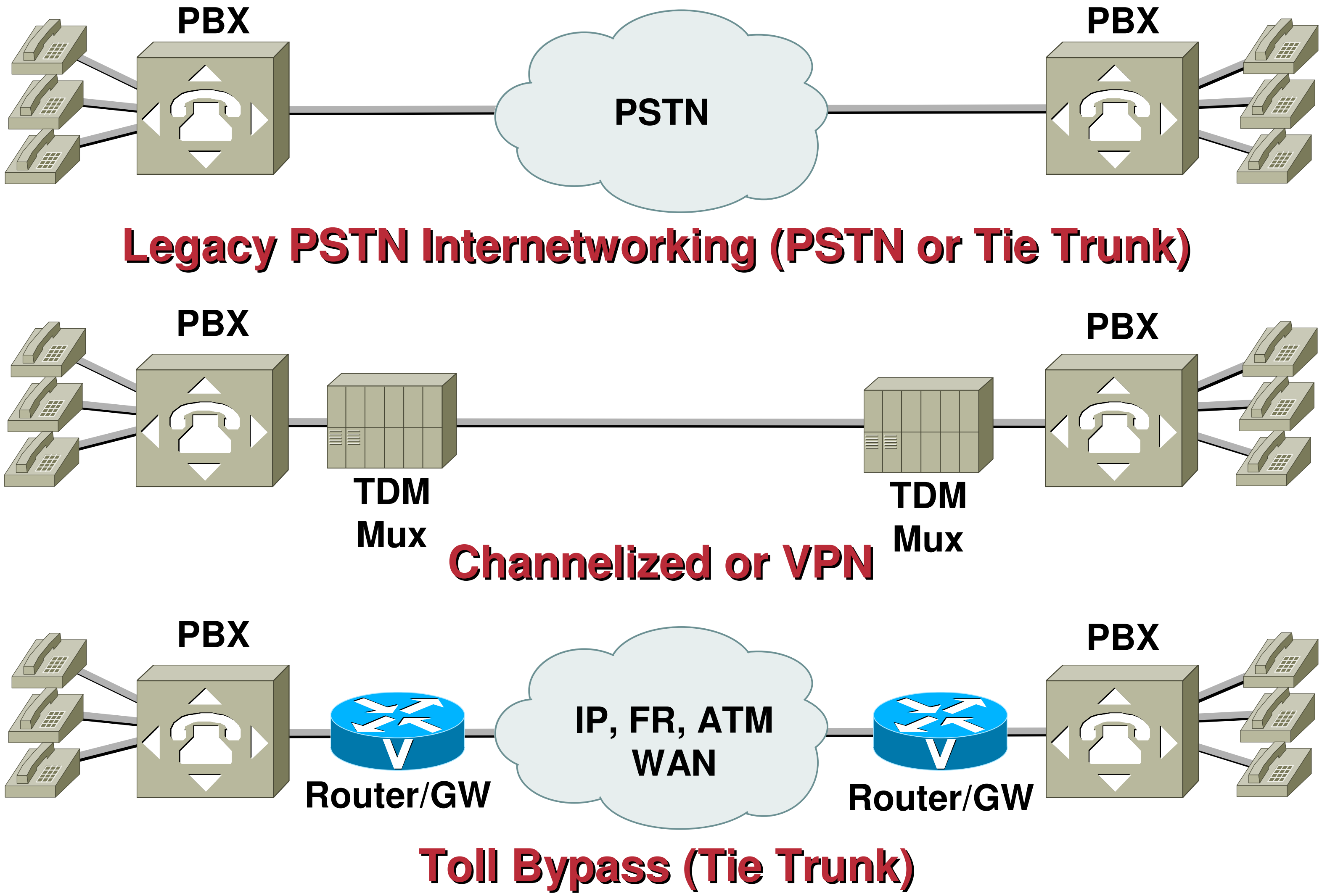
FRF.11/12

AAL1/2/5

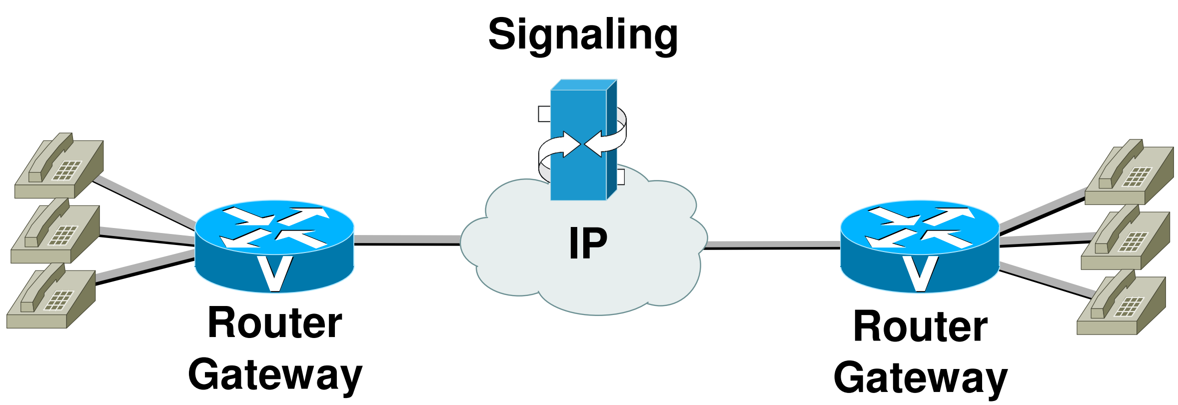


Digital Signal
Processor Advances

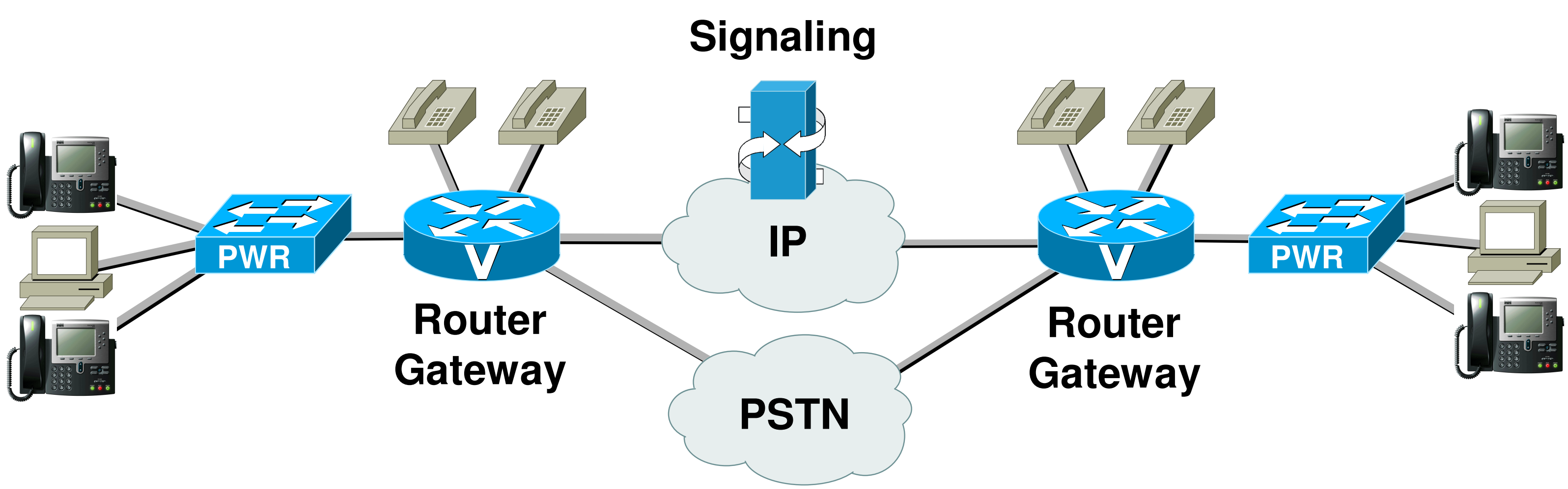
Voice Toll Bypass Evolution



IP Telephony



IP Switch (Softswitch)



IP Telephony

IP Telephony Application Integration

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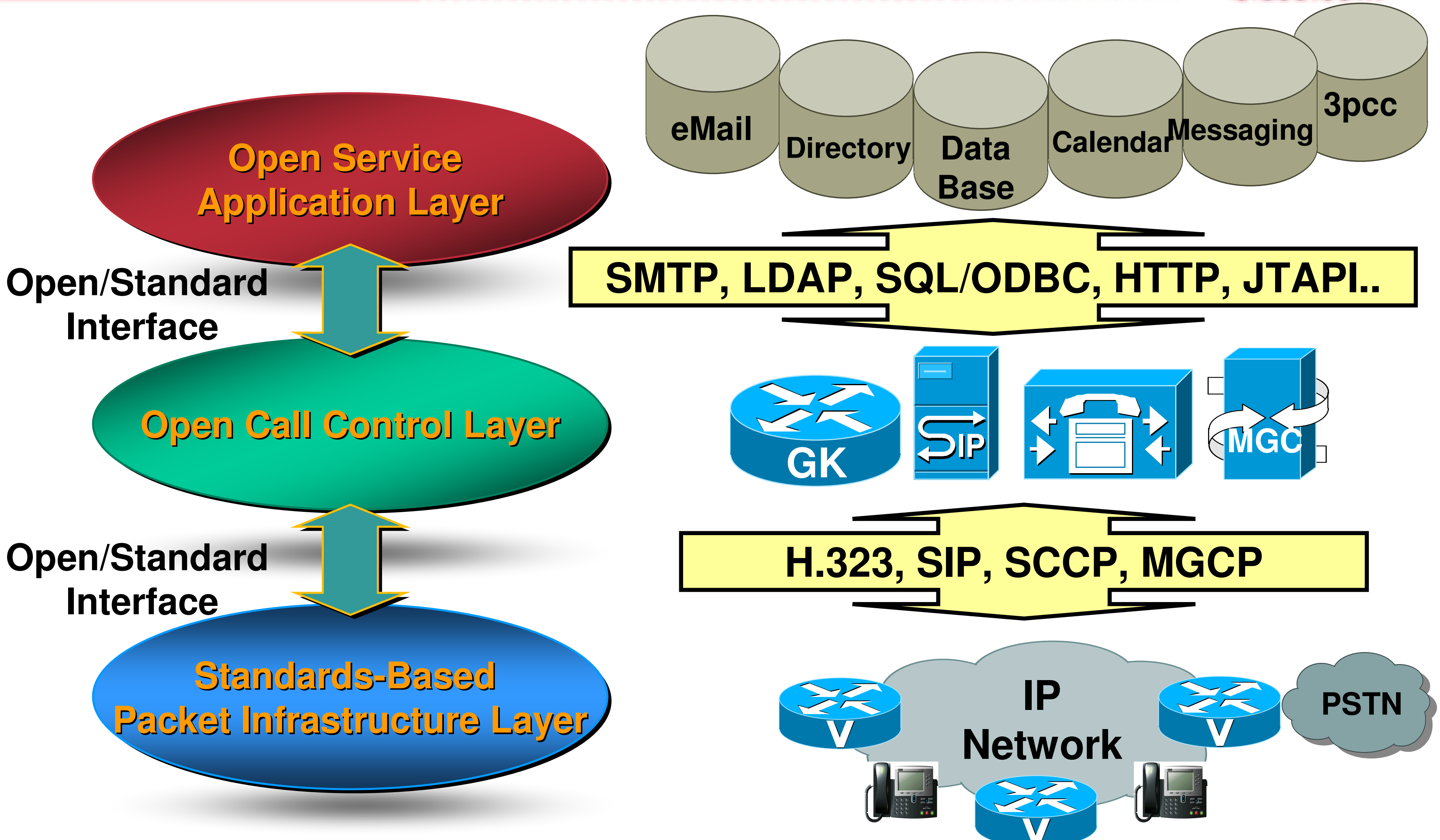

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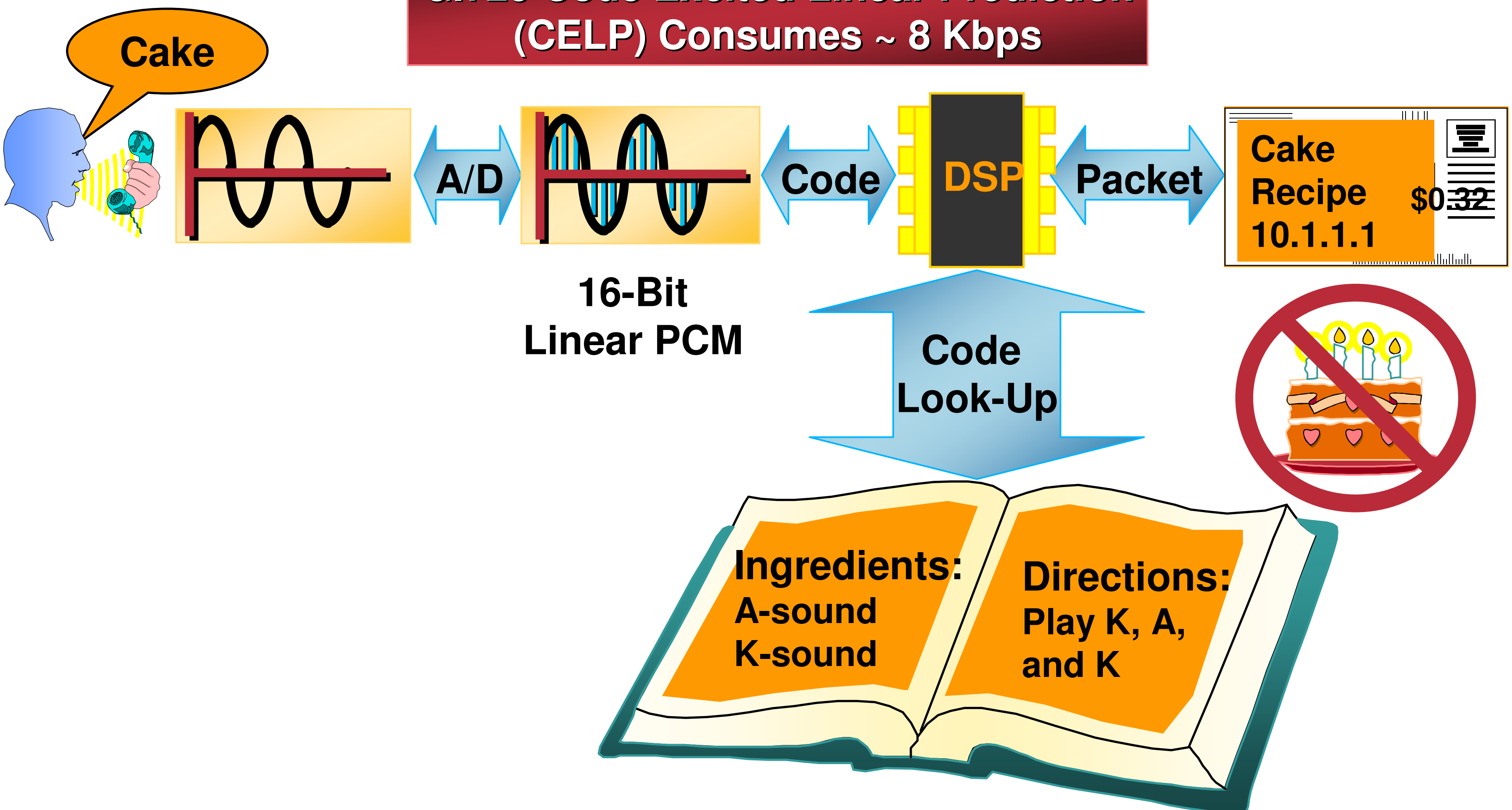
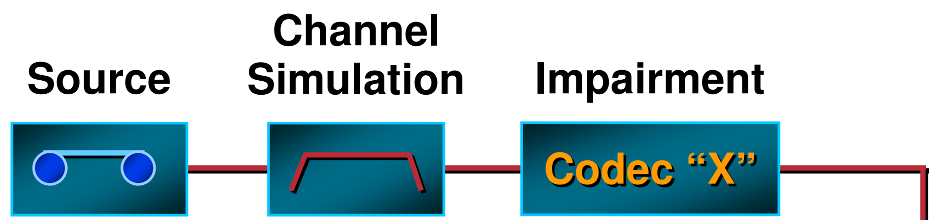
What Is an Advanced CODEC?

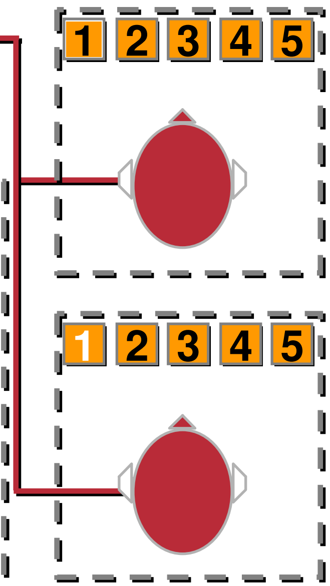
G.729 Code Excited Linear Prediction (CELP) Consumes ~ 8 Kbps



Recipe or Code Book

How Are CODECs Compared? (Mean Opinion Score)



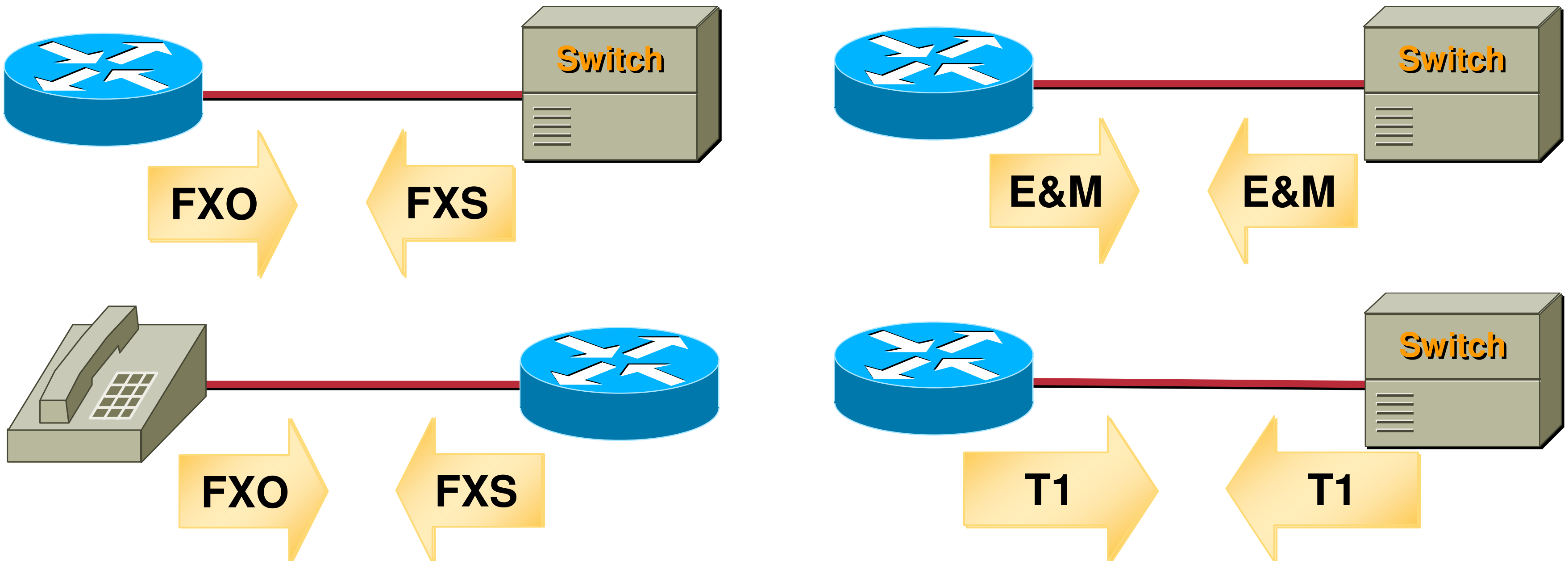


Rating	Speech Quality	Level of Distortion
5	Excellent	Imperceptible
4	Good	Just Perceptible—Not Annoying
3	Fair	Perceptible—Slightly Annoying
2	Poor	Annoying but Not Objectionable
1	Unsatisfactory	Very Annoying—Objectionable

Voice CODEC Cheat Sheet

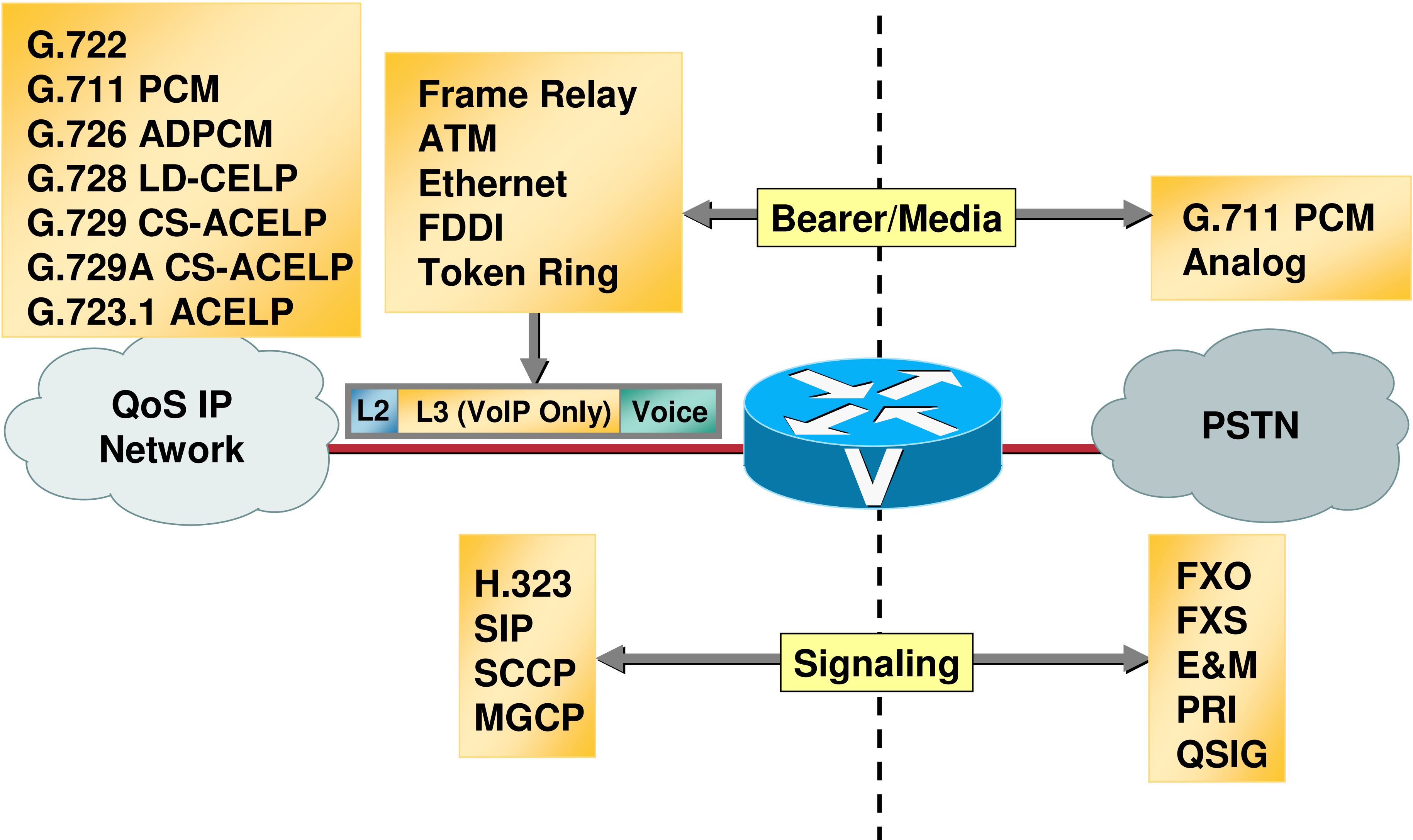
Encoding Compression	Mean Opinion Score	Native Bit Rate Kbps	Voice Quality	BW	DTMF	Dual Comp	CPU	Music on Hold
G.722 PCM	4.1	224	A	D	A	A	A	A
G.711 PCM	4.1	64	A	C	A	A	A	B
G.726 ADPCM	3.85	32	B	C	B	B	B	B
G.728 LD-CELP	3.61	16	C	B	B	C	C	C
G.729 CS-ACELP	3.92	8	A	A	B	B	C	C
G.729a CS-ACELP	3.7	8	B	A	C	C	B	D
G.723.1 ACELP	3.65	5.3	C	A	C	D	C	D

Router Voice Interfaces

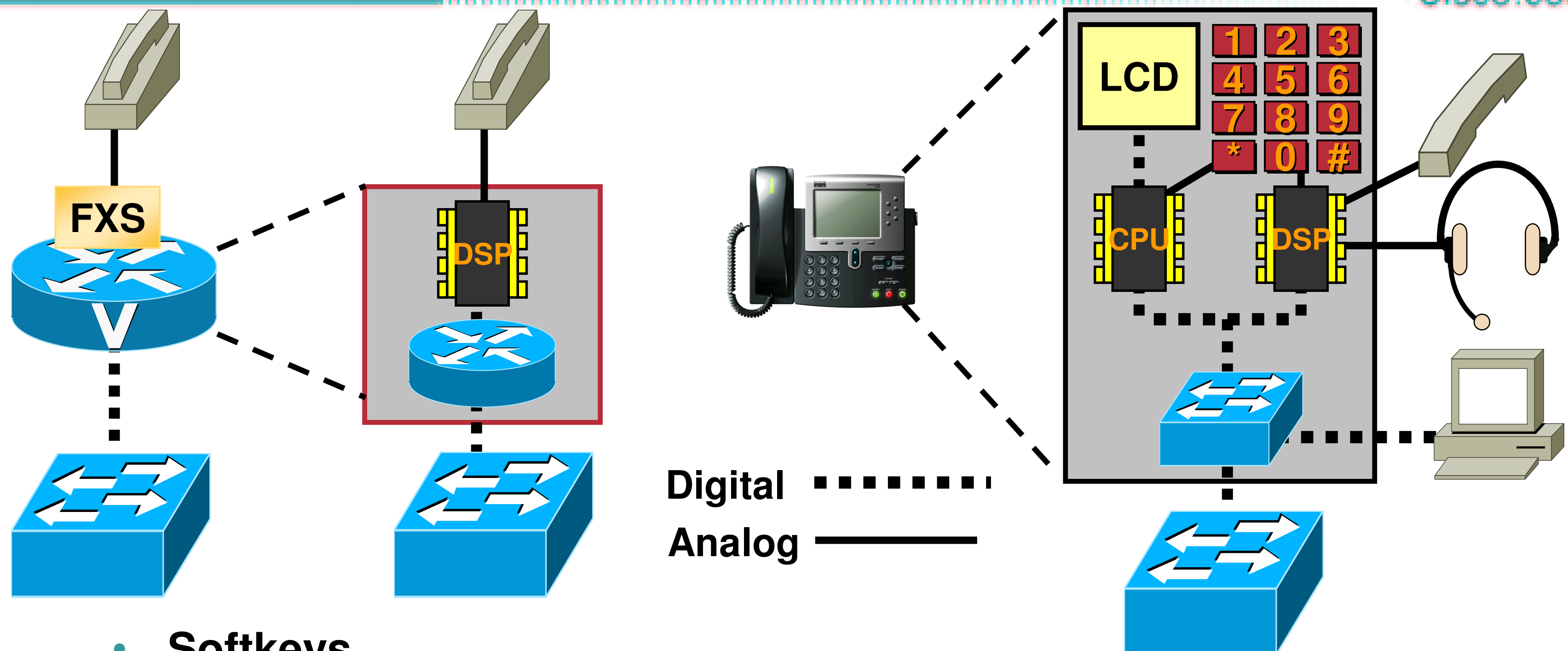


- FXO—Foreign Exchange Office
- FXS—Foreign Exchange Station
- E&M—Ear and Mouth
- T1—CAS and CCS (PRI and QSIG)

Voice-Enabled Gateways



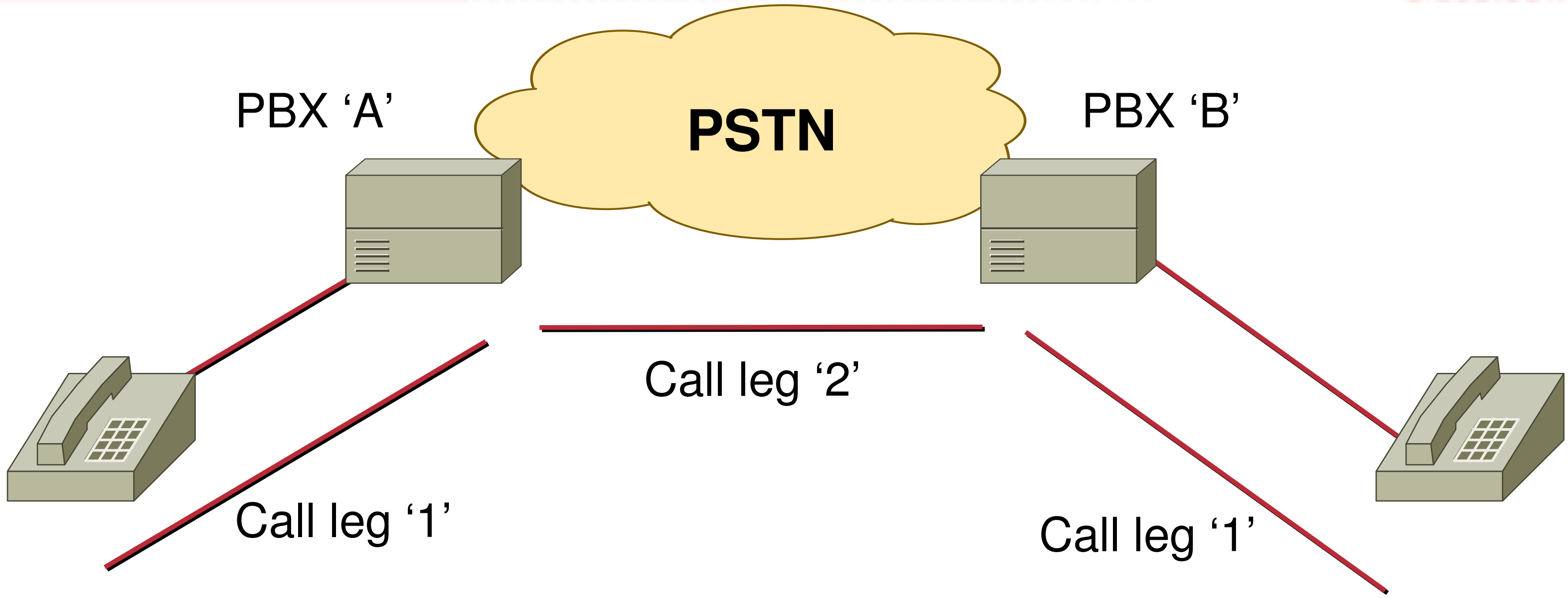
What's the Difference Between an Access Gateway and an IP Phone?



- Softkeys
- LCD
- HTTP client and server
- Extensible Markup Language (XML)
- Application Integration (LDAP, UM,..)

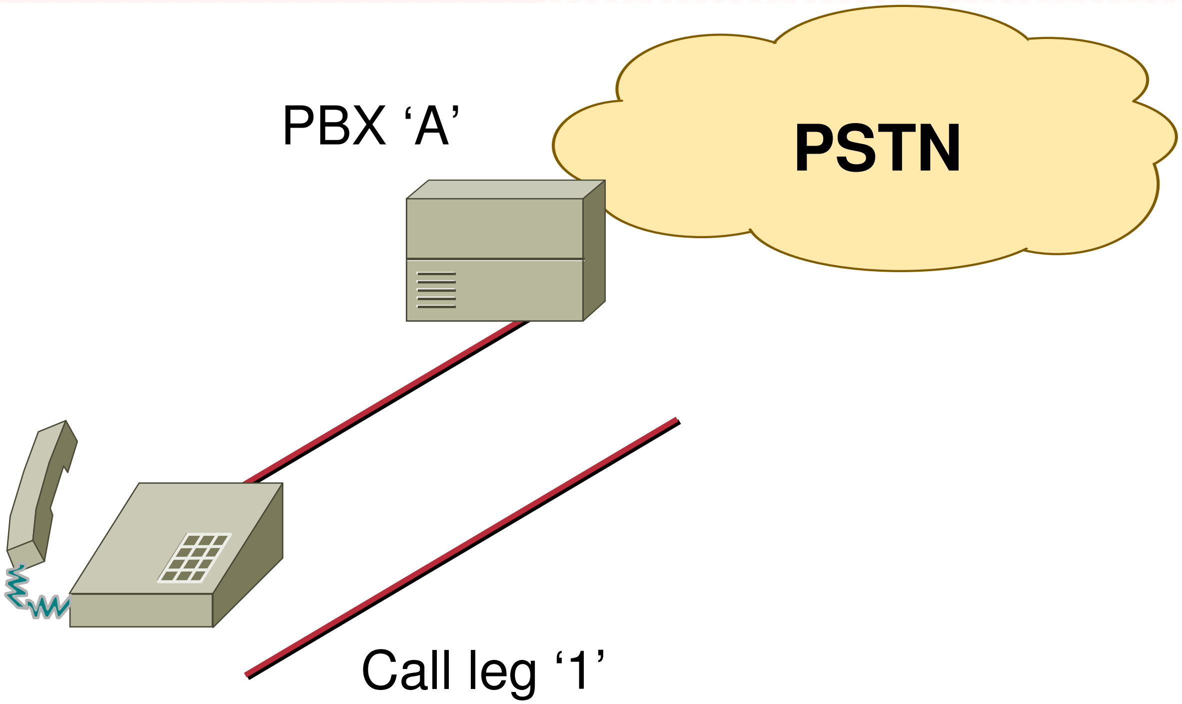
- Multiple line appearances
- Message Waiting Indicator
- Hands-free speaker
- Headphone jack
- Integrated Ethernet switch

Building a Call



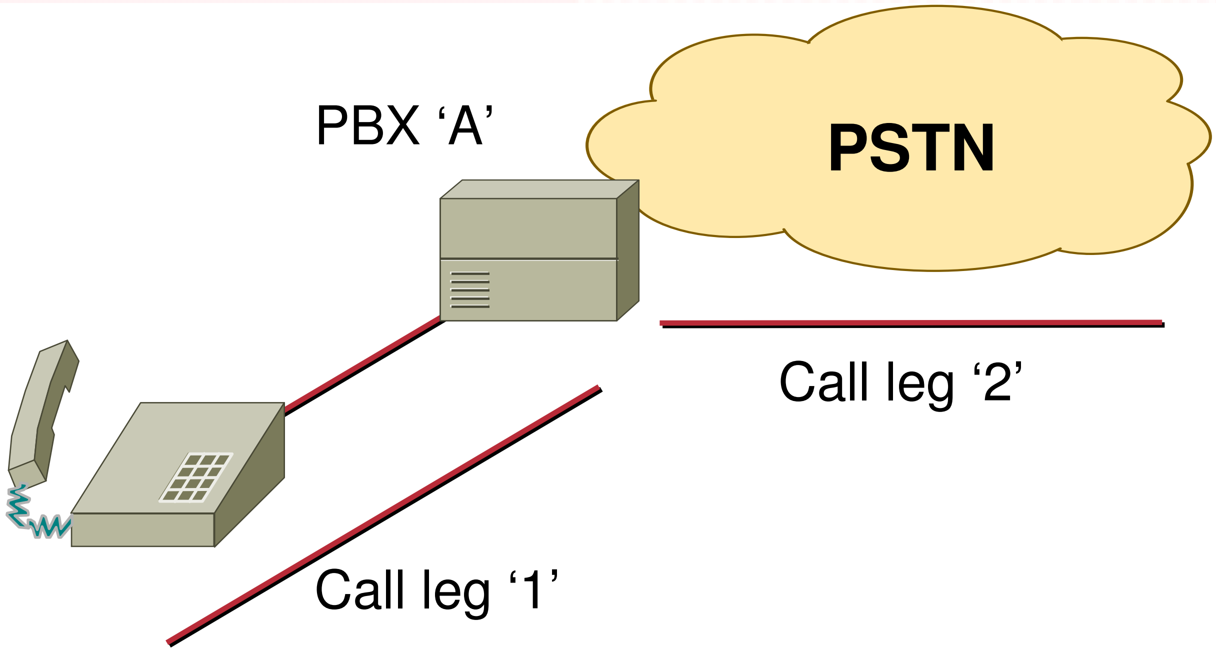
- **Two call legs bridged together**

Call Flow



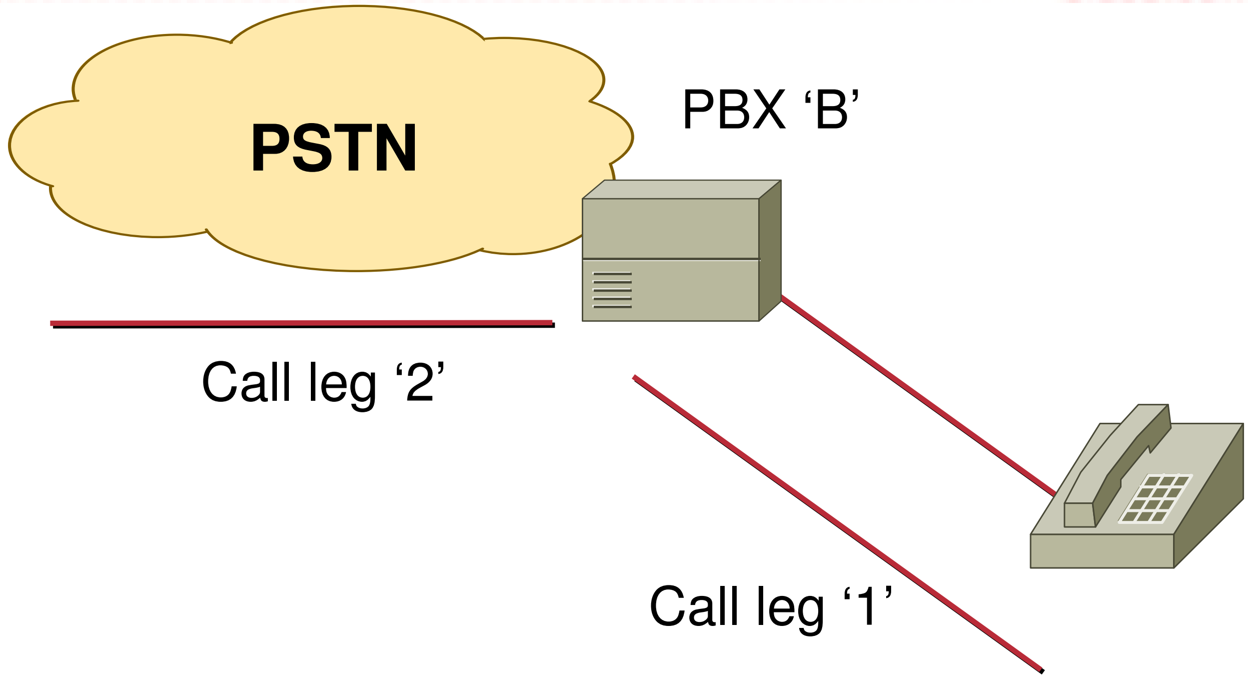
- **Caller A lifts receiver "off hook"**
- **PBX responds with dial-tone**
- **Call leg 1 is "created"**

Call Flow (Cont.)



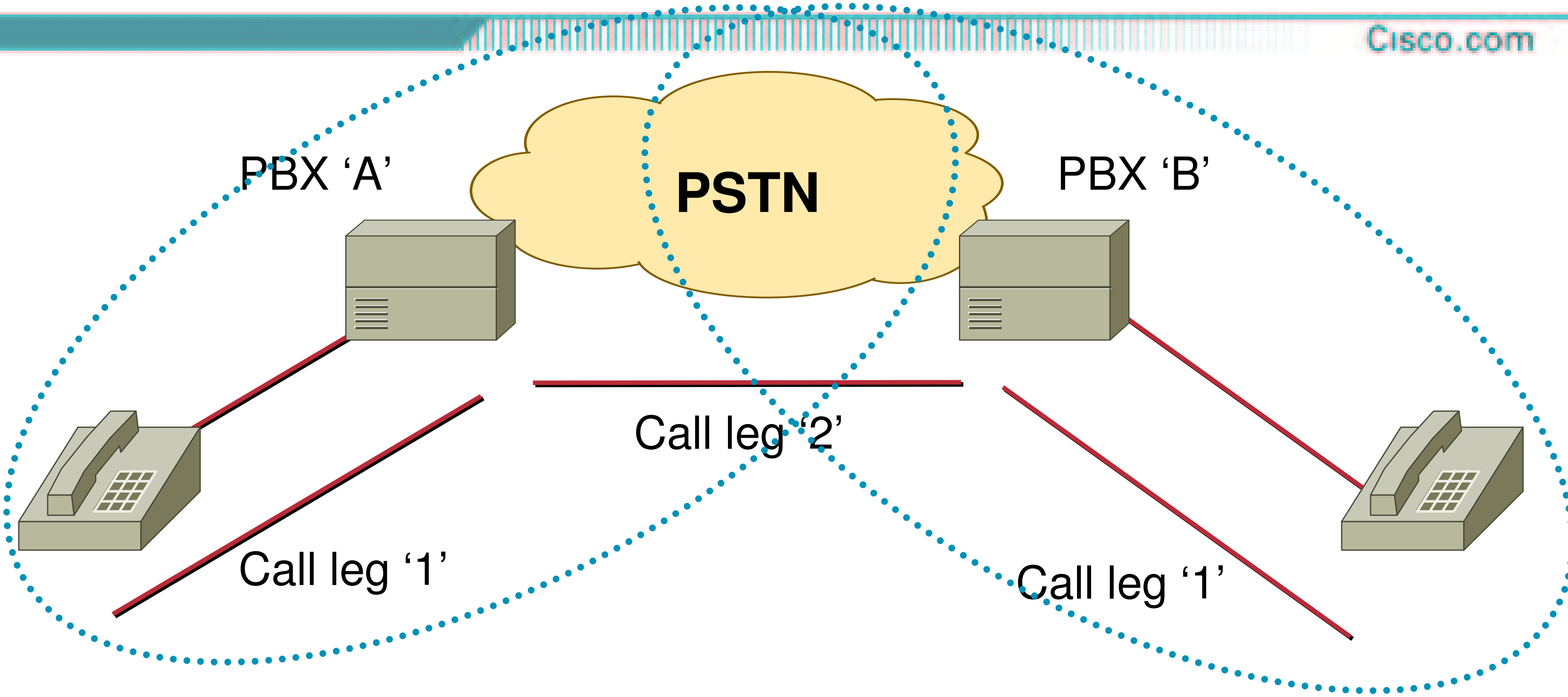
- **Caller A dials number**
- **PBX maps dialed number to trunk circuit**
- **Call leg 2 is "created"**
- **Two call legs "conferenced" together**

Call Flow (Cont.)



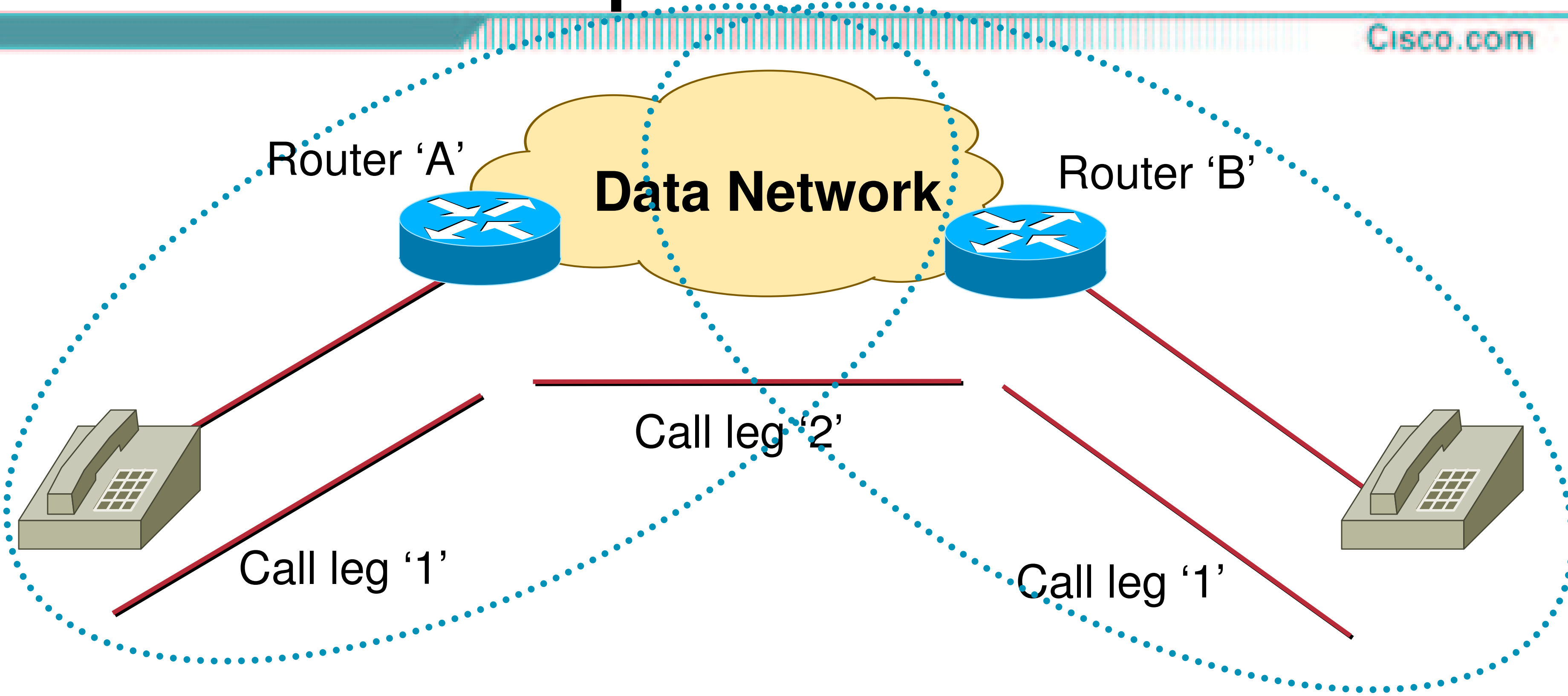
- **PBX 'B receives call setup from PSTN**
- **"Creates" first call leg ("2")**
- **Maps received digits to extension**
- **Alerts extension, "creating" second call leg ("1")**

Call Conferenced



- **Each PBX has bridged two call legs, each of local significance only**
- **Neither PBX has knowledge of the other PBX's second call leg**

Packet Voice Replacement



- **Simply replace PBX and PSTN with Router and data packet network**

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Vo* Bearer Technologies

- VoATM
- VoFR
- VoIP

Voice over ATM AAL1
Circuit Emulation Services (CES)

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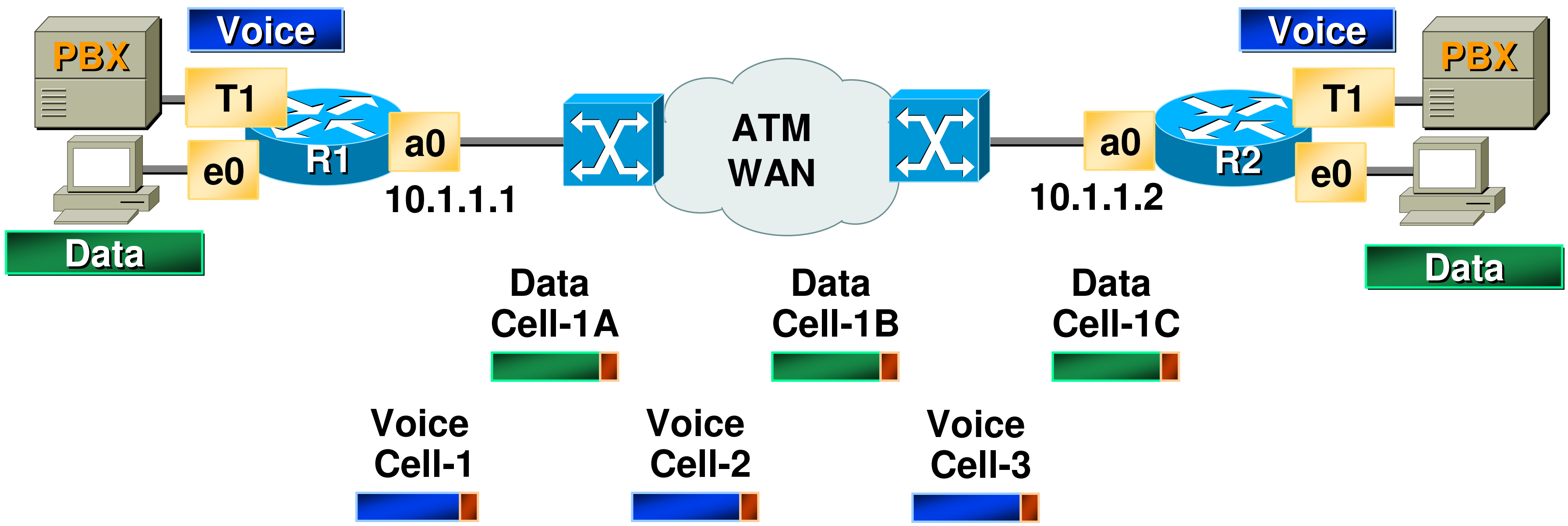
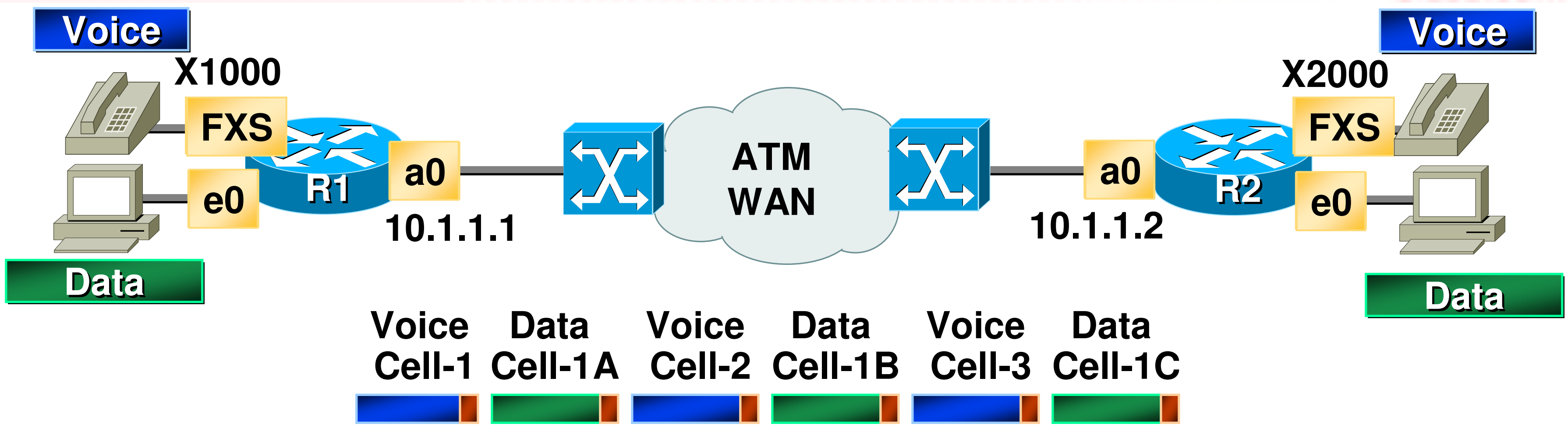


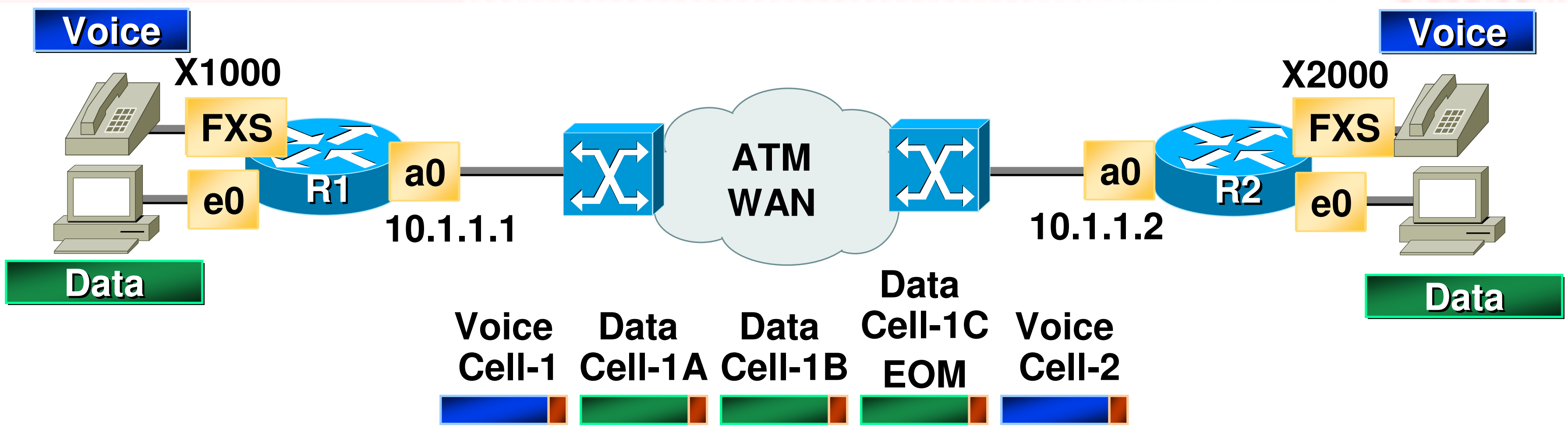
- AAL 1 ATM Constant Bit Rate (CBR) traffic type
- Dual VCs (Voice CBR and Data ABR/VBR/UBR)
- Dedicated bandwidth (not available to data)
- PBX feature transparency maintained

Voice over ATM AAL2
VToA



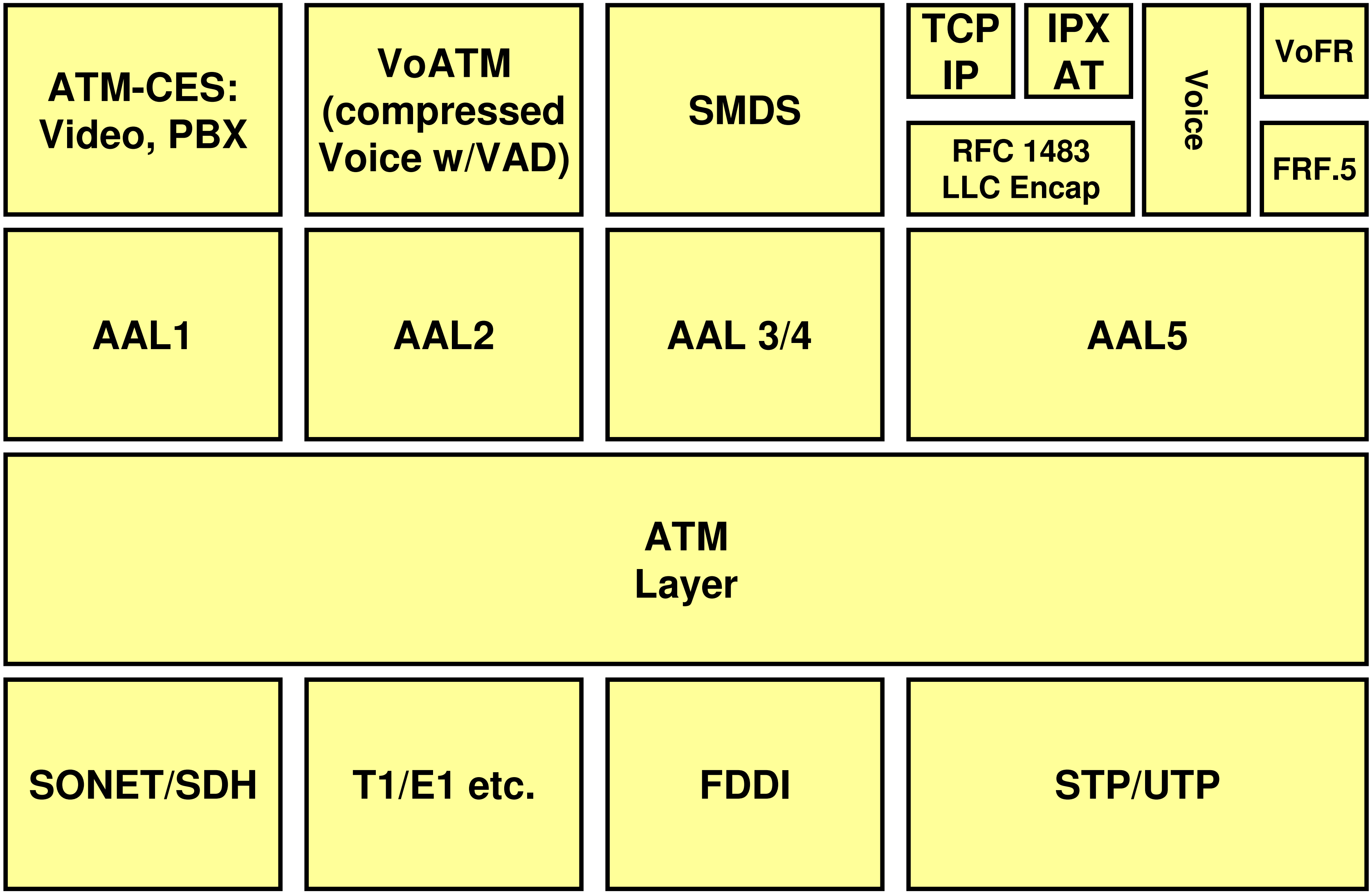
- Multiple channels within a single ATM connection (sub-cell multiplexing) using ATM Channel Identification (CID)
- Varying bandwidth requirements for the sub-cells with VBR traffic support
- Compressed voice
- Silence detection/suppression

Voice over ATM AAL5

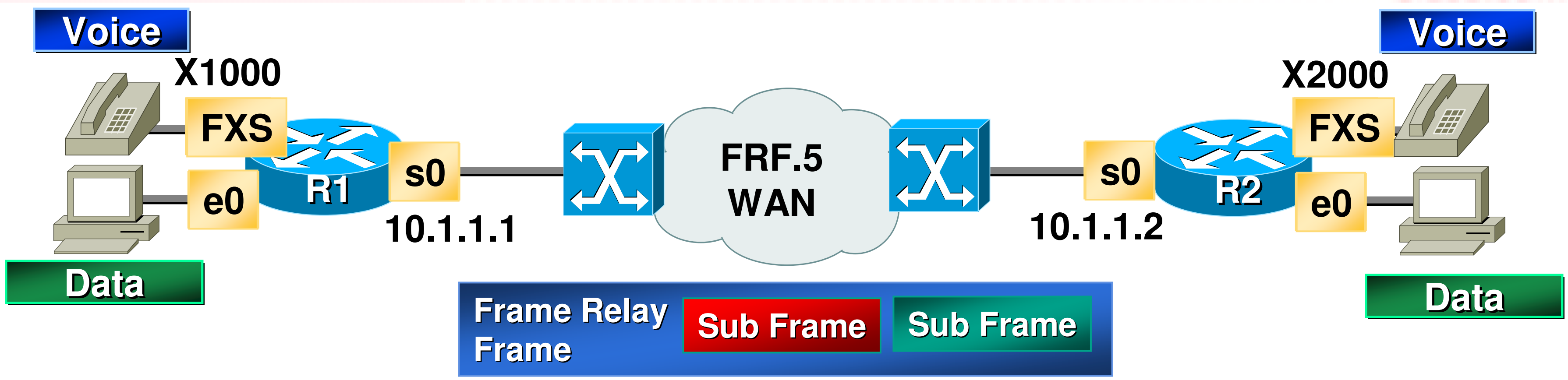


- Lower overhead (5 byte ATM header with 48 Byte payload)
- G.729 produces 20 byte payload with 20 ms voice sample
- 48 byte cell has 28 Bytes “overhead” due to padding
- No interleaving available on a single low bit rate VC

VoIPoATM or VoATM

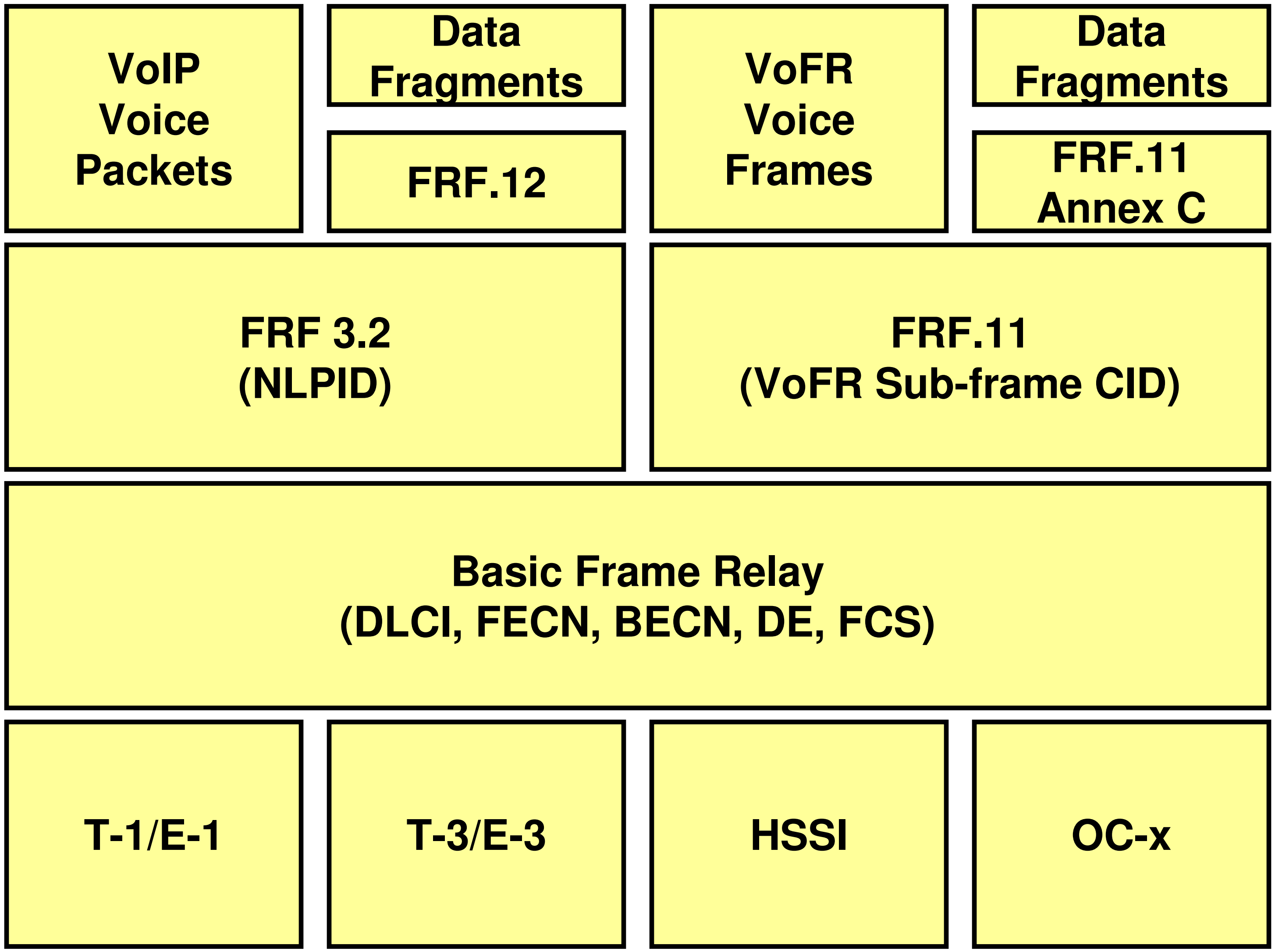


Voice over Frame Relay FRF.11

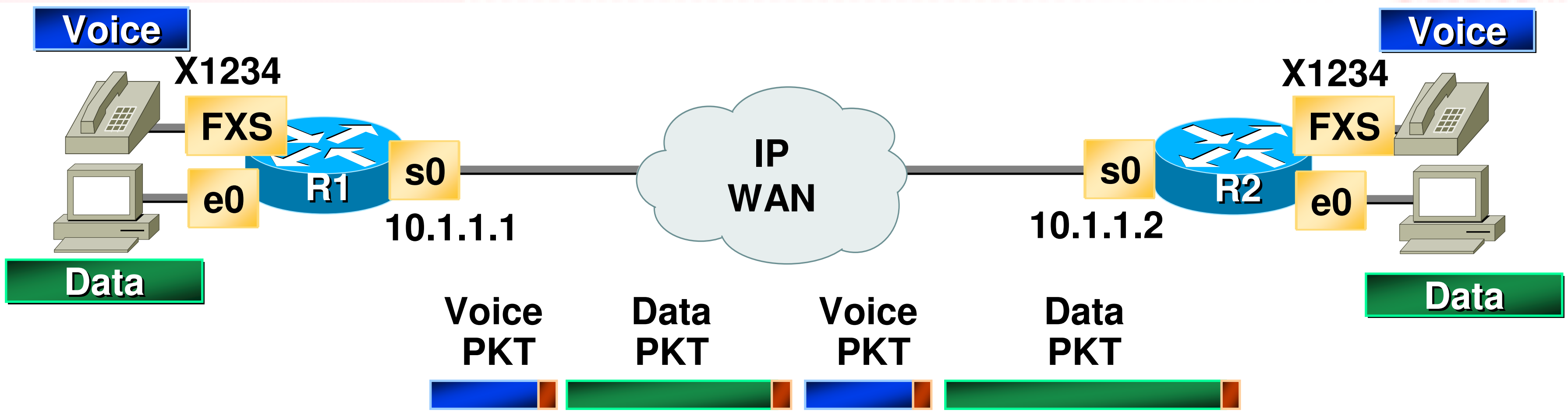


- FRF.11 creates many logical channels within one PVC
- FRF.11 sub-frame header is transparent to WAN
- FRF.11 sub-frame header may be viewed as an extension to the FR header DLCI address (like IP port numbers)
- CPE uses the sub-frame header to distinguish among different conversations and/or data streams
- Allows packing of many sub-frames in one FR frame
- FRF.12 provides Link Fragmentation and Interleaving

VoIPoFR or VoFR



Voice over Internet Protocol



- Data link layer independent
- Multiple trunks multiplexed by UDP port number
- Real Time Protocol (RTP)
- Real Time Transport Control Protocol (RTCP)
- RTP compression for improved bandwidth efficiency
- Enhances application integration

Real-Time Protocol RFC1889

- Payload type identification—Voice, video, compression type
- Sequence numbering
- Time stamping
- Delivery monitoring
- Carried on the odd port number with RTCP

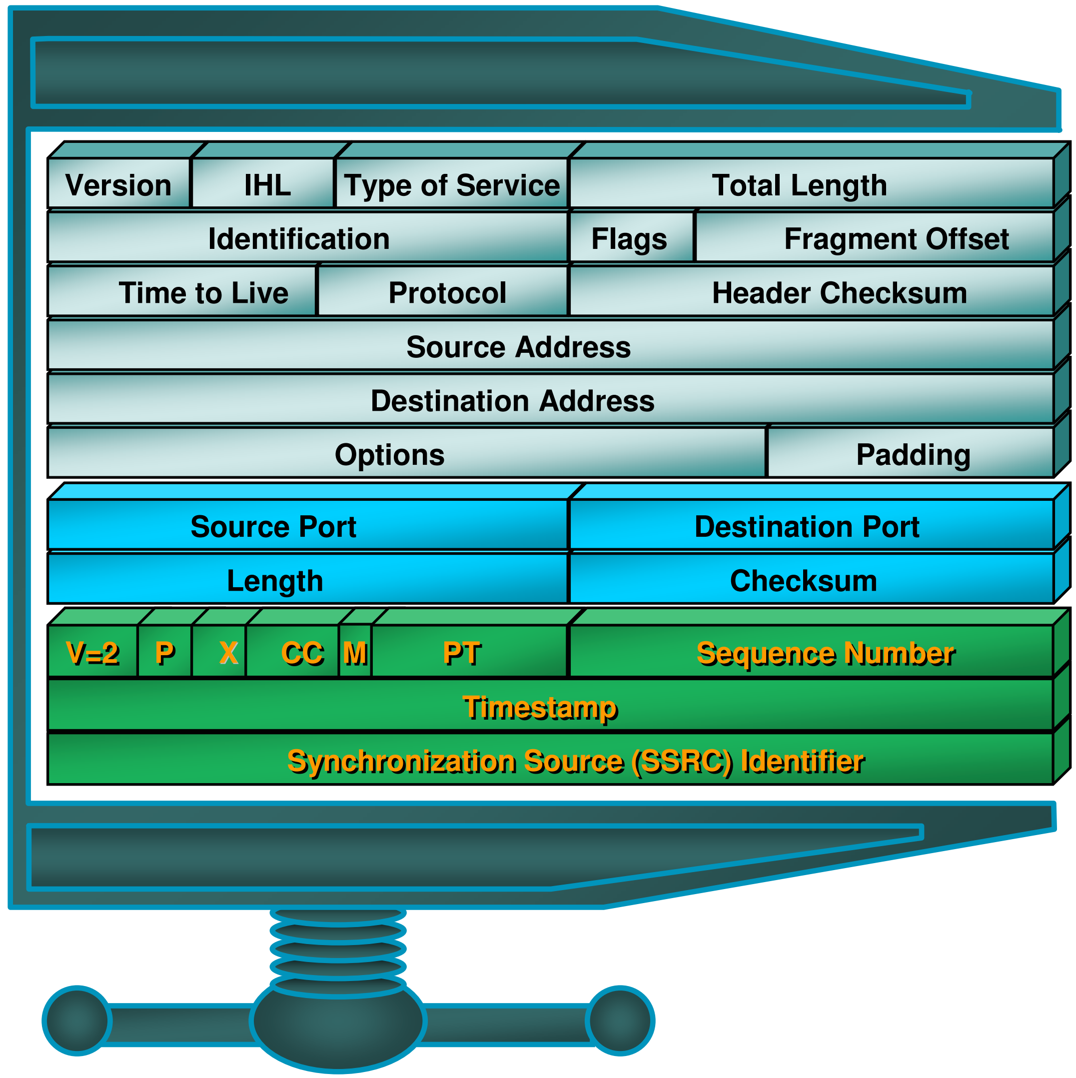
4 Bytes	VER		CC	M	Payload Type	Sequence Number
4 Bytes	RTP Timestamp					
4 Bytes	Synchronization Source (SSRC) ID					

VoIP Bandwidth Reduction
RTP Header Compression

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



Real-Time Transport Control Protocol (RTCP) RFC1890

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- Provides feedback on the quality of data distribution
- Conveys minimal session-control information (i.e. identifying a participant)
- Tracks participants in an RTP session
- Limits its own feedback send rate (< 5%)
- Carried on the even port number with RTP

Session Description Protocol (SDP) RFC2327

- A session description protocol for multimedia connections
- Developed by IETF music WG
- Simple/flexible
  - Text based
  - Extensible
- Announcements made by Session Announcement Protocol (SAP), E-mail, HTTP, etc.

- **V** = Version
- **O** = Originating device organization
- **S** = Description of the SDP message
- **C** = IP Address or Hostname that the originator expects the media to arrive at
- **T** = Time field
- **M** = Media description that the originator expects to receive
- **A** = Media attributes…

SDP Sample


```
v=0
o=- 49451 3 IN IP4 127.0.0.1
s=Sample Program
i=Sample Program
t=0 0
a=cat:3
a=tool:IP/TV Program Guide 1.6.0.71
a=type:broadcast
a=x-precept-type:scheduled
m=video 49152/1 RTP/AVP 31 32 96
c=IN IP4 239.255.0.1/15
a=rtpmap:96 WBIH/90000
m=audio 16384/1 RTP/AVP 0 3 5 14 96 97 98 99 100 101 102 103 104 105 106
c=IN IP4 239.255.0.1/15
a=rtpmap:96 X-WAVE/8000
a=rtpmap:97 X-WAVE/11025
…
```

Program Meta Data

Protocol and CODECS

UDP Port

Address

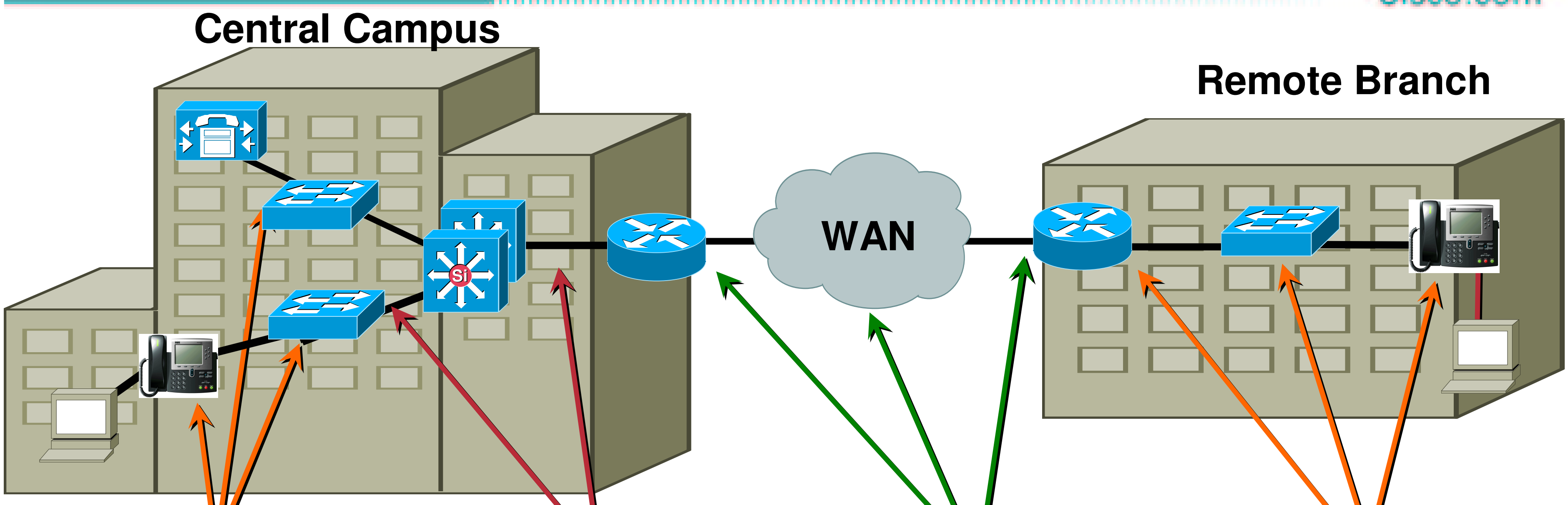
Vo* Transport Summary

Consideration	VoIP	VoFR	VoATM
Bandwidth Usage	Medium w/o cRTP Low w/cRTP (Depends on L2)	Low	AAL2—Low AAL5—Medium AAL1—High
Router Egress QoS	CBWFQ w/LLQ IP RTP Priority	Low Latency Queue (LLQ)	Low Latency Queue (LLQ)
WAN QoS	L3/L2 QoS Mapping	WAN Switch Proprietary QoS (High Priority VC)	WAN Switch Robust QoS (CBR, ABR, VBR-rt)
Serialization Delay (<768 Kbps)	LL—MLPPP FR—FRF.12 ATM—Dual VC	FRF.12	Dual VCs
Open Architecture	H.323, SIP, SCCP, MGCP Open APIs	Transport only	Transport only
Cost	Low (no VC mesh)	Low w/o mesh High w/mesh	Low w/o PVC mesh Medium w/SVCs High w/PVC mesh

Why QoS?

- Lost and delayed packets are caused by network quality, network congestion, delay, and jitter
- Solved with Cisco IOS classification, queuing, and network provisioning

QoS Planning (See VVT-211)



QoS - Campus Access

- Speed and Duplex settings
- Classification/Trust on IP Phone and Access switch
- Multiple queues on IP Phone and Access ports

QoS - Campus Distrib

- Layer 3 Policing
- Multiple queues on all ports; Priority Queuing for VoIP
- WRED within Data queue for Congestion Management

QoS - WAN

- Low-Latency Queuing
- Link Fragmentation and Interleave
- Bandwidth Provisioning
- Admission Control

QoS - Branch

- Classification and Trust Boundaries on IP Phone , Access Layer switch and router
- Multiple queues on IP Phone and all Access ports

Why VoIP?

- VoIP enables applications and new revenue generating services
- VoIP scales better than VoFR and VoATM
- VoIP is independent of network topology
- VoIP is the only option for Internet telephony
- VoIP is the only option for campus Ethernet
- VoIP is the only viable telecommuter solution via DSL, cable and ISDN

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IP Signaling Protocols

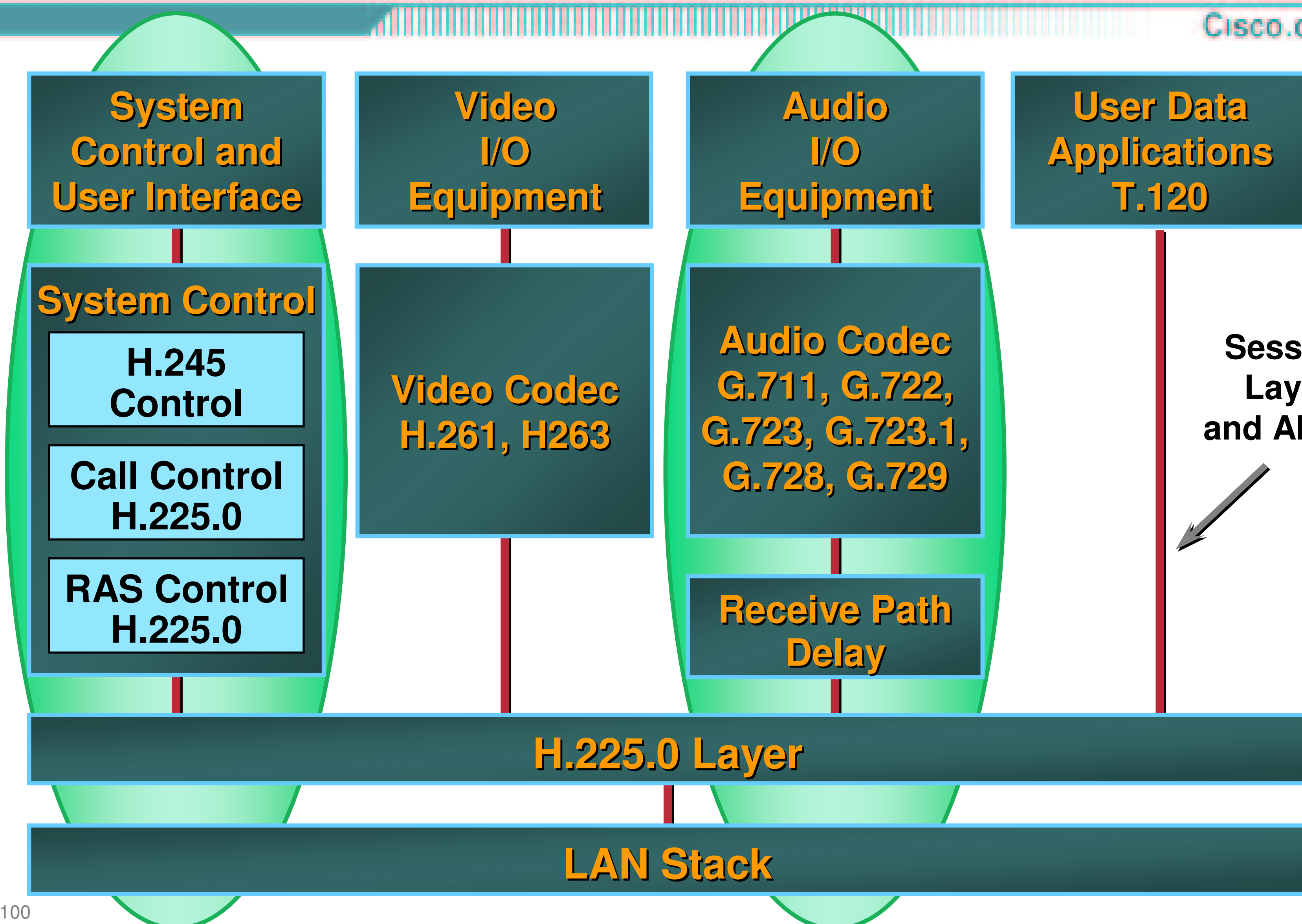
- H.323
- Session Initiation Protocol (SIP)
- Media Gateway Control Protocol (MGCP)
- Skinny Client Control Protocol (SCCP)

H.323

H.323 Background

- ITU H.323 issued in 1996
- H.323 was intended to be non-ISDN replacement of H.320 for packet-based multimedia communications
- Distributed call processing with intelligent endpoints
- Addressing by E.164 or account name

ITU H.323


System Control and User Interface
Video I/O Equipment
Audio I/O Equipment
User Data Applications T.120
System Control
H.245 Control
Call Control H.225.0
RAS Control H.225.0
Video Codec H.261, H263
Audio Codec G.711, G.722, G.723, G.723.1, G.728, G.729
Receive Path Delay
Session Layer and Above
H.225.0 Layer
LAN Stack

H.323 Components

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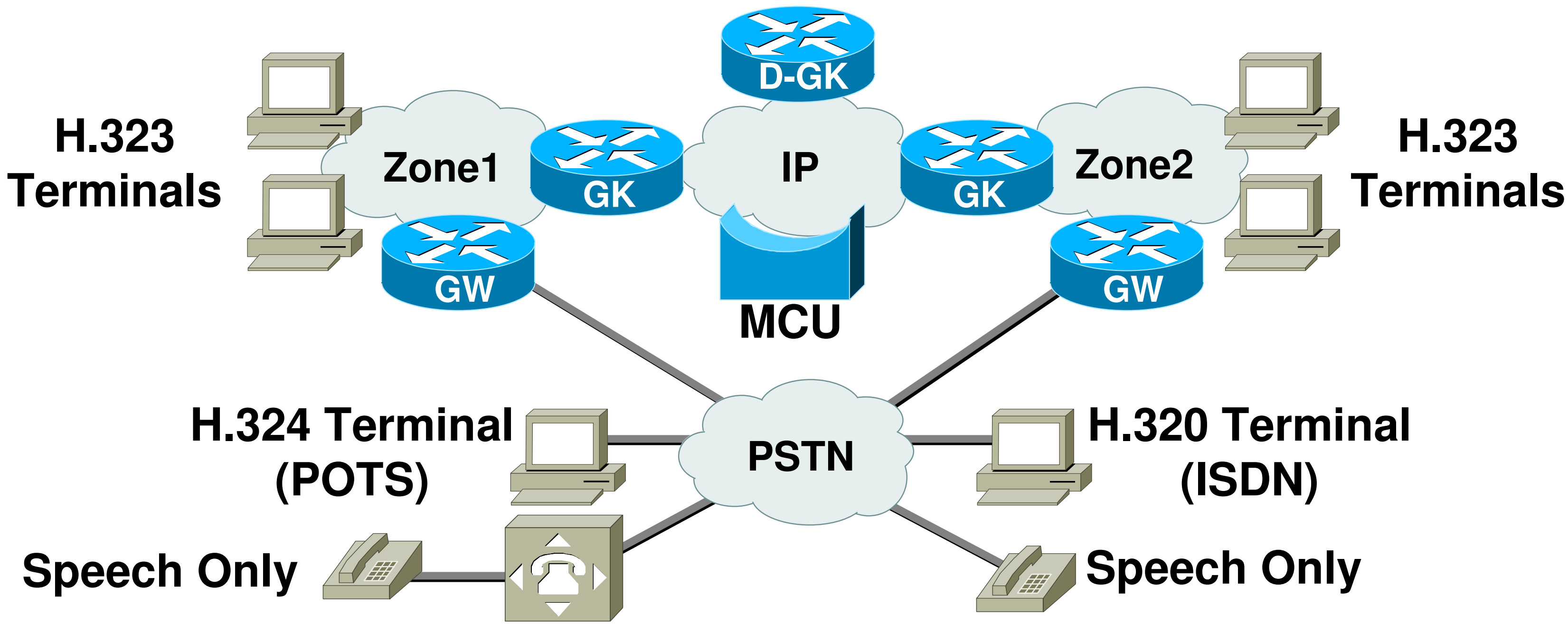


- Gateway (GW)—IP conversion
- Gatekeeper (GK)—Phone number and name to IP address lookup and zone bandwidth management/proxy
- Directory Gatekeeper (D-GK)—Dial plan database of GKs
- MCU—Multipoint Control Unit to mix audio and replicate video

H.225 RAS Protocol Elements

Gatekeeper Discovery

- GatekeeperRequest (GRQ)
- GatekeeperConfirm (GCF)
- GatekeeperReject (GRJ)

Terminal/Gateway Registration

- RegistrationRequest (RRQ)
- RegistrationConfirm (RCF)
- RegistrationReject (RRJ)

Terminal/Gateway Unregistration

- UnregistrationRequest (URQ)
- UnregistrationConfirm (UCF)
- UnregistrationReject (URJ)

Location Request

- LocationRequest (LRQ)
- LocationConfirm (LCF)
- LocationReject (LRJ)

Call Admission

- AdmissionRequest (ARQ)
- AdmissionConfirm (ACF)
- AdmissionReject (ARJ)

Disengage

- DisengageRequest (DRQ)
- DisengageConfirm (DCF)
- DisengageReject (DRJ)

Resource Availability

- Resource Availability Indicator (RAI)
- Resource Availability Confirm (RAC)

Bandwidth Change

- Bandwidth Change Request (BRQ)
- Bandwidth Change Confirm (BCF)
- Bandwidth Change Reject (BRJ)

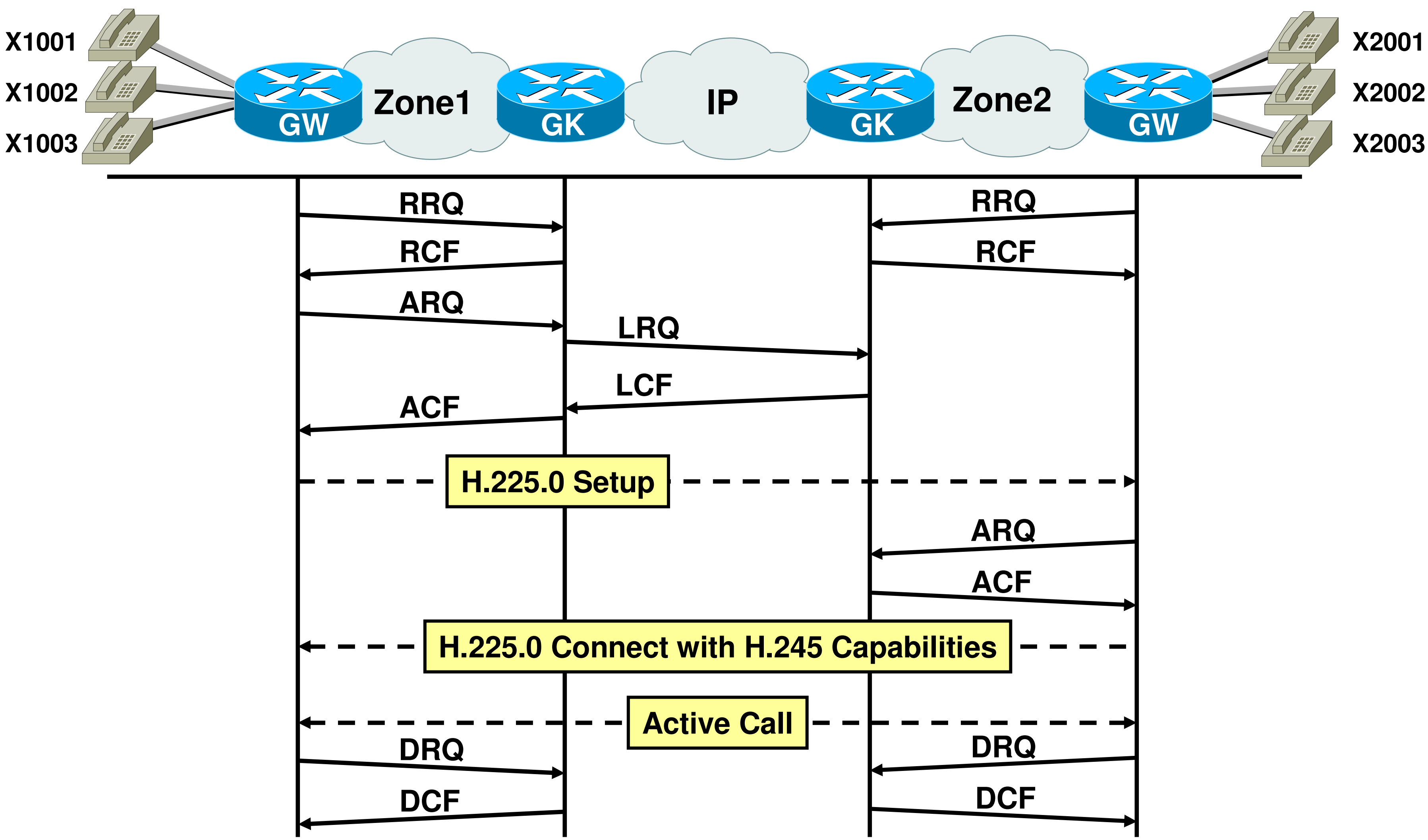
Request in Progress

- Request in Progress (RIP)

Status Queries

- InfoRequest (IRQ)
- InfoRequestResponse (IRR)
- InfoRequestAck (IACK)
- InfoRequestNak (INAK)

H.323 Signaling

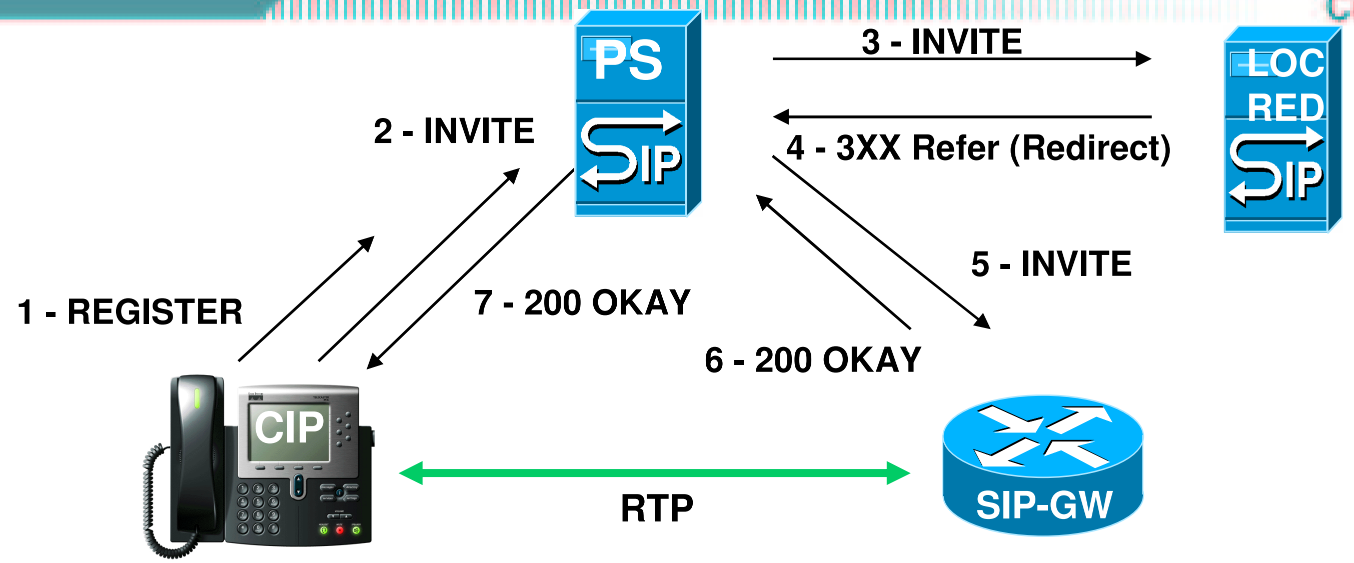


Session Initiation Protocol
SIP

Session Initiation Protocol Background

- IETF RFC3261 issued in 2002
- Protocol for creating, modifying and terminating sessions (Internet multimedia conferences, Internet telephone calls and multimedia distributions) with one or more participants
- Distributed call processing with intelligent endpoints
- Based on IETF RFCs (RTP, RTCP, HTTP, SDP, DNS, SAP, RTSP)
- Addressing by E.164, e-mail, or DNS SRV record

Cisco SIP Components



- **SIP Proxy Server (PS)**
  - **Registration Server (REG)—Accepts registration requests from UAs**
  - **Redirect Server (RED)—Maps SIP request to one or more addresses**
  - **Location Server (LOC)—Provides information on a callees locations**
- **User Agent (UA)**
  - **SIP Gateway (SIP-GW)**
  - **IP Phones (CIP)**

SIP Messages

- INVITE—Indicates a user or service is being invited to participate in a call session
- ACK—Confirms that the client has received a final response to an INVITE request
- BYE—Terminates a call and can be sent by either the caller or the callee
- CANCEL—Cancels any pending searches but does not terminate a call that currently is in progress
- OPTIONS—Queries the capabilities servers
- REGISTER—Registers the address listed in the To header field with a SIP server (not GWs)
- REFER

SIP Addressing

- Fully-Qualified Domain Names

  sip:jdoe@cisco.com

- E.164 addresses

  sip:14085551234@gateway.com; user=phone

- Mixed addresses

  sip:14085551234@10.1.1.1; user=phone

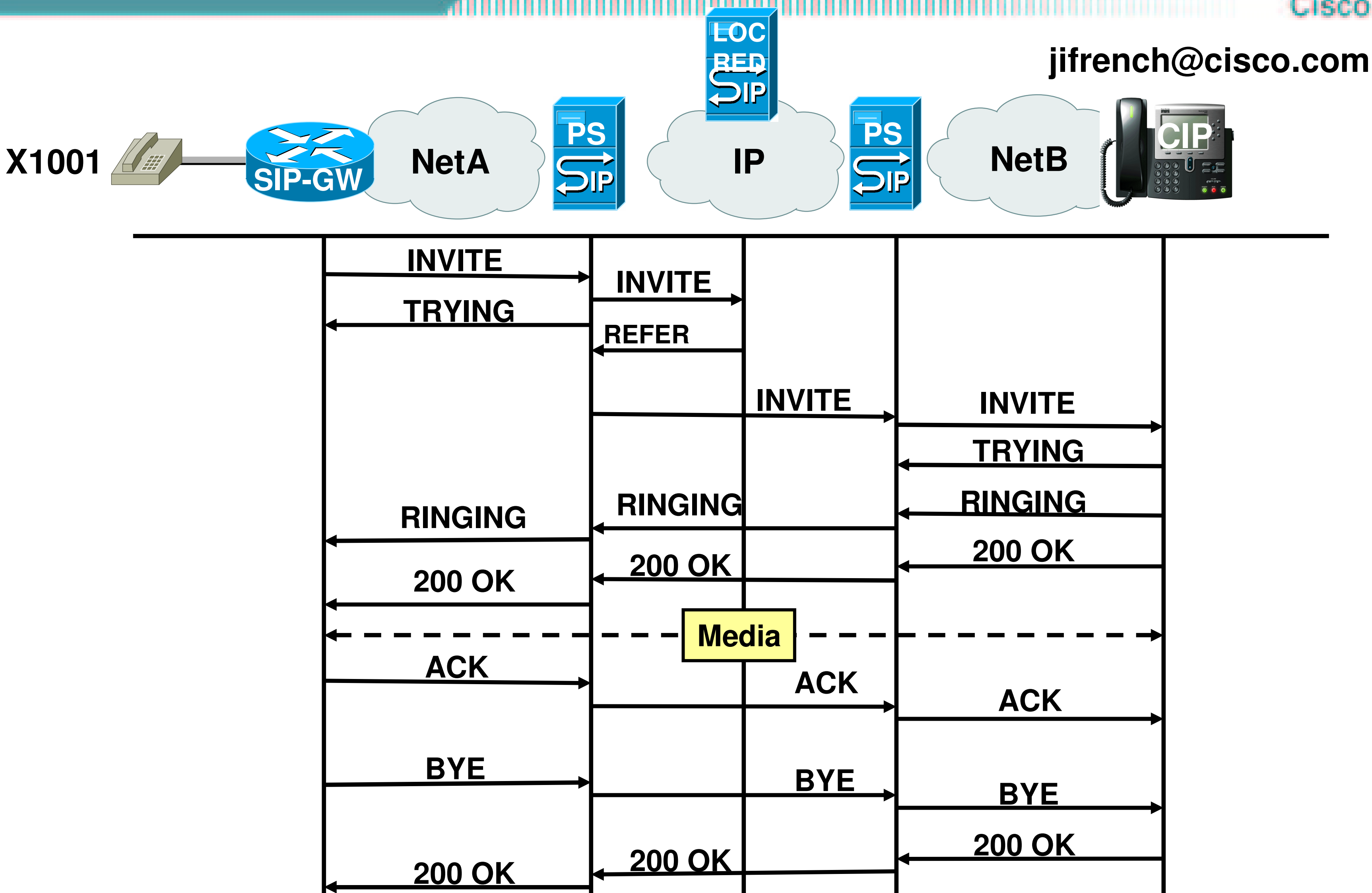
  sip:jdoe@10.1.1.1

- E.164 addresses

  tel:14085551234

Modeled after mailto URLs

SIP Signaling



H.248 Gateway Control Protocol MGCP/MEGACO

H.248/MGCP/MEGACO Background

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- IETF issued Media Gateway Control Protocol 1.0 RFC 2705 in Oct 1999
- MGCP is the result of merging SGCP and IPDC
- Centralized device control with simple endpoints for basic and enhanced telephony services
- Uses IETF Session Description Protocol (SDP)
- Addressing by E.164 phone number

Gateway Control Evolution

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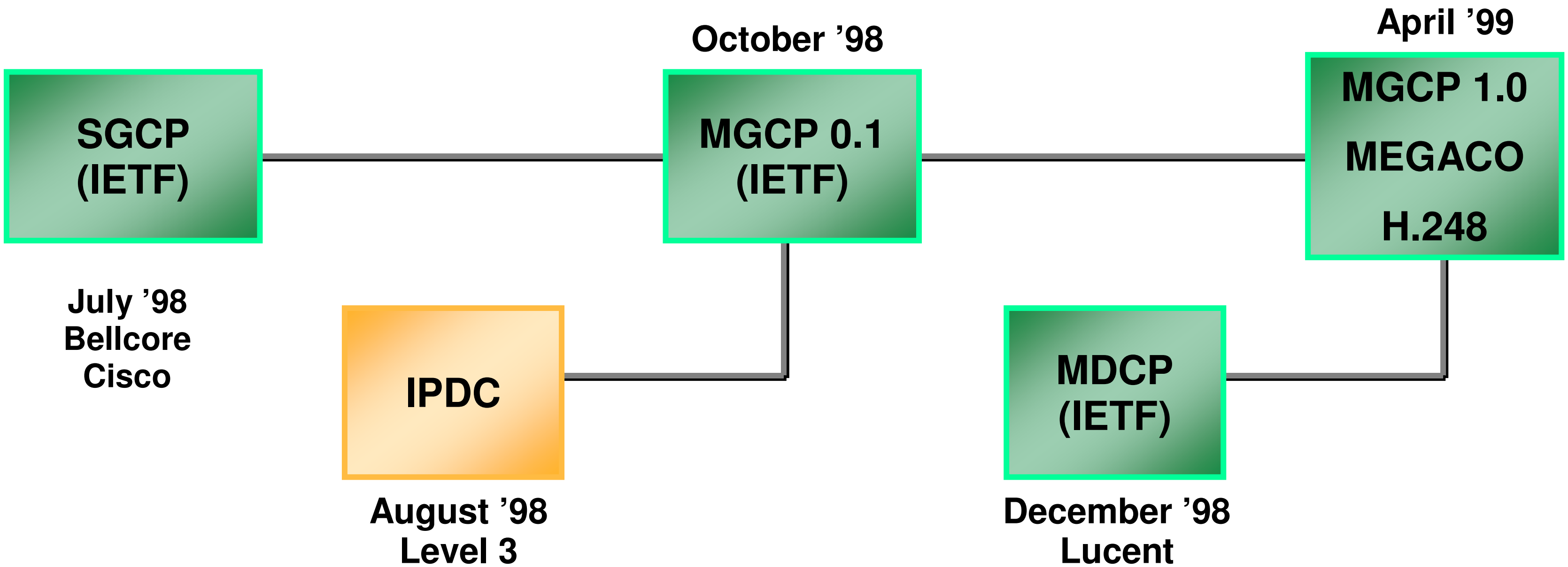
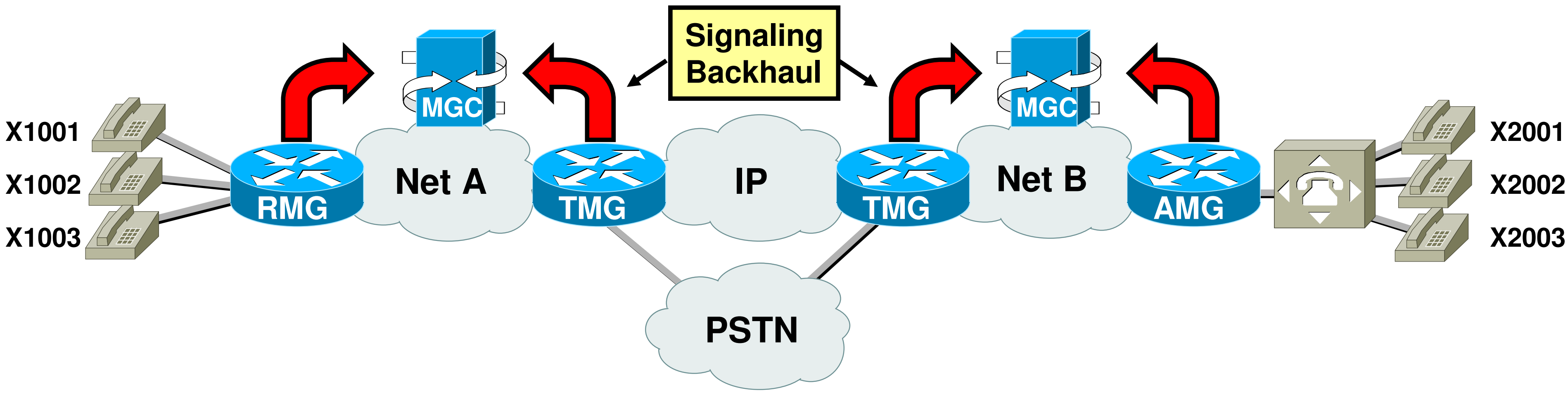


- SGCP—Simple Gateway Control Protocol
- IPDC—IP Device Control
- MGCP—Media Gateway Control Protocol
- MDCP—Media Device Control Protocol
- MEGACO—Media Gateway Controller

MGCP Components



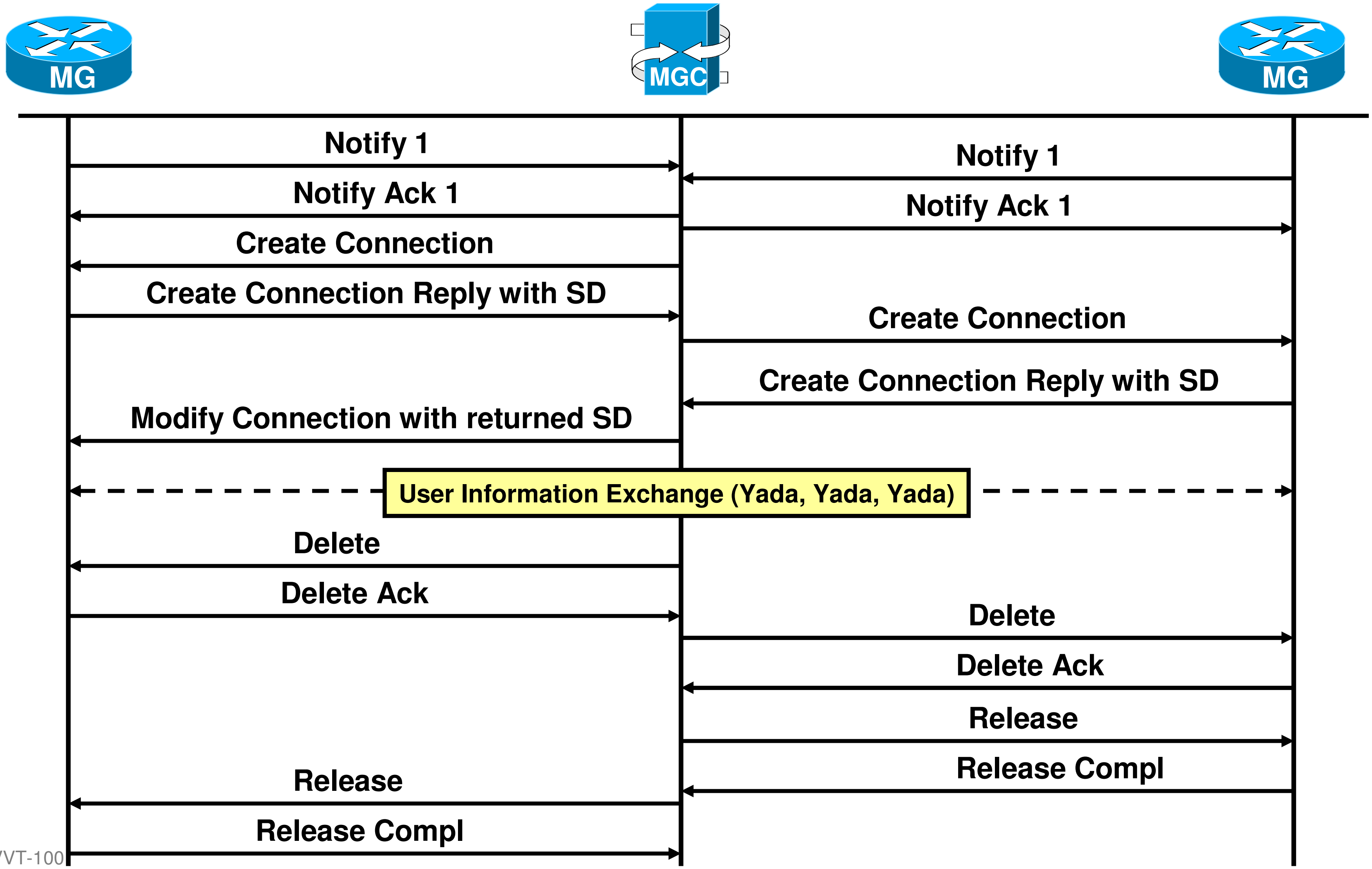
- Media Gateway Controller (MGC)
- Media Gateways (MG)

  Trunking Gateway (TMG)—Gateway interfacing SS7 bearer channels to IP network

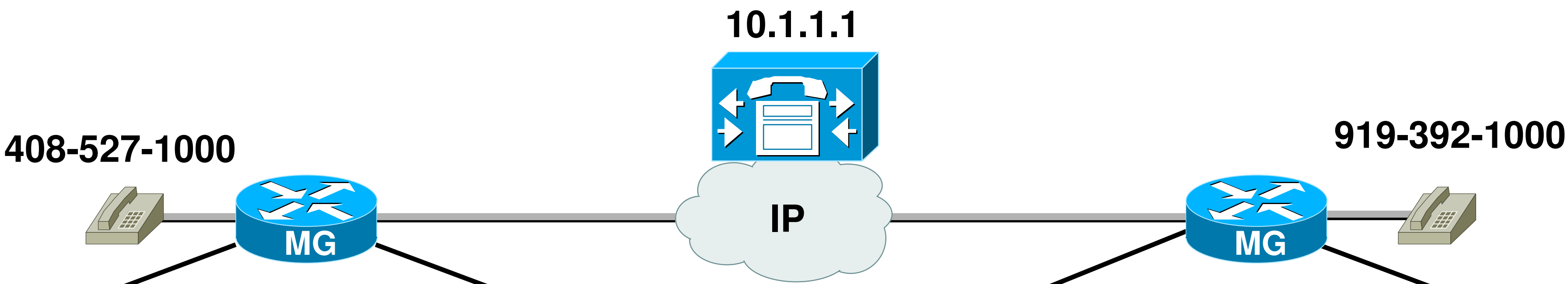
  Residential Gateway (RMG)—Gateway found in customer premise connecting POTS line to IP network

  Access Gateway (AMG)—Gateway connecting PBX trunks to IP network

MGCP/MEGACO Signaling



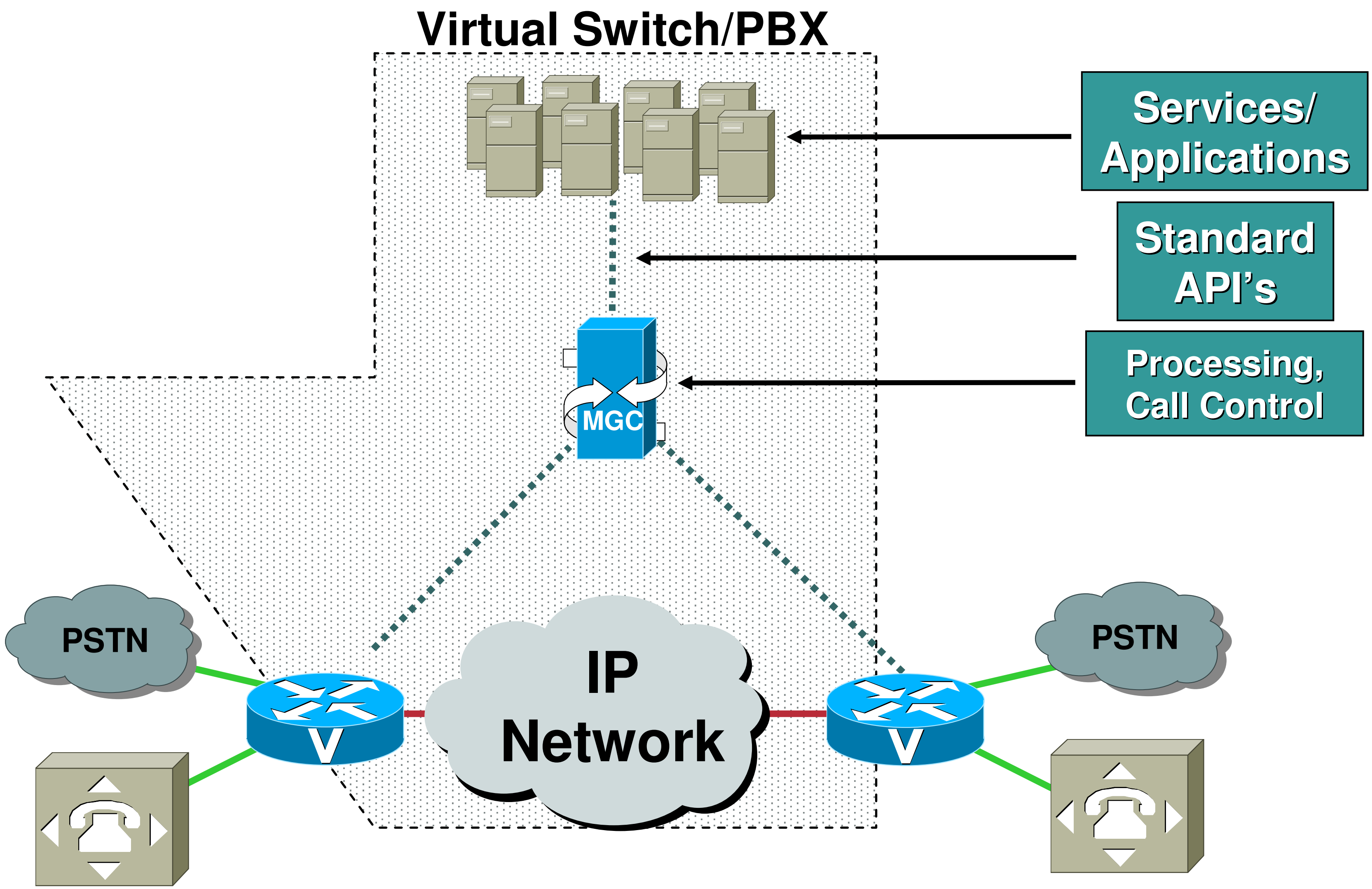
Call Manager
IOS MGCP Configuration




```
dial-peer voice 1 pots
 application mgcpapp
 port 1/1
!
mgcp
mgcp call-agent 10.1.1.1
mgcp dtmf-relay codec all
mode out-of-band
mgcp sdp simple
ccm-manager mgcp
```

```
dial-peer voice 1 pots
application mgcpapp
 port 1/1
!
mgcp
mgcp call-agent 10.1.1.1
mgcp dtmf-relay codec all
mode out-of-band
mgcp sdp simple
ccm-manager mgcp
```

MGCP Switch Replacement Softswitch (MGCP and SIP)

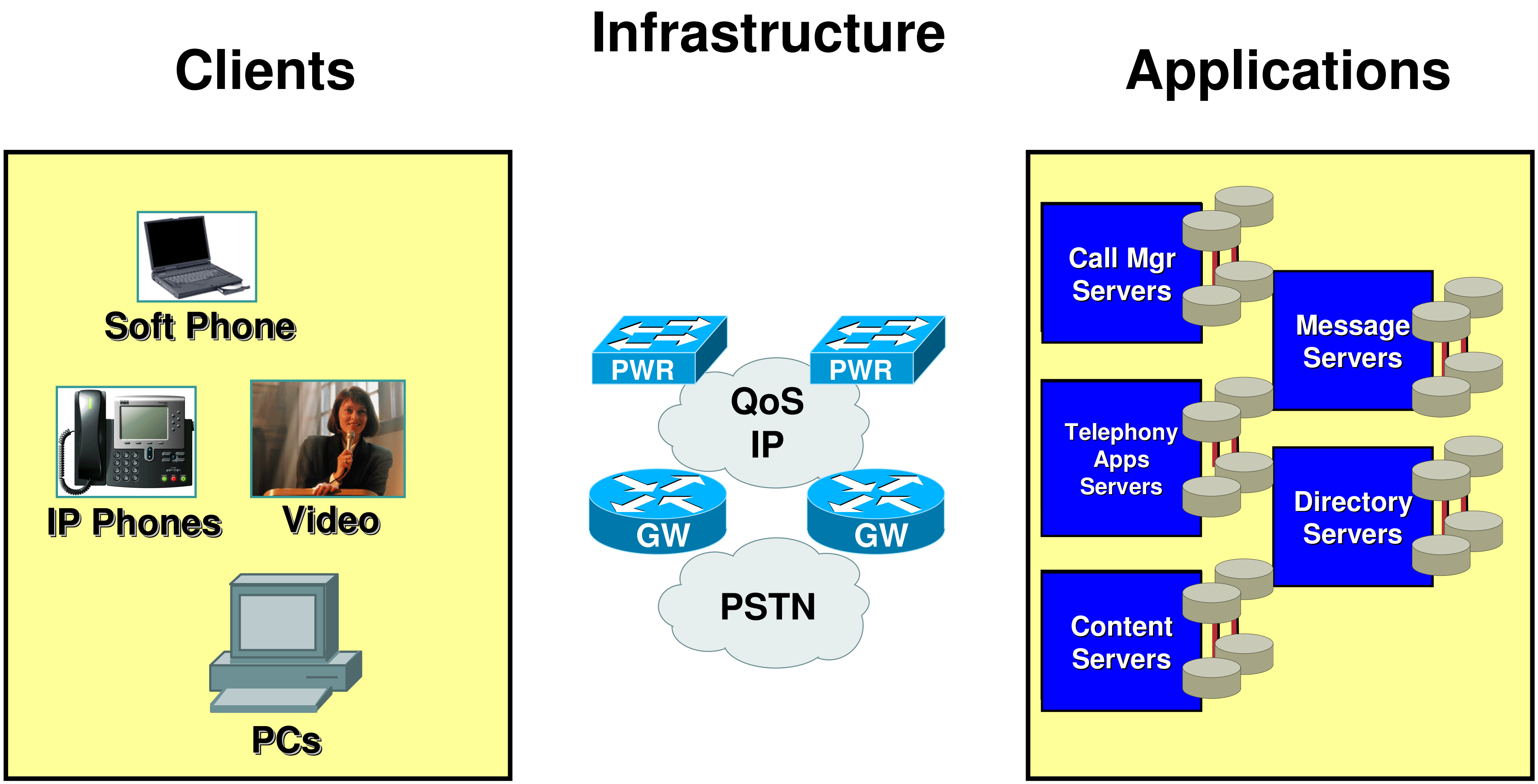


Skinny Client Control Protocol SCCP

SCCP Background

- Open protocol created by Selsius Corp. in 1996
- Protocol for creating, modifying and terminating multimedia conferences and telephone calls with simple, low cost, familiar endpoints
- Interfaces to other signaling protocols via SCCP proxy
- Addressing by E.164 or directory

AVVID Architecture



AVVID Components

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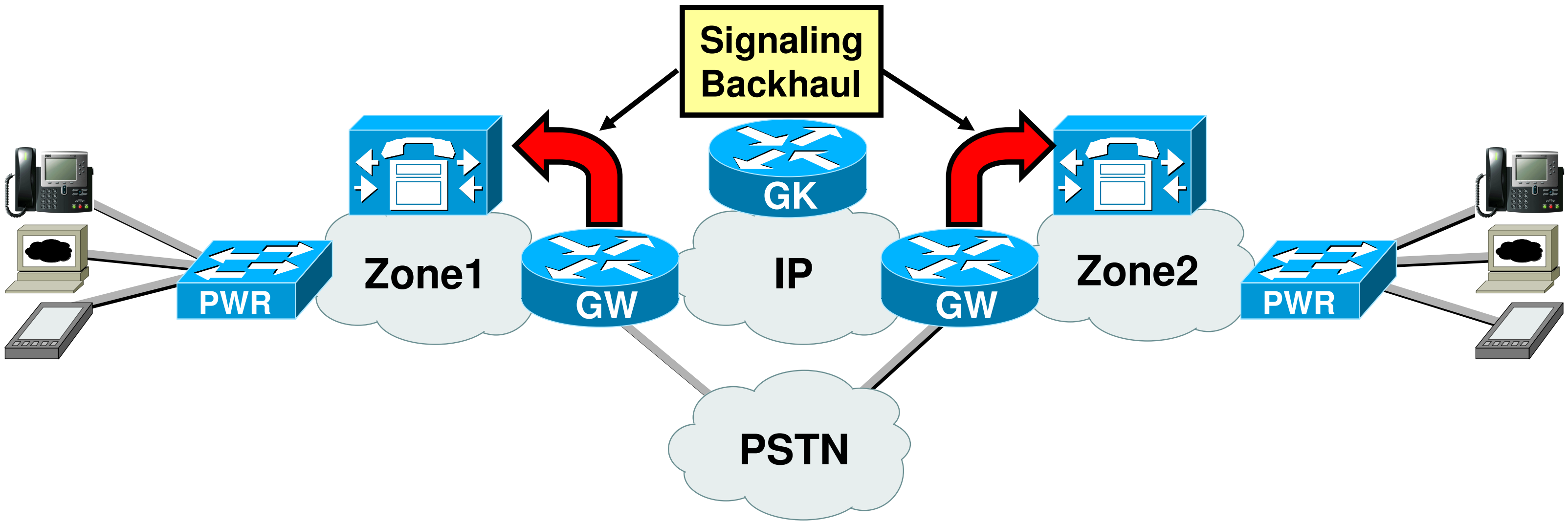
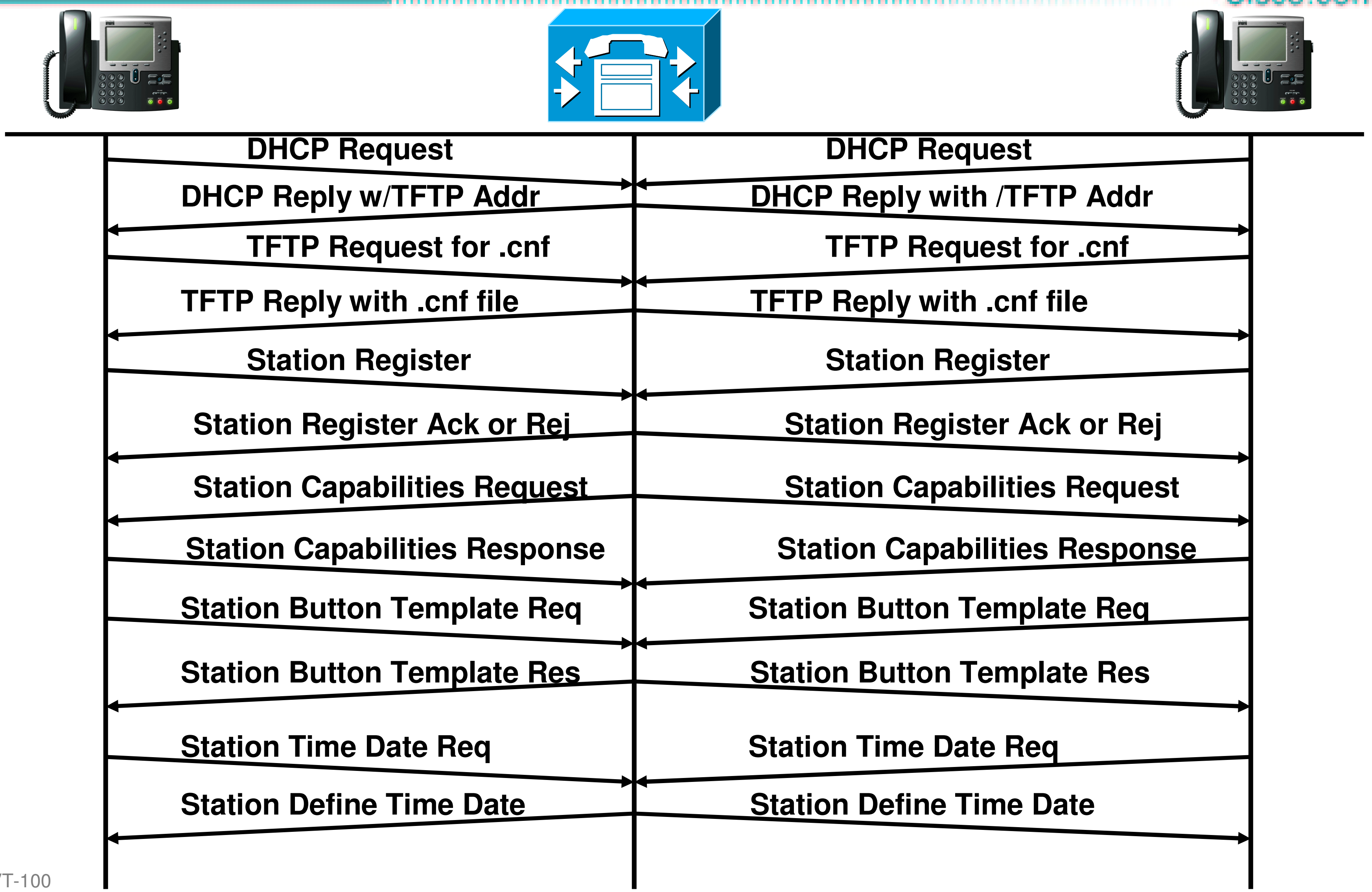


- Cisco Call Manager (CCM)
- Gateways (GW)
- Line Power Ethernet Switches (PWR)
- Clients (IP phones, Softphones, PDAs)

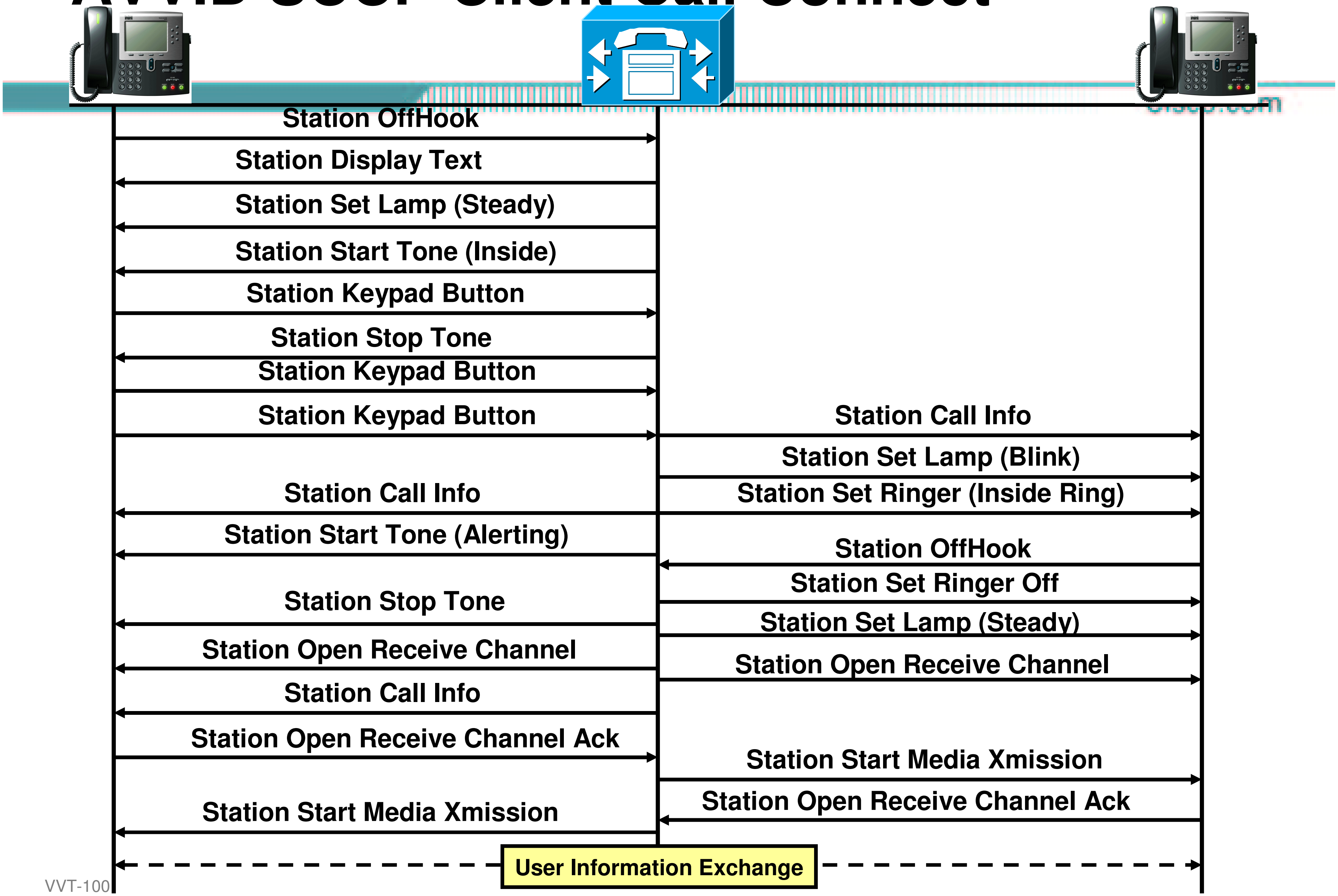
Call Manager Jobs

- Signaling (SCCP, H.323, MGCP)
- Signaling translation
- Dial Plan—Automated Call Routing (ARS)
- Phone number to IP address translation
- End point management
- User administration
- Directory management
- Clustering (2500 per CM, 5 CMs per cluster, 10000 per cluster, 100 clusters, 1,000,000)
- Database
- LDAP Directory Integration
- TFTP Server (as required)
- DHCP Server (as required)
- CDR logger
- Monitor

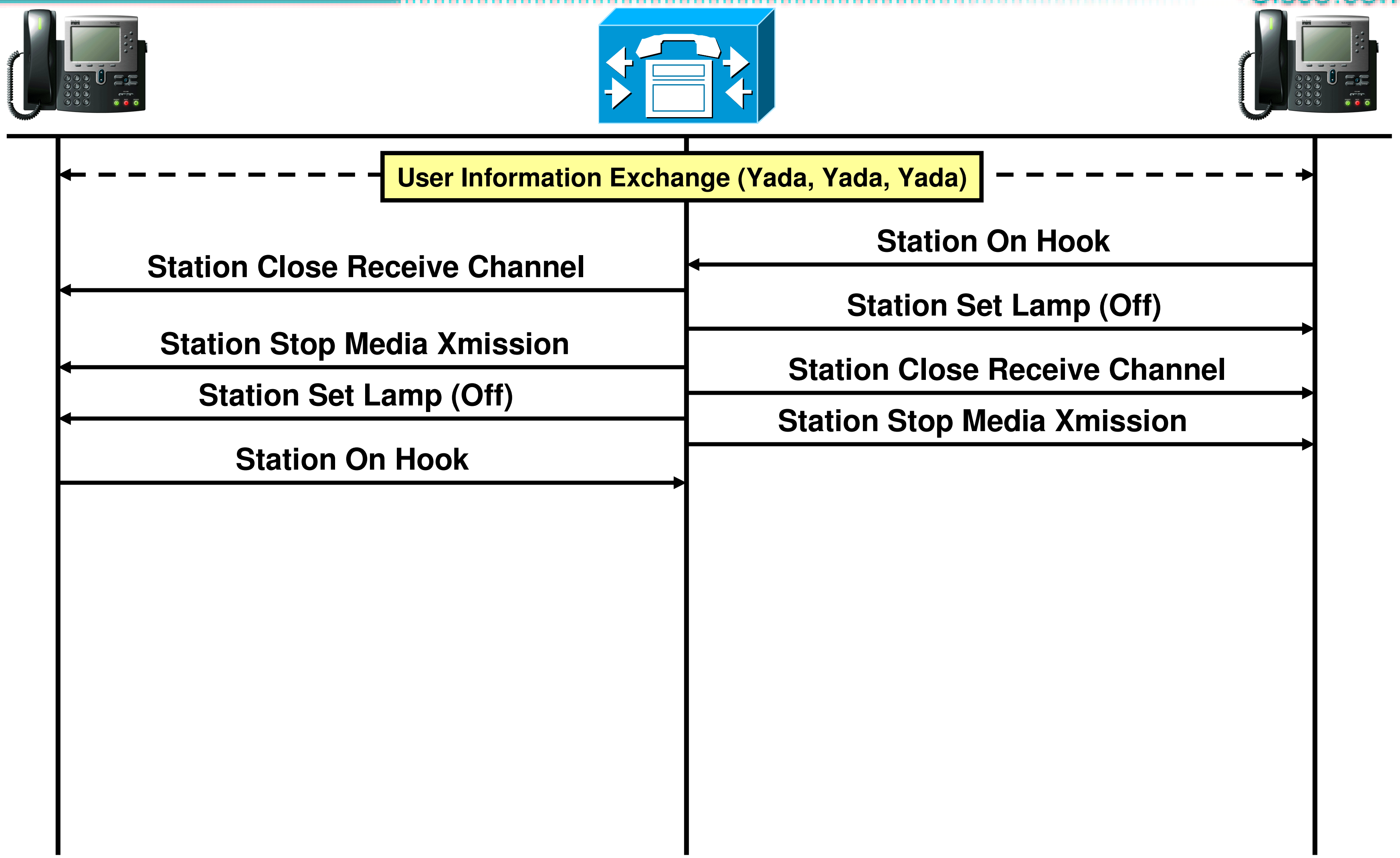
AVVID SCCP Client Registration



AVVID SCCP Client Call Connect



AVVID SCCP Client Call Disconnect



AVVID Distributed Call Processing

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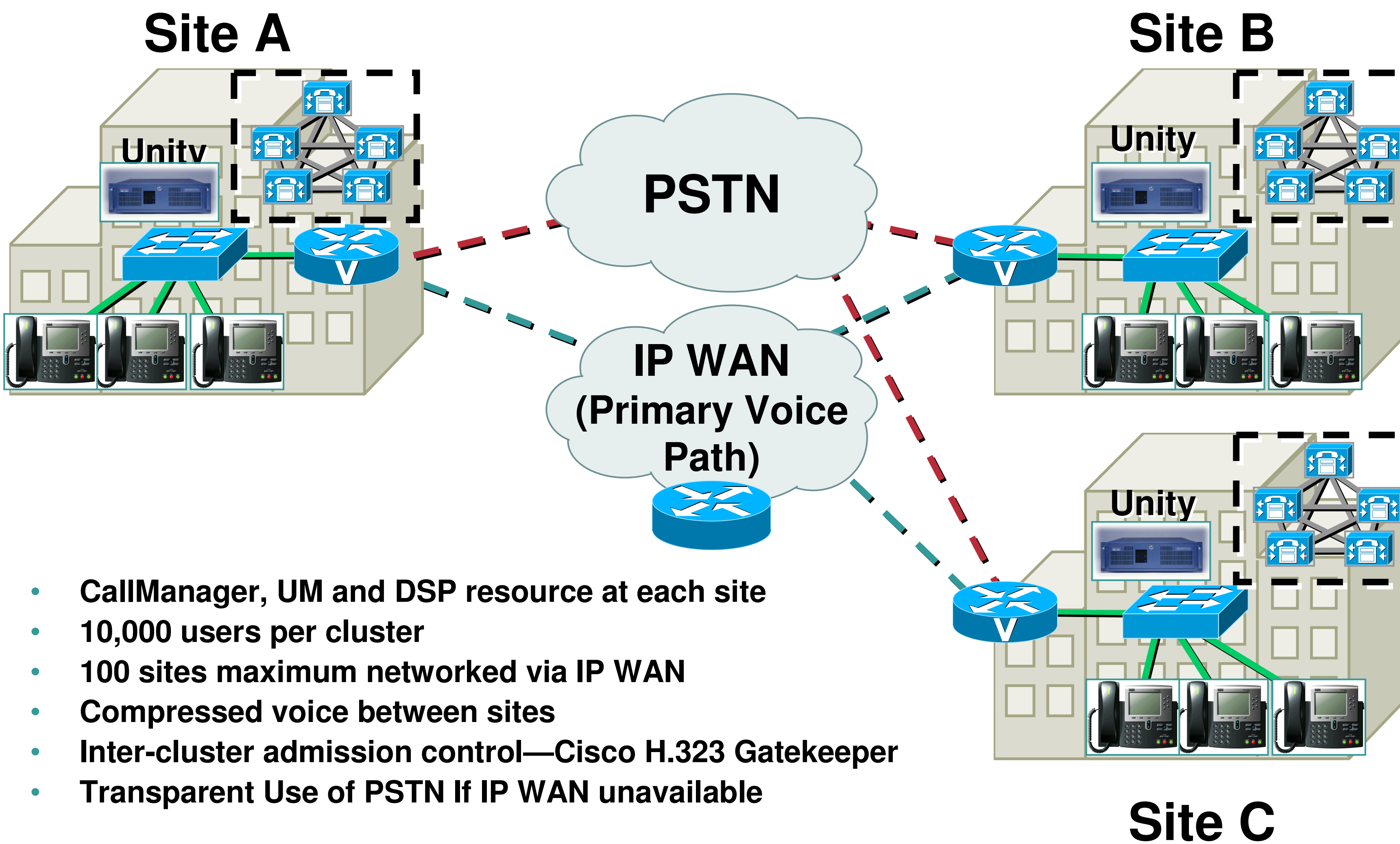
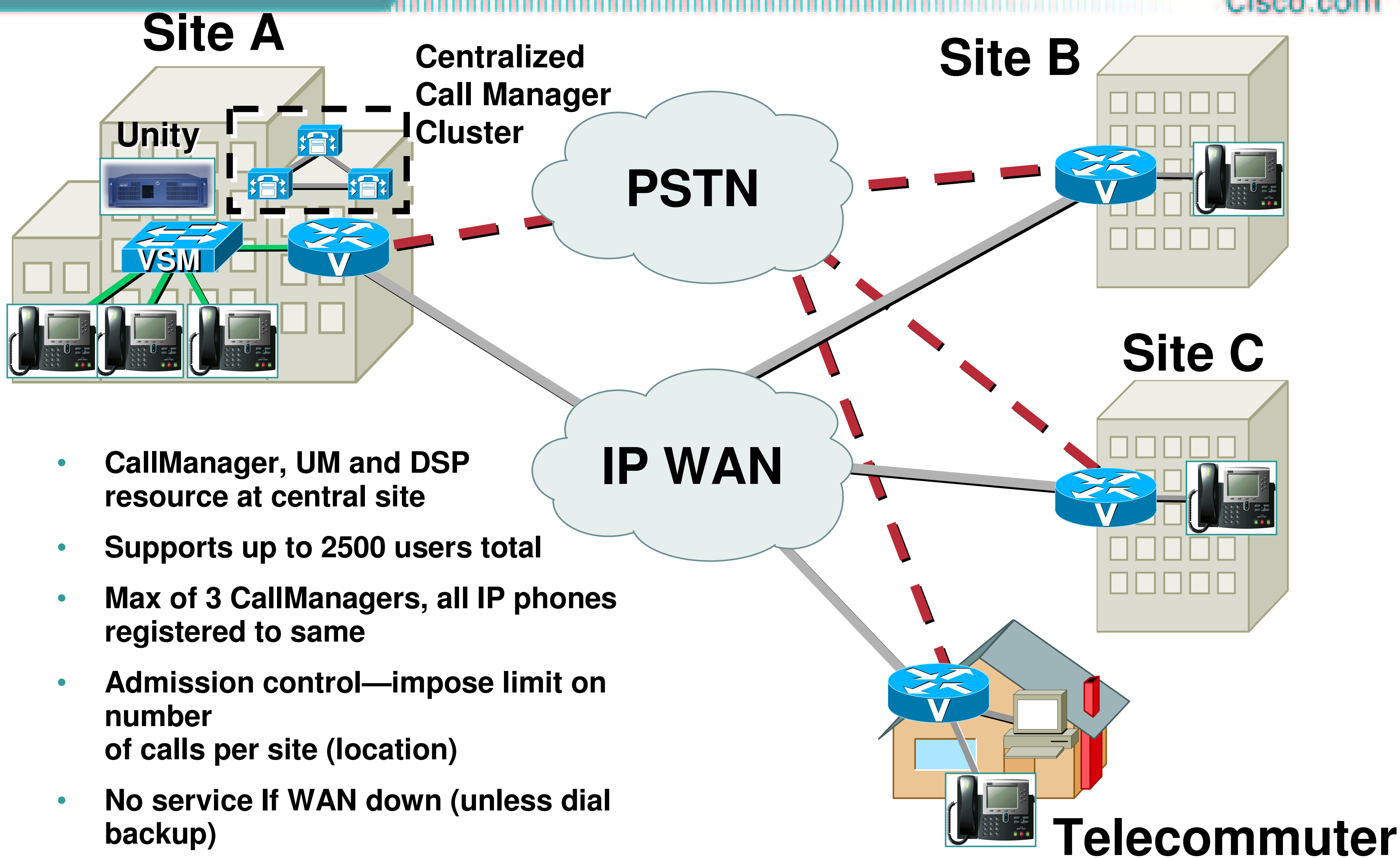


- CallManager, UM and DSP resource at each site
- 10,000 users per cluster
- 100 sites maximum networked via IP WAN
- Compressed voice between sites
- Inter-cluster admission control—Cisco H.323 Gatekeeper
- Transparent Use of PSTN If IP WAN unavailable

AVVID Centralized Call Processing



- CallManager, UM and DSP resource at central site
- Supports up to 2500 users total
- Max of 3 CallManagers, all IP phones registered to same
- Admission control—impose limit on number of calls per site (location)
- No service If WAN down (unless dial backup)

AVVID Survivable Remote Site Telephony

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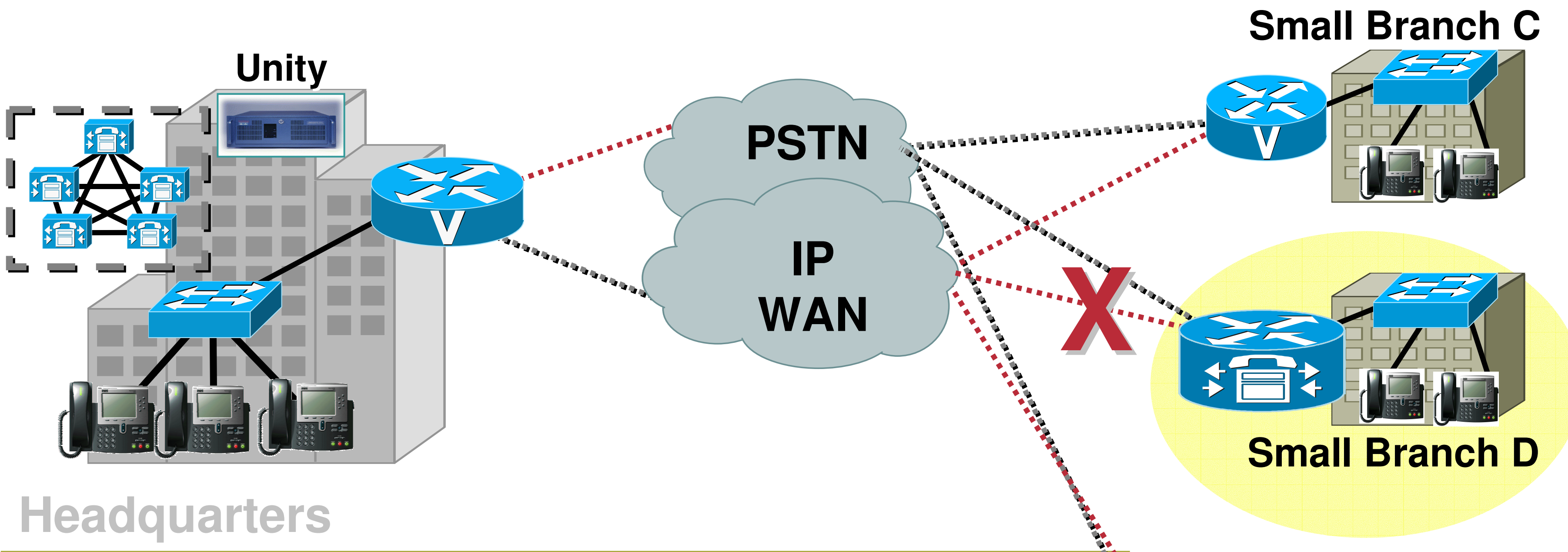
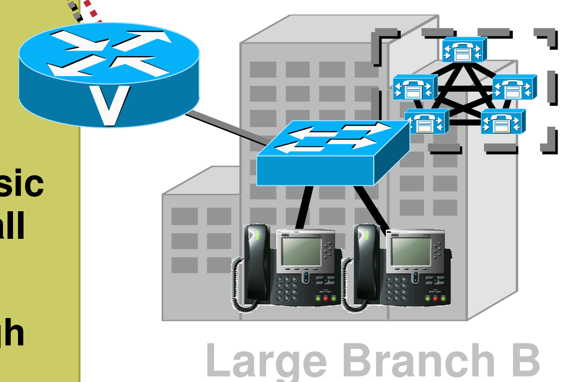
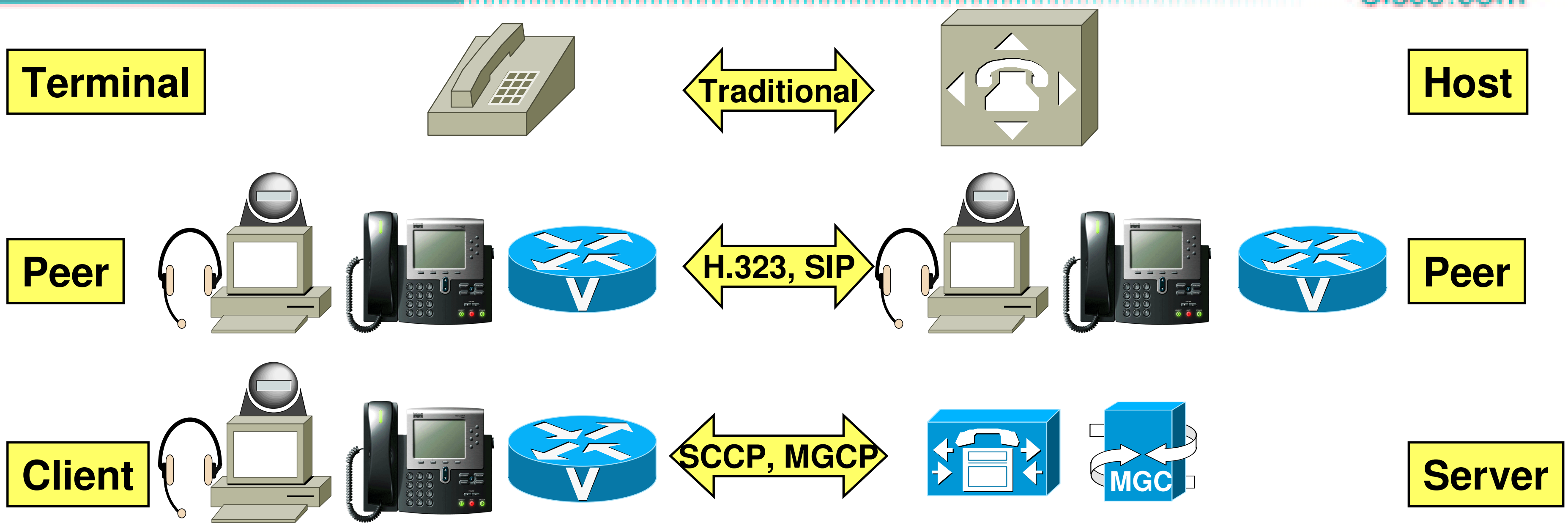


- Automatic detection of failure in the network
- Branch Office Router Intelligently reconfigures itself using System Network Auto Provisioning (SNAP) technology
- Robust functionality provided during WAN outage including basic calls, IP Phone to IP Phone and to PSTN, DID/DOD, Transfer, Call Forward, Hold, etc.
- Function added to deployed routers and 79X0 IP phones through Cisco IOS



Signaling Protocol Summary

Multimedia Application Models

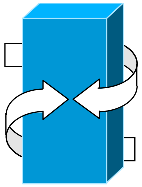


- Terminals are managed by the switch and cannot talk directly to other terminals
- Peer endpoints can place calls without a call agent but use call agents for name resolution/redirection
- Client endpoints cannot place calls without their call agent but create media streams peer to peer

Cisco Endpoint Signaling Support

- SCCP
- SIP
- MGCP

- H.323
- MGCP
- SIP
- SCCP



- SS7
- H.323
- MGCP
- SIP
- SCCP

Agenda

- Traditional Voice Review
- Market Evolution
- Packet Voice Basics
- Bearer Technologies
- Signaling Technologies
- Summary

It's All about Enabling Applications

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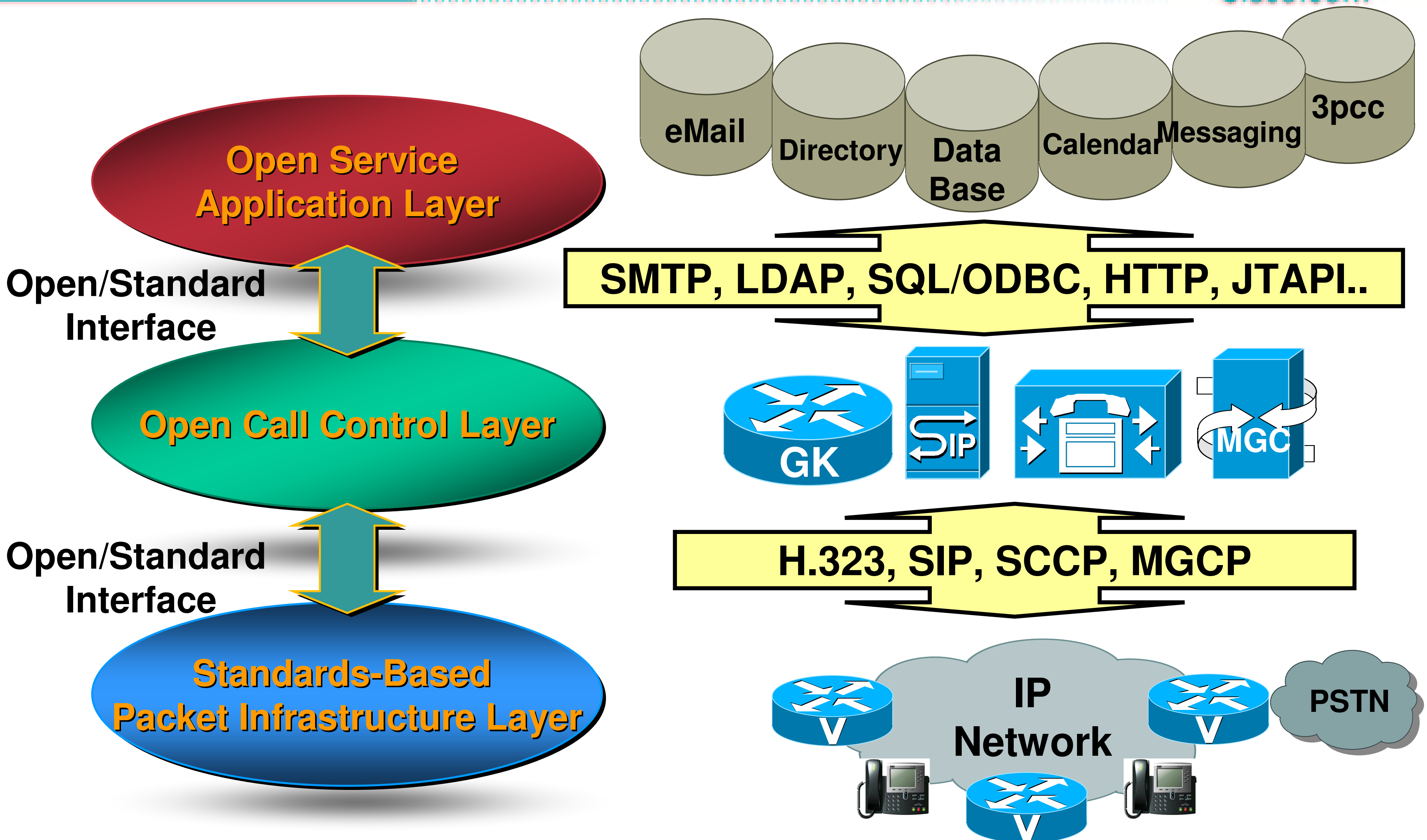

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Empowering the Internet Generation