



VoIP, AVVID and Unified Communications

Service Provider Seminar

21-23 November 2000

Nairobi , Kenya

Paul Williams

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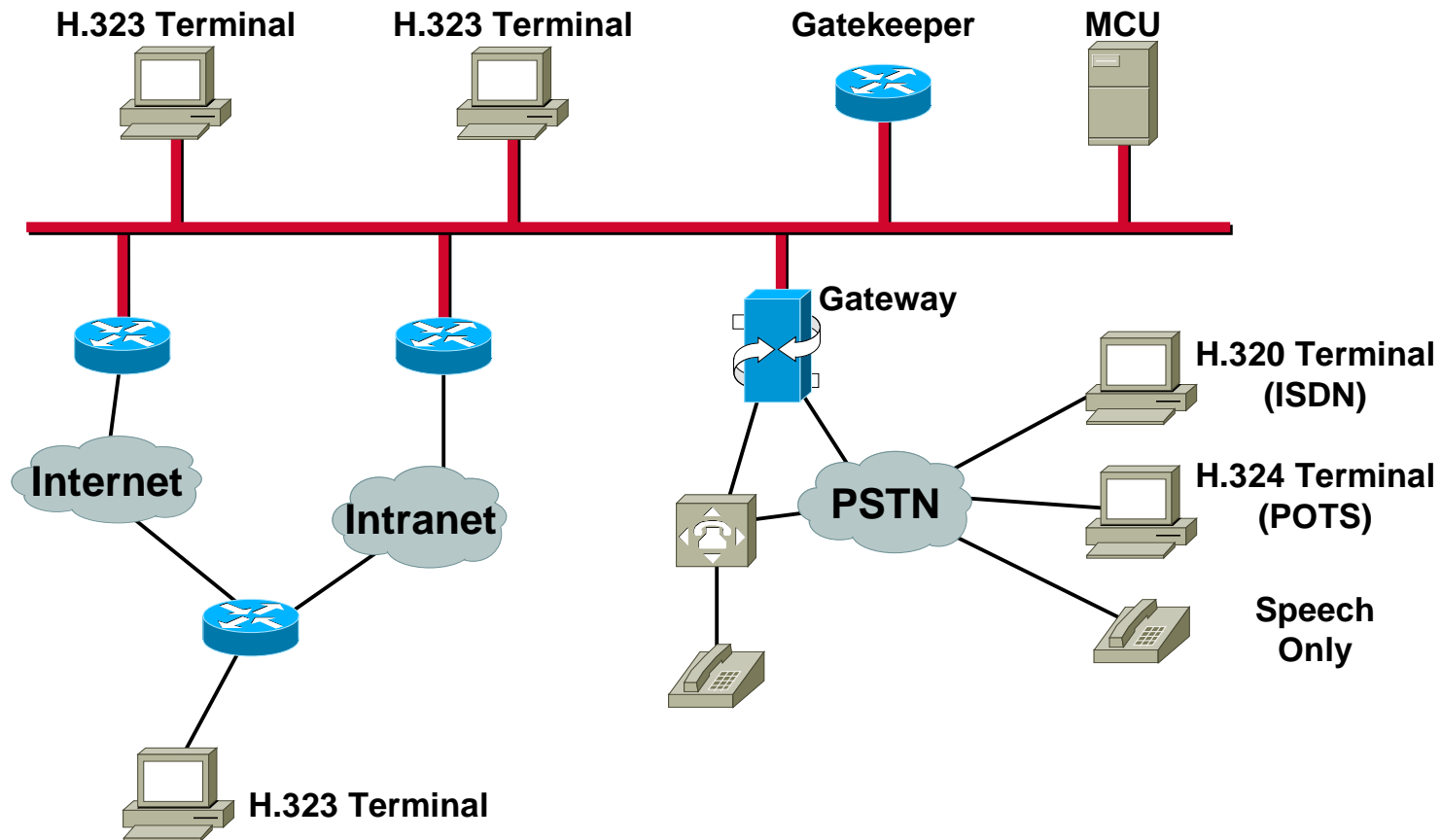


A man in a white shirt and red tie is holding a large red cable that loops around a globe. The globe is blue and green, representing Earth. The background is a textured, yellowish-brown surface.

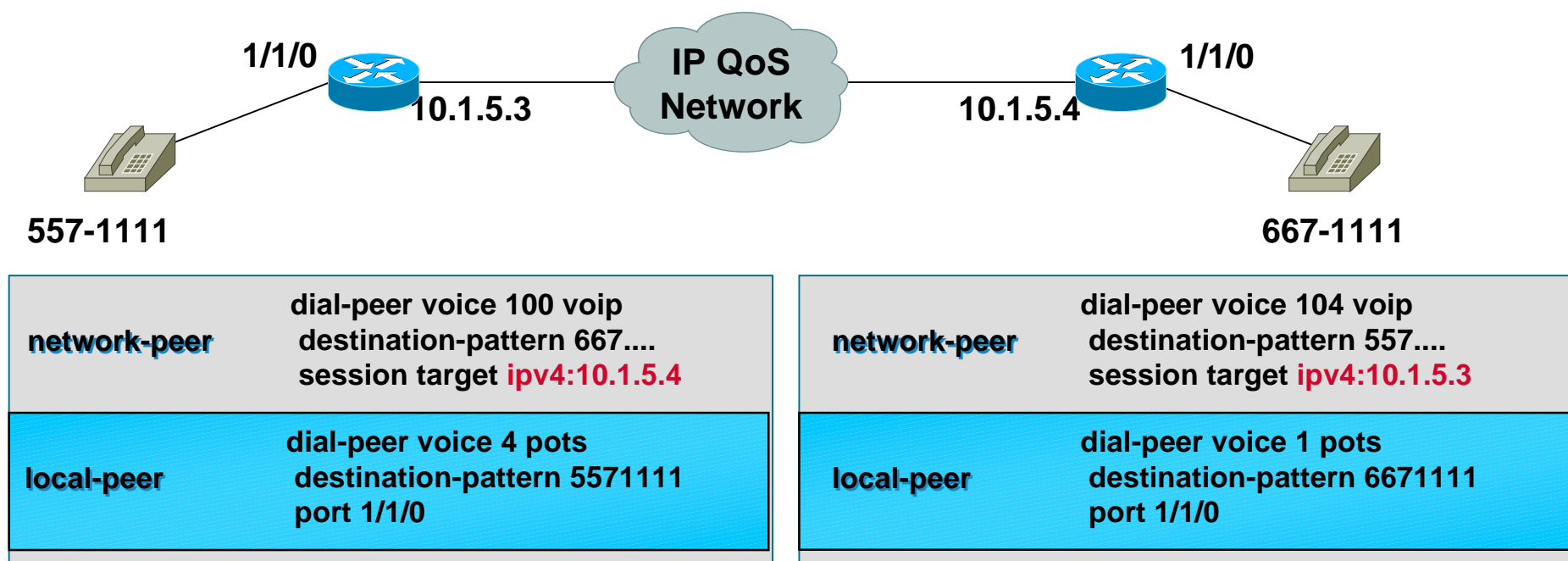
H.323 Primer



H.323 Network Overview

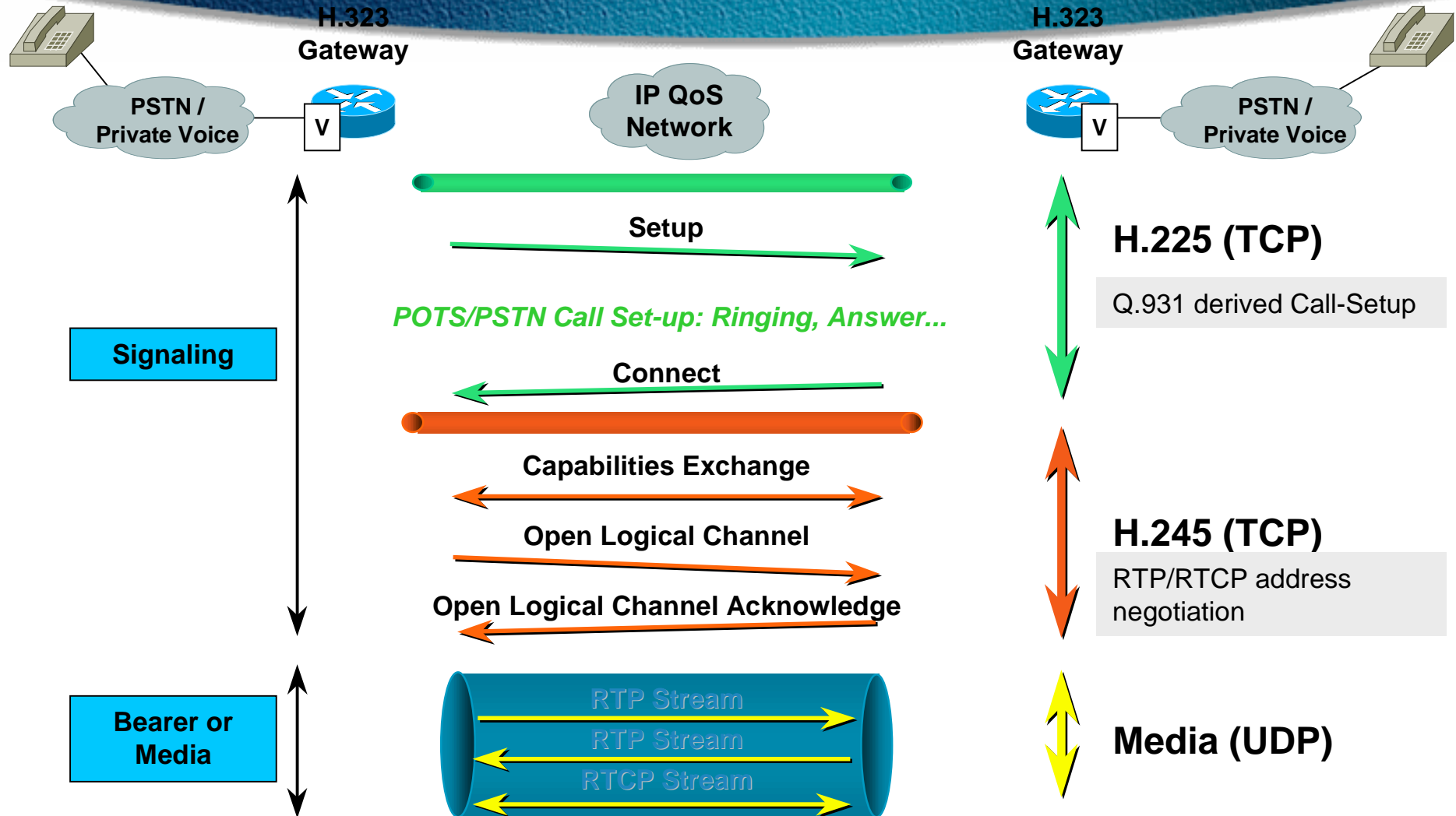


H.323 Without RAS - Configuration



- VoIP Dial Peer points directly to the destination GW's IP address
- Scaling to large networks becomes administratively burdensome

H.323 Without RAS - Call Set-up



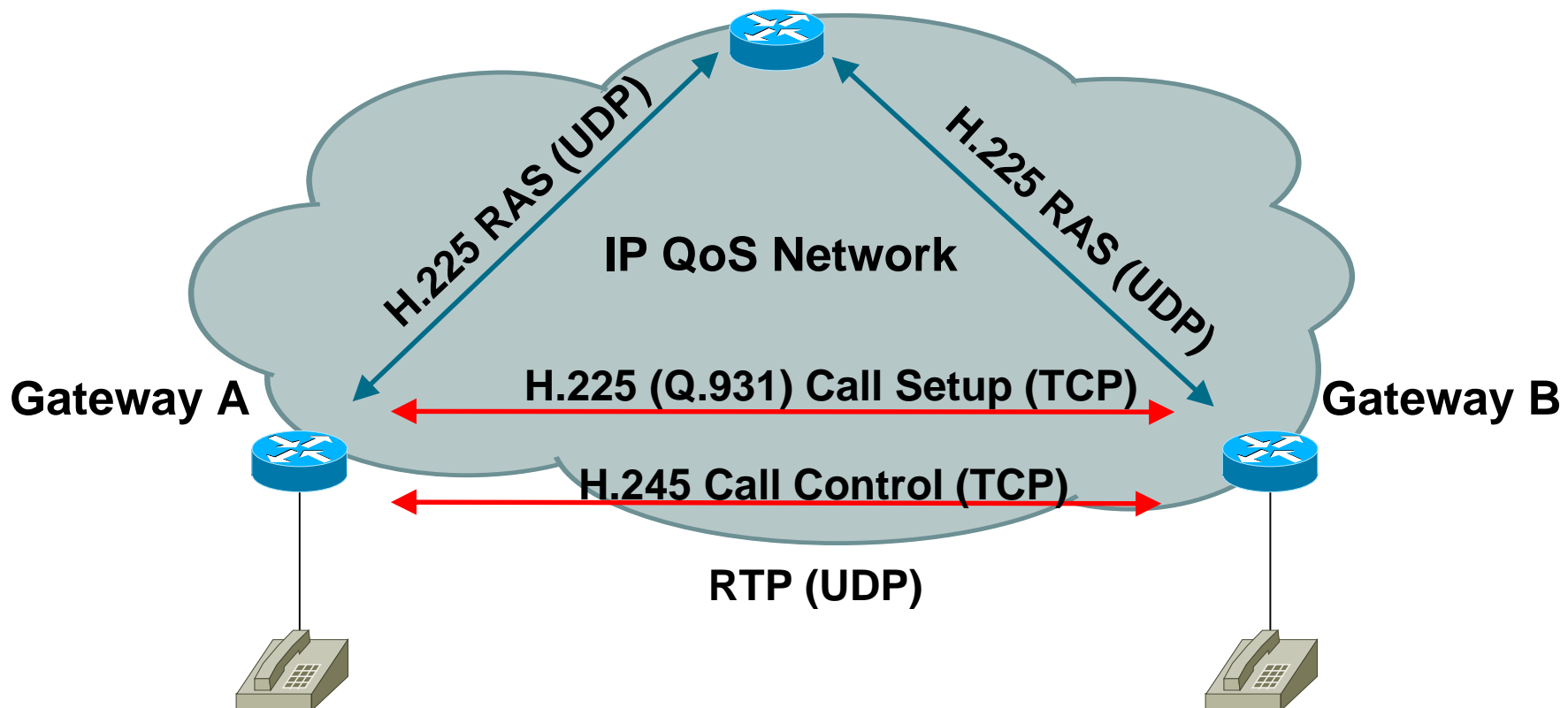
- Assumes Endpoints know each other's IP addresses
- The FastConnect v2 option cut down on messaging (see 12.1.1T section)

H.323 With RAS

Registration, Admission and Status

Gatekeeper

Address Translation: Every GW needs to know only about the GK, not about all other GWs



Gatekeeper Functionality

- **Gatekeeper is optional**
- **Logically separate from the H.323 endpoints**
- **H.323 ITU Specification**

Gatekeeper mandatory services are:

Address translation
Admissions control
Zone management

Gatekeeper optional services are:

Call control signaling
Call authorization
Bandwidth management and reservation

Gatekeeper Functionality

- **Cisco Gatekeeper Features**

Address Translation

H.323 ID and E.164 translation to an IP address

Admission Control

Terminal name registration

Gateway registration

E.164 Registration

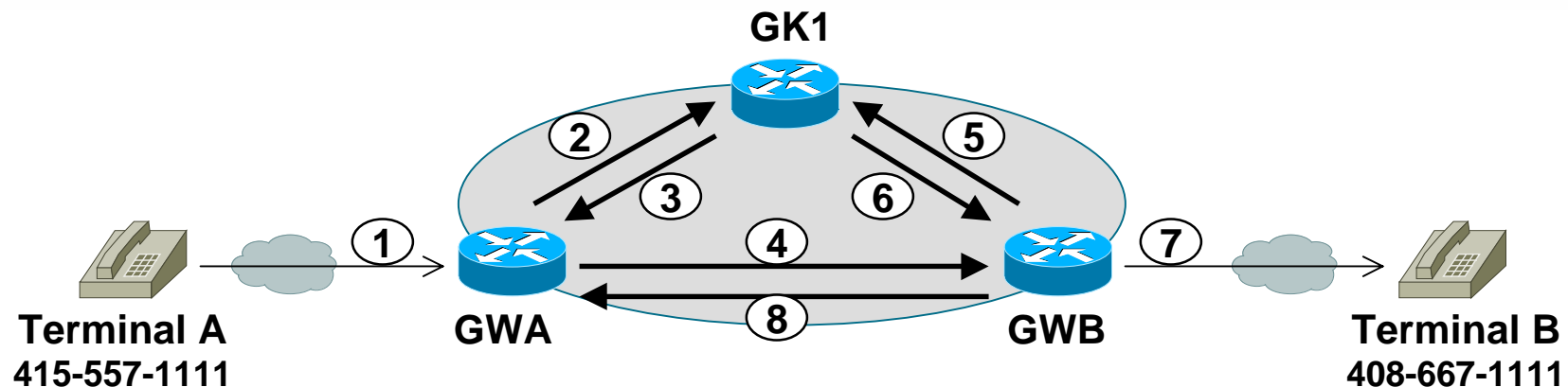
Zone Management

Zone and subnet configuration

Intra- and Inter-zone routing and communication (proxy assignment)

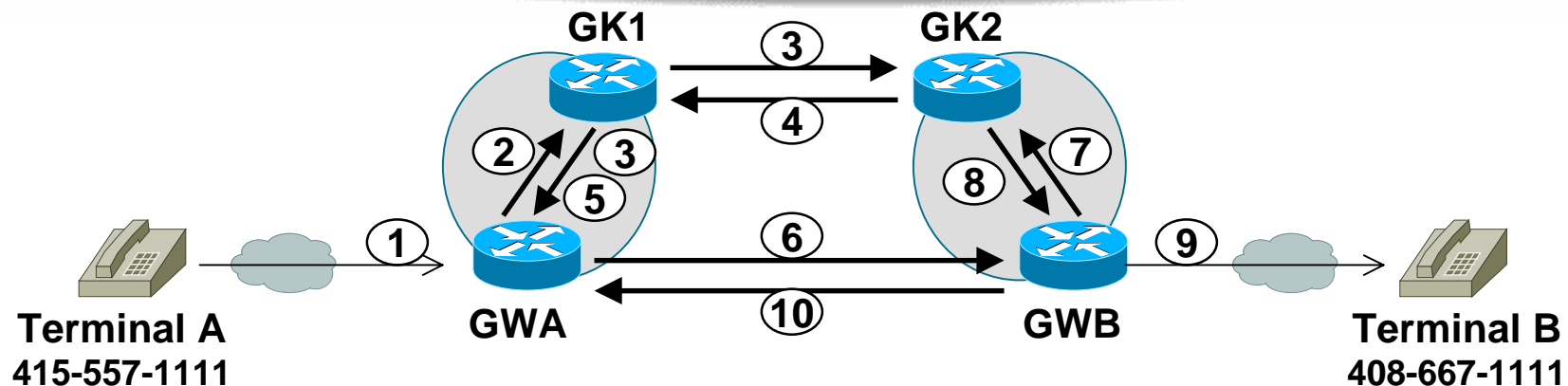
Zone access configuration

Intra-Zone Call Set-up



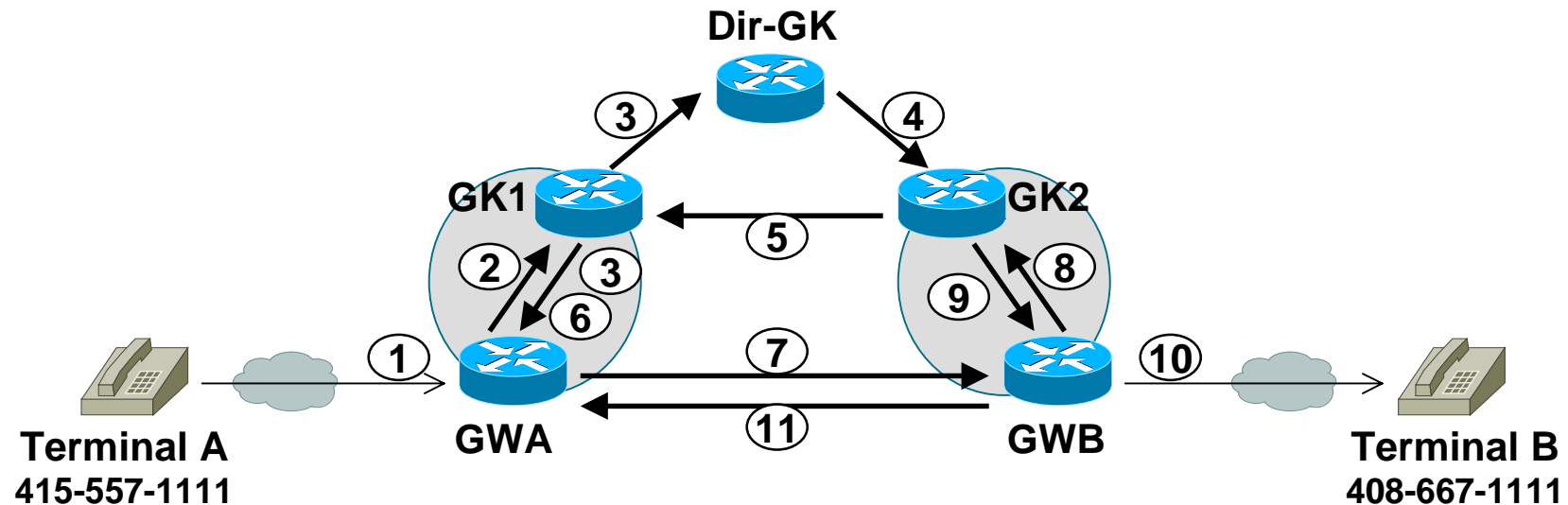
- 1) Terminal A **dials** the phone number 408-667-1111 for Terminal B
- 2) GWA sends GK1 an **ARQ**, asking permission to call Terminal B
- 3) GK1 does a look-up and finds Terminal B registered; returns an **ACF** with the IP address of GWB
- 4) GWA sends a **Q.931 Call-Setup** to GWB with Terminal B's phone number
- 5) GWB sends GK1 an **ARQ**, asking permission to answer GWA's call
- 6) GK1 returns an **ACF** with the IP address of GWA
- 7) GWB sets up a **POTS call** to Terminal B at 408-667-1111
- 8) When Terminal B answers, GWB sends **Q.931 Connect** to GWA

Inter-Zone Call Set-up



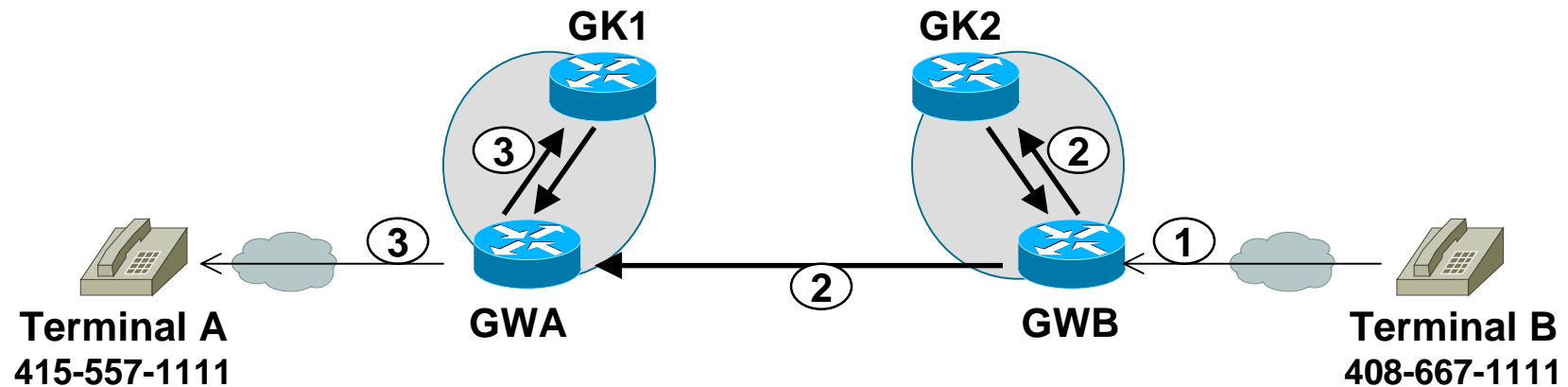
- 1) Terminal A **dials** the phone number 408-667-1111 for Terminal B
- 2) GWA sends GK1 an **ARQ**, asking permission to call Terminal B
- 3) GK1 does a look-up and does NOT find Terminal B registered; GK1 does a prefix look-up and finds a match with GK2; GK1 sends an **LRQ** to GK2, and **RIP** to GWA
- 4) GK2 does a look-up and finds Terminal B registered; returns an **LCF** with the IP address of GWB
- 5) GK1 returns an **ACF** with the IP address of GWB
- 6) GWA sends a **Q.931 Call-Setup** to GWB with Terminal B's phone number
- 7) GWB sends GK2 an **ARQ**, asking permission to answer GWA's call
- 8) GK2 returns an **ACF** with the IP address of GWA
- 9) GWB sets up a **POTS call** to Terminal B at 408-667-1111
- 10) When Terminal B answers, GWB sends **Q.931 Connect** to GWA

Inter-Zone Call Set-up: Directory Gatekeeper



- 1) Terminal A **dials** the phone number 408-667-1111 for Terminal B
- 2) GWA sends GK1 an **ARQ**, asking permission to call Terminal B
- 3) GK1 does a look-up and does NOT find Terminal B registered; GK1 does a prefix look-up and finds a wildcard match with Dir-GK; GK1 sends **LRQ** to Dir-GK, and **RIP** to GWA
- 4) Dir-GK does a prefix look-up and finds GK2; Forwards the **LRQ** to GK2
- 5-11) Same as steps 4-10 in previous scenario

Call Disconnect

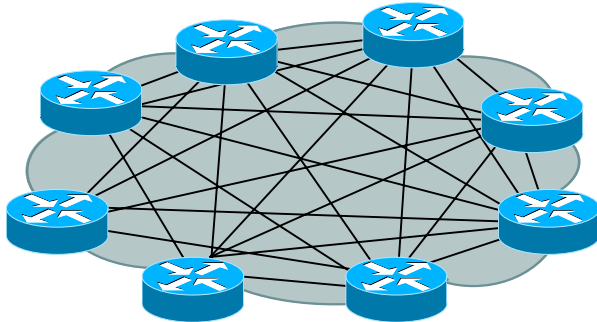


Terminals A and B are in active conversation...

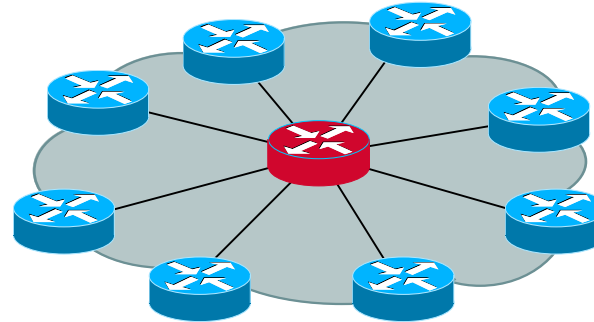
- 1) Terminal B **hangs up**
- 2) GWB sends **DRQ** to GK2, disconnecting the call between Terminals A and B.
A DCF is received some time later.
- 3) GWB sends a **Q.931 Release Complete** to GWA
- 4) GWA sends **DRQ** to GK1, disconnecting the call between Terminals A and B.
A DCF is received some time later.
- 5) GWA signals a **call disconnect** to the voice network (the mechanism differs depending on the trunk used on GWA. If it is a phone set (FXS), then there is no mechanism to signal the disconnect.

Directory Gatekeeper - Network Scaling

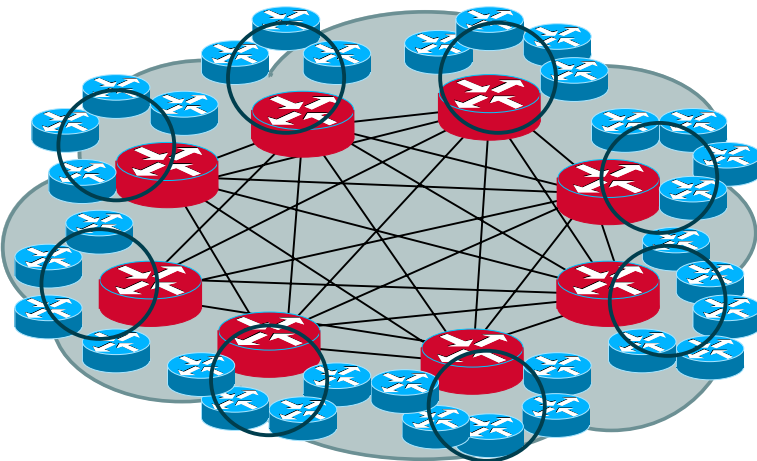
Small Network - Gateways only



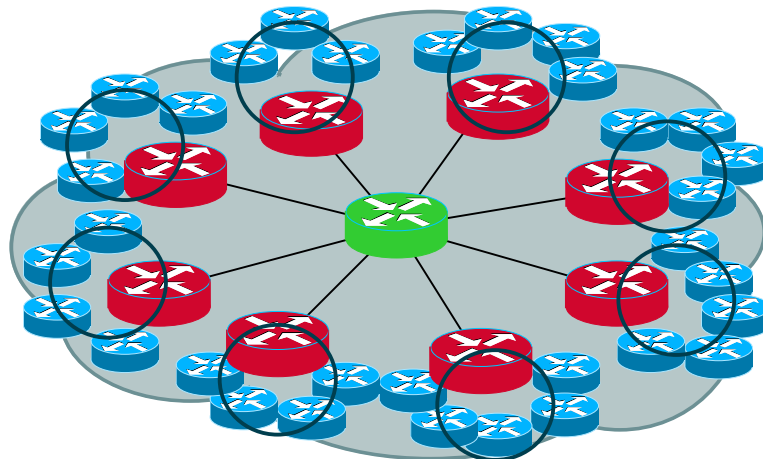
Small Network - simplified with a Gatekeeper



Medium Network - Multiple Gatekeepers



Medium-Large Network - Multiple Gatekeepers and a Directory Gatekeeper



Gateway



Gatekeeper



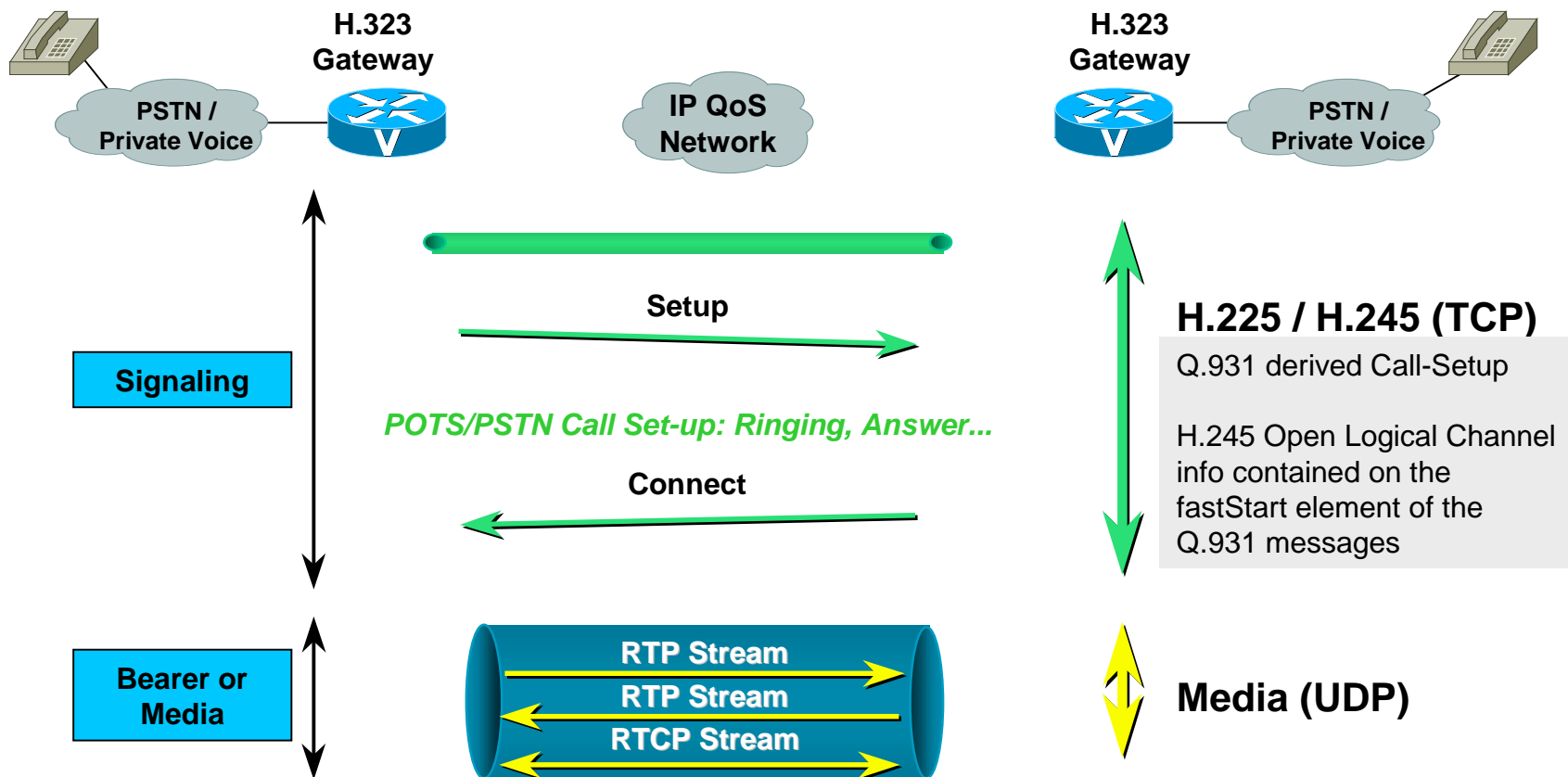
Directory Gatekeeper



cisco.com

FastConnect

12.1.1T



- Saves opening a separate H.245 TCP connection (sockets)
- Saves several message exchanges
- Default call-setup mechanism as of this SW release

FastConnect

12.1.1T



Only calling side can initiate fastConnect

Ready to Rx media

fastConnect Element with Open Logical Channel Info | Q.931 Setup

Called side can refuse by not returning fastConnect on the Q.931 message

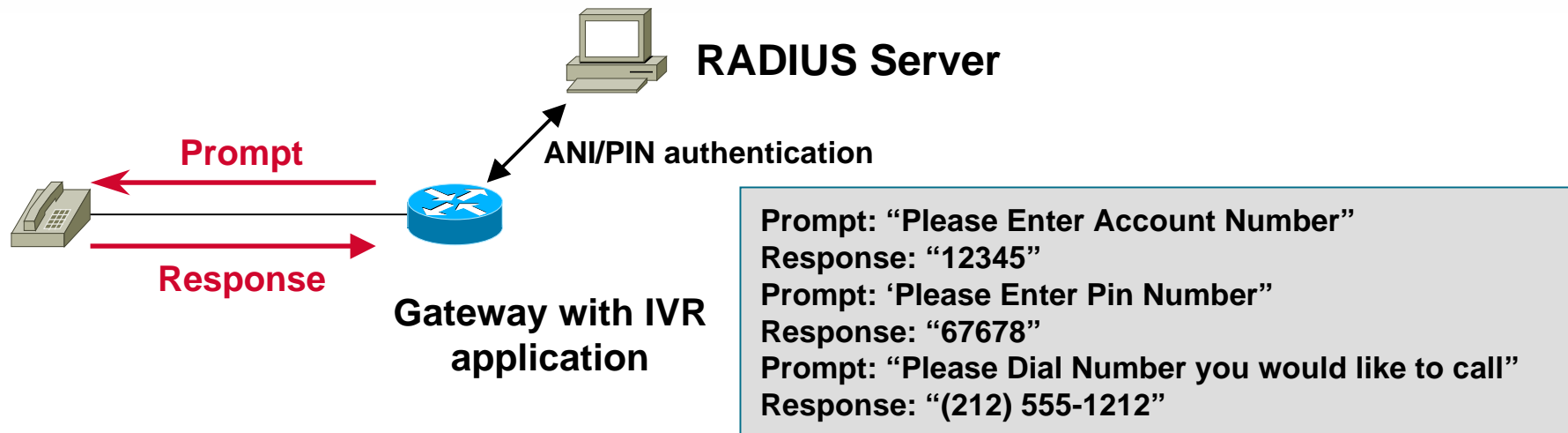
Any Q.931 message up to and including CONNECT

Q.931 Msg | fastConnect Element with Open Logical Channel Info

Ready to Rx media

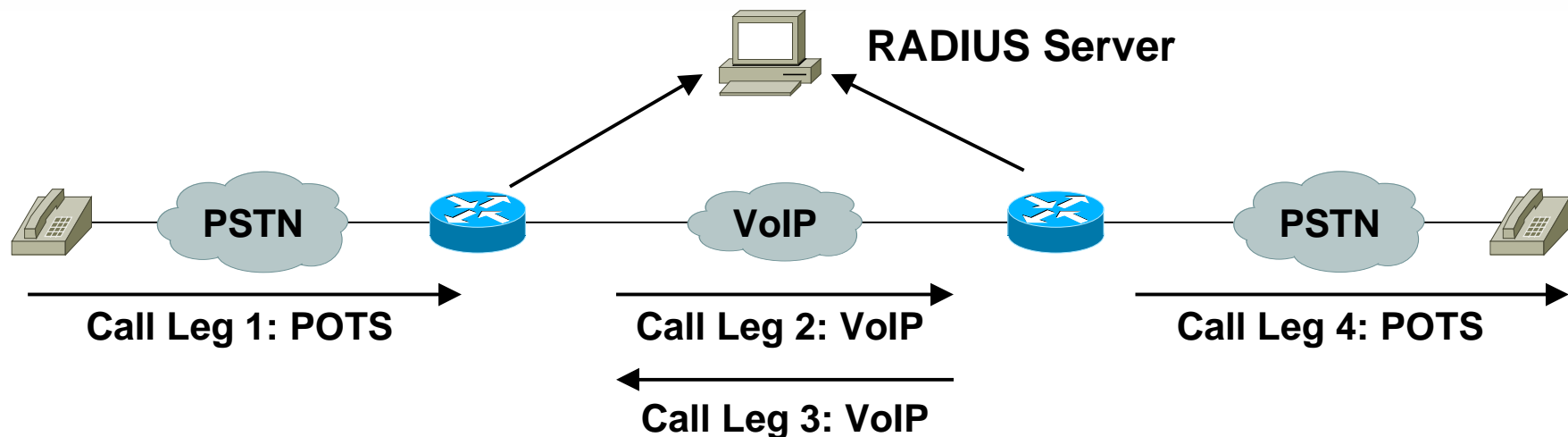
- Will work with any H.323 v2 endpoint that supports fastConnect

IVR Example



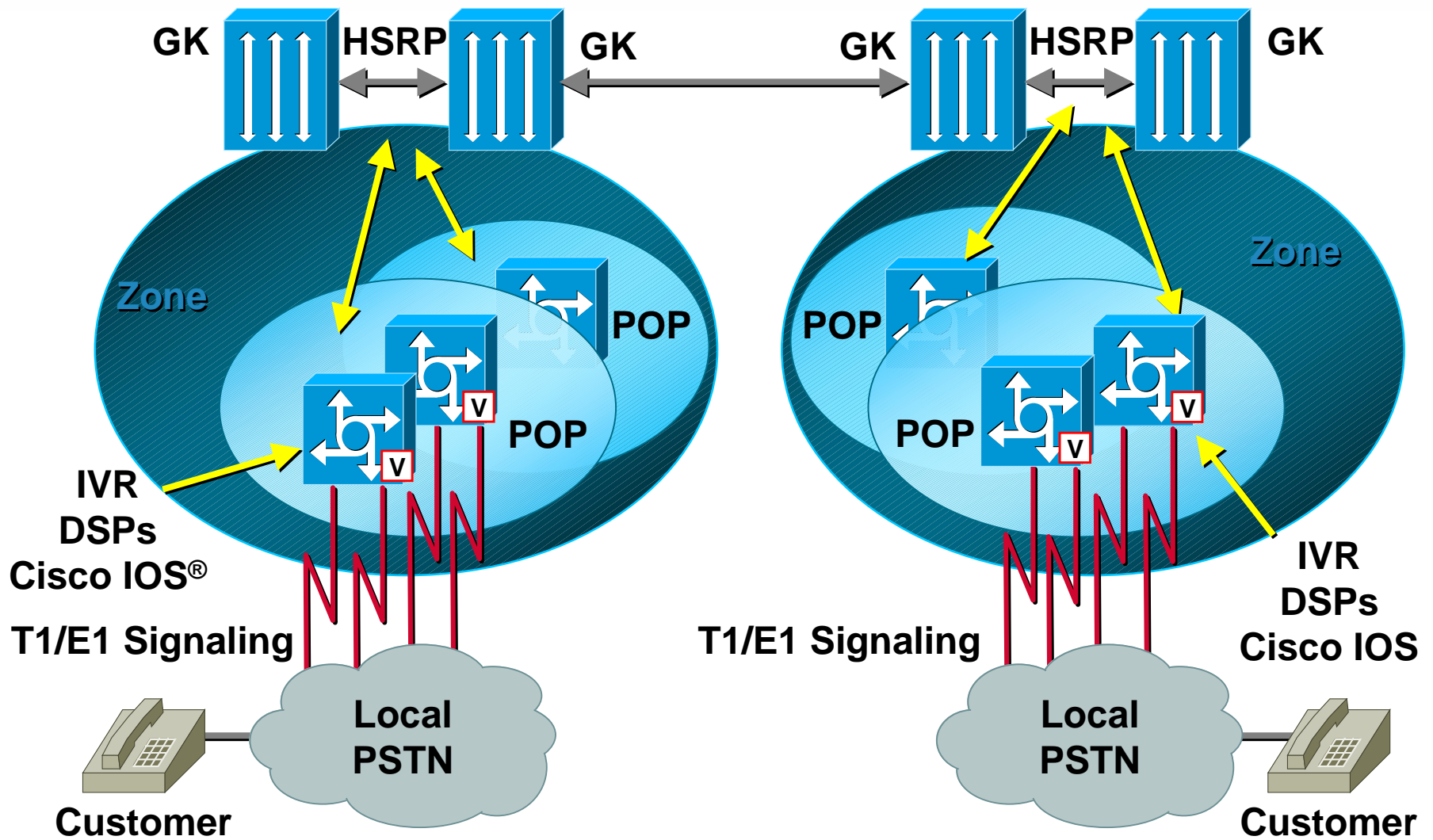
- Used for user authentication before entering the VoIP network
- IVR scripts are not modifiable in this release
the audio files for the prompts can be modified
- Script is a sequence of voice prompts and DTMF responses
 - Plays customized prompts
 - Collects account numbers, PINs and passwords
 - Collects the destination number
 - Perform AAA authentication
 - Place call
- IVR scripts can be invoke on voice ports, or called number

Billing

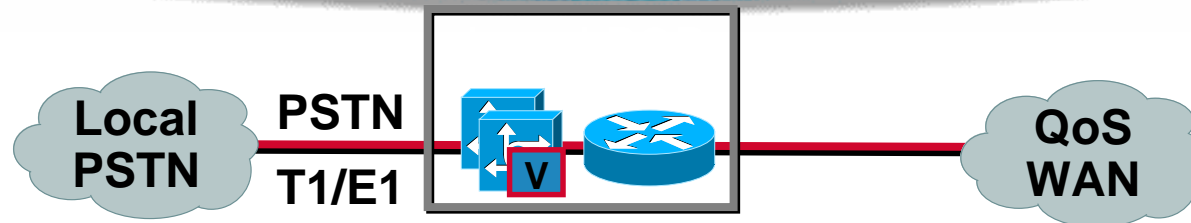


- Each of the Call Legs can generate Start and Stop records
- Each call leg reports the NTP time for when the **SETUP** was issued, the Call was **CONNECTED** and the **DISCONNECT** was received
- The Stop records have the required information for Billing
- IOS Accounting for Voice uses standard RADIUS attributes where possible

Anatomy of an H.323 Network



POP Sizing - Number Of GWs



Assumptions:

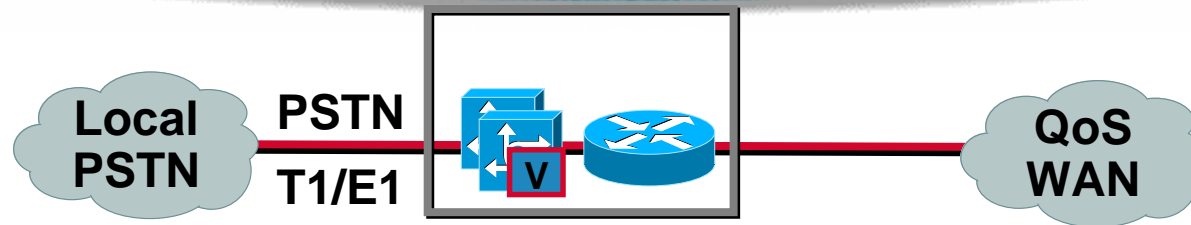
- POP must service X BHCA
- 3 Minute Hold Times (HT)
- AS5300 GWs can handle up to 2 calls per second
- 120* ports per AS5300

* Depends on T1/E1 and signaling type

Calculating Number Of GWs:

- Calls/DS0 per hour
= (60 min/hour) / (HT)
= 60 / 3
= **20 BHCA/DS0**
- BHCA capacity of AS5300
= (DS0/GW) * (BHCA/DS0)
= 120 * 20
= **2,400 BHCA/GW (0.67 calls/sec)**
- # AS5300 GWs Needed
= (POP BHCA) / (BHCA / GW)
= **X / 2400 AS5300 Needed**

POP Sizing - WAN Bandwidth



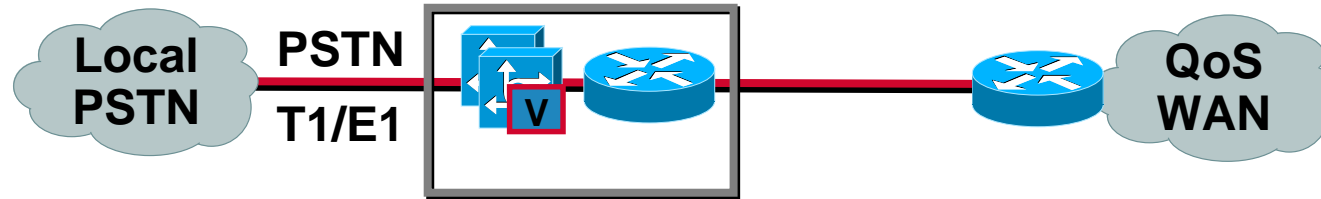
Assumptions:

- Voice calls
- 120* Ports / AS5300
- * Depends on T1/E1 and signaling type
- Voice Activity Detection 30% efficiency on bandwidth for standard voice calls at T1/E1 levels and up.
- G.729 CODEC used
- 60 byte packets + link layer (no header compression)

WAN Bandwidth/GW:

- Bandwidth per call
 $66 \text{ bytes} * 8 \text{ bits/byte} * 50 \text{ pps}$
 $= 26.4 \text{ kbps}$
- Total WAN bandwidth/GW
 $26.4 \text{ kbps} * 120 (96^*) \text{ calls/GW}$
 $= 3.17 \text{ Mbps} (2.53 \text{ Mbps}^*)$
- Assume 30% VAD Efficiency
 $2.53 \text{ Mbps} * 70\%$
 $= 2.22 \text{ Mbps} (1.77 \text{ Mbps}^*)$

POP WAN Sizing With CRTP



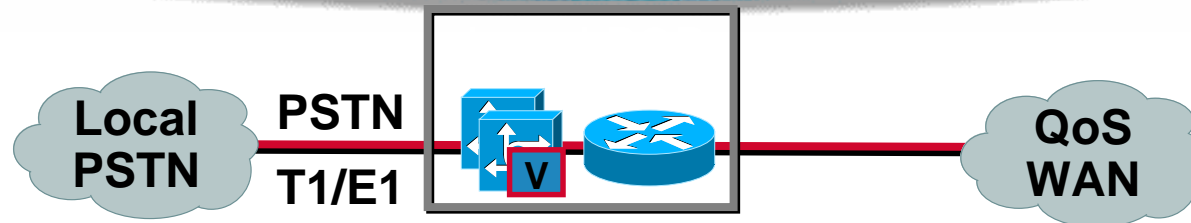
WAN Bandwidth:

- Bandwidth per call:
= **12 kbps**
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 $12 \text{ kbps} * 120 (96^*) \text{ calls}$
= **1.44 Mbps (1.15 Mbps*)**
- If Max = 160 (6.6 T1 or 5.3 E1)
 $12 \text{ kbps} * 160 \text{ calls}$
= **1.92 Mbps**
- Assume 30% VAD Efficiency
= **1.34 Mbps** < 1 T1/E1

CRTP:

- Performed on WAN backhaul router (e.g. 3660/7200)
- Reduces IP header from 40 bytes to 2-4 bytes much of the time
- Hop-By-Hop
- Fast/CEF switched cRTP
 - 12.0(7)T and 12.0(7)XK
 - Limited platforms / interfaces*
 - Limited # of cRTP sessions*
- http://wwwin.cisco.com/servpro/msa/products/docs/crt_web/crtp_web.html

POP Sizing - Number Of GWs



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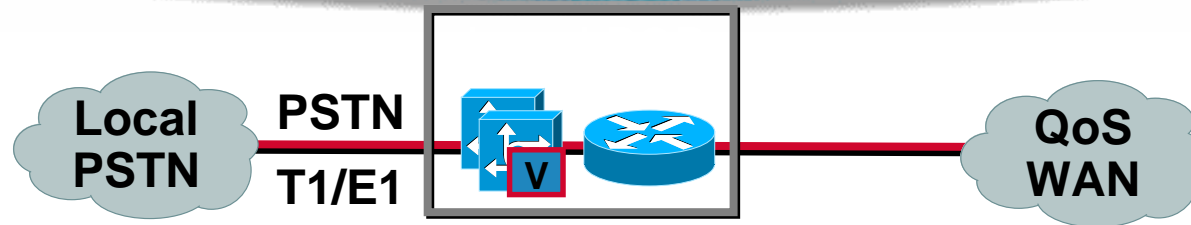
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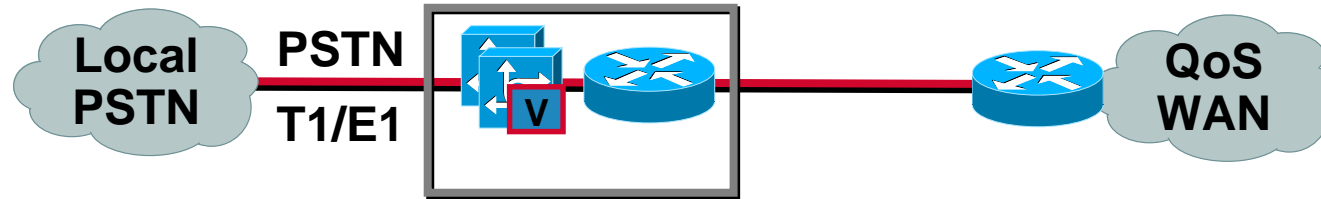
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POP WAN Sizing With CRTP



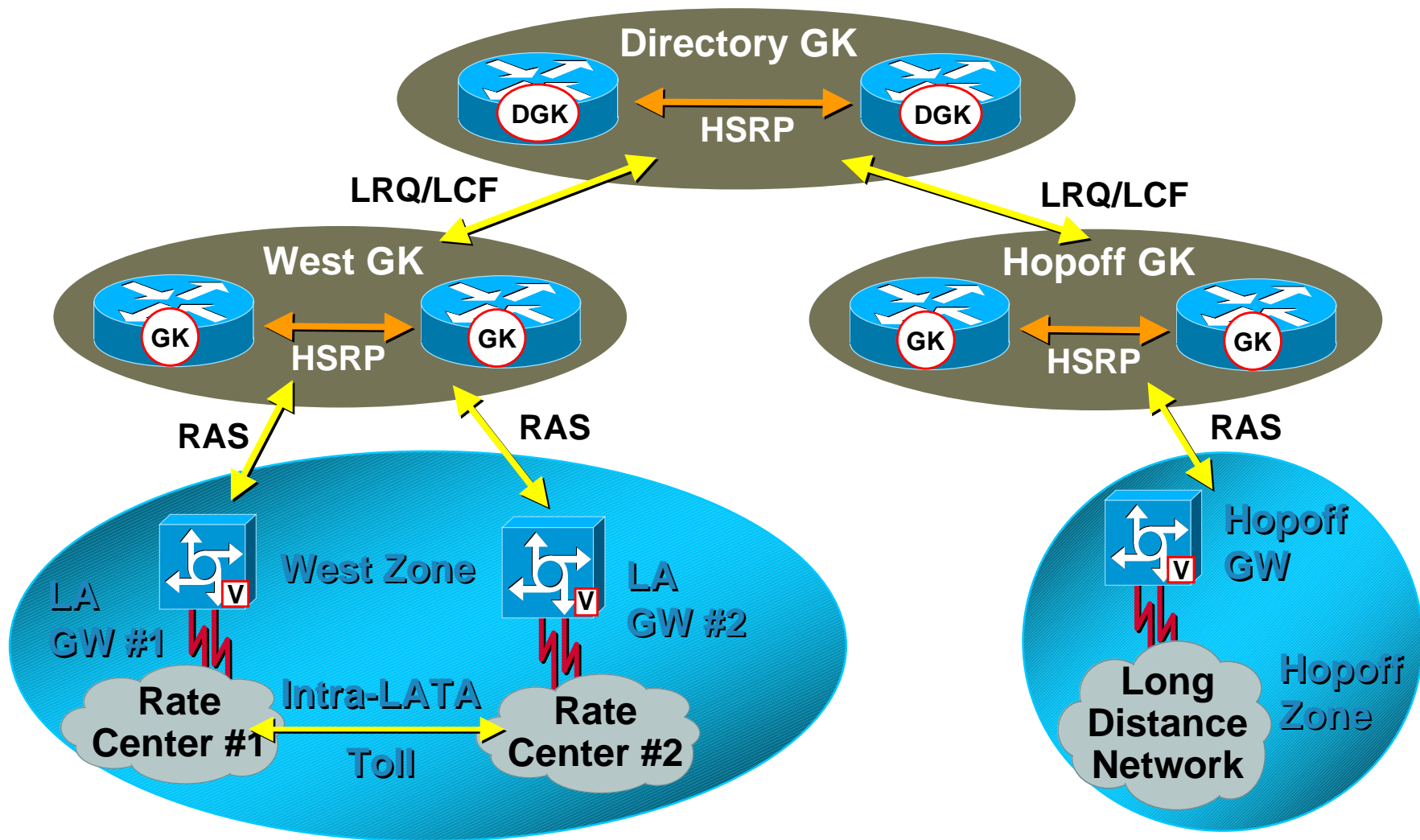
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Detailed Network Diagram



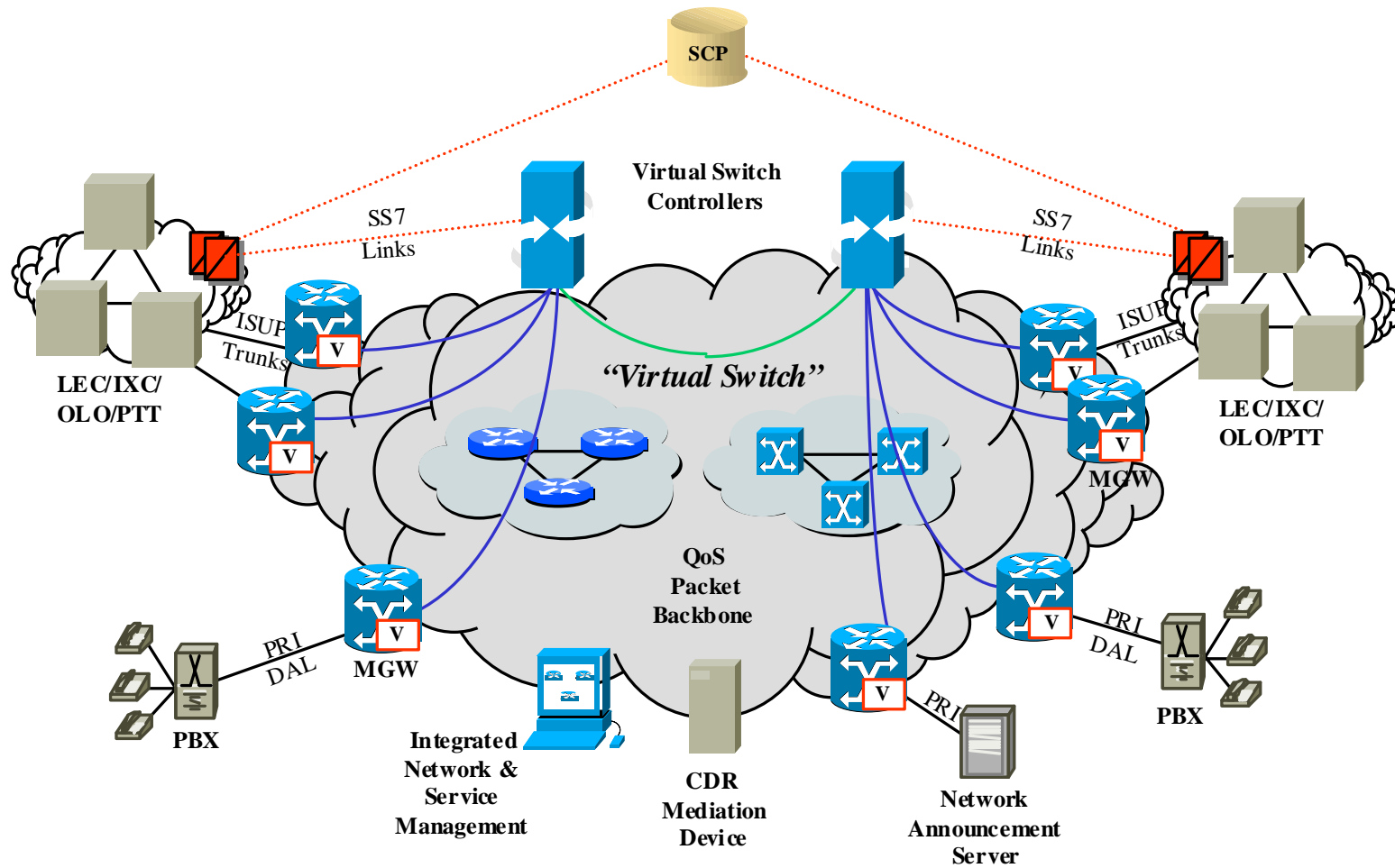


Packet Tandem Solution Overview

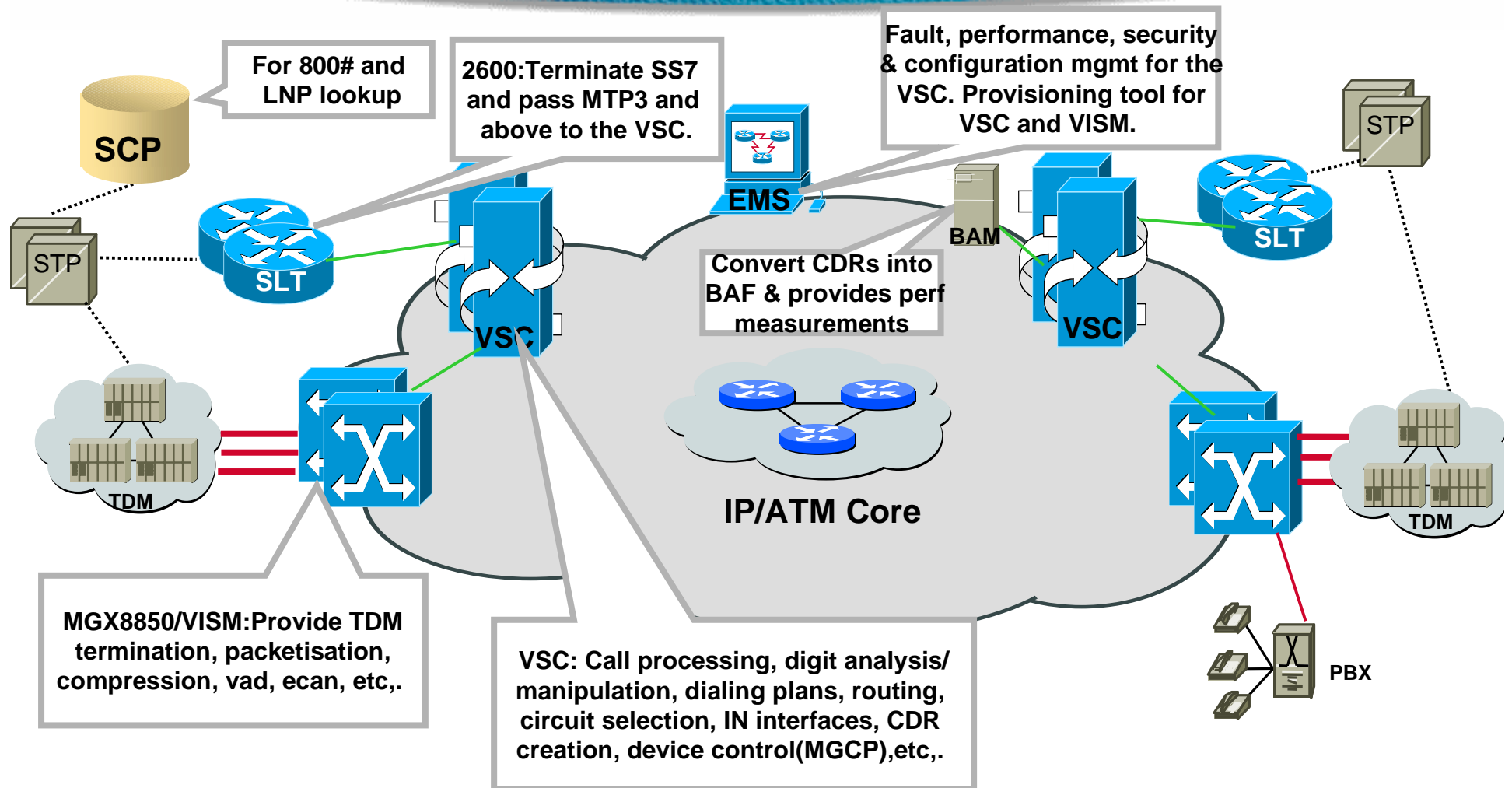
Drivers for packet tandem solution

- **Data traffic growing at exponential rate**
SPs have to optimize network for data
- **Deregulation and competition**
Need to focus on costs and services
VoIP non-regulation - bypass international settlement
Long-distance regulatory relief
Broadband access technologies
xDSL, cable and wireless (not TDM-based)
Core bandwidth explosion with WDM
- **Packet based end-systems and PBXs**
Multiservice access with data focus

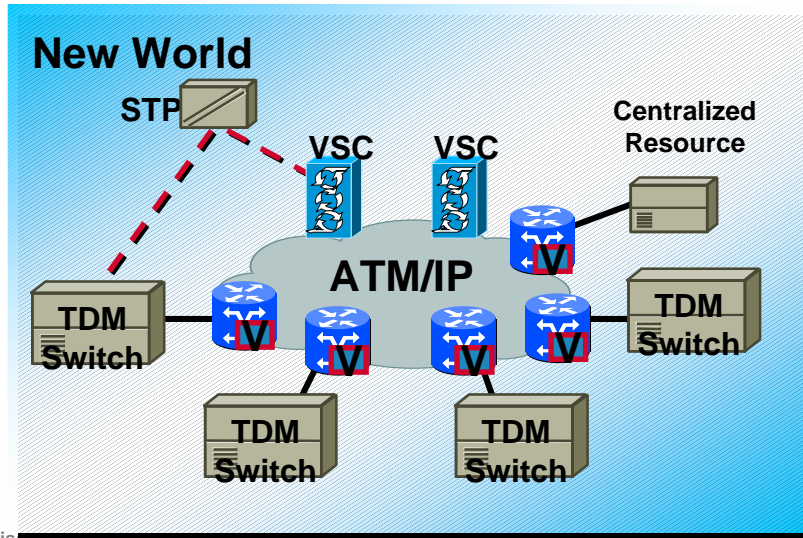
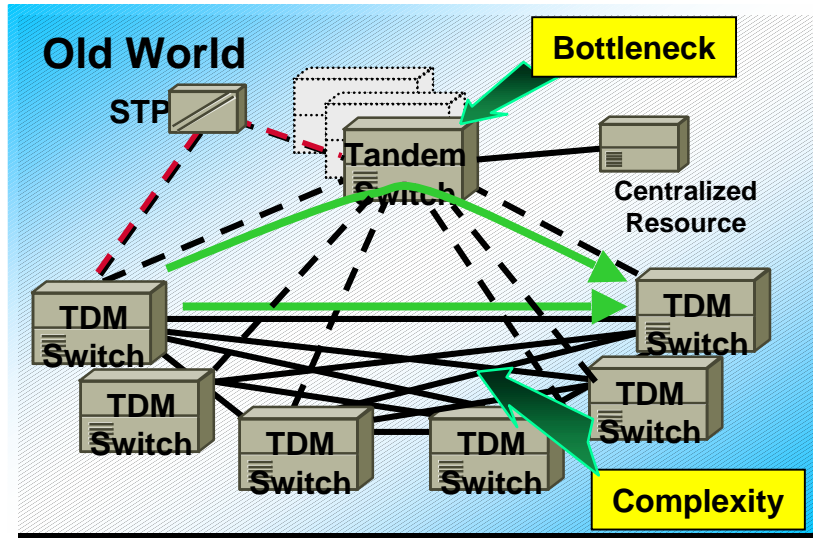
Packet Tandem solution Overview



Tandem/Transit Solution Components



Class 4 Tandem aka SuperTrunk



Service Description

- Replacement for existing voice (narrowband) Trunk and Tandem Network with a packet network

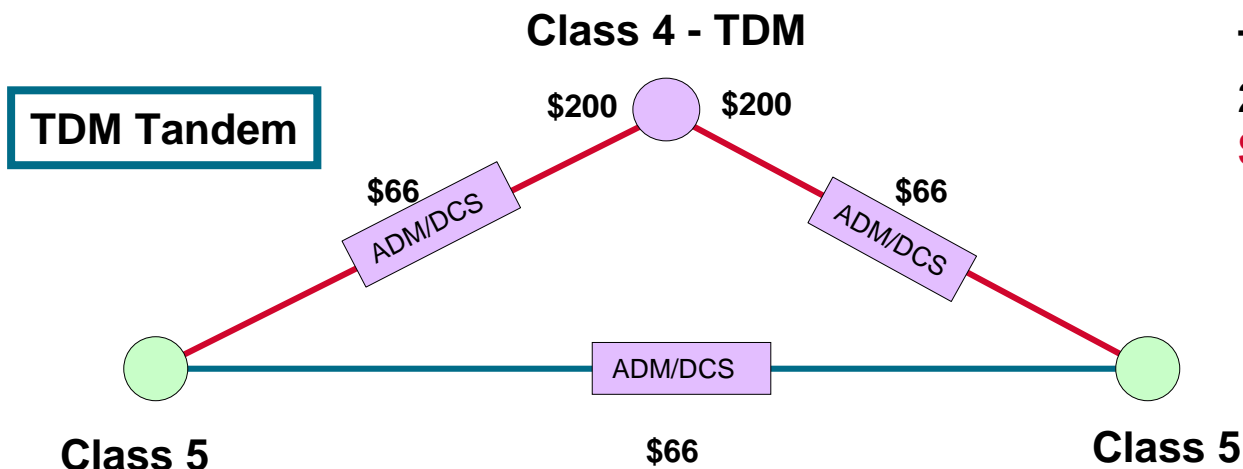
Value Proposition

- Reduces # of Tandems by providing a single large ATM Tandem Switch
- Reduces # of Trunks required through more efficient trunk utilization
- Lowers Operational Costs - Reduction in trunking operations for Translations, Forecasting, & Engineering
- Eliminates most circuit design for trunking & private lines

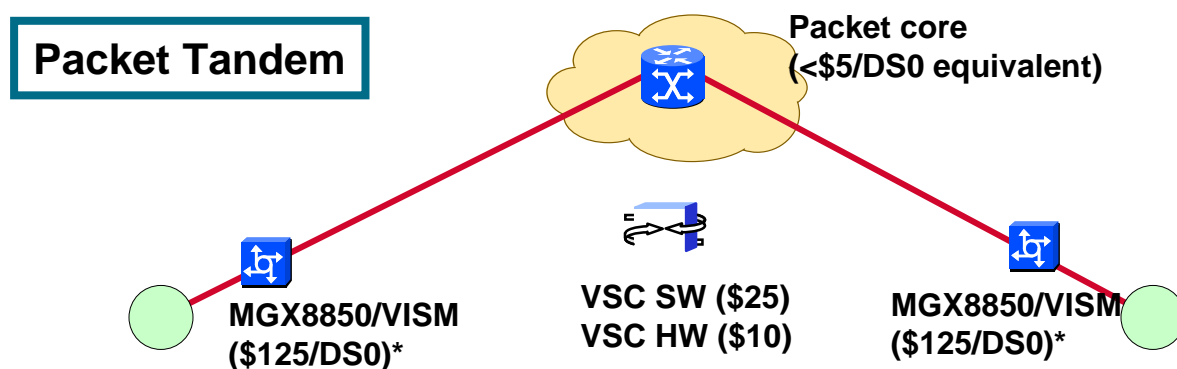
Solution Components

Packet Tandem Solution

Biz Case - Case Study



Tandem Path (TDM):
 $2x(\$200 + \$66) =$
\$532/DS0



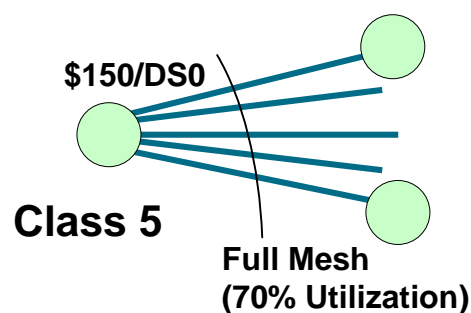
Tandem Path (Packet):
 $2x(\$125 + \$25 + \$10 + \$5) =$
\$330/DS0
Savings: 38%

*MGX8850/VISM: Redundant system, assuming 40% Discount, G.711 codec pricing
 VSC SW Pricing based on volume discount.
 VSC HW Pricing based on fully-redundant fault-tolerant Active & Standby system

Packet Tandem Solution

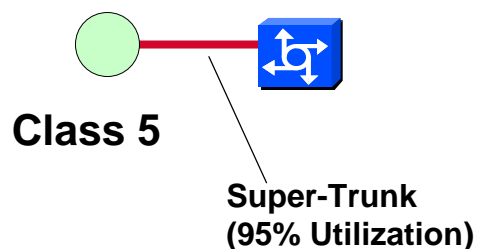
Biz Case - Case Study

Existing



Assuming 25K Trunk Ports / Class 5 Switch
Trunk port cost / DS0: \$150
Trunks in-use (70% Util): 17500

Packet Tandem



Trunks in-use (95% Util): 23750
Trunk ports saved / Class5: 6250
@ \$150/DS0 = ~ \$1M Savings / Class 5 Switch

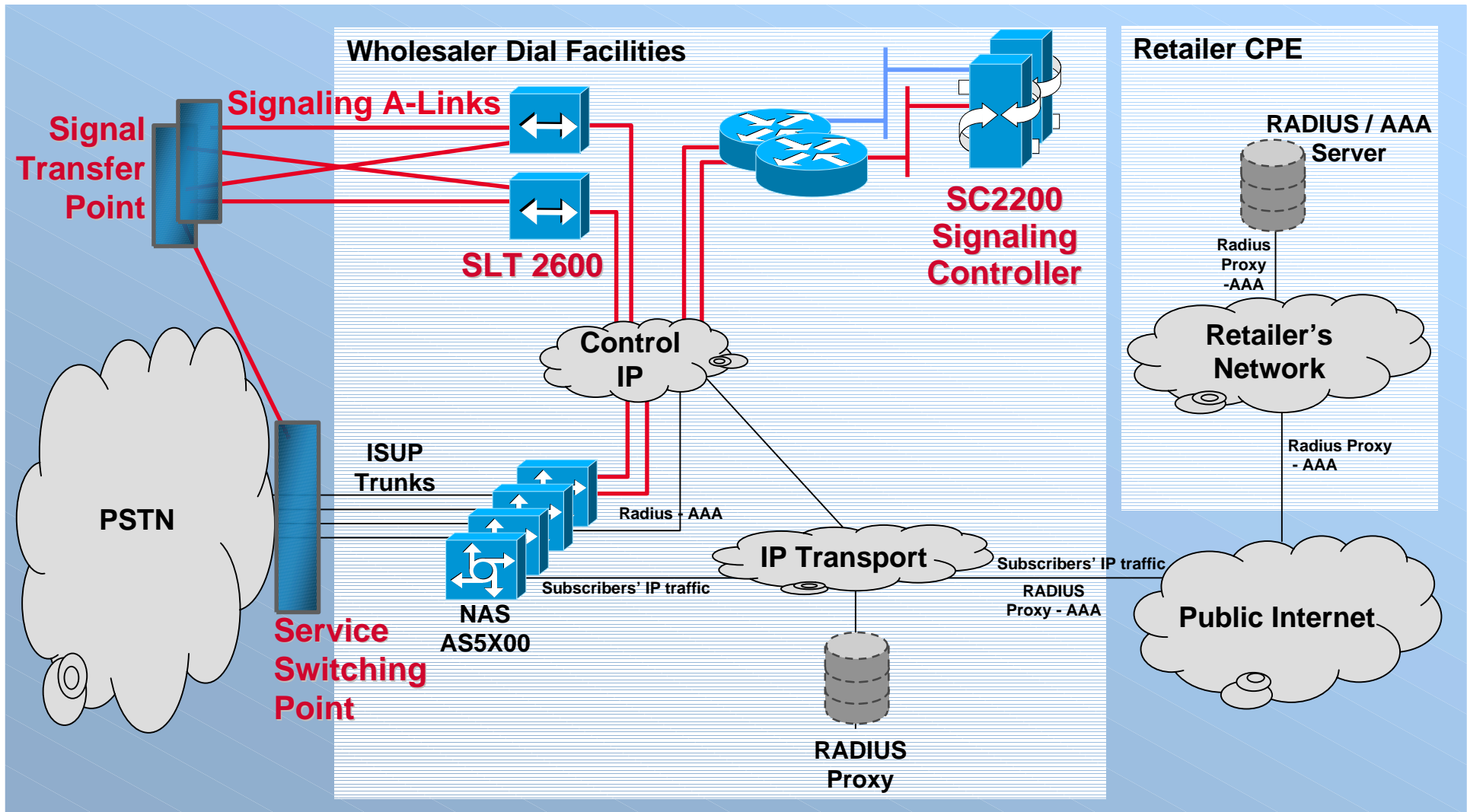
Plus Operational savings (trunk administration costs) makes a very compelling case for packet tandem solution



Wholesale Dial and Internet Offload

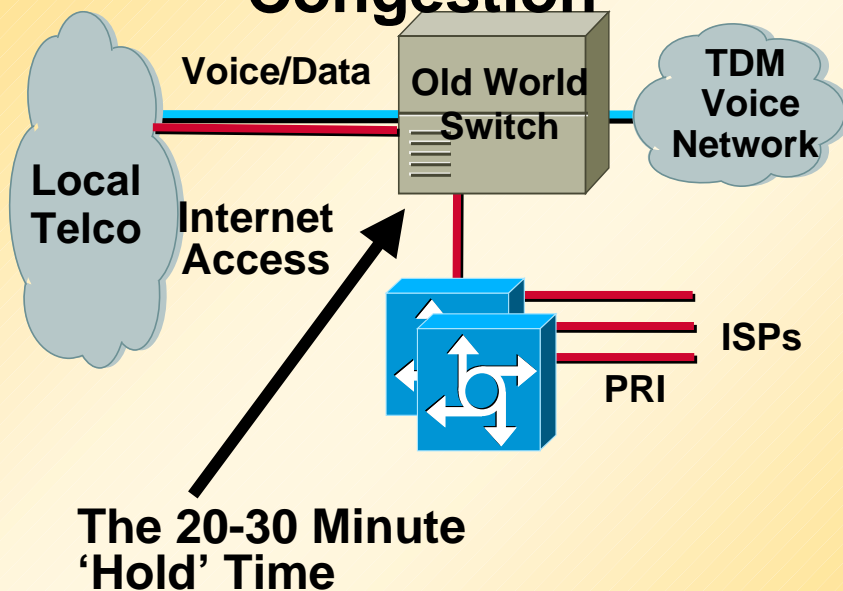


Wholesale Dial w/SS7

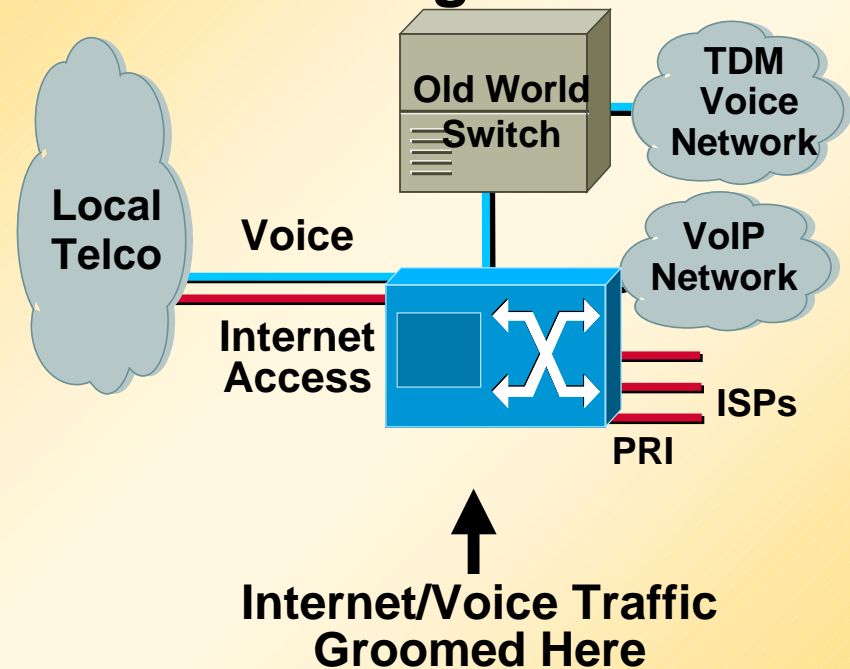


Internet Offload “Bulldog”

The Bottleneck - Congestion



Congestion Avoidance Traffic Grooming



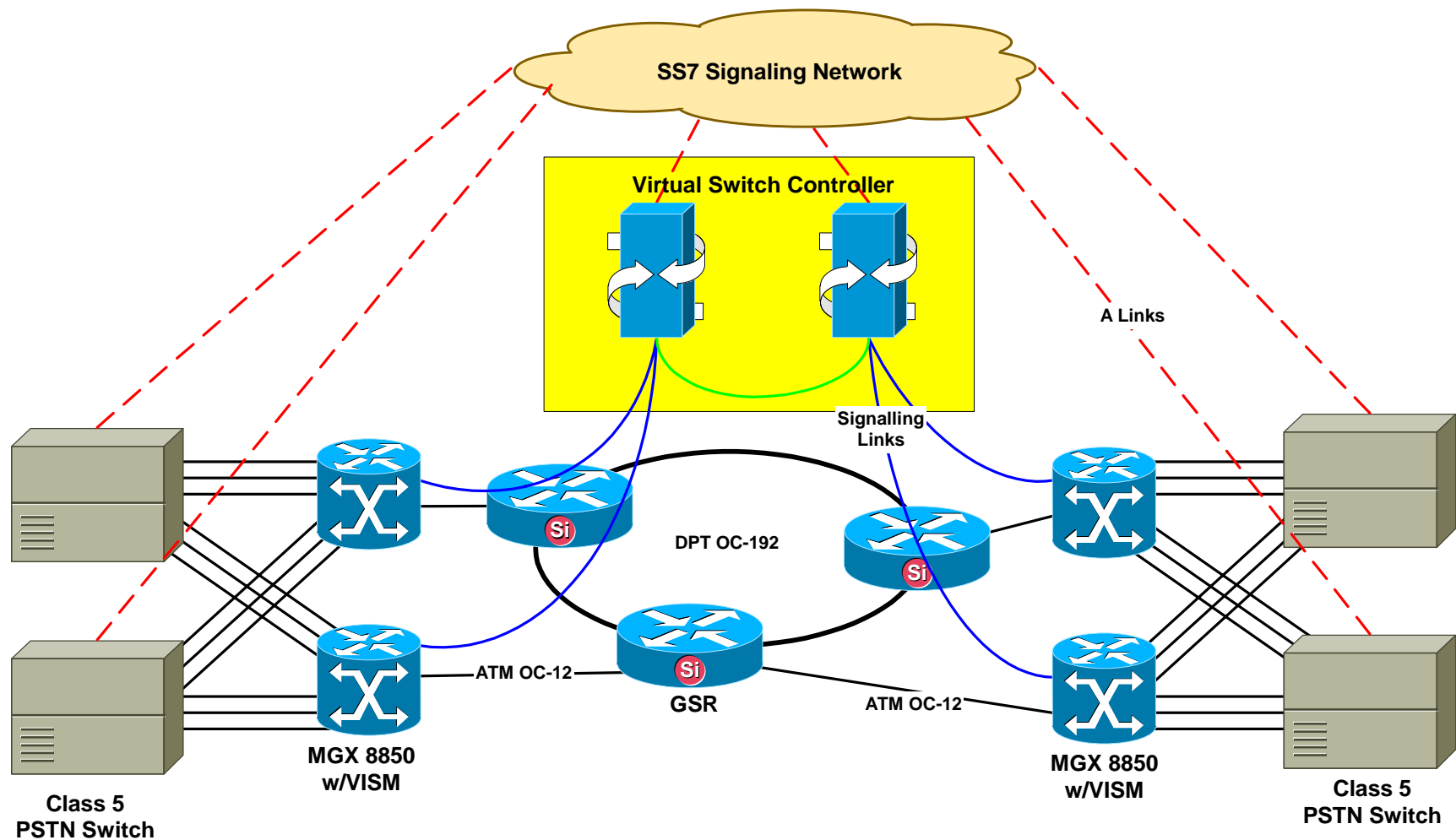


Voice Trunking over IP ILEC Design

Gary McNiel



VTolP





Debit/Prepaid Calling

Service Architectures

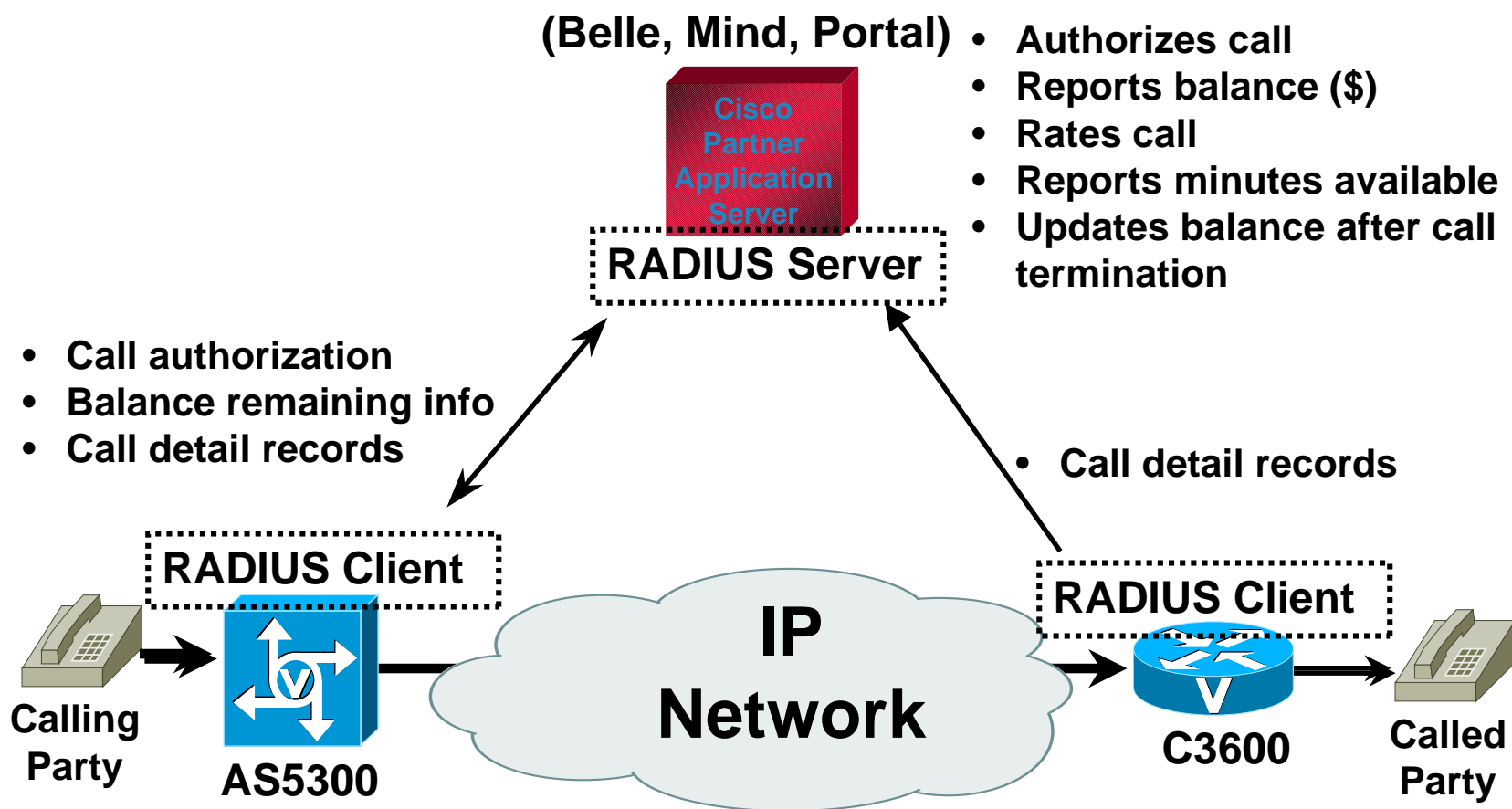
February 1, 2000

Internal Documentation



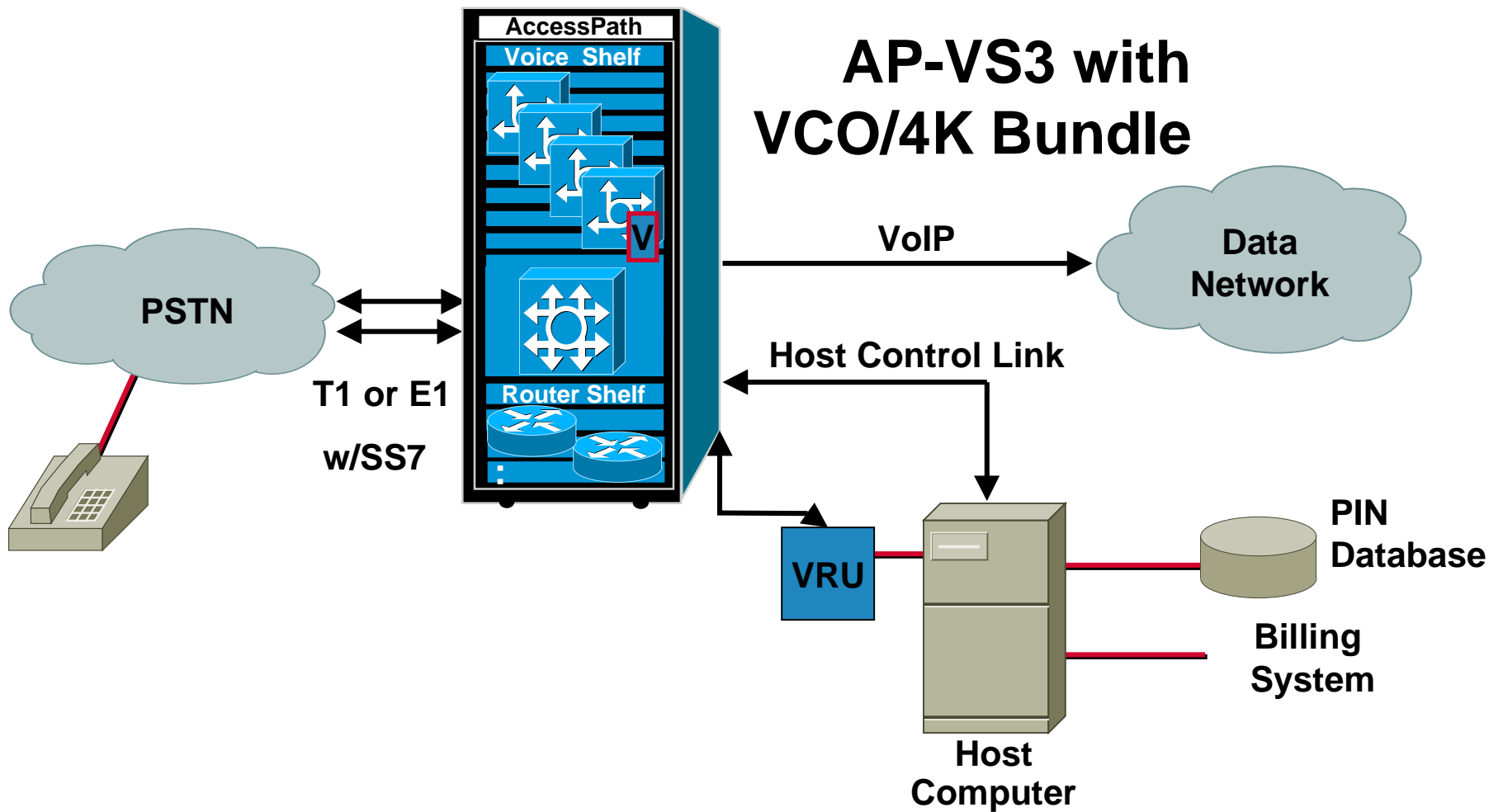
Debit/Prepaid Calling

Distributed Radius Service Architecture



Debit/Prepaid Calling

Service Node Architecture



A man in a white shirt and red tie is holding a large red cable that loops around a globe. The globe is blue and green, representing Earth. The background is a textured yellow and blue. The title "Product Updates" is written in large white letters across the center of the image.

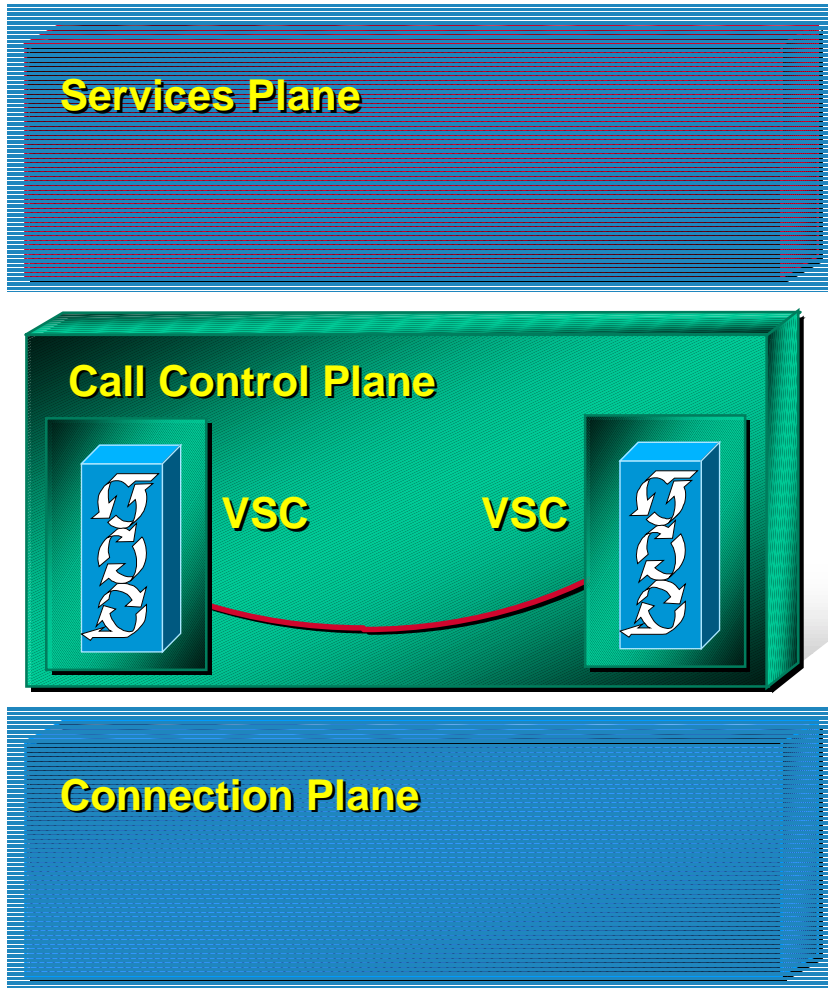
Product Updates





VSC3000

Call Control Plane— Virtual Switch Controller Concept



Call control

⇒ ***Call Routing***

⇒ ***Access to Services plane***

⇒ ***Gateway Control***

Signaling Mediation

Operation Support

⇒ ***Accounting***

Packet Telephony Signaling

Inter VSC Exchange

E-ISUP

- Signaling between VSC nodes
- Transparently pass TDM signaling between adjacent switches

Signaling Backhaul

Packet transport to deliver legacy SS7 and ISDN PRI signaling payload from gateway to VSC

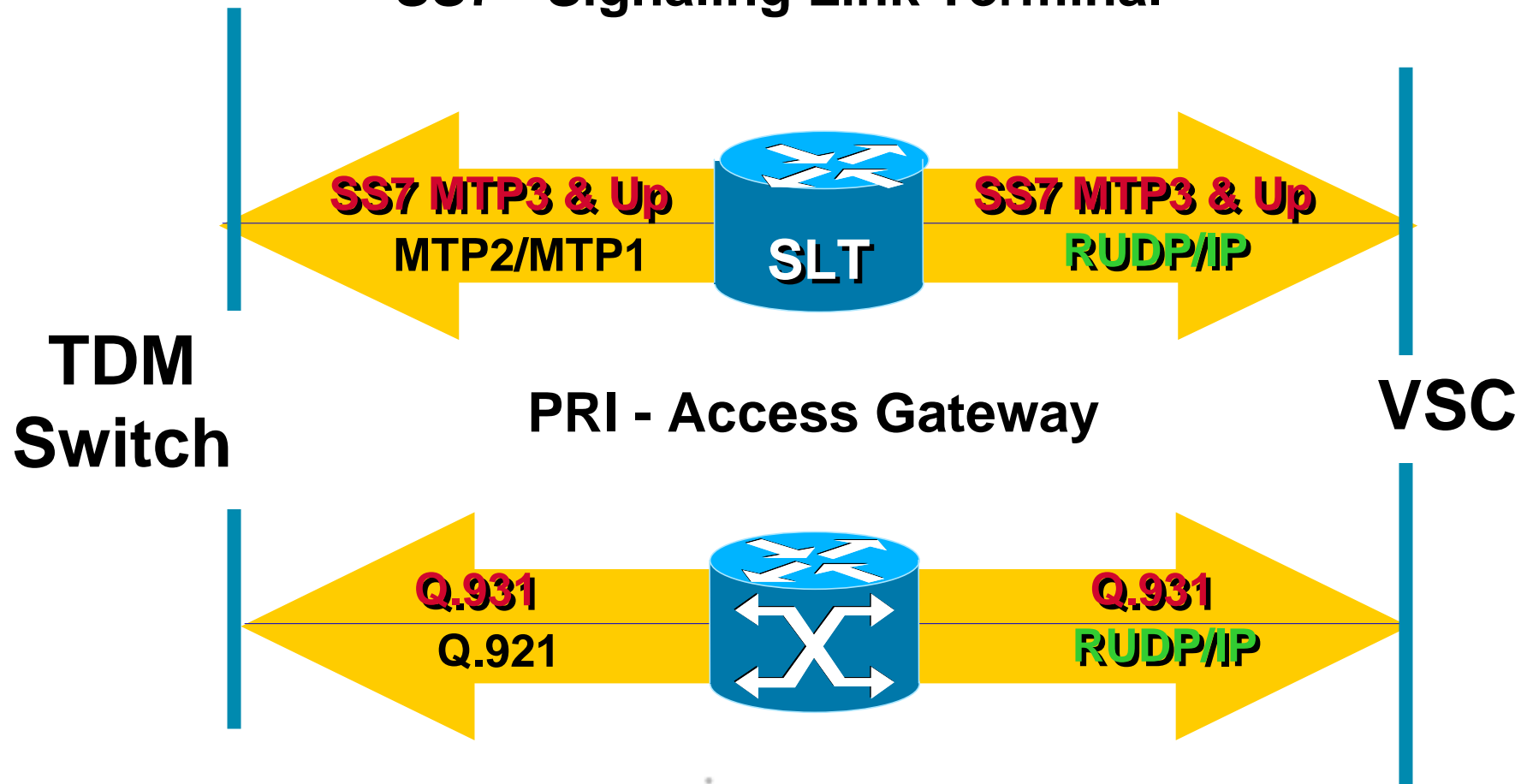
Gateway Control

MGCP

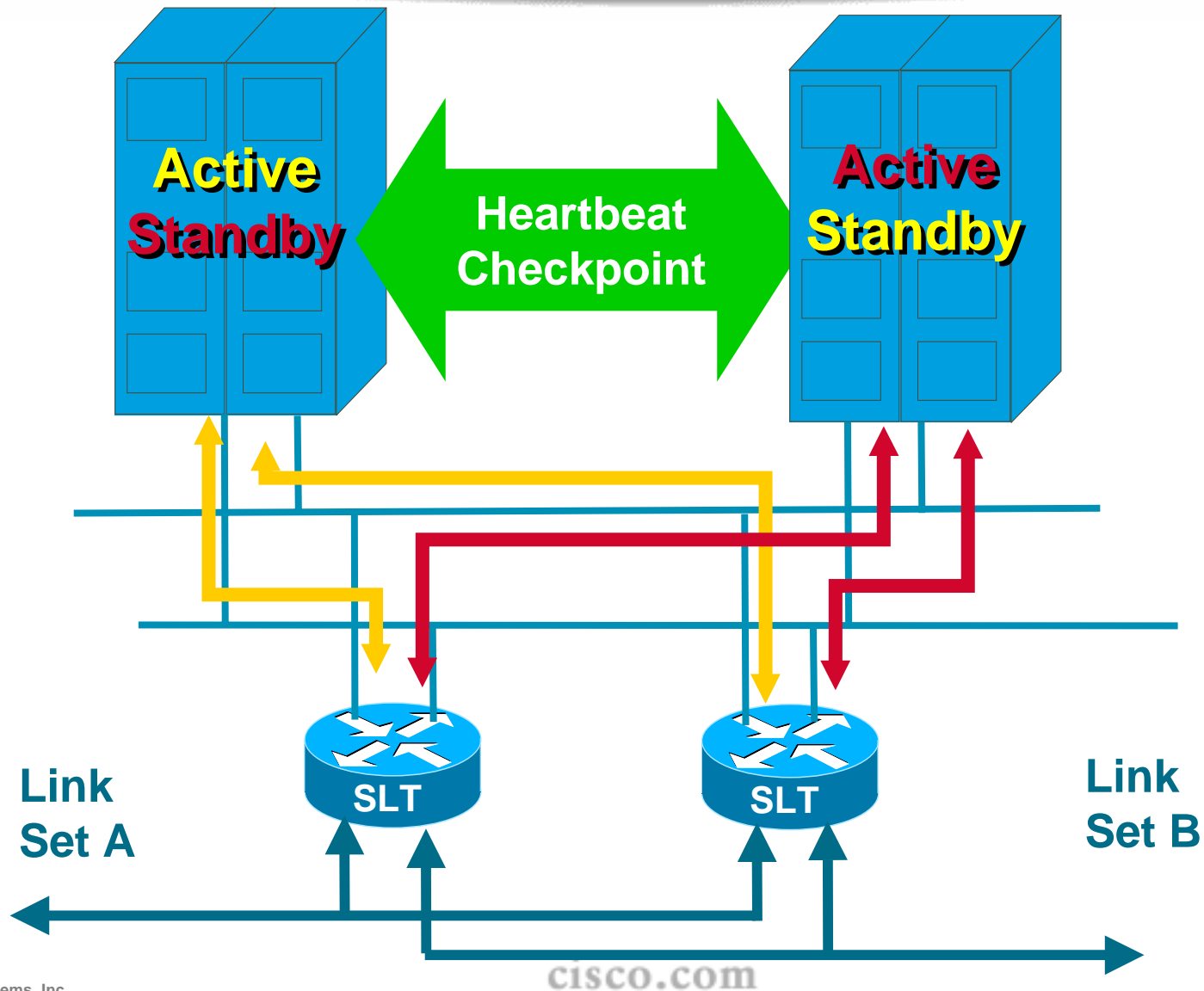
- Connection control: create/modify/delete
- Service commands: notify, audit GW
- Bearer agnostic: VoIP, VoATM

Signaling Backhaul

SS7 - Signaling Link Terminal

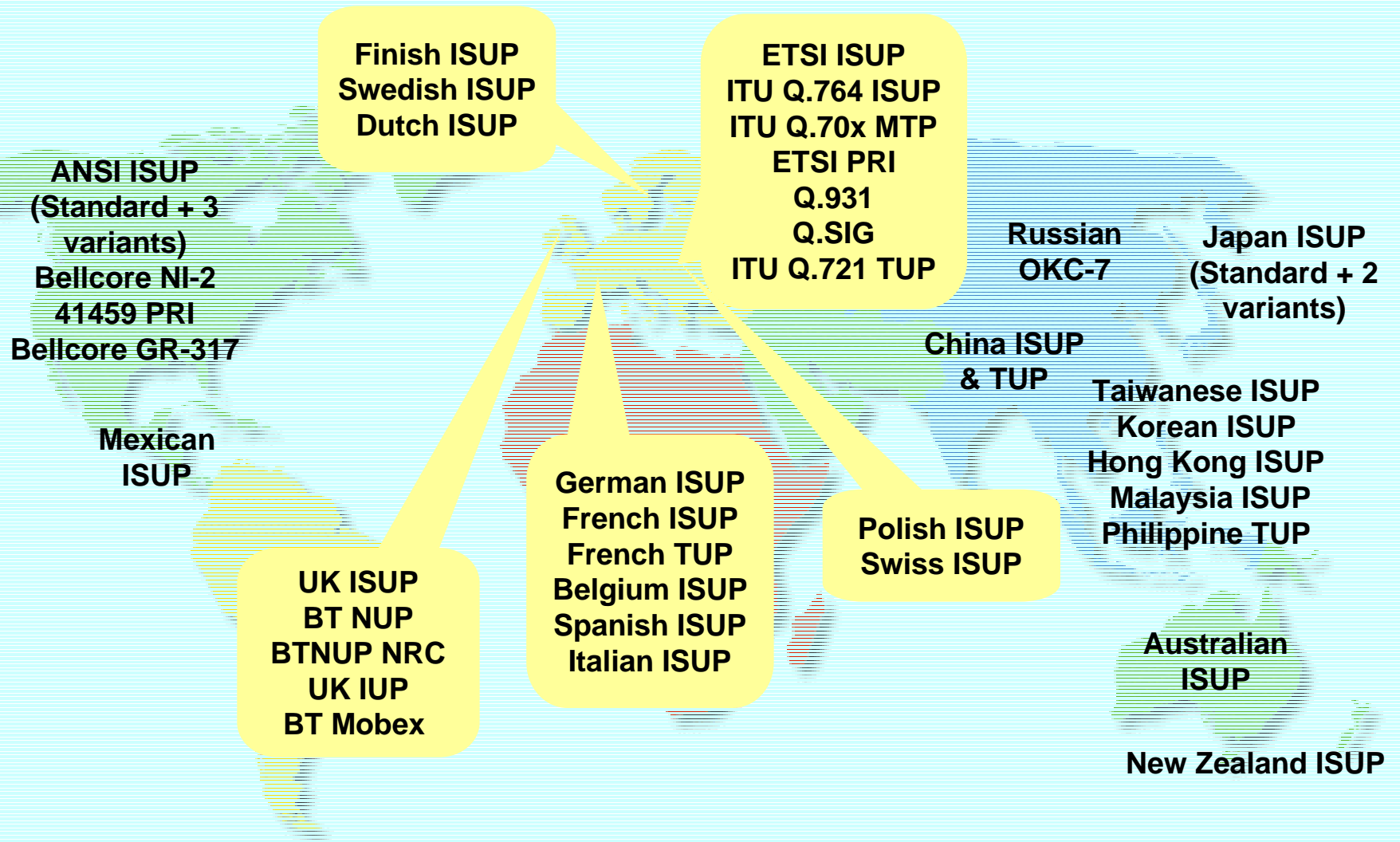


VSC Node Redundancy



VSC Signaling Support

Constantly Updated !





MGX 8850

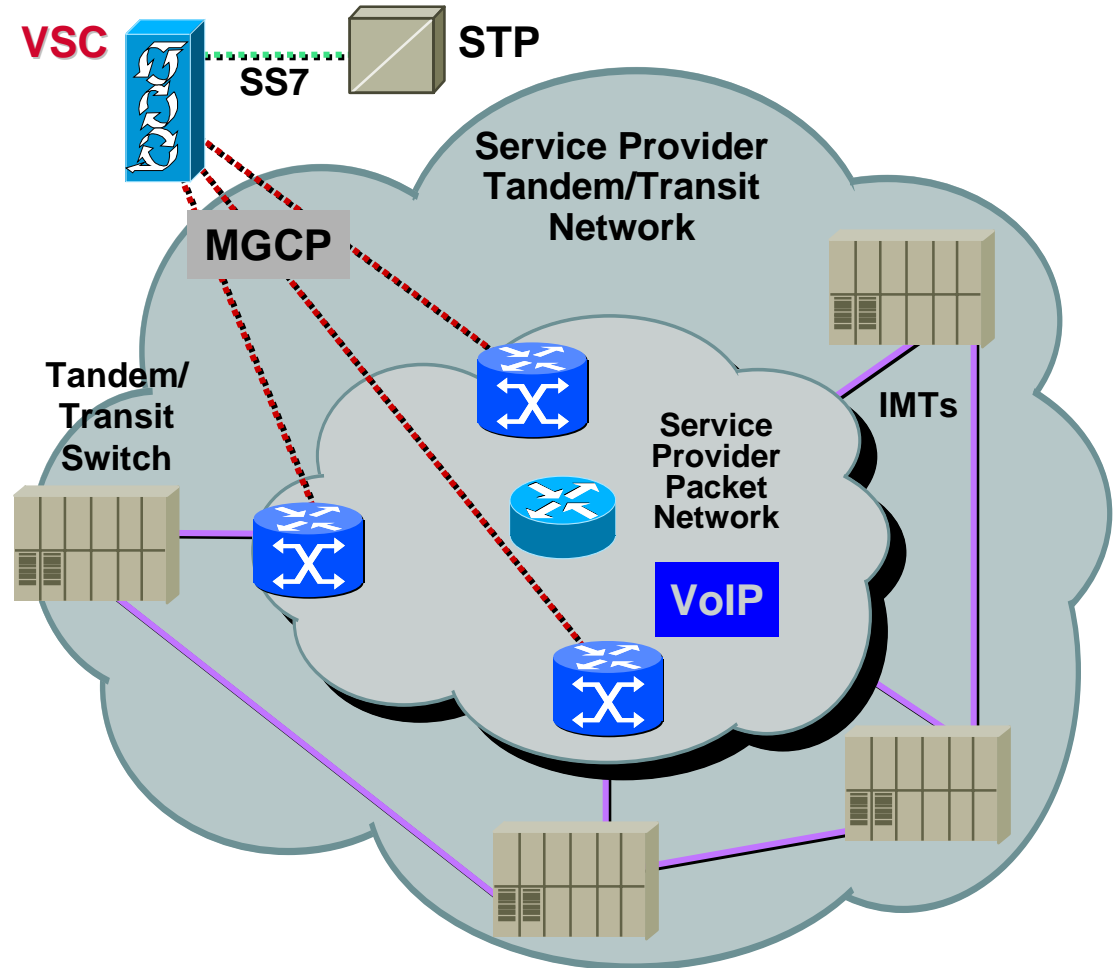
MGX 8850: First Integrated VoIP and VoATM Edge Switch

- Widest range of data and voice services on a carrier edge switch
- Most cost-effective multiservice solution for both small and large sites
- Widest range of data and voice services on a carrier edge switch
 - DS0 to OC-48c
 - ATM, FR, IP, TDM Services
- Carrier class reliability

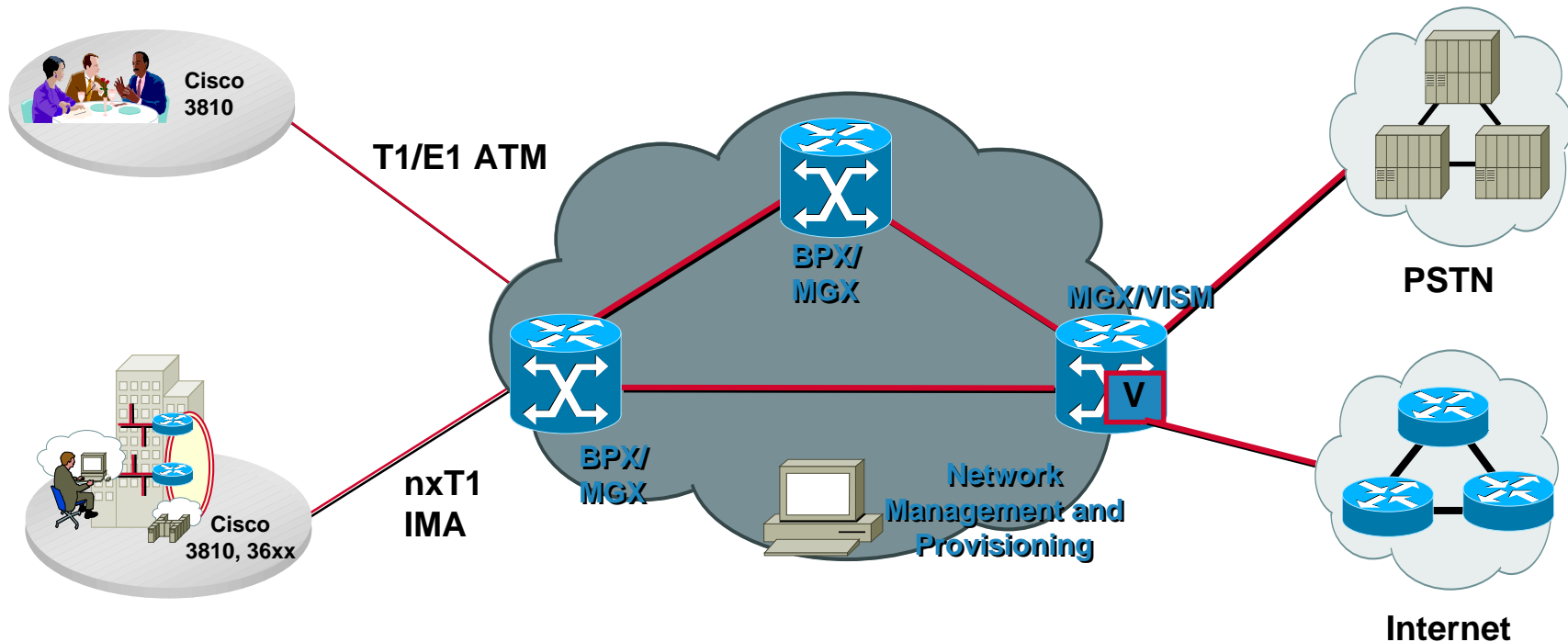


Packet Tandem Offload (VoIP)

- New packet revenue stream
- Unique capability to hairpin/groom voice traffic using TDM switching or conversion to VoIP
- Open interfaces: more choices, faster time to market
- SNMP-based EMS
 - Provides fault, performance, security, and configuration management
- CDR interface for billing mediation devices

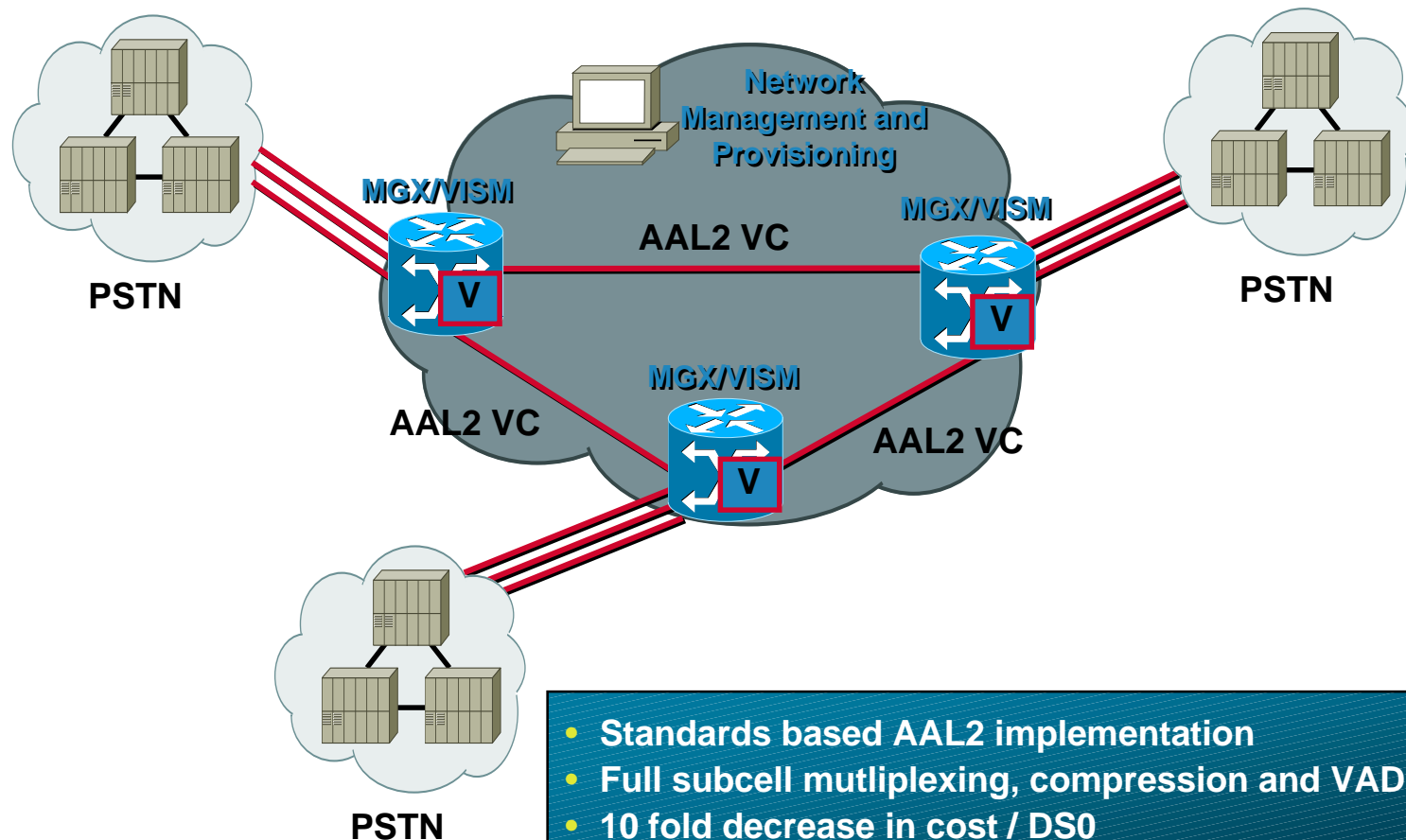


MS Access (T1-VoAAL2)



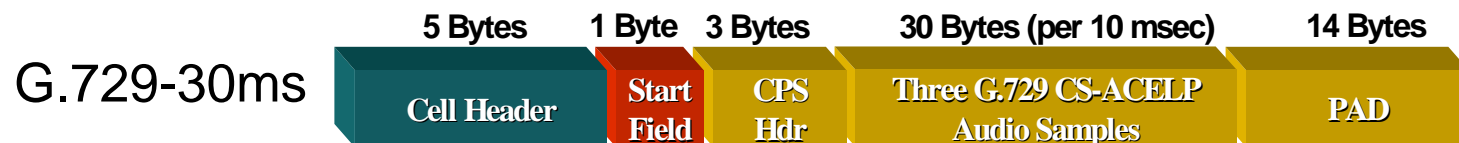
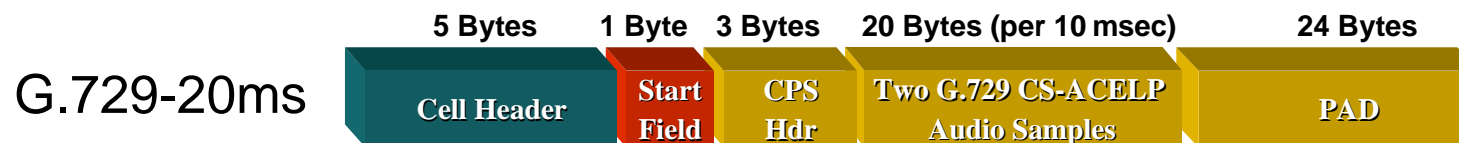
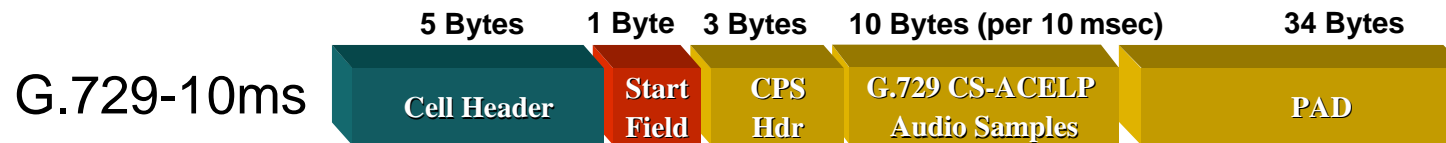
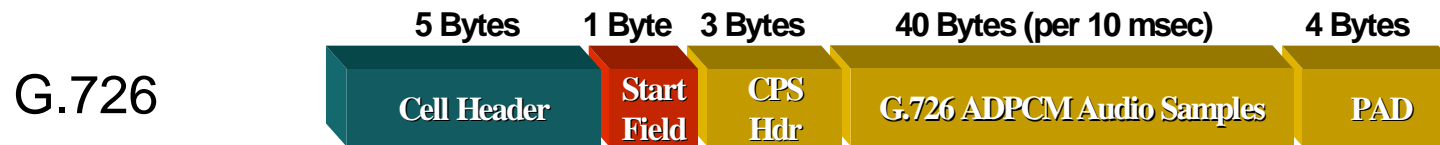
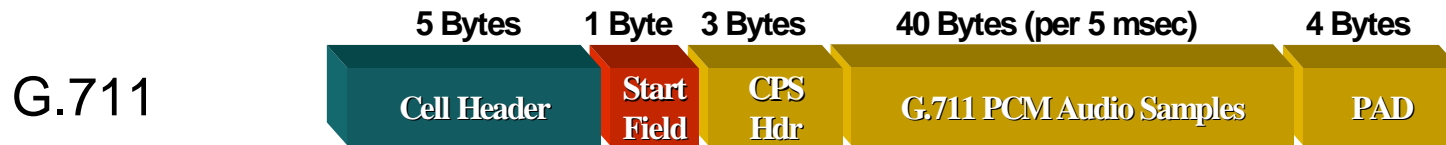
- Integrated IP+ATM voice and data access over a single line
- Compressed Voice over AAL2 using PVC Trunking Model
- Improved access economics

Bandwidth Efficient Point-to-Point Trunking



- Standards based AAL2 implementation
- Full subcell mutliplexing, compression and VAD
- 10 fold decrease in cost / DS0
- Integrated network for voice and data

AAL2 Coding Schemes



Cisco AS5300/VoIP

- **Services Supported**

PSTN & PBX to VoIP

Toll bypass

Direct and 2-stage dialing

Fax Relay

Integrated IVR

Modem & ISDN Internet access

- **Performance**

150-MHz R4700 RISC CPU

High-performance,
low-latency architecture

- **Carrier Class**

Toll quality

NEBS

Four T1/E1/PRI (supports 48/60 voice sessions)

10-Mb and 10/100-Mb Ethernet interface



Cisco AS5300 Voice/Fax Feature Card

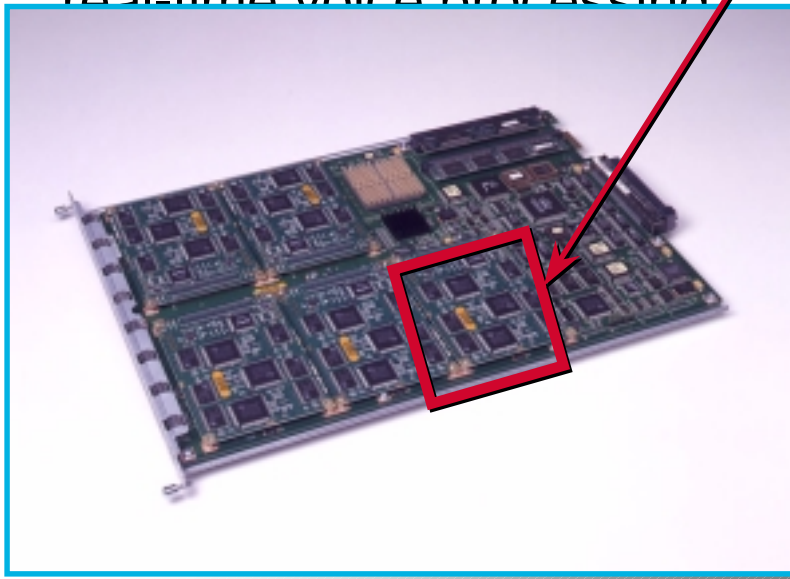
- **Voice/fax feature card**

One or two per AS5300

24/30 ports per feature card

100-MHz RISC controller

Specifically designed for real-time voice processing



- **DSP modules**

Snap-on DSP modules

Six DSPs per module

G.729 CS-ACELP—8 K

G.711 PCM

G.165 echo cancellation

Group III fax relay

- **VoIP protocol support**

H.323 standard

RTP, RTCP, CRTP, NTP

Cisco IOS™ QoS features

IP precedence, WFQ,
WRED, RSVP, MLF

Basic Interactive Voice Response (IVR)

- **IVR application integrated into the voice Gateway**
- **Simple voice prompting and digit collection from the caller to:**
 - authenticate the user**
 - identify destination**
- **Customer may record their own announcements and prompts (script logic cannot be customized)**
- **Scripts supported:**
 - Announcement**
 - ANI authorization**
 - Account number/PIN authorization**
 - Fax hop on/off (use of redialer boxes)**



Understanding Unified Communications for Service Providers

Session 2004



Agenda

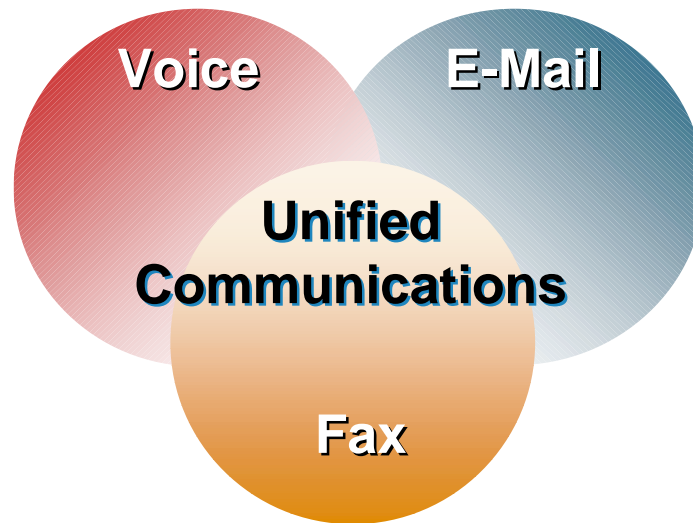
- **Introduction, Features, and Benefits**
- **Architecture and Components**
- **Typical Call Flows**
- **Deployment in a Service Provider Environment**
- **Redundancy and Load Balancing**

Agenda

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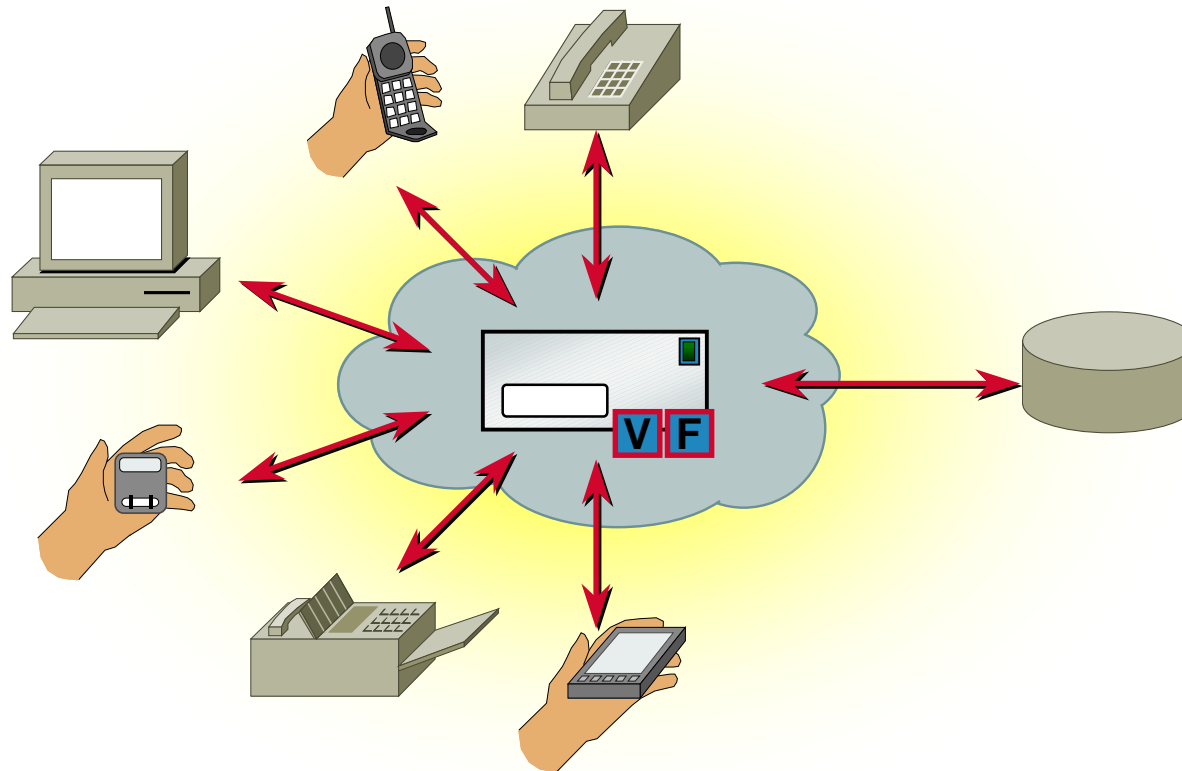
Unified Communications

Unified communications is an enhanced voice over IP solution that provides the ability to manage voice mail, e-mail and fax under a common message store, on an existing IP infrastructure.



Unified Communications

Multiple Device Types and Media



Non Real-Time Message Exchange

Unified Communications Features

Voice mail

- Multiple personalized greeting
- Handle all messages with a single call
- Designate and prioritize messages
- Leave messages for multiple subscribers
- Forward Voice messages as e-mail attachments
- Locate subscribers using name or phone number
- Message waiting indication by pager, stutter dial tone or indicator light

Unified Communications Features

E-Mail Messaging

- Ability to identify voice, e-mail and fax messages in mailbox
- Play voice messages as streaming audio
- Listen to e-mail messages over the phone using text to speech processing (TTS)
- Respond to an e-mail message over the phone as an audio attachment to the original sender
- Message waiting indication on arrival of new e-mail messages
- Print e-mail to a local fax machine

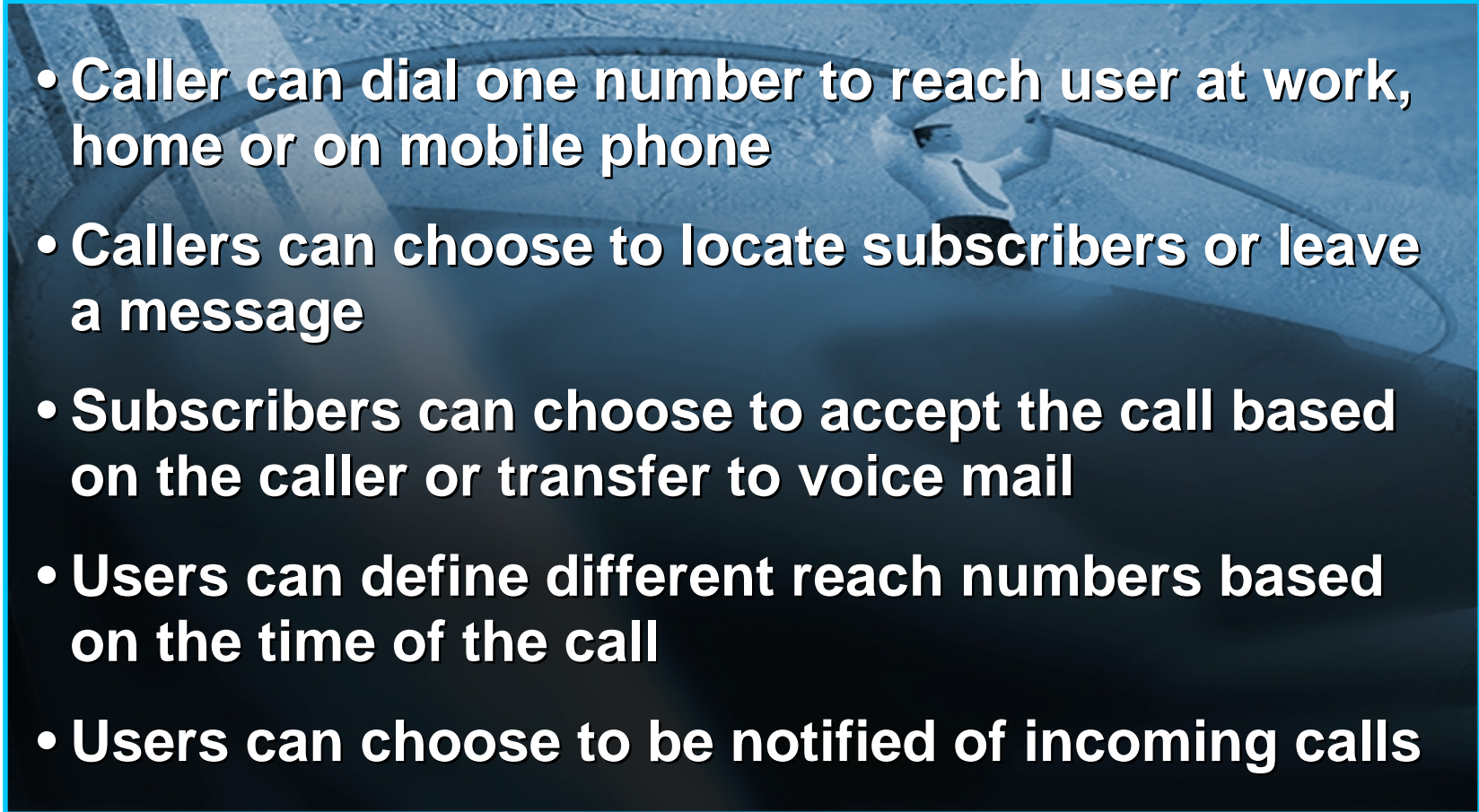
Unified Communications Features

Fax Messaging

- Ability to redirect fax messages to a local fax machine when ready
- Determine the time of arrival and sender of a fax message using a telephone
- View faxes as a (.tiff) attachment to an e-mail message
- Forward faxes as e-mail attachments to other users
- Message waiting indication on arrival of new fax messages

Unified Communications Features

Single Number Reach

- 
- Caller can dial one number to reach user at work, home or on mobile phone
 - Callers can choose to locate subscribers or leave a message
 - Subscribers can choose to accept the call based on the caller or transfer to voice mail
 - Users can define different reach numbers based on the time of the call
 - Users can choose to be notified of incoming calls

Service Provider Benefits

- **Brand services for greater recognition**
- **Drive minutes of use on the network, increasing total revenue per subscriber**
- **Reduce churn by strengthening customer relationships with value added services**
- **Reduce cost of ownership by utilizing a common platform to introduce new applications and services**

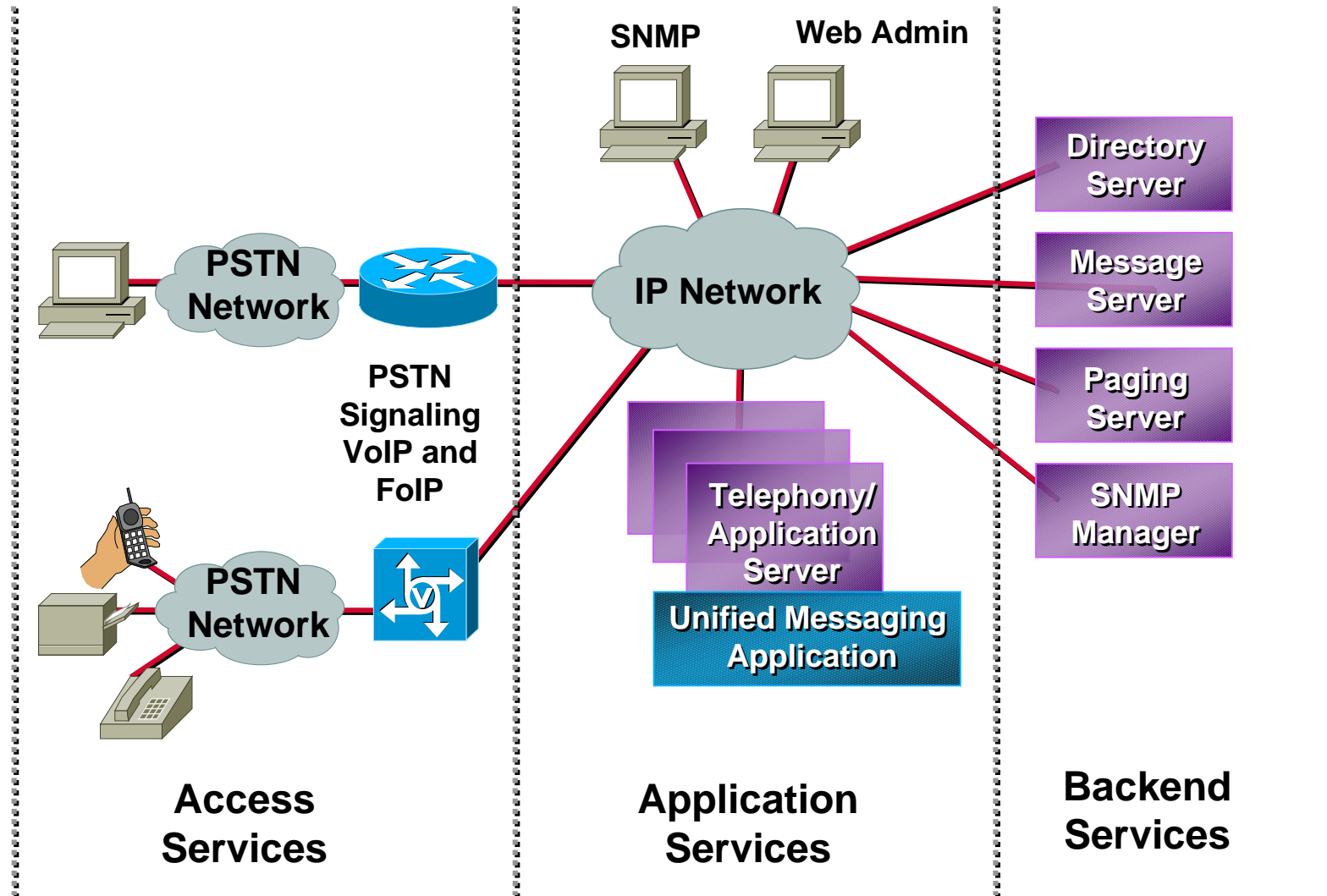
End User Benefits

- **Users can manage and access all of their messages regardless of the media type**
- **Remote users can access all of their messages with one phone call into their unified messaging system**
- **Voice mail, e-mail, and fax messaging are non-real-time means of communications, allowing users to access their messages at any time**
- **Media conversion allows users to access their messages in the media of their choice**

Agenda

- Introduction, Features, and Benefits
- **Architecture and Components**
- Typical Call Flows
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Unified Communications Network Components



Network Components

Access Services

- **Edge devices that provide telephony and data access to the network**

Cisco AS5300/5800 with Cisco IOS 12.0.5(T2) and vcw-vfc-mz.c542.4.04 as VoIP H.323 gateways which provide access to the network from traditional telephony devices

Cisco AS5300 Onramp and Offramp fax gateways that provide access to and from the network for group three fax machines

Access servers for dial in data access from PC's

Network components Application Services

- **Message management logic and H.323 call termination point**
- **Gateserver**

Sun Netra T 1125 Dual processor, 440 MHz, 512MB RAM, 9.1G hard drive with Solaris 2.6

uOne gateserver software version 4.2S

RadVision rel 2.1.2.3 H.323 Stack (Included with uOne)

L&H Telecom TTS (Text to Speech) V.100 for Solaris with American English, French and Spanish language sets

SNMP master agents (optional)—Solstice enterprise agents runtime V1.03

Network Components Backend Services

- **Iplanet Directory server 4.0 (LDAP)**
- **Iplanet Messaging server 4.1**
- **Hylafax paging server 4.0, patch 1**
- **Apache Web server 1.3.6**
- **Network management workstation**

Backend Services

Directory server

- **Storage of user profile information in a hierarchical tree like structure based on organizational or geographic boundaries**
- **Tuned to give quick response to high volume search operations**
- **LDAP (lightweight directory access protocol) used to manage user information on directory. Uses TCP port 389**

Backend Services

Messaging Server

- **Common message store for uOne with open access (IMAP4, HTTP, SMTP)**
- **Uses directory service for user account information/authentication**
- **Messages stored using SMTP in MIME format**
- **Retrieved using IMAP4, POP3 or HTTP (Web-based e-mail client)**

Backend Services

Paging Server

- **HylaFax paging server—interfaces with uOne using SNPP (Simple Network Paging Protocol)**
- **Solaris 2.6 based**
- **Connects to a modem server using a single ended SCSI 2 cable**

Gateserver Architecture

- **Distributed object-based framework**
- **Based on new and non-proprietary voice and information standards**
- **Several major components that can be distributed across multiple systems**

Gateserver Components

- **Agent Manager and Monitor (AMM)**

Scheduling, routing, launching, monitoring, and terminating services

Service libraries to access non proprietary services (IMAP, SMTP)

- **Call Control/Media Agent (CMA)**

Provides all H.323 services (H.225, RAS, H.245)

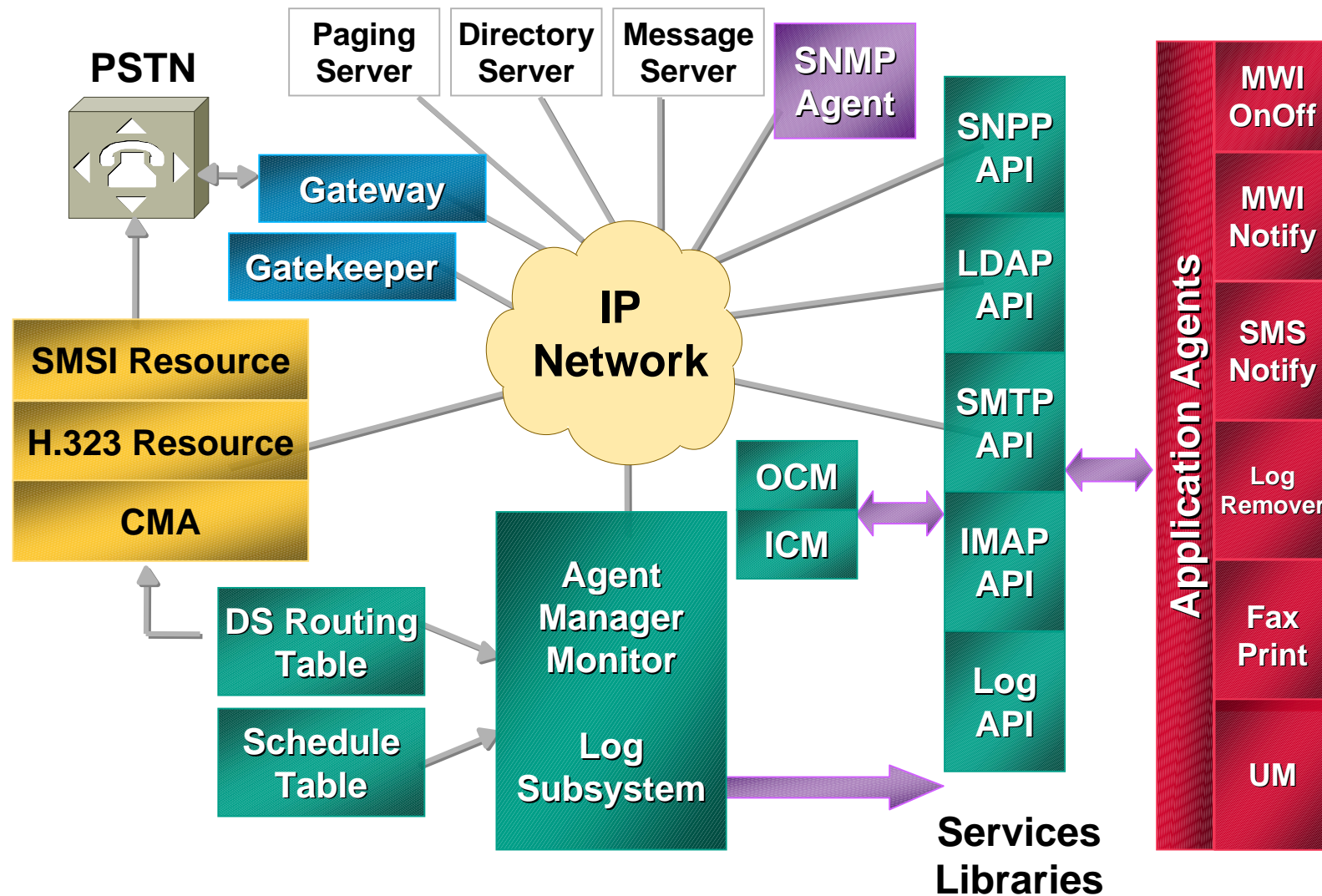
Media services such as playing and recording messages

- **Application agents**

Launched by AMM to provide specific tasks such as fax, notification

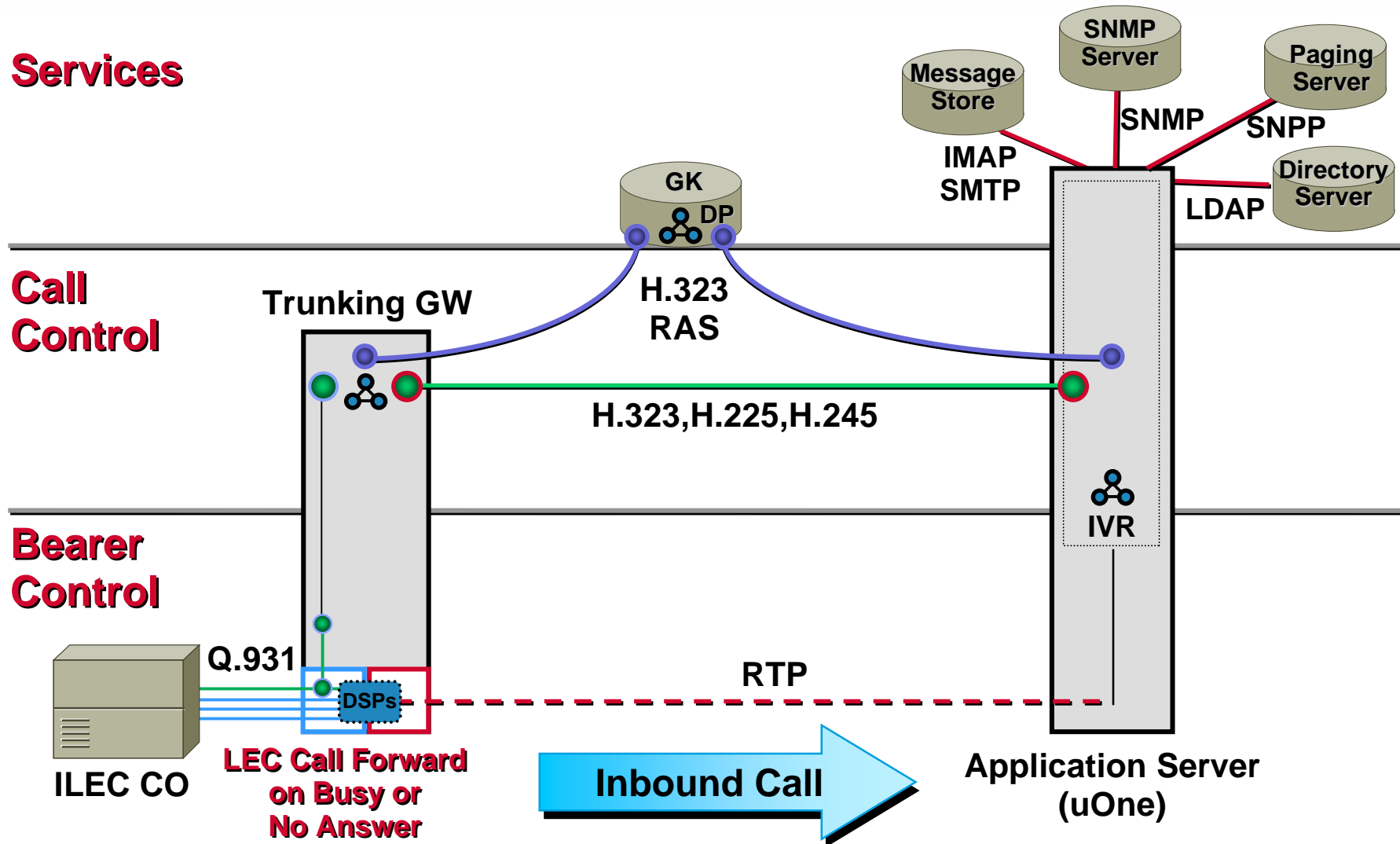
Provide access to back-end servers by using service library APIs

Gateserver Component Overview



Unified Communications Protocol Overview

Services

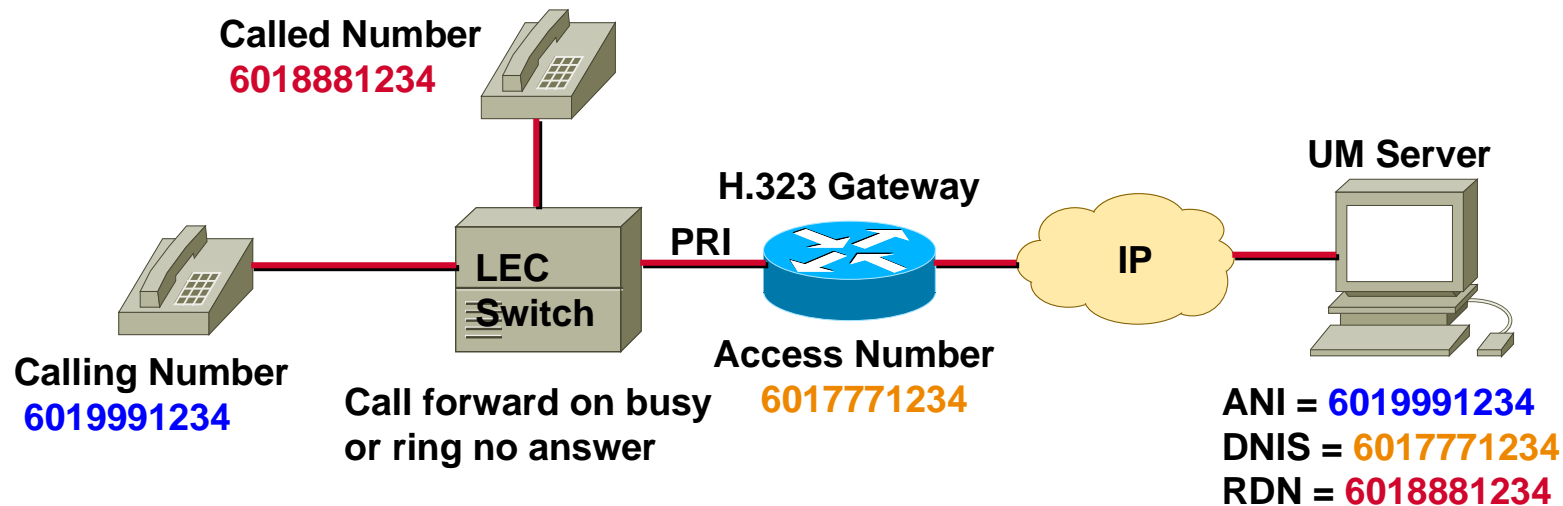


Agenda

- Introduction, Features, and Benefits
- Architecture and Components
- **Typical Call Flows**
- Deployment in a Service Provider Environment
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Call Flows

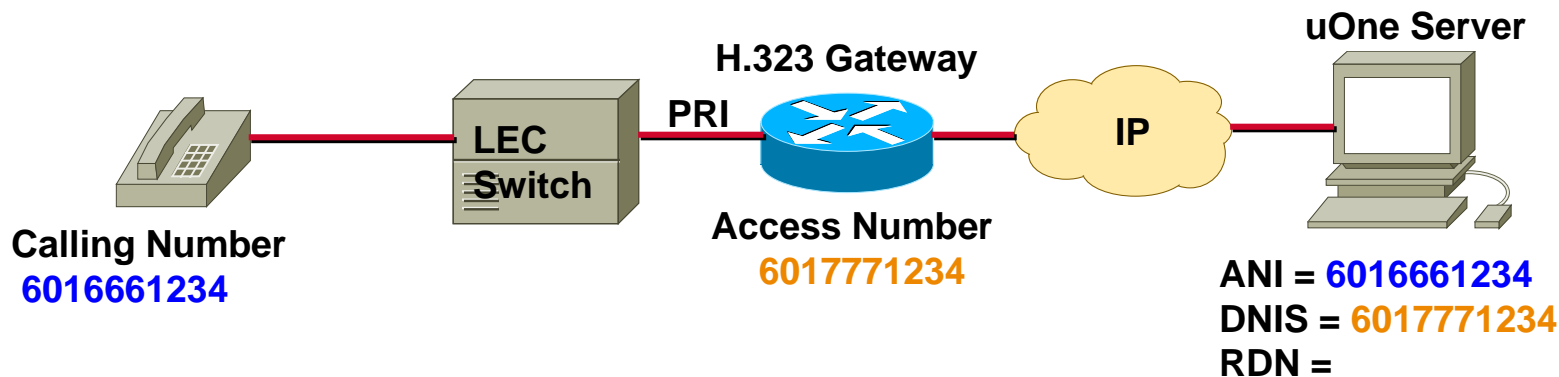
Identifying Callers



- Presence of RDN (Redirected Number) indicates call to subscriber
- uOne searches for subscriber profile using **6018881234**, retrieves and plays personal greeting

Call Flows

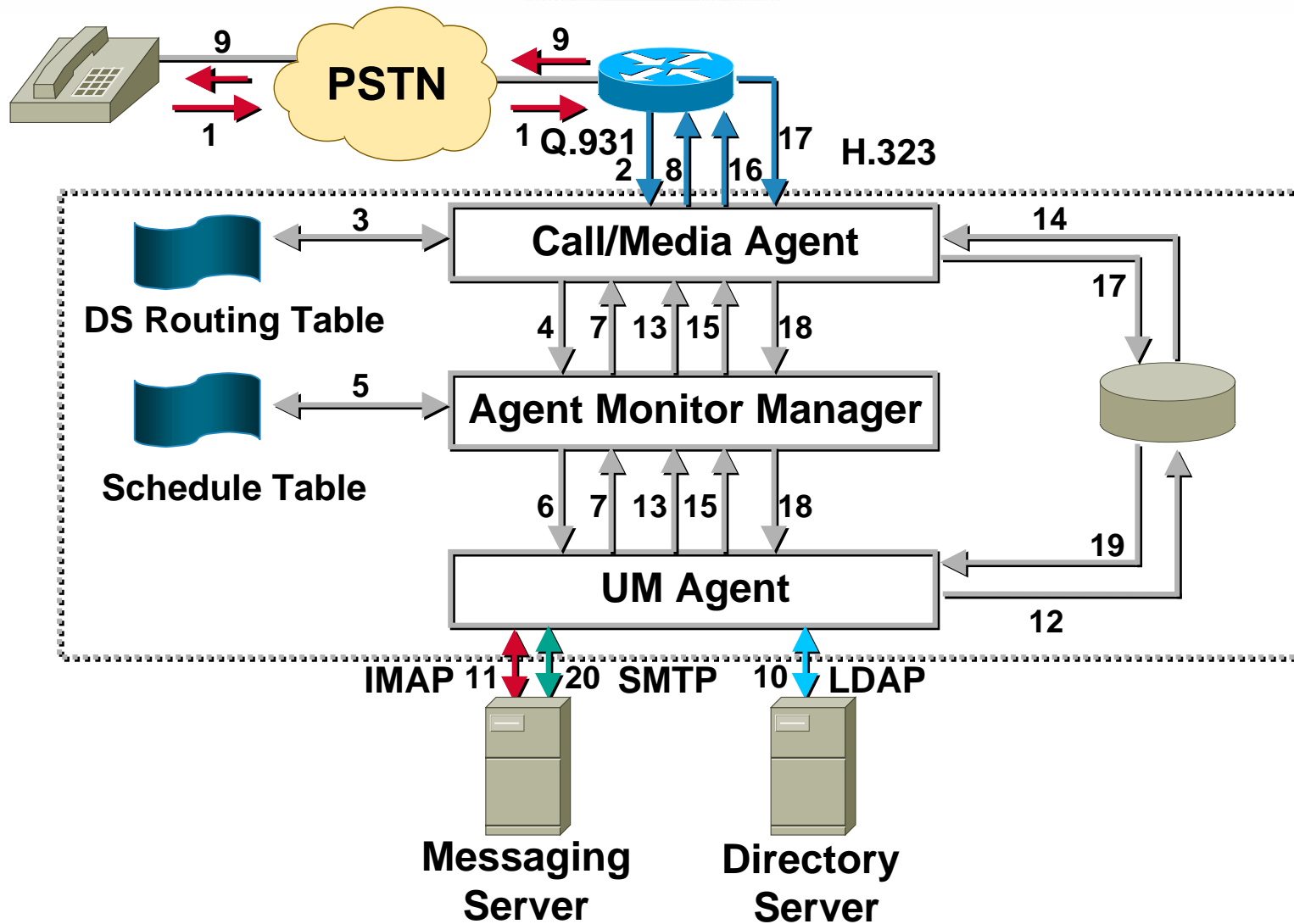
Identifying Subscribers



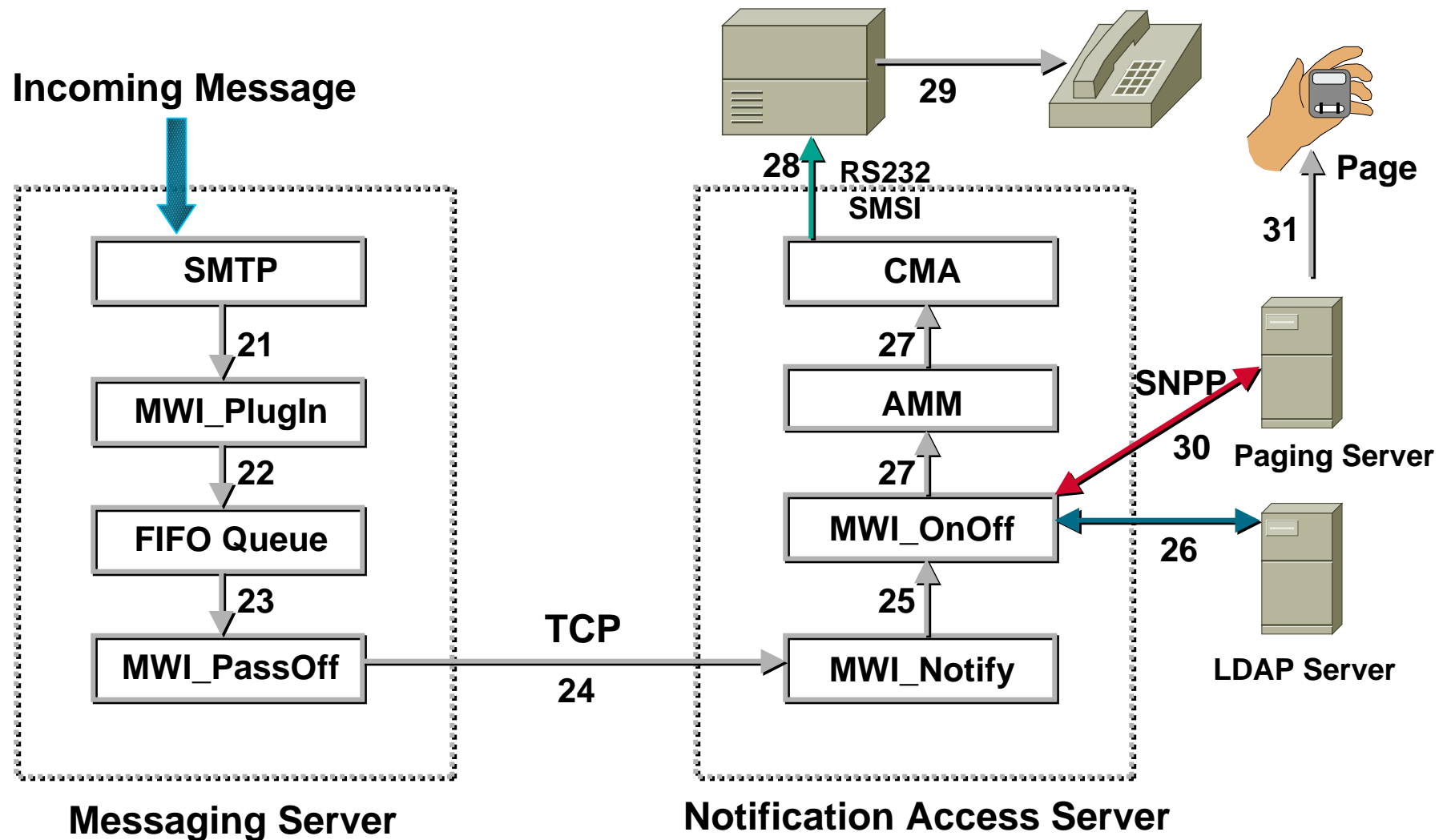
- Unpopulated RDN field indicates call from subscriber to retrieve messages
- uOne requests subscriber to enter phone number or simply press #
- If subscriber enters phone number, it is used in directory search (LDAP)
- If subscriber enters # 6016661234 is used to search directory for profile

Call Flows

Leaving a Message



Call Flows Notification



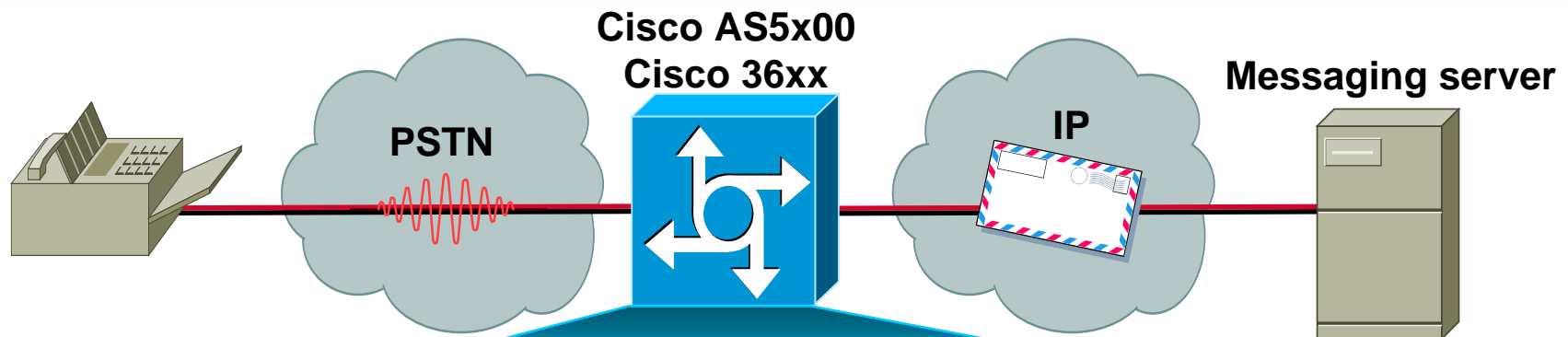
Call Flows

Inbound Fax

- **uOne creates an alias on messaging server mapping assigned fax number to e-mail address**
- **uOne does not participate in inbound fax to subscriber**
- **Fax delivered to subscriber mailbox by onramp fax gateway**

Call Flows

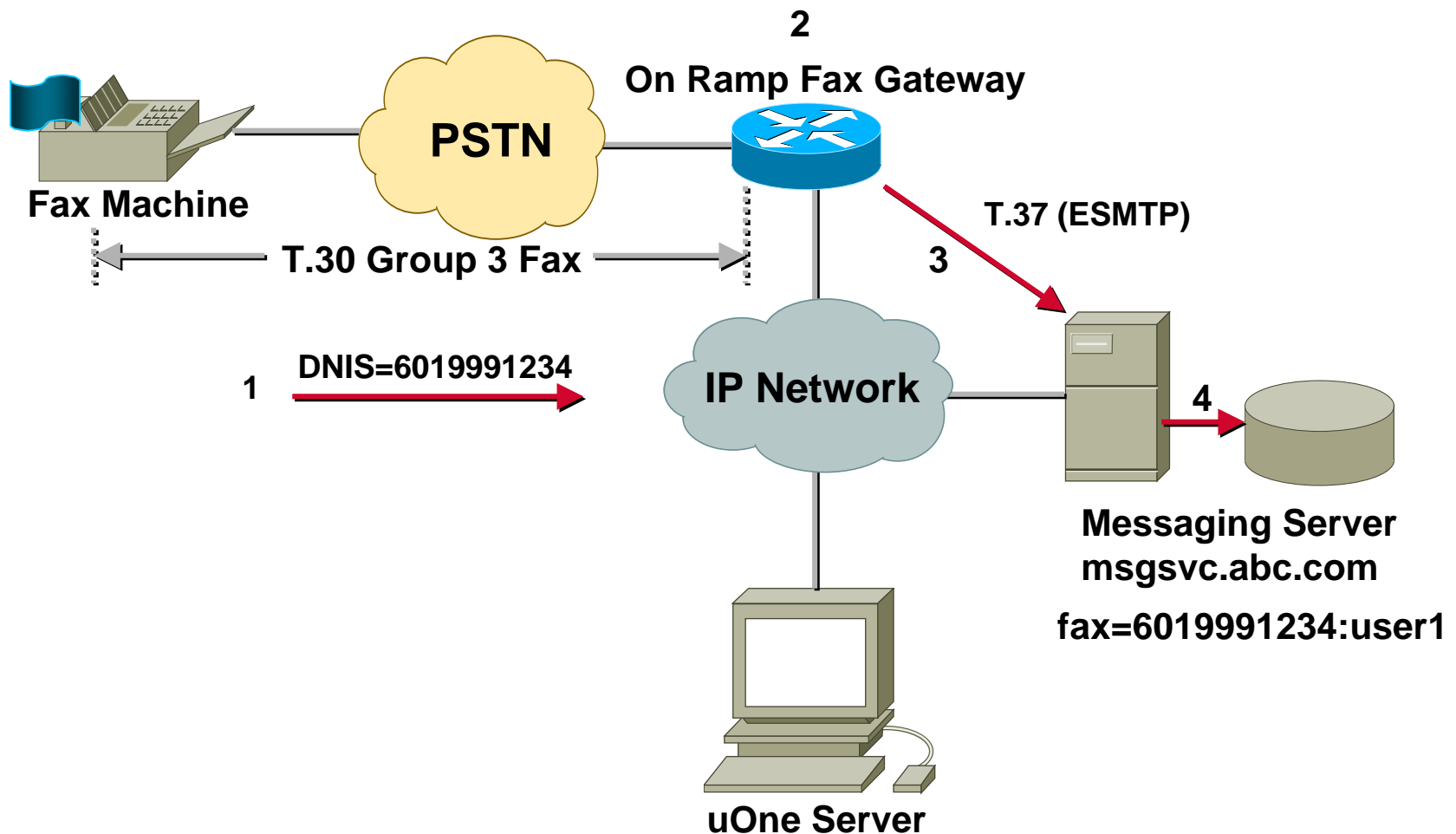
Onramp Function



- **Demodulate fax call**
- **ITU-T T.30 fax protocol handling**
- **Turn fax image into TIFF file**
- **Create MIME message with TIFF attachment**
- **Optionally re-write to: address**
- **Create call history record**
- **Forward to ESMTP mail server**

Call Flows

Inbound Fax



Call Flows

Inbound Fax

- 1. User sends a fax to the subscribers telephone number (6019991234). The fax connects to a fax gateway (AS5300)**
- 2. Incoming call is determined to be a fax call because the DNIS matches dial peer with information type set to fax. The gateway converts T.30 Group 3 fax to a .tif file**
- 3. The gateway creates a mail message, attaches the .tif file and delivers it using ESMTP to the messaging server. Destination e-mail: fax=6019991234@msgsvc.abc.com**

Call Flows

Inbound Fax

4. Messaging server deposits fax message in users mailbox

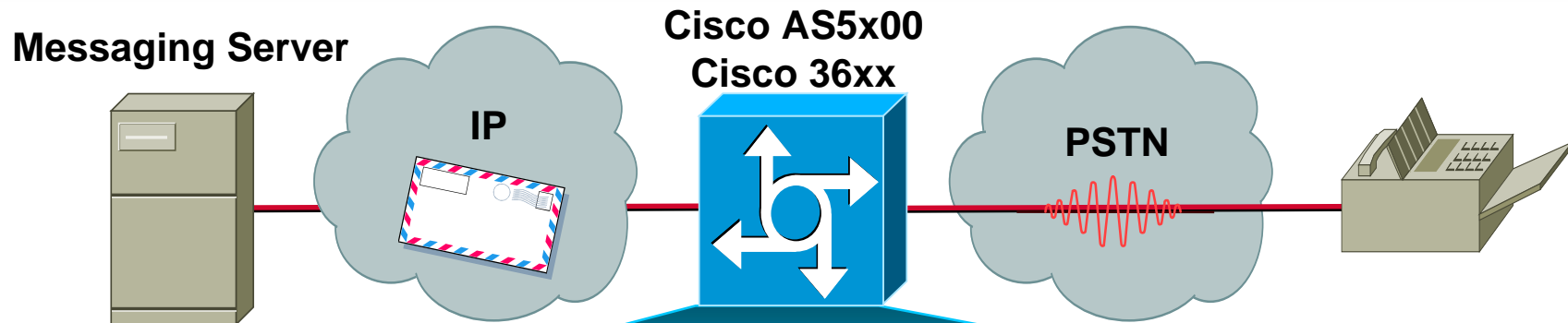
uOne has created an alias on the messaging server that maps fax=6019991234@msgsvc.abc.com to user1@msgsvc.abc.com at setup time

The receipt to e-mail address is maintained as the alias fax=6019991234@msgsvc.abc.com

The “fax=” alias in the “to:” address defines this message to be a fax message at retrieval time

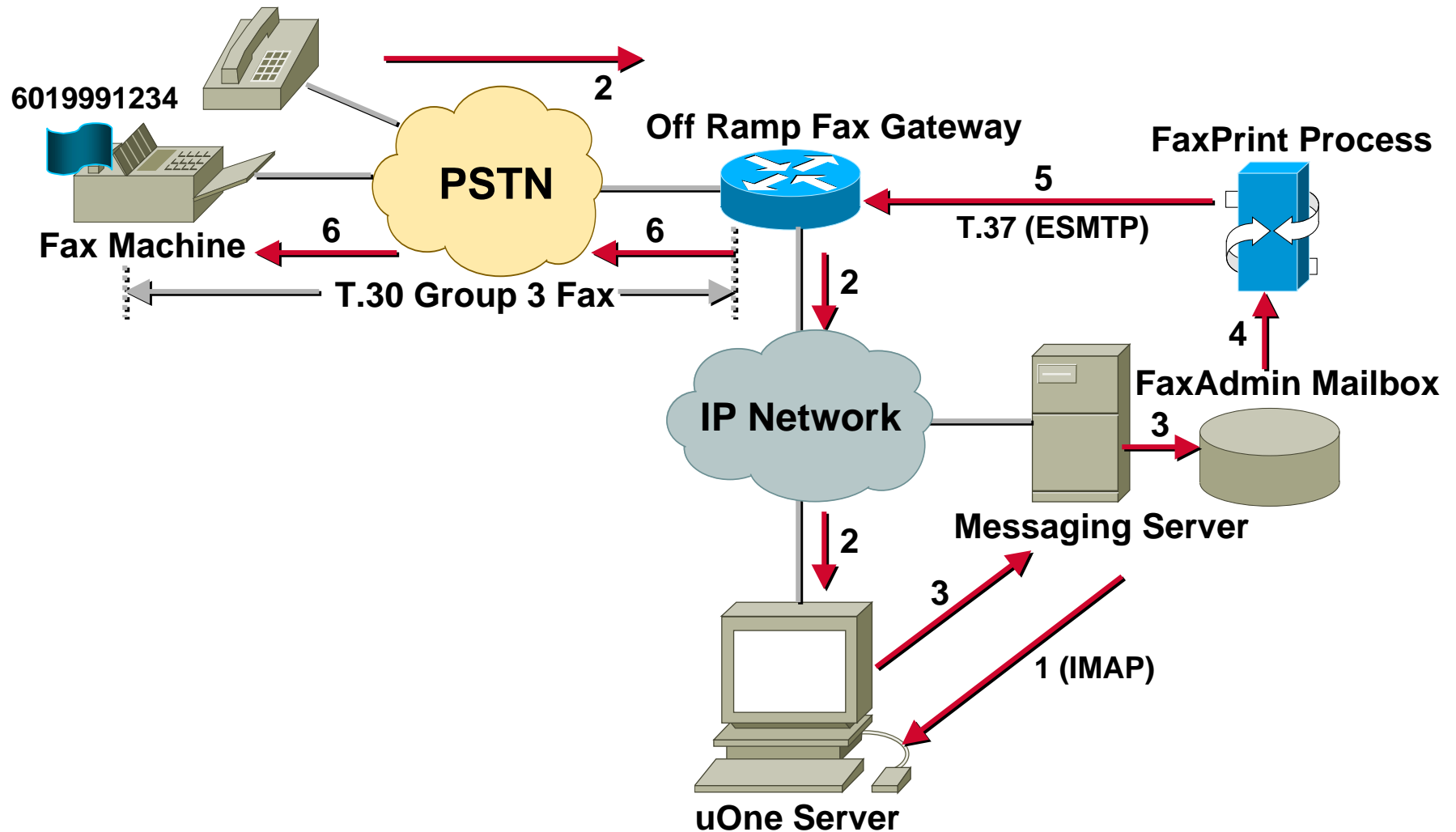
Call Flows

Offramp Function



- **Authenticate sender against AAA (optional)**
- **Rasterize text portions of e-mail (text->fax)**
- **Rasterize TIFF-F into fax pages**
- **Re-write fax destination number (optional)**
- **ITU-T T.30 fax protocol handling**
- **Modulate fax call**
- **Create call history record**
- **Delivery status notification**

Call Flows Outbound Fax



Call Flows

Outbound Fax

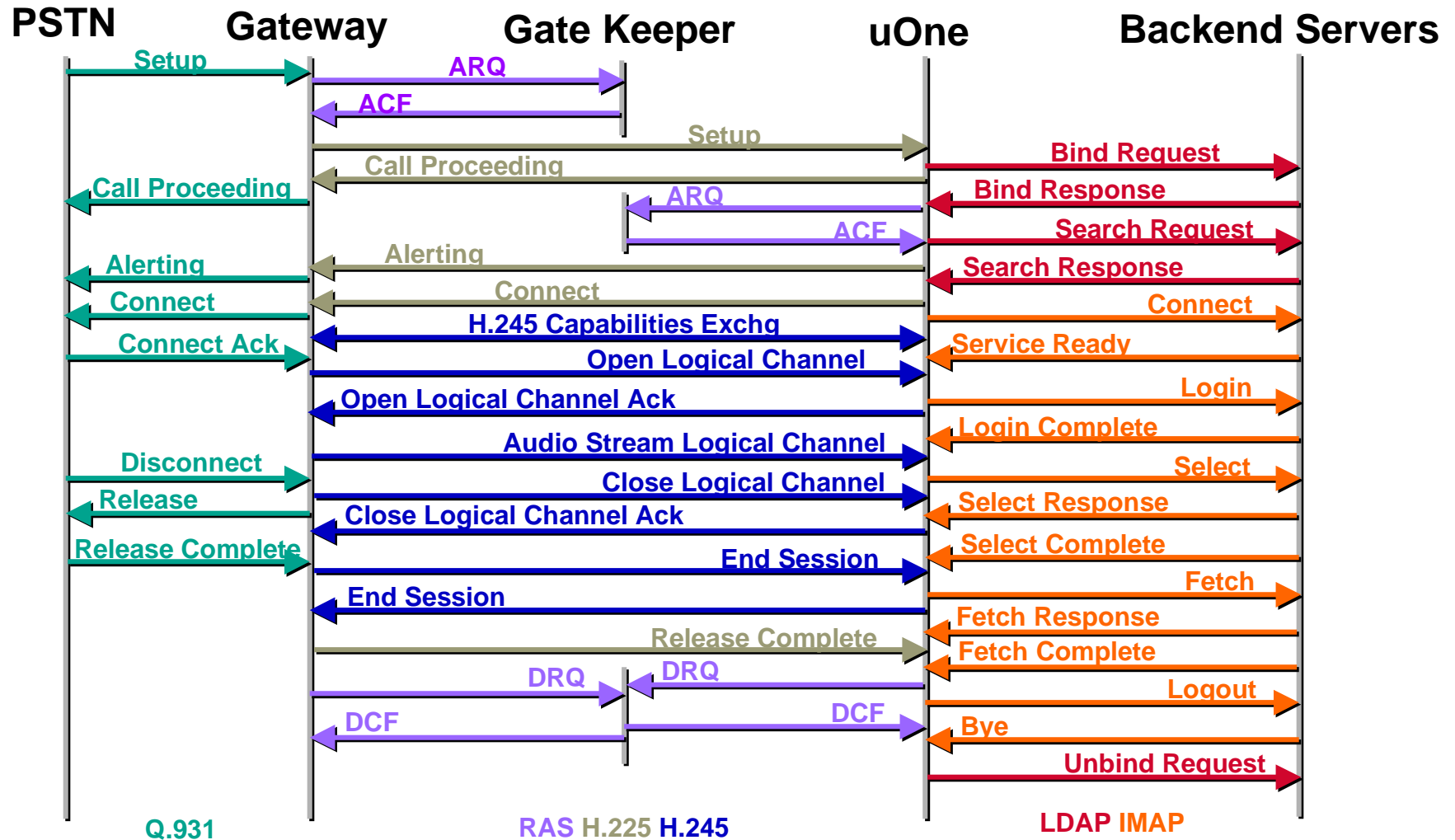
- 1. UM agent retrieves the fax or e-mail message from the messaging server using IMAP**
- 2. Subscriber chooses the option to print the message and keys in the phone number of the fax machine where the message is to be sent, example 6019991234**
- 3. UM agent adds destination fax information to the message and forwards it to subscriber's faxadmin e-mail account using SMTP**

Call Flows

Outbound Fax

4. FaxPrint application constantly monitors faxadmin's mailbox for new messages. It retrieves message sent in the previous step using IMAP
5. FaxPrint application sends the message to off Ramp fax gateway (ESMTP) addressed to (fax=6019991234@gw.abc.com)
6. Fax gateway extracts the destination phone number from e-mail address, converts any text to .tif format and sends the fax to the destination as T.30 Group 3 fax

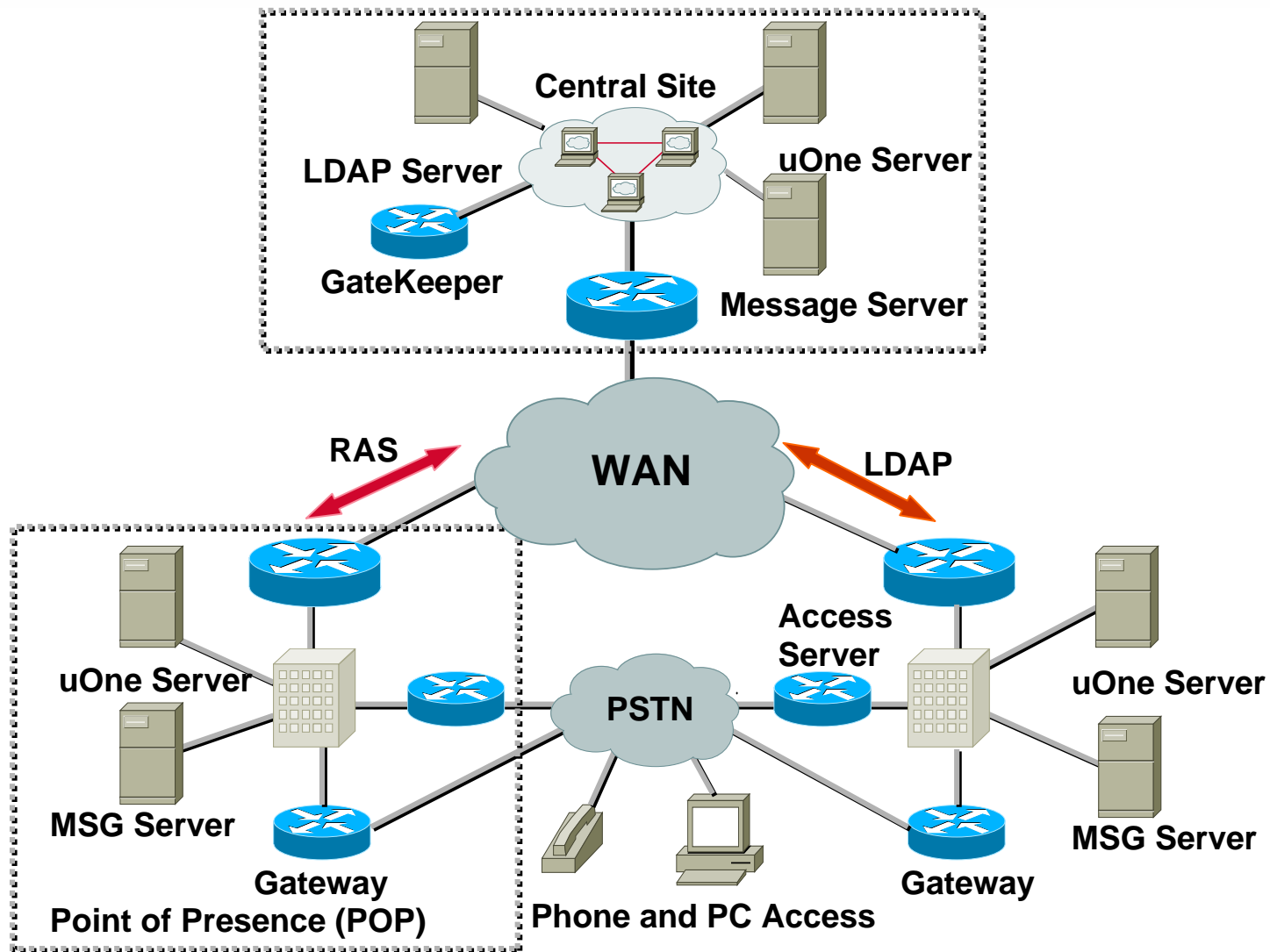
Call Flows Summary



Agenda

- Introduction, Features, and Benefits
- Architecture and Components
- Typical Call Flows
- **Deployment in a Service Provider Environment**
- Redundancy and Load Balancing

Deployment Overview



Deployment Overview

- **Service quality depends on where uOne components are placed in the network**
- **Inherent delays across WAN's**
- **Better quality and service by minimizing traffic across WAN's**

Deployment Overview

Service Quality

- **uOne server**

Call setup times, voice quality

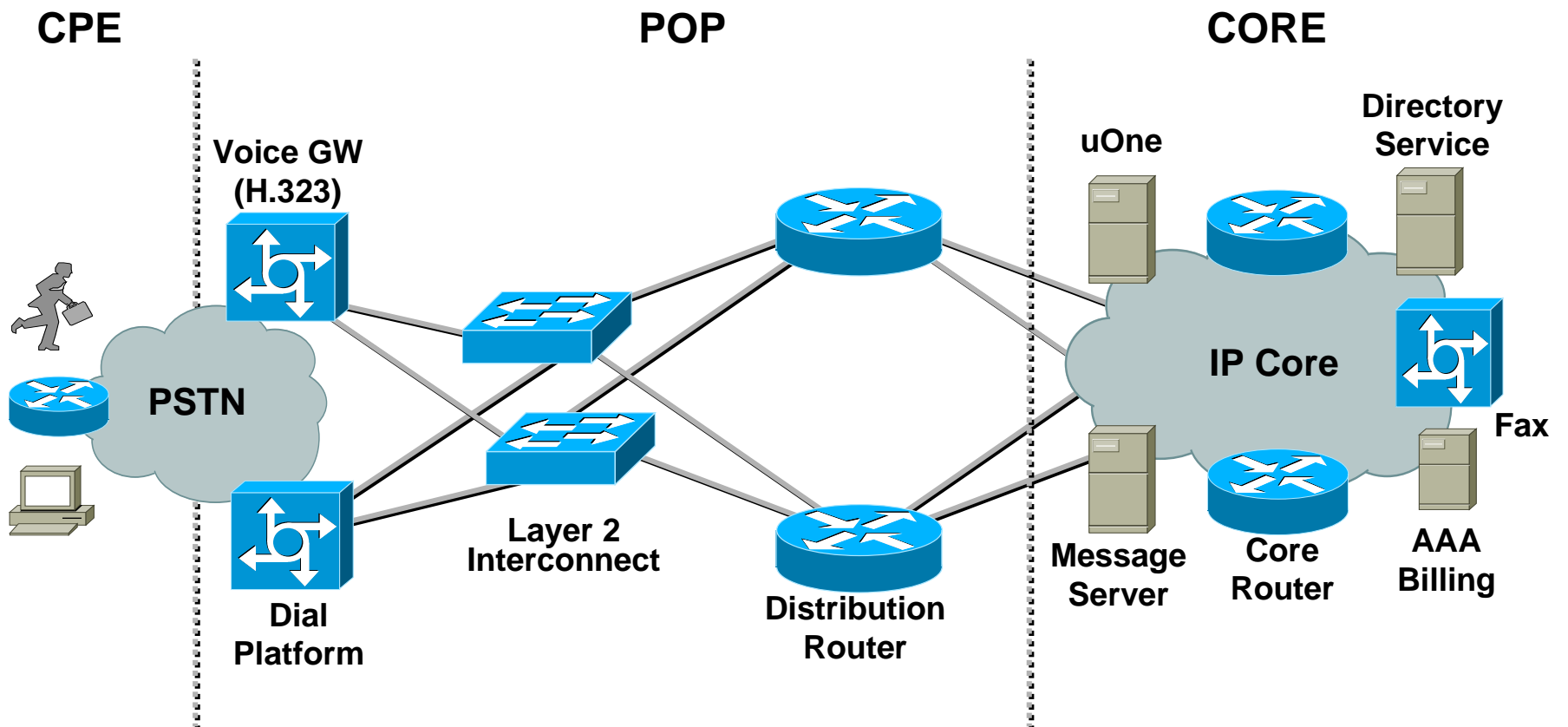
- **Directory server**

**Authentication, message response
(has to be centralized)**

- **Messaging server**

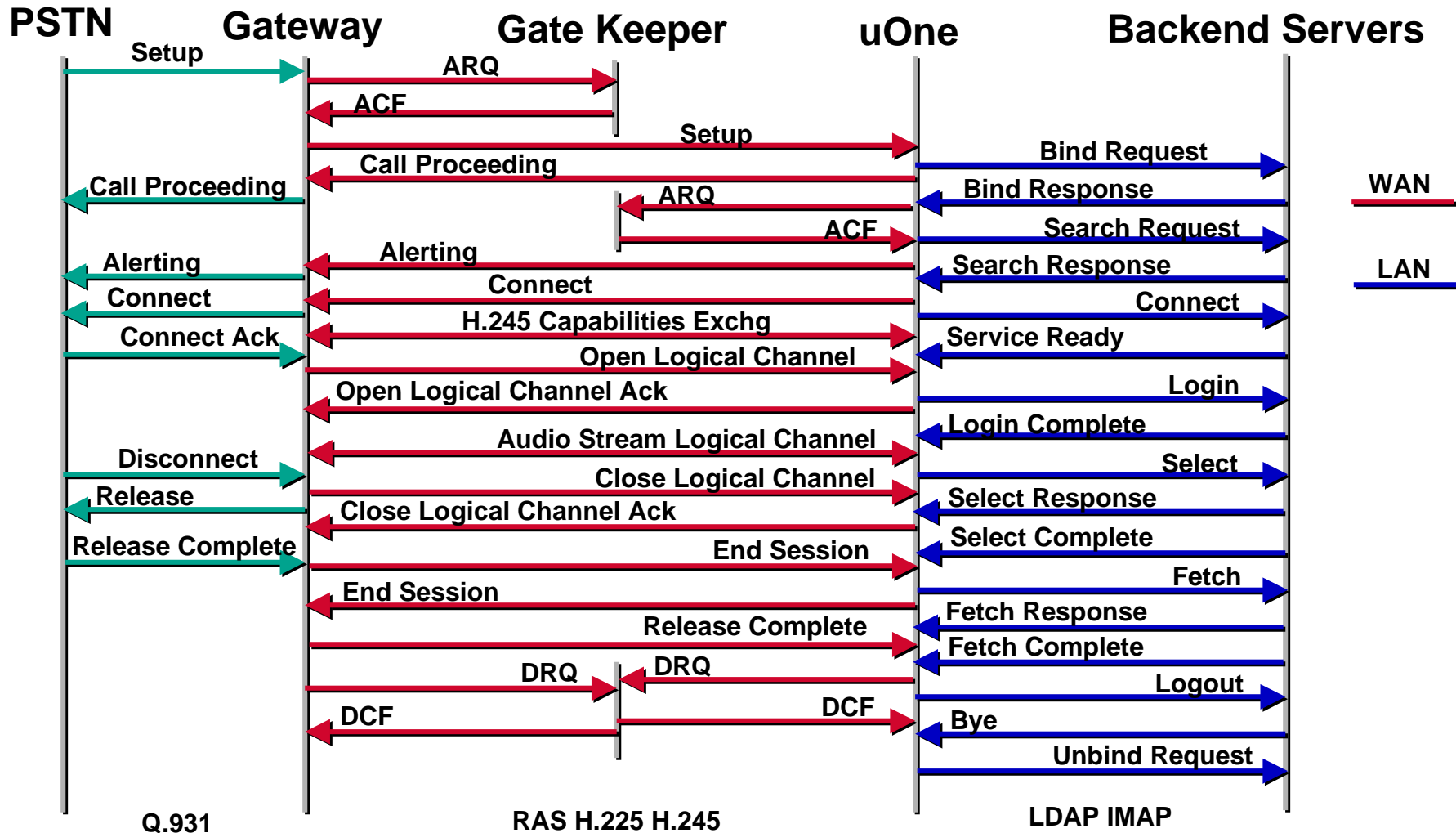
Message retrieval and response times

Dial Internet Access Fully Centralized

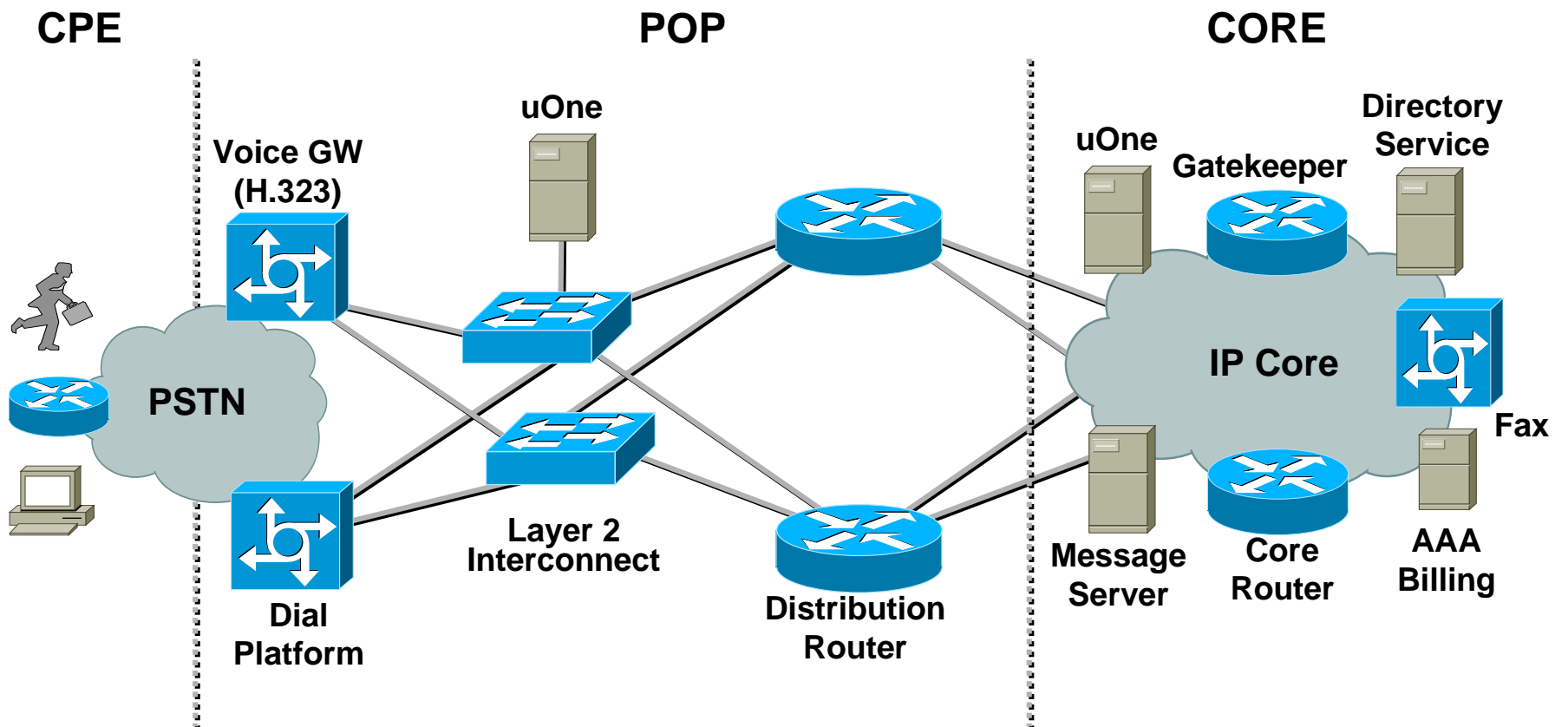


No uOne Components at POP

Dial Internet Access Fully Centralized

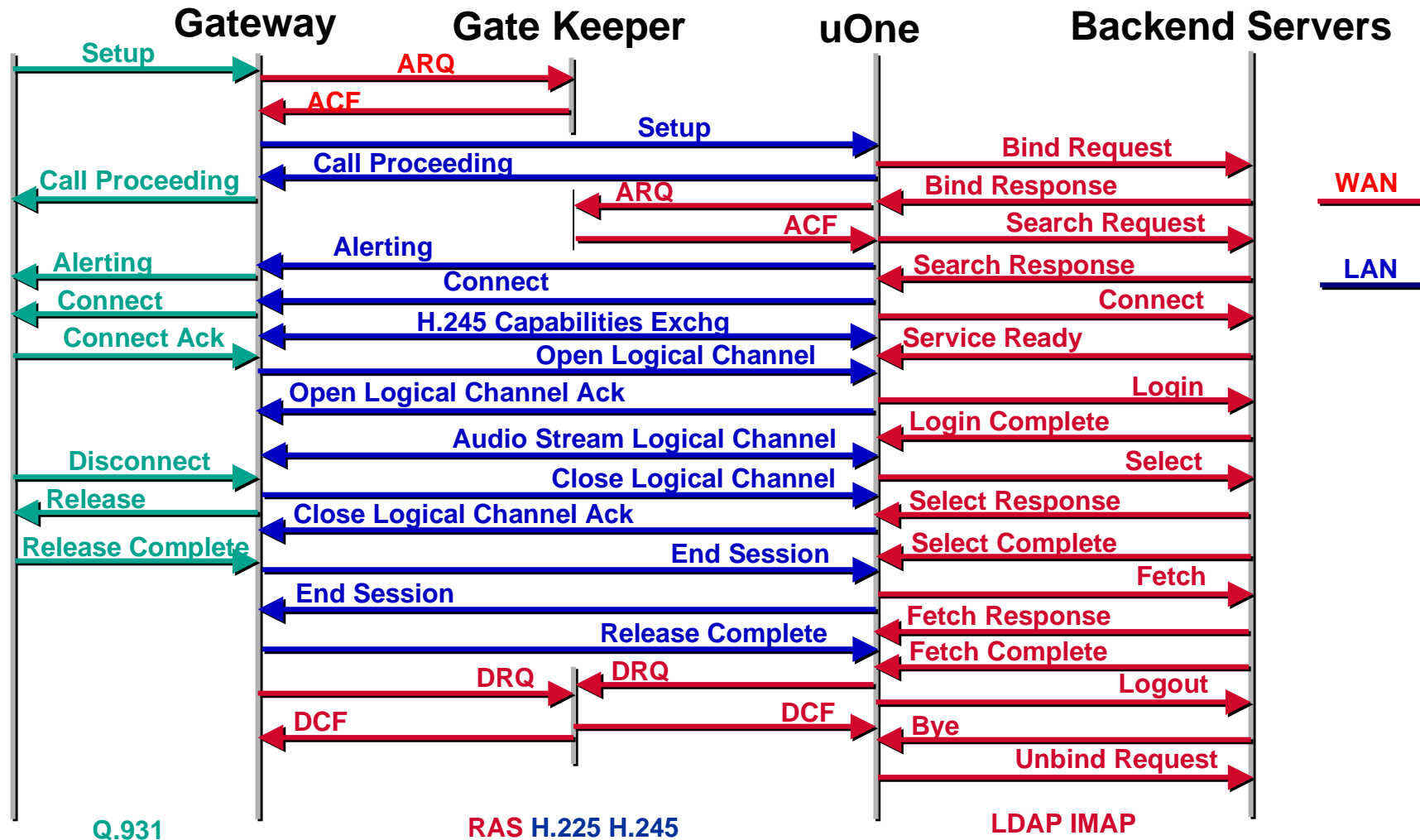


Dial Internet Access Partially Centralized

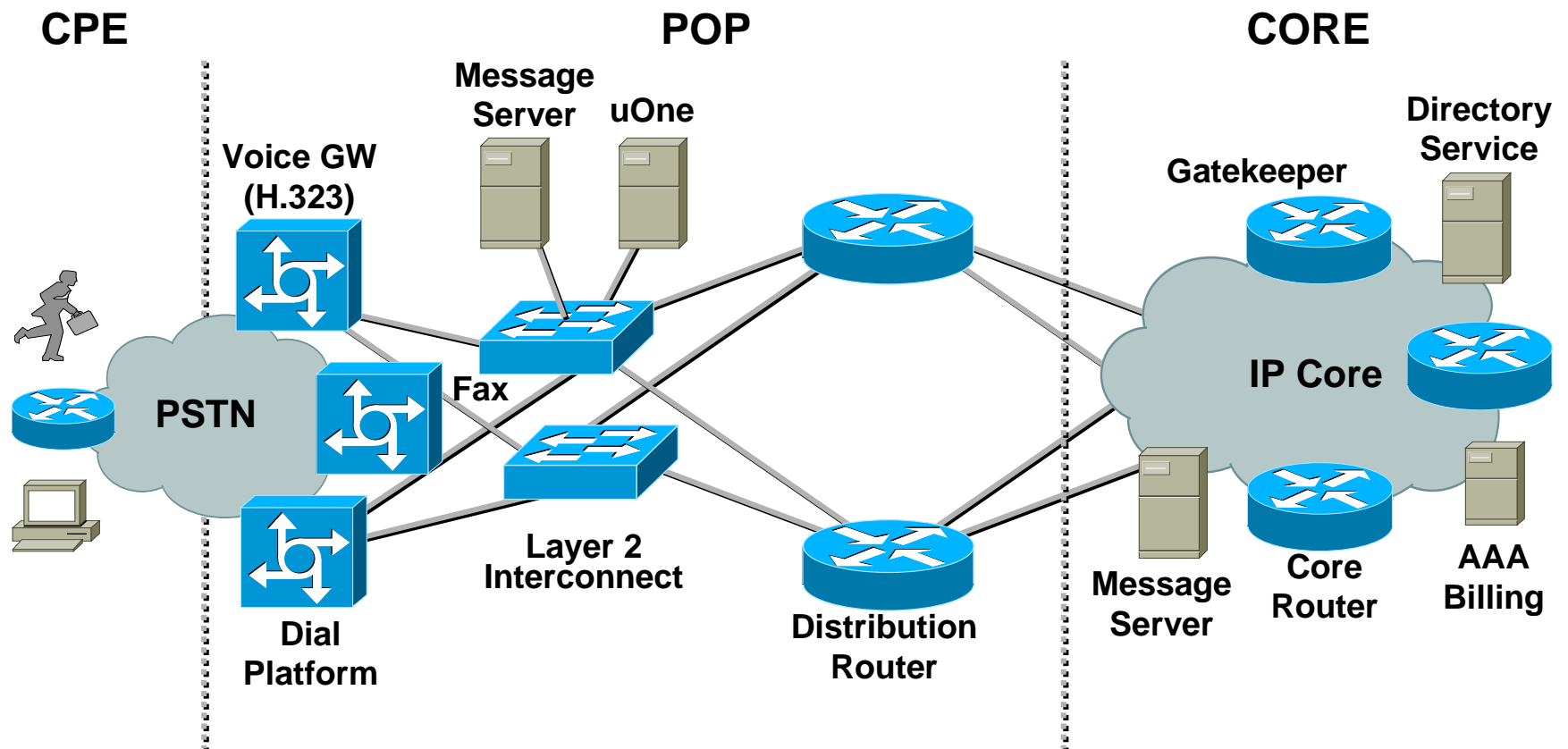


Centralized Back End Services

Dial Internet Access Partially Centralized

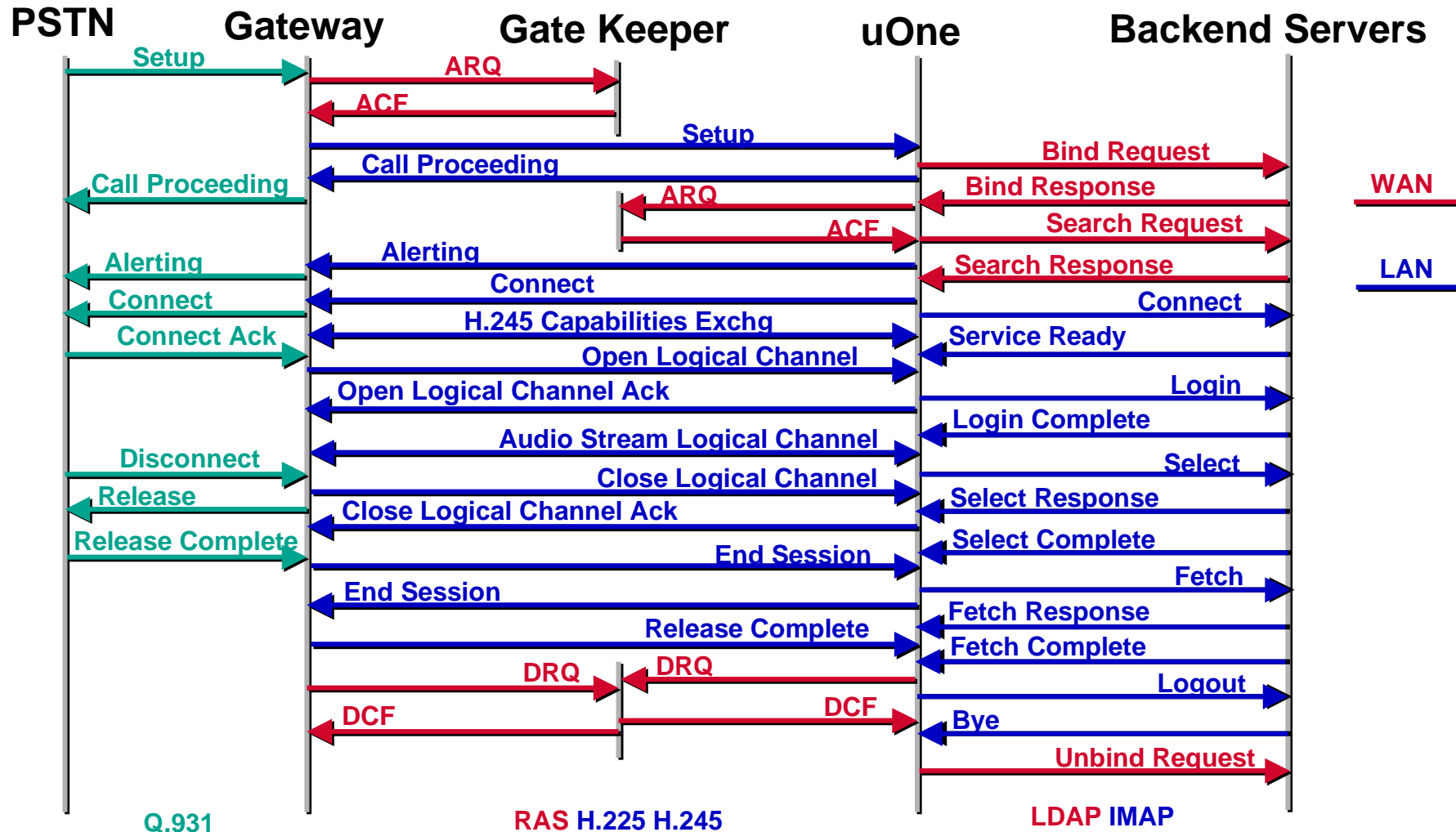


Dial Internet Access Partially Distributed

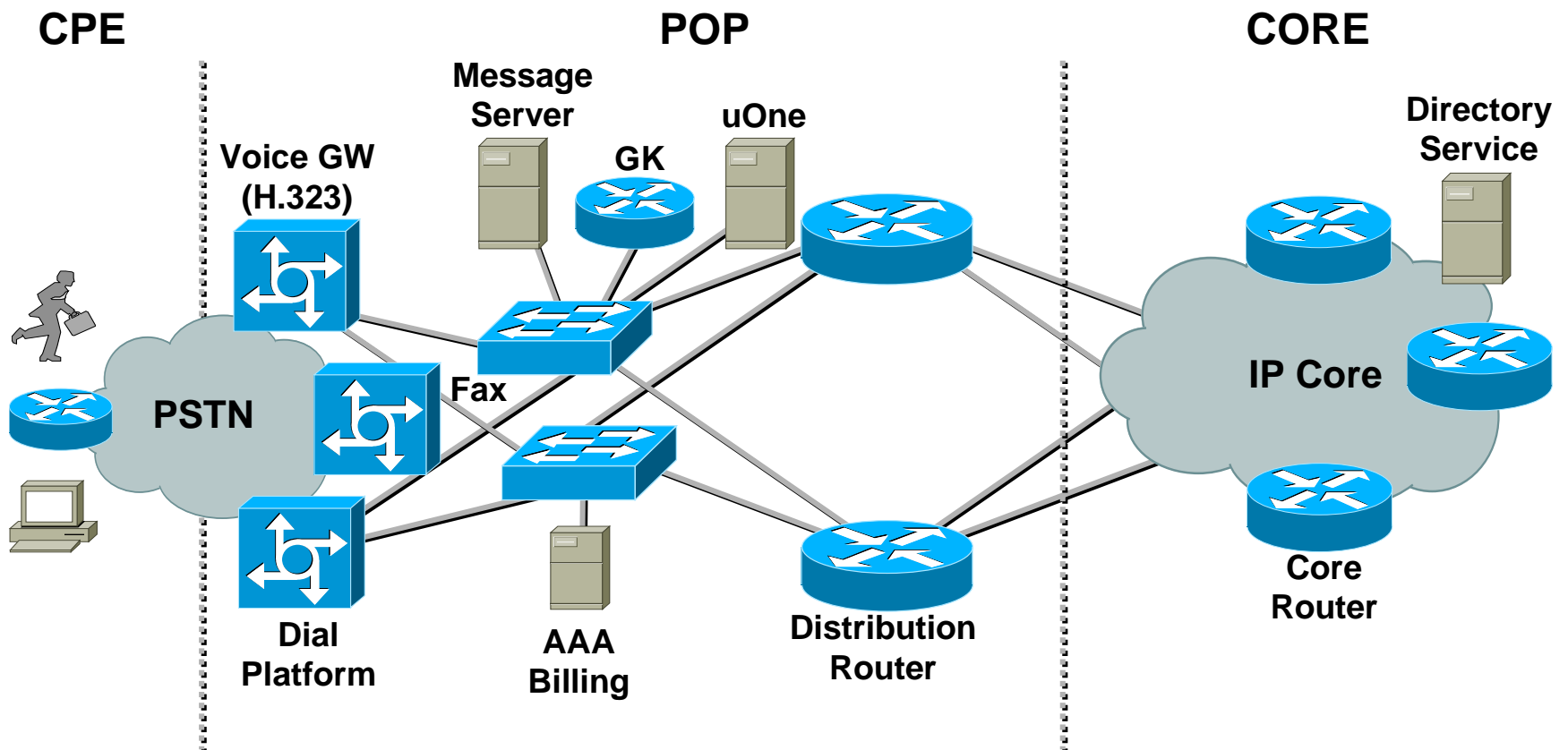


uOne and Messaging Server at Each POP

Dial Internet Access Partially Distributed

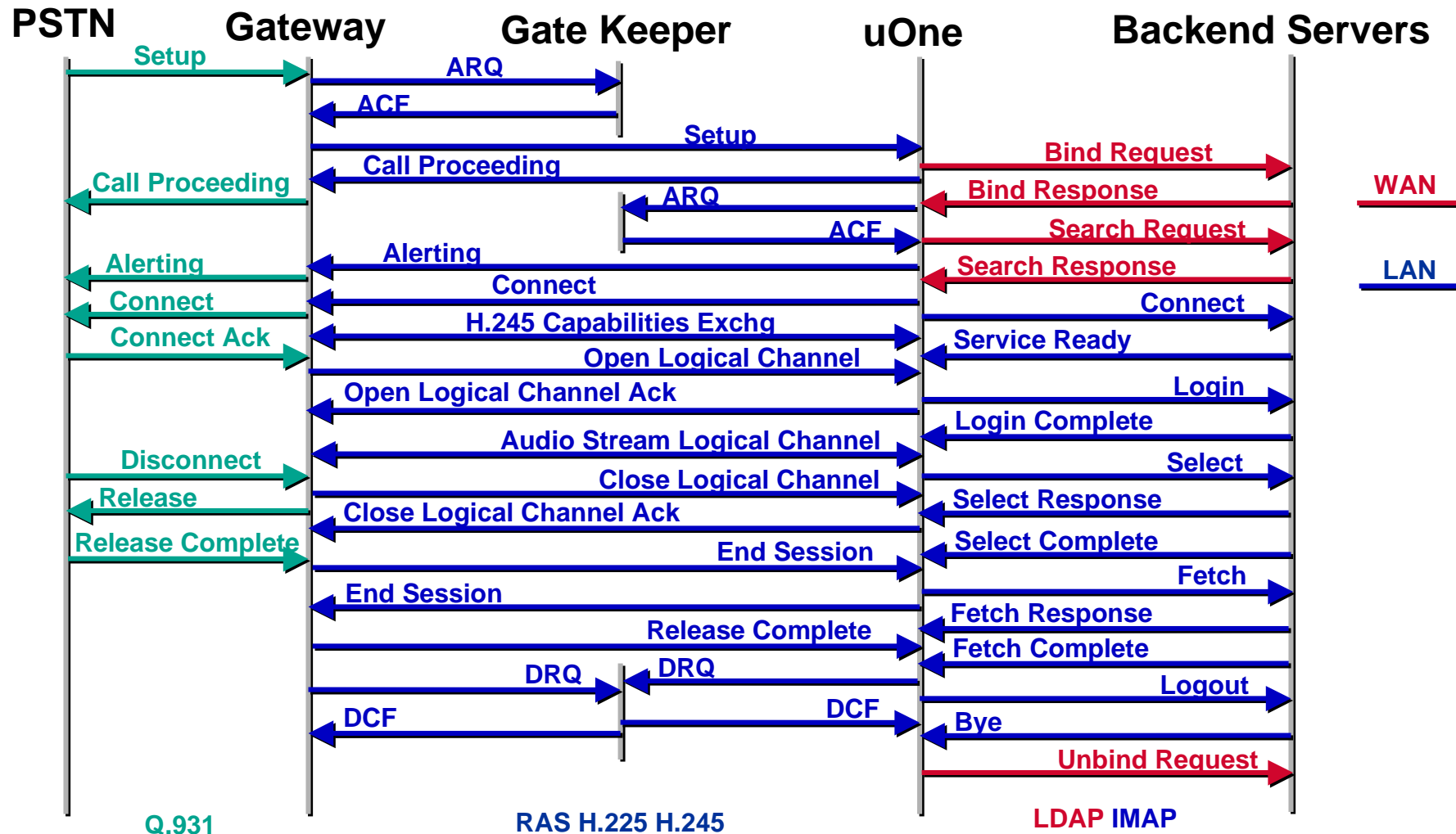


Dial Internet Access Fully Distributed



Local GK Zone for Each POP

Dial Internet Access Fully Distributed



Dial Internet Access Deployment Summary

Quality Feature	Fully Centralized	Partially Centralized	Fully Distributed	Partially Distributed
Call Setup Time	Delays Possible	Very Good	Best	Very Good
Voice Quality	Degraded Voice Possible	Very Good	Very Good	Very Good
Authentication	Good	Very Good	Very Good	Very Good
Message Response	Good	Good	Very Good	Very Good

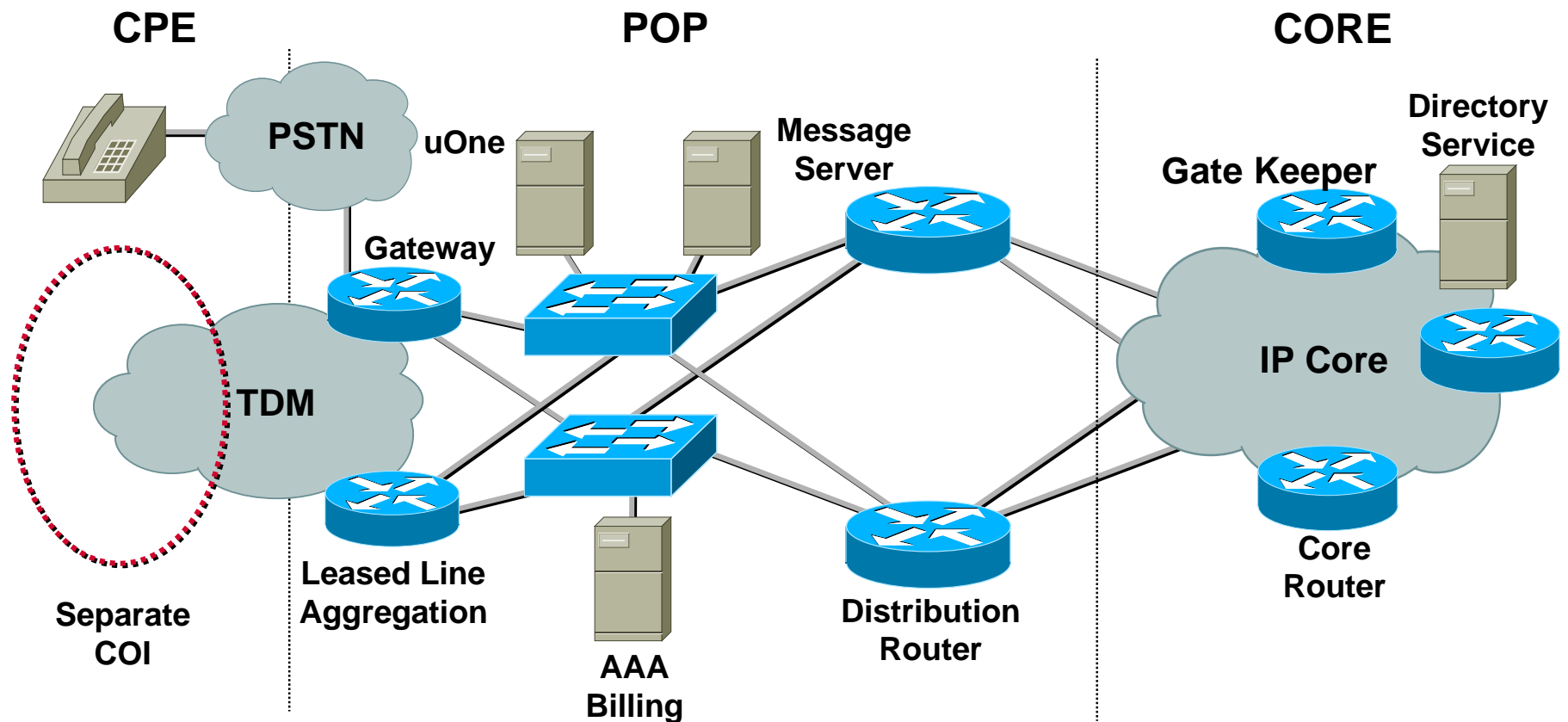
Call Setup Time: Time taken to setup call and hear ringing at the far end

Voice Quality: Quality of messages being played back from uOne

Authentication: Time subscriber has to wait for the system after entering user ID and pin

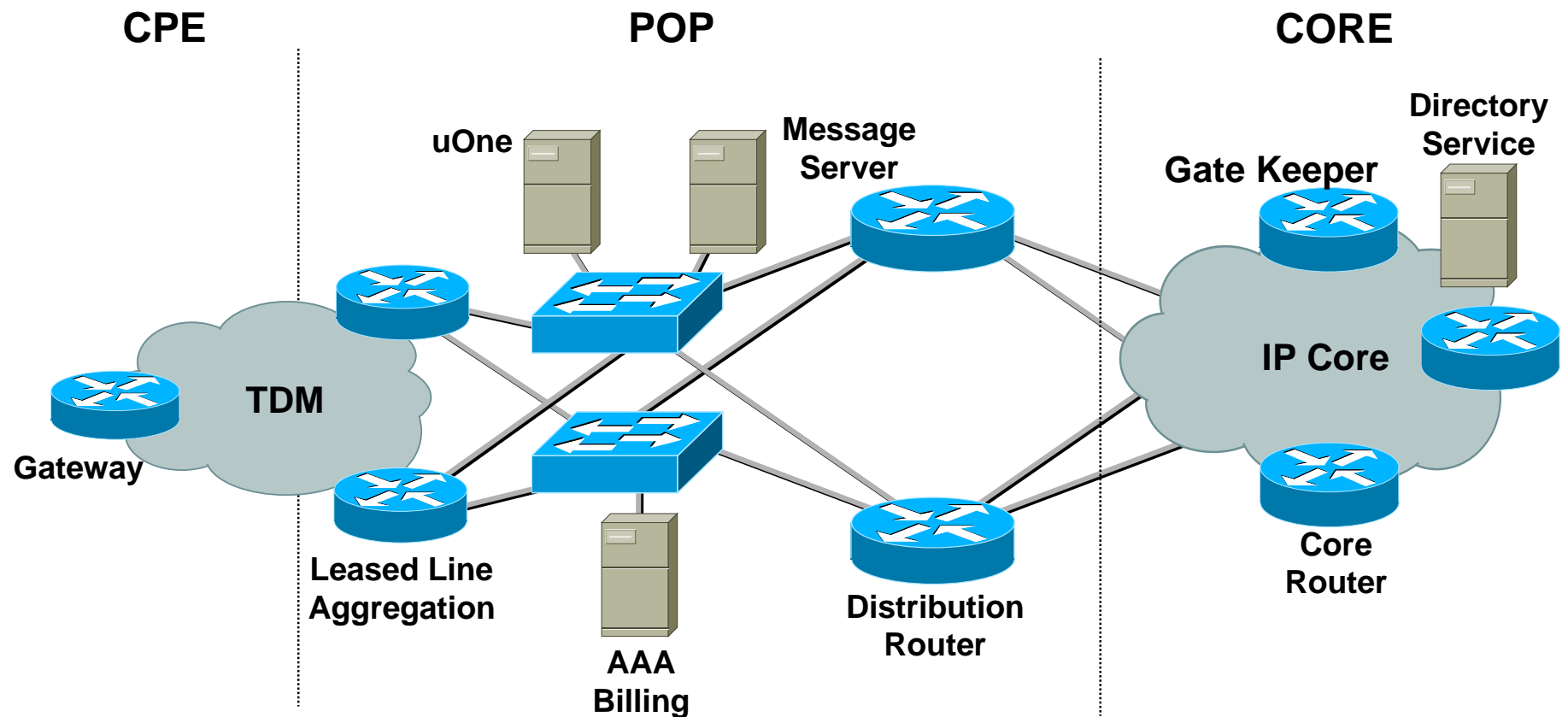
Message Response: Time subscriber has to wait to hear message after that message has been selected.

Dedicated Internet Access



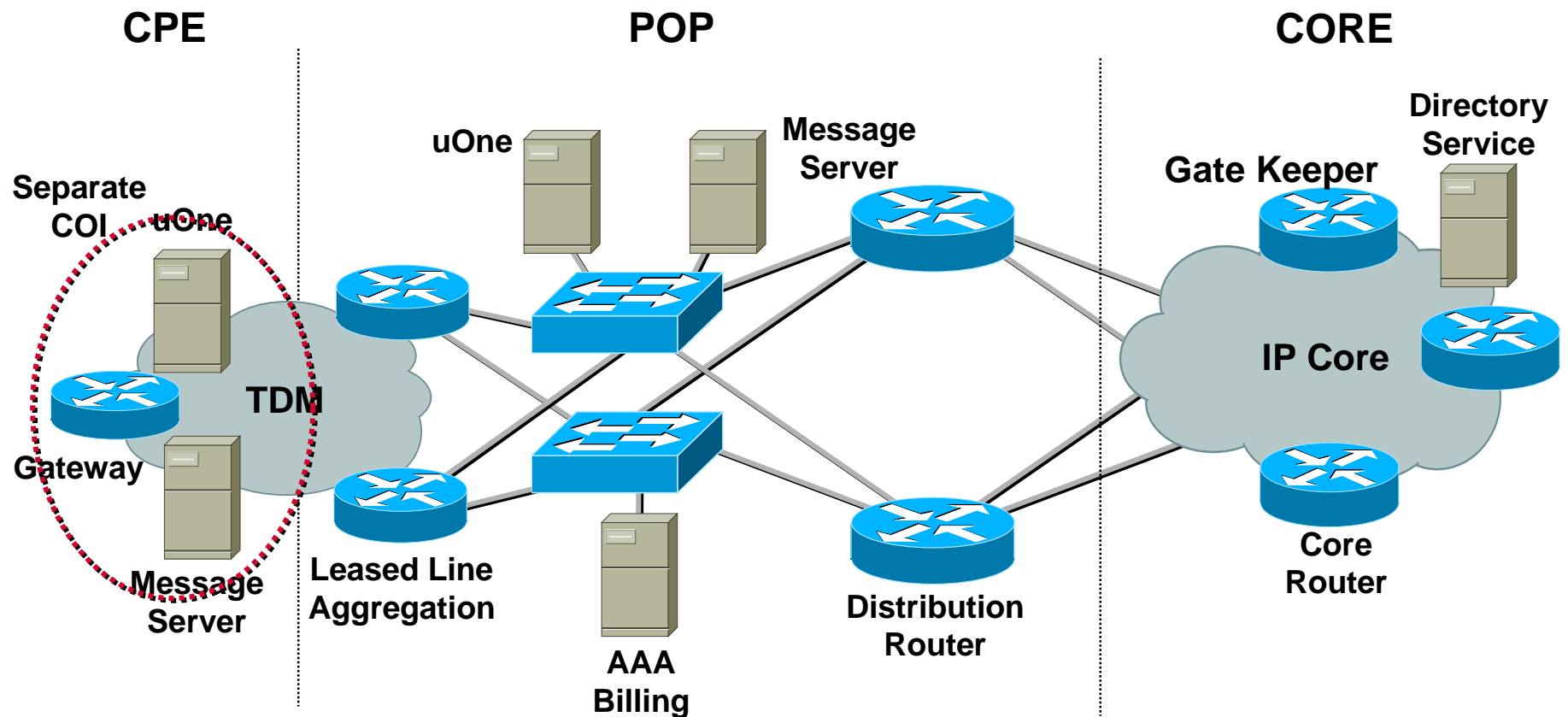
Corporation Share uOne and Gateway Resources in POP

Dedicated Internet Access



Corporation Share uOne Resources in POP and Has Local Number Access

Dedicated Internet Access



Corporation Has Dedicated uOne Resources



AVVID



Cisco AVVID Voice Attributes

- **Already delivered :**
 - Highly scaleable architecture**
 - High availability**
 - Distributed architecture**
 - Lower total cost of ownership**
 - Rapid application deployment**

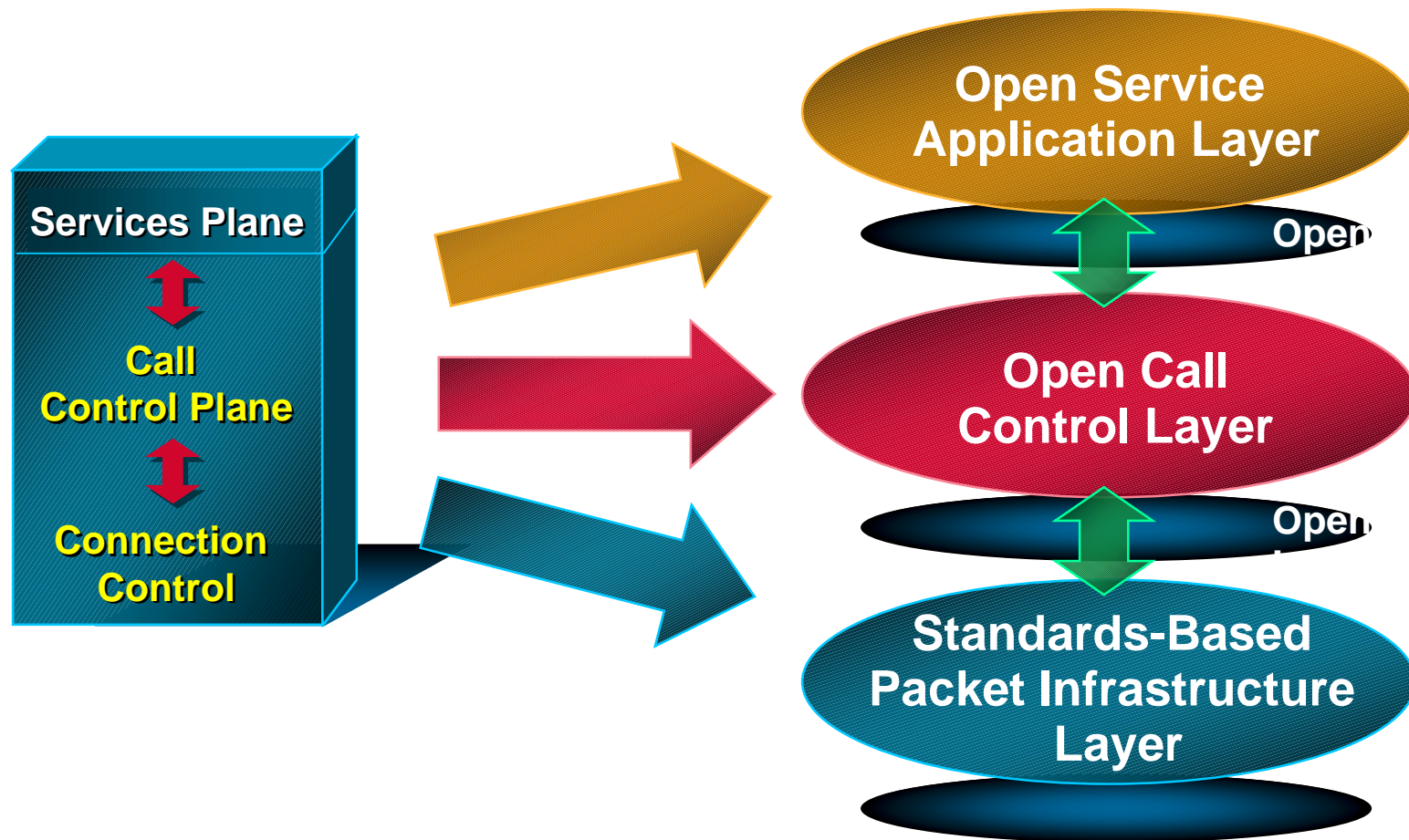


New Cisco AVVID Voice Attributes

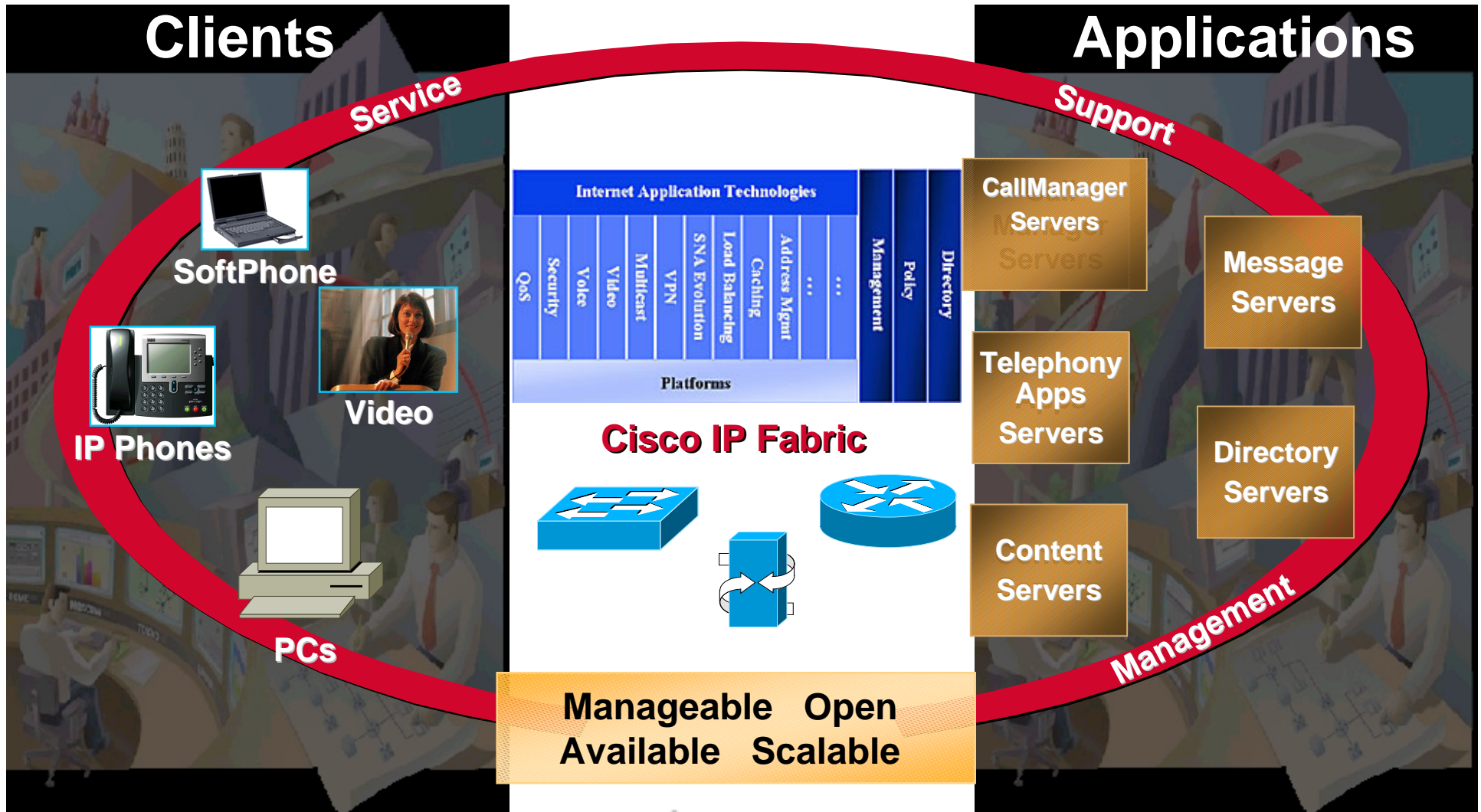
- **New World Applications**
 - Improved customer care
 - Enhanced workforce efficiency
 - Leverage converged infrastructure
- **Open Systems Partner Program**



Cisco's Open Telephony Strategy

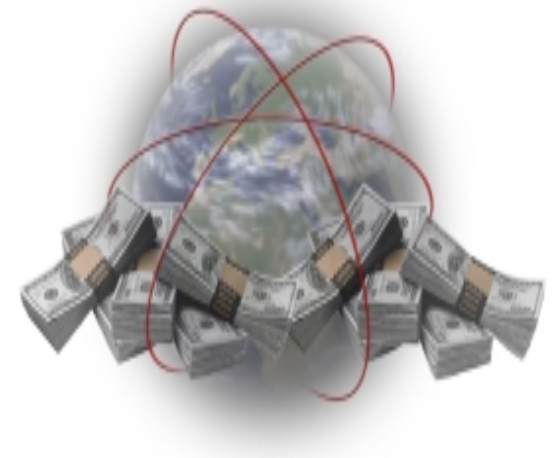


Cisco AVVID - an End-to-End Architecture



IP Application Deployment Advantages

- Scalable applications span time and distance
- Ease of integration
- Rapid deployment
- Leverages existing applications
directories, web sites, message stores, calendars



Announcing AVVID Applications

- **IP Phone 7960 Display Services**
- **IP SoftPhone**
- **IP Auto-Attendant**
- **IP Interactive Voice Response**
- **Cisco Web Attendant**
- **Cisco uOne Messaging for the enterprise**
- **IP Contact Center—IPCC**

AVVID Architecture

Partners

Applications

Call Processing

Infrastructure

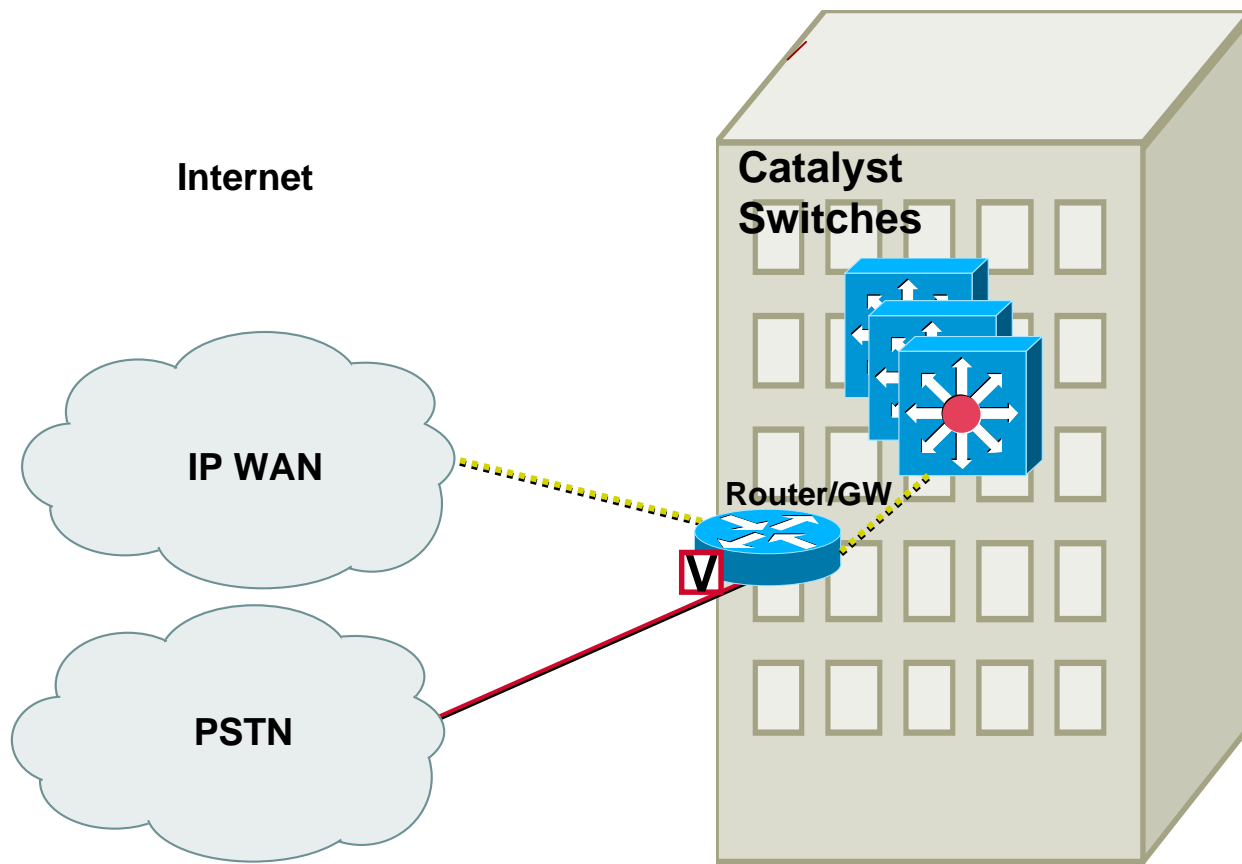
Clients



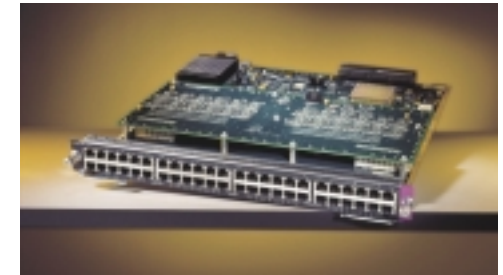
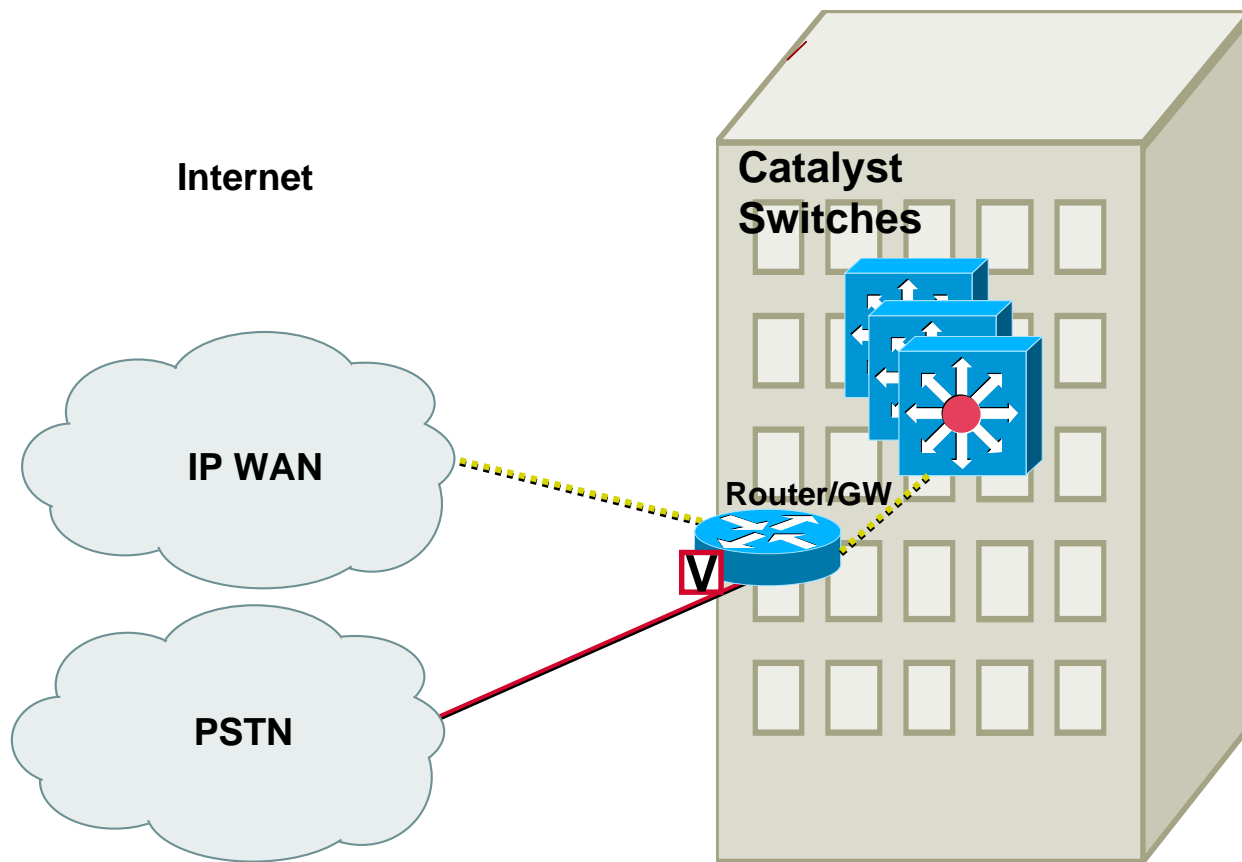
**The World Is Now Global—
All Applications Must Span Time and Distance**

cisco.com

Adding Voice and IP applications

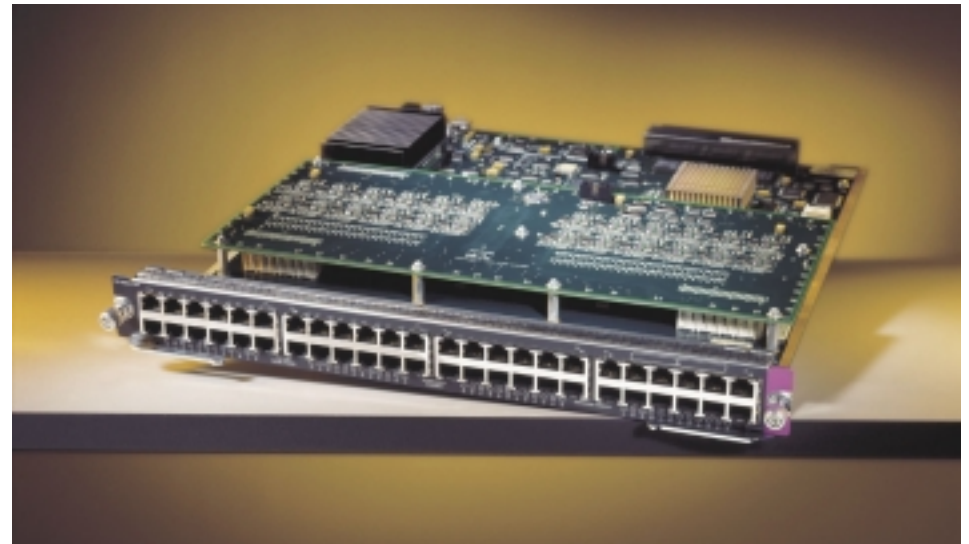


Voice Enabling the Network



Cisco Catalyst 4K/6K Ethernet Line Card with Inline Power Option

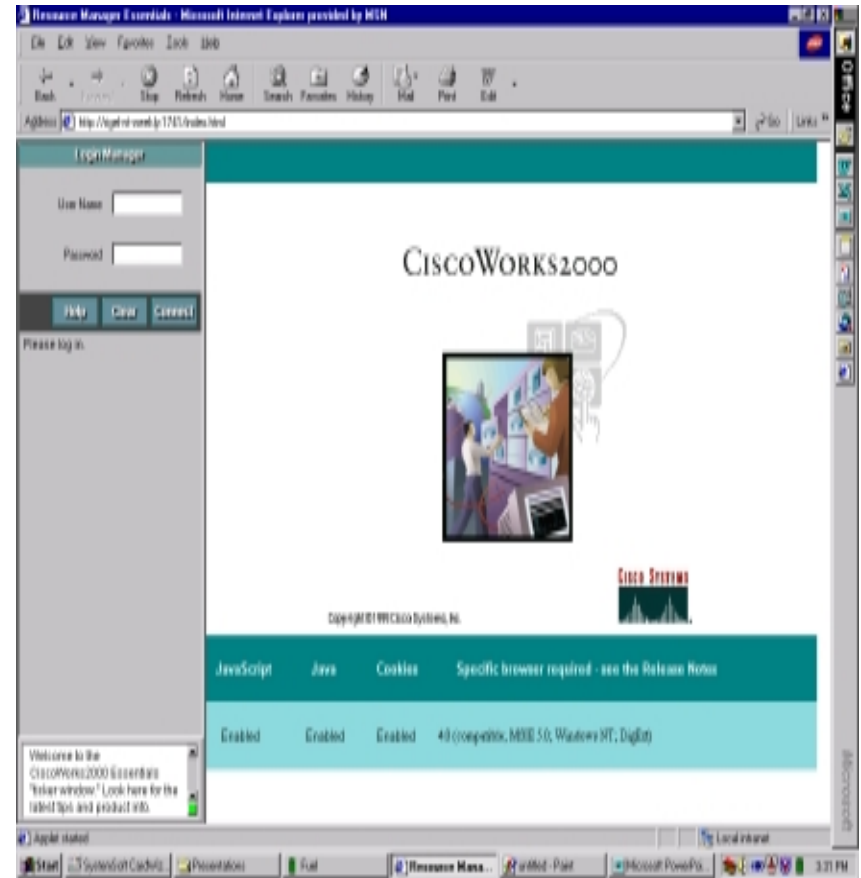
- 48-port 10/100 Ethernet switch line card, with optional Inline Power daughtercard
- Provides Inline Power to IP phones
- Intelligent auto-detection of IP phones
- Manageable via SNMP



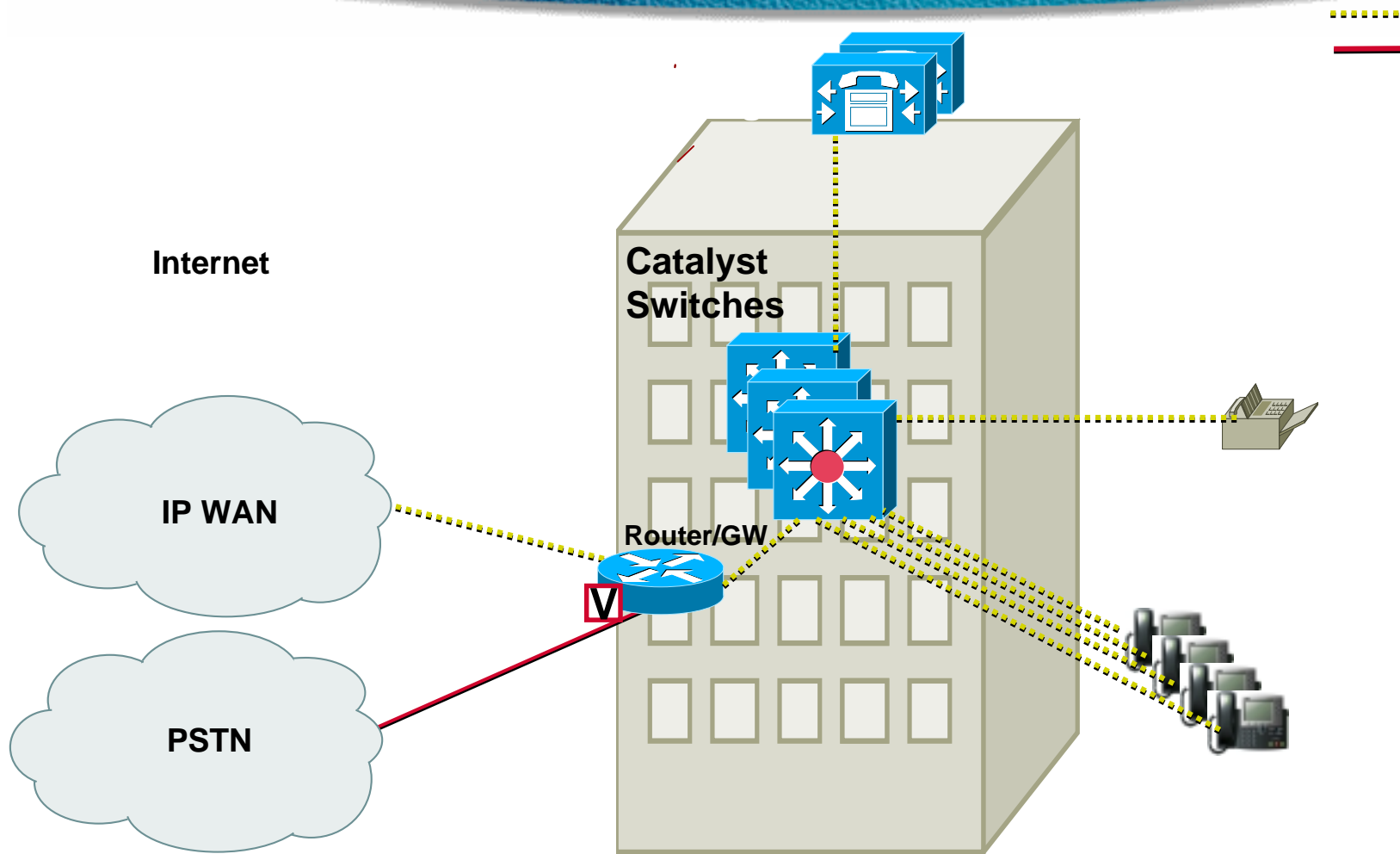
WS-X6348-RJ-45

Single System Management

- **Common management interfaces**
SNMP, XML, Telnet, TFTP
- **Common access via IP**
- **Common end host tracking:**
Track all end host devices within one system (phone, laptop, workstation)
- **Reduced operational costs**
No provisioning of adds, moves and changes



IP Telephony

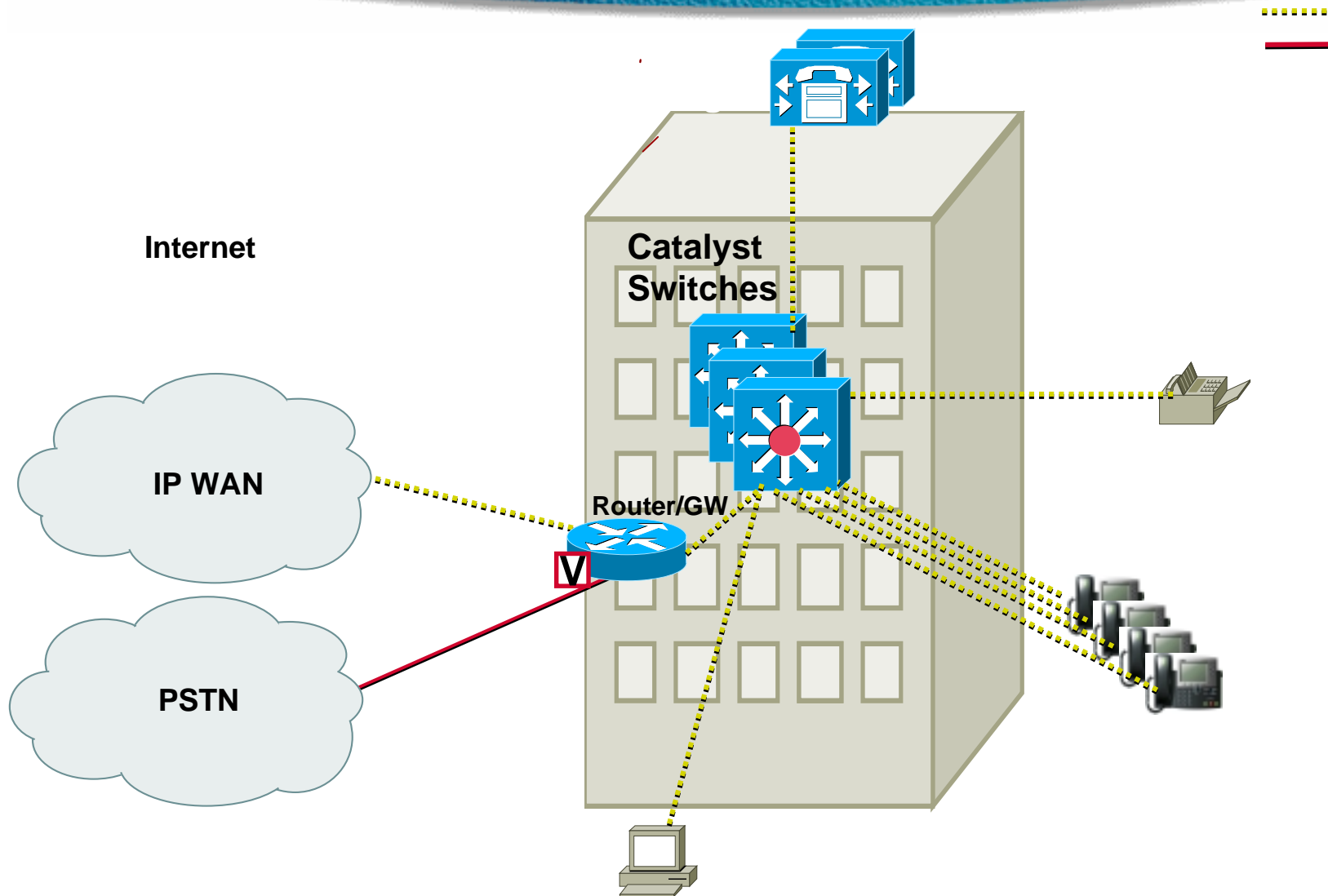


Individual Productivity IP 7960 Display Applications



- IP Telephony Appliance
- Corporate directory integration via LDAP
- Web site integration via XML
- Personalised menus via softkeys
- Context-sensitive help
- Open interface for 3rd party app development
- Intelligent, programmable services

Add Web Attendant

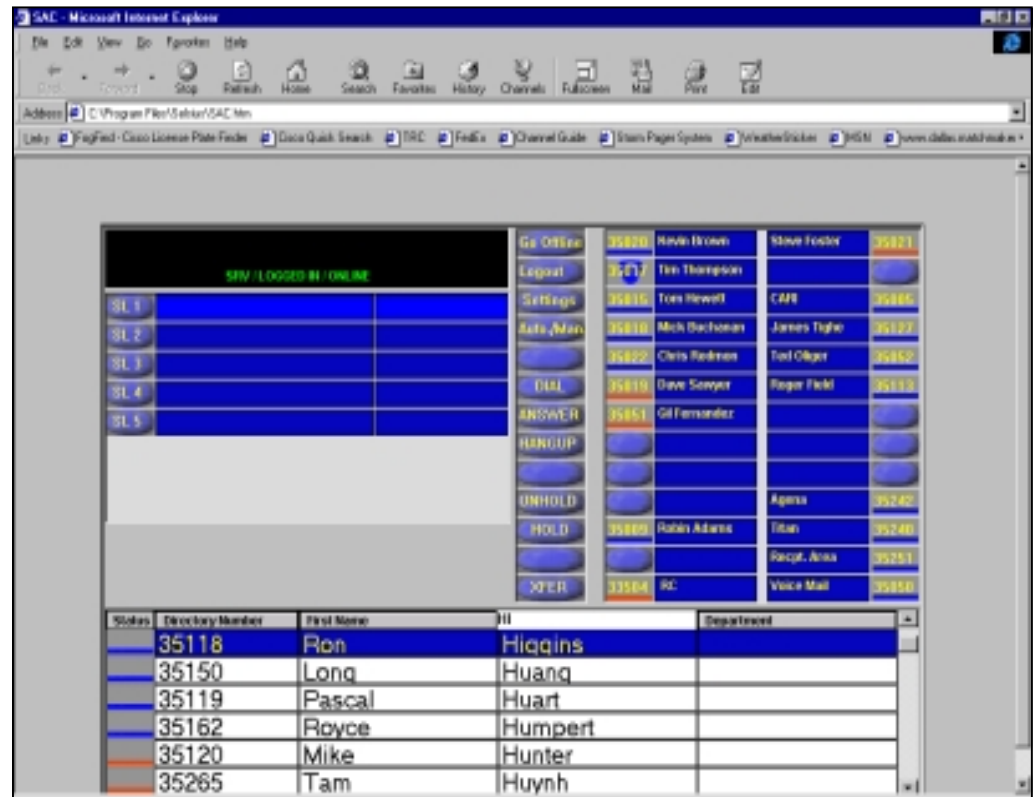


Cisco Web Attendant

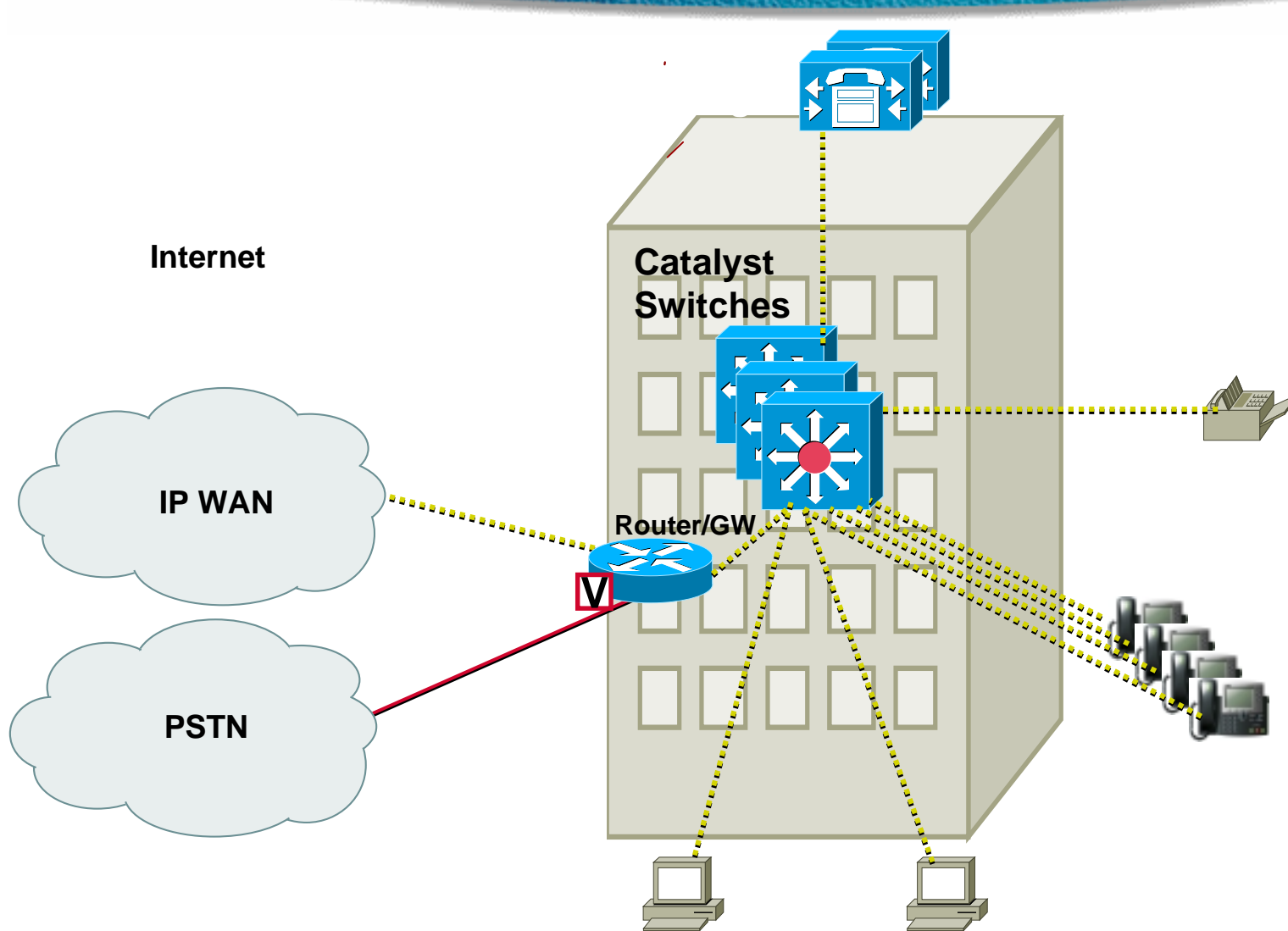
- Allows receptionists to use any phone as attendant console
- Drag and drop users via LDAP
- Benefits:

Expensive hardware not required

Can be run from any connected desk



Add IP Softphone

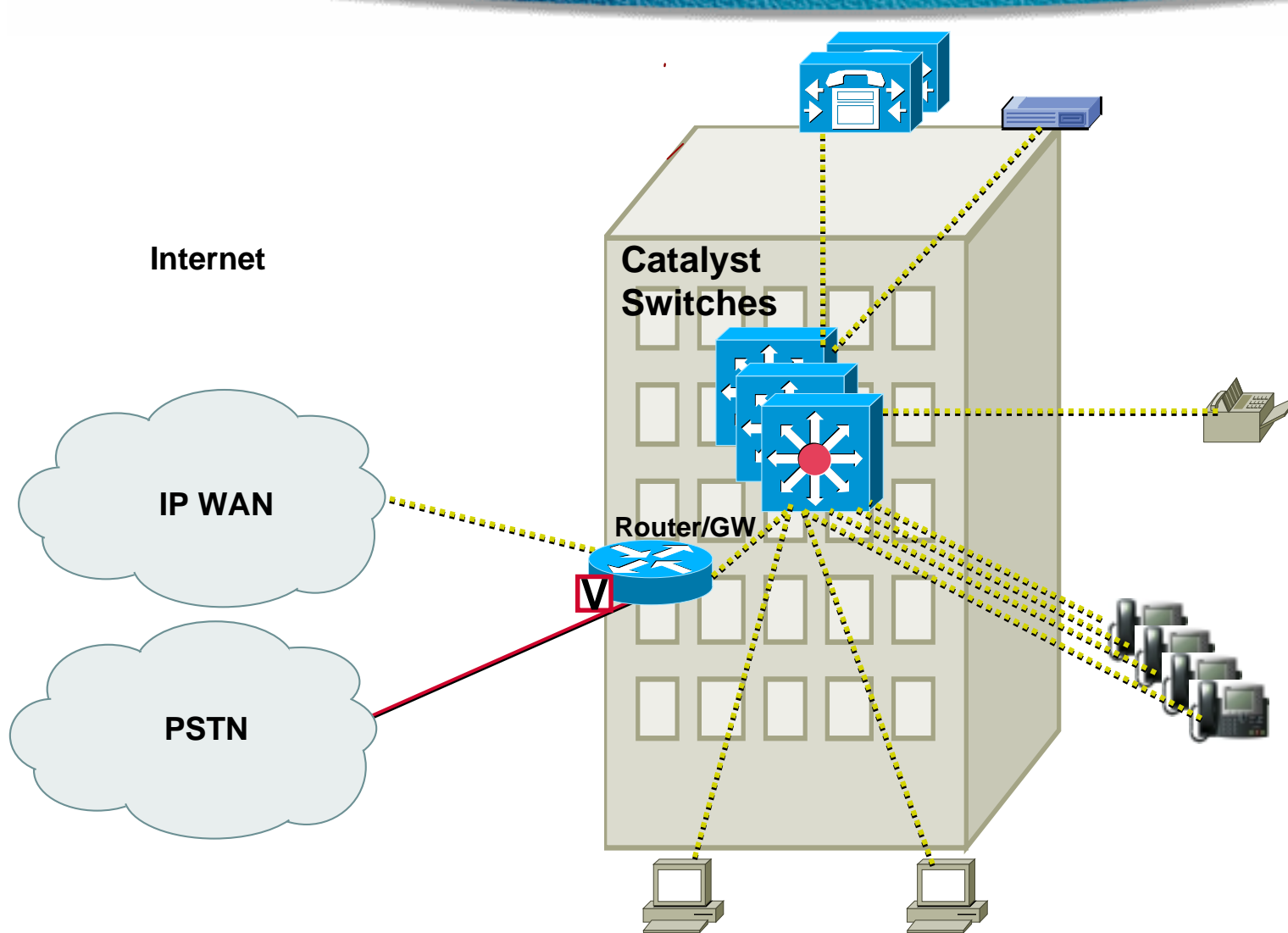


IP SoftPhone—Fingertip Features

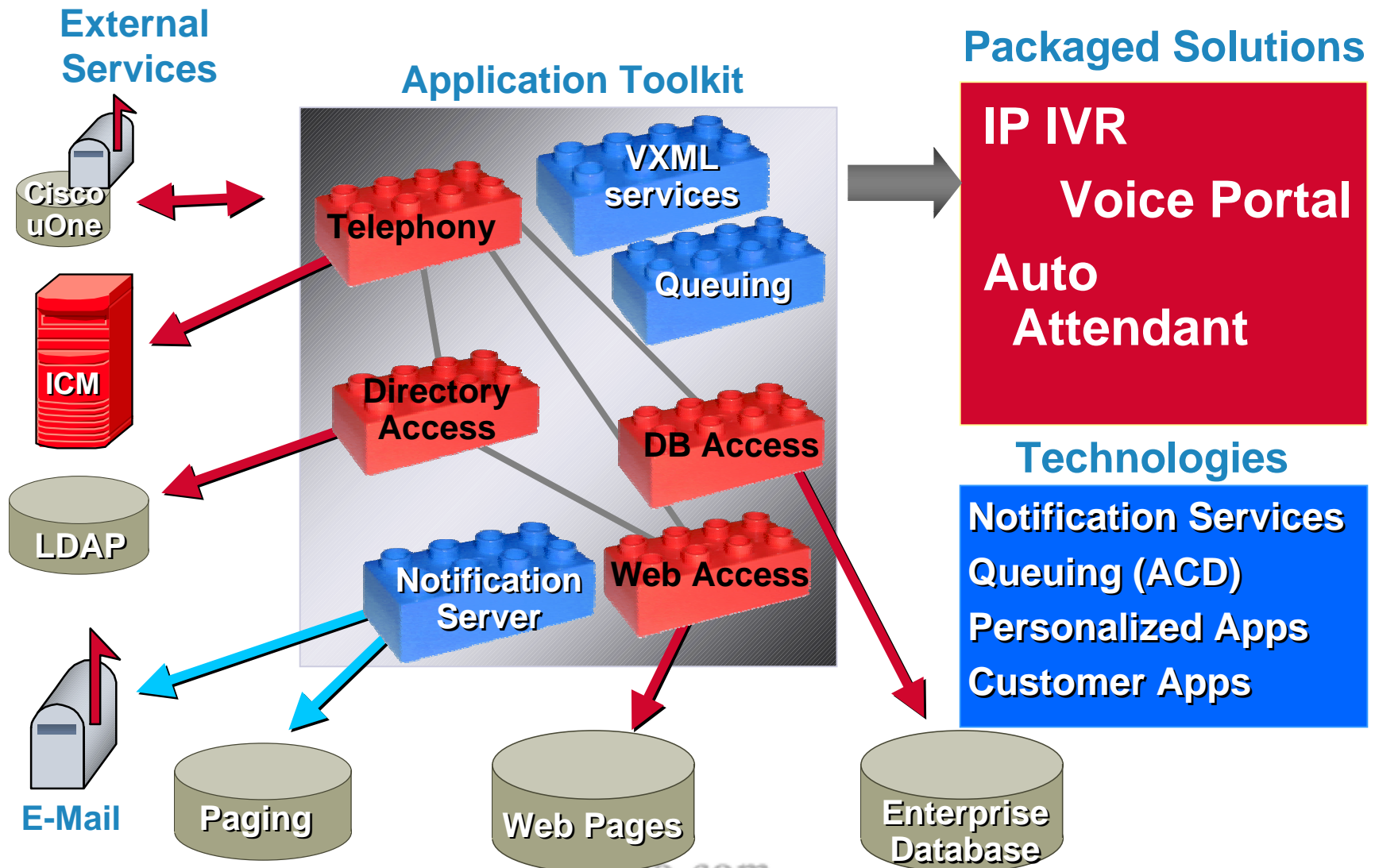
- **Media termination allows office extension mobility**
- **Improved productivity**
- **GUI based interface for phone control (drag and drop)**
- **Easy feature access**
 - One click conference, transfer & collaboration**
 - NetMeeting**
- **Directory integration**
 - Personal and Public (LDAP)**
 - Dial by name/email address**
- **Standards based TAPI integration**



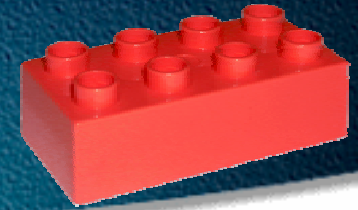
Add IP Auto Attendant and IP Interactive Voice Response



E-Services Application Engine



Rapid Update Productivity Voice Portal Solution



Call Manager IP IVR



IP Intranet



Cisco Stock
Quote



Press #1 to Hear
Cisco Stock Quote

- Extracts XML information from web page into IP IVR
- Benefit

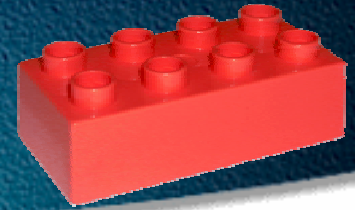
Only one place to configure
and maintain data

Consistency

Lower admin costs

cisco.com

Notification Services



- **Automatic notification**
 - triggers based on back-end system changes
 - thresholds pre-defined
- **Notification can be sent to:**
 - phone display
 - beeper
 - remote device
- **Ease of administration**
 - one place to configure and maintain data
 - provides basic e-CARE
 - easily personalized



CM
Apps

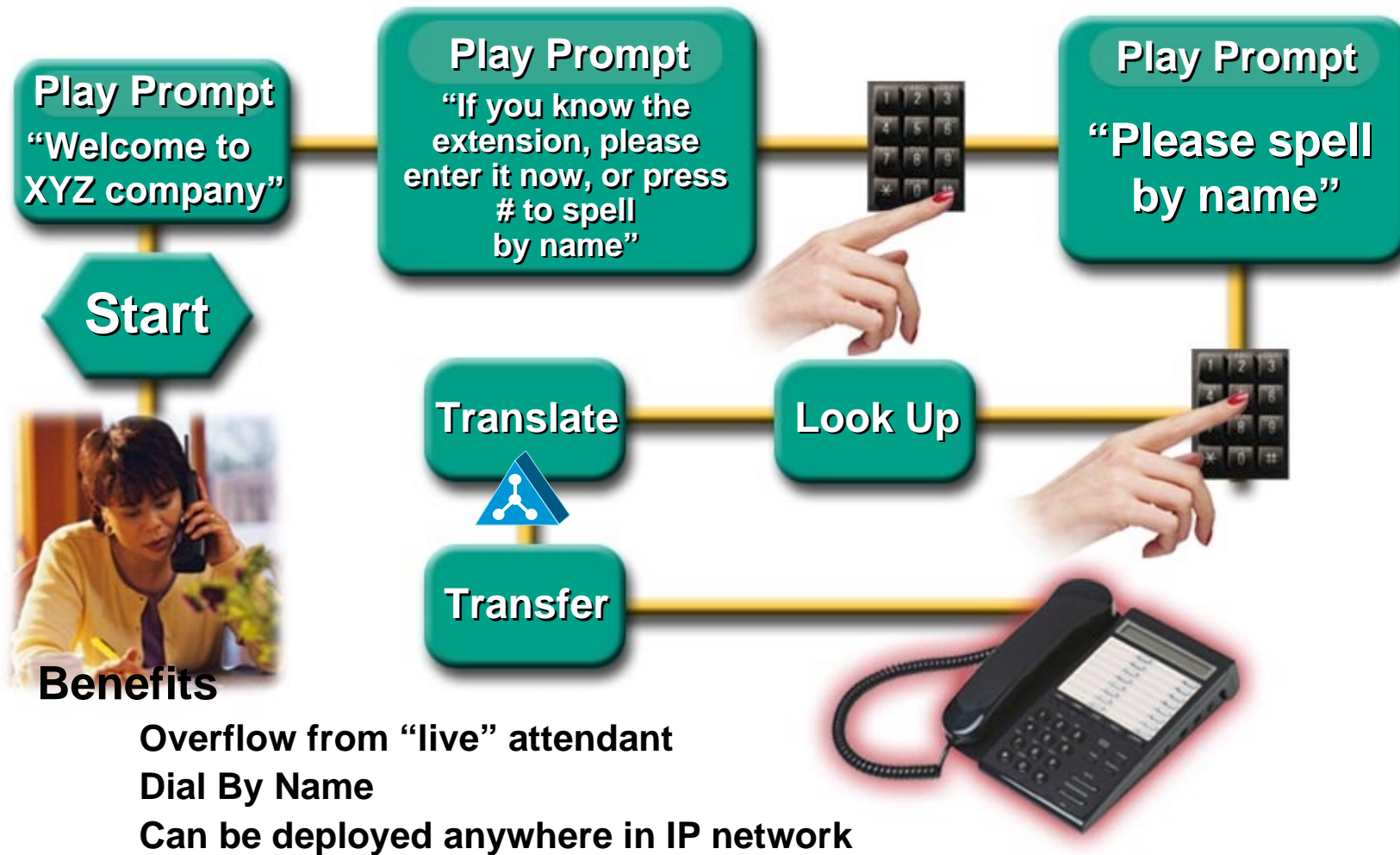


IP Intranet



Page Me when
Inventory Arrives

IP Auto Attendant



IP IVR

- Complete IP-based IVR
- Allows customer to find information: i.e. bank balance

- Benefits:

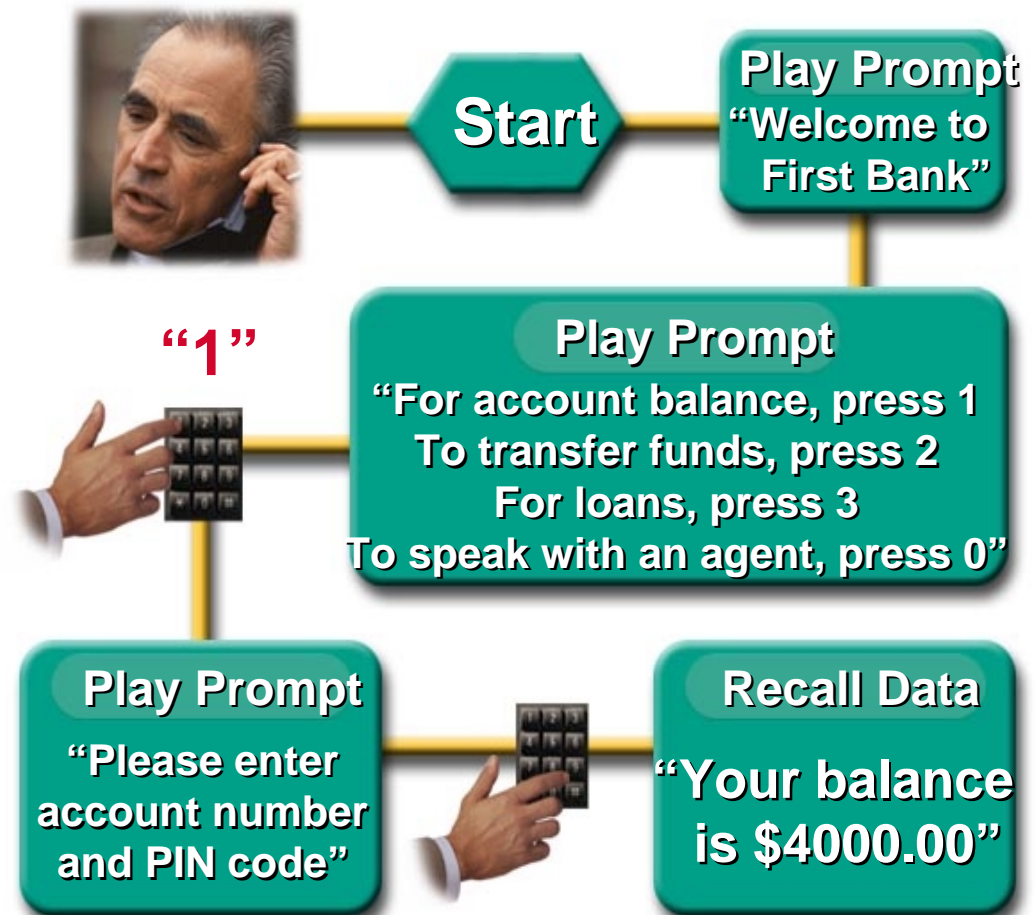
Easy to configure:
drag and drop

Location independent

Many solutions can
use same IP IVR

Can be deployed as
workgroup IVR

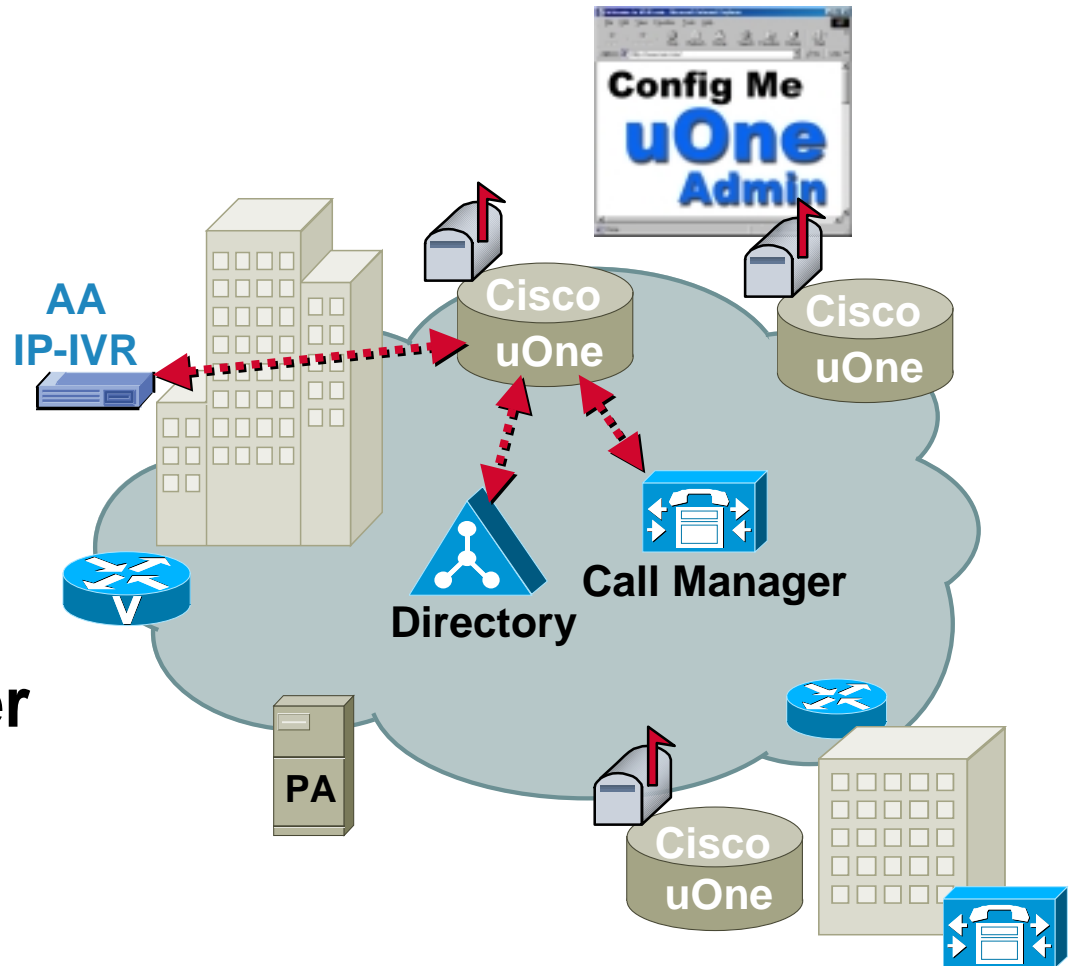
Easily personalised



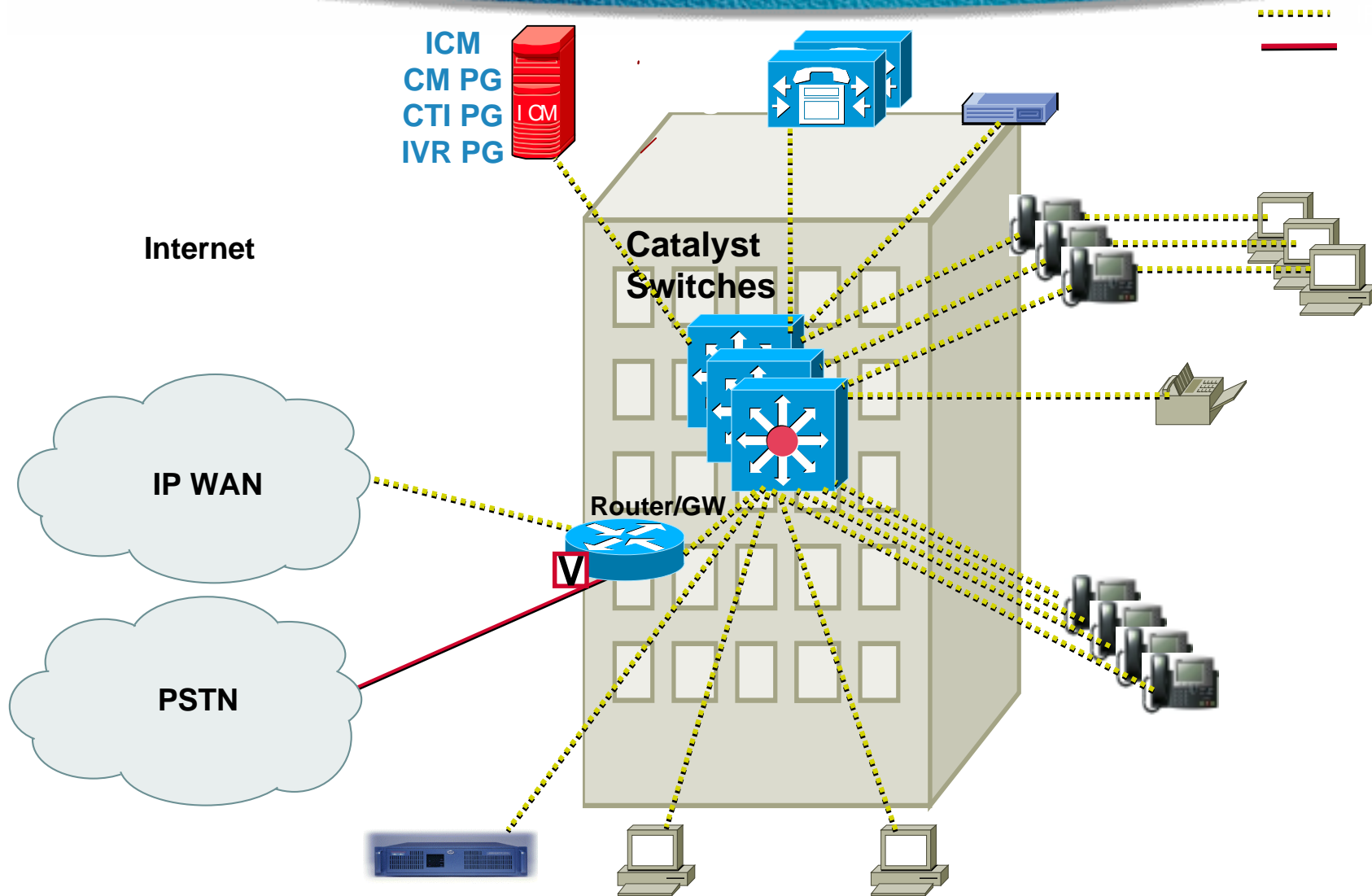


uOne Voice Mail Scalability

- Seamless dial by name, lists, distribution groups with common directory
- 10,000 users in one virtual system, single administration
- Interoperate with other AVVID applications including Personal Assistant (future)



Add ICM



The IP Contact Center

*Intelligent
Contact
Management*

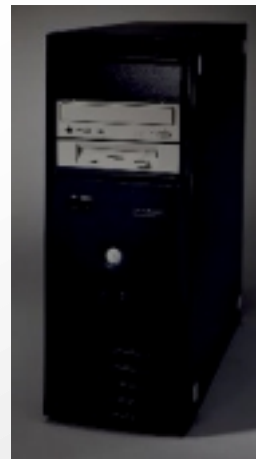


*Agent
Desktop*

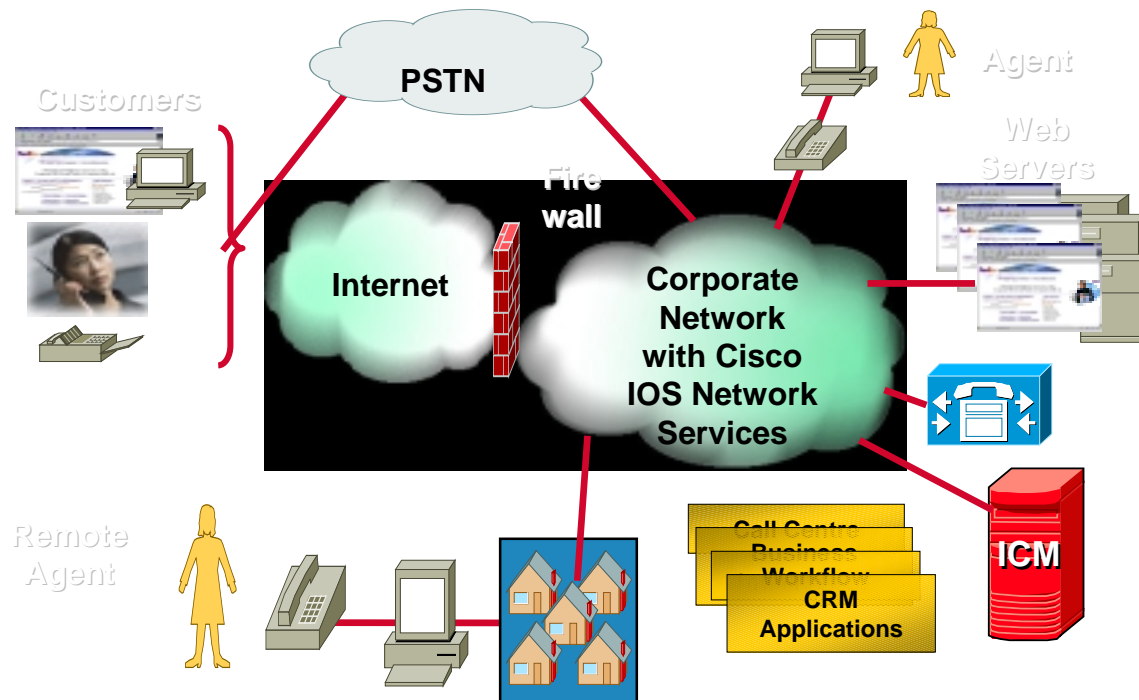
CallManager



*Integrated
Voice
Response*

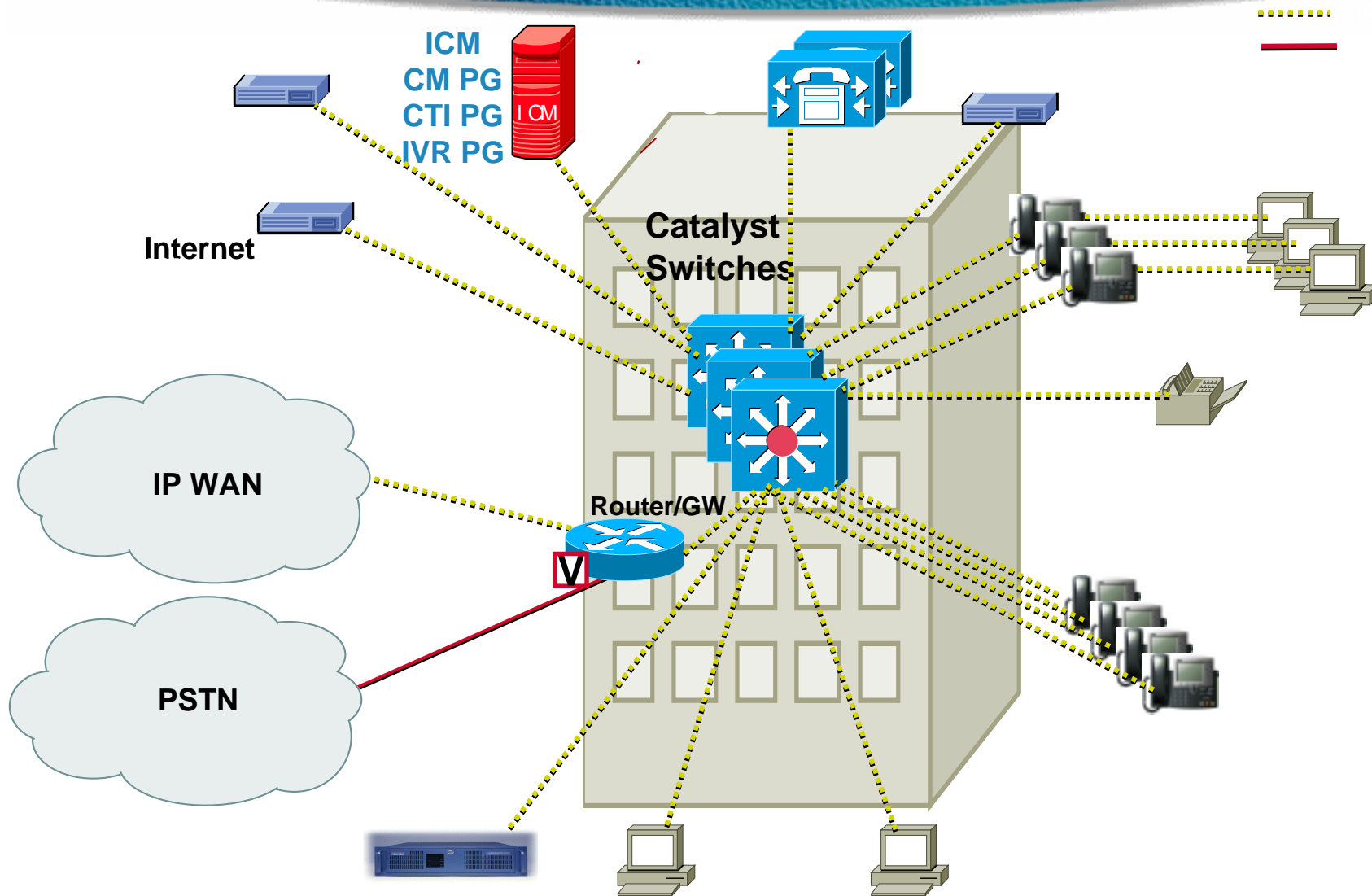


Customer Care IPCC

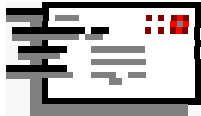


- **Location and media Independence**
- **Rapid application deployment**
- **Lower cost Total cost of ownership**
- **Contact Centre Specialisation**

Add Cisco Collaboration Server and Cisco email Manager



Cisco eMail Manager

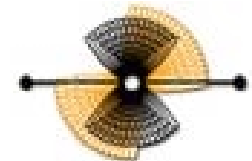


- Flexible, extensible and highly scalable email response management (ERM) system
- Allows companies to process email inquiries based on a flexible, customisable system of rules
- Key features:

Partner Categories



- Unified Messaging: Active Voice, AVT
- Call Centers: Interactive Intelligence, **EasyRun**, **Wicom**
- Call Acctg: ISI, **MIND CTI**, Telemate, Integratrak
- E-Conferencing: Latitude
- E911 Tracking: SCC
- End point access: Symbol, Circa
- Application Server Mgmt:
Integrated Research
- IVR Integrators: Gold Systems, Spanlink, NEC
- Network Analysis: Fluke, Shomiti



INTERACTIVE INTELLIGENCE



GOLD SYSTEMS
Customer interaction solutions that work

AVVID the complete solution

Partners

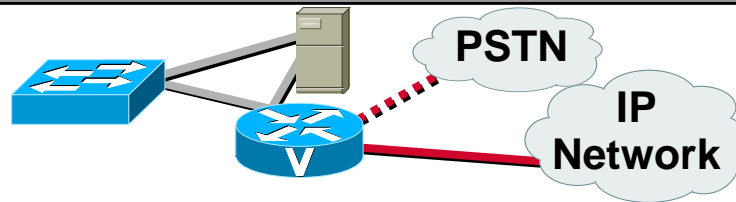
Applications



Call Processing



Infrastructure



- PSTN gateways
- Analog phone support
- DSP farms

Clients



The World Is Now Global—
All Applications Must Span Time and Distance

cisco.com

AVVID Installation in Retail

Sports Soccer

- Leading sports retailer in UK and Belgium
- Manufacturer of Donnay Sports equipment

Business Issues

- Poor communication between head office, 100 branches and regional managers
- Cost of separate voice, CCTV and data systems
- No branch access to corporate network
- Lack of real time information and control
- Unreliable dial up access to EPOS systems

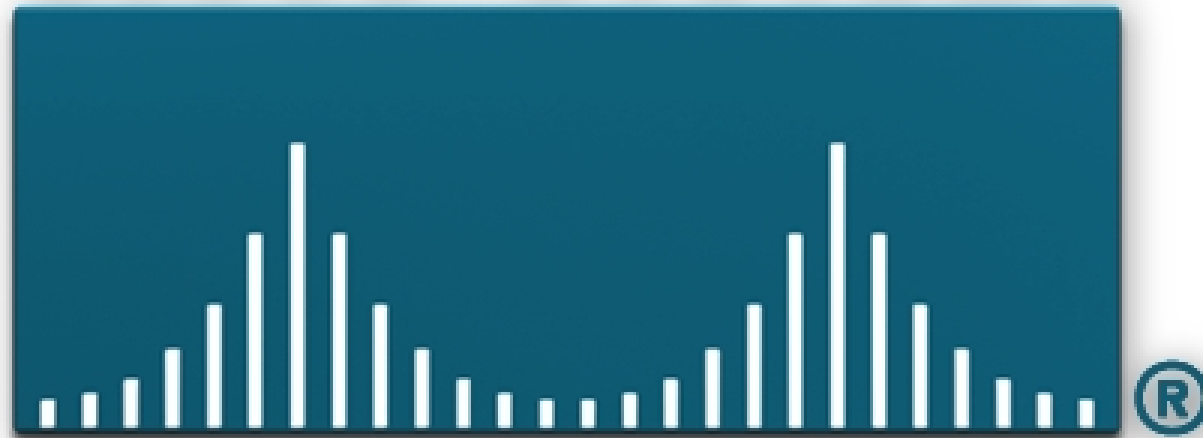


AVVID Summary

- **IP telephony has proven acceptance by Enterprise customers**
- **AVVID delivers high scalability and availability to the enterprise**
- **New World Applications improve productivity, customer care and deliver competitive advantage**
- **AVVID's open approach partner ecosystem enables choice**



CISCO SYSTEMS



EMPOWERING THE INTERNET GENERATIONSM