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Hosted IP Telephony

24 October 2002

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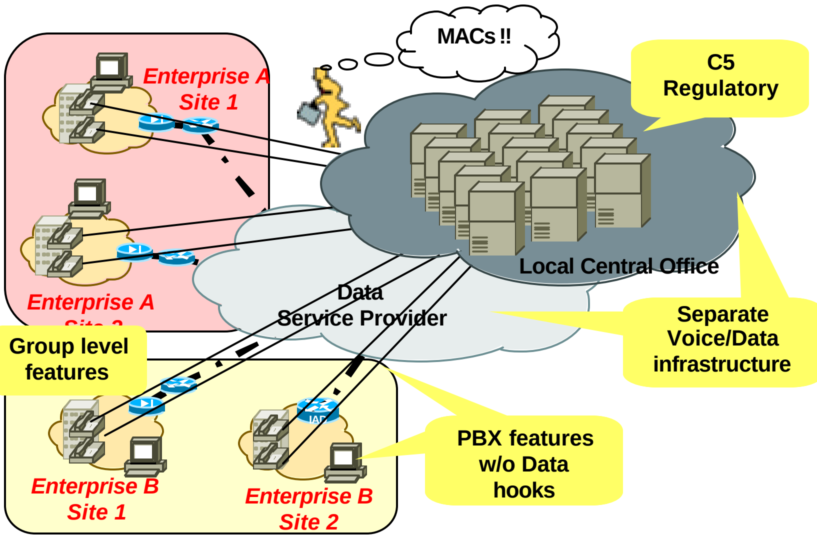
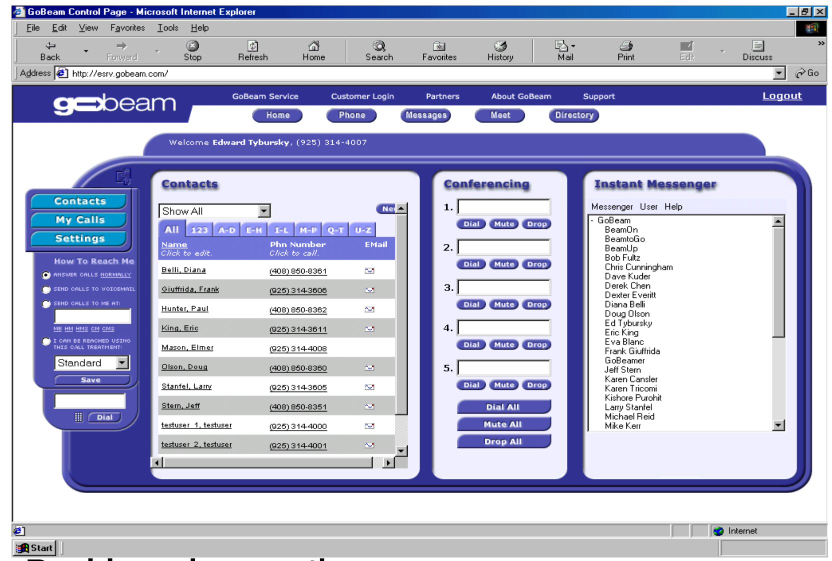
Service Provider Solutions Engineering

Agenda

- **Hosted IP Telephony is not Centrex**
- Solution Overview
- Hosted IP Telephony works
- Partner Overview

Centrex Replacement ? Hosted IP Telephony





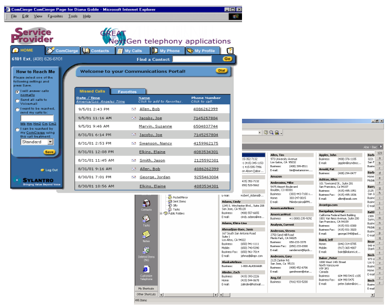
- •1 – End-user control
  - End-users don’t have to own CPE
- •2 – Radically lower OpEx
  - Moves, adds, and changes done via web interface
- •3 – New enhanced services
  - Generate more revenue per user
- •4 – Existing services usable
  - Better usability w/ web interface vs. star codes, flash hooks
- •5 – Improved enterprise networking
  - Voice VPN services to heterogeneous CPE
- •6 – Rapid service creation
  - Simple CPL scripting vs. complex AIN implementation
  - Enables 3rd party development, Only open, standard protocols
- •7 – Convergence cost savings
  - Voice and data on one pipe
- •8 – Services on one platform
  - Unified communications offering, Simpler back office integration
- •9 – Lower incremental cost
  - Modular, centralized deployment
- •10 – Future-proof network
  - Begin next-gen deployment

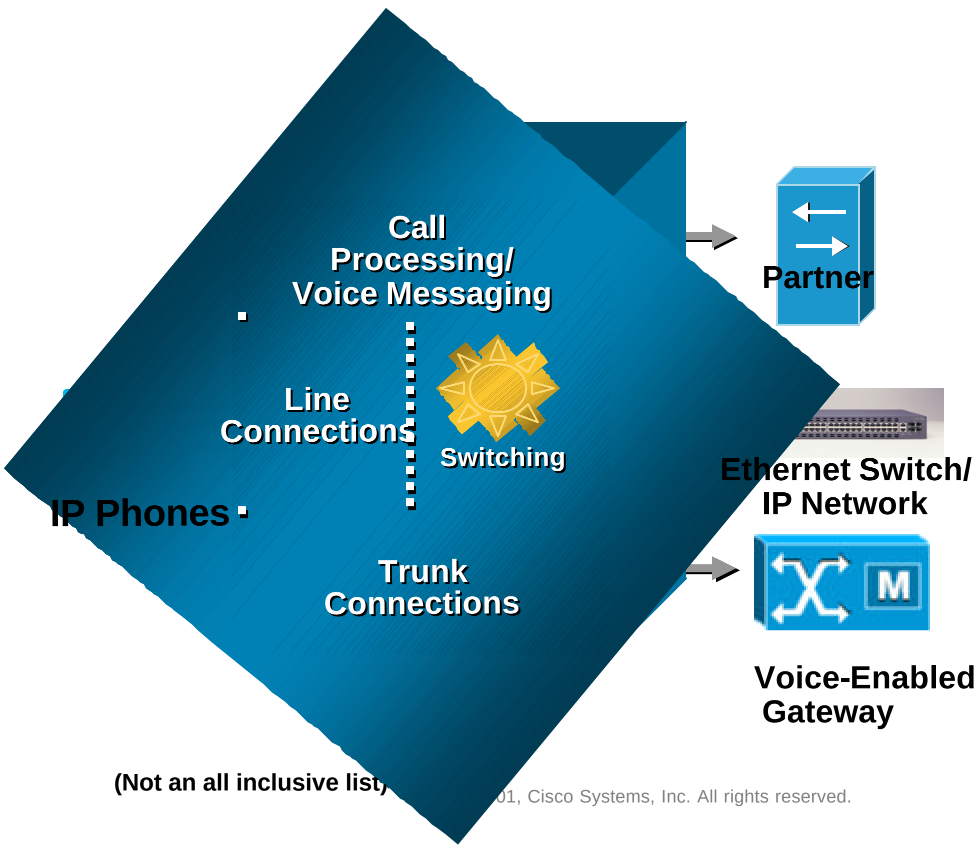
Agenda

- Hosted IP Telephony is not Centrex
- **Cisco HIPT Solution Overview**
- Hosted IP Telephony works
- Partner Overview

A lot like an IP PBX

GUI Interface
-Directory Svcs





Business Line Feat.

- Call Forward
- Call Transfer
- Redial
- Conferencing
- Malicious Call Trace
- Music On Hold
- Call Park
- Call Hold
- Speed Dial
- Anonymous Call Block
- Basic ACD functionality
- Extension dialing

Enhanced to leverage data.

- GUI End user portal
- Click to Dial
- Group Phone lists
- Presence Integration
- Unified Communications
- Outlook integration
- Personal Phone list
- Find me follow me
- PC Soft Phone
- WAP user portal for mobility

(Not an all inclusive list)

Carrier based Services

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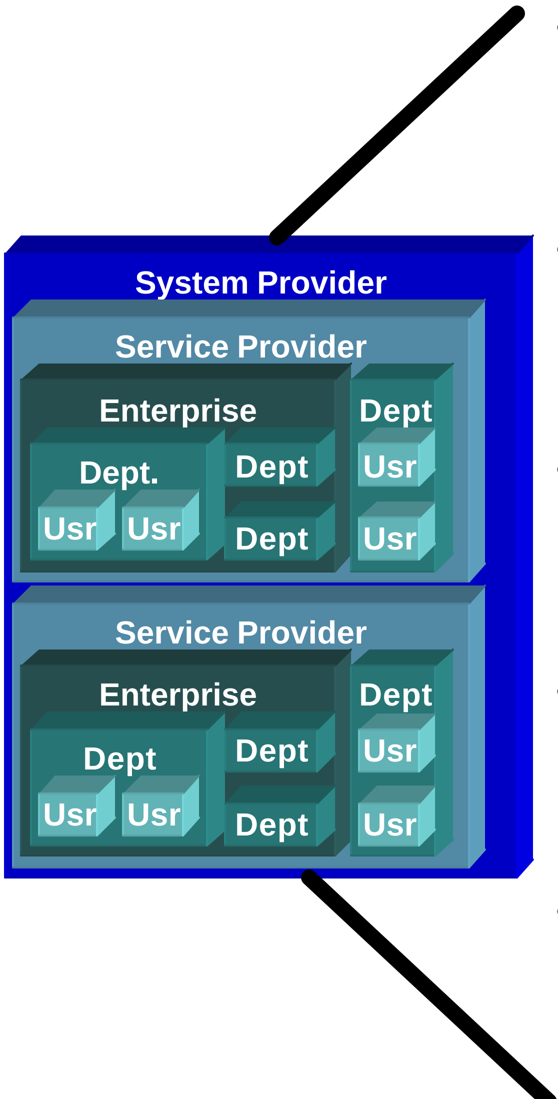
- E.164 Numbering Plan Translations
- Local Calling Area (LCA) Screening
- Service Center Routing
- Carrier Routing
- International Routing
- Alternate/Multiple Routes
- Flexible Routing (1-15 Digits)
- Equal Access Routing
- Casual Dialing
- URL Dialing
- Media Server Routing
- E911
- SS7 – ISUP, TCAP
- Toll Free
- Local Number Portability
- Origin Based Routing
- Destination Based Routing
- Lawful Intercept
- Operator Services
- Least Cost Routing
- Network Routing Prioritization
- Service Provider Provisioning via CLI, telnet or CORBA
- Alarms and statistics via SNMP
- Billing records Streamed or sent to FTP server

Hosted IP Telephony Structure



Layer	Services	Management
System Provider	Regulatory Services Translations/Routing Resource Management	Web Portal Integrated EMS Accounting
Service Provider	Reseller Support Least Cost Routing Virtual Provisioning	Web Portal Measurements Accounting
Enterprise	Voice VPNs	Web Portal Moves, Adds, Changes
Department	Call Centers Attendants	Web Portal Moves, Adds, Changes Service Management
User	Centrex Voice Mail Conferencing	Web Portal Personalization Customization

Agenda

- Hosted IP Telephony is not Centrex
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- Partner Overview

Hosted IP Telephony Architecture

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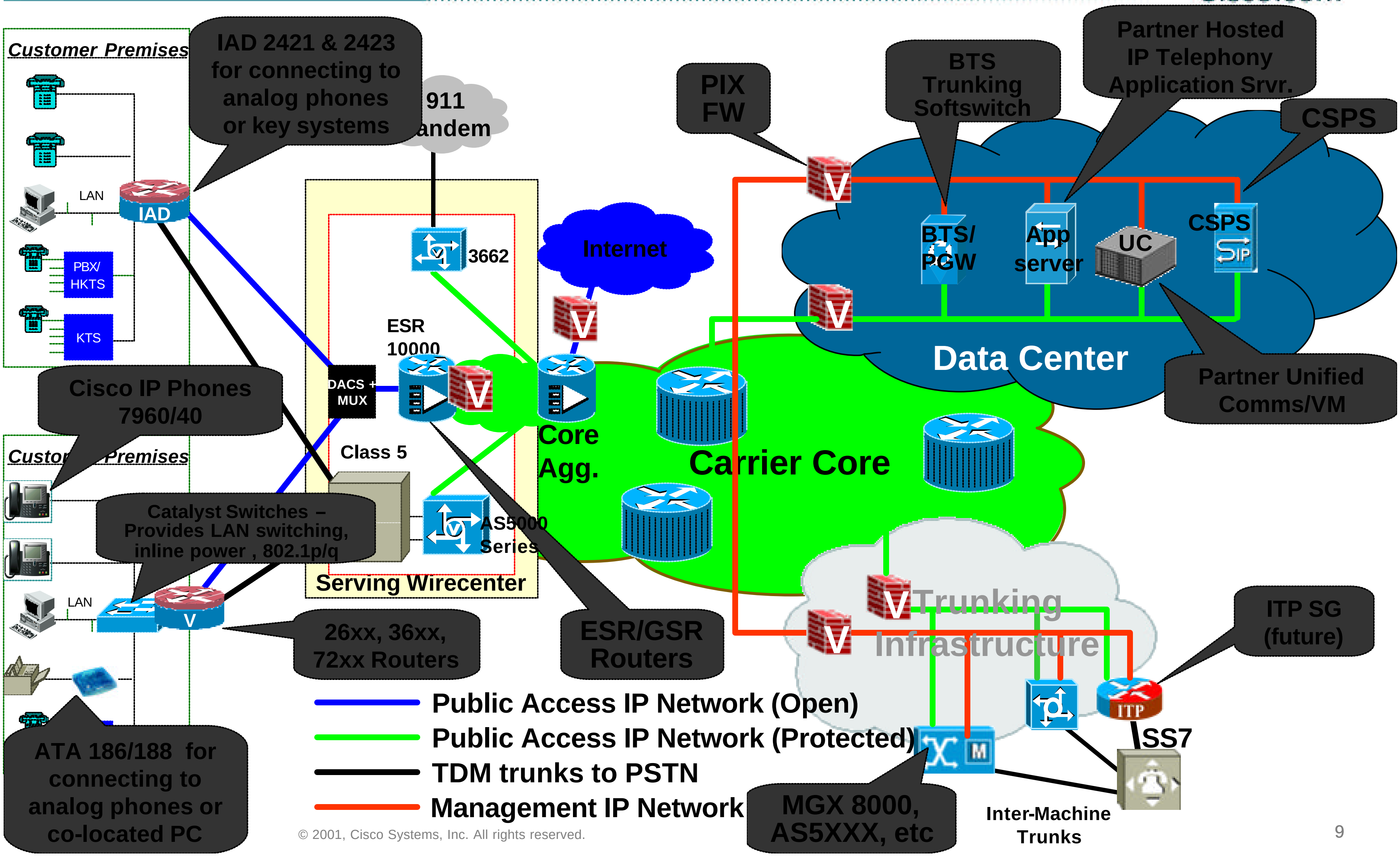
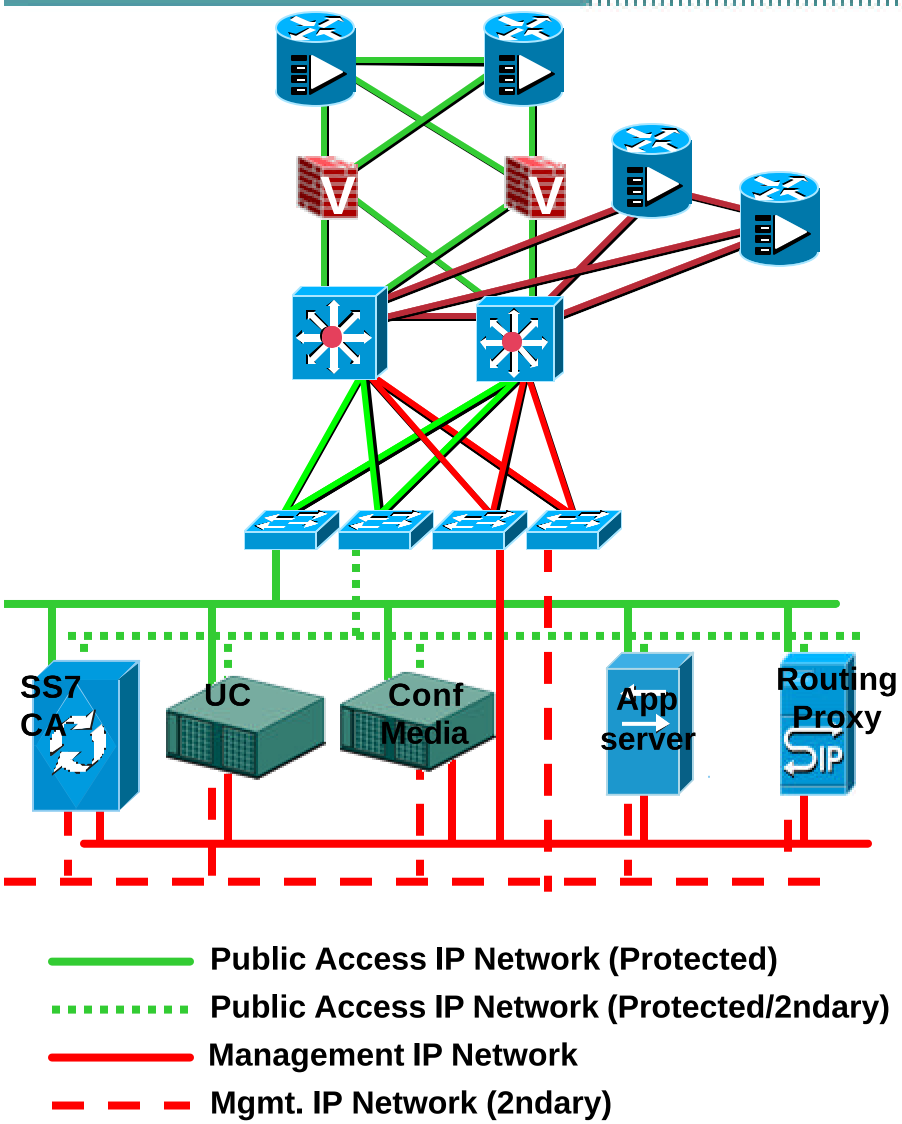


Solution Reliability and Security



- **High Availability Network design – router/switch redundancy**
- **PIX 535 - 10,000,000 MTBF (Est.)**
- **Servers in Active/standby pairs, clusters or pooling**
- **DNS SRV**
- **Self Healing IP network**
- **Secure management traffic through separate interfaces and networks**
- **Firewall with proper protocol ALG support (SIP or MGCP)**
- **QoS enabled IP network**

SIP Survivable Remote Site Telephony

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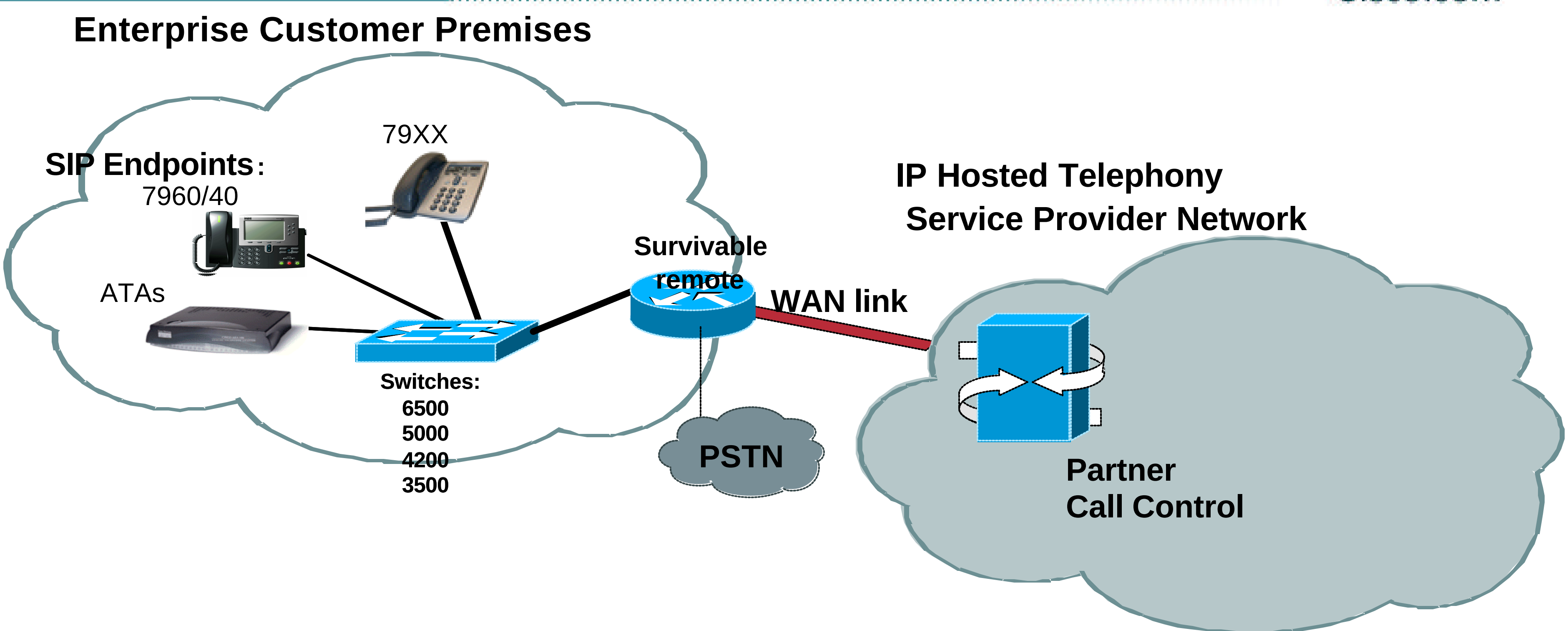
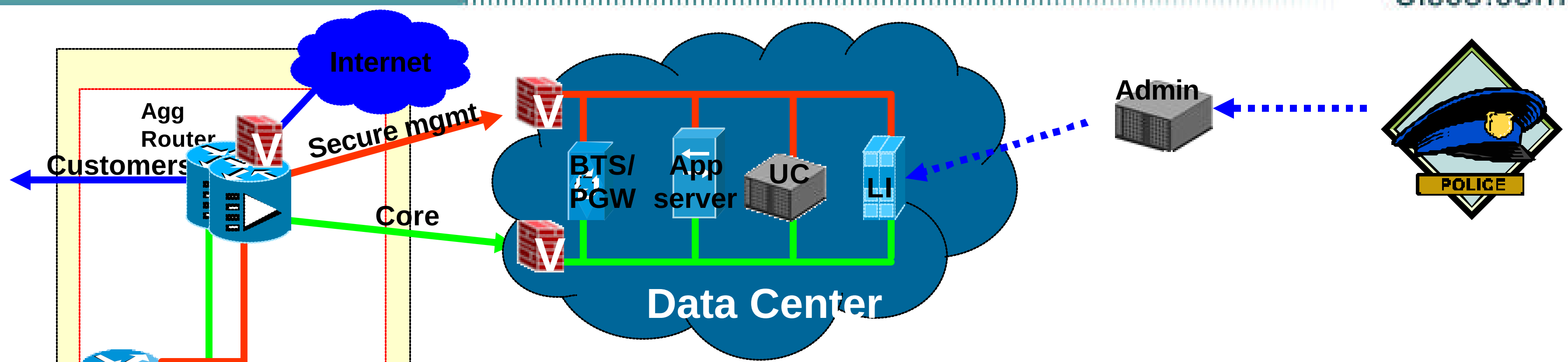
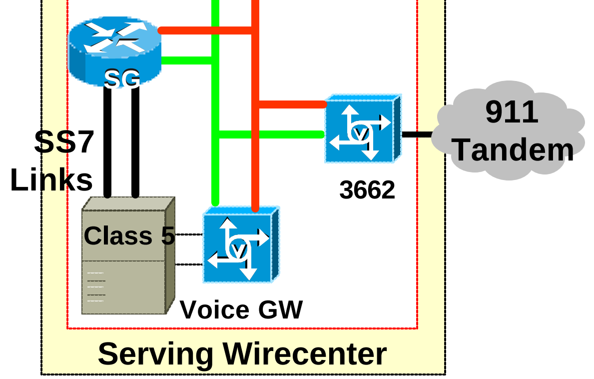


Survivable Remote – "the Ability to Provide Voice Services When a Remote Location Is 'Stranded' From the Core Service Provider Network"

- **UPS backup with Inline power phones**
- **Router CPE becomes call control entity**

Hosted IP Telephony Regulatory

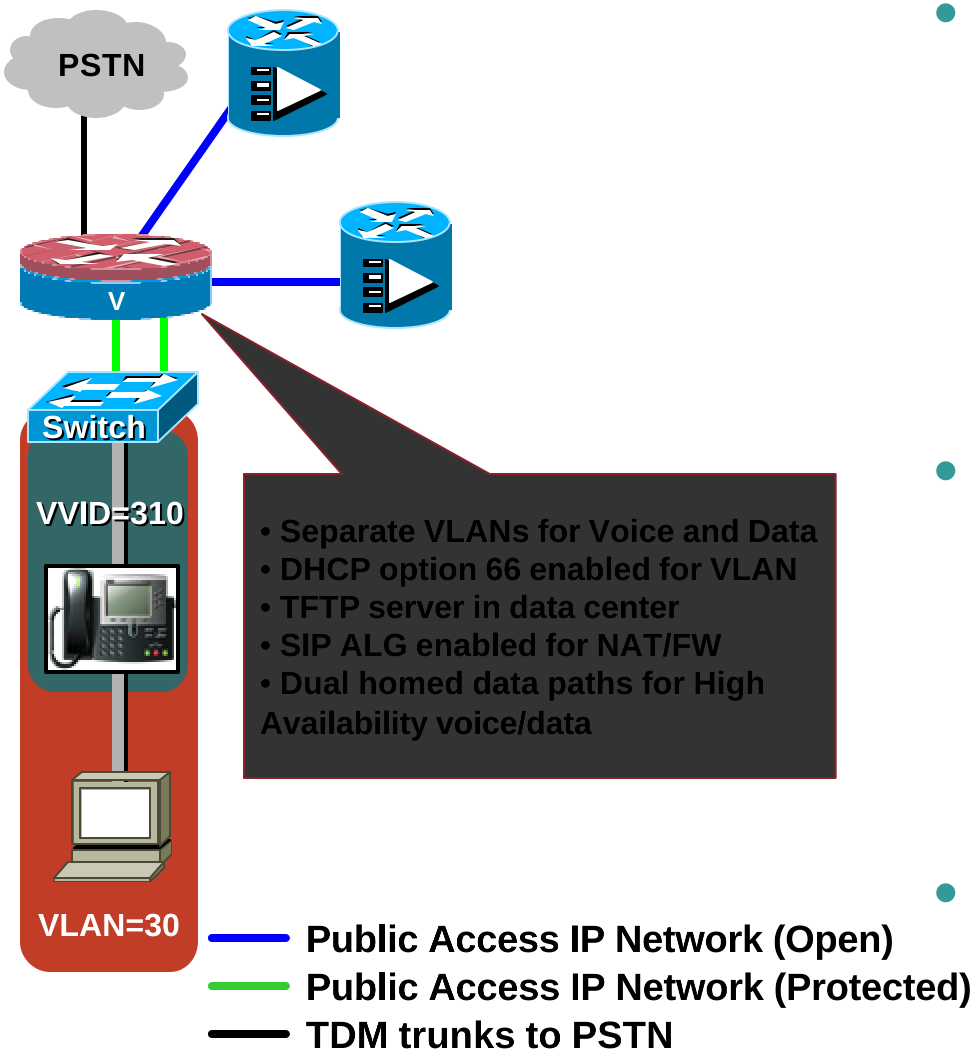




Public Access IP Network (Open)
Public Access IP Network (Protected)
TDM trunks to PSTN
Management IP Network

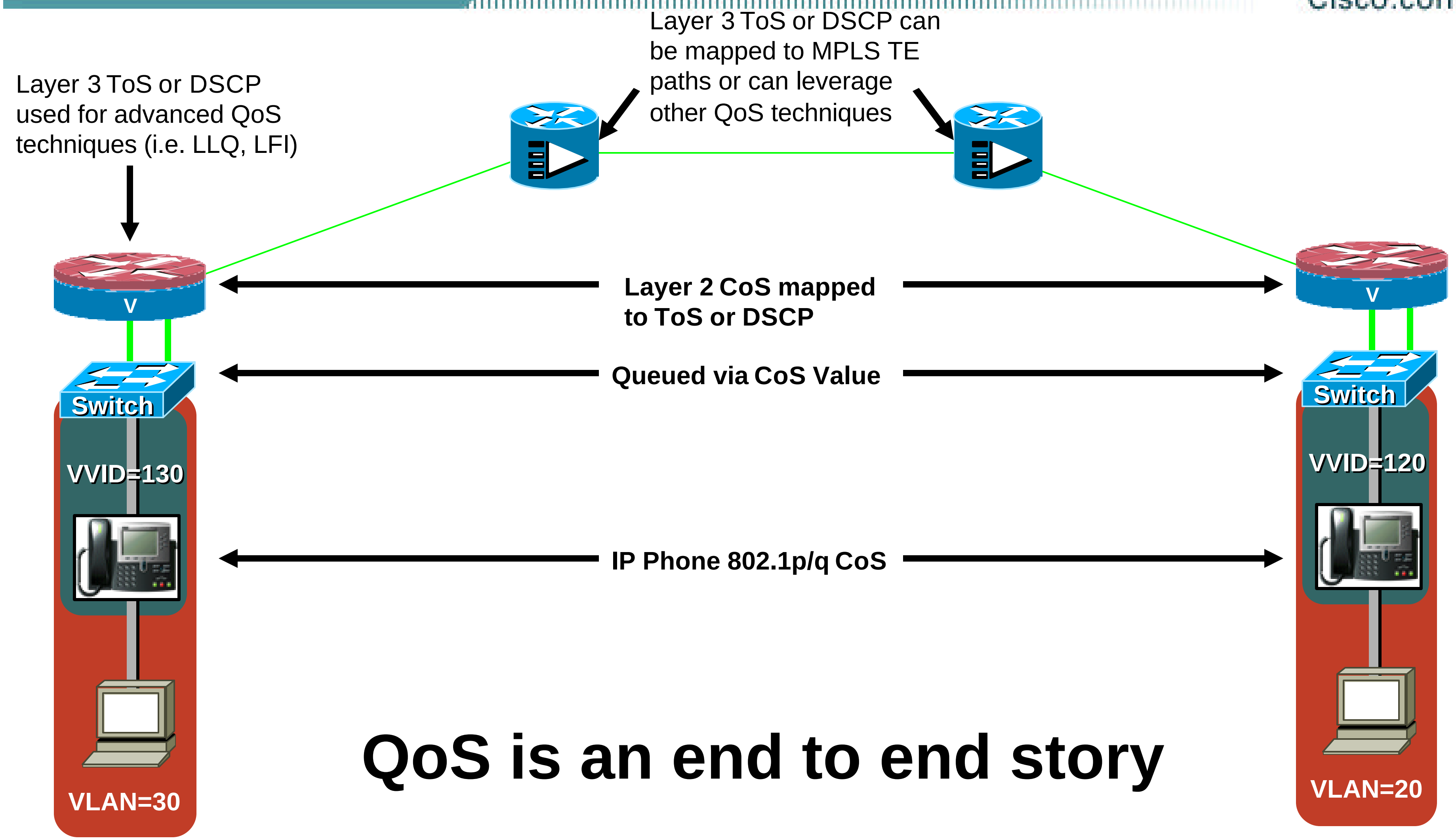
- Various LI partners to provide functionality
  - ESR packet replication techniques in place
  - End User blind to tap
  - Leveraged Packet Cable stds
- Emergency location services
  - Mixture of Partner and trunking softswitch

Access/CPE Flexibility



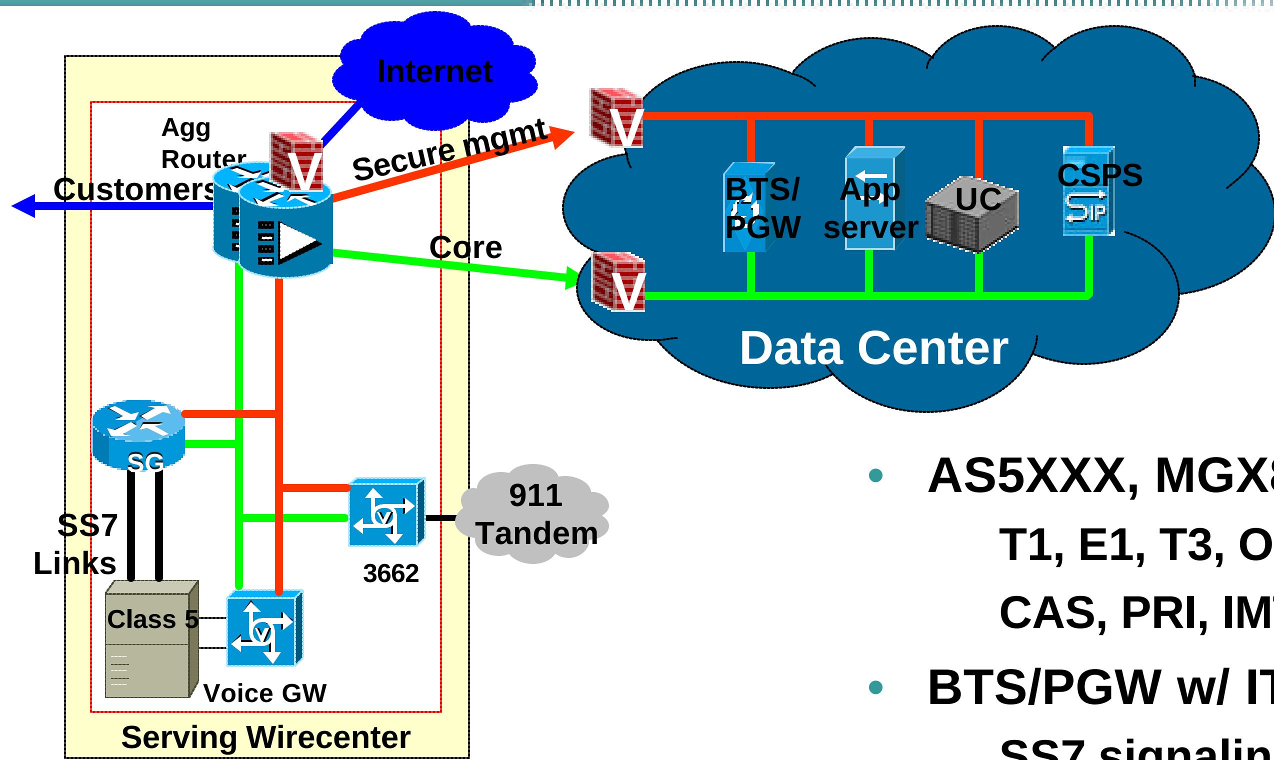
- **User Choice**
  - –ATA 186/188 – Analog
  - –IP Phone – 7960/40/05
  - –Key system w/ IAD trunking
- **Router flexibility**
  - – Various WIC support, xDSL, T1/E1 PPP, ATM, FR
  - – SIP Survivable Remote site Telephony (future)
- **SIP, MGCP Solution support.**

Hosted IP Telephony Solution Service Quality



QoS is an end to end story

Hosted IP Telephone Interoperability



Public Access IP Network (Open)
Public Access IP Network (Protected)
TDM trunks to PSTN
Management IP Network

- AS5XXX, MGX8800, C3660
  - T1, E1, T3, OC3 (GW dependent)
  - CAS, PRI, IMT support
- BTS/PGW w/ ITP/SLT
  - SS7 signaling
  - PGW Homologation over 60 countries
- Various products tested at Industry interop events (i.e. SIPit, ISC, IMTC)
- PIX FW used for Security and SIP/MGCP ALG support

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Sylantro Overview

Agenda

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- System Overview
- Features
- Components
- Scaling Models

Business Communication Ser

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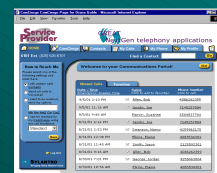
Service Provider Can package

Hosted PBX / IP Centrex

C-BUSINESS

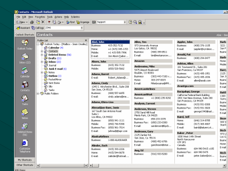
Call Hold
Call Forwarding
Call Waiting
Redial
Bridged Line
Call Transf
Speed Dia
Conferencing



COMPORTAL

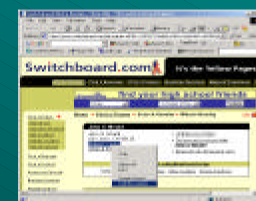
Web-based portal

Access Options



COMOFFICE

Full Outlook
Integration



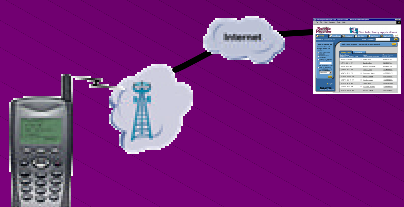
IE INTEGRATION

Click to call from
Any web page

Sylantro
Solutions

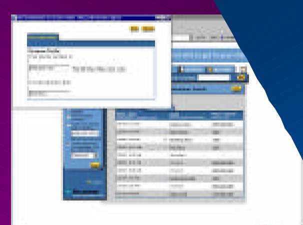
COMTRAVELLER

Desktop Applications in the palm of your hand



COMRIO

- Telecomm
- Multi-of
- Visiting s
- Remote em



- Che
- Statistic

Mobility

COMCIERGE

- Personalized call treatments
- VIP categories
- Single Mailbox

Advanced Options

Agenda

- System Overview
- **Features**
- Components
- Scaling Models

Sample Feature Set

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Sylantro's c-Business module represents the most popular Centrex & PBX features

c-Business SUBSCRIBER PHONE FEATURES - Productivity Features
- Call Hold
- Message Waiting Lamp
- Call Logs
- Missed Calls
- Incoming Calls
- Outgoing Calls
- Call Park
- Call Pick-up
- Call Transfer
- Call Waiting
- Conferencing
- DID
- DOD
- Favorites Contacts (Speed Dialing)
- Flexible Feature Mapping
- Hunt Groups
- Last Number Redial
- Station to Station Dialing
- Call Forwarding
- Do Not Disturb
- Handset Volume control
- Speaker Volume Control
- Hands-Free Dialing
- Hook Flash
- Click-to-Call
- Music on Hold
- Bridged Line Appearances
- Directed Pickup
- Billing Codes

COMMUNICATIONS PORTAL

Personal Communications Portal

(Personal) Welcome Kit

My Home (Page)
- How to Reach Me
- Missed Calls
- Favorites
- Single Number Service

Contacts (Page)
- Favorite Contacts
- Company Contacts

My Calls (Page)
- Call Logs
- Missed Calls
- Incoming Calls
- Outgoing Calls

My Phone (Page)

My Profile (Page)
- General
- Home Address
- Phone Numbers
- Security

Help (Pages)
- Context-sensitive Help Page(s)
- Online Access to User's Guide

MANAGEMENT PORTAL(S)

OA Setup Screens
- Secure Login Welcome (Page)

Company (Page)
- Intro
- General
- Billing Address

Employees (Page)
- Instructions
- Assignments

Services (Page)
- Instructions
- Assignments
- Options

Summary (Page)

Help (Pages)
- Help
- Context Sensitive Help
- Online Access to User's Guide

MANAGEMENT PORTAL(S)

Office Administration Portal
- Secure Login
- Forgotten Password Help

Employees (Home Page)
- Company Directory
- New
- New Fax

Contacts (Page)
- Company Contacts
- New

Services (Page)
- Hunt Groups
- Assign Extensions
- Answer Order
- Options

Reports (Page)
- System

Support (Page)
- Contact us
- Adapters
- Company Info
- Billing Address

Help (Pages)
- Help
- Context Sensitive Help
- Online Access to User's Guide

CARRIER Features
- Public dialing plans
  N11
  8XX
  1+, 0+
  101XXXX+
  011+
  Feature codes
- E911 routing
- Geographic-specific routing

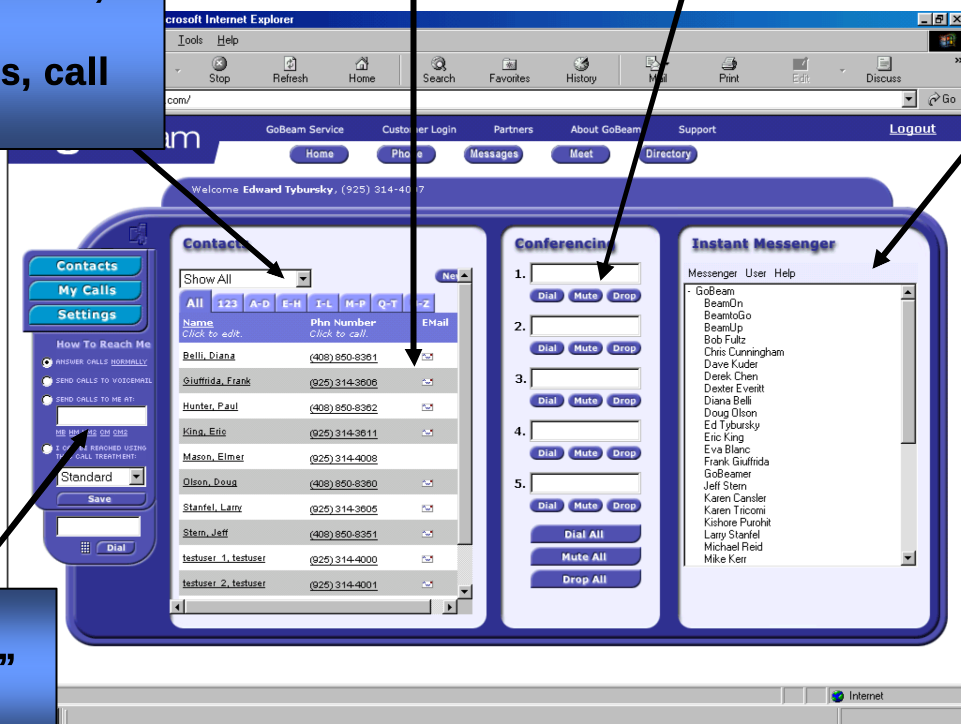
Service Provider Example

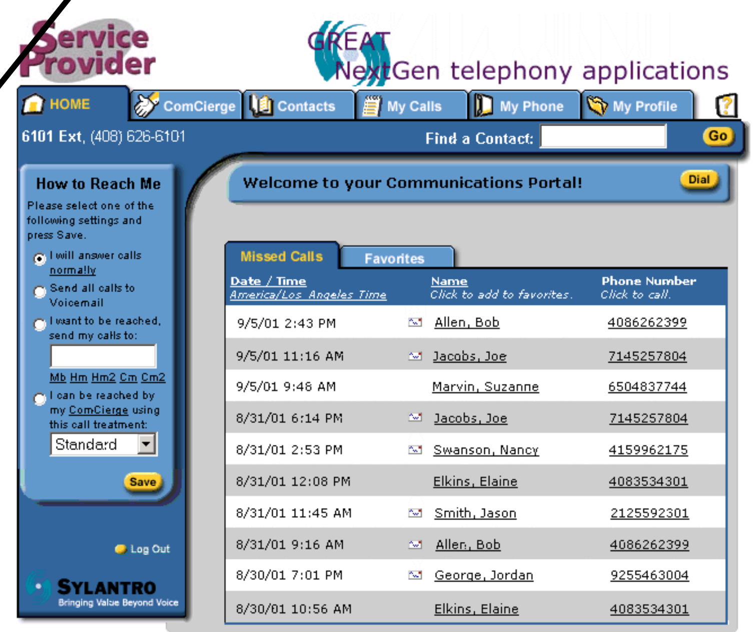
Just click to: call, email, return a call, add a contact

On-demand Conferencing

"chat" real-time with online users through Instant Messenger

Control "find me" feature, phone settings, call logs

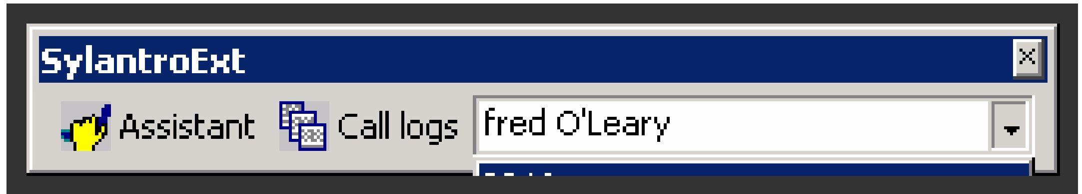


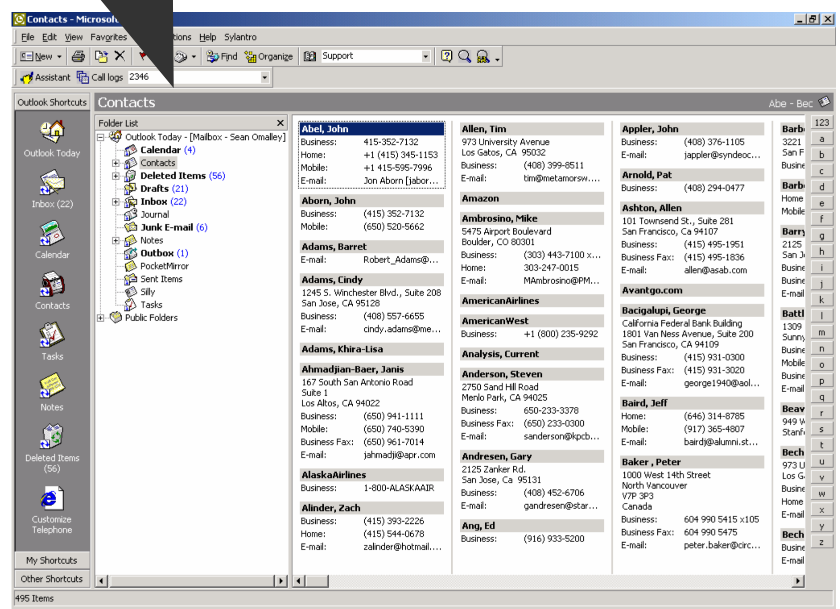


Manage "find me" feature

Outlook Phone Assistant

- Integrated Windows Toolbar
  - The Phone Assistant
    - Reach me settings
  - Call Logs
    - Missed, Incoming and Outgoing
  - Click to Call Directory Access
    - Fully synchronized with Outlook
    - Enter name or number to click-to-call





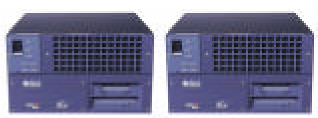
Agenda

- System Overview
- Features
- **Components**
- Scaling Models

Sylantro Servers

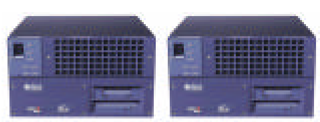
Application Switch

- Service execution environment
- Non-call applications (call logs)
- SNMP agents
- Watchdogs



Control Server

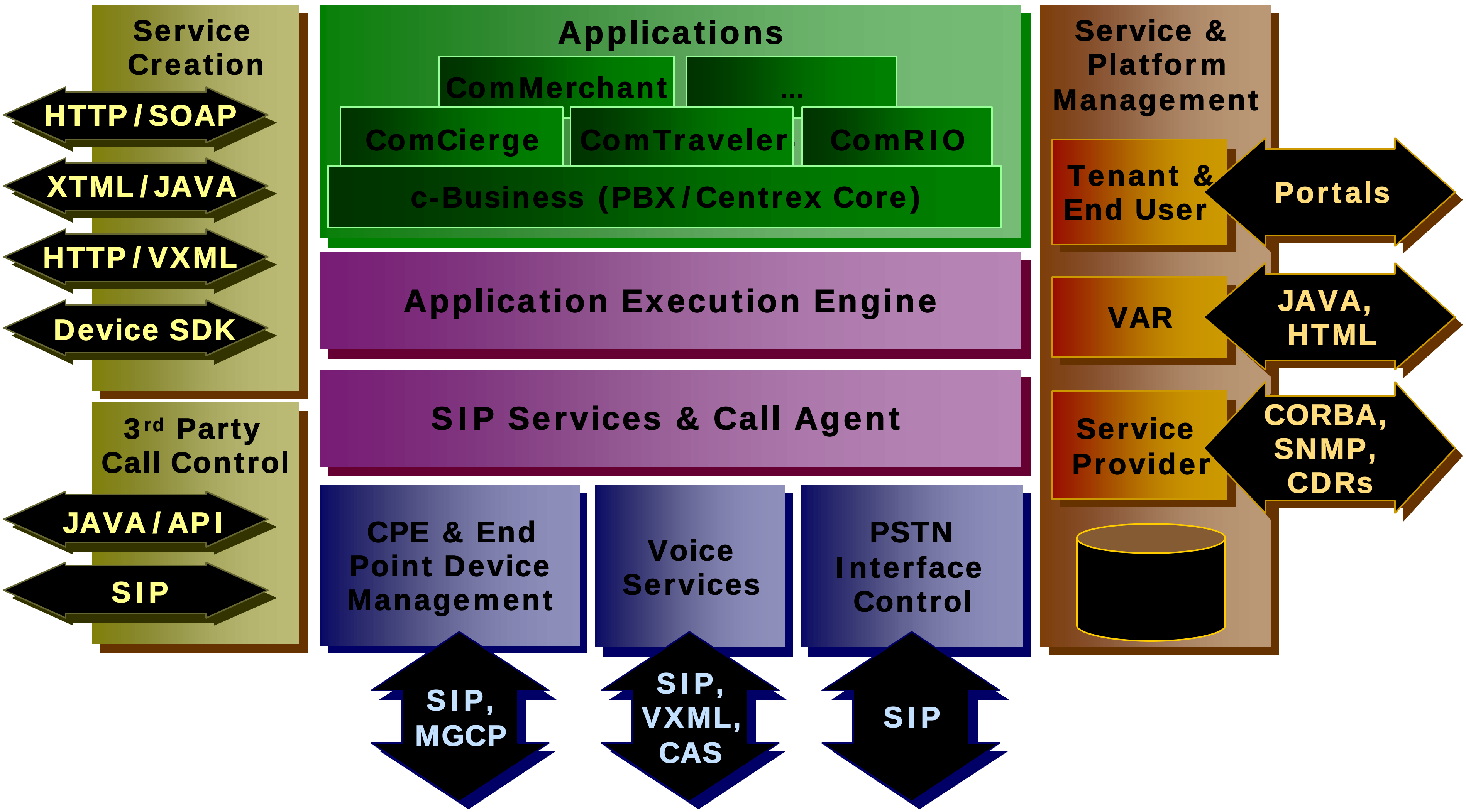
- Call processing
- Local change & resource management
- SNMP agents
- Watchdogs



Administration Server

- OSS interfaces ? CORBA
- CDR collection & interfaces ? flat file
- Configuration data & interfaces ? XML via CORBA
- SNMP agents
- Watchdogs
- Central change & resource management

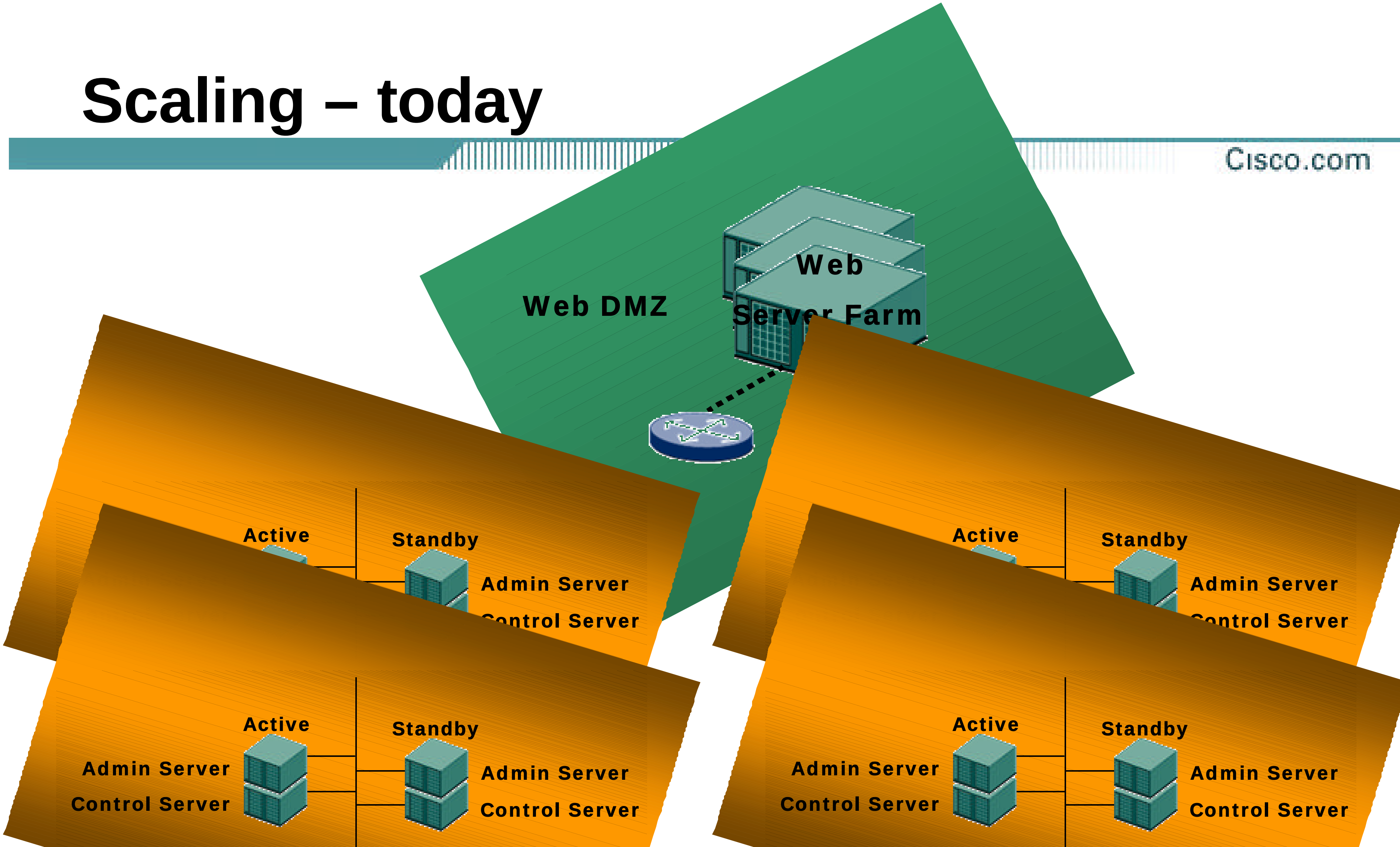
Architecture Overview
Functional Overview



Agenda

- System Overview
- Features
- Components
- **Scaling Models**

Scaling – today



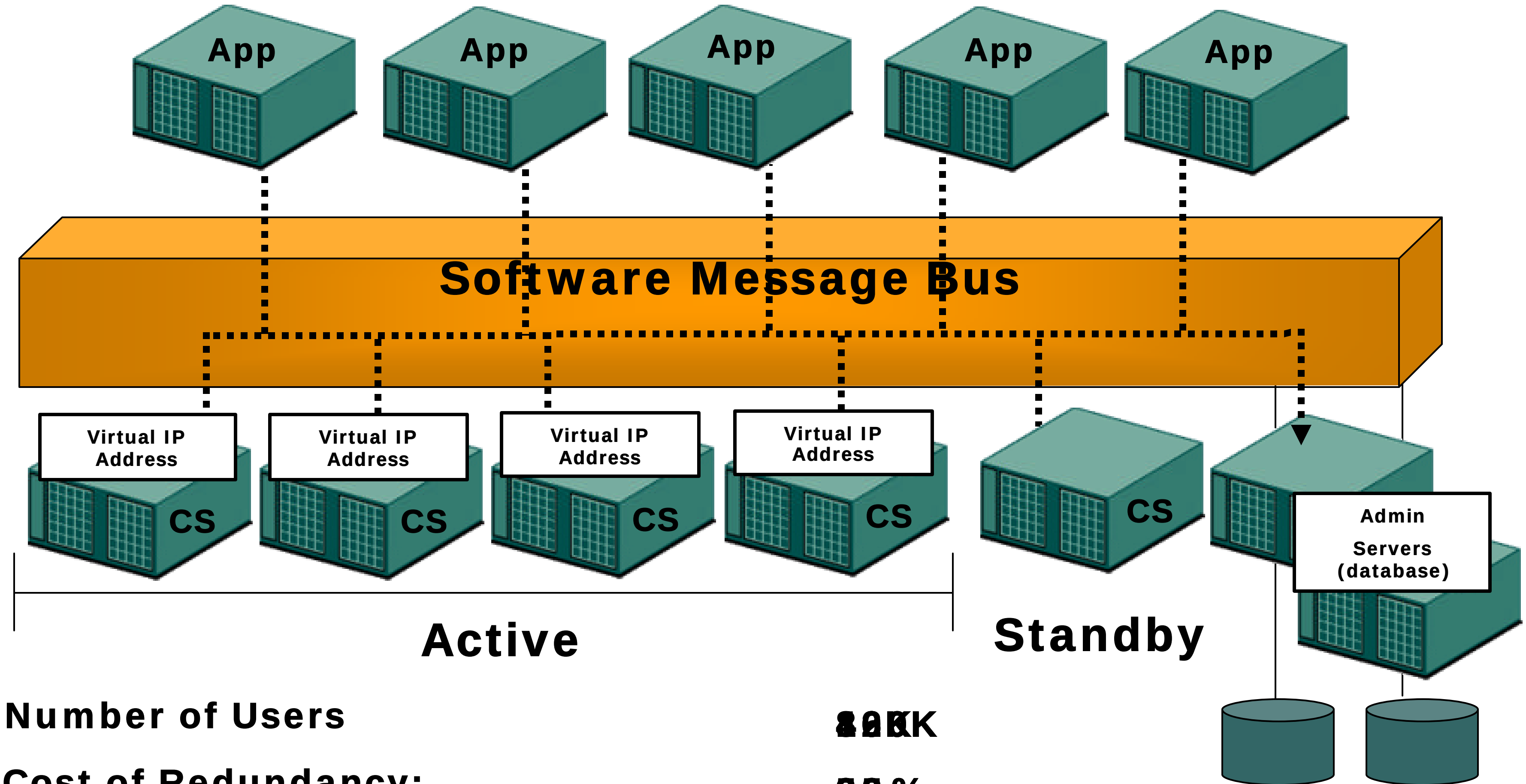
Number of Users **80K**

Cost of Redundancy: **50%**

Population impacted in case of failure*: **100%**

 ***No established call lost, impact limited to new calls experiencing delays**

Scale – Future



Number of Users [illegible]

Cost of Redundancy: [illegible]

Population impacted in case of failure*: [illegible]

***No established call lost, impact limited to new calls experiencing delays**

Broadsoft Overview

Agenda

- System Overview
- Features
- Components
- Scaling Models

Product is BroadWorks

Single communications services delivery platform

Dial Tone + Services

- PBX/Key
- Analog Phones
- IP PBX Set
- IP Phones

Enterprise Networking

- Site to Site
- Site to/from PSTN
- Remote Access

Hosted Enhanced Services

- Unified Messaging
- Call Centers
- Conferencing
- Remote Office

Service Provider Ready

- Carrier Routing
- OAMP
- Scalability, Reliability
- Service Creation/Differentiation

The BroadWorks Web interface for call control and service management

Agenda

- System Overview
- **Features**
- Components
- Scaling Models

BroadWorks Structure

Layer	Services	Management
System Provider	Regulatory Services Translations/Routing Resource Management	Web Portal Integrated EMS Accounting
Service Provider	Reseller Support Least Cost Routing Virtual Provisioning	Web Portal Measurements Accounting
Enterprise	Call Centers Attendants Voice VPNs	Web Portal Moves, Adds, Changes Service Management
User	Centrex Voice Mail Conferencing	Web Portal Personalization Customization

BroadWorks Network Services

Network Services

- North American Numbering Plan (NANP) Translation
- E.164 Numbering Plan Translations
- Local Calling Area (LCA) Screening
- Origin Based Routing
- Destination Based Routing
- Service Center Routing
- Carrier Routing
- International Routing
- VH Routing
- Least Cost Routing
- Network Routing Prioritization
- Alternate/Multiple Routes
- Flexible Routing (1-15 Digits)
- Equal Access Routing
- Casual Dialing
- URL Dialing
- Media Server Routing
- Lawful Intercept
- E911
- CommPilot Portal

SP branding

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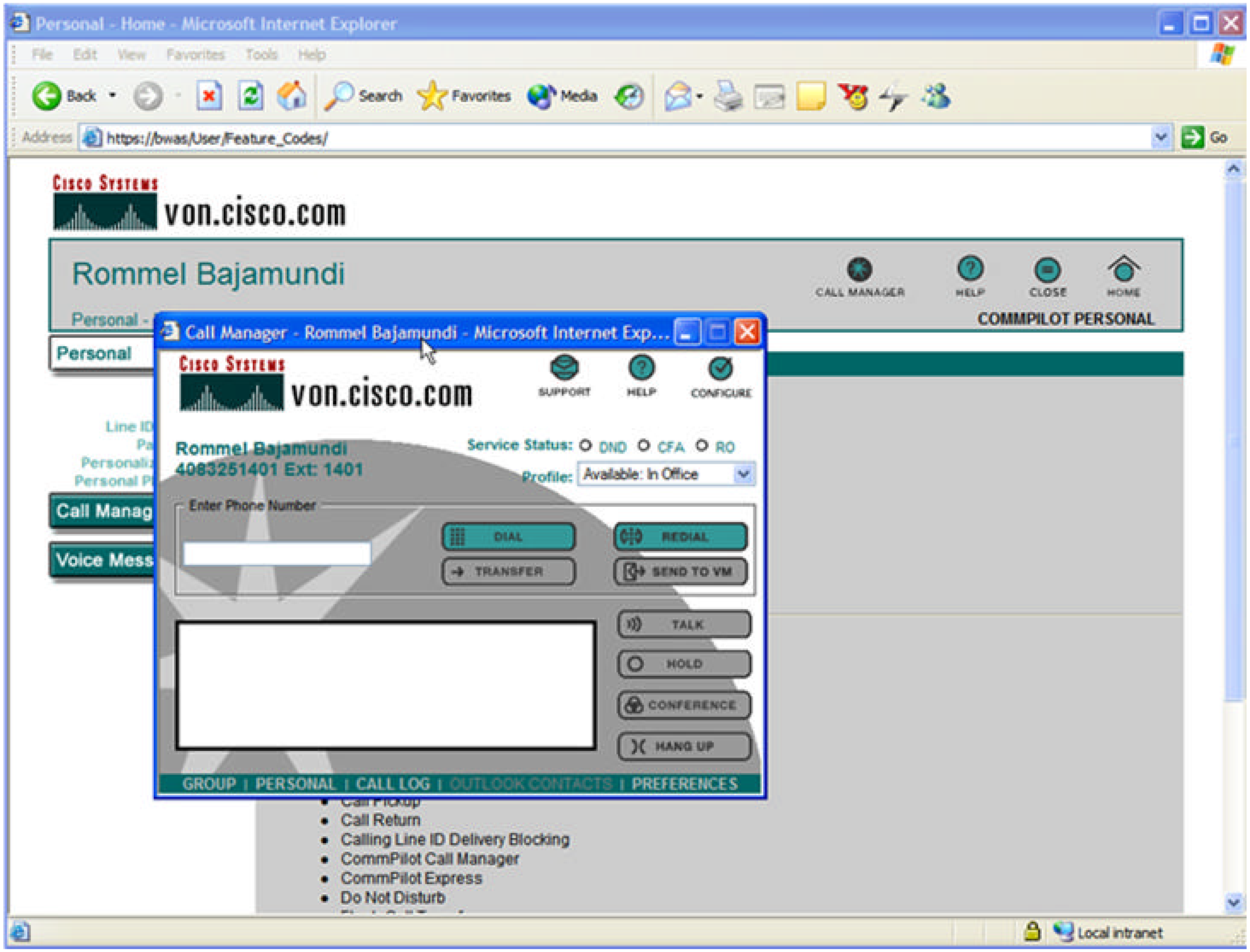


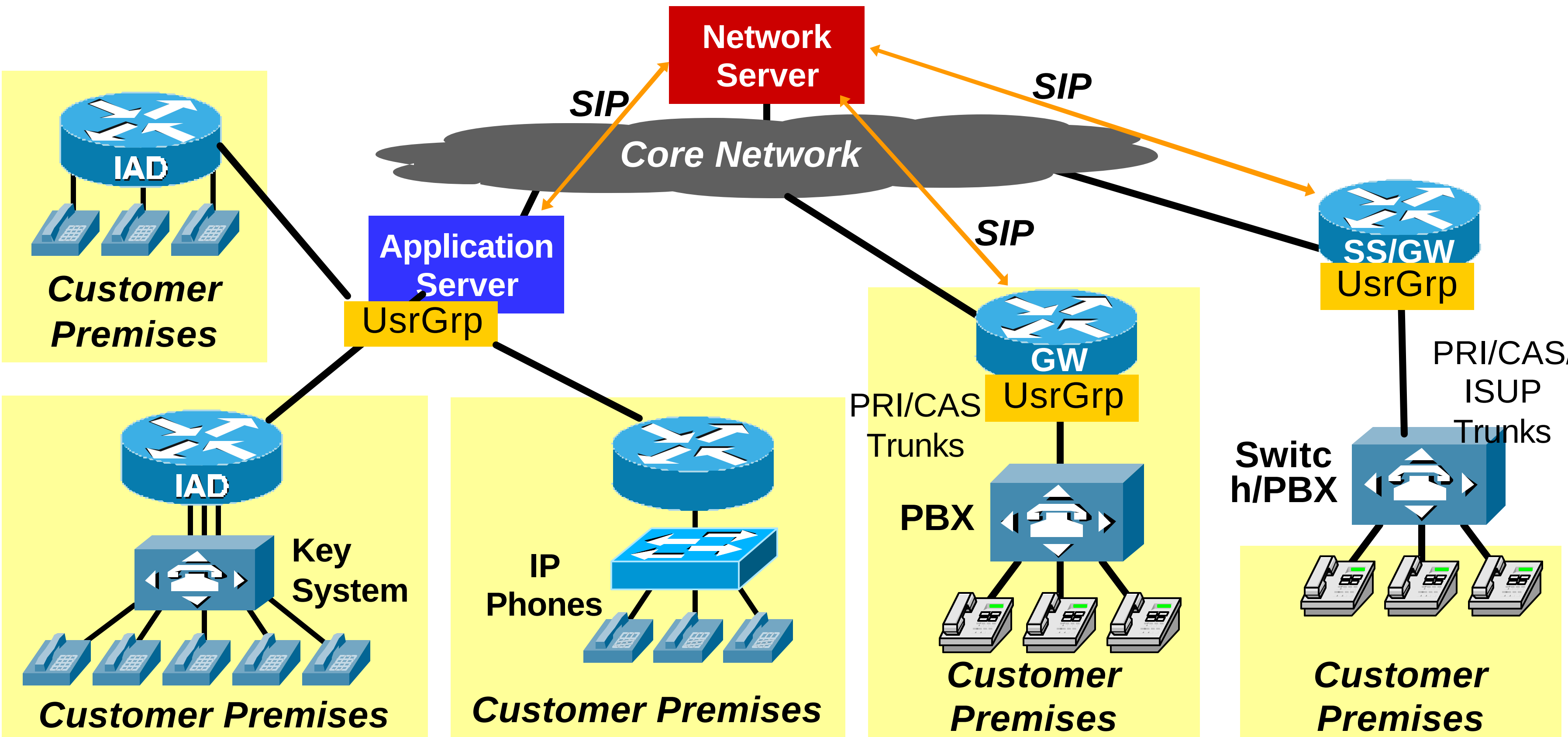
BroadWorks Enterprise Services

Enterprise Services

- Account Codes
- Authorization Codes
- Auto Attendants
- Call Centers
- Call Intercept
- Call Park
- Call Pickup
- Call Capacity Management
- Common Phone List
- CommPilot Web Portal
- Configurable Extension Dialing
- Configurable Feature Codes
- Equal Access
- Device Management
- Group Calling Identity
- Group Voice Mailbox
- Hunt Groups
- Incoming and Outgoing Calling Plans
- Instant Conferencing
- Least Cost Routing
- Loudspeaker Paging
- Private Dialing Plans
- Self Service Adds, Moves and Changes
- Series Completion
- Voice Configuration Portal
- Voice VPN Support

Voice VPN – Dial Plan Access

- Access to Voice VPN capabilities can occur through:
  - Hosted IP Telephony Application Server (via analog phones, SIP phones, IADs, etc)
  - Private gateway (i.e. customer premises located gateway) fronting a PBX
  - Public gateway (i.e. network located gateway) fronting a PBX or even a landline or mobile switch

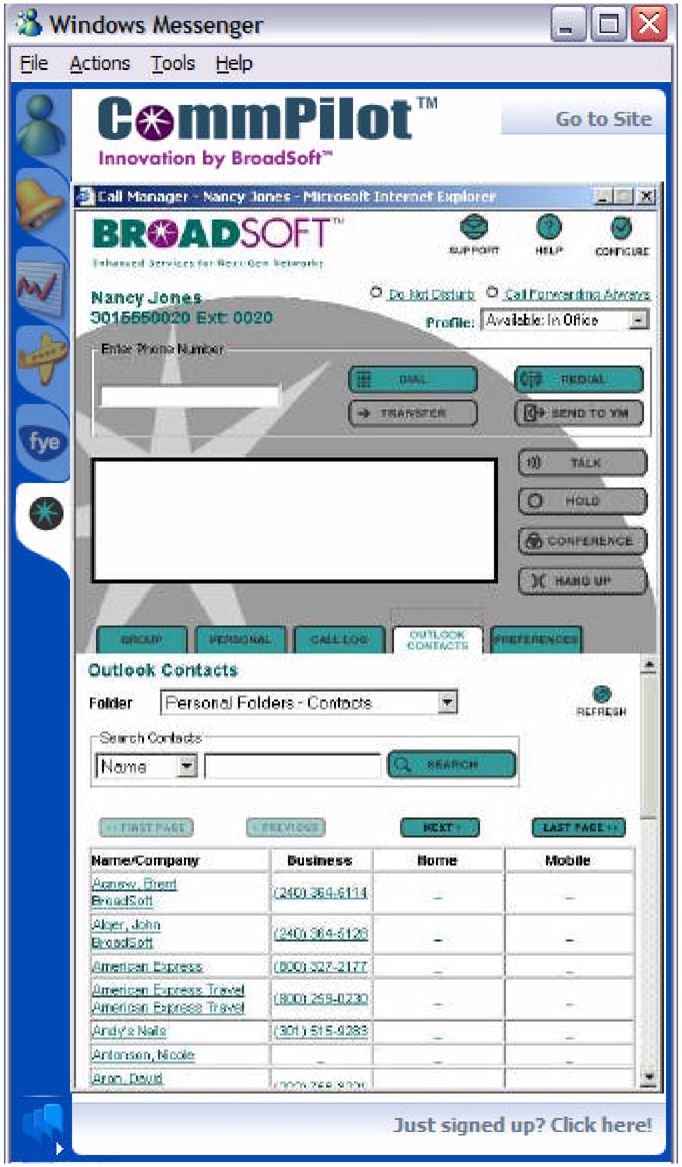


BroadWorks Personal Services

Personal Services

- Anonymous Call Rejection
- Authentication
- Call Forward Busy
- Call Forward No Answer
- Call Forward Always
- Call Hold/Retrieve
- Call Notify
- Call Park
- Call Pickup
- Call Return
- Call Transfer
- Call Waiting
- Calling Line ID Blocking
- Calling Name Delivery
- Calling Number Delivery
- Cancel Call Waiting
- CommPilot Call Manager
- CommPilot Express
- CommPilot Portal
- Distinctive Alerting
- Do Not Disturb
- Group Phone List
- IP Phone
- Last Number Redial
- Message Waiting Indication
- Outlook Dialing
- Outlook vCard Identity
- Personal Phone List
- Personal Service Creation (CPL Support)
- Phone List Import
- Priority Alert
- Recent Calls Phone List
- Remote Office
- Selective Call Acceptance
- Selective Call Forward
- Selective Call Rejection
- Simultaneous Ringing
- Three Way Call
- Voice Messaging
- Voice Mail to Email
- Voice Mail Notification

Windows Messenger Support



- **Enterprise Focused**
  - BroadWorks serves enterprise contacts
  - MSN serves external contacts
- **Multi Devices**
  - Use both desk phone and/or Windows Messenger
- **Instant Messaging and Presence**
  - Secure communications
  - Policy and accounting management
- **Integrated CommPilot**
  - CommPilot on Windows Messenger tab
- **Presence Integration**
  - "On The Phone" presence
  - Service profiles based on presence state

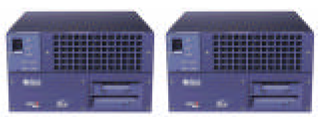
Agenda

- System Overview
- Features
- **Components**
- Scaling Models
- Call Flows

BroadWorks Servers

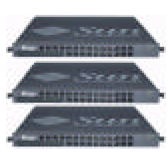
Application Server

- Services Delivery Platform → Access Services
- Line Side Softswitch → End User Focused
- Web Portal → Self Service Management



Network Server

- Centralized Routing → Easy to Manage
- Enterprise Services → Private Number Plans
- Location Register → Maps Users to Servers



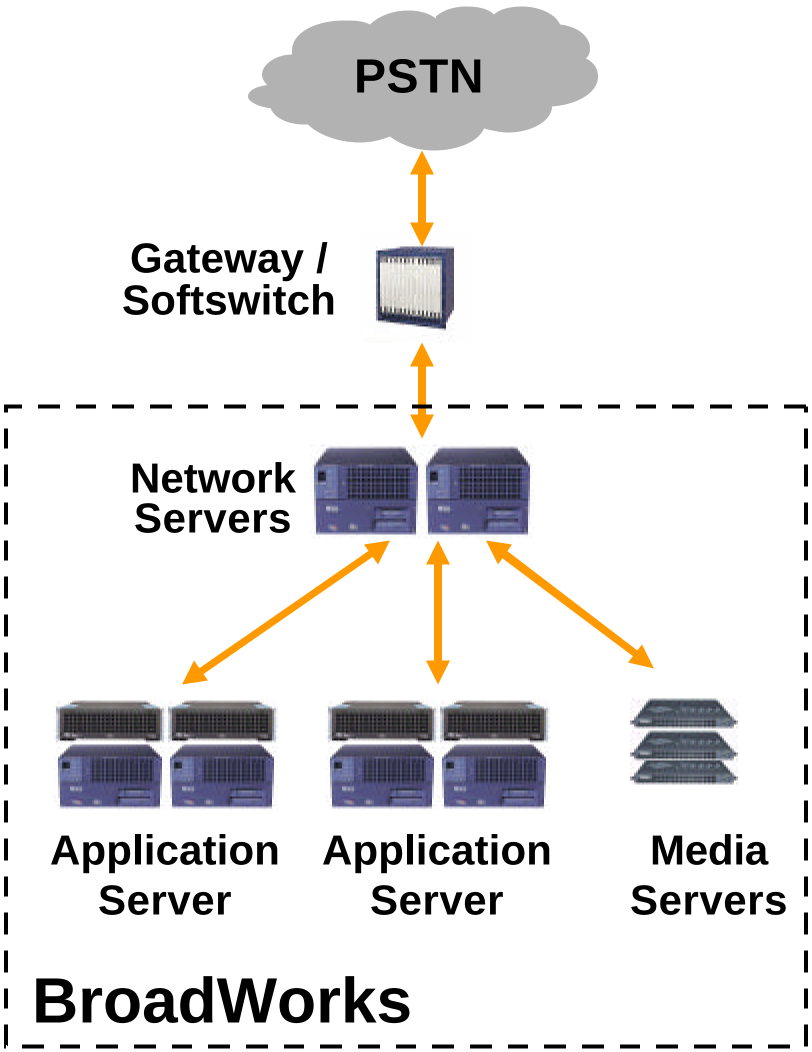
Media Server

- Media Resources → Multimedia Services
- IVR → DTMF, Prompt Playback/Recording
- Mixing → Conferencing

Agenda

- System Overview
- Features
- Components
- **Scaling Models**

Scalability Overview



- Highly Distributed System
- Network Server is "front end" to BroadWorks
- "Internet Model" scalability
  - Add servers when more capacity needed
- No changes required to Softswitch or Back Office
  - Network Server provides single point of contact

Scalability - Growth Path

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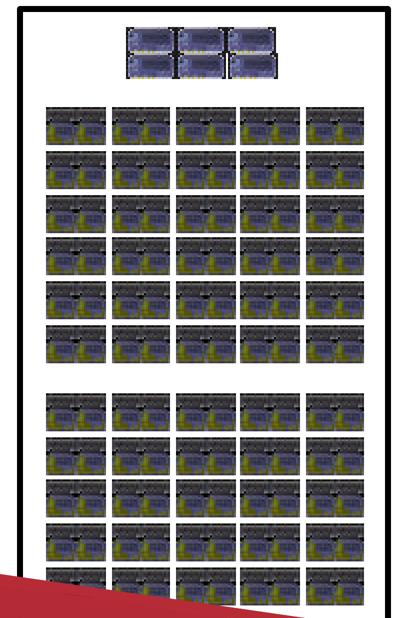
1 Network Server
1 App Server

1 Network Server
5 App Servers

7 Network Servers
35 App Servers



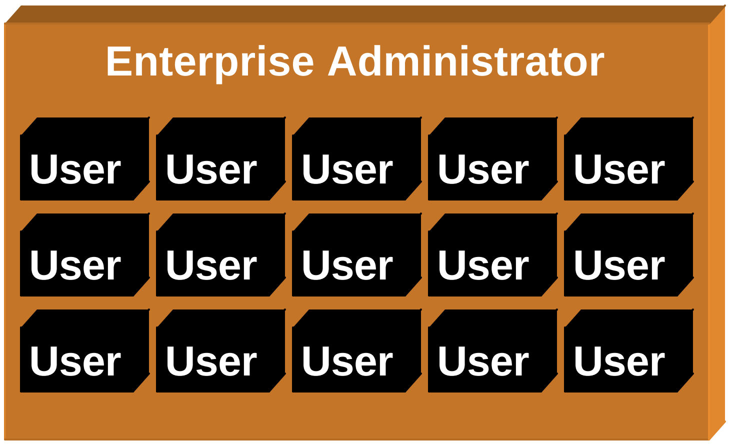
Scalability

Up to 50,000 users
200K BHCA

Up to 2
1M BHCA

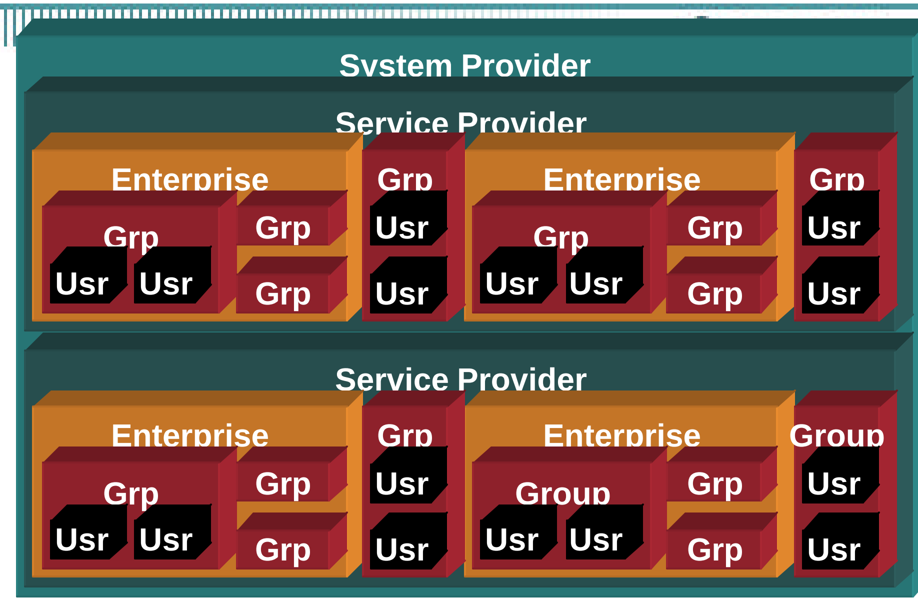
IP PBX vs. HIPT

Management



Managed iPBX

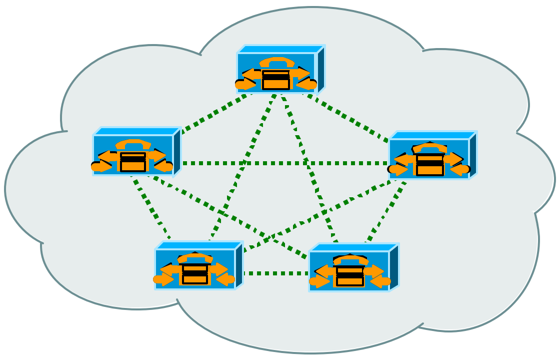
- Partitioning becomes complex and requires additional development
- Per iPBX billing



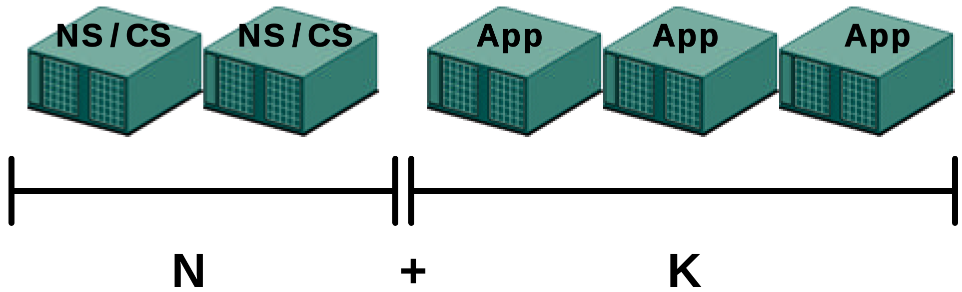
IP Centrex

- Four to five layers of provisioning
- Per logical group billing
- Designed for logical partitioning

Design Robustness



Clustered Approach



Managed iPBX

- Built for the Enterprise
- Windows OS
- SQL database (no real-time updating)
- Cluster for Scale
- Scales to Tens of thousands
- Pervasive Deployments

IP Centrex

- Built for the SP to manage but enterprise to use
- Unix OS
- Times Ten (real time DB)
- Server functions scaled independently
- Scales to hundreds of thousands

Features

Managed iPBX

- Regulatory not as important
- Rich enterprise feature set
- Cluster elements w/timing restraints
- JTAPI/XML for Service Creation
- No GUI branding
- No simple system language transition

IP Centrex

- Various regulatory features
- Feature set balanced between SP and Enterprise
- Geographic redundancy
- Robust Service Creation environment; CPL, XML, JAIN
- GUI Branding
- Defined system language transition

Summary

Cisco's Robust HIPT Solution

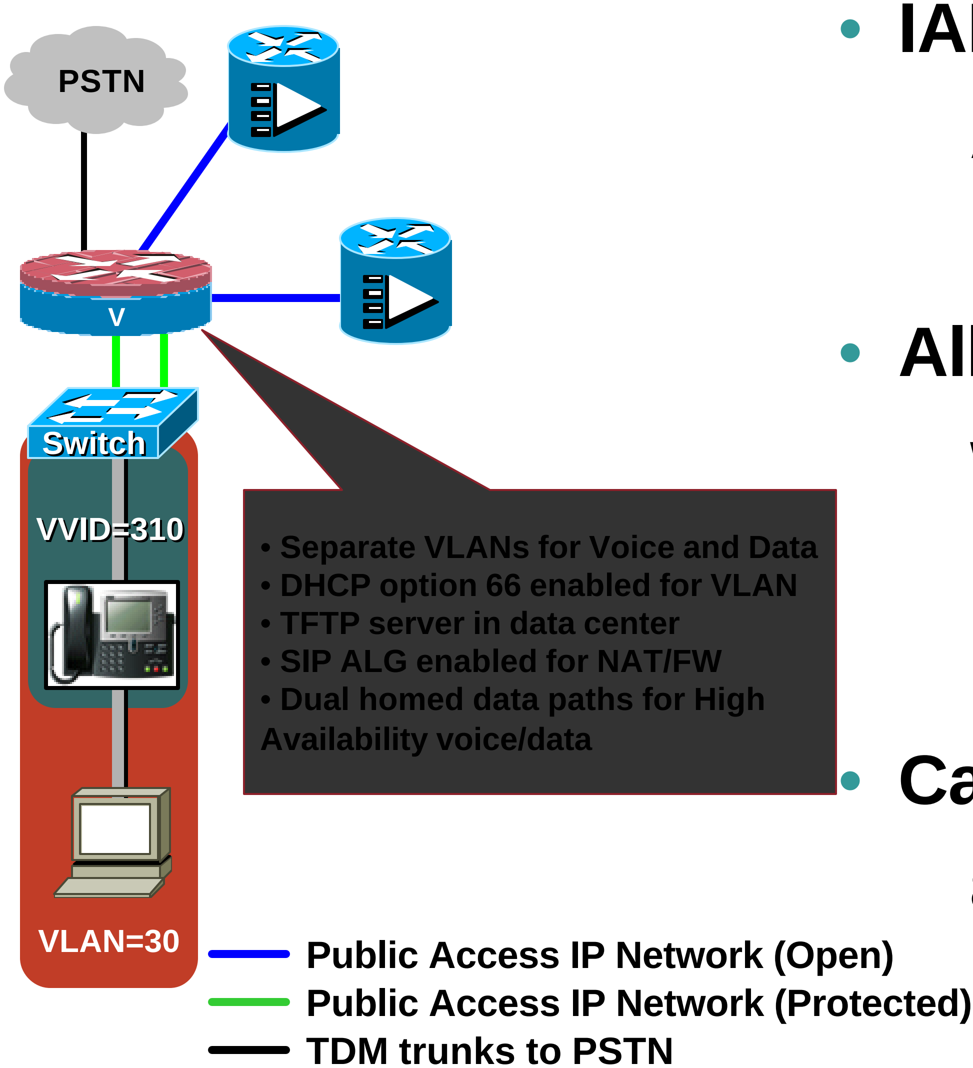
- **QoS End to End**
- **Regulatory feature readiness**
- **Wealth of Access Mechanisms**
- **Reliable network mechanisms**
- **Robust Secure Solution for service providers to deliver business telephony services to enterprise customers**

Cisco Hosted IP Telephony Advantages

- Market-leading experience in IP PBX & enterprise IP Telephony applications
- Product portfolio breadth spanning Enterprise & Service Provider
- Ability to scale as market demand dictates
- Multi-protocol, multi-environment support
- Leverage same infrastructure for other premium services such as multiservice VPNs
- Leverage the talent pool of Cisco certified networking specialists
- Opportunity to leverage Cisco’s Enterprise sales force & upgradeable installed base

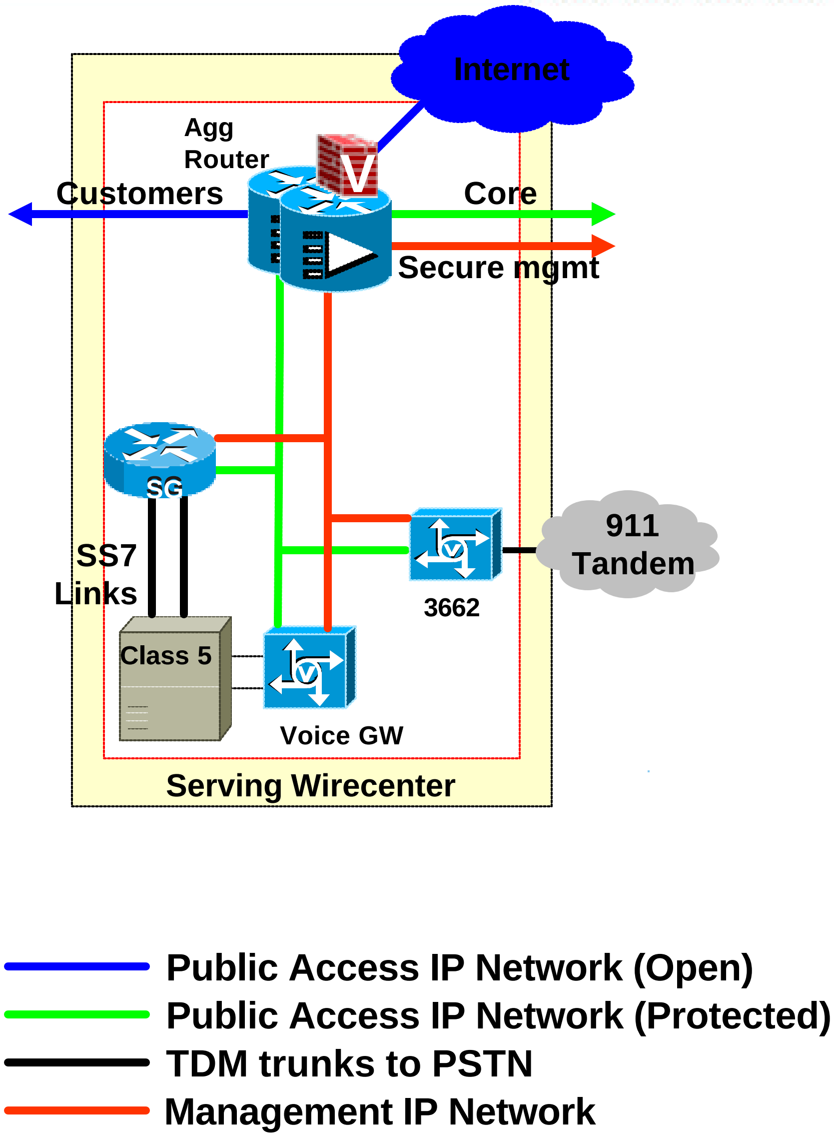
Backup

HIPT SMB CPE design



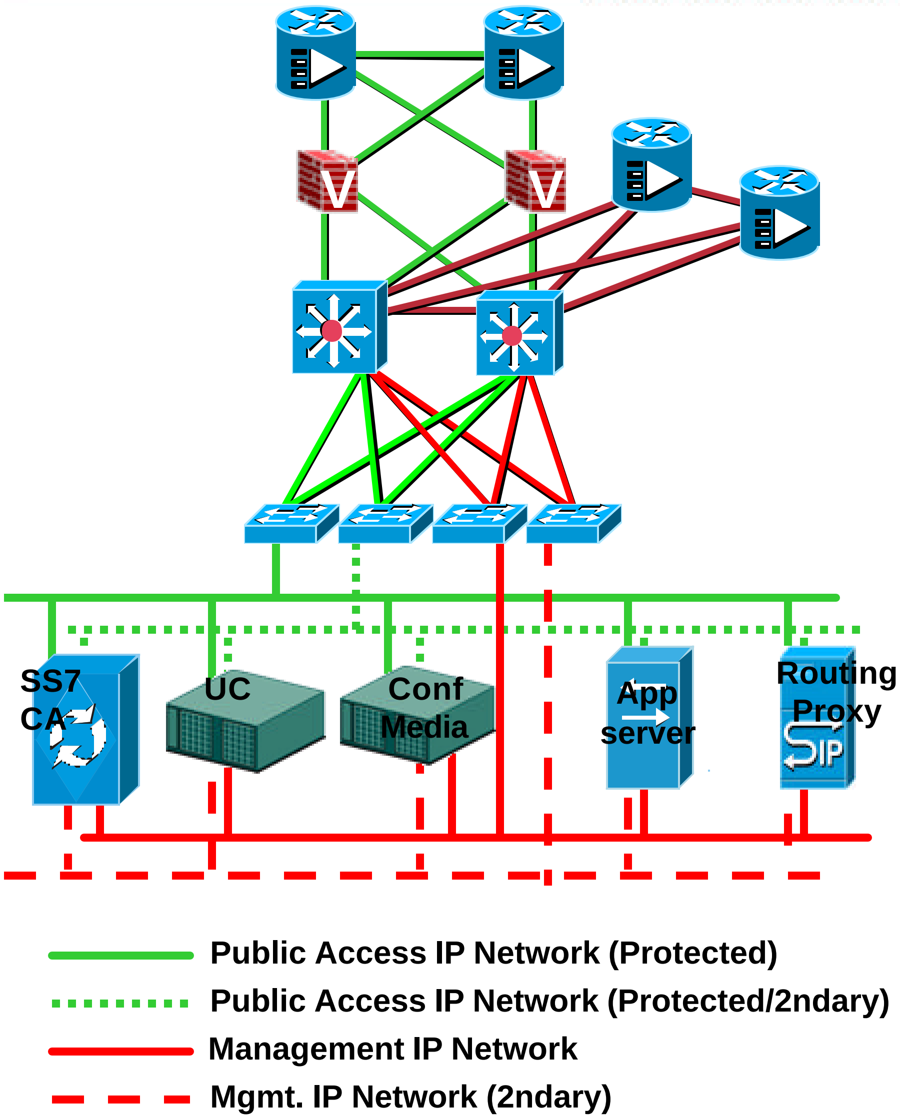
- **IAD24XX platforms**
  - Analog Lines
  - Key System-HIPT Integration
- **All IOS CPE**
  - WAN QoS
  - MPLS ability
  - NAT/FW ability
- **Catalyst Switches**
  - 802.1p/q support
  - Inline Power

HIPT POP design



- ESR 10k Agg routers
  - LI/CALEA packet replication
- PIX FW
  - Secure SP Network Elements
- Cisco VoIP GWs
  - Breadth of GW products
- ITP/SLT SG
  - SS7 link support

HIPT Data Center design



- Normal High availability methods for network
- Servers in Active/standby pairs, clusters or pooling
- DNS SRV
- Secure management traffic
- Firewall with proper protocol ALG support (SIP or MGCP)
- QoS enabled IP network

Product Stuff

Cisco Voice Equipment

Cisco.com

- CPE
  - IAD 2421/2423 - for inter-connecting analog lines
  - Catalyst Series of Modular Switches – for powered line cards for non IAD installations
  - 7940/7960 IP Phones
  - 26XX/36XX - for inter-connecting PBXs
  - ATA 186 - for SOHO and residential access
- Network
  - AS5XXX Intelligent Network Gateways - for inter-connecting to PSTN
  - MGX 8850 Voice Gateway – PSTN interconnection via IMT
  - 3660 Voice Gateway – Operator Services, Emergency Services Gateway
  - PGW 2200 – Signaling Controller for use with AS5XXX series voice gateways
  - BTS 10200 – Softswitch for use with MGX series voice gateways
  - Cisco SIP Proxy Server (CSPS) - Network device for routing, security and accounting

IAD 2421/2423

- **IAD 2421/2423**
  - 8 or 16 analog lines
  - T1 or ADSL access
  - Router with NAT, DHCP, Firewall
  - IAD Chaining via Serial Port
  - MGCP Control
  - Host Centrex and Business Lines

Cisco IP Phones 7960, 7940

7960

7940

- **Standards-based communication appliances.**
- **First in the industry IP Phone to offer investment protection of being a mutiprotocol device**
  - **Available with H.323, or Session Initiated Protocol (SIP) and, Media Gateway Control Protocol (MGCP), with system-initiated software updates.**
- **Displays accessible to XML based applications**
  - **Allows customization and branding**

26XX/36XX

- **26XX, 36XX**
  - **CAS/PRI CPE Inter-Connect**
  - **Multiple WAN interfaces**
  - **Router with NAT, DHCP, Firewall**
  - **SIP/MGCP/H.323 Control**
  - **Host PBXs via CAS or PRI trunks**
  - **Voice VPN services**

ATA 186

- **ATA 186**
  - **SOHO and residential use**
  - **2 analog line interfaces**
  - **SIP Control**
  - **Host analog lines**

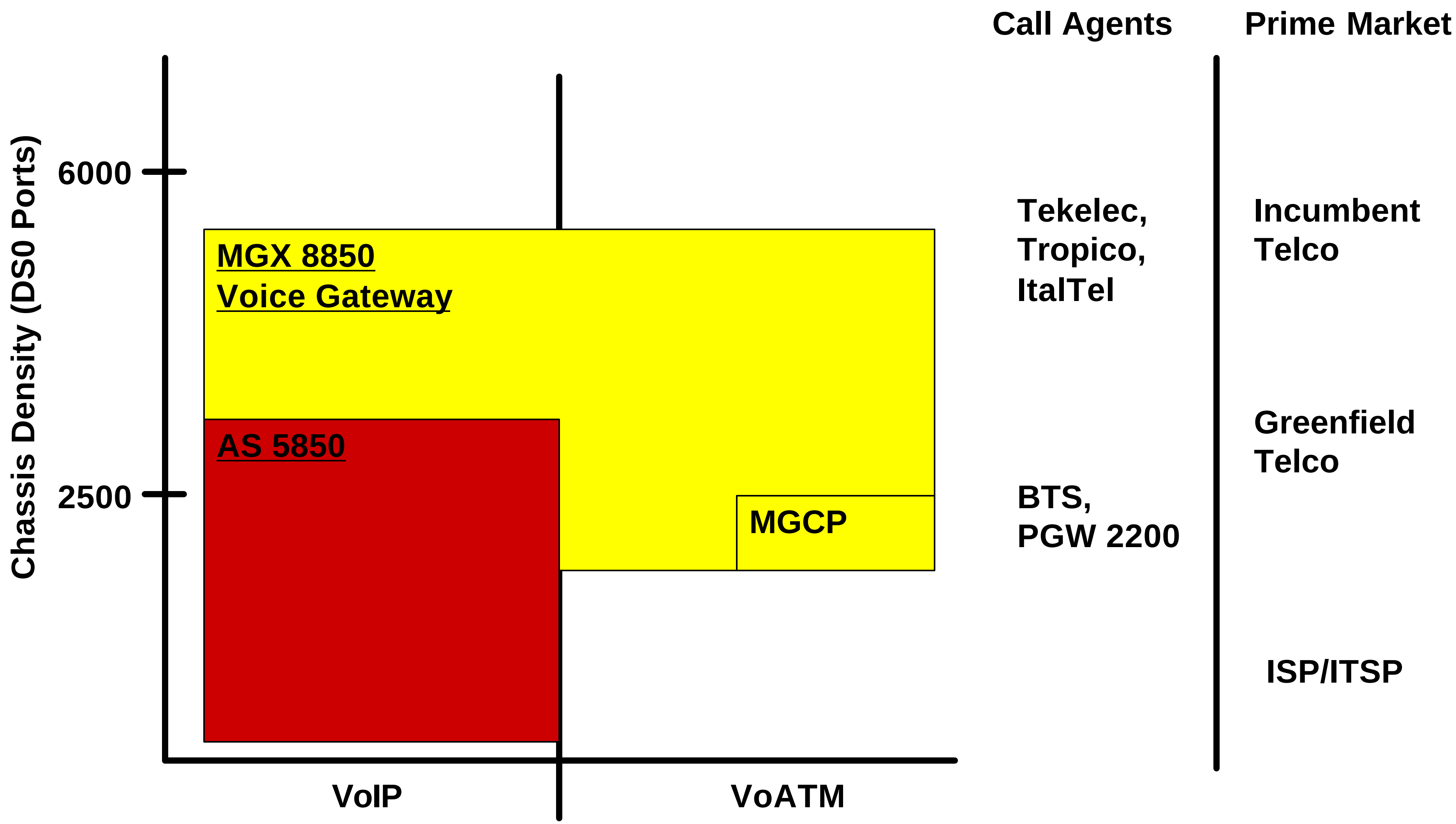
AS5XXX

- **AS5350, AS5400, AS5850 Industry's leading intelligent gateway**
  - **Full IOS feature support for H.323, SIP, and MGCP**
  - **Ideal for distributed architectures with on-board call control**
  - **Programmable for customized services**
  - **The leader for wholesale services**
  - **Control features to maximize call completions**
  - **Flexible and simple network engineering**
    - **Equal capacity for any codec. No pre-provisioning**
  - **Scalable**
    - **Up to 2688 concurrent users 14 RU chassis – three AS5850s per rack**
    - **10 Gig switching capacity**
  - **High performance – 5 msec packet latency**
  - **Highly available – 99.999%**
  - **Cisco Any Service, Any Port (ASAP)**
  - **Migrate from dial or voice services to dial *and* voice on one network**

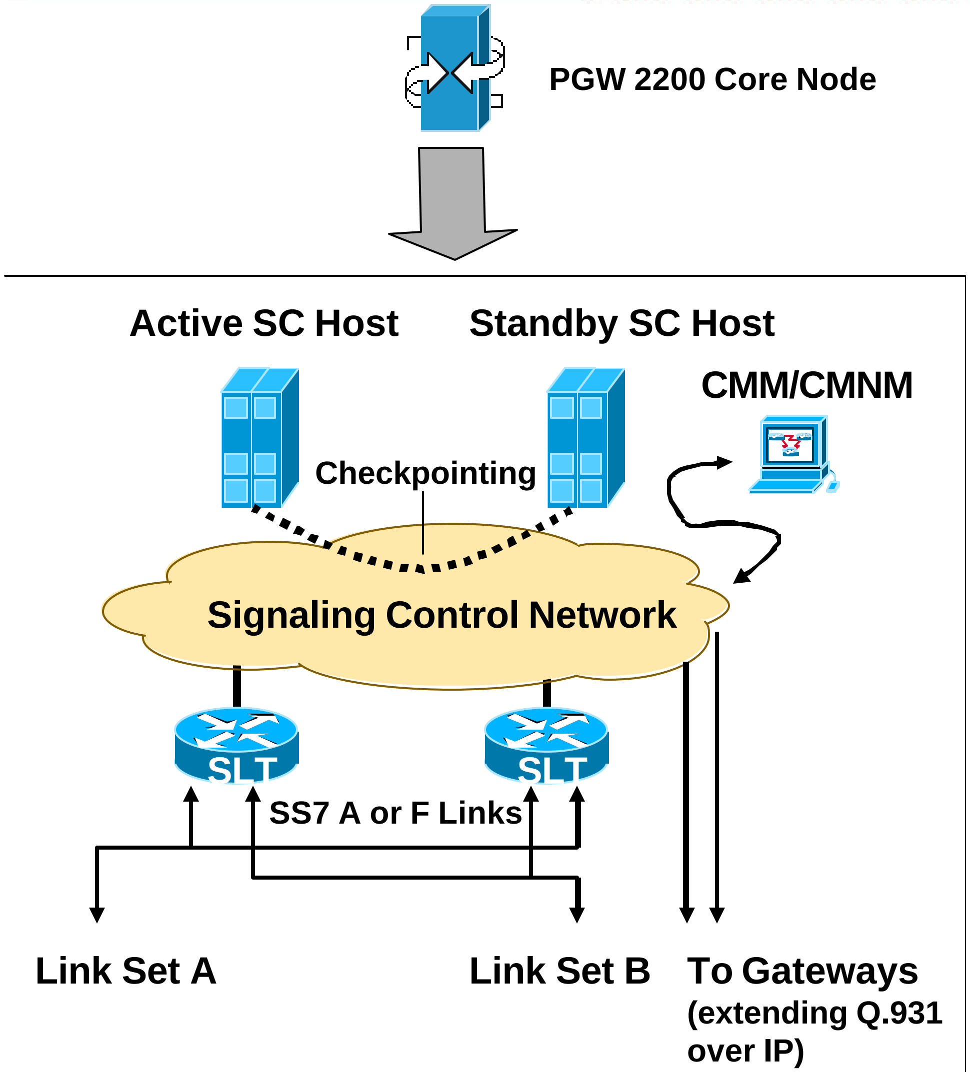
Cisco MGX 8850: Integrated VoIP and VoATM Gateway

- **1.2 Gbps / 45 Gbps Backplane Capacity**
- **Scalable packet voice gateway**
- **Integrated VoIP and VoATM Gateway**
- **Widest range of interfaces and services**
  - **DS0 to OC12**
  - **Integrated Layer-3 Routing**
  - **VoIP**
  - **VoATM (AAL1 and AAL2)**
  - **MPLS**
- **Carrier class reliability**
  - **1:1 Common Equipment Redundancy**
  - **1:N Service Module Redundancy**

Cisco High End Gateway Positioning



PGW 2200



- **Deployed since 1998, the Cisco PGW2200 Signaling Controller provides the SS7/C7 protocols for interconnecting the PSTN with a SIP based or H.323 based VoIP network.**
- **More than 50 SS7/C7 ISDN User Part (ISUP), National User Part (NUP), and Telephony User Part (TUP) protocol variants supported today**
- **Distributed redundant node architecture delivers carrier grade – 99.999% reliability**
- **Used in conjunction with the AS5XXX Intelligent Gateways**

BTS 10200

- The BTS 10200 Softswitch provides IP and ATM to PSTN calling using SS7 and MGCP. Call control and services software are provided on an open UNIX platform
- Designed for telephony-grade reliability. BTS 10200 Softswitch is carrier class - NEBS compliant, fault tolerant with fully redundant platform components
- Target Markets for Cisco BTS 10200 Softswitch:
    - Greenfield CLECs
    - Resellers moving to facilities based services
    - Facilities based CLECs
    - Fixed wireless (telco bypass)

Cisco SIP Proxy Server (CSPS)

- **SIP Proxy**

  Focus on simple routing, accounting and security

  Call-control software that enables service providers to build scalable, reliable packet voice networks

  Provides a full array of call-routing capabilities to maximize network performance in both small and large packet voice networks.

  Used to inter-connect Hosted IP Telephony Application Servers in basic Voice VPN configurations