Hosted IP Telephony

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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

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

- 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
Agenda

• Hosted IP Telephony is not Centrex
• Cisco HIPT Solution Overview
• Hosted IP Telephony works
• Partner Overview
Hosted IP Telephony Architecture

Customer Premises
- Cisco IP Phones 7960/40
- Catalyst Switches – Provides LAN switching, inline power, 802.1p/q
- ATA 186/188 for connecting to analog phones or co-located PC

IAD
- IAD 2421 & 2423 for connecting to analog phones or key systems

Serving Wirecenter
- ESR/GSR Routers

Core Agg.
- Class 5

Carrier Core
- ESR 10000
- DACS
- MUX

Infrastructure
- 26xx, 36xx, 72xx Routers
- 36xx

Public Access IP Network (Open)
- Blue

Public Access IP Network (Protected)
- Green

TDM trunks to PSTN
- Black

Management IP Network
- Red

Data Center
- MGX 8000, AS5XXX, etc

BTS/PGW
- Partner Hosted IP Telephony Application Srvr.
- Partner Unified Comms/VM

CSPS
- ITP SG (future)

Inter-Machine Trunks
- ISS7

Inter-Machine Trunks
- 911 tandem

Cisco.com
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
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

- 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.**
QoS is an end to end story

Layer 3 ToS or DSCP can be mapped to MPLS TE paths or can leverage other QoS techniques.

Layer 2 CoS mapped to ToS or DSCP

Queued via CoS Value

IP Phone 802.1p/q CoS

Layer 3 ToS or DSCP used for advanced QoS techniques (i.e. LLQ, LFI)
Hosted IP Telephone Interoperability

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

• System Overview
• Features
• Components
• Scaling Models
Business Communication Services

Service Provider Can package

Hosted PBX/IP Centrex

C-BUSINESS
- Call Hold
- Call Forwarding
- Call Waiting
- Redial
- Bridged Lines
- Call Transfer
- Speed Dialing
- Conferencing

COMTRAVELLER
- Desktop Applications in the palm of your hand

COMRIO
- Telecommuters
- Multi-offices
- Visiting sites
- Remote employees

COMOFFICE
- Personalized call treatments
- VIP categories
- Single Mailbox

COMCIERGE
- Web-based portal

COMPORTAL
- Call Center
- Display POP
- Check-in/out
- Statistics

COMMERCHANT
- IE INTEGRATION
  Click to call from Any web page

Sylantro Solutions

Access Options

Advanced Options

Mobility
Agenda

• System Overview
• Features
• Components
• Scaling Models
## Sample Feature Set

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

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<th>Feature</th>
<th>Communications Portal</th>
<th>Management Portal(S)</th>
<th>Services (Page)</th>
</tr>
</thead>
<tbody>
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<td>Call Hold</td>
<td>My Home (Page)</td>
<td>Intro</td>
<td>· Intro</td>
</tr>
<tr>
<td>Message Waiting Lamp</td>
<td>· How to Reach Me</td>
<td>· General</td>
<td>· General</td>
</tr>
<tr>
<td>Call Logs</td>
<td>· Missed Calls</td>
<td>· Billing Address</td>
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<tr>
<td>Missed Calls</td>
<td>· Favorites</td>
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<td>Incoming Calls</td>
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<td>Outgoing Calls</td>
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<td>Call Park</td>
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<tr>
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<td>Help (Pages)</td>
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<tr>
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<td>· Context Sensitive Help</td>
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<td>DID</td>
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<td>· Online Access to User's Guide</td>
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<td>MANAGEMENT PORTAL(S)</td>
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<td>Favorites Contacts</td>
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<td>Office Administration Portal</td>
<td>Office Administration Portal</td>
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<td>(Speed Dialing)</td>
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<td>· Secure Login</td>
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<tr>
<td>Flexible Feature Mapping</td>
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<td>· Forgotten Password Help</td>
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<td>Hunt Groups</td>
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<td>Employees (Home Page)</td>
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<td>Last Number Redial</td>
<td>· Phone Numbers</td>
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<td>· Company Directory</td>
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<td>· New</td>
<td>· New</td>
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<tr>
<td>Call Forwarding</td>
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<td>Do Not Disturb</td>
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<td>Contacts (Page)</td>
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<td>Handset Volume control</td>
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<tr>
<td>Speaker Volume Control</td>
<td>MANAGEMENT PORTAL(S)</td>
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<td>· New</td>
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<tr>
<td>Hands-Free Dialing</td>
<td>OA Setup Screens</td>
<td>Feature codes</td>
<td>Feature codes</td>
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<tr>
<td>Hook Flash</td>
<td>· Secure Login Welcome (Page)</td>
<td>E911 routing</td>
<td>E911 routing</td>
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<tr>
<td>Click-to-Call</td>
<td>· Company Info</td>
<td>Geographic-specific routing</td>
<td>Geographic-specific routing</td>
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<tr>
<td>Music on Hold</td>
<td>· Billing Address</td>
<td>· Adapters</td>
<td>· Adapters</td>
</tr>
<tr>
<td>Bridged Line Appearances</td>
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<tr>
<td>Directed Pickup</td>
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<td>· Assign Extensions</td>
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<tr>
<td>Billing Codes</td>
<td>· Help</td>
<td>· Answer Order</td>
<td>· Answer Order</td>
</tr>
</tbody>
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Service Provider Example

Control “find me” feature, phone settings, call logs

Manage “find me” feature

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

On-demand Conferencing

“chat” real-time with online users through Instant Messenger

On-demand Conferencing

“chat” real-time with online users through Instant Messenger

Manage “find me” feature

On-demand Conferencing

“chat” real-time with online users through Instant Messenger
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

Service Creation
- HTTP/SOAP
- XTM/L/JAVA
- HTTP/VXML
- Device SDK

3rd Party Call Control
- JAVA/API
- SIP

Applications
- ComMerchant
- ComCierge
- ComTraveler
- ComRIO
- c-Business (PBX/Centrex Core)

Application Execution Engine

SIP Services & Call Agent

CPE & End Point Device Management

Voice Services

PSTN Interface Control

SIP, MGCP

SIP, VXML, CAS

SIP

Service & Platform Management
- Tenant & End User
- VAR
- Service Provider

Portals
- JAVA, HTML
- CORBA, SNMP, CDRs

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Agenda

• System Overview
• Features
• Components
• Scaling Models
Scaling – today

Number of Users
Cost of Redundancy:
Population impacted in case of failure*: 20%

80K
50%

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

**Number of Users**
- 80K
- 80%
- 100%

**Cost of Redundancy:**
- 50%

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

*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

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

Up to 50,000 users
200K BHCA

Up to 250,000 users
1M BHCA

Up to 1.75M users
7 M BHCA

1 Network Server
5 App Servers

1 Network Server
1 App Server

7 Network Servers
35 App Servers
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

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

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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

- Separate VLANs for Voice and Data
- DHCP option 66 enabled for VLAN
- TFTP server in data center
- SIP ALG enabled for NAT/FW
- Dual homed data paths for High Availability voice/data

VLAN=30
VVID=310
Public Access IP Network (Open)
Public Access IP Network (Protected)
TDM trunks to PSTN
PSTN
Switch
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

• 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

- Standards-based communication appliances.
- First in the industry IP Phone to offer investment protection of being a multiprotocol 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

Chassis Density (DS0 Ports)

- **MGX 8850**
  - Voice Gateway
  - Incumbent Telco
  - Greenfield Telco

- **AS 5850**
  - BTS, PGW 2200

- **MGCP**
  - Tekelec, Tropico, ItalTel

Call Agents

Prime Market

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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.
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