Migrating to IP Telephony from Traditional Voice Technologies

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Agenda

- Introduction
- Dial Plan and PBX Migration
- Voicemail Migration
- Unified Messaging Integration
- Moving the Users
- Migration Examples
Introduction

- Why can’t we move 12,000 users in a day
- No flash cut,
- Plan/Test ... but before you do it, talk to you users, perform station reviews early in the process
- Don’t be afraid, IPT doesn’t bite
- Raffiky says “Change is good”
- Remember you are not just moving to another Phone System
- Do not perpetuate Chaos
- There are Tools Available to ease Migration
Typical Deployment Cycle

Complete Business Case
Complete Network Design
Finalize Migration Strategy
Identify Productivity Tools

Migrate Users
Hybrid Centrex/ IP Telephony Users

Pilot to Validate Design and Business Case

Migration Completed
Full Benefits Realized
Dial Plan

• It’s called Dial **PLAN** not Dial Improvise…
  So ..... PLAN IT

• Considerations
  Users want to keep phone numbers
  Local Number Portability
  Centrex Tie/CallFWD
  Integrated Dial Plan

• Other Tools that can help during migration
  IP Softphone, Directories, Personal Fast Dials
  Active Assistant, VMO/VMI
Local Number Portability

• 1. Local number portability key to migrating from Centrex to ISDN (with same Telco or a CLEC)
  http://www.crtc.gc.ca/cisc/eng/Portable.HTM

• 2. Alternate for Centrex users is to use a call forward feature for inbound on key published phone numbers where LNP not available.
CallManager and PBX Migration (The Basic Concept)

AVVID Network

Legacy PBX

Unity

PSTN

Legacy Voicemail System

Trunk between CCM and PBX

Gateway

User Migration

CallManager

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5 Digit Dialing Between all Users in a Site (IP Phones & Call Centre non-IP Phones)

- Site A IP Telephony
- Canadian Financial Institution Campus Network
- Call Manager Cluster
- Cisco Unity Campus VM and AA For Centrex & IPT
- Centrex TIE over Megalink
- PSTN
- DMS100

Allows Centrex phones to dial 5 digits to call between IP Phones and Call Centre non-IP Phones
External Route Elements in CallManager

**Route Pattern**
- Matches dialed number for external calls
- Performs digit manipulation (optional)
- Points to a Route List for routing

**Route List**
- Chooses path for call routing
- Points to prioritized Route Groups

**Route Group**
- Performs digit manipulation
- Points to the actual devices

**Devices**
- Gateways (H.323, MGCP)
- Gatekeeper
- Inter-cluster Trunk (remote CM)
Partitions/ Calling Search Spaces: Analogy with Subnets/ Access Lists

**Partition – “Where you are”**
- Collects devices with similar “reachability” characteristics
- Items placed in partitions: Directory Numbers (DN), Route Patterns, Voice Mail Ports...

**Calling Search Space – “Where you may call”**
- Set of rules to set call restrictions/ permissions
- Defines which partitions a device may search to reach a dialed number
- Is assigned to IP phones, GWs
Example of Composite Dial Plan View
Single Site PSTN Access

Calling Search Spaces
- Internal Only
- Local
- National
- International

Partitions
- Internal
  - All IP Phones
    - 911
    - 9.911

Route Lists
- Local
  - 9.[2-9]XXXXXX

Route Groups

Devices
- PSTN
- PSTN RL
- PSTN RG

Route Patterns

Calling Search Space assigned to IP Phone based on policy
Voicemail Migration

- SMDI Splitting with VG248
- DPA
- Overview of Networking
  - IVC, SMTP AMIS and Bridge Users
- AMIS
  - AMIS Definition, Sample Conversation, Bridgehead concept
- Analog Octelnet
  - Unity Bridge
- UM Considerations
- VPIM
SMDI for Voicemail—VG248

- Multiple SMDI links per Cisco CallManager
- SMDI fail-over capability
- Voicemail location independence
SMDI with PBX—VG248

• Multiple SMDI sources
• Smooth migration
• Single SMDI link to voicemail
**DPA for Octel with Avaya or Nortel PBX**

*Skinny Client Control Protocol, See Session VVT-220 for More Information*
DPA for Octel with CallManager

- DPA
- Octel Voicemail
- 24 DSE Interfaces
- SCCP
- CallManager
- Gateway
- PSTN

Allows Digital Set-Emulation Integration to CallManager
Unity Networking Overview

- Locations are Cisco Unity-specific objects that are used in networking.
- There are two types of locations: primary locations and delivery locations.
- Each Cisco Unity server is associated with one location—referred to as the default or primary.
- Each primary location contains the network information that identifies the Cisco Unity server to other Cisco Unity servers and other voice messaging systems.
Locations

Primary Location

Delivery Locations

1. N1
2. N2
3. N3

Bridge

Unity

AMIS

Unity

Unity

Unity
AMIS

- AMIS
  - Blind Addressing
  - AMIS Subscribers

- AMIS Bridgehead Model
  - Digital Network multiple Cisco Unity systems
  - Dedicated Cisco Unity for AMIS Traffic
  - Home Internet subscriber and AMIS users on that Cisco Unity
Cisco Unity: AMIS-A

- Legacy PBX
- Legacy voice mail system
- Analog Lines
- AMIS connection, via analog lines
- Legacy Phone
- PSTN
- T-1 line
- 3600 Router
- CallManager
- IP Phones
- Cisco Unity Server
- Microsoft Exchange message store
- Workstation with Microsoft Outlook

Legacy voice mail system connected via analog lines to the AMIS connection, and analog lines also connect to the Legacy PBX. The Legacy Phone is connected to the PSTN through a T-1 line. The 3600 Router connects to the CallManager, IP Phones, and Cisco Unity Server. The Cisco Unity Server is connected to the Microsoft Exchange message store and a workstation running Microsoft Outlook.
AMIS Conversation

- (Ringing)
- CCD*041116*05*1721#408#2327200#05*49*
  1630015555#9995#07*05
(Voice message plays here)*03417*05*049033
<table>
<thead>
<tr>
<th>Digits Sent</th>
<th>Sent By</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>N/A N/A</td>
<td>N/A</td>
<td>Ringing.</td>
</tr>
<tr>
<td>CC</td>
<td>Originating system</td>
<td>DTMF “C” tone for call setup. In this case, the “C” tone is repeated because no response is received within the Timeout period.</td>
</tr>
<tr>
<td>*041116</td>
<td>Destination system</td>
<td>DTMF “D” tone response.</td>
</tr>
<tr>
<td></td>
<td>Originating system</td>
<td>*: Start Digit. 04: Frame Length (4 digits, not including * and 04). 1: Function Code (1 = Start Session). 1: Data (always a 1 if a Start Session frame). 16: Checksum (sum of 10 + 4 + 1 + 1 = 16).</td>
</tr>
<tr>
<td>*05</td>
<td>Destination system</td>
<td>*: Start Digit. 0: Response Code (0 = accept and continue with Data frame). 5: Check Digit (always a 5 with a Response Code of 0).</td>
</tr>
</tbody>
</table>
AMIS Conversation

*1721#408#2327200#05  Originating system

*: Start Digit.
17: Frame Length (17 digits, not including * and 17).
2: Function Code (2 = System Number).
1#408#2327200#: Data
 (originating system phone number), plus # (the terminating character, needed because the Data is variable length).
05: Checksum (sum of all digits in the frame. 1 + 7 + 2 + 1 +
12 + 4 + 10 + 8 + 12 + 2 + 3 + 2 + 7 + 2 + 10 + 10 +
12 = 105, yielding a Checksum of 05).

*49  Destination system

*: Start Digit.
4: Response Code (4 = accepting messages but node response not allowed).
9: Check Digit (always a 9 with a Response Code of 4).
<table>
<thead>
<tr>
<th><strong>AMIS Conversation</strong></th>
</tr>
</thead>
<tbody>
<tr>
<td><em>1630015555#9995#07</em> Originating system</td>
</tr>
<tr>
<td>*: Start Digit.</td>
</tr>
<tr>
<td>3: Function Code (3 = Message Information).</td>
</tr>
<tr>
<td>0: Data, NDR Reason (0 = new message or node response message).</td>
</tr>
<tr>
<td>5555#: Data <em>(originating mailbox)</em>, plus # (the terminating character).</td>
</tr>
<tr>
<td>07: Checksum (sum of all digits in frame. 1 + 6 + 3 + 10 + 10 +1 + 5 + 5 + 5 + 12 + 9 + 9 + 9 + 5 + 12 = 107, yielding a Checksum of 07).</td>
</tr>
<tr>
<td><em>05</em> Destination system</td>
</tr>
<tr>
<td>5: Check Digit (always a 5 with a Response Code of 0).</td>
</tr>
</tbody>
</table>
### AMIS Conversation

<table>
<thead>
<tr>
<th>Conversation</th>
<th>PLAY MESSAGE</th>
<th>Example</th>
</tr>
</thead>
</table>
| *03417       | Originating system | *: Start Digit.  
03: Frame Length (3 digits, not including * and 03).  
4: Function Code (4 = **End Message**).  
17: Checksum (sum of all digits in frame. 10 + 3 + 4 = 17). |
| *05          | Destination system | *: Start Digit.  
0: Response Code (0 = accept and continue with Data frame).  
5: Check Digit (always a 5 with a Response Code of 0). |
| *049033      | Originating system | *: Start Digit.  
04: Frame Length (4 digits, not including * and 04).  
9: Function Code (9 = **End Session**).  
0: Data (0 = no more messages; normal termination disconnect).  
33: Checksum (sum of all digits in frame. 10 + 4 + 9 + 10 = 33).  
N/A Destination system No Response frame sent; call is disconnected as a successful call. |
Unity Bridge

- Networking GW between Unity and Octel
- The Bridge looks like another node in the network
- Communication between the Bridge and Octel is done using Analog OctelNet (based on DTMF tones)
- Communication between Unity and the Bridge is digital.
Cisco Unity With Unity Bridge
—Multi Node Octel

Cisco Unity IP connection to Cisco Unity Bridge

• The Bridge maintains:
  • A table for the Octel node that contains the Octel server name, unique serial number, and phone number.
  • Another table for the Unity node that contains the Cisco Unity server name, assigned serial number, and domain name.

• Octel nodes correspond to locations in Cisco Unity
Addressing options:

- **Blind addressing:**
  - No information about the individuals associated with the other Octel nodes (such as their extensions and recorded voice names).
  - To address a message, subscribers enter the delivery location Dial ID and the remote mailbox number of the recipient.
  - Cisco Unity sends the message without confirming that the recipient exists.

- **Bridge subscribers**
  - Cisco Unity has information about the remote users, such as their names, extensions, and recorded voice names.
  - Unity Subscribers address messages to Bridge subscribers the same way they do to regular Cisco Unity subscribers—by extension or by spelling the name of the recipient.
VPIM

- Allows the interchange of voice, fax and text messages between disparate voice messaging systems over a TCP/IP data network.
- Wraps encoded voice messages in MIME message parts, and uses SMTP to transport them over TCP/IP networks.
- The VPIM profile requires that these multi-part messages be formatted and used according to a specific set of conventions and rules as defined in these protocols.
VPIM

- VPIM messages are made up of one or more parts, at least one of which must be a voice message, and all of which are MIME encoded.

- The profile also allows adding optional parts for spoken name, forwarded messages and fax messages.

- Subscriber on one system sends VPIM mail to subscriber on the other system by addressing a SMTP message to TelePhoneNumber@Domain.com
Example # 1- To send voice mail to user Andres Martinez with telephone number 206-256-1234 at Cisco.com, VPIM compatible system will send VPIM message to 2062561234@cisco.com using SMTP protocol.

Example # 2- To send voice mail to user Andres Martinez with extension 1234 at telephone number 206-256-3000 at Cisco.com, VPIM compatible system will send VPIM message to 2062563000+1234@cisco.com using SMTP protocol.
# Protocol Comparison

<table>
<thead>
<tr>
<th>AMIS</th>
<th>Analog Octel (BRIDGE)</th>
<th>VPIM</th>
</tr>
</thead>
<tbody>
<tr>
<td>Analog Protocol</td>
<td>Analog Protocol</td>
<td>Digital Protocol</td>
</tr>
<tr>
<td>Most legacy systems support it.</td>
<td>Supported only by Octel Systems</td>
<td>Supported by newer voice mail systems including Avaya Interchange.</td>
</tr>
<tr>
<td>Does not support name confirmation</td>
<td>Supports name confirmation</td>
<td>May support name confirmation</td>
</tr>
<tr>
<td>Does not support fax</td>
<td>Support fax</td>
<td>Supports fax</td>
</tr>
<tr>
<td>Cost of ownership is Highest</td>
<td>Cost of ownership is Medium.</td>
<td>Cost of ownership is lowest.</td>
</tr>
<tr>
<td>5 minutes message to 10 recipients will take around 50 minutes of phone call.</td>
<td>5 minutes message to 10 recipients will take around 5 minutes of phone call.</td>
<td>5 minutes message to 10 recipients will take 0 minutes of phone call as messages are sent using SMTP.</td>
</tr>
</tbody>
</table>
UM Considerations

- AD integration
- Schema Updates
- Message Store
- The MCSE is your Friend
- VMO/VMI
Active Directory Integration Details

AD Terminology

**Forest:**
- Common Schema
- Inbuilt security trusts

**Tree:**
- Common Schema
- Common Namespace
- Multiple domains

**Domain:**
- Common Schema
- Common Namespace
- Common Security Policy

**Domain Controller:**
- Holds all domain information
- Provides authentication services

**“Dot” Domain**

**Domain Controllers**

**Schema Master**

**Child Domains**
- emea.ese.lab
- amer.ese.lab
- ese.lab

**“Dot” Domain**

**Domain:**
- avvid.info
Agenda  End User Migration

- Call Mgr cluster
- Dial plan
- gateways/PSTN
- voicemail integration
- voicemail templates
- CDR reporting
- network QoS
- implement Security policies

- Identify
- Users for move
- Station Review
  (eg. Phone type, special features,…)  
- Populate spreadsheet with users
- Bulk Load spreadsheet into Call Mgr DB and LDAP directory

Approx. 2-3 days prior to each move

- Installer unpacks and installs phone at desk
- phone auto-registers with temp number (eg. X1001) – no external calling allowed
- installer then picks up phone and TAPs service collects proper phone number
- TAPS service reconfigures phone with proper bulk loaded profile

(5 phones every 20 minutes per installer)

- User can bring their own phone to training (plug n’ play)
- user is trained on their own phone & voicemail box
- opportunity during training to correct any problems (eg. Name display)
- On-line tutorial for follow up training option
Bulk Administration Tool (BAT)

Allows bulk adds, deletes and updates of devices, lines and users

- Add or delete users by the thousands
- Operates on a comma-separated-value file
- Created from existing databases and directories
- Comes with spreadsheet macro
- Tool for Auto-registered Phone Support (TAPS) – updates auto-registered phones with predefined configurations
<table>
<thead>
<tr>
<th>B</th>
<th>C</th>
<th>D</th>
<th>E</th>
<th>F</th>
<th>G</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>Description</strong></td>
<td><strong>Location</strong></td>
<td><strong>Directory Number 1</strong></td>
<td><strong>Display 1</strong></td>
<td><strong>Forward Busy Destination 1</strong></td>
<td><strong>Call Pickup Group 1</strong></td>
</tr>
<tr>
<td>7444 - John Smith</td>
<td>Creekside</td>
<td>7944</td>
<td>John Smith</td>
<td>7000</td>
<td>Finance</td>
</tr>
<tr>
<td>7655 - Kate White</td>
<td>Creekside</td>
<td>7655</td>
<td>Kate White</td>
<td>7000</td>
<td>Finance</td>
</tr>
<tr>
<td>7301 - Mike Jones</td>
<td>Creekside</td>
<td>7301</td>
<td>Mike Jones</td>
<td>7000</td>
<td>Finance</td>
</tr>
<tr>
<td>7340 - Sally Barnes</td>
<td>Creekside</td>
<td>7340</td>
<td>Sally Barnes</td>
<td>7000</td>
<td>Finance</td>
</tr>
<tr>
<td>7642 - Cathy Ember</td>
<td>Creekside</td>
<td>7642</td>
<td>Cathy Ember</td>
<td>7000</td>
<td></td>
</tr>
</tbody>
</table>
**TAPs Process: Step 1 Bulk Load**

- Spreadsheet with users imported into Call Manager database

---

**Call Manager Publisher**

<table>
<thead>
<tr>
<th>Type</th>
<th>User</th>
<th>Phone/MAC</th>
</tr>
</thead>
<tbody>
<tr>
<td>Bulk Load - BAT</td>
<td>John Smith</td>
<td>7944</td>
</tr>
<tr>
<td>Bulk Load - BAT</td>
<td>Kate Jackson</td>
<td>7655</td>
</tr>
<tr>
<td></td>
<td></td>
<td></td>
</tr>
</tbody>
</table>
**TAPs Process: Step 2 Auto-Registers**

- Phone is plugged in and
- auto-registers with x1002
- Phone Software automatically updates
- Calls restricted internally
- Installer only needs to be at desk (no access to wiring closet required)

### Call Manager Publisher

<table>
<thead>
<tr>
<th>Type</th>
<th>User</th>
<th>MAC</th>
<th>Phone</th>
</tr>
</thead>
<tbody>
<tr>
<td>Bulk Load - BAT</td>
<td>John Smith</td>
<td>dummy</td>
<td>7944</td>
</tr>
<tr>
<td>Bulk Load - BAT</td>
<td>Kate Jackson</td>
<td>dummy</td>
<td>7655</td>
</tr>
<tr>
<td>Auto-Registered</td>
<td>Unknown</td>
<td>00c012341234</td>
<td>1001</td>
</tr>
<tr>
<td>......</td>
<td>......</td>
<td>............</td>
<td>.....</td>
</tr>
</tbody>
</table>
**TAPs Process: Step 3 Complete Registration**

- Calls TAPS service on Call Mgr
- Prompted for to enter phone number to complete registration
- Security feature to prevent hijacking phone numbers

---

**Call Manager Publisher**

<table>
<thead>
<tr>
<th>Type</th>
<th>User</th>
<th>MAC</th>
<th>Phone</th>
</tr>
</thead>
<tbody>
<tr>
<td>Final Configured</td>
<td>John Smith</td>
<td>00c012341234</td>
<td>7944</td>
</tr>
<tr>
<td>Bulk Load - BAT</td>
<td>Kate Jackson</td>
<td>dummy</td>
<td>7655</td>
</tr>
<tr>
<td>Auto-Registered</td>
<td>Unknown</td>
<td>00c012341234</td>
<td>1001</td>
</tr>
<tr>
<td>......</td>
<td>......</td>
<td>........</td>
<td>.....</td>
</tr>
</tbody>
</table>

**Process Steps:**
- Please enter your phone number
- Please re-enter your phone number
- Thank you your registration will be completed now
- Update bulk entry with New mac address
- Delete this record
End User Training

• User can bring phone to Class
  User trains on their own phone, voicemail
  any corrections to config can be done at class
  Option to give user phone at class and have them install at desk themselves
  Better retention of information if on their own phone

• Follow-up On-Line Phone Tutorial
  Full animated & audio interactive tutorial
Migration Examples
Canadian City ... Previous Design: Separate Data, Centrex Voice

Challenges:
- High operational costs
- Long lead times for changes
- MACs cost $$$
- Single points of failure
- Telco slow to deploy new features
- Centrex not available everywhere
New City Design: Converged Voice/Data

City Hall
- Preserved Existing Voicemail
- E911 Emergency Responder
- Tertiary Call Manager
- Call Manager Primary (Load Balancing)

City IP MAN
(Fibre, ADSL, 802.11b wireless)

PSTN

Backup Site Regional Office
- E911 Emergency Responder
- Secondary Call Manager (Load Balancing)

Small Site
- <24 users
- Survivable Remote Site Telephony Router (1751)
- Catalyst 3524PWR-XL
- In-line power, QoS
- Analog Terminal Adapters (ATA186)

Medium Site
- Cisco 3725 SRST Router
- Catalyst 354XL-PWR

Large Site
- >100 users
- Diverse MAN Paths
- To Catalyst 3524XL-PWR 1751 for 911
Campus Dial Plan Example

- **CSS**
  - CSS-BldgA
  - CSS-BldgB
  - CSS-BldgC
  - CSS-BldgD
  - CSS-BldgE
  - CSS-BldgF
  - CSS-BldgG
  - CSS-BldgH
  - CSS-BldgJ

- **Partition**
  - BldgA
  - BldgB
  - BldgC
  - BldgD
  - BldgE
  - BldgF
  - BldgG
  - BldgH
  - BldgJ

- **Route-List**
  - RL-911-BldgA
  - RL-911-BldgB
  - RL-911-BldgC
  - RL-911-BldgD
  - RL-911-BldgE
  - RL-911-BldgF
  - RL-911-BldgG
  - RL-911-BldgH
  - RL-911-BldgJ

- **RG**
  - RG-911-BldgA
  - RG-911-BldgB
  - RG-911-BldgC
  - RG-911-BldgD
  - RG-911-BldgE
  - RG-911-BldgF
  - RG-911-BldgG
  - RG-911-BldgH

- **SystemDNs**
  - Line1-10digit
  - Line2-10digit
  - M1DNs

- **CentrexDNs**
  - ICT-CM-3XX

- **TOR-LocalPSTN**
  - TOR-LDPSTN

- **Translation Pattern**
  - 4169YYXXXX -> Prefix 9, 4167YYXXXX -> Prefix 9
  - 4163YYXXXX -> Prefix 9, 4168YYXXXX -> Prefix 9
  - 4168QXXXXX -> Prefix 9, 4169QXXXXX -> Prefix 9
  - 4169ZXXXX -> Prefix 9

- **Route Pattern**
  - 9.[2-9]XXXXXXXXX
  - 9.611
  - 9.411

- **Unity VM Pilot, Ports, MWIOn, MWIOff, Meet Me Conference**

- **Migrating Large Networks to IPT**
Empowering the Internet Generation