SIP Trunking Configuration Guide
for
Cisco Unified Communications Manager
v7.1.3
TABLE OF CONTENTS

1 Audience ........................................................................................................................................5
2 Introduction ....................................................................................................................................5
3 Network Topology .........................................................................................................................6
4 System Components .....................................................................................................................7
  4.1 Hardware Components ............................................................................................................7
  4.2 Software Requirements ...........................................................................................................8
5 Features .........................................................................................................................................8
  5.1 Supported Features ..................................................................................................................8
  5.2 Unsupported Features ..............................................................................................................8
6 Caveats and Limitations ................................................................................................................9
7 Configuration ...............................................................................................................................9
  7.1 CUCM SIP Trunk Call Flows ..................................................................................................9
  7.2 SIP Trunk IP Address Assignments .....................................................................................10
  7.3 Cisco Unified Border Element (CUBE) Configuration .........................................................11
  7.4 Configuring the Cisco Unified Communications Manager (UCM) Overview .........................15
    7.4.1 Voice Services ................................................................................................................15
    7.4.2 Call Processing ................................................................................................................15
    7.4.3 Voicemail Services ........................................................................................................15
    7.4.4 Quality of Service ...........................................................................................................15
    7.4.5 Detail Communications Manager Configurations .............................................................16
  7.5 SIP Message Trace Examples ..............................................................................................57
    7.5.1 Incoming SIP INVITE – PSTN (E-SBC) to Cisco CUBE .................................................57
    7.5.2 Outgoing SIP INVITE – Cisco CUBE to PSTN (E-SBC) ..................................................57
    7.5.3 Transfer SIP UPDATE – Cisco CUBE to PSTN (E-SBC) ................................................58
    7.5.4 Call Forward SIP INVITE – Cisco CUBE to PSTN (E-SBC) ............................................59
  7.6 Full Detail IOS Configurations ..............................................................................................59

Table of Figures

Figure 1 – Cox Fiber Network ...........................................................................................................5
Figure 2 – Reference Network Architecture ..................................................................................6
Figure 3 – SIP Trunk Lab Reference Network ................................................................................7
Figure 4 – Geo-Redundant CUCM Network ................................................................................9
Figure 5 – Originating Call Flow Overview ..................................................................................9
Figure 6 – Terminating Call Flow Overview ................................................................................10
Figure 7 – Geo-Redundant Signaling Paths ................................................................................14
Figure 8 – Device Pools ................................................................................................................16
Figure 9 - Cisco Unified CM’s ........................................................................................................16
Figure 10 - Cisco Unified CM Groups ...........................................................................................17
Figure 11 – Regions .......................................................................................................................17
Figure 12 - Physical Locations .......................................................................................................18
Figure 13 - Enterprise Parameters Configuration ............................................................................19
Figure 14 - Enterprise Parameter Configuration – Continued .......................................................20
Figure 15 - Enterprise Parameters Configuration – Continued ....................................................21
Figure 16 - Enterprise Parameters Configuration – Continued ....................................................22
Figure 17 - Enterprise Parameters Configuration – Continued ....................................................23
Figure 18 - Enterprise Parameters Configuration – Continued ....................................................24
Figure 19 - Enterprise Phone Configuration ..................................................................................25
## Table of Tables

Table 1 – IP Addresses .......................................................................................................................... 10  
Table 2 – SIP Trunk Failover Timers .................................................................................................. 14
1 Audience
This document is intended for the SIP Trunk customer’s technical staff and Value Added Retailer (VAR) having installation and operational responsibilities.

2 Introduction

This Configuration Guide describes the configuration steps for Cox SIP Trunking with the Cisco Unified Communications Manager (CUCM) 7.1.3(b) and the Cisco Unified Border Element (CUBE) for connectivity to Cox’s SIP Trunking service. Cox’s SIP Trunking provides both inbound and outbound call services replacing traditional ISDN PRI services.

Cox SIP Trunking is a scalable and efficient IP trunking telecommunication solution for your business that provides all the traditional services such as Direct Inward Dialing, Hunting, Calling Name, Calling Number, Local/Long Distance and Business Continuity options, including:

- Burstable Trunk Capacity – Dynamically increases call capacity during peak busy periods so your customers never receive a busy signal.
- Call Forward Always – On the trunk group pilot number for all calls in case of an outage (flood, fire, power outage, etc.).
- Call Forward Not Reachable – On the trunk group pilot number that operates on a per-call contingency basis to forward the call to any PSTN number (i.e., call center or alternate office location) during temporary call completion impairments.
- Route Exhaustion – Automatic reroute of trunk group calls to any PSTN phone number (i.e., a call center) if calls can’t be completed to the PBX.
- Support for geo-redundant PBX deployments and automatic reroute of SIP Trunks to the backup customer data center.

All calls are routed over Cox’s national fiber network with guaranteed Quality of Service (QoS); calls never traverse the Internet.

```
Figure 1 – Cox Fiber Network
```
3 Network Topology

The high level Cox SIP Trunk network architecture is depicted below. The key network elements are:

- IP PBX – Customer PBX for terminating SIP Trunks.
- Cox Enterprise Session Border Controller (E-SBC) – The E-SBC is a smart service demarcation device and SIP Application Layer Gateway (ALG) installed and managed by Cox.
- Core Session Border Controllers and Broadsoft SIP Call Server for maximum survivability and reliability.
- PSTN Gateway for connections to the Public Switched Telephone Network (PSTN).

This SIP Trunk network architecture is replicated across the Cox operating regions for scalability and operational autonomy.

Cox will deploy one or more Enterprise Session Border Controllers (E-SBCs) to meet call capacity, customer data center geo-redundancy and trunk group requirements. The E-SBC is owned and managed by Cox and is the service demarcation point. The E-SBC performs SIP ALG, SIP normalization, NAT, security, traffic shaping/prioritization, performance reporting and remote diagnostic functions.
4 System Components

The lab network for the SIP Trunk reference configuration is illustrated in Figure 3 below and is representative of a CUCM network.

Figure 3 – SIP Trunk Lab Reference Network

4.1 Hardware Components

- Cisco IOS gateway running CUBE 1.2 (IOS image version 12.4)
- Cisco Unified Communications Manager cluster with two Cisco MCS 7800 Series servers
- Cisco 2811 router
- Cisco IP Phones. The topology diagram in Figure 3 depicts the 7960 and 7940, but any Cisco IP phone model may be used.
- Cisco VG202 Analog Voice Gateway
- EdgeMarc 6400 E-SBC
4.2 Software Requirements

- Cisco Unified Communications Manager 7.1.3(b)
- Cisco Unified Border Element (CUBE) version 1.2 IOS version 12.4
- Cisco GW IOS Release 12.4
- Cisco VG202 IOS Release version 12.4
- EdgeMarc 6400LF2 Release 9.12.0

5 Features

5.1 Supported Features

- Basic Calls with G.711ulaw CODEC
- Calling Party Number Presentation and Restriction
- Call Transfer
- Call Forwarding
- RFC2833 transcoding
- Calling party number presentation and privacy (P-Asserted-Identity)
- SIP UPDATE for call transfer support

5.2 Unsupported Features

- Codec negotiation of G.729, G.726 and others
- T.38 Fax relay

6 Caveats and Limitations

- A SIP header manipulation rule is required in the Cisco CUBE in for SIP Calls to proceed properly. A SIP profile was used to inject "user=phone" into the SIP INVITE and SIP RE-INVITE message headers that included: SIP Request-URI, Contact, To, and From header. Please refer to Section 7.3 for the SIP header rule definition.

- Cisco Analog Gateway(s) must be configured to support modem pass through for analog modem tones to work properly.
7 Configuration

The geo-redundant CUCM SIP Trunking lab network is illustrated in the figure below. The CUCM PBX at each site is assumed to be identically configured to carry the full call load in the event of a network failure at either location.

![Geo-Redundant CUCM Network Diagram]

Figure 4 – Geo-Redundant CUCM Network

7.1 CUCM SIP Trunk Call Flows

![Route Pattern Diagram]

Route Pattern
- For Voice Calls
  - Local (7 & 10 digits)
  - Long Distance
  - International
  - 911
  - Equal Access Code
  - NXX

![Route List Diagram]

Route List
- Site 1
- Site 2

![Route Group Diagram]

Route Group
- Site 1
- Site 2

![SIP Trunks Diagram]

SIP Trunks

COX SIP Network

Figure 5 – Originating Call Flow Overview

Page 9 of 71
The conceptual outgoing call flow is:

CUCM → CUBE → Cox E-SBC → Cox’s SIP Network → PSTN

The same SIP Trunks are utilized for all voice types calls between CUCM and CUBE as shown above. All outgoing calls are routed from the CUCM to CUBE through the E-SBC to Cox’s SIP Network and directed to the PSTN.

![Call Flow Diagram]

The Incoming call flow is:

PSTN → Cox’s SIP Network → Cox E-SBC → CUBE → CUCM

In the lab example, a test account DID ranges were created for Cisco Unified Communications Manager interoperability certification:

Site 1: 678.239.1xxx  
Site 2: 678.239.2xxx

All incoming calls are routed to CUCM Site 1 if the Called party number begins with 678.239.1xxx. Calls 678.239.2xxx are routed to CUCM Site 2.

7.2 SIP Trunk IP Address Assignments

To help organize your work, the IP addresses listed in Table 1 should be recorded for your deployment and referenced in the configuration steps described in this document.

<table>
<thead>
<tr>
<th>Table 1 – IP Addresses</th>
</tr>
</thead>
<tbody>
<tr>
<td>Component</td>
</tr>
<tr>
<td><strong>Cisco CUBE</strong></td>
</tr>
<tr>
<td>Site 1 CUBE Gateway IP Address</td>
</tr>
<tr>
<td>Site 1 CUBE Gateway IP Address</td>
</tr>
<tr>
<td><strong>Cox E-SBC EdgeMarc 6400’s</strong></td>
</tr>
<tr>
<td>LAN IP Address Site 1</td>
</tr>
<tr>
<td>LAN Subnet Mask Site 1</td>
</tr>
<tr>
<td>LAN IP Address Site 2</td>
</tr>
<tr>
<td>LAN Subnet Mask Site 2</td>
</tr>
</tbody>
</table>
7.3 Cisco Unified Border Element (CUBE) Configuration

Cisco Unified Border Element (CUBE) routers are utilized to hand off SIP calls to the Cox E-SBC. The CUBE feature set allowed for modification of key SIP headers using manipulation rules to format both SIP INVITE and SIP RE-INVITE SIP messages to proceed properly.

The following sip profile was used to inject “user=phone” into the INVITE and REINVITE headers.

```
voice class sip-profiles 1
request INVITE sip-header SIP-Req-URI modify " SIP/2.0" ;user=phone SIP/2.0"
request REINVITE sip-header SIP-Req-URI modify " SIP/2.0" ;user=phone SIP/2.0"
request INVITE sip-header Contact modify " >" ;user=phone>"
request REINVITE sip-header Contact modify " >" ;user=phone>"
request INVITE sip-header To modify " >" ;user=phone>"
request REINVITE sip-header To modify " >" ;user=phone>"
request INVITE sip-header From modify " >" ;user=phone>"
request REINVITE sip-header From modify " >" ;user=phone>"
```

The SIP-SIP calling, interface binding, in-call signaling and sip profile activation was enabled using the following commands.

```
voice service voip
allow-connections sip to sip
fax protocol pass-through g711ulaw
sip
bind control source-interface Loopback0
bind media source-interface Loopback0
min-se 2000
header-passing
asserted-id pai
privacy pstn
midcall-signaling passthru
sip-profiles 1
```

Detail description:
1. **fax protocol pass-through g711ulaw** - Fax pass-through takes place when incoming T.30 fax data is not demodulated or compressed for its transit through the packet network. The two endpoints (fax machines) communicate directly to each other over a transparent IP connection. The gateway does not distinguish fax calls from voice calls.
2. **asserted-id pai** - To enable the translation to PAID headers in the outgoing header at a global level.
3. **privacy pstn** – To support of User privacy policy on the UCM, this flag is set on CUBE to preserve the P-Asserted-Identity and Privacy header on the outgoing SIP INVITE.

Dial-Peers are used to provide both inbound and outbound call legs. Each call utilizes two peers. The following shows the Site 1 dial peers. In the event that the primary path is not available, preference 1 dial peer is used. Preference 0 is the default and therefore does not show in Cisco IOS.

```
dial-peer voice 1 voip
destination-pattern 6782392...
session protocol sipv2
session target ipv4:192.168.200.20
dtmf-relay rtp-nte
codec g711ulaw
ip qos dscp cs5 media
ip qos dscp cs4 signaling
no vad
```
! dial-peer voice 2 voip
  preference 1
  destination-pattern 6782392...
  session protocol sipv2
  session target ipv4:192.168.100.20
  dtmf-relay rtp-nte
  codec g711ulaw
  ip qos dscp cs5 media
  ip qos dscp cs4 signaling
  no vad

! dial-peer voice 10 voip
  translation-profile outgoing calling-mask
  destination-pattern 1[2-9]...[2-9]......
  session protocol sipv2
  session target ipv4:172.16.2.1
  dtmf-relay rtp-nte
  codec g711ulaw
  ip qos dscp cs5 media
  ip qos dscp cs4 signaling
  no vad

! dial-peer voice 11 voip
  translation-profile outgoing calling-mask
  preference 1
  destination-pattern [2-9]...[2-9]......
  session protocol sipv2
  session target ipv4:172.16.2.1
  dtmf-relay rtp-nte
  codec g711ulaw
  ip qos dscp cs5 media
  ip qos dscp cs4 signaling
  no vad

! dial-peer voice 12 voip
  translation-profile outgoing calling-mask
  destination-pattern 011T
  session protocol sipv2
  session target ipv4:172.16.2.1
  dtmf-relay rtp-nte
  codec g711ulaw
  ip qos dscp cs5 media
  ip qos dscp cs4 signaling
  no vad

! dial-peer voice 13 voip
  translation-profile outgoing calling-mask
  destination-pattern 911
  session protocol sipv2
  session target ipv4:172.16.2.1
  dtmf-relay rtp-nte
  codec g711ulaw
  ip qos dscp cs5 media
  ip qos dscp cs4 signaling
  no vad

! dial-peer voice 3 voip
  destination-pattern 6782391...
  session protocol sipv2
  session target ipv4:192.168.200.20
  dtmf-relay rtp-nte
  codec g711ulaw
  ip qos dscp cs5 media
  ip qos dscp cs4 signaling
  no vad
!  
dial-peer voice 4 voip  
  preference 1  
  destination-pattern 6782391...  
  session protocol sipv2  
  session target ipv4:192.168.100.20  
  dtmf-relay rtp-nte  
  codec g711ulaw  
  ip qos dscp cs5 media  
  ip qos dscp cs4 signaling  
  no vad

Redundant call paths for inbound calls require the following statement. The IOS default is too large and allows for a timeout before searching for the second peer (preference 1).

    sip-ua  
    retry invite 3  
    timers trying 100

Redundant call paths for outbound calls are handled by the Communication Manager. Communications Manager will route outbound calls based on the calling phones Calling Search Space (CSS). Each device assigned CSS provides for a primary and secondary path for outbound calls. The times that identify the primary path as unavailable were modified on the Communications Manager. The following values were modified in the Communications Manager Service Parameters.
Table 2 – SIP Trunk Failover Timers

<table>
<thead>
<tr>
<th>Parameter</th>
<th>Default</th>
<th>Set</th>
</tr>
</thead>
<tbody>
<tr>
<td>Retry Count for SIP Invite</td>
<td>6</td>
<td>1</td>
</tr>
<tr>
<td>SIP Trying Timer</td>
<td>500</td>
<td>100</td>
</tr>
</tbody>
</table>

See the following Cisco documentation for more information:

Failover Timer on SIP Trunks with CallManager Configuration Example:

7.4 Configuring the Cisco Unified Communications Manager (UCM) Overview

7.4.1 Voice Services
Site Call Processing, Voicemail Services as well as InterLATA and IntraLATA SCCP and SIP voice calls are provided as defined here.

7.4.2 Call Processing
Cisco Unified Communications Manager 7.1.3(b) was deployed using the distributed model. Communications Manager Server, which provides primary call processing and telephony services, can be deployed at one or two sites that maintain their own databases that are synchronized with each other. In the event that the local Communications Manager server becomes unavailable, IP Phones will attempt to register with the remote sites Communications Manager. Both Communications Manager servers share the same database and are fully aware of devices and phones registered to each other.

7.4.3 Voicemail Services
Site 1 utilizes a Cisco Unity Connection server for voicemail services. Unity Connection is integrated with Communications Manager and provides voicemail and auto attendant services for Site 1. Site 2 has access to the Unity Connection server, but instead uses a local Cisco Unity Express Network Module (CUE) for voicemail and auto attendant services. The CUE module is located within the Cisco 2811 router.

Voicemail services are independent of each other and are not “networked” together. This means that they are unable to forward messages to each others mail store and are unaware of users configured outside their system.

7.4.4 Quality of Service
Quality of Service (QOS) is provided at both, layer 2 and layer 3 boundaries using the Cisco Auto QOS feature. Both, RTP and signaling packets are tagged and passed on to the Cox’s IP Core Network for proper differential treatment.
7.4.5 Detail Communications Manager Configurations

Device pools are used to define sets of common characteristics for devices. The above screen shows a list of device pools.

Figure 8 – Device Pools

Device pools are used to define sets of common characteristics for devices. The above screen shows a list of device pools.

Figure 9 - Cisco Unified CM’s

Two Communications Managers – *Happy* and *Bashful*. Happy serves as both Publisher and Subscriber for Site 1, while Bashful acts as a Subscriber only for Site 2.
Figure 10 - Cisco Unified CM Groups

There are two CM Groups in this configuration and there are: Site 1 and Site 2 as shown above.

Figure 11 – Regions

There are four Regions defined: Default, CUBE, Site 1, and Site 2.
**Figure 12 - Physical Locations**

Two Locations – Site 1 and Site 2
### Figure 13 - Enterprise Parameters Configuration

Enterprise Parameters Configuration – relates to IP Phone Service

<table>
<thead>
<tr>
<th>Parameter Name</th>
<th>Parameter Value</th>
<th>Suggested Value</th>
</tr>
</thead>
<tbody>
<tr>
<td>Synchronization Between Auto Device, Profile, and Phone Configuration *</td>
<td>True</td>
<td>True</td>
</tr>
<tr>
<td>Max Number of Device Level Trace *</td>
<td>12</td>
<td>12</td>
</tr>
<tr>
<td>Trace Compression *</td>
<td>Disabled</td>
<td>Disabled</td>
</tr>
<tr>
<td>DSCP for Phone-based Services *</td>
<td>default DSCP (000000)</td>
<td>default DSCP (000000)</td>
</tr>
<tr>
<td>DSCP for Phone Configuration *</td>
<td>CS3(precedence 3) DSCP (011000)</td>
<td>CS3(precedence 3) DSCP (011000)</td>
</tr>
<tr>
<td>DSCP for Cisco CallManager to Device Interface *</td>
<td>CS3(precedence 3) DSCP (011000)</td>
<td>CS3(precedence 3) DSCP (011000)</td>
</tr>
<tr>
<td>Connection Monitor Duration *</td>
<td>120</td>
<td>120</td>
</tr>
<tr>
<td>Auto Registration Phone Protocol *</td>
<td>SCCP</td>
<td>SCCP</td>
</tr>
<tr>
<td>BLF For Call Lists *</td>
<td>Enabled</td>
<td>Disabled</td>
</tr>
<tr>
<td>Advertise G.722 Codec *</td>
<td>Enabled</td>
<td>Enabled</td>
</tr>
<tr>
<td>Phone Personalization *</td>
<td>Enabled</td>
<td>Disabled</td>
</tr>
<tr>
<td>Services Provisioning *</td>
<td>Internal</td>
<td>Internal</td>
</tr>
</tbody>
</table>
Figure 14 - Enterprise Parameter Configuration – Continued

Enterprise Parameters Configuration – relates to IP Phone Services
### Figure 15 - Enterprise Parameters Configuration – Continued

Enterprise Parameters Configuration – relates to IP Phone Services

<table>
<thead>
<tr>
<th>Parameter</th>
<th>Value</th>
</tr>
</thead>
<tbody>
<tr>
<td>Show Change Password Option</td>
<td>True</td>
</tr>
<tr>
<td>Show Change PIN Option</td>
<td>True</td>
</tr>
<tr>
<td>Show Download Plugin Option</td>
<td>True</td>
</tr>
<tr>
<td>Show Online Guide Option</td>
<td>True</td>
</tr>
<tr>
<td>Show Directory</td>
<td>True</td>
</tr>
<tr>
<td>Show Mobility Features Option</td>
<td>True</td>
</tr>
<tr>
<td>Show Manager Name in Directory</td>
<td>True</td>
</tr>
<tr>
<td>Show User Id in Directory</td>
<td>True</td>
</tr>
<tr>
<td>Show Extension in Directory</td>
<td>True</td>
</tr>
<tr>
<td>Show LDAP Extension in Directory</td>
<td>True</td>
</tr>
</tbody>
</table>

<table>
<thead>
<tr>
<th>Parameter</th>
<th>Value</th>
</tr>
</thead>
<tbody>
<tr>
<td>CDR Parameters</td>
<td></td>
</tr>
<tr>
<td>CDR File Time Interval</td>
<td>1</td>
</tr>
<tr>
<td>Cluster ID</td>
<td>StandAloneCluster</td>
</tr>
</tbody>
</table>

<table>
<thead>
<tr>
<th>Parameter</th>
<th>Value</th>
</tr>
</thead>
<tbody>
<tr>
<td>Localization Parameters</td>
<td></td>
</tr>
<tr>
<td>Default Network Locale</td>
<td>United States</td>
</tr>
<tr>
<td>Default User Locale</td>
<td>English United States</td>
</tr>
</tbody>
</table>
Figure 16 - Enterprise Parameters Configuration – Continued

Enterprise Parameters Configuration – relates to IP Phone Services
**Enterprise Parameters Configuration**

<table>
<thead>
<tr>
<th>Parameter</th>
<th>Value</th>
</tr>
</thead>
<tbody>
<tr>
<td>URL Messages</td>
<td></td>
</tr>
<tr>
<td>IP Phone Proxy Address</td>
<td></td>
</tr>
<tr>
<td>URL Services</td>
<td><a href="http://happy:8080/ccm/op/getservicesmenu.jsp">http://happy:8080/ccm/op/getservicesmenu.jsp</a></td>
</tr>
</tbody>
</table>

**User Search Parameters**

<table>
<thead>
<tr>
<th>Parameter</th>
<th>Value</th>
</tr>
</thead>
<tbody>
<tr>
<td>Enable All User Search *</td>
<td>True</td>
</tr>
<tr>
<td>User Search Limit *</td>
<td>64</td>
</tr>
</tbody>
</table>

**CCM Web Services Parameters**

<table>
<thead>
<tr>
<th>Parameter</th>
<th>Value</th>
</tr>
</thead>
<tbody>
<tr>
<td>Allowed Performance Queries Per Minute *</td>
<td>50</td>
</tr>
<tr>
<td>Allowed Device Queries Per Minute *</td>
<td>15</td>
</tr>
<tr>
<td>Performance Queue Limit *</td>
<td>100</td>
</tr>
<tr>
<td>Allowed CDRonDemand get _file Queries Per Minute *</td>
<td>10</td>
</tr>
<tr>
<td>Allowed CDRonDemand get _file _list Queries Per Minute *</td>
<td>20</td>
</tr>
</tbody>
</table>

**Trace Parameters**

<table>
<thead>
<tr>
<th>Parameter</th>
<th>Value</th>
</tr>
</thead>
<tbody>
<tr>
<td>File Close Thread Flag *</td>
<td>True</td>
</tr>
<tr>
<td>FileCloseThreadQueueWatermark *</td>
<td>100</td>
</tr>
</tbody>
</table>

Figure 17 - Enterprise Parameters Configuration – Continued

Enterprise Parameters Configuration – relates to IP Phone Services
---

**Figure 18 - Enterprise Parameters Configuration – Continued**

Enterprise Parameters Configuration – relates to IP Phone Services

<table>
<thead>
<tr>
<th>Parameter</th>
<th>Value</th>
</tr>
</thead>
<tbody>
<tr>
<td>Service Manager TCP server port number</td>
<td>8888</td>
</tr>
<tr>
<td>Service Manager TCP client port number</td>
<td>8889</td>
</tr>
<tr>
<td>CRS Application Parameters</td>
<td></td>
</tr>
<tr>
<td>Auto Attendant Installed</td>
<td>false</td>
</tr>
<tr>
<td>IPC Express Installed</td>
<td>false</td>
</tr>
<tr>
<td>Denial-of-Service Protection</td>
<td>True</td>
</tr>
<tr>
<td>TLS Handshake Timer</td>
<td>60</td>
</tr>
</tbody>
</table>

---
Figure 19 - Enterprise Phone Configuration

Enterprise Phone Configuration
Figure 20 - SIP Trunk Security Profiles

We Defined a Cox Business Security Profile for SIP Trunking

Figure 21 - Two Route Groups

Defined are two route groups
Figure 22 - Route Lists

Two site list are defined. One from Site 1 to Site 2 and the other from Site 2 to Site 1.

Figure 23 - Route Patterns

Defined Route patterns examples include: Local, Long Distance, International, etc.
Figure 24 - Route Patterns

Defined Route Patterns include: Local, Long Distance, International, etc. – Continued

Figure 25 - Line Groups

System - Call Routing - Media Resources - Voice Mail - Device - Application - User Manager

Find and List Line Groups

Find Line Groups where Line Group Name begins with

<table>
<thead>
<tr>
<th>Line Group Name</th>
</tr>
</thead>
<tbody>
<tr>
<td>CiscoUM1</td>
</tr>
<tr>
<td>HuntTest</td>
</tr>
<tr>
<td>Site 1 Broadcast</td>
</tr>
<tr>
<td>Site 1 Hunt</td>
</tr>
<tr>
<td>Site 2 Broadcast</td>
</tr>
<tr>
<td>Site 2 Hunt</td>
</tr>
</tbody>
</table>

Figure 25 - Line Groups
### Figure 26 - Hunt Lists

<table>
<thead>
<tr>
<th>Hunt List</th>
<th>Description</th>
<th>Enabled</th>
<th>Status</th>
</tr>
</thead>
<tbody>
<tr>
<td>Hunt Test List</td>
<td>Hunt Test List</td>
<td>true</td>
<td>Registered with 192.168.100.20</td>
</tr>
<tr>
<td>Site 1 Broadcast</td>
<td>Site 1 Broadcast Hunt</td>
<td>true</td>
<td>Registered with 192.168.100.20</td>
</tr>
<tr>
<td>Site 1 Hunt</td>
<td>Site 1 Hunt Top Down</td>
<td>true</td>
<td>Registered with 192.168.100.20</td>
</tr>
<tr>
<td>Site 2 Broadcast</td>
<td>Site 2 Broadcast Hunt</td>
<td>true</td>
<td>Registered with 192.168.100.20</td>
</tr>
<tr>
<td>Site 2 Hunt</td>
<td>Site 2 Top Down Hunt</td>
<td>true</td>
<td>Registered with 192.168.100.20</td>
</tr>
<tr>
<td>Unity</td>
<td>Unity Ports</td>
<td>true</td>
<td>Registered with 192.168.100.20</td>
</tr>
</tbody>
</table>

### Figure 27 - Hunt Pilots Numbers

<table>
<thead>
<tr>
<th>Pattern</th>
<th>Description</th>
<th>Partition</th>
<th>Route Filter</th>
<th>Hunt List</th>
</tr>
</thead>
<tbody>
<tr>
<td>1005</td>
<td>Site 1 Top Down Hunt Pilot</td>
<td>Internal</td>
<td>Site 1 Hunt</td>
<td></td>
</tr>
<tr>
<td>1006</td>
<td>Site 1 Broadcast Hunt Pilot</td>
<td>Internal</td>
<td>Site 1 Broadcast</td>
<td></td>
</tr>
<tr>
<td>2005</td>
<td>Site 2 Top Down Hunt Pilot</td>
<td>Internal</td>
<td>Site 2 Broadcast</td>
<td></td>
</tr>
<tr>
<td>2006</td>
<td>Site 2 Broadcast Hunt Pilot</td>
<td>Internal</td>
<td>Site 2 Broadcast</td>
<td></td>
</tr>
<tr>
<td>3000</td>
<td>Unity Connection</td>
<td>Internal</td>
<td>Unity</td>
<td></td>
</tr>
</tbody>
</table>
Figure 28 - Time Period

Defined Time Period for services, for testing we selected All the time.
### Find and List Partitions

#### Status
- 14 records found

<table>
<thead>
<tr>
<th>Partition Name</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>911Site1</td>
<td>911 Calls from Site 1</td>
</tr>
<tr>
<td>911Site2</td>
<td>911 Calls from Site 2</td>
</tr>
<tr>
<td>Feature</td>
<td>Calling Features</td>
</tr>
<tr>
<td>Internal</td>
<td>IP Phones and internal devices</td>
</tr>
<tr>
<td>International Site 1</td>
<td>International Calls from Site 1</td>
</tr>
<tr>
<td>International Site 2</td>
<td>International Calls from Site 2</td>
</tr>
<tr>
<td>Local Site 1</td>
<td>Local Calls from Site 1</td>
</tr>
<tr>
<td>Local Site 2</td>
<td>Local Calls from Site 2</td>
</tr>
<tr>
<td>Long Distance Site 1</td>
<td>Long distance Calls</td>
</tr>
<tr>
<td>Long Distance Site 2</td>
<td>Long distance Calls from Site 2</td>
</tr>
<tr>
<td>N11 Site 1</td>
<td>N11</td>
</tr>
<tr>
<td>N11 Site 2</td>
<td>N11 Site 2</td>
</tr>
<tr>
<td>UnityPorts</td>
<td>Unity Device Ports</td>
</tr>
<tr>
<td>Unreachable</td>
<td>Partition that is not in any CSS</td>
</tr>
</tbody>
</table>

Figure 29 - Partitions
### Calling Search Spaces

<table>
<thead>
<tr>
<th>CSS Name</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>COS1-Site1</td>
<td>Internal Only for Site 1</td>
</tr>
<tr>
<td>COS1-Site2</td>
<td>Internal Only for Site 1</td>
</tr>
<tr>
<td>COS2-Site1</td>
<td>Internal 911 and Local for Site 1</td>
</tr>
<tr>
<td>COS2-Site2</td>
<td>Internal 911 and Local for Site 2</td>
</tr>
<tr>
<td>COS3-Site1</td>
<td>Internal 911 Local and LD for Site 1</td>
</tr>
<tr>
<td>COS3-Site2</td>
<td>Internal 911 Local and LD for Site 2</td>
</tr>
<tr>
<td>COS4-Site1</td>
<td>Internal 911 Local and LD for Site 1</td>
</tr>
<tr>
<td>COS4-Site2</td>
<td>Internal 911 Local and LD for Site 2</td>
</tr>
<tr>
<td>Gateway Site 1</td>
<td>Inbound Calls from the Voice Gateway for Site 1</td>
</tr>
<tr>
<td>Gateway Site 2</td>
<td>Inbound Calls from the Voice Gateway for Site 2</td>
</tr>
<tr>
<td>Intercept-Site1</td>
<td>911 only access for Site 1</td>
</tr>
<tr>
<td>Intercept-Site2</td>
<td>911 only access for Site 2</td>
</tr>
<tr>
<td>UnityPorts</td>
<td>CSS for use with Unity CTI Ports</td>
</tr>
</tbody>
</table>

Figure 30 - Calling Search Spaces
### Figure 31 - Client Matter Codes

<table>
<thead>
<tr>
<th>Client Matter Code</th>
<th>Account Code</th>
<th>CMCA</th>
</tr>
</thead>
<tbody>
<tr>
<td>1234</td>
<td></td>
<td></td>
</tr>
<tr>
<td>5555</td>
<td></td>
<td></td>
</tr>
</tbody>
</table>

### Figure 32 - Forced Authorization Code

<table>
<thead>
<tr>
<th>Authorization Code Name</th>
<th>Authorization Code</th>
</tr>
</thead>
<tbody>
<tr>
<td>Authorization Code Name</td>
<td>Authorization Code</td>
</tr>
<tr>
<td>5678</td>
<td>0</td>
</tr>
</tbody>
</table>

Add New | Select All | Clear All | Delete Selected
Figure 33 - Translation Patterns

Figure 34 - Call Park Numbers
Figure 35 - Call Pickup Group

Figure 36 - Directory Numbers
### Figure 37 - Route Plan Report

<table>
<thead>
<tr>
<th>Pattern/Directory Number</th>
<th>Partition</th>
<th>Type</th>
<th>Route Detail</th>
</tr>
</thead>
<tbody>
<tr>
<td>*567.0411</td>
<td>International Site 1</td>
<td>Route Pattern</td>
<td>Site_1&lt;br&gt;Grumpy (All ports)</td>
</tr>
<tr>
<td></td>
<td></td>
<td></td>
<td>Site_2&lt;br&gt;Dopey (All ports)</td>
</tr>
<tr>
<td>*567.[2-9]XX[2-9]XXXXX</td>
<td>Long Distance Site 1</td>
<td>Route Pattern</td>
<td>Site_1_Private&lt;br&gt;Grumpy_Private (All ports)</td>
</tr>
<tr>
<td>*567.404[2-9]XXXXX</td>
<td>Local Site 1</td>
<td>Route Pattern</td>
<td>Site_1_Private&lt;br&gt;Grumpy_Private (All ports)</td>
</tr>
</tbody>
</table>

### Figure 38 - Route Plan Report - Continued

<table>
<thead>
<tr>
<th>Pattern/Directory Number</th>
<th>Partition</th>
<th>Type</th>
<th>Route Detail</th>
</tr>
</thead>
<tbody>
<tr>
<td>*567.6782391</td>
<td>Local Site 1</td>
<td>Route Pattern</td>
<td>Site_1&lt;br&gt;Grumpy (All ports)</td>
</tr>
<tr>
<td>*567.678[2-9]XXXXX</td>
<td>Local Site 1</td>
<td>Route Pattern</td>
<td>Site_1_Private&lt;br&gt;Grumpy_Private (All ports)</td>
</tr>
<tr>
<td>1001</td>
<td>Internal</td>
<td>Translation Pattern</td>
<td>SEP0023BEC89963&lt;br&gt;HuntTest&lt;br&gt;Site_1_Hunt&lt;br&gt;Site_1_Broadcast</td>
</tr>
<tr>
<td>1001</td>
<td>Internal</td>
<td>Directory Number</td>
<td>SEP0023BEC78105&lt;br&gt;HuntTest&lt;br&gt;Site_1_Hunt&lt;br&gt;Site_1_Broadcast</td>
</tr>
<tr>
<td>1001</td>
<td>Unreachable</td>
<td>Directory Number</td>
<td>SEP0023BEC89963&lt;br&gt;HuntTest&lt;br&gt;Site_1_Hunt&lt;br&gt;Site_1_Broadcast</td>
</tr>
<tr>
<td>1002</td>
<td>Internal</td>
<td>Directory Number</td>
<td>SEP0023BEC78105&lt;br&gt;HuntTest&lt;br&gt;Site_1_Hunt&lt;br&gt;Site_1_Broadcast</td>
</tr>
<tr>
<td>1002</td>
<td>Unreachable</td>
<td>Directory Number</td>
<td>SEP0023BEC89963&lt;br&gt;HuntTest&lt;br&gt;Site_1_Hunt&lt;br&gt;Site_1_Broadcast</td>
</tr>
<tr>
<td>1002</td>
<td>Internal</td>
<td>Directory Number</td>
<td>SEP0023BEC78105&lt;br&gt;HuntTest&lt;br&gt;Site_1_Hunt&lt;br&gt;Site_1_Broadcast</td>
</tr>
</tbody>
</table>
### Figure 39 - Route Plan Report - Continued

<table>
<thead>
<tr>
<th>System</th>
<th>Call Routing</th>
<th>Media Resources</th>
<th>Voice Mail</th>
<th>Device</th>
<th>Application</th>
<th>User Management</th>
</tr>
</thead>
<tbody>
<tr>
<td>1003</td>
<td>Internal</td>
<td>Directory Number</td>
<td></td>
<td></td>
<td></td>
<td>Site_1_Hunt</td>
</tr>
<tr>
<td>1004</td>
<td>Local Site 1</td>
<td>Directory Number</td>
<td></td>
<td></td>
<td></td>
<td>AH1DADAE6613000</td>
</tr>
<tr>
<td>1004</td>
<td>Internal</td>
<td>Directory Number</td>
<td></td>
<td></td>
<td></td>
<td>RossParkerBAC</td>
</tr>
<tr>
<td>1005</td>
<td>Internal</td>
<td>Hunt Pilot</td>
<td></td>
<td></td>
<td></td>
<td>Site_1_Hunt</td>
</tr>
<tr>
<td>1006</td>
<td>Internal</td>
<td>Hunt Pilot</td>
<td></td>
<td></td>
<td></td>
<td>Site_1_Broadcast</td>
</tr>
<tr>
<td>1008</td>
<td>Internal</td>
<td>Call Park</td>
<td></td>
<td></td>
<td></td>
<td>Site_1_Hunt</td>
</tr>
<tr>
<td>1009</td>
<td>Internal</td>
<td>Directory Number</td>
<td></td>
<td></td>
<td></td>
<td>1001, Internal</td>
</tr>
<tr>
<td>1010</td>
<td>Internal</td>
<td>Directory Number</td>
<td></td>
<td></td>
<td></td>
<td>1002, Internal</td>
</tr>
<tr>
<td>1011</td>
<td>Internal</td>
<td>Directory Number</td>
<td></td>
<td></td>
<td></td>
<td>1003, Internal</td>
</tr>
<tr>
<td>1015</td>
<td>Local Site 1</td>
<td>Call Pickup Group</td>
<td></td>
<td></td>
<td></td>
<td>Site_1_Broadcast</td>
</tr>
</tbody>
</table>

### Figure 40 - Route Plan Report - Continued

<table>
<thead>
<tr>
<th>System</th>
<th>Call Routing</th>
<th>Media Resources</th>
<th>Voice Mail</th>
<th>Device</th>
<th>Application</th>
<th>User Management</th>
</tr>
</thead>
<tbody>
<tr>
<td>1016</td>
<td>Local Site 2</td>
<td>Call Pickup Group</td>
<td></td>
<td></td>
<td></td>
<td>Site_2_Hunt</td>
</tr>
<tr>
<td>1050</td>
<td></td>
<td>Directory Number</td>
<td></td>
<td></td>
<td></td>
<td>Site_1_Broadcast</td>
</tr>
<tr>
<td>1051</td>
<td></td>
<td>Directory Number</td>
<td></td>
<td></td>
<td></td>
<td>Site_2_Broadcast</td>
</tr>
<tr>
<td>1052</td>
<td>Internal</td>
<td>Translation Pattern</td>
<td></td>
<td></td>
<td></td>
<td>Site_1_Broadcast</td>
</tr>
<tr>
<td>2000</td>
<td>UnityTransfer</td>
<td>Directory Number</td>
<td></td>
<td></td>
<td>Site_2_Broadcast</td>
<td></td>
</tr>
<tr>
<td>2001</td>
<td>Internal</td>
<td>Directory Number</td>
<td></td>
<td></td>
<td></td>
<td>Site_2_Hunt</td>
</tr>
<tr>
<td>2001</td>
<td>Local Site 1</td>
<td>Directory Number</td>
<td></td>
<td></td>
<td></td>
<td>Site_1_Broadcast</td>
</tr>
<tr>
<td>2001</td>
<td>Local Site 2</td>
<td>Directory Number</td>
<td></td>
<td></td>
<td></td>
<td>Site_2_Broadcast</td>
</tr>
<tr>
<td>2001</td>
<td>UnityTransfer</td>
<td>Directory Number</td>
<td></td>
<td></td>
<td></td>
<td>Site_1_Broadcast</td>
</tr>
<tr>
<td>2001</td>
<td>Site_2_Hunt</td>
<td>Directory Number</td>
<td></td>
<td></td>
<td></td>
<td>Site_2_Broadcast</td>
</tr>
<tr>
<td>2001</td>
<td>Unreacheable</td>
<td>Directory Number</td>
<td></td>
<td></td>
<td></td>
<td>Site_1_Broadcast</td>
</tr>
</tbody>
</table>
### Route Plan Report

<table>
<thead>
<tr>
<th>Year</th>
<th>Pattern/Directory Number</th>
<th>Partition</th>
<th>Type</th>
<th>Route Data</th>
</tr>
</thead>
<tbody>
<tr>
<td>2003</td>
<td>Internal</td>
<td>Call Park</td>
<td>Directory Number</td>
<td>Site_2_Hunt</td>
</tr>
<tr>
<td>2003</td>
<td>Internal</td>
<td>Directory Number</td>
<td>Site_2_Broadcast</td>
<td></td>
</tr>
<tr>
<td>2004</td>
<td>Feature</td>
<td>Directory Number</td>
<td>Site_2_Hunt</td>
<td></td>
</tr>
<tr>
<td>2004</td>
<td>N11 Site 1</td>
<td>Directory Number</td>
<td>Site_2_Broadcast</td>
<td></td>
</tr>
<tr>
<td>2005</td>
<td>UnityPorts</td>
<td>Directory Number</td>
<td>Site_2_Hunt</td>
<td></td>
</tr>
<tr>
<td>2006</td>
<td>UnityPorts</td>
<td>Directory Number</td>
<td>Site_2_Broadcast</td>
<td></td>
</tr>
<tr>
<td>2006</td>
<td>Internal</td>
<td>Directory Number</td>
<td>Site_2_Hunt</td>
<td></td>
</tr>
<tr>
<td>2007</td>
<td>UnityPorts</td>
<td>Directory Number</td>
<td>Site_2_Broadcast</td>
<td></td>
</tr>
</tbody>
</table>

**Figure 41 - Route Plan Report - Continued**

### Route Plan Report

<table>
<thead>
<tr>
<th>Find</th>
<th>All Patterns</th>
<th>Route Plan Report where Pattern/Directory Number begins with</th>
<th>Select item or enter search</th>
</tr>
</thead>
<tbody>
<tr>
<td></td>
<td></td>
<td>(51 - 100 of 132)</td>
<td></td>
</tr>
</tbody>
</table>

<table>
<thead>
<tr>
<th>Pattern/Directory Number</th>
<th>Partition</th>
<th>Call Park</th>
<th>Directory Number</th>
<th>Route Data</th>
</tr>
</thead>
<tbody>
<tr>
<td>Internal</td>
<td>Call Park</td>
<td>Directory Number</td>
<td>Site_2_Hunt</td>
<td></td>
</tr>
<tr>
<td>Internal</td>
<td>Directory Number</td>
<td>Site_2_Broadcast</td>
<td></td>
<td></td>
</tr>
<tr>
<td>Feature</td>
<td>Directory Number</td>
<td>Site_2_Hunt</td>
<td></td>
<td></td>
</tr>
<tr>
<td>N11 Site 1</td>
<td>Directory Number</td>
<td>Site_2_Broadcast</td>
<td></td>
<td></td>
</tr>
<tr>
<td>UnityPorts</td>
<td>Directory Number</td>
<td>Site_2_Hunt</td>
<td></td>
<td></td>
</tr>
<tr>
<td>UnityPorts</td>
<td>Directory Number</td>
<td>Site_2_Broadcast</td>
<td></td>
<td></td>
</tr>
<tr>
<td>Internal</td>
<td>Directory Number</td>
<td>Site_2_Hunt</td>
<td></td>
<td></td>
</tr>
<tr>
<td>Internal</td>
<td>Directory Number</td>
<td>Site_2_Broadcast</td>
<td></td>
<td></td>
</tr>
<tr>
<td>Internal</td>
<td>Directory Number</td>
<td>Site_2_Hunt</td>
<td></td>
<td></td>
</tr>
<tr>
<td>Internal</td>
<td>Directory Number</td>
<td>Site_2_Broadcast</td>
<td></td>
<td></td>
</tr>
<tr>
<td>Internal</td>
<td>Directory Number</td>
<td>Site_2_Hunt</td>
<td></td>
<td></td>
</tr>
<tr>
<td>Internal</td>
<td>Directory Number</td>
<td>Site_2_Broadcast</td>
<td></td>
<td></td>
</tr>
<tr>
<td>Internal</td>
<td>Directory Number</td>
<td>Site_2_Hunt</td>
<td></td>
<td></td>
</tr>
<tr>
<td>Unity</td>
<td>Directory Number</td>
<td>Site_2_Broadcast</td>
<td></td>
<td></td>
</tr>
<tr>
<td>CiscoUM1</td>
<td>Directory Number</td>
<td>Site_2_Hunt</td>
<td></td>
<td></td>
</tr>
<tr>
<td>6000, UnityPorts</td>
<td>Directory Number</td>
<td>Site_2_Broadcast</td>
<td></td>
<td></td>
</tr>
<tr>
<td>6000, UnityPorts</td>
<td>Directory Number</td>
<td>Site_2_Hunt</td>
<td></td>
<td></td>
</tr>
<tr>
<td>6002, UnityPorts</td>
<td>Directory Number</td>
<td>Site_2_Broadcast</td>
<td></td>
<td></td>
</tr>
</tbody>
</table>

**Figure 42 - Route Plan Report - Continued**
Figure 43 - Route Plan Report - Continued

<table>
<thead>
<tr>
<th>Route Plan Report</th>
<th>Related Links</th>
</tr>
</thead>
<tbody>
<tr>
<td>3001 Internal</td>
<td>Message Waiting</td>
</tr>
<tr>
<td>3002 Internal</td>
<td>Message Waiting</td>
</tr>
</tbody>
</table>

Figure 44 - Route Plan Report - Continued

<table>
<thead>
<tr>
<th>Route Plan Report</th>
<th>Related Links</th>
</tr>
</thead>
<tbody>
<tr>
<td>3001 Internal</td>
<td>Message Waiting</td>
</tr>
<tr>
<td>3002 Internal</td>
<td>Message Waiting</td>
</tr>
<tr>
<td>4000 Internal</td>
<td>Directory Number</td>
</tr>
<tr>
<td>4000 Internal</td>
<td>Directory Number</td>
</tr>
<tr>
<td>5000 Internal</td>
<td>Directory Number</td>
</tr>
<tr>
<td>5000 UnityPorts</td>
<td>Directory Number</td>
</tr>
<tr>
<td>5001 UnityPorts</td>
<td>Directory Number</td>
</tr>
<tr>
<td>5001 Internal</td>
<td>Directory Number</td>
</tr>
<tr>
<td>5002 Internal</td>
<td>Directory Number</td>
</tr>
<tr>
<td>5002 UnityPorts</td>
<td>Directory Number</td>
</tr>
<tr>
<td>5003 Internal</td>
<td>Directory Number</td>
</tr>
<tr>
<td>5003 UnityPorts</td>
<td>Directory Number</td>
</tr>
<tr>
<td>5004 UnityPorts</td>
<td>Directory Number</td>
</tr>
<tr>
<td>5004 Internal</td>
<td>Directory Number</td>
</tr>
<tr>
<td>5005 Internal</td>
<td>Directory Number</td>
</tr>
<tr>
<td>5005 UnityPorts</td>
<td>Directory Number</td>
</tr>
<tr>
<td>5006 Internal</td>
<td>Directory Number</td>
</tr>
<tr>
<td>5006 UnityPorts</td>
<td>Directory Number</td>
</tr>
<tr>
<td>5007 UnityPorts</td>
<td>Directory Number</td>
</tr>
<tr>
<td>5007 Internal</td>
<td>Directory Number</td>
</tr>
</tbody>
</table>
Figure 45 - Route Plan Report – Continued

<table>
<thead>
<tr>
<th>Route Plan Report</th>
<th>Related Links</th>
</tr>
</thead>
<tbody>
<tr>
<td>6001</td>
<td>Unity Ports</td>
</tr>
<tr>
<td>6002</td>
<td>Unity Ports</td>
</tr>
<tr>
<td>6003</td>
<td>Unity Ports</td>
</tr>
<tr>
<td>6004</td>
<td>Unity Ports</td>
</tr>
<tr>
<td>6005</td>
<td>Unity Ports</td>
</tr>
<tr>
<td>6006</td>
<td>Unity Ports</td>
</tr>
<tr>
<td>6007</td>
<td>Unity Ports</td>
</tr>
<tr>
<td>6008</td>
<td>Unity Ports</td>
</tr>
<tr>
<td>6009</td>
<td>Unity Ports</td>
</tr>
<tr>
<td>6010</td>
<td>Unity Ports</td>
</tr>
<tr>
<td>6011</td>
<td>Unity Ports</td>
</tr>
<tr>
<td>6012</td>
<td>Unity Ports</td>
</tr>
<tr>
<td>6013</td>
<td>Unity Ports</td>
</tr>
<tr>
<td>6014</td>
<td>Unity Ports</td>
</tr>
<tr>
<td>6015</td>
<td>Unity Ports</td>
</tr>
<tr>
<td>6016</td>
<td>Unity Ports</td>
</tr>
<tr>
<td>6017</td>
<td>Unity Ports</td>
</tr>
<tr>
<td>6018</td>
<td>Unity Ports</td>
</tr>
<tr>
<td>6019</td>
<td>Unity Ports</td>
</tr>
<tr>
<td>6020</td>
<td>Unity Ports</td>
</tr>
</tbody>
</table>

Figure 46 - Route Plan Report - Continued
<table>
<thead>
<tr>
<th>Route Plan Report</th>
<th>Related Links: View</th>
</tr>
</thead>
<tbody>
<tr>
<td>9.011</td>
<td>International Site 1</td>
</tr>
<tr>
<td>9.011#</td>
<td>International Site 2</td>
</tr>
<tr>
<td>9.011#</td>
<td>International Site 1</td>
</tr>
<tr>
<td>9.101XXXXX(2-9)XX(2-9)XXXXX</td>
<td>Local Site 1</td>
</tr>
</tbody>
</table>

Figure 47 - Route Plan Report – Continued

<table>
<thead>
<tr>
<th>Route Plan Report</th>
<th>Related Links: View</th>
</tr>
</thead>
<tbody>
<tr>
<td>9.1[2-9]XXXXX</td>
<td>Long Distance Site 1</td>
</tr>
<tr>
<td>9.1[2-9]XXXXX</td>
<td>Long Distance Site 2</td>
</tr>
<tr>
<td>9.404[2-9]XXXXX</td>
<td>Local Site 2</td>
</tr>
<tr>
<td>9.404[2-9]XXXXX</td>
<td>Local Site 1</td>
</tr>
</tbody>
</table>

Figure 48 - Route Plan Report - Continued
<table>
<thead>
<tr>
<th>Route Plan Report</th>
<th>Route Pattern</th>
<th>Related Links:</th>
<th>View</th>
</tr>
</thead>
<tbody>
<tr>
<td>9.675[2-0]xxxxxx</td>
<td>Local Site 1</td>
<td>Site_1 Site_2</td>
<td>Site_1 Site_2</td>
</tr>
<tr>
<td></td>
<td></td>
<td>Dopey (All ports) Dopey (All ports)</td>
<td>Dopey (All ports) Dopey (All ports)</td>
</tr>
<tr>
<td></td>
<td>Route Pattern</td>
<td>Grumpy (All ports) Grumpy (All ports)</td>
<td>Grumpy (All ports) Grumpy (All ports)</td>
</tr>
<tr>
<td>9.675[2-0]xxxxxx</td>
<td>Local Site 2</td>
<td>Site_2 Site_2</td>
<td>Site_2 Site_2</td>
</tr>
<tr>
<td></td>
<td>Route Pattern</td>
<td>Dopey (All ports) Dopey (All ports)</td>
<td>Dopey (All ports) Dopey (All ports)</td>
</tr>
<tr>
<td>9.700[2-0][xxxxxx</td>
<td>Local Site 1</td>
<td>Site_1 Site_1</td>
<td>Site_1 Site_1</td>
</tr>
<tr>
<td></td>
<td>Route Pattern</td>
<td>Grumpy (All ports) Grumpy (All ports)</td>
<td>Grumpy (All ports) Grumpy (All ports)</td>
</tr>
<tr>
<td>9.700[2-0][xxxxxx</td>
<td>Local Site 2</td>
<td>Site_2 Site_2</td>
<td>Site_2 Site_2</td>
</tr>
<tr>
<td></td>
<td>Route Pattern</td>
<td>Dopey (All ports) Dopey (All ports)</td>
<td>Dopey (All ports) Dopey (All ports)</td>
</tr>
<tr>
<td>9.775[2-0][xxxxxx</td>
<td>Local Site 1</td>
<td>Site_1 Site_1</td>
<td>Site_1 Site_1</td>
</tr>
<tr>
<td></td>
<td>Route Pattern</td>
<td>Grumpy (All ports) Grumpy (All ports)</td>
<td>Grumpy (All ports) Grumpy (All ports)</td>
</tr>
</tbody>
</table>

Figure 49 - Route Plan Report – Continued
Figure 50 - Route Plan Report - Continued

Figure 51 - Cisco Unity Voice Mail Ports Setup
Figure 52 - Message Waiting Numbers

Figure 53 - Voice Mail Pilot
Figure 54 - Voice Mail Profile

The CTI Route has three route points listed, they are Unity Express (Voice Mail), Unity Express Auto-Attendant, and Unity Connection Auto-Attendant.
Figure 56 - Detail Telephone Status Page

Figure 57 - SIP Trunks
Figure 58 - IP Phone Services

Figure 59 - SIP Profiles
COX Business SIP Profile

Figure 60 - Common Device Configuration
Figure 61 - Attendant Console User

Figure 62 - Detail Phone Configuration
Figure 63 - Detail Phone Configuration - Continued
Figure 64 - Detail Phone Configuration - Continued
### Expansion Module Information

<table>
<thead>
<tr>
<th>Module</th>
<th>Load Name</th>
</tr>
</thead>
<tbody>
<tr>
<td>Module 1</td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td>Module 2</td>
<td>&lt; None &gt;</td>
</tr>
</tbody>
</table>

### External Data Locations Information (Leave blank to use default)

<table>
<thead>
<tr>
<th>Information</th>
<th></th>
</tr>
</thead>
<tbody>
<tr>
<td>Directory</td>
<td></td>
</tr>
<tr>
<td>Messages</td>
<td></td>
</tr>
<tr>
<td>Services</td>
<td></td>
</tr>
<tr>
<td>Authentication Server</td>
<td></td>
</tr>
<tr>
<td>Proxy Server</td>
<td></td>
</tr>
<tr>
<td>Idle</td>
<td></td>
</tr>
<tr>
<td>Idle Timer (seconds)</td>
<td></td>
</tr>
</tbody>
</table>

### Extension Information

- [ ] Enable Extension Mobility
- Log Out Profile: **Use Current Device Settings**
- Log In Time: < None >
- Log out Time: < None >

### MLPP Information

<table>
<thead>
<tr>
<th>MLPP Domain</th>
<th>&lt; None &gt;</th>
</tr>
</thead>
<tbody>
<tr>
<td>Feature</td>
<td>Value</td>
</tr>
<tr>
<td>-------------------------------------------</td>
<td>-----------</td>
</tr>
<tr>
<td>Video Capabilities*</td>
<td>Disabled</td>
</tr>
<tr>
<td>Auto Line Select*</td>
<td>Disabled</td>
</tr>
<tr>
<td>Web Access</td>
<td>Enabled</td>
</tr>
<tr>
<td>Date/Display Not Active</td>
<td></td>
</tr>
<tr>
<td>Display On Time</td>
<td></td>
</tr>
<tr>
<td>Display On Duration</td>
<td></td>
</tr>
<tr>
<td>Display Idle Timeout</td>
<td></td>
</tr>
<tr>
<td>Span to PC Port</td>
<td></td>
</tr>
<tr>
<td>Logging Display</td>
<td></td>
</tr>
<tr>
<td>Load Server</td>
<td></td>
</tr>
<tr>
<td>Recording Time</td>
<td></td>
</tr>
<tr>
<td>Recording Time Local Volume</td>
<td></td>
</tr>
<tr>
<td>Recording Time Remote Volume</td>
<td></td>
</tr>
<tr>
<td>Recording Time Duration</td>
<td></td>
</tr>
<tr>
<td>Display On When Incoming Call</td>
<td></td>
</tr>
<tr>
<td>RTP*</td>
<td></td>
</tr>
<tr>
<td>&quot;more&quot; Soft Key Timer</td>
<td></td>
</tr>
<tr>
<td>Auto Call Select</td>
<td></td>
</tr>
<tr>
<td>Log Server</td>
<td></td>
</tr>
<tr>
<td>Asterphone 6722 Coder*</td>
<td></td>
</tr>
<tr>
<td>Wideband Headset UI Control</td>
<td></td>
</tr>
<tr>
<td>Wideband Headset UI Control</td>
<td></td>
</tr>
</tbody>
</table>

**Figure 66 - Detail Phone Configuration - Continued**

<table>
<thead>
<tr>
<th>Feature</th>
<th>Value</th>
</tr>
</thead>
<tbody>
<tr>
<td>Wideband Handset UI Control</td>
<td>Enabled</td>
</tr>
<tr>
<td>Wideband Headset</td>
<td>Enabled</td>
</tr>
<tr>
<td>Wideband Handset</td>
<td>Enabled</td>
</tr>
<tr>
<td>Peer Firmware Sharing</td>
<td></td>
</tr>
<tr>
<td>Cisco Discovery Protocol (CDP): Switch Port*</td>
<td></td>
</tr>
<tr>
<td>Cisco Discovery Protocol (CDP): PC Port*</td>
<td></td>
</tr>
<tr>
<td>Link-Layer Discovery Protocol - Media Endpoint Discover (LLDP-MED): Switch Port*</td>
<td></td>
</tr>
<tr>
<td>Link-Layer Discovery Protocol (LLDP): PC Port*</td>
<td></td>
</tr>
<tr>
<td>LLDP Asset ID</td>
<td></td>
</tr>
<tr>
<td>LLDP Power Priority</td>
<td></td>
</tr>
<tr>
<td>IPv4 Load Server</td>
<td></td>
</tr>
<tr>
<td>IPv4 Log Server</td>
<td></td>
</tr>
<tr>
<td>802.1X Authentication</td>
<td></td>
</tr>
<tr>
<td>Detect Unified CM Connection Failure*</td>
<td></td>
</tr>
<tr>
<td>Minimum Ring Volume</td>
<td></td>
</tr>
<tr>
<td>Headset Sidetone Level</td>
<td></td>
</tr>
<tr>
<td>Bibio Dialing</td>
<td></td>
</tr>
</tbody>
</table>

**Figure 67 - Detail Phone Configuration - Continued**
Figure 68 - Directory Number Services Configuration - Continued
### Directory Number Configuration

<table>
<thead>
<tr>
<th>Feature</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>No Answer Ring Duration (seconds)</td>
<td>10</td>
</tr>
<tr>
<td>Call Pickup Group</td>
<td>PickupGroup in Local Site 1</td>
</tr>
<tr>
<td>Forward No Coverage Internal</td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td>Forward No Coverage External</td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td>Forward on CFI Failure</td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td>Forward Unregistered Internal</td>
<td>COS4-Site1</td>
</tr>
<tr>
<td>Forward Unregistered External</td>
<td>COS4-Site1</td>
</tr>
<tr>
<td>Park Monitoring: Forward No Retrieve</td>
<td>A blank value means to call the partner’s line.</td>
</tr>
<tr>
<td>Destination Internal</td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td>Park Monitoring: Forward No Retrieve</td>
<td>A blank value means to call the partner’s line.</td>
</tr>
<tr>
<td>Destination Internal</td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td>Park Monitoring: Reversion Timer</td>
<td>A blank value will use value set in Park Monitoring Reversion Timer service parameter</td>
</tr>
</tbody>
</table>

**Figure 69 - Directory Number Services Configuration - Continued**
**Figure 70 - Directory Number Services Configuration - Continued**

<table>
<thead>
<tr>
<th>Line 1 on Device SEP902318C78105</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>Display (Internal Caller ID)</strong></td>
</tr>
<tr>
<td><strong>ASCII Display (Internal Caller ID)</strong></td>
</tr>
<tr>
<td><strong>Line Text Label</strong></td>
</tr>
<tr>
<td><strong>ASCII Line Text Label</strong></td>
</tr>
<tr>
<td><strong>External Phone Number Mask</strong></td>
</tr>
<tr>
<td><strong>Visual Message Waiting Indicator Policy</strong></td>
</tr>
<tr>
<td><strong>Audit Message Waiting Indicator Policy</strong></td>
</tr>
<tr>
<td><strong>Ring Setting (Phone Idle)</strong></td>
</tr>
<tr>
<td><strong>Ring Setting (Phone Active)</strong></td>
</tr>
<tr>
<td><strong>Call Pickup Group Audio Alert Setting (Phone Idle)</strong></td>
</tr>
<tr>
<td><strong>Call Pickup Group Audio Alert Setting (Phone Active)</strong></td>
</tr>
<tr>
<td><strong>Recording Option</strong></td>
</tr>
<tr>
<td><strong>Recording Profile</strong></td>
</tr>
<tr>
<td><strong>Monitoring Calling Search Space</strong></td>
</tr>
<tr>
<td><strong>Log Missed Calls</strong></td>
</tr>
</tbody>
</table>

**Multiple Call/Call Waiting Settings on Device SEP902318C78105**

Note: The range to select the Max Number of calls is: 1-100

Maximum Number of Calls: **4**
Figure 71 - Directory Number Services Configuration - Continued
7.5 SIP Message Trace Examples

7.5.1 Incoming SIP INVITE – PSTN (E-SBC) to Cisco CUBE

INVITE sip:6782391001@172.16.1.5:5060 SIP/2.0
Via: SIP/2.0/UDP 172.16.1.1:5060;branch=z9hG4bKm3d0m89bite5c408406o4rdqk7
Record-Route: <sip:6782391001@172.16.1.1;
From: "PSTNLINETEST1" <sip:4046691360@172.16.1.1;user=phone>;tag=SDjcurc01-1438275345-1273243620664-
To: "6782391001 6782391001" <sip:6782391001@172.16.1.1:5060>
Call-ID: SDjcurc01-5ad26d17fa988ebc728d8b51189bf9ea-vrvvfv3
CSeq: 966650525 INVITE
Contact: <sip:172.16.1.1:5060;transport=udp>
Supported: 100rel
Max-forwards: 69
Allow: ACK, BYE, CANCEL, INFO, INVITE, OPTIONS, PRACK, REFER, NOTIFY, UPDATE
Content-Type: application/sdpAccept: multipart/mixed, application/media_control+xml, application/sdp
Content-Length: 247

v=0
o=BroadWorks 89489 1 IN IP4 172.16.1.1
s=-c=IN IP4 172.16.1.1
t=0 0
m=audio 17468 RTP/AVP 0 101
a=rtpmap:0 PCMU/8000/1
a=sqn: 0
a=cdsc: 1 image udptl t38
a=mptime:20
a=ptime:20
a=rtpmap:101 telephone-event/8000/1
a=fmtp:101 0-15

7.5.2 Outgoing SIP INVITE – Cisco CUBE to PSTN (E-SBC)

INVITE sip:4046691360@172.16.1.1:5060;user=phone SIP/2.0
Via: SIP/2.0/UDP 172.16.1.5:5060;branch=z9hG4bK2531301
From: "James Dean" <sip:6782391001@172.16.1.5;user=phone>;tag=56E64F88-35E
To: <sip:4046691360@172.16.1.1;user=phone>
Date: Fri, 07 May 2010 14:50:06 GMT
Call-ID: A8DD7589-591E11DF-BFAF4A4A-35F18573@172.16.1.5
Supported: 100rel,timer,resource-priority,replaces,sdp-anat
Min-SE: 1800
Cisco-Guid: 2832845825-1495142879-3215567434-905020787
User-Agent: Cisco-SIPGateway/IOS-12.x
Allow: INVITE, OPTIONS, BYE, CANCEL, ACK, PRACK, UPDATE, REFER, SUBSCRIBE, NOTIFY, INFO, REGISTER
CSeq: 101 INVITE
Timestamp: 1273243806
Contact: <sip:6782391001@172.16.1.5:5060;user=phone>
Expires: 180
Allow-Events: telephone-event
Max-Forwards: 69
Session-Expires: 2200
P-Asserted-Identity: "James Dean" <sip:6782391001@172.16.1.5>
Content-Type: application/sdp
Content-Disposition: session;handling=required
Content-Length: 241
v=0
o=CiscoSystemsSIP-GW-UserAgent 3969 9006 IN IP4 172.16.1.5
s=SIP Callc=IN IP4 172.16.1.5
t=0 0
m=audio 18982 RTP/AVP 0 101
c=IN IP4 172.16.1.5
a=rtpmap:0 PCMU/8000
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-15
a=ptime:20

7.5.3 Transfer SIP UPDATE – Cisco CUBE to PSTN (E-SBC)

UPDATE sip:172.16.1.1:5060;transport=udp SIP/2.0
Via: SIP/2.0/UDP 172.16.1.1:5060;branch=z9hG4bK25B1BF9
From: "6782391001 6782391001" <sip:6782391001@172.16.1.1:5060>;tag=56E796F4-22DB
To: "PSTNLINESTEST1" <sip:4046691360@172.16.1.1:5060;user=phone>;tag=SDjcurc01-1438275345-1273243620664-
Date: Fri, 07 May 2010 14:51:33 GMT
Call-ID: SDjcurc01-5ad26d17fa988ebe728d8b51189bf9ea-vrvfvf3
User-Agent: Cisco-SIPGateway/IOS-12.x
Max-Forwards: 70
Route: <sip:6782391001@172.16.1.1;lr>
Supported: 100rel,timer,resource-priority,replaces,sdp-anat
Timestamp: 1273243911
Allow: INVITE, OPTIONS, BYE, CANCEL, ACK, PRACK, UPDATE, REFER, SUBSCRIBE, NOTIFY, INFO, REGISTER
CSeq: 101 UPDATE
Contact: <sip:6782391001@172.16.1.1:5060>
Remote-Party-ID: <sip:4046691323@172.16.1.1>;party=called;screen=no;privacy=off
Content-Length: 0

UPDATE sip:172.16.1.1:5060;transport=udp SIP/2.0
Via: SIP/2.0/UDP 172.16.1.1:5060;branch=z9hG4bK25CC1D
From: "James Dean" <sip:6782391001@172.16.1.5>;tag=56E7CDA0-48E
To: <sip:4046691323@172.16.1.1>;tag=SD3po2899-1307487053-1273243634795
Date: Fri, 07 May 2010 14:51:43 GMT
Call-ID: E32BF056-591E11DF-BFC4AA4A-35F18573@172.16.1.5
User-Agent: Cisco-SIPGateway/IOS-12.x
Max-Forwards: 70
Route: <sip:EWGW_0@172.16.1.1;lr>
Supported: 100rel,timer,resource-priority,replaces,sdp-anat
Timestamp: 1273243911
Allow: INVITE, OPTIONS, BYE, CANCEL, ACK, PRACK, UPDATE, REFER, SUBSCRIBE, NOTIFY, INFO, REGISTER
CSeq: 103 UPDATE
7.5.4 Call Forward SIP INVITE – Cisco CUBE to PSTN (E-SBC)

INVITE sip:4046691323@172.16.1.1:5060;user=phone SIP/2.0
Via: SIP/2.0/UDP 172.16.1.5:5060;branch=z9hG4bK26125E2
From: "PSTNLINETEST1" <sip:4046691360@172.16.1.5;user=phone>;tag=56E8B114-27C
To: <sip:4046691323@172.16.1.1;user=phone>
Date: Fri, 07 May 2010 14:52:42 GMT
Call-ID: 5E16542-591F11DF-BFD6AA4A-35F18573@172.16.1.5
Supported: 100rel,timer,resource-priority,replaces,sdp-anat
Min-SE: 1800
Cisco-Guid: 98417594-1495208415-3218123338-905020787
User-Agent: Cisco-SIPGateway/IOS-12.x
Allow: INVITE, OPTIONS, BYE, CANCEL, ACK, PRACK, UPDATE, REFER, SUBSCRIBE, NOTIFY, INFO, REGISTER
CSeq: 101 INVITE
Timestamp: 1273243962
Contact: <sip:4046691360@172.16.1.5:5060;user=phone>
Expires: 180
Allow-Events: telephone-event
Max-Forwards: 66
Session-Expires: 2200
P-Asserted-Identity: "PSTNLINETEST1" <sip:4046691360@172.16.1.5>
Diversion: <sip:6782391001@172.16.1.5>;privacy=off;reason=unconditional;screen=yes
Content-Type: application/sdp
Content-Disposition: session;handling=required
Content-Length: 240
v=0
o=CiscoSystems
SIP-GW-UserAgent 5232 531 IN IP4 172.16.1.5
s=SIP Callc=IN IP4 172.16.1.5
t=0 0
m=audio 16636 RTP/AVP 0 101
c=IN IP4 172.16.1.5
a=rtpmap:0 PCMU/8000
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-15
a=ptime:20

7.6 Full Detail IOS Configurations

! Site 1 -- Grumpy -- CUBE IOS Configurations
!
Current configuration : 6683 bytes
!
! Last configuration change at 11:30:27 EST Mon Jun 7 2010 by admin
! NVRAM config last updated at 11:30:28 EST Mon Jun 7 2010 by admin
!
version 12.4
service timestamps debug datatime msec
service timestamps log datatime msec
no service password-encryption
service sequence-numbers

hostname Grumpy

! boot-start-marker
boot system flash c2800nm-ipvoice_ivs-mz.124-24.T2.bin
boot-end-marker

! card type command needed for slot/vwic-slot 0/0
logging message-counter syslog
logging buffered 1000000
no logging console

no aaa new-model
clock timezone EST -5

! ip source-route

! ip cef

! no ip domain lookup
no ipv6 cef
multilink bundle-name authenticated

! voice service voip
allow-connections sip to sip
fax protocol cisco
modem passthrough nse codec g711ulaw
sip
bind control source-interface Loopback0
bind media source-interface Loopback0
min-se 2000
header-passing
asserted-id pai
privacy pstn
midcall-signaling passthru
sip-profiles 1

! voice class sip-profiles 1
request INVITE sip-header SIP-Req-URI modify " SIP/2.0" ;user=phone SIP/2.0"
request REINVITE sip-header SIP-Req-URI modify " SIP/2.0" ;user=phone SIP/2.0"
request INVITE sip-header Contact modify "">" ;user=phone">
request REINVITE sip-header Contact modify "">" ;user=phone">
request INVITE sip-header To modify "">" ;user=phone>"
request REINVITE sip-header To modify "">" ;user=phone"
request INVITE sip-header From modify "">" ;user=phone"
request REINVITE sip-header From modify "">" ;user=phone"
!
voice translation-rule 39
rule 2 /^1\(...\)$/ /6782391/1/
!
voice translation-profile calling-mask
translate redirect-target 39
translate redirect-called 39
!
voice-card 0
dspfarm
dsp services dspfarm
!
username admin privilege 15 secret $1$xscF$snZsx7K1jcqGNK6nW5PZn/
archive
log config
hidekeys
!
class-map match-any AutoQoS-VoIP-RTP-Trust
match ip dscp ef
!
class-map match-any AutoQoS-VoIP-Control-Trust
match ip dscp cs3
match ip dscp af31
!
!
policy-map AutoQoS-Policy-Trust
class AutoQoS-VoIP-RTP-Trust
  priority percent 70
class AutoQoS-VoIP-Control-Trust
  bandwidth percent 5
class class-default
  fair-queue
!
interface Loopback0
description SIP Interface
ip address 172.16.1.5 255.255.255.252
!
interface FastEthernet0/0
description Internal Interface
no ip address
duplex full
speed 100
auto qos voip trust
service-policy output AutoQoS-Policy-Trust
!
interface FastEthernet0/0.1
description Management VLAN 1
encapsulation dot1Q 1 native
ip address 192.168.1.1 255.255.255.0
!
interface FastEthernet0/0.2
description SIP Trunk Access
encapsulation dot1Q 10
ip address 172.16.10.2 255.255.255.252
!
interface FastEthernet0/1
description Edgemark
ip address 172.16.1.2 255.255.255.252
duplex auto
speed auto
auto qos voip trust
service-policy output AutoQoS-Policy-Trust
!
ip forward-protocol nd
ip route 0.0.0.0 0.0.0.0 172.16.10.1
ip route 192.168.0.0 255.255.0.0 172.16.10.1
!
ip http server
ip http access-class 23
ip http authentication local
ip http timeout-policy idle 60 life 86400 requests 10000
!
control-plane
!
rmon event 33333 log trap AutoQoS description "AutoQoS SNMP traps for Voice Drops" owner AutoQoS
rmon alarm 33333 cbQosCMDCropBitRate.18.3164929 30 absolute rising-threshold 1 33333 falling-threshold 0 owner AutoQoS
!
sccp local FastEthernet0/0.2
sccp ccm 192.168.200.20 identifier 2 version 7.0
sccp ccm 192.168.100.20 identifier 1 version 7.0
sccp
!
sccp ccm group 1
associate ccm 1 priority 1
associate profile 2 register TRAN-SITE-1
associate profile 1 register MTP-SITE-1
!
dspfarm profile 2 transcode
codec g711ulaw
codec g711alaw
codec g729ar8
codec g729abr8
maximum sessions 24
associate application SCCP
dspfarm profile 1 mtp
  codec g711ulaw
  maximum sessions software 100
  associate application SCCP
!

!  

dial-peer voice 1 voip
destination-pattern 6782391...
  session protocol sipv2
  session target ipv4:192.168.100.20
dtmf-relay rtp-nte
  codec g711ulaw
  ip qos dscp cs5 media
  ip qos dscp cs4 signaling
  no vad
!

!  

dial-peer voice 2 voip
  preference 1
destination-pattern 6782391...
  session protocol sipv2
  session target ipv4:192.168.200.20
dtmf-relay rtp-nte
  codec g711ulaw
  ip qos dscp cs5 media
  ip qos dscp cs4 signaling
  no vad
!

!  

dial-peer voice 10 voip
  translation-profile outgoing calling-mask
destination-pattern 1[2-9].[2-9]......
  session protocol sipv2
  session target ipv4:172.16.1.1
dtmf-relay rtp-nte
  codec g711ulaw
  ip qos dscp cs5 media
  ip qos dscp cs4 signaling
  no vad
!

!  

dial-peer voice 11 voip
  translation-profile outgoing calling-mask
  preference 1
destination-pattern [2-9].[2-9]......
  session protocol sipv2
  session target ipv4:172.16.1.1
dtmf-relay rtp-nte
  codec g711ulaw
  ip qos dscp cs5 media
  ip qos dscp cs4 signaling
  no vad
!

!  

dial-peer voice 12 voip
translation-profile outgoing calling-mask
destination-pattern 011T
session protocol sipv2
session target ipv4:172.16.1.1
dtmf-relay rtp-nte
codec g711ulaw
ip qos dscp cs5 media
ip qos dscp cs4 signaling
no vad
!
dial-peer voice 13 voip
translation-profile outgoing calling-mask
destination-pattern [2-9]11
session protocol sipv2
session target ipv4:172.16.1.1
dtmf-relay rtp-nte
codec g711ulaw
ip qos dscp cs5 media
ip qos dscp cs4 signaling
no vad
!
dial-peer voice 3 voip
destination-pattern 6782392...
session protocol sipv2
session target ipv4:192.168.100.20
dtmf-relay rtp-nte
codec g711ulaw
ip qos dscp cs5 media
ip qos dscp cs4 signaling
no vad
!
dial-peer voice 4 voip
preference 1
destination-pattern 6782392...
session protocol sipv2
session target ipv4:192.168.200.20
dtmf-relay rtp-nte
codec g711ulaw
ip qos dscp cs5 media
ip qos dscp cs4 signaling
no vad
!
dial-peer voice 14 voip
translation-profile outgoing calling-mask
preference 1
destination-pattern [2-9].....
session protocol sipv2
session target ipv4:172.16.1.1
dtmf-relay rtp-nte
codec g711ulaw
ip qos dscp cs5 media
ip qos dscp cs4 signaling
no vad
!
dial-peer voice 15 voip
  translation-profile outgoing calling-mask
  preference 1
  destination-pattern 101....1[2-9][2-9]......
  session protocol sipv2
  session target ipv4:172.16.1.1
  dtmf-relay rtp-nte
  codec g711ulaw
  ip qos dscp cs5 media
  ip qos dscp cs4 signaling
  no vad
!
!
sip-ua
  retry invite 3
  timers trying 100
!
!
gatekeeper
  shutdown
!
!
line con 0
  login local
line aux 0
line vty 0 4
  access-class 23 in
  privilege level 15
  login local
line vty 5 15
  access-class 23 in
  privilege level 15
  login local
!
scheduler allocate 20000 1000
ntp peer 192.168.100.20
ntp server 192.168.100.20
end

! Cisco Unified Express Voice Mail System
!
version 12.4
service timestamps debug datetime msec
service timestamps log datetime msec
no service password-encryption
!
hostname Sneezy
!
boot-start-marker
boot-end-marker
! card type command needed for slot/wwic-slot 0/0
logging buffered 51200 warnings
!
no aaa new-model
clock timezone EST -5
dot11 syslog
!
!
ip cef
!
!
no ip domain lookup
multilink bundle-name authenticated
!
!
voice-card 0
no dspfarm
!
!
crypto pki trustpoint TP-self-signed-1016059772
enrollment selfsigned
subject-name cn=IOS-Self-Signed-Certificate-1016059772
revocation-check none
rsakeypair TP-self-signed-1016059772
!
!
crypto pki certificate chain TP-self-signed-1016059772
certificate self-signed 01
!
!
username admin privilege 15 secret 5 $1$Qfk7$cwTNvyLG9jxGfXrAatbyc1
log config
  hidekeys
!
!
  class-map match-any AutoQoS-VoIP-RTP-Trust
    match ip dscp ef
  class-map match-any AutoQoS-VoIP-Control-Trust
    match ip dscp cs3
    match ip dscp af31
!
!
  policy-map AutoQoS-Policy-Trust
  class AutoQoS-VoIP-RTP-Trust
    priority percent 70
  class AutoQoS-VoIP-Control-Trust
    bandwidth percent 5
  class class-default
    fair-queue
!
!
  interface FastEthernet0/0
    description Internal Network
    no ip address
duplex auto
    speed auto
    auto qos voip trust
    service-policy output AutoQoS-Policy-Trust
!
  interface FastEthernet0/0.1
    encapsulation dot1Q 1 native
    ip address 192.168.1.5 255.255.255.0
    ip nat inside
    ip virtual-reassembly
!
  interface FastEthernet0/0.2
    encapsulation dot1Q 220
    ip address 192.168.220.1 255.255.255.0
    ip nat inside
    ip virtual-reassembly
!
  interface FastEthernet0/1
    description connection from COX Network
    ip address 10.100.50.200 255.255.255.0
    ip nat outside
    ip virtual-reassembly
duplex auto
    speed auto
    auto qos voip trust
    service-policy output AutoQoS-Policy-Trust
!
  interface FastEthernet0/1/0


interface FastEthernet0/1/1
!
interface FastEthernet0/1/2
!
interface FastEthernet0/1/3
!
interface Integrated-Service-Engine1/0
  ip unnumbered FastEthernet0/0.2
  service-module ip address 192.168.220.30 255.255.255.0
  service-module ip default-gateway 192.168.220.1
  no keepalive
!
interface Vlan1
  no ip address
!
ip forward-protocol nd
ip route 0.0.0.0 0.0.0.0 10.100.50.1
ip route 172.16.0.0 255.255.0.0 192.168.220.1
ip route 192.168.0.0 255.255.0.0 192.168.220.1
ip route 192.168.220.30 255.255.255.255 Integrated-Service-Engine1/0
!
ip http server
ip http access-class 23
ip http authentication local
ip http secure-server
ip http timeout-policy idle 60 life 86400 requests 10000
ip nat inside source static 192.168.1.2 10.100.50.201
ip nat inside source static 192.168.100.20 10.100.50.202
ip nat inside source static 192.168.200.20 10.100.50.203
ip nat inside source static 192.168.100.30 10.100.50.204
!
control-plane
!
line con 0
  login local
line aux 0
line 66
  no activation-character
  no exec
  transport preferred none
  transport input all
  transport output pad telnet rlogin lapb-ta mop udptn v120 ssh
line vty 0 4
  access-class 23 in
login local
length 0
transport input telnet ssh
line vty 5 15
access-class 23 in
privilege level 15
login local
transport input telnet ssh
!
scheduler allocate 20000 1000
ntp clock-period 17180827
ntp peer 192.168.100.20
!
end

/** Cisco Analog Voice Gateway VG202 configuration **/

!
version 12.4
no service pad
service timestamps debug datetime msec
service timestamps log datetime msec
no service password-encryption
!
hostname VG202
!
boot-start-marker
boot-end-marker
!
logging message-counter syslog
enable secret 5 $1$T8KF$UdA4F3MFMyAMzEMBLnZu1
!
no aaa new-model
ip source-route
!
ip cef
no ipv6 cef
!
stcapp ccm-group 1
stcapp
!
stcapp feature access-code
!
stcapp feature speed-dial
!
!
voice service voip
  modem passthrough nse codec g711ulaw
!
!
voice-card 0
!
username admin privilege 15 secret 5 $1$OTG6$gmEKkbulfgXiSRXA9Tqskg.
archive
  log config
  hidekeys

! interface FastEthernet0/0
  ip address dhcp
  duplex auto
  speed auto

! interface FastEthernet0/1
  no ip address
  shutdown
  duplex auto
  speed auto

! ip forward-protocol nd

! no ip http server

! control-plane

! voice-port 0/0
  timeouts initial 60
  timeouts interdigit 60
  timeouts ringing infinity

! voice-port 0/1
  timeouts initial 60
  timeouts interdigit 60
  timeouts ringing infinity

  ccm-manager fax protocol cisco
  ccm-manager config server 192.168.100.20
  ccm-manager sccp local FastEthernet0/0
  ccm-manager sccp

  mgcp fax t38 ecm
  mgcp behavior g729-variants static-pt

  sccp local FastEthernet0/0
  sccp ccm 192.168.200.20 identifier 1 version 7.0
  sccp ccm 192.168.100.20 identifier 2 version 7.0
  sccp

  sccp ccm group 1
  associate ccm 1 priority 1

! dial-peer voice 999000 pots
service stcapp
port 0/0
! dial-peer voice 999001 pots
  service stcapp
  port 0/1
!
!
  line con 0
    login local
    no modem enable
  line aux 0
  line vty 0 4
    privilege level 15
    login local
    length 0
!
end