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SIP Trunking Configuration Guide

for

Cisco Unified Communications Manager Express (CME)

8.5 CLI

formerly known as formerly known as Cisco Unified CallManager Express

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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 configuration steps for Cox SIP trunking to a Cisco Unified CME IP-PBX. 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 (i.e., flood, fire, loss of power, 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); <u>calls never traverse</u> the Internet.

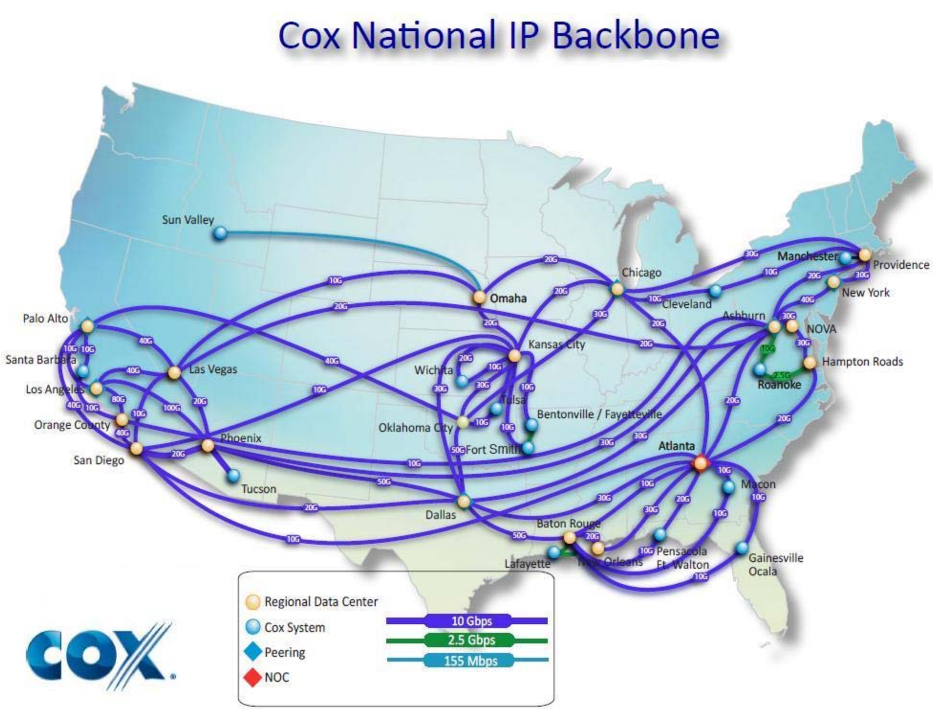

Figure 1 - Cox Fiber Network

2.1 tekVizion Labs

tekVizion Labs™ is an independent testing and Verification facility offered by tekVizion PVS, Inc. ("tekVizion"). tekVizion Labs offers several types of testing services including:

- Remote Testing – provides secure, remote access to certain products in tekVizion Labs for pre-Verification and ad hoc testing
- Verification Testing – Verification of interoperability performed on-site at tekVizion Labs between two products or in a multi-vendor configuration ("solution Verification")
- Product Assessment – independent assessment and verification of product functionality, interface usability, assessment of differentiating features as well as suggestions for added functionality, stress and performance testing, etc.

tekVizion is a systems integrator specifically dedicated to the telecommunications industry. Our core services include consulting/solution design, interoperability/Verification testing, integration, custom software development and solution support services. Our services helps service providers achieve a smooth transition to packet-voice networks, speeding delivery of integrated services. While we have expertise covering a wide range of technologies, we have extensive experience surrounding our FastForward>> practice areas which include: SIP Trunking, Packet Voice, Service Delivery, and Integrated Services.

The tekVizion team brings together experience from the leading service providers and vendors in telecom. Our unique expertise includes legacy switching services and platforms, and unparalleled product knowledge, interoperability and integration experience on a vast array of VoIP and other next-generation products. We rely on this combined experience to do what we do best: help our clients advance the rollout of services that excite customers and result in new revenues for the bottom line. tekVizion leverages this real-world, multi-vendor integration and test experience and proven processes to offer services to vendors, network operators, enhanced service providers, large enterprises and other professional services firms. tekVizion's headquarters, along with a state-of-the-art test lab and Executive Briefing Center, is located in the Telecom Corridor® in Richardson, Texas.
(*For more information on tekVizion and its practice areas, please visit tekVizion Labs's web site at www.tekVizionlabs.com.)*

3 SIP Trunking Network Components

The network for the SIP trunk reference configuration is illustrated below and is representative of a CME configuration

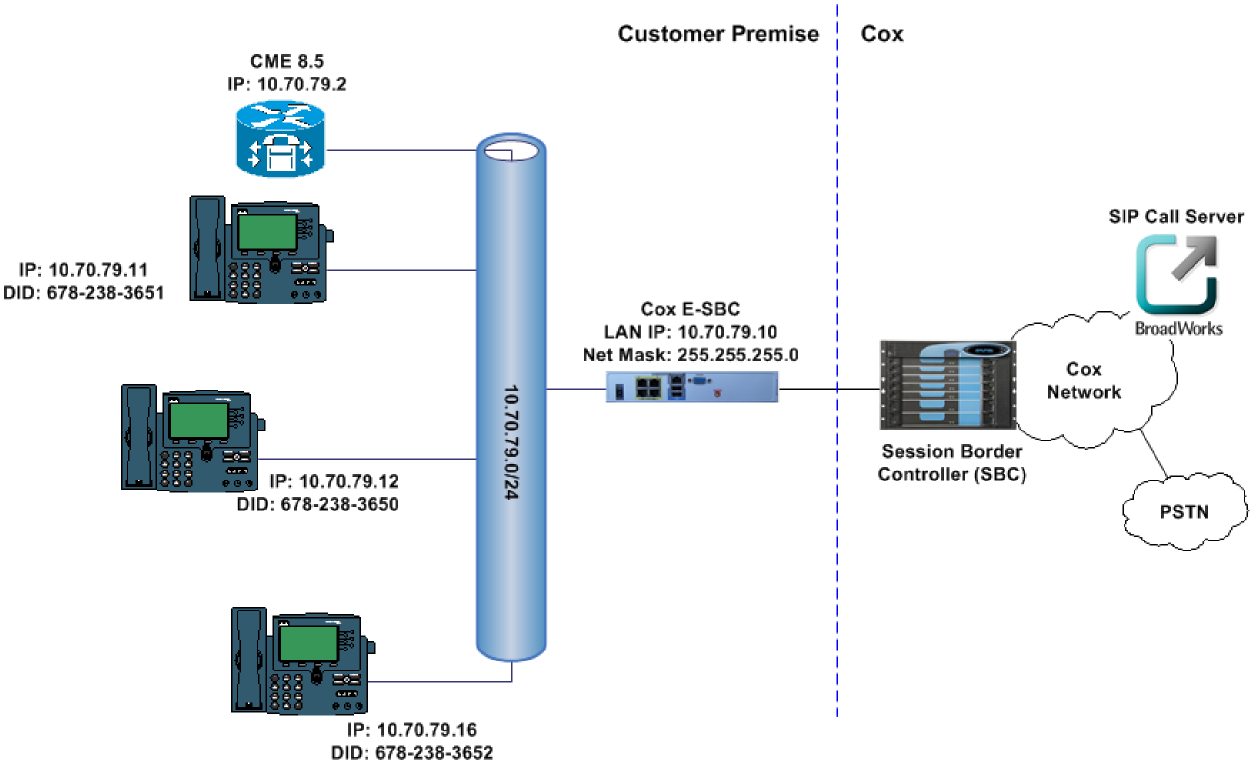

Figure 2 - SIP Trunk Lab Reference Network

Note: The CME does offer DHCP server for dynamic IP address assignment for the SIP phones; however, the Cox Enterprise Session Border Controller (E-SBC) requires a static LAN IP address that must be manually assigned by the LAN network administrator. The DHCP server is provisioned on the Ethernet switch. The DHCP's IP address pool is constrained so that the E-SBC can be assigned an IP address outside of the pool.

The lab network consists of the following components:

- CME IP PBX for voice features, SIP proxy and SIP trunk termination.
- Various SCCP phones on the local LAN.
- The Cox E-SBC is the Edgewater Networks (www.edgewaternetworks.com) EdgeMarc appliance. The EdgeMarc is the service demarcation point between customer's LAN network and Cox's WAN network and provides firewall/NAT traversal, B2BUA and SIP Application-level gateway. The EdgeMarc has diverse routes to a primary and secondary Acme SBC.
- Acme Packet Net-Net 9200 Session Border Controllers (SBC).

3.1 Hardware Components

- Cisco 3845
- Analog fax machine
- EdgeMarc 4550 E-SBC

3.2 Software Requirements

- CME Release V8.5
- Cisco 3845 Version 15.1(3)T
- EdgeMarc 4550 9.12.5 Release

4 Features

4.1 SIP Registration Method

Cox Network requires SIP REGISTER support to allow the IP-PBX to originate calls from the IP-PBX and to send calls to the PBX from the PSTN. CME supports SIP Register with authentication. Cox implementation team provides the Pilot number and the authentication key, which should be provisioned in the CME How to configure these in the CME are shown in Section 6.3.2

4.2 Features Supported

- Basic calls using G.711ulaw
- Calling Party Number Presentation
- Anonymous call
- Call Transfer
- Call Forwarding
- Call Hold and Resume
- Call Pickup
- Call Waiting
- DND
- Call Park
- Hunt groups (Simultaneous and Sequential Ring)
- Three-Way Calling
- G711 Passthru Fax
- PBX Account Codes
- PBX Auto Attendant to Off-net Numbers
- E911 Call
- RFC2833 transcoding
- PBX-Defined Caller ID (spoofing)

4.3 Features Not Supported

- T.38 Fax – roadmap item.
- Dial-Up Modem

5 Caveats and Limitations

- T.38 fax is supported by CME but at this time network issues did not allow completion. G711 fax is successful.
- Authorization codes can be configured and tested, but before entering the code, the audio is distorted and authorization code was not successfully completed and the call failed.
- Account Codes on the PBX are supported with 7960 Cisco IP phones no configuration is required. Once code is entered it will appear on the CDR if provisioned according to Cisco.
- Modem test did not pass. Test originated from PBX, and receiving side did not connect. This is most likely a lab environment artifact.

6 Configuration

6.1 Configuration Checklist

In this section we present an overview of the steps that are required to configure CME for SIP Trunking as well as all features that were tested.

Table 1 – PBX Configuration Steps

Step	Description	Reference
Step 1	System IP Address	Section 6.3.1
Step 2	SIP Registration	Section 6.3.2
Step 3	Sip Carrier Options	Section 6.3.3
Step 4	Configure Outbound Dial Peer	Section 6.3.4
Step 5	Configure ephone-dn	Section 6.3.5
Step 6	Configure ephone	Section 6.3.6
Step 7	Telephony Service	Section 6.3.7
Step 8	Translation Rules	Section 6.3.8
Step 9	Translation Profiles	Section 6.3.9
Step 10	Routing Services	Section 6.3.10
Step 11	Incoming Call	Section 6.3.11
Step 16	Assign Fax/Modem Station	Section 6.3.16
Step 17	CUBE Configuration	Section 6.4

6.2 IP Address Worksheet

The specific values listed in the table below and in subsequent sections are used in the lab configuration described in this document, and are for **illustrative purposes only**. The customer must obtain and use the values for your deployment.

Table 2 – IP Addresses

Component	Cox Lab Value	Customer Value
EdgeMarc E-SBC		
• LAN IP Address	10.70.79.10	
• LAN Subnet Mask	255.255.255.0	
CME IP PBX		
• System IP Address This is the IP address of the CME. This IP address is typically on the same subnet as the LAN IP Address of the E-SBC. If this is not the case, then Layer 3 routing must be in place.	10.70.79.2	
• Default Gateway The Default Gateway must be the LAN Network default Gateway. This will allow the administrator to log in via his\her workstation if the workstation is on a different network	10.70.95.1	
• DNS This is the DNS server for the Enterprise network. Cox Communications does not supply DNS services.	10.64.1.3	

6.3 CME Detailed Configuration Steps

Equipment used for configuration setup:

- Cisco 3845 Version 15.1(3)T
- CME software version release V8.5 CLI
- Cisco IP Phones (7970, 7975)

6.3.1 System IP Address

The IP address of the CME router is 10.70.79.2 with a subnet mask of 255.255.255.0. The following settings are used by the E-SBC for SIP Trunking Devices and the Default Dial Rules.

```
interface GigabitEthernet0/0[1]
    description $ES_LAN$[2]
    ip address 10.70.79.2 255.255.255.0[3]
    duplex full[4]
    speed 100[5]
    media-type rj45[6]
```

6.3.2 SIP Registration

These values allow for verification when communicating with the registrar server

```
sip-ua[7]
    credentials username 6782383650 password 0 6782383650 realm lab.tekvizion.com[8]
    authentication username 6782383650 password 0 6782383650[9]
    registrar 1 ipv4:10.70.79.10:5060 expires 60[10]
    sip-server ipv4:10.70.79.10:5060[11]
```

---

[1] Enters configuration mode for interface gigabit Ethernet 0/0
[2] Provides a description of the interface
[3] Assigns an IP address and subnet mask to the interface
[4] Sets duplex mode to full
[5] Sets the speed to 100 mbps
[6] Defines the media type as rj45
[7] Enters configuration mode for SIP User Agent

[8] Sets the username and password. For this example the realm is set to lab.tekvizion.com this domain name is not applicable to the registration but simply required by the configuration and may not be omitted. For example lab.tekvizion.com could be my.domain.com etc… The actual SIP Registration Password and Username will be provided by your Cox Account Representative and must be kept confidential! The Trunk Group Pilot Number (username) is used here for illustration purposes only!

[9] Sets the authentication username and password. The actual SIP Registration Password and Username will be provided by your Cox Account Representative and must be kept confidential! The Trunk Group Pilot Number (username) is used here for illustration purposes only!
[10] Points the ua to the registrar server, adds the port number and sets the registration expiration timer
[11] Points the ua to the SIP server and provides the port number

6.3.3 Sip Carrier Options

Remember that the E-SBC LAN IP address may/will be different from this example. Please see **Figure 2** and **Table 2** for the IP address scheme.

```
voice service voip [12]
    ip address trusted list [13]
    ipv4 10.70.79.0 255.255.255.0 [14]
    allow-connections h323 to h323[15]
    allow-connections h323 to sip
    allow-connections sip to h323
    allow-connections sip to sip
    supplementary-service h450.12
    modem passthrough protocol codec g711ulaw [16]
    sip[17]
    session refresh
    registrar server expires max 600 min 60[18]
```

6.3.4 Configure Outbound Dial Peer

The dial-peer voice number is determined by the administrator of the IP-PBX. For purposes of this example 200x is used for outgoing dial peers, 100x for incoming dial-peers, 10x for dial-peers being directed to CUE and 5x for POTS dial peers. Below is an example of the dial-peer that defines the Pilot Number being registered..

The common configuration lines will be documented as “common line“ and will be left out of the remaining outgoing dial-peers.

```
dial-peer voice 2000 voip[19]
    description **Registration Dial Peer**
    preference 1
    destination-pattern 6782383650[20]
    session protocol sipv2[21]
    session target sip-server[22]
    voice-class sip dtmf-relay force rtp-nte[23]
    dtmf-relay rtp-nte
    no vad
```

---

[12] Enters configuration mode for trunk settings
[13] Mode for listing the trusted addresses
[14] The IP address of the network that is trusted to receive calls from
[15] This allows for connections between various types of trunks
[16] Provides for modem traffic to be passed via clear channel and sets the codec for modem transmission
[17] Enters configuration mode for SIP commands
[18] Sets the expiration timers for the registrar server
[19] Common Line with exception of 2000.
[20] This destination pattern is the pilot number
[21] Common line that assigns SIPv2 as the session protocol
[22] Common line that sets the sip server as the session target
[23] Common line that forces RFC2833 for DTMF even if it was not requested in the initial invite. This is a hidden command

```
dial-peer voice 2001 voip
      description **10 digit national number**
      translation-profile outgoing PSTN_Outgoing
      destination-pattern 9[2-9]..[2-9]......[24]

dial-peer voice 2002 voip
      description **011 + International Number**
      translation-profile outgoing PSTN_Outgoing
      destination-pattern 9011T[25]

dial-peer voice 2003 voip
      description **CCA*North American-10-Digit*Service Numbers**
      translation-profile outgoing PSTN_Outgoing
      destination-pattern 9[2-9]11[26]

dial-peer voice 2004 voip
      description **1 + 10 digit LD number**
      translation-profile outgoing PSTN_Outgoing
      destination-pattern 91[2-9]..[2-9]......[27]

dial-peer voice 2005 voip
      description **0 or 00 calls to Local or International Operator**
      translation-profile outgoing PSTN_Outgoing
      destination-pattern 90T[28]

dial-peer voice 2006 voip
      description **Dial Carrier Access Code**
      translation-profile outgoing PSTN_Outgoing
      destination-pattern 9101....1[2-9]..[2-9]......[29]

dial-peer voice 1000 voip
      description ** Incoming call from SIP trunk (Cox Communications) **
      translation-profile incoming PSTN_Incoming
      incoming called-number .%[30]

dial-peer voice 50 pots
      destination-pattern 3674[31]
      fax rate voice[32]
```

[24] This pattern will strip off the 9, accept any number from 2-9 as the first of the 3-digit area code, and accept any number from 2-9 as the first number of a 7-digit phone number
[25] This pattern allows for dialing 9011 followed by a variable-length dial string
[26] This pattern allows for dialing 9 plus 311, 411,etc
[27] This pattern requires an additional 1 for long distance calls. Dial 91 plus a 3-digit area code that begins with 2-9, followed by the 7-digit dial string that must start with a 2-9
[28] This pattern allows for dialing 90 followed by a variable-length dial string
[29] This pattern allows for entering a Carrier Access Code with the format of 9101, a 4-digit code, followed by a 3-digit area code and the 7-digit phone number. The area code and phone number must begin with 2-9.
[30] This allows for receiving any dial string or nothing at all
[31] This is the extension number for the fax port
[32] This specifies the highest possible transmission speed allowed by the voice rate

```
    no digit-strip
    port 0/0/0

dial-peer voice 100 voip
    description **CUE Voice Mail Dial Peer **
    destination-pattern 3000[33]
    b2bua
    session target ipv4:10.70.79.50[34]

dial-peer voice 101 voip
    description **CUE Auto-Attendant Dial Peer **
    destination-pattern 3671[35]
```

6.3.5 Configure ephone-dn

```
ephone-dn  1  dual-line [36]
    number 3650[37] secondary 6782383650[38] no-reg both
    pickup-group 1[39]
    label 3650[40]
    description TestPhone One
    name TestPhone One
    call-forward busy 96782344101[41]
    call-forward noan 96782344101 timeout 20[42]
```

6.3.6 Configure ephone

```
ephone  1
    device-security-mode none[43]
    mac-address FCFB.FBCA.22FE
    type 7975
    button  1:1[44]
```

[33] This number directly accesses the CUE voice mail
[34] This is the IP address of the service module where the CUE resides
[35] This number directly accesses the CUE auto attendant
[36] Enters configuration mode and establishes ephone-dn 1 as a dual line
[37] Extension number of ephone-dn 1
[38] DID assigned to the secondary line
[39] Assigns rights to pick up incoming calls destined for any member of pickup-group 1
[40] This will show up as text on the phone display
[41] Defines behavior when phone is busy
[42] Defines behavior when a call is not answered within 20 seconds
[43] Disables device-security-mode
[44] Associates one line with one button

6.3.7 Telephony-Service

Telephony-Service configuration adjusts the characteristics of the CME. For testing purposes, a few sections of the running-config were modified.

```
telephony-service[45]
    moh-file-buffer 10000[46]
    internal-call moh-group 0[47]
    authentication credential admin admin[48]
    authentication credential cisco cisco
    max-ephones 15[49]
    max-dn 15[50]
    ip source-address 10.70.79.2 port 2000[51]
    caller-id block code *67[52]
    url services http://10.70.79.50/voiceview/common/login.do
    url authentication http://10.70.79.2/CCMCIP/authenticate.asp
    cnf-file location flash:[53]
    load 7960-7940 P00308010200
    load 7971 SCCP70.9-2-3S
    load 7975 SCCP75.9-1-1SR1S
    voicemail 3000[54]
    mwi relay
    max-conferences 12 gain -6
    call-park system application[55]
    moh flash:/RedHotChiliPeppersulaw.wav
    web admin system name cisco password cisco
    dn-webedit
    time-webedit
    transfer-system full-consult[56]
    transfer-pattern 9T[57]
    secondary-dialtone 9[58]
    after-hours block pattern 1 919003235555 7-24
    create cnf-files version-stamp Jan 01 2002 00:00:00
```

---

[45] Enters telephony-service configuration mode
[46] Sets the buffer size for moh to 10000. Default size is 64 KB. Range is 64 to 10000 KB
[47] Assigns MOH-group for all internal directory numbers
[48] Optional entry in the database used by the CME authentication server
[49] Sets maximum limit of 15 ephones
[50] Sets maximum limit of 15 ephone-dns
[51] IP address and port for IP PBX
[52] To place an anonymous call, dial *67 before dialing the outbound number
[53] Firmware for phones is located in flash
[54] Assigns 3000 as the extension for voicemail
[55] Defines call-park/pickup parameters for both SCCP and SIP lines
[56] Establishes full-consult as the call transfer setting. (Other options are full-blind and local-consult)
[57] Defines the digit string pattern for permitted non-local call transfers
[58] Creates another tone when 9 is dialed to place an outside call

6.3.8 Translation Rules

Translation Rules are created in global configuration mode with the function of matching and replacing digits. The translation rule number is a unique identifier that is used to associate a set of translation rules with a translation profile.

voice translation-rule 410
rule 1 /^9\(.*\)/ /\1/[59]
rule 15 /^...$/ /6782383650/[60]

voice translation-rule 1111
rule 1 /^678238\(....\)$/ /\1/[61]

voice translation-rule 2111
rule 1 /3653/ /8005551212/[62]
rule 2 /^\(....\)$/ /678238\1/[63]
rule 15 /3653/ /8005551212/[64]

voice translation-rule 2112
rule 1 /^9/ //[65]

6.3.9 Translation Profiles

Translation profiles are created in global configuration mode to define a voice profile for voice calls. This profile is associated with a translation rule to define the behavior of the call

voice translation-profile PSTN_Incoming
translate called 1111[66]

voice translation-profile PSTN_Outgoing
translate calling 2111
translate called 2112
translate redirect-target 410[67]
translate redirect-called 410[68]

---

[59] Strip off the 9, accept all other digits of any length
[60] Replaces any 3-digit number with the pilot number (6782383650)
[61] Replaces the 10-digit DID with the 4-digit extension
[62] Replace 3653 with 8005551212 for spoofing
[63] Replaces any 4-digit extension with the 10-digit DID associated with that extension
[64] Replaces any 3-digit number with the pilot number (6782383650)
[65] Strip off the 9, accept all other digits of any length
[66] Refers to translation rule 1111
[67] Refers to translation rule 410 which defines behavior for the transfer-to and call forwarding final destination numbers
[68] Refers to translation rule 410 which defines behavior for the redirect-called number

6.3.10 Routing Services

Routing services define the paths to reach the E-SBC

```
ip route 0.0.0.0 0.0.0.0 10.70.79.10[69]
ip route 10.70.79.50 255.255.255.255 Integrated-Service-Engine1/0[70]
ip route 10.70.90.0 255.255.255.0 10.70.79.1[71]
ip route 174.46.0.128 255.255.255.128 10.70.79.1
```

6.3.11 Incoming Call

```
dial-peer voice 1000 voip
    description ** Incoming call from SIP trunk (Cox Communications) **
    translation-profile incoming PSTN_Incoming[72]
    session protocol sipv2
    incoming called-number .%[73]
    voice-class codec 1
    voice-class sip dtmf-relay force rtp-nte
    dtmf-relay rtp-nte
    no vad
```

6.3.12 Assign Fax/Modem Station

```
voice-port 0/0/0[74]
    no echo-cancel enable [75]
    no comfort-noise [76]
    station-id number 3674[77]
    caller-id enable[78]
```

[69] Establishes a default route to the LAN interface of the E-SBC
[70] Establishes the route to the interface of Cisco Unity Express (CUE)
[71] Establishes a path for remote administration purposes
[72] Points to the PSTN_Incoming translation profile
[73] % wildcard states that the previous value is repeated any number of times including zero
[74] This section is for configuration of the analog voice-port for fax operation
[75] This disables the echo-cancel function which has minimal effect in relation to resources used
[76] Fax transmission does not utilize comfort noise
[77] This assigns an extension number to the endpoint
[78] Enables caller-id for outbound transmissions

6.4 Cisco Unified Border Element (CUBE) Configuration

Cisco Unified Border Element (CUBE) may be needed for routing between internal networks to the E-SBC on the external network.

The SIP-SIP calling, interface binding and in-call signaling is enabled using the following commands:

```
voice service voip
 allow-connections sip to sip
 early-offer forced[i]
 fax protocol pass-through g711ulaw[ii]
 sip
  bind control source-interface Loopback0
  bind media source-interface Loopback0
  min-se 2000
  header-passing
  asserted-id pai[iii]
  privacy pstn[iv]
```

[i] **`early-offer forced`** – This feature the alters the default configuration of the Cisco CUBE from not distinguishing SIP Delayed-Offer to Early-Offer call flows, to forcing the CUBE to generate an Early-Offer with the configured codecs for an incoming Delayed-Offer INVITE. This is required for RFC 2833 In-band DTMF from the CUCM to interwork with Cox's service.

[ii] **`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.

[iii] **`asserted-id pai`** - To enable the translation to PAID headers in the outgoing header at a global level.

[iv] **`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