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); calls never traverse the Internet.

![Cox Fiber Network](image)

**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](image)

**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](http://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

<table>
<thead>
<tr>
<th>Step</th>
<th>Description</th>
<th>Reference</th>
</tr>
</thead>
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<tr>
<td>Step 1</td>
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<td>Section 6.3.1</td>
</tr>
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<td>Step 2</td>
<td>SIP Registration</td>
<td>Section 6.3.2</td>
</tr>
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<td>Sip Carrier Options</td>
<td>Section 6.3.3</td>
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<td>Step 4</td>
<td>Configure Outbound Dial Peer</td>
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</tr>
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<td>Step 5</td>
<td>Configure ephone-dn</td>
<td>Section 6.3.5</td>
</tr>
<tr>
<td>Step 6</td>
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<td>Section 6.3.6</td>
</tr>
<tr>
<td>Step 7</td>
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</tr>
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</tr>
<tr>
<td>Step 16</td>
<td>Assign Fax/Modem Station</td>
<td>Section 6.3.16</td>
</tr>
<tr>
<td>Step 17</td>
<td>CUBE Configuration</td>
<td>Section 6.4</td>
</tr>
</tbody>
</table>
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

<table>
<thead>
<tr>
<th>Component</th>
<th>Cox Lab Value</th>
<th>Customer Value</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>EdgeMarc E-SBC</strong></td>
<td></td>
<td></td>
</tr>
<tr>
<td>• LAN IP Address</td>
<td>10.70.79.10</td>
<td></td>
</tr>
<tr>
<td>• LAN Subnet Mask</td>
<td>255.255.255.0</td>
<td></td>
</tr>
<tr>
<td><strong>CME IP PBX</strong></td>
<td></td>
<td></td>
</tr>
<tr>
<td>• System IP Address</td>
<td>10.70.79.2</td>
<td></td>
</tr>
<tr>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>• Default Gateway</td>
<td>10.70.95.1</td>
<td></td>
</tr>
<tr>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>• DNS</td>
<td>10.64.1.3</td>
<td></td>
</tr>
</tbody>
</table>

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.

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.

This is the DNS server for the Enterprise network. Cox Communications does not supply DNS services.
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.

```plaintext
interface GigabitEthernet0/0
  description $ES_LAN$2
  ip address 10.70.79.2 255.255.255.0
  duplex full
  speed 100
  media-type rj45
```

6.3.2 SIP Registration

These values allow for verification when communicating with the registrar server

```plaintext
sip-ua
  credentials username 6782383650 password 0 6782383650 realm lab.tekvizion.com
  authentication username 6782383650 password 0 67823836506
  registrar 1 ipv4:10.70.79.10:5060 expires 60
  sip-server ipv4:10.70.79.10:5060
```

---

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.

```text
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.

```text
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 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
b2bua
session target ipv4::10.70.79.50

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

6.3.5 Configure ephone-dn
ephone-dn 1 dual-line
number 3650 secondary 6782383650 no-reg both
pickup-group 1
label 3650
description TestPhone One
name TestPhone One
call-forward busy 96782344101
call-forward noan 96782344101 timeout 20

6.3.6 Configure ephone
ephone 1
device-security-mode none
mac-address FCFB.FBCA.22FE
type 7975
button 1

---

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.

```plaintext
telephony-service
  moh-file-buffer 10000
  internal-call moh-group 0
  authentication credential admin admin
  authentication credential cisco cisco
  max-ephones 15
  max-dn 15
  ip source-address 10.70.79.2 port 2000
  caller-id block code *67
  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:
  load 7960-7940 P00308010200
  load 7971 SCCP70.9-2-3S
  load 7975 SCCP75.9-1-1SR1S
  voicemail 3000
  mwi relay
  max-conferences 12 gain -6
  call-park system application
  moh flash:/RedHotChiliPeppersulaw.wav
  web admin system name cisco password cisco
dn-webedit
time-webedit
transfer-system full-consult
transfer-pattern 9T
secondary-dialtone
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/ 69
  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
```

---

69 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
ip route 10.70.79.50 255.255.255.255 Integrated-Service-Engine1/0
ip route 10.70.90.0 255.255.255.0 10.70.79.1
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
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
no echo-cancel enable
no comfort-noise
station-id number 3674
caller-id enable

---

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
  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
```

---

1. **early-offer forced** – This feature 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.

2. **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.

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

4. **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.