Mitel MI Voice Business [v7.2 SP1 PR1 13.2.1.27] to Verizon Business SIP Trunk via Cisco Unified Border Element (CUBE) 11.1.0 [IOS-XE 3.16.3 – 15.5(3)S3 on ISR4431]

July 2016
# Table of Contents

- **Introduction** .................................................................................................................. 3
- **Network Topology** .......................................................................................................... 4
- **System Components** ........................................................................................................ 5
  - Hardware Requirements ................................................................................................. 5
  - Software Requirements ................................................................................................. 5
- **Features** .......................................................................................................................... 5
  - Features Supported ....................................................................................................... 5
  - Features Not Supported ............................................................................................... 5
  - Caveats ......................................................................................................................... 5
- **Configuration** .................................................................................................................. 6
  - Configuring Cisco Unified Border Element ................................................................. 6
    - Network Interface ...................................................................................................... 6
    - Global Cisco UBE settings ......................................................................................... 9
    - Codecs ....................................................................................................................... 10
    - Dial peer .................................................................................................................... 11
    - Configuration example ............................................................................................ 14
  - Mitel 3300 PBX Configuration ...................................................................................... 39
    - Network Zones .......................................................................................................... 39
    - Add Cisco Cube in the Network Elements: ............................................................... 41
    - Class of Services Options ......................................................................................... 43
    - Trunk Attributes ........................................................................................................ 51
    - SIP Peer Profile ......................................................................................................... 52
- **Acronyms** ...................................................................................................................... 61
- **Important Information** .................................................................................................... 62
Introduction

Service Providers today, such as Verizon, are offering alternative methods to connect to the PSTN via their IP network. Most of these services utilize SIP as the primary signaling method and centralized IP to TDM POP gateways to provide on-net and off-net services.

Verizon Business SIP Trunk is a service provider offering that allows connection to the PSTN and may offer the end customer a viable alternative to traditional PSTN connectivity. A demarcation device between these services and customer owned services is recommended. As an intermediary device between Mitel 3300 and Verizon network, Cisco Unified Border Element (Cisco UBE 11.1.0) can be used. The Cisco Unified Border Element (Cisco UBE) [IOS-XE 3.16.3 – 15.5(3)S3] provides demarcation, security and inter-working and session control services for Mitel 3300 connected to Verizon IP network.

This document assumes the reader is knowledgeable with the terminology and configuration of Mitel 3300. Only configuration settings specifically required for Verizon interoperability are presented. Feature configuration and most importantly the dial plan are customer specific and need individual approach.

- This application note describes how to configure a Mitel 3300 and Cisco Unified Border Element (Cisco UBE v11.1.0) for connectivity to Verizon SIP Trunking service. The deployment model covered in this application note is Mitel 3300 to PSTN via Cisco Unified Border Element (CUBE)

- Testing was performed in accordance to Cisco generic SIP Trunking test methodology and among features verified were – basic calls, DTMF transport, Music on Hold (MOH), unattended and attended transfers, call forward, conferences and High Availability.

- The Cisco Unified Border Element (Cisco UBE) configuration detailed in this document is based on a lab environment with a simple dial-plan used to ensure proper interoperability between Verizon SIP network and Mitel 3300. The configuration described in this document details the important configuration settings to have enabled for interoperability to be successful and care must be taken by the network administrator deploying Mitel 3300 to interoperate to Verizon SIP Trunking network.
The network topology includes the Mitel 3300 and Cisco UBE. Verizon was used as the service provider with a SIP trunk to the Cisco UBE. 2 Cisco Unified Border Elements are used here for High Availability.

- SIP Trunk transport type used between Cisco Unified Border Element and Mitel 3300 is UDP. Configuration for each trunk type is explained in the Dial-peer section of this document.

**IP-PBX - CUBE Trunk Settings:**

<table>
<thead>
<tr>
<th>Setting</th>
<th>Value</th>
</tr>
</thead>
<tbody>
<tr>
<td>REFER Support</td>
<td>YES</td>
</tr>
<tr>
<td>Session Timer</td>
<td>ON</td>
</tr>
<tr>
<td>Early Media support with PRACK</td>
<td>NO</td>
</tr>
</tbody>
</table>
System Components

Hardware Requirements
- Cisco UBE on Cisco ISR 4431 router
- Mitel 3300

Software Requirements
- Cisco CUBE 11.1.0 IOS-XE 15.5(3)S3, XE – 3.16.03
- Mitel Mi Voice Business: 7.2 SP1 PR1 13.2.1.27

Features

Features Supported
- Incoming and outgoing off-net calls using G729
- International Calls and digit manipulations
- Call Conference
- CPE Voice Mail support
- Call hold & Resume with and without MoH
- Unattended and Attended Call transfer
- Call forward (all and no answer)
- DTMF (RFC2833)
- IP-PBX Calling number privacy
- High Availability

Features Not Supported
- Blind call transfer
- Fax SG3
- Tone on hold

Caveats
- With PRACK enabled, there is no voice for an outbound call from Mitel to PSTN after the call is established. PRACK has been disabled.

- Ring back tone is not heard in off-net phone when the Mitel user transfers the call.
- Caller ID updates are not observed on transfer scenarios and appears to be an ITSP issue.
- When CLIR is enabled and IP PBX user1 calls to IPPBX user2 through ITSP/Verizon, anonymous caller-id is not received. ITSP strips the privacy-id and sends it towards the IPPBX.
- Testing is done with only one IP PBX.
Configuring Cisco Unified Border Element

Network Interface
The IP addresses used are for illustration only, the actual IP addresses can vary. The Active/Standby pair share the same virtual IP address and continually exchange status messages.

Figure 2 High Availability topology
CUBE 1:

interface GigabitEthernet0/0/0
ip address 10.64.4.19 255.255.0.0
media-type rj45 negotiation auto

! interface GigabitEthernet0/0/1
no ip address
shutdown media-type rj45
negotiation auto

! interface GigabitEthernet0/0/2
description CUBE LAN interface
ip address 10.80.22.74 255.255.255.0
media-type rj45 negotiation auto
redundancy rii 1
redundancy group 1 ip 10.80.22.100 exclusive

! interface GigabitEthernet0/0/3
description CUBE WAN interface
ip address 192.65.79.114 255.255.255.224
media-type rj45 negotiation auto
redundancy rii 2
redundancy group 1 ip 192.65.79.123 exclusive

!
CUBE 2:

interface GigabitEthernet0/0/0
ip address 10.64.4.20 255.255.0.0
media-type rj45 negotiation auto
!
interface GigabitEthernet0/0/1
no ip address
shutdown
media-type rj45 negotiation auto
!
interface GigabitEthernet0/0/2
ip address 10.80.22.75 255.255.255.0
media-type rj45 negotiation auto
redundancy rii 1
redundancy group 1 ip 10.80.22.100 exclusive
!
interface GigabitEthernet0/0/3
description CUBE WAN Interface
ip address 192.65.79.115 255.255.255.224
media-type rj45 negotiation auto
redundancy rii 2
redundancy group 1 ip 192.65.79.123 exclusive
!

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Page 8 of 64
Global Cisco UBE settings
In order to enable Cisco UBE IP2IP SBC functionality, following command has to be entered:

voice service voip
no ip address trusted authenticate
rtp-port range 16384 32766
address-hiding
mode border-element
allow-connections sip to sip
redundancy-group 1
no supplementary-service sip refer
supplementary-service media-renegotiate
fax protocol pass-through g711ulaw
sip
session refresh
referto-passing
asserted-id pai
early-offer forced
no silent-discard untrusted
midcall-signaling passthru
privacy-policy passthru
privacy-policy send-always
g729 annexb-all
sip-profiles inbound
sip-profiles 100
copy-list 1

**Explanation**

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>allow-connections sip to sip</td>
<td>Allow IP2IP connections between two SIP call legs</td>
</tr>
<tr>
<td>redundancy-group 1</td>
<td>Enable High Availability for the VoIP service</td>
</tr>
<tr>
<td>rtcp keepalive</td>
<td>Enables the CUBE to send rtcp keepalive packets for the session keepalive</td>
</tr>
<tr>
<td>Bind control and media</td>
<td>SIP Global binding is done with the interface pointing to Mitel</td>
</tr>
</tbody>
</table>

**Codecs**

G729 is used as the preferred codec for this testing.

voice class codec 2

codec preference 1 g729r8
codec preference 2 g711ulaw
Dial peer

**Outgoing Dial-peer to Mitel:**

```
!  
dial-peer voice 101 voip
description "Outgoing to Mitel"
destination-pattern 719.......  
session protocol sipv2  
session target ipv4:10.35.31.85:5060  
session transport udp  
voice-class codec 1  
voice-class sip asserted-id pai  
voice-class sip privacy-policy passthru  
voice-class sip early-offer forced  
voice-class sip bind control source-interface GigabitEthernet0/0/2  
voice-class sip bind media source-interface GigabitEthernet0/0/2  
voice-class sip referto-passing  
dtmf-relay rtp-nte  
refer consume  
fax rate 14400  
fax protocol pass-through g711ulaw  
novad  
!```
Incoming Dial-peer From Verizon

dial-peer voice 100 voip
description incoming from VZ
session protocol sipv2
incoming called-number 719....... 
voice-class codec 1
voice-class sip bind control source-interface GigabitEthernet0/0/3
voice-class sip bind media source-interface GigabitEthernet0/0/3
dtmf-relay rtp-nte
fax-relay ecm disable
fax rate 14400
fax protocol pass-through g711ulaw
no vad
!

Outgoing Dial-peer to Verizon:
!
description outgoing call to VZ facing VZ network
huntstop
destination-pattern .T
translate-outgoing called 10
session protocol sipv2
session target ipv4:63.87.147.30:5074
session transport udp
voice-class codec 1
voice-class sip rel1xx disable
voice-class sip asserted-id pai
voice-class sip privacy-policy passthru
voice-class sip early-offer forced
voice-class sip bind control source-interface GigabitEthernet0/0/3
voice-class sip bind media source-interface GigabitEthernet0/0/3
dtmf-relay rtp-nte
fax rate 14400
fax protocol pass-through g711ulaw
no vad

Incoming from Mitel:

dial-peer voice 201 voip
description incoming from IP-PBX
session protocol sipv2
session transport udp
incoming called-number [2-9]T
voice-class codec 1
voice-class sip copy-list 1
voice-class sip bind control source-interface GigabitEthernet0/0/2
voice-class sip bind media source-interface GigabitEthernet0/0/2
dtmf-relay rtp-nte
fax rate 14400
fax protocol pass-through g711ulaw
no vad
Configuration example
The following configuration snippet contains a sample configuration of Cisco UBE with all parameters mentioned previously.

**Active Cisco UBE:**
CISCO_4K_ROUTER1#sh run

version 15.5

service timestamps debug datetime msec
service timestamps log datetime msec
no platform punt-keepalive disable-kernel-core
!
hostname CISCO_4K_ROUTER2
!
boot-start-marker
boot system flash isr4400-universalk9.03.16.01a.S.155-3.51a-ext.SPA.bin
boot-end-marker
!
vrf definition Mgmt-intf
!
address-family ipv4
exit-address-family
!
address-family ipv6
exit-address-family
aaa new-model

aaa session-id common

!

ipc zone default

association 1

no shutdown

!

ip name-server 8.8.8.8

no ip domain lookup

!

subscriber templating

multilink bundle-name authenticated

!

cts logging verbose

!

!
voice service voip
no ip address trusted authenticate
rtp-port range 16384 32766
address-hiding
mode border-element
allow-connections sip to sip
redundancy-group 1
no supplementary-service sip refer
supplementary-service media-renegotiate
fax protocol pass-through g711ulaw
sip
session refresh
refer-to-passing
asserted-id pai
early-offer forced
no silent-discard untrusted
midcall-signaling passthru
privacy-policy passthru
privacy-policy send-always
g729 annexb-all
sip-profiles inbound
sip-profiles 100
copy-list 1

voice class codec 1
codec preference 1 g729r8
codec preference 2 g711ulaw

voice class codec 2
codec preference 1 g711ulaw
codec preference 2 g729r8

voice class sip-profiles 1
request INVITE peer-header sip REFERRED-BY copy "sip:(.*)@" u01
request UPDATE sip-header P-Asserted-Identity modify ".*@(.*)" "P-asserted-identity <sip:\u01@\1"

voice class sip-profiles 100
request INVITE sip-header Allow-Header modify " UPDATE, " " "
request REINVITE sip-header Allow-Header modify " UPDATE, " " "
response 180 sip-header Allow-Header modify " UPDATE, " " "
response 200 sip-header Allow-Header modify " UPDATE, " " "

voice class sip-copylist 1
sip-header REFERRED-BY
license udi pid ISR4431/K9 sn FOC18232988
license boot level appxk9
license boot level uck9
license boot level securityk9
  
spanning-tree extend system-id
  
redundancy
  
mode none
  
anapplication redundancy
  
group 1
    
name voice-b2bha
    
priority 100 failover threshold 75
    
timers delay 30 reload 60
    
control GigabitEthernet0/0/0 protocol 1
    
data GigabitEthernet0/0/0
    
track 1 shutdown
    
track 2 shutdown
  
  
vlan internal allocation policy ascending
  
track 1 interface GigabitEthernet0/0/2 line-protocol
  
track 2 interface GigabitEthernet0/0/3 line-protocol
! translation-rule 2
Rule 1 2202 7192083392
!

! translation-rule 10
Rule 1 2141280877 7192083392
!

! translation-rule 11
Rule 1 7192083392 3392
!

interface GigabitEthernet0/0/0
  ip address 10.64.4.20 255.255.0.0
  media-type rj45
  negotiation auto
!

interface GigabitEthernet0/0/1
  no ip address
  shutdown
  media-type rj45
  negotiation auto
!
interface GigabitEthernet0/0/2
ip address 10.80.22.74 255.255.255.0
media-type rj45
negotiation auto
redundancy rii 1
redundancy group 1 ip 10.80.22.100 exclusive
!
interface GigabitEthernet0/0/3
description Wan Interface
ip address 192.65.79.114 255.255.255.224
media-type rj45
negotiation auto
redundancy rii 2
redundancy group 1 ip 192.65.79.123 exclusive
!
interface GigabitEthernet0
vrf forwarding Mgmt-intf
no ip address
shutdown
negotiation auto
!
interface Vlan1
no ip address
shutdown
!
ip forward-protocol nd
no ip http server
no ip http secure-server
ip rtcp report interval 60000
ip route 0.0.0.0 0.0.0.0 192.65.79.97
ip route 10.35.31.0 255.255.255.0 10.80.22.1
ip route 10.64.0.0 255.255.0.0 10.80.22.1
ip route 10.80.22.0 255.255.255.0 10.80.22.1
!
control-plane
!
mgcp behavior rsip-range tgcp-only
mgcp behavior comedia-role none
mgcp behavior comedia-check-media-src disable
mgcp behavior comedia-sdp-force disable
!
mgcp profile default
!
dial-peer voice 100 voip

description incoming from VZ

session protocol sipv2

incoming called-number 719.......

voice-class codec 1

voice-class sip bind control source-interface GigabitEthernet0/0/3

voice-class sip bind media source-interface GigabitEthernet0/0/3

dtmf-relay rtp-nte

fax-relay ecm disable

fax rate 14400

fax protocol pass-through g711ulaw

no vad

!

dial-peer voice 101 voip

description "Outgoing to IP-PBX"

destination-pattern 719.......

session protocol sipv2

session target ipv4:10.35.31.85:5060

session transport udp

voice-class codec 1

voice-class sip asserted-id pai

voice-class sip privacy-policy passthru
voice-class sip early-offer forced
voice-class sip bind control source-interface GigabitEthernet0/0/2
voice-class sip bind media source-interface GigabitEthernet0/0/2
voice-class sip referto-passing
dtmf-relay rtp-n-te
refer consume
fax rate 14400
fax protocol pass-through g711ulaw
no vad
!
dial-peer voice 200 voip
description outgoing call to VZ facing VZ network
huntstop
destination-pattern .T
translate-outgoing called 10
session protocol sipv2
session target ipv4:63.87.147.30:5074
session transport udp
voice-class codec 1
voice-class sip rel1xx disable
voice-class sip asserted-id pai
voice-class sip privacy-policy passthru
voice-class sip early-offer forced
voice-class sip bind control source-interface GigabitEthernet0/0/3
voice-class sip bind media source-interface GigabitEthernet0/0/3
dtmf-relay rtp-nte
fax rate 14400
fax protocol pass-through g711ulaw
no vad

!
dial-peer voice 201 voip
description incoming from IP-PBX
session protocol sipv2
session transport udp
incoming called-number [2-9]T
voice-class codec 1
voice-class sip copy-list 1
voice-class sip bind control source-interface GigabitEthernet0/0/2
voice-class sip bind media source-interface GigabitEthernet0/0/2
dtmf-relay rtp-nte
fax rate 14400
fax protocol pass-through g711ulaw
no vad

!
gateway
media-inactivity-criteria all
timer receive-rtcp 5
timer receive-rtp 86400
!
sip-ua
no remote-party-id
retry invite 1
timers expires 1800000
timers options 1000
connection-reuse
!
line con 0
stopbits 1
line aux 0
stopbits 1
line vty 0
session-timeout 90
exec-timeout 960 0
no activation-character
logging synchronous
transport preferred ssh
transport input all
stopbits 1
line vty 1 4
exec-timeout 960 0
logging synchronous
transport input all
!
end
Standby Cisco UBE:

version 15.5

service timestamps debug datetime msec

service timestamps log datetime msec

no platform punt-keepalive disable-kernel-core

!

hostname CISCO_4K_ROUTER2

!

boot-start-marker

boot system flash isr4400-universalk9.03.16.01a.S.155-3.51a-ext.SPA.bin

boot-end-marker

!

vrf definition Mgmt-intf

!

address-family ipv4

exit-address-family

!

address-family ipv6

exit-address-family

!

aaa new-model

aaa session-id common

!
ipc zone default
association 1
no shutdown
!
ip name-server 8.8.8.8
no ip domain lookup
!
subscriber templating
multilink bundle-name authenticated
cts logging verbose
!
voice service voip
no ip address trusted authenticate
rtp-port range 16384 32766
address-hiding
mode border-element
allow-connections sip to sip
redundancy-group 1
no supplementary-service sip refer
supplementary-service media-renegotiate
fax protocol pass-through g711ulaw
sip
session refresh
referred-passing
asserted-id pai
early-offer forced
no silent-discard untrusted
midcall-signaling passthru
privacy-policy passthru
privacy-policy send-always
g729 annexb-all
sip-profiles inbound
sip-profiles 100
copy-list 1
!
voice class codec 1
codec preference 1 g729r8
codec preference 2 g711ulaw
!
voice class codec 2
codec preference 1 g711ulaw
codec preference 2 g729r8
!
voice class sip-profiles 1
request INVITE peer-header sip REFERRED-BY copy "sip:(.*)@" u01
request UPDATE sip-header P-Asserted-Identity modify ".*@(.*).*P-asserted-identity <sip:\u01@\1"
!
voice class sip-profiles 100
request INVITE sip-header Allow-Header modify " UPDATE, " 
request REINVITE sip-header Allow-Header modify " UPDATE, "
response 180 sip-header Allow-Header modify " UPDATE, " " "
response 200 sip-header Allow-Header modify " UPDATE, " " "
!
voice class sip-copylist 1
sip-header REFERRED-BY
!
license udi pid ISR4431/K9 sn FOC18232988
license boot level appxk9
license boot level uck9
license boot level securityk9
!
spanning-tree extend system-id
!
redundancy
mode none
application redundancy
group 1
name voice-b2bha
priority 100 failover threshold 75
timers delay 30 reload 60
control GigabitEthernet0/0/0 protocol 1
data GigabitEthernet0/0/0
track 1 shutdown
track 2 shutdown
!
vlan internal allocation policy ascending
!
track 1 interface GigabitEthernet0/0/2 line-protocol
track 2 interface GigabitEthernet0/0/3 line-protocol
!
translation-rule 2
Rule 1 2202 7192083392
!
translation-rule 10
Rule 1 2141280877 7192083392
!
translation-rule 11
Rule 1 7192083392 3392
!
interface GigabitEthernet0/0/0
ip address 10.64.4.20 255.255.0.0
media-type rj45
negotiation auto
!
interface GigabitEthernet0/0/1
no ip address
shutdown
media-type rj45
negotiation auto
!
interface GigabitEthernet0/0/2
  ip address 10.80.22.75 255.255.255.0
  media-type rj45
  negotiation auto
  redundancy rii 1
  redundancy group 1 ip 10.80.22.100 exclusive
!
interface GigabitEthernet0/0/3
  description Wan Interface
  ip address 192.65.79.115 255.255.255.224
  media-type rj45
  negotiation auto
  redundancy rii 2
  redundancy group 1 ip 192.65.79.123 exclusive
!
interface GigabitEthernet0
  vrf forwarding Mgmt-intf
  no ip address
  shutdown
  negotiation auto
!
interface Vlan1
  no ip address
  shutdown
!
ip forward-protocol nd
no ip http server
no ip http secure-server
ip rtcp report interval 60000
ip route 0.0.0.0 0.0.0.0 192.65.79.97
ip route 10.35.31.0 255.255.255.0 10.80.22.1
ip route 10.64.0.0 255.255.0.0 10.80.22.1
ip route 10.80.22.0 255.255.255.0 10.80.22.1
!
control-plane
!
mgcp behavior rsip-range tgcp-only
mgcp behavior comedia-role none
mgcp behavior comedia-check-media-src disable
mgcp behavior comedia-sdp-force disable
!
mgcp profile default
!
dial-peer voice 100 voip
description incoming from VZ
session protocol sipv2
incoming called-number 719......
voice-class codec 1
voice-class sip bind control source-interface GigabitEthernet0/0/3
voice-class sip bind media source-interface GigabitEthernet0/0/3
dtmf-relay rtp-nnte
fax-relay ecm disable
fax rate 14400
fax protocol pass-through g711ulaw
no vad
!
dial-peer voice 101 voip
description "Outgoing to IP-PBX"
destination-pattern 719......
session protocol sipv2
session target ipv4:10.35.31.85:5060
session transport udp
voice-class codec 1
voice-class sip asserted-id pai
voice-class sip privacy-policy passthru
voice-class sip early-offer forced
voice-class sip bind control source-interface GigabitEthernet0/0/2
voice-class sip bind media source-interface GigabitEthernet0/0/2
voice-class sip refer-to-passing
dtmf-relay rtp-nte
refer consume
fax rate 14400
fax protocol pass-through g711ulaw
no vad
!
dial-peer voice 200 voip
description outgoing call to VZ facing VZ network
huntstop
destination-pattern .T
translate-outgoing called 10
session protocol sipv2
session target ipv4:63.87.147.30:5074
session transport udp
voice-class codec 1
voice-class sip rel1xx disable
voice-class sip asserted-id pai
voice-class sip privacy-policy passthru
voice-class sip early-offer forced
voice-class sip bind control source-interface GigabitEthernet0/0/3
voice-class sip bind media source-interface GigabitEthernet0/0/3
dtmf-relay rtp-nte
fax rate 14400
fax protocol pass-through g711ulaw
no vad
!
dial-peer voice 201 voip
description incoming from IP-PBX
session protocol sipv2
session transport udp
incoming called-number [2-9]T
voice-class codec 1
voice-class sip copy-list 1
voice-class sip bind control source-interface GigabitEthernet0/0/2
voice-class sip bind media source-interface GigabitEthernet0/0/2
dtmf-relay rtp-nte
fax rate 14400
fax protocol pass-through g711ulaw
no vad
!
dial-peer voice 786 voip
description test refer
shutdown
session protocol sipv2
session target ipv4:10.35.31.85:5060
session transport udp
voice-class codec 1
voice-class sip bind control source-interface GigabitEthernet0/0/2
voice-class sip bind media source-interface GigabitEthernet0/0/2
voice-class sip refer-to-passing
dtmf-relay rtp-nte
no vad
!
!
gateway
media-inactivity-criteria all
timer receive-rtcp 5
timer receive-rtp 86400
!
sip-ua
no remote-party-id
retry invite 1
timers expires 1800000
timers options 1000
connection-reuse
!
line con 0
stopbits 1
line aux 0
stopbits 1
line vty 0
session-timeout 90
exec-timeout 960 0
no activation-character
logging synchronous
transport preferred ssh
transport input all
stopbits 1
line vty 1 4
exectimeout 960 0
logging synchronous
transport input all
!
!
end
Mitel 3300 PBX Configuration

Network Zones

1. Navigate to Voice Network-> Network Zone.
2. Select a Network Zone and Click Change.

![Network Zones](image)

**Figure 3 Network Zones -1**

1. A new Pop-up window is opened as shown in the figure below.
2. Set Intra-Zone Compression: Yes is selected. This is where the preferred codec is configured. For this configuration G.729 is used as preferred codec.
3. Set Label: Cisco Cube is given for this example
<table>
<thead>
<tr>
<th>Network Zones</th>
<th></th>
</tr>
</thead>
<tbody>
<tr>
<td>Zone ID</td>
<td>2</td>
</tr>
<tr>
<td>Intra-zone Compression</td>
<td></td>
</tr>
<tr>
<td>Group Zone</td>
<td></td>
</tr>
<tr>
<td>Intra-zone Fax Profile</td>
<td>1</td>
</tr>
<tr>
<td>Label</td>
<td>Ciscocube</td>
</tr>
<tr>
<td>SMDR Tag</td>
<td></td>
</tr>
<tr>
<td>Time Zone</td>
<td>America/Chicago</td>
</tr>
<tr>
<td>LBN Prefix</td>
<td></td>
</tr>
<tr>
<td>Zone CESID</td>
<td></td>
</tr>
<tr>
<td>Default Billing Number</td>
<td></td>
</tr>
<tr>
<td>Default CPN</td>
<td></td>
</tr>
</tbody>
</table>

**Figure 3 Network Zones -2**
Add Cisco Cube in the Network Elements:

1. Navigate to Voice Network > Network Elements.
2. In the Right Frame of the window Click Add

3. A new pop-up window opens as shown in the figure below.
4. Set Name: Cube is given for this example.
5. Set Type: Other is selected from the drop down menu.
6. Set FQDN or IP Address: Assign the IP Address 10.80.22.74(Cube LAN IP).
7. Set Zone: 2 is given for this example. Refer to the Previous section
8. Set Peer Transport: UDP is selected from drop down menu.
10. All the other values are set default values.
11. Click Save.
**Figure 4 Configure Network elements -2**

<table>
<thead>
<tr>
<th><strong>Network Elements</strong></th>
<th></th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>Name</strong></td>
<td>Cube</td>
</tr>
<tr>
<td><strong>Type</strong></td>
<td>Other</td>
</tr>
<tr>
<td><strong>FQDN or IP Address</strong></td>
<td>10.80.22.74</td>
</tr>
<tr>
<td><strong>Local</strong></td>
<td>False</td>
</tr>
<tr>
<td><strong>Version</strong></td>
<td></td>
</tr>
<tr>
<td><strong>Zone</strong></td>
<td>2</td>
</tr>
<tr>
<td><strong>ARID</strong></td>
<td></td>
</tr>
<tr>
<td><strong>SIP Peer</strong></td>
<td>✅</td>
</tr>
<tr>
<td><strong>SIP Peer Specific</strong></td>
<td></td>
</tr>
<tr>
<td><strong>SIP Peer Transport</strong></td>
<td>UDP</td>
</tr>
<tr>
<td><strong>SIP Peer Port</strong></td>
<td>5060</td>
</tr>
<tr>
<td><strong>External SIP Proxy FQDN or IP Address</strong></td>
<td></td>
</tr>
<tr>
<td><strong>External SIP Proxy Transport</strong></td>
<td>UDP</td>
</tr>
<tr>
<td><strong>External SIP Proxy Port</strong></td>
<td>0</td>
</tr>
<tr>
<td><strong>SIP Registrar FQDN or IP Address</strong></td>
<td></td>
</tr>
<tr>
<td><strong>SIP Registrar Transport</strong></td>
<td>UDP</td>
</tr>
<tr>
<td><strong>SIP Registrar Port</strong></td>
<td>0</td>
</tr>
<tr>
<td><strong>SIP Peer Status</strong></td>
<td>Auto-Detect/Normal</td>
</tr>
</tbody>
</table>
Class of Services Options

1. Navigate to **System Properties-> System Feature Settings-> Class of Service Options**.
2. In right frame of the window, select a Class of Service Number and click on **Change**.

![Configure COS](image)

**Figure 5: Configure COS – 1**
### Figure 5: Configure COS – 2

<table>
<thead>
<tr>
<th>Class Of Service Number</th>
<th>10</th>
</tr>
</thead>
<tbody>
<tr>
<td>Comment</td>
<td>Cube</td>
</tr>
</tbody>
</table>

#### ACD

<table>
<thead>
<tr>
<th>ACD Agent Behavior on No Answer</th>
<th>Logout</th>
</tr>
</thead>
<tbody>
<tr>
<td>ACD Agent No Answer Timer</td>
<td>15</td>
</tr>
<tr>
<td>ACD Make Busy on Login</td>
<td></td>
</tr>
<tr>
<td>ACD Silent Monitor Accept</td>
<td></td>
</tr>
<tr>
<td>ACD Silent Monitor Accept Monitoring Non-Prime Lines</td>
<td></td>
</tr>
<tr>
<td>ACD Silent Monitor Allowed</td>
<td></td>
</tr>
<tr>
<td>ACD Silent Monitor Notification</td>
<td></td>
</tr>
<tr>
<td>Follow 2nd Alternate Reroute for Recall to Busy ACD Agent</td>
<td></td>
</tr>
<tr>
<td>Work Timer</td>
<td>0</td>
</tr>
</tbody>
</table>

#### Announce

| Call Announce Line                           |        |
| Off-Hook Voice Announce Allowed             |        |
| Handsfree AnswerBack Allowed                |        |

#### Busy Override

| Busy Override Security                       |        |
| Disable Executive Busy Override Tone        |        |
| Executive Busy Override                      |        |

#### Call Control Timer

<p>| Busy Tone Timer                             | 30     |
| Dialing Conflict Timer                      | 3      |
| First Digit Timer                           | 15     |
| Inter Digit Timer                           | 10     |
| Lockout Timer                               | 45     |</p>
<table>
<thead>
<tr>
<th>Feature</th>
<th>Value</th>
<th>Options</th>
</tr>
</thead>
<tbody>
<tr>
<td>Call Duration</td>
<td></td>
<td></td>
</tr>
<tr>
<td>Call Duration Forced Cleardown Timer</td>
<td>10</td>
<td></td>
</tr>
<tr>
<td>Enable Call Duration Limit on External Calls</td>
<td></td>
<td>No/Yes</td>
</tr>
<tr>
<td>Enable Call Duration Limit on Internal Calls</td>
<td></td>
<td>No/Yes</td>
</tr>
<tr>
<td>Call Forwarding/Rerouting</td>
<td></td>
<td></td>
</tr>
<tr>
<td>Call Forward - Delay</td>
<td>0</td>
<td></td>
</tr>
<tr>
<td>Call Forward No Answer Timer</td>
<td>15</td>
<td>No/Yes</td>
</tr>
<tr>
<td>Call Forward Override</td>
<td></td>
<td>No/Yes</td>
</tr>
<tr>
<td>Call Forwarding (External Destination)</td>
<td></td>
<td>No/Yes</td>
</tr>
<tr>
<td>Call Forwarding (Internal Destination)</td>
<td></td>
<td>No/Yes</td>
</tr>
<tr>
<td>Call Forwarding Accept</td>
<td></td>
<td>No/Yes</td>
</tr>
<tr>
<td>Call Reroute after CFFM to Busy Destination</td>
<td></td>
<td>No/Yes</td>
</tr>
<tr>
<td>Call Forwarding Reminder Ring (CFFM and CFIAH only)</td>
<td>0</td>
<td>No/Yes</td>
</tr>
<tr>
<td>Disable Call Reroute Chaining On Diversion</td>
<td></td>
<td>No/Yes</td>
</tr>
<tr>
<td>Group Call Forward Follow Me Accept</td>
<td></td>
<td>No/Yes</td>
</tr>
<tr>
<td>Group Call Forward Follow Me Allow</td>
<td></td>
<td>No/Yes</td>
</tr>
<tr>
<td>Third Party Call Forward Follow Me Accept</td>
<td></td>
<td>No/Yes</td>
</tr>
<tr>
<td>Third Party Call Forward Follow Me Allow</td>
<td></td>
<td>No/Yes</td>
</tr>
<tr>
<td>Use Held Party Device for Call Re-routing</td>
<td></td>
<td>No/Yes</td>
</tr>
<tr>
<td>Call Hold</td>
<td></td>
<td></td>
</tr>
<tr>
<td>Call Hold</td>
<td></td>
<td>No/Yes</td>
</tr>
<tr>
<td>Call Hold - Retrieve with Hold Key</td>
<td></td>
<td>No/Yes</td>
</tr>
<tr>
<td>Call Hold Remote Retrieve</td>
<td></td>
<td>No/Yes</td>
</tr>
<tr>
<td>Call Hold Timer</td>
<td>30</td>
<td></td>
</tr>
<tr>
<td>Local Music On Hold source</td>
<td></td>
<td>No/Yes</td>
</tr>
<tr>
<td>Music on Hold on Transfer</td>
<td></td>
<td>No/Yes</td>
</tr>
<tr>
<td>Use Called Party Call Hold Timer</td>
<td></td>
<td>No/Yes</td>
</tr>
</tbody>
</table>

**Figure 5: Configure COS – 3**
## Call Park

<table>
<thead>
<tr>
<th>Feature</th>
<th>Value</th>
</tr>
</thead>
<tbody>
<tr>
<td>Call Park Timer</td>
<td>180</td>
</tr>
<tr>
<td>Call Park-Allowed To Park</td>
<td>Yes</td>
</tr>
</tbody>
</table>

## Call Pickup

<table>
<thead>
<tr>
<th>Feature</th>
<th>Value</th>
</tr>
</thead>
<tbody>
<tr>
<td>Allow Directed Call Pickup Of Attendant Call</td>
<td>No</td>
</tr>
<tr>
<td>Call Pickup Dialed Accept</td>
<td>No</td>
</tr>
<tr>
<td>Call Pickup Directed Accept</td>
<td>Yes</td>
</tr>
</tbody>
</table>

## Call Privacy

<table>
<thead>
<tr>
<th>Feature</th>
<th>Value</th>
</tr>
</thead>
<tbody>
<tr>
<td>Call Privacy</td>
<td>Yes</td>
</tr>
<tr>
<td>Calling Party Name Substitution</td>
<td>No</td>
</tr>
<tr>
<td>Name Suppression on outgoing Trunk Call</td>
<td>Yes</td>
</tr>
<tr>
<td>Privacy Released</td>
<td>Yes</td>
</tr>
<tr>
<td>Public Network Identity Provided</td>
<td>Yes</td>
</tr>
</tbody>
</table>

## Call Waiting

<table>
<thead>
<tr>
<th>Feature</th>
<th>Value</th>
</tr>
</thead>
<tbody>
<tr>
<td>Call Waiting Swap</td>
<td>Yes</td>
</tr>
<tr>
<td>ONS CLASS/CLIP: Visual Call Waiting</td>
<td>Yes</td>
</tr>
</tbody>
</table>

## Campon

<table>
<thead>
<tr>
<th>Feature</th>
<th>Value</th>
</tr>
</thead>
<tbody>
<tr>
<td>Auto Campon Timer</td>
<td></td>
</tr>
<tr>
<td>Campon Recall Timer</td>
<td>10</td>
</tr>
</tbody>
</table>

## Direct Voice Call

<table>
<thead>
<tr>
<th>Feature</th>
<th>Value</th>
</tr>
</thead>
<tbody>
<tr>
<td>Direct Voice Call - Accept</td>
<td>Yes</td>
</tr>
<tr>
<td>Direct Voice Call - Allow</td>
<td>Yes</td>
</tr>
<tr>
<td>Direct Voice Call - Maximize Volume</td>
<td>Yes</td>
</tr>
</tbody>
</table>

---

**Figure 5: Configure COS – 4**
<table>
<thead>
<tr>
<th>Feature</th>
<th>Yes</th>
<th>No</th>
</tr>
</thead>
<tbody>
<tr>
<td>After Answer Display Time</td>
<td></td>
<td></td>
</tr>
<tr>
<td>Calling Name Display - Internal - ONS</td>
<td></td>
<td></td>
</tr>
<tr>
<td>Calling Number Display - Internal - ONS</td>
<td></td>
<td></td>
</tr>
<tr>
<td>Display ANI/DNIS/ISDN Calling/Called Number</td>
<td></td>
<td></td>
</tr>
<tr>
<td>Display ANI/ISDN Calling Number Only</td>
<td></td>
<td></td>
</tr>
<tr>
<td>Display Caller ID on multicall/keylines</td>
<td></td>
<td></td>
</tr>
<tr>
<td>Display Caller ID On Multicall/Keylines Timer</td>
<td>5</td>
<td></td>
</tr>
<tr>
<td>Display Caller ID On Single Line Displays For Forwarded Calls</td>
<td></td>
<td></td>
</tr>
<tr>
<td>Display Dialed Digits during Outgoing Calls</td>
<td></td>
<td></td>
</tr>
<tr>
<td>Display DNIS/Called Number Before Digit Modification</td>
<td></td>
<td></td>
</tr>
<tr>
<td>Display Held Call ID on Transfer</td>
<td></td>
<td></td>
</tr>
<tr>
<td>Display Transfer Destination on Recall</td>
<td></td>
<td></td>
</tr>
<tr>
<td>Hot Desk External User - Display Internal Calling ID</td>
<td></td>
<td></td>
</tr>
<tr>
<td>Maintain Ringing Party During Recall</td>
<td></td>
<td></td>
</tr>
<tr>
<td>Non-Prime Public Network Identity</td>
<td></td>
<td></td>
</tr>
<tr>
<td>Originator's Display Update In Call Forwarding/Rerouting</td>
<td></td>
<td></td>
</tr>
<tr>
<td>Suppress Delivery of Caller ID Display between Sets</td>
<td></td>
<td></td>
</tr>
<tr>
<td>Suppress Delivery of Caller ID Display between Sets - Override</td>
<td></td>
<td></td>
</tr>
<tr>
<td>Suppress Display Of Account Code Numbers</td>
<td></td>
<td></td>
</tr>
<tr>
<td>Suppress Redial Display</td>
<td></td>
<td></td>
</tr>
</tbody>
</table>

<table>
<thead>
<tr>
<th>Feature</th>
<th>Yes</th>
<th>No</th>
</tr>
</thead>
<tbody>
<tr>
<td>Fax</td>
<td></td>
<td></td>
</tr>
<tr>
<td>Campon Tone Security</td>
<td></td>
<td></td>
</tr>
<tr>
<td>External Trunk Standard Ringback</td>
<td></td>
<td></td>
</tr>
<tr>
<td>Fax Capable</td>
<td></td>
<td></td>
</tr>
<tr>
<td>Return Disconnect Tone When Far End Party Clears</td>
<td></td>
<td></td>
</tr>
</tbody>
</table>

**Figure 5: Configure COS – 5**
**Figure 5: Configure COS – 6**

<table>
<thead>
<tr>
<th>Category</th>
<th>Options</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>HCI</strong></td>
<td></td>
</tr>
<tr>
<td>HCI/CTI/TAPI Call Control Allowed</td>
<td>○ No ○ Yes</td>
</tr>
<tr>
<td>HCI/CTI/TAPI Monitor Allowed</td>
<td>○ No ○ Yes</td>
</tr>
<tr>
<td><strong>Hot Desk</strong></td>
<td></td>
</tr>
<tr>
<td>Green BLF Lamp for Logged in Hotdesk User</td>
<td>○ No ○ Yes</td>
</tr>
<tr>
<td>Hot Desk External User - Allow Mid-Call Features</td>
<td>○ No ○ Yes</td>
</tr>
<tr>
<td>Hot Desk External User - Answer Confirmation</td>
<td>○ No ○ Yes</td>
</tr>
<tr>
<td>Hot Desk External User - Dial Tone on Call Complete</td>
<td>○ No ○ Yes</td>
</tr>
<tr>
<td>Hot Desk External User - Permanent Login</td>
<td>○ No ○ Yes</td>
</tr>
<tr>
<td>Hot Desk External User - Remote MWI Enable Feature Access Code</td>
<td></td>
</tr>
<tr>
<td>Hot Desk External User - Remote MWI Disable Feature Access Code</td>
<td></td>
</tr>
<tr>
<td>Hot Desk Login Accept</td>
<td>○ No ○ Yes</td>
</tr>
<tr>
<td>Hot Desk Remote Logout Enabled</td>
<td>○ No ○ Yes</td>
</tr>
<tr>
<td><strong>Miscellaneous</strong></td>
<td></td>
</tr>
<tr>
<td>Backlighting - Enabled</td>
<td>○ No ○ Yes</td>
</tr>
<tr>
<td>Clear All Features Remote</td>
<td>○ No ○ Yes</td>
</tr>
<tr>
<td>Force Device Busy If Any Line In Use</td>
<td>○ No ○ Yes</td>
</tr>
<tr>
<td>Handset Volume Adjustment Saved</td>
<td>○ No ○ Yes</td>
</tr>
<tr>
<td>Head Set Switch Mute</td>
<td>○ No ○ Yes</td>
</tr>
<tr>
<td>Multi-Color LED Support - Disable</td>
<td>○ No ○ Yes</td>
</tr>
<tr>
<td>Phone Lock</td>
<td>○ No ○ Yes</td>
</tr>
<tr>
<td>Reseize Timer</td>
<td>180</td>
</tr>
<tr>
<td>Timed Reminder Allowed</td>
<td>○ No ○ Yes</td>
</tr>
<tr>
<td>User Inactivity Timer</td>
<td>0</td>
</tr>
<tr>
<td>Feature</td>
<td>Options</td>
</tr>
<tr>
<td>----------------------------------------------</td>
<td>-------------</td>
</tr>
<tr>
<td>Paging</td>
<td>No □ Yes □</td>
</tr>
<tr>
<td>Group Page Accept</td>
<td>No □ Yes □</td>
</tr>
<tr>
<td>Group Page Allow</td>
<td>No □ Yes □</td>
</tr>
<tr>
<td>Loudspeaker Pager Equivalent Zone Override Security</td>
<td>No □ Yes □</td>
</tr>
<tr>
<td>Loudspeaker Pager Override</td>
<td>No □ Yes □</td>
</tr>
<tr>
<td>Pager Access All Zones</td>
<td>No □ Yes □</td>
</tr>
<tr>
<td>Pager Access Individual Zones</td>
<td>No □ Yes □</td>
</tr>
<tr>
<td>PC Port</td>
<td>No □ Yes □</td>
</tr>
<tr>
<td>PC Port On IP Device - Disable</td>
<td>No □ Yes □</td>
</tr>
<tr>
<td>RAD</td>
<td></td>
</tr>
<tr>
<td>Answer Plus Delay To Message Timer</td>
<td>5</td>
</tr>
<tr>
<td>Answer Plus Expected Off-hook Timer</td>
<td>30</td>
</tr>
<tr>
<td>Answer Plus Message Length Timer</td>
<td>10</td>
</tr>
<tr>
<td>Answer Plus System Reroute Timer</td>
<td>0</td>
</tr>
<tr>
<td>Recorded Announcement Device</td>
<td>No □ Yes □</td>
</tr>
<tr>
<td>Recorded Announcement Device - Advanced</td>
<td>No □ Yes □</td>
</tr>
<tr>
<td>Ringing</td>
<td></td>
</tr>
<tr>
<td>Delay Ring Timer</td>
<td>10</td>
</tr>
<tr>
<td>No Answer Recall Timer</td>
<td>17</td>
</tr>
<tr>
<td>Ringing Line Select</td>
<td>No □ Yes □</td>
</tr>
<tr>
<td>Ringing Timer</td>
<td>180</td>
</tr>
<tr>
<td>SMDR</td>
<td></td>
</tr>
<tr>
<td>SMDR External</td>
<td>No □ Yes □</td>
</tr>
<tr>
<td>SMDR Internal</td>
<td>No □ Yes □</td>
</tr>
</tbody>
</table>

Figure 5: Configure COS – 7
## Trunk

<table>
<thead>
<tr>
<th>Feature</th>
<th>Value</th>
</tr>
</thead>
<tbody>
<tr>
<td>ANI/DNIS/ISDN Number Delivery Trunk</td>
<td>Yes</td>
</tr>
<tr>
<td>DASS II OLI/TLI Provided</td>
<td>No</td>
</tr>
<tr>
<td>Public Network Access via DPNSS</td>
<td>Yes</td>
</tr>
<tr>
<td>Public Network To Public Network Connection Allowed</td>
<td>Yes</td>
</tr>
<tr>
<td>Public Trunk</td>
<td>No</td>
</tr>
<tr>
<td>R2 Call Progress Tone</td>
<td>No</td>
</tr>
<tr>
<td>Suppress Simulated CCM after ISDN Progress</td>
<td>Yes</td>
</tr>
<tr>
<td>Trunk Calling Party Identification</td>
<td>No</td>
</tr>
<tr>
<td>Trunk Flash Allowed</td>
<td>No</td>
</tr>
<tr>
<td>Two B-Channel Transfer Allowed</td>
<td>No</td>
</tr>
</tbody>
</table>

### Voice Mail

<table>
<thead>
<tr>
<th>Feature</th>
<th>Value</th>
</tr>
</thead>
<tbody>
<tr>
<td>COV/ONS/E&amp;M Voice Mail Port</td>
<td>Yes</td>
</tr>
<tr>
<td>ONS VMail-Delay Dial Tone Timer</td>
<td>5</td>
</tr>
</tbody>
</table>

**Figure 5: Configure COS – 8**
Trunk Attributes

1. Navigate To **Trunks > Trunk Attributes**.
2. Right click on the Trunk Service number and click **Change**.

![Figure 6: Trunk Attributes](image)

3. A new pop-up window opens as shown in the figure below.
4. **Set Release Link Trunk**: No
5. **Set Call Recognition Service**: Off
6. **Set Class of Service**: Enter the Class of Service Number you have created in the previous section
7. **Set Class of Restriction**: Set the Class of Restriction, 2 is used here for example.
8. **Set Baud Rate**: 9600
9. **Set Intercept Number**: 2
10. Confirm Non-dial In Trunks Answer Point-Day: is blank (no data)
11. Confirm Non-dial In Trunks Answer Point-Night1: is blank (no data)
12. Confirm Non-dial In Trunks Answer Point-Night2: is blank (no data)
13. **Set Dial In Trunks Incoming Digit Modification-Absorb**: 0
14. Confirm Dial In Trunks Incoming Digit Modification-Insert: is blank (no data)
15. Confirm Dial In Trunks Answer Point: is blank (no data)
16. **Set Dial In Trunks Insert Forwarding Information**: No
17. **Set Trunk Label**: Set a label for the trunk you are currently creating, “Cisco Cube” is given for this example.
18. Click Save.

SIP Peer Profile

1. Navigate to Trunks > SIP > SIP Peer Profile.
2. Right click on the Trunk Service number and click Add.

![SIP Peer Profile](image)

Figure 7: SIP Peer Profile-1

1. Under Basic Tab:
   a. Set SIP Peer Profile Label: Cube is given for this example.
   b. Set Network element: Cube is selected from the drop down menu.
   c. Set Address Type: IP Address is selected for this example.
   Under Administration Options:
   d. Set Interconnect restriction: 1 is given for this example.
   e. Set SMDR Tag: 0.
   f. Set Trunk Service: 2 is given for this example.
   g. All the other values are set to default values.
2. Under **Call Routing Tab**:

   a. Set Alternation Destination Domain Enable: No
   b. Confirm Alternate Destination Domain IP Address: is disabled
   c. Set Enable special Re-invite Collision Handling: No
   d. Set Only Allow Outgoing Calls: No
   e. Set Private SIP Trunk: No
   f. Set Reject Incoming anonymous Calls: No
   g. Set Route Call Using To Header: No
3. Under **Calling Line ID**:
   a. Confirm **Default CPN**: is blank (no data)
   b. Confirm **Default CPN Name**: is blank (no data)
   c. Set **CPN Restriction**: No
   d. Set **Public Calling Party Number Passthrough**: No
   e. Set **Strip PNI**: No
   f. Set **Use Diverting Party Number as Calling Party Number**: No
   g. Set **Original Calling Party Number if Available**: No

---

**Figure 7: SIP Peer Profile-3**

**Figure 7: SIP Peer Profile-4**
4. Under SDP Options:
   a. Set Allow Peer To Use Multiple Active M-Lines: No
   b. Set Allow Using UPDATE For Early Media Renegotiation: No
   c. Set Avoid Signaling Hold to the Peer: No
   d. Set AVP Only Peer: Yes
   e. Set Enable Mitel Proprietary SDP: No
   f. Set Force sending SDP in initial Invite message: Yes
   g. Set Force sending SDP in initial Invite-Early Answer: No
   h. Set Ignore SDP Answers in provisional Responses: No
   i. Set Limit to one Offer/Answer per INVITE: No
   j. Set NAT Keepalive: No
   k. Set Prevent the Use of IP Address 0.0.0.0 in SDP Messages: Yes
   l. Set Renegotiate SDP To Enforce Symmetric Codec: No
   m. Set Repeat SDP Answer IF Duplicate Offer Is Received: No
   n. Set Restrict Audio Codec: No Restriction is given for this example.
   o. Set RTP Packetization Rate Override: No
   p. Confirm RTP Packetization Rate: is disabled (cannot edit)
   q. Set Special handling of Offers in 2XX responses(INVITE): No
   r. Set Suppress Use of SDP Inactive Media Streams: No
Figure 7: SIP Peer Profile-5

5. Under **Signaling and Header Manipulation**:
   a. Confirm Trunk Group Label: is blank (no data)
   b. Set **allow Display Update**: No
   c. Set **Build Contact Using Request URI Address**: No
   d. Set **De-register Using Contact Address not**: No
   e. Set **Disable Reliable Provisional Responses**: Yes
   f. Set **Disable Use of User-Agent and Server Headers**: No
   g. Set **E.164: Enable sending ‘+’**: No
   h. Confirm E.164: Add ‘+’ if digit length > N digits: is disabled (cannot edit)
   i. Set **E.164 Do not add ‘+’ to Emergency Called Party**: is disabled (cannot edit)
   j. Set **E.164 Do not add ‘+’ to Called Party**: is disabled (cannot edit)
   k. Set **Force Max-Forward: 70 on Outgoing Calls**: No
   l. Set **If TLS use ‘sips:’ Scheme**: No
m. Set Ignore Incoming Loose Routing Indication: No
n. Set Only use SDP to decide 180 or 183: Yes
o. Set prefer From Header from Caller ID: No
p. Set Require Reliable Provisional Responses on Outgoing Calls: NO
q. Set Use Privacy: none: NO
r. Set Use P-Asserted Identity Header: YES
s. Set Use P-Asserted Identity for Billing: No
t. Set Use P-Preferred Identity Header: No
u. Set Use Restricted Character Set for Authentication: No
v. Set Use to Address in From Header on Outgoing Calls: No
w. Set Use user=phone: No
### SIP Peer Profile

<table>
<thead>
<tr>
<th>Category</th>
<th>Options</th>
</tr>
</thead>
<tbody>
<tr>
<td>Trunk Group Label</td>
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</tr>
<tr>
<td>Allow Display Update</td>
<td>No □ Yes □</td>
</tr>
<tr>
<td>Build Contact Using Request URI Address</td>
<td>No □ Yes □</td>
</tr>
<tr>
<td>De-register Using Contact Address not *</td>
<td>No □ Yes □</td>
</tr>
<tr>
<td>Disable Reliable Provisional Responses</td>
<td>No □ Yes □</td>
</tr>
<tr>
<td>Disable Use of User-Agent and Server Headers</td>
<td>No □ Yes □</td>
</tr>
<tr>
<td>E.164: Enable sending ‘*’</td>
<td>No □ Yes □</td>
</tr>
<tr>
<td>E.164: Add ‘*’ if digit length &gt; N digits</td>
<td></td>
</tr>
<tr>
<td>E.164: Do not add ‘*’ to Emergency Called Party</td>
<td>No □ Yes □</td>
</tr>
<tr>
<td>E.164: Do not add ‘*’ to Called Party</td>
<td>No □ Yes □</td>
</tr>
<tr>
<td>Force Max-Forward: 70 on Outgoing Calls</td>
<td>No □ Yes □</td>
</tr>
<tr>
<td>If TLS use ‘sips:’ Scheme</td>
<td>No □ Yes □</td>
</tr>
<tr>
<td>Ignore Incoming Loose Routing Indication</td>
<td>No □ Yes □</td>
</tr>
<tr>
<td>Only use SDP to decide 180 or 183</td>
<td>No □ Yes □</td>
</tr>
<tr>
<td>Prefer From Header for Caller ID</td>
<td>No □ Yes □</td>
</tr>
<tr>
<td>Require Reliable Provisional Responses on Outgoing Calls</td>
<td>No □ Yes □</td>
</tr>
<tr>
<td>Suppress Redirection Headers</td>
<td>No □ Yes □</td>
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<tr>
<td>Use Fixed Retry Time for 491</td>
<td>No □ Yes □</td>
</tr>
<tr>
<td>Use Privacy: none</td>
<td>No □ Yes □</td>
</tr>
<tr>
<td>Use P-Asserted Identity Header</td>
<td>No □ Yes □</td>
</tr>
<tr>
<td>Use P-Asserted Identity for Billing</td>
<td>No □ Yes □</td>
</tr>
<tr>
<td>Use P-Call-Leg-ID Header</td>
<td>No □ Yes □</td>
</tr>
<tr>
<td>Use P-Preferred Identity Header</td>
<td>No □ Yes □</td>
</tr>
<tr>
<td>Use Restricted Character Set For Authentication</td>
<td>No □ Yes □</td>
</tr>
<tr>
<td>Use To Address In From Header on Outgoing Calls</td>
<td>No □ Yes □</td>
</tr>
<tr>
<td>Use user=phone</td>
<td>No □ Yes □</td>
</tr>
</tbody>
</table>

**Figure 7: SIP Peer Profile-6**
6. Under Timers Tab:
   a. Set Keep-Alive(OPTIONS) Period: 120
   b. Set Registration Period: 3600
   c. Set Registration Period Refresh(%): 50
   d. Set Registration Maximum Timeout: 90
   e. Set Session Timer: 90
   f. Set Session Timer: Local as Refresher: No
   g. Set Subscription Period: 3600
   h. Set Subscription Period Minimum: 300
   i. Set Subscription Period Refresh(%): 80
   j. Set Invite Ringing Response Timer: 0

![SIP Peer Profile](image)

Figure 7: SIP Peer Profile-7
7. Under **Key Press Event Tab:**
   a. Set **allow Inc Subscriptions for Local Digit Monitoring:** No
   b. Set **Allow Out Subscriptions for Remote Digit Monitoring:** No
   c. Confirm Force Out Subscriptions for Remote Digit Monitoring: is disabled (cannot edit)
   d. Confirm Request Outbound Proxy to Handle Out Subscriptions: is disabled (cannot edit)
   e. Confirm KPML Transport and Port: are disabled (cannot edit)

![SIP Peer Profile](image)

**Figure 7: SIP Peer Profile-8**
<table>
<thead>
<tr>
<th>Acronym</th>
<th>Definitions</th>
</tr>
</thead>
<tbody>
<tr>
<td>CPE</td>
<td>Customer Premise Equipment</td>
</tr>
<tr>
<td>Cisco UBE</td>
<td>Cisco Unified Border Element</td>
</tr>
<tr>
<td>Cisco UCM</td>
<td>Cisco Unified Communications Manager</td>
</tr>
<tr>
<td>MTP</td>
<td>Media Termination Point</td>
</tr>
<tr>
<td>POP</td>
<td>Point of Presence</td>
</tr>
<tr>
<td>PSTN</td>
<td>Public Switched Telephone Network</td>
</tr>
<tr>
<td>ESBC</td>
<td>Enterprise Session Border Controller</td>
</tr>
<tr>
<td>SCCP</td>
<td>Skinny Client Control Protocol</td>
</tr>
<tr>
<td>SIP</td>
<td>Session Initiation Protocol</td>
</tr>
</tbody>
</table>
Important Information

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