



# 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



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## 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.





## 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.



## Configuration

### Configuring Cisco Unified Border Element

#### Network Interface

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

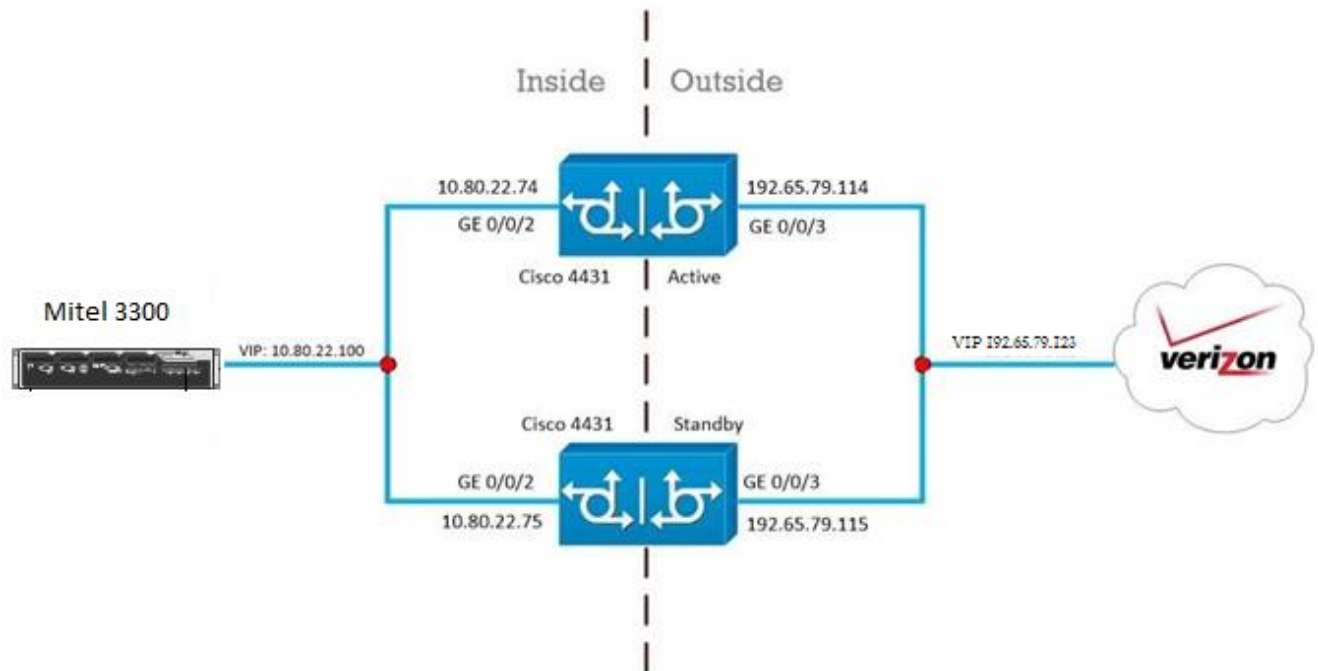


Figure 2 High Availability topology



### 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

!
```





## 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

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

## 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
no vad
```

!



### 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.S1a-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

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

!

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 10.80.22.74 255.255.255.0
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-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
!
```





!

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.S1a-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
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
!
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-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-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 referto-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
exec-timeout 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**.

The screenshot shows the Mitel 3300 PBX configuration interface. At the top, the Mitel logo is on the left, and the alarm status 'Node 'Local\_85' Alarm Status: No Alarm 2014-Nov-20 20:12:02' is on the right. Below the header, the left sidebar shows a navigation tree with 'Voice Network' expanded and 'Network Zones' selected. The main content area is titled 'Network Zones on Local\_85' and contains a table with three columns: 'Zone ID', 'Intra-zone Compression', and 'Intra-zone Fax Profile'. The table lists three zones, all with 'No' for compression and '1' for fax profile. Navigation buttons like 'Change', 'Change Page', and 'Clear' are visible above the table.

Zone ID	Intra-zone Compression	Intra-zone Fax Profile
1	No	1
2	No	1
3	No	1

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



**Change**

**Network Zones**

Zone ID	2
Intra-zone Compression	<input type="radio"/> No <input checked="" type="radio"/> Yes
Group Zone	
Intra-zone Fax Profile	1
Label	Ciscocube
SMDR Tag	
Time Zone	America/Chicago
LBN Prefix	
Zone CESID	
Default Billing Number	
Default CPN	

Save

Cancel

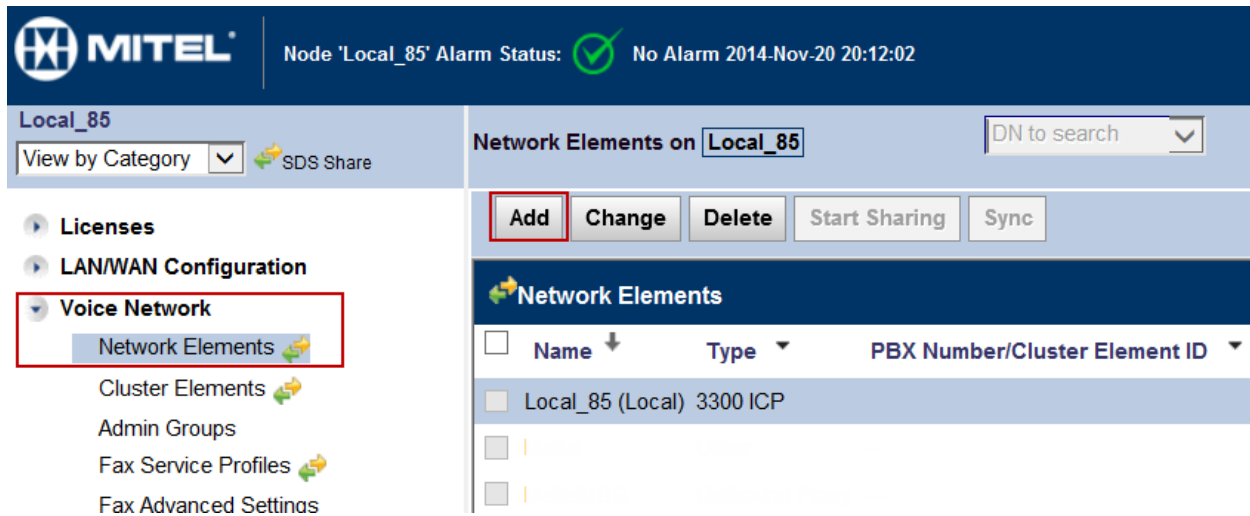
Figure 3 Network Zones -2





## Add Cisco Cube in the Network Elements:

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



**Figure 4 Configure Network elements -1**

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.
9. Set **SIP Peer Port**: 5060.
10. All the other values are set default values.
11. Click **Save**.



**Change**

**Network Elements**

<b>Name</b>	Cube
<b>Type</b>	Other <span>▼</span>
<b>FQDN or IP Address</b>	10.80.22.74 <span>×</span>
<b>Local</b>	False
<b>Version</b>	
<b>Zone</b>	2
<b>ARID</b>	
<b>SIP Peer</b>	<input checked="" type="checkbox"/>
<b>SIP Peer Specific</b>	
<b>SIP Peer Transport</b>	UDP <span>▼</span>
<b>SIP Peer Port</b>	5060
<b>External SIP Proxy FQDN or IP Address</b>	
<b>External SIP Proxy Transport</b>	UDP <span>▼</span>
<b>External SIP Proxy Port</b>	0
<b>SIP Registrar FQDN or IP Address</b>	
<b>SIP Registrar Transport</b>	UDP <span>▼</span>
<b>SIP Registrar Port</b>	0
<b>SIP Peer Status</b>	Auto-Detect/Normal <span>▼</span>

Save Cancel

Figure 4 Configure Network elements -2



## 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**.

The screenshot shows the Mitel System Properties configuration interface. The top header displays the Mitel logo and the status: "Node 'Local\_85' Alarm Status: No Alarm 2014-Nov-20 20:12:02". The left sidebar contains a navigation tree with the following items: Licenses, LAN/WAN Configuration, Voice Network, System Properties (highlighted with a red box), System Settings, System Feature Settings (highlighted with a red box), System Options, Shared System Options, Class of Service Options (highlighted with a red box), and SIP Device Capabilities. The main content area is titled "Class of Service Options on Local\_85" and includes "Change" and "Copy" buttons. Below these buttons is a pagination control showing "Page 1 of 11". The main table lists the "Class Of Service Number" with values 1, 2, 3, and 4. The first row (number 1) is highlighted in blue.

Class Of Service Number
1
2
3
4

Figure 5: Configure COS – 1



General		Advanced	
Class Of Service Number	10		
Comment	Cube		
<b>ACD</b>			
ACD Agent Behavior on No Answer	Logout		
ACD Agent No Answer Timer	15		
ACD Make Busy on Login	<input checked="" type="radio"/> No <input type="radio"/> Yes		
ACD Silent Monitor Accept	<input checked="" type="radio"/> No <input type="radio"/> Yes		
ACD Silent Monitor Accept Monitoring Non-Prime Lines	<input checked="" type="radio"/> No <input type="radio"/> Yes		
ACD Silent Monitor Allowed	<input checked="" type="radio"/> No <input type="radio"/> Yes		
ACD Silent Monitor Notification	<input checked="" type="radio"/> No <input type="radio"/> Yes		
Follow 2nd Alternate Reroute for Recall to Busy ACD Agent	<input checked="" type="radio"/> No <input type="radio"/> Yes		
Work Timer	0		
<b>Announce</b>			
Call Announce Line	<input checked="" type="radio"/> No <input type="radio"/> Yes		
Off-Hook Voice Announce Allowed	<input checked="" type="radio"/> No <input type="radio"/> Yes		
Handsfree AnswerBack Allowed	<input checked="" type="radio"/> No <input type="radio"/> Yes		
<b>Busy Override</b>			
Busy Override Security	<input type="radio"/> No <input checked="" type="radio"/> Yes		
Disable Executive Busy Override Tone	<input checked="" type="radio"/> No <input type="radio"/> Yes		
Executive Busy Override	<input checked="" type="radio"/> No <input type="radio"/> Yes		
<b>Call Control Timer</b>			
Busy Tone Timer	30		
Dialing Conflict Timer	3		
First Digit Timer	15		
Inter Digit Timer	10		
Lockout Timer	45		

Figure 5: Configure COS – 2



## Call Duration

Call Duration

10

Call Duration Forced Cleardown Timer

0

Enable Call Duration Limit on External Calls

☒ No ☐ Yes

Enable Call Duration Limit on Internal Calls

☒ No ☐ Yes

## Call Forwarding/Rerouting

Call Forward - Delay

0

Call Forward No Answer Timer

15

Call Forward Override

☐ No ☒ Yes

Call Forwarding (External Destination)

☐ No ☒ Yes

Call Forwarding (Internal Destination)

☐ No ☒ Yes

Call Forwarding Accept

☐ No ☒ Yes

Call Reroute after CFFM to Busy Destination

☐ No ☒ Yes

Call Forwarding Reminder Ring (CFFM and CFIAH only)

☒ No ☐ Yes

Disable Call Reroute Chaining On Diversion

☒ No ☐ Yes

Group Call Forward Follow Me Accept

☒ No ☐ Yes

Group Call Forward Follow Me Allow

☒ No ☐ Yes

Third Party Call Forward Follow Me Accept

☒ No ☐ Yes

Third Party Call Forward Follow Me Allow

☒ No ☐ Yes

Use Held Party Device for Call Re-routing

☐ No ☒ Yes

## Call Hold

Call Hold

☐ No ☒ Yes

Call Hold - Retrieve with Hold Key

☐ No ☒ Yes

Call Hold Remote Retrieve

☐ No ☒ Yes

Call Hold Timer

30

Local Music On Hold source

☐ No ☒ Yes

Music on Hold on Transfer

☐ No ☒ Yes

Use Called Party Call Hold Timer

☒ No ☐ Yes

Figure 5: Configure COS – 3



### Call Park

Call Park Timer

180

Call Park-Allowed To Park

☐ No ☒ Yes

### Call Pickup

Allow Directed Call Pickup Of Attendant Call

☒ No ☐ Yes

Call Pickup Dialed Accept

☐ No ☒ Yes

Call Pickup Directed Accept

☐ No ☒ Yes

### Call Privacy

Call Privacy

☐ No ☒ Yes

Calling Party Name Substitution

☒ No ☐ Yes

Name Suppression on outgoing Trunk Call

☒ No ☐ Yes

Privacy Released

☒ No ☐ Yes

Public Network Identity Provided

☐ No ☒ Yes

### Call Waiting

Call Waiting Swap

☒ No ☐ Yes

ONS CLASS/CLIP: Visual Call Waiting

☒ No ☐ Yes

### Campon

Auto Campon Timer

Campon Recall Timer

10

### Direct Voice Call

Direct Voice Call - Accept

☐ No ☒ Yes

Direct Voice Call - Allow

☐ No ☒ Yes

Direct Voice Call - Maximize Volume

☒ No ☐ Yes

Figure 5: Configure COS – 4



## Display

After Answer Display Time

Calling Name Display - Internal - ONS

Calling Number Display - Internal - ONS

Display ANI/DNIS/ISDN Calling/Called Number

Display ANI/ISDN Calling Number Only

Display Caller ID on multicall/keylines

Display Caller ID On Multicall/Keylines Timer

Display Caller ID On Single Line Displays For Forwarded Calls

Display Dialed Digits during Outgoing Calls

Display DNIS/Called Number Before Digit Modification

Display Held Call ID on Transfer

Display Transfer Destination on Recall

Hot Desk External User - Display Internal Calling ID

Maintain Ringing Party During Recall

Non-Prime Public Network Identity

Originator's Display Update In Call Forwarding/Rerouting

Suppress Delivery of Caller ID Display between Sets

Suppress Delivery of Caller ID Display between Sets -  
Override

Suppress Display Of Account Code Numbers

Suppress Redial Display

☐ No ☒ Yes

☐ No ☒ Yes

☒ No ☐ Yes

☒ No ☐ Yes

☒ No ☐ Yes

☒ No ☐ Yes

☒ No ☐ Yes

☒ No ☐ Yes

☒ No ☐ Yes

☒ No ☐ Yes

☒ No ☐ Yes

☒ No ☐ Yes

☒ No ☐ Yes

☒ No ☐ Yes

☒ No ☐ Yes

☒ No ☐ Yes

☒ No ☐ Yes

☒ No ☐ Yes

## Fax

Campon Tone Security

External Trunk Standard Ringback

Fax Capable

Return Disconnect Tone When Far End Party Clears

☒ No ☐ Yes

☒ No ☐ Yes

☒ No ☐ Yes

☒ No ☐ Yes

Figure 5: Configure COS – 5



## HCI

HCI/CTI/TAPI Call Control Allowed

☐ No ☒ Yes

HCI/CTI/TAPI Monitor Allowed

☐ No ☒ Yes

## Hot Desk

Green BLF Lamp for Logged in Hotdesk User

☒ No ☐ Yes

Hot Desk External User - Allow Mid-Call Features

☒ No ☐ Yes

Hot Desk External User - Answer Confirmation

☒ No ☐ Yes

Hot Desk External User - Dial Tone on Call Complete

☒ No ☐ Yes

Hot Desk External User - Permanent Login

☒ No ☐ Yes

Hot Desk External User - Remote MWI Enable Feature Access Code

Hot Desk External User - Remote MWI Disable Feature Access Code

Hot Desk Login Accept

☒ No ☐ Yes

Hot Desk Remote Logout Enabled

☒ No ☐ Yes

## Miscellaneous

Backlighting - Enabled

☐ No ☒ Yes

Clear All Features Remote

☒ No ☐ Yes

Force Device Busy If Any Line In Use

☒ No ☐ Yes

Handset Volume Adjustment Saved

☒ No ☐ Yes

Head Set Switch Mute

☒ No ☐ Yes

Multi-Color LED Support - Disable

☒ No ☐ Yes

Phone Lock

☒ No ☐ Yes

Reseize Timer

Timed Reminder Allowed

☐ No ☒ Yes

User Inactivity Timer

Figure 5: Configure COS – 6





## Paging

Group Page Accept

☒ No ☐ Yes

Group Page Allow

☒ No ☐ Yes

Loudspeaker Pager Equivalent Zone Override Security

☒ No ☐ Yes

Loudspeaker Pager Override

☐ No ☒ Yes

Pager Access All Zones

☐ No ☒ Yes

Pager Access Individual Zones

☒ No ☐ Yes

## PC Port

PC Port On IP Device - Disable

☒ No ☐ Yes

## RAD

Answer Plus Delay To Message Timer

5

Answer Plus Expected Off-hook Timer

30

Answer Plus Message Length Timer

10

Answer Plus System Reroute Timer

0

Recorded Announcement Device

☒ No ☐ Yes

Recorded Announcement Device - Advanced

☒ No ☐ Yes

## Ringing

Delay Ring Timer

10

No Answer Recall Timer

17

Ringing Line Select

☒ No ☐ Yes

Ringing Timer

180

## SMDR

SMDR External

☒ No ☐ Yes

SMDR Internal

☒ No ☐ Yes

Figure 5: Configure COS – 7



## Trunk

ANI/DNIS/ISDN Number Delivery Trunk

☐ No ☒ Yes

DASS II OLI/TLI Provided

☐ No ☒ Yes

Public Network Access via DPNSS

☐ No ☒ Yes

Public Network To Public Network Connection Allowed

☐ No ☒ Yes

Public Trunk

☐ No ☒ Yes

R2 Call Progress Tone

☐ No ☒ Yes

Suppress Simulated CCM after ISDN Progress

☒ No ☐ Yes

Trunk Calling Party Identification

☐ No ☒ Yes

Trunk Flash Allowed

☐ No ☒ Yes

Two B-Channel Transfer Allowed

☐ No ☒ Yes

## Voice Mail

COV/ONS/E&M Voice Mail Port

☒ No ☐ Yes

ONS VMail-Delay Dial Tone Timer

5

Figure 5: Configure COS – 8



## Trunk Attributes

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

MITEL

Node 'Local\_85' Alarm Status: No Alarm 2014-Nov-20 20:12:02

Local\_85

View by Category SDS Share

Trunk Attributes on Local\_85

**Change** **Change Page** **Change All** **Clear**

< Page 1 of 15 >

**Trunk Attributes**

Trunk Service Number	Release Link Trunk	Call Recognition Service	Direct Inward Dialing Service
1	No	Off	On
2	No	Off	On
3	No	Off	Off

Figure 6: Trunk Attributes

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**

MITEL

Node 'Local\_85' Alarm Status: No Alarm 2014-Nov-20 20:12:02

Local\_85

View by Category SDS Share

SIP Peer Profile on Local\_85

**Trunks**

- Trunk Attributes
- DTS Service Profiles
- Analog
- Digital
- IP/XNET
- SIP**
- DID Ranges for CPN Substitution
- SIP Peer Profile**
- SIP Peer Profile Assignment by

**Add** **Change** **Delete**

**SIP Peer Profile**

Network Element	SIP Peer Profile Label	Outbound Proxy Server	CPN Restriction	Trunk Service	Session Timer	Zone
-----------------	------------------------	-----------------------	-----------------	---------------	---------------	------

**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.



Basic	Call Routing	Calling Line ID	SDP Options	Signaling
Outgoing DID Ranges		Profile Information		
SIP Peer Profile Label		Cube		
Network Element		Cube		
<b>Local Account Information</b>				
Registration User Name				
Address Type		IP Address: 10.35.31.85		
<b>Administration Options</b>				
Interconnect Restriction		1		
Maximum Simultaneous Calls		10		
Minimum Reserved Call Licenses		0		
<b>Administration Options</b>				
Outbound Proxy Server				
SMDR Tag		0		
Trunk Service		2		
Zone		2		
User Name				
Password		*****		
Confirm Password		*****		
Authentication Option for Incoming Calls		No Authentication		
Subscription User Name				
Subscription Password		*****		
Subscription Confirm Password		*****		

Figure 7: SIP Peer Profile-2

2. Under **Call Routing Tab**:

- Set **Alternation Destination Domain Enable**: No
- Confirm Alternate Destination Domain IP Address: is disabled
- Set **Enable special Re-invite Collision Handling**: No
- Set **Only Allow Outgoing Calls**: No
- Set **Private SIP Trunk**: No
- Set **Reject Incoming anonymous Calls**: No
- Set **Route Call Using To Header**: No



Basic	Call Routing	Calling Line ID	SDP Options	Signaling a
Profile Information				
Alternate Destination Domain Enabled		<input checked="" type="radio"/> No <input type="radio"/> Yes		
Alternate Destination Domain FQDN or IP Address		<input type="text"/>		
Enable Special Re-invite Collision Handling		<input checked="" type="radio"/> No <input type="radio"/> Yes		
Only Allow Outgoing Calls		<input checked="" type="radio"/> No <input type="radio"/> Yes		
Private SIP Trunk		<input checked="" type="radio"/> No <input type="radio"/> Yes		
Reject Incoming Anonymous Calls		<input checked="" type="radio"/> No <input type="radio"/> Yes		
Route Call Using P-Called-Party-ID (if present)		<input checked="" type="radio"/> No <input type="radio"/> Yes		
Route Call Using To Header		<input checked="" type="radio"/> No <input type="radio"/> Yes		

Figure 7: SIP Peer Profile-3

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

SIP Peer Profile				
Basic	Call Routing	Calling Line ID	SDP Options	Signaling and T
Profile Information				
Default CPN		<input type="text"/>		
Default CPN Name		<input type="text"/>		
CPN Restriction		<input checked="" type="radio"/> No <input type="radio"/> Yes		
Public Calling Party Number Passthrough		<input checked="" type="radio"/> No <input type="radio"/> Yes		
Strip PNI		<input checked="" type="radio"/> No <input type="radio"/> Yes		
Use Diverting Party Number as Calling Party Number		<input checked="" type="radio"/> No <input type="radio"/> Yes		
Use Original Calling Party Number If Available		<input checked="" type="radio"/> No <input type="radio"/> Yes		

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



Basic	Call Routing	Calling Line ID	SDP Options	Signaling and Header Manipulation
Profile Information				
Allow Peer To Use Multiple Active M-Lines				<input checked="" type="radio"/> No <input type="radio"/> Yes
Allow Using UPDATE For Early Media Renegotiation				<input checked="" type="radio"/> No <input type="radio"/> Yes
Avoid Signaling Hold to the Peer				<input checked="" type="radio"/> No <input type="radio"/> Yes
AVP Only Peer				<input type="radio"/> No <input checked="" type="radio"/> Yes
Enable Mitel Proprietary SDP				<input checked="" type="radio"/> No <input type="radio"/> Yes
Force sending SDP in initial Invite message				<input type="radio"/> No <input checked="" type="radio"/> Yes
Force sending SDP in initial Invite - Early Answer				<input checked="" type="radio"/> No <input type="radio"/> Yes
Ignore SDP Answers in Provisional Responses				<input checked="" type="radio"/> No <input type="radio"/> Yes
Limit to one Offer/Answer per INVITE				<input checked="" type="radio"/> No <input type="radio"/> Yes
NAT Keepalive				<input checked="" type="radio"/> No <input type="radio"/> Yes
Prevent the Use of IP Address 0.0.0.0 in SDP Messages				<input type="radio"/> No <input checked="" type="radio"/> Yes
Renegotiate SDP To Enforce Symmetric Codec				<input checked="" type="radio"/> No <input type="radio"/> Yes
Repeat SDP Answer If Duplicate Offer Is Received				<input checked="" type="radio"/> No <input type="radio"/> Yes
Restrict Audio Codec				No Restriction <input type="button" value="v"/>
RTP Packetization Rate Override				<input checked="" type="radio"/> No <input type="radio"/> Yes
RTP Packetization Rate				20ms <input type="button" value="v"/>
Special handling of Offers in 2XX responses (INVITE)				<input checked="" type="radio"/> No <input type="radio"/> Yes
Suppress Use of SDP Inactive Media Streams				<input checked="" type="radio"/> No <input type="radio"/> Yes

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	
Basic	Call Routing
Calling Line ID	SDP Options
Signaling and Header Manipulation	
Profile Information	
Trunk Group Label	<input type="text"/>
Allow Display Update	<input checked="" type="radio"/> No <input type="radio"/> Yes
Build Contact Using Request URI Address	<input checked="" type="radio"/> No <input type="radio"/> Yes
De-register Using Contact Address not *	<input checked="" type="radio"/> No <input type="radio"/> Yes
Disable Reliable Provisional Responses	<input type="radio"/> No <input checked="" type="radio"/> Yes
Disable Use of User-Agent and Server Headers	<input checked="" type="radio"/> No <input type="radio"/> Yes
E.164: Enable sending '+'	<input checked="" type="radio"/> No <input type="radio"/> Yes
E.164: Add '+' if digit length > N digits	<input type="text" value="0"/>
E.164: Do not add '+' to Emergency Called Party	<input checked="" type="radio"/> No <input type="radio"/> Yes
E.164: Do not add '+' to Called Party	<input checked="" type="radio"/> No <input type="radio"/> Yes
Force Max-Forward: 70 on Outgoing Calls	<input checked="" type="radio"/> No <input type="radio"/> Yes
If TLS use 'sips:' Scheme	<input checked="" type="radio"/> No <input type="radio"/> Yes
Ignore Incoming Loose Routing Indication	<input checked="" type="radio"/> No <input type="radio"/> Yes
Only use SDP to decide 180 or 183	<input type="radio"/> No <input checked="" type="radio"/> Yes
Prefer From Header for Caller ID	<input checked="" type="radio"/> No <input type="radio"/> Yes
Require Reliable Provisional Responses on Outgoing Calls	<input checked="" type="radio"/> No <input type="radio"/> Yes
Suppress Redirection Headers	<input checked="" type="radio"/> No <input type="radio"/> Yes
Use Fixed Retry Time for 491	<input checked="" type="radio"/> No <input type="radio"/> Yes
Use Privacy: none	<input checked="" type="radio"/> No <input type="radio"/> Yes
Use P-Asserted Identity Header	<input type="radio"/> No <input checked="" type="radio"/> Yes
Use P-Asserted Identity for Billing	<input checked="" type="radio"/> No <input type="radio"/> Yes
Use P-Call-Leg-ID Header	<input checked="" type="radio"/> No <input type="radio"/> Yes
Use P-Preferred Identity Header	<input checked="" type="radio"/> No <input type="radio"/> Yes
Use Restricted Character Set For Authentication	<input checked="" type="radio"/> No <input type="radio"/> Yes
Use To Address in From Header on Outgoing Calls	<input checked="" type="radio"/> No <input type="radio"/> Yes
Use user=phone	<input checked="" type="radio"/> No <input type="radio"/> Yes

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			
Basic	Call Routing	Calling Line ID	SDP Options
Timers	Key Press Event	Profile Information	
Keep-Alive (OPTIONS) Period	120		
Registration Period	3600		
Registration Period Refresh (%)	50		
Registration Maximum Timeout	90		
Session Timer	90		
Session Timer: Local as Refresher	<input checked="" type="radio"/> No <input type="radio"/> Yes		
Subscription Period	3600		
Subscription Period Minimum	300		
Subscription Period Refresh (%)	80		
Invite Ringing Response Timer	0		

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	
Basic	Call Routing
Calling Line ID	SDP Options
Signaling and Header Manipulation	
Timers	Key Press Event
Profile Information	
Allow Inc Subscriptions for Local Digit Monitoring	<input checked="" type="radio"/> No <input type="radio"/> Yes
Allow Out Subscriptions for Remote Digit Monitoring	<input checked="" type="radio"/> No <input type="radio"/> Yes
Force Out Subscriptions for Remote Digit Monitoring	<input checked="" type="radio"/> No <input type="radio"/> Yes
Request Outbound Proxy to Handle Out Subscriptions	<input checked="" type="radio"/> No <input type="radio"/> Yes
KPML Transport	default
KPML Port	0

Figure 7: SIP Peer Profile-8



## Acronyms

Acronym	Definitions
CPE	Customer Premise Equipment
Cisco UBE	Cisco Unified Border Element
Cisco UCM	Cisco Unified Communications Manager
MTP	Media Termination Point
POP	Point of Presence
PSTN	Public Switched Telephone Network
ESBC	Enterprise Session Border Controller
SCCP	Skinny Client Control Protocol
SIP	Session Initiation Protocol



## Important Information

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