Application Note

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

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.

Network Topology

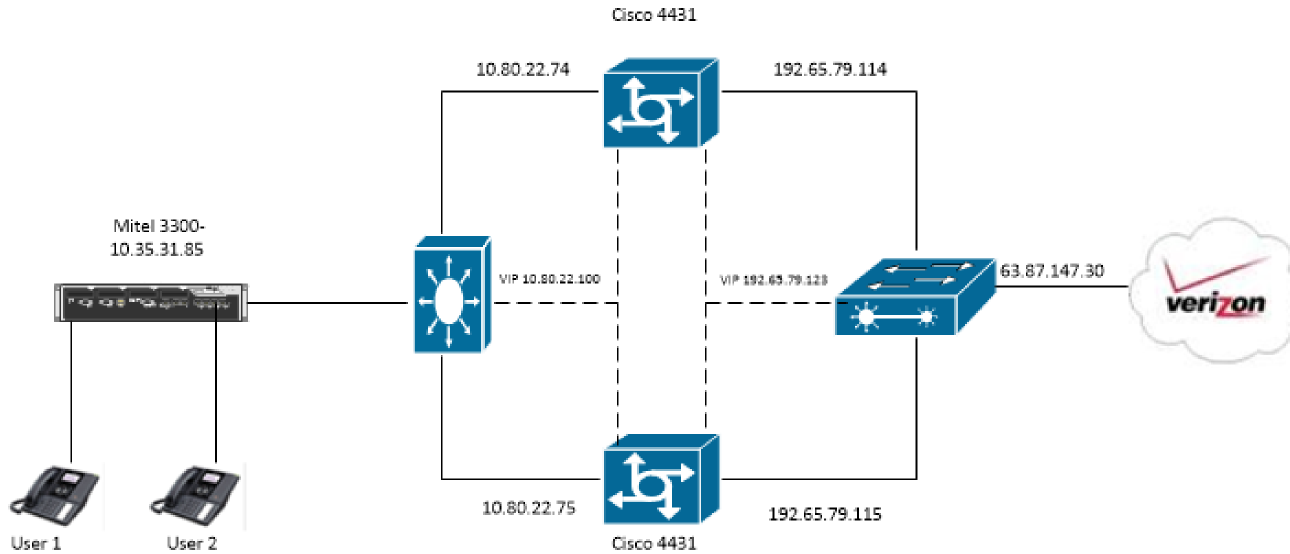


Figure 1 Network Topology

- 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:

Setting	Value
REFER Support	YES
Session Timer	ON
Early Media support with PRACK	NO

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.

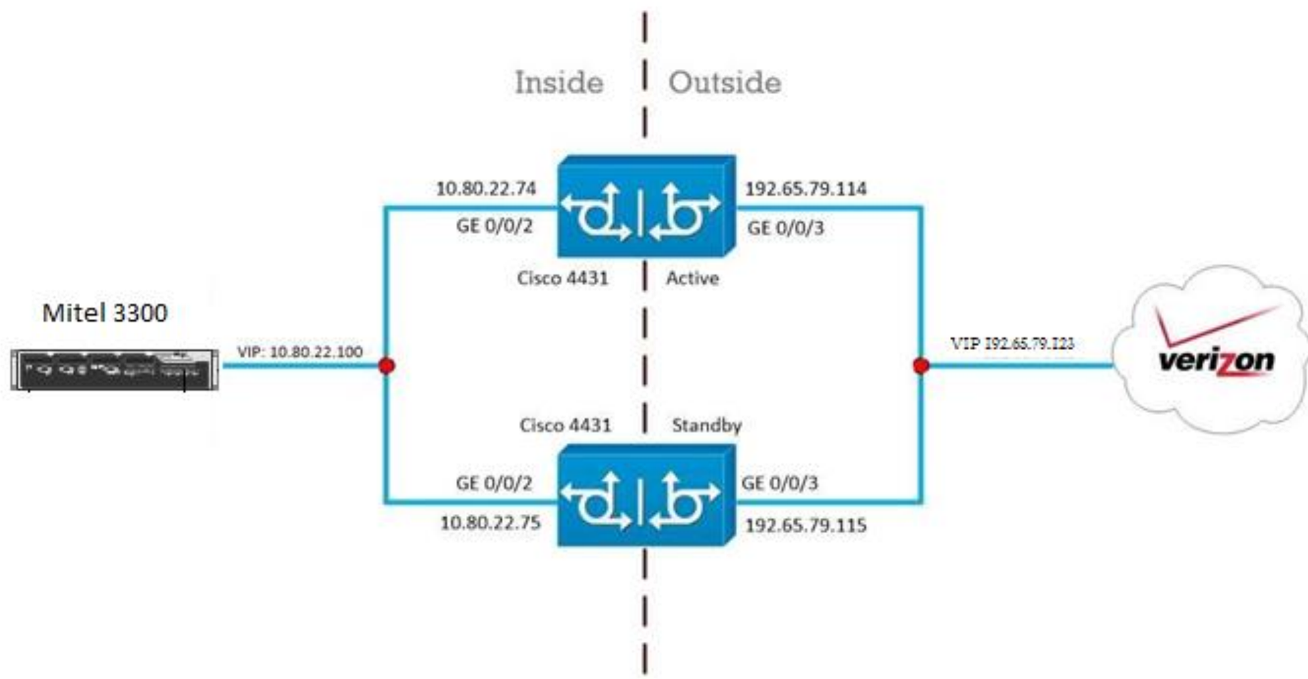


Figure 2 High Availability topology

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
```

Network Zones

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

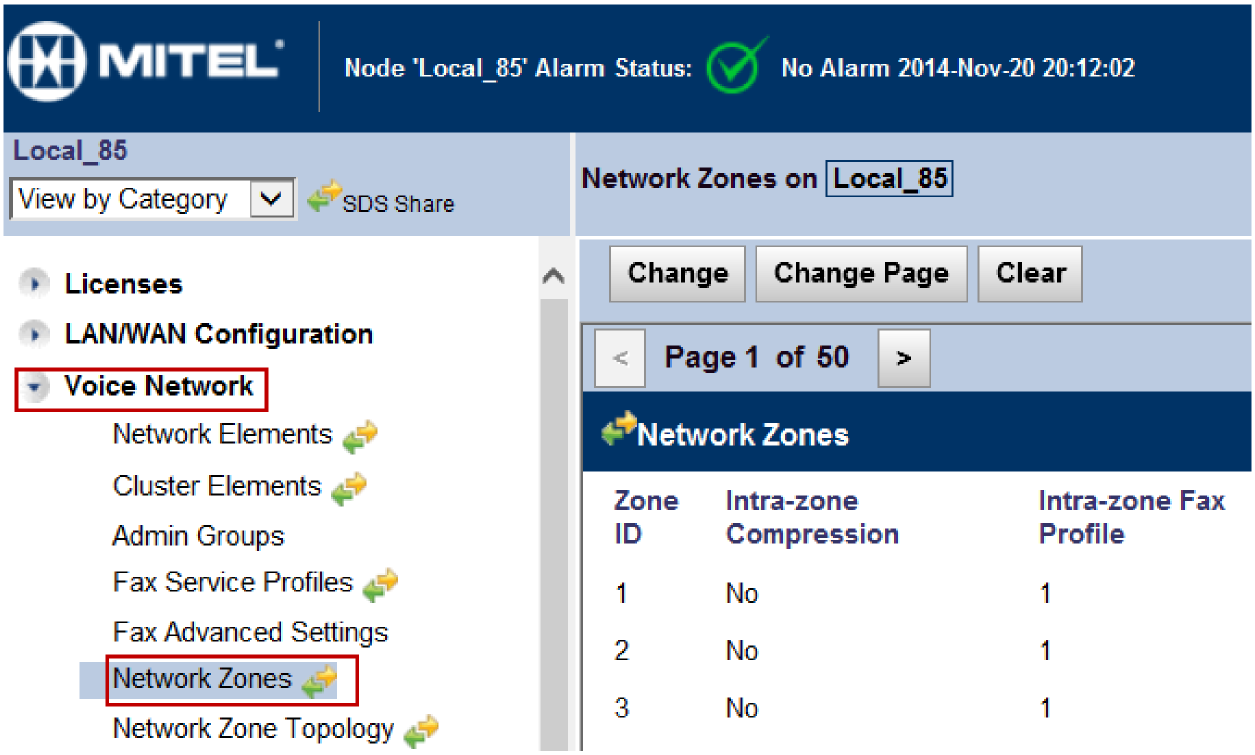


Figure 3 Network Zones -1

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

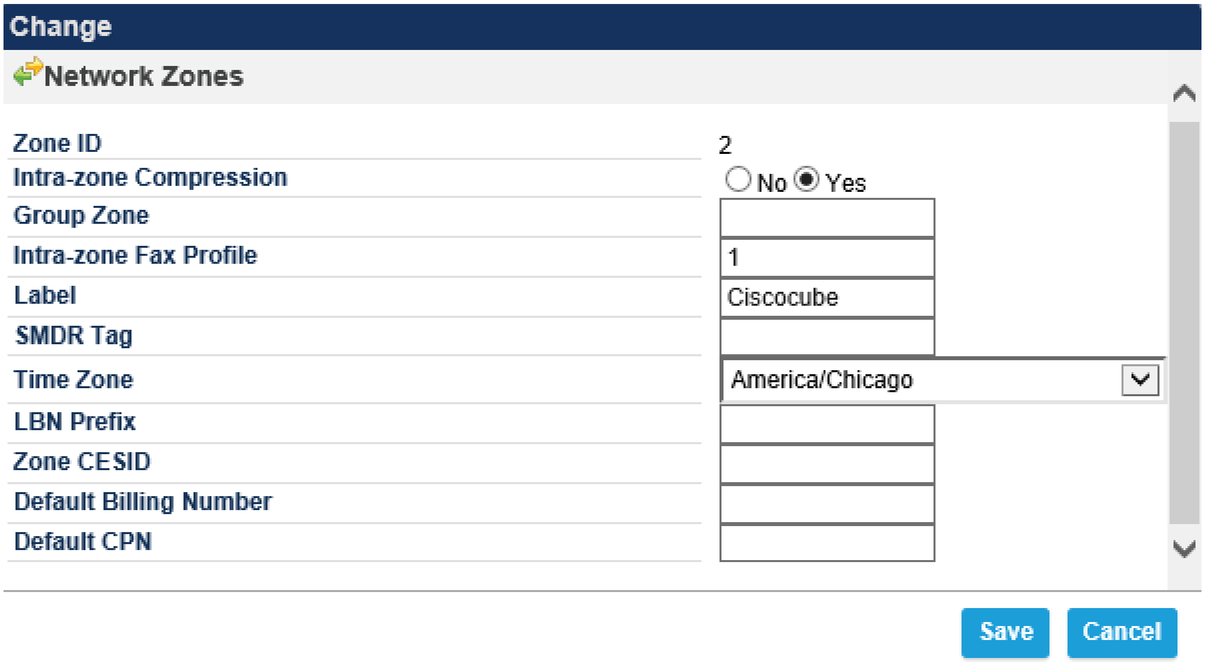


Figure 3 Network Zones -2

Add Cisco Cube in the Network Elements:

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

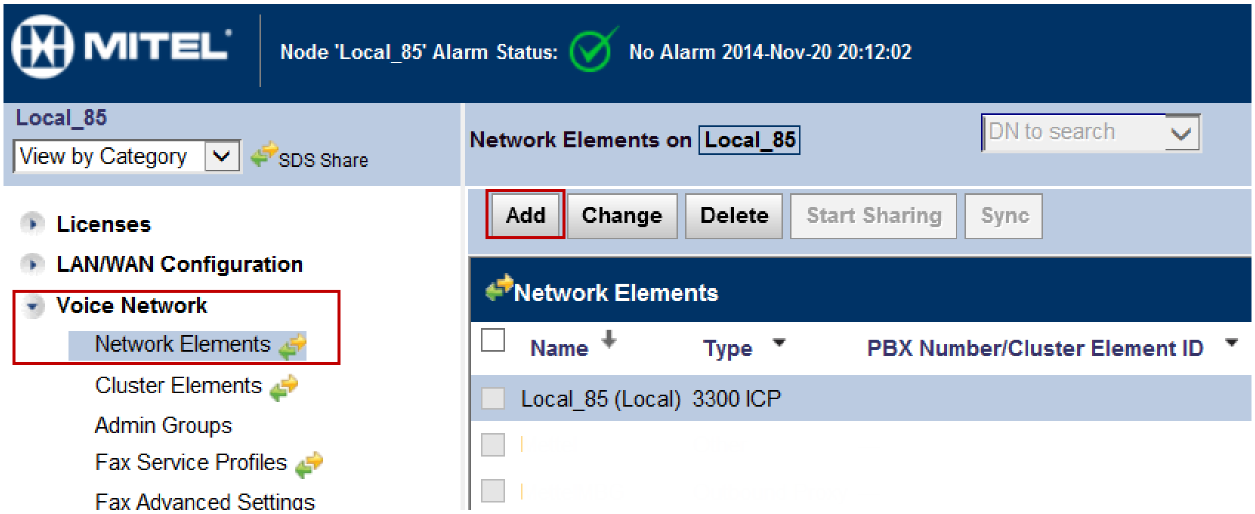


Figure 4 Configure Network elements -1

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**.

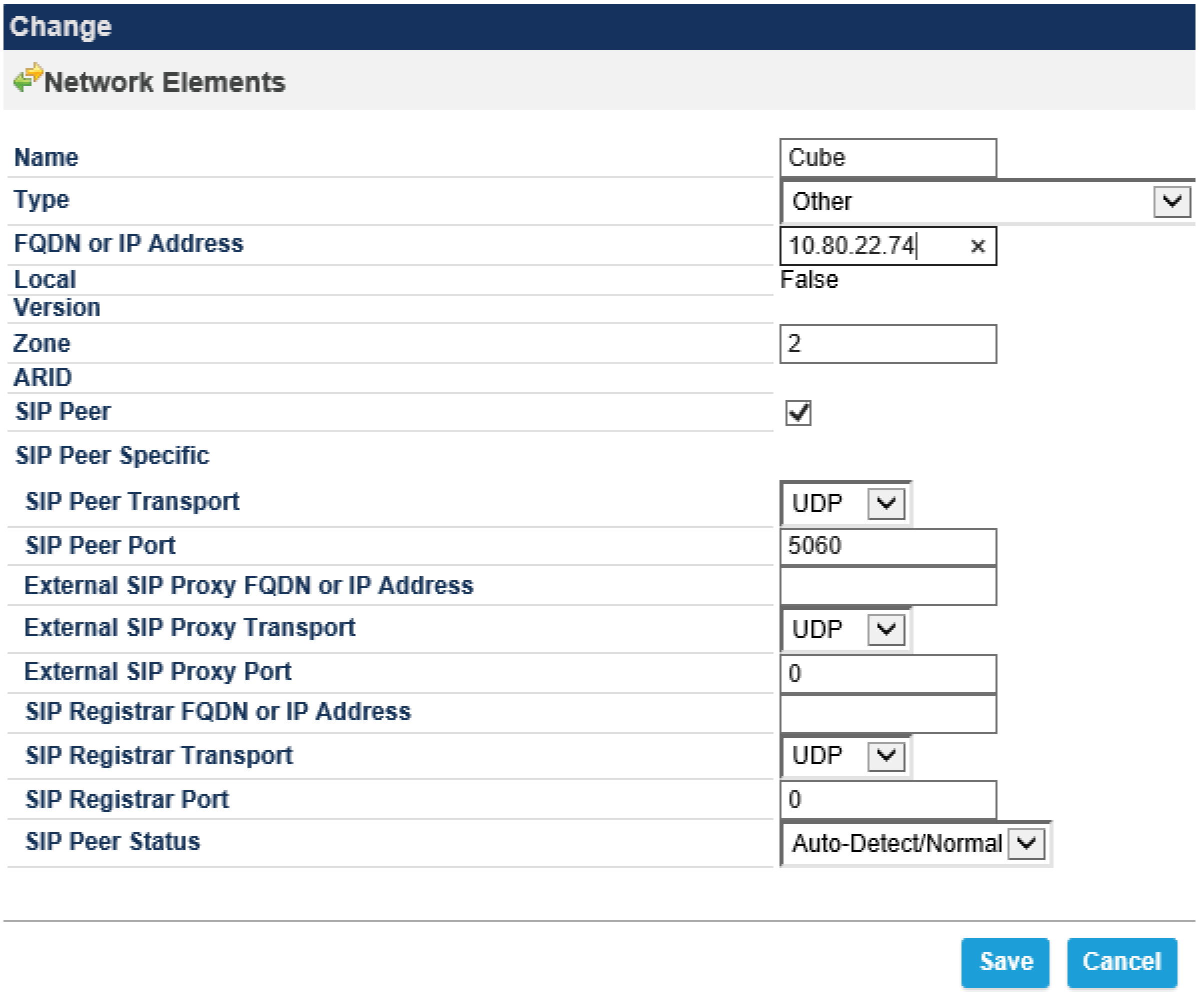


Figure 4 Configure Network elements -2

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**.

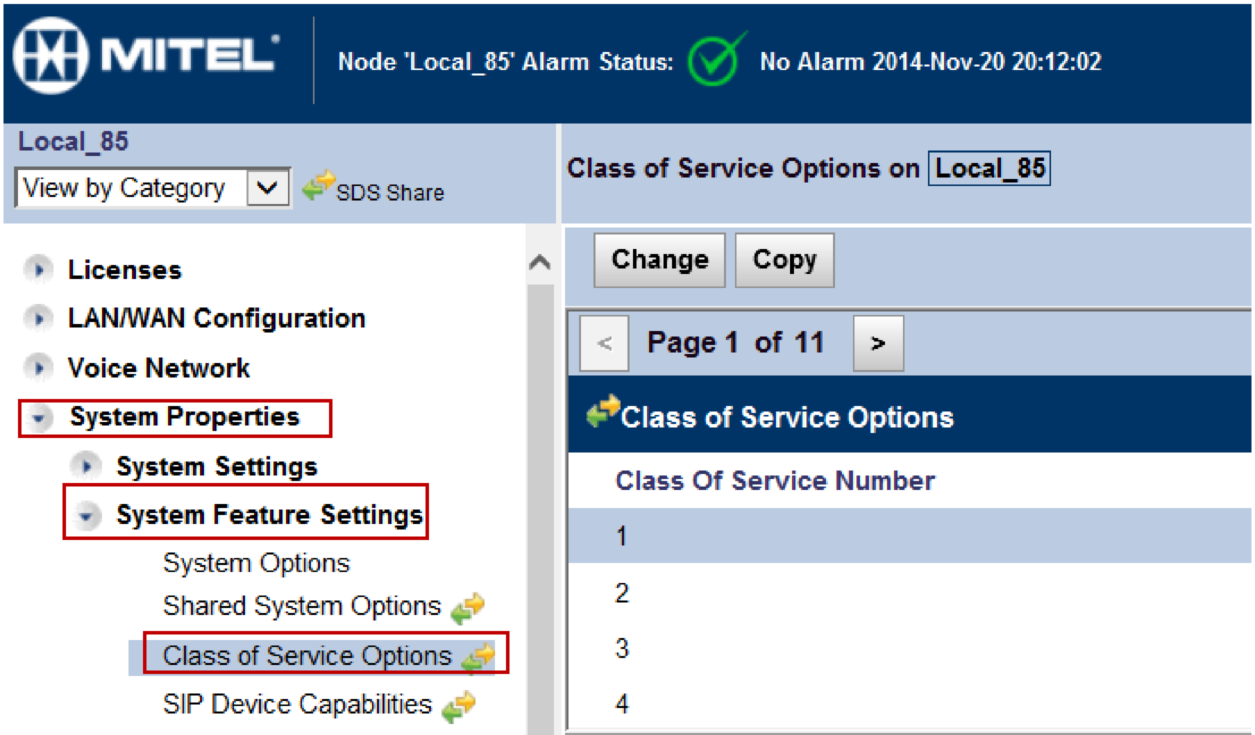


Figure 5: Configure COS – 1

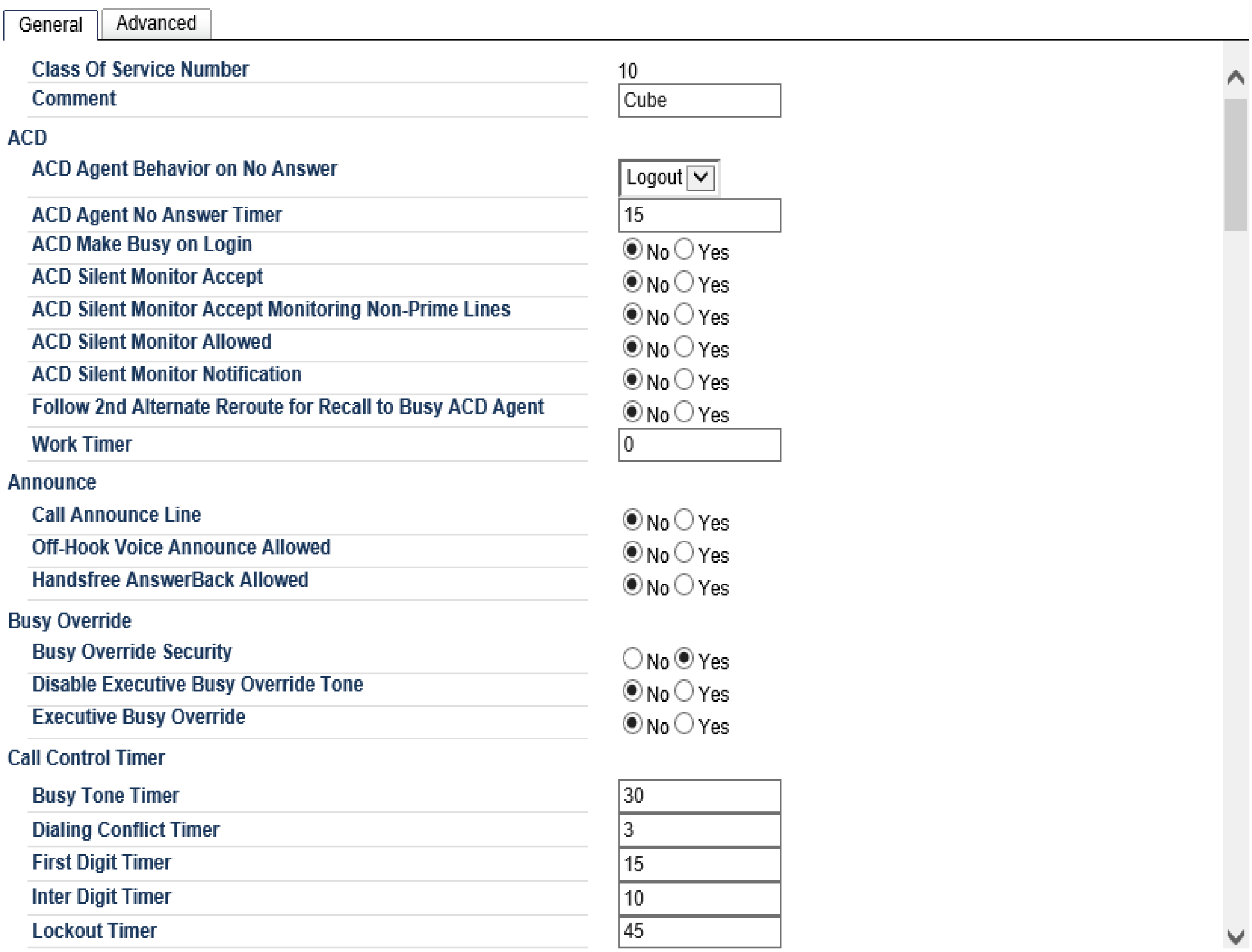


Figure 5: Configure COS – 2

Call Duration

Setting	Value
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

Setting	Value
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

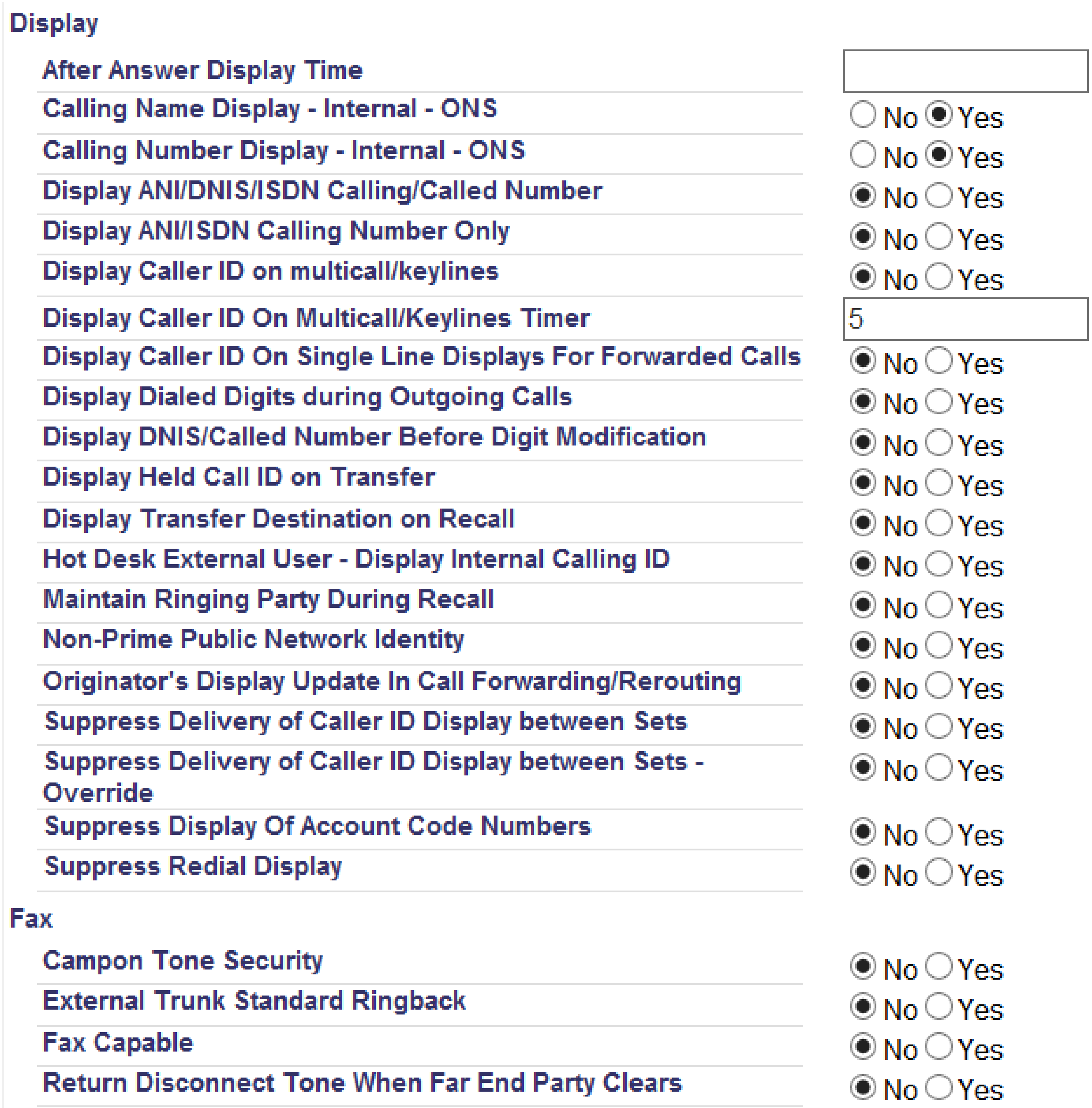
Call Hold

Setting	Value
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

Setting	Value
After Answer Display Time	
Calling Name Display - Internal - ONS	○ No ◉ Yes
Calling Number Display - Internal - ONS	○ No ◉ Yes
Display ANI/DNIS/ISDN Calling/Called Number	◉ No ○ Yes
Display ANI/ISDN Calling Number Only	◉ No ○ Yes
Display Caller ID on multicall/keylines	◉ No ○ Yes
Display Caller ID On Multicall/Keylines Timer	5
Display Caller ID On Single Line Displays For Forwarded Calls	◉ No ○ Yes
Display Dialed Digits during Outgoing Calls	◉ No ○ Yes
Display DNIS/Called Number Before Digit Modification	◉ No ○ Yes
Display Held Call ID on Transfer	◉ No ○ Yes
Display Transfer Destination on Recall	◉ No ○ Yes
Hot Desk External User - Display Internal Calling ID	◉ No ○ Yes
Maintain Ringing Party During Recall	◉ No ○ Yes
Non-Prime Public Network Identity	◉ No ○ Yes
Originator's Display Update In Call Forwarding/Rerouting	◉ No ○ Yes
Suppress Delivery of Caller ID Display between Sets	◉ No ○ Yes
Suppress Delivery of Caller ID Display between Sets - Override	◉ No ○ Yes
Suppress Display Of Account Code Numbers	◉ No ○ Yes
Suppress Redial Display	◉ No ○ Yes

Fax

Setting	Value
Campon Tone Security	◉ No ○ Yes
External Trunk Standard Ringback	◉ No ○ Yes
Fax Capable	◉ No ○ Yes
Return Disconnect Tone When Far End Party Clears	◉ No ○ Yes

Figure 5: Configure COS – 5

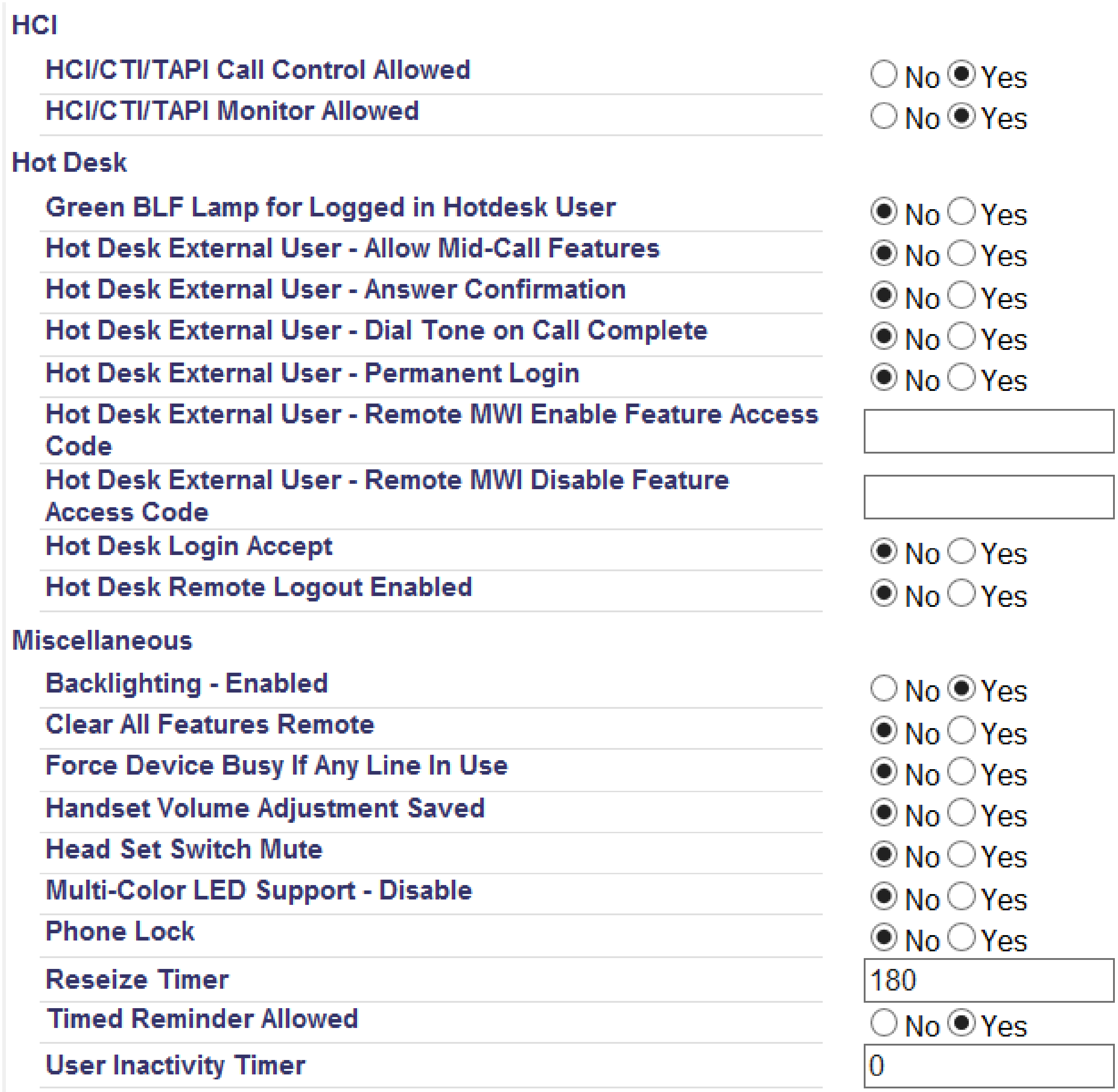


Figure 5: Configure COS – 6

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**

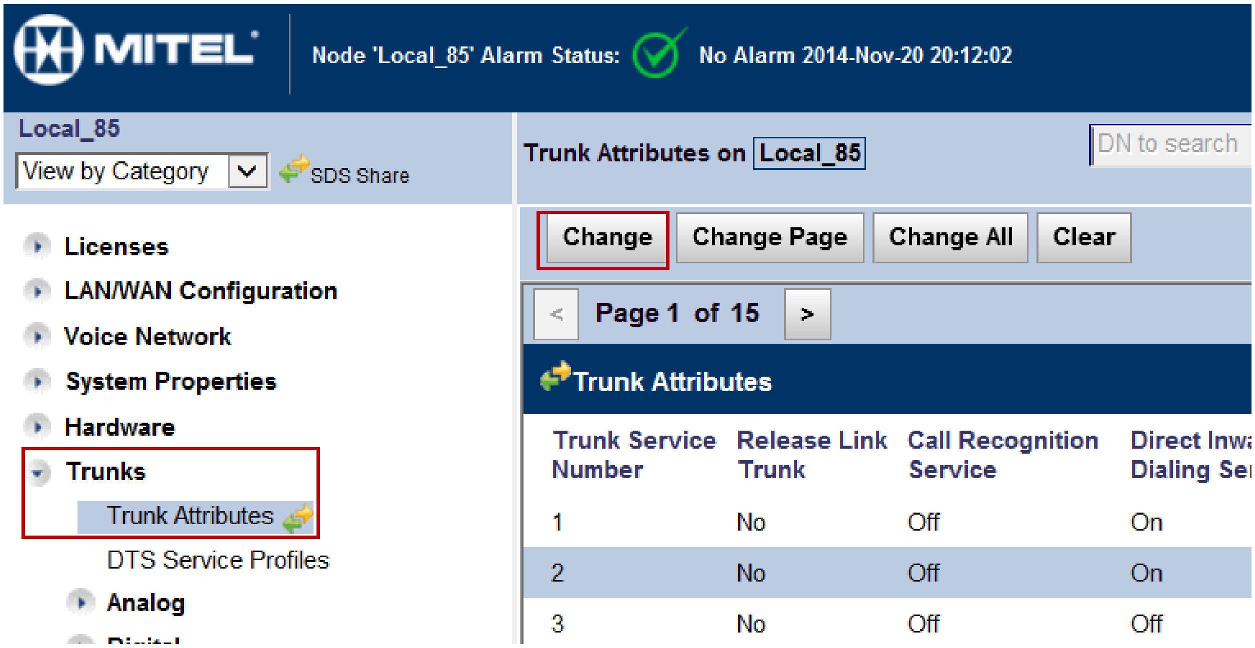


Figure 6: Trunk Attributes

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**

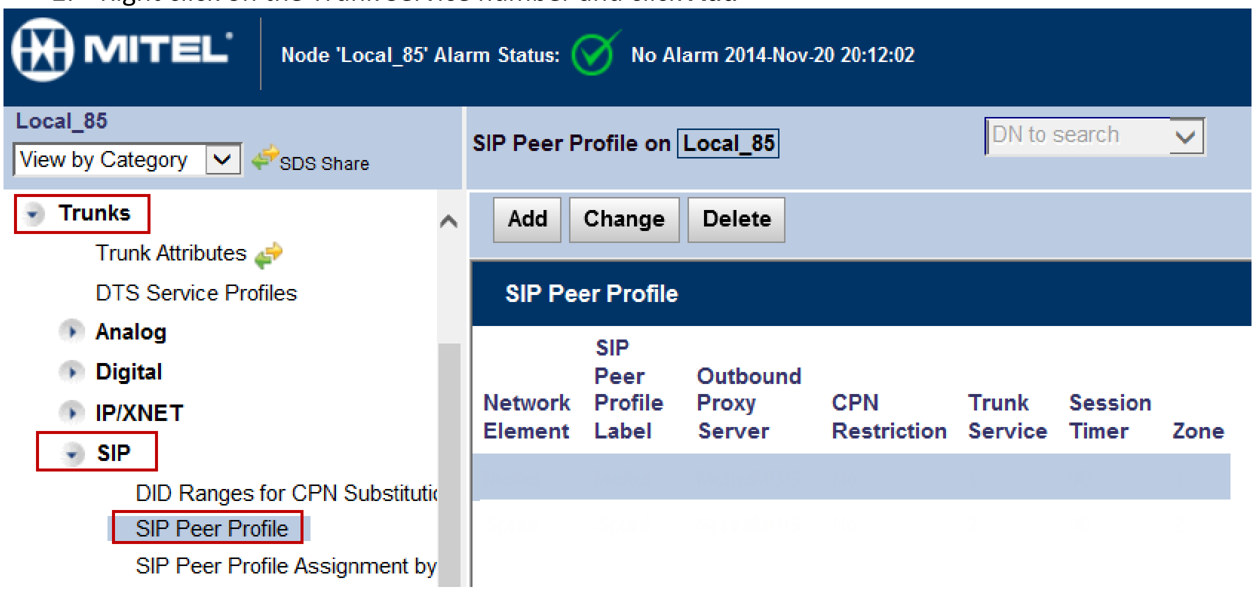


Figure 7: SIP Peer Profile-1

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.

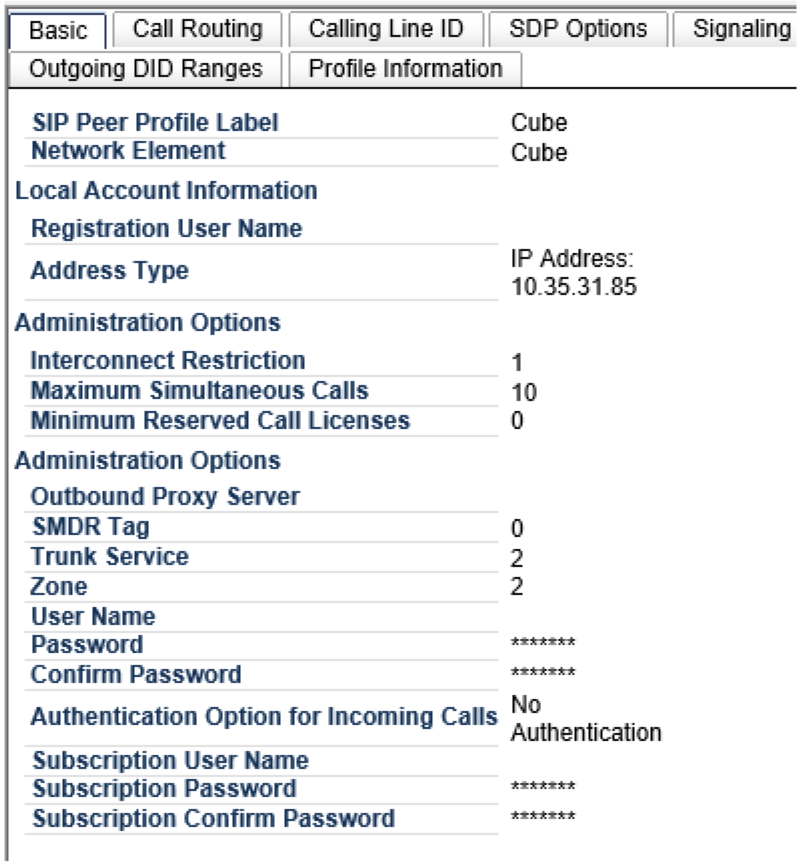


Figure 7: SIP Peer Profile-2

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

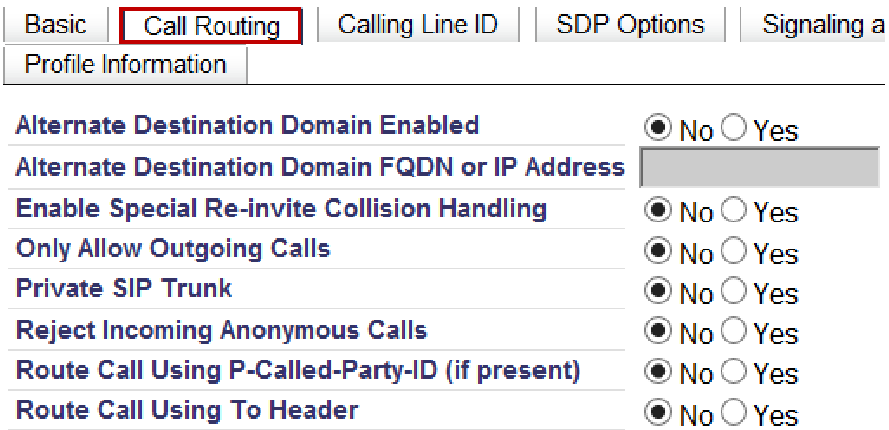


Figure 7: SIP Peer Profile-3

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

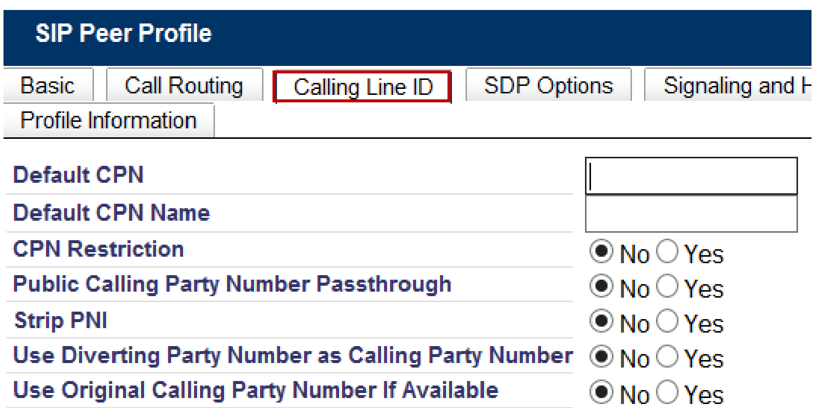


Figure 7: SIP Peer Profile-4

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

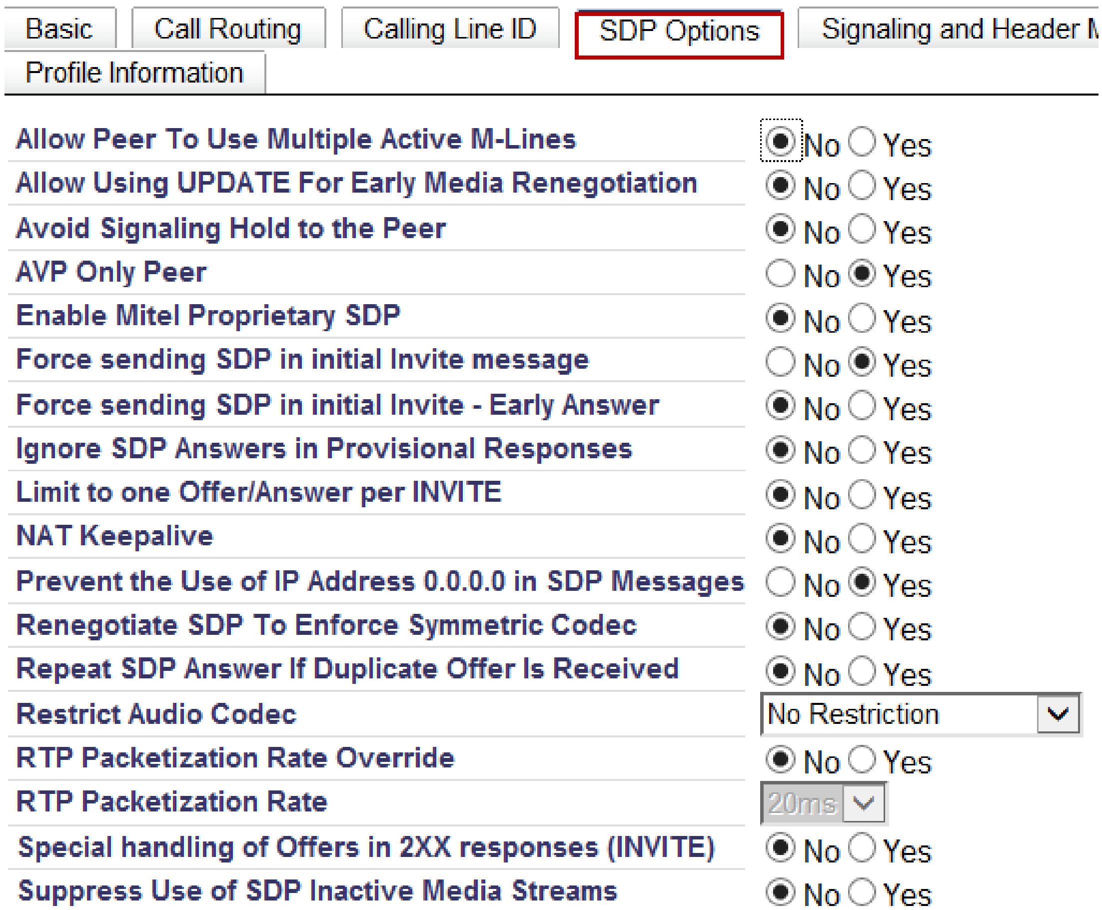


Figure 7: SIP Peer Profile-5

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

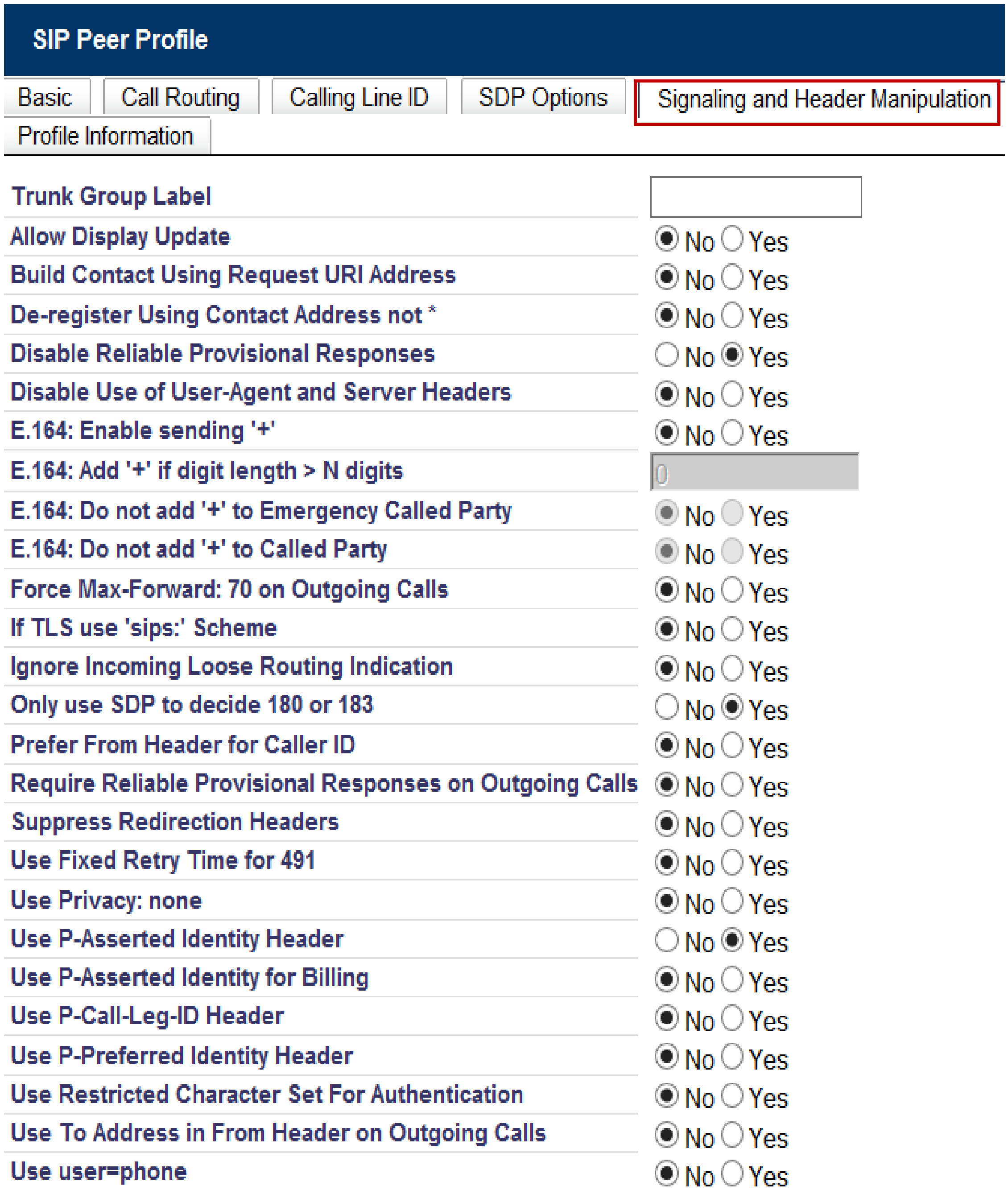


Figure 7: SIP Peer Profile-6

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

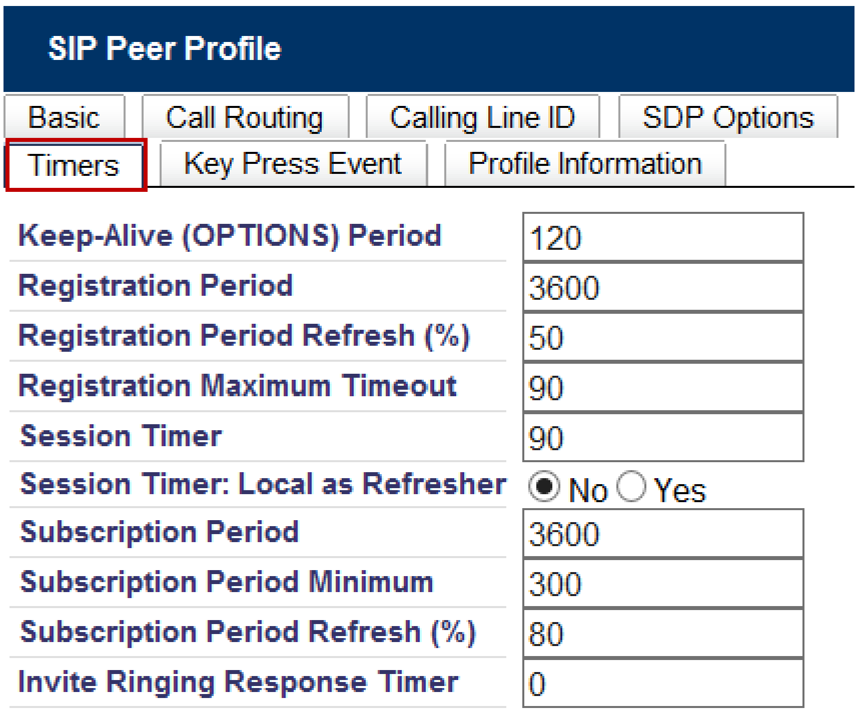


Figure 7: SIP Peer Profile-7

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)

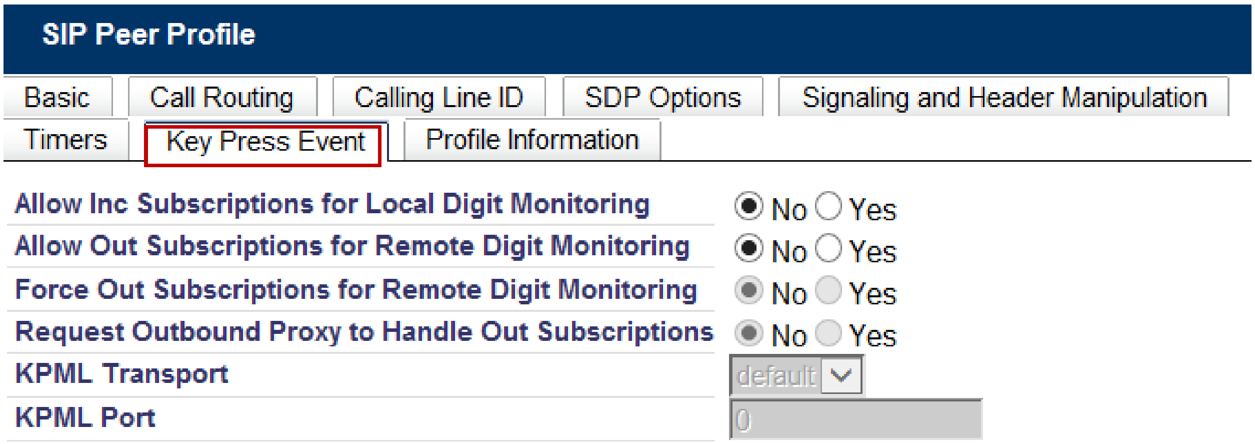


Figure 7: SIP Peer Profile-8

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

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Corporate Headquarters
Cisco Systems, Inc.
170 West Tasman Drive
San Jose, CA 95134-1706
USA
www.cisco.com
Tel: 408 526-4000
800 553-NETS (6387)
Fax: 408 526-4100

European Headquarters
CiscoSystems International BV
Haarlerbergpark
Haarlerbergweg 13-19
1101 CH Amsterdam
The Netherlands
www-europe.cisco.com
Tel: 31 0 20 357 1000
Fax: 31 0 20 357 1100

Americas Headquarters
Cisco Systems, Inc.
170 West Tasman Drive
San Jose, CA 95134-1706
USA
www.cisco.com
Tel: 408 526-7660
Fax: 408 527-0883

AsiaPacific Headquarters
Cisco Systems, Inc.
Capital Tower
168 Robinson Road
#22-01 to #29-01
Singapore 068912
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Tel: +65 317 7777
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