MetTel SIP Trunking:

Cisco Unified Communications Manager 10.5.2 with Cisco Unified Border Element 10.0.2 [IOS 15.4(3)M1] using SIP

May 18, 2015
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Introduction

Service Providers today, such as MetTel, 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.

MetTel 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 Cisco Unified Communications Manager and MetTel network, Cisco Unified Border Element (Cisco UBE) 15.4(3)M1 can be used. The Cisco Unified Border Element 15.4(3)M1 provides demarcation, security, interworking and session control services for Cisco Unified Communications Manager 10.5.2 connected to MetTel IP network.

This document assumes the reader is knowledgeable with the terminology and configuration of Cisco UCM (Cisco Unified Communications Manager). Only configuration settings specifically required for MetTel 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 Cisco Unified Communications Manager (Cisco UCM) 10.5.2 and Cisco Unified Border Element (Cisco UBE) 15.4(3)M1 for connectivity to MetTel SIP trunking service. The deployment model covered in this application note is CPE (Cisco UCM 10.5.2) to PSTN (MetTel).
- Testing was performed in accordance to MetTel 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 interoperability with Cisco Unity Connection (CUC)
- The Cisco UCM configuration detailed in this document is based on a lab environment with a simple dial-plan used to ensure proper interoperability between MetTel SIP network and Cisco Unified Communications. 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 Cisco UCM to interoperate to MetTel SIP trunking network.

This application note does not cover the use of Calling Search Spaces (CSS) or partitions on Cisco UCM. To understand and learn how to apply CSS and partitions refer to the cisco.com link below:

Cisco IP Phones 9971 and 7961 phones are the devices primarily used throughout the testing to place or receive calls.

VentaFax Soft Client is used to perform all fax related scenarios. The fax client is connected to SIP Gateway via FXS port which in turn communicates with Cisco UCM over SIP.
System Components

Hardware Requirements
- Cisco UCS-C240-M3S VMWare host running ESXi 5.5 Standard
- Cisco UCM and Cisco Unity Connection running on VMware
- Cisco UBE on Cisco ISR 2911 router
- Cisco 2901 Fax Gateway
- IP phones 9971(SIP) and 7961 (SCCP).

Software Requirements
- Cisco Unified Communications Manager 10.5.2
- Cisco Unity Connection 10.5.2
- IOS 15.4(3)M1 for ISR 2911 Cisco Unified Border Element
- IOS 15.4(3)M1 for Cisco 2901 Fax Gateway

Features

Features Supported
- Incoming and outgoing off-net calls using G729
- Call hold
- Call transfer (unattended and attended)
- Call forward (all, busy and no answer)
- Calling Line (number) Identification Presentation (CLIP)
- Calling Line (number) Identification Restriction (CLIR)
- DTMF (RFC2833)
- Media flow-through on Cisco UBE
- Fax (G.711 and T.38)

Features Not Supported
- Cisco IP phones used in this test do not support blind transfer. Unattended and attended transfer scenarios were tested

Caveats
- Cisco IOS Enhanced Conference Bridge is used to establish conference with PSTN using G729 codec
- CLID is not updated on PSTN phones for transfer scenarios
- Testing is done with only one PBX
Configuration

Configuring Cisco Unified Border Element

Network Interface
Configure Ethernet IP address and sub interface. The IP address and VLAN encapsulation used are for illustration only, the actual IP address can vary. For SIP trunks two IP addresses must be configured - for LAN and WAN.

interface GigabitEthernet0/0
description $ETH-LAN$$ETH-SW-LAUNCH$$INTF-INFO-GE 0/0$
ip address 192.65.79.140 255.255.255.128
standby delay minimum 30 reload 60
standby 1 ip 192.65.79.144
standby 1 priority 50
standby 1 preempt
standby 1 name SB
standby 1 track 1 decrement 10
duplex auto
speed auto
!
interface GigabitEthernet0/1
ip address 10.80.18.30 255.255.255.0
standby delay minimum 30 reload 60
standby 6 ip 10.80.18.10
standby 6 priority 50
standby 6 preempt
standby 6 track 2 decrement 10
duplex auto
speed auto
!

Global Cisco UBE settings
In order to enable Cisco UBE IP2IP gateway functionality, following command has to be entered:

voice service voip
ip address trusted list
ipv4 0.0.0.0 0.0.0.0
address-hiding
mode border-element
allow-connections sip to sip
redundancy
no supplementary-service sip handle-replaces
redirect ip2ip
fax protocol t38 version 0 ls-redundancy 0 hs-redundancy 0 fallback none
sip
header-passing
error-passthru
asserted-id pai
early-offer forced
no silent-discard untrusted
midcall-signaling passthru
privacy-policy passthru
!


Explanation

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
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<tr>
<td>allow-connections sip to sip</td>
<td>Allow IP2IP connections between two SIP call legs</td>
</tr>
<tr>
<td>fax protocol</td>
<td>Specifies the fax protocol</td>
</tr>
<tr>
<td>asserted-id</td>
<td>Specifies the type of privacy header in the outgoing SIP requests and response messages</td>
</tr>
<tr>
<td>early-offer forced</td>
<td>Enables SIP Delayed-Offer to Early-Offer globally</td>
</tr>
<tr>
<td>midcall-signaling passthru</td>
<td>Passes SIP messages from one IP leg to another IP leg</td>
</tr>
</tbody>
</table>

Media Passing through Cisco UBE (media flow-through vs. media flow-around)
Default Cisco UBE configuration enables Cisco UBE to work in flow-through mode (this test use Flow-through mode). In order to enable flow-around mode, please perform the following actions:

voice service voip
  media flow-around

Codecs
G729 is used as the preferred codec for this testing.

voice class codec 1
  codec preference 1 g729r8
  codec preference 2 g711ulaw.
Dial peer
Cisco UBE uses dial-peer to route the call accordingly based on the digits.

dial-peer voice 100 voip

description Outbound-from IP PBX to PSTN - LAN facing

session protocol sipv2

session transport udp

incoming called-number 214242....

voice-class codec 1

voice-class sip asserted-id pai

voice-class sip profiles 1

voice-class sip bind control source-interface GigabitEthernet0/1

voice-class sip bind media source-interface GigabitEthernet0/1

dtmf-relay rtp-nte

fax rate 14400

fax protocol t38 version 0 ls-redundancy 0 hs-redundancy 0 fallback none

no vad

!
dial-peer voice 200 voip

description Inbound-from PSTN to IP PBX - WAN facing

huntstop

session protocol sipv2

incoming called-number 917819....

voice-class codec 1

voice-class sip asymmetric payload full

voice-class sip asserted-id pai

voice-class sip privacy-policy passthru
voice-class sip profiles 1
voice-class sip bind control source-interface GigabitEthernet0/0
voice-class sip bind media source-interface GigabitEthernet0/0
dtmf-relay rtp-nte
fax rate 14400
fax protocol t38 version 0 ls-redundancy 0 hs-redundancy 0 fallback none
no vad
!
dial-peer voice 201 voip
description Inbound-from PSTN to IP PBX - LAN facing
huntstop
destination-pattern 917819....
session protocol sipv2
session target ipv4:10.80.18.3
session transport udp
voice-class codec 1
voice-class sip asymmetric payload full
voice-class sip asserted-id pai
voice-class sip privacy-policy passthru
voice-class sip early-offer forced
voice-class sip profiles 1
voice-class sip bind control source-interface GigabitEthernet0/1
voice-class sip bind media source-interface GigabitEthernet0/1
dtmf-relay rtp-nte
fax rate 14400
fax protocol t38 version 0 ls-redundancy 0 hs-redundancy 0 fallback none
no vad
!
dial-peer voice 101 voip
description Outbound-from IP PBX to PSTN - WAN facing
destination-pattern [2-9]T 
session protocol sipv2
session target sip-server
session transport udp
voice-class codec 2 
voice-class sip asymmetric payload full
voice-class sip asserted-id pai 
voice-class sip privacy-policy passthru
voice-class sip early-offer forced
voice-class sip profiles 1
voice-class sip bind control source-interface GigabitEthernet0/0
voice-class sip bind media source-interface GigabitEthernet0/0
dtmf-relay rtp-n-te
fax rate 14400
fax protocol t38 version 0 ls-redundancy 0 hs-redundancy 0 fallback none
no vad
!
Call flow

In the sample configuration presented here, Cisco UCM is provisioned with four-digit directory numbers corresponding to the last four DID digits. No digit manipulation is performed on the Cisco UBE.

For incoming PSTN calls, the Cisco UBE presents the full ten-digit DID number to Cisco UCM. The Cisco UCM Translation Pattern strips all but the last four digits and routes the call based on those digits. Voice calls are routed to IP phones; Fax calls are routed via a 4-digit route pattern over a SIP trunk that terminates on the Fax Gateway and in turn to the VentaFax client connected to the Fax Gateway.

CPE callers make outbound PSTN calls by dialing a “9” prefix followed by the destination number. For outbound fax calls from the analog fax endpoint, Cisco fax Gateway sends to Cisco UCM the DID with leading access code “9”. A “9.@” route pattern strips the prefix and routes the call with the remaining digits via a SIP trunk terminating on the Cisco UBE for Voice call or outbound Fax.

Figure 2 Outbound Voice Call

Figure 3 Inbound Voice Call
Configuration example
The following configuration snippet contains a sample configuration of Cisco UBE with all parameters mentioned previously.

**Active Cisco UBE:**
User Access Verification

Username: cisco
Password:
% Password expiration warning.

Cisco Configuration Professional (Cisco CP) is installed on this device and it provides the default username "cisco" for one-time use. If you have already used the username "cisco" to login to the router and your IOS image supports the "one-time" user option, then this username has already expired. You will not be able to login to the router with this username after you exit this session.
It is strongly suggested that you create a new username with a privilege level of 15 using the following command.

```
username <myuser> privilege 15 secret 0 <mypassword>
```

Replace `<myuser>` and `<mypassword>` with the username and password you want to use.
MetTelCube#show run
Building configuration...

Current configuration : 13734 bytes
!
! No configuration change since last restart
!
version 15.4
service timestamps debug datetime msec
service timestamps log datetime msec
service password-encryption
!
hostname MetTelCube
!
boot-start-marker
boot-end-marker
!
aqm-register-fnf
!
logging buffered 51200 warnings
enable secret 5 $1$CH86$KMS9W99Thge87.5iudgvy1
!
ipc zone default
association 1
no shutdown
protocol sctp
local-port 5000
  local-ip 10.80.18.30
remote-port 5000
  remote-ip 10.80.18.20
!
no aaa new-model
!
no ip domain lookup
ip cef
no ipv6 cef
!
multilink bundle-name authenticated
!
cts logging verbose
!
crypto pki trustpoint TP-self-signed-774086054
  enrollment selfsigned
  subject-name cn=IOS-Self-Signed-Certificate-774086054
  revocation-check none
  rsakeypair TP-self-signed-774086054
!
crypto pki certificate chain TP-self-signed-774086054
  certificate self-signed 01
  30820229 30820192 A0030201 02020101 300D0609 2A864886 F70D0101 05050030
30312E30 2C060355 04031325 494F532D 53656C66 2D536967 6E65642D 43657274
69666663 6174652D 3737343038 363035 401E 17 0D313530 33303720 38363054
305A170D 32303031 30303030 5A303031 17 0D32303031 30303030 5A303031 2E302C06
3550403 1325494F 532D5365 6C662D53 69676E65 642D4365 72746966 69636174 652D3737
34303836 30353430 819F300D 06092A86 4886F70D 01010105 0003818D 00308189 02818100
B2E766A5 82BD3422 25F6EB7B 5D4393DD F9577A7D CBEDB4BF 7775A94C B1D89535
09A55E99 9C3D3C6B E9688C17 242A977E B5E8055B 11238068 8D8DD86E 25964916
A51F70E0 B17A2CEC 714869A4 92A3528B DE4A48B E65EC342 42023284 1F9B7A83
D29D4A41 0789AD2A DC4B3EF0 D010E6A7 63EE80D7 9A98BD28 BF14CDF8 D2A20C85
02030100 01A35330 51300F06 03551D13 0101FF04 05300301 01FF301F 0603551D
23041830 16801496 638DBF0B 111BE8154 4A536831 2A6490C8 6E8E1F30 1D060355
1D0E0416 04149666 8D8F0B11 BEB1534A A568312A 6490C86E 8E1F300D 06092A86
4886F70D 01010505 00381810 001144F9 AC53A0E1 1289A22C 21F1E3D8 7D46E918
226CE2AE B5D3B141 2A5F4FBD FF6F3D68 C4BEADE6 24F404A5 FC62A18D B23AB556
AFC122C9 90F5C68D 09F33068 C584B4B4 FE5D656D DD5DC9AE 8BF6C2C3 9038F25B
B4C17289 F7F8C1C9 590C0439 E9C31825 D25BDD6E 09C6B6C8 DD6EFE4B D6CB93AF
EB9DA82A 34A67EBD BD3CD690 4D

quit

voice-card 0
dspfarm
dsp services dspfarm

!
voice service voip
ip address trusted list
ipv4 0.0.0.0 0.0.0.0
address-hiding
mode border-element
allow-connections sip to sip
redundancy
no supplementary-service sip handle-replaces
redirect ip2ip
fax protocol t38 version 0 ls-redundancy 0 hs-redundancy 0 fallback none
sip
header-passing
error-passthru
asserted-id pai
early-offer forced
no silent-discard untrusted
midcall-signaling passthru
privacy-policy passthru
!
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 sip-header Diversion modify "<sip:(.*)@(.*)>" "<sip:917819\1@\2>"

response ANY sip-header Allow-Header modify "UPDATE," ""

!

voice translation-rule 2

rule 1 /^.*\(51..\)/ /917819\1/

!

voice translation-profile Mettel

translate called 2

!

license udi pid CISCO2911/K9 sn FTX1807AJH3

license boot module c2900 technology-package securityk9

!

username cisco privilege 15 password 7 0010160D320A11575F2F

!

redundancy inter-device

scheme standby SB

!

redundancy

!

track 1 interface GigabitEthernet0/1 line-protocol

track 2 interface GigabitEthernet0/0 line-protocol

!

interface Embedded-Service-Engine0/0

no ip address

!
interface GigabitEthernet0/0
description $ETH-LAN$ETH-SW-LAUNCH$INTF-INFO-GE 0/0$
ip address 192.65.79.140 255.255.255.128
standby delay minimum 30 reload 60
standby 1 ip 192.65.79.144
standby 1 priority 50
standby 1 preempt
standby 1 name SB
standby 1 track 1 decrement 10
duplex auto
speed auto
!
interface GigabitEthernet0/1
ip address 10.80.18.30 255.255.255.0
standby delay minimum 30 reload 60
standby 6 ip 10.80.18.10
standby 6 priority 50
standby 6 preempt
standby 6 track 2 decrement 10
duplex auto
speed auto
!
interface GigabitEthernet0/2
no ip address
shutdown
duplex auto
speed auto
!
ip forward-protocol nd
!

ip http server

ip http authentication local

ip http secure-server

ip http timeout-policy idle 60 life 86400 requests 10000
!

ip route 0.0.0.0 0.0.0.0 192.65.79.129

ip route 10.64.0.0 255.255.0.0 10.80.18.1

ip route 172.16.0.0 255.255.0.0 10.80.18.1
!

color-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 Outbound-from IP PBX to PSTN - LAN facing

session protocol sipv2

session transport udp

incoming called-number 214242....

voice-class codec 1

voice-class sip profiles 1

voice-class sip bind control source-interface GigabitEthernet0/1
voice-class sip bind media source-interface GigabitEthernet0/1
dtmf-relay rtp-nte
fax rate 14400
fax protocol t38 version 0 ls-redundancy 0 hs-redundancy 0 fallback none
no vad
!
dial-peer voice 200 voip
description Inbound-from PSTN to IP PBX - WAN facing
huntstop
session protocol sipv2
incoming called-number 917819....
voice-class codec 1
voice-class sip asymmetric payload full
voice-class sip privacy-policy passthru
voice-class sip profiles 1
voice-class sip bind control source-interface GigabitEthernet0/0
voice-class sip bind media source-interface GigabitEthernet0/0
dtmf-relay rtp-nte
fax rate 14400
fax protocol t38 version 0 ls-redundancy 0 hs-redundancy 0 fallback none
no vad
!
dial-peer voice 201 voip
description Inbound-from PSTN to IP PBX - LAN facing
huntstop
destination-pattern 917819....
session protocol sipv2
session target ipv4:10.80.18.3
session transport udp
voice-class codec 1
voice-class sip asymmetric payload full
voice-class sip privacy-policy passthru
voice-class sip early-offer forced
voice-class sip profiles 1
voice-class sip bind control source-interface GigabitEthernet0/1
voice-class sip bind media source-interface GigabitEthernet0/1
dtmf-relay rtp-nte
fax rate 14400
fax protocol t38 version 0 ls-redundancy 0 hs-redundancy 0 fallback none
no vad
!
dial-peer voice 101 voip
description Outbound-from IP PBX to PSTN - WAN facing
destination-pattern [2-9]T
session protocol sipv2
session target sip-server
session transport udp
voice-class codec 1
voice-class sip asymmetric payload full
voice-class sip privacy-policy passthru
voice-class sip early-offer forced
voice-class sip profiles 1
voice-class sip bind control source-interface GigabitEthernet0/0
voice-class sip bind media source-interface GigabitEthernet0/0
dtmf-relay rtp-nte
fax rate 14400
fax protocol t38 version 0 ls-redundancy 0 hs-redundancy 0 fallback none
no vad
!
dial-peer voice 420 voip
description PBX to PBX dialing
translation-profile outgoing Mettel
destination-pattern 51..
session protocol sipv2
session target sip-server
session transport udp
voice-class codec 1
voice-class sip profiles 1
voice-class sip bind control source-interface GigabitEthernet0/0
voice-class sip bind media source-interface GigabitEthernet0/0
dtmf-relay rtp-npe
fax rate 14400
fax protocol t38 version 0 ls-redundancy 0 hs-redundancy 0 fallback none
no vad
!
dial-peer voice 110 voip
description Outbound-from IP PBX to PSTN - WAN facing
destination-pattern 1T
session protocol sipv2
session target sip-server
voice-class codec 1
voice-class sip profiles 1
voice-class sip bind control source-interface GigabitEthernet0/0
voice-class sip bind media source-interface GigabitEthernet0/0

dtmf-relay rtp-nte
fax rate 14400
fax protocol t38 version 0 ls-redundancy 0 hs-redundancy 0 fallback none
no vad
!
dial-peer voice 111 voip
description Outbound-from IP PBX to PSTN - LAN facing"
session protocol sipv2
incoming called-number 1T
voice-class codec 1
voice-class sip profiles 1
voice-class sip bind control source-interface GigabitEthernet0/1
voice-class sip bind media source-interface GigabitEthernet0/1
dtmf-relay rtp-nte
no vad
!
dial-peer voice 300 voip
description Int'l calls to MetTel - WAN facing
destination-pattern 011T
session protocol sipv2
session target sip-server
voice-class codec 1
voice-class sip profiles 1
voice-class sip bind control source-interface GigabitEthernet0/0
voice-class sip bind media source-interface GigabitEthernet0/0
dtmf-relay rtp-nte
fax-relay ecm disable
fax rate disable
fax protocol pass-through g711ulaw
no vad
!
dial-peer voice 400 voip
description " Int'l calls to MetTel - LAN facing
session protocol sipv2
incoming called-number 011T
voice-class codec 1
voice-class sip profiles 1
voice-class sip bind control source-interface GigabitEthernet0/1
voice-class sip bind media source-interface GigabitEthernet0/1
dtmf-relay rtp-nte
fax-relay ecm disable
fax rate disable
fax protocol pass-through g711ulaw
no vad
!
dial-peer voice 500 voip
description N11 Calls to MetTel - WAN facing
destination-pattern .11
session protocol sipv2
session target sip-server
voice-class codec 1
voice-class sip profiles 1
voice-class sip bind control source-interface GigabitEthernet0/0
voice-class sip bind media source-interface GigabitEthernet0/0
dtmf-relay rtp-nte
no vad
dial-peer voice 600 voip
description N11 Calls to MetTel - LAN facing
session protocol sipv2
incoming called-number .11
voice-class codec 1
voice-class sip profiles 1
voice-class sip bind control source-interface GigabitEthernet0/1
voice-class sip bind media source-interface GigabitEthernet0/1
dtmf-relay rtp-nte
no vad
!
dial-peer voice 122 voip
description "OPERATOR TESTING"
destination-pattern 0
session protocol sipv2
voice-class codec 1
session target sip-server
voice-class sip bind control source-interface GigabitEthernet0/0
voice-class sip bind media source-interface GigabitEthernet0/0
dtmf-relay rtp-nte
no vad
!
dial-peer voice 210 voip
description TollFree Calls to MetTel - WAN facing
destination-pattern 800T
session protocol sipv2
session target sip-server
voice-class codec 1
voice-class sip profiles 1
voice-class sip bind control source-interface GigabitEthernet0/0
voice-class sip bind media source-interface GigabitEthernet0/0
dtmf-relay rtp-nnte
no vad
!
dial-peer voice 211 voip
description TollFree Calls to MetTel - LAN facing
session protocol sipv2
session target sip-server
incoming called-number 800T
voice-class codec 1
voice-class sip profiles 1
voice-class sip bind control source-interface GigabitEthernet0/1
voice-class sip bind media source-interface GigabitEthernet0/1
dtmf-relay rtp-nnte
no vad
!
sip-ua
retry invite 2
timers expires 1800000
sip-server ipv4:72.11.195.82
!
gatekeeper
shutdown
!
banner exec ^C

% Password expiration warning.

-----------------------------------------------------------------------
Cisco Configuration Professional (Cisco CP) is installed on this device
and it provides the default username "cisco" for one-time use. If you have
already used the username "cisco" to login to the router and your IOS image
supports the "one-time" user option, then this username has already expired.
You will not be able to login to the router with this username after you exit
this session.

It is strongly suggested that you create a new username with a privilege level
of 15 using the following command.

username <myuser> privilege 15 secret 0 <mypassword>

Replace <myuser> and <mypassword> with the username and password you want to
use.

-----------------------------------------------------------------------
^C
banner login ^C

-----------------------------------------------------------------------
Cisco Configuration Professional (Cisco CP) is installed on this device.
This feature requires the one-time use of the username "cisco" with the
password "cisco". These default credentials have a privilege level of 15.

YOU MUST USE CISCO CP or the CISCO IOS CLI TO CHANGE THESE PUBLICLY-KNOWN
CREDENTIALS

Here are the Cisco IOS commands.

username <myuser> privilege 15 secret 0 <mypassword>
no username cisco

Replace <myuser> and <mypassword> with the username and password you want to use.

IF YOU DO NOT CHANGE THE PUBLICLY-KNOWN CREDENTIALS, YOU WILL NOT BE Able TO LOG INTO THE DEVICE AGAIN AFTER YOU HAVE LOGGED OFF.

For more information about Cisco CP please follow the instructions in the QUICK START GUIDE for your router or go to http://www.cisco.com/go/ciscocp

^C
!
line con 0
login local
line aux 0
line 2
no activation-character
no exec
transport preferred none
transport output pad telnet rlogin lapb-ta mop udptn v120 ssh
stopbits 1
line vty 0 4
privilege level 15
login local
transport input telnet ssh
line vty 5 15
privilege level 15
login local
transport input telnet ssh
!
scheduler allocate 20000 1000
ntp server 10.10.10.5
!
End

Standby Cisco UBE:

User Access Verification
Username: cisco
Password:
Mettel-HA#show run
Building configuration...
Current configuration : 11963 bytes
!
version 15.4
service timestamps debug datetime msec
service timestamps log datetime msec
no service password-encryption
!
hostname Mettel-HA

! boot-start-marker
boot-end-marker
!
aqm-register-fnf
!
card type t1 0 0
logging buffered 51200 warnings
enable secret 4 sKgCY/XPea3wk8xoe5Wo7UGFaNVwzXDEyXWhuDjeLk
!
!
ipc zone default
association 1
no shutdown
protocol sctp
local-port 5000
local-ip 10.80.18.20
remote-port 5000
remote-ip 10.80.18.30
!
no aaa new-model
network-clock-participate wic 0
!
!
!
!  
nocpdomainlookup  
ipv4cef  
nocp6cef  
multilinkbundle-nameauthenticated  
!  
cryptopkitrustpointTP-self-signed-1068156983  
enrollmentselfsigned  
subject-namecn=IOS-Self-Signed-Certificate-1068156983  
revocation-checknone  
rskakeypairTP-self-signed-1068156983  
!  
cryptopkicertificatechainTP-self-signed-1068156983  

certificateself-signed01  
  3082022B 30820194 A0030201 02020101 300D0609 2A864886 F70D0101 05050030  
  31312F30 2D060355 04031326 494F532D 53656666 25356967 6E65642D 43657274  
  69666663 6174652D 31303638 31353639 38333301E 170D3135 30323133 30303326  
  35315A17 0D323030 31303130 30303030 305A313201 312F302D 06035504 03132649  
  4F532D53 656C662D 5369676E 65642D 65727469 6669636174652D 31 30363831  
  35363938 3330819F 300D0609 2A864886 F70D0101 01050003 818D0030 81890281  
  8100A36C 4A091065 EE99508B 7E2F2D7D 9DD7EB88 E965EBD6 287BA16F 1C128CE2  
  0BFBD7E8 02C5B854 27F6016A 736B9AB7 B462EBB5 AAB61DC 4FB2EF6C F4566AC6  
  9BBEB905 A4F81660 052F8E49 640037A0 FCC0BB58 CE4AD5D2 94576607 DAF90E89  
  BAB3A516 F096231B 95FB1FB5 CB3074F2 23D0BD6 8AF1AC4A 4D1EC327 A02F4233
C9830203 010001A3 5330S130 0F060355 1D130101 FF040530 030101FF 301F0603
551D2304 18301680 14FC7820 960B27DF 9BB01B11 09617055 BB19A4E9 99301D06
03551D0E 04160414 FC782096 0B27DF9B B01B1109 617055BB 19A4E999 300D0609
2A864886 F70D0101 05050003 8181009C B27374D2 94EEF33A 85C1A570 36E1ED69
635A6DC8 3C02447B 51E3FDB0 11E25BEE 5BA0FED5 55A80630 37A77687 9A1F1258
79CC6BA3 B858C6D2 917685EE 7677073B 0DAC9974 BCF6C741 D057726D 4A9BA23C
C9E81898 7F47CB5F 5EBC4DB 53164DB2 C732E75 0DAA243C 6FBF1B9F 0D03DD76
F57B90C0 BFED12C5 7007C3DC 5212CE

   quit
voice-card 0

!

voice service voip
ip address trusted list
ipv4 0.0.0.0 0.0.0.0
address-hiding
mode border-element
allow-connections sip to sip
redundancy
no supplementary-service sip handle-replaces
redirect ip2ip
fax protocol t38 version 0 ls-redundancy 0 hs-redundancy 0 fallback none
sip
header-passing
error-passthru
asserted-id pai
early-offer forced
no silent-discard untrusted
midcall-signaling passthru
privacy-policy passthru
g729 annexb-all
!
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 sip-header Diversion modify "<sip:(.*)@(.*)>" "<sip:917819\1@\2>"
response ANY sip-header Allow-Header modify "UPDATE," ""
!
voice translation-rule 2
rule 1 /^.*\(51..\)/ /917819\1/
!
voice translation-profile Mettel
translate called 2
!
license udi pid CISCO2911/K9 sn FTX1807AJH2
hw-module pvdm 0/0
!
username cisco privilege 15 password 0 tekV1z10n

redundancy inter-device
  scheme standby SB

redundancy

track 1 interface GigabitEthernet0/1 line-protocol
track 2 interface GigabitEthernet0/0 line-protocol

interface Embedded-Service-Engine0/0
  no ip address
  shutdown

interface GigabitEthernet0/0
  description $ETH-LAN$$ETH-SW-LAUNCH$$INTF-INFO-GE 0/0$
  ip address 192.65.79.141 255.255.255.0
  standby delay minimum 30 reload 60
  standby 0 timers 2 40
  standby 1 ip 192.65.79.144
  standby 1 priority 50
  standby 1 preempt delay minimum 10
  standby 1 name SB
  standby 1 track 1 decrement 90
  duplex auto
speed auto
!
interface GigabitEthernet0/1
ip address 10.80.18.20 255.255.255.0
standby delay minimum 30 reload 60
standby 6 ip 10.80.18.10
standby 6 priority 50
standby 6 preempt delay minimum 10
standby 6 track 2 decrement 10
duplex auto
speed auto
!
interface GigabitEthernet0/2
no ip address
duplex auto
speed auto
!
ip forward-protocol nd
!
ip http server
ip http access-class 23
ip http authentication local
ip http secure-server
ip http timeout-policy idle 60 life 86400 requests 10000
!
ip rtcp report interval 3000
ip route 0.0.0.0 0.0.0.0 10.80.18.1
ip route 10.64.0.0 255.255.0.0 10.80.18.1
ip route 172.16.0.0 255.255.0.0 10.80.18.1

! control-plane

! voice-port 0/2/2

! voice-port 0/2/3

! 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 Outbound-from IP PBX to PSTN - LAN facing
session protocol sipv2
session transport udp
incoming called-number 214242....
voice-class codec 1
voice-class sip profiles 1
voice-class sip bind control source-interface GigabitEthernet0/1
voice-class sip bind media source-interface GigabitEthernet0/1
dtmf-relay rtp-nte
fax rate 14400
fax protocol t38 version 0 ls-redundancy 0 hs-redundancy 0 fallback none
no vad
!
dial-peer voice 200 voip
description Inbound-from PSTN to IP PBX - WAN facing
huntstop
session protocol sipv2
incoming called-number 917819....
voice-class codec 1
voice-class sip asymmetric payload full
voice-class sip privacy-policy passthru
voice-class sip profiles 1
voice-class sip bind control source-interface GigabitEthernet0/0
voice-class sip bind media source-interface GigabitEthernet0/0
dtmf-relay rtp-nte
fax rate 14400
fax protocol t38 version 0 ls-redundancy 0 hs-redundancy 0 fallback none
no vad
!
dial-peer voice 201 voip
description Inbound-from PSTN to IP PBX - LAN facing
huntstop
destination-pattern 917819....
session protocol sipv2
session target ipv4:10.80.18.3
session transport udp
voice-class codec 1
voice-class sip asymmetric payload full
voice-class sip privacy-policy passthru
voice-class sip early-offer forced
voice-class sip profiles 1
voice-class sip bind control source-interface GigabitEthernet0/1
voice-class sip bind media source-interface GigabitEthernet0/1
dtmf-relay rtp-npe
fax rate 14400
fax protocol t38 version 0 ls-redundancy 0 hs-redundancy 0 fallback none
no vad
!
dial-peer voice 101 voip
description Outbound-from IP PBX to PSTN - WAN facing
destination-pattern [2-9]T
session protocol sipv2
session target sip-server
session transport udp
voice-class codec 1
voice-class sip asymmetric payload full
voice-class sip privacy-policy passthru
voice-class sip early-offer forced
voice-class sip profiles 1
voice-class sip bind control source-interface GigabitEthernet0/0
voice-class sip bind media source-interface GigabitEthernet0/0
dtmf-relay rtp-npe
fax rate 14400
fax protocol t38 version 0 ls-redundancy 0 hs-redundancy 0 fallback none
no vad
!
dial-peer voice 420 voip
description PBX to PBX dialing
translation-profile outgoing Mettel
destination-pattern 51..
session protocol sipv2
session target sip-server
session transport udp
voice-class codec 1
voice-class sip profiles 1
voice-class sip bind control source-interface GigabitEthernet0/0
voice-class sip bind media source-interface GigabitEthernet0/0
dtmf-relay rtp-nte
fax rate 14400
fax protocol t38 version 0 ls-redundancy 0 hs-redundancy 0 fallback none
no vad
!
dial-peer voice 110 voip
description Outbound-from IP PBX to PSTN - WAN facing
destination-pattern 1T
session protocol sipv2
session target sip-server
voice-class codec 1
voice-class sip profiles 1
voice-class sip bind control source-interface GigabitEthernet0/0
voice-class sip bind media source-interface GigabitEthernet0/0
dtmf-relay rtp-nte
fax rate 14400
fax protocol t38 version 0 ls-redundancy 0 hs-redundancy 0 fallback none
no vad
!
dial-peer voice 111 voip
description Outbound-from IP PBX to PSTN - LAN facing"
session protocol sipv2
incoming called-number 1T
voice-class codec 1
voice-class sip profiles 1
voice-class sip bind control source-interface GigabitEthernet0/1
voice-class sip bind media source-interface GigabitEthernet0/1
dtmf-relay rtp-nte
no vad
!
dial-peer voice 300 voip
description Int'l calls to MetTel - WAN facing
destination-pattern 011T
session protocol sipv2
session target sip-server
voice-class codec 1
voice-class sip profiles 1
voice-class sip bind control source-interface GigabitEthernet0/0
voice-class sip bind media source-interface GigabitEthernet0/0
dtmf-relay rtp-nte
fax-relay ecm disable
fax rate disable
fax protocol pass-through g711ulaw
no vad!

dial-peer voice 400 voip
description " Int'l calls to MetTel - LAN facing
session protocol sipv2
incoming called-number 011T
voice-class codec 1
voice-class sip profiles 1
voice-class sip bind control source-interface GigabitEthernet0/1
voice-class sip bind media source-interface GigabitEthernet0/1
dtmf-relay rtp-nte
fax-relay ecm disable
fax rate disable
fax protocol pass-through g711ulaw
no vad!

!dial-peer voice 500 voip
description N11 Calls to MetTel - WAN facing
destination-pattern .11
session protocol sipv2
session target sip-server
voice-class codec 1
voice-class sip profiles 1
voice-class sip bind control source-interface GigabitEthernet0/0
voice-class sip bind media source-interface GigabitEthernet0/0
dtmf-relay rtp-nte
no vad
!  
dial-peer voice 600 voip  
description N11 Calls to MetTel - LAN facing  
session protocol sipv2  
incoming called-number .11  
voice-class codec 1  
voice-class sip profiles 1  
voice-class sip bind control source-interface GigabitEthernet0/1  
voice-class sip bind media source-interface GigabitEthernet0/1  
dtmf-relay rtp-nte  
no vad  
!  
dial-peer voice 122 voip  
description "OPERATOR TESTING"  
destination-pattern 0  
session protocol sipv2  
voice-class codec 1  
session target sip-server  
voice-class sip bind control source-interface GigabitEthernet0/0  
voice-class sip bind media source-interface GigabitEthernet0/0  
dtmf-relay rtp-nte  
no vad  
!  
dial-peer voice 210 voip  
description TollFree Calls to MetTel - WAN facing  
destination-pattern 800T  
session protocol sipv2  
session target sip-server  

voice-class codec 1
voice-class sip profiles 1
voice-class sip bind control source-interface GigabitEthernet0/0
voice-class sip bind media source-interface GigabitEthernet0/0
dtmf-relay rtp-nte
no vad
!
dial-peer voice 211 voip
description TollFree Calls to MetTel - LAN facing
session protocol sipv2
session target sip-server
incoming called-number 800T
voice-class codec 1
voice-class sip profiles 1
voice-class sip bind control source-interface GigabitEthernet0/1
voice-class sip bind media source-interface GigabitEthernet0/1
dtmf-relay rtp-nte
no vad
!
!
sip-ua
retry invite 2
timers expires 1800000
sip-server ipv4:72.11.195.82
!
gatekeeper
shutdown
!
! credentials
!
line con 0
login local
line aux 0
line 2
no activation-character
no exec
transport preferred none
transport output pad telnet rlogin lapb-ta mop udp tn v120 ssh
stopbits 1
line vty 0 4
exec-timeout 0 0
privilege level 15
login local
transport input telnet ssh
line vty 5 15
exec-timeout 0 0
privilege level 15
login local
transport input telnet ssh
!
scheduler allocate 20000 1000
!
end
Figure 6 Cisco UCM version
Cisco Call Manager Service Parameters

**Navigation path:** System -> Service Parameters

Select Server* = clus28pubsub—CUCM Voice/Video (Active)

Select Service*= Cisco CallManager (Active)

Select Duplex Streaming Enabled* = True

![Cisco Unified CM Administration](image)

**Figure 7 Service Parameter**
## System

<table>
<thead>
<tr>
<th>Parameter</th>
<th>Value 1</th>
<th>Value 2</th>
</tr>
</thead>
<tbody>
<tr>
<td>CDR Enabled Flag</td>
<td>True</td>
<td>False</td>
</tr>
<tr>
<td>CDR Log Calls with Zero Duration Flag</td>
<td>True</td>
<td>False</td>
</tr>
<tr>
<td>Digit Analysis Complexity</td>
<td>StandardAnalysis</td>
<td>StandardAnalysis</td>
</tr>
<tr>
<td>Database Payout Time</td>
<td>0</td>
<td>0</td>
</tr>
<tr>
<td>Maximum Phone Fallback Queue Depth</td>
<td>10</td>
<td>10</td>
</tr>
<tr>
<td>Maximum Number of Registered Devices</td>
<td>5000</td>
<td>5000</td>
</tr>
<tr>
<td>System Initialization Timer</td>
<td>60</td>
<td>60</td>
</tr>
</tbody>
</table>

There are hidden parameters in this group. Click on Advanced button to see hidden parameters.

## SDL Trace

<table>
<thead>
<tr>
<th>Parameter</th>
<th>Value 1</th>
<th>Value 2</th>
</tr>
</thead>
<tbody>
<tr>
<td>SDL Trace Data Flags</td>
<td>0x80000003F</td>
<td>0x000000111</td>
</tr>
<tr>
<td>SDL Trace Flush Immediately</td>
<td>False</td>
<td>False</td>
</tr>
<tr>
<td>SDL Trace Data Size</td>
<td>0</td>
<td>0</td>
</tr>
<tr>
<td>SDL Trace Flag</td>
<td>True</td>
<td>False</td>
</tr>
<tr>
<td>SDL TraceType Flags</td>
<td>0x010245FFS</td>
<td>0x80000015</td>
</tr>
</tbody>
</table>

There are hidden parameters in this group. Click on Advanced button to see hidden parameters.

## Clusterwide Parameters (Device - General)

<table>
<thead>
<tr>
<th>Parameter</th>
<th>Value 1</th>
<th>Value 2</th>
</tr>
</thead>
<tbody>
<tr>
<td>Show Line Group Member Env in FinalCalledPartyNumber CDR Field</td>
<td>Enabled Only When CDR Enabled Flag is True</td>
<td>Disabled</td>
</tr>
<tr>
<td>Show Line Group Member Non-Hanged Dn in FinalCalledPartyNumber CDR Field</td>
<td>True</td>
<td>False</td>
</tr>
<tr>
<td>CTT New Call Accept Timer</td>
<td>4</td>
<td>4</td>
</tr>
<tr>
<td>CTT Generate Busy Timer</td>
<td>456</td>
<td>250</td>
</tr>
<tr>
<td>CTT Call Digit Interval</td>
<td>155</td>
<td>250</td>
</tr>
<tr>
<td>CTT Await Further Digits</td>
<td>False</td>
<td>False</td>
</tr>
<tr>
<td>CTT Use Wildcard Pattern as calledPartyDigits</td>
<td>False</td>
<td>False</td>
</tr>
<tr>
<td>Retain Media on Disconnect with PI for Active Call</td>
<td>False</td>
<td>False</td>
</tr>
<tr>
<td>Station and Backup Server KeepAlive Interval</td>
<td>60</td>
<td>60</td>
</tr>
<tr>
<td>Station KeepAlive Interval</td>
<td>20</td>
<td>30</td>
</tr>
<tr>
<td>Status Enquiry Fail Flag</td>
<td>False</td>
<td>False</td>
</tr>
<tr>
<td>Strip # sign from Called Party Number</td>
<td>True</td>
<td>True</td>
</tr>
<tr>
<td>Session Handoff Alerting Timer</td>
<td>10</td>
<td>10</td>
</tr>
<tr>
<td>T301 Timer</td>
<td>180000</td>
<td>180000</td>
</tr>
<tr>
<td>T302 Timer</td>
<td>3000</td>
<td>15000</td>
</tr>
<tr>
<td>T303 Timer</td>
<td>4000</td>
<td>4000</td>
</tr>
<tr>
<td>T304 Timer</td>
<td>5000</td>
<td>30000</td>
</tr>
<tr>
<td>T305 Timer</td>
<td>5000</td>
<td>30000</td>
</tr>
<tr>
<td>T306 Timer</td>
<td>20000</td>
<td>30000</td>
</tr>
<tr>
<td>T308 Timer</td>
<td>4000</td>
<td>4000</td>
</tr>
<tr>
<td>T309 Timer</td>
<td>90000</td>
<td>90000</td>
</tr>
<tr>
<td>T310 Timer</td>
<td>90000</td>
<td>90000</td>
</tr>
<tr>
<td>T312 Timer</td>
<td>60000</td>
<td>60000</td>
</tr>
<tr>
<td>T314 Timer</td>
<td>4000</td>
<td>4000</td>
</tr>
<tr>
<td>T316 Timer</td>
<td>120000</td>
<td>120000</td>
</tr>
<tr>
<td>T317 Timer</td>
<td>100000</td>
<td>100000</td>
</tr>
<tr>
<td>T332 Timer</td>
<td>30000</td>
<td>30000</td>
</tr>
</tbody>
</table>

Figure 8 Service Parameter (cont.)
<table>
<thead>
<tr>
<th>Parameter</th>
<th>Value 1</th>
<th>Value 2</th>
</tr>
</thead>
<tbody>
<tr>
<td>3.3.3.3 Timer</td>
<td>4000</td>
<td>4000</td>
</tr>
<tr>
<td>Tone on Hold Timer</td>
<td>10</td>
<td>10</td>
</tr>
<tr>
<td>Unknown Caller ID Flag</td>
<td>True</td>
<td>True</td>
</tr>
<tr>
<td>Call Classification</td>
<td>Off</td>
<td>Off</td>
</tr>
<tr>
<td>Always Display Original Dialed Number</td>
<td>True</td>
<td>False</td>
</tr>
<tr>
<td>Name Display for Original Dialed Number When Translated</td>
<td>Show the Display Name for Original Dialed Number even if Translated</td>
<td>Show the Display Name for Original Dialed Number even if Translated</td>
</tr>
<tr>
<td>Always Use 7s with Original Dialed Number</td>
<td>False</td>
<td>False</td>
</tr>
<tr>
<td>Fall Call If Trusted Relay Point Allocation Failure</td>
<td>True</td>
<td>True</td>
</tr>
<tr>
<td>Display Calling ID When PI is Not Available</td>
<td>False</td>
<td>False</td>
</tr>
<tr>
<td>Enable Transit Counter Processing on OCS Trunks</td>
<td>False</td>
<td>False</td>
</tr>
<tr>
<td>Express Facility Count</td>
<td>6</td>
<td>6</td>
</tr>
</tbody>
</table>

There are hidden parameters in this group. Click on Advanced button to see hidden parameters.

### Clusterwide Parameters (Device - Phone)

- **Always Use Prime Line**: False
- **Always Use Prime Line for Voice Message**: False
- **Audio Bridge Enable**: Off
- **Device Mobility Mode**: Off
- **Doosler Device Mobility Location During Phone Registration**: True
- **Auto Answer Timer**: 1
- **Extension Display on Cisco IP Phone Model 7910**: False
- **Alternate Idle Phone Auto-Answer Behavior Enabled**: False
- **Hold Taps**: False

<table>
<thead>
<tr>
<th>Parameter</th>
<th>Value 1</th>
<th>Value 2</th>
</tr>
</thead>
<tbody>
<tr>
<td>Extension Display on Cisco IP Phone Model 7910</td>
<td>False</td>
<td>False</td>
</tr>
<tr>
<td>Alternate Idle Phone Auto-Answer Behavior Enabled</td>
<td>False</td>
<td>False</td>
</tr>
<tr>
<td>Hold Taps</td>
<td>False</td>
<td>False</td>
</tr>
</tbody>
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<table>
<thead>
<tr>
<th>Parameter</th>
<th>Value 1</th>
<th>Value 2</th>
</tr>
</thead>
<tbody>
<tr>
<td>Off-hook to First Digi Timer</td>
<td>15000</td>
<td>15000</td>
</tr>
<tr>
<td>Override Auto Answer If Speaker Is Disabled</td>
<td>Not Enough Bandwidth</td>
<td>Not Enough Bandwidth</td>
</tr>
<tr>
<td>SIP Network Congestion Recalculating Text</td>
<td>Only Apply Ring Setting of Busy Station When Incoming Call Arrives</td>
<td>Only Apply Ring Setting of Busy Station When Incoming Call Arrives</td>
</tr>
</tbody>
</table>

<table>
<thead>
<tr>
<th>Parameter</th>
<th>Value 1</th>
<th>Value 2</th>
</tr>
</thead>
<tbody>
<tr>
<td>Transfer On-hook Enabled</td>
<td>False</td>
<td>False</td>
</tr>
<tr>
<td>Ring Setting of Busy Station</td>
<td>Only Apply Ring Setting of Busy Station When Incoming Call Arrives</td>
<td>Only Apply Ring Setting of Busy Station When Incoming Call Arrives</td>
</tr>
<tr>
<td>Ring Setting of Busy Station</td>
<td>BEEP ONLY</td>
<td>BEEP ONLY</td>
</tr>
<tr>
<td>Transfer On-hook Enabled</td>
<td>True</td>
<td>True</td>
</tr>
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</table>

<table>
<thead>
<tr>
<th>Parameter</th>
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<th>Value 2</th>
</tr>
</thead>
<tbody>
<tr>
<td>SIP Station KeepAlive Interval</td>
<td>120</td>
<td>120</td>
</tr>
<tr>
<td>SIP Station Realm</td>
<td>ccspline</td>
<td>ccspline</td>
</tr>
<tr>
<td>Hunt Group Lookup Notification</td>
<td>None</td>
<td>None</td>
</tr>
<tr>
<td>Speed Dial Auto-Fill Digit</td>
<td>False</td>
<td>False</td>
</tr>
<tr>
<td>Display CTI Route Point Name or DN</td>
<td>False</td>
<td>False</td>
</tr>
<tr>
<td>Display Original Calling Number on Transfer from Cisco Phone</td>
<td>False</td>
<td>False</td>
</tr>
<tr>
<td>URI Dialing Display Preference</td>
<td>DN</td>
<td>DN</td>
</tr>
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</table>

Figure 9 Service Parameter (cont.)
### Service Parameter (cont.)

#### Clusterwide Parameters (Device - PRI and MGCP Gateway)

<table>
<thead>
<tr>
<th>Parameter</th>
<th>Description</th>
<th>Value 1</th>
<th>Value 2</th>
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<tbody>
<tr>
<td>CPNSI</td>
<td>Calling Party Number Screening Indicator</td>
<td>False</td>
<td>True</td>
</tr>
<tr>
<td>ONSRI</td>
<td>Enable Outbound NetworkTrunk CallingParty Restriction</td>
<td>False</td>
<td>True</td>
</tr>
<tr>
<td>CDRD</td>
<td>Clear Calls Flag When Data Link Is Down</td>
<td>True</td>
<td>True</td>
</tr>
<tr>
<td>DSAI</td>
<td>Device Status Poll Interval</td>
<td>3000</td>
<td>3000</td>
</tr>
<tr>
<td>DAII</td>
<td>Disable Alerting Progress Indicator</td>
<td>False</td>
<td>True</td>
</tr>
<tr>
<td>DNPH</td>
<td>Discard Non Inband Progress in Overlap Sending</td>
<td>False</td>
<td>True</td>
</tr>
<tr>
<td>DRSM</td>
<td>Disable Resume from Shared-line MGCP FXS Port</td>
<td>True</td>
<td>False</td>
</tr>
<tr>
<td>DST</td>
<td>DTMF Silence Tone Rug</td>
<td>False</td>
<td>True</td>
</tr>
<tr>
<td>EDIE</td>
<td>Enable Display IE in Codecset 6</td>
<td>False</td>
<td>True</td>
</tr>
<tr>
<td>ESF</td>
<td>Enable Sending PRI N1 Service Messages</td>
<td>False</td>
<td>True</td>
</tr>
<tr>
<td>FSHD</td>
<td>Flash Hook Duration</td>
<td>600</td>
<td>1000</td>
</tr>
<tr>
<td>GPT</td>
<td>Gateway Poll Timer</td>
<td>10</td>
<td>10</td>
</tr>
<tr>
<td>MUPI</td>
<td>Matching Calling Party with Attendant Flag</td>
<td>False</td>
<td>True</td>
</tr>
<tr>
<td>MDQD</td>
<td>MGCP Database Query Delay Timer</td>
<td>1000</td>
<td>1000</td>
</tr>
<tr>
<td>PMSO</td>
<td>MGCP Fks On-Hook Pending Timer</td>
<td>3</td>
<td>3</td>
</tr>
<tr>
<td>MRST</td>
<td>MGCP Response Timer</td>
<td>30</td>
<td>30</td>
</tr>
<tr>
<td>MT</td>
<td>MGCP Timer</td>
<td>3</td>
<td>3</td>
</tr>
<tr>
<td>NPI</td>
<td>Numbering Plan Info</td>
<td>1</td>
<td>1</td>
</tr>
<tr>
<td>ORF</td>
<td>Overlap Receiving Flag for PRI</td>
<td>True</td>
<td>True</td>
</tr>
<tr>
<td>OMCTD</td>
<td>Outgoing Media Connect Time for PRI</td>
<td>Connect ASAP</td>
<td>Connect ASAP</td>
</tr>
<tr>
<td>OR</td>
<td>Port Release Timer</td>
<td>0</td>
<td>0</td>
</tr>
<tr>
<td>SMCCD</td>
<td>SMCC Call Delay Timer</td>
<td>0</td>
<td>0</td>
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<tr>
<td>STS</td>
<td>Stable in State 4 Flag</td>
<td>False</td>
<td>True</td>
</tr>
<tr>
<td>MMGR</td>
<td>Optimize MGCP Registration</td>
<td>False</td>
<td>True</td>
</tr>
<tr>
<td>SOCA</td>
<td>Suppress Out-of-Channels Alarms</td>
<td>True</td>
<td>True</td>
</tr>
<tr>
<td>LT</td>
<td>LI-Frame Timer</td>
<td>2000</td>
<td>2000</td>
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<tr>
<td>U2UI</td>
<td>User-to-User IE Status</td>
<td>False</td>
<td>True</td>
</tr>
<tr>
<td>CEPM</td>
<td>Convert European Progress Message to Alerting</td>
<td>False</td>
<td>True</td>
</tr>
<tr>
<td>ECPM</td>
<td>Enable E1 PRI Notify Message from User to Network</td>
<td>True</td>
<td>True</td>
</tr>
<tr>
<td>AGC</td>
<td>Audit OCS Channels Interval</td>
<td>10</td>
<td>10</td>
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<tr>
<td>DAPN</td>
<td>Digital and Analog Ports Enabled</td>
<td>True</td>
<td>True</td>
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</table>

There are hidden parameters in this group. Click on Advanced button to see hidden parameters.

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Figure 10 Service Parameter (cont.)
### Figure 11 Service Parameter (cont.)

<table>
<thead>
<tr>
<th>Parameter</th>
<th>Value</th>
</tr>
</thead>
<tbody>
<tr>
<td>Call Present Disconnect Flag</td>
<td>False</td>
</tr>
<tr>
<td>Check Progress Indicator Before Establishing Media</td>
<td>False</td>
</tr>
<tr>
<td>H223 Block Setup Destination</td>
<td>False</td>
</tr>
<tr>
<td>H223 OSC Retry Timer</td>
<td>False</td>
</tr>
<tr>
<td>H223 Device Connect Timer</td>
<td>False</td>
</tr>
<tr>
<td>H223 OTMF Duration</td>
<td>100</td>
</tr>
<tr>
<td>H223 TopRef Retry</td>
<td>2</td>
</tr>
<tr>
<td>H223 Intercluster Call Throttle Timer</td>
<td>30</td>
</tr>
<tr>
<td>H225 T301 Timer</td>
<td>180000</td>
</tr>
<tr>
<td>H225 T302 Timer</td>
<td>15000</td>
</tr>
<tr>
<td>H225 T303 Timer</td>
<td>4000</td>
</tr>
<tr>
<td>H225 T304 Timer</td>
<td>30000</td>
</tr>
<tr>
<td>H225 T305 Timer</td>
<td>30000</td>
</tr>
<tr>
<td>H225 T310 Timer</td>
<td>60000</td>
</tr>
<tr>
<td>H225 TCP Timeout</td>
<td>5</td>
</tr>
<tr>
<td>H245 TCS Timeout</td>
<td>10</td>
</tr>
<tr>
<td>H323 Calling Party Number Screening Indicator</td>
<td>False</td>
</tr>
<tr>
<td>Apply External Phone Number Mask for H.323 Calls</td>
<td>False</td>
</tr>
<tr>
<td>Tone on Connect</td>
<td>False</td>
</tr>
<tr>
<td>Wait Time for G.722 with SD/RO Mode</td>
<td>3</td>
</tr>
<tr>
<td>RAS ARQ Timer</td>
<td>3</td>
</tr>
<tr>
<td>RAS RBO Timer</td>
<td>3</td>
</tr>
<tr>
<td>RAS DRQ Timer</td>
<td>3</td>
</tr>
<tr>
<td>RAS ARQ Timer</td>
<td>3</td>
</tr>
<tr>
<td>RAS RBO Timer</td>
<td>3</td>
</tr>
<tr>
<td>RAS DRQ Timer</td>
<td>3</td>
</tr>
<tr>
<td>Retry Count for ARQ</td>
<td>2</td>
</tr>
<tr>
<td>Retry Count for RBO</td>
<td>2</td>
</tr>
<tr>
<td>Retry Count for DRQ</td>
<td>2</td>
</tr>
<tr>
<td>Retry Count for ARQ</td>
<td>2</td>
</tr>
<tr>
<td>Retry Count for RBO</td>
<td>2</td>
</tr>
<tr>
<td>Retry Count for DRQ</td>
<td>2</td>
</tr>
<tr>
<td>CAB Product ID and Version ID</td>
<td>False</td>
</tr>
<tr>
<td>CAB Unified CM Version or Version ID in H225Setup</td>
<td>False</td>
</tr>
<tr>
<td>Send Progress Timer</td>
<td>30000</td>
</tr>
<tr>
<td>Send H225 User Info Message</td>
<td>False</td>
</tr>
<tr>
<td>Status Inquiry Poll Timer</td>
<td>False</td>
</tr>
<tr>
<td>Device Name of DRU-controlled Truck That Will Use Port 1720</td>
<td>None</td>
</tr>
<tr>
<td>Host Name/IP Address of DRU That Will Use RAS UDP Port 1720</td>
<td>None</td>
</tr>
<tr>
<td>cell_H245 MTP Allocation Fail</td>
<td>False</td>
</tr>
<tr>
<td>Overload Receiving Flag for H323</td>
<td>False</td>
</tr>
<tr>
<td>Allocate Transcoder for H.323 on Early Offer SIP Trunk for Calls with Early Media</td>
<td>False</td>
</tr>
</tbody>
</table>

There are hidden parameters in this group. Click on Advanced button to see hidden parameters.
### Clusterwide Parameters (Device - SIP)

<table>
<thead>
<tr>
<th>Parameter</th>
<th>Value 1</th>
<th>Value 2</th>
</tr>
</thead>
<tbody>
<tr>
<td>SIP Interoperability Enabled</td>
<td>True</td>
<td>True</td>
</tr>
<tr>
<td>Retry Count for SIP Bye</td>
<td>10</td>
<td>10</td>
</tr>
<tr>
<td>Retry Count for SIP Cancel</td>
<td>10</td>
<td>10</td>
</tr>
<tr>
<td>Retry Count for SIP Invite</td>
<td>6</td>
<td>6</td>
</tr>
<tr>
<td>Retry Count for SIP PRACK</td>
<td>6</td>
<td>6</td>
</tr>
<tr>
<td>Retry Count for SIP Reinvite</td>
<td>10</td>
<td>10</td>
</tr>
<tr>
<td>Retry Count for SIP Publish</td>
<td>6</td>
<td>6</td>
</tr>
<tr>
<td>SIP Connected Timer</td>
<td>500</td>
<td>500</td>
</tr>
<tr>
<td>SIP Disconnect Timer</td>
<td>500</td>
<td>500</td>
</tr>
<tr>
<td>SIP Expires Timer</td>
<td>180000</td>
<td>180000</td>
</tr>
<tr>
<td>SIP PRACK Timer</td>
<td>600</td>
<td>600</td>
</tr>
<tr>
<td>SIP Reinvite Timer</td>
<td>500</td>
<td>500</td>
</tr>
<tr>
<td>SIP Trunk Timer</td>
<td>500</td>
<td>500</td>
</tr>
<tr>
<td>SIP Publish Timer</td>
<td>500</td>
<td>500</td>
</tr>
<tr>
<td>SIP Min-SE Value</td>
<td>1800</td>
<td>1800</td>
</tr>
<tr>
<td>SIP URI Handling</td>
<td>Reject</td>
<td>Reject</td>
</tr>
<tr>
<td>SIP statistics Periodic update Timer</td>
<td>2</td>
<td>2</td>
</tr>
<tr>
<td>SIP Session Expire Timer</td>
<td>1800</td>
<td>1800</td>
</tr>
<tr>
<td>SIP Trunk TopRef Retry</td>
<td>2</td>
<td>2</td>
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</table>

### Clusterwide Parameters (Feature - General)

<table>
<thead>
<tr>
<th>Parameter</th>
<th>Value 1</th>
<th>Value 2</th>
</tr>
</thead>
<tbody>
<tr>
<td>Call Park Display Timer</td>
<td>10</td>
<td>10</td>
</tr>
<tr>
<td>Caller ID Display Priority Enabled</td>
<td>True</td>
<td>True</td>
</tr>
<tr>
<td>Call Park Reversion Timer</td>
<td>60</td>
<td>60</td>
</tr>
<tr>
<td>Park Monitoring Reversion Timer</td>
<td>60</td>
<td>60</td>
</tr>
<tr>
<td>Park Monitoring Period Reversion Timer</td>
<td>30</td>
<td>30</td>
</tr>
<tr>
<td>Park Monitoring Forward No Retrieve Timer</td>
<td>300</td>
<td>300</td>
</tr>
</tbody>
</table>

---

Figure 12 Service Parameter (cont.)
Figure 13 Service Parameter (cont.)

<table>
<thead>
<tr>
<th>Parameter Description</th>
<th>Value</th>
</tr>
</thead>
<tbody>
<tr>
<td>Preserve outbound CID for Parked Calls</td>
<td>True</td>
</tr>
<tr>
<td>Maximum Call Duration Timer</td>
<td>720</td>
</tr>
<tr>
<td>Maximum Hold Duration Timer</td>
<td>360</td>
</tr>
<tr>
<td>Party Entrance Tone</td>
<td>True</td>
</tr>
<tr>
<td>Message Waiting Lamp Policy</td>
<td>Primary Line - Light and Prompt</td>
</tr>
<tr>
<td>Audible Message Waiting Indication Policy</td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td>Message Waiting Indicator Inbound Calling Search Space</td>
<td>False</td>
</tr>
<tr>
<td>MWI Non Message Center Signaling Call Duration</td>
<td>False</td>
</tr>
<tr>
<td>Message Waiting Indicator RcdU E164 Translation CSS</td>
<td>False</td>
</tr>
<tr>
<td>Block Offset To Offset Transfer</td>
<td>False</td>
</tr>
<tr>
<td>Use Original Call Classification for Transferred Calls</td>
<td>True</td>
</tr>
<tr>
<td>Use Redirect attribute of ID/Name Presentation of Transferring Party</td>
<td>False</td>
</tr>
<tr>
<td>Local route group for redirected calls</td>
<td>False</td>
</tr>
<tr>
<td>Block Unencrypted Calls</td>
<td>False</td>
</tr>
<tr>
<td>Suppress MNT to Conference Bridge</td>
<td>True</td>
</tr>
<tr>
<td>Drop Ad Hoc Conference</td>
<td>Never</td>
</tr>
<tr>
<td>Maximum Ad Hoc Conference</td>
<td>4</td>
</tr>
<tr>
<td>Maximum Meetme Conference Unicast</td>
<td>4</td>
</tr>
<tr>
<td>Advanced Ad Hoc Conference Enabled</td>
<td>False</td>
</tr>
<tr>
<td>Choose Encrypted Audio Conference Instead Of Video Conference</td>
<td>True</td>
</tr>
<tr>
<td>Minimum Video Capable Participants To Allocate Video Conference</td>
<td>2</td>
</tr>
<tr>
<td>Enable Click-to-Conference for Third-Party Applications</td>
<td>False</td>
</tr>
<tr>
<td>IMS Conference Factory Unit</td>
<td><a href="mailto:cucm-conference-factory@cucm1.company.com">cucm-conference-factory@cucm1.company.com</a></td>
</tr>
<tr>
<td>Cluster Conference Prefix Identifier</td>
<td>False</td>
</tr>
<tr>
<td>Secure Call Icon Display Policy</td>
<td>All media except BCP and IX transports must be encryted</td>
</tr>
<tr>
<td>Forward Maximum Hop Count</td>
<td>12</td>
</tr>
<tr>
<td>Forward No Answer Timer</td>
<td>12</td>
</tr>
<tr>
<td>Max Forward Hops to DN</td>
<td>12</td>
</tr>
<tr>
<td>Return Forward Information</td>
<td>True</td>
</tr>
<tr>
<td>Forward By Recroute Enabled</td>
<td>True</td>
</tr>
<tr>
<td>Transform Forward by Recroute Detection</td>
<td>True</td>
</tr>
<tr>
<td>Always Forward Switch Voice Mail Calls</td>
<td>True</td>
</tr>
<tr>
<td>Forward By Recroute T1 Timer</td>
<td>10</td>
</tr>
<tr>
<td>Include Original Called Info for D-50G Call Diversions</td>
<td>False</td>
</tr>
<tr>
<td>Set Private Numbering Plan for Call Forward</td>
<td>False</td>
</tr>
<tr>
<td>Set Type of Number for Call Forward</td>
<td>Level1RegionalNumber</td>
</tr>
</tbody>
</table>

There are hidden parameters in this group. Click on Advanced button to see hidden parameters.
### Figure 14 Service Parameter (cont.)

<table>
<thead>
<tr>
<th>Clusterwide Parameters (Feature - Hold Reversion)</th>
</tr>
</thead>
<tbody>
<tr>
<td>Hold Reversion Duration *</td>
</tr>
<tr>
<td>Hold Reversion Notification Interval *</td>
</tr>
<tr>
<td>CPA Destination Override *</td>
</tr>
</tbody>
</table>

<table>
<thead>
<tr>
<th>Clusterwide Parameters (Feature - Call Pickup)</th>
</tr>
</thead>
<tbody>
<tr>
<td>Auto Call Pickup Enabled *</td>
</tr>
<tr>
<td>Call Pickup Locating Timer *</td>
</tr>
<tr>
<td>Call Pickup No Answer Timeout *</td>
</tr>
</tbody>
</table>

<table>
<thead>
<tr>
<th>Clusterwide Parameters (Feature - Redirect [3xx])</th>
</tr>
</thead>
<tbody>
<tr>
<td>Redirection Ring No Answer Reversion Timer *</td>
</tr>
<tr>
<td>Maximum Redirect Count *</td>
</tr>
</tbody>
</table>

<table>
<thead>
<tr>
<th>Clusterwide Parameters (Feature - Multilevel Precedence and Preemption)</th>
</tr>
</thead>
<tbody>
<tr>
<td>Location-based MPLP Enable *</td>
</tr>
<tr>
<td>Executive Override Call Preemptable *</td>
</tr>
<tr>
<td>Location-based Maximum Bandwidth Enforcement Level for MPLP CSS *</td>
</tr>
<tr>
<td>Non-Preemption Pattern CSS *</td>
</tr>
<tr>
<td>MPLP Exceedance Level *</td>
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</tbody>
</table>

<table>
<thead>
<tr>
<th>Clusterwide Parameters (Feature - Path Replacement)</th>
</tr>
</thead>
<tbody>
<tr>
<td>Path Replacement Enabled *</td>
</tr>
<tr>
<td>Path Replacement on Terminated Calls *</td>
</tr>
<tr>
<td>Start Path Replacement Minimum Delay Time *</td>
</tr>
<tr>
<td>Start Path Replacement Maximum Delay Time *</td>
</tr>
<tr>
<td>Path Replacement TTL Timer *</td>
</tr>
<tr>
<td>Path Replacement T2 Timer *</td>
</tr>
<tr>
<td>Path Replacement PINX ID *</td>
</tr>
<tr>
<td>Path Replacement Calling Search Space *</td>
</tr>
</tbody>
</table>

<table>
<thead>
<tr>
<th>Clusterwide Parameters (Feature - Call Back)</th>
</tr>
</thead>
<tbody>
<tr>
<td>Call Back Enabled Flag *</td>
</tr>
<tr>
<td>Call Back Notification Audio File Name *</td>
</tr>
<tr>
<td>Connection Proposal Type *</td>
</tr>
<tr>
<td>Connection Resource Type *</td>
</tr>
<tr>
<td>Call Back Requested T1 Timer *</td>
</tr>
<tr>
<td>Call Back Recall T3 Timer *</td>
</tr>
</tbody>
</table>

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<table>
<thead>
<tr>
<th>Parameter &amp; Value</th>
<th>Value</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>Call Back Calling Search Space</td>
<td>&lt; None &gt;</td>
<td></td>
</tr>
<tr>
<td>No Path Reservation</td>
<td>True</td>
<td></td>
</tr>
<tr>
<td>Set Private Numbering Plan for Call Back</td>
<td>False</td>
<td></td>
</tr>
<tr>
<td>Set Type of Number for Call Back</td>
<td>Level1RegionalNumber</td>
<td></td>
</tr>
</tbody>
</table>

There are hidden parameters in this group. Click on the Advanced button to see hidden parameters.

### Clusterwide Parameters (Feature - Call Recording)

<table>
<thead>
<tr>
<th>Parameter &amp; Value</th>
<th>Value</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>Play Recording Notification Tone To Observed Target</td>
<td>False</td>
<td></td>
</tr>
<tr>
<td>Play Recording Notification Tone To Observed Connected Parties</td>
<td>False</td>
<td></td>
</tr>
</tbody>
</table>

### Clusterwide Parameters (Feature - Monitoring)

<table>
<thead>
<tr>
<th>Parameter &amp; Value</th>
<th>Value</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>Play Monitoring Notification Tone To Observed Target</td>
<td>False</td>
<td></td>
</tr>
<tr>
<td>Play Monitoring Notification Tone To Observed Connected Parties</td>
<td>False</td>
<td></td>
</tr>
</tbody>
</table>

### Clusterwide Parameters (Feature - Join Across Lines And Single Button Barge Feature Set)

<table>
<thead>
<tr>
<th>Parameter &amp; Value</th>
<th>Value</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>Join Across Lines Policy</td>
<td>Off</td>
<td></td>
</tr>
<tr>
<td>Single Button Barge/Charge Policy</td>
<td>Off</td>
<td></td>
</tr>
<tr>
<td>Allow Barging When Ringing</td>
<td>False</td>
<td></td>
</tr>
</tbody>
</table>

### Clusterwide Parameters (Feature - Secure Tone)

<table>
<thead>
<tr>
<th>Parameter &amp; Value</th>
<th>Value</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>Play Tone To Indicate Secure/Non-Secure Call Status</td>
<td>False</td>
<td></td>
</tr>
</tbody>
</table>

There are hidden parameters in this group. Click on the Advanced button to see hidden parameters.

### Clusterwide Parameters (Feature - External Call Control)

<table>
<thead>
<tr>
<th>Parameter &amp; Value</th>
<th>Value</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>External Call Control Diversion Maximum Hop Count</td>
<td>12</td>
<td></td>
</tr>
<tr>
<td>External Call Control Diversion Hop to Pattern or Dn</td>
<td>12</td>
<td></td>
</tr>
<tr>
<td>External Call Control Routing Request Timer</td>
<td>2000</td>
<td></td>
</tr>
<tr>
<td>External Call Control Fully Qualified Role and Resource</td>
<td>CISCO:UC:UCMPolicy:VoiceOrVideoCall</td>
<td></td>
</tr>
<tr>
<td>External Call Control Initial Connection Count To POP</td>
<td>2</td>
<td></td>
</tr>
<tr>
<td>External Call Control Maximum Connection Count To POP</td>
<td>4</td>
<td></td>
</tr>
<tr>
<td>Always Use External Call Control-specified Called/Calling Party Names</td>
<td>True</td>
<td></td>
</tr>
</tbody>
</table>

### Clusterwide Parameters (Route Plan)

<table>
<thead>
<tr>
<th>Parameter &amp; Value</th>
<th>Value</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>Stop Routing on Out of Bandwidth Flap</td>
<td>False</td>
<td></td>
</tr>
<tr>
<td>Stop Routing on Unallocated Number Flap</td>
<td>False</td>
<td></td>
</tr>
<tr>
<td>Stop Routing on User Busy Flap</td>
<td>True</td>
<td></td>
</tr>
</tbody>
</table>

There are hidden parameters in this group. Click on the Advanced button to see hidden parameters.

### Clusterwide Parameters (Route Class Signaling)

<table>
<thead>
<tr>
<th>Parameter &amp; Value</th>
<th>Value</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>Route Class Trunk Signaling Enabled</td>
<td>True</td>
<td></td>
</tr>
<tr>
<td>SIP Route Class Naming Authority</td>
<td>cisco.com</td>
<td></td>
</tr>
</tbody>
</table>

There are hidden parameters in this group. Click on the Advanced button to see hidden parameters.

### Clusterwide Parameters (Hunt List)

<table>
<thead>
<tr>
<th>Parameter &amp; Value</th>
<th>Value</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>Stop Hunting on Out of Bandwidth Flap</td>
<td>False</td>
<td></td>
</tr>
</tbody>
</table>

Figure 15 Service Parameter (cont.)
Figure 16 Service Parameter (cont.)
<table>
<thead>
<tr>
<th>Parameter</th>
<th>Value</th>
</tr>
</thead>
<tbody>
<tr>
<td>DSCP for QoS Clear Cells</td>
<td>46 (101110)</td>
</tr>
<tr>
<td>DSCP for Priority QoS Clear Cells</td>
<td>45 (101101)</td>
</tr>
<tr>
<td>DSCP for Immediate QoS Clear Cells</td>
<td>44 (101100)</td>
</tr>
<tr>
<td>DSCP for Flash Override QoS Clear Cells</td>
<td>41 (101001)</td>
</tr>
<tr>
<td>DSCP for Executive Override QoS Clear Cells</td>
<td>42 (101010)</td>
</tr>
<tr>
<td>DSCP for Audio Cells when RSVP Fails</td>
<td>0 (000000)</td>
</tr>
<tr>
<td>DSCP for Video Cells when RSVP Fails</td>
<td>0 (000000)</td>
</tr>
<tr>
<td>DSCP for ICCP Protocol Links</td>
<td>24 (011000)</td>
</tr>
</tbody>
</table>

<table>
<thead>
<tr>
<th>Parameter</th>
<th>Value</th>
</tr>
</thead>
<tbody>
<tr>
<td>SDL Listening Port Number</td>
<td>8002</td>
</tr>
<tr>
<td>SDL Max Router Latency</td>
<td>20</td>
</tr>
<tr>
<td>Suppress Debug Info for Router Death</td>
<td>0 (000000)</td>
</tr>
<tr>
<td>Asynchronous SDL Looping Enabled</td>
<td>False</td>
</tr>
</tbody>
</table>

<table>
<thead>
<tr>
<th>Parameter</th>
<th>Value</th>
</tr>
</thead>
<tbody>
<tr>
<td>Enforce Millisecond Packet Size</td>
<td>True</td>
</tr>
<tr>
<td>Locations Trace Details Enabled</td>
<td>False</td>
</tr>
<tr>
<td>Preferred G.711 Millisecond Packet Size</td>
<td>20</td>
</tr>
<tr>
<td>Preferred G.722 Millisecond Packet Size</td>
<td>20</td>
</tr>
<tr>
<td>Preferred G.723.1 Millisecond Packet Size</td>
<td>30</td>
</tr>
<tr>
<td>Preferred G.729 Millisecond Packet Size</td>
<td>20</td>
</tr>
<tr>
<td>Always Use Preferred G.729 Packet Size For SIP Trunk Sessions</td>
<td>False</td>
</tr>
</tbody>
</table>

<table>
<thead>
<tr>
<th>Parameter</th>
<th>Value</th>
</tr>
</thead>
<tbody>
<tr>
<td>Preferred G.711 Codec Enabled</td>
<td>Enabled for All Devices</td>
</tr>
<tr>
<td>G.711 nu-law Codec Enabled</td>
<td>Enabled for All Devices</td>
</tr>
<tr>
<td>G.722 Codec Enabled</td>
<td>Enabled for All Devices</td>
</tr>
<tr>
<td>LPC Codec Enabled</td>
<td>Enabled for All Devices</td>
</tr>
<tr>
<td>GAC Codec Enabled</td>
<td>Enabled for All Devices</td>
</tr>
<tr>
<td>Default Interregion Max Audio Bit Rate</td>
<td>64 kbps (G.722, G.711)</td>
</tr>
<tr>
<td>Default Interregion Max Audio Bit Rate</td>
<td>8 kbps (G.729)</td>
</tr>
<tr>
<td>Default Interregion Max Video Call Bit Rate (Includes Audio)</td>
<td>364 (394)</td>
</tr>
<tr>
<td>Default Interregion Max Immersive Video Call Bit Rate (Includes Audio)</td>
<td>200000000000 (20000000000)</td>
</tr>
<tr>
<td>Use Video Bandwidth Pool for Immersive Video Cells</td>
<td>True</td>
</tr>
<tr>
<td>Default Interregion and Interregion Link Loss Type</td>
<td>Low Loss</td>
</tr>
<tr>
<td>Default Audio Codec List between Regions</td>
<td>Factory Default low loss</td>
</tr>
<tr>
<td>Default audio Codec List within Region</td>
<td>Factory Default low loss</td>
</tr>
<tr>
<td>Accept Audio Codec Preferences in Received Offer</td>
<td>Off</td>
</tr>
<tr>
<td>G.723 Bandwidth Override</td>
<td>False</td>
</tr>
</tbody>
</table>

<table>
<thead>
<tr>
<th>Parameter</th>
<th>Value</th>
</tr>
</thead>
<tbody>
<tr>
<td>Automated Alternate Routing Enable</td>
<td>False</td>
</tr>
<tr>
<td>Default Inter-location RSVP Policy</td>
<td>No Reservation</td>
</tr>
</tbody>
</table>

Figure 17 Service Parameter (cont.)
<table>
<thead>
<tr>
<th>Parameter</th>
<th>Value 1</th>
<th>Value 2</th>
</tr>
</thead>
<tbody>
<tr>
<td>RSVP Retry Timer</td>
<td>60</td>
<td>60</td>
</tr>
<tr>
<td>Mandatory RSVP Mid-call Retry Count</td>
<td>1</td>
<td>1</td>
</tr>
<tr>
<td>Mandatory RSVP mid-call error handle option</td>
<td>Call becomes best effort</td>
<td>Call becomes best effort</td>
</tr>
<tr>
<td>RSVP Video Time Spec Burst Size Factor</td>
<td>5</td>
<td>5</td>
</tr>
<tr>
<td>NLSP EXECUTIVE OVERRIDE To RSVP Priority Mapping</td>
<td>65535</td>
<td>65535</td>
</tr>
<tr>
<td>NLSP PLINK OVERRIDE To RSVP Priority Mapping</td>
<td>65534</td>
<td>65534</td>
</tr>
<tr>
<td>NLSP PLINK To RSVP Priority Mapping</td>
<td>65533</td>
<td>65533</td>
</tr>
<tr>
<td>NLSP IMMEDIATE To RSVP Priority Mapping</td>
<td>65532</td>
<td>65532</td>
</tr>
<tr>
<td>NLSP PL PRIORITY To RSVP Priority Mapping</td>
<td>65531</td>
<td>65531</td>
</tr>
<tr>
<td>NLSP PL ROUTINE To RSVP Priority Mapping</td>
<td>65530</td>
<td>65530</td>
</tr>
<tr>
<td>RSVP Audio Application ID</td>
<td>AudioStream</td>
<td>AudioStream</td>
</tr>
<tr>
<td>RSVP Video Application ID</td>
<td>VideoStream</td>
<td>VideoStream</td>
</tr>
<tr>
<td>RSVP Response Timer</td>
<td>2</td>
<td>2</td>
</tr>
</tbody>
</table>

**TLS Packet Capture Configurations**

<table>
<thead>
<tr>
<th>Parameter</th>
<th>Value 1</th>
<th>Value 2</th>
</tr>
</thead>
<tbody>
<tr>
<td>Packet Capture Enable</td>
<td>False</td>
<td>False</td>
</tr>
<tr>
<td>Packet Capture Max File Size (MB)</td>
<td>2</td>
<td>2</td>
</tr>
</tbody>
</table>

**Clusterwide Parameters (System - Presence)**

<table>
<thead>
<tr>
<th>Parameter</th>
<th>Value 1</th>
<th>Value 2</th>
</tr>
</thead>
<tbody>
<tr>
<td>Presence Subscription Throttling Threshold</td>
<td>60000</td>
<td>60000</td>
</tr>
<tr>
<td>Presence Subscription Resume Threshold</td>
<td>60</td>
<td>60</td>
</tr>
<tr>
<td>Default Inter-Presence Group Subscription</td>
<td>Disable Subscription</td>
<td>Disable Subscription</td>
</tr>
<tr>
<td>EPL Status Depicts DNS</td>
<td>False</td>
<td>False</td>
</tr>
</tbody>
</table>

**Clusterwide Parameters (System - Mobility)**

<table>
<thead>
<tr>
<th>Parameter</th>
<th>Value 1</th>
<th>Value 2</th>
</tr>
</thead>
<tbody>
<tr>
<td>Enterprise Feature Access Code for Hold</td>
<td>*81</td>
<td>*81</td>
</tr>
<tr>
<td>Enterprise Feature Access Code for Exclusive Hold</td>
<td>*82</td>
<td>*82</td>
</tr>
<tr>
<td>Enterprise Feature Access Code for Resume</td>
<td>*83</td>
<td>*83</td>
</tr>
<tr>
<td>Enterprise Feature Access Code for Transfer</td>
<td>*84</td>
<td>*84</td>
</tr>
<tr>
<td>Enterprise Feature Access Code for Conference</td>
<td>*85</td>
<td>*85</td>
</tr>
<tr>
<td>Enterprise Feature Access Code for Session Handoff</td>
<td>*74</td>
<td>*74</td>
</tr>
<tr>
<td>Enterprise Feature Access Code for Starting Selective Recording</td>
<td>*86</td>
<td>*86</td>
</tr>
<tr>
<td>Enterprise Feature Access Code for Stopping Selective Recording</td>
<td>*87</td>
<td>*87</td>
</tr>
<tr>
<td>Smart Mobile Phone Interdigit Timer</td>
<td>5000</td>
<td>5000</td>
</tr>
<tr>
<td>Non-Smart Mobile Phone Interdigit Timer</td>
<td>2000</td>
<td>2000</td>
</tr>
<tr>
<td>Send Call to Mobile Menu Timer</td>
<td>60</td>
<td>60</td>
</tr>
<tr>
<td>SIP Dual Mode Alert Timer</td>
<td>1500</td>
<td>1500</td>
</tr>
<tr>
<td>Call Screening Timer</td>
<td>4000</td>
<td>4000</td>
</tr>
<tr>
<td>Session Resumption Wait Timer</td>
<td>180</td>
<td>180</td>
</tr>
<tr>
<td>Inbound Calling Search Space for Remote Destination</td>
<td>Trunk or Gateway Inbound Calling Search Space</td>
<td>Trunk or Gateway Inbound Calling Search Space</td>
</tr>
<tr>
<td>Enable Enterprise Feature Access</td>
<td>False</td>
<td>False</td>
</tr>
<tr>
<td>Dial-voice-Office Forward Service Access Number</td>
<td>False</td>
<td>False</td>
</tr>
<tr>
<td>Enable Mobile Voice Access</td>
<td>False</td>
<td>False</td>
</tr>
<tr>
<td>Mobile Voice Access Number</td>
<td>10</td>
<td>10</td>
</tr>
</tbody>
</table>

Figure 18 Service Parameter (cont.)
<table>
<thead>
<tr>
<th>Parameter</th>
<th>System Remote Access Blocked Numbers</th>
<th>Enable Use of Called Party Transformed Number for Mobile-Derived Calls</th>
<th>False</th>
<th>False</th>
</tr>
</thead>
<tbody>
<tr>
<td></td>
<td></td>
<td>Host Gateway or Trunk Outbound Calling Party Selection for Mobile Connect Calls</td>
<td>False</td>
<td>False</td>
</tr>
<tr>
<td>Clusterwide Parameters (System - Mobility Single Number Reach Voicemail)</td>
<td></td>
<td>Timer Control</td>
<td>Timer Control</td>
<td></td>
</tr>
<tr>
<td></td>
<td></td>
<td>Timer Control</td>
<td>Timer Control</td>
<td></td>
</tr>
<tr>
<td></td>
<td></td>
<td>User Control Delayed Announcement Timer *</td>
<td>1000</td>
<td>1000</td>
</tr>
<tr>
<td></td>
<td></td>
<td>User Control Confirmed Answer Indication Timer *</td>
<td>10000</td>
<td>10000</td>
</tr>
<tr>
<td>Clusterwide Parameters (Feature - Reroute Remote Destination Calls to Enterprise Number)</td>
<td></td>
<td>Reroute Remote Destination Calls to Enterprise Number *</td>
<td>False</td>
<td>False</td>
</tr>
<tr>
<td></td>
<td></td>
<td>Ring All Shared Lines *</td>
<td>False</td>
<td>False</td>
</tr>
<tr>
<td></td>
<td></td>
<td>Transfer Call Forward All on Enterprise DN *</td>
<td>True</td>
<td>True</td>
</tr>
<tr>
<td>Clusterwide Parameters (Feature - Immediate Divert)</td>
<td></td>
<td>Use Legacy Immediate Divert *</td>
<td>True</td>
<td>True</td>
</tr>
<tr>
<td></td>
<td></td>
<td>Allow OS10 during iDivert *</td>
<td>False</td>
<td>False</td>
</tr>
<tr>
<td></td>
<td></td>
<td>Immediate Divert User Response Timer *</td>
<td>5</td>
<td>5</td>
</tr>
<tr>
<td>Clusterwide Parameters (Call Admission Control)</td>
<td></td>
<td>Call Counting CAC Enabled *</td>
<td>False</td>
<td>False</td>
</tr>
<tr>
<td></td>
<td></td>
<td>Audio Bandwidth For Call Counting CAC *</td>
<td>102</td>
<td>102</td>
</tr>
<tr>
<td></td>
<td></td>
<td>Video Bandwidth For Call Counting CAC *</td>
<td>500</td>
<td>500</td>
</tr>
</tbody>
</table>

Figure 19 Service Parameter (cont.)
Off-net calls via MetTel SIP Trunk

Off-net calls are served by SIP trunks configured between Cisco UCM and MetTel network and Calls are routed via Cisco UBE.

**SIP Trunk Security Profile**

**Navigation Path:** System -> Security -> SIP Trunk Security Profile

---

**Parameter** | **Value** | **Description**
--- | --- | ---
Incoming Transport Type | TCP + UDP | 
Outgoing Transport Type | UDP | SIP trunks to MetTel ESBC should use UDP as a transport protocol for SIP. This is configured using SIP Trunk Security profile, which is later assigned to the SIP trunk itself.

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SIP Profile Configuration

SIP Profile will be later associated with the SIP trunk.

Navigation Path: Device-> Device Settings-> SIP Profile

Figure 21 SIP Profile
Figure 22 SIP Profile (cont.)
Figure 23 SIP Profile (cont.)

<table>
<thead>
<tr>
<th>Parameter</th>
<th>Value</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>Default MTP Telephony Event Payload Type</td>
<td>101</td>
<td>RFC2833 DTMF payload type</td>
</tr>
<tr>
<td>SIP Re1XX Options</td>
<td>Send PRACK for 1xx Messages</td>
<td>Enable Provisional Acknowledgements (Reliable 100 messages)</td>
</tr>
<tr>
<td>Ping Interval for In-service and Partially In-service Trunks (seconds)</td>
<td>30</td>
<td>OPTIONS message parameters- interval time</td>
</tr>
<tr>
<td>Ping Interval for Out-of-service Trunks (seconds)</td>
<td>120</td>
<td>OPTIONS message parameters- interval time</td>
</tr>
</tbody>
</table>
SIP Trunk Configuration
Create SIP trunks to Cisco UBE. Create SIP trunks similarly to Cisco Unity Connection and Cisco Fax Gateway.

**Navigation:** Device-> Trunk

![SIP Trunks List](image)

**Figure 24 SIP Trunks List**
Figure 25 SIP Trunk to Cisco UBE
Figure 26 SIP Trunk to Cisco UBE (cont.)
<table>
<thead>
<tr>
<th>Parameter</th>
<th>Value</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>Device Name</td>
<td>SIP_Trunk_to_CUBE_for_Mettel</td>
<td>Name for the trunk</td>
</tr>
<tr>
<td>Device Pool</td>
<td>Mettel_Devicepool</td>
<td>Default Device Pool is used for this trunk</td>
</tr>
<tr>
<td>Media Resource Group List</td>
<td>MRGL_Default</td>
<td>MRG with resources:ANN, CFB, MOH and MTP</td>
</tr>
<tr>
<td>Significant Digits</td>
<td>4</td>
<td>4 digits Extension for all CPE phones</td>
</tr>
<tr>
<td>Destination Address</td>
<td>10.80.18.10</td>
<td>Virtual IP address of the Cisco UBE</td>
</tr>
<tr>
<td>SIP Trunk Security Profile</td>
<td>Non Secure SIP Trunk Profile</td>
<td>SIP Trunk Security Profile configured earlier</td>
</tr>
<tr>
<td>SIP Profile</td>
<td>Standard SIP Profile w/early media disabled</td>
<td>SIP Profile configured earlier</td>
</tr>
</tbody>
</table>
Dialplan

Route Pattern Configuration

**Navigation:** Call Routing-> Route/Hunt-> Route Pattern

Route patterns are configured as below, Cisco IP phone dial “9”+10 digits number to access PSTN via Cisco UBE, “9” is removed before send to Cisco UBE; for FAX call, Access Code “9” is used at Cisco Fax gateway, “9” is removed at Cisco UCM and 10 digits number is send to Cisco UBE to MetTel network. Incoming fax call to 5174 will be sent to Cisco Fax gateway. 2900 is the Pilot Number for Voice mail to Unity Connection.

![Figure 28 Route Patterns List](image-url)

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**Figure 29 Route Pattern for Voice**

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EDCS# xxx Rev # <edcs revision number> (to be filled in by Cisco when completed. Please leave blank)
Figure 30 Route Pattern for Voice (cont.)

Figure 31 Route Pattern for Unity Connection
### Calling Party Transformations

- **Use Calling Party’s External Phone Number Mask**: [ ]
- **Calling Party Transform Mask**: 
- **Prefix Digits (Outgoing Calls)**: 
- **Calling Line ID Presentation**: Default
- **Calling Name Presentation**: Default
- **Calling Party Number Type**: Cisco CallManager
- **Calling Party Numbering Plan**: Cisco CallManager

### Connected Party Transformations

- **Connected Line ID Presentation**: Default
- **Connected Name Presentation**: Default

### Called Party Transformations

- **Discard Digits**: < None >
- **Called Party Transform Mask**: 
- **Prefix Digits (Outgoing Calls)**: 
- **Called Party Number Type**: Cisco CallManager
- **Called Party Numbering Plan**: Cisco CallManager

### ISDN Network-Specific Facilities Information Element

<table>
<thead>
<tr>
<th>Network Service Protocol</th>
<th>Carrier Identification Code</th>
</tr>
</thead>
<tbody>
<tr>
<td>-- Not Selected --</td>
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</tr>
</tbody>
</table>

### ISDN Network-Specific Facilities Information Element

<table>
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<tr>
<th>Network Service Protocol</th>
<th>Service Parameter Name</th>
<th>Service Parameter Value</th>
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</thead>
<tbody>
<tr>
<td>-- Not Selected --</td>
<td>&lt; Not Exist &gt;</td>
<td></td>
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</tbody>
</table>

---

Figure 32 Route Pattern for Unity Connection (cont.)
Figure 33 Route Pattern for Fax
Figure 34 Route Pattern for Fax (cont.)
<table>
<thead>
<tr>
<th>Setting</th>
<th>Value</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>Route Pattern</td>
<td>9.@ for Voice call, 5174 for fax call and 2900 for Unity voice mail</td>
<td>Specify appropriate Route Pattern</td>
</tr>
<tr>
<td>Gateway/Route List</td>
<td>SIP_Trunk_to_CUBE_for_Mettel for Route Pattern 9.@ and 5174</td>
<td>SIP Trunk name configured earlier</td>
</tr>
<tr>
<td>Numbering Plan</td>
<td>NANP for Route Pattern 9.@</td>
<td>North American Numbering Plan</td>
</tr>
<tr>
<td>Call Classification</td>
<td>Offnet for Route Pattern 9.@ and 5174</td>
<td>Restrict the transferring of an external call to an external device</td>
</tr>
<tr>
<td>Discard Digits</td>
<td>PreDot for Route Pattern 9.@</td>
<td>Specifies how to modify digit before they are sending to MetTel network.</td>
</tr>
</tbody>
</table>
## Acronyms

<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>
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</tbody>
</table>
Important Information

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Appendix A: Test Results

SP_SIP_master_testplan_CUCM_10.5.2.xlsx