



TalkTalk SIP Trunking:

Cisco Unified Communications Manager 11.0.1 with Cisco Unified Border Element (CUBE 11.5.0) on ISR 4321 [IOS-XE - 15.6(1)S, 3.17] using SIP

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Introduction

Service Providers today, such as TalkTalk, 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.

TalkTalk 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 TalkTalk network, Cisco Unified Border Element (Cisco UBE) ISR 4321/K9 running IOS-XE 15.6(1)S/3.17 can be used. The Cisco Unified Border Element 15.6(1)S provides demarcation, security, interworking and session control services for Cisco Unified Communications Manager 11.0.1 connected to TalkTalk 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 TalkTalk 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) 11.0.1 and Cisco Unified Border Element (Cisco UBE) on ISR 4321/K9 [IOS-XE - 15.6(1)S, 3.17] for connectivity to TalkTalk SIP Trunking service. The deployment model covered in this application note is CPE (Cisco UCM 11.0.1) to PSTN (TalkTalk).
- Testing was performed in accordance to TalkTalk 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 TalkTalk 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 TalkTalk 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:

http://www.cisco.com/c/en/us/td/docs/voice_ip_comm/cucm/srnd/collab10/collab10/dialplan.html



Network Topology

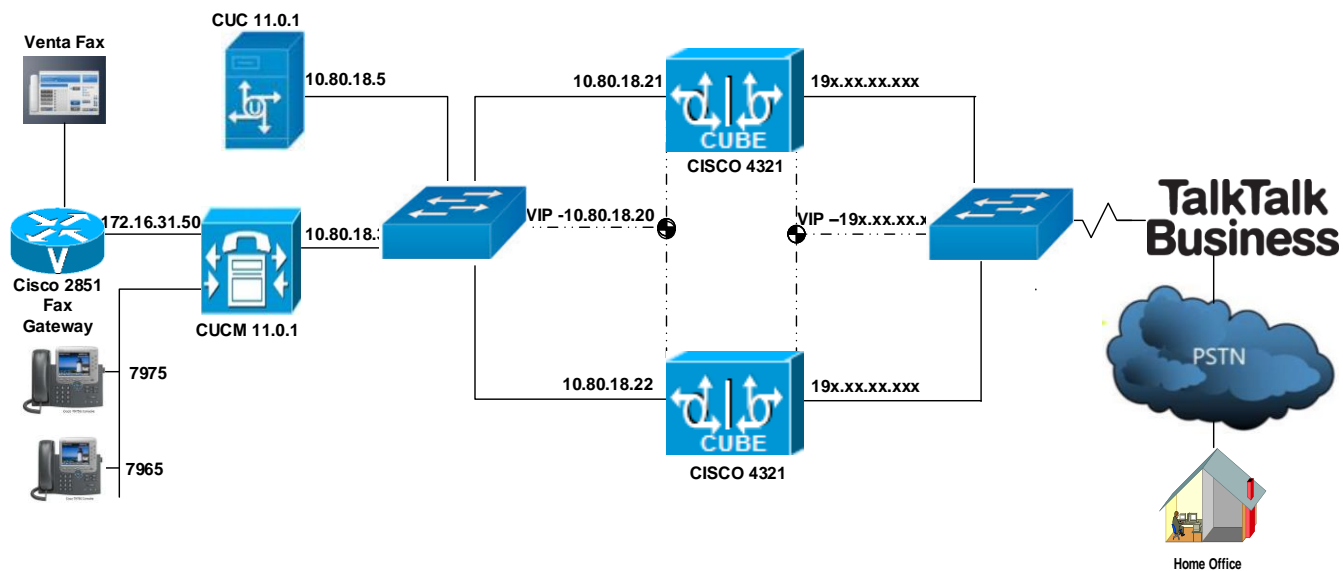


Figure 1: Network Topology

- Cisco IP Phones 7975 and 7965 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 ISR 4321/K9 router as CUBE
- Cisco 2851 Fax Gateway
- IP phones 7965 (SIP) and 7975 (SCCP)

Software Requirements

- Cisco Unified Communications Manager 11.0.1
- Cisco Unity Connection 11.0.1
- IOS 15.6(1)S for ISR 4321/K9 Cisco Unified Border Element
- IOS 15.1(4)M5 for Cisco 2851 Fax Gateway

Features

Features Supported

- Incoming and outgoing off-net calls using G711ALaw
- 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 pass-through)

Features Not Supported

- Cisco IP phones used in this test do not support blind transfer
- Fax (T.38) is not supported by Service Provider

Caveats

- CLID is not updated on PSTN phones for transfer (attended and unattended) OffNet PSTN scenarios. Caller ID is not updated at PSTN once transfer is completed by PBX. Cisco UBE modify PAI/PPI header and forward to network in the tested release. CISCO BUG ID: CSCuv04539.
- Call Forward Unconditional to OffNet PSTN scenarios was executed by configuring Calling Party Selection* to "First Redirect number (External)" under Trunk Settings



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 GigabitEthernet0/0/0
ip address 10.80.18.21 255.255.255.0
media-type rj45
negotiation auto
redundancy rii 1
redundancy group 1 ip 10.80.18.20 exclusive
!
interface GigabitEthernet0/0/1
ip address 192.65.79.140 255.255.255.128
negotiation auto
redundancy rii 2
redundancy group 1 ip 192.65.79.149 exclusive
!
```



Global Cisco UBE Settings

In order to enable Cisco UBE IP2IP gateway functionality, enter the following:

```
voice service voip
ip address trusted list
  ipv4 0.0.0.0 0.0.0.0
  ipv4 91.146.112.10
address-hiding
mode border-element license capacity 20
allow-connections sip to sip
redundancy-group 1
no supplementary-service sip handle-replaces
fax protocol pass-through g711alaw
sip
  bind control source-interface GigabitEthernet0/0/1
  bind media source-interface GigabitEthernet0/0/1
  rel1xx supported "rel100"
  session refresh
  asserted-id pai
  privacy pstn
  early-offer forced
  midcall-signaling passthru
  privacy-policy passthru
  privacy-policy send-always
  g729 annexb-all
!
```




Explanation

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

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 uses the flow-through mode). In order to enable flow-around mode, perform the following actions:

```
voice service voip
    media flow-around
```

Codecs

G711alaw is used as the preferred codec for this testing

```
voice class codec 1
    codec preference 1 g711alaw
    codec preference 2 g711ulaw
```



Dial Peer

Cisco UBE uses dial-peers to route the call accordingly based on the digits

dial-peer voice 200 voip

description Outbound-from IP PBX to PSTN - WAN facing

huntstop

destination-pattern .T

session protocol sipv2

session target sip-server

voice-class codec 1

voice-class sip asserted-id pai

voice-class sip profiles 101

voice-class sip bind control source-interface GigabitEthernet0/0/1

voice-class sip bind media source-interface GigabitEthernet0/0/1

dtmf-relay rtp-nte

fax-relay ecm disable

fax rate disable

fax nsf 000000

fax protocol pass-through g711alaw

no vad

!

dial-peer voice 100 voip

description Inbound-from PSTN to IP PBX - LAN facing

huntstop

destination-pattern 0203.....

session protocol sipv2

session target ipv4:10.80.18.3:5060

voice-class codec 1

voice-class sip asserted-id pai

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

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

dtmf-relay rtp-nte



```
fax-relay ecm disable
fax rate disable
fax nsf 000000
fax protocol pass-through g711alaw
no vad
!
dial-peer voice 420 voip
description PBX to PBX dialing
translation-profile outgoing TALK-TALK
huntstop
destination-pattern 61..
session protocol sipv2
session target sip-server
session transport udp
voice-class codec 1
voice-class sip asserted-id pai
voice-class sip profiles 101
voice-class sip bind control source-interface GigabitEthernet0/0/1
voice-class sip bind media source-interface GigabitEthernet0/0/1
dtmf-relay rtp-nte
fax-relay ecm disable
fax rate disable
fax protocol pass-through g711alaw
no vad
!
dial-peer voice 500 voip
description cube dp
huntstop
session protocol sipv2
session target sip-server
incoming called-number 0203.....
```



```
voice-class codec 1
voice-class sip asserted-id pai
voice-class sip profiles 101
dtmf-relay rtp-nte
fax-relay ecm disable
fax rate disable
fax nsf 000000
fax protocol pass-through g711alaw
no vad
!
dial-peer voice 210 voip
description outgoing call to TalkTalk - LAN facing
huntstop
session protocol sipv2
incoming called-number .T
voice-class codec 1
voice-class sip asserted-id pai
voice-class sip bind control source-interface GigabitEthernet0/0/0
voice-class sip bind media source-interface GigabitEthernet0/0/0
dtmf-relay rtp-nte
fax-relay ecm disable
fax rate disable
fax nsf 000000
fax protocol pass-through g711alaw
no vad
!
dial-peer voice 998 voip
description "Emergency and Operator call-LAN side"
session protocol sipv2
incoming called-number [1,9]..
voice-class codec 1
```



```
voice-class sip bind control source-interface GigabitEthernet0/0/0
voice-class sip bind media source-interface GigabitEthernet0/0/0
dtmf-relay rtp-nte
no vad
!
```

```
dial-peer voice 999 voip
description "Emergency and Operator call-WAN side"
destination-pattern [1,9]..
session protocol sipv2
session target sip-server
voice-class codec 1
voice-class sip early-offer forced
voice-class sip profiles 101
voice-class sip bind control source-interface GigabitEthernet0/0/1
voice-class sip bind media source-interface GigabitEthernet0/0/1
dtmf-relay rtp-nte
no vad
!
```

```
dial-peer voice 117 voip
description "information service call-LAN side"
session protocol sipv2
session target sip-server
incoming called-number 118500
voice-class codec 1
voice-class sip bind control source-interface GigabitEthernet0/0/0
voice-class sip bind media source-interface GigabitEthernet0/0/0
dtmf-relay rtp-nte
no vad
!
```

```
dial-peer voice 118 voip
description "information service call-WAN side"
```



```
destination-pattern 118500
session protocol sipv2
session target sip-server
voice-class codec 1
voice-class sip early-offer forced
voice-class sip profiles 101
voice-class sip bind control source-interface GigabitEthernet0/0/1
voice-class sip bind media source-interface GigabitEthernet0/0/1
dtmf-relay rtp-nte
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 picks up the last 4 significant Digits configured under SIP Trunk and routes the call based on those 4 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 “9.001@” 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 Fax. For PBX to PBX via TalkTalk, Caller dial 9 prefix followed by the target four-digits extension number, 9 was stripped and the four-digits extension number was send to Cisco UBE, Cisco UBE translate the 4 digits extension number to its full ten-digits DID under Dial Peer 420 and send to TalkTalk network which will direct back to Cisco UBE and handled same as normal incoming PSTN call.



Figure 2: Outbound Voice Call



Figure 3: Inbound Voice Call

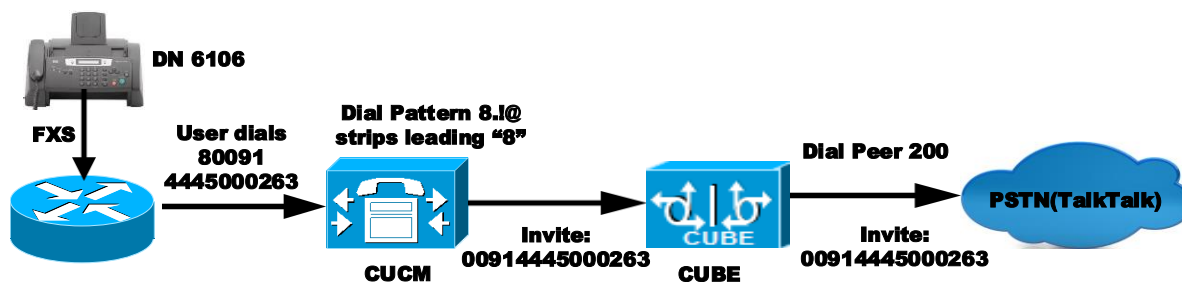


Figure 4: Outbound Fax Call

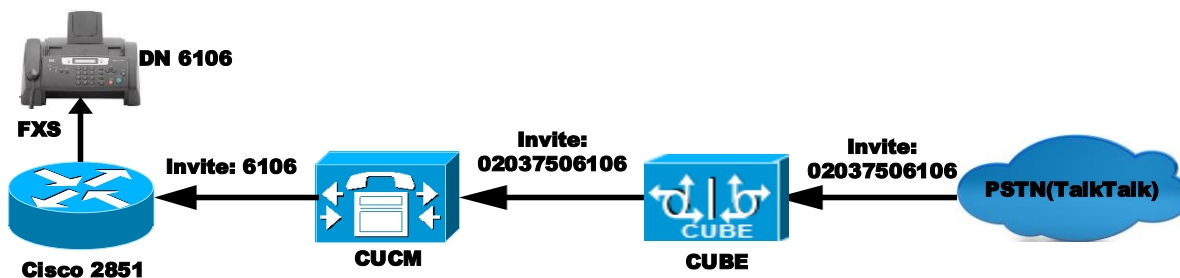


Figure 5: Inbound Fax Call

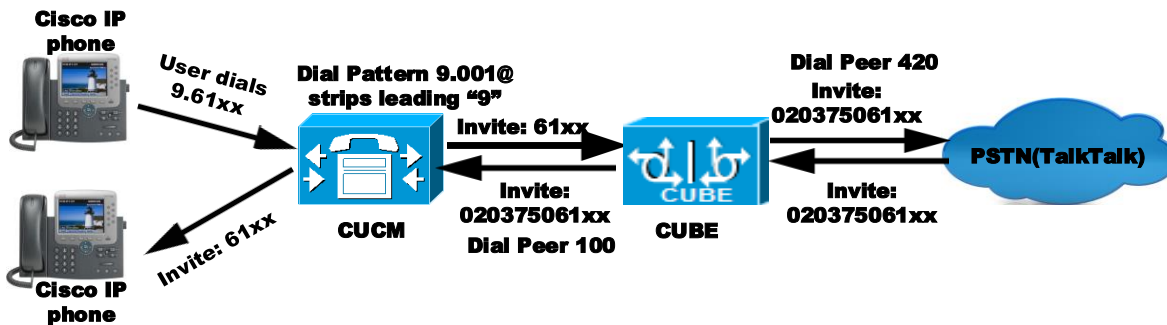


Figure 6: PBX to PBX via TalkTalk Call



Configuration Example

The following configuration snippet contains a sample configuration of Cisco UBE with all parameters mentioned previously

Active Cisco UBE

```
TalkTalk_CUBE1#sh running-config
```

```
Building configuration...
```

```
version 15.6
service timestamps debug datetime msec
service timestamps log datetime msec
no platform punt-keepalive disable-kernel-core
!
hostname TalkTalk_CUBE1
!
boot-start-marker
boot system bootflash:isr4300-universalk9.03.17.00.S.156-1.S-std.SPA.bin
boot-end-marker
!
vrf definition Mgmt-intf
!
address-family ipv4
exit-address-family
!
address-family ipv6
exit-address-family
!
no aaa new-model
!
no ip domain lookup
ip domain name tekvision.com
```



subscriber templating

!

multilink bundle-name authenticated

!

voice service voip

ip address trusted list

ipv4 0.0.0.0 0.0.0.0

ipv4 91.146.112.10

address-hiding

mode border-element license capacity 20

allow-connections sip to sip

redundancy-group 1

no supplementary-service sip handle-replaces

fax protocol pass-through g711alaw

sip

bind control source-interface GigabitEthernet0/0/1

bind media source-interface GigabitEthernet0/0/1

rel1xx supported "rel100"

session refresh

asserted-id pai

privacy pstn

early-offer forced

midcall-signaling passthru

privacy-policy passthru

privacy-policy send-always

g729 annexb-all

!

voice class codec 1

codec preference 1 g711alaw

codec preference 2 g711ulaw

!



```
!  
voice class sip-profiles 101  
  response ANY sip-header Allow-Header modify "UPDATE," ""  
  request INVITE sip-header Diversion modify "<sip:(.*)@(.*)>" "<sip:0203750\1@\2  
>"  
!  
voice translation-rule 2  
  rule 1 /^.*(61..\)/ /0203750\1/  
!  
voice translation-profile TALK-TALK  
  translate called 2  
!  
license udi pid ISR4321/K9 sn FDO19220MSQ  
!  
spanning-tree extend system-id  
!  
redundancy  
  mode none  
  application redundancy  
  group 1  
    name voice-b2bhaTalkTalk  
    priority 100 failover threshold 75  
    timers delay 30 reload 60  
    control GigabitEthernet0/1/0 protocol 1  
    data GigabitEthernet0/1/0  
    track 1 shutdown  
    track 2 shutdown  
!  
vlan internal allocation policy ascending  
!
```



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

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

```
!
```

```
interface GigabitEthernet0/0/0
```

```
ip address 10.80.18.21 255.255.255.0
```

```
media-type rj45
```

```
negotiation auto
```

```
redundancy rii 1
```

```
redundancy group 1 ip 10.80.18.20 exclusive
```

```
!
```

```
interface GigabitEthernet0/0/1
```

```
ip address 192.65.79.140 255.255.255.128
```

```
negotiation auto
```

```
redundancy rii 2
```

```
redundancy group 1 ip 192.65.79.149 exclusive
```

```
!
```

```
interface GigabitEthernet0/1/0
```

```
description CUBE HA MS5 3/0/37
```

```
ip address 10.89.20.9 255.255.255.0
```

```
negotiation auto
```

```
!
```

```
interface GigabitEthernet0
```

```
vrf forwarding Mgmt-intf
```

```
no ip address
```

```
negotiation auto
```

```
!
```

```
interface Vlan1
```

```
no ip address
```

```
shutdown
```

```
!
```



```
ip forward-protocol nd
no ip http server
no ip http secure-server
ip tftp source-interface GigabitEthernet0
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 172.16.24.0 255.255.248.0 10.80.18.1
!
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 200 voip
description Outbound-from IP PBX to PSTN - WAN facing
huntstop
destination-pattern .T
session protocol sipv2
session target sip-server
voice-class codec 1
voice-class sip asserted-id pai
voice-class sip profiles 101
voice-class sip bind control source-interface GigabitEthernet0/0/1
voice-class sip bind media source-interface GigabitEthernet0/0/1
dtmf-relay rtp-nte
fax-relay ecm disable
fax rate disable
```



```
fax nsf 000000
fax protocol pass-through g711alaw
no vad
!
dial-peer voice 100 voip
description Inbound-from PSTN to IP PBX - LAN facing
huntstop
destination-pattern 0203.....
session protocol sipv2
session target ipv4:10.80.18.3:5060
voice-class codec 1
voice-class sip asserted-id pai
voice-class sip bind control source-interface GigabitEthernet0/0/0
voice-class sip bind media source-interface GigabitEthernet0/0/0
dtmf-relay rtp-nte
fax-relay ecm disable
fax rate disable
fax nsf 000000
fax protocol pass-through g711alaw
no vad
!
dial-peer voice 420 voip
description PBX to PBX dialing
translation-profile outgoing TALK-TALK
huntstop
destination-pattern 61..
session protocol sipv2
session target sip-server
session transport udp
voice-class codec 1
voice-class sip asserted-id pai
```



```
voice-class sip profiles 101
voice-class sip bind control source-interface GigabitEthernet0/0/1
voice-class sip bind media source-interface GigabitEthernet0/0/1
dtmf-relay rtp-nte
fax-relay ecm disable
fax rate disable
fax protocol pass-through g711alaw
no vad
!
dial-peer voice 500 voip
description cube dp
huntstop
session protocol sipv2
session target sip-server
incoming called-number 0203.....
voice-class codec 1
voice-class sip asserted-id pai
voice-class sip profiles 101
dtmf-relay rtp-nte
fax-relay ecm disable
fax rate disable
fax nsf 000000
fax protocol pass-through g711alaw
no vad
!
dial-peer voice 210 voip
description outgoing call to TalkTalk - LAN facing
huntstop
session protocol sipv2
incoming called-number .T
voice-class codec 1
```



```
voice-class sip asserted-id pai
voice-class sip bind control source-interface GigabitEthernet0/0/0
voice-class sip bind media source-interface GigabitEthernet0/0/0
dtmf-relay rtp-nte
fax-relay ecm disable
fax rate disable
fax nsf 000000
fax protocol pass-through g711alaw
no vad
!
dial-peer voice 998 voip
description "Emergency and Operator call-LAN side"
session protocol sipv2
incoming called-number [1,9]..
voice-class codec 1
voice-class sip bind control source-interface GigabitEthernet0/0/0
voice-class sip bind media source-interface GigabitEthernet0/0/0
dtmf-relay rtp-nte
no vad
!
dial-peer voice 999 voip
description "Emergency and Operator call-WAN side"
destination-pattern [1,9]..
session protocol sipv2
session target sip-server
voice-class codec 1
voice-class sip early-offer forced
voice-class sip profiles 101
voice-class sip bind control source-interface GigabitEthernet0/0/1
voice-class sip bind media source-interface GigabitEthernet0/0/1
dtmf-relay rtp-nte
```




no vad

!

dial-peer voice 117 voip

description "information service call-LAN side"

session protocol sipv2

session target sip-server

incoming called-number 118500

voice-class codec 1

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

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

dtmf-relay rtp-nte

no vad

!

dial-peer voice 118 voip

description "information service call-WAN side"

destination-pattern 118500

session protocol sipv2

session target sip-server

voice-class codec 1

voice-class sip early-offer forced

voice-class sip profiles 101

voice-class sip bind control source-interface GigabitEthernet0/0/1

voice-class sip bind media source-interface GigabitEthernet0/0/1

dtmf-relay rtp-nte

no vad

!

!

sip-ua

keepalive target ipv4:91.146.112.10:5060

disable-early-media 180

timers keepalive active 180



```
sip-server ipv4:91.146.112.10:5060
```

```
connection-reuse
```

```
!
```

```
!
```

```
line con 0
```

```
stopbits 1
```

```
line aux 0
```

```
stopbits 1
```

```
line vty 0 4
```

```
exec-timeout 0 0
```

```
password tekV1z10n
```

```
login local
```

```
!
```

```
!
```

```
end
```



Standby Cisco UBE

TalkTalk_CUBE2#sh running-config

Building configuration...

version 15.6

service timestamps debug datetime msec

service timestamps log datetime msec

no platform punt-keepalive disable-kernel-core

!

hostname TalkTalk_CUBE2

!

boot-start-marker

boot system bootflash:isr4300-universalk9.03.17.00.S.156-1.S-std.SPA.bin

boot-end-marker

!

!

vrf definition Mgmt-intf

!

address-family ipv4

exit-address-family

!

address-family ipv6

exit-address-family

!

no aaa new-model

!

no ip domain lookup

ip domain name tekvision.com

!

subscriber templating



```
multilink bundle-name authenticated
!
voice service voip
  ip address trusted list
    ipv4 0.0.0.0 0.0.0.0
    ipv4 91.146.112.10
  address-hiding
  mode border-element license capacity 20
  allow-connections sip to sip
  redundancy-group 1
  no supplementary-service sip handle-replaces
  fax protocol pass-through g711alaw
  sip
    bind control source-interface GigabitEthernet0/0/1
    bind media source-interface GigabitEthernet0/0/1
    rel1xx supported "rel100"
    session refresh
    asserted-id pai
    privacy pstn
    early-offer forced
    midcall-signaling passthru
    privacy-policy passthru
    privacy-policy send-always
    g729 annexb-all
  !
voice class codec 1
  codec preference 1 g711alaw
  codec preference 2 g711ulaw
!
```



```
voice class sip-profiles 101
  response ANY sip-header Allow-Header modify "UPDATE," ""
  request INVITE sip-header Diversion modify "<sip:(.*)@(.*)>" "<sip:0203750\1@\2
>"
!
voice translation-rule 2
  rule 1 /^.*\((61..\)/ /0203750\1/
!
voice translation-profile TALK-TALK
  translate called 2
!
license udi pid ISR4321/K9 sn FDO19220MQ9
!
spanning-tree extend system-id
!
redundancy
  mode none
  application redundancy
  group 1
    name voice-b2bhaTalkTalk
    priority 100 failover threshold 75
    timers delay 30 reload 60
    control GigabitEthernet0/1/0 protocol 1
    data GigabitEthernet0/1/0
    track 1 shutdown
    track 2 shutdown
!
vlan internal allocation policy ascending
!
track 1 interface GigabitEthernet0/0/0 line-protocol
track 2 interface GigabitEthernet0/0/1 line-protocol
```



!

```
interface GigabitEthernet0/0/0
ip address 10.80.18.22 255.255.255.0
media-type rj45
negotiation auto
redundancy rii 1
redundancy group 1 ip 10.80.18.20 exclusive
```

!

```
interface GigabitEthernet0/0/1
ip address 192.65.79.141 255.255.255.128
negotiation auto
redundancy rii 2
redundancy group 1 ip 192.65.79.149 exclusive
```

!

```
interface GigabitEthernet0/1/0
description CUBE HA MS5 3/0/38
ip address 10.89.20.10 255.255.255.0
negotiation auto
```

!

```
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 tftp source-interface GigabitEthernet0
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 172.16.24.0 255.255.248.0 10.80.18.1
!
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 200 voip
description Outbound-from IP PBX to PSTN - WAN facing
huntstop
destination-pattern .T
session protocol sipv2
session target sip-server
voice-class codec 1
voice-class sip asserted-id pai
voice-class sip profiles 101
voice-class sip bind control source-interface GigabitEthernet0/0/1
voice-class sip bind media source-interface GigabitEthernet0/0/1
dtmf-relay rtp-nte
fax-relay ecm disable
fax rate disable
fax nsf 000000
fax protocol pass-through g711alaw
```



no vad

!

dial-peer voice 100 voip

description Inbound-from PSTN to IP PBX - LAN facing

huntstop

destination-pattern 0203.....

session protocol sipv2

session target ipv4:10.80.18.3:5060

voice-class codec 1

voice-class sip asserted-id pai

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

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

dtmf-relay rtp-nte

fax-relay ecm disable

fax rate disable

fax nsf 000000

fax protocol pass-through g711alaw

no vad

!

dial-peer voice 420 voip

description PBX to PBX dialing

translation-profile outgoing TALK-TALK

huntstop

destination-pattern 61..

session protocol sipv2

session target sip-server

session transport udp

voice-class codec 1

voice-class sip asserted-id pai

voice-class sip profiles 101

voice-class sip bind control source-interface GigabitEthernet0/0/1



```
voice-class sip bind media source-interface GigabitEthernet0/0/1
```

```
dtmf-relay rtp-nte
```

```
fax-relay ecm disable
```

```
fax rate disable
```

```
fax protocol pass-through g711alaw
```

```
no vad
```

```
!
```

```
dial-peer voice 500 voip
```

```
description cube dp
```

```
huntstop
```

```
session protocol sipv2
```

```
session target sip-server
```

```
incoming called-number 0203.....
```

```
voice-class codec 1
```

```
voice-class sip asserted-id pai
```

```
voice-class sip profiles 101
```

```
dtmf-relay rtp-nte
```

```
fax-relay ecm disable
```

```
fax rate disable
```

```
fax nsf 000000
```

```
fax protocol pass-through g711alaw
```

```
no vad
```

```
!
```

```
dial-peer voice 210 voip
```

```
description outgoing call to TalkTalk - LAN facing
```

```
huntstop
```

```
session protocol sipv2
```

```
session target sip-server
```

```
incoming called-number .T
```

```
voice-class codec 1
```

```
voice-class sip asserted-id pai
```



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

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

```
dtmf-relay rtp-nte
```

```
fax-relay ecm disable
```

```
fax rate disable
```

```
fax nsf 000000
```

```
fax protocol pass-through g711alaw
```

```
no vad
```

```
!
```

```
dial-peer voice 998 voip
```

```
description "Emergency and Operator call-LAN side"
```

```
session protocol sipv2
```

```
session target sip-server
```

```
incoming called-number [1,9]..
```

```
voice-class codec 1
```

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

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

```
dtmf-relay rtp-nte
```

```
no vad
```

```
!
```

```
dial-peer voice 999 voip
```

```
description "Emergency and Operator call-WAN side"
```

```
destination-pattern [1,9]..
```

```
session protocol sipv2
```

```
session target sip-server
```

```
voice-class codec 1
```

```
voice-class sip early-offer forced
```

```
voice-class sip profiles 101
```

```
voice-class sip bind control source-interface GigabitEthernet0/0/1
```

```
voice-class sip bind media source-interface GigabitEthernet0/0/1
```

```
dtmf-relay rtp-nte
```



```
no vad
!
dial-peer voice 117 voip
description "information service call-LAN side"
session protocol sipv2
session target sip-server
incoming called-number 118500
voice-class codec 1
voice-class sip bind control source-interface GigabitEthernet0/0/0
voice-class sip bind media source-interface GigabitEthernet0/0/0
dtmf-relay rtp-nte
no vad
!
dial-peer voice 118 voip
description "information service call-WAN side"
destination-pattern 118500
session protocol sipv2
session target sip-server
voice-class codec 1
voice-class sip early-offer forced
voice-class sip profiles 101
voice-class sip bind control source-interface GigabitEthernet0/0/1
voice-class sip bind media source-interface GigabitEthernet0/0/1
dtmf-relay rtp-nte
no vad
!
!
sip-ua
keepalive target ipv4:91.146.112.10:5060
disable-early-media 180
timers keepalive active 180
```



```
sip-server ipv4:91.146.112.10:5060
```

```
connection-reuse
```

```
!
```

```
!
```

```
line con 0
```

```
stopbits 1
```

```
line aux 0
```

```
stopbits 1
```

```
line vty 0 4
```

```
password tekV1z10n
```

```
login local
```

```
!
```

```
!
```

```
end
```



Configuring Cisco Unified Communications Manager

Cisco UCM Version

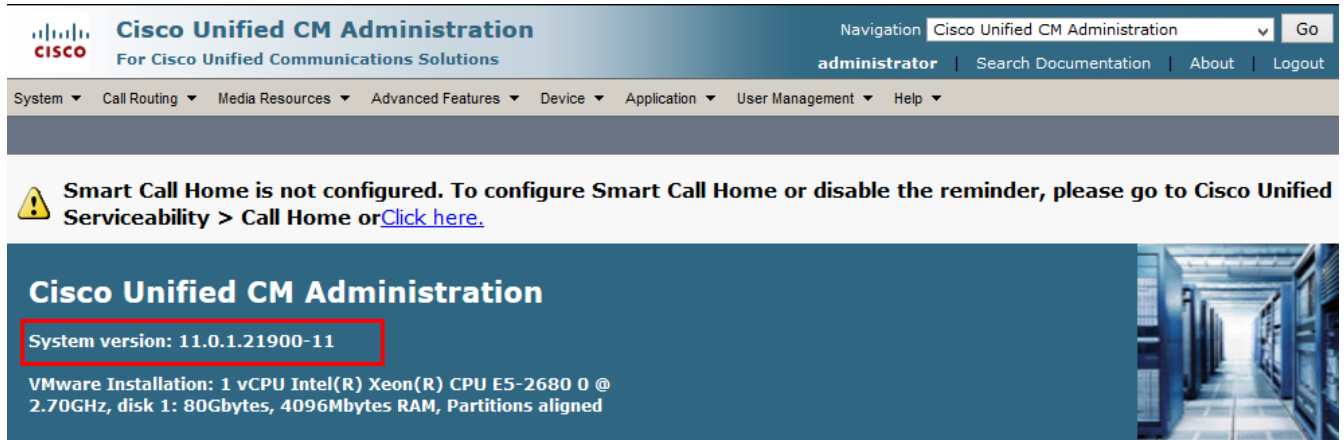


Figure 7: Cisco UCM Version

Cisco Call Manager Service Parameters

Navigation path: System > Service Parameters

Select Server* = Clus28Sub1--CUCM Voice/Video (Active)

Select Service* = Cisco CallManager (Active)

All other fields are set to default values



System ▾ Call Routing ▾ Media Resources ▾ Advanced Features ▾ Device ▾ Application ▾ User Management ▾ Help ▾

Service Parameter Configuration

Related Links: [Parameters for All Servers ▾](#) [Go](#)

Save Set to Default Advanced

Status

Status: Ready

Select Server and Service

Server* [Clus28Pub--CUCM Voice/Video \(Active\)](#) ▾

Service* [Cisco CallManager \(Active\)](#) ▾

All parameters apply only to the current server except parameters that are in the cluster-wide group(s).

Cisco CallManager (Active) Parameters on server Clus28Pub--CUCM Voice/Video (Active)

| Parameter Name | Parameter Value | Suggested Value |
|--|-----------------|-----------------|
| Call Throttling | | |
| Code Yellow Entry Latency * | 20 | 20 |
| Code Yellow Exit Latency Calculation * | 40 | 40 |
| Code Yellow Duration * | 5 ▾ | 5 |
| Max Events Allowed * | 2000 | 2000 |
| System Throttle Sample Size * | 10 | 10 |

Figure 8: Service Parameters

Memory Throttling

[Enable Memory Throttling](#) * [True](#) ▾ True

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

System

| | | |
|---|--------------------|------------------|
| CDR Enabled Flag * | False ▾ | False |
| CDR Log Calls with Zero Duration Flag * | False ▾ | False |
| Digit Analysis Complexity * | StandardAnalysis ▾ | StandardAnalysis |
| Database Debounce Timer * | 0 | 0 |
| Maximum Phone Fallback Queue Depth * | 10 | 10 |
| Maximum Number of Registered Devices * | 5000 | 5000 |
| System Initialization Timer * | 60 | 60 |

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



SDL Trace

| | | |
|---|------------|------------|
| SDL Trace Data Flags * | 0x00000111 | 0x00000111 |
| SDL Trace Flush Immediately * | False | False |
| SDL Trace Data Size * | 0 | 0 |
| SDL Trace Flag * | True | True |
| SDL TraceType Flags * | 0x8000EB15 | 0x8000EB15 |

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

Clusterwide Parameters (Device - General)

| | | |
|--|----------|----------|
| Call Diagnostics Enabled * | Disabled | Disabled |
| Show Line Group Member DN in finalCalledPartyNumber CDR Field * | False | False |
| Show Line Group Member Non Masked DN in finalCalledPartyNumber CDR Field * | False | False |
| CTI New Call Accept Timer * | 4 | 4 |
| CTI Generate Digits Interval * | 250 | 250 |
| CTI Dial Digits Interval * | 250 | 250 |
| CTI Await Further Digits * | False | False |
| CTI Use Wildcard Pattern as calledPartyDN * | False | False |
| CTI Report Ringback on SIP 183 with SDP * | True | True |
| Retain Media on Disconnect with PI for Active Call * | False | False |
| Station and Backup Server KeepAlive Interval * | 60 | 60 |
| Station KeepAlive Interval * | 30 | 30 |
| Status Enquiry Poll Flag * | False | False |
| Strip # Sign from Called Party Number * | True | True |

Figure 9: Service Parameters (Cont.)

| | | |
|--|--------|--------|
| Session Handoff Alerting Timer * | 10 | 10 |
| T301 Timer * | 180000 | 180000 |
| T302 Timer * | 15000 | 15000 |
| T303 Timer * | 4000 | 4000 |
| T304 Timer * | 30000 | 30000 |
| T305 Timer * | 30000 | 30000 |
| T306 Timer * | 30000 | 30000 |
| T308 Timer * | 4000 | 4000 |
| T309 Timer * | 90000 | 90000 |
| T310 Timer * | 60000 | 60000 |
| T313 Timer * | 4000 | 4000 |
| T316 Timer * | 120000 | 120000 |
| T317 Timer * | 100000 | 100000 |
| T321 Timer * | 30000 | 30000 |
| T322 Timer * | 4000 | 4000 |
| Tone on Hold Timer * | 10 | 10 |



| | | |
|--|--|---|
| Unknown Caller ID Flag * | True | True |
| Call Classification * | OffNet | OffNet |
| Always Display Original Dialed Number * | False | False |
| Name Display for Original Dialed Number When Translated * | Show the Display Name for Original Dialed Number even if | Show the Display Name for Original Dialed Number even if Translated |
| Always Use PIs With Original Dialed Number * | False | False |
| Fail Call If Trusted Relay Point Allocation Fails * | True | True |
| Display Calling/Called ID When PI is Not Available * | False | False |
| Enable Transit Counter Processing on QSIG Trunks * | False | False |
| Egress Facility IE Count * | 6 | 6 |
| Clusterwide Parameters (Device - Phone) | | |
| Always Use Prime Line * | False | False |
| Always Use Prime Line for Voice Message * | False | False |
| Builtin Bridge Enable * | Off | Off |
| Device Mobility Mode * | Off | Off |
| Display Device Mobility Location During Phone Registration * | True | True |
| Auto Answer Timer * | 1 | 1 |
| Extension Display on Cisco IP Phone Model 7910 * | False | False |
| Alternate Idle Phone Auto-Answer Behavior Enabled * | False | False |
| Hold Type * | False | False |
| Line State Update Enabled * | True | True |
| Off-hook to First Digit Timer * | 15000 | 15000 |
| Override Auto Answer If Speaker Is Disabled * | True | True |
| Out-of-Bandwidth Text * | Not Enough Bandwidth | Not Enough Bandwidth |
| Forced Authorization Code Prompt Text * | Enter Authorization Code | Enter Authorization Code |

Figure 10: Service Parameters (Cont.)

| | | |
|---|---|--|
| Client Matter Code Prompt Text * | Enter Client Matter Code | Enter Client Matter Code |
| AAR Network Congestion Rerouting Text * | Network Congestion. Rerouting. | Network Congestion. Rerouting. |
| Ring Setting of Busy Station Policy * | Only Apply Ring Setting of Busy Station When Incoming C | Only Apply Ring Setting of Busy Station When Incoming Call Arrives |
| Transfer On-hook Enabled * | False | False |
| Ring Setting of Busy Station * | Beep Only | Beep Only |
| Ring Setting of Idle Station * | Ring | Ring |
| Call Pickup Group Audio Alert Setting of Idle Station * | Ring Once | Ring Once |
| Call Pickup Group Audio Alert Setting of Busy Station * | Beep Only | Beep Only |
| BLF Pickup Audio Alert Setting of Idle Station * | Disable | Disable |
| BLF Pickup Audio Alert Setting of Busy Station * | Disable | Disable |
| Privacy Setting * | True | True |



| | | |
|---|--|--|
| Enforce Privacy Setting on Held Calls * | False | False |
| SIP Station KeepAlive Interval * | 120 | 120 |
| SIP Station Realm * | ccmsipline | ccmsipline |
| Hunt Group Logoff Notification * | None | None |
| Speed Dial Await Further Digits * | False | False |
| Display CTI Route Point Name or DN * | False | False |
| Display Original Calling Number on Transfer from Cisco Unity * | False | False |
| URI Dialing Display Preference * | DN | DN |
| | | |
| Insert Hyphens in 12-Digit Numbers * | False | False |
| Allow Call Waiting During an In-Progress Outbound Analog Call * | True | True |
| Clusterwide Parameters (Device - PRI and MGCP Gateway) | | |
| Calling Party Number Screening Indicator * | CallManager sets the screening indicator value - Default | CallManager sets the screening indicator value - Default setting |
| Enable Outbound NetworkTrunk CallingParty Restriction * | False | False |
| | | |
| Clear Calls Flag When Datalink Is Down * | True | True |
| Device Status Poll Interval * | 3000 | 3000 |
| Disable Alerting Progress Indicator * | False | False |
| Discard Non Inband Progress in Overlap Sending * | False | False |
| Disable Resume from Shared-line MGCP FXS Port * | True | True |
| DTMF Silence Tone Flag * | False | False |
| Enable Display IE in Codeset 6 * | False | False |
| Enable Sending PRI NI2 Service Message * | False | False |

Figure 11: Service Parameters (Cont.)



| | | |
|--|--|--|
| Flash Hook Duration * | 500 | 500 |
| Gateway Poll Timer * | 10 | 10 |
| Location In PRI Progress Indicator IE (User Side Only) * | Use the Network Side PRI progress indicator IE | Use the Network Side PRI progress indicator IE |
| Matching Calling Party with Attendant Flag * | False | False |
| MGCP Database Query Delay Timer * | 1000 | 1000 |
| MGCP FXS On-Hook Pending Timer * | 3 | 3 |
| MGCP Response Timer * | 30 | 30 |
| MGCP Timer * | 3 | 3 |
| Numbering Plan Info * | 1 | 1 |
| Overlap Receiving Flag for PRI * | True | True |
| Outgoing Media Connect Time for PRI * | Connect ASAP | Connect ASAP |
| Port Release Timer * | 0 | 0 |
| SMDI Call Delay Timer * | 0 | 0 |
| Stable in State 4 Flag * | False | False |
| Optimize MGCP Registration * | True | True |
| Suppress Out-of-Channels Alarms * | True | True |
| I-Frame Timer * | 2000 | 2000 |
| User-to-User IE Status * | False | False |
| Convert European Progress Message to Alerting * | False | False |
| Enable DMS PRI Notify Message from User to Network * | True | True |
| Audit OOS Channels Interval * | 10 | 10 |
| Digital and Analog Ports Enabled * | True | True |
| Clusterwide Parameters (Device - H323) | | |
| Accept Unknown TCP Connection * | False | False |
| BRQ Enabled * | False | False |
| Call Present Disconnect Flag * | False | False |
| Check Progress Indicator Before Establishing Media * | False | False |
| H225 Block Setup Destination * | False | False |
| H225 DB Retry Timer * | 0 | 0 |
| H225 Device Connect Timer * | 0 | 0 |
| H225 DTMF Duration * | 100 | 100 |
| H225 TspReq Retry * | 2 | 2 |
| H225 Intercluster Call Throttle Timer * | 30 | 30 |
| H225 T301 Timer * | 180000 | 180000 |
| H225 T302 Timer * | 15000 | 15000 |
| H225 T303 Timer * | 4000 | 4000 |

Figure 12: Service Parameters (Cont.)



| | | |
|---|------------------------------------|------------------------------------|
| H225 T304 Timer * | 30000 | 30000 |
| H225 T305 Timer * | 30000 | 30000 |
| H225 T310 Timer * | 60000 | 60000 |
| H225 TCP Timer * | 5 | 5 |
| H245 TCS Timeout * | 10 | 10 |
| H323 Calling Party Number Screening Indicator * | Calling number screened and passed | Calling number screened and passed |
| Apply External Phone Number Mask for H.323 Calls * | False | False |
| Tone on Connect * | False | False |
| Wait Time for SDP with SR/RO Mode * | 3 | 3 |
| RAS ARQ Timer * | 3 | 3 |
| RAS BRQ Timer * | 3 | 3 |
| RAS DRQ Timer * | 3 | 3 |
| RAS RRO Timer * | 3 | 3 |
| Ras URO Timer * | 3 | 3 |
| Retry Count for ARQ * | 2 | 2 |
| Retry Count for BRQ * | 2 | 2 |
| Retry Count for DRQ * | 2 | 2 |
| Retry Count for RRO * | 2 | 2 |
| Retry Count for URO * | 1 | 1 |
| Send Product ID and Version ID * | False | False |
| Send Unified CM Version as Version ID in H225Setup * | False | False |
| Send Progress Timer * | 3000 | 3000 |
| Send H225 User Info Message * | User Info for Call Progress Tone | User Info for Call Progress Tone |
| Status Enquiry Poll Timer * | 10000 | 10000 |
| Device Name of GK-controlled Trunk That Will Use Port 1720 * | None | None |
| Host Name/IP Address of GK That Will Use RAS UDP Port 1719 * | None | None |
| Fail Call If MTP Allocation Fails * | False | False |
| Overlap Receiving Flag for H323 * | False | False |
| Allocate Transcoder for H.323 on Early Offer SIP Trunk for Calls with Early Media * | False | False |

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

Clusterwide Parameters (Device - SIP)

| | | |
|--|--------|--------|
| SIP Interoperability Enabled * | True | True |
| Retry Count for SIP Bye * | 10 | 10 |
| Retry Count for SIP Cancel * | 10 | 10 |
| Retry Count for SIP Invite * | 6 | 6 |
| Retry Count for SIP PRACK * | 6 | 6 |
| Retry Count for SIP Rel1XX * | 10 | 10 |
| Retry Count for SIP Publish * | 6 | 6 |
| Retry Count for SIP Response * | 6 | 6 |
| SIP Connect Timer * | 500 | 500 |
| SIP Disconnect Timer * | 500 | 500 |
| SIP Expires Timer * | 180000 | 180000 |
| SIP PRACK Timer * | 500 | 500 |

Figure 13: Service Parameters (Cont.)



| | | |
|--|--------------|--------------|
| SIP Rel1XX Timer * | 500 | 500 |
| SIP Trying Timer * | 500 | 500 |
| SIP Publish Timer * | 500 | 500 |
| SIP Min-SE Value * | 1800 | 1800 |
| SIPS URI Handling * | Reject | Reject |
| SIP statistics Periodic update Timer * | 2 | 2 |
| SIP Session Expires Timer * | 1800 | 1800 |
| SIP Trunk TspReq Retry * | 2 | 2 |
| SIP TCP Unused Connection Timer * | 14 | 14 |
| SIP TCP Timer * | 5 | 5 |
| SIP Station TCP Port Throttle Threshold * | 100 | 100 |
| SIP Trunk TCP Port Throttle Threshold * | 500 | 500 |
| SIP V.150 Outbound SDP Offer Filtering * | No Filtering | No Filtering |
| Send SIP Multicast TTL in SDP * | False | False |
| Default PUBLISH Expiration Timer * | 3600 | 3600 |
| Minimum PUBLISH Expiration Timer * | 60 | 60 |
| IM and Presence Publish Trunk | < None > | |
| Send 181 Call Is Being Forwarded * | False | False |
| Delay Sending 181 until 180/183 message is received * | True | True |
| Fail Call Over SIP Trunk if MTP Allocation Fails * | False | False |
| Log Call-Related REFER/NOTIFY/SUBSCRIBE SIP Messages for Session Trace * | True | True |
| Port Received Timer for Outbound Call Setup * | 2 | 2 |

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

Clusterwide Parameters (Feature - General)

| | | |
|---|------|------|
| Call Park Display Timer * | 10 | 10 |
| Caller ID Display Priority Enabled * | True | True |
| Call Park Reversion Timer * | 60 | 60 |
| Park Monitoring Reversion Timer * | 60 | 60 |
| Park Monitoring Periodic Reversion Timer * | 30 | 30 |
| Park Monitoring Forward No Retrieve Timer * | 300 | 300 |
| Preserve globalCallId for Parked Calls * | True | True |
| Maximum Call Duration Timer * | 720 | 720 |
| Maximum Hold Duration Timer * | 360 | 360 |

Figure 14: Service Parameters (Cont.)



| | | |
|---|------------------------------------|------------------------------------|
| Party Entrance Tone * | True | True |
| Message Waiting Lamp Policy * | Primary Line - Light and Prompt | Primary Line - Light and Prompt |
| Audible Message Waiting Indication Policy * | OFF | OFF |
| Message Waiting Indicator Inbound Calling Search Space | < None > | |
| Multiple Tenant MWI Modes * | False | False |
| MWI Non Message Center Signaling Call Duration * | 0 | 0 |
| Message Waiting Indicator APDU Digit Translation CSS | < None > | |
| Block OffNet To OffNet Transfer * | False | False |
| Use Original Call Classification for Transferred Calls * | False | False |
| Use Restriction attribute of ID/Name Presentation of Transferring Party * | True | True |
| Local route group for redirected calls * | Local route group of calling party | Local route group of calling party |
| Block Unencrypted Calls * | False | False |

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

Clusterwide Parameters (Feature - Conference)

| | | |
|---|---|---|
| Suppress MOH to Conference Bridge * | True | True |
| Drop Ad Hoc Conference * | Never | Never |
| Maximum Ad Hoc Conference * | 4 | 4 |
| Maximum MeetMe Conference Unicast * | 4 | 4 |
| Advanced Ad Hoc Conference Enabled * | False | False |
| Choose Encrypted Audio Conference Instead Of Video Conference * | True | True |
| Minimum Video Capable Participants To Allocate Video Conference * | 2 | 2 |
| Enable Click-to-Conference for Third-Party Applications * | False | False |
| IMS Conference Factory URL * | cucm-conference-factory@cucm1.company.com | cucm-conference-factory@cucm1.company.com |
| Cluster Conferencing Prefix Identifier | | |

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

Clusterwide Parameters (Feature - Call Secure Status Policy)

| | | |
|---|---|---|
| Secure Call Icon Display Policy * | All media except BFCP and iX transports must be encrypted | All media except BFCP and iX transports must be encrypted |
|---|---|---|

Clusterwide Parameters (Feature - Forward)

| | | |
|--|--------------------------------|--------------------------------|
| Forward Maximum Hop Count * | 12 | 12 |
| Forward No Answer Timer * | 12 | 12 |
| Max Forward Hops to DN * | 12 | 12 |
| Retain Forward Information * | False | False |
| Forward By Reroute Enabled * | False | False |
| Transform Forward by Reroute Destination * | True | True |
| Always Forward Switch Voice Mail Calls * | True | True |
| Forward By Reroute T1 Timer * | 10 | 10 |
| Include Original Called Info for Q.SIG Call Diversions * | Only after the first diversion | Only after the first diversion |
| Set Private Numbering Plan for Call Forward * | False | False |
| Set Type of Number for Call Forward * | Level1RegionalNumber | Level1RegionalNumber |
| Max Forward UnRegistered Hops to DN * | 0 | 0 |

Figure 15: Service Parameters (Cont.)



| | | |
|--|---------------------|---------------------|
| CFA CSS Activation Policy * | With Configured CSS | With Configured CSS |
| Cause Code When Maximum Forward Hop Count is Triggered * | Normal Unspecified | Normal Unspecified |

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

Clusterwide Parameters (Feature - Hold Reversion)

| | | |
|--|-------|-------|
| Hold Reversion Duration * | 0 | 0 |
| Hold Reversion Notification Interval * | 30 | 30 |
| CFA Destination Override * | False | False |

Clusterwide Parameters (Feature - Call Pickup)

| | | |
|---|-------|-------|
| Auto Call Pickup Enabled * | False | False |
| Call Pickup Locating Timer * | 1 | 1 |
| Call Pickup No Answer Timer * | 12 | 12 |

Clusterwide Parameters (Feature - Refer)

| | | |
|---|-------------------------------------|-------------------------------------|
| Validate Refer-to URI * | Validate Except for Anonymous Users | Validate Except for Anonymous Users |
|---|-------------------------------------|-------------------------------------|

Clusterwide Parameters (Feature - Replaces)

| | | |
|---|-------|-------|
| Block OffNet To OffNet Replaces * | False | False |
|---|-------|-------|

Clusterwide Parameters (Feature - Redirection [3xx])

| | | |
|--|----|----|
| Redirection Ring No Answer Reversion Timer * | 24 | 24 |
| Maximum Redirection Count * | 70 | 70 |

Clusterwide Parameters (Feature - Multilevel Precedence and Preemption)

| | | |
|---|--------------------|--------------------|
| Locations-based MLPP Enable * | False | False |
| Executive Override Call Preemptable * | False | False |
| Location-based Maximum Bandwidth Enforcement Level for MLPP Calls * | Lenient | Lenient |
| Non-Preemption Pattern CSS | < None > | |
| MLPP Exception Level * | Executive Override | Executive Override |

Clusterwide Parameters (Feature - Path Replacement)

| | | |
|---|----------|-------|
| Path Replacement Enabled * | False | False |
| Path Replacement on Tromboned Calls * | True | True |
| Start Path Replacement Minimum Delay Time * | 0 | 0 |
| Start Path Replacement Maximum Delay Time * | 0 | 0 |
| Path Replacement T1 Timer * | 30 | 30 |
| Path Replacement T2 Timer * | 15 | 15 |
| Path Replacement PINX ID | | |
| Path Replacement Calling Search Space | < None > | |

Clusterwide Parameters (Feature - Call Back)

| | | |
|--|---------------------------------|---------------------------------|
| Call Back Enabled Flag * | True | True |
| Call Back Notification Audio File Name * | CallBack.raw | CallBack.raw |
| Connection Proposal Type * | Connection Retention | Connection Retention |
| Connection Response Type * | Default to Connection Retention | Default to Connection Retention |

Figure 16: Service Parameters (Cont.)



| | | |
|--|----------------------|----------------------|
| Call Back Request Protection T1 Timer * | 10 | 10 |
| Call Back Recall T3 Timer * | 20 | 20 |
| Call Back Calling Search Space | < None > | |
| No Path Reservation * | True | True |
| Set Private Numbering Plan for Call Back * | False | False |
| Set Type of Number for Call Back * | Level1RegionalNumber | Level1RegionalNumber |

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

Clusterwide Parameters (Feature - Call Recording)

| | | |
|--|-------|-------|
| Play Recording Notification Tone To Observed Target * | False | False |
| Play Recording Notification Tone To Observed Connected Parties * | False | False |

Clusterwide Parameters (Feature - Monitoring)

| | | |
|---|-------|-------|
| Play Monitoring Notification Tone To Observed Target * | False | False |
| Play Monitoring Notification Tone To Observed Connected Parties * | False | False |

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

| | | |
|---|-------|-------|
| Join Across Lines Policy * | Off | Off |
| Single Button Barge/CBarge Policy * | Off | Off |
| Allow Barging When Ringing * | False | False |

Clusterwide Parameters (Feature - Secure Tone)

| | | |
|---|-------|-------|
| Play Tone to Indicate Secure/Non-Secure Call Status * | False | False |
|---|-------|-------|

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

Clusterwide Parameters (Feature - External Call Control)

| | | |
|---|-------------------------------------|-------------------------------------|
| External Call Control Diversion Maximum Hop Count * | 12 | 12 |
| Maximum External Call Control Diversion Hops to Pattern or DN * | 12 | 12 |
| External Call Control Routing Request Timer * | 2000 | 2000 |
| External Call Control Fully Qualified Role And Resource * | CISCO:UC:UCMPolicy:VoiceOrVideoCall | CISCO:UC:UCMPolicy:VoiceOrVideoCall |
| External Call Control Initial Connection Count To PDP * | 2 | 2 |
| External Call Control Maximum Connection Count To PDP * | 4 | 4 |
| Always use External Call Control-specified Called/Calling Party Names * | True | True |

Clusterwide Parameters (Route Plan)

| | | |
|---|-------|-------|
| Stop Routing on Out of Bandwidth Flag * | False | False |
| Stop Routing on Unallocated Number Flag * | True | True |
| Stop Routing on User Busy Flag * | True | True |

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

Clusterwide Parameters (Route Class Signaling)

| | | |
|---|-----------|-----------|
| Route Class Trunk Signaling Enabled * | True | True |
| SIP Route Class Naming Authority * | cisco.com | cisco.com |

Figure 17: Service Parameters (Cont.)



| Clusterwide Parameters (Hunt List) | | |
|--|-------|-------|
| Stop Hunting on Out of Bandwidth Flag * | False | False |
| Use Pickup Group Of Line Group Member DN * | False | False |

| Clusterwide Parameters (External QoS) | | |
|--|-------|-------|
| External QoS Enabled * | False | False |

| Clusterwide Parameters (Service) | | |
|--|-------|-------|
| Default Network Hold MOH Audio Source ID * | 1 | 1 |
| Default User Hold MOH Audio Source ID * | 1 | 1 |
| Duplex Streaming Enabled * | True | False |
| Media Exchange Interface Capability Timer * | 8 | 8 |
| Send Multicast MOH in H.245 OLC Message * | True | True |
| Media Exchange Timer * | 12 | 12 |
| Media Exchange Stop Streaming Timer * | 8 | 8 |
| Open Video Channel Response Timer for SIP Interop * | 500 | 500 |
| Port Received Timer After Call Connection * | 500 | 500 |
| Media Resource Allocation Timer * | 12 | 12 |
| MTP and Transcoder Resource Throttling Percentage * | 95 | 95 |
| Intercluster Capabilities Mismatch Timer * | 1000 | 1000 |
| Silence Suppression * | False | False |
| Silence Suppression for Gateways * | False | False |
| Strip G.729 Annex B (Silence Suppression) from Capabilities * | False | False |
| Enable Source IP Address Verification for Software Media Devices * | True | True |

| Clusterwide Parameters (System - General) | | |
|---|---------|---------|
| Always Use Dial Tone Setting * | Default | Default |
| Restart Cisco CallManager on Initialization Exception * | True | True |
| Digit Analysis Timer * | 6 | 6 |
| Statistics Enabled * | True | True |

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

| Clusterwide Parameters (System - QOS) | | |
|--|-----------------|-----------------|
| Priority Class * | Normal Priority | Normal Priority |
| DSCP for Audio Calls * | 46 (101110) | 46 (101110) |
| DSCP for Video Calls * | 34 (100010) | 34 (100010) |
| DSCP for Audio Portion of Video Calls * | 34 (100010) | 34 (100010) |
| DSCP for TelePresence Calls * | 32 (100000) | 32 (100000) |
| DSCP for Audio Portion of TelePresence Calls * | 32 (100000) | 32 (100000) |
| DSCP for Priority Audio Calls * | 45 (101101) | 45 (101101) |
| DSCP for Immediate Audio Calls * | 44 (101100) | 44 (101100) |
| DSCP for Flash Audio Calls * | 41 (101001) | 41 (101001) |

Figure 18: Service Parameters (Cont.)



| | | |
|---|-------------------------|---------------------------|
| DSCP for Flash Override Audio Calls * | 42 (101010) | ▼ 42 (101010) |
| DSCP for Executive Override Audio Calls * | 42 (101010) | ▼ 42 (101010) |
| DSCP for Priority Video Calls * | 39 (100111) | ▼ 39 (100111) |
| DSCP for Immediate Video Calls * | 37 (100101) | ▼ 37 (100101) |
| DSCP for Flash Video Calls * | 35 (100011) | ▼ 35 (100011) |
| DSCP for Flash Override Video Calls * | 33 (100001) | ▼ 33 (100001) |
| DSCP for Executive Override Video Calls * | 33 (100001) | ▼ 33 (100001) |
| DSCP for G.Clear Calls * | 46 (101110) | ▼ 46 (101110) |
| DSCP for Priority G.Clear Calls * | 45 (101101) | ▼ 45 (101101) |
| DSCP for Immediate G.Clear Calls * | 44 (101100) | ▼ 44 (101100) |
| DSCP for Flash G.Clear Calls * | 41 (101001) | ▼ 41 (101001) |
| DSCP for Flash Override G.Clear Calls * | 42 (101010) | ▼ 42 (101010) |
| DSCP for Executive Override G.Clear Calls * | 42 (101010) | ▼ 42 (101010) |
| DSCP for Audio Calls when RSVP Fails * | 0 (000000) | ▼ 0 (000000) |
| DSCP for Video Calls when RSVP Fails * | 0 (000000) | ▼ 0 (000000) |
| DSCP for ICCP Protocol Links * | 24 (011000) | ▼ 24 (011000) |
| Clusterwide Parameters (System - SDL) | | |
| SDL Listening Port Number * | 8002 | 8002 |
| SDL Max Router Latency * | 20 | 20 |
| Suppress Debug Info for Router Death * | 0 | 0 |
| Asynchronous SDL Logging Enabled * | False | ▼ False |
| Clusterwide Parameters (System - Location and Region) | | |
| Enforce Millisecond Packet Size * | True | ▼ True |
| Locations Trace Details Enabled * | False | ▼ False |
| Preferred G.711 Millisecond Packet Size * | 20 | ▼ 20 |
| Preferred G.722 Millisecond Packet Size * | 20 | ▼ 20 |
| Preferred G.723.1 Millisecond Packet Size * | 30 | ▼ 30 |
| Preferred G.729 Millisecond Packet Size * | 20 | ▼ 20 |
| Always Use Preferred G.729 Packet Size For SIP Trunk Answers * | False | ▼ False |
| Preferred GSM EFR Bytes Packet Size * | 31 | ▼ 31 |
| G.711 A-law Codec Enabled * | Enabled for All Devices | ▼ Enabled for All Devices |
| G.711 mu-law Codec Enabled * | Enabled for All Devices | ▼ Enabled for All Devices |
| G.722 Codec Enabled * | Enabled for All Devices | ▼ Enabled for All Devices |
| iLBC Codec Enabled * | Enabled for All Devices | ▼ Enabled for All Devices |
| iSAC Codec Enabled * | Enabled for All Devices | ▼ Enabled for All Devices |
| Opus Codec Enabled * | Enabled for All Devices | ▼ Enabled for All Devices |
| Default Intra-region Max Audio Bit Rate * | 64 kbps (G.722, G.711) | ▼ 64 kbps (G.722, G.711) |
| Default Inter-region Max Audio Bit Rate * | 8 kbps (G.729) | ▼ 8 kbps (G.729) |
| Default Intra-region Max Video Call Bit Rate (Includes Audio) * | 384 | 384 |
| Default Inter-region Max Video Call Bit Rate (Includes Audio) * | 384 | 384 |
| Default Intra-region Max Immersive Video Call Bit Rate (Includes Audio) * | 2000000000 | 2000000000 |
| Default Inter-region Max Immersive Video Call Bit Rate (Includes Audio) * | 2000000000 | 2000000000 |

Figure 19: Service Parameters (Cont.)



| | | |
|--|--------------------------|----------------------------|
| Default Interregion Max Immersive Video Call Bit Rate (Includes Audio) * | 2000000000 | 2000000000 |
| Use Video BandwidthPool for Immersive Video Calls * | True | ▼ True |
| Default Intraregion and Interregion Link Loss Type * | Low Loss | ▼ Low Loss |
| Default Audio Codec List between Regions * | Factory Default low loss | ▼ Factory Default low loss |
| Default Audio Codec List within Region * | Factory Default low loss | ▼ Factory Default low loss |
| Accept Audio Codec Preferences in Received Offer * | Off | ▼ Off |
| G.Clear Bandwidth Override * | False | ▼ False |
| Clusterwide Parameters (System - CCM Automated Alternate Routing) | | |
| Automated Alternate Routing Enable * | False | ▼ False |
| Clusterwide Parameters (System - RSVP) | | |
| Default inter-location RSVP Policy * | No Reservation | ▼ No Reservation |
| RSVP Retry Timer * | 60 | 60 |
| Mandatory RSVP Mid-call Retry Counter * | 1 | 1 |
| Mandatory RSVP mid call error handle option * | Call becomes best effort | ▼ Call becomes best effort |
| RSVP Video Tspec Burst Size Factor * | 5 | 5 |
| MLPP EXECUTIVE_OVERRIDE To RSVP Priority Mapping * | 65535 | 65535 |
| MLPP FLASH_OVERRIDE To RSVP Priority Mapping * | 65534 | 65534 |
| MLPP FLASH To RSVP Priority Mapping * | 65533 | 65533 |
| MLPP IMMEDIATE To RSVP Priority Mapping * | 65532 | 65532 |
| MLPP PL PRIORITY To RSVP Priority Mapping * | 65531 | 65531 |
| MLPP PL ROUTINE To RSVP Priority Mapping * | 65530 | 65530 |
| RSVP Audio Application ID * | AudioStream | AudioStream |
| RSVP Video Application ID * | VideoStream | VideoStream |
| RSVP Response Timer * | 2 | 2 |
| TLS Packet Capture Configurations | | |
| Packet Capture Enable * | False | ▼ False |
| Packet Capture Max File Size (MB) * | 2 | 2 |
| Clusterwide Parameters(System - Presence) | | |
| Presence Subscription Throttling Threshold * | 60000 | 60000 |
| Presence Subscription Resume Threshold * | 80 | 80 |
| Default Inter-Presence Group Subscription * | Disallow Subscription | ▼ Disallow Subscription |
| BLF Status Depicts DND * | False | ▼ False |
| Clusterwide Parameters (System - Mobility) | | |
| Enterprise Feature Access Code for Hold * | *81 | *81 |
| Enterprise Feature Access Code for Exclusive Hold * | *82 | *82 |
| Enterprise Feature Access Code for Resume * | *83 | *83 |
| Enterprise Feature Access Code for Transfer * | *84 | *84 |
| Enterprise Feature Access Code for Conference * | *85 | *85 |

Figure 20: Service Parameters (Cont.)



| | | |
|--|---|---|
| Enterprise Feature Access Code for Session Handoff * | *74 | *74 |
| Enterprise Feature Access Code for Starting Selective Recording * | *86 | *86 |
| Enterprise Feature Access Code for Stopping Selective Recording * | *87 | *87 |
| Smart Mobile Phone Interdigit Timer * | 500 | 500 |
| Non-Smart Mobile Phone Interdigit Timer * | 2000 | 2000 |
| Send Call to Mobile Menu Timer * | 60 | 60 |
| SIP Dual Mode Alert Timer * | 1500 | 1500 |
| Call Screening Timer * | 4000 | 4000 |
| Session Resumption Await Timer * | 180 | 180 |
| Inbound Calling Search Space for Remote Destination * | Trunk or Gateway Inbound Calling Search Space | Trunk or Gateway Inbound Calling Search Space |
| Enable Enterprise Feature Access * | False | False |
| Dial-via-Office Forward Service Access Number | | |
| Enable Mobile Voice Access * | False | False |
| Mobile Voice Access Number | | |
| Matching Caller ID with Remote Destination * | Complete Match | Complete Match |
| Number of Digits for Caller ID Partial Match * | 10 | 10 |
| Enable Use of Called Party Transformed Number for Mobile-terminated Calls * | False | False |
| Honor Gateway or Trunk Outbound Calling Party Selection for Mobile Connect Calls * | False | False |
| Clusterwide Parameters (System - Mobility Single Number Reach Voicemail) | | |
| Single Number Reach Voicemail Policy * | Timer Control | Timer Control |
| Dial-via-Office Reverse Voicemail Policy * | Timer Control | Timer Control |
| User Control Delayed Announcement Timer * | 1000 | 1000 |
| User Control Confirmed Answer Indication Timer * | 10000 | 10000 |
| Clusterwide Parameters (Feature - Reroute Remote Destination Calls to Enterprise Number) | | |
| Reroute Remote Destination Calls to Enterprise Number * | False | False |
| Ring All Shared Lines * | False | False |
| Ignore Call Forward All on Enterprise DN * | True | True |
| Clusterwide Parameters (Feature - Immediate Divert) | | |
| Use Legacy Immediate Divert * | True | True |
| Allow QSIG during iDivert * | False | False |
| Immediate Divert User Response Timer * | 5 | 5 |

Figure 21: Service Parameters (Cont.)



| Clusterwide Parameters (Call Admission Control) | | |
|---|-----------------|-----------------|
| Call Counting CAC Enabled * | False | False |
| Audio Bandwidth For Call Counting CAC * | 102 | 102 |
| Video Bandwidth For Call Counting CAC * | 500 | 500 |
| UCM to LBM Periodic Reservation Refresh Timer * | 5 | 5 |
| Maximum Bandwidth Deduction Duration * | 720 | 720 |
| Call Treatment When No LBM Available * | Allow Calls | Allow Calls |
| Locations Media Resource Audio Bit Rate Policy * | Lowest Bit Rate | Lowest Bit Rate |
| Video Call QoS Marking Policy * | Default | Default |
| Deduct Audio Bandwidth Portion from Audio Pool for a Video Call * | False | False |

| Clusterwide Parameters (Emergency Calling for Require Off-premise Location) | |
|---|----------|
| Alternate Destination for Emergency Call | |
| Alternate Calling Search Space for Emergency Call | < None > |

Figure 22: Service Parameters (Cont.)



Offnet Calls via TalkTalk SIP Trunk

Off-net calls are served by SIP trunks configured between Cisco UCM and TalkTalk Network and calls are routed via Cisco UBE

SIP Trunk Security Profile

Navigation Path: System > Security > SIP Trunk Security Profile

Name*= TalkTalk Non Secure SIP Trunk Profile

Description = non Secure SIP Trunk Profile authenticated by null String

The screenshot shows the 'SIP Trunk Security Profile Configuration' page. The 'Name*' field is 'TalkTalk Non Secure SIP Trunk Profile'. The 'Description' field is 'Non Secure SIP Trunk Profile authenticated by null String'. The 'Device Security Mode' is 'Non Secure'. The 'Incoming Transport Type*' is 'TCP+UDP' and the 'Outgoing Transport Type' is 'UDP'. There are checkboxes for 'Enable Digest Authentication', 'Enable Application level authorization', 'Accept presence subscription', 'Accept out-of-dialog refer**', 'Accept unsolicited notification', 'Accept replaces header', 'Transmit security status', and 'Allow charging header'. The 'SIP V.150 Outbound SDP Offer Filtering*' dropdown is set to 'Use Default Filter'.

Figure 23: SIP Trunk Security Profile

Explanation

| Parameter | Value | Description |
|-------------------------|-----------|--|
| Incoming Transport Type | TCP + UDP | |
| Outgoing Transport Type | UDP | SIP trunks to TalkTalk SBC 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. |



SIP Profile Configuration

SIP Profile will be later associated with the SIP trunk

Navigation Path: Device > Device Settings > SIP Profile

Name* = TalkTalk Standard SIP Profile

Description = Default SIP Profile

System ▾ Call Routing ▾ Media Resources ▾ Advanced Features ▾ Device ▾ Application ▾ User Management ▾ Help ▾

SIP Profile Configuration Related Links: [Back To Find/List](#)

Save Delete Copy Reset Apply Config Add New

Status

Status: Ready

All SIP devices using this profile must be restarted before any changes will take affect.

SIP Profile Information

Name*

TalkTalk Standard SIP Profile

Description

Default SIP Profile

Default MTP Telephony Event Payload Type*

101

Early Offer for G.Clear Calls*

Disabled ▾

User-Agent and Server header information*

Send Unified CM Version Information as User-Agen ▾

Version in User Agent and Server Header*

Major And Minor ▾

Dial String Interpretation*

Phone number consists of characters 0-9, *, #, and ▾

Confidential Access Level Headers*

Disabled ▾

☐ Redirect by Application

☒ Disable Early Media on 180

☐ Outgoing T.38 INVITE include audio mline

☐ Use Fully Qualified Domain Name in SIP Requests

☐ Assured Services SIP conformance

SDP Information

SDP Session-level Bandwidth Modifier for Early Offer and Re-invites*

TIAS and AS ▾

SDP Transparency Profile

Pass all unknown SDP attributes ▾

Accept Audio Codec Preferences in Received Offer*

Default ▾

☒ Require SDP Inactive Exchange for Mid-Call Media Change

☐ Allow RR/RS bandwidth modifier (RFC 3556)

Parameters used in Phone

Timer Invite Expires (seconds)*

180

Timer Register Delta (seconds)*

5

Timer Register Expires (seconds)*

3600

Timer T1 (msec)*

500

Timer T2 (msec)*

4000

Retry INVITE*

6

Retry Non-INVITE*

10

Media Port Ranges

☒ Common Port Range for Audio and Video

☐ Separate Port Ranges for Audio and Video

Start Media Port*

16384

Stop Media Port*

32766

Figure 24: SIP Profile



| | |
|---|-----------------------------|
| DSCP for Audio Calls | Use System Default |
| DSCP for Video Calls | Use System Default |
| DSCP for Audio Portion of Video Calls | Use System Default |
| DSCP for TelePresence Calls | Use System Default |
| DSCP for Audio Portion of TelePresence Calls | Use System Default |
| Call Pickup URI* | x-cisco-serviceuri-pickup |
| Call Pickup Group Other URI* | x-cisco-serviceuri-opickup |
| Call Pickup Group URI* | x-cisco-serviceuri-gpickup |
| Meet Me Service URI* | x-cisco-serviceuri-meetme |
| User Info* | None |
| DTMF DB Level* | Nominal |
| Call Hold Ring Back* | Off |
| Anonymous Call Block* | Off |
| Caller ID Blocking* | Off |
| Do Not Disturb Control* | User |
| Telnet Level for 7940 and 7960* | Disabled |
| Resource Priority Namespace | < None > |
| Timer Keep Alive Expires (seconds)* | 120 |
| Timer Subscribe Expires (seconds)* | 120 |
| Timer Subscribe Delta (seconds)* | 5 |
| Maximum Redirections* | 70 |
| Off Hook To First Digit Timer (milliseconds)* | 15000 |
| Call Forward URI* | x-cisco-serviceuri-cfwdall |
| Speed Dial (Abbreviated Dial) URI* | x-cisco-serviceuri-abbrdial |

☒ Conference Join Enabled
☐ RFC 2543 Hold
☒ Semi Attended Transfer
☐ Enable VAD
☐ Stutter Message Waiting
☐ MLPP User Authorization

Normalization Script
 Normalization Script < None >
☐ Enable Trace

| | Parameter Name | Parameter Value | |
|---|----------------|-----------------|---|
| 1 | | | <input type="button" value="+"/> <input type="button" value="-"/> |

Incoming Requests FROM URI Settings
 Caller ID DN
 Caller Name

Figure 25: SIP Profile (Cont.)



Trunk Specific Configuration

Reroute Incoming Request to new Trunk based on* Never ▼

Resource Priority Namespace List < None > ▼

SIP Rel1XX Options* Send PRACK if 1xx Contains SDP ▼

Video Call Traffic Class* Mixed ▼

Calling Line Identification Presentation* Default ▼

Session Refresh Method* Invite ▼

Early Offer support for voice and video calls* Best Effort (no MTP inserted) ▼

☐ Enable ANAT

☐ Deliver Conference Bridge Identifier

☐ Allow Passthrough of Configured Line Device Caller Information

☐ Reject Anonymous Incoming Calls

☐ Reject Anonymous Outgoing Calls

☐ Send ILS Learned Destination Route String

SIP OPTIONS Ping

☒ Enable OPTIONS Ping to monitor destination status for Trunks with Service Type "None (Default)"

Ping Interval for In-service and Partially In-service Trunks (seconds)* 60

Ping Interval for Out-of-service Trunks (seconds)* 120

Ping Retry Timer (milliseconds)* 500

Ping Retry Count* 6

SDP Information

☒ Send send-receive SDP in mid-call INVITE

☐ Allow Presentation Sharing using BFCP

☐ Allow iX Application Media

☐ Allow multiple codecs in answer SDP

Figure 26: SIP Profile (Cont.)

Explanation

| Parameter | Value | Description |
|--|-----------------------------|---|
| Default MTP Telephony Event Payload Type | 101 | RFC2833 DTMF payload type |
| SIP Rel1XX Options | Send PRACK for 1xx Messages | Enable Provisional Acknowledgements (Reliable 100 messages) |
| Ping Interval for In-service and Partially In-service Trunks (seconds) | 60 | OPTIONS message parameters- interval time |
| Ping Interval for Out-of-service Trunks (seconds) | 120 | OPTIONS message parameters- interval time |



SIP Trunk Configuration

Create SIP trunks to Cisco UBE

Navigation Path: Device > Trunk

System ▾ Call Routing ▾ Media Resources ▾ Advanced Features ▾ Device ▾ Application ▾ User Management ▾ Help ▾

Find and List Trunks

+

Add New

Select All

Clear All

✖ Delete Selected

↺ Reset Selected

Trunks (1 - 7 of 7)

Rows per Page 50 ▾

Find Trunks where

Device Name ▾ begins with ▾ Find Clear Filter

⊕

⊖

Select item or enter search text ▾







| <input type="checkbox"/> | | Name ^ | Description | Calling Search Space | Device Pool | Route Pattern | Partition | Route Group | Priority | Trunk Type | SIP Trunk Status | SIP Trunk Duration | SIP Trunk Security Profile |
|--------------------------|---|--|----------------------------|----------------------|---------------------------|------------------------|-----------|-------------|----------|------------|------------------|---|---|
| <input type="checkbox"/> |  | SIP_trunk_to_fax_gateway | SIP_trunk_to_fax_gateway | | G711_pool | 6106 | | | | SIP Trunk | Full Service | Time In Full Service: 6 days 0 hour 23 minutes | TalkTalk Non Secure SIP Trunk Profile |
| <input type="checkbox"/> |  | TalkTalk | SIP trunk to TalkTalk CUBE | | G711_pool | [19]XX | | | | SIP Trunk | Full Service | Time In Full Service: 0 day 18 hours 51 minutes | TalkTalk Non Secure SIP Trunk Profile |
| <input type="checkbox"/> |  | TalkTalk | SIP trunk to TalkTalk CUBE | | G711_pool | 9.610X | | | | SIP Trunk | Full Service | Time In Full Service: 0 day 18 hours 51 minutes | TalkTalk Non Secure SIP Trunk Profile |
| <input type="checkbox"/> |  | TalkTalk | SIP trunk to TalkTalk CUBE | | G711_pool | 118500 | | | | SIP Trunk | Full Service | Time In Full Service: 0 day 18 hours 51 minutes | TalkTalk Non Secure SIP Trunk Profile |
| <input type="checkbox"/> |  | TalkTalk | SIP trunk to TalkTalk CUBE | | G711_pool | 8.1 | | | | SIP Trunk | Full Service | Time In Full Service: 0 day 18 hours 51 minutes | TalkTalk Non Secure SIP Trunk Profile |
| <input type="checkbox"/> |  | TalkTalk | SIP trunk to TalkTalk CUBE | | G711_pool | 9.001@ | | | | SIP Trunk | Full Service | Time In Full Service: 0 day 18 hours 51 minutes | TalkTalk Non Secure SIP Trunk Profile |

Figure 27: SIP Trunks List



Trunk Configuration

Related Links: [Back To Find/List](#)

Save Delete Reset Add New

-Status-

Status: Ready

-SIP Trunk Status-

Service Status: Full Service
Duration: Time In Full Service: 0 day 0 hour 1 minute

-Device Information-

| | |
|---|------------------------------|
| Product: | SIP Trunk |
| Device Protocol: | SIP |
| Trunk Service Type | None(Default) |
| Device Name* | TalkTalk |
| Description | SIP trunk to TalkTalk CUBE |
| Device Pool* | G711_pool |
| Common Device Configuration | < None > |
| Call Classification* | Use System Default |
| Media Resource Group List | MRGL_Default |
| Location* | Hub_None |
| AAR Group | < None > |
| Tunneled Protocol* | None |
| QSIG Variant* | No Changes |
| ASN.1 ROSE OID Encoding* | No Changes |
| Packet Capture Mode* | None |
| Packet Capture Duration | 0 |
| <input type="checkbox"/> Media Termination Point Required | |
| <input checked="" type="checkbox"/> Retry Video Call as Audio | |
| <input type="checkbox"/> Path Replacement Support | |
| <input type="checkbox"/> Transmit UTF-8 for Calling Party Name | |
| <input type="checkbox"/> Transmit UTF-8 Names in QSIG APDU | |
| <input type="checkbox"/> Unattended Port | |
| <input type="checkbox"/> SRTP Allowed - When this flag is checked, Encrypted TLS needs to be configured in the network to provide end to end security. Failure to do so will expose keys and other information. | |
| Consider Traffic on This Trunk Secure* | When using both sRTP and TLS |
| Route Class Signaling Enabled* | Default |
| Use Trusted Relay Point* | Default |
| <input checked="" type="checkbox"/> PSTN Access | |
| <input checked="" type="checkbox"/> Run On All Active Unified CM Nodes | |

Figure 28: SIP Trunk to Cisco UBE



Intercompany Media Engine (IME)

E.164 Transformation Profile < None >

MLPP and Confidential Access Level Information

MLPP Domain < None >

Confidential Access Mode < None >

Confidential Access Level < None >

Call Routing Information

☒ Remote-Party-Id

☒ Asserted-Identity

Asserted-Type* Default

SIP Privacy* Default

Inbound Calls

Significant Digits* 4

Connected Line ID Presentation* Default

Connected Name Presentation* Default

Calling Search Space < None >

AAR Calling Search Space < None >

Prefix DN

☒ Redirecting Diversion Header Delivery - Inbound

Incoming Calling Party Settings

If the administrator sets the prefix to Default this indicates call processing will use prefix at the next level setting (DevicePool/Service Parameter). Otherwise, the value configured is used as the prefix unless the field is empty in which case there is no prefix assigned.

[Clear Prefix Settings](#)

[Default Prefix Settings](#)

| Number Type | Prefix | Strip Digits | Calling Search Space | Use Device Pool CSS |
|-----------------|----------------------|----------------|-----------------------------|-------------------------------------|
| Incoming Number | Default | 0 | < None > | <input checked="" type="checkbox"/> |

Incoming Called Party Settings

If the administrator sets the prefix to Default this indicates call processing will use prefix at the next level setting (DevicePool/Service Parameter). Otherwise, the value configured is used as the prefix unless the field is empty in which case there is no prefix assigned.

[Clear Prefix Settings](#)

[Default Prefix Settings](#)

| Number Type | Prefix | Strip Digits | Calling Search Space | Use Device Pool CSS |
|-----------------|----------------------|----------------|-----------------------------|-------------------------------------|
| Incoming Number | Default | 0 | < None > | <input checked="" type="checkbox"/> |

Figure 29: SIP Trunk to Cisco UBE (cont.)



Connected Party Settings
Connected Party Transformation CSS < None >
☒ Use Device Pool Connected Party Transformation CSS

Outbound Calls
Called Party Transformation CSS < None >
☒ Use Device Pool Called Party Transformation CSS
Calling Party Transformation CSS < None >
☒ Use Device Pool Calling Party Transformation CSS
Calling Party Selection* First Redirect Number (External)
Calling Line ID Presentation* Default
Calling Name Presentation* Default
Calling and Connected Party Info Format* Deliver DN only in connected party
☒ Redirecting Diversion Header Delivery - Outbound
Redirecting Party Transformation CSS < None >
☒ Use Device Pool Redirecting Party Transformation CSS

Caller Information
Caller ID DN
Caller Name
☐ Maintain Original Caller ID DN and Caller Name in Identity Headers

SIP Information
Destination
☐ Destination Address is an SRV

| | Destination Address | Destination Address IPv6 | Destination Port |
|----|---------------------|--------------------------|------------------|
| 1* | 10.80.18.20 | | 5060 |

MTP Preferred Originating Codec* 711ulaw
BLF Presence Group* Standard Presence group
SIP Trunk Security Profile* TalkTalk Non Secure SIP Trunk Profile
Rerouting Calling Search Space < None >
Out-Of-Dialog Refer Calling Search Space < None >
SUBSCRIBE Calling Search Space < None >
SIP Profile* TalkTalk Standard SIP Profile [View Details](#)
DTMF Signaling Method* No Preference

Normalization Script
Normalization Script < None >
☐ Enable Trace

| | Parameter Name | Parameter Value | |
|---|----------------|-----------------|---|
| 1 | | | <input type="button" value="+"/> <input type="button" value="-"/> |

Recording Information
☒ None
☐ This trunk connects to a recording-enabled gateway
☐ This trunk connects to other clusters with recording-enabled gateways

Geolocation Configuration
Geolocation < None >
Geolocation Filter < None >
☐ Send Geolocation Information

Figure 30: SIP Trunk to Cisco UBE (Cont.)



Explanation

| Parameter | Value | Description |
|----------------------------|---------------------------------------|---|
| Device Name | TalkTalk | Name for the trunk |
| Device Pool | G711_pool | Default Device Pool is used for this trunk |
| Media Resource Group List | MRGL_Default | MRG with resources: ANN, CFB, MOH and MTP |
| Significant Digits | 4 | 4 digits Extension for all CPE phones |
| Destination Address | 10.80.18.20 | IP address of the Cisco UBE Virtual LAN |
| SIP Trunk Security Profile | TalkTalk Non Secure SIP Trunk Profile | SIP Trunk Security Profile configured earlier |
| SIP Profile | TalkTalk Standard SIP Profile | SIP Profile configured earlier |

Dial Plan

Route Pattern Configuration

Navigation: Call Routing > Route/Hunt > Route Pattern

Route patterns are configured as below:

- Cisco IP phone dial "9".001+10 digits number to access PSTN via Cisco UBE
 - "9" is removed before sending to Cisco UBE
- For FAX call, Access Code "8"+ 0091+10 digits number is used at Cisco Fax gateway
 - "8" is removed at Cisco UCM
 - The rest of the number is sent to Cisco UBE to TalkTalk network
- Incoming fax call to 6106 will be sent to Cisco Fax gateway
- Cisco IP phones dial 1XX and 9XX for emergency call and will send all digits to Cisco UBE to TalkTalk network

System ▾
Call Routing ▾
Media Resources ▾
Advanced Features ▾
Device ▾
Application ▾
User Management ▾
Help ▾

Find and List Route Patterns

+ Add New
Select All
Clear All
Delete Selected

- Status

1 records deleted
6 records found

Route Patterns (1 - 6 of 6)
Rows per Page 50 ▾

Find Route Patterns where
Pattern ▾
begins with ▾
Find
Clear Filter
+
-

| <input type="checkbox"/> | Pattern ^ | Description | Partition | Route Filter | Associated Device | Copy |
|--------------------------|------------------------|------------------------------|-----------|--------------|--|------|
| <input type="checkbox"/> | 118500 | information_services | | | TalkTalk | |
| <input type="checkbox"/> | 6106 | Route_pattern_to_fax_gateway | | | SIP trunk to fax gateway | |
| <input type="checkbox"/> | 8.! | international_calling | | | TalkTalk | |
| <input type="checkbox"/> | 9.001@ | to NA number | | | TalkTalk | |
| <input type="checkbox"/> | 9.610X | PBX_to_PBX_via_talk_talk | | | TalkTalk | |
| <input type="checkbox"/> | [19]XX | emergency_service_nos | | | TalkTalk | |



Figure 31: Route Patterns List

Route Pattern ConfigurationRelated Links: [Back To Find/List](#) ▼

Save Delete Copy Add New

- Pattern Definition -

Route Pattern*

9.001@

Route Partition

< None >

Description

to NA number

Numbering Plan*

NANP

Route Filter

< None >

MLPP Precedence*

Default

☐ Apply Call Blocking Percentage

Resource Priority Namespace Network Domain

< None >

Route Class*

Default

Gateway/Route List*

TalkTalk

[\(Edit\)](#)

Route Option

☒ Route this pattern

☐ Block this pattern

No Error

Call Classification*

OffNet

External Call Control Profile

< None >

☐ Allow Device Override

☒ Provide Outside Dial Tone

☐ Allow Overlap Sending

☐ Urgent Priority

☐ Require Forced Authorization Code

Authorization Level*

0

☐ Require Client Matter Code

- 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

PreDot

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-

Network Service Protocol

-- Not Selected --

Carrier Identification Code

Network Service

Service Parameter Name

Service Parameter Value

-- Not Selected --

< Not Exist >

Figure 32: Route Pattern for Voice



Route Pattern ConfigurationRelated Links: [Back To Find/List](#) ▾

Save Delete Copy Add New

-Pattern Definition-

Route Pattern*8.!

Route Partition< None > ▾

Descriptioninternational_calling

Numbering Plan-- Not Selected -- ▾

Route Filter< None > ▾

MLPP Precedence*Default ▾

☐ Apply Call Blocking Percentage

Resource Priority Namespace Network Domain< None > ▾

Route Class*Default ▾

Gateway/Route List*TalkTalk ▾ [\(Edit\)](#)

Route Option

☒ Route this pattern

☐ Block this pattern No Error ▾

Call Classification*OffNet ▾

External Call Control Profile< None > ▾

☐ Allow Device Override ☒ Provide Outside Dial Tone ☐ Allow Overlap Sending ☐ Urgent Priority

☐ Require Forced Authorization Code

Authorization Level*0

☐ Require Client Matter Code

-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 DigitsPreDot ▾

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-

Network Service Protocol-- Not Selected -- ▾

Carrier Identification Code

Network Service-- Not Selected -- ▾

Service Parameter Name< Not Exist >

Service Parameter Value

Figure 33: Route Pattern for Voice (Cont.)



Route Pattern ConfigurationRelated Links: [Back To Find/List](#)

Save Delete Copy Add New

Pattern Definition

Route Pattern*

9.610X

Route Partition

< None >

Description

PBX_to_PBX_via_talk_talk

Numbering Plan

-- Not Selected --

Route Filter

< None >

MLPP Precedence*

Default

☐ Apply Call Blocking Percentage

Resource Priority Namespace Network Domain

< None >

Route Class*

Default

Gateway/Route List*

TalkTalk

[\(Edit\)](#)

Route Option

☒ Route this pattern

☐ Block this pattern

No Error

Call Classification*

OffNet

External Call Control Profile

< None >

☐ Allow Device Override ☒ Provide Outside Dial Tone ☐ Allow Overlap Sending ☐ Urgent Priority

☐ Require Forced Authorization Code

Authorization Level*

0

☐ Require Client Matter Code

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

PreDot

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

Network Service Protocol

-- Not Selected --

Carrier Identification Code

Network Service

-- Not Selected --

Service Parameter Name

< Not Exist >

Service Parameter Value

Figure 34: Route Pattern for Voice (Cont.)

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Route Pattern ConfigurationRelated Links: [Back To Find/List](#)

Save Delete Copy Add New

Pattern Definition

Route Pattern*

[19]XX

Route Partition

< None >

Description

emergency_service_nos

Numbering Plan

-- Not Selected --

Route Filter

< None >

MLPP Precedence*

Default

☐ Apply Call Blocking Percentage

Resource Priority Namespace Network Domain

< None >

Route Class*

Default

Gateway/Route List*

TalkTalk

(Edit)

Route Option

☒ Route this pattern

☐ Block this pattern

No Error

Call Classification*

OffNet

External Call Control Profile

< None >

☐ Allow Device Override

☒ Provide Outside Dial Tone

☐ Allow Overlap Sending

☐ Urgent Priority

☐ Require Forced Authorization Code

Authorization Level*

0

☐ Require Client Matter Code

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

Network Service Protocol

-- Not Selected --

Carrier Identification Code

Network Service

-- Not Selected --

Service Parameter Name

< Not Exist >

Service Parameter Value

Figure 35: Route Pattern for Voice (Cont.)

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Route Pattern ConfigurationRelated Links: [Back To Find/List](#) ▾

Save Delete Copy Add New

-Pattern Definition-

Route Pattern*118500

Route Partition< None > ▾

Descriptioninformation_services

Numbering Plan-- Not Selected -- ▾

Route Filter< None > ▾

MLPP Precedence*Default ▾

☐ Apply Call Blocking Percentage

Resource Priority Namespace Network Domain< None > ▾

Route Class*Default ▾

Gateway/Route List*TalkTalk [\(Edit\)](#)

Route Option

☒ Route this pattern

☐ Block this pattern No Error ▾

Call Classification*OffNet ▾

External Call Control Profile< None > ▾

☐ Allow Device Override ☒ Provide Outside Dial Tone ☐ Allow Overlap Sending ☐ Urgent Priority

☐ Require Forced Authorization Code

Authorization Level*0

☐ Require Client Matter Code

-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-

Network Service Protocol-- Not Selected -- ▾

Carrier Identification Code

| Network Service | Service Parameter Name | Service Parameter Value |
|----------------------|------------------------|-------------------------|
| -- Not Selected -- ▾ | < Not Exist > | |

Figure 36: Route Pattern for Voice (Cont.)



Route Pattern ConfigurationRelated Links: [Back To Find/List](#)

Save

Delete

Copy

Add New

Pattern Definition

Route Pattern*

6106

Route Partition

< None >

Description

Route_pattern_to_fax_gateway

Numbering Plan

-- Not Selected --

Route Filter

< None >

MLPP Precedence*

Default

☐ Apply Call Blocking Percentage

Resource Priority Namespace Network Domain

< None >

Route Class*

Default

Gateway/Route List*

SIP_trunk_to_fax_gateway

[\(Edit\)](#)

Route Option

☒ Route this pattern

☐ Block this pattern

No Error

Call Classification*

OffNet

External Call Control Profile

< None >

☐ Allow Device Override

☒ Provide Outside Dial Tone

☐ Allow Overlap Sending

☐ Urgent Priority

☐ Require Forced Authorization Code

Authorization Level*

0

☐ Require Client Matter Code

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

0203750XXXX

Prefix Digits (Outgoing Calls)

Called Party Number Type*

Cisco CallManager

Called Party Numbering Plan*

Cisco CallManager

ISDN Network-Specific Facilities Information Element

Network Service Protocol

-- Not Selected --

Carrier Identification Code

Network Service

-- Not Selected --

Service Parameter Name

< Not Exist >

Service Parameter Value

Figure 37: Route Pattern for Fax



Explanation

| Setting | Value | Description |
|---------------------|---|---|
| Route Pattern | 9.001@ for Voice call, 8.! For International Calls, 6106 for fax call & 9.610X for CPE to CPE call via TALKTALK, 118500 for information services, [19]XX for operator call and emergency services | Specify appropriate Route Pattern |
| Gateway/Route List | TalkTalk for Route Pattern 9.001@, [19]XX ,118500, 9.610X, 8.!, SIP_trunk _to_ fax _ gateway for Route Pattern 6106 | SIP Trunk name configured earlier |
| Numbering Plan | NANP for Route Pattern 9.@ | North American Numbering Plan |
| Call Classification | Offnet for Route Pattern 9.001@, 6106, 118500, 9.610X, 8.! and [19]XX | Restrict the transferring of an external call to an external device |
| Discard Digits | PreDot for Route Pattern 9.001@, 8.! , 9.610X | Specifies how to modify digit before they are sending to TalkTalk network |

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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