Application Note

Skype for Business 2015 using SIP trunk (TLS) to Cisco Unified Communications Manager Release 11.5. (1) SU5

pplication Note

Table of Contents

Introduction 4
The following items were tested: 4
Listed below are the highlights of the integration issues: 4
Below are the key results: 5
Network Topology 5
Limitations 5
System Components 6
Hardware Requirements 6
Software Requirements 7
Features 7
Features Supported 7
Features Not Supported or Not Tested 8
Configuration 8
Global Trunk Configuration Highlights: 9
Configuring Sequence and Tasks: 9
Configuring the Skype for Business 10
Add Cisco UCM to Skype for Business Topology 10
Trunk Configuration 14
Route Configuration 18
Voice Policy and PSTN Usage Configuration 20
Dial Plan Configuration 22
Call Park Range Configuration 23
Global Media Bypass Configuration 24
User Configuration 25
Client Configuration 31
Configuring the Cisco Unified Communications Manager 32
SIP Trunk Security Profile for Trunk to Skype for Business 32
SIP Trunk Security Profile for Trunk to Unity Connection 34
SIP Profile 35
Media Resource Group 40
Media Resource Group List 44
Device Pool Configuration 46

Region Configuration 49
Normalization Script 50
SIP Trunk to Skype for Business Configuration 58
SIP Trunk to Cisco Unity Connection Configuration 63
Voice Mail Configuration 67
Route Pattern to Skype for Business Extensions 67
Route Pattern to invoke Jabber client with Remote Destination configured as Skype for Business Extensions 69
Route Pattern to Skype for Business Call Park range 72
Route Pattern to Unity Connection Voice Mail 74
Cisco UCM Extent and Connect 76
UC service Configuration 76
service Profile Configuration 78
Cisco Unified CM IM Presence – CCMCIP Profile Configuration 80
SIP trunk to Cisco IM&Presence Trunk Configuration 81
end user configuration 86
Remote Destination Configuration 91
Cisco UCM CTI Remote Device Configuration 92
Cisco Unity Connection 95
Telephony Integration – Add Phone System 95
Telephony Integration – Add Port Group 96
Telephony Integration – Add Ports 98
User Configuration 98
Acronyms 101

Introduction

This document describes the steps and configurations necessary for Cisco Unified Communications Manager (Cisco UCM) release 11.5.1 to interoperate with the Skype for Business 2015 on TLS using the following configuration:

The following items were tested:

- Basic call between the two systems and verification of voice path, using both SIP and Legacy phones on the Cisco side, and SIP client on the Skype for Business side (Refer to limitation section for more info)
- CLIP/CLIR/CNIP/CNIR features: Calling party Name and Number delivery (allowed and restricted) (Refer to limitation section for more info)
- COLP/CONP/COLR/CONR features: Connected Name and Number delivery (allowed and restricted) (Refer to limitation section for more info)
- Call Transfer: Attended and Early attended (Refer to limitation section for more info)
- Alerting Name Identification (Refer to limitation section for more info)
- Call forwarding: Call Forward Unconditional(CFU), Call Forward Busy (CFB), and Call Forward No Answer (CFNA)
- Hold and Resume with Music on Hold
- Three-way conferencing (Refer to limitation section for more info)
- Voice messaging and MWI activation-deactivation (Refer to limitation section for more info)
- Extend and Connect (Refer to limitation section for more info)
- Call Park (Refer to limitation section for more info)

Listed below are the highlights of the integration issues:

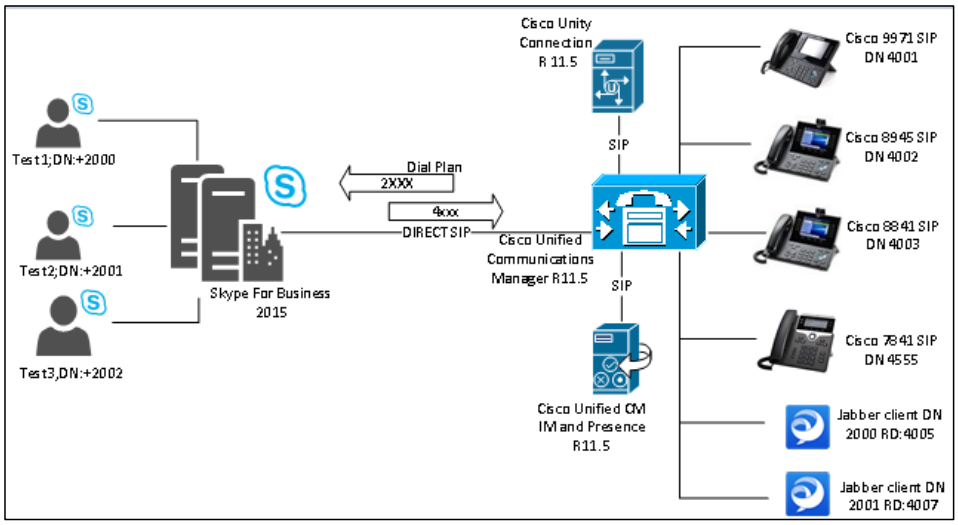
- Basic calls work from Cisco UCM to Skype for Business and vice versa. Testing has done with Cisco SIP phones only as SCCP phones do not support 80-bit crypto required by Skype for Business.
- Basic calls works from Cisco UCM to Skype for Business and vice versa using G711 ulaw and alaw
- Caller name and number is not updated correctly for basic calls and in the attended and early-attended transfer scenarios.
- Alerting name updates do not occur on Skype for Business.
- Video calls between the Cisco UCM and Skype for Business users were not tested.

- REFER support Skype for Business needs to be disabled to work Park scenario
- Skype for business does not support to update privacy:id sent by 18x and 2xx message sent from Cisco UCM

Below are the key results:

- Basic call, Call Transfer, Call Forwarding, Conference Call, and Hold and Resume tested successfully with a few caveats and limitations.
- Centralized voicemail, using Unity Connection server integrated with Cisco UCM via SIP is used for testing. This voicemail solution can provide centralized voicemail services, supporting both Skype for Business and Cisco end-users.

Network Topology



Limitations

These are the known limitations, caveats, or integration issues:

- Skype for Business do not support G729 codec. The trunk is tested with only G711 ulaw and alaw.
- Skype for Business and Cisco UCM do not support overlap dialing modes on their SIP endpoints
- Skype for Business does not support alerting name updates

- Skype for Business does not update the caller ID (connected Name) for a basic or privacy enabled call from the Cisco UCM. Therefore, only the connected party number is displayed on the Cisco UCM Phone
- Skype for Business does not consider Privacy ID, sent by Cisco UCM during 180 Ringing or 200 OK when Connected Name/ID is resticted on Cisco UCM. Subsequently, Skype for Business does not support updating Connected Party display as Private
- Skype for Business does not update the CLID in transfer/conference scenarios. After the transfer/conference is complete, Cisco UCM sends mid call INVITE and UPDATE messages that contain PAI and RPI. However, Skype for Business does not update this information on its clients
- Cisco CUM user parks a call with Skype for Business user. When another Skype for Business user retrieves that call, there is one-way audio. The Skype for Business user, who is retrieved from parking, could not hear the other user
- Skype for Business does not send PAI by default i.e. when restriction is not enabled
- Skype for Business sends incorrect details DN in history-info as a work around, DN configured in Skype for Business as prefix with "+"
- Cisco UCM Remote Destination is configured with a “+” prefix and a Route Pattern to route a DN with a preix ‘+’ is also added. (Refer Cisco UCM configuration section - Cisco Unified Communications Manager Route Pattern to invoke Jabber client with Remote Destination configured as Skype for Business Extensions)
- Skype for Business does not support MWI notification from Cisco Unity Connection. It responds with a “405 Method Not Allowed” to a NOTIFY Message from the Cisco UCM that has MWI information
- In Multiple Call Forwarding scenario between Skype for Business Users and Cisco UCM Users, wherein both originator and terminator being Skype for Business Users, originator does not display the Caller ID of terminator
- When Cisco UCM User completes the conference, the audio is on RTP. Pad-lock symbol on the Cisco phone disappears

System Components

Hardware Requirements

The following hardware are used:

- Cisco UCS-C240-M3S VMWare Host
- Cisco 8945,8841 ,7841, and 9971 IP phones

Software Requirements

The following software are used:

- Cisco UCSC-C240-M3S VMware vSphere Image Profile: ESXi-5.5.0-1331820-standard
- Cisco Unified Communications Manager release 11.5.1.15900-18
- Cisco Unified Communications Manager IM & P release 11.5.1.15900-33
- Cisco Unity Connection release 11.5.1.15900-18
- Cisco Jabber 12.5.1.2706 Build 277406
- Skype for Business 2015 6.0.9319.534
- Skype for Business Client version : 15.0.5111.1000

Features

This section lists supported and unsupported features. No deviation from the configuration presented in this document shall be supported by Cisco. Please see the Limitations section for more information.

Features Supported

- CLIP—calling line (number) identification presentation
- CLIR—calling line (number) identification restriction
- CNIP—calling Name identification presentation
- CNIR—calling Name identification restriction
- Attended call transfer
- Early attended call transfer
- CFU—call forwarding unconditional
- CFB—call forwarding busy
- CFNA—call forwarding no answer
- COLP—connected line (number) identification presentation
- COLR—connected line (number) identification restriction
- CONP—connected Name identification presentation
- CONR—connected Name identification restriction

- Hold and resume
- Conference call
- MWI—Message Waiting Indicator (only for Cisco Endpoints)
- Audio Codec Preference List
- Call Park/Pickup(see limitation section)
- Extend and Connect
- Shared Line on Cisco Endpoints

Features Not Supported or Not Tested

- G729 voice codec
- Call completion (callback, automatic callback)
- Shared Line on Skype for Business
- Message Waiting Indicator on Skype for Business Endpoints
- Blind transfer
- Video calls
- Scenarios that required theird PBX.
- Scenarios involving Non-SIP interfaces
- Scenarios involving Cisco UCM SCCP Phones
- Alerting Name on Skype for Business
- Connected party restriction sends and receives on Skype for Business Server is not supported

Configuration

The goal of this guide is to provide an overview of the integration between Cisco Unified Communication Manager and Skype for Business. The deployment will interconnect the UC systems using SIP. No PSTN connectivity has been tested with this integration. The following sections provide the required configurations for a successful integration.

Global Trunk Configuration Highlights:

Setting	Value
Skype for Business Media Bypass	DISABLED
Skype for Business Encryption Support	OPTIONAL
Skype for Business REFER Support	DISABLED
Cisco UCM SIP Trunk MTP	DISABLED
Cisco UCM PRACK	ENABLED
Cisco UCM Early Offer	ENABLED
Transport type Cisco UCM to Skype for Business	TLS
Cisco UCM SRTP allowed	Enabled

Configuring Sequence and Tasks:

Skype for Business:

Add Cisco UCM to Skype for Business Topology
Trunk Configuration
Route Configuration
Voice Policy and PSTN Usage Configuration
Dial Plan Configuration
Call Park range Configuration
Media Bypass Configuration
User Configuration
Client Configuration

Cisco Unified Communications Manager:

SIP trunk security profile
SIP profile
Media resource group and media resource group list
Assign media resource group list (MRGL) in the default device pool
Region configuration
Normalization script
SIP trunk to Skype for Business
SIP Trunk to Cisco Unity Connection
Assign User in Cisco Unity Connection
SIP phones device configuration
Route Group, Route List and SIP Route Pattern
Voice Mail
Route pattern to Skype for Business, Unity Connection and Skype for Business call Park Range

Extend and Connect Feature and User configuration

Cisco Unity Connection:

Cisco Unity Connection Telephony Integration
Cisco Unity Connection User Configuration

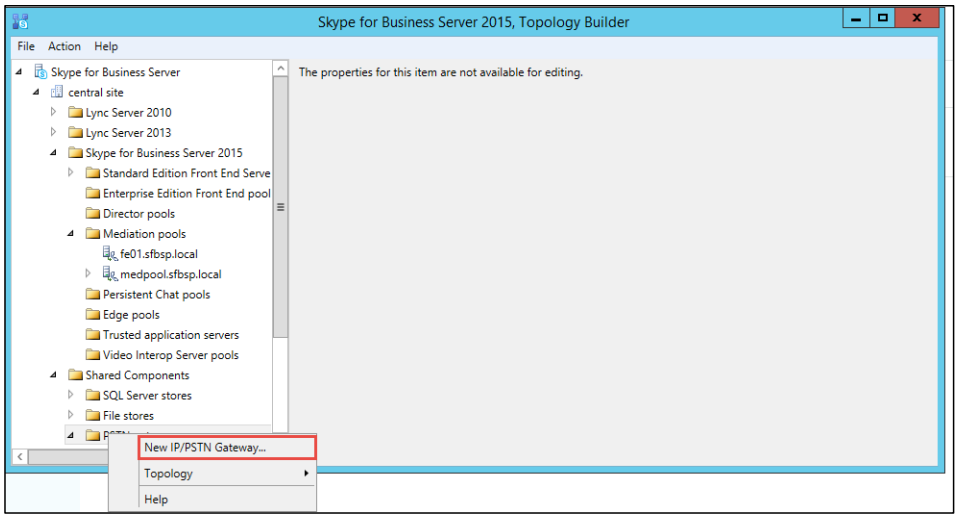
Configuring the Skype for Business

Add Cisco UCM to Skype for Business Topology

Run the Skype for Business 2015 Topology Builder as a user in the CSAdministrator group.

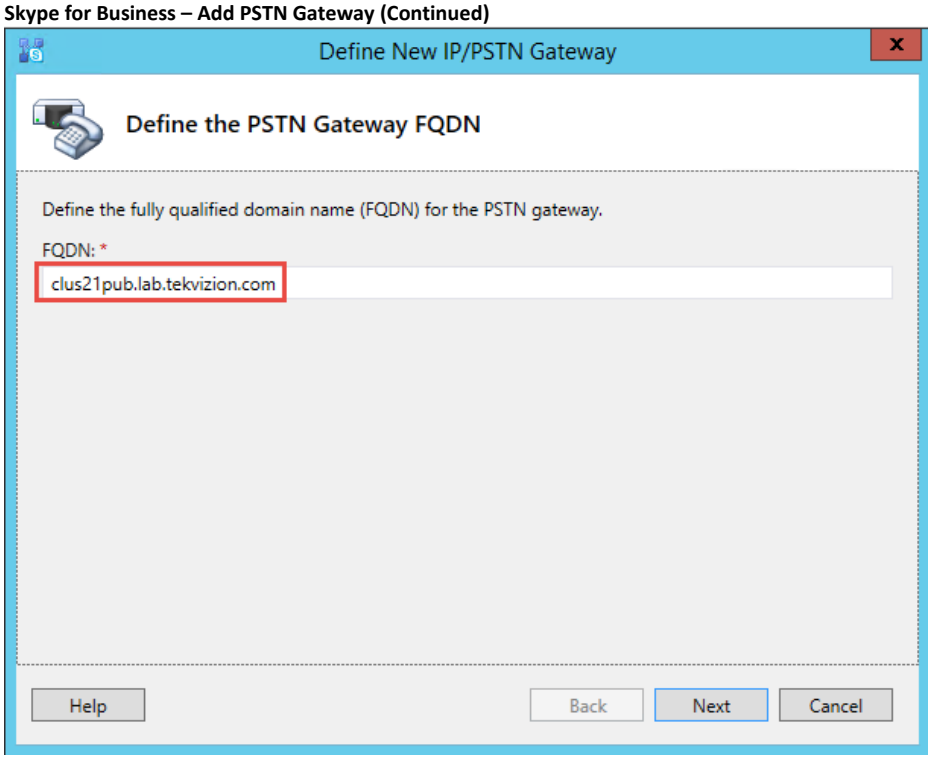
Navigation: Skype for Business Server→CleanDefaultTopology→Shared Components→PSTN gateways

Right click and select "New IP/PSTN Gateway"

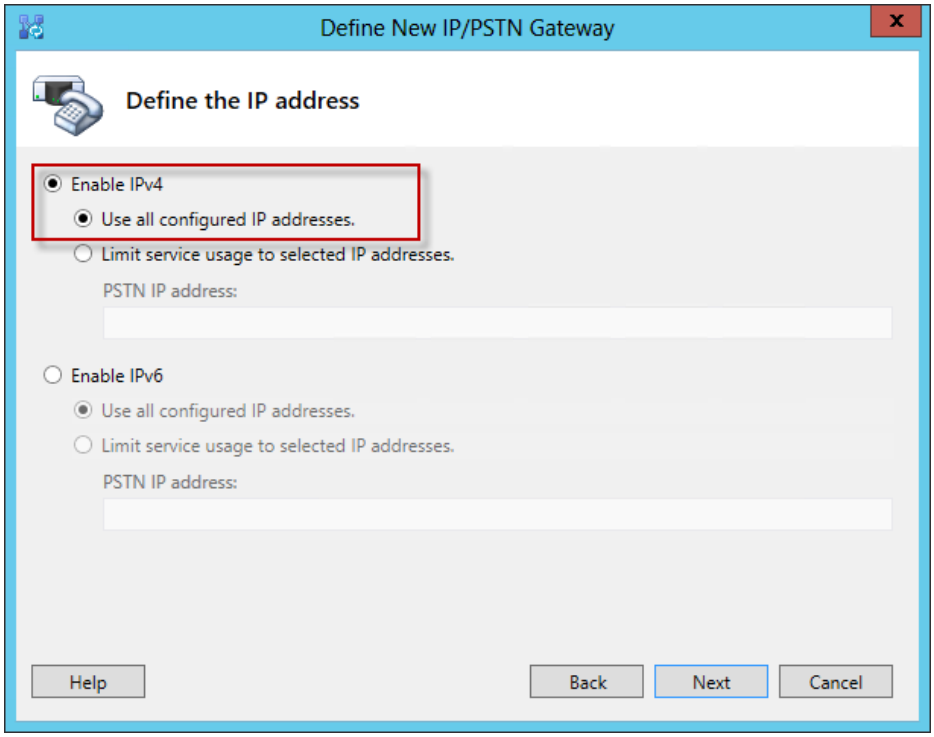


Set FQDN = <FQDN of the Cisco UCM>– clus21pub.sfbsp.local is used in this test.
Click Next.

Skype for Business – Add PSTN Gateway (Continued)

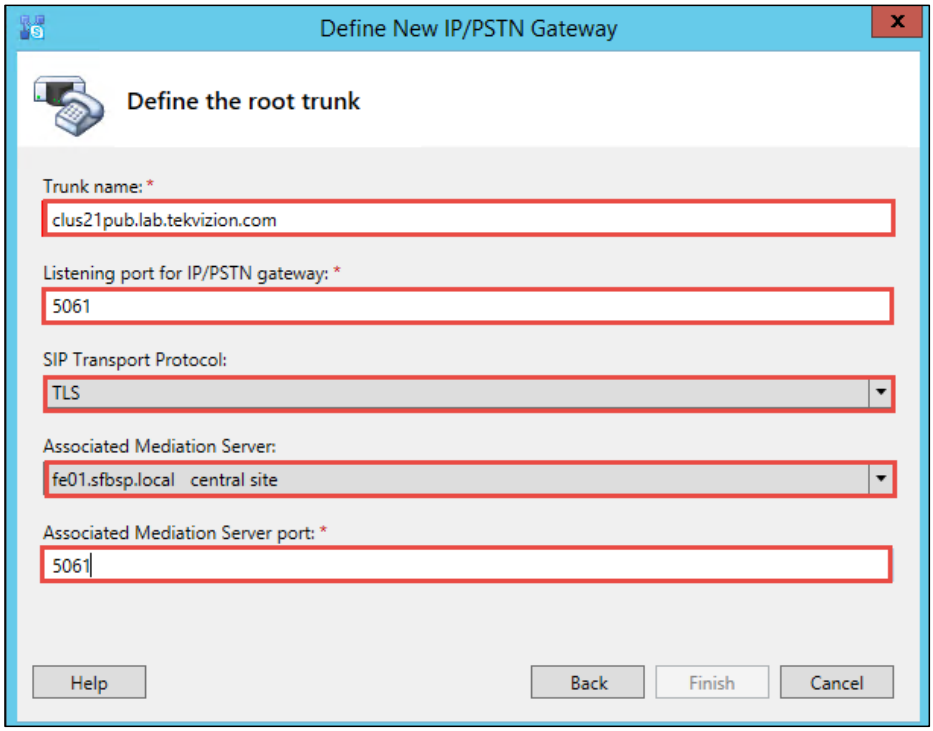


Check the Enable IPv4 and Use all configured IP addresses radio button
Click Next.



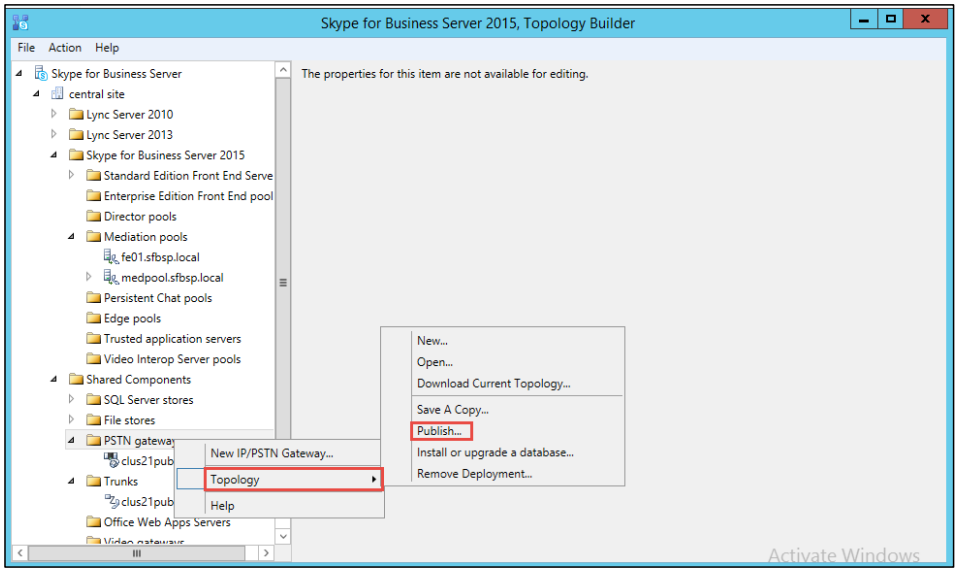
Skype for Business – Add PSTN Gateway (Continued)

Set **Trunk Name** = FQDN of the Cisco UCM – **clus21pub.sfbsp.local** is used for this test
Set **Listening port for IP/PSTN gateway =** The **Listening port** should match the **Incoming Port** setting in the CISCO UCM's **SIP Trunk Security Profile – 5061** is used for this test
Set **SIP Transport Protocol** = TLS
Set **Associate Mediation Server**: Assign this PSTN gateway to the Front End co-located mediation server – **fe01.sfbsp.local** is used for this test.
Click Finish.



Publish the topology so these new configurations take effect.

Skype for Business – Add PSTN Gateway (Continued)

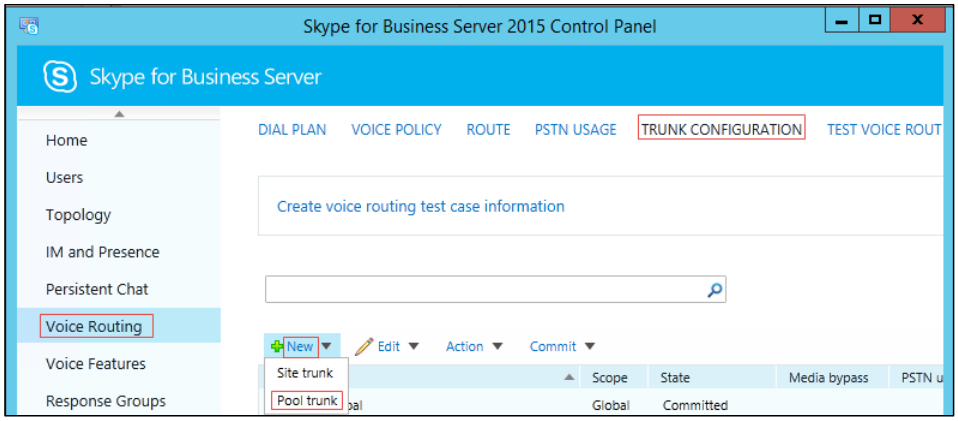


Trunk Configuration

Open the Skype for Business 2015 Control Panel.

Navigation: Voice Routing -> Trunk Configuration

Select New →Pool Trunk



Set **Service** = Trunk to Cisco UCM that was created earlier as a PSTN gateway in the topology builder – clus21pub.lab.tekvizion.com is used for the test.

Set **Maximum early dialogs supported** = **20**

Set **Encryption support level = Optional**

Set **Refer Support = None**

Uncheck **Enable media bypass**

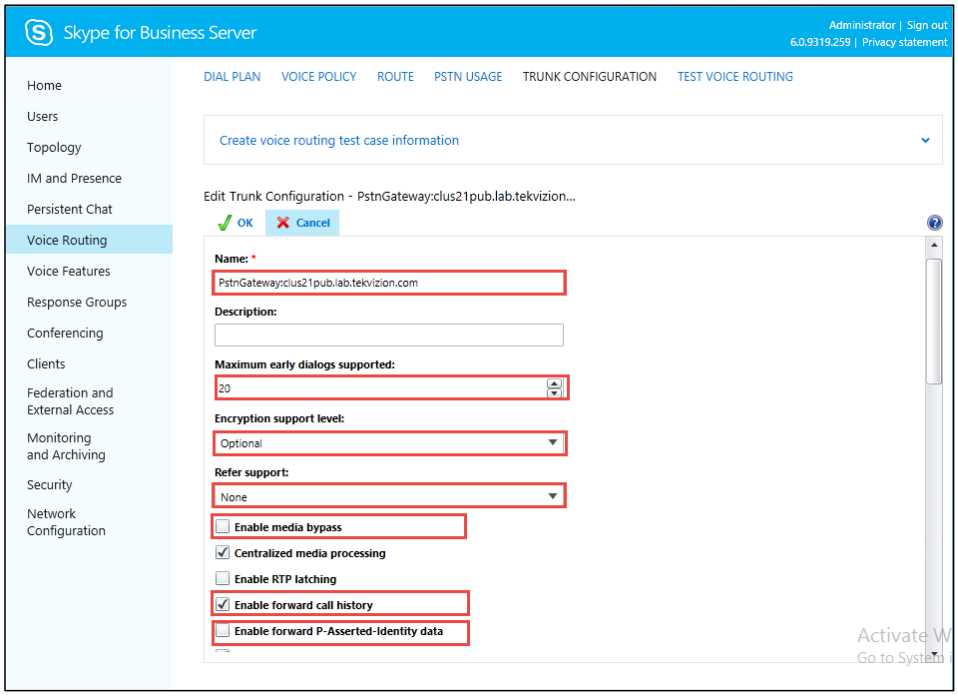
Check **Centralized media processing**

Uncheck **Enable RTP latching**

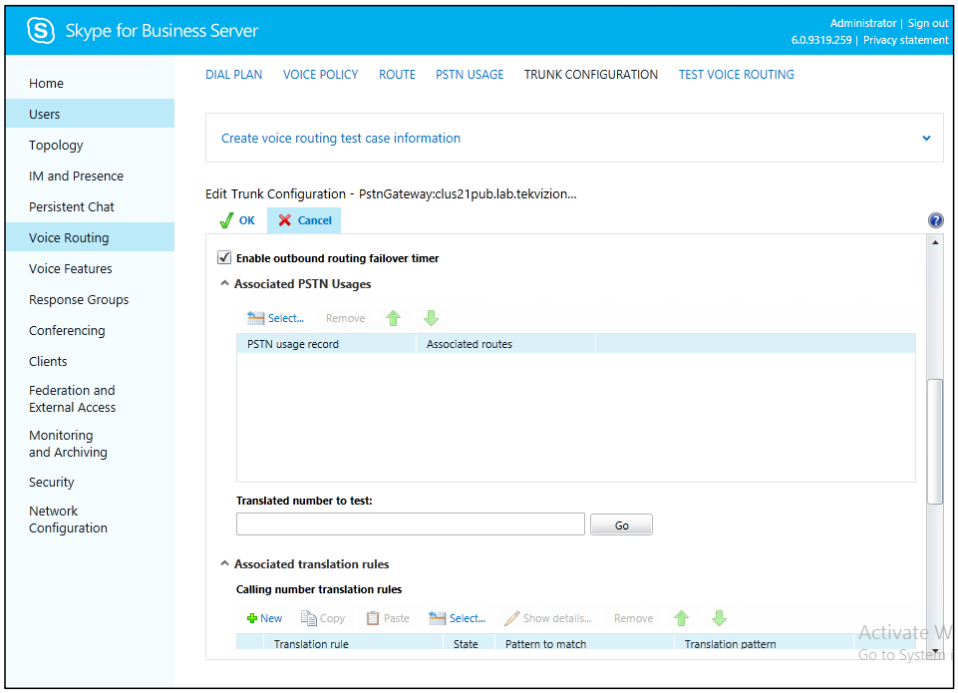
Check **Enable forward call history**

Uncheck **Enable forward P-Asserted-Identity data*** [Note: this is checked when test scenarios that involve restrict ID need to be executed]

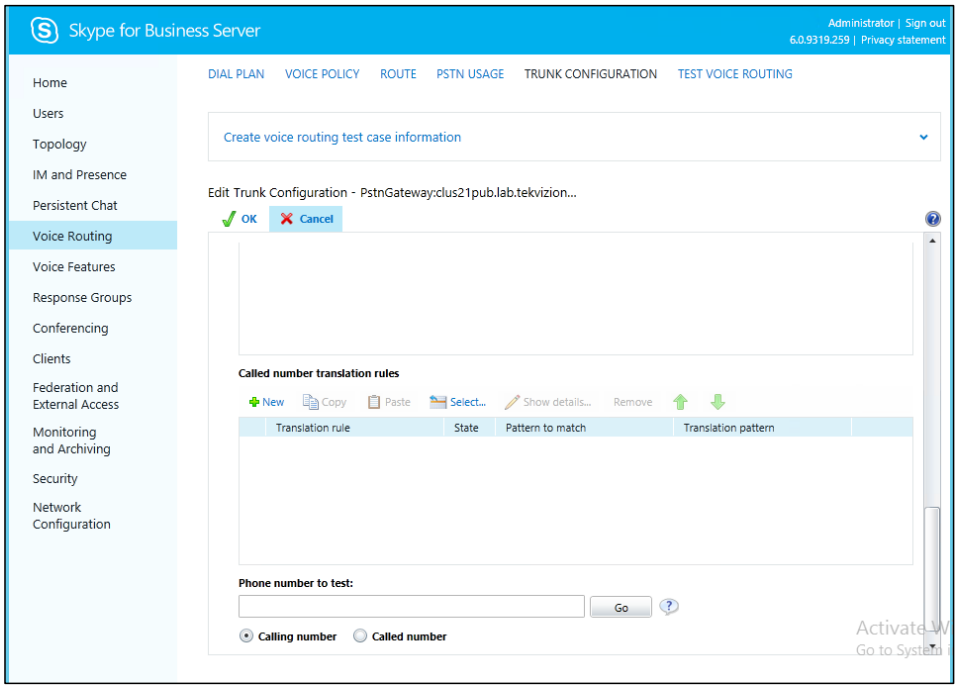
Uncheck **Enable outbound routing failover timer**



Skype for Business –Trunk Configuration (Continued)



Skype for Business –Trunk Configuration (Continued)



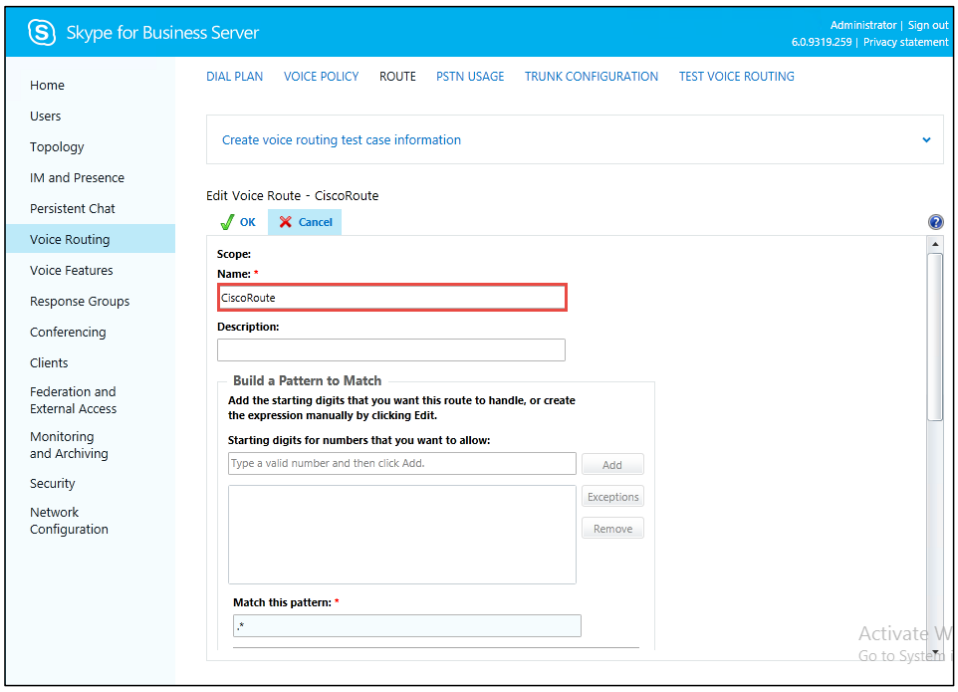
Route Configuration

Navigation: Voice Routing -> Route

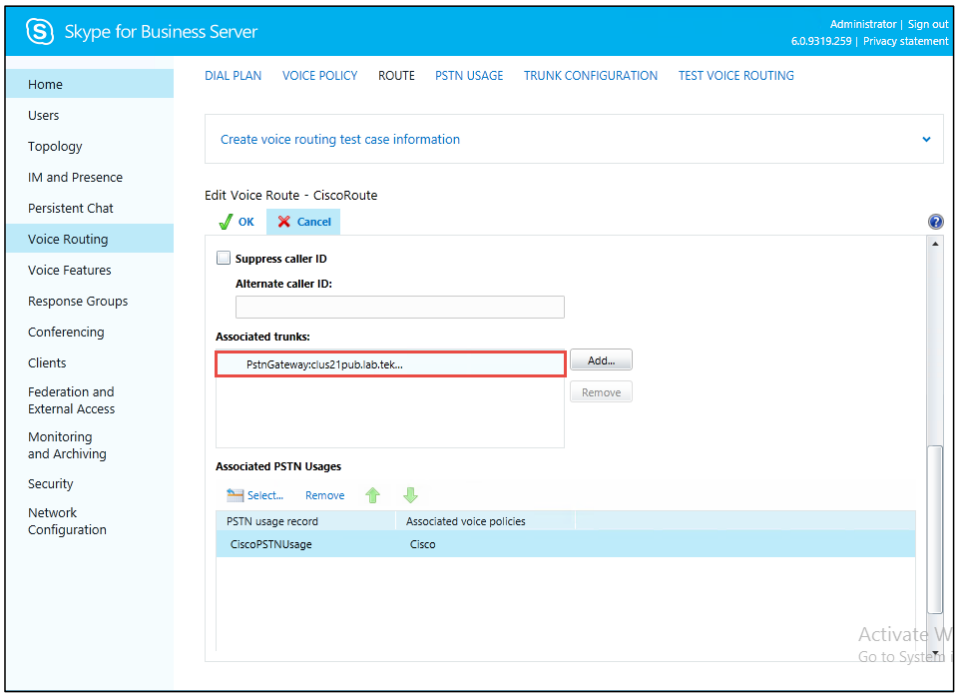
Click New

Set Name = enter a name to identify this Route. CiscoRoute is used for this test.

Add associated trunks = select the trunk configured earlier – PstnGateway: clus21pub.lab.tekvizion.com



Skype for Business –Route Configuration (Continued)



Voice Policy and PSTN Usage Configuration

Navigation: Voice Routing -> Voice Policy

Click New

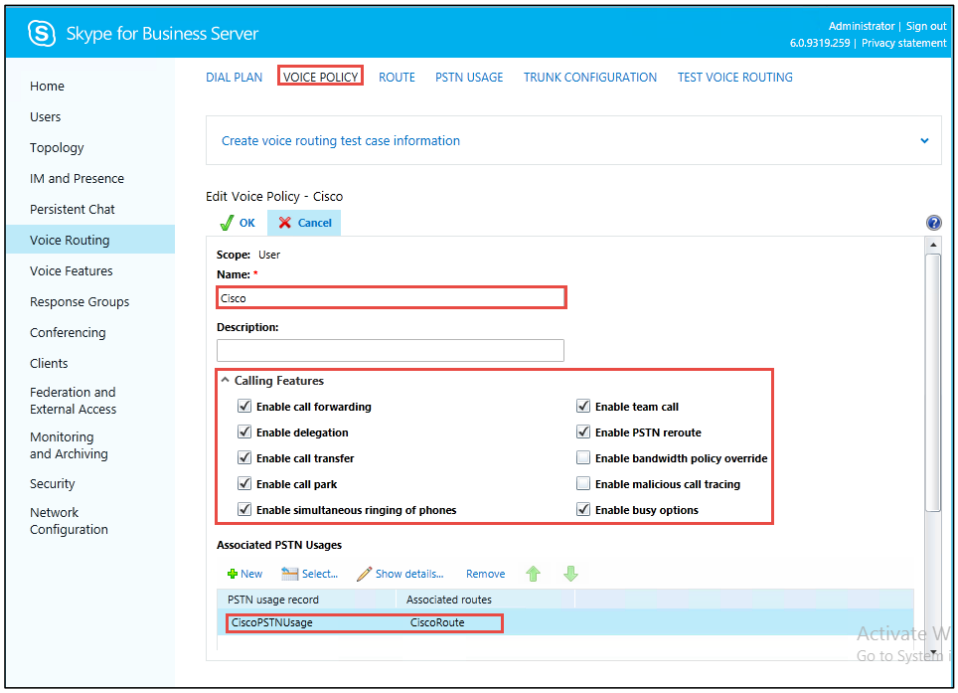
Set Name = enter a name to identify this voice policy – Cisco is used in this test.

Set Calling Features:

- Check Enable call forwarding
- Check Enable delegation
- Check Enable call transfer
- Check Enable call park
- Check Enable simultaneous ringing of phones
- Check Enable team call
- Check Enable PSTN reroute
- Uncheck Enable bandwidth policy override
- Uncheck Enable malicious call tracing
- Enable Busy options

Set Associated PSTN usages:

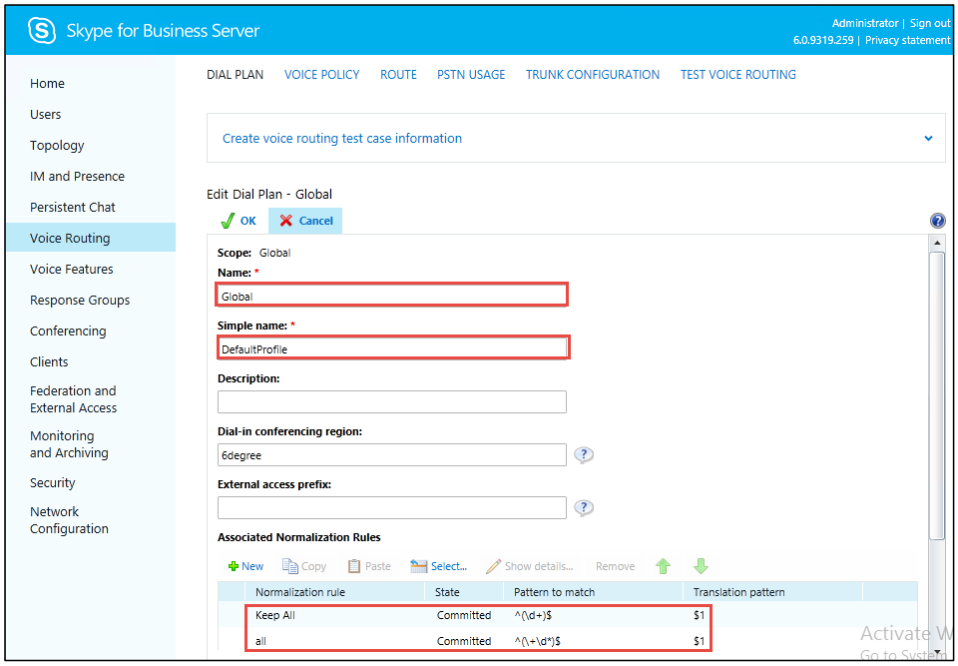
- Click New
- Set Name: enter a name to identify the PSTN Usage record – CiscoPSTNUsage is used in the test.
- Set Associated Routes = select the route created earlier= CiscoRoute


Skype for Business Server
Administrator | Sign out
6.0.9319.259 | Privacy statement
Home
Users
Topology
IM and Presence
Persistent Chat
Voice Routing
Voice Features
Response Groups
Conferencing
Clients
Federation and External Access
Monitoring and Archiving
Security
Network Configuration
DIAL PLAN
VOICE POLICY
ROUTE
PSTN USAGE
TRUNK CONFIGURATION
TEST VOICE ROUTING
Create voice routing test case information
Edit Voice Policy - Cisco
OK
Cancel
Scope: User
Name: *
Cisco
Description:
Calling Features
Enable call forwarding
Enable delegation
Enable call transfer
Enable call park
Enable simultaneous ringing of phones
Enable team call
Enable PSTN reroute
Enable bandwidth policy override
Enable malicious call tracing
Enable busy options
Associated PSTN Usages
New
Select...
Show details...
Remove
PSTN usage record
Associated routes
CiscoPSTNUsage
CiscoRoute

Dial Plan Configuration

Navigation: Voice Routing-> Dial Plan

Default Dial plan used for this topology



Call Park Range Configuration

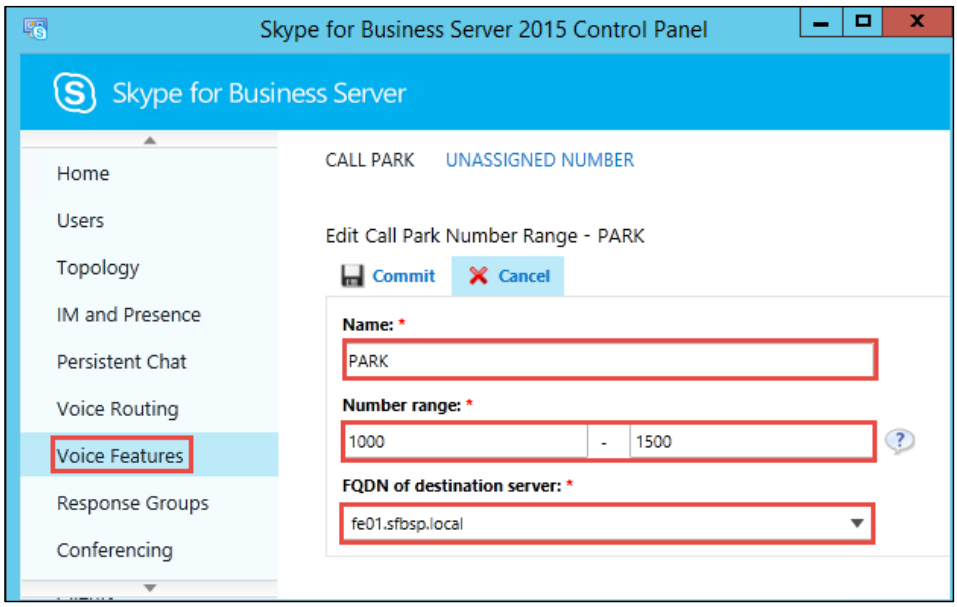
Navigation: Voice Features -> Call Park

Click New.

Set Name = enter text to identify the call Park Range – Orbit range is used in the test.

Set Number Range = 1000 to 1500 is used in the test.

Set FQDN of destination server= select the desired server – fe01.sfbsp.local is used in the test



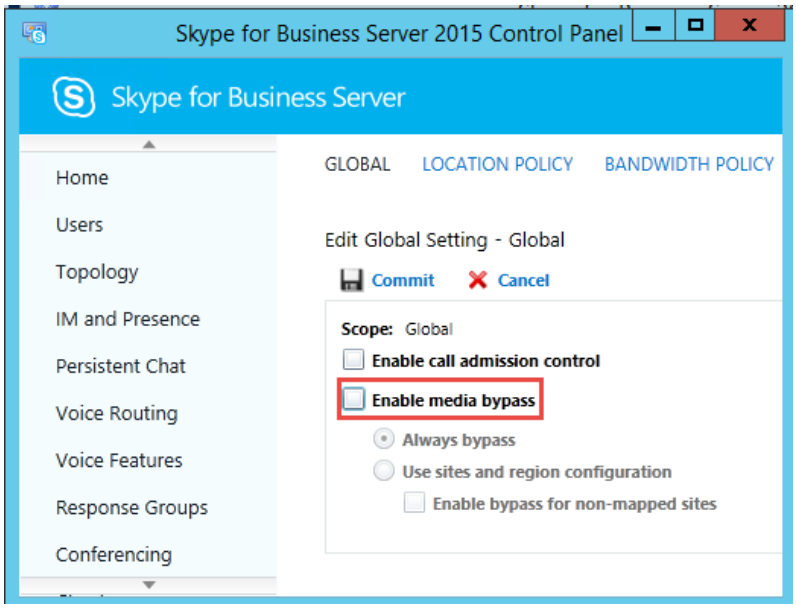
Global Media Bypass Configuration

Navigation: Network Configuration -> Global

Edit Global Setting –

- Uncheck Enable media bypass

Commit the configuration.

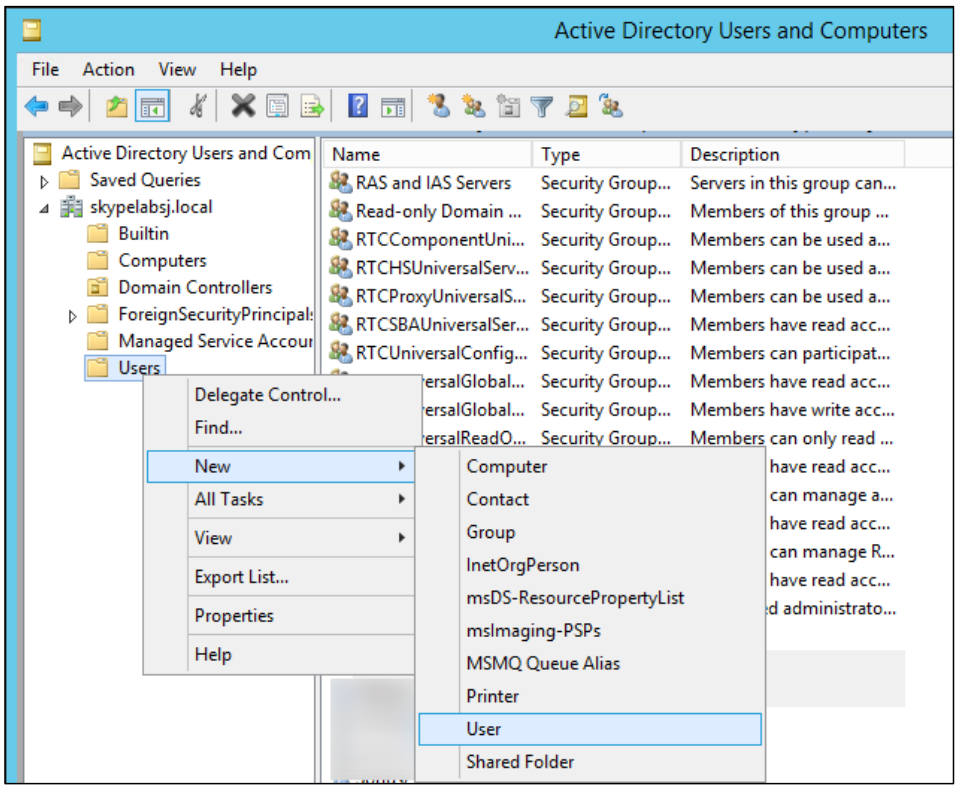


User Configuration

Login to the Skype for Business Active Directory

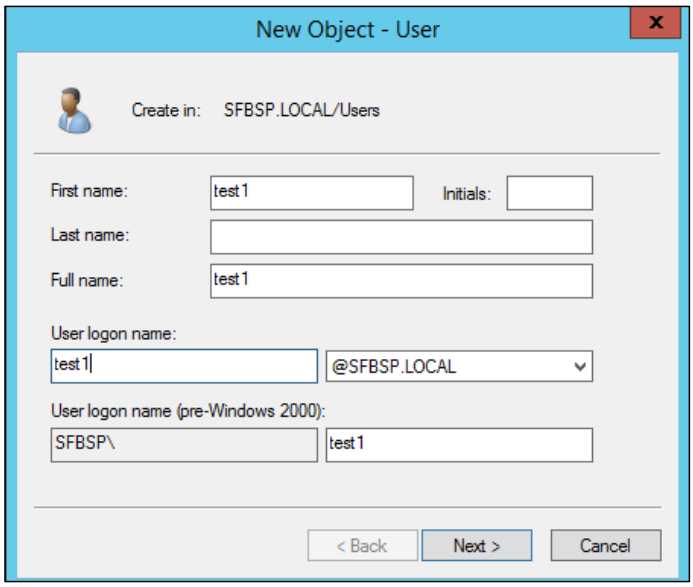
Navigation: Active Directory Users and Computers →Users

Add a New User

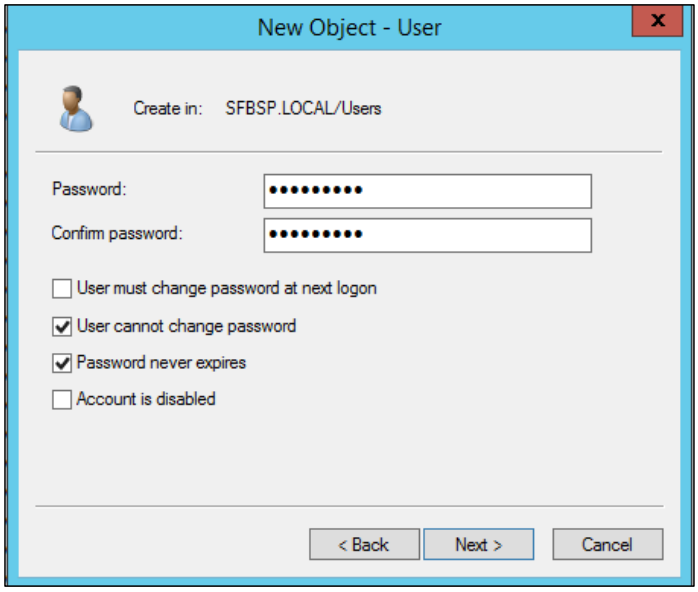


Skype for Business – New User configuration (continued)

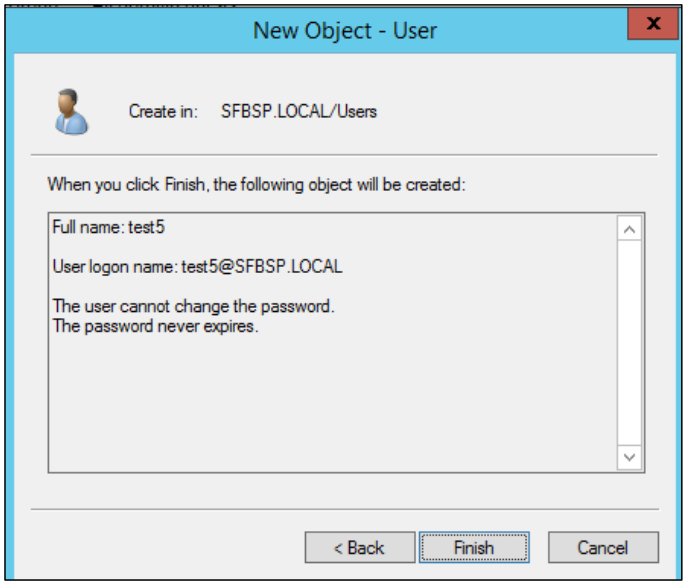
Follow the screenshots below to add a new user:



Skype for Business – New User configuration (continued)



Skype for Business – New User configuration (continued)

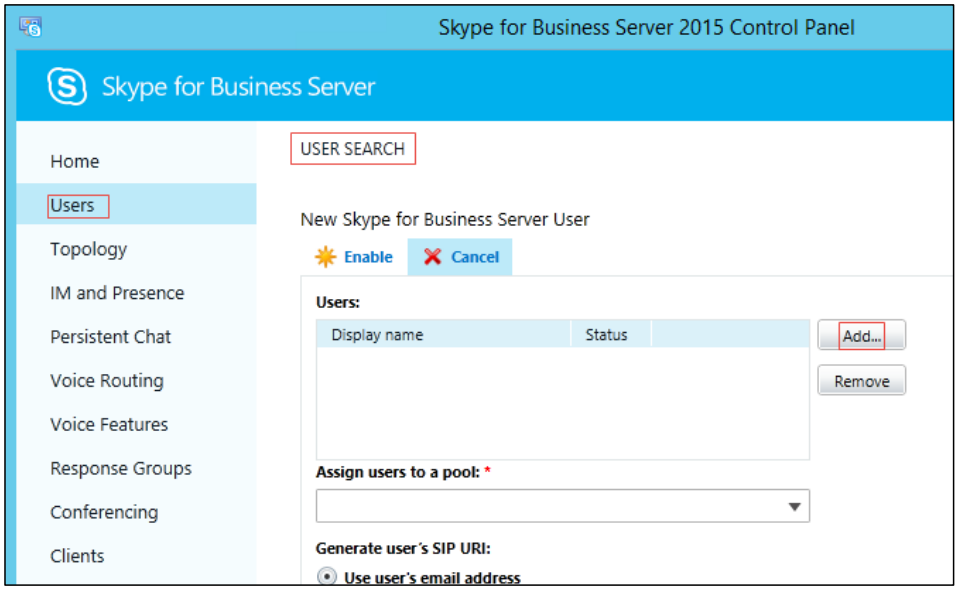


Once the user is created, login to the Skype for Business 2015 Control Panel

Navigation: Users→ Enable users

Click on the Add button and find the new user created earlier.

Skype for Business – New User configuration (continued)



Set Assign users to a **pool**= **fe01.sfbsp.local** from drop down menu

Set Generate user's **SIP URI**: Specify a SIP URI: sip:test1@sfbsp.local .This is used in this test

Set **Telephony**=**Enterprise Voice**

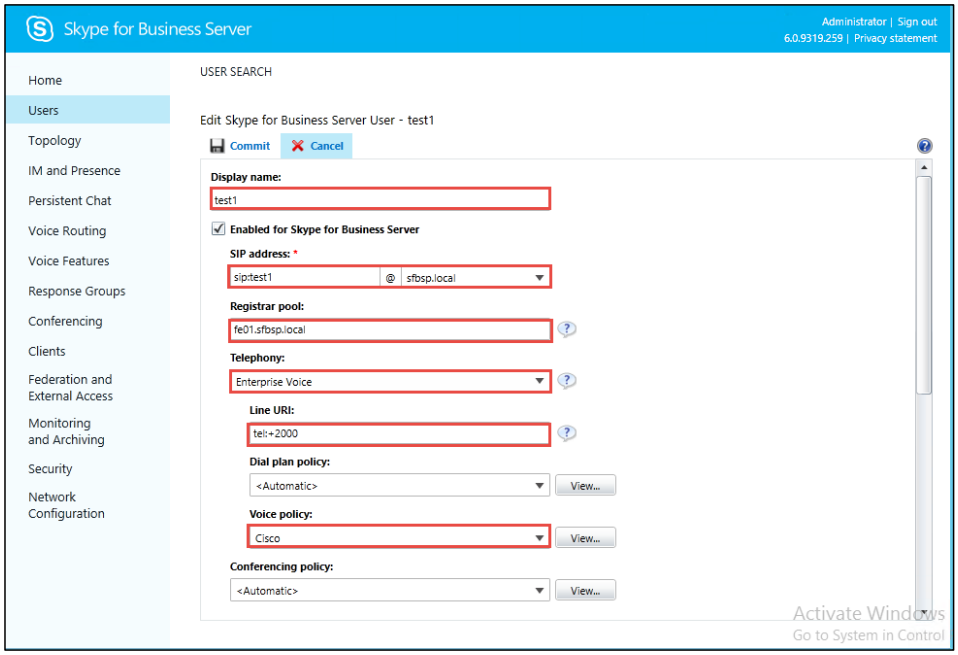
Set Line **URI**: = **tel: +2000** is used for the test. This is the DN for the user.

Set Dial **plan policy** = **Automatic** (as configured earlier)

Set **Voice policy**= **Cisco** (as configured earlier)

Click Enable.

Skype for Business – New User configuration (continued)

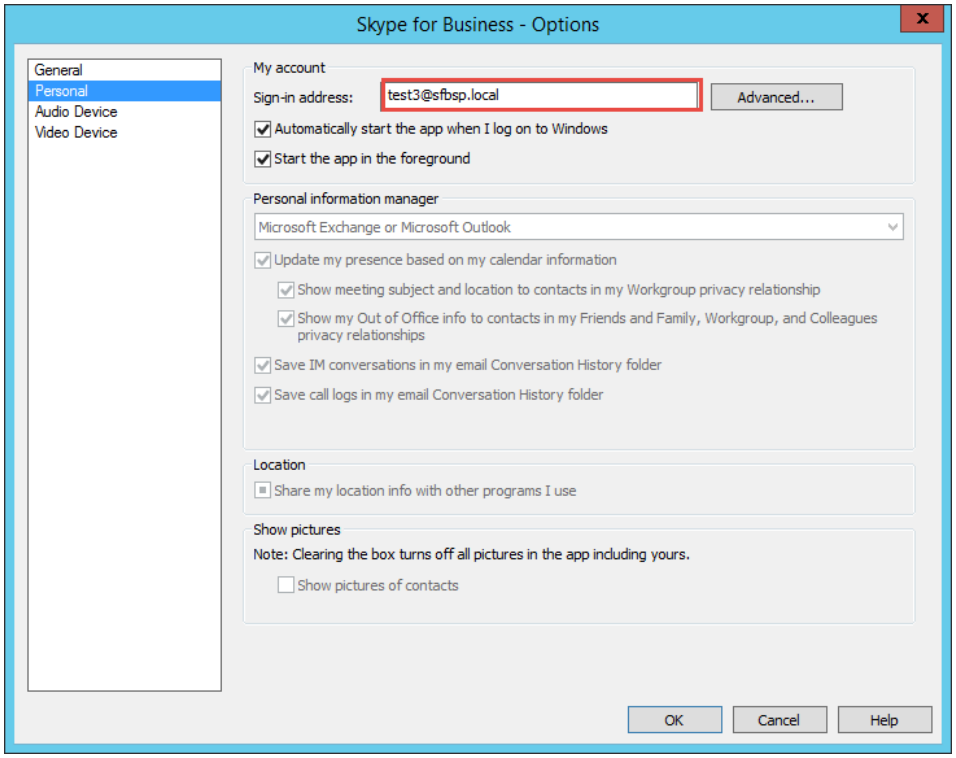


Client Configuration

Download the latest version of the Skype for Business client and launch the same.

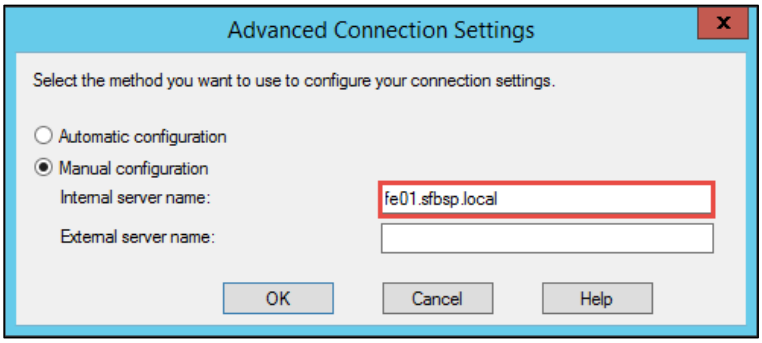
Navigation: Settings→Tools→Options→Personal→MyAccount

Set Sign-in-address= enter the sip uri of the user configured in username@domain format. test1@sfbsp.local is used for example.



Click Advanced. Select Manual Configuration.

Set Internal Server Name= Enter the FQDN of the domain (sfbsp.local.local is used for example)



Configuring the Cisco Unified Communications Manager

Cisco Unified Communications Manager Software Version

SIP Trunk Security Profile for Trunk to Skype for Business

Navigation: System→ Security → SIP trunk security profile

Set **Name*= SFB_SECURITY_PROFILE**. This is used for the test.

Set Device Security mode = Encrypted

Set **Incoming Transport Type = TLS**

Set **Outgoing Transport Type = TLS**

Set **X.509 Subject Name = fe01.sfbsp.local**

Set **Incoming Port = 5061**

Check **Accept Presence Subscription**

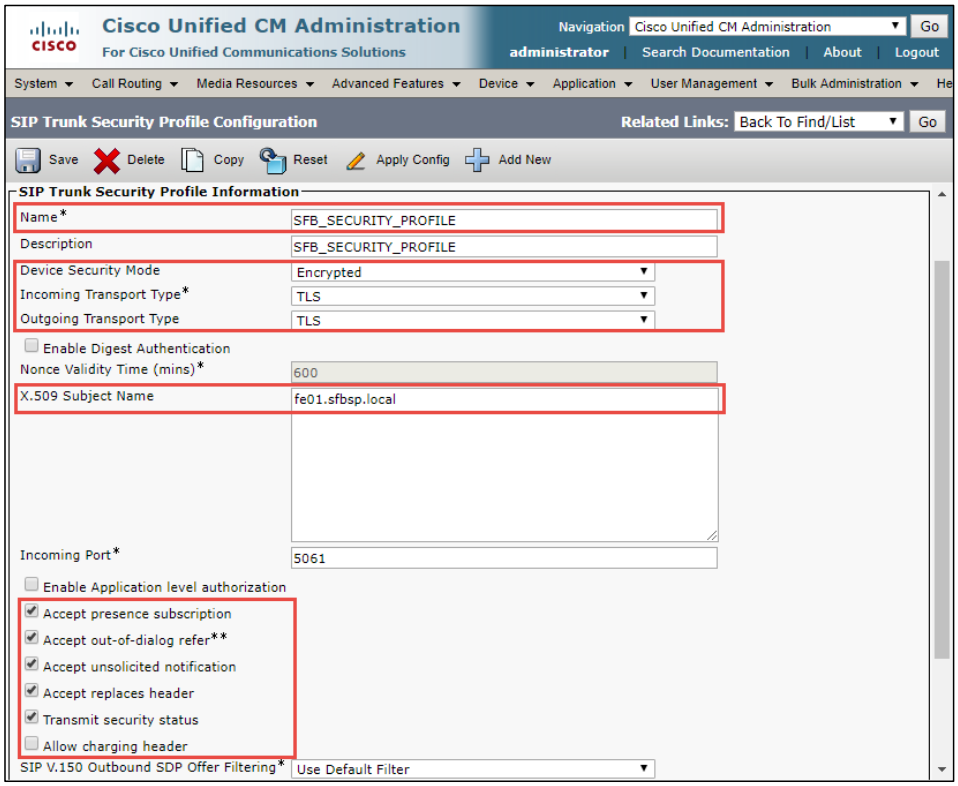
Check **Accept out of dialog refer**

Check **Accept unsolicited notification**

Check **Accept Replaces header**

Check **Transmit security status**

All other values are default.



SIP Trunk Security Profile for Trunk to Unity Connection

Navigation: System→ Security → SIP trunk security profile

Set **Name***= **TLS_CUC**. This is used for the test.

Set Device **Security mode** = **Encrypted**

Set **Incoming Transport Type = TLS**

Set **Outgoing Transport Type = TLS**

Set X**.509 Subject Name = clus21unity.lab.tekvizion.com**

Set **Incoming Port = 5061**

Uncheck **Accept Presence Subscription**

Check **Accept out-of-dialog refer****

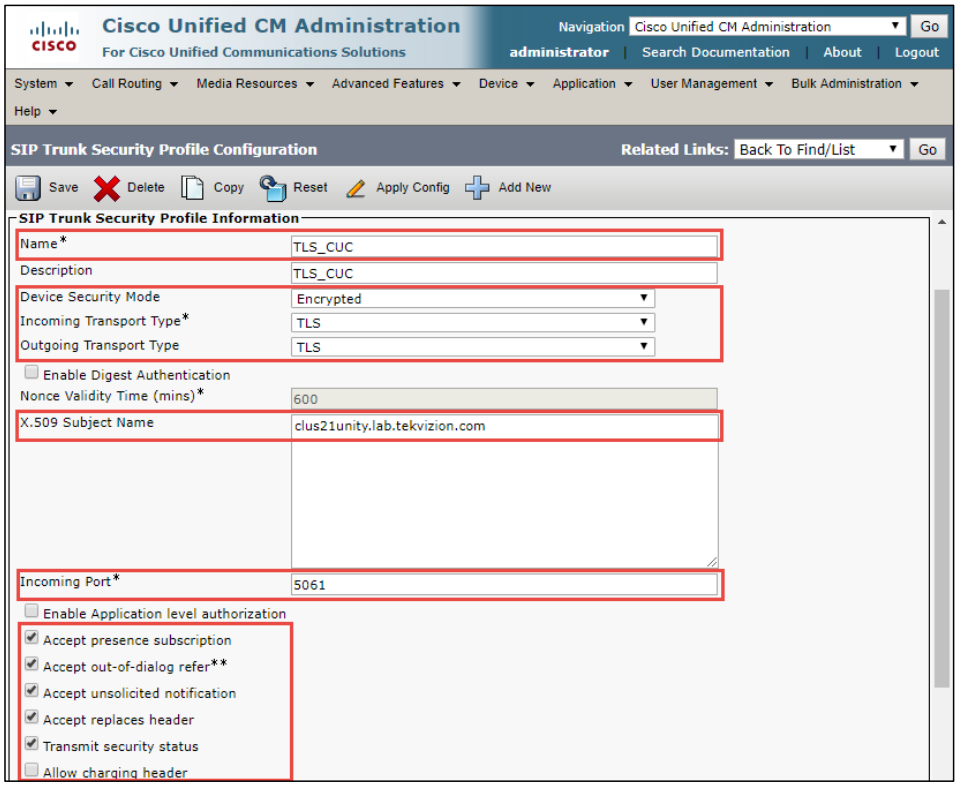
Check **Accept unsolicited notification**

Check **Accept Replaces header**

Check **Transmit security status**

All other values are default.

Commented [RSM1]: This has been checked in the screenshot



SIP Profile

Navigation: Device → Device Settings → SIP Profile

Set **Name*= SFB_SIP_PROFILE** - Standard SIP Profile. This is used for this test.

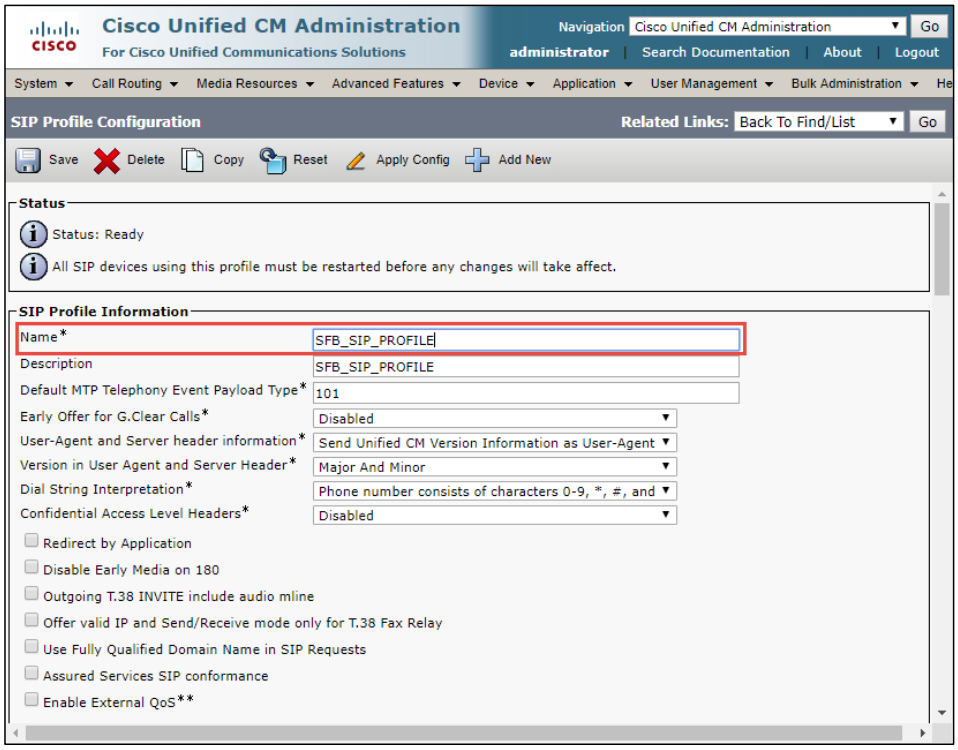
Set **SIP Rel1XX Options = Send PRACK if 1xx Contains SDP**

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

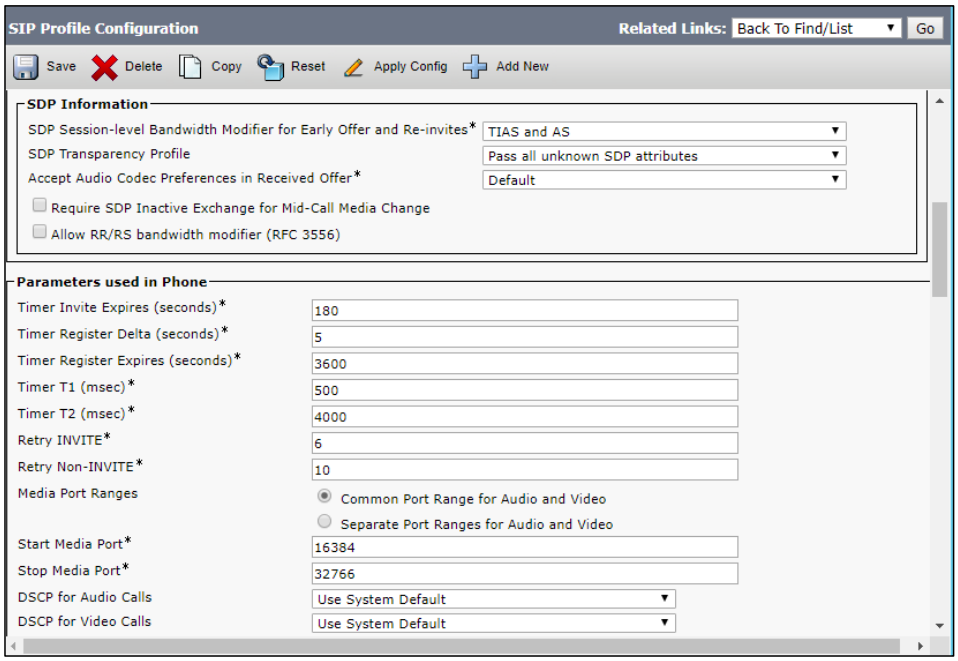
Check Enable OPTIONS Ping to monitor Destination status for Trunks with Service Type "None (Default)"

All other values are default.

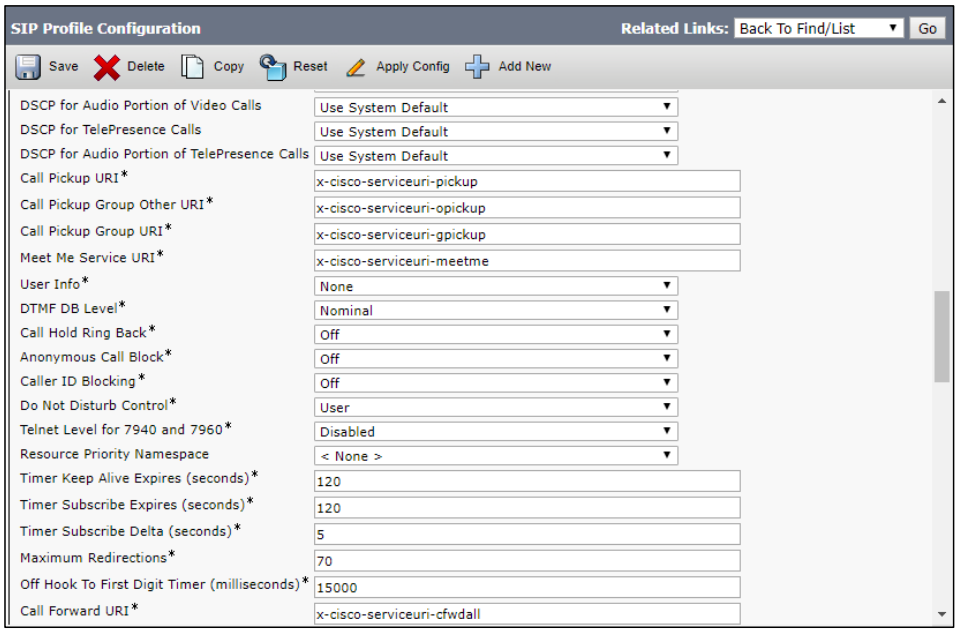
Commented [RSM2]: 1XX messages is used in the screenshot


Cisco Unified CM Administration
For Cisco Unified Communications Solutions
Navigation Cisco Unified CM Administration Go
administrator | Search Documentation | About | Logout
System Call Routing Media Resources Advanced Features Device Application User Management Bulk Administration
SIP Profile Configuration
Related Links: Back To Find/List Go
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* SFB_SIP_PROFILE
Description SFB_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-Agent
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
Offer valid IP and Send/Receive mode only for T.38 Fax Relay
Use Fully Qualified Domain Name in SIP Requests
Assured Services SIP conformance
Enable External QoS**

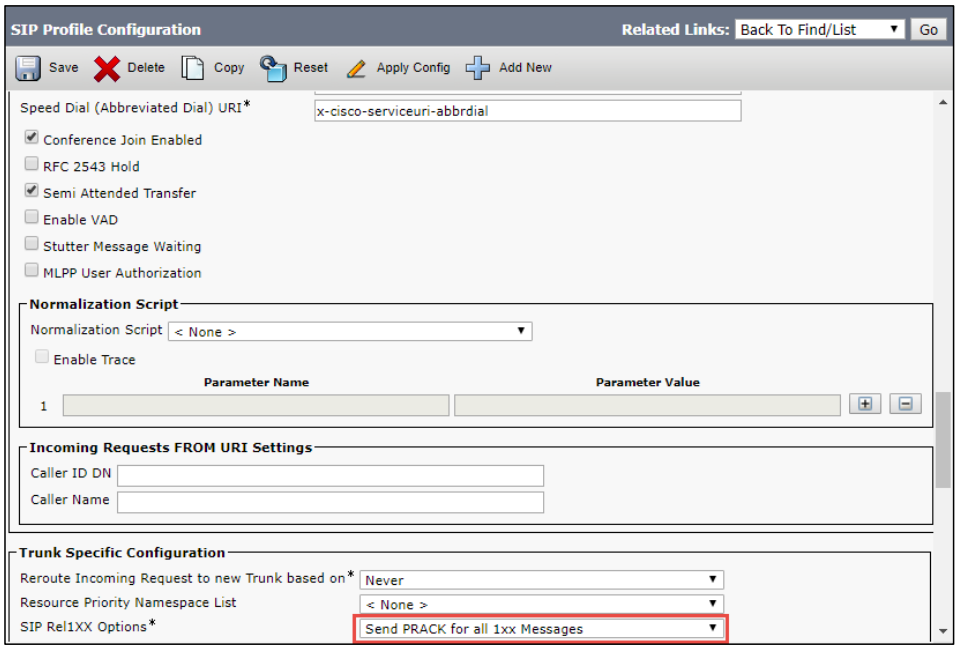
Cisco Unified Communications Manager SIP Profile (Continued)



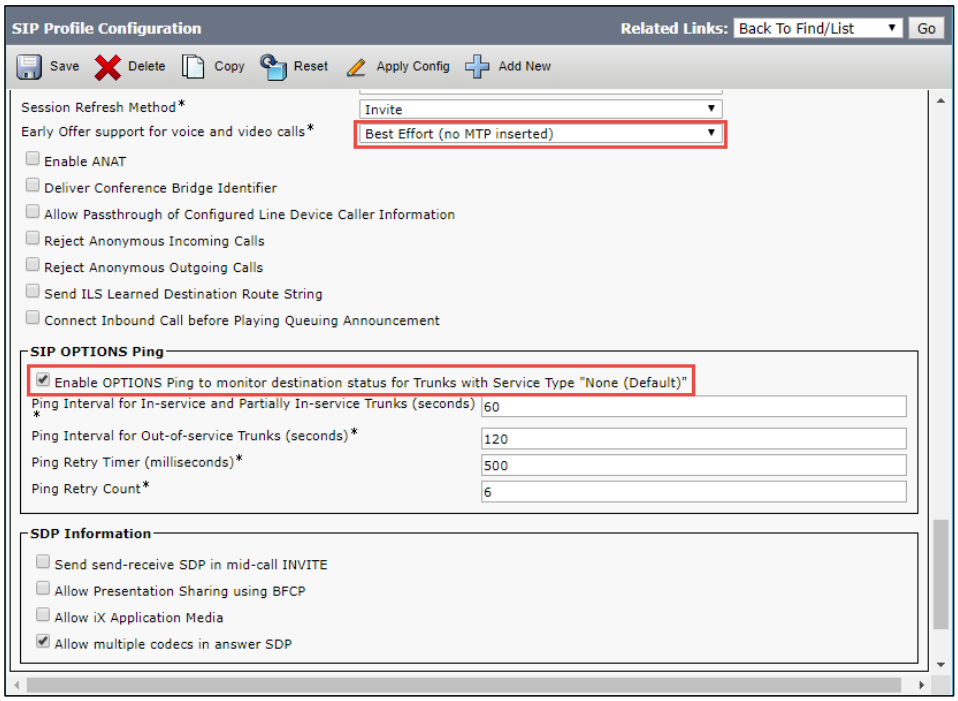
Cisco Unified Communications Manager SIP Profile (Continued)



Cisco Unified Communications Manager SIP Profile (Continued)



Cisco Unified Communications Manager SIP Profile (Continued)



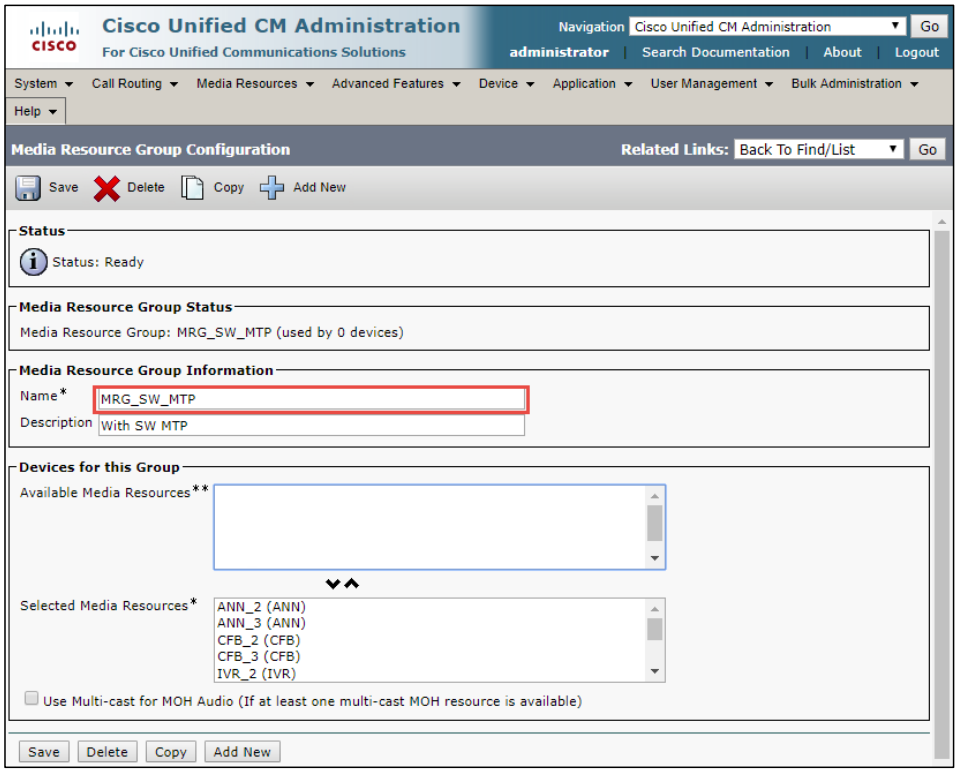
Media Resource Group

Navigation Path: Media Resources→ Media Resource Group; Add New
Media Resource Group MRG

Set **Name*= MRG_SW_MTP**, This is used for this test.

Set all resources in the Selected Media Resources* Box.

All other values are default.



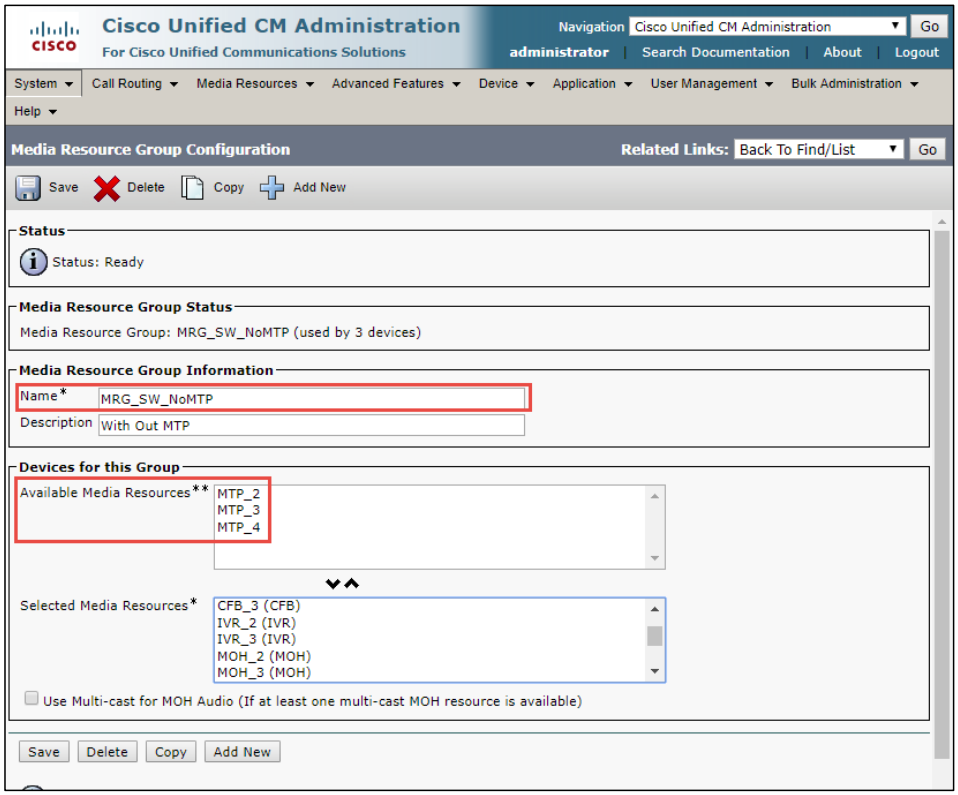
Resource Group for MRG_NoMTP

Set **Name*= MRG_NoMTP**. This is used for the test.

Set **Available Media Resources = MTP_2, MTP_3 and MTP_4**

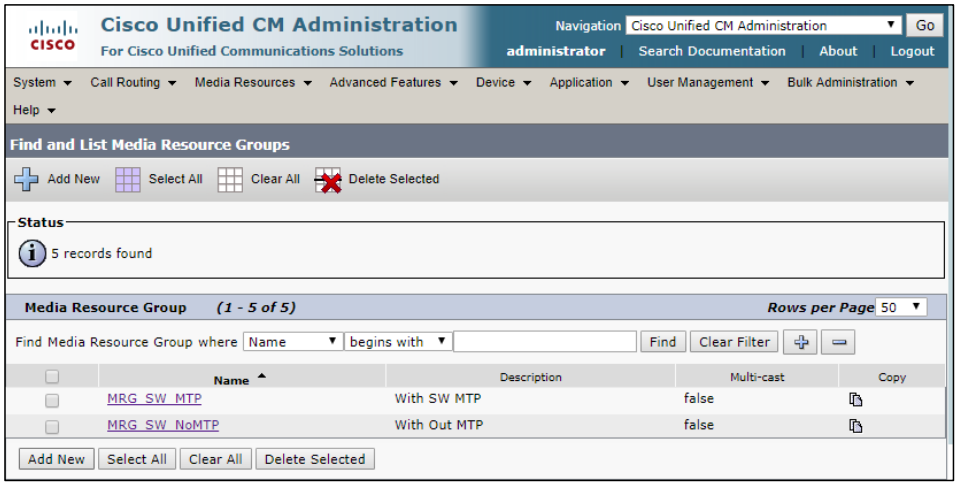
Set other resources in the Selected Media Resources*

All other values are default.


Cisco Unified CM Administration
For Cisco Unified Communications Solutions
Navigation Cisco Unified CM Administration Go
administrator | Search Documentation | About | Logout
System Call Routing Media Resources Advanced Features Device Application User Management Bulk Administration Help
Media Resource Group Configuration
Related Links: Back To Find/List Go
Save Delete Copy Add New
Status
Status: Ready
Media Resource Group Status
Media Resource Group: MRG_SW_NoMTP (used by 3 devices)
Media Resource Group Information
Name* MRG_SW_NoMTP
Description With Out MTP
Devices for this Group
Available Media Resources** MTP_2 MTP_3 MTP_4
Selected Media Resources* CFB_3 (CFB) IVR_2 (IVR) IVR_3 (IVR) MOH_2 (MOH) MOH_3 (MOH)
Use Multi-cast for MOH Audio (If at least one multi-cast MOH resource is available)
Save Delete Copy Add New

Cisco Unified Communications Manager Media Resource Group Configuration

Commented [RSM3]: The following screenshot is repetitive may be its correct



Media Resource Group List

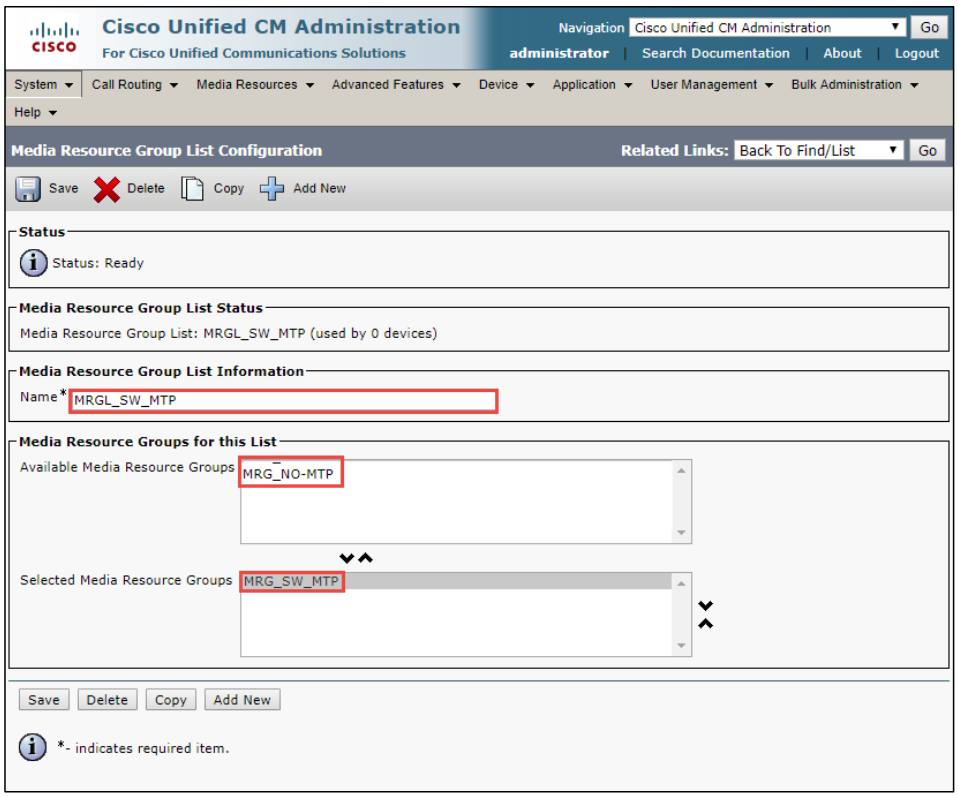
Navigation Path: Media Resources→ Media Resource Group List

Add New

Set **Name*= MRGL_SW_MTP**. This is used for this test.

Set **Available Media Resources = MRG_NO-MTP**

Set **Selected Media Resource Groups= MRGL_SW_MTP**

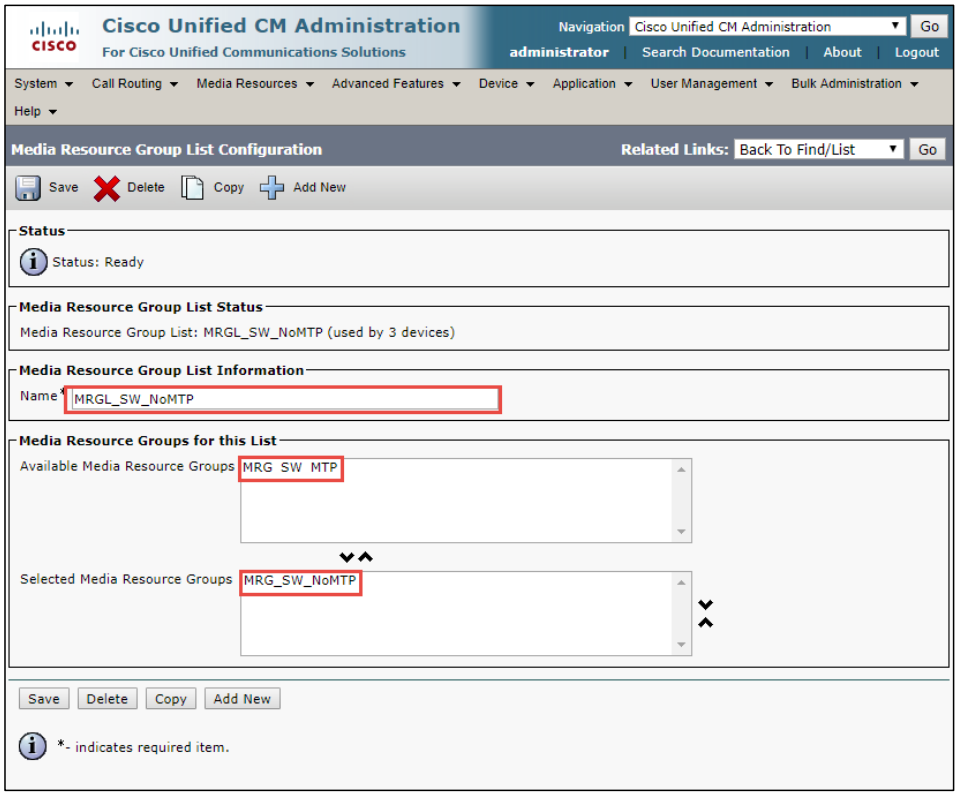


Add new

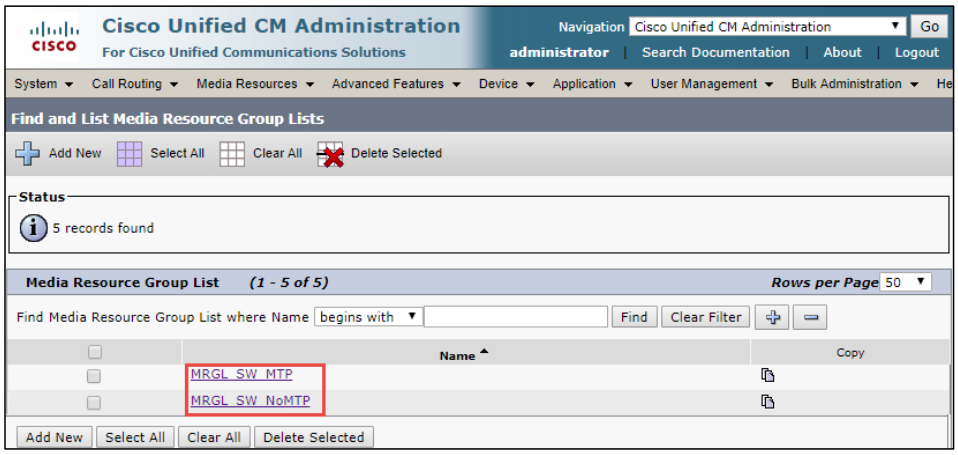
Set **Name*= MRGL_SW_NoMTP**. This is used for the test

Set **Available Media Resources MRG_SW_MTP**

Set **Selected Media Resource Groups= MRGL_SW_NoMTP**



Cisco Unified Communications Manager Media Resource Group List Configuration



Device Pool Configuration

Device Pool - **G711_Pool** is created in this test.

Navigation Path: System → Device Pool

Add New.

Set **Device Pool Name*= G711_pool**. This is used in the test.

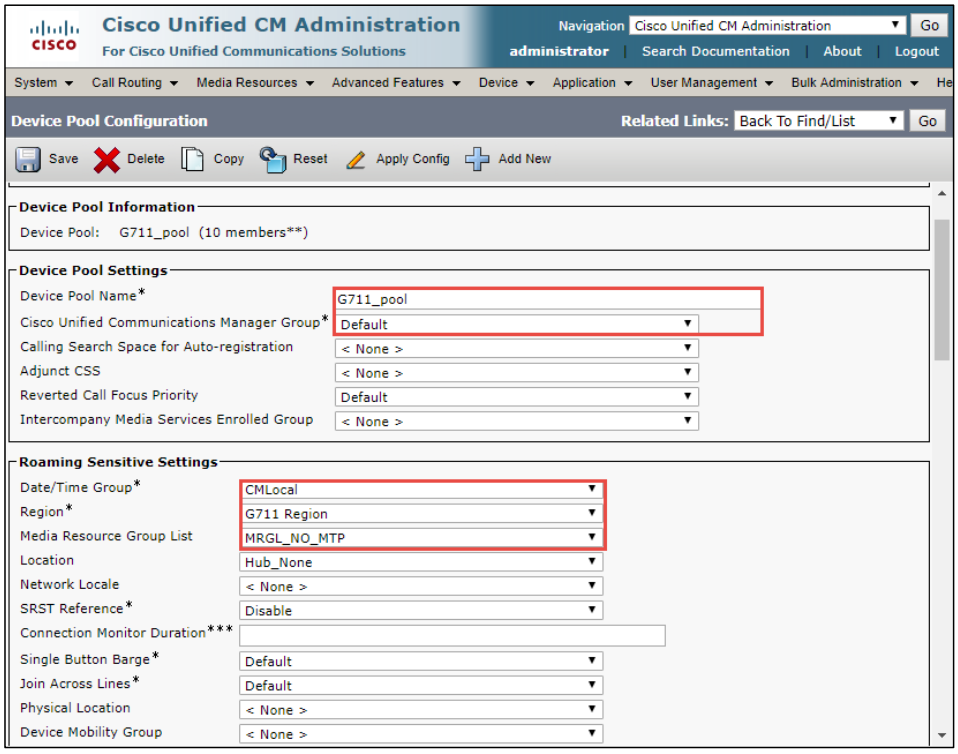
Set **Cisco Unified Communications Manager Group*= Default**

Set **Date/Time Group* = CMLocal**

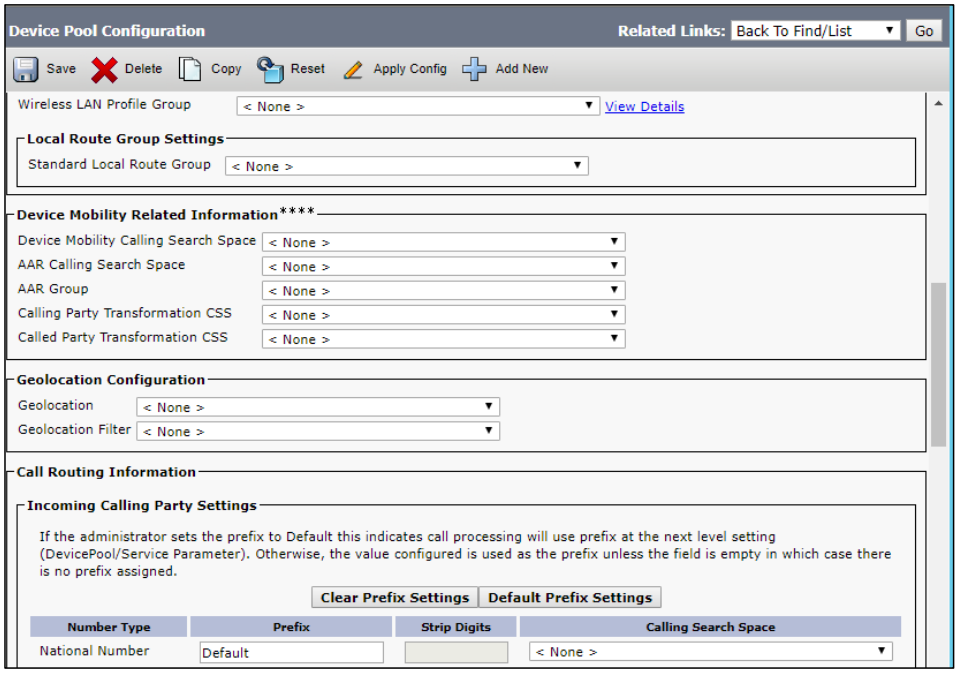
Set **Region* = G711 Region**. This is used in this example

Set **Media Resource Group List =MRGL_NO_MTP**. This is used in this example.

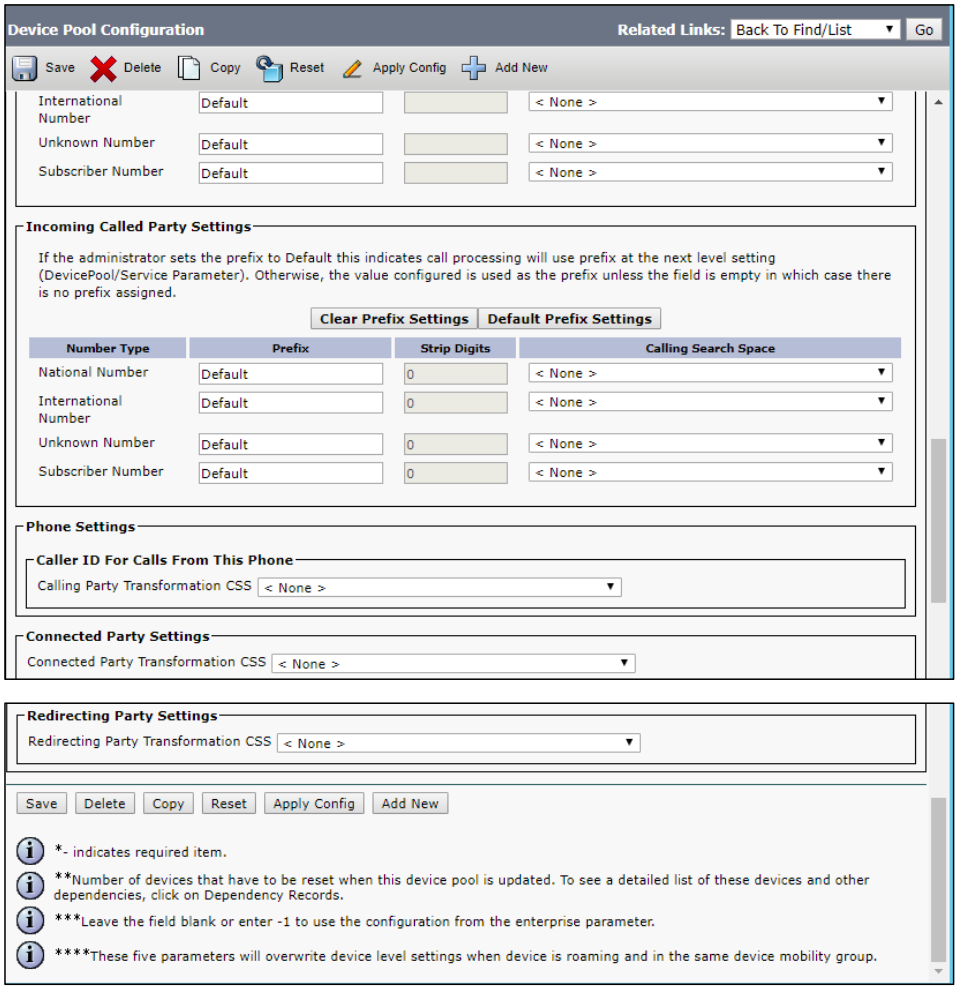
All other values are default.



Cisco Unified Communications Manager Device Pool Configuration (Continued)



Cisco Unified Communications Manager Device Pool Configuration (Continued)



Region Configuration

Navigation Path: System → Region Information → Region

Add New

G711 Region is created in this test.

Set **Name*= G711 Region**. This is used in this example

Set **Region= G711 Region**. This is used in this example

Set **Audio Codec Preference List= G711_Preferred Codec List**

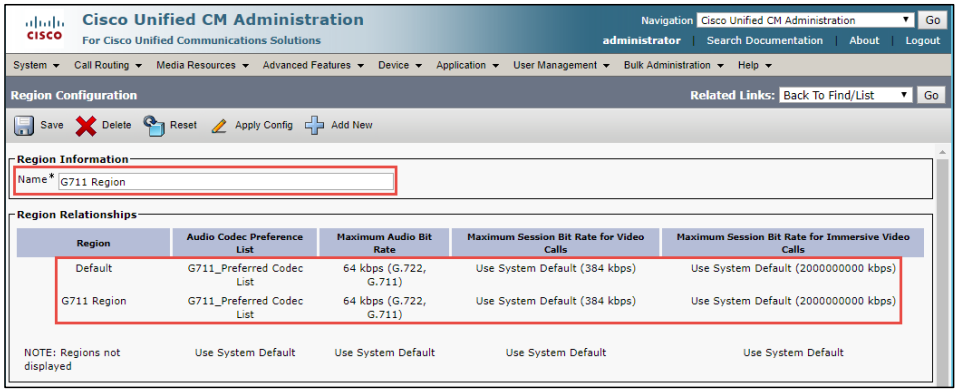
Set **Maximum Audio Bit Rate= 64 Kbps (G7.22, G7.11).** This is used in this example

Set **Region=Default**. This is used in this example

Set **Audio Codec Preference List= G711_Preferred Codec List**. This is used in this example

Set **Maximum Audio Bit Rate= 64 Kbps (G722, G7.11)**. This is used in this example

All other values are default



Normalization Script

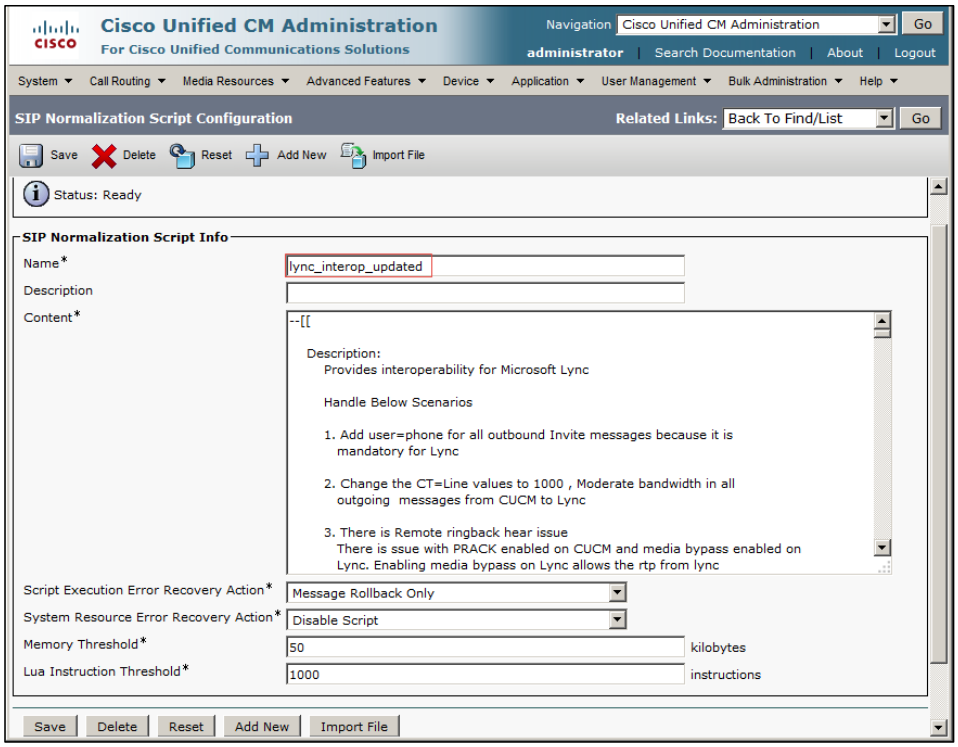
Navigation: Device->Device Settings->SIP Normalization Script

Add New

Set Name = enter text here to identify the normalization script for use on trunk. Lync_interop_updated is used in this test.

Set Content = add script content.

Note: “lync_interop” was the originally provided script by Cisco Support for Cisco UCM-Skype for Business TLS integration. However, with the script activated, the call from Skype for Business to Cisco UCM was not established on SRTP still. Cisco UCM sent AVP (that is, chose RTP instead of SRTP) when Skype for Business sent a=tcap: 1 RTP/SAVP through INVITE SDP. The script was updated to fix this issue and “lync_interop_updated” is the updated script.



Normalization Script

```
--[[
```

```
Description:
   Provides interoperability for Microsoft Lync

   Handle Below Scenarios

   1. Add user=phone for all outbound Invite messages because it is
      mandatory for Lync

   2. Change the CT=Line values to 1000 , Moderate bandwidth in all
      outgoing  messages from CUCM to Lync

   3. There is Remote ringback hear issue
      There is issue with PRACK enabled on CUCM and media bypass enabled on
      Lync. Enabling media bypass on Lync allows the rtp from lync
      endpoint to flow through CUCM directly instead of flowing through
      Mediation server. The problem with PRACK enabled is that Lync end
      point is now not able to answer the incoming call.Looking into the
      traces, it appears that even though Lync sent updated connection
      information in 183 w/sdp, the call manager is still sending rtp to the
      mediation server which seems to be incorrect" So In this scenario
      CUCM expects 180 Ringing not 183 Session progress.
      So added the Script to convert 183 Session Progress to 180 Ringing.

   4. There is incoming Invite from Lync and in From Header there is
      "user=phone" which cause CUCM to send malformed data in to
      different layers which cause call failure.So this is work around for that
      Scenario.

   5. Script modify the AS header which from outgoing messages because call forward
      fails due to bandwith negotiation value is A=64 is not supported

   6. Script convert the History info to diversion Header since call forward to unity
      Is not supported.

   7. Transfer Scenario: Referred-By in Incoming Invite is converted to Diversion Header.


   Script Parameters:

   Release: 9.1(2) , 10.0.(1)

   Copyright (c) 2009-2011 Cisco Systems, Inc. All rights reserved.
   All rights reserved.
--]]

M = {}
```

```
M.allowHeaders = {"History-Info"}

trace.enable()

local function getDisplayName (i_header)
   local position_of_uri=string.find(i_header, "<")
         if position_of_uri <= 2
         then
                  display_name=nil
         else
                  -- save display name which arrives in quotes
                  local display_name_tmp = string.sub(i_header,1, (position_of_uri - 1))
                  -- now remove the quotes
                  display_name_tmp = string.gsub(display_name_tmp,'"','')
                  -- now remove the space
                  display_name = string.gsub(display_name_tmp,' ','')
         end
         return display_name
end

local function modify_CT_bandwidth(msg)
   local sdp = msg:getSdp()
   if sdp
   then
      local b_CT_line = sdp:getLine("b=CT","64")
      if not b_CT_line
      then
         local b_CT_line = sdp:getLine("b=CT","0")
         if not b_CT_line
         then
            return
         end
         b_CT_line = b_CT_line:gsub("0", "1000")
         sdp = sdp:modifyLine("b=CT", "0", b_CT_line)
         msg:setSdp(sdp)
         return
      end
      b_CT_line = b_CT_line:gsub("64", "1000")
      sdp = sdp:modifyLine("b=CT", "64", b_CT_line)
      msg:setSdp(sdp)
   end
end

local function remove_AS_bandwidth(msg)
   local sdp = msg:getSdp()
   if sdp
   then
```

```
            local b_AS_line = sdp:getLine("b=AS","64")
        if b_AS_line
        then
           sdp = sdp:removeLine("b=AS", "64")
           msg:setSdp(sdp)
        end
    end
end

local function process_outbound_request(msg)

    local method, ruri, ver = msg:getRequestLine()

    if string.find(ruri, "@")
    then
        local uri = ruri .. ";user=phone"
        msg:setRequestUri(uri)
    end

    modify_CT_bandwidth(msg)
    remove_AS_bandwidth(msg)

end

local function process_outbound_message(msg)
    modify_CT_bandwidth(msg)
    remove_AS_bandwidth(msg)
end


local function process_inbound_progress(msg)

    msg:setResponseCode(180, "Ringing")
    local sdp = msg:getSdp()

            if sdp then
            sdp = sdp:removeMediaDescription("audio")
            msg:setSdp(sdp)
            end

    local req = msg:getHeader("Require")
    local reqHeader = req
    if req
    then
        msg:removeHeader("Require")
    end

    local rseq = msg:getHeader("Rseq")
```

```
   local rseqPresnt = rseq
   if rseq
   then
      seqVal = msg:getHeaderValues("Rseq")
      msg:removeHeader("Rseq")
   end

   local sdp = msg:getSdp()
   if sdp
   then
      msg:removeUnreliableSdp()
   end

   if reqHeader
   then
      msg:addHeader("Require", "100rel")
   end

   if rseqPresnt
   then
     msg:addHeader("RSeq",seqVal[1])
   end
end

-- Future reference for changing cause values in divertion header scenario
-- local HiCauseToDiversion = { }
-- HiCauseToDiversion["302"] = "unconditional"
-- HiCauseToDiversion["486"] = "user-busy"
-- HiCauseToDiversion["408"] = "no-answer"
-- HiCauseToDiversion["480"] = "deflection"
-- HiCauseToDiversion["487"] = "deflection"
-- HiCauseToDiversion["503"] = "unavailable"
-- HiCauseToDiversion["404"] = "unknown"

function convertHIToDiversion(msg)
   local historyInfos = msg:getHeaderValues("History-Info")

   for i, hi in ipairs(historyInfos)
   do
      hi = string.gsub(hi, "%%3B", ";")
      hi = string.gsub(hi, "%%3D", "=")
      hi = string.gsub(hi, "%%22", "\"")
      hi = string.gsub(hi, "%%20", " ")

      -- MS format: <sip:+19728522619@med02.lynclabsj.local;user=phone>;index=1;ms-retarget-
reason=forwarding

      local uri, index, reason = string.match (hi, "<(sip:.*@.*)>;index=(.*)reason=(.*)")
```

```
      trace.format("hi: uri '%s', reason '%s'", uri or "nil", reason or "nil")


      if uri
      then
         local diversion = string.format("<%s>", uri)
         if reason
         then
            diversion = string.format("<%s>;reason=\"unconditional\"", uri)
         end
         msg:addHeader("Diversion", diversion)
      end
   end
end

function convertReferredByToDiversion(msg)

   local refInfo = msg:getHeader("Referred-By")
   if refInfo
   then
      local diversion = string.format("%s;reason=\"unconditional\"", refInfo)
      msg:addHeader("Diversion", diversion)
   end

end

local function replaceHistoryHeader(msg)

   local hist = msg:getHeader("History-Info")

   if hist
   then
      convertHIToDiversion(msg)

      local di  = msg:getHeader("Diversion")

      if di
      then
         msg:removeHeader("History-Info")
      end
   end

end

local function replaceReferredByHeader(msg)

   local refby = msg:getHeader("Referred-By")
```

```
   if refby
   then
      convertReferredByToDiversion(msg)
   end

end

local function modifyUserFrom(msg)
-- get a data from "From" header and replace
   local removeUser= ""
   local value = msg:getHeader("From")

   if value
   then
      value = value:gsub(";user=phone", removeUser)
      if value
      then
         msg:modifyHeader("From", value)
      end
   end
end

local function process_inbound_request(msg)
   modifyUserFrom(msg)
   replaceHistoryHeader(msg)
   replaceReferredByHeader(msg)
   removecryptoline(msg)
   local sdp = msg:getSdp()
    if sdp
    then

        local tcap = sdp:getLine("a=tcap:", "RTP/SAVP")
if
   tcap
      then
      local a_m_line = sdp:getLine("m=audio", "RTP/AVP")
      a_m_line = a_m_line:gsub("AVP", "SAVP")
      sdp = sdp:modifyLine("m=audio", "RTP/AVP", a_m_line)
      end
       sdp=sdp:removeLine("a=crypto:", "|2^31|")
         msg:setSdp(sdp)
    end
end

function process_inbound_any_response(msg)
   msg:addHeader("SUPPORTED","X-cisco-srtp-fallback")
   local sdp = msg:getSdp()
```

```
   if sdp
   then
      trace.format("Inbound SDP")
      local tcap = sdp:getLine("a=tcap:", "RTP/SAVP")
      if tcap
      then
         local a_m_line = sdp:getLine("m=audio", "RTP/AVP")
         a_m_line = a_m_line:gsub("AVP", "SAVP")
         sdp = sdp:modifyLine("m=audio", "RTP/AVP", a_m_line)
      end
      trace.format("before removing crypto")
      sdp=sdp:removeLine("a=crypto:", "|2^31|")
      trace.format("after removing crypto")
      msg:setSdp(sdp)
   end
end

function removecryptoline(msg)
   local sdp = msg:getSdp()
   if sdp
   then
      trace.format("removecryptoline before removing crypto")
      sdp=sdp:removeLine("a=crypto:", "|2^31|")
      trace.format("removecryptoline after removing crypto")
      msg:setSdp(sdp)
   end

end


function process_inbound_any_request(msg)
  msg:addHeader("SUPPORTED","X-cisco-srtp-fallback")
end

M.outbound_INVITE         = process_outbound_request
M.outbound_ACK            = process_outbound_message
M.outbound_200_INVITE     = process_outbound_message
M.outbound_18X_INVITE     = process_outbound_message

M.inbound_183_INVITE      = process_inbound_progress
M.inbound_INVITE          = process_inbound_request
M.inbound_ANY_ANY         = process_inbound_any_response
M.inbound_ANY             = process_inbound_any_request

return M
```

SIP Trunk to Skype for Business Configuration

Navigation: Device → Trunk

Set **Device Name*= SFB**. This is used for the test

Set **Device Pool* = G711_pool**. This is used for the test

Set **Call Classification*= Use System Default**. This is used for the test

Set **Media Resource Group List = MRGL_SW_NoMTP**. This is used for the test

Uncheck Media Termination Point Required

Check Run On All Active Unified CM Nodes

Check Redirecting Diversion Header Delivery – Inbound

Set **Destination Address = 172.16.29.62** [FQDN of Skype for Business Mediaition Server] this is used in the test

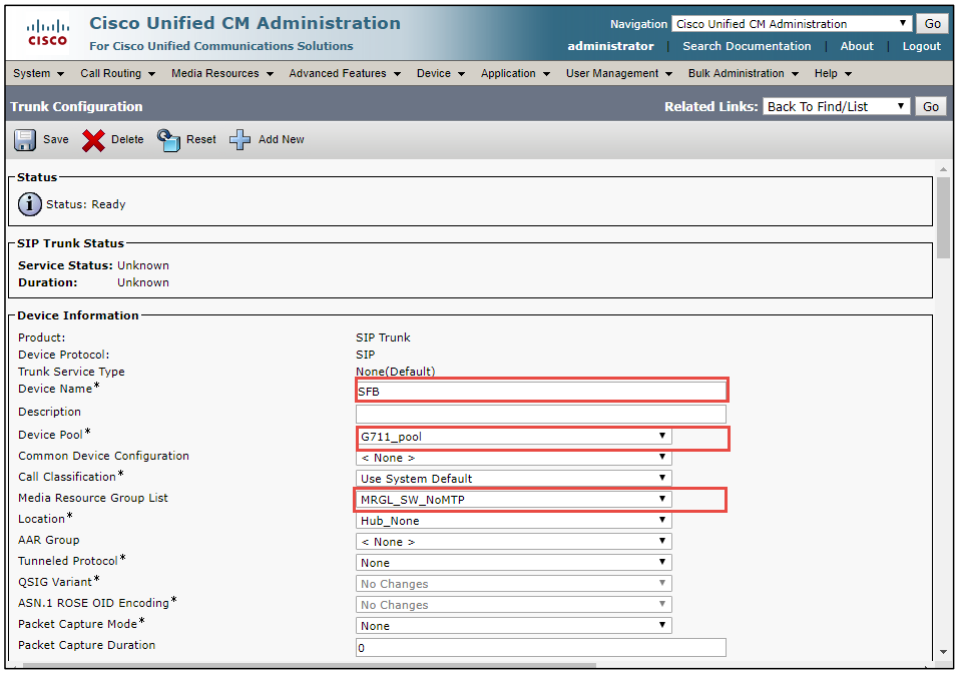
Set **SIP Trunk Security Profile*= SFB_SECURITY_PROFILE**

Set **SIP Profile*= SFB_SIP_PROFILE** – Standard SIP Profile

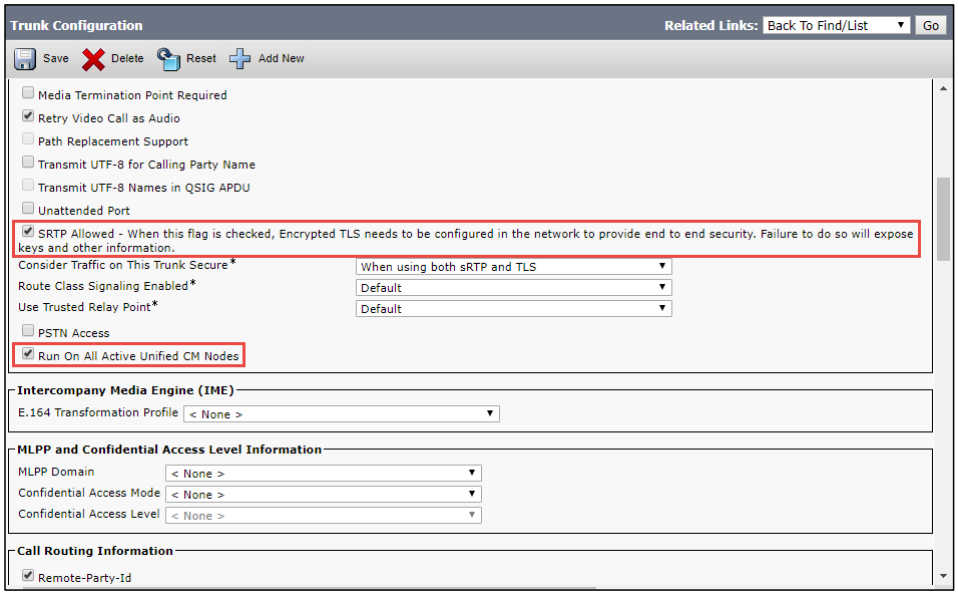
Set **DTMF Signaling Method*= No Preference**

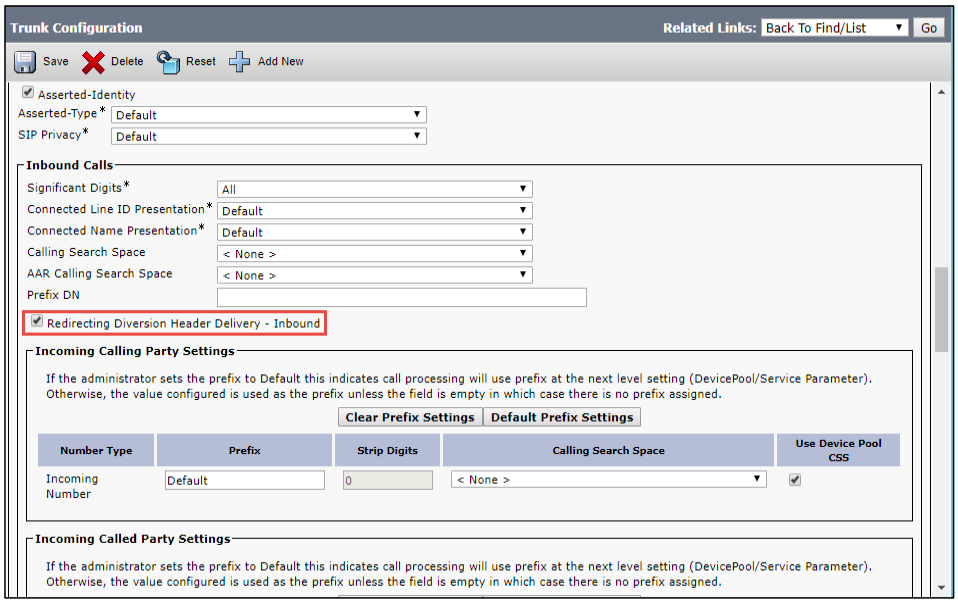
Set **Normalization Script = lync_interop_updated**

All other values are default.

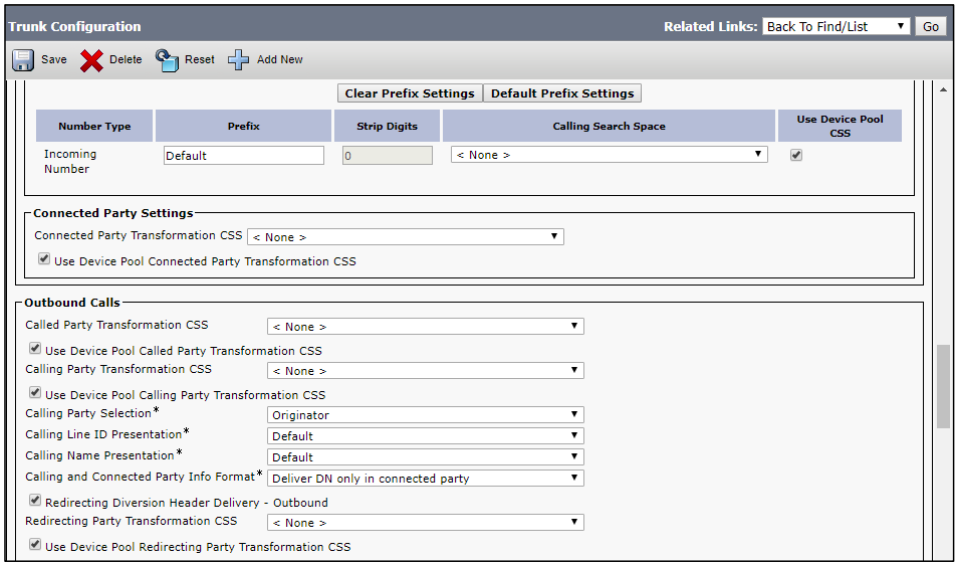

Cisco Unified CM Administration
For Cisco Unified Communications Solutions
Navigation Cisco Unified CM Administration Go
administrator | Search Documentation | About | Logout
System Call Routing Media Resources Advanced Features Device Application User Management Bulk Administration Help
Trunk Configuration
Related Links: Back To Find/List Go
Save Delete Reset Add New
Status
Status: Ready
SIP Trunk Status
Service Status: Unknown
Duration: Unknown
Device Information
Product: SIP Trunk
Device Protocol: SIP
Trunk Service Type None(Default)
Device Name* SFB
Description
Device Pool* G711_pool
Common Device Configuration < None >
Call Classification* Use System Default
Media Resource Group List MRGL_SW_NoMTP
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

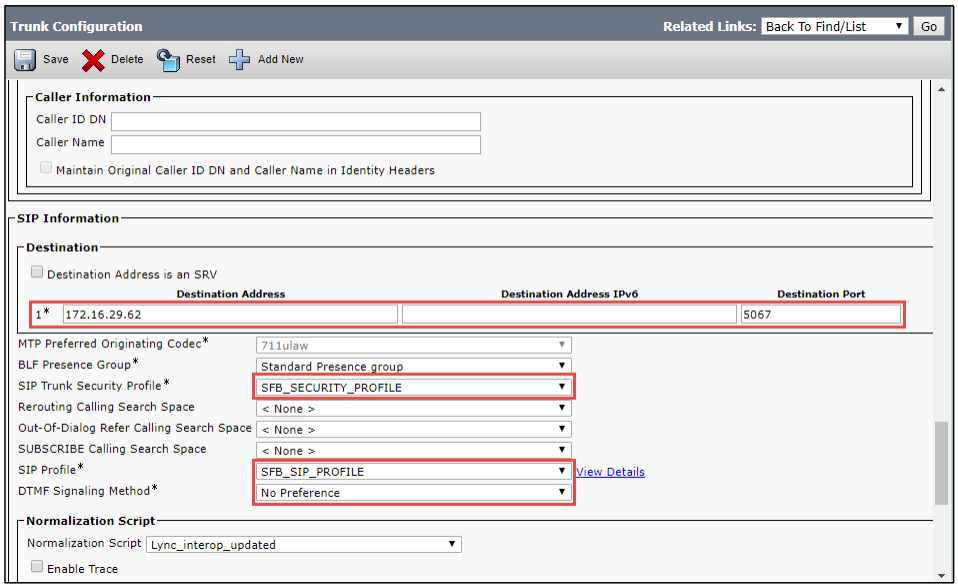
Cisco Unified Communications Manager SIP Trunk to Skype for Business Configuration (Continued)



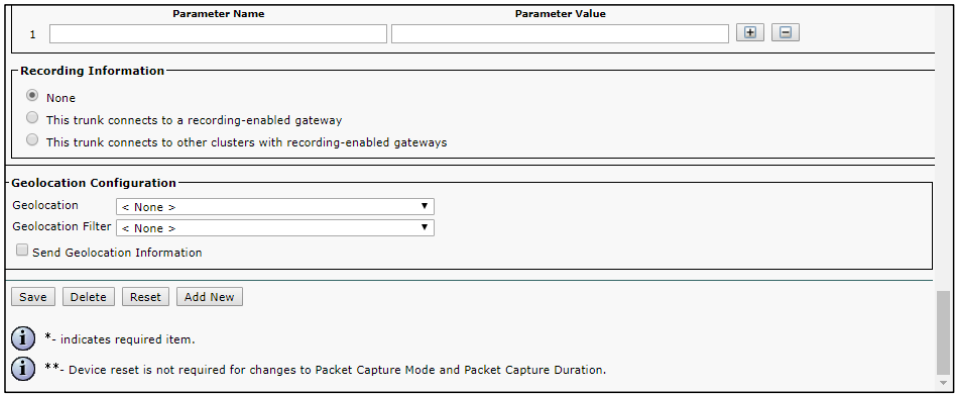


Cisco Unified Communications Manager SIP Trunk to Skype for Business Configuration (Continued)





Cisco Unified Communications Manager SIP Trunk to Skype for Business Configuration (Continued)



SIP Trunk to Cisco Unity Connection Configuration

Navigation: Device → Trunk

Set **Device Name*= Clus21unity**. This is used for the test.

Set **Device Pool* = Default**

Check Run On All Active Unified CM Nodes

Check Redirecting Diversion Header Delivery – Inbound

Check Redirecting Diversion Header Delivery – Outbound

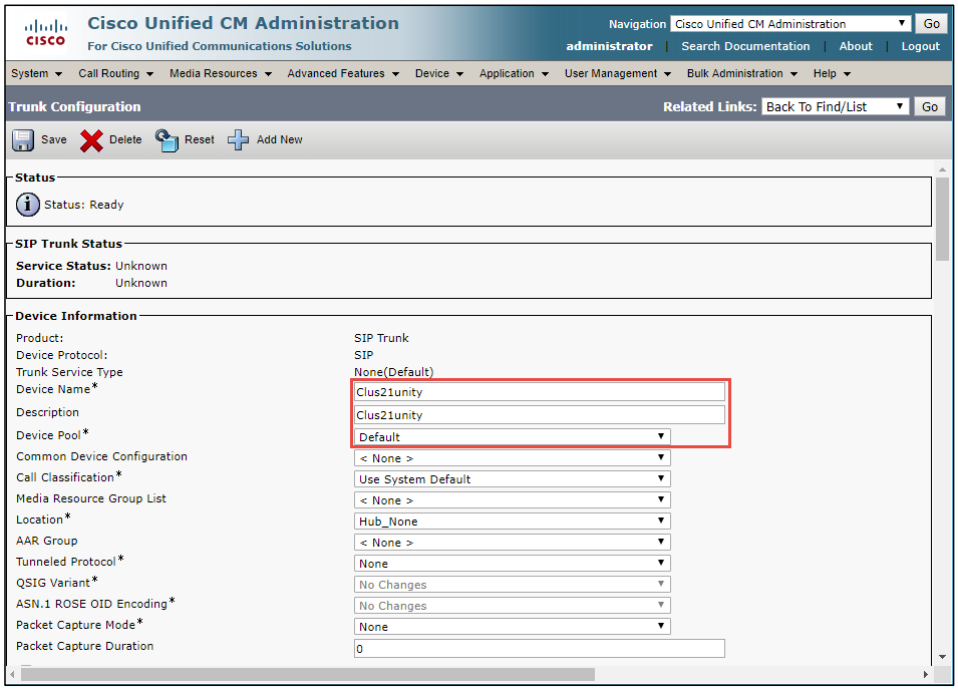
Set **Destination Address = 10.80.11.6**. This is used for the test.

Set **SIP Trunk Security Profile*= TLS_CUC**

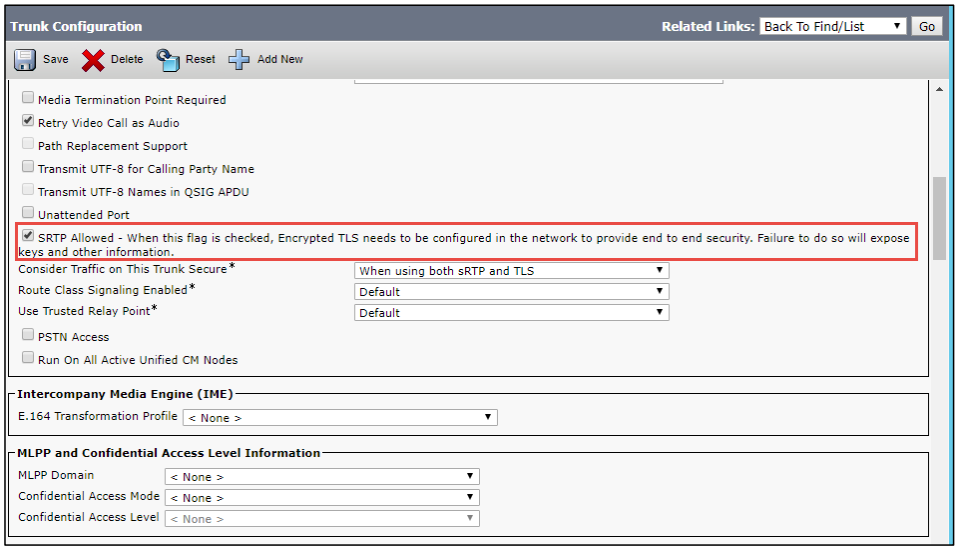
Set **SIP Profile*= Standard SIP Profile - OPTIONS**

Set **DTMF Signaling Method *= No Preference**

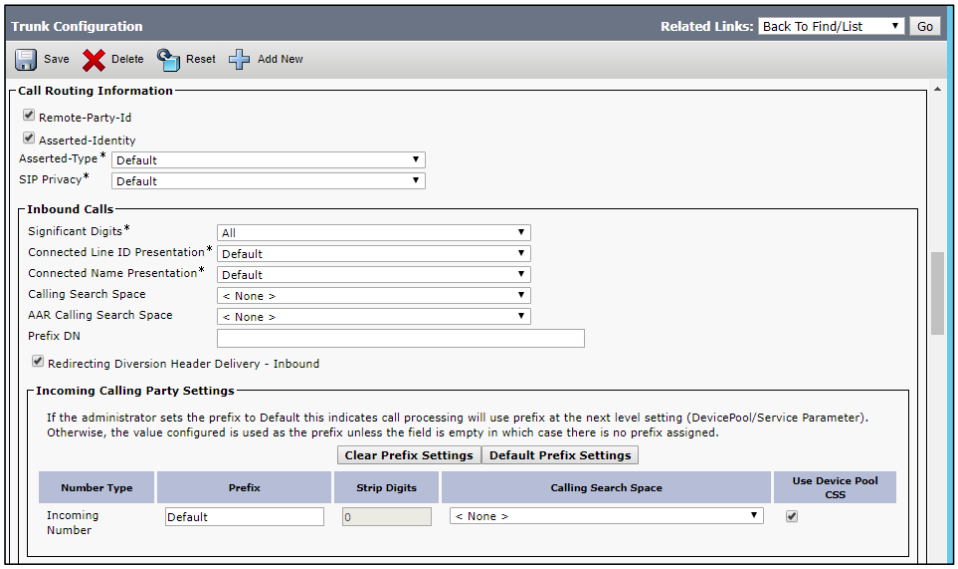
All other values are default



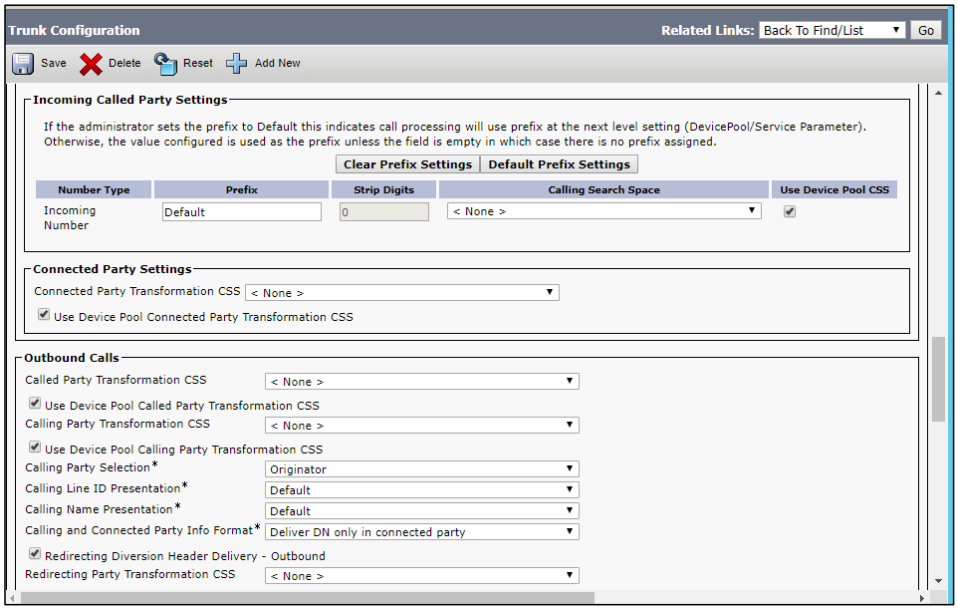
Cisco Unified Communications Manager SIP Trunk to Cisco Unity Connection Configuration (Continued)

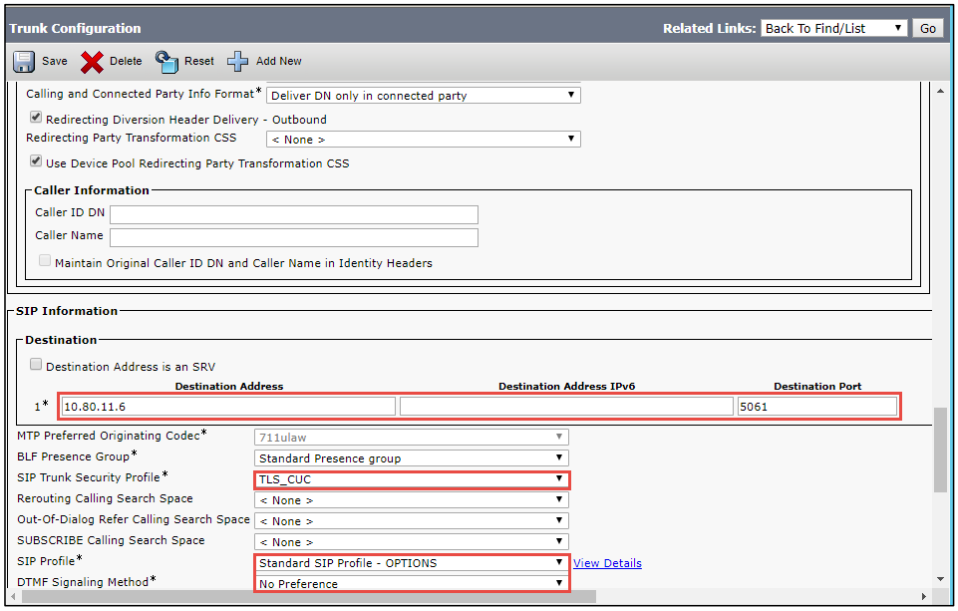


Cisco Unified Communications Manager SIP Trunk to Cisco Unity Connection Configuration (Continued)



Cisco Unified Communications Manager SIP Trunk to Cisco Unity Connection Configuration (Continued)





Voice Mail Configuration

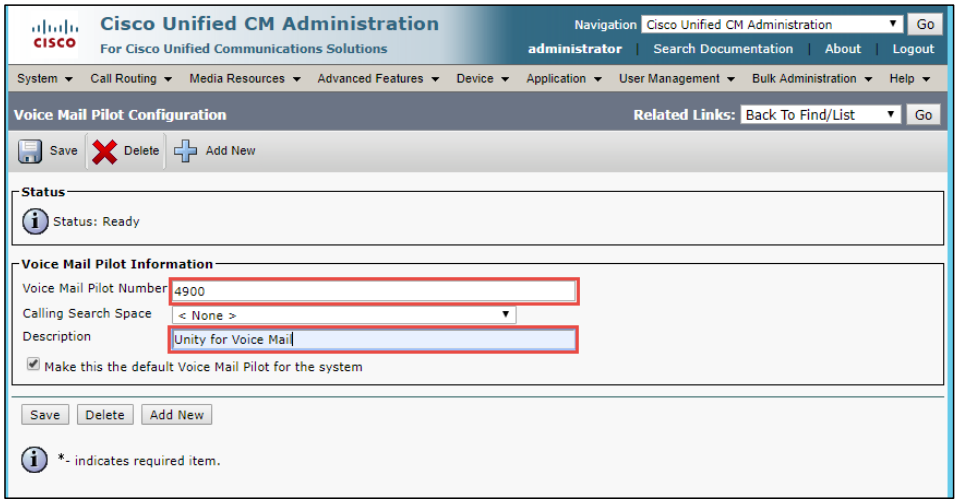
Configure Voice Mail Pilot:

Navigation: Advanced Features → Voice Mail → Voice Mail Pilot

Add new

Set **Voice Mail Pilot Number = 4900** .This is used for the test

Set **Description = Unity for Voice Mail** .This text is used to identify this SIP Profile



Route Pattern to Skype for Business Extensions

Navigation: Call Routing → Route/Hunt → Route Pattern

Add New

Set **Route Pattern* = 2XXX**. This is used to route to the Skype for Business in this test

Set **Gateway/Route List* = SFB**. This is used for the test

Uncheck Provide Outside Dial Tone

Commented [RSM4]: Checked in the screenshot

Set **Calling Line ID Presentation= Default**

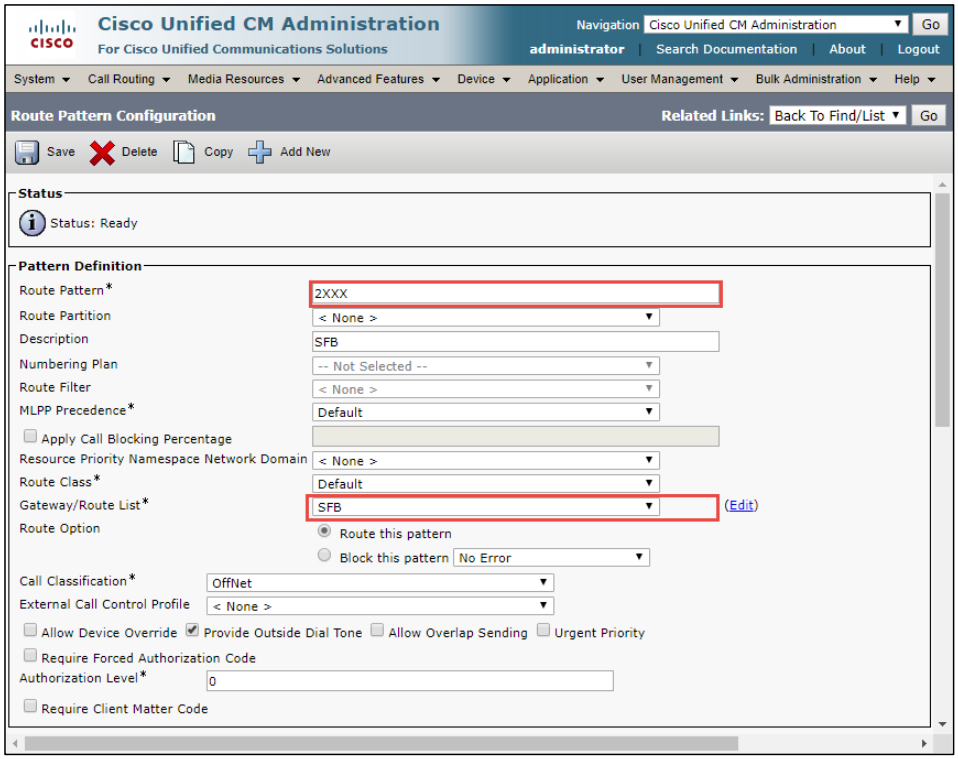
Set **Calling Name Presentation= Default**

Set **Connected Line ID Presentation*= Default**

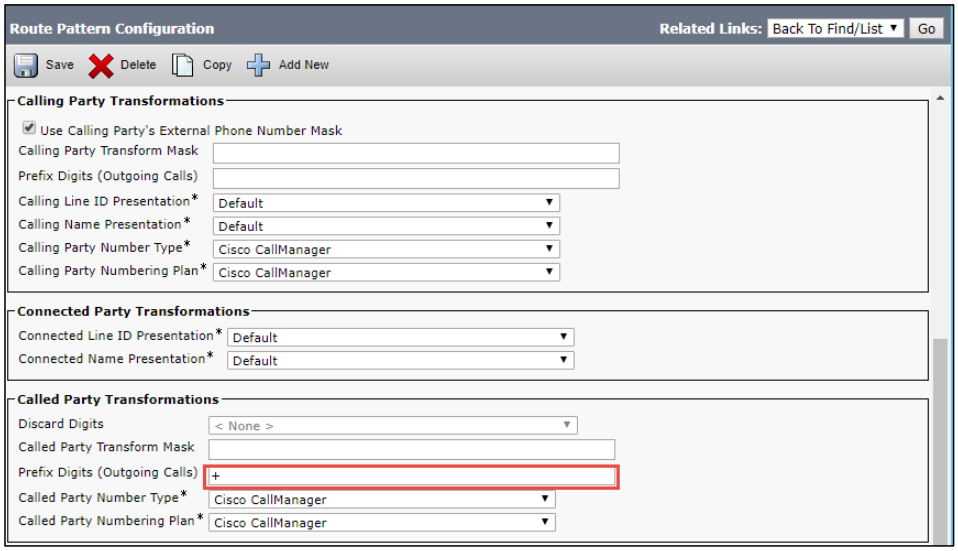
Set **Calling Name Presentation* = Default**

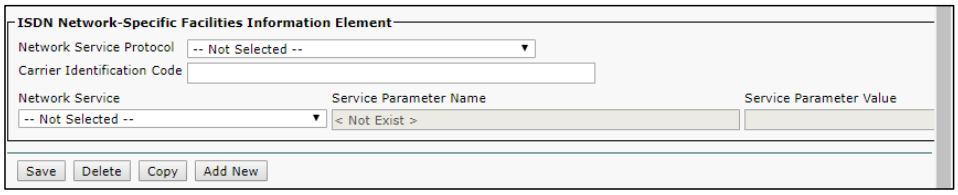
Prefix digit Outgoing Calls (Out Going) = +, Skype user configured + as prefix

All other values are default.



Route Pattern Configuration for 8xxx (Continued)





Route Pattern to invoke Jabber client with Remote Destination configured as Skype for Business Extensions

Navigation: Call Routing → Route/Hunt → Route Pattern

Set **Route Pattern* = \+.2XXX**. This is used to route to the Skype for Business when using the Extend and Connect functionality in this test

Set **Gateway/Route List* = SFB**. This is used for the test

Uncheck Provide Outside Dial Tone

Set **Calling Line ID Presentation= Default**

Set **Calling Name Presentation= Default**

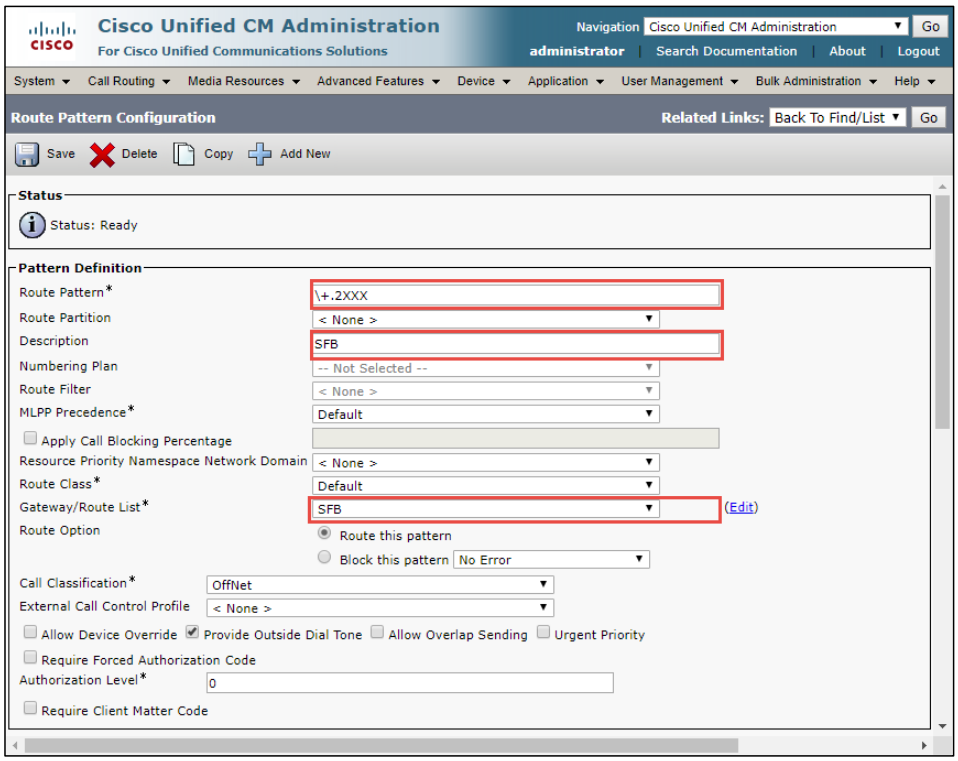
Set **Connected Line ID Presentation*= Default**

Set **Calling Name Presentation* = Default**

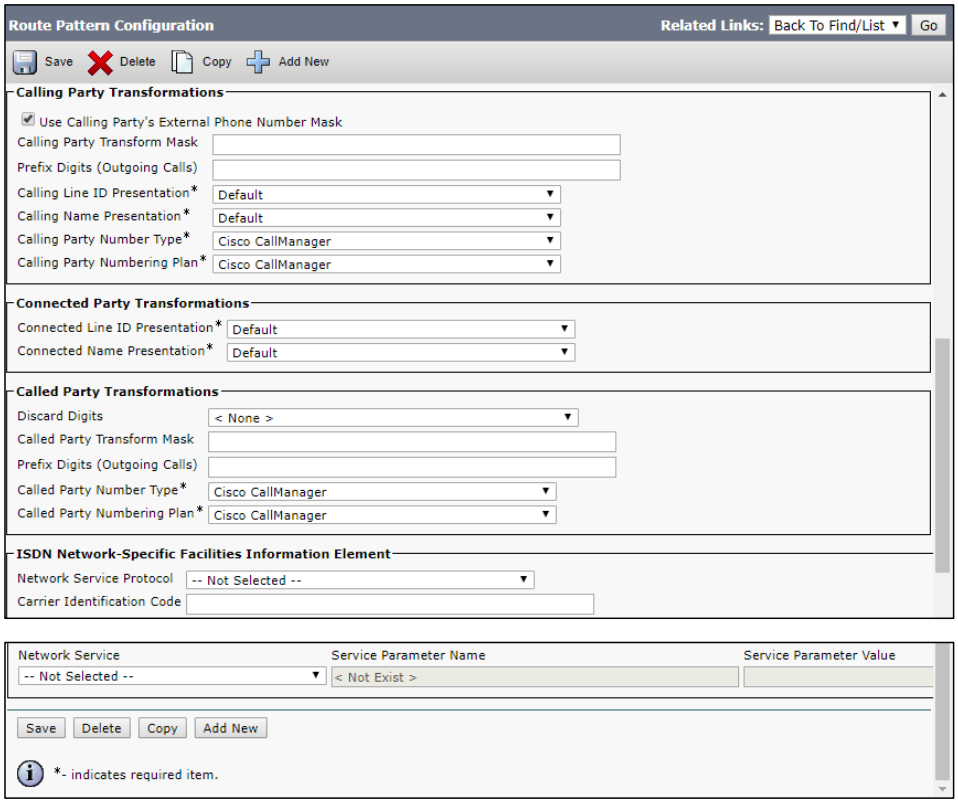
Set **Discard Digits = PreDot**

Commented [RSM5]: Not used in the screenshot

All other values are default

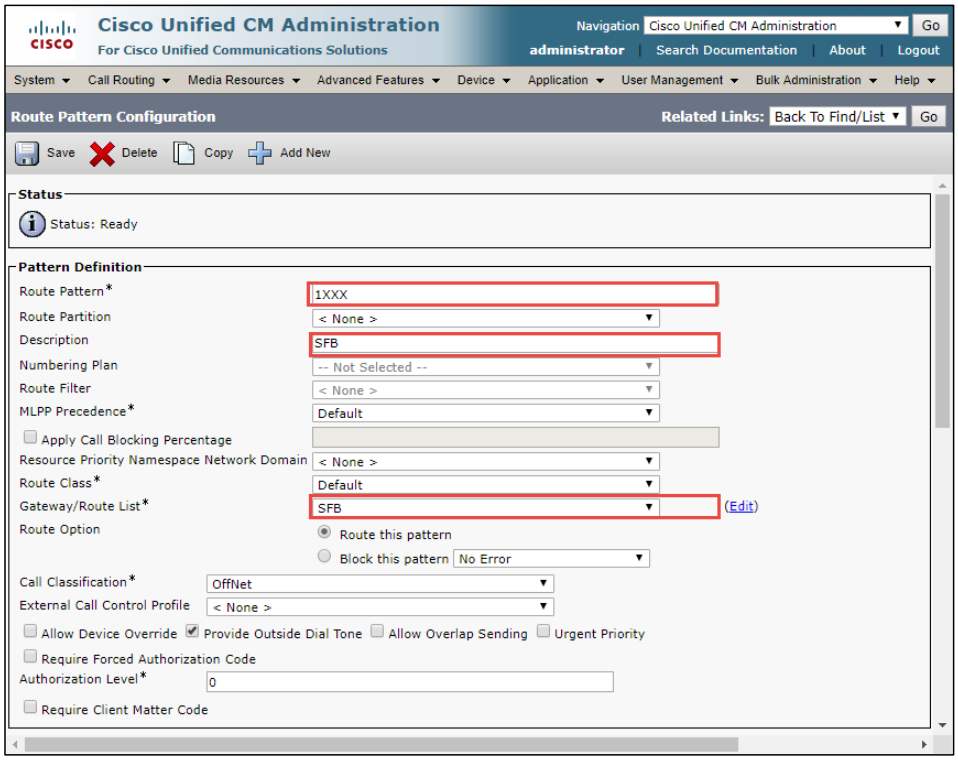


Cisco Unified Communications Manager Route pattern Configuration (Continued)

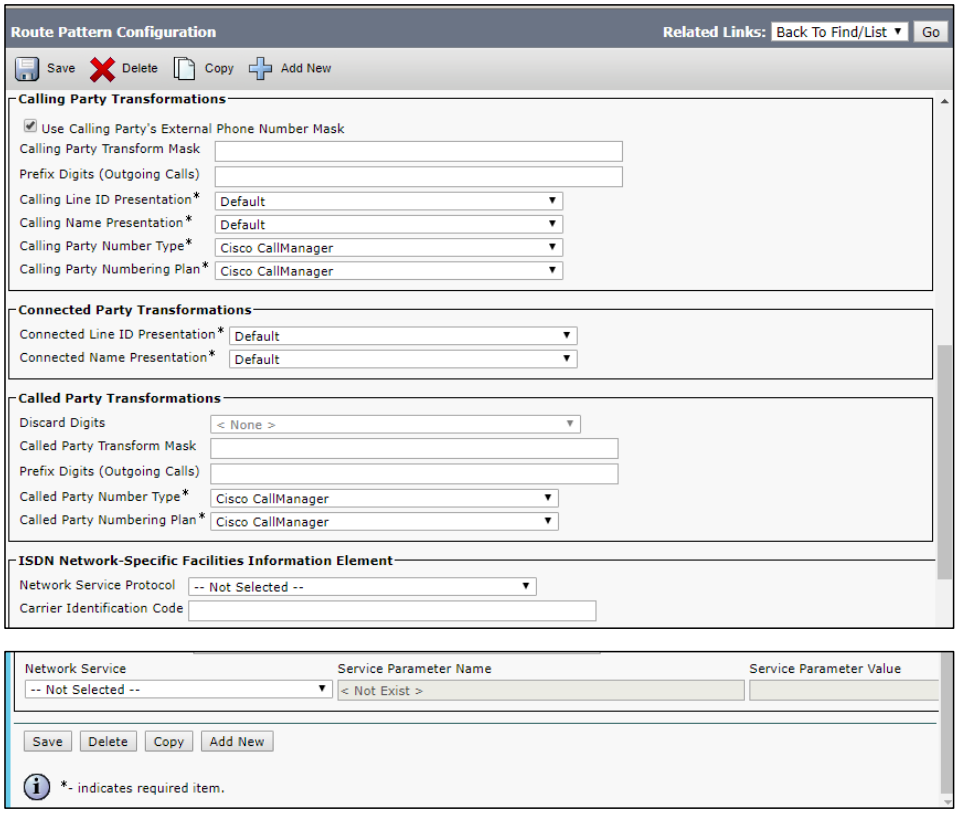


Route Pattern to Skype for Business Call Park range

The Skype for Business Call Park range configured is 1000-1999 .The following route pattern “1XXX” is therefore configured to enable a parked call to be retrieved from Cisco UCM.

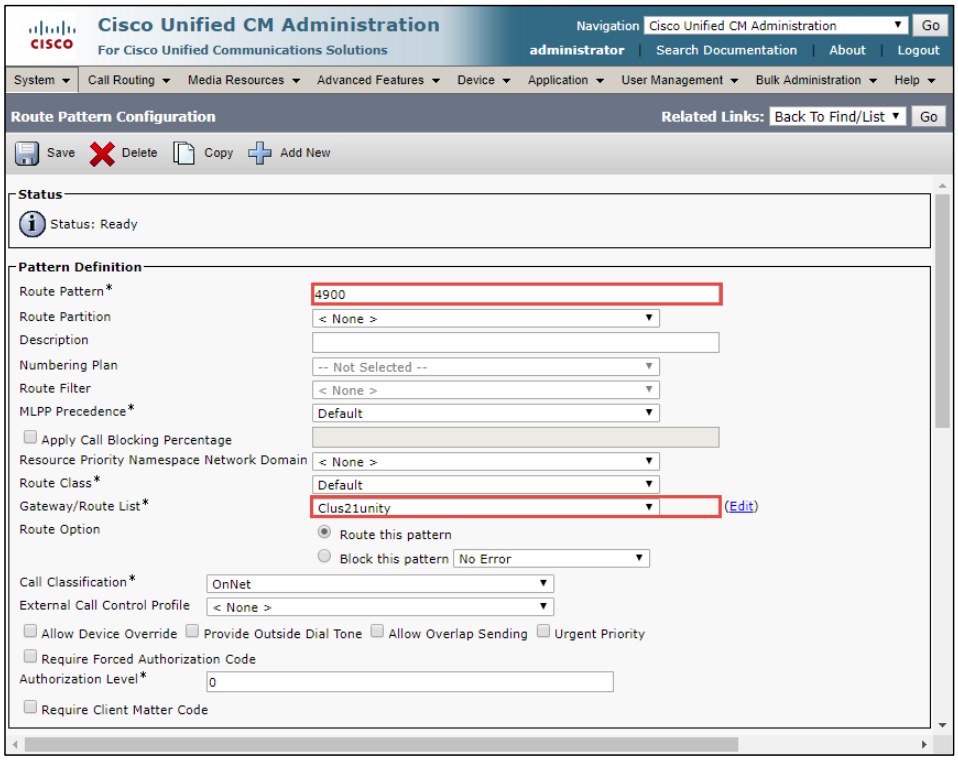


Cisco Unified Communications Manager Call Park Range Configuration (Continued)

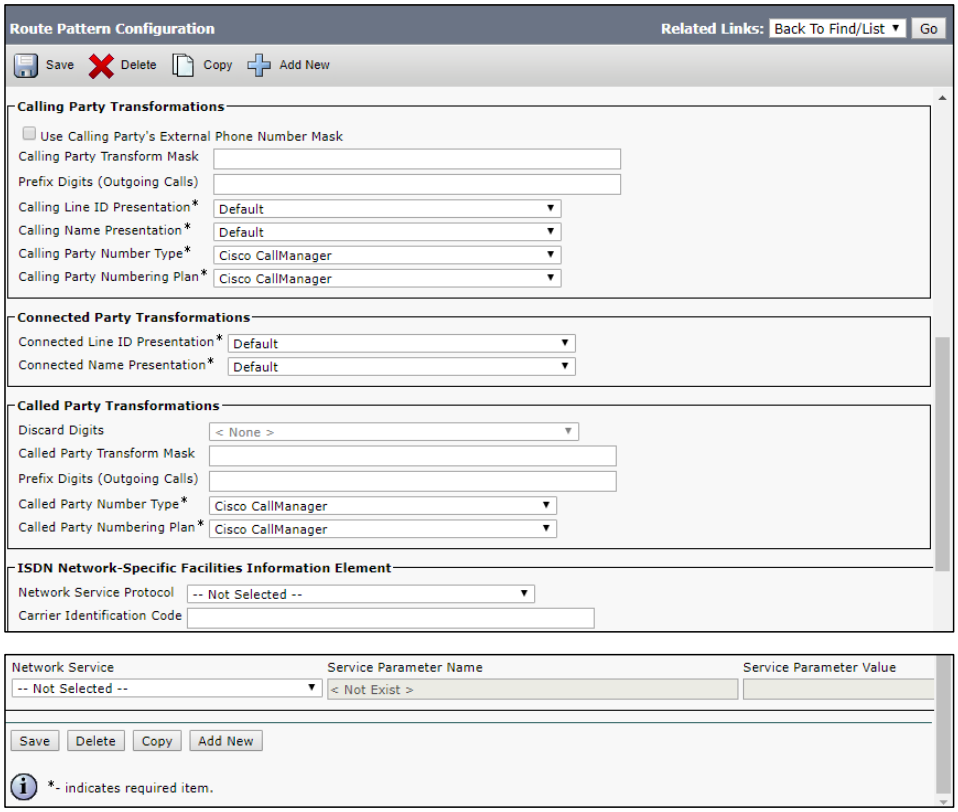


Route Pattern to Unity Connection Voice Mail

A route pattern 4900 (which is the voice mail pilot), is configured to reach Unity Connection Voice Mail.



Cisco Unified Communications Manager Route Pattern Configuration (Continued)



Cisco UCM Extent and Connect

Extend and Connect is a feature that allows administrators to rapidly deploy UC Computer Telephony Integration (CTI) applications, which interoperate with any endpoint. With Extend and Connect, users can leverage the benefits of UC applications from any location using any device. This feature also allows Interoperability between newer UC solutions and legacy systems, so customers can migrate to newer UC Solutions over time as existing hardware is deprecated.

UC service Configuration

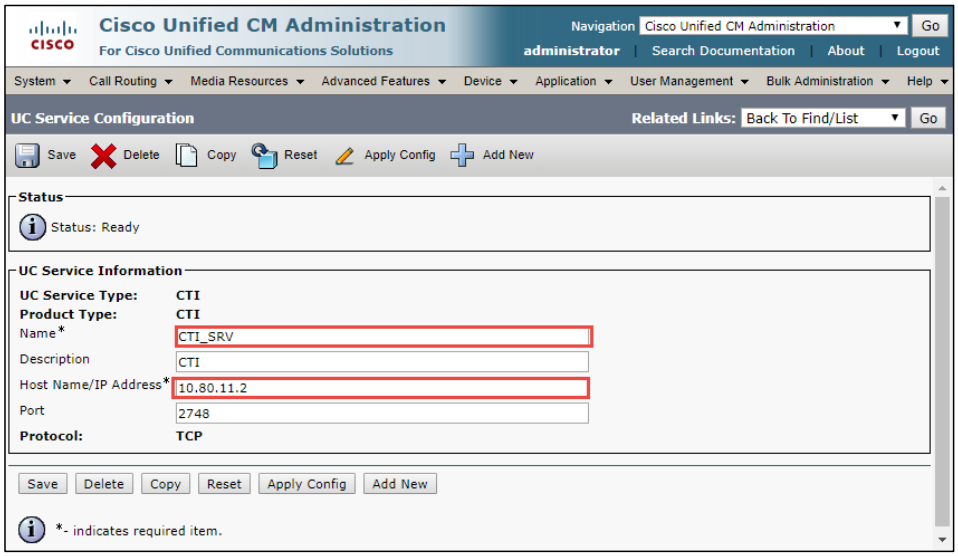
Navigation Path: User Management→ User setting → UC Service

Add New

Select Service Type as CTI

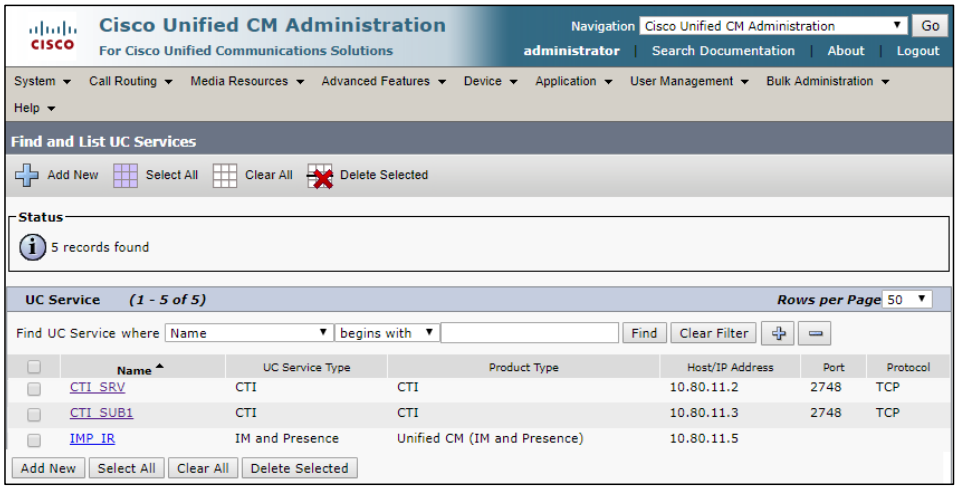
Set **Name = IMPTrunk**

Set **Host Name/IP Address* = 10.80.11.2**; this is the Cisco UCM publisher IP.



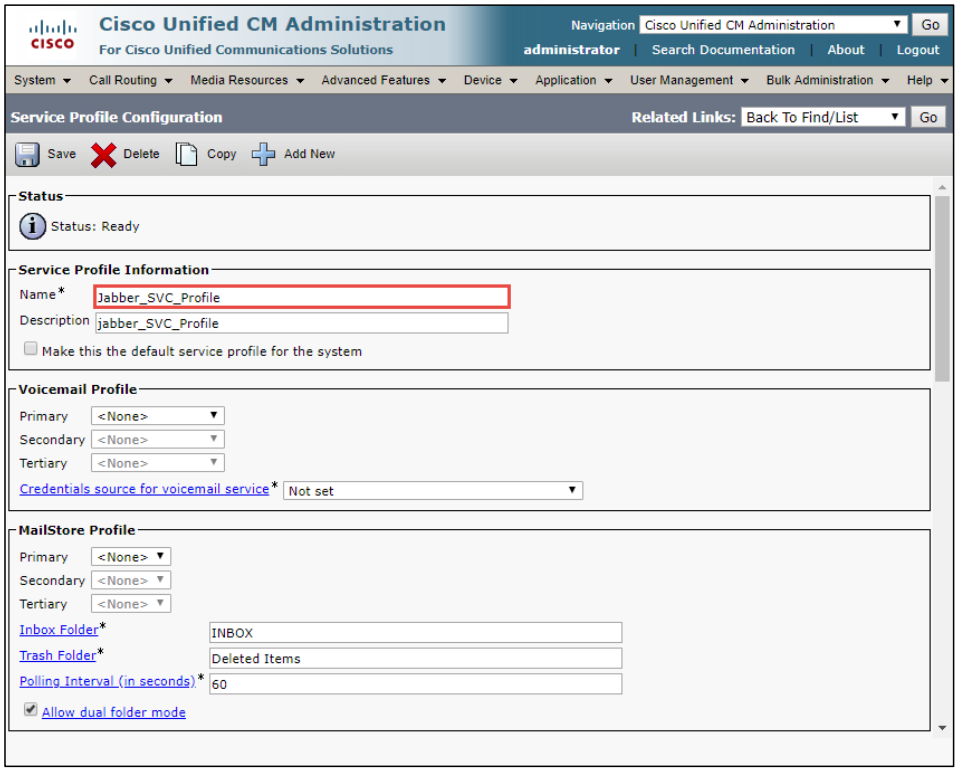
In the same manner, a UC Service is configured for the subscriber also.

A UC service of Type IM and Presence is configured with the IP of the Presence server.

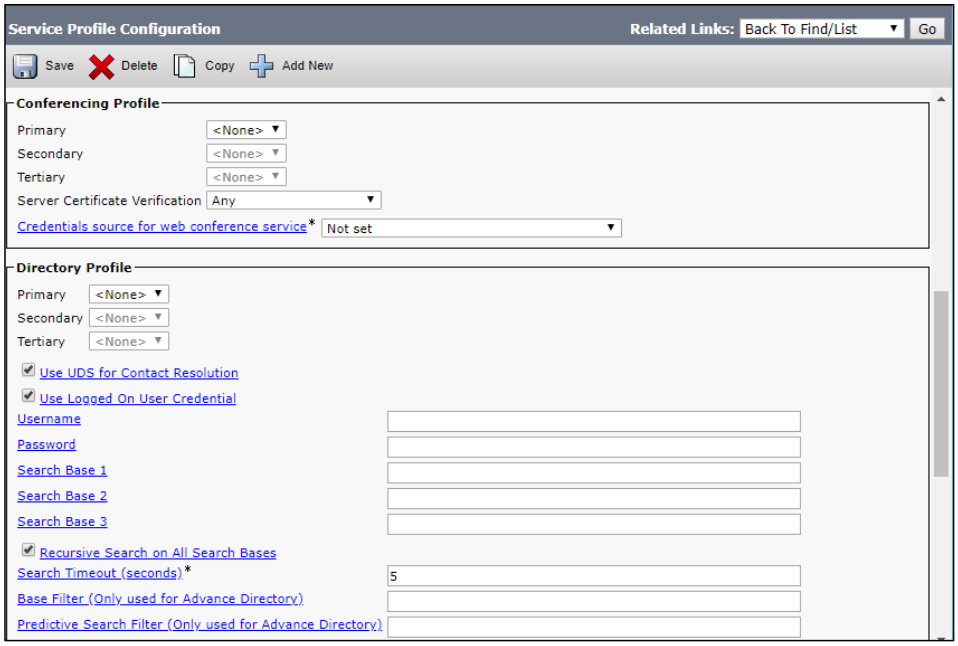

Cisco Unified CM Administration
For Cisco Unified Communications Solutions
Navigation Cisco Unified CM Administration Go
administrator | Search Documentation | About | Logout
System Call Routing Media Resources Advanced Features Device Application User Management Bulk Administration Help
Find and List UC Services
Add New Select All Clear All Delete Selected
Status
5 records found
UC Service (1 - 5 of 5)
Rows per Page 50
Find UC Service where Name begins with Find Clear Filter
Name UC Service Type Product Type Host/IP Address Port Protocol
CTI_SRV CTI CTI 10.80.11.2 2748 TCP
CTI_SUB1 CTI CTI 10.80.11.3 2748 TCP
IMP_IR IM and Presence Unified CM (IM and Presence) 10.80.11.5
Add New Select All Clear All Delete Selected

service Profile Configuration

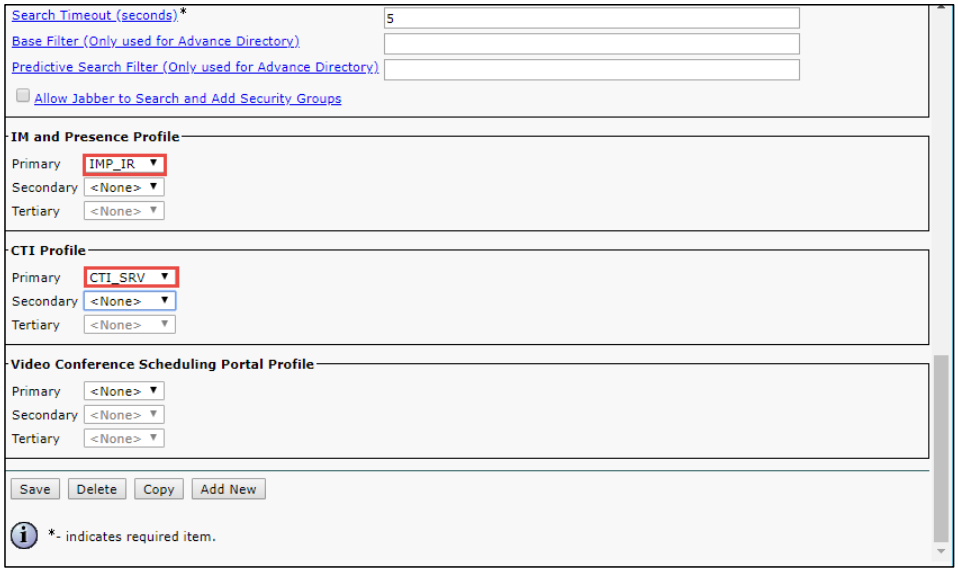
Navigation: User Management→ User setting → Service Profile



Cisco UCM service profile Configuration (Continued)



Cisco UCM service profile Configuration (Continued)



Cisco Unified CM IM Presence – CCMCIP Profile Configuration

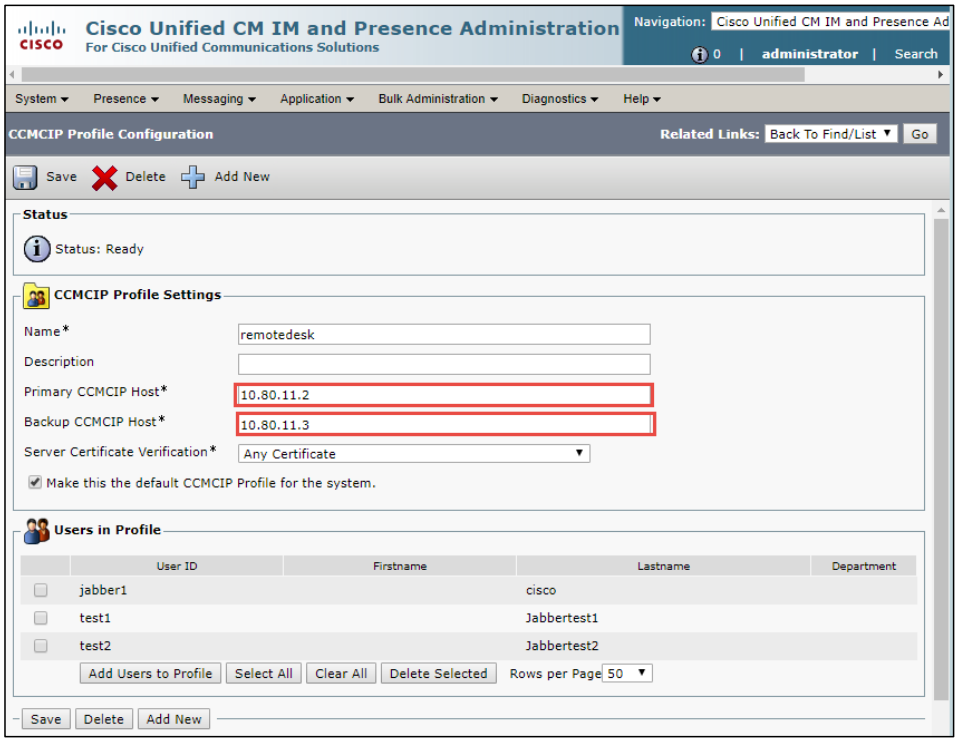
Navigation Path: Application → CCMCIP Profile

Set Name *: remotedesk. This is used in this example.

Set **Primary CCMCIP Host *: 10.80.11.2**.Cisco Publisher IP. This is used in this test.

Set **Backup CCMCIP Host *: 10.80.11.3**.Cisco Publisher IP. This is used in this test.

Add Users to Profile: jabber1, test2 and test3 .This is used in this test.



SIP trunk to Cisco IM&Presence Trunk Configuration

Navigation Path: Device→ Trunk

Set **Device Name*=IMPTrunk**. This is used for the test.

Set **Device Pool* = Default**. This is used for the test.

Set **Media Resource Group List = MRGL_SW_NoMTP**. This is used for the test.

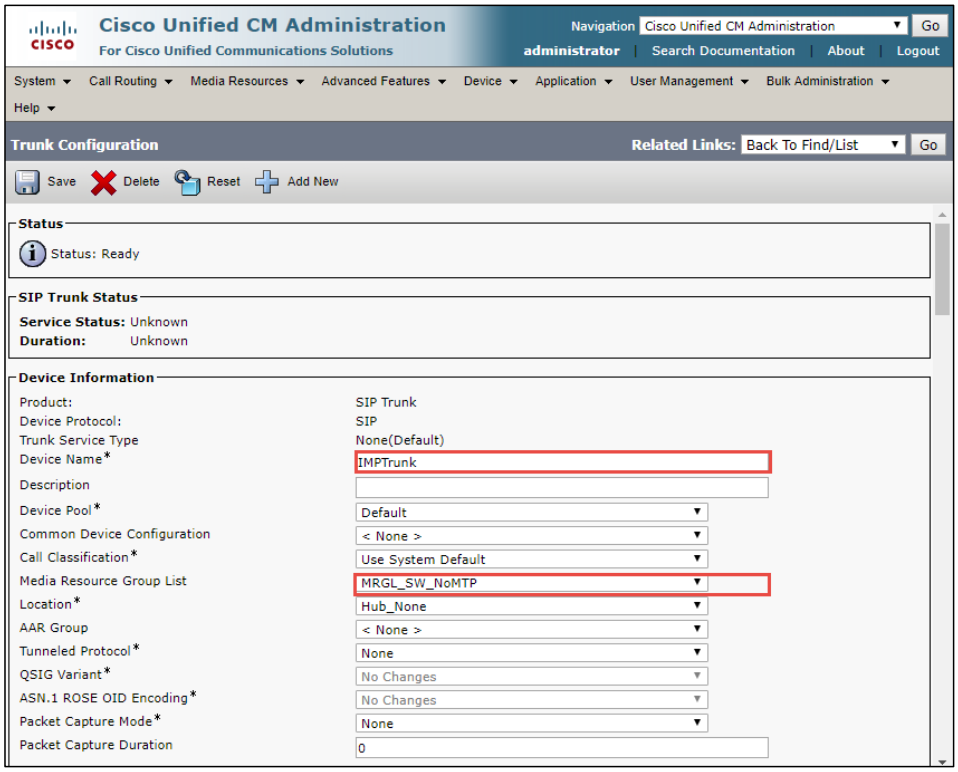
Set **Destination Address = 10.80.11.5**. This is used in this example.

Set **SIP Trunk Security Profile*= Non-Secure SIP Trunk Profile**.

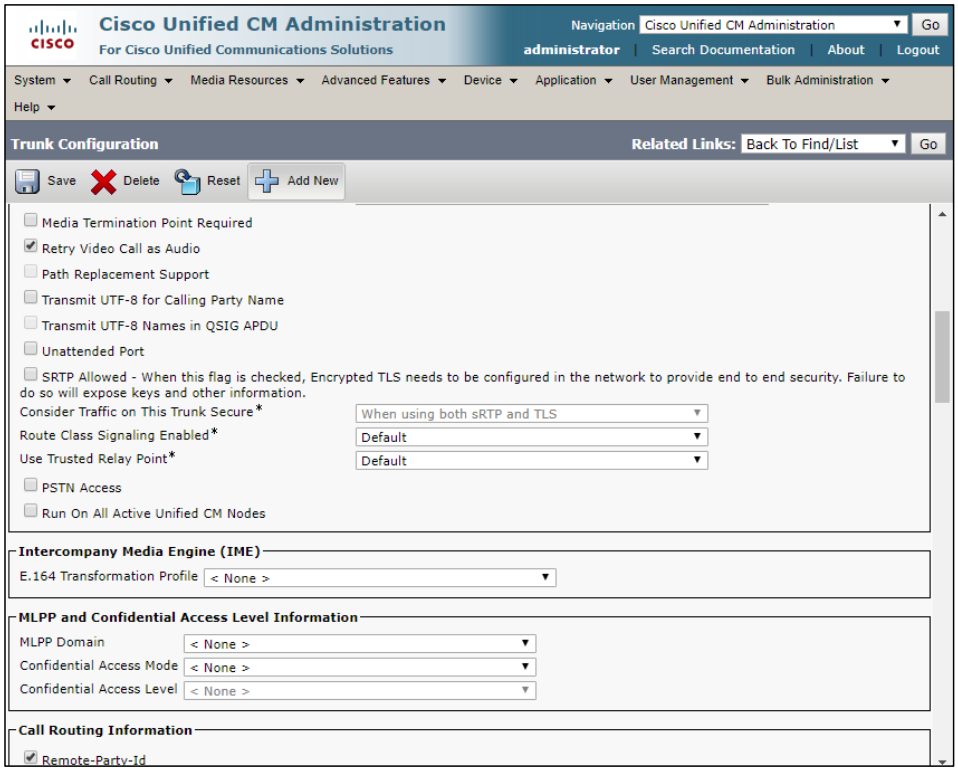
Set **SIP Profile*= Standard SIP Profile.**

Set **DTMF Signaling Method*= No Preference.**

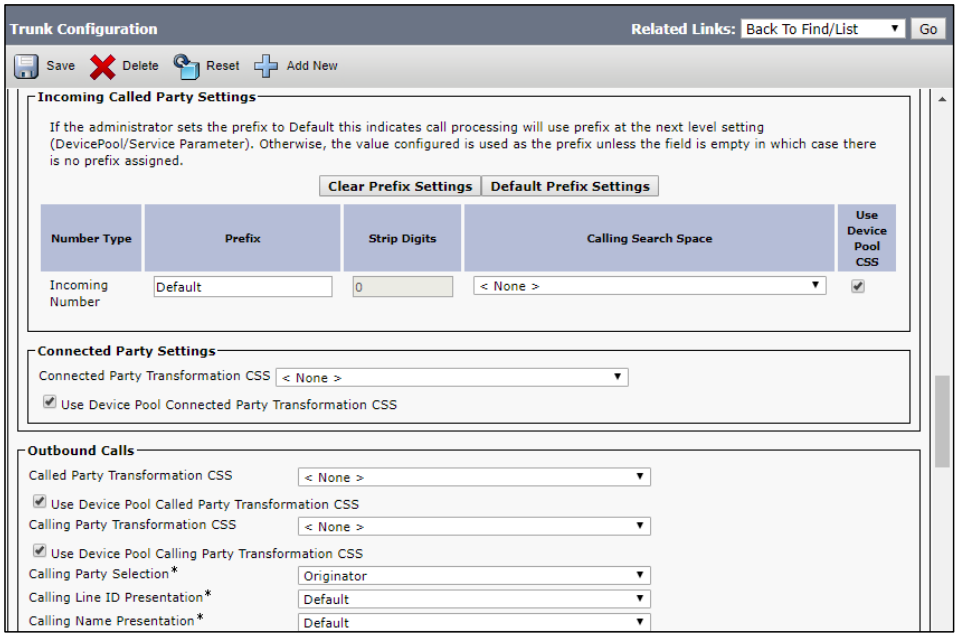
All other values are default.


Cisco Unified CM Administration
For Cisco Unified Communications Solutions
Navigation Cisco Unified CM Administration Go
administrator | Search Documentation | About | Logout
System Call Routing Media Resources Advanced Features Device Application User Management Bulk Administration Help
Trunk Configuration
Related Links: Back To Find/List Go
Save Delete Reset Add New
Status
Status: Ready
SIP Trunk Status
Service Status: Unknown
Duration: Unknown
Device Information
Product: SIP Trunk
Device Protocol: SIP
Trunk Service Type None(Default)
Device Name* IMPTrunk
Description
Device Pool* Default
Common Device Configuration < None >
Call Classification* Use System Default
Media Resource Group List MRGL_SW_NoMTP
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

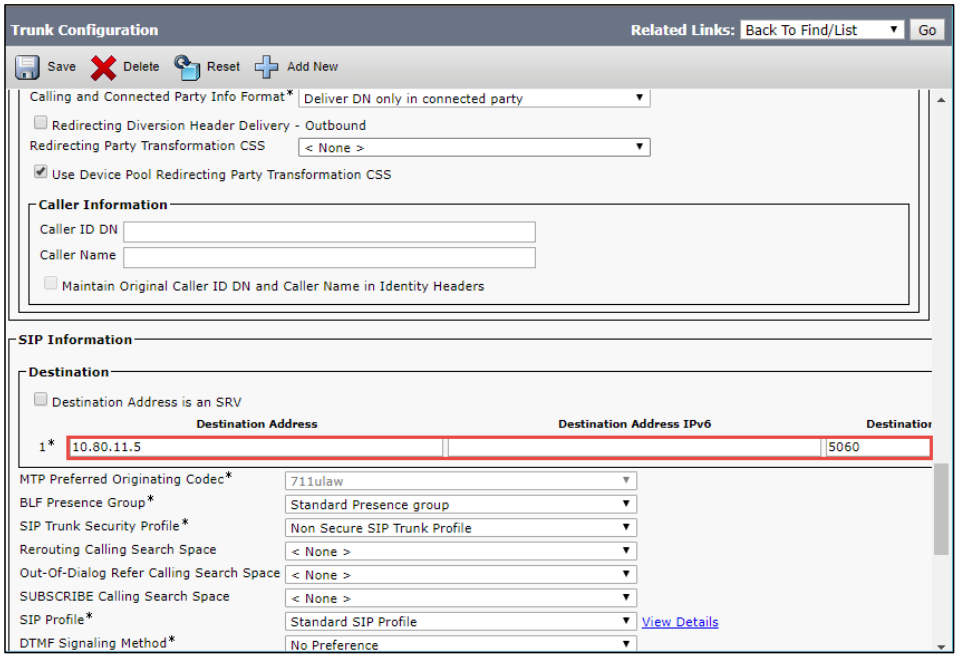
Cisco Unified Communications Manager SIP Trunk to CUP Configuration (Continued)

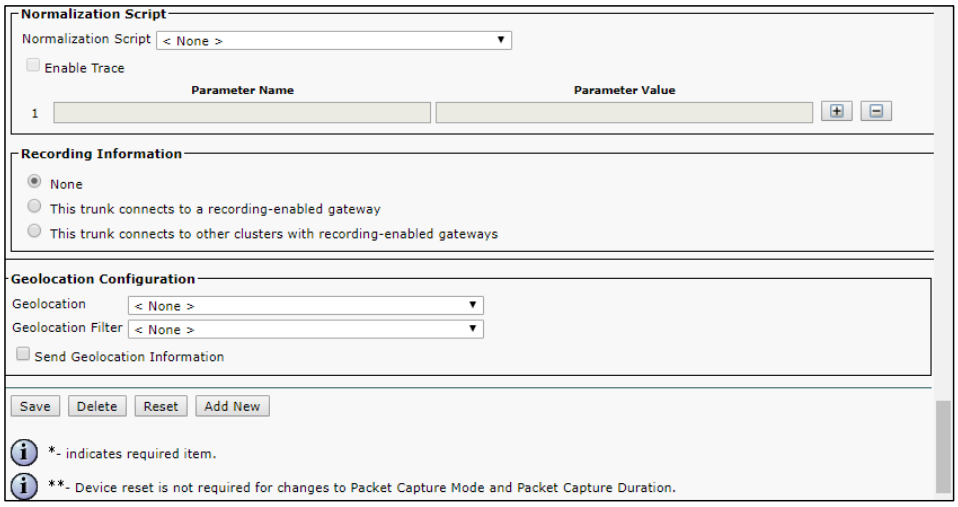


Cisco UCM SIP Trunk to CUP Configuration (Continued)



Cisco UCM SIP Trunk to CUP Configuration (Continued)





end user configuration

Add user to Cisco UCM

Navigation: User Management→ End user

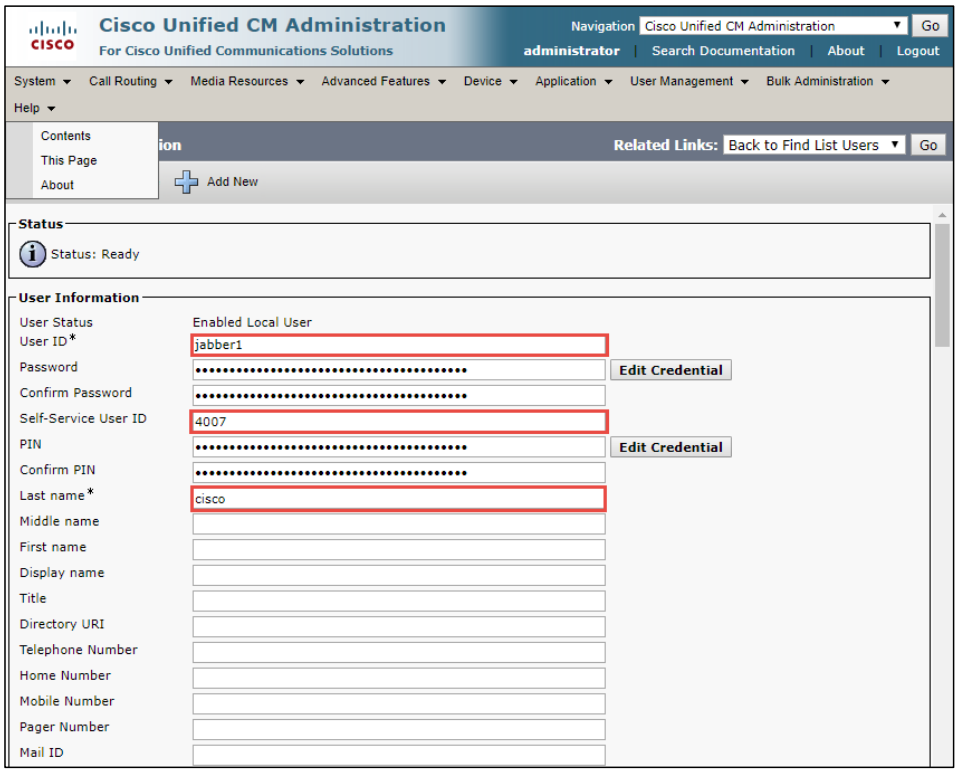
Set **User ID*= jabber1**. This is used for the test.

Set **Last Name = cisco**. This is used for the test.

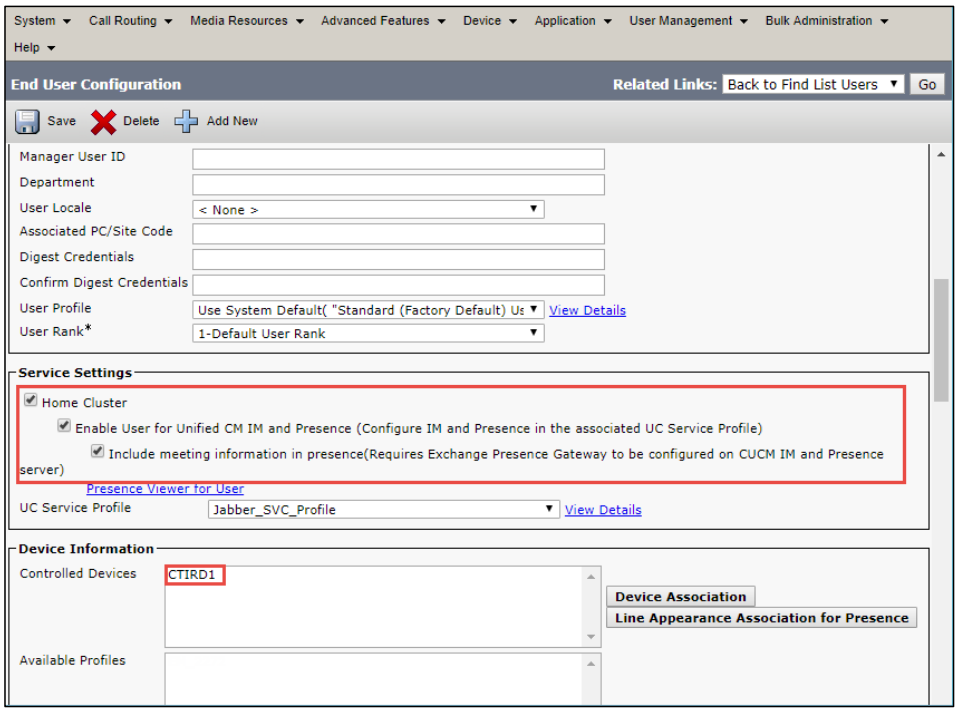
Check Home Cluster.

Click the Device Association

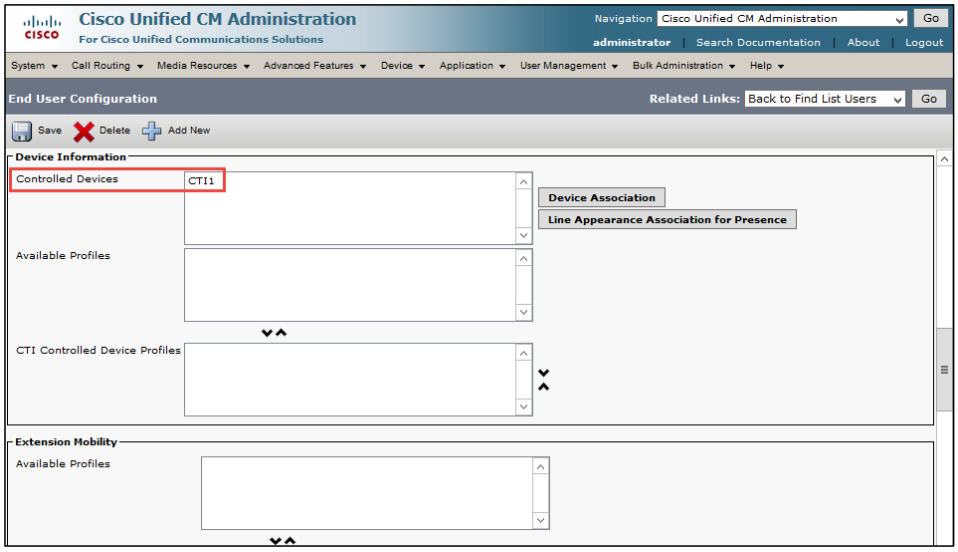
Select CTI1 from User Device Association screen



Cisco UCM end user Configuration (Continued)

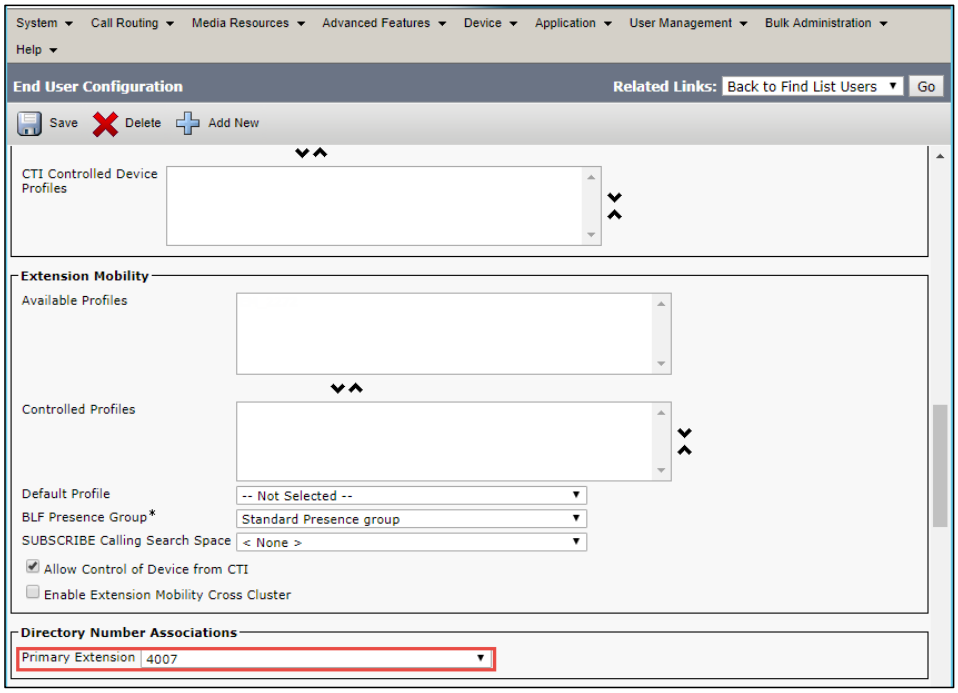


Cisco UCM end user Configuration (Continued)

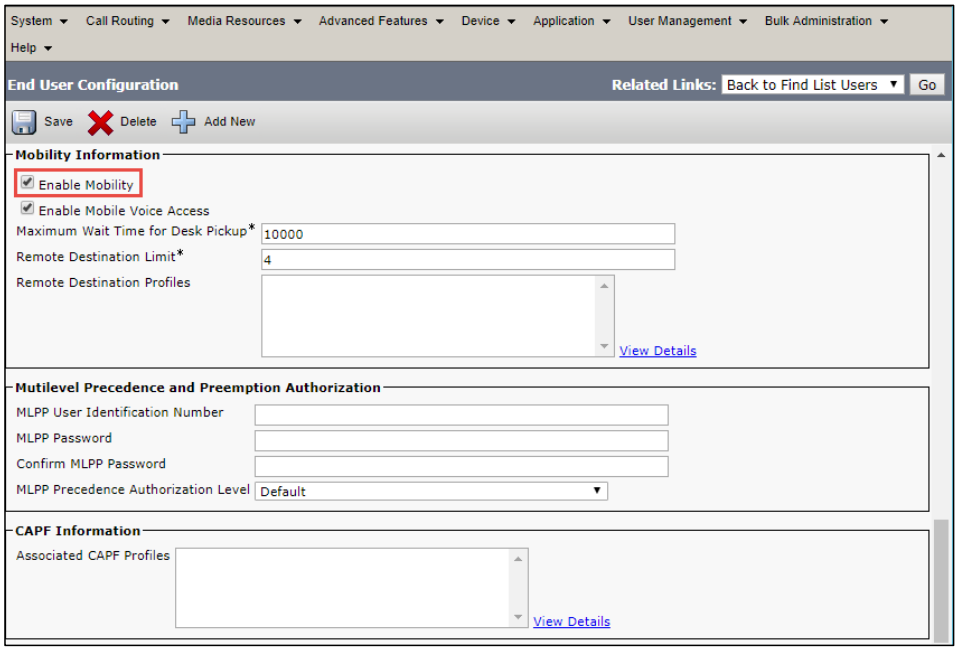


Cisco UCM end user Configuration (Continued)

Check Allow Control of Device from CTI
Select the Primary Extension for this user.4007 is used for this example.



Check Enable Mobility



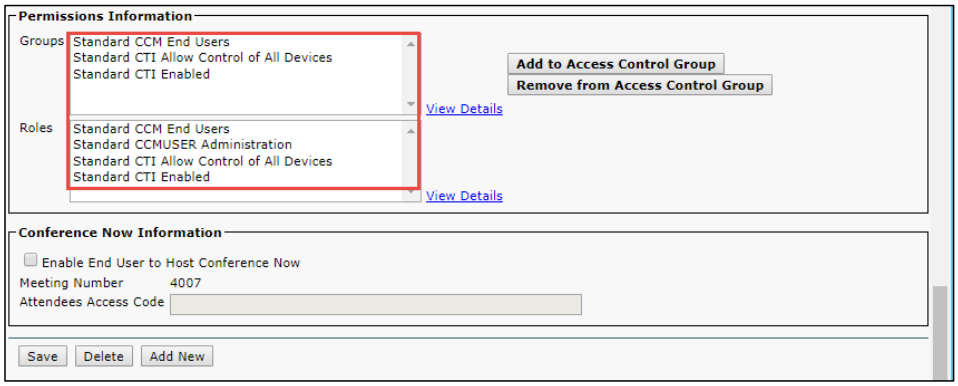
Add the following permissions for Standard Users:

–Standard CCM End-Users

–Standard CTI Enabled

–Standard CCMUSER Administration

Commented [RSM6]: Allow control of all devices in the sreenshot



Remote Destination Configuration

Navigation: Device→Remote Destination

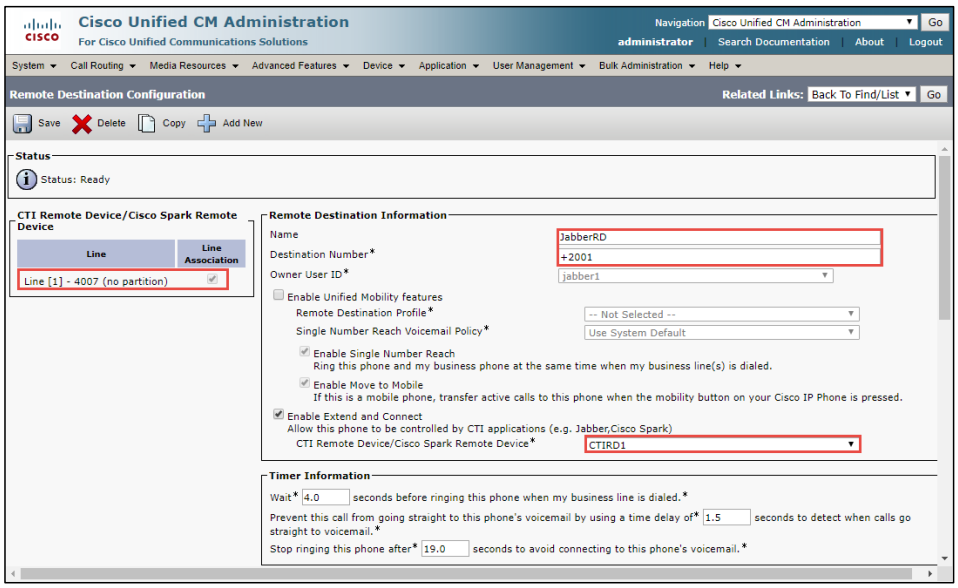
Add New

Set **name = Jabber RD** .This is used for the test

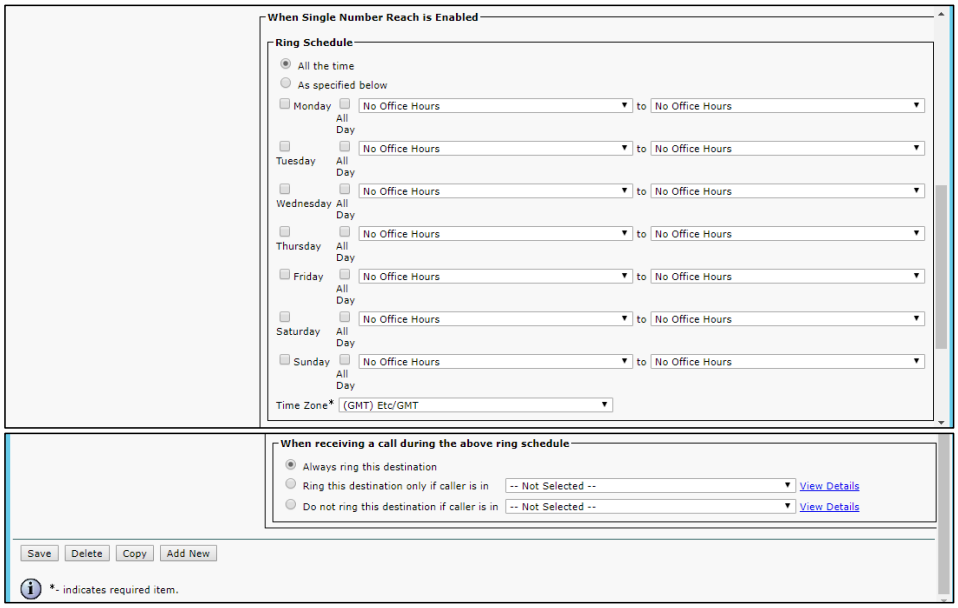
Set **Destination Number*= +2001**. This is used for the test. [8004 is a Skype for Business extension]

Check Enable Extend and Connect.

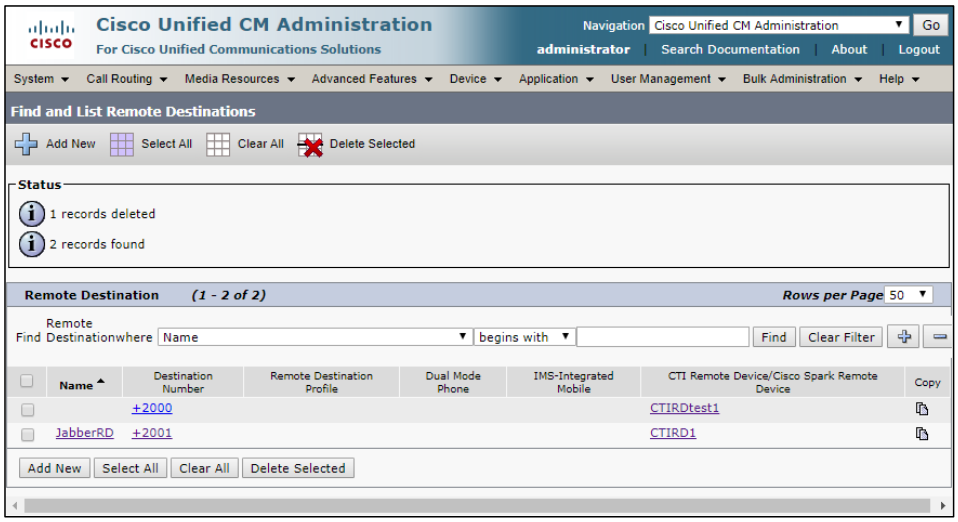
Set **CTI Remote Device = CTIRD1**



The CTI Remote Device configuration is updated with the remote destination:



Two Remote Destinations were configured for this test:



Cisco UCM CTI Remote Device Configuration

Navigation: Device→ Phone

Add New.

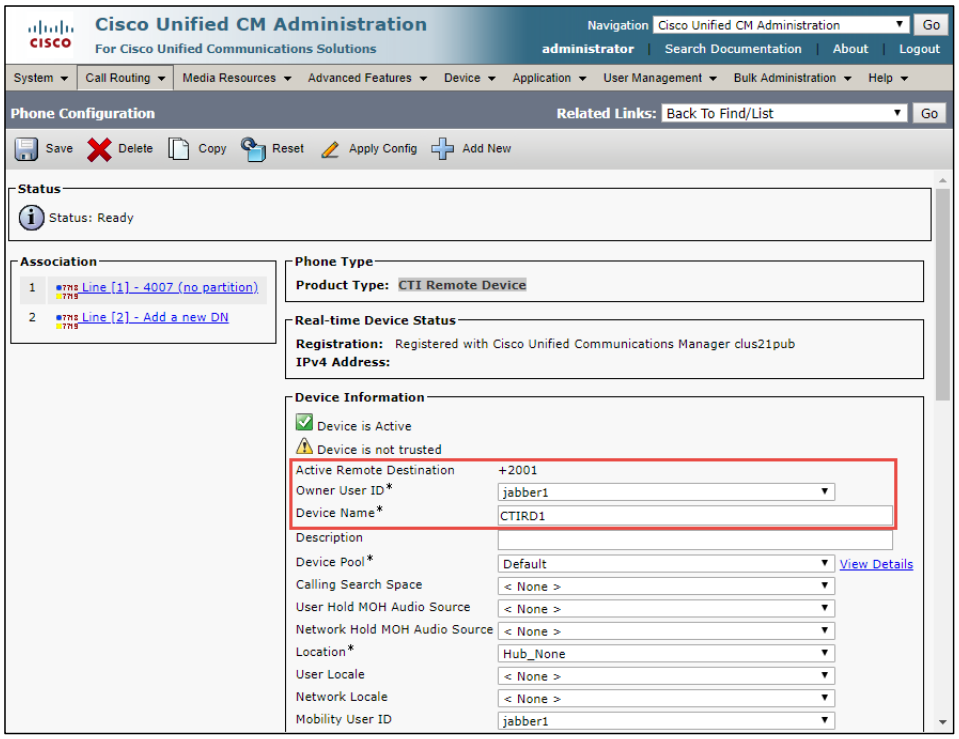
Select **Phone Type * = CTI Remote Device**

The CTI Remote Device type represents the user's remote device(s).

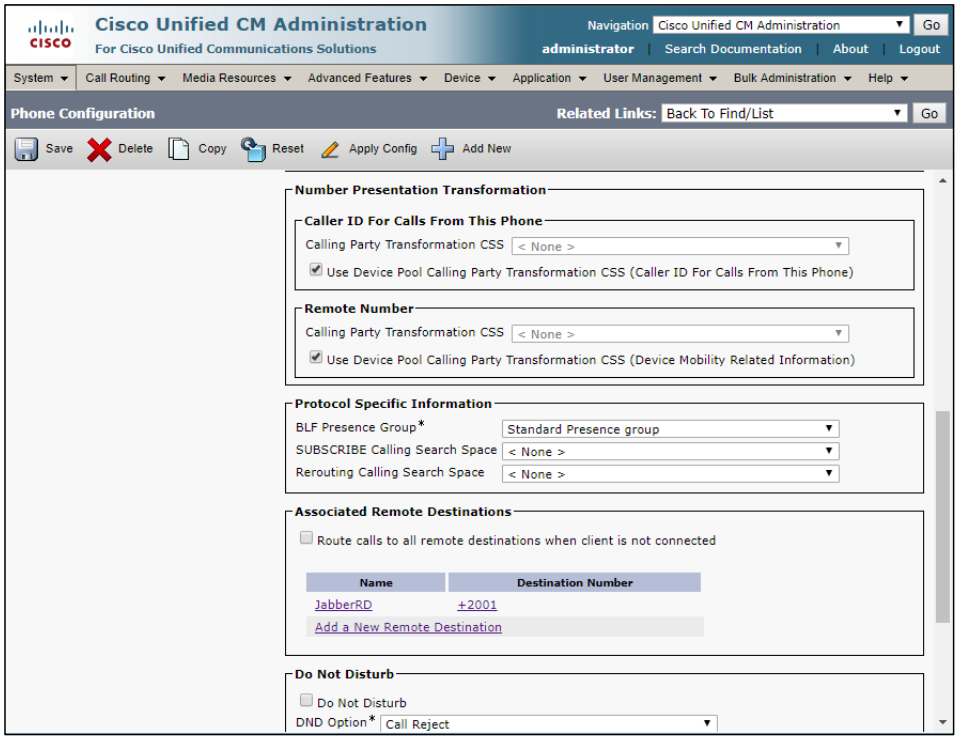
Select the desired Owner User ID. Jabber1 is used in this test.

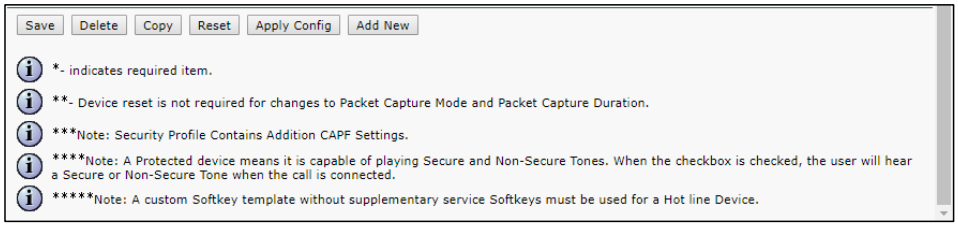
Set Device Pool: Default

Save.



Cisco UCM CTI Remote Device Configuration (Continued)





Add a DN to this device.

DN +2001 was configured for this test.

Cisco Unity Connection

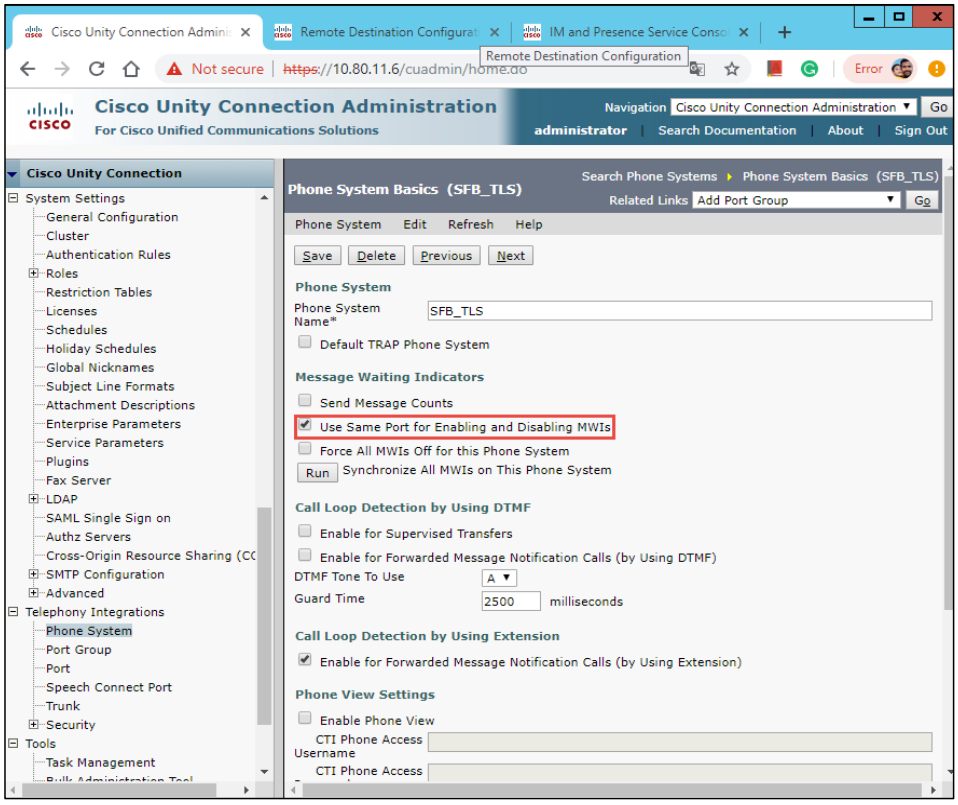
Telephony Integration – Add Phone System

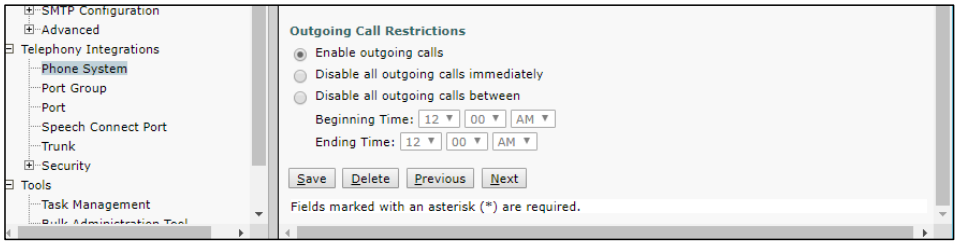
Navigation: Telephony Integrations → Phone system

Add New

Set Phone **System Name* = SFB_TLS**. This Name used for this test

Check Use Same Port for Enabling and Disabling MWIs





Telephony Integration – Add Port Group

Navigation: Telephony Integration → Port Group or from previous Screen, Related Links "Add Port Group" Go

Set **Phone System = SFB_TLS**

Set **Create From – Port group**

Set **Type = SIP**

Set Display Name* = SFB_TLS-1.This Name used for this example.

Set Ipv4 Address or Host Name = 10.80.10.2 [This is the Cisco UCM publisher IP]

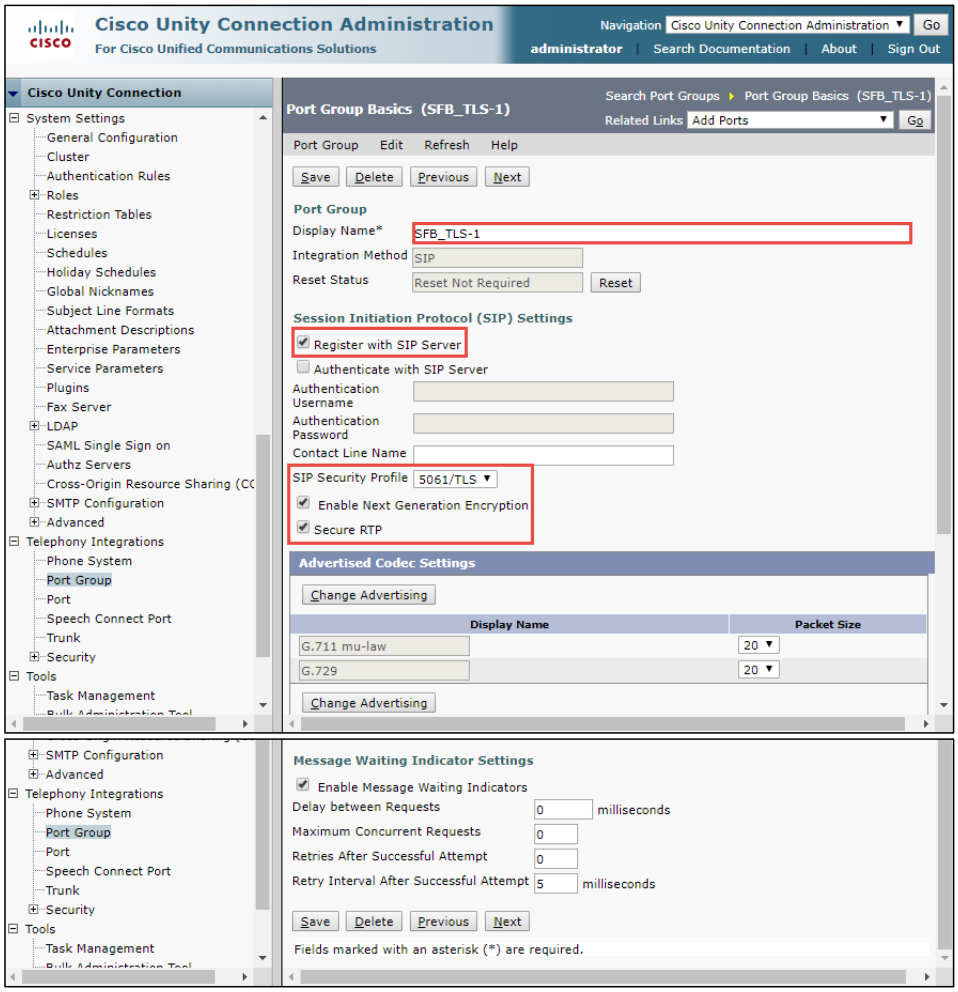
Check Register with SIP server

Set **Security Profile = 5061/TLS**

Set **Security Mode = Encrypted**

Check Secure RTP

Check Enable Message Waiting Indicators


Cisco Unity Connection Administration
For Cisco Unified Communications Solutions
Navigation Cisco Unity Connection Administration Go
administrator | Search Documentation | About | Sign Out
Cisco Unity Connection
System Settings
General Configuration
Cluster
Authentication Rules
Roles
Restriction Tables
Licenses
Schedules
Holiday Schedules
Global Nicknames
Subject Line Formats
Attachment Descriptions
Enterprise Parameters
Service Parameters
Plugins
Fax Server
LDAP
SAML Single Sign on
Authz Servers
Cross-Origin Resource Sharing (CC
SMTP Configuration
Advanced
Telephony Integrations
Phone System
Port Group
Port
Speech Connect Port
Trunk
Security
Tools
Task Management
Port Group Basics (SFB_TLS-1)
Search Port Groups ▸ Port Group Basics (SFB_TLS-1)
Related Links Add Ports Go
Port Group Edit Refresh Help
Save Delete Previous Next
Port Group
Display Name* SFB_TLS-1
Integration Method SIP
Reset Status Reset Not Required Reset
Session Initiation Protocol (SIP) Settings
Register with SIP Server
Authenticate with SIP Server
Authentication Username
Authentication Password
Contact Line Name
SIP Security Profile 5061/TLS
Enable Next Generation Encryption
Secure RTP
Advertised Codec Settings
Change Advertising
Display Name
Packet Size
G.711 mu-law 20
G.729 20
Change Advertising

SMTP Configuration
Advanced
Telephony Integrations
Phone System
Port Group
Port
Speech Connect Port
Trunk
Security
Tools
Task Management
Message Waiting Indicator Settings
Enable Message Waiting Indicators
Delay between Requests 0 milliseconds
Maximum Concurrent Requests 0
Retries After Successful Attempt 0
Retry Interval After Successful Attempt 5 milliseconds
Save Delete Previous Next
Fields marked with an asterisk (*) are required.

Click Save.

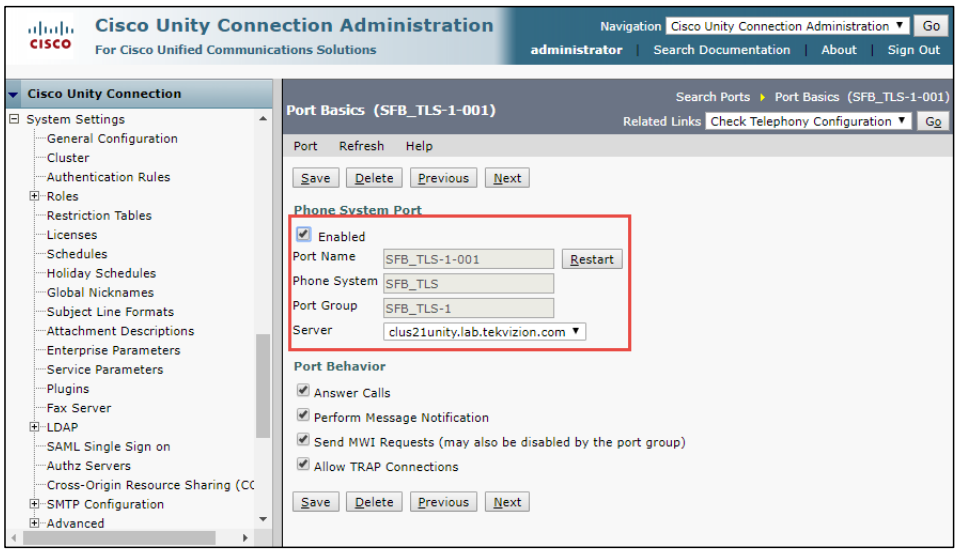
Telephony Integration – Add Ports

Set **Number of Ports = 10**

Set **Phone System = SFB_TLS**

Set **Port Group = SFB_TLS-1**

Set **Server = clus21unity.lab.tekvizion.com** (which is the FQDN of Cisco Unity Server)



User Configuration

Navigation: Cisco Unity Connection → Users → Users

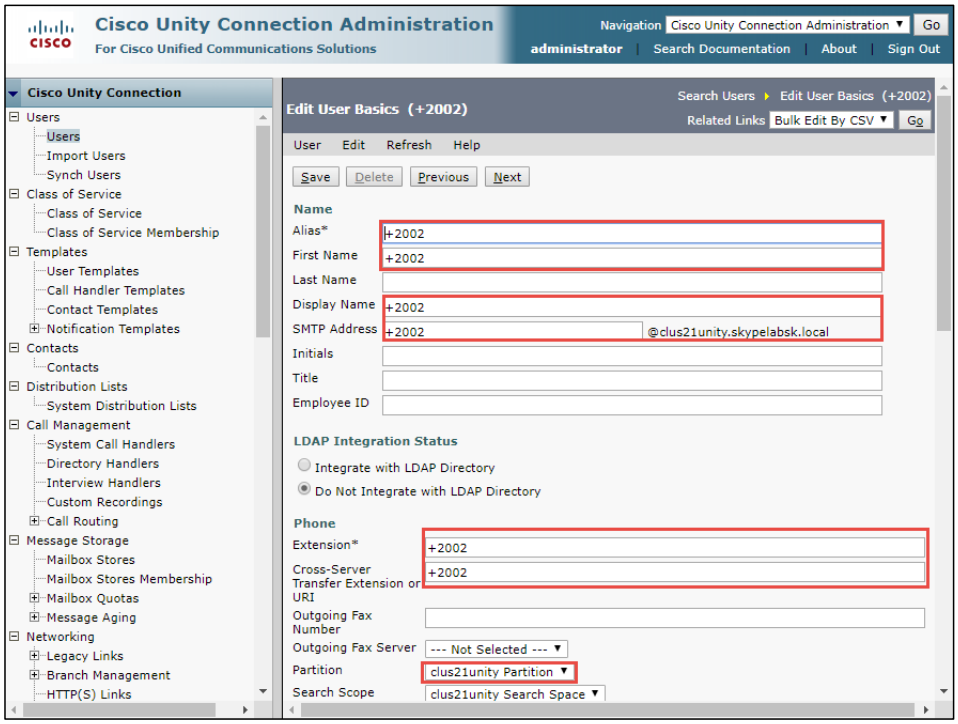
Set **Alias*= +2002** (This is used for the test)

Set **First Name = +2002** (This is used to identify the User)

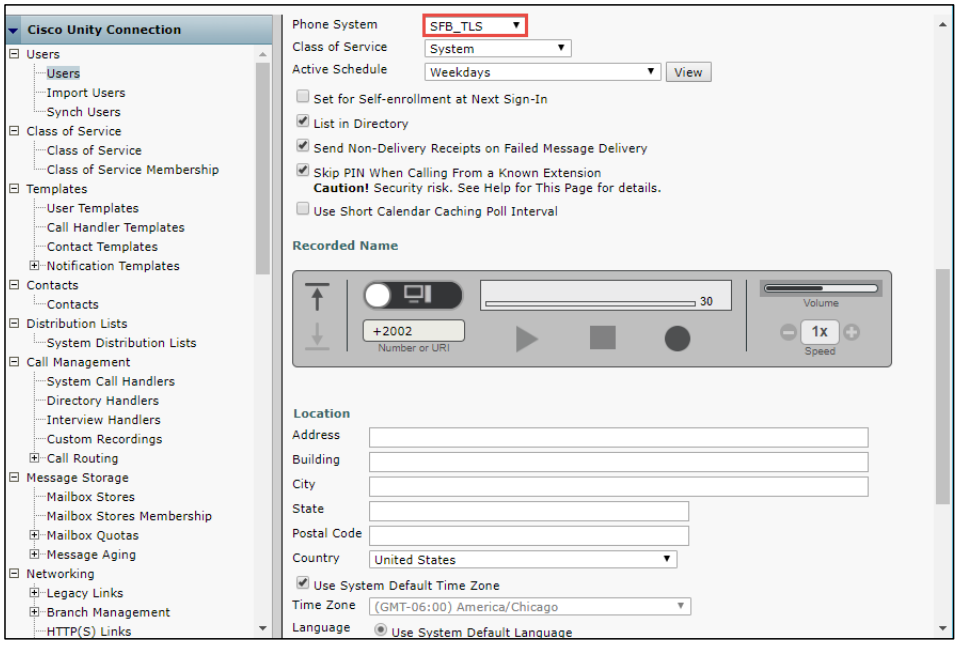
Set **Extension* = +2002** (This is user's extension number)

Set **Partition = clus21initypartition**

Set **Phone System = SFB_TLS**

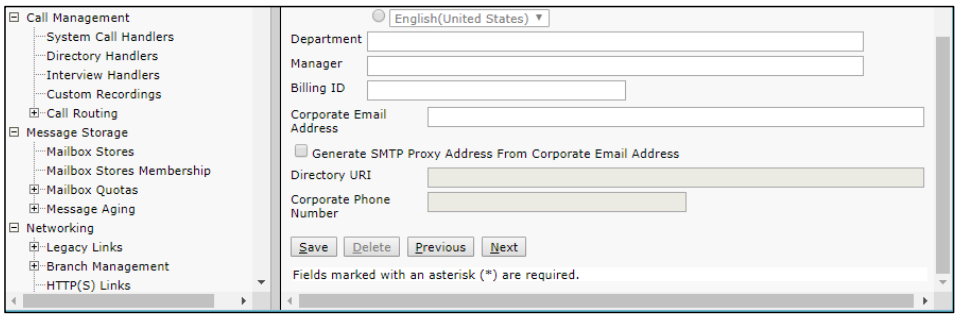

Cisco Unity Connection Administration
For Cisco Unified Communications Solutions
Navigation Cisco Unity Connection Administration Go
administrator | Search Documentation | About | Sign Out
Cisco Unity Connection
Users
Users
Import Users
Synch Users
Class of Service
Class of Service
Class of Service Membership
Templates
User Templates
Call Handler Templates
Contact Templates
Notification Templates
Contacts
Contacts
Distribution Lists
System Distribution Lists
Call Management
System Call Handlers
Directory Handlers
Interview Handlers
Custom Recordings
Call Routing
Message Storage
Mailbox Stores
Mailbox Stores Membership
Mailbox Quotas
Message Aging
Networking
Legacy Links
Branch Management
HTTP(S) Links
Edit User Basics (+2002)
Search Users Edit User Basics (+2002)
Related Links Bulk Edit By CSV Go
User Edit Refresh Help
Save Delete Previous Next
Name
Alias* +2002
First Name +2002
Last Name
Display Name +2002
SMTP Address +2002 @clus21unity.skypelabsk.local
Initials
Title
Employee ID
LDAP Integration Status
Integrate with LDAP Directory
Do Not Integrate with LDAP Directory
Phone
Extension* +2002
Cross-Server Transfer Extension or URI +2002
Outgoing Fax Number
Outgoing Fax Server --- Not Selected ---
Partition clus21unity Partition
Search Scope clus21unity Search Space

Cisco Unity Connection User Configuration (Continued)



Cisco Unity Connection User Configuration (Continued)

All values are default.



Similarly, create a user that has a Cisco extension.

Acronyms

Acronym	Definition
CCNR	Call Completion on No Reply
CFB	Call Forwarding on Busy
CFNA	Call Forwarding No Answer
CFU	Call Forwarding Unconditional
Cisco UCM	Cisco Unified Communications Manager
CLIP	Calling Line (Number) Identification Presentation
CLIR	Calling Line (Number) Identification Restriction
CNIP	Calling Name Identification Presentation
CNIR	Calling Name Identification Restriction
COLP	Connected Line (Number) Identification Presentation
COLR	Connected Line (Number) Identification Restriction
CONP	Connected Name Identification Presentation
CONR	Connected Name Identification Restriction
CT	Call Transfer
CUP	Cisco Unified Presence
DNS	Domain Name Server
EXT	Extension
FQDN	Fully Qualified Domain Name
MRGL	Media Resource Group List
MTP	Media Termination Point
MWI	Message Waiting Indicator
PBX	Private Branch Exchange
PSTN	Public Switched Telephone Network
RTP	Real Time Protocol
SCCP	Skinny Client Control Protocol
SFB	Skype for Business
SIP	Session Initiated Protocol
UDP	Uniform Dial Plan
VM	Voice Mail