Application Note

Spectrum Enterprise SIP Trunk: Connecting Cisco Unified Communication Manager Express (CME) 11.5 (IOS-XE 16.3.1) using SIP

November 19, 2018

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Introduction

Service Providers today, such as Spectrum Enterprise, 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.

Spectrum Enterprise 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.

- This application note describes how to configure a Cisco Unified Communications Manager Express (Cisco Unified CME) 11.5 – IOS 16.03.01 with connectivity to Spectrum Enterprise SIP Trunk service. The application note also covers support and configuration example Cisco Unity Connection (CUC) 12.0 messaging integrated into the Cisco Unified Communications Manager Express. The deployment model covered in this application note is Customer Premises Equipment (Cisco Unified CME/CUC) to PSTN (Spectrum Enterprise). Spectrum Enterprise provides inbound and outbound call service.
- Testing was performed in accordance to Cisco generic SIP Trunking test methodology and among features verified were – basic calls, DTMF transport, Music on Hold (MOH), unattended and attended transfers, call forward, conferences.
- Consult your Cisco representative for the correct IOS image and for the specific application and Device Unit License and Feature License requirements for all your Cisco Unified CME.

Network Topology

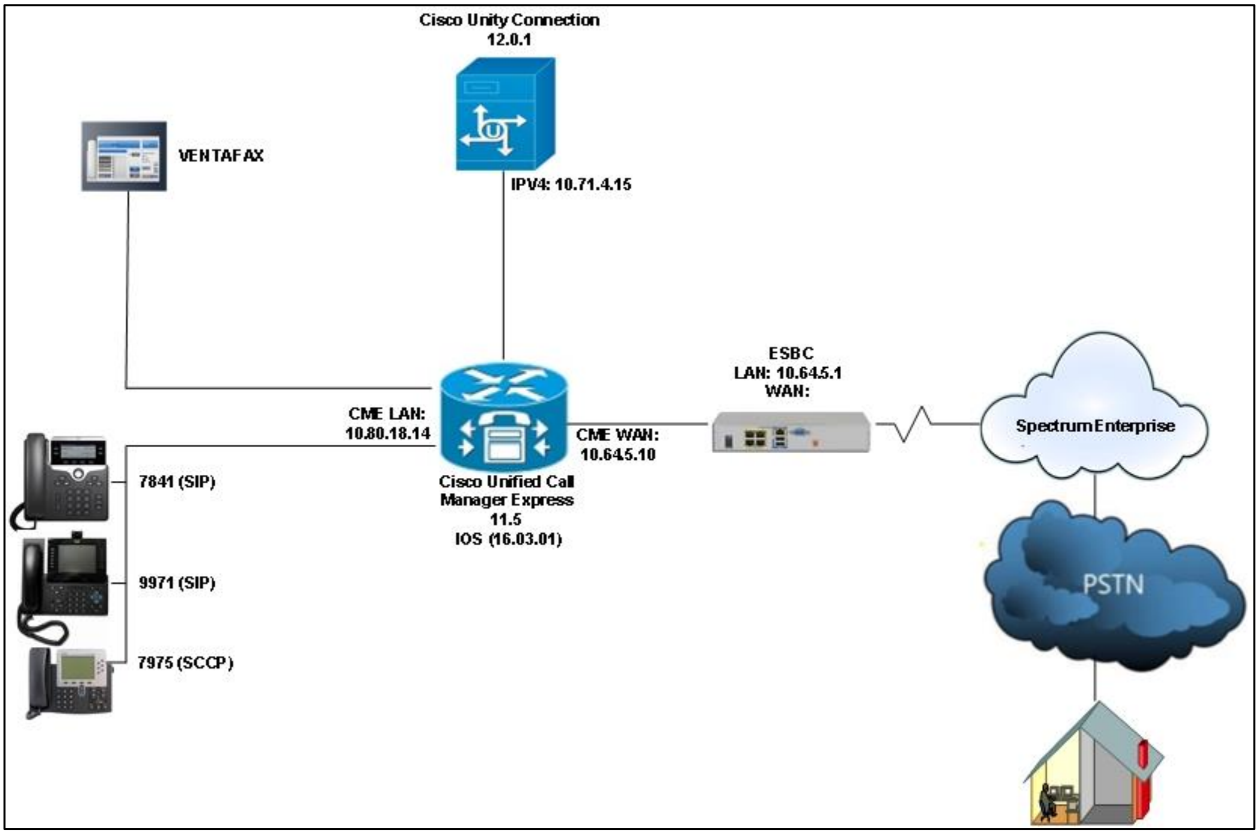

Figure 1 Network Topology

System Components

Hardware Requirements

- This solution was tested with Cisco ISR4331.
- Cisco IP Phones. This solution was tested with 7841, 7975, 9971 phones, but any Cisco IP Phone model supporting RFC2833 can be used.
- Cisco ISR4331/K9 (1RU) Process with 3700953K/6147K bytes of memory. Processor board ID FLM2141V251.
- 1 Virtual Ethernet interface and 3 Gigabit Ethernet interface.
- Cisco Unity Connection – VMware – 1 vCPU: Intel(R) Xeon(R) CPU X5675 @ 3.07GHz.
- HDD 160 GB, Memory 4096 Mbytes RAM.

Software Requirements

- Cisco IOS gateway running Cisco Unified CME Version 16.03.01, RELEASE SOFTWARE (fc3). This solution was tested with Cisco IOS image: “bootflash:isr4300-universalk9.16.03.01.SPA.bin”.
- This solution was tested with Cisco Unity Connection version (Version 12.0.1.21900-14)

Features

Features Supported

- Basic Call using G711ulaw.
- Calling Party Number Presentation and Restriction.
- Calling Name.
- Call Transfer.
- Conference.
- Call Hold and Resume.
- Call Forward All, Busy and No Answer.
- Fax over G.711.
- Fax over T.38 (See Note)
- Incoming DID Translation and Routing.
- Outbound calls and Inbound calls.
- Voicemail.
- Auto-attendant.

Features Not Supported

- Cisco IP phones used in this test do not support Blind Transfer, only Semi-attendant and Attendant transfer were tested.
- Spectrum Enterprise does not support G729 codec at the moment.
- Dual codecs (G711 & G729) are not supported by Spectrum Enterprise at the moment.

Note:

- Tested Fax Speed 14.4 reliably over the trunk for Both T.38 and G711ulaw passthrough, Spectrum Enterprise SIP Trunking supports V.34 specifications for SG3. However fax speeds higher than 14.4 is not guaranteed over the trunk and not tested.
- Spectrum Enterprise SIP Trunking will accept requests for t.38 fax if sent by the PBX, however Spectrum Enterprise SIP Trunking will not initiate a request for t.38. (For both INBOUND and OUTBOUND Fax scenario’s PBX must initiate t.38 request to complete the fax)

Configuration

Configuring CUCME

Cisco IOS-XE Version

SPECTRUM ENTERPRISECME#show version
Cisco IOS XE Software, Version 16.03.01
Cisco IOS Software [Denali], ISR Software (X86_64_LINUX_IOSD-UNIVERSALK9-M), Version 16.3.1, RELEASE SOFTWARE (fc3)
Technical Support: http://www.cisco.com/techsupport
Copyright (c) 1986-2018 by Cisco Systems, Inc.
Compiled Tue 02-Aug-16 18:49 by mcpre

Cisco IOS-XE software, Copyright (c) 2005-2018 by cisco Systems, Inc.
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with ABSOLUTELY NO WARRANTY. You can redistribute and/or modify such
GPL code under the terms of GPL Version 2.0. For more details, see the
documentation or "License Notice" file accompanying the IOS-XE software,
or the applicable URL provided on the flyer accompanying the IOS-XE
software.

ROM: IOS-XE ROMMON

SPECTRUM ENTERPRISECME uptime is 6 days, 21 hours, 48 minutes
Uptime for this control processor is 6 days, 21 hours, 50 minutes
System returned to ROM by reload at 09:35:50 cdt Thu Oct 4 2018
System restarted at 09:40:38 cdt Thu Oct 4 2018
System image file is "bootflash:isr4300-universalk9.16.03.01.SPA.bin"
Last reload reason: Reload Command

This product contains cryptographic features and is subject to United
States and local country laws governing import, export, transfer and
use. Delivery of Cisco cryptographic products does not imply
third-party authority to import, export, distribute or use encryption.
Importers, exporters, distributors and users are responsible for
compliance with U.S. and local country laws. By using this product you
agree to comply with applicable laws and regulations. If you are unable

to comply with U.S. and local laws, return this product immediately.

A summary of U.S. laws governing Cisco cryptographic products may be found at:
http://www.cisco.com/wwl/export/crypto/tool/stqrg.html

If you require further assistance please contact us by sending email to
export@cisco.com.

```
Suite License Information for Module:'esg'

--------------------------------------------------------------------------------
Suite                 Suite Current         Type           Suite Next reboot
--------------------------------------------------------------------------------
FoundationSuiteK9     None                  None           None
securityk9
appxk9

AdvUCSuiteK9          None                  None           None
uck9
cme-srst
cube

Technology Package License Information:

-----------------------------------------------------------------
Technology    Technology-package           Technology-package
              Current       Type           Next reboot
------------------------------------------------------------------
appxk9           None          None           None
uck9             uck9          Permanent      uck9
securityk9       None          None           None
ipbase           ipbasek9      Permanent      ipbasek9

cisco ISR4331/K9 (1RU) processor with 3700953K/6147K bytes of memory.
Processor board ID FLM2141V251
1 Virtual Ethernet interface
3 Gigabit Ethernet interfaces
32768K bytes of non-volatile configuration memory.
8388608K bytes of physical memory.
7057407K bytes of flash memory at bootflash:.
0K bytes of  at webui:.
```

Configuration register is 0x2102

SPECTRUM ENTERPRISECME#

SPECTRUMENTERPRISECME#**show running-config**
Building configuration...

```
Current configuration : 9627 bytes
!
version 16.3
service timestamps debug datetime msec
service timestamps log datetime msec
no platform punt-keepalive disable-kernel-core
!
hostname SPECTRUMENTERPRISECME
!
boot-start-marker
boot system flash bootflash:isr4300-universalk9.16.03.01.SPA.bin[1]
boot-end-marker
!
!
vrf definition Mgmt-intf
 !
 address-family ipv4
 exit-address-family
 !
 address-family ipv6
 exit-address-family
!
no aaa new-model
clock timezone cdt -6 0
clock summer-time cdt recurring
!
subscriber templating
!
multilink bundle-name authenticated
!
```

[1] If the CME has multiple IOS Bin files, this command specifies the Path of bootfile with which the CME has to be loaded.

```
voice service voip[2]
 ip address trusted list[3]
 no ip address trusted authenticate[4]
 allow-connections h323 to h323[5]
 allow-connections h323 to sip[6]
 allow-connections sip to h323[7]
 allow-connections sip to sip[8]
 no supplementary-service sip moved-temporarily[9]
 no supplementary-service sip refer[10]
no supplementary-service sip handle-replaces[11]
 redirect ip2ip[12]
 fax protocol t38 version 0 ls-redundancy 0 hs-redundancy 0 fallback none[13]
 sip
  session refresh[14]
  registrar server[15]
  privacy pstn[16]
!
voice class codec 1[17]
 codec preference 1 g711ulaw
 codec preference 2 g729r8
!
voice register global[18]
```

---

[2] This command enables the CUBE Application on this Device

[3] Use this command to manually add unique and multiple IP addresses to a list of trusted IP addresses

[4] To enable ip address trusted authentication for incoming VoIP (H.323/SIP) calls, use the **ip address trusted authenticate** command in voice service voip mode. To **disable** ip address trusted authentication, use the **no** form of this command.

[5] This command specifies Cisco Unified CME to perform basic h323 to h323 Voice communication

[6] This command specifies Cisco Unified CME to perform basic h323 to SIP Voice communication

[7] This command specifies Cisco Unified CME to perform basic SIP to h323 Voice communication

[8] This command enables Cisco Unified CME to perform basic SIP to SIP voice communication

[9] This command disables the use of SIP moved-temporarily response

[10] This command disables the use of SIP refer method for call transfer

[11] This command disables the replaces header handling for call transfer

[12] This command enables SIP phone calls to SIP phone calls redirect globally

[13] This command enables T.38 fax at global level, meaning all VoIP dial-peers not configured for specific fax protocol will use this setting. G711 fax protocol may be configured under appropriate dial-peers

[14] Use the SIP session refresh command to send the session refresh request

[15] Enable Local SIP Registrar which is required for SIP phones in Cisco Unified CME

[16] This command enables the use of Privacy ID globally for outbound anonymous calls. Prerequisite Caller ID block defined for a specific Phone or a specific dial-peer

[17] This command enables multiple codec support and performs codec filtering required for correct interoperability between Spectrum Enterprise SIP trunk and Cisco Unified CME

[18] Use this command to set provisioning parameters for all supported SIP phones in a Cisco Unified CME system

```
 mode cme[19]
 source-address 10.80.18.14 port 2000[20]
 max-dn 20[21]
 max-pool 10[22]
 load 9971 sip9971.9-2-2SR1-9[23]
 load 7841 sip78xx.12-0-1-11
 authenticate register[24]
 mwi stutter[25]
 tftp-path flash:[26]
 file text[27]
 create profile sync 0191072658043229[28]
 auto-register[29]
 !
!
voice register dn  1[30]
 number 8980[31]
 name SPECTRUMENTERPRISE Phone1[32]
 mwi[33]
!
voice register dn  2
 number 8981
 name SPECTRUMENTERPRISE Phone2
 mwi
!
```

[19] Enables the mode for configuring SIP IP phones in Cisco Unified CME
[20] This is the source address for SIP phone registration
[21] Configuration for maximum extensions
[22] Configuration for maximum phones
[23] Specify phone loads for each phone type
[24] All incoming registration requests are challenged and authenticated
[25] To generate a stutter tone for message-waiting indication (MWI) in a Cisco Unified CME system
[26] This command defines the location for configuration files that are generated by using the create profile command.
[27] Use this command to generate an ASCII text fils of the configuration profile for Cisco Unified IP Phone 7905s and 7905Gs, Cisco Unified IP Phone 7912s and 7912Gs, Cisco ATA-186s, or Cisco ATA-188s
[28] This command generates the configuration files used for provisioning SIP phones and writes the files to the location specified with the tftp-path command.
[29] This command is enabled by default and allows automatic registration of SIP phones on Unified CME, provided the administrator configures the password and DN range using the relevant sub-mode CLI options.
[30] Use this command to create directory numbers for SIP IP phones directly connected in Cisco Unified CME
[31] In **voice register dn** configuration mode, you assign an extension number by using the **number** command
[32] In **voice register dn** configuration mode, you assign a name to appear in the local directory by using the **name** command
[33] This command enables a particular extension on a SIP IP phone to receive MWI notification

```
voice register pool  1[34]
 busy-trigger-per-button 2[35]
 id mac B838.6116.3A2C[36]
 type 7841[37]
 number 1 dn 1[38]
 dtmf-relay rtp-nte[39]
 voice-class codec 1
 username 8980 password 1234[40]
!
voice register pool  2
 id mac C07B.BCA1.B811
 type 9971
 number 1 dn 2
 voice-class codec 1
 username 8981 password 1234
!
voice moh-group 1[41]
 moh enable-g711 "flash:music-on-hold.au"[42]
!
voice translation-rule 1[43]
 rule 1 /\(^8...\)/ /469573\1/
!
voice translation-rule 2[44]
 rule 1 /469573\(....\)/ /\1/
```

[34] Use this command to set phone-specific parameters for SIP phones in a Cisco Unified CME system

[35] This command limits the calls to an octo-line on the specified phone by triggering Call Forward Busy or a busy tone, after the number of active calls, incoming and outgoing

[36] To explicitly identify a locally available individual Cisco SIP IP phone using Phone's mac

[37] To define a phone type for a SIP phone, use the type command in voice register pool configuration mode

[38] To indicate the E.164 phone numbers that the registrar permits to handle the Register message from a Cisco Unified SIP IP phone, use the number command in voice register pool configuration mode.

[39] To specify the list of DTMF relay methods that can be used to relay dual-tone multifrequency (DTMF) audio tones between Session Initiation Protocol (SIP) endpoints, use the dtmf-relay command in voice register pool configuration mode

[40] To assign an authentication credential to a phone user so that the SIP phone can register in Cisco CME, use the username command in voice register pool configuration mode

[41] This command enters the voice-moh-group configuration mode for configuring music on hold (MOH) group parameters for SCCP IP phones in Cisco Unified CME

[42] The moh command allows you to specify the .au and .wav format music files that are played to callers who have been put on hold. MOH works only for G.711 calls and on-net VoIP and PSTN calls. For all other calls, callers hear a periodic tone

[43] This translation rule/profile is used to modify the calling number(used for this current test setup, might change per your environment)

[44] This translation rule/profile is used to modify the called number(used for this current test setup, might change per your environment)

```
!
voice translation-profile To-10Dig
translate calling 1
translate redirect-called 1
!
voice translation-profile to_ext
translate called 2
!
voice-card 0/4
 dsp services dspfarm
 no watchdog
!
license udi pid ISR4331/K9 sn FDO21781F1G
!
diagnostic bootup level minimal
spanning-tree extend system-id
!
!
redundancy
 mode none
!
!
vlan internal allocation policy ascending
!
ip tftp source-interface GigabitEthernet0/0/1
!
!interface GigabitEthernet0/0/0
 ip address 10.64.5.10 255.255.0.0
 negotiation auto
!
interface GigabitEthernet0/0/1
 ip address 10.80.18.14 255.255.255.0
 negotiation auto
!
interface GigabitEthernet0/0/2
 no ip address
 negotiation auto
!
interface Service-Engine0/4/0
!
interface GigabitEthernet0
 vrf forwarding Mgmt-intf
```

```
 no ip address
 negotiation auto
!
interface Vlan1
 no ip address
 shutdown
!
ip forward-protocol nd
ip http server
ip http authentication local
ip http secure-server
ip http client source-interface GigabitEthernet0/0/1
ip route 0.0.0.0 0.0.0.0 10.64.5.1
ip route 10.10.10.0 255.255.255.0 10.80.18.1[45]
ip route 10.71.4.0 255.255.255.0 10.80.18.1
ip route 172.16.24.0 255.255.248.0 10.80.18.1
!
!
tftp-server flash:kern2.78xx.12-0-1-11.sbn
tftp-server flash:kern78xx.12-0-1-11.sbn
tftp-server flash:rootfs2.78xx.12-0-1-11.sbn
tftp-server flash:rootfs78xx.12-0-1-11.sbn
tftp-server flash:sboot2.78xx.12-0-1-11.sbn
tftp-server flash:sboot78xx.12-0-1-11.sbn
tftp-server flash:sip78xx.12-0-1-11.loads
tftp-server flash:apps75.9-4-2ES26.sbn
tftp-server flash:cnu75.9-4-2ES26.sbn
tftp-server flash:cvm75sccp.9-4-2ES26.sbn
tftp-server flash:dsp75.9-4-2ES26.sbn
tftp-server flash:jar75sccp.9-4-2ES26.sbn
tftp-server flash:SCCP75.9-4-2SR3-1S.loads
tftp-server flash:term75.default.loads
tftp-server flash:music-on-hold.au
tftp-server flash:apps42.9-4-2ES26.sbn
tftp-server flash:cnu42.9-4-2ES26.sbn
tftp-server flash:cvm42sccp.9-4-2ES26.sbn
tftp-server flash:dsp42.9-4-2ES26.sbn
tftp-server flash:jar42sccp.9-4-2ES26.sbn
tftp-server flash:SCCP42.9-4-2SR3-1S.loads
tftp-server flash:term42.default.loads
```

[45] Static route used to define gateway IP to respond phone registration requests in a particular subnet. Might differ per your environment policies.

tftp-server flash:term62.default.loads
!
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
!
!
ccm-manager music-on-hold
!
!
telephony-service[46]
 conference transfer-pattern[47]
 max-ephones 20[48]
 max-dn 20[49]
 ip source-address 10.80.18.14 port 2000[50]
 load 7942 SCCP42.9-4-2SR3-1S[51]
 load 7975 SCCP75.9-4-2SR3-1S
 voicemail 7777[52]
 max-conferences 8 gain -6[53]
 call-forward pattern .T[54]

---

[46] This command enters the telephony-service configuration mode for configuring system wide parameters for SCCP IP phones in Cisco Unified CME

[47] There is no check for the conference numbers for call conferencing. Use this command to apply transfer-pattern for call conferencing

[48] The **max-ephones** command limits the number of Cisco IP phones supported on the Cisco Unified CME system

[49] The **max-dn** command limits the number of extensions (ephone-dns) available in a Cisco Unified CME system

[50] This command enables a router to receive messages from Cisco Unified IP phones through the specified IP address and port

[51] This command updates the Cisco Unified CME configuration file for the specified type of Cisco Unified IP phone to add the name of the firmware file to be loaded by a particular phone type

[52] This command configures the telephone number that is speed-dialed when the Messages button on a Cisco IP phone is pressed. The same telephone number is configured for voice messaging for all Cisco IP phones connected to the Cisco Unified CME system

[53] To set the maximum number of three-party conferences that are supported simultaneously by the Cisco Unified CME) system

[54] To specify a pattern for calling - party numbers that are able to support the ITU-T H.450.3 standard for call forwarding, **.T** is not recommended it is shown here as an example only. To avoid the interdigit timeout, a matching transfer pattern should be used

```
 moh enable-g711 "flash:music-on-hold.au"[55]
 transfer-system full-consult[56]
 transfer-pattern .T[57]
 create cnf-files version-stamp 7960 Oct 09 2018 07:21:38[58]
!
!
dial-peer voice 100 voip[59]
 description Outbound Call to SPECTRUMENTERPRISE
 translation-profile outgoing To-10Dig[60]
 destination-pattern .T[61]
 session protocol sipv2
 session target ipv4:10.64.5.1:5060[62]
 session transport udp
 voice-class codec 1[63]
 dtmf-relay rtp-nte
 fax protocol t38 version 0 ls-redundancy 0 hs-redundancy 0 fallback none
 no vad
!
dial-peer voice 101 voip[64]
 description inbound from SPECTRUMENTERPRISE
 translation-profile incoming to_ext[65]
 session protocol sipv2
 session transport udp
```

---

[55] This command enables MOH from .au and .wav format music files. MOH is played for G.711 callers and on-net VoIP and PSTN callers who are on hold in a Cisco CME system. Local callers within a Cisco CME system hear a repeating tone while they are on hold

[56] To specify the call transfer method to be used by Cisco Unified IP phones in Cisco Unified CME system, **full-consult** transfers calls using H.450.2 with consultation using a second phone line, if available. The calls fall back to full-blind if a second line is not available

[57] To allow transfer of telephone calls from Cisco IP phones to phones other than Cisco IP phones, .T is not recommended it is shown here as an example only. To avoid the interdigit timeout, a matching transfer pattern should be used

[58] Use this command to generate the XML configuration files used for provisioning SCCP phones and write the files to the location specified with the cnf-file location command

[59] Dial peer for Outbound calls towards Spectrum enterprise SIP trunk.

[60] Translate the calling Number from 4 digit extension to 10 digit DN for Outbound calls towards Spectrum enterprise SIP trunk.

[61] This command defines the matching pattern of the called number to route the call using this dial-peer. .T is not recommended it is shown here as an example only. To avoid the interdigit timeout, a matching transfer pattern should be used

[62] This command sets the SIP server target for outgoing SIP calls (towards Spectrum enterprise)

[63] This command assigns the voice class codec setting to this dial-peer

[64] Dial peer for Incoming calls from Spectrum enterprise SIP trunk.

[65] Translate the calling Number from 10 digit DN to 4 digit extension for Incoming calls from Spectrum enterprise SIP trunk

```
 incoming called-number 469573....[66]
 voice-class codec 1
 dtmf-relay rtp-nte
 fax protocol t38 version 0 ls-redundancy 0 hs-redundancy 0 fallback none
 no vad
!
dial-peer voice 102 voip[67]
 description towards CUC
 destination-pattern 7777[68]
 session protocol sipv2
 session target ipv4:10.71.4.15:5060[69]
 session transport udp
 voice-class codec 1
 dtmf-relay rtp-nte
 no vad
!
dial-peer voice 201 voip[70]
 description Trunk to ATA
 destination-pattern 8985
 session protocol sipv2
 session target ipv4:172.16.31.87:5060
 session transport udp
 voice-class codec 1
 fax protocol t38 version 0 ls-redundancy 0 hs-redundancy 0 fallback none
 no vad
!
sip-ua
 mwi-server ipv4:10.71.4.15 expires 3600 port 5060 transport udp unsolicited[71]
!
!
ephone-template  1[72]
 conference admin
 softkeys hold  Resume Join Newcall Select
 softkeys idle  Redial Cfwdall Dnd ConfList Gpickup HLog Join Login Newcall Pickup RmLstC
 softkeys seized  Redial Endcall Meetme Cfwdall CWOff Callback Gpickup HLog Pickup
```

[66] This command defines the matching pattern of the called number to route the call using this dial-peer
[67] Dial peer used to route the calls to Cisco unity connection server for Voicemail access
[68] This command defines the matching pattern of voicemail pilot number
[69] This command sets the Cisco unity connection server target for voicemail calls
[70] Dial peer for outbound fax towards Cisco ATA device on this setup.
[71] Use this command to configure the IP address of an external SIP MWI server
[72] To create an ephone template to configure a set of phone features and to enter ephone-template configuration mode, use the ephone-template command in global configuration mode

```
 softkeys connected  Endcall Trnsfer Hold Confrn Acct ConfList Flash HLog Join Park RmLstC Select
!
ephone-dn  1[73]
 number 8982[74]
 label 8982 for SPECTRUMENTERPRISE[75]
 name SPECTRUMENTERPRISE Phone[76]
 call-forward busy 4695738980
 call-forward noan 7777 timeout 20
 huntstop channel[77]
 mwi sip
 moh-group 1
!
ephone-dn  2
 number 8984
 label 8984 for SPECTRUMENTERPRISE
 no huntstop
 moh-group 1
!
ephone-dn  3
 number 8983
 label 8983 for SPECTRUMENTERPRISE
 name SPECTRUMENTERPRISE Phone
 no huntstop
 moh-group 1
!
ephone  1[78]
 mac-address 0008.3031.F5D4[79]
 ephone-template 1[80]
 max-calls-per-button 1[81]
```

[73] Use this command to enter **ephone-dn** configuration mode to create directory numbers
[74] In ephone-dn configuration mode, you assign an extension number using the **number** command
[75] One label is allowed per extension (ephone-dn). The ephone-dn must already have a number that was set using the **number** command before a label can be created for it
[76] In ephone-dn configuration mode, you assign a name to appear in the local directory using the **name** command
[77] This command disables call hunting to the second channel of this ephone-dn if the first channel is busy or does not answer
[78] Use the ephone command to enter ephone configuration mode. Use ephone configuration mode to create and configure Cisco Unified IP phones in Cisco Unified CME
[79] Use this command to specify the MAC address of a specific Cisco IP phone in order to physically identify the Cisco IP phone in a Cisco CME configuration
[80] This command in ephone configuration mode applies an ephone template to a particular phone
[81] This command limits the maximum number of calls, both incoming and outgoing, that can be active on each octo-line directory number on an SCCP phone

```
 username "8982" password 1234[82]
 type 7975[83]
 mwi-line 1[84]
 button  1:1[85]
!
ephone  2
 mac-address 0008.3031.F2A8
 ephone-template 1
 type 7975
 button  1:2
!
ephone  3
 mac-address 64D8.14A4.7C4D
 ephone-template 1
 type 7942
 button  1:3
!
line con 0
 exec-timeout 0 0
 password ***********
 login
 stopbits 1
line aux 0
 stopbits 1
line vty 0 4
 exec-timeout 0 0
 password ***********
 login
 transport input telnet
!
ntp server 10.10.10.5
!
end

SPECTRUMENTERPRISECME#
```

[82] This command assigns a login account username and password for a phone user and establishes a login account for each Cisco Unified IP phone

[83] To assign a phone type to an SCCP phone, use the type command in ephone or ephone-template configuration mode

[84] This command designates a phone line other than the primary line to activate the MWI lamp on the phone

[85] The button command assigns telephone extensions to Cisco Unified IP phones by associating a button number with one or more directory numbers (ephone-dns)

Cisco Unity Connection

Version Details

Cisco Unity Connection user configuration

Navigation: Cisco Unity Connection → Users → Users

Set Alias*= 8982. This is used for this example.

Set First Name = 8982 is used to identify this User.

Set Last Name* = 8982. This is used for this example

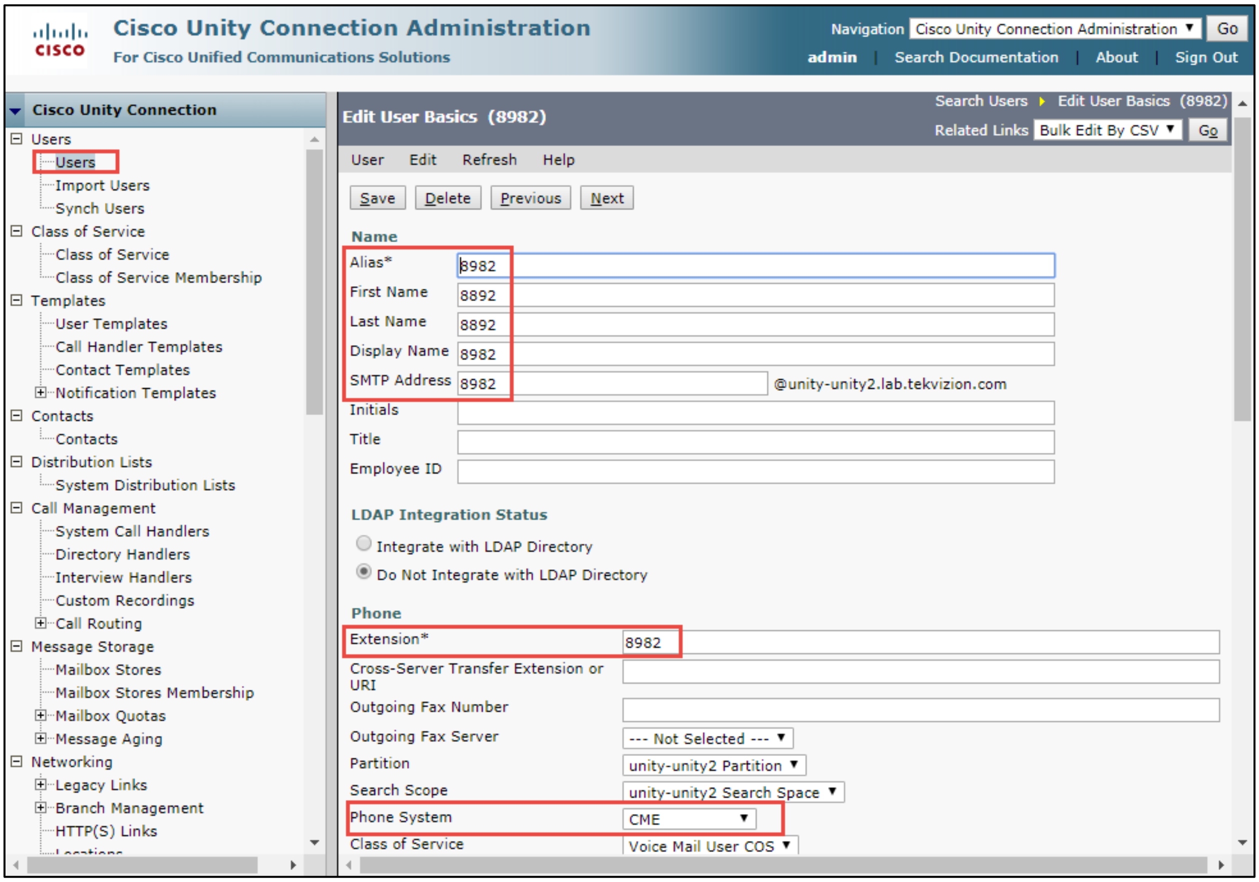
Set Display Name= 8982. This is used in this example.

Set SMTP Address =8982. This is used in this example.

Set Extension = 8982. This is used in this example.

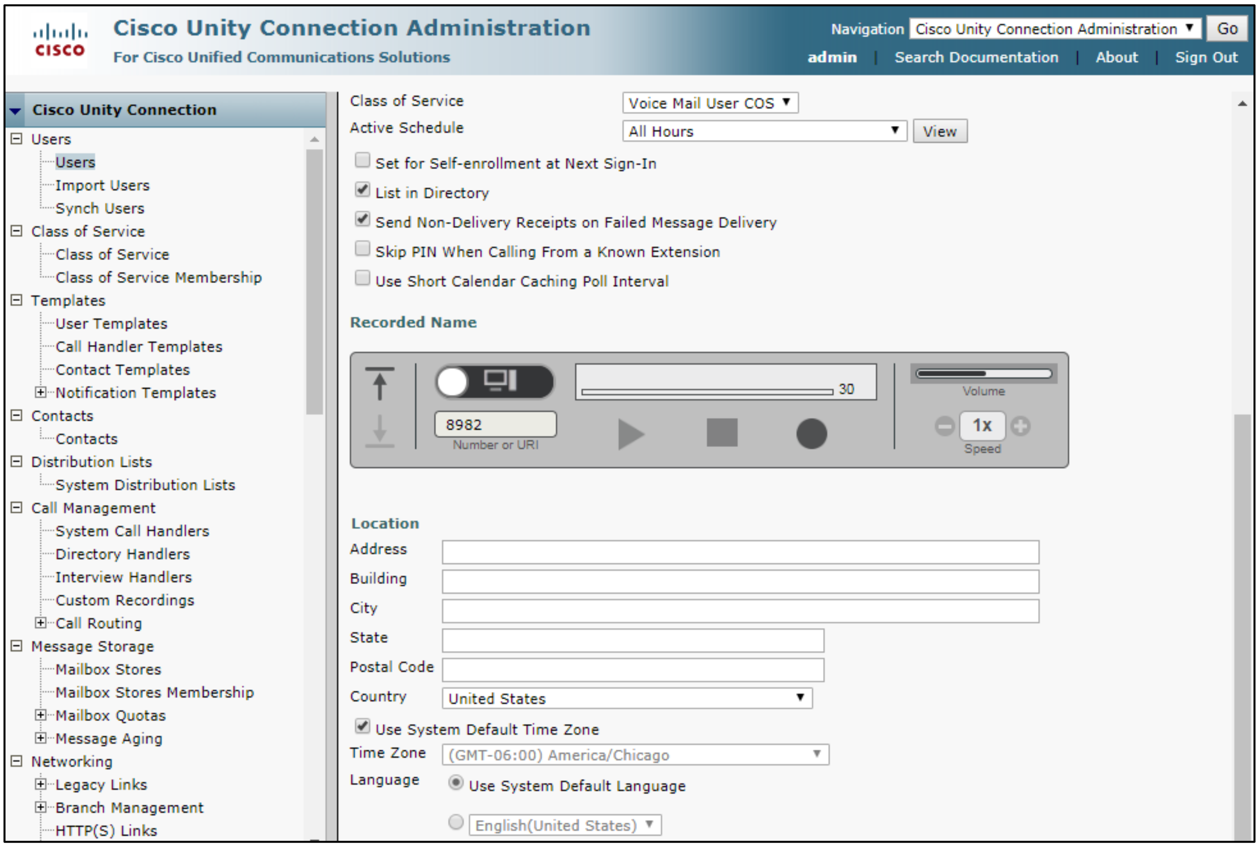
Set Phone System= CME. This is used in this example.

All other values are default.



Cisco Unity Connection User Configuration (Continued)

All values are default.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Postal Code
Country
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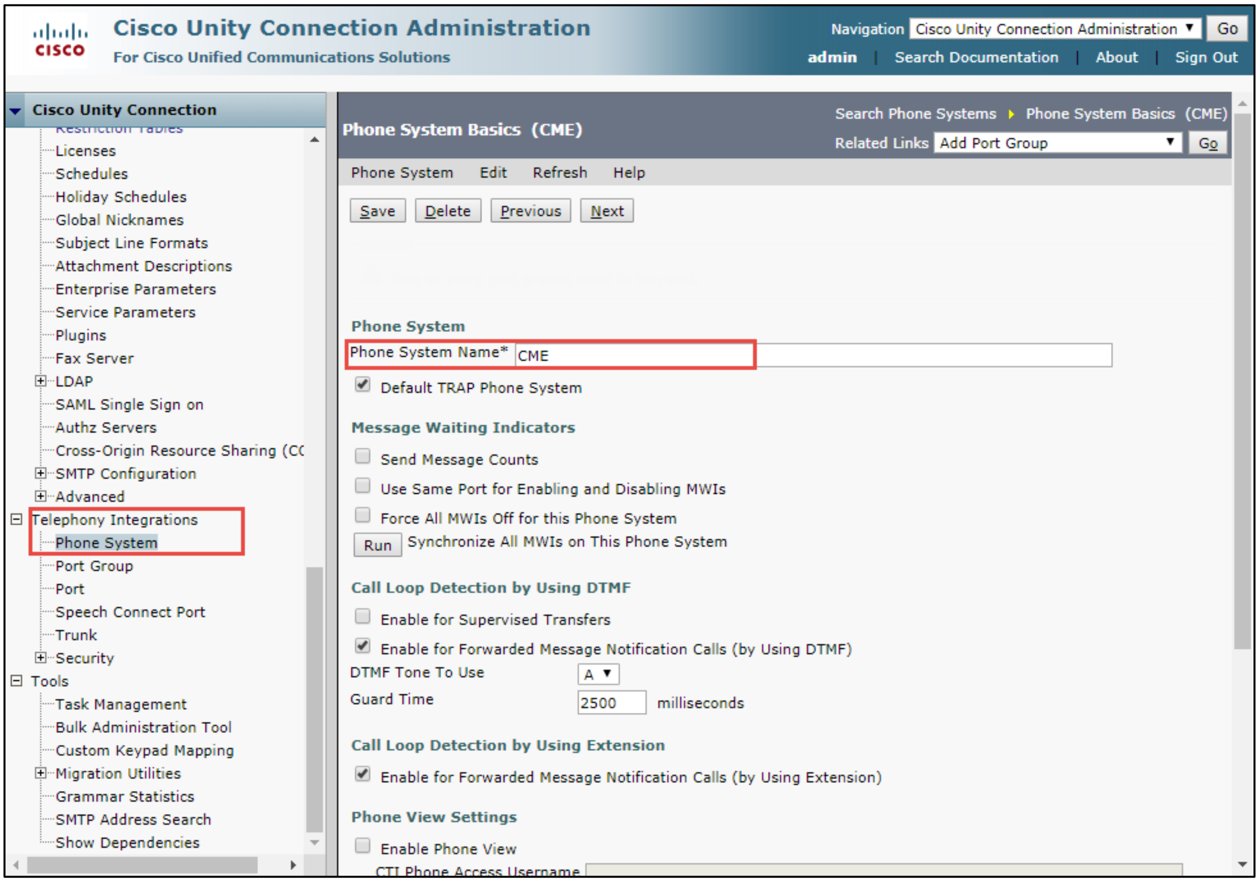
Use System Default Time Zone
Time Zone
(GMT-06:00) America/Chicago
Language
Use System Default Language
English(United States)

Cisco Unity Connection Telephony Integration

Navigation: Telephony Integrations → Phone system

Set System Name* = CME. This Name used for this example.

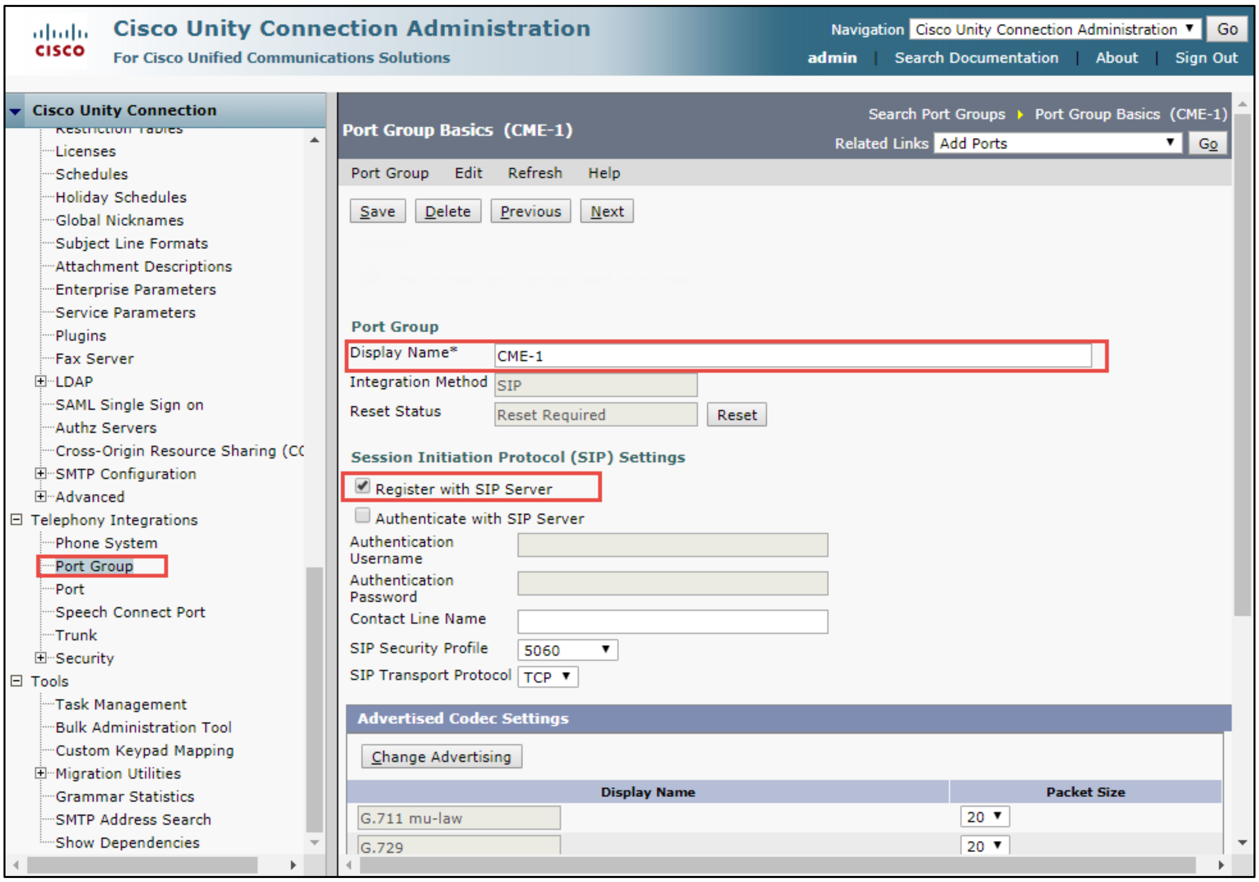


Port Group

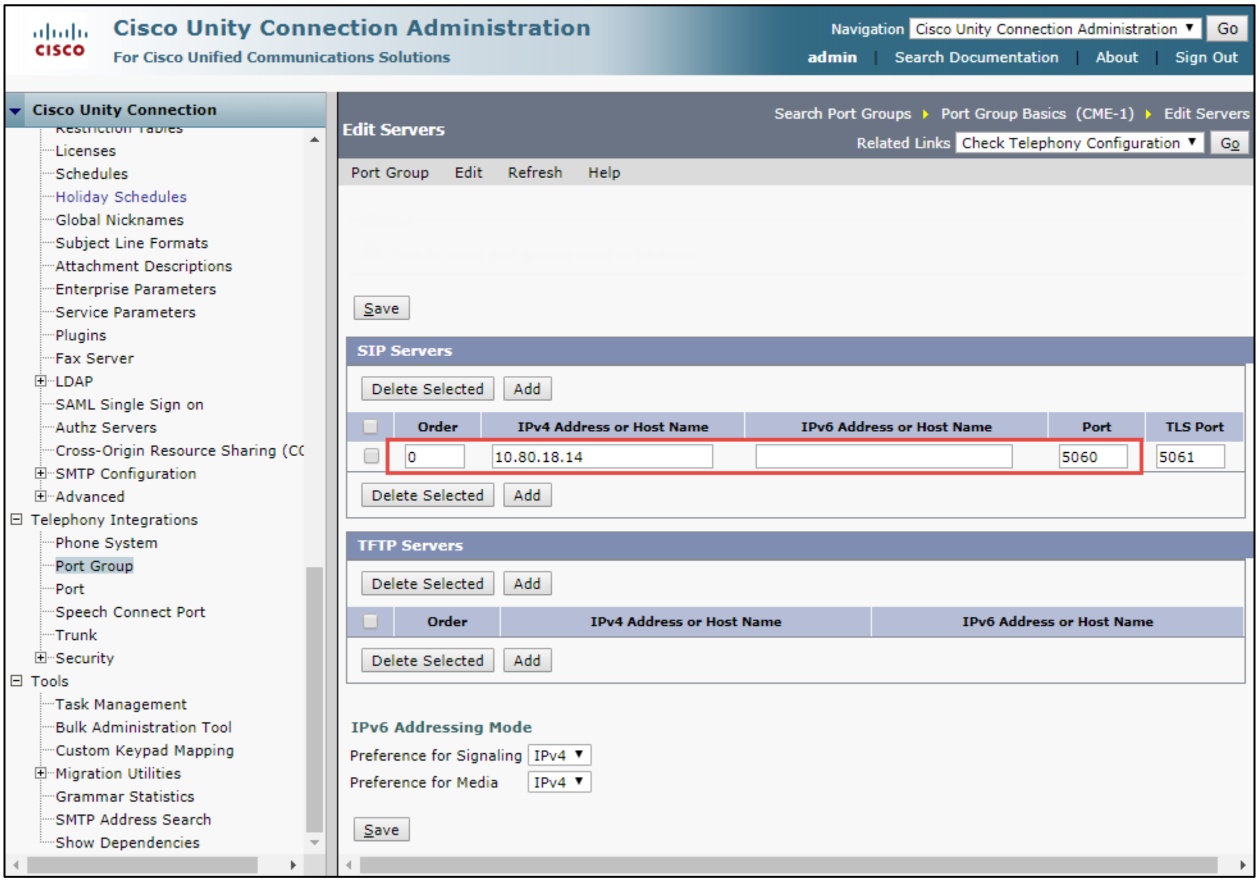
Navigation: Telephony Integration → Port Group.

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

Check Register with SIP server.



Navigation Path: Telephony Integration→ Port Group → Edit → Servers.



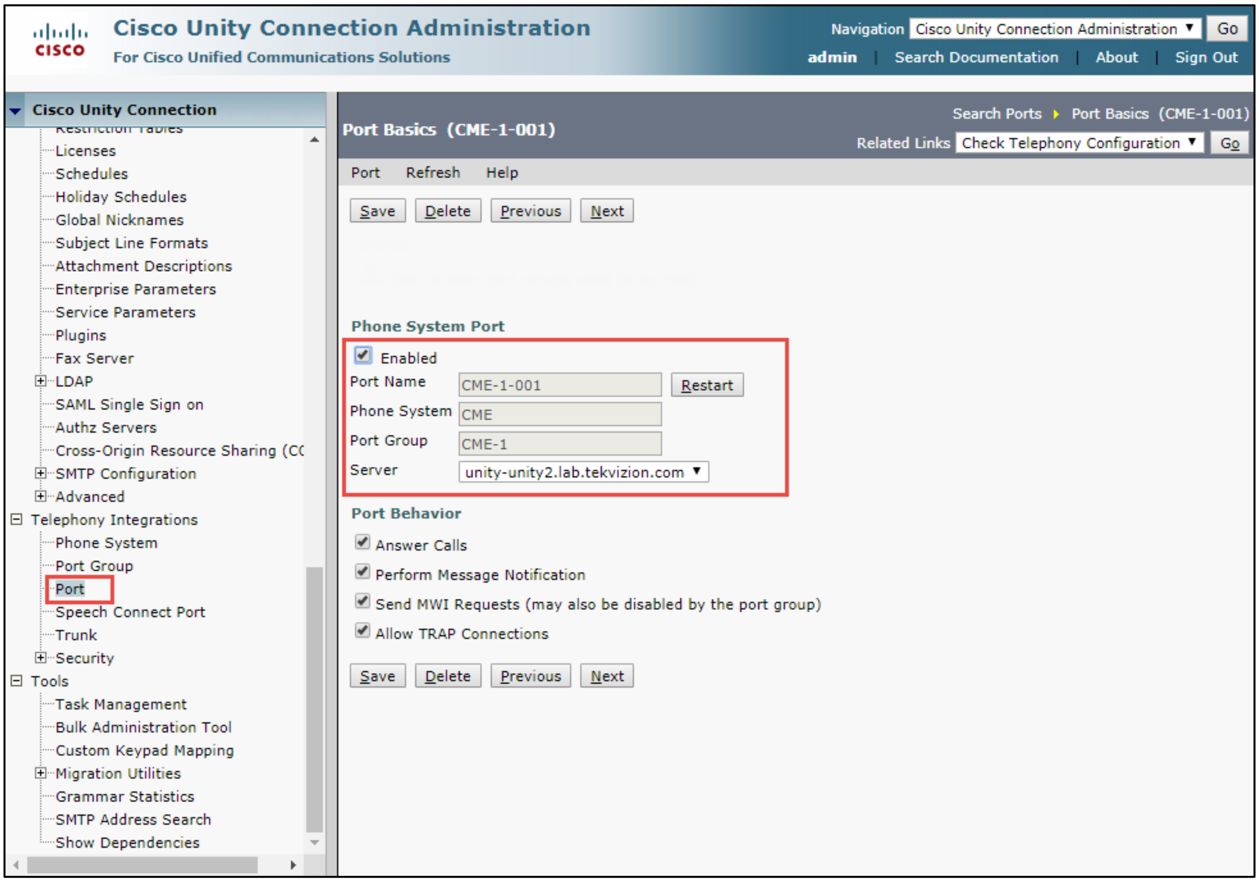
Port

Set Port Name = CME-1-001. This Name used for this example.

Phone System = CME.

Port Group = CME-1.

Server = unity-unity2.lab.tekvizion.com. This Name used for this example.



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Show Dependencies
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Search Ports Port Basics (CME-1-001)
Related Links Check Telephony Configuration Go
Port Refresh Help
Save Delete Previous Next
Phone System Port
Enabled
Port Name CME-1-001 Restart
Phone System CME
Port Group CME-1
Server unity-unity2.lab.tekvizion.com
Port Behavior
Answer Calls
Perform Message Notification
Send MWI Requests (may also be disabled by the port group)
Allow TRAP Connections
Save Delete Previous Next

Acronyms

Acronym	Definitions
CPE	Customer Premise Equipment
Cisco UBE	Cisco Unified Border Element
SfB	Skype for Business
MTP	Media Termination Point
POP	Point of Presence
PSTN	Public Switched Telephone Network
ESBC	Enterprise Session Border Controller
SIP	Session Initiation Protocol

Important Information

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