TECHNICAL GUIDE to access Business Talk & BTIP Cisco CUCME

versions addressed in this guide: 14.1

Version of 18/05/2021

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1 Goal of this document

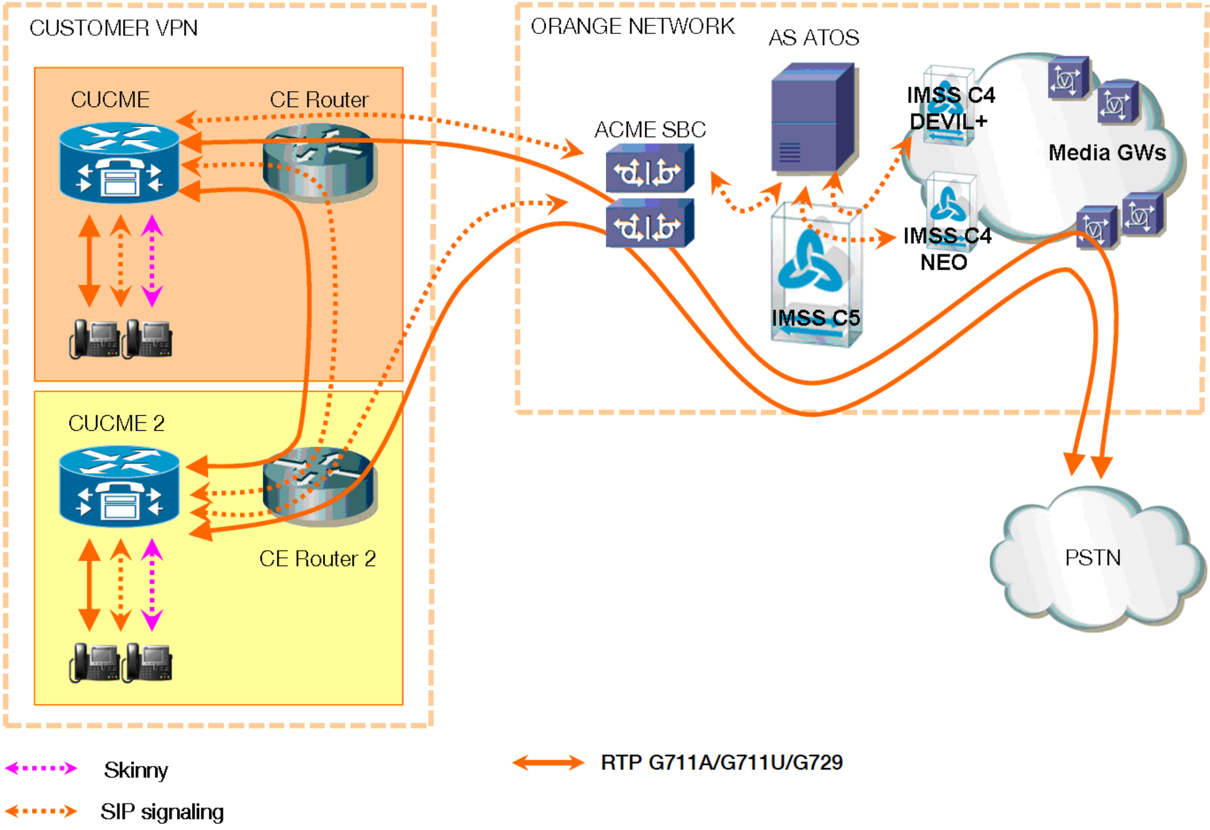
The aim of this document is to list technical requirements to ensure the interoperability between Cisco CUCME IPBX with Business Talk IP SIP, hereafter so-called “service”.

Note:

- This document describes “only” the main supported architectures either strictly used by our customers or that are used as reference to add specific usages often required in enterprise context (specific redundancy, specific ecosystems, multi-PBX environment, multi-codec and/or transcoding, recording…)

2 Architecture overview

2.1 Cisco Unified Communications Manager Express



In this architecture :

- all 'SIP trunking' signaling flows are carried by the CUCME server and routed on the main BVPN connection.
- Media flows are direct between endpoints and the Business Talk/BTIP but IP routing differs from one site to another :
  - For the Head Quarter site, media flows are just routed on the main BVPN connection
  - For Remote sites on BVPN, media flows are just routed on the local BVPN connection (= **distributed architecture**),
  - For Remote sites on Third Party WAN, media flows are routed through the Head Quarter (but not through the IPBX) and use the main BVPN connection (= **centralized architecture**).

2.2 Deployment models

This chapter describes the deployment models for CUCME, which have been validated for use with BT/BTIP customers. These deployment models are based on the official models outlined by Cisco in the SRND document and have been tested in Orange Engineering labs.

2.2.1 Single Site Model

The single-site model for Cisco Unified Communications consists of a CUCME router located at a single site or campus connected to VOIP IP VPN.

Characteristics:

- Single Cisco Unified Communications Manager Express router
  - Cisco ISR G3: 4000 series
  - Cisco ISR G2: 2900 and 3900 series are **not supported** for CUCME 12.1 or higher
- Maximum of 450 IP phones.
- Digital signal processor (DSP) resources for conferencing, and transcoding and PSTN termination
- Voice Mail - Cisco Unity Express 10.2 –Virtualized on UCS-E or as a KVM on ISR 4k
- High-bandwidth audio G.711 between devices within the site
- BTIP/BT for all calls outside the site, local PSTN for backup purpose.
- Capability to integrate with legacy private branch exchange (PBX) and voicemail systems

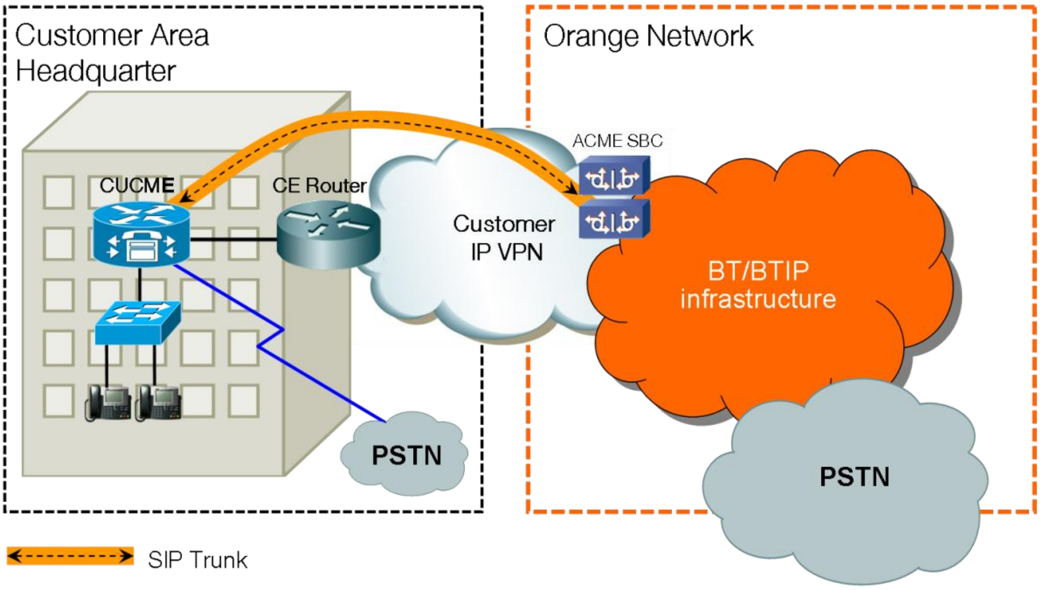


Figure 1. Single site model.

Benefits of a single site deployment model:

A single infrastructure for a converged network solution provides significant cost benefits and enables Cisco Unified Communications to take advantage of the many IP-based applications in the enterprise. Single-site deployment also allows each site to be completely self-contained. There is no dependency for service in the event of an IP WAN failure or insufficient bandwidth, and there is no loss of call processing service or functionality. In summary, the main benefits of the single-site model are:

- Ease of deployment
- A common infrastructure for a converged solution
- Simplified dial plan

2.2.2 Multisite WAN with Distributed Call Processing

The model for a multisite WAN deployment with distributed call processing consists of multiple independent sites, each with its own call processing agent cluster connected to an IP WAN that carries voice traffic between the distributed sites.

Characteristics

- Distributed Architecture is composed of two or more sites with Cisco Unified Communications Manager Express (CUCME) router.
- Communications between sites going via a direct SIP trunk.
- The same rules applies as for single site deployment.

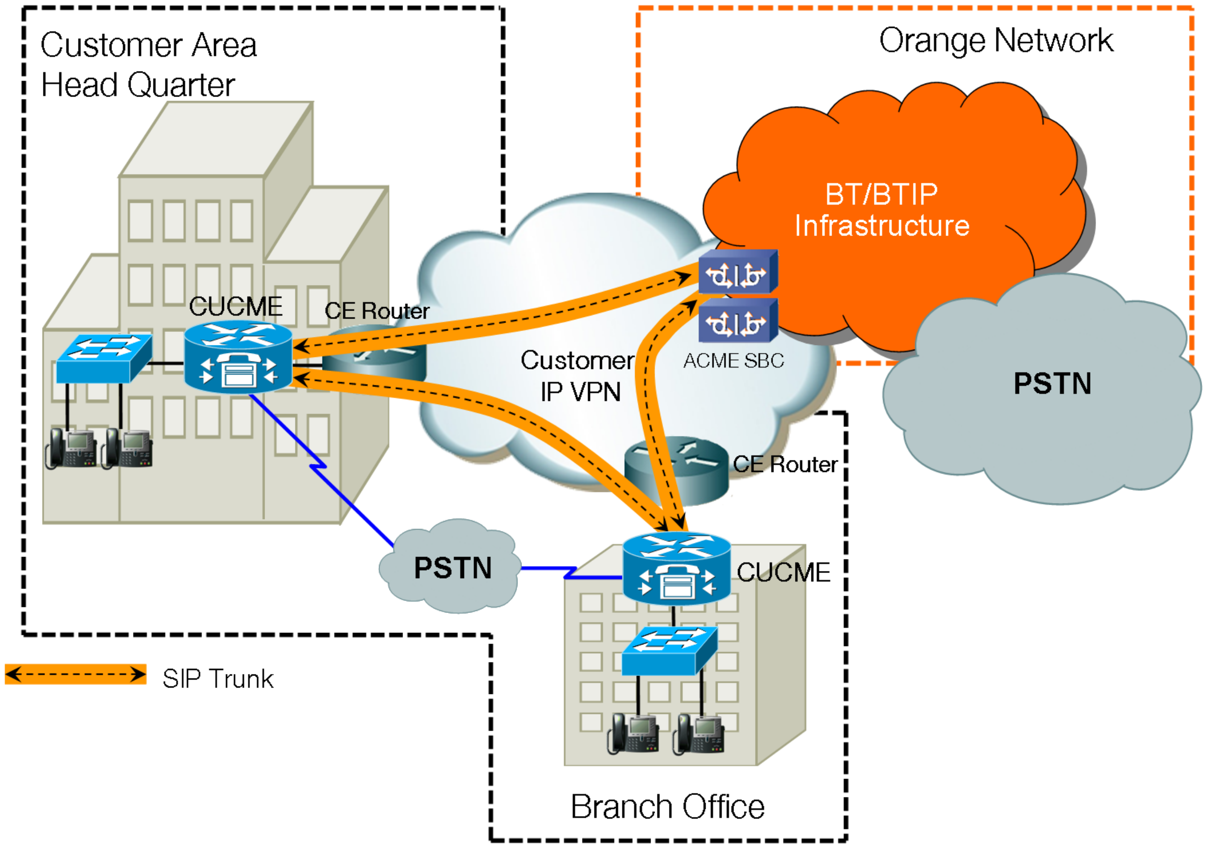


Figure 2. Multisite WAN with Distributed Call Processing.

2.3 Supported features

List of supported features:

- Basic calls (with & without call restriction) using G.711 a-law/G.711 u-law OR G.729 codec with 20ms payload (monocodec configuration – only one codec can be used in WAN for each customer)
- DTMF transport
- Music on Hold
- Call Transfer (supervised/blind)

- Call Forwarding (cfwall/on busy/on no answer)
- Ad-hoc conferencing
- Call Park/Call Pick-up/HuntGroup
- Voice Mail using Cisco Unity Express 10.2

3 Parameters to be provided by customer to access service

IP addresses marked in red have to be indicated by the customer, depending on customer architecture scenario.

3.1 CUCM Express

Head Quarter (HQ) or Branch Office (BO) architecture	Level of Service	Customer IP addresses used by service	
		Nominal	Backup
CUCM Express (1 server)	No redundancy (1 Publisher)	CUCME IP@	N/A

4 Certified software and hardware versions

4.1 CUCM Express certified versions

Cisco IPBX			
Equipment	Equipment Version	validation status	IPBX Version
CUCM Express	R14.1	✓	14.1

4.2 CUCM Express certified applications and devices versions

Cisco ecosystems					
Equipment		Equipment Version	validation status	IPBX Version	Comment
Voice Mail	Cisco Unity Express	10.2	✓	R14.1	
Integrated Services Router	ISR G3 (4000 series)	17.3.2	✓	R14.1	Starting with CUCME Release 12.1 ISR G2 are no longer supported
Phones	Cisco Unified Communication Manager Assistant (IPMA)		not supported	R14.1	
	All Cisco SCCP phones (skinny)		✓	R14.1	
	All Cisco SIP phones		✓	R14.1	
	IPCommunicator SCCP		not supported	R14.1	
	Jabber	12.7	✓	R14.1	

5 Cisco Unified Call Manager Express configuration

The checklists below present all the configuration steps required for interoperability between the service and CUCME.

5.1 Configuration for BT/BTIP

	Voice service voip configuration

Step 1	Important voice service voip configuration.
	`voice call send-alert`
	`voice service voip`
	` allow-connections sip to sip`
	` no supplementary-service sip moved-temporarily`
	` no supplementary-service sip refer`
	`sip`
	`  bind control source-interface <voice interface>`
	`  bind media source-interface <voice interface>`
	`  registrar server expires max 600 min 60`
	`  asserted-id pai`
	`  privacy pstn`
	`  sip-profiles <tag>`

	SNMP server configuration

Step 2	SNMP server is needed for IOSversion applet to work (described in a next step):
	`snmp-server community public RO`
	`snmp-server manager`

	Event Embedded Manager (EEM) applet configuration

Step 3	SIP profile will be create by Event Embedded Manager (EEM) applet when the CUCME boots:
	`event manager environment _sip_header_2 response 180 sip-header Server modify ".*" "Server: CUCME`
	`event manager environment _quote "`
	`event manager environment _sip_header_1 request INVITE sip-header User-Agent modify ".*" "User-Agent: CUCME"`
	`event manager environment _sip_header_3 response 183 sip-header Call-Info add "P-EARLY-MEDIA: sendrecv"`
	`event manager environment _sip_header_4 request INVITE sip-header Supported modify "timer," ""`
	`event manager environment _sip_header_5 response 180 sip-header Remote-Party-ID modify "privacy=full" "privacy=id"`
	`event manager environment _sip_header_6 response 183 sip-header Remote-Party-ID modify "privacy=full" "privacy=id"`
	`event manager environment _sip_header_7 response 200 sip-header Remote-Party-ID modify "privacy=full" "privacy=id"`
	
	`event manager applet IOSversion`
	` event timer countdown name IOSversion time 20`

```
 action 1.0 info type snmp oid 1.3.6.1.2.1.47.1.1.1.1.10.3
get-type exact
 action 2.0 cli command "enable"
 action 2.1 cli command "config t"
 action 3.0 cli command "no voice class sip-profiles 1"
 action 4.0 cli command "voice class sip-profiles 1"
 action 4.1 cli command "$_sip_header_1
$_info_snmp_value$_quote"
 action 4.2 cli command "$_sip_header_2
$_info_snmp_value$_quote"
 action 4.3 cli command "$_sip_header_3"
 action 4.4 cli command "$_sip_header_4"
 action 4.5 cli command "$_sip_header_5"
 action 4.6 cli command "$_sip_header_6"
 action 4.7 cli command "$_sip_header_7"
 action 5.0 cli command "voice service voip"
 action 5.1 cli command "sip"
 action 5.2 cli command "sip-profiles 1"
 action 6.0 cli command "end"
```

Codec configuration

Step 4

```
voice class codec <tag1>
 codec preference 1 <g711alaw/g711ulaw>
 codec preference 2 g729r8
voice class codec <tag2>
 codec preference 1 <g711alaw/g711ulaw or G729>
sip-ua
 g729-annexb override
```

Local (LAN) calls use codec G711. Remote calls use depend on customer can use one of following codecs G711alaw, G711ulaw or G729. Please configure two **voice class codec** lists: one with both codecs applied to SIP phones, and the other one applied to SIP trunks.

Multicast MOH configuration

Step 5

Telephony services configuration section

```
telephony-service
 multicast moh 239.10.16.4 port 20482
```

Note: Don't use low port numbers for multicast moh, especially don't use 2123. Instead of this use for example 20482

Dial-peer voice voip (to aSBC) configuration

Step 6

Create dial peers pointing to the primary and secondary aSBCs.

```
dial-peer voice <tag> voip
 preference <priority>
 progress_ind alert strip
 session protocol sipv2
 voice-class codec <tag>
```

	` voice-class sip options-keepalive up-interval 300 down-interval 300 retry 1`
	` dtmf-relay rtp-nte`
	` no vad`

	Voice translation rule

Step 7	Voice Translation rules manipulate digits of calling or called−numbers (depending on how they are referred to in the subsequent "voice translation−profile" command).
	
	`voice translation-rule <tag>`
	` rule 1 /matched_string/ /replacing_string/`
	
	`voice translation-profile <name>`
	` translate <called/calling> <tag>`
	
	`dial-peer voice <tag> voip`
	` translation-profile <outgoing/incoming> <name>`

Configuration of release/reroute behavior

Step 8

```
no voice hunt call-reject
sip-ua
set sip-status 400 pstn-cause 21
 set sip-status 401 pstn-cause 21
 set sip-status 403 pstn-cause 21
 set sip-status 404 pstn-cause 21
 set sip-status 405 pstn-cause 21
 set sip-status 406 pstn-cause 21
 set sip-status 409 pstn-cause 21
 set sip-status 410 pstn-cause 21
 set sip-status 413 pstn-cause 21
 set sip-status 414 pstn-cause 21
 set sip-status 415 pstn-cause 21
 set sip-status 416 pstn-cause 21
 set sip-status 417 pstn-cause 21
 set sip-status 420 pstn-cause 21
 set sip-status 422 pstn-cause 21
 set sip-status 480 pstn-cause 21
 set sip-status 481 pstn-cause 21
 set sip-status 482 pstn-cause 21
 set sip-status 483 pstn-cause 21
 set sip-status 484 pstn-cause 21
 set sip-status 485 pstn-cause 21
 set sip-status 486 pstn-cause 21
 set sip-status 487 pstn-cause 21
 set sip-status 488 pstn-cause 21
 set sip-status 580 pstn-cause 21
 set sip-status 600 pstn-cause 21
```

SIP-UA configuration

Step 9

```
sip-ua
 retry bye 2
```

	` retry invite 2` ` g729-annexb override`

5.2 Onnet calling between two CUCMEs of the same customer

	Dial-peer voice voip (to CUCME) configuration
Step 1	To interconnect two CUCMEs belonging to the same customer, a direct SIP Trunk need to be configured:

```
dial-peer voice <tag> voip
 preference <priority>
 session protocol sipv2
 voice-class codec <tag>
 voice-class sip options-keepalive up-interval 300 down-
interval 300 retry 1
 dtmf-relay rtp-nte
 no vad
```

5.3 Integration of CUCM and CUCME via direct sip trunk

	CUCM configuration
Step 1	Integration of Cisco Unified Communications Manager Express (CUCME) with Cisco Unified Communications Manager (CUCM) cluster requires a direct SIP trunk. Please consult CUCM configuration guidelines for detailed configuration steps. The configuration of such SIP Trunk is the same as the one described for off-net calls except that on trunk between sites there is no SIP Normalization Script.

	Dial-peer voice voip (CUCM) configuration
Step 2	Two dial peers are created pointing to CUCM cluster. Each of them can be customized according to customer dial plan:

```
dial-peer voice <tag> voip
 preference <priority>
 session protocol sipv2
 voice-class codec <tag>
 voice-class sip g729 annexb-all
 voice-class sip options-keepalive up-interval 300 down-
interval 300 retry 1
 dtmf-relay rtp-nte
 no vad
```

6 Cisco Unity Express configuration

For Cisco Unified Call Manager Express 14.1 validated release of Cisco Unity Express is 10.2. It is available as Virtual Machine that can be installed on a UCS-E or as a KVM on ISR 4k.

There are no specific configuration of CUE for VISIT BT/BTIP service.

To avoid usage of transcoder, please remember that there should be G.711u codec allowed on SIP trunks between CUCMEs of one customer. This codec should be also allowed to be used by dial peers assigned to SIP phones.

6.1.1 Cisco Unity Express 10.2

6.1.1.1 CUE instalation

Once UCS-E module is inserted and plugged into the router, configure CIMC in order to install ESXi operating system in order to be able to create CUE virtual machine.

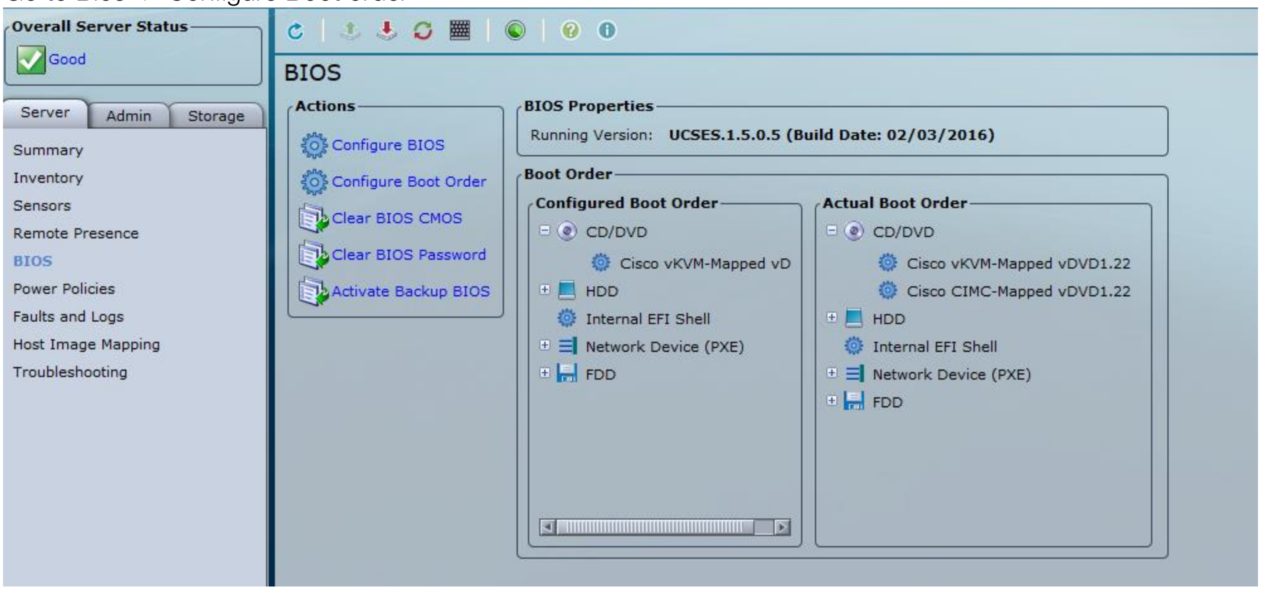
For ISR4k:

```
ucse subslot 1/0
imc ip address 10.25.0.16 255.255.255.0 default-gateway 10.25.0.120
imc access-port dedicated
```

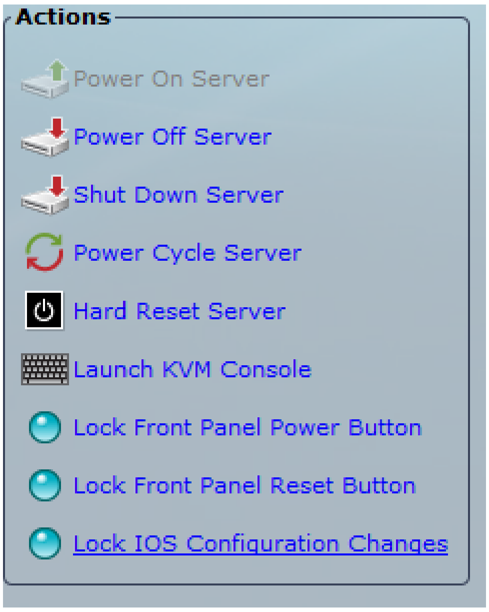
Please use dedicated mgmt port as in past some problems were encountered with CIMC monitoring.

Use your web browser to access CIMC. At first login use default credentials which are admin/password. You will be prompt to change them after first login.

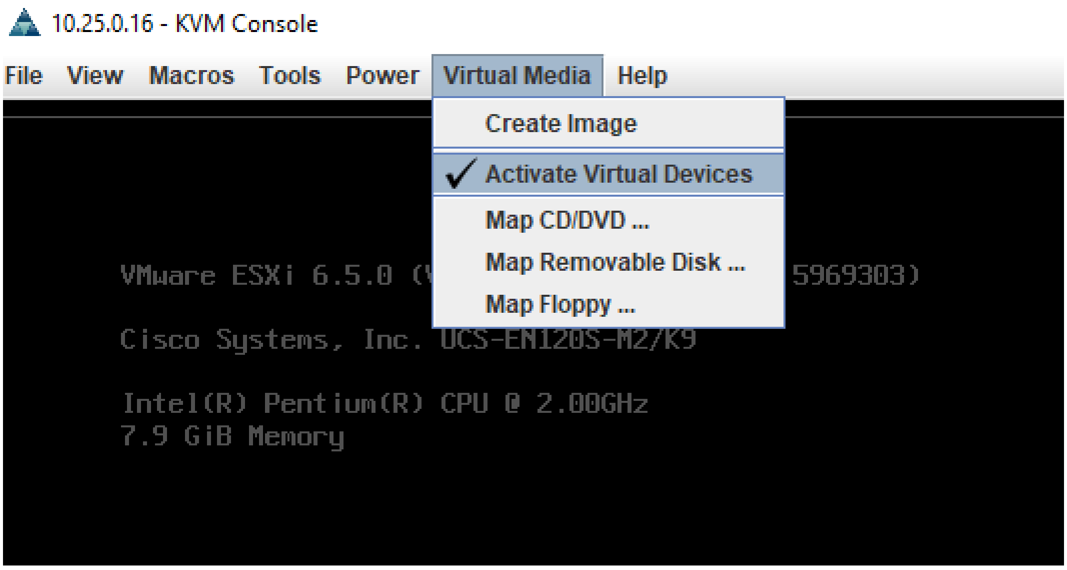
Once you have accessed CIMC, change the boot order prior to install ESXi:
Go to Bios -> Configure Boot order



Download ESXi software iso file to your desktop. Launch KVM console



Once connected, go to Virtual media and select activate virtual Devices



Than select map cd/dvd and browse to ESXi software you placed on your desktop.

Power cycle server. It should start from emulated CD/DVD drive with ESXi installation

Upon installation choose dedicated UCS-E interface for connection.

Once ESXi is configured and reachable, access it using either vCenter, or directly using vSphere Client, or web browser for ESXi 6.5.

Download CUE ova from Cisco support and deploy virtual machine.

Once virtual machine is deployed, open console and start virtual machine.

Step 1 In the vSphere Client GUI, in the left pane, select the Cisco Unity Express Virtual device. The name of the device is the name configured during installation.

Step 2 To open the console, click the Console icon in the vSphere toolbar.

A console window appears for the selected CUE instance.

Step 3 In the console window, click the Power On icon (appears as a green “Play” button).

The device boots, displaying the boot output in the console. When the start-up is complete, the console displays a message, prompting you to start configuration.

Step 4 At the prompt in the console window, confirm that you want to start the configuration process.

- y: If you enter y, the system asks you to confirm, then begin the interactive post installation configuration process.
- n: From Cisco Unity Express Virtual Release 9.0.3 onwards, if you select n, the initial setup wizard is skipped, and you are prompted to enter the administrator user ID and password. The default call agent is set to Unified CME.
- Timeout: If you do not enter any input for two minutes, the initial setup wizard is skipped, and you are prompted to enter the IP address, netmask, and default gateway address.

Step 5 When prompted, enter IP address and netmask of the device.

Note Cisco Unity Express Virtual requires IP communication access to the Cisco Unified Communications Manager and Remote sites.

Step 6 When prompted, enter the default gateway address. Confirm that the configuration is correct.

Step 7 When prompted for the host name, enter the name by which Cisco Unity Express Virtual appears within your network. Use a name that conforms to the fully qualified domain name (FQDN) rules.

Step 8 When prompted for a domain, enter a domain.

Note Configuring DNS server is optional. If DNS server is not configured, Cisco Unity Express Virtual gets the mapping of IP address to hostname and vice-versa, from “Extension: SubjectAltName” section of Unified Communications Manager certificate.

Step 9 When prompted regarding using DNS, enter y to configure Cisco Unity Express Virtual to use DNS.

Step 10 Enter the IP for the primary DNS server.

Step 11 Enter the IP for a secondary DNS server, if one is available. Otherwise, press Enter.

Step 12 When prompted for the primary network time protocol (NTP) server, enter the server domain name or the IP address. In some Cisco Unity Express Virtual software cases, a default server IP address appears automatically, and press Enter.

Note Cisco Unity Express Virtual requires a NTP server.

Step 13 When prompted for the secondary NTP server, enter the server domain name or IP if you have a secondary NTP. Otherwise, press Enter.

Step 14 When prompted for time zone information, use the menus to set your local time zone, and confirm when prompted.

Cisco Unity Express Virtual restarts.

Step 15 When prompted, enter an administrator user ID.

Step 16 Enter the password for the account, and confirm.

Note Ensure that you use this username and password to access Cisco Unity Express Virtual through SSH after installation.

When this procedure is complete, Cisco Unity Express Virtual indicates that the system is online and displays a command line prompt. For example:

SYSTEM ONLINE
CUE#

Verify network connectivity between CUE and CME.

Once your system is online and reachable, ssh into CUE, and activate evaluation

1. show license evaluation
2. license activate voicemail mailboxes
3. license activate ports
4. license activate ivr sessions
5. reload
6. show license in-use

6.1.1.2 CUCME for CUE configuration

The following example code present configuration Cisco Unified Communication Manager Express for Unity Express virtual machine.

Be aware that transcoder resources need to be configured and used if G711alaw or G729 codec is negotiated for calls to CUE.

```
ip http server

sip-ua
 mwi-server ipv4:6.1.0.5 expires 86400 port 5060 transport udp

voice register global
 voicemail 6666

telephony-service
 voicemail 6666
 web admin system name visit password visit1
```

```
dial-peer voice [tag] voip
 destination-pattern [voice mail number]
 session protocol sipv2
 session target ipv4:[IP_Address]
 dtmf-relay rtp-nte
 codec g711ulaw
 no vad
!
dial-peer voice [tag] voip
 destination-pattern [auto attendant number]
 session protocol sipv2
 session target ipv4:[IP_Address]
 dtmf-relay rtp-nte
 codec g711ulaw
 no vad
!
ephone-dn  2  dual-line
 number [number]
 name [name]
 call-forward busy [voice mail number]
 call-forward noan [voice mail number] timeout 10
 mwi sip
!
ephone-dn  [tag]
 number 8000....
 mwi sip
!
!
voice register dn  11
 number 1211
 mwi

voice register pool  11
 dtmf-relay rtp-nte
```

Note that MWI will not work if SNR is enabled on DN. Remove SNR first, configure MWI, and configure SNR back.

6.1.1.3 MWI

In order to enable MWI please use following command:

```
mwi-server ipv4:6.1.0.5 expires 3600 port 5060 transport udp
```

Explanation

Command	Description
mwi-server ipv4:6.3.25.5 expires 3600 port 5060 transport udp	Specifies MWI server (i.e. CUE), port and transport method how IP Phone is notified about voicemail.

6.1.1.4 CUE configuration (CLI)

The following example code present configuration that you should add to default configuration of the Unity Express.

```
username Kate create
username John create
username Sara create
username visit create
username Ronny create
username Bill create


username Kate phonenumber "1102"
username John phonenumber "1113"
```

```
ccn application voicemail aa
 description "voicemail"
 enabled
 maxsessions [number]
 end application

ccn subsystem sip
 gateway address "6.3.25.1"
 dtmf-relay rtp-nte
 mwi sip sub-notify
 transfer-mode attended
 end subsystem


ccn trigger sip phonenumber 6000
 application "voicemail"
 enabled
 maxsessions 2
 end trigger

snmp-server enable traps
snmp-server community public RO
snmp-server community public RW
snmp-server host 10.238.60.178 public
snmp-server host 10.25.0.225 public
snmp-server host 10.238.60.155 public

voicemail notification enable


voicemail mailbox owner "John" size 300
 description "John's Mailbox"
 expiration time 10
 messagesize 120
 end mailbox

voicemail mailbox owner "Kate" size 300
 description "Kate's mailbox"
 expiration time 10
 messagesize 120
 end mailbox


end
```

Explanation

Command	Description
username [name] create username lukas phonenumber "[number]"	Create user account
voicemail mailbox owner "[name]" size 400	Create mailbox for user and specify its size.
ccn subsystem sip	Specify the gateway address in this section
ccn trigger sip phonenumber [voice mail number]	Specify the number for the Voice Mail and its parameters
ccn application autoattendant aa	Enables Auto Attndant Script
ccn application ciscomwiapplication aa	Create numbers for Message Waiting Indicator (MWI) in this section.