CHAPTER 10

Cisco Unified CallManager Express Security Best Practices

Revised date: August 12, 2009

Cisco Unified CallManager Express (Cisco Unified CME) provides integrated IP communications on Cisco IOS routers. Therefore, the same security best practices recommended for all Cisco IOS voice-enabled routers also apply to Cisco Unified CME. In addition, you should implement Cisco Unified CME system-specific security practices to provide additional security protection.

This chapter describes how you can set up the Cisco Unified CME using the CLI to prevent users from intentionally or accidentally gaining system-level control from the GUI and local or remote CLI access. Specific sections presented in this chapter address the following Cisco Unified CME security considerations:

Note For additional information, see the "Related Documents and References" section on page xii.

Securing GUI Access

A Cisco IOS router authenticates an administrator CLI login against the enable password only, and the default setting for HTTP access is **ip http authentication enable**. If the system administrator, customer administrator, or phone user has the same password as the router's enable password, he or she can gain level 15 EXEC privilege access to Cisco IOS software by HTTP. A normal IP phone user can then accidentally change the Cisco Unified CME configuration, erase Flash, or reload the router when logging on to this URL:

http://*cme-ip-address*/

You should configure the following commands for Cisco Unified CME to use AAA or local authentication to prevent a normal user from gaining access to the enable password and therefore having access to the system administrator page:

ip http authentication aaa

or

ip http authentication local

System Administrator Account Authentication via AAA

Cisco Unified CME allows the system administrator username/password be authenticated by AAA. Use the following configuration to use AAA for system administrator user login:

```
ip http authentication
aaa new-model
aaa authentication login default group tacacs+ local
tacacs-server host 10.1.2.3
```

Note Normal username/password is not authenticated by AAA.

Using HTTPS for Cisco Unified CME GUI Management

HTTP over SSL (HTTPS) provides Secure Socket Layer (SSL) version 3.0 support for the HTTP 1.1 server and HTTP 1.1 client within Cisco IOS software. SSL provides server authentication, encryption, and message integrity to allow secure HTTP communications. SSL also provides HTTP client authentication. This feature is supported only in Cisco IOS software images that include the SSL feature. Specifically, SSL is supported in the Advanced Security, Advanced IP Services, and Advanced Enterprise Services images. Use the Advanced IP Services or Advanced Enterprise Services Cisco IOS images to get both the Cisco Unified CME and SSL features.

IP phones do not serve as HTTPS clients. If HTTPS is enabled on the Cisco Unified CME router, IP phones still attempt to connect to HTTP using port 80. Because the SSL default port is 443, the phones cannot display local directory and system speed dials. IP phones using HTTP can work with a system configured for SSL by enabling both HTTP and HTTPS, as shown in the following example.

```
ip http server
ip http secure-server
ip http secure-port port_number
!if https port is changed from default 443
ip http authentication AAA | TACACS | local
```

Use the following command to generate an RSA usage key pair with a length of 1024 bits or greater:

crypto key generate rsa usage 1024

If you do not generate an RSA usage key pair manually, an RSA usage key pair with a length of 768 bits is generated automatically when you connect to the HTTPS server for the first time. These auto generated RSA keys are not saved to the startup configuration; therefore, they are lost when the device is rebooted unless you save the configuration manually.

You should obtain an X.509 digital certificate with digital signature capabilities for the device from a certification authority (CA). If you do not obtain a digital certificate in advance, the device creates a self-signed digital certificate to authenticate itself.

If you change the device hostname after obtaining a device digital certificate, HTTPS connections to the device *fail* because the hostname does not match the hostname specified in the digital certificate. Obtain a new device digital certificate using the new hostname to fix this problem.

The **ip http secure-server** command prevents clear-text passwords from traveling across the network when a Cisco Unified CME administrator logs into the Cisco Unified CME GUI. However, communications between the phone and router remain in clear text.

The following are the suggested best practices for using HTTP interactive access to the Cisco Unified CME router:

- Use the **ip http access-class** command to allow only specified IP addresses to access the Cisco Unified CME GUI, thus restricting unwanted IP packets from connecting to Cisco Unified CME.
- Use the **ip http authentication** command with a central TACACS+ or RADIUS server for authentication purposes. Configuring authentication for the HTTP and HTTPS servers adds security to communication between clients and the HTTP and HTTPS servers on the device.
- Do not use the router enable password as a Cisco Unified CME login password (to prevent a regular user from gaining administrator privileges).

Configuring Basic Cisco Unified CME Access Security

This section summarizes the measures available to ensure only authorized users and systems can access Cisco Unified CME system-based resources. The following topics are addressed in this section:

Setting Local and Remote System Access

When in privileged EXEC mode, the **configure terminal** and **telephony-service** commands take a user into Cisco Unified CME configuration mode. The **show running-config** and **show telephony-service** commands show all registered phones and users, extension numbers, usernames, and passwords for Cisco Unified CME GUI access. An initial step to security control is at the system access level. Password encryption, user authentication, and command auditing are all critical to prevent security breaches.

Enabling Secret and Encrypt Passwords

The Enable password is presented in cleartext to provide access control to privileged EXEC mode of the router. Use Enable Secret to encrypt the enable password.

The following example illustrates this configuration:

```
enable secret secretword1
no enable password
```

The **enable secret** command takes precedence over the **enable password** command if both are configured; they cannot be used simultaneously.

To increase security access, passwords can be encrypted to prevent any unauthorized users from viewing the passwords when packets are examined by protocol analyzers:

The following example illustrates this configuration:

```
Service password-encryption
```

Creating Multiple Privilege Levels

By default, Cisco IOS software has two levels of access to commands: User EXEC mode (level 1) and privileged EXEC mode (level 15). Configuring up to 16 privilege levels (from 0, the most restricted level, to 15, the least restricted level) to protect the system from unauthorized access. Use the **privilege** *mode* **level** command.

The following example illustrates this configuration:

```
privilege exec level 14
enable secret level 2 secretword2
```

Restrict Access to VTY

Allow only certain users/locations to Telnet to the router via vty by defining and applying an access list for permitting or denying remote Telnet sessions.

The following example illustrates this configuration:

```
line vty 0 4
 access-class 10 in
 access-list 10 permit 10.1.1.0 0.0.0.255
```

Using AAA to Secure Access

An authentication server can be used to validate user access to the system. The following commands allow an AAA server, TACACS+ server, to be used for authentication services.

The following example illustrates this configuration:

```
aaa new-model
aaa authentication login default tacacs+ enable
aaa authentication enable default tacacs+ enable
ip tacacs source-interface Loopback0
tacacs-server host 10.17.1.2
tacacs-server host 10.17.34.10
tacacs-server key xyz
! Defines the shared encryption key to be xyz
```

Configuring Accounting and Auditing on AAA

The following commands use a TACACS+ server for command accounting and auditing purposes.

```
aaa new-model
aaa authentication login default tacacs+ enable
```

(login uses TACACS+, if not available, use enable password)

```
aaa authentication enable default tacacs+ enable
aaa accounting command 1 start-stop tacacs+
(runs accounting for commands at the specified privilege level 1)

aaa accounting exec start-stop tacacs+
ip tacacs source-interface Loopback0
tacacs-server host 10.17.1.2
tacacs-server host 10.17.34.10
tacacs-server key xyz (defines the shared encryption key to be xyz)
```

The example command log shows the information contained in a TACACS+ command accounting record for privilege level 1.

```
Wed Jun 25 03:46:47 1997 192.168.25.15 fgeorge tty3 5622329430/4327528 stop
task_id=3 service=shell priv-lvl=1 cmd=show version <cr>
Wed Jun 25 03:46:58 1997 192.168.25.15 fgeorge tty3 5622329430/4327528 stop
task_id=4 service=shell priv-lvl=1 cmd=show interfaces Ethernet 0 <cr>
Wed Jun 25 03:47:03 1997 192.168.25.15 fgeorge tty3 5622329430/4327528 stop
task_id=5 service=shell priv-lvl=1 cmd=show ip route <cr>
```

Configuring Local User Authentication When AAA Is Not Available

You should always require login-based authentication of users—even when the external AAA server is unreachable.

The following example illustrates this configuration:

```
username joe password 7 045802150C2E
username jim password 7 0317B21895FE
!
line vty 0 4
login local
```

Restricting Access to tty

You can allow only certain users and locations to Telnet to the router by using its terminal (tty) or virtual terminal (vty) lines. Define and apply an access list for permitting or denying remote Telnet sessions to your Cisco Unified CME router as shown in the following example.

```
line vty 0 4
 access-class 10 in
 access-list 10 permit 10.1.1.0 0.0.0.255
```

Configuring SSH Access

Use the following command to generate RSA key pairs for the router:

crypto key generate rsa

By default the vty's transport is Telnet. The following command disables Telnet and supports only SSH to the vty lines.

```
line vty 0 4
transport input ssh
```

Using ACLs for SNMP Access

The community access string can be set up to permit access to the Simple Network Management Protocol (SNMP). The following example assigns the *changeme-rw* string to SNMP, allowing read-write access and specifies that IP access list 10 can use the community string:

```
access-list 10 remark SNMP filter
access-list 10 permit 10.1.1.0 0.0.0.255
snmp-server community changeme-rw RW 10
snmp-server community changeme-ro RO 10
```

Because read and write are two common community strings for read and write access, respectively, change the community strings to different ones.

Disabling Cisco Discovery Protocol

Because Cisco Discovery Protocol (CDP) automatically discovers the neighboring network devices supporting CDP, disable CDP in an untrusted domain so that Cisco Unified CME routers will not appear in the CDP table of other devices. Disable CDP with the following command:

no cdp run

If CDP is needed, then consider disabling CDP on a per-interface basis, as in the following example:

```
Interface FastEthernet0/0
no cdp enable
```

Configuring COR for Incoming and Outgoing Calls

One of the ways to restrict unauthorized incoming and outgoing calls is to use the Class or Restriction (COR) commands. The configuration shown in the following example defines two groups of users: *user* and *superuser. Superuser* is allowed to make any calls, including local, long-distance, 411 directory lookup, and 911 calls. *User* is restricted from making 900, 411, and international calls.

```
dial-peer cor custom
 name 911
 name 1800
 name local-call
 name ld-call
 name 411
 name int-call
 name 1900
!
dial-peer cor list call911
 member 911
!
dial-peer cor list call1800
 member 1800
!
dial-peer cor list calllocal
 member local-call
!
dial-peer cor list callint
 member int-call
!
dial-peer cor list callld
 member ld-call
```

```
!
dial-peer cor list call411
 member 411
!
dial-peer cor list call1900
 member 1900
!
dial-peer cor list user
 member 911
 member 1800
 member local-call
 member ld-call
!
dial-peer cor list superuser
 member 911
 member 1800
 member local-call
 member ld-call
 member 411
 member int-call
 member 1900
!
dial-peer voice 9 pots
 corlist outgoing callld
 destination-pattern 91.........
 port 1/0
 prefix 1
!
dial-peer voice 911 pots
 corlist outgoing call911
 destination-pattern 9911
 port 1/0
 prefix 911
!
dial-peer voice 11 pots
 corlist outgoing callint
 destination-pattern 9011T
 port 2/0
 prefix 011
!
dial-peer voice 732 pots
 corlist outgoing calllocal
 destination-pattern 9732.......
 port 1/0
 prefix 732
!
dial-peer voice 800 pots
 corlist outgoing call1800
 destination-pattern 91800.......
 port 1/0
 prefix 1800
!
dial-peer voice 802 pots
 corlist outgoing call1800
 destination-pattern 91877.......
 port 1/0
 prefix 1877
!
dial-peer voice 805 pots
 corlist outgoing call1800
 destination-pattern 91888.......
 port 1/0
 prefix 1888
!
```

```
dial-peer voice 411 pots
 corlist outgoing call411
 destination-pattern 9411
 port 1/0
 prefix 411
!
dial-peer voice 806 pots
 corlist outgoing call1800
 destination-pattern 91866.......
 port 1/0
 prefix 1866

ephone-dn  1
 number 2000
 cor incoming user

ephone-dv 2
 number 2001
 cor incoming superuser
```

Restricting Outgoing Calling Patterns

You might use the **after-hours block** command to restrict incoming or outgoing calls after certain hours. You can also use after-hours blocking to restrict calls to numbers or area codes known to be fraudulent calling patterns. The commands shown in the following example block all calls at all times for patterns 2 to 6. Pattern 7 is blocked only during the configured after-hours period.

```
telephony-service
after-hours block pattern 2 .1264 7-24
after-hours block pattern 3 .1268 7-24
after-hours block pattern 4 .1246 7-24
after-hours block pattern 5 .1441 7-24
after-hours block pattern 6 .1284 7-24
after-hours block pattern 7 9011
after-hours day Sun 19:00 07:00
after-hours day Mon 19:00 07:00
after-hours day Tue 19:00 07:00
after-hours day Wed 19:00 07:00
after-hours day Thu 19:00 07:00
after-hours day Fri 19:00 07:00
after-hours day Sat 19:00 07:00
```

Cisco Unified CME Security for IP Telephony

The following topics are addressed in this section:

IP Phone Registration Control

Configure Cisco Unified CME to allow IP phones in the trusted domain for registration. Assuming that the local segment is a trusted domain, use the **strict-match** option in the **ip source-address** command, so that only locally attached IP phones will be able to register to the Cisco Unified CME router and get telephony services.

```
CME-3.0(config-telephony)# ip source-address 10.1.1.1 port 2000 strict-match
```

You can group a set of IP phones into one VLAN (such as 10.1.1.0/24), so that only IP phones in the specified VLAN can register to the Cisco Unified CME.

Block port 2000 access from the WAN side to prevent external SCCP phones from registering with Cisco Unified CME. Use the following **access-list** to block port 2000 access from WAN interfaces. The following example illustrates this configuration:

```
access-list 101 deny tcp any any eq 2000
```

You can also prevent unknown or unconfigured IP phones from being registered by disabling automatic registration using the following command:

```
CME-4.0(config-telephony)# no auto-reg-ephone
```

Note Disabling auto registration also disables the GUI ephone provisioning and Cisco Unified CME SRST Fallback. With Cisco Unified CME 3.x and prior releases, provision ephones before configuring the IP source address in order to workaround auto-registration behavior.

Prior to Cisco Unified CME 4.0, unknown phones or phones that are not configured in Cisco Unified CME are allowed to register with Cisco Unified CME by default for ease of management, but these phones do not provide a dial tone until you configure them by associating the buttons with the ephone-dns or configuring **auto assign** (from **telephony-service** configuration mode).

The following commands illustrates configuring ephone-dns with the **ephone-dn** command.

```
ephone-dn 1
number 1001

ephone-dn 2
number 1002

ephone 1
mac-address 1111.2222.3333
button 1:1 2:2
```

The following commands illustrate configuring the **auto assign** command:

```
CMEtest4-3745(config)# telephony-service
CMEtest4-3745(config-telephony)# auto assign 1 to 500
```

With Cisco Unified CME 4.0, you can configure **no auto-reg-ephone** in **telephony-service** configuration mode so that IP phones that are not explicitly configured with their MAC addresses in ephone configuration mode are prevented from automatically registering with the Cisco Unified CME system.

Monitoring IP Phone Registration

Cisco Unified CME 3.0 added the following syslog messages to generate and display all registration/deregistration events:

```
%IPPHONE-6-REG_ALARM
%IPPHONE-6-REGISTER
%IPPHONE-6-REGISTER_NEW
%IPPHONE-6-UNREGISTER_ABNORMAL
%IPPHONE-6-REGISTER_NORMAL
```

The following message indicates that a phone has registered and is not part of the explicit router configuration (ephone configuration has not been created or the MAC address has not been assigned):

```
%IPPHONE-6-REGISTER_NEW: ephone-3:SEP003094C38724 IP:10.4.170.6 Socket:1 DeviceType:Phone
has registered.
```

Note With Cisco Unified CME 4.0 and later releases, if you have configured the **no auto-reg-ephone** command, then the preceding message is not generated.

Cisco Unified CME allows unconfigured phones to register in order to make provisioning of the Cisco Unified CME system more convenient. By default, phones designated as “new” are not assigned phone lines and cannot make calls.

You can use the following configuration to enable syslogging to a router's buffer/console or a syslog server:

```
logging console | buffered
logging 192.168.153.129
! 192.168.153.129 is the syslog server
```

Call Activity Monitoring and Call History Logging

The Cisco Unified CME GUI provides call history table information so that a network administrator can monitor the call history information for unknown callers and use this information to disallow calling activities based on select calling patterns. The call history log should be configured to perform forensics and accounting and allow the administrator to track down fraudulent calling patterns. Configure the following commands to log call activity and call history:

```
dial-control-mib retain-timer 10080
dial-control-mib max-size 500
!
gw-accounting syslog
```

Toll Fraud Prevention

When a Cisco router platform is installed with a voice-capable Cisco IOS software image, appropriate features must be enabled on the platform to prevent potential toll fraud exploitation by unauthorized users. Deploy these features on all Cisco router Unified Communications applications that process voice calls, such as Cisco Unified Communications Manager Express (CME), Cisco Survivable Remote Site Telephony (SRST), Cisco Unified Border Element (UBE), Cisco IOS-based router and standalone analog and digital PBX and public-switched telephone network (PSTN) gateways, and Cisco contact-center VoiceXML gateways. These features include, but are not limited to, the following:

- Disable secondary dial tone on voice ports—By default, secondary dial tone is presented on voice ports on Cisco router gateways. Use private line automatic ringdown (PLAR) for foreign exchange office (FXO) ports and direct-inward-dial (DID) for T1/E1 ports to prevent secondary dial tone from being presented to inbound callers.
- Cisco router access control lists (ACLs)—Define ACLs to allow only explicitly valid sources of calls to the router or gateway, and therefore to prevent unauthorized Session Initiation Protocol (SIP) or H.323 calls from unknown parties to be processed and connected by the router or gateway.
- Close unused SIP and H.323 ports—If either the SIP or H.323 protocol is not used in your deployment, close the associated protocol ports. If a Cisco voice gateway has dial peers configured to route calls outbound to the PSTN using either time division multiplex (TDM) trunks or IP, close the unused H.323 or SIP ports so that calls from unauthorized endpoints cannot connect calls. If the protocols are used and the ports must remain open, use ACLs to limit access to legitimate sources.
- Change SIP port 5060—If SIP is actively used, consider changing the port to something other than well-known port 5060.
- SIP registration—If SIP registration is available on SIP trunks, turn on this feature because it provides an extra level of authentication and validation that only legitimate sources can connect calls. If it is not available, ensure that the appropriate ACLs are in place.
- SIP Digest Authentication—If the SIP Digest Authentication feature is available for either registrations or invites, turn this feature on because it provides an extra level of authentication and validation that only legitimate sources can connect calls.
- Explicit incoming and outgoing dial peers—Use explicit dial peers to control the types and parameters of calls allowed by the router, especially in IP-to-IP connections used on CME, SRST, and Cisco UBE. Incoming dial peers offer additional control on the sources of calls, and outgoing dial peers on the destinations. Incoming dial peers are always used for calls. If a dial peer is not explicitly defined, the implicit dial peer 0 is used to allow all calls.
- Explicit destination patterns—Use dial peers with more granularity than .T for destination patterns to block disallowed off-net call destinations. Use class of restriction (COR) on dial peers with specific destination patterns to allow even more granular control of calls to different destinations on the PSTN.
- Translation rules—Use translation rules to manipulate dialed digits before calls connect to the PSTN to provide better control over who may dial PSTN destinations. Legitimate users dial an access code and an augmented number for PSTN for certain PSTN (for example, international) locations.
- Tcl and VoiceXML scripts—Attach a Tcl/VoiceXML script to dial peers to do database lookups or additional off-router authorization checks to allow or deny call flows based on origination or destination numbers. Tcl/VoiceXML scripts can also be used to add a prefix to inbound DID calls. If the prefix plus DID matches internal extensions, then the call is completed. Otherwise, a prompt can be played to the caller that an invalid number has been dialed.
- Host name validation—Use the “permit hostname” feature to validate initial SIP Invites that contain a fully qualified domain name (FQDN) host name in the Request Uniform Resource Identifier (Request URI) against a configured list of legitimate source hostnames.
- Dynamic Domain Name Service (DNS)—If you are using DNS as the “session target” on dial peers, the actual IP address destination of call connections can vary from one call to the next. Use voice source groups and ACLs to restrict the valid address ranges expected in DNS responses (which are used subsequently for call setup destinations).

For more configuration guidance, see the “Cisco IOS Unified Communications Toll Fraud Prevention” paper.

COR for Incoming/Outgoing Calls to Prevent Toll Fraud

The following configuration example illustrates COR. There are two classes of service in the configuration: user and superuser along with various permissions allowed such as local calling, long distance calling, 911 access, and 411 access. In this example, *superuser* has access to everything and *user* has access to all resources with the exception of toll 1900, directory assistance 411, and international calling.

```
dial-peer cor custom
 name 911
 name 1800
 name local-call
 name ld-call
 name 411
 name int-call
 name 1900

dial-peer cor list call911
 member 911
!
dial-peer cor list call1800
 member 1800
!
dial-peer cor list calllocal
 member local-call
!
dial-peer cor list callint
 member int-call
!
dial-peer cor list callld
 member ld-call
!
dial-peer cor list call411
 member 411
!
dial-peer cor list call1900
 member 1900

dial-peer cor list user
 member 911
 member 1800
 member local-call
 member ld-call
!
dial-peer cor list superuser
 member 911
 member 1800
 member local-call
 member ld-call
 member 411
 member int-call
 member 1900

dial-peer voice 9 pots
 corlist outgoing callld
 destination-pattern 91..........
 port 1/0
 prefix 1
!
dial-peer voice 911 pots
 corlist outgoing call911
 destination-pattern 9911
```

```
 port 1/0
 prefix 911
!
dial-peer voice 11 pots
 corlist outgoing callint
 destination-pattern 9011T
 port 2/0
 prefix 011
!
dial-peer voice 732 pots
 corlist outgoing calllocal
 destination-pattern 9732.......
 port 1/0
 prefix 732
!
dial-peer voice 800 pots
 corlist outgoing call1800
 destination-pattern 91800.......
 port 1/0
 prefix 1800
!
dial-peer voice 802 pots
 corlist outgoing call1800
 destination-pattern 91877.......
 port 1/0
 prefix 1877
!
dial-peer voice 805 pots
 corlist outgoing call1800
 destination-pattern 91888.......
 port 1/0
 prefix 1888
!
dial-peer voice 411 pots
 corlist outgoing call411
 destination-pattern 9411
 port 1/0
 prefix 411
!
dial-peer voice 806 pots
 corlist outgoing call1800
 destination-pattern 91866.......
 port 1/0
 prefix 1866

ephone-dn 1
 number 2000
 cor incoming user

Ephone-dn 2
 number 2001
 cor incoming superuser
```

After-hours Blocking to Restrict Outgoing Calling Pattern-Toll Fraud

After-hours blocking can be added to restrict incoming calls after certain hours. After-hours blocking can also be used to restrict calls to numbers/area codes known as fraudulent calling patterns. The following configuration example can be used to restrict calls to certain area codes:

```
telephony-service
```

```
after-hours block pattern 1 .1242
after-hours block pattern 2 .1264
after-hours block pattern 3 .1268
after-hours block pattern 4 .1246
after-hours block pattern 5 .1441
after-hours block pattern 6 .1284
after-hours block pattern 7 .1345
after-hours block pattern 8 .1767
after-hours block pattern 9 .1809
after-hours block pattern 10 .1473
after-hours block pattern 11 .1876
after-hours block pattern 12 .1664
after-hours block pattern 13 .1787
after-hours block pattern 14 .1869
after-hours block pattern 15 .1758
after-hours block pattern 16 .1900
after-hours block pattern 17 .1976
after-hours block pattern 18 .1868
after-hours block pattern 19 .1649
after-hours block pattern 20 .1340
after-hours block pattern 21 .1784
after-hours block pattern 22 .1684
after-hours block pattern 23 .1590
after-hours block pattern 24 .1456
after-hours day Sun 00:00 23:59
after-hours day Mon 00:00 23:59
after-hours day Tue 00:00 23:59
after-hours day Wed 00:00 23:59
after-hours day Thu 00:00 23:59
after-hours day Fri 00:00 23:59
after-hours day Sat 00:00 23:59
```

Cisco Unified CME with NAT and Firewall

The following topics are addressed in this section:

Cisco Unified CME with NAT

Typically, Cisco Unified CME router's LAN interface (Ethernet interface) is used as a source IP address used by the IP phones and the Cisco Unified CME router to communicate with each other. However, when an internal switch module is used to connect IP phones, the VLAN's IP address can be used as a source IP address. A loopback interface's IP address is another option for a source IP address.

The IP addresses of the IP phones are internal addresses to the Cisco Unified CME router and are in a different segment that is not visible by the external devices or callers. Other devices including Cisco gateways or gatekeeper use the Cisco Unified CME router's IP address to communicate instead of directly communicating with the IP phones. The Cisco Unified CME router translates IP addresses back and forth for the traffic to route to the IP phones or outside of the network area. Therefore, no NAT

configuration is needed for two-way voice/audio from/to the IP phones locally attached to the Cisco Unified CME router. We recommend that NAT be deployed for data traffic only with Cisco Unified CME.

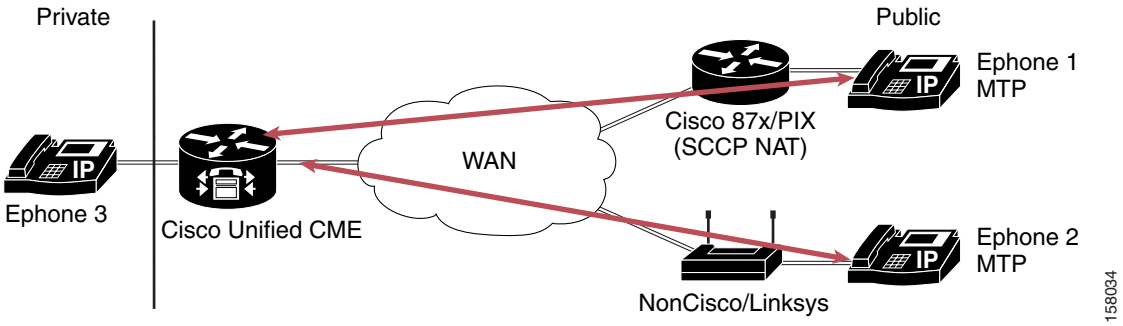
NAT may be required for IP phones deployed remotely which do not have routable IP addresses.

Note Cisco Unified CME IP address used as the source IP address needs to be routable and may be a loopback IP address in all the scenarios described in this section. Also, the UDP/TCP ports must be open between remote IP phones and Cisco Unified CME source address.

Remote Phones with Public IP Addresses

Remote phone support introduced in Cisco Unified CME 4.0 allows IP phones to be connected to Cisco Unified CME across a WAN link such as Frame Relay, DSL, and cable. Figure 10-1 shows a typical scenario for this connectivity arrangement.

Figure 10-1 Remote Phones with Public IP Addresses



In the scenario in Figure 10-1, *ephone 3* is in a private VLAN and uses Cisco Unified CME to reach *ephone 1* and *ephone 2* in remote sites with public IP addresses. However, because media streams are sent between the phones connected to the same Cisco Unified CME, Media Termination Point (MTP) should be configured on the remote phones in order to have Cisco Unified CME terminate the media stream—thereby ensuring two-way audio between *ephone 3* and *ephone 1* or *ephone 2*. Codec G729r8 is required for the remote phones. The configuration on *ephone 1* or *ephone 2* is as follows:

```
ephone 1
 mtp
 codec g729r8
```

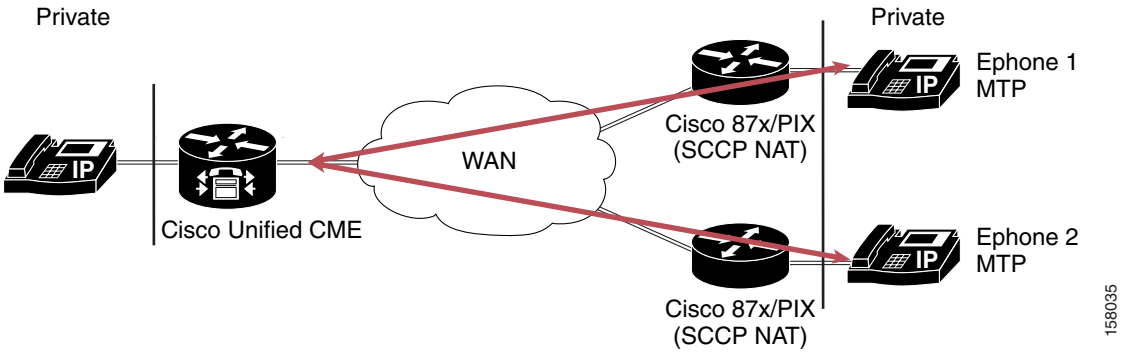
The *MTP* option under *ephone 1* causes the Cisco Unified CME router to act as a proxy. The Cisco Unified CME forwards media packets to other IP phones with the Cisco Unified CME router's address in the source address field. If another phones in the call is not an IP phone, Cisco Unified CME forwards the media packets.

Note If all phones have public IP addresses, then MTP configuration is not required and media will flow between phones (rather than through the Cisco Unified CME). Unless MTP is required for your implementation, we recommend that you do not use it. As in the prior scenario, the UDP/TCP ports must be open between remote IP phones and the Cisco Unified CME source address.

Remote Phones with Private IP Addresses

Figure 10-2 illustrates a typical scenario when remote phones are deployed with private IP addresses in the remote site.

Figure 10-2 Remote Phone Connection with Private IP Addresses



Remote phones can be connected via a traditional Cisco router (such as Cisco 87x or Cisco PIX) or using an alternative routing device (such as Linksys router). Both implementation require that NAT be configured if routable IP addresses are not used on the remote phones. NAT SCCP support is required to implement two-way audio between IP phones connected to the Cisco Unified CME. With NAT allowing for the translation of the embedded IP addresses and port numbers presented in the SCCP messages, a full NAT entry can be created to allow RTP traffic to flow between IP Phones. As a result, two-way voice/audio is permitted between the IP phones being connected via NAT. For a device such as Linksys router, which is not SCCP aware, a one-way audio issue exists between the two IP phone endpoints. A workaround is to connect the remote IP phone attached to the Linksys via a DMZ port with routable IP addresses or to establish a VPN connection to the Cisco Unified CME router to avoid having a one-way audio issue.

Caveats:

- NAT SCCP support is available in Cisco IOS Release 12.3(11)T and later in Cisco IOS routers.
- MTP is required to be configured on the remote phones.
- Remote phones attached through a Cisco router with SCCP NAT support also require the configuration of MTP in order to support two-way audio.
- Remote phones attached to a nonCisco SCCP NAT router will encounter a one-way audio issue even if MTP is configured on the remote phones. A workaround is to use VPN between Cisco Unified CME and the a nonCisco SCCP NAT router or obtain public IP addresses for the remote phones.

Note As in the prior examples, the UDP/TCP ports must be open between remote IP phones and Cisco Unified CME source address.

Remote Phones over VPN

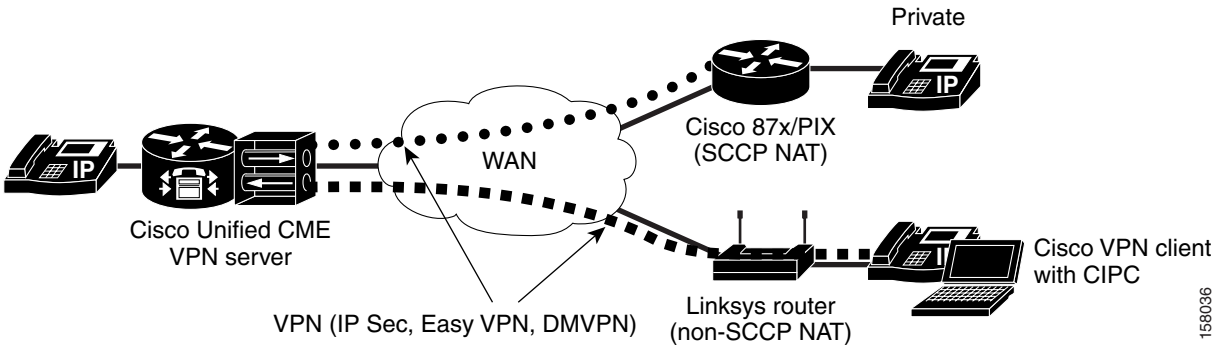
Remote phones with private IP addresses can be connected to phones attached to a Cisco Unified CME using a nonCisco router. However, in order to support two-way audio between these privately addressed remote phones and phones attached to a Cisco Unified CME (which have public IP addresses), a VPN IP Sec tunnel must be established between Cisco Unified CME and the nonCisco router.

VPN can also be used to connect Cisco Unified CME and Cisco SCCP NAT aware routers such as Cisco 87x/PIX, allowing for connections supported by QoS and VPN acceleration.

Figure 10-3 illustrates examples of these VPN-related environments.

Note As in the prior examples, the UDP/TCP ports must be open between remote IP phones and Cisco Unified CME source address.

Figure 10-3 Remote Phone Connection using VPN



Cisco Unified CME with Cisco IOS Firewall Implementation Considerations

This description of the Cisco Unified CME implementation with Cisco IOS firewall addresses the following topics:

- Overview of Cisco IOS Firewall with Cisco Unified CME, page 10-17
- Previous Problems on Cisco Unified CME with Cisco IOS Firewall, page 10-18
- Cisco Unified CME and Cisco IOS Firewall on the Same Router, page 10-19
- Other Alternatives for Ensuring Cisco Unified CME Security, page 10-20

Overview of Cisco IOS Firewall with Cisco Unified CME

The Cisco IOS Firewall, running on Cisco IOS routers, provides a network-based firewall solution with the functionality of Context-based Access Control (CBAC) or protocol inspection, Cisco Intrusion Detection System (Cisco IDS), authentication proxy, and URL filtering. A firewall provides access control between internal and external networks. It identifies networks as inside (private) or outside (public) in which packets can get from the inside to the outside, be blocked by default from outside to inside, and packets associated with an inside-originated connection are allowed to pass in. Many

firewalls work only if all outside traffic originates from well-known sockets and do not handle asymmetric traffic (such as UDP media). Cisco IOS firewalls allow packets to pass through based on source and destination IP addresses and the configured firewall policy.

Cisco Unified CME is a software feature added to the Cisco IOS routers that provides call processing for IP phones using Skinny Client Control Protocol (SCCP) for branch/SMB, and managed SP environments. There can be instances of SMB or branch office implementations in which a single router is required to provide Internet access, IP telephony service, and Cisco IOS Firewall functions. Cisco Unified CME requires that all IP phones be attached to the Cisco Unified CME router locally —before remote phone support was introduced.

Therefore, H.323 and SCCP support on the Cisco IOS Firewall are needed for locally generated traffic.

Previous Problems on Cisco Unified CME with Cisco IOS Firewall

SCCP is a Cisco proprietary small version of H.323. H.323 traffic can be classified into call signalling, call control, and media communication. H.323 uses Q.931, H.225, and H.245 to set up, manage/control, and tear down calls. The following descriptions address how signaling and media streams are affected by the Cisco IOS firewall.

Signaling Stream

An H.323 call requires a TCP connection for H.245 signalling that does not have an associated well-known port. The H.245 port is dynamically assigned. Because this port is not known ahead of time and cannot be configured when defining firewall policy, the Cisco IOS Firewall will block the H.245 message and the call signalling procedure will fail. When NAT is used in the H.323 signalling path, an inside IP address (which is behind the NAT and is not known to the rest of the world), will be used as the “calling party” information element in the H.225 signalling stream. As a result, an incoming call (attempts to make an H.225 connection back to that address) will fail.

Media Streams (RTP streams)

RTP streams run on top of UDP and do not have any fixed ports associated with them. Each type of media stream has one or more channels with dynamically assigned source, destination, and port numbers, which are not known ahead of time and cannot be preconfigured in the firewall policy. For the media stream to traverse the firewall, the firewall must open many UDP ports with source and destination pairs for each call session. This can open vulnerabilities to the network behind the firewall.

Because the Cisco IOS Firewall does not allow outside traffic to transverse to the inside destinations, VoIP calls (inbound calls) will fail. Furthermore, dynamic RTP/RTCP ports used by the endpoints are not automatically opened and allowed without modification of the security policy. The problems are summarized as follows:

- The firewall only looks at Layer 3 addresses.
- VoIP signalling protocols embed IP addresses at Layer 4 and above
  - RTP/RTCP works at Layer 5.
  - By default, firewalls do not allow outside to inside traffic.
  - Cisco IOS firewall feature set and NAT and PIX have application functionality called the Application Layer Gateway (ALG), or fixup, protocol which helps resolve these issues.
- The VoIP application is composed of a dynamic set of protocols.
  - SIP, MGCP, H.323, and SCCP for signalling
  - SDP, H.225, and H.245 for capability exchange

- RTP/RTCP for control and audio media
- RTP/RTCP both use a dynamic port for the audio media ranging from 16384 to 32767 for all Cisco products

Note The Cisco IOS Firewall did not previously support Skinny inspection, because outgoing packets are converted to H323 or SIP. As a result, there is no need for Skinny inspection. However, ACLs can be used to filter out unwanted packets/traffic as a way to support incoming Skinny packet inspection. Cisco IOS Firewall has added H.323 inspection support for any locally generated traffic, thus making it possible to deploy Cisco Unified CME and IOS Firewall on the same router.

Cisco Unified CME and Cisco IOS Firewall on the Same Router

As long as Cisco IOS Firewall is not applied to the interfaces that have voice traffic (signaling and media) coming in, Cisco Unified CME and Cisco IOS Firewall can co-exist on the same router. The inspection of router-generated traffic, available in Cisco Release IOS 12.3(14) T and later, enhances Cisco IOS Firewall functionality to inspect TCP, UDP, and H.323 connections that have a router or firewall as one of the connection endpoints. Inspection of TCP and UDP channels initiated from the router enables dynamic opening of pinholes on the interface access control list (ACL) to allow return traffic. Inspection of local H.323 connections enables the deployment of Cisco Unified CME and Cisco IOS Firewall on the same router. This also simplifies ACL configuration on Cisco Unified CME interface through which H.323 connections are made. Before this feature, multiple ACLs were required to allow all dynamically negotiated data and media channels—in addition to ACLs required to allow H.323 connections on a standard port such as 1720. With this feature, you configure the ACLs to allow H.323 control channels on port 1720. The Cisco IOS Firewall inspects all the traffic on the control channel and opens pinholes to allow dynamically negotiated data and media channels.

The following procedure illustrates ACL configuration to support this capability:

Step 1 Create the ACL. In this example, TCP traffic from subnet 10.168.11.1, 192.168.11.50, and 192.168.100.1 is permitted.

```
access-list 120 permit tcp host 10.168.11.1 any eq 1720
access-list 121 permit tcp host 192.168.11.50 host 10.168.11.1 eq 1720
access-list 121 permit tcp host 192.168.100.1 host 10.168.11.1 eq 1720
```

Step 2 Create the Cisco IOS Firewall inspection rule LOCAL-H323. This allows for the inspection of the protocol traffic specified by the rule. This inspection rule sets the timeout value to 180 seconds for each protocol (except for RPC). The timeout value defines the maximum time that a connection for a given protocol can remain active without any traffic passing through the router. When these timeouts are reached, the dynamic ACLs that are inserted to permit the returning traffic are removed, and subsequent packets (possibly even valid ones) are not permitted.

```
ip inspect name LOCAL-H323 tftp timeout 180
ip inspect name LOCAL-H323 h323 router-traffic timeout 180
```

Step 3 Apply the inspection rule and ACL. In this example, the inspection rule LOCAL-H323 is applied to traffic at interface Serial0/3/0:

```
interface Serial0/3/0
 ip address 10.168.11.2 255.255.255.0
 ip access-group 121 in
 ip access-group 120 out
 ip inspect LOCAL-H323 in
 ip inspect LOCAL-H323 out
 encapsulation frame-relay
```

```
frame-relay map ip 10.168.11.1 168 broadcast
 no frame-relay inverse-arp
 frame-relay intf-type dce
```

Step 4 The Cisco IOS Firewall supports only version 2 of the H.323 protocol. Configure the following in the Cisco Unified CME to support only version 2 features:

```
voice service voip
h323
session transport tcp calls-per-connection 1
h245 tunnel disable
h245 caps mode restricted
h225 timeout tcp call-idle value 0
```

Other Alternatives for Ensuring Cisco Unified CME Security

The following are four alternative solutions that you can use to provide security to the Cisco Unified CME users:

- Run Cisco IOS Firewall on a different router—it is not required to be on the same Cisco Unified CME.
- Set up a maximum number of connections in the Cisco Unified CME. This is available with the regular H.323 implementation in Cisco IOS software and can help control the maximum number of H.323 (H225 setup Inbound + Outbound) calls that will be processed (such as `dial-peer voice 10 voip; max-conn 5` limits calls to five connections).
- Set up ACLs to accept H.225 connections only from the gatekeeper (GK) if the GK in the network is using routed signaling.
- Use H.235 security to authenticate the callers and provide additional call security

Secure SCCP Signaling via TLS

Cisco Unified CME 4.0 introduced in Cisco IOS Release 12.4(4)XC provides phone authentication and secure SCCP signalling with Transport Layer Security (TLS).

Phone authentication is a security infrastructure for providing secure SCCP between Cisco Unified CME and IP phones. Phone authentication addresses the following security needs:

- Establishing the identity of each endpoint in the system
- Authenticating devices
- Providing signaling-session privacy
- Providing protection for configuration files

Note Secure RTP is not supported in Cisco Unified CME 4.0.

The secure phone authentication feature is supported in the following two Cisco IOS feature sets:

- Advanced IP Services (such as c3725-advipservicesk9-mz.124-4.XC.bin)
- Advanced Enterprise Services (c3725-adventerprisek9-mz.124-4.XC.bin)

Supported phones are Cisco Unified IP Phone 7911G, Cisco Unified IP Phone 7941G, Cisco 7961G, and Cisco Unified IP Phone7970/71G-GE.

Key considerations for secure SCCP signaling via TLS are as follows:

- Certificate Trust List (CTL) client is used to create the CTL file and makes it available in the TFTP directory The CTL file (CTLfile.tlv) contains the public key information of all the servers with which the IP phone will interact.
- A digitally signed configuration file (SEP<MAC-addr>.cnf.xml.sgh) is created by the telephony-service module in Cisco IOS software. The router's private key is used for signing this document.
- Certificate Authority Proxy Function (CAPF)—a proxy between the IP phone and the Certification Authority (CA)—is used to request for a certificate on behalf of the phone. It is through the CAPF protocol that the CAPF server gets all the required information from the phone (including the public key and phone ID). CAPF configuration status resides in the CNF file.
- Phone authentication occurs between the Cisco Unified CME and a supported device when each entity accepts the certificate of the other entity, and when a secure connection between the entities occurs. Phone authentication relies on the creation of a CTL file.
- File authentication validates digitally signed files that a phone downloads from a TFTP server: config files, ringist files, and locale and CTL files. When receiving these types of the files, the phone validates the file signatures to verify that file tampering did not occur after the files were created.
- Signaling authentication, also known as signaling integrity, uses the TLS protocol to validate that signaling packets have not been tampered with during transmission. Signaling authentication relies on the creation of the CTL file.

Use the following procedure to configure support for SCCP signaling using TLS:

Step 1 Configure NTP or manually set the software clock using the **clock set** command as in the following example:

```
clock timezone PST -8
clock summer-time PDT recurring
ntp clock-period 17247042
ntp server 171.68.10.80
ntp server 171.68.10.150
```

Step 2 Configure a Cisco IOS Certification Authority (CA) —The CA issues certificates to Cisco Unified CME, CAPF, TFTP, and SAST server functions:

The CA can be on the same Cisco Unified CME router or on an external router. The following example illustrates configuring a CA on the same Cisco Unified CME router:

```
crypto pki server laverda-ca
 grant auto
 database url flash:
!
crypto pki trustpoint laverda-ca
 enrollment url http://192.168.1.1:80
 revocation-check crl
 rsakeypair laverda-ca
```

Step 3 Certificate provisioning for Cisco Unified CME functions: *capf server*, *cme server*, *tftp server*, *sast1*, and *sast2* as illustrated in the following configuration examples.

a. Obtain a certificate for *capf server*:

```
!configuring a trust point
 crypto pki trustpoint capf-server
 enrollment url http://192.168.1.1:80
 revocation-check none
!authenticate w/ the CA and download its certificate
```

```
crypto pki authenticate capf-server
! enroll with the CA and obtain this trustpoint's certificate
 crypto pki enrollment capf-server
```

b. Obtain a certificate for *cme server*:

```
crypto pki trustpoint cme-server
 enrollment url http://192.168.1.1:80
 revocation-check none

crypto pki authenticate cme-server
crypto pki enrollment cme-server
```

c. Obtain a certificate for the *tftp server*:

```
crypto pki trustpoint tftp-server
 enrollment url http://192.168.1.1:80
 revocation-check none

crypto pki authenticate tftp-server
crypto pki enrollment tftp-server
```

d. Obtaining a certificate for *sast1*:

```
crypto pki trustpoint sast1
 enrollment url http://192.168.1.1:80
 revocation-check none

crypto pki authenticate sast1
crypto pki enrollment sast1
```

e. Obtaining a certificate for *sast2*:

```
crypto pki trustpoint sast2
 enrollment url http://192.168.1.1:80
 revocation-check none

crypto pki authenticate sast2
crypto pki enrollment sast2
```

Step 4 Configure Telephony Service with the following steps:

a. Configure the trustpoint label used for secure signaling:

```
secure-signaling trustpoint cme-server
```

b. Configure the TFTP server credentials (trustpoint) used for signing the configuration files:

```
tftp-server-credentials trustpoint tftp-server
```

c. Configure the security mode for the endpoints

```
 server-security-mode secure
 device-security-mode authenticated
```

The *authenticated* option will instruct the device to establish a TLS connection with no encryption. In this mode, there is no SRTP in the media path.

The *encrypted* option will instruct the device to establish a encrypted TLS connection to secure Media path using SRTP.

Note Use the *authenticated* option until SRTP is supported in the future.

d. Configure the system to generate the phone configuration XML files for each endpoint:

```
cnf-file perphone
```

e. Configure any ephone. For example:

```
ephone 1
 device-security-mode authenticated
```

Step 5 Configure the CTL client on a local Cisco Unified Cisco Unified CME in order to create a CTL file containing a list of known, trusted certificates and tokens.

The CTL client can either be run on the same Cisco Unified CME router or another standalone router. Here is an example for a CTL client on a local Cisco Unified CME router:

```
ctl-client
 server capf 192.168.1.1 trustpoint capf-server
 server tftp 192.168.1.1 trustpoint tftp-server
 server cme 192.168.1.1 trustpoint cme-server
 sast1 trustpoint sast1
 sast2 trustpoint sast2
```

After you have configured all the info above, use the **regenerate** command to create the CTL file:

regenerate

Step 6 Configure the CAPF server:

```
capf-server
 port 3804
 auth-mode null-string
 cert-enroll-trustpoint laverda-ca password 1 1511021F07257A767B
 trustpoint-label capf-server
 source-addr 192.168.1.1
!
```

Troubleshooting and Debugging

Use the following commands for troubleshooting and debugging your secure SCCP signaling via TLS setup:

- **show ephone registered**
- **show ctl-client**
- **show capf-server sessions**
- **show capf-server auth-strings**
- **show capf-server summary**
- **debug ctl-client**
- **debug credentials**
- **debug capf-server all|messages|error|events**

Note For details about these diagnostic commands, see your specific Cisco Unified CallManager Express command reference. The following is an example:

http://www.cisco.com/en/US/docs/voice_ip_comm/cucme/command/reference/cme_cr.html

Cisco Unified CME Commonly Used Ports

Table 10-1 and Table 10-2 illustrate Cisco Unified CME commonly used ports.

Table 10-1 Commonly Used Ports for Voice on Cisco Unified CME

Protocol	Port	Usage
SCCP	TCP 2000	Call control for SCCP phones
SIP	TCP 5060	Call control for SIP endpoints
RTP	UDP 16384-32767	Media from Cisco Unified CME to H.323/SIP endpoint, including Cisco Unity Express
RTP	UDP 2000	Media from Cisco Unified CME to SCCP phone
H.225	TCP 1720	H.323 Call Setup
H.245	TCP 11000-65535	H.323 Call control, port assignment random
H.323 RAS	UDP 1718	GK Discovery
H.323 RAS	UDP 1719	GK Call Control
H.323 RAS	UDP 223.0.1.4	GK Multicast discovery
TLS	TCP 3804	CAPF Authentication Request
TLS	TCP 2443	Secure Call control for SCCP phones

Table 10-2 Commonly Used Ports for Data on Cisco Unified CME

Protocol	Port	Usage
DHCP	UDP 67	IP addressing for IP phones
HTTP	TCP 80	Cisco Unified CME GUI access, IP phone local directory access
HTTPS/SSL	TCP 443	Secure Cisco Unified CME GUI access
NTP	UDP 123	Time sync for Cisco Unity Express, IP Phones
Radius	UDP 1645	Authentication for Cisco Unified CME CLI/GUI users
Radius	UDP 1646	CDR accounting
SNMP	UDP 161	Traps for Cisco Unified CME monitoring
SSH	TCP 22	Secure Cisco Unified CME CLI access
Syslog	UDP 514	System monitoring, CDR accounting
Telnet	TCP 23	Cisco Unified CME CLI access