

Troubleshooting Cisco IP Telephony and Video, v1.0

1. When a caller dials 9 plus an external seven-digit number, the caller hears a fast-busy tone after a period of silence. What is causing the silence?

A. The caller dialed the wrong number.

B. The T302 timer is waiting to expire.

C. The caller does not have the PSTN partition in the CSS.

D. The gateway is not dropping the leading 9, and the PSTN fails.

E. To dial successfully, the caller must enter a Forced Authorization Code.

F. There is no dial route for 9XXXXXXX on Cisco Unified Communications Manager.

Answer(s): B

2. When viewing the configuration of Intercluster Lookup Service, this command is used to look up the URI of a remote cluster user:

A. SIP route pattern for domain*.acme.com

B. SIP trunk with destination cucm1.cisco.com

C. SIP route pattern for domain*cisco.com

D. end user with directory URI

Answer(s): A

3. During a conference that is hosted on a Cisco TelePresence Server, which three circumstances determine that some participants do not have video, but they do have audio? (Choose three.)

A. No video ports are available in a slave Cisco TelePresence Server configuration.

B. In all Cisco TelePresence deployments, some participants are connected only via audio.

C. The participants have no video component.

D. The maximum number of participants is exceeded.

E. No free screen licensing ports are available.

F. No video ports are available in a single Cisco TelePresence Server configuration.

Answer(s): A,E,F

4. You are integrating a new video endpoint with Cisco VCS, but you find that the unit is failing to register. You assign extension 4000 to the device in the "vc.cisco.com" SIP domain, and you set its registration proxy to the IP address of 10.1.1.10 as the Cisco VCS. In order for the device to register via SIP, which format must you use when you set the SIP address of the device?

A. 4000@cisco

B.

C.

D. 4000

Answer(s): C

5. Which CLI command is used to troubleshoot ILS network connection issues within the local Cisco Unified CM cluster to determine which server within the cluster is the xnode?

A. utils ils lookup

B. utils ils display xnode

C. utils ils showpeerinfo

D. utils ils find xnode

Answer(s): D

6. A customer reports trial calls that are made through the PSTN gateway drop after few seconds of being placed on mute. Which MGCP configuration command can you issue in the gateway to resolve this problem?

A. no mgcp rtp unreachable timeout

B. no mgcp timer receive-rtcp

C. mgcp explicit hookstate

D. mgcp max-walling-delay

Answer(s): A

7. Phone A using G.729 is trying to call phone B that only supports G.711, and the phone rings, but the call drops as soon as it is answered. Which cause of the problem is most likely?

A. No software or hardware transcoding resources are configured in Cisco Unified Communications Manager.

B. No Media Termination Point resource partition is set up on Cisco Unified CM.

C. Mismatch is in the partition and CSS configuration.

D. No conference bridge resource is configured in Cisco Unified CM.

Answer(s): A

8. Refer to topology and Exhibits below:

A. show mgcp connection

B. show mgcp registration

C. show mgcp-gw

D. show ccm-manager

Answer(s): D

9. Which debug command analyzes messages that are produced by SIP during the call setup process in IOS?

A. Show isdn status

B. debug voip ccapi inout

C. debug voice dialpeer

D. show sip-ua register status

E. debug isdn q931

F. debug ccsip messages

Answer(s): F

10. In a Cisco UCM multisite WAN with centralized call-processing deployment model, what redundancy feature should be configured on remote site routers to provide basic IP telephony services in the event of a WAN outage?

A. V3PN

B. SRST

C. AAR

D. CAC

Answer(s): B

11. While troubleshooting a transcoding issue, where you need 32 G.711 to G.729a sessions, you realize the DSP capacity may be undersized.

A. PVDM4-48

B. PVDM4-32

C. PVDM4-128

D. PVDM4-64

E. PVDM4-16

Answer(s): B,D

12. When users in headquarters call branch office users over the WAN link, branch users report poor audio quality. Headquarters users consistently experience acceptable audio quality. Which troubleshooting approach most directly improves the audio quality of the branch users?

A. Make the branch router configuration for LLQ match the headquarters router.

B. Make the branch router configuration for CBWFQ match the headquarters router.

C. Make the headquarters router configuration for CBWFQ match the branch router.

D. Make the headquarters router configuration for LLQ match the branch router.

Answer(s): D

13. An engineer is troubleshooting a video conferencing endpoint that cannot successfully register to the desired VCS-C. It is using an SRV record for DNS resolution. Which action must be taken to fix this issue?

A. Restart DHCP server.

B. Restart DNS server

C. Configure endpoint to public DHCP

D. Configure endpoint to use a public DHCP

Answer(s): B

14. Two phones in the same cluster and at the same site have a call currently connected. The site local H.323 PSTN gateway loses connection with Cisco Unified Communications Manager. Which two results do you expect? (Choose two.)

A. SRST is active, and all the phones enter SRST mode.

B. The current call is not disconnected.

C. Cisco Unified SRST is able to receive incoming calls.

D. The phones display "CM Fallback Service Operating."

E. No incoming and outgoing calls are possible.

Answer(s): B,E

15. Which two statements about Cisco Unified CM location bandwidth deduction are true? (Choose two.)

A. If a call uses G.729, Cisco Unified Communications Manager subtracts 24k.

B. If a call uses G.711, Cisco Unified Communications Manager subtracts 64k.

C. If a call uses G.729, Cisco Unified Communications Manager subtracts 16k.

D. If a call uses G.723, Cisco Unified Communications Manager subtracts 16k.

E. If a call uses G.711, Cisco Unified Communications Manager subtracts 80k.

Answer(s): A,E

16. Which action is recommended when troubleshooting reports of reports of dropped calls between Cisco Unified communications manager and a voice gateway?

A. If the problem seems to occur only through a certain gateway, enable tracing and/or view the Real-time monitoring Tool. The RTMT files give a cause of termination that may help determine the cause of the problem.

B. Check the syslog viewer in the call detail records for phone or gateway resets.

C. Check the syslog viewer in APIC-EM for phone or gateway resets.

D. If the problem seems to occur only through a certain gateway, enable tracing and/or view the call detail Records. The CDR files give a cause of termination that may help determine the cause of the problem.

Answer(s): D

17. Which value is the default value that a SIP Phone will send a keep-alive to Cisco Unified Communications Manager and where do you modify the timers?

A. Default value 30 sService ParametersSIP Profile

B. Default value 3600 sSIP Security ProfileSIP Profile

C. Default value 115 sSIP ProfileEnterprise Parameters

D. Default value 120 sService ParametersSIP Profile

Answer(s): D

18. When a remote endpoint dials in to join a conference that is configured on a Cisco TelePresence Server bridge, the endpoint receives only audio. Other users can successfully join the call with Voice and Video. What is causing this issue?

A. The endpoint is assigned a region without enough configured bandwidth for video.

B. The endpoint does not have the multisite option installed.

C. The bridge is not able to host video calls.

D. The endpoint does not have the partition of the bridge in its CSS.

E. The bridge is out of all licenses.

Answer(s): A

19. Users of your local Cisco Unified Communications Manager cluster report that they receive error "Login is unavailable (23)" when they try to log in to Extension Mobility. Which reason for this error is true?

A. User has no Extension Mobility profiles assigned.

B. The given user ID is not found in any of the remote clusters.

C. User provided the wrong UserID or PIN

D. Phone is not subscribed to Extension Mobility phone service.

Answer(s): B

20. An engineer is troubleshooting a video call issue where a call is failing with the error message "488 Not Acceptable Media" and cause code 125/"Out Of Bandwidth". The video calls are not failing back to the audio due to this bandwidth issue. Which action fixes this issue?

A. Mark the device pool of the video calling device with an option "Retry Video Call As Audio"

B. Mark the trunk of the video calling device with an option "Retry Video Call As Audio"

C. Mark the SIP profile of the video calling device with an option "Retry Video Call As Audio"

D. Mark the video calling device with an option "Retry Video Call As Audio"

Answer(s): D
