Question #1 *Topic 1*

Which two functionalities does Cisco Expressway provide in the Cisco Collaboration architecture? (Choose two.)

A. Survivable Remote Site Telephony functionality

B. MGCP gateway registration

C. secure business-to-business communications

D. customer interaction management services

E. secure firewall traversal for remote devices

Correct Answer: *CE*

Community vote distribution

CE (100%)

Panda_man 11 months, 2 weeks ago

Selected Answer: CE

C & E is correct

upvoted 2 times

Myare 11 months, 2 weeks ago

correct answer is Cand E

upvoted 2 times

Question #2 *Topic 1*

An engineer must extend the corporate phone system to mobile users connecting through the internet with their own devices. One requirement is to keep that as simple as possible for end users. Which infrastructure element achieves these goals?

A. Cisco Express Mobility

B. Cisco Expressway-C and Expressway-E

C. Cisco Unified Border Element

D. Cisco Unified Instant Messaging and Presence

Correct Answer: *B*

Community vote distribution

B (100%)

JWMcInSC Highly Voted 3 years, 6 months ago

Shouldn't this be "B" C and E expressways?

upvoted 14 times

BarryR Highly Voted 3 years, 5 months ago

Correct answer is B Expressway-C and Expressway-E

upvoted 12 times

Panda_man Most Recent 11 months, 2 weeks ago

Selected Answer: B

It's B

upvoted 2 times

OSJAY 1 year, 4 months ago

Right, Correct answer is (B) Expressway-C and Expressway-E. This answer is wrong in every dump I found.

upvoted 2 times

Georges 1 year, 11 months ago

Selected Answer: B

The answer should be B as the Expressway secure internet connection to the CUCM

upvoted 2 times

Vijay_ABI 2 years, 1 month ago

Expressway E and C should be the answer

upvoted 2 times

VG224 2 years, 2 months ago

This dump is old .. I got may be 30 - 40% questions from here

upvoted 2 times

ccienetrider 2 years, 5 months ago

Correct answer is B Expressway-C and Expressway-E

upvoted 3 times

MrAshour 3 years ago

Cisco Unified Border Element (CUBE) provides connectivity between the enterprise network and service provider network. Not through the internet as in question.

The Answer must be B. Cisco Expressway-C and Expressway-E

upvoted 5 times

Mli2604 3 years, 3 months ago

I thing this belongs to Mobile and Remote access - so answer is B

upvoted 3 times

khader09 3 years, 3 months ago

Answer is B

upvoted 3 times

rishik 3 years, 5 months ago

Correct answer should be B

upvoted 5 times

nassar1 3 years, 6 months ago

This should be B expressway.

upvoted 6 times

Question #3 | *Topic 1*

A customer wants a video conference with five Cisco TelePresence IX5000 Series systems. Which media resource is necessary in the design to fully utilize the immersive functions?

A. Cisco PVDM4-128

B. software conference bridge on Cisco Unified Communications Manager

C. Cisco Webex Meetings Server

D. Cisco Meeting Server

Correct Answer: *D*

Reference:

https://www.examtopics.com/discussions/cisco/view/23793-exam-350-801-topic-1-question-3-discussion/

https://www.freecram.net/question/Cisco.350-801.v2020-10-02.q36/a-customer-wants-a-video-conference-with-five-cisco-telepresence-ix5000-series-system- which-media-resour# https://www.coursehero.com/file/phl3l2j/A-customer-wants-a-video-conference-with-five-Cisco-TelePresence-IX5000-series/

Community vote distribution

D (100%)

Rads25 Highly Voted 4 years, 6 months ago

Should be D : Cisco Meeting server

upvoted 17 times

dtmfjeff Highly Voted 4 years, 3 months ago

where are these wrong answers coming from? Surely they arent wrong on the exam.

upvoted 5 times

Twitchy Most Recent 11 months, 3 weeks ago

How is everyone agreeing with cisco's crappy way of phrasing questions (I'm convinced purposefully) and that D is the correct answer. While CMS is certainly a conference bridge, which is a media resource, CMS is a specific technology that is also a conference bridge but when they say media resource nowhere in documentation does anyone ever call CMS a media resource, CMS is a media resource, Media resource is not CMS. Sorry for the rant\

upvoted 2 times

dhia1706 1 year, 5 months ago

CMS is correct answer so D

upvoted 2 times

Panda_man 1 year, 11 months ago

Selected Answer: D

D is correct

upvoted 2 times

mcbesy 2 years, 7 months ago

The question here is, "Which MEDIA RESOURCE is necessary in the design to fully utilize the immersive functions?" Is Cisco Meeting Server a Media Resource?

upvoted 1 times

stefanahk 2 years, 6 months ago

is cisco webex meeting server a resource lol

upvoted 1 times

bcandrap 2 years, 10 months ago

Selected Answer: D

should be CMS

upvoted 1 times

Bazant 3 years ago

D seems correct. Webex Meetings Server is EOS/EOL and has nothing to do with IX5000

upvoted 1 times

sergioax88 3 years, 5 months ago

is D the correct answer, the

b not becouse

upvoted 1 times

0n3 3 years, 11 months ago

D is correct. Webex Meeting Server is no longer sold.

upvoted 4 times

alejandro00 3 years, 6 months ago

neither is the IX5000

upvoted 3 times

emersitro 4 years, 1 month ago

The correct answer is D

upvoted 3 times

rishik 4 years, 5 months ago

D is correct

upvoted 5 times

BarryR 4 years, 5 months ago

Correct answer is D

upvoted 4 times

Question #4 *Topic 1*

An engineer is designing a load balancing solution for two Cisco Unified Border Element routers. The first router (cube1.ab.com) takes 60% of the calls and the second router (cube2.abc.com) takes 40% of the calls. Assume all DNS A records have been created. Which two SRV records are needed for a load balanced solution? (Choose two.)

A. _sip._udp.abc.com 60 IN SRV 2 60 5060 cube1.abc.com

B. _sip._udp.abc.com 60 IN SRV 60 1 5060 cube1.abc.com

C. _sip._udp.abc.com 60 IN SRV 1 40 5060 cube2.abc.com

D. _sip._udp.abc.com 60 IN SRV 3 60 5060 cube2.abc.com

E. _sip._udp.abc.com 60 IN SRV 1 60 5060 cube1.abc.com

Correct Answer: *CE*

Community vote distribution

CE (100%)

BrunoRangel Highly Voted 2 years, 5 months ago

C,E are correct:

- Service: The symbolic name of the desired service
- Proto: The transport protocol of the desired service; this is usually either Transmission Control Protocol (TCP) or User Datagram Protocol (UDP)
- Name: The domain name for which this record is valid, it ends in a dot
- TTL: Standard DNS time to live field
- Class: Standard DNS class field (this is always IN)
- Priority: The priority of the target host, lower value means more preferred
- Weight: A relative weight for records with the same priority, higher value means more preferred
- Port: The TCP or UDP port on which the service is to be found
- Target: The canonical hostname of the machine that provides the service, it ends in a dot

Same priority and diferent weight

upvoted 5 times

Panda_man Most Recent 11 months, 2 weeks ago

Selected Answer: CE

C and E - correct - load balancing means same priority so both have number 1

upvoted 3 times

Omitted 2 years, 5 months ago

CE

These answers are trying to be confusing by making the TTL field "60" only the number right before the port (5060) matters for weight

upvoted 3 times

htruesdale 2 years, 5 months ago

Thank you BrunoRangel!

upvoted 1 times

Question #5 *Topic 1*

Which two functions are provided by Cisco Expressway Series? (Choose two.)

A. interworking of SIP and H.323

B. endpoint registration

C. intercluster extension mobility

D. voice and video transcoding

E. voice and video conferencing

Correct Answer: *AB*

Community vote distribution

AB (100%)

Janu82 Highly Voted 4 years, 6 months ago

The correct answers are A and B.

upvoted 27 times

BarryR Highly Voted 4 years, 5 months ago

Answer is A and B

upvoted 12 times

Kabimas66 Most Recent 9 months, 2 weeks ago

Answer: A & B

upvoted 1 times

GCISystemIntegrator 1 year, 3 months ago

A and B are correct

upvoted 1 times

CiscoSailor 1 year, 6 months ago

Selected Answer: AB

I agree with A & B

upvoted 1 times

Panda_man 1 year, 11 months ago

If we're taking this as VCS than B is correct but EXPRESSWAY (which is the question) - we don't register devices to expressway. Therefore its A and E

upvoted 2 times

Krachowsky 1 year, 9 months ago

Man you can register devices to the expressway series since many years. Your knowledge about this is outdated.

upvoted 1 times

genarouaaan 2 years, 2 months ago

Selected Answer: AB

A and B are correct

upvoted 3 times

Azrael4d 2 years, 7 months ago

Selected Answer: AB

The correct answers are A and B.

upvoted 2 times

Piji 2 years, 11 months ago

Selected Answer: AB

The correct ansers are A and B.

upvoted 3 times

msully 3 years, 2 months ago

CLCOR 350-801 Official Cert Guide, p, 468

In August 2016, Cisco released version X8.9, which allowed for registrations directly to the Expressway Core.

Then:

There have been several more upgrades since X8.9, and the Cisco Expressway Edge can now support both H.323 and SIP registration. Proxied registration still supports only SIP.

upvoted 2 times

timmyz 3 years, 4 months ago

B is NOT the answer. Expressways do not support device registration only VCS do.

The answer is A and E

upvoted 3 times

Shankarmuthu22 2 years, 3 months ago

Cisco did a u turn on this one and expressways now support endpoint registration as well.

upvoted 4 times

abelcollab 4 years, 3 months ago

Answer is A & B.

Recent Exam has question has made a clearer choice for B as follows :

Endpoint Registration over the Internet

upvoted 8 times

rishik 4 years, 5 months ago

AB are correct ones

upvoted 9 times

Question #6 *Topic 1*

An incoming off-net call to a user fails. An engineer notices that the off-net call is G.711, but the phone accepts only G.729. Which media resource on a Cisco
Unified Border Element and Cisco Unified Communications Manager must the engineer configure to manage the codec negotiation?

A. transcoder

B. CFB

C. MOH

D. MTP

Correct Answer: *A*

Community vote distribution

A (100%)

genarouaaan 8 months ago

Selected Answer: A

A is correct.

upvoted 2 times

BhaiKyare 1 year, 10 months ago

Why it can't be MTP on the SIP Trunk ?

upvoted 1 times

somebotty 1 year, 10 months ago

Media Termination Points can only handle G.711. They can't handle G.729. An MTP is mostly for providing supplementary services like hold, transfer, etc. They can also allow connection between a SIP device that requires Early offer and one that requires delayed offer. The MTP transcoding capability is limited to a to u law and packet size.

From Cisco.com "The MTP accepts two full duplex G.711 Coder-Decoder (CODEC) stream connections. MTPs bridge the media streams between two connections. The streaming data received from the input stream on one connection is passed to the output stream on the other connection, and vice versa. In addition, the MTP trancodes A-law to Mu-law (and vice versa) and adjusts packet sizes as required by the two connections."

https://www.cisco.com/en/US/docs/voice_ip_comm/cucm/admin/3_1_2/ccmsys/a05mtp.html

upvoted 6 times

Question #7 *Topic 1*

Which Cisco Unified Communications Manager service parameter should be enabled disconnect a multiparty call when the call initiator hangs up?

A. Drop Ad Hoc Conference

B. H.225 Block Setup Destination

C. Block OffNet To OffNet Transfer

D. Enterprise Feature Access Code for Conference

Correct Answer: *A*

Reference:

https://www.cisco.com/c/en/us/td/docs/voice_ip_comm/cucm/admin/10_0_1/ccmsys/CUCM_BK_SE5FCFB6_00_cucm-system-guide-100/CUCM_BK_SE5FCFB6_00_cucm-system-guide-100_chapter_011000.html#CUCM_TK_DFC66444_00

Community vote distribution

A (100%)

Panda_man 11 months, 2 weeks ago

Selected Answer: A

A is correct

upvoted 2 times

VitthalT 2 years, 4 months ago

Cisco Unified Communications Manager Administration provides the clusterwide service parameter, Drop Ad Hoc Conference, to allow the prevention of toll fraud (where an internal conference controller disconnects from the conference while outside callers remain connected). The service parameter settings specify conditions under which an ad hoc conference gets dropped.

upvoted 3 times

Question #8 *Topic 1*

A network administrator deleted a user from the LDAP directory of a company. The end user shows as Inactive LDAP Synchronized User in Cisco Unified
Communications Manager. Which step is next to remove this user from Cisco Unified Communications Manager?

A. Delete the user directly from Cisco Unified Communications Manager

B. Restart the Dirsync service after the user is deleted from LDAP directory.

C. Execute a manual sync to refresh the local database and delete the end user.

D. Wait 24 hours for the garbage collector to remove the user.

Correct Answer: *D*

Community vote distribution

D (100%)

micbosh Highly Voted 5 years ago

correct answer is D,

upvoted 18 times

Mert_kerna Most Recent 9 months ago

Selected Answer: D

D is correct. Sure, other things will remove the user, of course, but if there's a garbage collector that removes the user automatically by design, why would you NEED to remove it manually? If you select a different answer, you need to ask yourself, "why"? The question asks which step is next to remove the user. It doesn't ask, "how do you bypass the garbage collector before it automatically deletes the user?".

upvoted 3 times

b3532e4 9 months, 1 week ago

Correct answer : D
Base on E-Book CiscoPress-CCNP-and-CCIE-Collaboration-Core-CLCOR-350-801-Official-Cert-Guide

After the synchronization is completed, any LDAP synchronized accounts that were
not set to active are permanently deleted from the Cisco Unified Communications
Manager when the garbage collection process runs. Garbage collection is a process
that runs automatically at the fixed time of 3:15 a.m., and it is not configurable. However,
garbage collection will not delete any inactive accounts until they have been inactive
for at least 24 hours.

upvoted 2 times

ademozipek 1 year, 10 months ago

Selected Answer: D

Answer is D

upvoted 2 times

CiscoSailor 1 year, 12 months ago

Selected Answer: D

It's D. Restarting the service will not delete the users.

upvoted 2 times

MeowthL 2 years, 3 months ago

Selected Answer: D

Correct answer is D

upvoted 2 times

Panda_man 2 years, 5 months ago

Answer is D

upvoted 2 times

genarouaaan 2 years, 8 months ago

I am aggee Nacho721 with D is the answer. If a user is deleted and the LDAP sync runs, it will switch over to inactive, but won't leave the system until 24 hours have passed.
upvoted 1 times

Nacho721 3 years ago
D is the answer. If a user is deleted and the LDAP sync runs, it will switch over to inactive, but won't leave the system until 24 hours have passed.
upvoted 3 times

Georges 3 years, 5 months ago
Selected Answer: D
It should be D as the garbage collector will delete it 24hrs after.
upvoted 3 times

movalleuu 3 years, 8 months ago
Answer should be C, whenever you run the manual synch, all data from LDAP will be refreshed and there is no need to wait 24 hours since the user was already deleted
upvoted 1 times

Omitted 3 years, 5 months ago
Running the sync just makes the user show as inactive. You need to wait til garbage collection process runs for them to be deleted
upvoted 2 times

VitthalT 3 years, 10 months ago
D should be correct answer
upvoted 2 times

mbisi 3 years, 11 months ago
As long as the user is already INACTIVE, then obviously in 24hours (or less depending on when they started being inactive) they'll be deleted
upvoted 1 times

pasangawa 4 years ago
Answer is D. Garbage collector only deletes inactive account for more than 24hrs even if it runs at 3:15am.
upvoted 2 times

Ste1233 4 years, 2 months ago
Why would you restart the Dirsync to remove a user ? its correct it will remove it but would you?
upvoted 1 times

PunKike 4 years, 4 months ago
The documentation is clear, the answer is D
upvoted 1 times

parizek 4 years, 4 months ago
D, for sure
upvoted 2 times

Question #9 *Topic 1*

A customer has Cisco Unity Connections that is integrated with LDAP. As a Unity Connection administrator, you have received a request to change the first name for VM user. Where must the change be performed?

A. Cisco Unity Connection

B. Cisco Unified Communications Manager end user

C. Active Directory

D. Cisco IM and Presence

Correct Answer: *C*

Community vote distribution

C (80%) B (20%)

b3532e4 9 months, 1 week ago

Correct Answer:

C. Active Directory

This function is discussed in Chapter 16 of the Cisco Press guide on Page 376. It explains that in a Cisco Unity Connection integrated with LDAP, the user data imported from LDAP, including the first name, must be modified within the LDAP directory, such as Active Directory, and not in Cisco Unity Connection or CUCM directly.

upvoted 1 times

JoeC716 1 year, 1 month ago

Selected Answer: C

If it's integrated with LDAP, You're going through AD - It's C

upvoted 1 times

CiscoSailor 1 year, 12 months ago

Selected Answer: C

It is C. If it was coming from CUCM it would be AXL not LDAP.

upvoted 1 times

pacciolli 2 years, 1 month ago

Selected Answer: C

if VM is integrated with LDAP all de user information comes from the active directory

upvoted 2 times

Littlelarry123 2 years, 1 month ago

Selected Answer: B

I feel that its B. active directory have nothing to do with it

upvoted 1 times

driz 2 years, 1 month ago

If CUC is integrated with LDAP then the first name can only be changed on the LDAP system. Active Directory is the only choice that is an LDAP system.

upvoted 2 times

Littlelarry123 2 years, 1 month ago

Im wrong my bad g

upvoted 1 times

Question #10 *Topic 1*

Which configuration step is necessary for a Cisco SIP phone to synchronize its time with a specific source?

A. Add a Phone NTP Reference to the Date/Time Group.

B. Assign the device to the correct region.

C. Change the Time Format from 24-hour to 12-hour.

D. Change the Time Zone from ג€America/Los_Angeles" to ג€Etc/GMT+8".

Correct Answer: *A*

Reference:

https://www.cisco.com/c/en/us/td/docs/voice_ip_comm/cucm/admin/10_0_1/ccmcfg/CUCM_BK_C95ABA82_00_admin-guide-100/CUCM_BK_C95ABA82_00_admin-guide-100_chapter_0110.html

Community vote distribution

A (100%)

Mert_kerna 9 months ago

Selected Answer: A

Anything other than A is a situational preference, or just plain wrong.

If you're in the east coast of the US, why would it be a requirement to set your time zone to LA?

It doesn't matter whether you're using 12 or 24 hour format.

Assigning a device to a region doesn't change the time source for that device - if anything that's handled by Device Mobility.

-

The only sensible solution is to add the Phone NTP source to the Date/Time group.

upvoted 1 times

Panda_man 2 years, 5 months ago

Selected Answer: A

Answer is A

upvoted 1 times

Question #11 *Topic 1*

After an engineer runs the utils ntp status command on the Cisco Unified Communications Manager publisher, the stratum value is 16. Which issue can the
Cisco Unified CM cluster experience?

A. Unified CM sends an NTPv4 packet.

B. Database replication is not synchronized on the Unified CM nodes.

C. The cluster loses access to port 124 at the firewall.

D. The date/time group on all phones defaults to the time zone of the engineer.

Correct Answer: *B*

Community vote distribution

B (100%)

mzougari Highly Voted 4 years, 10 months ago

Correct answer is B:
16 is a invalid stratum value that means "this server is not considered a time provider". The stratum can be invalid for various reasons, the most common of which is that the "time provider not synchronized", the "configured source does not exist", or the "ntp server not running".
CUCM can communicate with the NTP server on Port 123

upvoted 8 times

b3532e4 Most Recent 9 months, 1 week ago

Correct Answer:

B. Database replication is not synchronized on the Unified CM nodes
Page Reference :
This issue is discussed in Chapter 14 of the Cisco Press guide on Pages 337-338. A stratum value of 16 indicates that the Cisco Unified CM cannot reach the NTP server, causing synchronization issues between the publisher and subscriber nodes. One of the critical issues that can arise is database replication not being synchronized across the cluster.

upvoted 1 times

pacciolli 2 years, 1 month ago

Selected Answer: B

B is correct

upvoted 1 times

Panda_man 2 years, 5 months ago

Selected Answer: B

B is correct

upvoted 2 times

ratbat 4 years, 7 months ago

B is the answer: DB synchronization would fail

upvoted 4 times

Grebec94 5 years ago

Your ntp test will fail at level 5, recommended to have is 3 or higher

upvoted 1 times

Question #12 *Topic 1*

When a new SIP phone is registered to Cisco Unified Communications Manager, it keeps failing and showing an `unprovisioned` error message in the phone display. Which problem is a possible cause of this issue?

A. Auto-registration is disabled on the Cisco Unified Communications Manager nodes and the phone device does not have a DN configured.

B. The DN assigned to the phone is already in use by another SIP phone.

C. The phone cannot download and install the latest firmware.

D. The DHCP settings are set incorrectly and the phone does not have an alternate TFTP defined.

E. The DN configuration for this phone is shared with an SCCP phone, which is not supported.

Correct Answer: *A*

Community vote distribution

A (100%)

virtu Highly Voted 4 years, 5 months ago

The correct Answer is A. I tested in my office, and 7821 without configured DN, is displaying "Unprovisioned". Auto-reg is disabled also.

upvoted 19 times

b3532e4 Most Recent 9 months, 1 week ago

Correct Answer:

A. Auto-registration is disabled on the Cisco Unified Communications Manager nodes, and the phone device does not have a DN configured.
Page Reference (English):
This issue is discussed in Chapter 17 of the Cisco Press guide on Page 399. The "unprovisioned" error message often occurs when auto-registration is disabled in Cisco Unified Communications Manager and the device does not have a directory number (DN) configured.

upvoted 1 times

Karthik_91 1 year, 1 month ago

Selected Answer: A

A is the correct ansewer.

upvoted 1 times

Kabimas66 1 year, 3 months ago

The question starts with " When a new SIP phone is registered to Cisco Unified Communications Manager"..........so phone was already registered. I think "unprovisioned" message comes when phone is unable to install last firmware . Answer could be: C

upvoted 1 times

martshep 1 year, 4 months ago

Selected Answer: A

Tested with 8861 and auto reg disabled. Assign the phone but don't put a DN on it, unprovisioned shown on phone.

upvoted 2 times

CiscoSailor 1 year, 12 months ago

Selected Answer: A

I agree with A

upvoted 2 times

paccIolli 2 years, 1 month ago

Selected Answer: A

A is correct

upvoted 1 times

MeowthL 2 years, 3 months ago

Selected Answer: A

The Answer is A

upvoted 2 times

Azrael4d 3 years, 1 month ago

Selected Answer: A

The correct Answer is A

upvoted 3 times

Slushed 3 years, 2 months ago

Selected Answer: A

Answer is A. A SIP-configured phone without a DN configured on Line 1 will remain unprovisioned.

upvoted 2 times

Brant 3 years, 9 months ago

Should be A.

upvoted 2 times

VitthalT 3 years, 10 months ago

Answer should be A

upvoted 1 times

Thandabantu380 4 years, 1 month ago

The possible answer should be A, Unprovisioned error is mostly referring to DN, nothing to do with TFTP or firmware

upvoted 2 times

IceSmack3r 4 years, 8 months ago

Answer is A.

upvoted 2 times

Testy1 4 years, 8 months ago

D is the correct answer. ...which problem is a possible cause of the issue... The phone having the wrong TFTP will cause the issue.

upvoted 1 times

Mtdaw 4 years, 8 months ago

In this answer alternate TFTP is mentioned not primary

upvoted 3 times

Jetnor 4 years, 11 months ago

Phone does not need latest firmware to register with cucm.

I searched some topics and found info that SIP phones require DN to register with cucm.

so it should be A.

upvoted 2 times

rishik 4 years, 11 months ago

Correct is C . The phone is already configured but got getting registered, so not auto-registration issue

upvoted 1 times

rishik 4 years, 11 months ago

Update on further reading, it should be A

upvoted 4 times

SDLOA14 4 years, 4 months ago

I believe auto-registration is mentioned in the question because with auto-registration the phone would be provided with a DN.

upvoted 1 times

Question #13 *Topic 1*

Which DHCP option must be set up for new phones to obtain the TFTP server IP address?

A. option 15

B. option 6

C. option 66

D. option 120

Correct Answer: *C*

Community vote distribution

C (100%)

Griswald Highly Voted 5 years ago

DHCP Option 150 is Cisco proprietary. Option 66 is an IEEE standard.

upvoted 18 times

MrAshour 4 years, 6 months ago

Thank you for examination

upvoted 3 times

b3532e4 Most Recent 9 months, 1 week ago

Correct Answer:

C. option 66

Page Reference (English):

This topic is covered in Chapter 9 of the Cisco Press guide on Page 205. Option 66 is used for provisioning the TFTP server address on Cisco phones. While Option 150 can also be used in Cisco networks, Option 66 is a more generic option for providing TFTP server information in non-Cisco environments.

upvoted 1 times

JoeC716 1 year, 1 month ago

Selected Answer: C

Execute Order 66

upvoted 1 times

Mert_kerna 2 years, 7 months ago

Selected Answer: C

The only reason option 66 is the answer is because option 150 isn't included in the list of answers.

upvoted 2 times

VitthalT 3 years, 10 months ago

It should be Option 150 for DHCP

upvoted 1 times

hhn53 3 years, 10 months ago

I was looking for option 150

upvoted 1 times

somedudebob 4 years, 2 months ago

Has anyone actually got this question in the exam? Is 15 a typo and should be 150? Because if 150 was an option I would probably select it over 66, as it provides a redundant option...

upvoted 1 times

DEFAULTNERD 4 years, 6 months ago

66 is correct but 150 is better as it has a redundant option. Unless you use dns than you can use multiple.

upvoted 2 times

Question #14 *Topic 1*

Which two conditions must a user meet to provision a new device using the Self-Provisioning feature? (Choose two.)

A. The user must have a primary extension.

B. At least two DNs must be assigned to the user device.

C. The user must be part of ג€Standard CCM Super Userג€.

D. The user must have the appropriate universal device template linked to the user profile.

E. The user must have at least one user device profile assigned.

Correct Answer: *AD*

Community vote distribution

AD (100%)

Griswald Highly Voted 4 years, 11 months ago

The user must have a primary extension.

The user must have the appropriate universal device template linked to the User Profile.

The total number of owned devices must be less than the Self-Provisioning limit that is specified on the associated User Profile.

upvoted 19 times

XalaGyan 4 years, 6 months ago

wonderfully said bro

upvoted 5 times

b3532e4 Most Recent 9 months, 1 week ago

Correct Answers:

A. The user must have a primary extension.
D. The user must have the appropriate universal device template linked to the user profile.
Page Reference (English):
This configuration is discussed in Chapter 17 of the Cisco Press guide on Pages 396-399. It explains that for self-provisioning, the user must have a primary extension and be associated with a user profile that includes the universal line and device templates.

upvoted 1 times

pcp84 1 year, 4 months ago

Selected Answer: AD

As others have stated, AD.
Just putting in a voting comment so it shows on the solution.

upvoted 1 times

VG224 3 years, 9 months ago

AD is correct

upvoted 1 times

devadarshan91730 3 years, 11 months ago

Before your end users can use self-provisioning, your end users be configured with the following items:
• Your end users must have a primary extension.
• Your end users must be associated to a user profile or feature group template that includes a universal
line template, universal device template. The user profile must be enabled for self-provisioning.

upvoted 3 times

Question #15 *Topic 1*

How many DNS SRV entries can be defined in the SIP trunk destination address field in Cisco Unified Communications Manager?

A. 1

B. 8

C. 16

D. 4

Correct Answer: *A*

Community vote distribution

A (100%)

nassar1 Highly Voted 5 years ago

Should be 1 SRV record

upvoted 20 times

Rads25 Highly Voted 5 years ago

https://www.cisco.com/c/en/us/td/docs/voice_ip_comm/cucm/admin/11_5_1/sysConfig/CUCM_BK_SE5DAF88_00_cucm-system-configuration-guide-1151/CUCM_BK_SE5DAF88_00_cucm-system-configuration-guide-1151_chapter_01110.html:

"You can assign up to 16 different destination addresses for a SIP trunk, using IPv4 or IPv6 addressing, fully qualified domain names, or you can use a single DNS SRV record."

upvoted 19 times

DaKenjee 2 years, 7 months ago

Answer is A

You can provide 16x entries, like IPv4 address or fully qualified domain name in SIP-Trunk > SIP Information > Destination

but as soon as you activate the >Destination Address is an SRV< check box, only one DNS-SRV entry is allowed to be configured, all other 2-16 destinations get grayed out

upvoted 2 times

b3532e4 Most Recent 9 months, 1 week ago

You can assign up to 16 different destination addresses for a SIP trunk, using IPv4 or IPv6 addressing, fully qualified domain names, or you can use a single DNS SRV record.

Answer: 16 C

upvoted 1 times

b3532e4 10 months ago

A Is correct

You can assign up to 16 different destination addresses for a SIP trunk, using IPv4 or IPv6 addressing, fully qualified domain names, or you can use a single DNS SRV record

upvoted 1 times

AbdurrahmanBNC 11 months, 4 weeks ago

I think you don't understand document which you give us link. It is written that "You can assign up to 16 different destination addresses for a SIP trunk, using IPv4 or IPv6 addressing, fully qualified domain names, or you can use a single DNS SRV record.". According to you send document right answer is Option A

upvoted 1 times

ademozipek 1 year, 10 months ago

Selected Answer: A

Answer is A

upvoted 1 times

JWMcInSC 2 years ago

Per the Cisco Sys Admin doc for v11.5:

You can assign up to 16 different destination addresses for a SIP trunk, using IPv4 or IPv6 addressing, fully qualified domain names, or you can use a single DNS SRV record.
upvoted 2 times

Panda_man 2 years, 5 months ago
Selected Answer: A
Answer is 1
upvoted 1 times

Mert_kerna 2 years, 7 months ago
Selected Answer: A
1 SRV record is allowed. Multiple FQDNs are allowed. CUCM, within SIP Trunk Configuration, literally grays out the capability to add more than one record if the checkbox to use a SRV record is set - At least on 12.x, which is what this exam is based off of.
upvoted 1 times

BigD2819 3 years, 7 months ago
Selected Answer: A
Should be 1 SRV record
upvoted 3 times

dave70 3 years, 7 months ago
the correct answer is A, because You can assign up to 16 different destination addresses for a SIP trunk, using IPv4 or IPv6 addressing, fully qualified domain names, or you can use a single DNS SRV record.

https://www.cisco.com/c/en/us/td/docs/voice_ip_comm/cucm/admin/11_5_1/sysConfig/CUCM_BK_SE5DAF88_00_cucm-system-configuration-guide-1151/CUCM_BK_SE5DAF88_00_cucm-system-configuration-guide-1151_chapter_01110.html
upvoted 3 times

Grolos 3 years, 8 months ago
For all people here who are confused, just do as follow: Open up CUCM, go the trunk page,click on any trunk that you may have, and check 'Destination Address is an SRV', YOU CANNOT ADD any other destination more than a single one. If you UNCHECK then you can add 16 >>IP's<< NOT SRV's. This is a fact and cannot be argued. The correct answer is A.
upvoted 2 times

CollabGuru 3 years, 10 months ago
You can assign up to 16 different destination addresses for a SIP trunk, using IPv4 or IPv6 addressing, fully qualified domain names, or you can use a single DNS SRV record
upvoted 2 times

xxCRONOxx 3 years, 10 months ago
This is another "lets me tricky jerks" question from Cisco. Per the documentation.. "You can assign up to 16 different destination addresses for a SIP trunk, using IPv4 or IPv6 addressing, fully qualified domain names, or you can use a single DNS SRV record"

Therefore i'll agree it's a you can use 16 destination IP addresses but only 1 DNS SRV entry.
upvoted 1 times

davidanibalmarcelino 4 years, 2 months ago
Should be C.
the question says "...entries can be defined.."
upvoted 1 times

> **davidanibalmarcelino** 4 years, 2 months ago
> After reading more it should be A
> You can assign up to 16 different destination addresses for a SIP trunk, using IPv4 or IPv6 addressing, fully qualified domain names, or you can use a single DNS SRV record.
> Also when checking off Destination address is an SRV under trunk configuration, you get a message "When the first destination is configured and saved as an SRV all other destinations on this trunk will be deleted"
> upvoted 3 times

Eolin 4 years, 7 months ago
the question say "..entries can be defined..", so you can define/configure/add until 16 entries. so, the correct answer is C.
upvoted 2 times

> **virtu** 4 years, 5 months ago

Correct answer is A, in a SIP trunk destination address, SRV record can be only one, you cant configure more than one record. Because SRV record it self can be pointed to "many" records.

upvoted 3 times

jaicar2509 4 years, 10 months ago

Reading further, 16 can be added but only one can be used. SRV records can provide redundancy.

upvoted 3 times

Question #16 — *Topic 1*

On which Cisco Unified Communications Manager nodes can the TFTP service be enabled?

A. any node

B. any two nodes

C. only nodes that have Cisco Unified CM service enabled

D. any subscriber nodes

Correct Answer: *A*

Community vote distribution

A (100%)

infamous476 Highly Voted 4 years, 11 months ago

Given the question, I would say the correct answer is A. The TFTP service can be enabled on any node. A phone can only point to (2) TFTP servers in it's settings, but that is not what this question is asking.

upvoted 19 times

Cllbrtr Highly Voted 5 years ago

The TFTP Service itself can be enabled on any node regardless of the node's purpose.

upvoted 9 times

Nila1 4 years, 2 months ago

Any node, I have this in my company cluster setup

upvoted 4 times

b3532e4 Most Recent 10 months ago

B. any two nodes

upvoted 1 times

Littlelarry123 1 year, 9 months ago

Why not C can anyone explain please?

upvoted 1 times

spag22500 2 years, 6 months ago

correct is A: any node

Find in study guide:
Depending on the number of devices that a server supports, you can run
the TFTP service on a dedicated server, on the database publisher server, or on any other
server in the cluster.

upvoted 2 times

Abdulkader2022 2 years, 7 months ago

correct Answer A

upvoted 1 times

H31d1 2 years, 7 months ago

Selected Answer: A

I designed this in various ways

upvoted 2 times

DaKenjee 2 years, 7 months ago

Selected Answer: A

(1) No Cisco Guide is explaining an active Cisco Unified CM service as mandatory for TFTP
(2) i checked it on one of our customers, it's not necessary.

We use two CUCM Subs only for CTI, TFTP, MTP, CFB, no Callmanager Service is active there

upvoted 4 times

jezg1976 3 years, 2 months ago

A.

From the SRND:

The Cisco TFTP service that provides this functionality can be enabled on any server node in the cluster. However, in a cluster with more than 1250 users, other services might be impacted by configuration changes that can cause the TFTP service to regenerate configuration files. Therefore, Cisco recommends that you dedicate a specific subscriber node to the TFTP service, as shown in Figure 9-1, for a cluster with more than 1250 users or any features that cause frequent configuration changes.

upvoted 3 times

MKZ 4 years ago

2 TFTP servers per cluster

upvoted 1 times

MKZ 4 years ago

It is possible to configure any number of remote clusters on the primary TFTP server; however, each remote cluster may contain only up to 3 TFTP IP addresses. The recommended design for redundancy is 2 TFTP servers per cluster, and thus 2 IP addresses per remote cluster on the Primary TFTP server for redundancy.

upvoted 1 times

Doys 4 years ago

correct answer is A ....B is incorrect,....C is wrong as running TFTP service with UCM service is NOT recommended for small deployments but rather [PUB/TFTP] ....D is wrong as any subscriber node implies any node BUT the Publisher which is incorrect.....BEST practices and Cisco recommendation has always been, deployments > 1200 endpoints to have a dedicated TFTP node otherwise Pub/TFTP same node. ....so the test answer is absolutely wrong. this is clearly spelled out in the SRND ....

upvoted 1 times

Chandanake 4 years, 3 months ago

The Cisco TFTP service that provides this functionality can be enabled on any server node in the cluster

https://www.cisco.com/c/en/us/td/docs/voice_ip_comm/cucm/srnd/collab11/collab11/callpros.html

upvoted 2 times

Tomilee 3 years, 4 months ago

Cisco recommends according to this link The Cisco TFTP service that provides this functionality can be enabled on any server node in the cluster. However, in a cluster with more than 1250 users, other services might be impacted by configuration changes that can cause the TFTP service to regenerate configuration files. Therefore, Cisco recommends that you dedicate a specific subscriber node to the TFTP service

upvoted 1 times

DEFAULTNERD 4 years, 6 months ago

It can be any node or dedicated node

upvoted 2 times

Ron_Berserker 4 years, 8 months ago

A is correct. The Cisco TFTP service can be enabled on any server node in the cluster.

B is not correct because a cluster can have more than two nodes not only two, C doesn't make any sense since Cisco Unified CM is a server, not a service and D is just a recommendation from Cisco so it is not the best choice.

Reference: https://www.cisco.com/c/en/us/td/docs/voice_ip_comm/cucm/srnd/collab11/collab11/callpros.html

upvoted 4 times

IceSmack3r 4 years, 8 months ago

Cisco recommends a dedicated publisher to prevent administrative operations from affecting the telephony services. A dedicated publisher does not provide call processing or TFTP services running on the node. Instead, other subscriber nodes within the cluster provide these services.

upvoted 1 times

jtg 4 years, 8 months ago

If there are any Cisco moderators out there, this is the type of question that drives me nuts. There are no restrictions as to how many servers in the cluster CAN run the TFTP service. All of the Cisco design guides indicate that TFTP SHOULD be run on two servers for redundancy purposes. I get the feeling Cisco wants you to answer "any two nodes" with the design guide in mind. The question asks "can" and not "should" and so I would answer "any node". I didn't know that being a voice engineer required an advanced degree in English.

upvoted 7 times

Mr_Kokonut 4 years, 10 months ago

Answer A. On larger systems with more than 1250 users, Cisco recommends a dedicated publisher to prevent administrative operations from affecting the telephony services. A dedicated publisher does not provide call processing or TFTP services running on the node. Instead, other subscriber nodes within the cluster provide these services.

TFTP Redundancy: Cisco recommends deploying more than one dedicated TFTP subscriber node for a large Unified CM cluster, thus providing redundancy for TFTP services. While two TFTP subscribers are typically sufficient, more than two TFTP server nodes can be deployed in a cluster.
SRND: https://www.cisco.com/c/en/us/td/docs/voice_ip_comm/cucm/srnd/collab11/collab11/callpros.html#15081
upvoted 4 times

Question #17 *Topic 1*

Which issue causes slips on a PRI?

A. incorrect clock source

B. incorrect encapsulation

C. incorrectly configured time zone

D. change in the line code

Correct Answer: *A*

Community vote distribution

A (100%)

AgshinA Highly Voted 4 years, 3 months ago

A is correct. A router with multiple Primary Rate Interfaces (PRIs) can experience slip errors on one of the T1 PRIs. This happens when you have multiple clock sources.

upvoted 7 times

milkman33 3 years, 6 months ago

Agreed A is correct. I have seen this many times out in the field.

upvoted 1 times

pcp84 Most Recent 10 months, 2 weeks ago

Selected Answer: A

A is correct.

Just putting in a voting comment so it shows on the solution.

upvoted 2 times

Question #18 *Topic 1*

An administrator recently upgraded a Cisco Webex DX80 through its web interface but discovered the next morning that the unit has received to its previous version. What must the administrator do to prevent this from happening again?

A. Assign a phone security profile with secure SIP.

B. Set the prepare cluster for rollback to pre-8.0 enterprise parameter to true.

C. Confirm the phone load name in the phone configuration.

D. Assign a universal device template to the phone.

Correct Answer: *C*

jancok Highly Voted 8 months, 3 weeks ago

The answer is C

upvoted 6 times

francknoel Most Recent 5 months, 1 week ago

Selected Answer: C

it's the only possible answer

upvoted 1 times

Question #19 *Topic 1*

An engineer is notified that the Cisco TelePresence MX800 that is registered in Cisco Unified Communications Manager shows an empty panel, and the Touch 10 shows a corresponding icon with no action when pressed. Where does the engineer go to remove the inactive custom panel?

A. The Software Upgrades page in CUCM OS Administration

B. The In-Room Control Editor on the webpage of the MX800

C. The phone configuration page in CUCM Administration

D. The SIP Trunk Security Profile page in CUCM Administration

Correct Answer: *B*

Community vote distribution

B (100%)

jorgeoscar90 Highly Voted 3 years ago

Should be B.

https://www.cisco.com/c/dam/en/us/td/docs/telepresence/endpoint/ce82/sx-mx-in-room-control-guide-ce82.pdf

https://www.cisco.com/c/dam/en/us/td/docs/telepresence/endpoint/ce82/sx-mx-in-room-control-guide-ce82.pdf

upvoted 23 times

> **jorgeoscar90** 3 years ago
>
> Perform the following steps to remove the in-room control panel and
> icon from Touch10:
> a. Launch the in-room control editor from the video system's
> web interface.
> b. Select the panel to be removed (Global, Homescreen or Incall)
> c. Click Recycle bin ( )in the lower right corner.
>
> upvoted 13 times

Rads25 Highly Voted 3 years ago

Should be B.

upvoted 9 times

usernamesarehard Most Recent 10 months ago

Selected Answer: B

Should be B

upvoted 1 times

Azrael4d 1 year, 1 month ago

Selected Answer: B

Should be B

upvoted 2 times

Georges 1 year, 6 months ago

Selected Answer: B

The answer is B

upvoted 4 times

fzfigueroa 1 year, 10 months ago

B is the only logical answer

upvoted 2 times

DEFAULTNERD 2 years, 6 months ago

Yeah B is the only Answer Why or how do they get so many wrong ?

upvoted 6 times

Question #20 | *Topic 1*

A presence redundancy group is deployed, and an engineer initiates a manual fallback. Which statement about Cisco Server Recovery Manager is true?

A. disconnects all users that had been failed over, and the users must log in again

B. disconnects all users that had been failed over

C. restarts critical services on the secondary node

D. restarts the Cisco Presence Engine

Correct Answer: *A*

Community vote distribution

A (75%) | B (25%)

jonycakes Highly Voted 3 years, 6 months ago

Thinking this should be A: Manual fallback—When you initiate a manual fallback, the Cisco Server Recovery Manager restarts critical services on the primary node and disconnects all users that had been failed over. Those users must then re-login to their assigned node.

Source: https://www.cisco.com/c/en/us/td/docs/voice_ip_comm/cucm/admin/11_0_1/sysConfig/CUCM_BK_C733E983_00_cucm-system-configuration-guide-1101/CUCM_BK_C733E983_00_cucm-system-configuration-guide-transformed_chapter_011010.html

upvoted 13 times

G0y0 Most Recent 3 months, 3 weeks ago

Selected Answer: A

After a failover, when the failed node comes online again, the clients automatically reconnect to the local IM and Presence Service node if you have configured automatic fallback. If you have not configured automatic fallback, you can manually initiate the fallback when the failed node comes online.

Now, when you initiate a manual fallback, the Cisco Server Recovery Manager restarts critical services on the primary node and disconnects all users that had been failed over. Those users must then re-login to their assigned node. So, answer is A.

Reference: Configure Redundancy and High Availability

https://www.cisco.com/c/en/us/td/docs/voice_ip_comm/cucm/im_presence/configAdminGuide/11_5_1/cup0_b_config-and-admin-guide-1151su5/cup0_b_imp-system-configuration-1151su5_chapter_0100.pdf

upvoted 1 times

G0y0 3 months, 3 weeks ago

B. is incomplete. If "Enable Automatic Fallback" is in False, you must perform manual fallback and we return to answer A. It "Enable Automatic Fallback" is True, then it makes that B. is totally incorrect.Whether if "Enable Automatic Fallback" is True or False, the users must re-login also whether automatically or manually.

upvoted 1 times

c6176b5 11 months, 2 weeks ago

Selected Answer: B

Did it many times, users must not relogin. Maybe a version dependancy?

upvoted 2 times

Rish 1 year, 6 months ago

A should be correct

upvoted 1 times

FG23 1 year, 8 months ago

Selected Answer: A

Answer is A

upvoted 1 times

Myare 1 year, 11 months ago

Correct answer is A:

Manual Failover, Fallback, and Recovery
Use Cisco Unified Communications Manager Administration to initiate a manual failover, fallback, and recovery for IM and Presence Service nodes in a presence redundancy group. You can also initiate these actions from Cisco Unified Communications Manager or IM and Presence Service using the CLI. See the
Command Line Interface Guide for Cisco Unified Communications Solutions for details.
• Manual failover: When you initiate a manual failover, the Cisco Server Recovery Manager stops the critical services on
upvoted 2 times

Mert_kerna 1 year, 11 months ago

Selected Answer: B

I believe B is the correct answer. The reason is that the default configuration for Manual Fallback is for the Jabber client to apply the re-log in upper and lower limits for the fallback. These values can be modified in CUCM at System > Service Parameters > Set the Server as the Imp Server and set the service as Cisco Server Recovery Manager.
Then look toward the bottom at Configure Re-Login limits.
The default value for Lower limit is 120 seconds.
The default value for Upper limit is 953 seconds.
This parameter prompts the Jabber client to log back into the services.
This works for both Manual and Automatic Fallback.
Automatic fallback is disabled by default, but can also be enabled by toggling true in the same service parameter as the Client Re-Login Upper and Lower Limit values.

I tested it and it works.
Also - Manually rehoming the user to between publisher and subscriber also automatically logs the user into that node's respective services.
upvoted 1 times

Mert_kerna 1 year, 11 months ago

Actually, my apologies. Just because the users don't have to "manually" log back into their primary node doesn't make B correct. A is the correct answer because the login is required, regardless of whether it's done automatically or not. This is a tricky one, but it's not B, the fell-back account doesn't simply stay disconnected. It uses the re-login setting within the Cisco Server Recovery Manager service parameter. Answer is A!!
upvoted 3 times

usernamesarehard 2 years, 4 months ago

Selected Answer: A

Should be A
upvoted 1 times

Azrael4d 2 years, 7 months ago

Selected Answer: A

The Answer is A
upvoted 4 times

mcbesy 2 years, 7 months ago

• Manual fallback: When you initiate a manual fallback, the Cisco Server Recovery Manager restarts
critical services on the primary node and disconnects all users that had been failed over. Those users
must then re-login to their assigned node.
upvoted 1 times

VG224 3 years, 2 months ago

B is correct , users doesn't have to re-login

User Failover Process
When a failover takes place(automatic or manual), the major point to remember is that the user account is not moved from one server to the other, but only the user session in Presence Engine is moved. In pre-10 versions of IM and Presence, the user assignment was moved from one server to the other. This user move was very expensive to server resources and added to the load that was on the server. In 10.X and later, the user stays homed on the server that they are assigned to, and the backend user session in the Presence Engine is moved from the failed node to the functional node. The user does not have to exit Jabber and re-log in when the change happens with Server Recovery Manager(SRM).
upvoted 2 times

CollabGuru 3 years, 4 months ago

• Manual fallback: When you initiate a manual fallback, the Cisco Server Recovery Manager restarts
critical services on the primary node and disconnects all users that had been failed over. Those users
must then re-login to their assigned node.

Answer found here:
https://www.cisco.com/c/en/us/td/docs/voice_ip_comm/cucm/im_presence/configAdminGuide/10_5_2/CUP0_BK_CEB3E82E_00_config-admin-guide-imp-1052/CUP0_BK_CEB3E82E_00_config-admin-guide-imp-1052_chapter_010000.pdf
upvoted 3 times

assemah 3 years, 4 months ago
Answer should be A "Manual fallback: When you initiate a manual fallback, the Cisco Server Recovery Manager restarts
critical services on the primary node and disconnects all users that had been failed over. Those users
must then re-login to their assigned node"
upvoted 2 times

Omitted 3 years, 5 months ago
Do the users really have to log in again? In my fallback experience they get disconnected then automatically find the primary server, takes about a minute.
upvoted 1 times

Omitted 3 years, 5 months ago
B is the best fit in my opinion.

I don't like A because it doesn't acknowledge that users shouldn't have to relog in.
"In 10.X and later...The user does not have to exit Jabber and re-log in when the change happens with Server Recovery Manager(SRM)."
https://www.cisco.com/c/en/us/support/docs/unified-communications/unified-communications-manager-im-presence-service/200958-IM-and-Presence-Server-High-Availability.html#anc7

It's not C bc this is only true for the primary node..
"Cisco Server Recovery Manager restarts critical services on the primary node and disconnects all users that had been failed over"
https://www.cisco.com/c/en/us/td/docs/voice_ip_comm/cucm/im_presence/configAdminGuide/11_5_1/cup0_b_config-and-admin-guide-1151su5/cup0_b_imp-system-configuration-1151su5_chapter_0100.html#task_6D63EA716B4BFA54377429B6C5D1E7C3
upvoted 2 times

WilliamC 3 years, 5 months ago
"the users(all)" is differente to "those users(users that had been failed over)" could be B. Tricky
upvoted 1 times

Komy 3 years, 6 months ago
Answer should be A
upvoted 1 times

Question #21 *Topic 1*

Which packet delay is the maximum supported between Cisco Unified Communications Manager nodes for clustering over WAN deployments?

A. 150 ms round tip

B. 510 ms round trip

C. 40 ms round trip

D. 80 ms round trip

Correct Answer: *D*

Reference:

https://www.cisco.com/c/en/us/td/docs/voice_ip_comm/cucm/srnd/collab11/collab11/callpros.html

Community vote distribution

D (100%)

domangez Highly Voted 3 years, 5 months ago

Cisco Collaboration System 12.x Solution Reference Network Designs (SRND):

The call processing service of Unified CM does support the deployment of a single cluster's call processing nodes across an IP WAN as long as the total end-to-end round-trip time between the nodes does not exceed 80 ms and an appropriate quantity of QoS-enabled bandwidth is provisioned. The maximum one-way delay between any two Unified CM servers should not exceed 40 ms, or 80 ms round-trip time.

https://www.cisco.com/c/en/us/td/docs/voice_ip_comm/cucm/srnd/collab12/collab12/models.html

upvoted 8 times

DaKenjee 2 years, 1 month ago

Answer D

Same here, only older Guide:

The maximum one-way delay between any two Unified CM servers should not exceed 40 msec, or 80 msec round-trip time

https://www.cisco.com/c/en/us/td/docs/voice_ip_comm/cucm/srnd/8x/uc8x/models.html#wp1044021

upvoted 2 times

pcp84 Most Recent 10 months, 2 weeks ago

Selected Answer: D

D is correct.

Just putting in a voting comment so it shows on the solution.

upvoted 1 times

a3c4c84 11 months, 2 weeks ago

Design Guidelines for Clustering over WAN Deployment Model

Two CUCM servers in a cluster must have a maximum round-trip delay of 80 ms between them

upvoted 1 times

Littlelarry123 1 year, 3 months ago

B. 510 ms round trip

This allows for communication between CUCM nodes across a wide area network with a delay of up to 510 milliseconds round trip time. This latency tolerance helps ensure that CUCM can function properly in geographically dispersed deployments

upvoted 1 times

Question #22 *Topic 1*

A user dials 9011841234567 to reach Vietnam. Which steps send the call to the PSTN provider as 011841234567?

A.

in the Called Party Transformation Pattern Configuration section,
configure the Pattern as 9.011841234567
configure the Discard Digits as Predot

B.

in the Calling Party Transformation Patterns section,
configure the Pattern as 9.011841234567
configure the Discard Digits as Predot 10-10-Dialing

C.

in the Called Party Transformation Pattern Configuration section,
configure the Pattern as 9.011841234567
configure the Discard Digits as Predot 10-10-Dialing

D.

in the Called Party Transformation Patterns section,
configure the Pattern as 9.011841234567
configure the Discard Digits as Predot

Correct Answer: *A*

MaxG Highly Voted 2 years ago

Correct Answer is D.

In Cisco Unified Communications Manager Administration, use the Call Routing > Transformation Pattern > Called Party Transformation Pattern menu path to configure called party transformation patterns.

https://www.cisco.com/c/en/us/td/docs/voice_ip_comm/cucm/admin/10_0_1/ccmcfg/CUCM_BK_C95ABA82_00_admin-guide-100/CUCM_BK_C95ABA82_00_admin-guide-100_chapter_0110010.html

upvoted 7 times

wwisp3422112 Highly Voted 2 years, 7 months ago

A here. Remember finding the documentation some days ago confirming A.

upvoted 5 times

Gary1968 Most Recent 4 months, 3 weeks ago

A.

Called Party Transformation Pattern contains Called Party Transformation Pattern Configuration

upvoted 2 times

decdca7 7 months, 2 weeks ago

D

https://www.cisco.com/c/en/us/td/docs/voice_ip_comm/cucm/admin/11_5_1/sysConfig/11_5_1_SU1/cucm_b_system-configuration-guide-1151su1/cucm_b_system-configuration-guide-1151su1_chapter_010111.html#CUCM_TK_CE4A835D_00

upvoted 2 times

Komy 1 year, 2 months ago

I would go with A:

Because in CUCM: , the page title is 'Called Party Transformation Pattern Configuration'

upvoted 2 times

c6176b5 1 year, 5 months ago

It is D!

https://www.cisco.com/c/en/us/td/docs/voice_ip_comm/cucm/admin/10_0_1/ccmcfg/CUCM_BK_C95ABA82_00_admin-guide-100/CUCM_BK_C95ABA82_00_admin-guide-100_chapter_0110010.html

upvoted 4 times

Reck12075 2 years ago

I would say A. If you go to add a new called party transformation pattern in CUCM, the page title is 'Called Party Transformation Pattern Configuration'

upvoted 5 times

Question #23 *Topic 1*

Where is the default Interregion Maximum Session Bit Rate for a region configured?

A. Service Parameter Configuration

B. Enterprise Phone Configuration

C. Enterprise Parameters Configuration

D. Region Configuration

Correct Answer: *A*

Community vote distribution

A (100%)

G0y0 3 months, 4 weeks ago

Selected Answer: A

yes, under the "Clusterwide Parameters (System - Location and Region" section, in Service Parameters, the aforementioned settings are thee.

upvoted 1 times

decdca7 7 months, 2 weeks ago

Selected Answer: A

It is talking about the CUCM default settings not the region to region settings

upvoted 1 times

b3532e4 10 months ago

Regions Overview

When you configure the maximum audio bit rate setting in the Region Configuration window (or use the service parameter in the Service Parameter Configuration window), this setting serves as a filter. When an audio codec is selected for a call, Unified Communications Manager takes the matching codecs from both sides of a call leg, filters out the codecs that exceed the configured maximum audio bit rate, and then picks the preferred codec among the codecs that are remaining in the list.

D is correct

upvoted 1 times

pcp84 1 year, 4 months ago

Selected Answer: A

A is correct.

Just putting in a voting comment so it shows on the solution.

upvoted 2 times

JOtac 2 years, 3 months ago

Yes , A is correct. There is service parameter to set the maximum audio intra and inter region

upvoted 1 times

Panda_man 2 years, 5 months ago

Selected Answer: A

A is correct

upvoted 1 times

Omitted 3 years, 1 month ago

Selected Answer: A

They are asking about the service parameter.

upvoted 3 times

Question #24 *Topic 1*

A Cisco TelePresence SX80 suddenly has issues displaying main video to a display over HDMI. Which command can you use from the SX80 admin CLI to check the video output status to the monitor?

A. xStatus Video Output

B. xCommand Video Status

C. xConfiguration Video Output

D. xStatus HDMI Output

Correct Answer: *A*

Community vote distribution

A (100%)

Vincentius Highly Voted 3 years, 1 month ago

Should be A, xStatus Video Output

https://www.cisco.com/c/dam/en/us/td/docs/telepresence/endpoint/sx-series/tc7/api-reference-guide/sx80-api-reference-guide-tc71.pdf

upvoted 22 times

DEFAULTNERD Highly Voted 2 years, 6 months ago

Who fixes the answers on this site. ? Please let me know .

upvoted 10 times

G0y0 Most Recent 3 months, 4 weeks ago

Selected Answer: A

it is an example with my MX300G2:

xstatus Video Output
*s Video Output Connector 1 Connected: True
*s Video Output Connector 1 MonitorRole: First
*s Video Output Connector 1 Resolution Height: 1080
*s Video Output Connector 1 Resolution RefreshRate: 60
*s Video Output Connector 1 Resolution Width: 1920
*s Video Output Connector 1 Type: LCD
*s Video Output Connector 2 Connected: False
*s Video Output Connector 2 ConnectedDevice Name: ""
*s Video Output Connector 2 ConnectedDevice PreferredFormat: ""
*s Video Output Connector 2 ConnectedDevice ScreenSize: -1
*s Video Output Connector 2 MonitorRole: Second
*s Video Output Connector 2 Resolution Height: 1080
*s Video Output Connector 2 Resolution RefreshRate: 60
*s Video Output Connector 2 Resolution Width: 1920
*s Video Output Connector 2 Type: HDMI
** end
OK
^C

upvoted 2 times

wwisp3422112 7 months ago

You can also use the command from answer C: "xConfiguration Video Output" to show the resolution, but you also get alot of other parameters.

A is correct here.

upvoted 1 times

usernamesarehard 10 months ago

Selected Answer: A

Should be A

upvoted 1 times

zzamchoi 11 months ago

Selected Answer: A

Correct answer is A

upvoted 2 times

[Removed] 1 year, 1 month ago

Selected Answer: A

can be easily tested, its A

upvoted 2 times

BarryR 2 years, 12 months ago

Correct answer is A

upvoted 6 times

Question #25 *Topic 1*

Which statement about Cisco Unified Communications Manager and Cisco IM and Presence backups is true?

A. Backups should be scheduled during off-peak hours to avoid system performance issues.

B. Backups are saved as .tar files and encrypted using the web administrator account.

C. Backups are saved as unencrypted.tar files.

D. Backups are not needed for subscriber Cisco Unified Communications Manager and Cisco IM and Presence servers.

Correct Answer: *A*
Reference:
https://www.cisco.com/c/en/us/td/docs/voice_ip_comm/cucm/admin/11_5_1_SU1/Administration/cucm_b_administration-guide-1151su1/cucm_b_administration-guide-1151su1_chapter_01010.html#CUCM_TK_S7FC26D5_00

Griswald Highly Voted 11 months, 1 week ago
TIP: Schedule backups during off-peak hours to avoid call-processing interruptions and impact to service.
upvoted 7 times

c956e37 Most Recent 1 week, 5 days ago
Selected Answer: A
the link attached to this question no longer works
upvoted 1 times

G0y0 3 months, 4 weeks ago
Selected Answer: A
B. is incorrect. Security Password is needed instead App or OS account.
C. incorrect. Backups are encrypted with the security password.
D. partially correct. You get a backup from all of your entire cluster. Is in the Restore where you can select what nodes to restore.
A. is correct. In peak hours you can get a backup in many hours cause the bandwidth consumption in peak hours. off-peak hours the backup, for example in a Large size cluster (a cluster of 9 nodes for example, in my experience), it could take 3 hours in a good WAN. The impact could be in the bandwidth, however, at level system I have not seen an impact in the performance of the cluster in the RTMT.
upvoted 1 times

Question #26 *Topic 1*

What is a software-based media resource that is provided by the Cisco IP Voice Media Streaming Application?

A. video conference bridge

B. auto-attendant

C. transcoder

D. annunciator

Correct Answer: *D*

Reference:

https://www.cisco.com/c/en/us/td/docs/voice_ip_comm/cucm/srnd/collab09/clb09/media.html

Community vote distribution

D (100%)

G0y0 3 months, 4 weeks ago

As described earlier in this chapter, CUCM offers software-based media resources. You must start the IP Voice Media Streaming Application Service to activate the following media resources:

■ Audio conferencing (G.711 only)

■ MTP

■ Annunciator

■ Video on hold (VOH)

■ Music on hold (MOH)

Implementing Cisco IP Telephony and Video, Part 1 (CIPTV1) Foundation Learning Guide

Chapter 10: Implementing Media Resources in Cisco Unified Communications Manager

upvoted 2 times

G0y0 3 months, 4 weeks ago

Answer: D.

upvoted 2 times

pcp84 10 months, 2 weeks ago

Selected Answer: D

D is correct.

Just putting in a voting comment so it shows on the solution.

upvoted 3 times

G0y0 3 months, 4 weeks ago

Read again, It is better to Upvote an existing comment IF you DON´T have anything to add.

upvoted 1 times

Ol_Mykhailiuk 2 years, 4 months ago

Selected Answer: D

An annunciator is a software function of the Cisco IP Voice Media Streaming Application that provides the ability to stream spoken messages or various call progress tones from the system to a user.

https://www.cisco.com/c/en/us/td/docs/voice_ip_comm/cucm/srnd/collab11/collab11/media.html#32774

upvoted 2 times

ocero 3 years, 8 months ago

The Cisco IP Voice Media Streaming Application provides the following software-based media resources:

Conference bridge

Music on Hold (MoH)

Annunciator

Media termination point (MTP)

Interactive Voice Response (IVR)

Probably a close answer would be A, but I think the most accurate is D
https://www.cisco.com/c/en/us/td/docs/voice_ip_comm/cucm/srnd/collab11/collab11/media.html
upvoted 4 times

mmollura 3 years, 6 months ago

I think the audio conference bridge is audio, not video. So A appears to be the right call.
upvoted 1 times

mmollura 3 years, 5 months ago

I meant D is the right call. Sorry about that.
upvoted 3 times

Question #27 *Topic 1*

When a user dials a number with a phone that is registered to the Cisco Unified Communications Manager, what is the default timeout before the number is sent?

A. 15 seconds

B. 5 seconds

C. 10 seconds

D. 3 seconds

Correct Answer: *A*

Community vote distribution

A (71%) C (29%)

examtopics_20 Highly Voted 5 years, 1 month ago

Should be C 15 second

upvoted 17 times

francesca Highly Voted 5 years ago

^update. Finding 15 sec is correct.

upvoted 11 times

b3532e4 Most Recent 9 months ago

When matching a variable-length pattern, the Cisco Unified Communications Manager does not know when dialing is complete, so it will wait for 15 seconds by default before processing the call. This post-dial delay can be reduced or eliminated as follows:

 Reduce the Cisco CallManager service parameter called the T302 Timer to allow earlier detection of the end of dialing. However, do not set this timer to less than 4 seconds, to prevent premature transmission of the call before the user finishes dialing.

 Configure a second route pattern, followed by the pound sign (#) wildcard, such as 9.011!# for North America or 0.00!# for Europe. Then educate users that they can indicate end of dialing by terminating the number with the # key. This action is analogous to pressing the Send button on a cell phone.

upvoted 1 times

JoeC716 1 year, 1 month ago

Selected Answer: A

Per the CCNP and CCIE Collaboration Core CLCOR 350-801 Official Cert Guide:

Chapter 19: Configuring Globalized Call Routing in Cisco Unified Communications Manager:

When matching a variable-length pattern, the Cisco Unified Communications Manager does not know when dialing is complete, so it will wait for 15 seconds by default before processing the call. A is the correct answer

upvoted 1 times

c6176b5 1 year, 5 months ago

Selected Answer: A

A 15s is correct

upvoted 2 times

ademozipek 1 year, 10 months ago

Selected Answer: A

15 is correct

upvoted 1 times

movalleuu 2 years, 3 months ago

T302 timer is set by default to 15 seconds

upvoted 1 times

Sergey 2 years, 4 months ago

Selected Answer: A

15 sec by default

upvoted 1 times

Panda_man 2 years, 5 months ago

default it 15 sec but it can be modified.

upvoted 1 times

Diego0169 2 years, 5 months ago

Selected Answer: A

15 sec is correct, this is choise A

upvoted 1 times

Mert_kerna 2 years, 5 months ago

In the current SRND, the CUCM Service Parameter default is set to 15000, which is 15 seconds.
Even when validating on the version of CUCM this test covers, hovering the mouse above the default T302 Timer value shows this javascript value, "javascript:getHelp('TimerT302_msec'), where msec = milliseconds. 15000 milliseconds to seconds converted is 15 seconds, respectively.

upvoted 1 times

Cujoka 2 years, 5 months ago

Selected Answer: A

15 sec is correct, this is choise A

upvoted 1 times

RdTx 2 years, 6 months ago

Selected Answer: A

15 seconds

upvoted 1 times

magdan 2 years, 7 months ago

Selected Answer: A

Now A is 15 second!

upvoted 2 times

DaKenjee 2 years, 7 months ago

Selected Answer: C

Answer C

This is T.302 Timer, no matter if SCCP or SIP

T302 Timer: Required Field This parameter specifies an interdigit timer for sending the SETUP ACK message.
The timer restarts each time Cisco CallManager receives a digit.
When this timer expires, Cisco CallManager routes the dialed digits.
This is a required field.
Default: 15000 ms
Minimum: 3000 ms
Maximum: 75000 ms

If the user dials digits but then does not press the Dial softkey or the # key,
the phone will wait for inter-digit timeout (15 seconds by default) before sending a SIP INVITE message to Unified CM.

https://www.cisco.com/c/en/us/td/docs/voice_ip_comm/cucm/srnd/collab10/collab10/dialplan.html

upvoted 1 times

usernamesarehard 2 years, 10 months ago

Selected Answer: C

Should be C

upvoted 1 times

stefanahk 3 years ago

Selected Answer: C

The valid delay is 15 seconds

upvoted 2 times

Question #28 *Topic 1*

An engineer deploys a Cisco Expressway-E server for a customer who wants to utilize all features on the server. Which feature does the engineer configure on the
Expressway-E?

A. H.323 endpoint registrations

B. VTC bridge

C. MRA

D. SIP gateway for PSTN providers

Correct Answer: *C*

Community vote distribution

C (100%)

MaxG 11 months ago

Selected Answer: C

Poorly worded question. I am back to thinking the Answer is C.

Both Expressway Core and Edge are always required. “All features” implies MRA, just because MRA is the overarching term with multiple capabilities underneath.

See About Mobile and Remote Access
https://www.cisco.com/c/en/us/td/docs/voice_ip_comm/expressway/config_guide/X12-5/exwy_b_mra-expressway-deployment-guide/exwy_b_mra-expressway-deployment-guide_chapter_00.html

upvoted 3 times

G0y0 2 years ago

what if H.323 registrations in Exp-E, whether or not there is Exp-C ?, answer A is more accurate. Furthermore there is nothing said about an Exp-C, so answer C is just speculative.

upvoted 1 times

G0y0 2 years ago

in addition, if MRA, so what about B2B calls? by my self, there is no reason to be answer C. If anyone has another interpretation of this question, please share.

upvoted 1 times

Omitted 2 years, 1 month ago

MRA isn't solely configured on expressway E. I know on newer versions of expressway you can register endpoints directly to the expressway E. Poorly worded question imo.

upvoted 2 times

Question #29 *Topic 1*

Which SNMP service must be activated manually on the Cisco UCM after installation?

A. Host Resources Agent

B. Cisco CallManager SNMP

C. Connection SNMP Agent

D. SNMP Master Agent

Correct Answer: *B*
SNMP Master Agent serves as the primary service for the MIB interface. You must manually activate Cisco CallManager SNMP service; all other SNMP services should be running after installation.
Reference:
https://www.cisco.com/c/en/us/td/docs/voice_ip_comm/cucm/service/9_0/admin/CUCM_BK_C136FE37_00_cisco-unified-serviceability-administration-90/CUCM_BK_C136FE37_00_cisco-unified-serviceability-administration-guide_chapter_0101.html

Community vote distribution
B (100%)

c956e37 1 week, 5 days ago
Selected Answer: B
come on its simple
upvoted 1 times

b3532e4 9 months, 1 week ago
SNMP Master Agent serves as the primary service for the MIB interface. You must manually activate Cisco CallManager SNMP service; all other SNMP services should be running after installation.
upvoted 1 times

Panda_man 2 years, 5 months ago
Selected Answer: B
Cisco CallManager SNMP bellongs to feature services who needs to be enabled manually.
upvoted 1 times

Panda_man 2 years, 5 months ago
Host Resources Agent and SNMP Master Agent are network services
upvoted 1 times

wwisp3422112 2 years, 7 months ago
B is correct here.
upvoted 1 times

Question #30 *Topic 1*

A company deploys centralized Cisco UCM architecture for a hub location and two remote sites.
✑ The company has only one ITSP connection at the hub location, and ITSP supports only G.711 calls.
✑ Remote site A has a 1-Gbps fiber connection to the hub location and calls to and from remote site A use G.711 codec.
✑ Remote site B has a 1-T1 connection to the hub location and calls to and from remote site B use G.729 codec.
Based on the provided guidance, a Cisco voice engineer must design media resource management for the customer. What is the method that needs to be followed?

A. configure the hardware transcoder on the site B router

B. configure the hardware transcoder on the site A router

C. configure the hardware transcoder on the hub location router

D. configure the software transcoder on Cisco UCM to support voice calls to and from both remote sites

Correct Answer: *C*

Community vote distribution

C (100%)

bdrewes Highly Voted 3 years, 6 months ago
I think it should be C, so you can maintain G729 calls across the T1.
upvoted 20 times

Vijay_ABI Highly Voted 3 years, 1 month ago
Correct answer is C. Site B has a T1 connection and we don't want to force them to limit the number of calls. As an administrator, i would take the incoming calls from site B in G729 and do a transcode at the hub location, and receive an Incoming call to Site B in G 711 and transcode it to G729 and traverse over T1. Win Win situation.
upvoted 8 times

pasangawa Most Recent 7 months, 3 weeks ago
Per SRND, planning media resources is conference remotely and transcode centrally therefor answer should be C
upvoted 1 times

Panda_man 1 year, 11 months ago
Selected Answer: C
C should be correct
upvoted 3 times

RdTx 2 years, 1 month ago
Selected Answer: C
Should be at the Hub site so that it can go thru the T1 using G729 and get transcoding at the Hub.
upvoted 2 times

Piji 2 years, 5 months ago
Site A not connected to Site B directly, and connected through Hub, so if we put the Hardware Transcoder in Hub, that also can be used for Site A talks to Site B through Hub with Hardware Transcoder.
Site B got limited bandwidth and that is why using G.729, so definitely the Hardware Transcoder should be in site Hub, so both Site A and Hub can use that Hardware Transcoder for communicate to site B. The correct answer is C.
upvoted 2 times

jarcoman99 3 years, 2 months ago
If you you are using G729 for remote location B, you are doing so to conserve bandwidth at the T1 link between Location B and the Hub Location. Converting from G729 to G711 at remote location B will force the traffic back to the hub location to be G711, which means that the G729 benefit (reduce bandwidth consumption) will not be used at all. This answer must be C (transcoder at the hub location).
upvoted 1 times

BhaiKyare 3 years, 4 months ago
I will go with C because it is centralized setup . Local transcoders are advised with multi site deployments right ?
upvoted 1 times

Butz1337 3 years, 4 months ago

I go or C, because the ITSP connection is not the connection connecting Site B and Hub, its a different one, therefor I would install trancoder @ hub location to maximize calls over T1 connecion.
upvoted 1 times

jay_c_an 3 years, 5 months ago
Definitely think it is B because the ISP can only support G711. Site B must transcode G729 into G711 before leaving the site.
upvoted 2 times

Komy 3 years, 6 months ago
I agree with @Collabhunter and @Landrey
First, we can not assume that Site A and Hub have transcoders. which means that they will need transcoders to communicate to Site B.
Scenario 1: if we place transcoders in site A --> Hub can't communicate to Site B
Scenario 2: if we place transcoders in Hub --> Site A can't communicate to site B

That's why we need to place transcoders in Site B, so both sites (hub and site A) can communicate to site B
upvoted 1 times

Piji 2 years, 5 months ago
Site A not connected to Site B directly, and connected through Hub, so if we put the Hardware Transcoder in Hub, that also can be used for Site A talks to Site B through Hub with Hardware Transcoder.
Site B got limited bandwidth and that is why using G.729, so definitely the Hardware Transcoder should be in site Hub, the correct answer is C.
upvoted 1 times

Collabhunter 3 years, 6 months ago
In real world I would say to put transconding on each site, due WAN bandwidth / consumption, this will be more feasible.. But since this is a Cisco Exam and we have one option to go... Im going with Landrey said, xcode needs to be place on site B, so both Hub and Site A (g711) could call site B
upvoted 3 times

davidanibalmarcelino 3 years, 6 months ago
does any one know how to add questions on here?
upvoted 1 times

MKZ 3 years, 6 months ago
I think it should be C
upvoted 1 times

Blue_Tektite 3 years, 6 months ago
What is the correct answer on this?
upvoted 2 times

Sharky1066 2 years, 10 months ago
The correct answer is C. You place the hardware (not software transcoder - there is no such thing) at the hub location. The trancoding resource takes the incoming G.711 media stream from the ISP (loacted at the hub location) and transcodes to a G.729 media stream that is routed to an endpoint located at site B thus conserving WAN bandwidth. There is no need to conserve WAN bandwith between the hub location and Site A therefore no need to transcode to G.729. If you were to place the trancoding resource at SiteB you would be sending a G.711 call across the T1 link (low bandwidth issues etc) only to be trancoded at site B - thats not something you do in the real world.
upvoted 4 times

landrey 3 years, 6 months ago
I think the hardware transcoder should be on the Site B, because all their calls no matter the destination will be using G729 against G711 so the transcoder source must be placed locally
upvoted 4 times

Question #31 *Topic 1*

What are two key features of the Expressway series? (Choose two.)

A. IP to PSTN call connectivity

B. B2B calls

C. VPN connection toward the internal UC resources

D. SIP header modification

E. device registration over the Internet

Correct Answer: *BE*

Community vote distribution

BE (100%)

Panda_man 11 months, 2 weeks ago

Selected Answer: BE

B and E - Registration directly to Expressway. This option is new in Expressway X8.9. Registering users and devices directly to Expressway (both SIP and H.323 registrations are supported with Expressway-C, and proxy SIP registration is supported with Expressway-E) enables you to extend video-centric services to users. and Feature is ofc b2b

upvoted 4 times

Question #32 *Topic 1*

When setting a new primary DNS server in the Cisco UCM CLI, what is required for the change to take effect?

A. restart of CallManager service

B. restart of DirSync service

C. restart of the network service

D. restart of TFTP service

Correct Answer: *C*

Community vote distribution

C (100%)

c956e37 1 week, 5 days ago

Selected Answer: A

The restart is necessary because CUCM does not dynamically update the DNS settings without rebooting.

upvoted 1 times

mcbesy 3 months, 1 week ago

Selected Answer: C

Cisco Unified Serviceability--> Tools--> Control Center - Network Services

upvoted 1 times

c37e2aa 4 months, 4 weeks ago

Selected Answer: A

You cannot restart the "network service".

Cisco says you need to restart the server > the best answer is A

upvoted 1 times

Panda_man 11 months, 2 weeks ago

Recommendation is to restart the cluster >therefore will go for C

upvoted 1 times

Sal007 1 year, 1 month ago

Selected Answer: C

if restarting the cluster means Call Manager service, then answer A is correct, else C is correct.

upvoted 1 times

usernamesarehard 1 year, 4 months ago

Selected Answer: C

restart network service

upvoted 4 times

Omitted 1 year, 7 months ago

Selected Answer: C

You don't have to restart the call manager service. Actually according to this guide (read at the bottom) you don't have to restart anything, it restarts network services for you.

https://www.cisco.com/c/en/us/td/docs/voice_ip_comm/cucm/admin/12_5_1SU1/adminGd/cucm_b_administration-guide-1251SU1/cucm_b_test-adminguide_chapter_011110.html

upvoted 4 times

ciscogeek 1 year, 7 months ago

Selected Answer: C

Referenced links are incorrect because CallManager service restart required for change in Domain Name of CUCM.

Here we are changing DNS, I would go for

C. restart of the network service

Else many docs suggest to restart cluster itself

upvoted 3 times

Question #33 *Topic 1*

When configuring Cisco UCM, which configuration enables phones to automatically reregister to a Cisco UCM publisher when the connection to the subscriber is lost?

A. SRST

B. Route Group

C. Device Pool

D. Cisco UCM Group

Correct Answer: *D*

Community vote distribution

D (100%)

pasangawa Highly Voted 2 years, 6 months ago

I believe this should be D. Cisco UCM Group dictates which server to register to. SRST pertains to the router registration when all servers have failed.

upvoted 14 times

Vijay_ABI Highly Voted 2 years, 1 month ago

Who actually answers these questions on this website? Most of the answers are incorrect and needs modification.

upvoted 5 times

Panda_man Most Recent 11 months, 2 weeks ago

Selected Answer: D

SRST is filed under device pool settings : " SRST Reference: This is an optional field. Survivable Remote Site Telephony is a redundancy feature available on Cisco IOS routers that allows Cisco Unified IP phones registered to the Cisco Unified Communications Manager from remote office locations to register to their local router during WAN network failure events. The SRST reference determines where those phones should register during these situations"

So answer is D

upvoted 2 times

DaKenjee 1 year, 1 month ago

Selected Answer: D

Question can be answered in both ways:
(1) what i can configure on device configuration page (device pool)
(2) for redundancy (Cisco UCM Group)

Help this page on CUCM answers this:
About Cisco Unified Communications Manager Group Setup
A Cisco Unified Communications Manager Group specifies a prioritized list of up to three Cisco Unified Communications Managers.
...
Each device pool has one Cisco Unified Communications Manager Group that is assigned to it.
When a device registers, it attempts to connect to the primary (first) Cisco Unified Communications Manager in the group
that is assigned to its device pool.
If the primary Cisco Unified Communications Manager is not available,
the device tries to connect to the next Cisco Unified Communications Manager that is listed in the group, and so on.

upvoted 1 times

usernamesarehard 1 year, 4 months ago

Selected Answer: D

Cisco UCM Group

upvoted 1 times

fundamentil 1 year, 9 months ago

Thinking its the device pool. The UCM groups are listed in the device pools and each phone has a device pool that it is associated with. I think this is a terrible question because the PUB may not be running Call Manger service anyway.

upvoted 1 times

msully 2 years, 2 months ago

D. Cisco UCM Group. SRST will cause the device to register with the router.

upvoted 4 times

sergioax88 2 years, 6 months ago

D

https://www.cisco.com/en/US/docs/voice_ip_comm/cucm/admin/4_1_3/ccmcfg/b02cmgrp.html

upvoted 2 times

MKZ 2 years, 6 months ago

should be D

upvoted 2 times

FlashNC 2 years, 6 months ago

Answer D. Call Manager group dictates the failover CUCM order.

upvoted 3 times

Question #34 *Topic 1*

Which version is used to provide encryption for SNMP management traffic in collaboration deployments?

A. SNMPv2c

B. SNMPv2

C. SNMPv1

D. SNMPv3

Correct Answer: *D*

Reference:

https://www.cisco.com/c/en/us/td/docs/ios-xml/ios/snmp/configuration/xe-16/snmp-xe-16-book/nm-snmp-encrypt-snmp-support.html

Currently there are no comments in this discussion, be the first to comment!

Question #35 *Topic 1*

What is the validity period of the ITL Recovery certificate in Cisco UCM?

A. 1 year

B. 20 years

C. 5 years

D. 10 years

Correct Answer: *B*

The validity of ITLRecovery has been extended from 5 years to 20 years to ensure that the ITLRecovery certificate remains same for a longer period

Reference:

https://www.cisco.com/c/en/us/td/docs/voice_ip_comm/cucm/security/12_0_1/secugd/cucm_b_cucm-security-guide-1201/cucm_b_cucm-security- guide-1201_chapter_011.html

Community vote distribution

B (100%)

Panda_man 11 months, 2 weeks ago

Selected Answer: B

As per the documentation it's 20 years.

Ref : https://www.cisco.com/c/en/us/td/docs/voice_ip_comm/cucm/security/12_0_1/secugd/cucm_b_cucm-security-guide-1201/cucm_b_cucm-security-guide-1201_chapter_011.html#CUCM_TK_V1AF72FD_00

upvoted 1 times

Ol_Mykhailiuk 1 year, 3 months ago

Selected Answer: B

The validity of ITLRecovery has been extended from 5 years to 20 years to ensure that the ITLRecovery certificate remains same for a longer period.

upvoted 1 times

Question #36 *Topic 1*

Which service must be enabled when LDAP on Cisco UCM is used?

A. Cisco Bulk Provisioning Service

B. Cisco AXL Web Service

C. Cisco CallManager SNMP Service

D. Cisco DirSync

Correct Answer: *D*

Reference:

https://www.cisco.com/c/en/us/td/docs/voice_ip_comm/cucm/im_presence/configAdminGuide/10_5_1/CUP0_BK_CE43108E_00_config-admin-guide- imp-105/CUP0_BK_CE43108E_00_config-admin-guide-imp-105_chapter_01000.html

Community vote distribution

D (100%)

Littlelarry123 1 year ago

Can anyone explain why it's D?

upvoted 1 times

unbelievable 8 months, 1 week ago

When configuring LDAP integration on Cisco Unified Communications Manager (UCM), the service that must be enabled is the "Cisco DirSync" service. The Cisco DirSync service is responsible for synchronizing the user data from the LDAP directory to the UCM database, ensuring that the user information on UCM is up-to-date with the LDAP directory.

upvoted 1 times

Panda_man 1 year, 5 months ago

Selected Answer: D

easy one

upvoted 1 times

Question #37 *Topic 1*

On a Cisco Catalyst Switch, which command is required to send CDP packets on a switch port that configures a Cisco IP phone to transmit voice traffic in 802.1Q frames, tagged with the voice VLAN ID 221?

A. Device(config-if)# switchport trunk allowed vlan 221

B. Device(config-if)# switchport vlan voice 221

C. Device(config-if)# switchport voice vlan 221

D. Device(config-if)# switchport access vlan 221

Correct Answer: *C*
Reference:
https://www.cisco.com/c/en/us/td/docs/switches/lan/catalyst2960/software/release/12-2_40_se/configuration/guide/scg/swvoip.pdf

Community vote distribution
C (100%)

JoeC716 7 months, 4 weeks ago
Selected Answer: C
C is right
upvoted 2 times

Panda_man 1 year, 11 months ago
Selected Answer: C
C is correct : switchport voice vlan {vlan-id | • vlan-id—Configure the phone to forward all voice traffic through the
specified VLAN. By default, the Cisco IP Phone forwards the voice
traffic with an IEEE 802.1Q priority of 5. Valid VLAN IDs are 1 to
4094.
upvoted 2 times

Question #38 *Topic 1*

Which datastore and protocol is used for saving back-up files within the Disaster Recovery System of Cisco UCM?

A. local disk on the Cisco UCM server

B. remote disk on the SFTP server

C. remote disk on a CIFS share

D. remote disk on an NFS share

Correct Answer: *B*

Reference:

https://www.learncisco.net/courses/cucm-basics/serviceability-backup-and-restore/backing-up-and-restoring-cucm.html

Community vote distribution

B (100%)

mcbesy 3 months, 1 week ago

Selected Answer: B

Before starting with the activities, you must have access to an SFTP server, which will be used to configure a network storage location. The SFTP path must exist before you create the backup, and the account that is used to access the SFTP server must have write permission for the selected path.

upvoted 1 times

Panda_man 11 months, 2 weeks ago

Selected Answer: B

from book : "The DRS includes the following
capabilities:
-A user interface for performing backup and restore tasks;
-A distributed system architecture for performing backup and restore functions, including
monitoring the current backup status and providing a history log;
-Scheduled or manual backups;
-Backups archived to a physical drive or remote SFTP server;

upvoted 2 times

Question #39 *Topic 1*

ip.addr==10.0.101.10

Time	Source	Destination	Info
18.683437	10.117.34.222	10.0.101.10	50310 --> 5060 [SYN] Seq=0 Win=64240 Len=0 MSS=1460 WS=256 SACK
18.938881	10.117.34.222	10.0.101.10	50314 --> 5060 [SYN] Seq=0 Win=64240 Len=0 MSS=1460 WS=256 SACK
21.686680	10.117.34.222	10.0.101.10	[TCP Retransmission] 50310 --> 5060 [SYN] Seq=0 Win=64240 Len=0
21.941993	10.117.34.222	10.0.101.10	[TCP Retransmission] 50314 --> 5060 [SYN] Seq=0 Win=64240 Len=0
21.687008	10.117.34.222	10.0.101.10	[TCP Retransmission] 50310 --> 5060 [SYN] Seq=0 Win=64240 Len=0
21.942784	10.117.34.222	10.0.101.10	[TCP Retransmission] 50314 --> 5060 [SYN] Seq=0 Win=64240 Len=0

Refer to the exhibit. An administrator is attempting to register a SIP phone to a Cisco UCM but the registration is failing. The IP address of the SIP Phone is
10.117.34.222 and the IP address of the Cisco UCM is 10.0.101.10. Pings from the SIP phone to the Cisco UCM are successful. What is the cause of this issue and how should it be resolved?

A. An NTP mismatch is preventing the connection of the TCP session between the SIP phone and the Cisco UCM. The SIP phone and Cisco UCM must be set with identical NTP sources.

B. The certificates on the SIP phone are not trusted by the Cisco UCM. The SIP phone must generate new certificates.

C. DNS lookup for the Cisco UCM FQDN is failing. The SIP phone must be reconfigured with the proper DNS server.

D. A network device is blocking TCP port 5060 from the SIP phone to the Cisco UCM. This device must be reconfigured to allow traffic from the IP phone.

Correct Answer: *D*

v1nhthanh 2 months, 1 week ago

Selected Answer: D

Trying multiple time to 5060

upvoted 1 times

n3rds 3 months, 2 weeks ago

Selected Answer: D

I could be wrong but the use case on this question is somewhat similar to this community thread https://community.cisco.com/t5/ip-telephony-and-phones/cisco-ip-phone-9971-sip-registration-failed/td-p/1632627

upvoted 1 times

Question #40 *Topic 1*

```
voice class dpg 2000
 dial-peer 2001 preference 1
 dial-peer 2002 preference 2
 dial-peer 2003 preference 3

dial-peer voice 1001 voip
 description INBOUND
 session protocol sipv2
 session target ipv4:10.0.0.1
 destination dpg 2000
 incoming called-number 5T

dial-peer voice 2001 voip
 destination-pattern 5506
 session protocol sipv2
 session target ipv4:10.0.0.2

dial-peer voice 2002 voip
 destination-pattern 55..
 session protocol sipv2
 session target ipv4:10.0.0.3

dial-peer voice 2003 voip
 destination-pattern 5507
 session protocol sipv2
 session target ipv4:10.0.0.4
```

Refer to the exhibit. A Cisco UCM user with directory number 4401 dials 5507, and the call is routed to a Cisco Unified Border Element. Which IP address will the call be sent to?

A. 10.0.0.2

B. 10.0.0.3

C. 10.0.0.4

D. 10.0.0.1

Correct Answer: *A*

Community vote distribution

A (83%) B (17%)

Nicetomeetyou Highly Voted 3 years, 12 months ago

Actually the correct answer is A, voice class dpg Completely ignores destination-pattern configured in the dial-peer section and follows preference. To be sure I build this setup in my lab which confirms my statement.

upvoted 18 times

WilliamC 3 years, 11 months ago

"The destination-pattern command is required on the outbound dial peer even though matching is not done based on this command." https://www.cisco.com/c/en/us/td/docs/ios-xml/ios/voice/dialpeer/configuration/xe-3s/vd-xe-3s-book/multiple-outBound-dial-peer.html

upvoted 3 times

MKZ 3 years, 11 months ago

Allows grouping of outbound dial-peers based on an incoming dial-peer, reducing existing outbound dial-peer provisioning requirements

• Eliminates the need to configure extra outbound dial-peers that are sometimes needed as workarounds to achieve desired call routing outcome

• Multiple outbound dial-peers are saved under a new “voice class dpg <tag>”. The new “destination dpg <tag>” command line of an inbound voip dial-peer can be used to reference the new dpg (dial-peer group)

• Once an incoming voip call is handled by an inbound voip dial-peer with an active dpg, dial-peers of a dpg will then be used as outbound dial-peers for an incoming call

• The order of outgoing call setups will be the sorted list of dial-peers from a dpg, i.e, the destination-patterns of the outgoing dial-peers is not relevant for selection

upvoted 1 times

MKZ Highly Voted 4 years ago

incoming call first hit dial-peer 1001 -> voice class DGP 2000 > Preference 2

I think 10.0.0.3.

upvoted 7 times

ciscogeek 3 years, 2 months ago

why not preference 1

upvoted 2 times

Mert_kerna 2 years, 5 months ago

Preference 1 doesn't match the dialed pattern. If you dial 5007, it doesn't match the pattern, 5006, respectively

upvoted 2 times

Mert_kerna 2 years, 5 months ago

HOWEVER, it is preference 1, because the destination pattern command is NOT evaluated when selecting an outbound dial-peer defined by a DPG.

upvoted 2 times

JWMcInSC 2 years ago

the destination is 5506 not the dialed number of 5507

upvoted 1 times

c956e37 Most Recent 1 week, 5 days ago

Selected Answer: C

come on the answer is C , the dialpeer configure has a destinatiuon pattern of 5507

upvoted 1 times

mcbesy 2 months, 2 weeks ago

Selected Answer: A

I know I last Choose Option "D" as the answer, but after researching more detailly, I come to the conclusion that Option "A" is the correct answer.

Explanation:

Once an INBOUND dial-peer with a dial-peer group configured is matched(Incoming called-number 5T), the call uses the dial-peer defined in the dial-peer group even if the destination-pattern does not match:dial-peer 2001 preference 1(destination-pattern 5506)

upvoted 1 times

mcbesy 2 months, 3 weeks ago

Selected Answer: D

Dial-peer grouping feature. A group of dial-peers can be made a part of a dial-peer group under a “voice class dpg <tag>”

“destination dpg <tag> will be used to reference a dial-peer group from an incoming dial-peer

Once an incoming dial-peer is matched. The dial-peers which are a part of the dpg defined under it will be used for outbound dial-peer matches on the particular call.

upvoted 1 times

G0y0 4 months, 2 weeks ago

The dial-peer preference command does not influence inbound dial peer selection when there are multiple dial peers with the same match criteria that could be selected, based on the ingress VoIP signaling message. CUBE uses the concept of the longest and most specific match to determine the priority.

Now, When attempting to route a call and perform an outbound dial peer selection, IOS uses the logic dictated by the dial-peer hunt command to determine which dial peer of a given match criteria should be used. The default configuration for dial-peer hunt is 0, which indicates “Longest match in phone number, explicit preference, random selection.” As this suggests, the concept of longest, most specific match applies to outbound dial peers just as it applies to inbound dial peers. In those conditions, the correct answer is C, 10.0.0.4

upvoted 1 times

58922c4 8 months ago

Selected Answer: A

matches 5T

upvoted 1 times

AbdurrahmanBNC 11 months, 3 weeks ago

Correct Answer: C. 10.0.0.4

Explanation:

The configuration provided in the exhibit includes multiple dial peers. When a call is made, the destination pattern of the dialed number is matched against the configured dial peers to determine the appropriate session target.

The user dials 5507.
The dial peer configuration includes:
dial-peer voice 2001 with pattern 5506.
dial-peer voice 2002 with pattern 55...
dial-peer voice 2003 with pattern 5507.
The destination pattern 5507 matches exactly with dial-peer voice 2003, which has the session target ipv4:10.0.0.4.

Thus, the call to 5507 will be sent to the IP address 10.0.0.4.
upvoted 2 times

MauroRey 12 months ago
Selected Answer: B
Correct answer is B!
upvoted 1 times

Tres_Iqus 1 year ago
I´m new to this so maybe this is an stupid post with an even more stupid question:
Do the system check all the patterns looking for a match? Or do it takes the first match option in the list as the correct one and then send the call?
If it looks all the patterns the correct answer is C as confirmed by the users.
But if it follows the order looking for the 1st match... "55.." pattern should be a match and the target IP would be option B.
upvoted 1 times

JWMcInSC 2 years ago
Option A at 5006 is not even the number dialed. The number dialed is 5007
upvoted 1 times

FG23 2 years, 2 months ago
Selected Answer: A
A is the answer
upvoted 1 times

Mbover 2 years, 4 months ago
Correct answer is A
"Once an inbound dial-peer with a dial-peer group configured is matched the call uses the dial-peer defined in the dial-peer group even if the destination-pattern doesn't match"
https://www.cisco.com/c/en/us/support/docs/voice/ip-telephony-voice-over-ip-voip/211306-In-Depth-Explanation-of-Cisco-IOS-and-IO.html#anc23
upvoted 4 times

WeNt48 2 years, 4 months ago
One of the most tricky questions in whole set.
This part you entered is most important. destination-pattern in dpg is mandatory and without it wouldn't treat dial-peer as valid destination.
Although it doesn't match pattern it will select first preference from the list. Thus answer will be A.
It would be enough if they would change preference 1 with dial-peer not having destination-pattern and the answer will be completely different.
upvoted 1 times

NNickyy 2 years, 5 months ago
Selected Answer: A
A, voice class dpg Completely ignores destination-pattern configured in the dial-peer section and follows preference.
upvoted 1 times

Panda_man 2 years, 5 months ago
correct but than it's B - first finding match , not A - A is clear match that is 5006 which is not correct
upvoted 1 times

da0011 3 years, 5 months ago
Selected Answer: A
The example in this document is similar to the test question.
https://www.cisco.com/c/en/us/support/docs/voice/ip-telephony-voice-over-ip-voip/211306-In-Depth-Explanation-of-Cisco-IOS-and-IO.html#anc23
upvoted 3 times

vvpark13 3 years, 8 months ago
Here is another reference. The example in this document is similar to the test question.
https://www.cisco.com/c/en/us/support/docs/voice/ip-telephony-voice-over-ip-voip/211306-In-Depth-Explanation-of-Cisco-IOS-and-IO.html#anc23

upvoted 3 times

Komy 3 years, 12 months ago

10.0.0.3 is correct

Snippet from Cisco: "Once an incoming call is matched by an inbound dial peer with an active destination dial-peer group, dial peers from this group are used to route the incoming call. No other outbound dial-peer provisioning to select outbound dial peers is used.

A preference can be defined for each dial peer in a dial-peer group"

upvoted 4 times

Question #41 *Topic 1*

Configuration of DNS is required to achieve a fully functional Cisco UCM system. Cisco UCM uses DNS to resolve fully qualified domain names to IP addresses for which destinations?

A. MRA

B. trunk

C. AAR

D. H.323

Correct Answer: *B*

Reference:

https://community.cisco.com/t5/ip-telephony-and-phones/2-part-question-regarding-dns-on-cucm/td-p/2256179

Community vote distribution

B (100%)

wwisp3422112 7 months, 1 week ago

Correct me if im wrong. MRA isn't a "destination", more of a concept to enable remote workers to connect to the enterprise and use the collab features provided by CUCM.

Ill go with B.

upvoted 2 times

Omitted 1 year, 1 month ago

Selected Answer: B

MRA isn't part of a fully functioning UCM system...

Unified CM can use DNS to:

- Resolve service (SRV) records to host names and then to IP addresses for SIP trunk destinations
- Resolve service (SRV) records to host names and then to IP addresses for SIP trunk destinations

https://community.cisco.com/t5/ip-telephony-and-phones/2-part-question-regarding-dns-on-cucm/td-p/2256179

upvoted 2 times

geroboamo 1 year, 3 months ago

definitely MRA. trunk can be configured with their IP

upvoted 1 times

santiagof 8 months ago

there is no MRA configuration on CUCM.

you may only need a trunk to VCS for B2b calls, but not for MRA

upvoted 1 times

ant71 1 year, 5 months ago

shoudn't be MRA?

upvoted 2 times

Question #42 Topic 1

```
Sent:
INVITE sip:2004@192.168.100.100:5060 SIP/2.0
Via: SIP/2.0/UDP 192.168.100.200:5060;branch=z9hG4bKF1FED
From: "7000" <sip:7000@192.168.100.200>;tag=43CDE-1A22
To: <sip:2004@192.168.100.100>
Call-ID: 12BCA00-3C3E11EA-01234567@192.168.100.200
Supported: 100rel,timer,resource-priority,replaces,sdp-anat
Min-SE: 1800
User-Agent: Cisco-SIPGateway/IOS-16.9.5
Allow: INVITE, OPTIONS, BYE, CANCEL, ACK, PRACK, UPDATE, REFER, SUBSCRIBE, NOTIFY, INFO, REGISTER
CSeq: 101 INVITE
Contact: <sip:7000@192.168.100.200:5060>
Expires: 180
Max-Forwards: 68
P-Asserted-Identity: "7000" <sip:7000@192.168.100.200>
Session-Expires: 1800
Content-Type: application/sdp
Content-Length: 254

v=0
o=CiscoSystemsSIP-GW-UserAgent 5871 9974 IN IP4 192.168.100.200
s=SIP Call
c=IN IP4 192.168.100.200
t=0 0
m=audio 8002 RTP/SAVP 0
c=IN IP4 192.168.100.200
a=rtpmap:0 PCMU/8000
a=ptime:20
```

Refer to the exhibit. Calls to Cisco Unity Connection are failing across Cisco Unified Border Element when callers try to select a menu prompt. Why is this happening and how is it fixed?

A. Cisco Unified Border Element is sending the incorrect media IP address. The IP address of the loopback interface must be advertised in the SDP.

B. Cisco Unity Connection is configured on G.729 only. Cisco Unity Connection must be reconfigured to support iLBC.

C. Cisco Unified Border Element is not sending any support for DTMF. DTMF configuration must be added to the appropriate dial peer.

D. The Cisco Unity Connection Call Handler is configured for a ג€Release to Switchג€ transfer type. Transfers must be disabled for the Cisco Unity Connection Call Handler.

Correct Answer: *C*

AbdurrahmanBNC 11 months, 3 weeks ago

• C. Cisco Unified Border Element is not sending any support for DTMF. DTMF configuration must be added to the appropriate dial peer: Correct. If the Cisco Unified Border Element (CUBE) is not properly configured to handle DTMF signals, the digits pressed by callers will not be transmitted to Cisco Unity Connection, causing the menu prompts to fail.

upvoted 2 times

Mbover 2 years, 4 months ago

In the INVITE there isn't any DTMF allowed, so the DTMF will not work. This is the issue.

upvoted 1 times

Mbover 2 years, 4 months ago

In the INVITE there isn't any DTMF allowed, so the DTMF will not work. This is the issue.

upvoted 1 times

wwisp3422112 2 years, 7 months ago

Anyone has a explenation for this question?

upvoted 1 times

Mert_kerna 2 years, 7 months ago

Yea, there's no attribute for DTMF

m = audio 35904 RTP/AVP 8 101
a = rtpmap:8 PCMA/8000
a = rtpmap:101 telephone-event/8000

101 being the DTMF attribute ETC

upvoted 2 times

Question #43 *Topic 1*

An administrator is configuring LDAP for Cisco UCM with Active Directory integration. A customer has requested to use `ipphone` instead of `telephoneNumber' as the phone number attribute. Where does the administrator specify this attribute mapping in Cisco UCM?

A. LDAP Custom Filter

B. LDAP Authentication

C. LDAP Directory user fields

D. LDAP Directory custom user fields

Correct Answer: *C*

Reference:

https://www.cisco.com/c/en/us/td/docs/voice_ip_comm/cucm/admin/10_0_1/ccmcfg/CUCM_BK_C95ABA82_00_admin-guide-100/CUCM_BK_C95ABA82_00_admin-guide-100_chapter_01101.pdf

Community vote distribution

C (100%)

skneff 11 months, 3 weeks ago

Selected Answer: C

C.

LDAP-synced CUCM users have their fields mapped to LDAP attributes in the 'LDAP Directory' configuration section.

The 'Phone number' CUCM field is what this question is referencing, and the two native selectable LDAP attributes to select from are 'telephoneNumber' and 'ipPhone'. These are native LDAP attributes and not custom LDAP attributes, therefore it is C, not D.

upvoted 3 times

Question #44 *Topic 1*

In the Cisco Expressway solution, which two features does Mobile and Remote Access provide? (Choose two.)

A. VPN-based enterprise access for a subnet of Cisco Unified IP Phone models

B. secure reverse proxy firewall traversal connectivity

C. the ability of Cisco IP Phones to access the enterprise through VPN connection

D. the ability to register third-party SIP or H.323 devices on Cisco UCM without requiring VPN

E. the ability for remote users and their devices to access and consume enterprise collaboration applications and services

Correct Answer: *BE*
Reference:
https://www.cisco.com/c/en/us/td/docs/solutions/CVD/Collaboration/enterprise/12x/120/collbcvd/edge.html

spag22500 Highly Voted 1 year ago

B&E

Mobile and Remote Access

The mobile and remote access feature of the Cisco Expressway solution provides secure reverse proxy firewall traversal connectivity, which enables remote users and their devices to access and consume enterprise collaboration applications and services.

upvoted 8 times

Question #45 *Topic 1*

What makes Cisco Unified Border Element a better choice than a conventional Session Border Controller?

A. DTMF interworking

B. SIP security

C. Voice policy

D. Address hiding

Correct Answer: *C*

Community vote distribution

C (100%)

G0y0 3 months, 4 weeks ago

Selected Answer: C

CUBE's integrated voice-policy goes beyond static white/black lists to policy-based evaluation of call patterns and media flows, to prevent abuses such as toll fraud or telephony denial of service.

I found that in this link, however, I do not know which is the source of this information:

https://www.tmcnet.com/voip/features/articles/378621-session-border-controller-roundup.htm

I remember that Voice Policy is the correct answer because I read it in a document that I do not remember unfortunately

upvoted 1 times

b3532e4 10 months ago

A. DTMF interworking

upvoted 1 times

Panda_man 2 years, 5 months ago

Selected Answer: C

Answer is C

upvoted 4 times

H31d1 2 years, 8 months ago

It's C. Other SBC provide the other options, too

https://www.cisco.com/c/en/us/td/docs/ios-xml/ios/voice/cube/configuration/cube-book/voi-cube-overview.html

upvoted 4 times

Omitted 3 years, 1 month ago

This could be A too. Not really sure what they want here.

upvoted 2 times

Question #46 | *Topic 1*

Which two features of Cisco Prime Collaboration Assurance require advanced licensing? (Choose two.)

A. customizable events

B. email notifications

C. multicluster support

D. real time alarm browser

E. call quality monitoring

Correct Answer: *CE*

Reference:

https://www.cisco.com/c/en/us/products/collateral/cloud-systems-management/prime-collaboration/white-paper-c11-731624.html

Community vote distribution

CE (100%)

ciao 10 months, 1 week ago

https://community.cisco.com/t5/collaboration-knowledge-base/cisco-prime-collaboration-assurance-feature-overview/ta-p/3140447

upvoted 1 times

Alan100 1 year, 9 months ago

Selected Answer: CE

C and E correct. https://www.cisco.com/c/en/us/td/docs/net_mgmt/prime/collaboration/11-6/assurance/advanced/guide/cpco_b_Cisco-Prime-Collaboration-Assurance-Guide-Advanced-11-6/cpco_b_Cisco-Prime-Collaboration-Assurance-Guide-Advanced-11-6_chapter_00.html#ID8

upvoted 1 times

Question #47 *Topic 1*

Which entity does Cisco UCM use DNS to resolve fully qualified domain names to an IP address?

A. application server name

B. SIP trunk

C. Cisco UCM Name

D. primary TFTP server for option 150

Correct Answer: *B*

Reference:

https://community.cisco.com/t5/ip-telephony-and-phones/2-part-question-regarding-dns-on-cucm/td-p/2256179

Community vote distribution

B (100%)

Daved90 9 months, 1 week ago

Selected Answer: B

once again poorly worded question, but answer is obviously B

upvoted 1 times

pcp84 10 months, 2 weeks ago

Selected Answer: B

B is correct.

Just putting in a voting comment so it shows on the solution.

upvoted 1 times

wwisp3422112 2 years, 1 month ago

Answer is B - SIP Trunk

https://community.cisco.com/t5/ip-telephony-and-phones/2-part-question-regarding-dns-on-cucm/td-p/2256179

Unified CM can use DNS to:

•Provide simplified system management

•Resolve fully qualified domain names to IP addresses for trunk destinations

•Resolve fully qualified domain names to IP addresses for SIP route patterns based on domain name

upvoted 3 times

Question #48 *Topic 1*

What are two functions of Cisco Expressway in the Collaboration Edge? (Choose two.)

A. Expressway-C provides encryption for Mobile and Remote Access but not for business-to-business communications.

B. Expressway-E provides a VPN entry point for Cisco IP phones with a Cisco AnyConnect client using authentication based on certificates.

C. Expressway-E provides a perimeter network that separates the enterprise network from the Internet.

D. The Expressway-C and Expressway-E pair can interconnect H.323-to-SIP calls for voice.

E. The Expressway-C and Expressway-E pair can enable connectivity from the corporate network to the PSTN via a T1/E1 trunk.

Correct Answer: *CD*

Community vote distribution

CD (100%)

pasangawa 7 months, 3 weeks ago

C&D

https://www.cisco.com/c/en/us/td/docs/solutions/CVD/Collaboration/enterprise/14/collbcvd/edge.html

Interworking — The capability to interconnect H.323-to-SIP calls for voice, video, and content sharing.

Boundary communications services — While Expressway-C sits in the corporate network, Expressway-E is in the enterprise DMZ and provides a distinct connection point for communication services between the enterprise network and the Internet.

Security — The capability to provide authentication and encryption for both mobile and remote access and business-to-business communications.

upvoted 2 times

MeowthL 1 year, 9 months ago

Selected Answer: CD

C and D is correct

https://www.cisco.com/c/en/us/td/docs/solutions/CVD/Collaboration/enterprise/12x/120/collbcvd/edge.html#:~:text=The%20core%20components%20of%

upvoted 4 times

Sergey 1 year, 11 months ago

Selected Answer: CD

C and D.

upvoted 1 times

Panda_man 1 year, 11 months ago

C and D is correct

upvoted 1 times

OSJAY 2 years, 4 months ago

Agree that should be CD. Answer A is wrong anyways cause you can setup the Expressways to provide B2B encryption. And the Expressway-E sits on the DMZ.

upvoted 3 times

Obama42 2 years, 5 months ago

Selected Answer: CD

C and D

upvoted 3 times

ciscogeek 2 years, 8 months ago

Selected Answer: CD

Correct Answers are C and D. From the first reference link:

The Expressway-C and Expressway-E pair performs the following functions:

Interworking — The capability to interconnect H.323-to-SIP calls for voice, video, and content sharing.

Boundary communications services — While Expressway-C sits in the corporate network, Expressway-E is in the enterprise DMZ and provides a distinct connection point for communication services between the enterprise network and the Internet.

Security — The capability to provide authentication and encryption for both mobile and remote access and business-to-business communications.

Hence, answer is C and D
upvoted 4 times

G0y0 2 years, 6 months ago
I am no so sure about C. Expressway E provides communications services between the enterprise and the Internet, but it is totally different to say that provides a "perimeter network". The only thing that provides a perimeter network is something called demilitarized zone (DMZ) that divides the corporate network and Intenet. Expressway E does not provides perimeter network, no way.
upvoted 1 times

WeNt48 1 year, 10 months ago
with A I've a problem that is clearly says Exp-C doesn't provide encryption for B2B calls. According to SRND for Edge infrastructure it is recommended to set media encryption on B2B traversal zone to best effort, as not all far end endpoint might support TLS. Answer says it doesn't provide encryption, where it is partial truth. Whereas for answer C I guess it again wording problem. Like you said it doesn't provide perimeter, but from all remaining answer this one sounds the most probably.
upvoted 2 times

Question #49 *Topic 1*

NAME	TTL	CLASS	TYPE	Priority	Weight	Port	Target Address
_sip._tcp.sample.com	86400	IN	SRV	10	60	5060	server1.sample.com
_sip._tcp.sample.com	86400	IN	SRV	10	30	5060	server2.sample.com
_sip._tcp.sample.com	86400	IN	SRV	5	20	5060	server3.sample.com

Refer to the exhibit. An administrator must fix the SRV records to ensure that server1.sample.com is always contacted first from the three servers. Which solution should the engineer apply to resolve this issue?

A. Priority = 5, Weight = 70

B. Priority = 10, Weight = 5

C. Priority = 100, Weight = 90

D. Priority = 10, Weight = 10

Correct Answer: *A*

Community vote distribution

A (83%) C (17%)

ciscogeek Highly Voted 1 year, 8 months ago

What is the difference between priority and weight in SRV records?

SRV records indicate the "priority" and "weight" of the various servers they list. The "priority" value in an SRV record enables administrators to prioritize one server that supports the given service over another. A server with a lower priority value will receive more traffic than other servers. However, the "weight" value is similar: a server with a higher weight will receive more traffic than other servers with the same priority.

The main difference between them is that priority is looked at first. If there are three servers, Server A, Server B, and Server C, and they have respective priorities of 10, 20, and 30, then their "weight" does not matter. The service will always query Server A first.

But suppose Servers A, B, and C all have a priority of 10 — how will a service choose between them? This is where weight becomes a factor: if Server A has a "weight" value of 5 and Servers B and C have a "weight" value of 3 and 2, Server A will receive the most traffic, Server B will receive the second-most traffic, and Server C the third-most.

upvoted 8 times

Alan100 Highly Voted 9 months, 1 week ago

Selected Answer: A

Priority checked first. Lower value priority is More Preffered. If there's a tie in priority, higher value weight is More Preffered. So answer is A.

upvoted 5 times

lpicardin Most Recent 9 months, 1 week ago

Selected Answer: C

The only way to server 1 to be contacted ALWAYS first is to upscale the Priority so it's the answer C

upvoted 1 times

lpicardin 9 months ago

Lower the priority sorry and then answer A :D

upvoted 1 times

patd1234 1 year, 3 months ago

The main difference between them is that priority is looked at first. If there are three servers, Server A, Server B, and Server C, and they have respective priorities of 10, 20, and 30, then their "weight" does not matter. The service will always query Server A first. So Server 3 is the answer

upvoted 1 times

Question #50 *Topic 1*

What are two differences between media flow-around and media flow-through on Cisco Unified Border Element? (Choose two.)

A. When using media flow-through, the call signaling and media are passed through the Cisco Unified Border Element.

B. When using media flow-through, call signaling goes through the Cisco Unified Border Element, but media does not.

C. When using media flow-around, both call signaling and media do not go through the Cisco Unified Border Element.

D. When using media flow-around, the call signaling goes through the Cisco Unified Border Element, but media is not passed through it.

E. When using media flow-through, the call signaling goes through the Cisco Unified Border Element, but media is not passed through it.

Correct Answer: *AD*

Reference:

https://www.cisco.com/c/en/us/td/docs/ios-xml/ios/voice/cube/configuration/cube-book/media-path.html

Community vote distribution

AD (100%)

mcbesy 2 months, 3 weeks ago

Selected Answer: AD

Media Flow-Through

Media Flow-Through is a media path mode where media and signaling packets terminate and originate on CUBE. As CUBE is an active participant of the call, this mode is recommended when connected outside an enterprise (untrusted endpoints).

Media Flow-Around

Media Flow-Around is a media path mode where signaling packets terminate and originate on CUBE. As media bypasses CUBE and flows directly between endpoints, this mode is recommended when connected within an enterprise (trusted endpoints). Media Flow-Around is supported for both audio and video calls.

upvoted 1 times

Panda_man 11 months, 2 weeks ago

Selected Answer: AD

A and D are correct

upvoted 4 times

Question #51 *Topic 1*

An employee of company ABC just quit. The IT administrator deleted the employee's user ID from the active directory at 10 a.m. on March 4
. The nightly sync
th
occurs at 10 p.m. daily. The IT administrator wants to troubleshoot and find a way to delete the user ID as soon as possible. How is this issue resolved?

A. Wait until 3:15 a.m. on March 6 for garbage collection to remove the user from Cisco UCM. th

B. Wait until 10 p.m. on March 5 when the user is automatically removed from Cisco UCM. th

C. Wait until 10 p.m. on March 4 when the user is automatically removed from Cisco UCM. th

D. Wait until 3:15 a.m. on March 5 for garbage collection to remove the user from Cisco UCM. th

Correct Answer: *A*

Community vote distribution

A (82%) D (18%)

Panda_man Highly Voted 2 years, 5 months ago

Selected Answer: A

A is correct, user must be marked 24 as inactive and than garbage collection will remove it

upvoted 5 times

ciscogeek Highly Voted 3 years, 1 month ago

Selected Answer: A

The answer is correct:

https://www.cisco.com/c/en/us/td/docs/voice_ip_comm/cucm/srnd/collab11/collab11/directry.html#pgfId-1045229

upvoted 5 times

b3532e4 Most Recent 9 months, 4 weeks ago

Nightly Sync: Cisco UCM is synchronized with Active Directory at 10 p.m. daily. However, the sync only disables the user in UCM; it doesn't immediately delete the user. UCM doesn't automatically remove disabled users right away.

Garbage Collection: Cisco UCM uses a process called garbage collection to fully remove objects (like deleted users) from the database. This process typically runs at 3:15 a.m. each day. So, in this case, after the nightly sync at 10 p.m. on March 4, the user will be disabled but not yet deleted. The garbage collection process at 3:15 a.m. on March 5 will then permanently remove the user from Cisco UCM.

upvoted 2 times

b3532e4 11 months, 2 weeks ago

The user ID is deleted from Active Directory at 10 a.m. on March 4.
The nightly sync at 10 p.m. on March 4 will detect the deletion.
Garbage collection will then run at 3:15 a.m. on March 5, which is the standard time for this process.

upvoted 1 times

meamme1 2 years, 6 months ago

Selected Answer: A

User must be marked inactive for 24 hours before garbage collection will remove

upvoted 4 times

RdTx 2 years, 6 months ago

Selected Answer: D

Garbage collection will run next at 3:15 am. So it should be D

upvoted 3 times

NPPR 2 years, 6 months ago

Account has deleted at 10 AM, March 4. So, at Sync Interval 10 PM, March 4. User on UCM is marked inactive. At 3:15 AM March 5, which is choice D, Garbage Collection run and fount that account is inactive less than 24 hours (10PM March 4 to 3:15AM March 5). So, the account have not been deleted yet. Once, the Garbage collection run again on 3:15AM March 6. The account will be deleted due to inactive >24 hours.

So, the answer must be choice A.
upvoted 5 times

Question #52 *Topic 1*

```
hqcucmpub.pkinane.com - PuTTY
login as: admin
admin@hqcucmpub.pkinane.com's password:
Command Line Interface is starting up, please wait ...

   Welcome to the Platform Command Line Interface

VMware Installation:
        2 vCPU: Intel(R) Xeon(R) CPU E5-2699 v3 @ 2.30GHz
        Disk 1: 110GB, Partitions aligned
        8192 Mbytes RAM
        WARNING: DNS unreachable

admin:
```

Refer to the exhibit. An administrator accesses the terminal of a Cisco UCM and starts a packet capture. Which two commands must the administrator use on
Cisco UCM to generate DNS traffic? (Choose two.)

A. show version active

B. utils diagnose module validate_network

C. utils diagnose test

D. utils ntp status

E. show cdp neighbor

Correct Answer: *BC*

Reference:

https://ccxguru.wordpress.com/2018/02/03/behind-the-utils-diagnose-test/

Community vote distribution

BC (100%)

Stevon 4 weeks, 1 day ago

Selected Answer: AB

To generate DNS traffic on a Cisco Unified Communications Manager (UCM) and validate network functionality, you can use the following commands: show version active and utils diagnose module validate_network.

upvoted 1 times

skneff 11 months, 3 weeks ago

Selected Answer: BC

B and C. Part of 'utils diagnose module validate_network' tests DNS functionality, and one of the modules that 'utils diagnose test' is 'utils diagnose module validate_network', so both in turn test DNS functionality.

upvoted 3 times

Question #53 *Topic 1*

A collaboration engineer adds a voice gateway to Cisco UCM. The engineer creates a new gateway device in Cisco UCM, selects VG320 as the device type, and selects MGCP as the protocol. What must be done next to add the gateway to the Cisco UCM database?

A. Select a device pool for the new gateway.

B. Add the FQDN or hostname of the device.

C. Configure the module in slot 0 of the new gateway.

D. Select the DTMF relay type for the gateway.

Correct Answer: *B*

Community vote distribution

B (62%) C (38%)

Komy Highly Voted 8 months, 2 weeks ago

Selected Answer: B

Answer is B: This snippet is captured from the link below:

"Now that you have selected the hardware and protocol used, you need to configure the Domain Name, Cisco Unified Communications Manager Group, and the Module Information. These are the major fields that are required to register an endpoint via MGCP."

So Cisco has listed the order as [Select Hardware, Protocol then FQDN/Hostname]. in the question, we can see it already told you that Hardware is: VG320, Protocol is :MGCP , the next step has to be Specify FQDN/Hostname

https://www.cisco.com/c/en/us/support/docs/voice/media-gateway-control-protocol-mgcp/214635-configure-and-troubleshoot-mgcp-gateways.html

upvoted 5 times

G0y0 Most Recent 5 months ago

B in incorrect because adding the hostname or FQDN is before adding the Gateway to the data base (ie, "save" and "apply conf"). The correct answer must be C.

Reference:

https://www.cisco.com/c/en/us/support/docs/voice/media-gateway-control-protocol-mgcp/214635-configure-and-troubleshoot-mgcp-gateways.html

upvoted 1 times

G0y0 4 months, 3 weeks ago

Another stuff I would like to add, is, to add the VG320 to the database, you need mandatory configure both, hostname/FQDN and Call Manager Group. Just setting the hostname/FQDN, CUCM does not allow you to save it and asks you to put a CMG. So, furthermore and in this scenario, I think the cuestion could be poorly written.

upvoted 1 times

TheBabu 8 months, 1 week ago

Selected Answer: B

Should be B, when you add a new gateway, and fill in the fields in the configuration page in order, the domain name is the first required item to fill.

upvoted 2 times

Kabimas66 9 months, 1 week ago

from: https://www.cisco.com/c/en/us/support/docs/voice/media-gateway-control-protocol-mgcp/214635-configure-and-troubleshoot-mgcp-gateways.html

Answer is B

upvoted 1 times

G0y0 5 months ago

No, B is incorrect. The Reference you share us is the correct one, however, the interpretation is wrong. When you set the FQDN or hostname of the gateway, it is BEFORE of adding the gateway of the CUCM database (ie, adding the device to the cucm database is when you do "save" and "apply config"). So B can not be any way, the correct answer is C.

upvoted 1 times

AgshinA 9 months, 3 weeks ago

Selected Answer: C

After selecting VG320 as the device type and MGCP as the protocol for a new gateway device in Cisco UCM, the next step to add the gateway to the Cisco UCM database is to configure the module in slot 0 of the new gateway1. This involves specifying the appropriate slots and vendor interface cards (VICs) for the gateway.

So, the correct action required would be: C. Configure the module in slot 0 of the new gateway.

upvoted 3 times

AgshinA 9 months, 2 weeks ago

Correcting: B It first asks for Domain Name as a mandatory field then Slot. So Its B

upvoted 2 times

ABHIJEET14 11 months ago

As per the attached below link , answer is C

https://www.cisco.com/c/en/us/td/docs/voice_ip_comm/cucm/admin/12_0_1/systemConfig/cucm_b_system-configuration-guide-1201/cucm_b_system-configuration-guide-1201_chapter_01001.html

Configure MGCP (IOS) Gateway

Perform the following procedure to add and configure an MGCP (IOS) gateway on the Unified Communications Manager.

upvoted 1 times

skneff 11 months, 3 weeks ago

Selected Answer: B

The answer is B.

https://community.cisco.com/t5/collaboration-knowledge-base/mgcp-gateway-integration-with-cucm-and-pstn-service-provider/ta-p/3116588#toc-hId-1719765046

"Step1: Enter the host name or fully qualified domain name (FQDN) of the gateway in the Domain name field. If a domain name is specified in the Cisco router you must provision an FQDN. If the FQDN specified in the CUCM does not match the “host name” dot “domain name” of the Cisco IOS router, then end point will never register. A router with the host name of Router1 and domain name of highpoint.com would have an FQDN of Router1.highpoint.come"

You can't use the IP address of the gateway either, so as written, B is correct. You need to configure a module to get the ports themselves to register, but not for the gateway to be simply entered into the database, which is what the question is asking.

upvoted 1 times

Littlelarry123 1 year, 2 months ago

why is it C?

upvoted 1 times

MaxG 1 year, 7 months ago

Selected Answer: B

B is correct per https://community.cisco.com/t5/collaboration-knowledge-base/mgcp-gateway-integration-with-cucm-and-pstn-service-provider/ta-p/3116588

upvoted 2 times

Firastoumi 1 year, 7 months ago

the answer is C check your CUCM Configuration

upvoted 1 times

pacciolli 1 year, 8 months ago

for me, B is correct

upvoted 2 times

glong 1 year, 9 months ago

my wrong,B is correct

upvoted 3 times

glong 1 year, 9 months ago

Selected Answer: C

must c

upvoted 2 times

Question #54 *Topic 1*

A collaboration engineer configures Global Dial Plan Replication for multiple Cisco UCM clusters. The local cluster acts as the hub cluster, and the remaining clusters act as spoke clusters. Which service must the engineer configure on the local cluster?

A. Location Conveyance on intercluster SIP trunks

B. Intercluster Lookup Service

C. Intra-Cluster Communication Signaling

D. Mobility Cross Cluster

Correct Answer: *B*

Community vote distribution

B (100%)

Slushed Highly Voted 2 years, 8 months ago

Selected Answer: B

The correct answer is B; Intercluster Lookup Service. Intra-Cluster Communication
Signaling (ICCS) is used for the propagation and replication of run-time data such as registration of devices, locations bandwidth, and shared media resources.

upvoted 12 times

Piji Highly Voted 2 years, 5 months ago

Selected Answer: B

Use global dial plan replication to create a global dial plan that spans across the Intercluster Lookup Service (ILS) network. When you enable Global DialPlan Replication, you configure the dial plan component on one cluster, and ILS replicates that information throughout the ILS network.
Use global dial plan replication to create a global dial plan that spans across the Intercluster Lookup Service (ILS) network. When you enable Global DialPlan Replication, you configure the dial plan component on one cluster, and ILS replicates that information throughout the ILS network.

https://www.cisco.com/c/en/us/td/docs/voice_ip_comm/cucm/admin/11_0_1/sysConfig/CUCM_BK_C733E983_00_cucm-system-configuration-guide-1101/cucm_mp_c9dc8050_00_configure-global-dial-plan-replication.pdf

upvoted 5 times

Stevon Most Recent 4 weeks, 1 day ago

Selected Answer: B

To configure Global Dial Plan Replication (GDPR) for multiple Cisco Unified Communications Manager (CUCM) clusters, you need to enable the Intercluster Lookup Service (ILS) and then configure each cluster as either a hub or a spoke, advertising the dial plan data. Each cluster in an ILS network advertises its global dial plan data, which includes both locally configured and learned data.

upvoted 1 times

G0y0 3 months, 4 weeks ago

Selected Answer: B

Learned global dial plan data is the GDPR data learned through ILS networking. Every CUCM cluster keeps a local database of the learned (and local) data.
Implementing Cisco IP Telephony and Video, Part 2 (CIPTV2) Foundation Learning Guide
Chapter 15: Cisco Inter-Cluster Lookup Service (ILS) and Global Dial Plan Replication (GDPR)

upvoted 1 times

skneff 11 months, 3 weeks ago

Selected Answer: B

Answer is B

upvoted 1 times

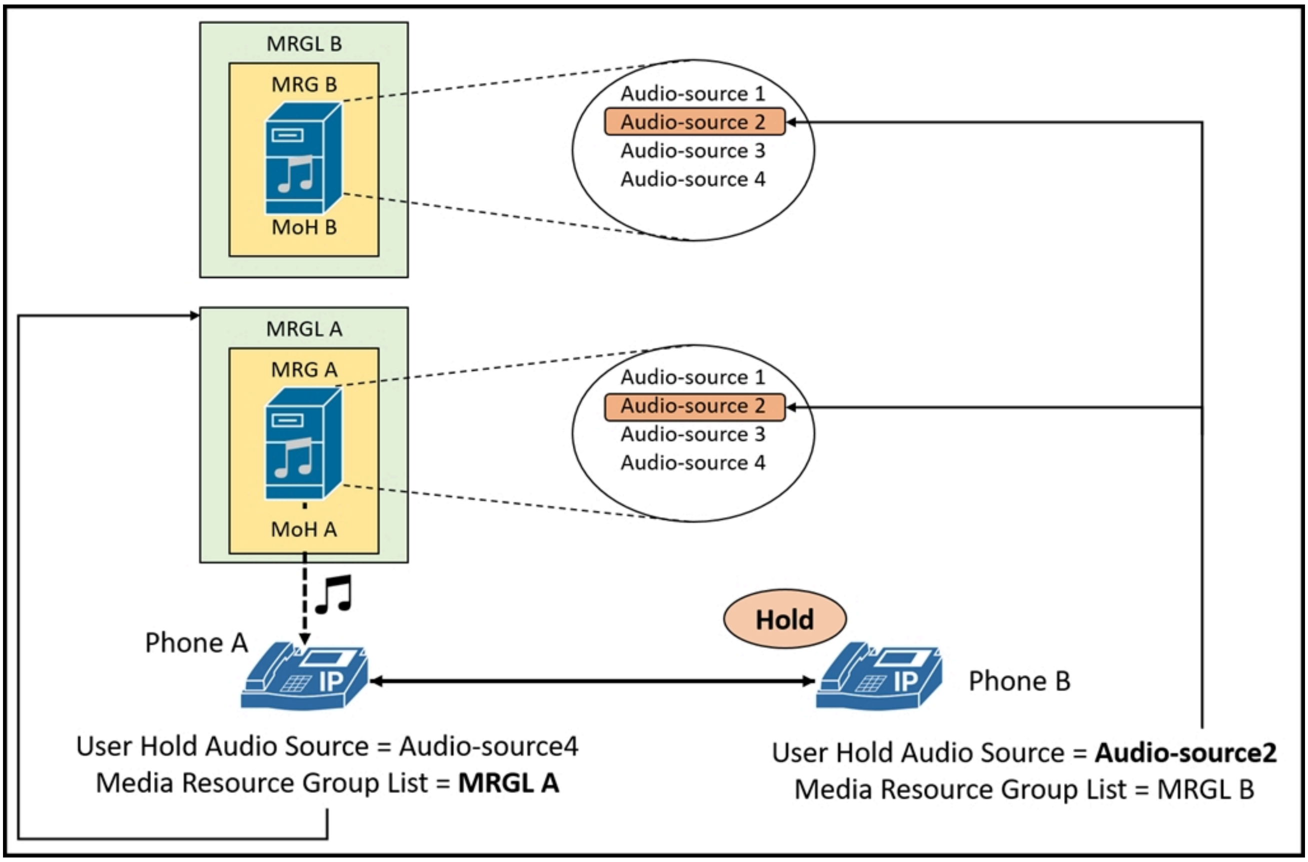
kitty73 1 year, 5 months ago

Selected Answer: B

https://www.cisco.com/c/en/us/td/docs/voice_ip_comm/cucm/admin/12_5_1SU3/systemConfig/cucm_b_system-configuration-guide-1251su3/cucm_m_intercluster-lookup-service.html

Question #55 Topic 1

Refer to the exhibit.



There is a call flow between Phone A and Phone B. Phone B (holder) places Phone A (holdee) on hold. Which MRGL and Audio Source are played to Phone A?

A. MRGL A and Audio Source 4

B. MRGL B and Audio Source 2

C. MRGL B and Audio Source 4

D. MRGL A and Audio Source 2

Correct Answer: *D*

Community vote distribution

D (100%)

ciscogeek Highly Voted 2 years, 8 months ago

Selected Answer: D

In simplest terms, the holder's configuration determines which audio file to play, and the holdee's configuration determines which resource or server will play that file. As illustrated by the example in Figure 18-1, if phones A and B are on a call and phone B (holder) places phone A (holdee) on hold, phone A will hear the MoH audio source configured for phone B (Audio-source2). However, phone A will receive this MoH audio stream from the MRGL (resource or server) configured for phone A (MRGL A).

https://www.cisco.com/c/en/us/td/docs/voice_ip_comm/cucm/srnd/8x/moh.html#wp1092723

upvoted 10 times

pasangawa Most Recent 11 months, 3 weeks ago

D

"The User Hold MoH Audio Source configured for the holder determines the audio file that will be streamed when the holder puts a call on hold, and the holdee's configured MRGL indicates the resource or server from which the holdee will receive the MoH stream."

upvoted 2 times

Panda_man 1 year, 11 months ago

Selected Answer: D

D is correct - if phones A and B are on a call and phone B (holder) places phone A (holdee) on hold, phone A will hear the MoH audio source configured for phone B (Audio-source2). However, phone A will receive this MoH audio stream from the MRGL (resource or server) configured for phone A (MRGL A).

upvoted 3 times

usernamesarehard 2 years, 4 months ago

Selected Answer: D

MRGL A and Audio Source 2

upvoted 1 times

Piji 2 years, 5 months ago

Selected Answer: D

I agree the answer is D, with provided link.

upvoted 2 times

Barney_best 2 years, 6 months ago

agree with ciscogeek. In fact in the link provided you have the answer.

upvoted 1 times

Question #56 *Topic 1*

A SIP phone has been configured in the system with MAC address 0030.96D2.D5CB. The phone retrieves the configuration file from the Cisco UCM. Which naming format is the file that is downloaded?

A. SEP0030407101980.cnf

B. SIP0030407101980.cnf

C. SIP003096D2D5CB.cnf.xml

D. SEP003096D2D5CB.cnf.xml

Correct Answer: *D*

Community vote distribution

D (75%) B (25%)

JoeC716 7 months, 3 weeks ago

Selected Answer: D

Looking at a file from my CM right now...It's D....

If you pick B then sign up for a dcloud demo and see for yourself...

upvoted 1 times

AgshinA 9 months, 3 weeks ago

Selected Answer: D

The configuration file that a SIP phone downloads from the Cisco Unified Communications Manager (UCM) follows the naming format SEP<mac-address>.cnf.xml. For the SIP phone with the MAC address 0030 96D2 D5CB, the configuration file would be named:

SEP003096D2D5CB.cnf.xml ---- D

upvoted 1 times

Mert_kerna 1 year, 10 months ago

Selected Answer: D

I'm looking the status messages of my Cisco 8851 phone right now. It's registered and received it's configuration file as SEPA4B239B7EF9C.cnf.xml - It's registered on a version 12.5 of CUCM. The answer is D.

upvoted 4 times

Myare 1 year, 11 months ago

–Phone-specific configuration file (SIPXXXXYYYYZZZZ.cnf)

XXXXYYYYZZZZ is the MAC address of the phone

upvoted 1 times

ciscogeek 2 years, 8 months ago

Selected Answer: B

https://www.cisco.com/c/en/us/td/docs/voice_ip_comm/cucme/admin/configuration/manual/cmeadm/cmefiles.html

The hunt algorithm searches for files in the following order:

CTLSEP <mac> file for a SCCP phone—For example, CTLSEP003094C25D2E.tlv

SEP <mac> file for a SCCP phone—For example, SEP003094C25D2E.cnf.xml

SIP <mac> file for a SIP phone—For example, SIP003094C25D2E.cnf or gk003069C25D2E

XML default file for SCCP phones—For example, SEPDefault.cnf.xmls

XML default file for SIP phones—For example, SIPDefault.cnf

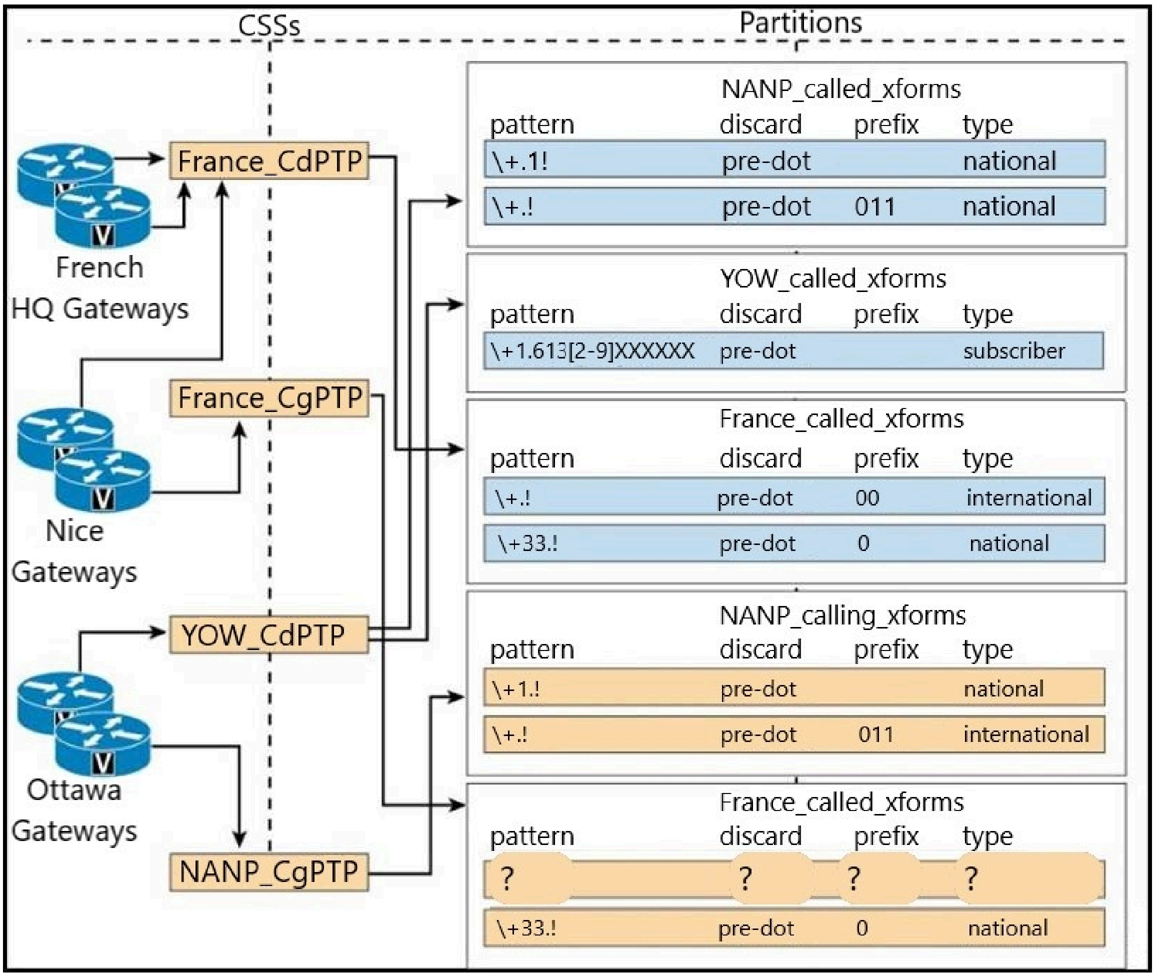
upvoted 2 times

Omitted 2 years, 7 months ago

That document is from CME. Since the question is asking about the naming format from UCM the answer is D. .cnf is only used on older phone models as far as i am aware.

Question #57 Topic 1

Refer to the exhibit.



A call from +1 613 555 1234 that is sent out through the Nice Gateways should result in a calling party of 001 613 555 1234 with the numbering type
`international`. Which configuration accomplishes this goal?

A. \+1.! none pre-dot 001 international

B. \+.001! pre-dot 1 international

C. 613XXXXXXX none +011 international

D. \+.! pre-dot 00 international

Correct Answer: *D*

Community vote distribution

D (83%) A (17%)

Deivich Highly Voted 1 year, 9 months ago

Correct answer D.
In this link you can see the same example with the solution:
https://www.cisco.com/c/en/us/td/docs/voice_ip_comm/cucm/srnd/collab12/collab12/dialplan.html
upvoted 10 times

cuernov Most Recent 9 months ago

Question is missing a image.
upvoted 1 times

Holko18 1 year, 2 months ago

Selected Answer: D

Correct answer is D
upvoted 1 times

⊟ **iExpo_91** 1 year, 3 months ago

The Answer is D reason being the Nice gw route to that \+.! pref 00 for international. so A only routes \+1.! for national.

upvoted 2 times

⊟ **MeowthL** 1 year, 3 months ago

Selected Answer: A

The Answer is A:

The question is asking 001 613 555 1234 with the numbering type "international"

it's totally different with the example. The example that link provided is "national", so have to read the question carefully.

upvoted 1 times

⊟ **MeowthL** 1 year, 3 months ago

sorry, the answer is D!! XD

upvoted 1 times

⊟ **wwisp3422112** 1 year, 7 months ago

Selected Answer: D

Correct answer is D

upvoted 4 times

Question #58 *Topic 1*

An administrator troubleshoots call flows and suspects that there are issues with the dial plan. Which tool enables a quick analysis of the dial plan and provides call flows of dialed digits?

A. Dial Plan Analyzer

B. Dialed Number Analyzer

C. Cisco Dial Plan Analyzer

D. Digit Analysis Analyzer

Correct Answer: *B*
Reference:
https://aurus5.com/blog/cisco/configuring-dialedconfiguring-dialed-number-analyzer-in-cucm/

Stevon 4 weeks, 1 day ago

Selected Answer: B

The tool to analyze a dial plan and view call flows in Cisco Unified Communications Manager (CUCM) is the Dialed Number Analyzer (DNA). This tool allows you to test dial plan configurations by inputting dialed digits, and then it provides a detailed analysis of how the dialed digits are processed and routed. The DNA can be accessed through the Cisco Unified Communications Manager Serviceability UI under Tools > Dial Number Analyze

upvoted 1 times

n3rds 3 months, 2 weeks ago

Selected Answer: B

Dialed Number Analyzer installs as a feature service along with Cisco Unified Communications Manager. The tool allows you to test a Cisco Unified Communications Manager dial plan configuration before deploying it. You can also use the tool to analyze dial plans after the dial plan is deployed.

https://www.cisco.com/c/en/us/td/docs/voice_ip_comm/cucm/dna/11_5_1/CUCM_BK_CBA47A6E_00_cucm-dna-guide-115/CUCM_BK_CBA47A6E_00_cucm-dna-guide-115_chapter_01.html

upvoted 1 times

Question #59 *Topic 1*

What are two characteristics of jitter in voice and video over IP communications? (Choose two.)

A. The packets arrive at uniform time intervals.

B. The packets arrive at varying time intervals.

C. The packets never arrive due to tail drop.

D. The packets arrive out of sequence.

E. The packets arrive with frame errors.

Correct Answer: *BD*

Stevon 4 weeks, 1 day ago

Selected Answer: BD

In voice and video over IP (VoIP/Video) communications, jitter refers to the variation in packet arrival time, causing delays or inconsistencies in the delivery of audio and video data. Two key characteristics of jitter are:

1. Variation in Latency:

Jitter is essentially the inconsistency in latency, meaning the time it takes for data packets to travel across the network is not consistent. Some packets may arrive with delays, while others may arrive on time, leading to a fluctuating and unpredictable flow of data.

2. Impact on Real-time Communication:

Jitter significantly impacts real-time applications like VoIP and video calls. High levels of jitter can lead to choppy audio, video lag, and even dropped packets, resulting in a poor user experience.

upvoted 1 times

b78405a 12 months ago

B & D are correct.

upvoted 2 times

Question #60 *Topic 1*

DRAG DROP -

Drag and drop the SNMPv3 message types from the left onto the corresponding definitions on the right.

Select and Place:

TRAP	messages used to modify a value of an object variable
SET	unreliable messages that alert the SNMP manager to a condition on the network
GET	reliable messages that alert the SNMP manager to a condition on the network
INFORM	messages used to retrieve an object instance

Correct Answer:

	SET
	TRAP
	INFORM
	GET

H31d1 Highly Voted 1 year, 1 month ago

Difference between Inform and Trap :

"Traps are identical to version 2 traps and are unacknowledged notifications sent by agents to managers. Informs are acknowledged notifications. Agent sends an inform notification and waits for acknowledgment."

https://snmpsharpnet.com/index.php/snmp-version-3-notifications-traps-and-informs/

upvoted 5 times

wwisp3422112 Most Recent 1 year, 1 month ago

Remember the name "STIG" as for this question :)

upvoted 2 times

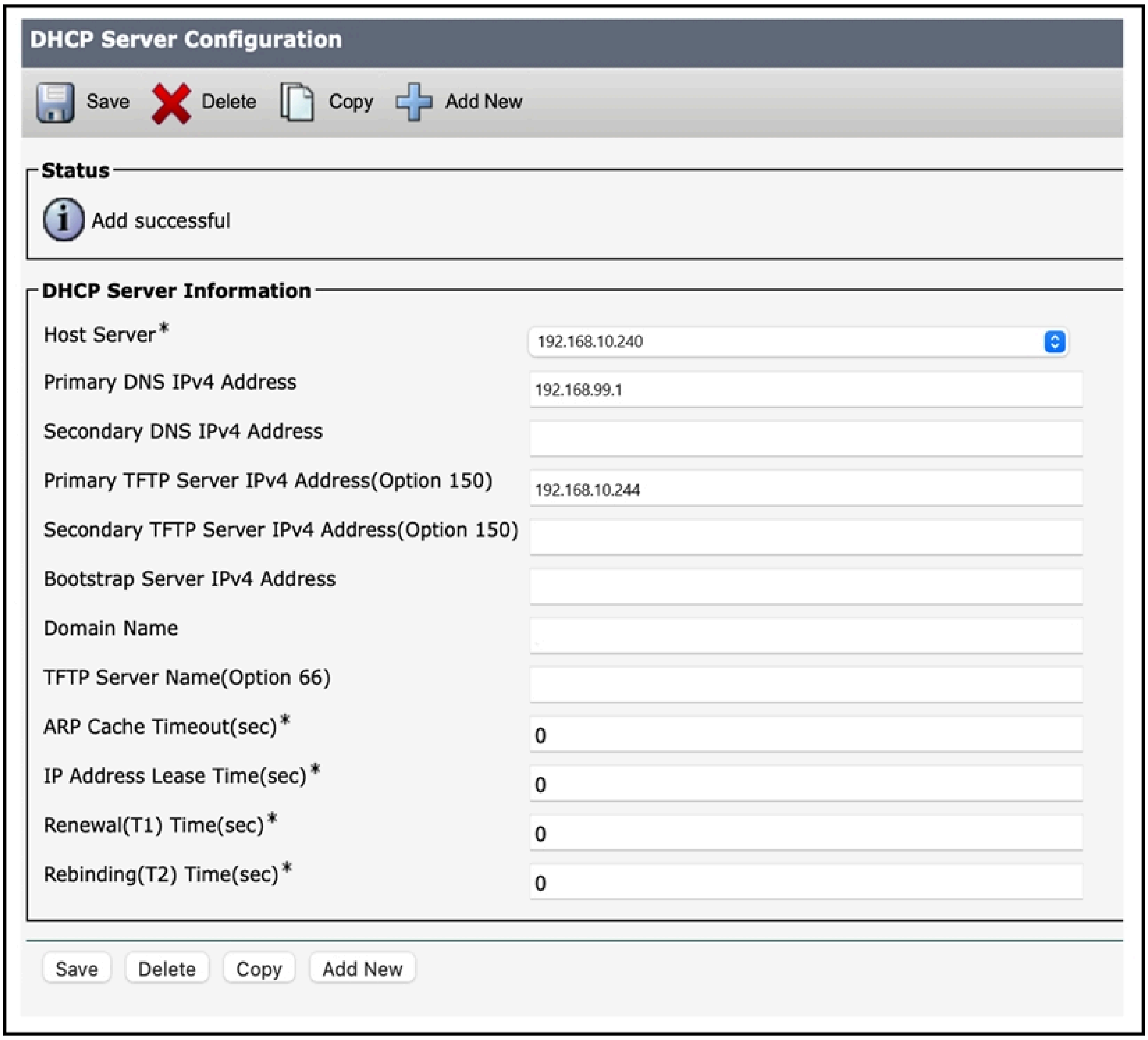
Alan100 9 months, 1 week ago

Bold of you to assume the choices cannot be reshuffled :)

upvoted 3 times

Question #61 Topic 1

Refer to the exhibit.



A collaboration engineer configures Cisco UCM to act as a DHCP server. What must be done next to configure the DHCP server?

A. Restart the TFTP service under Cisco Unified Serviceability.

B. Add a DHCP subnet to the DHCP server under Cisco UCM Administration.

C. Add the new DHCP server to the primary DNS server.

D. Restart the Cisco DHCP Monitor Service under Cisco Unified Serviceability.

Correct Answer: *B*

Community vote distribution

B (93%) 7%

ciscogeek Highly Voted 2 years, 7 months ago

Selected Answer: B

DHCP Subnet needs to be configured after DHCP Server configuration.

upvoted 8 times

G0y0 Most Recent 3 months, 4 weeks ago

Selected Answer: B

Reference:

CCNA Collaboration CICD 210-060 Official Cert Guide

Chapter 9: Managing Endpoints and End Users in CUCM

upvoted 1 times

pasangawa 7 months, 3 weeks ago

B

DHCP server and DHCP subnet are on separate menu in CUCM. DHCP subnet is where you define the subnet and IP range

upvoted 1 times

AgshinA 9 months, 3 weeks ago

Selected Answer: C

After configuring Cisco Unified Communications Manager (UCM) to act as a DHCP server, the next steps typically involve:

Defining DHCP Subnets: Create and configure DHCP subnets to define the range of IP addresses that the DHCP server can assign to clients.
Setting DHCP Options: Configure DHCP options such as the default gateway, DNS servers, and TFTP servers (Option 150 for Cisco IP phones).
Activating DHCP Monitor Service: Ensure that the DHCP Monitor Service is activated so that the DHCP server can start assigning IP addresses to clients

upvoted 1 times

AgshinA 9 months, 3 weeks ago

Sorry B. I choose wrong.

upvoted 1 times

MeowthL 1 year, 9 months ago

Selected Answer: B

B is the Correct Answer

upvoted 3 times

Piji 2 years, 5 months ago

Selected Answer: B

After "DHCP Server" configuration, the "DHCP Subnet" should be configured.

upvoted 4 times

AJBELL14 2 years, 5 months ago

Checked on other forums too. "Add a DHCP subnet to the DHCP server under Cisco UCM Administration: is the right answer

upvoted 2 times

Question #62 *Topic 1*

What are two features of Cisco Expressway that the customer gets if Expressway-I¡ and Expressway-E are deployed? (Choose two.)

A. session-based access to comprehensive collaboration for remote workers, without the need for a separate VPN client

B. highly secure firewall-traversal technology to extend organizational reach

C. complete endpoint registration and monitoring capabilities for devices that are local and remote

D. additional visibility of the edge traffic in an organization

E. utilization and adoption metrics of all remotely connected devices

Correct Answer: *AB*

Community vote distribution

AB (100%)

G0y0 3 months, 4 weeks ago

Selected Answer: AB

Cisco Expressway is designed specifically for comprehensive collaboration services. It features established firewall-traversal technology and helps redefine traditional enterprise collaboration boundaries, supporting Cisco's vision of any-to-any collaboration. As its primary features and benefits, Cisco Expressway offers proven and highly secure firewall-traversal technology to extend an organization's reach. It helps enable business-to-business, business-to- consumer, and business-to-cloud-service-provider connections. It provides session-based access to comprehensive collaboration for remote workers, without the need for a separate Virtual Private Network (VPN) client.
CCNP Collaboration Cloud and Edge Solutions CLCEI 300-820 Official Cert Guide
Chapter 1: Introduction to the Cisco Expressway Solution.

upvoted 2 times

b3532e4 9 months ago

Expressway offers the following primary features and benefits:

Provides proven, highly secure, firewall-traversal technology.

Facilitates connections for business-to-business, business-to-consumer, and business-to-cloud-service-provider.

Facilitates session-based access to collaboration services for remote workers, with no need for a separate VPN client.

Supports a wide range of devices, including Cisco Jabber for smartphones, tablets, and desktops.

Complements bring-your-own-device strategies and policies for remote and mobile workers.

upvoted 2 times

pcp84 1 year, 4 months ago

Selected Answer: AB

AB are correct.
Just putting in a voting comment so it shows on the solution.

upvoted 1 times

way2certs 2 years, 6 months ago

A and B are correct.
https://www.cisco.com/c/en/us/td/docs/voice_ip_comm/expressway/admin_guide/X14-0-2/exwy_b_cisco-expressway-administrator-guide-x1402/exwy_m_introduction.html

upvoted 2 times

Question #63 *Topic 1*

What is a capability of Cisco Expressway?

A. It functions as an analog telephony adapter.

B. It provides access to on-premises Cisco Unified Communications infrastructure for remote endpoints.

C. It has remote endpoint enrollment with Certificate Authority Proxy Function.

D. It gives directory access for remote users via Cisco Directory Integration

Correct Answer: *B*

Currently there are no comments in this discussion, be the first to comment!

Question #64 *Topic 1*

Refer to the exhibit.

```
admin:utils ntp status
ntpd (pid 17428) is running...

     remote           refid      st t when poll reach   delay   offset  jitter
==============================================================================
*192.168.1.1    17.253.14.125     2 u   39   64     3   0.456  -0.236   0.116
*192.168.1.2    17.253.14.125     2 u   38   64     3   0.817  -0.695   0.395
```

A collaboration engineer needs to replace the original, single NTP server that was configured during the initial install of a Cisco UCM server. What is the first step to accomplish this task?

A. Stop the NTP service on Cisco UCM.

B. Enable NTP authentication for the new NTP server on Cisco UCM.

C. Restart the NTP service on Cisco UCM.

D. Delete the original NTP server from Cisco UCM.

Correct Answer: *D*

Community vote distribution

D (78%) B (22%)

Piji Highly Voted 2 years, 11 months ago

Selected Answer: D

I am not agree with ciscogeek's answer, and the correct answer is D.

If you look at the exhibit on the "utils ntp status", it shows two original servers installed. And if you ready the question carefully, there is a "," after "original,", which means, from two original NTP servers, only single NTP server needs to be delete. So you can delete a single NTP server from the two listed NTP servers. However, even if there is only one server, you can't enable authentication for the new NTP server, you need first add it, then enable it with command "utils ntp auth symmetric-key enable" which you get prompt to select for enable the authentication for the listed servers:

admin:utils ntp auth symmetric-key enable

At the end, nothing in the question mentioned about NTP authentication, as you can see authentication report with the following command:

admin:utils ntp auth symmetric-key status

upvoted 9 times

MeowthL 2 years, 3 months ago

Agreed with you

upvoted 1 times

G0y0 Most Recent 3 months, 4 weeks ago

Selected Answer: D

A. is incorrect. You do no need to stop the NTP service.

B. is incorrect, it is optional.

C. is partially correct, it is not the first step, rather it is after of remove the original NTP server.

D. is correct. Just remove the actual NTP server.

upvoted 2 times

b3532e4 11 months, 2 weeks ago

Just C

upvoted 1 times

G0y0 3 months, 3 weeks ago

If you are going to make a statement without giving arguments, it is better not to give an opinion.

upvoted 1 times

OSJAY 2 years, 10 months ago

Still in doubt but I would agree with Piji. Q says there was a single original NTP server and we see 2 of them. That means the new one has already been configured and you need to Delete the original one. Or am I seeing this wrong? What makes it more confusing is that the Q asks "what is the first step". Hope I don't get this one in my upcoming test.

upvoted 1 times

ciscogeek 3 years, 1 month ago

Selected Answer: B

Original, Single NTP Server cannot be deleted unless there are two or more NTP servers. Hence we need to add one NTP server, before being able to delete the available NTP server as question says a 'single' NTP server is currently available.

upvoted 2 times

movalleuu 2 years, 3 months ago

on the exhibit, there are 2 ntp servers configured on CUCM already

upvoted 1 times

Question #65 *Topic 1*

Refer to the exhibit.

```
admin:utils ntp status
ntpd (pid 17428) is running...

     remote           refid          st t when poll reach   delay   offset   jitter
==============================================================================
*192.168.1.1     17.253.14.125     2 u   36   64   377    0.435   0.039    0.047
 192.168.1.2     .INIT.           16 u    -   64     0    0.000   0.000    0.000
```

A collaboration engineer adds a redundant NTP server to an existing Cisco Collaboration solution. On the Cisco UCM OS Administration page, the new NTP server shows as `Not Accessible`. Which action resolves this issue?

A. Delete and re-add the new NTP server via the Cisco UCM command-line interface.

B. Start the NTP service on the new NTP server.

C. Configure the ג€reachג€ value as ג€377ג€ for the new NTP server.

D. Restart NTPD on the Cisco UCM server.

Correct Answer: *B*

Community vote distribution

B (100%)

G0y0 3 months, 3 weeks ago

Selected Answer: B

It happens when the NTP server is not running its NTP service or because the new added NTP server is not a correct source. For example, if you try to add the IP address of your windows 10 PC, you can get this "Not Accessible".
A. could not be a good strategy if you are adding an incorrect NTP source.
C. does not have sense.
D. SEEMS to be incorrect, since they are not presenting the full output of "utils status", in the part that says if it is "synchronized" or "unsynchronized". It is stupid that they do not present the full output of the command.

So I think correct answer is B, NTP service is not running on the NTP server or a not supportable NTP source was added.

upvoted 1 times

G0y0 3 months, 3 weeks ago

errata: "utils ntp status"

upvoted 1 times

b3532e4 11 months, 2 weeks ago

Just B

upvoted 1 times

pasangawa 1 year, 1 month ago

B should be correct.
When you add NTP, it will automatically restart NTP service so there's no need to restart again so issue might be on new NTP server itself and not on CUCM

upvoted 1 times

skneff 1 year, 5 months ago

Selected Answer: B

Answer is B

upvoted 2 times

sandman0409 1 year, 10 months ago

Selected Answer: B

B should be the correct one

upvoted 2 times

aocstr 1 year, 11 months ago

Selected Answer: B

I agree this would be a ntp server not started or network issue. Answer B.

upvoted 2 times

pacciolli 2 years, 2 months ago

B is no correct. If the service were down the first NTP sever was not active. For me the A is correct

upvoted 1 times

driz 2 years, 1 month ago

they are two different servers, just because NTP is active on 192.168.1.1 does not mean it's active on 192.168.1.2. B, starting the service on 192.168.1.2 is the most appropriate answer.

upvoted 3 times

marjana_mirza 2 years, 2 months ago

Selected Answer: B

If its no accessibile then ntp service might need to start

upvoted 4 times

Mert_kerna 2 years, 5 months ago

It appears the service isn't running on the second NTP server. I would validate that the service is running. If it's not running, Start the service. If it is running, there's likely an issue with the service running. Proceed by restarting the service

upvoted 3 times

wwisp3422112 2 years, 7 months ago

https://www.examtopics.com/discussions/cisco/view/75414-exam-350-801-topic-1-question-65-discussion/

B

upvoted 3 times

wwisp3422112 2 years, 7 months ago

Is A correct here?

upvoted 1 times

papahawaii 2 years, 4 months ago

No, B is, as the question states not accessible = it is not running or a network issue. Only response in answers to fix this issue would be to start the server thus B.

upvoted 3 times

Question #66 *Topic 1*

Which action is required for a firewall configuration on a Mobile and Remote Access through Cisco Expressway deployment?

A. The external firewall must allow these inbound connections to Expressway: SIP: TCP 5061: HTTPS: TCP 8443; XMPP: TCP 5222; Media: UDP 36002 to 59999.

B. The internal firewall must allow these inbound and outbound connections between Expressway-I¡ and Expressway-E: SIP: HTTPS (tunneled over SSH between I¡ and E): TCP 2222: TCP 7001; Traversal Media: UDP 2776 to 2777 (or 36000 to 36011 for large VM/appliance); XMPP: TCP 7400.

C. Do not use a shared address for Expressway-E and Expressway-I¡, as the firewall cannot distinguish between them. If static NAT for IP addressing on Expressway-E is used, ensure that any NAT operation on Expressway-I¡ does not resolve the same traffic IP address. Shared NAT is not supported.

D. The traversal zone on Expressway-I¡ points to Expressway-E through the peer address field on the traversal zone, which specifies the Expressway-E server address. For dual NIC deployments, set the Expressway-E address using an FQDN that resolves the IP address of the internal interface.

Correct Answer: *A*

Community vote distribution

A (100%)

ciscogeek Highly Voted 3 years, 1 month ago

Selected Answer: A

https://www.cisco.com/c/en/us/td/docs/voice_ip_comm/expressway/config_guide/X14-0-2/mra/exwy_b_mra-deployment-guide-x1402.pdf

upvoted 5 times

H31d1 2 years, 8 months ago

P. 14 says A C and D are correct?

upvoted 2 times

G0y0 3 months, 3 weeks ago

C. and D. are correct, however, they apply as for B2B deployments as for MRA deployment. A. and B. are focused to RMA what is the focus of the answer, and finally just remain A.

upvoted 1 times

G0y0 3 months, 3 weeks ago

In fact, all of the four are correct, however, B makes the mistake of saying that the inbound firewall should allow inbound and outbound connections, which is a mistake. The internal firewall should only have outbound rules, from Exp-C to Exp-E.

upvoted 1 times

way2certs 2 years, 6 months ago

Indeed. As the question asks about firewall configuration , just A seems relevant out of the three.

upvoted 2 times

G0y0 Most Recent 3 months, 3 weeks ago

Selected Answer: A

Well, let us see:

Actually, all of the four answers are correct, they just differ in the context.

C. and D. are correct, even though they apply both as for B2B as for MRA, as for a Traversal Client/Server Zone as for a Unified Communications Traversal Zone. Remember the question is asking just for MRA.

B. is partially correct, even the port usage is correct, the truth is that no inbound ports are required to be opened on the internal firewall. The internal firewall must allow only outbound connections from the Expressway-C to the Expressway-E.

A. is the most appropriate. The external firewall must allow inbound connections to the Expressway-E: SIP (TCP 5061); HTTPS (TCP 8443); XMPP (TCP 5222); Media (UDP 36002 to 59999)

upvoted 1 times

G0y0 3 months, 3 weeks ago

Reference:

Cisco Expressway IP Port Usage Configuration Guide (X14.0);

CCNP Collaboration Cloud and Edge Solutions CLCEI 300-820 Official Cert Guide.

upvoted 1 times

b3532e4 10 months ago

A. The external firewall and B. The internal firewall is correct but in this Q not mention it Which firewall?

My opinion C is Correct

upvoted 1 times

AgshinA 1 year, 3 months ago

Selected Answer: A

Guide says exactly the same: The external firewall must allow the following inbound connections to Expressway: SIP: TCP 5061; HTTPS: TCP 8443; XMPP: TCP 5222; Media: UDP 36002 to 59999.

https://www.cisco.com/c/en/us/td/docs/voice_ip_comm/expressway/config_guide/X14-2/mra/exwy_b_mra-deployment-guide-x142/exwy_m_requirements-for-mra.html

upvoted 4 times

Question #67 *Topic 1*

The IP phones at a customer site do not pick an IP address from the DHCP. An engineer must temporarily disable LLDP on all ports of the switch to leave only
CDP. Which two commands accomplish this task? (Choose two.)

A. Switch(config)# no lldp transmit

B. Switch# configure terminal

C. Switch# copy running-config startup-config

D. Switch(config)# no lldp run

E. Switch(config)# interface GigabitEthernet1/0/1

Correct Answer: *BD*

Community vote distribution

BD (100%)

Ol_Mykhailiuk Highly Voted 10 months ago

Selected Answer: BD

To enable LLDP, use the lldp run command. To disable LLDP, use the no lldp run command.

https://www.cisco.com/c/en/us/td/docs/routers/nfvis/switch_command/b-nfvis-switch-command-reference/b-nfvis-switch-command-reference_chapter_010000.html#:~:text=To%20enable%20LLDP%2C%20use%20the,the%20no%20lldp%20run%20command.

upvoted 5 times

Mert_kerna Most Recent 1 year, 5 months ago

The answers are correct on how you'd disable LLDP, but what kind of question is this? Why would you even need to disable LLDP when CDP is the first protocol the phones will use.. I have myriads of active deployments where both CDP and LLDP are enabled on the switches. The phones ALWAYS use CDP unless CDP is disabled from the phone's configuration or if the switch doesn't support CDP. Power and VLAN configurations are negotiated by CDP if CDP is available. If it's not, LLDP will STILL work! So why is there a need to disable it? Because it's not proprietary to Cisco? So what?

upvoted 2 times

Mert_kerna 1 year, 5 months ago

They need to change the wording on this question to something like, "ABC company policy is that LLDP is not allowed to run on switching infrastructure. Out of the following configurations, select the procedures to disable LLDP." Please correct me if I'm wrong, but the wording on this question is misleading to how the technology even works.

upvoted 3 times

Question #68 *Topic 1*

To provide high-quality voice and take advantage of the full voice feature set, which two access layer switches provide support? (Choose two.)

A. Use multiple egress queues to provide priority queuing of RTP voice packet streams and the ability to classify or reclassify traffic and establish a network trust boundary.

B. Deploy RSVP to improve VoIP QoS only where it can have a positive impact on quality and functionality where there is limited bandwidth and frequent network congestion.

C. Map audio and video streams of video calls (AF41 and AF42) to a class-based queue with weighted random early detection.

D. Use 802.1Q trunking and 802.1p for proper treatment of Layer 2 CoS packet marking on ports with phones connected.

E. Implement IP RTP header compression on the serial interface to reduce the bandwidth required per voice call on point-to-point links.

Correct Answer: *AD*
Reference:
https://www.cisco.com/c/en/us/td/docs/voice_ip_comm/cucme/srnd/design/guide/cmesrnd/nstrct.html

wwisp3422112 Highly Voted 7 months ago
Correct A and D:

To provide high-quality voice and to take advantage of the full voice feature set, access layer switches should provide support for:

•802.1Q trunking and 802.1p for proper treatment of Layer 2 CoS packet marking on ports with phones connected

•Multiple egress queues to provide priority queuing of RTP voice packet streams

•The ability to classify or reclassify traffic and establish a network trust boundary

•Inline power capability (Although inline power capability is not mandatory, we highly recommend for the access layer switches.)

•Layer 3 awareness and the ability to implement QoS access control lists (These features are required if you are using certain IP telephony endpoints, such as a PC running a softphone application, that cannot benefit from an extended trust boundary.)

upvoted 8 times

Question #69 *Topic 1*

A customer wants to conduct B2B video calls with a partner using an on-premises conferencing solution. Which two devices are needed to facilitate this request?
(Choose two.)

A. Expressway-C

B. MGCP gateway

C. Cisco Unified Border Element

D. Cisco TelePresence Management Suite

E. Expressway-E

Correct Answer: *AE*

Community vote distribution

AE (100%)

skneff 11 months, 3 weeks ago

Selected Answer: AE

A and E are correct

upvoted 2 times

Question #70 *Topic 1*

A company wants to provide remote users with access to its on-premises Cisco collaboration features. Which components are required to enable Cisco Mobile and Remote Access for the users?

A. Cisco Unified Border Element, Cisco IM and Presence Server, and Cisco Video Communication Server

B. Cisco Unified Border Element, Cisco UCM, and Cisco Video Communication Server

C. Cisco Expressway-E, Cisco Expressway-C, and Cisco UCM

D. Cisco Expressway-E, Cisco IM and Presence Server, and Cisco Video Communication Server

Correct Answer: *C*

Reference:

https://www.cisco.com/c/en/us/td/docs/solutions/CVD/Collaboration/enterprise/12x/120/collbcvd/edge.html

Community vote distribution

C (100%)

skneff 11 months, 3 weeks ago

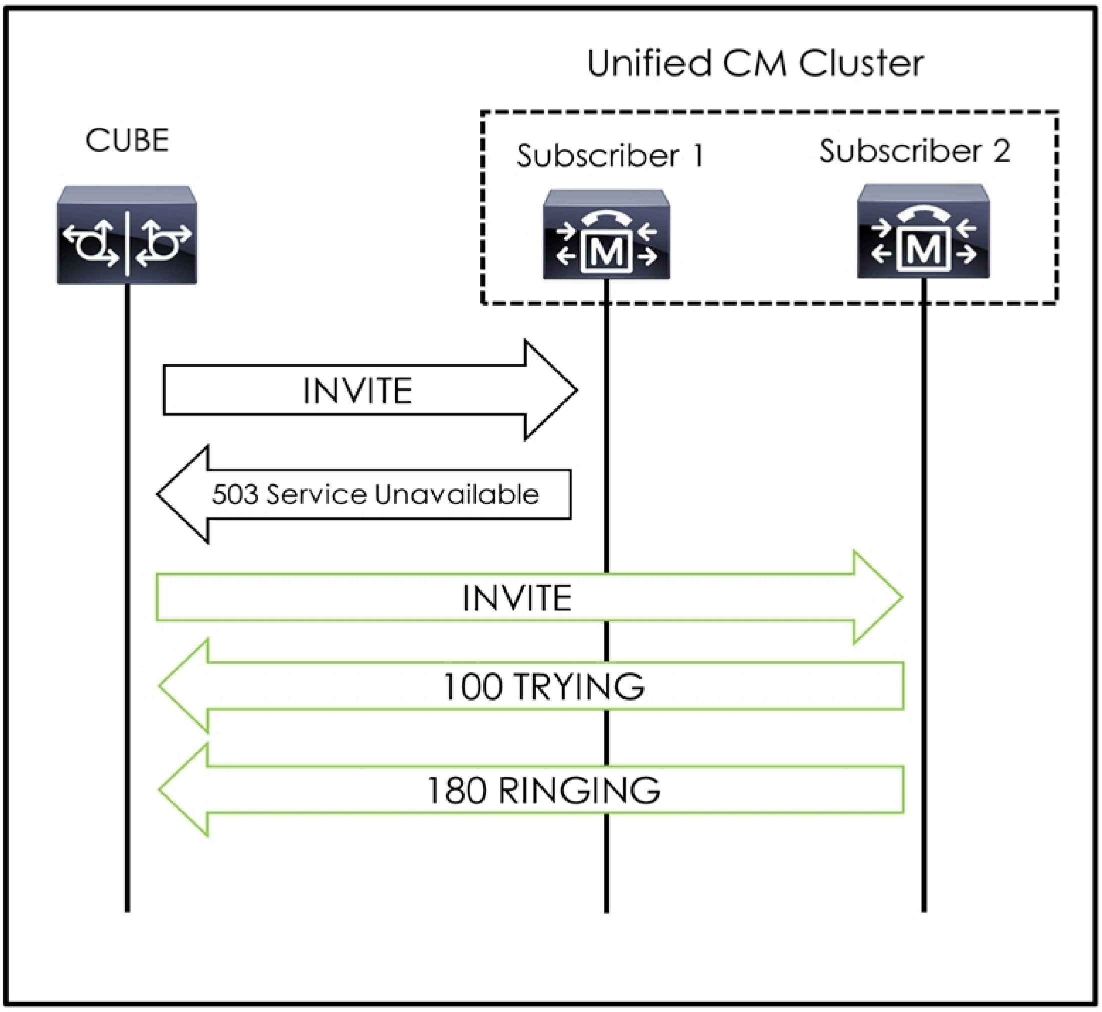
Selected Answer: C

C is correct

upvoted 1 times

Question #71 Topic 1

Refer to the exhibit.



Cisco Unified Border Element is attempting to establish a call with Subscriber 1, but the call fails. Cisco Unified Border Element then retries the same call with
Subscriber 2, and the call proceeds normally. Which action resolves the issue?

A. Verify that the Run On All Active Unified CM Nodes checkbox is enabled.

B. Verify that the correct calling search space is selected for the Inbound Calls section.

C. Verify that the Significant Digits field for Inbound Calls is set to All.

D. Verify that the PSTN Access checkbox is enabled.

Correct Answer: *A*

Community vote distribution

A (100%)

Slushed Highly Voted 2 years, 8 months ago

The correct answer is A. Run On All Unified CM Nodes allows Subscribers NOT configured within the Call Manager Group that is applied to Route Lists and/or SIP Trunks to process calls.

There is a whole rabbit whole with how this feature works when only applied to one or the other but per best practice, unless there is a VERY specific reason, you want this check box enabled on ALL Route Lists and SIP Trunks.

upvoted 11 times

G0y0 2 years, 7 months ago

I agree, it is A. 503 could be a symptom of overload or congestion in subscriber 1 due to bad call load balancing. Therefore, Run On All Unified CM Nodes allows a better call load balancing to avoid overloading over one server, and then a better call distribution.

upvoted 5 times

Obama42 Highly Voted 2 years, 5 months ago

Selected Answer: A

there is no reason to be inbound CSS, the sub2 get answered.Then the correct answer is A.

upvoted 5 times

skneff 11 months, 3 weeks ago

I was torn between A and B, but A makes sense for that exact reason. If both servers are in the same cluster, their CSS configuration would be the same, thus no reason to believe B would be the solution.

There could be other reasons that you get a 503 Service unavailable, but A is the only possible reason listed.

upvoted 1 times

G0y0 Most Recent 3 months, 3 weeks ago

Selected Answer: A

Commonly when there is a issue about Call Privileges, you get a reject or a 404 code. Meanwhile 5XX is a problem of the servers cause it can not provide temporally the service the client is looking for. So it could be that the call manager service is no running in that server, or could be that in that subscriber is not running an instance of that sip trunk, or any other issue with the application, so it can bot be B. or C. Meanwhile C. is a nonsense.

upvoted 1 times

DaKenjee 2 years, 1 month ago

Selected Answer: A

503 - Service Unavailable - The server's SIP service is temporarily unavailable.
If CSS is missing it would not find a proper DN, and will return <404 - not found<

Since there is no answer of several Trunks, it only explains one trunk and >Run On All Unified CM Nodes< should put in concern

upvoted 3 times

Piji 2 years, 4 months ago

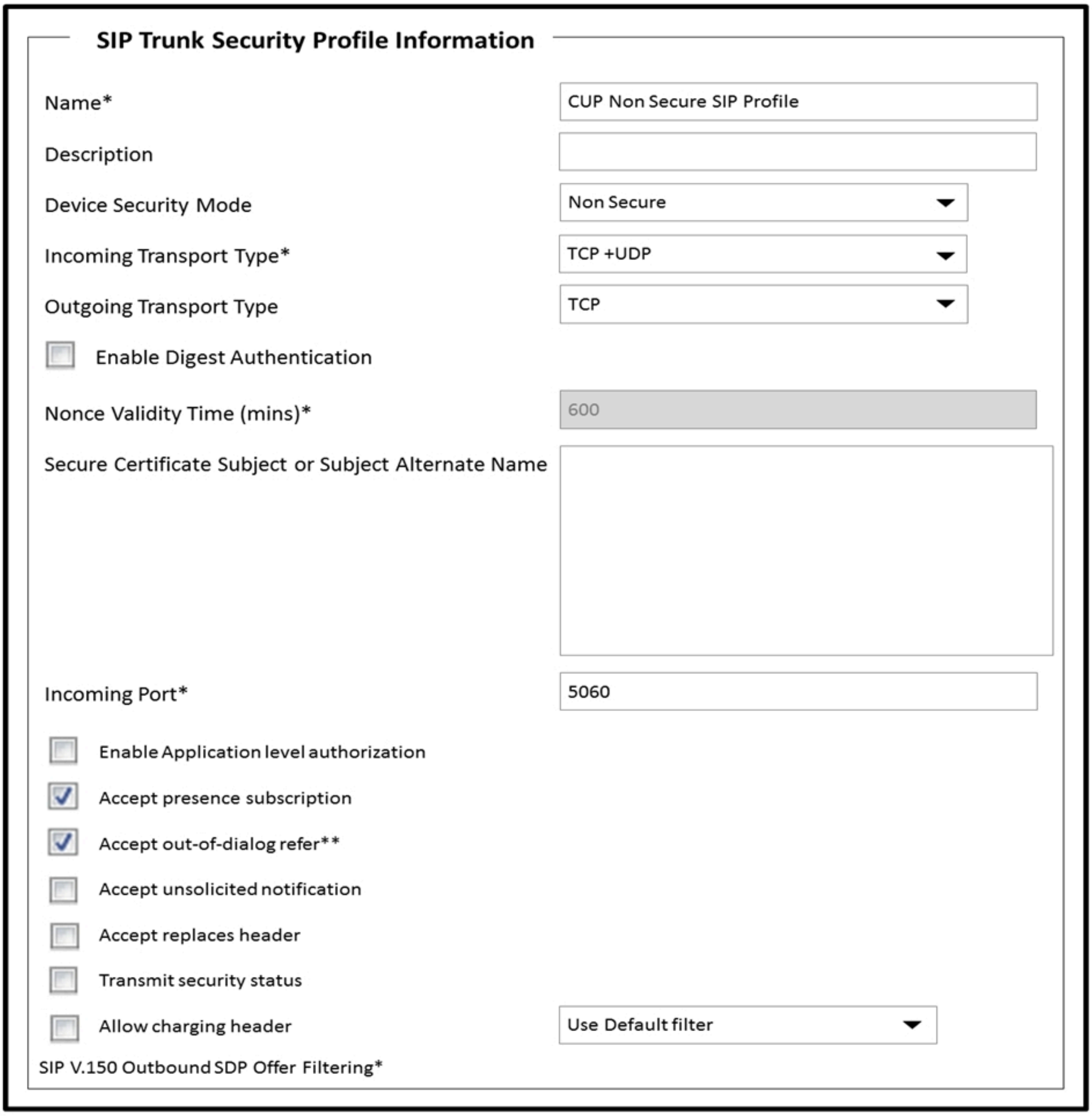
Selected Answer: A

Correct answer is A.

upvoted 3 times

Question #72 *Topic 1*

Refer to the exhibit.



SIP Trunk Security Profile Information

Name*	CUP Non Secure SIP Profile
Description	
Device Security Mode	Non Secure
Incoming Transport Type*	TCP +UDP
Outgoing Transport Type	TCP
☐ Enable Digest Authentication	
Nonce Validity Time (mins)*	600
Secure Certificate Subject or Subject Alternate Name	
Incoming Port*	5060
☐ Enable Application level authorization	
☑ Accept presence subscription	
☑ Accept out-of-dialog refer**	
☐ Accept unsolicited notification	
☐ Accept replaces header	
☐ Transmit security status	
☐ Allow charging header	
SIP V.150 Outbound SDP Offer Filtering*	Use Default filter

A collaboration engineer is configuring the Cisco UCM IM and Presence Service. Which two steps complete the configuration of the SIP trunk security profile?
(Choose two.)

A. Check the box to accept replaces header.

B. Check the box to allow charging header.

C. Check the box to enable application-level authorization.

D. Check the box to transmit security status.

E. Check the box to accept unsolicited notification.

Correct Answer: *AE*

Community vote distribution

AE (100%)

Omitted Highly Voted 3 years, 1 month ago

Selected Answer: AE
Check the following check boxes:

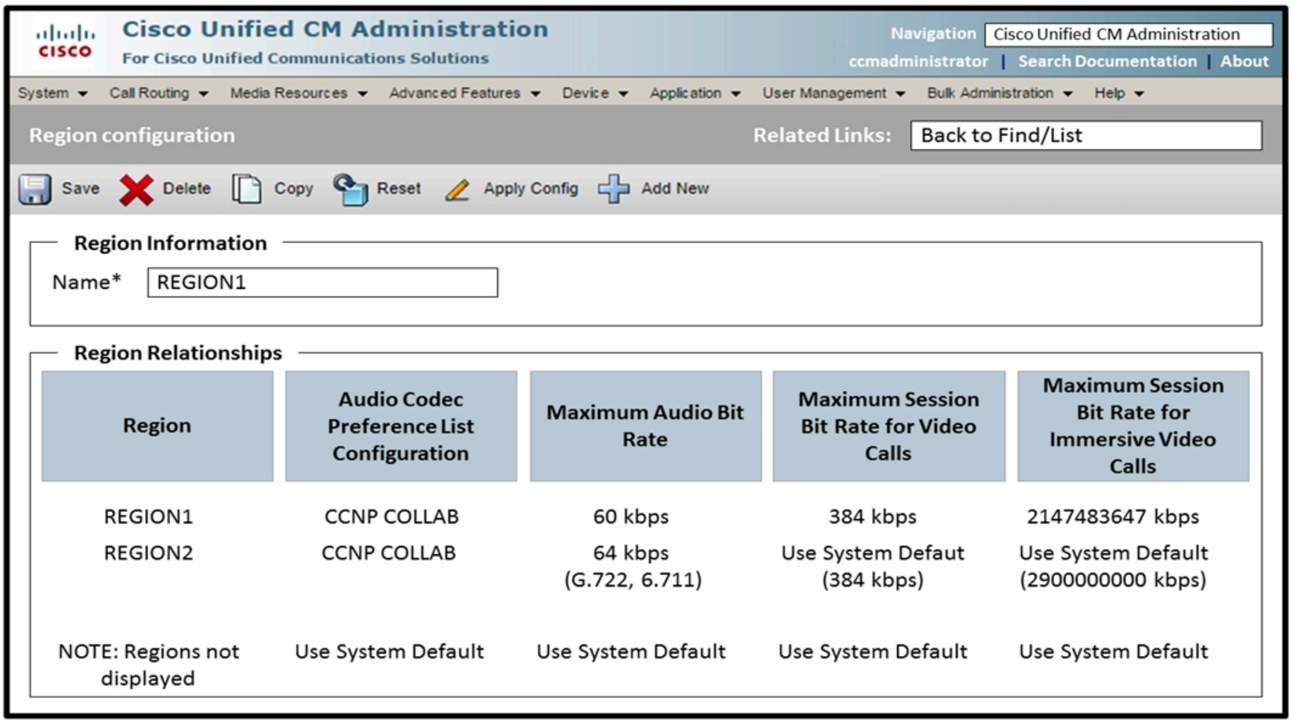
Accept Presence Subscription
Accept Out-of-Dialog REFER
Accept Unsolicited Notification
Accept Replaces Header

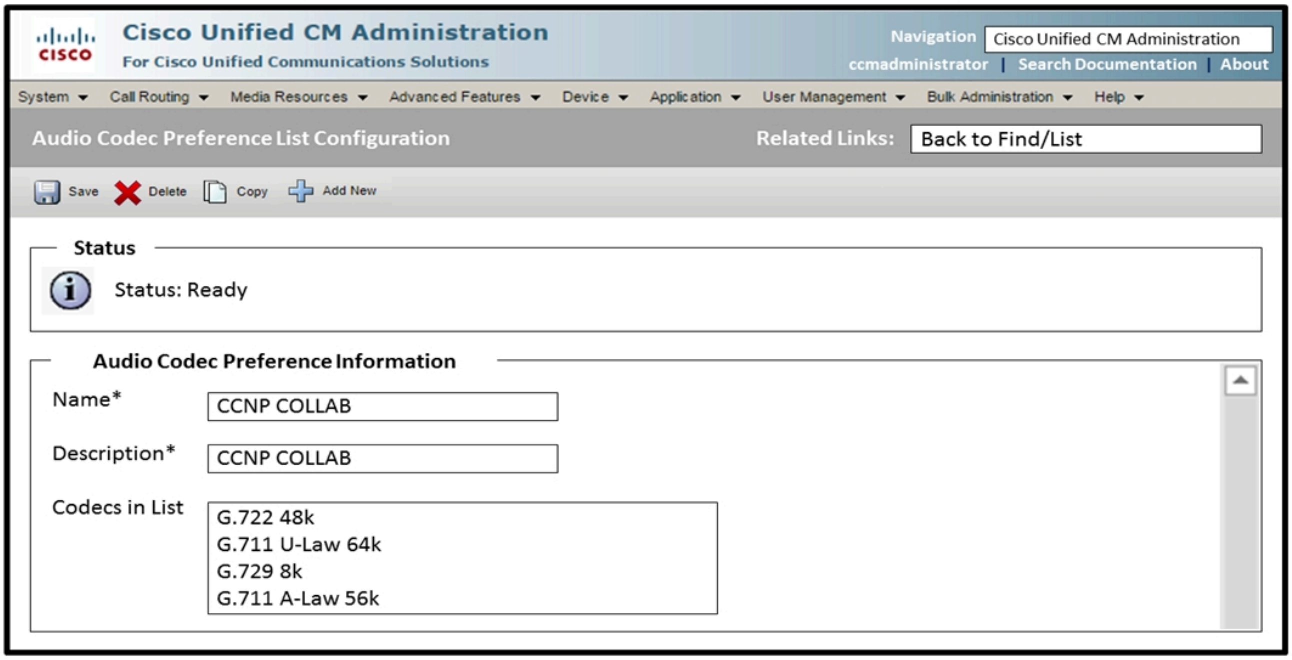
https://www.cisco.com/c/en/us/td/docs/voice_ip_comm/cucm/im_presence/configAdminGuide/11_5_1/cup0_b_config-and-admin-guide-1151su5/cup0_b_imp-system-configuration-1151su5_chapter_0111.html#task_441822D5A33E7A9AEB98901FCE5DA3D8
upvoted 6 times

b3532e4 Most Recent 9 months ago
Accept Presence Subscription
Accept Out-of-Dialog REFER
Accept Unsolicited Notification
Accept Replaces Header
upvoted 1 times

vip211 1 year ago
if i did not check anything what will happend
upvoted 1 times

Refer to the exhibit.





An engineer is troubleshooting this video conference issue:

✑ A video call between a Cisco 9971 in Region1 and another Cisco 9971 in Region1 works.

✑ As soon as the Cisco 9971 in Region1 conferences in a Cisco 8945 in Region2, the Region1 endpoint cannot see the Region2 endpoint video.

What is the cause of this issue?

A. Cisco 8945 does not have a camera connected.

B. Maximum Audio Bit Rate must be increased.

C. Maximum Session Bit Rate for Video Calls is too low.

D. Maximum Session Bit Rate for Immersive Video Calls is too low.

Correct Answer: *C*

Community vote distribution

C (70%) A (26%) 4%

Slushed Highly Voted 2 years, 8 months ago

Selected Answer: C

This is one of those "trick" questions that really does not have a clear answer and is more in "reading between the lines," in typical Cisco exam fashion.

Answer A: This cannot be right because that would be an assumption. 8945s have built-in video/camera.

Answer B: The codecs in the list and max bitrate show no issue, so this is not it.

Answer C: This is the best choice based on the information provided. The section in the question of "when the 9971 CONFERENCES IN the 8945..." indicating that the 8945 is being added to an already in-progress video call, totaling three or more video streams. This would mean 384kbps is definitely way too low.

Answer D: This is wrong because neither of these devices are immersive (not SX, DX, MX, IX, etc., aka Telepresence).

upvoted 18 times

ciscogeek Highly Voted 2 years, 7 months ago

Selected Answer: A

I believe the answer should be A, even with the 384 kbps bandwidth, the video calls and conferences are allowed and they do work, with lower resolution.

Answer C could be right when the question is about 'low video quality'

upvoted 6 times

MeowthL 1 year, 9 months ago

i'm agreed with you

upvoted 3 times

G0y0 Most Recent 5 months ago

Selected Answer: A

It is very "tricky" this question, it is my point of view:

A. could be just an assumption (as Slushed told us, because CP8945 has built-in camera.

B. The audio bit rate nothing to do here.

C. As Slushed told us, there are three video stream, however, there is not information about the locations configuration, just regions. Also SIF (352x240) @ 30fps and QCIF (176x144) @ 30fps are also below of 384 kbps.

D. it is not inmersive video.

So I think, in order of the presented information, it could fall in the assumption that the Phone has not a connected camera, the closest answer could be A.

upvoted 2 times

G0y0 5 months ago

both phones can support QCIF (176 x 144 pixels), which plus 48kbps (g722), the bandwitht of 384 could be enough

upvoted 1 times

Islam_Muhammad7 9 months, 3 weeks ago

I think C is correct not A, that because Cisco 8945 has a built-in Camera so, how the camera is not connected, maybe the shutter is closed but this is not an answer in the choices so, C is the best choise in this situation.

upvoted 2 times

Ruuddie 2 years, 2 months ago

Does a 9971 even have a video bridge built in? It's not stated anywhere they have a complete CMS solution running. Perhaps it's to be expected that the conference becomes audio-only.

upvoted 1 times

Piji 2 years, 4 months ago

Selected Answer: B

Another tricky question. I go with answer B, and the audio should be connected and then the video. Low bandwidth on audio REGION1 is 60kbps, and can't make audio to REGION2 which is 64kbps, no bandwidth the call disconnected.

upvoted 1 times

Panda_man 1 year, 11 months ago

As you can see region 2 has larger value for bandwidth for audio - there is no valid reason why audio call would fail.

upvoted 3 times

Question #74 *Topic 1*

If a phone needs to register with cucm1.cisco.com, which network service assists with the phone registration process?

A. SMTP

B. ICMP

C. DNS

D. SNMP

Correct Answer: *C*

OSJAY 9 months, 1 week ago

To resolve the CM fqdn cucm1.cisco.com you need DNS working

upvoted 1 times

Littlelarry123 1 year, 9 months ago

If you don't know this........... then why are you here cuz

upvoted 2 times

unbelievable 1 year, 8 months ago

The issue is always DNS

upvoted 1 times

Question #75 *Topic 1*

Which type of input is required when configuring a third-party SIP phone?

A. digest user

B. serial number

C. manufacturer

D. authorization code

Correct Answer: *A*

Community vote distribution

A (100%)

ademozipek 10 months, 1 week ago

Selected Answer: A

It's A.

upvoted 3 times

Panda_man 1 year, 5 months ago

Selected Answer: A

It's A following the documentation in this link :

https://www.cisco.com/c/en/us/td/docs/voice_ip_comm/cucm/admin/12_0_1/systemConfig/cucm_b_system-configuration-guide-1201/cucm_b_system-configuration-guide-1201_chapter_0101100.html

upvoted 4 times

Question #76 *Topic 1*

An administrator configures Cisco UCM to use UDP for SIP signaling and finds that an endpoint cannot make calls. Which action resolves this issue?

A. Change the common phone profile.

B. Change the SIP dial rules.

C. Change the SIP profile.

D. Change the phone security profile.

Correct Answer: *D*

Community vote distribution

D (100%)

ciscogeek Highly Voted 3 years, 1 month ago

Selected Answer: D

Answer seems right, as the particular phone's security profile might be configured for TCP only, hence , UDP can be allowed in phone security profile.

https://www.cisco.com/c/en/us/td/docs/voice_ip_comm/cucm/security/11_5_1/secugd/CUCM_BK_SEE2CFE1_00_cucm-security-guide-1151/CUCM_BK_SEE2CFE1_00_cucm-security-guide-1151_chapter_0111.html

upvoted 6 times

b3532e4 Most Recent 9 months, 1 week ago

D. Change the phone security profile

upvoted 1 times

Slushed 3 years, 1 month ago

None of these are correct as I am guessing whoever typed it up made a mistake. The correct answer, which is not present, should be SIP Trunk Security Profile. Within that configuration you can set the Incoming and Outgoing transport types (TCP, UDP, TLS where applicable).

upvoted 2 times

Omitted 3 years, 1 month ago

You can set the transport type on the SIP trunk security profile (incoming and outgoing) but also you can set the transport type on the any phone security profile. So the answer must be D here.

upvoted 3 times

skneff 1 year, 5 months ago

Exactly. It is not known from the question that the call needs to traverse a SIP trunk anyway. It could be a local call. Whereas if it was a Phone security profile issue it would be the problem no matter what.

upvoted 1 times

Question #77 *Topic 1*

An engineer implements a new Cisco UCM based telephony system per these requirements:
✑ The local Ethernet bandwidth is sized based on the total bandwidth per call.
✑ A G.736 codec is used.
✑ The bit rate is 64 kbps.
✑ The codec sample interval is 10 ms.
✑ The voice payload size is 160 bytes per 20 ms.
What should the size of the Ethernet bandwidth be per call?

A. 31.2 kbps

B. 38.4 kbps

C. 55.2 kbps

D. 87.2 kbps

Correct Answer: *D*

Community vote distribution

D (100%)

G0y0 3 months, 4 weeks ago

Selected Answer: D

It is just like the G.711 for ethernet frames.

upvoted 1 times

b3532e4 9 months ago

Calculating the overhead:

IP header: 20 bytes

UDP header: 8 bytes

RTP header: 12 bytes

Ethernet header: 18 bytes

Total overhead per packet:

20 (IP) + 8 (UDP) + 12 (RTP) + 18 (Ethernet) = 58 bytes

Total packet size:

Voice payload: 160 bytes

Total packet size: 160 (payload) + 58 (overhead) = 218 bytes

Bandwidth calculation:

Packet size: 218 bytes * 8 bits/byte = 1744 bits per packet

Since the codec sample interval is 20 ms, there are 50 packets per second (1000 ms / 20 ms).

Total bandwidth: 1744 bits/packet * 50 packets/second = 87,200 bps or 87.2 kbps

Thus, the size of the Ethernet bandwidth per call is:

D. 87.2 kbps

upvoted 3 times

AgshinA 1 year, 3 months ago

Selected Answer: D

Voice Payload Size: Given as 160 bytes every 20 ms.

Packets Per Second (PPS): Since the voice payload is sent every 20 ms, there are 50 packets per second (1000 ms / 20 ms = 50).

Total Packet Size: This includes the voice payload plus the overhead from the Layer 2 (L2) header and the IP/UDP/RTP headers. The typical L2 header for Ethernet is 18 bytes, and the IP/UDP/RTP header is 40 bytes.

Bandwidth Calculation: The total bandwidth per call is the total packet size multiplied by the PPS and then converted to bits per second (since 1 byte = 8 bits).

Using the formula:

Bandwidth=(L2 header+IP/UDP/RTP header+voice payload size)×PPS×8

We can plug in the values:
Bandwidth=(18 bytes+40 bytes+160 bytes)×50×8
Bandwidth=(218 bytes)×50×8
Bandwidth=10900 bytes/s×8
Bandwidth=87200 bits/s or 87.2 kbps
So, the Ethernet bandwidth size per call should be 87.2 kbps
upvoted 1 times

Testme1235 2 years, 4 months ago
Selected Answer: D
To calculate the Ethernet bandwidth per call, we need to take into account the total number of bytes per second in each direction (transmit and receive) and add additional overhead for Ethernet, IP, and UDP headers.

The total number of bytes per second is calculated as follows:

160 bytes per 20 ms = (160 bytes/20 ms) x (50 packets/s) = 8000 bytes/s

The bit rate is 64 kbps, so we need to add an additional 8 kbps for overhead:

64 kbps + 8 kbps = 72 kbps

The total number of bytes per second including overhead is:

72,000 bps / 8 bits per byte = 9,000 bytes/s

Adding additional overhead for Ethernet, IP, and UDP headers, we can estimate that the total number of bytes per second will be approximately 12,000 bytes/s in each direction. Therefore, the Ethernet bandwidth per call should be:

12,000 bytes/s x 8 bits per byte = 96 kbps

Therefore, the correct answer is D. 87.2 kbps is not sufficient to support the required bandwidth per call.
upvoted 3 times

DaKenjee 2 years, 7 months ago
Selected Answer: D
87.2 kbps; same here.
it is comparable with g.711, g.736 is not existing

https://www.cisco.com/c/en/us/support/docs/voice/voice-quality/7934-bwidth-consume.html
-> VoIP - Per Call Bandwidth
upvoted 2 times

Dailow 3 years, 1 month ago
Answer should be D.
G.736 doesn't exist as a codec. The figures line up with g711 or g722, but the codec isn't the important part. If we take the numbers given and run the calculations, the answer is 87.2 Kbps.

https://www.cisco.com/c/en/us/support/docs/voice/voice-quality/7934-bwidth-consume.html
upvoted 4 times

ciscogeek 3 years, 1 month ago
Selected Answer: D
All details match with G711 protocol, for which the calculation is :
(160+(40+18))/160 x 64 =87.2 kbps

https://www.cisco.com/c/en/us/support/docs/voice/voice-quality/7934-bwidth-consume.html

Codec Information Bandwidth Calculations
Codec & Bit Rate (Kbps) Codec Sample Size (Bytes) Codec Sample Interval (ms) Mean Opinion Score (MOS) Voice Payload Size (Bytes) Voice Payload

Size (ms) Packets Per Second (PPS) Bandwidth MP or FRF.12 (Kbps) Bandwidth w/cRTP MP or FRF.12 (Kbps) Bandwidth Ethernet (Kbps)
G.711 (64 Kbps) 80 Bytes 10 ms 4.1 160 Bytes 20 ms 50 82.8 Kbps 67.6 Kbps 87.2 Kbp

upvoted 3 times

Question #78 *Topic 1*

Which two steps are required for bulk configuration transactions on the Cisco UCM database utilizing BAT? (Choose two.)

A. A server template must be created in Cisco UCM.

B. A data file in Extensible Markup Language format by uploaded to Cisco UCM.

C. A data file in Abstract Syntax Notation One format by uploaded to Cisco UCM.

D. A device template must be created in Cisco UCM.

E. A data file in comma-separated values format must be uploaded to Cisco UCM.

Correct Answer: *DE*

Community vote distribution

DE (100%)

Stevon 4 weeks, 1 day ago

Selected Answer: DE

Two key steps are essential for performing bulk configuration transactions using the Bulk Administration Tool (BAT) on the Cisco Unified Communications Manager (UCM) database:

1. Creating a BAT Template:

This involves defining a template with the common settings for all the devices you're configuring, similar to how you'd manually add a device in the UCM administration interface. For example, a Cisco IP Phone 7960 template can be configured with specific device pool, location, and calling search space settings.

2. Using a CSV Data File:

A comma-separated value (CSV) data file is created to contain the unique or variable information for each individual device in the transaction. This might include details like directory numbers, MAC addresses, or other specific configurations. The CSV data file works in conjunction with the BAT template, providing the unique values that differentiate each device being configured.

upvoted 1 times

G0y0 3 months, 4 weeks ago

Selected Answer: DE

Reference: "Cisco Unified Communications Manager Bulk Administration Tool (BAT)"

https://www.cisco.com/en/US/docs/voice_ip_comm/cucm/bat/9_0_1/CUCM_BK_C22BD805_00_cucm-bulk-administration-guide_chapter_01.pdf

upvoted 1 times

JoeC716 7 months, 4 weeks ago

Selected Answer: DE

D & E would be the correct for backups since you need a CSV and a template

upvoted 1 times

Panda_man 1 year, 11 months ago

Selected Answer: DE

For bulk configuration transactions on the Cisco Unified Communications Manager database, the BAT process uses two components: a template for the device type and a data file in comma separated value (CSV) format that contains the unique values for configuring a new device or updating an existing record in the database. The CSV data file works in conjunction with the device template.

upvoted 2 times

Question #79 Topic 1

A collaboration engineer troubleshoots issues with a Cisco IP Phone 7800 Series. The IPv4 address of the phone is reachable via ICMP and HTTP, and the phone is registered to Cisco UCM. However, the engineer cannot reach the CLI of the phone. Which two actions in Cisco UCM resolve the issue? (Choose two.)

A. Enable Settings Access under Product Specific Configuration Layout in Cisco UCM.

B. Enable FIPS Mode under Product Specific Configuration Layout in Cisco UCM.

C. Enable SSH Access under Product Specific Configuration Layout in Cisco UCM.

D. Set a username and password under Secure Shell Information in Cisco UCM.

E. Disable Web Access under Product Specific Configuration Layout in Cisco UCM.

Correct Answer: *CD*

Community vote distribution

CD (100%)

ciscogeek Highly Voted 1 year, 1 month ago

Selected Answer: CD

Enable ssh for phone, and then setup username and password so that engineer can login to phone via CLI

upvoted 11 times

G0y0 Most Recent 4 months, 1 week ago

Selected Answer: CD

correct answers are C. and D. you set user and pass in ssh information and enable ssh acces. When you use putty, ingress the username and password you configured in ssh information in cucm, and the phone will be asking you again, a secound user and password. for 7800 phones, these credentialas are "debug" and "debug" for username and password, and you wil have gain access.

upvoted 1 times

DaKenjee 7 months, 1 week ago

Selected Answer: CD

had done this in past on 89xx/99xx only two settings necessary.

On phone configuration page:

(Answer C) Product Specific Configuration Layout > SSH Access= Enabled

(Answer D) Secure Shell Information > Secure Shell User + Secure Shell Password

upvoted 3 times

usernamesarehard 10 months ago

Selected Answer: CD

Should be C & D

upvoted 1 times

laumail 12 months ago

Selected Answer: CD

SSH must be enabled and credentials has to be set

upvoted 3 times

Question #80 *Topic 1*

A remote office has a less-than-optimal WAN connection and experiences packet loss, delay, and jitter. Which VoIP codec should be used in this situation?

A. G.711alaw

B. iLBC

C. G.722.1

D. G.729A

Correct Answer: *B*

Community vote distribution

B (71%) D (29%)

ring_phone Highly Voted 4 years, 6 months ago

B. same document
iLBC: iLBC provides audio quality between that of G.711 and G.729 at bit rates of 15.2 kbps (38-bytes or 20msec) and 13.3 kbps (50 bytes or 30 msec). iLBC handles lossy networks in better way than G729 because it treats each packet independently. G729 depends on the previous packet to handle packet loss, jitter and delay which doesn't tolerate well in lossy networks.

upvoted 12 times

[Removed] 2 years, 7 months ago

I Agree. The question says packet loss, delay, and jitter, there is no anything said about bandwith. G729 is for low bandwidth, and iLBC is for bad QoS wheter the bandwidth is low or not.

upvoted 1 times

BarryR 4 years, 5 months ago

Cisco recommends using G.729a as the low-bandwidth codec because it is supported on all Cisco Unified IP Phone models as well as most other Cisco Unified Communications devices, therefore it can eliminate the need for transcoding.

upvoted 1 times

Simpax_100 4 years, 5 months ago

But G.729A is a medium-complexity variant of G.729 with slightly lower voice quality and is more susceptible to network irregularities such as delay, variation, and "tandeming."
I think B is the answer

upvoted 1 times

Daved90 Most Recent 9 months, 1 week ago

Selected Answer: B

iLBC handles jitter and packet loss better

upvoted 2 times

Daved90 9 months, 1 week ago

and no mention about bandwith

upvoted 1 times

marjana_mirza 1 year, 8 months ago

Selected Answer: D

D - correct
Bandwidth estimation becomes an issue when voice is included in the calculation. Because WAN links are usually the lowest-speed circuits in an IP Telephony network, particular attention must be given to reducing packet loss, delay, and jitter where voice traffic is sent across these links. G.729 is the preferred codec for use over the WAN because the G.729 method forsampling audio introducesthe least latency (only 30 milliseconds) in addition to any other delays caused by the network.
ref: https://www.cisco.com/c/en/us/td/docs/voice_ip_comm/cust_contact/contact_center/crs/express_11_5/design/guide/uccx_b_soldg-for-unified-ccx/uccx_b_soldg-for-unified-ccx_chapter_0111.pdf

upvoted 2 times

Panda_man 1 year, 11 months ago

Selected Answer: B

iLBC: iLBC provides audio quality between that of G.711 and G.729 at bit rates of 15.2 kbps (38-bytes or 20msec) and 13.3 kbps (50 bytes or 30 msec). iLBC handles lossy networks in better way than G729 because it treats each packet independently. G729 depends on the previous packet to handle packet loss, jitter and delay which doesn't tolerate well in lossy networks.

upvoted 2 times

Piji 2 years, 4 months ago

Selected Answer: B

The correct answer is B. iLBC

no bandwidth issue, the quality issue packet loss, delay, and jitter. G.729 is good for bandwidth and iLBC for bad Qos.

There is similar question and only mention about bandwidth and the answer for that question is G.729.

upvoted 2 times

Tomilee 2 years, 10 months ago

Handles packet loss better than g729 & g711

upvoted 1 times

Tomilee 2 years, 10 months ago

iLBC is it

upvoted 1 times

Komy 3 years, 6 months ago

I will go with ILBC guys (information is captured from Cisco SRND)

https://www.cisco.com/c/en/us/td/docs/voice_ip_comm/cucm/srnd/collab10/collab10/netstruc.html

"When deploying voice in a WAN environment, Cisco recommends that you use the lower-bandwidth G.729 codec for any voice calls that will traverse WAN links because this practice will provide bandwidth savings on these lower-speed links.

Where calls are made over best-effort networks with no QoS guarantees for voice, consider using Internet Low Bit Rate Codec (iLBC), which enables graceful speech quality degradation and good error resilience characteristics in networks where frames can get lost."

upvoted 4 times

MKZ 3 years, 6 months ago

G729A & AB are medium Complexity

iLBC - High Complexity

I go with G729A

upvoted 1 times

CiscoCUCMKing 3 years, 10 months ago

Tricky question, as CollabGuy says. I don't think there is any arguement that iLBC is a marginally better codec for this kind of scenario. So it comes down to choosing between g729 and iLBC. Minimising transcoder usage should definitely be a factor. Since newer Cisco phones (8865 and 7975) support iLBC, in the exam I would answer B (iLBC) but how Cisco mark this question is hard to predict. As DEFAULTNERD says, Cisco may prefer g729.

upvoted 2 times

PunKike 3 years, 10 months ago

lol DEFAULTNERD

upvoted 1 times

DEFAULTNERD 4 years ago

Cisco does not like ILBC it did not create it and does not like it.

upvoted 2 times

ccosta7 4 years ago

As says at this referenced link I believes iLBC is the best option:

iLBC handles lossy networks in better way than G729 because it treats each packet independently. G729 depends on the previous packet to handle packet loss, jitter and delay which doesn't tolerate well in lossy networks.

upvoted 2 times

CollabGuy 4 years, 1 month ago

IMHO it has to be iLBC.

I think this is a very tricky question and almost evil to ask in an exam.

G729 and iLBC are both low bandwidth codecs (everybody knows that).

However, as they say that the network has jitter, delay and packet loss we have to use iLBC as "iLBC handles lossy networks in better way than G729 because it treats each packet independently. G729 depends on the previous packet to handle packet loss, jitter and delay which doesn't tolerate well in lossy networks."

upvoted 3 times

BarryR 4 years, 5 months ago

Cisco wants g.729 because no trans-coding is needed

upvoted 2 times

Testy1 4 years, 2 months ago

b is what I am going with. The question specifically asks about a network with loss, delay and jitter. Nothing about limited bandwidth. Sounds like iLBC.

upvoted 1 times

Mtdaw 4 years, 2 months ago

But states less-than-optimal WAN connection, that could mean limited bandwidth

upvoted 1 times

CollabGuy 4 years, 1 month ago

Debatable. Less than optimal WAN because it has jitter, packet loss, delay imho.

upvoted 1 times

Question #81 Topic 1

```
INVITE sip:2002@10.10.10.10:5060 SIP/2.0
[..truncated..]
v=0
o=UAC 6107 7816 IN IP4 10.10.10.11
s=SIP Call
c=IN IP4 10.10.10.11
t=0 0
m=audio 8190 RTP/AVP 18 110
c=IN IP4 10.10.10.11
a=rtpmap:18 G729/8000
a=fmtp:18 annexb=no
a=rtpmap:110 telephone-event/8000
a=fmtp:110 0-16
a=ptime:20

SIP/2.0 200 OK
[..truncated..]
v=0
o=UAS 4692 9609 IN IP4 10.10.10.10
s=SIP Call
c=IN IP4 10.10.10.10
t=0 0
m=audio 8056 RTP/AVP 18
c=IN IP4 10.10.10.10
a=rtpmap:18 G729/8000
a=fmtp:18 annexb=no
a=ptime:20
```

Refer to the exhibit. The SDP offer/answer has been completed successfully but there is no DTMF when users press keys. What is the cause of the issue?

A. DTMF was negotiated properly in these messages.

B. G.729 rather than G.711ulaw was negotiated.

C. Payload type 110 was negotiated rather than type 101.

D. DTMF was not negotiated on the call.

Correct Answer: *D*

BarryR Highly Voted 4 years, 12 months ago

D is correct answer DTMF was not negotiated

upvoted 14 times

J3ster 4 years, 9 months ago

As you can see... in the bottom it says:

a=fmtp:101 18

as you can see...

a=fmtp:101 0-15

DTMF tones (Events 0 through 15)

upvoted 1 times

rishik Highly Voted 4 years, 11 months ago

Correct answer is D. No DTMF was negotiated in 200 OK message although in the INVITE it sends payload type 110

upvoted 6 times

b3532e4 Most Recent 9 months ago

I don't know C & D

upvoted 1 times

BhaiKyare 3 years, 10 months ago

Out-of-Band using RTP, (RFC2833). This is usually the preferred method. Remember to set the payload type if different (e.g.: 96, 101 or 110 in some cases) so payload can be 110 also.

upvoted 1 times

somedudebob 4 years, 2 months ago

We can see in the INVITE that a=fmtp:18 and a fmtp:110 0-16
and in the OK we can only see a=fmtp:18
So ONLY G728 was negeotiated on this call.

A. NOT CORRECT - DTMF was not negeotiated
B. G729 was negeotiated however, we are not concerned with this codec because we are looking for DTMF
C. Payload 110 Telephone event was offered in the INVITE, but was not returned in the Ok, So this was not negeotiated.
D. The correct answer, NO DTMF exists
upvoted 3 times

skneff 1 year, 5 months ago
Correct. Even though using payload 101 is more common for negotiating NTE/DTMF, it wouldn't matter if it was instead used in this case, because the SIP response doesn't support payload 101 anyway.

There is no reason you can't use DTMF with G.729 in-band NTE per RFC2833. It's only an issue when using DTMF with G.729 in-band within the same RTP payload as the voice traffic, due to compression, which is not the case here, so B is incorrect.
upvoted 1 times

somedudebob 4 years, 2 months ago
We can see in the INVITE that a=fmtp:18 and a fmtp:110 0-16
and in the OK we can only see a=fmtp:18
So ONLY G728 was negeotiated on this call.

A. NOT CORRECT - DTMF was not negeotiated
B. G729 was negeotiated however, we are not concerned with this codec because we are looking for DTMF
C. Payload 110 Telephone event was offered in the INVITE, but was not returned in the Ok, So this was not negeotiated.
D. The correct answer, NO DTMF exists
upvoted 1 times

CollabGuy 4 years, 7 months ago
IMHO, D is the correct answer.
It is offered 18 (g729) and 110 (type-event), however the answer only contains 18. Therefore, DTMF was not negotiated.
upvoted 5 times

khader09 4 years, 9 months ago
Correct answer is C - C. Payload type 110 was negotiated rather than type 101.
upvoted 1 times

Testy1 4 years, 8 months ago
Correct answer is D. Look at the m=audio line in the invite, it sends 18 & 110. The same m=audio line in the 200 OK only has 18. No DTMF was successfully negotiated.
upvoted 8 times

egipty13 4 years, 11 months ago
Hi Guys, Somebody already applied this exam and know if this guide is very similar please? thankyou
upvoted 2 times

ring_phone 5 years ago
B.Dual-tone multi-frequency signaling (DTMF), fax transmissions, and high-quality audio cannot be transported reliably with this codec.
https://en.wikipedia.org/wiki/G.729
upvoted 1 times

valsrock 4 years, 10 months ago
It's wrong. DTMF requires the use of the named telephony events in the RTP payload for DTMF digits, telephony tones, and telephony signals as specified in RFC 4733.
upvoted 2 times

Janu82 5 years ago
DTMF will negotiate with payload type 101.
https://www.ciscolive.com/c/dam/r/ciscolive/us/docs/2018/pdf/BRKUCC-2006.pdf [page 47]
upvoted 1 times

J3ster 4 years, 9 months ago

As you can see... in the bottom it says:
a=fmtp:101 18

DTMF negotiation is...
a=fmtp:101 0-15
DTMF tones (Events 0 through 15)
upvoted 1 times

Griswald 5 years ago
kpml or RTP-NTE is DTMF so no Event: in SIP message = kpml or RTP-NTE
upvoted 1 times

Question #82 *Topic 1*

Which two types of device are supported by the Bulk Administration Tool? (Choose two.)

A. H.322

B. Cisco Unified IP phones (all models)

C. SIP trunks

D. H.225 trunks

E. music on hold servers

Correct Answer: *AB*

glong 8 months, 3 weeks ago

A. H.323 clients

upvoted 1 times

timmyz 2 years, 4 months ago

Typo, basically Phones and H323 Clients

upvoted 3 times

jonycakes 2 years, 6 months ago

Is A-H322 a typo?

Source: https://www.cisco.com/c/en/us/td/docs/voice_ip_comm/cucm/bat/12_5_1/cucm_b_bulk-administration-guide-1251/cucm_b_bulk-administration-guide-1251_chapter_01.html

You can use BAT to work with the following types of devices and records:

Add, update, and delete Cisco Unified IP Phones including voice gateway (VG) phones, computer telephony interface (CTI) ports, and H.323 clients, and migrate phones from Skinny Client Control Protocol (SCCP) to Session Initiation Protocol (SIP)

upvoted 2 times

Question #83 *Topic 1*

You are adding regions in Cisco Unified Communications Manager. Which codec(s) are selected when a call is placed if you set up the max audio bit rate to use 8 kbps?

A. G.729

B. G.729 and G.711ulaw

C. G.711ulaw and G.711alaw

D. G.722

Correct Answer: *A*

Griswald Highly Voted 11 months, 1 week ago

Becauseof bandwidthconstraintsat mostremote-sitedeployments,use 8 kb/s (G.729)asthe recommendedsettingbetweena new regionand existingregions.

upvoted 10 times

Stevon Most Recent 4 weeks, 1 day ago

Selected Answer: A

8 kbit/s

729 is mostly used in voice over Internet Protocol (VoIP) applications when bandwidth must be conserved. Standard G. 729 operates at a bit rate of 8 kbit/s, but extensions provide rates of 6.4 kbit/s (Annex D, F, H, I, C+) and 11.8 kbit/s (Annex E, G, H, I, C+) for worse and better speech quality, respectively. G.

upvoted 1 times

Question #84 *Topic 1*

How can an administrator stop Cisco Unified Communications Manager from advertising the OPUS codec for recording enabled devices?

A. Route recorded calls through Cisco Unified Border Element because it does not support OPUS.

B. Go to the phone's configuration page and set ג€Advertise OPUS Codecג€ to be ג€falseג€.

C. Integrate the Cisco Unified CM with 3 recording solution that does not support OPUS.

D. In CUCM Service Parameters set "Opus Codec Enabledג€ to ג€Enabled for all Devices Except Recording-Enabled Devices.ג€

Correct Answer: *D*
Reference:
https://www.cisco.com/c/en/us/support/docs/unified-communications/unified-communications-manager-callmanager/211297-Configure-Opus-Support- on-Cisco-Unified.pdf

Community vote distribution
D (100%)

Stevon 4 weeks, 1 day ago
Selected Answer: D
AI Overview
Learn more
To prevent Cisco Unified Communications Manager from advertising the OPUS codec for recording-enabled devices, you can configure the "Opus Codec Enabled" service parameter to "Enabled for all devices except recording-enabled devices". This setting restricts the use of OPUS to devices that are not configured for recording.
Here's a more detailed explanation:
1. Access Service Parameters:
Log into the Cisco Unified Communications Manager (CUCM) administration page and navigate to "System" > "Service Parameters".
2. Locate the Parameter:
Select your Call Manager server and the "Call Manager Service" from the dropdown menus.
3. Configure Opus Codec Enabled:
Search for the "Opus Codec Enabled" service parameter and set its value to "Enabled for all devices except recording-enabled devices".
4. Apply Configuration:
After making the change, click "Apply Config" and "Reset" to apply the new settings.
upvoted 1 times

Slushed 8 months ago
Selected Answer: D
D is correct. The setting is under the Cisco CallManager service specifically within the Service Parameters and you can choose to completely disable it as well.
upvoted 1 times

timmyz 1 year, 4 months ago
Answer is D
upvoted 2 times

Question #85 *Topic 1*

An engineer must manually provision a Cisco IP Phone 8845 using SIP. Which two fields must be configured for a successful provision? (Choose two.)

A. location

B. media resources group list

C. SIP profile

D. CSS

E. device security profile

Correct Answer: *CE*

Community vote distribution

CE (100%)

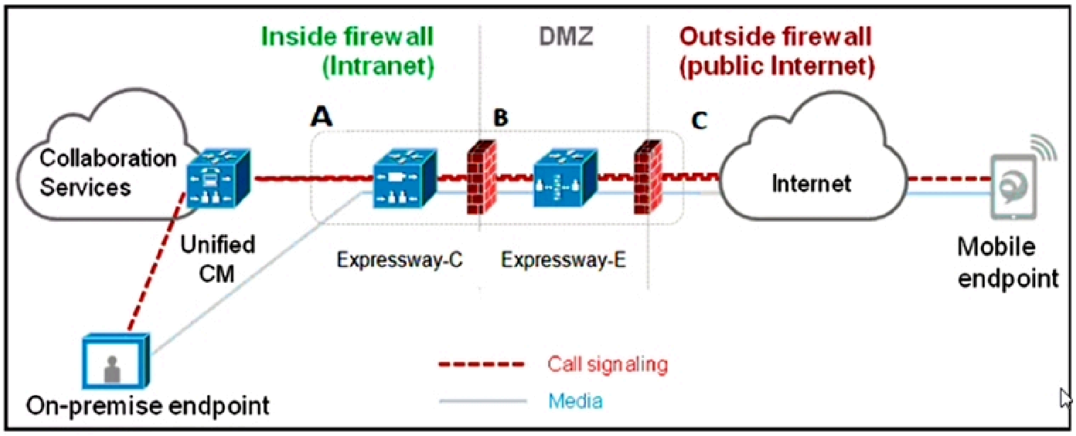
Omitted 11 months ago

Selected Answer: CE

These are mandatory fields on the device page in CUCM

upvoted 3 times

Question #86 Topic 1



Refer to the exhibit. When making a call to a MRA client, what are the combinations of protocol on each of the different sections A-B-C?

A. SIP TCP/TLS (A) + SIP TLS (B) + SIP TLS (C)

B. SIP TCP/TLS (A) + SIP TCP/TLS (B) + SIP TCP/TLS (C)

C. IP TCP/TLS (A) + SIP TCP/TLS (B) + SIP TLS (C)

D. SIP TLS (A) + SIP TLS (B) + SIP TLS (C)

Correct Answer: *A*

Reference:

https://www.cisco.com/c/dam/en/us/td/docs/voice_ip_comm/expressway/config_guide/X8-9/Mobile-Remote-Access-via-Expressway-Deployment-
Guide-X8-9-1.pdf

Community vote distribution

A (100%)

MKZ Highly Voted 1 year, 6 months ago

https://www.cisco.com/c/dam/en/us/td/docs/voice_ip_comm/expressway/config_guide/X12-5/Cisco-Expressway-IP-Port-Usage-for-Firewall-Traversal-Deployment-Guide-X12-5.pdf

Page 29

upvoted 6 times

Omitted Most Recent 7 months, 2 weeks ago

Selected Answer: A

The only thing that complicates this for me is that XMPP communication would be TCP at B. However, they mention only call signaling and media which means A should be correct.

upvoted 1 times

Question #87 *Topic 1*

An engineer is configuring a Cisco Unified Border Element to allow the video endpoints to negotiate without the Cisco Unified Border Element interfering in the process. What should the engineer configure on the Cisco Unified Border Element to support this process?

A. Configure codec transparent on the dial peers.

B. Configure a transcoder for video protocols.

C. Configure a hardcoded codec on the dial peers.

D. Configure pass-thru content sdp on the voice service.

Correct Answer: *D*

Community vote distribution

D (100%)

Kabimas66 9 months, 2 weeks ago

on the link provided by the exam: https://www.cisco.com/c/en/us/td/docs/ios-xml/ios/voice/cube/configuration/cube-book/cube-codec-basic.html
Confirms ANSWER D

upvoted 1 times

Myare 1 year, 11 months ago

Note
You can use 'pass-thru content sdp', if you do not want to involve CUBE in the codec negotiation.

upvoted 1 times

KZG 2 years, 4 months ago

Tricky bc it says Voice service instead of voice codec

upvoted 1 times

Piji 2 years, 4 months ago

Selected Answer: D

Correct answer D.

You can use 'pass-thru content sdp', if you do not want to involve CUBE in the codec negotiation.

https://www.cisco.com/c/en/us/td/docs/ios-xml/ios/voice/cube/configuration/cube-book/cube-codec-basic.html#id_128259

upvoted 3 times

G0y0 2 years, 6 months ago

About codec transparent, take in mind this: Only codecs that are supported by the IPIPGW or CUBE are transparently passed. If an unsupported codec is requested, call setup will fail.

upvoted 1 times

G0y0 2 years, 6 months ago

So, if you are supporting only cisco video endpoint, probably you only need codec transparent. But if you are supporting cisco video endpoint with third-party enpoints which might support different attributes or capabilities unsupported by CUBE, probably you will need pass-thru content sdp that will pass SDP untouched between different third-party endpoints. The problem is that the question do not say if the endpoints are just cisco, or different third-party.

upvoted 1 times

Sharky1066 2 years, 10 months ago

I think A could be the correct answer. The question is related to a specific requirement, ie video, not all voice/video services that are running through the CUBE. The voice-class codec transparent would be applied to a dial-peer for say a specific video requirement. The 'pass-thru content sdp' would be applied to the voice service effecting ALL traffic (voice and video)

upvoted 2 times

asdlfhqwoiefnwe 2 years, 11 months ago

Selected Answer: D

D

A is likely referring to voice-class codec transparent being applied to the dial-peers, but this is only for the initial offer and the response can still be altered by the CUBE with that method. https://www.cisco.com/c/en/us/td/docs/ios-xml/ios/voice/cube/configuration/cube-book/cube-codec-basic.html#id_128259

upvoted 4 times

davidanibalmarcelino 3 years, 6 months ago

D

https://www.cisco.com/c/en/us/td/docs/ios-xml/ios/voice/cube/configuration/cube-book/cube-codec-basic.html

"You can use 'pass-thru content sdp', if you do not want to involve CUBE in the codec negotiation."

upvoted 4 times

MKZ 3 years, 6 months ago

https://www.cisco.com/c/en/us/td/docs/ios-xml/ios/voice/cube/configuration/cube-book/cube-codec-basic.html

upvoted 1 times

jonycakes 3 years, 6 months ago

D: Configure pass-thru content sdp on the voice service.

Source: https://www.cisco.com/c/en/us/td/docs/ios-xml/ios/voice/cube/configuration/cube-book/cube-codec-basic.html

You can use 'pass-thru content sdp', if you do not want to involve CUBE in the codec negotiation.

You can use 'pass-thru content sdp', if you do not want to involve CUBE in the codec negotiation.

upvoted 3 times

Question #88 *Topic 1*

Due to service provider restriction, Cisco UCM cannot send video in the SDP. Which two options on Cisco UCM are configured to suppress video in the SDP in outgoing invites? (Choose two.)

A. Set Video Bandwidth in the Region settings to 0.

B. Check the Send send-receive SDP in mid-call INVITE check box on the SIP trunk SIP profile.

C. Change the Video Capabilities dropdown on the endpoint to Disabled.

D. Add the audio forced command to voice service VoIP on the Cisco Unified Border Element.

E. Check the Retry Video Call as Audio on the SIP trunk.

Correct Answer: *AE*

Community vote distribution

AE (82%) AC (18%)

Collabhunter Highly Voted 3 years, 6 months ago

Correct A and E, not only need to 0 the region but also check Retry Video Call as Audio on trunk

upvoted 9 times

Kabimas66 Most Recent 10 months, 2 weeks ago

from: https://www.cisco.com/c/en/us/td/docs/ios-xml/ios/voice/cube/configuration/cube-book/voi-audio-forced.html

It seems the answers are A and D.

upvoted 1 times

Kabimas66 9 months, 2 weeks ago

I read again, the question says ONLY on CUCM, so CUBE option is not valid.

I changed my ASNWER: A & E

upvoted 1 times

DaKenjee 2 years, 1 month ago

Selected Answer: AE

A+E

B is not suiting, has a different purpose

C would suit, but internal videos are gone too

upvoted 3 times

[Removed] 2 years, 2 months ago

Selected Answer: AE

Why would E be correct? It isnt configure TO SUPPRESS VIDEO. Its configured to offer an audio-only fallback when Video fails. Theoretically you can just leave it off and it wont matter for the goal of suppressing video.

On the other hand i dont know of such a dropdown as "Video Capabilities" on Video Endpoints on CUCM. So in Cisco Fashion it probably is A & E.

upvoted 2 times

DaKenjee 2 years, 1 month ago

i think they mean to deactivate camara, but that won't be a solution in general

upvoted 1 times

Piji 2 years, 4 months ago

Selected Answer: AE

A & E are correct answers.

C is not correct answer as you only want disable video to service provider and not on internal calls.

upvoted 1 times

Slushed 2 years, 8 months ago

Selected Answer: AE

A & E. The reason is it has to be only in CUCM per the question. You would create a specific Region relationship for the Device Pool applied to the trunk that cannot handle video and thus change the Maximum Video Bandwidth to 0 while setting retry a video call as audio on the Phone

Configuration (default setting is this is already checked).
upvoted 3 times

samconnects 2 years, 9 months ago
A and E should be the right answers. C would disable video for all calls. Not just the ITSP one.
E would disable video out to the trunk.
upvoted 1 times

asdlfhqwoiefnwe 2 years, 11 months ago
Selected Answer: AC
A and C
D is incorrect due to being a change on the CUBE rather than in CUCM
upvoted 2 times

Piji 2 years, 4 months ago
C is not correct, as they may want to have video for internal calls.
upvoted 1 times

Vijay_ABI 3 years, 1 month ago
The question clear says - what is done on the CUCM- So, Try Video as Audio call- is one option.. setting bandwidth to 0 on region is another option.
Drop down option on the Phone - is on the phone- so ignore.. anything we do on the router should be ignored.
upvoted 1 times

F3rnando 3 years, 4 months ago
A and C
B - is not related to video
C - I checked 9971 phone and there is definetly a dropdown about video capabilities .
D - This change is done in CUBE, not the CUCM
E will not disable video but instead will but mitigate consequence if its not working
upvoted 4 times

Sharky1066 2 years, 10 months ago
Your right you can modify the video behaviour on some endpoints but that would completely disable video for those device, eg also disable video for internal calls, etc. If you leave video enabled on the endpint and then set the trunk to Retry Video Call as Audio the call would be established as audio call with no video codec negotiation.
upvoted 2 times

Shazam021 3 years, 5 months ago
Answer is A and C.. There is definitely an option to disable Video Capabilities on the device (9971/8865 video phones).
upvoted 2 times

remmie_78 3 years, 5 months ago
I think it should be A and C. Answer A is clear. but C; Change the Video Capabilities dropdown on the endpoint to Disabled, will also bring no video.
The answer E (Retry the call) is only when Vido will not work....
upvoted 1 times

remmie_78 3 years, 5 months ago
second thought, there is no option C on the device.... so Yes: A and E are correct....
upvoted 2 times

gahhhhaccount 3 years, 6 months ago
I would go with A and D.

Support for Video Suppression
Cisco IOS 15.6(2)T
Cisco IOS XE Denali 16.3.1

This feature allows pass-through of only audio and application (for T.38 Fax) media types and drops all other media types in SDP.

The following commands are introduced: audio forced , voice-class sip audio forced

https://www.cisco.com/c/en/us/td/docs/ios-xml/ios/voice/cube/configuration/cube-book/voi-audio-forced.html
upvoted 2 times

gahhhhaccount 3 years, 6 months ago

On a second thought, the question ask what changes are done in CUCM, not the CUBE. So yeah, it may be A and E then.
upvoted 6 times

Question #89 *Topic 1*

A Cisco IP Phone 7841 that is registered to a Cisco UCM with default configuration receives a call setup message. Which codec is negotiated when the SDP offer includes this line of text? m=audio 49181 RTP/AVP 0 8 97

A. G.711alaw

B. iLBC

C. G.722

D. G.711ulaw

Correct Answer: *D*

Community vote distribution

D (100%)

skneff Highly Voted 11 months, 3 weeks ago

Selected Answer: D

D is correct. iLBC uses a dynamic payload and cannot be verified with the limited information provided, G.722 is payload type 9 (not listed). G.711a-law and G.711-ulaw are both acceptable, but G.711a-law is payload type 0, which is listed first, therefore is preferred

upvoted 5 times

kljw5 5 months, 1 week ago

Correcting your type. G.711u-law is payload type 0 so D is correct

upvoted 1 times

Question #90 *Topic 1*

Endpoint A is attempting to call endpoint B. Endpoint A only supports G.711ulaw with a packetization rate of 20 ms, and endpoint B supports packetization rate of
30 ms for G.711ulaw. Which two media resources are allocated to normalize packetization rates through transrating? (Choose two.)

A. software transcoder on Cisco UCM

B. hardware transcoder on Cisco IOS Software

C. software MTP on Cisco IOS Software

D. software MTP on Cisco UCM

E. hardware MTP on Cisco IOS Software

Correct Answer: *DE*

Community vote distribution

DE (52%) BE (29%) BD (19%)

gahhhaccount Highly Voted 4 years ago

It's B and E. This can only be done by using DSPs. Hardare MTP and Transcoder through a router. CUCM doesn't have a software transcoders so for sure A cannot be the answer.
https://www.cisco.com/c/en/us/td/docs/voice_ip_comm/cucm/srnd/collab10/collab10/media.html#77886

upvoted 14 times

SDLOA14 4 years ago

B and E

Agree

From Cisco SRND 11 "When Cisco IOS MTP resources are invoked by Unified CM for a call flow, a software session rather than a hardware DSP session is consumed unless the media legs of the call flow require transrating. Thus, for flows invoking an MTP, a DSP session [hardware MTP] is used only when transrating (conversion between media legs with the same codec but different packetization times) is required."

upvoted 6 times

skneff 1 year, 5 months ago

According to the Cisco e-Reader CLCOR v1.0.23 guide, under the "MTP and Transcoder Devices" section, where it talks about the Cisco IP Voice Media Streaming Application service in CUCM and it as a software MTP media resource, it says "This MTP type can perform transrating of a given codec; for example, when one call legs uses a sample size of 20 ms and the other call leg uses a sample size of 30 ms"

Thus, software MTP on Cisco UCM is ALSO a correct answer. To me, B, D, and E are all correct options. I don't see it mention that line in the SRND though, so I'm hesitant to believe that is one of the answers they are looking for. According to solely the SRND, B & E seem like the answers they want, but if we're talking older IOS firmware, D & E are correct as transcoding didn't support transrating until IOS release 15.0.1M.

upvoted 1 times

asdlfhqwoiefnwe Highly Voted 3 years, 5 months ago

Selected Answer: DE

D and E

While a transcoder could work, MTP is sufficient and the better answer. CUCM and IOS have MTP resources, but IOS can only handle transrating when using DSPs (which would be configured as a hardware MTP).

upvoted 9 times

cliftjeff 3 years, 4 months ago

I agree, CUCM has software based transcoder capabilities via the Cisco IP Voice Media Streaming Applications service. IOS Routers can be hardware or software based MTPs. Software based MTP IOS routers are limited by the CPU cycles available. Hardware based IOS MTP routers are for large scale and is limited by the number of DSP chips installed.

upvoted 1 times

cliftjeff 3 years, 4 months ago

edit: meant to say CUCM has software MTP capabilities via the Cisco IP Voice Media Streaming Applications service.
upvoted 2 times

Sharky1066 3 years, 4 months ago
spot-on, as per the 350-801 course notes - Ans D. Software MTP in CUCM can be used to transrate codecs of the same type using different sample sizes. Ans E. Hardware MTP on Cisco IOS router using DSP resource can be used when two call legs use the same audio codec but diffrent packetisation sizes per call leg. Starting with IOS version 15.0.1M a hardware trancoding (and there is only hardware transcoding - software transcoding doesnt exist!) a transcoder also supports transrating wherby it connects two streams that use the same codec but with different sampling sizes
upvoted 3 times

b3532e4 Most Recent 9 months ago
I Agree A & D
upvoted 1 times

Totor27 2 years, 4 months ago
Selected Answer: BE
As indicated in SRND collaboration 12.X, it's B and E. This can only be done by using DSPs. Hardare MTP and Transcoder through a router.

"When Cisco IOS MTP resources are invoked by Unified CM for a call flow, a software session rather than a hardware DSP session is consumed unless the media legs of the call flow require transrating. Thus, for flows invoking an MTP, a DSP session is used only when transrating (conversion between media legs with the same codec but different packetization times) is required."

https://www.cisco.com/c/en/us/td/docs/voice_ip_comm/cucm/srnd/collab12/collab12.pdf
upvoted 4 times

Piji 2 years, 10 months ago
Selected Answer: BD
The correct answer is BD.
Software MTP on CUCM, and Hardware transcoder on Cisco IOS Software.
upvoted 4 times

kljw5 5 months, 1 week ago
I'm coming around to this being the correct answer. As transrating is requied a software MTP does not handle this and a hardware or DSP is required.
upvoted 1 times

[Removed] 3 years ago
Selected Answer: DE
DE. Repacketization needs a DSP and cannot be done by software only. this means its obviously E on iOS, but also D, because on CUCM a hardware MTP is also refered to as a software MTP
(https://www.cisco.com/c/en/us/td/docs/ios/voice/cminterop/configuration/guide/15_1/vc_15_1_book/vc_enh_confr_vgr.html)

The following MTP resources are supported for Cisco Unified Communications Manager 4.0 (formerly known as Cisco CallManager 4.0) and later releases:

•Software MTP—Software-only implementation that does not use a DSP resource for endpoints using the same codec and the same packetization time.

•Hardware MTP—Hardware-only implementation that uses a DSP resource for endpoints using the same G.711 codec but a different packetization time. >>>> >>The repacketization requires a DSP resource so it cannot be done by software only. Cisco Unified Communications Manager also uses the term software MTP when referring to a hardware MTP. <<<<<<
upvoted 2 times

iamnoone 3 years, 2 months ago
Selected Answer: BE
B and E. Eventhough both endpoints use G711ulaw/alaw the sampling is different (20 and 30 ms). So in order for this to work a hardware IOS MTP will be required. An IOS transcoder can also do that.
upvoted 2 times

Bazant 3 years, 6 months ago
Based omen the below I'd say B D

https://www.cisco.com/c/en/us/td/docs/voice_ip_comm/cucm/srnd/collab10/collab10/media.html

A transcoder is a device that converts an input stream from one codec into an output stream that uses a different codec. Starting with Cisco IOS Release 15.0.1M, a transcoder also supports transrating, whereby it connects two streams that utilize the same codec but with a different packet size.

An MTP can be used to transcode G.711 a-law audio packets to G.711 mu-law packets and vice versa, or it can be used to bridge two connections that utilize different packetization periods (different sample sizes).
upvoted 2 times

jarcoman99 3 years, 8 months ago
I would say that the answer is B and E
•Software MTP—Software-only implementation that does not use a DSP resource for endpoints using the same codec and the same packetization time.

•Hardware MTP—Hardware-only implementation that uses a DSP resource for endpoints using the same G.711 codec but a different packetization time. The repacketization requires a DSP resource so it cannot be done by software only. Cisco Unified Communications Manager also uses the term software MTP when referring to a hardware MTP.
https://www.cisco.com/c/en/us/td/docs/routers/access/vgd1t3/rel1_0/software/configuration/guide/VGD_transcoding.html
upvoted 2 times

IMMohit 3 years, 8 months ago
How was the exam 350-801? Did you got questions from these questions. I have read comments, that very less questions are coming from here?
upvoted 1 times

jarcoman99 3 years, 5 months ago
The exam has a pool of about 250 questions now. And this dump covers roughly 70% of the exam.
upvoted 2 times

Brant 3 years, 9 months ago
For packetization you need MTP. Hence C & D. There is no hardware MTP on CISCO IOS, So E is incorrect.

https://www.cisco.com/c/en/us/td/docs/voice_ip_comm/cucm/srnd/collab10/collab10/media.html#67911
upvoted 1 times

BhaiKyare 3 years, 10 months ago
https://www.cisco.com/c/en/us/td/docs/voice_ip_comm/cucm/admin/11_5_1/sysConfig/CUCM_BK_SE5DAF88_00_cucm-system-configuration-guide-1151/CUCM_BK_SE5DAF88_00_cucm-system-configuration-guide-1151_chapter_01000010.html A and D is correct
upvoted 1 times

timmyz 3 years, 10 months ago
you can not have a software transcoder. Transcoders are only hardware based and register to CUCM. answer is B and E
upvoted 1 times

Question #91 *Topic 1*

An engineer with ID0123456789 is designing a new dial plan for a customer that has offices in several countries on four continents around the world. This client also wants to integrate with a Microsoft Lync backend. Which dial plan type does the engineer recommend?

A. H.323

B. SIP URI

C. E.164

D. TEHO

Correct Answer: *C*

Community vote distribution

C (100%)

G0y0 3 months, 3 weeks ago

Selected Answer: C

The only dial plan that is used for international communications is E.164

upvoted 1 times

Omitted 10 months, 3 weeks ago

Selected Answer: C

E.164 is the international numbering plan meant to ensure each number is globally unique.

upvoted 3 times

oyurchenko 1 year, 2 months ago

not clear question

B. SIP URI

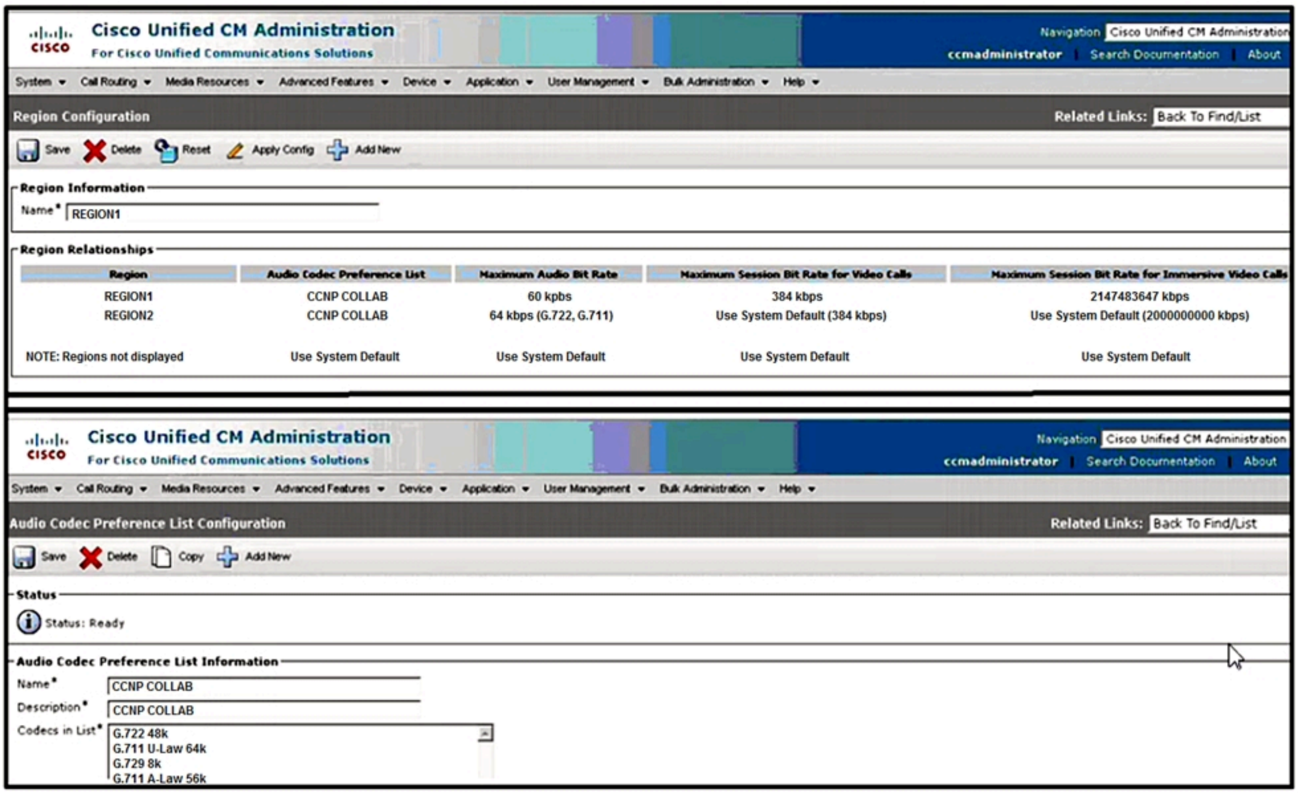
upvoted 3 times

G0y0 3 months, 3 weeks ago

SIP URI is a format to SIP transactions, it is a protocol, it is not a dial plan.

upvoted 1 times

Question #92 *Topic 1*



Refer to the exhibit. An engineer is troubleshooting a codec negotiation issue where both endpoints that are involved in the call support the codecs listed in the exhibit. Which audio codec is selected if a call between two endpoints in Region1 is placed?

A. G.722

B. G.729

C. G.711a

D. G.711u

Correct Answer: *A*

Community vote distribution

A (100%)

dbjim3 Highly Voted 3 years, 6 months ago

The correct answer is A. The codec used cannot exceed 60k. G.722 has the ability to use 48, 56, and 64k.

upvoted 6 times

kljw5 Most Recent 4 months ago

Selected Answer: A

Codes listed in priority G.722 is also labled as 48k meeting both bandwidth requirement and min audio bitrate constraint.

upvoted 1 times

G0y0 4 months, 1 week ago

Selected Answer: A

The Maximum Audio Bit Rate is still applied for calls within a region and between regions; but rather that using the highest audio quality codec (as in earlier Unified CM releases) based on the maximum bit rate setting, the codec selection is made based on the codec order in the audio codec preference list and the codecs that the endpoints support. In these condictions, the correct answer is A.

Please refer to: "Cisco Collaboration System 12.x Solution Reference Network Designs SRND", in section "Codec Selection over SIP Trunks"

upvoted 1 times

b3532e4 9 months ago

G.722 با سرعت 48 kbps

G.711 u-law با سرعت 64 kbps

G.729 با سرعت 8 kbps
G.711 A-law با سرعت 56 kbps
upvoted 2 times

b3532e4 9 months, 1 week ago
G.729 call uses 24 kb/s.
upvoted 1 times

[Removed] 1 year, 11 months ago
Based on the information provided in the image, if a call is placed between two endpoints in Region1, the audio codec that will be selected is G.711u. This means that the answer is D.

In the image, we can see that the Region1 to Region1 relationship has a maximum audio bit rate of 64 kbps. This means that the codec selected for a call between two endpoints in Region1 must have a bit rate of 64 kbps or lower. G.711u and G.711a both have bit rates of 64 kbps, while G.722 and G.729 have higher bit rates and would not be selected.

Since G.711u is listed first in the list of codecs supported by both endpoints, it will be selected over G.711a.
upvoted 1 times

sneff91 1 year, 3 months ago
Untrue. Region1 has a maximum permitted bandwidth of 60kbps to other devices in Region1, not 64kbps. Even if it was 64kbps though, it would still be G.722 as it is listed higher in the Audio Codec Preference list, which uses the FIRST codec in the list that is within the bandwidth threshold in a TOP DOWN fashion, NOT the highest permitted bandwidth. It's A.
upvoted 3 times

Alan100 2 years, 3 months ago
I believe A is best answer here. Order in preference list is factored in the choice. Also checks supported codecs, then the max bit rate. G722 and G711 alaw would both be selected since they are both supported and they are lower than the max audio bit rate then g722 is taken as the best since its higher in the preference list.
upvoted 1 times

Ol_Mykhailiuk 2 years, 9 months ago
Selected Answer: A
In the picture from the drop-down list (Codecs in List*) it is indicated that the codec is 722 use 48k.
upvoted 1 times

vvpark13 3 years, 8 months ago
The answer is B

https://www.cisco.com/c/en/us/td/docs/voice_ip_comm/cucm/admin/10_0_1/ccmsys/CUCM_BK_SE5FCFB6_00_cucm-system-guide-100/CUCM_BK_SE5FCFB6_00_cucm-system-guide-100_chapter_01000.html#CUCM_RF_B90D5F7F_00

Bandwidth Calculations

In performing location bandwidth calculations for purposes of call admission control, Cisco Unified Communications Manager assumes that each call stream consumes the following amount of bandwidth:

G.711 call uses 80 kb/s.
G.722 call uses 80 kb/s.
G.723 call uses 24 kb/s.
G.728 call uses 26.66 kb/s.
G.729 call uses 24 kb/s.
GSM call uses 29 kb/s.
Wideband call uses 272 kb/s.
iLBC call uses 24 kb/s.
AMR call uses 12.2 kb/s.
AMR-WB call uses 23.85 kb/s.
AAC call uses value that video mline specifies.
upvoted 1 times

jarcoman99 3 years, 8 months ago

The correct answer is A. Since there is a explicit intra-region bitrate relationship, the default 64kbps at the service parameters wont take effect, meaning that the 60kbps defined there is the max bitrate value to be used. The information that vvpark provided is related to Location bandwidth calculation, but not to how Region relationship work. Location and Region are two different features for the CUCM.

upvoted 2 times

F3rnando 3 years, 10 months ago

C - according to Preference list G711A is 56k that is the 'best' codec below 60k allowed.

upvoted 1 times

Shazam021 3 years, 11 months ago

On second thought,, yeah didn't notice the 48k G722 on the ACPL. So yeah A is the correct one

upvoted 1 times

Shazam021 3 years, 11 months ago

I think answer should be B. Since Intra-region max bandwidth is capped at 60kbps, G722 and G711 are not valid.

upvoted 1 times

basscov 3 years, 9 months ago

g722 48k (48 kbps) has a priority, so answer is A

upvoted 3 times

basscov 3 years, 9 months ago

Just reproduced this in the lab, it uses g729 , so answer is B

upvoted 2 times

Cirilo 3 years, 9 months ago

but g722 is first in the list

upvoted 2 times

hany20006 3 years, 6 months ago

are you sure you used the G722_48k and not the 64k ?

upvoted 2 times

Question #93 *Topic 1*

```
dspfarm profile 1 mtp
 codec g711ulaw
 maximum sessions software 50
 associate application SCCP
```

Refer to the Exhibit. Which command is required to allow this media resource to handle Video Media streams?

A. codec pass-through

B. associate application Cisco Unified Border Element

C. maximum sessions hardware 50

D. video codec h264

Correct Answer: *A*

Community vote distribution

A (100%)

gahhhaccount Highly Voted 4 years ago

It should be A. The command "codec pass-through" would let media streams be negotiated later in the call.

https://community.cisco.com/t5/ip-telephony-and-phones/mtp-codec-pass-through/td-p/2159568

upvoted 9 times

asdlfhqwoiefnwe Highly Voted 3 years, 5 months ago

Selected Answer: A

A - codec pass-through

dspfarm profile doesn't use the video codec command

upvoted 6 times

b3532e4 Most Recent 9 months ago

Limitations

• Video transcoding is not supported. This document only refers to transcoding for CUBE B2BUA calls.

Refer toSystem Configuration Guide for Cisco Unified Communications Manager for UCM MTPdetails.

So A correct answer

upvoted 1 times

[Removed] 2 years, 7 months ago

Selected Answer: A

codec pass-through

upvoted 1 times

Antioch_KF 3 years, 8 months ago

codec pass-through

upvoted 2 times

WilliamC 3 years, 11 months ago

video codec command is used on dial-peers or voice profile but not on dspfarm profile https://www.cisco.com/c/en/us/td/docs/ios-xml/ios/voice/cube/configuration/cube-book/cube-codec-basic.html

upvoted 2 times

landrey 4 years ago

correct one is codec pass-through

upvoted 3 times

Question #94 *Topic 1*

Which two protocols are proxied over an Expressway-E/C pair when a MRA login including phone services is performed? (Choose two.)

A. SCCP

B. HTTPS

C. H.323

D. SRTP

E. SIP

Correct Answer: *BE*

Community vote distribution

BE (100%)

Panda_man 11 months, 2 weeks ago

Selected Answer: BE

Should be B and E

upvoted 1 times

Omitted 1 year, 7 months ago

Selected Answer: BE

HTTPS for sure. SIP I assume is because of session establishment.

upvoted 1 times

VG224 2 years, 3 months ago

B and E are correct since the question is asking about protocol during Login.

upvoted 4 times

MKZ 2 years, 6 months ago

https://www.cisco.com/c/dam/en/us/td/docs/voice_ip_comm/expressway/config_guide/X8-10/Mobile-Remote-Access-via-Expressway-Deployment-Guide-X8-10.pdf

page 54

upvoted 4 times

Question #95 *Topic 1*

Which DTMF relay method configured on a SIP dial-peer will ensure that a media resource is not invoked by Unified CM for calls to UCCX IVRs?

A. sdtmf-relay h245-signal

B. dtmf-relay rtp-nte

C. dtmf-relay sip-notify

D. dtmf-relay sip-kpml

Correct Answer: *D*

Community vote distribution

D (75%) C (25%)

gahhhhaccount Highly Voted 4 years ago

It should be D "dtmf-relay sip-kpml". UCCX only supports out of band DTMF.

Here is a link about how somebody resolved it. https://community.cisco.com/t5/contact-center/dtmf-not-working-with-uccx/td-p/2317592

upvoted 12 times

Nicetomeetyou Highly Voted 4 years ago

I agree the correct answer should be D, also see best practices for some background info

https://www.cisco.com/c/dam/en/us/td/docs/ios-xml/ios/voice/cube_uccx/cube_uccx_interop_bestprac_guide.pdf

upvoted 8 times

Stevon Most Recent 4 weeks, 1 day ago

Selected Answer: D

configuring the DTMF relay method on a SIP dial-peer can prevent Unified CM from invoking a media resource (like an MTP) for calls to UCCX IVRs. Specifically, if the dial-peer is configured to use RTP NTE as the DTMF relay method, CUCM will attempt to convert in-band DTMF tones to out-of-band signaling messages, which requires an MTP to be present and invoked.

Explanation:

RTP NTE (RFC2833):

This method relays DTMF as in-band tones within the RTP stream.

UCCX and DTMF:

UCCX primarily relies on out-of-band DTMF signaling (like SIP INFO or KPML) to interpret DTMF digits.

CUCM and MTPs:

CUCM can convert in-band RTP NTE to out-of-band signaling if an MTP is configured and invoked.

Avoiding MTPs:

By configuring the dial-peer to use an out-of-band DTMF method (like SIP INFO or KPML) instead of RTP NTE, you can avoid having CUCM attempt to convert the DTMF, thus preventing the need for an MTP to be invoked

upvoted 1 times

G0y0 3 months, 3 weeks ago

Selected Answer: C

C. and D. are correct both.

However, if you want a tiebreaker take in mind the following:

++sip-kpml is DTMF Relay via KPML over SIP SUBCRIBE/NOTIFY, it is a standard (RFC 4730).

++sip-notify is DTMF Relay via SIP NOTIFY messages (also called unsolicited notify). This is a Cisco proprietary implementation and is not standardized in any IETF RFC. It is the same method that Unity and CUCM uses to MWI indicators.

In this scenario, I consider that the correct one, taking in account the consideration about Cisco proprietary, correct answer is C.

Reference:

Understanding SBC's Comprehensive Guide to Designing, Deploying, Troubleshooting, and Maintaining CUBE Solutions.

upvoted 1 times

G0y0 3 months, 3 weeks ago

Nother thing is that it is not mentioned what SBC is in use, it could be an Oracle, a Ribbon, etc. However in the question talking about the dial-peers, it targets to a Cisco Gateway or a CUBE. If we where talking about for example a Asterisk Call Control or a Ribbon SBC, we can consider to choose RFC 4730 (KPML). However in this case the SBC is a CUBE or a Cisco Gateway, so to be more specific and compatibility, C, could be the best answer. It is my criteria, you can choose D that it is correct too, but take in mind that one thing is to be facing a Cisco CCNP exam, and another think is facing a real scenario as for example an implementation or a project.

upvoted 1 times

b3532e4 9 months, 1 week ago

Dial Peer Facing CUCM:

dial-peer voice 10 voip

dtmf-relay sip-kpml sip-notify

Dial Peer Facing ITSP:

dial-peer voice 11 voip

dtmf-relay rtp-nte

upvoted 1 times

Brant 11 months, 1 week ago

B is the right answer.

Catch is media resources should not be invoked. Out of band Dtmf relay will invoke media resources.

upvoted 2 times

GCISystemIntegrator 1 year, 9 months ago

guys .... I had this issue, and my dial-peers were offering "dtmf-relay rtp-nte sip-notify sip-kpml", yet the DTMF was not being recognized by UCCX. Setting it to only "dtmf-relay sip-kpml" resolved the issue. than answer is D

upvoted 2 times

aocstr 1 year, 11 months ago

Selected Answer: D

right in the beginning we can see the answer is wrong here. Correct is D

https://www.cisco.com/c/dam/en/us/td/docs/ios-xml/ios/voice/cube_uccx/cube_uccx_interop_bestprac_guide.pdf

upvoted 2 times

MeowthL 2 years, 3 months ago

Selected Answer: C

i'm thinking the answer might be C

it's because the question is mentioning "is not invoked by Unified CM for calls to UCCX IVRs"

upvoted 2 times

asdlfhqwoiefnwe 3 years, 5 months ago

Selected Answer: D

D - dtmf-relay sip-kpml

UCCX only supports out-of-band DTMF. This can be applied at the CUBE to avoid a media resource being needed to address as it can directly handle the DTMF relay from RTP-NTE to SIP-KPML

https://www.cisco.com/c/dam/en/us/td/docs/ios-xml/ios/voice/cube_uccx/cube_uccx_interop_bestprac_guide.pdf

upvoted 4 times

jarcoman99 3 years, 8 months ago

Both C and D are valid answers from the CUBE stand point. CUCM has to convert the OOB DTMF into JTAPI and then send it to UCCX. For a more specific answer, the CUCM configuration has to be counted as well. From my experience working with CUBE, CUCM and UCCX, enabling dtmf-relay sip-kpml at the dial-peer is enough to make it work without an MTP.

upvoted 3 times

ring_phone 3 years, 8 months ago

C - https://www.cisco.com/c/dam/en/us/td/docs/ios-xml/ios/voice/cube_uccx/cube_uccx_interop_bestprac_guide.pdf

upvoted 2 times

Pashat 3 years, 5 months ago

Great document. Thanks for the link.

This document specifies both KPML and NOTIFY.

Both are OOB DTMF Methods.

upvoted 1 times

Question #96 *Topic 1*

Which two protocols can be configured for the Cisco Unity Connection and Cisco UCM integration? (Choose two.)

A. RTP

B. SCCP

C. MGCP

D. H.323

E. SIP

Correct Answer: *BE*

Reference:

https://www.cisco.com/c/en/us/td/docs/voice_ip_comm/connection/11x/design/guide/b_11xcucdg/b_11xcucdg_chapter_00.html#ID-2342-00000133

Currently there are no comments in this discussion, be the first to comment!

Question #97 *Topic 1*

Which task is required when configuring self-provisioning for an end user in Cisco UCM?

A. Associate the end user to a SIP Profile

B. Enable Auto-Registration

C. Disable Auto-Registration

D. Associate the end user to the Standard CCM Super Users group

Correct Answer: *B*

Community vote distribution

B (100%)

asdlfhqwoiefnwe Highly Voted 1 year, 11 months ago

Selected Answer: B

B is correct. This auto registration must be enabled for the self-provisioning to work due to needing the phone able to register to CUCM first.

upvoted 6 times

iExpo_91 Most Recent 9 months ago

Selected Answer: B

The Answer is B link below on the steps and requirements

https://www.cisco.com/c/en/us/td/docs/voice_ip_comm/cucm/admin/12_5_1/systemConfig/cucm_b_system-configuration-guide-1251/cucm_b_system-configuration-guide-1251_chapter_01001000.html

upvoted 2 times

Vijay_ABI 2 years, 1 month ago

The question is for END USER provisioning, not the phones. So this answer doesnt seem right

upvoted 1 times

Omitted 1 year, 10 months ago

Read again, “self provisioning for an End user”

upvoted 1 times

motsar111 10 months, 2 weeks ago

Self-provisioning is about registering phones on an automated way by the end-user. If auto-registration is not enabled it will not work

upvoted 1 times

Question #98 *Topic 1*

Which two steps should be taken to provision a phone after the Self-Provisioning feature was configured for end users? (Choose two.)

A. Dial the hunt pilot extension and associate the phone to an end user

B. Plug the phone into the network

C. Ask the Cisco UCM administrator to associate the phone to an end user

D. Enter settings menu on the phone and press *, *,#(star, star, pound)

E. Dial the self-provisioning IVR extension and associate the phone to an end user

Correct Answer: *BE*
Reference:
https://www.cisco.com/c/en/us/td/docs/voice_ip_comm/cucm/admin/12_0_1/systemConfig/cucm_b_system-configuration-guide-1201/cucm_b_system- configuration-guide-1201_chapter_01001000.pdf

jonycakes Highly Voted 1 year ago
B & E are correct:
Source: https://www.cisco.com/c/en/us/td/docs/voice_ip_comm/cucm/admin/12_0_1/systemConfig/cucm_b_system-configuration-guide-1201/cucm_b_system-configuration-guide-1201_chapter_01001000.pdf
Self-Provisioning Phones:
When the feature is configured,you can provision a phone by doing the following:
• Plug the phone into the network.
• Dial the self-provisioning IVR extension.
• Follow the prompts to configure the phone,and associate the phone to an end user. Depending on how you have configured self-provisioning,the end user may to enter the user password,PIN, or an administrative authentication code.
upvoted 7 times

devadarshan91730 Most Recent 12 months ago
• Plug the phone into the network.
• Dial the self-provisioning IVR extension.

https://www.cisco.com/c/en/us/td/docs/voice_ip_comm/cucm/admin/11_0_1/sysConfig/CUCM_BK_C733E983_00_cucm-system-configuration-guide-1101/Configure_Self_Provisioning.pdf
upvoted 3 times

Question #99 Topic 1

An engineer is configuring IP telephony. The network relies on DHCP to provide TFTP server addresses to the endpoints. Policy requires the endpoints to receive two server addresses. Which DHCP option must be configured?

A. 66

B. 143

C. 150

D. 166

Correct Answer: *C*

Community vote distribution

C (100%)

Stevon 2 weeks, 5 days ago

Selected Answer: C

The correct answer is C. DHCP Option 150.

Here's why:

DHCP Option 150 is used specifically for Cisco IP phones to provide one or more TFTP server addresses.

It supports multiple TFTP server addresses, making it ideal for environments requiring redundancy or load balancing.

Other options like Option 66 only support a single TFTP server address, which does not meet the requirement for two addresses.

upvoted 1 times

Gary1968 5 months, 1 week ago

Selected Answer: C

With Option 66 you can actually have multiple TFTP servers referenced within DNS if you use FQDN. However, the most applicable answer where both options are presented is C.

upvoted 1 times

AJBELL14 11 months ago

There is another question which only has the Option 66 for one TFTP server. In that instance Option 66 is correct. In this case, since we have 2 TFTP servers, it would be Option 150

upvoted 2 times

Omitted 1 year, 1 month ago

Selected Answer: C

Option 66 provides a single TFTP server address. Option 150 provides a list of TFTP server addresses. Also if both are configured, Cisco phones will target 150 by default before 66.

upvoted 4 times

Question #100 *Topic 1*

```
Voice class codec 1
codec preference 1 g711alaw
codec preference 2 g711ulaw
codec preference 3 g729r8

dial-peer voice 13 voip
description incoming dialpeer from ITSP
incoming called-number
voice-class codec 1

dial-peer voice 19 voip
description outgoing dialpeer to CUCM
destination-pattern .T
session protocol sipv2
session-target ipv4:3.3.3.3
voice-class codec 1

Incoming SDP from ITSP

v=0
0=sip:test@2.2.2.2 1 16 IN IP4 2.2.2.2
s=sip:test@2.2.2.2
c=IN IP4 2.2.2.2
t=0 0
m=audio 5000 RTP/AVP 18 0
a=rtpmap:0 PCMU/8000/1
a=rtpmap:18 G729/8000/1
```

Refer to the exhibit. Which outgoing m-line SDP is sent to Cisco UCM after matching the appropriate dial peers via Cisco Unified Border Element?

A. m=audio 16550 RTP/AVP 0 8 18 a=rtpmap:0 PCMU/8000/1 a=rtpmap:8 PCMA/8000/1 a=rtpmap:18 G729/8000/1

B. m=audio 16550 RTP/AVP 18 0 a=rtpmap:0 PCMU/8000/1 a=rtpmap:18 G729/8000/1

C. m=audio 16550 RTP/AVP 8 0 18 a=rtpmap:0 PCMU/8000/1 a=rtpmap:8 PCMA/8000/1 a=rtpmap:18 G729/8000/1

D. m=audio 16550 RTP/AVP 18 0 a=rtpmap:8 PCMA/8000/1 a=rtpmap:0 PCMU/8000/1 a=rtpmap:18 G729/8000/1

Correct Answer: *B*

Community vote distribution

B (83%) C (17%)

kljw5 5 months, 1 week ago

Selected Answer: B

This question seems to have no valid correct answer as the incoming SDP only offers 0 and 18 and not 8 so the outgoing SDP would match the available codecs in order of preference based on voice codec 1 which is 0 18. B is the closest available answer although the order of codec preference is incorrect.

upvoted 1 times

b3532e4 9 months, 4 weeks ago

The outgoing SDP will prefer PCMA (G.711alaw) first (RTP/AVP 8), followed by PCMU (G.711ulaw) (RTP/AVP 0), and finally G.729 (RTP/AVP 18), based on the codec preferences set in the voice-class codec 1 on both dial peers.

upvoted 1 times

kljw5 5 months, 1 week ago

This is wrong, The incoming SDP does not include 8 which is alaw. 0 and 18 so the outgoing SDP would prioritice ulaw 0 then 18 based on voice codec 1

upvoted 1 times

pasangawa 1 year, 1 month ago

Selected Answer: C

This should be C

https://en.wikipedia.org/wiki/RTP_payload_formats

8 - G711alaw

0 -G711ulaw

18 - G729

upvoted 1 times

kljw5 5 months, 1 week ago

Wrong, incoming SDP does not offer 8

upvoted 1 times

pasangawa 1 year, 1 month ago

change my mind, based on Cisco: https://www.cisco.com/c/en/us/td/docs/ios-xml/ios/voice/cube/ios-xe/config/ios-xe-book/m_voi-negt-aud-code.html#reference_D708A3F6727C4E47BBE1C0B9B417BFA7

In it, this note: Codec preference in the voice class codec on the outgoing call leg is not followed when the same codecs are available in the respective incoming invite with SDP with different codec preference. Cube prioritizes and follows the incoming invite with SDP codec preference when compared to the voice class codec preference on the outgoing dial-peer leg. So should be B.

upvoted 2 times

Piji 2 years, 10 months ago

Selected Answer: B

B is correct answer.

upvoted 3 times

Omitted 3 years, 1 month ago

Selected Answer: B

This should be B - it compares with the invite before sending to CUCM leaving G.711ulaw and G.729 remaining. For some reason the order is messed up but it seems like the best answer still.

upvoted 2 times

Question #101 *Topic 1*

An end user at a remote site is trying to initiate an Ad Hoc conference call to an end user at the main site. The conference bridge is configured to support G.711.
The remote user's phone only supports G.729. The remote end user receives an error message on the phone: `Cannot Complete Conference Call.`
What is the cause of the issue?

A. A software conference bridge is not assigned

B. The remote phone does not have the conference feature assigned

C. A Media Termination Point is missing

D. The transcoder resource is missing

Correct Answer: *D*

Community vote distribution

D (100%)

sneff91 9 months, 4 weeks ago

Selected Answer: D

Media resources not assigned are part of the NULL MRG and would still be used, so not A. Not B, as the conference feature is not selectable on a device/phone level setting. Not C, as MTP can only transcode calls between G.711ulaw/alaw and. A transcoding resource is required, so D.

upvoted 3 times

Question #102 *Topic 1*

Which transport protocol does the application layer protocol SNMP use?

A. XML

B. UDP

C. SIP

D. HTTP

Correct Answer: *B*

Reference:

https://www.geeksforgeeks.org/simple-network-management-protocol-snmp/

Griswald Highly Voted 11 months, 1 week ago

SNMP is an application layer protocol which uses UDP port number 161/162. SNMP is used to monitor the network, detect network faults and sometimes even used to configure remote devices. It is a software management software module installed on a managed device.

upvoted 15 times

Question #103 *Topic 1*

Which protocol does Cisco Prime Collaboration Assurance use to poll the health status of different systems in the Collaboration environment?

A. SIP

B. SNMP

C. SCCP

D. SMTP

Correct Answer: *B*

Reference:

https://www.cisco.com/c/en/us/products/collateral/cloud-systems-management/prime-collaboration/guide-c07-736946.html#_Toc446633083

Currently there are no comments in this discussion, be the first to comment!

Question #104 *Topic 1*

When a remote office location is set up with limited bandwidth resources, which codec would allow the most voice calls with the limited bandwidth?

A. G.711

B. G.722

C. G.723

D. G.729

Correct Answer: *D*
Reference:
https://www.cisco.com/c/en/us/td/docs/voice_ip_comm/cucm/srnd/collab10/collab10/media.html

Bazant 1 year ago
G.723 is NOT the same as G723.1 (available in CUCM). SO the answer for that question is correct. G.729 will allow for most number of calls.
upvoted 3 times

domangez 1 year, 6 months ago
G.723 (G.723.1) needs less BW (5.3/6.3 kbps) then G.729. It is not that well supported as G.729.
upvoted 1 times

jarcoman99 1 year, 2 months ago
G.723 is completely different than G.723.1

G.723 is an ITU standard for speech codecs that uses the ADPCM method and provides good quality audio at 24 and 40 Kbps.

Note: G.723 codec mainly used for digital circuit multiplication equipment (DCME) applications. And latter folded into G.726. Kindly see the G.726

G.723.1 is a speech codec that compresses voice audio in 30 ms frames. An algorithmic look-ahead of 7.5 ms duration means that total algorithmic delay is 37.5 ms.
upvoted 3 times

Question #105 *Topic 1*

```
INVITE sip:4000@172.16.1.1:5061 SIP/2.0
Via: SIP/2.0/TLS 172.16.2.143:5061;branch=z9hG4bK8FD315E7
Remote-Party-ID: <sip:+14088335000@172.16.2.143>;party=calling;screen=no; privacy=off
From: <sip:+14088335000@172.27.2.143>;tag=7B42E5F6-9B8
To: <sip:4000@172.16.1.1>
Date: Tue, 06 Aug 2019 15:03:05 GMT
Call-ID: 4EA4363-B77111E9-8A4AFFCF-10B6D71B@172.16.2.143
Supported: 100rel,timer,resource-priority, replaces,sdp-anat
Min-SE: 1800
Cisco-Guid: 0082391505-3077640681-2319777743-0280418075
User-Agent: Cisco-SIPGateway/IOS-15.5.3.S4b
Allow: INVITE, OPTIONS, BYE, CANCEL, ACK, PRACK, UPDATE, REFER, SUBSCRIBE, NOTIFY, INFO, REGISTER
CSeq: 101 INVITE
Timestamp: 1565089565
Contact: <sip:+ 14088335000@172.16.2.143:5061;transport=tls>
Expires: 180
Allow-Events: telephone-event
Max-Forwards: 68
Content-Type: application/sdp
Content-Disposition: session;handling=required
Content-Length: 416
v=0
o=CiscoSystemsSIP-GW-UserAgent 8486 8298 IN IP4 172.16.2.143
s=SIP Call
c=IN IP4 172.16.2.143
t=0 0
m=audio 44612 RTP/SAVP 0 101
c=IN IP4 172.16.2.143
a=crypto:XXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXX
a=crypto:XXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXX
a=rtpmap:0 PCMU/8000
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-16
a=ptime:20
```

Refer to the exhibit. This INVITE is sent to an endpoint that only supports G.729. What must be done for this call to succeed?

A. Nothing: both sides support G.729.

B. Add a transcoder that supports G.711ulaw and G.729.

C. Add a media termination point that supports G.711ulaw and G.729.

D. Nothing: both sides support payload type 101.

Correct Answer: *B*

Community vote distribution

B (100%)

BarryR Highly Voted 4 years, 12 months ago

Answer is B. "A transcoder takes the media stream of one codec and transcodes (converts) it from one compression type to another compression type. For example, it could take a stream from a G.711 codec and transcode (convert) it in real time to a G.729 stream. In addition to codec conversion, a transcoder resource can also provide MTP/TRP functionality to a call."

upvoted 18 times

AbdurrahmanBNC Most Recent 11 months, 3 weeks ago

Answer B

D. Nothing: both sides support payload type 101: Payload type 101 refers to DTMF events, not the actual audio codec

upvoted 1 times

BrianC 2 years, 3 months ago

Selected Answer: B

Correct answer is B.

upvoted 2 times

H31d1 2 years, 7 months ago

Selected Answer: B

101 is "telephone-event" --> DTMF

upvoted 1 times

Piji 2 years, 10 months ago

Selected Answer: B

Correct answer is B.

upvoted 4 times

[Removed] 3 years ago

https://www.cisco.com/c/en/us/td/docs/ios-xml/ios/voice/cube/configuration/cube-book/dtmf-relay.html

i think the reason it allows SUBSCRIBE/NOTIFY is because it allows kpml. subscribe is just the initial registration of the NOTIFY out-of-band method. the actual keys are transmitted in the NOTIFY then:
sip-kpml —This method is available only on SIP dial peers. The RFC 4730 defines the out-of-band DTMF relay mechanism to register the DTMF signals using the SIP-Subscribe messages. It transports the DTMF signals using the SIP-Notify messages containing an XML-encoded body. This method is calked Key Press Markup Language.

If you configure KPML on the dial peer, the gateway sends INVITE messages with KPML in the Allow-Events header.

NOTIFY sip:192.168.105.25:5060 SIP/2.0
Event: kpml
<?xml version="1.0" encoding="UTF-8"?>

upvoted 2 times

Niko11 2 years, 11 months ago

Hi, have you had exam recently? many new questions?

upvoted 1 times

ocero 4 years, 4 months ago

The answer is B.
The invite come with PCMU/8000, and not see G729/8000... transcoding is need it.

upvoted 2 times

Nebbiaman 4 years, 6 months ago

Lol D is not the answer.
The invite is offered with only g711. If the invited endpoint only support g729, how can one think the answer is D?
The answer is B.

upvoted 4 times

tebzades 4 years, 9 months ago

B is correct, transcoder is needed for translating

upvoted 3 times

rishik 4 years, 11 months ago

D is correct as the call would be negotiated with G729

upvoted 3 times

rishik 4 years, 11 months ago

On reading more a transcoder is needed, so B is correct

upvoted 6 times

Rolrik 4 years, 11 months ago

A call can be proceed without audio support so payload 101 is suffisent.

upvoted 1 times

CollabGuy 4 years, 8 months ago

I don't think a "no audio" call to be considered successful :)

upvoted 9 times

nassar1 5 years ago

I meant should be B.

upvoted 4 times

nassar1 5 years ago

Should be C. Sender supports G.711u only. Tranccoder is needed.

upvoted 1 times

Question #106 | *Topic 1*

```
Endpoint A:
m=audio 21796 RTP/AVP 108 9 104 105 101
  b=TIAS:64000
  a=extmap:14 http://protocols.cisco.com/timestamp#100us
  a=rtpmap:108 MP4A-LATM/90000
  a=fmtp:108 bitrate=64000;profile-level-id=24;object=23
  a=rtpmap:9 G722/8000
  a=rtpmap:104 G7221/16000
  a=fmtp:104 bitrate=32000
  a=rtpmap:105 G7221/16000
  a=fmtp:105 bitrate=24000
  a=rtpmap:101 telephone-event/8000
  a=fmtp:101 0-15
  a=trafficclass:conversational.audio.immersive.aq:admitted

Endpoint B:
m=audio 21796 RTP/AVP 105 0 8 18 101
  b=TIAS:64000
  a=extmap:14 http://protocols.cisco.com/timestamp#100us
  a=rtpmap:105 G7221/16000
  a=fmtp:105 bitrate=24000
  a=rtpmap:0 PCMU/8000
  a=rtpmap:8 PCMA/8000
  a=rtpmap:18 G729/8000
  a=fmtp:18 annexb=no
  a=rtpmap:101 telephone-event/8000
  a=fmtp:101 0-15
  a=trafficclass:conversational.audio.immersive.aq:admitted
```

Refer to the exhibit. Endpoint A calls endpoint B. What is the only audio codec that can be used for the call?

A. Telephone-event/8000

B. G7221/16000

C. PCMA/8000

D. G722/8000

Correct Answer: *B*

ccosta7 Highly Voted 2 years ago

B is correct

upvoted 15 times

XalaGyan 2 years ago

"G7221/16000" will be chosen. I am not sure about ABCD therefore i am pasting the text. the G722/8000 has a different sampling rate and would not adapt. Call will hung up immediately upon answering.

upvoted 2 times

DEFAULTNERD Highly Voted 2 years ago

a=rtpmap:105 G7221/16000

upvoted 5 times

IMMohit Most Recent 1 year, 2 months ago

As 101 is common, Option A. Telephone-event/8000 will be selected

upvoted 1 times

Bazant 1 year ago

It that case 101 is DTMF method not audio codec. The questions asks for audio codec which will be chosen. Correct answer is B

upvoted 4 times

cli4cli 1 year, 2 months ago

B is correct answer

upvoted 1 times

jumper2099 1 year, 11 months ago

why you think the correct answer is D? can someone clarify this for me, please?
upvoted 1 times

YarinBenAharon 1 year, 12 months ago
i cant understand why some pepole think that g722/8000 chosen.
g722 is not part of the supported codec of Device B
upvoted 1 times

MKZ 1 year, 6 months ago
Both devices support G722.1 - > a=rtpmap:105 G7221/16000
upvoted 3 times

DEFAULTNERD 2 years ago
Can anybody explain why they think that G722/800 would be chosen? Also the PT is in the dynamic for G7221 but I can;t fine where it has a static PT.
upvoted 2 times

DEFAULTNERD 2 years ago
However the PT is wrong for G7221
upvoted 1 times

DEFAULTNERD 2 years ago
PT Value is in the dynamic range.
upvoted 2 times

leeyy 2 years, 1 month ago
The answer should be D-G722/8000
upvoted 2 times

Question #107 Topic 1

```
INVITE sip:1@10.10.10.219;user=phone SIP/2.0
Via: SIP/2.0/TCP 10.10.10.84:50083;branch=z9hG4bK471df613
From: "1234 - My Phone" <sip:1234@10.10.10.219>;tag=381c1aba7a78002c558eda31-12b8af63
To: <sip:1@10.10.10.219>
Call-ID: 381c1aba-7a78000d-4ca6894a-41dd3e0f@10.10.10.84
Max-Forwards: 70
CSeq: 101 INVITE
Contact: <sip:1234@10.10.10.84:50083;transport=tcp>
Allow: ACK,BYE,CANCEL,INVITE,NOTIFY,OPTIONS,REFER,REGISTER,UPDATE,SUBSCRIBE,INFO
Allow-Events: kpml,dialog
Content-Type: application/sdp
Content-Length: 658

v=0
o=Cisco-SIPUA 26529 0 IN IP4 10.10.10.84
s=SIP Call
b=AS:4064
t=0 0
m=audio 32136 RTP/AVP 114 9 124 113 115 0 8 116 18
c=IN IP4 10.10.10.84
b=TIAS:64000
a=rtpmap:114 opus/48000/2
a=fmtp:114
maxplaybackrate=16000;sprop-maxcapturerate=16000;maxaveragebitrate=64000;stereo=0;sprop-
stereo=0;usedtx=0
a=rtpmap:9 G722/8000
a=rtpmap:124 ISAC/16000
a=rtpmap:113 AMR-WB/16000
a=fmtp:113 octet-align=0,mode-change-capability=2
a=rtpmap:115 AMR-WB/16000
a=fmtp:115 octet-align=1,mode-change-capability=2
a=rtpmap:0 PCMU/8000
a=rtpmap:8 PCMA/8000
a=rtpmap:116 iLBC/8000
a=fmtp:116 mode=20
a=rtpmap:18 G729/8000
a=fmtp:18 annexb=yes
a=sendrecv
```

Refer to the exhibit. When a UC Administrator is troubleshooting DTMF negotiated by this SIP INVITE, which two messages should be examined next to further troubleshoot the issue? (Choose two.)

A. REGISTER

B. UPDATE

C. PRACK

D. NOTIFY

E. SUBSCRIBE

Correct Answer: *DE*

Community vote distribution

DE (100%)

Griswald Highly Voted 3 years ago

The DTMF Events through SIP Signaling feature allows telephone event notifications to be sent through SIP NOTIFY messages, using the SIP SUBSCRIBE/NOTIFY method as defined in the Internet Engineering Task Force (IETF) draft, SIP-Specific Event Notification.

upvoted 17 times

v1nhthanh Most Recent 2 months, 1 week ago

Selected Answer: DE

No 101 event - therefore OOB. Notify and Subscribe

upvoted 1 times

Piji 10 months, 2 weeks ago

Selected Answer: DE

D,E correct answer.

upvoted 2 times

Konrad88 1 year, 10 months ago

I think it should be B & C (PRACK & UPDATE). Both of these include SDP where DTMF is negotiated.

upvoted 1 times

> **Konrad88** 1 year, 10 months ago
>
> After more study I believe they are looking for SUBSCRIBE & NOTIFY. Seems to be a Cisco thing. I haven't seen it in the general world of SIP.
>
> upvoted 3 times

MKZ 2 years ago

SUBSCRIBE - subscribes for an event of notification from the notifier (KPML & MWI)

NOTIFY - Notify the subscriber of a new event (used for KPML $ MWI)

upvoted 4 times

SDLOA14 2 years, 4 months ago

I'm going to have to investigate this question. I don't see any DTMF negotiation in this message. Nothing like telephony-event 101 configuration.

upvoted 1 times

MrAshour 2 years, 6 months ago

Thank you so much for the explanation :)

upvoted 2 times

Question #108 *Topic 1*

An engineer wants to manually deploy a Cisco Webex DX80 video endpoint to a remote user. Which type of provisioning is configured on the endpoint?

A. Cisco Unified Border Element

B. Cisco Meeting Server

C. Cisco Unity Connection

D. Edge

Correct Answer: *D*

Community vote distribution

D (100%)

Slushed Highly Voted 3 years, 1 month ago

Selected Answer: D

D is the correct answer. The provisioning modes/types are Off/Auto/CUCM/Edge/Spark/TMS/VCS.
https://www.cisco.com/c/dam/en/us/td/docs/telepresence/endpoint/ce95/dx70-dx80-administrator-guide-ce95.pdf#D1536209_DX70-DX80_Administrator_Guide_CE95.indd%3A.859768%3A307505

upvoted 7 times

R_oB3rT Most Recent 10 months, 4 weeks ago

Also for me the answer is D; but if in the answers is there also "CUCM" option, what is the best answer?

upvoted 1 times

JoeC716 1 year, 1 month ago

Selected Answer: D

Expressway EDGE

upvoted 1 times

Komy 1 year, 1 month ago

Selected Answer: D

D:

Edge is Correct

upvoted 1 times

MeowthL 2 years, 3 months ago

if the answer have CUCM and Edge, i will choose CUCM as the best answer

upvoted 1 times

Piji 2 years, 10 months ago

Selected Answer: D

D is the correct answer.

upvoted 1 times

AJBELL14 2 years, 11 months ago

Selected Answer: D

I agree with @Slushed ..Edge is the answer

upvoted 1 times

Question #109 *Topic 1*

```
v=0
o=Cisco-SIPUA 13439 0 IN IP4 10.10.10.10
s=SIP Call
b=AS:4064
t=0 0
m=audio 0 RTP/AVP 114 9 124 113 115 0 8 116 18 101
c=IN IP4 10.10.10.10
b=TIAS:64000
a=rtpmap:114 opus/48000/2
a=fmtp:114 maxplaybackrate=16000;sprop-maxcapturerate=16000;maxaveragebitrate=
64000;stereo=0;sprop-stereo=0;usedtx=0
a=rtpmap:9 G722/8000
a=rtpmap:124 ISAC/16000
a=rtpmap:113 AMR-WB/16000
a=fmtp:113 octet-align=0,mode-change-capability=2
a=rtpmap:115 AMR-WB/16000
a=fmtp:115 octet-align=1,mode-change-capability=2
a=rtpmap:0 PCMU/8000
a=rtpmap:8 PCMA/8000
a=rtpmap:116 iLBC/8000
a=fmtp:116 mode=20
a=rtpmap:18 G729/8000
a=fmtp:18 annexb=yes
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-15
a=sendrecv
```

Refer to the exhibit. A call is failing to establish between two SIP Devices. The called device answers with this SDP. Which SDP parameter causes this issue?

A. The payload for G.711ulaw must be 18.

B. The calling device did not offer a ptime value.

C. The media stream is set to sendonly.

D. The RTP port is set to 0.

Correct Answer: *D*

Community vote distribution

D (100%)

Alan100 9 months, 1 week ago

By Elimination: G711ulaw is 0, 18 is G729, ptime value is an optional field, media stream is set to sendrecv not send only. So D is the only viable answer

upvoted 1 times

Panda_man 11 months, 1 week ago

Selected Answer: D

the offer should contain valid port number

upvoted 1 times

MKZ 2 years, 6 months ago

The RTP port is set to 0 --- Where do you see it?

upvoted 2 times

Collabhunter 2 years, 6 months ago

M line

upvoted 3 times

MKZ 2 years, 6 months ago

The format of media announcement is as follows.

m=<media> <port> <transport> <fmt list>

The first sub-field is the media type. Currently defined media for TMedia is "audio".

The second sub-field is the transport port to which the media stream will be sent. In Smart Media we don't specify it in Profile SDP Description and therefore you should insert "0".

upvoted 5 times

Question #110 *Topic 1*

An engineer encounters third-party devices that do not support Cisco Discovery Protocol. What must be configured on the network to allow device discovery?

A. LACP

B. TFTP

C. LLDP

D. SNMP

Correct Answer: *C*

Community vote distribution

C (100%)

Mli2604 Highly Voted 3 years, 2 months ago

LLDP Link Layer Discovery Protocol

upvoted 10 times

Panda_man Most Recent 11 months, 1 week ago

Selected Answer: C

LLDP is support for third party devices

upvoted 1 times

Question #111 *Topic 1*

On which protocol and port combination does Cisco Prime Collaboration receive notifications (Traps and InformRequests) from several network devices in the
Collaboration infrastructure for which it has requested notifications?

A. UDP 162

B. TCP 80

C. UDP 161

D. TCP 161

Correct Answer: *A*
Reference:
https://www.cisco.com/c/en/us/td/docs/net_mgmt/prime/collaboration/12-1/assurance/advanced/guide/cpco_b_cisco-prime-collaboration-assurance- guide-advanced-12-1/cpco_b_cisco-prime-collaboration-assurance-guide-advanced-12-1_chapter_01111.html

Community vote distribution

A (100%)

Stevon 3 weeks, 2 days ago

Selected Answer: A

Cisco Prime Collaboration uses UDP 162 to receive notifications (Traps and InformRequests) from network devices. This protocol and port combination is crucial for monitoring and managing the Collaboration infrastructure. Other port and protocol combinations, such as TCP 80, UDP 161, and TCP 161, are not used for this specific purpose.
Here's a more detailed explanation:
UDP (User Datagram Protocol): UDP is a connectionless protocol that is faster than TCP, making it suitable for real-time notifications.
162: This is the standard port number for SNMP traps.
Traps and InformRequests: These are different types of SNMP notifications. Traps are sent by a device to indicate an event, while InformRequests are more reliable and require acknowledgement from the receiver.
SNMP (Simple Network Management Protocol): SNMP is a protocol used for managing and monitoring network devices.

upvoted 1 times

RdTx 11 months, 3 weeks ago

Selected Answer: A

UDP port 161 connects the SNMP Managers with SNMP Agents (i.e. polling)
UDP port 162 is used when SNMP Agents send unsolicited traps to the SNMP Manager

upvoted 4 times

Hussein1985 1 year, 5 months ago

A is correct answer

upvoted 1 times

rishik 3 years, 5 months ago

A is correct answer

upvoted 4 times

infamous476 3 years, 6 months ago

Correct Answer A:

· The SNMP manager receives notifications on UDP port 162 (TRAPS and INFORM).
From <https://snmpcenter.com/what-is-snmp/>

upvoted 2 times

Griswald 3 years, 6 months ago

Networking Devices PCA 162:UDP To receive SNMP traps.

upvoted 1 times

Question #112 *Topic 1*

Which two actions must be taken to provision a new device using the self-provisioning? (Choose two.)

A. Enable the self-provisioning IVR in the Cisco UCM.

B. Import the user profile to the corporate LDAP directory.

C. Link the appropriate universal device template to the user profile.

D. Ensure the user has a directory URI and a primary extension.

E. Link the appropriate service profile to the provisioning template.

Correct Answer: *AC*

Community vote distribution

AC (100%)

[Removed] Highly Voted 2 years, 6 months ago

Selected Answer: AC

It should be A and C by common sense, you do not need an URI.
Self-Provisioning of a phone to a user needs:

* an End user with a primary extension and a Feature Group Template
* the Feature Group Template needs a User Profile
* the User Profile needs a Universal Line and Universal Device Template

finally to activate it you need a CTI route point and a configured IVR service to actually register it.

HOWEVER, it looks like you dont actually NEED the IVR to technically fullfil the goal of self-registration, as users can use the URL based registration.
but i am not 100% sure if you wont need the IVR anyways, even if you arent using it - the guide is not 100% clear on that

https://www.cisco.com/c/en/us/support/docs/unified-communications/unified-communications-manager-callmanager/214228-configure-self-provisioning-feature-on-c.html

upvoted 6 times

G0y0 3 months, 3 weeks ago

whether if method URL or method IVR, should you need an user profile. Both methods are different and independent

upvoted 1 times

DaKenjee Highly Voted 2 years, 1 month ago

Selected Answer: AC

URL Based Explanation by FlyingThunder is another option,
but Guide below, explains other method with IVR, which is mentioned here and suitable.

Problem here is that Answer A,C and D suits

But when i assume question focus on activating self-provisioning, Answer D get skipt by Step 5:
Peer Guide
(Answer A) Activate Services for Self-Provisioning
Step 4 Under CTI Services, check Self Provisioning IVR.
(Answer C) Enable Autoregistration for Self-Provisioning
Step 3 Select the Universal Device Template that you want to be applied to provisioned phones.
Step 5 Use the Starting Directory Number and Ending Directory Number fields to enter a range of directory numbers to apply to provisioned phones.

https://www.cisco.com/c/en/us/td/docs/voice_ip_comm/cucm/admin/11_5_1/sysConfig/11_5_1_SU1/cucm_b_system-configuration-guide-1151su1/cucm_b_system-configuration-guide-1151su1_chapter_01001000.html

upvoted 5 times

pasangawa Most Recent 11 months, 3 weeks ago

Selected Answer: AC

https://www.cisco.com/c/en/us/td/docs/voice_ip_comm/cucm/admin/12_0_1/systemConfig/cucm_b_system-configuration-guide-1201/cucm_b_system-configuration-guide-1201_chapter_01001000.html#CUCM_TP_S5DD8938_00

"Your end users must be associated to a user profile or feature group template that includes a universal line template, universal device template."

" In Cisco Unified Serviceability, activate the Self-Provisioning IVR and CTI Manager services."

upvoted 1 times

usernamesarehard 2 years, 4 months ago

Selected Answer: AC

A and C is correct

upvoted 1 times

Piji 2 years, 4 months ago

Selected Answer: AC

A, C is correct Answer.

upvoted 2 times

AJBELL14 2 years, 5 months ago

Selected Answer: AC

A and C is correct

upvoted 2 times

aeiou5545 2 years, 7 months ago

I believe the answer is A & C. A primary extension is required, but a directory URI is not.

upvoted 1 times

Question #113 *Topic 1*

Which Cisco Collaboration Edge architecture product allows remote endpoints to leverage corporate on-premises Cisco Unified Communications infrastructure?

A. Cisco Umbrella

B. Cisco Unified Communications Mobile and Remote Access

C. Cisco VPN Client

D. Cisco Webex

Correct Answer: *B*

Reference:

https://www.cisco.com/c/en/us/td/docs/solutions/CVD/Collaboration/enterprise/12x/120/collbcvd/edge.html

Community vote distribution

B (100%)

sneff91 9 months, 4 weeks ago

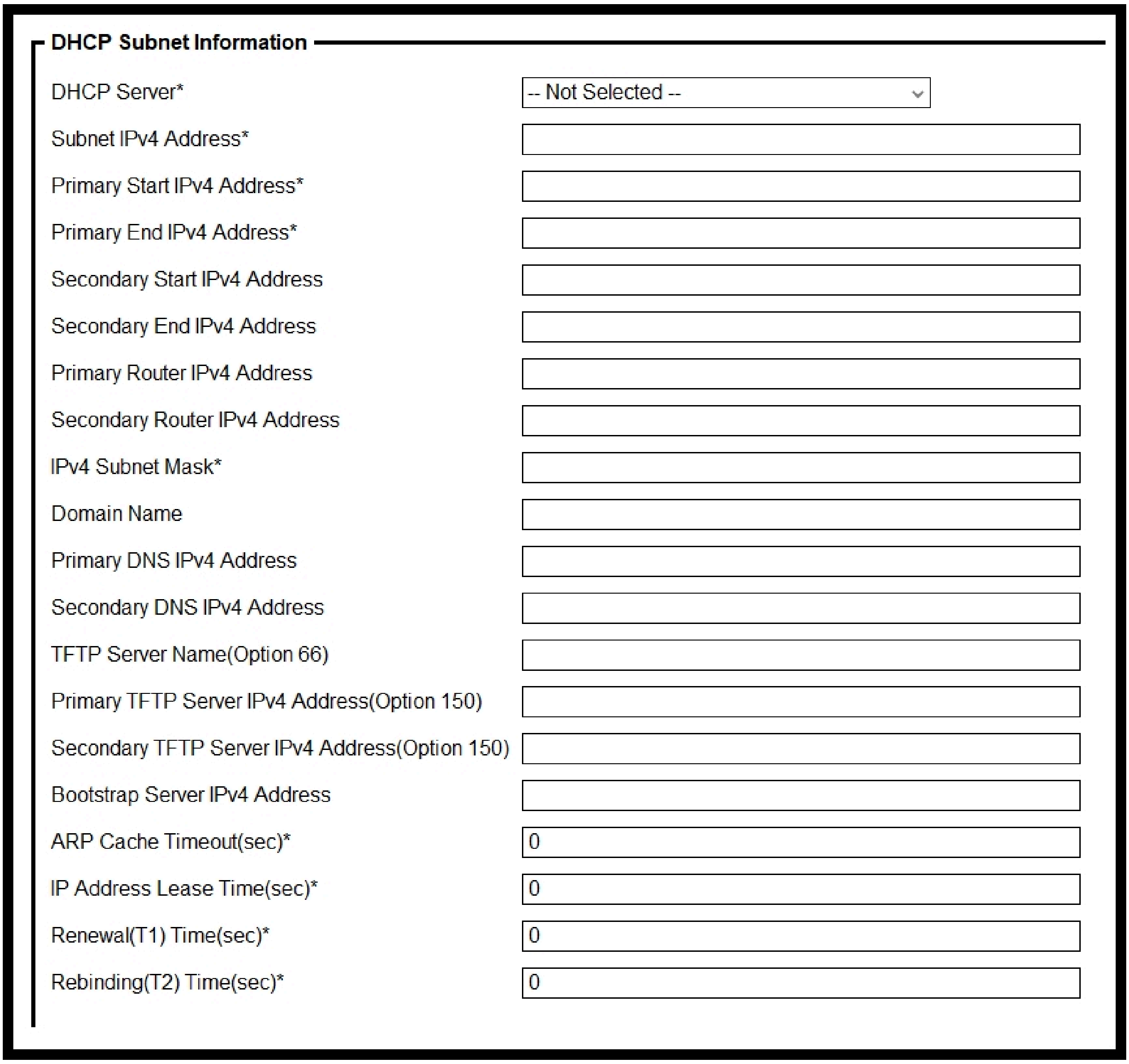
Selected Answer: B

I believe the answer that they are looking for is B, but MRA isn't a "product", it is a feature set PROVIDED by a product (Expressway). C feels less right as Cisco VPN client isn't a "Cisco Collaborate Edge" product. WebEx isn't really a "Cisco Collaborate Edge" product either.

Honestly, I feel like none of the answers are truly correct, but B feels the LEAST wrong.

upvoted 1 times

Question #114 Topic 1



DHCP Subnet Information

DHCP Server*	-- Not Selected --
Subnet IPv4 Address*	
Primary Start IPv4 Address*	
Primary End IPv4 Address*	
Secondary Start IPv4 Address	
Secondary End IPv4 Address	
Primary Router IPv4 Address	
Secondary Router IPv4 Address	
IPv4 Subnet Mask*	
Domain Name	
Primary DNS IPv4 Address	
Secondary DNS IPv4 Address	
TFTP Server Name(Option 66)	
Primary TFTP Server IPv4 Address(Option 150)	
Secondary TFTP Server IPv4 Address(Option 150)	
Bootstrap Server IPv4 Address	
ARP Cache Timeout(sec)*	0
IP Address Lease Time(sec)*	0
Renewal(T1) Time(sec)*	0
Rebinding(T2) Time(sec)*	0

Refer to the exhibit. An engineer must configure DHCP service on the Cisco UCM. In which field must the engineer configure the server IP to provide the configuration files to the endpoints?

A. Primary Router IPv4 Address

B. Primary TFTP Server IPv4 Address (Option 150)

C. Primary Start IPv4 Address

D. Bootstrap Server IPv4 Address

Correct Answer: *B*

Community vote distribution

B (100%)

Testme1235 10 months, 1 week ago

Selected Answer: B

The answer to the question is B. Primary TFTP Server IPv4 Address (Option 150).

In a Cisco Unified Communications Manager (UCM) environment, the server that provides the configuration files to the endpoints is the TFTP (Trivial File Transfer Protocol) server. To configure DHCP service on the UCM, the engineer must specify the IP address of the TFTP server in the "Primary TFTP Server IPv4 Address" field, which is also referred to as "Option 150" in the DHCP protocol.

Option A, "Primary Router IPv4 Address," is used to specify the IP address of the default gateway, which is used by the endpoint to reach other devices on the network.

Option C, "Primary Start IPv4 Address," is used to specify the starting IP address of the DHCP address pool.

Option D, "Bootstrap Server IPv4 Address," is used in some legacy protocols to specify the server that provides configuration files to the endpoint during the boot process, but it is not used in a modern Cisco UCM environment.
upvoted 3 times

Question #115 *Topic 1*

An administrator installs a new Cisco TelePresence video endpoint and receives this error: `AOR is not permitted by Allow/Deny list.` Which action should be taken to resolve this problem?

A. Correct the restriction policy settings.

B. Reboot the VCS server and attempt reregistration.

C. Upload a new policy in VCS.

D. Change the SIP trunk configuration.

Correct Answer: *A*
Reference:
https://www.cisco.com/c/en/us/td/docs/telepresence/infrastructure/articles/vcs_endpoint_registration_problems_kb_460.html

Community vote distribution
A (100%)

G0y0 3 months, 4 weeks ago
Selected Answer: A
The Expressway will also accept registration requests where the domain portion of the AOR is either the FQDN or the IP address of the Expressway. Whether or not the Expressway accepts a registration request depends on its registration control settings.
The first of these policies is the Registration Restriction Policy. This policy service controls which endpoints are allowed to register to the Expressway, and you can configure it to use an Allow List or a Deny List. There are two settings that you must configure when configuring the Registration Restriction Policy.
A registration restriction policy that uses either Allow Lists or Deny Lists or an external policy service to specify which aliases can and cannot register with the Expressway.
CCNP Collaboration Cloud and Edge Solutions CLCEI 300-820 Official Cert Guide
upvoted 1 times

wwisp3422112 7 months ago
Selected Answer: A
A - Know from experience
upvoted 1 times

vmandri 1 year ago
A
https://www.cisco.com/c/en/us/td/docs/telepresence/infrastructure/articles/vcs_endpoint_registration_problems_kb_460.html
upvoted 2 times

Question #116 *Topic 1*

```
dial-peer voice 2 voip
destination-pattern 5555678
sessiontarget ipv4:10.5.6.7
codec g729r8
fax protocol t38 ls-redundancy 0 hs-redundancy 0 fallback cisco
fax rate voice
```

Refer to the exhibit. An administrator configures fax dial-peers on a Cisco IOS gateway and finds that faxes are not working correctly. Which change should be made to resolve this issue?

A. codec g729br81

B. codec g723ar63

C. codec g726r32

D. codec g711ulaw

Correct Answer: *D*

Community vote distribution

D (100%)

Omitted Highly Voted 7 months, 2 weeks ago

Selected Answer: D

Don't use g729 for a fax machine. G.711ulaw or G.711alaw

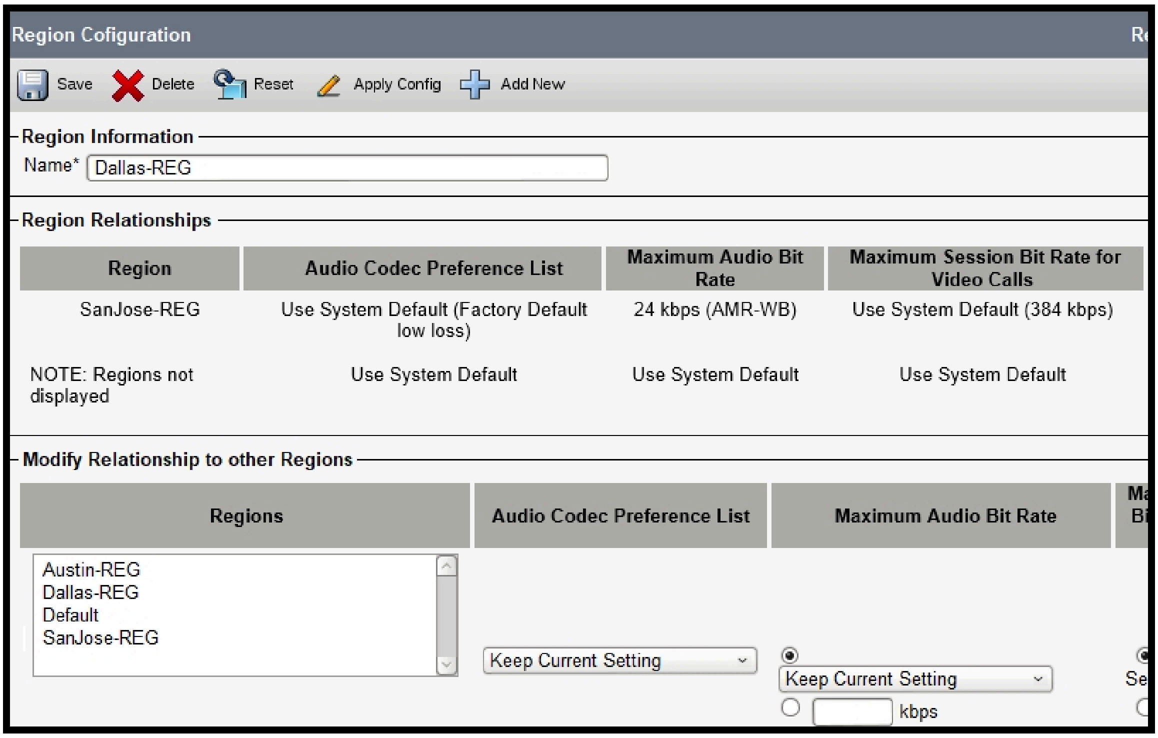
upvoted 5 times

G0y0 Most Recent 3 months, 4 weeks ago

Selected Answer: D

For any reason, Fax Relay is not working neither protocol t38 nor protocol cisco, and falling to passthrough. The only two options are related to change the codec, in this case G.711 is more secure to this works.

upvoted 1 times



Refer to the exhibit. Which codec should an engineer select for a call made between `Dallas-REG` & `Austin-REG`?

A. G.729

B. G.711

C. MP4A-LATM

D. OPUS

Correct Answer: *B*

Community vote distribution

B (63%) A (38%)

mcbesy 2 months, 1 week ago

Selected Answer: A

Table 7-1: For Region Configuration Settings, Default Codec with Other Regions

From the drop-down list box, choose a default codec to use for calls between this region and other regions. Due to bandwidth constraints at most remote-site deployments, the recommended default codec setting between a new region and existing regions is G.729.

https://www.cisco.com/en/US/docs/voice_ip_comm/cucm/admin/4_2_1/ccmcfg/b02regio.html

upvoted 1 times

Gary1968 5 months, 1 week ago

Selected Answer: B

The default audio codec for all calls through Cisco CallManager is G.711.

https://www.cisco.com/en/US/docs/voice_ip_comm/cucm/admin/4_2_1/ccmcfg/b02regio.html

upvoted 1 times

b3532e4 9 months, 1 week ago

Based on the exhibit, there is no explicit codec selection between Dallas-REG and Austin-REG, and the setting for both Audio Codec Preference List and Maximum Audio Bit Rate is left to "Keep Current Setting

Given that no specific codec has been pre-configured between these regions, it is reasonable to assume that the system default codec will be used. In most Cisco systems, G.711 is the default codec unless another codec is explicitly configured.

Therefore, the correct answer is:

B. G.711.

upvoted 1 times

frankieasantiago 1 year, 1 month ago

Selected Answer: B

Region Configuration

Note The default audio codec for all calls through Cisco CallManager is G. 711. If you do not plan to use any other audio codec, you do not need to use regions.

https://www.cisco.com/en/US/docs/voice_ip_comm/cucm/admin/4_2_1/ccmcfg/b02regio.html

upvoted 2 times

55c7877 1 year, 1 month ago

Selected Answer: A

This parameter specifies the default maximum audio bit rate for each call between a particular region and another region (interregion) when the Use System Default option is selected as the Max Audio Bit Rate in the Region Configuration window for the region's relationship with the other region. For example, assume there are two regions, one named Chicago and the other named Dallas. If Chicago has the Use System Default option set for the Max Audio Bit Rate for the relationship with Dallas, and the value in this parameter is set to 16 kbps (iLBC, G.728), then each call between devices in the Chicago region and devices in the Dallas region can use supported 16 kbps codecs like G.728, or lower bit rate codecs like the 8 kbps G.729, but not higher bit rate codecs like the 64 kbps G.722.
This is a required field.
Default: 8 kbps (G.729)

upvoted 2 times

pasangawa 1 year, 1 month ago

Selected Answer: B

it will use the default since it was not defined.
https://www.cisco.com/en/US/docs/voice_ip_comm/cucm/admin/4_2_1/ccmcfg/b02regio.html
the default audio codec for all calls through Cisco CallManager is G.711. If you do not plan to use any other audio codec, you do not need to use regions.

upvoted 1 times

pasangawa 1 year, 1 month ago

changing answe to A. G729 since it's the default max audio bit rate

upvoted 1 times

sneff91 1 year, 3 months ago

Selected Answer: A

It would be A - G.729. As It doesn't have an explicit inter-region setting to Austin-REG, it uses the System Default. The System Default list is referenced under Service Parameters, and the default Audio Codec Preference List specified is "Factory Default low loss". While G.711, MP4A-LATM, and OPUS are all higher in that list than G.729, they can't be used as the Dallas-REG region setting also use the System Default for the Maximum Audio Bit Rate, which references the "Default Interregion Max Audio Bit Rate" in Service Parameters. The system system value for that parameter is 8kbps, so only G.729 can be used.

upvoted 2 times

lanlord7 1 year, 8 months ago

Answer should be OPUS. Max bitrate is not defined between Dallas and Austin. Therefore, System Default of Inter-region Bitrate will apply which is 8kbps by default. As OPUS is a variable bitrate codec that can range from 6-150kbps, it meets the maximum bitrate requirement. As it's higher in the codec preference list by default, it should be chosen.

upvoted 3 times

spag22500 2 years, 6 months ago

A: G.729

=> Default Interregion Max Audio Bit Rate = 8 kbps (G.729)

Default Interregion Max Audio Bit Rate:
This parameter specifies the default maximum audio bit rate for each call between a particular region and another region (interregion) when the Use System Default option is selected as the Max Audio Bit Rate in the Region Configuration window for the region's relationship with the other region. For example, assume there are two regions, one named Chicago and the other named Dallas. If Chicago has the Use System Default option set for the Max Audio Bit Rate for the relationship with Dallas, and the value in this parameter is set to 16 kbps (iLBC, G.728), then each call between devices in the Chicago region and devices in the Dallas region can use supported 16 kbps codecs like G.728, or lower bit rate codecs like the 8 kbps G.729, but not higher bit rate codecs like the 64 kbps G.722.

This is a required field.
Default: 8 kbps (G.729)
upvoted 2 times

Mert_kerna 2 years, 6 months ago

Selected Answer: B

I believe B is the answer - G.711.
The question is asking what codec should be used between Dallas and Austin.
The Region configuration in this case only covers the relationship between Dallas and San Jose, which is set with a max bitrate of 24kbps.
All other regions in this case are set to "Use System Default" and aren't configured under this region configuration.
-
I also believe we shouldn't assume connection to the other regions requires bandwidth preservation for audio calls, because it's not specified in the question.

If the region configuration showed that the Austin region also required bandwidth preservation, then there wouldn't be an exact correct answer.

If you'd choose opus, which can be set down to 6kbps, why wouldn't you just use choose G.729, which has a bitrate of 8kbps?
-
Because we want to keep quality as optimal as possible, while not exceeding the Maximum audio bitrate for the region in question, calls between Dallas and Austin will be using G.711 by default, and that's how it should stay.
upvoted 2 times

Mert_kerna 2 years, 6 months ago

And if we're being precise, if a call was made to San Jose from Dallas, it would use OPUS or AMR-WB, with opus being higher in the audio codec preference list by default.
upvoted 1 times

wwisp3422112 2 years, 7 months ago

D: Opus
upvoted 1 times

Question #118 *Topic 1*

```
May  5 02:37:18.672: //-1/xxxxxxxxxxxx/SIP/Msg/ccsipDisplayMsg:
Sent:
SIP/2.0 200 OK
Via: SIP/2.0/TCP 10.8.140.6:5060;branch=z9hG4bK129bbd9bad
From: <sip:10.8.140.6>;tag=309206520
To: <sip:10.8.140.23>;tag=12237E50-1901
Date: Fri, 05 May 2017 02:37:18 GMT
Call-ID: c0710880-90b1e55e-d2d3-68c080a@10.8.140.6
Server: Cisco-SIPGateway/IOS-15.4.3.S4
CSeq: 101 OPTIONS
Allow: INVITE, OPTIONS, BYE, CANCEL, ACK, PRACK, UPDATE, REFER, SUBSCRIBE, NOTIFY,
INFO, REGISTER
Allow-Events: telephone-event
Accept: application/sdp
Supported: 100rel,timer,resource-priority,replaces,sdp-anat
Content-Type: application/sdp
Content-Length: 369

v=0
o=CiscoSystemsSIP-GW-UserAgent 6414 4717 IN IP4 10.8.140.23
s=SIP Call
c=IN IP4 10.8.140.23
t=0 0
m=audio 0 RTP/AVP 18 0 8 4 15
c=IN IP4 10.8.140.23
m=image 0 udptl t38
c=IN IP4 10.8.140.23
a=T38FaxVersion:0
a=T38MaxBitRate:9600
a=T38FaxRateManagement:transferredTCF
a=T38FaxMaxBuffer:200
a=T38FaxMaxDatagram:320
a=T38FaxUdpEC:t38UDPRedundancy
```

Refer to the exhibit. A customer wants the SIP 200 OK shown to advertise codecs in the following order:

✑ G.729

✑ G.711u

✑ G.711a

✑ G.723

G.728 -

.

After correcting the codec preferences, what should the audio payload show in the SIP Traces?

A. m=audio 0 RTP/AVP 0 8 18 4 15

B. m=audio 0 RTP/AVP 4 0 8 18 15

C. m=audio 0 RTP/AVP 0 18 8 4 15

D. m=audio 0 RTP/AVP 18 0 8 4 15

Correct Answer: *D*

Community vote distribution

D (100%)

Omitted Highly Voted 7 months, 2 weeks ago

Selected Answer: D

You could probably get thru the test just knowing the payload IDs for G.711u being 0 and G.729 being 18.

G.729 = 18

G.711u= 0

G.711a = 8

G.723 = 4

G.728 = 15

https://datatracker.ietf.org/doc/html/rfc3551#page-32

Question #119 *Topic 1*

A network administrator with ID0123456789 has determined that a WAN link between two Cisco UCM clusters supports only 1 Mbps of bandwidth for voice traffic.
How many calls does this link support if G.711 as the audio codec is used?

A. 12

B. 13

C. 15

D. 16

Correct Answer: *A*
Reference:
https://getvoip.com/blog/2013/05/31/voip-codecs-bandwidth-your-call-quality/

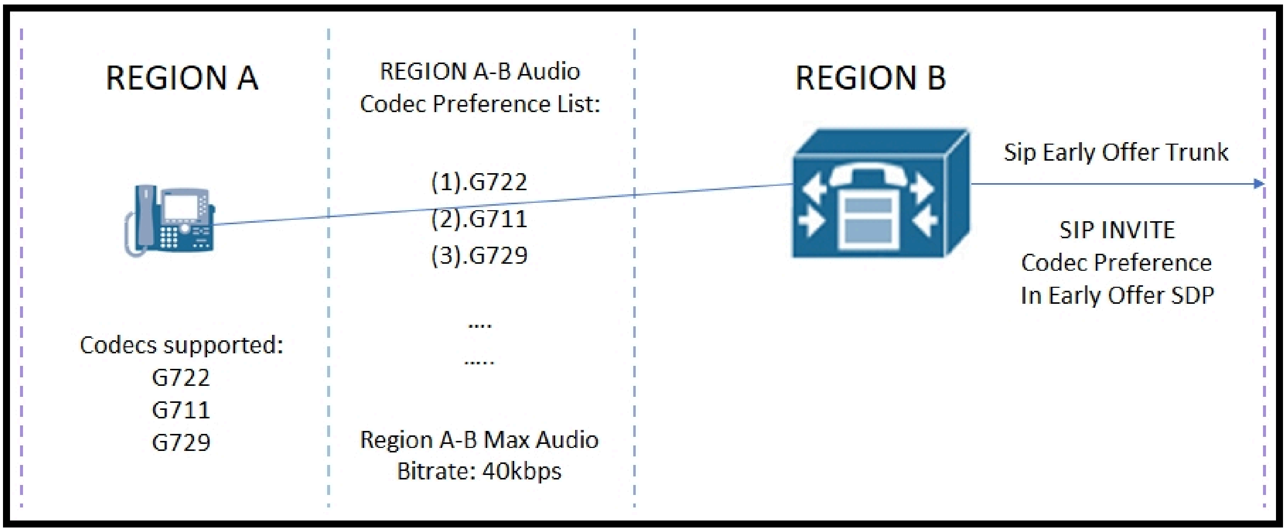
Gary1968 6 months, 2 weeks ago
Selected Answer: A
1024\87.2=11.74. A would be the closest match
upvoted 1 times

Omitted 1 year, 7 months ago
G.711 bandwidth per call should be 87.2 Kbps therefore i'd think the answer would be 11? what am i doing wrong?
upvoted 2 times

Piji 1 year, 4 months ago
I think if you round it 1000/80=12.5, but if you want to be accurate, 1024x1024/87.2=12.036 which will be the same thing, that is closer to 12.
upvoted 4 times

WeNt48 10 months ago
This is CAC question. Le'ts say we use 1 Mbps that is 1024. CUCM for g711 reduces 84 kbps from remaining bandwidth that was allocated. 1024/84 gives 12,19. This means CAC wouldn't allow 13th call to be placed, and options such as AAR could be triggered.
upvoted 3 times

Question #120 *Topic 1*



Refer to the exhibit. In this Cisco UCM setup configured for EARLY OFFER, what is the codec preference line in the initial SIP INVITE SDP?

A. m=audio 16444 RTP/AVP 9 0 8 18 101

B. m=audio 16444 RTP/AVP 0 8 18 101

C. m=audio 16444 RTP/AVP 9 18 101

D. m=audio 16444 RTP/AVP 18 101

Correct Answer: *D*

Community vote distribution

D (100%)

Slushed Highly Voted 2 years, 8 months ago

Selected Answer: D

D is correct. Since the Max Bitrate is only 40kbps, G722 (rtpmap:9) only goes down to 48kbps and G711 (rtpmap:0 ulaw/8 alaw) is 64kbps leaving only G729 (rtpmap:18).

upvoted 8 times

Iva9693 Most Recent 10 months, 1 week ago

A is correct

upvoted 1 times

Kabimas66 9 months, 2 weeks ago

Slushed is right. On this question you have a limitation of 40K for BW. G722 can use only 64k, 56K or 48K. So Answer is D

upvoted 3 times

Question #121

Topic 1

What is required for Cisco UCM to accept SIP calls with a URI in the format of `sip:2001@cucmpub.cisco.com`?

A. Change the Destination Address to a Fully Qualified Domain Name on the SIP trunk.

B. Set the SIPS URI Handling to True in CallManager Service Parameters.

C. Define Cluster Fully Qualified Domain Name under Servers in Cisco UCM.

D. Define Cluster Fully Qualified Domain Name in Enterprise Parameters.

Correct Answer: *D*

Community vote distribution

D (100%)

Panda_man 11 months, 1 week ago

Selected Answer: D

Source : https://www.cisco.com/c/en/us/td/docs/voice_ip_comm/cucm/admin/12_0_1/systemConfig/cucm_b_system-configuration-guide-1201/cucm_b_system-configuration-guide-1201_chapter_011011.html

upvoted 3 times

Question #122 *Topic 1*

An administrator installed a Cisco Unified IP 8831 Conference Phone that is failing to register. Which two actions should be taken to troubleshoot the problem?
(Choose two.)

A. Verify that the switch port of the phone is enabled.

B. Check the RJ-65 cable.

C. Verify that the phone's network can access the option 150 server.

D. Verify that the RJ-11 cable is plugged into the PC port.

E. Disable HSRP on the access layer switch.

Correct Answer: *AC*

Community vote distribution

AC (80%) BC (20%)

CiscoSailor Highly Voted 1 year, 12 months ago

Selected Answer: AC

I think option A is referring to the switch port the phone is connected to. I agree it is worded poorly but I think this is a better answer than assuming RJ-65 is just a typo.

upvoted 5 times

G0y0 Most Recent 3 months, 4 weeks ago

Selected Answer: BC

I understand that the display phone shows "unregistered", at least it receives PoE.

A. phone 8831 has only one port named "network port", not "switch port". If the port in the switch was disabled, Ip phone had not PoE, right?

B. If it is a typo, It would be RJ45. Many times the cables degrade and have some pins in their connector with false contact and the connection is intermittent even if they receive POE correctly. Again. if this is a typo, B could be a acceptable answer, and correct answer could be B. and C.

However, if B. is not a typo, then correct answers could be A and C.

BC or AC? any idea?

upvoted 1 times

decdca7 7 months, 2 weeks ago

Selected Answer: BC

All others are not relative

upvoted 1 times

lpicardin 2 years, 3 months ago

Selected Answer: BC

What's the link with the switchport ?? Only answer can be BC according the RJ65=RJ45 ... lol

upvoted 2 times

G0y0 3 months, 4 weeks ago

and what if in the real exam you see Rj65? would you still choose B?

upvoted 1 times

bigdog22 2 years, 4 months ago

why would the switchport on the phone have anthing to do with it....typically this is used to brdige in a PC?

upvoted 1 times

G0y0 3 months, 4 weeks ago

the phone has just one network port and two microphone ports

upvoted 1 times

JOtac 2 years, 4 months ago

Connector RJ-65. I suppose that this is an error. the correct connector name would be RJ-45

upvoted 2 times

G0y0 3 months, 4 weeks ago

I wish so

upvoted 1 times

Question #123 *Topic 1*

A Cisco UCM administrator wants to enable the Self-Provisioning feature for end users. Which two prerequisites must be met first? (Choose two.)

A. End users must have a primary extension.

B. End users must be associated to a user profile or feature group template that includes a universal line template and universal device template.

C. End users must have a secondary extension.

D. End users must belong to Standard CCM Admin Users group, the Standard CCM End Users group, and the Standard CCM Self-Provisioning group.

E. Cisco Extended Functions service must be running.

Correct Answer: *AB*
Reference:
https://www.cisco.com/c/en/us/td/docs/voice_ip_comm/cucm/admin/10_0_1/ccmcfg/CUCM_BK_C95ABA82_00_admin-guide-100/CUCM_BK_C95ABA82_00_admin-guide-100_chapter_01101001.html

iExpo_91 9 months ago
A & B both correct link below
https://www.cisco.com/c/en/us/td/docs/voice_ip_comm/cucm/admin/12_5_1/systemConfig/cucm_b_system-configuration-guide-1251/cucm_b_system-configuration-guide-1251_chapter_01001000.html
upvoted 3 times

Question #124 *Topic 1*

An administrator executes the debug isdn q931 command while debugging a failed call. After a test call is placed, the logs return a disconnect cause code of 1.
What is the cause of this problem?

A. The dialed number is not assigned to an endpoint.

B. The destination number rejects the call.

C. The destination number is busy.

D. The media resource is unavailable.

Correct Answer: *A*

Community vote distribution

A (100%)

Stevon 3 weeks, 1 day ago

Selected Answer: A

A disconnect cause code of "1" indicates that the called number is unallocated or unassigned. This means the number is not currently in use and cannot be reached. It doesn't necessarily mean the number doesn't exist, but rather that it hasn't been assigned to a specific device or user.
Here's a more detailed explanation:
Unallocated/Unassigned:
The called number is not in the system's routing table or it's not assigned to any device or user.
No Route to Destination:
The number may be in a valid format, but there's no established path or route to reach it.
Example Scenarios:
The number is a new number that hasn't been assigned to an employee's phone or a device yet.
The number was previously assigned but has been reassigned or removed from the system.

upvoted 1 times

Panda_man 11 months, 1 week ago

Selected Answer: A

Source https://support.digium.com/community/s/article/ISDN-Disconnect-cause-codes

upvoted 4 times

Omitted 1 year, 7 months ago

Selected Answer: A

Cause code 1: Unallocated or unassigned number

upvoted 4 times

Question #125 *Topic 1*

Where in Cisco UCM are codec negotiations configured for endpoints?

A. under device profiles in device settings

B. in in-service parameters

C. in enterprise parameters

D. under regions using preference lists

Correct Answer: *D*

Community vote distribution

D (100%)

Slushed Highly Voted 3 years, 1 month ago

Selected Answer: D

The correct answer is D. You choose your Codec Preference List in the Region configuration when creating new relationships between Regions.

upvoted 9 times

G0y0 Most Recent 4 months, 1 week ago

Selected Answer: D

The list of codecs used for codec negotiation during call setup, is the subset of codecs supported by the device and those in the codec preference list, limited by the maximum audio bit rate for the region or region pair. Answer is D.

Cisco Collaboration System 12.x Solution Reference Network Designs SRND.

Chapter 6 Cisco Unified CM Trunks

upvoted 1 times

G0y0 4 months, 1 week ago

It is very tricky. Service Parameters only determines whether CUCM allows negotiation for all, some, or no devices. However, negotiations are carried in regions and codec preference list. No obstant I consider answer is D.

upvoted 1 times

b3532e4 11 months, 1 week ago

The correct answer is D

upvoted 1 times

Piji 2 years, 10 months ago

Selected Answer: D

The correct answer is D, and B can't be at all, because we don't have "in-service parameters", as in "service parameters" also you got options to enable or disable codec.

upvoted 3 times

Question #126 Topic 1

An engineer configures a Cisco Unified Border Element and must ensure that the codecs negotiated meet the ITSP requirements. The ITSP supports G.711ulaw and G.729 for audio and H.264 for video. The preferred voice codec is G.711. Which configuration meets this requirement?

A.

```
voice class codec 10
 codec preference 1 g711ulaw
 codec preference 2 g729r8
 video codec h264

dial-peer voice 101 voip
 session protocol sipv2
 destination e164-pattern-map 1
 voice-class codec 100
```

B.

```
voice class codec 10
 codec preference 1 g711ulaw
 codec preference 2 g729r8

dial-peer voice 101 voip
 session protocol sipv2
 destination e164-pattern-map 1
 voice-class codec 10
```

C.

```
voice class codec 10
 codec preference 1 g711ulaw
 codec preference 2 g729r8
 video codec h264

dial-peer voice 101 voip
 session protocol sipv2
 destination e164-pattern-map 1
 voice-class codec 10
```

D.

```
voice class codec 10
 codec preference 1 g729r8
 codec preference 2 g711ulaw
 video codec h264

dial-peer voice 101 voip
 session protocol sipv2
 destination e164-pattern-map 1
 voice-class codec 10
```

Correct Answer: *C*
Reference:
https://www.cisco.com/c/en/us/td/docs/ios-xml/ios/voice/cube/configuration/cube-book/cube-codec-basic.html

WilliamC 9 months, 4 weeks ago
Usage Guidelines
Use this command to specify one or more video codecs for a voice class.

Examples
The following example shows configuration for voice class codec 10 with two audio codec preferences and three video codec preferences:

voice class codec 10
codec preference 1 g711alaw
codec preference 2 g722
video codec h261
video codec h263
video codec h264
upvoted 1 times

Komy 1 year, 1 month ago
Answer is [C]
upvoted 1 times

Ol_Mykhailiuk 1 year, 10 months ago
Cisco Codec Options(IOS)
H(config-dspfarm-profile)#codec ?

g711alaw G.711 A Law 64000 bps

g711ulaw G.711 u Law 64000 bps

g722-64 G722r64

g723r53 G.723.1 5300 bps, Not supported on PVDM3

g723r63 G.723.1 6300 bps, Not supported on PVDM3

g729abr8 G.729ab 8000 bps

g729ar8 G.729a 8000 bps

g729br8 G.729b 8000 bps

g729r8 G.729 8000 bps

gsmamr-nb GSMAMR codec

ilbc ILBC codec

isac ISAC codec

pass-through Stream Pass Through

upvoted 2 times

[Removed] 1 year, 11 months ago

NO voice-class codec 100 and 10

upvoted 1 times

[Removed] 1 year, 11 months ago

A C is same.

upvoted 1 times

Komy 1 year, 2 months ago

No They are not (the voice class codec defined is 10 while the voice class code referenced in the dial peer is 100)

upvoted 1 times

Question #127 *Topic 1*

What is a characteristic of a SIP endpoint configured in Cisco UCM with "Use Trusted Relay Point" set to "On"?

A. It enables Cisco UCM to insert an MTP or transcoder designated as a TRP.

B. It enables the Use Trusted Relay Point setting from the associated common device configuration.

C. If TRP is allocated and MTP is also required for the endpoint, calls fail.

D. It creates a trust relationship with the called party.

Correct Answer: *A*

Community vote distribution

A (100%)

Omitted Highly Voted 2 years, 1 month ago

Selected Answer: A

A Trusted Relay Point (TRP) is an MTP or transcoder that Cisco Unified Communications Manager can insert into the media stream to act as a control point for call media. The TRP can provide further processing on the stream and can ensure that the stream follows a specific path.

upvoted 6 times

G0y0 Most Recent 3 months, 4 weeks ago

Selected Answer: A

A Trusted Relay Point (TRP) is a device that can be inserted into a media stream to act as a control point for that stream. It may be used to provide further processing on that stream or as a method to ensure that the stream follows a specific path. If the signaling and media take different paths, a UDP pinhole is not opened. The solution might be a TRP. Subscribers in each data center can invoke TRPs that provide anchoring of the media and ensure that the media streams flow through the appropriate firewall. The TRP provides an IP address that enables a specific host route for media that can ensure the same routing path as the call signaling.

Implementing Cisco IP Telephony and Video, Part 1 (CIPTV1) Foundation Learning Guide

upvoted 1 times

Riktov 8 months, 1 week ago

Answer is A

https://www.cisco.com/c/en/us/td/docs/voice_ip_comm/cucm/admin/12_0_1/systemConfig/cucm_b_system-configuration-guide-1201/cucm_b_system-configuration-guide-1201_chapter_0111011.html

upvoted 2 times

Question #128 *Topic 1*

Which behavior occurs when Cisco UCM has a CallManager group that consists of two subscribers?

A. Endpoints attempt to register with both subscribers in a load-balanced method.

B. Endpoints attempt to register with the bottom subscriber in the list.

C. Endpoints attempt to register with the top subscriber in the list.

D. If a subscriber is rebooted, endpoints deregister until the rebooted system is back in service.

Correct Answer: *C*

Currently there are no comments in this discussion, be the first to comment!

Question #129 *Topic 1*

A customer is deploying a SIP IOS gateway for a customer who requires that in-band DTMF relay is first priority and out-of-band DTMF relay is second priority.
Which IOS entry sets the required priority?

A. dtmf-relay rtp-nte sip-notify

B. dtmf-relay cisco-rtp

C. sip-notify dtmf-relay rtp-nte

D. dtmf-relay sip-kpml cisco-rtp

Correct Answer: *A*

MKZ Highly Voted 1 year ago

In bound - rtp-nte
OOB - sip-notify, sip-kpml, sip-info

should be A

upvoted 7 times

MKZ 1 year ago

SIP GW doesn't support - cisco-rtp (H323 GW only)

upvoted 2 times

Question #130 *Topic 1*

A DTMF mismatch is occurring between an MGCP gateway registered FXS port and a Cisco Unified Communications Manager SIP trunk. Which media resource can be leveraged to interwork this mismatch?

A. Conference Bridge

B. Trusted Relay Point

C. Media Termination Point

D. Annunciator

Correct Answer: *C*

Currently there are no comments in this discussion, be the first to comment!

Question #131 *Topic 1*

```
dial-peer voice 10 voip
         destination-pattern 1…
         session target ipv4:10.1.1.1
         no vad
```

Refer to the exhibit. An engineer configures a VoIP dial peer on a Cisco gateway. Which codec is used?

A. G.711ulaw

B. No codec is used (missing codec command).

C. G.711alaw

D. G.729r8

Correct Answer: *D*

Community vote distribution

D (100%)

Chispas Highly Voted 3 years ago

Should be D

upvoted 15 times

rishik 2 years, 11 months ago

Correct default codec is G.729

upvoted 6 times

enashash Highly Voted 2 years, 10 months ago

Default codec for POTS is g.711

Default codec for VOIP is g.729

https://www.cisco.com/c/en/us/td/docs/voice_ip_comm/cucme/admin/configuration/manual/cmeadm/cmebasic.html#concept_2A67E83FAC004CF5AC02

upvoted 14 times

Piji Most Recent 10 months, 2 weeks ago

Selected Answer: D

Correct answer is D.

upvoted 2 times

asdlfhqwoiefnwe 1 year, 5 months ago

Selected Answer: D

Because this is a VoIP dial-peer, the codec is by default G.729. POTS dial-peers would utilize G.711.

upvoted 3 times

virtu 2 years, 5 months ago

Configures an audio codec at the dial peer level.

g729r8, 20-byte payload is configured by default.

upvoted 2 times

ratbat 2 years, 7 months ago

G729 is default codec when not explicitly specified folks

upvoted 4 times

Grebec94 3 years ago

it does not have a codec specified

upvoted 1 times

Question #132 *Topic 1*

An engineer must configure an MGCP gateway and register it to Cisco Unified Communications Manager. Which prerequisite must be met before applying the gateway commands to enable MGCP?

A. The MGCP gateway and the Cisco Unified CM must be able to communicate aver ports 5060 and 5061.

B. Cisco Unified CM and the MGCP gateway must utilize the SIP OPTIONS ping feature to monitor status.

C. The MGCP gateway must have voice service VoIP configured.

D. The MGCP gateway and the Cisco Unified CM must be able to communicate over ports 2427, 2428, and 2727.

Correct Answer: *D*
Reference:
https://www.cisco.com/c/en/us/td/docs/ios/voice/cminterop/configuration/guide/12_4t/vc_12_4t_book/vc_ucm_mgcp_gw.html

gottalearnsometime Highly Voted 4 years, 8 months ago
This is the only place I can find mention of port 2727: https://community.cisco.com/t5/collaboration-voice-and-video/mgcp-gateway-integration-with-cucm-and-pstn-service-provider/ta-p/3116588

It appears D is correct.
upvoted 11 times

CollabGuy 4 years, 7 months ago
Thanks, I was having a hard time to find this.
I really hate these questions because there's simply too many protocols/ports to remember :)
upvoted 5 times

VG224 3 years, 9 months ago
Use process of elimination .. all other answers are wrong
upvoted 2 times

Griswald Highly Voted 4 years, 11 months ago
MGCP Fundamentals
The Media Gateway Control Protocol (MGCP) is defined by RFC 2705. MGCP is a master/slave protocol, where the Endpoint (Slave) is controlled by a Call Agent (Master) of some type. The entire control intelligence is controlled by a Master who instructs the endpoint what action to take once an event is detected. MGCP uses TCP port 2428 and UDP port 2427.
TCP port 2428 in MGCP is used to open a new socket with the Call Agent to determine if the connection can be established. Without this new socket, subsequent MGCP Messages cannot be exchanged. It is also used to Send/Receive Backhaul Messages between PRI Endpoints and the Call Agent it is registered to. Finally, TCP port 2428 is used to failover to backup Call Agents in the event a Primary Call Agent is unresponsive.UDP Port 2427 in MGCP is used for MGCP Messages exchanged between the Endpoints and the Call Agents.
Basic Flow
upvoted 6 times

b3532e4 Most Recent 9 months ago
Prerequisites: For the configuration to work, the following prerequisites must be met.

MGCP Gateway and the CUCM TFTP server must have IP connectivity between each other. It's important to ensure that TCP and UDP ports 2427, 2428 and 2727 can communicate between the MGCP gateway and CUCM TFTP server. If there is a firewall then ensure these ports are enabled.
MGCP gateway and endpoints are configured in CUCM. Configuration has been saved and the Reset have been applied.
Host name or fully qualified domain name (FQDN) of the Cisco IOS MGCP gateway much match the CUCM MGCP gateway configuration. If the router's running configuration file has the <ip domain-name> command in it, an FQDN must be used , example : router1.highpoint.com
upvoted 1 times

mmollura 3 years, 11 months ago
I think it's C
upvoted 1 times

Question #133 *Topic 1*

```
000193: Dec 5 14:35:31.054: ISDN Se0/1/0:15 Q921: User TX -> SABMEp sapi=0 tei=0
000193: Dec 5 14:35:32.054: ISDN Se0/1/0:15 Q921: User TX -> SABMEp sapi=0 tei=0
000193: Dec 5 14:35:33.054: ISDN Se0/1/0:15 Q921: User TX -> SABMEp sapi=0 tei=0
000193: Dec 5 14:35:34.054: ISDN Se0/1/0:15 Q921: User TX -> SABMEp sapi=0 tei=0
```

Refer to the exhibit. Given this "debug isdn q921" output, what is the problem with the PRI?

A. Layer 1 is down on the controller.

B. PRI does not have an IP address configured on the interface.

C. Nothing, the PRI is sending keepalives.

D. Layer 2 is down on the controller.

Correct Answer: *D*

Community vote distribution

D (100%)

Ol_Mykhailiuk Highly Voted 9 months, 1 week ago

Selected Answer: D

If Layer 2 is stable, the router and switch must begin to synchronize with each other. The Set Asynchronous Balanced Mode Extended (SABME) message appears on the display. This message indicates that Layer 2 tries to initialize with the other side. Either side can send the message and try to initialize with the other side. If the router receives the SABME message, it must send back an Unnumbered Acknowledge frame (UAf).
https://www.cisco.com/c/en/us/support/docs/wan/t1-e1-t3-e3/8131-T1-pri.html

upvoted 5 times

Stevon Most Recent 3 weeks, 1 day ago

Selected Answer: D

https://www.cisco.com/c/en/us/support/docs/wan/t1-e1-t3-e3/14167-E1-pri.html

upvoted 1 times

Question #134 *Topic 1*

An administrator is trying to change the default LINECODE for a voice ISDN T1 PRI. Which command makes this change?

A. linecode esf

B. linecode ami

C. linecode hdb3

D. linecode b8zs

Correct Answer: *D*

Community vote distribution

D (62%) B (38%)

Chispas Highly Voted 4 years, 6 months ago

should be B default is b8zs

upvoted 17 times

Brant Highly Voted 3 years, 3 months ago

D is correct. Question is changing from default (which is AMI) for T1 line. So the correct answer is b8zs. hdb3 is for E1.

https://www.cisco.com/c/en/us/td/docs/ios/voice/cminterop/configuration/guide/12_4t/vc_12_4t_book/vc_mgcp_t1cas.html

upvoted 9 times

Stevon Most Recent 3 weeks, 1 day ago

Selected Answer: D

https://www.cisco.com/E-Learning/bulk/public/tac/cim/cib/using_cisco_ios_software/cmdrefs/linecode.htm

upvoted 1 times

Bazant 5 months, 3 weeks ago

Selected Answer: B

Read and understand the question. Correct answer is B

As per the guide we configure following linecodes:

ami—Specifies Alternate Mark Inversion (AMI) as the linecode type. Valid for T1 and E1 controllers.

b8zs—Specifies binary 8-zero substitution (B8ZS) as the linecode type. Valid for T1 controller only. This is the default for T1 lines.

hdb3—Specifies high-density binary 3 (HDB3) as the linecode type. Valid for E1 controller only. This is the default for E1 lines.

https://www.cisco.com/c/dam/en/us/td/docs/routers/ncs4200/configuration/guide/tdm/tdm-cfg-t1-e1-ncs4200-book.html

A - no esf is framing not linecode

B - fine it's non default framing possible to configure on T1

C - no because hdb3 is for E1 only

D - no because b8zs is default for T1

upvoted 1 times

Ste1233 7 months, 1 week ago

Selected Answer: B

should be B default is b8zs

upvoted 1 times

Ste1233 7 months, 1 week ago

B8ZS is the default linecode for T1. and changing from defualt

upvoted 1 times

Komy 8 months ago

Selected Answer: B

This snippet was taken from the link below:

"ami—Specifies Alternate Mark Inversion (AMI) as the linecode type. Valid for T1 and E1 controllers.

b8zs—Specifies binary 8-zero substitution (B8ZS) as the linecode type. Valid for T1 controller only. This is the default for T1 lines.

hdb3—Specifies high-density binary 3 (HDB3) as the linecode type. Valid for E1 controller only. This is the default for E1 lines."

So basically, for T1: default is b8zs (so the anser to this question should be AMI as hdb3 is only valid for E1)

https://www.cisco.com/c/dam/en/us/td/docs/routers/ncs4200/configuration/guide/tdm/tdm-cfg-t1-e1-ncs4200-book.html#:~:text=B8ZS%20is%20the%20default%20linecode%20for%20T1.
upvoted 1 times

CiscoSailor 1 year, 5 months ago
Selected Answer: D
Per the CLCOR book AMI is default so correct answer is B8ZS.
upvoted 2 times

AnelyP 1 year, 6 months ago
Update its B - AMI
upvoted 1 times

AnelyP 1 year, 6 months ago
I have checked the same question on questions provided by friend who have already passed the exam and its A - AMI
upvoted 2 times

istellas 1 year, 6 months ago
Selected Answer: B
https://content.cisco.com/chapter.sjs?uri=/searchable/chapter/www.cisco.com/content/en/us/td/docs/switches/metro/me3600x_3800x/software/release/15-2_4_S/chassis/configuration/guide/3600x_24cxscg/sw_T1-E1.html.xml

Selects the linecode type.
ami—Specifies Alternate Mark Inversion (AMI) as the linecode type. Valid for T1 and E1 controllers.
b8zs—Specifies binary 8-zero substitution (B8ZS) as the linecode type. Valid for T1 controller only. This is the default for T1 lines.
hdb3—Specifies high-density binary 3 (HDB3) as the linecode type. Valid for E1 controller only. This is the default for E1 lines.
esf – framing (not linecode)

Because Q asks for changing the default linecode (for T1), correct answer is B
upvoted 2 times

JWMcInSC 1 year, 6 months ago
Per Cisco: t1—Specifies T1 connectivity of 1.536 Mbps. B8ZS is the default line code for T1.

If B8ZS is the default, then the linecode needs to be set to AMI manually. Therefore B is correct.
upvoted 2 times

Ahochau 1 year, 7 months ago
Selected Answer: B
https://www.cisco.com/c/dam/en/us/td/docs/routers/ncs4200/configuration/guide/tdm/tdm-cfg-t1-e1-ncs4200-book.html
Selects the linecode type.

ami—Specifies Alternate Mark Inversion (AMI) as the linecode type. Valid for T1 and E1 controllers.
b8zs—Specifies binary 8-zero substitution (B8ZS) as the linecode type. Valid for T1 controller only. This is the default for T1 lines.
hdb3—Specifies high-density binary 3 (HDB3) as the linecode type. Valid for E1 controller only. This is the default for E1 lines.
upvoted 1 times

Alan100 1 year, 9 months ago
Selected Answer: D
Answer is D. Default for T1s is AMI (Alternate Mark Inversion), Default for E1s is hdb3 (High Density Bipolar 3). b8zs (Bipolar with eight-zero substitution) is a modified AMI for North American States. https://en.wikipedia.org/wiki/Integrated_Services_Digital_Network. According to Cisco, AMI is already the default for T1s so to change it, the only other option is b8zs.
https://www.cisco.com/c/en/us/td/docs/ios/voice/cminterop/configuration/guide/12_4t/vc_12_4t_book/vc_mgcp_t1cas.html
upvoted 2 times

Panda_man 1 year, 11 months ago
ami: Alternate mark inversion (AMI), valid for T1 or E1 controllers. Default for T1 lines.

b8zs: B8ZS, valid for T1 controllers only.

hdb3: High-density bipolar 3 (hdb3), valid for E1 controllers only. Default for E1 lines
upvoted 1 times

spag22500 2 years ago
correct is D

The keywords and descriptions are as follows:
- ami: Alternate mark inversion (AMI), valid for T1 or E1 controllers. Default for T1 lines.
- b8zs: B8ZS, valid for T1 controllers only.
- hdb3: High-density bipolar 3 (hdb3), valid for E1 controllers only. Default for E1 lines.
upvoted 1 times

azizkasmir 2 years ago
B - B8ZS is the default linecode for T1.

Linecode {ami | b8zs hbd3 }
From: https://www.cisco.com/c/dam/en/us/td/docs/routers/ncs4200/configuration/guide/tdm/tdm-cfg-t1-e1-ncs4200-book.html
upvoted 2 times

Question #135 *Topic 1*

Due to provider requirements, outgoing calls from the Enterprise to the PSTN must start with channel 1.
Which ISDN command changes the channel selection on IOS to meet this requirement?

A. isdn bchan-number-order descending

B. isdn bchan-number-order ascending

C. isdn protocol-emulate network

D. isdn incoming-voice voice

Correct Answer: *B*

Community vote distribution

B (100%)

Stevon 3 weeks, 1 day ago

Selected Answer: B

https://community.cisco.com/t5/ip-telephony-and-phones/question-on-isdn-b-channel-order/td-p/2105642

upvoted 1 times

Alan100 9 months, 1 week ago

Selected Answer: B

https://www.cisco.com/c/en/us/td/docs/ios-xml/ios/voice/isdn/configuration/15-mt/vi-15-mt-book/vi-isdn-vi-cfg.html

upvoted 2 times

Question #136 *Topic 1*

```
ISDN Serial1:23 interface
         dsl 1, interface ISDN Switchtype =
primary-5ess
     Layer 1 Status:
         ACTIVE
     Layer 2 Status:
         TEI = 0, Ces = 1, SAPI = 0, State =
TEI_ASSIGNED
     Layer 3 Status:
         0 Active Layer 3 Call(s)
     Activated dsl 1 CCBs = 0
     The Free Channel Mask: 0x807FFFFF
     Total Allocated ISDN CCBs = 5
```

Refer to the exhibit. What causes of the PRI issue?

A. The cable is unplugged.

B. The clock source is incorrect.

C. The controller shut down.

D. The framing is configured incorrectly.

Correct Answer: *B*

Community vote distribution

D (50%) B (50%)

santiagof Highly Voted 2 years, 8 months ago

D is correct

upvoted 8 times

ASSH 2 years, 7 months ago

Can you explain why ?

upvoted 1 times

santiagof 2 years, 7 months ago

Yes, A and C, are wrong, you wont have layer 1 up if they are true.

and B clock problems, cant turn down the layer 2 service.

upvoted 5 times

MeowthL 2 years, 3 months ago

https://www.gkhan.in/isdn-t1-e1-troubleshooting-tei_assigned/#:~:text=ISDN%20Status%20as%20TEI_ASSIGNED%20meaning%20Which%20indicates%20that,2%20problem%20with%20the%20debu

upvoted 2 times

MeowthL 2 years, 3 months ago

Symptoms :- ISDN layer 1 is active and Layer 2 going up and down with error “TEI_ASSIGNED”

There could be following missing command under signaling interface at Telco end if they are using Cisco Gateway.

upvoted 1 times

G0y0 Most Recent 3 months, 3 weeks ago

Selected Answer: D

A. and C. do not have sense.

B, is talking about a possibility of a intermittent service, it could be swinging between up and down. These errors can be viewed in the show controllers T1, however, the Layer 2 can be viewed in state MULTIPLE_FRAME_ESTABLISHED even though these slips errors.

D. could be the best answer, simply on side can not understand the frames from the other side, mean while while one side is sending SABME's like crazy and not receiving anything from the other side that could be solved with a correct configuration. I do not know if the problem is the framing or a bad configuration in the Serial interface (isdn protocol-emulate command), because it is stupid to be guessing without without having debug isdn q921and show controller t1 outputs, however, the only reason about the problem is a configuration issue, and just the answer D shows us "configured incorrectly". Something is configured incorrectly, without a doubt.

upvoted 1 times

G0y0 3 months, 3 weeks ago

Furthermore, the show isdn status command displays a summary of the current status. Therefore, Layer 2 can bounce up and down even though it indicates a MULTIPLE_FRAME_ESTABLISHED state. It is needed to use the debug isdn q921 command to ensure that Layer 2 is stable. This situation can be leveraged by a incorrect clock, however, as we do not have debug isdn q921, we are forced to assume that clock is correct and no slip erros.

Please refer to (it is amazing this document):

Troubleshoot T1 PPRI

https://www.cisco.com/c/en/us/support/docs/wan/t1-e1-t3-e3/8131-T1-pri.pdf

upvoted 1 times

decdca7 7 months, 2 weeks ago

Selected Answer: D

D is correct, Farming needs to be fixed,

Slips are the result of B.

upvoted 2 times

decdca7 7 months, 2 weeks ago

Selected Answer: B

D is the answer, B would be right if you had slips

upvoted 1 times

decdca7 7 months, 2 weeks ago

Changed my mind it is D

upvoted 1 times

JoeC716 1 year, 1 month ago

Selected Answer: D

Read MeowthL's comment and go to the link, in there read about the TEI_ASSIGNED state in the Layer 2 status and it talks about BAD FRAMES

upvoted 2 times

dbayt 1 year, 8 months ago

Selected Answer: D

https://www.cisco.com/c/en/us/support/docs/wan/t1-e1-t3-e3/8131-T1-pri.html

upvoted 2 times

kitty73 1 year, 11 months ago

Selected Answer: D

https://community.cisco.com/t5/ip-telephony-and-phones/isdn-layer-2-stays-on-quot-tei-assigned-quot-state/td-p/2928002

upvoted 2 times

Piji 2 years, 10 months ago

Selected Answer: B

The correct answer should be B.

upvoted 3 times

Question #137 Topic 1

```
hostname GATEWAY
ccm-manager config
ccm-manager config server 192.168.1.100
ccm-manager mgcp

mgcp call-agent CCMSub1.domain.com 2427 service-type mgcp version 0.1
```

Refer to the exhibit. An engineer verifies the configuration of an MGCP gateway. The commands are already configured. Which command is necessary to enable
MGCP?

A. Device(config)# mgcp enable

B. Device(config)# ccm-manager enable

C. Device(config)# ccm-manager active

D. Device(config)# mgcp

Correct Answer: *D*
Reference:
https://www.cisco.com/c/en/us/support/docs/voice-unified-communications/unified-communications-manager-callmanager/42105-vg200-cfg.html

Panda_man 11 months, 1 week ago
D is correct; Reference https://www.youtube.com/watch?v=fM9fdwQLzxw
upvoted 1 times

inilham 1 year, 7 months ago
Reverance:

https://www.cisco.com/c/en/us/support/docs/voice/media-gateway-control-protocol-mgcp/214635-configure-and-troubleshoot-mgcp-gateways.html
upvoted 1 times

Question #138 Topic 1

```
23031952: Apr  9 17:43:21.203 EDT: ISDN Se0/1/0:23 Q931: Applying typeplan for sw-type 0xD is 0x2 0x1, Calling num 4085556100
23031953: Apr  9 17:43:21.203 EDT: ISDN Se0/1/0:23 Q931: Sending SETUP callref = 0x12BE callID = 0xA3F5 switch = primary-ni interface = User
23031954: Apr  9 17:43:21.203 EDT: ISDN Se0/1/0:23 Q931: TX -> SETUP pd = 8 callref = 0x12BE
        Bearer Capability i = 0x8090A2
                Standard = CCITT
                Transfer Capability = Speech
                Transfer Mode = Circuit
                Transfer Rate = 64 kbit/s
        Channel ID i = 0xA98393
                Exclusive, Channel 19
        Progress Ind i = 0x8183 - Origination address is non-ISDN
        Calling Party Number i = 0x2181, '4085556100'
                Plan:ISDN, Type:National
        Called Party Number i = 0x91, '011443075552222'
                Plan:ISDN, Type:International
23031956: Apr  9 17:43:21.279 EDT: ISDN Se0/1/0:23 Q931: RX <- CALL_PROC pd = 8 callref = 0x92BE
        Channel ID i = 0xA98393
                Exclusive, Channel 19
23031957: Apr  9 17:43:21.283 EDT: ISDN Se0/1/0:23 Q931: RX <- PROGRESS pd = 8 callref = 0x92BE
        Cause i = 0x829F - Normal, unspecified
        Progress Ind i = 0x8488 - In-band info or appropriate now available
23031981: Apr  9 17:43:46.802 EDT: ISDN Se0/1/0:23 Q931: TX -> DISCONNECT pd = 8 callref = 0x12BE
        Cause i = 0x8090 - Normal call clearing
23031982: Apr  9 17:43:46.822 EDT: ISDN Se0/1/0:23 Q931: RX <- RELEASE pd = 8 callref = 0x92BE
23031983: Apr  9 17:43:46.822 EDT: ISDN Se0/1/0:23 Q931: TX -> RELEASE_COMP pd = 8 callref = 0x12BE
```

Refer to the exhibit. A call to an international number has failed. Which action corrects this problem?

A. Add the bearer-cap speech command to the voice port.

B. Strip the leading 011 from the called party number.

C. Assign a transcoder to the MRGL of the gateway.

D. Add the isdn switch-type primary-dms100 command to the serial interface.

Correct Answer: *B*

Community vote distribution

B (100%)

remmie_78 Highly Voted 2 years ago

Should be Answer B; Bearer-cap 0x8090A2 is already speech. Dialed nr starting 011 seems to me not an international number. Source; https://www.cisco.com/c/en/us/support/docs/voice/h323/14006-h323-isdn-callfailure.html

upvoted 6 times

Collabhunter 2 years ago

yeap, I'm going to B also. From eliminating other answers... bearer-cap as speech is already there as you said (0x8090A2)... no need for transcoder or changing switch-type as we got "normal call clearing" on disconnect cause, So probably we got "wrong number dialer from ITSP", then removing 011 will do the work to correct digits to pstn

upvoted 2 times

timmyz 1 year, 10 months ago

011 is the US international dial code. so that has to be ther to place the call even with the ITSP

upvoted 1 times

timmyz 1 year, 10 months ago

og but wait the ISDN PLAN is international so the leading 011 is not needed?

upvoted 1 times

VG224 1 year, 9 months ago

Not really .. specially in this case when ISDN call type is set to International

upvoted 1 times

DaKenjee Most Recent 7 months, 1 week ago

Selected Answer: B

From answers i would ignore A,C,D

A is not suiting, is already speach: 0x8090A2 Speech

https://www.cisco.com/c/en/us/support/docs/voice/h323/14006-h323-isdn-callfailure.html

C is not required, IDSN Debug is not showing codec issues on given cause codes

D is about Configuring the Switch Type, which i do not see as an issue, when focusing on given cause codes suspecious is 0x929F, which indicates issues in other related searches to signaled called number

I checked on our customer in USA, we send it with leading 011 + plan:unknown

I assume it is provider depending, but with plan:international it might be necessary stripping 011

upvoted 2 times

Sal007 7 months, 3 weeks ago

Selected Answer: B

B is the correct answer.

upvoted 2 times

Sal007 7 months, 3 weeks ago

Selected Answer: B

The answer is B

upvoted 1 times

[Removed] 1 year, 1 month ago

I am no so sure if it is B or C. If you see the stamps between the PROGRESS msg and the DISCONNECT msg, they are 17:43:21 and 17:43:46, so 25 seconds in difference doing ringing the called-phone in one side, and getting ringback the calling-phone in the other side. After 25 seconds, the caller-phone hangs-on, furthermore I do not see a problem of call routing or some unallocated number or stuff like that, just a normal call clearing, is really the problem the "011" (even when the hanging-on side was the GW, no the CO)?, if it were the "011" issue, then it wouldn't have ringing and ringbacking for 25 seconds. I am very confused.

By the way, indeed, A. and D. are absolutely discarded.

upvoted 1 times

[Removed] 1 year, 1 month ago

let us to consider the caller as the callref 0x12BE, and the called as the callref 0x92BE, and the caller was who hanged up after 25 seconds hearing ringback. If the called-side was who answered the call, It could has appeared the CONNECTED message, and then a DISCONNECT from the calling side for some issue of resources. In order of that, it seem that there was not any failed call, then may be I would be a bit more agree with B than C.

upvoted 1 times

[Removed] 1 year, 1 month ago

Another think to suspect is that PROGRESS, if it is tunneling a SIP message in a B2BUA configuration in the Gateway, is is similar a 183 Session in Progress (early media). There we do not know whats is happening in the caller side phone, if he is receiving ringback or just silence. Is the caller just hear silence, nothing, he might hang up impatiently. The same stuff if it is interworking H.323 with ISDN abut early media, we do not know. Whit the information provided by the question, it seems there is not failed call. And the solution would be a best practice, no such a fix, removing the "011", it is other reason I would be a bit more agree with B than C.

upvoted 1 times

nuno_paulo 1 year, 6 months ago

Selected Answer: B

A quick debug isdn q931 helps determine the issue:

A normal call will have:

Bearer Capability i = 0x8090A2

A failed call wil have the following:

Bearer Capability i = 0x8890

https://it-learn.io/2014/05/01/sip-devices-unable-to-stablish-pstn-calls/

upvoted 2 times

BhaiKyare 1 year, 10 months ago

If 011 is sent out we should get fast busy right ? Then why the question is not mentioning that but just say the call is failed ? So Answer A could be right

upvoted 2 times

VG224 1 year, 9 months ago

Its B . You do not need 011 in front when call type is set to International

upvoted 1 times

Question #139 *Topic 1*

```
05:50:14.102: ISDN BR0/1/1 Q921: User TX -> IDREQ ri=21653 ai=127
05:50:14.134: ISDN BR0/1/1 Q921: User RX <- SABMEp sapi=0 tei=0
05:50:14.150: ISDN BR0/1/1 Q921: User TX -> IDREQ ri=19004 ai=127
05:50:14.165: ISDN BR0/1/1 Q921 User RX <- SABMEp sapi=0 tei=0
```

Refer to the exhibit. A customer submits this debug output, captured on a Cisco IOS router. Assuming that an MGCP gateway is configured with an ISDN BRI interface, which BRI changes resolve the issue?

A. interface BRI0/1/1 no ip address isdn switch-type basic-net3 isdn incoming-voice voice isdn send-alerting isdn static-tei 0

B. interface BRI0/1/0 no ip address isdn switch-type basic-net3 isdn point-to-multipoint-setup isdn incoming-voice voice isdn send-alerting isdn static-tei 0

C. interface BRI0/1/1 no ip address isdn switch-type basic-net3 isdn point-to-point-setup isdn incoming-voice voice isdn send-alerting isdn static-tei 0

D. interface BRI0/1/1 no ip address isdn switch-type basic-net3 isdn point-to-multipoint-setup isdn incoming-voice voice isdn send-alerting isdn static-tei 0

Correct Answer: *C*

Community vote distribution

C (100%)

DaKenjee Highly Voted 7 months, 1 week ago

Selected Answer: C

B - is wrong by port and A has no isdn point-to-XX command
The line looks bad to read, but it only differs in
-> isdn point-to-point-setup
-> isdn point-to-multipoint-setup

SABME is the first message sent during a Layer 2 connection sequence.
Its basic purpose is to reset the link and prepare it for a new connection.
SABMEs must be acknowledged by a UA.

There is no UA Acknowledgment
Generally, multiple circuits with a single pilot number are point-to-point.

Router(config-if)# isdn point-to-point-setup
(Optional) Configures the ISDN port to send SETUP messages on the static TEI (point-to-point link).

upvoted 6 times

[Removed] Most Recent 7 months, 2 weeks ago

it is C - a tei is the "ISDN address" thats only needed when you are using a hub system with multiple endpoints. if you only have one endpoint connected then you are using tei=0, and are on a point-to-point connected

upvoted 4 times

Omitted 1 year, 1 month ago

Selected Answer: C

I think this is C. When you configure a static-tei then it should be point to point. Not point to multipoint.

upvoted 3 times

Question #140 *Topic 1*

Which command must be defined before an administrator has the ability to change the linecode value on an ISDN T1 PRI in slot 0/2 on an IOS-XE gateway?

A. voice-port 0/2/0:23

B. isdn incoming-voice voice

C. card type t1 0 2

D. pri-group timeslots 1-24

Correct Answer: *C*

Community vote distribution

C (100%)

Slushed Highly Voted 3 years, 1 month ago

Selected Answer: C

The correct answer is C. You cannot define a controller, and thus voice-ports or pri-timeslots, until the card-type is set.

upvoted 11 times

AbdurrahmanBNC Most Recent 11 months, 3 weeks ago

Do you want people choice false option? Who do answer to this question?

upvoted 1 times

OSJAY 7 months, 3 weeks ago

So, what is your answer?

upvoted 1 times

AJBELL14 2 years, 11 months ago

Selected Answer: C

C - Card Type t1 0 2 is the answer

upvoted 3 times

Question #141 *Topic 1*

```
voice translation-rule 1
 rule 1 /^[2-9]......$/ /\0/ type any subscriber
 rule 2 /^1[2-9]..[2-9]......$/ /\0/ type any subscriber
```

Refer to the exhibit. What is the result of applying these two rules to a voice translation profile for use with an ISDN T1 PRI on a Cisco Voice Gateway?

A. The leading Plus is stripped from the numeric phone number

B. The ISDN Type is modified to the administrator's defined value

C. Any zero is stripped from the numeric phone number

D. The ISDN Plan is modified to the administrators defined value

Correct Answer: *B*

Community vote distribution

B (100%)

G0y0 4 months ago

Selected Answer: B

The term "/\0/" is telling "do not modify anything of the original number and pass it". The only thing that is being changed is the TON. So Answer is B.

upvoted 1 times

ciscogeek 7 months, 3 weeks ago

Selected Answer: B

This example changes the number type and plan.

voice translation-rule 8
rule 1 /^2\(...$\)/ /01779345\1/ type unknown national plan unknown isdn
This rule matches any four-digit number that starts with "2". The rule removes the "2", adds the number "01779345" as a prefix, and sets the plan to "isdn" and the type to "national".

https://www.cisco.com/c/en/us/support/docs/voice/call-routing-dial-plans/61083-voice-transla-rules.html#type_plan

upvoted 4 times

OSCARX88 1 year ago

The correct is A

upvoted 1 times

Slushed 10 months, 1 week ago

B is actually the right answer.

Rule 1 will match any 7-digit number leading with a 2-9 (a local number) and rule 2 will match any number leading with a 1, 2-9, any any, 2-9, any any any any any any" which is the format for a long-distance number.
If you look at the documentation linked below, you can alter the TYPE or the PLAN of a number. The interpretation after the "replace digits" syntax is "match *type (any)* replace *type (subscriber)*.
https://www.cisco.com/c/en/us/support/docs/voice/call-routing-dial-plans/61083-voice-transla-rules.html#type_plan

upvoted 3 times

Question #142 Topic 1

An administrator needs to strip the leading 9 from outbound calls on an IOS Voice Gateway, and ensure that the system handles 911 emergency calls. Which configuration is needed to accomplish this task?

A.

```
voice translation-rule 1
 rule 1 /^9\(.*\)/ /\1/
 rule 2 /9?911/ /911/
```

B.

```
voice translation-rule 1
 rule 1 /9?911/ /911/
 rule 2 /^9\(.*\)/ /\0/
```

C.

```
voice translation-rule 1
 rule 1 /9?911/ /911/
 rule 2 /^9\(.*\)/ /&/
```

D.

```
voice translation-rule 1
 rule 1 /9?911/ /911/
 rule 2 /^9\(.*\)/ /\1/
```

Correct Answer: *B*

DaKenjee Highly Voted 1 year, 1 month ago

Answer D

A - incorrect- rule2 is to late, already stripped

B - rule2 wrong \1 is for argument (.*\), not 0

C - rule2 wrong, no argument /&/ is known to me or by guide

router#sh run | se voice translation-rule 20

voice translation-rule 20

rule 1 /9911/ /911/

rule 2 /^9\(.*\)/ /\1/

router#test voice translation-rule 20 95554444333

Matched with rule 2

Original number: 95554444333 Translated number: 5554444333

router#test voice translation-rule 20 9911

Matched with rule 1

Original number: 9911 Translated number: 911

upvoted 16 times

Slushed Highly Voted 1 year, 8 months ago

D is correct. The first rule will match 9911 and send it as 911. If that rule is not matched, rule 2 will match "9anything" and send it without the leading 9.

upvoted 8 times

G0y0 4 months ago

Actually the first rule can match 9911 and 911, since the "?" means occurrences from zero to any times. However, I could not agree more with you that D is correct.

upvoted 1 times

arinpas Most Recent 8 months, 2 weeks ago

Agree.. D is correct

upvoted 3 times

RdTx 1 year ago

Agree.. D is correct

upvoted 4 times

Omitted 1 year, 7 months ago

D is correct. But I'd be surprised if this question is still included given that none of these are compliant with Kari's law

upvoted 5 times

plazaliberdad 1 year, 8 months ago

D is correct

upvoted 3 times

Question #143 *Topic 1*

An engineer must configure a Cisco ISR 4000 as an MGCP gateway to download its MGCP-specific configuration from Cisco UCM. Which Cisco IOS configuration snippet accomplishes this task?

A. ISR4K(config)# ip tftp source-interface GigabitEthernet 0/0/1 ISR4K(config)# mgcp call-agent 10.1.2.3 2428 service-type mgcp version 0.1 ISR4K(config)# ccm-manager config server 10.1.2.3 ISR4K(config)# ccm-manager config ISR4K(config)# ccm-manager mgcp ISR4K(config)# mgcp

B. ISR4K(config)# ip tftp source-interface GigabitEthernet 0/0/1 ISR4K(config)# mgcp call-agent 10.1.2.3 2427 service-type mgcp version 0.1 ISR4K(config)# ccm-manager config server 10.1.2.3 ISR4K(config)# ccm-manager config download ISR4K(config)# ccm-manager mgcp ISR4K(config)# mgcp

C. ISR4K(config)# ip tftp source-interface GigabitEthernet 0/0/1 ISR4K(config)# mgcp call-agent 10.1.2.3 2427 service-type mgcp version 11.5.1 ISR4K(config)# ccm-manager config server 10.1.2.3 ISR4K(config)# ccm-manager config ISR4K(config)# ccm-manager mgcp ISR4K(config)# mgcp

D. ISR4K(config)# ip tftp source-interface GigabitEthernet 0/0/1 ISR4K(config)# mgcp call-agent 10.1.2.3 2427 service-type mgcp version 0.1 ISR4K(config)# ccm-manager config server 10.1.2.3 ISR4K(config)# ccm-manager config ISR4K(config)# ccm-manager mgcp ISR4K(config)# mgcp

Correct Answer: *D*

Community vote distribution

D (85%) B (15%)

ciscogeek Highly Voted 3 years, 1 month ago

Selected Answer: D

A is wrong because in " mgcp call-agent 10.1.2.3 2428" the port should be 2427

B is wrong because in "ccm-manager config download " there is no download keyword

C is wrong because in "mgcp call-agent 10.1.2.3 2427 service-type mgcp version 11.5.1 " , the version should be 0.1

D is the correct answer, as all commands are correct.

upvoted 13 times

frankieasantiago Most Recent 1 year, 1 month ago

Selected Answer: B

https://telephonynetworking.fandom.com/wiki/CUCM_set_up_MGCP_Gateway

unfortunately its a typo but the instructions state to download the config. Nice tutorial on how to setup the gateway and at the end the command is

Gateway side

use the command show CCM-manager host

use the command show ccm-manager config-download

use the command show mgcp endpoints

Anything other than download is wrong.

upvoted 1 times

Brant 11 months, 1 week ago

D is correct, Refer the cisco document.

https://www.cisco.com/c/en/us/support/docs/voice/media-gateway-control-protocol-mgcp/214635-configure-and-troubleshoot-mgcp-gateways.html

upvoted 2 times

ALLENNN 2 years, 2 months ago

Selected Answer: B

I think it is only typo error

show ccm-manager config-download -> shows the auto-configuration download status

upvoted 2 times

driz 2 years, 1 month ago

without what you say is a "typo" B becomes the same as D, so clearly it is not a typo and D is correct.

upvoted 1 times

MeowthL 2 years, 3 months ago

Selected Answer: D

D is the answer

upvoted 2 times

Omitted 3 years, 1 month ago

Selected Answer: D

It is D. A is wrong because of the port. B is wrong bc ccm-manager config "download" isn't a valid command. C is wrong bc version is the protocol version not CUCM version.

upvoted 3 times

ALLENNN 2 years, 2 months ago

I think it is only typo error

show ccm-manager config-download -> shows the auto-configuration download status

upvoted 1 times

Question #144 *Topic 1*

An engineer must manually register an analog port with Cisco UCM using the MGCP protocol. Which commands are required?

A. no mgcp mgcp call-agent 172.16.1.252 ccm-manager config server 172.16.1.252 ccm-manager config ccm-manager mgcp ! dial-peer voice 1 pots port 1/0/0 shut no shut !

B. mgcp mgcp call-agent 172.16.1.252 ccm-manager config server 172.16.1.252 ccm-manager config ccm-manager mgcp ! dial-peer voice 1 pots port 1/0/0 shut no shut !

C. mgcp ccm-manager config server 172.16.1.252 ccm-manager config ! dial-peer voice 1 pots application MGCPAPP port 1/0/0 !

D. mgcp mgcp call-agent 172.16.1.252 ccm-manager config server 172.16.1.252 ccm-manager config ccm-manager mgcp ! dial-peer voice 1 pots application MGCPAPP port 1/0/0 shut no shut !

Correct Answer: *D*

spag22500 1 year ago

D is correct

https://www.cisco.com/c/en/us/td/docs/ios-xml/ios/voice/mgcp/configuration/15-mt/vm-15-mt-book.pdf

upvoted 3 times

Question #145 *Topic 1*

Users dial a 9 before a 10-digit phone number to make an off-net call. All 11 digits are sent to the Cisco Unified Border Element before going out to the PSTN. The
PSTN provider accepts only 10 digits. Which configuration is needed on the Cisco Unified Border Element for calls to be successful?

A. voice translation-rule 1 rule 1 /^9ɪ¦€ɪ¦€ɪ€¦./ //

B. voice translation-rule 1 rule 1 /^9.+/ //

C. voice translation-rule 1 rule 1 /^9(ɪ¦€ɪ¦€ɪ€¦.)/ //

D. voice translation-rule 1 rule 1 /^9/ //

Correct Answer: *D*

Community vote distribution

D (90%) 10%

Omitted Highly Voted 2 years, 7 months ago

Selected Answer: D

D is the only answer that accomplishes this. Although not ideal as it matches anything beginning with a 9 but it is the only what that strips only the 9.
C is almost right but missing a / after the 9

upvoted 10 times

Omitted 2 years, 7 months ago

If it helps people this is how the answers should look

A) rule 1 /^9........../ // --> this removes everything

B) /^9.+/ // --> this also removes everything

C) /^9(..........)/ // --> this wouldn't match the pattern

D) /^9/ // -->this only removes the 9

upvoted 9 times

G0y0 Most Recent 3 months, 4 weeks ago

Selected Answer: D

#show run | sec voice translation-rule 1

voice translation-rule 1

rule 1 /^9/ //

#test voice translation-rule 1 91234567890

Matched with rule 1

Original number: 91234567890 Translated number: 1234567890

Original number type: none Translated number type: none

Original number plan: none Translated number plan: none

ahi esta su respuesta

upvoted 1 times

Slushed 2 years, 8 months ago

Selected Answer: B

B is the right answer. It will match 9 and any number of following digits.

upvoted 1 times

Slushed 2 years, 8 months ago

Ignore this, B is NOT right, I misread the question. It is like A or C are the right answer, but the patterns are all cryptic looking to me so I cannot determine what they actually are.

upvoted 3 times

G0y0 3 months, 4 weeks ago

agarra un router de tu chamba y haz el ejercicio, no adivines k

upvoted 1 times

santiagof 2 years, 1 month ago

hahahha, none of them are right, no pegaste una.
Correct answer is D
upvoted 1 times

Kabimas66 9 months, 1 week ago

Santiago pareces paisano de Venezuela
upvoted 1 times

Question #146 Topic 1

```
000142: *Apr 23 19:41:49.050: MGCP Packet received from 192.168.100.100:2427--->
AUEP 4 AALN/S0/SU0/0@VG320.cisco.local MGCP 0.1
F: X, A, I
<---

000143: *Apr 23 19:41:49.050: MGCP Packet sent to 192.168.100.101:2427--->
200 4
I:
X: 2
L: p:10-20, a:PCMU;PCMA;G.nX64, b:64, e:on, qc:1, s:on, t:10, r:g, nt:IN, v:T;G;D;L;H;R;ATM;SST;PRE
L: p:10-220, a:G.729;G.729a;G.729b, b:8. e:on, qc:1, s:on, t:10, r:g, nt:IN, v:T;G;D;L;H;R;ATM;SST;PRE
L: p:10-110, a:G.726-16;G.728; b:16, e:on, qc:1, s:on, t:10, r:g, nt:IN, v:T;G;D;L;H;R;ATM;SST;PRE
L: p:10-70, a:G.726-24; b:24, e:on, qc:1, s:on, t:10, r:g, nt:IN, v:T;G;D;L;H;R;ATM;SST;PRE
L: p:10-50, a:G.726-32; b:32, e:on, qc:1, s:on, t:10, r:g, nt:IN, v:T;G;D;L;H;R;ATM;SST;PRE
L: p:30-270, a:G.723.1-H;G.723;G.723.1a-H, b:6. e:on, qc:1, s:on, t:10, r:g, nt:IN, v:T;G;D;L;H;R;ATM;SST;PRE
L: p:30-330, a:G.723.1-L;G.723.1a-L, b:5, e:on, qc:1, s:on, t:10, r:g, nt:IN, v:T;G;D;L;H;R;ATM;SST;PRE
M: sendonly, recvonly, sendrecv, inactive, loopback, conttest, data, netwloop, netwtest
<---
```

Refer to the exhibit. What is the registration state of the analog port in this debug output?

A. The analog port is currently shut down.

B. The analog port is registered to Cisco UCM.

C. The MGCP Gateway is not communicating with the Cisco UCM.

D. The analog port failed to register to Cisco UCM with an error code 200.

Correct Answer: *B*

Community vote distribution

B (100%)

spag22500 1 year ago

B is correct:

https://www.cisco.com/c/en/us/support/docs/voice/media-gateway-control-protocol-mgcp/214635-configure-and-troubleshoot-mgcp-gateways.html#anc19

upvoted 2 times

Omitted 1 year, 8 months ago

Selected Answer: B

CUCM sends an AUEP (Audit Endpoint) to the Gateway to determine the status of the given Endpoint. The response from the Gateway is an ACK with the endpoints capabilities. Once this is complete the Endpoint is registered with the CUCM.

https://www.cisco.com/c/en/us/support/docs/voice/media-gateway-control-protocol-mgcp/214635-configure-and-troubleshoot-mgcp-gateways.html

upvoted 3 times

Question #147 *Topic 1*

Which command is used in Cisco IOS XE TDM gateway to configure the voice T1/E1 controller to provide clocking?

A. clock source line

B. clock source external

C. clock source internal

D. clock source network

Correct Answer: *D*

Community vote distribution

D (75%) C (25%)

Omitted Highly Voted 2 years, 7 months ago

Selected Answer: D

Clock source network - should be configured voice. If the t1 is data only then you would do clock source internal

upvoted 7 times

CiscoSailor Most Recent 1 year, 5 months ago

Selected Answer: D

https://community.cisco.com/t5/collaboration-knowledge-base/t1-pri-line-clock-design-considerations-on-isr4k/ta-p/3163800

upvoted 1 times

Ahochau 1 year, 7 months ago

Selected Answer: D

https://www.cisco.com/c/en/us/td/docs/routers/access/interfaces/NIM/software/configuration/guide/4gen-t1-e1-nim-guide.html

To provide clocking to the line use the command clock source network . The command clock source internal is applicable to data T1/E1 and is not used for T1/E1 voice.

upvoted 2 times

arinpas 1 year, 8 months ago

Looks like the correct answer is C: https://www.cisco.com/c/en/us/td/docs/ios-xml/ios/interface/command/ir-cr-book/ir-c2.html#wp7855926660

upvoted 2 times

sneff91 9 months, 4 weeks ago

Not sure where you got that from. I literally see "clock source network" on my IOS XE router running 17.06.05:

R1(config-controller)#clock source ?
internal Internal Clock
line Line Recovered Clock
network Internal TDMSwitch Source

R1(config-controller)#end
R1#sh vers
Cisco IOS XE Software, Version 17.06.05
Cisco IOS Software [Bengaluru], c8000be Software (X86_64_LINUX_IOSD-UNIVERSALK9-M), Version 17.6.5, RELEASE SOFTWARE (fc2)

upvoted 1 times

sneff91 9 months, 4 weeks ago

Oops - this was meant as a response to glong's comment.

upvoted 1 times

glong 1 year, 8 months ago

Selected Answer: C

XE IOS only have line and internal clock source

upvoted 3 times

sneff91 9 months, 4 weeks ago

Not sure where you got that from. I literally see "clock source network" on my IOS XE router running 17.06.05:

R1(config-controller)#clock source ?
internal Internal Clock
line Line Recovered Clock
network Internal TDMSwitch Source

R1(config-controller)#end
R1#sh vers
Cisco IOS XE Software, Version 17.06.05
Cisco IOS Software [Bengaluru], c8000be Software (X86_64_LINUX_IOSD-UNIVERSALK9-M), Version 17.6.5, RELEASE SOFTWARE (fc2)
upvoted 1 times

AJBELL14 2 years, 4 months ago
D - Clock source network. Not sure why so many answers are wrong here.
upvoted 2 times

leodev 2 years, 6 months ago
Its D. The command clock source internal is applicable to data T1/E1 and is not used for T1/E1 voice.
upvoted 3 times

Question #148 *Topic 1*

An administrator works with an ISDN PRI that is connected to a third-party PBX. The ISDN link does not come up, and the administrator finds that the third-part
PBX uses the QSIG signaling method. Which command enables the Cisco IOS Gateway to use QSIG signaling on the ISDN link?

A. isdn switch-type basic-qsig

B. isdn switch-type basic-ni

C. isdn switch-type primary-qsig

D. isdn incoming-voice voice

Correct Answer: *C*
Reference:
https://www.cisco.com/en/US/docs/ios/dial/configuration/guide/
dia_cfg_isdn_pri_external_docbase_0900e4b1806c752c_4container_external_docbase_0900e4b18216dd1b.html

Community vote distribution
C (100%)

Ol_Mykhailiuk 9 months, 1 week ago
Selected Answer: C
The switch type confiugred must be QSIG:
isdn switch-type primary-qsig
https://community.cisco.com/t5/collaboration-knowledge-base/qsig/ta-p/3126953
upvoted 1 times

Omitted 1 year, 1 month ago
C is correct. A is the answer if it's a BRI
upvoted 2 times

Question #149 *Topic 1*

An administrator needs to create a partial PRI consisting of the first seven timeslots available. Which configuration snippet configures the ISDN E1 PRI for this task?

A. config t 2900(config)#isdn switch-type primary-ni 2900(config)#pri-group timeslots 1-7

B. config t 2900(config)#isdn switch-type primary-ni 2900(config)#controller e1 0/0/0 2900(config-controller)#pri-group timeslots 1-7

C. config t 2900(config)#isdn switch-type primary-ni 2900(config)#interface Serial0/0/0:15 2900(config-controller)#pri-group timeslots 1-7

D. config t 2900(config)#isdn switch-type primary-ni 2900(config)#controller e1 0/0/0 2900(config-controller)#pri-timeslots 1-7

Correct Answer: *B*

Community vote distribution

B (100%)

Omitted 7 months, 3 weeks ago

Selected Answer: B

B is correct.

Enable the ISDN Switch type as Primary-ni globally:

config t

#isdn switch-type primary-ni

Then the controller E1 and Pri group timeslot configuration:

#controller e1 0/0/0

#pri-group timeslots 1-10

upvoted 3 times

Omitted 7 months, 3 weeks ago

except 1-7 for this example.

upvoted 5 times

Question #150 Topic 1

Refer to the exhibit.

```
dspfarm profile 1 transcode universal
  codec g729r8
   codec g729qr8
   codec g711ulaw
  codec g711alaw
  maximum sessions 8
  associate application SCCP
```

Which two codec permutations should be transcoded by this dspfarm? (Choose two.)

A. G.722 to G.729r8

B. G.729r8 to G.711ulaw

C. G.729br8 to G.711alaw

D. iLBC to G.711ulaw

E. G729ar8 to G.711alaw

Correct Answer: *BE*

Community vote distribution

BE (100%)

G0y0 4 months ago

Selected Answer: BE

Assuming that "g729qr8" is a typo (since in the keypad, the key "q" is next to the key "a", and the letter "b" do not look alike "a" or "q" at all), then it would mean "g729ar8", I think answer is BE. However, watch out, if it could be "g729qr8" is g729br8, son answers could be BC, so look carefully the exhibit in the exam to avoid surprises.

upvoted 1 times

ciao 8 months, 2 weeks ago

=> haystacknetworks.com

g729r8 = full g729 spec

g729br8 = g729r8 + built-in VAD

g729ar8 = simplified version of g729r8

g729abr8 = simplified g729r8 (g729ar8) + built-in VAD

g729r8 and g729ar8 are fully inter operable

g729br8 and g729abr8 are fully inter operable

upvoted 1 times

CiscoSailor 1 year, 5 months ago

Selected Answer: BE

I agree with B & E

upvoted 2 times

Janju 1 year, 6 months ago

ilbc is not in the dspfarm profile and g729r8 includes ar8 by default. Br8 is not included. C and D options are incorrect

upvoted 3 times

Panda_man 1 year, 11 months ago

Selected Answer: BE

B and E are correct

upvoted 4 times

wwisp3422112 2 years ago

since those codecs are part of the DSPFARM profile

upvoted 1 times

wwisp3422112 2 years ago

B and E correct

upvoted 1 times

Question #151 *Topic 1*

A company's employees have been complaining that they have been unable to select options on the internal IVR of the help desk. IT support has been given
Cisco UCM traces and below is the snippet of the SDP of the INVITE packet. m=audio 25268 RTP/AVP 18 101 a=rtpmap:0 PCMU/8000 a=rtpmap:18 G729/8000 a=ptime:20 a=fmtp:18 annexb=no a=rtpmap:101 telephone-event/8000 a=fmtp:101 0-15
How is this issue resolved?

A. Configure DTMF for KPML.

B. Configure CODEC for G.729.

C. Configure DTMF for RFC 2833.

D. Configure CODEC for G.722.

Correct Answer: *A*

Community vote distribution

A (100%)

6unc47 Highly Voted 3 years, 1 month ago

In m=audio 25268 RTP/AVP 18 101 not appears the 0 for g711, as rtp-nte should use a non compress codec, the g729 is not the ideal codec for that purpose, so you should configure KPLM

upvoted 7 times

Omitted Highly Voted 3 years, 1 month ago

Selected Answer: A

Configure KPML. You can see from the snippet that 2833 is already configured.

upvoted 5 times

b3532e4 Most Recent 9 months, 1 week ago

C. Configure DTMF for RFC 2833.

upvoted 1 times

Question #152 *Topic 1*

Which command in the MGCP gateway configuration defines the secondary Cisco Unified Communications Manager server?

A. mgcpapp

B. ccm-manager fallback-mgcp

C. mgcp call-agent

D. ccm-manager redundant-host

Correct Answer: *D*

Community vote distribution

D (100%)

Vincentius Highly Voted 4 years, 1 month ago

Should be D: ccm-manager redundant-host

https://www.cisco.com/c/en/us/td/docs/ios/voice/cminterop/configuration/guide/12_4t/vc_12_4t_book/vc_ucm_mgcp_gw.html

upvoted 26 times

Dread Highly Voted 3 years, 3 months ago

Guys has anyone passed recently after using these materials. They seem to be wrong most of them!?

upvoted 8 times

G0y0 Most Recent 3 months, 4 weeks ago

Selected Answer: D

Reference:

Cisco Voice Gateways and Gatekeepers

Chapter 2: Media Gateway Control Protocol

upvoted 1 times

CiscoSailor 12 months ago

Selected Answer: D

I agree it is D

upvoted 2 times

Piji 1 year, 10 months ago

Selected Answer: D

D is correct answer.

upvoted 3 times

somedudebob 3 years, 2 months ago

looking at this, A and C are obviously not correct

B is for configuring SRST

D is for configuring the Secondary CUCM IP

https://www.cisco.com/c/en/us/td/docs/routers/access/vgd1t3/rel1_0/software/configuration/guide/VGD_mgcp.html

See STEP 3 Configuration

upvoted 2 times

Puh 3 years, 4 months ago

D

https://www.cisco.com/c/en/us/td/docs/routers/access/vgd1t3/rel1_0/software/configuration/guide/VGD_mgcp.html

ccm-manager fallback-mgcp - Enables the MGCP fallback feature

ccm-manager redundant-host {ip-address | DNS-name} [ip-address | DNS-name] - Identifies up to two backup Cisco Unified Communications Manager servers.

upvoted 2 times

DEFAULTNERD 3 years, 6 months ago

Yeah B Is Wrong. Who programs these tests.

upvoted 4 times

MrAshour 3 years, 6 months ago

Guys, I am not sure if any of you get a chance to see my comment. but I really would like to thank you all for sharing your experiences, so helpful.

upvoted 3 times

MrAshour 3 years, 6 months ago

D it is

https://www.cisco.com/c/en/us/td/docs/ios-xml/ios/voice/cminterop/configuration/15-mt/dia-15-mt-book/vc-ucm-mgcp-gw.html#GUID-6FB38912-C5DF-448F-8462-7DA2DD44C754

upvoted 2 times

XpressMelo 3 years, 8 months ago

exactly , is should not be B as that answering configure the call for fallback , correct answer is ccm-manager redundant-host

upvoted 3 times

khader09 3 years, 9 months ago

should be D

upvoted 4 times

rishik 3 years, 11 months ago

ccm-manager mgcp

ccm-manager music-on-hold

ccm-manager config server X.X.X.X Primary server

ccm-manager redundant-host X.X.X.X Secondary server

ccm-manager config

!

mgcp

mgcp call-agent X.X.X.X 2427 service-type mgcp version 0.1

upvoted 4 times

rishik 3 years, 11 months ago

So its definitely D

upvoted 7 times

rishik 3 years, 11 months ago

D should be correct answer

B would configure SRST

upvoted 3 times

BarryR 3 years, 12 months ago

Answer is D...ccm-manager redundant-host {ip-address | DNS-name} [ip-address | DNS-name]

upvoted 4 times

Question #153 *Topic 1*

Which action is required if an engineer wants to have Cisco Unified Communications Manager control the configuration for an MGCP gateway?

A. Apply the ccm-manager configuration commands to the gateway.

B. Upload the custom configuration in the TFTP server in Cisco Unified CM.

C. From Cisco Unified CM > Device > Gateway > Add gateway, check the auto-configuration check box.

D. Configure the Cisco Unified CM's IP in voice service VoIP.

Correct Answer: *A*

Community vote distribution

A (100%)

Chispas Highly Voted 4 years, 6 months ago
Should be A
upvoted 15 times

francesca Highly Voted 4 years, 6 months ago
have never seen "auto-config" for a gwy
upvoted 11 times

G0y0 Most Recent 3 months, 4 weeks ago
Selected Answer: A
If you would like the gateway to download much of its MGCP configuration from CallManager, you must code that and give it the IP address or DNS name for the TFTP server (usually CallManager). This downloads XML files with configuration such as MGCP packages, RTP settings, and fax settings. The commands to enable this are as follows:
VoiceGW(config)#ccm-manager config server {ip-address | dns-name}
VoiceGW(config)#ccm-manager config
This file is downloaded when the gateway first communicates with CallManager, before it sends the RSIP message. The file is refreshed if changes are made to CallManager that require the gateway to be reset.
Cisco Voice Gateways and Gatekeepers
Chapter 2: Media Gateway Control Protocol
upvoted 1 times

Sanrio 8 months, 2 weeks ago
Selected Answer: A
Correct answer is A.
upvoted 1 times

WeNt48 1 year, 10 months ago
Selected Answer: A
Definitely go with A
upvoted 2 times

Piji 2 years, 4 months ago
Selected Answer: A
Correct answer is A.
upvoted 2 times

geroboamo 2 years, 9 months ago
Selected Answer: A
auto-configuration doesn't exists.. answer is A
upvoted 3 times

MKZ 3 years, 6 months ago
Should be A
upvoted 2 times

FlashNC 3 years, 6 months ago
Answer is A. to have the CUCM control the MGCP Gateway, you need "ccm-manager configuration"

upvoted 1 times

SDLOA14 3 years, 10 months ago

B) custom config from TFTP

I am going with B based on this article, after defining MGCP gateway in CUCM, you then enter the ccm-manager config commands. See 'Figure 1G-S MGCP Configuration Server Communication'

https://www.ccexpert.us/cisco-unified/mgcp-configuration-server.html

See text "Step 6 Click Save. Reset the gateway for configuration changes to be committed to the gateway"

Thinking about my experience with MGCP gateways, there are a number of parameters that I configure on CUCM & not on the gateway.

upvoted 1 times

SDLOA14 3 years, 8 months ago

Changed my selection. Going with A like everyone else

I got distracted by the 'TFTP server' term in the B) upload configuration configuration ...

upvoted 3 times

XpressMelo 4 years, 2 months ago

we cannot configure mgcp on auto configure on the cucm ; all parameter has to be set manually , so A is correct

upvoted 4 times

YohanesH 4 years, 2 months ago

my guess is A

upvoted 3 times

gottalearnsometime 4 years, 3 months ago

It is A

upvoted 3 times

valsrock 4 years, 4 months ago

Correct answer is letter A

upvoted 5 times

rishik 4 years, 5 months ago

Correct answer is A

upvoted 6 times

Question #154 *Topic 1*

A user reports that when receiving an inbound call on their IP Phone from the PSTN they are unable to transfer this call to another PSTN number. What is the reason for this failure?

A. The IP phone is configured with the wrong region.

B. The incoming calling search space of the SIP trunk does not include the partition of the line in the IP phone.

C. The service parameter related to Offnet to Offnet Call Transfer is set to TRUE.

D. The gateway is configured with the wrong device pool.

Correct Answer: *C*

Community vote distribution

C (100%)

sr_meck 11 months, 2 weeks ago

Correct answer is C.

upvoted 2 times

Piji 1 year, 4 months ago

Selected Answer: C

Correct answer is C.

upvoted 2 times

AJBELL14 1 year, 5 months ago

Selected Answer: C

C is the correct answer.

upvoted 1 times

Omitted 1 year, 7 months ago

Selected Answer: C

The service parameter is "Block OffNet To OffNet Transfer" so by setting it to true it will block the transfer.

upvoted 3 times

plazaliberdad 1 year, 8 months ago

C is correct

upvoted 2 times

Question #155 *Topic 1*

```
Gateway1#show sccp
SCCP Admin State: UP
Gateway Local Interface: Loopback0
        IPv4 Address: 192.168.12.1
        Port Number: 2000

Gateway1#
Gateway1#show ccm-manager
% Call Manager Application is not enabled
Gateway1#

Gateway1#show mgcp
MGCP Admin State DOWN, Oper State DOWN - Cause Code NONE
MGCP call-agent: none Initial protocol service is MGCP 0.1
MGCP validate call-agent source-ipaddr DISABLED
MGCP validate domain name DISABLED
MGCP block-newcalls DISABLED
MGCP send SGCP RSIP: forced/restart/graceful/disconnected DISABLED
```

Refer to the exhibit. A collaboration engineer adds an analog gateway to a Cisco UCM cluster. The engineer chooses MGCP over SCCP as the gateway protocol.

Which two actions ensure that the gateway registers? (Choose two.)

A. Delete and re-add the gateway configuration in Cisco UCM.

B. Enter "mgcp" on the gateway in configuration mode.

C. Enter "no sccp" on the gateway in configuration mode.

D. Enter "ccm-manager config" on the gateway in configuration mode.

E. Enter "ccm-manager mgcp" on the gateway in configuration mode.

Correct Answer: *BE*

Reference:

https://community.cisco.com/t5/unified-communications/cisco-router-2901-mgcp-x-callmanager-9-1-1/td-p/2419551

G0y0 4 months ago

Selected Answer: BE

Yo are right, the minimum MGCP configuration you need is the following:

(config)#mgcp

(config)#mgcp call-agent [ip-address|hostname] [port] service-type mgcp [version 0.1 | 1.0 | rfc3435-1.0]

(config)#ccm-manager mgcp

And (optional) if you want that the Gateway download XML config from CUCM (MGCP packages, RTP settings, and fax settings), you can use:

(config)#ccm-manager config server {ip-address | dns-name}

(config)#ccm-manager config

So, answers are B. and E.

Reference: "Cisco Voice Gateways and Gatekeepers", Cisco Press. Chapter 2: Media Gateway Control Protocol.

upvoted 2 times

wwisp3422112 7 months, 1 week ago

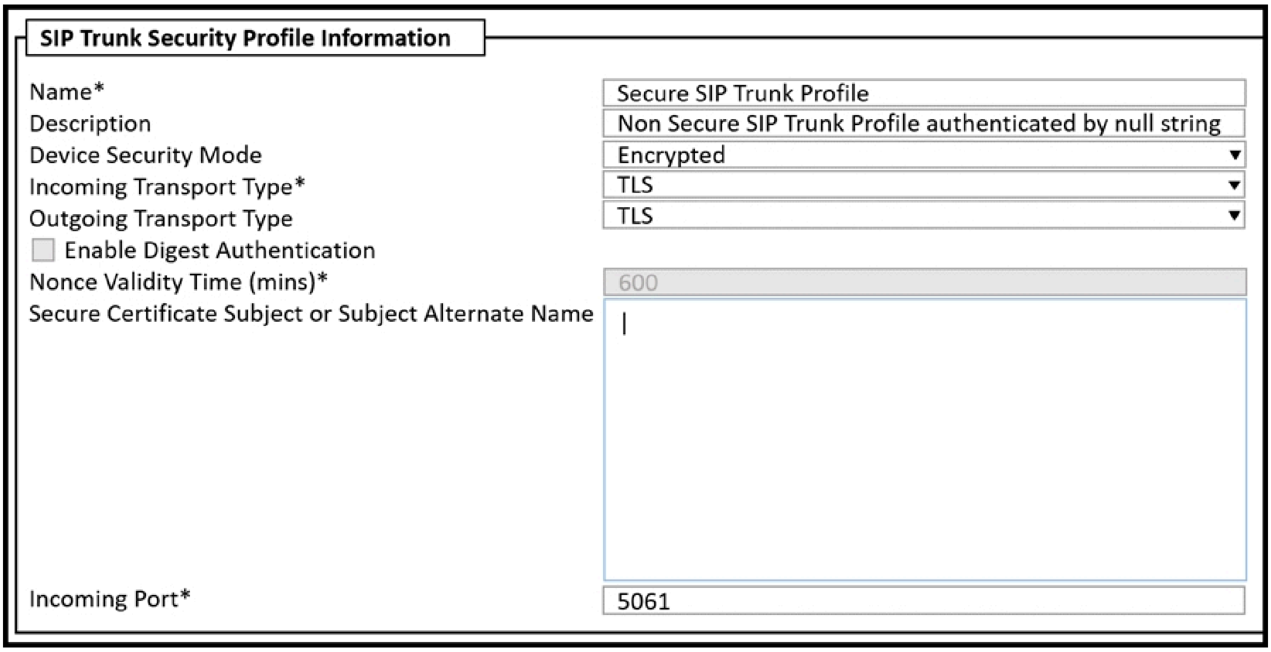
B and E correct:

https://www.cisco.com/en/US/docs/ios/12_3t/voice/command/reference/vrht_c4_ps5207_TSD_Products_Command_Reference_Chapter.html#wp1072910

upvoted 1 times

Question #156 Topic 1

Refer to the exhibit.


SIP Trunk Security Profile Information

Name*	Secure SIP Trunk Profile
Description	Non Secure SIP Trunk Profile authenticated by null string
Device Security Mode	Encrypted
Incoming Transport Type*	TLS
Outgoing Transport Type	TLS
Enable Digest Authentication	
Nonce Validity Time (mins)*	600
Secure Certificate Subject or Subject Alternate Name	
Incoming Port*	5061

An administrator configures a secure SIP trunk on Cisco UCM. Which value is needed in the Secure Certificate Subject or Subject Alternate Name field to accomplish this task?

A. the common name of the remote device certificates

B. the fully qualified domain name of all Cisco UCM nodes that run the CallManager service

C. the common name of the Cisco UCM CallManager certificate

D. the fully qualified domain name of the remote device that is configured on the SIP trunk

Correct Answer: *D*

Community vote distribution

D (48%) A (41%) 10%

Slushed Highly Voted 3 years, 1 month ago

Selected Answer: D

The correct answer is D.

https://video.cisco.com/video/6196744148001

upvoted 12 times

CiscoSailor Highly Voted 1 year, 12 months ago

Selected Answer: A

I think this is A. Yes, CN is often the same as FQDN, but not always. A is the better answer. This field refers to the certificate of the remote end, not the local CUCM, so B & C are incorrect.

upvoted 5 times

[Removed] 1 year, 11 months ago

NO, MR CHATGPT!

upvoted 1 times

G0y0 Most Recent 4 months ago

Selected Answer: A

Correct answer is A.

It is useless to specify the FQDN if it is not explicitly stated in the CN of the peer's certificate. If you put the FQDN, the CUCM will check if the FQDN is in the peer's certificate. If the FQDN is not in the CN, the CUCM will reject transactions with the peer.

You need to put there what is specified in the CN of the peer's certificate, be it the FQDN, the domain, the hostname, the IP address, etc. Common Name (CN) is not synonym of FQDN.

For example, if you do a TLS connection with an Expressway, the subject name or an subject alternate name provided, by the Expressway in its certificate. For Expressway clusters, ensure that this list includes all of the names contained within all of the peers' certificates. To specify multiple X.509 names, separate each name by a space, comma, semicolon or colon.

So , correct answer is A.
upvoted 2 times

G0y0 4 months ago
Reference: "Preferred Architecture for Cisco Collaboration 12.x Enterprise On-Premises Deployments", Chapter 7: Security, in the section "SIP Trunk Encryption", yo can read what is exactly X.509 Subject Name, that is "The common name (CN) of the remote party." and some examples.
upvoted 1 times

G0y0 4 months ago
Table 7-11 SIP Trunk Security Profile Parameters for Secure SIP Trunks
upvoted 1 times

DDPRE 4 months, 2 weeks ago
Selected Answer: D
As per Copilot:

For configuring a secure SIP trunk on Cisco Unified Communications Manager (UCM), the correct value needed in the Secure Certificate Subject or Subject Alternate Name field is:

D. the fully qualified domain name of the remote device that is configured on the SIP trunk.

This ensures that the certificate matches the domain name of the remote device, which is essential for establishing a secure TLS connection.
upvoted 1 times

decdca7 7 months, 2 weeks ago
Selected Answer: A
Common name or SAN that is how certificates work.
upvoted 1 times

cyberknock 9 months, 3 weeks ago
Selected Answer: A
A - Even the field Name indicates that the Common Name is meant...
upvoted 2 times

cyberknock 9 months, 3 weeks ago
C sorry
upvoted 1 times

JoeC716 1 year, 1 month ago
Selected Answer: D
Agree with Slushed - Look at 7:40 https://www.youtube.com/watch?v=mTEtVOHKf20
upvoted 2 times

G0y0 4 months ago
In the reference you provide, you are talking about a CUC certificate, which does match the FQDN, but the question does not tell you what is on the other side, whether it is a UNITY, an IM&T, a CUBE, a Gateway, an Expressway-C, or a cluster, or an Oracle SBC, etc. etc. etc. FQDN is not the same as CN.
If the question said that the remote device is a CUC, D would be correct. But assuming that it is the FQDN of a remote device that we do not know, the first thing we have to see is the CN and in this case it is precisely the one that is not used, therefore it is A.
It is recommended that if you are going to cite a source, it is valued. But please, interpret and read the source correctly before advertising it.
upvoted 1 times

TheBabu 1 year, 2 months ago
Selected Answer: A
If the FQDN that is configured on the SIP trunk is not present in the certificate, the TLS connection fails.
You gotta use an FQDN that exists on the remote device's certificate, that's how certificate validation works.
upvoted 2 times

Daved90 1 year, 3 months ago
Selected Answer: A
FQDN != CN or SAN, often it is but not always

upvoted 3 times

Kabimas66 1 year, 3 months ago

from https://www.cisco.com/c/en/us/td/docs/voice_ip_comm/cucm/security/11_5_1_SU3/cucm_b_security-guide-1151su3/cucm_b_security-guide-1151su3_chapter_011000.html, you can read explanation of "Secure Certificate Subject or Subject Alternate Name" field of SIP Trunk Security Profile Settings

It looks answer is C

upvoted 1 times

SergeantDuty 1 year, 11 months ago

Selected Answer: A

It's always the CN of the remote certificate. CN can bei the FQDN or Hostname or a MAC in case of Analogue Gateways. e.g. The CN is configured at the trustpoint configuration (Cisco IOS).

upvoted 5 times

[Removed] 1 year, 11 months ago

NO, MR CHATGPT!

upvoted 2 times

SergeantDuty 1 year, 11 months ago

Have a Look at this Link (https://www.cisco.com/c/en/us/support/docs/unified-communications/unified-communications-manager-callmanager/200180-Configure-SIP-TLS-Trunk-on-the-Communica.html). Section: Step 8. Create SIP Trunk Security Profiles. Here you can See that it is the CN of the remote Host/devices. At the screenshots below you can See that a CN is not always a FQDN. In this example it is CUCM10.

upvoted 3 times

TeeKay25 2 years, 1 month ago

Selected Answer: B

B is the correct answer

upvoted 1 times

[Removed] 1 year, 11 months ago

NO, MR CHATGPT!

upvoted 1 times

azizkasmir 2 years, 6 months ago

Selected Answer: D

https://www.google.com/url?
sa=t&rct=j&q=&esrc=s&source=web&cd=&cad=rja&uact=8&ved=2ahUKEwik1rGj9PT7AhVTH7cAHUcHBFgQFnoECBMQAw&url=https%3A%2F%2Fwww.youtube
mXaZI

upvoted 4 times

Daved90 1 year, 3 months ago

FQDN =! CN in a certificate, A is more correct

upvoted 1 times

KZG 2 years, 10 months ago

If you have a Unified Communications Manager cluster or if you use SRV lookup for the TLS peer, a single trunk may resolve to multiple hosts, which results in multiple Secure Certificate Subject or Subject Alternate Name for the trunks

https://www.cisco.com/c/en/us/td/docs/voice_ip_comm/cucm/security/12_5_1/cucm_b_security-guide-1251/cucm_b_security-guide-1251_chapter_011000.html

upvoted 1 times

Piji 2 years, 10 months ago

Selected Answer: D

Correct answer is D.

upvoted 2 times

AJBELL14 2 years, 11 months ago

Selected Answer: B

I couldn't find specific info on the cisco site. As per other forums - Option B - the fully qualified domain name of all Cisco UCM nodes that run the CallManager service is the right answer

upvoted 2 times

Question #157 *Topic 1*

A company hosts a conference call with no local users. How does the administrator stop the conference from continuing?

A. remove the transcoder

B. modifies the Block OffNet to OffNet Transfer service parameter

C. changes the codecs that are supported on the conference resource

D. modifies the Drop Ad Hoc Conference service parameter

Correct Answer: *D*

Community vote distribution

D (100%)

G0y0 4 months ago

Selected Answer: D

This parameter determines how an Ad Hoc conference terminates. Valid values follow:
Never - The conference remains active a) after the conference controller hangs, and b) after all OnNet parties hang up. Be aware that choosing this option means that if OnNet parties conference in OffNet parties and then disconnect, the conference stays active between the OffNet parties, which could result in potential toll fraud.
When Conference Controller Leaves - Terminate the conference when the conference controller hangs up or when the conference controller transfers, redirects, or parks the conference call and the retrieving party hangs up.
When No OnNet Parties Remain in the Conference - Terminate the conference when there are no OnNet parties remaining in the conference.
This is a required field.
Default: Never

upvoted 1 times

CiscoSailor 12 months ago

Selected Answer: D

I agree D is correct

upvoted 1 times

Piji 1 year, 10 months ago

Selected Answer: D

Correct Answers is D.

upvoted 1 times

ciscogeek 2 years, 1 month ago

Selected Answer: D

■ Drop Ad Hoc Conference: This parameter determines how an ad hoc conference terminates. This is an important toll-fraud prevention setting, because inside facilitators can set up a conference call to expensive international numbers and then drop out of the call. Without the conference controller, international tariffs are billed back to the company in which the conference call was set up. Valid values are as follows:

—Never (default): The conference remains active after the conference controller and all on-net parties hang up. This default setting could result in potential toll fraud.

—When Conference Controller Leaves: Terminate the conference when the conference controller hangs up.

—When No On-Net Parties Remain in the Conference: Terminate the conference when there are no on-net parties remaining in the conference. This distinction is important because the conference controller might have to drop out of the call, but other business partners on the call should continue the conference. The When Conference Controller Leaves option would hang up the call when the conference controller left the conference.

upvoted 4 times

Slushed 2 years, 1 month ago

Selected Answer: D

Correct answer is D.

upvoted 1 times

Question #158 *Topic 1*

After an engineer implements the FAC and CMC features together, users report that calls take almost one minute to complete and that they occasionally hear the reorder tone. Which two actions address this issue? (Choose two.)

A. Do not wait for the tones; immediately dial the FAC and CMC.

B. Advise the user to hang up and try again.

C. Advise the user to press the button after dialing the FAC and CMC codes.

D. Change the code if the problem persists.

E. Adjust the T302 timer from the default of 15 seconds to 5 seconds to shorten the interdigit timer.

Correct Answer: *CE*

Community vote distribution

CE (100%)

Omitted Highly Voted 2 years, 1 month ago

Selected Answer: CE

I think C is supposed to say press the # button

upvoted 7 times

MaxG Most Recent 1 year ago

Selected Answer: CE

Another one for C and E because:

Forced Authorization Codes (FAC) and Client Matter Codes (CMC) allow you to manage call access and accounting. CMC assists with call accounting and billing for billable clients, while Forced Authorization Codes regulate the types of calls that certain users can place.

Client matter codes force the user to enter a code to specify that the call relates to a specific client matter. You can assign client matter codes to customers, students, or other populations for call accounting and billing purposes. The Forced Authorization Codes feature forces the user to enter a valid authorization code before the call completes.

The delay is related to this interaction;

If you do not append the FAC or CMC with #, the system waits for the T302 timer to extend the call. So instruct users to press # at the end of the dial string.

http://www.cisco.com/en/US/docs/voice_ip_comm/cucm/admin/8_5_1/ccmfeat/fsfaccmc.html#wp1052254

upvoted 3 times

Question #159 *Topic 1*

```
rule 1 /^\(0[25]..\)\-\(...\)\-\(....$\)/ /\1\2\3/
```

Refer to the exhibit. The translation rule is configured on the voice gateway to translate DNIS. What is the outcome if the gateway receives 0255-343-1234 as
DNIS?

A. The translation rule is not matched because DNIS contains ג-€ג€.

B. The translation rule is not matched because DNIS does not end with a ג$€ג€.

C. The translation rule is matched and the translated number is 02553431234.

D. The translation rule is matched and the translated number is 025553431234.

Correct Answer: *C*

Community vote distribution

C (100%)

davidanibalmarcelino Highly Voted 3 years ago

Should be C

ISR4331(config)#voice translation-rule 10

ISR4331(cfg-translation-rule)#rule 1 /^\(0[25]..\)\-\(...\)\-\(....$\)/ /\1\2\3/

ISR4331#test voice translation-rule 10 0255-343-1234

Matched with rule 1

Original number: 0255-343-1234 Translated number: 02553431234

Original number type: none Translated number type: none

Original number plan: none Translated number plan: none

upvoted 20 times

timmyz 2 years, 10 months ago

I performed the same config/ test and received the same SHOULD BE C

upvoted 3 times

kitty73 Most Recent 11 months ago

Selected Answer: C

Scroll to the end.... https://www.cisco.com/c/en/us/support/docs/voice/call-routing-dial-plans/61083-voice-transla-rules.html

upvoted 2 times

TeeKay25 1 year, 1 month ago

Selected Answer: C

Correct Answer is C

upvoted 1 times

Mert_kerna 1 year, 6 months ago

Selected Answer: C

Keep in mind that any array of numbers with in the [] brackets means that it's any of the given numbers. In this case, you would read it as 2 or 5, not 2 and 5 and not 2 through 5. This is why its C and not D. And the REGEX simply matches the input.

upvoted 2 times

Piji 1 year, 10 months ago

Selected Answer: C

Correct answers is C.

upvoted 2 times

AJBELL14 1 year, 11 months ago

For those who cant see the first 2 options - here they are

A. The translation rule is not matched because DNIS contains “-“.

B. The translation rule is not matched because DNIS does not end with a “$”.

upvoted 4 times

Komy 2 years, 12 months ago

Answer is C

The rule is matched and the number is translated (however, in this case, the post translation number is the same as the original number)

upvoted 4 times

Alan100 1 year, 3 months ago

The translation strips off the "-" characters

upvoted 1 times

Question #160 *Topic 1*

What field is configured to change the caller ID information on a SIP route pattern?

A. Route Partition

B. Called Party Transformation Mask

C. Calling Party Transformation Mask

D. Connected Line ID Presentation

Correct Answer: *C*

Community vote distribution

C (100%)

rishik Highly Voted 4 years, 5 months ago

Correct answer should be C

CallingPartyTransformationMask

Enter a transformationmask value. Valid entriesincludethe digits 0 through9and the wildcardcharactersX, asterisk(*), and octothorpe(#). If this field isblankandtheprecedingfieldisnotchecked,nocallingpartytransformationtakesplace.

upvoted 15 times

PVDM Highly Voted 4 years, 6 months ago

is it C is the right answer?
from documentation:
Cisco Unified Communications Manager uses connected line ID presentation
(COLP/COLR) as a supplementary service to allow or restrict the called party
phone number on a call-by-call basis.

upvoted 8 times

c6176b5 Most Recent 11 months, 1 week ago

Selected Answer: C

Answer should be C

upvoted 1 times

norealchaos 1 year, 3 months ago

Selected Answer: C

correct answer is c

upvoted 1 times

CiscoSailor 1 year, 5 months ago

Selected Answer: C

I agree with C

upvoted 1 times

Piji 2 years, 4 months ago

Selected Answer: C

Correct Answer is C.

upvoted 2 times

BertaAle 2 years, 6 months ago

Is C, Line ID presentaion has 3 fixed value: default, allowed, restricted, you cannot change digit.

upvoted 1 times

Barney_best 2 years, 6 months ago

A. Is wrong
B. There is no Called Party transformation Mask in SIP Route Pattern
D. This is only Allow/Restrict and does not change the Caller ID

My best guess is C. Calling Party Transformation Mask.
upvoted 2 times

alphastate 3 years, 1 month ago
https://www.cisco.com/c/en/us/td/docs/voice_ip_comm/cucm/admin/11_0_1/sysConfig/CUCM_BK_C733E983_00_cucm-system-configuration-guide-1101/cucm_mp_t9cd64bd_00_transformation-pattern-chapter-map.pdf

"Configure Calling Party Transformation Patterns
Use this procedure to transform the calling number. For example, you can configure a transformation pattern that replaces a caller's extension with the office's main number when calling the PSTN."
upvoted 1 times

mmollura 3 years, 5 months ago
The correct answer is C
upvoted 1 times

Raman123 3 years, 7 months ago
Correct Answer shd be D ..
Unified Communications Manager uses connected line ID presentation (COLP) as a supplementary service to provide the calling party with the connected party number. The SIP trunk level configuration takes precedence over the call-by-call configuration.
upvoted 1 times

XalaGyan 3 years, 9 months ago
The question states "What field is configured to CHANGE the CALLER ID (Calling Partynumer) on a SIP Route Pattern?"

Whether SIP of SCCP or H323, in the basic call signaling they behave the same. Caller ID = Calling Party Number.

Connected Line ID Presentation of CLIP refers to ON or OFF, meaning it changes the way a CNG Party ID is displayed or not.

The question here is the WORD "CHANGE !!!!" and that is only doable with Route Patterns (Calling Party Number Mask) , Called Party Xformation Mask (CALLED PARTY but here we need CALLING Party) and finally the Calling Party Xformation Mask (CALLING PARTY + Modification)

I strongly believe the wording and question refers to Xformation Masks.

Lets not forget Xformation masks are NEW wheile CLIP or CLIR (Caller ID RESTRICTED the oposite of CLIP) has always been there since ISUP in SS7 or ISDN PRI.

C it is for me and in about 3 hours i have my exam and can be even more sure.
upvoted 5 times

DaKenjee 2 years, 1 month ago
same here, keyword is >change<
Answer D is not fitting, because details explain >called party phone number<
where Caller ID is a phone feature that provides the name and number of a calling party

As long as we deal with manipulating calling, it is Answer C -> Calling Party Transformation Mask

Connected Line ID Presentation:
Unified Communications Manager uses connected line ID presentation (COLP/COLR)
as a supplementary service to allow or restrict the
called party phone number
upvoted 1 times

0n3 3 years, 11 months ago
B is correct. Is the only place to change the information. The Connected Line ID Presentation determine if the information is presented or not. But does not manipulate/change what the information it self is.
upvoted 2 times

ocero 3 years, 11 months ago
Id D, the question says caller=calling....
upvoted 2 times

Vincez 4 years, 1 month ago

The answer is D.

The key is SIP Route Pattern and which field.

From Call Routing - SIP Route pattern add a new SIP Route pattern

The field to change is definitely Connected Line ID Presentation

upvoted 4 times

Jetnor 4 years, 5 months ago

D usage as described below should not be correct.

if we want to change dhe caller ID as the call is being forwarded, we should change it on the calling party transformation, so it C should be correct.

Connected Line ID Presentation:

Cisco Unified Communications Manager uses connected line ID presentation as a supplementary service to allow or restrict the called party phone number on a per-call basis. Choose one of the following options to allow or restrict the display of the connected party phone number on the calling party phone display for this translation pattern:

Default - This option does not change the connected line ID presentation.

Allowed - This option displays the connected party phone number.

Restricted - Cisco Unified Communications Manager blocks the display of the connected party phone number.

Note

If the incoming call goes through a translation or route pattern and the connected line ID presentation field is set to allowed or restricted, the system modifies the connected line presentation indicator with the translation or route pattern setting.

upvoted 2 times

BarryR 4 years, 5 months ago

Correct answer is D.

upvoted 1 times

BarryR 4 years, 5 months ago

After reading more I do believe it is C Calling party transformation mask

upvoted 5 times

Question #161 *Topic 1*

An engineer configures Cisco Unified Communications Manager to prevent toll fraud. At which two points does the engineer block the pattern in Cisco Unified CM to complete this task? (Choose two.)

A. route pattern

B. route group

C. translation pattern

D. partition

E. CSS

Correct Answer: *CE*

Community vote distribution

AC (100%)

virtu Highly Voted 3 years, 11 months ago

I`ve passed the exam yesterday, and my call control score was 100%.

I answered A and C. Route and Translation patterns. My thought was like "they are asking about, where admin can block particular pattern".

upvoted 29 times

Russel72 1 month ago

B.S. - of you block route pattern or TP , how do you route the call to outside?

upvoted 1 times

DaKenjee 2 years, 1 month ago

Translationpatter can block, does not matter, in which CSS it will reroute call

Routepattern can block, when call goes outside

All other mentioned value, are part of call routing, but not part of blocking

upvoted 1 times

PunKike 3 years, 10 months ago

What material did you use?

upvoted 1 times

Mli2604 3 years, 11 months ago

"at witch point" - for sure this points are route an translation pattern

Can someone confirm this ? every dump say PT & CCS but i also think Route and Translationspattern are the correct answer

upvoted 2 times

PunKike 3 years, 10 months ago

Maybe is Route Pattern and Translation Pattern, because "they are asking about, where admin can block particular pattern" Sounds logical

upvoted 2 times

Jetnor Highly Voted 4 years, 5 months ago

If we focus on patterns only, both translation pattern & route patern have the ability to check box (block this pattern). and the answer should be A & C

Also CSS and partitions are used to prevent toll fraud.

the question is tricky, but it says "block the pattern" so literally A&C have this option writen on it.

upvoted 13 times

mcbesy Most Recent 1 month, 3 weeks ago

Selected Answer: DE

Provide segmentation and control to the number that can be called, or vice versa. As a leading practice recommendation, either disable Call Forward All or limit it to an extension within your Collaboration network. Call Forward Busy and Call Forward No Answer should also be limited to internal partitions only. For phones with extension mobility, a logged-out CSS should be restricted to internal and emergency partitions only.

https://www.ciscopress.com/articles/article.asp?p=2218297&seqNum=10

upvoted 1 times

Russel1972 3 months, 1 week ago

Selected Answer: DE

You can definitely block calls through Route Pattern and Translation Pattern, but that should block legitimate calls along with toll frauds. We need to understand how the toll fraud would occur in the Telephony environment. Example would be: someone calling an extension (internal/ external) and that extension is forwarded to an international number. This should be blocked by proper Partitions and calling search spaces (CSS) designs.

upvoted 1 times

Afie 7 months, 3 weeks ago

A and C

CSS/call privileges are used to permit or allow calls. So you can grant or deny access to certain destinations. Only the selected partitions (dialable numbers) would be granted. There is no Block or allow option for CSS.

Route and Translation patterns on the other hand has the option to block or allow the Pattern.

upvoted 2 times

CiscoSailor 1 year, 5 months ago

Selected Answer: AC

Must be A & C

upvoted 2 times

marjana_mirza 1 year, 8 months ago

Selected Answer: AC

As per the discussing on this thread. The question is asking for "pattern" so the answer is route pattern and translation pattern

upvoted 2 times

vaniobesta 2 years, 7 months ago

correct answer is CSS and Partition

upvoted 2 times

Brant 3 years, 3 months ago

D&E are correct. in CUCM TF can be controlled via COS. Partition and CSS are CUCM COS components.

https://www.ciscopress.com/articles/article.asp?p=2218297&seqNum=10

upvoted 1 times

devadarshan91730 3 years, 5 months ago

Partition and CSS

upvoted 3 times

devadarshan91730 3 years, 5 months ago

on second thought, it should be route pattern and translation pattern

upvoted 1 times

Nila1 3 years, 6 months ago

A and C

Route Option are

1-Route this pattern

or

2-Block this pattern

upvoted 2 times

Ruslans 3 years, 10 months ago

Partition and CSSs provide segmentation and control for the number that can be called

upvoted 1 times

choomiksa 4 years ago

There is a similer question (answer is partition/CSS) - Which two configuration elements are part of the Cisco Unified Communications Manager toll-fraud prevention? (Choose two.)

A.

SIP trunk security profile.

B.

Partition

C.

Calling search space

D.

SUBSCRIBE calling search space

E.

Feature control policy

upvoted 3 times

MrAshour 4 years ago

You are right choomiksa (D & E) (Partitions and calling search spaces (CSS))

https://www.ciscopress.com/articles/article.asp?p=2218297&seqNum=10

upvoted 2 times

XalaGyan 4 years ago

It is actually very simple. you have a pattern for +1! as and your users shall use it. so now try you CSS and partition idea when asked to allow +1! however excluding +1888! numbers.

Translation Pattern is choice 1 to block

Route Pattern for lazy people like me who like to have it combined in one place.

upvoted 1 times

rishik 4 years, 5 months ago

CE can block calls together. Tricky question

upvoted 2 times

rishik 4 years, 5 months ago

On second look translation pattern and route pattern got ability to block calls. So it should be A & C

upvoted 5 times

BarryR 4 years, 5 months ago

Answer is D and E. When you attempt to make a call, Cisco CallManager looks into the CSS of the calling party and checks if the called party belongs to a partition within the CSS. If it does, the call is placed. If not, the call is rejected

upvoted 4 times

Grebec94 4 years, 6 months ago

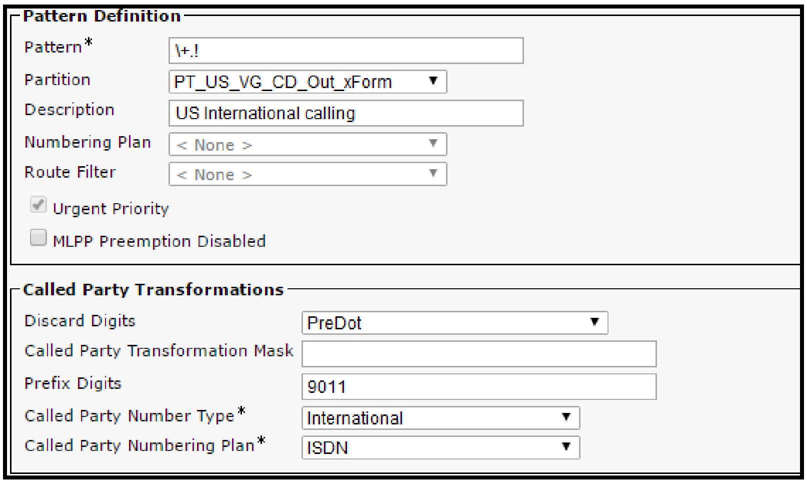
I think A, C is correct they both have block pattern option,

upvoted 7 times

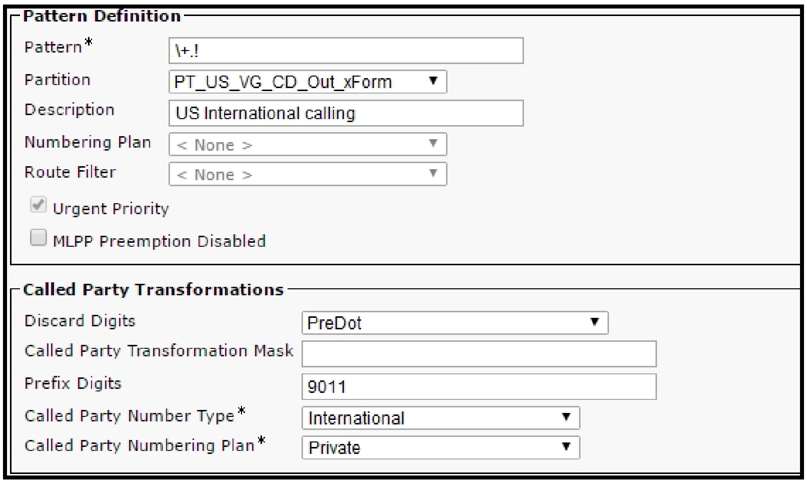
Given this H.323 gateway configuration and using Cisco best practices, how must the called party transformation pattern be configured to ensure that a proper
ISDN type of number is set?

```
voice translation-rule 40
 rule 1 /3…$/ /408555&/
!
voice translation-profile INT
 translate calling 40
!
dial-peer voice 9011 pots
 translation-profile outgoing INT
 destination-pattern 9011T
 port 0/1/0:23
```

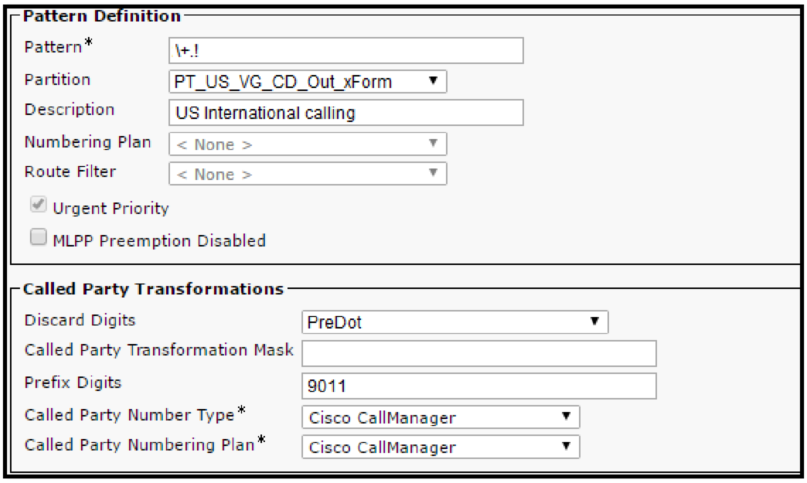
A.



B.

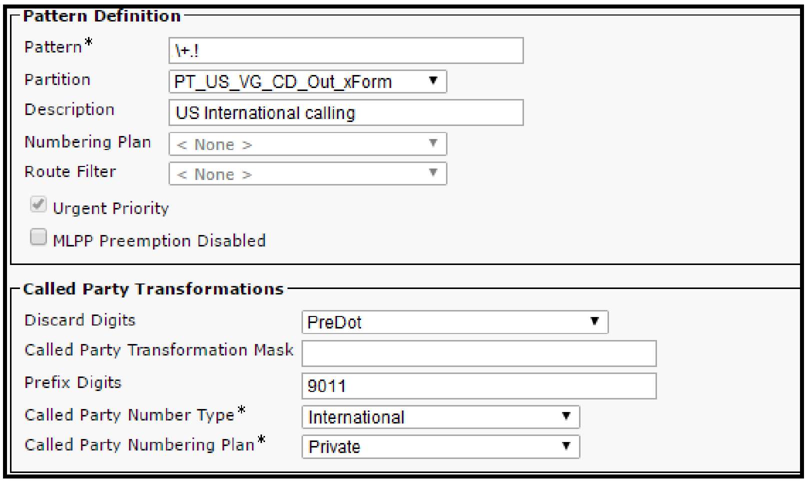


C.


Pattern Definition
Pattern* \+.!
Partition PT_US_VG_CD_Out_xForm
Description US International calling
Numbering Plan < None >
Route Filter < None >
Urgent Priority
MLPP Preemption Disabled

Called Party Transformations
Discard Digits PreDot
Called Party Transformation Mask
Prefix Digits 9011
Called Party Number Type* Cisco CallManager
Called Party Numbering Plan* Cisco CallManager

D.


Pattern Definition
Pattern* \+.!
Partition PT_US_VG_CD_Out_xForm
Description US International calling
Numbering Plan < None >
Route Filter < None >
Urgent Priority
MLPP Preemption Disabled

Called Party Transformations
Discard Digits PreDot
Called Party Transformation Mask
Prefix Digits 9011
Called Party Number Type* International
Called Party Numbering Plan* Private

Correct Answer: *C*

Janu82 Highly Voted 4 years ago

The correct answer is A. (Type ISDN)

upvoted 18 times

BarryR Highly Voted 3 years, 12 months ago

A is correct answer

upvoted 12 times

mtc_mnd Most Recent 2 months, 4 weeks ago

CUCM help page about the Route Pattern configuration says:

1) "Cisco recommends that you do not change the default value unless you have advanced experience with dialing plans such as NANP or the European dialing plan".
Default settings for Called Party Number Type and Called Party Numbering Plan are Cisco CallManager and I have many customers using such settings for international calls.

2) Called Party Number Type: "International—Use when you are dialing outside the dialing plan for your country."

3) Called Party Numbering Plan:"ISDN—Use when you are dialing outside the dialing plan for your country. "

1 and 2+3 seem to conflict. What are Cisco best practices, exactly?

upvoted 1 times

kitty73 11 months ago

Correct A because they are not using an internally globalized numbering plan. By document Called Party Number - International—Use when you are dialing outside the dialing plan for your country. and Called Party Numbering Plan - ISDN—Use when you are dialing outside the dialing plan for your country. https://www.cisco.com/c/en/us/td/docs/voice_ip_comm/cucm/admin/10_0_1/ccmcfg/CUCM_BK_C95ABA82_00_admin-guide-100/CUCM_BK_C95ABA82_00_admin-guide-100_chapter_0110010.html

upvoted 2 times

CiscoSailor 12 months ago

I agree with A

upvoted 3 times

Komy 2 years, 12 months ago

I would also say A, based on the below link:
https://www.cisco.com/c/en/us/td/docs/voice_ip_comm/cucm/admin/10_0_1/ccmcfg/CUCM_BK_C95ABA82_00_admin-guide-100/CUCM_BK_C95ABA82_00_admin-guide-100_chapter_0110010.html

Notice these two statements:
"Called party number type: International—Use when you are dialing outside the dialing plan for your country."
"Called party numbering plan: ISDN—Use when you are dialing outside the dialing plan for your country".

The route pattern and the dial peer show that we are dialing outside of the country's dial plan, thus choosing option (A)

upvoted 5 times

jimboo 3 years, 10 months ago

I would say A.
https://blog.ine.com/2010/01/11/building-global-dial-plans-in-cucm7-part-iii-mapping-global-calling-party-numbers-to-their-local-variant

upvoted 4 times

rishik 3 years, 11 months ago

Correct is C as per SRND

upvoted 3 times

onetruechamp 3 years, 11 months ago

Correct answer is C, the ISDN option is selected in the calling party transformation and not called party transformation.

upvoted 4 times

DaKenjee 1 year, 7 months ago

Correct is Answer A

All Called Party Transformation Pattern hits dialpeer with destination 9011
Diffrence is signaled called Party Number Type + Plan
H323 Protocol and translation-rule are not here relevant.
But in this example a POTS Dial-Peer hits a T1 PRI, where we have to signal for calling and called number a ISDN Type + Plan

I have customers, where we send it >unknown< (matchs Answer C, not explicit signaled).
By Answer C, we assume on diffrent ITSP requirements, that ITSP is aware of missing ISDN signaling and simply checks signaled number.
But This exam question is way to old, when we discuss H323 and ISDN in 2022, so ia ssume, it have to be signaled, like this is otherwise a missing signaling to ITSP and an error.

upvoted 1 times

kljw5 2 years, 1 month ago

The question clearly says "called" party. C is incorrect

upvoted 1 times

Simpax_100 3 years, 11 months ago

But; ISDN—Use when you are dialing outside the dialing plan for your country, for Called and Calling Party Numbering Plan.
https://www.cisco.com/c/en/us/td/docs/voice_ip_comm/cucm/admin/9_1_1/ccmcfg/CUCM_BK_A34970C5_00_admin-guide-91/CUCM_BK_A34970C5_00_admin-guide-91_chapter_0110001.html

upvoted 2 times

Question #163 *Topic 1*

An engineer with ID012345678 must build an international dial plan in Cisco Unified Communications Manager. Which action should be taken when building a variable-length route pattern?

A. reduce the T302 timer to less than 4 seconds

B. configure single route pattern for international calls

C. create a second route pattern followed by the # wildcard

D. set up all international route patterns to 0.!

Correct Answer: *C*

Community vote distribution

C (100%)

Vincentius Highly Voted 4 years, 7 months ago

This one should be C, create a secondary route pattern.

https://www.cisco.com/c/en/us/td/docs/voice_ip_comm/cucm/srnd/collab11/collab11/dialplan.html

upvoted 21 times

pasangawa Most Recent 12 months ago

Selected Answer: C

Should be C

upvoted 2 times

kitty73 1 year, 4 months ago

Selected Answer: C

The ! wildcard is also used for deployments in countries where the dialed numbers can be of varying lengths. In such cases, Unified CM does not know when the dialing is complete and will wait for 15 seconds (by default) before sending the call. You can reduce this delay in any of the following ways:

– Reduce the T302 timer (Service Parameter TimerT302_msec) to indicate end of dialing, but do not set it lower than 4 seconds to prevent premature transmission of the call before the user is finished dialing.

– Configure a second route pattern followed by the # wildcard (for example, 9.011!# for North America or 0.00!# for Europe), and instruct the users to dial # to indicate end of dialing. This action is analogous to hitting the "send" button on a cell phone.

https://www.cisco.com/c/en/us/td/docs/voice_ip_comm/cucm/srnd/collab11/collab11/dialplan.html

upvoted 2 times

CiscoSailor 1 year, 5 months ago

Selected Answer: C

It is C

upvoted 2 times

marjana_mirza 1 year, 8 months ago

Correct ans: C

+++

– Reduce the T302 timer (Service Parameter TimerT302_msec) to indicate end of dialing, but do not set it lower than 4 seconds to prevent premature transmission of the call before the user is finished dialing.

– Configure a second route pattern followed by the # wildcard (for example, 9.011!# for North America or 0.00!# for Europe), and instruct the users to dial # to indicate end of dialing. This action is analogous to hitting the "send" button on a cell phone.

+++

source: https://www.cisco.com/c/en/us/td/docs/voice_ip_comm/cucm/srnd/collab11/collab11/dialplan.html#88428

upvoted 2 times

BhaiKyare 3 years, 4 months ago

When you reduce T302 timer very less like 4 seconds you will have problem with UCCX IVR menus and also it times out for even local and long distance calls ?

upvoted 2 times

jonycakes 3 years, 6 months ago

C is correct: – Configure a second route pattern followed by the # wildcard (for example, 9.011!# for North America or 0.00!# for Europe), and instruct the users to dial # to indicate end of dialing. This action is analogous to hitting the "send" button on a cell phone.

A IS NOT CORRECT– Reduce the T302 timer (Service Parameter TimerT302_msec) to indicate end of dialing, but do not set it lower than 4 seconds to prevent premature transmission of the call before the user is finished dialing.

upvoted 3 times

Jetnor 4 years, 5 months ago

both A and C are correct

International and Variable-Length Route Patterns

International destinations are usually configured using the ! wildcard, which represents any quantity of digits. For example, in North America the route pattern 9.011! is typically configured for international calls. In most European countries, the same result is accomplished with the 0.00! route pattern. The ! wildcard is also used for deployments in countries where the dialed numbers can be of varying lengths. In such cases, Unified CM does not know when the dialing is complete and will wait for 15 seconds (by default) before sending the call. You can reduce this delay in any of the following ways:

– Reduce the T302 timer (Service Parameter TimerT302_msec) to indicate end of dialing, but do not set it lower than 4 seconds to prevent premature transmission of the call before the user is finished dialing.

– Configure a second route pattern followed by the # wildcard (for example, 9.011!# for North America or 0.00!# for Europe), and instruct the users to dial # to indicate end of dialing. This action is analogous to hitting the "send" button on a cell phone.

upvoted 1 times

DaKenjee 2 years, 1 month ago

Answer A should not be considered, because it shorts the time for entering a phone number on keypad.
Goal should be T302 on higher value and phone number should be marked as complete by # -> Answer C

upvoted 1 times

Jetnor 4 years, 5 months ago

Actually only C because alterantive A says less then 4 seconds so it is not correct

upvoted 8 times

XalaGyan 4 years ago

You are right 99% but Jetnor is correct a 100%. Lowering the T302 Interdigit Timer is not a best practice, but # Dialing is.

upvoted 2 times

rishik 4 years, 5 months ago

A is correct. Default T302 timer is 15 seconds so unless its reduced people while dialing international numbers (which are normally variable) would take very long to connect.

upvoted 1 times

rishik 4 years, 5 months ago

Update should be C

upvoted 2 times

BarryR 4 years, 5 months ago

C is correct

upvoted 2 times

Question #164 *Topic 1*

An engineer configures local route group names to simplify a dial plan. Where does the engineer set the route groups according to the local route group names that are configured?

A. CSS

B. route pattern

C. device pool

D. route list

Correct Answer: *C*

Community vote distribution

C (100%)

micbosh Highly Voted 3 years, 6 months ago

I guess its device pool!

upvoted 12 times

Piji Highly Voted 1 year, 4 months ago

Selected Answer: C

C. is correct answer

upvoted 5 times

TeeKay25 Most Recent 7 months, 3 weeks ago

Selected Answer: C

C. is correct answer

upvoted 2 times

msully 2 years, 2 months ago

CLCOR 350-801 Official Cert Guide, p. 453

Step 1. Configure local route group names

Step 2. Associate a local route group with a device pool

Step 3. Add a local route group to a route list

upvoted 3 times

DaKenjee 1 year, 1 month ago

Answer C is correct

CSS is wrong needs more then that -> CSS > Route pattern > Route List > Route Group

Route pattern is wrong, only address Gateway or Route List

Route list could be correct it allways includes a Route Group, but we talk about specific name >Local Route Group<

Device Pool mentions explicit Local Route Group

upvoted 2 times

basscov 2 years, 4 months ago

When you add Local Route Group Names on CUCM, they populate in Device Pool. So correct answer is C -device pool

upvoted 1 times

roony83 3 years, 3 months ago

I feel D. route list is correct, because question asked is where does the engineer set the route group.

upvoted 2 times

Sharky1066 1 year, 10 months ago

The configuration on the route list is used to specify which Local Route Group name to use. Local Route Groups names automatical appear in every Device Pool. You then have to apply or 'set' each individual Local Route Name with a Route Group. So the answer is C

upvoted 1 times

CollabGuy 3 years, 1 month ago

No. The question is regarding LOCAL Route Group, which is configured on the device pool. Check infamous476 answer

https://www.cisco.com/c/en/us/td/docs/voice_ip_comm/cucm/admin/10_0_1/ccmfeat/CUCM_BK_F3AC1C0F_00_cucm-features-services-guide-

100/CUCM_BK_F3AC1C0F_00_cucm-features-services-guide-100_chapter_0100110.html
upvoted 4 times

khader09 3 years, 4 months ago
C Device Pool
upvoted 2 times

enashash 3 years, 4 months ago
C Device pool
upvoted 2 times

rishik 3 years, 5 months ago
Definitely Device Pool is the place where you map the local route group

So correct answer is C
upvoted 4 times

infamous476 3 years, 6 months ago
Correct answer is C:

Use the System > Device Pool menu option in Cisco Unified Communications Manager Administration to configure the Local Route Group setting for the device pools in the Cisco Unified Communications Manager implementation. For each device pool that you configure, specify a route group to use as local route group for that device pool. For each device pool, users may also configure the Called Party Transformation CSS for the devices in that device pool.

https://www.cisco.com/c/en/us/td/docs/voice_ip_comm/cucm/admin/10_0_1/ccmfeat/CUCM_BK_F3AC1C0F_00_cucm-features-services-guide-100/CUCM_BK_F3AC1C0F_00_cucm-features-services-guide-100_chapter_0100110.html
upvoted 4 times

Question #165 *Topic 1*

A collaboration engineer must configure Cisco Unified Border Element to support up to five concurrent outbound calls across an Ethernet link with a bandwidth of
160 kb to the Internet Telephony Service Provider.
Which set of commands allows the engineer to complete the task without compromising voice quality?

A. dial-peer voice 1 voip translation-profile outgoing Strip9 max-conn 5 destination-pattern 91[2-9]..[2-9]ג¦€ג€¦$ session protocol sipv2 session target ipv4:142.45.10.1 dtmf-relay rtp-nte sip-notify sip-kpml

B. dial-peer voice 1 voip translation-profile outgoing Strip9 max-conn 5 destination-pattern 91[2-9]..[2-9]ג¦€ג€¦$ session protocol sipv2 session target ipv4:142.45.10.1 dtmf-relay rtp-nte sip-notify sip-kpml codec ilbc mode 20

C. dial-peer voice 1 voip translation-profile outgoing Strip9 max-conn 5 destination-pattern 91[2-9]..[2-9]ג¦€ג€¦$ session protocol sipv2 session target ipv4:142.45.10.1 dtmf-relay rtp-nte sip-notify sip-kpml codec aacld

D. dial-peer voice 1 voip translation-profile outgoing Strip9 max-conn 5 destination-pattern 91[2-9]..[2-9]ג¦€ג€¦$ session protocol sipv2 session target ipv4:142.45.10.1 dtmf-relay rtp-nte sip-notify sip-kpml codec mp4a-latm

Correct Answer: *A*

Community vote distribution

A (100%)

DaKenjee Highly Voted 7 months, 1 week ago

Selected Answer: A

As Omitted explains:

default for voip Dial-peer is G.729, which suits

all other dial-peers have same value, but use different codecs, which consume more bandwith

upvoted 5 times

G0y0 Most Recent 3 months, 1 week ago

Selected Answer: D

The problem is, if include 802.1Q tag or not in the ethernet header.

g729 Without 802.1q tag:

[20 bytes(payload) +40 bytes (ipheader) + 22 bytes (ethernet)] * 50 pps * 8bits/byte = 31200

g720 with 802.1q tag

[20 bytes(payload) +40 bytes (ipheader) + 22 bytes (ethernet)] * 50 pps * 8bits/byte = 32800

ILBC without 802.1Q tag:

[38 bytes (payload) + 40 bytes (ipheader) + 18 bytes (ethernet)] * 50 pps * 8bits/byte = 38400 bps

ILBC with 802.1q tag:

[38 bytes (payload) + 40 bytes (ipheader) + 22 bytes (ethernet)] * 50 pps * 8bits/byte = 40000 bps

it is there the homework, with or without 802.1q: I do not know if A or B

upvoted 1 times

G0y0 3 months, 1 week ago

Now, in my experience, I have worked with Gateways, CUBE's, I don't recall having to tag with 802.1q. Normally, we configure the access or distribution switch port either in routable mode or in switchport access mode without 802.1q. So, based on what I know and have done in my career, I wouldn't include the four bytes of 802.1q in the calculation, and in those cases, I would lean toward answer A. But be careful, that doesn't mean it's a general rule.

upvoted 1 times

G0y0 3 months, 1 week ago

Perhaps taking in count that it is a link to another organization, it is more suitable to not consider the 802.1Q. In fact, inside a LAN between switches and trunks, there could be suitable 802.1q to do intervlan trunking and stuff like that. So in this case, probably I would select A again.

upvoted 1 times

Gary1968 6 months, 2 weeks ago

Selected Answer: B

If the correct answer points to ILBC, shouldn't B be the correct answer?

upvoted 1 times

Piji 10 months, 2 weeks ago

Selected Answer: A

A. is correct answer.

upvoted 4 times

jmer311 1 year, 8 months ago

If you refer to Cisco's Ethernet Bandwidth table, only ilbc mode 30 would be able to run 5 concurrent calls on that link so answer should be A. As noted 5 x 31.2 = 156kb. Source: https://www.cisco.com/c/en/us/support/docs/voice/voice-quality/7934-bwidth-consume.html

upvoted 2 times

Omitted 1 year, 1 month ago

You linked the right document but might have been looking in the wrong spot for ilbc which should be 38.4*5 = 192kb. A should be correct as it uses the default codec of G.729 and your math checks out (5*31.2 - 156kb).

upvoted 3 times

F3rnando 1 year, 10 months ago

https://www.cisco.com/c/en/us/support/docs/voice/voice-quality/7934-bwidth-consume.html

G.729: 31.2×5= 156kb

ilbc: 38.4×5= 192kb

so its A for me

upvoted 4 times

brianccie 1 year, 11 months ago

iLBC is correct and acceptable answer - The internet Low Bit Rate Codec (iLBC) is designed for narrow band speech and results in a payload bit rate of 13.33 kbits per second for 30-millisecond (ms) frames and 15.20 kbits per second for 20 ms frames.

When the codec operates at block lengths of 20 ms, it produces 304 bits per block, which is packetized as defined in RFC 3952. Similarly, for block lengths of 30 ms it produces 400 bits per block, which is packetized as defined in RFC 3952.

The iLBC has built-in error correction functionality to provide better performance in networks with higher packet loss

mode frame_size—The iLBC operating frame mode that will be encapsulated in each packet. Valid entries are 20 (20ms frames for 15.2kbps bit rate) or 30 (30ms frames for 13.33 kbps bit rate). Default is 20

upvoted 1 times

valsrock 2 years, 10 months ago

Each call must be equal or less than 32 kbps, so iLBC is the correct

upvoted 1 times

spag22500 2 years, 10 months ago

https://www.cisco.com/c/en/us/td/docs/ios-xml/ios/voice/cube/configuration/cube-book/cube-codec-basic.html

upvoted 1 times

spag22500 2 years, 10 months ago

but iblc 20 it's 38,4 kbps => x5 = 192 kps => more than 160 kps

correct answer is not "A"?

Becouse without codec value in the dial-peer, by default it's use G729, and G729 is 31.2 kps => x5 = 156 kps

upvoted 11 times

Abhishek1610 2 years, 5 months ago

So the answer is option "A"?

upvoted 2 times

CollabGuy 2 years, 7 months ago

Damn. I chose iLBC instinctively, but I also think you are right. g729r8, 20-byte payload is configured by default and total would be 156kbps. Tricky question...

upvoted 5 times

Mli2604 2 years, 5 months ago

i also guess it is A - anyone else confirm this ?
upvoted 2 times

XalaGyan 2 years, 3 months ago
you nailed it with the explanation. thank you
upvoted 1 times

Griswald 2 years, 11 months ago
mode frame_size—The iLBC operating frame mode that will be encapsulated in each packet. Valid entries are 20 (20ms frames for 15.2kbps bit rate) or 30 (30ms frames for 13.33 kbps bit rate). Default is 20.•bytes payload_size—Number of bytes in an RTP packet. For mode 20, valid values are 38 (default), 76, 114, 152, 190, and 228. For mode 30, valid values are 50(default), 100, 150, and 200
upvoted 1 times

Question #166 *Topic 1*

Which settings are needed to configure the SIP route pattern in Cisco Unified Communications Manager?

A. pattern usage, IPv6 pattern, and SIP trunk/Route list

B. pattern usage, IPv4 pattern, IPv6 pattern, and description

C. pattern usage, IPv4 pattern, and SIP trunk/Route list

D. SIP trunk/Route list, description, and IPv4 pattern

Correct Answer: *C*

Community vote distribution

C (100%)

DaKenjee 1 year, 7 months ago

Selected Answer: C

gerald7 answered it -> C is correct. Check required in this Link

upvoted 3 times

Nkoundzi 1 year, 9 months ago

D is correct

upvoted 1 times

santiagof 1 year, 7 months ago

Do you really think that description is a required field?

upvoted 2 times

Littlelarry123 8 months, 3 weeks ago

you funny for that

upvoted 1 times

gerald7 2 years, 9 months ago

C is correct. Check required in this Link

https://www.cisco.com/c/en/us/td/docs/voice_ip_comm/cucm/admin/10_0_1/ccmcfg/CUCM_BK_C95ABA82_00_admin-guide-100/CUCM_BK_C95ABA82_00_admin-guide-100_chapter_0100111.pdf

upvoted 4 times

devadarshan91730 2 years, 11 months ago

answer is B.

pls refer page : 2

https://www.cisco.com/c/en/us/td/docs/voice_ip_comm/cucm/admin/10_0_1/ccmcfg/CUCM_BK_C95ABA82_00_admin-guide-100/CUCM_BK_C95ABA82_00_admin-guide-100_chapter_0100111.pdf

upvoted 1 times

devadarshan91730 2 years, 11 months ago

i take it back. Answer is C as SIP trunk / route list is REQUIRED attribute.

upvoted 4 times

F3rnando 2 years, 10 months ago

B,D are eliminated because description is not a required field.

C looks valid for me

upvoted 3 times

somedudebob 3 years, 2 months ago

Correct answer is D, pattern usage does not exist in configuration options.

upvoted 1 times

somedudebob 3 years, 2 months ago

Infact I did not read the question correctly, The correct answer is C

upvoted 5 times

Question #167 *Topic 1*

Which call routing pattern is used for phone numbers that are in the E.164 format?

A. \+.! Route Pattern

B. \+.! Translation Pattern

C. /+.! Route Pattern

D. \+1.[2-9]XX[2-9]XXXXXX Called Party Transformation Pattern

Correct Answer: *A*

Community vote distribution

A (100%)

Tlhoka Highly Voted 4 years, 6 months ago

Answer should be A

upvoted 15 times

c37e2aa Most Recent 4 months, 3 weeks ago

Selected Answer: D

CUCM will look for translation pattern and CPTP first, so route pattern is not a valid answer.

As CPTP is the best match, it will be used.

upvoted 1 times

Russel1972 3 months, 1 week ago

Transformation pattern doesn't do routing only manipulates number, can't be D

upvoted 1 times

JoeC716 7 months, 3 weeks ago

Selected Answer: A

A is the only correct answer in this scenario for a route pattern.

upvoted 1 times

DaKenjee 2 years, 1 month ago

Selected Answer: A

Answer is A.

Questin deals about call routing pattern, i would assume in general call routing,
but Route pattern is closer on E164 calls, which are adressing an outsite call to ITSP

B would suit, if it prepares it for a route pattern, which handels the call to ITSP, but we talk about call routing pattern
C is wrong because of Slash, instead of backslash
D is for NANP dial plan -> 10 digits, European dialplan have variable length

upvoted 2 times

Piji 2 years, 4 months ago

Selected Answer: A

A. is correct answer.

upvoted 2 times

zzamchoi 2 years, 4 months ago

Selected Answer: A

Answer is A.

upvoted 2 times

devadarshan91730 3 years, 5 months ago

Answer is A

\+.! Route pattern

upvoted 1 times

Botikus 4 years, 2 months ago

for CUCM there is no difference between patterns so if number start with +1 -- D is the best match. But if user dials not US number? then the correct answer is A.

upvoted 2 times

jherlitzke 4 years, 4 months ago

Seems to be A:

https://subscription.packtpub.com/book/networking_and_servers/9781849684323/1/ch01lvl1sec10/implementing-e-164-route-patterns-and-partitions

upvoted 3 times

rishik 4 years, 5 months ago

A is correct

upvoted 2 times

BarryR 4 years, 5 months ago

Answer is A

upvoted 2 times

Question #168 *Topic 1*

Which two recommendations are made to optimize Cisco UCM configuration to reduce the number of toll fraud incidents in an organization? (Choose two.)

A. Classify all route patterns as on-net and prohibit on-net to on-net call transfers in Cisco UCM service parameters.

B. Classify all route patterns as off-net and prohibit off-net to off-net call transfers in Cisco UCM service parameters.

C. Classify all route patterns as on-net or off-net and prohibit off-net to off-net call transfers in Cisco UCM service parameters.

D. Inbound CSS on any gateway typically should have access to internal destinations and PSTN destinations.

E. Inbound CSS on any gateway typically should have access to internal destinations only and not PSTN destinations.

Correct Answer: *CE*

Community vote distribution

CE (100%)

DaKenjee Highly Voted 7 months, 1 week ago

Selected Answer: CE

Answer C + E

E is defiantly correct. where toll fraud on unity occurs, with inbound css to pstn + voicemailbox use an alternate number

B+C are nearly the same, but i stick with C,
because a route pattern do not have to direct only to ITSP,
could be Unity, internal Fax Server, Contact Center and so on -> onnet

upvoted 8 times

santiagof Most Recent 7 months, 4 weeks ago

B & E is correct,

upvoted 2 times

FCBear 1 year ago

Should be C&E

upvoted 1 times

Question #169 *Topic 1*

Which wildcard must an engineer configure to match a whole domain in SIP route patterns?

A. .

B. !

C. @

D. *

Correct Answer: *D*

Reference:

https://www.cisco.com/c/en/us/td/docs/voice_ip_comm/cucm/admin/10_0_1/ccmcfg/CUCM_BK_C95ABA82_00_admin-guide-100/CUCM_BK_C95ABA82_00_admin-guide-100_chapter_0100111.html

timmyz 10 months, 1 week ago

Answer is D. There is no @ for a wildcard in SIP route patterns.

Because no default SIP route patterns exist in Cisco Unified Communications Manager, you must set them up. Domain name examples are cisco.com, my-pc.cisco.com, *.com, rtp-ccm[1-5].cisco.com. Valid characters for domain names are [, -, ., 0-9, A-Z, a-z, *, and].

IPv4 address examples 172.18.201.119 or 172.18.201.119/32 (explicit IP host address); 172.18.0.0/16 (IPsubnet); 172.18.201.18/21 (IP subnet).

Valid characters for IP addresses: 0-9, ., and /

upvoted 4 times

Question #170 *Topic 1*

Multiple route patterns match a number. How does Cisco Unified Communications Managers determine which pattern to use?

A. the one that comes first in numerical order

B. the one with the longest match

C. the one with the closest match

D. the one that discards everything PreDot

Correct Answer: *C*

Reference:

https://www.cisco.com/c/en/us/support/docs/voice-unified-communications/unified-communications-manager-callmanager/13920-call-routing.html#bcr

Community vote distribution

C (100%)

eddiechang Highly Voted 4 years, 9 months ago

the anwser is C

Cisco CallManager performs Closest-Match Routing.

upvoted 7 times

b3532e4 Most Recent 10 months, 3 weeks ago

C

When a number is dialed, the Cisco Unified Communications Manager uses closest-match

logic to select which pattern to match from among all the patterns in its call-routing table.

upvoted 1 times

Ahochau 2 years ago

Selected Answer: C

Technically, the most specific match. If there are identical patterns in multiple partitions it will take the pattern in the first matched partition.

upvoted 1 times

Pavel71 2 years, 2 months ago

Selected Answer: C

the anwser is C

upvoted 1 times

GUEZI 4 years, 10 months ago

B. The one with the longest match

upvoted 2 times

JS8 4 years, 10 months ago

C closest match would be correct

If the number is 1234567890

123XXXXXXX would be the longest match

1234! would be the closest match and would be used

From the reference: Cisco CallManager performs Closest-Match Routing.

https://www.cisco.com/c/en/us/support/docs/voice-unified-communications/unified-communications-manager-callmanager/13920-call-routing.html#bcr

upvoted 17 times

Question #171 *Topic 1*

Which action prevents toll fraud in Cisco UCM?

A. Implement toll fraud restriction in the Cisco IOS router.

B. Implement route patterns in Cisco UCM.

C. Allow off-net to off-net transfers.

D. Configure ad hoc conference restriction.

Correct Answer: *D*

Community vote distribution

D (100%)

b3532e4 10 months, 3 weeks ago

D

Drop Ad Hoc Conference, prevents toll fraud (where an internal conference controller disconnects from the conference while outside callers remain connected). The service parameter settings specify conditions under which an ad hoc conference gets dropped.

upvoted 1 times

JoeC716 1 year, 1 month ago

Selected Answer: D

D is correct

upvoted 1 times

auswar3ft 2 years, 4 months ago

Selected Answer: D

ad hoc conf restriction prevents Toll Fraud

upvoted 2 times

AJBELL14 2 years, 11 months ago

Selected Answer: D

D is the right answer

upvoted 2 times

Omitted 3 years, 1 month ago

Selected Answer: D

I would say D. Configuring drop ad hoc prevents toll fraud.

https://www.cisco.com/c/en/us/td/docs/voice_ip_comm/cucm/admin/12_0_1/featureConfig/cucm_b_cucm-feature-configuration-guide_1201/cucm_b_cucm-feature-configuration-guide_1201_chapter_010000.html

upvoted 2 times

ciscogeek 3 years, 1 month ago

Selected Answer: D

Drop Ad Hoc Conference, prevents toll fraud (where an internal conference controller disconnects from the conference while outside callers remain connected). The service parameter settings specify conditions under which an ad hoc conference gets dropped.

Never—The conference does not get dropped. (We recommend that you use the default option to avoid unintentional termination of a conference).

When No OnNet Parties Remain in the Conference—The system drops the active conference when the last on-network party in the conference hangs up or drops out of the conference. Unified Communications Manager releases all resources that are assigned to the conference.

Note

Drop Ad Hoc Conference feature in an ILS deployment will not drop the parties when it set at When No OnNet Parties Remain in the Conference because the route patterns learned are classified as On Net.

When Conference Controller Leaves—The active conference terminates when the primary controller (conference creator) hangs up. Unified Communications Manager releases all resources that are assigned to the conference.

upvoted 3 times

Question #172 *Topic 1*

A user forwards a corporate number to an international number. What are two methods to prevent this forwarded call? (Choose two.)

A. Set the Call Classification to OnNet for the international route pattern.

B. Block international dial patterns in the SIP trunk CSS.

C. Configure a Forced Authorization Code on the international route pattern.

D. Set Call Forward All CSS to restrict international dial patterns.

E. Check Route Next Hop By Calling Party Number on the international route pattern.

Correct Answer: *CD*

Community vote distribution

CD (100%)

Janu82 Highly Voted 3 years, 6 months ago

Answers should be C and D.

Configure FAC for International Dialing Patterns

Set Call Forward All to restrict International patterns

upvoted 18 times

rishik Highly Voted 3 years, 5 months ago

CD is correct answer

upvoted 8 times

auswar3ft Most Recent 11 months, 1 week ago

B is wrong. even though it was my first choice. but blocking international calls on the CSS would effectively block international calls for all users on that CSS.

upvoted 1 times

DaKenjee 1 year, 1 month ago

Selected Answer: CD

Forced Authorization Code, would require user input, which interrupts a Call forward All on Directory number to international

and a restricted CSS, always stops callrouting on specific patterns

D. Set Call Forward All CSS to restrict international dial patterns.

upvoted 3 times

Piji 1 year, 4 months ago

Selected Answer: CD

Corect answer is C,D.

upvoted 3 times

twelve12 3 years ago

Correct answers are C and D.

upvoted 3 times

tebzades 3 years, 3 months ago

cd are correct

upvoted 4 times

Griswald 3 years, 5 months ago

C and D

upvoted 4 times

jherlitzke 3 years, 5 months ago

I would tend to agree on C and D as well. If B was used you would block all international calls even if dialed straight up.

upvoted 5 times

Sharky1066 1 year, 10 months ago

I agree with you. The Trunk CSS however has nothing to do with the outbound call flow when setting a call forward on an IP handset. The Trunk CSS provides class of service for inbound calls in to CUCM (not outbound)

upvoted 1 times

BarryR 3 years, 5 months ago

Correct answer is C) Configure a Forced Authorization Code on the international router pattern... and E) Set call forward All CSS to restrict international dial patterns

upvoted 2 times

BarryR 3 years, 5 months ago

Opps I meant D) Set call forward All CSS to restrict international dial patterns

upvoted 4 times

Question #173 *Topic 1*

Calls are being delivered to the end user in the globalized format. Where does an engineer configure the calling number into a localized format?

A. route pattern

B. service parameters

C. IP phone

D. gateway

Correct Answer: *C*

Community vote distribution

C (83%) D (17%)

norealchaos Highly Voted 4 years, 10 months ago

The answer is C -Ip Phone - For the inbound call flow, globalization is done at ingress in the sip trunk/ gateway and localization at egress in the phone level with a calling party transformation.

https://www.cisco.com/c/en/us/td/docs/voice_ip_comm/cucm/srnd/collab11/collab11/dialplan.html#16685

upvoted 12 times

Komy Highly Voted 3 years, 12 months ago

I would say the answer is "Gateway". Here is my reasoning:

- Localization can be configured on Gateway and/or Device pool and/or phone level.

- However, In SRND 12, Under (Phone calling party number Localization) , the below statement shows that Cisco recommends placing the configuration in device pool and not on the phone:

"The Calling Party

Transformation Pattern is placed in a partition included in the destination phone's Calling Party Transformation Pattern CSS, configured at the device-pool level."

And because we do not have "device pool" as an available answer, i would go with "Gateway" because you can't just go to each and every single phone to apply config

upvoted 5 times

mtc_mnd Most Recent 3 weeks, 3 days ago

The question seems to point at a single end user (not end users).

Wouldn't this be enough to consider "IP Phone" the correct answer?

upvoted 1 times

decdca7 7 months, 2 weeks ago

Selected Answer: D

In the inboud configuration of the Gateway

upvoted 2 times

b3532e4 10 months, 3 weeks ago

D .gateway

upvoted 3 times

Komy 1 year, 2 months ago

Selected Answer: D

I would say the answer is "Gateway". Here is my reasoning:

- Localization can be configured on Gateway and/or Device pool and/or phone level.

- However, In SRND 12, Under (Phone calling party number Localization) , the below statement shows that Cisco recommends placing the configuration in device pool and not on the phone:

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And because we do not have "device pool" as an available answer, i would go with "Gateway" because you can't just go to each and every single phone to apply config
upvoted 3 times

Mert_kerna 2 years, 5 months ago
Selected Answer: C
The answer is C, because the calling party transformation CSS can only be applied to an IP phone or a device pool. Device pool is not a given answer, so, the answer is IP Phone.

The associated transformation pattern, however, can be applied directly on the device pool or the gateway.

The question is asking where the configuration can be applied. The final configuration (CSS) "Configuration" can only be applied to the Device Pool or an IP Phone. Not the gateway.
-
upvoted 3 times

Mert_kerna 2 years, 5 months ago
Additionally, the device pool is the only configuration that goes on to the Gateway. The device pool IS what the configuration is applied to [Or an IP Phone] (Not the actual configuration) And by Gateway, this question is referring to something like a MGCP gateway, not necessarily a SIP trunk etc - Both of which would be correct in this circumstance.
upvoted 1 times

DaKenjee 2 years, 7 months ago
Selected Answer: C
Answer is C - Komy, explained it, with SRND12
I searched and found a similar answer in SRND

route pattern is wrong, directs from CUCM away
service parameters, is possible -> service parameters to globalize calling party numbers on all devices clusterwide
IP phone, is possible with calling party transformation CSS
gateway, is possible by varios manipulations via CLI

i stick with Answer C, because gateway is ealiest ingrees in this callflow: ITSP > Gateway > Callmanager > Phone

SRND:
Thus, the guiding principle is:
Accept localized forms upon call ingress, and globalize them;
route the call based on the globalized form;
and localize the call to comply to the form required by the destination.
https://www.cisco.com/c/en/us/td/docs/voice_ip_comm/cucm/srnd/collab12/collab12/dialplan.html?bookSearch=true#48778
upvoted 2 times

bmne 2 years, 12 months ago
Answer is D. https://www.cisco.com/c/en/us/td/docs/voice_ip_comm/cucm/srnd/collab11/collab11/dialplan.html#16685
The globalization of the calling party number should be implemented by using the Incoming Calling Party Settings configured either on the gateway directly or in the device pool controlling the gateway.
upvoted 3 times

mnvx 3 years, 5 months ago
The question asks where a configuration is done.
The configuration in question is a calling number: wouldn't that be simply a DN on an IP Phone?

Is this explanation too simple?
upvoted 1 times

virtu 4 years, 5 months ago
Correct is C, localization of egress call make on ip phone, in the calling-party transformation patter that applicable to phone. Cisco`s digital learning and CLCRO cource says that
upvoted 2 times

mzougari 4 years, 10 months ago

Answer: C
Pre-transformation calling party numbers should be globalized and routable before modifying at phone: globalize on ingress (at gateway), localize on egress (at phone)
upvoted 2 times

Mr_Kokonut 4 years, 10 months ago
C. You cannot modify the called-party number during call egress (gateway), when a call is delivered to a phone by CUCM. Calling Party XFORM (\+1.!, Pre-Dot) is applied to phone Device pool via CSS and maintains full E.164 in missed/received call directory.
upvoted 4 times

jherlitzke 4 years, 11 months ago
sorry forgot link: https://www.cisco.com/c/en/us/td/docs/voice_ip_comm/cucm/admin/10_0_1/ccmfeat/CUCM_BK_F3AC1C0F_00_cucm-features-services-guide-100/CUCM_BK_F3AC1C0F_00_cucm-features-services-guide-100_chapter_01010.html
upvoted 1 times

jherlitzke 4 years, 11 months ago
Check out this link, am I reading at step 12 that maybe both gateway and phone configuration is where you set the call routing transformation pattern? Would that mean both C and D could be correct? My gut tells me gateway, but... just a thought.
upvoted 1 times

julioccamara 4 years, 7 months ago
"Phone" Configuration is different of the "IP PHONE" Configuration
upvoted 2 times

Griswald 4 years, 11 months ago
Core routing in the globalized dial plan approach is based on routing +E.164 patterns so that the native dialing habit for this dial plan approach is global +E.164 dialing.

Unified CM's translation patterns are used to convert localized user input as dialed from phones, to the global +E.164 form used to route the calls within the Unified Communications system.
upvoted 2 times

rishik 4 years, 11 months ago
Not sure how can someone normalize numbers in IP phone.

Gateway should be the correct answer
upvoted 3 times

mzougari 4 years, 10 months ago
Can be configured on endpoint or device pool (check “Use Device Pool Calling Party Transformation CSS” on endpoint config)
upvoted 3 times

Question #174 *Topic 1*

Which method is used to avoid toll fraud with Cisco Unified Communications Manager calls?

A. call policy service

B. TOLLFRAUD_APP

C. default zone access rules

D. class of service

Correct Answer: *D*

Griswald Highly Voted 1 year ago

To prevent toll fraud in a Cisco Collaboration network, you can employ various tools:

CUCM class of service (CoS)

Voice gateway toll fraud prevention application

Voice gateway class of restriction (CoR)

Cisco Unity Connection restriction rules

upvoted 10 times

Question #175

Topic 1

Which two configuration elements are part of the Cisco Unified Communications Manager toll-fraud prevention? (Choose two.)

A. SIP trunk security profile

B. Calling Search Space

C. SUBSCRIBE Calling Search Space

D. feature control policy

E. partition

Correct Answer: *BE*

ocero 8 months, 2 weeks ago

Partition and CCS

https://www.ciscopress.com/articles/article.asp?p=2218297&seqNum=10

upvoted 3 times

Question #176 *Topic 1*

How are E.164 called-party numbers normalized on a globalized call-routing environment in Cisco Unified Communications Manager?

A. Normalization is achieved by stripping or translating the called numbers to internally used directory numbers.

B. Normalization is achieved by setting up calling search spaces and partitions at the SIP trunks for PSTN connection.

C. Call ingress must be normalized before the call being routed.

D. Normalization is not required.

Correct Answer: *A*

Community vote distribution

A (100%)

G0y0 3 months, 3 weeks ago

Selected Answer: A

After think and think and think, about called numbers, E.164 format with the + prefix is used for external destinations. Therefore, called-number normalization is the result of globalization.

Now, Internal DNs are used for internal destinations. Normalization is achieved by stripping or translating the called number to internally used DNs.

■ External to internal:

■ Calling-party number: E.164

■ Called-party number: Directory number

■ External to external (if applicable):

■ Calling-party number: E.164

■ Called-party number: E.164

■ Internal to internal:

■ Calling-party number: Directory number

■ Called-party number: Directory number

■ Internal to external:

■ Calling-party number: E.164

■ Called-party number: E.164

upvoted 1 times

iExpo_91 8 months, 4 weeks ago

Selected Answer: A

While both A and C are correct, however the question is asking "how is it being normalize"

a good example will be assuming on cucm line/extensions you are using a 11 digit E164 format with +1 but the PSTN for incoming digit they are forwading you 10 digit, you will have to normalize the number by perfoming translation rules

voice translation-rule 1

rule 1 /[0-9].........$/ /+1\0/

So A is the most correct answer.

upvoted 3 times

DaKenjee 1 year, 1 month ago

Selected Answer: A

Answer is A - like devadarshan91730 mentioned

Normalized called-party numbers: E.164 format with the +prefix is used for external destination. Therefore, called-number normalization is the result of globalization. Internal directory numbers are used for internal destinations. Normalization is achieved by stripping or translating the called numbers to internally used directory numbers.

upvoted 3 times

G0y0 1 year, 7 months ago

The answer A says "stripping or translating the called numbers to internally used directory numbers" is closer to Localization than Globalization, so I would not be so sure that is the correct answer. Furthermore, prefixing of the number with "+" is called the globalized format of an E.164 number and is defined in the Internet Engineering Task Force RFC 2806. The best practice, the incoming call must be globalized, next routed and finally Localized. Now, E.164 must be normalized to +E.164 on a globalized environment, for example, from a ISDN BRI, you do no need to prefix 011 for international

call, just you can dial the E.164 format without access codes, and setting the TON to International but it is not globalization, so D can not be correct neither. If it were up to me, I would rather to choose C instead A.
upvoted 1 times

G0y0 1 year, 7 months ago
Well, I think I am changing my mind. Another think I would like to comment that E.164 number, in order to ITU-T E.164, is a external called-party number (without "+").
If it is a External incoming call (from GW), calling- and called-party numbers are usually provided in localized E.164 format. If Called-party number is internal, A. could be the best answer. If Called-party number is external (offnet to offnet call), C. could be the best answer.
If it is a Internal Outgoing Call (from internal IP phone DN) to external Called-Party number change the called number to +E.164 format if any other format was used (according to local dial rules, enterprise access codes, etc.) and this case, the best answer could be C.
In order of all of above, the closest answer would be A because it has a more specific scenario (PSTN Gateway to Internal DN). And C. is more general. So finally I prefer A.
upvoted 2 times

basscov 2 years, 4 months ago
Normalized called-party numbers: E.164 format with the +prefix is used for external destination. Therefore, called-number normalization is the result of globalization. Internal directory numbers are used for internal destinations. Normalization is achieved by stripping or translating the called numbers to internally used directory numbers. So answer A
upvoted 1 times

Komy 2 years, 5 months ago
Answer should be (A),
How can we say (D: no normalization is required) ?. IP phones within an organization will have DN configured to reach internal users. If we do not normalize the called number, should we expect to configure all phones with a Global number instead of a normal DN ?
upvoted 1 times

G0y0 1 year, 6 months ago
Because the question is not specifying if the called party number is external or internal, how we can be sure (speculation) that the called number is a 4 digit DN, even it could has implementations where DN's are in E.164 format. The question just says there is a E.164 called-party number, no anymore
upvoted 1 times

G0y0 1 year, 6 months ago
about if we should expect DN's in global format, there are destination
DN whit aliases as enterprise alternate number or +E.164 alternate number using Global Dial
Plan Replication (GDPR) and Inter-cluster Lookup Service (ILS).
The question is no so easy enough.
upvoted 1 times

devadarshan91730 2 years, 6 months ago
Answer is A.
https://community.cisco.com/t5/collaboration-voice-and-video/basics-of-globalized-call-routing/ta-p/3121674
upvoted 3 times

somedudebob 2 years, 8 months ago
I think the answer here is D, WE are asked about Called party numbers, so A is invalid as we would only strip or translate to internal if the call was inbound,
upvoted 2 times

CiscoCUCMKing 2 years, 10 months ago
I agree, the answer should be D since normalization of called party numbers should not be required in a globalized call routing environment.
upvoted 1 times

CiscoCUCMKing 2 years, 10 months ago
Actually, the answer could be A, if the question is referring to incoming call routing and the normalization of +e164 called party numbers to local DNs as per this document states
https://community.cisco.com/t5/collaboration-voice-and-video/basics-of-globalized-call-routing/ta-p/3121674. As usual, it all depends on what Cisco are looking for in this very vaguely phrased question
upvoted 4 times

G0y0 1 year, 6 months ago
The think is that the question does not specify if the called-party number is external or internal. Furthermore, it could have deployments where DN's are in E.164 format and then it would be necessary to do normalization if the localized format is not equal to the E.164 format. So, I am

not so sure about answer A, there is no mention in the question if the called number is just an internal 4 digits number, or external, or even the called DN is in E.164 format.

Now, returning at the first, of there is a called-party number, whether it be internal or external (no specified by the question), in E.164 format, it might not be necessary to normalize.

In the exam I could choose the answer D in order of my interpretation, it could be wrong or it could be right. If someone come up with another interpretation, It will be appreciated and I am looking forward to know more interpretations.

upvoted 1 times

virtu 2 years, 11 months ago

I think the answer is D, because called numbers are already in 164, as stated in question. And what I find in internet.

Number normalization

This term refers to the process of changing numbers to a well-defined, standardized (normalized) format. In this case, all external phone numbers are changed to +E.164 format (that is, the E.164 format with the plus [+] prefix).

upvoted 2 times

Mo_G 3 years ago

Normalized called-party numbers: E.164 format with the +prefix is used for external destination. Therefore, called-number normalization is the result of globalization. Internal directory numbers are used for internal destinations. Normalization is achieved by stripping or translating the called numbers to internally used directory numbers.

Answer is A

upvoted 1 times

Botikus 3 years ago

If it is a globalized envirement, all number should be in E.164 format so no normalization is required.

upvoted 4 times

Question #177

Topic 1

Which user group is targeted by MRA services?

A. production floor users who need wireless mobility in remote areas of the factory

B. mobile workers in a hot desk environment at HQ who log in every morning at possibly a different desk phone

C. on-the-go mobile workforce who connect to corporate phone services using their own mobile device

D. call center agents who dial out to remote customers

Correct Answer: *C*

Currently there are no comments in this discussion, be the first to comment!

Question #178 *Topic 1*

Which Cisco UCM feature is used to determine the maximum bit rate for a call between two video endpoints?

A. partitions

B. locations

C. regions

D. transformations

Correct Answer: *C*

Regions provide capacity controls for Unified Communications Manager multi-site deployments where you may need to limit the bandwidth for certain calls. For example, you can use regions to limit the bandwidth for calls that are sent across a WAN link, while maintaining a higher bandwidth for internal calls. You can use regions to limit the bandwidth for audio and video calls by setting the maximum bitrate for intraregional or interregional calls to whatever the region(s) can provide.

Reference:

https://www.cisco.com/c/en/us/td/docs/voice_ip_comm/cucm/admin/11_5_1/sysConfig/11_5_1_SU1/cucm_b_system-configuration-guide-1151su1/ cucm_b_system-configuration-guide-1151su1_chapter_0111.html

Community vote distribution

C (100%)

MaxG 10 months, 4 weeks ago

Selected Answer: C

Regions = maximum bandwidth per call between two endpoints

Locations = maximum available bandwidth for all calls

upvoted 1 times

Question #179 *Topic 1*

Which feature is enabled by Cisco Mobile and Remote Access?

A. Internal SIP clients registered to Cisco UCM can call external companies

B. Clients such as Cisco Jabber can use call control on Cisco UCM

C. VPN connectivity can be established to Cisco UCM

D. Cisco UCC Express clients can obtain VPN connectivity to Cisco UCC Enterprise

Correct Answer: *B*

Community vote distribution

B (60%) A (20%) C (20%)

Omitted Highly Voted 2 years, 7 months ago

B is the best answer but the wording on the question and answer could be better as MRA isn't enabling CTI control via jabber, that exists for onprem whether you have Expressway or not.

upvoted 8 times

TheBabu Most Recent 8 months, 1 week ago

Selected Answer: B

Worst question working ever, typical Cisco trying to make questions "hard" and ending up just writing nonsense. Had to get that out of my system.

Anyway, that said:
A: MRA is not needed for internal endpoints to call other companies, that's B2B instead.
B: Only possible right answer. Jabber doesn't necessarily need MRA to use call control on CUCM, but it's the one that makes a little bit of sense.
C, D: The whole point of MRA is for clients to connect without using a VPN, so no.

upvoted 2 times

sneff91 9 months, 4 weeks ago

Selected Answer: A

B doesn't require MRA and is possible internally with CUCM alone.
C is just untrue. MRA leverages proxy secure firewall traversal and resembles functionality of a VPN in some ways, but it's still NOT a VPN.
D... I don't even know what they're trying to say this. This just makes no sense.

Honestly, this question sucks. I'm almost inclined to say A because Expressway provides MRA which allows internal SIP calls to be routed from inside one organization, across an unsecure network to another (external) company using the B2B (Business to Business) methodology. Yes, it leaves some of the answer to the imagination, but the other answers just seem to much more wrong...

upvoted 1 times

c6176b5 11 months, 1 week ago

Selected Answer: B

IT is B

upvoted 1 times

ademozipek 1 year, 3 months ago

Selected Answer: B

It's B.

upvoted 1 times

MaxG 1 year, 6 months ago

Not C.
MRA provides a secure connection for Jabber application traffic and other devices with the required capabilities to communicate without having to connect to the corporate network over a VPN.

B is the correct answer:
MRA allows endpoints such as Cisco Jabber to have their registration, call control, provisioning, messaging and presence services provided by Cisco Unified Communications Manager (Unified CM) when the endpoint is outside the enterprise network. The Expressway provides secure firewall traversal and line-side support for Unified CM registrations.

https://www.cisco.com/c/en/us/td/docs/voice_ip_comm/expressway/config_guide/X12-7/exwy_b_mra-deployment/exwy_m_mra-overview-and-planning.pdf

upvoted 1 times

RdTx 1 year, 11 months ago

Selected Answer: C

Looks like C.. rest are wrong

upvoted 1 times

Panda_man 1 year, 11 months ago

there is no VPN when it comes to MRA

upvoted 6 times

Question #180 *Topic 1*

What dialed numbers match this Cisco UCM route pattern?
1[23]XX

A. 1200 through 1300 only

B. 12300 through 12399 only

C. 1200 through 1399 only

D. 1230 through 1239 only

Correct Answer: *C*

Community vote distribution

C (100%)

Testme1235 10 months, 1 week ago

Selected Answer: C

The correct answer is C. 1200 through 1399 only.

The route pattern "1[23]XX" will match any four-digit number that begins with "1" and has a second digit of either "2" or "3", with any value for the third and fourth digits.

Therefore, the following dialed numbers will match this route pattern:

1200
1201
1234
1399
Option A is incorrect because it only includes the numbers between 1200 and 1300, whereas the route pattern matches any number between 1200 and 1399.

Option B is incorrect because it includes numbers with five digits, whereas the route pattern only matches numbers with four digits.

Option D is incorrect because it only includes numbers between 1230 and 1239, whereas the route pattern matches any number between 1200 and 1399.

upvoted 2 times

Panda_man 11 months, 1 week ago

Selected Answer: C

C is true

upvoted 1 times

RdTx 11 months, 3 weeks ago

Selected Answer: C

Correct is C

upvoted 1 times

Omitted 1 year, 10 months ago

Selected Answer: C

I imagine on the test they give you this question in this exact order so you may accidentally pick A

upvoted 1 times

Question #181 *Topic 1*

Cisco UCM delays routing of a call during digit analysis with an overlapping dial plan. How long is the default wait time?

A. 5 seconds

B. 10 seconds

C. 15 seconds

D. 20 seconds

Correct Answer: *C*

Community vote distribution

C (100%)

Omitted 10 months, 3 weeks ago

Selected Answer: C

Default timeout for CUBEs/VGs is 10 seconds

Default timeout for CUCM is 15 seconds (T302)

upvoted 4 times

Question #182 *Topic 1*

An administrator is asked to implement toll fraud prevention in Cisco UCM, specifically to restrict off-net to off-net call transfers. How is this implemented?

A. Use the correct route filters.

B. Implement time-of-day routing.

C. Enforce ad-hoc conference restrictions.

D. Set the appropriate service parameter.

Correct Answer: *D*
Reference:
https://www.ciscopress.com/articles/article.asp?p=2218297&seqNum=10

Anky12 7 months, 3 weeks ago
D is correct - Block off-net to off-net call transfers needs to be set to true.
upvoted 1 times

Question #183 *Topic 1*

Which call flow matches traffic from a Mobile and Remote Access registered endpoint to central call control?

A. Endpoint > Expressway-E > Expressway-C > Cisco Unified CM

B. Endpoint > Expressway-E > Cisco Unified CM

C. Endpoint > Expressway-C > Cisco Unified CM

D. Endpoint > Expressway-C > Expressway-E > Cisco Unified CM

Correct Answer: *A*

Community vote distribution

A (100%)

Omitted 10 months, 3 weeks ago

Selected Answer: A

Some of the expressway questions are so simple.

upvoted 4 times

Question #184 *Topic 1*

How does Cisco Unified Communications Manager perform a digit analysis on-hook versus off-hook for an outbound call from a Cisco IP phone that is registered to Cisco Unified CM?

A. On-hook, Unified CM performs a digit-by-digit analysis; off-hook, Unified CM considers all digits were dialed and does not wait for additional digits.

B. On-hook, Unified CM considers all digits were dialed and does not wait for additional digits; off-hook, Unified CM performs a digit-by-digit analysis.

C. On-hook, by pressing the digits and entering "#" to process the call, Unified CM performs a digit-by-digit analysis; off-hook, Unified CM analyzes all digits as a string.

D. On-hook, no digit analysis is performed; off-hook, Unified CM requires the ג€#* to start the digit analysis.

Correct Answer: *B*

Community vote distribution

B (86%) | 14%

Janu82 Highly Voted 3 years, 6 months ago

The correct answer is B.

upvoted 26 times

rishik Highly Voted 3 years, 5 months ago

B is correct

upvoted 8 times

Stevon Most Recent 3 weeks, 1 day ago

Selected Answer: B

Digit Sending: On-hook sends all digits at once; off-hook sends digits one by one.

Digit Analysis: On-hook, CUCM considers the dialed number complete; off-hook, CUCM analyzes digits sequentially.

Call Initiation: On-hook, the call is initiated immediately after all digits are sent; off-hook, the call is initiated aft

upvoted 1 times

arinpas 8 months, 2 weeks ago

Selected Answer: C

B is correct

upvoted 1 times

H31d1 1 year, 1 month ago

Selected Answer: B

starts en bloc calling when dialing off-hook

upvoted 2 times

Piji 1 year, 4 months ago

Selected Answer: B

Correct answer is B.

upvoted 2 times

AJBELL14 1 year, 5 months ago

Selected Answer: B

B is the right answer. When you dial off hook you can actually feel the changes in the tones digit by digit, whereas during onhook dialing, it just dials everything at once

upvoted 2 times

DEFAULTNERD 3 years ago

Yeah B

upvoted 5 times

ratbat 3 years, 1 month ago

it sounds like B

upvoted 5 times

BarryR 3 years, 5 months ago

Correct answer is B

upvoted 7 times

Grebec94 3 years, 6 months ago

this sounds like A?

upvoted 1 times

Question #185 *Topic 1*

How does an administrator make a Cisco IP phone display the last 10 digits of the calling number when the call is in the connected state, and also display the calling number in the E.164 format within call history on the phone?

A. Configure a translation pattern that has a Calling Party Transform Mask of XXXXXXXXXX.

B. On the inbound SIP trunk, change Significant Digits to 10.

C. Change the service parameter Apply Transformations On Remote Number to True.

D. Configure a calling party transformation pattern that keeps only the last 10 digits.

Correct Answer: *D*

Community vote distribution

D (100%)

Tlhoka Highly Voted 3 years ago

C is the answer

upvoted 8 times

santiagof 7 months, 4 weeks ago

no its not

upvoted 1 times

ratbat Highly Voted 2 years, 7 months ago

D is the answer and here's why:

if your read scenario 2 on the below link, you will know it can only be D

https://blog.collabcert.com/ucm/understanding-the-5-uses-of-calling-party-transformation-pattern-in-ucm/

upvoted 8 times

mtc_mnd Most Recent 3 months ago

Selected Answer: D

Tested this. The answer D works exactly as per question requirements and with Service Parameters Apply Transformations On Remote Number set to False.

upvoted 1 times

fa2451d 3 months, 3 weeks ago

Selected Answer: C

I think it is C.

If you choose D, then you are changing the caller ID that the phone will receive and will be stored in history. The question asks to modify the connected number, which means to modify the caller ID while both parties are talking. On other words, when the called party left the receiver, the phone screen should show 10-digits only. The call history should stay untouched.

Service parameter is doing this trick.

upvoted 1 times

DaKenjee 7 months, 1 week ago

Selected Answer: D

Answer is D

Translationpattern might work, but have to be triggered as called number equals directory number, not suitable

sip-trung can change digits on calling number

Calling Party Transformations on IP Phones

Calling Party Transformation Patterns allow the system to adapt the calling party numbers to different formats.

The most typical use is to adapt from globalized to localized calling party numbers and vice versa.

https://www.cisco.com/c/en/us/td/docs/voice_ip_comm/cucm/srnd/collab11/collab11/dialplan.html

upvoted 3 times

cowell_53 1 year, 4 months ago

should be A

upvoted 1 times

Komy 1 year, 12 months ago

I will go with C: below is captured from SRND 12:

"The outbound call's calling party transformation CSS (also referred to as localization or remote number calling party transformation CSS) can also be used to localize remote connected party information.

To enable this, the advanced service parameter Apply Transformations On Remote Number must be enabled."

upvoted 4 times

pasangawa 2 years ago

C. 'Apply Transformations On Remote Number' allows the display on phone to be shown as the transformed number when connected. That being said, transformation pattern is still needed as combination.

upvoted 1 times

jonycakes 2 years ago

Answer D: On a phone in North America, Cisco Unified Communications Manager uses a calling party transformation to convert +12225551234 to 10 digits before the number displays on the phone; on a phone outside of North America, Cisco Unified Communications Manager may transform the number to only strip the + and to prefix the 00, as in 0012225551234.

Source: https://www.cisco.com/c/en/us/td/docs/voice_ip_comm/cucm/admin/10_0_1/ccmfeat/CUCM_BK_F3AC1C0F_00_cucm-features-services-guide-100/CUCM_BK_F3AC1C0F_00_cucm-features-services-guide-100_chapter_01010.html

upvoted 2 times

rishik 2 years, 11 months ago

Correct answer is D

Just changing the parameter under service parameter to won't change the calling number from E.164 to last 10 digits

upvoted 3 times

rishik 2 years, 11 months ago

Update C

upvoted 2 times

Steeve 2 years, 8 months ago

most of answers that talk about D , to keep the last 10 digits, so im still confusing, and i need help of this question

upvoted 1 times

rich19007 2 years, 2 months ago

Did you ever receive a solid answer to this question? is is C or D?

upvoted 1 times

G0y0 1 year ago

I think it must be both, C and D.

upvoted 1 times

santiagof 7 months, 4 weeks ago

the question is about a phone, if you change a service parameter, is a global configuration, right answer is D

upvoted 2 times

Question #186

Topic 1

Which endpoint feature is supported using Mobile and Remote Access through Cisco Expressway?

A. SRST

B. SSO

C. H.323 registration proxy to Cisco UCM

D. MGCP gateway registration

Correct Answer: *B*

Community vote distribution

B (100%)

arinpas 8 months, 2 weeks ago

Selected Answer: B

B is correct

upvoted 1 times

lpicardin 9 months, 1 week ago

Selected Answer: B

Definitively B

upvoted 1 times

AJBELL14 1 year, 5 months ago

Selected Answer: B

SSO is the right answer

upvoted 2 times

bmne 1 year, 5 months ago

B is the answer.

Beginning with X8.11 Expressway-E will support local SIP and H.323 video

registration

Expressway-E no longer required to proxy SIP registrations

Allows for remote H.323 registrations.

upvoted 1 times

Omitted 1 year, 7 months ago

B is supported no? It is an endpoint feature when you are using MRA phones that use their SSO credentials to connect.

upvoted 1 times

ciscogeek 1 year, 7 months ago

Selected Answer: B

There is SIP Proxy but not H323 Proxy, H323 devices can register to Expressway.

SSO works for MRA devices like jabber via expressway

upvoted 1 times

Question #187 *Topic 1*

When a phone is registered over Mobile and Remote Access, where does it register?

A. Cisco Unified Presence Server

B. Expressway-E

C. Cisco Unified Communications Manager

D. Expressway-C

Correct Answer: *C*

Community vote distribution

C (100%)

nassar1 Highly Voted 3 years, 6 months ago

should be C. Phone register with CUCM.

upvoted 22 times

rishik Highly Voted 3 years, 5 months ago

Phone registers with CUCM.

C is correct answer

upvoted 5 times

lpicardin Most Recent 9 months, 1 week ago

Selected Answer: C

MRA is always register with CUCM

upvoted 1 times

Piji 1 year, 4 months ago

Selected Answer: C

Anything login through MRA registerered to CUCM, Jaber/Phone.

So the correct answer is C.

upvoted 2 times

AJBELL14 1 year, 5 months ago

Selected Answer: C

MRA end point registers to CUCM

upvoted 2 times

omsssh 2 years, 9 months ago

Certainly CUCM - p.254 cisco reference CLCOR- the Cisco Unified IP phones cannot register to the Cisco Expressway. A proxy registration function using Mobile and Remote Access (MRA) allows Cisco Unified IP phones to register to the Cisco Unified Communications Manager through the Expressway

Core and Edge, but in this type of deployment, the registration is still to the Cisco Unified CM.

upvoted 3 times

Puh 2 years, 10 months ago

C

Just tested Jabber (BOT) MRA,device registered with CUCM

upvoted 2 times

MrAshour 3 years ago

Page 11 in the below link, I believe, cleared the confusion

Jabber An example remote endpoint, which is registering over the internet to Unified

CM via the Expressway-E and Expressway-C.

While

EX60 An example remote endpoint, which is registering to the Expressway-E via the

internet.

The correct answer appears to be (Expressway-E), because Jabber was not specifically mentioned in the Question???

https://www.cisco.com/c/dam/en/us/td/docs/voice_ip_comm/expressway/config_guide/X12-5/Cisco-Expressway-Basic-Configuration-Deployment-Guide-X12-5-4.pdf

upvoted 1 times

MrAshour 3 years ago

More reading , C is correct

upvoted 3 times

Piji 1 year, 4 months ago

Anything login through MRA registerered to CUCM, Jaber/Phone.

So the correct answer is C.

upvoted 1 times

abelcollab 3 years, 3 months ago

Detailed MRA / Jabber Client process @ https://netcraftsmen.com/cisco-mobile-remote-access-troubleshooting-basic-connectivity/

upvoted 1 times

khader09 3 years, 4 months ago

C is the correct answer

upvoted 3 times

Question #188 *Topic 1*

Which description of the Mobile and Remote Access feature is true?

A. Collaboration Edge feature that enables remote individuals to perform international calls from Jabber with a VPN connection.

B. Collaboration Edge feature that enables remote individuals to access all enterprise collaboration services using a PC within the corporate environment.

C. Collaboration Edge feature that enables remote individuals to access enterprise collaboration services via Jabber without the use of a VPN connection.

D. Collaboration Edge feature that enables remote individuals to access enterprise collaboration services via Jabber with the use of a VPN connection.

Correct Answer: *C*

Currently there are no comments in this discussion, be the first to comment!

Question #189

Topic 1

An administrator needs to help a remote employee make a free call to an international destination. The administrator calls the employee, then conferences in the international party. The administrator drops the call, and the employee and the international party continue their conversation. Which action prevents this type of toll fraud in the Cisco UCM?

A. Set service parameter ג€Advanced Ad Hoc Conferenceג€ to 2.

B. Set service parameter ג€Drop Ad Hoc Conferenceג€ to ג€Do not allow outside parties.ג€

C. Set service parameter ג€Advanced Ad Hoc Conferenceג€ to FALSE.

D. Set service parameter ג€Drop Ad Hoc Conferenceג€ to ג€When Conference Controller leaves.ג€

Correct Answer: *D*

Reference:

https://www.cisco.com/c/en/us/td/docs/voice_ip_comm/cucm/admin/10_5_2/ccmfeat/CUCM_BK_C3A84B33_00_cucm-feature-configuration-guide_1052/CUCM_BK_C3A84B33_00_cucm-feature-configuration-guide_chapter_010000.html

Currently there are no comments in this discussion, be the first to comment!

Question #190 *Topic 1*

An engineer is implementing toll fraud prevention for incoming calls cluster-wide on Cisco UCM. What is the first step to configure this feature?

A. Set service parameter ג€System Remote Access Blocked Numbersג€ to True.

B. Set service parameter ג€Block OffNet To OffNet Transferג€ to True.

C. Set service parameter ג€Block OffNet To OffNet Transferג€ to False.

D. Configure blocking for inbound calls based on caller ID.

Correct Answer: *B*

Reference:

https://www.cisco.com/en/US/docs/voice_ip_comm/cucm/admin/4_1_3/ccmfeat/fsxfer.html

Community vote distribution

B (100%)

Omitted 7 months, 2 weeks ago

Selected Answer: B

You want to set this to true. By default it is set to False.

upvoted 2 times

Question #191 *Topic 1*

A collaboration engineer must optimize the dial plan within Cisco UCM. There are multiple remote sites, and each site has its own route patterns and local gateways. What should the engineer do on the Cisco UCM to optimize the dial plan?

A. Implement ILS with GDPR so the dial plan can dynamically replicate across clusters.

B. Configure a Standard Local Route Group to use a single route pattern for all calls within the cluster.

C. Create a centralized dial plan with a Cisco UCM Session Management Edition cluster or a Cisco gatekeeper.

D. Leverage Cisco UCM Express as SRST so the phones can have more features at each remote site.

Correct Answer: *B*

Community vote distribution

B (100%)

iExpo_91 8 months, 4 weeks ago

Selected Answer: B

Answer is B, there are no multiple clusters is sites instead so ILS/GDRP wont apply on this design.

upvoted 3 times

lpicardin 9 months, 1 week ago

Selected Answer: B

There is no multiple cluster so the correct answer is B

upvoted 2 times

BrianC 10 months, 3 weeks ago

Selected Answer: B

B is the Answer

upvoted 2 times

DaKenjee 1 year, 1 month ago

Selected Answer: B

Standard Local Route Group addresses Route Group of location by device pool setting
All phones know Route Group in there location for PSTN access by device pool.

general route pattern can now address outgoing calls for all locations with "\+.!"
not by a single gateway/Route group, instead it is Standard Local Route Group

upvoted 2 times

AJBELL14 1 year, 5 months ago

It is a trick question. Generally for a main site and remote sites, the LRG would be the right answer

upvoted 1 times

bmne 1 year, 5 months ago

B is the Answer
The Local Route Group feature helps reduce the complexity and maintenance efforts of provisioning in a centralized Cisco Unified Communications Manager deployment that uses a large number of locations.

upvoted 1 times

Omitted 1 year, 7 months ago

Where does this question say there are multiple clusters? I think i might go with B here.

upvoted 1 times

Question #192 *Topic 1*

An administrator must implement toll fraud prevention on Cisco UCM using these parameters:
✑ Enable Forced Authorization Code 112211.
✑ Set an authorization level of 3 for the route pattern 8005551212.
✑ Require no access code to dial 10-digit numbers.
How must the route pattern be implemented?

A. Pattern = 8005xxxxxx

B. Pattern = 1122113.8005551212

C. Pattern = 3.800xxxxxxx

D. Pattern = 8005551212.1122113

Correct Answer: *A*

Community vote distribution

A (100%)

G0y0 4 months ago

Selected Answer: A

Correct Answer is A.
Neither the authorization level parameter nor the FAC have nothing to do into the pattern.
upvoted 1 times

Omitted 7 months, 2 weeks ago

Selected Answer: A

It is A, nothing special has to be done on the route pattern beside checking the Require Forced Authorization Code checkbox
https://www.cisco.com/c/en/us/support/docs/voice-unified-communications/unified-communications-manager-callmanager/81541-fac-config-ex.html
upvoted 2 times

Question #193 *Topic 1*

An engineer is asked to implement on-net/off-net call classification in Cisco UCM. Which two components are required to implement this configuration? (Choose two.)

A. SIP trunk

B. SIP route patterns

C. CTI route point

D. route group

E. route pattern

Correct Answer: *AE*

Reference:

https://www.cisco.com/c/en/us/td/docs/voice_ip_comm/cucm/admin/10_5_2/ccmfeat/CUCM_BK_C3A84B33_00_cucm-feature-configuration-guide_1052/CUCM_BK_C3A84B33_00_cucm-feature-configuration-guide_chapter_0100010.html

Community vote distribution

AE (100%)

b3532e4 9 months, 2 weeks ago

D & E correct

upvoted 1 times

TheBabu 1 year, 2 months ago

Selected Answer: AE

A and E seem correct according to this doc:

The External Call Transfer Restrictions feature allows you to configure gateways, trunks, and route patterns as OnNet (internal) or OffNet (external) devices at the system level.

...

For incoming calls, configure individual gateways or trunks as OffNet.

...

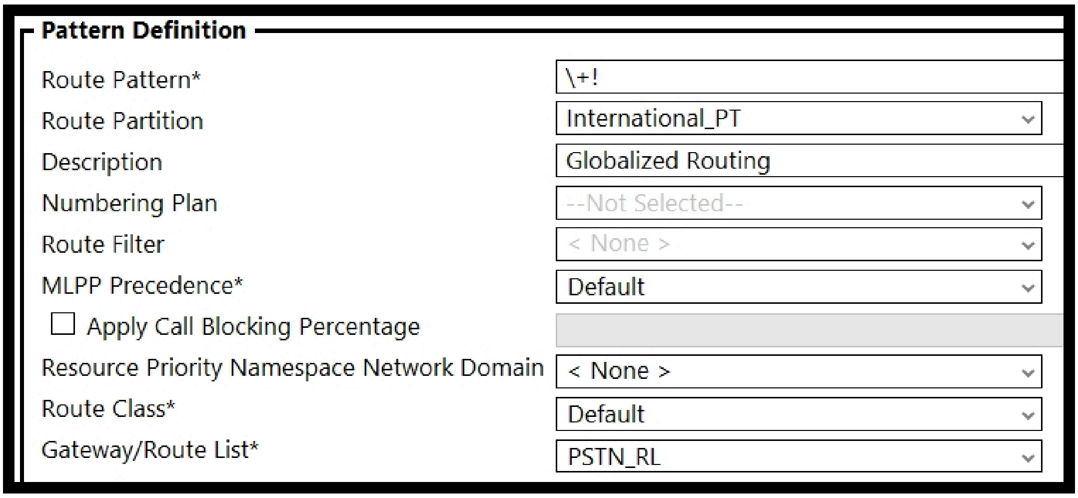
For outgoing calls, configure route pattern Call Classification field as OffNet.

https://www.cisco.com/c/en/us/td/docs/voice_ip_comm/cucm/admin/10_0_1/ccmfeat/CUCM_BK_F3AC1C0F_00_cucm-features-services-guide-100/CUCM_BK_F3AC1C0F_00_cucm-features-services-guide-100_chapter_011011.html

upvoted 2 times

An engineer must configure a route pattern that can route all +E.164 globalized international numbers for the dial plans of all countries. Which Cisco UCM configuration accomplishes this task?

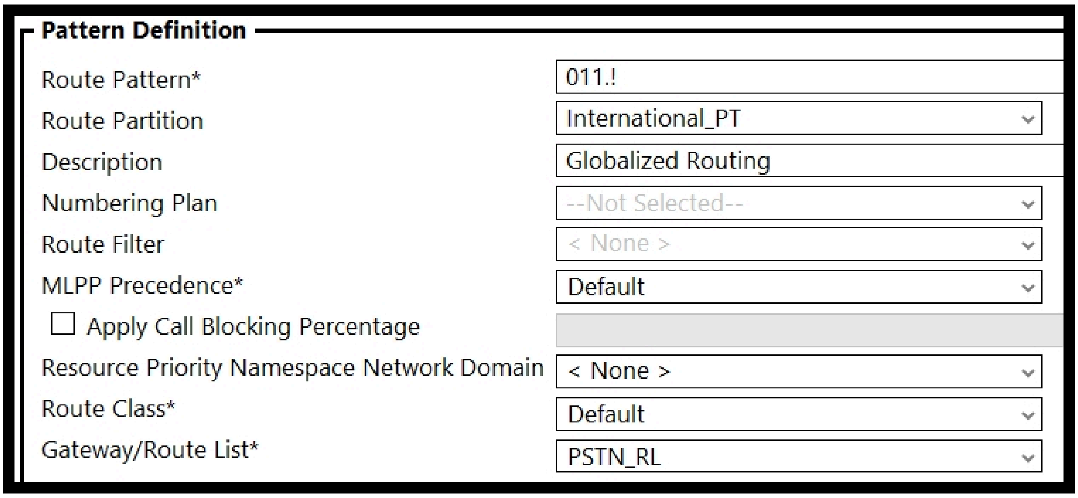
A.



Pattern Definition

Route Pattern*	\+!
Route Partition	International_PT
Description	Globalized Routing
Numbering Plan	--Not Selected--
Route Filter	< None >
MLPP Precedence*	Default
☐ Apply Call Blocking Percentage	
Resource Priority Namespace Network Domain	< None >
Route Class*	Default
Gateway/Route List*	PSTN_RL

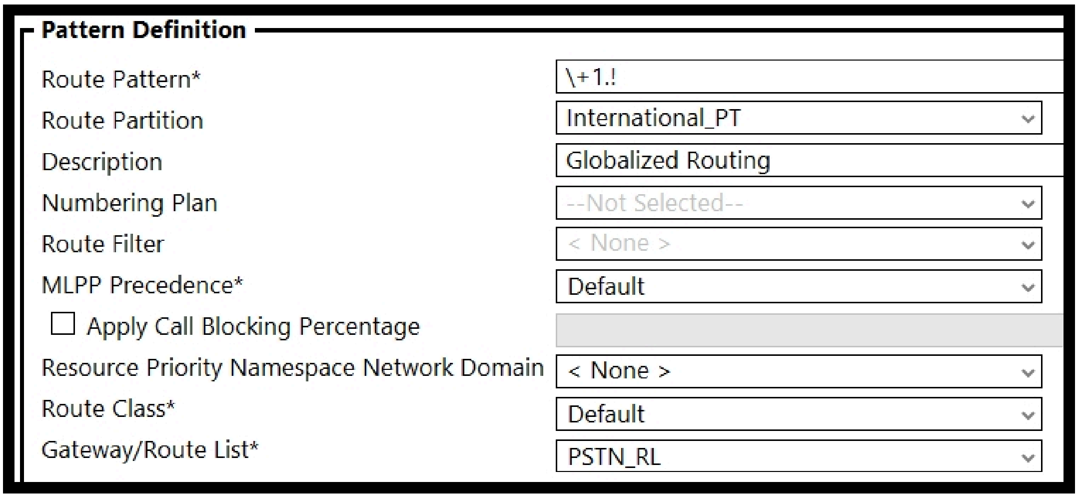
B.



Pattern Definition

Route Pattern*	011.!
Route Partition	International_PT
Description	Globalized Routing
Numbering Plan	--Not Selected--
Route Filter	< None >
MLPP Precedence*	Default
☐ Apply Call Blocking Percentage	
Resource Priority Namespace Network Domain	< None >
Route Class*	Default
Gateway/Route List*	PSTN_RL

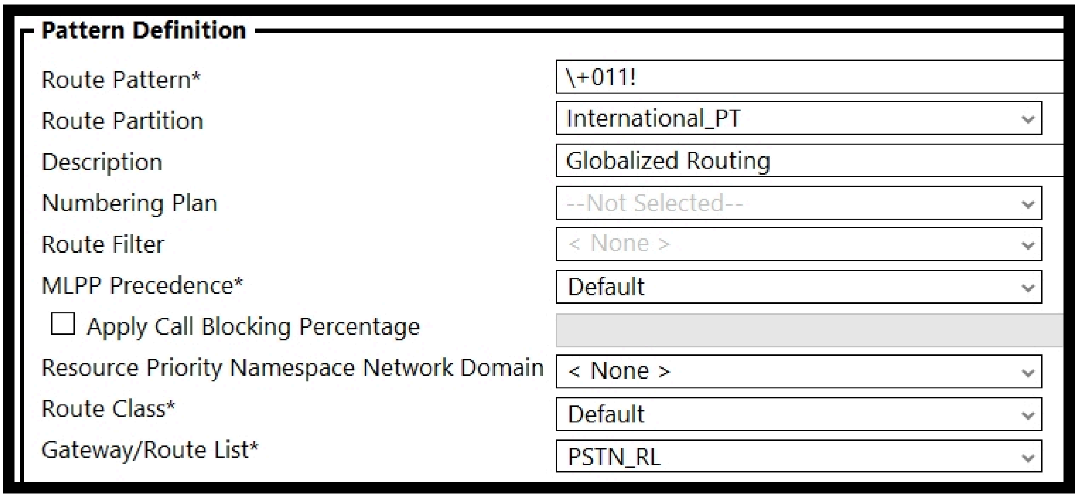
C.



Pattern Definition

Route Pattern*	\+1.!
Route Partition	International_PT
Description	Globalized Routing
Numbering Plan	--Not Selected--
Route Filter	< None >
MLPP Precedence*	Default
☐ Apply Call Blocking Percentage	
Resource Priority Namespace Network Domain	< None >
Route Class*	Default
Gateway/Route List*	PSTN_RL

D.



Pattern Definition

Route Pattern*	\+011!
Route Partition	International_PT
Description	Globalized Routing
Numbering Plan	--Not Selected--
Route Filter	< None >
MLPP Precedence*	Default
☐ Apply Call Blocking Percentage	
Resource Priority Namespace Network Domain	< None >
Route Class*	Default
Gateway/Route List*	PSTN_RL

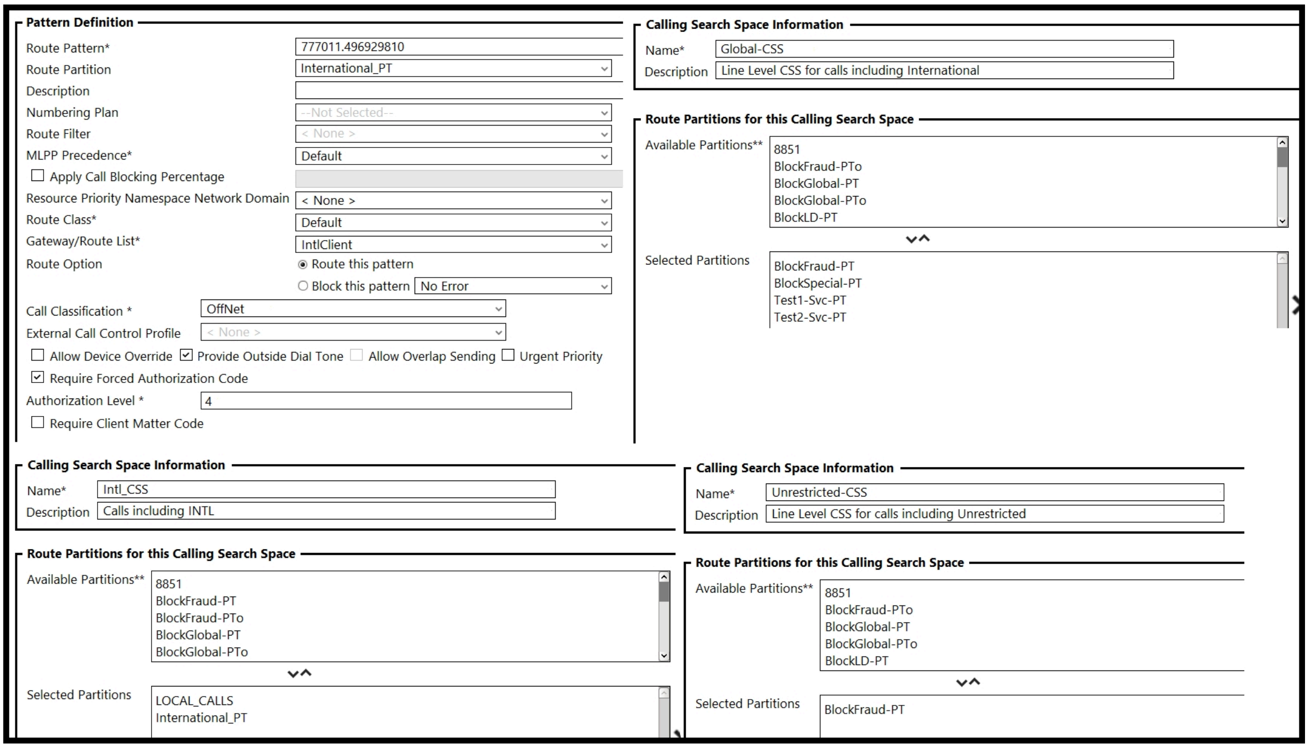
Correct Answer: *A*

sneff91 9 months, 4 weeks ago

A. "011" is used as the international dialing prefix in North America, but not all countries, so this is not valid. A relies on a translation pattern existing to normalize dialing to +E.164 format, but it does then route it in a valid manner. The question simply asks for a route pattern that ROUTEs internationally, not that it is also expected to normalize it.

upvoted 1 times

Question #195 | *Topic 1*



Refer to the exhibit. How must the +E.164 translation pattern be configured to reach international number 496929810?

A. Pattern= \+.496929810, CSS=Intl_CSS, PreDot, Prefix=777011

B. Pattern= \+.496929810, CSS=Unrestricted-CSS, PreDot, Prefix=777011

C. Pattem= \+.777011496929810, CSS=Intl_CSS

D. Pattern= \+.011496929810, CSS=Global-CSS, PreDot, Prefix=777

Correct Answer: *A*

Community vote distribution

A (100%)

ciscogeek Highly Voted 1 year, 7 months ago

Selected Answer: A

Should be A, because International Number is given and it should be in E164 format. Only A matches the format, and then prefixes the required digit for matching the route pattern.

upvoted 10 times

G0y0 Most Recent 5 months ago

The closest answer culd be A. However, if C had included "predot", it could be de best answer since 777 is the offnet access code and 011 is the international acces code. So, since C omit "predot" to complain the Route Pattern, A becomes the closest answer.

upvoted 1 times

G0y0 5 months ago

Another thing is, do not ouu think that it is not intuitive the user dials the "+" before access code 777 and international access code 011? it is ilogically redendant to put "+" and the "011", so C seems to be out of intuition.

upvoted 1 times

lpicardin 9 months, 1 week ago

Selected Answer: A

Answer A

upvoted 2 times

DaKenjee 1 year, 1 month ago

Selected Answer: A

It is a tricky question, i say Answer A.
Magic word is "international number 496929810"
Because both Answer A+C would trigger routepattern by later CSS and signaled number
Means we have to assume international goes with +E164 and therefore enduser calls +496929810

First question is, which Calling Search space (CSS) is working for getting displayed Route pattern; it is Intl_CSS
Intl_CSS is the only one with selected partition International_PT
Call should go first through a TP, which then should activate/hit mentioned route pattern
The route pattern could be understood by its dot, that 777011 will get stripped by predot, but this is not covered by screenshot
But does not matter because routepattern will do its job and sends call to ITSP
So TP should send call to RP as \+777011496929810, or RP won't be triggered.
upvoted 1 times

DaKenjee 1 year, 1 month ago
Besides Answer C is wrong, because a Translation pattern, which triggers same number like route pattern is useless and unnecessary
Which ends up in my thoughts, enduser calls +E164
upvoted 1 times

Padu_Home 1 year, 1 month ago
Selected Answer: A
Should be A, Answer B & D are wrong as CSS don't include the International_PT and Answer C is wrong as it is missing the PreDot to strip off the +
upvoted 3 times

Question #196 *Topic 1*

An engineer must configure a SIP route pattern using domain routing with destination +12345678901. The domain ciscocm1.jupiter.com resolves to 192.168.1.3.
How must the IPv4 Pattern be configured?

A. ciscocm1.jupiter.com

B. \+12345678901@192.168.1.3

C. 192.168.1.3

D. +12345678901@192.168.1.3

Correct Answer: *A*
Reference:
https://www.cisco.com/c/en/us/td/docs/voice_ip_comm/cucm/admin/10_0_1/ccmcfg/CUCM_BK_C95ABA82_00_admin-guide-100/CUCM_BK_C95ABA82_00_admin-guide-100_chapter_0100111.pdf

Community vote distribution

A (100%)

b3532e4 10 months, 3 weeks ago

D just D

upvoted 1 times

genarouaaan 2 years, 7 months ago

Selected Answer: A

A is correct becasue DNS is working and can resolve name.

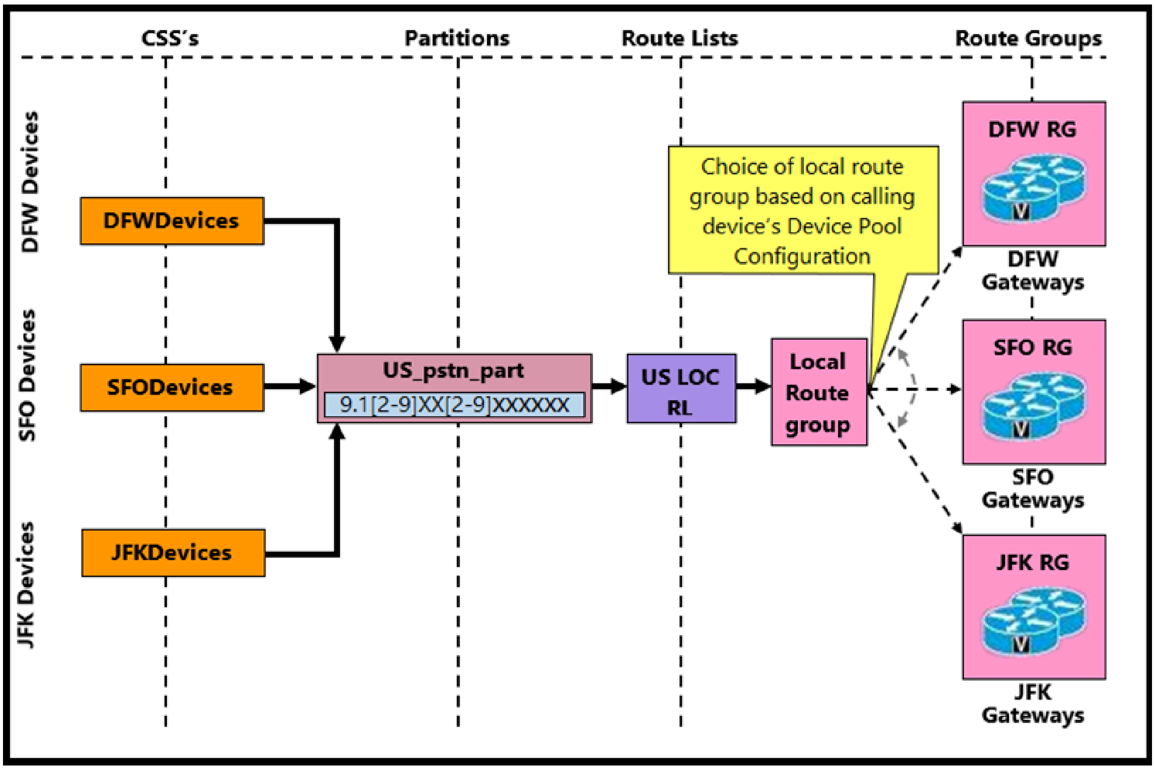
upvoted 2 times

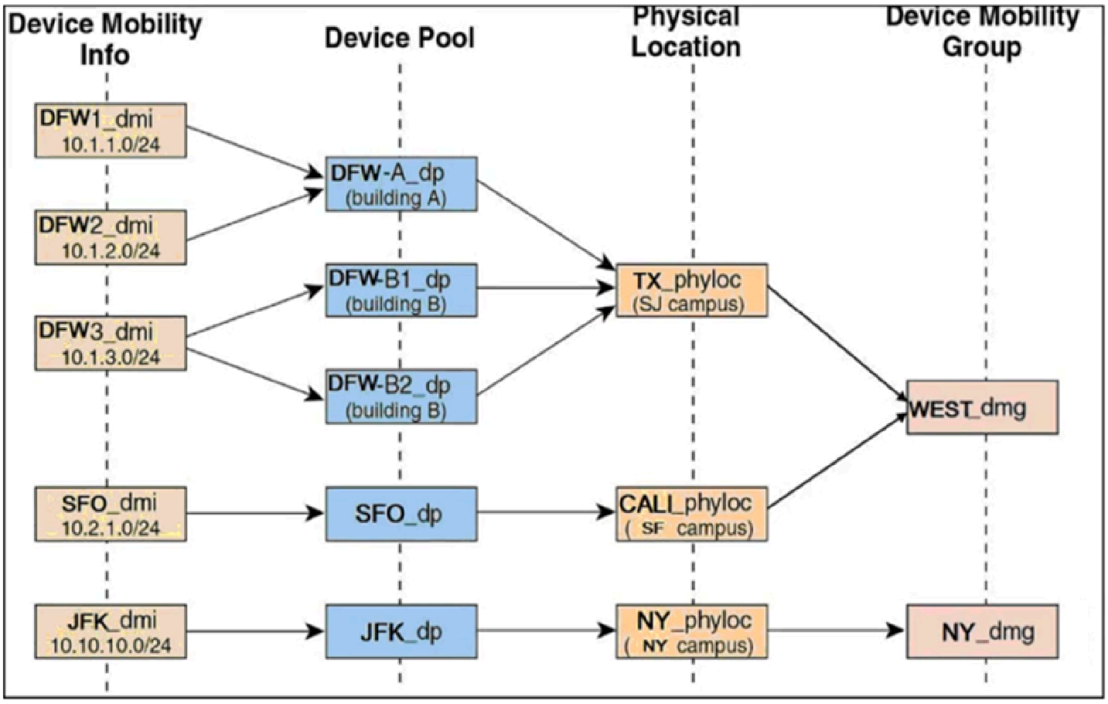
Omitted 3 years, 1 month ago

It is A. B&D would fail for invalid characters and you'd choose C if it was configured with IP address routing instead of domain routing.

upvoted 2 times

Question #197 *Topic 1*





Refer to the exhibit. A user takes a phone from San Francisco to New York for a short reassignment. The phone was set up to use the San Francisco device pool, and device mobility is enabled on the Cisco UCM. The user makes a call that matches a route pattern in a route list that contains the Standard Local Route Group.

To where does the call retreat?

A. The call egresses in San Francisco because the user uses device mobility and is allowed to roam while still keeping the number and resources assigned in San Francisco.

B. The call egresses in New York because the device automatically is assigned a New York device pool and uses the local gateway.

C. The call fails because device mobility is turned on, and the phone is not configured in New York. The engineer must configure which sites the device should be roaming to.

D. The call fails because the Standard Local Route Group is being used only if no configuration is set for the device pools.

Correct Answer: *B*

Community vote distribution

B (100%)

Slushed Highly Voted 2 years, 1 month ago

Selected Answer: B

Correct answer is B. The point of Device Mobility is for devices that are moved to a new location to utilize the resources local to that location.

upvoted 10 times

G0y0 Most Recent 4 months ago

Selected Answer: B

Interesting question, it is combining Standard Local Route Group and Device Mobility.

First, by step, remember that:

When a roaming device moves to another location in a different device mobility group:

Roaming Device Pool: yes

Location: Roaming device pool setting

Region: Roaming device pool setting

Media Resources Group List: Roaming device pool setting

Device CSS: Home location settings

AAR Group: Home location settings

AAR CSS: Home location settings.

upvoted 1 times

G0y0 4 months ago

In this case, the SFO user remains with his home CSS, but now with the NY DP.

Second step:

The Home CSS (SFODevices) have a partition pointing a RP (US_pstn_part) targeting a RL (US LOC RL) with a Standard Local Route Group. As the SFO user has now the Remote DP (JFK_dp, not the Home SFO_dp), then Standard Local Route Group will select NY Gateway for these national call, also it will take the media resources from NY Gateway. So Answer B is in compliance with the behavior of this framework

upvoted 1 times

G0y0 4 months ago

If certain calls from roaming endpoints need to be routed through gateways local to the home site of the roaming phone, then routing for these calls has to be implemented through route patterns pointing to route lists that use fixed site-specific route groups instead of Standard Local Group. If this were the case, then A would be the correct answer.

References:

"Feature Configuration Guide for Cisco Unified Communications Manager, Release 12.5(1)", Chapter 4: Device Mobility, Table 4: Device Mobility Scenarios.

"Cisco Collaboration System 12.x Solution Reference Network Designs SRND", Chapter 21: Mobile Collaboration, section: "Dial Plan Design Considerations"

upvoted 2 times

c37e2aa 4 months, 3 weeks ago

Selected Answer: A

DMG is not the same, so the Device Mobilty realted Information will not be applied.

upvoted 1 times

G0y0 4 months ago

That is correct, however, Roaming-sensitive settings are being applied, so it is taking the remote DP. Also, as the home CSS is pointing to a RP targeting a Standar Local Route Group, it will select the gateway with the current DP (in this case is the remote DP), and then it will select the remote gateway (NY gateway) to release the call and the media resources of the same remote gateway.

upvoted 1 times

cuernov 8 months ago

this question is missing a image.

upvoted 1 times

Panda_man 1 year, 5 months ago

Selected Answer: B

B for sure

upvoted 1 times

DaKenjee 1 year, 7 months ago

Selected Answer: B

Answer B

Standard local route group would use San Francisco device pool, if there is no device mobility active

Device Mobility Info Configuration delivers by IPv4 Subnet a selected Device Pool ; in this case new york device pool and its local gateway in new york

upvoted 4 times

DaKenjee 1 year, 7 months ago

They used pictures and explanation from SRND

Example 14-3 Device Mobility

https://www.cisco.com/c/en/us/td/docs/voice_ip_comm/cucm/srnd/collab12/collab12/dialplan.html

upvoted 3 times

MPLM 1 year, 9 months ago

B is the correct answer

https://community.cisco.com/t5/collaboration-knowledge-base/device-mobility/ta-p/3124062

upvoted 1 times

AAMM 2 years, 1 month ago

B is the answer

upvoted 1 times

Question #198 *Topic 1*

User's phone has:

Line level CSS which contains Partition_A
Device level CSS which contains Partition_B

Configured Route Patterns:

2XXX in Partition_A set to block call
21XX in Partition_B set to route call

Refer to the exhibit. An engineer is configuring class of control for a user in Cisco UCM. Which change will ensure that the user is unable to call 2143?

A. change line CSS to only contain Partition_B

B. set the user's line CSS to <None>

C. change line partition to Partition_A

D. set the user's device CSS to <None>

Correct Answer: *D*

Community vote distribution

D (100%)

DaKenjee Highly Voted 1 year, 1 month ago

Selected Answer: D

Checked with DNA, true, device CSS will be in use, because closer match

Now about answers

A is wrong, by assigning Partition_B i allow call, so device+line css both allow call

B is wrong, when none, there is still device CSS for allowing call

C is wrong, css is in use, and will hit a route pattern, does not matter which partion is in use for dn

upvoted 6 times

G0y0 Most Recent 3 months, 4 weeks ago

Selected Answer: D

Although route pattern 200X matches the dialed number 2143 and is listed in the first partition, it is not used to route the call. The first priority for the call routing decision is the best match. The order of partitions is important only if multiple best matches exist.

Implementing Cisco IP Telephony and Video, Part 1 (CIPTV1) Foundation Learning Guide

Chapter 8: Implementing Calling Privileges in Cisco Unified Communications Manager

upvoted 1 times

Myare 11 months, 1 week ago

Correct answer is A

by assigning both line and device to same CSS with same partition, it allow the call

upvoted 1 times

Learninp09 9 months ago

Yes, but we are trying the opposite (to block it)

upvoted 2 times

Omitted 1 year, 7 months ago

Selected Answer: D

The line level should take precedence over the device level. But i guess D since do nothing isn't an answer.

upvoted 4 times

Ruuddie 1 year, 2 months ago

Afaik it takes precedence, but only for exact matches. 2XXX is less specific, so it doesn't match.

upvoted 1 times

Question #199 *Topic 1*

An administrator must configure the Local Route Group feature on Cisco UCM. Which step will enable this feature?

A. For each route list, configure a route group to use as a Local Route Group.

B. For each route group, check the box for the Local Route Group feature.

C. For each route pattern, select the Local Route Group as the destination.

D. For each device pool, configure a route group to use as a Local Route Group for that device pool.

Correct Answer: *D*

Reference:

https://www.cisco.com/c/en/us/td/docs/voice_ip_comm/cucm/admin/10_0_1/ccmfeat/CUCM_BK_F3AC1C0F_00_cucm-features-services-guide-100/

CUCM_BK_F3AC1C0F_00_cucm-features-services-guide-100_chapter_0100110.html

Community vote distribution

D (100%)

MaxG 1 year ago

Selected Answer: D

Configure Local Route Groups – Step 4

https://www.cisco.com/c/en/us/td/docs/voice_ip_comm/cucm/admin/10_0_1/ccmfeat/CUCM_BK_F3AC1C0F_00_cucm-features-services-guide-100/CUCM_BK_F3AC1C0F_00_cucm-features-services-guide-100_chapter_0100110.html

upvoted 1 times

Question #200 *Topic 1*

Where is urgent priority enabled to bypass the T302 timer?

A. transformation pattern

B. directory number

C. route partition

D. CTI port

Correct Answer: *B*

Community vote distribution

B (100%)

c37e2aa 4 months, 3 weeks ago

Selected Answer: A

Urgent Priority can be enabled on Route-pattern and DN but it is enabled by default on Transformation Pattern.

upvoted 1 times

istellas 6 months, 3 weeks ago

I am a bit confused. It is either A or B.

B is the place where you can configure the urgent priority for a DN.

But Q asks "Where is urgent priority enabled". Cisco Unified Communications Manager sets all called/calling party transformation patterns with urgent priority, and you cannot change the priority of the patterns. It is enabled and cannot be changed on transformation patterns. So, why not A???

upvoted 1 times

ALLENNN 8 months, 3 weeks ago

Selected Answer: B

B DN has option to select priority.

upvoted 1 times

DaKenjee 1 year, 1 month ago

Selected Answer: B

Urgent Priority only exists as option on DN

upvoted 2 times

Omitted 1 year, 7 months ago

Selected Answer: B

It is B. There's an Urgent Priority checkbox right next to the directory number on the DN page.

upvoted 1 times

ciscogeek 1 year, 7 months ago

Selected Answer: B

Directory Number has 'Urgent Priority' Check Box after the DN field.

No other options here have urgent priority check box.

upvoted 2 times

Question #201 *Topic 1*

Callers from a branch report getting busy tones intermittently when trying to reach colleagues in other office branches during peak hours. An engineer collects
Cisco CallManager service traces to examine the situation. The traces show:
50805567.000 |07:35:39.676 |SdlSig |StationOutputDisplayNotify |restart0 |StationD(1,100,63,6382) |StationCdpc(1,100,64,4725) |1,100,40,7.101980^*^* |
[R:N-H:0,N:0,L:0,V:0,Z:0,D:0] TimeOutValue=10 Status=x807 UnicodeStatus= Locale=1 50805567.001 |07:35:39.676 |AppInfo | StationD: (0006382) DisplayNotify timeOutValue=10 notify='x807' content='Not Enough Bandwidth' ver=85720014.
What should be fixed to resolve the issue?

A. class of service configuration

B. geolocation configuration

C. region configuration

D. codec configuration

Correct Answer: *C*

Community vote distribution

C (100%)

G0y0 3 months, 2 weeks ago

Selected Answer: C

D. is totally incorrect. When when you are configuring regions in the CUCM, you do not select a codec specifically; rather, you set the per-call bandwidth rate that is allowed between each site. You can place a video call using 480p30 resolution at 384 kbps or at 2 Mbps. However, using a higher codec can provided better efficiency and thus consume less bandwidth.
In this case, obviously C. is correct.

upvoted 1 times

b3532e4 9 months ago

At the "peak time" ----> Not Enough Bandwidth. In my opinion Codec configuration

upvoted 1 times

G0y0 3 months, 2 weeks ago

And where would you configure the codecs?

upvoted 1 times

AgshinA 1 year, 3 months ago

Selected Answer: C

The issue indicated by the “Not Enough Bandwidth” message in the Cisco CallManager service traces suggests that the available bandwidth for calls between the branches is being exceeded during peak hours. To resolve this issue, you should:

Check and Adjust Locations and Regions: Verify the configuration of locations and regions in Cisco CallManager to ensure that they are set up correctly and that there is enough allocated bandwidth for the number of expected concurrent calls between the branches1.
Use Low Bandwidth Codecs: Consider configuring low bandwidth codecs like G.729 or G.723 for calls between the branches to reduce the amount of bandwidth each call consumes1.

upvoted 2 times

lpicardin 2 years, 3 months ago

Selected Answer: C

Answer C

upvoted 2 times

AAMM 3 years, 1 month ago

answer is B

upvoted 1 times

AAMM 3 years, 1 month ago

sorry for that it is C the correct answer

upvoted 2 times

ciscogeek 3 years, 1 month ago

Selected Answer: C

Class of Service is Partition/CSS which cannot solve the issue.

Region Configuration can set the maximum audio bit rate.

There is no page in CUCM called 'Codec' configuration, but it makes the question confusing. I would go with Region Config.

upvoted 4 times

Question #202 *Topic 1*

What is the purpose of Mobile and Remote Access (MRA) in the Cisco UCM architecture?

A. MRA is used to access Webex cloud services only if authenticated with on-premises LDAP service.

B. MRA is used to make secure PSTN calls by Cisco UCM only while on-premises authentication.

C. MRA is used to make B2B calls through Expressway registration.

D. MRA is used to access the collaboration services offered by Cisco UCM from off-premises network connections.

Correct Answer: *D*

Community vote distribution

D (100%)

TheBabu 9 months ago

Selected Answer: D

it's D. MRA and B2B are two different things.

Also "Expressway registration" implies endpoints are registering directly to Expressway, which is not the case when using MRA.

upvoted 1 times

0n3 12 months ago

C is the wrongest !!!!

upvoted 1 times

DaKenjee 2 years, 1 month ago

Selected Answer: D

MRA is for mobile remote access without vpn

upvoted 1 times

Piji 2 years, 4 months ago

Selected Answer: D

The correct answer is D.

upvoted 3 times

AAMM 2 years, 7 months ago

D is the answer

upvoted 2 times

Omitted 2 years, 7 months ago

Selected Answer: D

B2B is a separate configuration than MRA. The answer here is D

upvoted 3 times

Question #203 *Topic 1*

Which external DNS SRV record must be present for Mobile and Remote Access?

A. _cisco-uds._tcp.example.com

B. _collab-edge._tls.example.com

C. _collab-edge._tcp.example.com

D. _cisco-uds._tls.example.com

Correct Answer: *B*

Reference:

https://www.cisco.com/c/en/us/td/docs/voice_ip_comm/expressway/config_guide/X14-0-1/mra/exwy_b_mra-deployment-guide-x1401/exwy_m_requirements-for-mra.html

Community vote distribution

B (100%)

DaKenjee 7 months, 1 week ago

Selected Answer: B

Answer B

- > External Records

https://www.cisco.com/c/en/us/td/docs/voice_ip_comm/jabber/Windows/9_7/CJAB_BK_C606D8A9_00_cisco-jabber-dns-configuration-guide/CJAB_BK_C606D8A9_00_cisco-jabber-dns-configuration-guide_chapter_010.html

upvoted 1 times

Ol_Mykhailiuk 9 months, 1 week ago

Selected Answer: B

Cisco Expressway supports Mobile and Remote Access with multiple external domains. With this deployment, you will have more than one external domain where your MRA clients may reside. Expressway-E must be able to connect to all of them. To configure this deployment, do the following:

For Expressway-E:

On Expressway-E, configure _collab-edge._tls.<domain> and _sips_tcp.<domain>DNS SRV records for each Edge domain.

upvoted 1 times

Question #204 *Topic 1*

When a call is delivered to a gateway, the calling and called party number must be adapted to the PSTN service requirements of the trunk group. If a call is destined locally, the + sign and the explicit country code must be replaced with a national prefix. For the same city or region, the local area code must be replaced by a local prefix as applicable. Assuming that a Cisco UCM has a SIP trunk to a New York gateway (area code 917), which two combinations of solutions localize the calling and called party for a New York phone user? (Choose two.)

A. Configure the gateway to translate the calling number and apply it to the dial peer. Combine it with a translation profile for called numbers. ! voice translation-rule 1 rule 1 /^1917/ // rule 2 /^ [+]1917/ // ! voice translation-profile strip+1 translate calling 1 !

B. Configure two calling party transformation patterns: \+1917.CCCCCC, strip pre-dot, numbering type: subscriber \+!, strip pre-dot, numbering type: national

C. Configure two called party transformation patterns: \+1917.XXXXXXX, strip pre-dot, numbering type: subscriber \+1.!, strip pre-dot, numbering type: national

D. Configure the gateway to translate called numbers and apply it to the dial peer. Combine it with a translation profile for calling numbers. ! voice translation-rule 1 rule 1 /^1917/ // rule 2 /^[+]1917/ // ! voice translation-profile strip+1 translate called 1 !

E. Configure two calling party transformation patterns: \+1917.XXXXXXX, strip pre-dot, numbering type: subscriber \+1.!, strip pre-dot, numbering type: national

Correct Answer: *CE*

Community vote distribution

CE (64%) | AD (21%) | 14%

DaKenjee Highly Voted 2 years, 7 months ago

Selected Answer: CE

Answer C+E

First we have to adjust calling + called and a mixture by gateway normalization + callmanager normalization makes no sence.

Answer E is working for +1917 local->subscriber and +1 -> national signaling

Gateway Calling Party Number Localization:

As a call is delivered to a gateway, the calling party number must be adapted to the requirements of the PSTN service provider providing the trunk group to which the gateway is connected.

Calling Party Number Transformation patterns

can be used to change the calling party number digit string and numbering type.

They even use same wording like in question:

\+1.!, strip pre-dot, numbering type: national

When a call egresses to a gateway,

the calling and called party transformation patterns

are applied to the calling and called numbers respectively.

https://www.cisco.com/c/en/us/td/docs/voice_ip_comm/cucm/srnd/collab12/collab12/dialplan.html

upvoted 9 times

WeNt48 2 years, 4 months ago

Have you ever tried to pass type information over SIP Trunks?

https://www.cisco.com/c/en/us/td/docs/voice_ip_comm/cucm/admin/10_0_1/ccmsys/CUCM_BK_SE5FCFB6_00_cucm-system-guide-100/CUCM_BK_SE5FCFB6_00_cucm-system-guide-100_chapter_0101001.html

"SIP does not support the number type, so calls through SIP trunks only support the Incoming Calling Party Unknown Number (prefix and digits-to-strip) settings."

Key wording here is "Cisco UCM has a SIP Trunk to a New York gateway" and that means any setup going with type is invalid. Whole setup must be done on gateway otherwise it add unnecessary complexity to setup.

upvoted 1 times

wwisp3422112 2 years, 7 months ago

I agree with DaKenjee: CE correct.

If u want to remember the answers to the exam just think about: "XXXXXXX"

upvoted 4 times

b3532e4 Most Recent 9 months, 2 weeks ago

B: Configure transformation patterns to strip "+" and adapt to local numbers.

E: Apply transformation patterns for proper local routing.

upvoted 1 times

WeNt48 2 years, 4 months ago

Selected Answer: AD

https://www.cisco.com/c/en/us/td/docs/voice_ip_comm/cucm/admin/10_0_1/ccmsys/CUCM_BK_SE5FCFB6_00_cucm-system-guide-100/CUCM_BK_SE5FCFB6_00_cucm-system-guide-100_chapter_0101001.html

"SIP does not support the number type, so calls through SIP trunks only support the Incoming Calling Party Unknown Number (prefix and digits-to-strip) settings."

Key wording here is "Cisco UCM has a SIP Trunk to a New York gateway" and that means any setup going with type is incorrect. Whole configuration must be done on gateway otherwise it add unnecessary complexity.

upvoted 3 times

Panda_man 2 years, 5 months ago

Selected Answer: DE

D and E are correct

upvoted 2 times

santiagof 2 years, 7 months ago

A and D , are correct,

configuring 2 calling or 2 called translation (b c and d options), wont do it , because just one takes effect in the call.

upvoted 3 times

WeNt48 2 years, 4 months ago

Correct. SIP Trunks doesn't transmit numbering type information in signalization. Whole configuration must be completed on gateway, and that means only A and D are correct.

upvoted 3 times

OSJAY 2 years, 9 months ago

None of the answers convince me. I would do:

voice translation-rule 1

rule 1 /^1917/ //

rule 2 /^[+]1/ //

! voice translation-profile strip+1

translate called 1

translate calling 1

Apply as Outbound profile to outbound DP

upvoted 3 times

Question #205 *Topic 1*

A customer reports that the Cisco UCM toll-fraud prevention does not work correctly, and the customer is receiving charges for unapproved international calls as a result. Which two configuration changes resolve the issues? (Choose two.)

A. Use Cisco Unified Border Element to debug the calls.

B. Disable call forwarding on the phone.

C. Make the calls route through a firewall.

D. Mark patterns as off-net or on-net.

E. Modify the Block OffNet to OffNet Transfer service parameter.

Correct Answer: *DE*

Community vote distribution

DE (100%)

Padu_Home 7 months, 4 weeks ago

Selected Answer: DE

For the option Block OffNet to OffNet Transfer to work correctly we also need to classify the route patterns correctly. So for answer E to work we also need to do answer D.

upvoted 2 times

OSJAY 9 months, 3 weeks ago

Not 100% but I like B and E better. D says 'Mark patterns as off-net OR on-net'. If it would say only 'off-net' I would go with it.

upvoted 1 times

Obama42 11 months, 1 week ago

Selected Answer: DE

the corret answer

upvoted 2 times

chuck165 11 months, 2 weeks ago

I'd say correct answers are B & E.

D says mark patterns as Off-net or On-net. Marking them as On-net wouldn't really help.

It is possible, and somewhat common for users to forward their phones to an INTL number (such as family member), go home, call their desk phone number (a free local call) and have it connect to their family member in another country.

upvoted 2 times

Omitted 1 year, 1 month ago

Selected Answer: DE

A might be useful to troubleshoot the issue but the question is asking what will resolve the issue. The only two options that make sense for that are D & E

upvoted 2 times

ciscogeek 1 year, 1 month ago

Selected Answer: DE

We can classify the calls as offnet on route patterns and then enable the service parameter to block offnet to offnet calls

upvoted 2 times

Question #206 *Topic 1*

An administrator configures international calling on a Cisco UCM cluster and wants to minimize the number of route patterns that are needed. Which route pattern enables the administrator to match variable-length numbers?

A. 9.011@

B. 9.011#

C. 9.011*

D. 9.011!

Correct Answer: *D*

Community vote distribution

D (100%)

ciscogeek Highly Voted 2 years, 7 months ago

Selected Answer: D

D. 9.011! mathes variable length numbers

upvoted 5 times

Sanrio Most Recent 8 months, 2 weeks ago

Selected Answer: D

D. 9.011! mathes variable length numbers

upvoted 1 times

Korrr 1 year, 7 months ago

Selected Answer: D

* does not exist for cucm route patterns

upvoted 1 times

Testme1235 1 year, 10 months ago

Selected Answer: D

The route pattern that enables the administrator to match variable-length numbers is D. 9.011!

In Cisco Unified Communications Manager (UCM), the exclamation point (!) is used as a wild card character to represent any number of digits. By using 9.011! as a route pattern, the administrator can match any number that starts with 9.011, regardless of the number of digits that follow.

Option A, 9.011@, is not a valid route pattern as the at symbol (@) is not recognized as a wild card character in Cisco UCM.

Option B, 9.011#, is used to indicate the end of the number and is typically used when the user has finished dialing the number. It is not used to match variable-length numbers.

Option C, 9.011*, is not a valid route pattern as the asterisk (*) is not recognized as a wild card character in Cisco UCM.

Therefore, option D, 9.011!, is the correct answer as it allows the administrator to match any number that starts with 9.011, regardless of the number of digits that follow.

upvoted 4 times

DaKenjee 2 years, 1 month ago

Selected Answer: D

The exclamation point (!) wildcard matches one or more digits in the range 0 through 9.

and will be accepted as complete by T.302 Timer & addresses variable length numbers

upvoted 4 times

AJBELL14 2 years, 4 months ago

Selected Answer: D

* is a dial peer wild card. In this case, it should be D for route pattern configuration on the CUCM

upvoted 2 times

AAMM 2 years, 7 months ago

D is the answer

https://www.cisco.com/c/en/us/td/docs/voice_ip_comm/cucm/srnd/collab12/collab12/dialplan.html#:~:text=International%20and%20Variable,a%20cell%2

upvoted 1 times

Omitted 2 years, 7 months ago

Selected Answer: D

Never seen 9.011* before. 9.011! and 9.011!# are very common configured route patterns for international dialing.

upvoted 4 times

Question #207 *Topic 1*

A Cisco UCM administrator sets up new route patterns to support phones in four different locations, all with local gateways. The administrator wants to use the same route pattern for all four locations.
How must the system be configured to achieve this goal?

A. Use transforms in the route groups.

B. Use standard local route groups.

C. Add a CSS to each local gateway.

D. Use CSS alternate routing rules.

Correct Answer: *B*

Community vote distribution

B (100%)

Omitted 7 months, 2 weeks ago

Selected Answer: B

This would be the purpose of standard local route groups.

upvoted 4 times

Question #208 *Topic 1*

An engineer implements QoS in the enterprise network. Which command is used to verify the correct classification and marking on a Cisco IOS switch?

A. show class-map interface GigabitEthernet 1/0/1

B. show policy-map interface GigabitEthernet 1/0/1

C. show policy-map

D. show access-lists

Correct Answer: *B*

Community vote distribution

B (89%) 11%

b3532e4 10 months, 3 weeks ago

C is Correct because mentions it on a Cisco IOS switch NO PORT

upvoted 1 times

kitty73 1 year, 11 months ago

Selected Answer: B

B apply for the Switch, and C for a Router

https://www.cisco.com/c/en/us/td/docs/switches/lan/Denali_16-1/ConfigExamples_Technotes/Config_Examples/Misc/qos/m_qos_monitoring.html

upvoted 2 times

achio 2 years ago

Correct answer it's C

show policy-map

Example:

Device# show policy-map

upvoted 2 times

arinpas 2 years, 2 months ago

Selected Answer: C

I'd say it's C.

See the reference link of the question.

Also, in the link provided below by DaKenjee it's not about a specific interface:

Router# show policy-map interface input brief policy_map_1 timestamp

upvoted 1 times

DaKenjee 2 years, 7 months ago

Selected Answer: B

command needs interface specifically

https://www.cisco.com/c/en/us/td/docs/ios/12_2sb/feature/guide/sb_acpm.html

upvoted 2 times

Piji 2 years, 10 months ago

Selected Answer: B

Corrent answer is B.

upvoted 2 times

AJBELL14 2 years, 11 months ago

Selected Answer: B

the QOS is set on the interface level

upvoted 1 times

AAMM 3 years, 1 month ago

Selected Answer: B

upvoted 1 times

ciscogeek 3 years, 1 month ago

Selected Answer: B

B. show policy-map interface GigabitEthernet 1/0/1

this shows the packets matched

https://www.cisco.com/c/en/us/support/docs/quality-of-service-qos/qos-congestion-avoidance/10107-showpolicy.html

upvoted 2 times

Question #209 *Topic 1*

What is a characteristic of video traffic that governs QoS requirements for video?

A. Video is typically variable bit rate.

B. Voice and video traffic are different, but they have the same QoS requirements.

C. Video is typically constant bit rate.

D. Voice and video traffic are the same, so they have the same QoS requirements.

Correct Answer: *A*

Community vote distribution

A (100%)

Vincentius Highly Voted 3 years, 7 months ago

This should be A, Video is typically variable bitrate

upvoted 15 times

juanmacipag Highly Voted 2 years, 10 months ago

Suggestion: PLEASE Don't just answer by simply answering the question, provide your facts with relevant information, this way we can learn and understand from each other.

A. Video is typically variable bit rate.
https://www.cisco.com/c/en/us/td/docs/voice_ip_comm/cucm/srnd/collab11/collab11/cac.html
Video, on the other hand, has a variable bit rate (is bursty) and has a medium to large footprint when compared to audio, as well as a wide operational range of 1:40 (250p at 15 fps vs 1080p at 60 fps).

Cisco Press Book CLCOR 350-801 p.320
Cisco strongly recommends that you direct only voice traffic to it because voice traffic is well behaved, whereas other types of real-time traffic are not. Moreover, voice traffic requires that delay be nonvariable in order to avoid jitter. Real-time traffic such as video could introduce variation in delay, thereby thwarting the steadiness of delay required for successful voice traffic transmission.

upvoted 6 times

lpicardin Most Recent 9 months, 1 week ago

Selected Answer: A

voice = EF and video = AF41 so can't be the answer B. It's A

upvoted 3 times

Piji 1 year, 4 months ago

Selected Answer: A

Correct answer is A.

upvoted 3 times

G0y0 1 year, 6 months ago

question asks by a characteristic of video traffic.
Video traffic has following key characteristics:
■ Bandwidth: Variable bitrate (bursty) with medium/large footprint
■ Loss sensitive
■ Delay sensitive
So, one of them is requested to answer the question, obviously is "Variable bitrate (bursty) with medium/large footprint", therefore it is easy enough, correct answer is A.

upvoted 1 times

G0y0 1 year, 6 months ago

Reference: Implementing Cisco IP Telephony and Video, Part 1 (CIPTV1) , Chapter 12:, page 297.

upvoted 1 times

Ernisa 2 years, 7 months ago

On the same link where the answer is provided by the admin is written: "The important point to keep in mind is that audio and video, while similar in transport and sensitivity to loss and delay, are quite different with regard to managing their bandwidth requirements in the network. It should also be noted that, while video is pertinent to a full collaboration experience, audio is critical. If, for example, video is lost during a video call due to a network outage or some other network related event, communication can continue provided that audio is not lost during this outage. This is a critical concept when thinking through the network requirements of a collaboration design such as QoS classification and marking."
As so, voice and Video cannot have the same QoS requirements.
upvoted 1 times

mikelima 2 years, 8 months ago
any comment please
upvoted 1 times

CiscoCUCMKing 2 years, 10 months ago
A. B is not correct because voice and video have different QOS markings, voice=EF and video=AF41 or CS4
upvoted 2 times

Cobe43 3 years, 5 months ago
lets proceed by elimination: D is not correct (Audio is a constant bit rate and has a smaller footprint compared to video, as well as a narrower operational range of 1:6 ratio when comparing the lowest bit-rate audio codec to one of the highest bit-rate codecs. Video, on the other hand, has a variable bit rate (is bursty) and has a medium to large footprint when compared to audio, as well as a wide operational range of 1:40 (250p at 15 fps vs 1080p at 60 fps). C is definitely not correct. B is not correct neither because. although Voice and video traffic are different is true, they DO NOT have the same QoS requirements. So this leave us with A as we all know that Video traffic is busty.
upvoted 3 times

rishik 3 years, 5 months ago
B. Correct as the question is about "What is a characteristic of video traffic that governs QoS requirements for video ?" not characteristics of video
upvoted 2 times

BarryR 3 years, 5 months ago
Correct answer is A
upvoted 2 times

Question #210 *Topic 1*

What causes poor voice quality and video pixelization in a video call?

A. The QoS is configured incorrectly.

B. A firewall is blocking the RTP ports.

C. Cisco Unified Communications Manager is configured to use G.711 instead of G.729.

D. 1 Gbps network ports are used instead of 100 Mbps network ports.

Correct Answer: *A*

Currently there are no comments in this discussion, be the first to comment!

Question #211 *Topic 1*

What is a valid class included in the 8-Class QoS Strategy in a VoIP network?

A. Assured Forwarding

B. Broadcast Video

C. Multimedia Conferencing

D. Real-Time Interactive

Correct Answer: *C*

Reference:

https://www.ciscopress.com/articles/article.asp?p=2756478&seqNum=8

Community vote distribution

C (100%)

Nila1 Highly Voted 2 years, 2 months ago

Answer C:

8-Class QoS Strategy

The 8-class QoS strategy model builds upon the 4-class model and includes the following additional classes:

Multimedia conferencing

Multimedia streaming

Network control

Scavenger

The two additional multimedia traffic types in this model are multimedia conferencing and multimedia streaming. The explicitly defined network control traffic class is used for applications such as network routing protocol updates or network infrastructure control traffic such as OAM. The 8-class QoS strategy model is illustrated in Figure 16-6.

upvoted 8 times

Joe76 Most Recent 4 months, 3 weeks ago

Selected Answer: A

AF is part of the 8-Class QoS Strategy. It's a trick question.

upvoted 1 times

DaKenjee 7 months, 1 week ago

Selected Answer: C

8-Class QoS Strategy

The 8-class QoS strategy model builds upon the 4-class model and includes the following additional classes:

+ Multimedia conferencing

+ Multimedia streaming

+ Network control

+ Scavenger

https://www.ciscopress.com/articles/article.asp?p=2756478&seqNum=8

upvoted 1 times

msully 1 year, 8 months ago

I dont have an answer here, but the CLCOR 350-801 Official Cert Guide, page 310-311 states the 8 Class Baseline Model includes the following classes:

Voice

Video

Call Signaling

Network Control

Critical Data

Bulk Data
Best Effort
Scavenger
upvoted 2 times

F3rnando 1 year, 10 months ago

This q makes no sense, BCD are all included in 12 class model, not in 8 class model.
Anyway previous commenter found one book that mentioned second name
Multimedia conferencing (Interactive video)
https://www.ciscopress.com/articles/article.asp?p=2756478&seqNum=8

So lets gamble with answer C - Multimedia conferencing
upvoted 3 times

Question #212 *Topic 1*

Which issue can occur if QoS is not deployed on a Cisco Collaboration architecture across the WAN?

A. 403 Forbidden errors on SIP calls

B. excessive jitter

C. unexpected shut-down on Cisco Unified Communications Manager

D. packet fragmentation

Correct Answer: *B*

Currently there are no comments in this discussion, be the first to comment!

Question #213 *Topic 1*

What describes the outcome when the trust boundary is defined at the Cisco IP phone?

A. Packets or Ethernet frames are remarked at the access layer switch.

B. Packets or Ethernet frames are not remarked by the IP phone.

C. Packets or Ethernet frames are not remarked at the access layer switch.

D. Packets or Ethernet frames are remarked at the distribution layer switch.

Correct Answer: *C*
Reference:
https://networklessons.com/quality-of-service/how-to-configure-qos-trust-boundary-on-cisco-switches

ratbat Highly Voted 1 year, 7 months ago
C is the correct answer because if the trust boundary is the phone, it meant that phone serves a s security guard at the trust boundary enforcing QOS policies, but anything beyond the phone, simply trust the QOS markings and queues traffic based on receive DSCP or QOS values
upvoted 9 times

Stevon Most Recent 3 weeks ago
Selected Answer: C
https://networklessons.com/quality-of-service/how-to-configure-qos-trust-boundary-on-cisco-switches
upvoted 1 times

Stevon 3 weeks ago
Selected Answer: D
https://networklessons.com/quality-of-service/how-to-configure-qos-trust-boundary-on-cisco-switches
upvoted 1 times

brianccie 11 months, 1 week ago
The correct answer is C , where there will be not QOS remark to be done by the switch as the ip phone is trust boundary , thus all the marking done by the ip phone will be accepted. https://networklessons.com/quality-of-service/how-to-configure-qos-trust-boundary-on-cisco-switches ,
upvoted 3 times

juanmacipag 1 year, 4 months ago
C is correct

Cisco itself says:
QoS classification initially takes place in the Access layer; however, other QoS tools are needed to ensure voice and video quality is maintained throughout the network. In addition to traffic classification, queuing and bandwidth provisioning also ensure voice and video quality.

Therefore, QoS trust boundaries should be set up so that the switch will trust the QoS markings that phones place on their own packets. Layer 2 QoS uses a mechanism called class of service (CoS), which operates on the 802.1Q VLAN.

By default, QoS on a Cisco access switch is disabled. Once enabled, the switch does not trust QoS settings from a phone. Two simple commands can be entered under the global menu on a switch to enable QoS and change the trust boundary. Once it is enabled, you can use a show command to verify these settings. Example 13-1 illustrates the QoS enable and trust boundary commands and the show verification command.
upvoted 4 times

juanmacipag 1 year, 4 months ago
Sorry my Answers is - Packets or Ethernet frames are remarked at the access layer switch.
Not sure if that given answer A has a typo or what.

https://www.cisco.com/c/en/us/td/docs/solutions/Enterprise/WAN_and_MAN/QoS_SRND/QoS-SRND-Book/QoSDesign.html
Example 2-30 Catalyst 2970/3560/3750—Conditionally-Trusted IP Phone + PC + Scavenger (Basic) Model Configuration

CAT2970(config)#mls qos map cos-dscp 0 8 16 24 32 46 48 56
! Modifies CoS-to-DSCP mapping to map CoS 5 to DSCP EF

CAT2970(config)#mls qos map policed-dscp 0 24 to 8

! Excess VVLAN & DVLAN traffic will be remarked to Scavenger (CS1)

upvoted 1 times

Botikus 1 year, 6 months ago

D is correct.

There is no "layer" switch

upvoted 2 times

gottalearnsometime 1 year, 8 months ago

From the reference link: "In the picture above the trust boundary is at the Cisco IP phone, this means that we won't remark any packets or Ethernet frames anymore at the access layer switch."

upvoted 4 times

rishik 1 year, 11 months ago

D is correct

upvoted 1 times

rishik 1 year, 11 months ago

Update C is correct answer

upvoted 8 times

Question #214 *Topic 1*

Why would we not include an end user's PC device in a QoS trust boundary?

A. The end user could incorrectly tag their traffic to bypass firewalls.

B. The end user may incorrectly tag their traffic to be prioritized over other network traffic.

C. There is no reason not to include an end user's PC device in a QoS trust boundary.

D. The end user could incorrectly tag their traffic to advertise their PC as a default gateway.

Correct Answer: *B*

disdain Highly Voted 1 year, 6 months ago

I think the. answer is C. There is no reason not to include an end user's PC device m a QoS trust boundary

Because the first 2 choices are talking about the End User not their PC.

upvoted 5 times

xxCRONOxx Most Recent 10 months, 1 week ago

@disdain

I think they pretty much assume you know they refer to the End Users PC in the question.

upvoted 1 times

webby654 10 months, 2 weeks ago

I think the answer is B

upvoted 1 times

omsssh 1 year, 3 months ago

I think B is a better choice: p.351 CLCOR reference: "Most devices, such as computers and servers, cannot mark their own packets and should not be trusted even if they can. Cisco phones, however, can mark their own packets and can be trusted with the QoS markings they provide."

upvoted 4 times

SDLOA14 1 year ago

I'm going with B also

See what happens to your call quality when a user tags traffic for his PC's bittorrent site or MMRPG with voice EF.

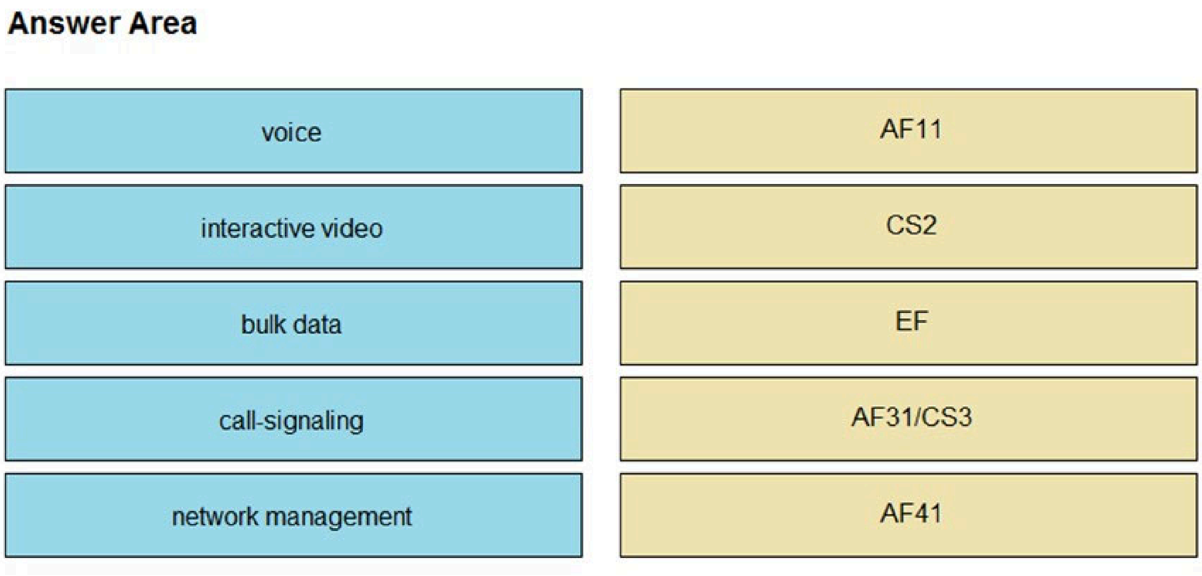
upvoted 1 times

Question #215 *Topic 1*

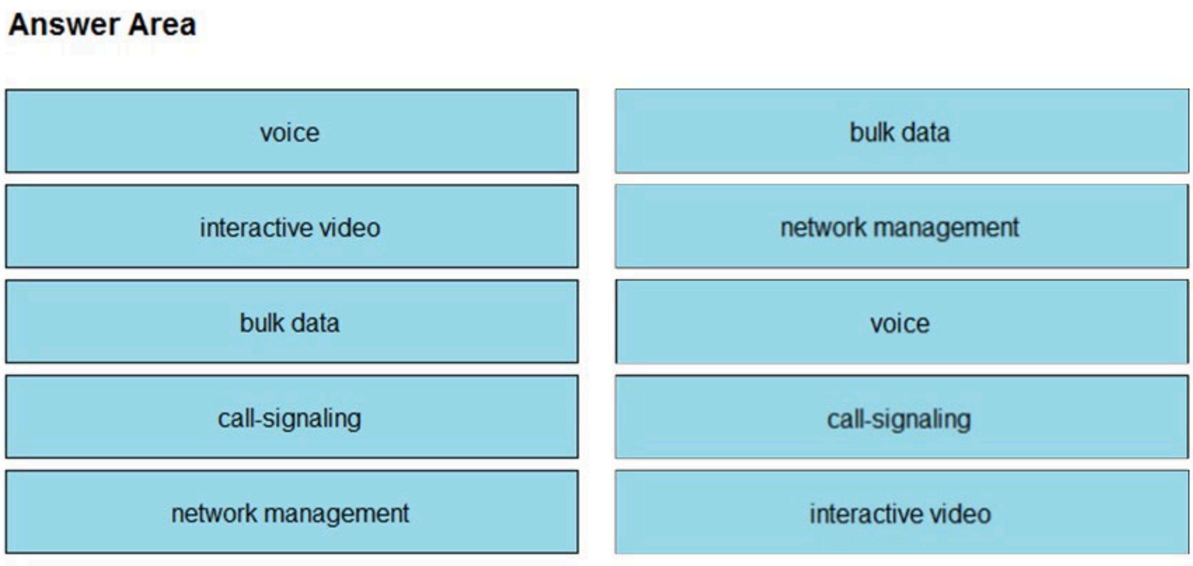
DRAG DROP -

According to the QoS Baseline Model, drag and drop the applications from the left onto the correct Per-Hop Behavior values on the right.

Select and Place:



Correct Answer:



Reference:

https://www.cisco.com/c/en/us/td/docs/solutions/Enterprise/WAN_and_MAN/QoS_SRND/QoS-SRND-Book/QoSIntro.html

b3532e4 10 months, 3 weeks ago

Voice → EF (Expedited Forwarding)

Interactive Video → AF41

Bulk Data → AF11

Call-Signaling → AF31/CS3

Network Management → CS2

upvoted 1 times

Panda_man 2 years, 5 months ago

The given answer is correct

upvoted 1 times

msully 3 years, 8 months ago

CLCOR 350-801 Official Cert Guide, p311-312

QoS Baseline Model (11 Class)

Voice: Refers to voice only, and it is marked with EF and limited to 10 percent of link bandwidth in a strict-priority queue.

Interactive Video: Refers to voice and video, and it is marked with AF41 or sometimes as EF and limited to 13 percent of link bandwidth.

Call Signaling: Marked with CS3 and provisioned with a minimum of 2 percent of link bandwidth.

Network Management: Marked with CS2 and provisioned as guaranteed 2 percent of link bandwidth.

Bulk Data: Marked with AF11 and provisioned with 10 percent of link bandwidth with WRED enabled.

Question #216 *Topic 1*

An engineer with troubleshoots poor voice quality on multiple calls. After looking at packet captures, the engineer notices high levels of jitter. Which two areas does the engineer check to prevent jitter? (Choose two.)

A. The network meets bandwidth requirements.

B. MTP is enabled on the SIP trunk to Cisco Unified Border Element.

C. Cisco UBE manages voice traffic, not data traffic.

D. All devices use wired connections instead of wireless connections.

E. Voice packets are classified and marked.

Correct Answer: *AE*
Reference:
https://www.cisco.com/c/en/us/support/docs/voice/voice-quality/20371-troubleshoot-qos-voice.html

b3532e4 9 months ago
A & E correct
upvoted 2 times

Question #217 *Topic 1*

Which configuration tells a switch port to send Cisco Discovery Protocol packets that configure an attached Cisco IP phone to trust tagged traffic that is received from a device that is connected to the access port on the Cisco IP phone?

A. Router# configure terminal Router(config)# interface gigabitethernet 5/1 Router(config-if)# platform qos trust extend

B. Router# configure terminal Router(config)# interface gigabitethernet 5/1 Router(config-if)# platform qos trust extend cos 3

C. Router# configure terminal Router(config)# interface gigabitethernet 5/1 Router(config-if)# platform qos trust extend cos 5

D. Router# configure terminal Router(config)# interface gigabitethernet 5/1 Router(config-if)# platform qos extend trust

Correct Answer: *A*

Community vote distribution

A (75%) B (25%)

Griswald Highly Voted 4 years ago

The trusted boundary feature is implemented with the platform qos trust extend command

upvoted 10 times

XalaGyan 3 years, 3 months ago

https://www.cisco.com/c/en/us/td/docs/switches/lan/catalyst6500/ios/15-5SY/config_guide/sup6T/15_5_sy_swcg_6T/voip.pdf

Here is the documentation of your correct answer Griswald

upvoted 4 times

Omitted Highly Voted 2 years, 4 months ago

I hate questions like this one. A good engineer doesn't need to know the exact command. Whether it is platform qos trust extend or platform qos extend trust. Some people get words and letters mixed up in their head. But they'll have no issues in front of the CLI for something like this...

upvoted 5 times

G0y0 2 years ago

Actually, whoever is not CCNP does not mean to be bad cisco engineer. However certification it is just a requisite to get the best salaries, see it as a good investment in money and time =).

Of course answer is A, but I am surprised to take a Catalyst 6T as a example of access switch, I was used to see tha command "switchport priority extend trust" in for example switches 3650 hahahahahaha

upvoted 1 times

G0y0 Most Recent 4 months ago

Selected Answer: A

This question is based on the document "Cisco IOS Software Configuration Guide, Release 15.0SY" Chapter 1 Cisco IP Phone Support. This is a document for a Catalyst 6500.

upvoted 1 times

kitty73 11 months ago

Selected Answer: A

To send CDP packets that configure an attached Cisco IP phone to trust tagged traffic received from a device connected to the access port on the Cisco IP phone, do not enter the cos keyword and CoS value.

• To send CDP packets that configure an attached Cisco IP phone to mark tagged ingress traffic received from a device connected to the access port on the Cisco IP phone, enter the cos keyword and CoS value (valid values are 0 through 7).

upvoted 1 times

vfilchev 1 year, 5 months ago

Reading more I suggest that right answer is A:

This example shows how to configure Gigabit Ethernet port 5/1 to send CDP packets that tell the Cisco IP phone to configure its access port as trusted:

Router# configure terminal

Router(config)# interface gigabitethernet 5/1
Router(config-if)# platform qos trust extend
upvoted 2 times

vfilchev 1 year, 5 months ago

Selected Answer: B

I go with B: This example shows how to configure Gigabit Ethernet port 5/1 to send CDP packets that tell the
Cisco IP phone to configure its access port as untrusted and to mark all tagged traffic received from a
device connected to the access port on the Cisco IP phone with CoS 3:
Router# configure terminal
Router(config)# interface gigabitethernet 5/1
Router(config-if)# platform qos trust extend cos 3

https://www.cisco.com/c/en/us/td/docs/switches/lan/catalyst6500/ios/15-5SY/config_guide/sup6T/15_5_sy_swcg_6T/voip.pdf
upvoted 1 times

papahawaii 1 year, 4 months ago

From the link you shared :

This example shows how to configure Gigabit Ethernet port 5/1 to send CDP packets that tell the
Cisco IP phone to configure its access port as trusted:
Router# configure terminal
Router(config)# interface gigabitethernet 5/1
Router(config-if)# platform qos trust extend

Thus A
upvoted 1 times

wwisp3422112 1 year, 7 months ago

Selected Answer: A

"The trusted boundary feature is implemented with the platform qos trust extend command"
upvoted 2 times

Question #218 *Topic 1*

Which recommendation is the best practice for marking video and voice media in a Cisco Unified Communications network?

A. Voice Cos 5 (IP Precedence 6, PHB AF41, or DSCP 16) Video Cos 4 (IP Precedence 5, PHB EF, or DSCP 32)

B. Voice Cos 6 (IP Precedence 4, PHB AF41, or DSCP 24) Video Cos 5 (IP Precedence 4, PHB EF, or DSCP 34)

C. Voice Cos 5 (IP Precedence 2, PHB EF, or DSCP 48) Video Cos 4 (IP Precedence 4, PHB AF41, or DSCP 46)

D. Voice Cos 5 (IP Precedence 5, PHB EF, or DSCP 46) Video Cos 4 (IP Precedence 4, PHB AF41, or DSCP 34)

Correct Answer: *D*

Reference:

https://www.cisco.com/c/en/us/td/docs/voice_ip_comm/cucm/srnd/collab10/collab10/netstruc.html

Community vote distribution

D (100%)

Stevon 3 weeks ago

Selected Answer: D

https://www.cisco.com/c/en/us/td/docs/voice_ip_comm/cucm/srnd/collab10/collab10/netstruc.html

upvoted 1 times

pcp84 10 months, 1 week ago

Selected Answer: D

Voting comment for D.

upvoted 1 times

Panda_man 1 year, 11 months ago

Correct is D

upvoted 1 times

msully 3 years, 2 months ago

CLCOR 350-801 Official Cert Guide, p 317.

Application L3 ToS/IPP PHB DSCP CoS

Voice Only 5 EF 46 5

Voice/Video 4 AF41 34 4

upvoted 3 times

Question #219 *Topic 1*

How can an engineer determine location-based CAC bandwidth requirements for Cisco Unified Communications Manager?

A. Set the requirements in the service parameters.

B. Add the requirements for each audio and video codec and multiply how many calls must be supported.

C. Execute the Resource Reservation Protocol to return location-based requirements.

D. Calculate the number of calls against the license for Cisco Unified Border Element to determine calls per location.

Correct Answer: *B*

Community vote distribution

B (100%)

BarryR Highly Voted 3 years, 5 months ago

Correct answer is B

upvoted 15 times

micbosh Highly Voted 3 years, 6 months ago

I guess B is right here

upvoted 9 times

arinpas Most Recent 8 months, 2 weeks ago

Selected Answer: B

B is correct

upvoted 2 times

lpicardin 9 months, 1 week ago

Selected Answer: B

Based on the location so definitely B

upvoted 4 times

AJBELL14 1 year, 5 months ago

Selected Answer: B

B it is

upvoted 3 times

msully 2 years, 2 months ago

CAC is configured in location, and is the total bandwidth for all calls. B is the answer.

upvoted 3 times

BhaiKyare 2 years, 4 months ago

https://www.cisco.com/c/en/us/td/docs/voice_ip_comm/cucm/admin/11_5_1/sysConfig/CUCM_BK_SE5DAF88_00_cucm-system-configuration-guide-1151/CUCM_BK_SE5DAF88_00_cucm-system-configuration-guide-1151_chapter_011101.html Answer is A

upvoted 1 times

YohanesH 3 years, 2 months ago

My guess is also B

upvoted 6 times

gottalearnsometime 3 years, 3 months ago

Definitely B

upvoted 6 times

Question #220 *Topic 1*

There is a saturated link that has traffic shaping configured. How is incoming traffic processed?

A. Traffic is compressed so that the traffic fits within the bandwidth of the link.

B. Excess traffic is queued, and then dropped after the timer expires.

C. Excess traffic is queued for later transmission.

D. Excess traffic is dropped.

Correct Answer: *C*

Community vote distribution

C (83%) D (17%)

AAMM Highly Voted 2 years, 1 month ago

C is the answer

https://www.cisco.com/c/en/us/support/docs/quality-of-service-qos/qos-policing/19645-policevsshape.html#:~:text=traffic%20shaping%20retains%20excess%20packets%20in%20a%20queue%20and%20then%20schedules%20the%20excess%20f

upvoted 6 times

refriednoodlenl Highly Voted 2 years, 5 months ago

Answer is C. Shaper only drops packets when the bucket overflows, not on a timer.

upvoted 6 times

G0y0 Most Recent 3 months, 3 weeks ago

Selected Answer: C

A. does not have sense.

B. is incorrect, the excess traffic is queued to later transmission, not dropped unless the capacity of the buffer is exceeded but not by a timer.

D. doe snot have sense. Also it is not traffic policy, rather traffic shaper.

C. is correct. Excess traffic is queued for later transmission

upvoted 1 times

Alex234 5 months, 3 weeks ago

Selected Answer: C

https://www.cisco.com/c/en/us/support/docs/quality-of-service-qos/qos-policing/19645-policevsshape.html#:~:text=traffic%20shaping%20retains%20excess%20packets%20in%20a%20queue%20and%20then%20schedules%20the%20excess%20f

upvoted 1 times

m2024 12 months ago

Selected Answer: D

From the link in Q "Only policing can be applied to inbound traffic on an interface."

upvoted 1 times

MaxG 1 year ago

Selected Answer: C

Traffic policing propagates bursts. When the traffic rate reaches the configured maximum rate, excess traffic is dropped (or remarked). The result is an output rate that appears as a saw-tooth with crests and troughs. In contrast to policing, traffic shaping retains excess packets in a queue and then schedules the excess for later transmission over increments of time. The result of traffic shaping is a smoothed packet output rate.

upvoted 1 times

bmne 1 year, 11 months ago

Think it is D:The question says - saturated link that has traffic shaping configured. How is incoming traffic processed.

Shaping is configured on the outbound interface and Policing is Inbound interface, so the question is about policing. It is not B because Policing does not use Queue.

upvoted 4 times

> **G0y0** 3 months, 3 weeks ago
>
> You are right, however, no body is talking about whether if policy is configured or not. traffic shape does make to traffic policy.
>
> upvoted 1 times

iamnoone 2 years, 2 months ago

Selected Answer: C

It's C, will queue packets for retransmission before it drops them.

But based on this "How is incoming traffic processed?" I would say shaping is applied to the egress so this question doesn't make sense.

upvoted 4 times