EXAMTOPICS

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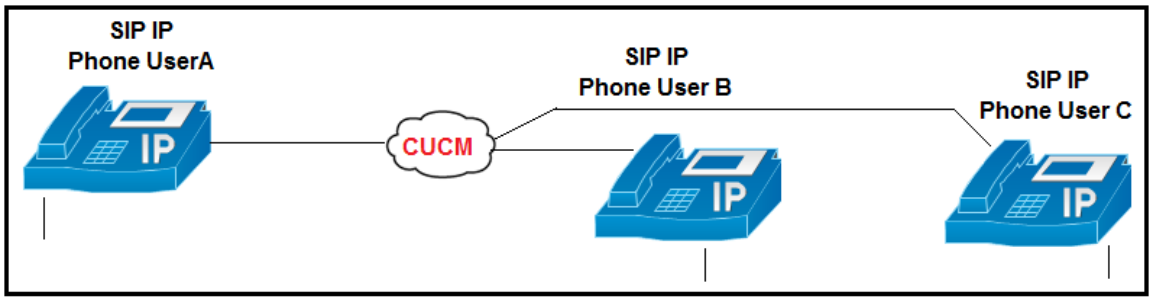
Question #1 *Topic 1*

Refer to the exhibit. In an active SIP call between phone user A and phone user B, phone A initiates a call transfer to phone user C. Which two scenarios are correct? (Choose two.)



A. Phone_A sends a SIP-REFER message to the Cisco Unified Communications Manager with Phone_C information in the Refer-To section.

B. Phone_B sends a SIP-REFER message to the Cisco Unified CM with Phone_C information in the Refer-To section.

C. As soon as Phone_A presses the Transfer button for the first time, Phone_B hears the MOH and the MOH audio is chosen from Phone_B User Hold MOH Audio Source settings.

D. As soon as Phone_A presses the Transfer button for the first time, Phone_B hears the music on hold and the MOH audio is chosen from Phone_A Network Hold MOH Audio Source settings.

E. As soon as Phone_A presses the Transfer button for the first time, Phone_B hears the MOH and the MOH audio is chosen from Phone_A User Hold MOH Audio Source settings.

Suggested Answer: *A, C*
Official Answer: AC

Community vote distribution
AD (75%) AC (25%)

SabSal Highly Voted 5 years, 8 months ago
Answer - A & D
upvoted 7 times

Puh Highly Voted 3 years, 8 months ago
outdated questions, tried to pass and I have around 50 % new question
upvoted 6 times

Littlelarry123 Most Recent 1 year, 3 months ago
Why not A&C?
upvoted 1 times

jatoja83 1 year, 5 months ago
AD, transfer a call is a network MOH, not user MOH
upvoted 1 times

b3532e4 1 year, 9 months ago
Answer A+D Network hold includes the following types: •Call transfer •Call Park •Conference setup •Application-based hold
upvoted 1 times

b7c9010 1 year, 11 months ago
Greetings, can someone please tell me if the Examtopics questions are up to date? It's urgent please.
upvoted 2 times

rsl4u 2 years, 4 months ago
I did on April 10, 2024 the exam 300-815. This was the last week for the version 1.1 of the Exam. There are changes in the next version 300-815 v1.2. I think, I had 60% to 65% of the exam questions (60 questions) which where equal to the questions in this examtopics. You also need to understand a lot of the exam subjects to get enough points from the remaining 40% of new questions in the exam to pass it. I passed the Exam with: 100% on subject Mobility 100% on Gateway Technologies 80% on Cisoc Unified Border Elements 80% on Call Control and Dial Planning 73% on CUCM Call Control Features 75% on Signalling and Media Protocols

upvoted 2 times

john_doe_9999 2 years, 7 months ago

A and D

upvoted 1 times

Wonderboy988 3 years, 4 months ago

Answer - A & D

upvoted 1 times

basscov 4 years, 3 months ago

Answer should be A & E. E because it's user who initiated call transfer and not the other device in the network

upvoted 1 times

enashash 5 years, 2 months ago

A & D User hold: With a user hold, a user presses the Hold button on a phone to explicitly place the caller on hold. If Phone A and Phone B are having a conversation, and the user of Phone B presses the Hold button, Phone B's user hold source is streamed to Phone A. ■ Network hold: A network hold occurs when a call is placed on hold as part of the processing of a supplementary service, such as park, transfer, or conference. If Phone B presses the Transfer button to transfer a call, Phone A still gets placed on hold but hears Phone B's network audio source while the rest of the transfer operations complete.

upvoted 3 times

Marco74 5 years, 3 months ago

Network hold includes transfer, conference, call park, and so forth. I think it's A / D

upvoted 2 times

AgshinA 5 years, 5 months ago

phone A answers the call and then transfers it to phone B (step 2). During the transfer process, phone C receives an MoH stream from the MoH server via the gateway (step 3). It should be A and D

upvoted 2 times

Frank31 5 years, 6 months ago

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 1 times

PVDM 5 years, 7 months ago

since the scenario is TRANSFER, is it should use network hold? https://www.cisco.com/c/en/us/td/docs/voice_ip_comm/cucm/srnd/8x/moh.html

upvoted 2 times

norealchaos 5 years, 9 months ago

Audio source is taken from Holder party, and MoH server from the Holdee MRGL configuration.Holder is Phone A in this case, so the correct answer must be A and E.

upvoted 1 times

norealchaos 5 years, 9 months ago

Audio source is taken from Holder party, and MoH server from the Holdee MRGL configuration.

upvoted 1 times

CCIE_Collab 5 years, 9 months ago

Holder is Phone A in this case, so the correct answer must be A and E.

upvoted 2 times

Question #2 *Topic 1*

Refer to the exhibit. Users report that when they dial to Cisco Unity Connection from an external network, they cannot enter any digits. Assuming only in-band
DTMF is supported, what is a reason for this malfunction?

```
SIP/2.0 200 OK
[..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

ACK sip:+123456789@10.10.20.20:5060 SIP/2.0
[..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
```

A. The negotiated RTP port is outside of the range described by RFC, so inband DTMFs do not work.

B. There is SIP Delayed Offer. DTMF is supported only in Early Offer.

C. The rtpmap:0 value for the negotiated codec is marking DTMF as inactive.

D. No DTMF is negotiated.

Suggested Answer: *D*
Official Answer: D

domangez Most Recent 1 year, 10 months ago
Only (18) G.729/8000 ist negotiated (D is correct)
upvoted 1 times

Question #3 *Topic 1*

The administrator of ABC company is troubleshooting a one-way audio issue for a call that uses H.323 protocol (slow-start mode). The administrator requests that you provide the IP and port information of the Real-Time Transport Protocol traffic that had the one-way audio call. You gather the H.225 and H.245 messages for one of the one-way audio calls. Where can you find the RTP IP and port information for both sides? (Note: This call flow has not invoked any media resources like MTP or transcoders).

A. H.245 Terminal Capability Set

B. H.245 Open Logical Channel

C. H.225 Connect

D. H.245 Open Logical Channel Ack

Suggested Answer: *B*

Reference: http://ccievoicehopeful.blogspot.com/2012/09/h323-notes.html

Community vote distribution

D (100%)

timmyz Highly Voted 3 years, 6 months ago

the right answer is D... the H245 Open Logical Channel Ack has both RTP and RTP port https://community.cisco.com/t5/collaboration-voice-and-video/h323-call-flow/ta-p/3160014

upvoted 6 times

domangez Most Recent 1 year, 10 months ago

H.245 OLC open logical channel - asks the endpoint to open a port for media communication. Endpoint responds with a H.245 OLC ACK message providing the RTP port number it has opened for communication for that call. (D is correct)

upvoted 2 times

john_doe_9999 2 years, 1 month ago

D is correct

upvoted 1 times

hbkmanu 4 years, 4 months ago

The answer should be D, Another Cisco ref for the same https://community.cisco.com/t5/collaboration-voice-and-video/h-323-basic-call-signalling-slow-start/ta-p/3155090

upvoted 2 times

hbkmanu 4 years, 4 months ago

H245 Open Logical Channel contains only RTCP port, H245 Open Logical Channel ACk will have both RTP and RTP Port.
https://community.cisco.com/t5/collaboration-voice-and-video/h323-call-flow/ta-p/3160014

upvoted 3 times

Question #4 *Topic 1*

Which two extended capabilities must be configured on dial peers for fast start-to-early media scenarios (H.323 to SIP interworking)? (Choose two.)

A. DTMF

B. BFCP

C. VIDEO

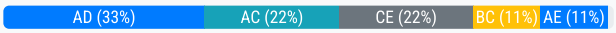
D. FAX

E. AUDIO

Suggested Answer: *A, B*

Official Answer: AB

Community vote distribution



f77380c Most Recent 4 months, 2 weeks ago

Video and BFCP. The actual extended capabilities that you can (and often must) configure are: VIDEO BFCP (Binary Floor Control Protocol) FECC (Far End Camera Control) T38 (for fax)

upvoted 1 times

v1nhthanh 1 year, 3 months ago

Audio is not extended capabilities

upvoted 2 times

jatoja83 1 year, 4 months ago

dtmf+audio - chatgpt said it

upvoted 1 times

decdca7 1 year, 5 months ago

Per Gemini, The extended capabilities that are most relevant to fast start-to-early media scenarios in H.323 to SIP interworking are those that involve media streams beyond basic audio. Therefore, the correct answers are: B. BFCP C. VIDEO

upvoted 2 times

Piji 1 year, 6 months ago

Fast start-to-early media scenarios involve setting up media (audio and/or video) as early as possible in the call signaling process. Audio is mandatory for any call, and video is required if video capabilities are involved. Configuring these capabilities ensures seamless media negotiation between H.323 and SIP endpoints.

upvoted 2 times

kljw5 1 year, 7 months ago

The question specifically ask what extended capabilities are required for fast start to early media scenarios. DTMF and Fax seem to be the correct answer due to rulling out Video and Audio as they are not considered Extended capabilities, BFCP is in fact extended capability as well however BFCP relates to content sharing and is not directly related to transport of audio, dtmf fax or other media streams required for h323 to SIP networking. Considering all the other comments all have different responses can we get some upvotes to validate???

upvoted 1 times

b3532e4 1 year, 10 months ago

Based on the information from the document, the two extended capabilities that must be configured on dial peers for fast start-to-early media scenarios in H.323 to SIP interworking are: DTMF AUDIO These are essential for ensuring proper media flow and signaling interworking between the H.323 and SIP protocols(h323-to-sip).

upvoted 1 times

Piji 2 years ago

Correct answers are A, E: The key point in this question is the requirement for "fast start-to-early media scenarios" in the context of H.323 to SIP interworking. Here's the reasoning: DTMF (Dual-Tone Multi-Frequency): This is crucial for sending dial tones, which are essential in many VoIP signaling scenarios, especially when interworking between different protocols. AUDIO: Audio is fundamental for any call setup. Without configuring audio capabilities, media negotiation cannot proceed correctly. While BFCP (Binary Floor Control Protocol) is important for scenarios involving conference control and content sharing, it is not typically required for basic media interworking scenarios like fast start-to-early media. So, the

revised correct answers should be: A. DTMF E. AUDIO These are the primary capabilities needed to ensure proper media negotiation and interworking in fast start-to-early media scenarios.

upvoted 1 times

FrankPic 2 years, 6 months ago

Fast start-to-early media scenarios refer to situations where two endpoints establish media connections and begin transmitting audio or video before the call signaling is fully completed. This allows for a faster call setup time and improved user experience. When H.323 and SIP protocols are used together for interworking, extended capabilities must be configured on the dial peers to support fast start-to-early media scenarios. Two of these extended capabilities are: BFCP (Binary Floor Control Protocol) AUDIO - This capability enables early media for audio streams, allowing audio to be transmitted even before the call is established. This allows users to hear ringback tones or announcements while waiting for the call to be connected. Therefore, options B (BFCP) and E (AUDIO) are the correct answers to this question. Options A (DTMF), C (VIDEO), and D (FAX) are not required for fast start-to-early media scenarios in H.323 to SIP interworking.@FrankPic - Where are these configured? For audio, I only see that this sets the db level. I cannot find any documentation on where BFCP (mainly used for sharing content during a video call) is configured.

upvoted 2 times

FrankPic 2 years, 6 months ago

Fast start-to-early media scenarios refer to situations where two endpoints establish media connections and begin transmitting audio or video before the call signaling is fully completed. This allows for a faster call setup time and improved user experience.

When H.323 and SIP protocols are used together for interworking, extended capabilities must be configured on the dial peers to support fast start-to-early media scenarios. Two of these extended capabilities are:

BFCP (Binary Floor Control Protocol)

AUDIO - This capability enables early media for audio streams, allowing audio to be transmitted even before the call is established. This allows users to hear ringback tones or announcements while waiting for the call to be connected.

Therefore, options B (BFCP) and E (AUDIO) are the correct answers to this question. Options A (DTMF), C (VIDEO), and D (FAX) are not required for fast start-to-early media scenarios in H.323 to SIP interworking.

upvoted 2 times

Schmidlap 2 years, 6 months ago

@FrankPic - Where are these configured? For audio, I only see that this sets the db level. I cannot find any documentation on where BFCP (mainly used for sharing content during a video call) is configured.

upvoted 2 times

htruesdale 4 years, 9 months ago

Is the answer then A and B because the other choices are not extended capabilities?

upvoted 1 times

htruesdale 4 years, 9 months ago

Everything I have read only references that it is a restriction to configure extended capabilities on dial peers for fast-start to early media. https://www.cisco.com/c/en/us/td/docs/ios-xml/ios/voice/cube/configuration/cube-book/h323-to-sip.html

upvoted 2 times

Question #5 *Topic 1*

When you troubleshoot H.323 call setup, which message informs you that the called party is being notified about the call?

A. ALERTING

B. PROCEEDING

C. CONNECT

D. RINGING

Suggested Answer: *C*

Official Answer: C

Community vote distribution

A (100%)

vishaljhaveri Most Recent 4 months, 3 weeks ago

A. Alerting Here is the breakdown of the H.323 call flow: SETUP: Initiates the call. ALERTING: Sent back to the calling party to indicate that the called party is being notified (ringing). CONNECT: Sent when the called party answers (goes off-hook), notifying the caller that two-way communication is ready.

upvoted 2 times

FCBear 1 year, 11 months ago

The answer is A : Alerting https://community.cisco.com/t5/collaboration-knowledge-base/h323-call-flow/ta-p/3160014

upvoted 2 times

Afaik 2 years ago

The device knows about the call with the proceeding, by alerting the device actually should ring which the question is refering to, connect is to late as the call is answered by that. Ringing should irritate you as we don't have SIP here I guess.

upvoted 1 times

RC31 2 years, 7 months ago

Proceeding, the slow start call flow goes.... Call Proceeding Alerting Connect Keepalives

upvoted 1 times

themis 2 years, 8 months ago

Procceding. The called endpoint will first respond with the Q.931/H.225.0 message Call Proceeding, to indicate it has started working on setting up the call, to the calling endpoint.

upvoted 1 times

Mert_kerna 3 years, 3 months ago

The CONNECT message is sent from the called party to the calling party to inform the caller that the call has been answered.

upvoted 1 times

Mert_kerna 3 years, 3 months ago

The alerting message is an optional status message issued by the called endpoint to the caller. This message confirms that the called endpoint has initiated an indication of the incoming call to the called user, which is that the called phone is ringing. The answer is A; Alerting.

upvoted 1 times

jsantiago2pk 4 years, 12 months ago

The answer is ALERTING for H323. RINGING is for SIP. Question is related to H323

upvoted 3 times

Telcom_Jnk 5 years ago

which message informs you that the called party is being notified ? The answer is RINGING

upvoted 1 times

Grebec94 5 years ago

alerting message when the other end rings

upvoted 3 times

Question #6 *Topic 1*

End users at a new site report being unable to hear the remote party when calling or being called by users at headquarters. Calls to and from the PSTN work as expected. To investigate the SIP signaling to troubleshoot the problem, which field can provide a hint for troubleshooting?

A. Contact: header of the 200 OK response

B. Allow: header if the 200 OK response

C. o= line of SDP content

D. c= line of SDP content

Suggested Answer: *C*

Official Answer: C

Community vote distribution

D (100%)

AKREM86 Highly Voted 4 years, 7 months ago

Correct Answer D. c= attributes contained the connection media information. It contains the IP address that will be used for RTP.

https://tools.ietf.org/html/rfc4566#section-5.7

upvoted 6 times

jatoja83 Most Recent 1 year, 5 months ago

The C Field indicates the IP address where the media RTP should be sent to by the other end.

upvoted 1 times

Afaik 2 years ago

There are two attributes required for each side to negotiate the RTP stream in SDP and this information is passed on the c= for IP and m= for Port and Codec(s).

upvoted 2 times

enashash 4 years, 2 months ago

D C=connection information while o=originator and session identifier

upvoted 1 times

juanmacipag 4 years, 4 months ago

Correct Answer D. c= (Connection/IP address for RTP stream) Reference: SDP Parameter

upvoted 2 times

Question #7 *Topic 1*

Why would RTP traffic that is sent from the originating endpoint fail to be received on the far endpoint?

A. The far end connection data (c=) in the SDP was overwritten by deep packet inspection in the call signaling path.

B. Cisco Unified Communications Manager invoked media termination point resources.

C. The RTP traffic is arriving beyond the jitter buffer on the receiving end.

D. A firewall in the media path is blocking TCP ports 16384-32768.

Suggested Answer: *D*

Official Answer: D

Community vote distribution

A (100%)

Mormon Highly Voted 6 years ago

Answer is A, RTP uses UDP port range

upvoted 14 times

grnmad Most Recent 1 year, 4 months ago

Some routers and firewalls have SIP and H.323 ALG capabilities. ALG is also referred to as Fixup, Inspection, Application Awareness, Stateful Packet Inspection, Deep Packet Inspection, and so forth. This means that the router/firewall is able to identify SIP and H.323 traffic as it passes through and inspect, and in some cases modify, the payload of the SIP and H.323 messages. The purpose of modifying the payload is to help the H.323 or SIP application from which the message originated to traverse NAT; Source:CCNP Collaboration Cloud and Edge Solutions CLCEI 300-820 Official Cert Guide

upvoted 1 times

b3532e4 1 year, 10 months ago

A. The far end connection data (c=) in the SDP was overwritten by deep packet inspection in the call signaling path. Explanation: RTP traffic may fail to reach the far endpoint if the c= line in the SDP (which specifies the media IP address) is modified or overwritten by deep packet inspection (DPI) in the signaling path. This can cause the media to be sent to an incorrect or unreachable IP address, leading to one-way audio issues. Other options, such as firewall blocking, would typically involve UDP ports, not TCP ports.

upvoted 1 times

john_doe_9999 2 years, 7 months ago

I would agree with A

upvoted 1 times

Afaik 3 years ago

A. Is the only reasonfull answer and is also seen in the wild as a common issue, if the sender sends to a wrong destination no audio will be receieved. B. Only if a is happening, normaly if MTP was invoked successfull RTP goes to MTP and then destination. C. It is arriving so not matching the questions. D. Normally this is no problem as RTP should be send via UDP, if it happens by TCP there is already something wrong, also this reference to Cisco most used UDP RTP range the answer makes sense in a weired way but not really due to TCP being mentioned.

upvoted 2 times

TangoDown 3 years, 4 months ago

Answer is C

"If the jitter is so large that it causes packets to be received out of the range of this buffer, the out-of-range packets are discarded and dropouts are heard in the audio. "

https://www.cisco.com/c/en/us/support/docs/voice/voice-quality/18902-jitter-packet-voice.html

upvoted 1 times

TangoDown 3 years, 4 months ago

Answer is C "If the jitter is so large that it causes packets to be received out of the range of this buffer, the out-of-range packets are discarded and dropouts are heard in the audio. " https://www.cisco.com/c/en/us/support/docs/voice/voice-quality/18902-jitter-packet-voice.htmlThe question doesn't indicate drops in audio but "fail to be received" indicating complete audio failure.

upvoted 1 times

kljw5 1 year, 7 months ago

The question doesn't indicate drops in audio but "fail to be received" indicating complete audio failure.

upvoted 1 times

AKREM86 5 years, 1 month ago

Answer is A. Regarding D it is not correct as RTP uses most of the cases UDP instead of TCP for better Data delivery.

upvoted 1 times

movalleuu 5 years, 5 months ago

I think the answer is correct since the question says that packets fail to be delivered, which means packets do not arrive, no matter the jitter time

upvoted 1 times

SabSal 5 years, 8 months ago

Answer C: Check Jitter section in the link given https://www.cisco.com/c/en/us/support/docs/voice-unified-communications/unified-ip-phone-7900-series/7415-telecaster-trouble.html

upvoted 4 times

SabSal 5 years, 8 months ago

Answer C: Check Jitter section in the link given https://www.cisco.com/c/en/us/support/docs/voice-unified-communications/unified-ip-phone-7900-series/7415-telecaster-trouble.htmlI'm going with C as well. If the buffer is too small at the receiving end, it will discard the packets. Thus, the RTP traffic will not be received by the end device.

upvoted 4 times

Jajo 4 years, 2 months ago

I'm going with C as well. If the buffer is too small at the receiving end, it will discard the packets. Thus, the RTP traffic will not be received by the end device.

upvoted 1 times

Question #8 *Topic 1*

An administrator is troubleshooting call failures on an H.323 gateway via the CLI. To see signaling for media and call setup, which debug must the Administrator turn on?

A. debug H.323 messages

B. debug H.225 asn1

C. debug H.246 asn 1

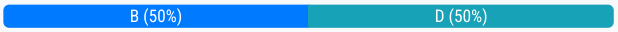
D. debug H.225 media

E. debug H.323 asn 1

Suggested Answer: *B*

Official Answer: B

Community vote distribution



zeroblasted Highly Voted 4 years, 9 months ago

There's no such thing as 'debug h323' commands, nor 'h246.' There's also no h225 or 245 media debug, hence the only correct answer is B. Verified on a cisco ISR4331 CUBE Router.I agree to you. But what the second?

upvoted 5 times

Marco74 4 years, 9 months ago

I agree to you. But what the second?

upvoted 1 times

Akularn Most Recent 8 months, 1 week ago

debug h225 asn1

upvoted 1 times

eran1911 4 years, 4 months ago

in the test need two answer , but only answer exists in router cisco . what is the second answer correct?

upvoted 1 times

jatoja83 1 year, 5 months ago

B and D debug h225 asn1 debug h225 media

upvoted 1 times

domangez 1 year, 10 months ago

- debug cch323 ? all / error / h225 / h245 / session / video / RAS / Capacity ... - debug h225 ? asn1 / events / q931 - debug h245 ? asn1 / events / srtp

upvoted 1 times

zeroblasted 4 years, 9 months ago

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upvoted 4 times

eran1911 4 years, 4 months ago

in the test need two answer , but only answer exists in router cisco . what is the second answer correct?debug h225 asn1 debug h245 asn1

upvoted 1 times

davidanibalmarcelino 4 years, 3 months ago

debug h225 asn1 debug h245 asn1

upvoted 3 times

alert2003 4 years, 10 months ago

real exam require two answers. Answer B and may be E (I can't find any other commands except debug h323-annexg asn1).

upvoted 1 times

AgshinA 4 years, 11 months ago

B is correct: https://www.ccexpert.us/ip-telephony/selected-debug-commands.html

upvoted 1 times

Question #9 *Topic 1*

What is first preference condition matched in a SIP-enabled incoming dial peer?

A. incoming uri

B. target carrier-id

C. answer-address

D. incoming called-number

Suggested Answer: *A*

Reference: 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#anc8

Community vote distribution

A (100%)

Akularn Most Recent 8 months, 1 week ago

A. incoming uri

upvoted 1 times

b3532e4 1 year, 10 months ago

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#toc-hId--209982026 A. incoming uri

upvoted 1 times

b3532e4 1 year, 10 months ago

The correct answer is: A. incoming uri Explanation: In a SIP-enabled incoming dial peer, the first preference condition matched is typically the incoming URI. This allows the router or gateway to match SIP requests based on the Uniform Resource Identifier (URI) provided in the SIP message, making it an important criterion for incoming call handling. Other options like target carrier-id, answer-address, and incoming called-number are valid but generally used after the URI matching.

upvoted 1 times

FrankPic 2 years, 7 months ago

A is correct Inbound SIP Dial-Peer Selection Preference: 1 URI incoming uri via 2 URI incoming uri request 3 URI incoming uri to 4 URI incoming uri from 5 Called Number incoming called-number / incoming called e164-pattern-map 6 Calling Number incoming calling e164-pattern-map / answer-address 7 Destination-pattern (ANI) destination-pattern 8 Carrier-ID carrier-id source

upvoted 2 times

Question #10 *Topic 1*

Cisco SIP IP telephony is implemented on two floors of your company. Afterward, users report intermittent voice issues in calls established between floors. All calls are established, and sometimes they work well, but sometimes there is one-way audio or no audio. You determine that there is a firewall between the floors, and the administrator reports that it is allowing SIP signaling and UDP ports from 20000 to 22000 bidirectionally. What are two possible solutions? (Choose two.)

A. Go to the SIP profile assigned to these IP phones in Cisco Unified CM and change the range of media ports to 16384-32767

B. Ask the firewall administrator to change the ports to TCP.

C. Ask the firewall administrator to change the range of UDP ports to 16384-32767.

D. Go to the SIP profile assigned to these IP phones in Cisco Unified CM and change the range of media ports to 20000-22000.

E. Go to System Parameters in Cisco Unified Communications Manager and change the range of media ports to 20000-22000.

Suggested Answer: *A, C*

Reference: https://www.cisco.com/c/en/us/td/docs/voice_ip_comm/cucm/port/9_1_1/CUCM_BK_T2CA6EDE_00_tcp-port-usage-guide-91/CUCM_BK_T2CA6EDE_00_tcp-port-usage-guide-91_chapter_01.html

Community vote distribution



thili Highly Voted 5 years, 11 months ago

Answer is C,D

upvoted 16 times

Mert_kerna Highly Voted 4 years, 3 months ago

This question is tricky because it's asking for two separate solutions. The wording suggests that it's not looking for two tasks to become one single solution. In this case, one solution is to change the ports on the SIP profile of the phones to match the ports on the firewall. The other solution is to ask the firewall admin to change his ports to match the ports in UCM.I agree, you either change CUCM to match the firewall or change the firewall to match CUCM. C and D

upvoted 7 times

Mert_kerna 4 years, 3 months ago

This question is tricky because it's asking for two separate solutions. The wording suggests that it's not looking for two tasks to become one single solution. In this case, one solution is to change the ports on the SIP profile of the phones to match the ports on the firewall. The other solution is to ask the firewall admin to change his ports to match the ports in UCM.

upvoted 7 times

RC31 3 years, 7 months ago

I agree, you either change CUCM to match the firewall or change the firewall to match CUCM. C and D

upvoted 3 times

Akularn Most Recent 8 months, 1 week ago

Answer is C,D

upvoted 1 times

b3532e4 1 year, 10 months ago

Both FW and sip profiles must have the same port, so A and C is the correct answer If we need an answer, just D is fine

upvoted 1 times

729edae 2 years, 3 months ago

C and D are the answers. Mert,kerna is correct, they are looking for a single solution. Either change the FW or change the SIP Profile.

upvoted 1 times

john_doe_9999 2 years, 7 months ago

C and D

upvoted 2 times

Afaik 3 years ago

If you change the ports on the CUCM for the devices to use different ones they need to match the ones that can be used on the firewall so A) is wrong and D) is correct.

upvoted 1 times

7ArchAngel7 3 years, 3 months ago

C and D...why are there so many wrong on this site?

upvoted 1 times

7ArchAngel7 3 years, 3 months ago

C and D...why are there so many wrong on this site?16384 - 32767 is default cisco use of ephemeral

upvoted 1 times

7ArchAngel7 3 years, 3 months ago

16384 - 32767 is default cisco use of ephemeral

upvoted 1 times

AgshinA 5 years, 5 months ago

It is correct, A and C. Both must be 16384 - 32767. No need to configure 16384 - 32767 in the firewall but 20000 - 22000 in CM. They should be the same.

upvoted 1 times

AgshinA 5 years, 5 months ago

It is correct, A and C. Both must be 16384 - 32767. No need to configure 16384 - 32767 in the firewall but 20000 - 22000 in CM. They should be the same.if the firewall admin has permitted only udp 20000 until 22000, how can changing something on cucm affect that? C and D are correct.

upvoted 1 times

XalaGyan 5 years, 4 months ago

if the firewall admin has permitted only udp 20000 until 22000, how can changing something on cucm affect that? C and D are correct.

upvoted 7 times

Question #11 *Topic 1*

Which section under the Real-Time Monitoring Tool allows for reviewing the call flow and signaling for a SIP call in real time?

A. Analysis Manager > Inventory > Trace File Repositories

B. System > Tools > Trace and Log Central

C. Voice/Video > Session Trace Log View > Real Time Data

D. Voice/Video > Session Trace Log View > Open From Local Disk

Suggested Answer: *C*
Reference: https://www.cisco.com/c/en/us/support/docs/unified-communications/unified-communications-manager-callmanager/213583-procedure-to-analyse-call- flow-of-sip-ca.html

Marco74 Highly Voted 4 years, 9 months ago
The correct answer is C. What do you think reading that link? https://www.cisco.com/c/en/us/support/docs/unified-communications/unified-communications-manager-callmanager/213583-procedure-to-analyse-call-flow-of-sip-ca.html
upvoted 5 times

Marco74 Highly Voted 4 years, 9 months ago
The correct answer is C. What do you think reading that link? https://www.cisco.com/c/en/us/support/docs/unified-communications/unified-communications-manager-callmanager/213583-procedure-to-analyse-call-flow-of-sip-ca.htmlYes. C is the correct answer. All steps ftom your link: System > Voice/Video > SIP activity > Session trace log view > Real Time data
upvoted 5 times

dansecu 4 years, 9 months ago
Yes. C is the correct answer. All steps ftom your link: System > Voice/Video > SIP activity > Session trace log view > Real Time data
upvoted 1 times

domangez Most Recent 1 year, 10 months ago
System > Tools > Trace and Log Central >> Real Time Trace > View Real Time Data shows the Logs as the come in real time. Voice/Video > Session Trace Log View > Real Time Data loads saved Logs for selected period (max 60 minutes long) in the past.
upvoted 1 times

htruesdale 4 years, 3 months ago
I think the correct answer is C. Cisco labs for this course focus on gathering info as described by C.
upvoted 1 times

alert2003 4 years, 10 months ago
Correct answer is B
https://www.cisco.com/c/en/us/td/docs/voice_ip_comm/cups/7_0/english/serviceability/administration/guide/rttlc.html#wp1148063Real-Time data in section Voice/Video is located: Voice/Video->Call Process->Session Trace log ViewTested on CUCM: Really B- because in C answer you can see data only after call, but in B you can see the call progress in real-timeI think the most appropriate answer is C . Reason - B (You need to gather the logs and analyze with tools) C (You can view the SIP call flow etc from RTMT itself, no need of any tool to find the call flow and signaling)
upvoted 1 times

alert2003 4 years, 10 months ago
Correct answer is B
https://www.cisco.com/c/en/us/td/docs/voice_ip_comm/cups/7_0/english/serviceability/administration/guide/rttlc.html#wp1148063
upvoted 1 times

alert2003 4 years, 10 months ago
Real-Time data in section Voice/Video is located: Voice/Video->Call Process->Session Trace log View
upvoted 1 times

alert2003 4 years, 10 months ago
Real-Time data in section Voice/Video is located: Voice/Video->Call Process->Session Trace log ViewTested on CUCM: Really B- because in C answer you can see data only after call, but in B you can see the call progress in real-timeI think the most appropriate answer is C . Reason - B (You need to gather the logs and analyze with tools) C (You can view the SIP call flow etc from RTMT itself, no need of any tool to find the call flow and signaling)
upvoted 1 times

alert2003 4 years, 10 months ago

Tested on CUCM: Really B- because in C answer you can see data only after call, but in B you can see the call progress in real-timeI think the most appropriate answer is C . Reason - B (You need to gather the logs and analyze with tools) C (You can view the SIP call flow etc from RTMT itself, no need of any tool to find the call flow and signaling)

upvoted 1 times

alert2003 4 years, 10 months ago

Tested on CUCM: Really B- because in C answer you can see data only after call, but in B you can see the call progress in real-time

upvoted 1 times

hbkmanu 4 years, 4 months ago

I think the most appropriate answer is C . Reason - B (You need to gather the logs and analyze with tools) C (You can view the SIP call flow etc from RTMT itself, no need of any tool to find the call flow and signaling)

upvoted 1 times

Question #12 *Topic 1*

Which description of RTP timestamps or sequence numbers is true?

A. The sequence number is used to detect losses.

B. Timestamps increase by the time "carrying" by a packet.

C. Sequence numbers increase by four for each RTP packet transmitted.

D. The timestamp is used to place the incoming audio and video packets in the correct timing order (playout delay compensation).

Suggested Answer: *D*

Reference: https://www.cs.columbia.edu/~hgs/rtp/faq.html

Community vote distribution

D (100%)

AgshinA Highly Voted 5 years, 5 months ago

The timestamp is used to place the incoming audio and video packets in the correct timing order (playout delay compensation). The sequence number is mainly used to detect losses. Sequence numbers increase by one for each RTP packet transmitted, timestamps increase by the time "covered" by a packet. For video formats where a video frame is split across several RTP packets, several packets may have the same timestamp. In some cases such as carrying DTMF (touch tone) data (RFC 2833), RTP timestamps may not be monotonic. D is corectso also A is correct. The sequence number is mainly used to detect lossesRight, but this is a Cisco exam. If it's not an absolute answer to the exact question, and there's another answer that is more correct, it's wrong.

upvoted 8 times

AgshinA Highly Voted 5 years, 5 months ago

The timestamp is used to place the incoming audio and video packets in the correct timing order (playout delay compensation). The sequence number is mainly used to detect losses. Sequence numbers increase by one for each RTP packet transmitted, timestamps increase by the time "covered" by a packet. For video formats where a video frame is split across several RTP packets, several packets may have the same timestamp. In some cases such as carrying DTMF (touch tone) data (RFC 2833), RTP timestamps may not be monotonic.

D is corect

upvoted 8 times

stocaxx 4 years, 6 months ago

so also A is correct. The sequence number is mainly used to detect lossesRight, but this is a Cisco exam. If it's not an absolute answer to the exact question, and there's another answer that is more correct, it's wrong.

upvoted 1 times

stocaxx 4 years, 6 months ago

so also A is correct. The sequence number is mainly used to detect losses

upvoted 1 times

Mert_kerna 4 years, 3 months ago

Right, but this is a Cisco exam. If it's not an absolute answer to the exact question, and there's another answer that is more correct, it's wrong.

upvoted 2 times

Piji Most Recent 1 year, 6 months ago

Why D is correct: The RTP timestamp is primarily used for playout delay compensation, which means it ensures that audio and video packets are played back at the correct time. It helps synchronize media streams and smooth out network jitter (variations in packet arrival time). Without timestamps, audio and video playback would be choppy or out of sync. Why A is not the best choice: The RTP sequence number does help detect packet loss, but its primary function is not related to timing or playback order. The sequence number only detects missing packets (e.g., gaps in numbering), but it does not help with timing adjustments or delay compensation. If packets arrive out of order, the sequence number alone cannot fix playback timing—timestamps are required for that.

upvoted 2 times

ricsterr 1 year, 9 months ago

A. is also true in that the sequence number is used to detect packet loss, but D is the most relevant to the timestamp's purpose.

upvoted 1 times

b3532e4 1 year, 10 months ago

Definitely A

upvoted 1 times

FCBear 2 years, 10 months ago

Why D and not A, because A is not a description. D is the exact description

upvoted 1 times

Question #13 *Topic 1*

A support engineer is troubleshooting a voice network. When conducting a search for call setup details related to calling search space issues, which trace files should be investigated?

A. CallManager traces

B. CTI Manager traces

C. Cisco IP Manager Assistant

D. Call logs

Suggested Answer: *A*
Official Answer: A

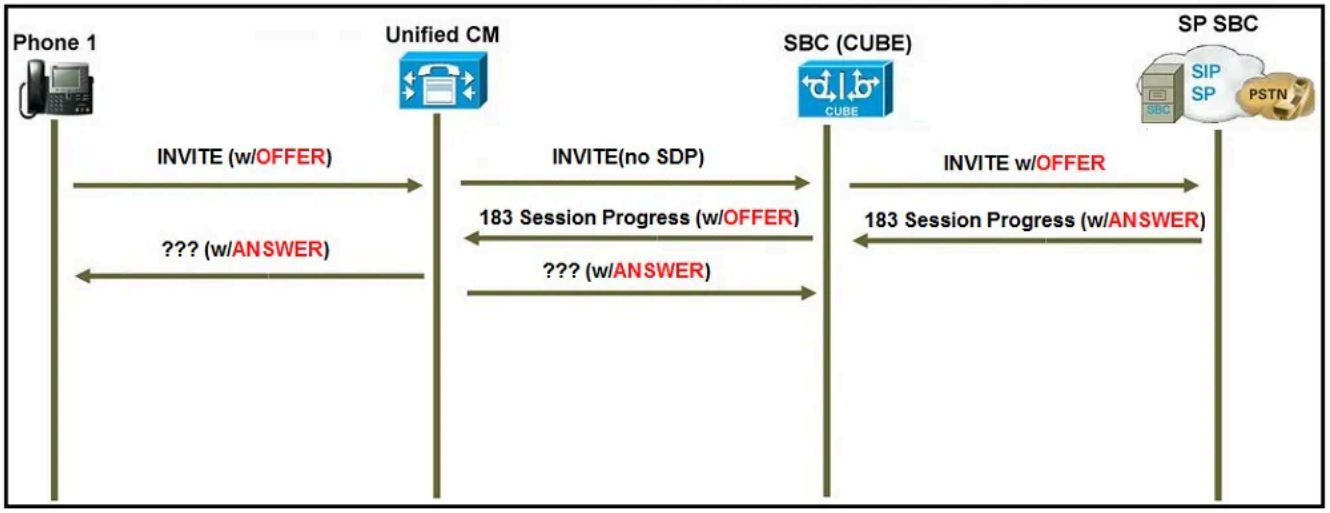
Community vote distribution
A (100%)

SabSal Highly Voted 2 years, 2 months ago
Answer : A https://www.cisco.com/c/en/us/support/docs/voice-unified-communications/unified-communications-manager-callmanager/22325-part-css-tn.html
upvoted 7 times

b3532e4 Most Recent 1 year, 7 months ago
Answer : A When you look into the Cisco CallManager traces, this line displays when a phone attempts to make a call: Digit analysis: match(fqcn="2001", cn="2001", pss=":Internal:No-International", dd="") 'cn' stands for the calling number. In this case, it is 2001. 'pss' stands for partition search space, and has the information about the partition contained in the CSS assigned to the phone.)
upvoted 2 times

Question #14 *Topic 1*

Refer to the exhibit. A user reports that when they call a specific phone number, no one answers the call, but when they call from a mobile phone, the call is answered. The engineer troubleshooting the issue is expecting the far-end gateway to cut through audio on the 183 Session Progress SIP message. Which SIP
Profile configuration element is necessary for the Cisco Unified Communications Manager to send acknowledgement of provisional responses?



A. Allow Passthrough of Configured Line Device Caller Information must be enabled.

B. Accept Audio Codec Preferences in Received Offer must be set to On.

C. On the SIP Profile, the configuration parameter SIP Rel1XX Options must be set to Send PRACK for all 1xx Messages.

D. Early Offer for G Clear Calls must be enabled.

Suggested Answer: *C*
Official Answer: C

Community vote distribution
C (100%)

SabSal Highly Voted 5 years, 8 months ago
Answer C: https://community.cisco.com/t5/ip-telephony-and-phones/ring-back-while-183-session-progress/m-p/2813529#M315884
upvoted 5 times

f1ab921 Most Recent 1 year, 10 months ago
This configures cucm to send pracks to all 1xx messages via the sip trunk to which this sip profile is assigned.
upvoted 2 times

Question #15 *Topic 1*

A company has an SRST gateway running an IOS XE image. The company plans to enable the IPv6 addressing companywide. To enable the IPv6 in a unified
SRST gateway to support SIP phones, what are two supported supplementary features for an IPv6 fallback scenario? (Choose two.)

A. three-way conference

B. secure SIP lines

C. T.38 fax relay

D. transcoding

E. SIP trunk

Suggested Answer: *A, C*
Reference:
https://www.cisco.com/c/en/us/td/docs/voice_ip_comm/cusrst/admin/sccp_sip_srst/configuration/guide/SCCP_and_SIP_SRST_Admin_Guide/ srst_sip_isr4000.html

Community vote distribution
AC (100%)

b3532e4 Most Recent 1 year, 7 months ago
Selected Answer: AC
upvoted 1 times

Afaik 2 years ago
https://www.cisco.com/c/en/us/td/docs/voice_ip_comm/cusrst/admin/sccp_sip_srst/configuration/guide/SCCP_and_SIP_SRST_Admin_Guide/srst_sip_isr4000.html
upvoted 1 times

AgshinA 4 years, 5 months ago
The following supplementary services are supported as part of IPv6 in Unified SRST IP Phones: Hold/Resume Call Forward Call Transfer Three-way Conference (with BIB conferencing only) Line to T1/E1 Trunk and Trunk to Line with Supplementary Service Features Fax to and from PSTN (IPv4 ATA to ISDN T1/E1) for both T.38 Fax Relay and Fax Passthrough So A and C is correct
upvoted 4 times

Question #16 *Topic 1*

Which action is correct with respect to toll fraud prevention configuration in the Cisco Unified Communications Manager Express?

A. Configure Direct Inward Dial for Incoming ISDN Calls with overlap dialing.

B. Configure IP Address Trusted Authentication for Incoming VoIP Calls.

C. Configure the command no ip address trusted authenticate under "voice service voip".

D. Enable Secondary Dial tone on Analog and Digital FXO Ports.

Suggested Answer: *B*

Reference: https://www.cisco.com/c/en/us/td/docs/voice_ip_comm/cucme/admin/configuration/manual/cmeadm/cmetoll.html#concept_ECC4F4E7ED0F45C594B703EEF34762F2

Community vote distribution

B (100%)

Mert_kerna Highly Voted 1 year, 9 months ago

Correct answer is, in fact B. Be careful on these questions, because the answers may differ between SRST and CUCME configs.

upvoted 5 times

decdca7 Most Recent 1 year, 5 months ago

Only B

upvoted 2 times

alert2003 2 years, 10 months ago

Correct is B.

https://www.cisco.com/c/en/us/td/docs/voice_ip_comm/cucme/admin/configuration/manual/cmeadm/cmetoll.html#concept_E435095B0E2845C78D60332753288337 * Direct-inward-dial isdn is not supported for incoming ISDN overlap dialing call.

upvoted 4 times

Question #17 *Topic 1*

You see the voice register pool 1 command in your Cisco Unified Communications Manager Express configuration. Which configuration is occurring in this section?

A. configuration for a single SIP phone

B. configuration items common for all SIP phones

C. configuration for a pool of SIP phones (similar to device pool on Cisco Unified Communications Manager)

D. configuration for SIP registrar service

Suggested Answer: *C*
Reference:
https://www.cisco.com/c/en/us/td/docs/voice_ip_comm/cusrst/admin/sccp_sip_srst/configuration/guide/SCCP_and_SIP_SRST_Admin_Guide/ srst_setting_up_using_sip.html

Community vote distribution
A (100%)

akosmakos Highly Voted 5 years, 10 months ago
According to the the CME command reference, the answer should be 'A': To enter voice register pool configuration mode and create a pool configuration for a SIP IP phone in Cisco Unified CME or for a set of SIP phones in Cisco Unified SIP SRST, use the voice register pool command in global configuration mode.
https://www.cisco.com/c/en/us/td/docs/voice_ip_comm/cucme/command/reference/cme_cr/cme_v1ht.html#wp2339729225
upvoted 8 times

thili Highly Voted 5 years, 11 months ago
Answer should be A
upvoted 6 times

ricsterr Most Recent 1 year, 9 months ago
The correct answer is: A. configuration for a single SIP phone In Cisco Unified Communications Manager Express (CME), the voice register pool command is used to configure settings specific to a single SIP phone. Each pool represents a single phone registration, including the phone's number, username, and password, as well as other device-specific parameters like the button layout or features. B refers to configuration items common for all SIP phones, which is done in the voice register global section. C is incorrect, as the voice register pool command is for configuring individual phones, not a group or pool of phones. D relates to the SIP registrar service but is not directly tied to the voice register pool command.
upvoted 1 times

b3532e4 1 year, 9 months ago
voice register pool 1 id mac F029.2959.558F type 7945 number 1 dn 1 username simon password 1234 description SIMON codec g711ulaw no vad
upvoted 1 times

f1ab921 1 year, 10 months ago
Option C is so wrong.
upvoted 1 times

729edae 2 years, 3 months ago
Answer is A. Even though you could use the id-network command to configure a single Voice Register Pool for multiple devices SRST or otherwise, the answer is A. Very Cisco...
upvoted 1 times

MaxG 3 years ago
Command syntax is "voice register pool pool-tag". For Cisco Unified CME systems, the "pool-tag" specifies the max number of phones in the pool. Upper limit for this argument is defined by the max-pool command. Use "max-pool" command to set phone-specific parameters for SIP phones in a Cisco Unified CME system. Before using this command, enable the "mode cme" command and set the maximum number of SIP phones supported in your system by using the "max-pool" command.
https://www.cisco.com/c/en/us/td/docs/voice_ip_comm/cucme/command/reference/cme_cr/cme_v1ht.html#wp2339729225
upvoted 2 times

enashash 5 years, 2 months ago

A o enter voice register pool configuration mode and create a pool configuration for a SIP IP phone in Cisco Unified CME or for a set of SIP phones in Cisco Unified SIP SRST, use the voice register pool command in global configuration mode. To remove the pool configuration, use the no form of this command. REF: https://www.cisco.com/c/en/us/td/docs/voice_ip_comm/cucme/command/reference/cme_cr/cme_v1ht.html#wp2339729225

upvoted 1 times

ALLENN 5 years, 11 months ago

Answer is D . Backup registrar service to SIP IP phones can be provided by configuring a voice register pool on SIP gateways. https://www.cisco.com/c/en/us/td/docs/voice_ip_comm/cusrst/admin/sccp_sip_srst/configuration/guide/SCCP_and_SIP_SRST_Admin_Guide/srst_setting_up_using_sip.html

upvoted 1 times

ALLENN 5 years, 11 months ago

Answer is D . Backup registrar service to SIP IP phones can be provided by configuring a voice register pool on SIP gateways. https://www.cisco.com/c/en/us/td/docs/voice_ip_comm/cusrst/admin/sccp_sip_srst/configuration/guide/SCCP_and_SIP_SRST_Admin_Guide/srst_setting_up_using_sip.htmlThe sip registrar service is configured under voice service voip > sip. The voice register pool command identifies configuration related to a specific pool member. Correct answer is A.No, what you're referring to is SRST, which is able to fail over a pool of phones based on a subnet provided in the configuration. Within the parameters of CME, it is intended only for a single phone by specifying the MAC address of the phone.

upvoted 1 times

zeroblasted 5 years, 3 months ago

The sip registrar service is configured under voice service voip > sip. The voice register pool command identifies configuration related to a specific pool member. Correct answer is A.

upvoted 1 times

Mert_kerna 4 years, 3 months ago

No, what you're referring to is SRST, which is able to fail over a pool of phones based on a subnet provided in the configuration. Within the parameters of CME, it is intended only for a single phone by specifying the MAC address of the phone.

upvoted 2 times

Question #18 *Topic 1*

Which top-level IOS command is needed to begin the configuration of a Cisco Unified Communications Manager Express gateway to enable phones to be registered via SIP?

A. allow-connections sip to sip

B. voice service voip

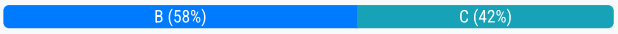
C. voice register global

D. voice register dn

Suggested Answer: *C*

Reference: https://www.cisco.com/c/en/us/support/docs/voice-unified-communications/unified-communications-manager-express/99946-cme-sip-guide.html

Community vote distribution



jn4voip Highly Voted 5 years, 4 months ago

Answer should be B. You have to enable the registrar service before you can provide the address in voice register global voice service voip allow-connections sip to sip !--- Enable SIP to SIP calls. sip registrar server expires max 1200 min 300 !--- Enable Cisco IOS SIP registrar.

upvoted 6 times

Afaik Highly Voted 3 years ago

B and C are both global commands, to use C you need to enter B, but the questions aimes at phone registration, B would be correct basically if the beginn would be a stronger word like first step I assume.

upvoted 5 times

729edae Most Recent 1 year, 4 months ago

This is a Cisco Exam...so the answer is B. IRL...we all know that you can access the "global" SIP config from (conf-voi-serv) but, "to allow for Registration/Provisioning" that is done via the Voice Register. SIP services can be configured on the Dial-Peer. Another Cisco "gotcha" with impunity. The exam is not only testing your VOIP skills but, your reading comprehension. (Insert side-eye emoji).

upvoted 1 times

kljw5 1 year, 4 months ago

To enable phones to register via SIP on a Cisco Unified Communications Manager Express (CME) gateway, the first required top-level command is: arduino Copy Edit Router(config)# voice service voip Why This is Correct: voice service voip is the global configuration mode that enables and configures VoIP services on the router. Inside this mode, you can configure protocols such as SIP and H.323. It is a mandatory starting point before configuring SIP registration or dial-peers.

upvoted 1 times

Piji 1 year, 7 months ago

B. Voice Service Voip

upvoted 1 times

f1ab921 1 year, 10 months ago

The most voted question is actually B but C is shown as the most voted... strange. The answer is B, because the registrar server command is configured under the voice service voip command. The registrar server command is required to allow the cme to process SIP REGISTER requests.

upvoted 1 times

b3532e4 1 year, 10 months ago

Top-Level command Voice service Voip

upvoted 1 times

Piji 2 years ago

The correct answer is B: voice service voip: This is the top-level command used to enter the VoIP configuration mode, where you can configure protocols such as SIP and H.323. From this mode, you set global parameters for SIP, including codec settings, call admission control, and other general VoIP services. This command is necessary to begin configuring the CME gateway for SIP registration. voice register global: This command is used after entering the VoIP configuration mode (voice service voip) and is specifically for configuring SIP phones in CME. It defines global parameters for SIP registrations, such as the maximum number of phones, system messages, and timers for SIP phones. While this command is essential for SIP phone configuration, it comes after you've initiated VoIP configuration with voice service voip. Therefore, voice service voip (B) is the

top-level command needed to start the SIP configuration, and voice register global (C) is a subsequent step in configuring SIP phone registrations within CME.

upvoted 2 times

Piji 2 years ago

Correct answer is B.

upvoted 1 times

john_doe_9999 2 years, 7 months ago

Going with B as it would be the first thing you'd need to do.

upvoted 2 times

enashash 5 years, 2 months ago

B is the correct answer

upvoted 2 times

zeroblasted 5 years, 3 months ago

Correct answer is B, you have to enter 'voice service voip' to begin to configure CUBE as a registrar server

upvoted 3 times

Question #19 *Topic 1*

For s SIP to SIP call flow, when does Cisco Unified Border Element require transcoding resources for DTMF?

A. interworking between an OOB method and RFC2833 for flow-around calls

B. interworking between h245-signal and rtp-nte

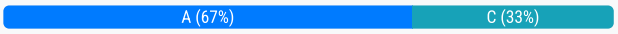
C. interworking between an OOB method and RFC2833 for flow-through calls

D. interworking between h245-alpha numeric and sip-kpml

Suggested Answer: *A*

Reference: https://www.cisco.com/c/en/us/support/docs/unified-communications/unified-border-element/200412-DTMF-Relay-and-Interworking-on- CUBE.html#anc35

Community vote distribution



zeroblasted Highly Voted 5 years, 3 months ago

I stand corrected, A is correct CUBE requires transcoding resources registered locally in these scenarios ... Interworking between an OOB method and RFC2833 for flow-around calls https://www.cisco.com/c/en/us/support/docs/unified-communications/unified-border-element/200412-DTMF-Relay-and-Interworking-on-CUBE.html

upvoted 6 times

AgshinA Highly Voted 5 years, 5 months ago

CUBE requires transcoding resources registered locally in these scenarios Interworking between RFC2833 and Voice In-band Interworking between an OOB method and RFC2833 for flow-around calls CUBE is able to interwork between all other DTMF relay methods with flow-through calls without the need for a transcoder. A is correct

upvoted 5 times

Gary1968 Most Recent 2 months, 3 weeks ago

https://www.cisco.com/c/en/us/support/docs/unified-communications/unified-border-element/200412-DTMF-Relay-and-Interworking-on-CUBE.html#anc35 hen Does CUBE Require Transcoding Resources for DTMF CUBE requires transcoding resources registered locally in these scenarios Interworking between RFC2833 and Voice In-band Interworking between an OOB method and RFC2833 for flow-around calls CUBE is able to interwork between all other DTMF relay methods with flow-through calls without the need of a transcoder.

upvoted 1 times

f77380c 4 months, 2 weeks ago

Trancode resources are needed specifically: When there is DTMF interworking between an Out-of-Band (OOB) method (such as SIP-KPML, SIP-Notify, or H.245) and RFC 2833 (rtp-nte), and the call is using flow-through mode.

upvoted 1 times

decdca7 1 year, 5 months ago

When Does CUBE Require Transcoding Resources for DTMF CUBE requires transcoding resources registered locally in these scenarios Interworking between RFC2833 and Voice In-band Interworking between an OOB method and RFC2833 for flow-around calls CUBE is able to interwork between all other DTMF relay methods with flow-through calls without the need of a transcoder.

upvoted 1 times

b3532e4 1 year, 9 months ago

Supported In-band DTMF Relay Methods In-band audio DTMF through G711 RFC2833 Supported Out-Of-Band DMTF Relay Methods H.245 Alphanumeric H.245 Signal SIP Unsolicited NOTIFY SIP KPML SIP INFO Correct answer C , (media flow-around call , flow-through calls)

upvoted 1 times

FCBear 2 years, 11 months ago

Correct answer is A. @zeroblasted : We're not talking about transcoding codecs but DTMF relay. Interworking between an OOB method and RFC2833 for flow-around calls CUBE is able to interwork between all other DTMF relay methods with flow-through calls without the need of a transcoder.

upvoted 1 times

zeroblasted 5 years, 3 months ago

media flow-around indicates the media does NOT flow through the CUBE but rather directly between UAC and UAS. Hence, you cannot transcode resources that don't touch the CUBE. Correct answer is C

upvoted 4 times

Question #20 *Topic 1*

Where is the dtmf-relay command configured on Cisco Unified Border Element?

A. in the voice-class VoIP configuration

B. in the VoIP dial peer

C. in global SIP configuration

D. in the VoIP or POTS dial peers

Suggested Answer: *B*
Reference: https://www.cisco.com/c/en/us/td/docs/ios-xml/ios/voice/cube/configuration/cube-book/dtmf-relay.html

Aravi1234 Most Recent 1 year, 10 months ago

B - in Dial-peer

upvoted 3 times

Question #21 *Topic 1*

Refer to the exhibit. Calls incoming from the provider are not working through newly set up Cisco Unified Border Element. Provider engineers get the 404 Not
Found SIP message. Incoming calls are coming from the provider with called number "222333444" and Cisco Unified Communications Manager is expecting the called number to be delivered as "444333222". The administrator already verified that the IP address of the Cisco Unified CM is set up correctly and there are no dial peers configured other than those shown in the exhibit. Which action must the administrator take to fix the issue?

```
voice translation-profile incoming
     translate called 999
!
voice translation-rule 999
     rule 1/\ (^[1-2] [1-2] [1-2]\ ) 333\ ([4-5] [4-5] .\) $ /  / \2333\1/
!
dial-peer voice 999 voip
     translation-profile outgoing incoming
     session protocol sipv2
     incoming called-number
     dtmf-relay rtp-nte
     codec transparent
     destination dpg 888
     no vad
!
voice class dpg 888
     dial-peer 888
!
dial-peer voice 888 voip
     destination-pattern 888
     session protocol sipv2
     session target ipv4:192.168.0.1
     codec transparent
     dtmf-relay rtp-nte
     no vad
```

A. Change the destination-pattern on the outgoing dial peer to match "444333222".

B. Set up translation-profile on the incoming dial peer to match incoming traffic.

C. Create specific matching for "222333444" on the incoming dial peer.

D. Fix the voice translation-rule to match specifically number "222333444" and change it to "444333222".

Suggested Answer: *B*
Official Answer: B

Community vote distribution
B (50%) A (25%) C (12%) D (12%)

zeroblasted Highly Voted 5 years, 3 months ago
Correct answer is C - The configuration is intentionally missing the incoming called-number. If you try to leave that blank on a router it will reject the command. The dial peer group DOES NOT have to match the outgoing digits because it is associated. And the translation rule was written correctly (minus the spaces). Verified on a 4331 - voice translation-rule 999 rule 1 /\(^[1-2][1-2][1-2]\)333\([4-5][4-5].\)$/ /\2333\1/ #test voice translation-rule 999 222333444 Matched with rule 1 Original number: 222333444 Translated number: 444333222No this would output 2333222i can't understand how does this rule matches this pattern??
upvoted 7 times

zeroblasted Highly Voted 5 years, 3 months ago
Correct answer is C - The configuration is intentionally missing the incoming called-number. If you try to leave that blank on a router it will reject the command. The dial peer group DOES NOT have to match the outgoing digits because it is associated. And the translation rule was written correctly (minus the spaces). Verified on a 4331 -

voice translation-rule 999 rule 1 /\(^[1-2][1-2][1-2]\)333\([4-5][4-5].\)$/ /\2333\1/ #test voice translation-rule 999 222333444 Matched with rule 1 Original number: 222333444 Translated number: 444333222

upvoted 7 times

jimboo 4 years, 7 months ago

No this would output 2333222

upvoted 1 times

Godan 4 years, 8 months ago

i can't understand how does this rule matches this pattern??

upvoted 1 times

729edae Most Recent 1 year, 4 months ago

rule 1 /|(^[1-2][1-2][1-2]\) 333\ ([4-5][4-5].\)$/ /\2333\1/ The translation pattern creates a Grouping so its right and it works: rule 1 /| (^[1-2][1-2][1-2]\) 333 \ ([4-5][4-5].\)$ / /\2333\1/ Group 1 then 333 Group 2 Group2, then 3333, then group So 2223334444 becomes 444333222 Regardless of the fact that the incoming called-number is missing, you will still have to change the direction of the aptly named translation-profile. Because you want to translate called 999 on calls incoming, not outgoing.

upvoted 1 times

decdca7 1 year, 5 months ago

Sorry D is the correct answer, there others are wrong

upvoted 1 times

decdca7 1 year, 5 months ago

None of the others are correct. The translation does not translate the number to 444333222

upvoted 1 times

Piji 1 year, 7 months ago

The translation rule is correct and it translates 222333444 to 44433322. The issue is "destination-pattern 888" which should be "destination-pattern 4443333222". So the correct answer is the A. Bear in your mind "incoming called-number" missing "." which that is the typo and should be "incoming called-number ."

upvoted 2 times

b3532e4 1 year, 10 months ago

I think 2 have problems: 1. voice translation-rule 999 ---> Original number: 222333444 Translated number:2333444 2. dial-peer voice 999 voip incoming called-number . (DOT)

upvoted 2 times

b3532e4 1 year, 10 months ago

I think 2 have problems: 1. voice translation-rule 999 ---> Original number: 222333444 Translated number:2333444 2. dial-peer voice 999 voip incoming called-number . (DOT)sorry 222333444----> voice Translation-profile incoming (444333222)--->DP voice 999 -->DGP888-->DP voice 888---> Destination-pattern 888 (route call 888 = not found 404) Correct answer AWhen you use DPG the destination pattern does not matter

upvoted 2 times

b3532e4 1 year, 7 months ago

sorry 222333444----> voice Translation-profile incoming (444333222)--->DP voice 999 -->DGP888-->DP voice 888---> Destination-pattern 888 (route call 888 = not found 404) Correct answer AWhen you use DPG the destination pattern does not matter

upvoted 1 times

b3532e4 1 year, 7 months ago

sorry 222333444----> voice Translation-profile incoming (444333222)--->DP voice 999 -->DGP888-->DP voice 888---> Destination-pattern 888 (route call 888 = not found 404)

Correct answer A

upvoted 1 times

decdca7 1 year, 5 months ago

When you use DPG the destination pattern does not matter

upvoted 2 times

FrankPic 2 years, 7 months ago

In my opinion the best answer is B. The incoming dial-peer is missing the "." parameter after the incoming called-number command. This is quite common to set up a match-any incoming dial-peer, both single digit o entire dial-string for en-block dialing like sip (https://community.cisco.com/t5/unified-communications-infrastructure/quot-incoming-called-number-quot-what-the-quot-dot-quot-stands/td-p/4435303) Voice-translation rule 99 is correct but on the incoming dial-peer (voip dial-peer 999) you need to change the translation-profile direction to incoming, like following: translation-profile incoming incoming I also have some doubts about the outgoing dial (voip dial-peer 888) as destination-

pattern 888 will never be matched by called number after voice translation-rule 99. So also A, on a second step, can be right or anyway need to be applied.

upvoted 3 times

john_doe_9999 2 years, 7 months ago

This one is B. The translation works, but is being applied in the wrong direction on the incoming dial-peer.

upvoted 2 times

Afaik 3 years ago

There is so much wrong information, do we actually have a inbound dial-peer match here? translation is also not correct but doesn't work either assuming we hit the according dial-peer, basically reading the question it points towards CUCM and explaining detailed information what CUCM expect, D provides the best solution answer to that, ignoring this mess of configuration.

upvoted 1 times

7ArchAngel7 3 years, 2 months ago

It's obvious to anyone who has an intimate knowledge of CUBE/IOS Call Processing that information provided in the configuration output is either missing or omitted. It's sad but some assumptions have to be made here. The incoming called-number is missing its matching parameter (IMO I assume this is unintentional as the command by itself is always rejected). I would also assume that the original question has the command as "incoming called-number 22233344.." or some type of extended matching, which would make B the Answer because the translation rule works as intended but it is obviously configured in the wrong direction and its name is made with the intention to confuse, which is a very Cisco-Like question. So you are left with a conundrum.... The answer is to either configure specific matching as there is no matching for the incoming called number 222333444 but either way you have to change the direction of the translation profile. I would assume the answer is B.

upvoted 1 times

basscov 4 years, 2 months ago

Think here is a bunch of ssues. Firstly, actual translation rule is setup incorrectly, this will not give you 444333222 in the output. Secondly, translation profile is applied incorrectly ,in outbound direction. And third, description says cucm is expecting called number 444333222 , but destination pattern 888 will not work for it, so it needs changed as well. Should be a few answers here

upvoted 1 times

basscov 4 years, 2 months ago

Think here is a bunch of ssues. Firstly, actual translation rule is setup incorrectly, this will not give you 444333222 in the output. Secondly, translation profile is applied incorrectly ,in outbound direction. And third, description says cucm is expecting called number 444333222 , but destination pattern 888 will not work for it, so it needs changed as well. Should be a few answers hereI agree with you on the first two statements. The third one "dial-peer voice 888 voip" will work, because the routing is based on "destination dpg 888", which will send the call through the most preferred dial-peer in the list, doesn't matter what is configured on the "destination-pattern". So I think option D is the best choice.

upvoted 1 times

bitdesaia 3 years, 4 months ago

I agree with you on the first two statements. The third one "dial-peer voice 888 voip" will work, because the routing is based on "destination dpg 888", which will send the call through the most preferred dial-peer in the list, doesn't matter what is configured on the "destination-pattern". So I think option D is the best choice.

upvoted 3 times

Mert_kerna 4 years, 3 months ago

I think we can all just assume there was a typo in the voice translation-rule, because it doesn't match correctly. However, there is not an answer that corresponds to the translation-rule being incorrect, or that would be the correct answer. The answer is B, because the translation-profile is being applied in the wrong direction. The name of the translation-profile is meant to trip you up in this question because it's "outgoing incoming" Outgoing is the name direction of the translation-profile and incoming is the name. It needs to be changed to become an incoming translation-profile. It's an incoming dial-peer as it doesn't have a session-target. So it needs to be translation-profile incoming incoming, where incoming represents the direction and name of the translation-profile

upvoted 2 times

Mert_kerna 4 years, 3 months ago

I think we can all just assume there was a typo in the voice translation-rule, because it doesn't match correctly. However, there is not an answer that corresponds to the translation-rule being incorrect, or that would be the correct answer. The answer is B, because the translation-profile is being applied in the wrong direction. The name of the translation-profile is meant to trip you up in this question because it's "outgoing incoming" Outgoing is the name direction of the translation-profile and incoming is the name. It needs to be changed to become an incoming translation-profile. It's an incoming dial-peer as it doesn't have a session-target. So it needs to be translation-profile incoming incoming, where incoming represents the direction and name of the translation-profileI may have been a little tipsy when I wrote this. What I meant to type is Outgoing is the direction of the translation-profile and incoming is the name of the translation-profile. It needs to change to translation-profile incoming incoming in order to facilitate the corresponding direction of the call leg and manipulate the digits. The answer HAS to be B.

upvoted 2 times

Mert_kerna 4 years, 3 months ago

I may have been a little tipsy when I wrote this. What I meant to type is Outgoing is the direction of the translation-profile and incoming is the name of the translation-profile. It needs to change to translation-profile incoming incoming in order to facilitate the corresponding direction of the call leg and manipulate the digits. The answer HAS to be B.

upvoted 1 times

DoubleK 5 years, 3 months ago

I think "B". The translation rule is set for outbound but this is an inbound dial peer which only uses inbound translations. (The name of the translation rule is "inbound" but the name does not change the direction it is actually used in). The incoming called-number command missing the dot is probably a typo. Even if you make it more specific and match 2223334444 the translation rule would not be applied because it is assigned the wrong direction. This config uses Dial Peer Groups which ignores the outgoing destination pattern so changing 888 to 444333222 would do nothing.

upvoted 2 times

Marco74 5 years, 3 months ago

I think that correct is B

upvoted 2 times

jn4voip 5 years, 4 months ago

The answer is A. The destination pattern on the outgoing dial peer is 888, but the called number after the correct translation is 444333222. \2 takes the second group of 444 and adds 333, then \1 adds the first group of 222

upvoted 3 times

jn4voip 5 years, 4 months ago

The answer is A. The destination pattern on the outgoing dial peer is 888, but the called number after the correct translation is 444333222. \2 takes the second group of 444 and adds 333, then \1 adds the first group of 222I agree with A, because there is no valid pattern on outgoing to cucm dial peer to accommodate 444333222This is actually a trick question. If you look at the "incoming" dial-peer. It has a translation profile set to outgoing: translation-profile outgoing incoming. After the translation-profile command, you need to specify either incoming or outgoing when it hits that dial-peer. It's tricky because the profile is named "incoming". This is not indicative of the direction... Only the profile name. Thus, an incoming profile must be created for that incoming dial-peer.

upvoted 3 times

basscov 4 years, 3 months ago

I agree with A, because there is no valid pattern on outgoing to cucm dial peer to accommodate 444333222This is actually a trick question. If you look at the "incoming" dial-peer. It has a translation profile set to outgoing: translation-profile outgoing incoming. After the translation-profile command, you need to specify either incoming or outgoing when it hits that dial-peer. It's tricky because the profile is named "incoming". This is not indicative of the direction... Only the profile name. Thus, an incoming profile must be created for that incoming dial-peer.

upvoted 1 times

basscov 4 years, 3 months ago

I agree with A, because there is no valid pattern on outgoing to cucm dial peer to accommodate 444333222

upvoted 1 times

Jajo 4 years, 2 months ago

This is actually a trick question. If you look at the "incoming" dial-peer. It has a translation profile set to outgoing: translation-profile outgoing incoming. After the translation-profile command, you need to specify either incoming or outgoing when it hits that dial-peer. It's tricky because the profile is named "incoming". This is not indicative of the direction... Only the profile name. Thus, an incoming profile must be created for that incoming dial-peer.

upvoted 1 times

Emil7 5 years, 5 months ago

Shouldn't this be D? Assuming " Cisco Unified Communications Manager is expecting the called number to be delivered as "444333222"." I don't see this happening in the configured Translation rule.I think D , but translation profile in dial-peer 999 match outgoing traffic and I think Answer B is correct too...

upvoted 2 times

Emil7 5 years, 5 months ago

Shouldn't this be D? Assuming " Cisco Unified Communications Manager is expecting the called number to be delivered as "444333222"." I don't see this happening in the configured Translation rule.

upvoted 2 times

alert2003 5 years, 4 months ago

I think D , but translation profile in dial-peer 999 match outgoing traffic and I think Answer B is correct too...

upvoted 2 times

Question #22 *Topic 1*

Refer to the exhibit. Users report that outbound PSTN calls from phones registered to Cisco Unified Communications Manager are not completing. The local service provider in North America has a requirement to receive calls in 10-digit format. The Cisco Unified CM sends the calls to the Cisco Unified Border Element router in a globalized E.164 format. There is an outbound dial peer on Cisco Unified Border Element configured to send the calls to the provider. The dial peer has a voice translation profile applied in the correct direction but an incorrect voice translation rule applied, which is shown in the exhibit. Which rule modified DNIS in the format that the provider is expecting?

```
voice translation-rule 84
        rule 1 /^\ ([2-9]..[2-9].....$\)/ /\2/
```

A. rule 1 /^/+\([^1].*\)/ /011\1/

B. rule 1/^\+1\([2-9]..[2-9]"¦"¦$\)/ /\1/

C. rule 1 /^\([2-9]..[2-9]"¦"¦$\)/ /\1/

D. rule 1 /^\+1\([2-9]..[2-9]"¦"¦$\)/ /\0/

Suggested Answer: *B*

Official Answer: B

Community vote distribution

B (100%)

john_doe_9999 Highly Voted 2 years, 7 months ago

The correct translation without the incorrect characters is: rule 1 /^\+1\([2-9]..[2-9].....$\)/ /\1/

upvoted 7 times

f1ab921 Most Recent 1 year, 10 months ago

The options are not showing correctly here, john_doe_9999 is 100% correct, and his translation is the right answer, which will be one of the options in the exam.

upvoted 2 times

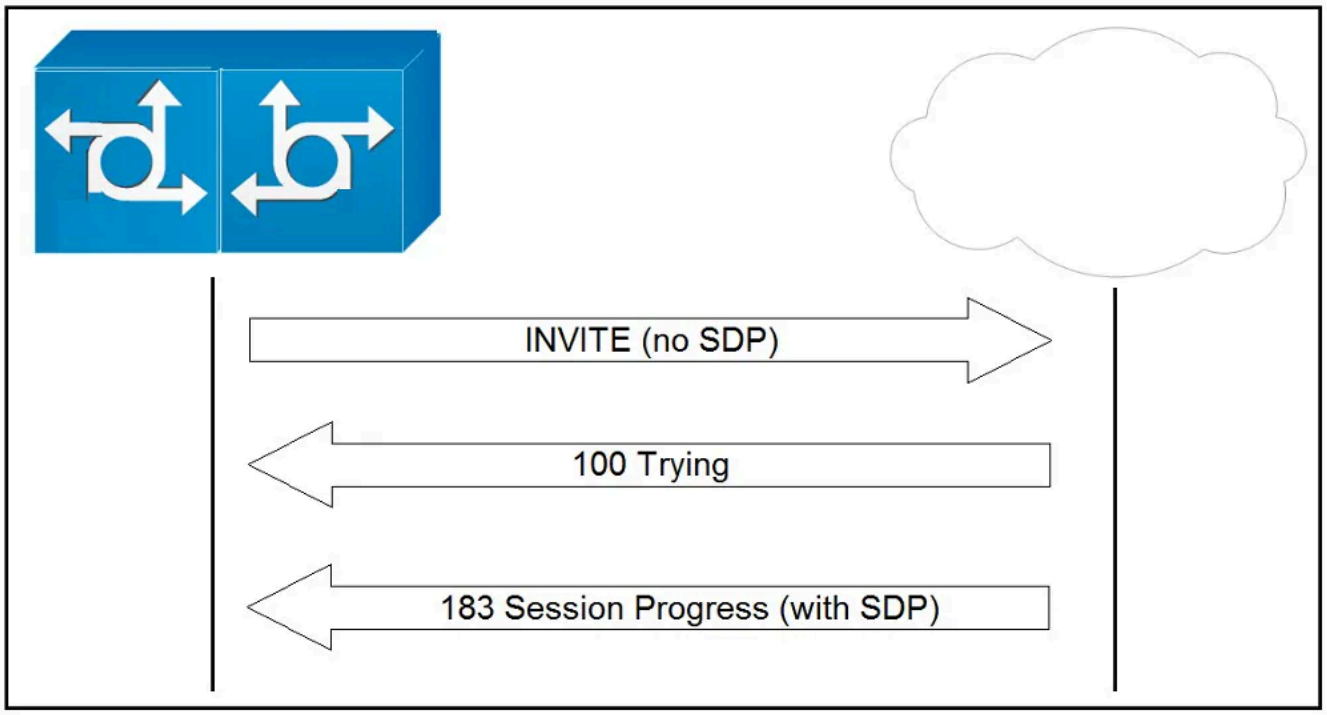
FrankPic 2 years, 11 months ago

In my opinion there is no valid answer. A is not correct since it contains invalid escape character (/+ instead of \+) B, C and D contain invalid wildcard " and | C is also missing the \+ in order to match incoming E164 called number from CUCM

upvoted 1 times

Question #23 *Topic 1*

Refer to the exhibit. An administrator is troubleshooting why users are not hearing audio when dialing long distance numbers across their Cisco Unified Border
Element. The customer's carrier has a requirement that dialing long distance requires an access code to be entered. Looking at the exhibit, what two actions can be taken to correct signaling? (Choose two.)



A. Enanle PRACK.

B. Enable Early Offer on the Cisco Unified Border Element.

C. Enable the supplementary-service media-renegotiate command.

D. Enable Media Flow Around

E. Enable Mid-Call Signaling Consumption.

Suggested Answer: *A, B*
Official Answer: AB

Community vote distribution
AB (100%)

john_doe_9999 Most Recent 2 years, 1 month ago
A and B are correct
upvoted 2 times

Question #24 *Topic 1*

Which IOS command creates a SIP-enabled dial peer?

A. voice dial-peer 20 sip

B. dial-peer voice 20 voip

C. dial-peer voice 20 pots

D. dial peer voice 20 sip

Suggested Answer: *B*

Reference: https://www.ciscopress.com/articles/article.asp?p=664148&seqNum=6

Community vote distribution

B (100%)

729edae Most Recent 1 year, 9 months ago

B is correct but...lol that command alone will create an H.323 dial peer. Once the command session protocol sipv2 is added to the dial peer...it becomes a SIP dial peer. Very Cisco.

upvoted 1 times

john_doe_9999 2 years, 1 month ago

B is correct

upvoted 1 times

Testme1235 2 years, 10 months ago

The IOS command to create a SIP-enabled dial peer is: php Copy code dial-peer voice voip Once you enter this command, you can configure SIP-specific parameters for the dial peer using the following commands: bash Copy code session protocol sipv2 destination You can then configure other parameters as needed, such as codecs, dial peer preferences, and call routing options.

upvoted 2 times

Question #25 *Topic 1*

A user in location X dials an extension at location Y. The call travels through a QoS-enabled WAN network, but the user experiences choppy or clipped audio.
What is the cause of this issue?

A. missing Call Admission Control

B. codec mismatch

C. ptime mismatch

D. phone class of service issue

Suggested Answer: *B*
Official Answer: B

Community vote distribution



thili Highly Voted 5 years, 11 months ago
Answer should be A
upvoted 16 times

movalleuu Highly Voted 5 years, 5 months ago
Answer should be A, codec mismatch generally drops the call
upvoted 9 times

f77380c Most Recent 4 months, 2 weeks ago
All other options would cause call to fail. Answer is C.
upvoted 1 times

729edae 1 year, 4 months ago
Answer A (The question mentions Q0S enabled WAN Network, not LAN) B - Call would not work - SDP would not be negotiated Cause=65 (most likely) no audio at all C - Could actually provide those symptoms but, usually complete playback failure. D- COS (Layer 2, think LAN) WAN cares not at L3 (DSCP-IP-Prec) (Put there to trip you up) I would need qos mapping cos to ip prec (old mls qos commands) on L3 Switch. Too much assumption here. Then again were talking Cisco questions, not known for making sense when it comes to testing your actual abilities.
upvoted 2 times

decdca7 1 year, 5 months ago
Phone marks the packet witha wrong COS thus QoS is not working
upvoted 1 times

Piji 1 year, 6 months ago
A is correct answer as two different location,
upvoted 1 times

jarcoman99 1 year, 8 months ago
Question states that the WAN is QoS enabled. Answers B and C will not have audio. Answer A still doesn't fix the real root cause. Answer is D, because CoS setting will properly tag audio traffic so that it is prioritized over the WAN.
upvoted 2 times

b3532e4 1 year, 9 months ago
A. missing Call Admission Control
upvoted 1 times

Piji 2 years ago
The question asks about the cause of choppy or clipped audio when a user in location X calls an extension at location Y over a QoS-enabled WAN network. The most likely cause of this issue is: B. codec mismatch Thus, B. codec mismatch is the most plausible cause of choppy or clipped audio in this scenario.
upvoted 1 times

basscov 4 years, 2 months ago
I think answer D. Because A is something that could help to overcome the issue but it's not a route cause of it
upvoted 2 times

timmyz 4 years, 11 months ago

Maybe D ? if the phone is not set for trust then there is no QoS markings on the phone which in turn never get prioritized across the WAN. even though QoS is enabled on the WAN

upvoted 3 times

XalaGyan 5 years, 4 months ago

Best Answer would have been if there was something related to QoS misconfigured or missing but Call Admission Control is the closest it gets to QoS. So i agree with you its (A)

upvoted 4 times

Question #26 *Topic 1*

An engineer must route all SIP calls in the form of @example.com to the SIP trunk gateway corporate local. Which two SIP route patterns can be used to accomplish this task? (Choose two.)

A. example.com@gateway.corporate.local

B. *@example.com

C. gateway.corporate.local

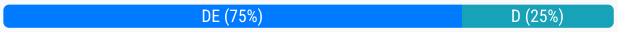
D. example.com

E. *.*

Suggested Answer: *B, E*

Official Answer: BE

Community vote distribution



Grebec94 Highly Voted 5 years, 6 months ago

e and d, pattern cannot accept @

upvoted 14 times

Grebec94 Highly Voted 5 years, 6 months ago

e and d, pattern cannot accept @You are correct. I attempted it in CUCM just to make sure and it denied me because of the @

upvoted 14 times

Mert_kerna 3 years, 9 months ago

You are correct. I attempted it in CUCM just to make sure and it denied me because of the @

upvoted 3 times

kljw5 Most Recent 1 year, 4 months ago

Correct Answers for This Question: □ D. example.com — This is the required pattern to match the domain correctly. □ (No valid second choice among the options given.)

upvoted 1 times

kljw5 1 year, 7 months ago

Wouldn't *.* be generic and allow anything through including but not excluding any other domain???? Must route calls using example.com. E. doesn't do this

upvoted 2 times

john_doe_9999 2 years, 1 month ago

D and E

upvoted 1 times

MaxG 2 years, 6 months ago

. is a valid pattern. Here it is in action: https://cmslab.ciscolive.com/pod1/cucm/cucmtrunk

upvoted 2 times

flexster 3 years, 7 months ago

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].

upvoted 2 times

Question #27 *Topic 1*

Which two statements are correct with respect to the Client Matter Code setting in the route pattern configuration? (Choose two.)

A. The Client Matter Code feature does not support overlap sending because the Cisco Unified CM cannot determine when to prompt the user for the code.

B. If you check the Allow Overlap Sending check box, the Require Client Matter Code check box becomes disabled.

C. If you check the Allow Overlap Sending check box, you can also check the Require Client Matter Code check box.

D. The Client Matter Code feature does support overlap sending because the Cisco Unified Communications Manager can determine when to prompt the user for the code.

E. The Client Matter Code has the option to configure Authorization Level such as in the Forced Authorization Code.

Suggested Answer: *A, B*

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_010000.pdf

Community vote distribution

AB (100%)

john_doe_9999 Most Recent 2 years, 1 month ago

A and B are correct

upvoted 1 times

Mert_kerna 3 years, 9 months ago

Tested this in CUCM and these answers are correct.

upvoted 4 times

Question #28 *Topic 1*

A network engineer designs a new dial plan and wants to block a certain range of numbers (8135100 through 8135105). What is the most specific route pattern that can be configured to block only the numbers in this range?

A. 813510[012345]

B. 813510[12345]

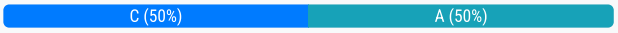
C. 813510[^0-5]

D. 81XXXXX

Suggested Answer: *A*

Official Answer: A

Community vote distribution



Mert_kerna Highly Voted 4 years, 3 months ago

This is correct. Your brain probably tricked you and assumed C may have been the correct answer. The problem is that the ^ symbol implies that the following digit is at the start of the string. Put yourself in the shoes of the exam writers and you'll see where they're trying to trip you on questions. Failures generate revenue or the tests would be easier.

upvoted 6 times

fd53486 Most Recent 9 months, 3 weeks ago

The circumflex (^) character, used with the square brackets, negates a range of values. It must be the first first character following the opening bracket ([). Each route pattern can have only one ^ character. -> The route pattern 813510[^0-5] routes or blocks all numbers in the range 8135106 through 8135109. https://www.cisco.com/en/US/docs/voice_ip_comm/cucm/admin/3_1_2/ccmcfg/b03spchr.html#wp1029834 So option C. is the correct choice!

upvoted 2 times

1b9a51d 2 years, 1 month ago

https://www.cisco.com/en/US/docs/voice_ip_comm/cucm/admin/3_1_2/ccmcfg/b03spchr.html#wp1029834

upvoted 1 times

john_doe_9999 2 years, 7 months ago

A is correct

upvoted 2 times

Question #29 *Topic 1*

Which two descriptions of the Standard Local Route Group deployment are true? (Choose two.)

A. can be associated under the route group

B. can be associated only under the route list

C. chooses the route group that is configured under the device pool of the calling-party device

D. chooses the route group that is configured under the device pool of the called-party device

E. can be assigned directly to the route pattern

Suggested Answer: *B, D*

Official Answer: BD

Community vote distribution

BC (100%)

thili Highly Voted 5 years, 5 months ago

Answer should be B,CNot true .. BD are correct answers

upvoted 27 times

thili Highly Voted 5 years, 5 months ago

Answer should be B,C

upvoted 27 times

VG224 4 years, 4 months ago

Not true .. BD are correct answers

upvoted 1 times

b3532e4 Most Recent 1 year, 7 months ago

Answer should be B,C

upvoted 1 times

john_doe_9999 2 years, 1 month ago

B and C are correct

upvoted 2 times

FCBear 2 years, 5 months ago

Correct answer is BC, I know this for a fact as I configured them a thousand times.

upvoted 2 times

basscov 3 years, 8 months ago

Called party is destination number, calling party is source number. So answer B and C

upvoted 2 times

Mert_kerna 3 years, 9 months ago

B and D are not correct answers. From the perspective of CUCM, phones in a device pool will be both the called and calling-party device at any given time. However, this doesn't count the probability that this question is geared toward the device in the device pool utilizing the route pattern associated with the standard local route group. If a device is using SLRG, it is likely calling out to an external location. It is safe to presume, based on the logic of the question, that the given device in this equation would be the calling-party device. If another device, external or internal, is calling said device, then it becomes the called-party device. The answer definitely should be B and C.

upvoted 2 times

AKREM86 4 years, 6 months ago

B, C are the correct answers:
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 3 times

AKREM86 4 years, 6 months ago

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100/CUCM_BK_F3AC1C0F_00_cucm-features-services-guide-100_chapter_0100110.htmlyou need to read the link you suggested .. correct ans are BDIt is B and C. Usual the called device will be external so that is not the answer.you'd better study the theory a little bit more. BC

upvoted 3 times

VG224 4 years, 4 months ago

you need to read the link you suggested .. correct ans are BD

upvoted 2 times

VG224 4 years, 4 months ago

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upvoted 2 times

mmollura 3 years, 5 months ago

It is B and C. Usual the called device will be external so that is not the answer.

upvoted 3 times

stocaxx 3 years, 12 months ago

you'd better study the theory a little bit more. BC

upvoted 3 times

Marco74 4 years, 10 months ago

I'm not sure about that answers. Someone did it lately?

upvoted 1 times

Question #30 *Topic 1*

Refer to the exhibit. An engineer configures Cisco Unified Border Element to connect the enterprise VoIP network with a SIP telephony provider. Calls are not working in either direction. What must be configured in the dial peer 1 to fix the issue?

```
!

dial-peer voice 1 voip
 description to ITSP
 destination-pattern 555.....
 session target ipv4:209.110.110.1
 incoming called-number .
 codec g711ulaw
!
!
```

A. answer-address 555 "¦"¦..

B. codec g729

C. session-protocol sipv2

D. incoming called number 555"¦"¦.

Suggested Answer: *D*

Official Answer: D

Community vote distribution

C (100%)

rishik Highly Voted 5 years, 11 months ago

Correct answer should be C

upvoted 28 times

Piji Most Recent 2 years ago

C. session-protocol sipv2 The dial peer is configured to handle VoIP traffic, but it is missing a key configuration element: the session protocol. Since this setup is for a SIP telephony provider, the dial peer must explicitly specify that it will use the SIP protocol. The session-protocol sipv2 command will enable SIP for this dial peer.

upvoted 1 times

john_doe_9999 2 years, 7 months ago

C is correct

upvoted 2 times

FrankPic 2 years, 11 months ago

I agree with C also because A and D both contain invalid wildcard " and | (seen also in other questions...).

upvoted 1 times

7ArchAngel7 3 years, 3 months ago

The answer should be C. The default protocol for a VoIP Dial Peer is H.323 unless the command "session protocol sipv2" is entered in the dial-peer configuration. Codec would determine the RTP/SDP negotiation and the incoming called number and answer address would only determine matching characteristics. The problem is the signaling protocol required by the provider is SIP and its not being used here.

upvoted 4 times

Question #31 *Topic 1*

After configuring a Cisco CallManager Express with Cisco Unity Express, inbound calls from the PSTN SIP trunk receive a ring tone for 20 seconds and then a busy signal instead of voicemail. Which configuration fixes this problem?

A. Router(config)# voice service voip Router(conf-voi-serv)#allow-connections h323 to h323

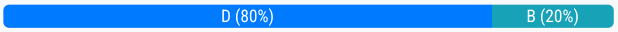
B. Router(config)#dial-peer voice 2 voip Router(config-dial-peer)#no vad

C. Router(config)# voice service voip Router(conf-voi-serv)#allow-connections voice-mail mod

D. Router(config)# voice service voip Router(conf-voi-serv)#no supplementary-service sip moved-temporarily

Suggested Answer: *A*
Reference:
https://www.cisco.com/c/en/us/td/docs/voice_ip_comm/cusrst/admin/sccp_sip_srst/configuration/guide/SCCP_and_SIP_SRST_Admin_Guide/srst_call_handling.html

Community vote distribution



decdca7 Most Recent 1 year, 5 months ago
https://www.cisco.com/c/en/us/support/docs/voice-unified-communications/unified-communications-manager-express/91535-cme-sip-trunking-config.html Call Forward When a call comes in on a SIP trunk and gets forwarded (CFNA / CFB / CFA), then the default behavior is for the CME to send the 302 "Moved Temporarily" SIP message to the Service Provider (SP) proxy. The user portion of the Contact Header in the 302 message might need to be translated to reflect a DID that the SP proxy can route to. The host portion of the Contact Header in the 302 message should be modified to reflect the Address of Record (AOR) using the host-registrar CLI under sip-ua and the b2bua CLI under the VoIP dial peer going to the CUE. Some SIP proxies might not support this. If so, then you need to add this: Router(config)#voice service voip Router(conf-voi-serv)#no supplementary-service sip moved-temporarily
upvoted 1 times

jatoja83 1 year, 5 months ago
Explanation: The issue you're describing suggests that inbound calls from the PSTN SIP trunk are not being routed correctly to voicemail and instead receive a busy signal after 20 seconds. This often happens because the Voice Activity Detection (VAD) feature, which can cause interruptions in call routing, may be enabled. Disabling VAD is a typical fix in such scenarios. VAD (Voice Activity Detection): This feature can cause issues in call routing if it incorrectly detects silence and prematurely ends the call. Disabling VAD ensures that the call stays active, allowing it to be routed correctly to voicemail.
upvoted 1 times

rsl4u 2 years, 1 month ago
Reference URL does not exist anymore: Other URL with explaination: https://www.cisco.com/c/en/us/support/docs/voice-unified-communications/unified-communications-manager-express/91535-cme-sip-trunking-config.html (Section: Call Forward)
upvoted 1 times

john_doe_9999 2 years, 1 month ago
I'd say D
upvoted 1 times

Mert_kerna 3 years, 9 months ago
no supplementary-service sip moved temporarily is to remove the default behavior where CME sends the 302 moved temporarily SIP message to the Service Provider.
upvoted 2 times

AgshinA 4 years, 11 months ago
it should be B. VAD is enabled by default. Because there is no comfort noise during periods of silence, the call may seem to be disconnected. You may prefer to set no vad on the SIP phone pool.Actually it is D. https://community.cisco.com/t5/ip-telephony-and-phones/help-please-callers-get-busy-signal-instead-of-aim-cue-voicemail/td-p/1911590
upvoted 2 times

AgshinA 4 years, 11 months ago
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upvoted 2 times

AgshinA Highly Voted 4 years, 11 months ago

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upvoted 12 times

Grebec94 5 years, 6 months ago

A voice activity detection (VAD) must be switched off and a codec must be designated.

upvoted 4 times

Question #32 *Topic 1*

An engineer must configure a secure SIP trunk with a remote provider, with a specific requirement to use port 5065 for inbound and otubound traffic. Which two items must be configured to complete this configuration? (Choose two.)

A. Incoming Port in SIP Information section of the SIP Trunk configuration.

B. Incoming Port in Security Information of the SIP Profile configuration.

C. Destination Port in SIP Information section of the SIP Trunk configuration

D. Incoming Port in SIP Trunk Security Profile configuration

E. Destination Port in SIP Trunk Security Profile configuration

Suggested Answer: *C, D*
Official Answer: CD

john_doe_9999 Most Recent 2 years, 1 month ago

C and D are correct

upvoted 2 times

Mert_kerna 4 years ago

100% correct. Incoming Port of a SIP Trunk Security Profile is Mandatory. Destination Port is configured to the right of the Destination Address in the SIP Information portion of the SIP Trunk Configuration.

upvoted 4 times

Question #33 *Topic 1*

In Cisco Unified Communications Manager, which tool do you use to check SIP traces?

A. MTP

B. CCSIP

C. RTMT

D. OS Administration Page

Suggested Answer: *C*
Official Answer: C

timmyz Most Recent 2 years ago

Answer is C

upvoted 3 times

Question #34 *Topic 1*

If all patterns below are configured in Cisco Unified Communications Manager which would be used when dialing the pattern "123"?

A. 12!

B. 12X (urgent priority set)

C. 1XX (urgent Priority Set)

D. 12[2-5]

Suggested Answer: *B*
Official Answer: B

Community vote distribution
D (100%)

jtlzhang Highly Voted 5 years, 6 months ago
D is right?
upvoted 9 times

jtlzhang Highly Voted 5 years, 6 months ago
D is right?No, its BSorry , its D. you are correct
upvoted 9 times

thili 5 years, 5 months ago
No, its BSorry , its D. you are correct
upvoted 2 times

thili 5 years, 5 months ago
No, its B
upvoted 2 times

thili 5 years, 5 months ago
Sorry , its D. you are correct
upvoted 8 times

PiperP 5 years ago
answer is D, tested it on CUCM
upvoted 7 times

Dariusz_N Most Recent 1 year, 4 months ago
Dialed Digits = 123 Match Result = RouteThisPattern Matched Pattern Information Pattern = 12[2-5] This is what Dialed Number Analyzer returns
upvoted 1 times

john_doe_9999 2 years, 1 month ago
D is correct
upvoted 1 times

certified_bunny 2 years, 4 months ago
closest match
upvoted 1 times

Mert_kerna 4 years ago
Correct answer is D. Urgent priority isn't used to prioritize which pattern will be used when dialing digits. It's used to route the call as soon as the match is detected, without waiting for the T302 timer to expire. If a closer match is found, even without urgent priority checked, it will ALWAYS use the closest match.
upvoted 5 times

enashash 4 years, 8 months ago
correct answer ASorry It`s D
upvoted 1 times

enashash 4 years, 8 months ago
correct answer A
upvoted 1 times

enashash 4 years, 8 months ago

Sorry It`s D

upvoted 1 times

Question #35 *Topic 1*

Which configuration must an administrator perform to display Translation Pattern operations in Cisco Unified Communications Manager SDL traces?

A. Enable the Detailed Call Analysis option under Enterprise Parameters for Unified CM.

B. Set up the Digit Analysis Complexity in Service Parameters for Cisco Unified CM to TranslationAndAlternatePatternAnalysis.

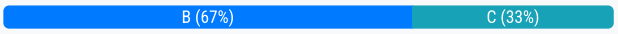
C. Check the Translation Patterns Analysis check box in Micro Traces on the Cisco Unified CM Serviceability page.

D. By default, the Translation Patterns operations are printed in SDL traces, so no additional configuration is necessary.

Suggested Answer: *A*

Reference: https://community.cisco.com/t5/collaboration-voice-and-video/taking-sip-call-trace-on-cisco-unified-cm-using-rtmt/ta-p/3161200

Community vote distribution



alert2003 Highly Voted 4 years, 11 months ago

refer for B: https://www.cisco.com/c/en/us/td/docs/voice_ip_comm/hcs/12_5/HCS_Solution/Troubleshooting/chcs_b_troubleshooting-guide/chcs_m_troubleshooting-voice-application-components.html?dtid=osscdc000283

upvoted 13 times

jatoja83 Most Recent 1 year, 5 months ago

he correct answer is: C. Check the Translation Patterns Analysis check box in Micro Traces on the Cisco Unified CM Serviceability page. Explanation: In Cisco Unified Communications Manager (CUCM), to capture detailed information related to Translation Patterns in SDL (Signaling Debug Log) traces, the administrator must enable the "Translation Patterns Analysis" option in the Micro Traces settings. This allows the system to display translation pattern operations in the SDL traces.

upvoted 1 times

dirkske78 2 years ago

Digit Analysis Complexity: Required Field This parameter specifies the amount of digit analysis information that CCM trace file will provide. Cisco CallManager offers three digit analysis modes: StandardAnalysis - Provides detail about the final matched pattern without the translation pattern and alternate matches detail. TranslationAndAlternatePatternAnalysis - Provides detail about the translation pattern and the alternate matches in the call flow. This parameter displays details of only up to two Translation Patterns. CompleteTranslationAndAlternatePatternAnalysis - Provides detail about all the translation patterns and the alternate matches in the call flow.

upvoted 2 times

Freddan 5 years, 4 months ago

Sorry I mean DNA!

upvoted 1 times

Freddan 5 years, 4 months ago

There is no "Detailed Call Analysis" under enterprise parameter. But to see translation pattern in CAR you need to choose "TranslationAndAlternatePatternAnalysis" under service parameters.

upvoted 2 times

ALLENN 5 years, 4 months ago

@Freddan do you have any reference to support your answer? thank you

upvoted 1 times

ALLENN 5 years, 4 months ago

@Freddan do you have any reference to support your answer? thank youThere is a bug CSCvb32314 in CUCM. Symptom: Translation patterns are not present in cucm SDL traces even parameter "Digit Analysis Complexity" under Service parameters -> Call Manager -> Advanced is set to "TranslationAndAlernatePatternAnalysis". This is causing troubleshooting of HCS looping dial plan issues (or complex dialplan issues) difficult as translation patterns applied before reaching endpoint are not visible. So it is B. I had the same bug in cucm 11.5 and investigated it deeply then found this bugAdditionally, if you reference Service Parameter Configuration under the Cisco Call Manager (Active) service, within System, you'll find "Digit Analysis Complexity". You can toggle "StandardAnalysis", "TranslationAndAlternatePatternAnalysis", and "CompleteTranslationAndAlternatePatternAnalysis".

upvoted 1 times

AgshinA 4 years, 11 months ago

There is a bug CSCvb32314 in CUCM. Symptom: Translation patterns are not present in cucm SDL traces even parameter "Digit Analysis Complexity" under Service parameters -> Call Manager -> Advanced is set to "TranslationAndAlernatePatternAnalysis". This is causing troubleshooting of HCS looping dial plan issues (or complex dialplan issues) difficult as translation patterns applied before reaching endpoint are not visible. So it is B. I had the same bug in cucm 11.5 and investigated it deeply then found this bug

upvoted 1 times

Mert_kerna 3 years, 9 months ago

Additionally, if you reference Service Parameter Configuration under the Cisco Call Manager (Active) service, within System, you'll find "Digit Analysis Complexity". You can toggle "StandardAnalysis", "TranslationAndAlternatePatternAnalysis", and "CompleteTranslationAndAlternatePatternAnalysis".

upvoted 3 times

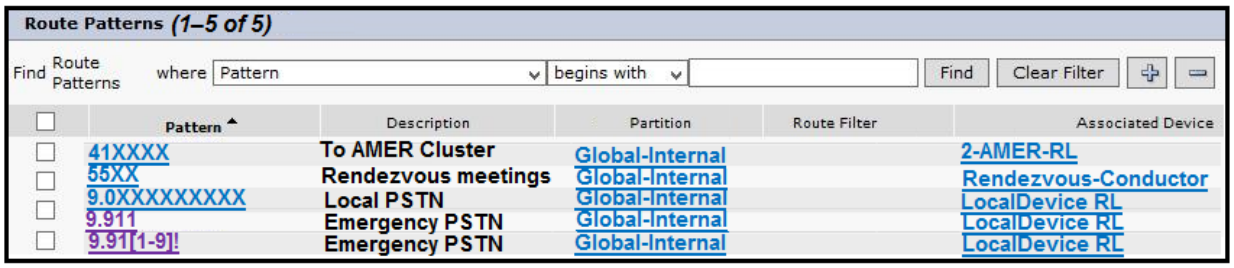
Freddan 5 years, 4 months ago

Should be B

upvoted 3 times

Question #36 *Topic 1*

Refer to the exhibit. Users report that when they dial the emergency number 9911 from any internal phone, it takes a long time to connect with the emergency operator. Which action resolves this issue?



Route Patterns *(1–5 of 5)*

Find Route Patterns where Pattern begins with Find Clear Filter

Pattern	Description	Partition	Route Filter	Associated Device
41XXXX	To AMER Cluster	Global-Internal		2-AMER-RL
55XX	Rendezvous meetings	Global-Internal		Rendezvous-Conductor
9.0XXXXXXXXX	Local PSTN	Global-Internal		LocalDevice RL
9.911	Emergency PSTN	Global-Internal		LocalDevice RL
9.91[1-9]!	Emergency PSTN	Global-Internal		LocalDevice RL

A. Adjust the service parameter T302 timet to the desired value.

B. Adjust the service parameter T204 timer to the desired value.

C. Check the Urgent Priority check box under 9.911 pattern.

D. Point the emergency pattern directly to the PSTN gateway.

Suggested Answer: *C*

Official Answer: C

Community vote distribution

C (100%)

Palo111 Most Recent 1 year, 9 months ago

C is right

upvoted 3 times

Question #37 *Topic 1*

The Cisco Unified Communications Manager Dialed Number Analyzer allows analysis of calls from which two devices? (Choose two.)

A. translation patterns

B. device pools

C. CTI ports

D. CTI route points

E. IP phones

Suggested Answer: *C, E*

Reference: 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#CUCM_TP_A5DA99E0_00

Community vote distribution

CE (100%)

jatoja83 Most Recent 1 year, 5 months ago

CE is correct

upvoted 2 times

b3532e4 1 year, 7 months ago

Cisco Unified Communications Manager Dialed Number Analyzer allows selection of specific devices that act as calling parties and called parties to test the dial plan. It allows analysis of calls from devices such as IP phones, CTI ports, and gateways.

upvoted 2 times

MaxG 2 years ago

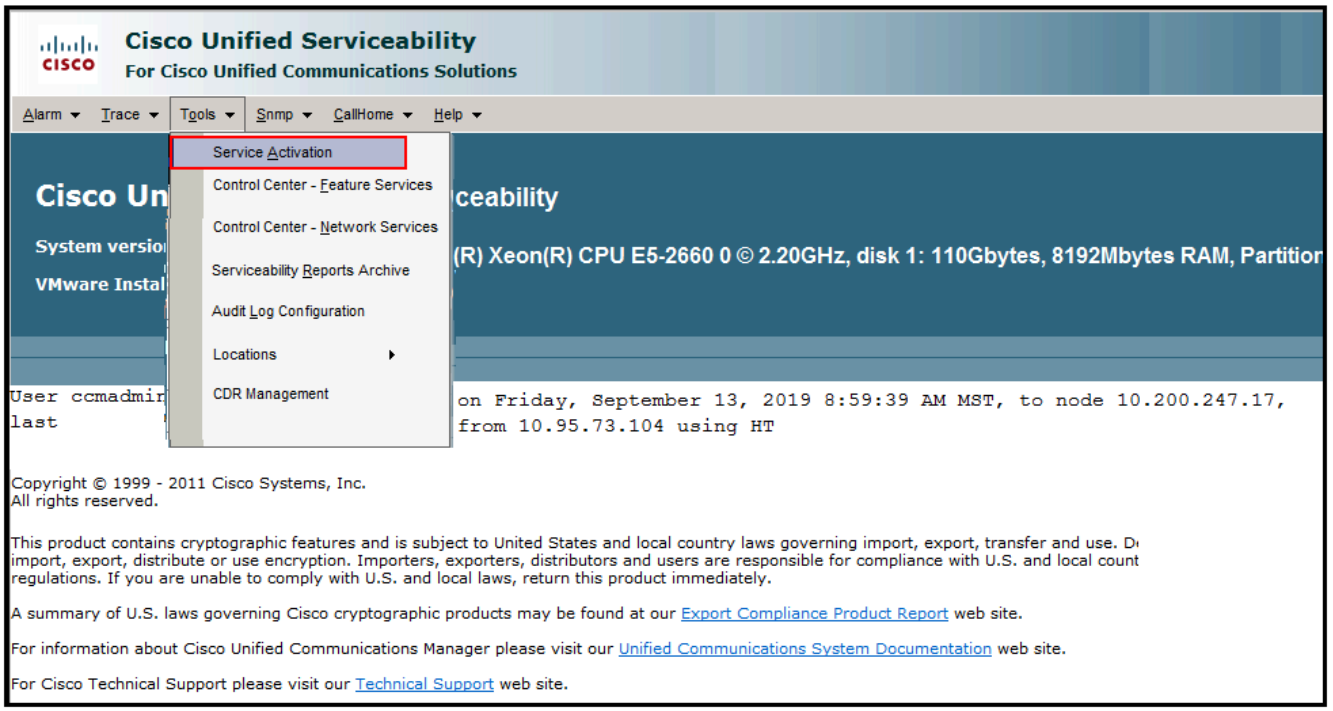
Cisco Unified Communications Manager Dialed Number Analyzer allows selection of specific devices that act as calling parties and called parties to test the dial plan. It allows analysis of calls from devices such as IP phones, CTI ports, and gateways.

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#CUCM_TP_A5DA99E0_00

upvoted 1 times

Question #38 *Topic 1*

Refer to the exhibit. An administrator is troubleshooting a situation where a call placed from a phone registered to Cisco Unified Communications Manager does not complete. The administrator wants to use the Dialed Number Analyzer on Cisco Unified CM to check which translation pattern the call is matching. However, when logging in to Cisco Unified Serviceability there is no option for Dialed Number Analyzer under the tool menu. Which two steps must be performed to resolve this issue? (Choose two.)



A. Restart the subscriber

B. Activate the Cisco Extended Functions service.

C. Activate the Cisco CallManager service.

D. Activate the Cisco Dialed Number Analyzer service.

E. Activate the Cisco Dialed Number Analyzer Server service.

Suggested Answer: *D, E*

Official Answer: DE

Community vote distribution

DE (100%)

b3532e4 Most Recent 1 year, 7 months ago

Cisco Dialed Number Analyzer Server The Cisco Dialed Number Analyzer Server service along with the Cisco Dialed Number Analyzer service supports Cisco Unified Communications Manager Dialed Number Analyzer. This service needs to be activated only on the node that is dedicated specifically for the Cisco Dialed Number Analyzer service. Unified Communications Manager clusters only: Cisco does not recommend that you activate the service on all the servers in a cluster. Cisco recommends that you activate this service only on one of the servers of a cluster where call-processing activity is the least.

upvoted 1 times

FrankPic 2 years, 1 month ago

D & E should be correct answers: https://www.cisco.com/c/en/us/td/docs/voice_ip_comm/cucm/admin/11_5_1/Admin/cucm_b_serviceability-admin-guide-1151su1/cucm_b_serviceability-admin-guide-1151su1_chapter_0101.html#CUCM_RF_C984F493_00

upvoted 1 times

MaxG 2 years, 6 months ago

This video confirms the answer: https://youtu.be/0eYO90kC7F0?t=51

upvoted 1 times

Question #39 *Topic 1*

In Cisco Unified Communications Manager globalized call routing is implemented and must confirm that it is correctly implemented without making a call. Which tool do you use for verification?

A. Dialed Number Analyzer

B. Real-Time Monitoring Tool

C. SDI trace

D. SDL trace

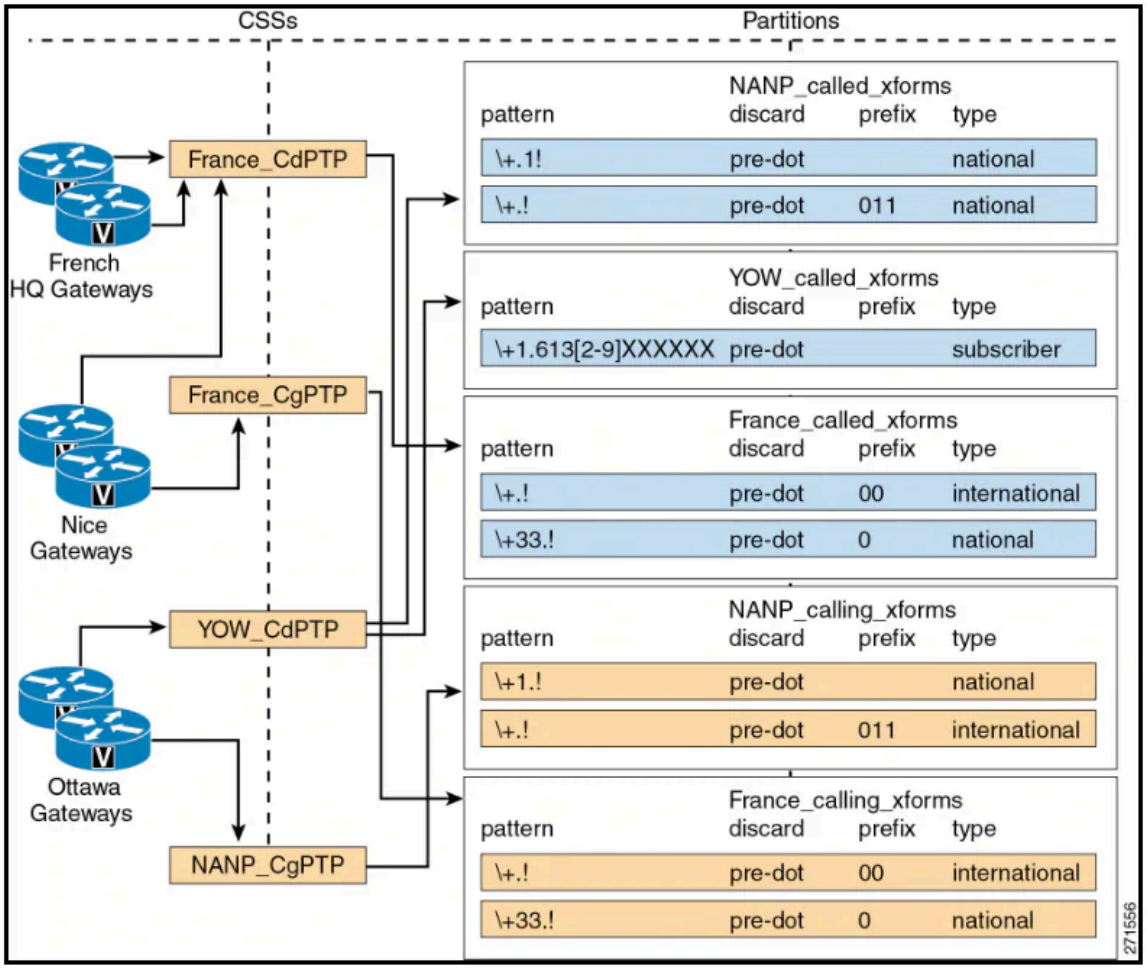
Suggested Answer: *A*
Official Answer: A

Community vote distribution
A (100%)

Akularn Most Recent 8 months, 1 week ago
A is correct
upvoted 1 times

Question #40 *Topic 1*

Refer to the exhibit. Within the North American Numbering Plan, gateways located in Ottawa, Canada and marked as "YOW" are assigned to the Calling Party
Transformation CSS NANP_CgPTP, which contains partition NANP_calling_xforms. What is the calling-party number and the numbering type if the calling user
+1613-555-1234 dials the number?



A. calling number 613-555-1234 and numbering type "subscriber"

B. calling number 011-1-613-555-1234 and numbering type "subscriber"

C. calling number 011613-555-1234 and numbering type "international"

D. calling number 613-555-1234 and numbering type "national"

Suggested Answer: *D*
Official Answer: D

Aravi1234 Most Recent 1 year, 9 months ago
Correct answer is D
upvoted 2 times

Question #41 *Topic 1*

Where on Cisco Unified Communications Manager do you configure the standard local route group for a group of devices?

A. System > Location Info

B. Call Routing > Route/Hunt > Local Route Group Names

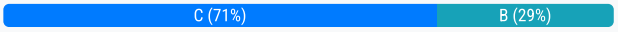
C. System > Device Pool

D. Call Routing > Emergency Location > Emergency Location (ELIN) Groups

Suggested Answer: *B*

Reference: https://www.uccollabing.com/configuring-standard-local-route-group-cucm/

Community vote distribution



decdca7 Most Recent 1 year, 5 months ago

Device pool is where you configure your standard route group!

upvoted 1 times

Piji 1 year, 7 months ago

C. is correct answer, this is the place you configure Standard Local Router group for group of devices. On B. you only configure the Local Router Group names.

upvoted 2 times

b3532e4 1 year, 10 months ago

C is the best answer

upvoted 2 times

dirkske78 2 years, 6 months ago

C is the best answer imho

upvoted 2 times

john_doe_9999 2 years, 7 months ago

C is the best answer here

upvoted 3 times

FCBear 2 years, 11 months ago

Correct answer is C B us wrong. You can't configure Standard local Route group from Route\Hunt

upvoted 3 times

Afaik 3 years ago

for a group of devices => device pool, if it is B you can leave out that information imho, so answer should be C

upvoted 2 times

Testme1235 3 years, 4 months ago

Selected Answer: B

Log in to Cisco Unified Communications Manager Administration.

Navigate to Call Routing > Route/Hunt > Route Group.

Click the Add New button to create a new route group.

In the Route Group Configuration window, enter a name for the route group in the Route Group Name field.

From the Distribution Algorithm drop-down menu, select the distribution algorithm that you want to use for the route group.

From the Available Devices list, select the devices that you want to include in the route group, and click the Add button to move them to the Selected Devices list.

Click Save to save the route group configuration.

To apply the route group to a route pattern, navigate to Call Routing > Route/Hunt > Route Pattern.

Select the route pattern that you want to configure, and click the Edit button.

In the Route Pattern Configuration window, select the appropriate standard local route group from the Route Group drop-down menu.

Click Save to save the route pattern configuration.

upvoted 2 times

Testme1235 3 years, 4 months ago

Log in to Cisco Unified Communications Manager Administration. Navigate to Call Routing > Route/Hunt > Route Group. Click the Add New button to create a new route group. In the Route Group Configuration window, enter a name for the route group in the Route Group Name field. From the Distribution Algorithm drop-down menu, select the distribution algorithm that you want to use for the route group. From the Available Devices list, select the devices that you want to include in the route group, and click the Add button to move them to the Selected Devices list. Click Save to save the route group configuration. To apply the route group to a route pattern, navigate to Call Routing > Route/Hunt > Route Pattern. Select the route pattern that you want to configure, and click the Edit button. In the Route Pattern Configuration window, select the appropriate standard local route group from the Route Group drop-down menu. Click Save to save the route pattern configuration.You are describing how to configure a plain old route group. This question asks where to configure the standard local route group for a group of devices, or in other words, where you assign a standard local route group to a group of devices, which can only be done within the device pool configuration and makes C correct.

upvoted 2 times

f1ab921 1 year, 10 months ago

You are describing how to configure a plain old route group. This question asks where to configure the standard local route group for a group of devices, or in other words, where you assign a standard local route group to a group of devices, which can only be done within the device pool configuration and makes C correct.

upvoted 1 times

Mert_kerna 4 years, 3 months ago

Selected Answer: B

If the answer was C, the question would have been "where do you select the Local Route Group for a group of devices" instead of "Where do you configure the Local Route Group for a group of devices".

upvoted 2 times

Mert_kerna 4 years, 3 months ago

If the answer was C, the question would have been "where do you select the Local Route Group for a group of devices" instead of "Where do you configure the Local Route Group for a group of devices".Additionally, when you add a new Local Route Group under Call Routing > Route/Hunt > Local Route Group Names, CUCM adds it as an additional local route group setting within all device pools. You actually configure the standard local route group in Standard Local Route Group Names under Call Routing > Route/Hunt. You select the "route group" (not the "local route group", which is named standard local route group by default)of the local route group in the individual device pool. This is the difference between attention to the question and assuming you know what they're asking.Question is "where do you configure STANDARD local route group for group of devices". You don't need to configure standard local route group as it's already there by default and you can't delete it. So the only place you can do something with it it's device pool where you select route group for it. So I think more likely answer is B - device poolsorry C -device pool

upvoted 2 times

Mert_kerna 4 years, 3 months ago

Additionally, when you add a new Local Route Group under Call Routing > Route/Hunt > Local Route Group Names, CUCM adds it as an additional local route group setting within all device pools. You actually configure the standard local route group in Standard Local Route Group Names under Call Routing > Route/Hunt. You select the "route group" (not the "local route group", which is named standard local route group by default)of the local route group in the individual device pool. This is the difference between attention to the question and assuming you know what they're asking.

upvoted 2 times

basscov 4 years, 1 month ago

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upvoted 2 times

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upvoted 2 times

basscov 4 years, 1 month ago

sorry C -device pool

upvoted 2 times

enashash 5 years, 2 months ago

System-> device pool -> local route group

upvoted 1 times

pin1987 5 years, 4 months ago

Answer is C. B is wrong, because we cannot find Standard local route group their!!! 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. 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#CUCM_TK_CB653D2D_00

upvoted 3 times

SabSal 5 years, 8 months ago

Answer is B

upvoted 3 times

Freddan 5 years, 11 months ago

You can remove my comment.

upvoted 1 times

Freddan 5 years, 11 months ago

Should be C because its for a group of devices and this is configured in the device pool/Local Route Group Settings.

upvoted 3 times

Question #42 *Topic 1*

How does an engineer globalize routing for ingress calls coming from the PSTN to internal DNs?

A. At the PSTN gateway, put the calling number in PSTN format and the called number in DN format.

B. At Cisco Unified CM, put the calling number in E.164 format and the called number in PSTN format.

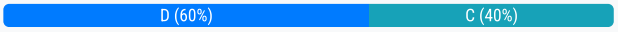
C. At the PSTN gateway, put the calling number in E.164 format and the called number in localized (DN) format.

D. At Cisco Unified Communications Manager, put the calling number in E.164 format and the called number in E.164 format.

Suggested Answer: *B*

Official Answer: B

Community vote distribution



AgshinA Highly Voted 5 years, 5 months ago

I suppose it should be D as it asks for globalizing. Should be E.164 in both places.But assuming by internal dn, they mean non DID, why would you globalized the called number coming in from the pstn. What would you even globalize the called number to? I think it should be C. Globalize the calling number at the gw and localize the 10 digit called number from pstn too match internal dn

upvoted 8 times

AgshinA Highly Voted 5 years, 5 months ago

I suppose it should be D as it asks for globalizing. Should be E.164 in both places.

upvoted 8 times

phoneguy99 4 years ago

But assuming by internal dn, they mean non DID, why would you globalized the called number coming in from the pstn. What would you even globalize the called number to? I think it should be C. Globalize the calling number at the gw and localize the 10 digit called number from pstn too match internal dn

upvoted 2 times

dansecu 5 years, 3 months ago

I think should be C: C. At the PSTN gateway, put the calling number in E.164 format and the called number in localized (DN) format. https://community.cisco.com/t5/collaboration-voice-and-video/basics-of-globalized-call-routing/ta-p/3121674 1. External to Internal: Calling-party number: E.164 Called-party number: Directory number

upvoted 7 times

dansecu 5 years, 3 months ago

I think should be C: C. At the PSTN gateway, put the calling number in E.164 format and the called number in localized (DN) format. https://community.cisco.com/t5/collaboration-voice-and-video/basics-of-globalized-call-routing/ta-p/3121674 1. External to Internal: Calling-party number: E.164 Called-party number: Directory numberGlobalized Routing in CUCM use E.164 format. With globalized call routing, numbers must use the following formats: Normalized called-party numbers: The +E.164 format is used for all external destinations during call routing. Per the recommendation in Preferred Architecture for Cisco Collaboration, you should also use the +E.164 format for calling internal phones that have a dedicated external DID number. Normalized calling-party numbers: When a call involves at least one external party, then the +E.164 global format is used for the source of the call. Per the recommendation in Preferred Architecture for Cisco Collaboration, you should also use the +E.164 format for calls coming from internal phones that have a dedicated external DID number.From your link: Normalized calling-party numbers: E.164 global format is used for all calling-party numbers except calls from a internal number to another internal number. Such purely internal calls use the internal directory number for the calling party number. If source of calls (users at phones, incoming PSTN Calls at gateways, calls received through trunks, and so on) do not use the normalized format, the localizes call ingress must be normalized before being routed. This requirement applies to all received calls (coming from gateways, trunks, and phones), and it applies to both the calling and called-party numbers. Note: Except for the internal calls that were mentioned (where the destination is a directory number and in the case of an internal source, the source is a directory number), all numbers are normalized to the E.164 global format. Therefore, call routing based on the normalized numbers is ref

upvoted 7 times

stocaxx 4 years, 6 months ago

Globalized Routing in CUCM use E.164 format. With globalized call routing, numbers must use the following formats: Normalized called-party numbers: The +E.164 format is used for all external destinations during call routing. Per the recommendation in Preferred Architecture for Cisco Collaboration, you should also use the +E.164 format for calling internal phones that have a dedicated external DID number. Normalized calling-party

numbers: When a call involves at least one external party, then the +E.164 global format is used for the source of the call. Per the recommendation in Preferred Architecture for Cisco Collaboration, you should also use the +E.164 format for calls coming from internal phones that have a dedicated external DID number.

upvoted 1 times

FrankPic 2 years, 7 months ago

From your link: Normalized calling-party numbers: E.164 global format is used for all calling-party numbers except calls from a internal number to another internal number. Such purely internal calls use the internal directory number for the calling party number. If source of calls (users at phones, incoming PSTN Calls at gateways, calls received through trunks, and so on) do not use the normalized format, the localizes call ingress must be normalized before being routed. This requirement applies to all received calls (coming from gateways, trunks, and phones), and it applies to both the calling and called-party numbers. Note: Except for the internal calls that were mentioned (where the destination is a directory number and in the case of an internal source, the source is a directory number), all numbers are normalized to the E.164 global format. Therefore, call routing based on the normalized numbers is referred to as globalized call routing. Correct Answer should be D

upvoted 1 times

decdca7 Most Recent 1 year, 5 months ago

it said globalized!, Internal DN is localiziation!

upvoted 1 times

kljw5 1 year, 7 months ago

When globalizing routing for ingress calls, the primary goal is to standardize the format of phone numbers for consistent call routing and feature application. Using the E.164 format, which is the international standard for phone numbers, ensures compatibility and simplifies routing. Ingress Call Handling: The calling number (caller ID) is normalized into E.164 format for consistent application of policies (e.g., call blocking, forwarding). The called number is also normalized into E.164 format to ensure the call can be routed correctly, even across different locations or sites. Where the Configuration Happens: This normalization is typically done in Cisco Unified Communications Manager (CUCM) using translation patterns or transformation masks. By normalizing both the calling and called numbers into E.164 format, CUCM provides a consistent and scalable approach to call routing, especially in environments with multiple sites and a mix of internal and external calls.

upvoted 2 times

b3532e4 1 year, 10 months ago

I think C and D are the correct answers, but D is better

upvoted 1 times

Testme1235 3 years, 4 months ago

Option A is incorrect because the calling number should be in E.164 format, not PSTN format. Option B is incorrect because the called number should be in localized (DN) format, not PSTN format. Option C is correct because the calling number should be in E.164 format and the called number should be in localized (DN) format. Option D is incorrect because both the calling and called numbers should be in E.164 format.

upvoted 2 times

eran1911 4 years, 10 months ago

any idea?

upvoted 1 times

Question #43 *Topic 1*

Which two types of distribution algorithm are within a line group? (Choose two.)

A. random

B. circular

C. highest preference

D. top down

E. bottom up

Suggested Answer: *B, D*

Reference: https://www.cisco.com/c/en/us/td/docs/voice_ip_comm/cucm/admin/9_0_1/ccmcfg/CUCM_BK_CDF59AFB_00_admin-guide-90/CUCM_BK_CDF59AFB_00_admin-guide_chapter_0100011.html

Community vote distribution

BD (100%)

MaxG Most Recent 2 years ago

A line group allows you to designate the order in which directory numbers are selected. Cisco Call Manager distributes a call to idle or available members of a line group and/or route group based on a call distribution algorithm and on the Ring No Answer Reversion (RNAR) setting. Available distribution algorithms: - Top Down (Default) - Circular - Longest Idle Time - Broadcast

https://www.cisco.com/en/US/docs/voice_ip_comm/cucm/admin/4_0_1/ccmcfg/b03lngrp.html#wp1027362

upvoted 2 times

Question #44 *Topic 1*

An engineer is configuring a call park feature in Cisco Unified Communications Manager Express. Which command does the engineer use to ensure that the call is reverted to the user after 60 seconds?

A. R2(config-ephone-dn)#park reservation-group 60

B. R2(config-ephone-dn)#park-slot timeout 60 limit 2 recall alternate 3002

C. R2(config-ephone-dn)#park reservation-group 1

D. R2(config-ephone-dn)#park-slot timeout 30 limit 2 recall alternate 3002

Suggested Answer: *B*

Official Answer: B

Community vote distribution

D (100%)

vic1 Highly Voted 5 years, 6 months ago

Answer: D

https://www.cisco.com/c/en/us/td/docs/voice_ip_comm/cucme/admin/configuration/manual/cmeadm/cmepark.html#concept_6DE6A02D258F4BA48D2F94A1053563AF The timeout and limit keywords and arguments also set the maximum time that calls stay parked. For example, a timeout interval of 10 seconds and a limit of 5 timeout intervals (park-slot timeout 10 limit 5) will park calls for approximately 50 seconds.

upvoted 17 times

kljw5 Most Recent 1 year, 7 months ago

R2(config-ephone-dn)#park-slot timeout 30 limit 2 recall alternate 3002 Timeout = 30 seconds: This means each interval is 30 seconds long. Limit = 2: This means there are 2 timeout intervals, resulting in a total parked duration of: 30 seconds (timeout) × 2 intervals (limit) = 60 seconds total . 30seconds (timeout)×2intervals (limit)=60seconds total. Recall Alternate 3002: The call will revert to the alternate DN (3002) after the total timeout. Analysis: This configuration results in the parked call staying parked for exactly 60 seconds before reverting. Therefore, Option D correctly fulfills the requirement.

upvoted 1 times

b3532e4 1 year, 7 months ago

Answer is definitely D.

upvoted 1 times

f1ab921 1 year, 10 months ago

configuration on B calls user after 120 seconds (2 timeouts x 60 sec). The answer is D. 2 timeouts x 30 sec = 60 sec.

upvoted 1 times

b3532e4 1 year, 10 months ago

Answer: D

upvoted 2 times

729edae 2 years, 3 months ago

Answer is definitely D.

upvoted 3 times

FrankPic 2 years, 7 months ago

Answer: D https://www.cisco.com/c/en/us/td/docs/voice_ip_comm/cucme/admin/configuration/manual/cmeadm/cmepark.html A reminder ring can be sent to the extension that parked the call by using the timeout keyword with the park-slot command. The timeout keyword and argument set the interval length during which the call-park reminder ring is timed out or inactive. If the timeout keyword is not used, no reminder ring is sent to the extension that parked the call. The number of timeout intervals and reminder rings are configured with the limit keyword and argument. For example, a limit of 3 timeout intervals sends 2 reminder rings (interval 1, ring 1, interval 2, ring 2, interval 3). The timeout and limit keywords and arguments also set the maximum time that calls stay parked. For example, a timeout interval of 10 seconds and a limit of 5 timeout intervals (park-slot timeout 10 limit 5) will park calls for approximately 50 seconds.

upvoted 1 times

Question #45 *Topic 1*

Which call pickup feature allows users to pick up incoming calls in a group that is associated with their own group?

A. Other Group Pickup

B. BLF Call Pickup

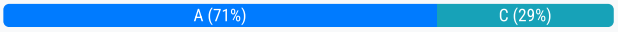
C. Group Call Pickup

D. Directed Call Pickup

Suggested Answer: *A*

Reference: https://www.cisco.com/en/US/docs/voice_ip_comm/cucm/admin/4_1_3/ccmsys/a07cpick.html#wp1022865

Community vote distribution



kljw5 Most Recent 1 year, 7 months ago

A is 100% correct based on MaxG's post read the feature descriptions. The Other Group Pickup feature allows users to pick up incoming calls in a group that is associated with their own group

upvoted 1 times

MaxG 2 years ago

Other Group Pickup feature allows users to pick up incoming calls in a group that is associated with their own group. The Unified Communications Manager automatically searches for the incoming call in the associated groups to make the call connection when the user activates this feature from a Cisco Unified IP Phone. Users use the softkey, OPickUp, for this type of call pickup.
https://www.cisco.com/c/en/us/td/docs/voice_ip_comm/cucm/admin/11_5_1/cucm_b_feature-configuration-guide-cisco1151su8/cucm_b_feature-configuration-guide-cisco1151su8_chapter_011010.html#CUCM_CN_ODD272E5_00

upvoted 4 times

Testme1235 2 years, 4 months ago

The call pickup feature that allows users to pick up incoming calls in a group that is associated with their own group is typically referred to as "group call pickup" or "group pickup." This feature enables members of a particular group or department to answer calls that are ringing on another member's phone within the same group. When a call is received, any member of the group can pick up the call by dialing a specific code or pressing a pre-assigned button on their phone. This feature is commonly used in businesses and organizations to improve communication and collaboration among team members.Look into it again, the correct answer is AThe group pickup is called Other PickUp oPickup in the softkey templates

upvoted 2 times

Testme1235 2 years, 4 months ago

Selected Answer: C

The call pickup feature that allows users to pick up incoming calls in a group that is associated with their own group is typically referred to as "group call pickup" or "group pickup." This feature enables members of a particular group or department to answer calls that are ringing on another member's phone within the same group. When a call is received, any member of the group can pick up the call by dialing a specific code or pressing a pre-assigned button on their phone. This feature is commonly used in businesses and organizations to improve communication and collaboration among team members.

upvoted 2 times

Testme1235 2 years, 4 months ago

Look into it again, the correct answer is A

upvoted 3 times

Afaik 2 years ago

The group pickup is called Other PickUp oPickup in the softkey templates

upvoted 2 times

Question #46 *Topic 1*

When locations-based Call Admission Control denies the call, which two masks can AAR apply when routing the call through the PSTN? (Choose two.)

A. AAR destination mask

B. called party transform mask

C. external phone number mask

D. +E.164 alternate number mask

E. enterrise alternate number mask

Suggested Answer: *A, C*

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

Community vote distribution

AC (100%)

MaxG Most Recent 2 years ago

The rerouting of calls requires using a destination number that can be routed through the alternate network (for example, the PSTN). Automated Alternate Routing (AAR) uses the dialed digits to establish the on-cluster destination of the call and then combines them with the called party's AAR Destination Mask; if it is not configured, the External Phone Number Mask is used instead. The combination of the dialed digits and the applicable mask must yield a fully qualified number that can be routed by the alternate network.

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

upvoted 1 times

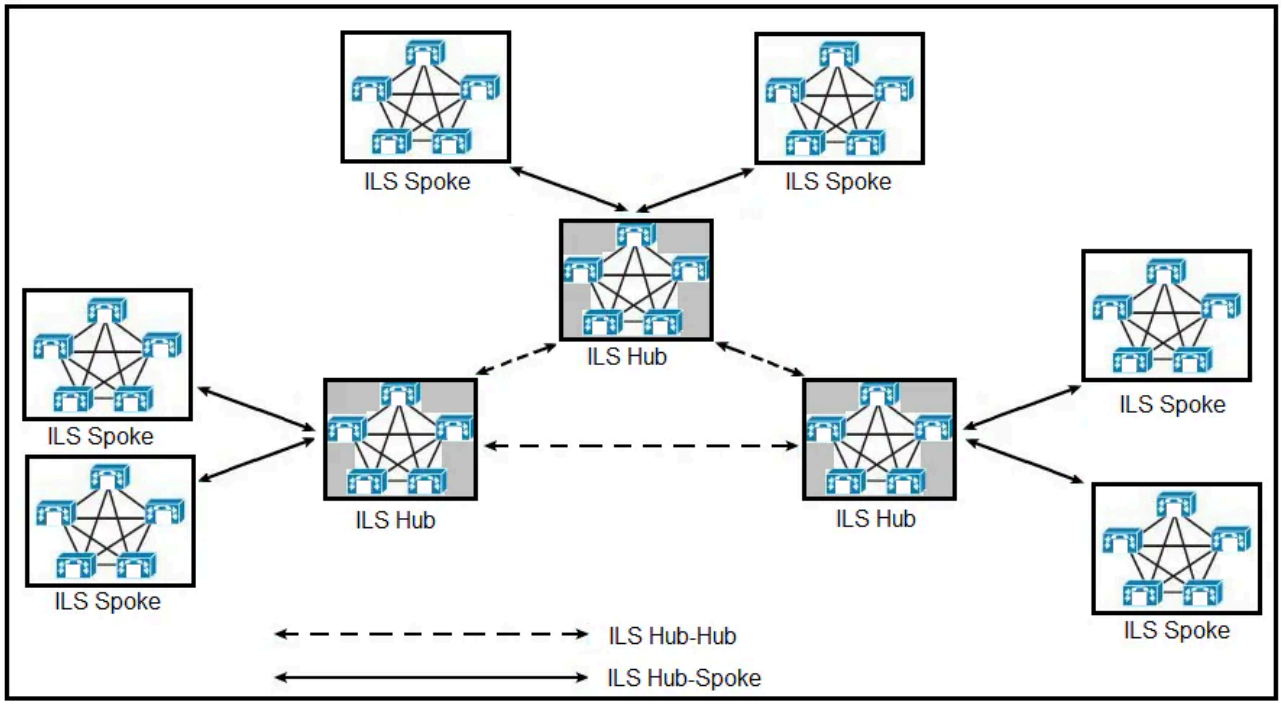
Testme1235 2 years, 4 months ago

The two masks that AAR can apply when routing the call through the PSTN are: The External Phone Number Mask The AAR Destination Mask The External Phone Number Mask is a mask that is used to manipulate the called party number. This mask can be used to add or remove digits from the called party number. The AAR Destination Mask is a mask that is used to manipulate the destination number. This mask can be used to add or remove digits from the destination number. AAR can also apply the dial prefix defined under the AAA Group when routing the call through the PSTN.

upvoted 2 times

Question #47 *Topic 1*

Refer to the exhibit. How many maximum hops can an ILS updarte traverse?



A. 3

B. 6

C. 9

D. 12

Suggested Answer: *A*

Official Answer: A

Community vote distribution

A (75%) C (25%)

timmyz 2 years ago

answer is A Allow enough time for end-to-end replication based on synchronization intervals (set on the ILS Configuration page) that are configured for all the clusters involved in the path. All clusters in an ILS network are a maximum of three hops from every other cluster in the network. 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_mp_ife214af_00_intercluster-lookup-service.html#:~:text=All%20clusters%20in%20an%20ILS,other%20cluster%20in%20the%20network.

upvoted 1 times

john_doe_9999 2 years, 1 month ago

A is correct

upvoted 3 times

MaxG 2 years, 6 months ago

All clusters in Intercluster Lookup Service (ILS) network are a maximum of three hops from every other cluster in the network. https://www.cisco.com/c/en/us/td/docs/voice_ip_comm/cucm/srnd/collab10/collab10/dialplan.htmlSo answer is A.

upvoted 1 times

MaxG 2 years, 6 months ago

Selected Answer: C

All clusters in Intercluster Lookup Service (ILS) network are a maximum of three hops from every other cluster in the network.

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

upvoted 1 times

f77380c Most Recent 4 months, 1 week ago

So answer is A.

upvoted 1 times

Question #48 *Topic 1*

An administrator is configuring a cluster for ILS and wants to limit the amount of entities that Cisco Unified Communications Manager can write to the database for data that is learned through ILS. Which service parameter is used to adjust this limit?

A. ILS Max Number of Learned Objects in Database

B. ILS Active Learned Object Upper Limit

C. Global Data Service Parameter Limit

D. Imported Dial Plan Replication Database Object Lower Limit

Suggested Answer: *A*

Reference: https://www.cisco.com/c/en/us/td/docs/voice_ip_comm/cucm/admin/12_5_1SU1/systemConfig/cucm_b_system-configuration-guide-1251su1/ cucm_b_system-configuration-guide-1251su1_restructured_chapter_0100011.html#CUCM_TK_I7C708C2_00

Community vote distribution

A (100%)

john_doe_9999 Most Recent 2 years, 1 month ago

A is correct

upvoted 1 times

MaxG 2 years, 6 months ago

See Set Database Limits for Learned Data

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_011010.html#CUCM_TK_I7C708C2_00

upvoted 1 times

Question #49 *Topic 1*

When configuring hunt groups, where do you add the individual directory numbers that will be part of the group?

A. route group

B. line group

C. hunt list

D. hunt pilot

Suggested Answer: *B*

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_010101.html

Community vote distribution

B (100%)

MaxG Most Recent 2 years ago

We have a pilot number for the Hunt Group. So there's a Hunt Pilot, it points to a Hunt List, a line group, and then directory numbers. In other words we have to have directory numbers created first, those directory numbers get added to a Line Group, which then gets added to a Hunt List which then gets pointed to by that Hunt Pilot number. https://www.learncisco.net/courses/icomm-ccna-voice/call-flows-in-cuc-systems/hunt-groups-and-line-groups.html

upvoted 2 times

Question #50 *Topic 1*

Which configuration element of a hunt group allows for changing Calling Party Transformations settings?

A. line group

B. hunt pilot

C. route group

D. hunt list

Suggested Answer: *B*

Reference: https://community.cisco.com/t5/ip-telephony-and-phones/call-alerting-on-hunt-group-as-shared-line/td-p/2658015

Community vote distribution

B (100%)

b3532e4 Most Recent 1 year, 7 months ago

Selected Answer: B

upvoted 1 times

MaxG 2 years ago

https://www.cisco.com/en/US/docs/voice_ip_comm/cucm/admin/4_1_3/ccmcfg/b03htpil.html#wp1023973

upvoted 3 times

Question #51 *Topic 1*

Which two types of authentication are supported for the configuration of Intercluster Lookup Service? (Choose two.)

A. TokenID

B. username and secret key

C. TLS certificates

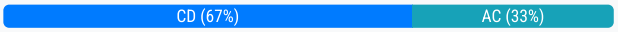
D. passwords

E. FQDN of the servers defined in DNS

Suggested Answer: *C, D*

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_011001.pdf

Community vote distribution



jatoja83 1 year, 5 months ago

Selected Answer: AC

Explanation: When configuring Intercluster Lookup Service (ILS) in Cisco Unified Communications Manager (CUCM), two types of authentication are supported to secure communication between clusters:

TokenID (A): This is a unique identifier used for authentication between clusters in ILS. It helps to ensure secure communication by verifying that the clusters are authorized to exchange information.

TLS certificates (C): Transport Layer Security (TLS) certificates are used for encrypted communication between clusters. TLS provides secure encryption of the data transmitted between CUCM clusters, ensuring that the ILS data is protected during transit.

upvoted 1 times

jatoja83 1 year, 5 months ago

Explanation: When configuring Intercluster Lookup Service (ILS) in Cisco Unified Communications Manager (CUCM), two types of authentication are supported to secure communication between clusters: TokenID (A): This is a unique identifier used for authentication between clusters in ILS. It helps to ensure secure communication by verifying that the clusters are authorized to exchange information. TLS certificates (C): Transport Layer Security (TLS) certificates are used for encrypted communication between clusters. TLS provides secure encryption of the data transmitted between CUCM clusters, ensuring that the ILS data is protected during transit.The two types of authentication supported for the configuration of the Intercluster Lookup Service (ILS) are: C. TLS certificates D. passwords

upvoted 1 times

jatoja83 Most Recent 1 year, 5 months ago

The two types of authentication supported for the configuration of the Intercluster Lookup Service (ILS) are: C. TLS certificates D. passwords

upvoted 1 times

MaxG 2 years ago

- Use TLS authentication between clusters in the ILS network. - Use password authentication between remote clusters in the ILS network. - Use TLS and password authentication to setup a ILS network using common Certificate Authority (CA) signed certificates without exchanging self-signed certificates between clusters. https://www.cisco.com/c/en/us/td/docs/voice_ip_comm/cucm/admin/11_5_1_SU7/cucm_b_system-configuration-1151su7-1151su8/cucm_b_system-configuration-guide-1151su1_chapter_011001.html#CUCM_TK_I2D5987B_00

upvoted 2 times

Question #52 *Topic 1*

Which two configuration parameters are prerequisites to set Native Call Queuing on Cisco Unified Communications Manager? (Choose two.)

A. Cisco IP Voice Media Streaming Service must be activated on at least one node in the cluster.

B. A unicast music on hold audio source must be configured.

C. Cisco RIS data collector service must be running on the same server as the Cisco CallManager service.

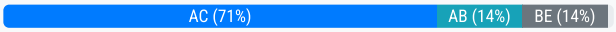
D. The maximum number of callers allowed in queue must be 10.

E. The phone button template must have the Queue Status Softkey configured.

Suggested Answer: *A, C*

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_01001101.html#CUCM_RF_C960BC9A_00

Community vote distribution



f77380c Most Recent 4 months, 2 weeks ago

Only A & B are required. Some of the other options are not necessarily required.

upvoted 1 times

b3532e4 1 year, 7 months ago

Call Queuing Prerequisites Cisco IP Voice Media Streaming (IPVMS) Application, which should be activated on at least one node in the cluster Cisco CallManager service that is running on at least one server in the cluster Cisco RIS Data Collector service that is running on the same server as the Cisco CallManager service Cisco Unified Communications Manager Locale Installer, if you want to use non-English phone locales or country-specific tones

upvoted 1 times

MaxG 2 years ago

Call Queuing Prerequisites - Cisco IP Voice Media Streaming (IPVMS) Application, which should be activated on at least one node in the cluster - Cisco CallManager service that is running on at least one server in the cluster - Cisco RIS Data Collector service that is running on the same server as the Cisco CallManager service - Cisco Unified Communications Manager Locale Installer, if you want to use non-English phone locales or country-specific tones 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_01001101.html#CUCM_RF_C960BC9A_00

upvoted 4 times

Afaik 2 years ago

The question does not make any sense. A. is required for native B. but basically no parameter B. despite unicast is required C. not relevant for native call queuing? D. Can be 1-100. E. If you want to use it it has to be applied. Imho B and E are at least related configuration parameters for native call queuing...

upvoted 1 times

Question #53 *Topic 1*

What is the relationship between partition, time schedule, and time period in Time-of-Day routing in Cisco Unified Communications Manager?

A. A partition can have multiple time schedules assigned. A time schedule contains one or more time periods.

B. A partition can have one time schedule assigned. A time schedule contains one or more time periods.

C. A partition can have multiple time schedules assigned. A time schedule contains only one time period.

D. A partition can have one time schedule assigned. A time schedule contains only one time period.

Suggested Answer: *A*

Official Answer: A

Community vote distribution

B (100%)

rishik Highly Voted 4 years, 11 months ago

Correct answer is B

upvoted 16 times

Grebec94 Highly Voted 5 years, 1 month ago

Partition can have only one schedule setup

upvoted 6 times

MaxG Most Recent 2 years ago

Time Period: Defines a time frame as 24 hours of a day. Time Schedule: This is a collection of one or more time periods. Partition: A partition comprises of a logical grouping of Directory Numbers (DNs) and route patterns with similar reachability characteristics. The time schedule comprises of one or more time periods assigned to a partition. It defines a time frame when the partition is logically active.
https://www.cisco.com/c/en/us/support/docs/voice-unified-communications/unified-communications-manager-callmanager/109588-tod-routing-example00.html

upvoted 1 times

enashash 4 years, 2 months ago

As mentioned earlier, partitions allow you to assign a time schedule to a partition. A time schedule contains a list of time periods. Figure 4-19 shows a configuration example for a time period named Work Day that includes the times from 9 a.m. to 6 p.m. on Monday through Friday. This time period can then be added to a time schedule, as shown in Figure 4-20. The time schedule can include more than one time period. For example, you might create a time period for each holiday and then create a time schedule named Holidays that includes all the holiday time periods. page 162 cert. study guide CORRECT ANSWER IS B

upvoted 2 times

Question #54 *Topic 1*

Configure Call Queuing in Cisco Unified Communications Manager. Where do you set the maximum number of callers in the queue?

A. in the telephony service configuration

B. in the queuing configuration

C. in Cisco Unified CM Enterprise Parameters

D. in Cisco Unified CM Service Parameters

Suggested Answer: *B*

Reference: https://www.cisco.com/c/en/us/support/docs/unified-communications/unified-communications-manager-callmanager/200453-Configure-CUCM-Native- Call-Queuing-Featu.html

Community vote distribution

B (100%)

v1nhthanh Most Recent 1 year, 3 months ago

Max 100 default 32

upvoted 1 times

MaxG 2 years ago

Another reference. Picture all the way at the bottom shoing the "Queuing" configuration https://geekstuff.org/native-call-queuing-cucm/

upvoted 1 times

Question #55 *Topic 1*

A user reports that when they attempt to log out from the Cisco Extension Mobility service by pressing the Services button, they cannot log out. What is the most likely cause of this issue?

A. The Cisco Extension Mobility service has not been configured on the phone.

B. There might be a significant delay between the button being pressed and the Cisco Extension Mobility service recognizing it. It would be best to check network latency.

C. The user device profile has not been assigned to the user.

D. The user device profile is not subscribed to the Cisco Extension Mobility service.

Suggested Answer: *D*
Official Answer: D

Community vote distribution
D (100%)

f77380c Most Recent 4 months, 2 weeks ago
A & D is possible. But if A is correct, the user would not even be able to login which does not seem to be a problem. So it must be D.
upvoted 1 times

MaxG 2 years ago
Subscribe to Extension Mobility Subscribe IP phones and device profiles to the extension mobility service so that users can log in, use, and log out of extension mobility. 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_011101.html#CUCM_TK_S7548027_00
upvoted 2 times

Question #56 *Topic 1*

What is a component of Cisco Unified Mobility?

A. Unified IVR

B. Mobile Connect

C. Smart Client Support

D. Single Number Connect

Suggested Answer: *B*

Official Answer: B

Community vote distribution

B (100%)

john_doe_9999 Most Recent 2 years, 1 month ago

B is correct

upvoted 1 times

MaxG 2 years, 6 months ago

Cisco Unified Mobility makes Cisco Mobile Connect services available to Cisco Unified Communications Manager users who want to consolidate all their business calls with a single enterprise IP phone number and immediately connect wherever they are working. Cisco Mobile Connect service helps mobile workers direct their inbound business calls to their IP phone number and initiate outbound business calls as if they were at their IP phone, all from the mobile phone (or other remote phone destination). https://www.cisco.com/c/en/us/products/collateral/unified-communications/unified-mobility/product_data_sheet0900aecd80410f2d.html

upvoted 2 times

Question #57 *Topic 1*

When the services key is pressed Cisco Extension Mobility does not show up. What is the cause of the issue?

A. The URL configured for Cisco Extension Mobility is not correct.

B. Cisco Extension Mobility Service is not running.

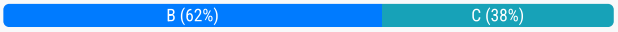
C. The phone is not subscribed to Cisco Extension Mobility Service.

D. Cisco Extension Mobility is not enabled in the Phone Configuration Window (Device > Phone)

Suggested Answer: *C*

Official Answer: C

Community vote distribution



f77380c 4 months, 2 weeks ago

Selected Answer: C

C is correct

upvoted 1 times

Gary1968 3 months, 1 week ago

This would typically happen when the phone is not subscribed to the EM service.

upvoted 1 times

f77380c 4 months, 2 weeks ago

C is correctB is not correct - if the service is not running and the phone is subscribed, the EM will be listed but result in error if selected.

upvoted 2 times

v1nhthanh 1 year, 3 months ago

Selected Answer: B

What would be the result if it was C? They will not see Extension Mobility?

upvoted 3 times

f77380c 4 months, 2 weeks ago

B is not correct - if the service is not running and the phone is subscribed, the EM will be listed but result in error if selected.

upvoted 1 times

v1nhthanh 1 year, 3 months ago

What would be the result if it was C? They will not see Extension Mobility?Wouldn't C be the answer in that case?

upvoted 4 times

Gary1968 **Most Recent** 1 month, 2 weeks ago

Wouldn't C be the answer in that case?

upvoted 1 times

jatoja83 1 year, 5 months ago

he cause of the issue where Cisco Extension Mobility does not show up when the services key is pressed is most likely: B. Cisco Extension Mobility Service is not running. Explanation: Cisco Extension Mobility relies on the Extension Mobility Service to function. If this service is not running, the feature will not be available on the phone, and pressing the services key will not show Cisco Extension Mobility as an option.

upvoted 1 times

Question #58 *Topic 1*

A user reports when they press the services key they do not receive a user ID and password prompt to assign the phone extension. Which action resolves the issue?

A. Create the default device profiles for all phone models that are used.

B. Subscribe the phone to the Cisco Extension Mobility service.

C. Create the end user and associate it to the device profile.

D. Assign the extension as a mobile extension.

Suggested Answer: *B*
Official Answer: B

Rustynailz Most Recent 2 years ago

B should be correct. After unchecking "enable extension mobility" under phone in CUCM, the user name and password prompt disappeared from my phone.

upvoted 2 times

Question #59 *Topic 1*

What are the elements for Device Mobility configuration?

A. physical location, device pool, and Device Mobility group

B. device pool, Device Mobility group, and region

C. physical location. Device Mobility group, and region

D. device pool, Device Mobility group, and Cisco IP phone

Suggested Answer: *A*

Reference: https://www.ciscopress.com/articles/article.asp?p=1249228&seqNum=4

Community vote distribution

C (100%)

Akularn Most Recent 8 months, 2 weeks ago

C: https://www.cisco.com/c/en/us/td/docs/voice_ip_comm/cucm/admin/cucm_b_feature-configuration-guide-for-15/cucm_m_configure-device-mobility-1251.html

upvoted 1 times

john_doe_9999 2 years, 1 month ago

A is correct

upvoted 3 times

Question #60 *Topic 1*

Which services are needed to successfully implement Cisco Extension Mobility in a standalone Cisco Unified Communications Manager server?

A. Cisco Extended Functions, Cisco Extension Mobility, and Cisco AXL Web Service

B. Cisco CallManager, Cisco TFTP, and Cisco CallManager SNMP Service

C. Cisco CallManager, Cisco TFTP, and Cisco Extension Mobility

D. Cisco TAPS Service, Cisco TFTP, and Cisco Extension Mobility

Suggested Answer: *C*

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_011101.html#CUCM_TK_A337E035_00

Community vote distribution

C (100%)

MaxG Most Recent 2 years ago

Activate the following services as needed: a. Cisco CallManager b. Cisco Tftp c. Cisco Extension Mobility

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_011101.html#CUCM_TK_A337E035_00

upvoted 2 times

Question #61 *Topic 1*

Refer to the exhibit. While troubleshooting call failures on the Cisco Unified Border Element, an administrator notices that messages are being sent to the service provide, but there is no response. The administrator later learns that this SIP provider does not support PRACK. Which header should be removed from the SIP message to resolve this issue?

```
SIP/2.0 183 Session Progress
Via: SIP/2.0/UDP 192.168.100.100:5060
From: <sip:+123456789@192.168.100.100>;
To: <sip:987654321@192.168.100.200>
Date: Fri, 28 Jun 2019 08:30:32 GMT
Call-ID: fce8c980-d151d028-19cf3-325900a@192.168.100.100
CSeq: 101 INVITE
Require: 100rel
RSeq: 101
Allow: INVITE, OPTIONS, BYE, CANCEL, ACK, PRACK, UPDATE, REFER, SUBSCRIBE, NOTIFY, INFO, REGISTER
Contact: <sip:987654321@192.168.100.200:5060>
Content-Type: application/sdp
Content-Disposition: session;handling=required
Content-Length: 247

v=0
o=CiscoSystemsSIP-GW-UserAgent 4780 5245 IN IP4 192.168.100.200
s=SIP Call
c=IN IP4 192.168.100.200
t=0 0
m=audio 16384 RTP/AVP 8 101
c=IN IP4 192.168.100.200
a=rtpmap:8 PCMA/8000
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-15
a=ptime:20
```

A. Require: 100rel

B. Content-Type: application/sdp

C. Contact:

D. Content-Disposition: session;handling=required

Suggested Answer: *A*

Official Answer: A

Community vote distribution

A (100%)

MaxG Most Recent 2 years ago

The 100rel is a SIP option tag used to indicate support for reliability of provisional responses. The CSP supports 100rel in the following headers: - Supported - Required https://www.dialogic.com/webhelp/csp1010/8.4.1_ipn3/sip_software_chap_-_prack_support.htm

upvoted 2 times

Question #62 *Topic 1*

The SIP session refresh timer allows the RTP session to stay active during an active call. The Cisco UCM sends either SIP-INVITE or SIP-UPDATE messages in a regular interval of time throughout the active duration of the call. During a troubleshooting session, the engineer finds that the Cisco UCM is sending SIP-UPDATE as the SIP session refresher, and the engineer would like to use SIP-INVITE as the session refresher. What configuration should be made in the Cisco UCM to achieve this?

A. Change Session Refresh Method on the SIP profile to INVITE.

B. Increase Retry INVITE to 20 seconds on the SIP profile.

C. Enable Send send-receive SDP in mid-call INVITE on the SIP profile.

D. Enable SIP Rel1XX Options on the SIP profile.

Suggested Answer: *A*

Official Answer: A

Community vote distribution

A (100%)

domangez Most Recent 1 year, 10 months ago

SIP profile > Session Refresh Method : INVITE or UPDATE (A is correct)

upvoted 1 times

MaxG 2 years, 6 months ago

See Avoid Interoperability Issues with Update Refresh section: https://www.cisco.com/c/en/us/support/docs/unified-communications/unified-border-element/213843-troubleshoot-session-refresh-on-cube.html

upvoted 1 times

Question #63 *Topic 1*

Refer to the exhibit. An engineer is troubleshooting an issue where inbound calls are failing after they are transferred. The provider reports that UPDATE is not supported, and this is causing the calls to fail. Which command should resolve this issue?

```
Received
UPDATE sip:192.168.100.101:5060;transport=udp SIP/20.0
Via: SIP/2.0/UDP 192.168.200.101:5060;branch=
From: "Amy" <sip:2001@192.168.100.101:5060;user=phone>;tag=
To: "Bob" <sip:2002@192.168.100.101:5060;user=phone>;tag=
Call-ID: abcd1234@192.168.200.101
Max-Forwards: 70
Timestamp: 123456789
Allow: INVITE, OPTIONS, BYE, CANCEL, ACK, PRACK, UPDATE, REFER,
SUBSCRIBE, NOTIFY, INFO, REGISTER
Cseq: 101 UPDATE
Contact: <sip:2001@192.168.200.101:5060>
Min-SE: 2000
P-Asserted-Identity: "Joe" <sip:3010@192.168.200.101>
Content-Length: 0
```

A. no midcall-signaling passthru

B. no update-callerid

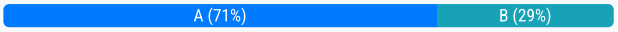
C. no contact-passing

D. rel1xx require "100rel"

Suggested Answer: *B*

Official Answer: B

Community vote distribution



decdca7 Most Recent 1 year, 5 months ago

no update-callerid – This command disables caller ID updates mid-call, but the issue is with the unsupported UPDATE method, not caller ID updates.

upvoted 1 times

Piji 1 year, 6 months ago

The SIP UPDATE method is used to modify session parameters during an ongoing call without affecting the media. Some providers do not support the SIP UPDATE method, which can cause call failures. The "no midcall-signaling passthru" command ensures that mid-call signaling (such as UPDATE) is not passed through, preventing unsupported UPDATE messages from being sent to the provider.

upvoted 2 times

b3532e4 1 year, 7 months ago

no midcall-signaling passthru and Session Refresh Method change to INVITE Totally, this Q is freaky

upvoted 2 times

b3532e4 1 year, 10 months ago

Actually is A

upvoted 2 times

b3532e4 1 year, 10 months ago

Actually is Asorry B

upvoted 2 times

b3532e4 1 year, 9 months ago

sorry B

upvoted 1 times

blude 2 years, 9 months ago

Actually is B

upvoted 1 times

blude 2 years, 9 months ago

Tha document show the answer is D

upvoted 1 times

MaxG 3 years ago

If no update-callerid command is enabled and UPDATE request contains only caller-ID changes, then re-negotiation does not happen for any early dialog caller-ID changes. If UPDATE request contains transcoder changes or video escalation or de-escalation, re-negotiation happens even if no update-callerid command is enabled. https://www.cisco.com/c/en/us/td/docs/ios-xml/ios/voice/cube/configuration/cube-book/voi-cube-edupdate-block.html

upvoted 2 times

Question #64 *Topic 1*

Refer to the exhibit. An administrator has configured a SIP trunk between two Cisco UCM clusters. For calls that should use the trunk, the calls fail with a fast busy. The administrator checks the Cisco CallManager SDL traces and found that the cluster to which the calling device is registered never sends an INVITE to the destination cluster. The administrator also verifies that all nodes from both clusters are powered on, and the CallManager service is running. How is this issue resolved?

```
SIPHandler/ccbId=0/scbId=0/wait_SIPTimer: TimerExpired
type=SIP_TIMER_WAIT_CONNECT value=5000 retries=0
Stack/Transport/0x0xee9c8980/sipTransportPostInternalMsg: Posting Internal Msg
type=1
Stack/Transport/0x0/sipTransportPostCloseConnection: Posting TCP conn close for
addr=10.10.5.11, port=5060, connid=20
Stack/Transport/0x0/sipDeleteConnInstance: Deleted conn=0xe7ac06c0, connid=20,
addr=10.10.5.11, port=5060, transport=ICP
Stack/Info/0x0/ccsip_process_sipspi_queue_event: ccsip_spi_get_msg_type returned:
2 (SIP_NETWORK_MSG), for event 64 (SIPSPI_EV_INTERNAL_MSG)
Stack/Error/0x0xee9c8950/sipTransportPostSendFailure: Posting send failure msg
with tcb: (nil) reason=4
Stack/Infor/0x0/ccsip_process_sipspi_queue_event: ccsip_spi_get_msg_type
returned: 2 (SIP_NETWORK_MSG), for event 55 (SIPSPI_EV_SEND_FAILURE_MSG)
Stack/Info/0x0xee9c8980/ccsip_spi_process_event: Send Error for event
(0xee9cb8b0)
Stack/Error/0x0/act_idle_send_msg_failure: Send Error to 10.10.5.11:5060 for
transport TCP
Stack/Info/0x0xee9c8980/ccsip_set_oo_cause_for_spi_err: Categorized cause: 38,
category:186
Stack/Info/0x0xee9c8980/sipSPIInitiateDisconnect: Initiate call disconnect (38)
for outgoing call
SIPHandler/ccbId=22609/scbId=0/ccsip_api_call_disconnected: ccb->cc_disc_cause
(38): ccb->sip_disc_cause (503)
SIPHandler/ccbId=22609/scbId=0/findDevicePID: Routed to SIPD by ccbId/scbId
Stack/States/0x0xee9c8980/sipSPIChangeState: 0xee9c8980 : State change from
(STATE_IDLE, SUBSTATE_NONE) to (STATE_DISONNECTING, SUBSTATE_NONE)
```

A. The administrator must associate the route pattern with a calling search space the device can dial.

B. The administrator needs to enable OPTIONS pings on the SIP trunks for both clusters.

C. The administrator must allow connectivity so that TCP connections do not fail between the nodes.

D. The administrator needs to disable OPTIONS pings on the SIP trunks for both clusters.

Suggested Answer: *C*

Official Answer: C

Community vote distribution

C (100%)

Piji Most Recent 1 year, 6 months ago

The logs indicate that the SIP trunk is using TCP transport, and the failure occurs when trying to establish the connection. If the TCP connection fails, SIP messages (like INVITE) cannot be sent between the clusters. The administrator should check firewall rules, ACLs, and routing to ensure TCP traffic is allowed between both clusters on port 5060 (for SIP) or 5061 (if using secure SIP). A (Route Pattern) is incorrect because the issue is not about dialing permissions but about network connectivity. B (OPTIONS pings) is unrelated, as disabling OPTIONS would not solve a connectivity failure. D (OPTIONS pings disabling) is not a relevant fix here.

upvoted 1 times

b3532e4 1 year, 7 months ago

My opinion B and C are correct, but a don't know

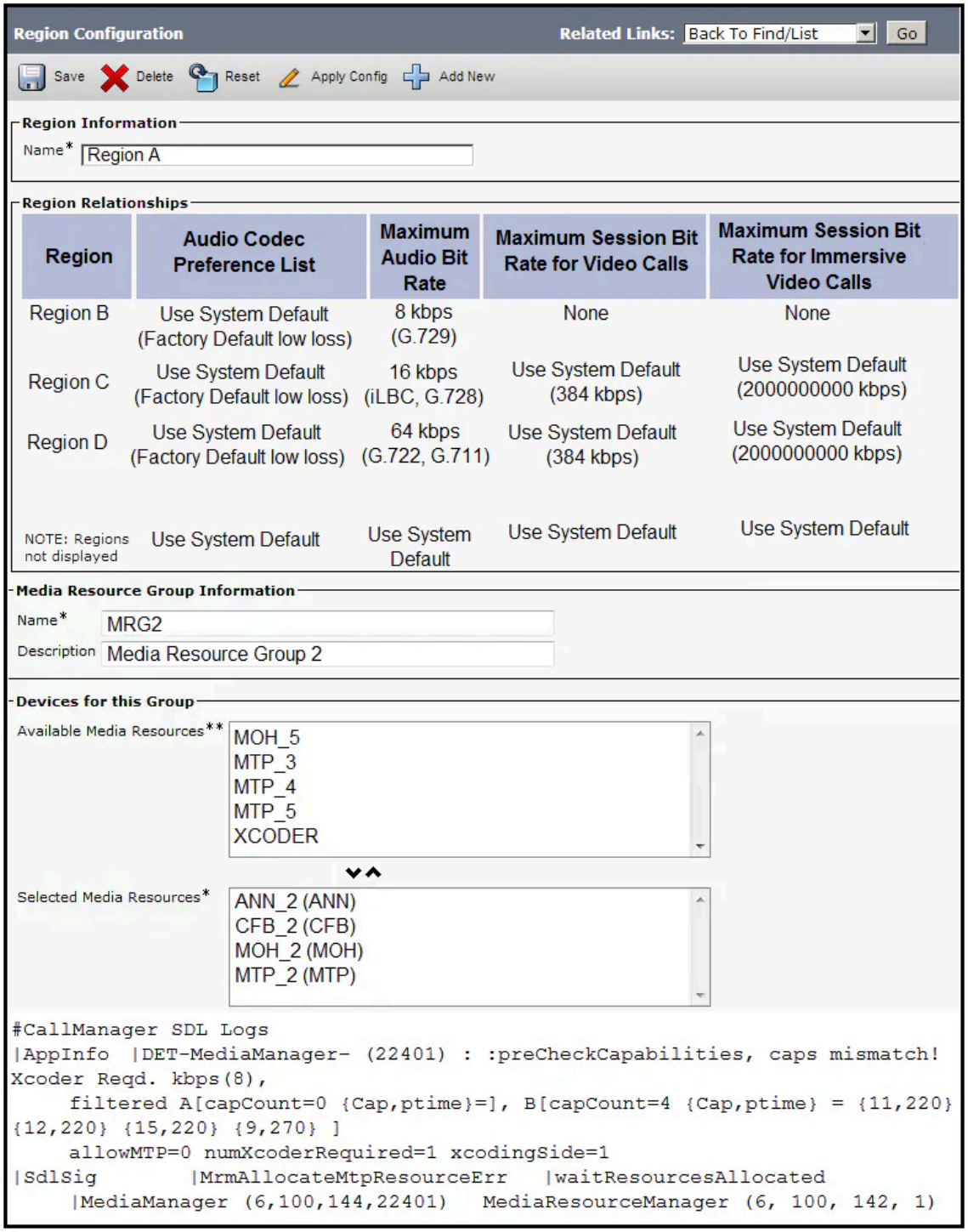
upvoted 1 times

domangez 1 year, 10 months ago

Destination IP is in the log, so route pattern has the destination device. But the connection (TCP) to that IP is failing. No indication of Options Ping problem.

upvoted 1 times

Refer to the exhibit. Regions have been configured for all major branches based on the available circuit bandwidth. Some calls from Region A endpoints to Region B endpoints are failing to connect. How is the issue resolved?



Region Configuration Related Links: Back To Find/List Go

Save Delete Reset Apply Config Add New

Region Information

Name* Region A

Region Relationships

Region	Audio Codec Preference List	Maximum Audio Bit Rate	Maximum Session Bit Rate for Video Calls	Maximum Session Bit Rate for Immersive Video Calls
Region B	Use System Default (Factory Default low loss)	8 kbps (G.729)	None	None
Region C	Use System Default (Factory Default low loss)	16 kbps (iLBC, G.728)	Use System Default (384 kbps)	Use System Default (2000000000 kbps)
Region D	Use System Default (Factory Default low loss)	64 kbps (G.722, G.711)	Use System Default (384 kbps)	Use System Default (2000000000 kbps)
NOTE: Regions not displayed	Use System Default	Use System Default	Use System Default	Use System Default

Media Resource Group Information

Name* MRG2

Description Media Resource Group 2

Devices for this Group

Available Media Resources** MOH_5, MTP_3, MTP_4, MTP_5, XCODER

Selected Media Resources* ANN_2 (ANN), CFB_2 (CFB), MOH_2 (MOH), MTP_2 (MTP)

```
#CallManager SDL Logs
|AppInfo  |DET-MediaManager- (22401) : :preCheckCapabilities, caps mismatch!
Xcoder Reqd. kbps(8),
     filtered A[capCount=0 {Cap,ptime}=], B[capCount=4 {Cap,ptime} = {11,220}
{12,220} {15,220} {9,270} ]
     allowMTP=0 numXcoderRequired=1 xcodingSide=1
|SdlSig         |MrmAllocateMtpResourceErr    |waitResourcesAllocated
     |MediaManager (6,100,144,22401)   MediaResourceManager (6, 100, 142, 1)
```

A. Update the calling search space for affected endpoints to none.

B. Update all regions to 8 kbps maximum audio bitrate.

C. Increase the number of available media termination points.

D. Add a media resource to transcode between available capabilities.

Suggested Answer: *D*

Official Answer: D

Community vote distribution

D (100%)

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

Question #66 *Topic 1*

55697959.007 |12:20:50.913 |AppInfo |RouteListCdrc::createPartyTransformedCcSetupReqMsg - before DAapplyCdpnXform() preXformCdpn=11112222 preTag=SUBSCRIBER prePos=11112222 crCdpnMask=33334444 crPrefixDigit= crDDI=2
55697959.008 |12:20:50.913 |AppInfo |RouteListCdrc::createPartyTransformedCcSetupReqMsg - after DAapplyCdpnXform() xformCdpn=33334444 xformTag=SUBSCRIBER xformPos=11112222
55697959.009 |12:20:50.913 |AppInfo |RouteListCdrc::transformed cdpn (without unconsumpt digits) = 33334444, unconsumed digit=
Refer to the exhibit. Which INVITE is sent to 10.10.100.123 as a result of this log?

A. 55698034.001 |12:20:50.922 |AppInfo |SIPTcp - wait_SdlSPISignal: Outgoing SIP TCP message to 10.10.100.123 on port 5060 index 41 [95992364,NET] INVITE sip:33334444@10.10.100.123:5060 SIP/2.0 Via: SIP/2.0/TCP 10.122.200.50:5060;branch=z9hG4bK268d6e4e48f3ae From: "1000" ;tag=32412716~41f7 To: Date: Thu, 01 Apr 2021 17:20:50 GMT Call-ID: 99878a80-66100f2-265e57-67071d0a@10.122.200.50 Supported: timer,resource-priority,replaces Min-SE: 1800 - User-Agent: Cisco-CUCM12.0 -

B. 55698034.001 |12:20:50.922 |AppInfo |SIPTcp - wait_SdlSPISignal: Outgoing SIP TCP message to 10.10.100.123 on port 5060 index 41 [95992364,NET] INVITE sip:33334444@10.10.100.123:5060 SIP/2.0 Via: SIP/2.0/TCP 10.122.200.50:5060;branch=z9hG4bK268d6e4e48f3ae From: "11112222" ;tag=32412716~41f7 To: Date: Thu, 01 Apr 2021 17:20:50 GMT Call-ID: 99878a80-66100f2-265e57-67071d0a@10.122.200.50 Supported: timer,resource-priority,replaces Min-SE: 1800 - User-Agent: Cisco-CUCM12.0 -

C. 55698034.001 |12:20:50.922 |AppInfo |SIPTcp - wait_SdlSPISignal: Outgoing SIP TCP message to 10.10.100.123 on port 5060 index 41 [95992364,NET] INVITE sip:11112222@10.10.100.123:5060 SIP/2.0 Via: SIP/2.0/TCP 10.122.200.50:5060;branch=z9hG4bK268d6e4e48f3ae From: "1000" ;tag=32412716~41f7 To: Date: Thu, 01 Apr 2021 17:20:50 GMT Call-ID: 99878a80-66100f2-265e57-67071d0a@10.122.200.50 Supported: timer,resource-priority,replaces Min-SE: 1800 - User-Agent: Cisco-CUCM12.0 -

D. 55698034.001 |12:20:50.922 |AppInfo |SIPTcp - wait_SdlSPISignal: Outgoing SIP TCP message to 10.10.100.123 on port 5060 index 41 [95992364,NET] INVITE sip:11112222@10.10.100.123:5060 SIP/2.0 Via: SIP/2.0/TCP 10.122.200.50:5060;branch=z9hG4bK268d6e4e48f3ae From: "11112222" ;tag=32412716~41f7 To: Date: Thu, 01 Apr 2021 17:20:50 GMT Call-ID: 99878a80-66100f2-265e57-67071d0a@10.122.200.50 Supported: timer,resource-priority,replaces Min-SE: 1800 - User-Agent: Cisco-CUCM12.0

Suggested Answer: *C*
Official Answer: C

Community vote distribution
A (100%)

FCBear Highly Voted 2 years, 11 months ago
Correct answer is A. From the log it shows that called party is transformed from 11112222 to 33334444.I agree on the transformation but... why A and not B? There is no info about the calling party number (from header) so both A or B can be correcti guess since the FROM in option B is 11112222 doesn't make sense, since you will not be calling yourself, so we assume the caller is 1000 instead of 11112222.
upvoted 7 times

FCBear Highly Voted 2 years, 11 months ago
Correct answer is A. From the log it shows that called party is transformed from 11112222 to 33334444.
upvoted 6 times

FrankPic 2 years, 7 months ago
I agree on the transformation but... why A and not B? There is no info about the calling party number (from header) so both A or B can be correct
upvoted 1 times

FrankPic 2 years, 7 months ago
I agree on the transformation but... why A and not B? There is no info about the calling party number (from header) so both A or B can be correcti guess since the FROM in option B is 11112222 doesn't make sense, since you will not be calling yourself, so we assume the caller is 1000 instead of 11112222.
upvoted 1 times

hany20006 1 year, 10 months ago
i guess since the FROM in option B is 11112222 doesn't make sense, since you will not be calling yourself, so we assume the caller is 1000 instead of 11112222.
upvoted 3 times

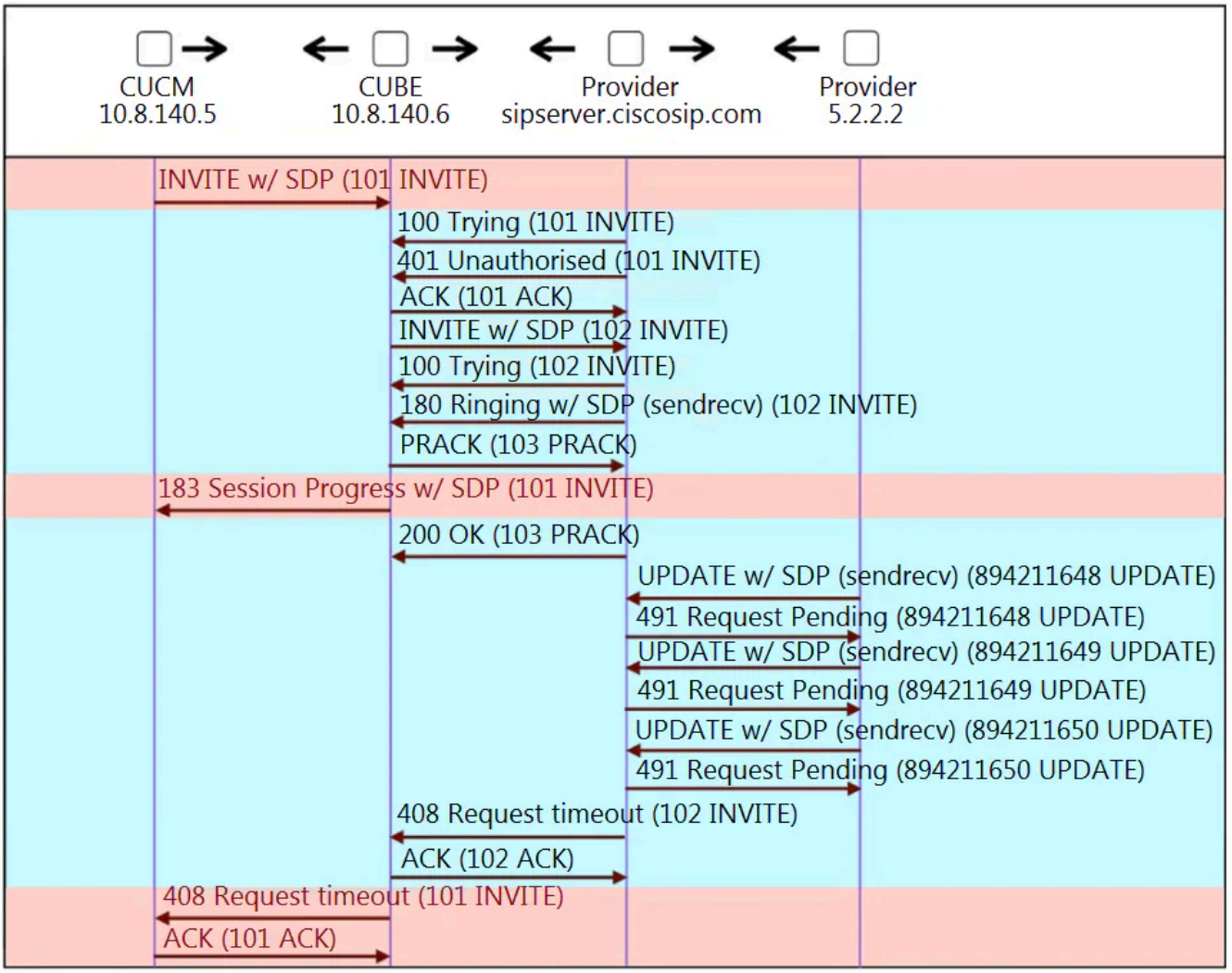
b3532e4 Most Recent 1 year, 7 months ago

Correct answer is A.

upvoted 2 times

Question #67 *Topic 1*

Refer to the exhibit. A Cisco Unified Border Element continues to send 180/183 with the required: 100rel header to Cisco UCM, and the call eventually disconnects. How is the issue resolved?



A. Disable "SIP Rel1XX Options" and "Early Offer Support" on the SIP Profile Configuration Page in Cisco UCM.

B. Enable "SIP Rel1XX Options" and "Early Offer Support" on the SIP Profile Configuration Page in Cisco UCM.

C. Disable "Send send-receive SDP in mid-call INVITE" on the SIP Profile Configuration Page in Cisco UCM.

D. Enable "Early Offer support for voice and video calls" on the SIP Profile Configuration Page in Cisco UCM.

Suggested Answer: *D*

Official Answer: D

Community vote distribution

B (100%)

bitdesaia Highly Voted 2 years, 4 months ago

https://www.cisco.com/c/en/us/support/docs/voice/session-initiation-protocol-sip/116086-configure-cube-cucm-sip-00.pdf Symptom 2: CUBE continues to send 180/183 with the Require: 100rel header to CUCM. This issue usually occurs when CUCM does not support reliable response. In order to resolve this issue, enable Rel1xx on CUCM.

upvoted 7 times

MaxG Most Recent 2 years ago

Should be B, per Cisco documentation under "Troubleshooting": https://www.cisco.com/c/en/us/support/docs/voice/session-initiation-protocol-sip/116086-configure-cube-cucm-sip-00.html

upvoted 3 times

Question #68 *Topic 1*

Refer to the exhibit. An engineer is troubleshooting an issue with the caller not hearing a PSTN announcement before the SIP call has completed setup. How must the engineer resolve this issue using the reliable provisional response of the SIP?

```
SIP/2.0 180 Ringing
Via: SIP/2.0/UDP 10.1.60.105:5060;branch=z9hG4bK721ed5d4
From: "1001" <sip:1001@10.88.247.229>;tag=6cfa89726ac700b569ec133a-7e6cd8aa
To: <sip:2005@10.88.247.229>;tag=47B5F70-438
Date: Fri, 19 Apr 2019 12:13:40 GMT
Call-ID: 6cfa8972-6ac7002b-5af19a5c-0de23108@10.1.60.105
CSeq: 101 INVITE
Require: 100rel
RSeq: 3344
Allow: INVITE, OPTIONS, BYE, CANCEL, ACK, PRACK, UPDATE, REFER, SUBSCRIBE, NOTIFY,
INFO, REGISTER
Remote-Party-ID: <sip:2005@10.88.247.229>;party=called;screen=yes;privacy=off
Contact: <sip:2005@10.88.247.229:5060>
Server: Cisco-SIPGateway/IOS-16.6.2
Content-Length: 0
```

A. voice service voip sip no rel1xx

B. sip-ua disable-early-media 180

C. voice service voip sip rel1xx require 100rel

D. voice service voip sip send 180 sdp

Suggested Answer: *C*

Official Answer: C

Community vote distribution

C (100%)

domangez Most Recent 1 year, 10 months ago

voice service voip sip rel1xx require 100rel / supported 100rel / disable

upvoted 1 times

MaxG 2 years, 6 months ago

https://www.cisco.com/c/en/us/support/docs/voice/session-initiation-protocol-sip/116086-configure-cube-cucm-sip-00.html

upvoted 1 times

Question #69 *Topic 1*

An IP Telephony administrator is deploying IP phones. The administrator has an existing Cisco UCME router with several SCCP & SIP phones registered. The administrator receives a request for a new SIP phone with MAC address 1111.2222.3333 and directory number 2050 to be added in the Cisco UCME. Which two configurations should be added in CME to support this request? (Choose two.)

```
voice register pool 1
 id mac 1111.2222.3333
 type 8941
 number 2 dn 1
```

```
ephone 1
 mac-address 1111.2222.3333
 type 8941
 button 1:2
```

```
ephone-dn 2
 number 2050
```

```
voice register dn 2
 number 2050
```

```
voice register pool 1
 id mac 1111.2222.3333
 type 8941
 number 1 dn 2
```

A.

B.

C.

D.

E.

Suggested Answer: *D, E*

Official Answer: DE

Community vote distribution

DE (100%)

MaxG Most Recent 2 years ago

Per SIP Implementation Guide: https://www.cisco.com/c/en/us/support/docs/voice-unified-communications/unified-communications-manager-express/99946-cme-sip-guide.html

upvoted 2 times

Question #70 *Topic 1*

An engineer must implement call restriction to toll-free numbers using a class of restriction in a branch Cisco UCME. In which two places is the corlist incoming or cor incoming command configured? (Choose two.)

A. “voice register pool ” configuration mode

B. “ephone-dn ” configuration mode

C. “dial-peer cor custom ” configuration mode

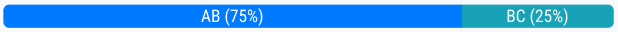
D. “voice register global ” configuration mode

E. “telephony-service ” configuration mode

Suggested Answer: *A, B*

Official Answer: AB

Community vote distribution



f77380c Most Recent 4 months, 2 weeks ago

A for SIP Phones, B for SCCP phones

upvoted 1 times

Piji 1 year, 6 months ago

"ephone-dn" configuration mode The corlist incoming command is applied at the ephone-dn level to control incoming call restrictions for specific directory numbers. "dial-peer cor custom" configuration mode The cor incoming command is used under dial-peer configuration mode to enforce call restriction rules based on COR lists.

upvoted 1 times

FrankPic 2 years ago

https://www.cisco.com/c/en/us/support/docs/voice/call-routing-dial-plans/42720-configuring-cor.html Voice Register Pool are device definition for SIP phones on CME Ephone-DN are device definition for SCCP phones on CME

upvoted 1 times

FrankPic 2 years ago

https://www.cisco.com/c/en/us/support/docs/voice/call-routing-dial-plans/42720-configuring-cor.html Voice Register Pool are device definition for SIP phones on CME Ephone-DN are device definition for SCCP phones on CME

Correct Answers are A+B

upvoted 2 times

FrankPic 2 years ago

https://www.cisco.com/c/en/us/support/docs/voice/call-routing-dial-plans/42720-configuring-cor.html Voice Register Pool are device definition for SIP phones on CME Ephone-DN are device definition for SCCP phones on CME Correct Answers are A+Bsorry not 100% correct: ephone-DN is directory number definition while voice register-pool define SIP endpoint...

upvoted 2 times

FrankPic 2 years ago

sorry not 100% correct: ephone-DN is directory number definition while voice register-pool define SIP endpoint...

upvoted 2 times

MaxG 2 years, 6 months ago

cor (voice register pool) To configure a class of restriction (COR) on the VoIP dial peers associated with directory numbers, use the cor command in voice register pool configuration mode. Also see Configuring Class of Restrictions (COR): https://www.cisco.com/c/en/us/support/docs/voice/call-routing-dial-plans/42720-configuring-cor.html

upvoted 1 times

Question #71 *Topic 1*

A customer has multisite deployments with a globalized dial plan. The customer wants to route PSTN calls via the gateway assigned to each site. Which two actions will fulfill the requirement? (Choose two.)

A. Create one global route list for PSTN calls that points to one global PSTN route group.

B. Create a route group which has all the gateways and associate it to the device pool of every site.

C. Assign one route group as a local route group in the device pool of the corresponding site.

D. Create one route group for each site and one global route list for PSTN calls that point to the local route group.

E. Create a hunt group and assign it to each side route pattern.

Suggested Answer: *A, C*

Official Answer: AC

Community vote distribution

CD (100%)

FCBear 2 years, 5 months ago

Answer should be C and D.

upvoted 4 times

Afaik 2 years, 6 months ago

Answer should be C and D, D basically describes on how to access the configuration made in C.D is not 100% correct: RL should point to Standard Local Route Group and not Local Route Group...

upvoted 4 times

Afaik 2 years, 6 months ago

Selected Answer: CD

Answer should be C and D, D basically describes on how to access the configuration made in C.

upvoted 3 times

FrankPic Most Recent 2 years ago

D is not 100% correct: RL should point to Standard Local Route Group and not Local Route Group...

upvoted 2 times

Question #72 *Topic 1*

An engineer is troubleshooting local ringback on a Cisco SIP gateway. The gateway is not ignoring the SIP 180 response with SDP from the service provider, and the far end device is sending the 180 with SDP to play ringback from the IP address specified in the SDP. Which configuration change must be made on the gateway to resolve the issue?

A. Router(config-sip-ua)# no disable-early-media 180

B. Router(conf-voi-serv)# no disable-early-media 180

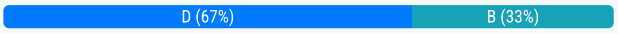
C. Router(conf-voi-serv)# disable-early-media 180

D. Router(config-sip-ua)# disable-early-media 180

Suggested Answer: *D*

Official Answer: D

Community vote distribution



vishaljhaveri Most Recent 4 months, 3 weeks ago

D. Router(config-sip-ua)# disable-early-media 180. By default, Cisco gateways treat a SIP 180 response containing SDP similarly to a 183 Session Progress response. It assumes the far end will provide early media (like a custom ringback or announcement) and cuts through the audio path to play whatever is being sent from the IP address in the SDP. The disable-early-media 180 command, configured under sip-ua configuration mode, explicitly instructs the gateway to ignore the presence of SDP in a 180 nresponse and provide a local ringback tone instead.

upvoted 2 times

blackozkr 8 months, 3 weeks ago

disable-early-media 180 command does not exist on conf-voi-serv

upvoted 1 times

v1nhthanh 1 year, 3 months ago

You configure on voice service voip

upvoted 1 times

decdca7 1 year, 5 months ago

Sorry I meant B

upvoted 1 times

MaxG 2 years ago

To specify which call treatment, early media or local ringback, is provided for 180 responses with Session Description Protocol (SDP), use the disable-early-media 180 command in sip-ua configuration mode. To enable early media cut-through for 180 messages with SDP, use the no form of this command. The following example disables early media cut-through for SIP 180 responses with SDP: Router(config-sip-ua)# disable-early-media 180 https://www.cisco.com/en/US/docs/ios/12_3t/voice/command/reference/vrht_d2_ps5207_TSD_Products_Command_Reference_Chapter.html#wp1452642

upvoted 1 times

Question #73 *Topic 1*

An engineer must configure a Cisco UCM hunt list so that calls to users in a line group are routed to the first idle user and then the next. Which distribution algorithm must be configured to accomplish this task?

A. broadcast

B. top down

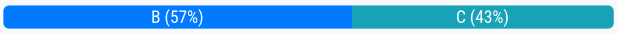
C. longest idle time

D. circular

Suggested Answer: *C*

Official Answer: C

Community vote distribution



blude Highly Voted 2 years, 9 months ago

This one should be B, the word "first" make the different , is goign to be the first one not the one longest-idle. Top Down—If you choose this distribution algorithm, Cisco CallManager distributes a call to idle or available members starting from the first idle or available member of a line group to the last idle or available member.

upvoted 7 times

blude Highly Voted 2 years, 9 months ago

This one should be B, the word "first" make the different , is goign to be the first one not the one longest-idle. Top Down—If you choose this distribution algorithm, Cisco CallManager distributes a call to idle or available members starting from the first idle or available member of a line group to the last idle or available member.First idle, just means that person will have the longest idle time, so they will be routed to first. Answer is C

upvoted 8 times

johnathanm 2 years, 7 months ago

First idle, just means that person will have the longest idle time, so they will be routed to first. Answer is C

upvoted 2 times

f77380c Most Recent 4 months, 2 weeks ago

C is wrong, the word idle is a trick. Top down will choose 1st idle on the list, while Longest idle time will choose longest but not first. Answer is B.

upvoted 1 times

decdca7 1 year, 5 months ago

Not the longest idle, first idle, which is the first one available in a list.

upvoted 1 times

Piji 1 year, 7 months ago

The question ask "First Idle User" and then "Next" which means "Next Ideal User", because it uses word "idle", can't be "Top Down" as that is always follow the list and can't find the "idle".

upvoted 1 times

f1ab921 1 year, 10 months ago

Surprised to find the answer is actually B. Here's a copy and paste from Cisco url at the bottom: TOP DOWN—If you choose this distribution algorithm, Cisco CallManager distributes a call to idle or available members STARTING FROM THE FIRST IDLE or available member of a line group to the last idle or available member. LONGEST IDLE—If you choose this distribution algorithm, Cisco CallManager only distributes a call to idle members, STARTING FROM THE LONGEST IDLE member to the least-idle member of a line group.
https://www.cisco.com/en/US/docs/voice_ip_comm/cucm/admin/4_0_1/ccmcfg/b03lngrp.html#wp1027384

upvoted 3 times

7ArchAngel7 2 years, 3 months ago

Answer is C. Longest Idle Time. Top Down will just go through the line group member list as its configured from the Top-Down, regardless of whether the user is idle or not.

upvoted 1 times

MaxG 3 years ago

Longest Idle Time - If you choose this distribution algorithm, Cisco CallManager only distributes a call to idle members, starting from the longest-idle member to the least-idle member of a line group.

https://www.cisco.com/en/US/docs/voice_ip_comm/cucm/admin/4_0_1/ccmcfg/b03lngrp.html#wp1027384

upvoted 1 times

Question #74 *Topic 1*

An engineer has temporarily disabled toll fraud prevention for SIP line calls on a Cisco CME12.6x and must enforce security and toll fraud prevention for the SIP line side on Cisco Unified CME. Which configuration must be used to start this process?

A. voice service voip enable ip address trust list

B. voice service voip ip address trusted list

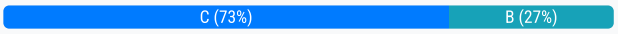
C. voice service voip ip address trusted authenticate

D. voice service voip enable ip address trust authentication

Suggested Answer: *B*

Official Answer: B

Community vote distribution



john_doe_9999 Highly Voted 2 years, 7 months ago

The answer is C. The command to disable is "no ip address trusted authenticate" under voice service voip config. To enable you would use "ip address trusted authenticate" under voice service voip config. If you simply type "ip address trusted list" you get put in to that sub-config mode and do not re-enable it.

upvoted 6 times

b3532e4 Most Recent 1 year, 10 months ago

Unified CME 12.6 or a later version for SIP line calls. Unified CME 8.1 or a later version for secure trunk calls. SUMMARY STEPS enable configure terminal voice service voip ip address trusted authenticate ip-address trusted call-block cause code end show ip address trusted list

upvoted 2 times

johnathanm 2 years, 7 months ago

"In Unified CME, IP address trusted authentication is enabled by default. When IP address trusted authenticate is enabled, Unified CME accepts incoming VoIP (SIP/H.323) calls only if the remote IP address of an incoming VoIP call is successfully validated from the system IP address trusted list ." So, Answer is B, as you only need to start with these commands

upvoted 3 times

MaxG 3 years ago

Configure IP Address Trusted Authentication for Incoming VoIP Calls. SUMMARY STEPS 1. enable 2. configure terminal 3. voice service voip 4. ip address trusted authenticate 5. ip-address trusted call-block cause code 6. end

https://www.cisco.com/c/en/us/td/docs/voice_ip_comm/cucme/admin/configuration/manual/cmeadm/cmetoll.html#task_151A890CF831459986B1DDE915C6FEA8My only thought is in Unified CME IP address trusted authentication is enabled by default. so to start the process is B. voice service voip ip address trusted list"An engineer has temporarily disabled toll fraud prevention for SIP line calls on a Cisco CME12.6x"

upvoted 2 times

MaxG 3 years ago

Selected Answer: C

Configure IP Address Trusted Authentication for Incoming VoIP Calls. SUMMARY STEPS 1. enable 2. configure terminal 3. voice service voip 4. ip address trusted authenticate 5. ip-address trusted call-block cause code 6. end

https://www.cisco.com/c/en/us/td/docs/voice_ip_comm/cucme/admin/configuration/manual/cmeadm/cmetoll.html#task_151A890CF831459986B1DDE915C6FEA8

upvoted 2 times

johnathanm 2 years, 7 months ago

My only thought is in Unified CME IP address trusted authentication is enabled by default. so to start the process is B. voice service voip ip address trusted list"An engineer has temporarily disabled toll fraud prevention for SIP line calls on a Cisco CME12.6x"

upvoted 1 times

johnathanm 2 years, 7 months ago

My only thought is in Unified CME IP address trusted authentication is enabled by default. so to start the process is B. voice service voip ip address trusted list

upvoted 1 times

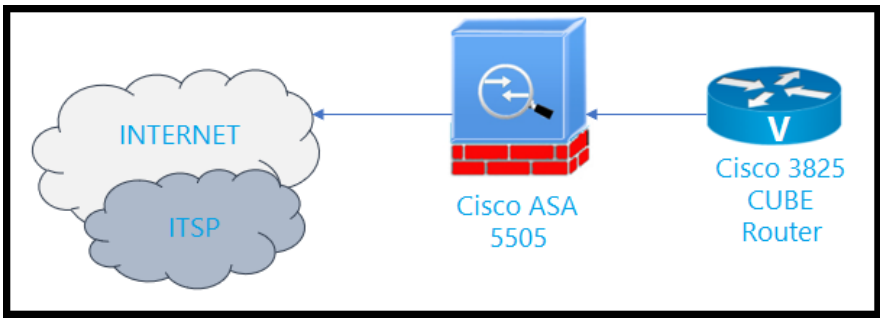
john_doe_9999 2 years, 7 months ago

"An engineer has temporarily disabled toll fraud prevention for SIP line calls on a Cisco CME12.6x"

upvoted 1 times

Question #75 *Topic 1*

Refer to the exhibit. An administrator is troubleshooting a problem in which some outbound calls from an internal network to the Internet telephony service provider are not getting connected, but some others connect successfully. The firewall team found that some call attempts on port 5060 came from an unrecognized IP that has not been defined in the firewall rule. What should the administrator configure in the Cisco Unified Border Element to fix this issue?



A. use of port 5061 for SIP secure

B. access list allowing the firewall IP

C. bind signaling and media to the loopback interface

D. ip prefix-list to filter the unwanted IP address

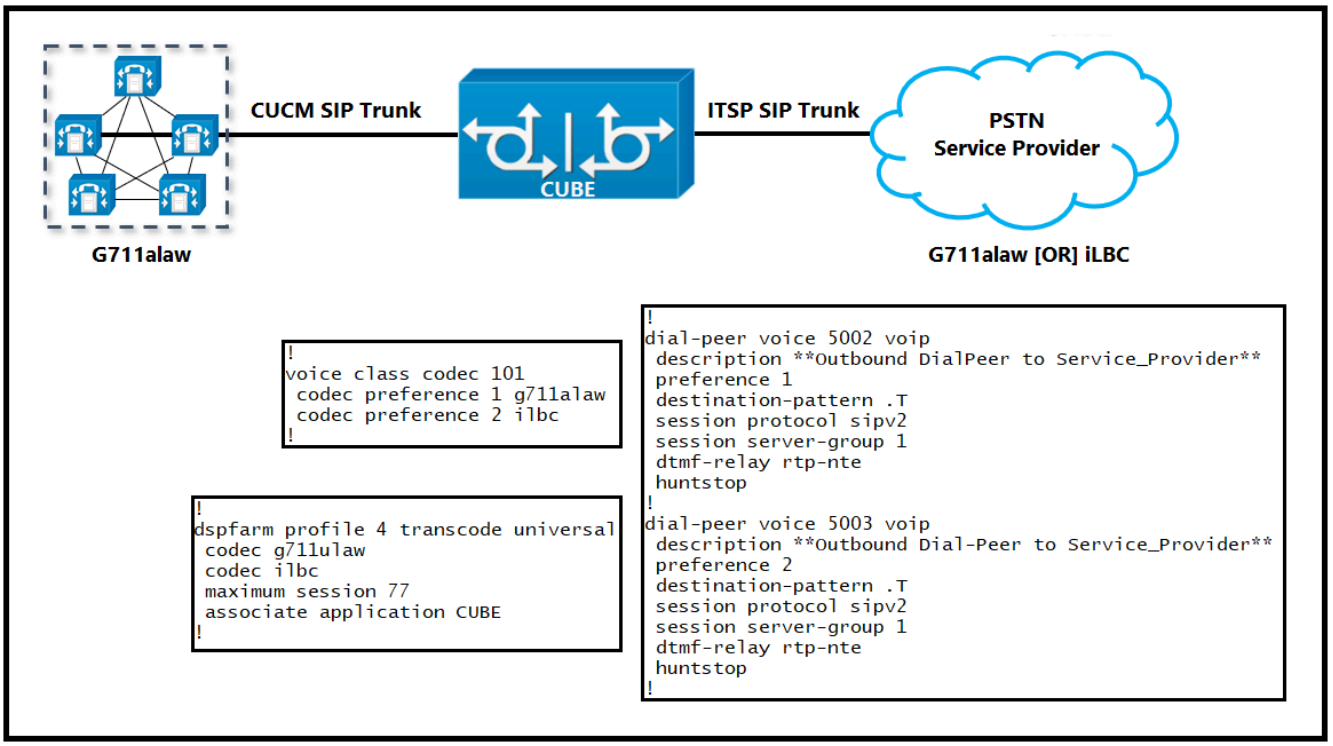
Suggested Answer: *C*
Official Answer: C

Community vote distribution
C (100%)

john_doe_9999 Most Recent 2 years, 1 month ago
The answer is C
upvoted 3 times

Question #76 *Topic 1*

Refer to the exhibit. Outbound calls to the service provider cause intermittent errors due to a codec mismatch. The internal network sends early offer SDP that contains only G.711 A-law. The service provider reports that some destinations support only G.711 A-law while others support only iLBC. The service provider also allows only 20 active calls at a time. Which configuration allows successful media negotiation for all calls using outbound dial peers 5002 and 5003?



A. dial-peer voice 5002 voip codec g711alaw ilbc ! dial-peer voice 5003 voip codec g711alaw ilbc

B. dial-peer voice 5002 voip voice-class codec 101 offer-all ! dial-peer voice 5003 voip voice-class codec 101 offer-all

C. dial-peer voice 5002 voip voice-class codec 101 ! dial-peer voice 5003 voip voice-class codec 101

D. dial-peer voice 5002 voip codec g711alaw ! dial-peer voice 5003 voip codec ilbc

Suggested Answer: *B*

Official Answer: B

Community vote distribution

B (88%) C (12%)

CCNPVoice Most Recent 8 months, 3 weeks ago

I thought it was B; however, the "offer all" is not necessary when there are codecs explicitly configured. Adding "offer-all" at the end of the command is optional. Furthermore, the "Offer all" will kill the preference list instantly. Answer should be C

upvoted 1 times

john_doe_9999 2 years, 1 month ago

Without being able to see the inbound dial-peer I think B is the best answer here.

upvoted 3 times

MaxG 2 years, 6 months ago

All codecs are offered by default only for delayed offer calls. For Early Offer you either need to configure a codec preference list with a specific list of codec or "offer all". https://community.cisco.com/t5/unified-communications-infrastructure/cube-quot-offer-all-quot-for-voice-class-codec/td-p/4686998

upvoted 4 times

Question #77 *Topic 1*

Refer to the exhibit. An engineer is troubleshooting an issue where inbound calls to Cisco UCM with early media fail to establish. While investigating the issue, the engineer finds that Cisco UCM is set to require a PRACK, but the Cisco Unified Border Element is not sending it. Which command is causing this issue?

```
!
dial-peer voice 100 voip
 description Outbound to CUCM
 translation-profile outgoing CUCM
 session protocol sipv2
 session target ipv4:192.168.100.200
 voice-class sip transport switch udp tcp
 voice-class sip conn-reuse
 voice-class sip rel1xx disable
 voice-class sip session refresh
 voice-class sip midcall-signaling block
 voice-class sip early-media update block
 dtmf-relay rtp-nte
 codec g711ulaw
 no vad
!
```

A. voice-class midcall-signaling block

B. voice-class sip rel1xx disable

C. voice-class sip early-media update block

D. voice-class sip conn-reuse

Suggested Answer: *B*
Official Answer: B

b3532e4 Most Recent 1 year, 10 months ago

PRACK to Cisco UCM is: B. voice-class sip rel1xx disable Explanation: The command voice-class sip rel1xx disable disables the handling of reliable provisional responses (like 183 Session Progress) that require PRACK. When this is disabled, the CUBE will not send PRACK messages, which is likely causing the failure in early media call establishment with Cisco UCM, as UCM expects PRACK in response to provisional responses. To fix the issue, you would need to either remove or modify the voice-class sip rel1xx disable command so that the CUBE properly handles reliable provisional responses and sends the required PRACK messages.

upvoted 2 times

Question #78 *Topic 1*

Refer to the exhibit. The Cisco Unified Border Element receives an INVITE matching inbound dial peer 5002. The outbound dial peer supports only iLBC, and a Local Transcoding Interface is allocated. Based on the configuration and SDP from the INVITE message, which codec is chosen by Cisco Unified Border Element for the inbound call leg?

```
voice class codec 100
 codec preference 1 g711alaw
 codec preference 2 g729r8
 codec preference 3 g729br8
 codec preference 4 g711ulaw
!
dial-peer voice 5002 voip
 session protocol sipv2
 session server-group 1
 incoming called-number 5...
 voice-class codec 100
 dtmf-relay rtp-nte
 no vad

m=audio 30104 RTP/AVP 0 9 124 116 18 101
a=rtpmap:0 PCMU/8000
a=rtpmap:9 G722/8000
a=rtpmap:124 iSAC/16000
a=rtpmap:116 iLBC/8000
a=maxptime:20
a=fmtp:116 mode=20
a=rtpmap:18 G729/8000
a=fmtp:18 annexb=no
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-15
```

A. G.729r8

B. G.711 A-law

C. G.711 U-law

D. G.729br8

Suggested Answer: *C*

Official Answer: C

Community vote distribution

A (88%) C (12%)

john_doe_9999 Highly Voted 2 years, 7 months ago

After a lot of research on this one i'm going with A, G.729r8. The codec chosen will be the codec that is of highest preference in the voice-class code list that is also present in the SDP of the Invite. The invite has G.729 in it (no annex b). The spec for G.729r8 covers regular G.729 (no annex b) and G.729a.

upvoted 7 times

Musienkoalex Most Recent 7 months ago

Preference: For SIP-to-SIP calls, CUBE typically prioritizes the incoming INVITE's codec preference over its own local voice-class codec configuration.

upvoted 1 times

b3532e4 1 year, 9 months ago

B. G.711 A-law becouse

upvoted 1 times

Question #79 *Topic 1*

Due to a shortage of physical interfaces on a device, the administrator requires that a loopback for RTP is used. Which command is required when using a loopback interface for RTP?

A. voice-class sip early-offer forced

B. voice-class sip bind control source-interface Loopback0

C. voice-class sip bind media source-interface Loopback0

D. voice-class sip resource priority mode passthrough

Suggested Answer: *C*
Official Answer: C

1b9a51d Most Recent 2 years, 1 month ago

Configuring SIP Binding 6. Use one of the following commands to configure SIP binding: - voice-class sip bind media {source-address ipv4 ipv4-address | source-interface interface-id [ipv6-address ipv6-address]} in dial-peer configuration mode. https://www.cisco.com/c/en/us/td/docs/ios-xml/ios/voice/cube/configuration/cube-book/voi-sip-bind.html#GUID-89132FC8-B523-420B-A48F-0A5FC853E55A

upvoted 1 times

Question #80 *Topic 1*

An administrator is implementing a new dial-plan on Cisco Unified Border Element. The administrator must ensure that incoming dial-peers are matched based on the IP address from where the incoming request originates. Which dial-peer configuration should be applied to accomplish this requirement?

A. dial-peer voice 1 voip incoming uri to

B. dial-peer voice 1 voip incoming called-number

C. dial-peer voice 1 voip incoming uri via

D. dial-peer voice 1 voip incoming uri request

Suggested Answer: *C*

Official Answer: C

Community vote distribution



b3532e4 Most Recent 1 year, 7 months ago

Selected Answer: C

upvoted 1 times

MaxG 2 years ago

Matching Against IPv4 Address and VIA CUBE is configured to use incoming dial-peer 101 for incoming SIP calls from remote SIP endpoint having an IP address of 10.10.10.1 ---------- voice class uri 201 sip host ipv4:10.10.10.1 dial-peer voice 101 voip session protocol sipv2 incoming uri via 201 ---------- https://www.cisco.com/c/en/us/td/docs/ios-xml/ios/voice/cube/configuration/cube-book/voi-inbnd-dp-match-uri.html

upvoted 4 times

Question #81 *Topic 1*

When a third-party SIP Phone System is dialed inbound across a Cisco Unified Border Element, DTMF is failing. The third-party vendor accepts only out-of-band DTMF. Which configuration should be added to the outgoing dial peer to resolve this issue?

A. dtmf-relay rtp-nte

B. dtmf-relay cisco-rtp

C. dtmf-relay h245-signal

D. dtmf-relay sip-kpml

Suggested Answer: *C*
Official Answer: C

Community vote distribution
D (100%)

MaxG Highly Voted 2 years, 6 months ago

Selected Answer: D

We are talking about a SIP Phone System, so the answer cannot be C. "dtmf-relay sip-kpml" is the only option.

upvoted 6 times

MaxG Highly Voted 2 years, 6 months ago

We are talking about a SIP Phone System, so the answer cannot be C. "dtmf-relay sip-kpml" is the only option.You are correct! I would really be looking for dtmf-relay sip-notify here as its "Third Party" as well.

upvoted 7 times

729edae Most Recent 1 year, 9 months ago

You are correct! I would really be looking for dtmf-relay sip-notify here as its "Third Party" as well.

upvoted 2 times

Question #82 *Topic 1*

Refer to the exhibit. A call made through the Cisco Unified Border Element to pilot 2000 is failing. What is causing the call to fail?

```
!
dial-peer voice 10 voip
 description Inbound
 session protocol sipv2
 incoming called-number 2000
 dtmf-relay rtp-nte
 no vad
!
dial-peer voice 20 voip
 description Outbound
 destination-pattern 2.
 session protocol sipv2
 session target ipv4:192.168.100.101
 voice-class sip options-keepalive
 dtmf-relay rtp-nte
!

CUBE#show dial-peer voice summary
dial-peer hunt 0
              AD                                    PRE    PASS   SESS-SER-GRP\    OUT
TAG    TYPE   MIN    OPER PREFIX    DEST-PATTERN    FER    THRU   SESS-TARGET      STAT PORT    KEEPALIVE  VRF
10     voip   up     up                             0      syst                                    NA
20     voip   up     up             2.              0      syst   ipv4:192.168.100.101             busyout  NA
```

A. The Cisco Unified Border Element is not receiving a response to its OPTION keepalives.

B. The destination pattern is incorrect for the dialed number.

C. VAD was not disabled on the outgoing dial peer.

D. No codecs are configured on the dial peers.

Suggested Answer: *D*

Official Answer: D

Community vote distribution

A (67%) B (27%)

timmyz Highly Voted 2 years ago

If you look at the output in the question under the KEEPALIVE field it shows "busyout" that means the receiving end is not UP and the call fails. It really has nothing to do with the dailpeer. it's just configured for SIP-keepalives. check out this blog post https://www.uccollabing.com/configure-cisco-cube-sip-options-ping/ the answer is A

upvoted 5 times

Gary1968 3 months, 1 week ago

destination-pattern 2. will only match 2 digits. Therefore, won't be a match for 2000.Correction A.

upvoted 1 times

Gary1968 Most Recent 1 month, 3 weeks ago

Correction A.

upvoted 1 times

b3532e4 1 year, 7 months ago

I have 2 problems with this Q my opinion, A correct and B

upvoted 1 times

729edae 1 year, 9 months ago

LOL codecs have no bearing on this. The answer is A. The pattern matters not. There is no response to the SIP OPTIONS being sent, meaning the trunk to the D/E is down. Wrong pattern or no, the status would not be busyout unless the D/E was responding to the keepalives. The CUBE/CME/VGR here considers it down. Also, no information is given about the requirement for SDP so the default codec of G729r8 is being used. The answer is A.

upvoted 4 times

john_doe_9999 2 years, 1 month ago

Selected Answer: D

The pattern 2. does match 2000 since it isn't terminated with a $. It's matching on "20" alone. g729 would be used as the codec since it isn't defined, but we aren't getting that far as the dial peer is in busyout due to failing keepalives. Can't negotiate or fail on codec if the dial peer isn't even able to communicate. D should be the correct answer here.

upvoted 1 times

rsl4u 2 years, 1 month ago

I found this document: Troubleshoot Busyout Dial Peers on CUBE or IOS Voice Gateway https://www.cisco.com/c/en/us/support/docs/unified-communications/unified-border-element/217860-troubleshooting-busyout-dial-peers-on-cu.html In the bottom of the document you see 2 steps to fix the problem. Step 2 stated: Remove the 'sip options-keepalive' on the dial-peer. dial-peer voice 1000 voip no voice-class sip options-keepalive profile 1 I think answer A is the right one

upvoted 1 times

johnathanm 2 years, 1 month ago

This is not correct. the "." is a single digit placeholder.

upvoted 1 times

johnathanm 2 years, 1 month ago

After looking into it and what John Doe said. It is A. Look this https://pbxbook.com/cisco/Dial_Peer_Config.pdf Variable-Length Matching will cause it to match the 2. This means it is hitting the correct dial peer, but is failing because the peer is busyout.

upvoted 4 times

john_doe_9999 2 years, 1 month ago

The pattern 2. does match 2000 since it isn't terminated with a $. It's matching on "20" alone. g729 would be used as the codec since it isn't defined, but we aren't getting that far as the dial peer is in busyout due to failing keepalives. Can't negotiate or fail on codec if the dial peer isn't even able to communicate. D should be the correct answer here.This is not correct. the "." is a single digit placeholder.Yes it is. 2000 will match pattern "2." because it isn't terminated. Here it is on my CUBE: dial-peer voice 555 voip destination-pattern 2. session protocol sipv2 session server-group 1 sh dialplan number 2000 tag = 555, destination-pattern = `2.'Correction, A

upvoted 1 times

johnathanm 2 years, 1 month ago

This is not correct. the "." is a single digit placeholder.Yes it is. 2000 will match pattern "2." because it isn't terminated. Here it is on my CUBE: dial-peer voice 555 voip destination-pattern 2. session protocol sipv2 session server-group 1 sh dialplan number 2000 tag = 555, destination-pattern = `2.'

upvoted 1 times

john_doe_9999 2 years, 1 month ago

Yes it is. 2000 will match pattern "2." because it isn't terminated. Here it is on my CUBE: dial-peer voice 555 voip destination-pattern 2. session protocol sipv2 session server-group 1 sh dialplan number 2000 tag = 555, destination-pattern = `2.'

upvoted 3 times

john_doe_9999 2 years, 1 month ago

Correction, A

upvoted 1 times

MaxG 2 years, 6 months ago

VoIP dial peers default to G.729 if no codec is specified, so D is NOT correct. Correct answer is B. Destination pattern is incorrect, as noted below by mmollura.

upvoted 3 times

stephen_b 2 years, 8 months ago

D. is correct.

upvoted 1 times

mmollura 2 years, 12 months ago

Destination pattern is incorrect. Should be 2... as one '.' is only 1 digit.

upvoted 2 times

Question #83 *Topic 1*

Refer to the exhibit. An engineer is troubleshooting a call-establishment problem between Cisco Unified Border Element and Cisco UCM. Which command set corrects the issue?



```
interface GigabitEthernet0/0/0
 description to CUCM
 ip address 10.10.150.1 255.255.255.0
 negotiation auto
!
interface GigabitEthernet0/0/1
 description to ITSP
 ip address 192.168.10.78 255.255.255.0
 negotiation auto
!
dial-peer voice 100 voip
 incoming called-number 8005532447
 session protocol sipv2
 codes g711ulaw
 voice-class sip bind control source-interface GigabitEthernet0/0/1
 voice-class sip bind media source-interface GigabitEthernet0/0/1
 dtmf-relay rtp-nte
!
dial-peer voice 200 voip
 destination-pattern 8005532447
 session target ipv4:192.168.10.100
 session protocol sipv2
 codec g711ulaw
 voice-class sip bind control source-interface GigabitEthernet0/0/1
 voice-class sip bind media source-interface GigabitEthernet0/0/1
 dtmf-relay rtp-nte
!
dial-peer voice 300 voip
 answer-address 8005532447
 session protocol sipv2
 codec g711ulaw
 voice-class sip bind control source-interface GigabitEthernet0/0/1
 voice-class sip bind media source-interface GigabitEthernet0/0/1
 dtmf-relay rtp-nte
```

```
Received:
INVITE sip:8005532447010.10.150.1:5060 SIP/2.0
Via: SIP/2.0/UDP 10.10.150.11:5060;branch-z9hG4bK1046d36216b0de
From: <sip:1001010.10.150.11>;tag-23125042-8a7bedal-fb5d-4d82-bdb6-4b07a7393aff-27428388
To: "CISCO SYSTEMS" <sip:8005532447010.10.150.1>;tag=D974B1S2=FAS
Date: Tue, 30 Mar 2021 22:14:00 GMT
Call-ID: C57C1746-90D511EB-826BBE69-C6943E02010.10.150.1
User-Agent: Cisco-CUCM11.5
Allow: INVITE, OPTIONS, INFO, BYE, CANCEL, ACK, PRACK, UPDATE, REFER, SUBSCRIBE, NOTIFY
CSeq: 103 INVITE
[..Omitted for brevity..]
Contact: <sip:1001010.10.150.11:5060>;
Content-Type: application/adp
Content-Length: 235

v=0
o=CiscoSystemsCCM-SIP 23125042 1 IN IP4 10.10.150.11
s=SIP Call
c=IN IP4 10.10.2.254
b=TIAS:64000
b=AS:64
t=0 0
m=audio 35023 RTP/AVP 0 101
a=ptime:20
a=rtpmap:0 PCMU/8000
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-15

  Calling Number=1001,(Calling Name-) (TON-Unknown, NPI-Unknown, Screening-User, Passed,
  Called Number-8005532447(TON-Unknown, NPI-Unknown),
  Calling Translated-FALSE, Subscriber Type Str-Unknown, FinalDestinationFlag-FALSE,
  Incoming Dial-peer-100, Progress Indication-NULL(0), Calling IE Present-TRUE,
```

A. SIP binding in SIP configuration mode: voice service voip sip bind control source-interface GigabitEthernet0/0/1 bind media source-interface GigabitEthernet0/0/1

B. SIP binding in dial-peer configuration mode: dial-peer voice 100 voip voice-class sip bind control source-interface GigabitEthernet0/0/0 voice-class sip bind media source-interface GigabitEthernet0/0/0

C. SIP binding in dial-peer configuration mode: dial-peer voice 300 voip voice-class sip bind control source-interface GigabitEthernet0/0/1 voice-class sip bind media source-interface GigabitEthernet0/0/1

D. SIP binding in SIP configuration mode: voice service voip sip bind control source-interface GigabitEthernet0/0/0 bind media source-interface GigabitEthernet0/0/0

Suggested Answer: *B*

Official Answer: B

Community vote distribution

B (100%)

FrankPic Most Recent 2 years, 1 month ago

Correct Answer: B https://www.cisco.com/c/en/us/support/docs/voice/call-routing-dial-plans/14074-in-dial-peer-match.html The router or gateway matches the information elements in the setup message with the dial peer attributes to select an inbound dial peer. The router or gateway matches these items in this order: 1. Called number (DNIS) with the incoming called-number command 2. Calling Number (ANI) with the answer-address command 3. Calling Number (ANI) with the destination-pattern command 4. Voice-port (associated with the incoming call setup request) with configured dial peer port (applicable for inbound POTS call legs) 5. If no match is found in the first four steps, then the default dial peer 0 (pid:0) command is used.

upvoted 2 times

john_doe_9999 2 years, 1 month ago

I'm thinking B is correct

upvoted 2 times

Question #84 *Topic 1*

A new deployment is using MVA for a specific user on the sales team, but the user is having issues when dialing DTMF. Which DTMF method must be configured in resolve the issue?

A. in-band

B. out-of-band

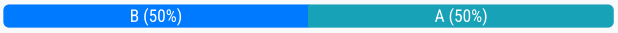
C. gateway

D. channel

Suggested Answer: *B*

Official Answer: B

Community vote distribution



f77380c 4 months, 1 week ago

Answer is B.

upvoted 1 times

Musienkoalex 7 months ago

Selected Answer: B

The following VOIP dial-peer will be used for all MVA calls.

dial-peer voice 33 voip destination-pattern 3000 session target ipv4:192.100.65.11 codec g722-64 << or voice-class codec containing codecs dtmf-relay h245-alphanumeric no vad

upvoted 2 times

Musienkoalex 7 months ago

The following VOIP dial-peer will be used for all MVA calls. dial-peer voice 33 voip destination-pattern 3000 session target ipv4:192.100.65.11 codec g722-64 << or voice-class codec containing codecs dtmf-relay h245-alphanumeric no vadWhere are you getting the specific dial-peer information, as it isn't recorded in the question?

upvoted 2 times

Gary1968 Most Recent 3 months, 1 week ago

Where are you getting the specific dial-peer information, as it isn't recorded in the question?

upvoted 1 times

729edae 1 year, 4 months ago

Why OOB? Option A (in-band) should be the correct choice. Configuring the gateway to use in-band DTMF (specifically RFC 2833/4733) ensures that the gateway can detect the in-band tones from the mobile phone and relay them to CUCM as RTP events. This is the most compatible and standard method for DTMF in a VoIP environment, especially for MVA scenarios involving the PSTN. Option B (out-of-band) would likely cause the issue to persist. If the gateway is set to out-of-band DTMF, it might expect DTMF to be sent as signaling messages (e.g., KPML, SIP INFO), which isn't how the mobile phone sends DTMF over the PSTN. This mismatch would prevent the gateway from correctly detecting the tones.

upvoted 3 times

Question #85 *Topic 1*

An administrator is working on an issue between the customer's Cisco Unified Border Element and the service provider. The provider only wants to see mid-call signaling from the Cisco Unified Border Element for fax calls. Which command must be configured on Cisco Unified Border Element?

A. midcall-signaling passthru

B. no update-callerid

C. midcall-signaling passthru media-change

D. midcall-signaling preserve-codec

Suggested Answer: *C*

Official Answer: C

Community vote distribution

C (100%)

729edae Most Recent 1 year, 4 months ago

Agreed, C.

upvoted 1 times

MaxG 2 years ago

Passthrough media change method optimizes or consumes mid-call, media-related signaling within the call. Mid-call signaling changes will be passed through only when bidirectional media like T.38 (aka fax) or video is added. The command midcall-signaling passthru media-change needs to be configured to enable passthrough media change. https://www.cisco.com/c/en/us/td/docs/ios-xml/ios/voice/cube/configuration/cube-book/voi-cube-midcall-reinvite.html#GUID-F5641A05-14F7-4396-B04B-5F1D4F527457

upvoted 2 times

Question #86 *Topic 1*

CollabCorp is a global company with two clusters, emea.collab.corp and apac.collab.corp. URI dialing is implemented and working in each cluster. The company configured routing between clusters to make inter-cluster calls via URI, but this is not working as expected. Which two configuration elements should be checked to resolve this issue? (Choose two.)

A. intercluster trunk

B. directory URI partition

C. SIP route pattern

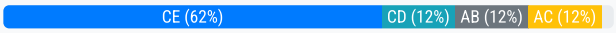
D. calling search space and partition

E. SIP trunk

Suggested Answer: *C, E*

Official Answer: CE

Community vote distribution



f77380c Most Recent 4 months, 2 weeks ago

Intracluster works - the only other factors to troubleshoot for intercluster is C & E.

upvoted 2 times

729edae 1 year, 4 months ago

SIP Route Pattern (C): This is the most likely culprit. The EMEA cluster needs a SIP route pattern to route *.apac.collab.corp to the intercluster trunk. If this is misconfigured, missing, or inaccessible, the call will fail. Intercluster Trunk (A): The ICT is the connection between the clusters. If it's down or misconfigured, the call can't be routed, even with a correct SIP route pattern. D. Calling Search Space and Partition: This is a secondary issue. The SIP route pattern itself is the more direct element to check. B. Directory URI Partition: This is relevant in the destination cluster, but intra-cluster URI dialing is working, so the Directory URI partitions are likely correct. Remember, the question explicitly states "URI dialing is implemented and working in each cluster". E. SIP Trunk: This is a distractor, as the intercluster trunk (option A) is the relevant SIP trunk for this scenario. There is no mention of ILS in the question. ONLY "Inter-cluster calls via URI"

upvoted 1 times

Piji 1 year, 6 months ago

CollabCorp is a global company with two clusters, emea.collab.corp and apac.collab.corp. URI dialing is implemented and working in each cluster. The company configured routing between clusters to make inter-cluster calls via URI, but this is not working as expected. Which two configuration elements should be checked to resolve this issue? (Choose two.) A. intercluster trunk B. directory URI partition C. SIP route pattern D. calling search space and partition E. SIP trunk

upvoted 1 times

FrankPic 2 years, 1 month ago

Correct Answers: CE When a user dials a learned directory URI that is registered to a remote cluster, Cisco Unified Communications Manager pulls the route string that is associated to that directory URI, matches that route string to a SIP route pattern, and routes the call to the outbound trunk that is specified by the SIP route pattern. In order for Cisco Unified Communications Manager to route calls to a route string, you must configure SIP route patterns that route the destination route strings to the next-hop clusters in your ILS network.

upvoted 4 times

john_doe_9999 2 years, 1 month ago

Going with C and D on this one. You have to have the correct SIP route patterns in place and have the privileges to reach them before anything else can be accomplished.

upvoted 1 times

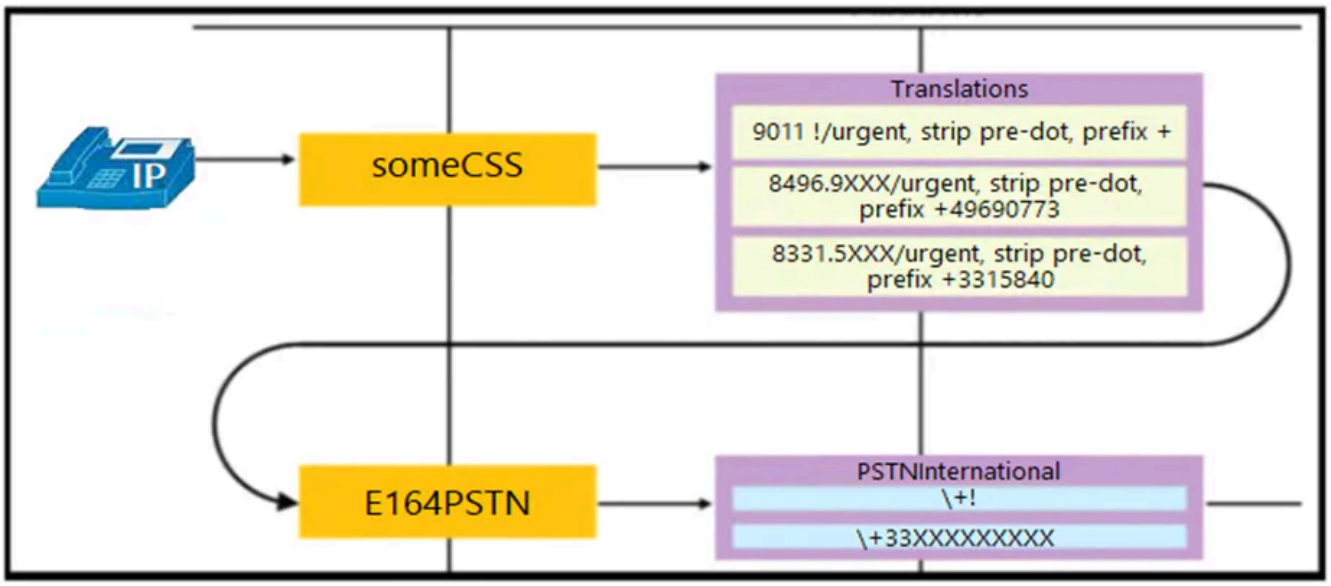
MaxG 2 years, 6 months ago

See Set Up URI Dialing: 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_0110011.html#CUCM_TK_SBE2D597_00

upvoted 3 times

Question #87 *Topic 1*

Refer to the exhibit. A user dials 84969010 and observes that the call is not routed immediately. The administrator notices that after matching the fixed-length translation pattern, the call hits the \+! pattern and waits for interdigit timeout. What should be configured to ensure that the call routes out immediately?



A. Allow Device Override on the route pattern

B. Do Not Wait For Interdigit Timeout On Subsequent Hops on the route pattern

C. Do Not Wait For Interdigit Timeout On Subsequent Hops on the translation pattern

D. Route Next Hop By Calling Party Number on the translation pattern

Suggested Answer: *C*
Official Answer: C

Community vote distribution
C (100%)

1b9a51d Most Recent 2 years, 1 month ago
the option Do Not Wait For Interdigit Timeout On Subsequent Hops on the translation pattern has to be set. If this option is set, then after matching the translation pattern, Unified CM will not wait for any further digits and will just match the translated called party number against the patterns identified by the CSS defined on the intermediate translation pattern.
https://www.cisco.com/c/en/us/td/docs/voice_ip_comm/cucm/srnd/collab12/collab12/dialplan.html#11439
upvoted 1 times

domangez 2 years, 4 months ago
"Do Not Wait For Interdigit Timeout On Subsequent Hops" option exist in Translation Pattern, but not in Route Pattern
upvoted 1 times

john_doe_9999 2 years, 7 months ago
C is correct
upvoted 1 times

Question #88 *Topic 1*

A customer is using a SIP trunk to route calls to ITSP. To decrease the possibility of downtime, the customer invested in a failover device. How does the customer ensure reachability to ITSP, so that if one device on ITSP fails, the calls will be routed to another device?

A. Enable SIP Option Ping on the SIP profile.

B. Monitor the link using network management tools, and if it fails, manually change the routing to another working device.

C. Enable ANAT on the SIP profile.

D. Enable transmit security status on the SIP security profile.

Suggested Answer: *A*

Official Answer: A

Community vote distribution

A (100%)

john_doe_9999 Most Recent 2 years, 1 month ago

A makes the most sense in a collaboration context

upvoted 2 times

Question #89 *Topic 1*

A customer routes PSTN calls to ITSP through a SIP trunk on Cisco UCM that forwards and receives calls to and from ITSP. ITSP is set to send an E.164 number when the customer's extension is four digits. Which action should be taken to route the incoming calls to four-digit extensions?

A. Configure a voice translation profile to map the E.164 number to four digits and assign it to the incoming dial-peer on Cisco Unified Border Element.

B. Set the Significant Digits to 8 on the SIP trunk.

C. Set the Significant Digits to 4 on the SIP trunk.

D. Configure a voice translation rule to map the E.164 number to four digits and assign it to the incoming dial-peer on Cisco Unified Border Element.

Suggested Answer: *C*

Official Answer: C

Community vote distribution

C (100%)

MaxG Most Recent 2 years ago

The Significant Digits feature instructs CUCM to analyze the configured number of digits (from right to left) of the called number for incoming calls received by a gateway or trunk. Setting the significant digits to 5 on a PSTN gateway instructs CUCM to ignore all but the last five digits of the called party number for routing incoming gateway or trunk calls. The Significant Digits feature is the easiest approach to convert incoming PSTN called numbers to an internal extension, but the setting affects all calls received from the gateway. https://www.ciscopress.com/articles/article.asp?p=1745737&seqNum=6

upvoted 3 times