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# 642-457

## Cisco

*Implementing Cisco Unified Communications Manager,  
Part 2 v8.0 (CIPT2 v8.0)*

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**QUESTION: 1**

Which two features require or may require configuring a SIP trunk? (Choose two.)

- A. SIP gateway
- B. Call Control Discovery between a Cisco Unified Communications Manager and Cisco Unified Communications Manager Express
- C. Cisco Device Mobility
- D. Cisco Unified Mobility
- E. registering a SIP phone

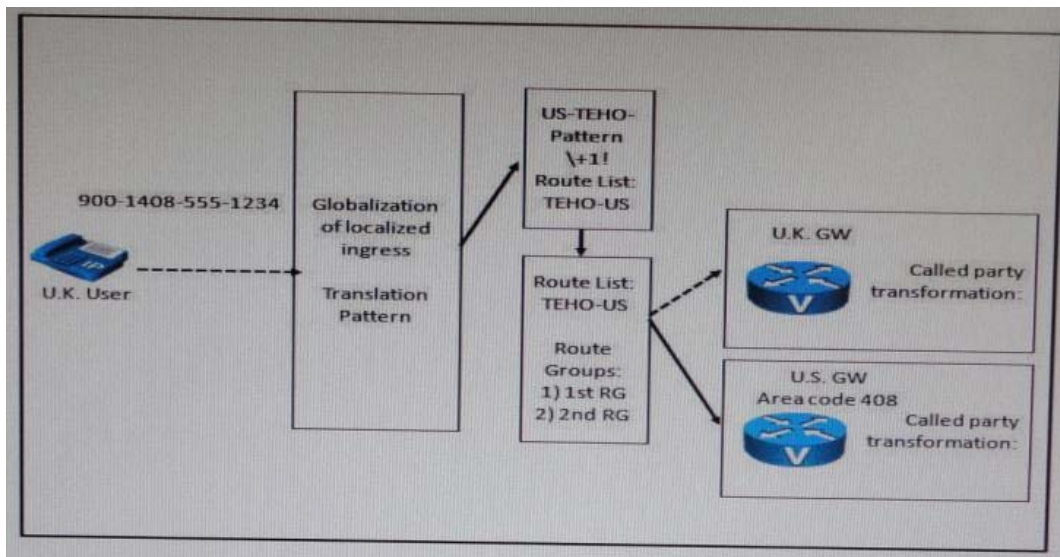
**Answer:** A, B

**Incorrect Answer:**

CDE All protocols require that either a signaling interface (trunk) or a gateway be created to accept and originate calls. Device mobility allows Cisco Unified Communications Manager to determine whether the phone is at its home location or at a roaming location. Cisco Unified Mobility gives users the ability to redirect incoming IP calls from Cisco Unified Communications Manager to different designated phones, such as cellular phones. Link: [http://www.cisco.com/en/US/docs/voice\\_ip\\_comm/cucm/admin/8\\_6\\_1/ccmsys/a08sip.html#wpxref77849](http://www.cisco.com/en/US/docs/voice_ip_comm/cucm/admin/8_6_1/ccmsys/a08sip.html#wpxref77849)

**QUESTION: 2**

Refer to the exhibit.



The exhibit shows centralized Cisco Unified Communications Manager configuration components for TEHO calls to U.S. area code J08 from the U.K. The PSTN access code for the U.K is 9 and 001 for international calls to the U.S. What should the TEHO-US route list configuration consist of?

- A. First route group should point only to the U.K. gateway. The second route group should point to the U.S. gateway.
- B. First route group should be only the local route group. The second route group should point to the U.S gateway.
- C. First route group should point only to the U.S. gateway. The second route group should be the local route group.
- D. The TEHO-US route list should contain only the local route group. The globalized configuration means that the appropriate gateway will be elected automatically.
- E. The \+! route pattern should point directly to the U.S gateway.

**Answer:** C

**Incorrect Answer:**

ABD The route group points to one or more gateways and can choose the gateways for call routing based on preference. The route group can serve as a trunk group by directing all calls to the primary device and then using the secondary devices when the primary is unavailable. One or more route lists can point to the same route group.  
 Link:[http://www.ciscosystems.com/en/US/docs/voice\\_ip\\_comm/cucm/admin/8\\_6\\_1/ccmsys/a08gw.html#wp1167274](http://www.ciscosystems.com/en/US/docs/voice_ip_comm/cucm/admin/8_6_1/ccmsys/a08gw.html#wp1167274)

**QUESTION: 3**

When Cisco Extension Mobility is implemented, how is the audio source for the MOH selected?

- A. The audio source that is configured at the user device profile is selected.
- B. The audio source that is configured at the home phone of the user is selected.
- C. The audio source that is configured at the physical phone used for the Cisco Extension Mobility login is selected.
- D. The audio source that is configured in the IP Voice Media Streaming parameters is selected.

**Answer:** A

**Incorrect Answer:**

BCD To specify the audio source that plays when a user initiates a hold action, choose an audio source from the User Hold MOH Audio Source drop-down list box from device profile configuration settings.  
 Link:[http://cisco.biz/en/US/docs/voice\\_ip\\_comm/cucmbe/admin/8\\_6\\_1/ccmcf/b06dvprf.html](http://cisco.biz/en/US/docs/voice_ip_comm/cucmbe/admin/8_6_1/ccmcf/b06dvprf.html)

**QUESTION: 4**

In what Cisco solution is Simple Network-Enabled Auto Provision technology used?

- A. Cisco Unified Gateway Duplication
- B. Cisco Unified CallManager Redundancy
- C. Cisco Unified SRST
- D. Cisco Unified Call Survivability

**Answer:** C

**Incorrect Answer:**

ABD When the system automatically detects a failure, Cisco Unified SRST uses Simple Network Auto Provisioning (SNAP) technology to auto-configure a branch office router to provide call processing for the Cisco Unified IP phones that are registered with the router

Link:

[http://www.cisco.com/en/US/docs/voice\\_ip\\_comm/cucme/admin/configuration/guide/cmestst.html](http://www.cisco.com/en/US/docs/voice_ip_comm/cucme/admin/configuration/guide/cmestst.html)

**QUESTION: 5**

Which method can be used to address variable-length dial plans?

- A. Overlap sending and receiving.
- B. Add a prefix for all calls that are longer than 10-digits long
- C. Use nested translation patterns to eliminate inter-digit timeout
- D. Use the @macro on the route pattern
- E. Use MGCP gateways, which support variable-length dial plans

**Answer:** A

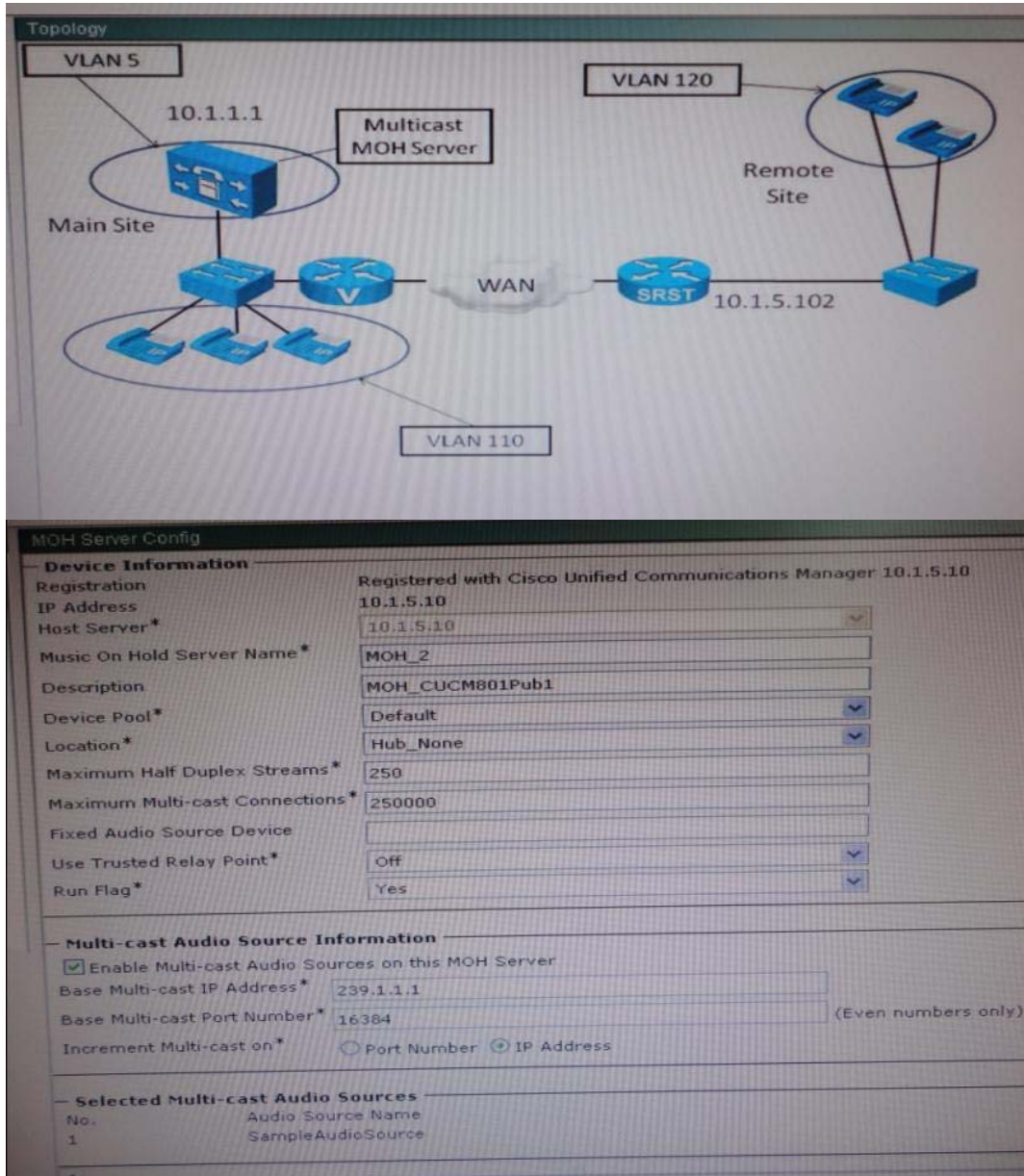
**Incorrect Answer:**

BCDE If the dial plan contains overlapping patterns, Cisco Unified Communications Manager does not route the call until the interdigit timer expires (even if it is possible to dial a sequence of digits to choose a current match). Check this check box to interrupt interdigit timing when Cisco Unified Communications Manager must route a call immediately. By default, the Urgent Priority check box displays as checked. Unless your dial plan contains overlapping patterns or variable length patterns that contain !, Cisco recommends that you do not uncheck the check box.

Link:[http://www.cisco.com/en/US/docs/voice\\_ip\\_comm/cucm/admin/8\\_6\\_1/ccmfeat/fsintrcm.html](http://www.cisco.com/en/US/docs/voice_ip_comm/cucm/admin/8_6_1/ccmfeat/fsintrcm.html)

**QUESTION: 6**

Refer to the exhibit.



To stream multicast MOH to the remote site across the WAN, what should the minimum value for the Max Hops be configured as?

- A. 1
- B. 2
- C. 3
- D. 4

**Answer:** D

**Incorrect Answer:**

ABC The Max Hops field in the Music On Hold (MOH) Server Configuration window indicates the maximum number of routers that an audio source is allowed to cross. If max hops is set to zero, the audio source must remain in its own subnet. If max hops is set to one, the audio source can cross up to one router to the next subnet. Cisco recommends setting max hops to two.

Link:

[http://www.cisco.com/en/US/docs/voice\\_ip\\_comm/cucm/admin/8\\_6\\_1/ccmfeat/fsmoh.html](http://www.cisco.com/en/US/docs/voice_ip_comm/cucm/admin/8_6_1/ccmfeat/fsmoh.html)

### QUESTION: 7

Refer to the exhibit.

Region Information			
Name: Default			
Region Relationships			
Region	Max Audio Bit Rate	Max Video Call Bit Rate (Includes Audio)	Link Loss Type
BK	8 kbps (G.729)	None	Lossy
Default	64 kbps (G.722, G.711)	None	Use System Default
SAF	8 kbps (G.729)	None	Use System Default
NOTE: Region(s) not displayed      Use System Default      Use System Default      Use System Default			
Modify Relationship to other Regions			
Regions	Max Audio Bit Rate	Max Video Call Bit Rate (Includes Audio)	Link Loss Type
BK Default SAF	Keep Current Setting	<input type="radio"/> Keep Current Setting <input type="radio"/> Use System Default <input checked="" type="radio"/> None <input type="radio"/> kbps	Lossy

Which statement about the configuration between the Default and BK regions is true?

- A. Calls between the two regions can use either 64 kbps or 8 kbps.
- B. Calls between the two regions can use only the G 729 codec
- C. Only 64 kbps will be used between the two regions because the link is "lossy"
- D. Both codecs can be used depending on the loss statistics of the link, when lossy conditions are high, the G.711 codec will be used.

**Answer: B**

### Incorrect Answer:

ACD

Due to bandwidth constraints to remote site ,use 8 kb/s (G.729) as the recommended setting between a new region and existing regions.

Link:[http://www.ciscosystems.com/en/US/docs/voice\\_ip\\_comm/cucm/admin/8\\_6\\_1/ccmcfg/b02regio.html](http://www.ciscosystems.com/en/US/docs/voice_ip_comm/cucm/admin/8_6_1/ccmcfg/b02regio.html)

### QUESTION: 8

Refer to the exhibit.



CCD Requesting Service

**CCD Requesting Service Info**

Name\*

Description

Route Partition

Learned Pattern Prefix

PSTN Prefix

Available SAF Trunks

▼ ▲

Selected SAF Trunks

Activated Feature

---

RTMT

Learned Pattern

Select a Node

Pattern	TimeStamp	Status	Protocol	AgentId	IP Address	ToDID	CUCMNodeId
300X	2010/05/07 14:52:06	Reachable	SIP	CID10.1.5.11	10.1.5.11(5060)	0.44228822	1

When the user of a phone registered to the Cisco Unified Communications Manager places a call to 3001 when the SAF network is down, what happens?

- A. The call fails.
- B. The call is rerouted to the PSTN with the constructed PSTN number as +442288223001
- C. The call is rerouted to the PSTN with the constructed PSTN number as 442288223001
- D. The call is rerouted to the PSTN with the constructed PSTN number as 0002288223001
- E. The call is rerouted to the PSTN with the constructed PSTN number as +0002288223001

**Answer:** A

**Incorrect Answer:** B, C, D

When the SAF forwarder loses network connection with its call-control entity, the SAF forwarder withdraws those learned patterns that were published by the call control entity. In this case, CCD requesting service marks those learned patterns as unreachable via IP, and the calls gets routed through the PSTN gateway.  
 Link:[http://www.cisco.com/en/US/docs/voice\\_ip\\_comm/cucm/admin/8\\_6\\_1/ccmfeat/fscallcontroldiscovery.html](http://www.cisco.com/en/US/docs/voice_ip_comm/cucm/admin/8_6_1/ccmfeat/fscallcontroldiscovery.html)

**QUESTION:** 9

Refer to the exhibit.



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and many others.. See complete list [Here](#)

