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QUESTION & ANSWER

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Exam : 350-030

Title : CCIE Voice Written

Version : DEMO

1.What are two advantages of multicast technologies? (Choose two.)

- A.Denial of service attacks in the network are prevented.
- B.They eliminate multipoint applications.
- C.They reduce traffic by delivering a separate stream of information to each corporate recipient or home environment, which reduces bandwidth.
- D.They control network traffic and reduce server and CPU load.
- E.They eliminate traffic redundancy.

Answer: DE

2.Which two descriptions apply to the Calling Search Space function in Cisco Unified Communications Manager? (Choose two.)

- A.It defines which numbers are available for a device to call.
- B.It provides a group of dial patterns to look through when making a call.
- C.Within a partition, each CSS has a directory number.
- D.It defines route patterns and directory numbers from which calls can be received.
- E.It defines the search for directory numbers in assigned partitions according to dial patterns.

Answer: AE

3.Which two statements apply to the partitions function in Cisco Unified Communications Manager? (Choose two.)

- A.When a directory number or route pattern is placed into a certain partition, this creates a rule for who can call that device or route list.
- B.A partition is a logical grouping of directory numbers and route patterns that have similar reachability characteristics.
- C.Calling Search Spaces are assigned to partitions.
- D.A directory number may appear in only one partition.
- E.Within the partition, each CSS has a directory number.

Answer: AB

4.Which three statements are true about multicast IGMP snooping? (Choose three.)

- A.When a host in a multicast group sends an IGMP leave message, only that port is deleted from the multicast group.
- B.An IP multicast stream to the IP host can be stopped only by an IGMP leave message.
- C.IGMP snooping does not examine or snoop Layer 3 information in packets that are sent between the hosts and the router.
- D.When the switch hears the IGMP host report from a host for a particular multicast group, the switch adds the host's port number to the associated multicast table entry.
- E.IGMP control messages are transmitted as IGMP multicast packets so that they can be distinguished from normal multicast data at Layer 2.
- F.A switch that is running IGMP snooping examines every multicast data packet to verify whether it contains any pertinent IGMP "must control" information.

Answer: ADF

5.Which three options are valid SCCP call states sent to an IP phone?

- A. Ring Off
- B. On Hook
- C. Call Transmit
- D. Connected
- E. Disconnected
- F. In Use Remotely

Answer: BDF

6. Which three statements are true about Cisco Discovery Protocol? (Choose three.)

- A. It is an excellent tool for displaying the interface status on switches.
- B. It works on top of the network layer and data link level.
- C. It uses a multicast packet with a destination MAC address of 01-00-CC-CC-CC.
- D. The platform TLV (TLV type 0x0006) contains an ASCII character string that describes the hardware platform of the device.
- E. You can use the CDP timer feature to change update times. The default is 60 seconds.
- F. It uses a broadcast packet with a destination MAC address of 01-00-CC-CC-CC.

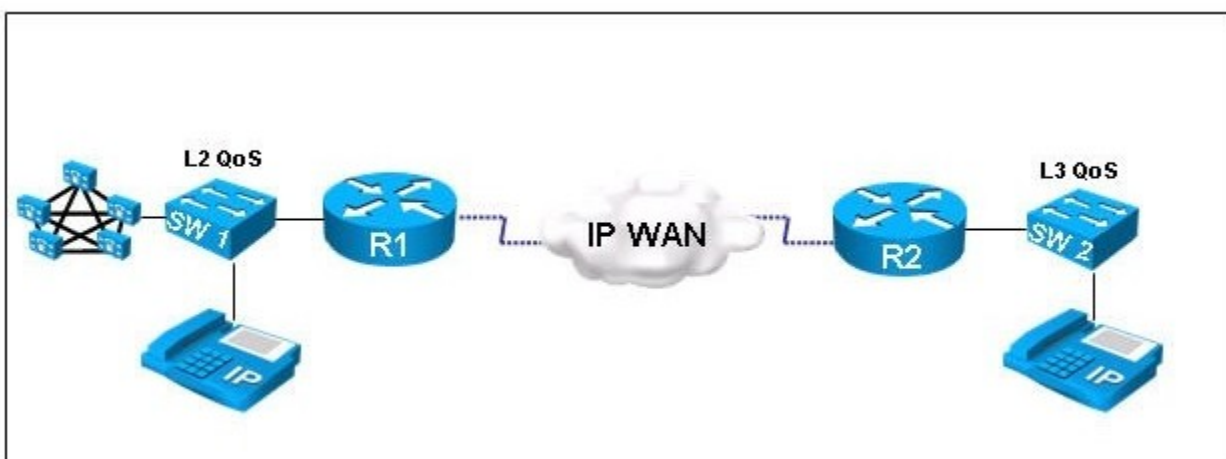
Answer: ADE

7. Which two of the following are functions of DHCP snooping? (Choose two.)

- A. relies on already discovered trusted and untrusted ports
- B. dynamic ARP inspection
- C. defines trusted and untrusted ports
- D. uses existing binding tables
- E. builds a binding table
- F. automatically builds ACLs

Answer: CE

8. Refer to the exhibit.



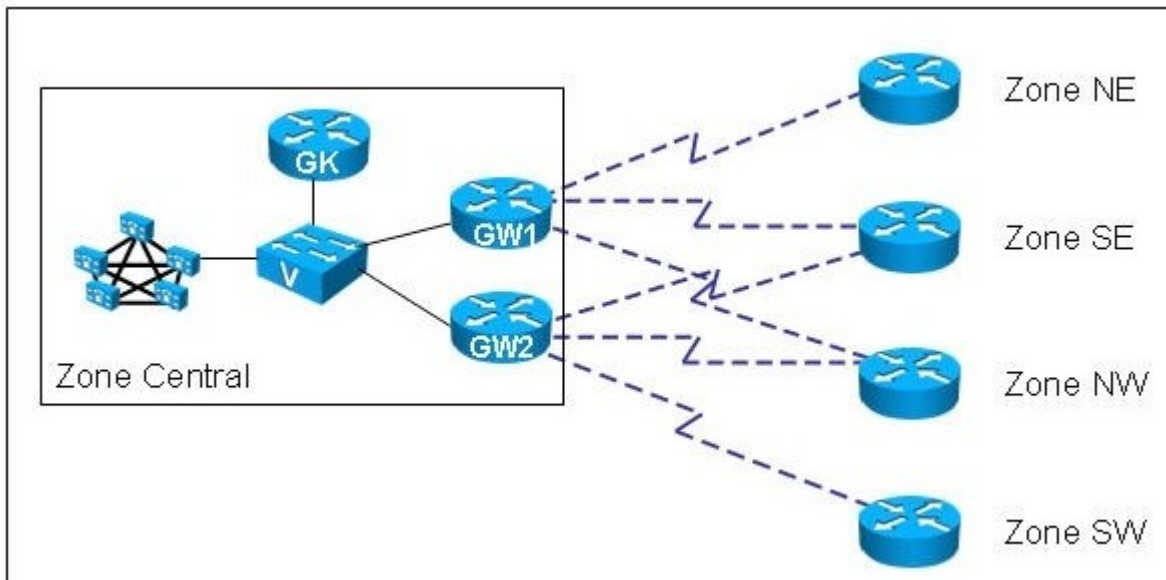
Traffic flows from the IP phone that is connected to SW1 to the IP phone on SW2. If the trust boundary has been extended to the IP phone on SW1, in what two places will traffic be marked and classified so that the proper QoS settings may be carried through the network? (Choose two.)

- A. IP phone attached to SW1

- B.SW1 ingress port
- C.R1 ingress port
- D.SW1 egress port
- E.R1 egress port

Answer: BC

9.Refer to the exhibit.



Which gatekeeper mechanism prevents the gatekeeper from using all the resources on either gateway 1 or gateway 2 when sending calls to zones SE and NW?

- A.bandwidth remote
- B.resource availability indicator
- C.bandwidth total
- D.bandwidth zone
- E.Irq immediate advance
- F.ras timeout brq

Answer: B

10.When implementing a Cisco Unified Communications Manager solution over an MPLS WAN, which two rules must be observed to prevent overrunning the priority queue? (Choose two.)

- A.RSVP will transparently pass application IDs from the customer network across the MPLS WAN.
- B.The media streams must be the same size in both directions.
- C.Only the connection to the MPLS WAN where the Cisco Unified Communications Manager resides must be enabled as a CE device.
- D.The media has to be symmetrically routed.
- E.If the CE is under corporate control, it may support either topology-aware or measurement- based CAC.

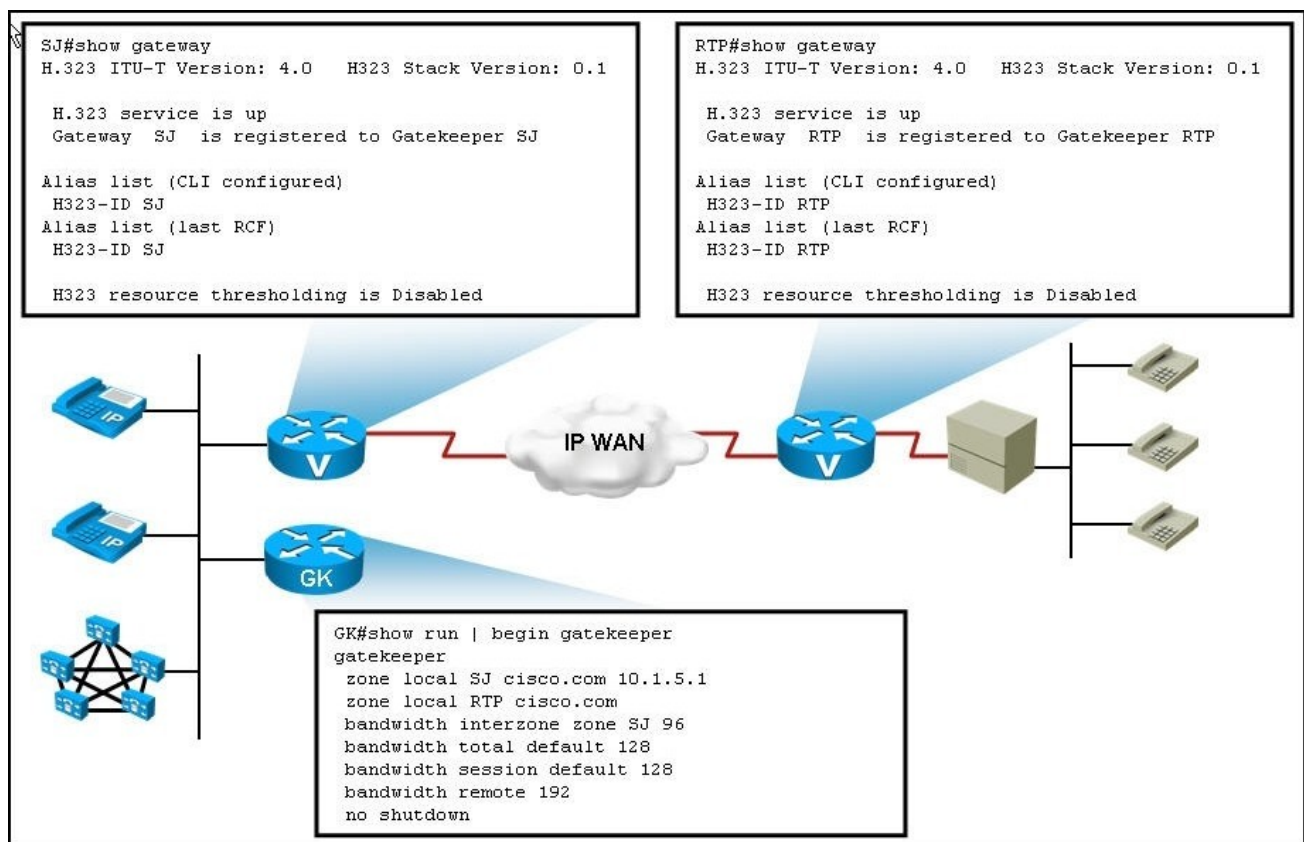
Answer: BD

11.How is fax pass-through traffic treated over IP WAN connections that use the G.729 codec?

- A.The fax traffic is demodulated and sent with VAD and echo cancellor disabled.
- B.When the TGW detects the CED tone from the fax machine that has been contacted, the TGW changes to the G.711 codec with echo cancellor and VAD disabled.
- C.When the OGW detects the CED tone from the fax machine that is making the call, the OGW is informed by the contacted device of the Cisco NSF features and switches to the G.711 codec with VAD disabled.
- D.The contacting fax machine sends a TCF message to the contacted fax machine and waits for a CFR message. When the CFR message is received, the fax tones sent by the contacting fax machine cause the OGW to send an NSF message to the TGW, instructing it to switch to the
- E.711 codec with echo cancellor and VAD disabled.

Answer: B

12.Refer to the exhibit.



How many simultaneous G.729 calls can be established between sites SJ and RTP?

- A.4
- B.5
- C.6
- D.8
- E.12

Answer: C

13.Company Alpha has a central office and a branch office that utilize a central call processing topology. Calls between the two sites are using the G.729 codec; calls within each site are using the G.711 codec.

To conference an existing call between two phones at the central site with a phone at the remote office, which two of the following are possible solutions? (Choose two.)

- A.a software conference bridge that is configured in Cisco Unified Communications Manager
- B.a software conference bridge that is configured in Cisco Unified Communications Manager and a HW transcoder
- C.a hardware conference bridge
- D.a hardware transcoder and a hardware conference bridge
- E.No extra configurations required--phones automatically negotiate using the lowest common denominator codec (G.729)

Answer: BC

14/Refer to the exhibit.

The screenshot shows the Cisco Unity Express Editor interface. The left pane displays a tree view of call flow elements. The right pane shows the script for 'aa_sample1.aef'. The script includes the following steps:

- Start
- Accept (contact: --Triggering Contact--)
- /* Initialize Prompts */
- Set menuPrompt = SP[AA/AAMainMenu]
- Set extnPrompt = SP[AA/AAEnterExtn]
- Set namePrompt = SP[AA/AANameDial]
- /* Play Alternate (Emergency) ... */
- Call Subflow -- checkAltGreet.aef
- /* Play Welcome Prompt without ... */
- Play Prompt (contact: --Triggering Contact--, prompt: welcomePrompt)
- MainMenu:
- Create Container Prompt (output prompt: prompt)
- /* Clear Prefix */
- Set prefixPrompt = P[]
- Menu (contact: --Triggering Contact--, prompt: prompt)
- DialByExtn
- DialByName
- Operator
- Timeout
- Unsuccessful
- /* We could not recognize ... */
- Sorry:
- Play Prompt (contact: --Triggering Contact--, prompt: prefixPrompt + SP[AA/ASorry])
- Call Redirect (contact: --Triggering Contact--, extension: operExtn)
- If (attempts < MaxRetry) Then
- /* We're not able to transfer ... */
- Play Prompt (contact: --Triggering Contact--, prompt: prefixPrompt + DP[1000])
- End

At the bottom, a table shows the values for the prompts:

Name	Type	Value	Attrib
welco...	Prompt	P[AA/welcome....	Para
extr\fer	String		
user	User	null	

You have been asked to edit the sample auto attendant script so that callers are prompted to press 1 for sales, 2 for service, or 3 for the directory. If callers select 3, they should hear the existing menu choices to dial by extension, dial by name, or transfer to the operator. What steps can you take to create this nested

menu?

- A. Drag a new Menu step from the palette and drop it on the Start step. Drag the existing Menu step and drop it on Output 3 of the new Menu.
- B. Drag a new Menu step from the palette and drop it on the existing Menu step. This will make the existing Menu subordinate to the new Menu.
- C. Drag a new Menu step from the palette and drop it on the existing Menu step. Drag the existing Menu step and drop it on Output 3 of the new Menu.
- D. Delete the existing Menu. Drag a new Menu step from the palette and drop it on the Set prefixPrompt=P[] step. Recreate the existing directory menu as the third option of the new Menu step.

Answer: C

15. Which two of these are possible reasons why a JTAPI subsystem might have the status PARTIAL_SERVICE? (Choose two.)

- A. Cisco Unified Contact Center is not able to resolve the host name of Cisco Unified Communications Manager.
- B. A referenced CTI Route Point is not associated with the JTAPI user.
- C. The JTAPI user password is not correct.
- D. There is an error in one of the scripts being loaded.
- E. The CTI Manager service is not running on Cisco Unified Communications Manager.

Answer: BD

16. Which three of these are mandatory sub-commands of the call-manager-fallback command and will help an IP phone register to an IOS router in SRST mode? (Choose three.)

- A. access-code
- B. dialplan-pattern
- C. ip source-address
- D. keepalive
- E. max-dn
- F. max-ephones

Answer: CEF

17. Refer to the exhibit.

```
May  9 02:13:06.370:
Send:
  SIP/2.0 415 Unsupported media type
Via: SIP/2.0/UDP 10.0.1.3:54475
From: sip:7300001@10.0.1.3
To: <sip:7900001@10.0.2.3;user=phone>;tag=9B9374-18BE
Date: Tue, 09 May 1993 02:13:06 UTC
Call-ID: 7B8D93F2-6EE5001E-0-9BCB54@10.0.1.3
Server: Cisco VoIP Gateway/ IOS 12.x/ SIP enabled
Content-Type: application/sdp
CSeq: 100 INVITE
Content-Length: 442
```

You are debugging a problem on a SIP network and have run the debug ccsip messages command. One of the messages returned is shown in the exhibit. What information will the server return to the caller?

- A.the acceptable media type
- B.a list of acceptable media types
- C.a list of acceptable formats
- D.a correct directory number
- E.an acceptable language code

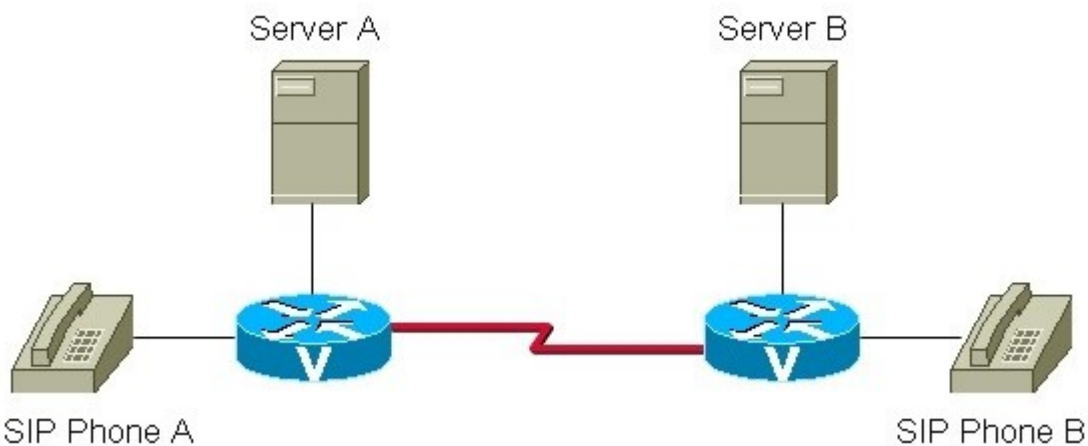
Answer: C

18.Which type of SIP responses would indicate that a server encountered an error in attempting to complete a SIP request?

- A.1xx
- B.3xx
- C.4xx
- D.5xx
- E.6xx

Answer: D

19.To hide its identity when initiating calls, SIP Phone B requests that Server B place its calls for it.



What kind of device is Server B?

- A.proxy
- B.redirect
- C.registrar
- D.user agent client
- E.user agent server

Answer: A

20.Which of the following three messages could be sent by the UAC in response to the 180 Ringing?
(Choose three.)

- A.PRACK
- B.ACK
- C.BYE
- D.CANCEL
- E.INVITE

Answer: ABD

