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問題集

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Exam : **920-432**

Title : Communication Server 1000
RIs.5.0-BCM RIs.4.0
Multi-site

Version : DEMO

1. What may be a cause of echo on a VoIP network?

- A. the VoIP products on the network
- B. the CODEC used on the VoIP network
- C. poor-quality headphones used on the VoIP network
- D. MCDN network features configured on the VoIP network

Answer: C

2. All systems on centralized voicemail must use what kind of dialing plan?

- A. CDP
- B. FNP
- C. ESN
- D. UDP

Answer: A

3. Click on the Exhibit button.

Based on the exhibit showing the H.323 settings, what is causing problems with the Communication Server (CS) 1000 - Business Communications Manager (BCM) integration?

H323 GW Settings

Primary gatekeeper IP address: 47.104.36.76

Alternate gatekeeper IP address: 0.0.0.0

Primary Network Connect Server IP address: 47.104.36.76

Primary Network Connect Server Port number: 16500 Range: 1024 to 65535

Alternate Network Connect Server IP address: 0.0.0.0

Alternate Network Connect Server Port number: 16500 Range: 1024 to 65535

Primary Network Connect Server timeout: 10 Range: 1 to 30

Telephony Resources

Modules

Bus	Prog Type	Actual Type	Dip Sw	State	Devices	Low	High	Total	Busy
0	N/A	IP Trunks	N/A	N/A	Lines		1	60	N/A
1	N/A	IP & App Sets	N/A	Enabled	Sets	N/A	N/A	8	0

Disable Enable

Details for Module: 0

Routing Table | IP Trunk Settings | **H323 Settings** | H323 Media Parameters | SIP Settings | SIP Media Parameters | SIP URI Map

Telephony Settings

Fallback to circuit-switched: Disabled

Gateway protocol: CSE

Gatekeeper digits: []

Gatekeeper wildcard:

Configuration

Call signaling: Gatekeeper Resolved

Call signaling port: 1720

Enable H245 tunnelling:

RAS port: 0

Primary Gatekeeper IP: 47.104.36.80

Registration TTL (s): 60

Backup Gatekeeper(s): 0.0.0.0

Gatekeeper TTL (s): 0

Alias names: Name:BCM-1

Modify...

Status: Attempting to discover gatekeeper at 47.104.36.80

- A. An alternate gatekeeper has not been defined.
- B. The RAS port has not been defined on the BCM.
- C. The Primary Gatekeeper IP address does not match.
- D. The Primary Network Connect Server Port number does not match the Call signaling port.

Answer: C

4. When should you change the RTP over UDP port range configuration on a Business Communications Manager (BCM)?

- A. when you are configuring a CS 1000 and BCM VoIP network integration
- B. when only absolutely necessary in instances where port configurations are causing conflicts
- C. when you have multiple BCM systems in a CS 1000 and BCM VoIP network integration
- D. when you are configuring a multisite BCM VoIP network integration

Answer: B

5. Click on the Exhibit button.

The Message Wait Indication is not working on a Business Communications Manager remote site.

Based on the exhibit showing Centralized Voice Messaging, what is the issue?

Center	External Number ▲	Message Waiting Indication String	Message Waiting Cancellation String
1	2000	2001#	AN^0#
2		AN^1#	AN^0#
3		AN^1#	AN^0#
4		AN^1#	AN^0#
5		AN^1#	AN^0#

Mailbox Number*: 2000

Mailbox Class: Regular User [Class Details](#)

Language: English(American)

Location Name: CP-204

Mailbox File System Volume ID: 103

Linked to external Directory: Not linked [Link...](#)

A. The Link to the external directory is not active.

- B. The External Number does not contain the proper access code.
- C. The Message Waiting Cancellation String has not been configured.
- D. The Message Wait Indication String must use the mailbox number assigned by CallPilot.

Answer: D

6. Click on the Exhibit button.

The remote Business Communications Manager is having difficulty getting its voicemail to work.

Based on the exhibit showing Capabilities for DN 4000, what is the problem?

Details for DN: 4000

Properties	Capabilities	SWCA, Call Group	Preferences	Button Programming Table	Button Programming	User Speed Dial
Handsfree	Auto		HF answerback	<input checked="" type="checkbox"/>		
Pickup group			DND on Busy	<input type="checkbox"/>	Allow redirect	<input type="checkbox"/>
Page zone	1		Paging	<input checked="" type="checkbox"/>	Redirect ring	<input checked="" type="checkbox"/>
Direct dial	1		Auto hold for incoming page	<input type="checkbox"/>	Receive short tones	<input type="checkbox"/>
Intrusion protection level	None		Priority call	<input type="checkbox"/>	Silent monitor supervisor	<input type="checkbox"/>
			Auto hold	<input checked="" type="checkbox"/>		

- A. Redirect ring is enabled.
- B. Priority call is not enabled.
- C. Allow redirect is not enabled.
- D. Pickup group has not been defined.

Answer: C

7. Click on the Exhibit button.

There is an error with the SIP settings of a Communication Server 1000 - Business Communications Manager integrated network.

Based on the information in the exhibit, what is the error?

The screenshot shows a configuration window titled "SIP Server Settings". It contains the following settings:

- Mode: Redirect (dropdown menu)
- UDP transport enabled: (unchecked)
- UDP port: 5060 (text input)
- UDP maximum transmission unit (MTU): 1500 (text input)
- TCP transport enabled: (checked)
- TCP port: 5060 (text input)
- TCP maximum transmission unit (MTU): 1500 (text input)

- A. The TCP MTU is set too low.
- B. UDP transport is not enabled.
- C. UDP and TCP ports are the same port.
- D. The SIP server mode should be set to Direct.

Answer: B

8. How do jitter buffers help enhance Quality of Experience (QoE)?

- A. Jitter buffers transmit voice frames at a fixed rate, enhancing the QoE.
- B. Jitter buffers assign packets to the appropriate queue, allowing for the packets to be played immediately.
- C. Jitter buffers can be configured to discard packets that are low quality thereby playing only high quality packets.
- D. Jitter buffers hold arriving packets in a buffer long enough to allow the slowest packets to arrive, enabling the packets to be played in the correct sequence.

Answer: D

9. What happens if a Business Communications Manager (BCM) is not configured for fallback and the IP call quality is below threshold?

- A. The IP call fails.
- B. The IP call is rerouted to a PSTN line pool.

- C. The IP call proceeds with low-quality transmission.
- D. The IP call is held until call quality is above threshold.

Answer: C

10. Which components are the two main components within SIP architecture?

- A. the INVITE and the REGISTER
- B. the SIP user agent and the SIP network server
- C. the User Agent Client and the User Agent Server
- D. the SIP proxy server and the SIP redirect server

Answer: B