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Exam : 920-432

Title : Communication Server 1000
Rls.5.0-BCM Rls.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 Gatekeeper digits

Gateway protocol Gatekeeper wildcard ☐

Configuration

Call signaling Call signaling port

Enable H245 tunnelling ☐ RAS port

Primary Gatekeeper IP Registration TTL (s)

Backup Gatekeeper(s) Gatekeeper TTL (s)

Alias names

Status

- 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?

Task Navigation Panel
Configuration Administration
Welcome
System
Administrator Access
Resources
Telephony
Global Settings
Sets
Lines
Active Physical Lines
Active VoIP Lines
Target Lines
Inactive Lines
All Lines
Loops
Scheduled Services
Dialing Plan
Ring Groups
Call Security
Hospitality
Hunt Groups
Call Detail Recording
Data Services
Applications
Voice Messaging/Contact Center
LANCOTE

Voice Messaging / Contact Center
Centralized Voice Messaging
Voice Message Centers

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#

Local Voice Messaging / Contact Center
Launch CallPilot Manager

Mailbox
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			
Capabilities			
Handsfree	Auto	HF answerback	☒
Pickup group		DND on Busy	☐
Page zone	1	Paging	☒
Direct dial	1	Auto hold for incoming page	☐
Intrusion protection level	None	Priority call	☐
		Auto hold	☒
		Allow redirect	☐
		Redirect ring	☒
		Receive short tones	☐
		Silent monitor supervisor	☐

- 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 'SIP Server Settings' window with the following configuration:

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

- 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