Internetwork Expert
Voice Workbook Volume II
Lab 1

Difficulty Rating (10 highest): 6

Lab Overview:

This lab scenario is a mock lab exam designed to simulate the conditions of Cisco Systems' CCIE Voice Lab exam. This lab should be completed within 8 hours. The only resource that should be used while configuring this lab is Cisco's documentation set. This documentation is available in both DVD format, and online at http://www.cisco.com/go/support.

Lab Instructions:

Prior to starting, ensure that the initial configuration scripts for this lab have been applied. For a current copy of these scripts, see the Internetwork Expert members site at http://members.internetworkexpert.com. The initial configurations for all routers and switches include IP addressing and routing configuration.

Refer to the lab diagram available under your member's account for information on the topology and IP addressing. In addition to the lab diagram, please download and reference the *Table Reference Guide* while completing the lab. Both the diagram and reference guide are available under the Voice Volume II section of your member's account.

The lab could be completed using either soft-phones (e.g. Cisco IP Communicator or IP Blue's VTGO) or by using Cisco 7960, 61, 62, and 65 IP phones. Refer to your rack rental user's guide for detailed instructions on connecting the remote soft-phones to the rack.

Grading:

This practice lab consists of various sections totaling 79 points. A score of 59 points is required to achieve a passing score. A section must work 100% with the requirements given in order to be awarded the points for that section. No partial credit is awarded. If a section has multiple possible solutions, choose the solution that best meets the requirements.

Point Values:

The point values for each section are as follows:

Section	Point Value
Infrastructure Configuration	9
Station Devices	7
Trunks & Gateways	9
Call Routing	16
CAC & Codecs	3
Media Resources	9
Applications & Services	4
High Availability	3
QoS	6
Voice Mail	9
CUCCX	4
Total:	79

1. Infrastructure Configuration

1.1. VLAN Assignment

- Configure the respective on-site switches to support Cisco IP phones
- Make sure you provide support for future PCs to be connected to the Cisco IP Phones internal switch port
- Use **switchport mode trunk** command on the Catalyst 3550 and 3560 switchports connected to the Cisco IP Phones
- Eliminate the STP forwarding delay on ports connecting the IP Phones

3 Points

1.2. DHCP

- Configure the routers at all three sites to allocate IP addresses via DHCP per the following requirements:

    - Create DHCP pools for voice VLANs at the HQ, BR1 and BR2 sites
    - Configure the DHCP pools to allocate IP addresses from the respective /24 subnet using the diagram
    - Allocate the respective router IP address as a default gateway
    - Set the TFTP server IP address to the CallManager Publisher

- Allocate IP addresses with the last octet in ranges “50-100” and “200-254” only

3 Points

1.3. NTP

- Configure R1 to act as NTP master in stratum 2 and R2, R3 as NTP clients to R1 in order to synchronize their clocks
- Refer to **Table 1** for information on time-zones to be configured on the routers
- Authenticate the NTP adjacencies using the password value of “CISCO”

3 Points

2. Station Devices

2.1. Cisco IP Phones

- The company is planning to use a centralized call processing deployment model with Cisco CallManager being the main call-processing agent
- Using the following table as your reference, register the IP phones at all three sites with the CallManager cluster and assign directory numbers

Phone	Extension	User	DID Number
HQ IP Ph1	1001	hquser1	7752011001
HQ IP Ph2	1002	hquser2	7752011002
HQ IP Ph3	1003	hquser3	7752011003
BR1 IP Ph1	2001		3123012001
BR2 IP Ph1	3001		21313001
BR2 IP Ph2	3002		21313002

- Additionally, create users in the corporate directory and associate them with their phones
- Use a default password value of “cisco” and a default PIN “12345” for all users

3 Points

2.2. Cisco SIP Phones

- Br1 Phone 2 should be converted from a SCCP phone to SIP Phone (**If the phone is already running SIP Firmware, first auto register the phone to the CUCM system, which will convert it to SCCP**)
- Ensure only this phone is converted to SIP
- Register the phone with the following parameters:

Phone	Extension	User	DID Number
BR1 IP Ph2	2002		3123012001

3 Points

2.3. Presentation Settings

- Configure the CallManager so that IP Phones display correct date and time according to their time zone settings
- The IP phones should present their full number (10 digits for HQ and BR1, and 8 digits for BR2) when calling an external (PSTN) destinations unless a task explicitly instructs not to do so
- Make sure BR1 IP Phone LCD Displays have the following appearance:

Display layout for BR1 IP Phone 1:

HH:MMp	YY/MM/DD	3123012001
		2001
Your Current Options		

1 Points

3. Trunks & Gateways

3.1. MGCP Gateway

- Register R1 as a MGCP gateway with the CallManager cluster per the following requirements:
    - o The CallManager should use ascending order, when looking for available ISDN bearer channels
    - o Make sure you busy-out the unused channels in the CallManager Service Parameters

- To verify the task, place a call from the PSTN phone to any on-site IP phone

3 Points

3.2. SIP Gateway

- Configure the router at BR1 location (R2) as a SIP gateway and add a corresponding device to the CallManager cluster
- Use Loopback0 IP address of R2 to source all SIP signaling and media packets
- Minimize the delay of waiting for a provisional response from a failed CallManager

3 Points

3.3. H.323 Gateway

- Configure the router at the BR2 location as a H.323 gateway within the CUCM system
- Use the Loopback0 IP address of the respective router to source H.323 VoIP signaling packets
- Ensure that the time needed to detect a failed CallManager is set to the minimum

3 Points

4. Call Routing

4.1. Route Plan Description

- Phones at all locations use code “9” to access the PSTN. Outside dial tone should be provided as soon as “9” has been dialed
- Users should be able to dial “911” and “9911” to reach emergency services where applicable
- Make sure “9” is stripped for calls going to PSTN
- Local area PSTN calls are placed to 7-digit numbers at the HQ and BR1 locations, and to 8-digit numbers at BR2 location
- Long-distance (national) calls at the HQ and BR1 are placed to 10-digit numbers using code “1” for toll-alert, and to 10 digit numbers at BR2 using code “0” for toll-alert
- International calls at the HQ and BR1 are signaled using access code “011” along with the variable length number
- International calls at BR2 are placed using toll-alert code “00” and variable length number
- Users may terminate their international number dialing using the “#” sign, in order to place the call immediately, without waiting for the inter-digit timeout
- You may only configure two route patterns to complete all call routing tasks. One for Emergency calls, and one for all remaining calls

0 Points

4.2. Local Gateway

- All calls from the HQ and BR1 site should use their local gateway for calls to Emergency Services (911)
- Configure only one Route Pattern for this task
- Ensure the Calling Party number type is set to Subscriber for these calls

4 Points

4.3. HQ Call Routing

- Use the following table as a reference on dialing patterns for the HQ site:

Call Type	HQ Pattern
Emergency	911
Local	[2-9]XXXXXX
National/LD	1[2-9]XX[2-9]XXXXXX
International	011 + Variable length

- Configure call routing for the HQ users per the following requirements:
    - The phones should use the on-site gateway as the primary PSTN access point
    - Hide the calling number when calling international destinations
- No special classes of restriction apply to the HQ phones, every phone is allowed to call any PSTN number

3 Points

4.4. BR1 Call Routing

- Configure call routing for BR1 phones per the following requirements:
    - The phones at BR1 location should use the on-site gateway as the primary PSTN access point
    - Hide the calling number when calling international destinations
- Use the following table as a reference on dialing patterns for the BR1 site

Call Type	BR1 Pattern
Emergency	911
Local	[2-9]XXXXXX
National/LD	1[2-9]XX[2-9]XXXXXX
International	011 + Variable length

3 Points

4.5. BR2 Call Routing

- Use the following table as a reference on BR2 dialing patterns

Call Type	BR2 Pattern
Emergency	999
Local	[1-8]XXXXXXX
National/LD	0[1-8]XXXXXXXXX
International	00 + Variable length

- Configure call routing for BR2 phones per the following requirements:
  - The IP Phones should use R3 as the only PSTN gateway
  - Ensure that BR2 users may use speed-dials “*1” and “*2” to reach HQ Phone 1 and HQ Phone 2 over the PSTN

- The IP Phones should present their full DID numbers but not names when calling any PSTN number

3 Points

4.6. Advanced Call Routing

- All incoming calls from the PSTN should appear in the called parties Missed/Received calls directory as +XXXXXXXXXX (ex. +17755011111)
- Any incoming call to BR1 phones that has no matching internal extension should be re-routed to BR1 IP Phone 2

4 Points

5. CAC & Codecs

5.1. Call Quality Control

- Ensure that a low bit rate codec is used for calls placed to the BR2 location over IP, and the best possible quality is provided to voice calls made between phones on the same site and between the HQ and BR1 sites
- Allow two VoIP calls towards BR1 devices and three VoIP calls towards BR2 phones from the HQ site
- Ensure this restriction applies to PSTN calls redirected from an on-site gateway to an off-site location

3 Points

6. Media Resources

6.1. Hardware Resources

- Register hardware DSP resources with the CallManager per the following requirements:
    - The HQ DSP resources should be registered as a conference bridge with the CallManager cluster
    - R2's on-board DSP resources should be used as a hardware transcoder by the CallManager cluster
- Restrict the number of transcoding sessions to 3 under the router configuration

3 Points

6.2. Media Resources Allocation

- All sites should use the HQ conference bridge as their hardware conferencing resource
- Additionally, all devices should be able to use BR1's hardware transcoder
- Allow fallback to software conference bridge if the hardware is not available or has exhausted it's capacity

3 Points

6.3. Music on Hold

- Configure the Publisher and Subscriber to stream Music on Hold
- Subscriber should be used first and Publisher as a fallback media streaming agent
- Music should be sent using the G.711 format inside the HQ region and using G.729 when streaming to BR1 and BR2 locations
- Use multicast media delivery only to stream MoH to BR1 and BR2 devices

3 Points

7. Applications & Services

7.1. IPMA Proxy Mode

- Use the following table as reference for user names and extension numbers to complete this task

Phone	Extension	Comment
HQ IP Ph1	1001	IPMA Manager Line
HQ IP Ph2	1002	Assistant Primary Line
	1011	Assistant Proxy Line

- HQ Phone 1 is assigned to a manager and HQ Phone 2 to the manager’s assistant
- Configure IPMA service in proxy mode to intercept calls coming to HQ Phone 1’s primary line
- Install IPMA Console application on the Subscriber CallManager to test your configuration

6 Points

8. High Availability

8.1. AAR

- Reroute VoIP calls over the PSTN in case of CallManager CAC denying the call
- Ensure AAR is working with the IPMA application

3 Points

9. QoS

9.1. Packet Marking

- Enable MLS QoS on SW1 and SW2
- Configure switchports to trust the CoS marking assigned by IP phones and ensure this marking translates correctly to a CoS value of CS3
- Appliance packets should have an explicit value of CoS 1 assigned to them
- For IOS VoIP gateways, ensure the signaling uses the DSCP value of CS3

3 Points

9.2. Traffic Shaping

- Use the following table to obtain information on Frame-Relay PVC CIR values and physical port speeds

Connection	Speed/CIR
R1 to R2	384K
R1 to R3	256K
R2 to R1	384K
R3 to R1	256K

- Configure the routers connected to the Frame-Relay cloud to conform to their provisioned CIR values
- Take into account the Cisco QoS SRND recommendation for Frame-Relay traffic-shaping CIR values
- Do not use the “frame-relay traffic-shaping” interface-level command to accomplish this task

3 Points

10. Voice Mail

10.1. Unity VM Integration

- You are planning to deploy a centralized Voice Mail system for your sites using the Unity server installed at the HQ location
- Integrate the Unity system with the CallManager cluster per the following requirements:
    - Use the voice-mail pilot number: “1500”
    - The voice-mail ports should be in the range of “1501-1502”
    - Use numbers “1998” and “1999” for MWI on and off settings

3 Points

10.2. Unity Subscribers

- Create voice mailboxes for HQ IP Phones 1, 3 and BR1 IP Phone 2
- Use the password value “123456” for every user and simple names which allow you to distinguish between users, such as “HQ Phone 1” etc.
- Make sure that a when a user presses the “Messages” button on their phone it reaches the specific voice mailbox greeting, not the self-enrollment dialog or the Unity general greeting
- An unanswered phone call or calls to a busy phone line on the above mentioned phones should be redirected to the Voice Mail system

3 Points

11. CUCCX

11.1. CUCCX

- Integrate your existing CUCCX system with the CallManager cluster per the following requirements:
    - JTAPI user name prefix/password: “jtapi/cisco”
    - RMCM user name prefix/password: “rmcm/cisco”
    - CTI port range: “1401-1402”
    - Default application script name: “icd.aef”
    - Default ICD Application CTI RP DN: “1400”

- No CTI manager redundancy is required for this task
- Configure HQ Phone 3 for an ICD agent, and assign it to call service queue

4 Points