

CCNA Voice 640-461

Number: 640-461
Passing Score: 800
Time Limit: 120 min
File Version: 12.69



<http://www.gratisexam.com/>



Extracted from the book **CCNA Voice 640-461 Official Cert Guide** by Jeremy Cioara and Mike Valentine.

Copyright © 2012 Pearson Education, Inc. Published by Cisco Press.

Exam A

QUESTION 1

Users report that all external callers are leaving urgent voice-mail messages. Where can this behavior be changed?

- A. under the Phone Menu Configuration > Unidentified Callers Message Urgency
- B. under the Opening Greeting > Unidentified Callers Message Urgency
- C. under the Message Settings > Unidentified Callers Message Urgency
- D. under the System Call Handlers > Unidentified Callers Message Urgency
- E. under the Voice-mail Box Settings > Unidentified Callers Message Urgency

Correct Answer: C

Section: (none)

Explanation

Explanation/Reference:

Message urgency indicates the action that Cisco Unity Connection allows when a message has been left by an unidentified caller or by a user who has not explicitly signed in.

Link:

http://www.cisco.com/en/US/docs/voice_ip_comm/connection/8x/administration/guide/8xcucsag170.html

QUESTION 2

Which two fields are required parameters when manually creating users on Cisco Unity Connection with predefined templates? (Choose two.)

- A. username (alias)
- B. extension
- C. first name and last name
- D. employee ID
- E. title

Correct Answer: AB

Section: (none)

Explanation

Explanation/Reference:

The following items need to be configured when creating new user from predefined template: User Template Type (extension), Based on Template, Alias, Display Name, Mailbox Store, Outgoing Fax Server, Phone System, Generate SMTP Proxy Address From Corporate Email Address (Cisco Unity Connection 8.5 and Later only)

QUESTION 3

An administrator is attempting to add a new user in Cisco Unified Communications Manager Administration but does not see the option to add a new user. What is the most likely cause of this issue?

- A. The SQL User database is not running.
- B. The system is synchronized with an LDAP server.
- C. BAT is not enabled.
- D. The administrator has the aceno user adda rights box checked.
- E. The SIP Realm is not defined in User Management.

Correct Answer: B

Section: (none)

Explanation

Explanation/Reference:

If you enable LDAP synchronization in Cisco Unified Communications Manager Administration, you thereby configure your system to use the LDAP corporate directory as the end user directory for Cisco Unified Communications Manager.

In this scenario, you cannot add or delete users in Cisco Unified Communications Manager Administration. You add and remove end users in the corporate LDAP directory.

QUESTION 4

Which call processing agent is based on Cisco IOS software and works with ISR platforms?

- A. Cisco Unified Presence Server
- B. Cisco Unity Connection
- C. Cisco Unified Communications Manager Express
- D. Cisco Unified Communications Manager
- E. Cisco UnifiedContact Center Express

Correct Answer: C

Section: (none)

Explanation

Explanation/Reference:

Cisco unified presence server, unity connection, communication manager and contact center express requires appliance based system based on Cisco media convergence server (MCS), UCS (unified computing system) or VMware.

QUESTION 5

Which two options allow the maximum message length to be adjusted in Cisco Unity Connection? (Choose two.)



<http://www.gratisexam.com/>

- A. Message Settings under individual users
- B. User Templates >VoiceMailUserTemplate > Message Settings
- C. Contacts > Message Settings
- D. Enterprise Parameters > Maximum Message length
- E. Service Parameters > Voicemail Settings

Correct Answer: AB

Section: (none)

Explanation

Explanation/Reference:

- 1.Message Settings under **individual** users
- 2.User Templates ---> VoiceMailUserTemplate ---> Message Settings

QUESTION 6

An administrator wants to locate and remove all unassigned directory numbers on the Cisco Unified Communications Manager system. Which method is the best method to accomplish this task?

- A. Choose Device > Phone. Search all phones and remove the undesired directory numbers.
- B. Use the Dial Plan Installer to remove the directory numbers.
- C. Use the Disaster Recovery System to restore only valid directory numbers.
- D. Choose Call Routing > Route Plan Report, choose the Unassigned DN drop-down menu, and then remove all orphaned directory numbers.
- E. Choose Device > Device Settings > Device Defaults and use the wizard to locate and remove the orphaned directory numbers.

Correct Answer: D

Section: (none)

Explanation

Explanation/Reference:

Step 1 - Choose Call Routing > Route Plan Report.

The Route Plan Report window displays. Use the three drop-down list boxes to specify a route plan report that lists all unassigned DNs.

Step 2 - Three ways exist to delete directory numbers:

- a. Click the directory number that you want to delete. When the Directory Number Configuration window displays, click Delete.
- b. Check the check box next to the directory number that you want to delete. Click Delete Selected.
- c. To delete all found unassigned directory numbers, click Delete All Found Items.

A warning message verifies that you want to delete the directory number.

Step 3 - To delete the directory number, click OK. To cancel the delete request, click Cancel.

QUESTION 7

When troubleshooting a phone that is unable to get an IP address from a DHCP server, what is the first thing to check for on the phone?

- A. Make sure that DHCP Enabled is disabled on the phone.
- B. Make sure that the phone is getting the proper VLAN information
- C. Make sure that the TFTP server address is correct on the phone.
- D. Make sure that the DHCP scope has enough addresses left in the range.
- E. Make sure the phone has the correct phone load ID.

Correct Answer: B

Section: (none)

Explanation

Explanation/Reference:

IP phones rely on a TFTP-based process to acquire configuration files, software images, and other endpoint-specific information. The Cisco TFTP service is a file serving system that can run on one or more Unified CM servers. **(Actual test explanation)**

If your IP phone is not able to obtain an IP address via DHCP, check for any network connectivity problems between the IP phone and the DHCP server. If the DHCP server is not the same subnet as the IP phone and the DHCP server. If the DHCP server is not on the same subnet as the IP phone voice VLAN, ensure that you have properly configured an IP helper address on the local router.

QUESTION 8

Which issue does CAS signaling on a T1 circuit create?

- A. Signaling bits are subtracted from each frame, which causes a significant loss of voice quality.
- B. An extra signaling bit is added to the sixth frame to carry signaling information.
- C. A signaling bit is subtracted from every sixth frame to carry signaling information.
- D. Signaling bits are added to the signaling stream to create extended super frames.

Correct Answer: C

Section: (none)

Explanation

Explanation/Reference:

Each T1 channel carries a sequence of frames. These frames consist of 192 bits and an additional bit designated as the framing bit, for a total of 193 bits per frame.

Super Frame (SF) groups twelve of these 193 bit frames together and designates the framing bits of the even numbered frames as signaling bits.

CAS looks specifically at every sixth frame for the timeslot's or channel's associated signaling information

QUESTION 9

Which description describes the weighted fair queuing algorithm?

- A. Empty queue 1. If queue 1 is empty, empty queue 2, then empty queue 3, unless a packet for a higher queue arrives.
- B. An administrator defines the traffic classes based on match criteria, including protocols, access control lists, and input interfaces.
- C. A flow-based algorithm that simultaneously schedules interactive traffic to the front of a queue to reduce response time and fairly shares the remaining bandwidth among high-bandwidth flows.
- D. This feature brings strict priority queuing to CBWFQ.
- E. Packets are placed into a single queue and serviced in the order they were received.

Correct Answer: C

Section: (none)

Explanation

Explanation/Reference:

WFQ provides traffic priority management that automatically sorts among individual traffic streams without requiring that you first define access lists. WFQ can also manage duplex data streams such as those between pairs of applications, and simplex data streams such as voice or video. There are two categories of WFQ sessions: high bandwidth and low bandwidth. Low-bandwidth traffic has effective priority over high-bandwidth traffic, and high-bandwidth traffic shares the transmission service proportionally according to assigned weights.

When WFQ is enabled for an interface, new messages for high-bandwidth traffic streams are discarded after the configured or default congestive messages threshold has been met. However, low-bandwidth conversations, which include control message conversations, continue to enqueue data. As a result, the fair queue may occasionally contain more messages than its configured threshold number specifies.

With standard WFQ, packets are classified by flow. Packets with the same source IP address, destination IP address, source Transmission Control Protocol (TCP) or User Datagram Protocol (UDP) port, or destination TCP or UDP port belong to the same flow. WFQ allocates an equal share of the bandwidth to each flow. Flow-based WFQ is also called fair queueing because all flows are equally weighted.

The Cisco IOS software provides two forms of WFQ:

- Standard WFQ, which is enabled by default on all serial interfaces that run at or below 2 Mbps, and can run on all Cisco serial interfaces.

• Distributed WFQ, which runs only on Cisco 7000 series routers with a Route Switch Processor-based RSP7000 interface processor or Cisco 7500 series routers with a Versatile Interface Processor-based VIP2-40 or greater interface processor. (A VIP2-50 interface processor is strongly recommended when the aggregate line rate of the port adapters on the VIP is greater than DS3. A VIP2-50 interface processor is required for OC-3 rates.)

Link:

http://www.cisco.com/en/US/docs/ios/12_0/qos/configuration/guide/qcwfq.html

QUESTION 10

Which utility should you use when you need to add a large number of users into Cisco Unified Communications Manager?

- A. Cisco Unified User Administration
- B. Application User CAPF Profile
- C. Cisco Unified Communications Manager Bulk Administration Tool
- D. Cisco Unified Telephony User Administration

Correct Answer: C

Section: (none)

Explanation

Explanation/Reference:

The Cisco Unified Communications Manager Bulk Administration Tool (BAT), a web-based application, performs bulk transactions to the Cisco Unified Communications Manager database.

BAT lets you add, update, or delete a large number of similar phones, users, or ports at the same time. When you use Cisco Unified Communications Manager Administration, each database transaction requires an individual manual operation,

while BAT automates the process and achieves faster add, update, and delete operations.

Link:

http://www.cisco.com/en/US/docs/voice_ip_comm/cucm/bat/7_1_2/t01batow.html

QUESTION 11

Which type of user in Cisco Unified Communications Manager has an interactive login?

- A. administrator
- B. end user
- C. application user
- D. phone user

Correct Answer: B

Section: (none)

Explanation

Explanation/Reference:

End users — End users are associated with an individual and have an interactive login. End users can have administrative roles based on the user group role configuration.

Application users — Application users are associated with applications such as Cisco Unified Attendant Console, Cisco Unified Contact Center Express (UCCX), or Cisco Unified Manager Assistant.

The mentioned applications need to authenticate with CUCM, but application users do not have the ability to interactively log in. Application users are leveraged for internal process-level communications between applications

Link:

http://www.cisco.com/en/US/products/sw/voicesw/ps556/products_configuration_example09186a0080b3dd05.shtml

QUESTION 12

When you configure a shared extension in Cisco Configuration Professional, which two characteristics should be the same on both phones, so that the phones can successfully use a shared extension? (Choose two.)

- A. monitor mode
- B. MAC address
- C. ephone-dn
- D. number
- E. user ID

Correct Answer: CD

Section: (none)

Explanation

Explanation/Reference:

You can set up one or more lines with a shared-line appearance. A Cisco CallManager system considers a directory number to be a shared line if it appears on more than one device in the same partition. For example, if directory number 9600 on phone A is in the partition called Dallas and on phone B in the partition called Texas, that directory number does not represent a shared-line appearance. (Ensure the directory number 9600 for phone A and phone B are in the same partition; for example, Dallas.)

In a shared-line appearance, for example, you can set up a shared line, so a directory number appears on line 1 of a manager phone and also on line 2 of an assistant phone. Another example of a shared line involves a single incoming 800 number that is set up to appear as line 2 on every sales representative phone in an office. You can also choose to update a directory number and have the updates apply to all devices that share the directory number.

The following information provides tips about and lists the restrictions for using shared-line appearances with Cisco CallManager.

Shared Line Tips

Use the following tips when configuring shared lines:

- You create a shared-line appearance by assigning the same directory number and route partition to different devices.
- If multiple devices share a line, each device name displays in the Associated Devices pane of the directory number in the Directory Number Configuration window in Cisco CallManager Administration.
- If you change the Calling Search Space or Call Forward and Pickup settings on any device that uses the shared line, the changes apply to all devices that use that shared line.
- To stop sharing a line appearance on a device, change the directory number or partition name for the line and update the directory number in the Directory Number Configuration window in Cisco CallManager Administration.
- In the case of a shared-line appearance, Remove From Device removes the directory number on the current device only and does not affect other devices.
- Most devices with a shared-line appearance can make or receive new calls or resume held calls at the same time. Incoming calls display on all devices that share a line, and anyone can answer the call. Only one call remains active at a time on a device.

Link:

http://www.cisco.com/en/US/docs/voice_ip_comm/cucm/admin/5_0_1/ccmsys/a03dn.html

QUESTION 13

Refer to the exhibit. The exhibit shows a partial screen shot for a Cisco Unified Client Services Framework device. If the username that is associated with this device is jdoe, what should the device name be?

Phone Configuration

Save

Status

Status: Ready

Phone Type

Product Type: Cisco Unified Client Services Framework
Device Protocol: SIP

Device Information

Device is trusted

Device Name *

Description

Device Pool * Default

Common Device Configuration < None >

Phone Button Template * -- Not Selected --

Common Phone Profile * Standard Common Phone Profile

Calling Search Space < None >

- A. The device name should be JDOEUPC.
- B. The device name should be UPCJDOE.
- C. The device name should be JDOE.
- D. The device name should be UPCCUPC.
- E. The device name should be UPCCSF.
- F. The device name has no naming convention.

Correct Answer: F

Section: (none)

Explanation

Explanation/Reference:

When you create a softphone device for each Cisco Unified Personal Communicator user, you must specify a device name. Make sure that the device name conforms to these guidelines:

Link:

http://docwiki.cisco.com/wiki/Cisco_Unified_Presence,_Release_7.x_-_How_to_Configure_Cisco_Unified_Personal_Communicator_on_Cisco_Unified_Communications_Manager#Table:_Username_Conversion_for_Cisco_Unified_Personal_Communicator_Softphone_Device

QUESTION 14

When implementing a plan of action, what should you do?

- A. Make all of the changes at once to minimize the impact to users.
- B. Limit the impact of the changes to users.

- C. Completely remove access lists to ensure that they will not impact the changes.
- D. Even if a change adversely affects the users, keep moving forward with the plan of action.

Correct Answer: B

Section: (none)

Explanation

Explanation/Reference:

Making all of the change at once is not good decision. Making changes is a step by step process, make the change, apply the change and watch for the behaviour of the system or the impact to the user or users. This way if something goes wrong the change can be reversed or corrected quickly and timely with out wondering which change caused the problem if many changes are made at once.

QUESTION 15

Refer to the exhibit. Where can the phone menu be accessed from?

Phone Menu

User Template Edit Refresh Help

Save

Phone Menus

Touchtone Conversation Menu Style Full

Conversation Volume Medium

Conversation Speed Normal

Time Format

12-Hour Clock (12:00 AM - 11:59 PM)

24-Hour Clock (00:00 - 23:59)

Conversation Style

Touchtone Conversation Classic Conversation

Finding Messages with Message Locator

Enable

Message Locator Sort Order Last In, First Out

- A. from the individual users or user templates
- B. from the Class of Service configuration screen
- C. from the user contacts
- D. from the Interview Handler configuration screen
- E. from the Message Store configuration screen

Correct Answer: A

Section: (none)

Explanation

Explanation/Reference:

Cisco Unity Connection offers several versions of the phone conversation that users hear and use. The version you select determines whether Connection responds only to phone keypad input or also uses voice recognition to interpret spoken commands

QUESTION 16

A user in Cisco Unified Communications Manager Administration has been added to the Standard CCM Admin Users group, which includes the Standard CCMADMIN Administration role, but the user cannot add new users. What is the cause of this issue?

- A. The add user capability has been disabled for the group
- B. The incorrect group and role were assigned.
- C. The add user capability has been disabled for the role.
- D. Only theCCMAdmin user can add users.
- E. Users can be added only via LDAP

Correct Answer: B

Section: (none)

Explanation

Explanation/Reference:

The Standard CCM Admin Users role includes no permissions beyond logging into Cisco Unified Communications Manager Administration.

The administrator must add another authorization role to define the parts of the Cisco Unified Communications Manager Administration that the user can administer.

The Standard CCMADMIN Administration role allows a user to access and make changes in all of Cisco Unified Communications Manager Administration.

Link:

http://www.cisco.com/en/US/docs/voice_ip_comm/cucmbe/admin/8_6_1/ccmsys/a02mla.html

QUESTION 17

Which two statistics can be generated with the Server Report in Cisco Unity Connection? (Choose two)

- A. Percentage of CPU per Server
- B. Percentage of Memory Usage per Server
- C. Percentage of Hard Disk Usage of the Active Partition per Server
- D. Cisco Unity Connection Port Activity Report
- E. Percentage of Port Usage

Correct Answer: AB

Section: (none)

Explanation

Explanation/Reference:**Server Statistics Report**

The Server Statistics Report provides the following line charts:

- **Percentage of CPU per Server**
- **Percentage of Memory Usage per Server**
- **Percentage of Hard Disk Usage of the Largest Partition per Server**

In a Cisco Unified Communications Manager Business Edition 5000 system, the Server Statistics Report supports both Cisco Unified Communications Manager and Cisco Unity Connection.

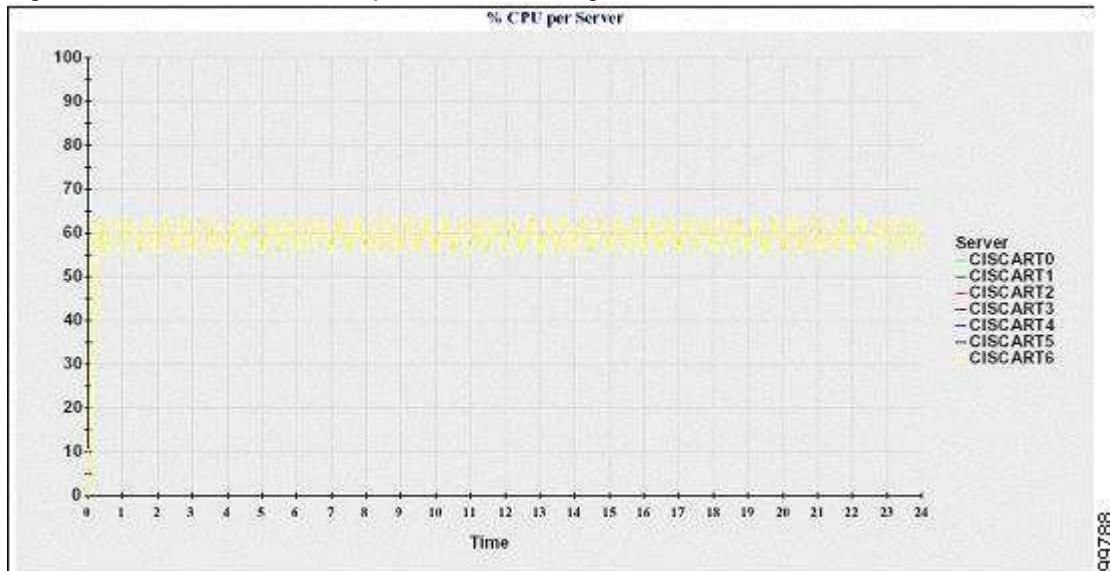
Percentage of CPU per Server

A line chart displays the percentage of CPU usage for the server (or for each server in a Cisco Unified Communications Manager cluster). The line in the chart represents the data for the server (or one line for each

server in a Cisco Unified Communications Manager cluster) for which data is available. Each data value in the chart represents the average CPU usage for a 15-minute duration. If no data exists for the server (or for any one server in a Cisco Unified Communications Manager cluster), Reporter does not generate the line that represents that server. If there are no lines to generate, Reporter does not create the chart. The message "No data for Server Statistics report available" displays.

Figure 10-5 shows a line chart example that represents the percentage of CPU usage per server in a Cisco Unified Communications Manager cluster configuration.

Figure 10-5 Line Chart That Depicts the Percentage of CPU Per Server

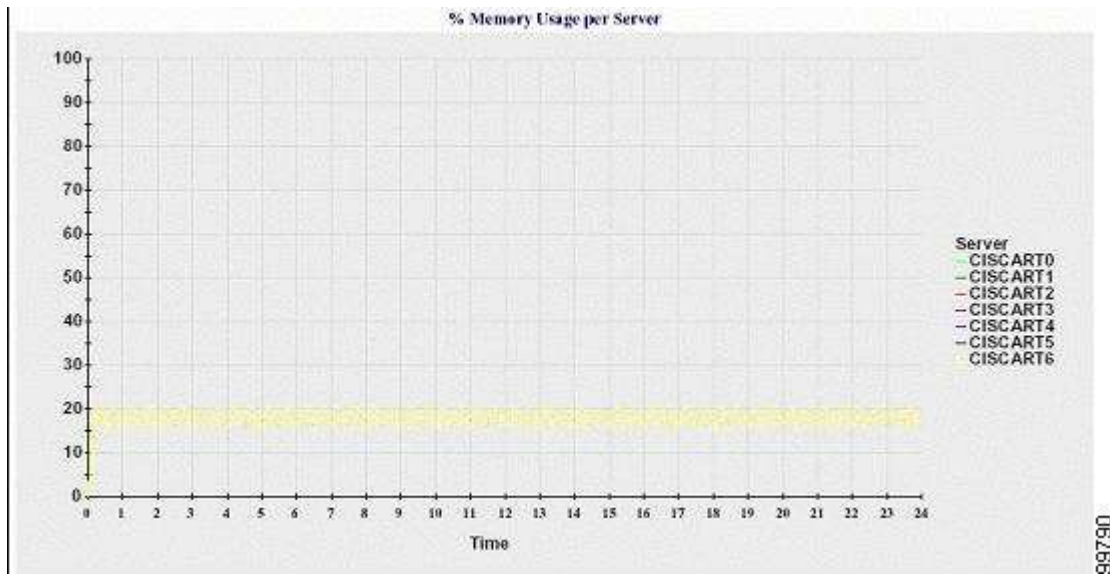


Percentage of Memory Usage per Server

A line chart displays the percentage of Memory Usage for the Cisco Unified Communications Manager server (%MemoryInUse). In a Cisco Unified Communications Manager cluster configuration, there is one line per server in the cluster for which data is available. Each data value in the chart represents the average memory usage for a 15-minute duration. If no data exists, Reporter does not generate the chart. If no data exists for any server in a Cisco Unified Communications Manager cluster configuration, Reporter does not generate the line that represents that server.

Figure 10-6 shows a line chart example that represents the percentage of memory usage per Cisco Unified Communications Manager server in a Cisco Unified Communications Manager cluster configuration.

Figure 10-6 Line Chart That Depicts Percentage of Memory Usage Per Server

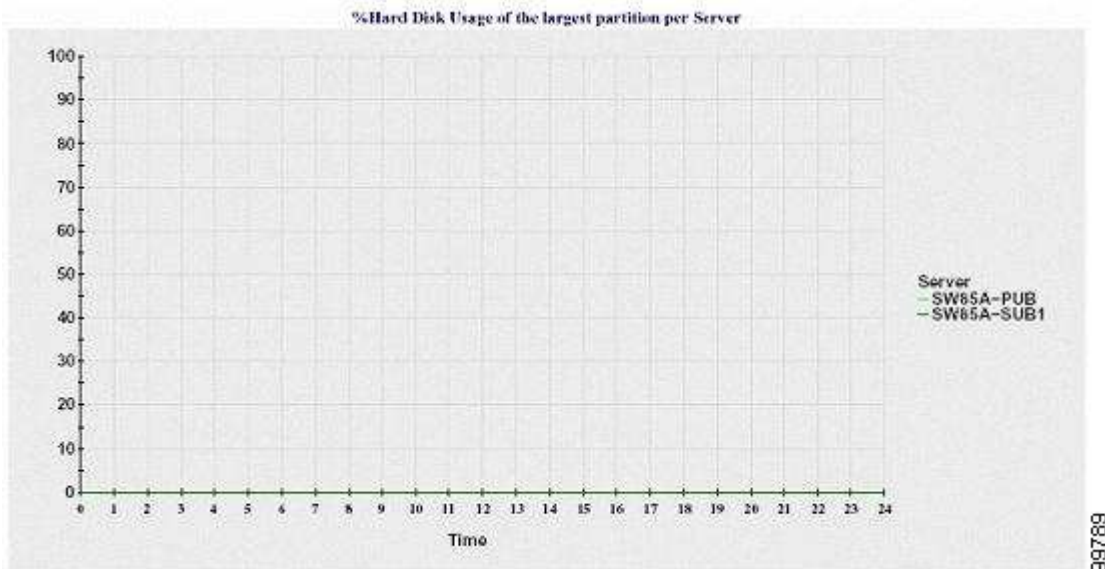


Percentage of Hard Disk Usage of the Largest Partition per Server

A line chart displays the percentage of disk space usage for the largest partition on the server (% DiskSpaceInUse), or on each server in a Cisco Unified Communications Manager cluster configuration. Each data value in the chart represents the average disk usage for a 15-minute duration. If no data exists, Reporter does not generate the chart. If no data exists for any one server in a cluster configuration, Reporter does not generate the line that represents that server.

Figure 10-7 shows a line chart example that represents the percentage of hard disk usage for the largest partition per server in a Cisco Unified Communications Manager cluster configuration.

Figure 10-7 Line Chart That Depicts Percentage of Hard Disk Usage of the Largest Partition Per Server



The server (or each server in a Cisco Unified Communications Manager cluster configuration) contains log files that match the file name pattern ServerLog_mm_dd_yyyy_hh_mm.csv. The following information exists in the log file:

- % CPU usage on the server (or each server in a Cisco Unified Communications Manager cluster)

- % Memory usage (%MemoryInUse) on the server (or on each server in a Cisco Unified Communications Manager cluster)
- % Hard disk usage of the largest partition (%DiskSpaceInUse) on the server (or on each server in a Cisco Unified Communications Manager cluster)

Link:

http://www.cisco.com/en/US/docs/voice_ip_comm/cucm/service/8_5_1/admin/sasrdes.html

QUESTION 18

Refer to the exhibit. The exhibit shows a partial screen shot for a Cisco Unified Client Services Framework device. When should this device be configured?



- A. when configuring the Service Advertisement Framework feature for Call Control Discovery
- B. when Cisco Unified Personal Communicator is used in desk-phone mode.
- C. when Cisco Unified Personal Communicator version 7.0 is used in soft-phone mode
- D. when Cisco Unified Personal Communicator version 8.0 is used in soft-phone mode
- E. when Cisco Unified Personal Communicator version 8.0 is used in desk-phone mode

Correct Answer: A

Section: (none)

Explanation

Explanation/Reference:

The device is being configured for Service Advertisement Framework feature for call control.

QUESTION 19

How can an end user be defined as a manager in the CAR tool?

- A. Configure a Manager ID field in end user configuration
- B. Add the end user to the Standard CCM Manager group
- C. Check the Manager check box in the LDAP directory
- D. Enable an end user as a manager in the CAR tool

Correct Answer: A

Section: (none)

Explanation

Explanation/Reference:

Any user can act as a CAR administrator (including application users); however, you must add the end user to the Cisco CAR Administrators User Group in Cisco Unified Communications Manager Administration (Standard CAR Admin Users).

End users who have been identified as CAR administrators have full control over the CAR system. The administrator can modify all the parameters that relate to the system and the reports.

End users who have not been identified as CAR administrators can access only designated CAR reports.

Link:

http://www.cisco.com/en/US/docs/voice_ip_comm/cucm/service/8_0_2/car/caranrpt.html#wp1072381

QUESTION 20

Which of the following is not a source for Cisco Unified Reporting?

- A. Cisco Unified CM database
- B. Cisco Unified Presence database
- C. Cisco Unified CM RTMT counters
- D. Cisco Unified CM CDR/CAR

Correct Answer: B

Section: (none)

Explanation

Explanation/Reference:

What is Cisco Unified Reporting?

Cisco Unified Reporting web application, which is accessed at the Cisco Unified CallManager console, generates reports for troubleshooting or inspecting cluster data.

This convenient tool provides a snapshot of cluster data without requiring multiple steps to get the data. The tool design facilitates gathering data from existing sources, comparing the data, and reporting irregularities.

A report combines data from one or more sources on one or more servers into one output view. For example, you can view a report that shows the hosts file for all servers in the cluster.

The application gathers information from the publisher server and each subscriber server. A report provides data for all active cluster nodes that are accessible at the time that the report is generated.

Some reports run checks to identify conditions that could impact cluster operations. Status messages indicate the outcome of every data check that is run.

Only authorized users can access the Cisco Unified Reporting application. By default, this includes administrator users in the Standard Unified CM Super Users group. As an authorized user, you can view reports, generate new reports, or download reports at the graphical user interface (GUI).

Note: Administrator users in the Standard Unified CM Super Users group can access all administrative applications in the Cisco Unified CallManager Administration navigation menu, including Cisco Unified Reporting, with a single sign on to one of the applications.

Cisco Unified Reporting includes the following capabilities:

- A user interface for generating, archiving, and downloading reports
- Notification message if a report will take excessive time to generate or consume excessive CPU

Data Sources

Reports that Cisco Unified Reporting generates may use any of the following sources:

- RTMT counters
- CDR_CAR
- CUCM DB
- disk files
- OS API calls
- network API calls
- prefs (Windows registry)
- CLI
- RIS

The Report Descriptions link in the System Reports list provides the information sources that are used in a report, as described in Accessing On-Line and Report Help.

Navigation

Cisco Unified Reporting For Cisco Unified Communications Solutions

System Reports Help

System Reports

- Unified CM Cluster Overview
- Unified CM Database Status
- Unified CM Data Inventory Summary
- Report Descriptions
- Unified CM Database Replication Debug
- CUCM Count By Server

OK: Report opened successfully.

Unified CM Cluster Overview

Provide an overview of the Unified CM cluster.
Created on Thu Aug 09 09:45:54 CDT 2007

Unified CM Cluster Name
Lists the cluster name from the Enterprise Parameter and the publisher server name.

Cluster Name	Publisher Name/IP
StandAloneCluster	sw006a-118

Unified CM Provisioned Servers
Lists all servers in the cluster by either name or IP as provisioned in the database, installed or not. The IP address is obtained from the host file on the local server.

The 10.54.68.22 server is provisioned in the Database but not installed.

Name	Description	IP Address
sw006a-118		10.89.87.118
sw006a-119	sw006a-119	10.89.87.119
sw006a-120	sw006a-120	10.89.87.120
10.54.68.22		Not Installed

Link:

http://www.cisco.com/en/US/docs/voice_ip_comm/cucm/service/5_1_3/report/curptg.html

QUESTION 21

Which two steps are not necessary when enabling an end user for Cisco Unified Personal Communicator?
(Choose two)

- A. Assign license capabilities
- B. Subscribe phone service
- C. Create a CSF device
- D. Configure the owner user ID

Correct Answer: BD

Section: (none)

Explanation

QUESTION 22

A new phone has been added to the Cisco Unified Communications Manager server. The phone display shows Your Current Options, but when the New Call softkey is pressed, no dial tone is heard and the call cannot be placed.

What could be the cause of this issue?

- A. An incorrect MAC address has been entered for the new phone.
- B. No directory number has been assigned to a line.
- C. The end user is not associated with the device.
- D. No calling search space has been configured on the line.
- E. An incorrect device pool has been configured on the phone.

Correct Answer: B

Section: (none)

Explanation

Explanation/Reference:

A Cisco Unified CME system uses the following basic building blocks:

- **Ephone or voice register pool** - A software concept that usually represents a physical telephone, although it is also used to represent a port that connects to a voice-mail system, and provides the ability to configure a physical phone using Cisco IOS software. Each phone can have multiple extensions associated with it and a single extension can be assigned to multiple phones. Maximum number of ephones and voice register pools supported in a Cisco Unified CME system is equal to the maximum number of physical phones that can be connected to the system.

- **Directory number** - A software concept that represents the line that connects a voice channel to a phone. A directory number represents a virtual voice port in the Cisco Unified CME system, so the maximum number of directory numbers supported in Cisco Unified CME is the maximum number of simultaneous call connections that can occur. This concept is different from the maximum number of physical lines in a traditional telephony system.

Link:

http://www.cisco.com/en/US/docs/voice_ip_comm/cucme/admin/configuration/guide/cmeover.html

QUESTION 23

What Cisco client application allows administrators to interact with performance monitoring counters to assist in determining the overall health of the Cisco Unified Communications Manager server?

- A. Cisco Unified Communications Manager Administration
- B. Cisco Unified Real-Time Monitoring Tool
- C. Cisco Unified OS Administration
- D. CAR Tool
- E. BAT Tool

Correct Answer: B

Section: (none)

Explanation

Explanation/Reference:

Cisco Unified Communications Manager Administration Guide provides instructions for administering the Cisco Unified Communications Manager. Cisco Unified Communications Operating System Administration allows you to configure and manage the Cisco Unified Communications Operating System.

CAR denotes to CDR analysis and reporting tool, The Cisco Unified Communications Manager Bulk

Administration Tool (BAT), a web-based application, performs bulk transactions to the Cisco Unified Communications Manager database.

BAT lets you add, update, or delete a large number of similar phones, users, or ports at the same time.

When you use Cisco Unified Communications Manager Administration, each database transaction requires an individual manual operation, while BAT automates the process and achieves faster add, update, and delete operations.

Link:

http://www.cisco.com/en/US/docs/voice_ip_comm/cucm/service/8_0_2/rtmt/RTMT.html

QUESTION 24

Which report can be generated by using the User Reports feature of the CAR tool?

- A. Traffic
- B. Top N
- C. Malicious Call Details
- D. CDR Error
- E. FAC/CMC

Correct Answer: B

Section: (none)

Explanation

Explanation/Reference:

Users, managers, and CAR administrators can generate user reports. CAR includes the following user reports: Bills, Top N, FAC/CMCC, CDR error and malicious call details are system report.

Link:

http://www.cisco.com/en/US/docs/voice_ip_comm/cucm/service/8_6_1/car/carustop.html

QUESTION 25

After changes are made to an IP phone, which reset method is the fastest to bring the phone back into service?

- A. drop
- B. restart
- C. reset
- D. shutdown
- E. shut and no shut

Correct Answer: B

Section: (none)

Explanation

Explanation/Reference:

reset Command----->Takes longer to process when updating multiple phones.

restart Command----->Faster processing for multiple phones.

restart- Performs a fast reboot of the specified phone or all phones running SCCP associated with this Cisco Unified CME router. Does not contact the DHCP server for updated information

Link:

http://www.cisco.com/en/US/docs/voice_ip_comm/cucme/admin/configuration/guide/cmreset.html#wpixref87974

QUESTION 26

In which two locations can an end user configure their Call Forward All settings? (Choose two.)

- A. Cisco Unified Serviceability

- B. Cisco Unified Communications Manager User Options Interface
- C. Directly on the Cisco Unified IP phone
- D. Cisco Unified Communications Manager Administration
- E. Cisco Unified User Serviceability

Correct Answer: BC

Section: (none)

Explanation

Explanation/Reference:

Cisco Unified Serviceability, a web-based troubleshooting tool.

Cisco unified communication manager administration comprise all the features

QUESTION 27

What is the quickest way to test the Cisco Unified Communications Manager configuration part of MWI to see if MWI On and MWI Off is working?

- A. Dial into Cisco Unity Connection from an IP phone. Enter the MWI On numbers, then enter the MWI Off numbers.
- B. Call a voice-mail user and ask them if their MWI light is on, and then disconnect the call. Call the user back and ask if the MWI light is off.
- C. In Unity Connection, issue the MWI Flash command to turn all MWI lights on, then off.
- D. If MWI numbers are dialable from an IP phone, dial the MWI On number. If the light comes on, then dial the MWI Off number to see if the light goes off.
- E. MWI cannot be tested directly from the Cisco Unified Communications Manager or an IP phone.

Correct Answer: D

Section: (none)

Explanation

Explanation/Reference:

The Phone System Is Not Able To Turn MWIs On and Off

To test whether the phone system can turn MWIs on and off (Cisco CallManager integration only)

Step 1 - From a test phone whose MWI is off, dial the number that turns MWIs on. The indicator should turn on. If the indicator turns on, continue with Step 2. If not, skip to Step 3.

Step 2 - Dial the number that turns MWIs off. The indicator should turn off. If it is off, continue with the next possible cause in the "Subscriber Phone System Assignment Is Incorrect (Dual Integrations Only)" section. If it is not off, continue with Step 3.

Step 3 - Confirm that the MessageWaitingOnDN and MessageWaitingOffDN extensions are not used by any device on CallManager.

Step 4 - Confirm that there are no other dialable extensions in the same range as the MessageWaitingOnDN and MessageWaitingOffDN extensions. For example, confirm that extensions 1xxx do not interfere with extensions 1xxxx.

Step 5 - Confirm that all of the CallManager voice messaging ports are in the same Calling Search Space as all of the subscriber phones. Change the Calling Search Space assignments if necessary.

Step 6 - Confirm that CallManager port forwarding and hunt groups do not include the MWI ports. Change the port forwarding or hunt group programming if necessary.

Step 7 - Confirm that you are using the correct version of the TSP for your Cisco Unity release. For a list of the

TSP versions that each version of Cisco Unity supports, refer to Cisco Unity-CM TSP Release Notes, available on Cisco.com at http://www.cisco.com/univercd/cc/td/doc/product/voice/c_unity/tsp/index.htm and on the Cisco Documentation CD-ROM.

Step 8 - Confirm that the MessageWaitingOnDN and MessageWaitingOffDN extensions specified in the Cisco CallManager Service Parameters are the same as those specified in the TAPI Service Provider settings. If they are not the same, change the TAPI Service Provider settings.

Step 9 - If you made a change to the Cisco CallManager or TAPI Service Provider settings, shut down and restart the Cisco CallManager service and the Cisco Unity server for the changes to take effect.

Step 10 - Perform Step 1 and Step 2 again. If you hear a reorder tone, then the MessageWaitingOnDn and MessageWaitingOffDn settings are not correct on CallManager. Return to Step 3.

If you do not hear reorder tone, but the MWIs are not on or turned off, then your route plan may be causing the problem. Review the route plan and return to Step 5.

Step 11 - If MWIs still are not working, contact Cisco TAC.

Link:

http://www.cisco.com/en/US/docs/voice_ip_comm/unity/3x/troubleshooting/guide/312/TSG_0500.html

QUESTION 28

Which tools allow the administrator to migrate users from Cisco Unity to Cisco Unity Connection?

- A. Cisco Object Backup and Restore Application Suite
- B. Cisco Disaster Recovery Framework Tool
- C. Cisco Real Time Monitoring Tool
- D. Cisco Unity Serviceability Tool

Correct Answer: A

Section: (none)

Explanation

Explanation/Reference:

To migrate messages and data from Cisco Unity 4.0(5) or later to Connection 8.x, we recommend that you use the Cisco Unified Backup and Restore Application Suite (COBRAS) instead of the Migrate Messages and Migrate Users utilities.

Download the latest version, and view training videos and Help at http://www.ciscounitytools.com/App_COBRAS.htm.

QUESTION 29

Which statement describes the difference between SIP and SCCP integration of Cisco Unity Connection?

- A. SCCP integrations cannot be secured
- B. SIP integrations do not need MWI configuration
- C. SCCP integrations do not need MWI configuration
- D. SIP integration cannot be secured

Correct Answer: B

Section: (none)

Explanation

Explanation/Reference:

While both the methods are pretty good and matured, SCCP integration is best option in my opinion. It is because of a couple of reasons which are as follows:

1 - In case of SCCP integration there is 1 to 1 correlation between the voicemail ports on the CUCM and that on Unity Connection. This makes the troubleshooting of Call Flows fairly simple. Unlike SCCP, SIP integration involves one SIP trunk per unity connection server.

2 - The way the MWI dialout works in SCCP integration works is much simpler compared to how it works with SIP.

In SCCP, VM ports simulate a phone extension dialing an MWI on/ off number which is fairly simple to troubleshoot.

In SIP integration, the MWI dialouts are performed using SIP NOTIFY messages.

Having said that, I would again like to emphasize that kind of integration is a matter of choice and convenience. It is difficult to determine one being better over other.

Link:

<https://supportforums.cisco.com/thread/2098717>

QUESTION 30

Which is indicated as the calling number for a call that is placed from a remote destination to an internal directory number?

- A. the Mobile Voice Access number
- B. the number of the remote destination
- C. the directory number of the office phone with which the remote destination is associated
- D. the directory number of the called office phone, if it is associated with the calling remote destination

Correct Answer: C

Section: (none)

Explanation

Explanation/Reference:

Mobile Connect allows users to manage business calls using a single phone number and pick up in-progress calls on the desktop phone and cellular phone. Mobile Voice Access is the associated integrated voice response (IVR) system, which allows users to turn Mobile Connect on or off and to initiate calls from a cellular phone or other remote phone as if the call were initiated from the desktop phone.

A user's remote destination profile contains the parameters that apply to all the remote destinations (cellular or other phones) available for in-progress call transfers and initiation of calls by way of Mobile Voice Access.

For more information on Mobile Connect and Mobile Voice Access and how to configure remote destination profiles, refer to the Mobile Connect and Mobile Voice Access chapter in the Cisco Unified Communications Manager Features and Services Guide.

Link:

http://www.cisco.com/en/US/docs/voice_ip_comm/cucm/admin/6_1_1/ccmcfg/b06rdprf.html

QUESTION 31

What component does Cisco Unified Communications Manager Express use to match outbound dial peers?

- A. destination pattern
- B. incoming called-number
- C. calling number ANI
- D. answer-address
- E. port or session target

Correct Answer: A

Section: (none)
Explanation

Explanation/Reference:

When a router receives a voice call, it selects an outbound dial peer by comparing the called number (the full E.164 telephone number) in the call information with the number configured as the destination pattern for the POTS dial peer

Link:

http://www.cisco.com/en/US/docs/voice_ip_comm/cucme/admin/configuration/guide/cmedialp.html

QUESTION 32

A customer is in the planning stages of deploying a Cisco Unified Communications solution for their company. Previously, they were leasing a traditional PBX system from the telco and they have very little experience with voice. The customer wants to know what two signaling methods between the IP phone and the Cisco Unified Communications Manager Express are available for their use. (Choose two)

- A. H.323
- B. SCCP
- C. MGCP
- D. SIP
- E. RTP

Correct Answer: BD

Section: (none)

Explanation

Explanation/Reference:

Correct Answer: B & D

Skinnny Client Control Protocol (SCCP)

SCCP uses Cisco-proprietary messages to communicate between IP devices and Cisco CallManager. SCCP easily coexists in a multiple protocol environment. The Cisco IP Phone is an example of a device that registers and communicates with Cisco CallManager as an SCCP client. During registration, a Cisco IP phone receives its line and all other configurations from Cisco CallManager. Once it registers, it is notified of new incoming calls and can make outgoing calls. The SCCP protocol is used for VoIP call signaling and enhanced features such as Message Waiting Indication (MWI).

Session Initiation Protocol (SIP)

The Internet Engineering Task Force (IETF) developed the SIP standard for multimedia calls over IP. ASCII-based SIP works in client/server relationships as well as in peer-to-peer relationships. SIP uses requests and responses to establish, maintain, and terminate calls (or sessions) between two or more end points. Refer to the Understanding Session Initiation Protocol (SIP) chapter for more information on SIP and the interaction between SIP and Cisco Call Manager.

H.323 Protocol

The International Telecommunications Union (ITU) developed the H.323 standard for multimedia communications over packet networks. As such, the H.323 protocol is a proven ITU standard and provides multivendor interoperability. The H.323 protocol specifies all aspects of multimedia application services, signaling, and session control over an underlying packet network. Audio is standard on H.323 networks, but networks can be scaled to include both video and data. The H.323 protocol can be implemented in large enterprise networks or can be deployed over an existing infrastructure, which makes H.323 an affordable solution.

The basic components of the H.323 protocol are terminals, gateways, and gatekeepers (which provide call

control to H.323 endpoints). Similar to other protocols, H.323 applies to point-to-point or multipoint sessions. However, compared to MGCP, H.323 requires more configuration on the gateway.

Media Gateway Control Protocol (MGCP)

MGCP provides Cisco CallManager a powerful, flexible and scalable resource for call control. Cisco CallManager uses MGCP to control media on the telephony interfaces of a remote gateway and also uses MGCP to deliver messages from a remote gateway to appropriate devices.

MGCP enables a call agent (media gateway controller) to remotely control and manage voice and data communication devices at the edge of multiservice IP packet networks. Because of its centralized architecture, MGCP simplifies the configuration and administration of voice gateways and supports multiple (redundant) call agents in a network. MGCP does not provide security mechanisms such as message encryption or authentication.

Using MGCP, Cisco CallManager controls call processing and routing and provides supplementary services to the gateway. The MGCP gateway provides call preservation (the gateway maintains calls during failover and fallback), redundancy, dial-plan simplification (the gateway requires no dial-peer configuration), hookflash transfer, and tone on hold. MGCP-controlled gateways do not require a media termination point (MTP) to enable supplementary services such as hold, transfer, call pickup, and call park. If the MGCP gateway loses contact with its Cisco Call Manager, it falls back to using H.323 control to support basic call handling of FXS, FXO, T1 CAS, and T1/E1 PRI interfaces.

Link:

http://www.cisco.com/en/US/docs/voice_ip_comm/cucm/admin/4_0_1/ccmsys/a08procl.html

QUESTION 33

Which best describes the auto-attendant in a Cisco Unified Communications environment?

- A. A set of call processing instructions that automatically tell the system what to do when it reaches a particular system ID
- B. A system that automatically allows inside or outside callers to leave voice-mail messages 24 hours, 7 days a week, even when no operator is on duty
- C. A function that greets and guides callers through a telephony system in a friendly and timely manner, allowing them to reach an endpoint, leave messages, or speak to an operator
- D. A function that allows the option of listening to, composing, replying to, forwarding, or deleting calls or voice-mail messages through a website without the need of a live telephone operator

Correct Answer: C

Section: (none)

Explanation

Explanation/Reference:

Cisco Unified Communications Manager Auto-Attendant, a simple automated attendant, allows callers to locate people in your organization without talking to a receptionist. You can customize the prompts that are played for the caller, but you cannot customize how the software interacts with the customer.

Link:

http://www.cisco.com/en/US/docs/voice_ip_comm/cucm/admin/7_0_1/ccmfeat/fsaa.html

QUESTION 34

Which statement about the Cisco Unity Express default AutoAttendant is true?

- A. The default AutoAttendant must be enabled during the Initialization Wizard process otherwise voice-mail services will not function.
- B. Enabling the default AutoAttendant is not mandatory during the Initialization Wizard process.
- C. The default AutoAttendant is enabled by default with the exception of the prompts that must be recorded via AvT.

D. The default AutoAttendant cannot be used as is, and it must be customized for the particular environment it will be used in.

Correct Answer: B

Section: (none)

Explanation

Explanation/Reference:

The AutoAttendant can be configured at a later stage.

QUESTION 35

You have been tasked to configure a Cisco Unity Express system with a voicemail pilot number of 1900, an AutoAttendant with pilot number of 2900, and an Administration via Telephone pilot number of 3900. What is the minimum number of SIP dial peers required?

- A. 1
- B. 2
- C. 3
- D. 4
- E. 5

Correct Answer: A

Section: (none)

Explanation

Explanation/Reference:

QUESTION 36

How can an administrator determine which codec is being used between two endpoints while a call is in progress?

- A. Run the codec trace in Cisco Unified Communication Manager.
- B. Use Cisco Unified Serviceability network trace.
- C. Can only be seen in Cisco SDI traces.
- D. Can only be seen in a sniffer trace.
- E. Press the [?] button twice on one of the IP phones.

Correct Answer: E

Section: (none)

Explanation

Explanation/Reference:

Pressing the button (?) twice in quick succession during an active call. The display shows you receive and transmit statistics for the call

Link:

http://docwiki.cisco.com/wiki/Cisco_Unified_Communications_-_One-Way_Audio

QUESTION 37

An administrator wants to see how long a specific IP phone extension is in use during a given period of time. What CAR tool feature would the administrator use?

- A. System Reports > Traffic > Summary by Phone Number
- B. CDR > Search > By Call Precedence Level
- C. Device Reports > Route Patterns/Hunt Groups > Route and Line Group Utilization

D. User Reports > Top N > By Duration

Correct Answer: A

Section: (none)

Explanation

Explanation/Reference:

Only CAR administrators generate the Traffic Summary by Phone Number report.

The report provides information about the call volume for a period and set of phone numbers that you specify.

Choose System Reports > Traffic > Summary By Phone Number.

Link:

http://www.cisco.com/en/US/docs/voice_ip_comm/cucm/service/8_6_1/car/carsytra.html

QUESTION 38

Which three commands does Cisco recommend to use first when setting up phones for Cisco Unified Communications Manager Express? (Choose three)

- A. Load
- B. Max-dn
- C. Keepalive
- D. Max-ephones
- E. Create cnf files
- F. Telephony-service

Correct Answer: BDF

Section: (none)

Explanation

Explanation/Reference:

Basic Manual CME Setup Using the CLI

Router(config)# **telephony-service** - Enters telephony-service configuration mode.

Router(config-telephony-service)# **max-ephones 24** - Sets the maximum number of supported IP phones.

Router(config-telephony-service)# **max-dn 48** - Sets the maximum number of extensions.

NOTE: The maximum number of phones and extensions is platform and IOS version dependent.

Router(config-telephony-service)# **load 79xx P0xxxxxxxx** - Identifies the Cisco IP phone firmware file to be used by specific Cisco IP phones when they register.

Router(config-telephony-service)# **ip source-address 192.168.31.1** - Identifies the IP address and port number for IP phone registration.

NOTE: The default port is 2000.

Router(config-telephony-service)# **create cnf-files** - Builds the XML configuration files required for Cisco CME phones.

Router(config-telephony-service)# **reset sequence-all** - Resets all phones one at a time.

Router(config-telephony-service)# **exit** - Returns to global configuration mode.

Link:

<http://www.ciscopress.com/articles/article.asp?p=1182471&seqNum=2>

QUESTION 39

What statement about the Cisco Unity Express AIM Module and the Cisco Unity Express Network Module is true?

- A. The AIM module takes up a router slot whereas the network module does not.
- B. The network module takes up a router slot whereas the AIM module does not.
- C. The AIM module runs the Linux OS whereas the network module runs the Cisco IOS software.
- D. The network module requires the Cisco Unified Communications Manager Express to be preconfigured whereas the AIM module does not.

Correct Answer: B

Section: (none)

Explanation

Explanation/Reference:

AIM - AIMS are modules that you install inside a compatible router. Think of an AIM as an internal expansion card like for your computer. You can get AIMS that do different things like VPN acceleration

Network Module - The network module takes up a router slot on the router.

Cisco Access Routers and Cisco Network Modules

The Cisco network modules described in this document are supported on the following Cisco routers:

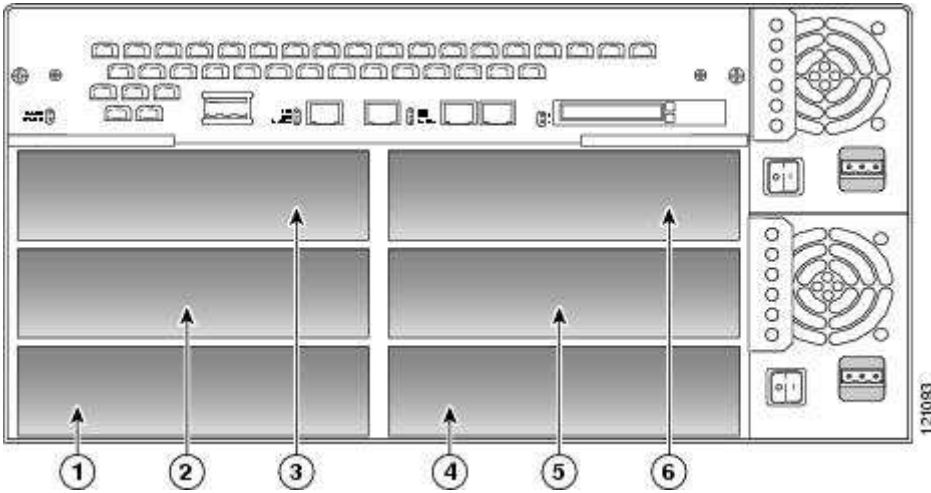
- Cisco 2600 series, including the Cisco 2610, Cisco 2610XM, Cisco 2611, Cisco 2611XM, Cisco 2612, Cisco 2613, Cisco 2620, Cisco 2620XM, Cisco 2621, Cisco 2621XM, Cisco 2650, Cisco 2650XM, Cisco 2651, Cisco 2651XM, and Cisco 2691 routers

Note: References to the Cisco 2600XM routers apply to the following routers: Cisco 2610XM, Cisco 2611XM, Cisco 2620XM, Cisco 2621XM, Cisco 2650XM, and Cisco 2651XM.

- Cisco 2811, Cisco 2821, and Cisco 2851 routers
- Cisco 3600 series, including the Cisco 3620, Cisco 3631, Cisco 3640, Cisco 3640A, Cisco 3661, and Cisco 3662 routers
- Cisco 3700 series, including the Cisco 3725 and Cisco 3745 routers
- Cisco 3800 series, including the Cisco 3825 and Cisco 3845 routers
- Cisco MWR 1941-DC routers
- Cisco 2900 series and Cisco 3900 series routers support some network modules with the addition of a network module adapter. See "Using Network Modules in Service Module Slots" section.

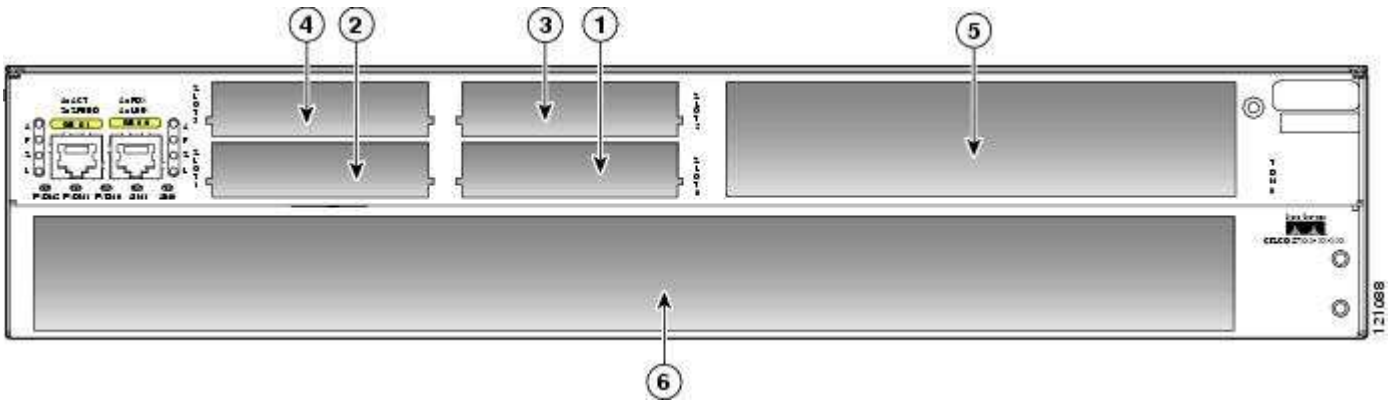
For information on these Cisco routers, see the hardware documentation described in the "Related Documents" section.

Cisco 3660 Router Rear View



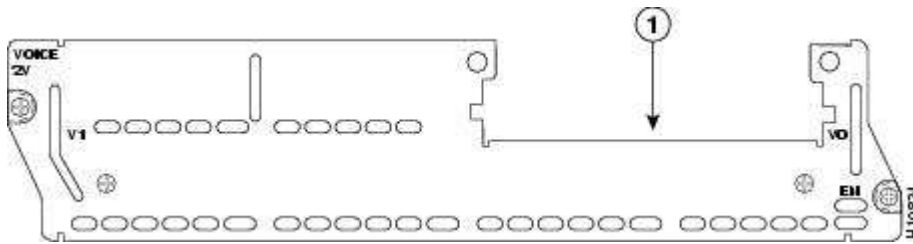
- 1 - Network module slot 2
- 4 - Network module slot 1
- 2 - Network module slot 4
- 5 - Network module slot 3
- 3 - Network module slot 6
- 6 - Network module slot 5

Chassis Slot Locations in Cisco 2851 Routers



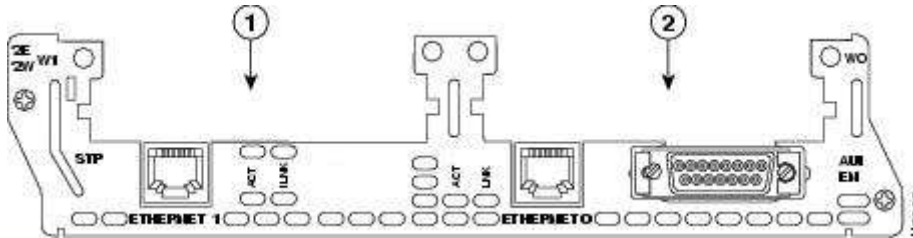
- 1 - High-speed WAN interface card slot 0
- 4 - High-speed WAN interface card slot 3
- 2 - High-speed WAN interface card slot 1
- 5 - Extension module slot
- 3 - High-speed WAN interface card slot 2
- 6 - Network module slot

Sample 1-Slot Network Module



1 - Interface card slot 0

Sample 2-Slot Network Module



1 - Interface card slot 1

2 - Interface card slot 0

Link:

<http://www.cisco.com/en/US/docs/routers/access/interfaces/nm/hardware/installation/guide/OvrNetM.html>

QUESTION 40

The SBB Company is setting up Call Transfer for its Cisco Unified Communications Manager Express solution. The company uses five-digit extensions and would like to be able to transfer calls outside the network to the CEO's home.

The CEO's telephone number is 866-555-2222.

Which configuration command will allow this to occur?

- A. SBB(config-ephone-dn)# transfer-pattern .T
- B. SBB(config-telephony-service)# transfer-pattern .T
- C. SBB(config-ephone-dn)# transfer-pattern51212
- D. SBB(config-ephone-dn)# transfer-pattern 8665552222
- E. SBB(config-telephony-service)# transfer-pattern 86655.....
- F. SBB(config-telephony-service)# transfer-pattern 8665552222

Correct Answer: F

Section: (none)

Explanation

Explanation/Reference:

Configuring CME Transfer Patterns to allow Outside Transfers

CME_Voice(config)# **telephony-service**

CME_Voice(config-telephony)# transfer-pattern ?

WORD digit string pattern for permitted non-local call transfer

CME_Voice(config-telephony)# transfer-pattern 8665552222

Link:

http://books.google.ca/books?id=zHwL_jOhdcgC&pg=PA181&lpg=PA181&dq=cisco+cme+transfer-pattern&source=bl&ots=-SgiK-KtBX&sig=90iMxqU9hmgAVkv_Q5CFc885zol&hl=en&sa=X&ei=e4_8UJLiMMSi2AXykYGyDQ&sqi=2&ved=0C

FsQ6AEwBw#v=onepage&q=cisco%20cme%20transfer-pattern&f=false

QUESTION 41

What is the maximum number of phones are supported on Call Manager Express?

- A. 48
- B. 28
- C. 240
- D. 500

Correct Answer: C

Section: (none)

Explanation

Explanation/Reference:

The maximum number of phones supported depends on the Cisco router you have chosen to use in your CME deployment.

Cisco is referring to the Cisco 3845 router which can support a maximum of 240 IP phones.

Supported Platforms and Memory Specifications: SCCP IP Phones*

Cisco Platform	-	Maximum Total IP Phones
Cisco IAD 2430	-	24
Cisco 1751-V	-	24
Cisco 1760, Cisco 1760-V	-	24
Cisco 261xXM, 262xXM	-	36
Cisco 2650XM, 2651XMx	-	48
Cisco 2691	-	72
Cisco 2801	-	24
Cisco 2811	-	36
Cisco 2821	-	48
Cisco 2851	-	96
Cisco 3725	-	144
Cisco 3745	-	192
Cisco 3825	-	168
Cisco 3845	-	240

Link:

http://www.cisco.com/en/US/docs/voice_ip_comm/cucme/requirements/guide/cme34spc.html

QUESTION 42

In Cisco Unified Communications Manager Native Presence, what two things is a watcher monitoring in real time? (Choose two.)

- A. registration status of a specific IP phone
- B. registration status of the hunt group
- C. registration status of the MGCP gateway
- D. registration status of Cisco Extension Mobility of the IP phone
- E. status of a registered directory number

Correct Answer: AE

Section: (none)

Explanation

Explanation/Reference:

RTMT provide the following reports , Call Activity Status, Device Status, server status, Service Status, Alert

Link:

http://www.cisco.com/en/US/docs/voice_ip_comm/cucm/service/6_0_1/rtmt/rtintro.html

QUESTION 43

Which protocol is used for communication between Cisco Unity Express and Cisco Unified Communications Manager Express?

- A. H.323
- B. G.711
- C. MGCP
- D. Q.931
- E. SIP

Correct Answer: E

Section: (none)

Explanation

Explanation/Reference:

Figure 14-3 Protocols Used Between Cisco Unity Express and Cisco Unified CallManager

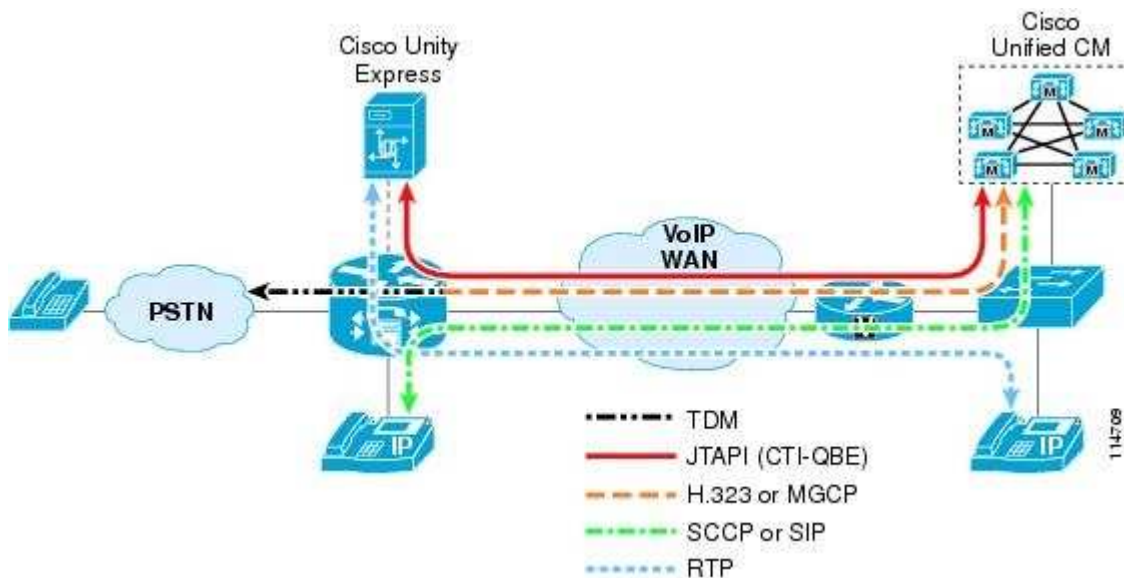


Figure 14-3 illustrates the following signaling and media flows:

- Phones are controlled via SCCP from Cisco Unified CallManager.
- Cisco Unity Express is controlled via JTAPI (CTI-QBE) from Cisco Unified CallManager.
- The Message Waiting Indicator (MWI) on the phone is affected by Cisco Unity Express communicating a change of mailbox content to Cisco Unified CallManager via CTI-QBE, and by Cisco Unified CallManager in turn sending a MWI message to the phone to change the state of the lamp.
- The voice gateway communicates via either H.323, SIP, or MGCP to Cisco Unified CallManager.
- Real-Time Transport Protocol (RTP) stream flows carry the voice traffic between endpoints.

Figure 14-4 shows the protocols involved in the call flow between the SRST router and Cisco Unity Express

when the WAN link is down.

Link:

http://www.cisco.com/en/US/docs/voice_ip_comm/cucm/srnd/4x/42unityx.html

QUESTION 44

Refer to the exhibit. The error message was displayed when the administrator attempted to set the default user password in the user template to four digits long. Which option can rectify this issue?

The screenshot shows a web interface titled "Change Password (Voice Mail)". At the top, there are navigation links: "User Template", "Edit", "Refresh", and "Help". Below this is a "Status" section with a red error icon and the message "Password does not have enough characters". Underneath is a "Choose Password" section with a dropdown menu currently set to "Voice Mail" and a "Save" button. At the bottom, there are two input fields labeled "Voice Mail Password" and "Confirm Password". A large watermark "Lead2pass.com" is visible across the center of the interface.

- A. The minimum password length in Cisco Unity Connection should be at least five digits long.
- B. The password length needs to be configured under the Authentication Rules settings using the Minimum Credential Length configuration field.
- C. The password length needs to be configured under the user template settings using the Minimum Credential Length configuration field.
- D. The password length cannot be adjusted under the user template; the password length can only be adjusted under the individual users.

Correct Answer: B

Section: (none)

Explanation

Explanation/Reference:

A minimum length requirement (as set on the Edit Authentication Rule page, in the Minimum Credential Length field)

- Inclusion of at least one character from each of the following categories: upper-case letter, lowercase letter, number, and symbol (~ ! @ # \$ % ^ & * " ` , . : ; ? - _ () [] < > { } + = / \ |)
- No characters repeated consecutively more than three times (for example, aaaaB1C9 is invalid)
- No inclusion of the alias or name of the administrator

QUESTION 45

What is the correct URL to use to add a user to Cisco Unified Communications Manager Express?

- A. <http://ipaddress/cucme.html>
- B. <https://ipaddress/cue.html>
- C. <https://www.ipaddress/cme.html>
- D. <http://ipaddress/ccme.html>
- E. <http://ipaddress/cme.html>

Correct Answer: D

Section: (none)

Explanation

Explanation/Reference:

CCME – Cisco call manager express

Link:

http://www.cisco.com/en/US/docs/voice_ip_comm/cucme/admin/configuration/guide/cmegui.html#wp1069477

QUESTION 46

Refer to the exhibit. What protocol is being used to send and receive instant messaging between the Cisco Unified Personal Communicator and Cisco Unified Presence?



- A. Enterprise Instant Messaging Protocol
- B. Extensible Messaging and Presence Protocol
- C. SIP
- D. SCCP
- E. CTI

Correct Answer: B

Section: (none)

Explanation

Explanation/Reference:

Cisco Unified Presence consists of many components that enhance the value of a Cisco Unified Communications system.

The main presence component of the solution is the Cisco Unified Presence server, which incorporates the Jabber Extensible Communications Platform and supports SIP/SIMPLE and Extensible Messaging and Presence Protocol (XMPP) for collecting information regarding a user's availability status and communications capabilities

Link:

http://www.cisco.com/en/US/docs/voice_ip_comm/cucm/srnd/8x/presence.html

QUESTION 47

In which location is Cisco Unified Presence enabled for a specific user in Cisco Unified Communications Manager?

- A. User Administration
- B. Application
- C. Advanced Features
- D. Capabilities Assignment
- E. on the IP phone

Correct Answer: D

Section: (none)

Explanation

Explanation/Reference:

Capabilities Assignment allows system administrators to enable the Cisco Unified Presence (CUP) and Cisco Unified Personal Communicator (CUPC) capabilities for users. You must ensure that licenses for CUP and CUPC are available.

Link:

http://www.cisco.com/en/US/docs/voice_ip_comm/cucm/admin/8_6_1/ccmfeat/fslicense.html#wp1158223

QUESTION 48

What is the Cisco Unified Communications Manager implementation of one-way intercom referred to as?

- A. One-Way Intercom monitor
- B. Whisper Intercom
- C. Secure Intercom
- D. Silent monitor

Correct Answer: B

Section: (none)

Explanation

Explanation/Reference:

Whisper intercom means that only one-way audio exists from the caller to the called party. The called party must manually press a key to talk to the caller

Link:

http://www.cisco.com/en/US/docs/voice_ip_comm/cucm/admin/8_6_1/ccmfeat/fsintrcm.html

QUESTION 49

Which protocol should you use to securely access Cisco Configuration Professional?

- A. HTTPS
- B. Telnet
- C. SCP
- D. TLS

Correct Answer: A

Section: (none)

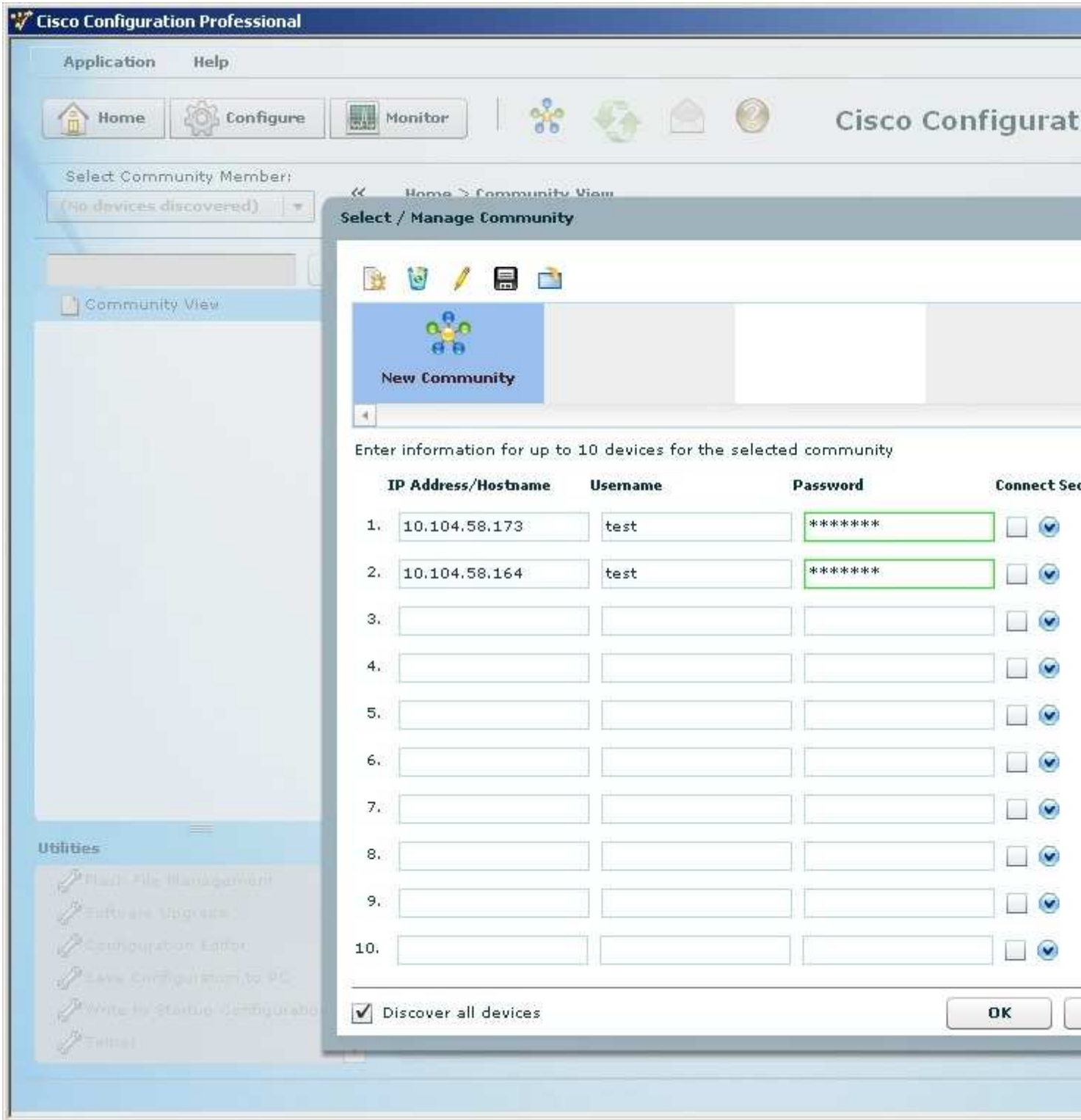
Explanation

Explanation/Reference:

If you want Cisco Configuration Professional (CCP) to connect securely with the device, check the Connect Securely check box.

When you check the Connect Securely check box, HTTPS port 443 and SSH port 22 information is automatically added to the device. To view the port information, click the down-arrow next to the Connect Securely check box.

If you did not check the Connect Securely check box, the HTTP port 80 and Telnet port 23 information is automatically added to the device. To view the port information, click the down-arrow next to the Connect Securely check box.



Link:

http://www.cisco.com/en/US/docs/net_mgmt/cisco_configuration_professional/v2_5/guides/getting_start/creatcom.html

QUESTION 50

Which type of ephone-dn is typically used for intercoms and paging?

- A. dual-line
- B. multiline
- C. single-line
- D. special-line

Correct Answer: C

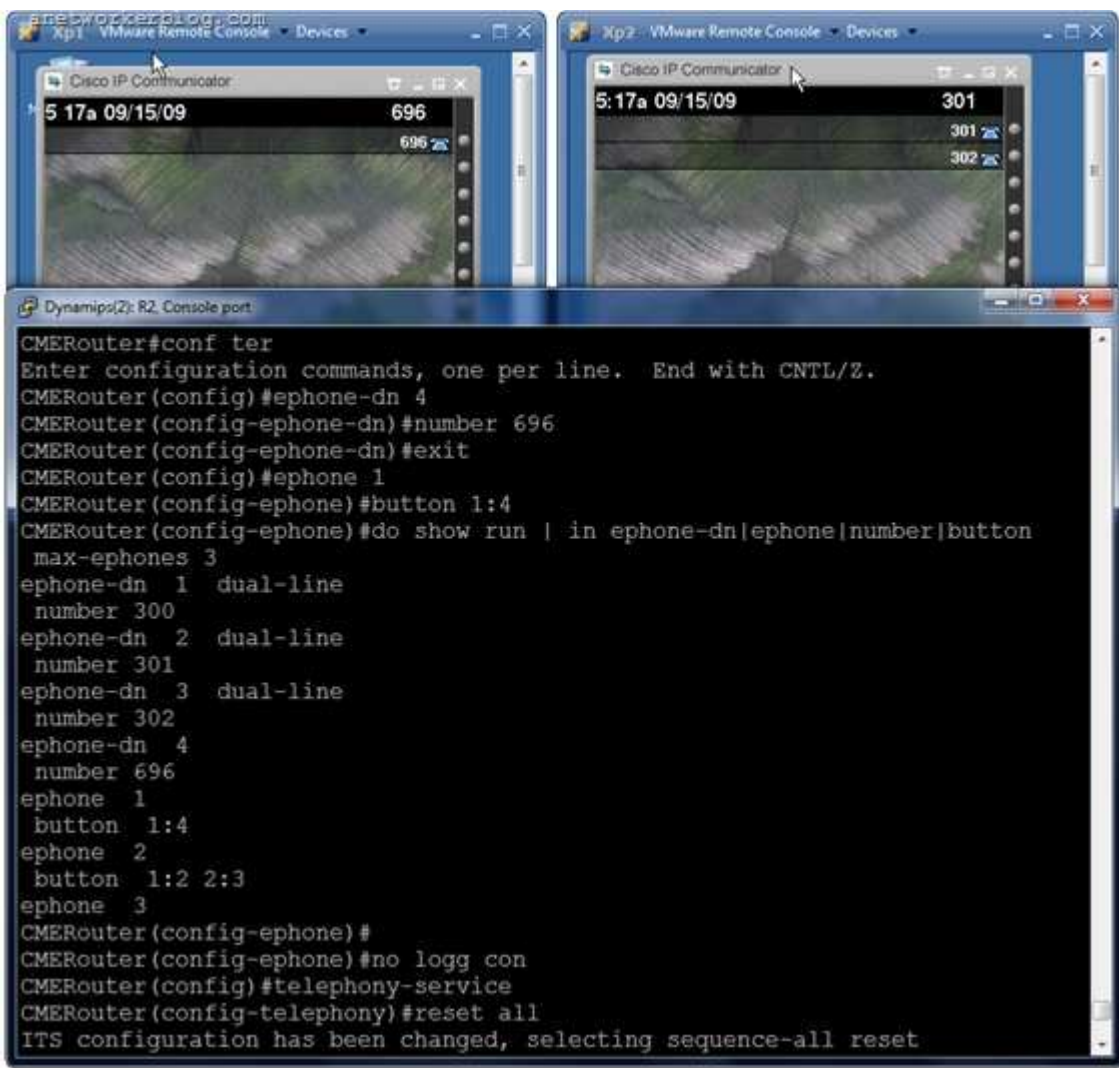
Section: (none)

Explanation

Explanation/Reference:

Single-Line ephone-dn

Only 1 call to or from this ephone-dn can occur at one time, there is no support for call waiting, and is commonly used on paging, intercoms, call parking, MOH feeds, it is important to note that with this ephone-dn we create a virtual voice port.



Dual-line ephone-dn

Using this ephone type we can have 2 calls connections at the same time, since it has 2 channels, it could have 1 number or 2 (– Dual-number ephone-dn – primary and secondary), with this type of ephone you can have features like call waiting, call transfer and conferencing, it is important to know that you should not use dual-line

for call waiting, call transfer or conferencing.



Link:

<http://anetworkerblog.com/2009/09/14/types-of-ephone-dns/>

QUESTION 51

A phone is configured with an ephone-dn number of A100. Which CLI command is used in ephone-dn configuration mode to enable the Intercom feature to dial this phone?

- A. intercom extension A100
- B. intercom number A100
- C. intercom A100 enable
- D. intercom A100

Correct Answer: D

Section: (none)

Explanation

Explanation/Reference:

Configuring Intercom

Intercom configurations are common in traditional phone systems. This feature allows an administrative assistant and executive to work closely together by having a speakerphone “tether” between them.

Figure 7-15 Configuring Call Pickup Groups in CCP

Technically, the way intercom deployments work is through a speed-dial and auto-answer speed-dial configuration. If the administrative assistant presses the button configured as an intercom, it speed dials the executive’s phone, which auto-answers the call on muted speakerphone. To establish two-way communication, the executive deactivates mute (by pressing the Mute button). Understanding this helps make the intercom configuration much clearer.

To configure intercom functionality, you must configure two new ephone-dns, one for each side of the intercom connection. These intercom lines should be assigned a number, just like any other ephone-dn. However, to prevent others from accidentally (or purposely) dialing the intercom and ending up on muted speakerphone for a random IP phone, the number should be something users cannot dial from other IP phones. The configuration in

Example 7-8 Configuring Intercom

```
CME_Voice(config)# ephone-dn 60
CME_Voice(config-ephone-dn)# number A100
CME_Voice(config-ephone-dn)# intercom A101 label “Manager”
CME_Voice(config-ephone-dn)# exit
CME_Voice(config)# ephone-dn 61
CME_Voice(config-ephone-dn)# number A101
CME_Voice(config-ephone-dn)# intercom A100 label “Assistant”
CME_Voice(config-ephone-dn)# exit
CME_Voice(config)# ephone 1
CME_Voice(config-ephone)# button 2:60
CME_Voice(config-ephone)# restart
restarting 0014.1C48.E71A
CME_Voice(config-ephone)# exit
CME_Voice(config)# ephone 2
CME_Voice(config-ephone)# button 2:61
CME_Voice(config-ephone)# restart
restarting 0019.D122.
```

Notice the number assigned to ephone-dn 60 is A100. You cannot dial this number from a Cisco IP Phone keypad, but you can assign it to a speed-dial button. The **intercom** command acts like a speed-dial button on the ephone-dn. In the case of ephone-dn 60, the command **intercom A101** dials the number A101, which is assigned to ephone-dn 61. Because ephone-dn 61 is also configured with the **intercom** command, it auto-answers the incoming call on muted speakerphone. The label syntax allows you to assign a logical name to the speed-dial; otherwise, the A101 or A100 label will show up next to the line button on the phone. There are three other arguments you can use with the **intercom** command to tune the functionality:

- **barge-in**: Automatically places an existing call on hold and causes the intercom to immediately answer.
- **no-auto-answer**: Causes the phone to ring rather than auto-answer on speakerphone.
- **no-mute**: Causes the intercom to answer with unmuted speakerphone rather than muted. Although this is beneficial to allow immediate two-way conversation, you run the risk of one side barging into existing conversations or background noise.

CCP can also configure Intercom functionality. To do this, navigate to **Unified Communications > Telephony Features > Intercom** and click the **Create** button. The configuration window shown in Figure 7-16 allows you to select the user and speed-dial button you want to assign Intercom functionality.

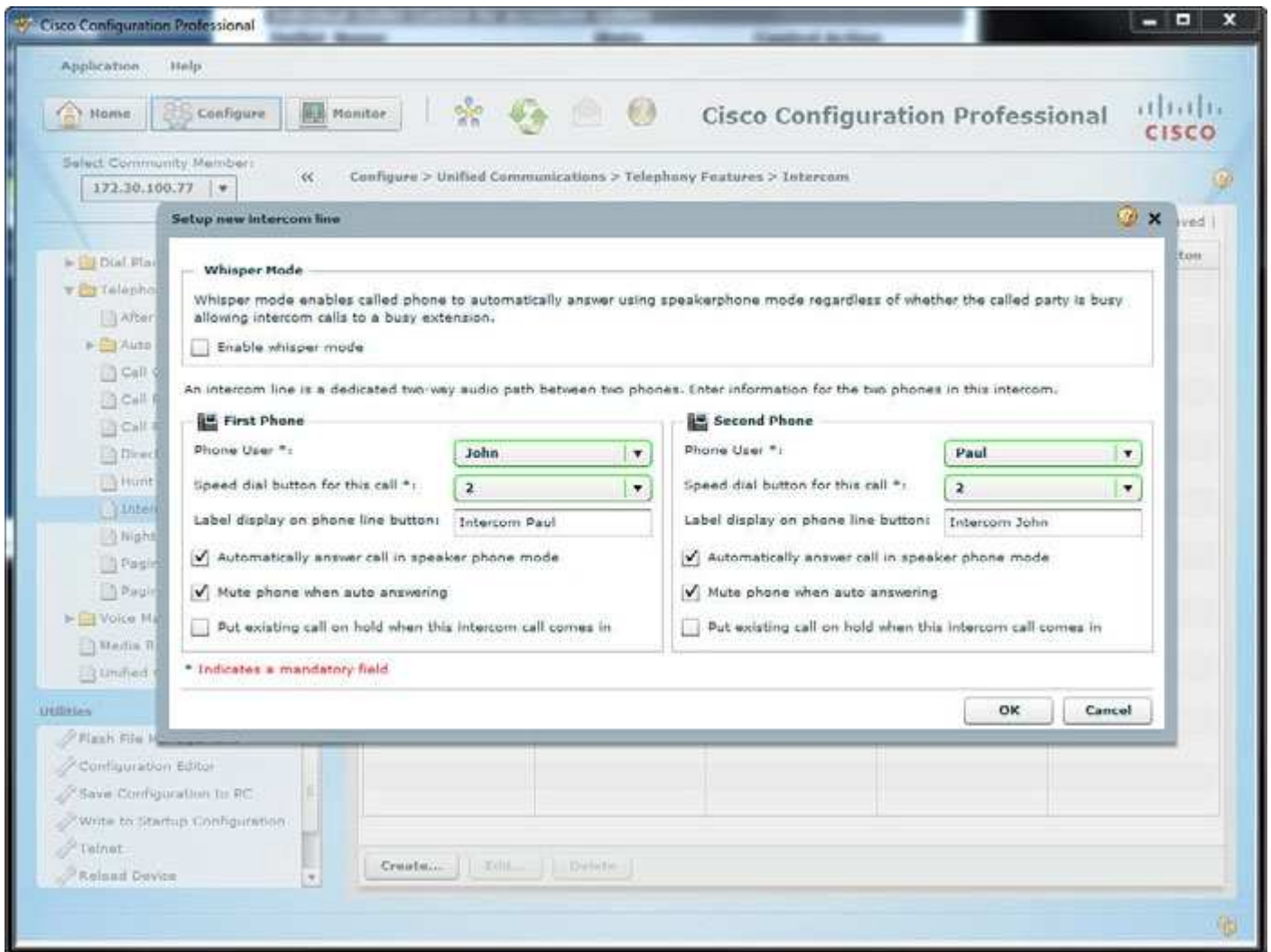


Figure 7-16 Configuring Intercom in CCP

After you click the **Deliver** button, CCP applies the following syntax to the router. Notice that CCP also uses alpha-numeric speed dials to prevent other users from accessing the intercom functionality inadvertently:

```
ephone-dn 5
number A105002
description Intercom
intercom A105102 label "Intercom Paul"
exit
```

```
ephone-dn 4
number A105102
description Intercom
intercom A105002 label "Intercom John"
exit
```

```
ephone-dn 5
name ""
exit
```

```
ephone 2
button 2:5
restart
exit
```

```
ephone-dn 4
name ""
exit
```

```
ephone 3
button 2:4
restart
exit
```

Link: CCNA Voice 640-461 Official Cert Guide

http://books.google.ca/books?id=zHwL_jOhdcgC&pg=PA190&lpg=PA190&dq=A+phone+is+configured+with+an+ephone-dn+number+of+A100.&source=bl&ots=-SgjK0DtAW&sig=554-3vSclgtMPEN-lvSZo5WzR0g&hl=en&sa=X&ei=Kpv8UOqTCrOK2QW07YGICQ&ved=0CDYQ6AEwAg#v=onepage&q=A%20phone%20is%20configured%20with%20an%20ephone-dn%20number%20of%20A100.&f=false

QUESTION 52

When creating a Cisco Unity Connection user template, which element should you configure to automatically play a "This department is closed" message at specific hours?

- A. greeting schedule
- B. extension greetings
- C. schedule
- D. active schedule

Correct Answer: D

Section: (none)

Explanation

Explanation/Reference:

Understanding Time-of-Day Routing

Time-of-Day routing comprises individual time periods that the administrator defines and groups into time schedules. The administrator associates time schedules with a partition. In the Partition Configuration window, the administrator chooses either the time zone of the originating device or any specific time zone for a time schedule. The system checks the chosen time zone against the time schedule when the call gets placed to directory numbers in this partition. The Time Period and Time Schedule menu items exist in the Route Plan menu under the Class of Control submenu. The Partition and Calling Search Space menu items also have moved to the Class of Control submenu.

Time Periods

A time period comprises a start time and end time. The available start times and end times comprise 15-minute intervals on a 24-hour clock from 00:00 to 24:00. Additionally, a time period requires definition of a repetition interval. Repetition intervals comprise the days of the week (for example, Monday through Friday) or a day of the calendar year (for example, June 9).

Examples:

You can define time period weekdayofficehours as 08:00 to 17:00 from Monday to Friday.

You can define time period newyearsday as 00:00 to 24:00 on January 1.

You can define time period noofficehours that has no office hours on Wednesdays. For this time period, the associated partition is not active on Wednesdays.

Note: In defining a time period, the start time must precede (be less than) the end time.

Tip: To define an overnight time span that starts on Monday through Friday at 22:00 and ends at 04:00 the next morning, create two time periods, such as lateevening (from 22:00 to 24:00 on Monday through Friday) and earlymorning (from 00:00 to 04:00 on Tuesday through Saturday). Use the Time Schedule Configuration window to associate the lateevening and earlymorning time periods into a single time schedule that spans the overnight hours.

After the administrator creates a time period, the administrator must associate the time period with a time schedule.

Link:

http://www.cisco.com/en/US/docs/voice_ip_comm/cucm/admin/7_1_2/ccmsys/a03tod.html

QUESTION 53

In a Cisco UCM multisite WAN with centralized call-processing deployment model, what redundancy feature should be configured on remote site routers to provide basic IP telephony services in the event of a WAN outage?

- A. AAR
- B. SRST
- C. CAC
- D. V3PN

Correct Answer: B

Section: (none)

Explanation

Explanation/Reference:

A survivable remote site telephony (SRST) reference comprises the gateway that can provide limited Cisco CallManager functionality when all other Cisco CallManager servers for a device are unreachable.

Typically assigned to device pools, SRST references determine the gateways where calling devices search when they attempt to complete a call if Cisco CallManager is unavailable.

For more detailed information on SRST references, refer to the "Survivable Remote Site Telephony References" section in the Cisco CallManager System Guide.

Link:

http://www.cisco.com/en/US/docs/voice_ip_comm/cucm/admin/4_0_1/ccmcf/b02srst.html

QUESTION 54

Which three options are valid for creating users in Cisco Unity Connection? (Choose three.)

- A. manual creation
- B. bulk using *.csv file
- C. bulk using enterprise parameters
- D. Cisco Unity Connection Serviceability
- E. automatic creation through TUI by users dialing into voice mail
- F. import through Active Directory

Correct Answer: ABF

Section: (none)

Explanation

Explanation/Reference:

Cisco Unity Connection Serviceability, a web-based troubleshooting tool for Cisco Unity Connection.

Link:

http://www.cisco.com/en/US/docs/voice_ip_comm/connection/2x/serviceability/cuc_administration/guide/2xcucservag010.html

QUESTION 55

Refer to the exhibit. The exhibit shows a partial screen shot for a Cisco Unified Client Services Framework device. If the username that is associated with this device is jdoe, what should the device name be?

Phone Type	
Product Type:	Cisco Unified Personal Communicator
Device Protocol:	SIP
Device Information	
⚠ Device is not trusted	
Device Name*	<input type="text"/>
Description	<input type="text"/>
Device Pool*	-- Not Selected --
Common Device Configuration	< None >
Phone Button Template*	-- Not Selected --
Common Phone Profile*	Standard Common Phone Profile
Calling Search Space	< None >
Media Resource Group List	< None >
User Hold MOH Audio Source	< None >
Network Hold MOH Audio Source	< None >

- A. The device name should be JDOEUPC.
- B. The device name should be UPCJDOE.
- C. The device name should be JDOE.
- D. The device name should be UPCCUPC.

Correct Answer: B

Section: (none)

Explanation

Explanation/Reference:

When you create a softphone device for each Cisco Unified Personal Communicator user, you must specify a device name. Make sure that the device name conforms to these guidelines:

Derives from the username.

Starts with UPC.

Contains only uppercase letters, or numerals.

Contains no more than 12 additional characters after UPC.

Table: Username Conversion for Cisco Unified Personal Communicator Softphone Device provides some example device names.

You must create username that do not collide when converted, for example, the usernames john_doe and

john doe convert to the same softphone device name and therefore are said to collide.

Caution! If Cisco Unified Personal Communicator is unable to derive its softphone device name, it cannot properly register and cannot function as expected. You might have to reconfigure a user to use a name other than their normal username to avoid this problem.

QUESTION 56

After an IP phone has been added to a Cisco Unified Communications Manager server, the administrator notices that the phone has a directory number that is not in the number range in use by the organization. He also notes that the directory number is 1000. What is the most likely cause?

- A. The phone number was misconfigured.
- B. The phone may have auto-registered.
- C. The phone is configured on another server.
- D. DHCP gave the phone the wrong directory number
- E. TFTP server is misconfigured

Correct Answer: B

Section: (none)

Explanation

Explanation/Reference:

Organizations have multiple server which work as a single cluster.

Link:

http://www.cisco.com/en/US/docs/voice_ip_comm/cucme/admin/configuration/guide/cmehbasic.html#wp1013086

QUESTION 57

Which action must be taken for Cisco Unified Personal Communicator clients to access Cisco Unity Connection voice mail?

- A. Cisco Unity Connection must be integrated with LDAP.
- B. Cisco Unity Connection must be integrated with Cisco Unified Communications Manager using SIP integration.
- C. A Microsoft Exchange mailbox store must be configured in Cisco Unified Presence.
- D. IMAP must be enabled on Cisco Unity Connection for users that need to access voice mail through Cisco Unified Personal Communicator clients.
- E. Voice mail is automatically enabled for users who log in through Cisco Unified Personal Communicator clients.

Correct Answer: D

Section: (none)

Explanation

Explanation/Reference:

Cisco Unity Connection can receive and process SMTP messages that are generated by IMAP clients, for example, a voice message recorded in a Microsoft Outlook email client by using ViewMail for Outlook.

When an authorized IMAP client tries to send a message to Connection through SMTP, Connection attempts to categorize the message as a voice mail, email, fax, or delivery receipt.

Connection also attempts to map the sender to a user and the message recipients to users or contacts by comparing the SMTP addresses in the message header to its list of SMTP proxy addresses.

Link:

http://www.cisco.com/en/US/docs/voice_ip_comm/connection/7x/administration/guide/7xcucsag175.html

QUESTION 58

In which two ways can end users send an active call to the remote destination? (Choose two)

- A. using softkey in a softkey template
- B. with a feature button in a phone button template
- C. with a service URL associated with a phone service
- D. by dropping the call and calling both parties again
- E. by connecting the originator and the remote destination into a conference call

Correct Answer: AB

Section: (none)

Explanation

Explanation/Reference:

Introducing Mobile Connect and Mobile Voice Access

Mobile Connect allows users to answer incoming calls on the desktop phone or cellular phone, to pick up in-progress calls on the desktop phone or cellular phone without losing the connection, and to originate enterprise calls from the cellular phone.

Mobile Voice Access extends Mobile Connect capabilities by allowing users to originate a call from a remote destination such as a cellular phone as if dialing from the desktop phone.

A remote destination is a phone that is designated as available for Mobile Connect responses and pickup. The user dials Mobile Voice Access from the remote destination.

The user is prompted for the PIN assigned to the user in Cisco Unified Communications Manager. Once authenticated, the user can make a call using the same Mobile Connect features that would be available if the user originated the call from the enterprise desktop phone.

When calling Mobile Voice Access, the system prompts the user for the originating phone number in addition to the PIN if any of the following is true:

- The number the user is calling from is not one of the user's remote destinations.
- The number is blocked by the user or the user's carrier (shown as "Unknown Number").
- The number is not accurately matched in the Cisco Unified Communications Manager database; for example, if the number is 510-666-9999, but it is listed as 666-9999 in the database, or the number is 408-999-6666, but it is entered as 1-408-999-6666 in the database.

If the user incorrectly enters any requested information (such as cellular phone number or PIN) three times in a row, the Mobile Voice Access call disconnects, and the user is locked out for a period of time.

Note: Existing cellular phones, including Code Division Multiple Access (CDMA) and Global System for Mobile Communications (GSM) phones can be used for Mobile Connect and Mobile Voice Access. In some cases, however, it may be necessary to modify timer settings in Cisco Unified Communications Manager to assure compatibility. See the "Adding Remote Destinations" section.

Link:

http://www.cisco.com/en/US/docs/voice_ip_comm/cucm/admin/6_0_1/ccmfeat/fsmobmgr.html

QUESTION 59

How can visibility of the presence-enabled speed dials be limited?

- A. presence groups
- B. Subscribe CSS
- C. calling search space
- D. partition

Correct Answer: A

Section: (none)

Explanation

Explanation/Reference:

Unified CM Presence Groups

The second aspect of the presence policy for Unified CM is presence groups. Devices, directory numbers, and users can be assigned to a presence group, and by default all users are assigned to the Standard Presence Group. A presence group controls the destinations that a watcher can monitor, based on the user's association with their defined presence group (for example, Contractors watching Executives is disallowed, but Executives watching Contractors is allowed). The presence group user setting applies for originating subscriptions when the user is logged in to the device via Extension Mobility or when the user is administratively assigned to the device.

When multiple presence groups are defined, the Inter-Presence Group Subscribe Policy service parameter is used. If one group has a relationship to another group via the Use System Default setting rather than being allowed or disallowed, this service parameter's value will take effect. If the Inter-Presence Group Subscribe Policy service parameter is set to Disallowed, Unified CM will block the request even if the SUBSCRIBE calling search space allows it. The Inter-Presence Group Subscribe Policy service parameter applies only for presence status with call history lists and is not used for BLF speed dials.

Presence groups can list all associated directory numbers, users, and devices if you enable dependency records. Dependency records allow the administrator to find specific information about group-level settings. However, use caution when enabling the Dependency Record Enterprise parameter because it could lead to high CPU usage.

Link:

http://www.cisco.com/en/US/docs/voice_ip_comm/cucm/srnd/8x/presence.html

QUESTION 60

What is a benefit of using FRF.12 in a Frame Relay network?

- A. provides a Layer 3 mechanism for reducing latency in the network
- B. fragments packets into equal sizes to reduce fixed-network delay
- C. reduces delay and jitter by expediting the transfer of smaller frames through the hardware transmit queue
- D. eliminates the need for prioritization of delay-sensitive traffic

Correct Answer: B

Section: (none)

Explanation

Explanation/Reference:

The FRF.12 Implementation Agreement allows long data frames to be fragmented into smaller pieces. This process allows real-time traffic and non-real-time traffic to be carried together on lower-speed links without causing excessive delay to the real-time traffic.

Link:

http://www.cisco.com/en/US/docs/ios/12_1t/12_1t2/feature/guide/dtfragsw.html#wp1015329

QUESTION 61

All users report that when they press the Messages button on their IP phone, nothing happens. What is the most likely cause of this issue?

- A. Cisco Unity Connection is not configured.
- B. The default Voicemail Profile does not have a Pilot number configured.
- C. The Voicemail Pilot does not have the VoiceMail Profile configured.

- D. The Integrated Service Engine is offline.
- E. The Voicemail Pilot is incorrect.

Correct Answer: B

Section: (none)

Explanation

Explanation/Reference:

The voice-mail pilot number specifies the directory number that you dial to access your voice messages. Cisco Unified Communications Manager automatically dials the voice-messaging number when you press the messages button on your phone.

Each voice-mail pilot number can belong to a different voice-messaging system

Link:

http://www.cisco.com/en/US/docs/voice_ip_comm/cucm/admin/6_0_1/ccmsys/a06vmccm.html#wp1043604

QUESTION 62

In which scenario is it possible to have the same directory number configured on two different lines or phones and not be a shared line?

- A. directory number assigned to different partitions
- B. directory number assigned to the same partition
- C. directory number assigned to different calling search spaces
- D. directory number assigned to the same calling search space

Correct Answer: A

Section: (none)

Explanation

Explanation/Reference:

You can set up one or more lines with a shared-line appearance. A Cisco Unified Communications Manager system considers a directory number to be a shared line if it appears on more than one device in the same partition.

Link:

http://cisco.biz/en/US/docs/voice_ip_comm/cucmbe/admin/8_6_1/ccmsys/a03dn.html#wp1100362

QUESTION 63

Which type of single switch port can support data and voice VLANs and is recommended for Cisco Unified IP phones?

- A. multiflex port
- B. trunk port
- C. access port
- D. ISL trunking port

Correct Answer: C

Section: (none)

Explanation

Explanation/Reference:

A port on a Cisco switch is either an access port or a trunk port. Access ports belong to a single VLAN and do not provide any identifying marks on the frames that are passed between switches

Link:

http://www.cisco.com/en/US/products/hw/switches/ps4324/products_configuration_example09186a0080722cdb.shtml

QUESTION 64

Refer to the exhibit. Is this configuration correct and is the T1 operational? If not, what is the cause?

```
HQ#sh isdn stat
Global ISDN Switchtype = primary-ni

%Q.931 is backhauled to CCM MANAGER 0x0003 on DSL 0. Layer 3 output may not apply

ISDN Serial0/0/0:23 interface
  dsl 0, interface ISDN Switchtype = primary-ni
  L2 Protocol = Q.921 0x0000 L3 Protocol(s) = CCM MANAGER 0x0003
  Layer 1 Status:
    ACTIVE
  Layer 2 Status:
    TEI = 0, Ces = 1, SAPI = 0, State = TEI_ASSIGNED
  Layer 3 Status:
    0 Active Layer 3 call(s)
  Active dsl 0 CCBS = 0
  The Free Channel Mask: 0x807FFFFFFF
  Number of L2 Discards = 0, L2 Session ID = 3
  Total Allocated ISDN CCBS = 0
```

```
card type t1 0 0
enable password cisco
!
isdn switch-type primary-ni
!
controller T1 0/0/0
cablelength short 110
pri-group timeslots 1-24 service mgcp
!
interface Serial0/0/0:23
no ip address
encapsulation hdlc
isdn switch-type primary-ni
isdn incoming-voice voice
no cdp enable
!
```

- A. Yes, the configuration is correct and it is operational.
- B. No, the configuration is incorrect and the T1 is not operational because MGCP is not the proper service statement.
- C. No, the configuration is incorrect and the T1 is not operational because, the isdn switch-type is incorrect.
- D. No, the configuration is incorrect and the T1 is not operational because the isdn-bind-13 ccm-manager command is missing from the serial0/0/0:23 interface.
- E. No, the configuration is incorrect and the T1 is not operational because the no ip address command is applied to the serial0/0/0:23 interface.

Correct Answer: D

Section: (none)

Explanation

Explanation/Reference:

To bind Layer 3 of the ISDN PRI interface of the Media Gateway Control Protocol (MGCP) voice gateway to the Cisco CallManager for PRI Q.931 signaling backhaul support, use the **isdn bind13 ccm-manager** command in interface configuration mode

Link:

http://www.cisco.com/en/US/docs/ios/12_3t/voice/command/reference/vrht_i2_ps5207_TSD_Products_Command_Reference_Chapter.html#wp1094910

QUESTION 65

Which network component would cause variable network delay?

- A. dejitter buffer
- B. DSP delay
- C. processing delay
- D. serialization delay
- E. propagation delay

Correct Answer: C

Section: (none)

Explanation

Explanation/Reference:

Processing Delay:

In a network based on packet switching, processing delay is the time it takes routers to process the packet header. Processing delay is a key component in network delay.

Propagation delay:

Propagation delay is a technical term that can have a different meaning depending on the context. It can relate to networking, electronics or physics.

In general it is the length of time taken for the quantity of interest to reach its destination

Link:

http://en.wikipedia.org/wiki/Propagation_delay

QUESTION 66

Which headers are compressed when cRTP is used?

- A. the UDP and RTP headers
- B. the IP header only
- C. the TCP header only
- D. the RTP header only
- E. the IP, UDP, RTP headers and the first byte of the payload
- F. the IP, UDP, and RTP headers

Correct Answer: F

Section: (none)

Explanation

Explanation/Reference:

Compresses a header at various layers of the Open System Interconnection (OSI) reference model. Examples include Transmission Control Protocol (TCP) header compression, compressed RTP (cRTP), and compressed Internet Protocol/User Datagram Protocol (IP/UDP).

QUESTION 67

Which statement about Cisco Unity Connection user templates is true?

- A. Changes in user templates affect only new users to be created.
- B. Changes in user templates affect only existing users.
- C. Changes in user templates affect new and existing users.

D. Changes in user templates have no impact on users unless those users are imported through Active Directory.

Correct Answer: A

Section: (none)

Explanation

Explanation/Reference:

Changes to the template will not affect any currently existing user accounts that were based on the template when they were created.

Changes to a user template affect only those accounts that are created subsequent to the changes.

QUESTION 68

When adding an IP phone to a Cisco Unified Communications Manager system, what two choices does an administrator have to add the phone to the system? (Choose two.)

- A. auto-registration
- B. FHSS provisioning
- C. IP phone configuration assistant
- D. manual provisioning
- E. Cisco Unified Serviceability

Correct Answer: AD

Section: (none)

Explanation

Explanation/Reference:

You can automatically add phones that support either SCCP or SIP to the Cisco Unified Communications Manager database by using autoregistration, manually by using the phone configuration windows, or in groups with the Bulk Administration Tool (BAT). If you do not use autoregistration, you must manually add phones to the Cisco Unified Communications Manager database.

QUESTION 69

What would be the result if a user logs into Cisco Extension Mobility using a device profile that is not subscribed to the Cisco Extension Mobility Service?

- A. Normal Cisco Extension Mobility operation will occur.
- B. The user will not be able to log into Cisco Extension Mobility on the phone.
- C. The phone will not use the correct device profile.
- D. The user will not be able to log out of Cisco Extension Mobility on the phone.
- E. The phone will reboot continuously.

Correct Answer: D

Section: (none)

Explanation

Explanation/Reference:

User Device Profile

You can assign a user device profile to a user so that when a user logs in or out of a device, the corresponding user device profile is loaded onto that device. Devices can adopt user device profiles when a user logs into a device or when there is no user logged in. You can modify, or delete a user device profile in the Cisco CallManager administration pages.

Logging In and Logging Out

When a user logs in, the current configuration of a device is replaced by a particular user device profile.

When a user logs out, the current configuration of a device (the user default device profile) is replaced by the logout profile.

Cisco CallManager Extension Mobility supports only one login at a time on a device. Subsequent logins will fail.

You cannot log out of a device which does not have anyone logged in.

Link:

http://www.cisco.com/en/US/products/sw/voicesw/ps556/products_administration_guide_chapter09186a0080153e60.html

QUESTION 70

Which configuration must be configured on the Cisco Unified Communications Manager Express router prior to Graphical Administration?

```
ip http server

telephony-service
 web admin system name admin password cisco
 dn-webedit
 time-webedit
```

A.

```
ip http server

telephony-service
 web admin system name admin password cisco
 dn-webedit
 time-webedit
```

B.

```
ip http server
ip http path flash:path

telephony-service
 web admin system name admin password cisco
 dn-webedit
 time-webedit
```

C.

```
ip https server
ip https path flash:path

telephony-service
 web admin system name admin password cisco
 dn-webedit
 time-webedit
```

D.

```
ip http server
ip http path flash:path

telephony-service
 dn-webedit
 time-webedit
```

Correct Answer: B

Section: (none)

Explanation

Explanation/Reference:

To enable the HTTP server, and specify the path to files for the GUI and a method of user authentication for security, perform the following steps. The HTTP server on a router is disabled by default.

SUMMARY STEPS

- 1.enable
- 2.configure terminal
- 3.ip http server
- 4.ip http path flash:
- 5.ip http authentication {aaa | enable | local | tacacs}
- 6.exit

To enable the HTTP server, and specify the path to files for the GUI and a method of user authentication for security, perform the following steps. The HTTP server on a router is disabled by default.

QUESTION 71

On an IP phone line appearance, for which two purposes is the Display parameter designed to be used? (Choose two.)

- A. display a number other than the directory number of the IP phone
- B. display a name instead of the directory number of the IP phone
- C. internal caller ID
- D. external caller ID
- E. full directory number ID for outgoing calls

Correct Answer: BC

Section: (none)

Explanation

QUESTION 72

Users report that when they press the Messages button on their phones, they often get a busy tone. Which option can rectify this issue?

- A. Disable the Callers Can Edit Messages check box under the Message settings
- B. The CSS for the phones does not contain the voice-mail port partitions
- C. The CSS for the phone does not contain the voice-mail pilot partition.
- D. Precede all Cisco Unity Connection greetings to announce that each message is limited to 90 seconds long to free up voice-mail ports.

Correct Answer: D

Section: (none)

Explanation

Explanation/Reference:

The key term is "they often get a busy tone"....but not always

Answer "C" would indicate that some phones would not be able to reach voicemail at all, not sometimes...

Answer "D" would indicate that users are experiencing intermittent congestion reaching voicemail during busy periods, and limiting the message length to 90secs would reduce this problem but not fix it (may look like its fixed short term)...really need to buy more voicemail ports (probably not cheap) or educate users to be brief

But then if people are leaving messages >90secs, you'd need a coffee (or 3) to stay awake

Link:

<http://www.techexams.net/forums/ccna-voice/76886-help-cisco-640-461-preparation-exam.html>

QUESTION 73

Which user parameter is used as a password when connecting through the telephony user interface?

- A. PIN
- B. Passphrase
- C. PassKey
- D. Key ID

Correct Answer: A

Section: (none)

Explanation

Explanation/Reference:

Mobility — In the past, your phone has been that object on your desk. Now your phone number is your key to mobility. With Extension Mobility, you can log in to any Cisco VoIP phone on campus using your PIN and access the information you've configured on your web portal.

With Mobility, you can pass a call from your desk phone to your cell phone (and back). With Enable Mobile Connect, you can set up your mobile phone (or another target number) to ring simultaneously with your desk phone.

What to set up on the web portal — Most of us are used to using our desk phone without interacting with our computer. Your new Cisco VoIP phone works with a web portal to enable more options than you could have with the phone alone.

On the portal, you can change your PIN, set up mobility, speed dials, Personal Address Book, search Stanford Directory — and more

Link:

<http://itservices.stanford.edu/service/phone/departement/voip/cisco/userguides/deskphone>

QUESTION 74

Which Cisco Unified Personal Communicator mode should you use when connecting via an existing IP phone?

- A. office mode
- B. deskphone mode
- C. softphone mode
- D. IP phone mode

Correct Answer: B

Section: (none)

Explanation

Explanation/Reference:

Desk Phone Control and the CTI Connection Failures

The CTI gateway provides desk phone control (phone-association mode) to Cisco Unified Personal Communicator users. You must specify CTI gateway server names, addresses, ports, and protocol types on Cisco Unified Presence so that the information required to reach the CTI gateway server can be downloaded when the user logs in to Cisco Unified Personal Communicator.

If the CTI connection to Cisco Unified Communications Manager is lost while Cisco Unified Personal Communicator is operating in desk phone mode, the application tries to reestablish the connection to the primary and then to the backup servers. Connection attempts continue on a round-robin basis, beginning again with the primary server. Successive attempts to reconnect to a server occur at intervals of 4, 8, 16, 32, and 60

seconds (maximum) until a connection is re-established.

Link:

http://www.cisco.com/en/US/docs/voice_ip_comm/cups/8_6/english/install_upgrade/deployment/guide/dgclient.html#wp1100542

QUESTION 75

Which license capability must be enabled for Cisco Unified Presence to work with a specific user?

- A. Enable Cisco Unified Presence
- B. Enable Cisco Unified Presence Server
- C. Enable Cisco Unified Presence Communicator
- D. Enable Extensible Messaging and Presence Protocol

Correct Answer: A

Section: (none)

Explanation

Explanation/Reference:

Requirements

CUP 8.x

Obtaining a License File

When you place an order for Cisco software, Cisco provides a Product Authorization Key (PAK). The PAK provides the software activation key and the license file.

- Obtain the MAC address or the License MAC value of the server for which you are requesting the licenses.
- Obtain a valid email address.
- Determine the number of device licenses that you need by using the license unit calculator in Cisco Unified Communications Manager Administration (System > Licensing > License Unit Calculator).

Step 1 - Register the PAK that you received using the License Registration web tool that is provided on CCO.

Step 2 - Enter the following information:

- The MAC address or the License MAC value of the server for which you are requesting the licenses.
- Your valid email address.
- The number of servers and devices for which you want licenses.
- The server type, physical server or VMware.
- The install type, fresh install or upgrade.

Step 3 Upload the license file to Cisco Unified Communications Manager.

Link:

<https://supportforums.cisco.com/docs/DOC-26718>

QUESTION 76

Which web-based tool is used to access Cisco Unified Communications Manager CDR Analysis and Reporting?

- A. Cisco Unified CDR Administration
- B. Cisco Unified CM Administration
- C. Cisco Unified Reporting
- D. Cisco Unified Serviceability

Correct Answer: D

Section: (none)**Explanation****Explanation/Reference:****Cisco Unified Serviceability**

Administrators can use the Cisco Unified Serviceability web-based tool to troubleshoot problems with the Cisco Unified Communications Manager system. Cisco Unified Serviceability provides the following services:

- Saves Cisco CallManager services alarms and events for troubleshooting and provides alarm message definitions.
- Saves Cisco CallManager services trace information to various log files for troubleshooting. Administrators can configure, collect, and view trace information.
- Monitors real-time behavior of the components in a Cisco Unified Communications Manager system.
- Generates reports for Quality of Service, traffic, and billing information through Cisco CDR Analysis and Reporting (CAR) application.
- Provides feature services that you can activate, deactivate, and view through the Service Activation window.
- Provides an interface for starting and stopping feature and network services.
- Archives reports that are associated with Cisco Unified Serviceability tools.
- Allows Cisco Unified Communications Manager to work as a managed device for SNMP remote management and troubleshooting.
- Monitors the disk usage of the log partition on a server (or all servers in the cluster).

To access Serviceability from the Cisco Unified Communications Manager Administration window, choose Cisco Unified Serviceability from the Navigation drop-down list box that displays in the upper, right corner of the window and click Go.

For more information, refer to the Cisco Unified Serviceability Administration Guide.

Link:

http://www.cisco.com/en/US/docs/voice_ip_comm/cucm/admin/8_0_1/ccmsys/a10tools.html

QUESTION 77

When generating QoS reports in the CAR tool, what two parameters are valid for report generation? (Choose two.)

- A. route lists
- B. route patterns/hunt pilots
- C. route groups
- D. gateway types
- E. partitions
- F. IP phone directory numbers

Correct Answer: BD

Section: (none)

Explanation

Explanation/Reference:**Configuring Route Pattern and Hunt Pilot Device Reports**

Only CAR administrators generate the Route/Hunt List Utilization report. The Route/Hunt List Utilization report provides an estimate of the maximum utilization percentage of the route/hunt list (cumulative utilization of all the gateways under the route/hunt list) for the period and not the exact utilization. The system calculates the cumulative utilization of all the gateways under the route lists and all the lines under the hunt lists.

You can examine the usage based on each hour of a day or on a specified number of days for each week or month. Reports generate for each of the selected route/hunt lists.

You can either view reports that the system automatically generates or generate new reports. Only CAR administrators can schedule reports for automatic generation.

Link:

http://www.cisco.com/en/US/docs/voice_ip_comm/cucm/service/8_6_1/car/cardvrou.html

Configuring QoS System Reports

CAR provides reporting capabilities for three levels of users:

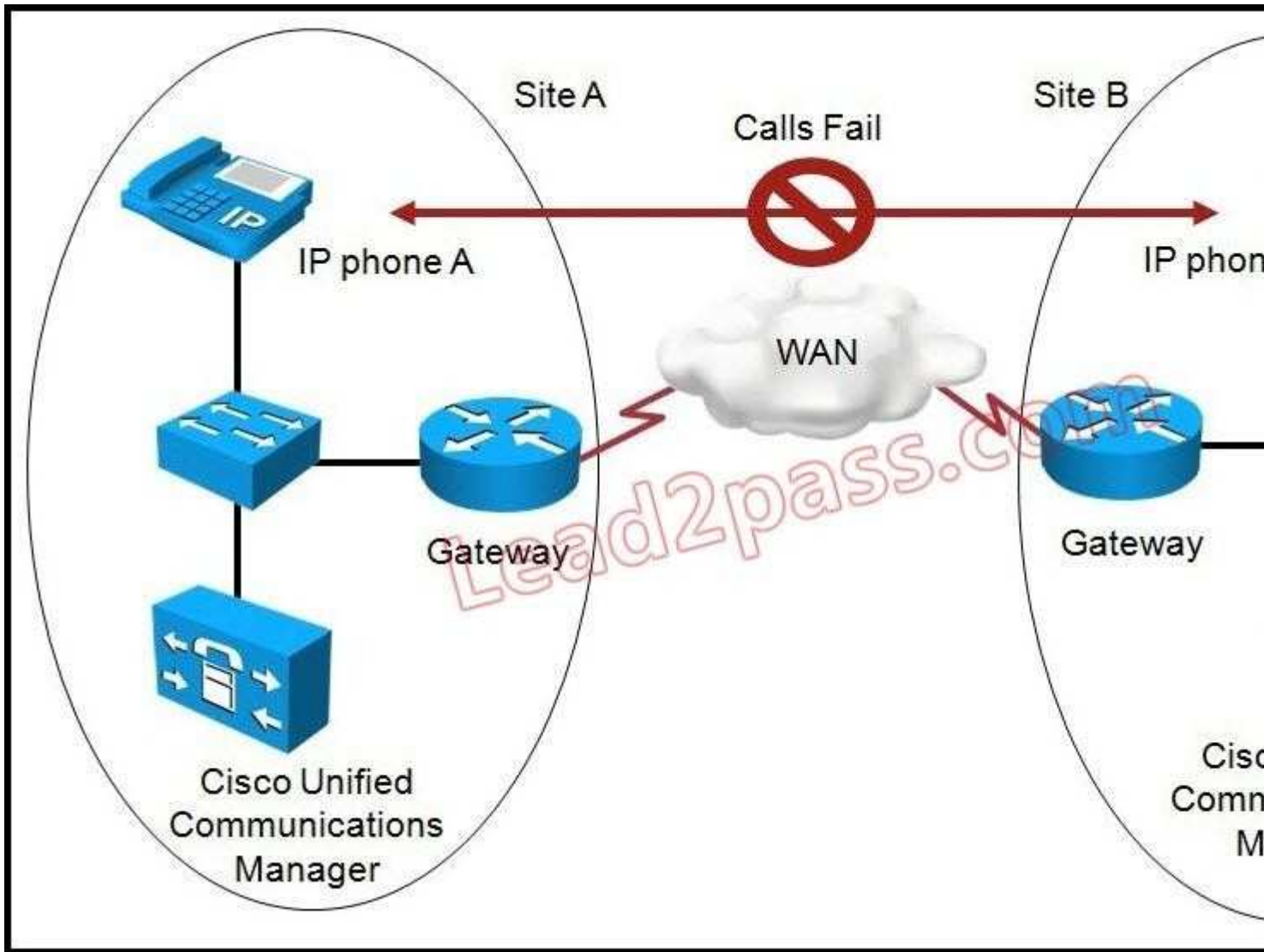
- Administrators—Generate system reports to help with load balancing, system performance, and troubleshooting.
- Managers—Generate reports for users, departments, and QoS to help with call monitoring for budgeting or security purposes and for determining the voice quality of the calls.
- Individual users—Generate a billing report for calls by each user.

Link:

http://www.cisco.com/en/US/docs/voice_ip_comm/cucm/service/7_0_1/car/carsyqos.html

QUESTION 78

Refer to the exhibit. The user of IP phone A has opened a trouble ticket stating that he cannot call IP phone B. Where is the best place to start troubleshooting this issue?



- A. IP phone B
- B. the Cisco Unified Communication Manager system in site A
- C. IP phone A
- D. IP phone A user
- E. the local VoIP voice gateway of IP phone B
- F. IP phone B user

Correct Answer: D

Section: (none)

Explanation

Explanation/Reference:

Always start with the user as he/she could be doing something wrong, i.e. pressing the wrong key sequence.

QUESTION 79

Which three characteristics are associated with data?

- A. Greedy
- B. TCP retransmits

- C. UDP priority
- D. delay sensitive
- E. drop insensitive
- F. benign
- G. benign or greedy.

Correct Answer: BEG

Section: (none)

Explanation

Explanation/Reference:

Data traffic always acknowledge TCP packets for confirm delivery E2E. It is not drop sensitive because it's not real time traffic.

QUESTION 80

Which command is useful to see if network layer information is being received at a PSTN gateway?



<http://www.gratisexam.com/>

- A. show gateway status
- B. show isdn q931
- C. show ccm-manager status
- D. show isdn status
- E. show isdn q921

Correct Answer: D

Section: (none)

Explanation

Explanation/Reference:

Using the show isdn status Command

The **show isdn status** command displays a summary of all ISDN interfaces. It also displays the status of Layers 1, 2, and 3. Complete the following steps to check the status of the layers:

1. Verify that Layer 1 is in the ACTIVE state. The status of Layer 1 should always be ACTIVE unless the E1 is down.

If the **show isdn status** command output indicates that Layer 1 is DEACTIVATED, then there is a problem with the physical connectivity of the E1 line. If the line is administratively down, use the **no shutdown** command to restart the interface.

2. Ensure that Layer 2 is in the MULTIPLE_FRAME_ESTABLISHED state. This is the desired state for Layer 2, indicating that Layer 2 frames are being exchanged and Layer 2 initialization has finished.

If Layer 2 is not in the MULTIPLE_FRAME_ESTABLISHED state, use the **show controller e1 EXEC** command to diagnose the problem. For more information, see the E1 Alarm Troubleshooting document.

Since the **show isdn status** command displays a summary of the current status, it is possible that Layer 2 is bouncing up and down despite indicating a MULTIPLE_FRAME_ESTABLISHED state. Use the **debug isdn**

q921 command to verify that Layer 2 is stable.
Following is an example of **show isdn status** output:

```
bru-nas-03#show isdn status
Global ISDN Switchtype = primary-net5
ISDN Serial0:15 interface
    dsl 0, interface ISDN Switchtype = primary-net5
    Layer 1 Status:
        ACTIVE
    Layer 2 Status:
        TEI = 0, Ces = 1, SAPI = 0, State = MULTIPLE_FRAME_ESTABLISHED
    Layer 3 Status:
        5 Active Layer 3 Call(s)
    Activated dsl 0 CCBs = 5
        CCB:callid=7D5, sapi=0, ces=0, B-chan=9, calltype=DATA
        CCB:callid=7D6, sapi=0, ces=0, B-chan=10, calltype=DATA
        CCB:callid=7DA, sapi=0, ces=0, B-chan=11, calltype=DATA
        CCB:callid=7DE, sapi=0, ces=0, B-chan=1, calltype=DATA
        CCB:callid=7DF, sapi=0, ces=0, B-chan=2, calltype=DATA
    The Free Channel Mask: 0xFFFF78FC
ISDN Serial1:15 interface
    dsl 1, interface ISDN Switchtype = primary-net5
    Layer 1 Status:
        ACTIVE
    Layer 2 Status:
        TEI = 0, Ces = 1, SAPI = 0, State = TEI_ASSIGNED
    Layer 3 Status:
        0 Active Layer 3 Call(s)
    Activated dsl 1 CCBs = 0
    The Free Channel Mask: 0xFFFF7FFF
    Total Allocated ISDN CCBs = 5
```

Notice that E1 0 (whose D-channel is Serial 0:15) has Layer 1 as ACTIVE and Layer 2 as MULTIPLE_FRAME_ESTABLISHED indicating that the signaling channel is functioning correctly and is exchanging Layer 2 frames with the Telco switch. The D-channel (Serial1:15) for E1 1 has Layer 1 ACTIVE, but Layer 2 is TEI_ASSIGNED. This indicates that the PRI is not exchanging Layer 2 frames with the switch. Use the **show controller e1 x** command to troubleshoot. Refer to the E1 Troubleshooting flowchart for more information.

Link:

http://www.cisco.com/en/US/tech/tk713/tk628/technologies_tech_note09186a00800f9e3a.shtml

QUESTION 81

Which device would allow you to place calls from a Cisco Unified Communications Manager that is configured with SCCP phone to a Cisco Unified communication Manager Express that is configured with SIP phones?

- A. gatekeeper
- B. gateway
- C. H.323 trunk
- D. SIP trunk
- E. Cisco Unified Border Element

Correct Answer: A

Section: (none)

Explanation

Explanation/Reference:

Gatekeeper

Cisco gatekeepers are used to group gateways into logical zones and perform call routing between them.

Gateways are responsible for edge routing decisions between the Public Switched Telephone Network (PSTN) and the H.323 network.

Cisco gatekeepers handle the core call routing among devices in the H.323 network and provide centralized dial plan administration.

Without a Cisco gatekeeper, explicit IP addresses for each terminating gateway would have to be configured at the originating gateway and matched to a Voice over IP (VoIP) dial-peer.

With a Cisco gatekeeper, gateways query the gatekeeper when trying to establish VoIP calls with remote VoIP gateways.

Link:

http://www.cisco.com/en/US/tech/tk1077/technologies_tech_note09186a00800a8928.shtml

Cisco Unified Border Element

The Cisco® Unified Border Element (SP Edition) is a high-scale, carrier-grade Session Border Controller (SBC), which is integrated into Cisco routing platforms and can use a huge number of router functions to provide a very feature-rich and intelligent SBC application. In Cisco earlier known as Session Border Controller, Cisco Unified Border Element (SP Edition) provides a network-to-network demarcation interface for signaling interworking, media interworking, address and port translations, billing, security, quality of service, call admission control, and bandwidth management.

Link:

<http://www.webtorials.com/content/2009/07/cisco-unified-border-element.html>

QUESTION 82

Which four characteristics are associated with video? (Choose four)

- A. greedy
- B. TCP retransmits
- C. UDP priority
- D. delay sensitive
- E. drop sensitive
- F. benign
- G. bursty

Correct Answer: CDEG

Section: (none)

Explanation

Explanation/Reference:

Video traffic uses UDP packets for network transfer and normally they are sensitive to delay (Latency), drop (Packet drops) and Jitter because it's a real time traffic. Video traffic can be bursty as well.

QUESTION 83

Can an IP phone be configured so that calls to that IP phone can be rerouted if the IP phone become unregistered?

- A. Yes. configuring call forward All provides this coverage
- B. No. if a phone is unregistered, the re-order tone is played to the caller
- C. Yes. configuring Call Forward Busy Internal provides this coverage
- D. Yes. configuring Call Forward Unregistered provides this coverage
- E. No. the call is dropped at the gateway

Correct Answer: D

Section: (none)

Explanation

Explanation/Reference:
Call Forward Unregistered

The Call Forward Unregistered (CFU) feature allows you to forward a call to a different number if the directory number (DN) is not associated with a phone or if the associated phone is not registered to Cisco Unified CME. The CFU feature is very useful for wireless phone users when the wireless phone is out of the access point or phone shuts down automatically because of an automatic shutdown feature. The service is not available and the call can be forwarded to the CFU destination. Any unregistered or floating DN can be forwarded using the CFU feature.

An unregistered DN indicates that none of its associated phones are registered to the Cisco Unified CME. A registered phone will become unregistered when the Cisco Unified CME sends an unregistration request or responds to a phone's unregistration request. Cisco Unified CME sends an unregistration request under the following circumstances:

–When the keepalive timer expires.

–When a user issues a **reset** or **restart** command on the phone.

–When an extension mobility (EM) user logs into the phone. (All DNs configured under the logout-profile are unregistered except for the shared ones that are associated with other registered phones.)

–When an EM user logs out of the phone. (All DNs configured under the user-profile are unregistered except for the shared ones that are associated with other registered phones.)

There is always a gap between the time the phone loses its connection with Cisco Unified CME and the time when Cisco Unified CME claims the phone is unregistered. The length of the gap depends on the keepalive timer. Cisco Unified CME considers the phone as registered and tries to associate DNs until the keepalive timer expires. You can configure the expiration for the keepalive timer using the **registrar server expires max <seconds> min <seconds>** command under **sip** in voice service voip mode for SIP IP phones.

For more information, see the, "Configuring Keepalive Timer Expiration in SIP Phones: Example" section.

Cisco Unified CME 8.6 supports the CFU feature on SIP IP phones using the **call-forward b2bua unregistered** command under **voice register dn** tag. The CFU feature supports overlap dialing and en-bloc dialing. A call to a floating DN is forwarded to its CFU destination, if configured. Calls to a DN out of service point or phones losing connection are not forwarded to a CFU number until the phone becomes unregistered. **For more information on configuring call-forward unregistered, see the "Configuring Call Forward Unregistered for SIP IP Phones: Example" section.**

Note: In earlier versions of Cisco Unified CME, a busy tone was played for callers when the callers are unable to reach the SCCP phone number. In Cisco Unified CME 8.6 and later versions, a fast busy tone is played instead of a busy tone for callers who are unable to reach the phone.

Link:

http://www.cisco.com/en/US/docs/voice_ip_comm/cucme/admin/configuration/guide/cmetrans.html

QUESTION 84

Which statement about Cisco Unified Presence and LDAP is true?

- A. Cisco Unified Presence must be integrated with LDAP.
- B. LDAP integration is mandatory if Cisco Unified Personal Communicator is used as a client.
- C. LDAP integration with Cisco Unified Presence is optional. However, user search functionality in Cisco Unified Personal Communicator will not be available.
- D. LDAP integration with Cisco Unified Presence is optional. However, instant messaging functionality in Cisco Unified Personal Communicator will not be available.

Correct Answer: C

Section: (none)

Explanation

Explanation/Reference: LDAP Integrations

You can configure a corporate LDAP directory in this integration to satisfy a number of different requirements:

- **User provisioning:** You can provision users automatically from the LDAP directory into the Cisco Unified Communications Manager database.
Cisco Unified Communications Manager synchronizes with the LDAP directory content so you avoid having to add, remove, or modify user information manually each time a change occurs in the LDAP directory.
- **User authentication:** You can authenticate users using the LDAP directory credentials.
Cisco Unified Presence synchronizes all the user information from Cisco Unified Communications Manager to provide authentication for users of the Cisco Unified Personal Communicator client and Cisco Unified Presence user interface.
- **User lookup:** You can enable LDAP directory lookups to allow Cisco Unified Personal Communicator client users, or third-party XMPP clients, to search for and add contacts from the LDAP directory.

As the scope of the LDAP integration is dependent on customer requirements and it can vary between companies, there are a number of potential LDAP integration scenarios:

1. You integrate Cisco Unified Communications Manager and Cisco Unified Personal Communicator with an LDAP directory. We strongly recommend this configuration.
2. You integrate Cisco Unified Communications Manager with an LDAP directory, but you do not integrate Cisco Unified Personal Communicator. We do not recommend this configuration because it will impact Cisco Unified Personal Communicator functionality and you will experience performance issues.
3. You integrate Cisco Unified Personal Communicator with an LDAP directory, but you do not integrate Cisco Unified Communications Manager. We do not recommend this configuration because you will have to manually configure all your users on Cisco Unified Communications Manager at initial installation, and each time a change is made on the LDAP directory.

Note When Cisco Unified Communications Manager is not integrated with LDAP, you must verify that the username is exactly the same in Active Directory and Cisco Unified Communications Manager before deploying Cisco Unified Presence. If the letter case does not match, the presence status will not work properly in Cisco Unified Personal Communicator Release 7.x. Correct the username in Cisco Unified Communications Manager to match Active Directory.

Link:

http://www.cisco.com/en/US/docs/voice_ip_comm/cups/8_0/english/install_upgrade/deployment/guide/dgldap.html

QUESTION 85

Which three rules are valid transfer rules in Cisco Unity Connection? (Choose Three)

- A. standard
- B. alternate
- C. closed
- D. holiday
- E. nonstandard

Correct Answer: ABC

Section: (none)

Explanation

Explanation/Reference:
Standard Transfer Rule

This transfer rule applies during the work hours that your Cisco Unity administrator specified for your organization, or in other situations when no other transfer rule is enabled. By design, the standard transfer rule cannot be disabled.

Alternate Transfer Rule

Enable this transfer rule to apply during a specific time period when you want to override the other transfer rules. For example, you may want to route all your calls directly to voice mail while you are out of the office or you may want to transfer your calls to a different extension if you are temporarily working from another location. As long as it is enabled, the alternate transfer rule overrides all other transfer rules.

Closed Transfer Rule

Enable this transfer rule if you want Cisco Unity to perform different transfer actions during the nonwork hours that your Cisco Unity administrator specified for your organization. (For example, you may want to route all your calls directly to voice mail during nonwork hours.) As long as it is enabled, the closed transfer rule overrides the standard transfer rule during nonbusiness hours.

QUESTION 86

With GETVPN, if a key server is configured to use multicast as the rekey transport mechanism, then under which of these conditions will the key server retransmit the rekey messages?

- A. It never retransmit the rekey messages
- B. it only retransmit the rekey message when it does not receive the rekey acknowledgment from at least one group member
- C. it only retransmit the rekey message when it does not receive the rekey acknowledgment from all group member
- D. it only retransmit the rekey message when DPD to the group members fails
- E. it always retransmit the rekey message

Correct Answer: E

Section: (none)

Explanation

Explanation/Reference:

Cisco Group Encrypted Transport VPN

Retransmitting a Rekey

Multicast rekeys are retransmitted by default. For unicast rekeys, if the key server does not receive the ACK, it retransmits the rekey. In either case, before retransmitting a rekey, the key server checks if there is a TEK or KEK rekey scheduled in the next 120 seconds. If so, it stops the current retransmission and waits for the scheduled rekey to happen.

Group Member Access Control List

For GET VPN, the traffic that has to be protected is defined statically on the key server using the ACL. The group member gets information about what has to be protected from the key server. This structure allows the key server to choose and change the policy dynamically as needed. In Secure Multicast, the key server ACL is defined inclusively. The ACL includes only the exact traffic that should be encrypted, with an implicit deny causing all other traffic to be allowed in the clear (that is, if there is no permit, all other traffic is allowed).

GET VPN employs a different philosophy: The definition of which packets should be encrypted is delivered independently. GET VPN supports only statically defined traffic selectors. Policy can be defined by using both deny and permit ACLs on the key server. Only the deny ACL is allowed to be manually configured on a group member. The policies that are downloaded from the key server and configured on the group member are merged. Any ACL that is configured on the group member has predominance over what is downloaded from the key server.

After the group member gets the ACL from the key server, the group member creates a temporary ACL and inserts it into the database. This ACL will be deleted if the group member is removed from the GDOI group for any reason. The packets that are going out of the interface are dropped by the group member if a packet matches the ACL but no IPsec SA exists for that packet.

The key server can send a set of traffic selectors, which may not exactly match the group member ACL on the group member. If such differences occur, the differences have to be merged and resolved. Because the group member is more aware of its topology than the key server, the downloaded ACLs are appended to the group member ACL. The group member ACL (except the implicit deny) is inserted into the database first, followed by the downloaded key server ACL. The database is prioritized, and the database search stops whenever a matched entry is found.

Link:

http://www.cisco.com/en/US/docs/ios-xml/ios/sec_conn_getvpn/configuration/xs-3s/sec-get-vpn.html

QUESTION 87

An administrator is attempting to add a new IP phone to the network. The phone does not register and continues to cycle through the registration process. The administration checks and notices that the IP address assigned to the phone is not correct network.

What is the cause of this issue?

- A. The TFTP server is reconfigured
- B. The DHCP server is giving out false IP addresses
- C. The Cisco Unified Communication Manager is down
- D. The switch port that the phone is connected to is configured with the wrong voice LAN
- E. The PSTN gateway is down

Correct Answer: D

Section: (none)

Explanation

Explanation/Reference:

The voice VLAN feature enables access ports to carry IP voice traffic from an IP phone. The switch can connect to a Cisco 7960 IP Phone and carry IP voice traffic.

Because the sound quality of an IP phone call can deteriorate if the data is unevenly sent, the switch supports quality of service (QoS) based on IEEE 802.1P class of service (CoS).

QoS uses classification and scheduling to send network traffic from the switch in a predictable manner. For more information on QoS, see "Configuring QoS."

The Cisco 7960 IP Phone is a configurable device, and you can configure it to forward traffic with an 802.1P priority. You can configure the switch to trust or override the traffic priority assigned by an IP Phone.

Link:

http://www.cisco.com/en/US/docs/switches/lan/catalyst3550/software/release/12.1_19_ea1/configuration/guide/swvoip.html

QUESTION 88

In which mode is CTI used with Cisco Unified Personal Communicator?

- A. soft-phone mode
- B. desk-phone mode
- C. IP communicator mode
- D. IP phone mode
- E. CTI mode

Correct Answer: B

Section: (none)

Explanation

Explanation/Reference:

Desk Phone Control and the CTI Connection Failures

The CTI gateway provides desk phone control (phone-association mode) to Cisco Unified Personal Communicator users. You must specify CTI gateway server names, addresses, ports, and protocol types on Cisco Unified Presence so that the information required to reach the CTI gateway server can be downloaded when the user logs in to Cisco Unified Personal Communicator.

If the CTI connection to Cisco Unified Communications Manager is lost while Cisco Unified Personal Communicator is operating in desk phone mode, the application tries to reestablish the connection to the primary and then to the backup servers. Connection attempts continue on a round-robin basis, beginning again with the primary server. Successive attempts to reconnect to a server occur at intervals of 4, 8, 16, 32, and 60 seconds (maximum) until a connection is re-established.

Link:

http://www.cisco.com/en/US/docs/voice_ip_comm/cups/8_6/english/install_upgrade/deployment/guide/dgclient.html#wp1100542

QUESTION 89

What happens when a user associated with a device is deleted?

- A. The user is removed, but the DN and the Device remains in the system
- B. The user is removed, and the DN and the Device also removed
- C. The user is removed, and the DN and the Device become orphan in the system
- D. The user is removed, and the DN and the Device are automatically assigned to other user
- E. The user is removed, and the DN and the Device are assigned to the administrator

Correct Answer: A

Section: (none)

Explanation

Explanation/Reference:

Deleting an End User

Tips About Deleting End Users

Before you delete an end user, determine whether the devices or profiles that are associated with the end user need to be removed or deleted.

You can view the devices and profiles that are assigned to the end user from the Device Associations, Extension Mobility, Directory Number Associations, CAPF Information, and Permissions Information areas of the End User Configuration window.

You can also choose Dependency Records from the Related Links drop-down list box in the End User Configuration window. If the dependency records are not enabled for the system, the dependency records summary window displays a message.

For more information about dependency records, see the "Accessing Dependency Records" section.

Link:

http://www.cisco.com/en/US/docs/voice_ip_comm/cucm/admin/8_6_1/ccmcfg/b08user.html

QUESTION 90

In which two ways can an administrator reset an IP phone that is registered with Cisco Unified Communications Manager? (Choose two.)

- A. Phone can be reset in Cisco Unified Communications Manager Administration.
- B. Phone can be reset in the Cisco Unified Communications Manager Express CLI.

- C. Press the * * #* * key combination on the IP phone keypad.
- D. Enter the reset ephone command in the switch.
- E. Press the * * #* key combination on the IP phone keypad.
- F. Press the ##**# key combination on the IP phone keypad.

Correct Answer: AC

Section: (none)

Explanation

Explanation/Reference:

Verify

Complete these steps in order to reset the Cisco 7970 IP Phone:

1. Press the **Settings** button on the Cisco 7970 IP Phone.
2. Enter the **##** key sequence on the phone keypad in order to reset the phone.

The correct time displays on the Cisco 7970 IP Phone after the reset.

Link:

http://www.cisco.com/en/US/products/sw/voicesw/ps4625/products_configuration_example09186a0080631683.shtml

QUESTION 91

Which two options are features in Cisco Unified Presence? (Choose two.)

- A. IP Phone Messenger
- B. Native Presence
- C. BLF speed-dial
- D. Enterprise Instant Messaging
- E. BLF speed-dial pickup

Correct Answer: AB

Section: (none)

Explanation

Explanation/Reference:

- A. IP Phone Messenger —————> Feature
- B. Native Presence —————> Feature
- C. BLF speed-dial —————> Service
- D. Enterprise Instant Messaging —> Service
- E. BLF speed-dial pickup —————> Service

CUPS is tightly integrated with CUCM, which provides call control and native Presence signaling (on-/off-hook status). CUPS itself provides a central collection point for user capabilities and status by way of standards-based signaling using Session Initiation Protocol (SIP), SIP for Instant Messaging and Presence Leveraging Extensions (SIMPLE), and Extensible Messaging and Presence Protocol (XMPP). A variety of client interfaces are available, including Cisco Unified Personal Communicator (CUPC), which provides a rich and tightly integrated user experience.

CUPC provides TLS-secured Chat and Group Chat capability. Ad-Hoc Group Chat sessions are stored in memory on the CUPC server. The Persistent Chat feature enables group chat rooms where the conversation persists even when all participants have left the chat session. If Persistent Chat is enabled, it requires an external database to store the chat rooms and conversations. Offline IM allows chat messages to be sent to users who are currently offline; these messages are stored in the local IDS database on the CUPS server.

According to the Cisco exam A & B is the correct answer

QUESTION 92

Which level of users in Cisco Unified Communications Manager CDR Analysis and Reporting can generate reports for quality of service?

- A. administrators
- B. managers
- C. auditors
- D. individual users

Correct Answer: B

Section: (none)

Explanation

Explanation/Reference:

CAR System Reports Configuration

CAR provides reporting capabilities for three levels of users:

- Administrators — Generate system reports to help with load balancing, system performance, and troubleshooting.
- Managers — Generate reports for users, departments, and QoS to help with call monitoring for budgeting or security purposes and for determining the voice quality of the calls.
- Individual users — Generate a billing report for each user's calls.

Link:

http://www.cisco.com/en/US/docs/voice_ip_comm/cucm/service/6_0_1/car/carsyrrpt.html

QUESTION 93

When performing backups within Cisco Unified Communications Manager, which component stores backups on a remote SFTP server?

- A. Local Agent
- B. Master Agent
- C. Scheduler
- D. Backup Controller

Correct Answer: B

Section: (none)

Explanation

Explanation/Reference:

What is the Disaster Recovery System?

The Disaster Recovery System (DRS), which can be invoked from Cisco Unified Communications Manager Administration, provides full data backup and restore capabilities for all servers in a Cisco Unified Communications Manager cluster.

The Disaster Recovery System allows you to perform regularly scheduled automatic or user-invoked data backups.

The Disaster Recovery System performs a cluster-level backup, which means that it collects backups for all servers in a Cisco Unified Communications Manager cluster to a central location and archives the backup data to physical storage device.

When performing a system data restoration, you can choose which nodes in the cluster you want to restore.

The Disaster Recovery System includes the following capabilities:

- A user interface for performing backup and restore tasks.
- A distributed system architecture for performing backup and restore functions.
- Scheduled backups.
- Archive backups to a physical tape drive or remote sftp server.

The Disaster Recovery System contains two key functions, Master Agent (MA) and Local Agent (LA). The Master Agent coordinates backup and restore activity with Local Agents.

The system automatically activates both the Master Agent and the Local Agent on all nodes in the cluster.

Note: The Disaster Recovery System does not migrate data from Windows to Linux or from Linux to Linux. A restore must run on the same product version as the backup. For information on data migration from a Windows-based platform to a Linux-based platform, refer to the Data Migration Assistant User Guide.

Link:

http://www.cisco.com/en/US/docs/voice_ip_comm/cucm/drs/6_0_1/drs/drsag601.html

QUESTION 94

Which Cisco IOS command should you use to view the configuration of voice dial peer 911?

- A. show dialplan dialpeer 911
- B. show dialplan number 911
- C. show dial-peer voice 911
- D. show event-manager consumers 911

Correct Answer: C

Section: (none)

Explanation

Explanation/Reference:

Router# show dial-peer voice 911

QUESTION 95

Which type of server is used to deliver the configuration to an IP phone?

- A. TFTP
- B. DHCP
- C. FTP
- D. Cisco Discovery Protocol

Correct Answer: A

Section: (none)

Explanation

Explanation/Reference:

Cisco TFTP

The Cisco TFTP service builds and serves files consistent with the trivial file transfer protocol, which is a simplified version of the File Transfer Protocol (FTP). Cisco TFTP builds configuration files and serves

embedded component executables, ringer files, and device configuration files.

A configuration file contains a prioritized list of Cisco CallManagers for a device (telephones and gateways), the TCP port on which the device connects to those Cisco CallManagers, and an executable load identifier. Configuration files for Cisco IP Phone 7960 and 7940 models also contain URLs for the phone buttons: messages, directories, services, and information. Configuration files for gateways contain all their configuration information.

Configuration files may be in a .cnf format or a .cnf.xml format, depending on the device type and your TFTP service parameter settings. When you set the BuildCNFFlag service parameter to True, the TFTP server builds both .cnf.xml and .cnf format configuration files for devices. When you set the parameter to False, the TFTP server builds only .cnf.xml files for devices.

Link:

http://www.cisco.com/en/US/docs/voice_ip_comm/cucm/admin/3_1_2/ccmsys/a02tftp.html

QUESTION 96

An end user reports that they are unable to control their Cisco IP phone using Cisco Unified Personal Communicator and cannot make any calls. Which situation can cause this issue?

- A. The Cisco Unified Personal Communicator is not registered in the Cisco Unified Presence server.
- B. The LDAP integration is incorrect.
- C. The Cisco Unified Personal Communicator is configured in desk-phone mode.
- D. The Allow Control of Device from CTI checkbox in the device configuration on Cisco Unified Communications Manager is disabled.
- E. The Allow Control of Device from CTI checkbox in the device configuration on the Cisco Unified Presence is disabled.

Correct Answer: D

Section: (none)

Explanation

Explanation/Reference:

Complete these steps in order to resolve this issue:

1. Go to User Management > End User in order to verify that the user logged in is a member of the Standard CTI Enabled group in CUCM.
2. Go to User Management > End User and verify the Device Association.

This verifies that the user device is associated with their CUCM user profile.
3. Go to Application > Cisco Unified Personal Communicator > User Settings in order to verify that the device is the Preferred CTI Device in CUPS.
4. Go to User Management > End User in order to verify that the CUCM user has a primary extension specified.
5. The DN of the device you are trying to control must be specified in that user's Active Directory (AD) profile under Telephone Number.
6. With CUP 6.0 and CUPC 1.2, specify the TFTP server address of the CUCM under Cisco Unified Presence - Settings.
7. Ensure that the DNS server the PC uses can resolve the fully qualified name of the CUPS server.

QUESTION 97

To obtain CDR information, which application is used?

- A. Cisco Unified Communications Manager Administration
- B. Cisco Unified Serviceability
- C. Cisco Unified Operating System Administration
- D. Disaster Recovery System
- E. Cisco Unified Communications Manager Call Detail Record Analysis and Reporting tool
- F. Cisco Unified Reporting

Correct Answer: E

Section: (none)

Explanation

Explanation/Reference:

Cisco Unified Serviceability supports CDR Analysis and Reporting (CAR) under the Tools menu. CAR generates reports for Quality of Service, traffic, and billing information.

Call detail records (CDRs) detail the called number, the number that placed the call, the date and time that the call was started, the time that it connected, and the time that it ended. Call management records (CMRs, or diagnostic records) detail the jitter, lost packets, the amount of data sent and received during the call, and latency. CDR data comprises CDRs and CMRs collectively. A single call can result in the generation of several CDRs and CMRs. Cisco Unified Communications Manager records information regarding each call in CDRs and CMRs. CDRs and CMRs, known collectively as CDR data, serve as the basic information source for CAR.

Link:

http://www.cisco.com/en/US/docs/voice_ip_comm/cucm/service/6_0_1/car/carovrvw.html#wp1101768

QUESTION 98

Which two protocols are used by Cisco Unified Presence? (Choose two.)

- A. SIP/SIMPLE
- B. XMPP
- C. SCCP
- D. PPPoX
- E. IMPP

Correct Answer: AB

Section: (none)

Explanation

Explanation/Reference:

1. The **Skinny Call Control Protocol (SCCP, or short Skinny)** is a proprietary network terminal control protocol.

2. **PPPoX** (PPP over X) designates a family of encapsulating communications protocols implementing Point-to-Point Protocol.

3. **IMPP** is not a protocol.

QUESTION 99

What are the two differences between the IntServ and DiffServ models of QoS? (Choose two.)

- A. DiffServ provides absolute QoS guarantees.
- B. IntServ is the default QoS mechanism for all routers, because applications signal the router with the QoS level they require.
- C. DiffServ inherits the connection-oriented approach from telephony network design. Every individual communication must explicitly specify its traffic descriptor and requested resources to the network.
- D. With IntServ, packet delivery is guaranteed. However, the use of IntServ can severely limit the scalability of a network.

E. DiffServ was designed to overcome the limitations of both the best-effort and IntServ models and can provide an "almost guaranteed" QoS.

Correct Answer: DE

Section: (none)

Explanation

Explanation/Reference:

Integrated Services

Cisco IOS Software supports two fundamental Quality of Service architectures: Differentiated Services (DiffServ) and Integrated Services (IntServ). In the DiffServ model a packet's "class" can be marked directly in the packet, which contrasts with the IntServ model where a signaling protocol is required to tell the routers which flows of packets requires special QoS treatment. DiffServ achieves better QoS scalability, while IntServ provides a tighter QoS mechanism for real-time traffic. These approaches can be complimentary and are not mutually exclusive.

The IntServ architecture model (RFC 1633, June 1994) was motivated by the needs of real-time applications such as remote video, multimedia conferencing, visualization, and virtual reality. It provides a way to deliver the end-to-end Quality of Service (QoS) that real-time applications require by explicitly managing network resources to provide QoS to specific user packet streams (flows). It uses "resource reservation" and "admission control" mechanisms as key building blocks to establish and maintain QoS.

IntServ uses Resource Reservation Protocol (RSVP) to explicitly signal the QoS needs of an application's traffic along the devices in the end-to-end path through the network. If every network device along the path can reserve the necessary bandwidth, the originating application can begin transmitting.

Besides end-to-end signaling, IntServ requires several functions on routers and switches along the path:

Admission Control: determine whether a new flow can be granted the requested QoS without impacting existing reservations

Classification: recognize packets that need particular levels of QoS

Policing: take action, including possibly dropping packets, when traffic does not conform to its specified characteristics

Queuing and Scheduling: forward packets according to those QoS requests that have been granted

Link:

http://www.cisco.com/en/US/products/ps6611/products_ios_protocol_group_home.html

QUESTION 100

Which three characteristics are associated with voice? (Choose three.)

- A. greedy
- B. TCP retransmits
- C. UDP priority
- D. delay sensitive
- E. drop insensitive
- F. benign
- G. benign or greedy

Correct Answer: CDF

Section: (none)

Explanation

Explanation/Reference:

Voice traffic uses UDP packets for network transfer and normally they are sensitive to delay (Latency) because it's real time traffic. Voice traffic can be benign as well.

QUESTION 101

Which type of packet-oriented network has the characteristic of being drop-tolerant and delayinsensitive?

- A. data
- B. voice
- C. video
- D. converged
- E. All packet-oriented networks share these characteristics.

Correct Answer: A

Section: (none)

Explanation

Explanation/Reference:**Three Characteristics of Data**

TCP retransmits
drop insensitive
benign or greedy

QUESTION 102

What will happen if an end user is deleted from the Cisco Unified Communications Manager system?

- A. The user will be removed, but the associated device and directory number will remain in the system.
- B. The user and the associated device will be removed, but the directory number will become orphaned.
- C. The user, the associated device, and the directory number will be removed.
- D. The user will be removed, and the associated device and directory number will be automatically assigned to the administrator.
- E. The user will be removed, and the associated device and directory number will be allocated to the next user added to the system.

Correct Answer: A

Section: (none)

Explanation

QUESTION 103

How does the Cisco Unified Communications Manager Express match an outbound VoIP dial peer?

- A. Outbound dial-peer matching uses the entire string of digits en bloc to match the dial peer with the longest match.
- B. Outbound dial-peer matching is completed on a digit-by-digit basis.
- C. It matches outbound dial peers by placing all the dial peers into a hunt group and then uses the entire dialed number en bloc to match the first dial peer.
- D. Cisco Unified Communications Manager Express creates a hunt group that contains all the configured dial peers and then applies the dial digits in a digit-by-digit manner to match a dial peer.

Correct Answer: A

Section: (none)

Explanation

Explanation/Reference:**According to Actual test Answer is A.**

When a router receives a voice call, it selects an outbound dial peer by comparing the called number (the full E.164 telephone number) in the call information with the number configured as the destination pattern for the POTS dial peer. The router then strips out the left-justified numbers corresponding to the destination pattern matching the called number. If you have configured a prefix, the prefix will be put in front of the remaining numbers, creating a dial string, which the router will then dial. If all numbers in the destination pattern are stripped-out, the user will receive (depending on the attached equipment) a dial tone. **(Actual test explanation)**

The method a router uses to select an outbound dial peer depends on **whether DID is configured in the inbound POTS dial peer. If DID is not configured in the inbound POTS dial peer**, the router collects the incoming dialed string digit by digit. When one dial peer is matched, the router immediately places the call using the configured attributes in the matching dial peer.

If DID is configured in the inbound POTS dial peer, the router uses the full incoming dial string to match the destination pattern in the outbound dial peer. With DID, the setup message contains all the digits necessary to route the call; no additional digit collection is required. If more than one dial peer matches the dial string, all of the matching dial peers are used to form a rotary group. The router attempts to place the outbound call leg using all of the dial peers in the rotary group until one is successful.

Link:

http://docwiki.cisco.com/wiki/Cisco_IOS_Voice_Troubleshooting_and_Monitoring_-_Call_Flow_Overview#Outbound_Dial_Peer_Matching

QUESTION 104

In which situation are user PINs maintained by the local Cisco Unified Communications servers?

- A. Only when the system is using LDAP synchronization.
- B. Only when the system is using LDAP authentication.
- C. Only when the system is using Global Directory.
- D. PINs are always maintained by the local Cisco Unified Communications Manager servers
- E. PINs are never maintained by the local Cisco Unified Communications Manager servers.

Correct Answer: D**Section: (none)****Explanation****Explanation/Reference:**

Cisco Unified Communications Manager assigns the system Default Credential Policy to end user passwords, end user PINS, and application user passwords

QUESTION 105

An IP phone has a line calling search space and a device calling search space. If a call is made from the IP phone, which calling search space is used?

- A. Neither calling search space is used.
- B. The line calling search space takes precedence and is used.
- C. The device calling search space takes precedence and is used.
- D. The line and device calling search spaces are combined and the line calling search space has precedence.

Correct Answer: D**Section: (none)****Explanation****Explanation/Reference:**

If you configure a calling search space both on an IP phone line and on the device (IP phone) itself, Cisco Unified Communications Manager concatenates the two calling search spaces and places the line calling

search space in front of the device calling search space

QUESTION 106

Which command is used to determine if an MGCP gateway is registered with a Cisco Unified Communications Manager server?

- A. show gateway status
- B. show isdn q931
- C. show ccm-manager
- D. show isdn status
- E. show isdn q921

Correct Answer: C

Section: (none)

Explanation

Explanation/Reference:

Router# **show ccm-manager**

MGCP Domain Name: router.cisco.com

Total number of host: 2

Priority Status Host

```
=====
Primary Registered 10.0.0.201
First backup Backup polling 10.0.0.50
Second backup Undefined
Current active Communications Manager: 10.0.0.201
Current backup Communications Manager: 10.0.0.50
Redundant link port: 2428
Failover Interval: 30 seconds
Keepalive Interval: 15 seconds
Last keepalive sent: 00:20:18 (elapsed time: 00:00:06)
Last MGCP traffic time: 00:20:18 (elapsed time: 00:00:06)
Last switchover time: None
Switchback mode: Immediate
```

QUESTION 107

A user exists within an office of IP phones, all of which have the same pickup group. A call comes into a phone of another office worker who is currently at lunch. What should the user do to have the call redirected to his or her phone?

- A. Press the "Pickup" softkey and then the line that is ringing.
- B. Press the "Call Pickup" softkey only.
- C. Press the "Call Pickup" softkey and then the line that is ringing.
- D. Press the "Pickup" softkey only.

Correct Answer: B

Section: (none)

Explanation

Explanation/Reference:

Cisco IP Phones provide three types of call pickup: call pickup, group call pickup, and other group call pickup.

- Call pickup - Allows users to pick up incoming calls within their own group. Cisco CallManager automatically dials the appropriate call pickup group number when the user activates this feature from a Cisco IP Phone. Use the softkey, Pickup, for this type of call pickup.
- Group call pickup - Allows users to pick up incoming calls in another group. User must dial the appropriate call

pickup group number when activating this feature from a Cisco IP Phone. Use the softkey, GPickUp, for this type of call pickup.

Note: The same procedures apply for configuring call pickup and group call pickup features. Group call pickup numbers apply to lines or directory numbers.

- Other group call pickup - Allows users to pick up incoming calls in a group that is associated with their own group. The Cisco CallManager automatically searches for the incoming call in the associated groups to make the call connection when the user activates this feature from a Cisco IP Phone. Use the softkey, oPickup, for this type of call pickup.

Link:

http://www.cisco.com/en/US/docs/voice_ip_comm/cucm/admin/5_0_1/ccmfeat/fscpickg.html

QUESTION 108

What is the preferred analog signaling method to reduce glare?

- A. loop start
- B. ground start
- C. on-hook
- D. off-hook

Correct Answer: B

Section: (none)

Explanation

Explanation/Reference:

Loop-Start Signaling

Loop-start signaling sends an off-hook signal that starts a call and an on-hook signal that closes the loop and ends the call. Loop-start trunks lack positive disconnect supervision, and users can experience glare when two calls seize the trunk at the same time.

Ground-Start Signaling

Ground-start signaling provides current detection mechanisms at both ends of the trunk to detect off-hook signals. This mechanism permits endpoints to agree on which end is seizing the trunk before it is seized, and minimizes the chance of glare. Ground start provides positive recognition of connects and disconnects and is the preferred signaling method for PBX connections. Some PBXs do not support ground-start signaling, so then you must use loop-start signaling for the trunk interface.

Link:

http://www.cisco.com/en/US/docs/voice_ip_comm/cucm/admin/4_0_1/ccmsys/a08procl.html

QUESTION 109

What is the accepted maximum limit for good-quality voice connection delay?

- A. 100 ms
- B. 200 ms
- C. 250 ms
- D. 300 ms

Correct Answer: C

Section: (none)

Explanation

Explanation/Reference:

Delay Specifications

<u>Range in Milliseconds</u>	-	<u>Description</u>
0-150	-	Acceptable for most user applications.
150-400	-	Acceptable provided that administrators are aware of the transmission time and the impact it has on the transmission quality of user applications.
Above 400	-	Unacceptable for general network planning purposes. However, it is recognized that in some exceptional cases this limit is exceeded.

Link:

http://www.cisco.com/en/US/tech/tk652/tk698/technologies_white_paper09186a00800a8993.shtml

QUESTION 110

Which two directory services are supported by Cisco Unified Communications Manager for Lightweight Directory Access Protocol integration? (Choose two.)

- A. Windows Active Directory 2008
- B. Novell eDirectory
- C. iPlanet Directory Server 4.0
- D. Sun ONE Directory Server
- E. Open Lightweight Directory Access Protocol 2.1

Correct Answer: AD

Section: (none)

Explanation

Explanation/Reference:

LDAP Directory Configuration Checklist

If you want to do so, you can add users from your corporate directory to the Cisco Unified Communications Manager database by synchronizing the user data to the database.
Cisco Unified Communications Manager allows synchronization from the following directories to the database:

- Microsoft Active Directory 2000
- Microsoft Active Directory 2003
- Microsoft Active Directory 2008
- Microsoft Active Directory Application Mode 2003
- Microsoft Lightweight Directory Services 2008
- iPlanet Directory Server 5.1
- Sun ONE Directory Server 5.2
- Sun ONE Directory Server 6.x
- OpenLDAP 2.3.39
- OpenLDAP 2.4

Note: Microsoft Active Directory Application Mode support is limited to those directory topologies already supported with a native Active Directory connection. No additional topologies, such as multi-forest, multi-tree single forest, or global catalog are supported.

Cisco Unified Communications Manager supports the following types of synchronization:

- Automatic synchronization, which synchronizes the data at regular intervals.
- Manual synchronization, which allows forcing the synchronization.
- Stop synchronization, which stops the current synchronization. If synchronization is in progress, check for agreement.

Link:

http://www.cisco.com/en/US/docs/voice_ip_comm/cucm/admin/8_6_1/ccmsys/a04direc.html

QUESTION 111

When creating a new softphone for a Cisco Unified Presence user, which phone type should you select if you are using Cisco Unified Personal Communicator Release 7.1?

- A. Cisco Unified Personal Communicator
- B. Cisco Unified Client Services Communicator
- C. Cisco Unified Client Services Framework
- D. Cisco Unified Personal Communicator Framework

Correct Answer: A

Section: (none)

Explanation

Explanation/Reference:

(Cisco Unified Personal Communicator Release 7.1) Guidelines for Configuring the Softphone Device Name

When you create a softphone device for each Cisco Unified Personal Communicator user, you must specify a device name. Make sure that the device name conforms to these guidelines:

- Derives from the username.
- Starts with UPC.
- Contains only uppercase letters, or numerals.
- Contains no more than 12 additional characters after UPC.

Link:

http://www.cisco.com/en/US/docs/voice_ip_comm/cups/8_0/english/install_upgrade/deployment/guide/dgclient.html#wp1135232

QUESTION 112

One user from your company wants to use a signaling protocol on the voice gateways that require registration with the Cisco Unified Communications Manager. Which protocol should you suggest to him?

- A. SIP
- B. Frame-Relay
- C. SRTP
- D. MGCP

Correct Answer: D

Section: (none)

Explanation

Explanation/Reference:

Media Gateway Control Protocol (MGCP)

MGCP provides Cisco Unified Communications Manager with a powerful, flexible and scalable resource for call control. Cisco Unified Communications Manager uses MGCP to control media on the telephony interfaces of a remote gateway and also uses MGCP to deliver messages from a remote gateway to appropriate devices.

MGCP enables a call agent (media gateway controller) to remotely control and manage voice and data communication devices at the edge of multiservice IP packet networks. Because of its centralized architecture, MGCP simplifies the configuration and administration of voice gateways and supports multiple (redundant) call agents in a network. MGCP does not provide security mechanisms such as message encryption or authentication.

Using MGCP, Cisco Unified Communications Manager controls call processing and routing and provides supplementary services to the gateway. The MGCP gateway provides call preservation (the gateway maintains calls during failover and fallback), redundancy, dial-plan simplification (the gateway requires no dial-peer configuration), hookflash transfer, and tone on hold. MGCP-controlled gateways do not require a media termination point (MTP) to enable supplementary services such as hold, transfer, call pickup, and call park. If the MGCP gateway loses contact with its Cisco Unified Communications Manager, it falls back to using H.323 control to support basic call handling of FXS, FXO, T1 CAS, and T1/E1 PRI interfaces.

Link:

http://www.cisco.com/en/US/docs/voice_ip_comm/cucm/admin/9_0_1/ccmsys/CUCM_BK_CD2F83FA_00_system-guide_chapter_0100111.html

QUESTION 113

Which value should you assign to option 150 for DHCP to operate correctly in a Cisco Unified Communications Manager Express environment?

- A. FTP server of the DNS server
- B. MAC address of the DHCP server
- C. MAC address of the TFTP server
- D. IP address of the TFTP server

Correct Answer: D

Section: (none)

Explanation

Explanation/Reference:

Configuring a Router-Based DHCP Server

RTR# **configure terminal**

RTR(config)# **ip dhcp excluded-address 172.16.1.1 172.16.1.9**

RTR(config)# **ip dhcp excluded-address 172.16.2.1 172.16.2.9**

RTR(config)# **ip dhcp pool DATA_SCOPE**

RTR(dhcp-config)# **network 172.16.2.0 255.255.255.0**

RTR(dhcp-config)# **default-router 172.16.2.1**

RTR(dhcp-config)# **dns-server 4.2.2.2**

RTR(dhcp-config)# **exit**

RTR(config)# **ip dhcp pool VOICE_SCOPE**

RTR(dhcp-config)# **network 172.16.1.0 255.255.255.0**

RTR(dhcp-config)# **default-router 172.16.1.1**

RTR(dhcp-config)# **option 150 ip 172.16.1.1**

RTR(dhcp-config)# **dns-server 4.2.2.2**

QUESTION 114

What device enables Call Admission Control in a CME environment?

- A. Gateway
- B. Transcoder
- C. Bridge
- D. Gatekeeper

Correct Answer: D

Section: (none)

Explanation

Explanation/Reference:

Cisco IOS Gatekeeper Zones

A Cisco IOS gatekeeper can provide call routing and call admission control between devices such as Cisco Unified CM, Cisco Unified Communications Manager Express (Unified CME), or H.323 gateways connected to legacy PBXs. It uses the H.323 Registration Admission Status (RAS) protocol to communicate with these devices and route calls across the network.

Gatekeeper call admission control is a policy-based scheme requiring static configuration of available resources. The gatekeeper is not aware of the network topology, so it is limited to simple hub-and-spoke topologies. Refer to the section on Design Considerations for Call Admission Control, for detailed topology examples.

For a listing of the available Cisco IOS gatekeeper platforms and the features supported on each platform, refer to the Cisco IOS H323 Gatekeeper Data Sheet at

http://www.cisco.com/en/US/prod/collateral/voicesw/ps6788/vcallcon/ps4139/data_sheet_c78_561921.html

The call admission control capabilities of a Cisco IOS gatekeeper are based on the concept of gatekeeper zones. A zone is a collection of H.323 devices, such as endpoints, gateways, or Multipoint Control Units (MCUs), that register with a gatekeeper. There can be only one active gatekeeper per zone, and you can define up to 100 local zones on a single gatekeeper. A local zone is a zone that is actively handled by that gatekeeper - that is, all H.323 devices assigned to that zone register with that gatekeeper.

When multiple gatekeepers are deployed in the same network, a zone is configured as a local zone on only one gatekeeper. On the other gatekeepers, that zone is configured as a remote zone. This configuration instructs the gatekeeper to forward calls destined for that zone to the gatekeeper that "owns it" (that is, the gatekeeper on which that zone is configured as a local zone).

For details on configuring the gatekeeper, refer to the Cisco IOS H.323 Configuration Guide at

http://www.cisco.com/en/US/docs/ios/voice/h323/configuration/guide/15_0/vh_15_0_book.html

The bandwidth value deducted by the gatekeeper for every active call is double the bit-rate of the call, excluding Layer 2, IP, and RTP overhead. For example, a G.711 audio call that uses 64 kbps would be denoted as 128 kbps in the gatekeeper, and a 384-kbps video call would be denoted as 768 kbps. Table 11-4 shows the bandwidth values used by the gatekeeper feature for some of the most popular call speeds.

Link:

http://www.cisco.com/en/US/docs/voice_ip_comm/cucm/srnd/8x/cac.html

QUESTION 115

In a Cisco UCM single-site deployment, what is the maximum number of IP phones that can register with a UCM cluster?

- A. 2500
- B. 7500
- C. 10000
- D. 30000

Correct Answer: D

Section: (none)

Explanation

Explanation/Reference:

Single Site

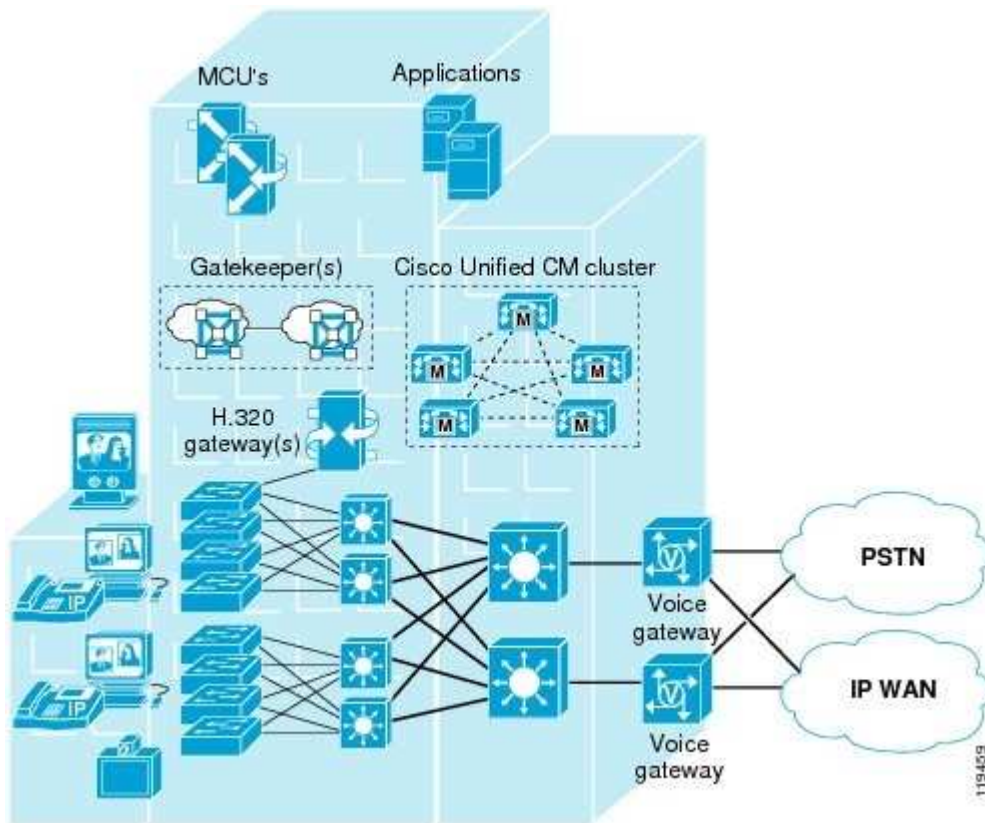
The single-site model for IP telephony consists of a call processing agent located at a single site, or campus, with no telephony services provided over an IP WAN. An enterprise would typically deploy the single-site model over a LAN or metropolitan area network (MAN), which carries the voice traffic within the site. In this model, calls beyond the LAN or MAN use the public switched telephone network (PSTN).

The single-site model has the following design characteristics:

- Single Cisco Unified CallManager or Cisco Unified CallManager cluster.
- Maximum of 30,000 Skinny Client Control Protocol (SCCP) IP phones or SCCP video endpoints per cluster.
- Maximum of 500 H.323 devices (gateways, MCUs, trunks, and clients) per Cisco Unified CallManager cluster.
- PSTN for all external calls.
- Digital signal processor (DSP) resources for conferencing, transcoding, and media termination point (MTP).
- Voicemail or unified messaging components.
- Capability to integrate with legacy private branch exchange (PBX) and voicemail systems.
- H.323 clients, MCUs, and H.323/H.320 gateways that require a gatekeeper to place calls must register with a Cisco IOS Gatekeeper (Cisco IOS Release 12.3(8)T or greater). Cisco Unified CallManager then uses an H.323 trunk to integrate with the gatekeeper and provide call routing and bandwidth management services for the H.323 devices registered to it. Multiple Cisco IOS Gatekeepers may be used to provide redundancy.
- MCU resources are required for multipoint video conferencing. Depending on conferencing requirements, these resources may be either SCCP or H.323, or both.
- H.323/H.320 video gateways are needed to communicate with H.320 videoconferencing devices on the public ISDN network.
- High-bandwidth audio (for example, G.711, G.722, or Cisco Wideband Audio) between devices within the site.
- High-bandwidth video (for example, 384 kbps or greater) between devices within the site. The Cisco Unified Video Advantage Wideband Codec, operating at 7 Mbps, is also supported.

Figure 2-1 illustrates the model for an IP telephony network within a single campus or site.

Figure 2-1 Single-Site Deployment Model



Link:

http://www.cisco.com/en/US/docs/voice_ip_comm/cucm/srnd/4x/42models.html

QUESTION 116

An administrator wants to import users using the Bulk Administration menu in Cisco Unified Communication is valid for this operation?

- A. .PDF
- B. .DOC
- C. .DOCX
- D. .CSV
- E. .XLS

Correct Answer: D

Section: (none)

Explanation

Explanation/Reference:

CSV is the format in which administrator can import users using the Bulk Administration menu in Cisco Unified Communications.

QUESTION 117

Which statement accurately describes a calling search space?

- A. a group of object with similar reach ability characteristics.
- B. a calling feature that finds mobile users.
- C. a toll that is used to track calls to certain numbers.

D. a feature that defines which partitions are reachable from a device.

Correct Answer: D

Section: (none)

Explanation

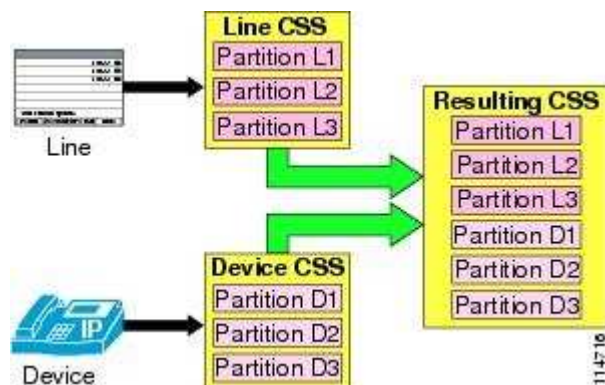
Explanation/Reference:

Calling Search Spaces

A calling search space defines which partitions are accessible to a particular device. Devices that are assigned a certain calling search space can access only the partitions listed in that calling search space. Attempts to dial a DN in a partition outside that calling search space will fail, and the caller will hear a busy signal.

If you configure a calling search space both on an IP phone line and on the device (phone) itself, Cisco Unified CallManager concatenates the two calling search spaces and places the line's calling search space in front of the device's calling search space, as shown in Figure 10-6.

Figure 10-6 Concatenation of Line and Device Calling Search Spaces for IP Phones



Note: In versions of Cisco Unified CallManager prior to Release 4.2, the device calling search space remains the same as the device is moved to different parts of the network. With Cisco Unified CallManager 4.2, the device calling search space can be determined dynamically based on where in the network the phone is physically located, as determined by the phone's IP address. See Device Mobility, for more details.

If the same route pattern appears in two partitions, one contained in the line's calling search space and one contained in the device's calling search space, then according to the rules described in the section on Partitions, Cisco Unified CallManager selects the route pattern listed first in the concatenated list of partitions (in this case, the route pattern associated with the line's calling search space).

For recommendations on how to set the line and device calling search spaces, refer to the sections on Building Classes of Service for Cisco Unified CallManager, and Classes of Service with the Line/Device Approach.

The maximum length of the combined calling search space (device plus line) is 1024 characters, including separator characters between each partition name. (For example, the string "partition_1:partition_2:partition_3" contains 35 characters.) Thus, the maximum number of partitions in a calling search space varies, depending on the length of the partition names. Also, because the calling search space clause combines the calling search space of the device and that of the line, the maximum character limit for an individual calling search space is 512 (half of the combined calling search space clause limit of 1024 characters).

Therefore, when you are creating partitions and calling search spaces, keep the names of partitions short relative to the number of partitions that you plan to include in a calling search space. For more details on configuring calling search spaces, refer to the Cisco Unified CallManager Administration Guide, available online at

<http://www.cisco.com>

Before you configure any partitions or calling search spaces, all DNs reside in a special partition named <None>, and all devices are assigned a calling search space also named <None>. When you create custom partitions and calling search spaces, any calling search space you create also contains the <None> partition, while the <None> calling search space contains only the <None> partition.

Note: Any dial plan entry left in the <None> partition is implicitly reachable by any device making a call. Therefore, to avoid unexpected results, Cisco strongly recommends that you do not leave dial plan entries in the <None> partition.

Note: Cisco strongly recommends that you do not leave any calling search space defined as <None>. Doing so can introduce dial plan behavior that is difficult to predict.

Link:

http://www.cisco.com/en/US/docs/voice_ip_comm/cucm/srnd/4x/42dialpl.html

QUESTION 118

An engineer is configuring a new Cisco Unified Communications Manager server. However, when the engineer tries to register the IP phones, the registrations are unsuccessful.

When the engineer checks one of the phones, there is no status that is shown along with the Cisco Unified Communications Manager server IP address.

What is the probable cause?

- A. The server connection is established, but the information is encrypted.
- B. There is no current connection with the Cisco Unified Communications Manager server.
- C. The Cisco Unified Communications Manager server is currently available.
- D. The currently receiving call-processing services are running on the phone.

Correct Answer: B

Section: (none)

Explanation

Explanation/Reference:

QUESTION 119

Which Cisco Unity Connection report provides a summary view of the current size, last error condition, and status of the mailbox store?

- A. Users
- B. Message Traffic
- C. Mailbox Store
- D. System Configuration

Correct Answer: C

Section: (none)

Explanation

Explanation/Reference:

Reports Available in Cisco Unity Connection

Includes the following information about the specified mailbox stores:

- Mail database name.
- Display name.
- Server name.

- Whether access is enabled.
- Mailbox store size.
- Last error.
- Status.
- Whether the mail database can be deleted.

Link:

http://www.cisco.com/en/US/docs/voice_ip_comm/connection/8x/administration/guide/8xcucsag300.html

QUESTION 120

If two phones share a line between them, what happens if the user of the first phone is using the line and the user of the second phone uses the Barge feature?

- The user of the first phone is notified of the Barge request.
- A three-way conference begins.
- The call is discontinued, but the first phone user and the second phone user are able to talk.
- The user on the first phone is knocked off the line, and the second phone user takes over the line.

Correct Answer: B

Section: (none)

Explanation

Explanation/Reference:

Barge and Privacy

The Single Button Barge/cBarge, Barge, Privacy, and Privacy on Hold features work with each other. These features work with only shared lines.

Barge adds a user to a call that is in progress. Pressing a softkey automatically adds the user (initiator) to the shared-line call (target), and the users currently on the call receive a tone (if configured). Barge supports built-in conference and shared conference bridges.

The Single Button Barge/cBarge feature allows the user to simply press the shared-line button to be added to the call. The Single Button Barge/cBarge feature supports built-in conferences and shared conference bridges.

The administrator enables or disables privacy and privacy on hold features. Privacy must be enabled for a device to activate privacy on hold. Users toggle the privacy feature on or off.

The administrator enables or disables Privacy setting. When Privacy is enabled, the system removes the call information from all phones that share lines and blocks other shared lines from barging in on its calls. When Privacy is disabled, the system displays call information on all phones that have shared line appearances and allows other shared lines to barge in on its calls. Administrators can configure Privacy for all devices or configure Privacy for each device. Users toggle the Privacy feature on or off.

The privacy on hold feature preserves privacy when a private call on a shared line is put on hold. When privacy on hold is enabled, the calling name and number that are blocked when privacy is enabled remain blocked when the call is put on hold, and the system blocks other shared lines from resuming the held call. When privacy on hold is disabled and a private call is put on hold, the system displays calling name and number on all phones that have shared line appearances and allows other shared lines to resume the held call.

If privacy on hold is enabled, users can activate the feature while the call is on hold by toggling privacy on; likewise, users can deactivate privacy on hold by toggling privacy off while the call is on hold. If privacy on hold is disabled, toggling privacy on or off does not affect the held call.

If a private call is put on hold, retrieved at the same phone, and privacy is then toggled off, the system displays the call information on all phones that have shared line appearances but does not allow another phone to resume or barge the held call.

Administrators can configure privacy for all devices or for each device. Administrators configure privacy on hold for the cluster.

Link:

http://www.cisco.com/en/US/docs/voice_ip_comm/cucm/admin/7_0_1/ccmfeat/fsbarge.html

QUESTION 121

Which protocol is being used when you are utilizing the chat feature of Cisco Unified Personal Communicator?

- A. SIMPLE
- B. SIP
- C. XMPP
- D. HTTPS

Correct Answer: C

Section: (none)

Explanation

Explanation/Reference:

Quickly Communicate with Borderless Enterprise-Class Instant Messaging

Instant messaging is an important communication option that lets you efficiently interact in today's multitasking business environment. Cisco Unified Personal Communicator delivers enterprise-class instant messaging capabilities that are based on the Extensible Messaging and Presence Protocol (XMPP) open standard and technology from the Jabber, Inc. acquisition. The solution provides personal chat, group chat, and persistent chat rooms so you can quickly connect with individuals and groups and conduct ongoing conversations. Client-side history and server-based logging capabilities allow you to view the content of prior chats and to store messages for convenience, compliance, and regulation purposes. Instant messaging is smoothly integrated with other communication capabilities so you can simply move between individual and group chats to audio and video conversations and web conferences.

Cisco Unified Personal Communicator lets you securely communicate with other organizations that use Cisco Unified Personal Communicator, Cisco WebEx™ Connect, Microsoft Office Communicator, IBM Lotus Sametime, GoogleTalk, AOL, or any XMPP-based service. The enterprise-class instant messaging capabilities of Cisco Unified Personal Communicator provide more-efficient, secure, flexible, and borderless collaboration.

Link:

http://www.cisco.com/en/US/prod/collateral/voicesw/ps6789/ps6836/ps6844/data_sheet_c78-647911.html

QUESTION 122

When deploying Cisco Unified Presence, which service is a basic service that can be considered as optional?

- A. Cisco Unified Presence Engine
- B. Cisco Unified Presence Sync Agent
- C. Cisco Unified Presence Session Initiation Protocol Proxy
- D. Cisco Unified Protocol Extensible Messaging Presence and Protocol Agent

Correct Answer: D

Section: (none)

Explanation

QUESTION 123

Which option should you use in the Cisco Unified Communications Manager End Users Configuration page to ensure that a user can use a desk phone both for calls and for Cisco Unified Presence?

- A. Enable Cisco Unified Presence Communicator
- B. Allow Control of Device from Cisco Computer Telephony Integration
- C. Allow Cisco Unified Personal Communicator Integration
- D. Allow Cisco Unified Presence Control over IP Phone

Correct Answer: B

Section: (none)

Explanation

Explanation/Reference:

Computer Telephony Integration

Computer telephony integration (CTI) enables you to leverage computer-processing functions while making, receiving, and managing telephone calls. CTI applications allow you to perform such tasks as retrieving customer information from a database on the basis of information that caller ID provides. CTI applications can also enable you to use information that an interactive voice response (IVR) system captures, so the call can be routed to the appropriate customer service representative or so the information is provided to the individual who is receiving the call.

Computer Telephony Integration Applications

The following list contains descriptions of some Cisco CTI applications that are available:

- Cisco IP Communicator—Cisco IP Communicator, a desktop application, turns your computer into a full-feature telephone with the added advantages of call tracking, desktop collaboration, and one-click dialing from online directories. You can also use Cisco IP Communicator in tandem with a Cisco Unified IP Phone to place, receive, and control calls from your desktop PC. All features function in both modes of operation.
- Cisco Unified Communications Manager Auto-Attendant—The Cisco Unified Communications Manager Auto-Attendant application works with Cisco Unified Communications Manager to receive calls on specific telephone extensions and to allow the caller to choose an appropriate extension.
- Cisco Web Dialer—Cisco Web Dialer, which is installed on a Cisco Unified Communications Manager server and is used in conjunction with Cisco Unified Communications Manager, allows Cisco Unified IP Phone users to make calls from web and desktop applications.
- Cisco Unified Communications Manager Assistant—The Cisco Unified Communications Manager Assistant feature enables managers and their assistants to work together more effectively. The feature comprises a call-routing service, enhancements to phone capabilities for the manager and the assistant, and assistant console interfaces that are primarily used by the assistant.

Link:

http://www.cisco.com/en/US/docs/voice_ip_comm/cucm/admin/8_0_1/ccmsys/a08cti.html

QUESTION 124

Which protocol should you use to configure Cisco Unified Personal Communicator for secure voice messaging with Cisco Unity Connection?

- A. TCP
- B. SSL
- C. TLS
- D. UDP

Correct Answer: C

Section: (none)

Explanation

Explanation/Reference: Introduction:

Transport Layer Security (TLS).

TLS is a successor to Secure Sockets Layer protocol. TLS provides secure communications on the Internet for such things as e-mail, Internet faxing, and other data transfers. There are slight differences between SSL 3.0 and TLS 1.0, but the protocol remains substantially the same. It is good idea to keep in mind that TLS resides on the Application Layer of the OSI model. This will save you a lot of frustrations while debugging and troubleshooting encryption problems related to TLS.

TLS Handshake:

The TLS Handshake Protocol allows the server and client to authenticate each other and to negotiate an encryption algorithm and cryptographic keys before data is exchanged. In a typical scenario, only the server is authenticated and its identity is ensured while the client remains unauthenticated. The mutual authentication of the servers requires public key deployment to clients. When a server and client communicate, TLS protocol ensures that no third party may eavesdrop, tamper with any message, and message forgery. A TLS message may span multiple TLS records.

Link:

<https://supportforums.cisco.com/docs/DOC-1157>

QUESTION 125

Which report in Cisco Unified Reporting should you use to track the number of users with one or more phones?

- A. Unified CM User Device Count
- B. Unified CM Device Distribution Summary
- C. Unified CM Table Count Summary
- D. Unified CM Data Summary

Correct Answer: A

Section: (none)

Explanation

Explanation/Reference: Unified CM reports

Unified CM User Device Count - Provides information about associated devices; for example, this report lists the number of phones with no users, the number of users with one phone, and the number of users with more than one phone.

Unified CM Device Distribution Summary - Provides a summary of how devices are distributed throughout the cluster; for example, this report shows which devices are associated with the primary, secondary, tertiary servers, and so on.

Unified CM Data Summary - Provides a summary of data that exists in the Unified CM database, according to the structure of the menus in Unified CM Administration. For example, if you configure three credential policies, five conference bridges, and ten shared-line appearances, you can see that type of information in this report.

Unified CM Table Count Summary - Provides a database-centric view of data. This report proves useful for administrators or AXL API developers that understand database schema.

Link:

http://www.cisco.com/en/US/docs/voice_ip_comm/cucm/service/9_1_1/report/

QUESTION 126

Which Cisco IOS CLI command should you use to perform an IP phone cold reboot?

- A. router(config-ephone)#reset
- B. router(config-ephone)#restart
- C. router(config-ephone-dn)#restart
- D. router(config-phone)#reset

Correct Answer: A

Section: (none)

Explanation

Explanation/Reference:

```
Router> enable
Router# configure terminal
Router(config-ephone)# reset
Router(config-ephone)# end
```

Link:

http://www.cisco.com/en/US/docs/voice_ip_comm/cucme/admin/configuration/guide/cmreset.html

QUESTION 127

By default, how many failed attempts at signing into Cisco Unity Connection is a user allowed before their account is locked out?

- A. 2
- B. 3
- C. 4
- D. 5
- E. 6
- F. No limit

Correct Answer: B

Section: (none)

Explanation

Explanation/Reference:

Edit Authentication Rule

Failed Logon Attempts:

Enter the number of failed logon attempts after which users cannot access Cisco Unity Connection. When set to 0 (zero), there is no limit to the number of failed logon attempts, and the user is not locked out of the account.

Default setting: 3 attempts.

Link:

http://www.cisco.com/en/US/docs/voice_ip_comm/connection/7x/gui_reference/guide/7xcucgrg100.html

QUESTION 128

When you reset an IP phone via the Cisco Unified Communications Manager Administration page, which method briefly shuts down a registered phone and brings it back up?

- A. drop

- B. restart
- C. reset
- D. shutdown
- E. shut & no shut

Correct Answer: C

Section: (none)

Explanation

Explanation/Reference:

reset Command----->Takes longer to process when updating multiple phones.

reset - Performs a reboot of the specified phone or all phones running SCCP associated with this Cisco Unified CME router. Does contact the DHCP server for updated information

restart Command----->Faster processing for multiple phones.

restart - Performs a fast reboot of the specified phone or all phones running SCCP associated with this Cisco Unified CME router. Does not contact the DHCP server for updated information

Link:

http://www.cisco.com/en/US/docs/voice_ip_comm/cucme/admin/configuration/guide/cmreset.html#wpxref87974

QUESTION 129

Which generating QoS reports CAR tool, what two parameters are valid for report generation? (Choose two)

- A. route lists
- B. route patterns/hunt pilots
- C. route groups
- D. gateway types
- E. partitions
- F. IP phone directory numbers

Correct Answer: BD

Section: (none)

Explanation

Explanation/Reference:

Configuring Route Pattern and Hunt Pilot Device Reports

Only CAR administrators generate the Route/Hunt List Utilization report. The Route/Hunt List Utilization report provides an estimate of the maximum utilization percentage of the route/hunt list (cumulative utilization of all the gateways under the route/hunt list) for the period and not the exact utilization. The system calculates the cumulative utilization of all the gateways under the route lists and all the lines under the hunt lists.

You can examine the usage based on each hour of a day or on a specified number of days for each week or month. Reports generate for each of the selected route/hunt lists.

You can either view reports that the system automatically generates or generate new reports. Only CAR administrators can schedule reports for automatic generation.

Link:

http://www.cisco.com/en/US/docs/voice_ip_comm/cucm/service/8_6_1/car/cardvrou.html

Configuring QoS System Reports

CAR provides reporting capabilities for three levels of users:

- Administrators—Generate system reports to help with load balancing, system performance, and troubleshooting.
- Managers—Generate reports for users, departments, and QoS to help with call monitoring for budgeting or security purposes and for determining the voice quality of the calls.
- Individual users—Generate a billing report for calls by each user.

Link:

http://www.cisco.com/en/US/docs/voice_ip_comm/cucm/service/7_0_1/car/carsyqos.html

QUESTION 130

Which description describes the low latency queuing algorithm?

- A. Empty queue 1. If queue 1 is empty, empty queue 2, then empty queue 3, unless a packet for a higher queue arrives.
- B. An administrator defines the traffic classes based on match criteria, including protocols, access control lists, and input interfaces.
- C. A flow-based algorithm that simultaneously schedules interactive traffic to the front of a queue to reduce response time and fairly shares the remaining bandwidth among high-bandwidth flows.
- D. This feature brings strict priority queuing to CBWFQ.
- E. Packets are placed into a single queue and serviced in the order they were received.

Correct Answer: D

Section: (none)

Explanation

Explanation/Reference:

The Low Latency Queueing feature brings strict priority queueing to Class-Based Weighted Fair Queueing (CBWFQ). Strict priority queueing allows delay-sensitive data such as voice to be dequeued and sent first (before packets in other queues are dequeued), giving delay-sensitive data preferential treatment over other traffic.

Without Low Latency Queueing, CBWFQ provides weighted fair queueing based on defined classes with no strict priority queue available for real-time traffic. CBWFQ allows you to define traffic classes and then assign characteristics to that class. For example, you can designate the minimum bandwidth delivered to the class during congestion.

For CBWFQ, the weight for a packet belonging to a specific class is derived from the bandwidth you assigned to the class when you configured it. Therefore, the bandwidth assigned to the packets of a class determines the order in which packets are sent. All packets are serviced fairly based on weight; no class of packets may be granted strict priority. This scheme poses problems for voice traffic that is largely intolerant of delay, especially variation in delay. For voice traffic, variations in delay introduce irregularities of transmission manifesting as jitter in the heard conversation.

The Low Latency Queueing feature provides strict priority queueing for CBWFQ, reducing jitter in voice conversations. Configured by the priority command, Low Latency Queueing enables use of a single, strict priority queue within CBWFQ at the class level, allowing you to direct traffic belonging to a class to the CBWFQ strict priority queue. To enqueue class traffic to the strict priority queue, you configure the priority command for the class after you specify the named class within a policy map. (Classes to which the priority command is applied are considered priority classes.) Within a policy map, you can give one or more classes priority status. When multiple classes within a single policy map are configured as priority classes, all traffic from these classes is enqueued to the same, single, strict priority queue.

One of the ways in which the strict priority queueing used within CBWFQ differs from its use outside CBWFQ is in the parameters it takes. Outside CBWFQ, by using the `ip rtp priority` command, you specify the range of UDP ports whose voice traffic flows are to be given priority service. Using the `priority` command, because you can configure the priority status for a class within CBWFQ, you are no longer limited to a UDP port number to stipulate priority flows. Instead, all of the valid match criteria used to specify traffic for a class now applies to priority traffic. These methods of specifying traffic for a class include matching on access lists, protocols, and input interfaces. Moreover, within an access list you can specify that traffic matches are allowed based on the IP Differentiated Services Code Point (DSCP) value that is set using the first six bits of the Type of Service (ToS) byte in the IP header.

Although it is possible to enqueue various types of real-time traffic to the strict priority queue, we strongly recommend that you direct only voice traffic to it. This recommendation is made because voice traffic is well-behaved, whereas other types of real-time traffic are not. Moreover, voice traffic requires that delay be nonvariable in order to avoid jitter. Real-time traffic such as video could introduce variation in delay, thereby thwarting the steadiness of delay required for successful voice traffic transmission.

QUESTION 131

Which behavior can be configured with the `snr calling-number local` command?

- A. displaying the called number instead of the remote destination at the calling party
- B. displaying the original caller at the remote destination
- C. displaying the enterprise directory number instead of the original caller at the calling party
- D. displaying the enterprise directory number instead of the original caller at the remote destination

Correct Answer: D

Section: (none)

Explanation

QUESTION 132

What application uses the Cisco Unified Operating System for administration and configuration?

- A. Cisco Unity Express
- B. Cisco Unified Messaging Gateway
- C. Cisco Unified Communications Manager Express
- D. Cisco Unified Presence

Correct Answer: D

Section: (none)

Explanation

Explanation/Reference:

Cisco unity is for voice mail feature.

Cisco unified operating system is CUCM/CCME feature.

QUESTION 133

Click and drag the description of the signaling protocol on the left to the appropriate category on the right.

Uses call agents and provides the signaling capability for less-expensive edge devices.

Clients consume less processing overhead and can also be used to control FXS ports in a router to provide supplementary features.

Uses a modified form of the URL addressing scheme that is used within email that is based on the Simple Mail Transfer Protocol.

A standard that provides multimedia communication services—real-time audio, video, and data communications—over packet networks.

H.323

MGCP

SIP

SCCP

Exhibit:

Click and drag the description of the signaling protocol on the left to the appropriate category on the right.

Uses call agents and provides the signaling capability for less-expensive edge devices.

Clients consume less processing overhead and can also be used to control FXS ports in a router to provide supplementary features.

Uses a modified form of the URL addressing scheme that is used within email that is based on the Simple Mail Transfer Protocol.

A standard that provides multimedia communication services—real-time audio, video, and data communications—over packet networks.

H.323

MGCP

SIP

SCCP

Select and Place:

Click and drag the description of the signaling protocol on the left to the appropriate category on the right.

Uses call agents and provides the signaling capability for less-expensive edge devices.

Clients consume less processing overhead and can also be used to control FXS ports in a router to provide supplementary features.

Uses a modified form of the URL addressing scheme that is used within email that is based on the Simple Mail Transfer Protocol.

A standard that provides multimedia communication services—real-time audio, video, and data communications—over packet networks.

H.323

MGCP

SIP

SCCP

Correct Answer:

Click and drag the description of the signaling protocol on the left to the appropriate category on the right.

Four empty rectangular boxes for descriptions of signaling protocols, arranged vertically.

H.323

A standard that provides multimedia time audio, video, and data commun

MGCP

Uses call agents and provides th expensive edge

SIP

Uses a modified form of the URL a within email that is based on the s

SCCP

Clients consume less processing o control FXS ports in a router to pro

Section: (none)
Explanation

Explanation/Reference:

H.323
A standard that provides multimedia communication services—real-time audio, video, and data communications—over packet networks.
MGCP
Uses call agents and provides the signaling capability for less-expensive edge devices.
SIP
Uses a modified form of the URL addressing scheme that is used within email that is based on the Simple Mail Transfer Protocol.
SCCP
Clients consume less processing overhead and can also be used to control FXS ports in a router to provide supplementary features.

QUESTION 134

Which two types of device can the Cisco Unified Disaster Recovery use as a backup target? (Choose two.)

- A. DVD ROM drive
- B. FTP server
- C. tape device
- D. TFTP server
- E. WebDAV server
- F. SFTP server
- G. CD ROM

Correct Answer: CF

Section: (none)

Explanation

Explanation/Reference:

Cisco unified disaster recovery choose one of the following backup devices.

Tape Device - Stores the backup file on a locally attached tape drive

Network Directory - Stores the backup file on a networked drive that is accessed through an SFTP connection

QUESTION 135

To monitor the service health and performance, which service should you activate in Cisco Unity Connection?

- A. CUC Performance Service
- B. CUC System Auditing
- C. Real-Time Monitoring Tool
- D. Cisco Serviceability Reporter

Correct Answer: D

Section: (none)

Explanation

Explanation/Reference:

Activate the Cisco Serviceability Reporter service.

Link:

http://www.cisco.com/en/US/docs/voice_ip_comm/connection/2x/serviceability/unified_administration/guide/2xcucuniag100.html

QUESTION 136

Which report can you use to display all currently unassigned directory numbers?

- A. Route Plan Report
- B. Directory Number Assignment
- C. Unassigned Objects Report
- D. IP Phone Number Assignment

Correct Answer: A

Section: (none)

Explanation

Explanation/Reference:

Route Plan Report

The route plan report lists all assigned and unassigned directory numbers (DN), call park numbers, call pickup numbers, conference numbers, route patterns, translation patterns, message-waiting indicators, voice mail ports, and Cisco CallManager Attendant Console pilot numbers in the system. The route plan report allows you to view either a partial or full list and to go directly to the associated configuration windows by clicking the Pattern/Directory Number, Partition, or Route Detail fields.

In addition, the route plan report allows you to save report data into a .csv file that you can import into other applications. The .csv file contains more detailed information than the web pages, including directory numbers for phones, route patterns, pattern usage, device name, and device description.

Cisco CallManager uses the route plan to route both internal calls and external public switched telephone network (PSTN) calls. For more detailed information on the route plan, refer to the "Understanding Route Plans" section in Cisco CallManager System Guide.

Link:

http://www.cisco.com/en/US/docs/voice_ip_comm/cucm/admin/4_1_3/ccmcfg/b03rtrep.html

QUESTION 137

An engineer is installing an IP phone in a remote location. When the engineer plugs the IP phone into the network, the phone does not power up. What is the first thing that should be checked?

- A. Power over Ethernet switch
- B. Cisco Unified Communications Manager Server
- C. Cisco Unified Presence
- D. DHCP server

Correct Answer: A

Section: (none)

Explanation

Explanation/Reference:

Symptom: The Cisco Unified IP Phone Does Not Go Through its Normal Startup Process

When you connect a Cisco Unified IP Phone into the network port, the phone should go through its normal startup process as described in "Verifying the Phone Startup Process" section and the LCD screen should display information. If the phone does not go through the startup process, the cause may be faulty cables, bad connections, network outages, lack of power, and so on. Or, the phone may not be functional.

To determine whether the phone is functional, follow these suggestions to systematically eliminate these other potential problems:

1. Verify that the network port is functional:

- Exchange the Ethernet cables with cables that you know are functional.
- Disconnect a functioning Cisco Unified IP Phone from another port and connect it to this network port to verify the port is active.
- Connect the Cisco Unified IP Phone that will not start up to a different network port that is known to be good.
- Connect the Cisco Unified IP Phone that will not start up directly to the port on the switch, eliminating the patch panel connection in the office.

2. Verify that the phone is receiving power:

- If you are using external power, verify that the electrical outlet is functional.
- If you are using in-line power, use the external power supply instead.
- If you are using the external power supply, switch with a unit that you know to be functional.

3. If the phone still does not start up properly, power up the phone with the handset off-hook. When the phone is powered up in this way, it attempts to launch a backup software image.

4. If the phone still does not start up properly, perform a factory reset of the phone. For instructions, see the "Performing a Factory Reset" section.

If after attempting these solutions, the LCD screen on the Cisco Unified IP Phone does not display any characters after at least five minutes, contact a Cisco technical support representative for additional assistance.

Link:

http://www.cisco.com/en/US/docs/voice_ip_comm/cuipph/7962g_7942g/6_0/english/administration/guide/7962trb.html

QUESTION 138

An organization is using a centralized DHCP server for all IP phones. However, when the IP phones are powered up, the phones are unable to obtain an IP address.

Which CLI command should be in the router configuration to ensure that the IP phones are able to reach the DHCP server?

- A. `router(config)#helper-address`
- B. `router(config-if)#ip helper-address`
- C. `router(config-if)#helper-address`
- D. `router(config)#ip helper-address`

Correct Answer: B

Section: (none)

Explanation

Explanation/Reference:

Configuring IP Helper Addresses

To configure the helper address, identify the router interface that will receive the broadcasts for UDP services. In interface configuration mode, use the **ip helper-address** command to define the address to which UDP broadcasts for services should be forwarded.

By default, the **ip helper-address** command forwards the eight UDP services listed

What if Company XYZ needs to forward requests for a service not on this list? The Cisco IOS software provides the global configuration command **ip forward-protocol** to allow an administrator to forward any UDP port in addition to the default eight. To forward UDP on port 517, use the global configuration command **ip forward-protocol udp 517**. This command is used not only to add a UDP port to the default eight, but also to subtract an unwanted service from the default group. When forwarding DHCP, TFTP, and DNS without forwarding Time, TACACS, and NetBIOS, the Cisco IOS software requires that the router be configured according to the syntax shown in Example 2-5.

Example 2-5 Forwarding UDP Services

```
RTA(config-if)#ip helper-address 192.168.1.254
RTA(config-if)#exit
RTA(config)#ip forward-protocol udp 517
RTA(config)#no ip forward-protocol udp 37
RTA(config)#no ip forward-protocol udp 49
RTA(config)#no ip forward-protocol udp 137
RTA(config)#no ip forward-protocol udp 138
```

Link:

<http://www.ciscopress.com/articles/article.asp?p=330807&seqNum=9>

QUESTION 139

An engineer is troubleshooting call quality issues between central headquarters and a remote branch location. When on an active call, the engineer watches the call statistics on the IP phone and notices that the max jitter is 100 ms.

What is the maximum amount of jitter that the engineer should set to maintain a high-quality call?

- A. 5 ms
- B. 50 ms
- C. 10 ms
- D. 30 ms

Correct Answer: D

Section: (none)

Explanation

Explanation/Reference:

Voice (Bearer Traffic)

The following list summarizes the key QoS requirements and recommendations for voice (bearer traffic):

- Voice traffic should be marked to DSCP EF per the QoS Baseline and RFC 3246.
- Loss should be no more than 1 percent.
- One-way latency (mouth to ear) should be no more than 150 ms.
- Average one-way jitter should be targeted at less than 30 ms.
- A range of 21 to 320 kbps of guaranteed priority bandwidth is required per call (depending on the sampling

rate, the VoIP codec, and Layer 2 media overhead).

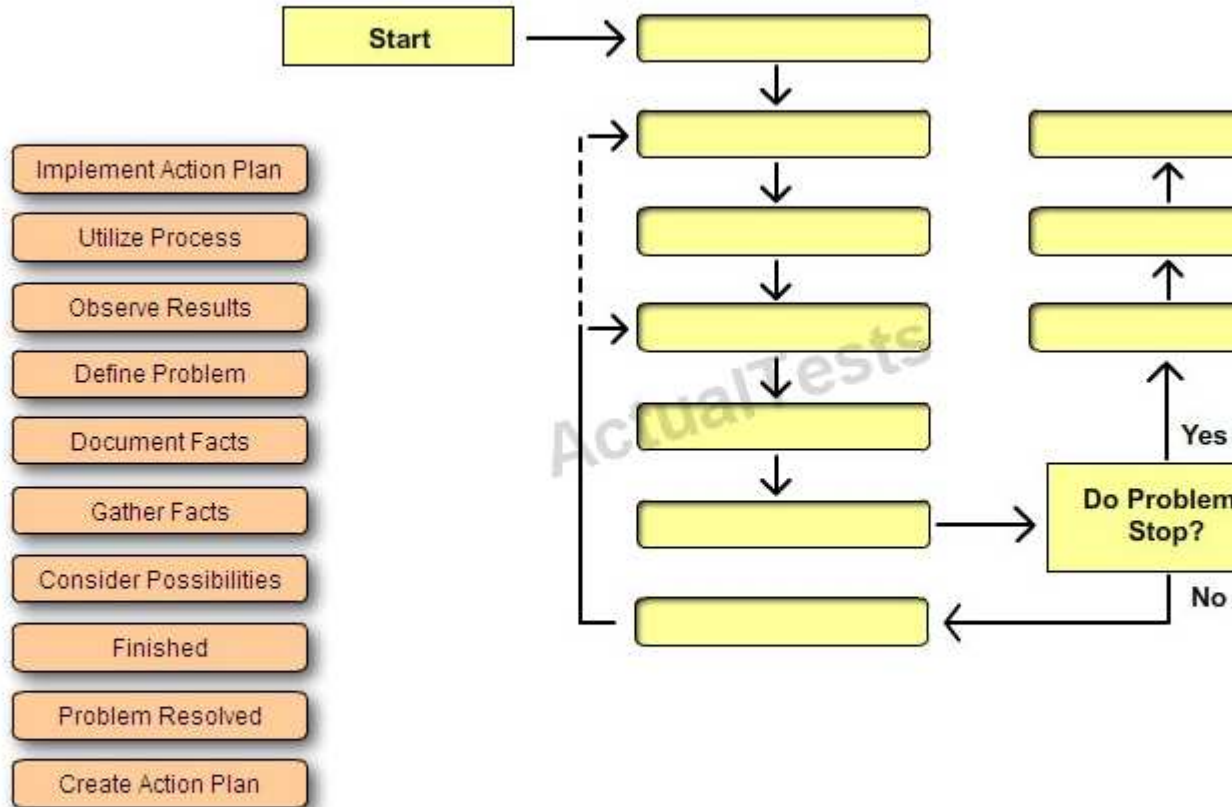
Voice quality directly is affected by all three QoS quality factors: loss, latency, and jitter.

Link:

<http://www.ciscopress.com/articles/article.asp?p=357102>

QUESTION 140

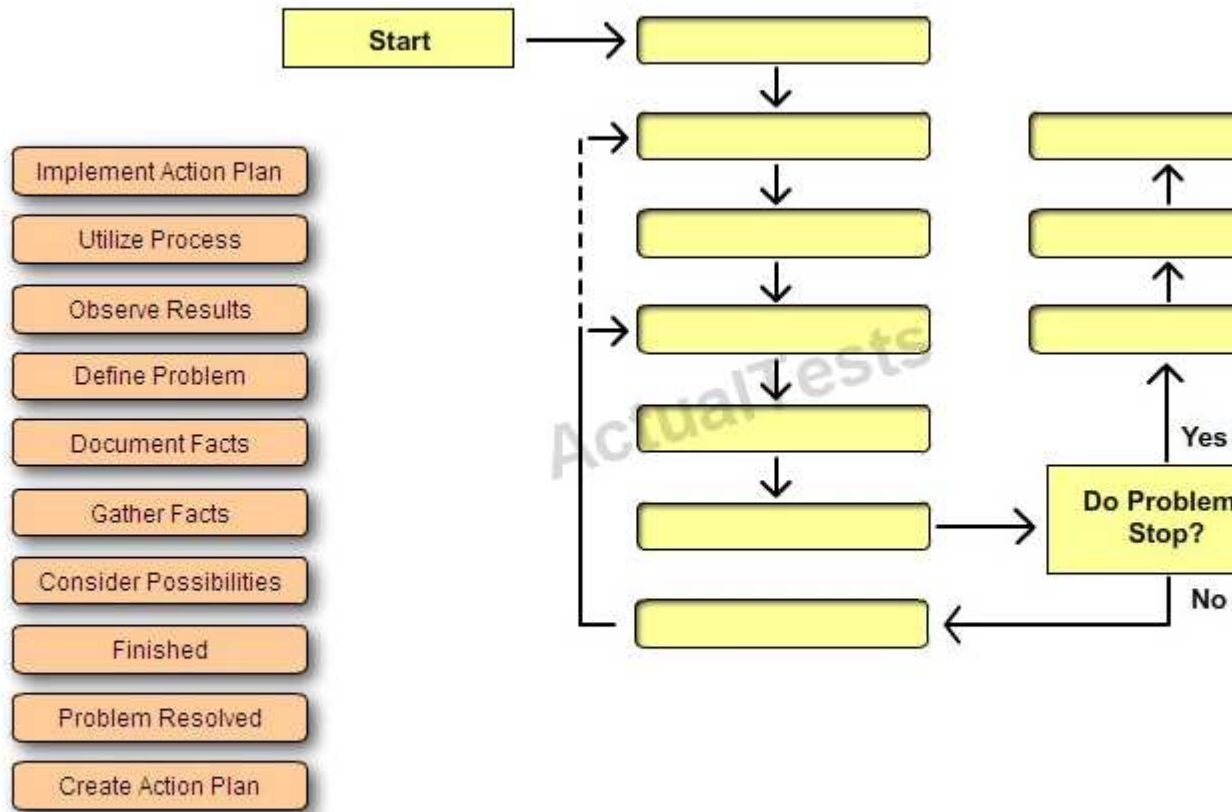
Click and drag the items on the left, into the correct order on the right for a systematic troubleshooting method.



1.0.1.10

Exhibit:

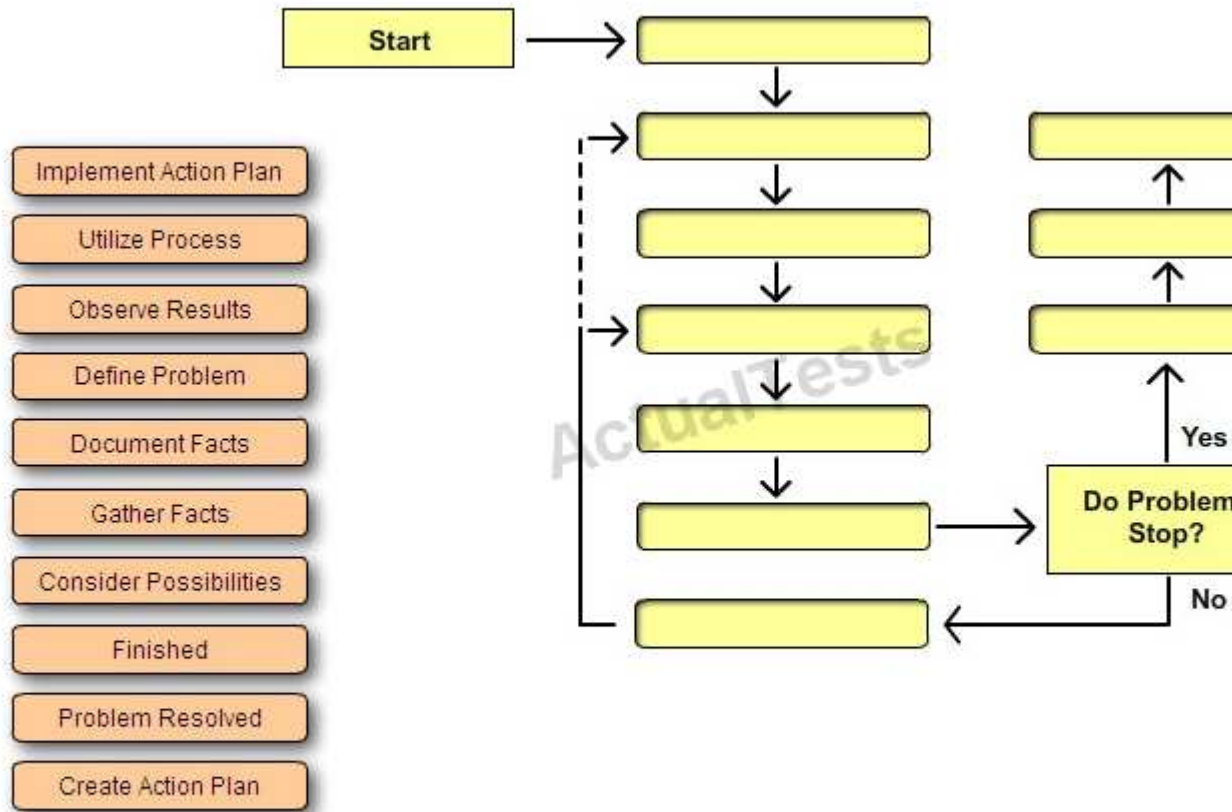
Click and drag the items on the left, into the correct order on the right for a systematic troubleshooting method.



1.0.1.10

Select and Place:

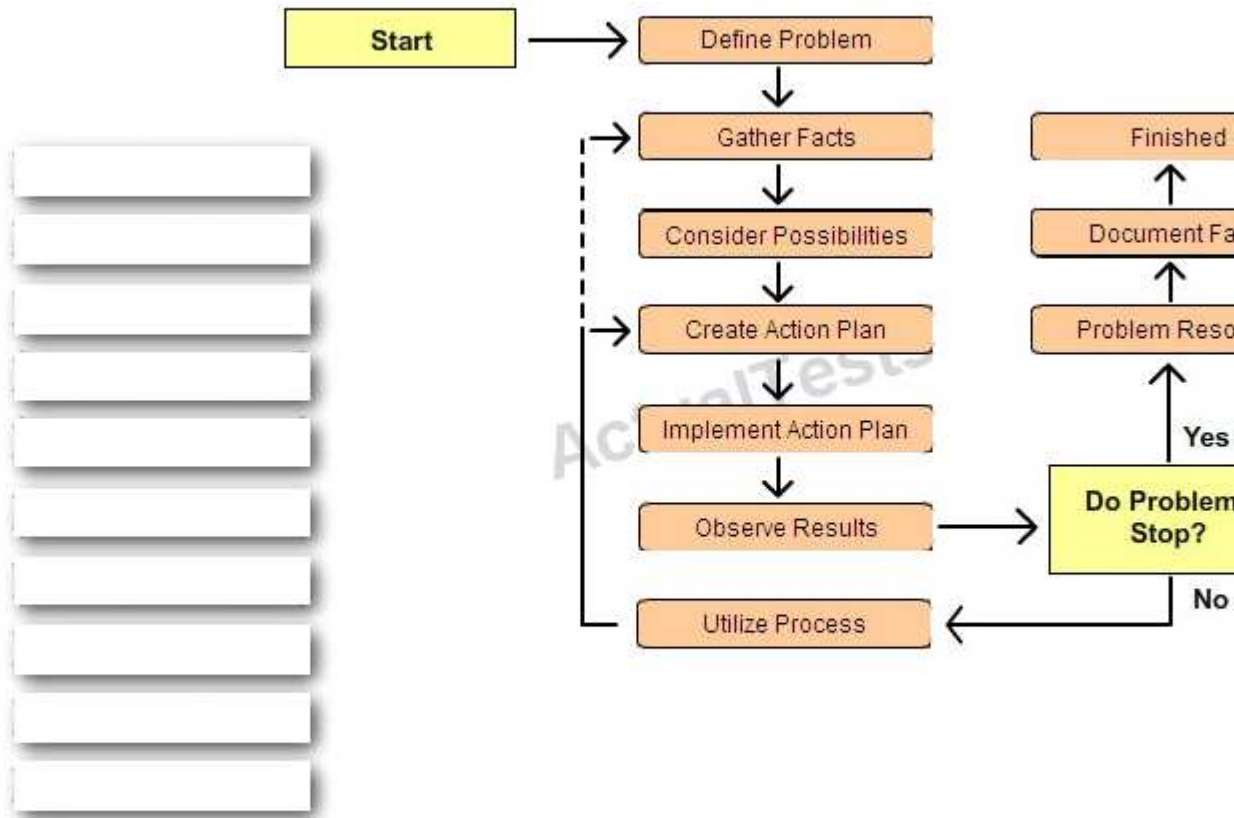
Click and drag the items on the left, into the correct order on the right for a systematic troubleshooting method.



1.0.1.10

Correct Answer:

Click and drag the items on the left, into the correct order on the right for a systematic troubleshooting method.

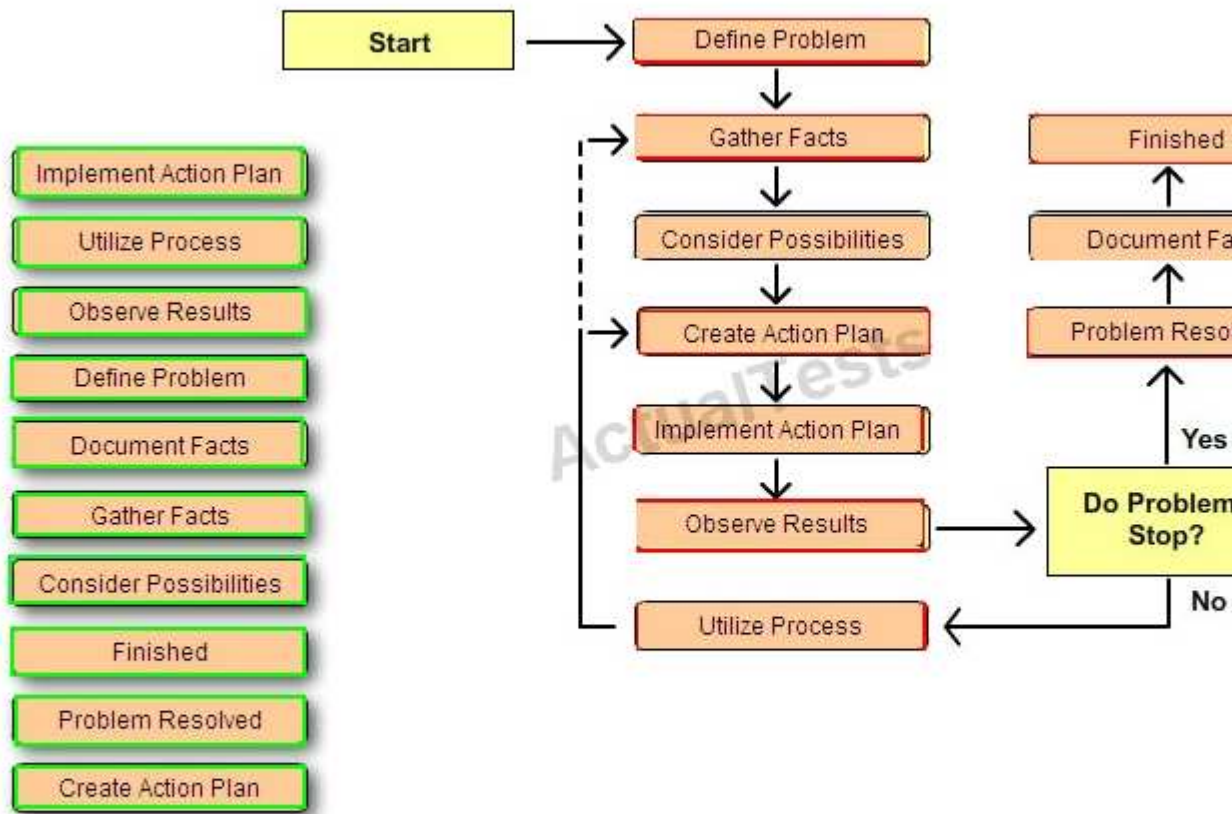


1.0.1.10

Section: (none)
Explanation

Explanation/Reference:

Click and drag the items on the left, into the correct order on the right for a systematic troubleshooting method.



1.0.1.10