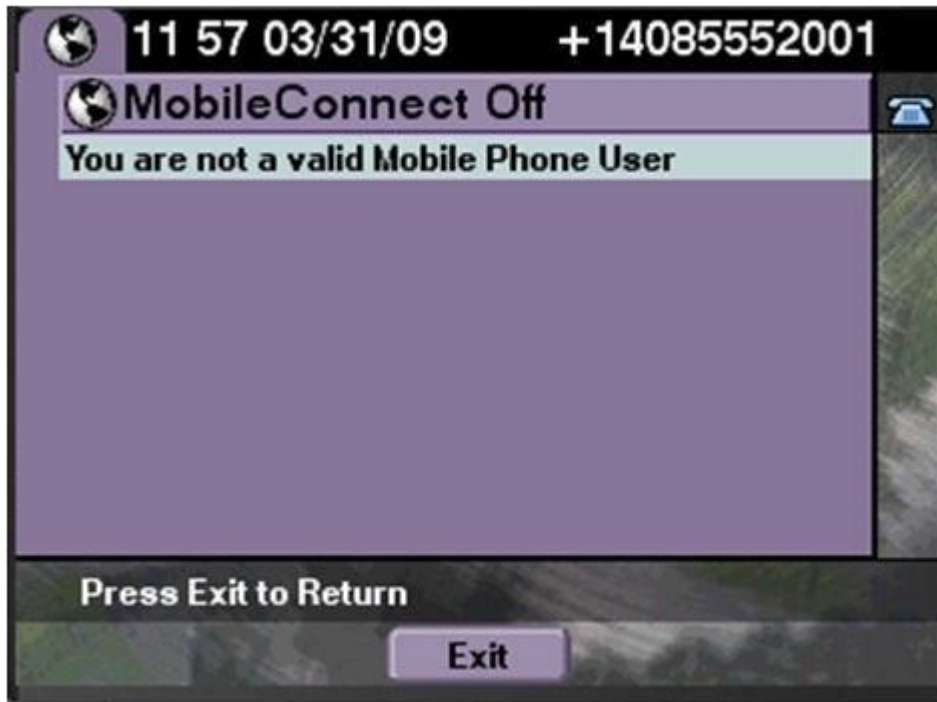**Vendor:** Cisco

Exam Code: 642-427

Exam Name: Troubleshooting Cisco Unified
Communications v8.0 (TVOICE v8.0)

Version: DEMO

QUESTION 1
Refer to the exhibit. Which course of action will resolve the Mobile Connect issues that are shown in the exhibit?

A. Configure the Mobility softkey on the phone.
B. Enable the user Cisco Mobile Connect
C. Make the user an owner of the phone device in phone device configuration page.
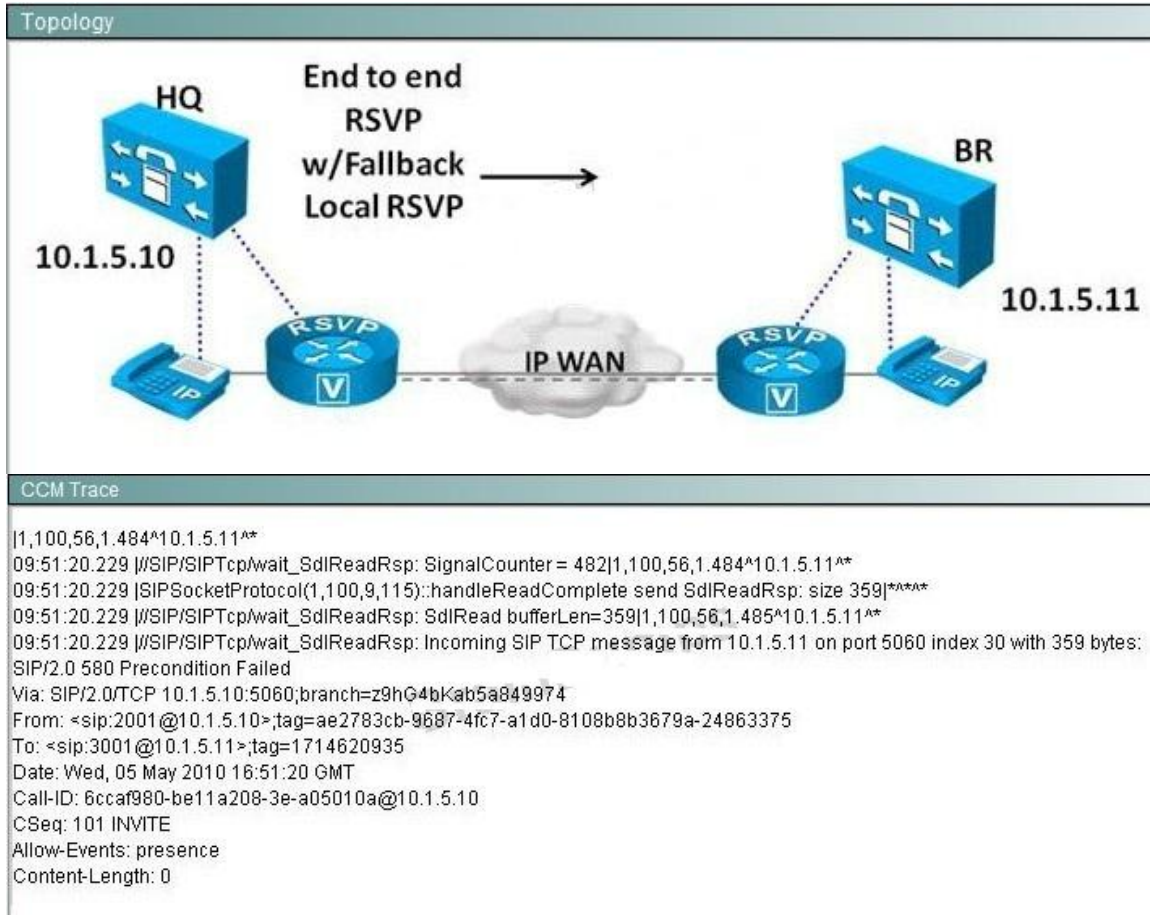D. Enable the device mobility mode on the phone since it is disable.

Answer: C
Explanation:
https://learningnetwork.cisco.com/docs/DOC-4140

QUESTION 2
Refer to the exhibits. End-to-end RSVP with local RSVP fallback has been configured on the HO Cisco Unified Communications Manager. RSVP between the locations that are assigned to the IP phones and SIP trunk at the HQ site is configured as mandatory. When a call is placed from an HQ phone to a BR phone, the end-to-end RSVP request fails.

Topology

End to end
RSVP
w/Fallback
Local RSVP →

HQ
10.1.5.10

BR
10.1.5.11

IP WAN

CCM Trace

|1,100,56,1.484^10.1.5.11^*
09:51:20.229 |//SIP/SIPTcp/wait_SdlReadRsp: SignalCounter = 482|1,100,56,1.484^10.1.5.11^*
09:51:20.229 |SIPSocketProtocol(1,100,9,115)::handleReadComplete send SdlReadRsp: size 359|*^*^*
09:51:20.229 |//SIP/SIPTcp/wait_SdlReadRsp: SdlRead bufferLen=359|1,100,56,1.485^10.1.5.11^*
09:51:20.229 |//SIP/SIPTcp/wait_SdlReadRsp: Incoming SIP TCP message from 10.1.5.11 on port 5060 index 30 with 359 bytes:
SIP/2.0 580 Precondition Failed
Via: SIP/2.0/TCP 10.1.5.10:5060;branch=z9hG4bKab5a849974
From: <sip:2001@10.1.5.10>;tag=ae2783cb-9687-4fc7-a1d0-8108b8b3679a-24863375
To: <sip:3001@10.1.5.11>;tag=1714620935
Date: Wed, 05 May 2010 16:51:20 GMT
Call-ID: 6ccaf980-be11a208-3e-a05010a@10.1.5.10
CSeq: 101 INVITE
Allow-Events: presence
Content-Length: 0

Refer to the SDI trace. Which statement about the call is true?

A. The Cisco Unified Communications Manager at HO will fall back to local RSVP and place the call No RSVP end-to-end will occur.
B. Only local RSVP will take place at both the HQ and BR Cisco Unified Communications Managers.
C. The Cisco Unified Communications Manager at HO will use end to end RSVP. The BR Cisco Unified Communications Manager will use local RSVP.
D. The call will fail.
E. The call will proceed as a normal call with no RSVP reservation.

Answer: D
Explanation:
We think the call will fail, because RSVP on the SIP Trunk is mandatory.The CCM Trace show an unsuccessful call setup with SIP Preconditions

QUESTION 3
In order for a third-party SIP phone to register with a Cisco Unified CallManager cluster, which three configuration parameters must match between the phone and the cluster? (Choose three.)

A. device pool
B. DN
C. username
D. password

E. MAC address
F. SIP profile

Answer: BCE

QUESTION 4
You have configured an ISR at a branch office to register as an Enhanced IOS media resource providing transcoding services. The transcoder has not been placed into a media resource group. Which statement best describes which devices will be able to utilize this transcoder?

A. No devices will be able to utilize the transcoder until it is placed in a media resource group.
B. Only devices at the branch office will be able to utilize the transcoder.
C. Only devices configured to use G.729 will be able to utilize the transcoder.
D. Only devices that have not been assigned an MRGL will be able to utilize the transcoder.
E. All devices will be able to utilize the transcoder.

Answer: E

QUESTION 5
You have configured the Enable Keep Alive check box under Trace Filter Settings. How does this change the trace output?

A. It adds TCP socket numbers between the endpoint and Cisco Unified CallManager for the session.
B. It maps the unique TCP handle for the endpoint to the MAC address of the endpoint in the trace output.
C. It adds the IP address of the endpoint in hex.
D. It adds the SCCP messages and all fields sent as part of that message.

Answer: D
Explanation:
Enable Keep Alive Trace Activates trace for keepalive trace information in theCisco CallManager traces. Because each SCCP device reports keepalive messagesevery 30 seconds, and each keepalive message creates 3 lines of trace data.

QUESTION 6
You have just obtained a list of the following options:

```
- All Patterns
- Unassigned DN
- Call Park
- Conference
- Directory Number
- Translation Pattern
- Call Pickup Group
- Route Pattern
- Message Waiting
- Voice Mail
- Attendant Console
```

What have you selected in order to produce this list?

A. Control Center > Feature Services

B. Dialed Number Analyzer
C. Route Plan > Route Plan Report
D. Route Plan > External Route Plan Wizard

Answer: C
Explanation:
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.
Viewing Route Plan Records
This section describes how to view route plan records. Because you might have several records in your network, Cisco CallManager Administration lets you locate specific route plan records on the basis of specific criteria. Use the following procedure to generate customized route plan reports.
Procedure
Step 1Choose Route Plan > Route Plan Report.
The Route Plan Report window displays. Use the three drop-down list boxes to specify a route plan report that meets your requirements.
Step 2 From the first Find drop-down list box, choose one of the following criteria:
http://www.cisco.com/en/US/docs/voice_ip_comm/cucm/admin/4_1_3/ccmcfg/b03rtrep.html

QUESTION 7
LSC validation is failing on a new SCCP IP phone you have just added to the Cisco Unified CallManager 5.0 cluster. No other IP phones are experiencing any problems with LSC validation. What can you do to help pinpoint the problem?

A. View the SDI trace output.
B. Check for security alarms.
C. Use the security configuration menu on the IP phone to verify that an LSC has been downloaded to the IP phone.
D. Verify that the authentication string is correct in the Cisco Unified CallManager device configuration screen.

Answer: C
Explanation:
Troubleshooting If the Locally Significant Certificate Validation Fails On the phone, the locally significant certificate validation may fail if the certificate is not the version that CAPF issued, the certificate has expired, the CAPF certificate does not exist on all servers in the cluster, the CAPF certificate does not exist in the CAPF directory, the phone is not registered to CiscoCallManager, and so on. If the locally significant certificate validation fails, review the SDL trace files and the CAPF trace files for errors.
You can verify that the locally significant certificate installed on the phone by choosingSettings > Model Informationand viewing the LSC setting. The LSC setting displaysInstalledorNot Installed, depending on the circumstances.
http://www.cisco.com/en/US/docs/voice_ip_comm/cucm/trouble/6_0_1/tbsystem.html#wpmkr 1159616

QUESTION 8
Which Cisco Unified Communications Manager troubleshooting tool can be used to look at detailed specific events, such as dial plan digit analysis, as they die happening?

A. traceroutes
B. RTMT real-time trace
C. Cisco Unified Communications Manager alerts
D. Cisco Unified Dialed Number Analyzer
E. RTMT performance log viewer
F. syslog output

Answer: B

QUESTION 9
An IP phone that is connected through a Cisco Catalyst 3750 Series Switch is failing to register with the subscriber as a backup server. When the user presses the settings button on the phone, only the Cisco Unified Communications Manager publisher shows as registered. What is the most likely cause for this issue?

A. The phone does not have the correct Cisco Unified Communications Manager group in the device configuration page.
B. The Cisco Unified Communications Manager group that is applied through the device pool is misconfigured.
C. The ip-helper address command for the subscriber is not configured on the switch port.
D. The subscriber does not have the correct device pool configured.
E. The enterprise phone configuration does not have the call control redundancy enabled.

Answer: B

QUESTION 10
Which step in the problem-solving model is important to accurately interview end users to get all the pertinent details of the problem?

A. Implement Action Plan
B. Define the Problem
C. Consider the Possibilities
D. Create Action Plan
E. Gather Facts
F. Observe Results
G. Restart Problem-Solving Process
H. Problem Resolved

Answer: E

QUESTION 11
You have received a trouble ticket stating that an IP phone is not working. When asked, the user informs you the phone is displaying the message "Registration rejected." Which two issues are possible causes of this problem? (Choose two.)

A. The IP phone is not getting an IP address.

B. The IP phone's primary Cisco Unified CallManager has a database replication issue.
C. The primary Cisco Unified CallManager is unavailable and the CallManager group assigned to the IP phone does not include a secondary CallManager.
D. The IP phone has not been defined in Cisco Unified CallManager.
E. The IP phone is not associated with a valid user profile.

Answer: BD

QUESTION 12
When using trace output to troubleshoot a Cisco Unified CallManager 5.0 problem, how can you collect and view the trace files?

A. Download the RTMT plug-in from the Cisco Unified CallManager Serviceability page to view the preconfigured trace files.
B. Configure the proper trace settings on the Cisco Unified CallManager Serviceability page and then use the embedded RTMT tool to view the trace files.
C. Configure the proper alarms and traces on the Cisco Unified CallManager Administration page and view the output with the RTMT plug-in.
D. Configure the proper trace settings on the Cisco Unified CallManager Serviceability page and download the RTMT plug-in from the CallManager Administration page to view the trace output.

Answer: D

QUESTION 13
Phone calls have been successful with acceptable audio quality over a WAN link. Recent configuration changes in the WAN routers have resulted in the inability to hear any audio on new calls across the WAN.
Which configuration changes directly cause a loss of audio?

A. a WAN router access list that blocks all UDP ports greater than 10,000
B. a WAN router access list that blocks all UDP ports lesser than 10,000
C. a WAN router access list that blocks all TCP ports greater than 10,000
D. a WAN router access list that blocks all TCP ports lesser than 10,000

Answer: A

QUESTION 14
Up to 10 phone calls over a 1 Mb/s WAN link have had consistently acceptable audio quality, even with considerable data traffic. Recently, when over 10 calls are made, audio quality has declined.
Which solution would you implement to maintain acceptable audio quality for the first 10 calls only while allowing for changes in WAN data utilization?

A. Cisco Unified Communications Manager CAC
B. RSVP CAC on the routers integrated with Cisco Unified Communications Manager
C. CAC implemented directly on the Cisco IP Phones
D. RSVP CAC operating on the routers independently of Cisco Unified Communications Manager

Answer: B

QUESTION 15
Refer to the exhibits. When a remote Cisco Unified Communications Manager learns the advertised patterns that are shown in the exhibit, which patterns would be shown in the Cisco Unified Communications Manager RTMT tool?

Domain_6.2_HostedDN

Hosted DN Pattern Info

Hosted Pattern*	2XXX
Description	
Hosted DN Group*	HQ_DN
PSTN Failover Strip Digits	0
PSTN Failover Prepend Digits	+498950555

☐ Use HostedDN as PSTN Failover

Domain_6.2_HostedGroup

Hosted DN Group Info

Name*	HQ_DN
Description	
PSTN Failover Strip Digits	0
PSTN Failover Prepend Digits	+498953121

☐ Use HostedDN as PSTN Failover

A. 2XXX and the ToDiD will be 0:+498950555
B. 2XXX and the ToDiD will be 0+498953121
C. +4989505552XXX and the ToDiD will be 0:
D. +498953121 2XXX and the ToDiD will be 0:
E. Both +4989505552XXX and +4989531 21 2XXX will be advertised with ToDID of 0:

Answer: A

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