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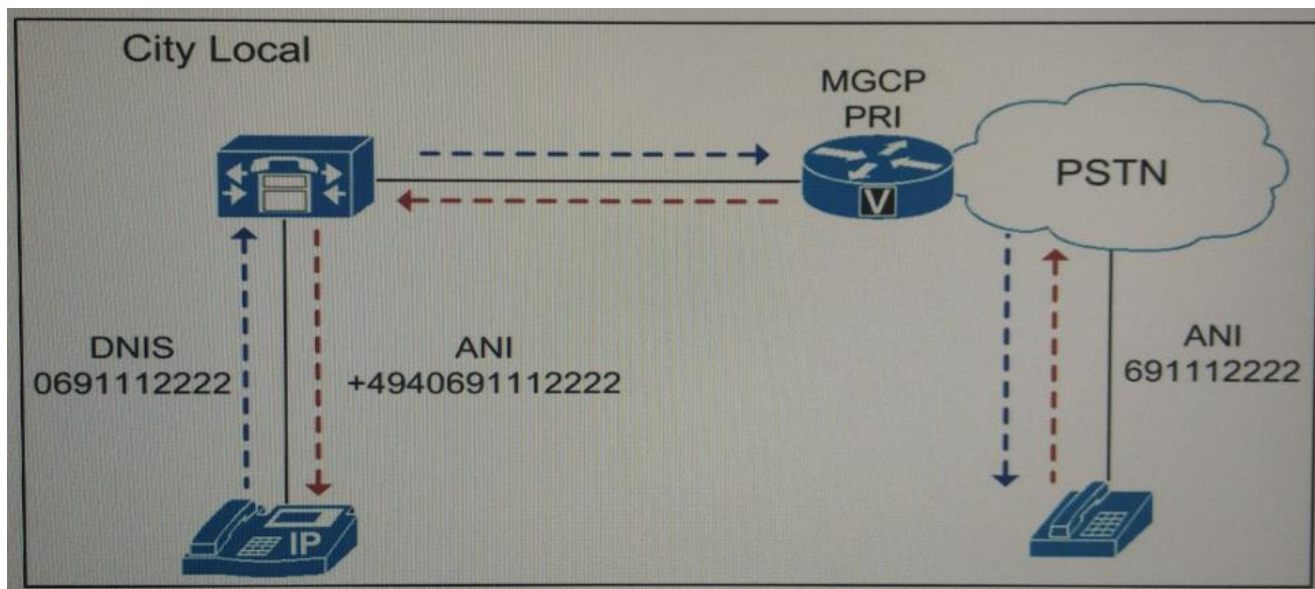
**Exam** : **400-051**

**Title** : **CCIE Collaboration Written**

**Vendor** : **Cisco**

**Version** : **DEMO**

**NO.1** Refer to the exhibit.



Refer to the exhibit. An engineer is working with globalization and has these requirements:

- A. Set the subscriber number prefix with +4940.
- B. allow redialing from the call history without manually manipulating the digit sting
- C. Set the prefix digit with 0 and calling number type national.
- D. Set the national number prefix with +4940.
- E. Set the calling-party transformation pattern with \+4940.!, and discard digits preDot.
- F. Set the prefix digit with 0 and calling number type subscriber
- G. the ANI to be presented in the E.164 format to the phone on the Cisco Unified Communications Manager
- H. Which three configuration steps in Cisco Unified Communications Manager are needed to meet these requirements? (Choose three)
- I. Set the calling party transformation pattern with /+4940.! And discard digits preAT

**Answer:** E,G,H

**NO.2** A client wants to play and compose voice messages from Microsoft Outlook. What is required for this functionality?

- A. Single inbox with ViewMail.
- B. Single inbox user message delivery with folder deletion.
- C. Single inbox with mailboxes larger than 2 G
- D. Single inbox synchronization with send and draft messages.

**Answer:** A

**NO.3** Refer to the exhibit.

<u>G.711 A-law Codec Enabled</u> *	Disabled
<u>G.711 mu-law Codec Enabled</u> *	Enabled for All Devices
<u>G.722 Codec Enabled</u> *	Disabled
<u>iLBC Codec Enabled</u> *	Enabled for All Devices
<u>iSAC Codec Enabled</u> *	Enabled for All Devices
<u>Default Intraregion Max Audio Bit Rate</u> *	64 kbps (G.722, G.711)
<u>Default Interregion Max Audio Bit Rate</u> *	64 kbps (G.722, G.711)
<u>Default Intraregion Max Video Call Bit Rate (Includes Audio)</u> *	384
<u>Default Interregion Max Video Call Bit Rate (Includes Audio)</u> *	384
<u>Use Video BandwidthPool for Immersive Video Calls</u> *	True
<u>Default Intraregion and Interregion Link Loss Type</u> *	Low Loss
<u>Default Audio Codec List between Regions</u> *	Factory Default low loss
<u>Default Audio Codec List within Region</u> *	Factory Default low loss
<u>Accept Audio Codec Preferences in Received Offer</u> *	Off

Refer to the exhibit. What is the preferred audio codec for calls between different regions?

- A. iSAC codec
- B. G711ulaw codec
- C. iLBC codec
- D. low loss configured codec

**Answer:** B

**NO.4** Which SIP request is used by Cisco Unified Communications Manager to signal DND status changes to a Cisco 9971 IP Phone?

- A. UPDATE
- B. NOTIFY
- C. OPTIONS

D. INFO

E. REFER

**Answer:** E

**NO.5** Which Software component in a Cisco Meeting Server deployment supports clustering for scalability and resilience?

A. Web Bridge

B. Web Admin

C. Database

D. XMPP Server

E. Call Bridge

F. TURN Server

**Answer:** E

**NO.6** Exhibit:

```

!
voice register dn 1
  number 2001
  call-forward b2bua busy 2100
  shared-line
  huntstop channel 6
!
voice register pool 1
  busy-trigger-per-button 4
  id mac 1111.1111.1111
  type 7965
  number 1 dn 1
!
voice register pool 2
  busy-trigger-per-button 5
  id mac 2222.2222.2222
  type 7965
  number 1 dn 1
!

```

How many simultaneous inbound calls can be handled by these two IP phones?

A. 9

B. 6

C. 2

D. 10

E. 4

**Answer:** C

**NO.7** Which description of route list digit manipulation behavior in Cisco unified Communications Manager is true?

- A. Called party transformations at route list level is not shown on the display of the calling phone
- B. Called party transformation at route list level is replaced by called party transformations at route pattern level
- C. Digit manipulations occur once per route list
- D. Called part transformations at route list level is reflected on the display of the calling phone
- E. Only called party transformation is available at route list level

**Answer:** D

**NO.8** Which two are the requirements of database cluster client and server certificates when configuring database clustering on Cisco Meeting Servers? (Choose two.)

- A. The CN of a database cluster client certificate must include the Call bridge server name
- B. The CN of a database cluster client certificate must include the domain name of the Call Bridge
- C. The CN of a database cluster client certificate must include the "postgres" keyword
- D. The CN of a database cluster server certificate must include the FQDN of the Call Bridge
- E. The CN of a database cluster server certificate must include the hostname of the Call Bridge
- F. The CN of a database cluster server certificate must include the domain name of the Call Bridge

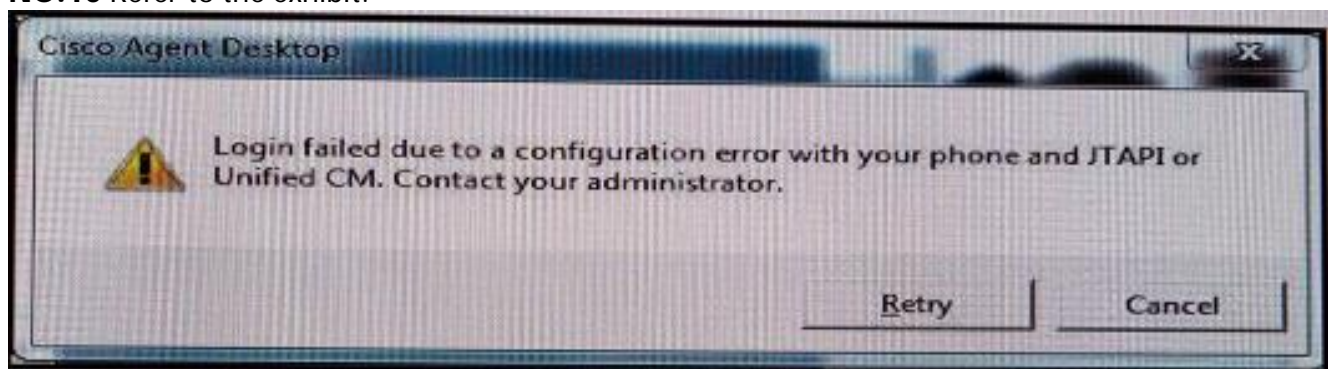
**Answer:** C,F

**NO.9** When a SIP message does not contain the "Allow" header, what is the sender telling the receiver about supported SIP methods?

- A. The sender supports all standard based SIP methods
- B. It is malformed SIP message
- C. It is coming from an intermediate SIP entity
- D. The sender does not support any SIP methods
- E. The sender is not providing any information on what methods it supports

**Answer:** E

**NO.10** Refer to the exhibit:



Refer to the exhibit, which two are possible causes of the error message when an agent attempted to log into the Cisco Agent Desktop? (Choose two)

- A. The RMCM subsystem is stuck in the initializing state.
- B. The resource is not available under Cisco Desktop Administrator.

- C. The incorrect extension was entered by the agent while logging onto Cisco Agent Desktop.
- D. The IPCC extension is not associated with the end user.
- E. The MAC of the agent phone is not associated with RMCM application user on the Cisco Unified Communications Manager.

**Answer:** A,E

**NO.11** Refer to the exhibit.

Input Message	Output Message
<pre> INVITE sip:2222@10.106.106.131:5060 SIP/2.0 Via: SIP/2.0/TCP 10.106.106.171:5060;branch=z9hG4bK8b337b7046 From: &lt;sip:5000@10.106.106.171&gt;;tag=67890~08d2afcc-154f-465b-8999-1889b75f43d8-17576409 To: &lt;sip:2222@10.106.106.131&gt; Date: Wed, 04 Nov 2015 10:21:52 GMT Call-ID: dc44e580-6391dc40-4f-ab6a6a0a@10.106.106.171 Supported: timer,resource-priority,replaces Min-SE: 1800 User-Agent: Cisco-CUCM10.5 Allow: INVITE, OPTIONS, INFO, BYE, CANCEL, ACK, PRACK, UPDATE, REFER, SUBSCRIBE, NOTIFY CSeq: 101 INVITE Expires: 180 Allow-Events: presence, kpml Supported: X-cisco-srtp-fallback Supported: Geolocation Call-Info: &lt;sip:10.106.106.171:5060&gt;;method="NOTIFY";Event=telephone-event;Duration=500 Call-Info: &lt;urn:x-cisco-remotecallinfo&gt;;x-cisco-video-traffic-class=VIDEO_UNSPECIFIED  Cisco-Guid: 3695502720-0000065536-0000000018-2875877898 Session-Expires: 1800 Diversion: &lt;sip:5002@10.106.106.171&gt;;reason=unconditional;privacy=off;screen=yes P-Asserted-Identity: &lt;sip:5000@10.106.106.171&gt; Remote-Party-ID: &lt;sip:5000@10.106.106.171&gt;;party=calling;screen=yes;privacy=off Contact: &lt;sip:5000@10.106.106.171:5060;transport=tcp&gt; Max-Forwards: 70 Content-Length: 0                     </pre>	<pre> INVITE sip:2222@10.106.106.131:5060 SIP/2.0 Via: SIP/2.0/TCP 10.106.106.171:5060;branch=z9hG4bK8b337b7046 From: &lt;sip:5002@10.106.106.171&gt;;tag=67890~08d2afcc-154f-465b-8999-1889b75f43d8-17576409 To: &lt;sip:2222@10.106.106.131&gt; Date: Wed, 04 Nov 2015 10:21:52 GMT Call-ID: dc44e580-6391dc40-4f-ab6a6a0a@10.106.106.171 Supported: timer,resource-priority,replaces Min-SE: 1800 User-Agent: Cisco-CUCM10.5 Allow: INVITE, OPTIONS, INFO, BYE, CANCEL, ACK, PRACK, UPDATE, REFER, SUBSCRIBE, NOTIFY CSeq: 101 INVITE Expires: 180 Allow-Events: presence, kpml Supported: X-cisco-srtp-fallback Supported: Geolocation Call-Info: &lt;sip:10.106.106.171:5060&gt;;method="NOTIFY";Event=telephone-event;Duration=500 Call-Info: &lt;urn:x-cisco-remotecallinfo&gt;;x-cisco-video-traffic-class=VIDEO_UNSPECIFIED  Cisco-Guid: 3695502720-0000065536-0000000018-2875877898 Session-Expires: 1800 Diversion: &lt;sip:5002@10.106.106.171&gt;;reason=unconditional;privacy=off;screen=yes P-Asserted-Identity: &lt;sip:5000@10.106.106.171&gt; Remote-Party-ID: &lt;sip:5000@10.106.106.171&gt;;party=calling;screen=yes;privacy=off Contact: &lt;sip:5000@10.106.106.171:5060;transport=tcp&gt; Max-Forwards: 70 Content-Length: 0                     </pre>

Refer to the exhibit. Which Cisco IOS SIP profile is valid for copying value from the "Diversion" header to the "From" header in a SIP INVITE message?

**A.** Option

```

voice class sip-profiles 1
 request INVITE sip-header Diversion copy "<sip:(.*)@.*" u01
 request INVITE sip-header From copy "<sip:(.*)@.*" u02
 request INVITE sip-header From modify "(.*)<sip:.*@(.*)" "\2<sip:\u01@1"
                    
```

**B.** Option

```

voice class sip-profiles 1
 request INVITE sip-header Diversion copy "<sip:(.*)@.*" u01
 request INVITE sip-header From copy "<sip:(.*)@.*" u02
 request INVITE sip-header From modify "(.*)<sip:.*@(.*)" "\1<sip:\u01@1"
                    
```

**C.** Option

```

voice class sip-profiles 1
 request INVITE sip-header Diversion copy "<sip:(.*)@.*" u02
 request INVITE sip-header From copy "<sip:(.*)@.*" u01
 request INVITE sip-header From modify "(.*)<sip:.*@(.*)" "\1<sip:\u01@2"
                    
```

**D.** Option

```

voice class sip-profiles 1
 request INVITE sip-header Diversion copy "<sip:(.*)@.*" u01
 request INVITE sip-header From copy ".*<sip:(.*)@.*" u02
 request INVITE sip-header From modify "(.*)<sip:.*@(.*)" "\1<sip:\u01@2"
                    
```

**E.** Option

```

voice class sip-profiles 1
 request INVITE sip-header Diversion copy "<sip:(.*)@.*" u01
 request INVITE sip-header From copy "<sip:(.*)@.*" u02
 request INVITE sip-header From modify "(.*)<sip:.*@(.*)" "\1<sip:\u02@2"
                    
```

**Answer:** D

**NO.12** Refer to the exhibit.

```
SIP/2.0 200 OK
Call-ID: b8bdd5f22eacf87c@127.0.0.1
CSeq: 12049 SERVICE
From: <sip:serviceproxy@10.10.10.1>;tag=7be6e74dbefae446
To: <sip:serviceserver@10.10.10.1>;tag=58c07d2f83ea795b
From: <sip:72253001@ucmpub>;tag=081196545e6500020000428b-00005ddf
To: <sip:cucm>
Route: <sip:ucmpub;transport=tcp;lr>
<?xml version="1.0" encoding="utf-8"?> <methodResponse><params><id>1571751309</id><result>success</result>
<sa>50ff19a9493b1f5670f47225047a77ab</sa></params><methodName>DigestSA</methodName><version>1.0</version>
<msgid>1408286445</msgid></methodResponse>
```

Refer to the exhibit. Which Mobile Remote Access (MRA) solution component is the recipient of this SIP message?

- A. CUCM
- B. Expressway-E
- C. Jabber client
- D. IM&P
- E. Expressway-C

**Answer: B**