# Troubleshooting Cisco Unified Communications v8.0 (TVOICE v8.0)

Number: 642-427
Passing Score: 800
Time Limit: 120 min
File Version: 1.0



**Troubleshooting Cisco Unified Communications v8.0 (TVOICE v8.0)**
**Version: 7.0**

**By J-Pack**

**Exam A**

**QUESTION 1**
Where does an IP phone obtain the extension number and speed-dial settings from?

A. the settings that are configured on the physical phone
B. the registration file that the phone receives from the Cisco Unified Communications Manager
C. the device and line configuration in Cisco Unified Communications Manager, during the registration process
D. the default device profile that is configured in Cisco Unified Communications, Manager

**Correct Answer:** C
**Section: (none)**
**Explanation**

**Explanation/Reference:**
**Explanation:** When we configure IP phone profile in CUCM that time we also configure extension number and speed dial as per requirement. When IP reachability gets establish between IP phone and CUCM then phone will download config file from CUCM during initial registration process. Link: http://www.cisco.com/en/US/docs/voice_ip_comm/cucm/admin/3_1_2/ccmcfg/b06phone.html

**QUESTION 2**
Which web-based application that is accessed via the Cisco Unified Communications Manager Administration GUI generates reports for troubleshooting or inspecting cluster data?

A. Cisco Unified Serviceability alarms
B. Cisco Unified RTMT Trace and Log Central
C. Cisco Unified RTMT Monitor
D. Cisco Unified Reporting tool

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

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

**QUESTION 3**
Which statement about device mobility is true?

A. When local route groups are used, there is no need to configure device mobility groups orphone device CSSs as long as phone line CSSs are used
B. When local route groups are used, you must configure device mobility groups and phone device CSSs
C. When the device mobility group at the home device pool and roaming device pool are not the same, the Phone will keep the home region.
D. When device mobility groups at the home device pool and roaming device pool are the same, the phone will keep the home MRGL setting.

**Correct Answer:** A
**Section: (none)**
**Explanation**

**Explanation/Reference:**
**Explanation:**

**QUESTION 4**

## Refer to the SDI trace in the exhibit:

A PSTN call arrived at the MGCP gateway that is shown in the SDI trace. If the caller ID that is displayed on the IP phone is 087071 222 and the HQ_clng__ pty_CSS contains the HQ_cing_pty_Pt partition, which exhibit shows the correct gateway digit manipulation"?

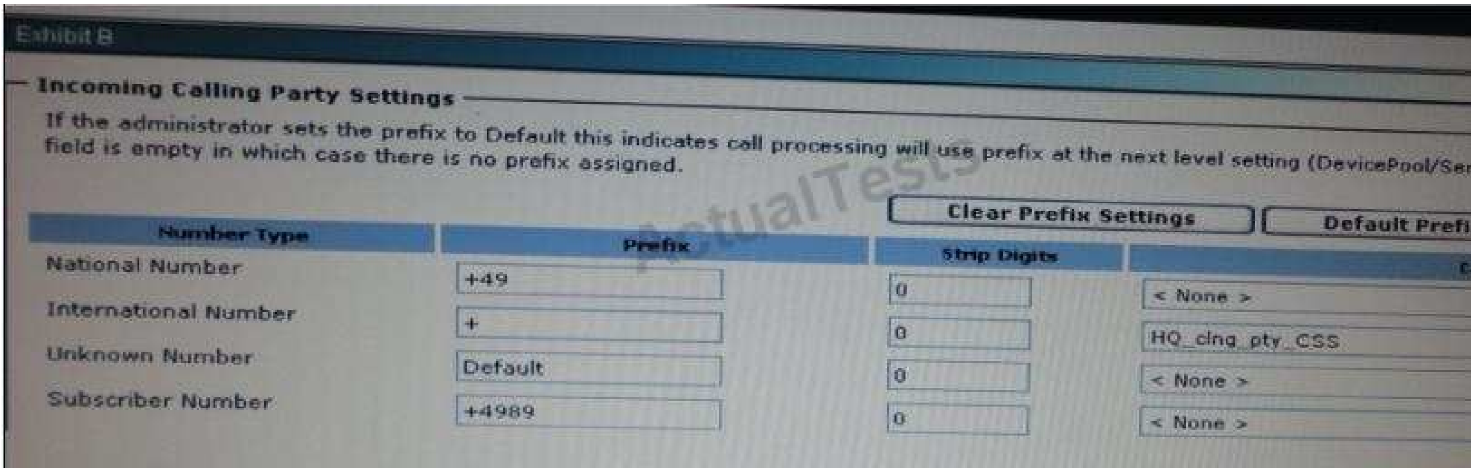


Exhibit B

Incoming Calling Party Settings

If the administrator sets the prefix to Default this indicates call processing will use prefix at the next level setting (DevicePool/Ser field is empty in which case there is no prefix assigned.

Clear Prefix Settings | Default Prefi

| Number Type | Prefix | Strip Digits | C |
|---|---|---|---|
| National Number | +49 | 0 | < None > |
| International Number | + | 0 | HQ_clng_pty_CSS |
| Unknown Number | Default | 0 | < None > |
| Subscriber Number | +4989 | 0 | < None > |

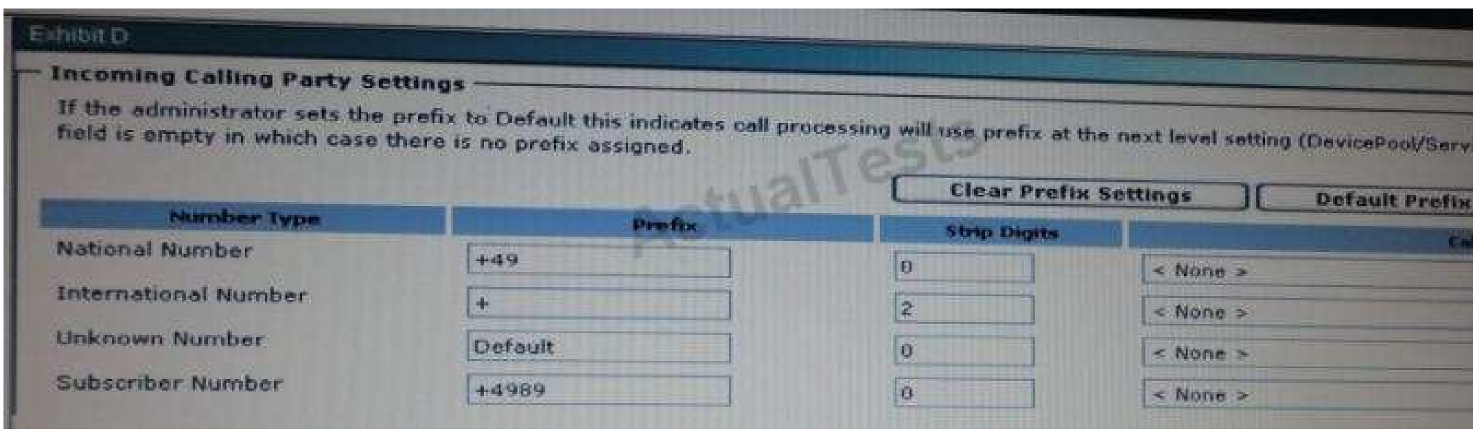


Exhibit D

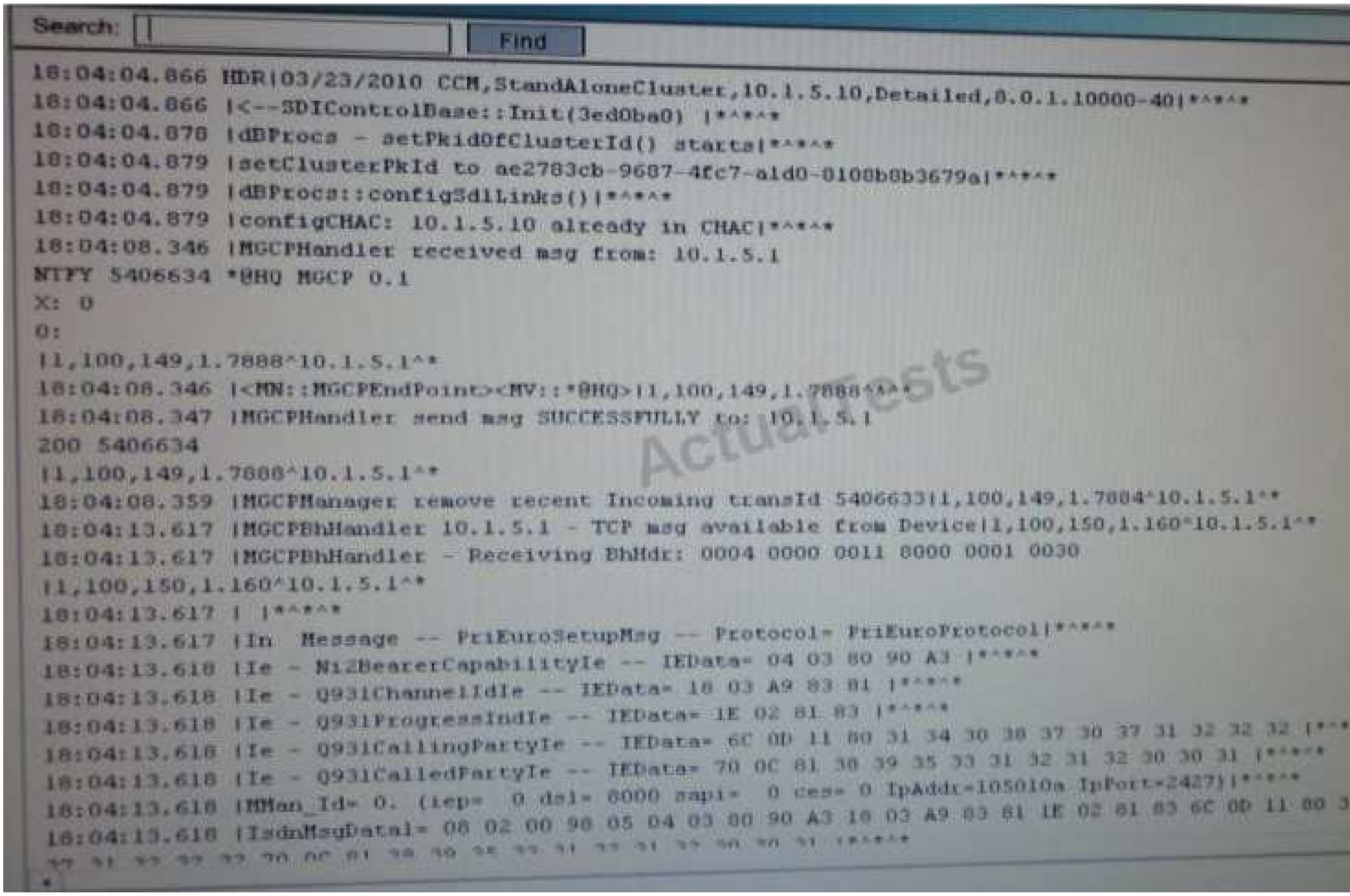
Incoming Calling Party Settings

If the administrator sets the prefix to Default this indicates call processing will use prefix at the next level setting (DevicePool/Serv field is empty in which case there is no prefix assigned.

Clear Prefix Settings | Default Prefix

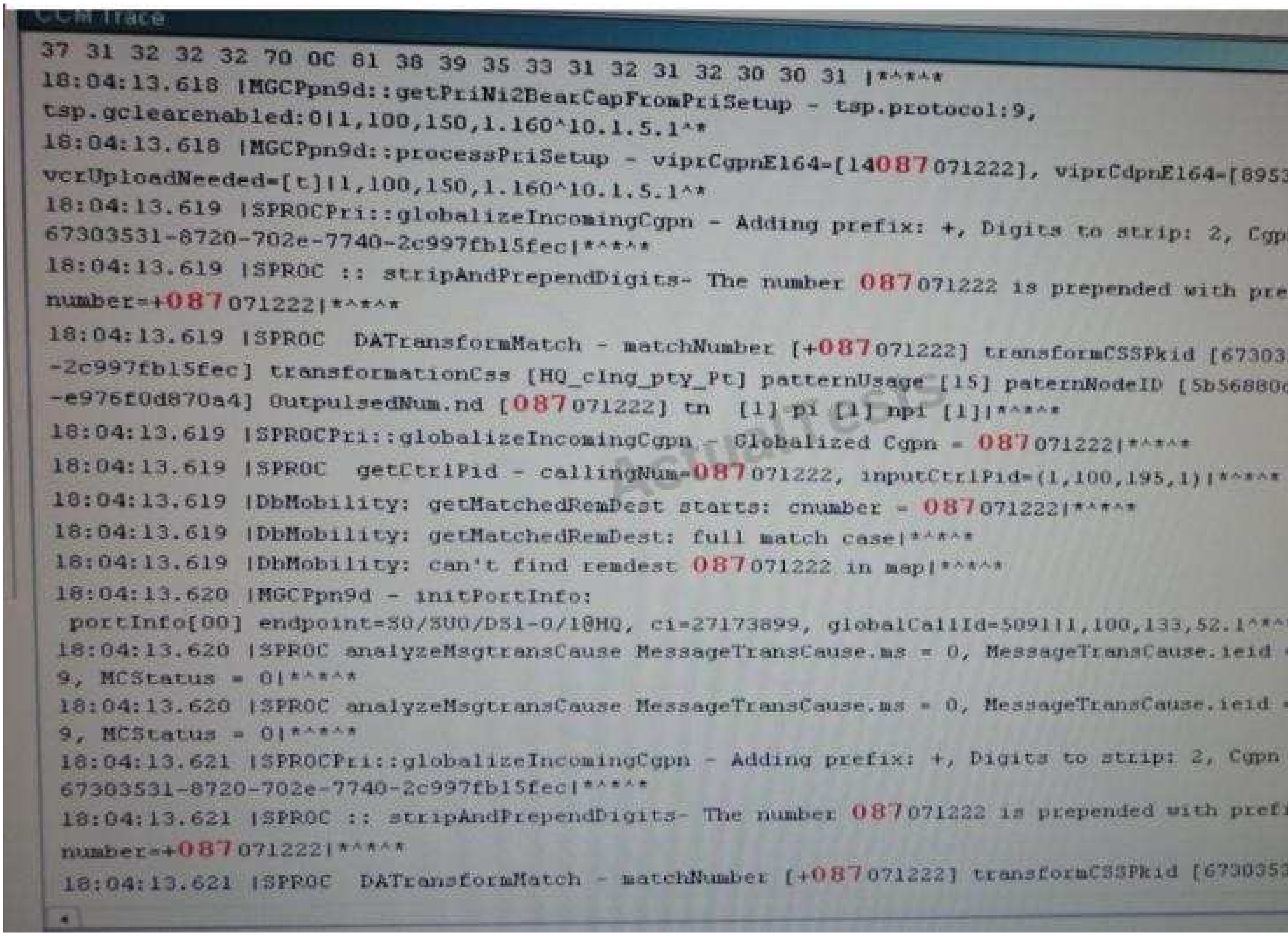
| Number Type | Prefix | Strip Digits | Ca |
|---|---|---|---|
| National Number | +49 | 0 | < None > |
| International Number | + | 2 | < None > |
| Unknown Number | Default | 0 | < None > |
| Subscriber Number | +4989 | 0 | < None > |

**1 (exhibit):**



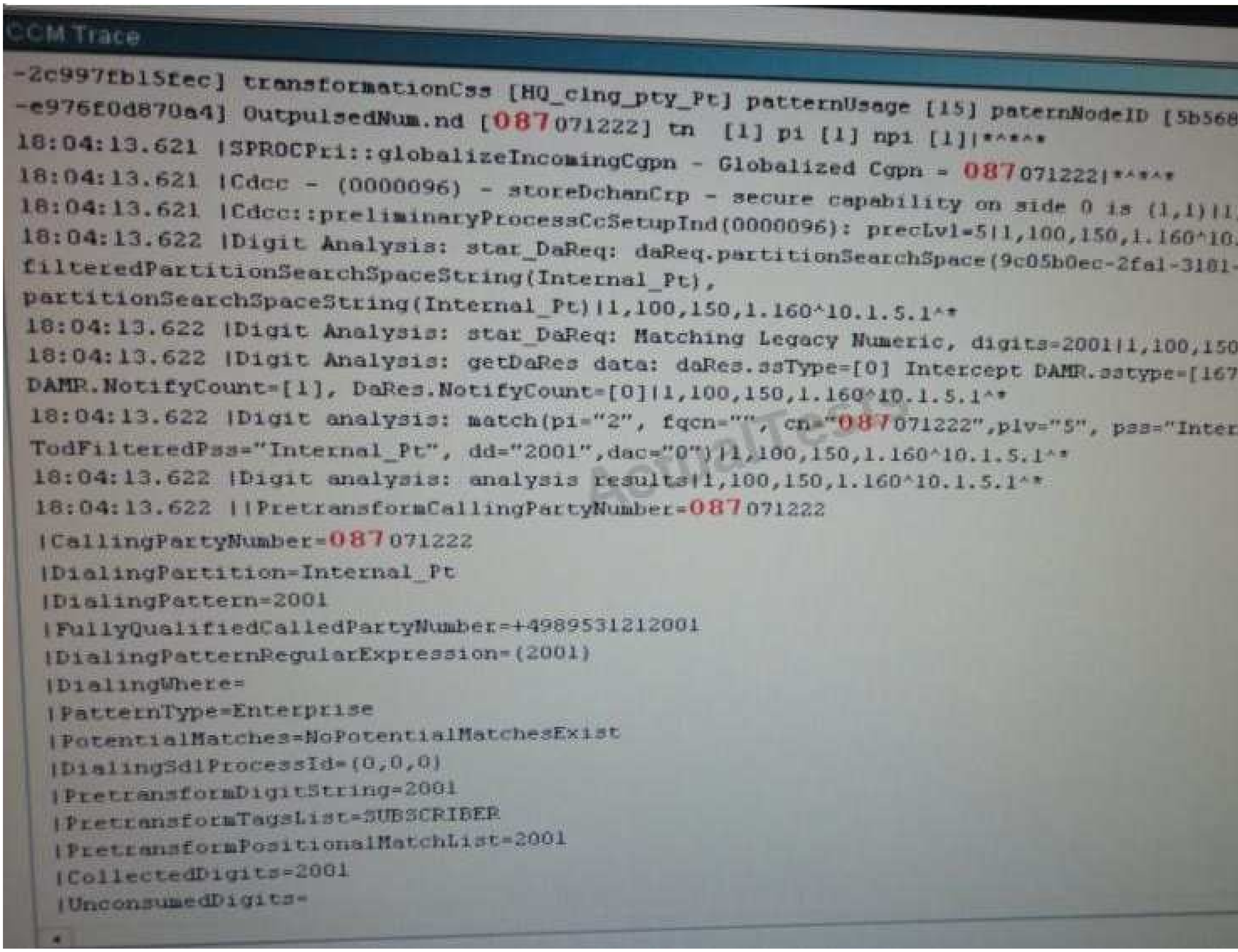
```
Search: [          ] Find
18:04:04.866 HDR|03/23/2010 CCM,StandAloneCluster,10.1.5.10,Detailed,8.0.1.10000-40|*^*^*
18:04:04.866 |<--SDIControlBase::Init(3ed0ba0) |*^*^*
18:04:04.878 |dBProcs - setPkidOfClusterId() starts|*^*^*
18:04:04.879 |setClusterPkId to ae2783cb-9687-4fc7-a1d0-0108b8b3679a|*^*^*
18:04:04.879 |dBProcs::configSdlLinks()|*^*^*
18:04:04.879 |configCHAC: 10.1.5.10 already in CHAC|*^*^*
18:04:08.346 |MGCPHandler received msg from: 10.1.5.1
NTFY 5406634 *@HQ MGCP 0.1
X: 0
O:
|1,100,149,1.7888^10.1.5.1^*
18:04:08.346 |<MN::MGCPEndPoint><MV::*@HQ>|1,100,149,1.7888^*^*
18:04:08.347 |MGCPHandler send msg SUCCESSFULLY to: 10.1.5.1
200 5406634
|1,100,149,1.7888^10.1.5.1^*
18:04:08.359 |MGCPManager remove recent Incoming transId 5406633|1,100,149,1.7884^10.1.5.1^*
18:04:13.617 |MGCPBhHandler 10.1.5.1 - TCP msg available from Device|1,100,150,1.160^10.1.5.1^*
18:04:13.617 |MGCPBhHandler - Receiving BhHdr: 0004 0000 0011 8000 0001 0030
|1,100,150,1.160^10.1.5.1^*
18:04:13.617 | |*^*^*
18:04:13.617 |In  Message -- PriEuroSetupMsg -- Protocol= PriEuroProtocol|*^*^*
18:04:13.618 |Ie - Ni2BearerCapabilityIe -- IEData= 04 03 80 90 A3 |*^*^*
18:04:13.618 |Ie - Q931ChannelIdIe -- IEData= 18 03 A9 83 81 |*^*^*
18:04:13.618 |Ie - Q931ProgressIndIe -- IEData= 1E 02 81 83 |*^*^*
18:04:13.618 |Ie - Q931CallingPartyIe -- IEData= 6C 0D 11 80 31 34 30 38 37 30 37 31 32 32 32 |*^*
18:04:13.618 |Ie - Q931CalledPartyIe -- IEData= 70 0C 81 38 39 35 33 31 32 31 32 30 30 31 |*^*^*
18:04:13.618 |MMan_Id= 0. (iep= 0 dsl= 8000 sapi= 0 ces= 0 IpAddr=105010a IpPort=2427)|*^*^*
18:04:13.618 |IsdnMsgData1= 08 02 00 98 05 04 03 80 90 A3 18 03 A9 83 81 1E 02 81 83 6C 0D 11 80 3
18:04:13.618 |IsdnMsgData1= 08 02 00 98 05 04 03 80 90 A3 18 03 A9 83 81 1E 02 81 83 6C 0D 11 80 3
```

**2 (exhibit):**



```
CCM Trace
37 31 32 32 32 70 0C 81 38 39 35 33 31 32 31 32 30 30 31 |*^*^*
18:04:13.618 |MGCPpn9d::getPriNi2BearCapFromPriSetup - tsp.protocol:9,
tsp.gclearenabled:0|1,100,150,1.160^10.1.5.1^*
18:04:13.618 |MGCPpn9d::processPriSetup - viprCgpnE164=[14087071222], viprCdpnE164=[8953
vcrUploadNeeded=[t]|1,100,150,1.160^10.1.5.1^*
18:04:13.619 |SPROCPri::globalizeIncomingCgpn - Adding prefix: +, Digits to strip: 2, Cgp
67303531-8720-702e-7740-2c997fb15fec|*^*^*
18:04:13.619 |SPROC :: stripAndPrependDigits- The number 087071222 is prepended with pre
number=+087071222|*^*^*
18:04:13.619 |SPROC  DATransformMatch - matchNumber [+087071222] transformCSSPkid [67303
-2c997fb15fec] transformationCss [HQ_clng_pty_Pt] patternUsage [15] paternNodeID [5b56880
-e976f0d870a4] OutpulsedNum.nd [087071222] tn  [1] pi [1] npi [1]|*^*^*
18:04:13.619 |SPROCPri::globalizeIncomingCgpn - Globalized Cgpn = 087071222|*^*^*
18:04:13.619 |SPROC  getCtrlPid - callingNum=087071222, inputCtrlPid=(1,100,195,1)|*^*^*
18:04:13.619 |DbMobility: getMatchedRemDest starts: cnumber = 087071222|*^*^*
18:04:13.619 |DbMobility: getMatchedRemDest: full match case|*^*^*
18:04:13.619 |DbMobility: can't find remdest 087071222 in map|*^*^*
18:04:13.620 |MGCPpn9d - initPortInfo:
 portInfo[00] endpoint=S0/SU0/DS1-0/1@HQ, ci=27173899, globalCallId=509111,100,133,52.1^*^
18:04:13.620 |SPROC analyzeMsgtransCause MessageTransCause.ms = 0, MessageTransCause.ieid
9, MCStatus = 0|*^*^*
18:04:13.620 |SPROC analyzeMsgtransCause MessageTransCause.ms = 0, MessageTransCause.ieid
9, MCStatus = 0|*^*^*
18:04:13.621 |SPROCPri::globalizeIncomingCgpn - Adding prefix: +, Digits to strip: 2, Cgpn
67303531-8720-702e-7740-2c997fb15fec|*^*^*
18:04:13.621 |SPROC :: stripAndPrependDigits- The number 087071222 is prepended with prefi
number=+087071222|*^*^*
18:04:13.621 |SPROC  DATransformMatch - matchNumber [+087071222] transformCSSPkid [6730353
```

**3 (exhibit):**



CCM Trace

```
-2c997fb15fec] transformationCss [HQ_cing_pty_Pt] patternUsage [15] paternNodeID [5b568
-e976f0d870a4] OutpulsedNum.nd [087071222] tn  [1] pi [1] npi [1]|*^*^*
18:04:13.621 |SPROCPri::globalizeIncomingCgpn - Globalized Cgpn = 087071222|*^*^*
18:04:13.621 |Cdcc - (0000096) - storeDchanCrp - secure capability on side 0 is (1,1)|1
18:04:13.621 |Cdcc::preliminaryProcessCcSetupInd(0000096): precLvl=5|1,100,150,1.160^10
18:04:13.622 |Digit Analysis: star_DaReq: daReq.partitionSearchSpace(9c05b0ec-2fa1-3181-
filteredPartitionSearchSpaceString(Internal_Pt),
partitionSearchSpaceString(Internal_Pt)|1,100,150,1.160^10.1.5.1^*
18:04:13.622 |Digit Analysis: star_DaReq: Matching Legacy Numeric, digits=2001|1,100,150
18:04:13.622 |Digit Analysis: getDaRes data: daRes.ssType=[0] Intercept DAMR.sstype=[167
DAMR.NotifyCount=[1], DaRes.NotifyCount=[0]|1,100,150,1.160^10.1.5.1^*
18:04:13.622 |Digit analysis: match(pi="2", fqcn="", cn="087071222",plv="5", pss="Inter
TodFilteredPss="Internal_Pt", dd="2001",dac="0")|1,100,150,1.160^10.1.5.1^*
18:04:13.622 |Digit analysis: analysis results|1,100,150,1.160^10.1.5.1^*
18:04:13.622 ||PretransformCallingPartyNumber=087071222
|CallingPartyNumber=087071222
|DialingPartition=Internal_Pt
|DialingPattern=2001
|FullyQualifiedCalledPartyNumber=+4989531212001
|DialingPatternRegularExpression=(2001)
|DialingWhere=
|PatternType=Enterprise
|PotentialMatches=NoPotentialMatchesExist
|DialingSdlProcessId=(0,0,0)
|PretransformDigitString=2001
|PretransformTagsList=SUBSCRIBER
|PretransformPositionalMatchList=2001
|CollectedDigits=2001
|UnconsumedDigits=
```

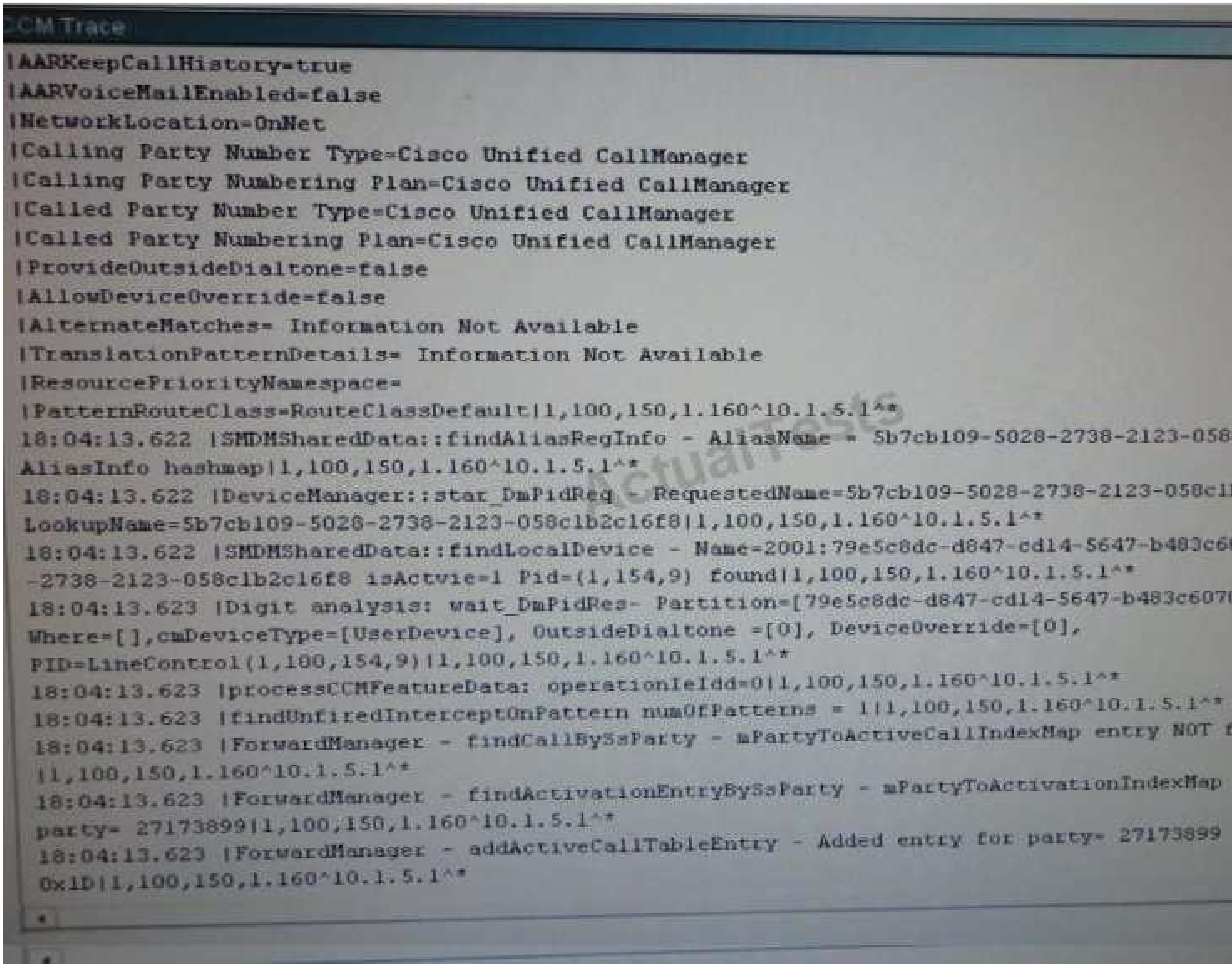
**4 (exhibit):**

CCM Trace

```
|TagsList=SUBSCRIBER
|PositionalMatchList=2001
|VoiceMailbox=
|VoiceMailCallingSearchSpace=Internal_Pt
|VoiceMailPilotNumber=2000
|RouteBlockFlag=RouteThisPattern
|RouteBlockCause=0
|AlertingName=
|UnicodeDisplayName=
|DisplayNameLocale=1
|InterceptPartition=Internal_Pt
|InterceptPattern=2001
|InterceptWhere=
|InterceptSdlProcessId=(0,0,0)
|InterceptSsType=16777222
|InterceptSsKey=0
|InterceptSsNotifyType=1
|OverlapSendingFlagEnabled=0
|WithTags=
|WithValues=
|CallingPartyNumberPi=NotSelected
|ConnectedPartyNumberPi=NotSelected
|CallingPartyNamePi=NotSelected
|ConnectedPartyNamePi=NotSelected
|CallManagerDeviceType=NoDeviceType
|PatternPrecedenceLevel=Routine
|CallableEndPointName=[5b7cb109-5028-2738-2123-058c1b2c16f8]
|PatternNodeId=[5b7cb109-5028-2738-2123-058c1b2c16f8]
|AARNeighborhood=[]
|AARDestinationMask=[]
```



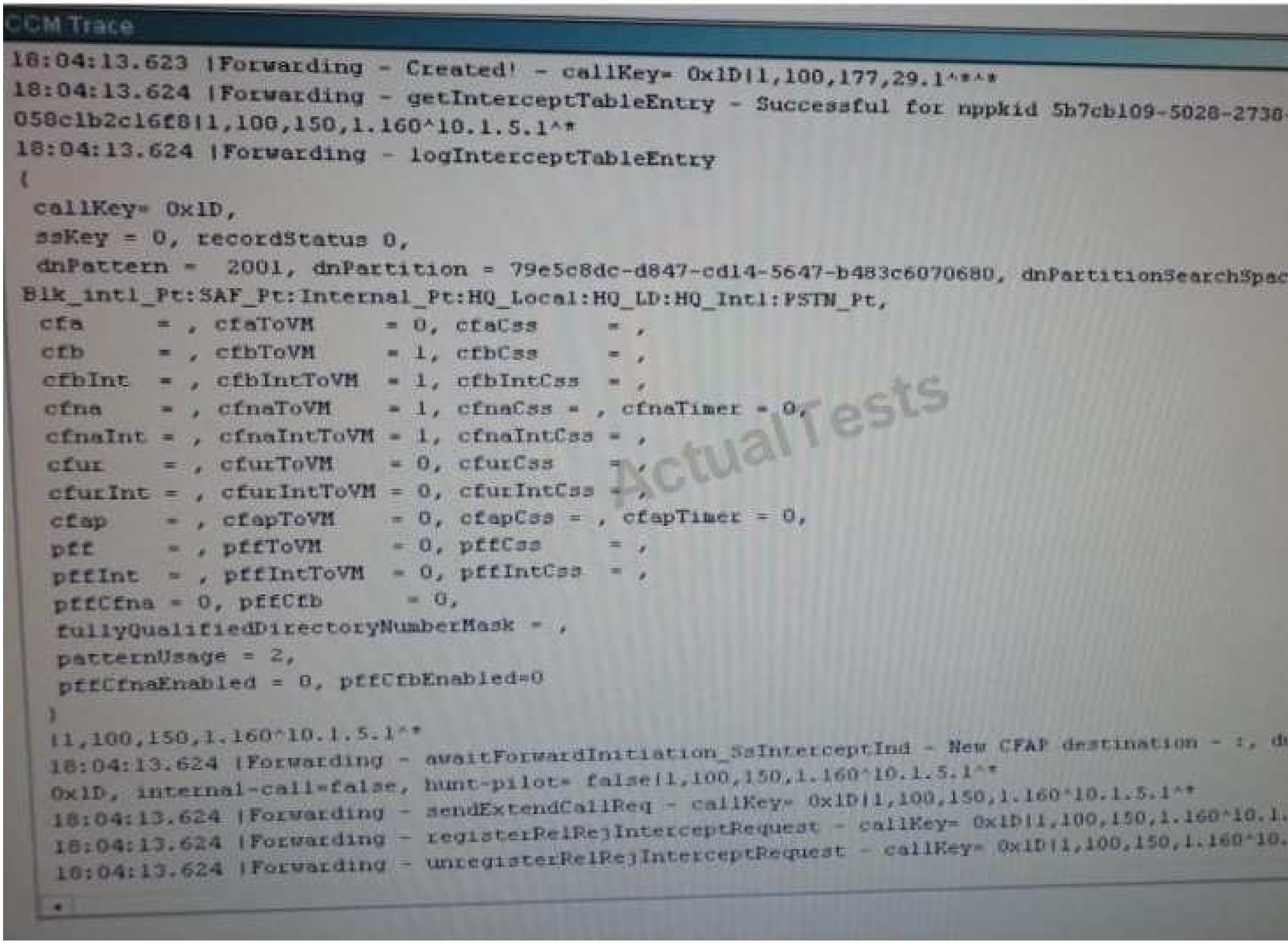
**5 (exhibit):**



CCM Trace

```
|AARKeepCallHistory=true
|AARVoiceMailEnabled=false
|NetworkLocation=OnNet
|Calling Party Number Type=Cisco Unified CallManager
|Calling Party Numbering Plan=Cisco Unified CallManager
|Called Party Number Type=Cisco Unified CallManager
|Called Party Numbering Plan=Cisco Unified CallManager
|ProvideOutsideDialtone=false
|AllowDeviceOverride=false
|AlternateMatches= Information Not Available
|TranslationPatternDetails= Information Not Available
|ResourcePriorityNamespace=
|PatternRouteClass=RouteClassDefault|1,100,150,1.160^10.1.5.1^*
18:04:13.622 |SMDMSharedData::findAliasRegInfo - AliasName = 5b7cb109-5028-2738-2123-058
AliasInfo hashmap|1,100,150,1.160^10.1.5.1^*
18:04:13.622 |DeviceManager::star_DmPidReq - RequestedName=5b7cb109-5028-2738-2123-058c1
LookupName=5b7cb109-5028-2738-2123-058c1b2c16f8|1,100,150,1.160^10.1.5.1^*
18:04:13.622 |SMDMSharedData::findLocalDevice - Name=2001:79e5c8dc-d847-cd14-5647-b483c6
-2738-2123-058c1b2c16f8 isActvie=1 Pid=(1,154,9) found|1,100,150,1.160^10.1.5.1^*
18:04:13.623 |Digit analysis: wait_DmPidRes- Partition=[79e5c8dc-d847-cd14-5647-b483c607
Where=[],cmDeviceType=[UserDevice], OutsideDialtone =[0], DeviceOverride=[0],
PID=LineControl(1,100,154,9)|1,100,150,1.160^10.1.5.1^*
18:04:13.623 |processCCMFeatureData: operationIeIdd=0|1,100,150,1.160^10.1.5.1^*
18:04:13.623 |findUnfiredInterceptOnPattern numOfPatterns = 1|1,100,150,1.160^10.1.5.1^*
18:04:13.623 |ForwardManager - findCallBySsParty - mPartyToActiveCallIndexMap entry NOT f
|1,100,150,1.160^10.1.5.1^*
18:04:13.623 |ForwardManager - findActivationEntryBySsParty - mPartyToActivationIndexMap
party= 27173899|1,100,150,1.160^10.1.5.1^*
18:04:13.623 |ForwardManager - addActiveCallTableEntry - Added entry for party= 27173899
0x1D|1,100,150,1.160^10.1.5.1^*
```

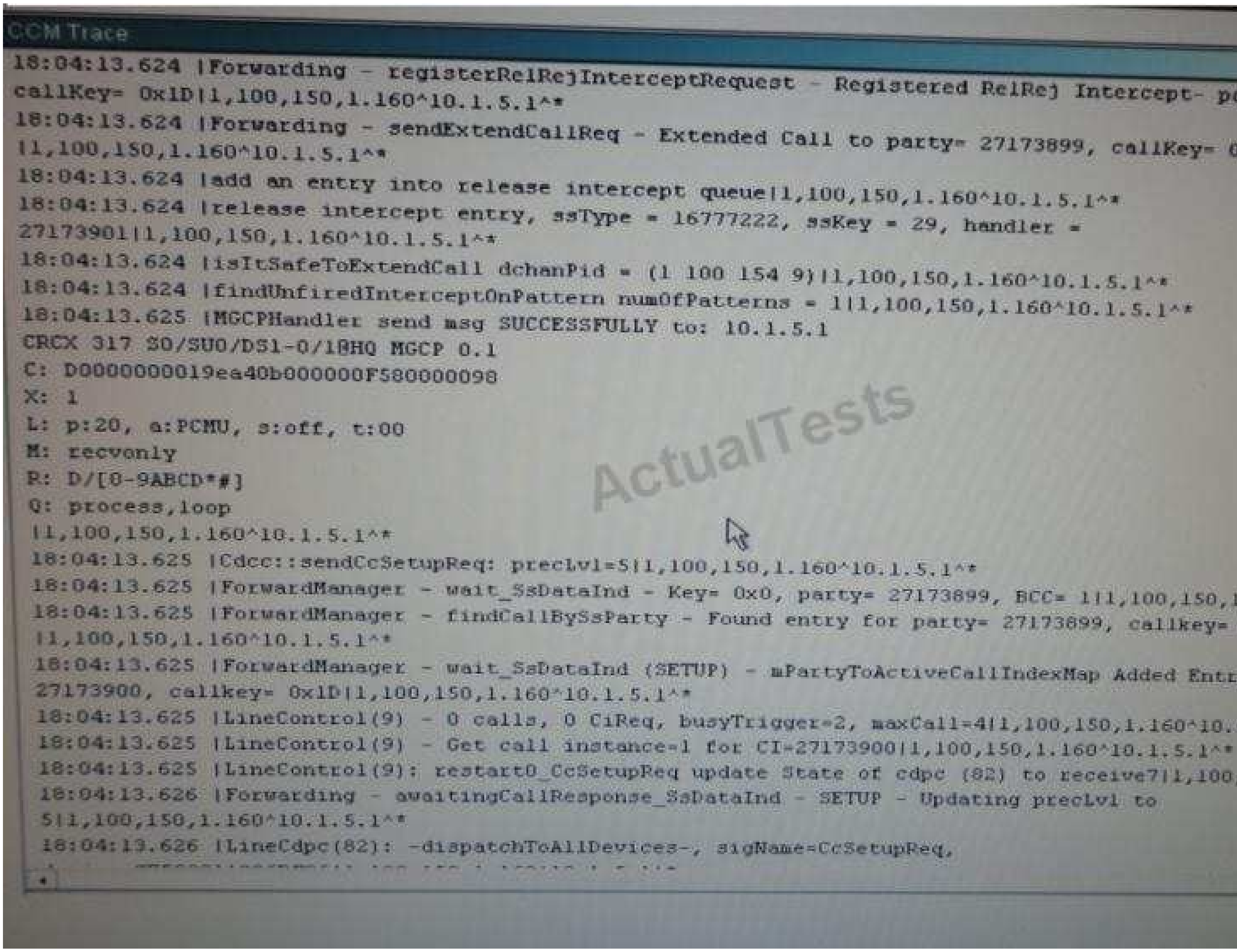
**6 (exhibit):**



CCM Trace

```
18:04:13.623 |Forwarding - Created! - callKey= 0x1D|1,100,177,29.1^*^*
18:04:13.624 |Forwarding - getInterceptTableEntry - Successful for nppkid 5b7cb109-5028-2738-
058c1b2c16f8|1,100,150,1.160^10.1.5.1^*
18:04:13.624 |Forwarding - logInterceptTableEntry
{
 callKey= 0x1D,
 ssKey = 0, recordStatus 0,
 dnPattern =  2001, dnPartition = 79e5c8dc-d847-cd14-5647-b483c6070680, dnPartitionSearchSpace
Blk_intl_Pt:SAF_Pt:Internal_Pt:HQ_Local:HQ_LD:HQ_Intl:PSTN_Pt,
 cfa      = , cfaToVM     = 0, cfaCss      = ,
 cfb      = , cfbToVM     = 1, cfbCss      = ,
 cfbInt   = , cfbIntToVM  = 1, cfbIntCss   = ,
 cfna     = , cfnaToVM    = 1, cfnaCss = , cfnaTimer = 0,
 cfnaInt = , cfnaIntToVM = 1, cfnaIntCss = ,
 cfur     = , cfurToVM    = 0, cfurCss     = ,
 cfurInt = , cfurIntToVM = 0, cfurIntCss = ,
 cfap     = , cfapToVM    = 0, cfapCss = , cfapTimer = 0,
 pff      = , pffToVM     = 0, pffCss      = ,
 pffInt   = , pffIntToVM  = 0, pffIntCss   = ,
 pffCfna = 0, pffCfb      = 0,
 fullyQualifiedDirectoryNumberMask = ,
 patternUsage = 2,
 pffCfnaEnabled = 0, pffCfbEnabled=0
}
|1,100,150,1.160^10.1.5.1^*
18:04:13.624 |Forwarding - awaitForwardInitiation_SsInterceptInd - New CFAP destination - :, du
0x1D, internal-call=false, hunt-pilot= false|1,100,150,1.160^10.1.5.1^*
18:04:13.624 |Forwarding - sendExtendCallReq - callKey= 0x1D|1,100,150,1.160^10.1.5.1^*
18:04:13.624 |Forwarding - registerRelRejInterceptRequest - callKey= 0x1D|1,100,150,1.160^10.1.
18:04:13.624 |Forwarding - unregisterRelRejInterceptRequest - callKey= 0x1D|1,100,150,1.160^10.
18:04:13.624 |Forwarding -
```

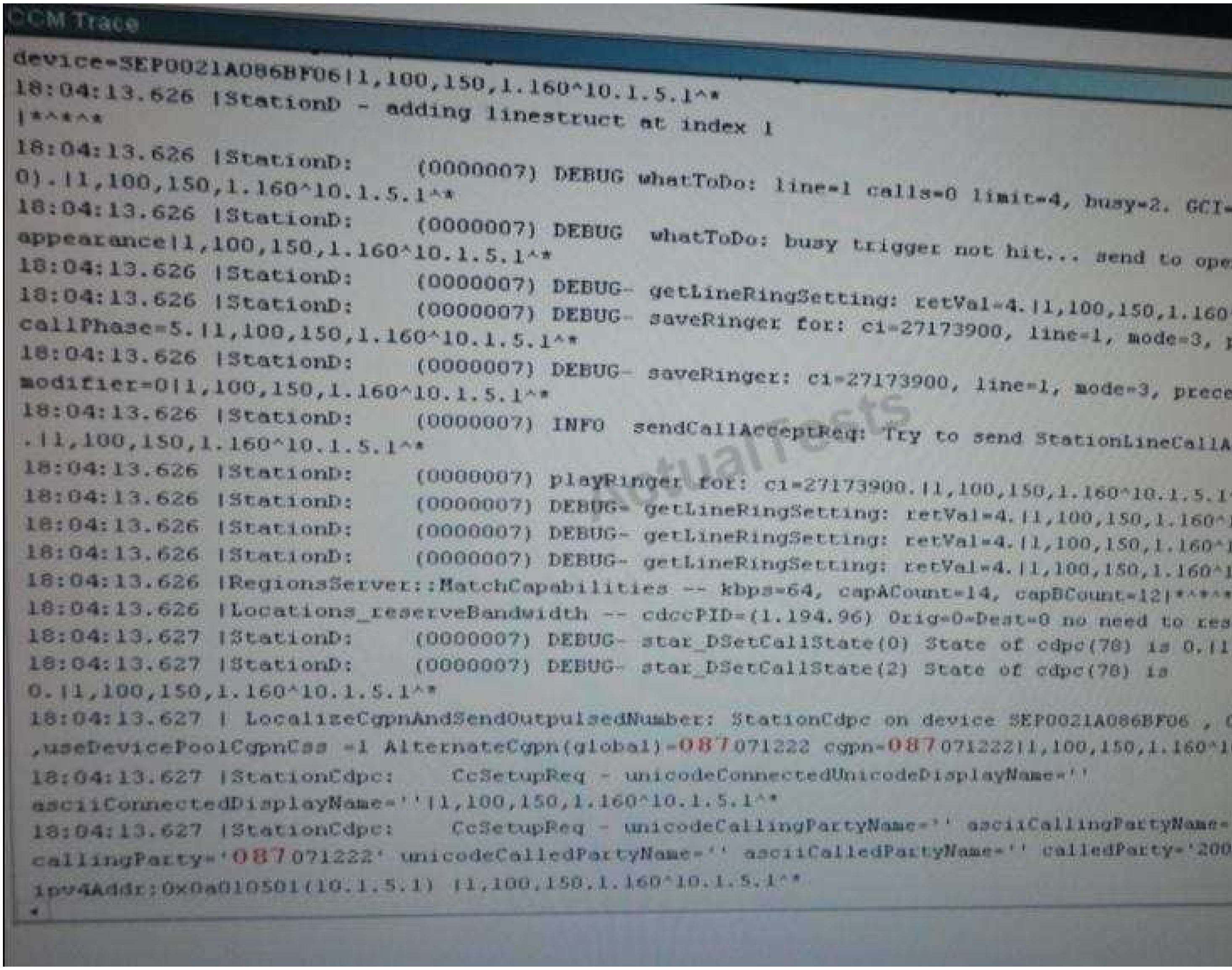
**7 (exhibit):**



```
CCM Trace
18:04:13.624 |Forwarding - registerRelRejInterceptRequest - Registered RelRej Intercept- pa
callKey= 0x1D|1,100,150,1.160^10.1.5.1^*
18:04:13.624 |Forwarding - sendExtendCallReq - Extended Call to party= 27173899, callKey= 0x
|1,100,150,1.160^10.1.5.1^*
18:04:13.624 |add an entry into release intercept queue|1,100,150,1.160^10.1.5.1^*
18:04:13.624 |release intercept entry, ssType = 16777222, ssKey = 29, handler =
27173901|1,100,150,1.160^10.1.5.1^*
18:04:13.624 |isItSafeToExtendCall dchanPid = (1 100 154 9)|1,100,150,1.160^10.1.5.1^*
18:04:13.624 |findUnfiredInterceptOnPattern numOfPatterns = 1|1,100,150,1.160^10.1.5.1^*
18:04:13.625 |MGCPHandler send msg SUCCESSFULLY to: 10.1.5.1
CRCX 317 S0/SU0/DS1-0/18HQ MGCP 0.1
C: D0000000019ea40b000000F580000098
X: 1
L: p:20, a:PCMU, s:off, t:00
M: recvonly
R: D/[0-9ABCD*#]
Q: process,loop
|1,100,150,1.160^10.1.5.1^*
18:04:13.625 |Cdcc::sendCcSetupReq: precLvl=5|1,100,150,1.160^10.1.5.1^*
18:04:13.625 |ForwardManager - wait_SsDataInd - Key= 0x0, party= 27173899, BCC= 1|1,100,150,1
18:04:13.625 |ForwardManager - findCallBySsParty - Found entry for party= 27173899, callkey=
|1,100,150,1.160^10.1.5.1^*
18:04:13.625 |ForwardManager - wait_SsDataInd (SETUP) - mPartyToActiveCallIndexMap Added Entry
27173900, callkey= 0x1D|1,100,150,1.160^10.1.5.1^*
18:04:13.625 |LineControl(9) - 0 calls, 0 CiReq, busyTrigger=2, maxCall=4|1,100,150,1.160^10.1
18:04:13.625 |LineControl(9) - Get call instance=1 for CI=27173900|1,100,150,1.160^10.1.5.1^*
18:04:13.625 |LineControl(9): restart0_CcSetupReq update State of cdpc (82) to receive7|1,100,
18:04:13.626 |Forwarding - awaitingCallResponse_SsDataInd - SETUP - Updating precLvl to
5|1,100,150,1.160^10.1.5.1^*
18:04:13.626 |LineCdpc(82): -dispatchToAllDevices-, sigName=CcSetupReq,
```

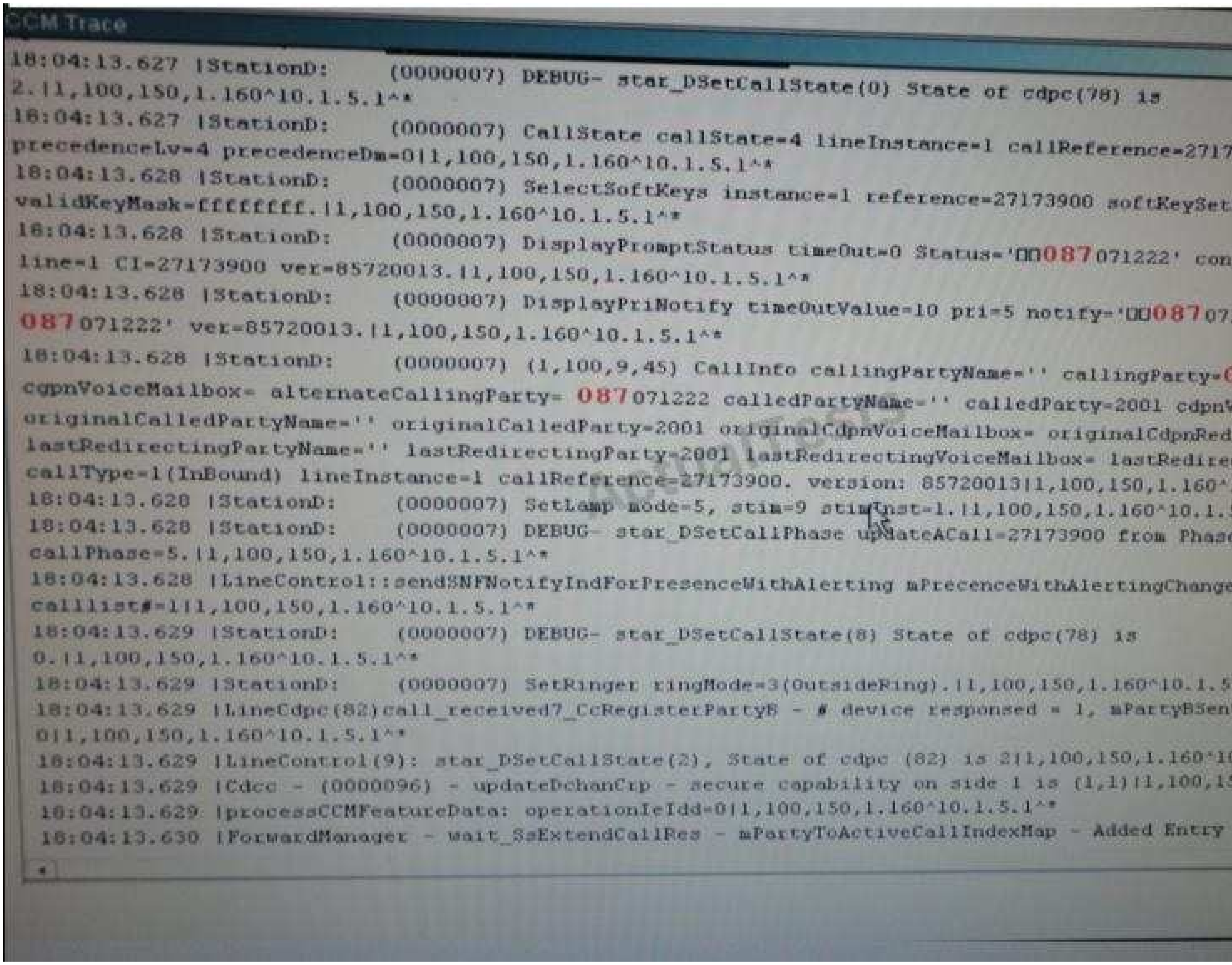


8 (exhibit):



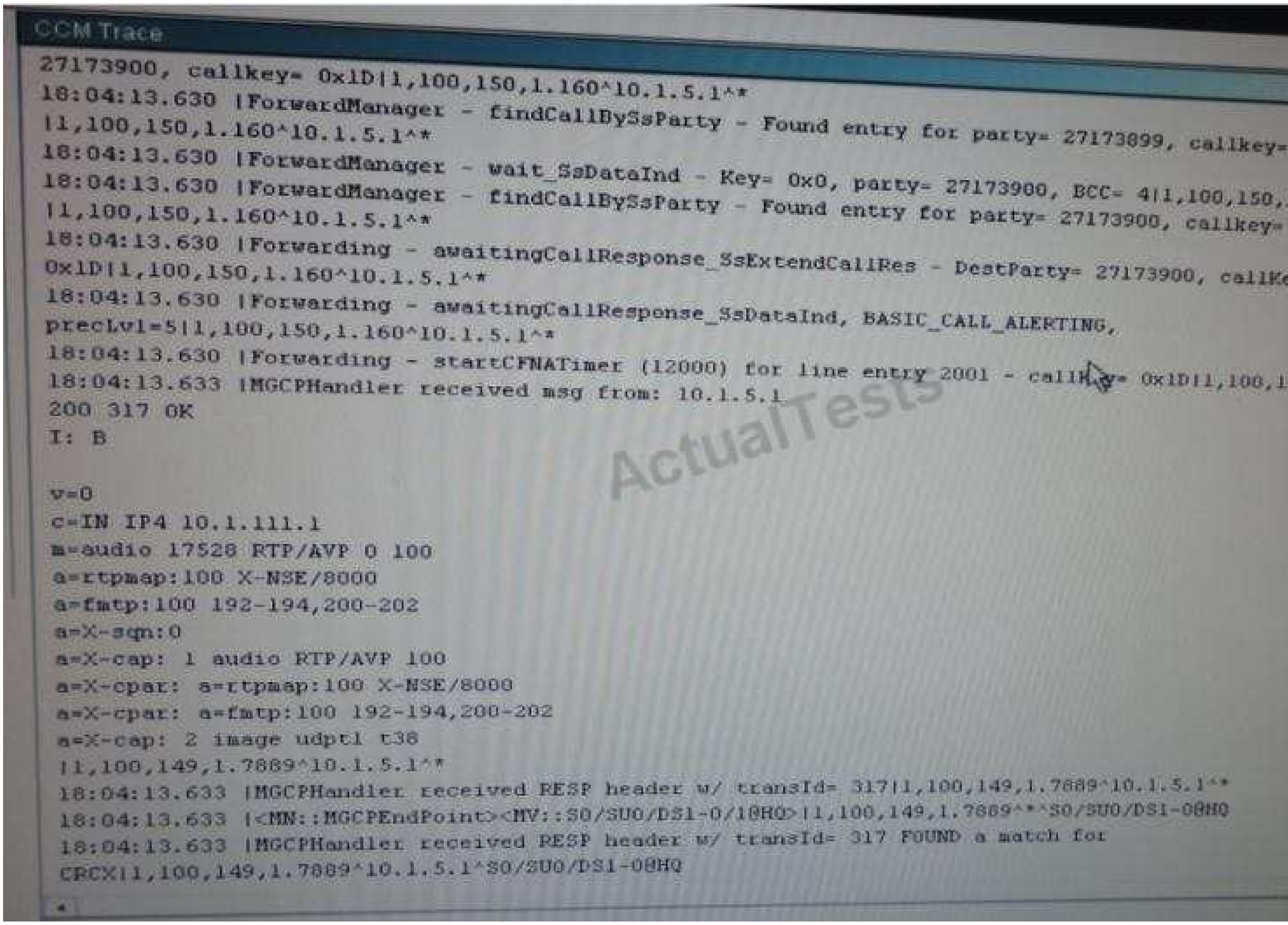
```
CCM Trace
device=SEP0021A086BF06|1,100,150,1.160^10.1.5.1^*
18:04:13.626 |StationD - adding linestruct at index 1
|*^*^*
18:04:13.626 |StationD:    (0000007) DEBUG whatToDo: line=1 calls=0 limit=4, busy=2. GCI=
0).|1,100,150,1.160^10.1.5.1^*
18:04:13.626 |StationD:    (0000007) DEBUG  whatToDo: busy trigger not hit... send to oper
appearance|1,100,150,1.160^10.1.5.1^*
18:04:13.626 |StationD:    (0000007) DEBUG- getLineRingSetting: retVal=4.|1,100,150,1.160^
18:04:13.626 |StationD:    (0000007) DEBUG- saveRinger for: ci=27173900, line=1, mode=3, p
callPhase=5.|1,100,150,1.160^10.1.5.1^*
18:04:13.626 |StationD:    (0000007) DEBUG- saveRinger: ci=27173900, line=1, mode=3, prece
modifier=0|1,100,150,1.160^10.1.5.1^*
18:04:13.626 |StationD:    (0000007) INFO  sendCallAcceptReq: Try to send StationLineCallAc
.|1,100,150,1.160^10.1.5.1^*
18:04:13.626 |StationD:    (0000007) playRinger for: ci=27173900.|1,100,150,1.160^10.1.5.1^
18:04:13.626 |StationD:    (0000007) DEBUG- getLineRingSetting: retVal=4.|1,100,150,1.160^1
18:04:13.626 |StationD:    (0000007) DEBUG- getLineRingSetting: retVal=4.|1,100,150,1.160^1
18:04:13.626 |StationD:    (0000007) DEBUG- getLineRingSetting: retVal=4.|1,100,150,1.160^1
18:04:13.626 |RegionsServer::MatchCapabilities -- kbps=64, capACount=14, capBCount=12|*^*^*
18:04:13.626 |Locations_reserveBandwidth -- cdccPID=(1.194.96) Orig=0=Dest=0 no need to rese
18:04:13.627 |StationD:    (0000007) DEBUG- star_DSetCallState(0) State of cdpc(78) is 0.|1,
18:04:13.627 |StationD:    (0000007) DEBUG- star_DSetCallState(2) State of cdpc(78) is
0.|1,100,150,1.160^10.1.5.1^*
18:04:13.627 | LocalizeCgpnAndSendOutpulsedNumber: StationCdpc on device SEP0021A086BF06 , C
,useDevicePoolCgpnCss =1 AlternateCgpn(global)=087071222 cgpn=087071222|1,100,150,1.160^10
18:04:13.627 |StationCdpc:    CcSetupReq - unicodeConnectedUnicodeDisplayName=''
asciiConnectedDisplayName=''|1,100,150,1.160^10.1.5.1^*
18:04:13.627 |StationCdpc:    CcSetupReq - unicodeCallingPartyName='' asciiCallingPartyName='
callingParty='087071222' unicodeCalledPartyName='' asciiCalledPartyName='' calledParty='200
ipv4Addr:0x0a010501(10.1.5.1) |1,100,150,1.160^10.1.5.1^*
```

**9 (exhibit):**

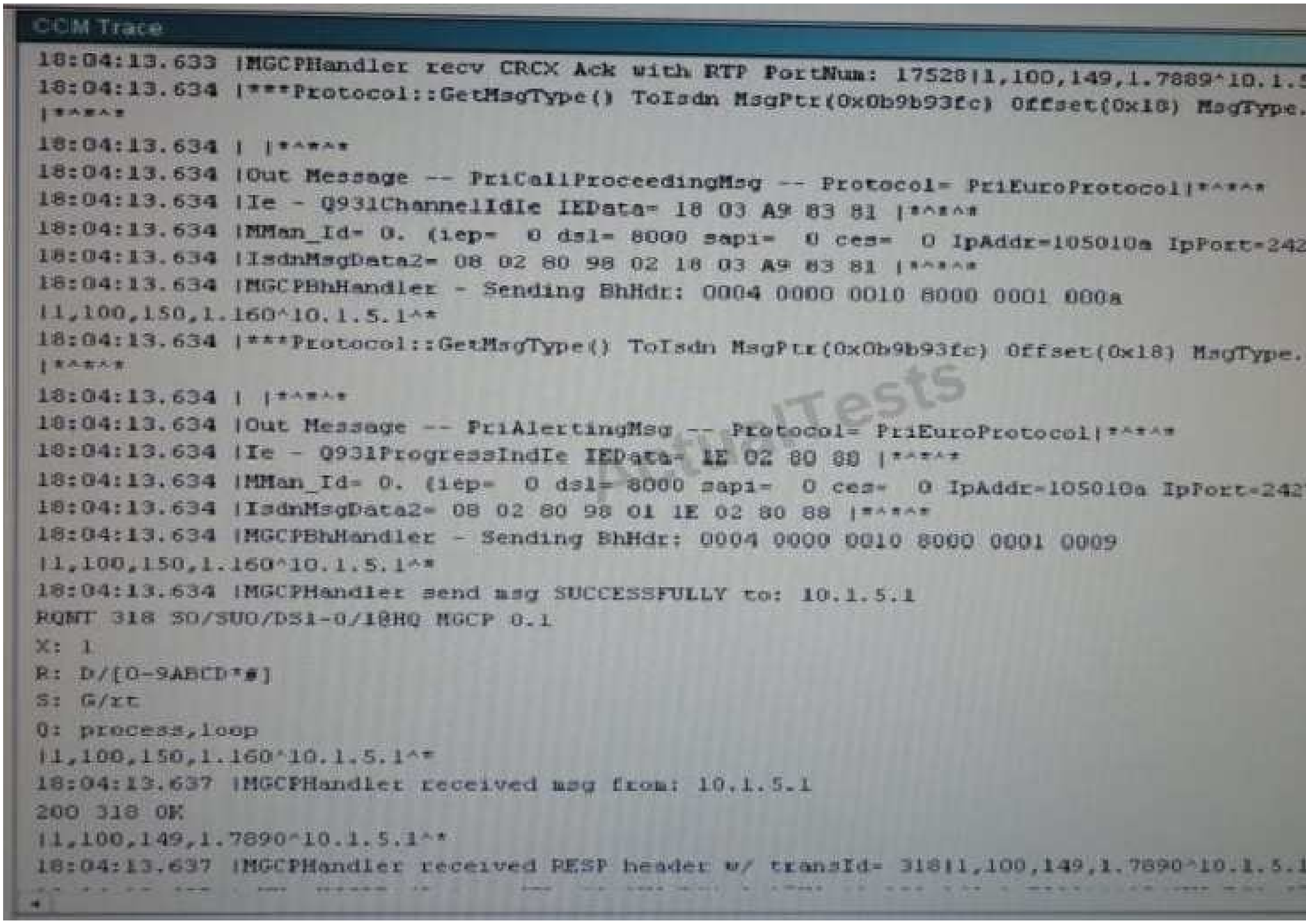

CCM Trace

```
18:04:13.627 |StationD:     (0000007) DEBUG- star_DSetCallState(0) State of cdpc(78) is
2.|1,100,150,1.160^10.1.5.1^*
18:04:13.627 |StationD:     (0000007) CallState callState=4 lineInstance=1 callReference=2717
precedenceLv=4 precedenceDm=0|1,100,150,1.160^10.1.5.1^*
18:04:13.628 |StationD:     (0000007) SelectSoftKeys instance=1 reference=27173900 softKeySet
validKeyMask=ffffffff.|1,100,150,1.160^10.1.5.1^*
18:04:13.628 |StationD:     (0000007) DisplayPromptStatus timeOut=0 Status='□□087071222' con
line=1 CI=27173900 ver=85720013.|1,100,150,1.160^10.1.5.1^*
18:04:13.628 |StationD:     (0000007) DisplayPriNotify timeOutValue=10 pri=5 notify='□□08707
087071222' ver=85720013.|1,100,150,1.160^10.1.5.1^*
18:04:13.628 |StationD:     (0000007) (1,100,9,45) CallInfo callingPartyName='' callingParty=(
cgpnVoiceMailbox= alternateCallingParty= 087071222 calledPartyName='' calledParty=2001 cdpn
originalCalledPartyName='' originalCalledParty=2001 originalCdpnVoiceMailbox= originalCdpnRed
lastRedirectingPartyName='' lastRedirectingParty=2001 lastRedirectingVoiceMailbox= lastRedire
callType=1(InBound) lineInstance=1 callReference=27173900. version: 85720013|1,100,150,1.160^
18:04:13.628 |StationD:     (0000007) SetLamp mode=5, stim=9 stimInst=1.|1,100,150,1.160^10.1.
18:04:13.628 |StationD:     (0000007) DEBUG- star_DSetCallPhase updateACall=27173900 from Phas
callPhase=5.|1,100,150,1.160^10.1.5.1^*
18:04:13.628 |LineControl::sendSNFNotifyIndForPresenceWithAlerting mPrecenceWithAlertingChange
calllist#=1|1,100,150,1.160^10.1.5.1^*
18:04:13.629 |StationD:     (0000007) DEBUG- star_DSetCallState(8) State of cdpc(78) is
0.|1,100,150,1.160^10.1.5.1^*
18:04:13.629 |StationD:     (0000007) SetRinger ringMode=3(OutsideRing).|1,100,150,1.160^10.1.5
18:04:13.629 |LineCdpc(82)call_received7_CcRegisterPartyB - # device responsed = 1, mPartyBSen
0|1,100,150,1.160^10.1.5.1^*
18:04:13.629 |LineControl(9): star_DSetCallState(2), State of cdpc (82) is 2|1,100,150,1.160^1
18:04:13.629 |Cdcc - (0000096) - updateDchanCrp - secure capability on side 1 is (1,1)|1,100,1
18:04:13.629 |processCCMFeatureData: operationIeIdd=0|1,100,150,1.160^10.1.5.1^*
18:04:13.630 |ForwardManager - wait_SsExtendCallRes - mPartyToActiveCallIndexMap - Added Entry
```

**10 (exhibit):**


```
CCM Trace
27173900, callkey= 0x1D|1,100,150,1.160^10.1.5.1^*
18:04:13.630 |ForwardManager - findCallBySsParty - Found entry for party= 27173899, callkey=
|1,100,150,1.160^10.1.5.1^*
18:04:13.630 |ForwardManager - wait_SsDataInd - Key= 0x0, party= 27173900, BCC= 4|1,100,150,
18:04:13.630 |ForwardManager - findCallBySsParty - Found entry for party= 27173900, callkey=
|1,100,150,1.160^10.1.5.1^*
18:04:13.630 |Forwarding - awaitingCallResponse_SsExtendCallRes - DestParty= 27173900, callK
0x1D|1,100,150,1.160^10.1.5.1^*
18:04:13.630 |Forwarding - awaitingCallResponse_SsDataInd, BASIC_CALL_ALERTING,
precLvl=5|1,100,150,1.160^10.1.5.1^*
18:04:13.630 |Forwarding - startCFNATimer (12000) for line entry 2001 - callKey= 0x1D|1,100,1
18:04:13.633 |MGCPHandler received msg from: 10.1.5.1
200 317 OK
I: B

v=0
c=IN IP4 10.1.111.1
m=audio 17528 RTP/AVP 0 100
a=rtpmap:100 X-NSE/8000
a=fmtp:100 192-194,200-202
a=X-sqn:0
a=X-cap: 1 audio RTP/AVP 100
a=X-cpar: a=rtpmap:100 X-NSE/8000
a=X-cpar: a=fmtp:100 192-194,200-202
a=X-cap: 2 image udptl t38
|1,100,149,1.7889^10.1.5.1^*
18:04:13.633 |MGCPHandler received RESP header w/ transId= 317|1,100,149,1.7889^10.1.5.1^*
18:04:13.633 |<MN::MGCPEndPoint><MV::S0/SU0/DS1-0/1@HQ>|1,100,149,1.7889^*^S0/SU0/DS1-0@HQ
18:04:13.633 |MGCPHandler received RESP header w/ transId= 317 FOUND a match for
CRCX|1,100,149,1.7889^10.1.5.1^S0/SU0/DS1-0@HQ
```

**11 (exhibit):**



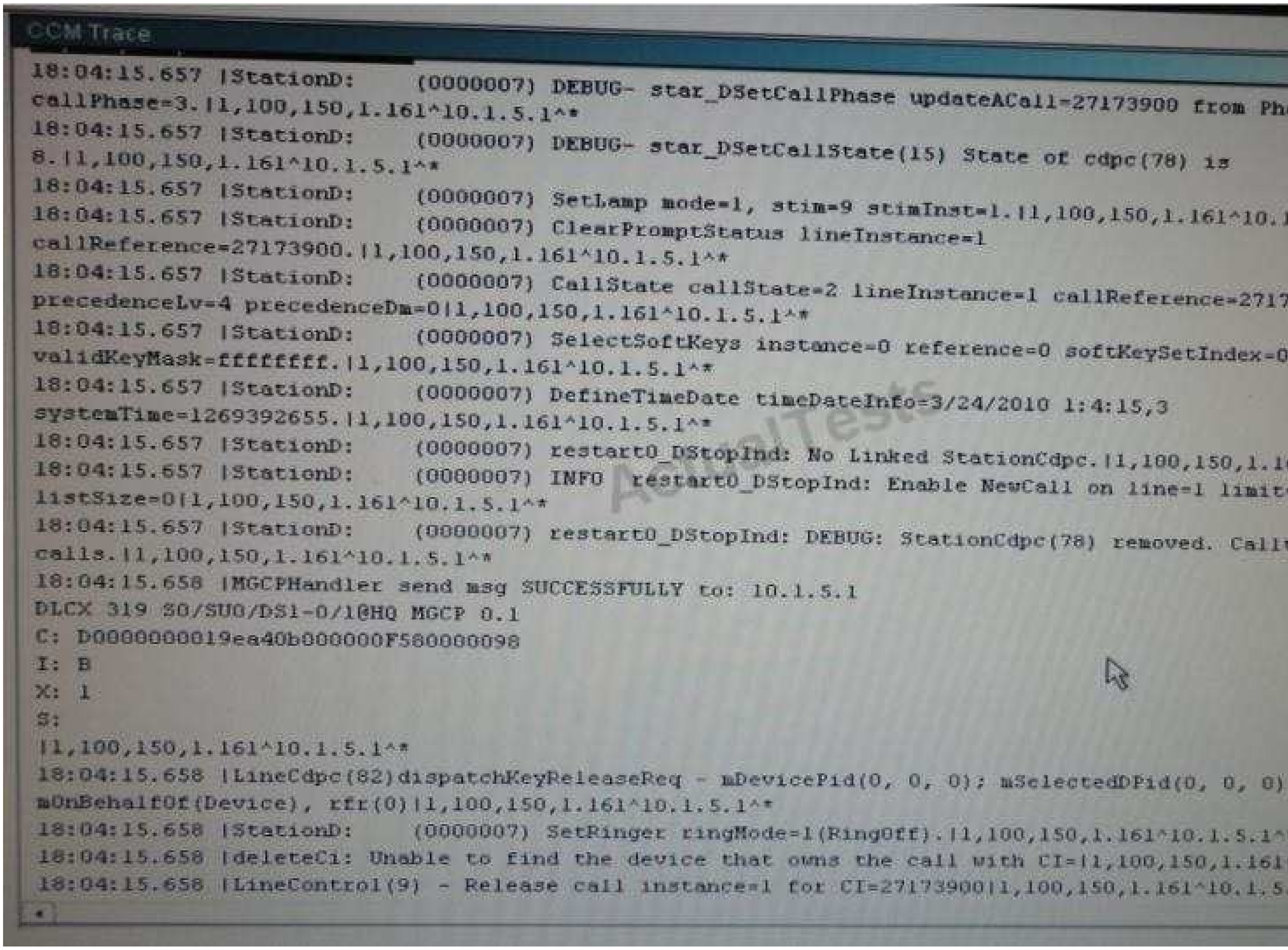
```
CCM Trace
18:04:13.633 |MGCPHandler recv CRCX Ack with RTP PortNum: 17528|1,100,149,1.7889^10.1.5
18:04:13.634 |***Protocol::GetMsgType() ToIsdn MsgPtr(0x0b9b93fc) Offset(0x18) MsgType.
|*^*^*
18:04:13.634 | |*^*^*
18:04:13.634 |Out Message -- PriCallProceedingMsg -- Protocol= PriEuroProtocol|*^*^*
18:04:13.634 |Ie - Q931ChannelIdIe IEData= 18 03 A9 83 81 |*^*^*
18:04:13.634 |MMan_Id= 0. (iep= 0 dsl= 8000 sapi= 0 ces= 0 IpAddr=105010a IpPort=242
18:04:13.634 |IsdnMsgData2= 08 02 80 98 02 18 03 A9 83 81 |*^*^*
18:04:13.634 |MGCPBhHandler - Sending BhHdr: 0004 0000 0010 8000 0001 000a
|1,100,150,1.160^10.1.5.1^*
18:04:13.634 |***Protocol::GetMsgType() ToIsdn MsgPtr(0x0b9b93fc) Offset(0x18) MsgType.
|*^*^*
18:04:13.634 | |*^*^*
18:04:13.634 |Out Message -- PriAlertingMsg -- Protocol= PriEuroProtocol|*^*^*
18:04:13.634 |Ie - Q931ProgressIndIe IEData= 1E 02 80 88 |*^*^*
18:04:13.634 |MMan_Id= 0. (iep= 0 dsl= 8000 sapi= 0 ces= 0 IpAddr=105010a IpPort=242
18:04:13.634 |IsdnMsgData2= 08 02 80 98 01 1E 02 80 88 |*^*^*
18:04:13.634 |MGCPBhHandler - Sending BhHdr: 0004 0000 0010 8000 0001 0009
|1,100,150,1.160^10.1.5.1^*
18:04:13.634 |MGCPHandler send msg SUCCESSFULLY to: 10.1.5.1
RQNT 318 S0/SU0/DS1-0/1@HQ MGCP 0.1
X: 1
R: D/[0-9ABCD*#]
S: G/rt
Q: process,loop
|1,100,150,1.160^10.1.5.1^*
18:04:13.637 |MGCPHandler received msg from: 10.1.5.1
200 318 OK
|1,100,149,1.7890^10.1.5.1^*
18:04:13.637 |MGCPHandler received RESP header w/ transId= 318|1,100,149,1.7890^10.1.5.1
```

**12 (exhibit):**

CCM Trace

```
18:04:13.637 |<MN::MGCPEndPoint><MV::S0/SU0/DS1-0/1@HQ>|1,100,149,1.7890^**^S0/SU0/DS1-0@HQ
18:04:13.637 |MGCPHandler received RESP header w/ transId= 318 FOUND a match for
RQNT|1,100,149,1.7890^10.1.5.1^S0/SU0/DS1-0@HQ
18:04:13.637 |MGCPHandler recv RQNT Ack from 10.1.5.1|1,100,149,1.7890^10.1.5.1^S0/SU0/DS1-0
18:04:15.655 |MGCPBhHandler 10.1.5.1 - TCP msg available from Device|1,100,150,1.161^10.1.5.
18:04:15.655 |MGCPBhHandler - Receiving BhHdr: 0004 0000 0011 8000 0001 0009
|1,100,150,1.161^10.1.5.1^*
18:04:15.655 | |*^*^*
18:04:15.655 |In  Message -- PriDisconnectMsg -- Protocol= PriEuroProtocol|*^*^*
18:04:15.655 |Ie - Q931CauseIe -- IEData= 08 02 82 90 |*^*^*
18:04:15.655 |MMan_Id= 0. (iep=  0 dsl= 8000 sapi=  0 ces= 0 IpAddr=105010a IpPort=2427)|*^*
18:04:15.655 |IsdnMsgData1= 08 02 00 98 45 08 02 82 90 |*^*^*
18:04:15.655 |SPROC analyzeMsgtransCause MessageTransCause.ms = 0, MessageTransCause.ieid =
9, MCStatus = 0|*^*^*
18:04:15.655 |MGCPpn9cuser - sendCcDisconnInd, Q931Cause.cv:16, CcDisconnInd.c.cv:16|1,100,1
18:04:15.655 |Cdcc::isStaticTransactionApplicable |1,100,150,1.161^10.1.5.1^*
18:04:15.655 |processCCMFeatureData: operationIeIdd=0|1,100,150,1.161^10.1.5.1^*
18:04:15.656 |ForwardManager - wait_SsDataInd - Key= 0x0, party= 27173899, BCC= 6|1,100,150,
18:04:15.656 |ForwardManager - findCallBySsParty - Found entry for party= 27173899, callkey=
|1,100,150,1.161^10.1.5.1^*
18:04:15.656 |ConnectionManager - wait_AuDisconnectRequest ERROR:NO ENTRY FOUND IN
TABLE,CI(27173899,27173900),dcType=1,IFCreated(0,0),PID(0-0,0
-0),IFHandling(0,0),MCNode(0,0)|1,100,150,1.161^10.1.5.1^*
18:04:15.656 |MatrixControl:updatePartyMediaCoordinatorNodeId: party1 videoCapable=0, party
videocapable=0|1,100,150,1.161^10.1.5.1^*
18:04:15.656 |Cdcc - (0000096) - resetMediaSecurity|1,100,150,1.161^10.1.5.1^*
18:04:15.656 |LineCdpc(82): -dispatchToAllDevices-, sigName=CcDisconnReq,
device=SEP0021A086BF06|1,100,150,1.161^10.1.5.1^*
18:04:15.657 |LineControl TEST DEBUGS: Number of entries in CallTable is =  1
|1,100,150,1.161^10.1.5.1^*
```

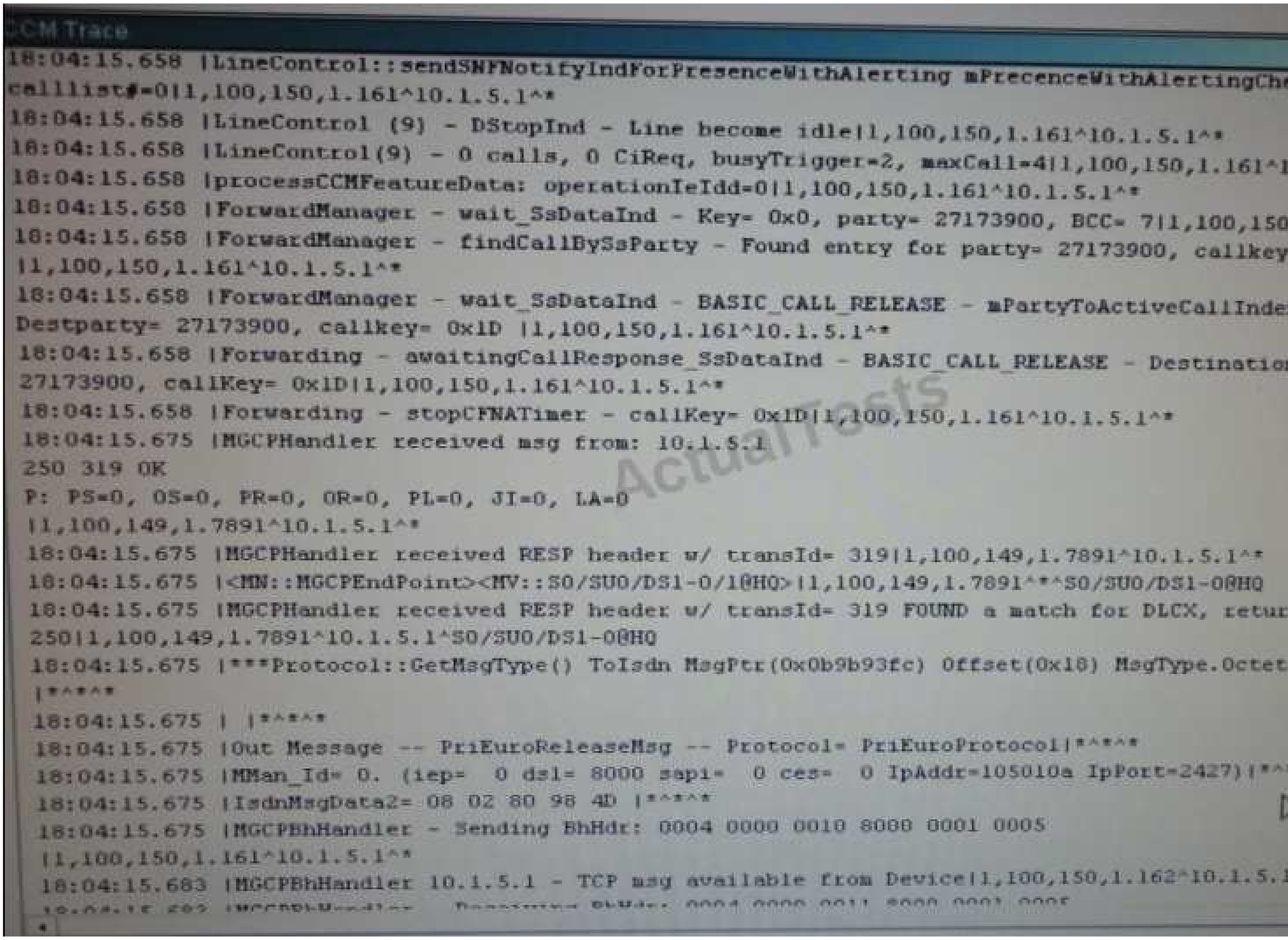
**13 (exhibit):**



CCM Trace

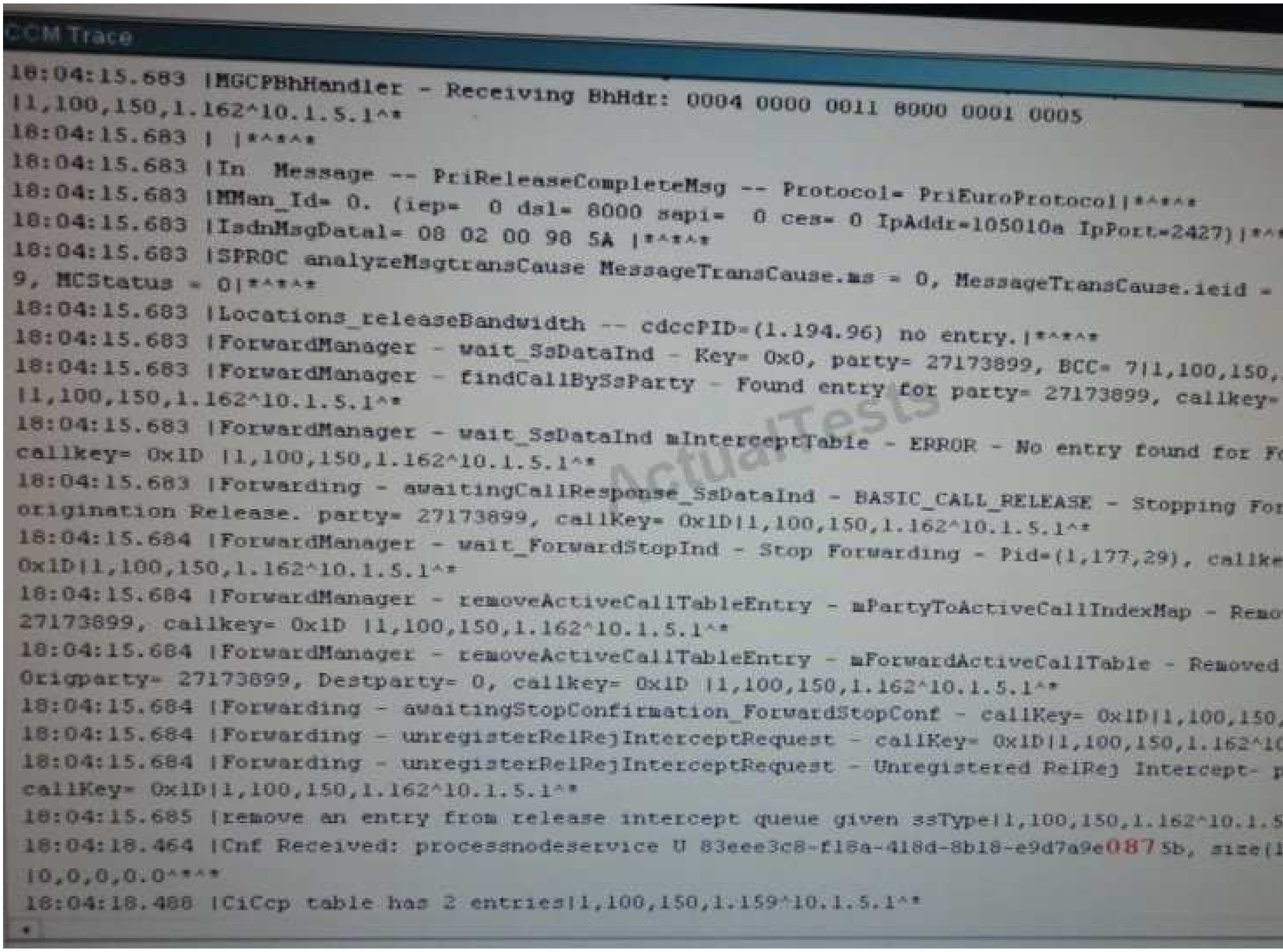
```
18:04:15.657 |StationD:     (0000007) DEBUG- star_DSetCallPhase updateACall=27173900 from Ph
callPhase=3.|1,100,150,1.161^10.1.5.1^*
18:04:15.657 |StationD:     (0000007) DEBUG- star_DSetCallState(15) State of cdpc(78) is
8.|1,100,150,1.161^10.1.5.1^*
18:04:15.657 |StationD:     (0000007) SetLamp mode=1, stim=9 stimInst=1.|1,100,150,1.161^10.1
18:04:15.657 |StationD:     (0000007) ClearPromptStatus lineInstance=1
callReference=27173900.|1,100,150,1.161^10.1.5.1^*
18:04:15.657 |StationD:     (0000007) CallState callState=2 lineInstance=1 callReference=2717
precedenceLv=4 precedenceDm=0|1,100,150,1.161^10.1.5.1^*
18:04:15.657 |StationD:     (0000007) SelectSoftKeys instance=0 reference=0 softKeySetIndex=0
validKeyMask=ffffffff.|1,100,150,1.161^10.1.5.1^*
18:04:15.657 |StationD:     (0000007) DefineTimeDate timeDateInfo=3/24/2010 1:4:15,3
systemTime=1269392655.|1,100,150,1.161^10.1.5.1^*
18:04:15.657 |StationD:     (0000007) restart0_DStopInd: No Linked StationCdpc.|1,100,150,1.1
18:04:15.657 |StationD:     (0000007) INFO  restart0_DStopInd: Enable NewCall on line=1 limit
listSize=0|1,100,150,1.161^10.1.5.1^*
18:04:15.657 |StationD:     (0000007) restart0_DStopInd: DEBUG: StationCdpc(78) removed. Call
calls.|1,100,150,1.161^10.1.5.1^*
18:04:15.658 |MGCPHandler send msg SUCCESSFULLY to: 10.1.5.1
DLCX 319 S0/SU0/DS1-0/1@HQ MGCP 0.1
C: D0000000019ea40b000000F580000098
I: B
X: 1
S:
|1,100,150,1.161^10.1.5.1^*
18:04:15.658 |LineCdpc(82)dispatchKeyReleaseReq - mDevicePid(0, 0, 0); mSelectedDPid(0, 0, 0)
mOnBehalfOf(Device), rfr(0)|1,100,150,1.161^10.1.5.1^*
18:04:15.658 |StationD:     (0000007) SetRinger ringMode=1(RingOff).|1,100,150,1.161^10.1.5.1^
18:04:15.658 |deleteCi: Unable to find the device that owns the call with CI=|1,100,150,1.161
18:04:15.658 |LineControl(9) - Release call instance=1 for CI=27173900|1,100,150,1.161^10.1.5
```

**14 (exhibit):**



```
CCM Trace
18:04:15.658 |LineControl::sendSNFNotifyIndForPresenceWithAlerting mPrecenceWithAlertingChe
calllist#=0|1,100,150,1.161^10.1.5.1^*
18:04:15.658 |LineControl (9) - DStopInd - Line become idle|1,100,150,1.161^10.1.5.1^*
18:04:15.658 |LineControl(9) - 0 calls, 0 CiReq, busyTrigger=2, maxCall=4|1,100,150,1.161^1
18:04:15.658 |processCCMFeatureData: operationIeIdd=0|1,100,150,1.161^10.1.5.1^*
18:04:15.658 |ForwardManager - wait_SsDataInd - Key= 0x0, party= 27173900, BCC= 7|1,100,150
18:04:15.658 |ForwardManager - findCallBySsParty - Found entry for party= 27173900, callkey
|1,100,150,1.161^10.1.5.1^*
18:04:15.658 |ForwardManager - wait_SsDataInd - BASIC_CALL_RELEASE - mPartyToActiveCallInde
Destparty= 27173900, callkey= 0x1D |1,100,150,1.161^10.1.5.1^*
18:04:15.658 |Forwarding - awaitingCallResponse_SsDataInd - BASIC_CALL_RELEASE - Destinatio
27173900, callKey= 0x1D|1,100,150,1.161^10.1.5.1^*
18:04:15.658 |Forwarding - stopCFNATimer - callKey= 0x1D|1,100,150,1.161^10.1.5.1^*
18:04:15.675 |MGCPHandler received msg from: 10.1.5.1
250 319 OK
P: PS=0, OS=0, PR=0, OR=0, PL=0, JI=0, LA=0
|1,100,149,1.7891^10.1.5.1^*
18:04:15.675 |MGCPHandler received RESP header w/ transId= 319|1,100,149,1.7891^10.1.5.1^*
18:04:15.675 |<MN::MGCPEndPoint><MV::S0/SU0/DS1-0/1@HQ>|1,100,149,1.7891^*^S0/SU0/DS1-0@HQ
18:04:15.675 |MGCPHandler received RESP header w/ transId= 319 FOUND a match for DLCX, retur
250|1,100,149,1.7891^10.1.5.1^S0/SU0/DS1-0@HQ
18:04:15.675 |***Protocol::GetMsgType() ToIsdn MsgPtr(0x0b9b93fc) Offset(0x18) MsgType.Octet
|*^*^*
18:04:15.675 | |*^*^*
18:04:15.675 |Out Message -- PriEuroReleaseMsg -- Protocol= PriEuroProtocol|*^*^*
18:04:15.675 |MMan_Id= 0. (iep= 0 dsl= 8000 sapi= 0 ces= 0 IpAddr=105010a IpPort=2427)|*^
18:04:15.675 |IsdnMsgData2= 08 02 80 98 4D |*^*^*
18:04:15.675 |MGCPBhHandler - Sending BhHdr: 0004 0000 0010 8000 0001 0005
|1,100,150,1.161^10.1.5.1^*
18:04:15.683 |MGCPBhHandler 10.1.5.1 - TCP msg available from Device|1,100,150,1.162^10.1.5.1
```

**15 (exhibit):**



```
CCM Trace
18:04:15.683 |MGCPBhHandler - Receiving BhHdr: 0004 0000 0011 8000 0001 0005
|1,100,150,1.162^10.1.5.1^*
18:04:15.683 | |*^*^*
18:04:15.683 |In  Message -- PriReleaseCompleteMsg -- Protocol= PriEuroProtocol|*^*^*
18:04:15.683 |MMan_Id= 0. (iep=  0 dsl= 8000 sapi=  0 ces= 0 IpAddr=105010a IpPort=2427)|*^*
18:04:15.683 |IsdnMsgData1= 08 02 00 98 5A |*^*^*
18:04:15.683 |SPROC analyzeMsgtransCause MessageTransCause.ms = 0, MessageTransCause.ieid =
9, MCStatus = 0|*^*^*
18:04:15.683 |Locations_releaseBandwidth -- cdccPID=(1.194.96) no entry.|*^*^*
18:04:15.683 |ForwardManager - wait_SsDataInd - Key= 0x0, party= 27173899, BCC= 7|1,100,150,
18:04:15.683 |ForwardManager - findCallBySsParty - Found entry for party= 27173899, callkey=
|1,100,150,1.162^10.1.5.1^*
18:04:15.683 |ForwardManager - wait_SsDataInd mInterceptTable - ERROR - No entry found for Fo
callkey= 0x1D |1,100,150,1.162^10.1.5.1^*
18:04:15.683 |Forwarding - awaitingCallResponse_SsDataInd - BASIC_CALL_RELEASE - Stopping For
origination Release. party= 27173899, callKey= 0x1D|1,100,150,1.162^10.1.5.1^*
18:04:15.684 |ForwardManager - wait_ForwardStopInd - Stop Forwarding - Pid=(1,177,29), callke
0x1D|1,100,150,1.162^10.1.5.1^*
18:04:15.684 |ForwardManager - removeActiveCallTableEntry - mPartyToActiveCallIndexMap - Remo
27173899, callkey= 0x1D |1,100,150,1.162^10.1.5.1^*
18:04:15.684 |ForwardManager - removeActiveCallTableEntry - mForwardActiveCallTable - Removed
Origparty= 27173899, Destparty= 0, callkey= 0x1D |1,100,150,1.162^10.1.5.1^*
18:04:15.684 |Forwarding - awaitingStopConfirmation_ForwardStopConf - callKey= 0x1D|1,100,150,
18:04:15.684 |Forwarding - unregisterRelRejInterceptRequest - callKey= 0x1D|1,100,150,1.162^1
18:04:15.684 |Forwarding - unregisterRelRejInterceptRequest - Unregistered RelRej Intercept- p
callKey= 0x1D|1,100,150,1.162^10.1.5.1^*
18:04:15.685 |remove an entry from release intercept queue given ssType|1,100,150,1.162^10.1.5
18:04:18.464 |Cnf Received: processnodeservice U 83eee3c8-f18a-418d-8b18-e9d7a9e0875b, size(1
|0,0,0,0.0^*^*
18:04:18.488 |CiCcp table has 2 entries|1,100,150,1.159^10.1.5.1^*
```

A. Exhibit A
B. Exhibit B
C. Exhibit C
D. Exhibit D

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

**Explanation/Reference:**
**Explanation:** Actual incoming number is 14-087071 222 but next to this information in trace we can see two digits are stripped which is international code hence D is valid answer.

**QUESTION 5**
When a database replication issue is suspected, which three tools can be used to check the database replication status? (Choose three.)

A. Cisco Unified Communications Manager RTMT tool
B. Cisco Unified Communications Manager Serviceability interface
C. Cisco Unified Reporting

D. Cisco Unified Communications Manager CLI interface
E. Cisco IP Phone Device Stats from the Settings button
F. Cisco Unified OS Administration interface

**Correct Answer:** ACD
**Section: (none)**
**Explanation**

**Explanation/Reference:**
**Reference:** http://www.cisco.com/en/US/products/sw/voicesw/ps556/
products_tech_note09186a00809643e8. shtml

**QUESTION 6**
Which of these reasons can cause intrasite calls within a Cisco Unified Communications Manager cluster to fail?



A. The route partition that is configured in the CCD requesting service is not listed in the calling phone CSS
B. The trunk CSS does not include the partition for the called directory number
C. The MGCP gateway is not registered
D. The calling phone does not have the correct CSS configured
E. The calling phone does not have the correct partition configured.

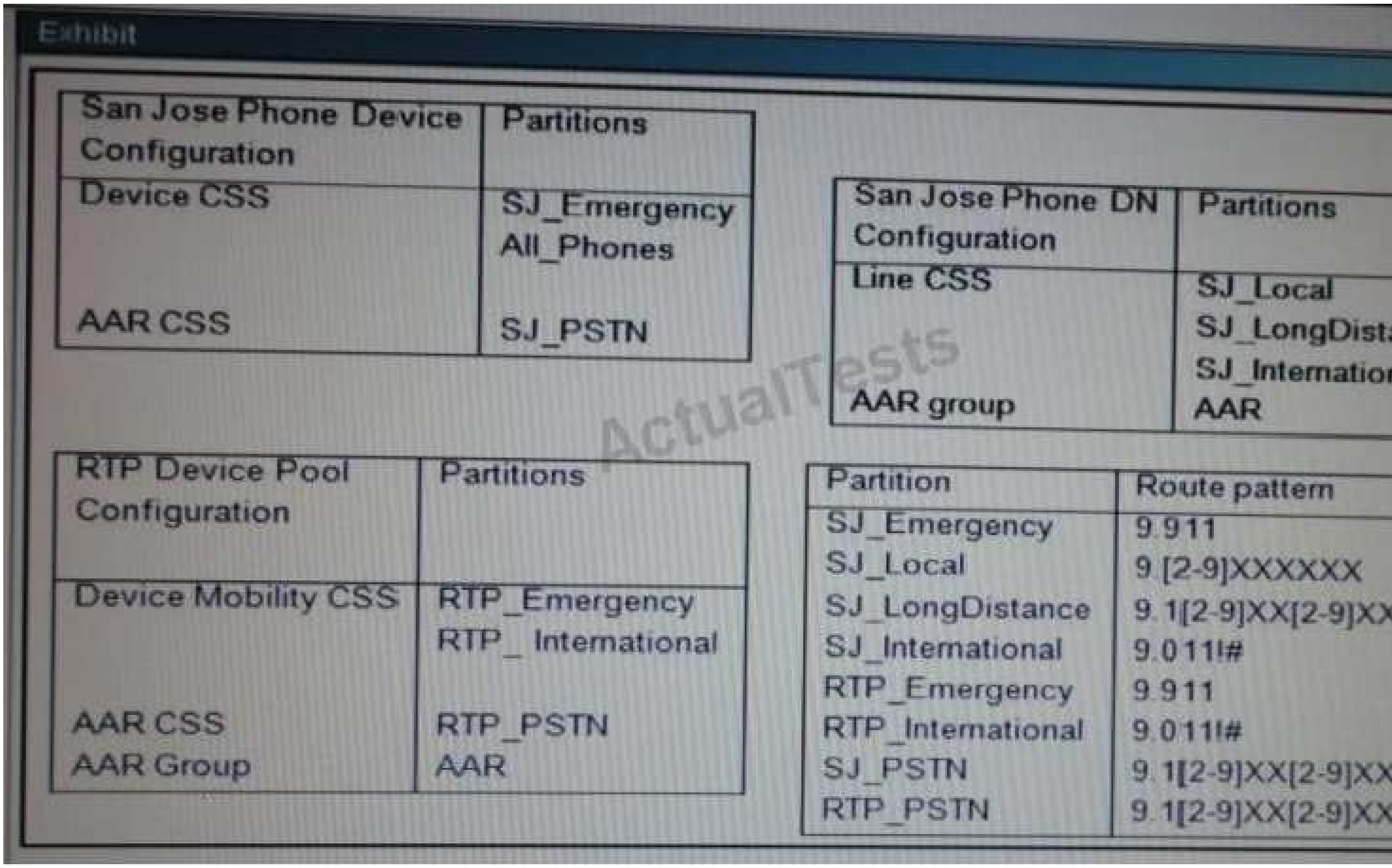
**Correct Answer:** D
**Section: (none)**
**Explanation**

**Explanation/Reference:**
**Explanation:**To make a successful call within CUCM cluster following condition should satisfy.
**Link-** http://www.cisco.com/en/US/products/sw/voicesw/ps556/products_tech_note09186a0080094b53. shtml

**QUESTION 7**
**Refer to the exhibit.**

When a Cisco IP Communicator phone roams from San Jose (SJ) to RTP, the Cisco IP Communicator physical location and the device mobility group change from SJ to RTP All route patterns are assigned a route list that points to the local route group All device pools are configured to use the local route group Which statement is true when the roaming phone places an AAR call?

**Exhibit:**



Exhibit

| San Jose Phone Device Configuration | Partitions |
|---|---|
| Device CSS | SJ_Emergency<br>All_Phones |
| AAR CSS | SJ_PSTN |

| San Jose Phone DN Configuration | Partitions |
|---|---|
| Line CSS | SJ_Local<br>SJ_LongDist<br>SJ_Internatio |
| AAR group | AAR |

| RTP Device Pool Configuration | Partitions |
|---|---|
| Device Mobility CSS | RTP_Emergency<br>RTP_ International |
| AAR CSS | RTP_PSTN |
| AAR Group | AAR |

| Partition | Route pattern |
|---|---|
| SJ_Emergency | 9.911 |
| SJ_Local | 9.[2-9]XXXXXX |
| SJ_LongDistance | 9.1[2-9]XX[2-9]XX |
| SJ_International | 9.011!# |
| RTP_Emergency | 9.911 |
| RTP_International | 9.011!# |
| SJ_PSTN | 9.1[2-9]XX[2-9]XX |
| RTP_PSTN | 9.1[2-9]XX[2-9]XX |



A. Since globalized call routing is not configured, then the SJ gateway will be used in this case
B. The phone will use the AAR CSS that contains the SJ_PSTN partition. The call will egress at the SJ gateway
C. The phone will use the AAR CSS that contains the RTP_PSTN partition. The call will egress at the SJ gateway
D. The phone will use the AAR CSS that contains the SJ_PSTN partition. The call will egress at the RTP gateway.
E. The phone will use the AAR CSS that contains the RTP_PSTN partition The call will egress at the RTP gateway

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

**Explanation/Reference:**

## Explanation:

Cisco Unified Communications Manager Version 7.0 introduced the Local Route Group feature.When using local route groups, gateway selection is totally independent of the matched route pattern and referenced route list and routegroup. The use of the Local Route Group feature makes no changes regarding roaming-sensitive settings. The application of these settings always makes sense when roaming between sites.

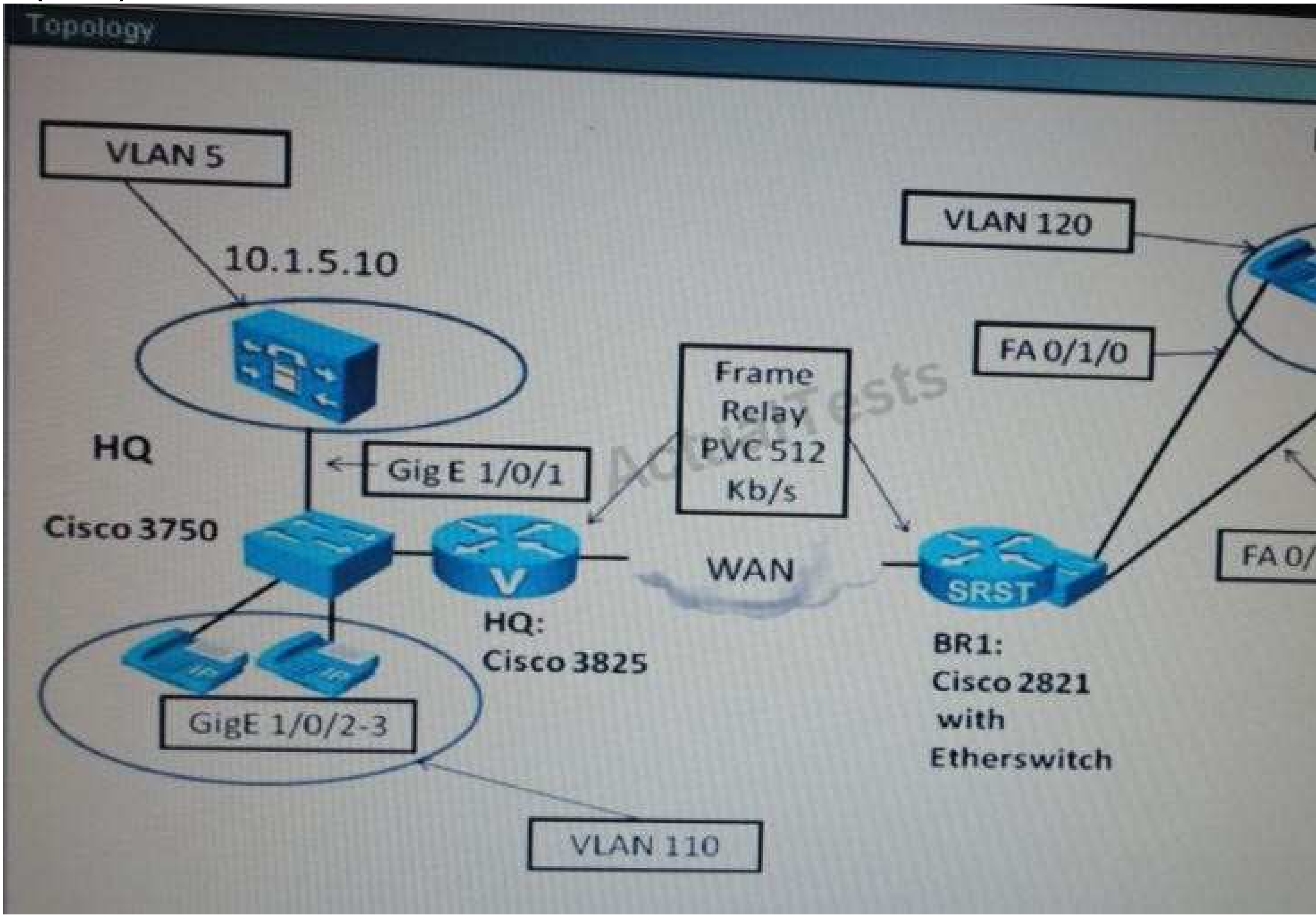
The settings have no influence to the gateway selection and the dial rules that a user must follow. However, the dial planrelated part of Device Mobility changes substantially withthe new dial plan concept, This concept allows a roaming user to follow the home dial rules for external calls but use the local gateway of the roaming site In this case, When the device mobility group is not the same for San Jose and RTP, the Device Mobility related settings are not applied. The phone device keeps its San Jose-specific configuration Despite the San Jose-specific configuration on the phone, the PSTN calls that originate from the roaming phone are routed via the local PSTN gateway (RTP GW) and are based on the route list and device pool local route group settings.

The San Jose-specific dial plan is used. Also, AAR
remains configured with the San Jose-specific configuration, but if the San Jose dial plan and San Jose AAR CSS permit and if the AAR group contains the prefix that can be applied in RTP, then AAR can work

**QUESTION 8**
## Refer to the exhibits.

Low latency queuing has been implemented on the HO and BR1 routers to allow five G.729 calls. Callers are still experiencing poor audio, in particular choppy and delayed audio during traffic congestion. This problem occurs even with just one active call. Which two actions will solve the issue?

**16 (exhibit):**




**17 (exhibit):**

HQ Router Config

```
version 12.4
service timestamps debug datetime msec
service timestamps log datetime msec
no service password-encryption
!
hostname HQ
!
boot-start-marker
boot system flash:c3825-ipvoice_ivs-mz.124-22.T.bin
boot-end-marker
!
card type t1 0 0
logging message-counter syslog
enable password cisco123
!
no aaa new-model
clock timezone PST -8
clock summer-time pst recurring
network-clock-participate wic 0
!
ip source-route
ip cef
!
no ipv6 cef
multilink bundle-name authenticated
no ip domain lookup
!
!
isdn switch-type primary-ni
!
voice-card 0
```



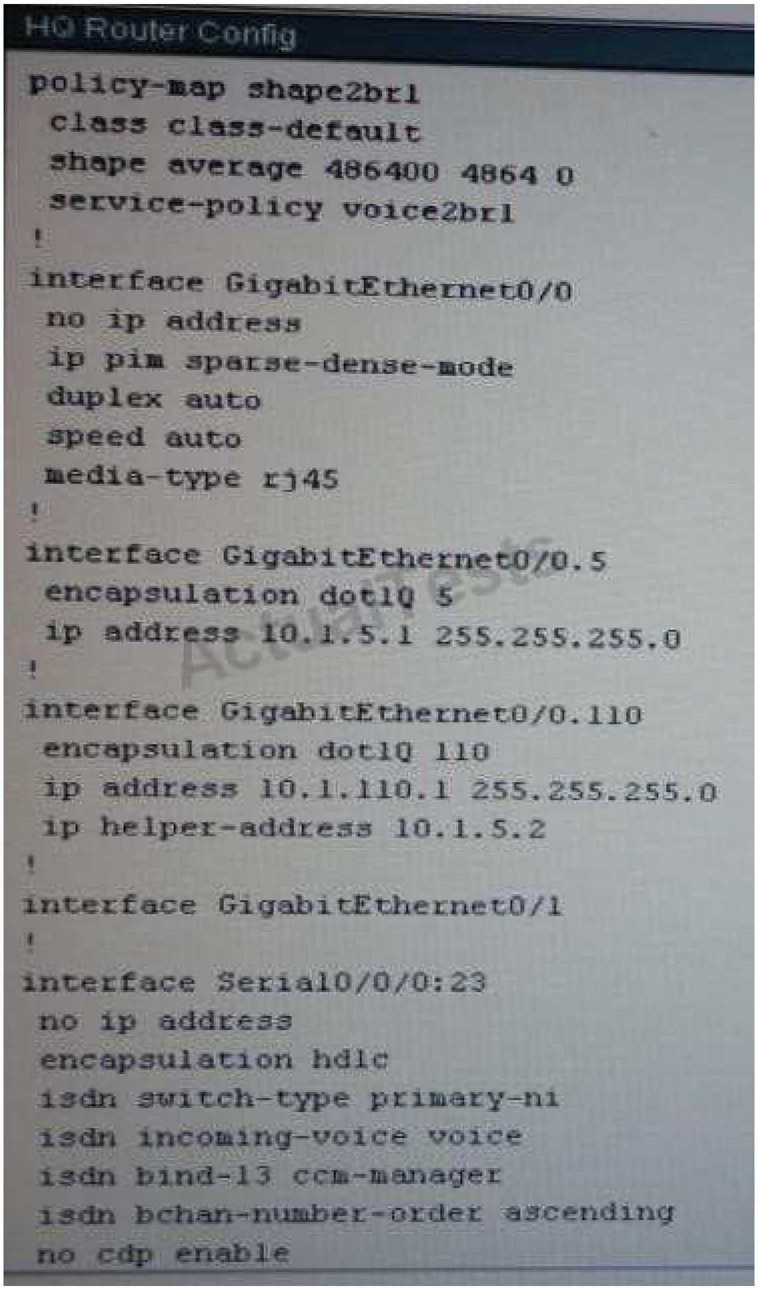
**18 (exhibit):**

HQ Router Config

```
voice-card 0
 dsp services dspfarm
!
!
!
!
archive
 log config
 hidekeys
!
!
controller T1 0/0/0
 cablelength short 110
 pri-group timeslots 1-12,24 service mgcp
!
!
class-map match-all ctraffic
 match ip dscp cs3
class-map match-all vtraffic
 match ip dscp ef
!
policy-map voice2br1
 class vtraffic
 priority 64
 class ctraffic
 bandwidth 8
 class class-default
 fair-queue
!
policy-map shape2br1
 class class-default
```



**19 (exhibit):**



HQ Router Config

```
policy-map shape2br1
 class class-default
 shape average 486400 4864 0
 service-policy voice2br1
!
interface GigabitEthernet0/0
 no ip address
 ip pim sparse-dense-mode
 duplex auto
 speed auto
 media-type rj45
!
interface GigabitEthernet0/0.5
 encapsulation dot1Q 5
 ip address 10.1.5.1 255.255.255.0
!
interface GigabitEthernet0/0.110
 encapsulation dot1Q 110
 ip address 10.1.110.1 255.255.255.0
 ip helper-address 10.1.5.2
!
interface GigabitEthernet0/1
!
interface Serial0/0/0:23
 no ip address
 encapsulation hdlc
 isdn switch-type primary-ni
 isdn incoming-voice voice
 isdn bind-l3 ccm-manager
 isdn bchan-number-order ascending
 no cdp enable
```

**20 (exhibit):**

```
!
map-class frame-relay frts2brl
 frame-relay fair-queue
 service-policy output shape2brl
!
control-plane
!
!
voice-port 0/0/0:23
!
ccm-manager redundant-host 10.1.5.2
ccm-manager mgcp
ccm-manager fax protocol cisco
ccm-manager music-on-hold
ccm-manager config server 10.1.5.3
!
mgcp
mgcp call-agent 10.1.5.3 2427 service-type mgcp version 0.1
mgcp rtp unreachable timeout 1000 action notify
mgcp modem passthrough voip mode nse
mgcp package-capability rtp-package
mgcp package-capability sst-package
mgcp package-capability pre-package
no mgcp package-capability res-package
no mgcp timer receive-rtcp
mgcp sdp simple
mgcp fax t38 ecm
mgcp rtp payload-type g726r16 static
!
mgcp profile default
!
```



21 (exhibit):

```
mgcp fax t38 ecm
mgcp rtp payload-type g726r16 static
!
mgcp profile default
!
gateway
 timer receive-rtp 1200
!
!
gatekeeper
 shutdown
!
!
line con 0
 exec-timeout 0 0
 logging synchronous
line aux 0
line vty 0 4
 exec-timeout 0 0
 password cisco123
 login
!
scheduler allocate 20000 1000
end
```



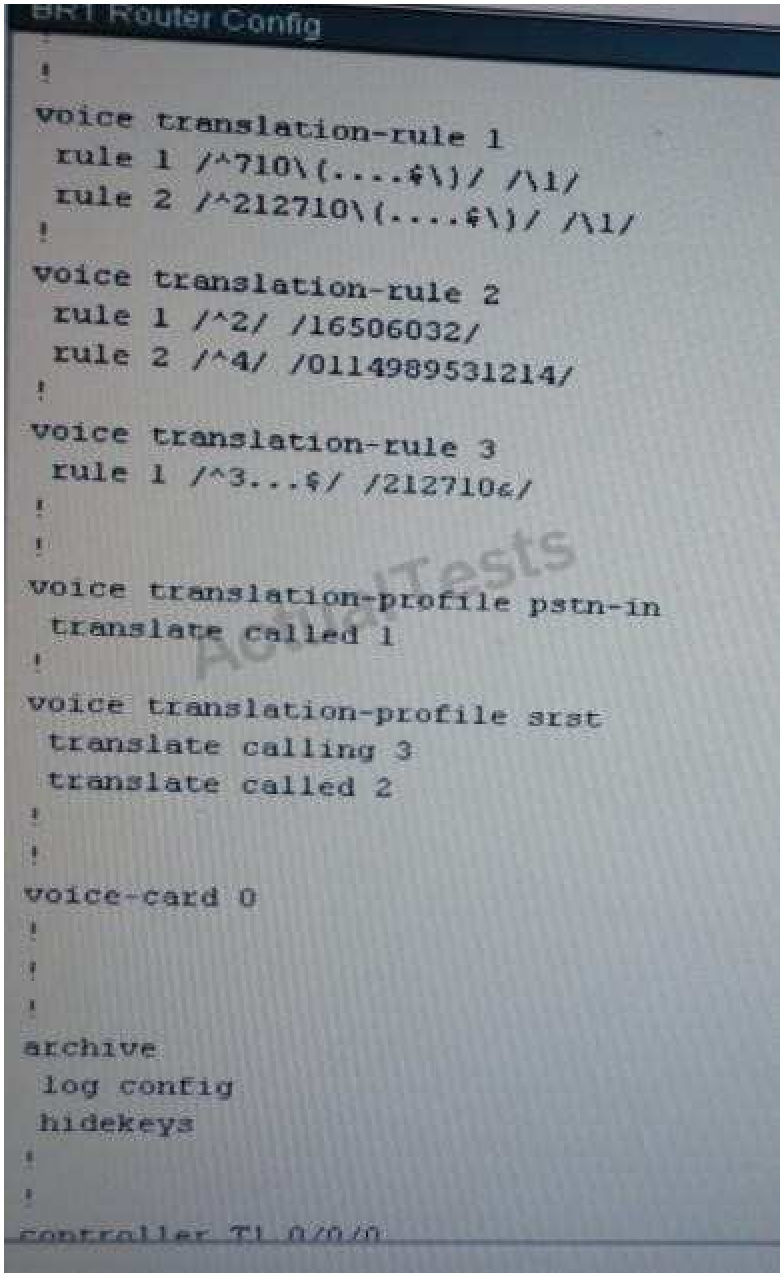
**22 (exhibit):**

BR1 Router Config

```
version 12.4
service timestamps debug datetime msec
service timestamps log datetime msec
no service password-encryption
!
hostname BR1
!
boot-start-marker
boot-end-marker
!
card type t1 0 0
logging message-counter syslog
enable password cisco
!
no aaa new-model
network-clock-participate wic 0
!
ip source-route
!
!
ip cef
no ip domain lookup
no ipv6 cef
multilink bundle-name authenticated
!
isdn switch-type primary-ni
!
!
voice class h323 1
 h225 timeout tcp establish 3
!
```



**23 (exhibit):**



BRT Router Config

```
!
voice translation-rule 1
 rule 1 /^710\(....$\)/ /\1/
 rule 2 /^212710\(....$\)/ /\1/
!
voice translation-rule 2
 rule 1 /^2/ /16506032/
 rule 2 /^4/ /0114989531214/
!
voice translation-rule 3
 rule 1 /^3...$/ /212710&/
!
!
voice translation-profile pstn-in
 translate called 1
!
voice translation-profile srst
 translate calling 3
 translate called 2
!
!
voice-card 0
!
!
!
archive
 log config
 hidekeys
!
!
controller T1 0/0/0
```

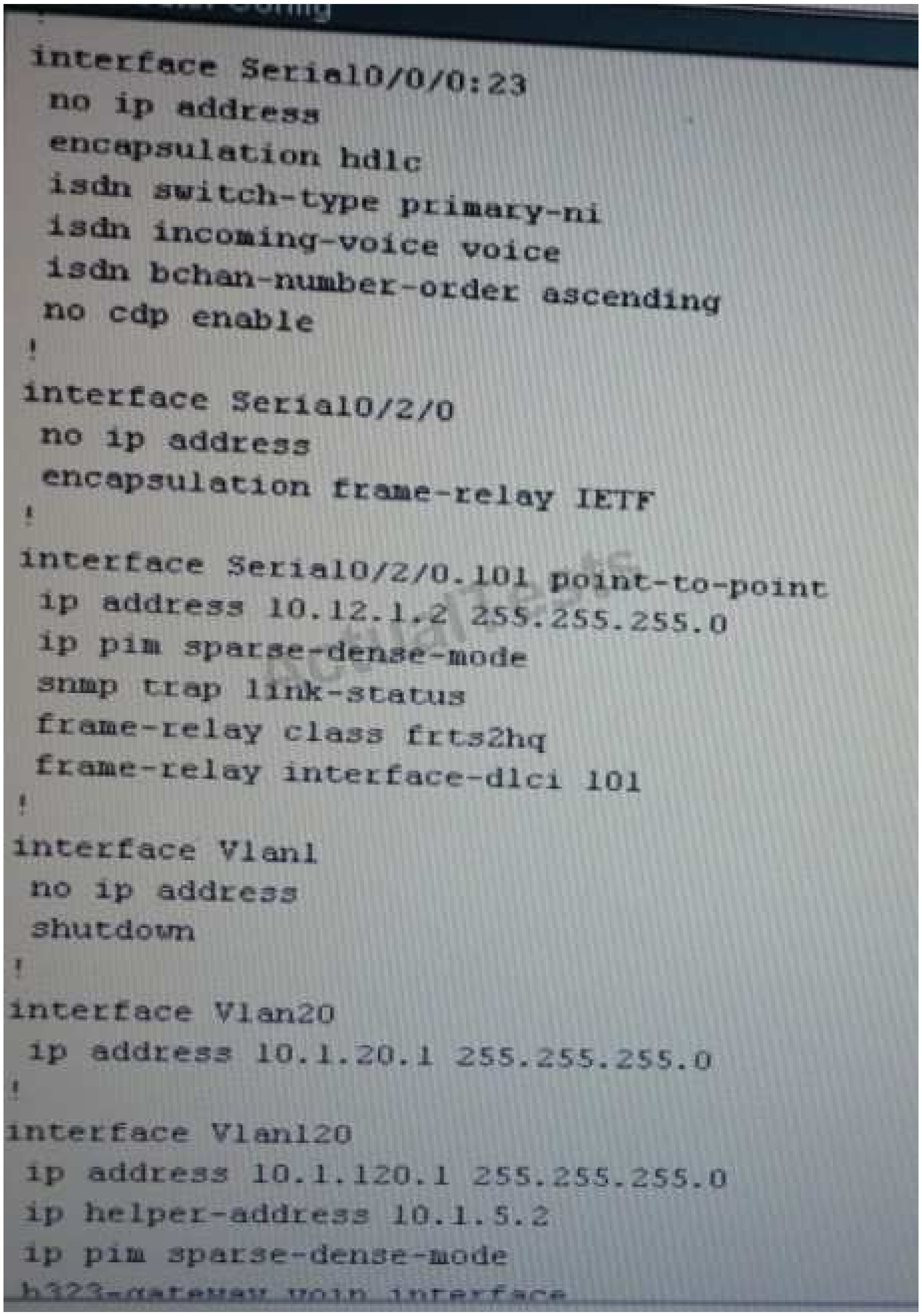
**24 (exhibit):**

```
controller T1 0/0/0
 cablelength short 110
 pri-group timeslots 1-12,24
!
vtp mode transparent
!
vlan 20
 name BR1-Data
!
vlan 120
 name BR1-Voice
!
!
class-map match-all ctraffic
 match ip dscp cs3
class-map match-all vtraffic
 match ip dscp ef
!
policy-map voice2hq
 class vtraffic
 priority 64
 class ctraffic
 bandwidth 8
 class class-default
 fair-queue
policy-map shape2hq
 class class-default
 shape average 486400 4864 0
 service-policy voice2hq
!
```

**25 (exhibit):**

```
!
interface GigabitEthernet0/0
 no ip address
 shutdown
 duplex auto
 speed auto
!
interface GigabitEthernet0/1
 no ip address
 shutdown
 duplex auto
 speed auto
!
interface FastEthernet0/1/0
 description BR1 Phone1
 switchport access vlan 20
 switchport voice vlan 120
 spanning-tree portfast
!
interface FastEthernet0/1/1
 description BR1 Phone2
 switchport access vlan 20
 switchport voice vlan 120
 spanning-tree portfast
!
interface FastEthernet0/1/2
!
interface FastEthernet0/1/3
!
interface Serial0/0/0:23
 no ip address
```

**26 (exhibit):**



```
interface Serial0/0/0:23
 no ip address
 encapsulation hdlc
 isdn switch-type primary-ni
 isdn incoming-voice voice
 isdn bchan-number-order ascending
 no cdp enable
!
interface Serial0/2/0
 no ip address
 encapsulation frame-relay IETF
!
interface Serial0/2/0.101 point-to-point
 ip address 10.12.1.2 255.255.255.0
 ip pim sparse-dense-mode
 snmp trap link-status
 frame-relay class frts2hq
 frame-relay interface-dlci 101
!
interface Vlan1
 no ip address
 shutdown
!
interface Vlan20
 ip address 10.1.20.1 255.255.255.0
!
interface Vlan120
 ip address 10.1.120.1 255.255.255.0
 ip helper-address 10.1.5.2
 ip pim sparse-dense-mode
 h323-gateway voip interface
```

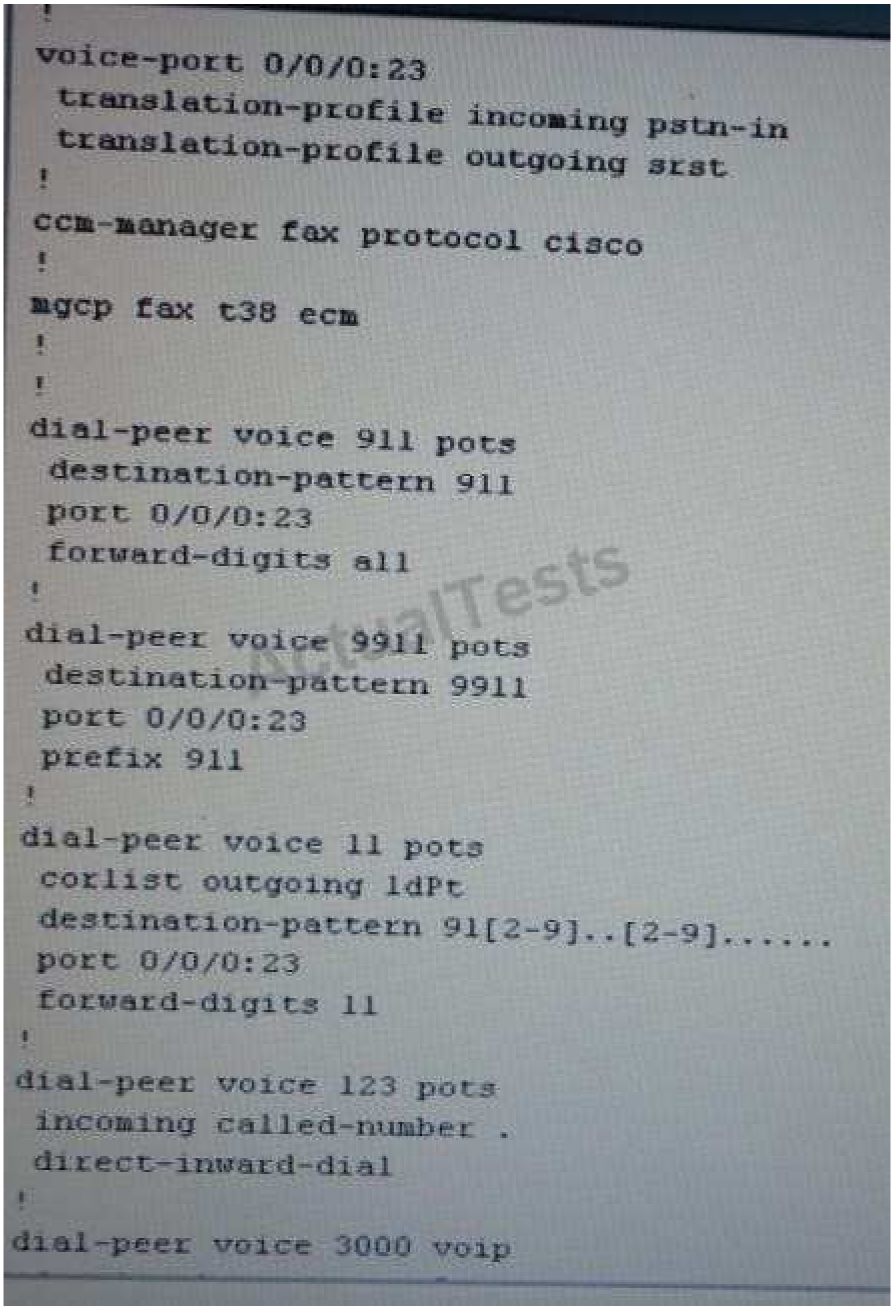
**27 (exhibit):**

BR1 Router Config

```
!
interface Vlan120
 ip address 10.1.120.1 255.255.255.0
 ip helper-address 10.1.5.2
 ip pim sparse-dense-mode
 h323-gateway voip interface
 h323-gateway voip bind srcaddr 10.1.120.1
!
router eigrp 10
 network 10.0.0.0
 no auto-summary
!
ip forward-protocol nd
!
no ip http server
!
!
map-class frame-relay frts2hq
 frame-relay fair-queue
 service-policy output shape2hq
!
control-plane
!
voice-port 0/0/0:23
 translation-profile incoming pstn-in
 translation-profile outgoing srst
!
ccm-manager fax protocol cisco
!
mgcp fax t38 ecm
!
!
```



**28 (exhibit):**



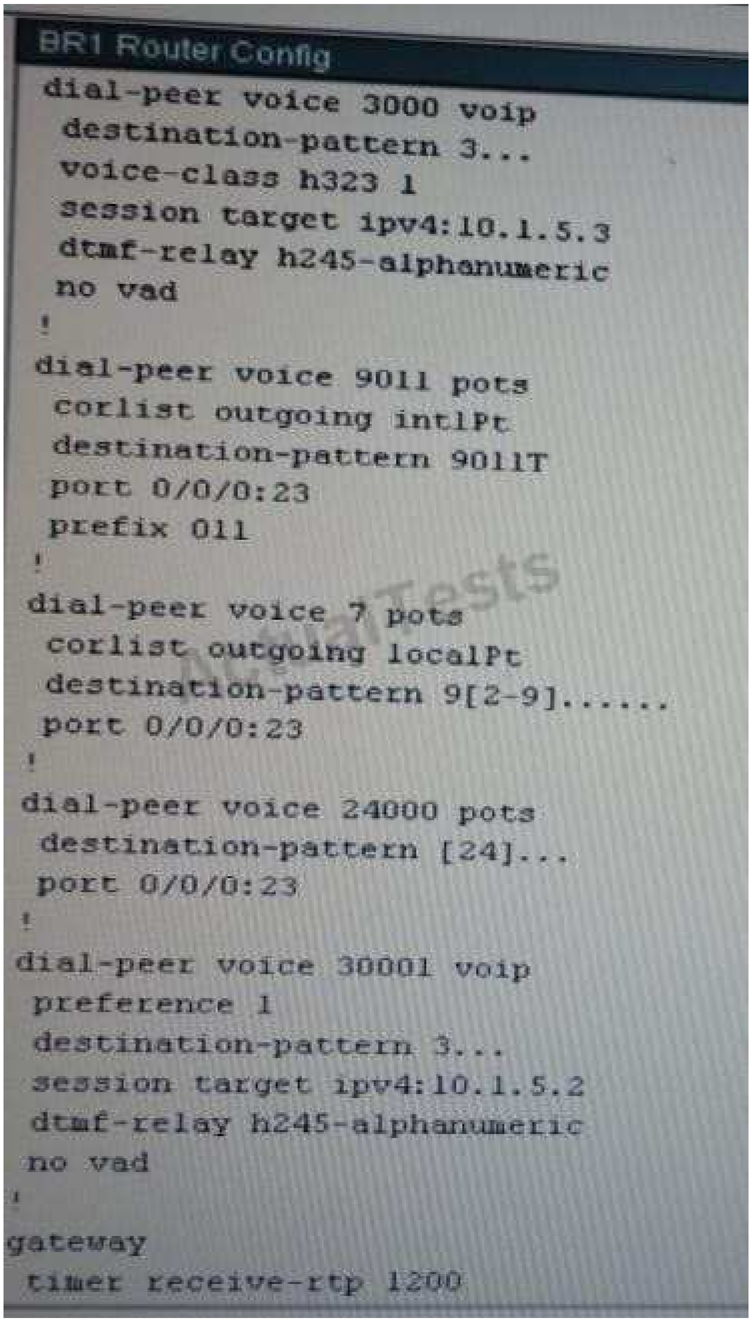
```
!
voice-port 0/0/0:23
 translation-profile incoming pstn-in
 translation-profile outgoing srst
!
ccm-manager fax protocol cisco
!
mgcp fax t38 ecm
!
!
dial-peer voice 911 pots
 destination-pattern 911
 port 0/0/0:23
 forward-digits all
!
dial-peer voice 9911 pots
 destination-pattern 9911
 port 0/0/0:23
 prefix 911
!
dial-peer voice 11 pots
 corlist outgoing ldPt
 destination-pattern 91[2-9]..[2-9]......
 port 0/0/0:23
 forward-digits 11
!
dial-peer voice 123 pots
 incoming called-number .
 direct-inward-dial
!
dial-peer voice 3000 voip
```



**29 (exhibit):**

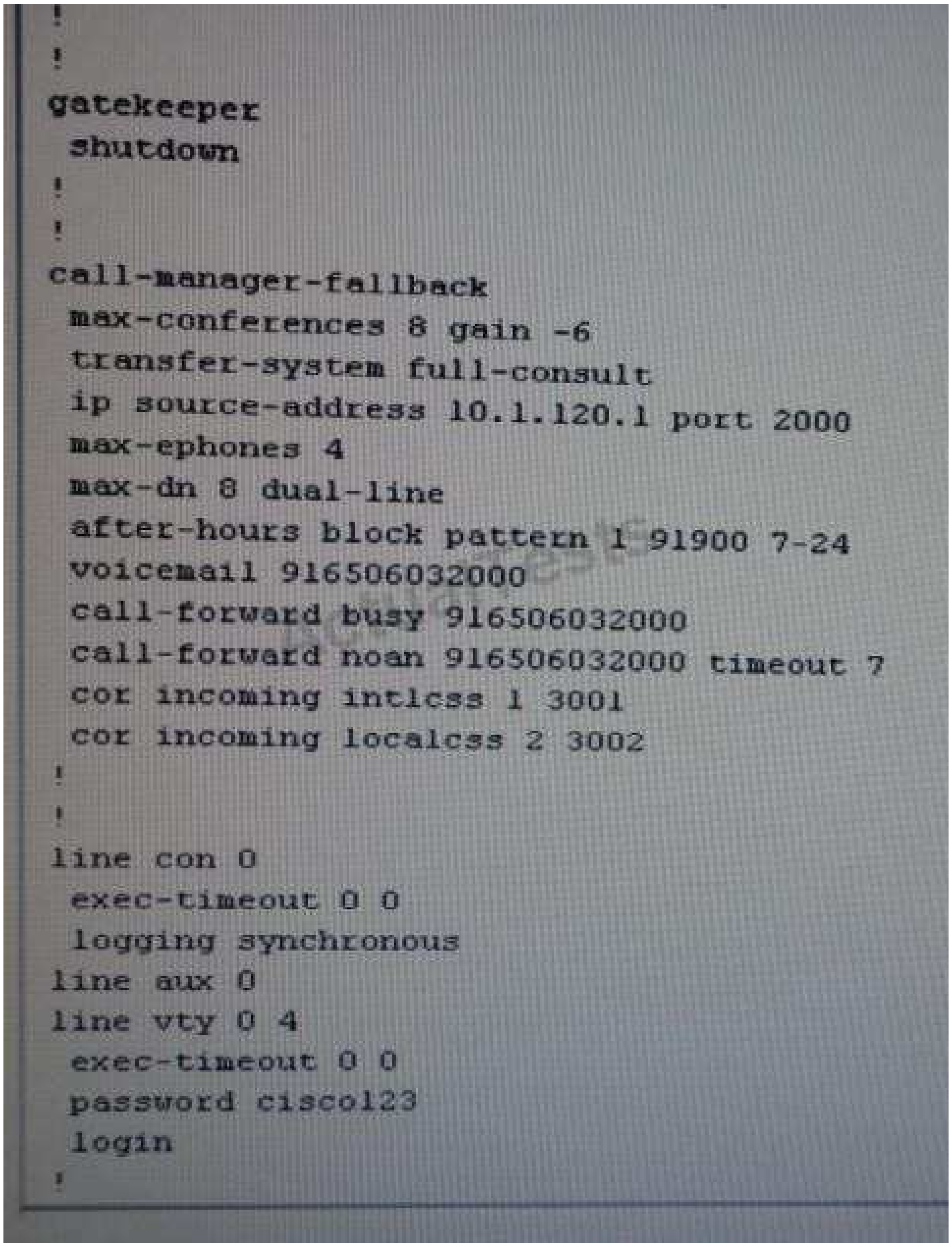
```
 destination-pattern 3...
 voice-class h323 1
 session target ipv4:10.1.5.3
 dtmf-relay h245-alphanumeric
 no vad
!
dial-peer voice 9011 pots
 corlist outgoing intlPt
 destination-pattern 9011T
 port 0/0/0:23
 prefix 011
!
dial-peer voice 7 pots
 corlist outgoing localPt
 destination-pattern 9[2-9]......
 port 0/0/0:23
!
dial-peer voice 24000 pots
 destination-pattern [24]...
 port 0/0/0:23
!
dial-peer voice 30001 voip
 preference 1
 destination-pattern 3...
 session target ipv4:10.1.5.2
 dtmf-relay h245-alphanumeric
 no vad
!
gateway
 timer receive-rtp 1200
```

**30 (exhibit):**


BR1 Router Config

```
dial-peer voice 3000 voip
 destination-pattern 3...
 voice-class h323 1
 session target ipv4:10.1.5.3
 dtmf-relay h245-alphanumeric
 no vad
!
dial-peer voice 9011 pots
 corlist outgoing intlPt
 destination-pattern 9011T
 port 0/0/0:23
 prefix 011
!
dial-peer voice 7 pots
 corlist outgoing localPt
 destination-pattern 9[2-9]......
 port 0/0/0:23
!
dial-peer voice 24000 pots
 destination-pattern [24]...
 port 0/0/0:23
!
dial-peer voice 30001 voip
 preference 1
 destination-pattern 3...
 session target ipv4:10.1.5.2
 dtmf-relay h245-alphanumeric
 no vad
!
gateway
 timer receive-rtp 1200
```

**31 (exhibit):**



```
!
!
gatekeeper
 shutdown
!
!
call-manager-fallback
 max-conferences 8 gain -6
 transfer-system full-consult
 ip source-address 10.1.120.1 port 2000
 max-ephones 4
 max-dn 8 dual-line
 after-hours block pattern 1 91900 7-24
 voicemail 916506032000
 call-forward busy 916506032000
 call-forward noan 916506032000 timeout 7
 cor incoming intlcss 1 3001
 cor incoming localcss 2 3002
!
!
line con 0
 exec-timeout 0 0
 logging synchronous
line aux 0
line vty 0 4
 exec-timeout 0 0
 password cisco123
 login
!
```

A. Change the codec type to G 711. J
B. Configure RSVP call admission control
C. Configure L ink Fragmentation and Interleave on the WAN links
D. Configure RTP header compression on the WAN links
E. Increase the priority queue bandwidth to 80 Kb/s
F. Configure location settings in Cisco Unified Communications Manager to 1 20 Kb/s

**Correct Answer:** CD
**Section: (none)**
**Explanation**

**Explanation/Reference:**
**Explanation:** below link is very good to understand this concept.
Link-http://www.cisco.com/en/US/docs/ios/12_2/qos/configuration/guide/qcflem.html

**QUESTION 9**
**Refer to the exhibit**.

When calling 911, which gateway/route list is defined in the route pattern in Cisco Unified Communications Manager and used to route matched digits to the PSTN?

**Exhibit:**

```
11:01:10.482 |StationD:    (0000008) DialedNumber dialedNumber=911 lineInstance=1 callRefer
11:01:10.482 |StationD:    (0000008) CallState callState=12 lineInstance=1 callReference=20
11:01:10.482 |StationD:    (0000008) (1,100,9,17) CallInfo callingPartyName='' callingParty
11:01:10.482 |StationD:    (0000008) DEBUG- star_DSetCallState(6) State of cdpc(13) is 5.|1
11:01:10.482 |RouteListControl::idle_CcSetupReq - RouteList(LRG_RL), numberSetup=0 numberMe
11:01:10.483 |RouteListControl::idle_CcSetupReq -  RouteList(LRG_RL), RouteListCdrc::create
11:01:10.483 |RouteListCdrc::StartTransition non EMCC call|1,100,74,11.1^*^*
11:01:10.483 |RoutePlanServer::getRouteList() - RouteListName(87da585a-9c00-acc9-a136-6de66
11:01:10.483 |RoutePlanServer::getRouteGroup: standardLocalRG = 00000000-1111-0000-0000-000
11:01:10.483 |RoutePlanServer::getRouteGroup: LRG flag = 1, lRouteGroupName = 00000000-1111
11:01:10.483 |RoutePlanServer::getRouteGroup: standardLocalRG = 00000000-1111-0000-0000-000
11:01:10.483 |RoutePlanServer::getRouteGroup: mDeviceInfoList size =1|*^*^*
11:01:10.483 |RouteListCdrc - RouteList Info, by RouteGroups|*^*^*
11:01:10.484 |RouteList - RouteListName=''LRG_RL'' CallableEndPointName=''87da585a-9c00-acc
11:01:10.484 |TDCLdb.hpp - CallManagerGroup - serverCount = 1|*^*^*
11:01:10.484 |TDCLdb.hpp - CallManagerGroup - nodeId = 1|*^*^*
11:01:10.484 |RouteList - RouteGroup count=''1''|*^*^*
11:01:10.484 |RouteListCdrc - RouteGroup count = 1|*^*^*
11:01:10.484 |RouteListCdrc - Device count = 1|*^*^*
11:01:10.484 |RouteListCdrc::null0_CcSetupReq check vipr call mViprReroute=0 mViprAlreadyAt
11:01:10.484 |RouteListCdrc::null0_CcSetupReq - gets a next group while 1 groups remain. mA
11:01:10.484 |RouteListCdrc::algorithmCategorization -- CDRC_SERIAL_DISTRIBUTION type=1|1,1
11:01:10.484 |RouteListCdrc::createDistributedDeviceInfoList check vipr call flag mViprRero
11:01:10.484 |RouteListCdrc::null0_ccsetupreq -- newBusyRejFlag = 0, last route_setting = 1
11:01:10.484 |RouteListCdrc::null0_CcSetupReq - Selecting a device.|1,100,49,1.13149^10.1.1
11:01:10.484 |RouteListCdrc::selectDevices -- mTemporaryDeviceInfoList.size = 1.|1,100,49,1
11:01:10.484 |RouteListCdrc::null0_CcSetupReq - RNAR timeout = 0.|1,100,49,1.13149^10.1.110
11:01:10.484 |SMDMSharedData::findAliasRegInfo - AliasName = b87f9c1e-e8e8-0b90-4460-1611fc
11:01:10.484 |DeviceManager::star_DmPidReq - RequestedName=b87f9c1e-e8e8-0b90-4460-1611fc8b
11:01:10.484 |SMDMSharedData::findRemoteDeviceAny - Key=b87f9c1e-e8e8-0b90-4460-1611fc8b19c
11:01:10.484 |RouteListCdrc::select_facility_DmPidErr; Unable to locate DeviceName = b87f9c
11:01:10.484 |RouteListCdrc::markDeviceAsDown|1,100,49,1.13149^10.1.110.20^SEP002290BA361B
11:01:10.485 |RouteListCdrc::select_facility_DmPidErr: Execute a route action.|1,100,49,1.1
11:01:10.485 |RouteListCdrc::algorithmCategorization -- CDRC_SERIAL_DISTRIBUTION type=1|1,1
11:01:10.485 |RouteListCdrc::whichAction -- DOWN (Current Group) = 1|1,100,49,1.13149^10.1.
11:01:10.485 |RouteListCdrc::routeAction -- current device name=b87f9c1e-e8e8-0b90-4460-161
11:01:10.485 |RouteListCdrc::executeRouteAction: SKIP_TO_NEXT_MEMBER|1,100,49,1.13149^10.1.
11:01:10.485 |RouteListCdrc::skipToNextMember|1,100,49,1.13149^10.1.110.20^SEP002290BA361B
```

A. SEP002290BA361B
B. standardLocalRG
C. RouteListCdrc
D. LRG_RL
E. nodeId = 1
F. BRANCH

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

**Explanation/Reference:**
**Explanation:** logs clearly showing route list name

**QUESTION 10**
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

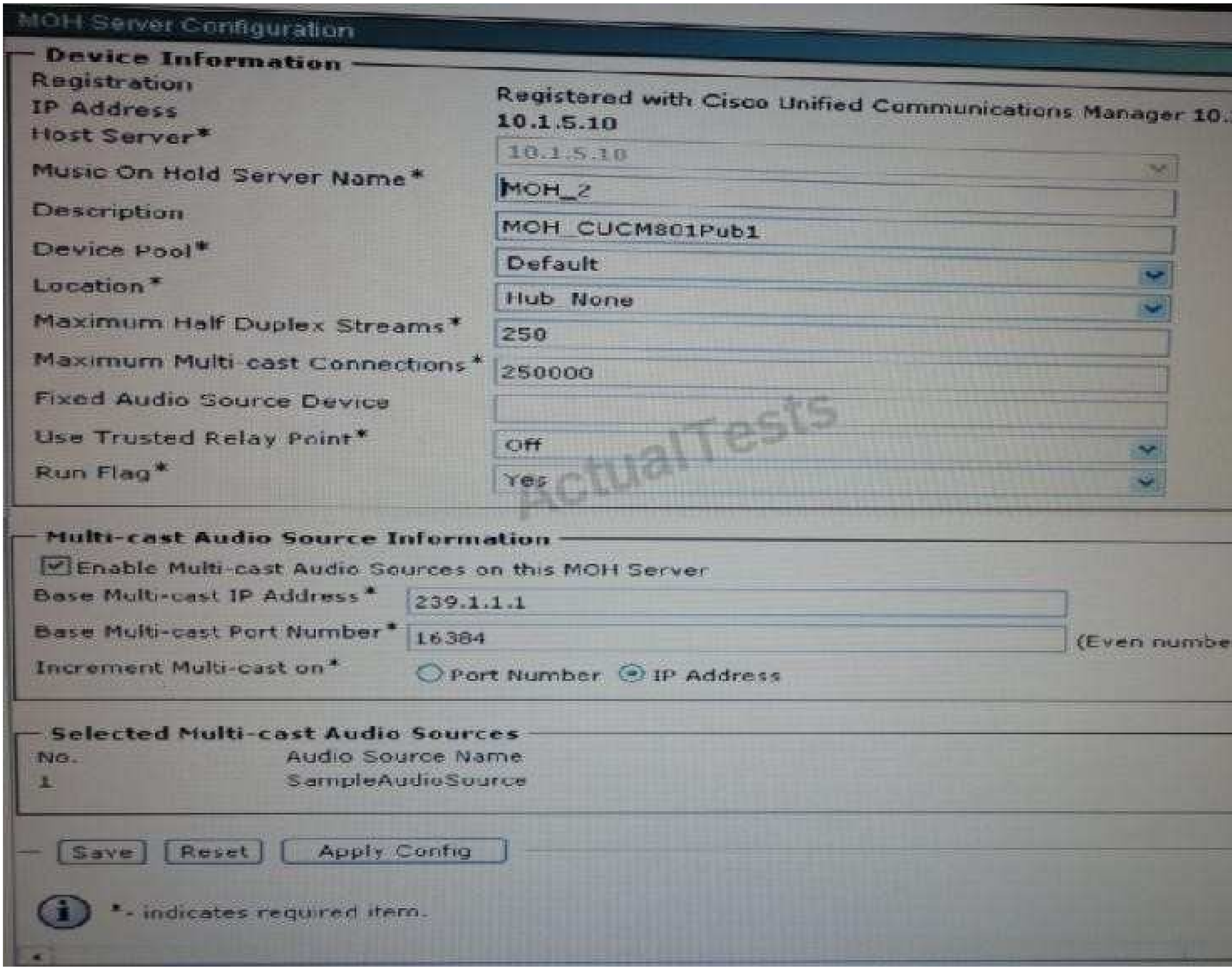
**Correct Answer:** B
**Section: (none)**
**Explanation**

**Explanation/Reference:**

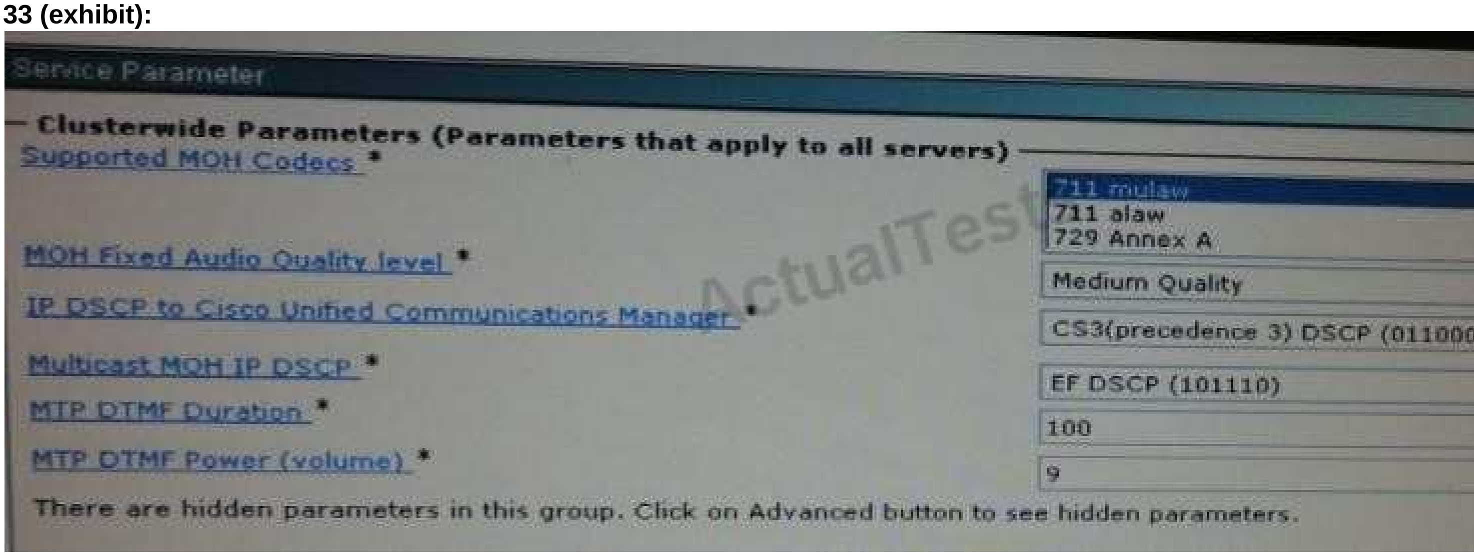
**QUESTION 11**
## Refer to the exhibits.

MOH has been configured to run from flash at the BR1 site. The HQ phones and MOH server are placed in the Default region through the Default device pool. The BR1 phones are placed in the BR1 region through the BR1 device pool. The region configuration between Default and BR1 only permits G.729 codec. When an IP phone user at the HQ site places a BR1 caller on hold, the BR1 caller hears tone on hold. Which of the following can cause this issue?

**32 (exhibit):**


MOH Server Configuration
Device Information
Registration
IP Address
Host Server*
Music On Hold Server Name*
Description
Device Pool*
Location*
Maximum Half Duplex Streams*
Maximum Multi-cast Connections*
Fixed Audio Source Device
Use Trusted Relay Point*
Run Flag*
Registered with Cisco Unified Communications Manager 10.
10.1.5.10
10.1.5.10
MOH_2
MOH_CUCM801Pub1
Default
Hub_None
250
250000
Off
Yes
Multi-cast Audio Source Information
Enable Multi-cast Audio Sources on this MOH Server
Base Multi-cast IP Address*
239.1.1.1
Base Multi-cast Port Number*
16384
(Even numbe
Increment Multi-cast on*
Port Number
IP Address
Selected Multi-cast Audio Sources
No.
Audio Source Name
1
SampleAudioSource
Save
Reset
Apply Config
*- indicates required item.

**33 (exhibit):**


Service Parameter
Clusterwide Parameters (Parameters that apply to all servers)
Supported MOH Codecs *
711 mulaw
711 alaw
729 Annex A
MOH Fixed Audio Quality level *
Medium Quality
IP DSCP to Cisco Unified Communications Manager *
CS3(precedence 3) DSCP (01100
Multicast MOH IP DSCP *
EF DSCP (101110)
MTP DTMF Duration *
100
MTP DTMF Power (volume) *
9
There are hidden parameters in this group. Click on Advanced button to see hidden parameters.

**34 (exhibit):**

SRST Config

```
version 12.4
service timestamps debug datetime msec
service timestamps log datetime msec
no service password-encryption
!
hostname BR1
!
card type t1 0 0
logging message-counter syslog
enable password cisco123
!
no aaa new-model
network-clock-participate wic 0
!
ip source-route
!
!
ip cef
ip dhcp excluded-address 10.1.20.1 10.1.20.9
ip dhcp excluded-address 10.1.20.21 10.1.20.254
!
ip dhcp pool Data
  network 10.1.20.0 255.255.255.0
  default-router 10.1.20.1
!
!
no ip domain lookup
no ipv6 cef
multilink bundle-name authenticated
!
!
```



**35 (exhibit):**

```
!
isdn switch-type primary-ni
!
!
!
!
!
voice translation-rule 1
 rule 1 /^710(....$)/ /□/
 rule 2 /^212710(....$)/ /□/
!
voice translation-rule 2
 rule 1 /^2/ /6506032/ type any national
 rule 2 /^4/ /4989531214/ type any international
 rule 3 /^9011/ // type any international
!
voice translation-rule 3
 rule 1 /^3...$/ /212710&/
!
!
voice translation-profile pstn-in
 translate called 1
!
voice translation-profile srst
 translate calling 3
 translate called 2
!
!
voice-card 0
 dspfarm
 dsp services dspfarm
```

**36 (exhibit):**

```
!
vtp mode transparent
archive
 log config
  hidekeys
!
!
controller T1 0/0/0
 cablelength short 110
 pri-group timeslots 1-12,24
!
vlan 20
 name BR1-Data
!
vlan 120
 name BR1-Voice
!
!
!
!
interface FastEthernet0/1/0
 description BR1 Phone1
 switchport access vlan 20
 switchport voice vlan 120
 spanning-tree portfast
!
interface FastEthernet0/1/1
 description BR1 Phone2
 switchport access vlan 20
 switchport voice vlan 120
 spanning-tree portfast
```

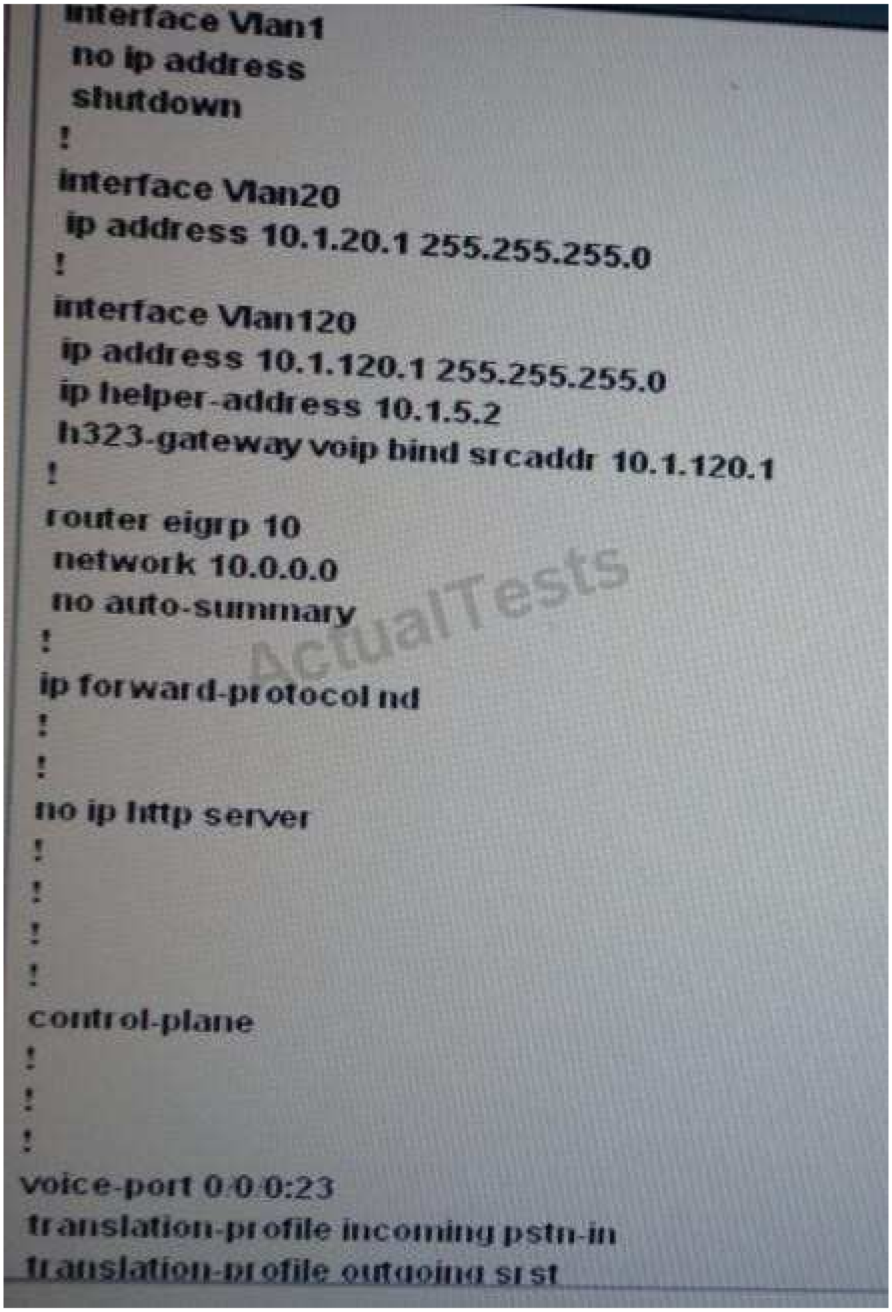


**37 (exhibit):**

```
!
interface FastEthernet0/1/1
 description BR1 Phone2
 switchport access vlan 20
 switchport voice vlan 120
 spanning-tree portfast
!
!
interface Serial0/0/0:23
 no ip address
 encapsulation hdlc
 isdn switch-type primary-ni
 isdn incoming-voice voice
 isdn bchan-number-order ascending
 no cdp enable
!
interface Serial0/2/0
 no ip address
 encapsulation frame-relay IETF
 !
interface Serial0/2/0.101 point-to-point
 ip address 10.12.1.2 255.255.255.0
 ip pim sparse-dense-mode
 snmp trap link-status
 frame-relay interface-dlci 101
!
interface Vlan1
 no ip address
 shutdown
!
interface Vlan20
 ip address 10.1.20.1 255.255.255.0
```



**38 (exhibit):**



```
interface Vlan1
 no ip address
 shutdown
!
interface Vlan20
 ip address 10.1.20.1 255.255.255.0
!
interface Vlan120
 ip address 10.1.120.1 255.255.255.0
 ip helper-address 10.1.5.2
 h323-gateway voip bind srcaddr 10.1.120.1
!
router eigrp 10
 network 10.0.0.0
 no auto-summary
!
ip forward-protocol nd
!
!
no ip http server
!
!
!
!
control-plane
!
!
!
voice-port 0/0/0:23
 translation-profile incoming pstn-in
 translation-profile outgoing srst
```



**39 (exhibit):**

```
voice-port 0/0/0:23
 translation-profile incoming pstn-in
 translation-profile outgoing srst
!
ccm-manager fax protocol cisco
!
mgcp fax t38 ecm
!
!
dial-peer voice 911 pots
 destination-pattern 911
 port 0/0/0:23
 forward-digits all
!
dial-peer voice 9911 pots
 destination-pattern 9911
 port 0/0/0:23
 forward-digits all
!
dial-peer voice 123 pots
 incoming called-number .
 direct-inward-dial
!
dial-peer voice 3000 voip
 destination-pattern 3...
 session target ipv4:10.1.5.10
 dtmf-relay h245-alphanumeric
 no vad
!
dial-peer voice 9011 pots
 corlist outgoing intlPt
```



**40 (exhibit):**

```
dial-peer voice 9011 pots
 corlist outgoing intlPt
 destination-pattern 9011T
 port 0/0/0:23
!
dial-peer voice 7 pots
 corlist outgoing localPt
 destination-pattern 9[2-9]......
 port 0/0/0:23
!
dial-peer voice 24000 pots
 destination-pattern [24]...
 port 0/0/0:23
!
!
dial-peer voice 11 pots
 corlist outgoing ldPt
 destination-pattern 91[2-9]..[2-9]......
 port 0/0/0:23
!
!
gateway
 timer receive-rtp 1200
!
!
gatekeeper
 shutdown
!
!
call-manager-fallback
 max-conferences 8 gain -6
```



**41 (exhibit):**

```
gatekeeper
 shutdown
!
!
call-manager-fallback
 max-conferences 8 gain -6
 transfer-system full-consult
 ip source-address 10.1.120.1 port 2000
 max-ephones 4
 max-dn 8 dual-line
 moh music-on-hold.au
 multicast moh 239.1.1.1 port 16384
!
line con 0
 exec-timeout 0 0
 logging synchronous
line aux 0
line vty 0 4
 exec-timeout 0 0
 password cisco123
 login
!
scheduler allocate 20000 1000
end
```



A. Multicast routing is not enabled on the BR1 router.
B. The command ip pim separate-dense-mode is missing from interface VLAN 120 at the SRST router in BR1
C. The MOH server is unable to stream MOH using G.711 codec because of the regions configuration.
D. The command route 10.1.120.1 must be added to the multicast moh 239.1.1.1 port 16384 command at the SRST router in BR1
E. The Max Hops is too small in the MOH configuration

**Correct Answer:** B
**Section: (none)**
**Explanation**

**Explanation/Reference:**
**Explanation:** The router runs IP Multicast routing and IP PIM sparse-dense mode on any physical interface that must participate in multicast (PIM is in either sparse or dense mode, but the interface can be configured to

forward sparse mode, dense mode, or both).
**Link-** http://www.cisco.com/en/US/technologies/tk436/tk428/
technologies_white_paper0900aecd80131281_ns465_Networking_Solutions_White_Paper.html

**QUESTION 12**
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.

**Correct Answer:** B
**Section: (none)**
**Explanation**

**Explanation/Reference:**
**Explanation: Explanation-**Yes if The Cisco Unified Communications Manager group that is applied through the device pool is misconfigured then IP phone doesn’t recognized the subscriber IP address.
Link-http://www.cisco.com/en/US/docs/voice_ip_comm/cucm/admin/7_0_1/ccmcfg/b02devpl.html

**QUESTION 13**
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

**Correct Answer:** E
**Section: (none)**
**Explanation**

**Explanation/Reference:**
**Explanation:** http://www.cisco.com/en/US/docs/internetworking/troubleshooting/guide/tr1901.html Step 2 Gather the facts that you need to help isolate possible causes. Ask questions of affected users, network administrators, managers, and other key people. Collect information from sources such as network management systems, protocol analyzer traces, output from router diagnostic commands, or software release notes.

**QUESTION 14**
**Refer to the exhibit.**

```
*Mar 24 16:17:54.190: ISDN Se0/0/0: 15 Q931: RX <- SETUP pd = 8 callref = 0x00AA
  Beaere Capability i = 0x8090A3
    Standard = CCITT
    Transfer Capability = Spee
    Transfer Mode = Circuit
    Transfer Rate = 64 kbit/s
 Channel ID i = 0xA98381
 Exclusive, Channel 1
 Progress Ind i = 0x8183 - Origination address is non-ISDN
 Calling Party Number i = 0x1180, '4940302156001'
 Plan:ISDN, Type:International
 Called Party Number i = 0x81, '2288223001'
 Plan:ISDN, Type:Unknown
 *Mar 24 16:17:54:210: ISDN Se0/0/0 15 Q931: TX-> RELEASE_COMP pd=8 callref=
 0x80AA
 Cause i = 0x8081 = Unallocated/unaligned number
```

The exhibit shows the output of debug isdn q931. An inbound PSTN call was received by a SIP gateway that is reachable via a SIP trunk that is configured in Cisco Unified Communications Manager. The call failed to ring extension 3001. If the phone at extension 3001 is registered and reachable through the gateway inbound CSS, which three actions can resolve this issue? (Choose three.)

A. Change the significant digits for inbound calls to 4 on the SIP trunk configuration in Cisco Unified Communications Manager
B. Configure the digit strip 4 on the SIP trunk under Incoming Called Party Settings in Cisco Unified Communications Manager
C. Configure a translation pattern in Cisco Unified Communications Manager that can be accessed by the trunk CSS to truncate the called number to four digits
D. Configure a called-party transformation CSS on the gateway in Cisco Unified Communications Manager that includes a pattern that transforms the number from ten digits tofour digits
E. Configure a voice translation profile in the SIP Cisco IOS gateway with a voice translation rule that truncates the number from ten digits to four digits
F. Configure the Cisco IOS command num-exp 2288223001 3001 on the gateway ISDN interface.

**Correct Answer:** ACE
**Section: (none)**
**Explanation**

**Explanation/Reference:**

**QUESTION 15**
Which of these is used by the Cisco IP phone to relay to the switch the information regarding how much power is needed?

A. the Cisco Discovery Protocol
B. IEEE 802.10 protocol
C. Cisco IP phones always use a fixed power consumption hased on the resistor, which is specific to the model
D. The switch model determines how much power is consumed by the different phone models

**Correct Answer:** A
**Section: (none)**

**Explanation**

**Explanation/Reference:**
**Explanation**-if CDP is enabled on the switch, 15.4W is initially allocated, and then further refined when the CDP message is received from the PD **Link-** http://www.cisco.com/en/US/products/hw/phones/ps379/products_qanda_item09186a00808996f3. shtml

**QUESTION 16**
## Refer to the exhibit

| RTP Phone Device Configuration | Partitions |
|---|---|
| Device CSS | RTP_Emergency<br>ALL_Phones |
| AAR CSS | RTP_LongDistance |

| RTP Phone DN Configuration | Partitions |
|---|---|
| Line CSS | RTP_Local<br>RTP_LongDistance<br>RTP_International |
| AAR Group | AAR |



| U.K. User Device Profile | Partitions |
|---|---|
| Line CSS | U.K_Emergency<br>ALL_Phones |
| AAR Group | AAR |

| Partition | Route Pattern |
|---|---|
| RTP_Emergency | 9.911 |
| RTP_Local | 9.[2-9]XXXXXX |
| RTP_LongDistance | 9.1[2-9]XX[2-9]XXXXXX |
| RTP_International | 9.011!# |
| U.K_Emergency | 0.000 |
| U.K_PSTN | 9.! |

Assume a centralized Cisco Unified Communications Manager topology with the headquarters at RTP and remote located at the U.K. All route patterns are assigned a route list that contains a route group pointing to the local gateway. RTP route patterns use the RTP gateway, and U.K. route patterns use the U.K. gateway.

When a U.K. user logs into an RTP phone using the Cisco Extension Mobility feature and places an emergency call to 0000, which statement about the emergency call is true?

A. The call will match the U.K_Emergency route pattern partition and will egress at the RTP gateway.
B. The call will match the U.K_Emergency route pattern partition and will egress at the U.K. gateway.
C. The call will match the RTP_Emergency route pattern partition and will egress at the RTP gateway
D. The call will match the RTP_Emergency route pattern partition and will egress at the U.K. gateway.
E. The call will fail

**Correct Answer:** B
**Section: (none)**
**Explanation**

**Explanation/Reference:**

**QUESTION 17**
Which issue would cause an MGCP gateway to fail to register with Cisco Unified Communications Manager?

A. missing the configuration command isdn bind-13 ccm-manager under the ISDN interface
B. mismatched domain name on the MGCP gateway and Cisco Unified Communications Manager gateway configuration
C. misconfigured route group in Cisco Unified Communications Manager
D. incorrect MGCP IP address specified in the gateway configuration in Cisco Unified Communications Manager

**Correct Answer:** B
**Section: (none)**
**Explanation**

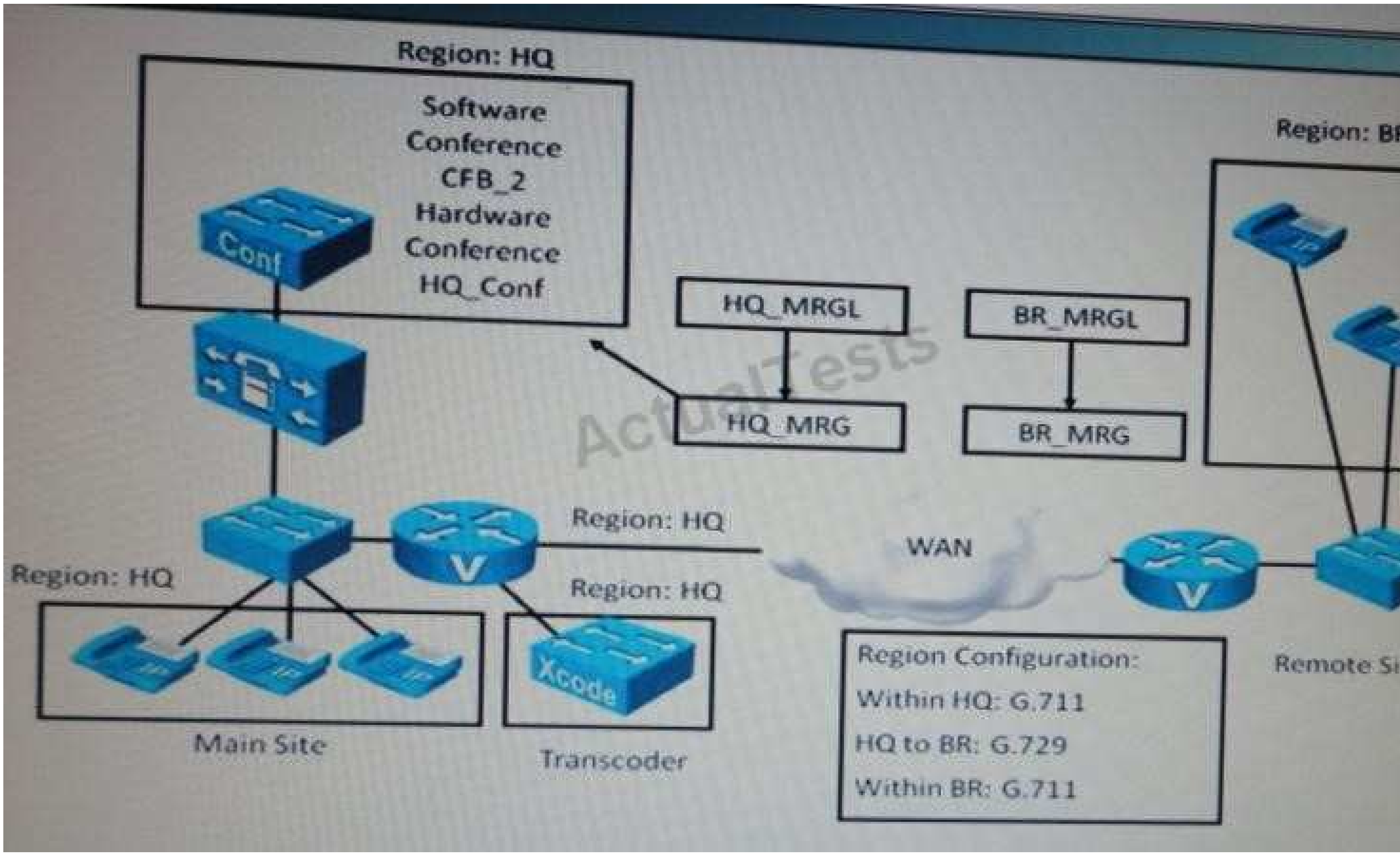
**Explanation/Reference:**
**Explanation**-
This problem is a domain name issue. If a domain name is configured on the MGCP gateway, the domain name for the gateway configuration on Cisco CallManager must be the same.
**Link-** http://www.cisco.com/en/US/products/sw/voicesw/ps556/products_tech_note09186a00805a316c.s html

**QUESTION 18**
## Refer to the exhibits.

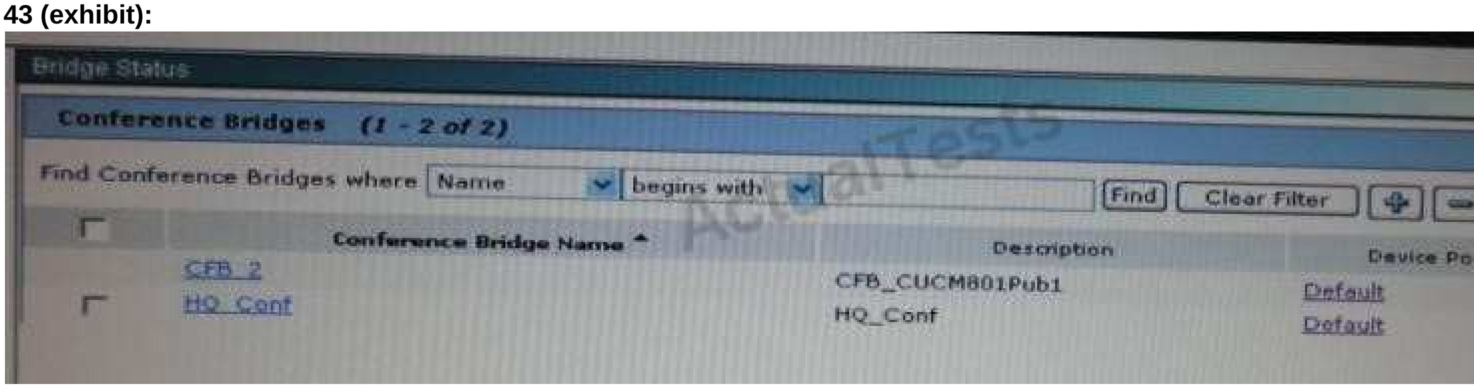
The HG_MRG that is shown in the exhibit is assigned to an MRGL, which is configured at the HQ phones.

A call exists between two HQ phones that use G.711 codec. When one of the HQ users attempts to conference a BR phone across the WAN, the conference fails. The SDI trace shows an error "No transcoder device configured."
Which statement indicates the correct resolution or reason for the issue?

**42 (exhibit):**




**43 (exhibit):**




**44 (exhibit):**


**45 (exhibit):**

IOS Config

```
!
sccp local GigabitEthernet0/0.110
sccp ccm 10.1.5.10 identifier 1 version 7.0
sccp
!
sccp ccm group 1
 associate ccm 1 priority 1
 associate profile 1 register HQ_Conf
!
dspfarm profile 1 conference
 codec g711ulaw
 codec g711alaw
 codec g729ar8
 codec g729abr8
 codec g729r8
 codec g729br8
 maximum sessions 5
 associate application SCCP
!
|
```

**46 (exhibit):**

MRG

**Media Resource Group Information**

Name* MRG_HQ

Description

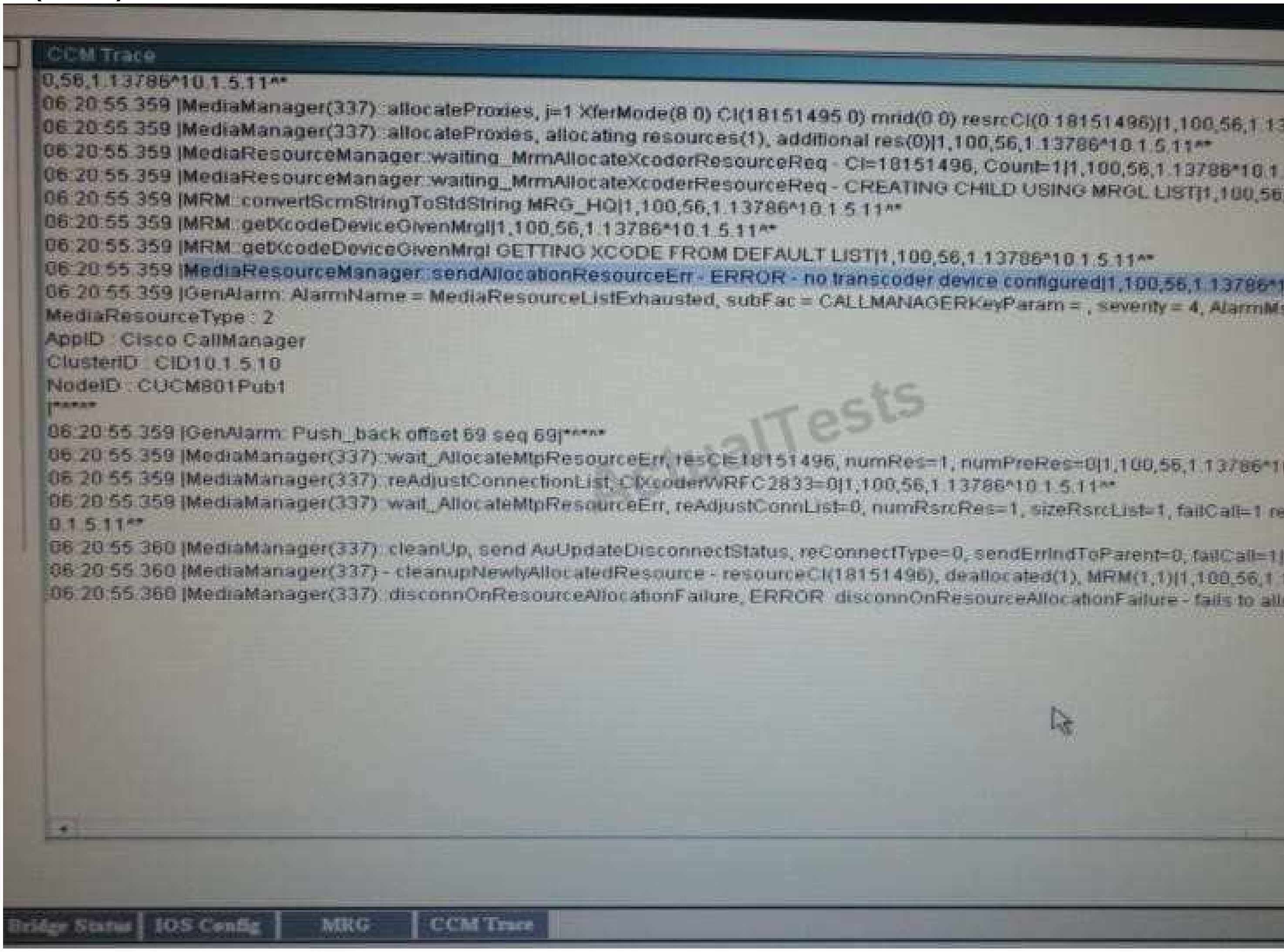
**Devices for this Group**

Available Media Resources**
ANN_2
HQ_MTP
HQ_SIP_MTP
MOH_2[Multicast]
MTP_2

Selected Media Resources*
CFB_2 (CFB)
HQ_Conf (CFB)

☐ Use Multi-cast for MOH Audio (If at least one multi-cast MOH resource is available)

**47 (exhibit):**



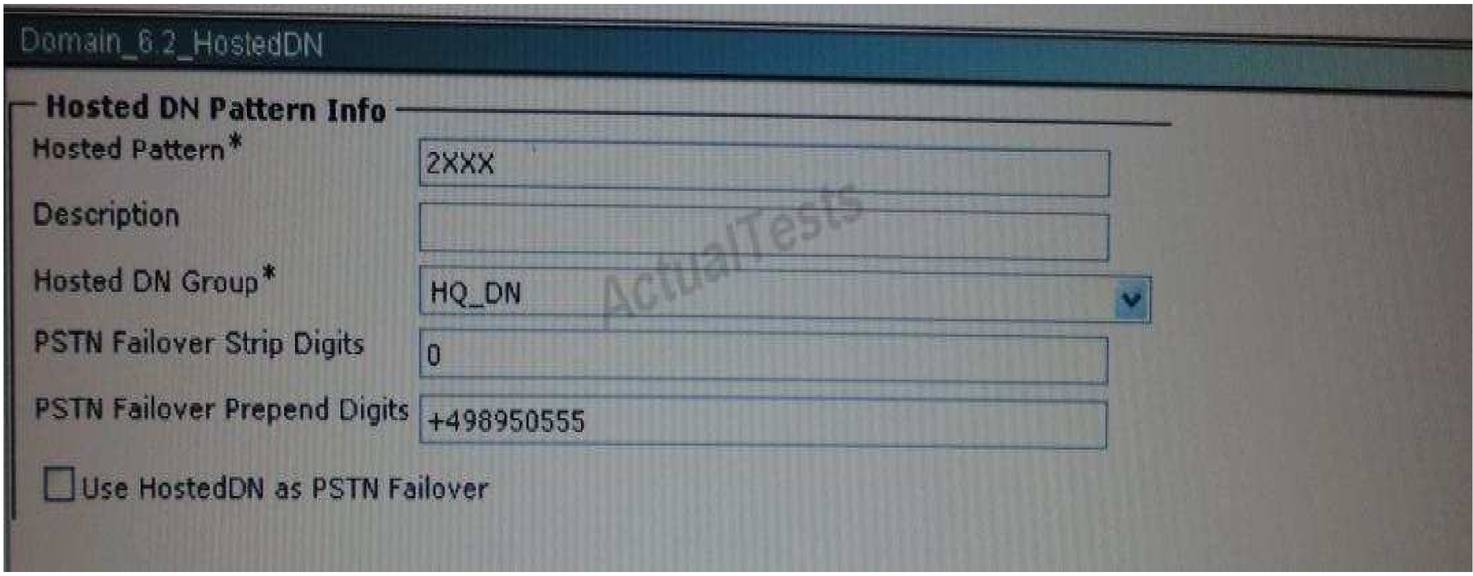

A. The BR phone does not have access to the HO_Conf bridge
B. The BR phone does not have access to the CFB_2 bridge
C. The BR phone does not have access to a transcoder
D. The CFB_2 bridge should be removed from the HQ_MRG and assigned to an MRG that is not assigned to an MRGL
E. The CFB_2 bridge should be listed last in the HO_MRG

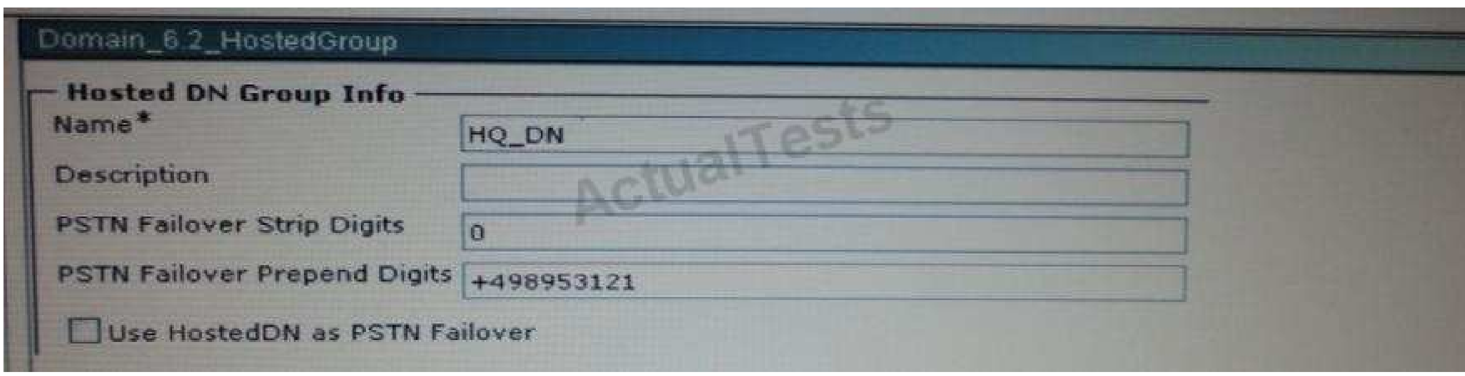
**Correct Answer:** E
**Section: (none)**
**Explanation**

**Explanation/Reference:**
**Explanation**:
In the group MRG_HQ are two conference system in the following sequence is entered:1. Software = CFB_22. Hardware = HQ_ConfIt is as always the first group CFB_2 used. But as they only support G711 calls the call will fail. Only the conference originator need access to the transcoderSee TVOICE V 2 6-71

**QUESTION 19**

**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?

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:

**Correct Answer:** A
**Section: (none)**
**Explanation**

**Explanation/Reference:**

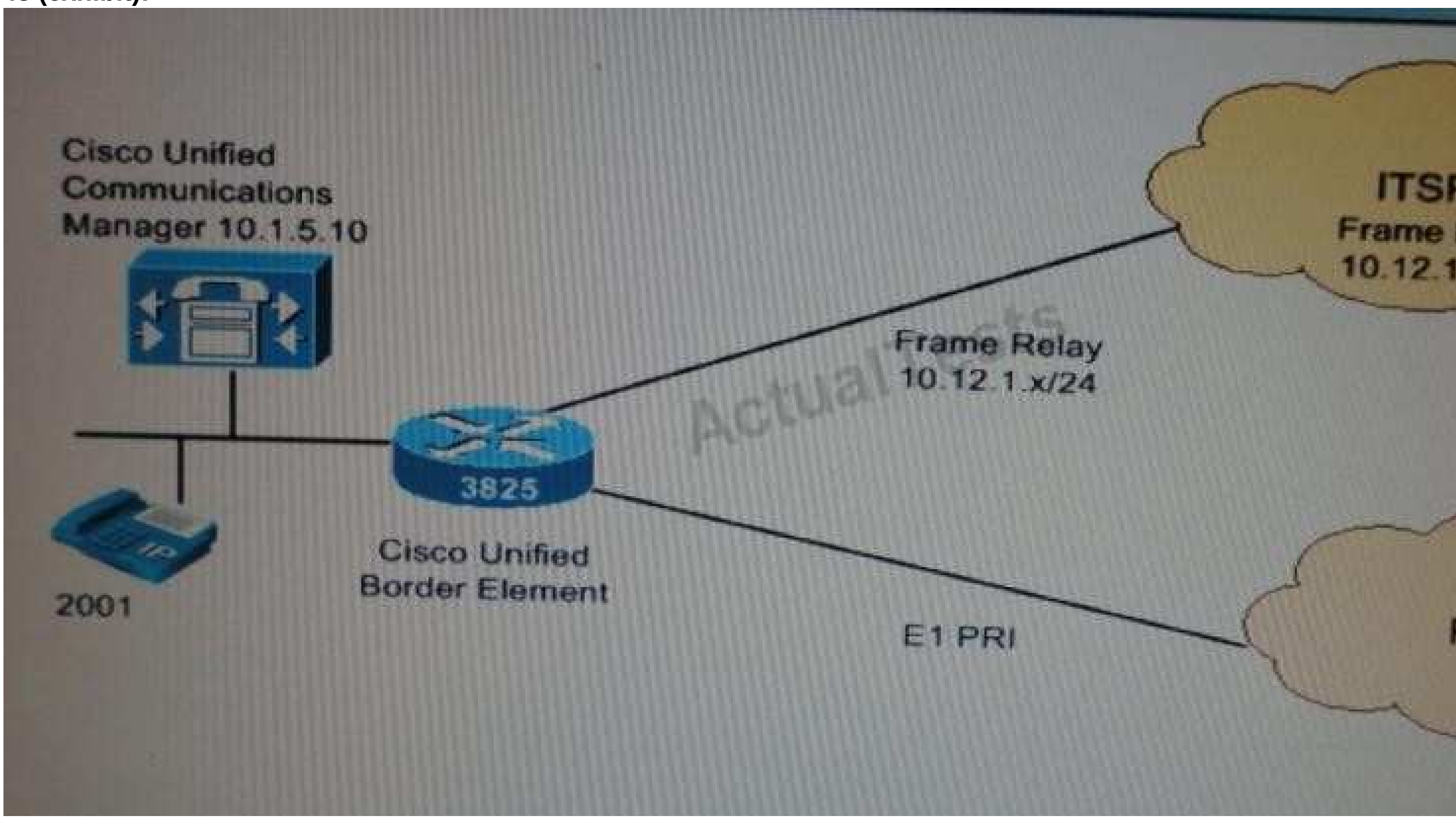
**QUESTION 20**
**Refer to the exhibits.**

When the IP phone 2001 places a call to 9011 49403021 56001, the call is sent to the Cisco Unified Border

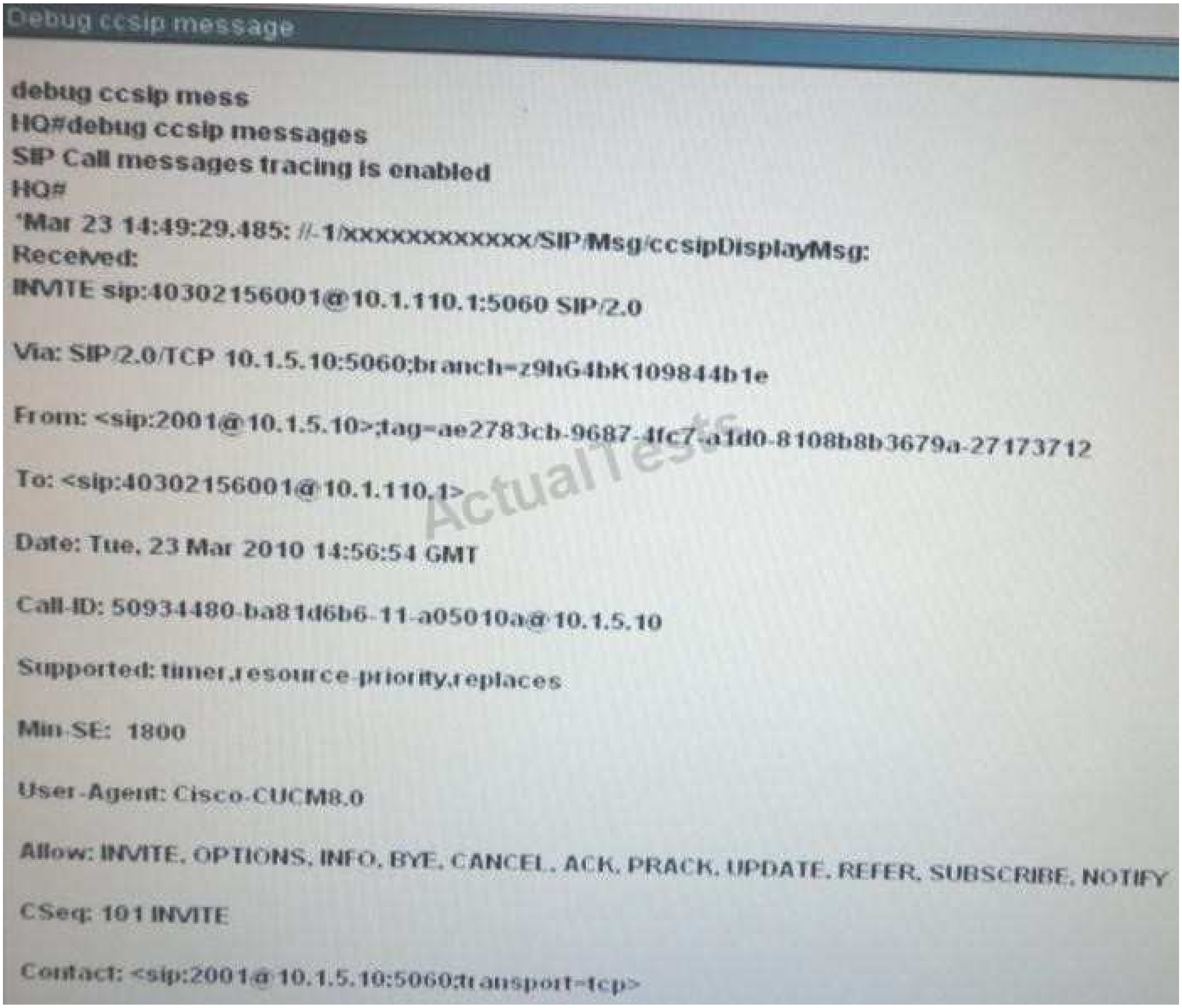
Element as 40302156001 which is what the ITSP expects to receive. The ITSP support personnel claim that they never saw the call. Issuing the debug CCSIP message command on the Cisco Unified Border Element results in the message "SIP/2 0 404 Not Found".

Refer to the Cisco Unified Border Element configuration, debug voice dial and ccsip messages exhibits. Which situation can cause this issued?

**48 (exhibit):**




**49 (exhibit):**



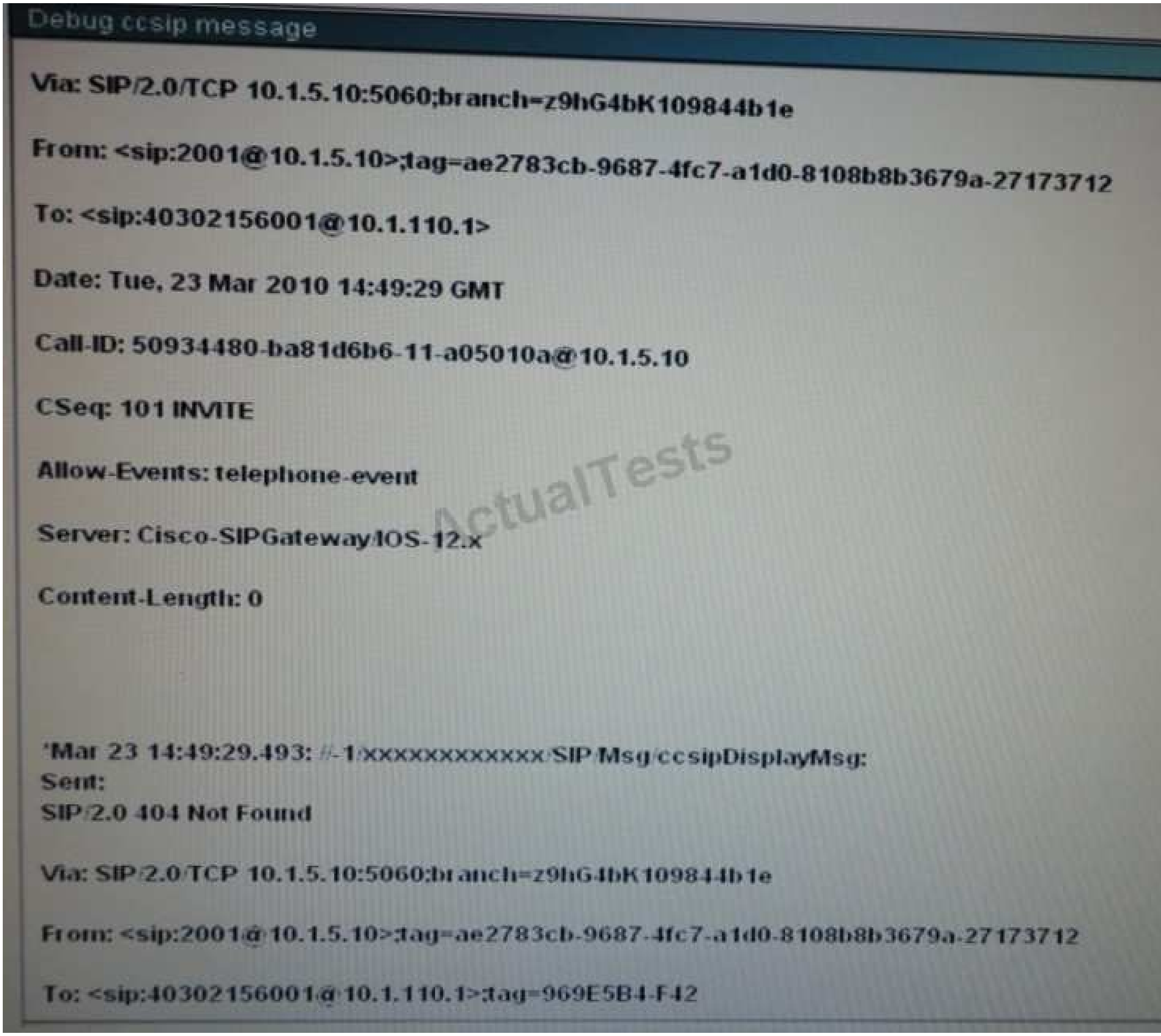
```
Debug ccsip message

debug ccsip mess
HQ#debug ccsip messages
SIP Call messages tracing is enabled
HQ#
*Mar 23 14:49:29.485: //-1/xxxxxxxxxxxx/SIP/Msg/ccsipDisplayMsg:
Received:
INVITE sip:40302156001@10.1.110.1:5060 SIP/2.0

Via: SIP/2.0/TCP 10.1.5.10:5060;branch=z9hG4bK109844b1e

From: <sip:2001@10.1.5.10>;tag=ae2783cb-9687-4fc7-a1d0-8108b8b3679a-27173712

To: <sip:40302156001@10.1.110.1>

Date: Tue, 23 Mar 2010 14:56:54 GMT

Call-ID: 50934480-ba81d6b6-11-a05010a@10.1.5.10

Supported: timer,resource-priority,replaces

Min-SE:  1800

User-Agent: Cisco-CUCM8.0

Allow: INVITE, OPTIONS, INFO, BYE, CANCEL, ACK, PRACK, UPDATE, REFER, SUBSCRIBE, NOTIFY

CSeq: 101 INVITE

Contact: <sip:2001@10.1.5.10:5060;transport=tcp>
```

**50 (exhibit):**

```
Expires: 180

Allow-Events: presence, kpml

Supported: X-cisco-srtp-fallback

Supported: Geolocation

Call-Info: <sip:10.1.5.10:5060>;method="NOTIFY;Event=telephone-event;Duration=500"

Cisco-Guid: 1351828608-3129071286-0000000015-0168100106

Session-Expires:  1800

P-Asserted-Identity: <sip:2001@10.1.5.10>

Remote-Party-ID: <sip:2001@10.1.5.10>;part
HQ#y=calling;screen=yes;privacy=off

Max-Forwards: 70

Content-Length: 0



*Mar 23 14:49:29.493: //-1/xxxxxxxxxxxx/SIP/Msg/ccsipDisplayMsg:
Sent:
SIP/2.0 100 Trying

Via: SIP/2.0/TCP 10.1.5.10:5060;branch=z9hG4bK109844b1e
```



**51 (exhibit):**



```
Debug ccsip message

Via: SIP/2.0/TCP 10.1.5.10:5060;branch=z9hG4bK109844b1e

From: <sip:2001@10.1.5.10>;tag=ae2783cb-9687-4fc7-a1d0-8108b8b3679a-27173712

To: <sip:40302156001@10.1.110.1>

Date: Tue, 23 Mar 2010 14:49:29 GMT

Call-ID: 50934480-ba81d6b6-11-a05010a@10.1.5.10

CSeq: 101 INVITE

Allow-Events: telephone-event

Server: Cisco-SIPGateway/IOS-12.x

Content-Length: 0



*Mar 23 14:49:29.493: //-1/xxxxxxxxxxxx/SIP/Msg/ccsipDisplayMsg:
Sent:
SIP/2.0 404 Not Found

Via: SIP/2.0/TCP 10.1.5.10:5060;branch=z9hG4bK109844b1e

From: <sip:2001@10.1.5.10>;tag=ae2783cb-9687-4fc7-a1d0-8108b8b3679a-27173712

To: <sip:40302156001@10.1.110.1>;tag=969E5B4-F42
```

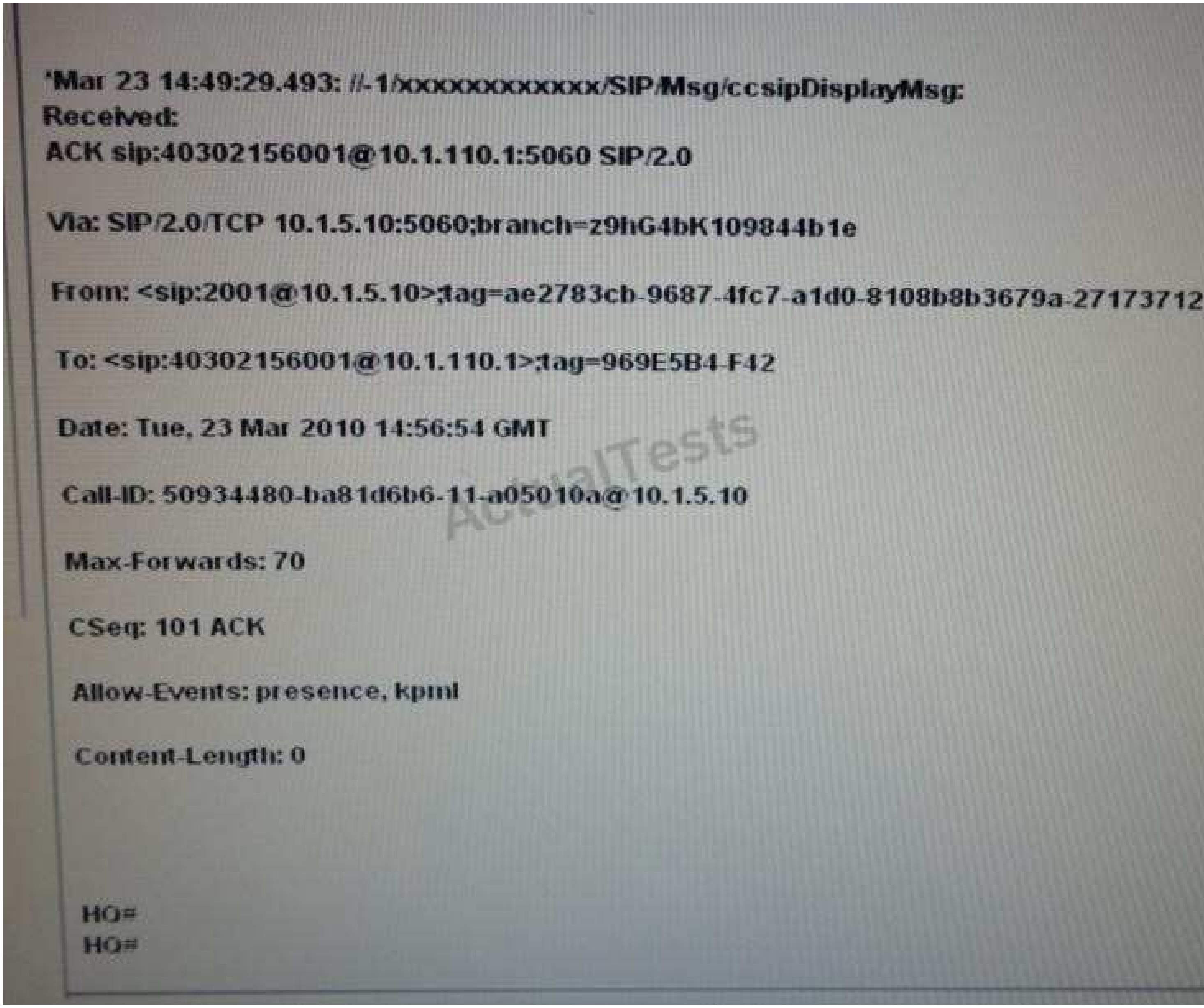


**52 (exhibit):**

```
Call-ID: 50934480-ba81d6b6-11-a05010a@10.1.5.10

CSeq: 101 INVITE

Allow-Events: telephone-event

Server: Cisco-SIPGateway/IOS-12.x

Reason: Q.850;cause=3

Content-Length: 0

*Mar 23 14:49:29.493: //-1/xxxxxxxxxxxx/SIP/Msg/ccsipDisplayMsg:
Received:
ACK sip:40302156001@10.1.110.1:5060 SIP/2.0

Via: SIP/2.0/TCP 10.1.5.10:5060;branch=z9hG4bK109844b1e

From: <sip:2001@10.1.5.10>;tag=ae2783cb-9687-4fc7-a1d0-8108b8b3679a-27173712

To: <sip:40302156001@10.1.110.1>;tag=969E5B4-F42

Date: Tue, 23 Mar 2010 14:56:54 GMT

Call-ID: 50934480-ba81d6b6-11-a05010a@10.1.5.10

Max-Forwards: 70
```



**53 (exhibit):**



```
*Mar 23 14:49:29.493: //-1/xxxxxxxxxxxx/SIP/Msg/ccsipDisplayMsg:
Received:
ACK sip:40302156001@10.1.110.1:5060 SIP/2.0

Via: SIP/2.0/TCP 10.1.5.10:5060;branch=z9hG4bK109844b1e

From: <sip:2001@10.1.5.10>;tag=ae2783cb-9687-4fc7-a1d0-8108b8b3679a-27173712

To: <sip:40302156001@10.1.110.1>;tag=969E5B4-F42

Date: Tue, 23 Mar 2010 14:56:54 GMT

Call-ID: 50934480-ba81d6b6-11-a05010a@10.1.5.10

Max-Forwards: 70

CSeq: 101 ACK

Allow-Events: presence, kpml

Content-Length: 0



HQ#
HQ#
```



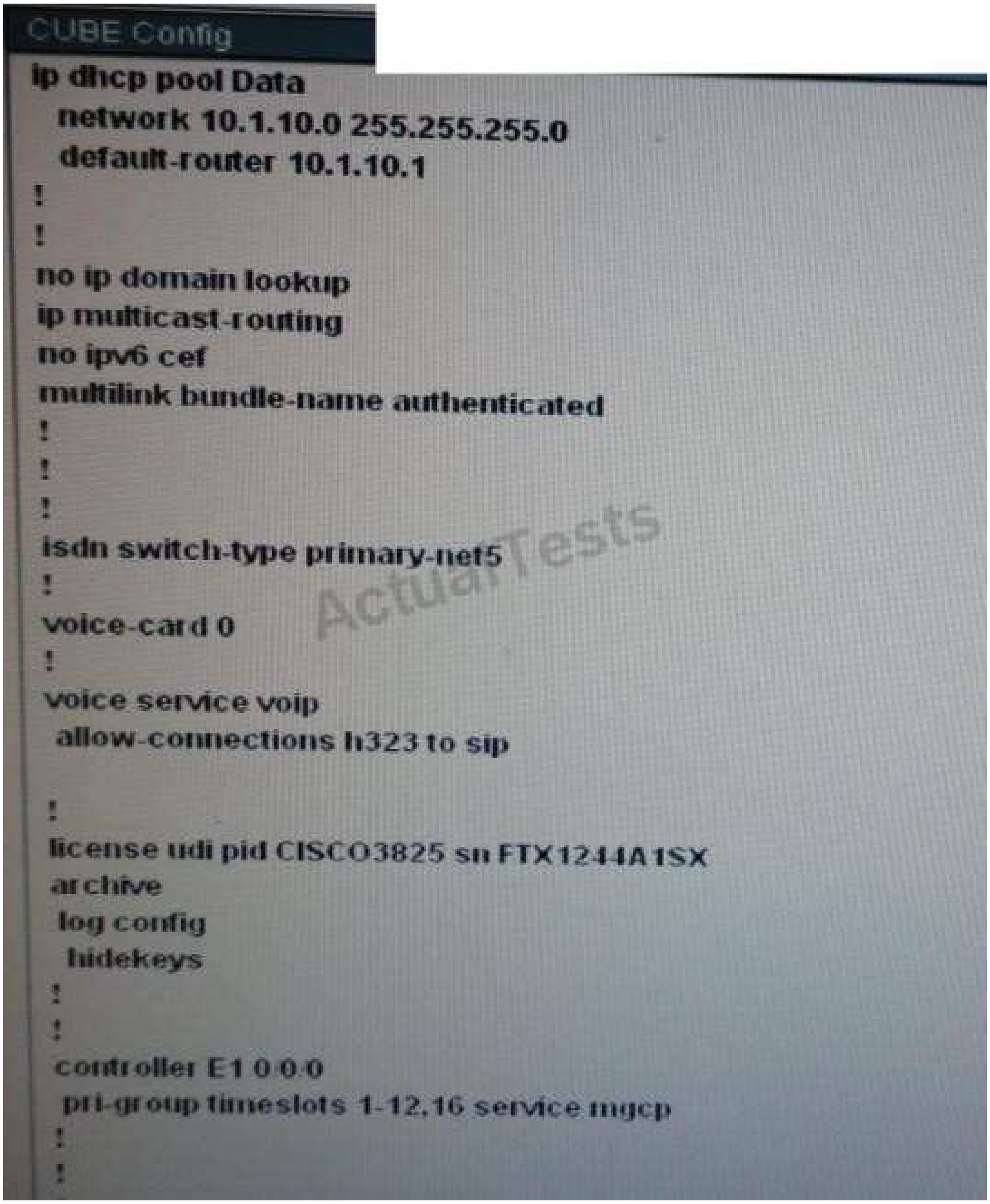
**54 (exhibit):**

CUBE Config

```
Building configuration...


Current configuration : 4713 bytes
!
! Last configuration change at 14:47:51 UTC Tue Mar 23 2010
!
version 15.0
service timestamps debug datetime msec
service timestamps log datetime msec
no service password-encryption
!
hostname HQ
!
boot-start-marker
boot system flash c3825-ipvoice_ivs-mz.150-1.XA1.bin
boot-end-marker
!
card type e1 0 0
enable password cisco123
!
no aaa new-model
network-clock-participate wic 0
!
ip source-route
ip cef
!
!
ip dhcp excluded-address 10.1.10.1 10.1.10.9
```



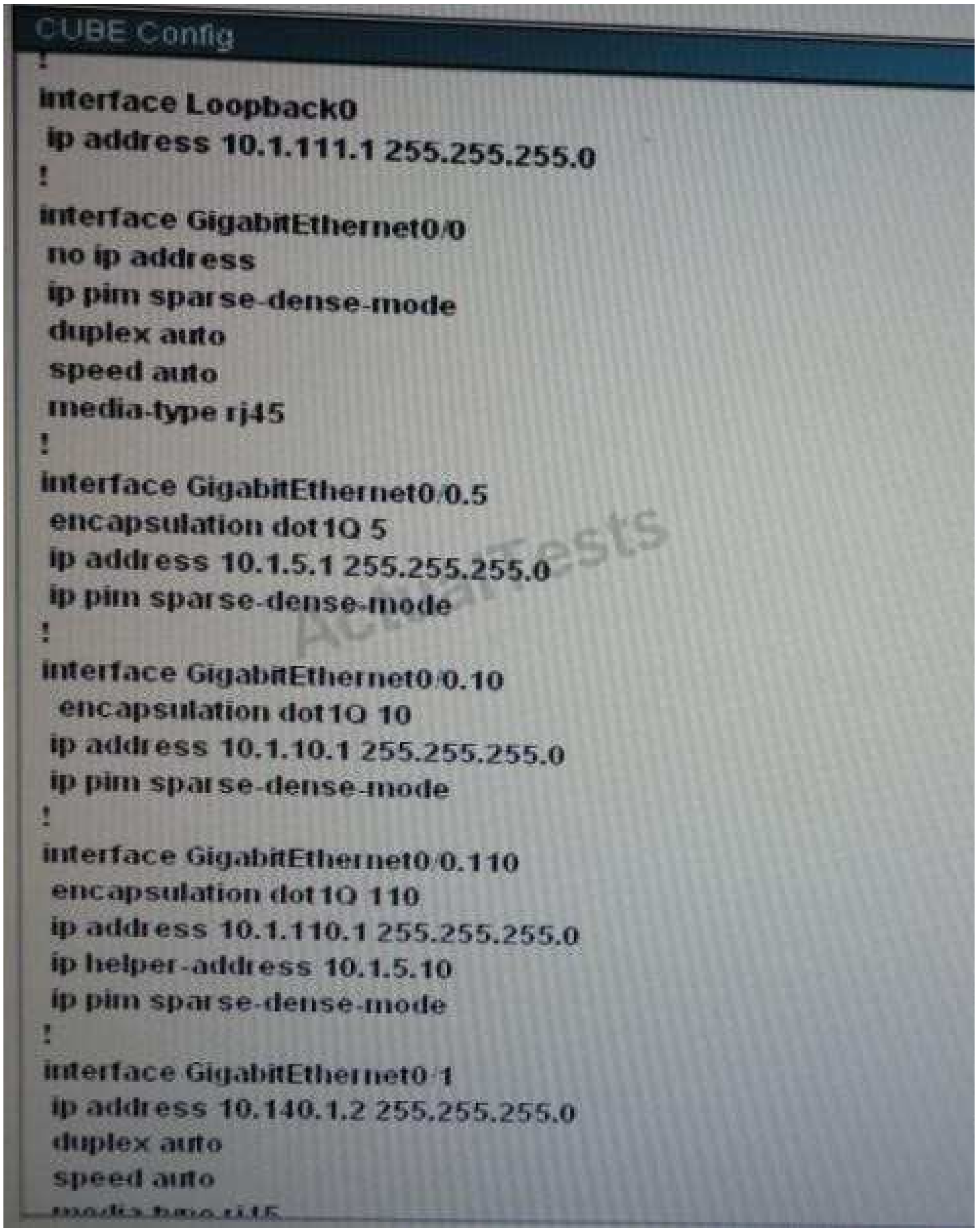
**55 (exhibit):**



CUBE Config

```
ip dhcp pool Data
 network 10.1.10.0 255.255.255.0
 default-router 10.1.10.1
!
!
no ip domain lookup
ip multicast-routing
no ipv6 cef
multilink bundle-name authenticated
!
!
!
isdn switch-type primary-net5
!
voice-card 0
!
voice service voip
 allow-connections h323 to sip

!
license udi pid CISCO3825 sn FTX1244A1SX
archive
 log config
  hidekeys
!
!
controller E1 0/0/0
 pri-group timeslots 1-12,16 service mgcp
!
!
```

56 (exhibit):



```
CUBE Config
!
interface Loopback0
 ip address 10.1.111.1 255.255.255.0
!
interface GigabitEthernet0/0
 no ip address
 ip pim sparse-dense-mode
 duplex auto
 speed auto
 media-type rj45
!
interface GigabitEthernet0/0.5
 encapsulation dot1Q 5
 ip address 10.1.5.1 255.255.255.0
 ip pim sparse-dense-mode
!
interface GigabitEthernet0/0.10
 encapsulation dot1Q 10
 ip address 10.1.10.1 255.255.255.0
 ip pim sparse-dense-mode
!
interface GigabitEthernet0/0.110
 encapsulation dot1Q 110
 ip address 10.1.110.1 255.255.255.0
 ip helper-address 10.1.5.10
 ip pim sparse-dense-mode
!
interface GigabitEthernet0/1
 ip address 10.140.1.2 255.255.255.0
 duplex auto
 speed auto
 media-type rj45
```



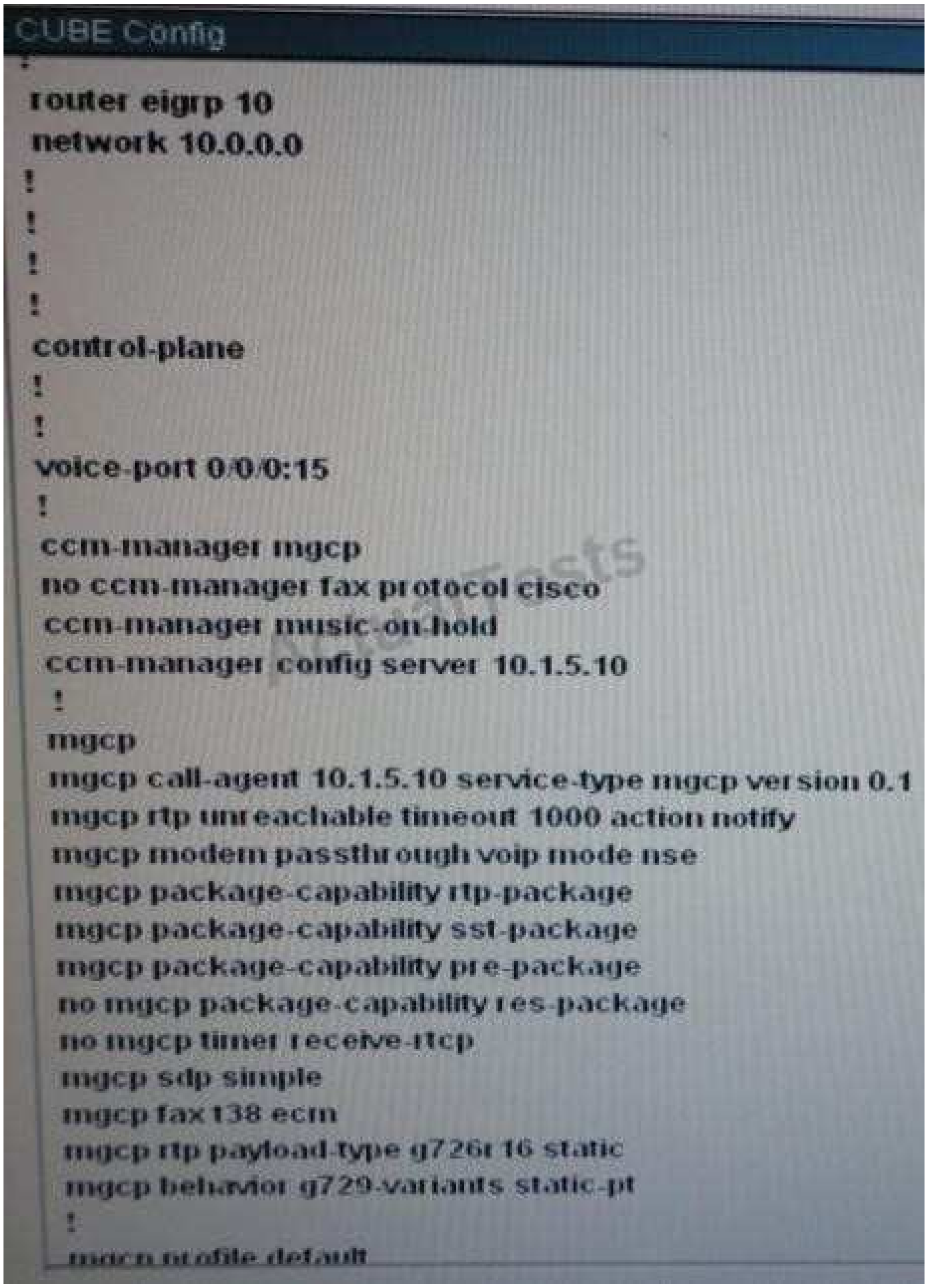
**57 (exhibit):**

CUBE Config

```
!
interface Serial0/0/0:15
 no ip address
 encapsulation hdlc
 isdn switch-type primary-net5
 isdn incoming-voice voice
 isdn bind-l3 ccm-manager
 no cdp enable
!
interface Serial0/1/0
 no ip address
 ip pim sparse-dense-mode
 encapsulation frame-relay IETF
!
interface Serial0/1/0.101 point-to-point
 ip address 10.12.1.1 255.255.255.0
 ip pim sparse-dense-mode
 snmp trap link-status
 frame-relay interface-dlci 101
!
interface Serial0/1/0.102 point-to-point
 ip address 10.13.1.1 255.255.255.0
 snmp trap link-status
 frame-relay interface-dlci 102
!
interface Serial0/1/1
 no ip address
 shutdown
 clock rate 2000000
!
!
```



**58 (exhibit):**



CUBE Config

```
!
 router eigrp 10
 network 10.0.0.0
!
!
!
!
 control-plane
!
!
 voice-port 0/0/0:15
 !
 ccm-manager mgcp
 no ccm-manager fax protocol cisco
 ccm-manager music-on-hold
 ccm-manager config server 10.1.5.10
 !
 mgcp
 mgcp call-agent 10.1.5.10 service-type mgcp version 0.1
 mgcp rtp unreachable timeout 1000 action notify
 mgcp modem passthrough voip mode nse
 mgcp package-capability rtp-package
 mgcp package-capability sst-package
 mgcp package-capability pre-package
 no mgcp package-capability res-package
 no mgcp timer receive-rtcp
 mgcp sdp simple
 mgcp fax t38 ecm
 mgcp rtp payload-type g726r16 static
 mgcp behavior g729-variants static-pt
 !
 mgcp profile default
```

**59 (exhibit):**

```
CUBE Config
!
dial-peer voice 1111 voip
 session protocol sipv2
 incoming called-number .
!
dial-peer voice 222 voip
 destination-pattern 40.........
 session target ipv4:10.12.1.2
!
!
gateway
 timer receive-rtp 1200
!
!
gatekeeper
 shutdown
!
!
line con 0
 exec-timeout 0 0
 logging synchronous
line aux 0
line vty 0 4
 exec-timeout 0 0
 password cisco123
 login
!
scheduler allocate 20000 1000
end

HQ#
```

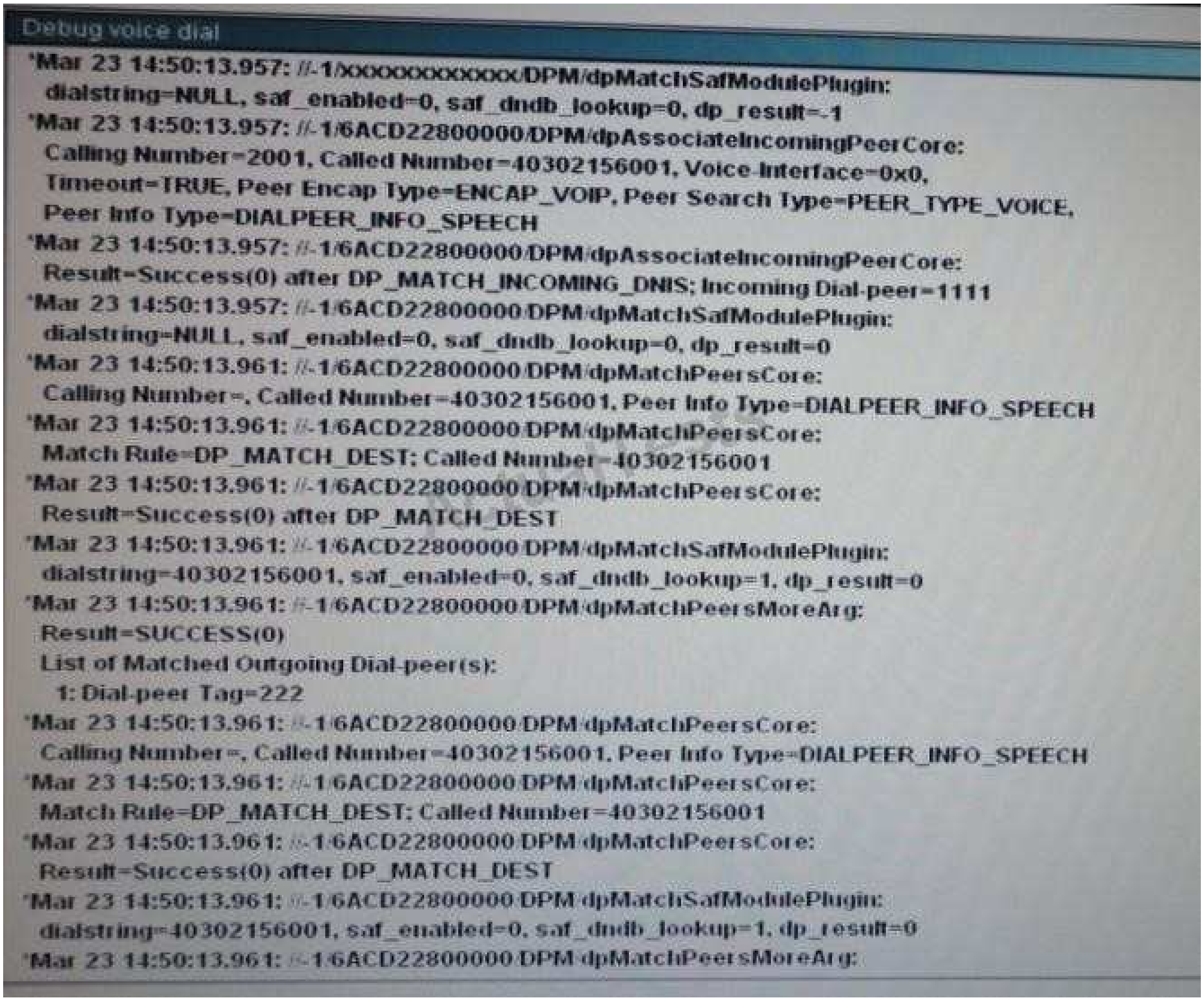


**60 (exhibit):**

Debug voice dial

```
HQ#debug voice dial
voip dialpeer default debugging is on
HQ#
*Mar 23 14:50:13.953: //-1/xxxxxxxxxxxx/DPM/dpMatchPeersCore:
  Calling Number=40302156001, Called Number=40302156001, Peer Info Type=DIALPEER_INFO_SPEECH
*Mar 23 14:50:13.953: //-1/xxxxxxxxxxxx/DPM/dpMatchPeersCore:
  Match Rule=DP_MATCH_DEST; Called Number=40302156001
*Mar 23 14:50:13.957: //-1/xxxxxxxxxxxx/DPM/dpMatchPeersCore:
  Result=Success(0) after DP_MATCH_DEST
*Mar 23 14:50:13.957: //-1/xxxxxxxxxxxx/DPM/dpMatchSafModulePlugin:
  dialstring=40302156001, saf_enabled=1, saf_dndb_lookup=1, dp_result=0
*Mar 23 14:50:13.957: //-1/xxxxxxxxxxxx/DPM/dpMatchPeersMoreArg:
  Result=SUCCESS(0)
  List of Matched Outgoing Dial-peer(s):
    1: Dial-peer Tag=222
*Mar 23 14:50:13.957: //-1/xxxxxxxxxxxx/DPM/dpAssociateIncomingPeerCore:
  Calling Number=2001, Called Number=, Voice-Interface=0x0,
  Timeout=TRUE, Peer Encap Type=ENCAP_VOIP, Peer Search Type=PEER_TYPE_VOICE,
  Peer Info Type=DIALPEER_INFO_SPEECH
*Mar 23 14:50:13.957: //-1/xxxxxxxxxxxx/DPM/dpAssociateI
HQ#ncomingPeerCore:
  Result=NO_MATCH(-1) After All Match Rules Attempt
*Mar 23 14:50:13.957: //-1/xxxxxxxxxxxx/DPM/dpMatchSafModulePlugin:
  dialstring=NULL, saf_enabled=0, saf_dndb_lookup=0, dp_result=-1
*Mar 23 14:50:13.957: //-1/xxxxxxxxxxxx/DPM/dpAssociateIncomingPeerCore:
  Calling Number=2001, Called Number=, Voice-Interface=0x0,
  Timeout=TRUE, Peer Encap Type=ENCAP_VOIP, Peer Search Type=PEER_TYPE_VOICE,
  Peer Info Type=DIALPEER_INFO_SPEECH
*Mar 23 14:50:13.957: //-1/xxxxxxxxxxxx/DPM/dpAssociateIncomingPeerCore:
  Result=NO_MATCH(-1) After All Match Rules Attempt
```

**61 (exhibit):**



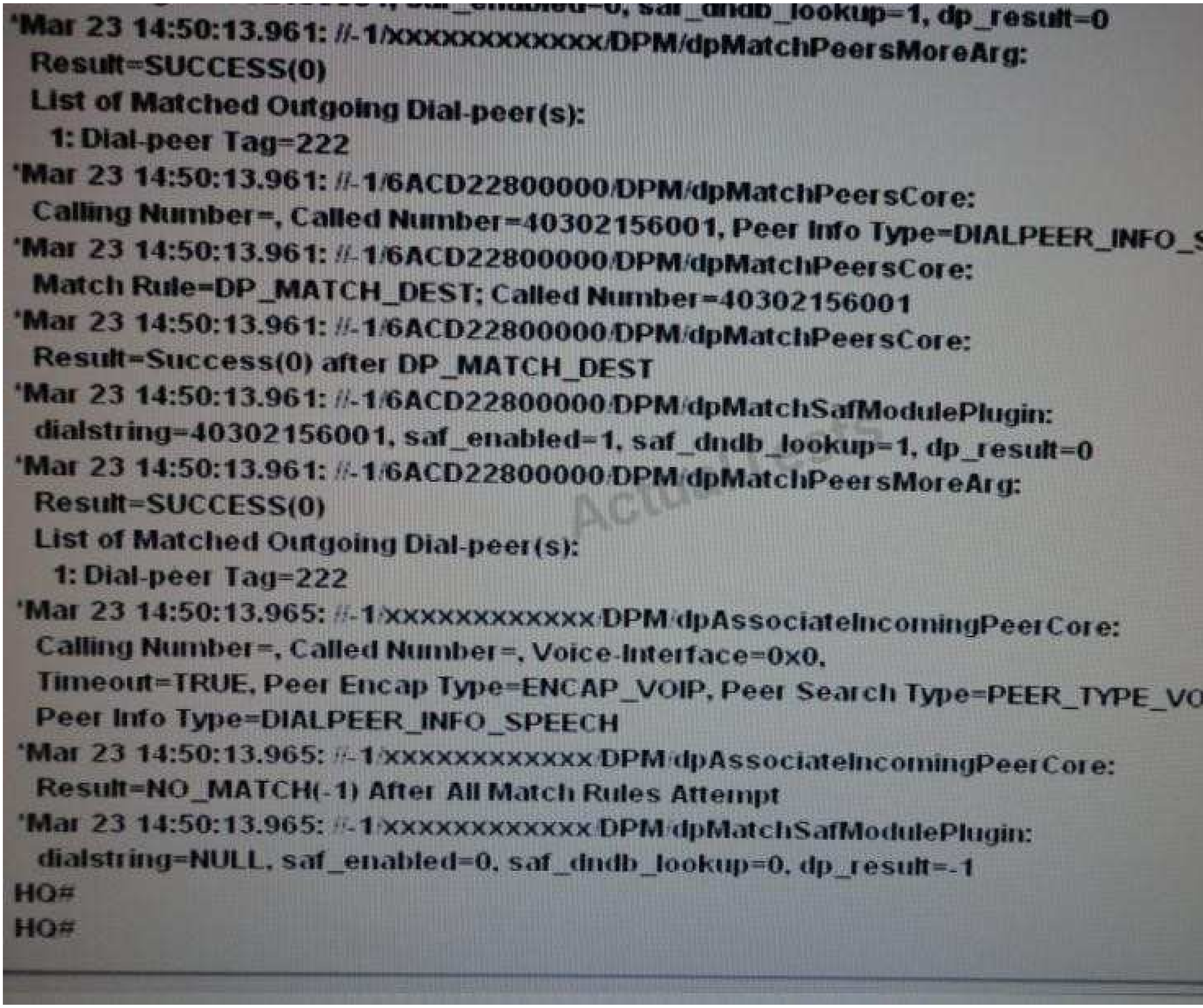
```
Debug voice dial
*Mar 23 14:50:13.957: //-1/xxxxxxxxxxxx/DPM/dpMatchSafModulePlugin:
  dialstring=NULL, saf_enabled=0, saf_dndb_lookup=0, dp_result=-1
*Mar 23 14:50:13.957: //-1/6ACD22800000/DPM/dpAssociateIncomingPeerCore:
  Calling Number=2001, Called Number=40302156001, Voice-Interface=0x0,
  Timeout=TRUE, Peer Encap Type=ENCAP_VOIP, Peer Search Type=PEER_TYPE_VOICE,
  Peer Info Type=DIALPEER_INFO_SPEECH
*Mar 23 14:50:13.957: //-1/6ACD22800000/DPM/dpAssociateIncomingPeerCore:
  Result=Success(0) after DP_MATCH_INCOMING_DNIS; Incoming Dial-peer=1111
*Mar 23 14:50:13.957: //-1/6ACD22800000/DPM/dpMatchSafModulePlugin:
  dialstring=NULL, saf_enabled=0, saf_dndb_lookup=0, dp_result=0
*Mar 23 14:50:13.961: //-1/6ACD22800000/DPM/dpMatchPeersCore:
  Calling Number=, Called Number=40302156001, Peer Info Type=DIALPEER_INFO_SPEECH
*Mar 23 14:50:13.961: //-1/6ACD22800000/DPM/dpMatchPeersCore:
  Match Rule=DP_MATCH_DEST; Called Number=40302156001
*Mar 23 14:50:13.961: //-1/6ACD22800000/DPM/dpMatchPeersCore:
  Result=Success(0) after DP_MATCH_DEST
*Mar 23 14:50:13.961: //-1/6ACD22800000/DPM/dpMatchSafModulePlugin:
  dialstring=40302156001, saf_enabled=0, saf_dndb_lookup=1, dp_result=0
*Mar 23 14:50:13.961: //-1/6ACD22800000/DPM/dpMatchPeersMoreArg:
  Result=SUCCESS(0)
  List of Matched Outgoing Dial-peer(s):
   1: Dial-peer Tag=222
*Mar 23 14:50:13.961: //-1/6ACD22800000/DPM/dpMatchPeersCore:
  Calling Number=, Called Number=40302156001, Peer Info Type=DIALPEER_INFO_SPEECH
*Mar 23 14:50:13.961: //-1/6ACD22800000/DPM/dpMatchPeersCore:
  Match Rule=DP_MATCH_DEST; Called Number=40302156001
*Mar 23 14:50:13.961: //-1/6ACD22800000/DPM/dpMatchPeersCore:
  Result=Success(0) after DP_MATCH_DEST
*Mar 23 14:50:13.961: //-1/6ACD22800000/DPM/dpMatchSafModulePlugin:
  dialstring=40302156001, saf_enabled=0, saf_dndb_lookup=1, dp_result=0
*Mar 23 14:50:13.961: //-1/6ACD22800000/DPM/dpMatchPeersMoreArg:
```

**62 (exhibit):**

```
Debug voice dial
  Result=SUCCESS(0)
  List of Matched Outgoing Dial-peer(s):
   1: Dial-peer Tag=222
*Mar 23 14:50:13.961: //-1/xxxxxxxxxxxx/DPM/dpMatchPeersCore:
  Calling Number=40302156001, Called Number=40302156001, Peer Info Type=DIALPEER_INFO_S
*Mar 23 14:50:13.961: //-1/xxxxxxxxxxxx/DPM/dpMatchPeersCore:
  Match Rule=DP_MATCH_DEST; Called Number=40302156001
*Mar 23 14:50:13.961: //-1/xxxxxxxxxxxx/DPM/dpMatchPeersCore:
  Result=Success(0) after DP_MATCH_DEST
*Mar 23 14:50:13.961: //-1/xxxxxxxxxxxx/DPM/dpMatchSafModulePlugin:
  dialstring=40302156001, saf_enabled=0, saf_dndb_lookup=1, dp_result=0
*Mar 23 14:50:13.961: //-1/xxxxxxxxxxxx/DPM/dpMatchPeersMoreArg:
  Result=SUCCESS(0)
  List of Matched Outgoing Dial-peer(s):
   1: Dial-peer Tag=222
*Mar 23 14:50:13.961: //-1/xxxxxxxxxxxx/DPM/dpAssociateIncomingPeerCore:
  Calling Number=40302156001, Called Number=, Voice-Interface=0x0,
  Timeout=TRUE, Peer Encap Type=ENCAP_VOIP, Peer Search Type=PEER_TYPE_VOICE,
  Peer Info Type=DIALPEER_INFO_SPEECH
*Mar 23 14:50:13.961: //-1/xxxxxxxxxxxx/DPM/dpAssociateIncomingPeerCore:
  Result=Success(0) after DP_MATCH_ORIGINATE; Incoming Dial-peer=222
*Mar 23 14:50:13.961: //-1/xxxxxxxxxxxx/DPM/dpMatchSafModulePlugin:
  dialstring=NULL, saf_enabled=0, saf_dndb_lookup=0, dp_result=0
*Mar 23 14:50:13.961: //-1/xxxxxxxxxxxx/DPM/dpAssociateIncomingPeerCore:
  Calling Number=40302156001, Called Number=, Voice-Interface=0x0,
  Timeout=TRUE, Peer Encap Type=ENCAP_VOIP, Peer Search Type=PEER_TYPE_VOICE,
  Peer Info Type=DIALPEER_INFO_SPEECH
*Mar 23 14:50:13.961: //-1/xxxxxxxxxxxx/DPM/dpAssociateIncomingPeerCore:
  Result=Success(0) after DP_MATCH_ORIGINATE; Incoming Dial-peer=222
*Mar 23 14:50:13.961: //-1/xxxxxxxxxxxx/DPM/dpMatchSafModulePlugin:
  dialstring=NULL, saf_enabled=0, saf_dndb_lookup=0, dp_result=0
```



**63 (exhibit):**



```
saf_dndb_lookup=1, dp_result=0
*Mar 23 14:50:13.961: //-1/xxxxxxxxxxxx/DPM/dpMatchPeersMoreArg:
  Result=SUCCESS(0)
  List of Matched Outgoing Dial-peer(s):
   1: Dial-peer Tag=222
*Mar 23 14:50:13.961: //-1/6ACD22800000/DPM/dpMatchPeersCore:
  Calling Number=, Called Number=40302156001, Peer Info Type=DIALPEER_INFO_S
*Mar 23 14:50:13.961: //-1/6ACD22800000/DPM/dpMatchPeersCore:
  Match Rule=DP_MATCH_DEST; Called Number=40302156001
*Mar 23 14:50:13.961: //-1/6ACD22800000/DPM/dpMatchPeersCore:
  Result=Success(0) after DP_MATCH_DEST
*Mar 23 14:50:13.961: //-1/6ACD22800000/DPM/dpMatchSafModulePlugin:
  dialstring=40302156001, saf_enabled=1, saf_dndb_lookup=1, dp_result=0
*Mar 23 14:50:13.961: //-1/6ACD22800000/DPM/dpMatchPeersMoreArg:
  Result=SUCCESS(0)
  List of Matched Outgoing Dial-peer(s):
   1: Dial-peer Tag=222
*Mar 23 14:50:13.965: //-1/xxxxxxxxxxxx/DPM/dpAssociateIncomingPeerCore:
  Calling Number=, Called Number=, Voice-Interface=0x0,
  Timeout=TRUE, Peer Encap Type=ENCAP_VOIP, Peer Search Type=PEER_TYPE_VO
  Peer Info Type=DIALPEER_INFO_SPEECH
*Mar 23 14:50:13.965: //-1/xxxxxxxxxxxx/DPM/dpAssociateIncomingPeerCore:
  Result=NO_MATCH(-1) After All Match Rules Attempt
*Mar 23 14:50:13.965: //-1/xxxxxxxxxxxx/DPM/dpMatchSafModulePlugin:
  dialstring=NULL, saf_enabled=0, saf_dndb_lookup=0, dp_result=-1
HQ#
HQ#
```

A. The Cisco Unified Bolder Element is configured as an MGCP gateway also so that the call is attempted via the PSTN
B. The command allow-connections sip to h323 is missing
C. SIP error 404 means that a codec mismatch occurred Cisco Unified Communications Manager is sending the call as an early offer with G.711 codec.
D. The Cisco Unified Communications Manager is rnisconfigured. The SIP invite should be sent to the ITSP at 10.1.2.1.2. The debug ccsip message shows the SIP invite being sent to 10.12.1.2.

**Correct Answer:** B
**Section: (none)**
**Explanation**

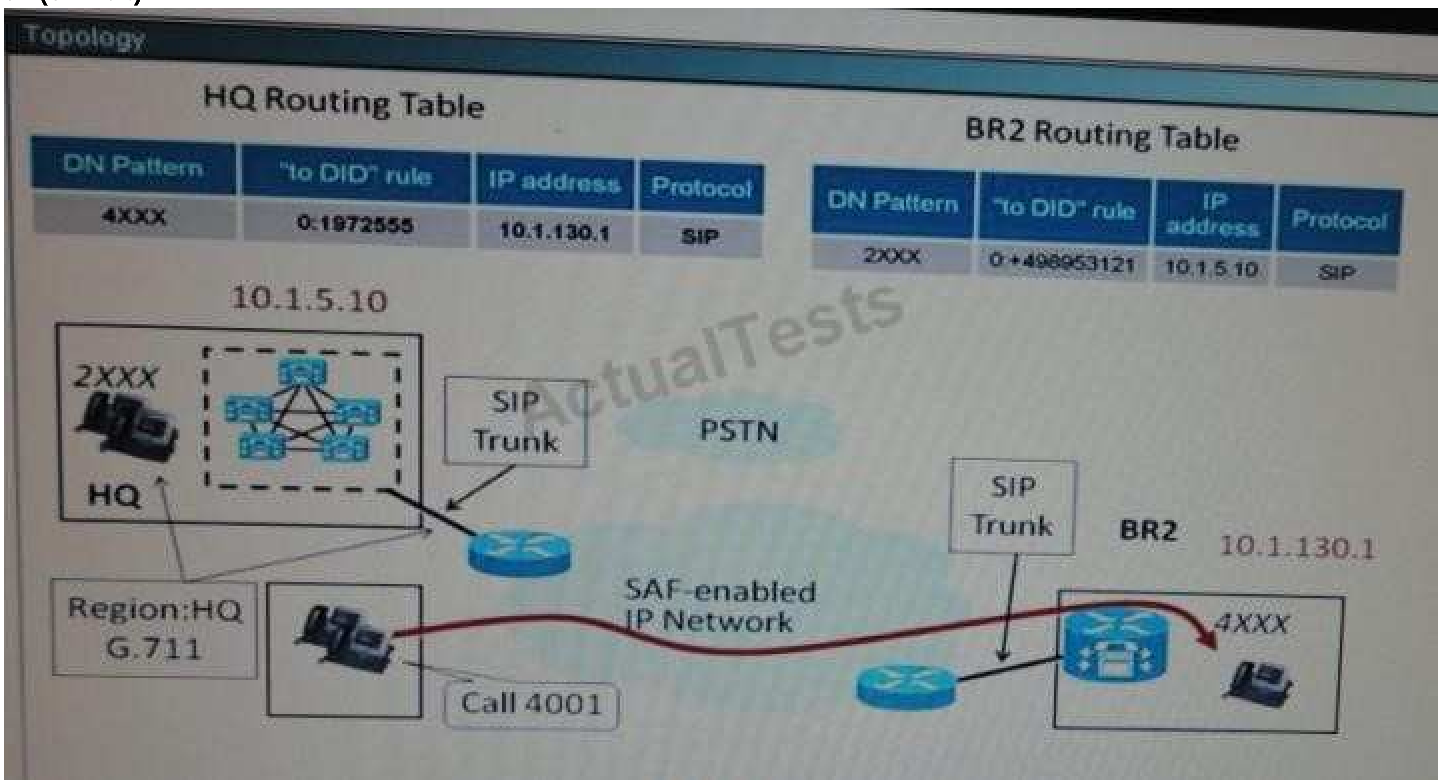
**Explanation/Reference:**
**Explanation**- As we can see in logs, the call is between two different signaling devices i.e. SIP and H.323 hence The command allow-connections sip to h323 is mandatory. **Link**-http://www.cisco.com/en/US/docs/ios/voice/cube/configuration/guide/vb-gw-h323sip.html

**QUESTION 21**

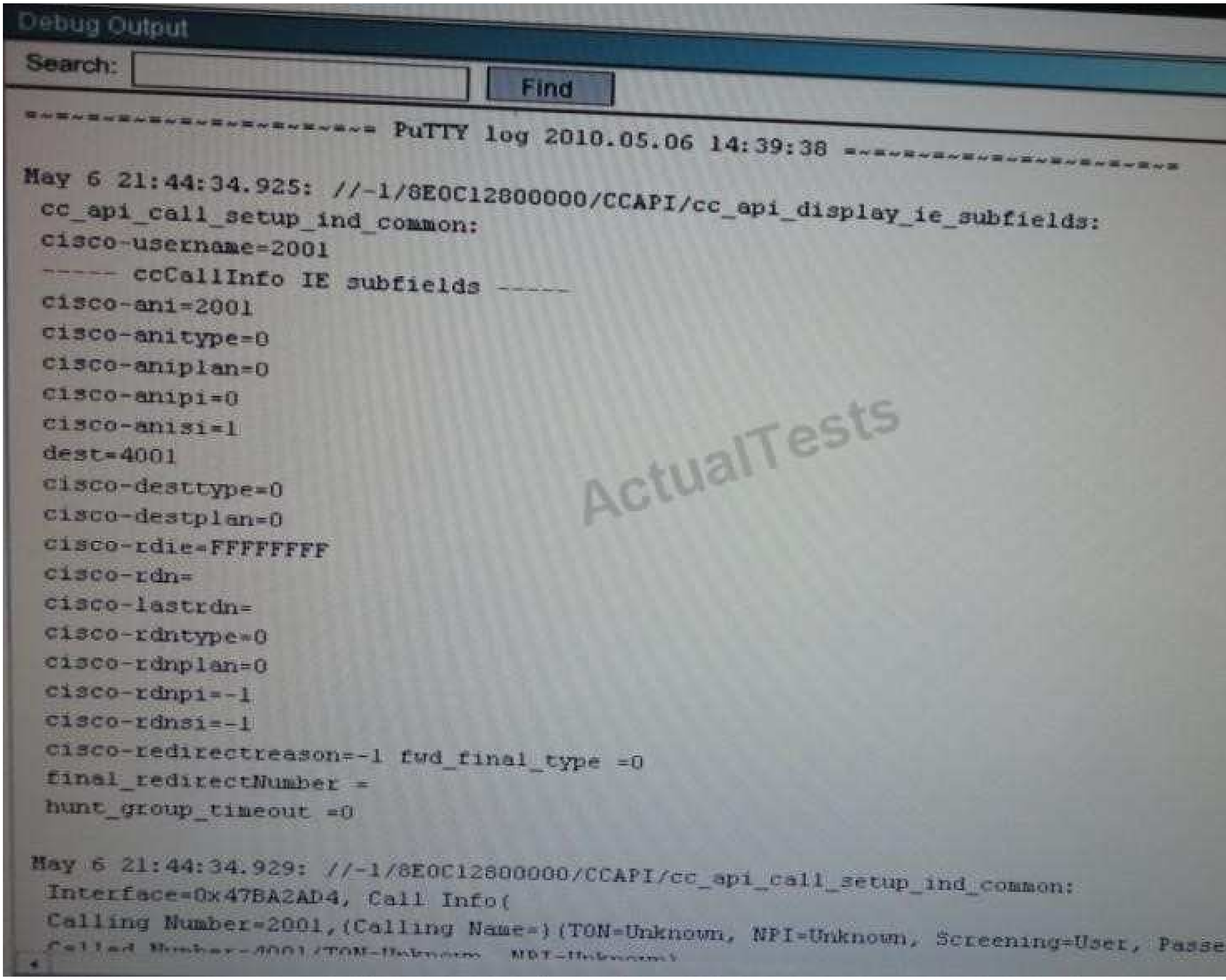
## Refer to the exhibit.

When an HQ caller places a call to extension 4001 the phone at BR2 rings. Alter the user at BR2 goes off hook, the call is disconnected. Examine the debug voice ccapi coice inout command output. What caused this issue?

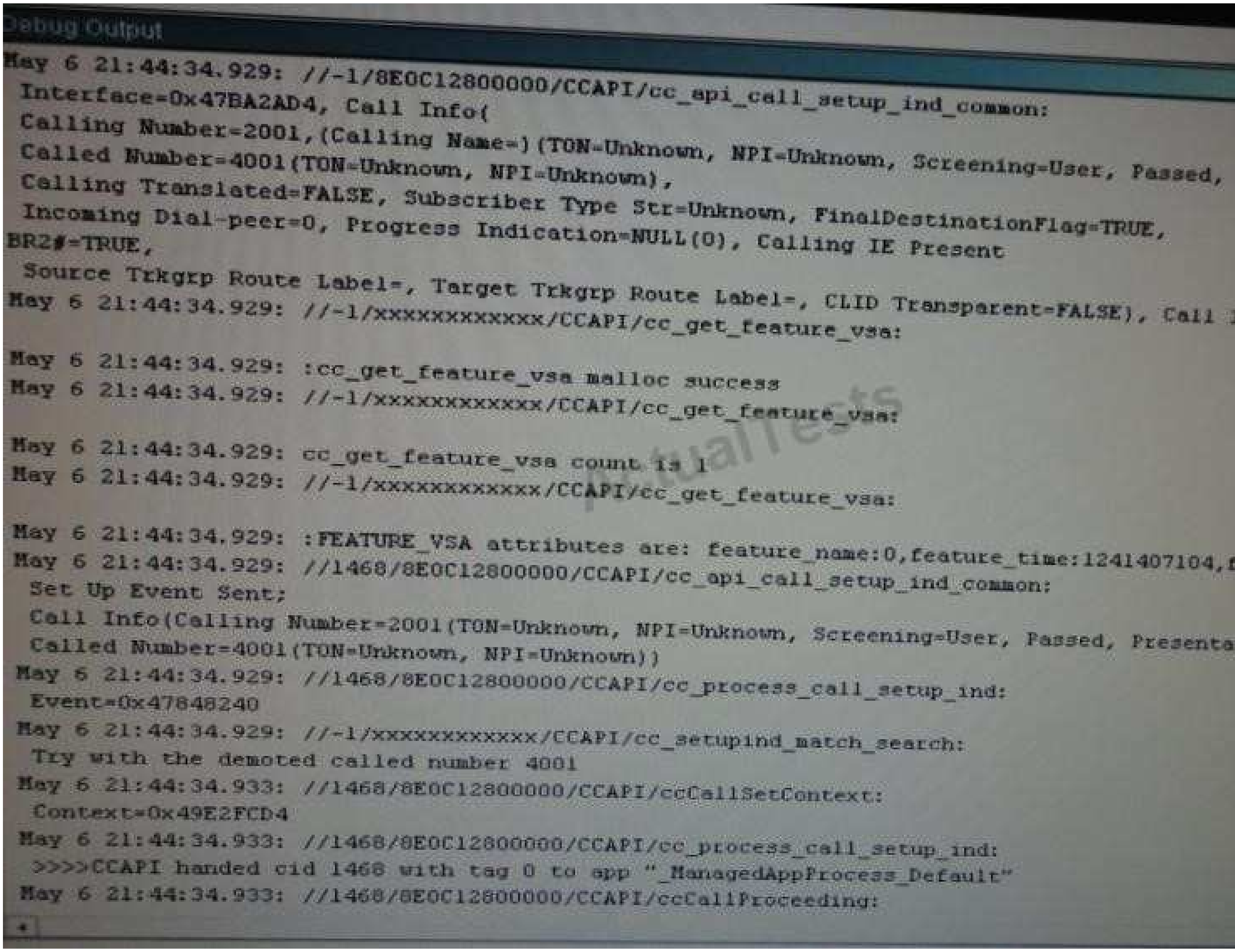
**64 (exhibit):**




**65 (exhibit):**

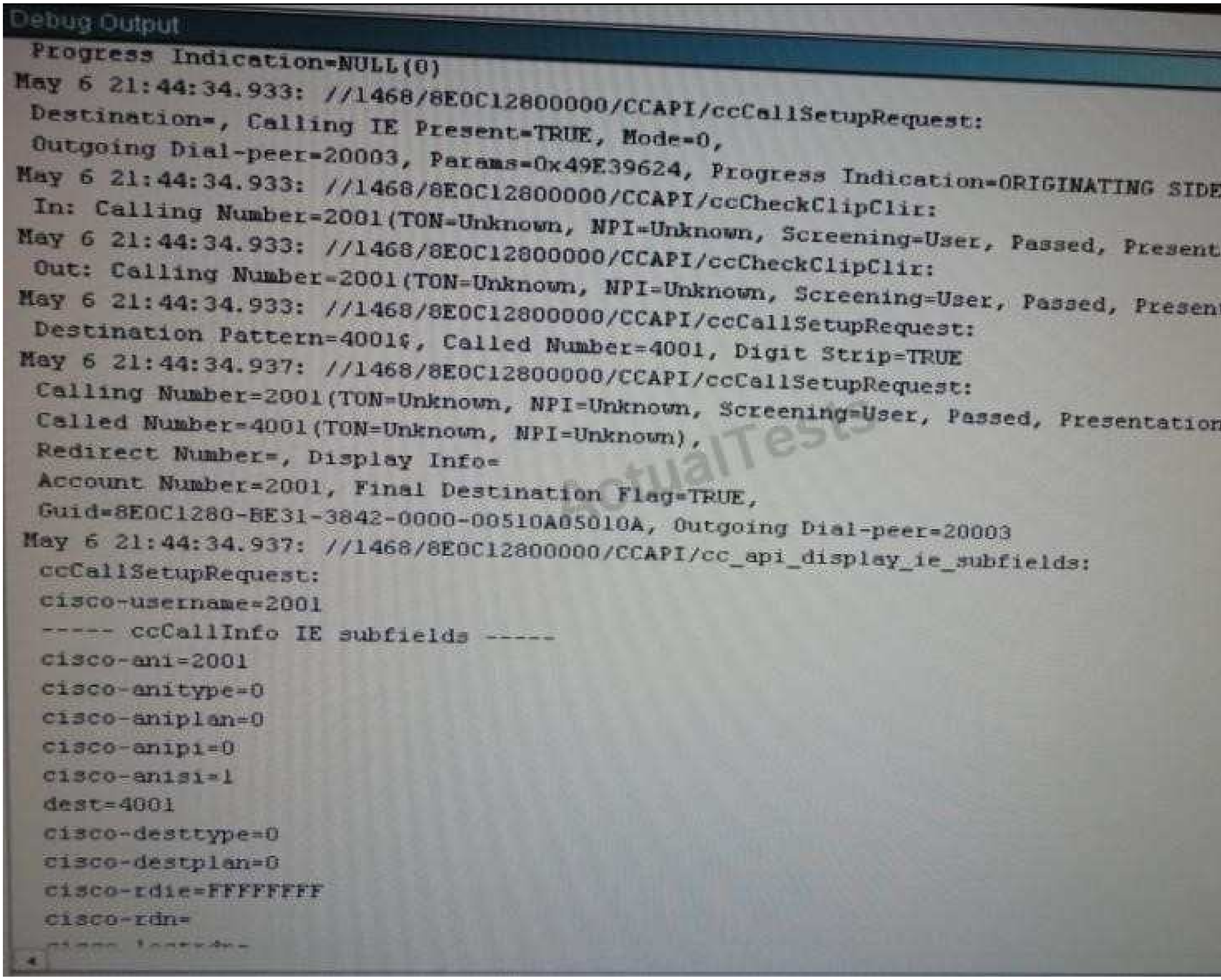

Debug Output
Search:
Find
=~=~=~=~=~=~=~=~=~=~=~= PuTTY log 2010.05.06 14:39:38 =~=~=~=~=~=~=~=~=~=~=~=
May 6 21:44:34.925: //-1/8E0C12800000/CCAPI/cc_api_display_ie_subfields:
cc_api_call_setup_ind_common:
cisco-username=2001
----- ccCallInfo IE subfields -----
cisco-ani=2001
cisco-anitype=0
cisco-aniplan=0
cisco-anipi=0
cisco-anisi=1
dest=4001
cisco-desttype=0
cisco-destplan=0
cisco-rdie=FFFFFFFF
cisco-rdn=
cisco-lastrdn=
cisco-rdntype=0
cisco-rdnplan=0
cisco-rdnpi=-1
cisco-rdnsi=-1
cisco-redirectreason=-1 fwd_final_type =0
final_redirectNumber =
hunt_group_timeout =0
May 6 21:44:34.929: //-1/8E0C12800000/CCAPI/cc_api_call_setup_ind_common:
Interface=0x47BA2AD4, Call Info(
Calling Number=2001,(Calling Name=)(TON=Unknown, NPI=Unknown, Screening=User, Passe

**66 (exhibit):**



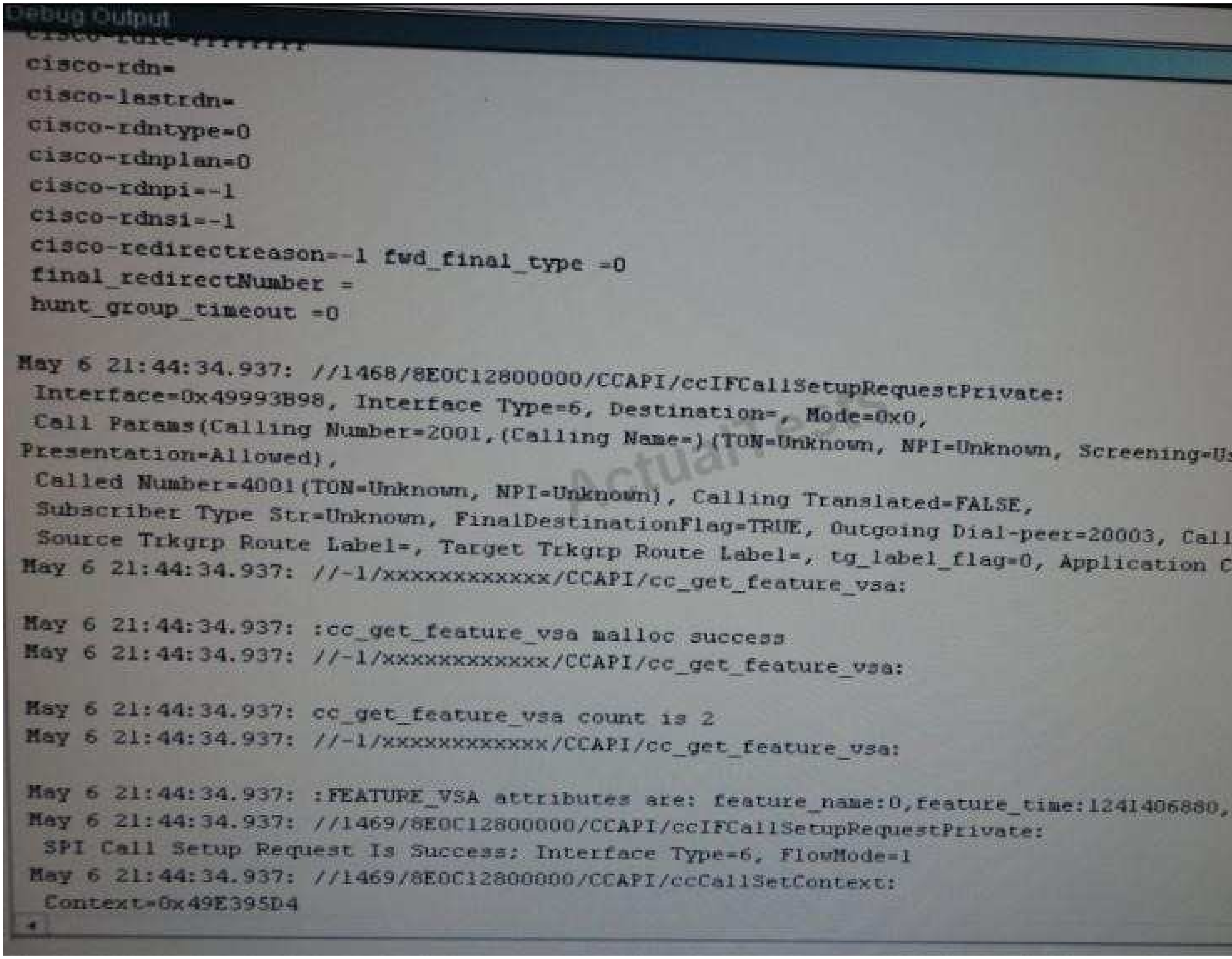
```
Debug Output
May 6 21:44:34.929: //-1/8E0C12800000/CCAPI/cc_api_call_setup_ind_common:
 Interface=0x47BA2AD4, Call Info(
 Calling Number=2001,(Calling Name=)(TON=Unknown, NPI=Unknown, Screening=User, Passed,
 Called Number=4001(TON=Unknown, NPI=Unknown),
 Calling Translated=FALSE, Subscriber Type Str=Unknown, FinalDestinationFlag=TRUE,
 Incoming Dial-peer=0, Progress Indication=NULL(0), Calling IE Present
BR2#=TRUE,
 Source Trkgrp Route Label=, Target Trkgrp Route Label=, CLID Transparent=FALSE), Call
May 6 21:44:34.929: //-1/xxxxxxxxxxxx/CCAPI/cc_get_feature_vsa:

May 6 21:44:34.929: :cc_get_feature_vsa malloc success
May 6 21:44:34.929: //-1/xxxxxxxxxxxx/CCAPI/cc_get_feature_vsa:

May 6 21:44:34.929: cc_get_feature_vsa count is 1
May 6 21:44:34.929: //-1/xxxxxxxxxxxx/CCAPI/cc_get_feature_vsa:

May 6 21:44:34.929: :FEATURE_VSA attributes are: feature_name:0,feature_time:1241407104,f
May 6 21:44:34.929: //1468/8E0C12800000/CCAPI/cc_api_call_setup_ind_common:
 Set Up Event Sent;
 Call Info(Calling Number=2001(TON=Unknown, NPI=Unknown, Screening=User, Passed, Presenta
 Called Number=4001(TON=Unknown, NPI=Unknown))
May 6 21:44:34.929: //1468/8E0C12800000/CCAPI/cc_process_call_setup_ind:
 Event=0x47848240
May 6 21:44:34.929: //-1/xxxxxxxxxxxx/CCAPI/cc_setupind_match_search:
 Try with the demoted called number 4001
May 6 21:44:34.933: //1468/8E0C12800000/CCAPI/ccCallSetContext:
 Context=0x49E2FCD4
May 6 21:44:34.933: //1468/8E0C12800000/CCAPI/cc_process_call_setup_ind:
 >>>>CCAPI handed cid 1468 with tag 0 to app "_ManagedAppProcess_Default"
May 6 21:44:34.933: //1468/8E0C12800000/CCAPI/ccCallProceeding:
```

**67 (exhibit):**



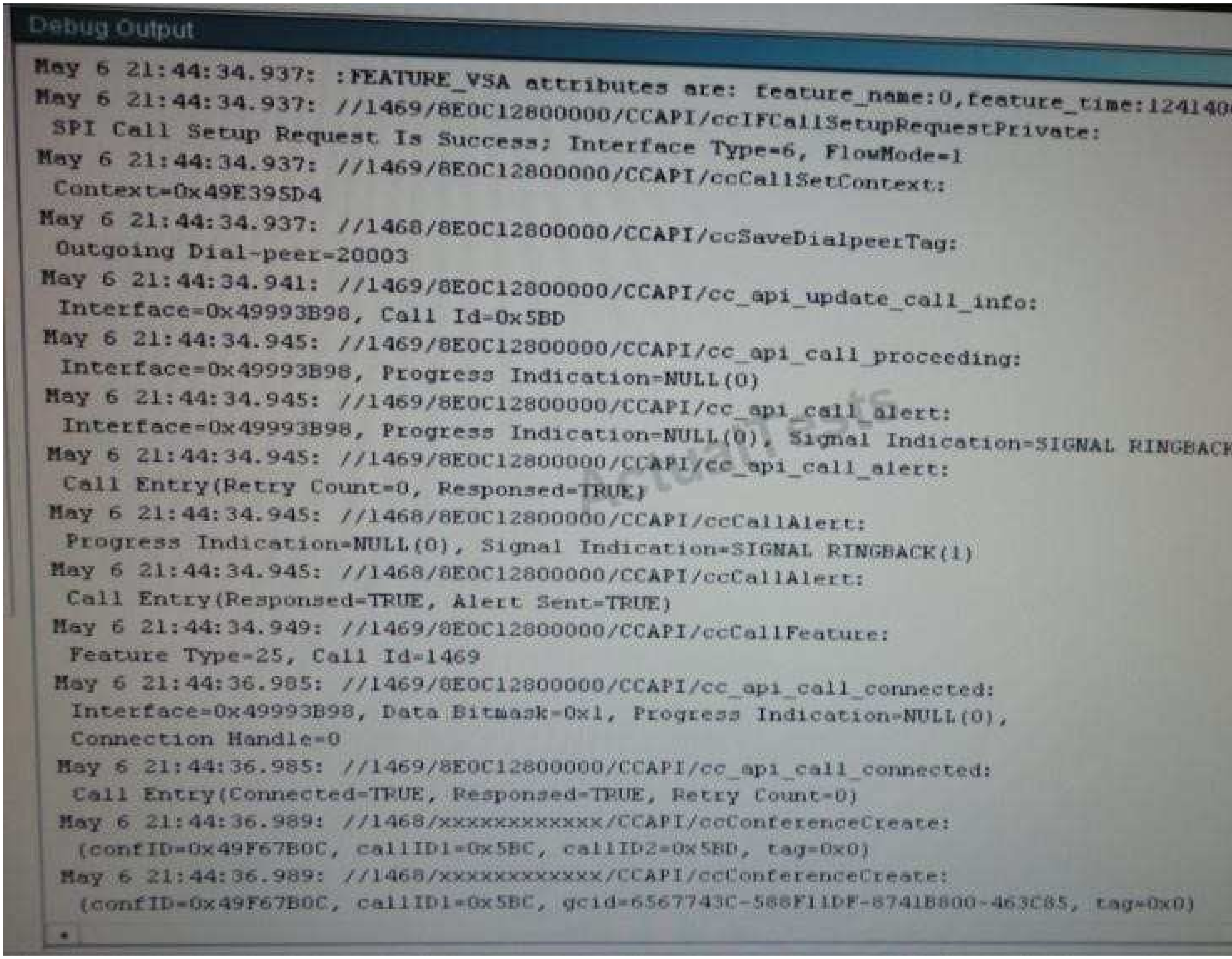
```
Debug Output
 Progress Indication=NULL(0)
May 6 21:44:34.933: //1468/8E0C12800000/CCAPI/ccCallSetupRequest:
 Destination=, Calling IE Present=TRUE, Mode=0,
 Outgoing Dial-peer=20003, Params=0x49E39624, Progress Indication=ORIGINATING SIDE
May 6 21:44:34.933: //1468/8E0C12800000/CCAPI/ccCheckClipClir:
 In: Calling Number=2001(TON=Unknown, NPI=Unknown, Screening=User, Passed, Present
May 6 21:44:34.933: //1468/8E0C12800000/CCAPI/ccCheckClipClir:
 Out: Calling Number=2001(TON=Unknown, NPI=Unknown, Screening=User, Passed, Presen
May 6 21:44:34.933: //1468/8E0C12800000/CCAPI/ccCallSetupRequest:
 Destination Pattern=4001$, Called Number=4001, Digit Strip=TRUE
May 6 21:44:34.937: //1468/8E0C12800000/CCAPI/ccCallSetupRequest:
 Calling Number=2001(TON=Unknown, NPI=Unknown, Screening=User, Passed, Presentation
 Called Number=4001(TON=Unknown, NPI=Unknown),
 Redirect Number=, Display Info=
 Account Number=2001, Final Destination Flag=TRUE,
 Guid=8E0C1280-BE31-3842-0000-00510A05010A, Outgoing Dial-peer=20003
May 6 21:44:34.937: //1468/8E0C12800000/CCAPI/cc_api_display_ie_subfields:
 ccCallSetupRequest:
 cisco-username=2001
 ----- ccCallInfo IE subfields -----
 cisco-ani=2001
 cisco-anitype=0
 cisco-aniplan=0
 cisco-anipi=0
 cisco-anisi=1
 dest=4001
 cisco-desttype=0
 cisco-destplan=0
 cisco-rdie=FFFFFFFF
 cisco-rdn=
```

**68 (exhibit):**



```
Debug Output
cisco-rdn=
cisco-lastrdn=
cisco-rdntype=0
cisco-rdnplan=0
cisco-rdnpi=-1
cisco-rdnsi=-1
cisco-redirectreason=-1 fwd_final_type =0
final_redirectNumber =
hunt_group_timeout =0

May 6 21:44:34.937: //1468/8E0C12800000/CCAPI/ccIFCallSetupRequestPrivate:
 Interface=0x49993B98, Interface Type=6, Destination=, Mode=0x0,
 Call Params(Calling Number=2001,(Calling Name=)(TON=Unknown, NPI=Unknown, Screening=U
Presentation=Allowed),
 Called Number=4001(TON=Unknown, NPI=Unknown), Calling Translated=FALSE,
 Subscriber Type Str=Unknown, FinalDestinationFlag=TRUE, Outgoing Dial-peer=20003, Call
 Source Trkgrp Route Label=, Target Trkgrp Route Label=, tg_label_flag=0, Application C
May 6 21:44:34.937: //-1/xxxxxxxxxxxx/CCAPI/cc_get_feature_vsa:

May 6 21:44:34.937: :cc_get_feature_vsa malloc success
May 6 21:44:34.937: //-1/xxxxxxxxxxxx/CCAPI/cc_get_feature_vsa:

May 6 21:44:34.937: cc_get_feature_vsa count is 2
May 6 21:44:34.937: //-1/xxxxxxxxxxxx/CCAPI/cc_get_feature_vsa:

May 6 21:44:34.937: :FEATURE_VSA attributes are: feature_name:0,feature_time:1241406880,
May 6 21:44:34.937: //1469/8E0C12800000/CCAPI/ccIFCallSetupRequestPrivate:
 SPI Call Setup Request Is Success; Interface Type=6, FlowMode=1
May 6 21:44:34.937: //1469/8E0C12800000/CCAPI/ccCallSetContext:
 Context=0x49E395D4
```

**69 (exhibit):**



Debug Output

```
May 6 21:44:34.937: :FEATURE_VSA attributes are: feature_name:0,feature_time:124140
May 6 21:44:34.937: //1469/8E0C12800000/CCAPI/ccIFCallSetupRequestPrivate:
 SPI Call Setup Request Is Success; Interface Type=6, FlowMode=1
May 6 21:44:34.937: //1469/8E0C12800000/CCAPI/ccCallSetContext:
 Context=0x49E395D4
May 6 21:44:34.937: //1468/8E0C12800000/CCAPI/ccSaveDialpeerTag:
 Outgoing Dial-peer=20003
May 6 21:44:34.941: //1469/8E0C12800000/CCAPI/cc_api_update_call_info:
 Interface=0x49993B98, Call Id=0x5BD
May 6 21:44:34.945: //1469/8E0C12800000/CCAPI/cc_api_call_proceeding:
 Interface=0x49993B98, Progress Indication=NULL(0)
May 6 21:44:34.945: //1469/8E0C12800000/CCAPI/cc_api_call_alert:
 Interface=0x49993B98, Progress Indication=NULL(0), Signal Indication=SIGNAL RINGBACK
May 6 21:44:34.945: //1469/8E0C12800000/CCAPI/cc_api_call_alert:
 Call Entry(Retry Count=0, Responsed=TRUE)
May 6 21:44:34.945: //1468/8E0C12800000/CCAPI/ccCallAlert:
 Progress Indication=NULL(0), Signal Indication=SIGNAL RINGBACK(1)
May 6 21:44:34.945: //1468/8E0C12800000/CCAPI/ccCallAlert:
 Call Entry(Responsed=TRUE, Alert Sent=TRUE)
May 6 21:44:34.949: //1469/8E0C12800000/CCAPI/ccCallFeature:
 Feature Type=25, Call Id=1469
May 6 21:44:36.985: //1469/8E0C12800000/CCAPI/cc_api_call_connected:
 Interface=0x49993B98, Data Bitmask=0x1, Progress Indication=NULL(0),
 Connection Handle=0
May 6 21:44:36.985: //1469/8E0C12800000/CCAPI/cc_api_call_connected:
 Call Entry(Connected=TRUE, Responsed=TRUE, Retry Count=0)
May 6 21:44:36.989: //1468/xxxxxxxxxxxx/CCAPI/ccConferenceCreate:
 (confID=0x49F67B0C, callID1=0x5BC, callID2=0x5BD, tag=0x0)
May 6 21:44:36.989: //1468/xxxxxxxxxxxx/CCAPI/ccConferenceCreate:
 (confID=0x49F67B0C, callID1=0x5BC, gcid=6567743C-588F11DF-8741B800-463C85, tag=0x0)
```

**70 (exhibit):**

Debug Output

```
 (confID=0x49F67B0C, callID1=0x5BC, gcid=6567743C-588F11DF-8741B800-463C85,
May 6 21:44:36.989: //1469/xxxxxxxxxxxx/CCAPI/ccConferenceCreate:
 (confID=0x49F67B0C, callID2=0x5BD, gcid=6567743C-588F11DF-8741B800-463C85,
May 6 21:44:36.989: //1468/8E0C12800000/CCAPI/ccConferenceCreate:
 Conference Id=0x49F67B0C, Call Id1=1468, Call Id2=1469, Tag=0x0
May 6 21:44:36.989: //1468/xxxxxxxxxxxx/CCAPI/cc_api_get_xcode_stream:

May 6 21:44:36.989: cc_api_get_xcode_stream : 4546
May 6 21:44:36.989: //1468/xxxxxxxxxxxx/CCAPI/cc_api_bridge_done:
 Conference Id=0x3D, Source Interface=0x47BA2AD4, Source Call Id=1468,
 Destination Call Id=1469, Disposition=0x0, Tag=0x0
May 6 21:44:36.989: //1469/xxxxxxxxxxxx/CCAPI/cc_api_bridge_done:
 Conference Id=0x3D, Source Interface=0x49993B98, Source Call Id=1469,
 Destination Call Id=1468, Disposition=0x0, Tag=0xFFFFFFFF
May 6 21:44:36.989: //1468/8E0C12800000/CCAPI/cc_generic_bridge_done:
 Conference Id=0x3D, Source Interface=0x49993B98, Source Call Id=1469,
 Destination Call Id=1468, Disposition=0x0, Tag=0xFFFFFFFF
May 6 21:44:36.989: //1468/8E0C12800000/CCAPI/ccConferenceCreate:
 Call Entry(Conference Id=0x3D, Destination Call Id=1469)
May 6 21:44:36.989: //1469/8E0C12800000/CCAPI/ccConferenceCreate:
 Call Entry(Conference Id=0x3D, Destination Call Id=1468)
May 6 21:44:36.989: //1469/8E0C12800000/CCAPI/cc_api_caps_ind:
 Destination Interface=0x47BA2AD4, Destination Call Id=1468, Source Call Id=
 Caps(Codec=0x1, Fax Rate=0x1, Fax Version:=0, Vad=0x1,
 Modem=0x0, Codec Bytes=20, Signal Type=3)
May 6 21:44:36.989: //1469/8E0C12800000/CCAPI/cc_api_caps_ind:
 Caps(Playout Mode=1, Playout Initial=60(ms), Playout Min=40(ms),
 Playout Max=1000(ms), Fax Nom=300(ms))
May 6 21:44:36.989: //1468/8E0C12800000/CCAPI/cc_api_caps_ind:
 Destination Interface=0x49993B98, Destination Call Id=1469, Source Call Id=
```

**71 (exhibit):**

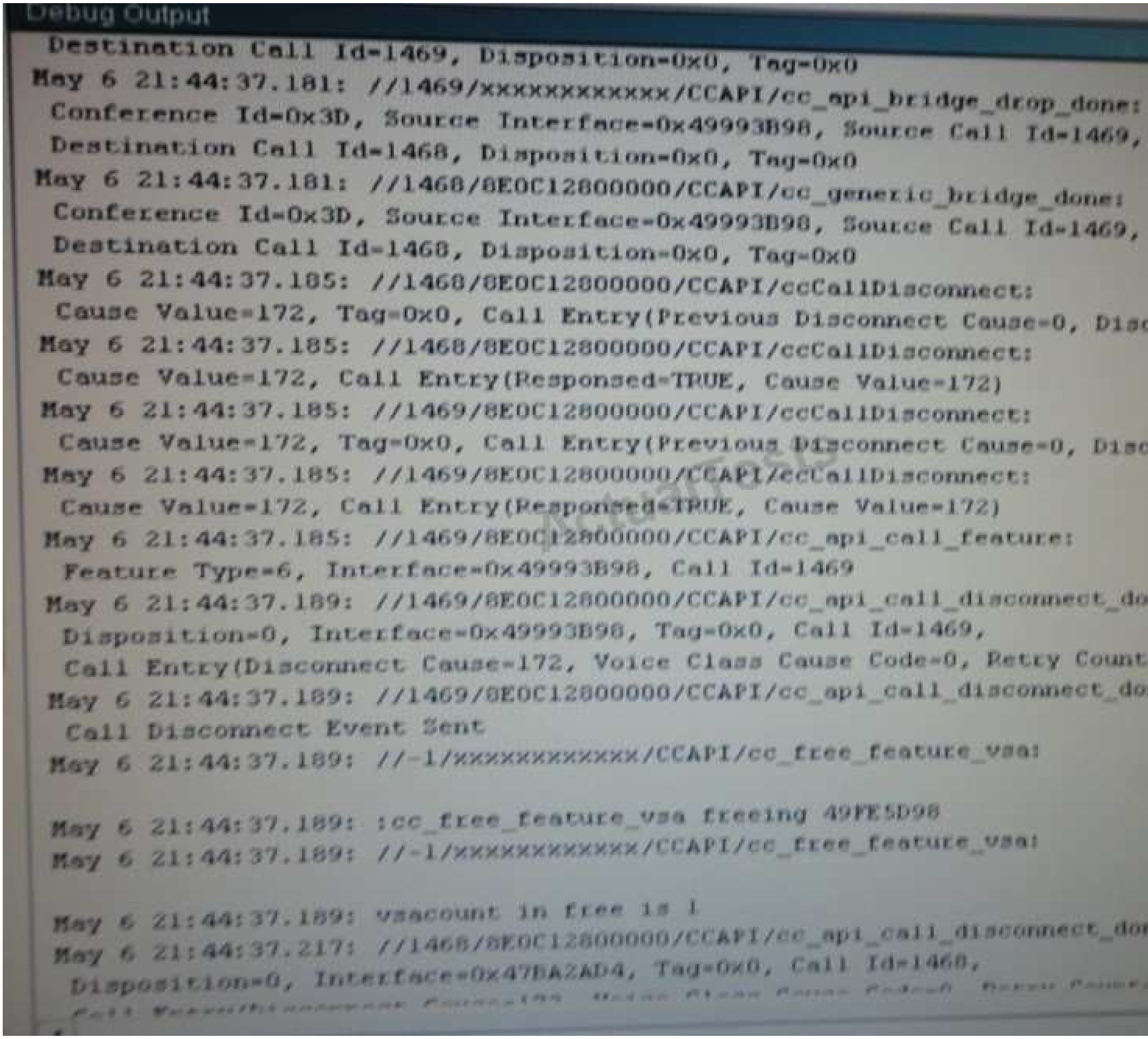
```
Debug Output
 Modem=0x0, Codec Bytes=20, Signal Type=2)
May 6 21:44:36.989: //1468/8E0C12800000/CCAPI/cc_api_caps_ind:
 Caps(Playout Mode=1, Playout Initial=60(ms), Playout Min=40(ms),
 Playout Max=1000(ms), Fax Nom=300(ms))
May 6 21:44:36.989: //1468/8E0C12800000/CCAPI/cc_api_caps_ack:
 Destination Interface=0x49993B98, Destination Call Id=1469, Source Call
 Caps(Codec=g729r8(0x4), Fax Rate=FAX_RATE_VOICE(0x2), Fax Version:=0, V
 Modem=OFF(0x0), Codec Bytes=20, Signal Type=2, Seq Num Start=283)
May 6 21:44:36.989: //1469/8E0C12800000/CCAPI/cc_api_caps_ack:
 Destination Interface=0x47BA2AD4, Destination Call Id=1468, Source Call
 Caps(Codec=g729r8(0x4), Fax Rate=FAX_RATE_VOICE(0x2), Fax Version:=0, Va
 Modem=OFF(0x0), Codec Bytes=20, Signal Type=2, Seq Num Start=283)
May 6 21:44:36.993: //1468/8E0C12800000/CCAPI/ccCallConnect:
 Progress Indication=NULL(0), Data Bitmask=0x1
May 6 21:44:36.993: //1468/8E0C12800000/CCAPI/ccCallConnect:
 Call Entry(Connected=TRUE, Responsed=TRUE)
May 6 21:44:36.993: //1469/8E0C12800000/CCAPI/ccCallFeature:
 Feature Type=25, Call Id=1469
May 6 21:44:36.993: //1468/8E0C12800000/CCAPI/cc_process_notify_bridge_do
 Conference Id=0x3D, Call Id1=1468, Call Id2=1469
May 6 21:44:36.997: //1469/8E0C12800000/CCAPI/cc_api_voice_mode_event:
 Call Id=1469
May 6 21:44:36.997: //1469/8E0C12800000/CCAPI/cc_api_voice_mode_event:
 Call Entry(Context=0x49E395D4)
May 6 21:44:37.105: //1468/8E0C12800000/CCAPI/cc_api_caps_ind:
 Destination Interface=0x49993B98, Destination Call Id=1469, Source Call I
 Caps(Codec=0x10000000, Fax Rate=0x2, Fax Version:=0, Vad=0x2,
 Modem=0x0, Codec Bytes=160, Signal Type=2)
May 6 21:44:37.109: //1468/8E0C12800000/CCAPI/cc_api_caps_ind:
 Caps(Playout Mode=1, Playout Initial=60(ms), Playout Min=40(ms),
```

**72 (exhibit):**

Debug Output

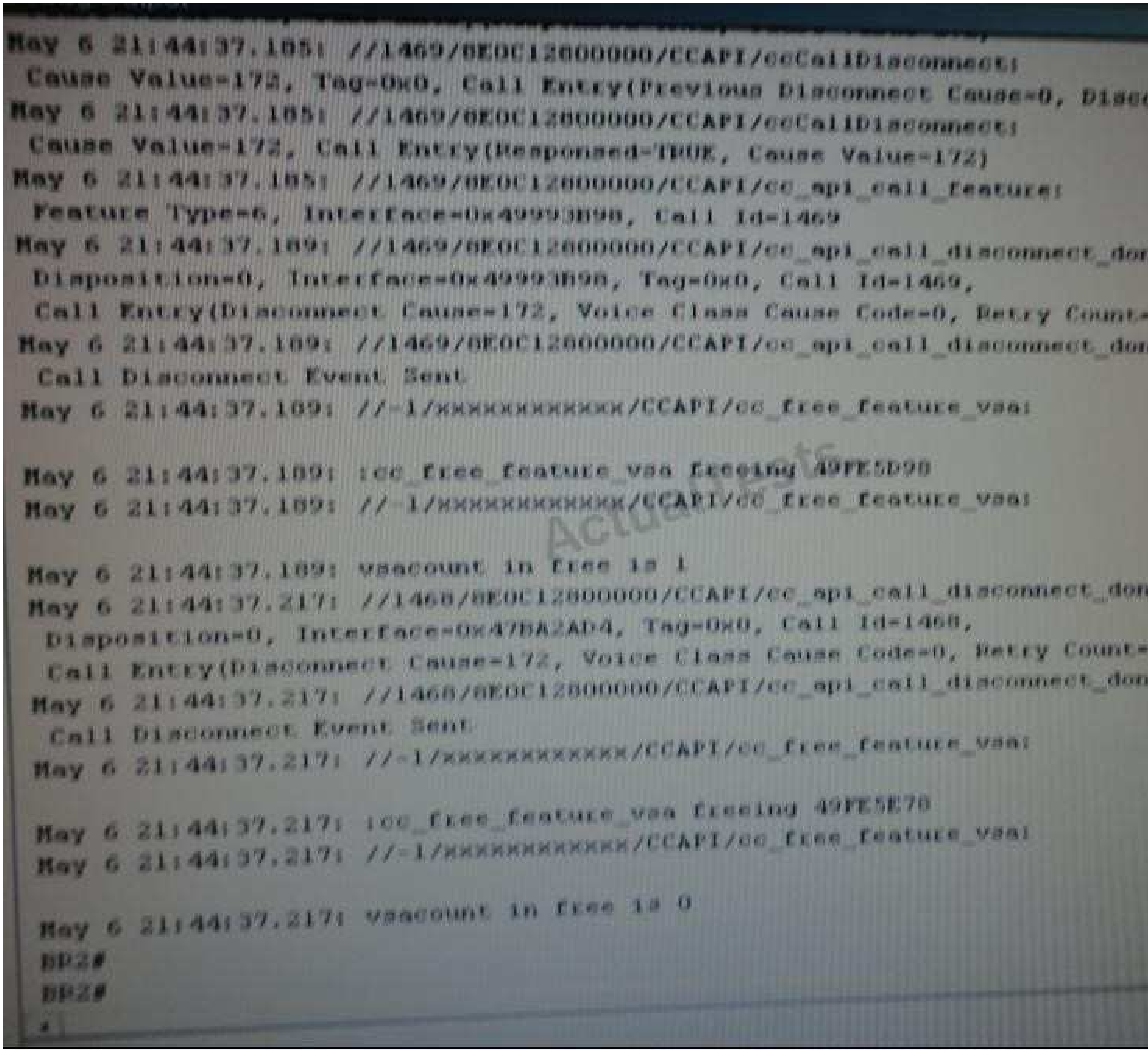
```
 Caps(Codec=0x10000000, Fax Rate=0x2, Fax Version:=0, Vad=0x2,
 Modem=0x0, Codec Bytes=160, Signal Type=2)
May 6 21:44:37.109: //1468/8E0C12800000/CCAPI/cc_api_caps_ind:
 Caps(Playout Mode=1, Playout Initial=60(ms), Playout Min=40(ms),
 Playout Max=1000(ms), Fax Nom=300(ms))
May 6 21:44:37.109: //1468/8E0C12800000/CCAPI/cc_api_caps_ack:
 Destination Interface=0x49993B98, Destination Call Id=1469, Source Cal
 Caps(Codec=g722-64(0x10000000), Fax Rate=FAX_RATE_VOICE(0x2), Fax Versi
 Modem=OFF(0x0), Codec Bytes=160, Signal Type=2, Seq Num Start=5132)
May 6 21:44:37.109: //1469/8E0C12800000/CCAPI/cc_api_caps_ack:
 Destination Interface=0x47BA2AD4, Destination Call Id=1468, Source Call
 Caps(Codec=g722-64(0x10000000), Fax Rate=FAX_RATE_VOICE(0x2), Fax Versi
 Modem=OFF(0x0), Codec Bytes=160, Signal Type=2, Seq Num Start=5132)
May 6 21:44:37.113: //1469/8E0C12800000/CCAPI/cc_api_voice_mode_event:
 Call Id=1469
May 6 21:44:37.113: //1469/8E0C12800000/CCAPI/cc_api_voice_mode_event:
 Call Entry(Context=0x49E395D4)
May 6 21:44:37.181: %DSMP-3-DSPALARM: Alarm on DSP : status=0x0 message=
May 6 21:44:37.181: //1469/8E0C12800000/CCAPI/cc_api_call_disconnected:
 Cause Value=172, Interface=0x49993B98, Call Id=1469
May 6 21:44:37.181: //1469/8E0C12800000/CCAPI/cc_api_call_disconnected:
 Call Entry(Responsed=TRUE, Cause Value=172, Retry Count=0)
May 6 21:44:37.181: //1468/8E0C12800000/CCAPI/ccConferenceDestroy:
 Conference Id=0x3D, Tag=0x0
May 6 21:44:37.181: //1468/xxxxxxxxxxxx/CCAPI/cc_api_bridge_drop_done:
 Conference Id=0x3D, Source Interface=0x47BA2AD4, Source Call Id=1468,
 Destination Call Id=1469, Disposition=0x0, Tag=0x0
May 6 21:44:37.181: //1469/xxxxxxxxxxxx/CCAPI/cc_api_bridge_drop_done:
 Conference Id=0x3D, Source Interface=0x49993B98, Source Call Id=1469,
 Destination Call Id=1468, Disposition=0x0, Tag=0x0
```

**73 (exhibit):**



```
Debug Output
 Destination Call Id=1469, Disposition=0x0, Tag=0x0
May 6 21:44:37.181: //1469/xxxxxxxxxxxx/CCAPI/cc_api_bridge_drop_done:
 Conference Id=0x3D, Source Interface=0x49993B98, Source Call Id=1469,
 Destination Call Id=1468, Disposition=0x0, Tag=0x0
May 6 21:44:37.181: //1468/8E0C12800000/CCAPI/cc_generic_bridge_done:
 Conference Id=0x3D, Source Interface=0x49993B98, Source Call Id=1469,
 Destination Call Id=1468, Disposition=0x0, Tag=0x0
May 6 21:44:37.185: //1468/8E0C12800000/CCAPI/ccCallDisconnect:
 Cause Value=172, Tag=0x0, Call Entry(Previous Disconnect Cause=0, Disc
May 6 21:44:37.185: //1468/8E0C12800000/CCAPI/ccCallDisconnect:
 Cause Value=172, Call Entry(Responsed=TRUE, Cause Value=172)
May 6 21:44:37.185: //1469/8E0C12800000/CCAPI/ccCallDisconnect:
 Cause Value=172, Tag=0x0, Call Entry(Previous Disconnect Cause=0, Disc
May 6 21:44:37.185: //1469/8E0C12800000/CCAPI/ccCallDisconnect:
 Cause Value=172, Call Entry(Responsed=TRUE, Cause Value=172)
May 6 21:44:37.185: //1469/8E0C12800000/CCAPI/cc_api_call_feature:
 Feature Type=6, Interface=0x49993B98, Call Id=1469
May 6 21:44:37.189: //1469/8E0C12800000/CCAPI/cc_api_call_disconnect_do
 Disposition=0, Interface=0x49993B98, Tag=0x0, Call Id=1469,
 Call Entry(Disconnect Cause=172, Voice Class Cause Code=0, Retry Count
May 6 21:44:37.189: //1469/8E0C12800000/CCAPI/cc_api_call_disconnect_do
 Call Disconnect Event Sent
May 6 21:44:37.189: //-1/xxxxxxxxxxxx/CCAPI/cc_free_feature_vsa:

May 6 21:44:37.189: :cc_free_feature_vsa freeing 49FE5D98
May 6 21:44:37.189: //-1/xxxxxxxxxxxx/CCAPI/cc_free_feature_vsa:

May 6 21:44:37.189: vsacount in free is 1
May 6 21:44:37.217: //1468/8E0C12800000/CCAPI/cc_api_call_disconnect_do
 Disposition=0, Interface=0x47BA2AD4, Tag=0x0, Call Id=1468,
```

**74 (exhibit):**



```
May 6 21:44:37.185: //1469/8E0C12800000/CCAPI/ccCallDisconnect:
 Cause Value=172, Tag=0x0, Call Entry(Previous Disconnect Cause=0, Disc
May 6 21:44:37.185: //1469/8E0C12800000/CCAPI/ccCallDisconnect:
 Cause Value=172, Call Entry(Responsed=TRUE, Cause Value=172)
May 6 21:44:37.185: //1469/8E0C12800000/CCAPI/cc_api_call_feature:
 Feature Type=6, Interface=0x49993B98, Call Id=1469
May 6 21:44:37.189: //1469/8E0C12800000/CCAPI/cc_api_call_disconnect_don
 Disposition=0, Interface=0x49993B98, Tag=0x0, Call Id=1469,
 Call Entry(Disconnect Cause=172, Voice Class Cause Code=0, Retry Count=
May 6 21:44:37.189: //1469/8E0C12800000/CCAPI/cc_api_call_disconnect_don
 Call Disconnect Event Sent
May 6 21:44:37.189: //-1/xxxxxxxxxxxx/CCAPI/cc_free_feature_vsa:

May 6 21:44:37.189: :cc_free_feature_vsa freeing 49FE5D98
May 6 21:44:37.189: //-1/xxxxxxxxxxxx/CCAPI/cc_free_feature_vsa:

May 6 21:44:37.189: vsacount in free is 1
May 6 21:44:37.217: //1468/8E0C12800000/CCAPI/cc_api_call_disconnect_don
 Disposition=0, Interface=0x47BA2AD4, Tag=0x0, Call Id=1468,
 Call Entry(Disconnect Cause=172, Voice Class Cause Code=0, Retry Count=
May 6 21:44:37.217: //1468/8E0C12800000/CCAPI/cc_api_call_disconnect_don
 Call Disconnect Event Sent
May 6 21:44:37.217: //-1/xxxxxxxxxxxx/CCAPI/cc_free_feature_vsa:

May 6 21:44:37.217: :cc_free_feature_vsa freeing 49FE5E78
May 6 21:44:37.217: //-1/xxxxxxxxxxxx/CCAPI/cc_free_feature_vsa:

May 6 21:44:37.217: vsacount in free is 0
BR2#
BR2#
```

A. The BR2 router does not have a matching inbound dial peer.
B. There is a codec mismatch between the source and destination. Codec negotiation failed
C. The BR2 router is sending the call to dial-peer 20003 which is a POTS dial-peer.
D. The destination phone is not registered.

**Correct Answer:** B
**Section: (none)**
**Explanation**

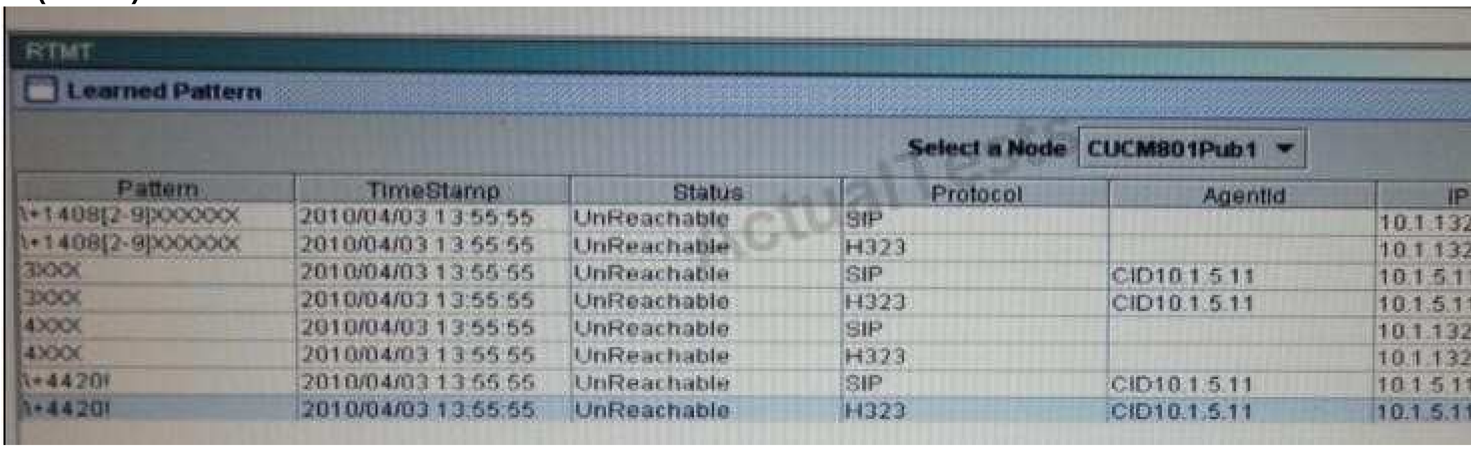
**Explanation/Reference:**
**Explanation**- Logs are clearly define the answer. **Link**- http://www.cisco.com/en/US/tech/tk1077/technologies_tech_note09186a0080094045.shtml#codec negvalues

**QUESTION 22**

## Refer to the exhibit.

An intercluster call was placed from extension 2001 to 3001. The SDI trace from the calling cluster. Which RTP port was used by the calling phone?

**75 (exhibit):**



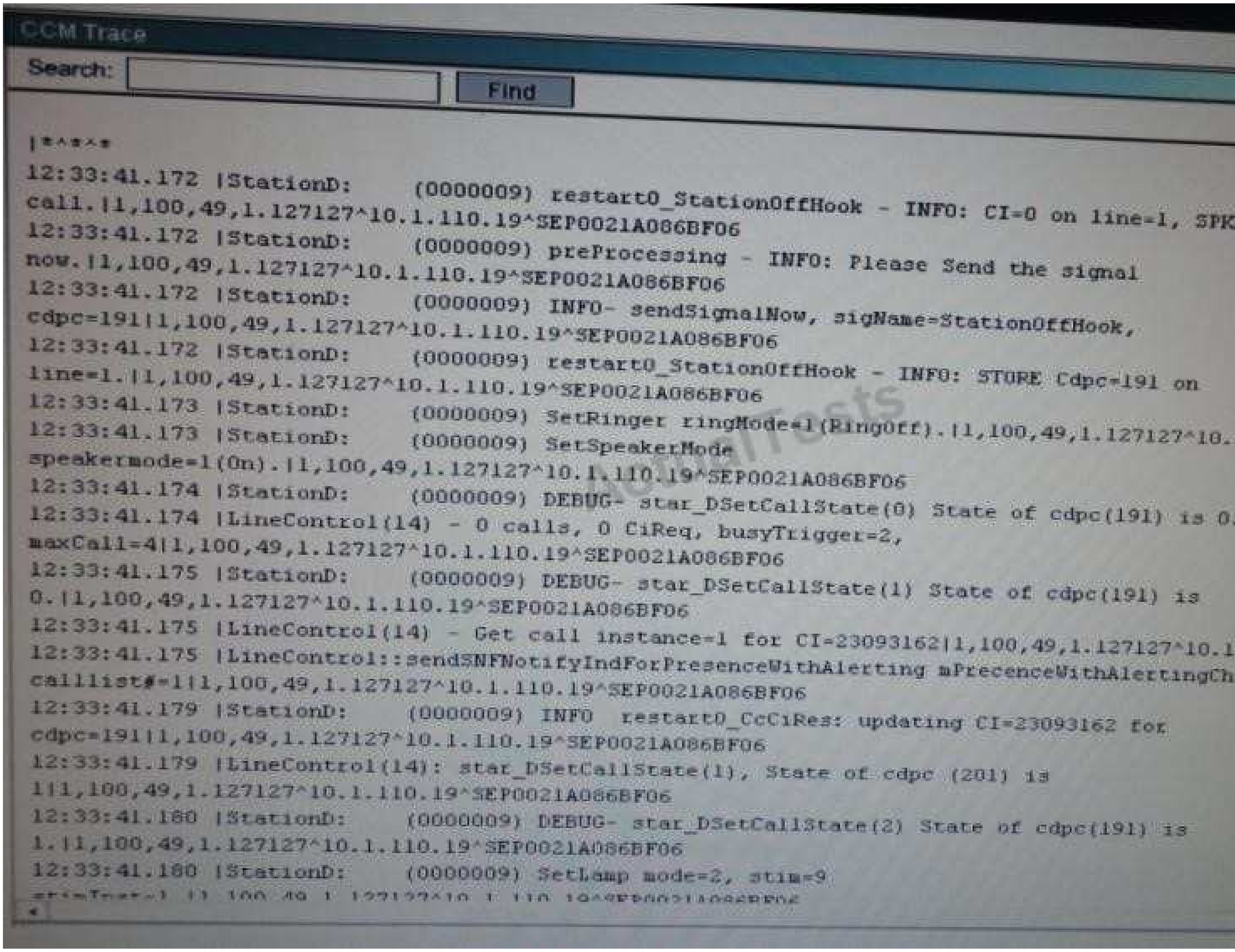
RTMT

Learned Pattern

Select a Node CUCM801Pub1

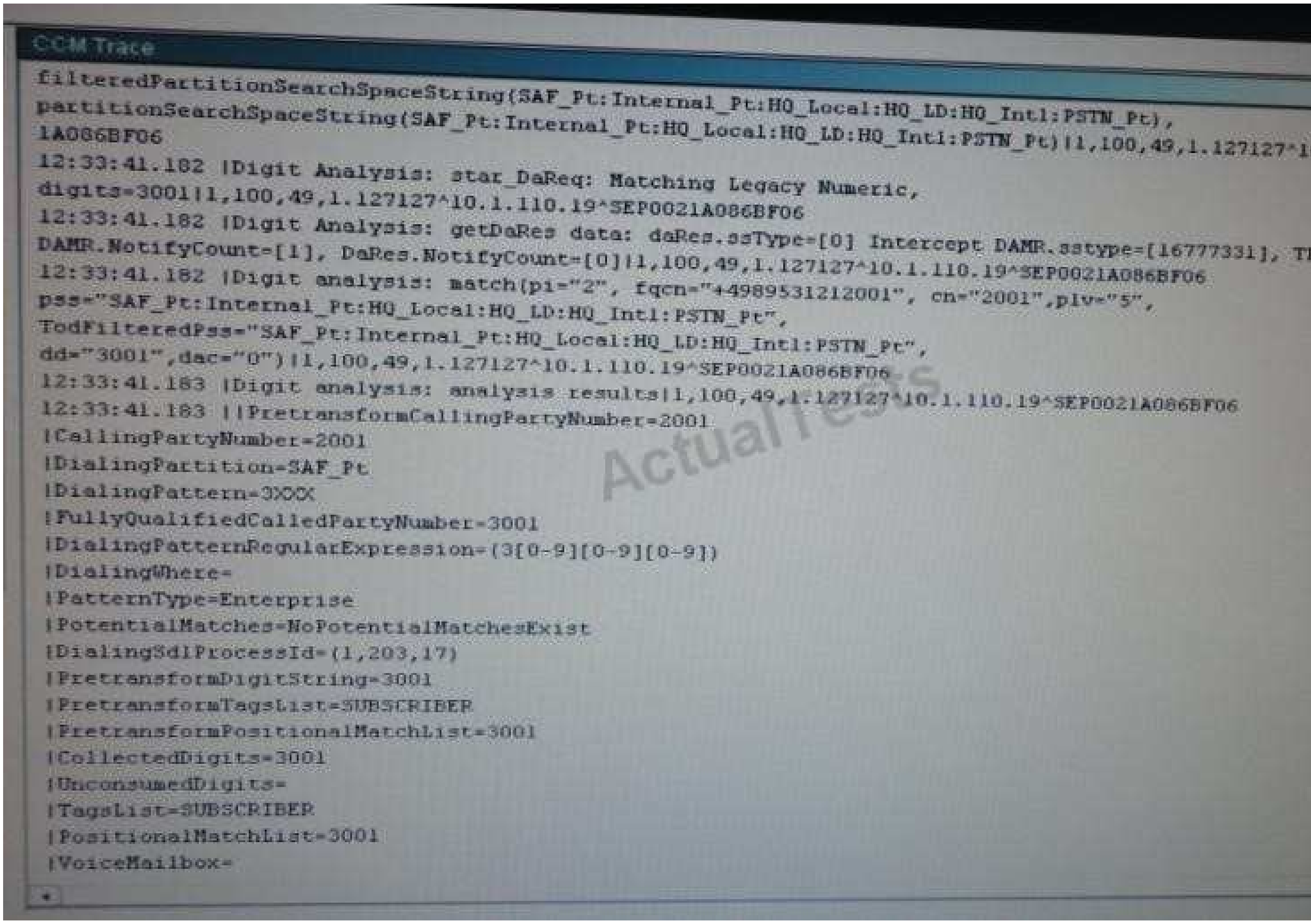
| Pattern | TimeStamp | Status | Protocol | AgentId | IP |
|---|---|---|---|---|---|
| \+1408[2-9]XXXXXX | 2010/04/03 13:55:55 | UnReachable | SIP | | 10.1.132 |
| \+1408[2-9]XXXXXX | 2010/04/03 13:55:55 | UnReachable | H323 | | 10.1.132 |
| 3XXX | 2010/04/03 13:55:55 | UnReachable | SIP | CID10.1.5.11 | 10.1.5.11 |
| 3XXX | 2010/04/03 13:55:55 | UnReachable | H323 | CID10.1.5.11 | 10.1.5.11 |
| 4XXX | 2010/04/03 13:55:55 | UnReachable | SIP | | 10.1.132 |
| 4XXX | 2010/04/03 13:55:55 | UnReachable | H323 | | 10.1.132 |
| \+44201 | 2010/04/03 13:55:55 | UnReachable | SIP | CID10.1.5.11 | 10.1.5.11 |
| \+44201 | 2010/04/03 13:55:55 | UnReachable | H323 | CID10.1.5.11 | 10.1.5.11 |

**76 (exhibit):**



CCM Trace
Search:
Find
12:33:41.172 |StationD: (0000009) restart0_StationOffHook - INFO: CI=0 on line=1, SPK
call.|1,100,49,1.127127^10.1.110.19^SEP0021A086BF06
12:33:41.172 |StationD: (0000009) preProcessing - INFO: Please Send the signal
now.|1,100,49,1.127127^10.1.110.19^SEP0021A086BF06
12:33:41.172 |StationD: (0000009) INFO- sendSignalNow, sigName=StationOffHook,
cdpc=191|1,100,49,1.127127^10.1.110.19^SEP0021A086BF06
12:33:41.172 |StationD: (0000009) restart0_StationOffHook - INFO: STORE Cdpc=191 on
line=1.|1,100,49,1.127127^10.1.110.19^SEP0021A086BF06
12:33:41.173 |StationD: (0000009) SetRinger ringMode=1(RingOff).|1,100,49,1.127127^10.
12:33:41.173 |StationD: (0000009) SetSpeakerMode
speakermode=1(On).|1,100,49,1.127127^10.1.110.19^SEP0021A086BF06
12:33:41.174 |StationD: (0000009) DEBUG- star_DSetCallState(0) State of cdpc(191) is 0.
12:33:41.174 |LineControl(14) - 0 calls, 0 CiReq, busyTrigger=2,
maxCall=4|1,100,49,1.127127^10.1.110.19^SEP0021A086BF06
12:33:41.175 |StationD: (0000009) DEBUG- star_DSetCallState(1) State of cdpc(191) is
0.|1,100,49,1.127127^10.1.110.19^SEP0021A086BF06
12:33:41.175 |LineControl(14) - Get call instance=1 for CI=23093162|1,100,49,1.127127^10.1
12:33:41.175 |LineControl::sendSNFNotifyIndForPresenceWithAlerting mPrecenceWithAlertingCh
calllist#=1|1,100,49,1.127127^10.1.110.19^SEP0021A086BF06
12:33:41.179 |StationD: (0000009) INFO restart0_CcCiRes: updating CI=23093162 for
cdpc=191|1,100,49,1.127127^10.1.110.19^SEP0021A086BF06
12:33:41.179 |LineControl(14): star_DSetCallState(1), State of cdpc (201) is
1|1,100,49,1.127127^10.1.110.19^SEP0021A086BF06
12:33:41.180 |StationD: (0000009) DEBUG- star_DSetCallState(2) State of cdpc(191) is
1.|1,100,49,1.127127^10.1.110.19^SEP0021A086BF06
12:33:41.180 |StationD: (0000009) SetLamp mode=2, stim=9


**77 (exhibit):**

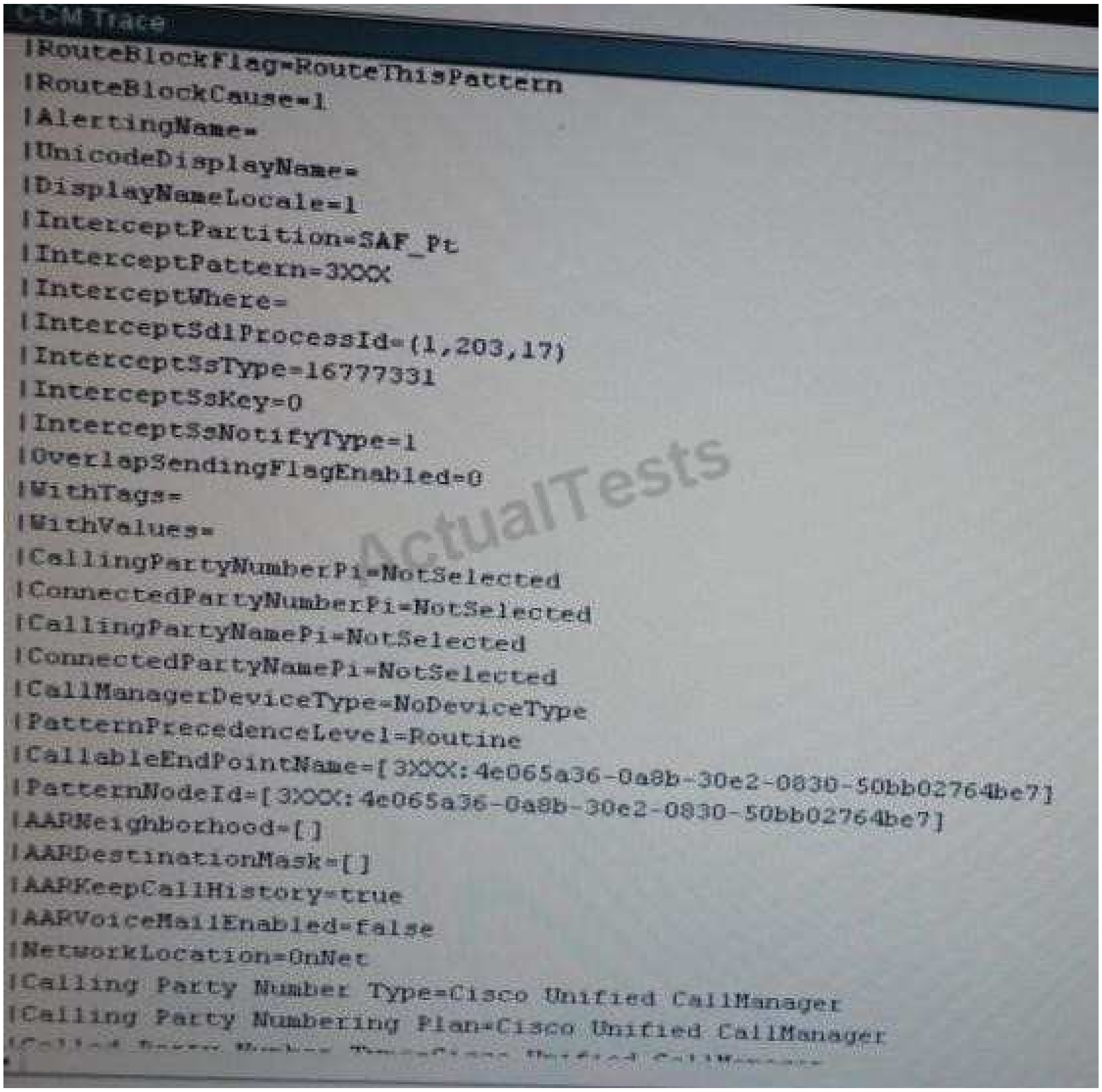


CCM Trace

```
filteredPartitionSearchSpaceString(SAF_Pt:Internal_Pt:HQ_Local:HQ_LD:HQ_Intl:PSTN_Pt),
partitionSearchSpaceString(SAF_Pt:Internal_Pt:HQ_Local:HQ_LD:HQ_Intl:PSTN_Pt)|1,100,49,1.127127^1
1A086BF06
12:33:41.182 |Digit Analysis: star_DaReq: Matching Legacy Numeric,
digits=3001|1,100,49,1.127127^10.1.110.19^SEP0021A086BF06
12:33:41.182 |Digit Analysis: getDaRes data: daRes.ssType=[0] Intercept DAMR.sstype=[16777331], T
DAMR.NotifyCount=[1], DaRes.NotifyCount=[0]|1,100,49,1.127127^10.1.110.19^SEP0021A086BF06
12:33:41.182 |Digit analysis: match(pi="2", fqcn="+4989531212001", cn="2001",plv="5",
pss="SAF_Pt:Internal_Pt:HQ_Local:HQ_LD:HQ_Intl:PSTN_Pt",
TodFilteredPss="SAF_Pt:Internal_Pt:HQ_Local:HQ_LD:HQ_Intl:PSTN_Pt",
dd="3001",dac="0")|1,100,49,1.127127^10.1.110.19^SEP0021A086BF06
12:33:41.183 |Digit analysis: analysis results|1,100,49,1.127127^10.1.110.19^SEP0021A086BF06
12:33:41.183 ||PretransformCallingPartyNumber=2001
|CallingPartyNumber=2001
|DialingPartition=SAF_Pt
|DialingPattern=3XXX
|FullyQualifiedCalledPartyNumber=3001
|DialingPatternRegularExpression=(3[0-9][0-9][0-9])
|DialingWhere=
|PatternType=Enterprise
|PotentialMatches=NoPotentialMatchesExist
|DialingSdlProcessId=(1,203,17)
|PretransformDigitString=3001
|PretransformTagsList=SUBSCRIBER
|PretransformPositionalMatchList=3001
|CollectedDigits=3001
|UnconsumedDigits=
|TagsList=SUBSCRIBER
|PositionalMatchList=3001
|VoiceMailbox=
```

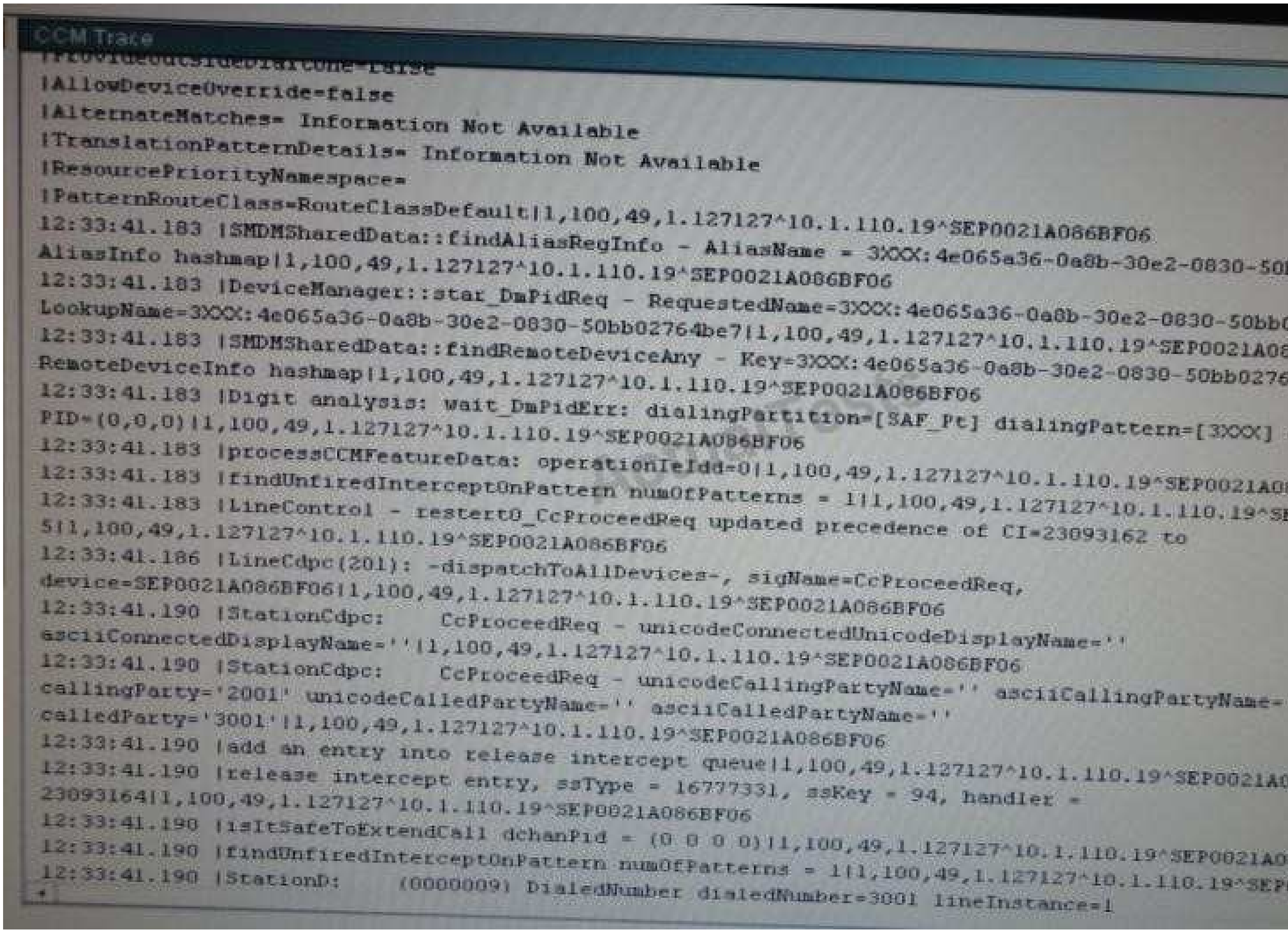


**78 (exhibit):**


CCM Trace
|RouteBlockFlag=RouteThisPattern
|RouteBlockCause=1
|AlertingName=
|UnicodeDisplayName=
|DisplayNameLocale=1
|InterceptPartition=SAF_Pt
|InterceptPattern=3XXX
|InterceptWhere=
|InterceptSdlProcessId=(1,203,17)
|InterceptSsType=16777331
|InterceptSsKey=0
|InterceptSsNotifyType=1
|OverlapSendingFlagEnabled=0
|WithTags=
|WithValues=
|CallingPartyNumberPi=NotSelected
|ConnectedPartyNumberPi=NotSelected
|CallingPartyNamePi=NotSelected
|ConnectedPartyNamePi=NotSelected
|CallManagerDeviceType=NoDeviceType
|PatternPrecedenceLevel=Routine
|CallableEndPointName=[3XXX:4e065a36-0a8b-30e2-0830-50bb02764be7]
|PatternNodeId=[3XXX:4e065a36-0a8b-30e2-0830-50bb02764be7]
|AARNeighborhood=[]
|AARDestinationMask=[]
|AARKeepCallHistory=true
|AARVoiceMailEnabled=false
|NetworkLocation=OnNet
|Calling Party Number Type=Cisco Unified CallManager
|Calling Party Numbering Plan=Cisco Unified CallManager



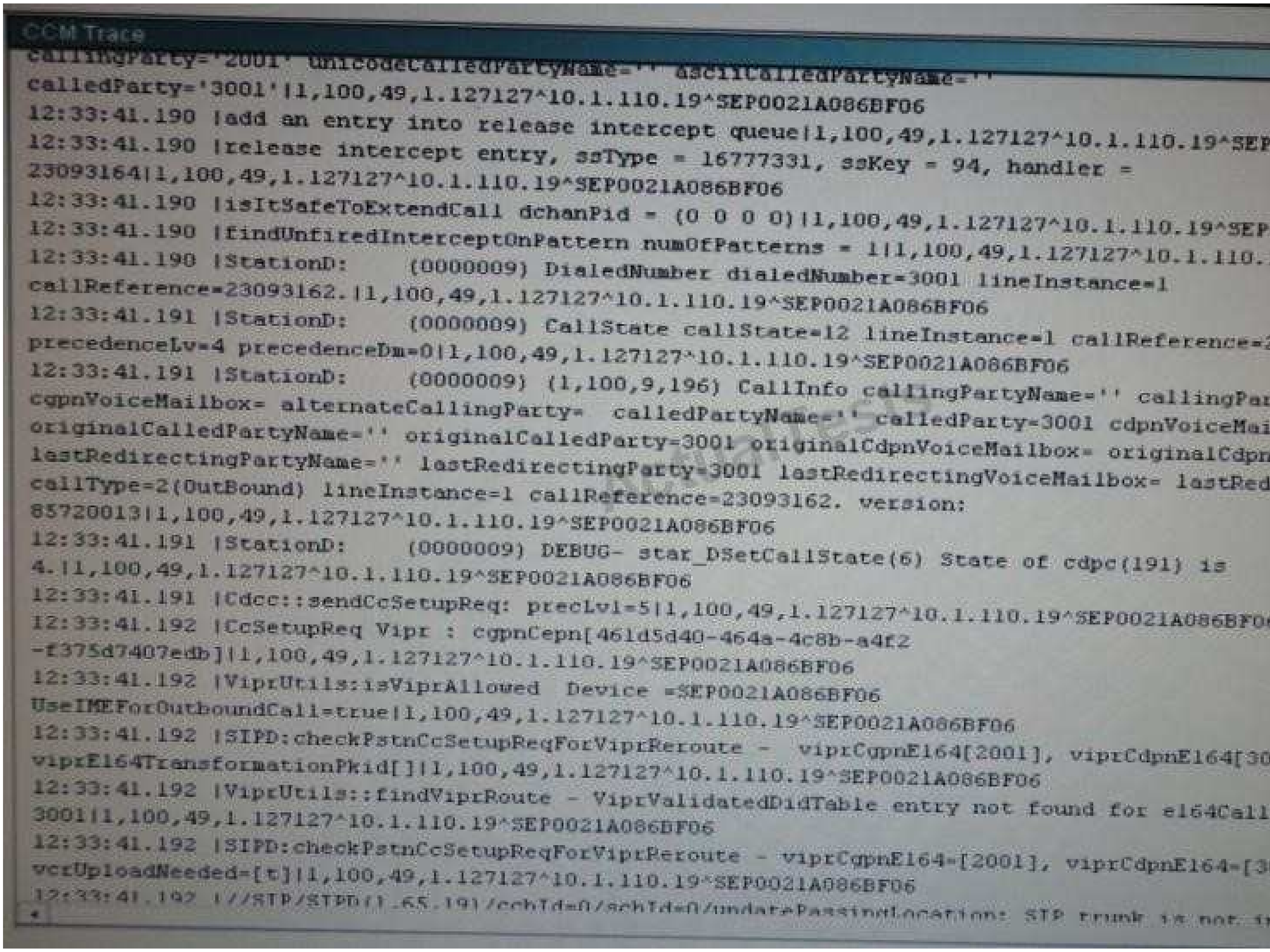
**79 (exhibit):**



CCM Trace

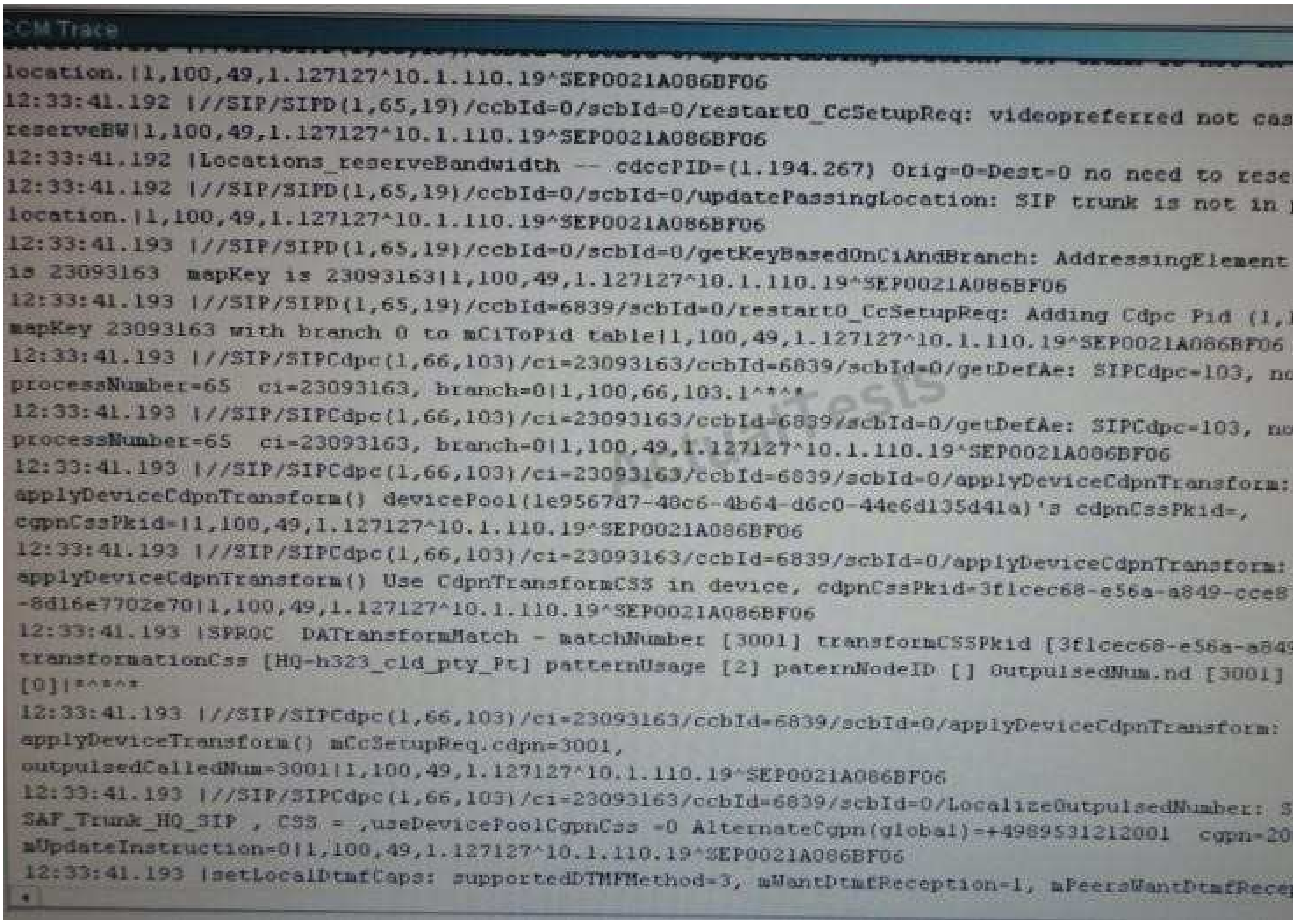
```
|ProvideOutsideDialtone=false
|AllowDeviceOverride=false
|AlternateMatches= Information Not Available
|TranslationPatternDetails= Information Not Available
|ResourcePriorityNamespace=
|PatternRouteClass=RouteClassDefault|1,100,49,1.127127^10.1.110.19^SEP0021A086BF06
12:33:41.183 |SMDMSharedData::findAliasRegInfo - AliasName = 3XXX:4e065a36-0a8b-30e2-0830-50b
AliasInfo hashmap|1,100,49,1.127127^10.1.110.19^SEP0021A086BF06
12:33:41.183 |DeviceManager::star_DmPidReq - RequestedName=3XXX:4e065a36-0a8b-30e2-0830-50bb0
LookupName=3XXX:4e065a36-0a8b-30e2-0830-50bb02764be7|1,100,49,1.127127^10.1.110.19^SEP0021A08
12:33:41.183 |SMDMSharedData::findRemoteDeviceAny - Key=3XXX:4e065a36-0a8b-30e2-0830-50bb0276
RemoteDeviceInfo hashmap|1,100,49,1.127127^10.1.110.19^SEP0021A086BF06
12:33:41.183 |Digit analysis: wait_DmPidErr: dialingPartition=[SAF_Pt] dialingPattern=[3XXX] c
PID=(0,0,0)|1,100,49,1.127127^10.1.110.19^SEP0021A086BF06
12:33:41.183 |processCCMFeatureData: operationIeIdd=0|1,100,49,1.127127^10.1.110.19^SEP0021A08
12:33:41.183 |findUnfiredInterceptOnPattern numOfPatterns = 1|1,100,49,1.127127^10.1.110.19^SEP0021A08
12:33:41.183 |LineControl - restert0_CcProceedReq updated precedence of CI=23093162 to
5|1,100,49,1.127127^10.1.110.19^SEP0021A086BF06
12:33:41.186 |LineCdpc(201): -dispatchToAllDevices-, sigName=CcProceedReq,
device=SEP0021A086BF06|1,100,49,1.127127^10.1.110.19^SEP0021A086BF06
12:33:41.190 |StationCdpc:    CcProceedReq - unicodeConnectedUnicodeDisplayName=''
asciiConnectedDisplayName=''|1,100,49,1.127127^10.1.110.19^SEP0021A086BF06
12:33:41.190 |StationCdpc:    CcProceedReq - unicodeCallingPartyName='' asciiCallingPartyName='
callingParty='2001' unicodeCalledPartyName='' asciiCalledPartyName=''
calledParty='3001'|1,100,49,1.127127^10.1.110.19^SEP0021A086BF06
12:33:41.190 |add an entry into release intercept queue|1,100,49,1.127127^10.1.110.19^SEP0021A0
12:33:41.190 |release intercept entry, ssType = 16777331, ssKey = 94, handler =
23093164|1,100,49,1.127127^10.1.110.19^SEP0021A086BF06
12:33:41.190 |isItSafeToExtendCall dchanPid = (0 0 0 0)|1,100,49,1.127127^10.1.110.19^SEP0021A08
12:33:41.190 |findUnfiredInterceptOnPattern numOfPatterns = 1|1,100,49,1.127127^10.1.110.19^SEP0
12:33:41.190 |StationD:    (0000009) DialedNumber dialedNumber=3001 lineInstance=1
```

**80 (exhibit):**


CCM Trace

```
callingParty='2001' unicodeCalledPartyName='' asciiCalledPartyName=''
calledParty='3001'|1,100,49,1.127127^10.1.110.19^SEP0021A086BF06
12:33:41.190 |add an entry into release intercept queue|1,100,49,1.127127^10.1.110.19^SEP
12:33:41.190 |release intercept entry, ssType = 16777331, ssKey = 94, handler =
23093164|1,100,49,1.127127^10.1.110.19^SEP0021A086BF06
12:33:41.190 |isItSafeToExtendCall dchanPid = (0 0 0 0)|1,100,49,1.127127^10.1.110.19^SEP
12:33:41.190 |findUnfiredInterceptOnPattern numOfPatterns = 1|1,100,49,1.127127^10.1.110.
12:33:41.190 |StationD:     (0000009) DialedNumber dialedNumber=3001 lineInstance=1
callReference=23093162.|1,100,49,1.127127^10.1.110.19^SEP0021A086BF06
12:33:41.191 |StationD:     (0000009) CallState callState=12 lineInstance=1 callReference=
precedenceLv=4 precedenceDm=0|1,100,49,1.127127^10.1.110.19^SEP0021A086BF06
12:33:41.191 |StationD:     (0000009) (1,100,9,196) CallInfo callingPartyName='' callingPar
cgpnVoiceMailbox= alternateCallingParty=  calledPartyName='' calledParty=3001 cdpnVoiceMai
originalCalledPartyName='' originalCalledParty=3001 originalCdpnVoiceMailbox= originalCdpn
lastRedirectingPartyName='' lastRedirectingParty=3001 lastRedirectingVoiceMailbox= lastRed
callType=2(OutBound) lineInstance=1 callReference=23093162. version:
85720013|1,100,49,1.127127^10.1.110.19^SEP0021A086BF06
12:33:41.191 |StationD:     (0000009) DEBUG- star_DSetCallState(6) State of cdpc(191) is
4.|1,100,49,1.127127^10.1.110.19^SEP0021A086BF06
12:33:41.191 |Cdcc::sendCcSetupReq: precLvl=5|1,100,49,1.127127^10.1.110.19^SEP0021A086BF0
12:33:41.192 |CcSetupReq Vipr : cgpnCepn[461d5d40-464a-4c8b-a4f2
-f375d7407edb]|1,100,49,1.127127^10.1.110.19^SEP0021A086BF06
12:33:41.192 |ViprUtils:isViprAllowed  Device =SEP0021A086BF06
UseIMEForOutboundCall=true|1,100,49,1.127127^10.1.110.19^SEP0021A086BF06
12:33:41.192 |SIPD:checkPstnCcSetupReqForViprReroute -  viprCgpnE164[2001], viprCdpnE164[30
viprE164TransformationPkid[]|1,100,49,1.127127^10.1.110.19^SEP0021A086BF06
12:33:41.192 |ViprUtils::findViprRoute - ViprValidatedDidTable entry not found for e164Call
3001|1,100,49,1.127127^10.1.110.19^SEP0021A086BF06
12:33:41.192 |SIPD:checkPstnCcSetupReqForViprReroute - viprCgpnE164=[2001], viprCdpnE164=[3
vcrUploadNeeded=[t]|1,100,49,1.127127^10.1.110.19^SEP0021A086BF06
12:33:41.192 |//SIP/SIPD(1,65,19)/cchId=0/schId=0/updatePassingLocation: SIP trunk is not i
```

**81 (exhibit):**



```
CCM Trace
location.|1,100,49,1.127127^10.1.110.19^SEP0021A086BF06
12:33:41.192 |//SIP/SIPD(1,65,19)/ccbId=0/scbId=0/restart0_CcSetupReq: videopreferred not cas
reserveBW|1,100,49,1.127127^10.1.110.19^SEP0021A086BF06
12:33:41.192 |Locations_reserveBandwidth -- cdccPID=(1.194.267) Orig=0=Dest=0 no need to rese
12:33:41.192 |//SIP/SIPD(1,65,19)/ccbId=0/scbId=0/updatePassingLocation: SIP trunk is not in
location.|1,100,49,1.127127^10.1.110.19^SEP0021A086BF06
12:33:41.193 |//SIP/SIPD(1,65,19)/ccbId=0/scbId=0/getKeyBasedOnCiAndBranch: AddressingElement
is 23093163  mapKey is 23093163|1,100,49,1.127127^10.1.110.19^SEP0021A086BF06
12:33:41.193 |//SIP/SIPD(1,65,19)/ccbId=6839/scbId=0/restart0_CcSetupReq: Adding Cdpc Pid (1,
mapKey 23093163 with branch 0 to mCiToPid table|1,100,49,1.127127^10.1.110.19^SEP0021A086BF06
12:33:41.193 |//SIP/SIPCdpc(1,66,103)/ci=23093163/ccbId=6839/scbId=0/getDefAe: SIPCdpc=103, no
processNumber=65  ci=23093163, branch=0|1,100,66,103.1^*^*
12:33:41.193 |//SIP/SIPCdpc(1,66,103)/ci=23093163/ccbId=6839/scbId=0/getDefAe: SIPCdpc=103, no
processNumber=65  ci=23093163, branch=0|1,100,49,1.127127^10.1.110.19^SEP0021A086BF06
12:33:41.193 |//SIP/SIPCdpc(1,66,103)/ci=23093163/ccbId=6839/scbId=0/applyDeviceCdpnTransform:
applyDeviceCdpnTransform() devicePool(1e9567d7-48c6-4b64-d6c0-44e6d135d41a)'s cdpnCssPkid=,
cgpnCssPkid=|1,100,49,1.127127^10.1.110.19^SEP0021A086BF06
12:33:41.193 |//SIP/SIPCdpc(1,66,103)/ci=23093163/ccbId=6839/scbId=0/applyDeviceCdpnTransform:
applyDeviceCdpnTransform() Use CdpnTransformCSS in device, cdpnCssPkid=3f1cec68-e56a-a849-cce8
-8d16e7702e70|1,100,49,1.127127^10.1.110.19^SEP0021A086BF06
12:33:41.193 |SPROC  DATransformMatch - matchNumber [3001] transformCSSPkid [3f1cec68-e56a-a849
transformationCss [HQ-h323_cld_pty_Pt] patternUsage [2] patenNodeID [] OutpulsedNum.nd [3001]
[0]|*^*^*
12:33:41.193 |//SIP/SIPCdpc(1,66,103)/ci=23093163/ccbId=6839/scbId=0/applyDeviceCdpnTransform:
applyDeviceTransform() mCcSetupReq.cdpn=3001,
outpulsedCalledNum=3001|1,100,49,1.127127^10.1.110.19^SEP0021A086BF06
12:33:41.193 |//SIP/SIPCdpc(1,66,103)/ci=23093163/ccbId=6839/scbId=0/LocalizeOutpulsedNumber: S
SAF_Trunk_HQ_SIP , CSS = ,useDevicePoolCgpnCss =0 AlternateCgpn(global)=+4989531212001  cgpn=20
mUpdateInstruction=0|1,100,49,1.127127^10.1.110.19^SEP0021A086BF06
12:33:41.193 |setLocalDtmfCaps: supportedDTMFMethod=3, mWantDtmfReception=1, mPeersWantDtmfRece
```

**82 (exhibit):**

CCM Trace

```
policy[1], resvStatus[1], video[0]|1,100,49,1.127127^10.1.110.19^SEP0021A086BF06
12:33:41.194 |//SIP/SIPCdpc(1,66,103)/ci=23093163/ccbId=6839/scbId=0/handleCcSetupReq: mtpReq
mGClearCall=0|1,100,49,1.127127^10.1.110.19^SEP0021A086BF06
12:33:41.194 |//SIP/SIPCdpc(1,66,103)/ci=23093163/ccbId=6839/scbId=0/handleCcSetupReq: mixing
mP=0, eM=0, eP=0|1,100,49,1.127127^10.1.110.19^SEP0021A086BF06
12:33:41.194 |//SIP/SIPCdpc(1,66,103)/ci=23093163/ccbId=6839/scbId=0/getDefCcRegister: Secure
mSrtpPresent=0|1,100,49,1.127127^10.1.110.19^SEP0021A086BF06
12:33:41.194 |//SIP/SIPCdpc(1,66,103)/ci=23093163/ccbId=6839/scbId=0/getDefSetup: NumberPi 1 ,
1|1,100,49,1.127127^10.1.110.19^SEP0021A086BF06
12:33:41.194 |//SIP/SIPCdpc(1,66,103)/ci=23093163/ccbId=6839/scbId=0/setIdentityOutgoingSIPMsg
identityFlag=[PAI;RPID;], privacyType[0]|1,100,49,1.127127^10.1.110.19^SEP0021A086BF06
12:33:41.194 |//SIP/SIPCdpc(1,66,103)/ci=23093163/ccbId=6839/scbId=0/getDefSetup: ReqURI is no
present|1,100,49,1.127127^10.1.110.19^SEP0021A086BF06
12:33:41.194 |//SIP/SIPCdpc(1,66,103)/ci=23093163/ccbId=6839/scbId=0/getDefSetup: fSetup.mDevi
07dd-672e-1324-468369cb850b|1,100,49,1.127127^10.1.110.19^SEP0021A086BF06
12:33:41.194 |//SIP/SIPCdpc(1,66,103)/ci=23093163/ccbId=6839/scbId=0/handleDynamicSetup: CcSetu
feature data is present|1,100,49,1.127127^10.1.110.19^SEP0021A086BF06
12:33:41.194 |CMSIPUtility::getDynamicRouteAndUpdList Dynamic target is address 10.1.5.11, rout
-0f08-f242-4a3022a291ac@CUCM801Pub2, transport is 1, and port = 5060|*^*^*
12:33:41.194 |//SIP/SIPCdpc(1,66,103)/ci=23093163/ccbId=6839/scbId=0/buildCallInfoHeader: Encod
support in Call-Info header.|1,100,49,1.127127^10.1.110.19^SEP0021A086BF06
12:33:41.195 |//SIP/SIPCdpc(1,66,103)/ci=23093163/ccbId=6839/scbId=0/appendRPHdr: appendRPHdr -
Network Domain set and not precedence call, return|1,100,49,1.127127^10.1.110.19^SEP0021A086BF0
12:33:41.195 |//SIP/SIPCdpc(1,66,103)/ci=23093163/ccbId=6839/scbId=0/appendGuidHeader: adding G
sipContainerWrapper: 51775E00BA31D195000000370A05010A|1,100,49,1.127127^10.1.110.19^SEP0021A086
12:33:41.195 |//SIP/SIPCdpc(1,66,103)/ci=23093163/ccbId=6839/scbId=0/getRedirectingNumIE:
outboundRedirectingNumIEDeliveryFlag is not enabled, no Diversion header will be
sent|1,100,49,1.127127^10.1.110.19^SEP0021A086BF06
12:33:41.195 |Cdcc - (0000267) - updateDchanCrp - secure capability on side 1 is
(1,1)|1,100,49,1.127127^10.1.110.19^SEP0021A086BF06
```

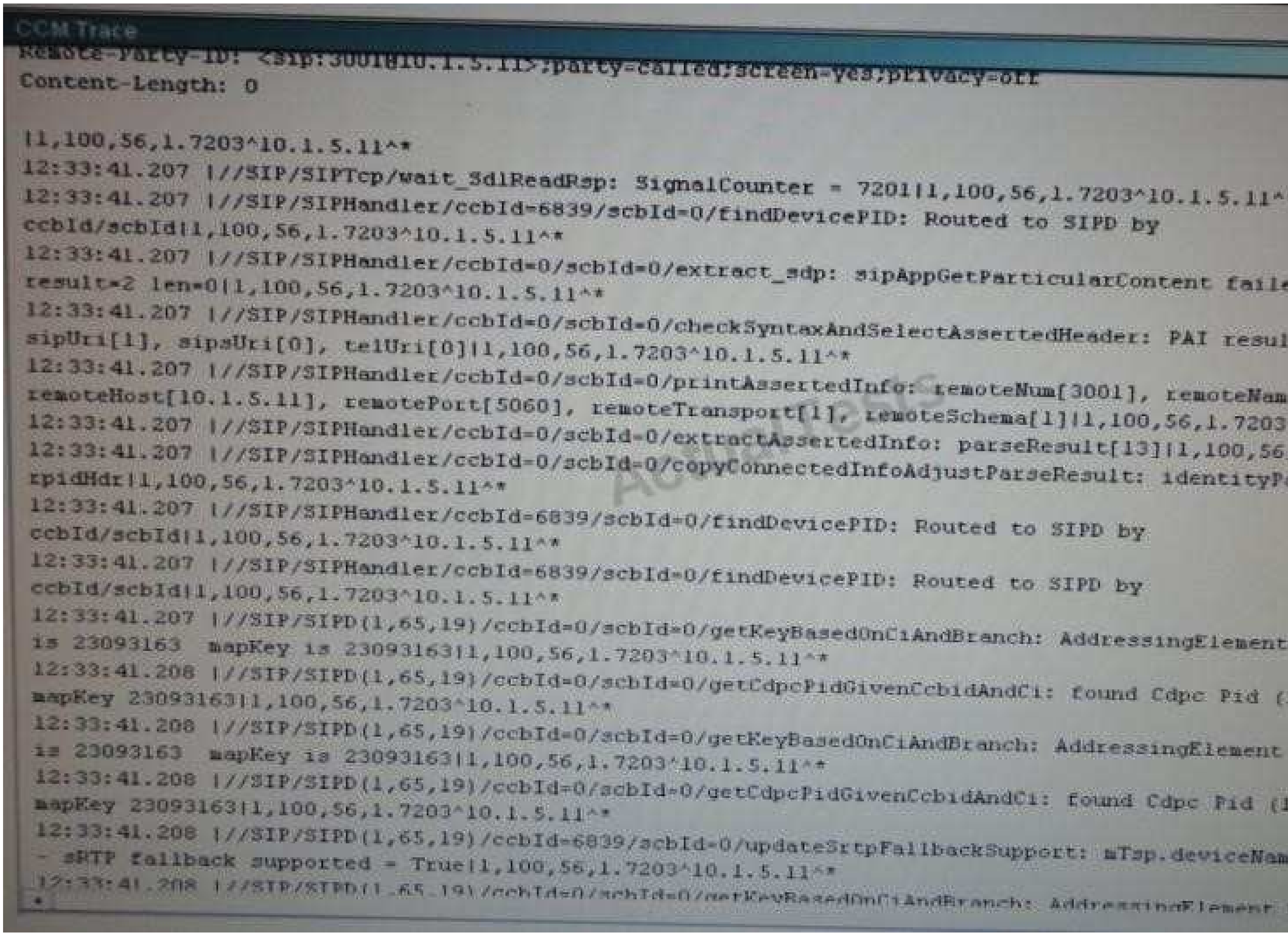
**83 (exhibit):**

CCM Trace

```
->pid|1,100,49,1.127127^10.1.110.19^SEP0021A086BF06
12:33:41.196 |//SIP/SIPHandler/ccbId=0/scbId=0/sipSPIGetCallExtensionSupported:
SIPRel1xxEnabledServiceParamSetting=0 , ccb->pid.outboundRel1xx=1|1,100,49,1.127127^10.1.11
12:33:41.196 |//SIP/SIPHandler/ccbId=0/scbId=0/sip_stop_timer: type=SIP_TIMER_TRYING value=
retries=6|1,100,49,1.127127^10.1.110.19^SEP0021A086BF06
12:33:41.196 |//SIP/SIPHandler/ccbId=0/scbId=0/sip_start_timer: type=SIP_TIMER_TRYING value=
retries=6|1,100,49,1.127127^10.1.110.19^SEP0021A086BF06
12:33:41.197 |//SIP/SIPD(1,65,19)/ccbId=0/scbId=0/getKeyBasedOnCiAndBranch: AddressingElemen
is 23093163  mapKey is 23093163|1,100,49,1.127127^10.1.110.19^SEP0021A086BF06
12:33:41.197 |//SIP/SIPD(1,65,19)/ccbId=0/scbId=0/getCdpcPidGivenCcbidAndCi: found Cdpc Pid
mapKey 23093163|1,100,49,1.127127^10.1.110.19^SEP0021A086BF06
12:33:41.197 |//SIP/SIPTcp/wait_SdlSPISignal: received a spi signal
...|1,100,49,1.127127^10.1.110.19^SEP0021A086BF06
12:33:41.197 |//SIP/SIPTcp/wait_SdlSPISignal: Outgoing SIP TCP message to 10.1.5.11 on port
INVITE sip:3001@10.1.5.11:5060 SIP/2.0
Via: SIP/2.0/TCP 10.1.5.10:5060;branch=z9hG4bK7347799947d
From: <sip:2001@10.1.5.10>;tag=ae2783cb-9687-4fc7-a1d0-8108b8b3679a-23093163
To: <sip:3001@10.1.5.11>
Date: Fri, 19 Mar 2010 19:33:41 GMT
Call-ID: 51775e00-ba31d195-44-a05010a@10.1.5.10
Supported: timer,resource-priority,replaces
Min-SE:  1800
User-Agent: Cisco-CUCM8.0
Allow: INVITE, OPTIONS, INFO, BYE, CANCEL, ACK, PRACK, UPDATE, REFER, SUBSCRIBE, NOTIFY
CSeq: 101 INVITE
Contact: <sip:2001@10.1.5.10:5060;transport=tcp>
Expires: 180
Allow-Events: presence, kpml
Route: sip:aff96646-1cc6-0f08-f242-4a3022a291ac@CUCM801Pub2
Supported: X-cisco-srtp-fallback
```



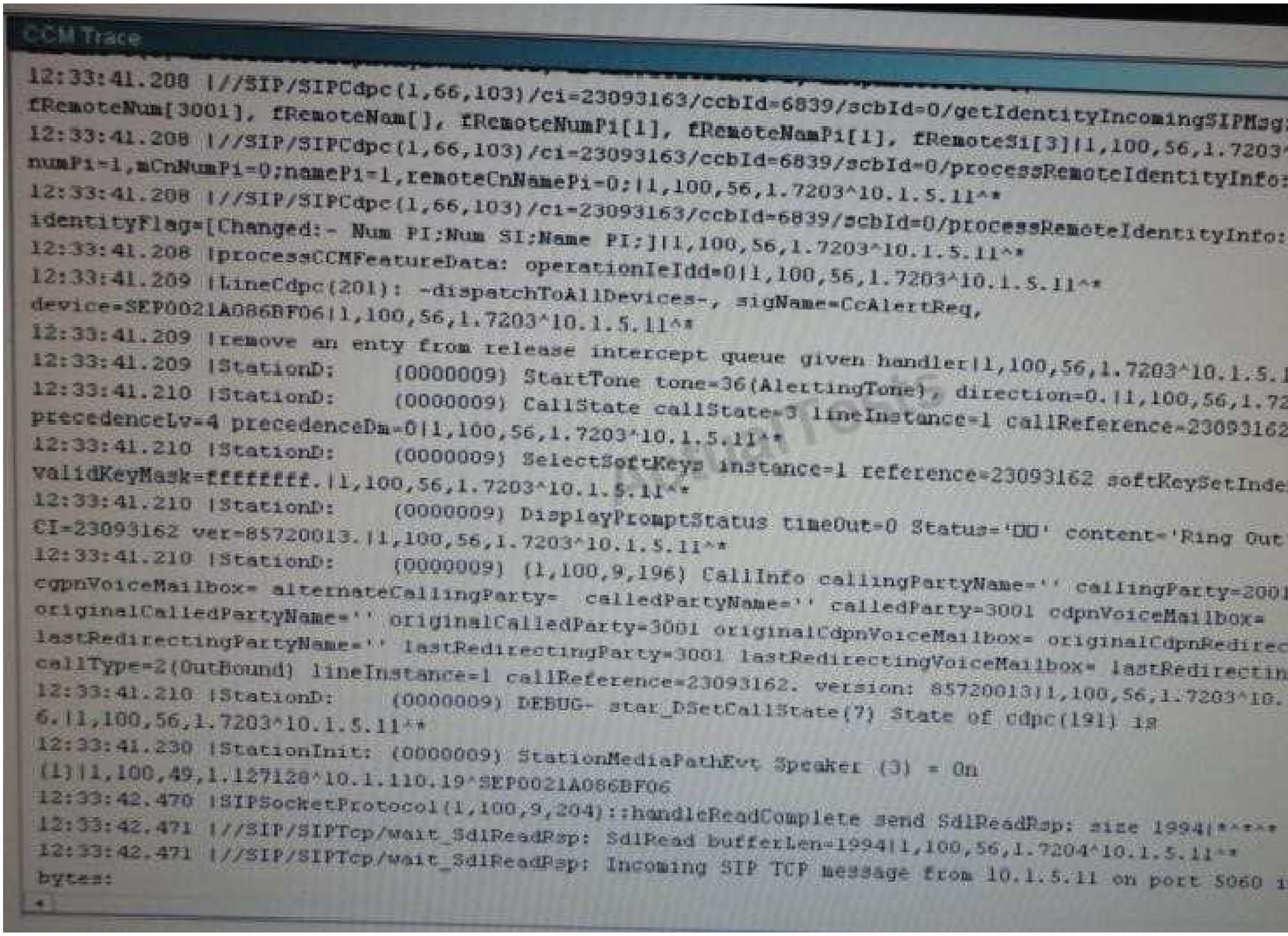
**84 (exhibit):**



CCM Trace

```
Remote-Party-ID: <sip:3001@10.1.5.11>;party=called;screen=yes;privacy=off
Content-Length: 0

|1,100,56,1.7203^10.1.5.11^*
12:33:41.207 |//SIP/SIPTcp/wait_SdlReadRsp: SignalCounter = 7201|1,100,56,1.7203^10.1.5.11^
12:33:41.207 |//SIP/SIPHandler/ccbId=6839/scbId=0/findDevicePID: Routed to SIPD by
ccbId/scbId|1,100,56,1.7203^10.1.5.11^*
12:33:41.207 |//SIP/SIPHandler/ccbId=0/scbId=0/extract_sdp: sipAppGetParticularContent faile
result=2 len=0|1,100,56,1.7203^10.1.5.11^*
12:33:41.207 |//SIP/SIPHandler/ccbId=0/scbId=0/checkSyntaxAndSelectAssertedHeader: PAI resul
sipUri[1], sipsUri[0], telUri[0]|1,100,56,1.7203^10.1.5.11^*
12:33:41.207 |//SIP/SIPHandler/ccbId=0/scbId=0/printAssertedInfo: remoteNum[3001], remoteNam
remoteHost[10.1.5.11], remotePort[5060], remoteTransport[1], remoteSchema[1]|1,100,56,1.7203
12:33:41.207 |//SIP/SIPHandler/ccbId=0/scbId=0/extractAssertedInfo: parseResult[13]|1,100,56
12:33:41.207 |//SIP/SIPHandler/ccbId=0/scbId=0/copyConnectedInfoAdjustParseResult: identityP
rpidHdr|1,100,56,1.7203^10.1.5.11^*
12:33:41.207 |//SIP/SIPHandler/ccbId=6839/scbId=0/findDevicePID: Routed to SIPD by
ccbId/scbId|1,100,56,1.7203^10.1.5.11^*
12:33:41.207 |//SIP/SIPHandler/ccbId=6839/scbId=0/findDevicePID: Routed to SIPD by
ccbId/scbId|1,100,56,1.7203^10.1.5.11^*
12:33:41.207 |//SIP/SIPD(1,65,19)/ccbId=0/scbId=0/getKeyBasedOnCiAndBranch: AddressingElement
is 23093163  mapKey is 23093163|1,100,56,1.7203^10.1.5.11^*
12:33:41.208 |//SIP/SIPD(1,65,19)/ccbId=0/scbId=0/getCdpcPidGivenCcbidAndCi: found Cdpc Pid (
mapKey 23093163|1,100,56,1.7203^10.1.5.11^*
12:33:41.208 |//SIP/SIPD(1,65,19)/ccbId=0/scbId=0/getKeyBasedOnCiAndBranch: AddressingElement
is 23093163  mapKey is 23093163|1,100,56,1.7203^10.1.5.11^*
12:33:41.208 |//SIP/SIPD(1,65,19)/ccbId=0/scbId=0/getCdpcPidGivenCcbidAndCi: found Cdpc Pid (1
mapKey 23093163|1,100,56,1.7203^10.1.5.11^*
12:33:41.208 |//SIP/SIPD(1,65,19)/ccbId=6839/scbId=0/updateSrtpFallbackSupport: mTsp.deviceNam
- sRTP fallback supported = True|1,100,56,1.7203^10.1.5.11^*
12:33:41.208 |//SIP/SIPD(1,65,19)/ccbId=0/scbId=0/getKeyBasedOnCiAndBranch: AddressingElement
```

**85 (exhibit):**

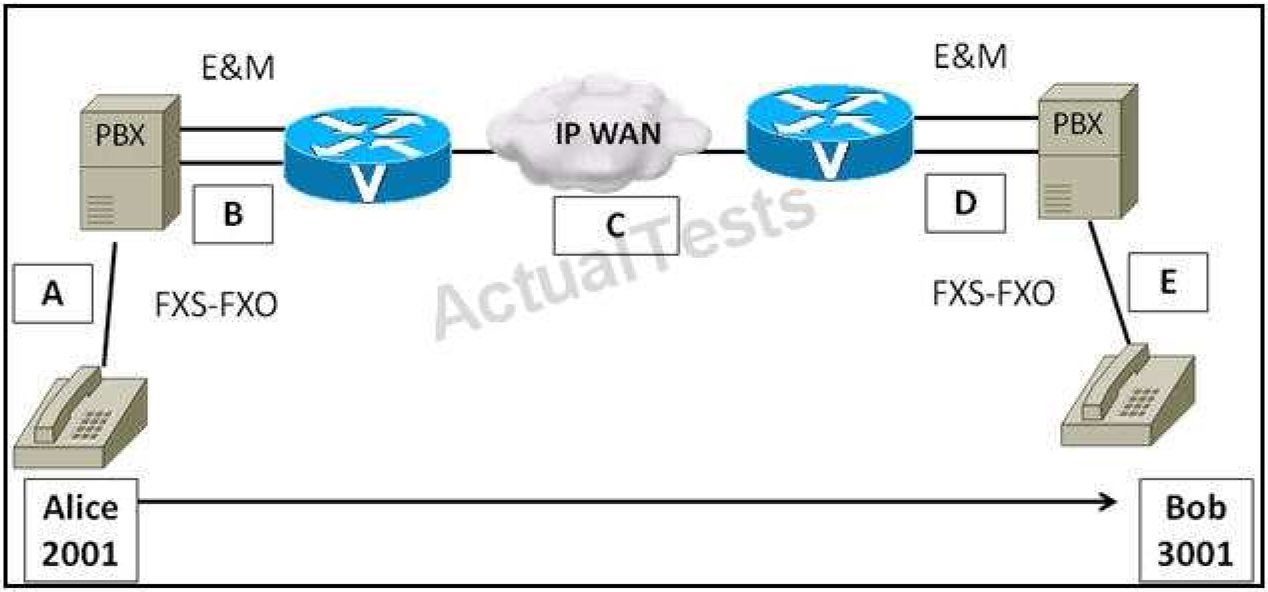



A. 4000
B. 8000
C. 18462
D. 19470
E. Not possible to tell because a second invite was sent because of call failure

**Correct Answer:** C
**Section: (none)**
**Explanation**

**Explanation/Reference:**
**Explanation**- RTP port shows in the logs.

**QUESTION 23**
**Refer to the exhibit.**



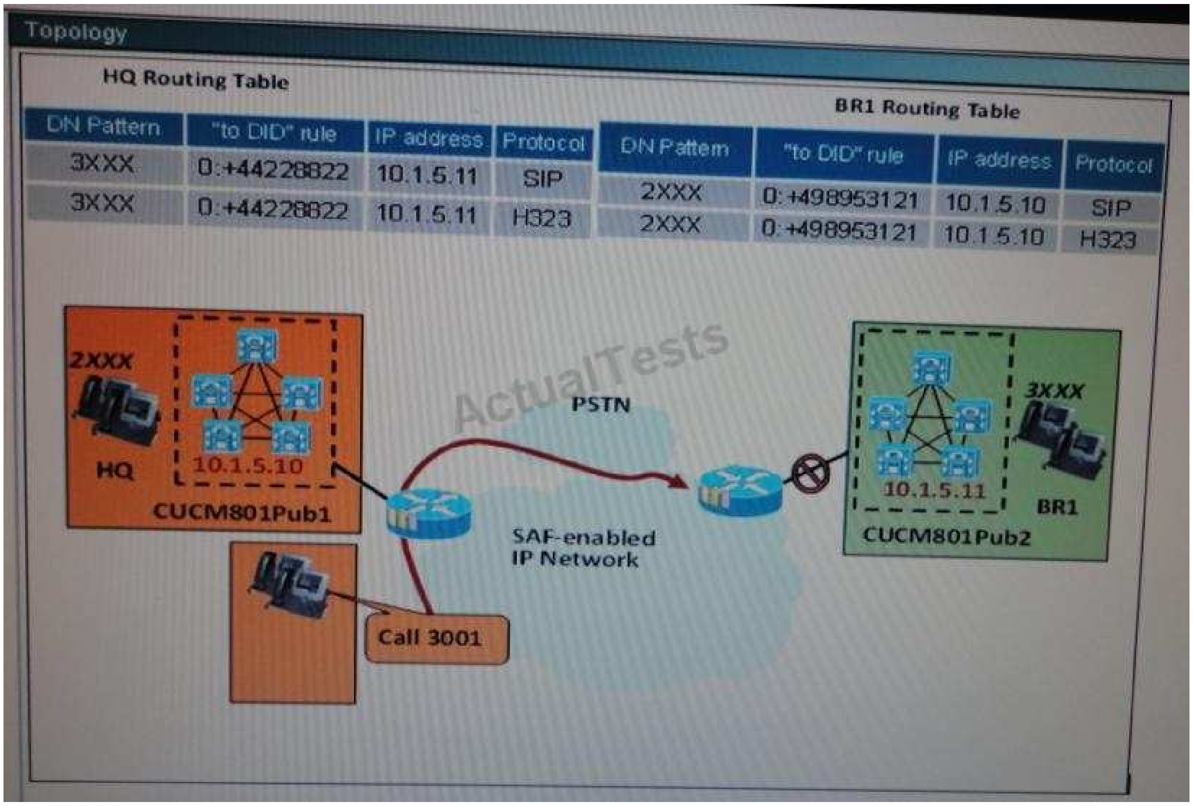

When Alice, at extension 2001, places a call to Bob, at extension 3001, Alice hears her own voice reflected. Which type of echo is this classified as?

A. talker echo
B. listener echo
C. tail circuit echo
D. front-end circuit echo

**Correct Answer:** A
**Section: (none)**
**Explanation**

**Explanation/Reference:**

**QUESTION 24**
Refer to the topology exhibit.



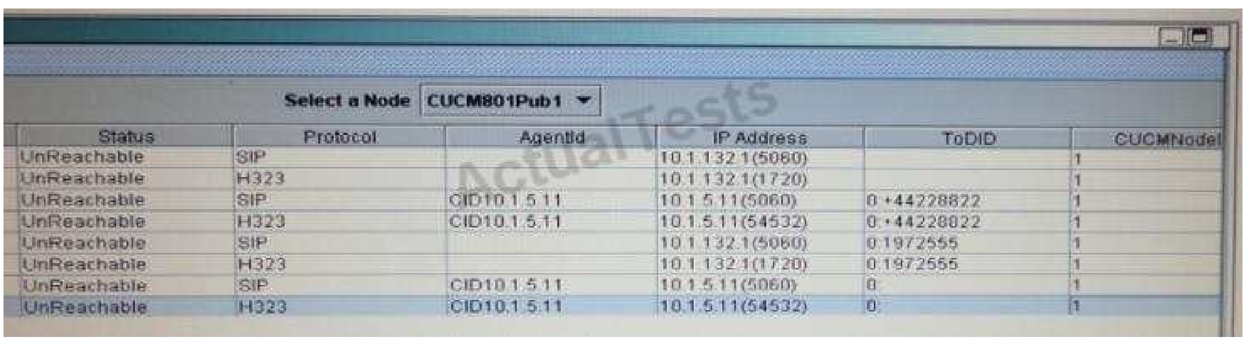


HQ Routing Table

| DN Pattern | "to DID" rule | IP address | Protocol |
|---|---|---|---|
| 3XXX | 0:+44228822 | 10.1.5.11 | SIP |
| 3XXX | 0:+44228822 | 10.1.5.11 | H323 |

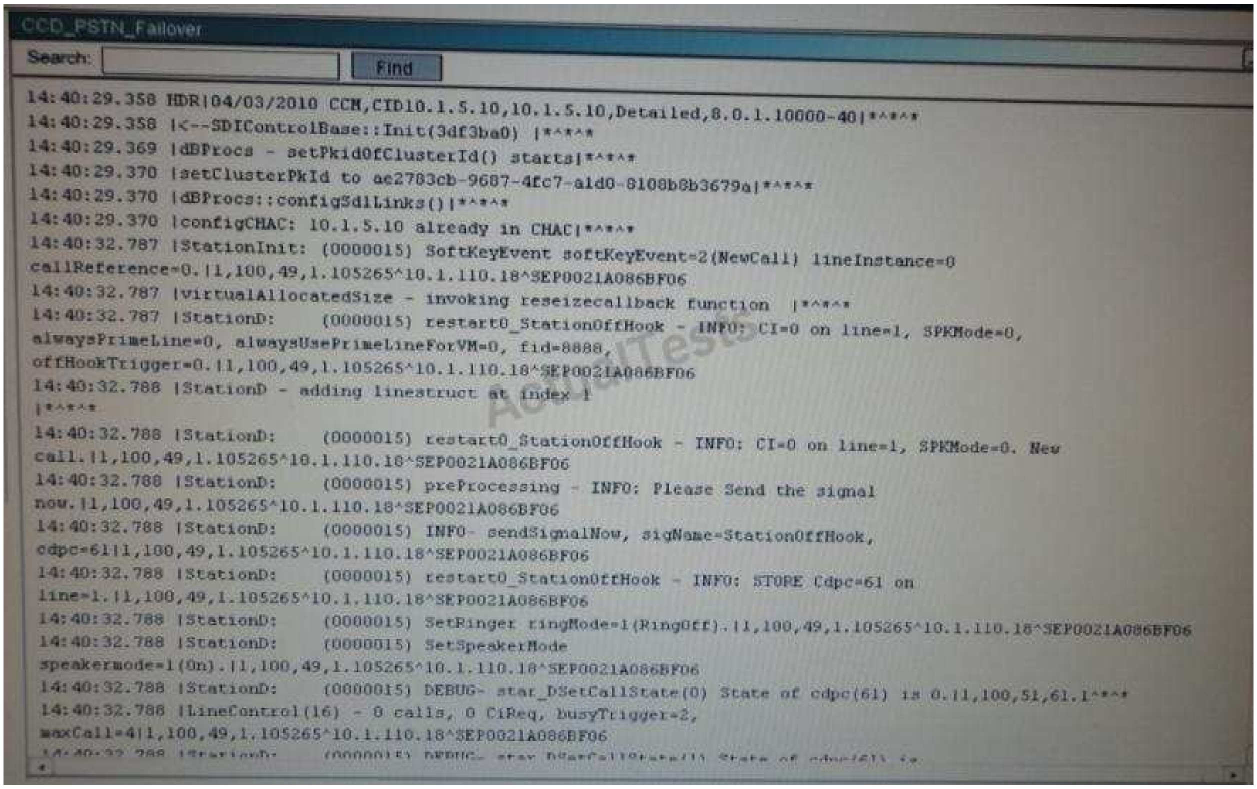
BR1 Routing Table

| DN Pattern | "to DID" rule | IP address | Protocol |
|---|---|---|---|
| 2XXX | 0:+498953121 | 10.1.5.10 | SIP |
| 2XXX | 0:+498953121 | 10.1.5.10 | H323 |

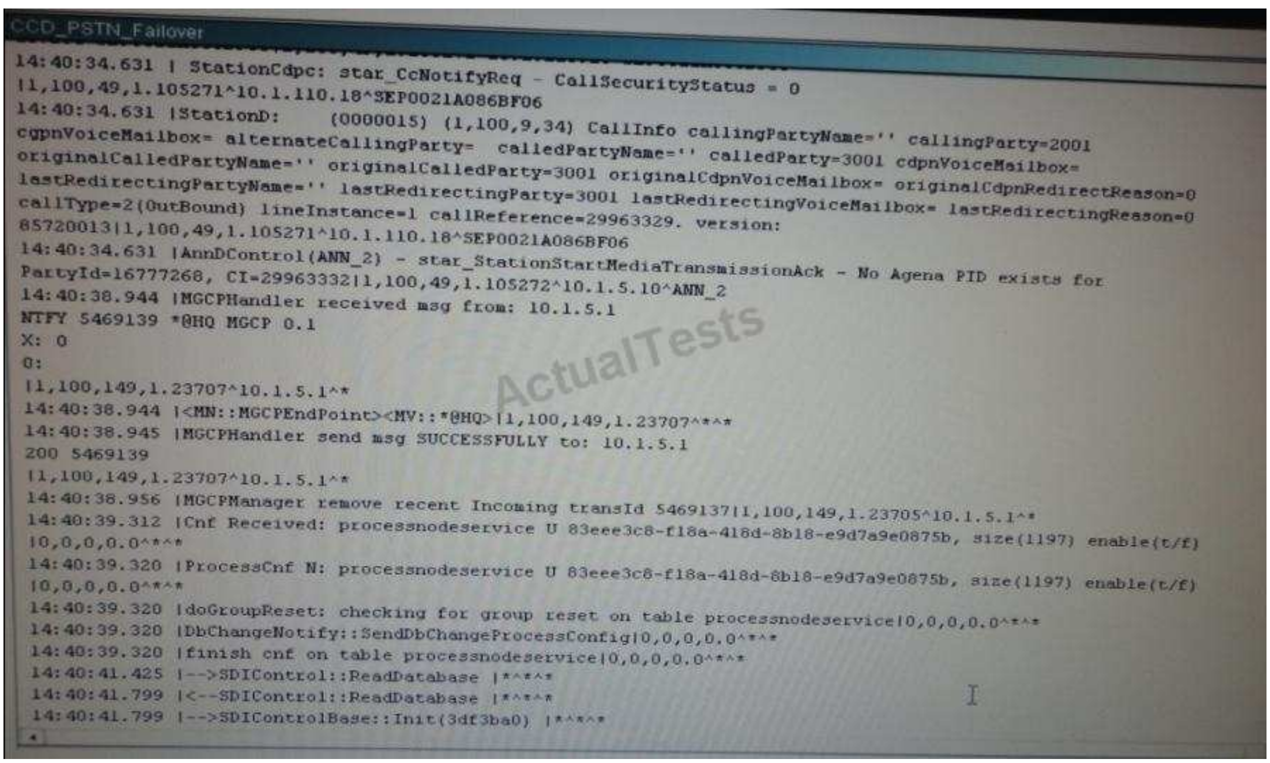


Select a Node CUCM801Pub1

| Status | Protocol | AgentId | IP Address | ToDID | CUCMNodel |
|---|---|---|---|---|---|
| UnReachable | SIP | | 10.1.132.1(5060) | | 1 |
| UnReachable | H323 | | 10.1.132.1(1720) | | 1 |
| UnReachable | SIP | CID10.1.5.11 | 10.1.5.11(5060) | 0:+44228822 | 1 |
| UnReachable | H323 | CID10.1.5.11 | 10.1.5.11(54532) | 0:+44228822 | 1 |
| UnReachable | SIP | | 10.1.132.1(5060) | 0:1972555 | 1 |
| UnReachable | H323 | | 10.1.132.1(1720) | 0:1972555 | 1 |
| UnReachable | SIP | CID10.1.5.11 | 10.1.5.11(5060) | 0: | 1 |
| UnReachable | H323 | CID10.1.5.11 | 10.1.5.11(54532) | 0: | 1 |



```
CCD_PSTN_Failover
Search: [          ] Find
14:40:29.358 HDR|04/03/2010 CCM,CID10.1.5.10,10.1.5.10,Detailed,8.0.1.10000-40|*^*^*
14:40:29.358 |<--SDIControlBase::Init(3df3ba0) |*^*^*
14:40:29.369 |dBProcs - setPkidOfClusterId() starts|*^*^*
14:40:29.370 |setClusterPkId to ae2783cb-9687-4fc7-a1d0-8108b8b3679a|*^*^*
14:40:29.370 |dBProcs::configSdlLinks()|*^*^*
14:40:29.370 |configCHAC: 10.1.5.10 already in CHAC|*^*^*
14:40:32.787 |StationInit: (0000015) SoftKeyEvent softKeyEvent=2(NewCall) lineInstance=0
callReference=0.|1,100,49,1.105265^10.1.110.18^SEP0021A086BF06
14:40:32.787 |virtualAllocatedSize - invoking reseizecallback function  |*^*^*
14:40:32.787 |StationD:     (0000015) restart0_StationOffHook - INFO: CI=0 on line=1, SPKMode=0,
alwaysPrimeLine=0, alwaysUsePrimeLineForVM=0, fid=8888,
offHookTrigger=0.|1,100,49,1.105265^10.1.110.18^SEP0021A086BF06
14:40:32.788 |StationD - adding linestruct at index 1
|*^*^*
14:40:32.788 |StationD:     (0000015) restart0_StationOffHook - INFO: CI=0 on line=1, SPKMode=0. New
call.|1,100,49,1.105265^10.1.110.18^SEP0021A086BF06
14:40:32.788 |StationD:     (0000015) preProcessing - INFO: Please Send the signal
now.|1,100,49,1.105265^10.1.110.18^SEP0021A086BF06
14:40:32.788 |StationD:     (0000015) INFO- sendSignalNow, sigName=StationOffHook,
cdpc=61|1,100,49,1.105265^10.1.110.18^SEP0021A086BF06
14:40:32.788 |StationD:     (0000015) restart0_StationOffHook - INFO: STORE Cdpc=61 on
line=1.|1,100,49,1.105265^10.1.110.18^SEP0021A086BF06
14:40:32.788 |StationD:     (0000015) SetRinger ringMode=1(RingOff).|1,100,49,1.105265^10.1.110.18^SEP0021A086BF06
14:40:32.788 |StationD:     (0000015) SetSpeakerMode
speakermode=1(On).|1,100,49,1.105265^10.1.110.18^SEP0021A086BF06
14:40:32.788 |StationD:     (0000015) DEBUG- star_DSetCallState(0) State of cdpc(61) is 0.|1,100,51,61.1^*^*
14:40:32.788 |LineControl(16) - 0 calls, 0 CiReq, busyTrigger=2,
maxCall=4|1,100,49,1.105265^10.1.110.18^SEP0021A086BF06
```



```
CCD_PSTN_Failover
14:40:34.631 | StationCdpc: star_CcNotifyReq - CallSecurityStatus = 0
|1,100,49,1.105271^10.1.110.18^SEP0021A086BF06
14:40:34.631 |StationD:     (0000015) (1,100,9,34) CallInfo callingPartyName='' callingParty=2001
cgpnVoiceMailbox= alternateCallingParty= calledPartyName='' calledParty=3001 cdpnVoiceMailbox=
originalCalledPartyName='' originalCalledParty=3001 originalCdpnVoiceMailbox= originalCdpnRedirectReason=0
lastRedirectingPartyName='' lastRedirectingParty=3001 lastRedirectingVoiceMailbox= lastRedirectingReason=0
callType=2(OutBound) lineInstance=1 callReference=29963329. version:
85720013|1,100,49,1.105271^10.1.110.18^SEP0021A086BF06
14:40:34.631 |AnnDControl(ANN_2) - star_StationStartMediaTransmissionAck - No Agena PID exists for
PartyId=16777268, CI=29963332|1,100,49,1.105272^10.1.5.10^ANN_2
14:40:38.944 |MGCPHandler received msg from: 10.1.5.1
NTFY 5469139 *@HQ MGCP 0.1
X: 0
O:
|1,100,149,1.23707^10.1.5.1^*
14:40:38.944 |<MN::MGCPEndPoint><MV::*@HQ>|1,100,149,1.23707^*^*
14:40:38.945 |MGCPHandler send msg SUCCESSFULLY to: 10.1.5.1
200 5469139
|1,100,149,1.23707^10.1.5.1^*
14:40:38.956 |MGCPManager remove recent Incoming transId 5469137|1,100,149,1.23705^10.1.5.1^*
14:40:39.312 |Cnf Received: processnodeservice U 83eee3c8-f18a-418d-8b18-e9d7a9e0875b, size(1197) enable(t/f)
|0,0,0,0.0^*^*
14:40:39.320 |ProcessCnf N: processnodeservice U 83eee3c8-f18a-418d-8b18-e9d7a9e0875b, size(1197) enable(t/f)
|0,0,0,0.0^*^*
14:40:39.320 |doGroupReset: checking for group reset on table processnodeservice|0,0,0,0.0^*^*
14:40:39.320 |DbChangeNotify::SendDbChangeProcessConfig|0,0,0,0.0^*^*
14:40:39.320 |finish cnf on table processnodeservice|0,0,0,0.0^*^*
14:40:41.425 |-->SDIControl::ReadDatabase |*^*^*
14:40:41.799 |<--SDIControl::ReadDatabase |*^*^*
14:40:41.799 |-->SDIControlBase::Init(3df3ba0) |*^*^*
```

When the Cisco Unified Communications Manager at the BR1 site loses connectivity to the SAF network, the call to extension 3001 fails to reroute via the PSTN. Examine the SDI trace in the exhibit. What is the most likely cause of the failure?

A. The full E.164 number for extension 3001 could not be constructed

B. The Cisco Unified Communications Manager at the HQ site was unable to determine the ToDID number to extension 3001.
C. The calling phone line CSS does not have access to the V route pattern.
D. CCD requesting service does not have the correct partition configured for CCD learned patterns
E. The AAR CSS for the HQ phone does not have access to the \\ route pattern.

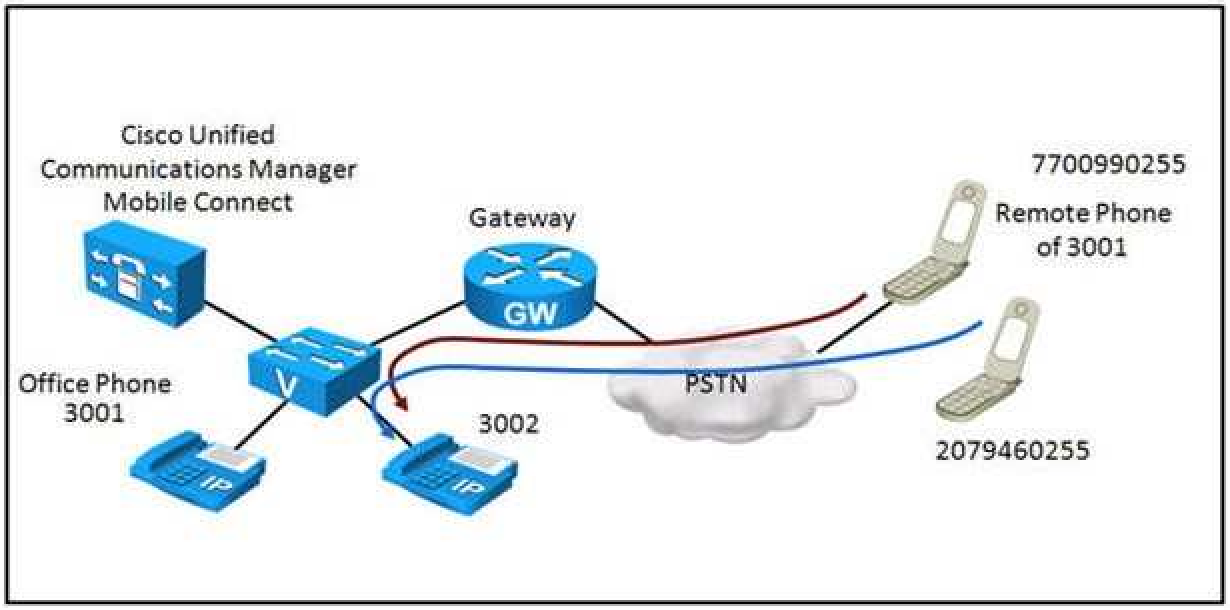
**Correct Answer:** E
**Section: (none)**
**Explanation**

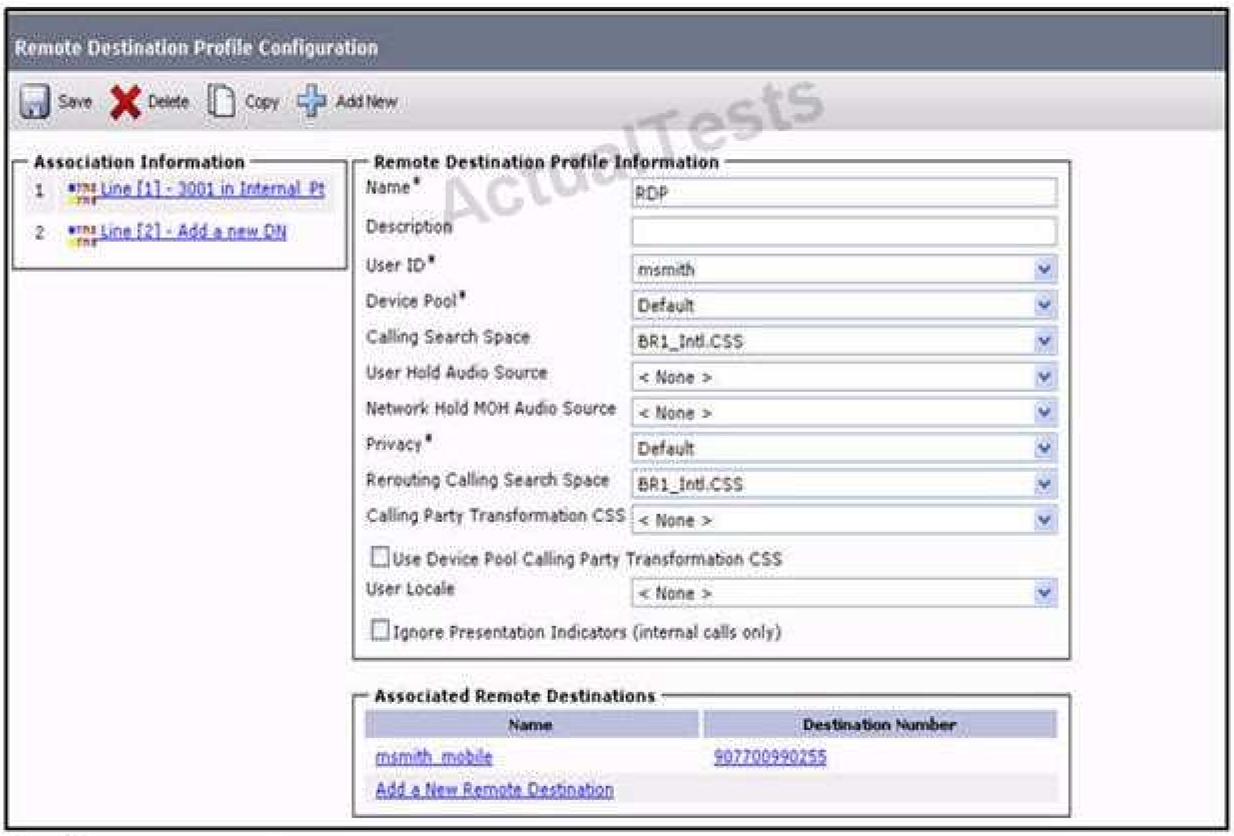
**Explanation/Reference:**
**Explanation**- Logs states that AAR CSS for HQ phone doesn’t have the access to the route pattern.

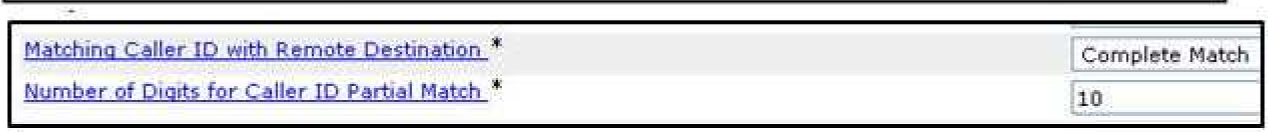
**QUESTION 25**
## Refer to the exhibit.

All directory number extensions are assigned to the Internal_LP partition. All phones are configured with only a line CSS, BR1 Jntl.CSS. Assuming the gateway has been configured correctly. When the two PSTN callers place a call to 02288223002, which of the following statements is true?

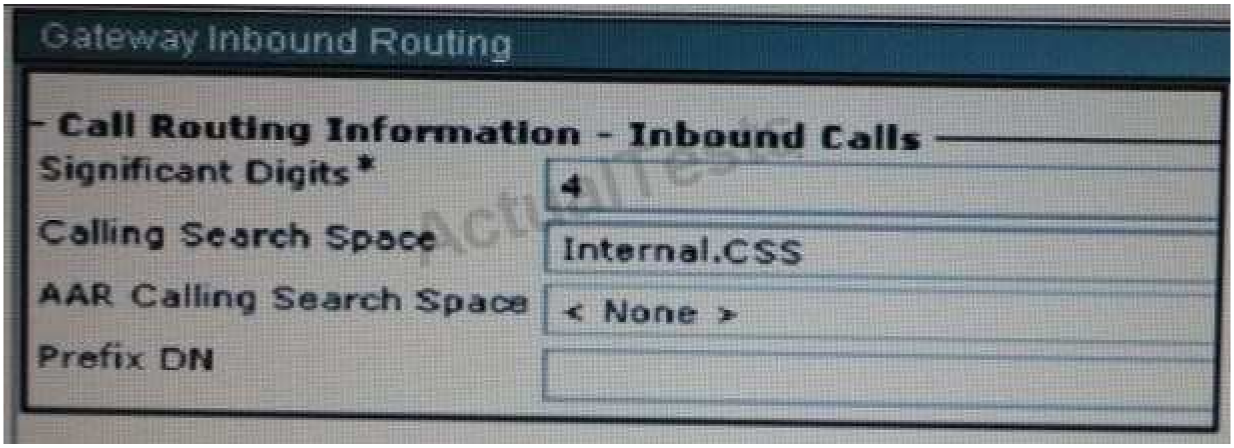
**86 (exhibit):**


Cisco Unified
Communications Manager
Mobile Connect
Gateway
GW
7700990255
Remote Phone
of 3001
Office Phone
3001
PSTN
3002
2079460255

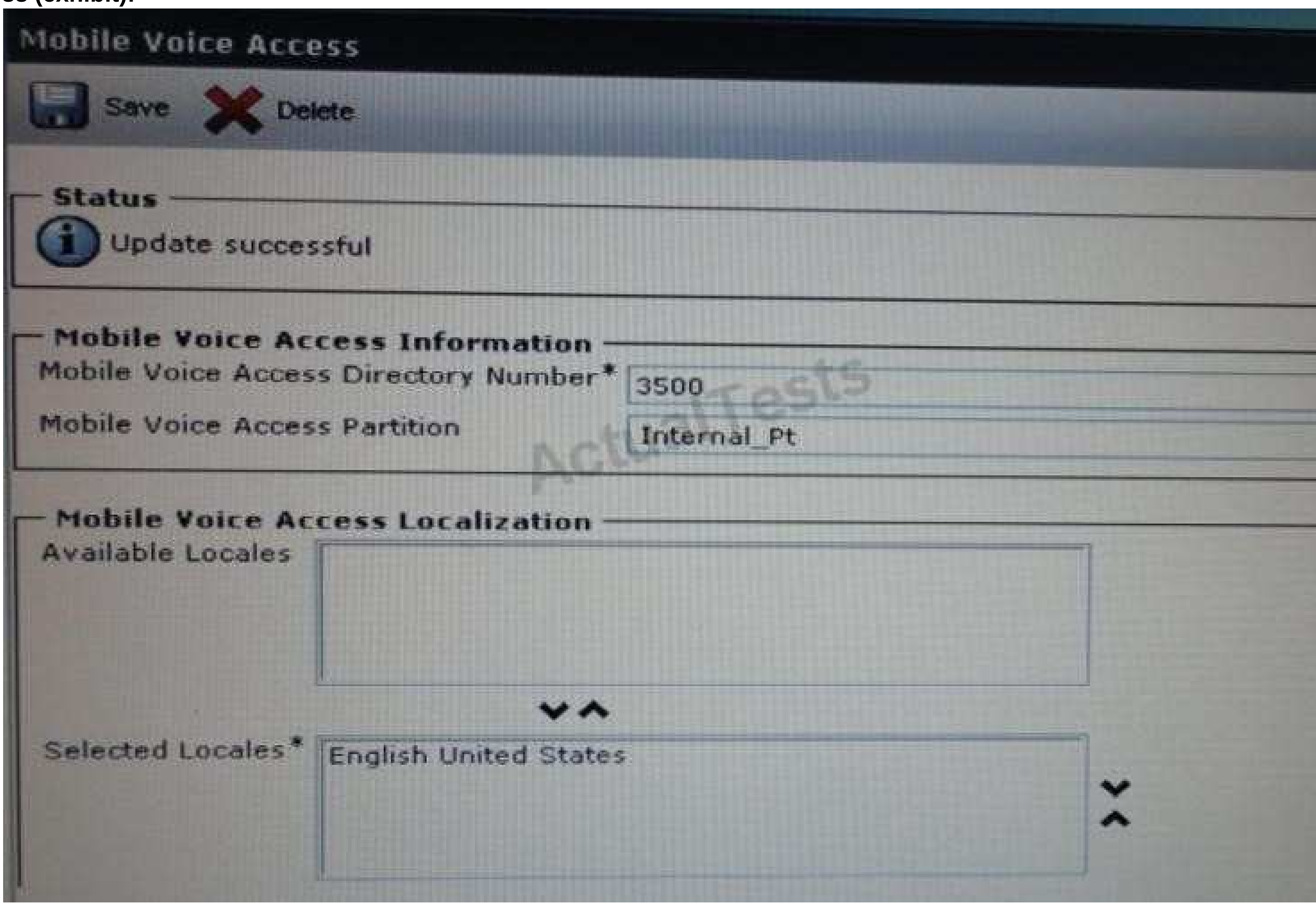

Remote Destination Profile Configuration
Save
Delete
Copy
Add New
Association Information
1 Line [1] - 3001 in Internal_Pt
2 Line [2] - Add a new DN
Remote Destination Profile Information
Name* RDP
Description
User ID* msmith
Device Pool* Default
Calling Search Space BR1_Intl.CSS
User Hold Audio Source < None >
Network Hold MOH Audio Source < None >
Privacy* Default
Rerouting Calling Search Space BR1_Intl.CSS
Calling Party Transformation CSS < None >
Use Device Pool Calling Party Transformation CSS
User Locale < None >
Ignore Presentation Indicators (internal calls only)
Associated Remote Destinations
Name
Destination Number
msmith_mobile
907700990255
Add a New Remote Destination


Matching Caller ID with Remote Destination *
Complete Match
Number of Digits for Caller ID Partial Match *
10

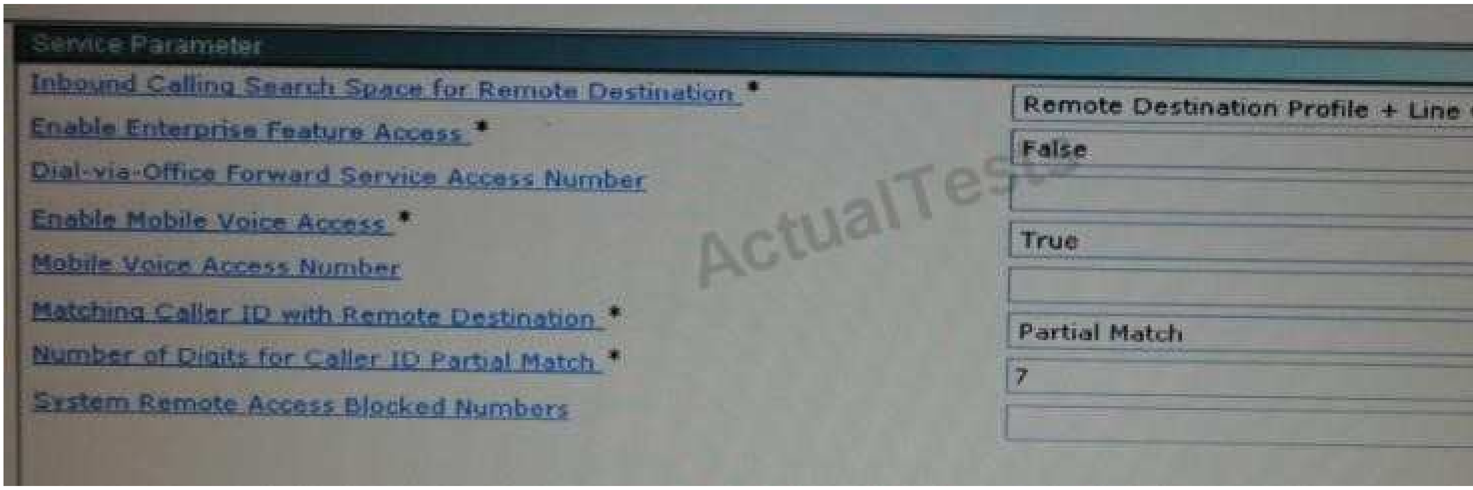
**87 (exhibit):**



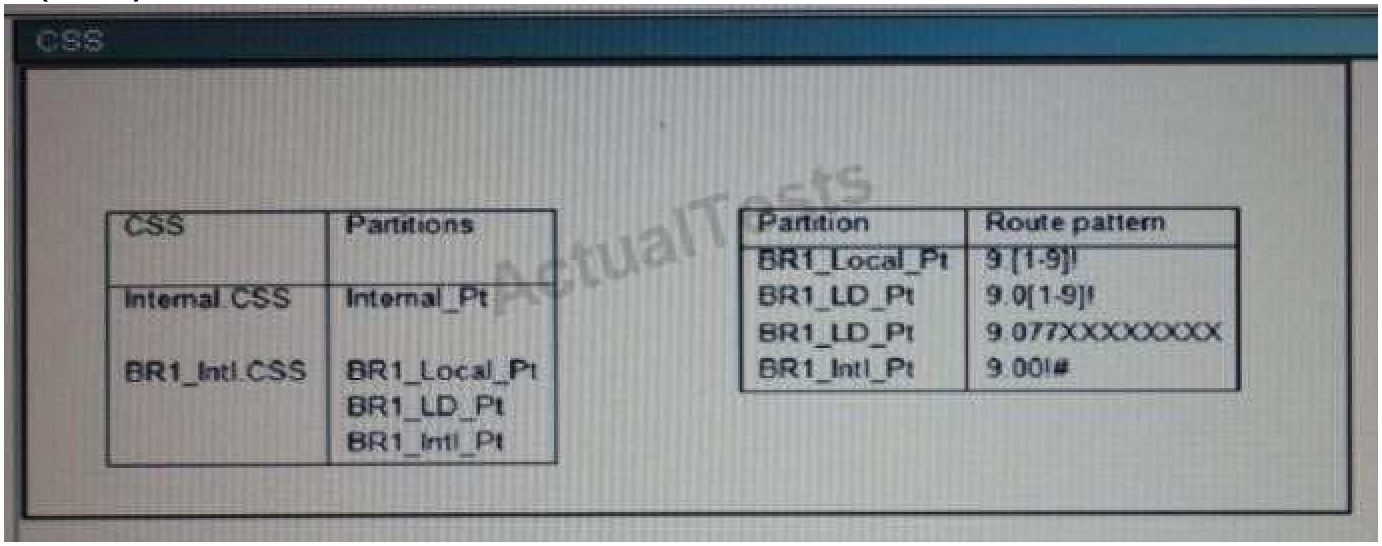

**88 (exhibit):**




**89 (exhibit):**




**90 (exhibit):**




| CSS | Partitions |
|---|---|
| Internal.CSS | Internal_Pt |
| BR1_Intl.CSS | BR1_Local_Pt<br>BR1_LD_Pt<br>BR1_Intl_Pt |

| Partition | Route pattern |
|---|---|
| BR1_Local_Pt | 9.[1-9]! |
| BR1_LD_Pt | 9.0[1-9]! |
| BR1_LD_Pt | 9.077XXXXXXXX |
| BR1_Intl_Pt | 9.00!# |

A. The caller at 7700990255 will succeed. The caller at 2079460255 will fail

B. The caller at 2079460255 will succeed. The caller at 7700990255 will fail

C. Both calls will succeed.

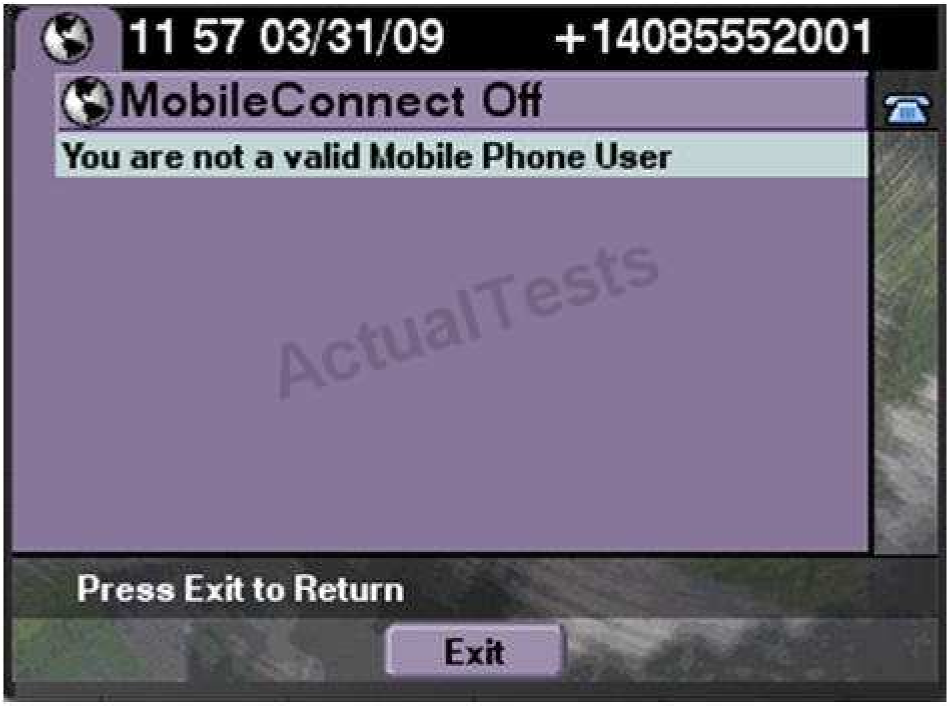
D. Both calls will fail

E. Both calls will fail. However, the caller at 7700990255 can still reach 3001 because 7700990255 is the Remote Destination of 3001

**Correct Answer:** B
**Section: (none)**
**Explanation**

**Explanation/Reference:**
**Explanation:** Both calls will succeedThe CSS of Gateway Inbound Routing an Mobile Voice Access ist Internal.css. The phones should thus be accessible for both.The remote destination can use the Mobile Voice access. Matching Caller ID = Partial Match and the Calling Number = msmith mobil number.

**QUESTION 26**
**Refer to the exhibit.**



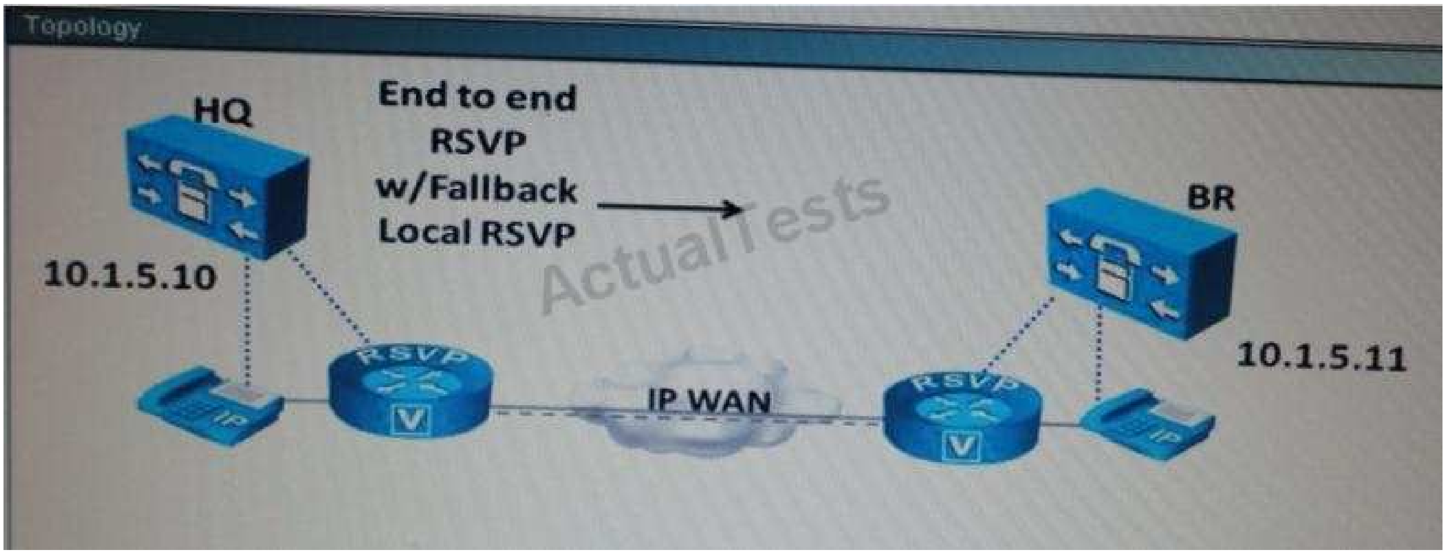

Which course of action will resolve the Mobile Connect issues that are shown in the exhibit?

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.

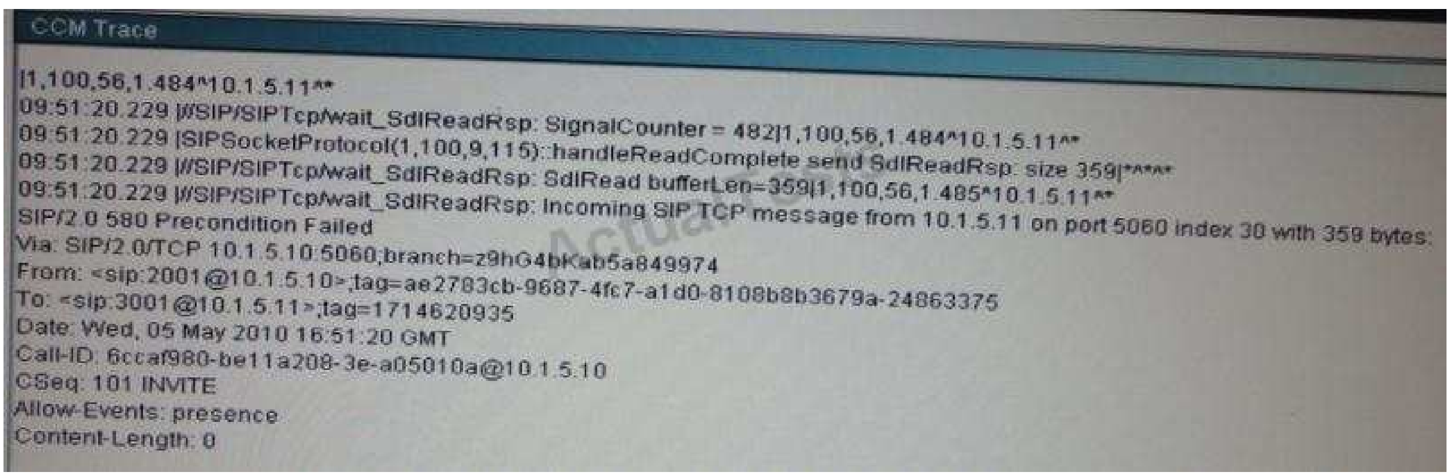
**Correct Answer:** C
**Section: (none)**
**Explanation**

**Explanation/Reference:**
**Explanation: Link**-https://learningnetwork.cisco.com/docs/DOC-4140

**QUESTION 27**
**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.



```
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
```

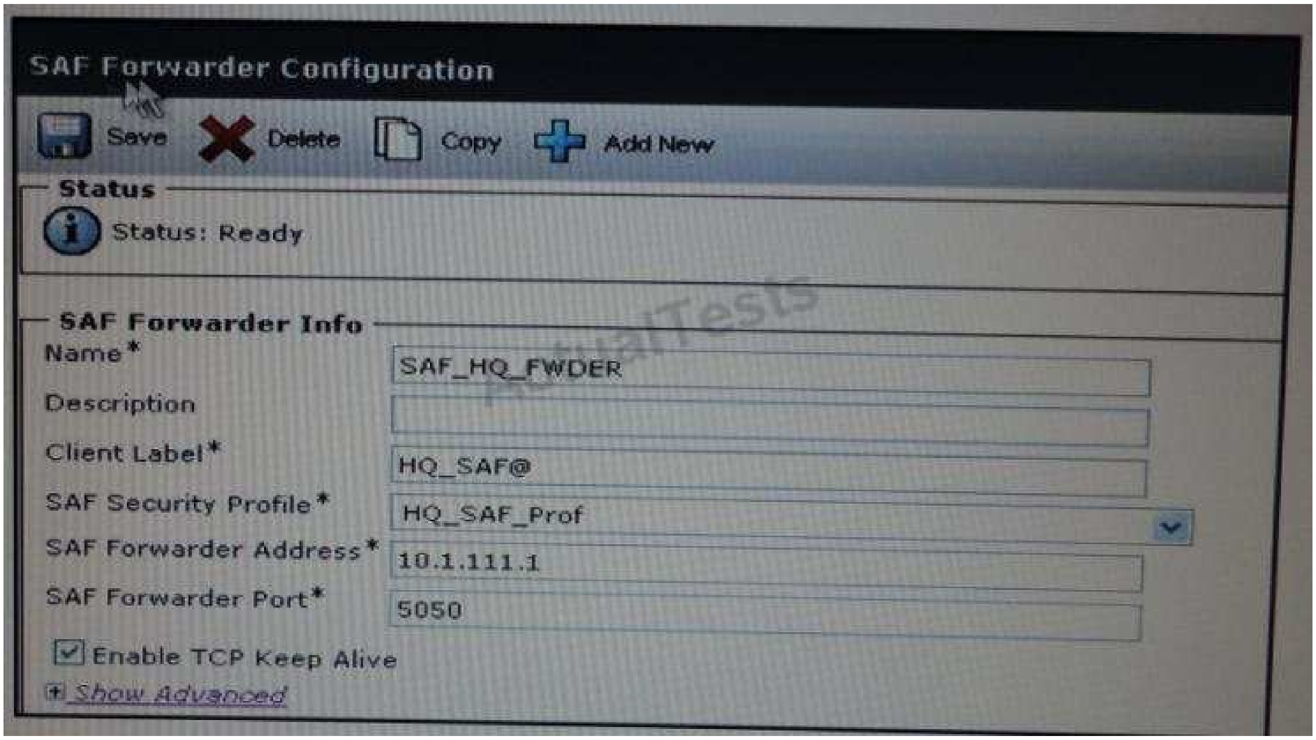
Refer to the SDI trace. Which statement about the call is true?

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.

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

**Explanation/Reference:**
**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 See TVOICE V II 6-152

**QUESTION 28**
Refer to the exhibit.




Which configuration is the correct Cisco IOS configuration that corresponds to the Cisco SAF Forwarder that is shown in the exhibit?

**A (exhibit):**

A.

```
interface Loopback0
 ip address 10.1.111.1 255.255.255.0
!
router eigrp SAF
 !
 service-family ipv4 autonomous-system 1
  !
  topology base
   external-client HQ_SAF
  exit-sf-topology
 exit-service-family
!
service-family external-client listen ipv4 5050
 external-client HQ_SAF basename
  username SAFUSER
  password SAFPASSWORD
  keepalive 3599999
```

**B (exhibit):**

B.

```
interface Loopback0
 ip address 10.1.111.1 255.255.255.0
!
router eigrp SAF
 !
 service-family ipv4 autonomous-system 1
  !
  topology base
   external-client HQ_SAF
  exit-sf-topology
 exit-service-family
!
service-family external-client listen ipv4 5050
 external-client HQ_SAF basename
  username SAFUSER
  password SAFPASSWORD
  keepalive 3599999
```

**C (exhibit):**

C.

```
interface Loopback0
 ip address 10.1.111.1 255.255.255.0
!
router eigrp SAF
 !
 service-family ipv4 autonomous-system 1
  !
  topology base
   external-client HQ_SAF
  exit-sf-topology
 exit-service-family
!
service-family external-client listen ipv4 5050
 external-client HQ_SAF basename
  username SAFUSER
  password SAFPASSWORD
  keepalive 3599999
```



**D (exhibit):**

D.

```
interface Loopback0
 ip address 10.1.111.1 255.255.255.0
!
router eigrp SAF
 !
 service-family ipv4 autonomous-system 1
  !
  topology base
   external-client HQ_SAF
  exit-sf-topology
 exit-service-family
!
service-family external-client listen ipv4 5050
 external-client HQ_SAF basename
  username SAFUSER
  password SAFPASSWORD
  keepalive 3599999
```



A. Exhibit A
B. Exhibit B
C. Exhibit C
D. Exhibit D

**Correct Answer:** A
**Section: (none)**
**Explanation**

**Explanation/Reference:**

**QUESTION 29**
Cisco Unified Communications Manager failed to register with the Cisco SAF Forwarder. Assuming that the Cisco IOS SAF Forwarder is configured correctly, which minimum configuration would be needed on Cisco Unified Communications Manager to test registration?

A. SAF trunk, SAF security profile, Cisco SAF Forwarder, and CCD advertising service
B. SAF trunk, SAF security profile, Cisco SAF Forwarder, and CCD requesting service
C. SAF trunk, SAF security profile, Cisco SAF Forwarder, CCD requesting service, and CCD advertising service
D. SAF trunk, SAF security profile, and Cisco SAF Forwarder
E. SAF trunk, CCD requesting service, and CCD advertising service

**Correct Answer:** B
**Section: (none)**
**Explanation**

**Explanation/Reference:**

**QUESTION 30**
Which two types of Cisco Unified Communications Manager trace files contain Call Processing information that is helpful for troubleshooting outbound and inbound calling issues?(Choose two)

A. Cisco Unified Communications Manager syslog trace
B. Cisco Unified Communications Manager Dialed Number Analyzer trace
C. Real Time Monitoring Tool Processes trace
D. Cisco Unified Communications Manager SDL trace
E. Cisco Unified Communications Manager Log4Jtrace
F. Cisco Unified Communications Manager SDI trace

**Correct Answer:** DF
**Section: (none)**
**Explanation**

**Explanation/Reference:**
**Explanation**-The Trace and Alarm tools work together. You configure trace and alarm settings for Cisco Unified CallManager services. A Cisco TAC engineer receives the results. You can direct alarms to the Microsoft Windows 2000 Event Viewer, CiscoWorks2000 Syslog,

system diagnostic interface (SDI) or signal distribution layer (SDL) trace log files, or to all destinations. You can base traces for Cisco Unified CallManager services on debug levels, specific trace fields, and Cisco Unified CallManager devices such as phones or gateways. You can perform a trace on the alarms that are sent to the SDI or SDL trace log files.

**Link-**http://www.cisco.com/en/US/docs/voice_ip_comm/cucm/service/4_2_3/ccmsrva/satracea.html

**QUESTION 31**
**Refer to the exhibit.**

```
*Mar 24 16:17:54.190: ISDN Se0/0/0: 15 Q931: RX <- SETUP pd = 8 callref = 0x00AA
  Beaere Capability i = 0x8090A3

     Standard = CCITT
     Transfer Capability = Spee
     Transfer Mode = Circuit
     Transfer Rate = 64 kbit/s

 Channel ID i = 0xA98381
 Exclusive, Channel 1
 Progress Ind i = 0x8183 - Origination address is non-ISDN
 Calling Party Number i = 0x1180, '4940302156001'
 Plan:ISDN, Type:International
 Called Party Number i = 0x81, '2288223001'
 Plan:ISDN, Type:Unknown
 *Mar 24 16:17:54:210: ISDN Se0/0/0 15 Q931: TX-> RELEASE_COMP pd=8 callref=
 0x80AA
 Cause i = 0x8081 = Unallocated/unaligned number
```



The exhibit shows the output of debug isdn q931. An inbound PSTN call was received by an H.323 gateway that is configured in Cisco Unified Communications Manager. The call failed to ring extension 3001. If the phone at extension 3001 is registered and reachable through the gateway inbound CSS, which four actions can resolve this issue? (Choose four.)

A. Change the significant digits for inbound calls to 4 on the gateway configuration in Cisco Unified Communications Manager.
B. Configure the digit strip 4 on H.323 gateway in the Incoming Called Party Settings in Cisco Unified Communications Manager.
C. Configure a translation pattern in Cisco Unified Communications Manager that can be accessed by the phone CSS to truncate the called number to four digits
D. Configure a called-party transformation CSS on the gateway in Cisco Unified Communications Manager that includes a pattern that transforms the number from ten digits to four digits
E. Configure a voice translation profile in the H.323 Cisco IOS gateway with a voice translation rule that truncates the number from ten digits to four.
F. Configure the Cisco IOS command num-exp 2288223001 3001 on the gateway ISDN interface.

**Correct Answer:** ABDE
**Section: (none)**
**Explanation**

**Explanation/Reference:**

**QUESTION 32**
**Refer to the exhibit.**

```
HQ-1#show policy-map interface serial 0/1/0.101

...truncated...

        Class-map: voice (match-all)
          2073 packets, 183124 bytes
          5 minute offered rate 98701 bps, drop rate 14763 bps
          Match: ip dscp ef (46)
          Priority: 106 kbps, burst bytes 2650, b/w exceed drops: 21
...truncated...
        Class-map: signaling (match-any)
          50232 packets, 2692896 bytes
          5 minute offered rate 17254 bps, drop rate 851 bps
          Match: ip dscp cs3 (24)
            50232 packets, 2692896 bytes
            5 minute rate 17254 bps
          Queueing
          queue limit 64 packets
          (queue depth/total drops/no-buffer drops) 59/1567/239
          (pkts output/bytes output) 50232/2692896
          bandwidth 24 kbps
```

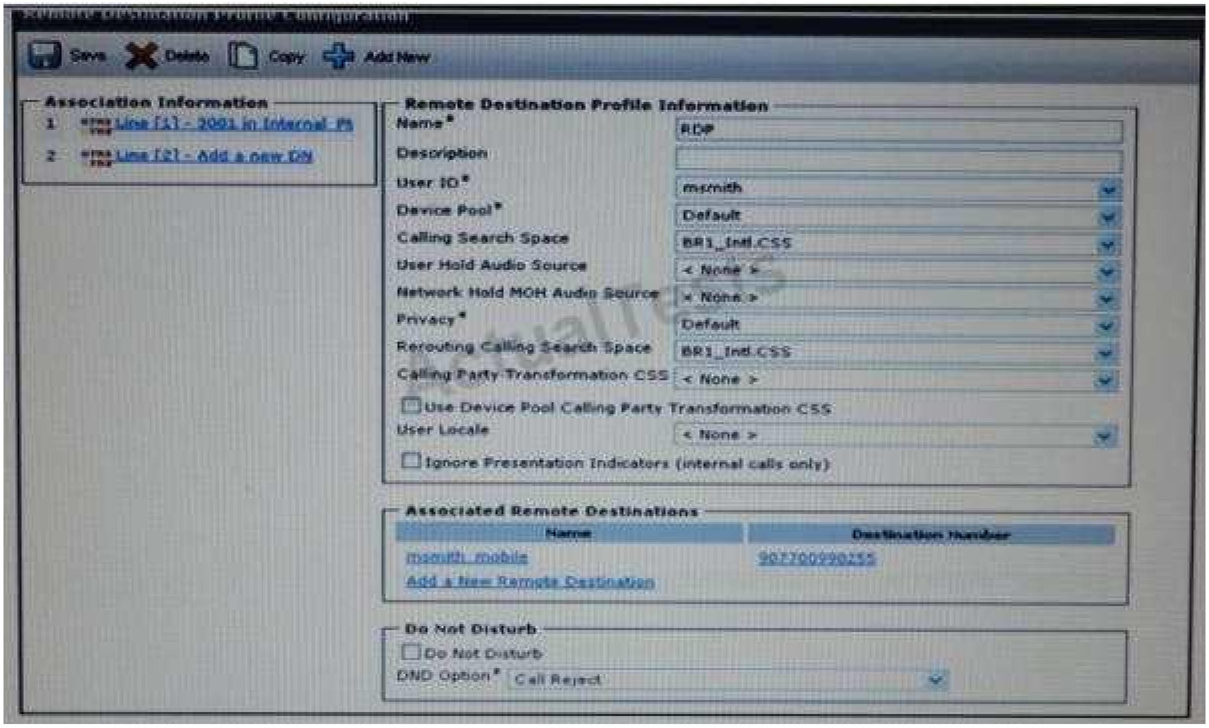
Assume a centralized Cisco Unified Communications Manager topology. From the output of the show policy-map interface command, it can be seen that the voice traffic is experiencing drops. The router is configured for low latency queuing and is located at the central site. Which course of action would rectify this issue?

A. Configure a higher rate bandwidth codec on all voice calls
B. Configure Call Admission Control to limit the maximum calls that are sent to the priority queue.
C. Place the signaling traffic in the priority queue
D. split some of the excess voice traffic into other queues K avoid oversubscription of the priority queue.

**Correct Answer:** B
**Section: (none)**
**Explanation**

**Explanation/Reference:**
**Explanation: Link**-http://www.iphelp.ru/faq/27/ch07lev1sec3.html

**QUESTION 33**
**Refer to the exhibit.**



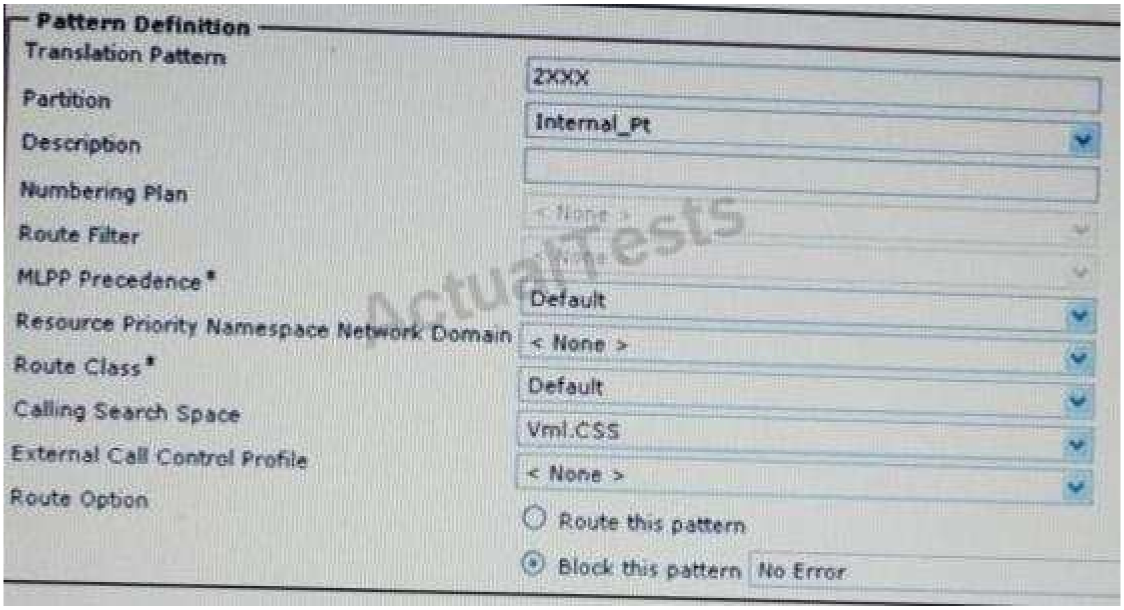

When a call is placed to extension 3001, the associated remote destination 907700990255 does not ring. Which CSS is responsible for extending the call to the remote destination?

A. The Calling Search Space is responsible
B. The Rerouting Calling Search Space is responsible
C. The Calling Party Transformation CSS is responsible
D. The Calling Search Space and the Rerouting Calling Search Space are concatenated to determine where the call should be extended

**Correct Answer:** B
**Section: (none)**
**Explanation**

**Explanation/Reference:**

**QUESTION 34**
**Refer to the exhibit.**



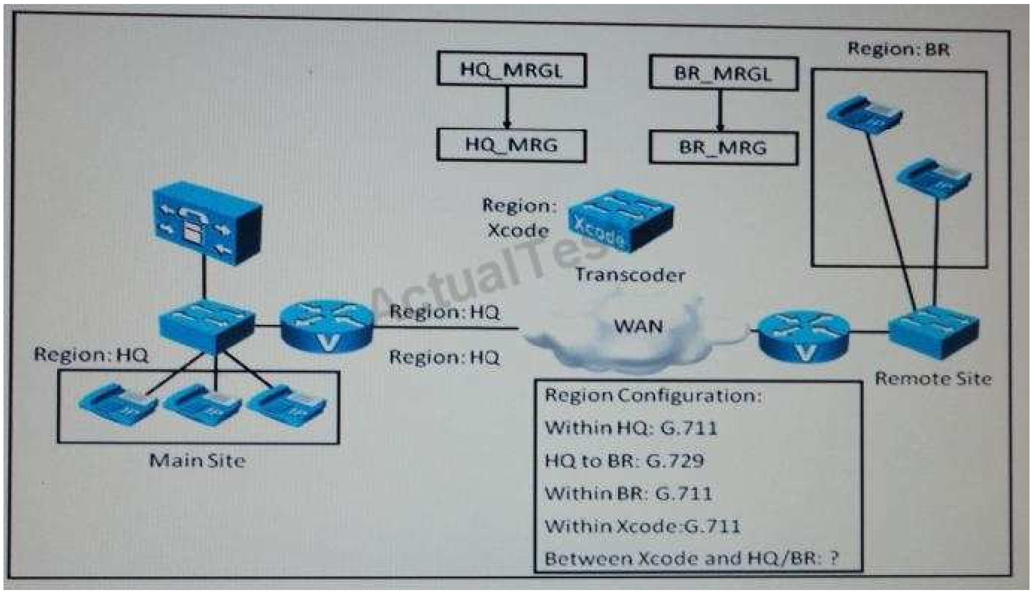

Refer to the exhibit. All phones are placed in the Internal_Pt partition. The CSS for all phones contains the partition Internal_Pt, and Vml.CSS contains the voice-mail hunt pilot. When a call is placed from extension 2001 to 2002, which statement is true?

A. Extension 2002 will ring.
B. The call will be blocked.
C. The call will be answered by voice mail.
D. Extension 2002 will ring, and if the call is not answered, the call will match the translation pattern and then be blocked.
E. Extension 2002 will ring, and if the call is not answered, the call will match the translation pattern and then be forwarded to voice mail

**Correct Answer:** A
**Section: (none)**
**Explanation**

**Explanation/Reference:**

**QUESTION 35**
**Refer to the exhibit.**




The BR phones only support G.711 codec. When a call from HQ is placed to a BR phone, the call fails. The network administrator would like to use a hardware transcoder that is only reachable via the WAN from the HQ and BR sites. How should the transcoder region be configured?

A. Configure G.711 codec from HQ to Xcode and G.711 from BR to Xcode
B. Configure G.729 codec from HQ to Xcode and G.729 from BR to Xcode
C. Configure G.711 codec from HQ to Xcode and G.729 from BR to Xcode
D. Configure G.729 codec from HQ to Xcode and G.711 from BR to Xcode

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

**Explanation/Reference:**
**Explanation**- Region configuration in diagram explains the statement D, G.729 codec from Hq and G.711 from B

**QUESTION 36**
Which two statements indicate something that can cause an IP phone to fail roaming when device mobility has been configured? (Choose two)

A. Device Mobility Mode is set to Off in the Cisco Unified Communications Manager service parameters while the device mobility configuration on the phone is set to default.
B. No device mobility groups have been configured.
C. No locations have been configured and assigned to the device pools.
D. No physical locations have been configured and assigned to the device pools.
E. No device mobility-related information settings were configured under the device pools.

**Correct Answer:** AD
**Section: (none)**
**Explanation**

**Explanation/Reference:**

**QUESTION 37**

Which tool allows the administrator to analyze call routing in Cisco Unified Communications Manager without physically placing a call?

A. Cisco Unified Communications Manager Dialed Number Analyzer.
B. Cisco IOS Gateway debug commands.
C. Cisco Unified Communications Manager RTMT trace output.
D. base configuration information for this user that specifies Class of Restriction, Partition and Calling Search Space information
E. Cisco Unified Communications Manager Serviceability tools
F. Cisco Unified Communications Manager OS Administration.

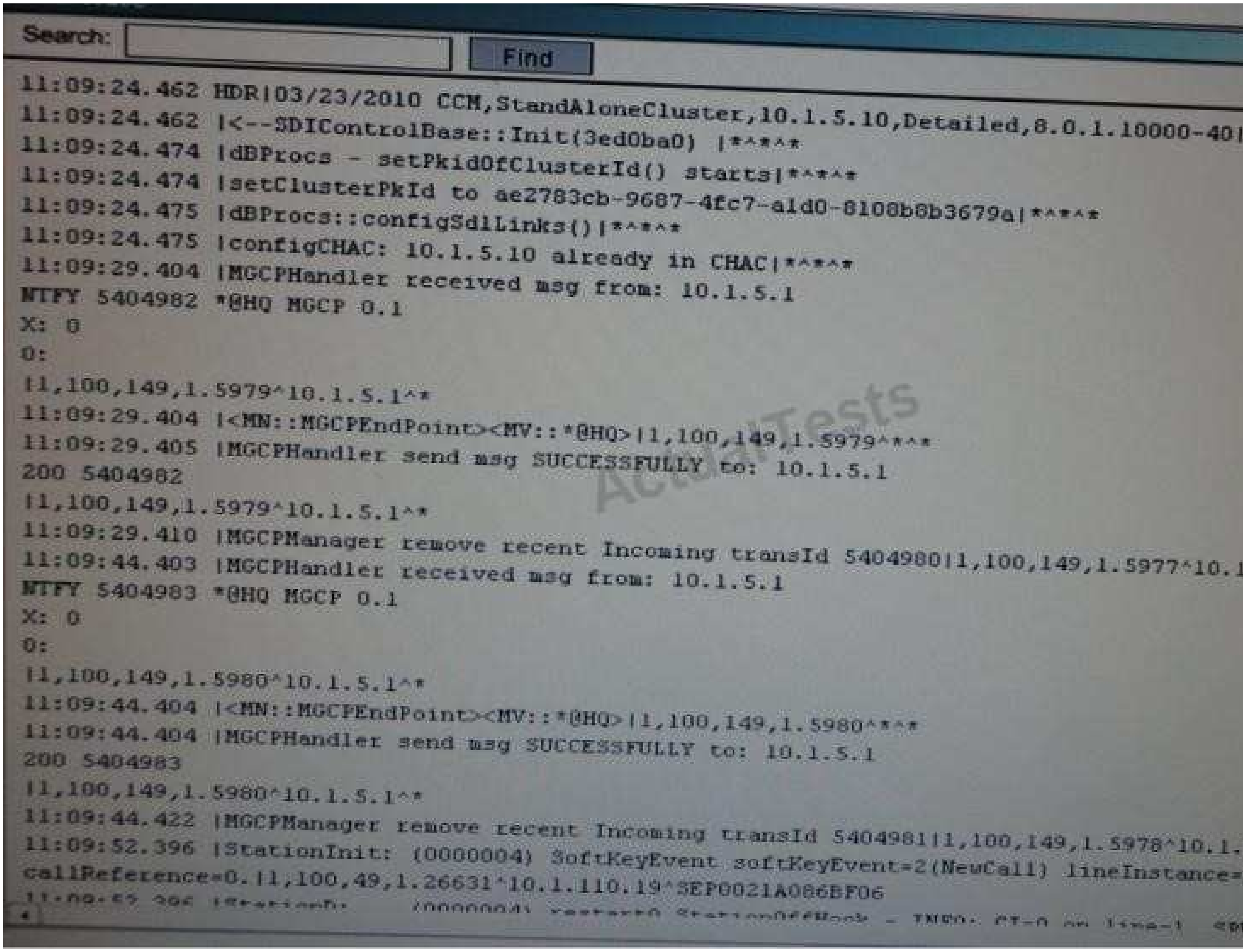
**Correct Answer:** A
**Section: (none)**
**Explanation**

**Explanation/Reference:**
**Explanation**-Dialed Number Analyzer installs as a feature service along with Cisco Unified Communications Manager. The tool allows you to test a Cisco Unified Communications Manager dial plan configuration prior to deploying it. You can also use the tool to analyze dial plans after the dial plan is depl Link-http://www.cisco.com/en/US/docs/voice_ip_comm/cucm/dna/6_1_1/dnai.html

**QUESTION 38**
## Refer to the exhibits

Examining the SDI trace shows that extension 2001 was configured with both a device and a line CSS. A can is placed to 00014087071222. Which statement about the call is true?

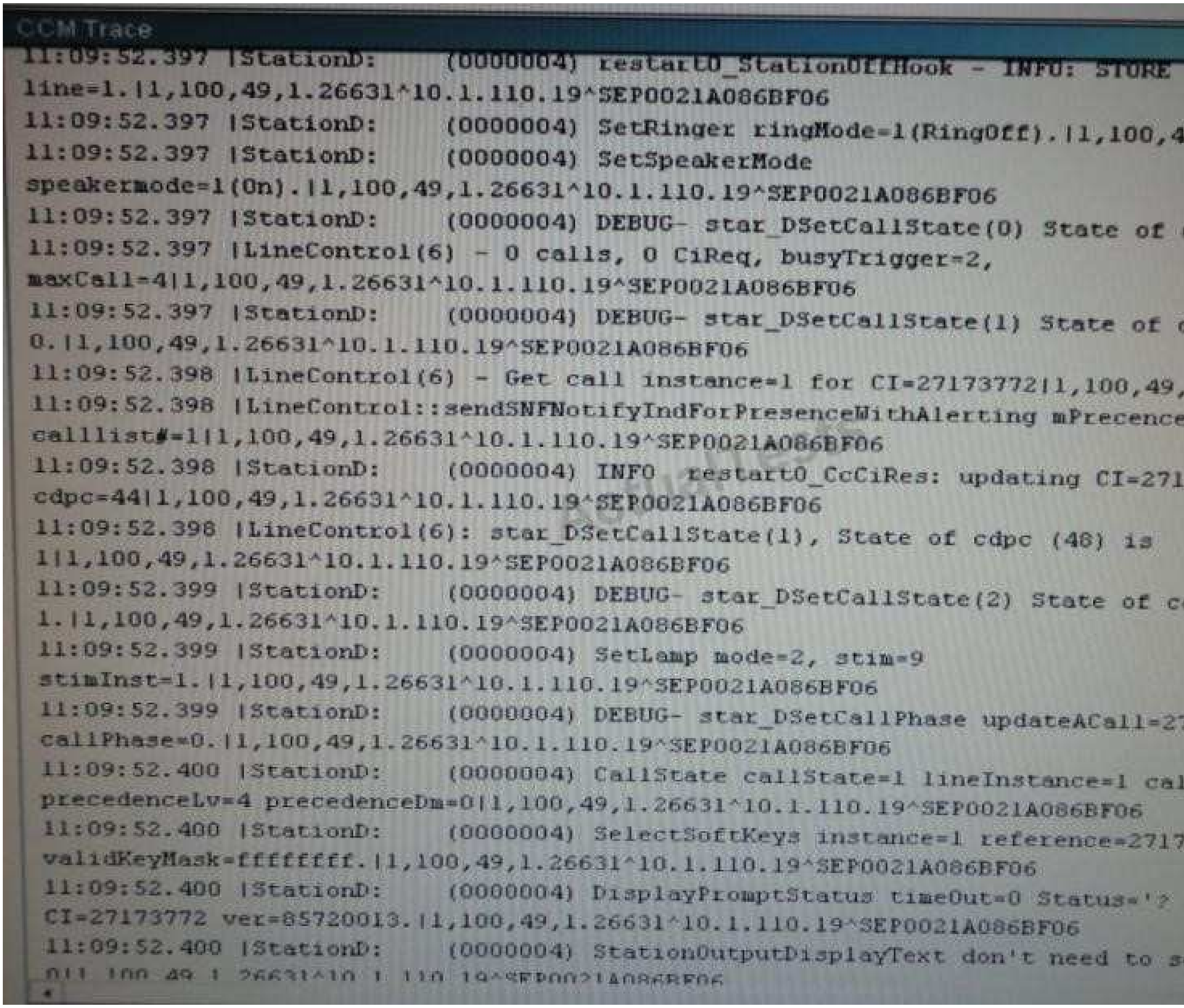
**91 (exhibit):**



Search: Find

```
11:09:24.462 HDR|03/23/2010 CCM,StandAloneCluster,10.1.5.10,Detailed,8.0.1.10000-40|
11:09:24.462 |<--SDIControlBase::Init(3ed0ba0) |*^*^*
11:09:24.474 |dBProcs - setPkidOfClusterId() starts|*^*^*
11:09:24.474 |setClusterPkId to ae2783cb-9687-4fc7-a1d0-8108b8b3679a|*^*^*
11:09:24.475 |dBProcs::configSdlLinks()|*^*^*
11:09:24.475 |configCHAC: 10.1.5.10 already in CHAC|*^*^*
11:09:29.404 |MGCPHandler received msg from: 10.1.5.1
NTFY 5404982 *@HQ MGCP 0.1
X: 0
O:
|1,100,149,1.5979^10.1.5.1^*
11:09:29.404 |<MN::MGCPEndPoint><MV::*@HQ>|1,100,149,1.5979^*^*
11:09:29.405 |MGCPHandler send msg SUCCESSFULLY to: 10.1.5.1
200 5404982
|1,100,149,1.5979^10.1.5.1^*
11:09:29.410 |MGCPManager remove recent Incoming transId 5404980|1,100,149,1.5977^10.1
11:09:44.403 |MGCPHandler received msg from: 10.1.5.1
NTFY 5404983 *@HQ MGCP 0.1
X: 0
O:
|1,100,149,1.5980^10.1.5.1^*
11:09:44.404 |<MN::MGCPEndPoint><MV::*@HQ>|1,100,149,1.5980^*^*
11:09:44.404 |MGCPHandler send msg SUCCESSFULLY to: 10.1.5.1
200 5404983
|1,100,149,1.5980^10.1.5.1^*
11:09:44.422 |MGCPManager remove recent Incoming transId 5404981|1,100,149,1.5978^10.1.
11:09:52.396 |StationInit: (0000004) SoftKeyEvent softKeyEvent=2(NewCall) lineInstance=
callReference=0.|1,100,49,1.26631^10.1.110.19^SEP0021A086BF06
```

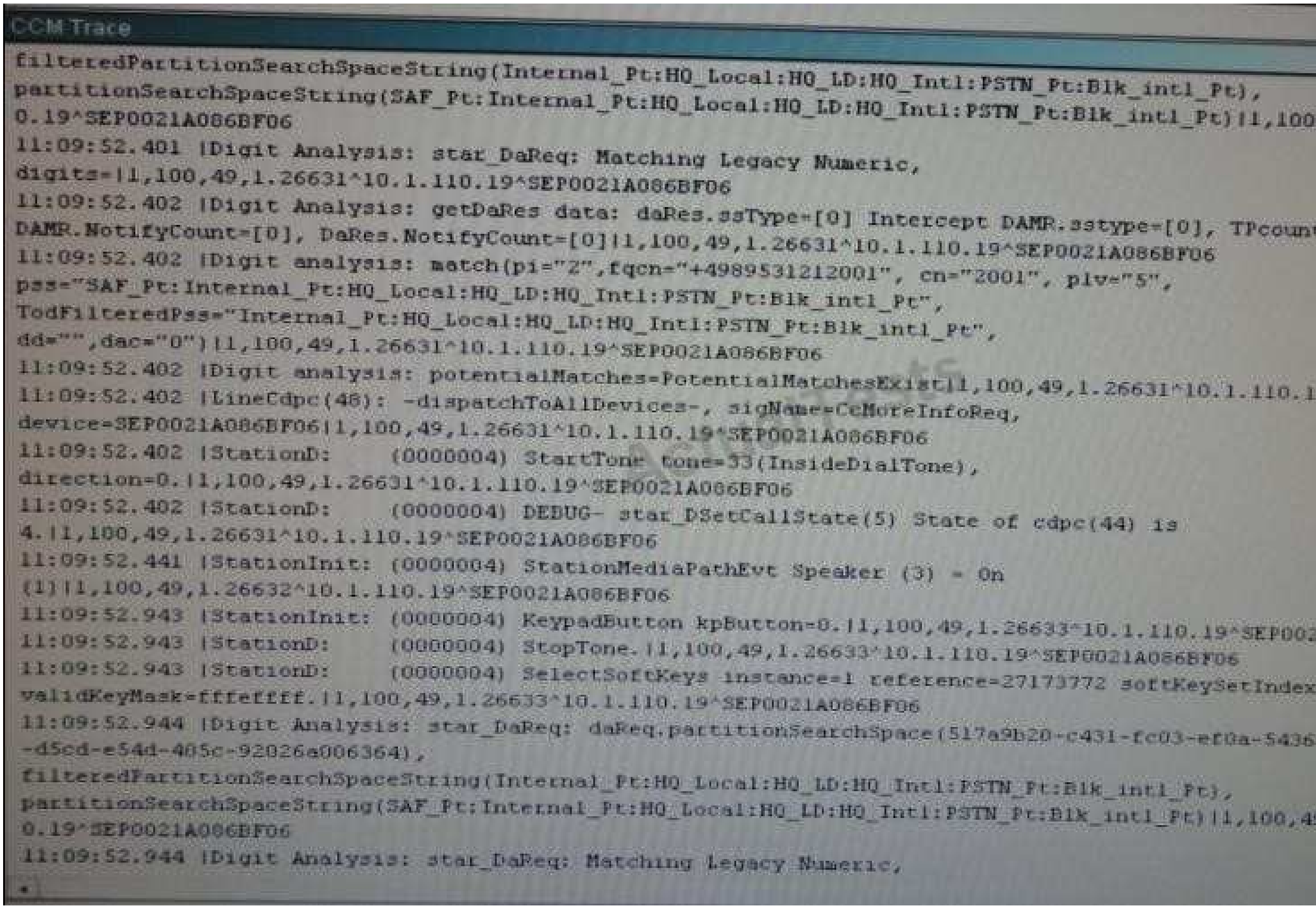
**92 (exhibit):**



CCM Trace

```
11:09:52.397 |StationD:     (0000004) restart0_StationOffHook - INFO: STORE
line=1.|1,100,49,1.26631^10.1.110.19^SEP0021A086BF06
11:09:52.397 |StationD:     (0000004) SetRinger ringMode=1(RingOff).|1,100,4
11:09:52.397 |StationD:     (0000004) SetSpeakerMode
speakermode=1(On).|1,100,49,1.26631^10.1.110.19^SEP0021A086BF06
11:09:52.397 |StationD:     (0000004) DEBUG- star_DSetCallState(0) State of
11:09:52.397 |LineControl(6) - 0 calls, 0 CiReq, busyTrigger=2,
maxCall=4|1,100,49,1.26631^10.1.110.19^SEP0021A086BF06
11:09:52.397 |StationD:     (0000004) DEBUG- star_DSetCallState(1) State of
0.|1,100,49,1.26631^10.1.110.19^SEP0021A086BF06
11:09:52.398 |LineControl(6) - Get call instance=1 for CI=27173772|1,100,49,
11:09:52.398 |LineControl::sendSNFNotifyIndForPresenceWithAlerting mPrecence
calllist#=1|1,100,49,1.26631^10.1.110.19^SEP0021A086BF06
11:09:52.398 |StationD:     (0000004) INFO  restart0_CcCiRes: updating CI=271
cdpc=44|1,100,49,1.26631^10.1.110.19^SEP0021A086BF06
11:09:52.398 |LineControl(6): star_DSetCallState(1), State of cdpc (48) is
1|1,100,49,1.26631^10.1.110.19^SEP0021A086BF06
11:09:52.399 |StationD:     (0000004) DEBUG- star_DSetCallState(2) State of c
1.|1,100,49,1.26631^10.1.110.19^SEP0021A086BF06
11:09:52.399 |StationD:     (0000004) SetLamp mode=2, stim=9
stimInst=1.|1,100,49,1.26631^10.1.110.19^SEP0021A086BF06
11:09:52.399 |StationD:     (0000004) DEBUG- star_DSetCallPhase updateACall=27
callPhase=0.|1,100,49,1.26631^10.1.110.19^SEP0021A086BF06
11:09:52.400 |StationD:     (0000004) CallState callState=1 lineInstance=1 cal
precedenceLv=4 precedenceDm=0|1,100,49,1.26631^10.1.110.19^SEP0021A086BF06
11:09:52.400 |StationD:     (0000004) SelectSoftKeys instance=1 reference=2717
validKeyMask=fffffff.|1,100,49,1.26631^10.1.110.19^SEP0021A086BF06
11:09:52.400 |StationD:     (0000004) DisplayPromptStatus timeOut=0 Status='?
CI=27173772 ver=85720013.|1,100,49,1.26631^10.1.110.19^SEP0021A086BF06
11:09:52.400 |StationD:     (0000004) StationOutputDisplayText don't need to s
0|1,100,49,1.26631^10.1.110.19^SEP0021A086BF06
```

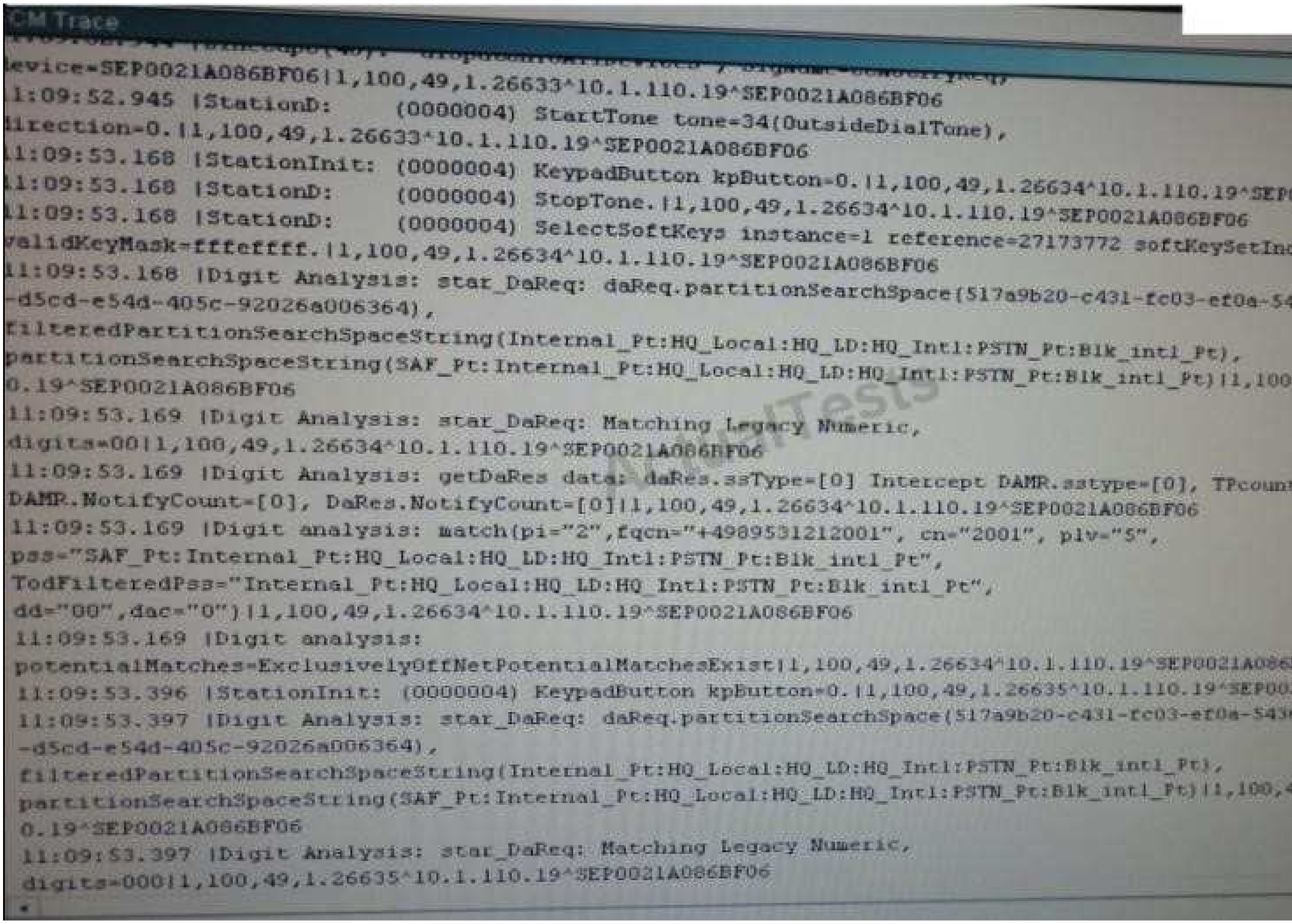
**93 (exhibit):**



CCM Trace

```
filteredPartitionSearchSpaceString(Internal_Pt:HQ_Local:HQ_LD:HQ_Intl:PSTN_Pt:Blk_intl_Pt),
partitionSearchSpaceString(SAF_Pt:Internal_Pt:HQ_Local:HQ_LD:HQ_Intl:PSTN_Pt:Blk_intl_Pt)|1,100
0.19^SEP0021A086BF06
11:09:52.401 |Digit Analysis: star_DaReq: Matching Legacy Numeric,
digits=|1,100,49,1.26631^10.1.110.19^SEP0021A086BF06
11:09:52.402 |Digit Analysis: getDaRes data: daRes.ssType=[0] Intercept DAMR.sstype=[0], TPcount
DAMR.NotifyCount=[0], DaRes.NotifyCount=[0]|1,100,49,1.26631^10.1.110.19^SEP0021A086BF06
11:09:52.402 |Digit analysis: match(pi="2",fqcn="+4989531212001", cn="2001", plv="5",
pss="SAF_Pt:Internal_Pt:HQ_Local:HQ_LD:HQ_Intl:PSTN_Pt:Blk_intl_Pt",
TodFilteredPss="Internal_Pt:HQ_Local:HQ_LD:HQ_Intl:PSTN_Pt:Blk_intl_Pt",
dd="",dac="0")|1,100,49,1.26631^10.1.110.19^SEP0021A086BF06
11:09:52.402 |Digit analysis: potentialMatches=PotentialMatchesExist|1,100,49,1.26631^10.1.110.1
11:09:52.402 |LineCdpc(48): -dispatchToAllDevices-, sigName=CcMoreInfoReq,
device=SEP0021A086BF06|1,100,49,1.26631^10.1.110.19^SEP0021A086BF06
11:09:52.402 |StationD:    (0000004) StartTone tone=33(InsideDialTone),
direction=0.|1,100,49,1.26631^10.1.110.19^SEP0021A086BF06
11:09:52.402 |StationD:    (0000004) DEBUG- star_DSetCallState(5) State of cdpc(44) is
4.|1,100,49,1.26631^10.1.110.19^SEP0021A086BF06
11:09:52.441 |StationInit: (0000004) StationMediaPathEvt Speaker (3) = On
(1)|1,100,49,1.26632^10.1.110.19^SEP0021A086BF06
11:09:52.943 |StationInit: (0000004) KeypadButton kpButton=8.|1,100,49,1.26633^10.1.110.19^SEP002
11:09:52.943 |StationD:    (0000004) StopTone.|1,100,49,1.26633^10.1.110.19^SEP0021A086BF06
11:09:52.943 |StationD:    (0000004) SelectSoftKeys instance=1 reference=27173772 softKeySetIndex
validKeyMask=fffeffff.|1,100,49,1.26633^10.1.110.19^SEP0021A086BF06
11:09:52.944 |Digit Analysis: star_DaReq: daReq.partitionSearchSpace(517a9b20-c431-fc03-ef0a-5436
-d5cd-e54d-405c-92026a006364),
filteredPartitionSearchSpaceString(Internal_Pt:HQ_Local:HQ_LD:HQ_Intl:PSTN_Pt:Blk_intl_Pt),
partitionSearchSpaceString(SAF_Pt:Internal_Pt:HQ_Local:HQ_LD:HQ_Intl:PSTN_Pt:Blk_intl_Pt)|1,100,4
0.19^SEP0021A086BF06
11:09:52.944 |Digit Analysis: star_DaReq: Matching Legacy Numeric,
```

**94 (exhibit):**

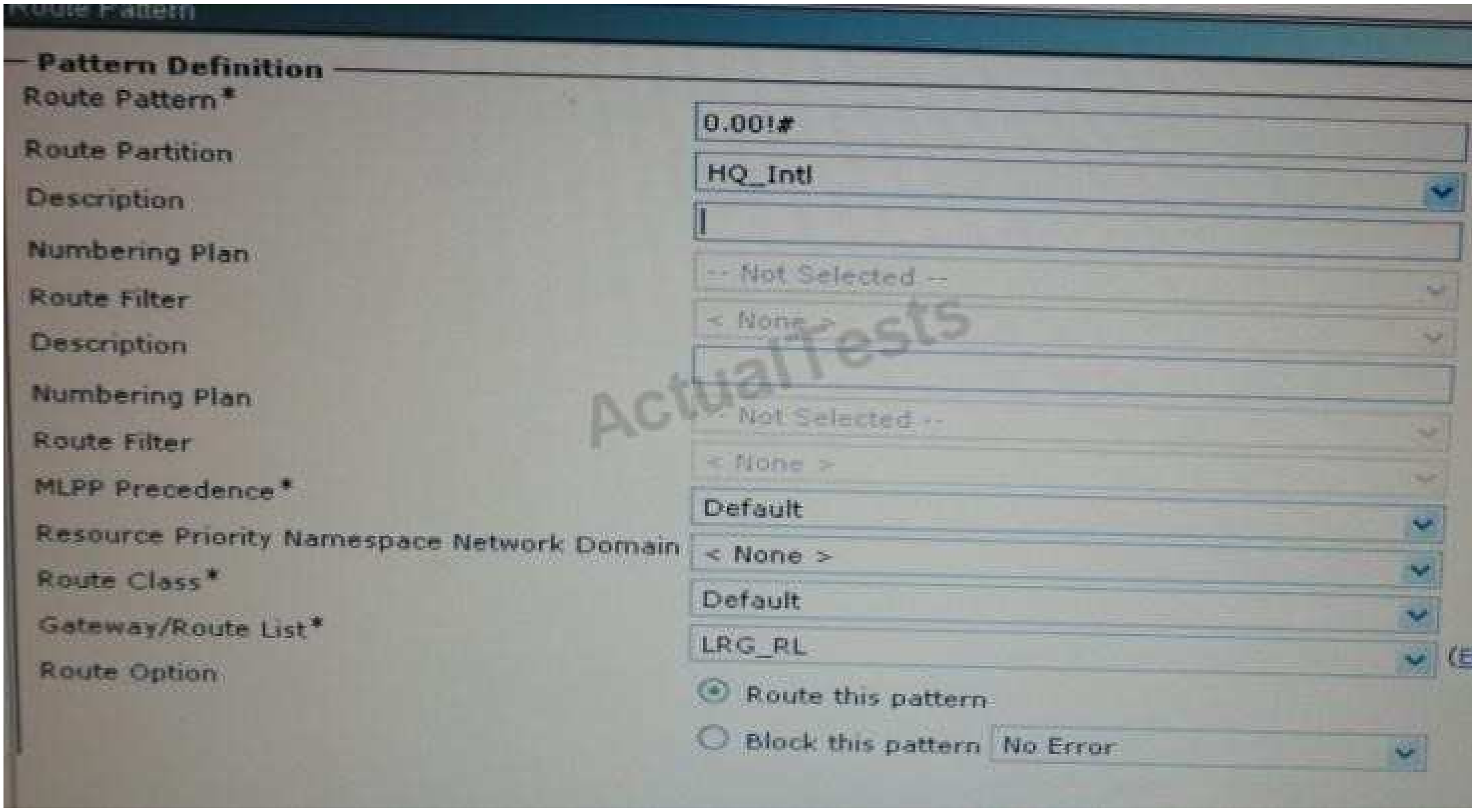


```
CM Trace
device=SEP0021A086BF06|1,100,49,1.26633^10.1.110.19^SEP0021A086BF06
11:09:52.945 |StationD:      (0000004) StartTone tone=34(OutsideDialTone),
direction=0.|1,100,49,1.26633^10.1.110.19^SEP0021A086BF06
11:09:53.168 |StationInit:  (0000004) KeypadButton kpButton=0.|1,100,49,1.26634^10.1.110.19^SEP
11:09:53.168 |StationD:      (0000004) StopTone.|1,100,49,1.26634^10.1.110.19^SEP0021A086BF06
11:09:53.168 |StationD:      (0000004) SelectSoftKeys instance=1 reference=27173772 softKeySetInd
validKeyMask=fffeffff.|1,100,49,1.26634^10.1.110.19^SEP0021A086BF06
11:09:53.168 |Digit Analysis: star_DaReq: daReq.partitionSearchSpace(517a9b20-c431-fc03-ef0a-54
-d5cd-e54d-405c-92026a006364),
filteredPartitionSearchSpaceString(Internal_Pt:HQ_Local:HQ_LD:HQ_Intl:PSTN_Pt:Blk_intl_Pt),
partitionSearchSpaceString(SAF_Pt:Internal_Pt:HQ_Local:HQ_LD:HQ_Intl:PSTN_Pt:Blk_intl_Pt)|1,100
0.19^SEP0021A086BF06
11:09:53.169 |Digit Analysis: star_DaReq: Matching Legacy Numeric,
digits=00|1,100,49,1.26634^10.1.110.19^SEP0021A086BF06
11:09:53.169 |Digit Analysis: getDaRes data: daRes.ssType=[0] Intercept DAMR.sstype=[0], TPcount
DAMR.NotifyCount=[0], DaRes.NotifyCount=[0]|1,100,49,1.26634^10.1.110.19^SEP0021A086BF06
11:09:53.169 |Digit analysis: match(pi="2",fqcn="+4989531212001", cn="2001", plv="5",
pss="SAF_Pt:Internal_Pt:HQ_Local:HQ_LD:HQ_Intl:PSTN_Pt:Blk_intl_Pt",
TodFilteredPss="Internal_Pt:HQ_Local:HQ_LD:HQ_Intl:PSTN_Pt:Blk_intl_Pt",
dd="00",dac="0")|1,100,49,1.26634^10.1.110.19^SEP0021A086BF06
11:09:53.169 |Digit analysis:
potentialMatches=ExclusivelyOffNetPotentialMatchesExist|1,100,49,1.26634^10.1.110.19^SEP0021A086
11:09:53.396 |StationInit: (0000004) KeypadButton kpButton=0.|1,100,49,1.26635^10.1.110.19^SEP00
11:09:53.397 |Digit Analysis: star_DaReq: daReq.partitionSearchSpace(517a9b20-c431-fc03-ef0a-543
-d5cd-e54d-405c-92026a006364),
filteredPartitionSearchSpaceString(Internal_Pt:HQ_Local:HQ_LD:HQ_Intl:PSTN_Pt:Blk_intl_Pt),
partitionSearchSpaceString(SAF_Pt:Internal_Pt:HQ_Local:HQ_LD:HQ_Intl:PSTN_Pt:Blk_intl_Pt)|1,100,4
0.19^SEP0021A086BF06
11:09:53.397 |Digit Analysis: star_DaReq: Matching Legacy Numeric,
digits=000|1,100,49,1.26635^10.1.110.19^SEP0021A086BF06
```

**95 (exhibit):**

```
11:09:55.783 |StationInit: (0000004) KeypadButton kpButton=2.|1,100,49,1.26646^10.1.110.1
-d5cd-e54d-405c-92026a006364),
filteredPartitionSearchSpaceString(Internal_Pt:HQ_Local:HQ_LD:HQ_Intl:PSTN_Pt:Blk_intl_Pt)
partitionSearchSpaceString(SAF_Pt:Internal_Pt:HQ_Local:HQ_LD:HQ_Intl:PSTN_Pt:Blk_intl_Pt)|
0.19^SEP0021A086BF06
11:09:55.786 |Digit Analysis: star_DaReq: Matching Legacy Numeric,
digits=00014087071222|1,100,49,1.26646^10.1.110.19^SEP0021A086BF06
11:09:55.786 |Digit Analysis: getDaRes data: daRes.ssType=[0] Intercept DAMR.sstype=[0], T
DAMR.NotifyCount=[0], DaRes.NotifyCount=[0]|1,100,49,1.26646^10.1.110.19^SEP0021A086BF06
11:09:55.786 |Digit analysis: match(pi="2",fqcn="+4989531212001", cn="2001", plv="5",
pss="SAF_Pt:Internal_Pt:HQ_Local:HQ_LD:HQ_Intl:PSTN_Pt:Blk_intl_Pt",
TodFilteredPss="Internal_Pt:HQ_Local:HQ_LD:HQ_Intl:PSTN_Pt:Blk_intl_Pt",
dd="00014087071222",dac="0")|1,100,49,1.26646^10.1.110.19^SEP0021A086BF06
11:09:55.786 |Digit analysis:
potentialMatches=ExclusivelyOffNetPotentialMatchesExist|1,100,49,1.26646^10.1.110.19^SEP002
11:09:56.228 |StationInit: (0000004) KeypadButton kpButton=f.|1,100,49,1.26647^10.1.110.19^
11:09:56.229 |Digit Analysis: star_DaReq: daReq.partitionSearchSpace(517a9b20-c431-fc03-ef0
-d5cd-e54d-405c-92026a006364),
filteredPartitionSearchSpaceString(Internal_Pt:HQ_Local:HQ_LD:HQ_Intl:PSTN_Pt:Blk_intl_Pt),
partitionSearchSpaceString(SAF_Pt:Internal_Pt:HQ_Local:HQ_LD:HQ_Intl:PSTN_Pt:Blk_intl_Pt)|1,
0.19^SEP0021A086BF06
11:09:56.229 |Digit Analysis: star_DaReq: Matching Legacy Numeric,
digits=00014087071222#|1,100,49,1.26647^10.1.110.19^SEP0021A086BF06
11:09:56.230 |Digit Analysis: getDaRes data: daRes.ssType=[0] Intercept DAMR.sstype=[0], TPc
DAMR.NotifyCount=[0], DaRes.NotifyCount=[0]|1,100,49,1.26647^10.1.110.19^SEP0021A086BF06
11:09:56.230 |Digit analysis: match(pi="2", fqcn="+4989531212001", cn="2001",plv="5",
pss="SAF_Pt:Internal_Pt:HQ_Local:HQ_LD:HQ_Intl:PSTN_Pt:Blk_intl_Pt",
TodFilteredPss="Internal_Pt:HQ_Local:HQ_LD:HQ_Intl:PSTN_Pt:Blk_intl_Pt",
dd="00014087071222#",dac="0")|1,100,49,1.26647^10.1.110.19^SEP0021A086BF06
```

**96 (exhibit):**




A. The call will be blocked because the blkJntLPt partition appears in the combined CSS.
B. The call will work because the PSTN__ Pt partition appears before the blk_intl_Pt partition.
C. The call will work because the blk_intl_Pt partition appears last in the combined CSS.
D. The call will not work because the blk_intl_Pt partition appears first in the combined CSS.
E. It is not possible to tell because furthertrace analysis is required

**Correct Answer:** C
**Section: (none)**
**Explanation**

**Explanation/Reference:**
**Explanation: Link**- http://www.cisco.com/en/US/products/sw/voicesw/ps556/products_tech_note09186a0080094b53. shtml

**QUESTION 39**
**Refer to the exhibit.**

```
voice service saf
 profile trunk-route 1
  session protocol sip interface Loopback1 transport tcp port 5060
!
profile dn-block 1 alias-prefix 1972555
 pattern 1 type extension 4XXX
!
profile callcontrol 1
 dn-service
  trunk-route 1
  dn-block 1
!
channel 1 vrouter SAF asystem 1
 subscribe callcontrol wildcarded
 publish callcontrol 1
```

When a Cisco Unified Communications Manager Express advertises the directory number pattern in the exhibit, what would the learned pattern be in the RTMT tool on the Cisco Unified Communications Manager?

A. 4XXX and the ToDID will be 0:+1972555
B. 4XXX and the ToDID will be 0:+19725554XXX
C. 4XXX and the ToDID will be 0:19725554XXX
D. 4XXX and the ToDID will be 0:1972555
E. 19725554XXX and the ToDID will be 0:+1972555

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

**Explanation/Reference:**
**Explanation**: The answer is 4XXX and the ToDID will be 0:1972555.
Exhibit explain profile dn-block1 alias-prefix 1972555 and pattern 1 type extension 4xxx.

**QUESTION 40**
When a user attempts to log out from Cisco Extension Mobility service by pressing the services button and selecting the Cisco Extension Mobility service, the user is not able to log out. What is causing this issue?

A. The Cisco Extension Mobility service has not been configured on the phone
B. The user device profile is not subscribed to the Cisco Extension Mobility service.
C. The CTI service is not running
D. The logout URL that is defined for the Cisco Extension Mobility service is incorrect or does not exist under the IP Phone Services configuration.

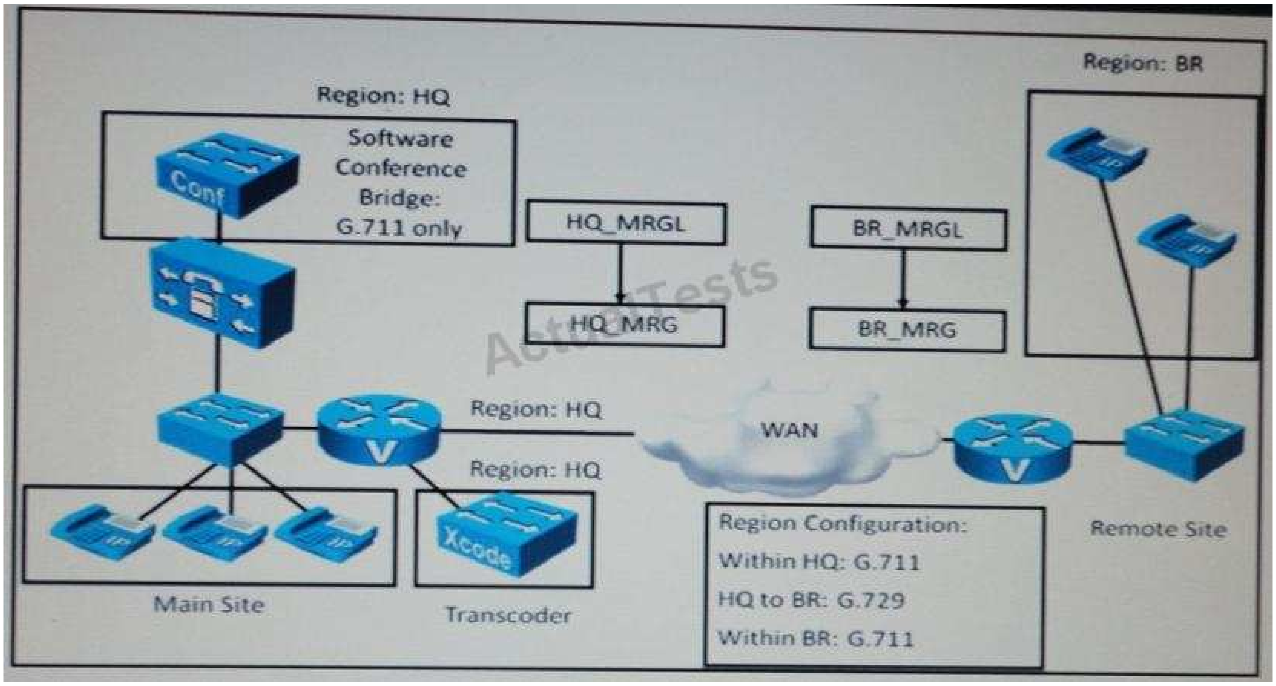
**Correct Answer:** B
**Section: (none)**
**Explanation**

**Explanation/Reference:**
Cisco CallManager Extension Mobility looks up the URL in the Cisco CallManager Directory on the first instance only; the URL is then stored as a static variable.

**Link-**http://www.cisco.com/en/US/docs/voice_ip_comm/cucm/admin/4_0_1/ccmfeat/fsem.html#wp1105
592

**QUESTION 41**

## Refer to the exhibit.




Refer to the exhibit. When a call between two HQ users was being conferenced with a remote user at the BR site, the conference failed. Which configuration would be needed to solve the problem?

A. The BR_MRG must contain the transcoder device. The BR_MRGL must be assigned to the BR phones.
B. The HQ_MRG must contain the transcoder device. The HQ_MRGL must be assigned to the HQ phones.
C. A transcoder should be configured at the remote site and assigned to all remote phones through the BR_MRGL.
D. The HQ_MRG must contain the transcoder device. The HQ_MRGL must be assigned to the software conference bridge.
E. Enable the software conference bridge to support G.711 and G.729 codecs in Cisco Unified

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

**Explanation/Reference:**
**Link-**http://www.cisco.com/en/US/products/sw/voicesw/ps556/products_tech_note09186a008020f198.s html

**QUESTION 42**

## Refer to the exhibit.


An engineer is troubleshooting an outbound call and needs to see each step of the dial plan as it is being parsed in the system. The engineer is not able to see all of the steps in the trace output. How can this problem be resolved?

A. Change the SDL Trace Flush Immediately to True.
B. No change is needed; the steps are automatically shown in trace files.
C. Change the SDL TraceType Flag to 0x9000EC44
D. Change the SDL Trace Max File Size to a higher number because it is not large enough to display dial plan steps.
E. Change the Digit Analysis Complexity toTranslation And Alternate Pattern Analysis.

**Correct Answer:** E
**Section: (none)**
**Explanation**

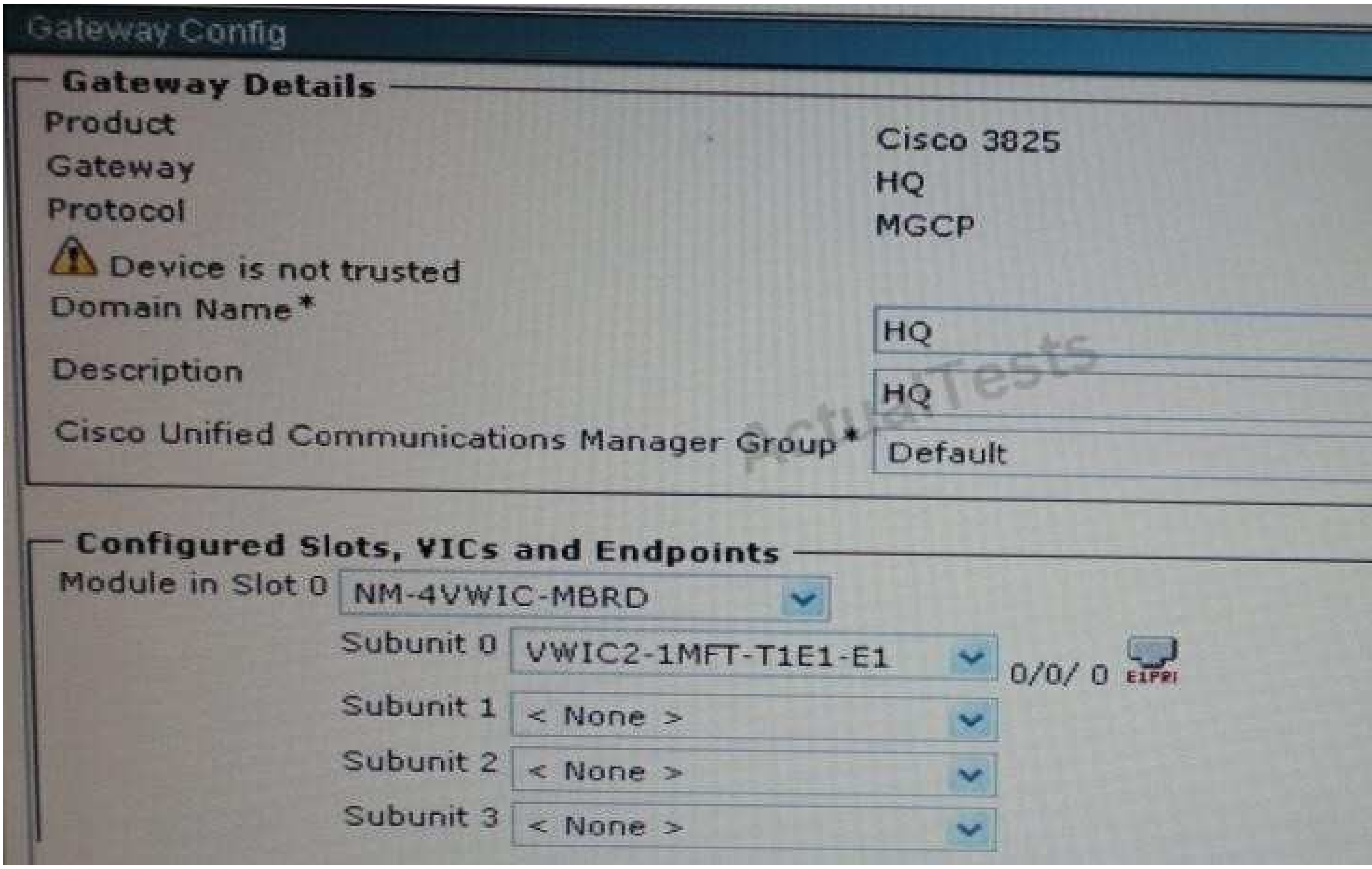
**Explanation/Reference:**
**Explanation**-DIGIT ANALYSY COMPLEXITY
determines whether detailed digit analysis information is included in the trace output and Cisco DNA. If you want to see translation pattern and alternative pattern match information, set this

parameter to TranslationAndAlternatePatternAnalysis. The default is StandardAnalysis, which does not show pattern transformations.
To troubleshoot dial plan issues, change to Translational Pattern and Alternate Analysis. Setting this option creates detailed digit analysis in CCM traces. http://www.iphelp.ru/faq/1/ch08lev1sec3.html

**QUESTION 43**
## Refer to the exhibits.

The gateway CSS for inbound calls is configured using internal. CSS, which contains all the directory number partitions for the phones. When a PSTN call that is destined for extension 2001 arrives at the MGCP gateway, the call falls. Use the debug and configuration information in the exhibits to determine what might be causing this issue?

**97 (exhibit):**


Gateway Config
Gateway Details
Product
Cisco 3825
Gateway
HQ
Protocol
MGCP
Device is not trusted
Domain Name*
HQ
Description
HQ
Cisco Unified Communications Manager Group*
Default
Configured Slots, VICs and Endpoints
Module in Slot 0
NM-4VWIC-MBRD
Subunit 0
VWIC2-1MFT-T1E1-E1
0/0/ 0
E1PRI
Subunit 1
< None >
Subunit 2
< None >
Subunit 3
< None >

**98 (exhibit):**

MGCP IOS Config

```
hostname HQ
!
boot-start-marker
boot system flash c3825-ipvoice_ivs-mz.150-1.XA1.bin
boot-end-marker
!
card type e1 0 0
enable password cisco123
!
no aaa new-model
network-clock-participate wic 0
!
ip source-route
ip cef
!
!
ip dhcp excluded-address 10.1.10.1 10.1.10.9
ip dhcp excluded-address 10.1.10.21 10.1.10.254
!
ip dhcp pool Data
  network 10.1.10.0 255.255.255.0
  default-router 10.1.10.1
!
!
no ip domain lookup
ip multicast-routing
no ipv6 cef
multilink bundle-name authenticated
!
!
!
```



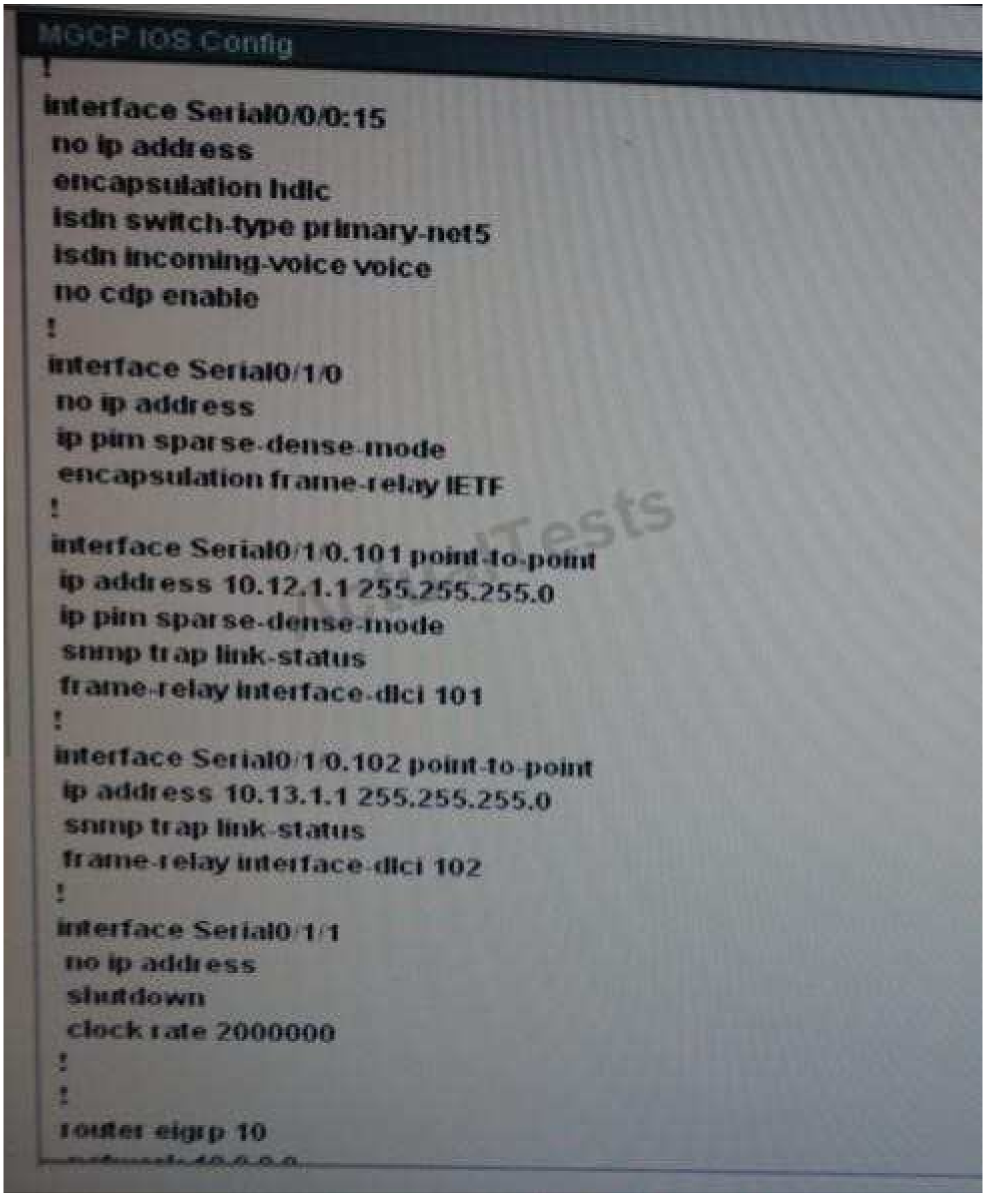
**99 (exhibit):**

MGCP IOS Config

```
!
!
controller E1 0/0/0
 pri-group timeslots 1-12,16 service mgcp
!
interface GigabitEthernet0/0
 no ip address
 ip pim sparse-dense-mode
 duplex auto
 speed auto
 media-type rj45
!
interface GigabitEthernet0/0.5
 encapsulation dot1Q 5
 ip address 10.1.5.1 255.255.255.0
 ip pim sparse-dense-mode
!
interface GigabitEthernet0/0.10
 encapsulation dot1Q 10
 ip address 10.1.10.1 255.255.255.0
 ip pim sparse-dense-mode
!
interface GigabitEthernet0/0.110
 encapsulation dot1Q 110
 ip address 10.1.110.1 255.255.255.0
 ip helper-address 10.1.5.10
 ip pim sparse-dense-mode
!
interface Serial0/0/0:15
 no ip address
 encapsulation hdlc
```

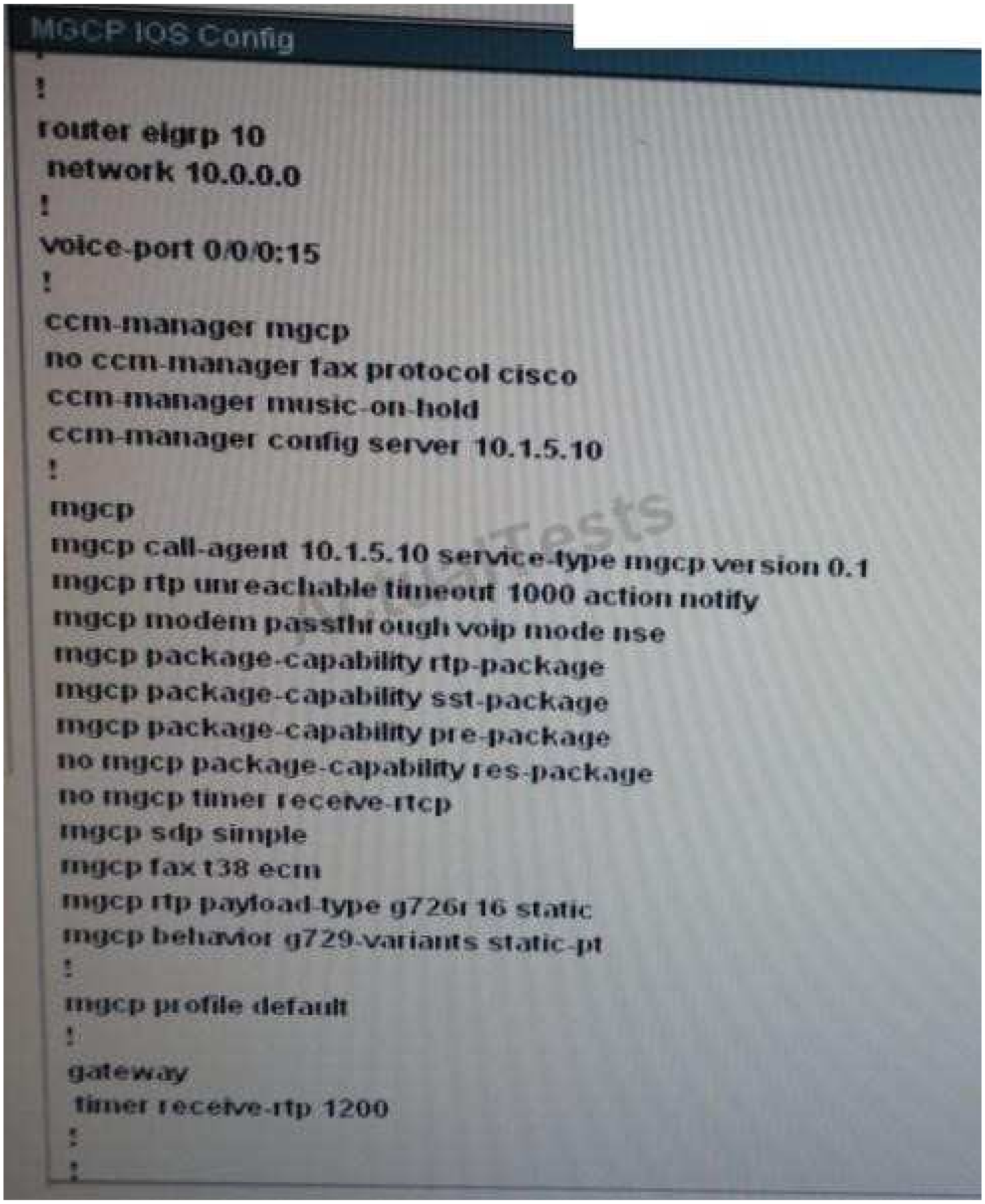


**100 (exhibit):**



```
MGCP IOS Config
!
interface Serial0/0/0:15
 no ip address
 encapsulation hdlc
 isdn switch-type primary-net5
 isdn incoming-voice voice
 no cdp enable
!
interface Serial0/1/0
 no ip address
 ip pim sparse-dense-mode
 encapsulation frame-relay IETF
!
interface Serial0/1/0.101 point-to-point
 ip address 10.12.1.1 255.255.255.0
 ip pim sparse-dense-mode
 snmp trap link-status
 frame-relay interface-dlci 101
!
interface Serial0/1/0.102 point-to-point
 ip address 10.13.1.1 255.255.255.0
 snmp trap link-status
 frame-relay interface-dlci 102
!
interface Serial0/1/1
 no ip address
 shutdown
 clock rate 2000000
!
!
router eigrp 10
```

**101 (exhibit):**



MGCP IOS Config

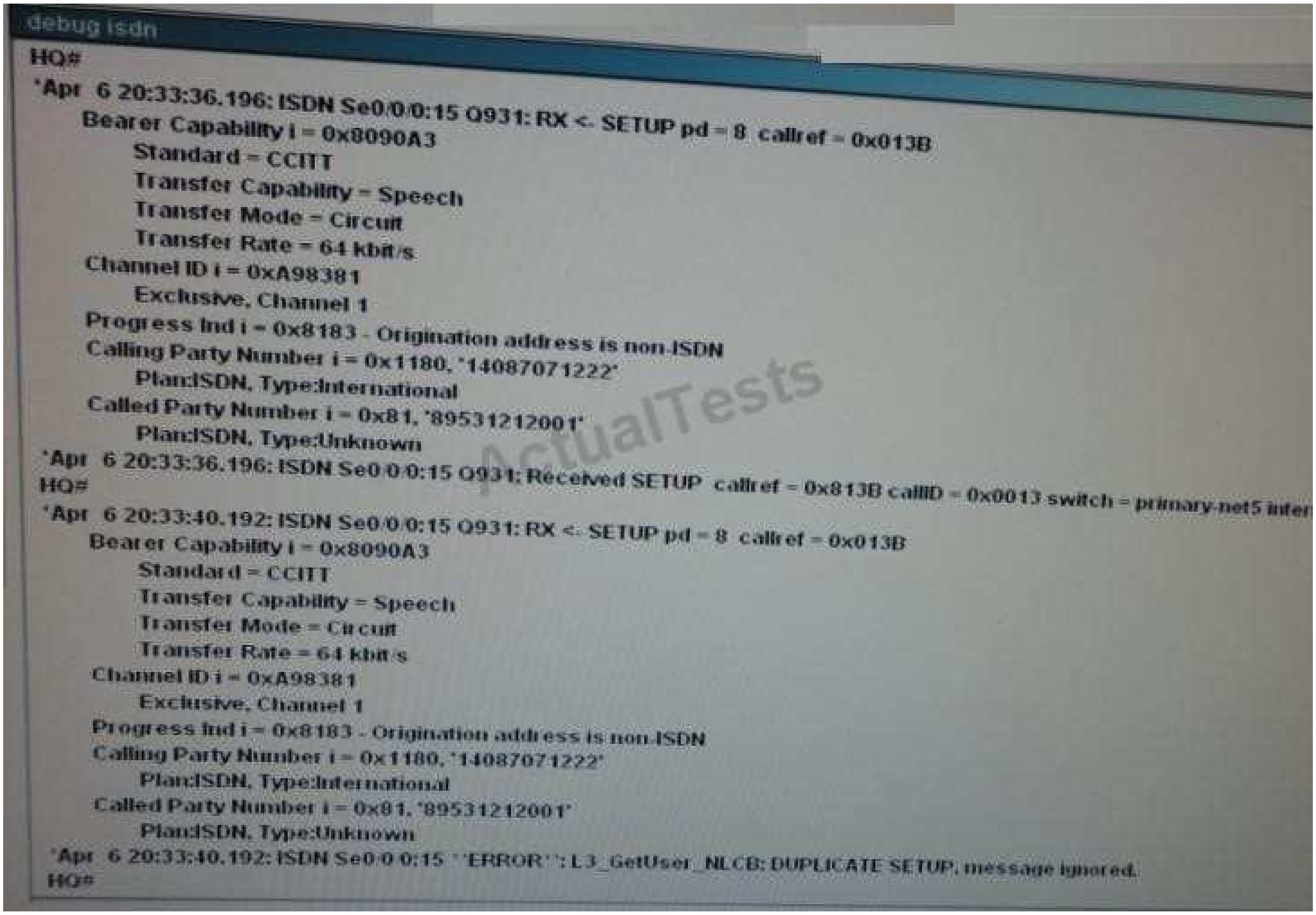
```
!
router eigrp 10
 network 10.0.0.0
!
voice-port 0/0/0:15
!
ccm-manager mgcp
no ccm-manager fax protocol cisco
ccm-manager music-on-hold
ccm-manager config server 10.1.5.10
!
mgcp
mgcp call-agent 10.1.5.10 service-type mgcp version 0.1
mgcp rtp unreachable timeout 1000 action notify
mgcp modem passthrough voip mode nse
mgcp package-capability rtp-package
mgcp package-capability sst-package
mgcp package-capability pre-package
no mgcp package-capability res-package
no mgcp timer receive-rtcp
mgcp sdp simple
mgcp fax t38 ecm
mgcp rtp payload-type g726r16 static
mgcp behavior g729-variants static-pt
!
mgcp profile default
!
gateway
 timer receive-rtp 1200
!
!
```

**102 (exhibit):**

```
mgcp rtp payload-type g726r16 static
mgcp behavior g729-variants static-pt
!
mgcp profile default
!
gateway
 timer receive-rtp 1200
!
!
gatekeeper
 shutdown
!
!
line con 0
 location CIEV-GA-LAB04, SJ
 exec-timeout 0 0
 logging synchronous
line aux 0
line vty 0 4
 exec-timeout 0 0
 password cisco123
 login
!
scheduler allocate 20000 1000
end
```



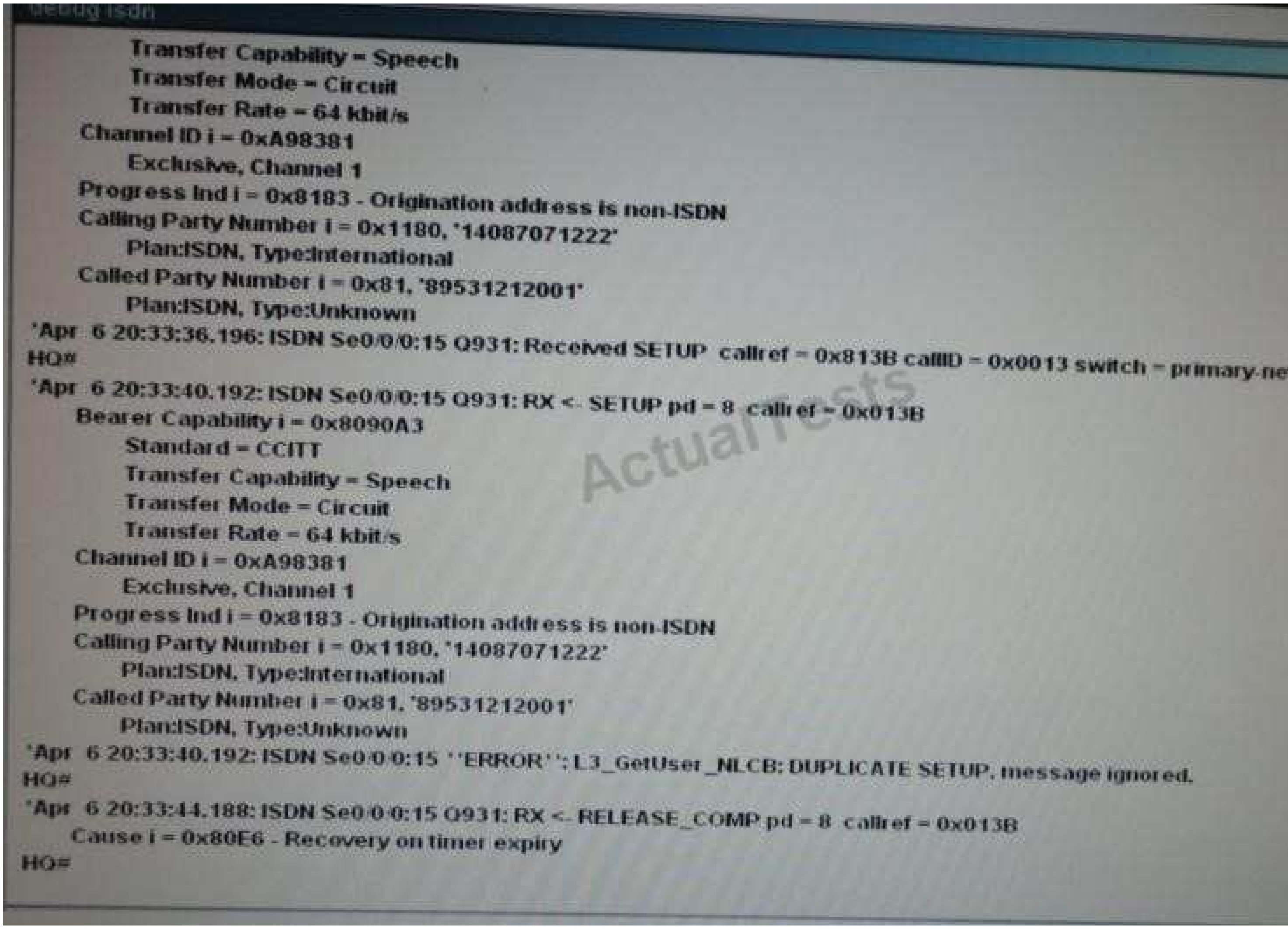
**103 (exhibit):**



```
debug isdn
HQ#
*Apr  6 20:33:36.196: ISDN Se0/0/0:15 Q931: RX <- SETUP pd = 8  callref = 0x013B
        Bearer Capability i = 0x8090A3
                Standard = CCITT
                Transfer Capability = Speech
                Transfer Mode = Circuit
                Transfer Rate = 64 kbit/s
        Channel ID i = 0xA98381
                Exclusive, Channel 1
        Progress Ind i = 0x8183 - Origination address is non-ISDN
        Calling Party Number i = 0x1180, '14087071222'
                Plan:ISDN, Type:International
        Called Party Number i = 0x81, '89531212001'
                Plan:ISDN, Type:Unknown
*Apr  6 20:33:36.196: ISDN Se0/0/0:15 Q931: Received SETUP  callref = 0x813B callID = 0x0013 switch = primary-net5 inter
HQ#
*Apr  6 20:33:40.192: ISDN Se0/0/0:15 Q931: RX <- SETUP pd = 8  callref = 0x013B
        Bearer Capability i = 0x8090A3
                Standard = CCITT
                Transfer Capability = Speech
                Transfer Mode = Circuit
                Transfer Rate = 64 kbit/s
        Channel ID i = 0xA98381
                Exclusive, Channel 1
        Progress Ind i = 0x8183 - Origination address is non-ISDN
        Calling Party Number i = 0x1180, '14087071222'
                Plan:ISDN, Type:International
        Called Party Number i = 0x81, '89531212001'
                Plan:ISDN, Type:Unknown
*Apr  6 20:33:40.192: ISDN Se0/0/0:15 **ERROR**: L3_GetUser_NLCB: DUPLICATE SETUP, message ignored.
HQ#
```

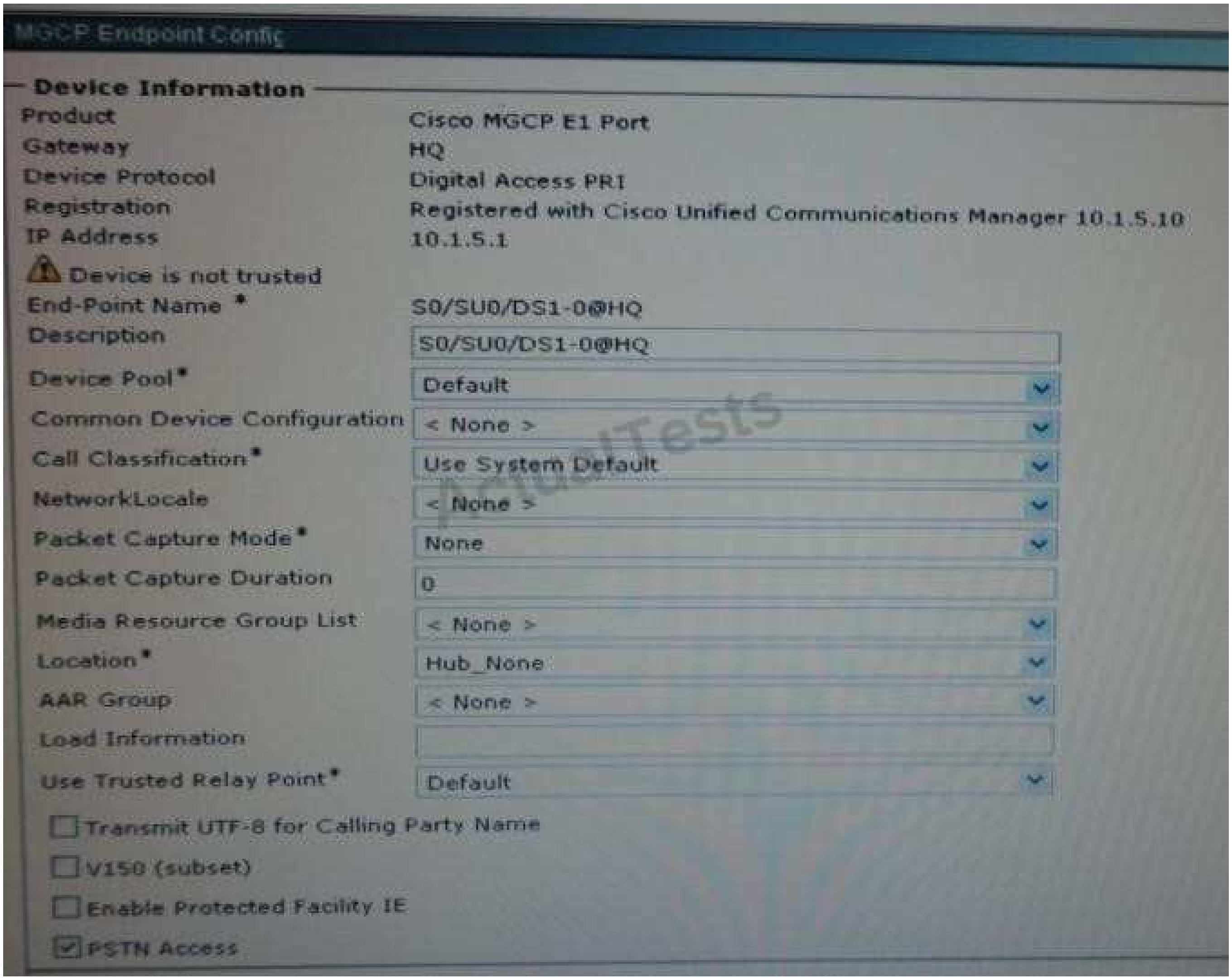


**104 (exhibit):**

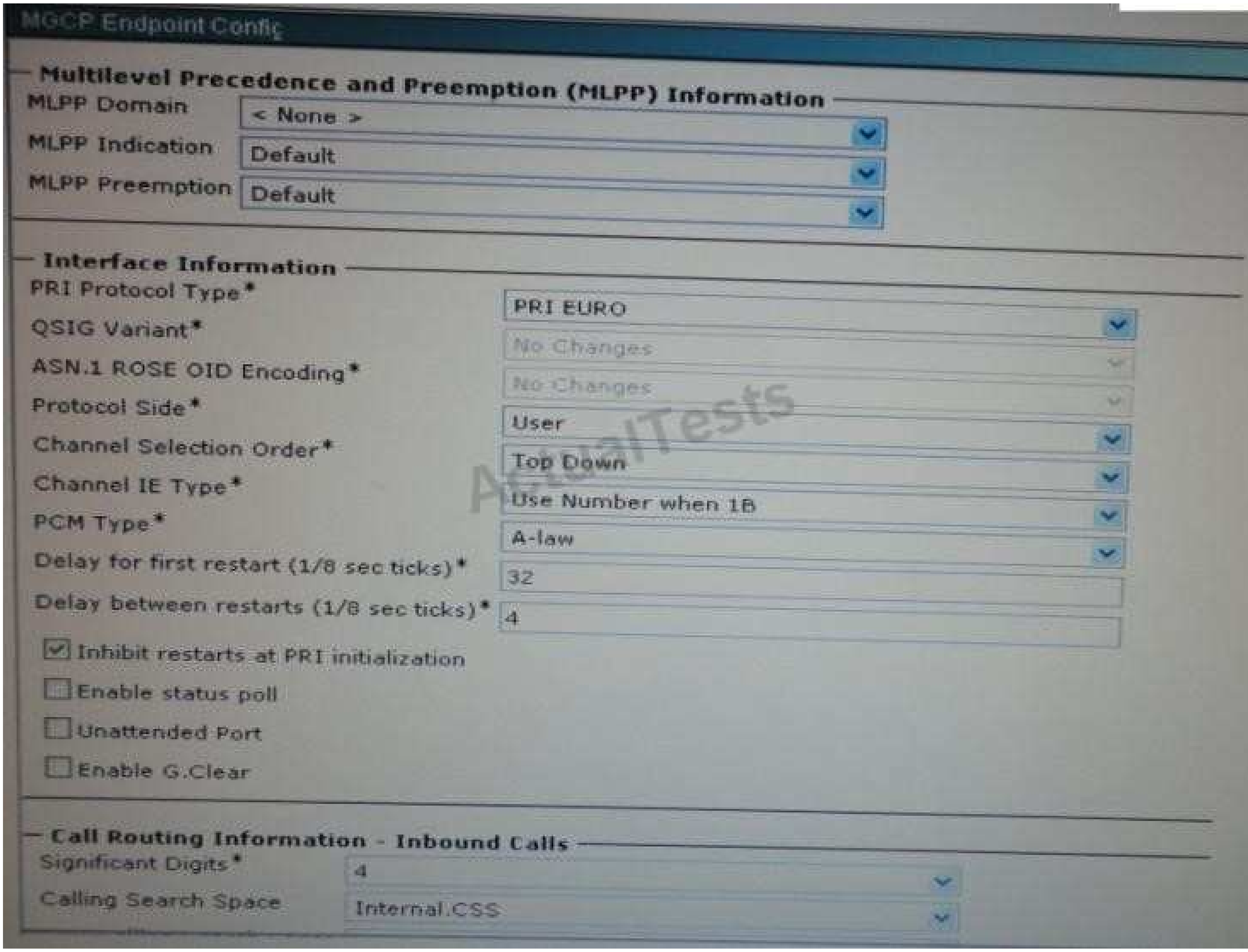


```
debug isdn
            Transfer Capability = Speech
            Transfer Mode = Circuit
            Transfer Rate = 64 kbit/s
        Channel ID i = 0xA98381
            Exclusive, Channel 1
        Progress Ind i = 0x8183 - Origination address is non-ISDN
        Calling Party Number i = 0x1180, '14087071222'
            Plan:ISDN, Type:International
        Called Party Number i = 0x81, '89531212001'
            Plan:ISDN, Type:Unknown
*Apr  6 20:33:36.196: ISDN Se0/0/0:15 Q931: Received SETUP  callref = 0x813B callID = 0x0013 switch = primary-ne
HQ#
*Apr  6 20:33:40.192: ISDN Se0/0/0:15 Q931: RX <- SETUP pd = 8  callref = 0x013B
        Bearer Capability i = 0x8090A3
            Standard = CCITT
            Transfer Capability = Speech
            Transfer Mode = Circuit
            Transfer Rate = 64 kbit/s
        Channel ID i = 0xA98381
            Exclusive, Channel 1
        Progress Ind i = 0x8183 - Origination address is non-ISDN
        Calling Party Number i = 0x1180, '14087071222'
            Plan:ISDN, Type:International
        Called Party Number i = 0x81, '89531212001'
            Plan:ISDN, Type:Unknown
*Apr  6 20:33:40.192: ISDN Se0/0/0:15 **ERROR**: L3_GetUser_NLCB: DUPLICATE SETUP, message ignored.
HQ#
*Apr  6 20:33:44.188: ISDN Se0/0/0:15 Q931: RX <- RELEASE_COMP pd = 8  callref = 0x013B
        Cause i = 0x80E6 - Recovery on timer expiry
HQ#
```

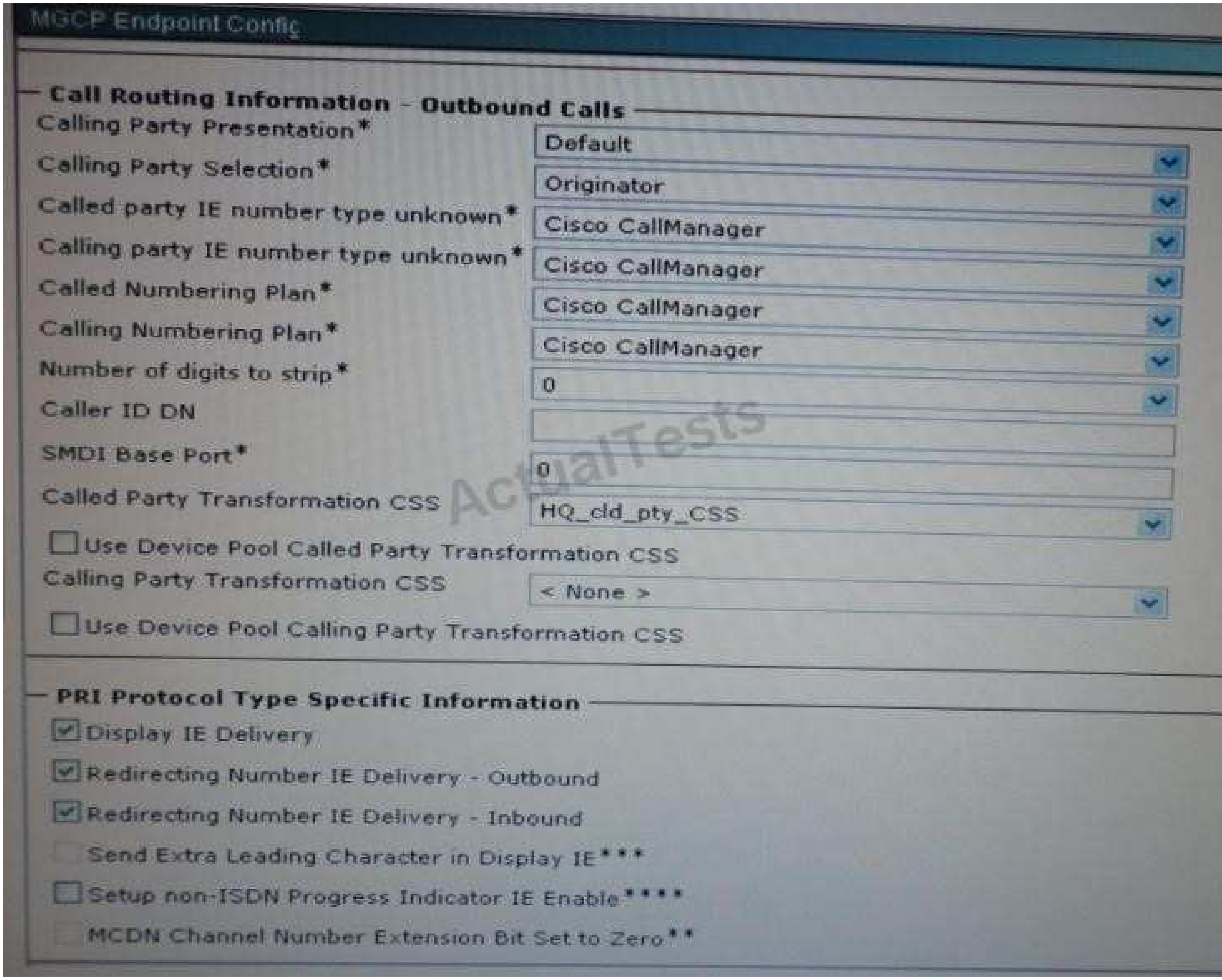
**105 (exhibit):**


MGCP Endpoint Config
Device Information
Product
Cisco MGCP E1 Port
Gateway
HQ
Device Protocol
Digital Access PRI
Registration
Registered with Cisco Unified Communications Manager 10.1.5.10
IP Address
10.1.5.1
Device is not trusted
End-Point Name *
S0/SU0/DS1-0@HQ
Description
S0/SU0/DS1-0@HQ
Device Pool*
Default
Common Device Configuration
< None >
Call Classification*
Use System Default
NetworkLocale
< None >
Packet Capture Mode*
None
Packet Capture Duration
0
Media Resource Group List
< None >
Location*
Hub_None
AAR Group
< None >
Load Information
Use Trusted Relay Point*
Default
Transmit UTF-8 for Calling Party Name
V150 (subset)
Enable Protected Facility IE
PSTN Access

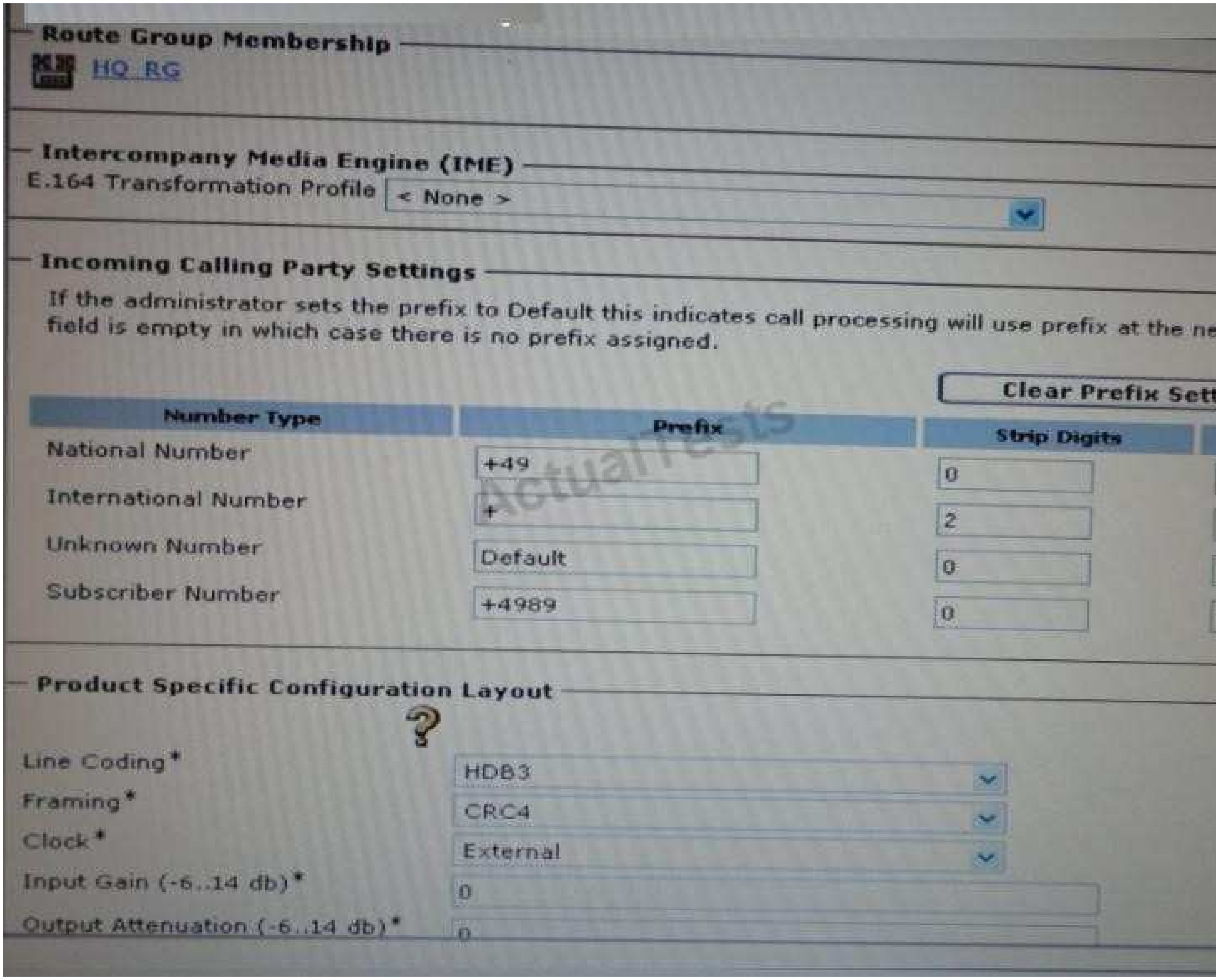
**106 (exhibit):**


MGCP Endpoint Config
Multilevel Precedence and Preemption (MLPP) Information
MLPP Domain
< None >
MLPP Indication
Default
MLPP Preemption
Default
Interface Information
PRI Protocol Type*
PRI EURO
QSIG Variant*
No Changes
ASN.1 ROSE OID Encoding*
No Changes
Protocol Side*
User
Channel Selection Order*
Top Down
Channel IE Type*
Use Number when 1B
PCM Type*
A-law
Delay for first restart (1/8 sec ticks)*
32
Delay between restarts (1/8 sec ticks)*
4
Inhibit restarts at PRI initialization
Enable status poll
Unattended Port
Enable G.Clear
Call Routing Information - Inbound Calls
Significant Digits*
4
Calling Search Space
Internal.CSS

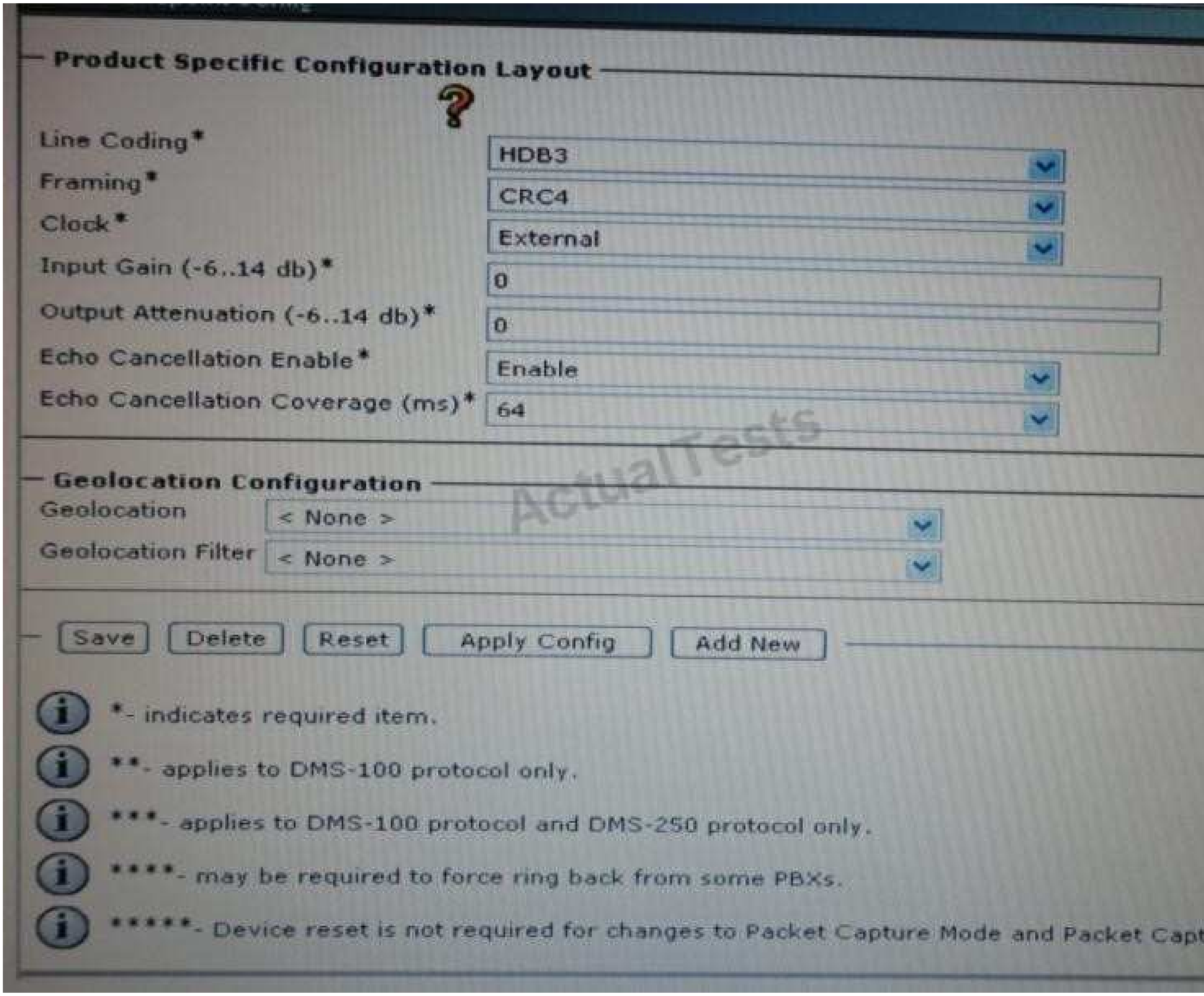
**107 (exhibit):**


MGCP Endpoint Config
Call Routing Information - Outbound Calls
Calling Party Presentation*
Default
Calling Party Selection*
Originator
Called party IE number type unknown*
Cisco CallManager
Calling party IE number type unknown*
Cisco CallManager
Called Numbering Plan*
Cisco CallManager
Calling Numbering Plan*
Cisco CallManager
Number of digits to strip*
0
Caller ID DN
SMDI Base Port*
0
Called Party Transformation CSS
HQ_cld_pty_CSS
Use Device Pool Called Party Transformation CSS
Calling Party Transformation CSS
< None >
Use Device Pool Calling Party Transformation CSS
PRI Protocol Type Specific Information
Display IE Delivery
Redirecting Number IE Delivery - Outbound
Redirecting Number IE Delivery - Inbound
Send Extra Leading Character in Display IE***
Setup non-ISDN Progress Indicator IE Enable****
MCDN Channel Number Extension Bit Set to Zero**

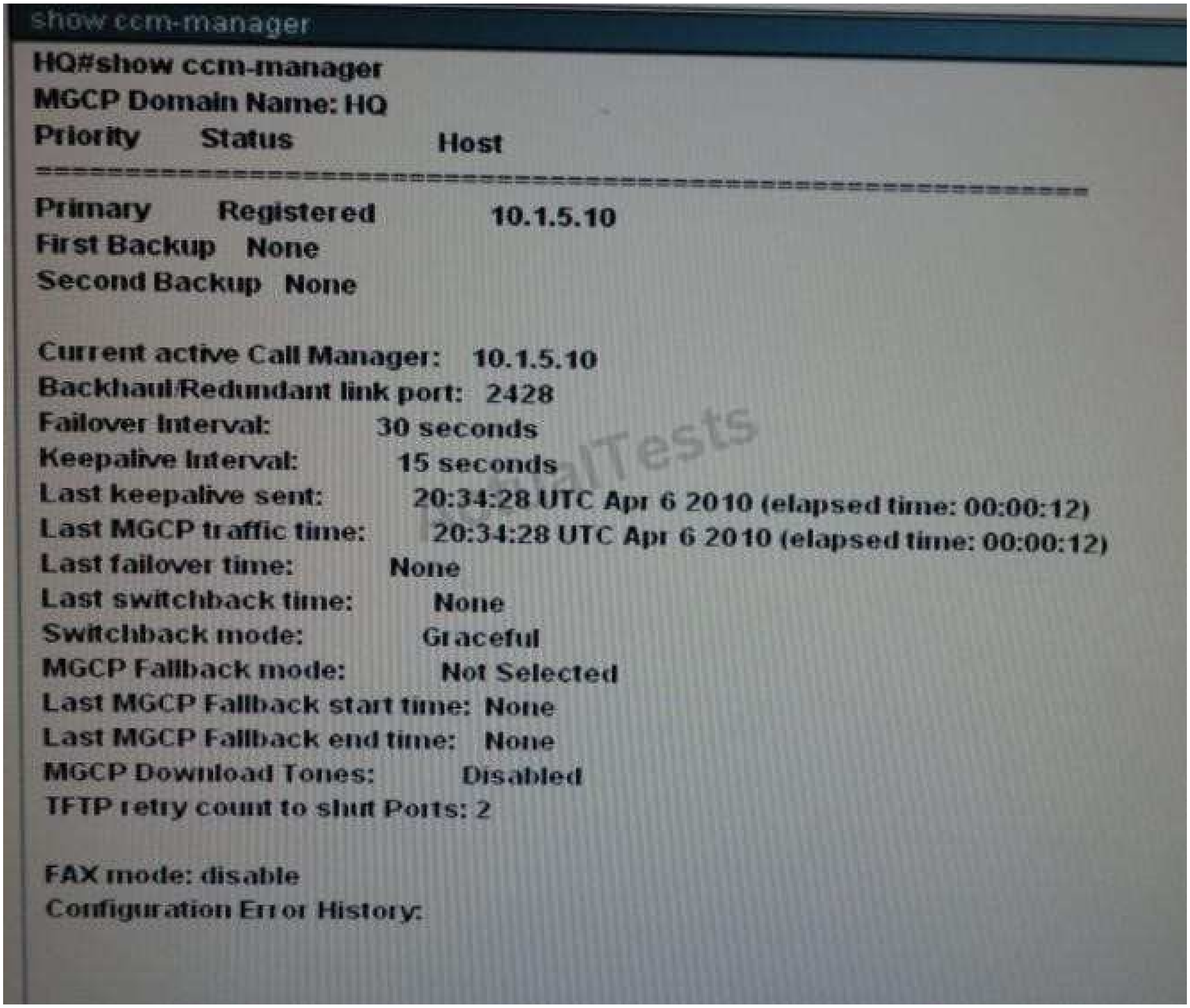
**108 (exhibit):**


Route Group Membership
HQ_RG
Intercompany Media Engine (IME)
E.164 Transformation Profile
< None >
Incoming Calling Party Settings
If the administrator sets the prefix to Default this indicates call processing will use prefix at the ne
field is empty in which case there is no prefix assigned.
Clear Prefix Sett
Number Type
Prefix
Strip Digits
National Number
+49
0
International Number
+
2
Unknown Number
Default
0
Subscriber Number
+4989
0
Product Specific Configuration Layout
Line Coding*
HDB3
Framing*
CRC4
Clock*
External
Input Gain (-6..14 db)*
0
Output Attenuation (-6..14 db)*
0

**109 (exhibit):**


Product Specific Configuration Layout
Line Coding*
HDB3
Framing*
CRC4
Clock*
External
Input Gain (-6..14 db)*
0
Output Attenuation (-6..14 db)*
0
Echo Cancellation Enable*
Enable
Echo Cancellation Coverage (ms)*
64
ActualTests
Geolocation Configuration
Geolocation
< None >
Geolocation Filter
< None >
Save
Delete
Reset
Apply Config
Add New
*- indicates required item.
**- applies to DMS-100 protocol only.
***- applies to DMS-100 protocol and DMS-250 protocol only.
****- may be required to force ring back from some PBXs.
*****- Device reset is not required for changes to Packet Capture Mode and Packet Capt

**110 (exhibit):**



```
show ccm-manager
HQ#show ccm-manager
MGCP Domain Name: HQ
Priority      Status          Host
===================================================================
Primary       Registered         10.1.5.10
First Backup    None
Second Backup   None

Current active Call Manager:    10.1.5.10
Backhaul/Redundant link port:   2428
Failover Interval:          30 seconds
Keepalive Interval:          15 seconds
Last keepalive sent:         20:34:28 UTC Apr 6 2010 (elapsed time: 00:00:12)
Last MGCP traffic time:       20:34:28 UTC Apr 6 2010 (elapsed time: 00:00:12)
Last failover time:         None
Last switchback time:          None
Switchback mode:            Graceful
MGCP Fallback mode:           Not Selected
Last MGCP Fallback start time:  None
Last MGCP Fallback end time:    None
MGCP Download Tones:          Disabled
TFTP retry count to shut Ports: 2

FAX mode: disable
Configuration Error History:
```

A. The MGCP gateway L3 timers need to be increased.
B. The backhaul link is not open.
C. The gateway is using partial PRI, which is not supported in MGCP mode.
D. Incoming dial-peer configuration is needed.
E. The hostname for the gateway does not match the configuration in Cisco Unified Communications Manager.

**Correct Answer:** A
**Section: (none)**
**Explanation**

**Explanation/Reference:**
**Explanation**:On the picture „show ccm-manager“ you can recognize that the backhaul link is not open. You can also see that the „isdn bind-l3 ccm-manager“ is missing on controller. That is the cause of issue. The answer should therefore be "The MGCP gateway L3 timers need to be increased"?http://www.cisco.com/en/US/tech/tk1077/technologies_configuration_example09186a00801ad22f.shtml--

**QUESTION 44**
Refer to the exhibit.

```
HQ1#show call active voice
-   snip -
OutSignalLevel=-15
InSignalLevel=-15
ERLLevel=25
-  snip –
```

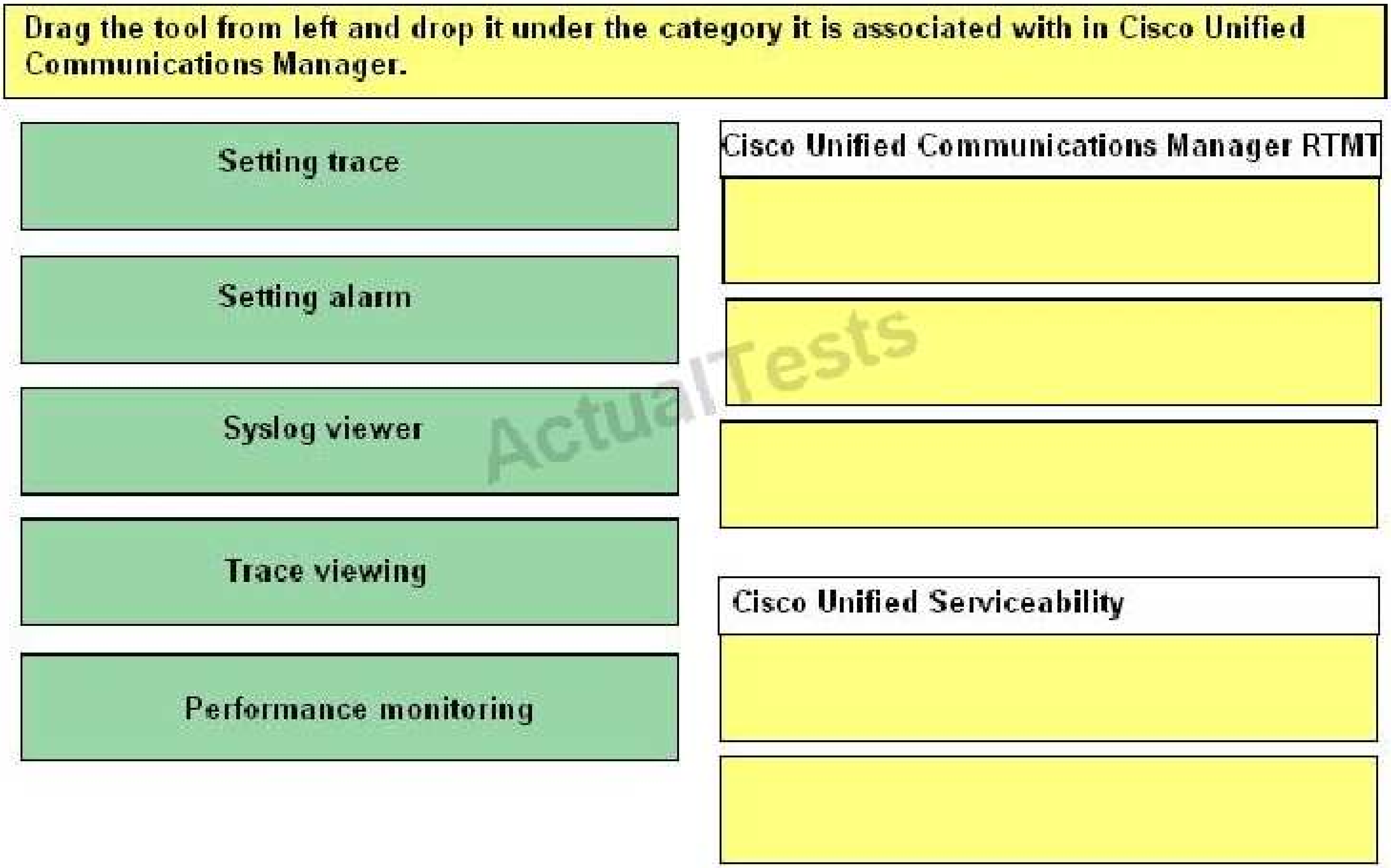
The output of the show call active voice command shows an ERL level of 25. Which statement about the ERL is true?

A. The ERL level is the absolute sum of OutSignalLevel and InSignalLevel, which equals 30. The DSP has an error margin of 5 dB.
B. ERL 25 shows how much echo cancellation has been performed by the echo canceller in the router.
C. The ERL value of 25 is not valid and hence should be ignored; however, the echo canceller is being organized.
D. The correct ERL could not be calculated by the echo canceller. The signal is being reflected with 0 dB loss.

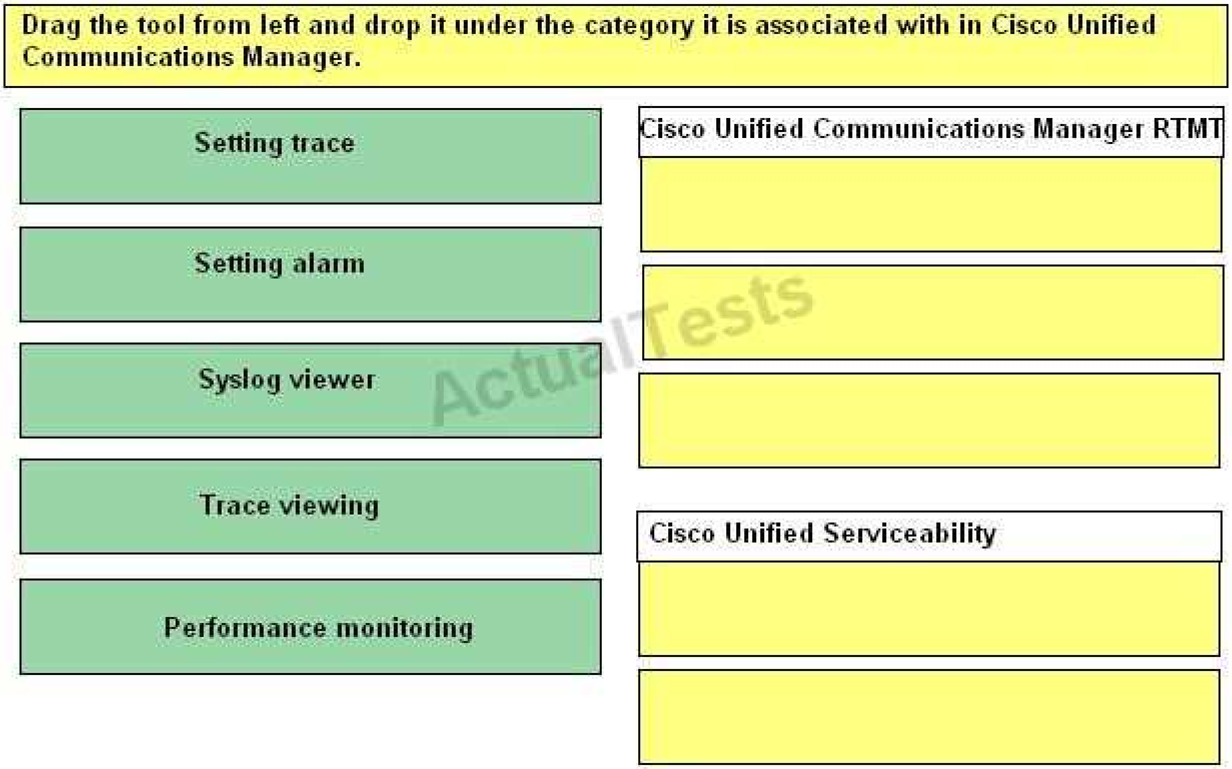
**Correct Answer:** D
**Section: (none)**
**Explanation**

**Explanation/Reference:**

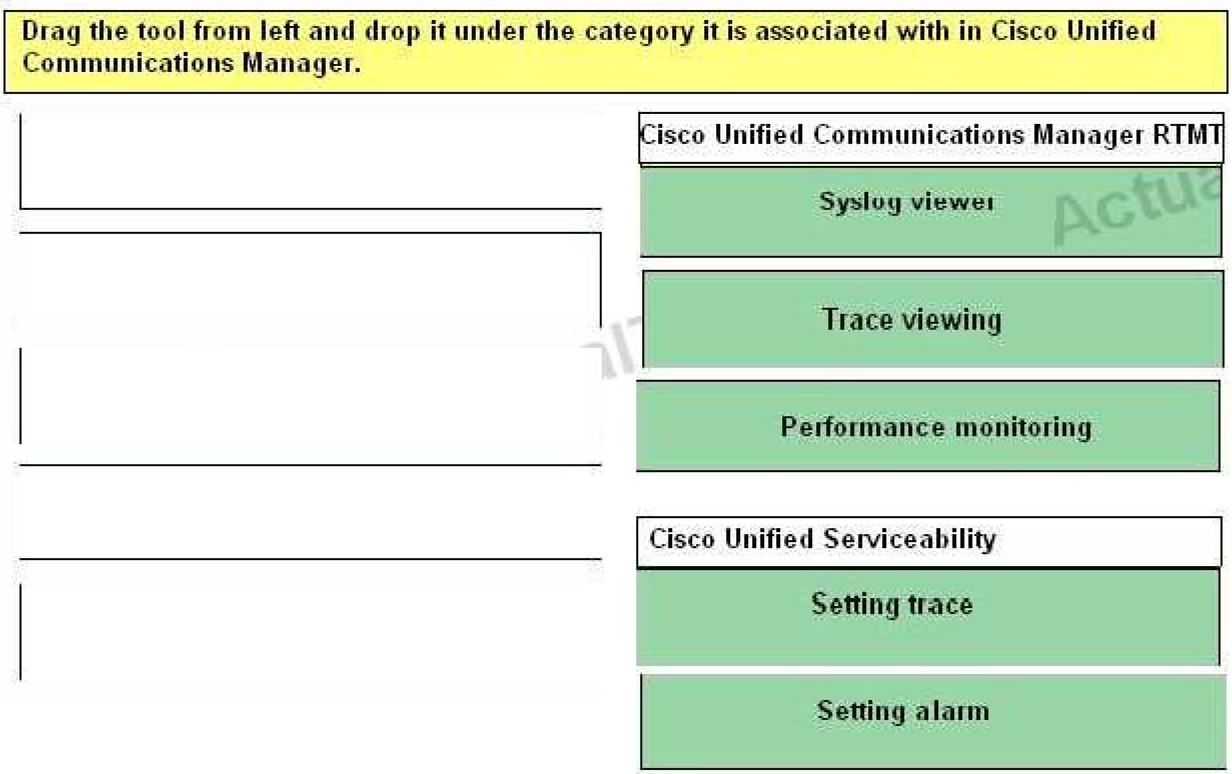
**QUESTION 45**




**Select and Place:**



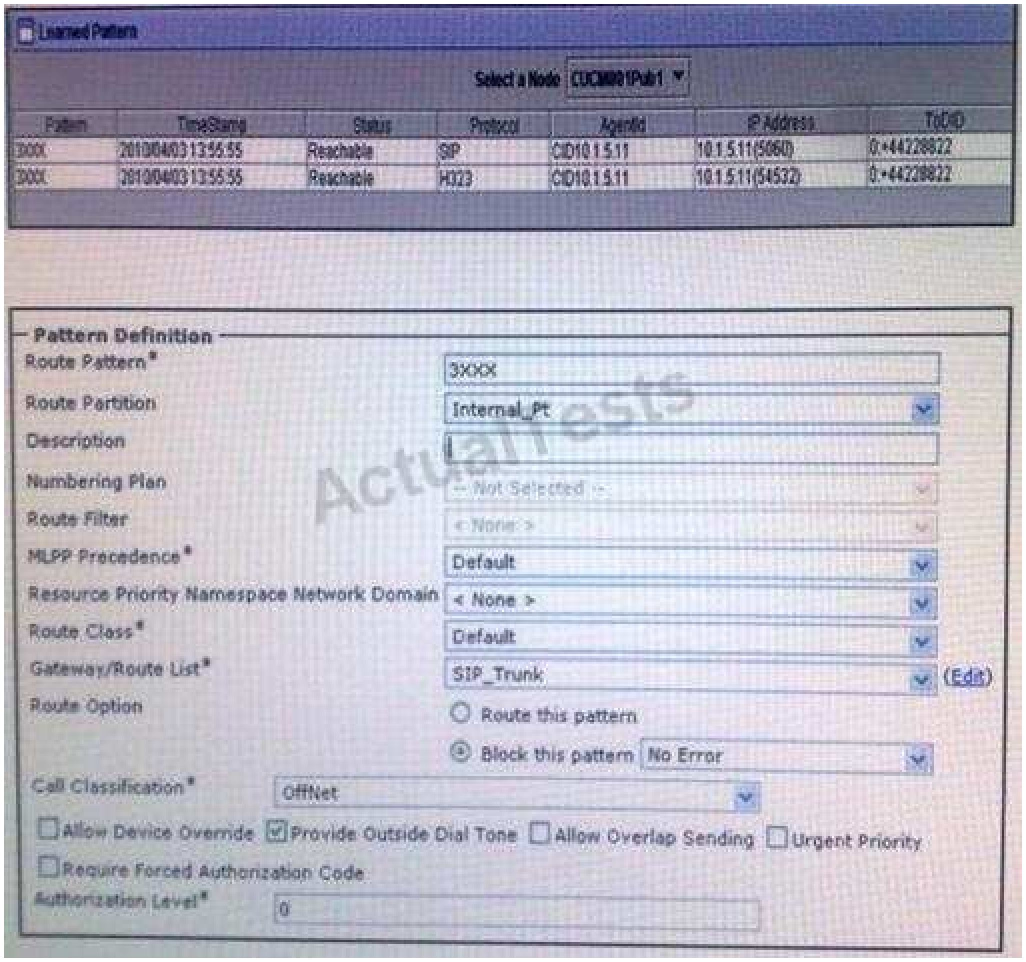

**Correct Answer:**




**Section: (none)**
**Explanation**

**Explanation/Reference:**
**Explanation**- CUCM RTMT tools are basically used to get different traces, logs and monitoring where serviceability is used for setting alarm to different levels and trace files.

**QUESTION 46**
## Refer to the Exhibits.




Refer to the exhibits Assume that all learned SAF routes are placed in the SAF_Pt partition. An IP phone CSS contains the following partitions in this order Internal_Pt, 3AF_Pt When the IP phone places a call to 3001. What will occur?

A. The call will succeed and will be placed via the SAF network SAF-learned routes always take precedence.
B. The call will fail because it will be blocked by the route pattern.
C. The call will be placed in a round-robin fashion between the SAF network and SIP_Trunk.
D. The call will be placed in a round-robin fashion between the SAF network and SIP_Trunk. Every other call will fail.

**Correct Answer:** B
**Section: (none)**
**Explanation**

**Explanation/Reference:**
**Explanation**: If partition is listed first in CSS, it has priority for equal qualified matches.If no single best match exist, the call-routing entry with the partition that is listed first in thecalling-device CSS is used.See CIPT II V II 5-61 and TVOICE V I 3-20

**QUESTION 47**
Which two troubleshooting tools would initially be the best to use when troubleshooting the PSTN gateway side of a call routing issue while using Cisco Unified Communications Manager? (Choose two)

A. RTMT trace output
B. Cisco IOS debug commands
C. Dialed Number Analyzer output
D. Cisco Unified Communications Manager alerts
E. Cisco IOS show commands

**Correct Answer:** BE
**Section: (none)**
**Explanation**

**Explanation/Reference:**
**Link**- http://www.cisco.com/en/US/docs/voice_ip_comm/cucm/service/5_0_1/ccmsrva/sartmt.html http://www.cisco.com/en/US/docs/voice_ip_comm/cucm/dna/5_0_4/dnai.html

**QUESTION 48**
**Refer to the Exhibit.**

```
router eigrp SAF
!
Service-family ipv4 autonomous-system 1
!
topology base
exit-sf-topology
exit-service-family
!
```

```
router eigrp CUCME
!
Service-family ipv4 autonomous-system 1
!
topology base
exit-sf-topology
exit-service-family
!
```

Assuming that the two Cisco SAF Forwarders are adjacent to each other and that no SAF clients have been configured, which statement is true?

A. The Cisco SAF Forwarders will not establish a neighbor relationship because the service-family external-client configuration is missing
B. The Cisco SAF Forwarders will not establish a neighbor relationship because the eigrp label CUCME should be replaced with SAF
C. The Cisco SAF Forwarders will not establish a neighbor relationship because the service-family external-client configuration is missing as well as the static neighbor configurations
D. The Cisco SAF Forwarders will establish a neighbor relationship. No further configuration is required.
E. Cisco SAF Forwarders will not establish a neighbor relationship until the SAF clients are configured and registered to the Cisco SAF Forwarders

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

**Explanation/Reference:**
**Explanation**: EIGRP label should match with these two routers otherwise SAF forwarded doesn't establish neighbor relationship. EIGRP process ID has only local significance. See CIPT II V2 5-35

**QUESTION 49**
Which three types of information are included in an alarm? (Choose three)

A. explanation of event and recommended action
B. application type, machine location, and cluster location
C. an ordered list of locations to which an alarm was sent
D. application name, machine name, and cluster name
E. user-defined text that can be custom-defined
F. the list of alarms and alarm levels for each location to which the alarm is sent

**Correct Answer:** ADE
**Section: (none)**
**Explanation**

**Explanation/Reference:**

**QUESTION 50**
The CLI utils dbreplication reset all was run on a Cisco Unified Communications Manager. However, replication failed to restart. Which course of action should be taken to resolve this issue?

A. Restart the replication from the Cisco Unified Communications Manager RTMT tool.
B. Issue the CLI command utils dbreplication runtimestate
C. Issue the CLI command utils dbreplication reset
D. Issue the CLI command utils dbreplication clusterreset
E. Issue the CLI command utils dbreplication clusterreset all

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

**Explanation/Reference:**
Database Replication Does Not Occur When Connectivity Is Restored on Lost Node
Symptom Database replication does not occur when connectivity is restored on lost node
recovery. You can verify the state of replication by using the methods given in the topic Replication Fails Between the Publisher and the Subscriber. Only use the following procedure if you have already tried to reset replication on the node, and have been unsuccessful.

Possible Cause: The CDR check remains stuck in a loop, due to a delete on device table. Recommended Action

Step1 Run utils dbreplication stop on the affected subscribers. You can run them all at once.

Step2 Wait until Step1completes, then, run utils dbreplication stop on the affected publisher server.

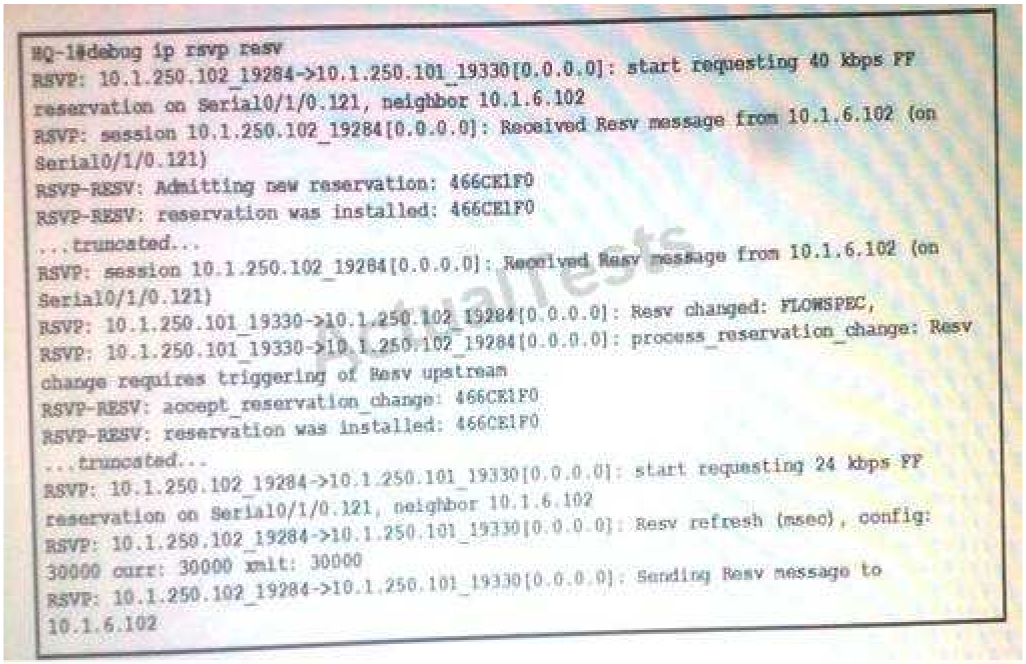
Step3 Run utils dbreplication clusterreset from the affected publisher server. When you run the command, the log name gets listed in the log file. Watch this file to monitor the process status. The path to the follows:**/var/log/active/cm/trace/dbl/sdi**

Step4 From the affected publisher, runutils dbreplication reset all.

Step5 Stop and restart all the services on all the subscriber servers [or restart/reboot all the systems(subscriber servers)] in the cluster to get the service changes. Do this only afterutils dbreplication status shows Status 2.

**Link**-http://www.cisco.com/en/US/docs/voice_ip_comm/cucm/trouble/7_0_1/tbsystem.html

**QUESTION 51**
## Refer to the exhibit.



```
HQ-1#debug ip rsvp resv
RSVP: 10.1.250.102_19284->10.1.250.101_19330[0.0.0.0]: start requesting 40 kbps FF
reservation on Serial0/1/0.121, neighbor 10.1.6.102
RSVP: session 10.1.250.102_19284[0.0.0.0]: Received Resv message from 10.1.6.102 (on
Serial0/1/0.121)
RSVP-RESV: Admitting new reservation: 466CE1F0
RSVP-RESV: reservation was installed: 466CE1F0
...truncated...
RSVP: session 10.1.250.102_19284[0.0.0.0]: Received Resv message from 10.1.6.102 (on
Serial0/1/0.121)
RSVP: 10.1.250.101_19330->10.1.250.102_19284[0.0.0.0]: Resv changed: FLOWSPEC,
RSVP: 10.1.250.101_19330->10.1.250.102_19284[0.0.0.0]: process_reservation_change: Resv
change requires triggering of Resv upstream
RSVP-RESV: accept_reservation_change: 466CE1F0
RSVP-RESV: reservation was installed: 466CE1F0
...truncated...
RSVP: 10.1.250.102_19284->10.1.250.101_19330[0.0.0.0]: start requesting 24 kbps FF
reservation on Serial0/1/0.121, neighbor 10.1.6.102
RSVP: 10.1.250.102_19284->10.1.250.101_19330[0.0.0.0]: Resv refresh (msec), config:
30000 curr: 30000 xmit: 30000
RSVP: 10.1.250.102_19284->10.1.250.101_19330[0.0.0.0]: Sending Resv message to
10.1.6.102
```

The exhibit shows the output of a successful RSVP call setup that uses G.729 codec. What should the minimum bandwidth configuration on interface serial 0/1/0.121 for the command ip rsvp bandwidth be?

A. 8 Kb/S
B. 16 Kb/S
C. 24 Kb/S
D. 32 Kb/S
E. 40 Kb/S

**Correct Answer:** E
**Section: (none)**
**Explanation**

**Explanation/Reference:**
**Explanation**-The call setup message indicate that Per this link it should be 24Kb/sbut RSVP Reservation adds 16k on top (which is not used really and released after the booking), so 24 +16 = 40k http://fengnet.com/book/voip/ch08lev1sec2.html

**QUESTION 52**
What do you do next if after observing the result of your troubleshooting, the problem still exists?

A. Implement Acton Plan
B. Define the Problem
C. Consider the Possibilities
D. Create Action Plan
E. Gather Facts
F. Observe Results

G. Restart Problem-Solving

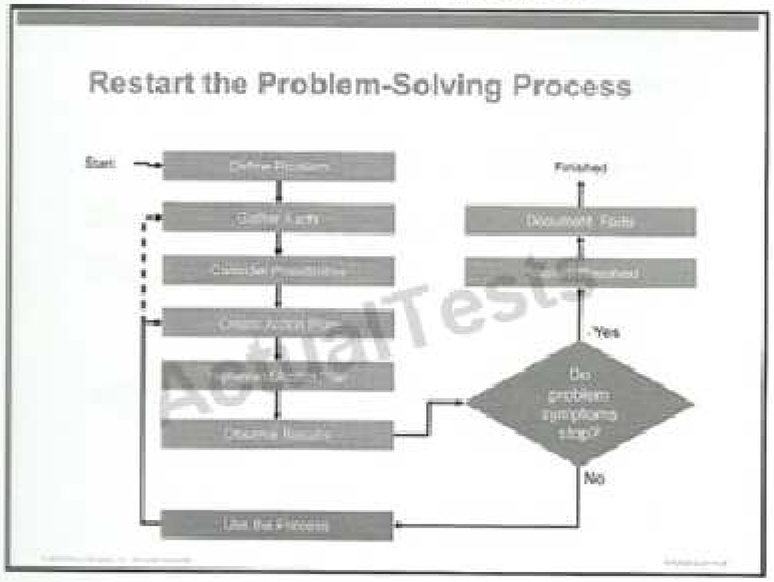
H. Process Problem Resolved

**Correct Answer:** C
**Section: (none)**
**Explanation**

**Explanation/Reference:**
See CIPT I V 1 1-29. You will see that we continue to make in the process.

## Restart the Problem-Solving Process

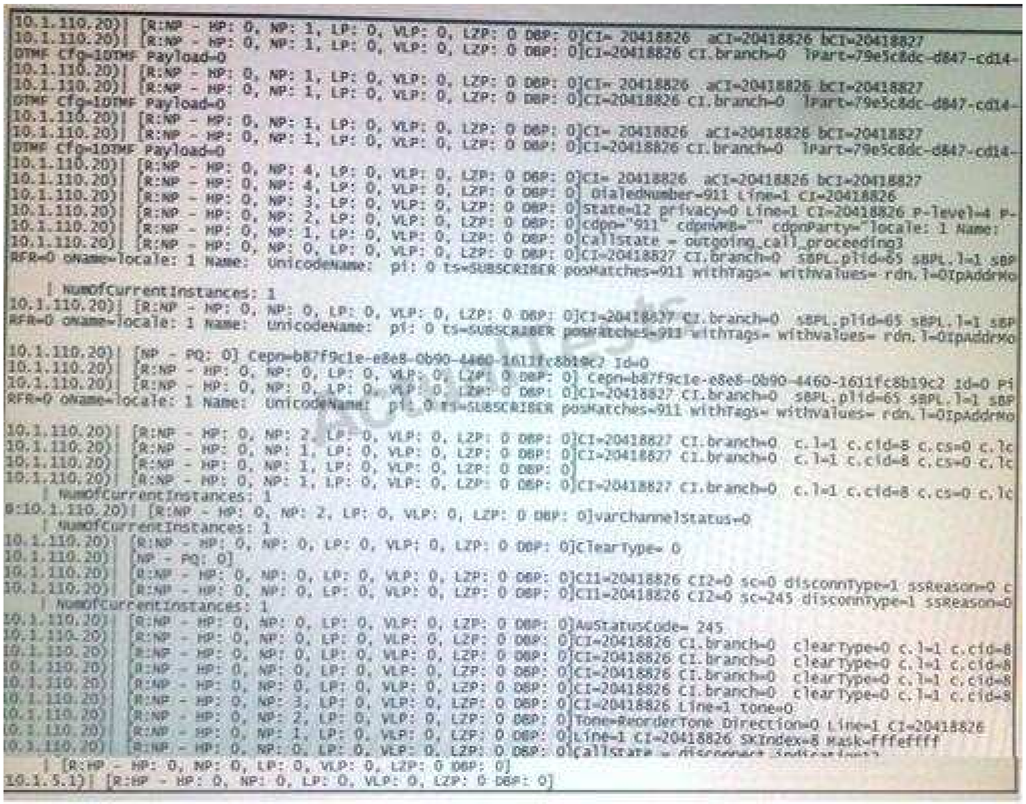
This topic explains what to do if the troubleshooting task fails.




After you observe the results and determine that the problem still exists, restart the problem-solving process with the remaining possibilities that you identified when gathering facts. With the result of the last action plan, you can narrow the possibilities. Your narrowing of the possibilities should be an ongoing process.

If the previous action plan is a valid action that results in a desirable configuration, but does no solve the root cause, then leave the action implemented and create another action plan. If the previous action does not solve the problem, and you do not consider the results a desired permanent state, then back out of the action before you reiterate the process of creating a new action plan.

**QUESTION 53**



```
10.1.110.20)| [R:NP - HP: 0, NP: 1, LP: 0, VLP: 0, LZP: 0 DBP: 0]CI= 20418826  aCI=20418826 bCI=20418827
10.1.110.20)| [R:NP - HP: 0, NP: 1, LP: 0, VLP: 0, LZP: 0 DBP: 0]CI=20418826 CI.branch=0  lPart=79e5c8dc-d847-cd14-
DTMF Cfg=1DTMF Payload=0
10.1.110.20)| [R:NP - HP: 0, NP: 1, LP: 0, VLP: 0, LZP: 0 DBP: 0]CI= 20418826  aCI=20418826 bCI=20418827
10.1.110.20)| [R:NP - HP: 0, NP: 1, LP: 0, VLP: 0, LZP: 0 DBP: 0]CI=20418826 CI.branch=0  lPart=79e5c8dc-d847-cd14-
DTMF Cfg=1DTMF Payload=0
10.1.110.20)| [R:NP - HP: 0, NP: 1, LP: 0, VLP: 0, LZP: 0 DBP: 0]CI= 20418826  aCI=20418826 bCI=20418827
10.1.110.20)| [R:NP - HP: 0, NP: 1, LP: 0, VLP: 0, LZP: 0 DBP: 0]CI=20418826 CI.branch=0  lPart=79e5c8dc-d847-cd14-
DTMF Cfg=1DTMF Payload=0
10.1.110.20)| [R:NP - HP: 0, NP: 4, LP: 0, VLP: 0, LZP: 0 DBP: 0]CI= 20418826  aCI=20418826 bCI=20418827
10.1.110.20)| [R:NP - HP: 0, NP: 4, LP: 0, VLP: 0, LZP: 0 DBP: 0] DialedNumber=911 Line=1 CI=20418826
10.1.110.20)| [R:NP - HP: 0, NP: 3, LP: 0, VLP: 0, LZP: 0 DBP: 0]State=12 privacy=0 Line=1 CI=20418826 P-level=4 P-
10.1.110.20)| [R:NP - HP: 0, NP: 2, LP: 0, VLP: 0, LZP: 0 DBP: 0]cdpn="911" cdpnVMB="" cdpnParty="locale: 1 Name:
10.1.110.20)| [R:NP - HP: 0, NP: 1, LP: 0, VLP: 0, LZP: 0 DBP: 0]callstate = outgoing_call_proceeding3
10.1.110.20)| [R:NP - HP: 0, NP: 0, LP: 0, VLP: 0, LZP: 0 DBP: 0]CI=20418827 CI.branch=0  sBPL.plid=65 sBPL.l=1 sBP
RFR=0 oName=locale: 1 Name:  UnicodeName:  pi: 0 ts=SUBSCRIBER posMatches=911 withTags= withValues= rdn.l=0IpAddrMo

   | NumOfCurrentInstances: 1
10.1.110.20)| [R:NP - HP: 0, NP: 0, LP: 0, VLP: 0, LZP: 0 DBP: 0]CI=20418827 CI.branch=0  sBPL.plid=65 sBPL.l=1 sBP
RFR=0 oName=locale: 1 Name:  UnicodeName:  pi: 0 ts=SUBSCRIBER posMatches=911 withTags= withValues= rdn.l=0IpAddrMo

10.1.110.20)| [NP - PQ: 0] Cepn=b87f9c1e-e8e8-0b90-4460-1611fc8b19c2 Id=0
10.1.110.20)| [R:NP - HP: 0, NP: 0, LP: 0, VLP: 0, LZP: 0 DBP: 0] Cepn=b87f9c1e-e8e8-0b90-4460-1611fc8b19c2 Id=0 Pi
10.1.110.20)| [R:NP - HP: 0, NP: 0, LP: 0, VLP: 0, LZP: 0 DBP: 0]CI=20418827 CI.branch=0  sBPL.plid=65 sBPL.l=1 sBP
RFR=0 oName=locale: 1 Name:  UnicodeName:  pi: 0 ts=SUBSCRIBER posMatches=911 withTags= withValues= rdn.l=0IpAddrMo

10.1.110.20)| [R:NP - HP: 0, NP: 2, LP: 0, VLP: 0, LZP: 0 DBP: 0]CI=20418827 CI.branch=0  c.l=1 c.cid=8 c.cs=0 c.lc
10.1.110.20)| [R:NP - HP: 0, NP: 1, LP: 0, VLP: 0, LZP: 0 DBP: 0]CI=20418827 CI.branch=0  c.l=1 c.cid=8 c.cs=0 c.lc
10.1.110.20)| [R:NP - HP: 0, NP: 1, LP: 0, VLP: 0, LZP: 0 DBP: 0]
10.1.110.20)| [R:NP - HP: 0, NP: 1, LP: 0, VLP: 0, LZP: 0 DBP: 0]CI=20418827 CI.branch=0  c.l=1 c.cid=8 c.cs=0 c.lc
   | NumOfCurrentInstances: 1
8:10.1.110.20)| [R:NP - HP: 0, NP: 2, LP: 0, VLP: 0, LZP: 0 DBP: 0]varChannelStatus=0
   | NumOfCurrentInstances: 1
10.1.110.20)| [R:NP - HP: 0, NP: 0, LP: 0, VLP: 0, LZP: 0 DBP: 0]ClearType= 0
10.1.110.20)| [NP - PQ: 0]
10.1.110.20)| [R:NP - HP: 0, NP: 0, LP: 0, VLP: 0, LZP: 0 DBP: 0]CI1=20418826 CI2=0 sc=0 disconnType=1 ssReason=0 c
10.1.110.20)| [R:NP - HP: 0, NP: 0, LP: 0, VLP: 0, LZP: 0 DBP: 0]CI1=20418826 CI2=0 sc=245 disconnType=1 ssReason=0
   | NumOfCurrentInstances: 1
10.1.110.20)| [R:NP - HP: 0, NP: 0, LP: 0, VLP: 0, LZP: 0 DBP: 0]AuStatusCode= 245
10.1.110.20)| [R:NP - HP: 0, NP: 0, LP: 0, VLP: 0, LZP: 0 DBP: 0]CI=20418826 CI.branch=0  clearType=0 c.l=1 c.cid=8
10.1.110.20)| [R:NP - HP: 0, NP: 0, LP: 0, VLP: 0, LZP: 0 DBP: 0]CI=20418826 CI.branch=0  clearType=0 c.l=1 c.cid=8
10.1.110.20)| [R:NP - HP: 0, NP: 0, LP: 0, VLP: 0, LZP: 0 DBP: 0]CI=20418826 CI.branch=0  clearType=0 c.l=1 c.cid=8
10.1.110.20)| [R:NP - HP: 0, NP: 0, LP: 0, VLP: 0, LZP: 0 DBP: 0]CI=20418826 CI.branch=0  clearType=0 c.l=1 c.cid=8
10.1.110.20)| [R:NP - HP: 0, NP: 3, LP: 0, VLP: 0, LZP: 0 DBP: 0]CI=20418826 Line=1 tone=0
10.1.110.20)| [R:NP - HP: 0, NP: 2, LP: 0, VLP: 0, LZP: 0 DBP: 0]Tone=ReorderTone Direction=0 Line=1 CI=20418826
10.1.110.20)| [R:NP - HP: 0, NP: 1, LP: 0, VLP: 0, LZP: 0 DBP: 0]Line=1 CI=20418826 SKIndex=8 Mask=fffeffff
10.1.110.20)| [R:NP - HP: 0, NP: 0, LP: 0, VLP: 0, LZP: 0 DBP: 0]Callstate = disconnect indication
   | [R:HP - HP: 0, NP: 0, LP: 0, VLP: 0, LZP: 0 DBP: 0]
10.1.5.1)| [R:HP - HP: 0, NP: 0, LP: 0, VLP: 0, LZP: 0 DBP: 0]
```

Refer to the exhibit. Which two pieces of information can you gather from the Cisco Unified Communications Manager trace file shown in the exhibit? (Choose two)

A. The calling number is 2001
B. The call proceeded normally
C. The calling number is 911
D. The called number is 911
E. The called number is 2001
F. The call did not complete

**Correct Answer:** DF
**Section: (none)**
**Explanation**

**Explanation/Reference:**

**QUESTION 54**

## Refer to the Exhibit

```
voice service saf
 profile trunk-route 1
  session protocol sip interface Loopback1 transport tcp port 5060
 !
 profile dn-block 1
  pattern 1 type global 1408[2-9]XXXXXX
 !
 profile callcontrol 1
  dn-service
   trunk-route 1
   dn-block 1
 !
 channel 1 vrouter SAF asystem 1
  subscribe callcontrol wildcarded
  publish callcontrol 1
```

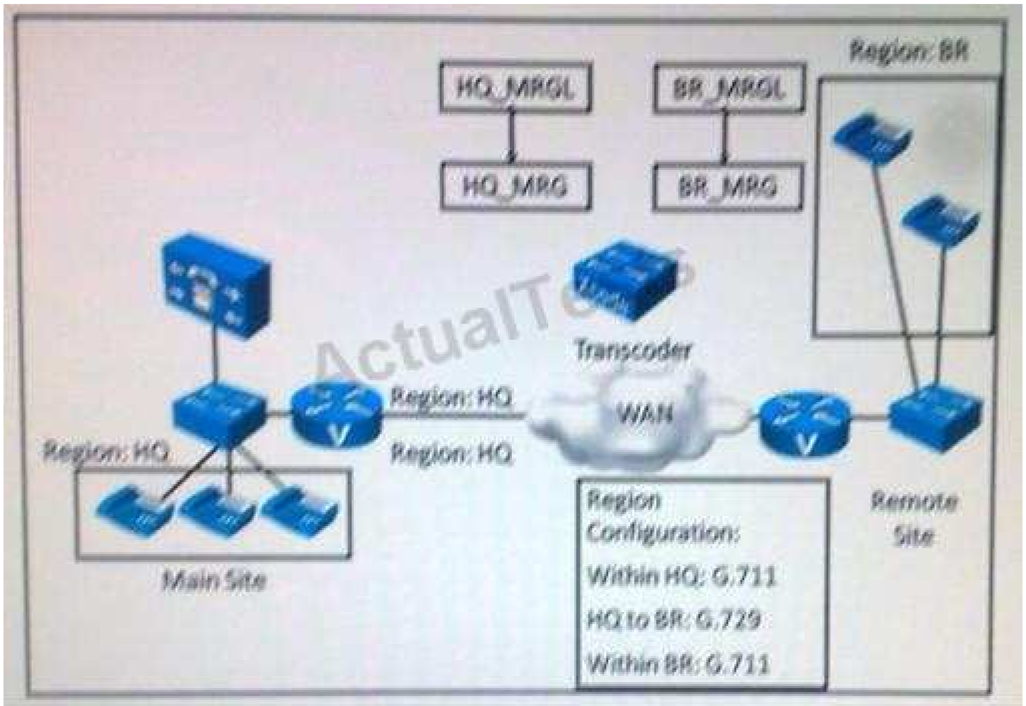
When a Cisco Unified Communications Manager Express advertises directory number pattern in the exhibit. What would the learned pattern be in the RTMT tool on the Cisco Unified Communication Manager?

A. \+1408[2-9]XXXXXX and the ToDID will be 0:
B. 1408[2-9]XXXXXX and the ToDID will be 0:+1408[2-9]XXXXXX
C. \+1408[2-9]XXXXXX and the ToDID will be 0:+1408[2-9]XXXXXX
D. \+1408[2-9]XXXXXX and the ToDID will be empty
E. [2-9] XXXXXX and the ToDID will be 0:+1408[2-9]XXXXXX

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

**Explanation/Reference:**
The only way to advertise a pattern with + is to use the pattern tag type global command instead the pattern tag type extension. In this case however, ToDID is always unset, regardless of the configured alias prefix at the directory number block profile. See CIPT II V V 5-62The answer is:\+1408[2-9] XXXXXX and the ToDID will be empty

**QUESTION 55**
## Refer to the exhibit




HQ_MPQL is assigned to the HQ IP phones. BR_MRPGL is assigned to the BR IP phones. The remote site BR IP phones only support G711 codec. When a call is placed from HQ phone, the call fails. Which statement indicates how this issue is resolved?

A. Configure the Transcoder at the HQ site and assign it to HQ_MRG
B. Configure the Transcoder at the BR site and assign it to BR_MRG
C. The Transcoder should be assigned to its own MRG, which should then be assigned to the default device pool at HQ.
D. A Transcoder is not needed. The HQ phones will automatically change over to G 711 codec.

**Correct Answer:** B
**Section: (none)**
**Explanation**

**Explanation/Reference:**
The phone in BR ony support G711. The region configuration only allow G729 for HQ to BR. So you need a transcoder in BR.

**QUESTION 56**
**Refer to the exhibit.**

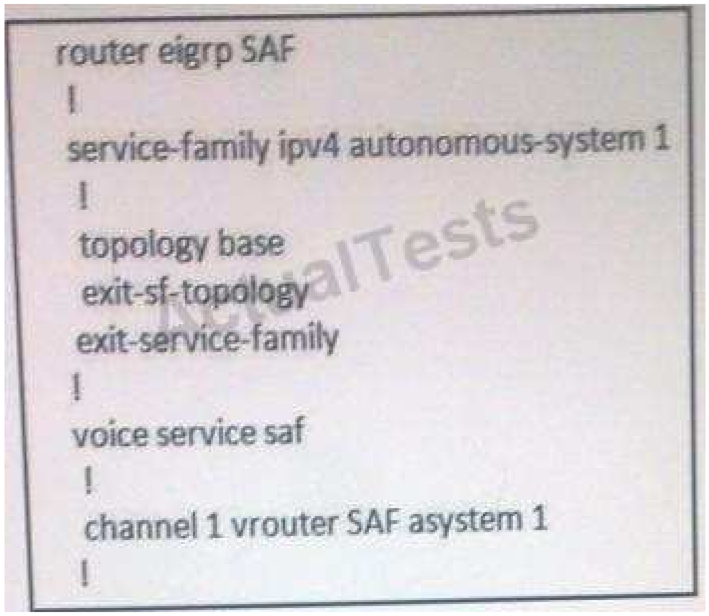

Which Cisco Unified Communications Manager trace file level should be selected when enabling traces to send to Cisco TAC for analysis?

A. State Transition
B. Arbitrary
C. Significant
D. Error
E. Detailed
F. Special

**Correct Answer:** E
**Section: (none)**
**Explanation**

**Explanation/Reference:**
**Explanation**- Cisco TAC always request for detailed traces.

**QUESTION 57**
**Refer to the exhibit**



```
router eigrp SAF
 !
 service-family ipv4 autonomous-system 1
  !
  topology base
  exit-sf-topology
 exit-service-family
 !
 voice service saf
  !
  channel 1 vrouter SAF asystem 1
  !
```

Which statement is true about the Cisco SAF Forwarder configuration on a Cisco Unified Communications Manager Express?

A. The Cisco Unified Communications Manager Express will not be able to register with the Cisco SAF Forwarder because the service-family external-client configuration is missing.
B. The Cisco Unified Communications Manager Express will be able to register with the Cisco SAF Forwarder
C. The QSCC Unified Communications Manager Express will not be able to register with the Cisco SAF Forwarder because the trunk configuration under voice service saf is missing
D. The Cisco Unified Communications Manager Express will not be able to register with the Cisco SAF Forwarder because call control services not been configured
E. The Cisco Communications Manager Express will be able to register with the Cisco SAF Forwarder, but an error message will be displayed regarding the missing trunk configuration

**Correct Answer:** B
**Section: (none)**
**Explanation**

**Explanation/Reference:**
**Explanation**- configuration is ok hence CUCME will be able to register with SAF forwarder. **Link**- http://www.cisco.com/en/US/docs/voice_ip_comm/cucme/feature/guide/SAF_FeatureModule.html#wp1235309

**QUESTION 58**
When the command utils dbreplication status is executed on the Cisco Unified Communications Manager CLI, which step should be taken next to check the database replication status?

A. View the activelog file
B. Run the same command on all nodes of the cluster
C. Restart the Cisco CallManager service.
D. The command utils dbreplication runtimetstate must be run on the publisher.
E. The command utils dbreplication runtimestate must be run on the subscriber.

**Correct Answer:** A
**Section: (none)**
**Explanation**

**Explanation/Reference:**

**QUESTION 59**

Which QoS methodology combines strict priority queuing with class-based weighted fair queuing?

A. IP RTP Priority
B. Multilink
C. IP Frame Relay RTP Priority
D. RSVP
E. LLQ

**Correct Answer:** E
**Section: (none)**
**Explanation**

**Explanation/Reference:**
**Explanation: Link-** http://www.cisco.com/en/US/tech/tk543/tk544/tk399/tsd_technology_support_sub-protocol_home.html

**QUESTION 60**

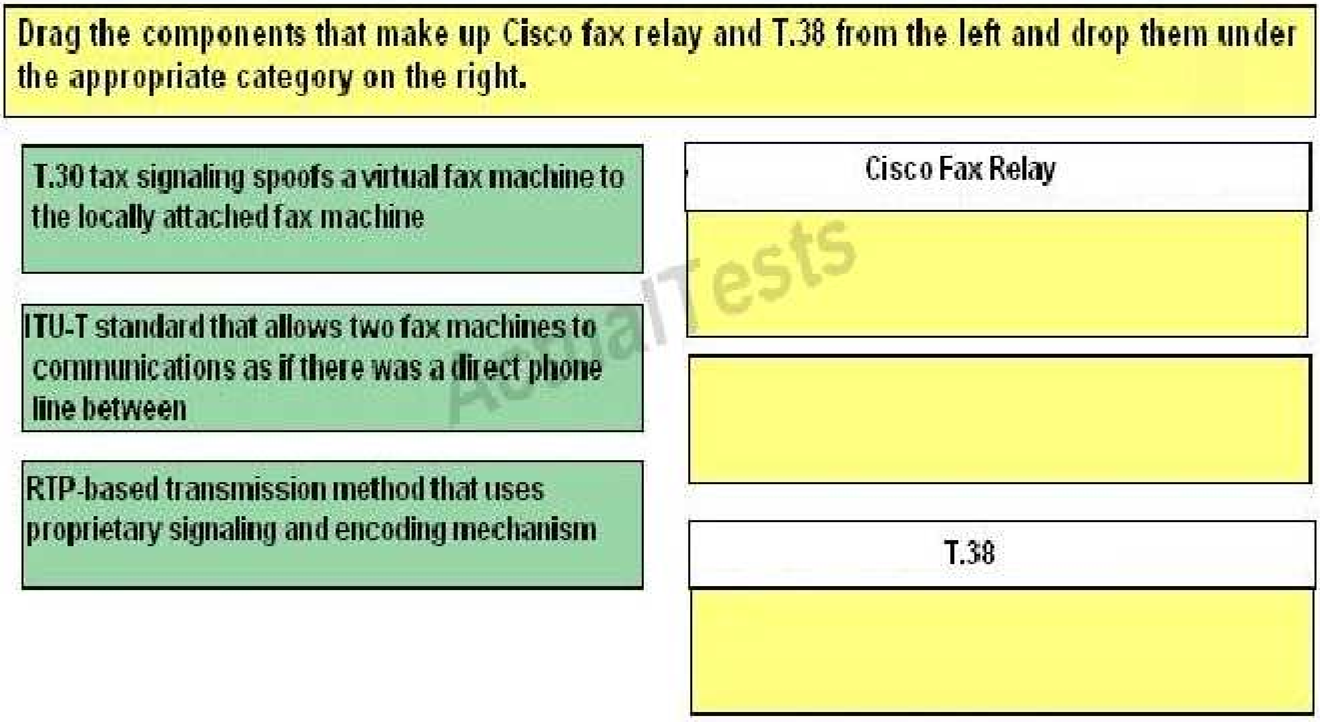
How does LLQ ensure that voice traffic is always expedited?

A. LLQ adds a strict priority class to CBWQF. The class allows delay-sensitive data such as voice to be dequeued and sent first.
B. LLQ uses CBWFQ to prioritize voice traffic and dequeue the voice packets so that they can be handled first.
C. The strict priority queue has a higher weight than the queues in CBWFQ. This weight allows the delay sensitive data such as voice to be dequeued and sent first.
D. The LLQ strict priority queue allows delay-sensitive data such as voice to be dequeued and sent first (before packets in order queue are dequeued), giving delay sensitive data preferential treatment over other traffic.

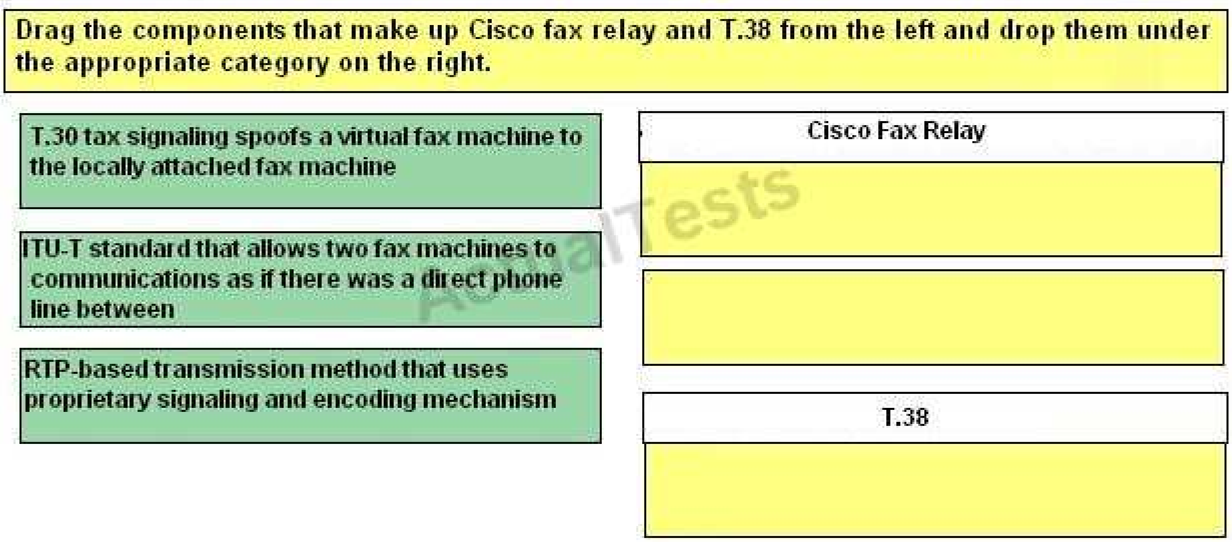
**Correct Answer:** D
**Section: (none)**
**Explanation**

**Explanation/Reference:**
**Explanation: Link**-http://en.wikipedia.org/wiki/Low_Latency_Queuing

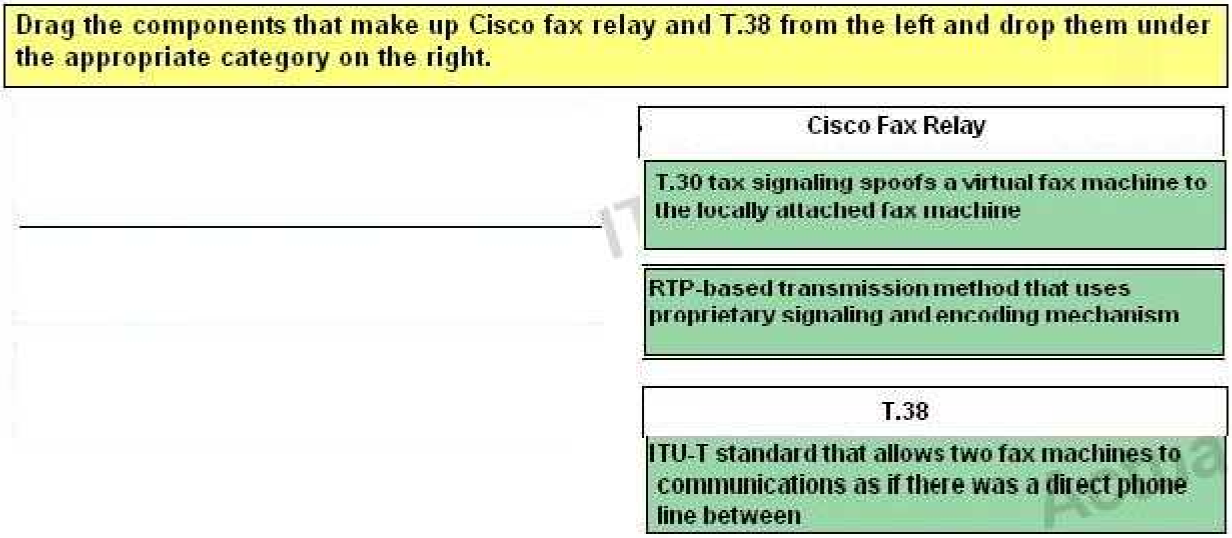
**QUESTION 61**




**Select and Place:**




**Correct Answer:**




**Section: (none)**
**Explanation**

**Explanation/Reference:**

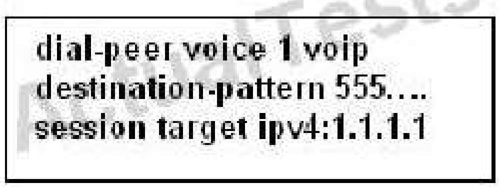
**QUESTION 62**
Which three Cisco IOS commands are required to configure a voice gateway as a DHCP server to support a data subnet with the IP address of 101.30.0/24 default gateway of 10.1.30.1/24? (Choose three)

A. ip dhcp pool
B. subnet 10.1.30.1 255.255.255.0
C. ip dhcp pool data
D. network 10.1.30.1/24
E. network 10.1.30.0 255.255.255.0
F. default-gw 10.1.30.1.24
G. default-router 10.1.30.1

**Correct Answer:** CEG
**Section: (none)**
**Explanation**

**Explanation/Reference:**
**Explanation**- Per question, CGE are required commands to configure a voice gateway. **Link**- http://www.cisco.com/en/US/docs/routers/access/vg202_vg204/software/2_vg4_voip_ps2250_TSD_Products_Configuration_Guide_Chapter.html

**QUESTION 63**
**Refer to the exhibit.**



When 5551234 is being matched with the outgoing dial peer that is shown in the exhibit, which of the following called numbers will be sent to the VoIP network?

A. 5551234
B. 1234
C. 555
D. Null
E. 5
F. 51234

**Correct Answer:** A
**Section: (none)**
**Explanation**

**Explanation/Reference:**
**Explanation**- no rules are applied in this dial peer hence calling number will not be changed and same number will be sent to VoIP network.

**QUESTION 64**
When a Cisco Unified Border Element connects two VoIP streams using flow-around media, which of the following options describe the components of the call that flow around and the components that flow through the device?

A. All security information flows through the Cisco Unified Border Element, and all call signaling and RTP flows around the device.
B. Call signaling flows through and call media flows around the device
C. Call media flows through and call signaling flows around the device.
D. The initial call-signaling traffic flows through the device to initiate the call and then all subsequent calls flow around the device.

**Correct Answer:** B
**Section: (none)**
**Explanation**

**Explanation/Reference:**
**Explanation:** Configuring Media Flow-AroundThis feature adds media flow-around capability on the Cisco Unified Border Element by supporting the processing of call setup and teardown requests (VoIP call signaling) and for media streams (flow-through and flow-around). Media flow-

around can be configured the global level or it must be configured on both incoming and outgoing dial peers. If configured only on either the incoming or outgoing dialpeer, the call will become a flow-through call.

Media flow-around is a good choice to improve scalability and performance when network-topology hiding and bearer-level interworking features are not requiredWith the default configuration, the Cisco UBE receives media packets from the inbound call leg, terminates them, and then reoriginates the media stream on an outbound call leg.

Media flow-around enables media packets to be passed directly between the endpoints, without the intervention of the Cisco UBE. The Cisco UBE continues to handle routing and billing functions.To specify media flow-around for voice class, all VoIP calls, or individual dial peers, perform the steps in this section.References:

http://www.cisco.com/en/US/prod/collateral/voicesw/ps6790/gatecont/ps5640/
prod_qas09186a00801da69b.htmlhttp://www.cisco.com/en/US/docs/ios/voice/cube/configuration/g uide/vb-gw-sipsip.html#wp1392896

**QUESTION 65**

Which codec complexity type will offer the greatest number of voice channels, provided that the complexity type is compatible with the particular codecs that are in use?

A. low complexity
B. medium complexity
C. high complexity
D. flex complexity

**Correct Answer:** A
**Section: (none)**
**Explanation**

**Explanation/Reference:**
**Explanation**-a lower codec complexity to support the greatest number of voice channels, provided that the lower complexity is compatible with the particular codecs in use. **Link-**http://www.cisco.com/en/US/docs/ios/12_3t/voice/command/reference/vrht_c6_ps5207_TSD_Prod ucts_Command_Reference_Chapter.html

**QUESTION 66**
When a Cisco Unified Border Element is deployed to support RSVP-based CAC, which method is required?

A. RSVP-based CAC can be supported with either media flow-through or media flow-around if the Cisco Unified Communication manager is configured as an RSVP agent B. RSVP-based CAC only supports media flow-around
B. The Cisco Unified Border Element does not have to participate in the RSVP message exchange and will pass RSVP message through unchanged using media flow-around.
C. RSVP-based CAC requires Cisco Unified Border Element to use media flow-through.

**Correct Answer:** A
**Section: (none)**
**Explanation**

**Explanation/Reference:**
**Explanation: Link**- http://www.cisco.com/en/US/prod/collateral/voicesw/ps6789/ps7046/ps6832/solution_overview_c22-521818.html

**QUESTION 67**
Which voice translation rule will expand extensions that are in the 3000-3999 range to a 10-digit number?

A. /^…\(… .$\)/^1/
B. /3…//4085553/
C. /^3…\(… .$\)//408555\1/
D. /^3…\(… .$\)//408555\
E. /^3…$//408555&/

**Correct Answer:** C
**Section: (none)**
**Explanation**

**Explanation/Reference:**
**Link-**http://www.cisco.com/en/US/products/sw/voicesw/ps4625/products_tech_note09186a0080816cac. shtml

**QUESTION 68**
Which of the following describes SIP Early Offer?

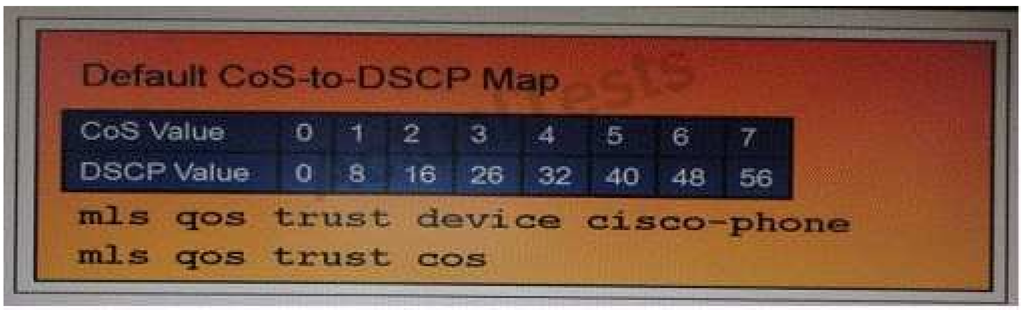
A. In SIP Early Offer mode, the SDP media capabilities are sent in the INVITE message calling device.

B. Early Offer always uses session indicator 183
C. In SIP Early Offer mode, the SDP media capabilities are sent in the 200 OK messages of the calling device.
D. In SIP Early Offer mode, the INVITE and the 200 OK messages use non-SDP message format to indicate SIP Early Offer

**Correct Answer:** A
**Section: (none)**
**Explanation**

**Explanation/Reference:**
**Explanation: Link**-http://www.cisco.com/en/US/docs/ios/voice/cube/configuration/guide/vb-gw- sipsip.html

**QUESTION 69**
## Refer to the exhibit.



Default CoS-to-DSCP Map

| CoS Value | 0 | 1 | 2 | 3 | 4 | 5 | 6 | 7 |
|---|---|---|---|---|---|---|---|---|
| DSCP Value | 0 | 8 | 16 | 26 | 32 | 40 | 48 | 56 |

```
mls qos trust device cisco-phone
mls qos trust cos
```

How does a switch port that receives marked traffic from a Cisco IP phone use the mls qos trust cos command?

A. The CoS setting is modified according to the CoS-to-DSCP map.
B. CoS is used to select the ingress and egress queues.
C. For non-IP packets, the CoS is set to 7 and DSCP-to-CoS mapping is not applied.
D. The DSCP-to-CoS map is applied.

**Correct Answer:** A
**Section: (none)**
**Explanation**

**Explanation/Reference:**
**Explanation**- mls qos trustcosdefines that cos setting is modified according to COS- DSCP map.

**QUESTION 70**
What are the three acceptable values for one-way delay, jitter, and packet loss in a VoIP network? (Choose three)

A. 0-400 ms for delay
B. 1 packet loss
C. 20 ms for jitter
D. 0-150 ms for delay
E. 1 percent packet loss
F. 30 ms for jitter

**Correct Answer:** DEF
**Section: (none)**
**Explanation**

**Explanation/Reference:**
0-150 ms for delay1 percent packet loss30 ms for jitter See TVOICE V I 6-165

**QUESTION 71**
Which statement best describes dial peers in voice gateway?

A. Dial peers are call legs that are used to identity call source and destination endpoints and to define the characteristics that are applied to each call leg in the call connection.
B. Dial peers are configured with call legs that are essential to implementing dial plans and providing voice services over an IP packet network.
C. dial peer is a physical addressable endpoint in a voice gateway.
D. Dial peers create physical connections called call legs to complete an end-to-end call.

**Correct Answer:** A
**Section: (none)**
**Explanation**

**Explanation/Reference:**
**Link-** http://www.cisco.com/en/US/tech/tk652/tk90/technologies_tech_note09186a008010ae1c.shtml

**QUESTION 72**
Which of the following best describe implementation challenges that are associated with variable- length numbering plans?

A. the variable number of extensions that need to be implemented
B. the number of trunks that need to be assigned
C. the mapping between IP addresses and extension numbers
D. the identification of the number of digits that need to be dialed before the call is routed
E. the degree in which the dial plan varies.

**Correct Answer:** E
**Section: (none)**
**Explanation**

**Explanation/Reference:**
**Explanation**- An open numbering plan, as found in countries that have not yet standardized on numbering plans, features variance in the length of the area code or the local number, or both.
**Link**-http://www.ciscopress.com/articles/article.asp?p=1715059

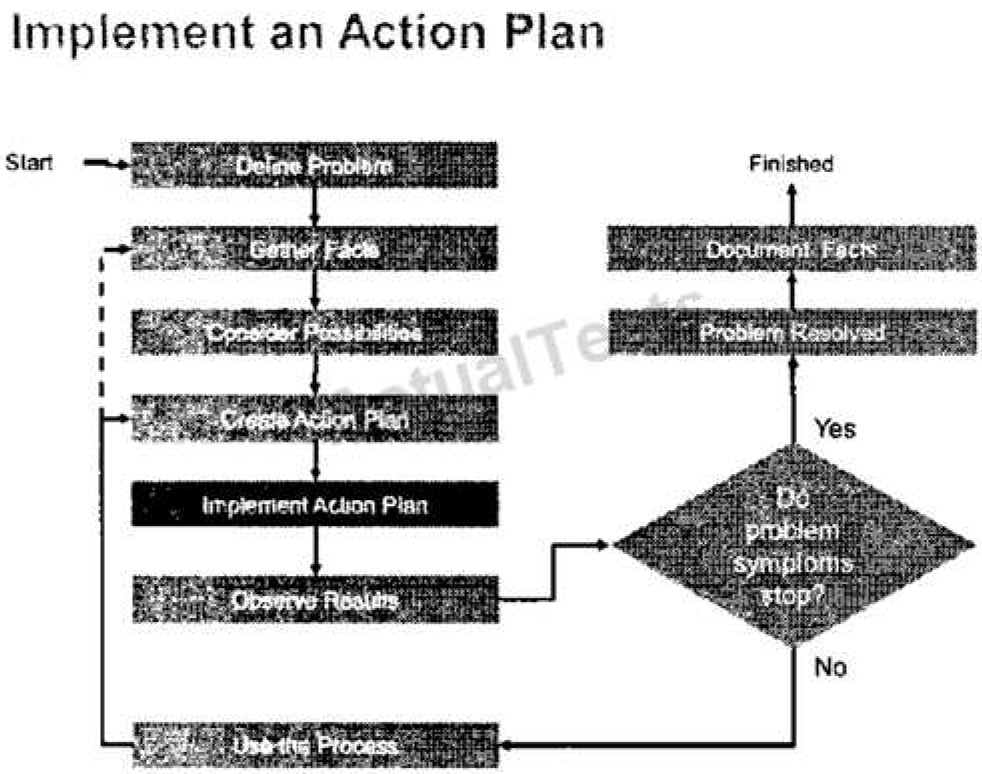
**QUESTION 73**
At what step do you restart the troubleshooting process if after observing the result of your troubleshooting, the problem still exists?

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

**Correct Answer:** A
**Section: (none)**

**Explanation**

**Explanation/Reference:**
The diagram below from the Cisco TVOICE course, which we believe shows the correct answer to be “restart the troubleshooting process after implementing an action plan and observing it has not solved the problem” this also makes more sense than restarting the troubleshooting process after creating an action plan.




Note:
The following steps detail the problem-solving process outlined:

Step 1 When analyzing a network problem, make a clear problem statement. You should define the problem in terms of a set of symptoms and potential causes.
To properly analyze the problem, identify the general symptoms and then ascertain what kinds of problems (causes) could result in these symptoms. For example, hosts might not be responding to service requests from clients (a symptom). Possible causes might include a misconfigured host, bad interface cards, or missing router configuration commands.

Step 2 Gather the facts that you need to help isolate possible causes.
Ask questions of affected users, network administrators, managers, and other key people. Collectinformation from sources such as network management systems, protocol analyzer traces, output from router diagnosticcommands, or software release notes.

Step 3 Consider possible problems based on the facts that you gathered. Using the facts, you can eliminate some of the potential problems from your list. Depending on the data, for example, you might be able to eliminate hardware as a problem so that you can focus on software problems. At every opportunity, try to narrow the number of potential problems so that you can create an efficient plan of action.

Step 4 Create an action plan based on the remaining potential problems. Begin with the most likely problem,

and devise a plan in which only one variable is manipulated.
Changing only one variable at a time enables you to reproduce a given solution to a specific problem. If you alter more than one variable simultaneously, you might solve the problem, but identifying the specific change that eliminated the symptom becomes far more difficult and will not help you solve the same problem if it occurs in the future.

Step 5 Implement the action plan, performing each step carefully while testing to see whether the symptom disappears.

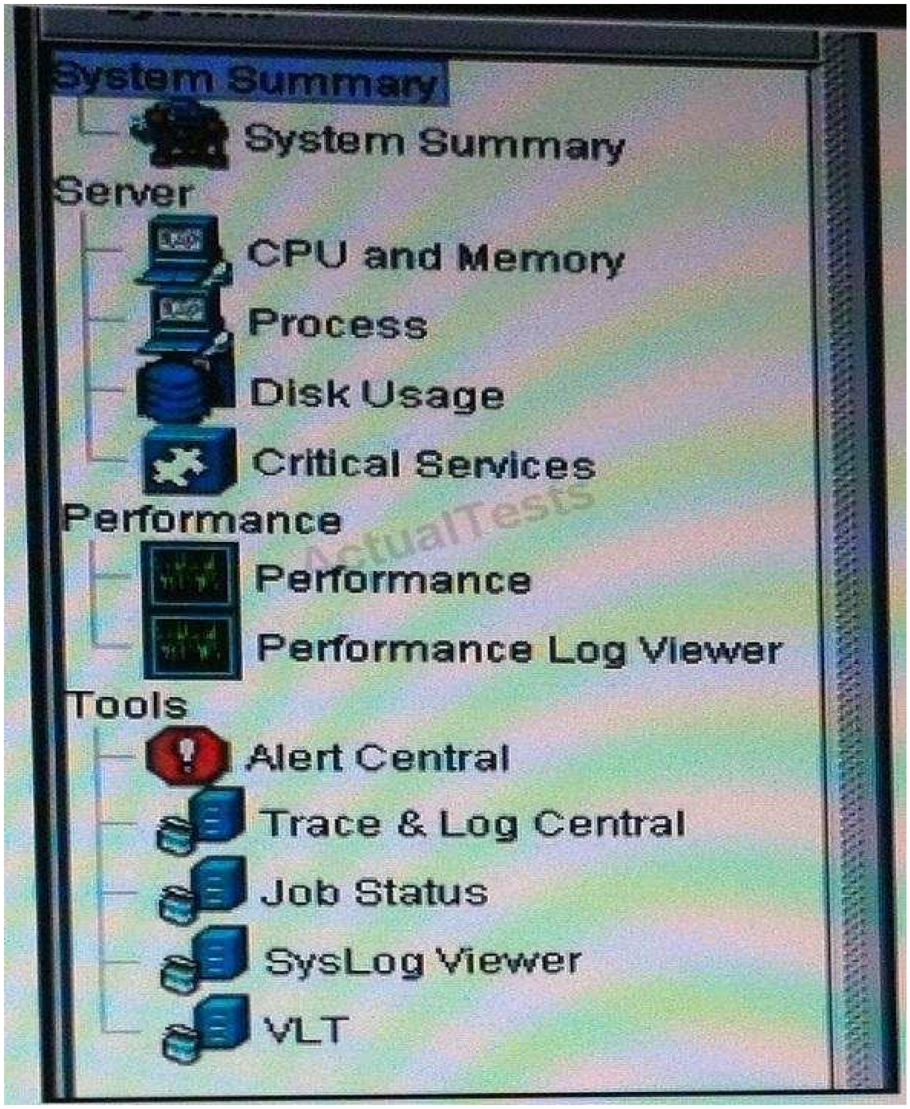
Step 6 Whenever you change a variable, be sure to gather results. Generally, you should use the same method of gathering facts that you used in Step 2 (that is, working with the key people affected, in conjunction with utilizing your diagnostic tools).

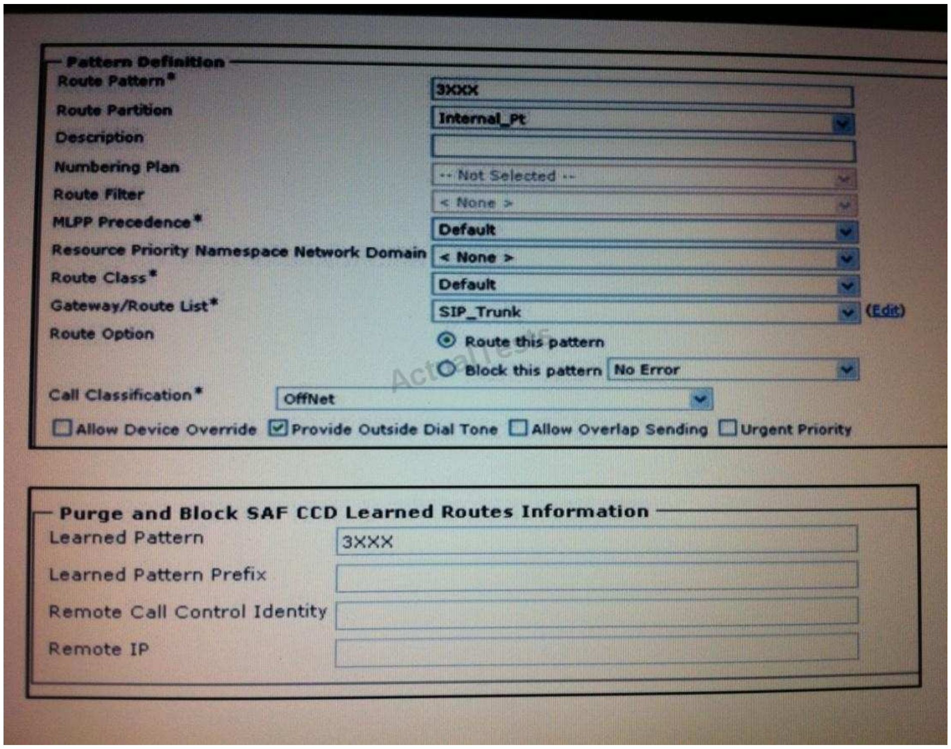
Step 7 Analyze the results to determine whether the problem has been resolved. If it has, then the process is complete.

Step 8 If the problem has not been resolved, you must create an action plan based on the next mostlikely problem in your list. Return to Step 4, change one variable at a time, and repeat the process untilthe problem is solved. http://www.cisco.com/en/US/docs/internetworking/troubleshooting/guide/tr1901.html#wp1020562

**QUESTION 74**

**Refer to Exhibit.**


System Summary
System Summary
Server
CPU and Memory
Process
Disk Usage
Critical Services
Performance
Performance
Performance Log Viewer
Tools
Alert Central
Trace & Log Central
Job Status
SysLog Viewer
VLT



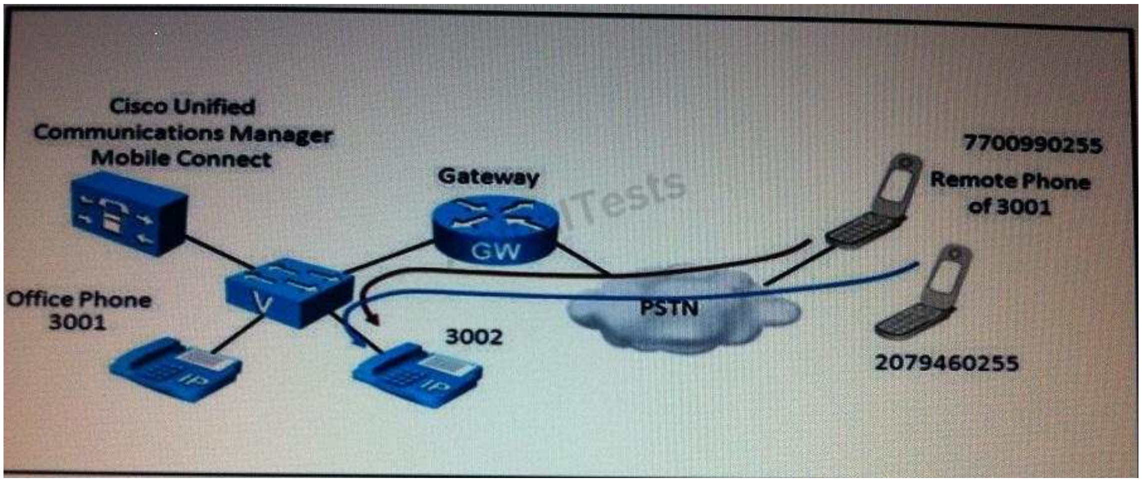

Which Cisco Unified Communications Manager Real-Time Monitoring Tool component can be used to view DHCP requests and responses from a Cisco Unified Communications Manager DHCP server?

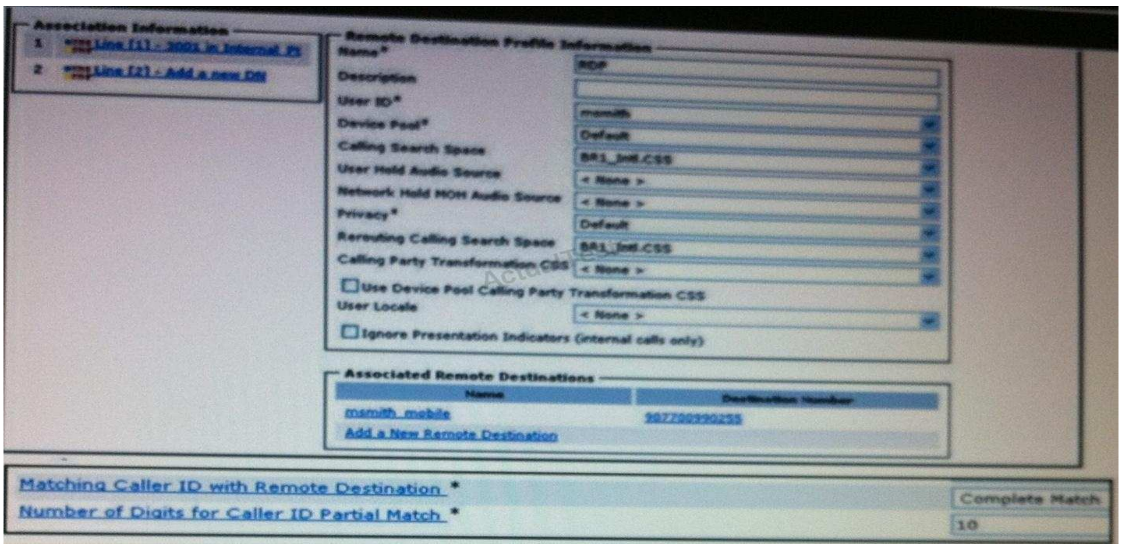
A. Performance Log Viewer
B. Processor
C. System Summary
D. Job Status
E. SysLog Viewer
F. VLT

**Correct Answer:** E
**Section: (none)**
**Explanation**

**Explanation/Reference:**

**QUESTION 75**
**Refer to Exhibits.**



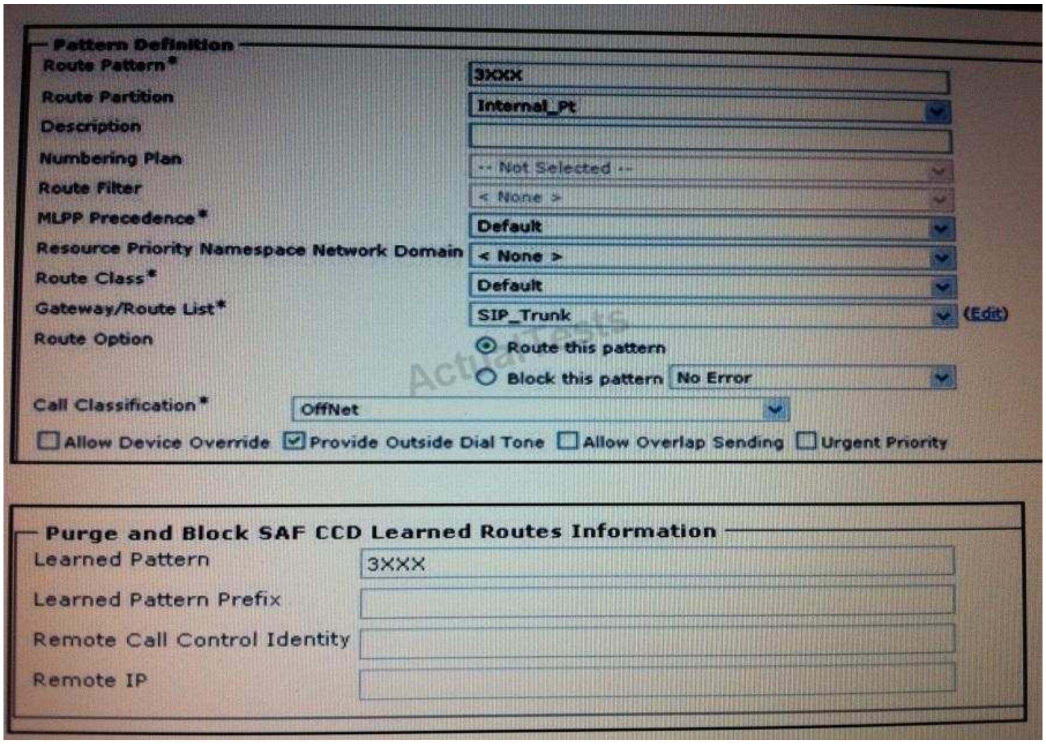

When the remote destination of IP phone 3001 places an inbound PSTN call to extension 3002, the caller ID that is displayed on extension 3002 is 7700990255 of 3001. Which course of action can resolve this issue?

A. Change the remote destination number to 7700l in the configuration.
B. Change the service parameter matching Caller ID with remote Destination to Partial Match.
C. Change the service parameter Number of Digits for caller ID partial Match to 12.
D. Change the Privacy Setting from Default to Off.

**Correct Answer:** B
**Section: (none)**
**Explanation**

**Explanation/Reference:**

**QUESTION 76**
**Refer to Exhibit.**




Assume that all learned SAF routes are placed in the SAF_Pt partition. The 3XXXdirectory number pattern is being advertised by a remote cluster and is also being blocked by the local cluster that is shown in the exhibit. An IP Phone is attached to the local cluster and is configured with a CSS that contains the following partitions: SAF_Pt and internal_Pt in this order. When the IP phone places a call to 3001, what will occur?

A. The call will succeed and will be placed via the SIP_Trunk.
B. The call will fail because it will be blocked by the CCO blocked Learned Route configuration.
C. The Call will be placed in a round-robin fashion between the SAF network and SIP_trunk.
D. The Call will be placed in a round -robin fashion between the SAF network and SIP network and SIP_Trunk Every other call will fail.

**Correct Answer:** A
**Section: (none)**
**Explanation**

**Explanation/Reference:**

**QUESTION 77**
**Refer to Exhibit.**

```
hostname HQ
!
card type e1 0 0
!
network-clock-participate wic 0
!
isdn switch-type primary-net5
!
voice-card 0
!
sccp local GigabitEthernet0/0.110
sccp ccm 10.1.5.10 identifier 1 version 7.0
sccp
!
sccp ccm group 1
 associate ccm 1 priority 1
 associate profile 1 register HQ_Conf
!
dspfarm profile 1 conference
 maximum conference-participants 0
 shutdown
```

```
HQ(config-dspfarm-profile)# maximum sessions 5
                            ^
% Invalid input detected at '^' marker.
```

```
HQ(config-dspfarm-profile)#maximum sessions ?
 <0-0>  Number of sessions assigned to this profile
```

When the user tried to configure the command maximum session 5 under the dspfarm profile 1, the error shown in the exhibit was reported. Which course of action will resolve this issue?

A. The maximum conference-participants value must be configured > 0
B. Ensure that the conference bridge is not registered in Cisco Unified Communications Manager.
C. The Command dsp service dspfarm must be configured under the voice-card configuration.
D. Configure the dspfarm all command under the voice card.

**Correct Answer:** C
**Section: (none)**
**Explanation**

**Explanation/Reference:**

**QUESTION 78**
## Refer to the Exhibit

```
HQ-1#debug ip rsvp resv
Apr 15 09:22:25.256: RSVP: session 10.1.250.101_16858[0.0.0.0]: Received Resv
message from 127.0.0.1 (on receiver host)
Apr 15 09:22:25.256: RSVP: 10.1.250.102_16546->10.1.250.101_16858[0.0.0.0]:
Successfully parsed Resv message from 127.0.0.1 (on receiver host)
Apr 15 09:22:25.256: RSVP-MSG: 10.1.250.102_16546-
>10.1.250.101_16858[0.0.0.0]: no matching path state for Resv
Apr 15 09:22:25.264: RSVP: session 10.1.250.101_16858[0.0.0.0]: Received Resv
message from 127.0.0.1 (on receiver host)
Apr 15 09:22:25.268: RSVP: 10.1.250.102_16546->10.1.250.101_16858[0.0.0.0]:
Successfully parsed Resv message from 127.0.0.1 (on receiver host)
Apr 15 09:22:25.268: RSVP-RESV: Admitting new reservation: 466CE358
Apr 15 09:22:25.268: RSVP-RESV: Locally created reservation. No
admission/traffic control needed
Apr 15 09:22:25.268: RSVP: 10.1.250.102_16546->10.1.250.101_16858[0.0.0.0]:
start requesting 96 kbps FF reservation on Serial0/1/0.121, neighbor
10.1.6.102
Apr 15 09:22:25.268: RSVP: 10.1.250.102_16546->10.1.250.101_16858[0.0.0.0]:
start requesting 96 kbps FF reservation on Serial0/1/0.121, neighbor
10.1.6.102
Apr 15 09:22:25.272: RSVP: session 10.1.250.102_16546[0.0.0.0]: Received Resv
message from 10.1.6.102 (on Serial0/1/0.121)
Apr 15 09:22:25.272: RSVP: 10.1.250.101_16858->10.1.250.102_16546[0.0.0.0]:
Successfully parsed Resv message from 10.1.6.102 (on Serial0/1/0.121)
Apr 15 09:22:25.272: RSVP: 10.1.250.101_16858->10.1.250.102_16546[0.0.0.0]:
reservation not found--new one
Apr 15 09:22:25.272: RSVP-RESV: Admitting new reservation: 466CE1F0
Apr 15 09:22:25.276: RSVP: 10.1.250.101_16858->10.1.250.102_16546[0.0.0.0]:
building error_spec object with err-node addr: 10.1.6.101
Apr 15 09:22:25.276: RSVP: 10.1.250.101_16858->10.1.250.102_16546[0.0.0.0]:
Sending ResvError message to 10.1.6.102
Apr 15 09:22:25.276: RSVP-RESV: reservation was not installed: 466CE1F0
Apr 15 09:22:25.280: RSVP-RESV: Deleting reservation: 466CE1F0
Apr 15 09:22:25.280: RSVP: 10.1.250.101_16858->10.1.250.102_16546[0.0.0.0]:
Expiring Serial0/1/0.121 RESV state, reason: Traffic control error
Apr 15 09:22:25.280: RSVP: 10.1.250.101_16858->10.1.250.102_16546[0.0.0.0]:
Expiring Serial0/1/0.121 RESV state, reason: Traffic control error (17:16858)
Apr 15 09:22:25.280: RSVP: 10.1.250.101_16858->10.1.250.102_16546[0.0.0.0]:
```

```
10.1.6.102           reservation on Serial0/1/0.121, neighbor
Apr 15 09:22:25.272: RSVP: session 10.1.250.102_16546[0.0.0.0]: Received Resv
message from 10.1.6.102 (on Serial0/1/0.121)
Apr 15 09:22:25.272: RSVP: 10.1.250.101_16858->10.1.250.102_16546[0.0.0.0]:
Successfully parsed Resv message from 10.1.6.102 (on Serial0/1/0.121)
Apr 15 09:22:25.272: RSVP: 10.1.250.101_16858->10.1.250.102_16546[0.0.0.0]:
reservation not found--new one
Apr 15 09:22:25.272: RSVP-RESV: Admitting new reservation: 466CE1F0
Apr 15 09:22:25.276: RSVP: 10.1.250.101_16858->10.1.250.102_16546[0.0.0.0]:
building error_spec object with err-node addr: 10.1.6.101
Apr 15 09:22:25.276: RSVP: 10.1.250.101_16858->10.1.250.102_16546[0.0.0.0]:
Sending ResvError message to 10.1.6.102
Apr 15 09:22:25.276: RSVP-RESV: reservation was not installed: 466CE1F0
Apr 15 09:22:25.280: RSVP-RESV: Deleting reservation: 466CE1F0
Apr 15 09:22:25.280: RSVP: 10.1.250.101_16858->10.1.250.102_16546[0.0.0.0]:
Expiring Serial0/1/0.121 RESV state, reason: Traffic control error
Apr 15 09:22:25.280: RSVP: 10.1.250.101_16858->10.1.250.102_16546[0.0.0.0]:
Expiring Serial0/1/0.121 RESV state, reason: Traffic control error (17:16858)
Apr 15 09:22:25.280: RSVP: 10.1.250.101_16858->10.1.250.102_16546[0.0.0.0]:
Sending PathTear message to 10.1.250.102
Apr 15 09:22:25.284: RSVP: 10.1.250.102_16546->10.1.250.101_16858[0.0.0.0]:
Refresh RESV, req=466D33A0 [cleanup timer is not awake]
```

The Exhibit shows the output of an unsuccessful RSVP call setup that uses G.711 codec. Which configuration on serial 0/1/0.121 would resolve that issue?

A. ip rsvp bandwidth 40 40
B. ip rsvp bandwidth 40 24
C. ip rsvp bandwidth 96 80
D. ip rsvp bandwidth 96 96
E. ip rsvp bandwidth 80 16

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

**Explanation/Reference:**

**QUESTION 79**
You have been presented with a trouble ticket from an end user who works at a remote location that is served by a Cisco Unified Communications Manager Express. The user reports being unable to place calls to international numbers, but all other calls work properly and other users at this location can place international calls. Which two troubleshooting techniques would be helpful in resolving this issue? (Choose Two)

A. Cisco IOS debug tools
B. Class of Restriction baseline configuration for the user on Cisco Unified CommunicationsManager Express
C. show output of the ephone and ephone -dn configurations.
D. show output of the voice translation rules in the voice gateway
E. show output for the T1 controller and voice port configuration in the voice gateway

**Correct Answer:** AB
**Section: (none)**
**Explanation**

**Explanation/Reference:**

**QUESTION 80**
Which statement indicates something that can cause an inbound PSTN call to an H.323 gateway that is configured in Cisco Unified Communications Manager to fail to ring an IP Phone?

A. The gateway is not registered in Cisco Unified Communications Manager
B. The gateway IP address that is configured in Cisco Unified Communications Manager does not match the IP address that is configured at the gateway in the h323-gataway voip bind srcaddr command.
C. The Cisco Unified Communications Manager does not have a matching route pattern to match the called number.
D. The gateway is missing the command allow-connections h323 to h323 under the voice service voip configuration

**Correct Answer:** B
**Section: (none)**
**Explanation**

**Explanation/Reference:**

**QUESTION 81**
**Refer to the exhibit.**

```
interface GigabitEthernet1/0/4
description HQ Phone 1
switchport access vlan 10
switchport voice lan 110
spanning-tree portfast
```

An IP phone that is connected through a Cisco Catalyst 3750 Series Switch is failing to register with Cisco Unified Communications Manager. When the user presses the settings button on the phone, the Operational VLAN ID shows a blank entry. What is the most likely cause for this issue?

A. The switch may not be supplying inline power.
B. The spanning tree portfast command needs to be removed.
C. The trunk encapsulation is missing. The trunk must be configured for dot1.Q
D. Cisco Discovery Protocol is disabled on the switch
E. The Operational VLAN ID of the phone always shows as blank. The Admin VLAN ID should be 110

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

**Explanation/Reference:**

**QUESTION 82**
If after observing the results of your action plan, the problem still remains, at what point should you restart the troubleshooting process?

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

**Correct Answer:** C
**Section: (none)**
**Explanation**

**Explanation/Reference:**

**QUESTION 83**
After a successful login using Cisco Extension Mobility, an IP phone performs a restart followed by a reset. What can cause this issue?

A. The phone model to which the user logged in is a different model than the model that is configured in the user device profile.

B. The locale that is configured on the phone is different than the locale that is configured in the user device profile.
C. The security profile that is specified in the user device profile does not match the security profile for the phone where the user logged in.
D. The user device profile and the phone that is used for the Cisco Extension Mobility log in do not use the same phone protocol.

**Correct Answer:** B
**Section: (none)**
**Explanation**

**Explanation/Reference:**

**QUESTION 84**
Your Cisco Unified CallManager 5.0 cluster is using your corporate Windows 2003 Active Directory for user information. Over the weekend you updated the Windows 2003 Active Directory Server and added a small group of new users. Cisco Unified CallManager is configured to synchronize with the Active Directory server every 8 hours and it has been 32 hours since the last successful synchronization. The configuration on Cisco Unified CallManager did not change during the Active Directory server upgrade and the remainder of the Windows network is functioning properly.

What are two possible causes of this synchronization issue? (Choose two.)

A. The synchronization on the AD server was set to manual
B. The domain controllers are down.
C. There is a username and or password mismatch between the Cisco Unified CallManager cluster and the Windows AD server.
D. Authorization has not been configured for a third party LDAP service
E. The LMHOSTS file has been corrupted.

**Correct Answer:** BC
**Section: (none)**
**Explanation**

**Explanation/Reference:**

**QUESTION 85**
Refer to the exhibit. You have received a trouble ticket stating that users cannot place calls to the PSTN. During testing you discover the gateway is not switching to the secondary call agent when the primary call agent is unreachable.

What needs to be done to allow the MGCP gateway to use a different call agent if the primary fails?

```
!
mgcp
mgcp call-agent 10.1.44.4 2427 service-type mgcp version 0.1
mgcp dtmf-relay voip codec all mode out-of-band
mgcp rtp unreachable timeout 1000 action notify
mgcp modem passthrough voice mode cisco
mgcp sdp simple mgcp package-capability rtp-package
mgcp package-capability sst-packateno mgcp timer receive-rtcp
no mgcp explicit hookstate
!
ccm-manager mgcp
!
```

A. The ccm-manager fallback-mgcp command needs to be added to the gateway.
B. The ccm-manager redundant-host command needs to be added to the gateway.
C. A Cisco Unified CallManager group that includes the secondary call agent needs to be assigned to the gateway.
D. The gateway needs to be defined as a non-gatekeeper-controlled intercluster trunk with the secondary Cisco Unified CallManager defined.

**Correct Answer:** B
**Section: (none)**
**Explanation**

**Explanation/Reference:**

**QUESTION 86**
You have received a trouble ticket stating that when callers dial the internal Help Desk queue at extension 2300, they hear a message that their calls cannot be completed as dialed. Which two issues could cause this problem? (Choose two.)

A. There are no agents logged in to the Help Desk queue.
B. The script associated with the Help Desk queue is corrupt.
C. There is a connectivity issue between Cisco Unified CallManager and the Cisco Unified Contact Center Express server.
D. The route point for 2300 has been modified or deleted in Cisco Unified CallManager, resulting in a synchronization issue.
E. The CSS of the route point for 2300 is incorrect.

**Correct Answer:** CD
**Section: (none)**
**Explanation**

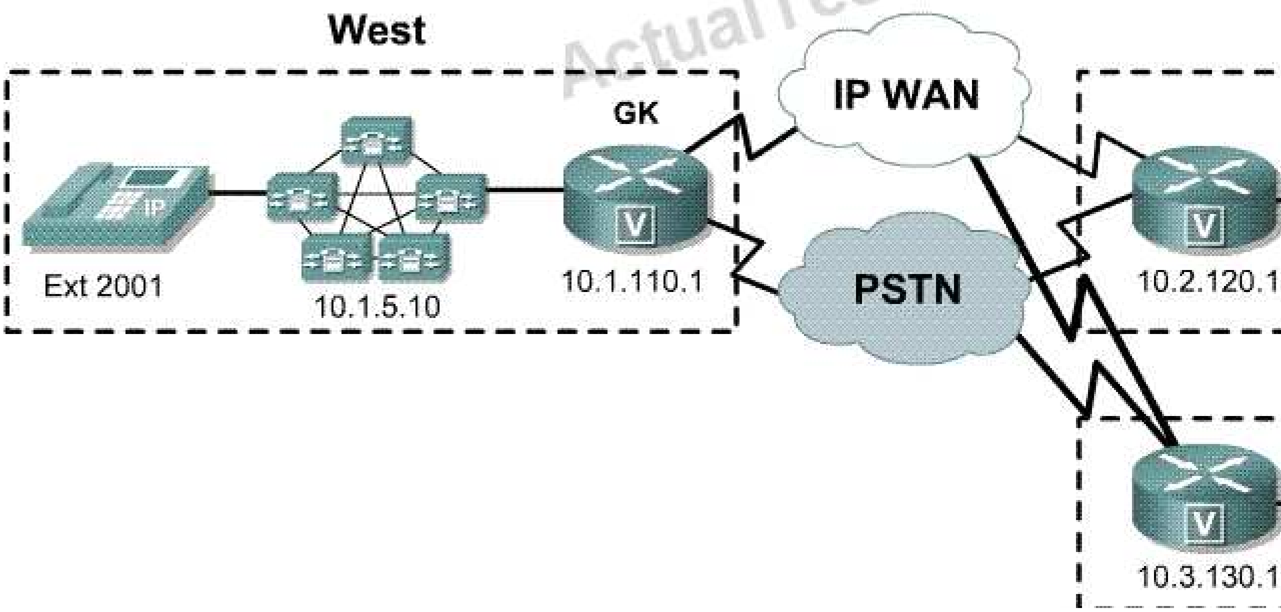
**Explanation/Reference:**

**QUESTION 87**

1

Refer to the exhibit. Click on the GK icon to see the output of the **show gatekeeper endpoint** comm
CallManager cluster to view the trunk configuration screens. Click the X in each popup to return to th
A gatekeeper has been configured on the 10.1.110.1 router to support three local zones, West, East
you do a **show gatekeeper endpoints** command the West zone device is missing. What needs to b
configuration in the CallManager for it to register with the gatekeeper in zone West?

- Restart the CallManager so it can register with the gatekeeper
- Change the Device Name to be the IP address of the gatekeeper in the CallManager Trunk Conf
- Set the Terminal Type to terminal in the Gatekeeper Information field of the Trunk Configuration
- Configure a zone name on the CallManager trunk




**Point and Shoot:**

1

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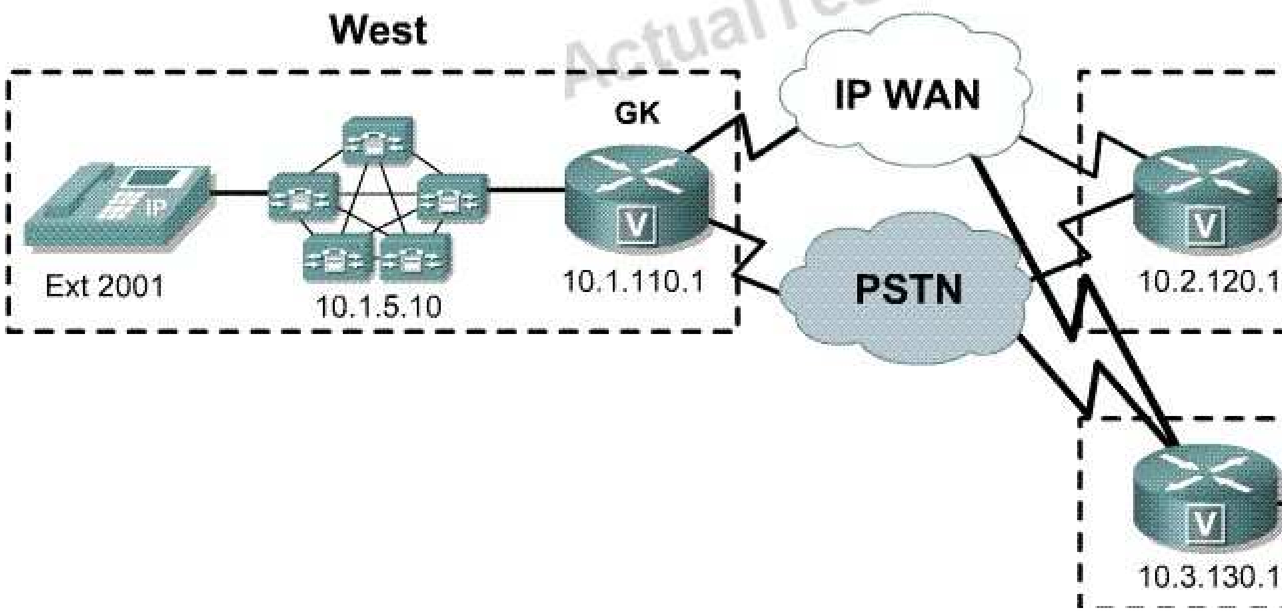
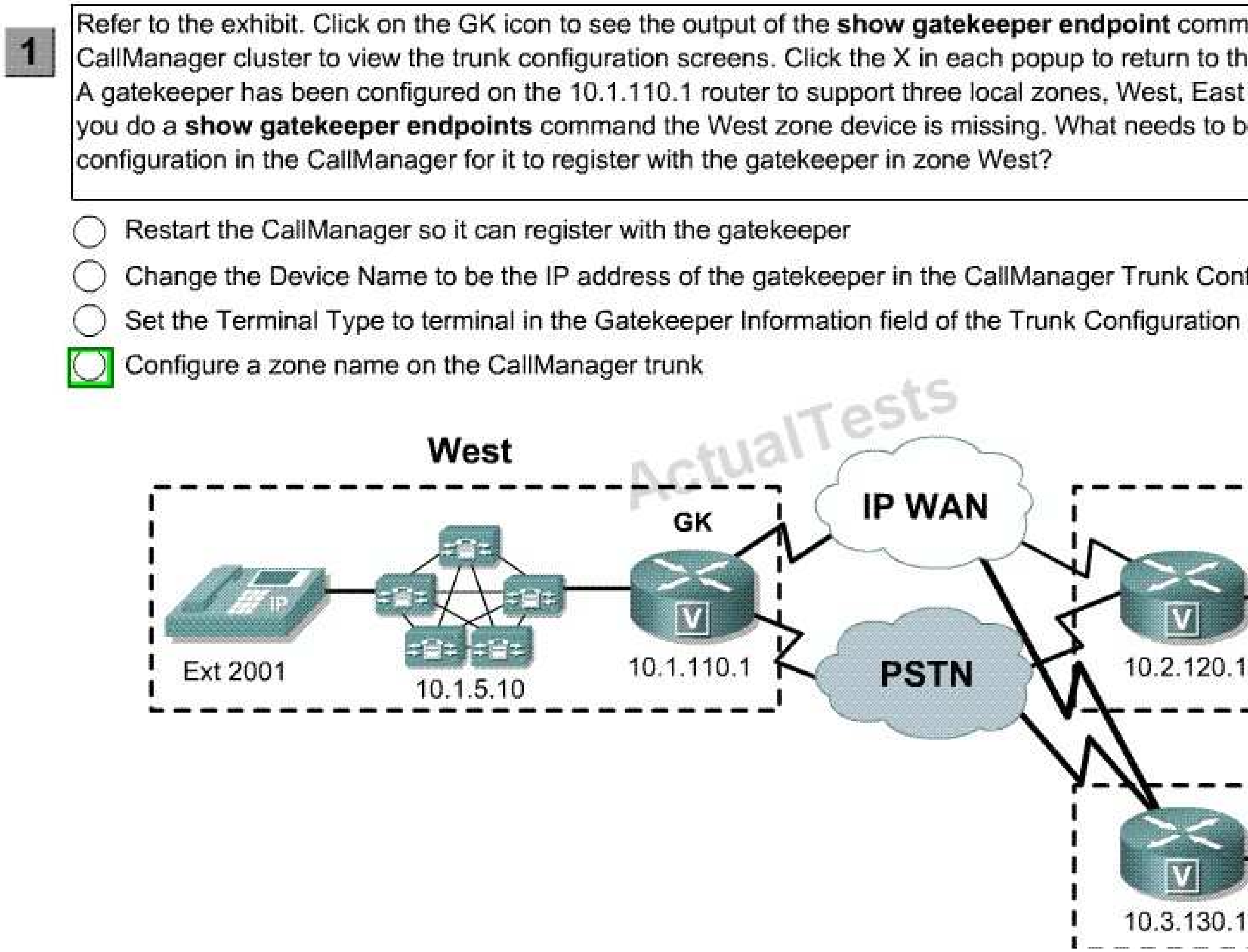



**Correct Answer:**




**Section: (none)**
**Explanation**

**Explanation/Reference:**

**QUESTION 88**
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.

**Correct Answer:** D

**Section: (none)**
**Explanation**

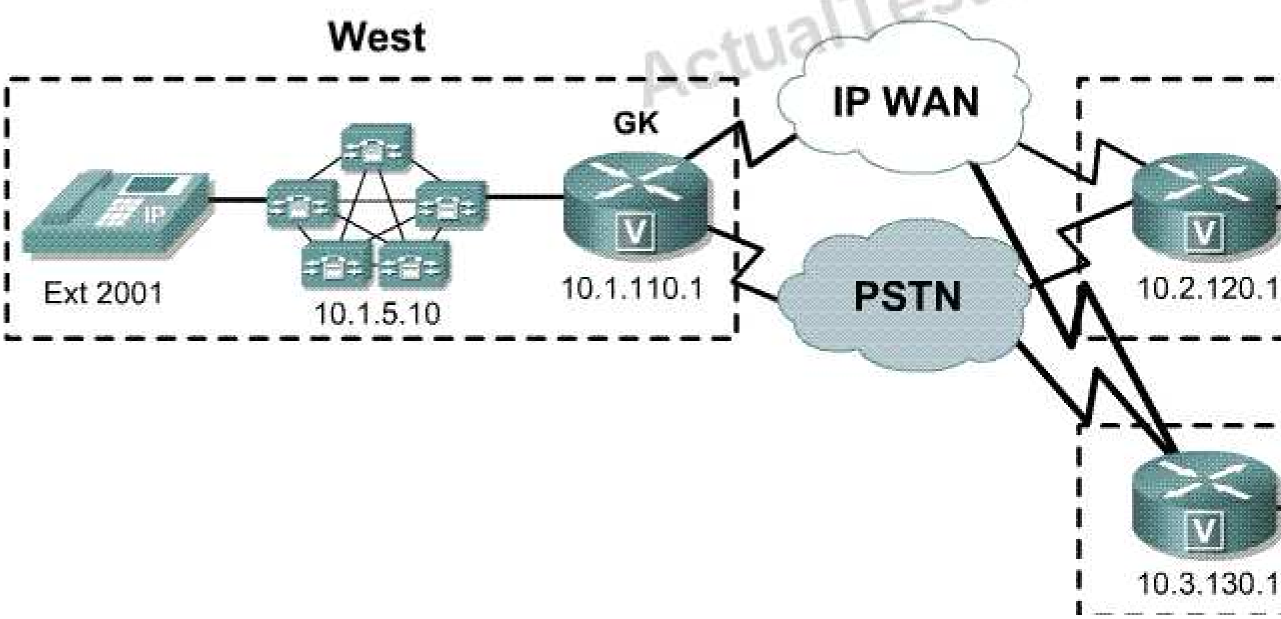
**Explanation/Reference:**

**QUESTION 89**

1

Refer to the exhibit. Click on the GK icon to see the output of the **show gatekeeper endpoint** comm
the voice gateways to see the configuration of the interfaces. Click the X in each popup to return to th
You have configured a gatekeeper with three local zones named East, West and EMEA. CallManag
the East and EMEA zones are all registered in the West zone but they should each be registered in t
needs to be done to resolve this issue?

- The gatekeeper needs to be stopped and restarted for the changes to take effect
- The IP address in the gateway command **h323-gateway voip ID** needs to be changed to that of
- The zone name in the gateway command **h323-gateway voip ID** needs to be changed to the co
- The gatekeeper can only support one zone and all registering endpoints are being placed correc




**Point and Shoot:**

**1**

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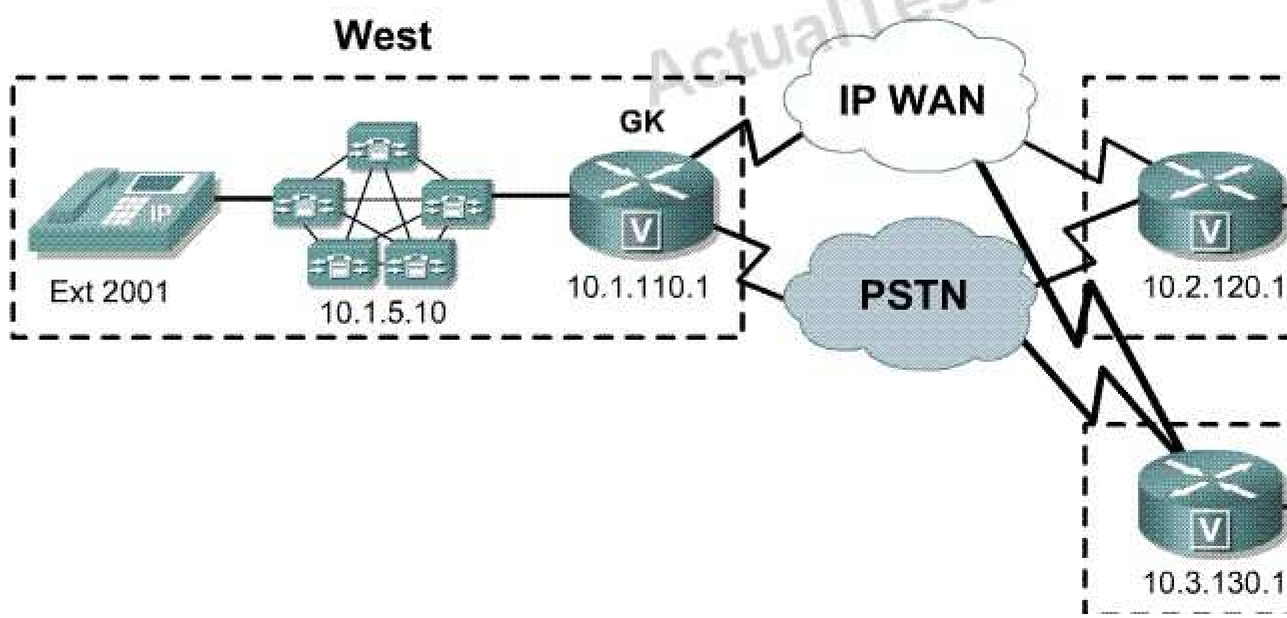
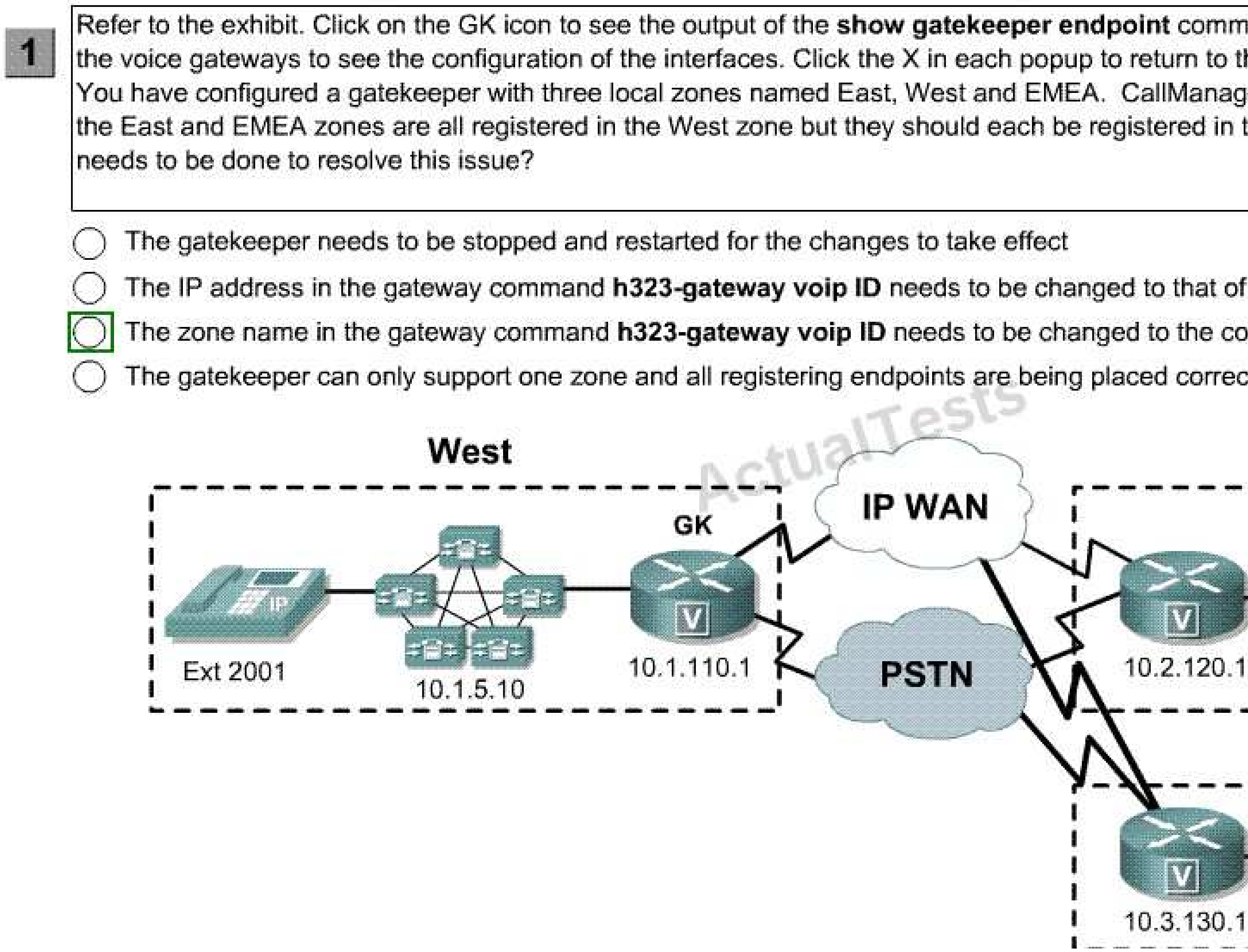



**Correct Answer:**




**Section: (none)**
**Explanation**

**Explanation/Reference:**

**QUESTION 90**
Which type of echo is found mostly on tail circuits and is due to reflection that causes the Tx signal to appear on the Rx signal?

A. Hybrid echo
B. Talker echo
C. Listener echo
D. Tail-end echo

**Correct Answer:** A
**Section: (none)**
**Explanation**

**Explanation/Reference:**

IntroductionThis document describes how to troubleshoot and eliminate echo where possible inIP Telephony networks with Cisco IOS® gateways.There are two sources of echo:* Hybrid echo* Acoustic echoHybrid echo is caused by an impedance mismatch in the hybrid circuit, such as atwo-wire to four-wire interface. This mismatch causes the Tx signal to appear onthe Rx signal.Acoustic echo is caused by poor acoustic isolation between the earpiece and themicrophone in handsets and hands-free devices

**QUESTION 91**
Your Cisco Unified CallManager 5.0 cluster has just started to use a third-party LDAP service. Users complain that they are unable to make changes to their passwords in their Cisco Unified CallManager user web pages. How should you resolve this problem?

A. Restart the phones that are having problems to reinitialize the LDAP database
B. Have the users make changes to their passwords in the LDAP database.
C. Configure automatic synchronization of the LDAP database.
D. Change the passwords on the IP phone screen using the TUI

**Correct Answer:** B
**Section: (none)**
**Explanation**

**Explanation/Reference:**

**QUESTION 92**
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.

**Correct Answer:** BD
**Section: (none)**
**Explanation**

**Explanation/Reference:**



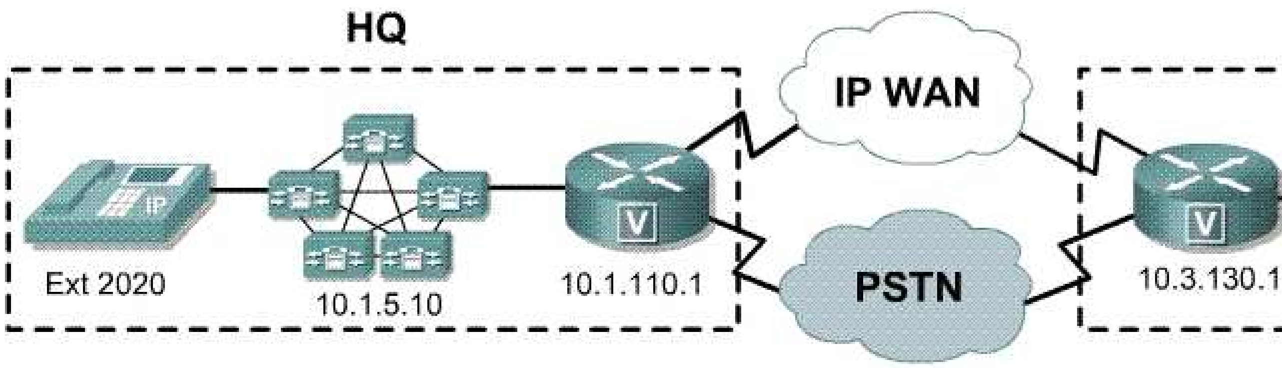
**QUESTION 93**

**1**

Refer to the exhibit. Click the Voice Gateway for the BR2 location to view the output from the **debug**
command and click on 10.1.5.10 to view and search the trace file output. Output can be searched by
the Search box and clicking the Find button. Click the X to return to the item.
A trunk has been configured between the Cisco Unified CallManager cluster at 10.1.5.10 and a CME
testing you find that calls are completed when dialing from ext. 2020 to ext. 4001, but calls from ext.
receive a fast busy. What is the issue that is preventing calls from ext. 4001 to ext. 2020 from being

- The **h323-gateway voip bind srcaddr 10.3.130.1** command has been omitted from the BR2 co
- A translation rule has been applied that is keeping the call from being completed
- A CSS has been omitted from the trunk configured to BR2
- An incorrect CSS has been applied to the gateway at HQ
- The trunk IP address in the Cisco Unified CallManager Information is field is incorrect






**Point and Shoot:**

**1**

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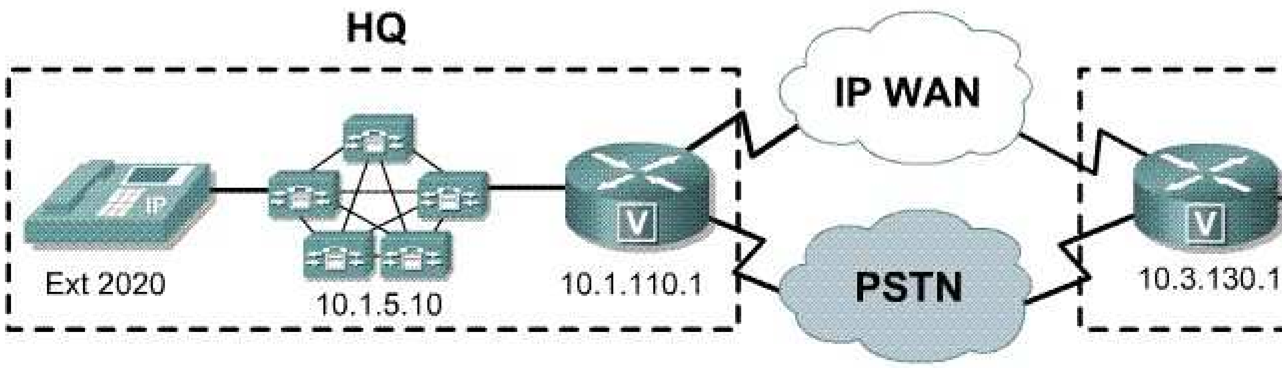


**Correct Answer:**

1

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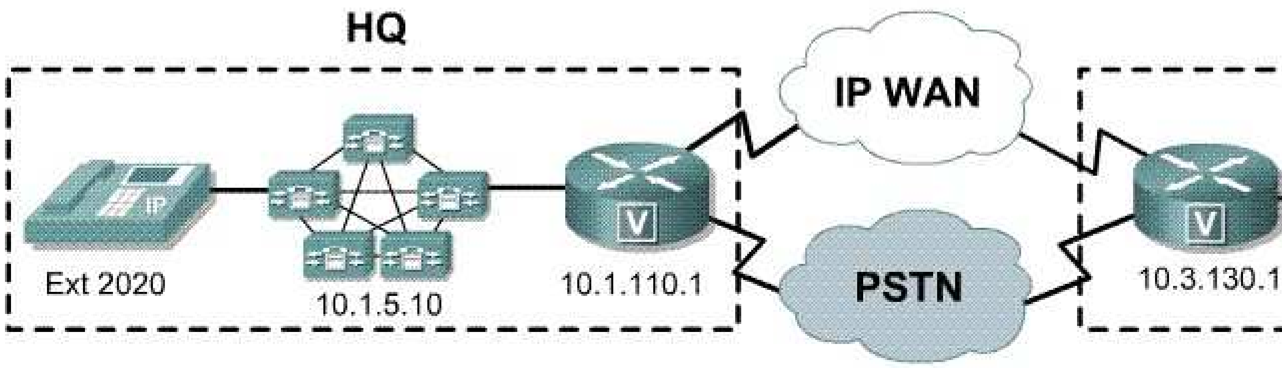
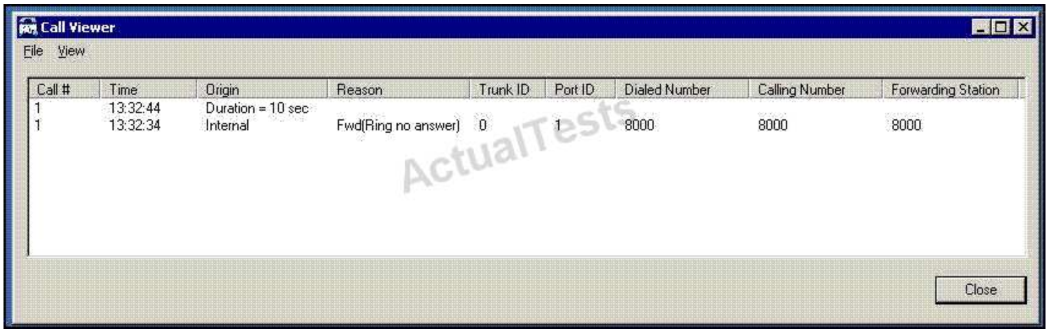



**Section: (none)**
**Explanation**

**Explanation/Reference:**

**QUESTION 94**
Refer to the exhibit. A site is using four-digit extensions for internal calling. The voice-mail pilot number is 8000. Calls to extension 2001 hear the Cisco Unity opening greeting instead of the subscriber's greeting when forwarded to Cisco Unity.

What is the probable cause?




A. The mailbox is configured with the E.164 number instead of the extension.

B. A call routing rule has been added that is preempting the Attempt Forward rule

C. Extension 2001 has not been defined in Cisco Unity

D. A greeting has not been recorded for mailbox 2001.

E. The Voice Mail Box Mask setting in Cisco Unified CallManager is set to 8000 instead of XXXX

**Correct Answer:** E
**Section: (none)**
**Explanation**

**Explanation/Reference:**

**QUESTION 95**
What happens if CDP is not enabled on a switch port to which an IP phone is connected?

A. The phone is unable to acquire an IP address.

B. The phone cannot get its VLAN ID assignments.

C. The phone cannot learn the address of the TFTP server.

D. The switch will put the port into the errDisable state until CDP is enabled.

**Correct Answer:** B
**Section: (none)**
**Explanation**

**Explanation/Reference:**

**QUESTION 96**
Refer to the exhibit. You have received a trouble ticket that users cannot place calls to the PSTN. During testing you discover the gateway is not switching to the secondary call agent when the primary call agent is unreachable.

What needs to be done to allow the MGCP gateway to use a different call agent if the primary fails?

```
!
ccm-manager mgcp
ccm-manager music-on-hold
ccm-manager config server 10.1.5.2
ccm-manager config
!
mgcp
mgcp call-agent 10.1.5.2 2427 service-type mgcp version 0.1
mgcp dtmf-relay voip codec all mode out-of-band
mgcp rtp unreachable timeout 1000 action notify
mgcp modem passthrough voip mode nse
mgcp package-capability rtp-package
mgcp package-capability sst-package
no mgcp package-capability fxr-package
mgcp package-capability pre-package
no mgcp timer receive-rtcp
mgcp sdp simple
mgcp rtp payload-type g726r16 static
!
mgcp profile default
!
```

A. The ccm-manager fallback-mgcp command needs to be added to the gateway.
B. The ccm-manager redundant-host command needs to be added to the gateway.
C. A Cisco Unified CallManager group that includes the secondary call agent needs to be assigned to the gateway.
D. The gateway needs to be defined as a non-gatekeeper-controlled intercluster trunk with the secondary Cisco Unified CallManager defined.

**Correct Answer:** B
**Section: (none)**
**Explanation**

**Explanation/Reference:**

**QUESTION 97**
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.

**Correct Answer:** C
**Section: (none)**
**Explanation**

**Explanation/Reference:**
**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 choosing**Settings > Model Information**and viewing the LSC setting. The LSC setting displays**Installed**or**Not Installed**, depending on the circumstances. Link:http://www.cisco.com/en/US/docs/voice_ip_comm/cucm/trouble/6_0_1/
tbsystem.html#wpmkr
1159616

**QUESTION 98**
Refer to the exhibit. Voice bearer traffic is mapped to which queue in FastEthernet0/2?

```
mls qos map cos-dscp 0 8 12 16 28 32 40 46
mls qos
!
spanning-tree mode pvst
spanning-tree extend system-id
!
interface FastEthernet0/1
switchport trunk encapsulation dot1q
switchport mode trunk
no ip address
wrr-queue cos-map 1 6 7
wrr-queue cos-map 2 5
wrr-queue cos-map 3 2 3 4
wrr-queue cos-map 4 0 1
spanning-tree portfast
!
interface FastEthernet0/2
switchport access vlan 20
switchport trunk encapsulation dot1q
switchport mode dynamic desirable
switchport voice vlan 120
no ip address
mls qos trust device cisco-phone
mls qos trust cos
wrr-queue cos-map 1 6 7
wrr-queue cos-map 2 5
wrr-queue cos-map 3 2 3 4
wrr-queue cos-map 4 0 1
spanning-tree portfast
!
```



A. Queue 1
B. Queue 2
C. Queue 3
D. Queue 4

**Correct Answer:** B
**Section: (none)**

**Explanation**

**Explanation/Reference:**
To map CoS values to drop thresholds for a queue, use thewrr-queue cos-mapcommand. wrr-queue cos-map**queue-id**[**threshold-id**][cos-1 ... cos-n
queue-id :Queue number; the valid values are from 1 to 2. threshold-idThreshold ID; valid values are from 1 to 2.
cos-1 ... cos-nCoS value; valid values are from 0 to 7

According with this command, le correct answer is Queue 2 http://www.cisco.com/en/US/docs/switches/lan/catalyst6500/ios/12.1E/native/command/reference/ W1.pdf

**QUESTION 99**
Which log file contains call-processing information from services such as Cisco Unified CallManager and Cisco Unified CallManager CTI Manager?

A. CTI trace
B. SDI trace
C. CDROnDemand
D. SDL trace

**Correct Answer:** B
**Section: (none)**
**Explanation**

**Explanation/Reference:**

**QUESTION 100**
You have received a trouble ticket from a user who could not leave a message for John Doe at extension 2001. The user stated that when the call was forwarded to voice-mail, the Cisco Unity opening greeting was played instead of John Doe's greeting. Which three issues could cause this problem? (Choose three.)

A. The exchange server was down and Cisco Unity was operating in UMR mode.
B. A Cisco Unity call routing rule was added that prevented the Attempt Forward rule from being applied to the call.
C. The voice-mail port used was configured only for Message Notification and MWI Outdial.
D. Extension 2001 was not configured with the correct voice-mail profile.
E. John Doe's mailbox does not have the correct extension configured in Cisco Unity.
F. A translation pattern modified the redirecting number.

**Correct Answer:** BDE
**Section: (none)**
**Explanation**

**Explanation/Reference:**

**QUESTION 101**
Partition A contains four route patterns. The calling search space assigned to Device B contains only Partition A. When Device B dials 1136, which of the route patterns will be selected?

A. 1[14]XX
B. 11X!
C. 1[^2-8]XX

D. 1[1-4]XX

**Correct Answer:** B
**Section: (none)**
**Explanation**

**Explanation/Reference:**

**QUESTION 102**
Which two methods can be used to correct database replication issues in a cluster running Cisco Unified CallManager 4.1? (Choose two.)

A. Enter the utils dbreplication repair command at the command-line prompt.
B. Execute the dblhelper utility on the publisher
C. Use the SQL Server Enterprise Manager application to recreate the database subscription.
D. Run the Cisco Unified CallManager BARS utility to restore the database to the subscriber.
E. Use the Informix database utility to recreate the database subscription.

**Correct Answer:** BC
**Section: (none)**
**Explanation**

**Explanation/Reference:**

**QUESTION 103**
Your customer has implemented a multipoint control unit to allow conferencing between existing Cisco Unified Video Advantage users. The multipoint control unit appears to be properly configured, but users are complaining that when they conference they get audio but no video. What is the most likely cause?

A. The region configuration is selecting an incorrect codec.
B. The partition or CSS configuration configuration is preventing video setup.
C. The QoS policy is placing video packets in the default queue.
D. The MRGL configuration is selecting the incorrect conference resource.

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

**Explanation/Reference:**

**QUESTION 104**
You are troubleshooting why a user cannot make calls to the PSTN. You are reviewing trace files and you have found where the user's IP phone initiates the call but you never see the call go out the gateway. What is the next step in troubleshooting this issue?

A. Look in the SDL trace file to see if there is a signal to another Cisco Unified CallManager node with the same time-stamp.
B. Look in the SDL trace file to see if there is a signal to another Cisco Unified CallManager node with the same TCP handle.
C. Look in the IP Voice Media Streaming App trace file to see if an MTP was invoked.
D. Look in the MGCP trace file to determine which MGCP gateway the call was sent to.

**Correct Answer:** A
**Section: (none)**
**Explanation**

**Explanation/Reference:**

**QUESTION 105**
You have two Cisco Unified CallManager clusters each using endpoint authentication. You have moved a small group of users from Cluster 1 to Cluster 2 and now their phones are failing to register with Cluster 2. Each phone displays the error message "failure to authenticate on the CTL file".

Which step should be taken to resolve this issue?

A. Restart the Cisco CTL Provider and the Cisco Certificate Authority Proxy Function.
B. Reset each IP phone so the correct CTL file can be downloaded.
C. Perform a factory reset on each IP phone so the correct CTL file can be downloaded.
D. Press the Settings button on each IP phone, select the Security Configuration menu, and reset the trust list.

**Correct Answer:** C
**Section: (none)**
**Explanation**

**Explanation/Reference:**

**QUESTION 106**
Your company has recently installed a Cisco Unified CallManager cluster and a Cisco Unity voice mail platform. You have received complaints from users that the red MWI light never comes on, even when there are new voice-mail messages in the voice mailboxes. Which two steps must be taken to resolve this issue? (Choose two.)

A. Verify, using the Port Usage tool, that the ports dedicated to MWI on/off are not over-utilized. Add another dedicated port if the current port is over-utilized.
B. Verify that the MWI on/off numbers are unique within the Cisco Unified CallManager cluster dial plan. If they are not, change the MWI on/off numbers in the Cisco Unified CallManager cluster so they are unique and configure the Cisco Unity server so they match
C. Ensure that the number of ports licensed for the Cisco Unity server is greater than or equal to the number of configured ports.
D. Verify that the calls are being sent to the correct ports on the Cisco Unity server. If they are being sent to the incorrect ones by the Cisco Unified CallManager cluster, correct the values in the cluster.
E. Verify that the same numbers are being used for MWI on/off in both the Cisco Unified CallManager cluster and Cisco Unity server. If they are different, change the Cisco Unity server to match the Cisco Unified CallManager cluster.

**Correct Answer:** BE
**Section: (none)**
**Explanation**

**Explanation/Reference:**

**QUESTION 107**
You have developed a dial plan for a Cisco Unifed CallManager 5.0 solution. All the route patterns, partitions, calling search spaces, and translation rules have been configured. Before starting up the system you wish to test the dial plan for errors.

Which Cisco Unifed CallManager tool will simplify this testing?

A. Dial Plan Installer
B. RTMT Traces and Alarms
C. Route Plan Report
D. Dialed Number Analyzer

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

**Explanation/Reference:**

**QUESTION 108**
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

**Correct Answer:** BCE
**Section: (none)**
**Explanation**

**Explanation/Reference:**

**QUESTION 109**
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.

**Correct Answer:** E
**Section: (none)**
**Explanation**

**Explanation/Reference:**

**QUESTION 110**
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.

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

**Explanation/Reference:**
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 111**
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

**Correct Answer:** C
**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.

Viewing Route Plan RecordsThis section describes how to view route plan records. Because you might have several records in

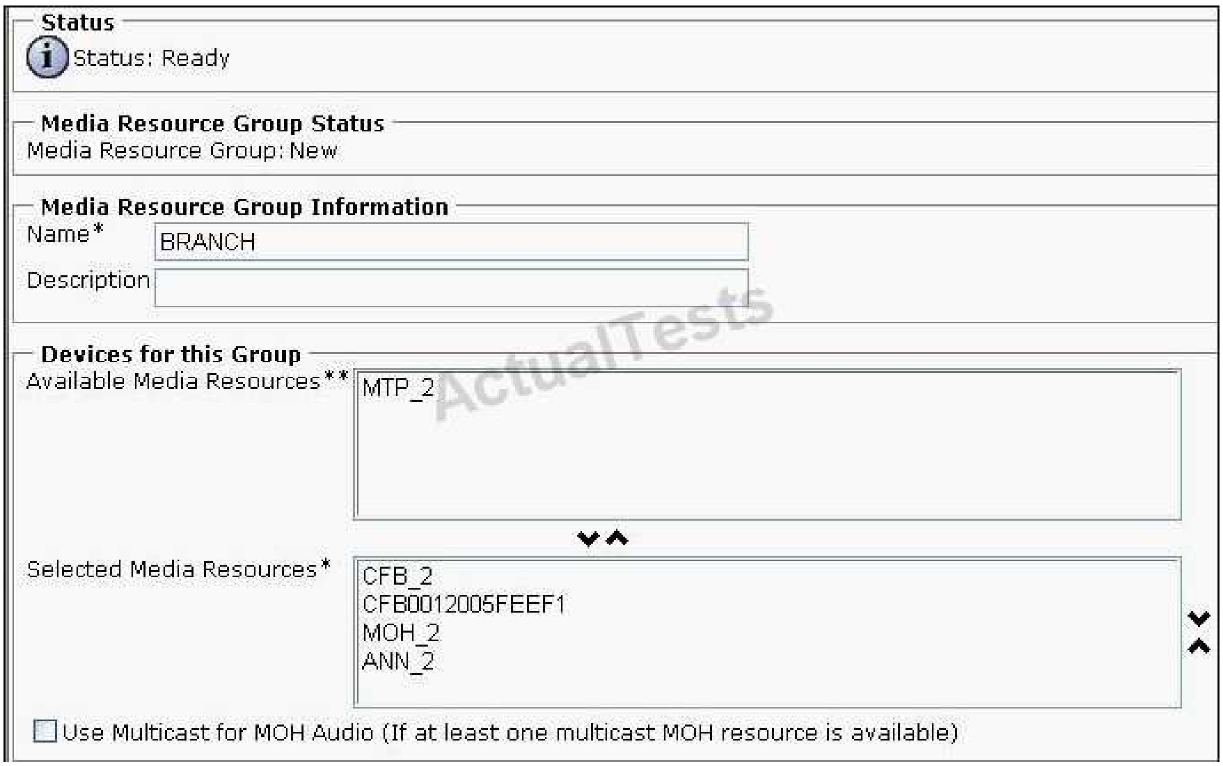
yournetwork, Cisco CallManager Administration lets you locate specific route plan records on the basis of specific criteria. Use the following procedure to generate customized route plan reports.

**Procedure**

Step 1**Choose 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 112**
Refer to the exhibit. You have received a trouble ticket that two engineers at the HQ site tried to conference at a user location at the Branch site and the ad-hoc conference failed.




Further investigation reveals that the remote user can occasionally be conferenced in. Phones are configured to use G.711 for intra-site calls and G.729 for inter-site calls. All phones at the remote site are configured to use MRGL_Branch, which contains only the media resource group named BRANCH.

What should be done to correct this issue?

A. The software conference bridge CFB_2 should be removed from the Branch media resource group.
B. The hardware conference bridge CFB0012005FEEF1 should be listed first in the Branch media resource group so it is utilized if it is available.
C. Additional hardware conference resources should be added.
D. The number of sessions allowed per conference should be increased.

**Correct Answer:** A
**Section: (none)**
**Explanation**

**Explanation/Reference:**

The software conference bridge can support only G.711

**QUESTION 113**
You have added a subscriber server to your Cisco Unified CallManager 5.0 cluster. The server is functioning properly, but regular updates from the publisher are failing. All other publisher- subscriber communications are working properly in the cluster.

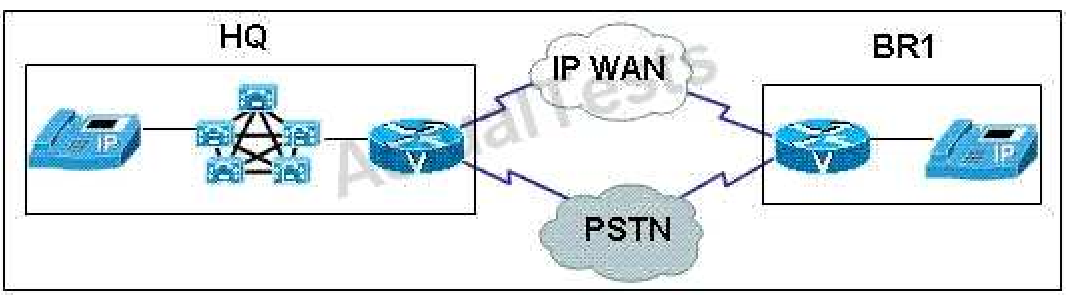
What is the problem?

A. the service isn't included in the current CTL client
B. the correct username and password are not being applied in the subscriber
C. the Cisco Unified CallManager Administrator username and password have been changed in the publisher
D. the CTL client wasn't signed with the security token when the service was included in it
E. the MIC for the service needs to be included in the CTL client

**Correct Answer:** B
**Section: (none)**
**Explanation**

**Explanation/Reference:**

**QUESTION 114**
Refer to the exhibit. A user at BR1 is on a call to the PSTN through the PRI interface on the local H.323 gateway. The IP WAN has just failed and the call has been dropped.




What is causing the call to be dropped?

A. The H.323 gateway has lost the D channel on the PRI.
B. SRST hasn't been configured on the BR1 gateway.
C. The no h225 timeout keepalive command has not been configured on the gateway.
D. The IP phone hasn't registered with SRST.

**Correct Answer:** C
**Section: (none)**
**Explanation**

**Explanation/Reference:**

**QUESTION 115**
You have received a trouble ticket stating that an executive with an account at a bank cannot retrieve account information by phone. When the executive calls the bank, the call is answered and the executive is prompted to enter the account code. However, the bank does not seem to recognize the DTMF tones and disconnects the call.

What is a possible solution to this problem?

A. Configure the voice rtp send-recv command in the gateway.
B. Set the Cisco Unified CallManager service parameter ToSendH225UserInfoMsg to True.
C. Configure the progress_ind setup enable 3 command under the gateway VoIP dial peer.
D. Configure the progress_ind alert enable 8 command under the gateway POTS dial peer.

**Correct Answer:** A
**Section: (none)**
**Explanation**

**Explanation/Reference:**
Using thevoice rtp send-recvCommand to Establish Early Two-Way Audio
The voice path is established in the backward direction as soon as the RTP stream is started. The forward audio path is not cut through until the Cisco IOS gateway receives a connect message from the remote end. In some cases it is necessary to establish a two-way audio path as soon as the RTP channel is opened, before the connect message is received. To achieve this, use thevoice rtp send-recvglobal configuration command you can have a situation where you only have a 1 way path to listen to announcements, it's usually used to fix 1-way audio issues when

they're not supposed to happen intentionally Also helps on DTMF issues

**QUESTION 116**
The following is a partial configuration of an access layer switch:

mls qos map cos-dscp 0 8 12 16 28 32 40 48

mls qos
!

Voice bearer traffic will be set to use which per-hop behavior?

A. EF
B. CS4
C. AF32
D. AF12
E. BE

**Correct Answer:** B
**Section: (none)**
**Explanation**

**Explanation/Reference:**
The LAN CoS maps voice to 8 classes 0 through 7, example below:
cos: 0 1 2 3 4 5 6 7
you then map values to each of these classes for dscp, below is the default values (note that there is no EF (46) equivalent):
cos: 0 1 2 3 4 5 6 7
-
dscp: 0 8 16 24 32 40 48 56
So the question has mapped the class "5" (the voice bearer class) to a value of32 "0 8 12 16 283240 48". This will give it a per hop behavior of CS4/Prec 4

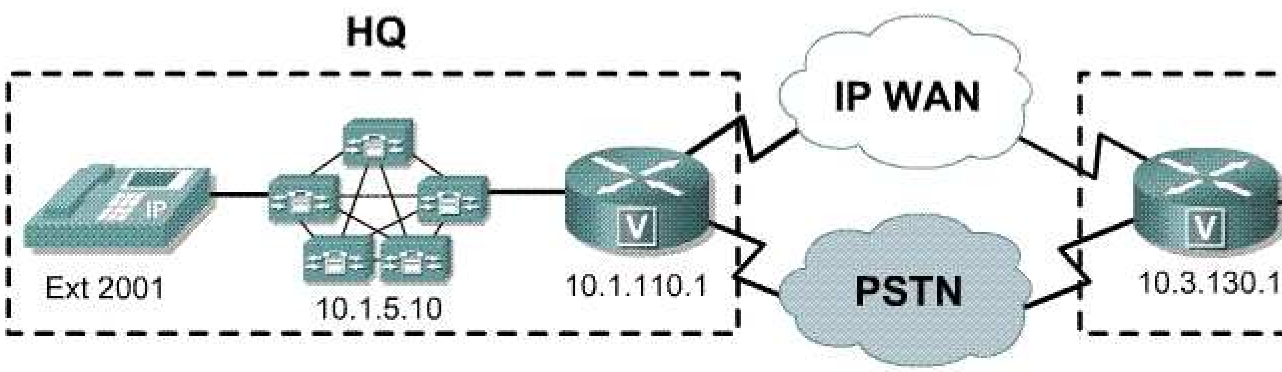
**QUESTION 117**

**1**

Refer to the exhibit. Click the Voice Gateway for the BR2 location to view the output from the **debug**
command and click on 10.1.5.10 to view and search the trace file output. Output can be searched by
Search box and clicking the Find button. Click the X to return to the item.
A trunk has been configured between the Cisco Unified CallManager cluster 10.1.5.10 and a CME at
testing you find that calls cannot be completed when dialing from ext. 2001 to ext. 4001 or from ext.
What is the issue that is preventing calls from being completed in either direction?

- ◯ The **h323-gateway voip bind srcaddr 10.3.130.1** command has been omitted from the BR2 co
- ◯ A translation rule has been applied that is keeping the call from being completed
- ◯ The CSS has been omitted from the trunk configured to BR2
- ◯ An incorrect CSS has been applied to the gateway at HQ
- ◯ The trunk IP address in the Cisco Unified CallManager Information field is incorrect






**Point and Shoot:**

**1**

Refer to the exhibit. Click the Voice Gateway for the BR2 location to view the output from the **debug**
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What is the issue that is preventing calls from being completed in either direction?

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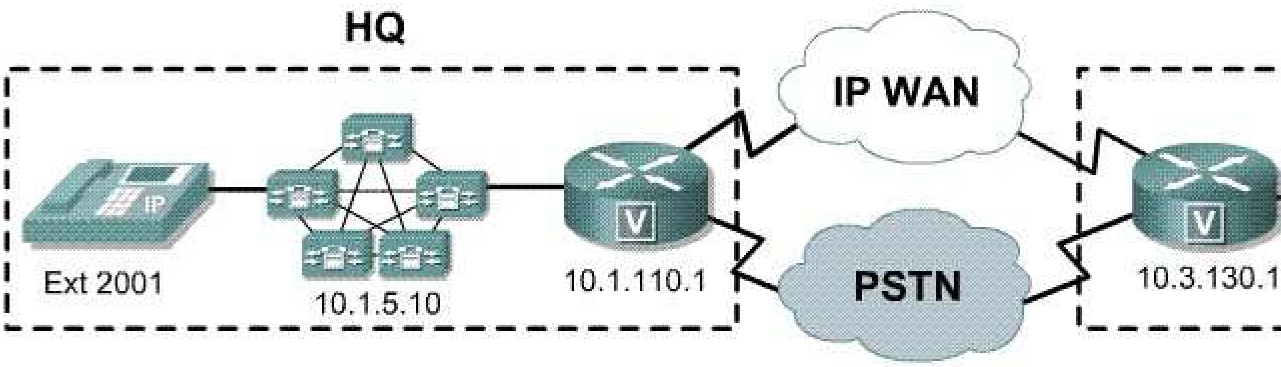
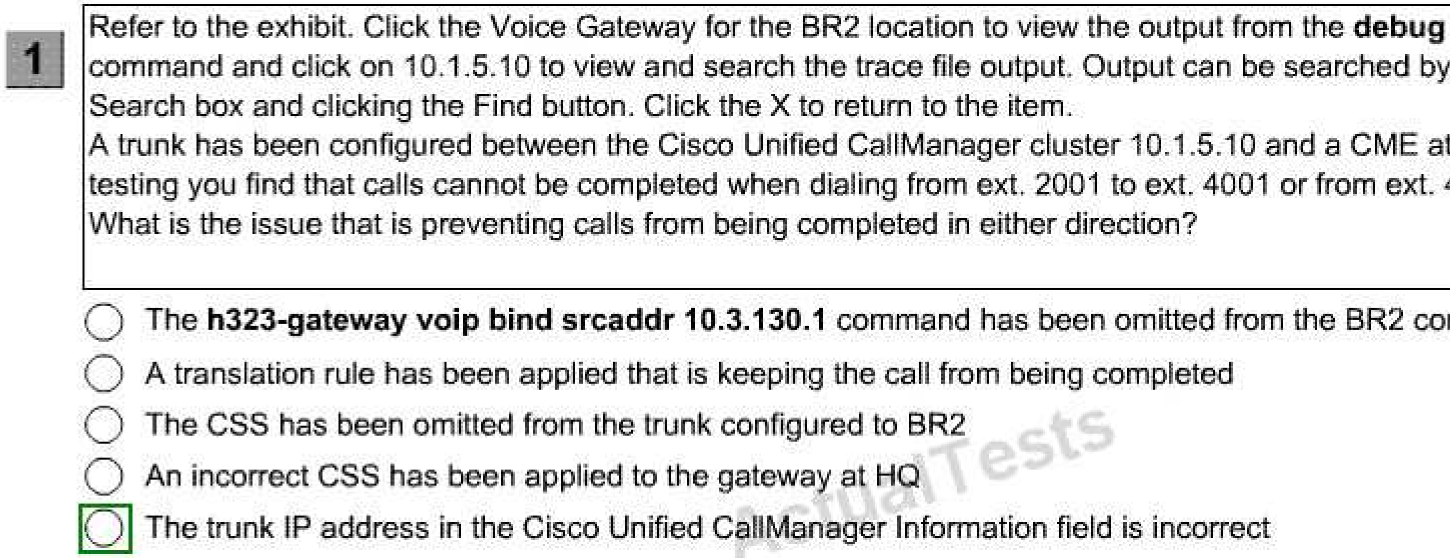
- ◯ A translation rule has been applied that is keeping the call from being completed
- ◯ The CSS has been omitted from the trunk configured to BR2
- ◯ An incorrect CSS has been applied to the gateway at HQ
- ◯ The trunk IP address in the Cisco Unified CallManager Information field is incorrect






**Correct Answer:**



1

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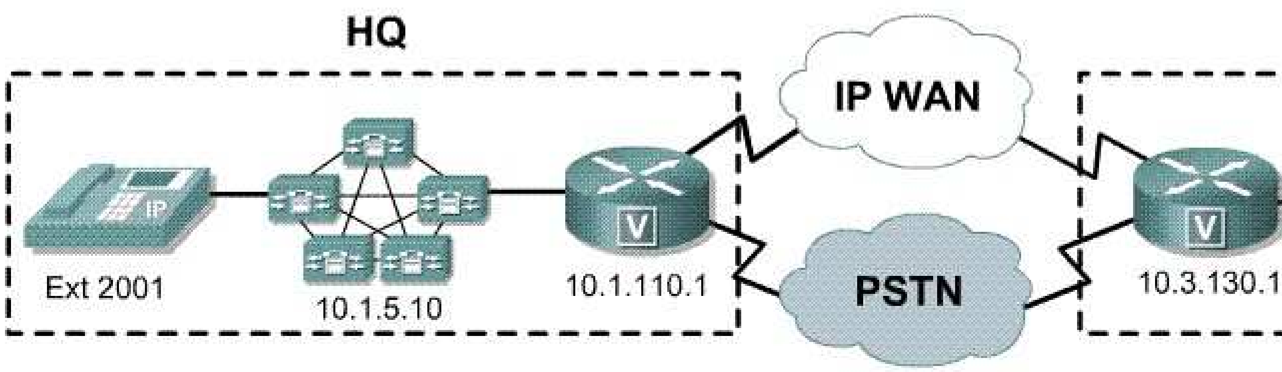
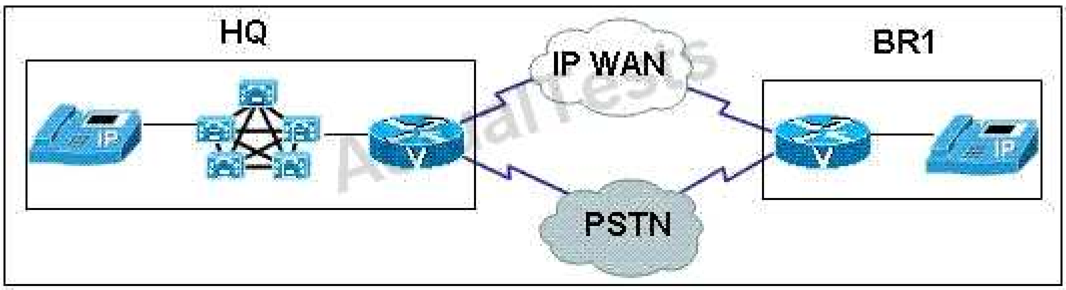
- The trunk IP address in the Cisco Unified CallManager Information field is incorrect




**Section: (none)**
**Explanation**

**Explanation/Reference:**

**QUESTION 118**
Refer to the exhibit. A user is on a call to the PSTN through the MGCP gateway at BR1. The gateway and the IP phone lose the connection to the Cisco Unified CallManager cluster at HQ and the call is dropped. What is causing the call to be dropped?



A. The CallManager cluster has lost control of the D channel on the PRI gateway at BR1.
B. SRST has not been configured on the BR1 gateway.
C. The IP phone has not registered with SRST
D. The no h225 timeout keepalive command has not been configured on the gateway.

**Correct Answer:** A
**Section: (none)**
**Explanation**

**Explanation/Reference:**

**QUESTION 119**
Which method can be used to correct database replication issues in a cluster running Cisco Unified CallManager 5.0?

A. Enter the utils dbreplication repair command at the command-line prompt.
B. Execute the dblhelper utility on the publisher.
C. Use the SQL Server Enterprise Manager application to recreate the database subscription.
D. Run the Cisco Unified CallManager BARS utility to restore the database to the subscriber.
E. Use the Informix database utility to recreate the database subscription.

**Correct Answer:** A
**Section: (none)**
**Explanation**

**Explanation/Reference:**

**QUESTION 120**
You have received a trouble ticket stating that users in accounting are not able to use the CFwdAll softkey to forward their calls to voice mail. Their phones continue to ring when they receive inbound calls, even after they have pressed the CFwdAll softkey and the Messages button.

What is a possible cause of this issue?

A. The users have not been enabled to use CTI.
B. The users have not been associated with their phones.
C. The voice-mail ports are not registered.
D. There is a database replication issue with the subscriber the phones are registered to.
E. The phones are not configured to use the standard phone template.

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

**Explanation/Reference:**
**Explanation/Reference:**
Symptom: IP Phones will not be able to set Call Forward All (CFA) and phones that are already CFA will be unable to clear the condition, even if the phone indicates that it is not forwarded. Further Problem Description: Many times you will be able to tell when the problem first started by looking in the Application Event Viewer on all nodes for an Error message from CallManager. This message should state that an SDL connection to one or more nodes went OOS (out of service). Ideally the low level networking part of CallManager should alert all other processes when the SDL link goes down. This will allow them to reestablish their connections when the CallManager

process recovers or is restarted. In this case they are n ever notified that the service went down, so they do not attempt to recover.

Conditions: After a stop or crash of the CallManager service on any node the connection between the processes that monitor the**database**for changes and the CallManager processes on all nodes will fail to re-establish. This results in the CFA changes being written to the**database**(and can beseen in the CCMAdmin web page) but the CallManager process on each node will not be notified that the change occured.

Workaround: Restarting the**Database**Layer Monitor service on all nodes in the cluster should recover the connection (this is not service impacting).
In extreme cases a restart of CallManager may be necessary as well, which will be service impacting. Note that after the connection is restored the forwarding information internal to CM memory may be corrupted for phones that were "stuck" forwarded during the outage. To fix this change the CallingSearchSpace for CFA under the line that is "stuck". Update the line, then reset the devices. You can then change the CFA CSS back to its original value and the phone should display the correct CFA information and calls should be routed appropriately.

**QUESTION 121**
You have placed all DNs in the Phones partition. During testing you discover that you cannot place calls between IP phones, but you can place calls to the PSTN and voice mail. What is one possible cause of this issue?

A. A database replication issue is preventing calls between phones.
B. An access list is blocking RTP streams in your voice VLAN.
C. An access list is blocking SCCP packets in your voice VLAN.
D. The IP phones have not been assigned a CSS.

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

**Explanation/Reference:**

**QUESTION 122**
You have received a trouble ticket stating that the MWI light is not coming on for a group of users. Further investigation reveals that the affected users are connected to the same subscriber in the cluster. Users that are connected to other subscribers in the same cluster are not experiencing this issue.

What is causing this problem?

A. The MWI numbers for this subscriber have been changed.
B. The Cisco Unity voice-mail services need to be restarted for the subscriber so all the users will receive proper MWI message indication.
C. Database replication has an error or has failed.
D. The voice-mail ports assigned to the subscriber are down.

**Correct Answer:** C
**Section: (none)**
**Explanation**

**Explanation/Reference:**

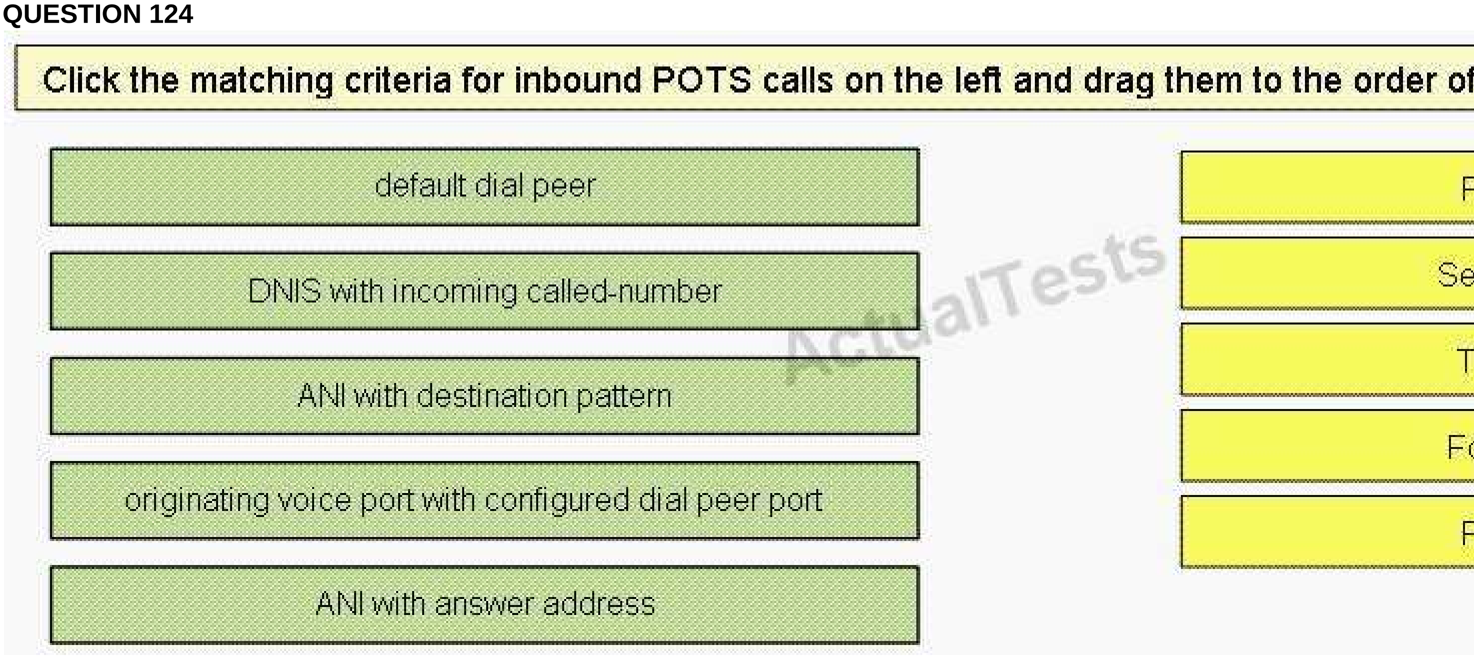
**QUESTION 123**
How does the echo canceller interpret a signal when the ERL value is too low?

A. The echo canceller treats the signal as echo and applies the configured values.
B. The echo canceller considers the signal returning to the gateway as comfort noise generated during periods of silence and does not act on it.
C. The echo canceller will apply the maximum echo-cancel coverage time to the signal to determine if this is echo or voice.
D. The echo signal retuning to the gateway is too loud and the echo canceller interprets it as normal voice instead of echo.

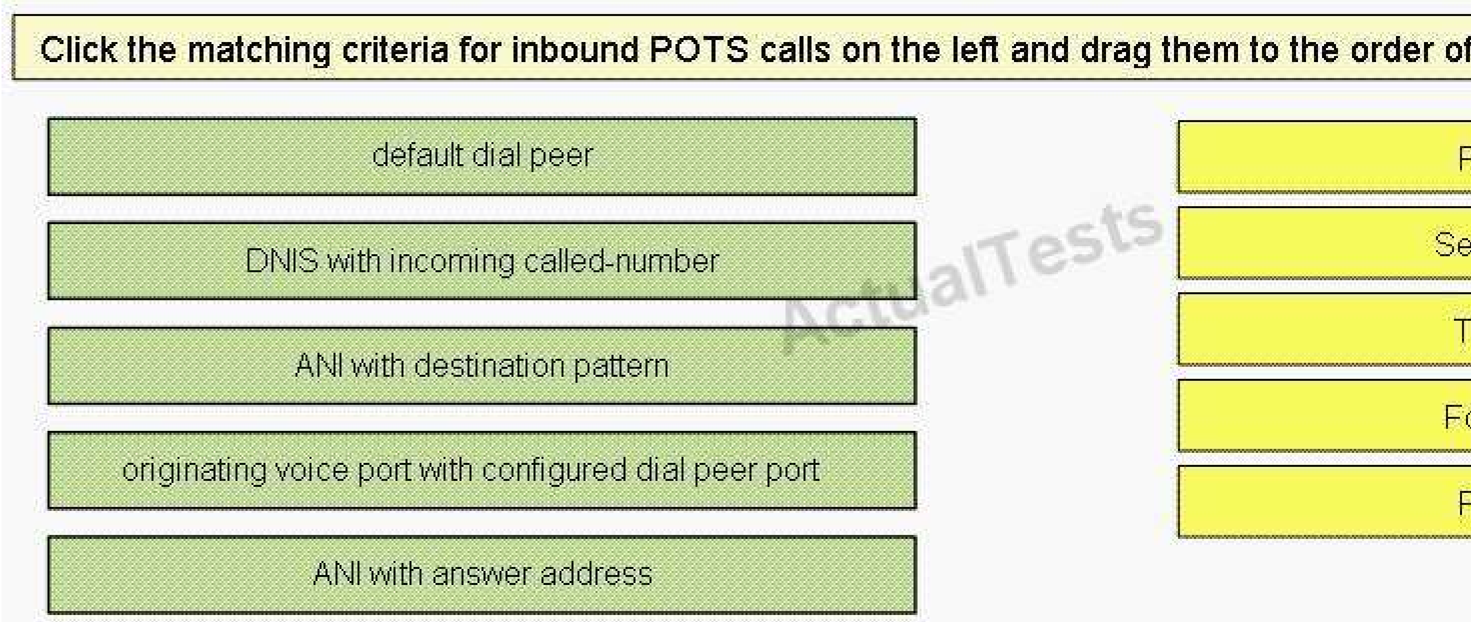
**Correct Answer:** D
**Section: (none)**
**Explanation**

**Explanation/Reference:**
Insufficient echo return loss (ERL) to handle the echo might cause these problems:
Echo canceler does not cancel, but not enough to make echo inaudible. If the ERL value is too low, the total echo return loss seen by the IP network (ACOM) might be insufficient to suppress the echo. ERL needs to be approximately 20 dB (at least 15 dB)
Echo canceler does not cancel.**If the ERL value is too low, the echo signal that returns to the gateway might be too loud**(within 6 dB of the talker signal). This causes the echo canceler to consider it as voice (double-talk) instead of echo. As a consequence, the echo canceler does not cancel it. ERL needs to be approximately 6 dB or higher for the echo canceler to engage. In Cisco IOS Software Release 12.2.13T, you can configure this ERL level. See theEcho Canceler Enhancements in Cisco IOS Software Releases 12.2.11T and 12.2.13Tsection of this document. http://www.cisco.com/en/US/tech/tk652/tk698/technologies_tech_note09186a0080149a1f.shtml

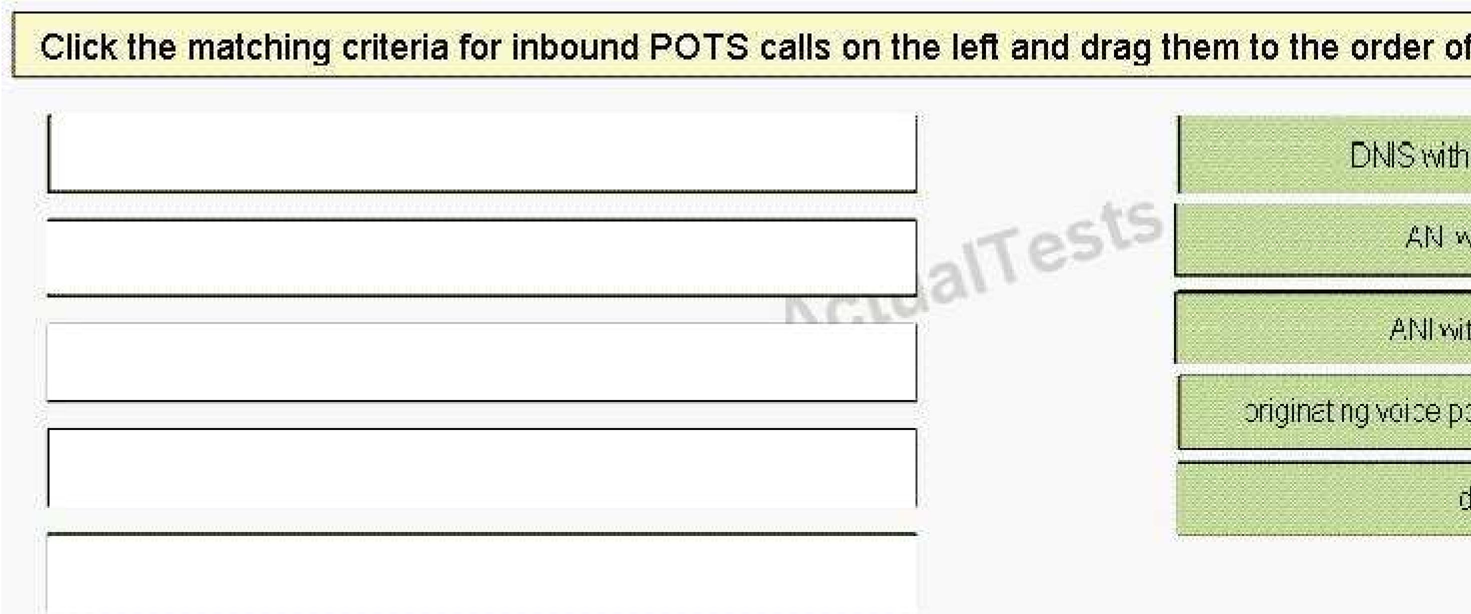
**QUESTION 124**




**Select and Place:**




**Correct Answer:**




**Section: (none)**
**Explanation**

**Explanation/Reference:**
The router or gateway matches these items in this order:
1. Called number (**DNIS**) with the**incoming called-number**command. First, the router or gateway attempts to match the called number of the call setup request with the configured**incoming called-number**of each dial peer. Because call setups always include DNISinformation, it is recommended to use the**incoming callednumber**command for inbound dial peer matching
2. Calling Number (**ANI**)**with theanswer-address**command If no match is found in step 1, the router orgateway attempts to match the calling number of the call setup request with the**answer- address**of each dial peer. This attribute can be useful in situations where you want to match calls based on the calling number (originating)
3. Calling Number**(ANI) with the destination-pattern**command If no match is found in step 2, the router or gateway attempts to match the calling number of the call setup request to the **destination-pattern**of each dial

peer. For more information about this, see the first bullet in theDial Peer Additional Informationsection of this document.
4.**Originating Voice-port**(associated with the incoming call setup request)**with configured dial peer port** ( applicable for inbound POTS call legs) If no match is found in the step 3, the router or gateway attempts to match the configured dial peer**port**to the voice-port associated with the incoming call. If multiple dial peers have the same port configured, the dial peer first added in the configuration is matched.
5. If no match is found in the first four steps, then the**default dial peer 0 (pid:0)**command is used. LINK:http:// www.cisco.com/en/US/tech/tk652/tk90/technologies_tech_note09186a008010fed1.sht ml

**QUESTION 125**
Three calls can successfully be made across a WAN link. When a fourth call is made, the quality of all four calls degrades. Which QoS mechanism can help avoid this problem?

A. CAC
B. FRF.12
C. LFI
D. LLQ
E. priority queuing

**Correct Answer:** A
**Section: (none)**
**Explanation**

**Explanation/Reference:**

**QUESTION 126**
Cisco CallManager 5.0 has just been deployed in two locations across a wide-area link. A distributed model with an intercluster trunk has been used. When you call an IP phone at the remote site the phone rings, but as soon as the other person picks up the phone, the call is dropped.

Where should you look to diagnose the problem?

A. locations
B. system parameters
C. media resource group
D. Cisco Unified CallManager group
E. Cisco Unified CallManager CTI traces

**Correct Answer:** C
**Section: (none)**
**Explanation**

**Explanation/Reference:**

**QUESTION 127**
What are the two most common causes of echo? (Choose two.)

A. impedance mismatch at the two-wire to four-wire hybrid
B. misconfigured tail circuits
C. excessive packet loss
D. signal reflection
E. incorrect IP phone software loads

**Correct Answer:** AD
**Section: (none)**
**Explanation**

**Explanation/Reference:**

**QUESTION 128**
Echo is always present to some extent in all voice networks. For echo to be a problem, which three conditions must exist? (Choose three.)

A. analog leakage between analog Tx and Rx paths
B. sufficient echo amplitude to be perceived as annoying
C. sufficient power from the talker's side to cause listener echo
D. sufficient delay in echo return for echo to be perceived as annoying
E. an analog two-wire to four-wire hybrid operating below 600 ohm impedance

**Correct Answer:** ABD
**Section: (none)**
**Explanation**

**Explanation/Reference:**
Echo is perceived as annoying when all of these conditions are true: http://www.cisco.com/en/US/tech/tk652/tk698/technologies_tech_note09186a0080149a1f.shtml

**QUESTION 129**
Refer to the exhibit. You have received a trouble ticket stating that calls to international numbers are failing. To place an international call, users dial the access code "9," followed by "011," the country code, and the destination number. After entering the debug isdn q931 command on the MGCP gateway, you have the user attempt the call again.

```
BR1#deb isdn q931
  debug isdn q931 is              ON.
  BR1#
  *Apr 23 05:08:36.464: ISDN Se1/0:23 Q931: TX -> SETUP pd = 8  callref = 0x0002
          Bearer Capability i = 0x8090A2
                  Standard = CCITT
                  Transfer Capability = Speech
                  Transfer Mode = Circuit
                  Transfer Rate = 64 kbit/s
          Channel ID i = 0xA98397
                  Exclusive, Channel 23
          Calling Party Number i = 0x0181, '3002'
                  Plan:ISDN, Type:Unknown
          Called Party Number i = 0x80, '01133205551234'
                  Plan:Unknown, Type:Unknown
  *Apr 23 05:08:36.484: ISDN Se1/0:23 Q931: RX <- RELEASE_COMP pd = 8  callref = 0
  x8002
          Cause i = 0x82AC18 - Requested circuit/channel not available
BR1#
```

Based on the debug output, what is the most likely cause of this problem?

A. Cisco Unified CallManager is not stripping the access code before sending the call to the gateway.
B. The gateway dial peer needs to prefix "011" to the called number so the PSTN knows this is an international call.

C. The user's CSS does not permit international calls.
D. The TON is incorrect.
E. The circuit is not configured correctly or has a physical layer issue.

**Correct Answer:** E
**Section: (none)**
**Explanation**

**Explanation/Reference:**
This question is to test the troubleshooting of the call failure via the debug information. According to the debug information, the channel requested by the call is unavailable. There are two reasons:
1. The channels used by two endpoints are different;
2. Because of the configuration error, the connection cannot be established.
Cause No. 44 - requested circuit/channel not available [Q.850] This cause is returned
when the circuit or channel indicated by the requesting entity**cannot be provided by the other side of the interface**.

**QUESTION 130**
You have received a trouble ticket from John Doe at extension 2001, which states that he cannot call a new employee who has been assigned extension 2005. When John Doe attempts to call 2005 he hears a message that his call cannot be completed as dialed. What is one possible cause of this issue?

A. The new employee's phone is not registered with Cisco Unified CallManager due to a physical cable or IP connectivity issue.
B. John Doe's CSS does not contain the partition assigned to extension 2005.
C. Extension 2005 was not assigned a partition.
D. A Cisco Unity voice-mail box has not yet been created for the new employee.

**Correct Answer:** B
**Section: (none)**
**Explanation**

**Explanation/Reference:**

**QUESTION 131**
A company has migrated to a Cisco Unified CallManager IP Telephony system and is now replacing the existing voice-mail system with a Cisco Unity voice-mail system. A small group of users has been established to test the new voice mail system. The users were able to initialize their mailboxes and record greetings; however, during testing the callers were sometimes unable to leave voice-mail messages for Cisco Unity users.

Which two issues could cause this problem? (Choose two.)

A. There is a mismatch in the number of ports configured in Cisco Unified CallManager and Cisco Unity.
B. There is a mismatch in the MWI on/off numbers configured in Cisco Unified CallManager and Cisco Unity.
C. The call transfer call handlers are not configured correctly.
D. Cisco Unity is in a G.729 region and has not been configured to support G.729.
E. The hunt group is hunting to Cisco Unity ports that have been dedicated for message notification.

**Correct Answer:** BE
**Section: (none)**
**Explanation**

**Explanation/Reference:**
http://www.cisco.com/en/US/products/sw/voicesw/ps2237/products_tech_note09186a0080711ae3.
shtml#topic1

**QUESTION 132**
Which Cisco Unity troubleshooting tool would be used to diagnose problems with skinny and MWI messages?

A. Cisco Unity Performance Information and Diagnostics
B. Unity Diagnostic Tool
C. Integration Monitor
D. SysCheck

**Correct Answer:** B
**Section: (none)**
**Explanation**

**Explanation/Reference:**

**QUESTION 133**
Your company has a centralized IP Telephony system and branch offices in eight major cities. The manager of your company help desk recently published local access numbers for external customers to reduce the costs associated with your company's toll-free 800 service. However, when customers call the new local numbers they hear a reorder tone instead of your centralized IVR.

What is the best solution to this issue?

A. Deploy transcoders at each remote location.
B. Deploy transcoders at the central location.
C. Deploy Cisco Unified IP IVRs at each remote location.
D. Increase the number of ports in the centralized IVR.

**Correct Answer:** B
**Section: (none)**
**Explanation**

**Explanation/Reference:**

**QUESTION 134**
Refer to the exhibit. You have received a trouble ticket stating that calls to local PSTN numbers are failing. To place a call, Cisco Unified CallManager users dial the access code "9," followed by seven digits. After entering the debug voice dialpeer command on the H.323 gateway, you have the user attempt the call again.

```
BR2#sh run | begin dial-peer
dial-peer voice 9 pots
 destination-pattern 9T
 port 1/0:1
BR2#
BR2#debug voice dialpeer
voip dialpeer default debugging is on
BR2#
*Jul 20 03:42:06.478: //-1/802B59BE0500/DPM/dpAssociateIncomingPeerCore:
   Calling Number=3002, Called Number=9625432, Voice-Interface=0x0,
   Timeout=TRUE, Peer Encap Type=ENCAP_VOIP, Peer Search Type=PEER_TYPE_VOICE,
   Peer Info Type=DIALPEER_INFO_SPEECH
*Jul 20 03:42:06.482: //-1/802B59BE0500/DPM/dpAssociateIncomingPeerCore:
   Result=NO_MATCH(-1) After All Match Rules Attempt
*Jul 20 03:42:06.482: //-1/802B59BE0500/DPM/dpAssociateIncomingPeerCore:
   Calling Number=3002, Called Number=9625432, Voice-Interface=0x0,
   Timeout=TRUE, Peer Encap Type=ENCAP_VOIP, Peer Search Type=PEER_TYPE_VOICE,
   Peer Info Type=DIALPEER_INFO_SPEECH
*Jul 20 03:42:06.482: //-1/802B59BE0500/DPM/dpAssociateIncomingPeerCore:
   Result=NO_MATCH(-1) After All Match Rules Attempt
*Jul 20 03:42:06.486: //-1/802B59BE0500/DPM/dpMatchPeersCore:
   Calling Number=, Called Number=9625432, Peer Info Type=DIALPEER_INFO_SPEECH
*Jul 20 03:42:06.486: //-1/802B59BE0500/DPM/dpMatchPeersCore:
   Match Rule=DP_MATCH_DEST; Called Number=9625432
*Jul 20 03:42:06.486: //-1/802B59BE0500/DPM/dpMatchPeersCore:
   Result=Success(0) after DP_MATCH_DEST
*Jul 20 03:42:06.486: //-1/802B59BE0500/DPM/dpMatchPeersMoreArg:
   Result=SUCCESS(0)
   List of Matched Outgoing Dial-peer(s):
     1: Dial-peer Tag=9
```

Based on the debug output, what is the most likely cause of this problem?

A. The call is not matching an inbound dial peer resulting in a codec mismatch.
B. Cisco Unified CallManager is stripping the access code, resulting in only seven digits being sent to the gateway.
C. There is a physical layer issue with the circuit.
D. The gateway dial peer needs to prefix the access code to the called number.

**Correct Answer:** B
**Section: (none)**
**Explanation**

**Explanation/Reference:**

**QUESTION 135**
You have received a trouble ticket stating that users are no longer hearing a second dial tone after pressing 9 to initiate an external call. The trouble ticket states that the second dial tone is played only after several additional digits are dialed. What is the mostly likely cause of this problem?

A. The Cisco Unified CallManager server is experiencing CPU spikes, causing a delay in playing the second dial tone.
B. A route pattern beginning with the digit "9" has been added to the route plan without the Provide Outside Dial Tone check box selected.

C. A route pattern beginning with the digit "9" has been added to the route plan with the Call Classification parameter set to OnNet.
D. The first gateway in the route group is not available, causing a delay in playing the second dial tone while Cisco Unified CallManager queries the second gateway.

**Correct Answer:** B
**Section: (none)**
**Explanation**

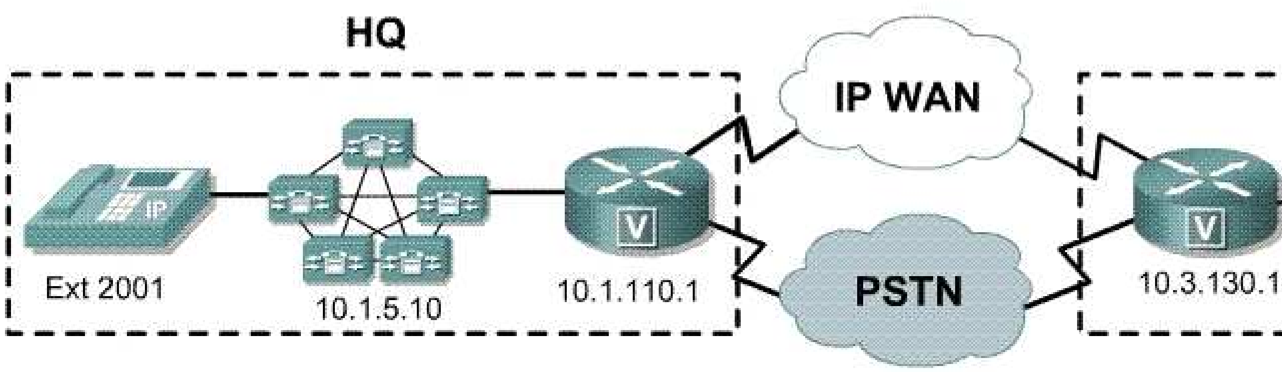
**Explanation/Reference:**

**QUESTION 136**

1

Refer to the exhibit. Click the Voice Gateway for the BR2 location to view the output from the **debug**
command and click on 10.1.5.10 to view and search the trace file output. Output can be searched by
Search box and clicking the Find button. Click the X to return to the item.
A trunk has been configured between the Cisco Unified CallManager cluster at 10.1.5.10 and a CME
testing you find that calls are completed when dialing from ext. 2001 to ext. 4001, but calls from ext. 4
receive a fast busy. What is the issue that is preventing calls from ext. 4001 to ext. 2001 from being c

- The **h323-gateway voip bind srcaddr 10.3.130.1** command needs to be added to BR2
- A translation rule has been applied that is keeping the call from being completed
- The CSS has been omitted from the trunk configured to BR2
- An incorrect CSS has been applied to the gateway at HQ
- The trunk IP address in the Cisco Unified CallManager Information field is incorrect




**Point and Shoot:**

**1**

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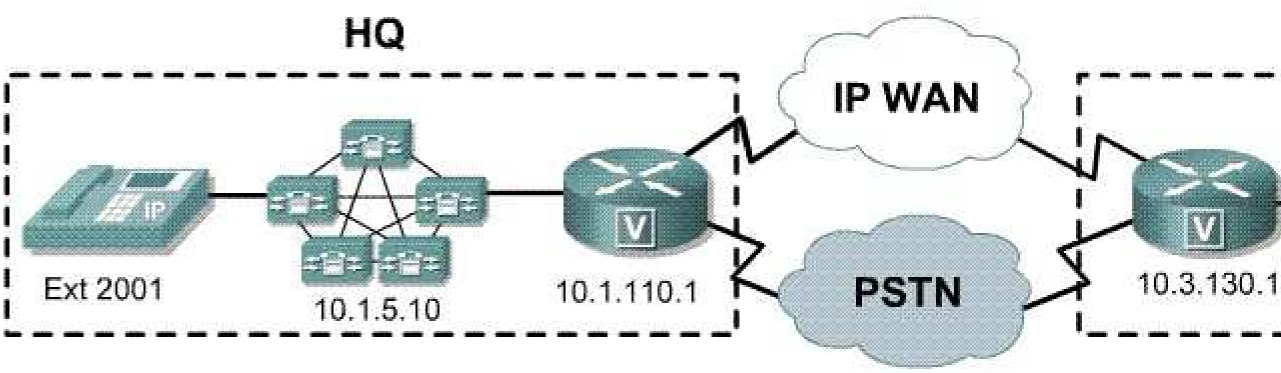


**Correct Answer:**

1

Refer to the exhibit. Click the Voice Gateway for the BR2 location to view the output from the **debug**
command and click on 10.1.5.10 to view and search the trace file output. Output can be searched by
Search box and clicking the Find button. Click the X to return to the item.
A trunk has been configured between the Cisco Unified CallManager cluster at 10.1.5.10 and a CME
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- An incorrect CSS has been applied to the gateway at HQ
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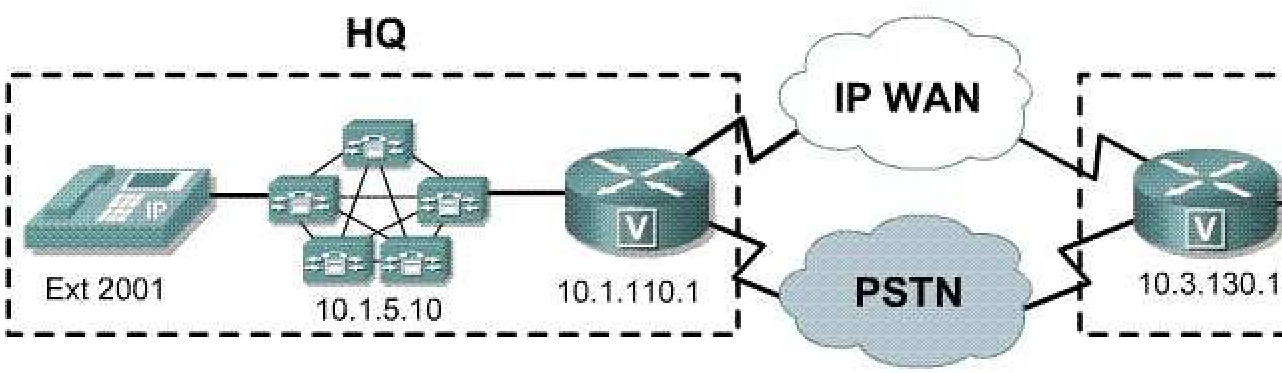



**Section: (none)**
**Explanation**

**Explanation/Reference:**

**QUESTION 137**
Which three capabilities cannot be configured if the default dial peer is matched? (Choose three.)

A. disable DID
B. invoke a Tcl application
C. enable dtmf-relay
D. disable VAD
E. set codec to G.711
F. set preference to 1

**Correct Answer:** BCD
**Section: (none)**
**Explanation**

**Explanation/Reference:**
**Explanation/Reference:**

The Default Dial-Peer 0 peer_tag=0, pid:0
If no incoming dial peer is matched by the router or gateway, the inbound call leg is automatically routed to a default dial peer (POTS or Voice-Network). This default dial peer is referred to as**dial- peer 0**or**pid:0**.
**Dial-peer 0**(pid:0) has a default configuration that cannot be changed.
**The defaultdial-peer 0fails to negotiate non-default capabilities, services, and applications such as:**
http://www.cisco.com/en/US/tech/tk652/tk90/technologies_tech_note09186a008010fed1.shtml

**QUESTION 138**
Refer to the exhibit. You have received a trouble ticket stating that calls to international numbers are failing. To place an international call, users dial the access code "9," followed by "011," the country code, and the destination number. After entering the debug isdn q931 command on the MGCP gateway, you have the user attempt the call again.

```
BR1#deb isdn q931
debug isdn q931 is              ON.
BR1#
*Apr 23 05:41:07.828: ISDN Se1/0:23 Q931: TX -> RELEASE pd = 8  callref = 0x0003
*Apr 23 05:41:07.840: ISDN Se1/0:23 Q931: RX <- RELEASE_COMP pd = 8  callref = 0
x8003[OK]
*Apr 23 05:41:16.368: ISDN Se1/0:23 Q931: TX -> SETUP pd = 8  callref = 0x0004
        Bearer Capability i = 0x8090A2
                Standard = CCITT
                Transfer Capability = Speech
                Transfer Mode = Circuit
                Transfer Rate = 64 kbit/s
        Channel ID i = 0xA98381
                Exclusive, Channel 1
        Calling Party Number i = 0x0181, '3002'
                Plan:ISDN, Type:Unknown
        Called Party Number i = 0x80, '01133205551234'
                Plan:Unknown, Type:Unknown
*Apr 23 05:41:16.416: ISDN Se1/0:23 Q931: RX <- CALL_PROC pd = 8  callref = 0x80
04
        Channel ID i = 0xA98381
                Exclusive, Channel 1
*Apr 23 05:41:16.428: ISDN Se1/0:23 Q931: RX <- DISCONNECT pd = 8  callref = 0x8
004
        Cause i = 0x82BE - Service not allowed
*Apr 23 05:41:16.468: ISDN Se1/0:23 Q931: TX -> RELEASE pd = 8  callref = 0x0004
*Apr 23 05:41:16.484: ISDN Se1/0:23 Q931: RX <- RELEASE_COMP pd = 8  callref = 0
x8004
BR1#
```



Based on the debug output, what is the most likely cause of this problem?

A. Cisco Unified CallManager is not stripping the access code before sending the call to the gateway.
B. The gateway dial peer needs to prefix "011" to the called number so the PSTN knows this is an international call.
C. The user's CSS does not permit international calls.
D. The TON is incorrect.
E. The circuit is not configured correctly or has a physical layer issue.

**Correct Answer:** C
**Section: (none)**
**Explanation**

**Explanation/Reference:**
**Explanation/Reference:**
Cause Code Origination Point
Look @ message**Chanel ID i = 0x828381**
The first byte (most significant) after**0x**indicates the point in the circuit path where the disconnect cause code appears.

| | |
|---|---|
| 82 | 82—the public network near the local user (local telco switch |

The next byte (83 in the sample output) that follows the cause code origination point byte is the **Disconne** **Cause Code**. This byte helps you to troubleshoot the disconnection. Use this table to associate a Disconn Cause Code (in Hex) and the Cause Description to determine the disconnect reason:

| | | |
|---|---|---|
| 83 | No route to destination | The call routes through an intermediate network that does not serve the destination address. Th**is cause indicates that the called user is not reachable. A user is not reachable when the network used to route the call does not serve the required destination**. This cause is supported on a network-dependent basis. |

The last two hexadecimal digits (**BE**in the example) are optional. You do not commonly use these digits for diagnostic purposes. However, you can sometimes use this byte to furnish additional information for the Disconnect Cause Code. The**debug isdn q931**output can sometimes contain these digits. http://www.cisco.com/en/US/tech/tk801/tk379/technologies_tech_note09186a008012e95f.shtml

**QUESTION 139**
Which of these is a correct definition of "Alert"? Select the best response.

A. an event that provides the run-time status of a system, notifications when problems have occurred, and sometimes problem resolution
B. an event that can be configured to send information to multiple destinations, each of which has its own alert event level
C. an event, either a preconfigured or user-defined, that provides the current status and history of events in the Cisco Unified CallManager cluster
D. an event that provides the current view and history of SDI/SDL messages, information from which is stored by Cisco Unified CallManager in an SQL server database

**Correct Answer:** C
**Section: (none)**
**Explanation**

**Explanation/Reference:**

**QUESTION 140**
Refer to the exhibit. What is the TCP handle of this device? Select the best response.

```
01/29/07 07:32:26.564 CCM|InboundStim – KeepAliveMessage - Send
KeepAlive to Device Controller. DeviceName=SEP0002FD3BAB01,TCPPid =
[1.100.7.173], IPAdder=10.1.120.149, Port 49985, Device Controller=[1,60,87]
```

A. SEP0002FD3BAB01
B. 1.100.7.173
C. 1,60,87
D. 10.1.120.149
E. 87

**Correct Answer:** E
**Section: (none)**
**Explanation**

**Explanation/Reference:**
The Correct answer is 87 (the last number in the diagram).

Because the TCP handle represents a unique value that identifies a specific IP pone registered to this CM Server.
With The TCP handle, you can follow every message sent and received from a particular IP phone. The TCP handle of a device can also be found by looking up the IP phone's MAC Address until you find a keep alive message to that phone. Example:
05/23/2005 12:47:01.874 CCM|StationInit - InboundStim - KeepAliveMessage – Send KeepAlive to Device Controller. DeviceName=SEP000F8F59D304,**TCPHandle=000000004**, IPAddr=192.168.1.128, Port=52952, Device Controller=[1,92,1] <CLID::
StandAloneCluster><NID::192.168.1.140><CT::1,100,93,1.984><IP::192.168.1.128><DEV::SEP0
00F8F59D304>

**QUESTION 141**
Users are complaining that they hear echo when they place calls to the PSTN through the HQ_PSTN gateway. The voice port configuration for the gateway is as follows:

voice-port 1/0:23 input gain -1
output attenuation 1

echo-cancel erl worst-case 3

Why are users hearing echo?

Select the best response.

A. The echo canceller is functioning properly and some echo is to be expected on every call.
B. The actual ERL is 2dB; however, the echo canceller still does not acknowledge the input signal as echo because it is lower than the echo-cancel erl worst-case 3 command.
C. The echo canceller should be configured so that SIN and SOUT have a 2 dB difference in order to function properly.
D. InSignalLevel needs to be adjusted so that it is within the range from -20 dB to -25 dB

**Correct Answer:** B
**Section: (none)**
**Explanation**

**Explanation/Reference:**

**QUESTION 142**

Which method can be used to correct database replication issues in a cluster running Cisco Unified CallManager 5.0? Select the best response.

A. Enter the utils dbreplication repair command at the command-line prompt.
B. Execute the dblhelper utility on the publisher.
C. Use the SQL Server Enterprise Manager application to recreate the database subscription.
D. Run the Cisco Unified CallManager BARS utility to restore the database to the subscriber.
E. Use the Informix database utility to recreate the database subscription.

**Correct Answer:** A
**Section: (none)**
**Explanation**

**Explanation/Reference:**

**QUESTION 143**
As a network technician, you should know the VoIP network components well. Which one of the following components can provide the CAC, bandwidth control and management, and address translation?

A. Gateway
B. Gatekeeper
C. MCU
D. Call agent

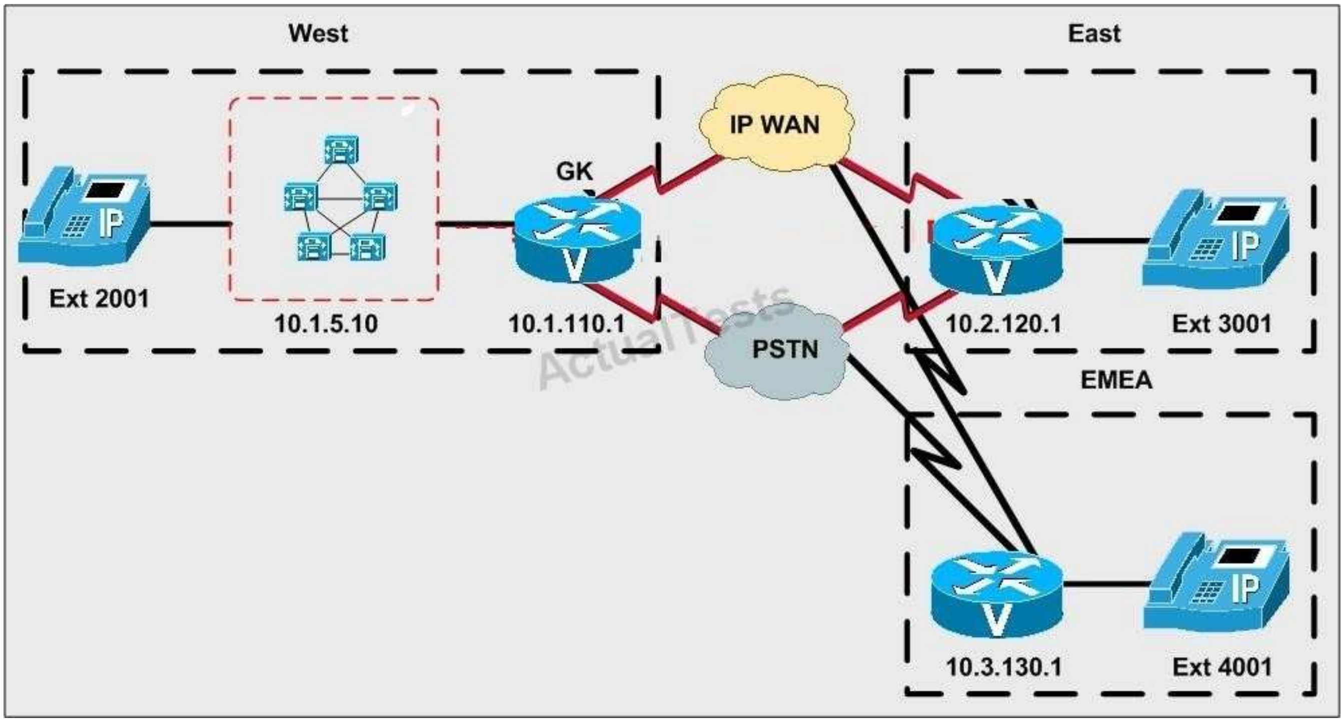
**Correct Answer:** B
**Section: (none)**
**Explanation**

**Explanation/Reference:**

**QUESTION 144**
The following statements describe the icons' functions displayed in the exhibit.

The GK icon can be clicked to view the output of the show gatekeeper endpoint command. The CallManager cluster can be clicked to see the trunk configuration screens.
The X of each popup can be clicked to return to the item.

In order to support three local zones West, East and EMEA, the 10.1.110.1 router is configured with a gatekeeper. After issuing the show gatekeeper endpoints command, you find that the West zone device is lost. Please choose the proper measure that should be taken to the configuration in the CallManager for it to register with the gatekeeper of the zone West.




A. Restart the callmanger so it can register with the gatekeeper
B. Change the Device Name to be the IP address of the gatekeeper in the CallManager Trunk Configuration page
C. Set the Terminal Type to terminal in the Gatekeeper Information field of the Trunk Configuration
D. Configure a zone name on the CallManager trunk

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

**Explanation/Reference:**

**QUESTION 145**
CRTP can compress some specific headers. What are they? (Select three.)

A. Data link
B. IP
C. TCP
D. UDP
E. RTP
F. sRTP

**Correct Answer:** BDE
**Section: (none)**
**Explanation**

**Explanation/Reference:**

**QUESTION 146**

Which two of the following VoIP gateway platforms are considered to be ISRs? (Select two.)

A. Cisco 2600XM Series
B. Cisco 2800 Series
C. Cisco Catalyst 6500 Series
D. Cisco 3800 Series

**Correct Answer:** BD
**Section: (none)**
**Explanation**

**Explanation/Reference:**

**QUESTION 147**
Which two of the following items are the gateway supplementary services? (Select two.)

A. Hold
B. DTMF relay
C. Transfer
D. Transcoding

**Correct Answer:** AC
**Section: (none)**
**Explanation**

**Explanation/Reference:**

**QUESTION 148**
Please choose the function of an H.323 gateway from the following items

A. Converts an alias address to an IP address
B. Responds to bandwidth requests and modifications
C. Transmits and receives G.711 PCM-encoded voice
D. Performs translation between audio, video, and data formats

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

**Explanation/Reference:**

**QUESTION 149**
As a network administrator, you should be familiar with various commands. Which command can be used to enable a SIP user agent on a Cisco router?

A. sip-ua interface configuration mode subcommand
B. sip-ua dial-peer configuration mode subcommand
C. sip-ua global configuration mode command
D. No special command is required. SIP is on by default.

**Correct Answer:** C
**Section: (none)**

**Explanation**

**Explanation/Reference:**
The SIP UA does not require configuration to function, but you might want to make some adjustments. Enter UA configuration mode by issuing the**sip-ua**command. As with dial peers, the options vary by Cisco IOS and device. Table 4-2 shows some common UA commands. http://www.ciscopress.com/articles/article.asp?p=664148&seqNum=6

**QUESTION 150**
Please make the correct selection order for inbound POTS calls.

1. default dial peer

2. DNS with incoming called-number

3. ANI with destination pattern

4. originating voice port with configured dial peer port

5. ANI with answer address

A. 3 > 5 > 2 > 4 > 1
B. 1 > 3 > 5 > 4 > 2
C. 2 > 3 > 5 > 4 > 1
D. 2 > 5 > 3 > 4 > 1

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

**Explanation/Reference:**

**QUESTION 151**
Please choose four mandatory features of an H.323 gatekeeper from the following items.

A. Admission control
B. Address resolution
C. Bandwidth control
D. Zone management

**Correct Answer:** ABCD
**Section: (none)**
**Explanation**

**Explanation/Reference:**

**QUESTION 152**
As a network administrator, you should know the features of H.323 gatekeeper. Which H.323 gatekeeper feature allows calls to be routed to a specific zone, regardless of the zone prefix in the address?

A. Technology prefix with hop-off
B. Default technology prefix
C. E.164 registration
D. Subnet scoping

**Correct Answer:** A
**Section: (none)**
**Explanation**

**Explanation/Reference:**

**QUESTION 153**
As a network administrator, you should be familiar with various commands. Which VoIP dial-peer configuration mode command can be used to specify a technology prefix of 2#, indicating that when the dial peer is used as an outgoing dial peer, a 2# will be prepended to the dial string sent to the gatekeeper?

A. technology-prefix 2#
B. h323-gateway voip tech-prefix 2#
C. gw-type-prefix 2#
D. tech-prefix 2#

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

**Explanation/Reference:**
**Explanation:** VoIP Dial-peerThis command prepends a technology prefix to the called number matched by the dial-peer. It is not used for registration, but for call setup with the Cisco gatekeeper. For example, called number 5551010 becomes 1#5551010.GWY-B1(config)#dial- peer voice 2 voipGWY-B1(config-dial-peer) #tech-prefix ?WORD: A string.VoIP InterfaceThis command registers the Cisco gateway with the defined technology prefix. The technology prefix registration information is sent to the Cisco gatekeeper in the RAS Registration Request (RRQ) message. For example:GWY-B1(config)#interface ethernet 0/0GWY-B1(config-if) #h323-gateway voip tech-prefix ?WORD: A technology prefix that the interface will registerwith the Gatekeeper.http://www.cisco.com/en/US/tech/tk1077/technologies_tech_note09186a00800a8928. shtml

**QUESTION 154**
An H.323 gatekeeper maintains a separate gateway list, ordered by priority, for each of its zone prefixes. Suppose that a gateway does not have an assigned priority for a zone prefix. Please choose the default priority of this gateway.

A. 0
B. 5
C. 10
D. 32

**Correct Answer:** B
**Section: (none)**
**Explanation**

**Explanation/Reference:**
The gatekeeper maintains a separate gateway list, ordered by priority, for each of its zone prefixes. If a gateway does not have an assigned priority for a zone prefix, it**defaults to priority 5**

, which is the median. To explicitly bar the use of a gateway for a zone prefix, the gateway must be defined as having a priority 0 for that zone prefix. When selecting gateways, the gatekeeper identifies a target pool of gateways by performing a longest zone prefix match; then it selects from

the target pool according to priorities and resource availability. If all high-priority gateways are busy, a low-priority gateway might be selected. http://www.cisco.com/en/US/docs/ios/12_3/vvf_c/cisco_ios_h323_configuration_guide/old_archive s_h323/5gkconf.html

**QUESTION 155**
How an H.323 gatekeeper can assume which bandwidth that will be required by a G.729 call?

A. 8 kbps
B. 16 kbps
C. 24 kbps
D. 64 kbps

**Correct Answer:** B
**Section: (none)**
**Explanation**

**Explanation/Reference:**

**QUESTION 156**
As a network administrator, you should be familiar with the bandwidth command. Which parameter of the bandwidth command is used in gatekeeper configuration mode to specify the maximum amount of bandwidth that can be allocated in a zone?

A. interzone
B. total
C. session
D. remote

**Correct Answer:** B
**Section: (none)**
**Explanation**

**Explanation/Reference:**
To specify the maximum aggregate bandwidth for H.323 traffic and verify the available bandwidth of the destination gatekeeper, use thebandwidthcommand in gatekeeper configuration mode. bandwidth{interzone|total|session} {default|zone**zone-name**}**bandwidth-size**

| | |
|---|---|
| **interzone** | Total amount of bandwidth for H.323 traffic from the zone to any other zone. |
| **total** | Total amount of bandwidth for H.323 traffic allowed in the zone. |
| **session** | Maximum bandwidth allowed for a session in the zone. |
| **default** | Default value for all zones. |
| **zone** | A particular zone. |
| *zone-name* | Name of the particular zone. |
| *bandwidth-size* | Maximum bandwidth, in kbps. For**interzone**and**total**, range : 1to10000000. For**session**, range:1 to5000. |

**QUESTION 157**
Please choose the location of RAI configuration from the following options.

A. On a gatekeeper
B. On a gateway

C. On both a gatekeeper and a gateway
D. On a Cisco Unified Communications Manager server

**Correct Answer:** B
**Section: (none)**
**Explanation**

**Explanation/Reference:**
To allow gatekeepers to make intelligent call routing decisions, the gateway reports the status of its resource availability to its gatekeeper. Resources that are monitored are DS0 channels and DSP channels.
The gateway reports its resource status to the gatekeeper with the use of RAS Resource Availability Indication (RAI). When a monitored resource falls below a configurable threshold, the gateway sends an RAI to the gatekeeper that indicates that the gateway is almost out of resources. When the available resources then cross above another configurable threshold, the gateway sends an RAI that indicates that the resource depletion condition no longer exists.
This feature was included in Cisco IOS Software Release 12.0(5)T on the Cisco AS5300 gateway, and Cisco IOS Software Release 12.1(1)T for other gateways in H.323 version 2 http://www.cisco.com/en/US/tech/tk1077/technologies_tech_note09186a0080093f67.shtml

**QUESTION 158**
You have been employed as a network technician in a middle-sized company. Suppose that the default dial peer is matched. Please choose a capability that you must configure.

A. disable DID
B. invoke a Tcl application
C. enable dtmf-relay
D. disable VAD

**Correct Answer:** A
**Section: (none)**
**Explanation**

**Explanation/Reference:**
**Dial-peer 0**(pid:0) has a default configuration that cannot be changed. The default**dial-peer 0**fails to negotiate non-default capabilities, services, and applications such

**QUESTION 159**
Which four of the following options are Cisco-supported IP telephony deployment models?

A. Single site
B. Multisite with distributed call processing
C. Multisite with centralized call processing
D. Clustering over the IP WAN
E. Transcoding

**Correct Answer:** ABCD
**Section: (none)**
**Explanation**

**Explanation/Reference:**

**QUESTION 160**
MGCP use which call control model?

A. Distributed
B. Centralized
C. Ad hoc
D. Hybrid

**Correct Answer:** B
**Section: (none)**
**Explanation**

**Explanation/Reference:**

**QUESTION 161**
If you are required to configure a router to use MGCP on a digital port, which measure will you take?

A. Add the application mgcpapp subcommand to the dial peer
B. Add the service mgcp subcommand to the dial peer
C. Add the parameter application mgcpapp to the ds0-group controller subcommand.
D. Add the service mgcp parameter to the ds0-group controller subcommand

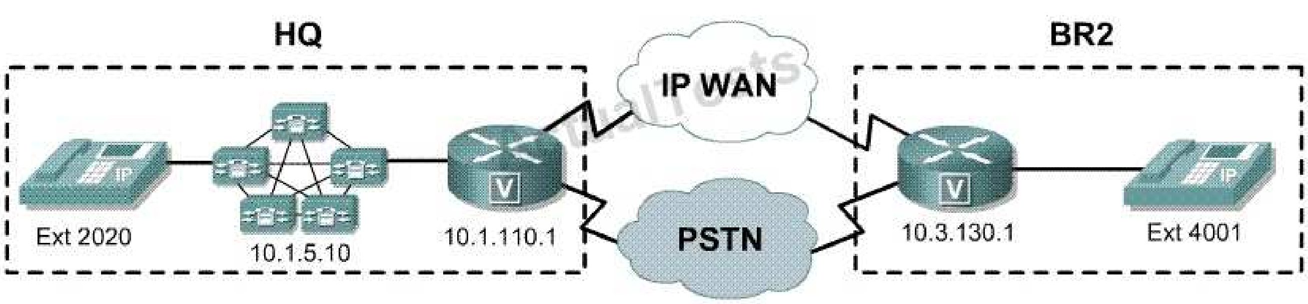
**Correct Answer:** D
**Section: (none)**
**Explanation**

**Explanation/Reference:**

**QUESTION 162**
Look at the following exhibit carefully.

You can click the Voice Gateway for the BR2 location to see the output from the debug voice ccapi inout command and click on 10.1.5.10 to view and search the trace file output. You can also enter a string in the Search box and click the Find button to search the output. X can be clicked to back to the item.

As a network technician, you have recently configured a trunk between the Cisco Unified CallManager cluster at 10.1.5.10 and a CME at a 10.3.130.1. However, in the testing of this configuration, you discover that you cannot complete any call when dialing from ext. 2020 to ext.4001 or from ext.4001 to ext.2020. Please choose the most possible reason from the following statements.




A. A transaction rule has been applied that is keeping the call from being completed.
B. The CSS has been omitted from the trunk configured to BR2
C. An incorrect CSS has been applied to the gateway at HQ.
D. The trunk IP address in the Cisco Unified CallManager information field is incorrect.

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

**Explanation/Reference:**

**QUESTION 163**
As a network technician, you should be familiar with various commands. Which command displays a count of successful and unsuccessful control commands?

A. show mgcp calls
B. show mgcp statistics
C. show mgcp
D. debug mgcp statistics

**Correct Answer:** B
**Section: (none)**
**Explanation**

**Explanation/Reference:**

**QUESTION 164**
In a SIP direct call setup, which message will be sent by the originating UAC to the UAS of the recipient?

A. INVITE
B. RINGING
C. ACK
D. OK

**Correct Answer:** A
**Section: (none)**
**Explanation**

**Explanation/Reference:**

**QUESTION 165**
Which two of the following signaling protocols are peer-to-peer protocols? (Select two.)

A. H.323
B. MGCP
C. SIP
D. SCCP

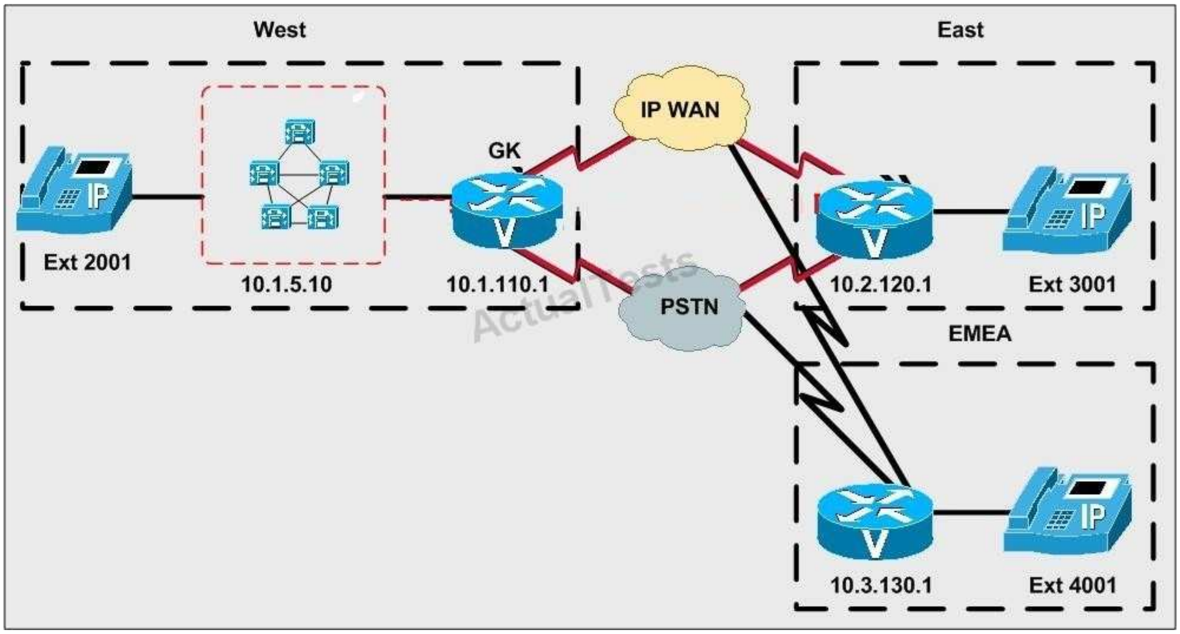
**Correct Answer:** AC
**Section: (none)**
**Explanation**

**Explanation/Reference:**

**QUESTION 166**
Look at the following exhibit carefully.

You can click on the GK icon to view the output of the show gatekeeper endpoint command and click on each voice gateway to view the interface configuration. X of each popup can be clicked to back to the item.

As a network technician, you are recently responsible for configuring a gatekeeper with three local zones called East, West and EMEA. You have accomplished this task. In the East and EMEA zones, CallManager and the gateways are all registered in the West Zone, however, each of them should be registered in their own zone. In order to fix this issue, which measure should you take?




A. The gatekeeper needs to be stopped and restarted for the changes to take effect.
B. The IP address in the gateway command h323-gateway voip ID needs to be changed to that of the individual gateways in both gateways.
C. The zone name in the gateway command h323-gateway voip ID needs to be changed to the correct zone in both gateways.
D. The gatekeeper can only support one zone and all registering endpoints are being placed correctly in the first configured zone.

**Correct Answer:** C
**Section: (none)**
**Explanation**

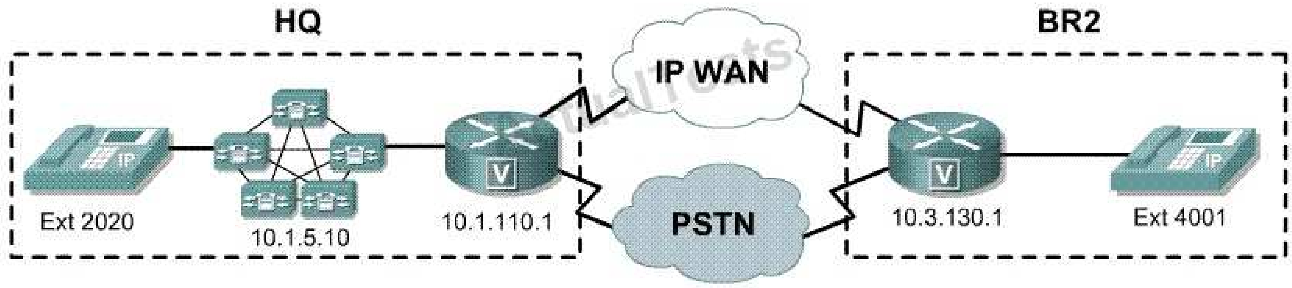
**Explanation/Reference:**

**QUESTION 167**
Look at the following exhibit carefully.

You can click the Voice Gateway for the BR2 location to see the output from the debug voice ccapi inout command and click on 10.1.5.10 to see and search the trace file output. You can also enter a string in the Search box and click the Find button to search the output. X can be clicked to back to the item.

As a network technician, you have recently configured a trunk between the Cisco Unified CallManager cluster at 10.1.5.10 and a CME at a 10.3.130.1. In the testing of this configuration, you discover that you can complete calls when dialing from ext.2020 to ext.4001, however, a fast busy can only be received when dialing from ext.4001 to ext.2020.

Please choose the most possible reason of this issue from the following statements.




A. The h323-gateway voip bind srcaddr 10.3.130.1 command has been omitted from the BR2 configuration.
B. A transaction rule has been applied that is keeping the call from being completed.
C. A CSS has been omitted from the trunk configured to BR2.
D. An incorrect CSS has been applied to the gateway at HQ

**Correct Answer:** A
**Section: (none)**
**Explanation**

**Explanation/Reference:**

**QUESTION 168**
In a Cisco UCM single-site deployment, please choose the maximum number of IP phones that can register with a UCM cluster.

A. 2500
B. 7500
C. 10,000
D. 30,000

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

**Explanation/Reference:**

**QUESTION 169**
In a Cisco UCM multisite WAN with centralized call-processing deployment model, what redundancy feature should be configured on remote site routers to supply 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:**

**QUESTION 170**
Look at the following options carefully. Which two tasks are performed by the RAS signaling function of H.225.0? (Select two.)

A. Performs bandwidth changes.
B. Transports audio messages between endpoints.
C. Performs disengage procedures between endpoints and a gatekeeper.
D. Allows endpoints to create connections between call agents.

**Correct Answer:** AC
**Section: (none)**
**Explanation**

**Explanation/Reference:**

**QUESTION 171**
As a network administrator, you should be familiar with various commands. Which command can be used to designate a source IP address for a voice gateway?

A. h323-gateway voip interface
B. h323-gateway voip h323-id
C. h323-gateway voip bind srcaddr
D. Voice service

**Correct Answer:** C
**Section: (none)**
**Explanation**

**Explanation/Reference:**

**QUESTION 172**
Look at the following options. Which are SIP servers? (Select four.)

A. Registrar
B. Redirect
C. Location
D. Proxy

**Correct Answer:** ABCD
**Section: (none)**
**Explanation**

**Explanation/Reference:**

**QUESTION 173**
The knowledge about RAS message is very important. Which of the following RAS messages can be sent by using either unicast or multicast?

A. RRQ
B. ARQ

C. GRQ
D. RIP

**Correct Answer:** C
**Section: (none)**
**Explanation**

**Explanation/Reference:**
Typically, RAS communications is carried out via UDP through port 1719 (unicast) and 1718 (multicast)

**QUESTION 174**
Given the following configuration, what IP address will GK1 use to send and receive RAS messages?

GK1 (config)#interface serial 0/0/0

GK1 (config-if)#ip address 192.168.0.2 255.255.255.0

GK1 (config-if)#exit

GK1 (config)#interface serial 0/0/1

GK1 (config-if)#ip address 172.16.0.2 255.255.255.0

GK1 (config-if)#exit

GK1 (config)#gatekeeper

GK1 (config-gk)#zone local SanJose cisco.com 172.16.0.2

GK1 (config-gk)#zone remote Austin cisco.com 192.168.0.1

GK1 (config-gk)#zone prefix SanJose 2... GK1 (config-gk)#zone prefix Austin 3...

A. 192.168.0.2
B. 172.16.0.2
C. 192.168.0.1
D. RAS messages will be load balanced between 192.168.0.2 and 172.16.0.2

**Correct Answer:** B
**Section: (none)**
**Explanation**

**Explanation/Reference:**

**QUESTION 175**
You are a network technician working in the Network Company. Recently, users complain that they cannot call the PSTN. With the help of testing, you find that the gateway is not switching to the secondary call agent when the primary call agent is unreachable. In order to permit the MGCP gateway to take use of a different call agent once the primary fails, which configuration should you make?

A. Add ccm-manager fallback-mgcp command to the gateway.
B. Add ccm-manager redundant-host command to the gateway
C. Assign a Cisco Unified CallManager group including the secondary call agent to the gateway
D. Define gateway as a non-gatekeeper-controlled intercluster trunk with the secondary Cisco Unified CallManager defined.

**Correct Answer:** B
**Section: (none)**
**Explanation**

**Explanation/Reference:**

**QUESTION 176**
Which RAS message does a gateway use to request admission to a network and to also request phone number to IP address resolution?

A. ARQ
B. IRQ
C. LRQ
D. RRQ

**Correct Answer:** A
**Section: (none)**
**Explanation**

**Explanation/Reference:**
Admission messages between endpoints (**like a gateway**) and gatekeepers provide the basis for call admissions and bandwidth control. Gatekeepers authorize access to H.323 networks with the confirmation of or rejection of an admission request. This table defines the RAS admission messages

| Admission Messages | |
|---|---|
| ARQ (Admission_Request) | An attempt by an endpoint to initiate a call. |
| ACF (Admission_Confirm) | An authorization by the gatekeeper to admit the call. This message contains the IP address of the terminating gateway or gatekeeper and enables the original gateway to initiate call control signaling procedures. |
| ARJ (Admission_Reject) | Denies the request of the endpoint to gain access to the network for this particular call. |

http://www.cisco.com/en/US/tech/tk1077/technologies_tech_note09186a00800c5e0d.shtml

**QUESTION 177**
As a network technician, you should be familiar with RTCP. Which of the following statements best describes a function of RTCP?

A. RTCP provides encryption, message authentication and integrity, and anti-replay service for voice streams.
B. RTCP uses even-numbered UDP ports in the range 16,384??0?10?0?43??ì?0?1C32,767 to transport voice payloads
C. RTCP provides out-of-band control information for an RTP flow
D. RTCP caches an RTP packet-Layer 3 and Layer 4 headers in the routers at each end of a link, resulting in lower bandwidth demand for subsequent RTP packets.

**Correct Answer:** C
**Section: (none)**
**Explanation**

**Explanation/Reference:**

**QUESTION 178**
You are a voice technician. If you are required to solve latency issues in a VoIP network, which measures will you take? (Select three.)

A. Use dejitter buffers
B. Increase bandwidth
C. Prioritize voice packets
D. Fragment data packets

**Correct Answer:** BCD
**Section: (none)**
**Explanation**

**Explanation/Reference:**

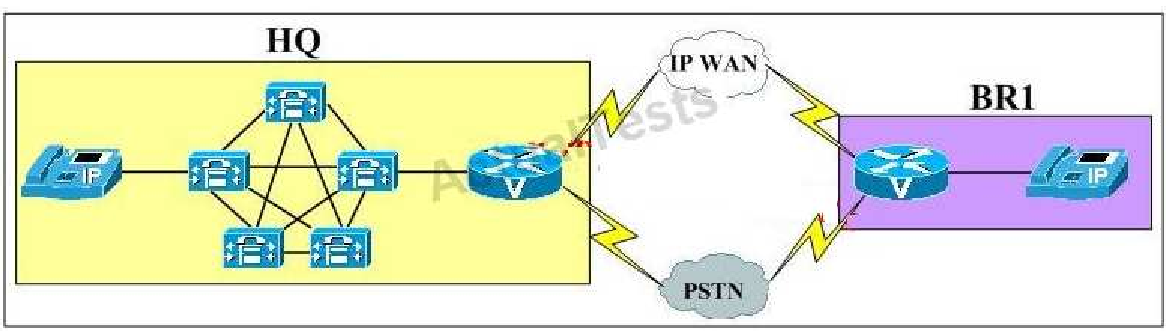
**QUESTION 179**
Please choose two methods of LRQ forwarding from the following items. (Select two.)

A. LRQ init
B. LRQ blast
C. LRQ static
D. LRQ sequential

**Correct Answer:** BD
**Section: (none)**
**Explanation**

**Explanation/Reference:**

**QUESTION 180**
Look at the following exhibit carefully. You are a network technician working in a middle-sized company. Some users complain that the call is always dropped as soon as it connects when they dial from BR1 to HQ through the IP WAN. Please choose the most possible reason of this issue from the following statements.




A. H.245 capabilities exchange has failed due to insufficient bandwidth.

B. A codec mismatch has occurred and transcoders or additional transcoding resources need to be configured at HQ.
C. Packet loss has kept H.225 setup messages from completing the call.
D. MTP resources need to be configured on the BR1 gateway.

**Correct Answer:** B
**Section: (none)**
**Explanation**

**Explanation/Reference:**

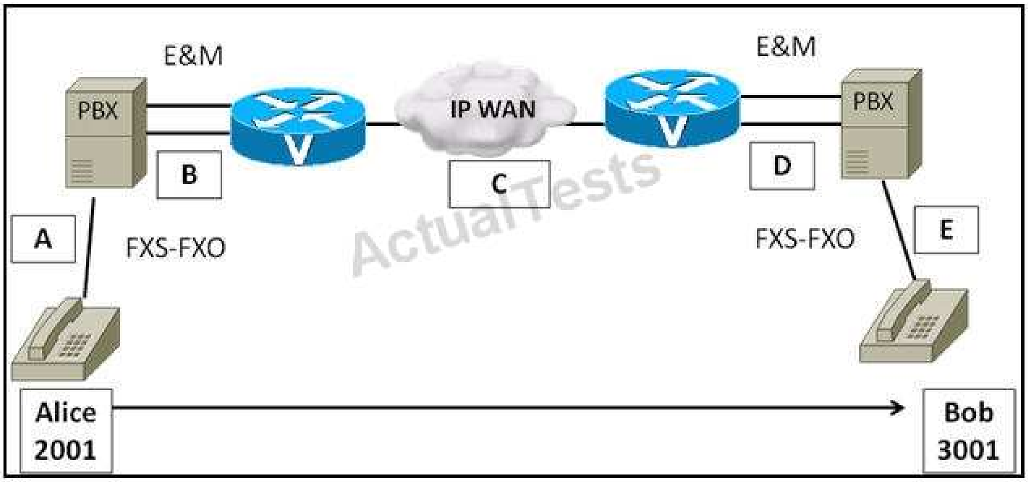
**QUESTION 181**
You are a network technician with many years' experience. Many users complain that they can hear echo when their calls go out an H.323 gateway. You have made some testing for the gateway and have changed the configuration. So the ERL level turns to be 6 dB. Furthermore, the echo-cancel coverage value is raised to 64 ms. Please choose the effect on the voice quality after this modification.

A. Consonants will be chopped by the echo canceller.
B. The increase in echo-cancel coverage will have no effect on voice quality.
C. The ends of sentences will be chopped by the echo canceller.
D. The echo canceller will take 2-3 seconds longer to converge at the beginning of the call.

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

**Explanation/Reference:**
echo-cancel coveragecommand Adjusts the coverage size of the echo canceller. This command enables cancellation of voice that is sent out through the interface and received back on the same interface within the configured amount of time. If the**local loop**(the distance from the interface to the connected equipment that is producing the echo) is longer, the configured value of this command should be extended. If you configure a longer value for this command, it takes the echo canceller longer to converge. In this case, the user might hear a slight echo when the connection is initially set up. If the configured value for this command is too short, the user might hear some echo for the duration of the call because the echo canceller is not canceling the longer-delay echoes. There is no echo or echo cancellation on the network side (for example, the non-POTS side of the connection).

**QUESTION 182**
**Refer to the exhibit**




When Alice at extension 2001 places a call to Bob at extension 3001, Bob hears Alice's voice twice. What type of echo is this classified as?

A. Talker echo.
B. Listener echo.
C. Tail circuit echo.
D. Front end circuit echo.

**Correct Answer:** B
**Section: (none)**
**Explanation**

**Explanation/Reference:**

**QUESTION 183**
**Refer to the exhibit.**

```
*Mar 24 16:17:54.190: ISDN Se0/0/0:15 Q931: RX <- SETUP pd = 8  callref = 0x00AA
        Bearer Capability i = 0x8090A3
                Standard = CCITT
                Transfer Capability = Speech
                Transfer Mode = Circuit
                Transfer Rate = 64 kbit/s
        Channel ID i = 0xA98381
                Exclusive, Channel 1
        Progress Ind i = 0x8183 - Origination address is non-ISDN
        Calling Party Number i = 0x1180, '4940302156001'
                Plan:ISDN, Type:International
        Called Party Number i = 0x81, '2288223001'
                Plan:ISDN, Type:Unknown
*Mar 24 16:17:54.210: ISDN Se0/0/0:15 Q931: TX -> RELEASE_COMP pd = 8  callref =
                        0x80AA
        Cause i = 0x8081 - Unallocated/unassigned  number
```



The exhibit shows the output of debug isdn q931. An inbound PSTN call was received by an MGCP gateway that is registered with a Cisco Unified Communications Manager. The call failed to ring extension 3001. If the phone at extension 3001 is registered and reachable through the gateway inbound CSS, which two actions can resolve this issue? (Choose two.)

A. Change the significant digits for inbound calls to 4 in the gateway configuration in CiscoUnified Communications Manager.
B. Configure the digit strip 4 on the MGCP gateway configuration in Cisco UnifiedCommunications Manager under Incoming Called Party Settings
C. Configure a translation pattern in Cisco Unified Communications Manager that can beaccessed by the gateway CSS to truncate the called number to four digits.
D. Configure a called-party transformation CSS on the gateway in Cisco UnifiedCommunications Manager that includes a pattern that transforms the number from ten digits to four digits.
E. Configure a voice translation profile in the MGCP Cisco IOS gateway with a voice translation rule that truncates the number from ten digits to four digits.
F. Configure the Cisco IOS command num-exp 2288223001 3001 on the gateway.

**Correct Answer:** AC
**Section: (none)**
**Explanation**

**Explanation/Reference:**

**QUESTION 184**
Which command should you use to resolve a jerky speed issue?

A. playout-delay
B. show voice port
C. comfort-noise
D. echo-cancel enable

E. echo-cancel coverage
F. comfort-echo

**Correct Answer:** A
**Section: (none)**
**Explanation**

**Explanation/Reference:**

**QUESTION 185**
You are trying to access the GUI of Cisco Unified Communications Manager. However, it displays a "not accessible" error. In Cisco Unified Serviceability, which two services should you check for and ensure are running on the Control Center – Network Services page? (Choose two.)

A. Cisco Certificate Expiry Monitor
B. Cisco CallManager
C. Cisco Trust Verification Service
D. System Application Agent
E. Cisco Tomcat Stats Servlet
F. Cisco Tomcat

**Correct Answer:** BF
**Section: (none)**
**Explanation**

**Explanation/Reference:**

**QUESTION 186**
As a voice administrator, you have received reports on issues with call dropping and call failures over a period of time. While troubleshooting, you find that there is a Code Yellow alert due to high CPU usage. You collect the logs that are shown below from the CLI of Cisco Unified Communications Manager.

Nov5 05:12:15, cm01, Error, Cisco CallManager, ccm: 147897: Nov

05 05:12:15.268 UTC: %CCM_CALLMANAGER-CALLMANAGER-3-CodeYellowExit: CodeYellowExit
Expected Average Delay:0 Entry Latency:20 Exit Latency:8
Sample Size:10Time Spent in Code Yellow:2 Number of Calls Rejected Due to

Call Throttling:60 Total Code Yellow Exit:14 High Priority Queue Depth:0

Normal Priority Queue Depth:5 Low Priority Queue Depth:4 Cluster

ID:StandAloneCluster Node ID:cms01, 3653

From these logs, what does "Time Spent in Code Yellow" indicate?

A. A critical overload condition exists that may impact phone registration after 2 hours of this alert.
B. The server stayed in a Code Yellow state for 2 seconds.
C. The server stayed in a Code Yellow state for 2 milliseconds.
D. The server stayed in a Code Yellow state for 2 minutes.
E. The server needs a reboot within 2 hours.
F. There is a call failure and, as a result, one call is rejected every 2 milliseconds.

**Correct Answer:** C
**Section: (none)**
**Explanation**

**Explanation/Reference:**

**QUESTION 187**
A customer is trying to register an IP phone. During the registration process, the IP phone receives the configuration file (.xml) from the TFTP server. Which input can you find in the configuration file that is downloaded to the IP phone?

A. firmware to be loaded on IP phone
B. extension number
C. speed dials
D. valid locally significant certificate
E. location of the DHCP server
F. IP address of the IP phone

**Correct Answer:** A
**Section: (none)**
**Explanation**

**Explanation/Reference:**

**QUESTION 188**
Which port number is used as a backhaul for Media Gateway Control Protocol?

A. 2426
B. 2427
C. 2428
D. 2429
E. 2456
F. 2458

**Correct Answer:** C
**Section: (none)**
**Explanation**

**Explanation/Reference:**

**QUESTION 189**
You are in the final stages of upgrading the Cisco Unified Communications Manager, and you are waiting for dbreplication to complete. Which command should you execute from the Cisco Unified Communications Manager publisher to verify status reports and to check that all the tables are synchronized?

A. utils dbreplication runtimestate
B. utils dbreplication status all
C. utils dbreplication status
D. utils service list
E. utils dbreplication quickaudit
F. utils core active

**Correct Answer:** A
**Section: (none)**
**Explanation**

**Explanation/Reference:**

**QUESTION 190**
In a Cisco Unified Communications Manager cluster, you make a few changes to the publisher server. However, the phones that are registered with the subscriber server do not receive these changes. You verify that the publisher and subscriber servers are up and running in the cluster.

What do you need to do to resolve this problem?

A. Reboot the publisher server.
B. Reboot the subscriber servers.
C. Manually reload the configuration on the phones.
D. Fix the replication between the publisher and subcriber servers.
E. Manually copy the database changes from the publisher to the subscriber.
F. Re-set up the database replication between the publisher and subscriber.

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

**Explanation/Reference:**

**QUESTION 191**
**Refer to the exhibit.**

```
voice service saf
 profile trunk-route 1
  session protocol sip interface Loopback1 transport tcp port 5060
!
 profile dn-block 1
  pattern 1 type global 1408[2-9]XXXXXX
 !
 profile callcontrol 1
  dn-service
   trunk-route 1
   dn-block 1
!
channel 1 vrouter SAF asystem 1
  subscribe callcontrol wildcarded
  publish callcontrol 1
```

When a Cisco Unified Communications Manager Express advertises the directory number pattern in the exhibit, what would the learned pattern be in the RTMT tool on the Cisco Unified Communications Manager?

A. \+1408[2-9]XXXXXX and the ToDID will be 0:
B. 1408[2-9]XXXXXX and the ToDID will be 0:+1408[2-9]XXXXXX
C. \+1408[2-9]XXXXXX and the ToDID will be 0:+1408[2-9]XXXXXX
D. \+1408[2-9]XXXXXX and the ToDID will be empty
E. [2-9] XXXXXX and the ToDID will be 0:+1408[2-9]XXXXXX

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

**Explanation/Reference:**

**QUESTION 192**
Which default switchover method is used by the SCCP client to connect to another Cisco Unified Communications Manager after losing connectivity with the first Cisco Unified Communications Manager?

A. immediate
B. urgent
C. graceful
D. panic
E. recovery
F. static

**Correct Answer:** C
**Section: (none)**
**Explanation**

**Explanation/Reference:**

**QUESTION 193**
What are three requirements for Quality of Service for voice calls? (Choose three.)

A. jitter less than or equal to 30 ms
B. PoE-supported Layer 2 switches used to connect IP Phones
C. one-way latency less than or equal to 150 ms
D. jitter less than or equal to 45 ms
E. guaranteed bandwidth of 384 kbps for a voice call
F. loss less than or equal to 1 percent

**Correct Answer:** ACF
**Section: (none)**
**Explanation**

**Explanation/Reference:**

**QUESTION 194**
If you need to avoid choppy speech, what is the maximum tolerable round-trip delay between two VoIP endpoints?

A. 100 ms
B. 200 ms

C. 300 ms
D. 400 ms
E. 500 ms
F. 800 ms

**Correct Answer:** C
**Section: (none)**
**Explanation**

**Explanation/Reference:**