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NEW QUESTION: 1
Two phones in the same cluster and at the same site have a call currently connected. The site local
A. SRST is active, and all the phones enter SRST mode.
B. No incoming and outgoing calls are possible.
C. 323 PSTN gateway loses connection with Cisco Unified Communications Manager. Which two results do you expect? (Choose two.)
D. The current call is not disconnected.
E. The phones display "CM Fallback Service Operating."
F. Cisco Unified SRST is able to receive incoming calls.
Answer: (SHOW ANSWER)

NEW QUESTION: 2
Which CLI command monitors ILS replication progress?
A. utils ils show peer info
B. utils ils findxnode
C. utils ils lookup
D. utils ils showpeerinfo
Answer: (SHOW ANSWER)

NEW QUESTION: 3
In a SAF deployment, the registration status looks correct and the learned patterns appear reachable, but calls are not routed. What is causing this issue?
A. TCP connection failure with the primary SAF Forwarder
B. network connection failure between the SAF Forwarder and Cisco Unified Communications Manager
C. TCP connection failure with the backup SAF Forwarder

D. network connection failure between the primary and backup SAF Forwarders
Answer: (SHOW ANSWER)

NEW QUESTION: 4
Cisco Unified Mobile Connect has been enabled, but users are not able to switch an in-progress call from their mobile phone to their desk phone. You find out that the Resume softkey option does not appear on the desk phone after users hang up the call on their mobile phone. What do you do to resolve this issue?
A. Issue the voice call disc-pi-off command in the gateway.
B. Enable mobile connect on the user profile.
C. Issue the progress_ind progress disable command in the gateway.
D. Assign Resume softkey on the desk phone.
Answer: A (LEAVE A REPLY)

NEW QUESTION: 5
To maintain proper database integrity, what is the recommended maximum round-trip delay between multiple Cisco VCS appliances in a cluster?
A. 15 ms
B. 25 ms
C. 50 ms
D. 30 ms
E. 80 ms
F. 10 ms
Answer: (SHOW ANSWER)

NEW QUESTION: 6
A user using a CP-9971 SIP phone reports that during a video call, the video portion of the call freezes.
What should you do in order to point to and troubleshoot the issue?
A. Make sure that the camera is connected to the USB.
B. Restart the phone.
C. Verify if the camera shutter is open.
D. Perform a factory reset of the phone.
E. On the phone, navigate to Administrator Settings > Status > Call Statistics > Video > Video statistics > Rcvr Packets statistics. Verify if the phone is receiving packets.
Answer: (SHOW ANSWER)

NEW QUESTION: 7
Some users report that they cannot dial out from headquarters on their Cisco IP Phones to PSTN users, but others can. Which troubleshooting approach is the most direct to isolate the source of the failure of the users that cannot dial out to the PSTN?

A. Use DNA to analyze the dialing permissions of the Cisco IP Phones.
B. Use RTMT to analyze the dialing permissions of the Cisco IP Phones.
C. Use RTMT to generate actual calls to the PSTN.
D. Use DNA to generate actual calls to the PSTN.
Answer: (SHOW ANSWER)

NEW QUESTION: 8
You are trying to register an H.323-based Cisco TelePresence system to Cisco Unified Communications Manager and a Cisco DX70 system to the Cisco VCS Control. Why do neither of the units want to register?
A. The H.323-based system needs to register to the Cisco VCS Control with an E.164 number, and the Cisco DX70 needs the TFTP address to register on the Cisco Unified Communications Manager.
B. Both systems need to register to the Cisco Unified Communications Manager, as the Cisco VCS Control is used only for firewall traversal.
C. Both systems need to register to the Cisco VCS Control, but the H.323-based system needs to have the gatekeeper setting set to "Direct."
D. The H.323-based system needs an E164 number to register to Cisco Unified Communications Manager, and the Cisco DX70 needs to have the MAC address configured first on the Cisco VCS Control.
E. You need Cisco TelePresence Server to register Cisco TelePresence systems.
F. You need Cisco TelePresence Management Suite to register Cisco TelePresence systems.
Answer: (SHOW ANSWER)

NEW QUESTION: 9
Which of these reasons can cause intrasite calls within a Cisco Unified Communications Manager cluster to fail?
A. The MGCP gateway is not registered.
B. The calling phone does not have the correct partition configured.
C. The route partition that is configured in the CCD requesting service is not listed in the calling phone CSS.
D. The calling phone does not have the correct CSS configured.
E. The trunk CSS does not include the partition for the called directory number.
Answer: (SHOW ANSWER)

NEW QUESTION: 10
Refer to topology and Exhibits below:
What is the reason that this MGCP gateway is not registered with Cisco Unified Communications Manager?
A. The primary server address is incorrect.

B. The MGCP domain name is incorrect on either the Cisco Unified Communications Manager or the router.
C. Backhaul/Redundant link port is incorrect
D. This MGCP gateway is not down; it is operational.
Answer: (SHOW ANSWER)
Explanation/Reference:

NEW QUESTION: 11
Refer to the exhibit.
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. 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.
B. Change the significant digits for inbound calls to 4 on the SIP trunk configuration in Cisco Unified Communications Manager.
C. 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.
D. Configure the Cisco IOS command num-exp 2288223001 3001 on the gateway ISDN interface.
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 digit strip 4 on the SIP trunk under Incoming Called Party Settings in Cisco Unified Communications Manager.
Answer: (SHOW ANSWER)

NEW QUESTION: 12
Of the following persistent settings for Cisco TMS-controlled endpoints, TMS overwrites these settings if which five of them are altered on the endpoint? (Choose five.)
A. System Contact
B. Configuration Template
C. E.164 alias
D. H.323 ID
E. SIP URI
F. Active Cisco Unified Communications Manager Address
G. IEEE 802.1x Authentication Password
H. System Name
Answer: B,C,D,E,H (LEAVE A REPLY)

NEW QUESTION: 13

Cisco TelePresence System EX90-A and EX90-B are in a call. EX90-A tries to call EX90-C. When the call is dialed, EX90-B is put on hold. EX90-A and EX90-C are connected, but there is no merge button on the touch panel. What is causing this issue?

A. The multisite option key is missing.

B. The conference configuration is missing.

C. The multisite configuration is missing.

D. The conference option key is missing.

E. The multipoint option key is missing.

F. CiscoTelePresence systems cannot make multipoint calls without a Cisco TelePresence Server.

Answer: (SHOW ANSWER)

NEW QUESTION: 14

Endpoints are configured for both H.323 and SIP using the same URI and Cisco VCS settings, but the endpoints register only as H.323 endpoints. What is causing this issue?

A. The Cisco VCS is blocking the endpoints because of duplicate ID entries.

B. The Cisco VCS has no SIP domains configured.

C. The endpoints do not have the SIP option key installed.

D. SIP does not work, because SIP is used for Cisco Unified Communications Manager registration only.

E. A firewall is blocking all traffic from the endpoints to the Cisco VCS.

Answer: (SHOW ANSWER)

NEW QUESTION: 15

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 logout URL that is defined for the Cisco Extension Mobility service is incorrect or does not exist under the IP Phone Services configuration.

C. The CTI service is not running.

D. The user device profile is not subscribed to the Cisco Extension Mobility service.

Answer: (SHOW ANSWER)

NEW QUESTION: 16

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. Special

B. Error

C. Significant
D. State Transition
E. Detailed
F. Arbitrary
Answer: (SHOW ANSWER)

NEW QUESTION: 17
You are troubleshooting video quality issues on a Cisco TelePresence TX9000 Series system. Which CLI command shows the total number of lost video packets and the received jitter during a call in progress?
A. show call statistics video
B. show call statistics detail
C. show call statistics video detail
D. show call statistics all
Answer: (SHOW ANSWER)

NEW QUESTION: 18
You must integrate a third-party H.323 system with your existing Cisco Unified Communications Manager cluster. When you create an H.323 trunk from the cluster, calls from the cluster to the third- party H.323 system are failing. The vendor of the third-party H.323 device has confirmed that the H.323 call setup time must be reduced. Which two approaches reduce the call setup time from Cisco Unified Communications Manager to the third-party H.323 device? (Choose two.)
A. Implement a software MTP.
B. Implement transcoding with the router DSP resources.
C. Implement transcoding with the Cisco Unified Communications Manager resources.
D. Implement a hardware MTP.
Answer: (SHOW ANSWER)

NEW QUESTION: 19
Refer to Exhibit:
After reviewing the trace in the exhibit, what was the Directory number of the called party?
A. 1905

B. 5010
C. 2003
D. 2001
Answer: (SHOW ANSWER)

NEW QUESTION: 20
When parsing trace output after the call routing decision and path selection have been made, which two records can be found in the CCM|RouteList? (Choose two.)
A. PretransfromDigitString
B. findLocalDevice
C. CallingPartyNumber
D. RouteListCdrc
E. RouteListName
F. PretransformCallingPartyNumber
Answer: (SHOW ANSWER)

NEW QUESTION: 21
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 HQ_MRG must contain the transcoder device. The HQ_MRGL must be assigned to the software conference bridge.
B. Enable the software conference bridge to support G.711 and G.729 codecs in Cisco Unified Communications Manager service parameters.
C. The BR_MRG must contain the transcoder device. The BR_MRGL must be assigned to the BR phones.
D. A transcoder should be configured at the remote site and assigned to all remote phones through the BR_MRGL.
E. The HQ_MRG must contain the transcoder device. The HQ_MRGL must be assigned to the HQ phones.
Answer: (SHOW ANSWER)

NEW QUESTION: 22
Refer to the exhibit.
When the user tried to configure the command maximum sessions 5 under the dspfarm profile 1, the error shown in the exhibit was reported. Which course of action will resolve this issue?
A. Configure the dspfarm all command under the voice card.
B. Ensure that the conference bridge is not registered in Cisco Unified Communications Manager.
C. The maximum conference-participants value must be configured >0.
D. The command dsp services dspfarm must be configured under the voice-card configuration.
Answer: (SHOW ANSWER)

NEW QUESTION: 23

In an MCU call with three Cisco TelePresence MX800 systems and a mobile phone calling in, the three TelePresence MX800 systems suddenly experience low audio levels, but the mobile phone audio levels are correct. What can you do to correct this issue?

A. Turn on AGC on the MCU to adjust the audio levels.

B. Turn on the Auto Adjust levels under "Settings > Audio" on the MCU.

C. Use the mobile phone audio option on the TelePresence MX800 to adjust the mobile phone levels.

D. Mobile phone audio levels can vary, so you cannot correct the issue.

E. Turn on ALG on the MCU to adjust the audio levels.

F. Turn off the audio processors on the TelePresence MX800.

Answer: (SHOW ANSWER)

NEW QUESTION: 24

Refer to Exhibit:

After reviewing the trace in the exhibit, what was the Directory number of the calling party?

A. 1905

B. 2003

C. 5010

D. 2001

Answer: (SHOW ANSWER)

NEW QUESTION: 25

Refer to topology and Exhibits below:

Which command can be used to obtain the status of an MGCP gateway from the IOS device, as shown in the given output?

A. show mgcp-gw

B. show ccm-manager

C. show mgcp registration

D. show mgcp connection

Answer: (SHOW ANSWER)

NEW QUESTION: 26

You configured a Cisco ISR G2 as a SIP gateway, but the gateway does not show that it is registered with Cisco Unified Communications Manager. What is causing this issue?

A. Cisco Unified Communications Manager never shows a SIP gateway as registered even when it is properly configured.

B. The Cisco ISR G2 cannot be a SIP gateway.

C. The gateway does not have Cisco Unified Border Element session licensing.

D. The gateway does not have the UC license installed.

E. Cisco Unified Communications Manager does not show a SIP gateway as registered if it is not properly configured.
F. Cisco Unified Communications Manager does not support SIP gateways.
Answer: (SHOW ANSWER)

NEW QUESTION: 27
Refer to the exhibit.
You're tasked with staging configuration changes to add conference bridge functionality to an existing IOS voice gateway deployment. What command is missing for the configuration to be accepted by the IOS CLI?
A. The command dsp services dspfarm must be configured under the voice-card configuration.
B. The dsp tdm pooling command under the voice-card is missing.
C. The Enhanced IOS Conference Bridge is not configured in Cisco Unified Communications Manager.
D. The dspfarm command under the voice card is missing.
E. The command maximum conference-participants must be configured under the dspfarm profile.
Answer: (SHOW ANSWER)

NEW QUESTION: 28
What is a common reason that an IP Phone cannot get its configuration from Cisco Unified Communications Manager after it obtains the correct IP address information?
A. The DHCP scope has the incorrect Option 150 or 66 defined.
B. The DHCP server is not reachable.
C. The DHCP scope is on the wrong subnet.
D. The DHCP scope is exhausted.
Answer: (SHOW ANSWER)

NEW QUESTION: 29
Which tool allows the administrator to analyze call routing in Cisco Unified Communications Manager without physically placing a call?
A. Cisco IOS Gateway debug commands
B. Cisco Unified Communications Manager Dialed Number Analyzer
C. base configuration information for this user that specifies Class of Restriction, Partition, and Calling Search Space information
D. Cisco Unified Communications Manager OS Administration
E. Cisco Unified Communications Manager Serviceability tools
F. Cisco Unified Communications Manager RTMT trace output
Answer: (SHOW ANSWER)

NEW QUESTION: 30

Where in Cisco TMS would you see if a system is registered to a Cisco VCS or a Cisco Unified Communications Manager?

A. under Registration on the System Administration tab
B. System Overview
C. where you start the Cisco Unified Communications Manager RTMT under Systems and Reports
D. Settings > Provisioning
E. Systems > Registration
F. Navigation > Systems > Registrations
Answer: (SHOW ANSWER)

NEW QUESTION: 31
When a remote endpoint dials in to join a conference that is configured on a Cisco TelePresence Server bridge, the endpoint receives only audio. Other users can successfully join the call with Voice and Video.
What is causing this issue?
A. The endpoint is assigned a region without enough configured bandwidth for video.
B. The endpoint does not have the multisite option installed.
C. The bridge is out of all licenses.
D. The endpoint does not have the partition of the bridge in its CSS.
E. The bridge is not able to host video calls.
Answer: (SHOW ANSWER)

NEW QUESTION: 32
Refer to Exhibit:
What was the cause for the call termination?
A. No route to called number
B. Called party cancelled the call
C. Call completed successfully
D. Outbound gateway was not found
E. Calling party abandoned the call
Answer: (SHOW ANSWER)

NEW QUESTION: 33

During a business-to-business video call through the Cisco Expressway solution, the internal endpoint can call out to the remote endpoint on the Internet, but it does not receive audio or video. The remote endpoint receives both audio and video. What is causing the issue?

A. The firewall is blocking inbound RTP ports.

B. The Cisco Unified Communications Manager is not configured for business-to-business calling.

C. The Cisco Expressway does not have a Rich Media Session license.

D. The Advanced Networking option is not installed on the Expressway Edge.

E. The firewall is blocking SIP signaling.

Answer: (SHOW ANSWER)

NEW QUESTION: 34

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 CLI interface

B. Cisco Unified Reporting

C. Cisco Unified Communications Manager Serviceability interface

D. Cisco IP Phone Device Stats from the Settings button

E. Cisco Unified OS Administration interface

F. Cisco Unified Communications Manager RTMT tool

Answer: (SHOW ANSWER)

NEW QUESTION: 35

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. show output for the T1 controller and voice port configuration in the voice gateway

C. Class of Restriction baseline configuration for the user on Cisco Unified Communications Manager Express

D. show output of the voice translation rules in the voice gateway

E. show output of the ephone and ephone-dn configurations

Answer: (SHOW ANSWER)

NEW QUESTION: 36

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, SAF_Pt. When the IP phone places a call to 3001, what will occur?
A. The call will be placed in a round-robin fashion between the SAF network and SIP_Trunk.
B. The call will be placed in a round-robin fashion between the SAF network and SIP_Trunk. Every other call will fail.
C. The call will fail because it will be blocked by the route pattern.
D. The call will succeed and will be placed via the SAF network. SAF-learned routes always take precedence.
Answer: C (LEAVE A REPLY)

NEW QUESTION: 37
When dialing any external SIP URI for a business-to-business call, an endpoint that is registered to the Cisco VCS Control fails to locate the remote endpoint. The same endpoint can successfully call another endpoint that is registered to the Cisco VCS Expressway. How do you resolve this issue?
A. Add a multisite option to the endpoint.
B. Add traversal call licensing on the Cisco VCS Control.
C. Configure a traversal zone between the Cisco VCS Control and the Cisco VCS Expressway.
D. Configure a proper DNS zone on the Cisco VCS Expressway.
E. Add traversal call licensing on the Cisco VCS Expressway.
F. Configure a SIP route pattern in Cisco Unified Communications Manager.
Answer: (SHOW ANSWER)

NEW QUESTION: 38
When identifying Cisco TelePresence Endpoint traffic characteristics, which three statements are true?
(Choose three.)
A. Latency, jitter, and loss are measured in a round-trip fashion.
B. Latency and jitter are measured at a packet level, based on RTP header sequence numbers and time stamps.
C. Latency, jitter, and loss are measured unidirectionally.
D. Jitter is measured at a packet level, by measuring the arrival time of the packet versus the expected arrival time.
E. Latency and loss are measured at a packet level, based on RTP header sequence numbers and time stamps.
F. Jitter is measured at a video frame level, by measuring the arrival time of the video frame versus the expected arrival time.
Answer: C,E,F (LEAVE A REPLY)

NEW QUESTION: 39

When users in headquarters call branch office users over the WAN link, branch users report poor audio quality. Headquarters users consistently experience acceptable audio quality. Which troubleshooting approach most directly improves the audio quality of the branch users?

A. Make the branch router configuration for CBWFQ match the headquarters router.

B. Make the headquarters router configuration for CBWFQ match the branch router.

C. Make the headquarters router configuration for LLQ match the branch router.

D. Make the branch router configuration for LLQ match the headquarters router.

Answer: C (LEAVE A REPLY)

NEW QUESTION: 40

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-gateway 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.

Answer: B (LEAVE A REPLY)

NEW QUESTION: 41

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 for Cisco Mobile Connect.

C. Enable the device mobility mode on the phone since it is disabled.

D. Make the user an owner of the phone device in the phone device configuration page.

Answer: (SHOW ANSWER)

NEW QUESTION: 42

Refer to the exhibits.

Assume that all learned SAF routes are placed in the SAF_Pt partition. The 3XXX directory 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 be placed in a round-robin fashion between the SAF network and SIP_Trunk.

C. The call will placed in a round-robin fashion between the SAF network and SIP_Trunk. Every other call will fail.
D. The call will fail because it will be blocked by the CCD Blocked Learned Route configuration.
Answer: (SHOW ANSWER)

NEW QUESTION: 43
Which three network conditions and equipment should you avoid to ensure a high-quality Cisco TelePresence experience? (Choose three.)
A. redundant network trunks
B. 10/100 access ports
C. high utilization link with QoS
D. duplex mismatch connections
E. Layer 3 switches
F. network loops
G. network hubs
Answer: (SHOW ANSWER)

NEW QUESTION: 44
Which four performance counters are available when monitoring a Cisco MTP device using the Cisco Unified Communications Manager RTMT? (Choose four.)
A. Resource Idle
B. MTP Streams Active
C. Resource Available
D. MTP Connection Lost
E. MTP Instances Active
F. Resource Total
G. Resource Active
H. Out of Resources
Answer: (SHOW ANSWER)

NEW QUESTION: 45
You have 50 hardware MTP resources and 200 software MTP resources. You want to use hardware resources first, but software is being used first. Where can you confirm the MTP selection order?
A. MTP list
B. Media Resource Group List
C. Cisco Unified Real-Time Monitoring Tool
D. calling search space
E. MGCP gateway
F. phone device pool
Answer: (SHOW ANSWER)

NEW QUESTION: 46

Which Cisco Unified Communications Manager troubleshooting tool can be used to determine the digit manipulation path a call takes within the Cisco Unified Communications Manager system from the perspective of a specific directory number, without having the actual device at hand?

A. Cisco IOS debugs

B. Cisco Unified Communications Manager Serviceability

C. Cisco Unified Syslog Viewer

D. Cisco Unified Communications Manager Dialed Number Analyzer

E. Cisco Unified Communications Manager Real Time Monitoring Tool

Answer: (SHOW ANSWER)

NEW QUESTION: 47

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

A. mismatched domain name on the MGCP gateway and Cisco Unified Communications Manager gateway configuration

B. incorrect MGCP IP address specified in the gateway configuration in Cisco Unified Communications Manager

C. misconfigured route group in Cisco Unified Communications Manager

D. missing the configuration command isdn bind-l3 ccm-manager under the ISDN interface

Answer: A (LEAVE A REPLY)

NEW QUESTION: 48

Refer to topology and Exhibits below:

From the perspective of the Cisco Unified Communications Manager, what is the status of the MGCP gateway?

A. registered

B. initializing

C. registering

D. unknown

Answer: (SHOW ANSWER)

NEW QUESTION: 49
After an IP Phone gets IP address information from DHCP, what is the next step in the initialization process?
A. The phone requests its VLAN information.
B. Nothing else is required, the phone is operational at this stage
C. The TFTP server is contacted for configuration information.
D. CTL and ITL files are downloaded.
E. The DHCP offer is sent from the phone
Answer: (SHOW ANSWER)

NEW QUESTION: 50
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 voicemail hunt pilot. When a call is placed from extension 2001 to 2002, which statement is true?
A. Extension 2002 will ring, and if the call is not answered, the call will match the translation pattern and then be forwarded to voicemail.
B. The call will be answered by voicemail.
C. Extension 2002 will ring, and if the call is not answered, the call will match the translation pattern and then be blocked.
D. Extension 2002 will ring.
E. The call will be blocked.
Answer: (SHOW ANSWER)

NEW QUESTION: 51
To achieve 720p (HD) quality at 30 frames per second on an endpoint that is running TC software, what is the minimum configured call rate?
A. 768 kbps
B. 2560 kbps
C. 1152 kbps
D. 512 kbps
Answer: (SHOW ANSWER)

NEW QUESTION: 52
Refer to the exhibit.
Users are reporting that inbound calls from the PSTN are dropping when not answered within 10 seconds.
Calls come in via ISDN T1 PRI. Which configuration change is needed to prevent the calls from dropping?
A. Remove the timeouts initial 10 command from under the voice-port.

B. Remove the timer receive-rtp 10000 command from under the gateway.
C. Modify the Call Forward No Answer setting in CUCM to redirect calls to Voicemail or another extension.
D. Remove the timer receive-rtcp 2 command from under the gateway.
E. Remove the timeouts wait-release 10 command from under the voice-port.
Answer: (SHOW ANSWER)

NEW QUESTION: 53
Refer to Exhibit:
What protocol was used in this call?
A. H.323
B. SCCP
C. MGCP
D. SIP
Answer: (SHOW ANSWER)