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For example, if Hold was the pressed soft key,CallManager places the active call on user hold.In some trace files you might see a soft key number without the corresponding description. The following list defines each soft key number. QUESTION 2Which device is the initiator of a StationInit message in a Cisco Unified Communications Manager SDI trace? A. Cisco Unified Communications ManagerB. MGCP gatewayC. Cisco Music on Hold serverD. SCCP IP phoneE. SIP Proxy Server Answer: DExplanation:StationInit means that an inbound Transmission Control Protocol (TCP) messagefrom a Skinny station reached CallManager. A Skinny station is any endpoint that uses the Skinny protocol to communicate with CallManager QUESTION 3Refer to the exhibit. You received this debug output to troubleshoot a Cisco IOS MGCP gateway problem at a customer site. Which statement about this endpoint on the Cisco MGCP gateway is true? A. This endpoint is on a T1 Controller 0/1/0.B. This endpoint is on an E1 Controller 0/1/0.C. This endpoint is on a T1 Controller 0/1/1.D. This endpoint is on an E1 Controller 0/1/2.E. This endpoint is on an T1 Controller 0/1/2. Answer: A QUESTION 4Refer to the exhibit. You received this debug output to troubleshoot a Cisco IOS MGCP gateway media-related problem at a customer site. What is the purpose of this message? A. The MGCP gateway is responding to an RQNT message from Cisco Unified Communications Manager to poll the media capabilities on its endpoints.B. The MGCP gateway is responding to an AUEP message from Cisco Unified Communications Manager to poll the media capabilities on its endpoints.C. The MGCP gateway is responding to an AUCX message from Cisco Unified Communications Manager to poll the active calls on its endpoints.D. The MGCP gateway is responding to an MDCX message from Cisco Unified Communications Manager during a call setup.E. The MGCP gateway is responding to a DLCX message from Cisco Unified Communications Manager during a call setup. Answer: E QUESTION 5To which SIP response class do the SIP response codes 300 to 399 belong? A. ProvisionalB. Client FailureC. Server FailureD. SuccessfulE. Redirection Answer: EExplanation:Redirection -- further action needs to be taken in order to complete the request. That is what this class implies. QUESTION 6Which SIP request method enables reliability of SIP 1xx response types? A. ACKB. PRACKC. OPTIONSD. CANCELE. REGISTER Answer: BExplanation:In order to achieve reliability for provisional responses, we do nearly the same thing. Reliable provisional responses are retransmitted by the TU with an exponential backoff. Those retransmissions cease when a PRACK message is received. The PRACK request plays the same role as ACK, but for provisional responses. There is an important difference, however. PRACK is a normal SIP message, like BYE. QUESTION 7Which SIP response is considered a final response? A. 183 Session in ProgressB. 199 Early Dialog TerminatedC. 200 OKD. 180 RingingE. 100 Trying Answer: CExplanation:Indicates the request was successful.Wheather other options state the request is still in progress or request is intiated. QUESTION 8Which two SDP content headers can be found in a SIP INVITE message? (Choose two.) A. ExpiresB. ContactC. Connection InfoD. Media AttributesE. AllowF. CSeq Answer: CDExplanation:Connection info is optional field in sdp wheather Media attributes decide the codec and media type for that call. QUESTION 9Refer to the exhibit. If this SIP call is initiated using early offer, which SIP message will UA#2 use to communicate its media capability to UA#1? A. INVITEB. 180 RingingC. 200 OKD. ACKE. RTP Media Answer: CExplanation:In Early offer, SIP Send SDP in the invite , the other node will send the SDP in the 200 message. QUESTION 10Refer to the exhibit. If this SIP call is initiated using delayed offer, which SIP message will UA#1 use to communicate its media capability to UA#2? A. INVITEB. 180 RingingC. 200 OKD. ACKE. RTP Media Answer: DExplanation:In the Delayed Offer process, the calling does not send its offer in the SIP INVITE Message. The callee sends the offer within the SDP fields of its answer (SIP 200 OK). The calling answers within the ACK message. QUESTION 11Which two responses are examples of client error responses in SIP protocol? (Choose two.) A. 302 Moved TemporarilyB. 404 Not FoundC. 503 Service UnavailableD. 502 Bad GatewayE. 604 Does Not Exist AnywhereF. 408 Request Timeout Answer: BFExplanation:Client Error (400 to 499)--Request contains bad syntax or cannot be fulfilled at this server. This class of 400 to 499 contains only error messages. QUESTION 12Which H.245 information is exchanged within H.225 messages in H.323 Fast Connect? A. Terminal Capability SetB. Open Logical ChannelC. Master-Slave

DeterminationD. Call SetupE. Call Progress Answer: BExplanation:With the standard H.245 negotiation, the two endpoints need three round- trips before they agree on the parameters of the audio/video channels (1. master/slave voting, 2. terminal capability set exchange, and finally, 3. opening the logical channels). In certain situations and especially with high-latency network links, this can last too long and users will notice the delay. QUESTION 13Which two compression formats for high-definition video have technical content that is identical to H.264? (Choose two.) A. MPEG-4 Part 10B. MPEG-4 Part 14C. MPEG-2 Part 7D. AVCE. VC3F. VP8 Answer: ADExplanation:MPEG-4 Part 10, also known as MPEG-4 AVC (Advanced Video Coding), is actually defined in an identical pair of standards maintained by different organizations, together known as the Joint Video Team (JVT). While MPEG-4 Part 10 is a ISO/IEC standard, it was developed in cooperation with the ITU, an organization heavily involved in broadcast television standards. Since the ITU designation for the standard is H.264, you may see MPEG-4 Part 10 video referred to as either AVC or H.264. Both are valid, and refer to the same standard. QUESTION 14Refer to the exhibit. A user is going through a series of dialing steps on an SCCP IP phone (extension 1001) to call another SCCP IP phone (extension 2003). Both phones are registered to the same Cisco Unified Communications Manager cluster. Which user inputs are sent from the calling IP phone to the Cisco Unified Communications Manager, in forms of SCCP messages, after the user pressed the Dial softkey? Note that the commas in answer choices below are logical separators, not part of the actual user input or SCCP messages. A. A separate SCCP message is sent to Cisco Unified Communications Manager for each of the following user inputs: 2, 0, 0, 3.B. A separate SCCP message is sent to Cisco Unified Communications Manager for each of the following user inputs: 2, 0, 1, <<, 0, 3.C. A single SCCP message is sent to Cisco Unified Communications Manager to report that digits 2003 have been dialed.D. A single SCCP message is sent to Cisco Unified Communications Manager to report that digits 201<<03 have been dialed.E. A separate SCCP message is sent to Cisco Unified Communications Manager for each of the following user inputs: 2, 0, 1, <<, 2, 0, 0, 3. Answer: CExplanation:After the user delete phone stop the digit by digit dialing and send it as a whole setup. QUESTION 15How are DTMF digits transported in RFC 2833? A. In the RTP stream with the named telephone events payload format.B. In the RTP stream with the regular audio payload format.C. In SIP NOTIFY messages.D. In SIP INFO messages.E. In SIP SUBSCRIBE messages. Answer: AExplanation:DTMF digits and named telephone events are carried as part of the audio stream, and MUST use the same sequence number and time-stamp base as the regular audio channel to simplify the generation of audio waveforms at a gateway. The default clock frequency is 8,000 Hz, but the clock frequency can be redefined when assigning the dynamic payload type. QUESTION 16Refer to the exhibit. Which DTMF relay method is advertised when the originating SIP gateway sends an INVITE message with a Call-Info header shown? A. RFC 2833B. SIP INFOC. SIP NOTIFYD. SIP KPMLE. In-band audio Answer: CExplanation:You can develop user-specific applications that reside on your network entity and have the ability to subscribe for event services supported by the IMG. If the network entity wants the ability to detect an entered DTMF digit (only telephone event of "###" are currently supported) from the TDM-side of a call to the IP side of a call, the entity can subscribe to the IMG for these events and receive SIP NOTIFY events containing the digit event. QUESTION 17What is the maximum length of any numeric geographic area address in ITU recommendation E.164? A. 15B. 18C. 21D. 22E. 25 Answer: AExplanation:E.164 defines a general format for international telephone numbers. Plan- conforming numbers are limited to a maximum of 15 digits. The presentation of numbers is usually prefixed with the character + (plus sign), indicating that the number includes the international country calling code (country code), and must typically be prefixed when dialing with the appropriate international call prefix, which is a trunk code to reach an international circuit from within the country of call origination. QUESTION 18According to ITU-T E.164 recommendations, which two fields in the National Significant Number code may be further subdivided? (Choose two.) A. Country CodeB. National Destination CodeC. Subscriber NumberD. Regional Significant NumberE. Local User CodeF. National Numbering Plan Answer: BCExplanation:A telephone number can have a maximum of 15 digits .The first part of the telephone number is the country code (one to three digits) .The second part is the national destination code (NDC). The last part is the subscriber number (SN). The NDC and SN together are collectively called the national (significant) number QUESTION 19Refer to the exhibit. A user is going through a series of dialing steps on a SIP Type B IP phone (for example, a Cisco 7975) to call an SCCP IP phone. Both phones are registered to the same Cisco Unified Communications Manager cluster. Assuming that the calling SIP phone is not associated with any SIP dial rules, which statement about how digits are forwarded to Cisco Unified Communications Manager for further call processing is true? A. Each digit is sent to Cisco Unified Communications Manager in a SIP NOTIFY message KPML event, at the time that the user enters the digit on the keypad.B. The SIP IP phone will wait for the interdigit timer to expire, or for the Dial softkey to be selected before sending each digit to Cisco Unified Communications Manager as a separate KPML event in a SIP NOTIFY message.C. The SIP IP phone will wait for the interdigit timer to expire, or for the Dial softkey to be selected before sending all digits to Cisco Unified Communications Manager in a SIP INVITE message.D. The SIP IP phone will wait for the interdigit timer to expire or for the Dial softkey to be selected before sending the first digit in a SIP

INVITE and the subsequent digits in SIP INFORMATION messages.E. The SIP IP phone will send all digits to Cisco Unified Communications Manager in a SIP INVITE message as soon as the fourth digit is pressed. Answer: AExplanation:KPML procedures use a SIP SUBSCRIBE message to register for DTMF digits. The digits themselves are delivered in NOTIFY messages containing an XML encoded body. And it is Out of Band DTMF QUESTION 20Refer to the exhibit. A user is going through a series of dialing steps on a SIP Type B IP phone (for example, a Cisco 7975) to call an SCCP IP phone. Both phones are registered to the same Cisco Unified Communications Manager cluster. Assuming the calling SIP phone is associated with a SIP dial rule with a pattern value of 2001, which statement about how digits are forwarded to Cisco Unified Communications Manager for further call processing is true? A. As each digit is pressed on the SIP IP phone, it is sent to Cisco Unified Communications Manager in a SIP NOTIFY message as a KPML event.B. The SIP IP phone will wait for the interdigit timer to expire, and then send each digit to Cisco Unified Communications Manager as a separate KPML event in a SIP NOTIFY message.C. The SIP IP phone will wait for the interdigit timer to expire, or for the Dial softkey to be selected before sending all digits to Cisco Unified Communications Manager in a SIP INVITE message.D. The SIP IP phone will wait for the interdigit timer to expire, or for the Dial softkey to be selected before sending the first digit in a SIP INVITE and the subsequent digits in SIP INFORMATION messages.E. The SIP IP phone will wait for the interdigit timer to expire, and then send all digits to Cisco Unified Communications Manager in a SIP INVITE message. Answer: EExplanation:Cisco Type B SIP Phones offer functionality based SIP INVITE Message. Every key the end user presses triggers an individual SIP message. The first event is communicated with a SIP INVITE, but subsequent messages use SIP NOTIFY messages. The SIP NOTIFY messages send KPML events corresponding to any buttons or soft keys pressed by the user. Cisco Type B SIP IP Phones with SIP dial rules operate in the same manner as Cisco Type A phones with dial rules. Lead2pass helps millions of candidates pass the Cisco 400-051 exam and get the certification. We have tens of thousands of successful stories. Our dumps are reliable, affordable, updated and of really best quality to overcome the difficulties of Cisco 400-051 certifications. Lead2pass exam dumps are latest updated in highly outclass manner on regular basis and material is released periodically. 400-051 new questions on Google Drive: https://drive.google.com/open?id=0B3Syig5i8gpDQ1ZudWVBRHk3bDQ 2017 Cisco 400-051 exam dumps (All 542 Q&As) from Lead2pass: http://www.lead2pass.com/400-051.html [100% Exam Pass Guaranteed]