CCNA Voice Practice Tests – Describe VoIP Components and Technologies (11-15)

Topic 3 ? Describe VoIP Components and Technologies Question 11 Which two of the following signaling protocols are peer-to-peer protocols?(Choose 2.) A.H.323 B.MGCP C.SIP D.SCCP Answer: A C Explanation: This question tests the classification of IP signaling. The IP signaling can be divided into two categories: peer-to-peer and client-server. Peer-to-peer: There are no registration relations between the terminals, terminal and call agent. It is only necessary to ensure the reachability on IP. The call establishment and parameter selection are both decided after mutual negotiation. The peer-to-peer signaling includes H323 and SIP. Client-Server: The terminal must be registered in the call agent and all the negotiations should be completed by the call agent. The terminal completes the related action according to the direction of call agent. The client-server signaling includes SCCP and MGCP. Question 12 Which three headers are compressed by cRTP? (Choose 3.) A.Data link B.IP C.UDP D.RTP Answer: B C D Explanation: The main point of this question is about RTP header compression. The three headers compressed by cRTP refer to IP, UDP and RTP headers. Then depending on whether the header contains check field, the size of the header will be 2 or 4 bytes after the compression. Question 13 Which of the following 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 16,384-32,767 to transport voice payloads C.RTCP provides out-of-band control information for an RTP flow D.RTCP caches an RTP packet's Layer 3 and Layer 4 headers in the routers at each end of a link, resulting in lower bandwidth demand for subsequent RTP packets. Answer: C Explanation: Explanation: This question tests the function of RTCP. RTCP is usually working in conjunction with RTP, which is mainly used for feedback of line use and congestion during the QOS process. It can be regarded as an out-band management to RTP stream. Question 14 An ITSP (Internet Telephony Service Provider) offers an Internet data service for making telephone calls using VoIP (Voice over IP) technology. Most ITSPs use SIP, H.323, or IAX (although H.323 use is declining)[citation needed] for transmitting telephone calls as IP data packets. Customers may use traditional telephones with an analog telephony adapter (ATA) providing RJ11 to Ethernet connection.What are two benefits of using an ITSP for long distance telephony services? A.The circuits are dedicated only to voice B.Connection to an ITSP is only available in full T1/E1/PRI circuit quantities. C.Connection to an ITSP is very granular and can provision from 1 to 100 simultaneous calls. D.Connection to an ITSP allows you to use the bandwidth that is guaranteed to the voice traffic for data when voice is not using the bandwidth. Answer: C D Explanation: A: false. According to different applied circuit and interface card, T1/E1 circuit can transmit voice or data; B: false. The channelized T1/E1 can be used for the application of data lines and the full time slot is not necessary. C: right. The multi-way calling can be established simultaneously via the connection between T1/E1 circuit and ITSP. D: right. The applied IP circuit can simultaneously transmit voice and data. Question 15 You are CCNA VOICE associate in Lead2pass.com.You are planning stages of deploying a Cisco Unified Communications solution. Previously, Lead2pass was leasing a traditional PBX system from the telcom and very little experience with voice. What two signaling methods between the IP phone and the Cisco Unified Communications Manager Express are available for usage? A.H.323 B.SCCP C.MGCP D.SIP Answer: B D Explanation: This question tests the signaling used for IP Phone registration. For Cisco UC solutions, IP Phone can register to CUCM by two kinds of signalings: SCCP and SIP. SCCP is a client-server signaling. CUCM determines the configuration of IP Phone and call-established negotiation. SIP is a peer-to-peer signaling. No matter which signaling is used by IP Phone to register to the CUCM, it has nothing to do with CUCM, PBX and the communication of gateway. Therefore, we should only take the signaling available for IP Phone registration into consideration.