CCNA Voice Practice Tests – Describe VoIP Components and Technologies (21-23)

Topic 3 ? Describe VoIP Components and Technologies Question 21 Identify four SIP servers. (Choose 4.) A.Registrar B.Gateway C.Redirect D.Location E.Proxy Answer: A C D E Explanation: This question is to examine the application of SIP. There are four types of servers, registrar, redirect, location and proxy. A registrar is a server that accepts REGISTER requests and places the information it receives in those requests into the location service for the domain it handles. A redirect server is a user agent server that generates 3xx responses to requests it receives, directing the client to contact an alternate set of URIs. A proxy server is an intermediary entity that acts as both a server and a client for the purpose of making requests on behalf of other clients. The proxy and redirect servers are used in connection negotiation, while the registrar and location servers are used in the terminal registration and searching. Question 22 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 .Which types of high-density trunk can be used to connect to an ITSP? (Choose all the apply.) A.T1/E1 CAS B.FXS C.T1/E1 PRI D.E&M   Answer: A B C D Explanation: This question is to examine the types of high-density trunk to connect to an ITSP. Channel Associated Signaling (CAS), also known as Per-Trunk Signaling (PTS), is a form of digital communication signaling. FXS is used to connect traditional analog phone or PBX. T1/E1 PRI is also a form of digital signaling. E & M is a traditional line to connect PBX. Question 23 You are CCNA VOICE associate in Lead2pass.com. Your company has a few slow links in its voice and data network. Which two techniques can be used to reduce delay in voice transmission? A.Framerelay B.buffering voice packets C.fragmentation of large packets D.compression of IP, RTP, and UDP headers Answer: C D Explanation? This question is to examine the methods to reduce delay. A: Frame relay is a transmission technology; B: Buffering voice packet is just able to reduce jitter; C: Fragmentation of large packets is able to reduce the serialization delay during data transmission; D: Compression of IP, RTP, and UDP headers can compress the data packets into two or four bytes so as to reduce serialization delay.