Exam Code: 920-804
Technology Standards and Protocols for Converged Networks

Demo Version

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1. While working on a private VoIP network to improve QoS, one of the main impairments turns out to be intermittent muted voice on most VoIP calls. Packet loss has also been detected at about 2%. The network uses the G.711 CODEC for the LAN's and the G.729 (a) CODEC on the WAN. To improve QoS on this network, which recommendation should you make?
A. Increase the overall network bandwidth by 1%.
B. Use the G.729 (a) CODEC end-to-end since G.729 has built in PLC.
C. Change the WAN CODEC to G.723.1 and maximize the speech frame size.
D. Use PLC with the G.711 CODEC and optimize for less than 1% packet loss.
Answer: D

2. An organization that has Internet Telephones and VoIP applications throughout its network is having issues with equipment that has been recently deployed. Given the following network information:
• Recently a disabled firewall that has been forwarding all traffic was discovered on the network. It is located between the segment on which all executives are connected and the rest of the organization's backbone.
• Per the organization's security policy, this firewall was immediately re-enabled to protect the executive data. As a result, the executive network is NO longer able to log in to the call servers.
• The call servers and the Internet Telephones use the SIP and the G.711 CODEC for the voice portion of the connection.
• Upon review of the firewall policies and specifications, it has been identified as a proxy firewall for File Transfer Protocol (FTP) and Hypertext Transfer Protocol (HTTP) traffic. The policies on the firewall permit all traffic from the executive LAN into the organization's backbone and drop all other traffic.
Which recommendation would enable the executive LAN to make VoIP calls again, maximize network security, and require minimal IT support?
A. Install an additional PC-based gateway device to act as a SIP proxy for the VoIP calls.
B. Replace the proxy firewall with a stateful-inspection firewall, which is able to understand SIP.
C. Leave the proxy firewall in place and install an ESP/AH based-VPN (Virtual Private Network) to get the VoIP calls to the call server.
D. Replace the proxy firewall with a router that has packet filtering enabled. Then open up the appropriate ports needed for communication.
Answer: B

3. After assessing a customer's network in preparation for a multi-site LAN/WAN VoIP deployment, you have determined the following:
• Average 5% dropped packets in end to end tests
• Intermittent voice muting
• Unacceptable delay
The customer wishes to upgrade their network and correct these deficiencies. Suggest one solution that will give the most noticeable improvement and prepare the network for future growth with the least complexity.
A. Implement an all optical network.
B. Reduce frame size for all packets.
C. Increase LAN/WAN bandwidth by 30%.
D. Implement QoS technique like DiffServ.
Answer: C

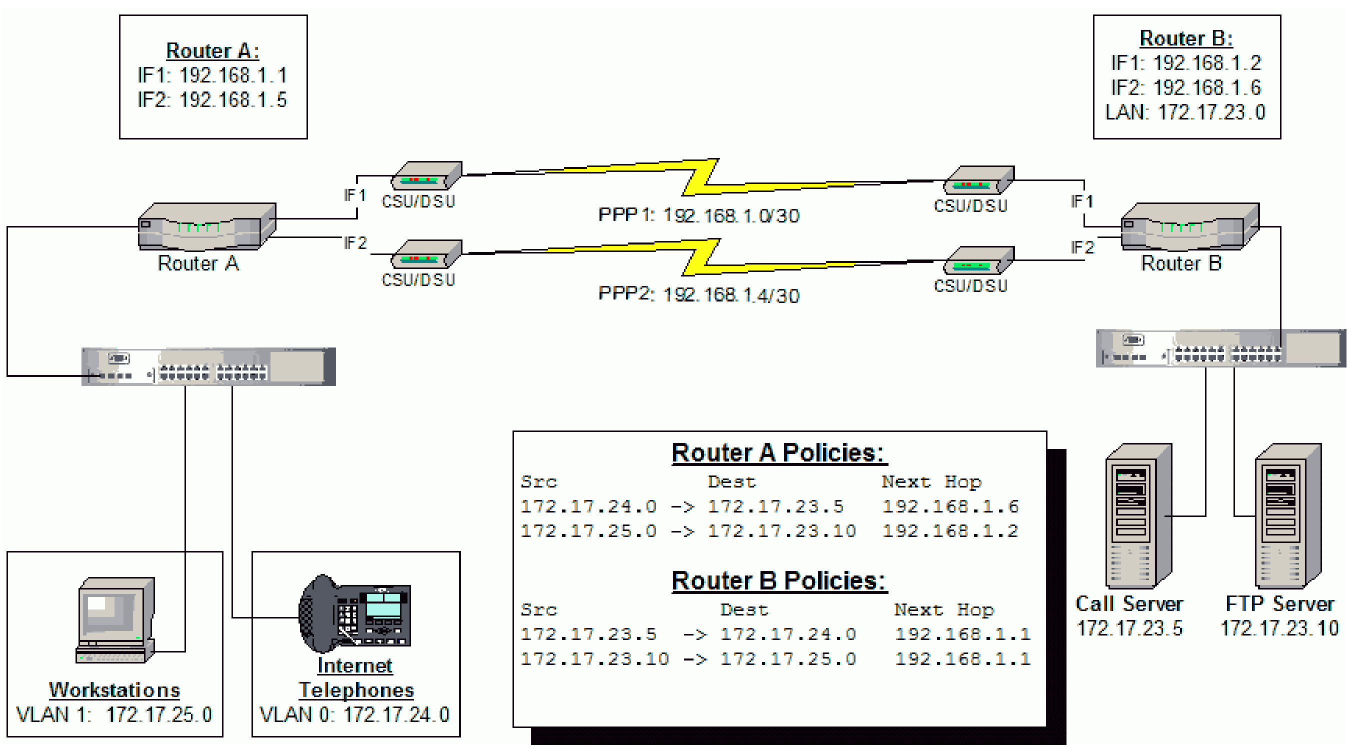
4. A customer is performing VoIP trunking between two IP-enabled Private Branch Exchanges (PBXs) systems over a WAN.
Given the following network information:
• One Frame Relay circuit of adequate bandwidth connects the sites.
• Recently, a second Point-to-Point Protocol (PPP) link provided by a different carrier has been installed for redundancy and load sharing. Since its installation, customers have been reporting that their VoIP calls sound like a bad cell phone call.
• A continuous ping between the two sites reveals varying delay.
Which configuration change should you make to resolve this issue? (Choose two)
A. Increase the jitter buffer size.
B. Increase the route cost of the PPP link.
C. Configure a route policy for VoIP trunks.
D. Disable the Virtual Router Redundancy Protocol (VRRP) master interface.
Answer: BC

5. A customer is performing VoIP trunking between two IP enabled Private Branch Exchanges (PBXs) over a WAN. Some customers have reported that frequently they will place a VoIP call to someone at the remote site and experience one-way speech path or NO speech path at all. Examination of the issue reveals the following:
• Calls with speech-path issues are undergoing proper call-setup.

• When the issue occurs, the Forward Explicit Congestion Notification/Backward Forward Explicit Congestion Notification (FECN/BECN) counters of the routers increment.
• When the issue occurs, the Discard Eligibility (DE) counter increments.
• A continuous ping between the two sites reveals spikes in delay and minor loss.
Which action should you take to resolve the speech-path issue?
A. Increase the jitter-buffer size.
B. Select a different CODEC.
C. Map non-VoIP traffic to a lower-priority queue.
D. Increase the Committed Information Rate (CIR).
Answer: D

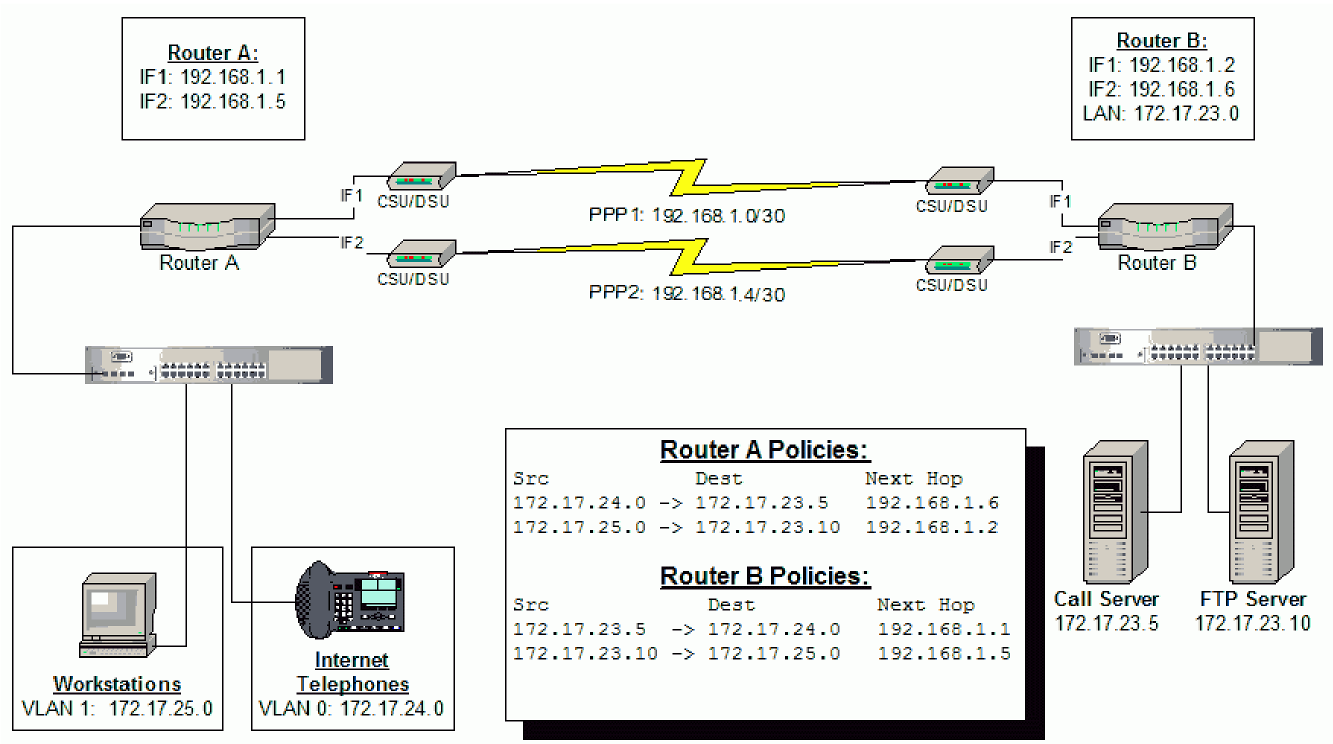
6. A company deployed VoIP in its multisite WAN and has been unsuccessful in getting the VoIP system to work properly.
Given the following network information:
• Each site is connected in a partial mesh topology using 512 kbps fractional T1s.
• The network has been engineered to handle four simultaneous VoIP trunks to each directly connected site to leave room for data traffic.
• So far, the company has only been able to establish two simultaneous voice trunks to any one directly connected site.
Which configurable parameter on the network needs to be modified to solve the customer's issues?
A. voice CODEC
B. VLAN assignment
C. Voice Activity Detection (VAD)
D. Differentiated Services (DiffServ) queuing
Answer: A

7. Click the exhibit button.



Given the network in the exhibit and the following network information:
• NO QoS mechanisms are in place.
• The File Transfer Protocol (FTP) server is receiving high levels of traffic.
• Users are reporting bad voice quality.
Which QoS method/traffic flow change will resolve the VoIP network issues?
A. routing policies in Router A
B. 802.1p prioritization services
C. routing policies in both routers
D. Equal Cost Multi-Path (ECMP) routings
Answer: C

8. Click the exhibit button.



Given the network diagram in the exhibit and the following network information:
- 802.1p is implemented on all LANs and VLANs.
- Call quality degrades as data traffic increases.

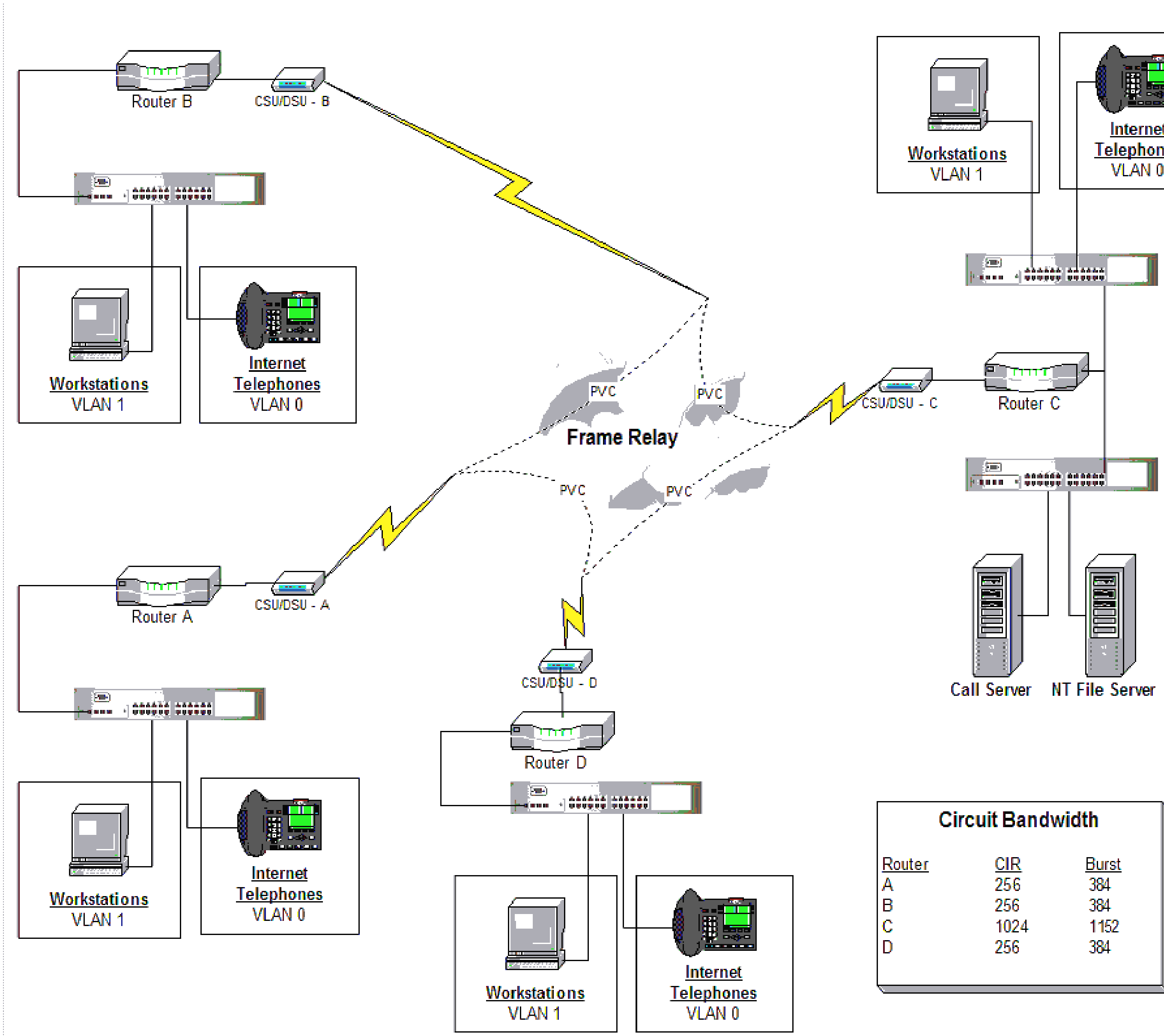
Which QoS method/traffic flow change will resolve the VoIP network issue?
A. Enable packet fragmentation so that serialization delay is NOT an issue.
B. Institute PPP Congestion Control Protocol (PCCP) to eliminate link congestion.
C. Implement Equal Cost Multi-Path (ECMP) routing so traffic is load balanced evenly.
D. Change routing policies so VoIP and data traffic take separate Point-to-Point Protocol (PPP) links.
Answer: D

9. Given the following network information:
- A customer has a 15-site VoIP network connected via Frame Relay.
- VLANs have been implemented at each site to separate VoIP traffic from data traffic.
- Policy switches have been installed to prioritize traffic.

Which QoS method will allow prioritization of information to be preserved from end-to-end?
A. Differentiated Services (DiffServ)
B. Resource Reservation Protocol (RSVP)
C. Layer 2 over Frame Relay Encapsulation
D. Multicast Open Shortest Path First (OSPF) Routing (MOSPF)
Answer: A

10. A customer wants to implement VoIP over a 56 kbps Point-to-Point Protocol (PPP) dial-up link using the G.729 CODEC. Which fragment size (Maximum Transmittable Unit [MTU]) should you recommend to the customer to ensure the lowest serialization delay?
A. Set the MTU to equal 70 bytes. This will fragment all packets that exceed the specified size.
B. Set the MTU to equal 80 bytes. This will allow interleaving of voice and data packets.
C. Set the MTU to equal 160 bytes. This ensures that only the data traffic will be fragmented.
D. Set the MTU to equal 320 bytes. This ensures that the VoIP traffic does NOT get fragmented.
Answer: A

11. Click the exhibit button.



A customer with the Frame Relay (FR) network depicted in the exhibit is having issues with serialization delay when interleaving data and VoIP traffic. They have decided to implement fragmentation.
Which fragmentation implementation provides the least amount of delay and the most efficient transport of fragments?
A. fragmentation on Router C
B. fragmentation on Routers A, B, and D
C. fragmentation on all Virtual Circuits (VCs)
D. end-to-end fragmentation on all Permanent Virtual Circuits (PVCs)
Answer: B

12. A customer is planning to implement VoIP on their existing data (WAN) network. The goal is to give priority to VoIP traffic and to reduce voice delay and jitter by implementing a Layer 2 packet fragmentation method. The routers on the network support the DLL fragmentation protocol. The chosen fragmentation method must enable higher-priority VoIP packets to be transmitted ahead of the lower-priority data packets and NOT be limited to a specific link layer technology. Which fragmentation method will give the best results for this application?
A. IP fragmentation
B. ATM fragmentation
C. PPP fragmentation
D. FRF.12 fragmentation
Answer: C

13. Given the following customer needs and current network design:
• Priority Queuing (PQ) is being used over the WAN circuits.
• Differentiated Services (DiffServ) marking is used to prioritize the different traffic streams into general queues by changing the Type of Service (TOS) bits of the IP headers.
• VoIP is configured for the “high” queue File Transfer Protocol (FTP) traffic that is asserted on the “medium” queue and all other traffic is left in the default “low” queue.
• Although NOT an issue at this time, it is expected that VoIP and FTP traffic will use 90% of the available WAN circuit bandwidth.
• All other business data will be buffered or dropped when bandwidth is limited.
• Increasing the bandwidth of the WAN circuits to the remote offices is NOT financially feasible.
Which recommendation for this customer's network will improve QoS?
A. Leave PQ in place and continue to allow VoIP and FTP to dominate the WAN link.
B. Configure the WAN link end-point routers with a First-In First-Out (FIFO) queuing mechanism.
C. Implement Weighted Fair Queuing (WFQ) and remove the DiffServ marking function from the WAN line routers.
D. Implement Weighted Fair Queuing (WFQ) to better control the traffic flows to allow business traffic to have an opportunity to traverse the WAN circuits.
Answer: D

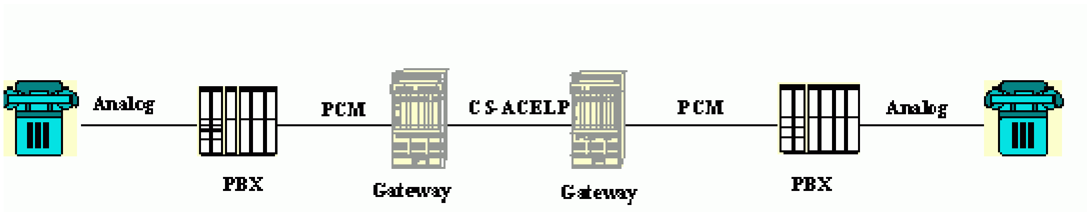
14. Given the following VoIP network information and customer voice quality requirements:
• Traffic across customer Frame Relay WAN links is being buffered by a First-In First-Out (FIFO) mechanism.
• The FIFO buffer is only used when traffic intermittently bursts over the available bandwidth of the WAN circuits.
• It is assumed that the FIFO buffering is NOT adversely affecting the VoIP traffic at this time.
• Network usage is expected to grow to twice its current level on all LAN and WAN circuits.
• All traffic has the Type of Service (TOS) bits set for priority of traffic.
• All traffic across the WAN links has the same TOS bit setting except File Transfer Protocol (FTP), which is given a lower TOS bit value.
In comparison to TOS, what are the advantages of implementing Differentiated Services (DiffServ) to improve the customer's voice quality?
A. DiffServ Weighted Fair Queuing (WFQ) replaces FIFO for improved handling of the available bandwidth.
B. DiffServ allows separation of the different types of traffic into respective queues by placing VoIP traffic in a higher queue value than all other types of traffic.
C. There would NOT be an advantage to implementing DiffServ in this scenario because the current FIFO settings are working fine with the current TOS bit settings.
D. There would NOT be an advantage to implementing DiffServ in this scenario because FIFO buffering can be re-configured to use a larger buffer for the expected increase in typical traffic.
Answer: A

15. Given the following customer network information:
• They have a high-speed campus network with Frame Relay WAN circuits to all remote office networks.
• All WAN circuits are 256 kbps with a Committed Information Rate (CIR) of 64 kbps.
• All calls across the WAN circuits are using the G.729 CODEC.
• Local campus network calls are using the G.711 CODEC.
• Currently the WAN routers are providing enough bandwidth to handle all business data traffic as well as VoIP traffic.
How can you improve the QoS of the VoIP enabled network?
A. Implement a First-In Last-Out (FILO) buffer on the WAN circuit to handle bursty traffic.
B. Implement Differentiated Service (DiffServ) marking and WFQ on the WAN circuit routers.
C. Implement DiffServ marking and First-In First-Out (FIFO) buffers on the WAN circuit routers.
D. Implement DiffServ and Weighted Fair Queuing (WFQ) to keep time-sensitive traffic moving.
Answer: B

16. A customer with a VoIP solution has two gateways located in the same location. There are two T1s/E1s connecting the VoIP system to the Public Switched Telephone Network (PSTN). Which three reasons support placing the two T1s/E1s in separate gateways? (Choose three.)
A. Traffic distributed through two gateways decreases processor load on each gateway.
B. Splitting the T1s/E1s between two gateways will decrease the potential down time on each T1/E1.
C. If one gateway fails, traffic can be routed through the other gateway removing one point of failure.
D. If the network connection is lost to one gateway, the traffic will still have access to the PSTN through the other gateway.
Answer: ACD

17. A customer with a VoIP system has the voice mail ports and public T1s/E1s trunks in one gateway. What is the most significant issue that the VoIP system could encounter with this gateway/port design?
A. This design does NOT have any potential issues.
B. The gateway could experience a bottleneck at the network connection into the gateway.
C. Congestion could occur between those trying to access voice mail boxes and those calling out to the Public Switched Telephone Network (PSTN).
D. The gateway is a single point of failure, which could cause the VoIP system to lose communication with the Public Switched Telephone Network (PSTN).
Answer: D

18. Click the exhibit button.



In the exhibit, which components are involved in transcoding?
A. Passport to Passport
B. all components shown
C. PBX and both Passports
D. analog telephone, PBX and connecting Passport
Answer: A

19. You have been tasked with improving voice quality on a customer's VoIP network. One of the main issues has been voice delay caused by transcoding. Presently VoIP calls travel through the company's LAN (1) via the G.711 CODEC to the company's WAN via the G.729(a) CODEC and then finally to another company LAN (2) using the G.723.1 CODEC. Which CODEC scheme willl reduce the negative effects of transcoding and improve the voice quality without increasing the required overall per call bandwidth?
A. LAN (1) G.711 through WAN to LAN (2) G.711
B. LAN (1) G.729 (a) through WAN to LAN (2) G.729 (a)
C. LAN (1) G.723.1 through WAN via G.711 to LAN (2) G.723.1
D. LAN (1) G.729 (a) through WAN via G.723.1 to LAN (2) G.723.1
Answer: B

20. Click the exhibit button.

Bandwidth per Voice Calls with Standard IP Header						
Codec	G.711			G.729		
Codec Bit Rate	64kbps			8kbps		
Voice Sample (ms)	10	20	30	10	20	30
IP Payload size (bytes)	80	160	240	10	20	30
IP Packet size (40 byte header)	120	200	280	50	60	70
Ethernet						
Ethernet bytes (per packet)	150	230	310	80	90	100
Ethernet bandwidth per voice flow (kbps)	130	96.8	85.9	73.6	40.8	29.9
Number of Voice Calls, Assuming 50% Link Utilization for Voice Traffic						
10 Mbps	38	51	58	67	122	167
100Mbps	385	516	582	679	1225	1674
1 Gbps	3858	5165	5823	6793	12254	16742
Frame Relay						
Frame Relay bytes (per packet)	124	204	284	54	64	74
Frame Relay bandwidth per voice flow (kbps)	100	82.0	76.0	44.0	26.0	20.0

Given the following customer network information:
• A customer has five locations connected to a Frame Relay (FR) cloud.
• Site A is the headquarter's location and it has a voice mail system.
• Sites B, C, D, and E will use the voice mail system from site A.
• All sites are connected via VoIP gateways.
• The number of concurrent calls that will come to the voice mail system are as follows:
• Site A = 12
• Site B = 2
• Site C = 3
• Site D = 2
• Site E = 4
Given the following network specifications, what is the minimum required bandwidth for the FR connection coming into Site E?
• CODEC = G.711
• Packetization = 20 milliseconds
• Voice Activity Detection (VAD) = Off
• Header compression is NOT being used.
A. 49.6 kbps
B. 164.2 kbps
C. 273.6 kbps
D. 329.6 kbps
Answer: D