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Exam : **920-804**

Title : Nortel technology standards and protocols for converged networks

Version : DEMO

1. A customer wants to implement VoIP network in their LAN/WAN environment. Following the network assessment, you have to make the appropriate recommendations to address network issues that can cause problems for real-time VoIP traffic. Based on the following, what should be your first recommendation?

A. Use media shared hubs for LAN connection.

B. Change the CODEC selection for LAN devices only.

C. Put call servers, signaling servers and media gateways on one VLAN.

D. Configure all interfaces to eliminate duplex mismatch and set to auto negotiate.

Answer: D

2. You are using Net IQ Chariot on a customer's LAN to complete QoS testing with pre-defined data flow emulation templates. You want to check that the current DiffServ settings will achieve the desired results. Which parameters will provide you with the required information? (Choose two.)

A. transaction rate

B. network throughput

C. precedence settings

D. resource reservation

Answer: AB

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. Given the following network assessment information: ?At all times of the day there is sufficient bandwidth between all test points sitting on the LAN drops as well as the remote offices across the WAN circuits where Internet Telephones will be placed. ?All the call servers, media gateways and signaling servers are on the same VLAN along with the data network traffic. ?All of the servers are duplicated on the network to ensure five nines (99.999%) of reliability and call handling. ?There is only one physical/logical path to each LAN drop and remote office. ?The architecture of the network is in a physical star topology with all VoIP processing servers at the center of the star, including the duplicate servers. ?Each of the remote sites were built from the ground up for VoIP and data traffic with NO regard for any call processing locally. All calls must go back to the campus network. Which three recommendations should you make to the customer concerning their network to enable it to support VoIP traffic? (Choose three.)

A. Add data path redundancy to give the network survivability.

B. Ensure that a backup generator sustains the power of all the network equipment.

C. Recommend a re-evaluation to obtain network delay and jitter information to ensure that these properties are within the appropriate bounds.

D. Segment the call server, media gateway, and signaling server to appropriate VLANs to ensure signaling traffic is NOT impacted by standard IP data.

Answer: ACD

5. Which statement about the requirements of VoIP network assessments is true?

A. Physical diagrams define the network operating systems that will be tested.

B. Calling patterns define the current peak-period usages of the data networks that will be tested.

C. Physical diagrams define the network-interface speeds, modes, and protocols that will be tested.

D. Calling patterns help determine the potential dates for shutting down the network to perform the testing.

Answer: C

6. A company wants its network to be configured and if necessary upgraded to enable it to handle the addition of VoIP traffic along with other IP data. During the network assessment you identify: ?The core of the network is a high-speed data center where there is ample bandwidth to support the addition of VoIP

traffic. ?The remote offices that the data center services are all connected via Frame Relay (FR). After setting up several NetIQ Chariot VoIP endpoints at remote - office locations, you identify the following issues: ?Excessive delays during peak hours, as much as 200 milliseconds on round trips. ?There are dropped packets even at low usage rates. ?The jitter across the FR links is excessive when it is greater than 30 milliseconds. Which two changes should occur before the customer's network is declared VoIP - ready? (Choose two.)

A. Upgrade the data center to Layer 3 switching.

B. Modify the 802.1p values on the VLANs to reduce the dropped packets to the remote sites.

C. Increase the Committed Information Rate (CIR) to guarantee that enough bandwidth is available to the remote sites.

D. Implement traffic prioritization for shaping and policing on the routers (data center and remote offices) to ensure that VoIP traffic is prioritized over non-time-sensitive traffic.

Answer: CD

7. A customer is planning a new VoIP network based on their existing PBX and corporate network. They need historical information about calling patterns in order to plan for VoIP traffic. What is the best source of this type historical information?

A. Operational Measurements from switch

B. Error messages from all networking devices

C. Alarm indications from all networking devices

D. Log reports from switch and all networking devices

Answer: A

8. Given the following network information: ?A company with a mixed VoIP and data network is experiencing bad voice quality with VoIP calls during normal business hours. ?Tests within their LAN segments warn of excessive jitter, delay, and in some cases packet retransmissions. ?There is a pair of switches that all traffic converges upon. The link between the two switches is a 100/Full Fast Ethernet connection that is operating at near maximum capacity for the majority of the day. ?At any given time, the estimated traffic on the link between the two switches is 200 Mbps. Which two configuration changes will improve the voice quality of the VoIP calls and prevent possible VoIP issues in the future between these

two switches? (Choose two.)

A. Place a router between the two switches and configure the 100 Mbps links to the two switches with a higher QoS for VoIP traffic.

B. Prioritize the traffic on the switches to give VoIP improved QoS. Prioritize only the traffic on the switches to give VoIP calls improved QoS.

C. Provide a larger aggregated bandwidth of 300 Mbps by trunking three interfaces together on the two switches using Multilink Trunks (MLTs).

D. Purchase two new Layer 2 switches that are capable of gigabit Ethernet to replace the Fast Ethernet connection with a single gigabit Ethernet connection.

Answer: BC

9. A company requests that their VoIP network be assessed to determine if some changes can be made to alleviate bandwidth limitations. Given the following network information: ?The CODEC is G.711. ?Voice Activity Detection (VAD) is NOT being used. ?The company wants to keep the bandwidth usage on its Frame Relay (T1/E1 and T3/E3) circuits less than 50% of the overall bandwidth. ?The company will allow the quality of the intelligible voice to degrade slightly in favor of a significant increase in total number of calls possible on their circuits. ?They will NOT accept additional packetization delay since their network is already at the maximum limit of acceptable delay (maximum delay budget) between the Internet Telephones and the call server. Which configuration change will increase the number of VoIP calls possible on the company's Frame Relay circuits?

A. Increase the packetization rate from 20 milliseconds to 30 milliseconds and decrease the jitter buffer.

B. Increase the packetization rate from 20 milliseconds to 30 milliseconds to gain approximately 20% more calls.

C. Change the CODEC to G.729 with the same packetization factor of 20 milliseconds to more than double the possible calls on the Frame Relay circuits.

D. There is NO way to increase the number of calls possible within the bandwidth restrictions unless they are willing to re-engineer their network with a larger-bandwidth circuits.

Answer: C

10. Click the exhibit button. An organization with a newly deployed VoIP network is complaining of poor

VoIP call quality. Given the following issues that have been reported by the company management: ?VoIP call quality on the first four Internet Telephones is good during non-peak hours of business. ?VoIP call quality degrades during the day around 10:00 a.m. and 2:00 p.m. ?VoIP call quality degrades in the evening hours when more than six Internet Telephones are making calls at the same time. The organization has sent you a protocol decode of a Real Time Protocol Control Protocol (RTPCP) packet (see the exhibit). It shows some information about the quality of the VoIP calls during the peak hours. Which two conclusions should you reach from analyzing this datagram? (Choose two.)

```
RTCP: ----- RTP Control Protocol -----
  RTCP:
  RTCP: Ver, Pad, RC:          = 81
  RTCP:          10.. ....     = Version = 2 (RFC 1889)
  RTCP:          ..0. ....     = Padding = 0
  RTCP:          ...0 0001     = Reception report count = 1
  RTCP: Packet type            = 200 (Sender Report)
  RTCP: Length                 = 13 (32-bit words)
  RTCP: SSRC of sender         = 205291315
  RTCP:
  RTCP: NTP reference timestamp    = 271480.77500 sec
  RTCP: RTP timestamp              = 2171846080
  RTCP: Sender's packet count      = 727
  RTCP: Sender's octet count       = 116320
  RTCP:
  RTCP: SSRC                           = 26415
  RTCP: Fraction lost                  = 0.00000
  RTCP: Cumulative packets lost        = 16777070
  RTCP: Extended highest sequence #    = Cycle:0, Seq:26926
  RTCP: Interarrival jitter            = 234
  RTCP: Last SR timestamp              = 3495568576
  RTCP: Delay since last SR            = 0 (Sec)
  RTCP:
```

A. This capture is inconclusive and a long-term detailed capture is required.

B. The indication of dropped packets is NOT an issue because the fraction-lost value is zero.

C. Cumulative packets-lost value indicates that the network is dropping VoIP data resulting in lost audio information.

D. The interarrival-jitter value indicates network issues and the likelihood of delayed audio between Internet Telephones.

E. The interarrival-jitter value is within typical VoIP functional ranges and does NOT impact Internet Telephone performance.

Answer: CD