Exam Report: 9.2.5 Practice Questions

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Overall Performance

Your Score: 54%

Passing Score: 80%

View results by: ○ Objective Analysis ● Individual Responses

Individual Responses

Question 1: Correct

Which VoIP device helps establish the connection between two VoIP phones?

- ○ VoIP gateway
- ○ VoIP endpoint
- ○ VoIP codec
- ➡ ◉ VoIP server

Explanation

A VoIP server helps establish the connection between two VoIP phones. Once the connection is established, the two phones communicate directly with each other.

A VoIP gateway converts voice calls between the PSTN and an IP network. A VoIP endpoint is a client of a VoIP server, such as a VoIP phone. A VoIP codec is not a device, but a special algorithm that compresses VoIP data to reduce bandwidth consumption.

References

LabSim for Network Pro, Section 9.2.
[netpro18v5_all_questions_en.exm VOIP_FACTS_01]

Question 2: Incorrect

What are other names for a VoIP server? (Select two.)

- ➡ ☐ VoIP PBX
- ☐ Jitter
- ☐ QoS
- ☑ ~~Hard phone~~
- ➡ ☑ IP-PBX

Explanation

A VoIP server is also known as a VoIP PBX or an IP-PBX since a VoIP server provides many of the functions of a traditional phone system PBX.

Quality of service (QoS) can be configured on network devices to give priority to VoIP traffic. Jitter can cause unusual sound effects in a VoIP call. A hard phone is a VoIP endpoint that is really a computer built to look like and work like a phone.

References

LabSim for Network Pro, Section 9.2.
[netpro18v5_all_questions_en.exm VOIP_FACTS_02]

Question 3: Incorrect

What is one benefit of placing VoIP gateways in geographically separated branch offices that have an existing WAN connection?

- ~~Data can be transported between VoIP gateways over the PSTN instead of via the expensive WAN.~~
- ➡ Long-distance PSTN charges can be reduced by switching VoIP calls to the PSTN in locations where only local call charges would be incurred.
- VoIP gateways can aggregate multiple VoIP calls for more efficient transmission to the home office.
- Less costly VoIP gateways can be used in place of expensive routers and modems to provide WAN connectivity.

Explanation

VoIP gateways convert voice and fax calls between the PSTN and an IP network. A WAN connection can carry VoIP calls from a distant location to a geographically separated branch office. A VoIP gateway located at the branch office can switch the call to the PSTN, where only local phone charges would be incurred.

References

LabSim for Network Pro, Section 9.2.
[netpro18v5_all_questions_en.exm VOIP_FACTS_04]

Question 4: Incorrect

When would you consider changing the codec used in your VoIP system? (Select two.)

- ➡ [x] When VoIP data consumes too large a portion of your network bandwidth.
- [ ] When excessive jitter causes unusual sound effects in VoIP calls.
- [ ] When network latency causes callers to talk over each other.
- [x] ~~When an open source VoIP protocol requires a different codec.~~
- ➡ [ ] When sound quality is poor.

Explanation

A special algorithm called a codec compresses VoIP data to reduce bandwidth consumption. A codec determines the sound quality of the VoIP call and the amount of bandwidth that it will require.

A more efficient codec can better compress VoIP data so that it will not consume as much bandwidth. Often, better compression will reduce the sound quality of VoIP calls. Conversely, a codec can preserve sound quality at the cost of using more bandwidth. A good codec can preserve sound quality and reduce bandwidth consumption.

References

LabSim for Network Pro, Section 9.2.
[netpro18v5_all_questions_en.exm MCM7]

Question 5: Incorrect

How can QoS be configured so that large data transfers will not block VoIP calls by using too much network bandwidth?

- ~~QoS can be configured on network devices to set a bandwidth threshold on selected ports.~~
- QoS can be configured on network devices to only allow network protocols that throttle network bandwidth usage.
- ➡ QoS can be configured on network devices to give priority to VoIP traffic.
- QoS can be configured on network devices to limit the size of a file that can be transferred on the network.

Explanation

Network devices can examine the type of service or precedence bits in the header of an IP packet to determine the type of traffic. QoS settings can be configured on a network devices to give VoIP traffic priority over normal computer traffic.

References

LabSim for Network Pro, Section 9.2.
[netpro18v5_all_questions_en.exm MCS7]

Question 6: Incorrect

Upper management has asked you if there is a way to integrate phone calls, emails, and instant messaging into a single platform.

Which of the following systems should you recommend?

- ~~Voice over IP~~
- ➡ Unified communication
- PSTN
- Quality of service

Explanation

Unified communications (UC) integrates multiple types of communications into a single system. UC systems can integrate the following real-time communications:

- Voice calls
- Audio conferencing
- Video conferencing (VTC)
- Desktop sharing
- Instant messaging

UC systems can also provide non-real-time communication integration, including:

- Texting
- Voicemail
- Email
- Faxing

Voice over IP only provides voice calling integration with an IP network. Quality of Service is used to ensure that voice data is given higher priority on the network. The PSTN is the traditional method used for phone calls.

References

LabSim for Network Pro, Section 9.2.

[netpro18v5_all_questions_en.exm *NP15_VOICE_OVER_IP_02]

Question 7: Correct

Which of the following protocols is an open source protocol used by most manufacturers of VoIP systems?

- User datagram protocol (UDP)
- Stream control transmission protocol (SCTP)
- Transmission control protocol (TCP)
- ➡ Session initiation protocol (SIP)

Explanation

The session initiation protocol (SIP) is one of the protocols used during the call control process of multimedia communications, such as a VoIP call. SIP is used to set up, maintain, and tear down multimedia communications.

SIP rides on top of the TCP, UDP, and SCTP transport layer protocols.

References

LabSim for Network Pro, Section 9.2.
[netpro18v5_all_questions_en.exm *NP15_VOIP_FACTS_01]

Question 8: Correct

Which of the following protocols is used by VoIP to set up, maintain, and terminate a phone call?

- SSH
- NTP
- TLS
- ➡ SIP
- RTP

Explanation

The session initiation protocol (SIP) is used to set up, maintain, tear down, and redirect the call. The real-time transport protocol (RTP) contains the actual voice data.

SSH is used for secure remote administration of a network device. TLS is used to add security to other protocols. NTP is used for synchronizing clocks on network devices.

References

LabSim for Network Pro, Section 9.2.
[netpro18v5_all_questions_en.exm NP09_1-1 #8]

Question 9: Correct

Your company uses VoIP for phone calls. Recently, employees have been complaining about phone calls with unusual sound effects.

Which type of problem is occurring on the VoIP system?

- Echo
- ➡ Jitter
- Latency

Packet loss

Explanation

Because VoIP transmits call data using IP packets over a packet-switched network, VoIP is susceptible to the following problems:

- Latency occurs when data takes a long time to arrive at the receiving device. Delays cause long pauses between speaking and receiving and can result in callers continually interrupting each other.
- Jitter is a variation in the delay of individual packets. Jitter causes strange sound effects as the delay of packets fluctuates.
- Packet loss occurs when packets do not arrive at all. Packet loss causes drop-outs in the conversation.
- Echo occurs when you hear your own voice in the telephone receiver while you are talking. Excessive delay can cause unacceptable levels of echo.

References

LabSim for Network Pro, Section 9.2.
[netpro18v5_all_questions_en.exm *NP15_VOICE_OVER_IP_04]

Question 10: Correct

You are on a phone call using VoIP. You notice that it takes several seconds for the person on the other end to respond to questions you ask.

Which type of problem is occurring?

- Echo
- ➡ Latency
- Packet loss
- Jitter

Explanation

Because VoIP transmits call data using IP packets over a packet-switched network, VoIP is susceptible to the following problems:

- Delay (or latency), which occurs when data takes a long time to arrive at the receiving device. Delays cause long pauses between speaking and receiving and can result in callers continually interrupting each other. International standards call for a delay of 150 milliseconds or less.
- Jitter, which is a variation in the delay of individual packets. Jitter causes strange sound effects as the delay of packets fluctuates.
- Packet loss, which occurs when packets do not arrive at all. Packet loss causes dropouts in the conversation.
- Echo, which occurs when you hear your own voice in the telephone receiver while you are talking. Excessive delay can cause unacceptable levels of echo.

References

LabSim for Network Pro, Section 9.2.
[netpro18v5_all_questions_en.exm *NP15_VOICE_OVER_IP_06]

Question 11: Correct

What is a soft phone?

- A traditional or VoIP phone that has a padded handset that is more comfortable for a user.
- ➡ A software application that runs on a computer or other device that accesses a VoIP server to make real-time phone calls.

A device that converts fax calls and other soft calls between the PSTN and an IP network.

A software algorithm that compresses VoIP data prior to transmission on an IP network.

Explanation

A soft phone is a software application that is installed on a computing device such as a computer or a handheld device.

References

LabSim for Network Pro, Section 9.2.
[netpro18v5_all_questions_en.exm VOIP_FACTS_03]

Question 12: Incorrect

Which switch features are typically used with VoIP? (Select two.)

- ~~Spanning tree~~
- VLAN
- Mirroring
- PoE

Explanation

When configuring Voice over IP (VoIP), switches with Power over Ethernet (PoE) capabilities provide power to the VoIP phone through an Ethernet cable, the same cable that is used for transmitting data signals. Virtual LANs (VLANs) are often used to distinguish voice traffic from data traffic so that Quality of Service (QoS) measures can be applied to traffic that is part of the voice VLAN.

Bonding allows multiple switch ports to be used at the same time to reach a specific destination. Spanning tree is a protocol on a switch that allows the switch to maintain multiple paths between switches within a subnet. The spanning tree protocol runs on each switch and is used to select a single path between any two switches. Mirroring sends traffic from all switch ports to a switch port you designate as the mirrored port.

References

LabSim for Network Pro, Section 9.2.
[netpro18v5_all_questions_en.exm NP09_3-3 #5]

Question 13: Correct

Which of the following features is used with digital IP phones to supply power through a switch port?

- Spanning tree
- 802.1x
- PoE
- Trunking
- VPN

Explanation

Power over Ethernet (PoE) supplies power to end devices through the RJ45 Ethernet switch port. Power to the phone is carried on unused wires within the drop cables.

Spanning tree is a protocol on a switch that allows the switch to maintain multiple paths between switches within a subnet. The spanning tree protocol runs on each switch and is used to select a single path between any two switches. Trunking allows a switch to forward VLAN traffic between switches. 802.1x is an authentication protocol used with port security or port authentication.

References

LabSim for Network Pro, Section 9.2.
[netpro18v5_all_questions_en.exm NP09_3-3 #1]