Exam : 642-432

Title : Cisco Voice Over IP

Ver : 08.28.07

QUESTION 1:

Which type of signaling is DTMF?

A. Supervisory
B. Route
C. Informational
D. Address

Answer: D

Explanation:
Start dial supervision is the line protocol that defines how the equipment seizes the E&M trunk and passes the address signaling information such as dual tone multifrequency (DTMF) digits.
Address Signaling
Address signaling typically represents the digits dialed (called party's number). There are two options used to pass address information. Either Pulse dial (rotary dialing) or Tone dial (DTMF) can be used. The default for Cisco routers and gateways is DTMF.

QUESTION 2:

You have set up a complex dial plan using translation rules. The following translation rule has been configured.
What output would correspond to the test translation-rule command?
translation-rule 1
rule 0 ^0.. 215550210
rule 1 ^1.. 215550211
rule 2 ^2.. 215550212
rule 3 ^3.. 215550213
rule 4 ^4.. 215550214
rule 5 ^5.. 215550215
rule 6 ^6.. 215550216
rule 7 ^7.. 215550217
rule 8 ^8.. 215550218
rule 9 ^9.. 215550210

A. test translation-rule 1512
The replaced number: 21555021512
B. test translation-rule 1555
The replaced number: 55521555021
C. test translation-rule 1617
The replaced number: 61721555021
D. test translation-rule 1910
The replaced number: 21555021910

Answer: A

Explanation:
New Delhi(2-digit indial range)
!--- Only relevant "IOS translation rule" output is presented
!
translation-rule 1
!-- The "1" above is the tag for the set.
rule 0 ^0. 1011000
rule 1 ^1. 1011001
rule 2 ^2. 1011002
rule 3 ^3. 1011003
rule 4 ^4. 1011004
rule 5 ^5. 1011005
rule 6 ^6. 1011006
rule 7 ^7. 1011007
rule 8 ^8. 1011008
rule 9 ^9. 1011009
!
!-- These rules replace the first digit of a 2-digit number with the corresponding
!-- translation. The router looks for a 2-digit number starting with a leading [0-9].
!-- The caret, "^" ensures the match only happens at the start of the digit string
!-- rather than any occurrence in a digit string. This ensures the router makes the
!-- translation only for the leading digits. By default, if an explicit match is made
!-- on a digit (in this case the first digit) the router replaces it with the new
!-- digits. Therefore, to keep the original numbering, the matched digit needs to be
!-- replaced with the same digit at the end of the modified string. Once the call
!-- comes in, the called number prepended with 101100 followed by the
!-- original 2 digits.
!
voice-port 1/0:1
translate called 1
cptone IN
compand-type a-law
!
!-- The translation rule is applied to the voice port where the
!-- call comes in to the router. When a call comes in from the
!-- telephone network towards the router, the called number
!-- is translated before it is matched on any dial peers.
!
dial-peer voice 100 voip
destination-pattern 101100..
session target ipv4:main site IP address
ip precedence 5
dtmf-relay h245-alphanumeric

!
!-- The VoIP dial peer needs to be configured to match on the new numbering plan
This output was captured from the NewDelhi router which shows the translation rules applied while dialing from the NewDelhi site.
NewDelhi- Output
!-- It is possible to confirm the translation rules are working:!!NewDelhi#test translation-rule 1 99!-- Original called number is "99"The replaced number: 10110099!-- Translated to 8 digits

---

QUESTION 3:

What is the optimal end-to-end delay that should be achieved in a VoIP network?

A. 20 ms
B. 100 ms
C. 150 ms
D. 400 ms

Answer: C

Explanation:
Delay Specifications

Range in Milliseconds	Description
0-150	**Acceptable for most user applications.**
150-400	Acceptable provided that administrators are aware of the transmission time and it's impact on the transmission quality of user applications.
Above 400	Unacceptable for general network planning purposes, however, it is recognized that in some exceptional cases this limit will be exceeded.

QUESTION 4:

Which three are supervisory signals? (Choose three)

A. busy
B. on hook
C. off hook
D. call waiting

E. ring

Answer: B, C, E

Explanation:
1. Supervisory Signalling - electrical voltages and tones that can be heard are used to signify call status as follows:
2. 1. On-hook - produces an open circuit which does not allow any signalling, only the ringer can operate.
2. Off-hook - lifting the handset closes the circuit and allows the telephone switch to send an audible dial tone to the receiver.
3. Ringing - the switch sends a ringing voltage to the destination telephone as notification of an incoming call. Also an audible ringing tone is sent to the caller telephone to indicate that the call is progressing. This tone takes the form of a pattern called Cadence In Europe this Cadence takes the form of a double ring (duration of 0.4s separated by 0.2s) followed by two seconds of silence, whereas in the US it takes the form of two seconds of ring followed by four seconds of silence.

QUESTION 5:

What is the E.164 numbering plan?

A. A proprietary PBX number plan.
B. The IETF North American number plan.
C. The European PBX standard telephony number plan.
D. The ITU worldwide number plan.

Answer: D

Explanation:
Numbering Scheme
The standard PSTN is a large, circuit-switched network. It uses a specific numbering scheme, which complies with theITU-T international public telecommunications numbering plan (E.164) recommendations. For example, in North America, the North American Numbering Plan (NANP) is used, which consists of an area code, an office code, and a station code. Area codes are assigned geographically, office codes are assigned to specific switches, and station codes identify a specific port on that switch. The format in
North America is 1Nxx-Nxx-xxxx, with N = digits 2 through 9 and x = digits 0 through 9. Internationally, each country is assigned a one- to three-digit country code; the country's dialing plan follows the country code. In Cisco's voice implementations, numbering schemes are configured using the destination-pattern command.
E.164 is an ITU-T recommendation which defines the international public telecommunication numbering plan used in the PSTN and some other data networks.

QUESTION 6:

On the MOS scale, what does a 5 represent?

A. poor
B. fair
C. average
D. extra medium
E. excellent

Answer: E

QUESTION 7:

Which of the following best describes the main difference between G.729 and G.729a?

A. G.729 has higher complexity.
B. G.729 requires a higher bit rate.
C. G.729a has built-in echo cancellation.
D. G.729a has improved speech performance.
E. G.729a is designed for use with 3DES encryption

Answer: A

QUESTION 8:

What type of signalling is used for a circuit transmitted within the same channel as the voice?

A. PCM
B. SS7 Signalling
C. In-line Signalling
D. Common Channel Signalling
E. Channel Associated Signalling

Answer: E

QUESTION 9:

You are the Voice technician at Certkiller .com. The Certkiller network uses VoIP.
Your newly appointed Certkiller trainee wants to knowwhat the disadvantage of using VoIP rather than VoFR or VoATM are.
What will your reply be?

A. Data can arrive out of sequence.

B. Networks are complicated to design.
C. Data units can arrive out of sequence.
D. Network failures are not automatically found.

Answer: C

QUESTION 10:

You are the network engineer at Certkiller .com. You have configured real-time call control processing on the Certkiller VoIP network. You want to verify this configuration.
What command should you use?

A. debug voip rtcp
B. debug call control
C. debug voip ccapi inout
D. debug voip call control
E. debug voice call control

Answer: C

QUESTION 11:

How do a-law and mu-law reduce quantization error?

A. Use smaller step functions at lower amplitudes.
B. Use smaller step functions at higher frequencies.
C. Increase code points at lower amplitudes.
D. Increase code points at higher frequencies.

Answer: A

QUESTION 12:

You have a pair of voice enabled routers that have the capability of supporting only medium complexity codecs. You need to conserve bandwidth on WAN links without major impact to call quality.
Which codecs will satisfy these requirements? (Choose two)

A. G.729
B. G.729A
C. G.729B
D. G.729.AB

Answer: B, D

QUESTION 13:

You are the network technician at Certkiller .com. Your newly appointed Certkiller trainee wants to know whatdoes the connection tie-line command emulates.
What will your reply be?

A. A temporary connection to a PBX.
B. A permanent connection to a PBX.
C. A temporary connection to the PSTN.
D. A permanent connection to the PSTN.
E. A permanent connection to the network.

Answer: A

QUESTION 14:

What is the most important piece to implement if you are considering a VoIP infrastructure?

A. QoS
B. Reinstallation of the PBX
C. A new Help Desk trained on Voice technologies
D. POTS installation documentation.

Answer: A

QUESTION 15:

Currently, unlike traditional phone service, IP telephone service is relatively unregulated by government.

A. True
B. False

Answer: A

QUESTION 16:

You are the network engineer at Certkiller .com. Certkiller has its headquarters in New York and branch offices in Delaware, Detroit and Denver. Certkiller has a traditional time-division multiplexing (TDM) based network with automated call distribution (ACD) private branch exchanges (PBX) and desktop systems. You want to add an IP telephony solution to the Certkiller network.
What will enable server and agent-level IP telephony to coexist with the existing network?

A. IPCC
B. XLD-t
C. PBX-R
D. CiscoWorks
E. Call Manager

Answer: A

QUESTION 17:

Calls between IP and PBX users can use all of the features provided by each system, and that subset is defined by the level of complexity of the voice interface between the IP network and the PBX.

A. Fake
B. True

Answer: A

QUESTION 18:

Which will provide your IP Phones with an IP Address?

A. The technician, it has to be statically assigned.
B. Any DHCP server
C. The Cisco CallManager DHCP snap-in with CiscoWorks.
D. IP Phones do not need an IP address.

Answer: B

QUESTION 19:

What explains how the Cisco IP SoftPhone uses the Cisco CallManager?

A. The IP SoftPhone does not work with the Cisco CallManager
B. Cisco IP SoftPhone uses the services of the Cisco CallManager to route calls through an IP telephony network.
C. Any IP SoftPhone plugs directly into the CallManager IPSP jack for onsite support
D. The IP SoftPhone will use CallManager to reset all voicemail on the PBX.

Answer: B

QUESTION 20:

Which organization approved the H.323 standard?

A. ITEF
B. IEEE
C. FEMA
D. Bell Atlantic
E. ITU

Answer: E

QUESTION 21:

When they are booted, the Cisco Access Digital Trunk Gateway DT-24+, the Cisco Access Digital Trunk Gateway DE-30+, and the Catalyst 6000 digital gateway are provisioned with Cisco CallManager location information. When these gateways initialize, a list of Cisco CallManager's, referred to as a ______________ is downloaded to the gateways.

A. Cisco IPSP group
B. Call managed Cisco redundancy group
C. IPNC redundancy group
D. Cisco CallManager redundancy group

Answer: D

QUESTION 22:

Which are three required steps in digitizing voice? (Choose three)

A. Companding the signal.
B. Quantizing the amplitude.
C. Filtering the signal.
D. Sampling the soundwave.
E. Encoding the results in binary form.

Answer: B, D, E
The answer on "Question 1" on page 15 dealing with digitizing voice should be:
1. Quantizing the amplitude
2. Sampling the soundwave
3. Encoding the results in binary form
See page 2-45 of CVoice version 4.1 class books.
Not C: "Filtering" is part of the process to go from digital to analog not analog to digital (p. 2-47)

QUESTION 23:

You are the VoIP engineer at Certkiller .com. A Certkiller user complains that she gets a busy tone instead of a dial tone when she tries to call another user. You want

to troubleshoot this problem.
What command should you use?

A. show voice dsp
B. show voice path
C. show voice connection
D. show voice port summary
E. show dial-peer voice summary

Answer: A

QUESTION 24:

Your manager asks you for a worksheet defining items that need to be addressed for the future VOIP and IP telephone rollout.
What items do you put on the worksheet that need to be addresses for the wiring closets? (Choose all that apply.)

A. Switches with Inline Power
B. A 7000 series router to backup the switch with HSRP
C. PBX failover
D. UPS systems and Backup power
E. Cooling Requirements (a heat profile)

Answer: A, D, E

QUESTION 25:

Which device listed below has an intelligent power management system that grants or denies power to various system components based on power availability in the system for use with IP telephony?

A. Cisco 7309 VXD Router
B. Cisco Works plug in
C. CallManager
D. Catalyst 6000 switch

Answer: D

QUESTION 26:

From the list below, what allows a Cisco IP phone to detect the absence of audio and therefore does not transmit packets over the network?

A. Ipng
B. PBX Filters

C. Voice Activation Detection
D. DHE
E. Call Waiting

Answer: C

QUESTION 27:

What Cisco Catalyst Switch command produces the following inline power output?
Defalut Inline Power allocation per port: 10.00 Watts (0.23
Amps @42V)
Port InlinePowered PowerAllocated
Admin Oper Detected mWatt mA @42V
---- ---- ---- ---- ----
7/1 auto off no 0 0
7/2 auto on yes 5040 120
7/3 auto faulty yes 12600 300
7/4 auto deny yes 0 0
7/5 off off no 0 0

A. show cam inlinepower <mod>|<mod/port>
B. show port inline <mod>|<mod/port>
C. show port inlinepower <mod>|<mod/port>
D. show port power <mod>|<mod/port>

Answer: C

QUESTION 28:

What are two constraints that you may encounter when trying to design a IP
Telephone infrastructure?

A. Upper level management acceptance
B. Budgetary Constraints
C. STP reliability
D. IP convergence

Answer: A, B

QUESTION 29:

What is the biggest issue affecting voice transport when you implement IPSec VPNs
in a converged network?

A. Hop count.
B. Using G.729 as the codec.

C. Throughput considerations.
D. Ensuring only software encryption is running.

Answer: C

QUESTION 30:

What factors must be considered in the overall design when implementing an IPSec VPN for transport of voice?

A. Port numbers and added delay.
B. Added delay and added overhead.
C. Port numbers and longer dial plan.
D. Port numbers and added overhead.
E. Added overhead and longer dial plan.

Answer: D

QUESTION 31:

When analyzing the WAN for IP Telephone deployment, you need to collect information from the WAN and LAN devices. You need to determine current Bandwidth usage before rolling out a solution. From the analysis you are performing, there are categories you can collect information from.
Select two from the list below.

A. Device information, which includes router models, memory, CPU, interface card modules versions and software versions
B. The serial numbers from the Meridian Phone System
C. The existing WAN topology, which includes logical design information and bandwidth subscription rates
D. LAN information as in the make and model of the Remote site closet Nortel equipment
E. You need all the CiscoWorks server configurations to make sure you can install Call Manager

Answer: A, C

QUESTION 32:

If you were troubleshooting No Ringback Tone on ISDN-VoIP (H.323) Calls and had problem that POTS (PSTN/PBX) user places a call (through Cisco router/gateways) and does not hear ringback tone before call is answerered, what would you do?

A. Use the conf inline powercommand because it is not set in the terminating router.

B. Reset all the phone connections on the IP SoftPhones
C. Configure CiscoWorks CallManager to handle all errors automatically.
D. Configure the Cisco IOS global configuration command voice call send-alert in the terminating router

Answer: D

QUESTION 33:

Which preference key word assigns top precedence to a dial peer in a hunt-group?

A. 0
B. priority
C. 1
D. high

Answer: A

QUESTION 34:

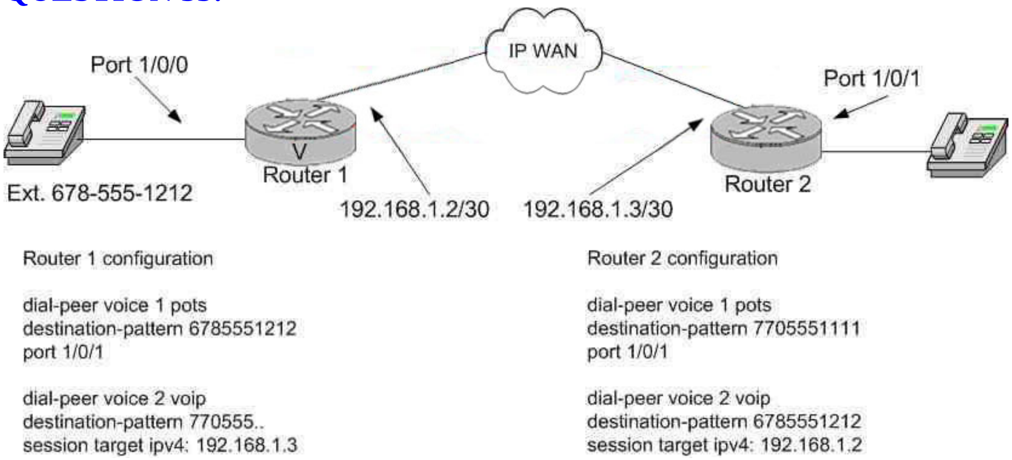
What are two basic parameters needed to setup a dial peer connected to the PSTN? (Choose two)

A. voice port
B. signaling type
C. interface bandwidth
D. destination pattern

Answer: A, D

Explanation:
Depending on the call leg, a call is routed using one of the two types of dial peers:
POTS-Dial peer that defines the characteristics of a traditional telephony network connection. POTS dial peers map a dialed string to a specific voice port on the local router, normally the voice port connecting the router to the local PSTN, PBX, or telephone.
Voice-network-Dial peer that defines the characteristics of a packet network connection.
Voice-network dial peers map a dialed string to a remote network device, such as the destination router that is connected to the remote telephony device.
The following examples show basic configurations for POTS and VoIP dial peers:
dial-peervoice 1 potsdestination-pattern555....port1/0:1dial-peervoice 2 voip destination-pattern 555....session target ipv4:192.168.1.1

QUESTION 35:



Users are complaining that they are unable to complete a call from 678-555-1212 to 770-555-1111 from Router 1 to Router 2.
Select the correct answer to resolve the problem.

A. Incorrect dial-peer statement in Router 1.
B. Incorrect port statement in Router 1 pots dial peer.
C. Incorrect session-target statement in Router 2.
D. Incorrect destination-pattern in Router 1.

Answer: B
Given the output the correct answer would be "Incorrect port assignment in router one". Voice port 1/0/1 does not exist, according to the drawing on router 1, Voice port 1/0/0 is the correct port.under the router 1 dial-peer the port assignment is port 1/0/1.
There is no problem with the destination-patterns (not D)

QUESTION 36:

Which dial plan characteristic is most obviously improved by dropping a number translation step?

A. Availability
B. Post-dial delay
C. Scalability
D. Hierarchical design

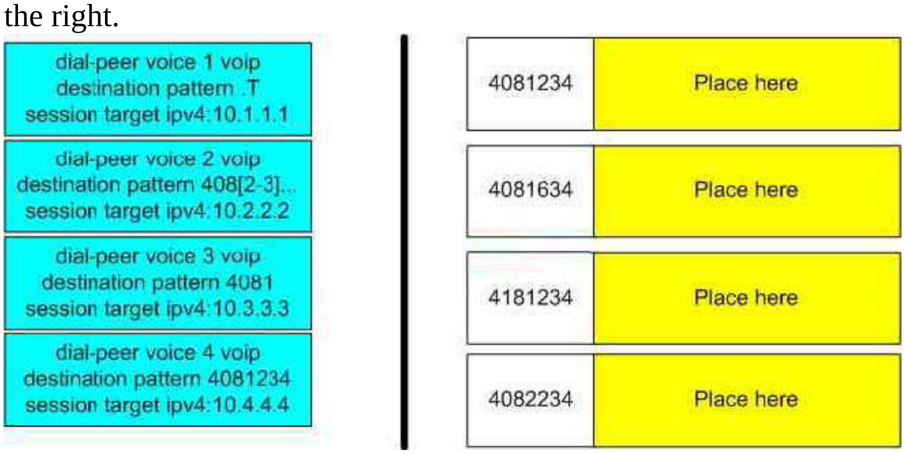
Answer: C

Explanation:
Introduction
This document provides a sample configuration for creating scalable dial plans for a VoIP network using IOS translation rules. As you install integrated voice and data networks, one issue frequently encountered is how to manage the numbering plans of the indial ranges at different locations. Depending on the type of exchange, signaling

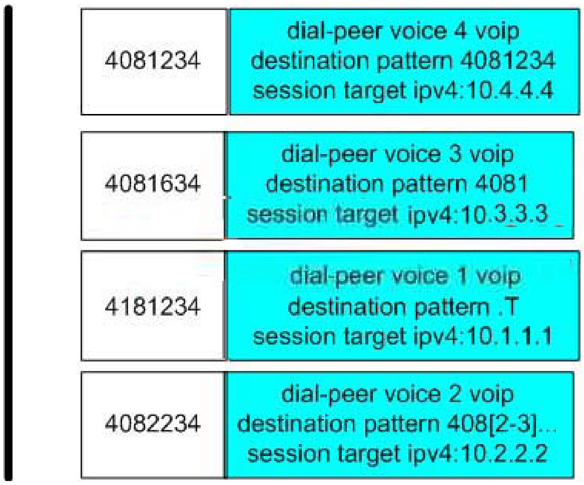
protocol standards and even location, the service provider could pass similar number ranges to the subscriber equipment at each remote site. If these calls are being routed back to a central site, there could be an overlap in the called numbers that originate from each of the remote sites. Since the PBX makes the routing decision based on unique called numbers, this could cause problems with automatic call distribution (ACD) queues on private branch exchange (PBX) systems . For example, calls from each site may need to be directed to particular operators who speak the local language from where the call originated. If the called numbers from each site overlap, there is not any way of identifying the origin of a call, therefore the PBX is not able to route the call to the correct ACD queue.
Some remote sites may be provided with a 2-digit indial number range while other sites may have 3- or 4-digit indial ranges, so the called numbers could be from [00 - 99] to [0000 - 9999]. With these number ranges, the main site router would need configurations to handle 2-, 3- and 4-digit numbering plans. This could add to the overall complexity of the router configuration.
The solution to these issues is to use IOS digit translation rules at each remote site to prepend digits to the number range that comes in from the telephone network. This then creates a standard numbering plan across the customer's network and allows new sites to be gradually added without major changes to the rest of the network.

QUESTION 37:

DRAG DROP
Drag each of the dial peers on the left to the phone number that it would match on the right.



Answer:



Explanation:
Destination pattern 4081234" would be an exact match for 4081234 and therefore would be the correct answer.
Destination pattern .T" would match 4181234 not 4081234 due to length of match. "
Note:
Destination Pattern
The destination pattern associates a dialed string with a specific telephony device. It is configured in a dial peer by using the destination-pattern command. If the dialed string matches the destination pattern, the call is routed according to the voice port in POTS dial peers, or the session target in voice-network dial peers. For outbound voice-network dial peers, the destination pattern may also determine the dialed digits that the router collects and then forwards to the remote telephony interface, such as a PBX, a telephone, or the PSTN. You must configure a destination pattern for each POTS and voice-network dial peer that you define on the router.
The destination pattern can be either a complete telephone number or a partial telephone number with wildcard digits, represented by a period (.) character. Each "." represents a wildcard for an individual digit that the originating router expects to match. For example, if the destination pattern for a dial peer is defined as "555....", then any dialed string beginning with 555, plus at least four additional digits, matches this dial peer.
In addition to the period (.), there are several other symbols that can be used as wildcard characters in the destination pattern. These symbols provide additional flexibility in implementing dial plans and decrease the need for multiple dial peers in configuring telephone number ranges.
Fixed- and Variable-Length Dial Plans
Fixed-length dialing plans, in which all the dial-peer destination patterns have a fixed length, are sufficient for most voice networks because the telephone number strings are of known lengths. Some voice networks, however, require variable-length dial plans, particularly for international calls, which use telephone numbers of different lengths.
If you enter the timeout T-indicator at the end of the destination pattern in an outbound voice-network dial peer, the router accepts a fixed-length dial string and then waits for additional dialed digits. The timeout character must be an uppercase T. The following dial-peer configuration shows how the T-indicator is set to allow variable-length dial strings:
dial-peervoice 1 voipdestination-pattern2222Tsessiontarget ipv4:10.10.1.1In the example

above, the router accepts the digits 2222, and then waits for an unspecified number of additional digits. The router can collect up to 31 additional digits, as long as the interdigit timeout has not expired. When the interdigit timeout expires, the router places the call. The default value for the interdigit timeout is 10 seconds. Unless the default value is changed, using the T-indicator adds 10 seconds to each call setup because the call is not attempted until the timer has expired (unless the # character is used as a terminator). You should therefore reduce the voice-port interdigit timeout value if you use variable-length dial plans. You can change the interdigit timeout by using the timeouts inter-digit voice-port command.

Table 11 Wildcard Symbols Used in Destination Patterns

Symbol	Description
.	Indicates a single-digit placeholder. For example, 555.... Matches any dialed string beginning with 555, plus at least four additional digits.
[ ]	Indicates a range of digits. A consecutive range is indicated with a hyphen (-); for example, [5-7]. A nonconsecutive range is indicated with a comma (,); for example, [5,8]. Hyphens and commas can be used in combination; for example, [5-7,9].
()	Indicates a pattern; for example, 408(555). It is used in conjunction with the symbol? %, or +.
?	Indicates that the preceding digit occurred zero or one time. Enter ctrl-v before entering? From your keyboard.
%	Indicates that the preceding digit occurred zero or more times. This functions the same as the "*" used in regular expression.
+	Indicates that the preceding digit occurred one or more times.
T	Indicates the interdigit timeout. The router pauses to collect additional dialed digits.

QUESTION 38:

When does an IP Phone receive the ring tones on the phone?

A. The phone downloads the wave file on boot.
B. The phone downloads based upon user selection.
C. The phone downloads the wave file on every request.
D. The phone downloads based on CallManager request.

Answer: A

QUESTION 39:

Which two tools are most appropriate for configuring 4,000 IP Phones prior to deploying the phones and allow phones to auto register? (Choose two.)?

A. ART
B. BAT

C. AST
D. TAPS

Answer: B, D

QUESTION 40:

Which field can be manually entered into the database when using the BAT tool, but not when using the TAPS tool??

A. Partiition
B. Directory number
C. Device MAC address
D. Calling Search Space

Answer: C

QUESTION 41:

Name two factors that will need to be considered and may provide a hurdle to move forward with your VOIP rollout and IP telephony implementation:

A. Money Flow issues.
B. Business Requirement
C. Technical Constraints
D. Budget constraints
E. Project Management Meetings

Answer: A, C

QUESTION 42:

When the Directories button on the 7960 phone is pressed, what does the 7960 use to retrieve the Dirctory information?

A. XML
B. SQL
C. LDAP
D. Skinny

Answer: A

QUESTION 43:

Which file does the BAT tool use to import users into the CallManager database?

A. CSV
B. Microsoft Word
C. Microsoft Excel
D. Tab-delimited text file

Answer: A

QUESTION 44:

You are the network technician at Certkiller .com. Your newly appointed Certkiller trainee wants to knowwhat the attributes of a scalable dialing plan are.
What will your reply be? (Choose four)

A. Logic distribution
B. Hierarchical design
C. Simplicity in provisioning
D. Reduction in pre-dial delay
E. reduction in post-dial delay

Answer: A, B, C, E

QUESTION 45:

What happens if no incoming dial peer matches a router or gateway?

A. The incoming call leg takes an alternate path.
B. The incoming call legmatches the default dial peer.
C. The incoming call leg sends a busy to the originator.
D. The incoming call leg is denied and the call is dropped.

Answer: B

QUESTION 46:

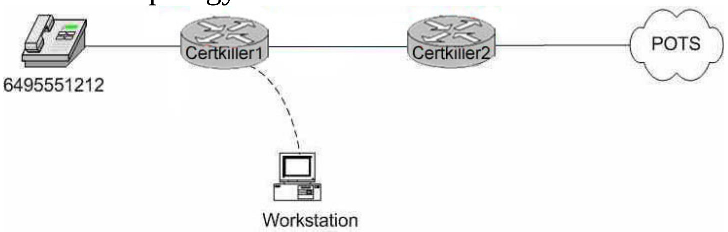
A 9 digit number must be dialed to reach numbers on the PSTN.
What process makes sure that the first 9 digit is not transmitted as part of the called number?

A. digit alternating
B. digit masking
C. digit manipulation
D. digit seizing

Answer: C

QUESTION 47:

SIMULATION
Network topology exhibit:



You're a CCNP certified employee at Deutsche Telekom. You are working with the Certkiller .com Telephone Company, in Rammstein Germany, to assist them to set up a simple configuration to demonstrate an FXS-to-PSTN connection across an IP network. The Ethernet interfaces and routing protocols are already configured on both routers. The Certkiller 1 router will act a s gateway to the PSTN to the PSTN and is correctly configured for this task.
You are required to add the voice portion to the Certkiller 2 router. Configure the pots and VoIP dial peers and insure that the pots telephone on the Certkiller 2 router can reach the PSTN connected to the Certkiller 1 router. The analog telephone is connected to voice port 1/0/0. The customer uses the access code 9 to dial out the PSTN. Insure that the Certkiller 1 and Certkiller 2 recognize the dialed digits as standard E.164 numbers. The telephone number of the pots telephone connected to the Certkiller 1 router is 49648455554321.
After the configuration is complete you may click on the telephone to check your configuration. If the call is successful, you will get a ringing message and a completed call. If the call is unsuccessful, you will get a busy signal.
The Certkiller 1 router has the following configured ports:
Ethernet: 0/0 172.16.1.11 255.255.255.0
To configure the router click on a host icon that is connected to a router by a serial console cable.

Answer:

Explanation:
en
config t
dial-peervoice 1 pots
destination-pattern 6495551212
port1/0/0
dial-peervoice 2 voip
destination-pattern +9T
sessiontarget ipv4:172.16.1.11

Explanation: Reference CVoice ver 4.1 class books page 4-21 and Cisco Voice over Frame Relay, ATM, and IP (from Cisco Press second printing May 2002) page 225. The "+" sign is optionaland is usedas the first digit to indicate an E.164 standard number. We

place the "+" in front if the 9 digit in the voip dial peer because the question states to make sure that Certkiller 1 and Certkiller 2 recognize the dialed digits as standard E.164 numbers. The pots phone will not be calling itself so therefore the logical placement of the "+" would be in front of the 9 digit in the voip dial peer.
Note 1: The simulation does not allow the use of the "register" command.
Note 2: Instead of destination-pattern +9T it might be destination-pattern 9+T. You will know when you have it correct if, after clicking on the telephone icon, you receive a message that says ringing.

QUESTION 48:

Which gateway interface connects to the standard station port of a PBX?

A. FXS
B. E&M
C. POTS
D. FXO

Answer: D

QUESTION 49:

What from the list below combines voice mail, e-mail, and fax into a single application suite where a single application can be used to store and retrieve entire suite of message types?

A. PBSX Listing
B. Name Resolution IPTC
C. Call Manager 3.01
D. Cat 4000 STP v3
E. Unified messaging

Answer: E

QUESTION 50:

In a distributed call processing model, which three are located at each site? (Choose three.)

A. gatekeeper
B. voice messaging
C. media resources
D. Cisco CallManager cluster

Answer: B, C, D

QUESTION 51:

What can be used not only to restrict dialing, but also to identify a subset of a subset of a wildcard pattern (when using the @ wildcard in the North American Dialing Plan)?

A. An IS sheet
B. A ACL
C. A Route Filter
D. A DN top

Answer: C

QUESTION 52:

What does the Digit Discard Instruction of PreDot do to the pattern 9.2148134444?

A. prefix a 9 before the "-" if none is dialled
B. discard 2148134444 and send the 9 access code
C. only collect the first four digits counting right to left.
D. change it to 2148134444 before presenting it to the PSTN

Answer: D

QUESTION 53:

What is the key element in call admission control when interconnecting CallManager sites via the IP WAN?

A. gatekeepers
B. voice messaging
C. media resources
D. call processing agents

Answer: A

QUESTION 54:

Certkiller has its headquarters in New York and branch offices in Delaware, Delhi and Dakar. Headquarters and the Delaware branch office has IP Phones. The other two offices have analog phones that are connected to FXS port on the router in the site´s administration building. Users at these offices complain that they are unable to call out in the PSTN or to each other.
You receive the following output:
2611#s voice port 1/0/0
Foreign Exchange Station 1/0/0 Slot is 1, Sub-unit is 0,

Port is 0
Type of VoicePort is FXS
Operation State is DORMANT
Administrative State is UP
No Interface Down Failure
Description is not set
Noise Regeneration is enabled
Non Linear Processing is enabled
Non Linear Mute is disabled
Non Linear Threshold is -21 dB
Music On Hold Threshold is Set to 38 dBm
In Gain is Set to 0 dB
Out Attention is Set to 3 dB
Echo Cancellation is enabled
Echo Cancellation NLP mute is disabled
Echo Cancellation NLP threshold is -21 dB
Echo Cancel Coverage is set to default
Playout-delay Mode is set to default
Playout-delay Nominal is set to 60 ms
Playout-delay Maximal is set to 200 ms
Playout-delay Minimum mode is set to default, value 40 ms
Playout-delay Fax is set to 300 ms
Connection Mode is normal
Connection Number is not set
Initial Time Out is set to 10 s
Interdigit Time Out is set to 10 s
Call Disconnect Time Out is set to 60 s
Ringing Time Out is set to 180 s
Wait Release Time Out is set to 30 s
Companding Type is u-law
Region Tone is set for US
Analog Info Follows:
Currently processing none
Maintenance Mode Set to None (not in mtc mode)
Number of signaling protocol errors are 0
Impedance is set to 600r Ohm
Station name None, Station number None
Voice card specific Info Follows:
Signal Type is groundStart
Ring Frequency is 25 Hz
Hook Status is On Hook
Ring Active Status is inactive
Ring Ground Status is inactive
Tip Ground Status is inactive
Digit Duration Status is inactive
Digit Duration Timing is set to 100 ms

InterDigit Duration Timing is set to 100 ms
No disconnect acknowledge
Ring Cadence is defined by CPTone Selection
Ring Cadance are [20 40] * 100 msec
2611#
What is the cause of this problem?

A. The cptone is incorrect
B. The dial-type is incorrect
C. The signal type is incorrect
D. The playout-delay is incorrect
E. The disconnect-ack is incorrect

Answer: C

QUESTION 55:

You are the Voice technician at Certkiller .com. Your newly appointed Certkiller trainee wants to know on what type of port you would set impedance.
What will your reply be?

A. T1
B. E1
C. FXS
D. FXO
E. E&M

Answer: E
Source Cisco CVOICE book - page 3-48
VoicePortTuning Parameters
E&M voice port parameters
- input-gain
- no echo-cancel enable
- impedance
FXO voice port parameters
- echo-cancel coverage
- output-attenuation

QUESTION 56:

Which type of delay is caused by the line speed of the interface?

A. Queuing delay
B. Serialization delay
C. Propagation delay
D. Packetiziation delay

Answer: B

Explanation:
Serialization Delay
Serialization delay (?n) is the fixed delay required to clock a voice or data frame onto the network interface, and It is directly related to the clock rate on the trunk. Remember that at low clock speeds and small frame sizes the extra flag needed to separate frames is significant.
Queuing/Buffering Delay
After the compressed voice payload is built, a header is added and the frame is queued for transmission on the network connection. Because voice should have absolute priority in the router/gateway, a voice frame must only wait for either a data frame already playing out, or for other voice frames ahead of it. Essentially the voice frame is waiting for the serialization delay of any preceding frames in the output queue. Queuing delay (ßn) is a variable delay and is dependent on the trunk speed and the state of the queue. Clearly there are random elements associated with the queuing delay.
PacketizationDelay
Packetization delay(?n) is the time taken to fill a packet payload with encoded/compressed speech. This delay is a function of the sample block size required by the vocoder and the number of blocks placed in a single frame. Packetization delay may also be called Accumulation delay, as the voice samples accumulate in a buffer before being released.

QUESTION 57:

Certkiller has its headquarters in New York and branch offices in Delaware, Detroit and Denver.Each office has an analog phone at each location. These phones are connected to an FXS port on the on-site router. The Finance department at the Denver office is unable to make any phone class from these analog phones.
You receive the following output:
2611#s voice port 1/0/0
Foreign Exchange Station 1/0/0 Slot is 1, Sub-unit is 0,
Port is 0
Type of VoicePort is FXS
Operation State is DORMANT
Administrative State is UP
No Interface Down Failure
Description is not set
Noise Regeneration is enabled
Non Linear Processing is enabled
Non Linear Mute is disabled
Non Linear Threshold is -21 dB
Music On Hold Threshold is Set to 38 dBm
In Gain is Set to 0 dB
Out Attention is Set to 3 dB

Echo Cancellation is enabled
Echo Cancellation NLP mute is disabled
Echo Cancellation NLP threshold is -21 dB
Echo Cancel Coverage is set to default
Playout-delay Mode is set to default
Playout-delay Nominal is set to 60 ms
Playout-delay Maximal is set to 200 ms
Playout-delay Minimum mode is set to default, value 40 ms
Playout-delay Fax is set to 300 ms
Connection Mode is normal
Connection Number is not set
Initial Time Out is set to 10 s
Interdigit Time Out is set to 10 s
Call Disconnect Time Out is set to 60 s
Ringing Time Out is set to 180
Wait Release Time Out is set to 30 s
Companding Type is u-law
Region Tone is set for US
Analog Info Follows:
Currently processing none
Maintenance Mode Set to None (not in mtc mode)
Number of signaling protocol errors are 0
Impedance is set to 600r Ohm
Station name None, Station number None
Voice card specific Info Follows:
Signal Type is groundStart
Ring Frequency is 25 Hz
Hook Status is On Hook
Ring Active Status is inactive
Ring Ground Status is inactive
Tip Ground Status is inactive
Digit Duration Status is inactive
Digit Duration Timing is set to 100 ms
InterDigit Duration Timing is set to 100 ms
No disconnect acknowledge
Ring Cadence is defined by CPTone Selection
Ring Cadance are [20 40] * 100 msec
2611#
What is the cause of this problem?

A. The cptone is incorrect
B. The dial-type is incorrect
C. The signal type is incorrect
D. The playout-delay is incorrect
E. The disconnect-ack is incorrect

Answer: C

QUESTION 58:

You are the Voice technician at Certkiller .com. Your newly appointed Certkiller trainee wants to know what types of trunks Cisco support with the connection trunk command.
What will your reply be? (Choose three)

A. FXS to FXS trunks, FXS to FXO trunks, and FXS to E&M trunks
B. FXS to FXS trunks, FXS to FXO trunks, and E&M to E&M trunks
C. FXS to FXS trunks, FXO to FXO trunks, and E&M to E&M trunks
D. FXO to FXS trunks, FXO to FXO trunks, and E&M to E&M trunks
E. FXS to FXS trunks, FXS to E&M trunks, and E&M to E&M trunks

Answer: B

QUESTION 59:

You are the voice technician at Certkiller .com. Certkiller has its offices in Great Britain. You need to install a Cisco router to support IP Telephony services with direct-connected analog phones. You need to emulate the local PSTN provider.
What FXS port parameter do you need to change?

A. Pulse
B. Signal
C. Cptone
D. Busyout
E. Description

Answer: C

QUESTION 60:

You are the network technician at Certkiller .com. Your newly appointed Certkiller trainee wants to knowwhen glare occurs.
What will your reply be?

A. When echo cancellers fail to synchronize.
B. When two phones go off-hook at the same time.
C. When two optical wavelengths collide in the same fiber.
D. When both ends of a telephone line or trunk experience echo.
E. When both ends of a telephone line or trunk are seized by different users.

Answer: E

QUESTION 61:

You are the Voice technician at Certkiller .com. The Certkiller network uses VoIP. Your newly appointed Certkiller trainee wants to know what themodes of the playout delay buffer are.
What will your reply be?

A. Percent and Unit.
B. Nominal and Full.
C. Dynamic and Static.
D. Smooth and Serrated.
E. Minimum and Maximum.

Answer: C

QUESTION 62:

You are the network engineer at Certkiller .com. Certkiller has its headquarters in New York and a branch office in Delaware. In the branch office, one VoIP dial-peer has been configured to point to headquarters over a low speed serial link. You want to limit the maximum number of concurrent calls to 3.
Which command would you use?

A. interface serial 3/3
ip rsvp bandwidth 3
B. interface serial 3/3
max-con 3
C. dial-peer voice 1000 voip
max-conn 3
D. dial-peer voice 1000 voip
max-concurrent 3
E. dial-peer voice 1000 voip
ip rsvp neighbor 3

Answer: C

QUESTION 63:

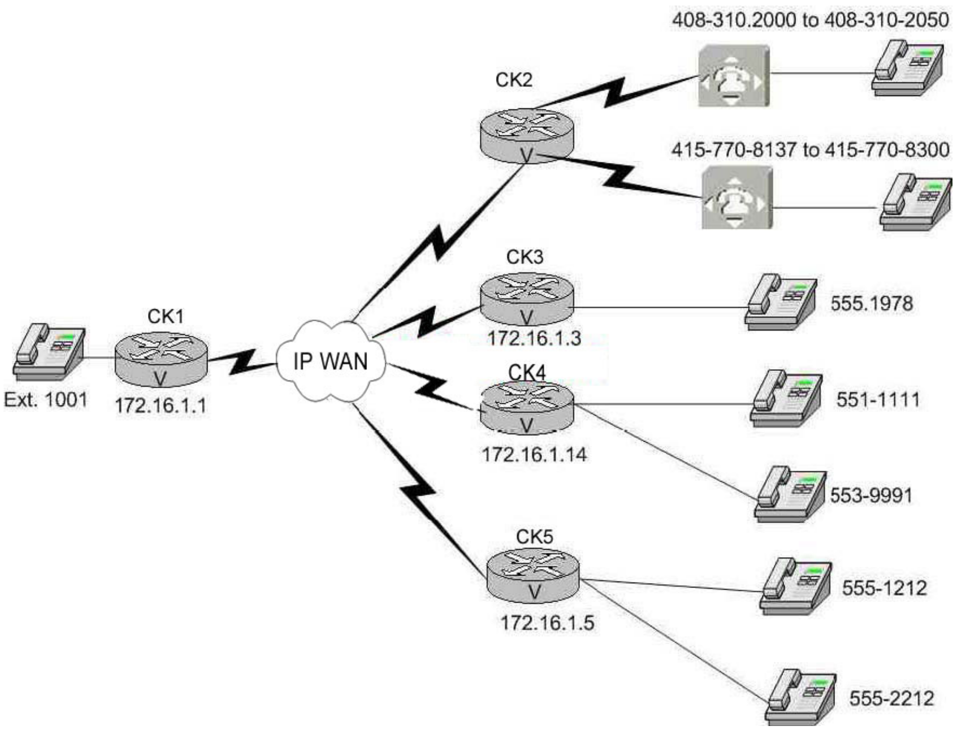
You are the network engineer at Certkiller .com. Certkiller has its headquarters in New York and branch offices in Delaware, Detroit and Denver. You have deployed VoIP over the Certkiller WAN. Certkiller user at headquarters complain that early in the day, the quality of calls between headquarters and the branch offices is very good, but as the day progresses and more calls are placed to the branch offices, the quality degrades.
The Certkiller network is using RSVP. The WAN bandwidth to the branch offices

allows 4 calls to the Delaware office, 6 calls to the Detroit office, and 8 calls to the Denver office. You want to verify the configuration of Call Admission Control on the headquarters router.
What command should you use?

A. show call cac conf
B. show call rsvp-sync logs
C. show call rsvp-sync conf
D. show call rsvp-sync stats
E. show call rsvp-sync events

Answer: C

QUESTION 64:



Use the exhibit to answer the following questions.
When a call is placed from extension 1001 to 555-2212, which outbound dial peer is matched?

A. dial-peer voice 5 voip
destination-pattern55[1-5]5[01][0-4].
B. dial-peer voice 1 voip
destination-pattern55[0-1]0[1-3]..
C. dial-peer voice 2 voip
destination-pattern .!5551978
D. dial-peer voice 4 voip
destination-pattern55[153][19]...[19][19][1]

E. dial-peer voice 3 voip
destination-pattern.T

Answer: E

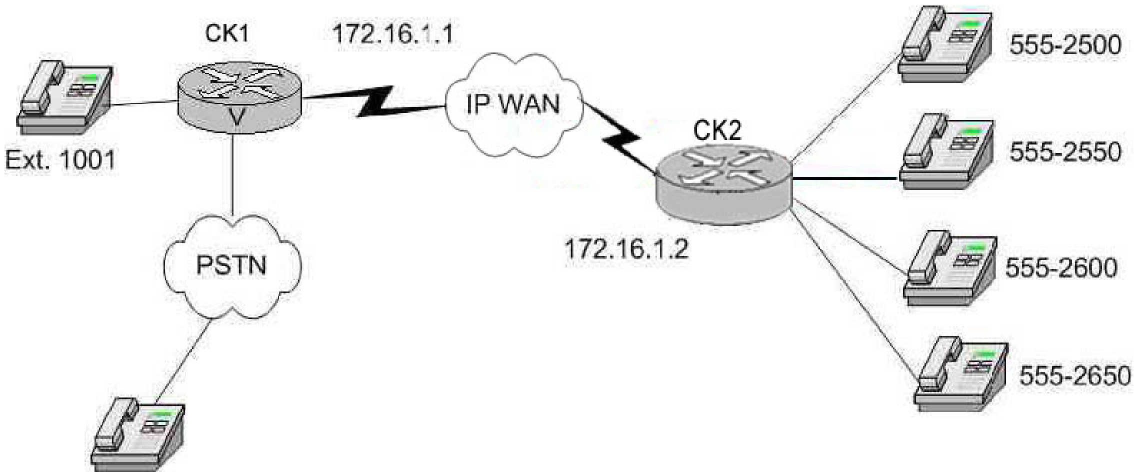
QUESTION 65:

What will be the outcome of an incoming VoIP call arriving at CK2 from CK1 ,
given the following router configurations?
CK1 Configuration
dial-peer voice 1 pots
destination-pattern 1111
port 1/0/0
CK2 Configuration
dial-peer voice 1 pots
destination-pattern 2222
port 1/0/0
!
dial-peer voice 2 voip
destination-pattern 1111
session-targetipv4:172.16.1.1

A. The call setup will proceed by matching dial-peer 1 pots, but will have one-way audio.
B. The call setup will fail.
C. The call setup will proceed and audio path will be established by matching the inbound call to the default dial peer.
D. The call setup will proceed, but will have no audio path.

Answer: B
Based on the configuration shown the correct answer concern a VoIP call arriving at CK2 from CK1 should be "The call will fail". The reason is that CK1 does not have a dial-peer statement defining a session target with the "session target ipv4:" command. The telephone on CK1 has no route defined on how to reach CK2 . Reference page 4-24 of CVoice version 4.1 class books. If the call was originating from CK2 to CK1 the correct answer would be "C" OR if the configurations were reversed then the answer would be "C".

QUESTION 66:

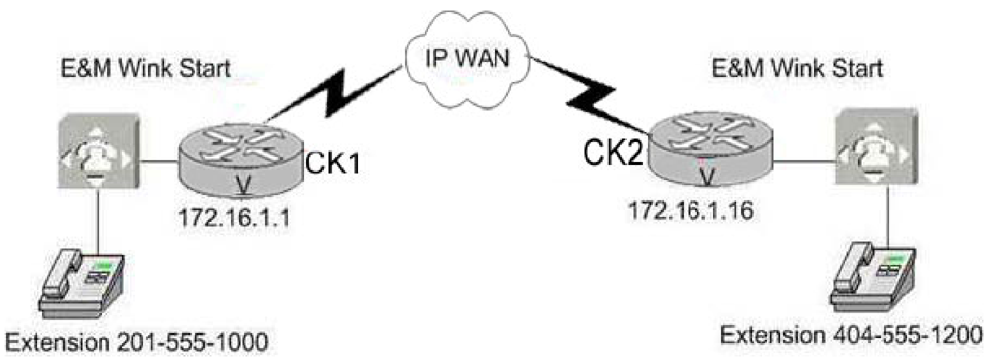


In router CK2 which dial peer statement will match only the four extensions?

A. dial-peer voice 1 pots
destinationpattern 5552[5-6].
B. dial-peer voice 1 pots
destination-pattern5552[5-6][05]0
C. dial-peer voice 1 pots
destination-pattern5552.[0-5]0
D. dial-peer voice 1 pots
destination-pattern555[2-5][56]0

Answer: C
Note: "C" is a correct answer but "B" would also work based upon the statements here.

QUESTION 67:



hostname CK1
!
interface serial0/0
ip address 172.16.1.1 255.255.255.248
!
controller t1
framing esp
clock source line

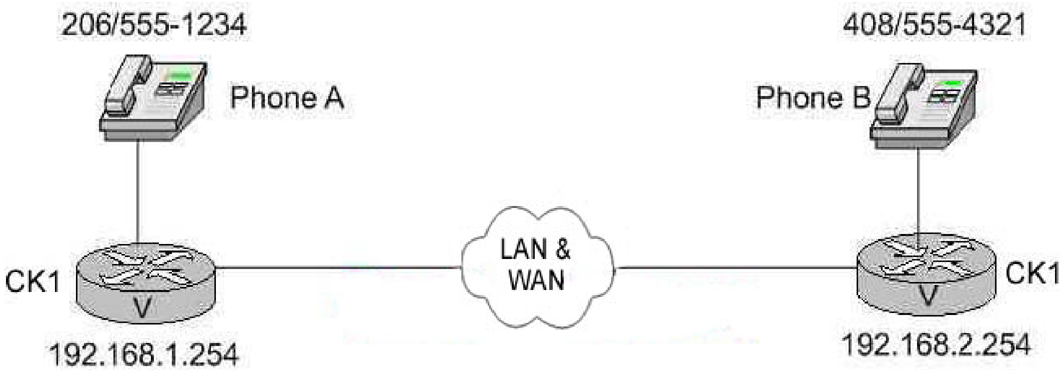
```
linecode b8zs
ds0-group 1timeslots 1-24 type e&m-wink-start
!
voice port 1/0:1
!
dial-peer voice 1 voip
destination-pattern 404555.....
session-target ipv4:172.16.1.6
!
dial-peer voice 2 pots
destination-pattern 201555.....
port 1/0:1
hostname CK2
!
interface serial0/0
ip address 172.16.1.6 255.255.255.248
!
controller t1
framing esp
clock source line
linecode b8zs
ds0-group 1timeslots 1-24 type e&m-wink-start
!
voice port 1/0:1
!
dial-peer voice 1 voip
destination-pattern 201555.....
session-target ipv4:172.16.1.1
!
dial-peer voice 2 pots
destination-pattern 404555.....
port 1/0:1
```

Your customer has forwarded this diagram and configuration. The customer wishes to have a connection between its PBXs, a connection that is created and dropped as required. There is one configuration statement missing from each router.
What are the two missing statements? (Choose two)

A. connection trunk 20155510004555... .
B. connection trunk 4045551200
C. connection tie-line 4045551200
D. connection tie-line 404555... .
E. connection tie-line 2015551000

Answer: C, E

QUESTION 68:



The following are the original dial peer configurations for routers CK1 and CK2 :
CK1 :
dial-peer voice 20 voip
destination-pattern 408.......
session target ipv4: 192.168.2.254
!
CK2
dial-peer voice 21 ports
destination-pattern 4085554321
port 1/0/1
!
Which phones can call to the other?

A. Only Phone A can call Phone B.
B. Only Phone B can call Phone A.
C. Both phones can call each other.
D. Neither phone can call the other.

Answer: A

QUESTION 69:

How are inbound and outbound call legs handled from the perspective of the source router?

A. Only the inbound call leg is established by the source router.
B. Only the outbound call leg is established by the source router.
C. The inbound call leg and outbouond call leg are matched to the same dial peer.
D. The outbound call leg is matched first. Then, once the source is known, an inbound call leg is established.
E. The inbound call leg is matched first. Then, once the destination is known, an outbound call leg is established.

Answer: E

QUESTION 70:

You are the Voice technician at Certkiller .com. The Certkiller network uses VoIP. Your newly appointed Certkiller trainee wants to know what themodes of the playout delay buffer are.
What will your reply be?

A. Percent and Unit.
B. Nominal and Full.
C. Dynamic and Static.
D. Smooth and Serrated.
E. Minimum and Maximum.

Answer: C

QUESTION 71:

You are the network technician at Certkiller .com. Your newly appointed Certkiller trainee wants to know what factors affects audio quality.
What will your reply be?

A. Echo and delay variation
B. Infidelity and delay variation
C. Echo and playout delay buffer
D. Infidelity and transmission medium

Answer: A

QUESTION 72:

When a call is placed, it is routed toward the destination.
Which call legs are created on that router for the call?

A. long legs
B. short legs
C. inbound call legs only
D. outbound call legs only
E. inbound and outbound call legs

Answer: E

QUESTION 73:

Which of the following parameter is checked first when matching inbound dial peers?

A. called number (DNIS) with voice-port
B. calling number (ANI) with answer-address
C. calling number (ANI) with destination pattern
D. calling number (ANI) with incoming called-number
E. called number (DNIS) with incoming called-number

Answer: E

QUESTION 74:

What is used to translate called (DNIS) and calling automatic number identification (ANI) numbers before routing the call?

A. IR IP internetworking
B. Transitional Pattern
C. PIM Routing
D. Translation Pattern

Answer: D

QUESTION 75:

Choose all functions available to you with the IP SoftPhone. (Choose all that apply.)

A. Automatic IPX blocking ASICs
B. Displays caller name
C. Displays the caller address
D. Resets all calls every 1 hour
E. Logs calls to the call log
F. Displays caller phone number

Answer: B, C, E, F

QUESTION 76:

What could happen if the playout delay buffer size is configured too large?

A. The overall echo on the connection may rise to unacceptable levels.
B. The overall delay on the connection may rise to unacceptable levels.
C. The overall stress on the connection may rise to unacceptable levels.
D. The overall volume on the connection may rise to unacceptable levels.

Answer: B

QUESTION 77:

You are the network engineer at Certkiller .com. You have configured dial peers in a hunt group for a Support team that answers when the number 5952215 is dialled. The Support team consists of one senior agent and three junior agents. You want the senior agent to receive the incoming call first.
Which dial peer should you configure to point to the senior agent?

A. dial-peer voice 1 pots
destination-pattern 5952215
port 1/0/0
preference 1
B. dial-peer voice 2 pots
destination-pattern 5952215
port 1/0/1
preference 0
C. dial-peer voice 3 pots
destination-pattern 5952215
port 1/1/0
preference 9
D. dial-peer voice 4 pots
destination-pattern 5952215
port 1/1/1
preference 0

Answer: D
Note: "D" is a valid answer but based on the configuration statements shown "B" would work. Both have the preference set to 0 and all other statements in each answer are correct.

QUESTION 78:

You are the network engineer at Certkiller .com. Certkiller has its headquarters in New York and a branch office in Delaware.
In the branch office, one VoIP dial-peer has been configured to point to headquarters over a low speed serial link. You want to limit the maximum number of concurrent calls to 3.
Which command would you use?

A. interface serial 3/3
ip rsvp bandwidth 3
B. interface serial 3/3
max-con 3
C. dial-peer voice 1000 voip
max-conn 3
D. dial-peer voice 1000 voip

max-concurrent 3
E. dial-peer voice 1000 voip
ip rsvp neighbor 3

Answer: C

QUESTION 79:

With Cisco CallManager Release 3.0, the term "route point" is replaced with which term from the list below?

A. Call list
B. Phone/list
C. Route list
D. IP_PHONE_SET
E. Phone-all
F. Route Print

Answer: C

QUESTION 80:

For very low-speed links (those with a link speed of less than 768 K), it is necessary to use techniques that provide link fragmentation and interleaving of packets. This prevents voice traffic from being delayed behind large data frames and hence bounds jitter.
What are two techniques that exist for this?

A. Ipng for DSL links
B. LECS for ATM links
C. Multilink PPP (MLP) for serial links
D. FRF.12 for Frame Relay

Answer: C, D

QUESTION 81:

You are the network engineer at Certkiller .com. Certkiller has been using the following dial peer codec command:
Codec g729r8
You reconfigure the dial peers with the following command:
Codec g729ar8 bytes 10
How will this reconfiguration affect the voice network bandwidth and delay characteristics? (Choose two.)

A. There will be no change.

B. Delay will increase on a per call basis.
C. Delay will decrease on a per call basis.
D. Bandwidth consumption will decrease on a per call basis.
E. Bandwidth consumption will increase on a per call basis.

Answer: C, E

QUESTION 82:

What happens if no incoming dial peer matches a router or gateway?

A. The incoming call leg takes an alternate path.
B. The incoming call legmatches the default dial peer.
C. The incoming call leg sends a busy to the originator.
D. The incoming call leg is denied and the call is dropped.

Answer: B

QUESTION 83:

If a PC connected to an IP Phone is having trouble obtaining an IP address, which setting on the phone might help resolve the problem?

A. Admin VLAN
B. Spanning Tree
C. Default Gateway
D. Forwarding Delay

Answer: D

QUESTION 84:

What are two characteristics of a distributed call processing model? (Choose two.)

A. sites connected via the PSTN
B. sites connected via the IP WAN
C. call processing agent at one site
D. call processing agent at each site

Answer: B, D

QUESTION 85:

You have all ten digits being sent to your CM from the PSTN (via a gateway). If you have four digit extensions, how do you make sure that the call gets routed?

A. update the Phone Calling Parity mask
B. -change the Route Group configuration
C. configure the GW to only collect four digits
D. change the Network Side/User Side Parameter on the gateway

Answer: C

QUESTION 86:

Cisco is making every effort to ensure that the gateways, applications, and client produced integrate and operate seamlessly with third party products. From the list below, select which protocols are being used to ensure this effort.

A. H.323
B. Session Initiation Protocol (SIP)
C. Media Gateway Control Protocol (MGCP)
D. Simple Gateway Control Protocol (SGCP)
E. All choices are correct.

Answer: E

QUESTION 87:

Which of the following statements is correct when discussing how the Cisco CallManager works with IP Phone registration? (Choose all that apply.)

A. On initial configuration, an IP phone is assigned a DSNP listing, which it loses when moved.
B. On initial configuration, an IP phone is assigned a directory number (DN), which it loses when moved
C. On initial configuration, an IP phone is assigned a DSNP, which it maintains wherever it resides
D. On initial configuration, an IP phone is assigned a directory number (DN)k, which it maintains wherever it resides.

Answer: D

QUESTION 88:

When discussing Route Groups, we know that they control specific devices such as gateways. On which protocols can gateways be based?

A. H.323
B. MGCP
C. IPNCP
D. Skinny Gateway Protocol

E. SNA
F. SAA
G. DecLat

Answer: A, B, D

QUESTION 89:

Which statement is true about VoIP packet loss?

A. Lost packets are simply retransmitted.
B. Even minimal packet loss causes echo.
C. IP phones can reconstruct up to three consecutive loss packets.
D. Codec algorithms can overcome minimal packet loss.

Answer: D

QUESTION 90:

Which is the best way to achieve a scalable dial plan?

A. Group numbers for a particular area.
B. Variable number of extension digits.
C. Single number prefixing.
D. Hunt groups.

Answer: A

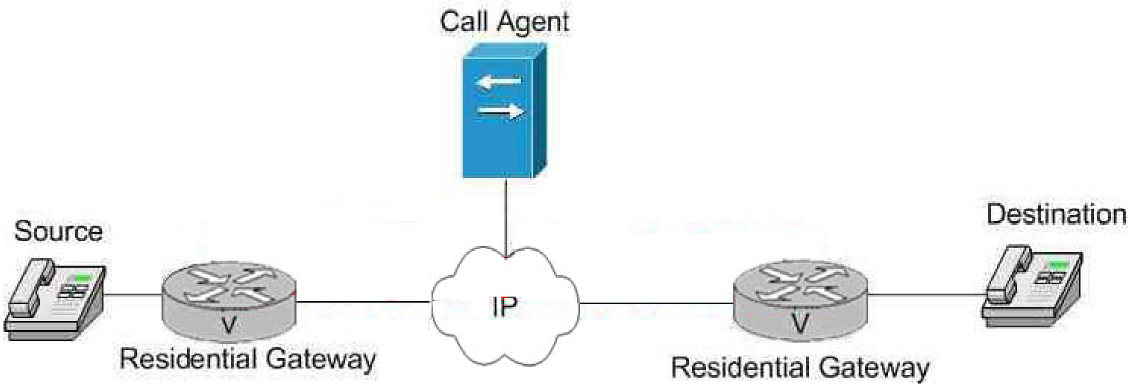
QUESTION 91:

Which channel carries Q.931 signals in a T1 connection from a PBX to a Cisco gateway?

A. 0
B. 16
C. 24
D. 31

Answer: C

QUESTION 92:



At what point does the MGCP call agent turn over to the residential gateways the setup of the call path?

A. After the call agent has been notified that an event has occurred at the source residential gateway.
B. After the call agent has been notified of an event and has instructed the source residential gateway to create a connection.
C. The call agent is never out of the call path setup.
D. After the call agent has sent a connection requests to both the source and destination and has relayed a modify-connection request to the source so that the source and destination can set up the call path.
E. After the call agent has forwarded session description protocol information to the destination from the source and has sent a modify connection to the destination and a create-connection request to the source.

Answer: D

QUESTION 93:

What is true about H.323 endpoint call setup?

A. Endpoints always do their own call setup.
B. Endpoints require a gatekeeper to do call setup.
C. Endpoints can either do their own setup or be assisted by a gatekeeper.
D. Endpoints require a proxy server to do call setup.

Answer: C

Explanation:
A gatekeeper is an H.323 entity on the network that provides services such as address translation and network access control for H.323 terminals, gateways, and MCUs. Also, they can provide other services such as bandwidth management, accounting, and dial plans that can be centralized to provide salability.
Gatekeepers are logically separated from H.323 endpoints such as terminals and gateways. They are optional in an H.323 network, but if a gatekeeper is present, endpoints must use the services provided.

QUESTION 94:

Examine the example output
hostname GW1
!
interface Ethernet 0/0
ip address 172.16.2.1 255.255.255.0
h323-gateway voip interface
h323-gateway voip id GK1-zone1 .abc.com abc.com ipaddr 172.16.2.2
h323-gateway voip h323-id GW1
h323-gateway voip bind srcaddr 172.16.2.1
!
dial-peer voice 1 voip
destination-pattern 12.12... ... .
session-target ras
!
dial-peer voice 2 pots
destination-pattern 2125551212
no register e164
!
end
Choose the command that will restore communication with gatekeeper functionality to this device.

A. h323-gateway voip h323-id GK1
B. gateway
C. h323-gateway voip bind srcaddr 172.16.2.2
D. h323-gateway voip GW1-zone2.abc.com abc.com ipaddr 172.16.2.1

Answer: B

QUESTION 95:

What does a gateway router match to a dialed number when setting up a VoIP call?

A. IP route
B. Destination pattern
C. Call leg
D. Session target

Answer: B

Explanation:
The router selects a dial peer for a call leg by matching the string that is defined by using

the answer-address, destination-pattern, or incoming called-number command in the dial
peer configuration.

QUESTION 96:

What is used in the Cisco implementation of T.37?

A. Special gateways configured as IVRs
B. Special gateways configured as TIFFs
C. Special gateways configured as on-ramps and off ramps
D. Special gateways configured as MTA, MDN, and DSN parameters

Answer: C

QUESTION 97:

You are the network engineer at Certkiller .com. Your newly appointed Certkiller
trainee wants to know how an endpoint determines the address of the gatekeeper.
What will your reply be? (Choose two.)

A. The endpoint issues a GCP.
B. The endpoint issues a GRQ.
C. The endpoint queries the registrar server.
D. The endpoint is preconfigured to recognize the domain name or IP address of its
gatekeeper.

Answer: B, D

QUESTION 98:

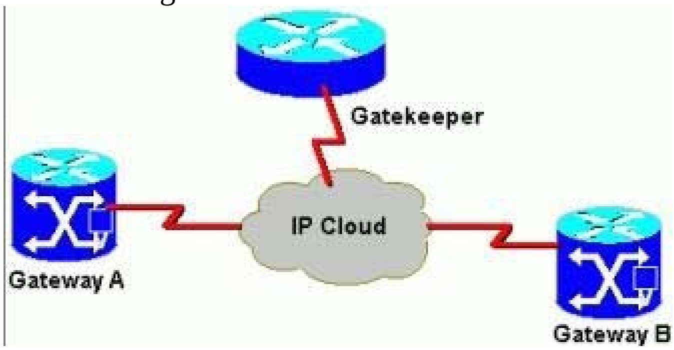
You are the Voice engineer at Certkiller .com. Certkiller has an H.323
gatekeeper.Your newly appointed Certkiller trainee wants to know what functions
are supported by this gatekeeper.
What will your reply be? (Choose four.)

A. It provides services to registered endpoints.
B. It converts an alias address to an IP address.
C. It responds to bandwidth requests and modifications.
D. It provides translation between audio, video, and data formats.
E. It provides conversion between call setup signals and procedures.
F. It limits access to network resources based on call bandwidth restrictions.
G. It provides conversion between communication control signals and procedures.

Answer: A, B, C, F

QUESTION 99:

You are the network engineer at Certkiller .com. The Certkiller network is shown in the following exhibit:



If the show gatekeeper callscommand shows a total of five active calls on the gatekeeper, how many call legs would the show call active voice command display on Gateway A?

A. 2
B. 5
C. 6
D. 10
E. 15

Answer: D

QUESTION 100:

You are the Voice technician at Certkiller .com. Your newly appointed Certkiller trainee wants to know what makes it possible for gatekeepers to communicate with each other.
What will your reply be?

A. RTP
B. RAS channel
C. call signaling channel
D. H.245 control channel
E. Q.931 control channel

Answer: B

QUESTION 101:

Your newly appointed Certkiller trainee wants to knowwhat protocol negotiates the codec type for H.323 sessions.
What will your reply be?

A. H.225
B. H.245
C. Q.931
D. Q.932
E. H.320

Answer: B

QUESTION 102:

You are the network engineer at Certkiller .com. Certkiller has its offices in London.
You are installing a voice gateway.
What do you need to verify? (Choose two.)

A. The PSTN standards in England.
B. Encryption capabilities legalities.
C. The service provider installing the gateway.
D. Supplementary service including fax and modem.

Answer: A, B

QUESTION 103:

You are the network engineer at Certkiller .com. Your newly appointed Certkiller trainee wants to knowwhat a voice gateway is.
What will your reply be?

A. It is a device that connects two dissimilar networks.
B. It is a device that transports voice and restricts data.
C. It is a device that can support only a distributed call processing model.
D. It is a device that cannot be connected to the traditional PSTN network.

Answer: B

QUESTION 104:

What would Receiving an Alarm Indication Signal of Blue indicate on your T1 connection where your voice traffic is going over?

A. Blue means there is an alarm occurring in the building, it is part of your disaster plan.
B. Blue means there is an alarm occurring on the line downstream from the equipment that is connected to the port
C. There is no blue alarm, only red and yellow.
D. Blue means there is an alarm occurring on the line upstream from the equipment that is connected to the port

Answer: D

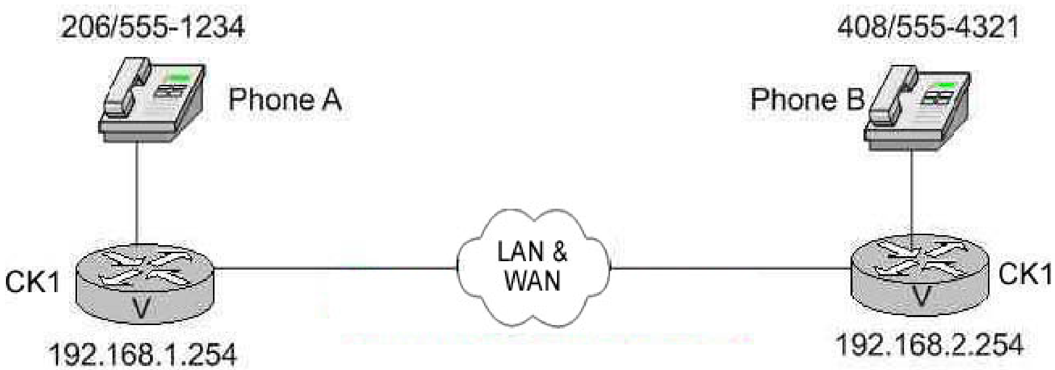
QUESTION 105:

You are the network technician at Certkiller .com. Your newly appointed Certkiller trainee wants to knowwhen glare occurs.
What will your reply be?

A. When echo cancellers fail to synchronize.
B. When two phones go off-hook at the same time.
C. When two optical wavelengths collide in the same fiber.
D. When both ends of a telephone line or trunk experience echo.
E. When both ends of a telephone line or trunk are seized by different users.

Answer: E

QUESTION 106:



One voice packet is lost between Phone A and Phone B.
What will be the result to the listener?

A. The call is terminated.
B. The listener will experience a gap in the received audio stream.
C. The listener will hear the audio normally.
Packet loss concealment will make the loss inaudible.
D. The listener will hear the audio out of order when the lost packet is retransmitted.

Answer: B

QUESTION 107:

What will happen when a network link is oversubscribed?

A. The link goes down.
B. All voice calls suffer.

C. Voice packets are fragmented.
D. Excess voice calls are dropped.
E. Data packets are given priority.

Answer: B

QUESTION 108:

Your newly appointed Certkiller trainee wants to know what CAC applies to.
What will your reply be?

A. Latency
B. Data traffic
C. Voice traffic
D. TCP networks
E. Voice and data traffic

Answer: C

QUESTION 109:

You are the network engineer at Certkiller .com. Certkiller has its headquarters in New York and a branch office in New Hamshire. You want to configure a permanent connection between the PBX at headquarters and the PBX at the branch office.
The following configuration is used at the New York site:
dial-peer voice 20 pots
destination-pattern 20
port 1.0:1
dial-peer voice 41 voip
destination-pattern 41
session target ipv4:10.2.0.20
The following configuration is used at the New Hamshire site:
dial-peer voice 40 pots
destination-pattern 41
port 1.0:1
dial-peer voice 20 voip
destination-pattern 20
session target ipv4:10.4.1.41
What must be added to the voice port configuration at the New York site?

A. connection trunk 20
B. connection trunk 41
C. connection tie-line 20
D. connection tie-line 41

Answer: B

Explanation: You must specify the same number in the connection trunk voice port command as in the appropriate dial peer destination-pattern command in order to create a permanent trunk.

QUESTION 110:

You are the Voice technician at Certkiller .com. Your newly appointed Certkiller trainee wants to know what configuration would define a destination pattern for all of the 1000 and 2000 range of extensions starting with the numbers 555.
What will your reply be?

A. 5551...
B. 5552...
C. 555[1-2]...
D. 555[100-200]...
E. 555[1000-2000]...

Answer: C

QUESTION 111:

Certkiller distributes computer components and has warehouses in New York and Chicago. Headquarters is located in Washington, DC. To keep costs low, all inside sales associates are located at headquarters.
Your want to provide a direct analog telephone connection to the inside sales teams from the pick-up counters at the warehouses.This connection should not require the inside sales teams to dial any digits.
One of the warehouses is having a problem with their sales phone.
You receive the following output:
altwhse#show voice port 1/0:1
Foreign Exchange Office
Type of VoicePort is E&M
Operation State is DORMANT
Administrative State is UP
The Last Interface Down Failure Cause is Administrative Shutdown
Description is not set
Noise Regeneration is enabled
Non Linear Processing is enabled
Music On Hold Threshold is Set to -38 dBm
In Gain is Set to 0 dB
Out Attenuation is Set to 0 dB
Echo Cancellation is enabled
Echo Cancel Coverage is set to 8 ms
Connection Mode is plar

Connection Number is 2000
Initial Time Out is set to 10 s
Interdigit Time Out is set to 10 s
Call-Disconnect Time Out is set to 60 s
Ringing Time Out is set to 180 s
Region Tone is set for US
What is the cause of the problem?

A. VoicePort type is incorrect.
B. Echo cancellation is enabled.
C. Connection Number is not required.
D. Interdigit Time Out is set to 10 seconds.

Answer: A

QUESTION 112:

You are the network engineer at Certkiller .com. Your newly appointed Certkiller trainee wants to knowwhich features render VAD ineffective.
What will your reply be? (Choose two.)

A. Fax
B. CNG
C. Call waiting
D. Music on hold
E. Call forwarding

Answer: A, D

QUESTION 113:

What is a logical grouping of directory numbers (DN) and route patterns with similar reachability characteristics when working with IP Telephony?

A. Call Manager
B. CiscoWorks IP set
C. A DSN
D. A Partition

Answer: D

QUESTION 114:

What unlocks the 7960 configuration menu?

A. **3

B. **#
C. **4
D. **#*

Answer: B

QUESTION 115:

Name two standards that are being adopted from by the telecommunicates industry that are used to communicate between applications such as the Cisco CallManager providing IP PBX functionality and unified products such as the GateServer products acquired through the acquisition of Amteva. (Select two.)

A. The Java Telephone Application Programmable Interface (JTAPI)
B. The IP Telephone Call protocol (IPTC)
C. The Telephony Application programmable Interface (TAPI)
D. The System Architecture Voice Telephony Architecture (SAVTA)

Answer: A, C

QUESTION 116:

Which network protocols does an IP Phone use to communicate?

A. TCP/IP for both skinny signalling and RTP voice streams
B. UDP/IP for both skinny signalling and RTP voice streams
C. TCP/IP for skinny signalling and UDP/IP for RTP voice streams.
D. TCP/IP for skinny signalling and TCP/IP for RTP voice streams.

Answer: C

QUESTION 117:

There are six major steps for WAN deployment when preparing IP telephony. From the list below, please select which of the following are valid pre deployment choices. (Choose all that apply.)

A. Choosing Wiring Closets carefully
B. Determining Voice Bandwidht Requirements
C. Assessing Results
D. Selecting the right handset for the IP SoftPhone
E. Analyzing Upgrade Requirements
F. Collecting Information on the Current WAN Environment

Answer: B, C, E, F

QUESTION 118:

Before voice and video can be placed on a network, it is necessary to ensure that adequate bandwidth exists for all required applications. To begin, the minimum bandwidth requirements for each major application (for example, the voice media streams, video streams, voice control protocols, and all data traffic) should be summed. This sum represents the minimum bandwidth requirement for any given link, and it should consume no more than what percentage of the total bandwidth available on that link?

A. 25%
B. 50%
C. 100%
D. 75%

Answer: D

QUESTION 119:

You need to prefix any outbound number dialled by a user with a 9. Where can you do this? (Choose two.)

A. in a Route Filter
B. on a Route Pattern
C. on a Translation Pattern
D. on the phone configuration mask

Answer: B, C

QUESTION 120:

Your Manager asks you as the Lead Network Designer to give a status on the VOIP integration project. Your Manager specifically asks what you need to replace the PBX. From the list below, what are you going to need to replace the PBX to roll out the VOIP solution? (Choose all that apply.)

A. TCP/IP
B. Cisco CallManager
C. IPX/SPX compatible
D. IP Telephones
E. All of the answers
F. Cat 4000's

Answer: A, B, D, F

QUESTION 121:

What is the major advantage of designing and placing VoIP and Internet telephony in a clients organization?

A. It is cheap but you still need a PBX regardless
B. The PSTN is doomed to be EOL in 5 years and this is the replacement.
C. It avoids the tolls charged by ordinary telephone service
D. Even without QoS it is much clearer that PSTN technology.

Answer: C

QUESTION 122:

You are the network technician at Certkiller .com. Your newly appointed Certkiller trainee wants to know what function T-CCS performs.
What will your reply be?

A. It allows a PBX to pass signalling to the PSTN switch.
B. It allows a PBX to pass analog signalling to the router
C. It allows a PBX to pass signalling to the router for compression and processing
D. It allows a PBX to pass proprietary signalling to another PBX across the IP network.

Answer: D

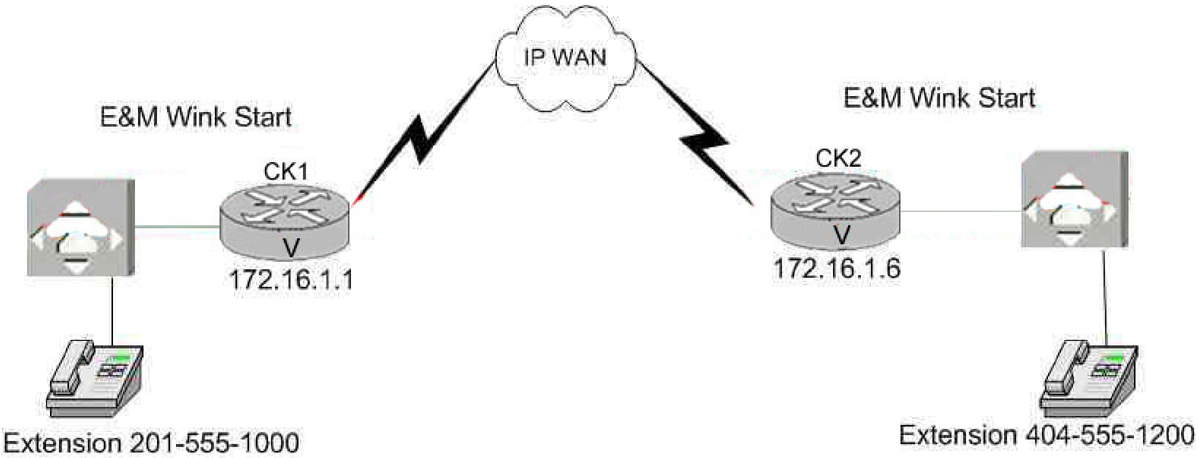
QUESTION 123:

You are the network engineer at Certkiller .com. You to connect a Cisco voice gateway to a PBX or the PSTN via ISDN (PRI, QSIG, BRI).
What are two attributes of the PBX/PSTN switch that must be known to understand which features to configure on the voice gateway to connect successfully to it?
(Choose two)

A. Whether Q.921 or Q.931 is supported by the PBX/PSTN switch.
B. Whether Symmetric mode is supported by the PBX/PSTN switch.
C. Which PRI/BRI switch-type is supported by the PBX/PSTN switch.
D. Whether network or user side is supported by the PBX/PSTN switch.
E. Whether wink, delay dial, or immediate dial is supported by the PBX/PSTN switch.

Answer: C, D

QUESTION 124:




```
hostname CK1
!
isdn switch-type primary-qsig
!
interface serial0/0
ip address 172.16.1.1 255.255.255.248
!
control t10/0
pri-group 1timeslots 1-23
!
interface serial0/0:23
isdn incoming-voice voice
!
voice-port 1/0:1
dial-peer voice 2 pubs
destination-pattern 20155....
port 1/0:1
```

```
hostname CK2
!
isdn switch-type primary-qsig
!
interface serial0/0
ip address 172.16.1.6 255.255.255.248
!
control t10/0
pri-group 1timeslots 1-23
!
interface serial0/0:23
isdn incoming-voice voice
!
voice-port 1/0:1
connection tie-line 201555
destination-pattern 404556....
port 1/0:1
```

You are working with a potential customer that would like to integrate its existing PBX telephone system into its IP network. The accompanying figure shows that the customer has two offices that need to be connected to the IP network so that the customer can exchange telephone calls without using the PSTN. Both PBXs are currently connected to T1 ISDN circuits.
Which signaling type will allow you to support your customer?

A. QSIG
B. CCS
C. CAS
D. T-CCS
E. E&M
F. FXO

Answer: A

QUESTION 125:

Which statement is an example of in-band signaling?

A. Uses a single channel for synchronization and hook status.
B. Transports synchronization signals within the voice channel.
C. Carries hook status in a dedicated signaling channel.
D. Robs bits from some frames to provide signaling states.

Answer: D

QUESTION 126:

You are the network technician at Certkiller .com. VoIP is implemented on the Certkiller network. Your newly appointed Certkiller trainee wants to know what this implementation uses to carry the payload across the network.
What will your reply be?

A. Only RTP
B. Only UDP
C. UDP inside RTP
D. RTP inside UDP

Answer: D

QUESTION 127:

In a VoIP environment when speech samples are framed every 20 ms. a payload of 20 bytes is generated. Assuming a total packet length of 60 bytes, what is the length of the packet header if cRTP is deploued without redundancy checks?

A. 1 byte
B. 2 bytes
C. 3 bytes
D. 4 bytes
E. 20 bytes
F. 40 bytes

Answer: B

QUESTION 128:

What does the PBX use to determine the destination of a call?

A. An ISDN ANI packet
B. A blocked/permitted call list
C. An analysis of the dialled digits
D. Historic requests from the specific phone extension

Answer: C

QUESTION 129:

Which of the following are CS-ACELP coding schemes? (Choose two)

A. G.711
B. G.728
C. G.729
D. Q.931
E. G-729A

Answer: C, E

QUESTION 130:

Which of the following is the worst-case compression delay for CD-ACELP?

A. 2.5 ms
B. 5 ms
C. 7.5ms
D. 10 ms
E. 20 ms

Answer: E

QUESTION 131:

What type of connection is considered a call leg?

A. A digital connection
B. A virtual connection
C. A logical connection
D. A physical connection
E. A hardwired connection

Answer: C

QUESTION 132:

To which layer of the OSI model does Q.921 signaling equates to in ISDN?

A. Session
B. Network
C. Transport

D. Data-Link
E. Application

Answer: D

QUESTION 133:

Certkiller has a PBX at corporate HQ and one at a branch office. You to replace the PBX-to-PXB TDM trunk connection with IP connectivity. The PBXs use proprietary signalling method.
The following is a partial configuration of the HQ router that connect to the PBX:
controller t1 1/0
ds0-group 1 timeslots 1-24 type ext-sig
dial-peer voice 1 voip
destination-pattern 1001
session target ipv4:10.10.0.1
dial-peer voice 2 pots
destination-pattern 2001
port 1/0:1
connection trunk 1001
Which command is missing from the above configuration?

A. transparent-ccs in the voice port configuration
B. signal wink-start in the controller t1 configuration
C. auto-cut-through in the pots dial peer configuration
D. codec clear-channel in the voip dial peer configuration

Answer: D

QUESTION 134:

You are the network engineer at Certkiller .com. The Certkiller ISDN network has two PBX systems from different manufactures.
Which protocol allows functionality between these two PBX systems?

A. QSIG
B. Q.921
C. Q.931
D. T-CCS

Answer: A

QUESTION 135:

You are the network technician at Certkiller .com. Your newly appointed Certkiller trainee wants to knowwhich application conveys fax using T.37 fax relay.

What will your reply be?

A. IVR
B. TCL
C. TIFF
D. SNMP
E. SMTP

Answer: E

QUESTION 136:

What will happen when a network link is oversubscribed?

A. The link goes down.
B. All voice calls suffer.
C. Voice packets are fragmented.
D. Excess voice calls are dropped.
E. Data packets are given priority.

Answer: B

QUESTION 137:

Certkiller sells managed IP Phone service to businesses in multi-tenant units.
Certkiller has POPs in many cities, so all of their dial peer patterns are based on 10 digit numbers. Users dial 9 for local calls, followed by the 7 digital local number.
The following dial peer has been configured in a New York POP:
dial-peer voice 595 pots
destination-pattern 595
port 1/0:24
A user dials a local number, 9-638-4422.
What command must be configured in the gateway to allow the call to complete?

A. prefix 595
B. forward-digits 7
C. rule 1 9.......595.......
D. forward 9.......595.......
E. num-exp 9.......595.......

Answer: E

QUESTION 138:

IP Telephony uses which protocol that does not accommodate re-transmission?

A. RIP (Routing Information Protocol)
B. IP (Internet Protocol)
C. RTP (real time protocol)
D. TCP (Transmission Control Protocol)

Answer: C

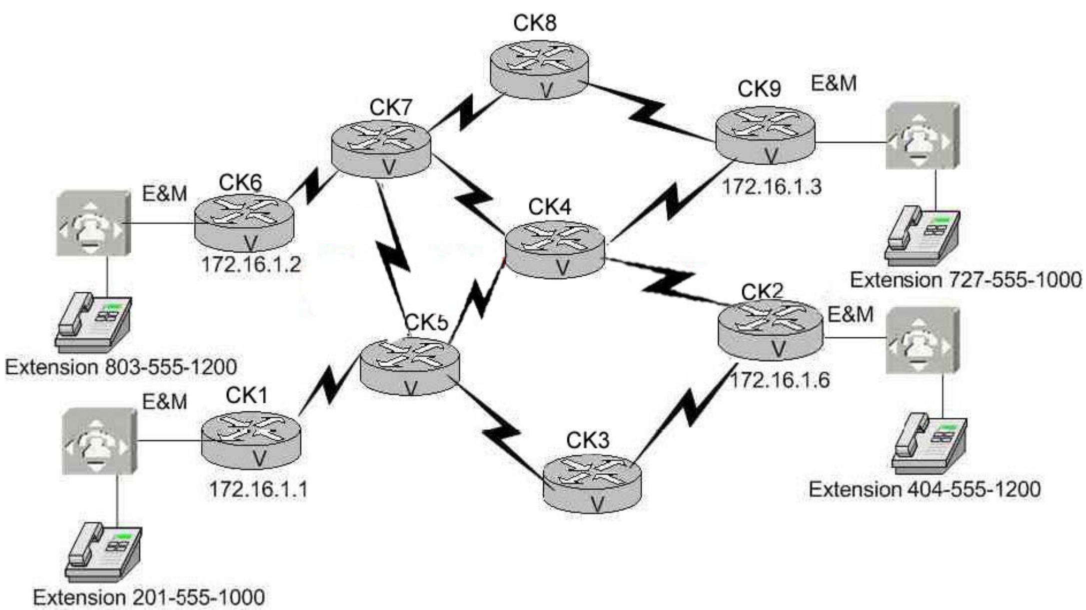
QUESTION 139:

When placing a call from an IP Phone to another IP Phone, how is ringback generated??

A. CallManager generates an RTP stream to play ringback on the originated phone.
B. CallManager sends a command to the originating IP Phone to play ringback locally.
C. The originating IP Phone plays ringback locally until the RTP stream has been established.
D. The phone is connected to an audio file server that generates the inband ringback tones.

Answer: B

QUESTION 140:



In the VoIP network above, which protocol provides the necessary sequence numbers so voice packets originating at CK1 are played in the correct order to CK5 ?

A. UDP
B. TCP
C. RTCP

D. RTP
E. CRTP

Answer: D

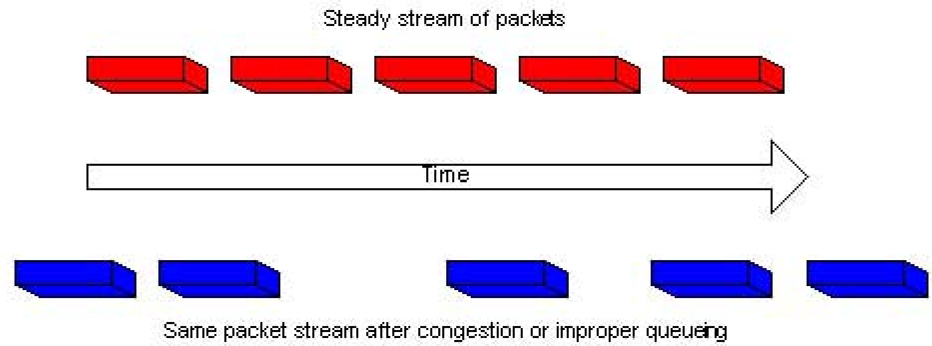
QUESTION 141:

What is the most probable cause of jitter?

A. Variable delay
B. Dropped packets
C. Impedance mismatch
D. Excessive delay

Answer: A

Explanation:
Jitter in Packet Voice Networks
Jitter is defined as a variation in the delay of received packets. At the sending side, packets are sent in a continuous stream with the packets spaced evenly apart. Due to network congestion, improper queuing, or configuration errors, this steady stream can become lumpy, or the delay between each packet can vary instead of remaining constant. This diagram illustrates how a steady stream of packets is handled.



When a router receives a Real-Time Protocol (RTP) audio stream for Voice over IP (VoIP), it must compensate for the jitter that is encountered. The mechanism that handles this function is the playout delay buffer. The playout delay buffer must buffer these packets and then play them out in a steady stream to the digital signal processors (DSPs) to be converted back to an analog audio stream. The playout delay buffer is also sometimes referred to as the de-jitter buffer.

QUESTION 142:

When an IP phone says "Configuration CM List", what is it doing?

A. downloading a .cnf.xmk file via TFTP
B. retrieving the OS79XX.txt files from TFTP
C. downloading the application load from the TFTP server

D. attempting to register with the first two CallManagers onits list of configure CallManagers

Answer: A

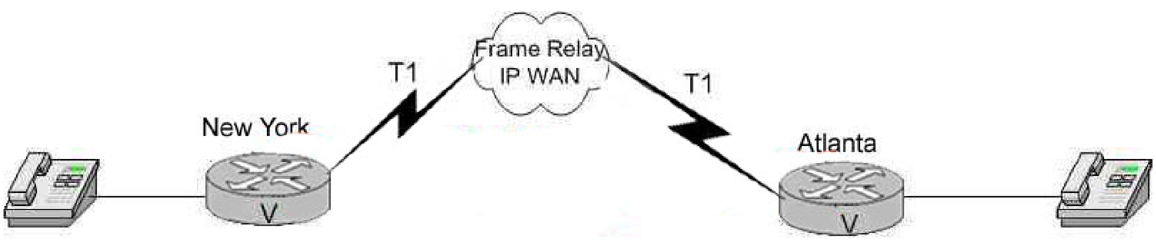
QUESTION 143:

Name two sensitivities that Voice traffic has that data traffic is not necessarily affected by.

A. TPI
B. RFI
C. Delay
D. EMI
E. Jitter
F. Noise

Answer: C, E

QUESTION 144:



Your customer would like to investigate converging voice and data on their existing T1 Frame Relay WAN link between New York and Atlanta. The following applications are consuming no more bandwidth than what is in the list on this segment of the network.
T1 link 1536 Kbps
e-mail75 Kbps
Internet 200 Kbps
Oracle 500 Kbps
FTP 250 Kbps
Total 1025 Kbps
The customer has allocated 25% of the WAN link for routing updated and other overhead. Assuming 6 bytes overhead for Frame Relay, no cRTP and using the G.729 codec, how many calls could be placed on this link?

A. 2 calls
B. 3 calls
C. 4 calls
D. 5 calls
E. 6 calls

Answer: C
Based upon a total bandwidth of 1536 Kbps and 1025 Kbps being used by other applications you can only have 4 calls not 5. The reason is that of the 1536 Kbps of bandwidth only 75% of it is available (or 1152 Kbps). 1152 minus 1025 leaves just 127 Kbps available for voice traffic. Assuming that you are using FRF.12, G.729 (stated in this scenario), and no cRTP (also stated in this scenario) then you will need approximately 28.14 Kbps per call with 5% overhead included (26.8 Kbps without overhead). 26.8 x 5 = 134 Kbps and 28.14 x 5 = 140.7 Kbps. Both exceed the 127 Kbps available for voice. To calculate the required bandwidth reference the "Voice Codec Bandwidth Calculator" available on Cisco's web site (requires a CCO sign-on to access the calculator).

QUESTION 145:

You have set up Call Admission Control for a customer between their headquarters and manufacturing facility over their Frame Relay WAN. You are using the G.726r16 codec with a 40 byte sample, CRTP without CRC, and 90 kbps configured as the maximum bandwidth for CAC to use.
What will happen when 7 calls try to call the remote office?

A. All the calls will go through without any quality issues.
B. Only 4 calls will go through and the remainder will get a reorder tone.
C. Six calls will go through, and the seventh call will be placed on hold until bandwidth is available.
D. Three calls will cross the Frame Relay WAN link, and four will use the PSTN with AAR.

Answer: B

QUESTION 146:

You have designed a complex dial plan using digit manipulation. Given the following snippet of your configuration file, what action would you expect to result when a call beginning with the digits "5501" is received?
dial-peer voice 1 pots
destination-pattern 5501... ...
prefix
port 1/0/0

A. A nine digit number beginning with 5501 will be forwarded.
B. A ten digit number beginning with 5501 will be forwarded.
C. A nine digit number beginning with 5501612 will be forwarded.
D. A ten digit number beginning with 5501612 will be forwarded.

Answer: B

Explanation:
Destination Pattern
The destination pattern associates a dialed string with a specific telephony device. It is configured in a dial peer by using the destination-pattern command. If the dialed string matches the destination pattern, the call is routed according to the voice port in POTS dial peers, or the session target in voice-network dial peers. For outbound voice-network dial peers, the destination pattern may also determine the dialed digits that the router collects and then forwards to the remote telephony interface, such as a PBX, a telephone, or the PSTN. You must configure a destination pattern for each POTS and voice-network dial peer that you define on the router.
The destination pattern can be either a complete telephone number or a partial telephone number with wildcard digits, represented by a period (.) character. Each "." represents a wildcard for an individual digit that the originating router expects to match. For example, if the destination pattern for a dial peer is defined as "555....", then any dialed string beginning with 555, plus at least four additional digits, matches this dial peer.

QUESTION 147:

What transport layer protocol does RTP utilize?

A. TCP
B. UDP
C. IP
D. ICMP

Answer: B

Explanation:
RTP typically runs on top of UDP to utilize its multiplexing and checksum services. Other transport protocols besides UDP can carry RTP as well.
Real-Time Transport Protocol, an Internet protocol for transmitting real-time data such as audio and video. RTP itself does not guarantee real-time delivery of data, but it does provide mechanisms for the sending and receiving applications to support streaming data. Typically, RTP runs on top of the UDP protocol, although the specification is general enough to support other transport protocols.

QUESTION 148:

You are the network technician at Certkiller .com. VoIP is implemented on the Certkiller network. Your newly appointed Certkiller trainee wants to know what is used to carry VoIP voice packets on this network.
What will your reply be?

A. ICMP/IP
B. RTP/TCP

C. RTP/UDP
D. STP/UDP
E. RTP/RCMP

Answer: C

QUESTION 149:

Which lower layer protocol does the Real-Time Protocol (RTP) use?

A. TCP
B. UDP
C. WDP
D. HTTP
E. RTCP

Answer: B

QUESTION 150:

You are the network technician at Certkiller .com. Your newly appointed Certkiller trainee wants to know what TCP's reliable deliver service provides.
What will your reply be?

A. Connectionless service, flow control, sequenced delivery, and automatic error recovery
B. Flow control, sequenced delivery, automatic error recovery, and transmission window management
C. Unregulated send rate, automatic error recovery, and transmission window management
D. Connectionless service, unregulated send rate, automatic error recovery, and transmission window management

Answer: B

QUESTION 151:

You are the Voice technician at Certkiller , Inc. You want to deploy an IP telephony solution for the company. The Certkiller network is currently a traditional LAN/WAN based on Frame Relay.
Your CEO has read about the issues of converging both data and voice traffic onto a single network. She is concerned about the quality of their calls that need to cross the WAN in particularly.
What would you need to implement to ensure QoS for VoIP over Frame Relay?

A. Traffic shaping, priority queuing, Call Admission Control, and Class Based Weighted

Fair Queuing
B. Traffic shaping, priority queuing, Call Admission Control, and Weighted Random Early Detection
C. Fragmentation, traffic shaping, priority queuing, Low Latency Queuing, and link efficiency with cRTP.
D. Fragmentation, traffic shaping, priority queuing, Call Admission Control, and Weighted Random Early Detection

Answer: C

QUESTION 152:

On what is system capacity planning based?

A. On calculations and measurements of packet length distributions.
B. On calculations and measurements of busy hour call volume/estimates.
C. On calculations and measurements of the phone costs from phone bills.
D. On calculations and measurements of the total number of calls placed during a month.

Answer: B

QUESTION 153:

You have a customer that is interested in determining the number of VoIP calls their Frame Relay WAN links can support. Each of their Frame Relay WAN links has 54 kbps of bandwidth available outside all other applications and overhead. How many G.726 calls using the 32 kbps codec and 80 byte sample size can be supported?

A. 1
B. 2
C. 3
D. 4

Answer: A

QUESTION 154:

You are the network engineer at Certkiller .com. Your newly appointed Certkiller trainee wants to know which functions use UDP as their transport mechanism. What will your reply be? (Choose two)

A. RTP
B. RAS control function
C. call signaling function
D. H.245 control function

Answer: A, B

QUESTION 155:

What does gateway require to function as a translating gateway?

A. The capacity to translate the audio.
B. The ability to recognize the call control procedures of both connecting endpoints.
C. The ability to establish separate RTP sessions with the originating and terminating endpoints.
D. The ability to recognize the call control procedures for at least one of the connecting endpoints.

Answer: B

QUESTION 156:

You are the Voice engineer at Certkiller .com. Your newly appointed Certkiller trainee wants to knowwhat compressed RTP does.
What will your reply be?

A. It significantly reduce packet delay
B. It significantly reduce total bandwidth
C. It significantly reduce Frame Relay overhead
D. It significantly reduce the total number of packets

Answer: B

QUESTION 157:

You are the network engineer at Certkiller .com. You are implementing Frame Relay traffic shaping on the Certkiller network. Your newly appointed Certkiller trainee wants to know why Frame Relay traffic shaping is important.
What will your reply be?

A. It ensures that excess traffic above the CIR on the link is dropped.
B. It ensures that voice packets are not trapped behind large data packets.
C. It ensures that the priority of the voice packet is higher than the data packets.
D. It ensures that the RTP headed is reduced in size to reduce the overall size of the voice packet.
E. It ensures that excess traffic above the CIR on the link is not dropped, but is buffered and sent when there is capacity on the link.

Answer: E

QUESTION 158:

You are the network engineer at Certkiller .com. Certkiller has its headquarters in New York and a branch office in Delaware. The branch office is using a 128 kbps Frame Relay link to connect to headquarters. You want to ensure good voice quality on this link.
Which two QoS mechanisms should you implement on the Frame Relay interface? (Choose two.)

A. CIR
B. LLQ
C. WFQ
D. WRED
E. Fragmentation

Answer: B, E

QUESTION 159:

You are the Voice technician at Certkiller .com. The Certkiller network uses RTCP. Your newly appointed Certkiller trainee wants to knowwhat RTCP does.
What will your reply be?

A. It provides independent services irrespective of RTP.
B. It provides compression techniques to save bandwidth.
C. It provides in-band control information for an RTP flow.
D. It provides out-of-band control information for an RTP flow.

Answer: D

Explanation: RTCP provides out-of-band control information for an RTP flow.

QUESTION 160:

Which statement is true about the MGCP call agent?

A. Acts only as a recorder of call details.
B. Provides only call signaling and call setup.
C. Manages all aspects of the call and voice stream.
D. Monitors the quality of each call after setup.

Answer: B

Explanation:
In the MGCP model, the gateways focus on the audio signal translation function, while the Call Agent handles the signaling and call processing functions.

QUESTION 161:

The Cisco CallManager dial plan architecture is set up to handle two general types of calls. What are they? (Choose all that apply.)

A. External calls through a SAA Gateway
B. External calls through a PSTN gateway or to another Cisco CallManager cluster
C. Internal calls From the source router to the PBX-1
D. Internal calls to Cisco IP phones registered to the Cisco CallManager cluster itself-
E. Internal calls from the IP SoftPhone to the 7200 VXR2
F. External calls through the last downstream CallManager phone set.

Answer: B, D

QUESTION 162:

From the list below, what protocol is designed to provide end-to-end network transport functions for applications transmitting real-time data, such as audio, video, or simulation data, over Multicast or Unicast network services.

A. CAM
B. IPTV
C. STP
D. RTP
E. DMVRP
F. PIM
G. IS-IS

Answer: D

QUESTION 163:

Which statement represents the definition of an MGCP endpoint?

A. The interconnection between packet and traditional telephone networks.
B. Any analog telephony device (PBX, switch, ect).
C. IP hones
D. The gatekeepers in a VoIP network.

Answer: A

Explanation:
A typical MGCP gateway environment connects on one side with a public switched telephone network (PSTN), and on the other side with an IP network. Specialized call agent applications control the flow of media data across the distributed environment.

Call agents determine the route that data follows as it flows through the system. Multiple call agents can control call processing and data transfer. These call agents use a separate protocol to synchronize with each other and to send coherent commands to modules under their control.
MGCP assumes a connection model where the basic constructs are endpoints and connections. Endpoints are sources or sinks of data and could be physical or virtual.
Examples of physical endpoints are:
* An interface on a gateway that terminates a trunk connected to a PSTN switch (e.g., Class 5, Class 4, etc.). A gateway that terminates trunks is called a trunk gateway.
* An interface on a gateway that terminates analog POTS connection to a phone, key system, PBX, etc. A gateway that terminates residential POTS lines (to phones) is called a residential gateway.
An example of a virtual endpoint is an audio source in an audio- content server. Creation of physical endpoints requires hardware installation, while creation of virtual endpoints can be done by software.

QUESTION 164:

What are the three components in an MGCP environment? (Choose three)

A. Gateway
B. Gatekeeper
C. Endpoint
D. Call agent
E. Proxy server

Answer: A, C, D

Explanation:
A typical MGCP gateway environment connects on one side with a public switched telephone network (PSTN), and on the other side with an IP network. Specialized call agent applications control the flow of media data across the distributed environment. Call agents determine the route that data follows as it flows through the system. Multiple call agents can control call processing and data transfer. These call agents use a separate protocol to synchronize with each other and to send coherent commands to modules under their control.
Each call agent usually controls a set of gateway applications, including at least one media gateway. Media gateways convert media signals to an appropriate format depending on whether the signals are directed to a circuit switched network format or a packet switched network. Media gateways primarily perform audio signal translation functions in accordance with call agent commands.
Note: Gateways connected to an SS7 controlled network must also include at least one signaling gateway for controlling SS7 signaling.
The MGCP connection model consists of endpoints and connections. Endpoints represent physical or virtual sources through which data can flow
(for example, PSTN ports on a media gateway). Call agents combine sets of endpoints

under their control to create point-to-point or multipoint connections. Connections provide data paths for transferring and processing the data that flows through the gateway environment.
In the MGCP model, call control intelligence resides in the call agents, not in the media gateways. In effect, the MGCP standard defines a master/slave relationship between call agents and media gateways, where gateways execute commands sent by the call agents.
MGCP is a client-server protocol. The CA handles all aspects of setting up calls to and from endpoints. CAs or control servers provide the feature capabilities that a particular endpoint will be able to use. Endpoints connected to different CAs will likely have a different set of features they can use. Since all of the call control features are in the control server, each control server vendor decides which features are most important, and therefore different control server vendors differ in "essential features."
MGCP relies on a control server, or call agent (CA), to control call progression, tones to apply, and call characteristics. MGCP endpoints carry out instructions from the CA, which controls how calls proceed.

QUESTION 165:

With regard to MGCP, what is a call?

A. It is the path between two telephones.
B. It is the RTP sessions between the endpoints.
C. It is a connection between an endpoint and the call agent.
D. It is two or more endpoints sharing the same Call ID and the same media stream.

Answer: D

QUESTION 166:

You are the network engineer at Certkiller .com. You are deploying an IP telephony solution using MGCP. The call agent expects the gateway to use UDP port 2427 but an application on the Certkiller network is already using that port. You want to use port 4662 instead.
Which command would allow you to change the UDP port that the call agents and gateway communicate on?

A. Router(config)# mgcp UDP 4662
B. Router(config)# mgcp gateway 4662
C. Router(config)# mgcp call-agent 4662
D. Router(config-dial-peer)#application MGCPAPP 4662
E. Router(config)# mgcp default-package gm-package 4662

Answer: C

QUESTION 167:

You are the Voice engineer at Certkiller .com. Numerous Certkiller users complain that they are unable to complete calls through the MGCP network. You want to verify the extent of the problem by reviewing a count of the successful and unsuccessful control commands.
Which command should you use?

A. show mgcp
B. show mgcp count
C. show mgcp statistics
D. show call active voice
E. show call history voice

Answer: C

QUESTION 168:

You are the network engineer at Certkiller .com. You want to verify the registration of the gateway with the call agent.
Which show command should you use?

A. show mgcp
B. show call agent
C. show gateway mgcp
D. show endpoint mgcp
E. show call active voice

Answer: A

QUESTION 169:

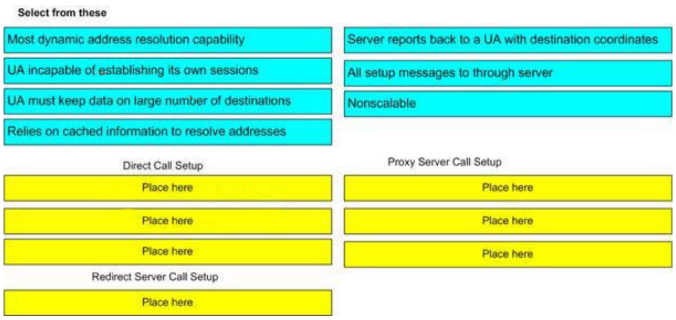
What identifies an MGCP endpoint?

A. A two part identifier that consists of the telephone number and local name of the user.
B. A two part identifier that consists of the telephone number and remote name of the user.
C. A two part identifier that consists of the domain name of the user and the IP address of the gateway.
D. A two part identifier that consists of the local name of the user and the domain name of the gateway.

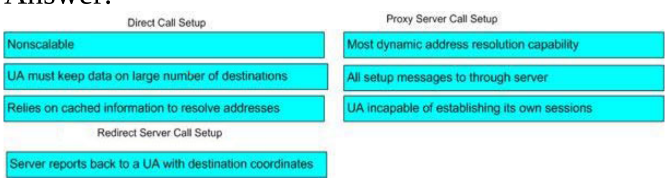
Answer: D

QUESTION 170:

Assume a SIP voice network. Drag each characteristic to the type of SIP call setup the characteristics best describes.

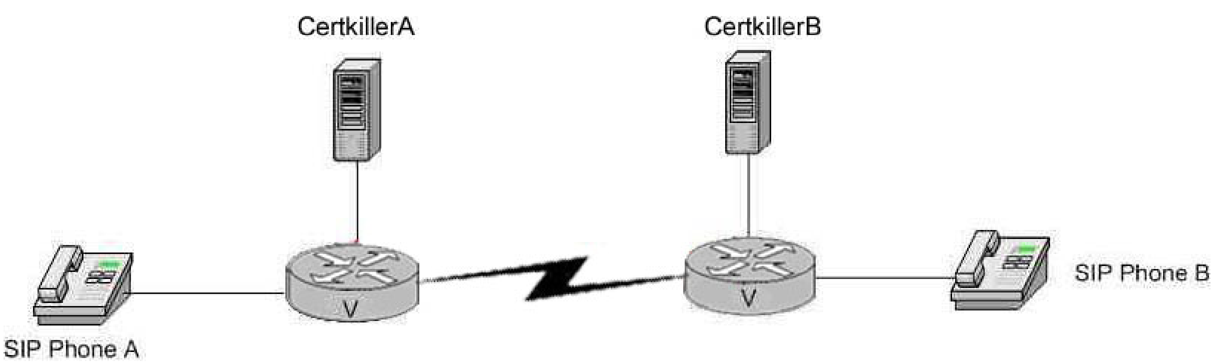


Answer:



Explanation:
"Server reports back to a UA with destination coordinates" is a function of the a Redirect Server (p. 6-94 of CVoice version 4.1 class books). Reference pages 6-91 - 6-94 of CVoice version 4.1 class books.

QUESTION 171:



For Scalability and ease of management, the decision has been made to centralize the location of all SIP endpoints in servers.
When phone A wants to call Phone B. it asks Certkiller A how to find Phone B.
What kind of device is Certkiller A?

A. Proxy
B. Redirect
C. Registrar
D. User agent client
E. User agent server

Answer: B

Explanation:
SIP ServersSIP servers include:
1. Proxy server-the proxy server is an intermediate device that receives SIP requests from

a client and then forwards the requests on the client's behalf. Basically, proxy servers receive SIP messages and forward them to the next SIP server in the network. Proxy servers can provide functions such as authentication, authorization, network access control, routing, reliable request retransmission, and security.
2. Redirectserver-Providesthe client with information about the next hop or hops that a message should take and then the client contacts the next hop server or UAS directly.
3. Registrar server-Processes requests from UACs for registration of their current location. Registrar servers are often co-located with a redirect or proxy server.
Redirect server: A redirect server is a server that accepts a SIP request, maps the address into zero or more new addresses and returns these addresses to the client

QUESTION 172:

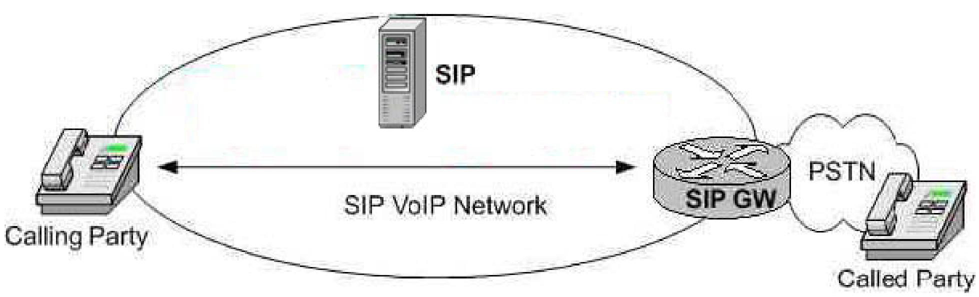
What is the function of a SIP location server?

A. Resolves active endpoint addresses
B. Routes service requests
C. Acquires active endpoint addresses
D. Resolves text addresses to IP addresses

Answer: A

Explanation:
The correct answer should be "Resolves active endpoint addresses" based on the following from CVoice version 4.1 class books on pages 6-84 and 6-89. A Location Server is defined (on page 6-84) as: An abstraction of a service providing address resolution services to SIP proxy or redirect servers. A location server embodies mechanisms to resolve addresses. On page 6-89 a Registrar Server is described as a server that acquires addresses for the location server.

QUESTION 173:



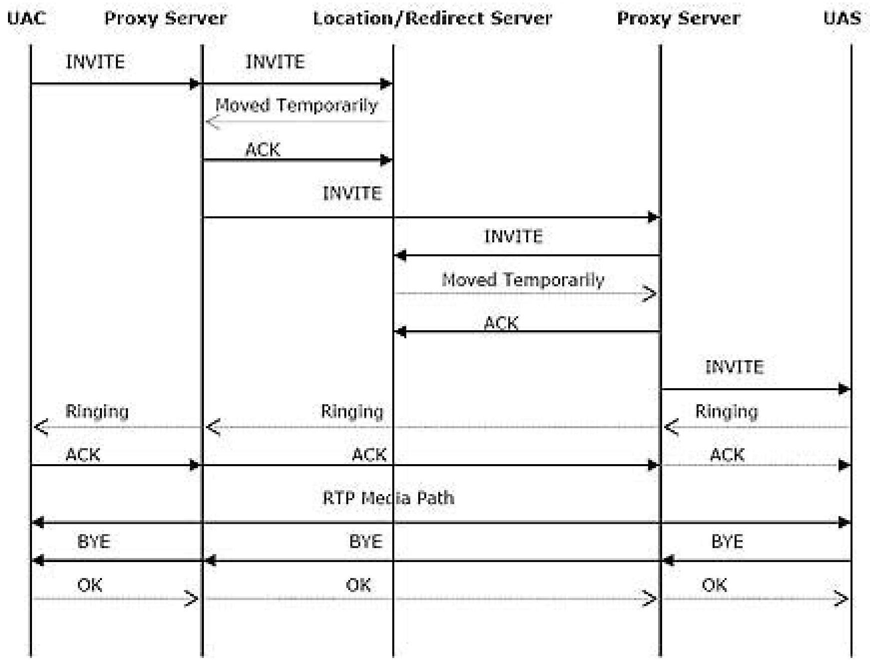
Given the SIP network shown in the diagram identify which three actions are initiated by the UAC (user agent client)? (Choose three)

A. Initiates a SIP requests.
B. Originated the BYE method to indicate call termination.
C. Originates the ACK method to indicate that it has receives a response to its invitation.
D. Contacts the user when a SIP invitation is receives.

E. Returns a response on behalf of the user to the invitation originator.

Answer: A, B, C

Explanation:
1.4.4 SIP Invitation
A successful SIP invitation consists of two requests, INVITE followed by ACK. The INVITE (Section 4.2.1) request asks the callee to join a particular conference or establish a two-party conversation. After the callee has agreed to participate in the call, the caller confirms that it has received that response by sending an ACK (Section 4.2.2) request. If the caller no longer wants to participate in the call, it sends a BYE request instead of an ACK.



QUESTION 174:

Which characteristic is true about SIP protocol messages?

A. Binary
B. Text-based
C. Numeric
D. Encrypted

Answer: B

Explanation:
Format
All SIP messages are either requests from a server or client or responses to a request. The messages are formatted according to RFC 822,

"Standard for the format of ARPA internet text messages." For all messages, the general format is:
1. A start line
2. One or more header fields
3. An empty line
4. A message body (optional)
Each line must end with a carriage return-line feed (CRLF).

QUESTION 175:

Upon which protocol model is the SIP protocol based?

A. HTML
B. H.323
C. Q.931
D. MGCP
E. HTPP/WWW

Answer: E

QUESTION 176:

With regard to SIP and SDP, which of the following statements is true?

A. SIP is similar to RAS and SDP is similar to RTP
B. SIP is similar to RTP and SDP is similar to RAS
C. SIP is similar to H.225 and SDP is similar to H.245
D. SIP is similar to H.245 and SDP is similar to H.323
E. SIP is similar to H.323 and SDP is similar to H.225

Answer: C

QUESTION 177:

You are the network engineer at Certkiller .com. You are configuring a connection to a SIP proxy server.
Which command would you use to specify the IP address of the server?

A. sip-ua
sip-server ipv4:1.2.3.4
B. sip-ua
sip-server target:1.2.3.4
C. dial-peer voice 1 voip
session target sip:1.2.3.4
D. dial-peer voice 1 voip
session target sip-server:1.2.3.4

Answer: A

QUESTION 178:

Which of the following call control models are based on decentralized call control? (Choose two.)

A. SIP
B. CAS
C. H.323
D. Q.931
E. MGCP

Answer: A, C

QUESTION 179:

You are meeting with a customer that has deployed IP telephony at their headquarters location. They would like to roll out IP telephony to their regional office as well. They are now using the G.711 codec at headquarters. They want to be able to maximize the number of calls carried without impacting voice quality or forcing a WAN upgrade.
Which codec would be appropriate for their WAN?

A. G.726
B. G.723.1
C. G.711
D. G.729B

Answer: D

QUESTION 180:

Examine the output.
ccm-manager mgcp
!
mgcp 5036
!
voice-port 1/0/0
!
voice-port 1/0/1
!
dial-peer voice 1 pots
application MGCPAPP
port 1/0/0

!
dial-peer voice 2 ports
application MGCPAPP
port 1/0/1
!
Your customer has sent you their MGCP gateway configuration. They are unable to get the gateway to communicate with the call agent.
What command needs to be inserted to resolve the problem?

A. ccm-manager mgcp 172.16.1.1
B. mgcpcall-agent 172.16.1.1
C. application MGCPAPP 172.16.1.1
D. mgcp5036 172.16.1.1

Answer: B

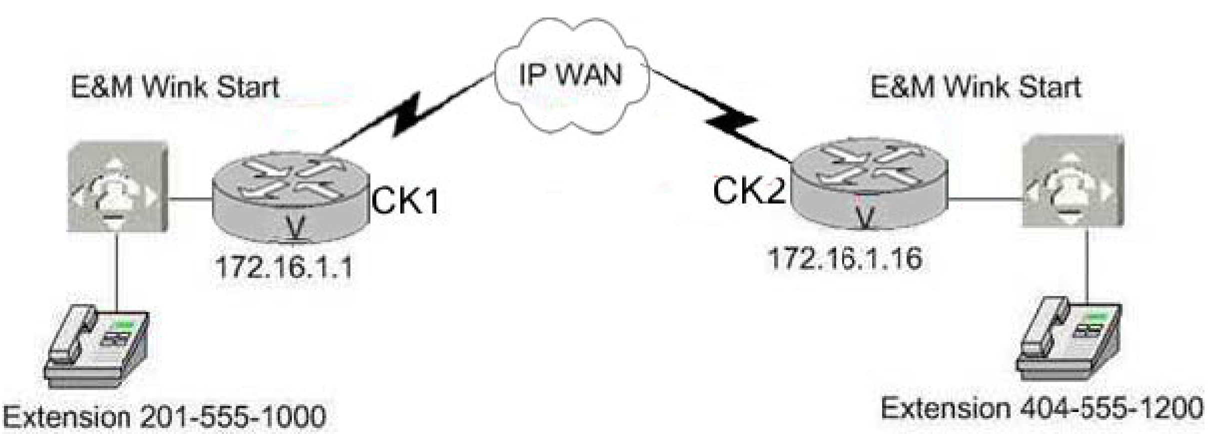
QUESTION 181:

You are the Voice technician at Certkiller .com. Your newly appointed Certkiller trainee wants to knowwhat request method initiates a SIP call setup.
What will your reply be?

A. ACK
B. INVITE
C. OPTIONS
D. REGISTER
E. DISCOVER

Answer: B

QUESTION 182:



hostname CK1
!
interface serial 0/0

```
ip address 172.16.1.1 255.255.255.248
!
controller t1
framing esp
clock source line
linecode b8zs
ds0-group 1timeslots 1-24 type e&m-wink-start
!
voice-port 1/0:1
!
dial-peer voice 1 voip
destination-pattern 404555....
session-targetipv4:172.16.1.6
!
dial-peer voice 2 ports
destination-pattern 201555....
port 1/0:1
hostname CK2
!
interface serial 0/0
ip address 172.16.1.6 255.255.255.248
!
controller t1
framing esp
clock source line
linecode b8zs
ds0-group 1timeslots 1-24 type e&m-wink-start
!
voice-port 1/0:1
!
dial-peer voice 1 voip
destination-pattern 201555....
session-targetipv4:172.16.1.1
!
dial-peer voice 2 ports
destination-pattern 404555....
port 1/0:1
```

Use the figure above to answer this question.
When extension 201-555-1000 dials 404-555-1200, how are digits manipulated in R1 so they are presented correctly at CK2 ?

A. When extension 201-555-1000 dials 404-555-1200, the digits 404-555 are stripped off prior to matching the outbound POTS dial peer.
B. When extension 202-555-1000 dials 404-555-1200, the digits 404-555 are stripped off by the connection trunk and CK2 receives only 1200.
C. When extension 201-555-1000 dials 404-555-1200, the outbound VoIP dial peer is

matched and all digits are sent.
D. When extension 201-555 1000 dials 404-555-1200, CK1 collects the 1200 and prepends the tie-line digits 404555.
That number is matched to a VoIP dial peer and sent to the appropriate address.

Answer: D

QUESTION 183:

How is CAS different on E1 and T1?

A. T1 has more signaling channels.
B. E1 CAS signaling is out-of-band while T1 is in-band.
C. E1 uses robbed-bit signaling.
D. T1 uses the D channel for CAS signaling.

Answer: B

QUESTION 184:

When impendence is mismatched in a two-wire to four-wire circuit, what is the common result?

A. glare
B. jitter
C. echo
D. clipping

Answer: C

QUESTION 185:

In the connection between a Cisco router and an E&M port on a PBX, which side is generally the Cisco side?

A. loop start
B. trunk circuit
C. switch port
D. signaling unit

Answer: D

Explanation:
Analog trunk circuits connect automated systems, such as a private branch exchange (PBX) and the network, such as a central office (CO). The most common form of analog trunking is the E&M interface. E&M Signaling is commonly refer to as "ear & mouth" or

"recEive and transMit", but its origin comes from the term earth and magnet. Earth represents electrical ground and magnet represents the electromagnet used to generate tone.
E&M signaling defines a trunk circuit side and a signaling unit side for each connection similar to the data circuit-terminating equipment (DCE) and data terminal equipment (DTE) reference type. Usually the PBX is the trunk circuit side and the telco, CO, channel-bank, or Cisco voice enabled platform is the signaling unit side.
Note:Cisco's analog E&M interface functions as the signaling unit side, so it expects the other side to be a trunk circuit.

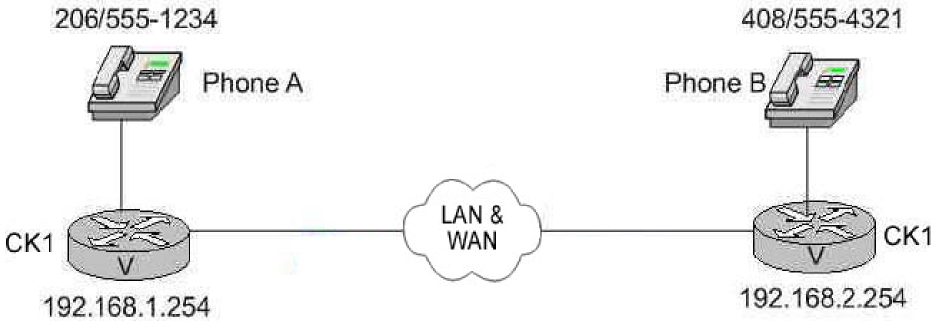
QUESTION 186:

You are the network technician at Certkiller .com. Your newly appointed Certkiller trainee wants to knowwhich signal types are used by E&M.
What will your reply be?

A. wink start, delay start, and loop start
B. wink start, loop start, and immediate start
C. wink start, delay start, and immediate start
D. delay start, and loop start, and immediate start

Answer: C

QUESTION 187:



In an effort to consume less bandwidth across the WAN, the decision was made at Certkiller to change the voice packet size. They changed from two voice frames per packet to one voice frame per packet.
What effect did this have on Certkiller 's voice traffic?

A. Per call bandwidth consumption decreased and end-to-end delay increased.
B. Per call bandwidth consumption increased and end-to-end delay decreased.
C. Per call bandwidth consumption decreased and end-to-end delay decreased.
D. Per call bandwidth consumption increased and end-to-end delay also increased.
E. There was no effect on voice traffic.

Answer: B

QUESTION 188:

You have been forwarded some questions by a prospective VoIP customer who would like to know the Cisco default sample size for the G.729 codec.
What is it?

A. 40 ms
B. 30 ms
C. 20 ms
D. 10 ms

Answer: D

Explanation:
Codec Sample Interval (ms) This is the sample interval at which the codec operates. For example, the G.729 coder operates on sample intervals of 10 ms, corresponding to 10 bytes (80 bits) per sample at a bit rate of 8 Kbps. (codec bit rate = codec sample size / codec sample interval).

QUESTION 189:

What component can be used to compensate for jitter?

A. FIFO queuing
B. Ethernet hubs
C. DSP algorithms
D. Playout delay buffer
E. Transmission medium

Answer: D

QUESTION 190:

You are the network engineer at Certkiller .com. Certkiller has its headquarters in New York and a branch office in Delaware. Users at headquarters must be able to call users at the branch office and users at the branch office must be able to call headquarters.
How many dial peers must you configure to meet these requirements?

A. 1
B. 2
C. 3
D. 4
E. none

Answer: D

QUESTION 191:

You are the network engineer at Certkiller .com. Certkiller has an IP network. Your newly appointed Certkiller trainee wants to know which issues would adversely affect voice quality on the Certkiller network.
What will your reply be?

A. Jitter, delay, and packet loss
B. Jitter, prioritization, and acknowledgment
C. Prioritization, delay, and delivery guarantee
D. Packet loss, acknowledgment, and delivery guarantee

Answer: A

QUESTION 192:

In accordance with the G.114 standard, which of the following delay ranges is acceptable?

A. 0 - 150 ms
B. 0 - 250 ms
C. 0 - 300 ms
D. 0 - 400 ms
E. 0 - 500 ms

Answer: A

QUESTION 193:

Which application allows you to communicate to multiple remote offices simultaneously?

A. IP Phone
B. IP Centrex
C. Toll Bypass
D. Multi-tenant
E. Hoot and Holler

Answer: E

QUESTION 194:

You are the network engineer at Certkiller .com. Your newly appointed Certkiller trainee wants to know under which standard the fragmentation for VoIP over Frame Relay is defined.

What will your reply be?

A. FRF.5
B. FRF.6
C. FRF.9
D. FRF.11
E. FRF.12

Answer: E

QUESTION 195:

What type of multiplexing is packet switching an example of?

A. Statistical
B. Time division
C. Phase division
D. Frequency division

Answer: A

QUESTION 196:

Which of the following is the correct formula for encoding PCM?

A. 2 states per bit x 8000 Hz frequency coded into 4 bits = 64 kbps
B. 8000 Hz frequency encoded in 4 bits, each expanded by two = 64 kbps
C. 3400 Hz voice frequency represented in 8 bits x 2 states per bit = 56 bkps
D. 4000 Hz frequency sampled at two times the frequency and each sample is represented in 8 bits = 64 kbps

Answer: D

QUESTION 197:

Your newly appointed Certkiller trainee wants to know what CAC applies to.
What will your reply be?

A. Latency
B. Data traffic
C. Voice traffic
D. TCP networks
E. Voice and data traffic

Answer: C

QUESTION 198:

In a campus network, which of the following are categories for QoS?

A. Separation of queues, queue scheduling, and pruning of queues
B. Pruning of queues, and marking control and management traffic
C. Queue scheduling, pruning of queues and marking control and management traffic
D. Separation of queues, queue scheduling, and marking control and management traffic

Answer: D

QUESTION 199:

What is the recommended configuration for the transmit interface in switchwide queuing?

A. CoS
B. PFC
C. FIFO
D. TIFF
E. 2Q1T

Answer: E

QUESTION 200:

Which tool can be applied to the Campus Switches to help eliminate traffic congestion?

A. RDP
B. CDP
C. LMI
D. QoS
E. PIM
F. DVRMP

Answer: D

QUESTION 201:

You are the Lead Network Engineer for your company. Your manager asks what needs to be upgraded on the Network to begin Preparing for the VOIP upgrade.
Your Network Campus Consists of One Core Rack of Cat 4000 Switches and 4 closets with BayStack and Synoptics hubs. IP phones will be placed on every desk of the organization and the entire Campus is wired with Cat 5e.
From the list below, what needs to be done ? (Select all that apply.)

A. Upgrade all the Wire to Cat 6
B. Install 7200 VXR's in Every Close to Route all IP traffic to the Core Switch
C. Upgrade the Cat 4000's to 6000 series
D. Upgrade all the Hubbed gear with Cat 4000 series Switches

Answer: D

QUESTION 202:

Which of the following QoS measures affect the outbound queue when implemented? (Choose three.)

A. LLQ
B. RSQ
C. WFQ
D. FIFO
E. FRF.12
F. CBWFQ

Answer: A, C, F

QUESTION 203:

Standard PCM encodes voice at which sampling rate?

A. 16 kbps
B. 32 kbps
C. 64 kbps
D. 128 kbs

Answer: C

Explanation:
Standards-Based PCM Encoding Standards-based ITU-T G.711 PCM encoding provides 64 kbps analog to digital conversion using u-law or A-law

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QUESTION 204:

Within a distributed call processing environment, what can you use to achieve call admission control across the WAN? (Choose all that apply.)

A. You can use a 720VXR to diversify the IPN1 traffic.
B. You can use a gatekeeper.
C. You can use a H.333 Line card.
D. You can use a CiscoWorks RME package to keep lines clear.

Answer: B

QUESTION 205:

From the following list, please select the Cisco Hardware that is able to terminate Voice Traffic:

A. Cisco 500
B. MC3810
C. Cisco 1700
D. 2600 Series Router
E. Cisco cat 2950XL
F. 7200 VXR

Answer: B, C, D; F

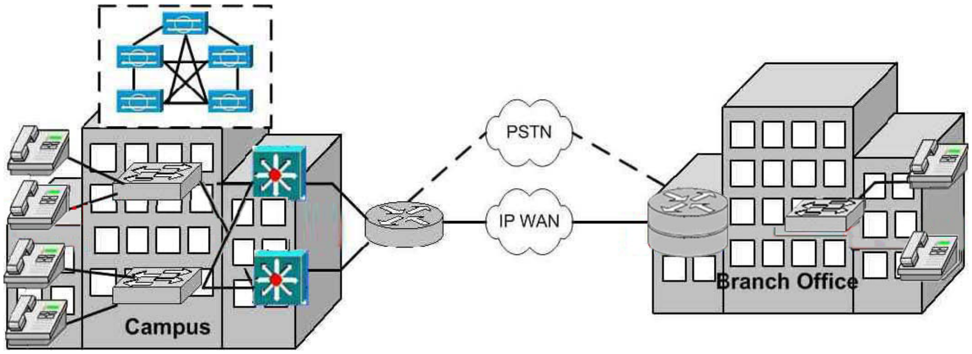
QUESTION 206:

You are using class-based weighted fair queuing (CBWFQ) to manage queues in your network. Voice quality has been inconsistent over lower speed WAN links and users have been complaining. You have decided to configure low latency queuing (LLQ) on the lower speed links to help improve voice quality in the network. Which three of the following steps should you take to implement LLQ in the network? (Choose three)

A. Ensure voice traffic is marked with a value EF in the DSCP.
B. Assign voice traffic control protocol traffic to its own queue with a DSCP value of AF31.
C. Assign all voice traffic to the priority queue in LLQ.
D. Assign all non time-sensitive, nonvoice traffic to a default queue with a DSCP value of 0.
E. Ensure voice traffic is given a minimum of 20% of available bandwidth through policing.
F. Ensure that all nonvoice traffic does not exceed more than 75% of available bandwidth.

Answer: A, B, C

QUESTION 207:



Given the network shown in the exhibit, select the recommended QoS configuration for the slow speed WAN segment connecting the campus and branch office.

A. LLQ
Voice traffic marked as EF
Voice control traffic marked as AF31
Implement Admission Control
B. CBWFQ
Voice traffic marked as EF
Implement policing on input
Non-time-sensitive traffic marked as Best Effort
C. LLQ
Voice traffic marked as AF
Implement policing on input
Non-time-sensitive traffic marked as Best Effort
D. CDWFQ
Voice traffic marked as AF
Voice control traffic marked as AF31
Implement Admission Control

Answer: A

QUESTION 208:

You are the network engineer at Certkiller .com. Certkiller is using LLQ on the serial interface 3/3 on the Certkiller router. You want to verify the status if LLQ on the interface.
Which command would you use?

A. show class 11q
B. show interface serial 3/3
C. show queue interface serial 3/3
D. show interface serial 3/3 class 11q
E. show policy-map interface serial 3/3

Answer: E

QUESTION 209:

On what is traffic engineering for voice based?

A. Peak of service.
B. Class of service.
C. Grade of service.
D. Speed of service.
E. Quality of service.

Answer: C

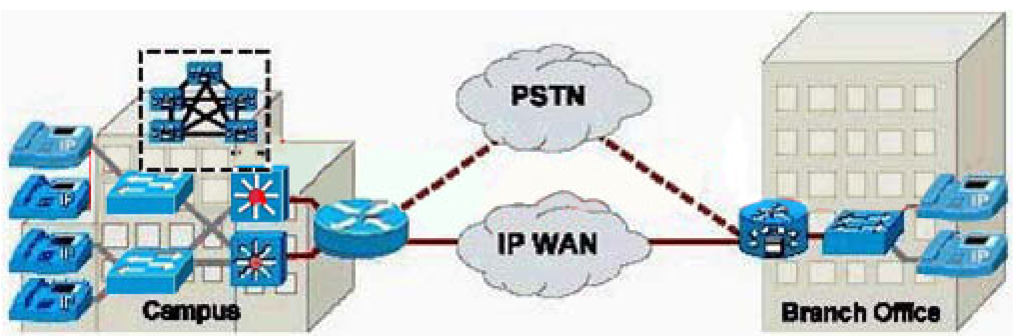
QUESTION 210:

Which process changes an internal extension into a fully qualified external PSTN number before matching to a dial peer?

A. digit masking
B. forward digits
C. number expansion
D. prefix extension

Answer: C

QUESTION 211:



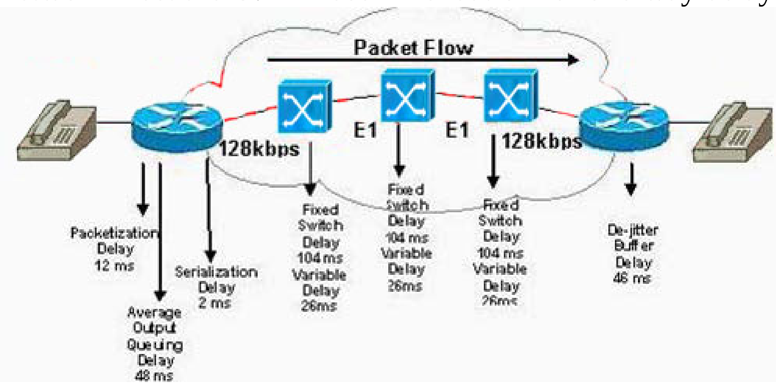
Given the network shown in the exhibit, select the recommended QoS configuration for the slow speed WAN segment connecting the campus and branch office.

A. LLQ Voice traffic marked as EF Voice control traffic marked as AF31 Implement Admission Control
B. CBWFQ Voice traffic marked as EF Implement policing on input Non-time-sensitive traffic marked as Best Effort
C. LLQ Voice traffic marked as AF Implement policing on input Non-time-sensitive traffic marked as Best Effort
D. CBWFQ Voice traffic marked as AF Voice control traffic marked as AF31 Implement Admission Control

Answer: A

QUESTION 212:

Considering only the delays actually shown in the exhibit, how will the proposed network meet the G.114 recommendation for one-way delay?



A. unacceptable for general network purposes
B. acceptable for most user applications
C. acceptable, provided administrators understand implications
D. unacceptable for voice, but acceptable for time-sensitive transaction processing

Answer: A

QUESTION 213:

Site A uses three-digit internal numbers and Site B uses four-digit internal numbers. All calls to the PSTN are routed through Site B. What dial plan below best represents provision simplicity?

A. Translate all called numbers within Site A to four digits.
B. Translate all called numbers within Site B to three digits.
C. Translate all called numbers leaving Site A to ten digits.
D. Translate all called numbers at either site to ten digits.

Answer: A

QUESTION 214:

Examine the following PBX system parameters: The calling side seizes the line by going off-hook on its E-lead and sends information as DTMF digits. The voice path is 4-wires, and the voice enabled router is in another building from the PBX. Select the correct set of commands to allow communication between a voice enabled router and a PBX.

A. voice port 1/0/0
signal immediate-start

operation 4-wire
type 2
B. voice port 1/0/0
signal delay-dial
operation 4-wire
type 1
C. voice port 1/0/0
signal wink-start
operation 4-wire
type 3
D. voice port 1/0/0
signal immediate-start
operation 4-wire
type 4

Answer:

QUESTION 215:

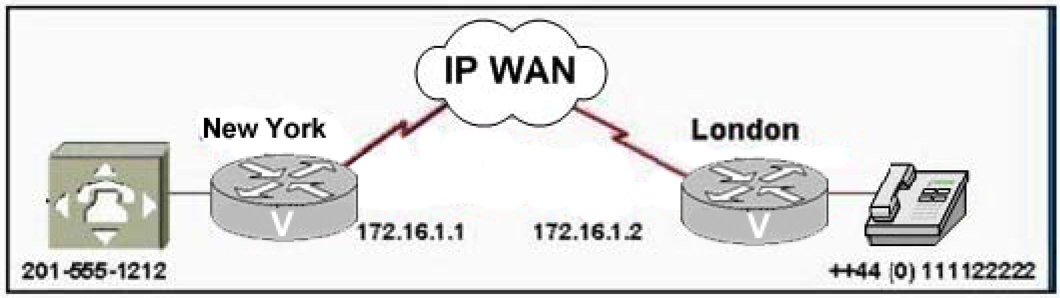
Which device is used to allow an H.323 stream to transit a firewall?

A. gatekeeper
B. gateway
C. proxy
D. MCU

Answer:

QUESTION 216:

Refer to the graphic for IP addresses and telephone numbers. You are working with a customer that is opening a small sales office in London. You would like to be able to have the user in London be able to dial into the PBX in New York over the IP WAN. The New York PBX uses loop start, a two-wire operation, and DTMF dialing. Please choose the correct FXO port configuration for New York.
Exhibit:



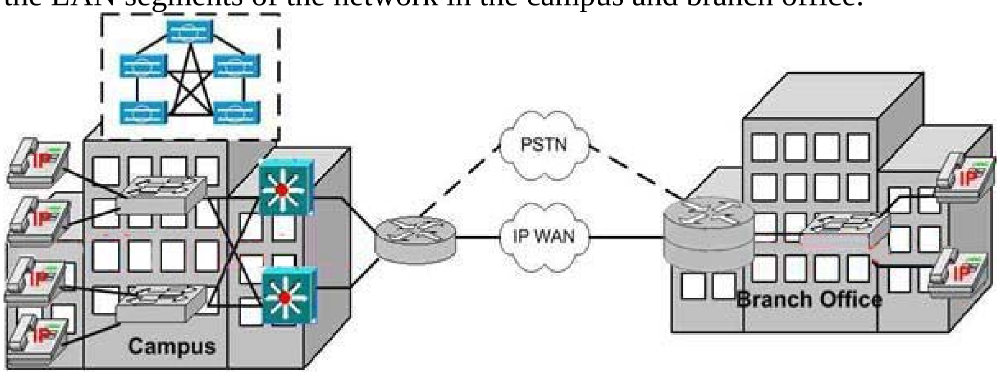
A. voice port 1/0/0

signal loop-start
operation 2-wire
dial-type dtmf
B. voice port 1/1/1
destination 2015551212
signal loop-start
operation 2-wire
type 1
dial-type dtmf
C. voice-port 1/0/0
session target ipv4:172.16.1.1
destination 2015551212
signal loop-start
operation 2-wire
dial-type dtmf

Answer:

QUESTION 217:

Given the network shown in the exhibit, select the recommended QoS configuration for the LAN segments of the network in the campus and branch office.



A. Configure WRR with voice as highest priority. Use ACLs to classify voice traffic. Isolate voice traffic in its own VLAN. Configure access switches to trust traffic from IP phones.
B. Configure a PQ with WRR. Use ACLs to classify voice control traffic. Isolate voice traffic in its own VLAN. Configure access switches to trust traffic from IP phones
C. Configure a PQ with WRR. Use ACLs to classify voice traffic. Isolate voice traffic in its own VLAN. Configure access switches to trust traffic from IP phones.
D. Configure WRR with voice as highest priority. Use ACLs to classify voice traffic. Isolate voice traffic in its own VLAN. Configure access switches to trust traffic from IP phones.

Answer:

QUESTION 218:

Which problem does a-law and mu-law address?

A. quantization produces distortion at higher frequencies
B. encoding loses signal integrity at lower frequencies
C. quantization produces higher signal-to-noise ratios at smaller amplitudes
D. encoding produces greater signal loss at higher amplitude

Answer:

QUESTION 219:

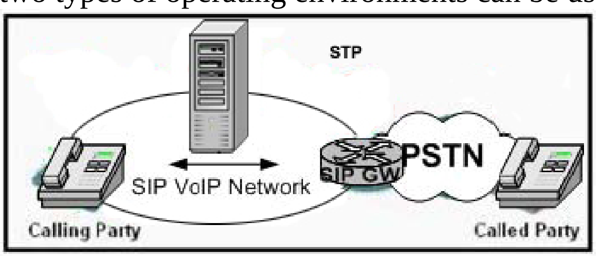
Which statement is true about out-of-band signaling?

A. A signaling bit is robbed from each frame.
B. Signaling bits are sent in a special order in a dedicated signaling frame.
C. All signaling is directly associated with its corresponding voice frame.
D. All voice packets carry their own signaling.

Answer:

QUESTION 220:

If the Cisco SIP server shown in the exhibit is used only for SIP redirect call setup, which two types of operating environments can be used for the server? (Choose two.)



A. Linux (Redhat)
B. Windows
C. Solaris
D. Exchange
E. Microsoft Live Communication Server

Answer:

QUESTION 221:

A network has the following characteristics:
* the use of the G.711 codec with codec speed of 64kbps
* a 160-byte sample size

* the use of Frame Relay without compressed Real-Time Transport Protocol (CRTP)
* FRF.12 with 6 bytes of overhead
What minimum WAN bandwidth would be required to support three simultaneous VoIP calls?

A. 247200 bps
B. 19200 bps
C. 79200 bps
D. 51600 bps

Answer:

QUESTION 222:

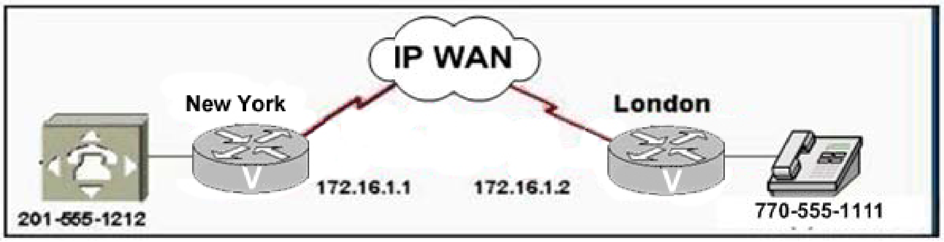
What is the purpose of RTCP?

A. provides monitoring and control for RTP flows
B. provides RTP header compression
C. passes RTP route status information
D. provides RTP with packet retransmission

Answer:

QUESTION 223:

Refer to the graphic for IP addresses and telephone numbers. You are working with a customer that is opening a small sales office in Atlanta. You would like to have the user in Atlanta be able to dial into the PBX in New York over the IP WAN. The New York PBX uses ground start, a two-wire operation, and DTMF dialing. Please choose the correct FXO port configuration commands for New York.



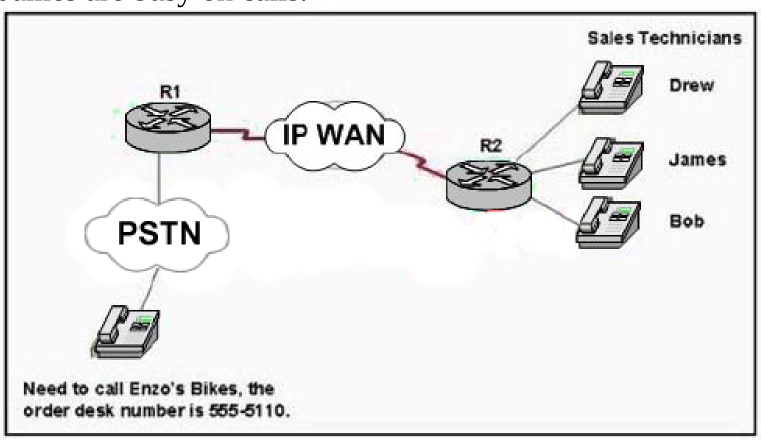
A. voice-port 1/0/0
signal ground-start
operation 2-wire
dial-type dtmf
B. voice-port 1/1/1
destination 2015551212
signal ground-start
operation 2-wire
type 1

dial-type dtmf
C. voice-port 1/0/0
session target ipv4:172.16.1.1
destination 2015551212
signal ground-start
operation 2-wire
dial-type dtmf

Answer:

QUESTION 224:

Refer to the graphic. You have a customer that manufactures high end bicycle frames. Until recently they sold only to bicycle shops. Now they are starting to sell to end users. They need a way to add two additional sale staff and ensure that the senior sales technician always gets the first call. Drew is the senior sales technician. Bob is the newest sales technician. Select the correct dial-peer command set so that Bob will always be the last chosen for incoming sales calls, after Drew and James, or when Drew and James are busy on calls.



A. dial-peer voice 3 pots
destination-pattern 5555110
preference 2
B. dial-peer voice 3 pots
destination-pattern 5555110
preference 0
C. dial-peer voice 3 pots
destination-pattern 5555110
preference firstlast
D. dial-peer voice 3 pots
destination-pattern 5555110
preference huntstcp
E. dial-peer voice 3 pots
destination-pattern 5555110
preference high

Answer:

QUESTION 225:

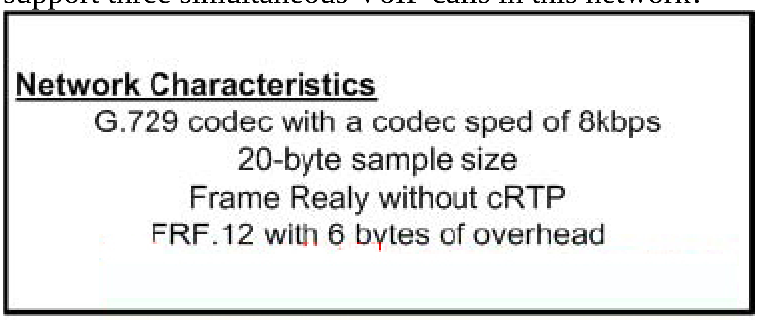
What is the approximate frequency range of human speech?

A. 20 Hz to 20,000 Hz
B. 40 Hz to 15,000 Hz
C. 200 Hz to 9000 Hz
D. 600 Hz to 5400 Hz

Answer:

QUESTION 226:

Refer to the exhibit. What is the minimum WAN bandwidth that would be required to support three simultaneous VoIP calls in this network?



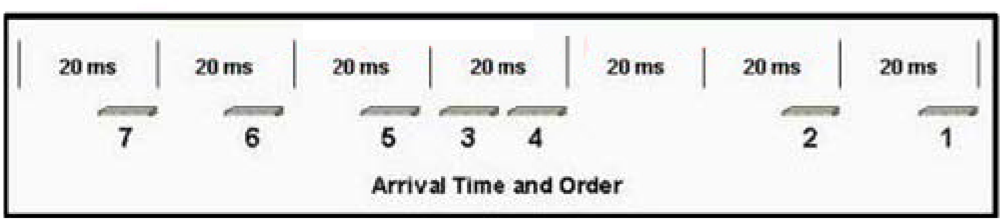
Network Characteristics
G.729 codec with a codec sped of 8kbps
20-byte sample size
Frame Realy without cRTP
FRF.12 with 6 bytes of overhead

A. 19,200 bps
B. 51,600 bps
C. 79,200 bps
D. 247,200 bps

Answer:

QUESTION 227:

Exhibit:



The illustration above shows the timing and order of arrival of voice packets at the end device. Each voice packet contains 20 ms of voice data. Assuming a play-out buffer of 20 ms, what will happen to voice packet number 3?

A. It will be played out in its proper location.
B. It will be played out in the order it is received.
C. It will be dropped and treated as a lost packet.
D. Immediately moved to the front of the play-out buffer.

Answer:

QUESTION 228:

Which two codes together make up the number that follows the E.164 recommendation numbering scheme? (Choose two.)

A. country code
B. subscriber code
C. national destination code
D. provider code

Answer: